Project Documentation

VERSION 1.0

PROJECT MINUTEMAN

A network of audio sensors capable of detecting gunshots in a building.

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1. Executive Summary

For years, our country has bared witnessed to countless massacres committed by mass shooters. The public response to these events have varied, but have almost always been the same. Gun control activists have called for more legislation, forcing pro-gun groups to respond by attempting to defend against these legislation attempts. This project does not attempt to answer which of these stances is correct. However, it is clear that after all this time, it is very unlikely we are going to find effective legislation that we will all agree with. A technological solution to this problem is necessary, not just because all groups can agree on it, but because it can reliably respond to these events and save lives regardless of legislation issues.

Our society has advanced to a stage that technological solutions to difficult problems are becoming a reality almost every day. Already several solutions to the issue of mass shootings have been revealed such as door blocking systems [1], inflatable slides allowing occupants on the upper floor of a building to escape [2], and more. These solutions are not divisive, inexpensive, and are signs that the market is capable and willing to create more technological solutions to this issue in the future.

This project will aim to fulfill two primary goals. The first goal is to provide a solution that can provide rapid but accurate detection of a firearm discharge within an operational area. The second is to rapidly send alerts, informing authorities and/or facility security. This information will include the location of where the shots detected occured, as well as the type of weapon fired. It is the hope of this project that it will become a starting point that will enable other technological solutions to the threat of mass shooters.

This project was inspired by a previous Senior Design project called GLASS [3] which was an outdoor audio detection system that could accurately pick up the sound of a gunshot, discern the type of firearm used, triangulate the location of the gunshot, and then report that information to authorities.

2. Project Description

The features of our project are to create an audio sensor system that can be installed outdoors. This system is to accurately detect gunfire discharges, being able to discern them from other sounds in the environment such as engine backfires or glass breaking. It is to be able to discern the general location of the gunfire within the operational area of the sensor. Then it is to inform the authorities.

This system in its basic form will be affordable for most facilities such as schools, public buildings, or places of large public gatherings. Although this is not the primary intention of the project, the devices will also be available for home security or other private facilities looking for security as well. This system will have a lot of potential for expansion, either through additional technologies created by this design team or by third-parties wanting to make a difference.

2.1 Goals and Objectives

The project will fulfill the following goals and/or objectives as shown in Table 2-1:

A single multi-directional audio sensor to detect events that could include a gunshot. Connected to a building or vehicle's power supply.
A microcontroller to connect the sensor and analyze events. Connected to the building or vehicle's power supply or to the host computer.
A program/app that can be downloaded to a host computer to manage the sensor or network of sensors and devices, and confirm potential events.
Sensor should be mountable to tripods, pole mounts, or vehicles.

Table 2-1: Goals and Objectives

2.2 Requirements

The project will fulfill the following requirements as shown in Table 2-2:

Identify gunshots with greater than:	90%	Accuracy
Identify gunshots at least within:	250	Meters

Identify and confirm a gunshot within:	2.5	Seconds
Identify the direction of gunshots to within:	5	Degrees
Identify the location of gunshots to within:	5	Meters
Maintain a cost of less than:	500	USD

Table 2-2: Requirements

2.3 Specifications:

The project will fulfill the following specifications as shown in Table 2-3:

The sensors are connected to the central hub, which is connected to the host computer program.
The sensors send events of an intensity over 100 dB within 150m of the sensor to the hub for analysis. [5]
The hub analyzes the event and confirms if it is a firearm discharge by filtering out other noises (that do not range in the intensity or the pulse signature of firearms) and searching for a specific pulse signature in the event.
Multiple inputs of the same event or similar event from other sensors in the same time-frame will be used to confirm the original event.
The hub is supported by machine learning and a database of audio samples consisting of discharge samples of multiple handguns, rifles, and shotguns, as well as different calibers or gauges (including 9×19mm Parabellum, 5.56mm, and 12ga)
The hub triangulates the general location of the gunshot from sensor data of the event between multiple sensors to within 5-10m.
The hub sends out an alert to host computer within 0.5 seconds of the event.
The program contacts authorities, building security, and other individuals inside the building within 1 second of the event.
The program provides data to other devices connected to the network through the program within 1.5 seconds of the event.
The program provides an administrator/staff member with the ability to cancel an alert/call a false alarm.

Table 2-3: Specifications

2.4 Constraints

The project will be held within the following constraints as shown in Table 2-4:

Requires at least one sensor module
All components must share the same clock rate when communicating
There is a required minimum sample rate required for analysis
The sensor module should not weigh more than 5 pounds.
The entire project cannot cost more than 1000 USD

Table 2-4: The Constraints

2.4.1 Ethical Constraints

Given the nature of the situations a gunshot detection system will be required in, there are some ethical constraints to consider. While a system failure is not always within the designers ability to manage, at worst a failure in this case can potentially be catastrophic for several lives, especially if they are expecting the system to be reliable in this regard. At the least, it can create false alarms, wasting potentially thousands of dollars of resources in responding to a situation that is not in fact dangerous. Therefore, making the system as reliable as possible in correctly identifying shootings and doing so as rapidly as possible is a design constraint.

There are also other ethical constraints to consider with regards to privacy. As this is a device that nearly always listens, efforts should be made to prevent individuals from using the device to spy on others. This is especially true for

3. Project Management

Within our group, we planned on which programs we were gonna use to produce organizational duties within each person and make sure we do our priorities right and have a to-do list for each one. The team is split up into hardware and software so we utilize apps like Trello or Google Drive to store our documents and keep updated with each other pertaining to research tools and references regarding our project. We use Google Drive mainly to store any articles we may have found all throughout the depths of the internet and then we Trello to organize our priorities and have our milestones or deadlines for any major priority we have to attend to currently or even later on in Senior Design 2.

We planned on having weekly meetings on a certain day like Friday so we make sure everyone in the team is well prepared for each milestone laid out for us and no one misses a deadline or else there will be a consequences. We also decided on having core values between the group because without these values, everyone will be in disarray or someone has the potential to be lost or he doesn't know what to do a certain stage of the implementation of the design. Trello is a useful tool as it is one of the most popular organizational modes that is also accessible easily and portable because users can use in their mobile phones. Google Drive is a sureshot go-to for document storing and document/spreadsheet creation since it is the most popular out of the bunch that is used by millions of people around the world.

We also use Discord as a means of communication between each other since it has its handy feature of voice chat and screen sharing. We meet up on discord every week to discuss the necessary details that we to deal with for our next milestone. There is constant agreement and we make sure that nothing gets passed through without the approval of everyone in the group. It's expected that everyone has their own fair and share of other obstacles and we expect each other to remind the whole team if something unexpected pops up we have to understand all the circumstances regarding the situation if ever one arises.

There will be proper meeting times that is to be finalized as the final deadline looms in to work on the design and then the real implementation of the Minuteman system. The projected days of gathering wherein we will plan accordingly what the next steps are for a successful stage of development for both the hardware and software sides of the spectrum whether it may be a certain change in software source code to be used or if there's a decision to change a programming board or even a microcontroller for future use. If a meeting is not met, we will assess each possible solution to make our flow better as a team and make sure we have ample teamwork so that no one gets left behind in our progress. Continuous communication and trust is key in a smooth and successful project and so we will implement these key factors as some things to rely on when things get really tough.

There are numerous key points that are well stated when it comes to setting up a precise and well resourced project especially in the caliber this of a class in senior design. Everyone in the group have come into a union that this project will be not be a "piece of cake" because of the astounding amount of research information needed and the requires expertise from each of the members whether it be pure knowledge about how a part of the device functions or to bridge out to the software world and meddle in with critical analysis of a certain software function or runtime efficiency execution hoping it won't crash or have a runtime error while making the two ends meet properly. If there is no cohesion between the two main giants, there is going to be major consequences in the final design because there is potential that the requirements might switch unexpectedly where the group scrambles into panic mode that will cause even more unknown obstacles that will need attention as soon as possible because it all declared decisions aren't met the project will crumble into pieces.

One of the major risks in project planning is that of the lack of it. If there is no planning at all to begin the project there will be people wondering where they got things wrong and be focused on blaming things where they shouldn't be blamed. This cause can effect into frustration along with poor project performance.There always should be a full pledged way to make decisions and not go spontaneously especially where the expertise of the team isn't that high since we are only students. It will take in more time to fully access our capabilities and know our strengths and weaknesses as the projects flow by so indeed there will be minor setbacks along the way but, with the said operation for planning, it will not be that hard to fix the holes as long as there is quality teamwork between everyone in the group.

There is a preeminent structure that properly breaks down project development and planning. This is called the Work Breakdown Structure. According to teamgantt, a Work Breakdown Structure is a "hierarchical (from general to specific) tree structure of deliverables and tasks that need to be performed to complete a project". This structure is the foundation for good planning in projects, the fears of over costing, and management. This is one of the most important aspect in setting up a project. This is the foundation where everything else roots from and gets built. The lowest level of the work breakdown structure is the Work Package. WP is defined as the allowance of a certain package that can be costed, resourced, and also schedules. The WP contains a list of tasks that can be performed that form a basis for the schedule created by the team themselves. This package also allows the fair assignment of the roles for a team. WBS allows build up of costs and schedule as expected from a team of our caliber. With the WBS any cost or schedule modification can be reported at any level of the hierarchy.

This facilities strong management in project execution. This kind of process can only be used for many other various things such as Document management, Risk management, etc.

Another keypoint in good project planning is involving the right people. Especially in Senior Design 2, instead of research, it will mainly focus on actually creating the project and the team would need the right resources in order to create it in the first place and this involves having the suitable acquaintances. Without suitable acquaintances, reliability in self working the project will be really low. As long as there are good sources, reference, or even dependencies, the flow of the project should be smooth because of the accountability that these provide to a concise and clear stream for the project finalization.

Lastly, we will always allow enough time for each stage to be done. We won't be stressing out if a certain deadline is not met because we will be working in advance and will ample time to tasks offered by what is required. We will be working systematically wherein we will methodologically implement our own work ethics in to play.

3.1 House of Quality Table:

Table 3-1 properly illustrates our discussed customer requirements along with the desired technical equivalents into a relationship matrix showing how strong of a relationship two different requirements relate to. There is also a row called Direction Of Improvement where it shows how much relative improvement we need for each field and a certain weight on how important each field is for our project.

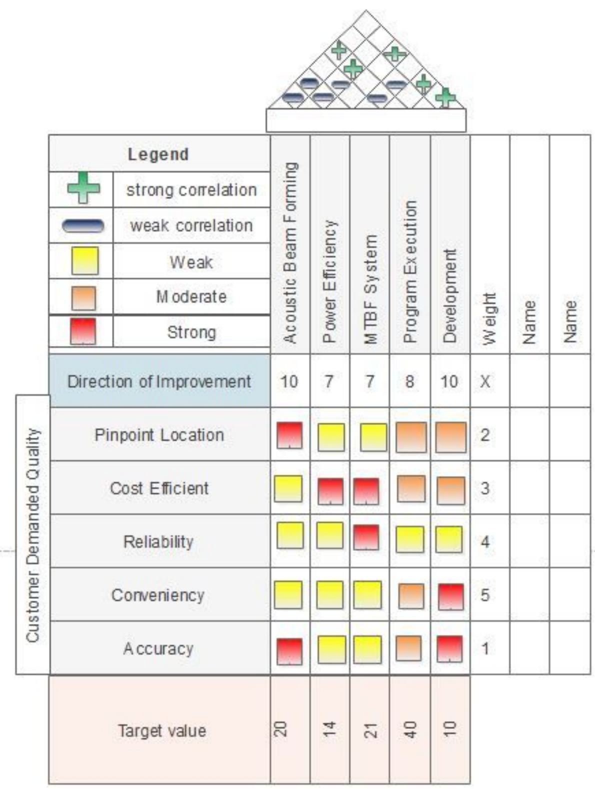

Table 3-1: House of Quality

In our HOQ table, this is intended to compare and contrast then analyze with determination in both the engineering requirements and marketing requirements for oue Minuteman device. The HOQ table is a crucial part of our product development process which is also widely used in the world of engineering. A House of Quality table is a conceptual map that allows for inter-functional planning and communications where it produces a dynamic system of goals to determine which goals are more important than the others and to achieve these goals where in other goals would be affected.

3.1.1 Understanding the House of Quality Table:

There are exactly 5 engineering requirements and 5 customer centered requirements laid out in a matrix like structure. These will then be compared to each other to determine which pairs of relationship extract the most attention and which do not. As can be seen from the House of Quality table above, there are different color coded regions in the mapping of the table which corresponds to how strong a relationship is between two requirements. The red color stands for a strong relationship which means a pair of requirements correlate with a stronger bond. The orange color stands for a middle ground between requirements meaning that this has a decent relationship while not as robust as a pair with a perfect correlation. Lastly, the yellow color stands for a weak relationship meaning that they do not correlate whatsoever by any means necessary.

With these correlations, these provide the necessary information for customers alike in how our priorities are aligned. For example, Pinpoint location and Acoustic Beamforming have such a tight relationship that there is no more reason to worry about how to stoutly perform that certain task. Compared to the weaker relationships, they tend to be the ones that need more critical attention because they usually are going to be harder to deal with and, in effect, would need to do more focused research into.

The Direction of Improvement row in our HOQ table signifies which pairs would be put in the priority queue first and which come in last. From the table there is a digit that represents the priority sequence for our project. A 10 corresponds to more improvement in a part where a lower digit would correspond to a lesser worry for improvement. For example, Power Efficiency and Pinpoint Location got a 7 which means that those two won’t be in need of improving since they really do not offer much relationship with each other because they don’t relate to each other at all.

The top side of the HOQ table corresponds to the need for execution in parallel to desired prioritized actions regarding the requirements chosen. There are symbols like a plus and minus sign to fully express the need for prioritization for all the requirements decided upon by the team. A plus symbol designates a strong need for action and a better mindset for future reference because this is needed for necessary enhancements down the line when something bad happens with the product creation. A minus symbol is the ultimate opposite which provides weaker need for attention for future use. The weights in which these aligned is very crucial because these serves as the engineering specifications just perfect for the customer needs and wants or else the relationship with the specific customer will crumble when it matters most. We have decided on which requirements are more suitable for our project extensively so that we have our minds set for what's to come in this journey of creating our Minuteman system.

3.2 Project Block Diagram:

For this project, it is definitely necessary that there is a widespread approval of everyone else's roles in the development and the design stages alike. We have carefully decided on which sectors that we encapsulate for the project so it is properly shown that what every member is doing whether it might be in the power module, central module where a microcontroller is included, sensor module or even the software side of things.

What is a block diagram? A block diagram constitutes a "quick, high level view of a system to rapidly identify points of interest or trouble spots. With this kind of structure, it has the chance to not offer much detail that is required for the likes of comprehensive planning or the implementation of the design. Block diagrams are essentially focused on specific parts of a project such as input or output of a design. This principle is called "black box" in the world of engineering. Block diagrams are similar to making flowcharts but, in this case, it is shown that every block in the diagram is color coded and, according to the professors, that each of the initials of the members of the team so it is clear that the roles are laid out properly. Our block diagram is spread properly as though it looks to be categorized in different modules. There are arrows that indicate where each module traverses into. For example, the Power Control Unit (PCU) module flows through the FPGA / Microcontroller that goes into the user interface module that is led by a variety of combinations of students.

There are varying practices for making these block diagrams enticing and better looking for the readers to have a look at in the future. Firstly, we have to identify the system that we are working on. We have to determine the system of roles that needs to be assigned so that each of the members have a clear look of what needs to be done because of the potential consequences that emerge if everyone thinks they have a concise view of their tasks that need to be

completed. We also have to determine which kinds of components that are required to be used, what are the flow of the different modules that are included in the system and what the outputs are for each of the inputs.
Next, we have to create and also label the block diagram. We had to determine carefully what kind of modules are needed for the minuteman system. It didn't take long for us to be sure of what the different pieces are in designing the system because all that really needed to be done was to identify the core components that blends in properly in creating a flaw free prototype for presentation come next semester. The arrows are a crucial part of the diagram because, without them, we wouldn't know what the other modules of the project relate to in terms of the other modules. The relationships between the modules decided on by team is vital in the development process of the project because it will be the deciding factor where the team can process into themselves what the system will look like for the final product. While this is the case, the team has to make sure there is ample accuracy from the professors so as though there will not be risks taken by the group and that each of the components are set properly.

Figure 3-1 shows the diagram, which shows the modules and division of tasks given to each individual member of the group. The blocks were made by looking at the objectives and requirements of the project. Each module represents a necessary function of the system. Each block in the diagram contains a description of the function and an abbreviation of initials. This divulges each group members' tasks for the project and the level of priority they have in that area. The priority is given from right to left.

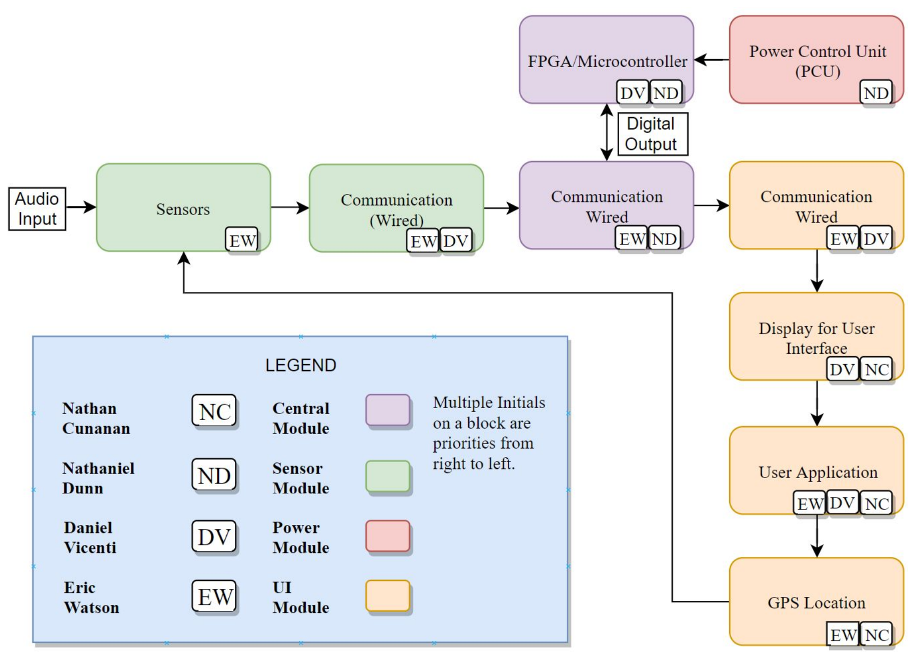

Figure 3-1: Project Block Diagram of Modules and Division of Tasks

3.3 Project Budget and Financing:

For the financing and budget of the project, each part was carefully analyzed and reviewed upon approval for each member of the group so that there is full dependency for everyone. This is one of the main factors in creating the project: money. As students, money is an aspect in current life where it isn't flourishing since all the focus is finishing and getting a degree. The whole team knows that this is going be a big focus in managing the project properly and that money would be an immediate issue because of the lack of it. We had the ideas of implementing sponsors for our project so that we don't have to worry about where to get the financing from but we have decided to stick with getting the finance naturally so, in that way, once our product gets finalized come summer, we will have the path of managing our own business in regards to our Minuteman system without having to depend or give back to our sponsors especially in the share in profits between the companies end and our end as well. No project plan is complete without a concrete plan for our budget. Figuring out the bottom line is what matters and we have to extensively process as a unit on what we are willing to spend based on the parts that we decide to purchase.

With a proper budget, it is important to come up with detailed estimates because it is essential that there is a foundation for where we start from in terms of which parts to get and where to source them as well. While the work is ongoing, it is definitely possible to track the project even if there possible changes that can appear during the design implementation. Our plan is to purchase all the materials needed for the project before the second semester starts because if we procrastinate and get our parts late, we can possibly lose time in the process since we don't have much leeway for our Senior Design 2 time span. The estimate of these costs will vary as the development of the system gets deeper and deeper. We will get a more proper view of what the costs really are as we learn more about the project.

There is an inclusion of resource cost rates. Since there is work needed for the project to be done, there are also rates that need to be calculated. All the materials that is used for the project also are counted for cost rates. The rate of the materials that are purchased will account for the labor needed by the members as well. While there material cost rates, there are also costs of quality. The quality related activities come into play in this kind of project in a way where the quality of the work process of each member is monitored so that the group doesn't get affected by a lack of performance which can result in a bad product. The cost of quality is just a way of tracking the cost of each activity that the team will encounter throughout this whole process. With this, it is super important to keep track of the estimates needed. In this way, we will know the assumptions made when we decided on making the numbers.

With all these steps, we can concur that our project creation will be smoother because we don't have to worry about where to get the money and plus we will learn from the initial makeup of the design that we can eventually change for the better later.

Table 3-2 shows the material costs and quantity of our project, providing some perspective on what will be necessary for the project's budget. This budget will change as the project gets more dense and more as we go further along the journey of making this project. We hope to have a targeted budget of $1000 or less by the time we have to purchase the parts.

Material/Part	**Quantity**	**Price**	**Total**
Arduino Due	1	$48.00	$48.00
AT91SAM3X8E	3	$11.54	$34.62
Power Adaptor	3	$2.99	$8.97
Analog Device LMT4622	2	$17.31	$34.62
MAX1044	2	$1.43	$2.86
CMC-6035-130T	12	$2.12	$25.44
BNO055	1	$34.34	$34.34
NEO-6 u-blox 6 GPS Modules	1	$15.99	$15.99
Maxim Integrated DS18B20	2	$8.20	$16.40
TL072	50	$0.15	$7.50
PCB X 5	2	$100.00	$200.00
Miscellaneous		$90.00	$90.00
		Total	$518.74

Table 3-2: Material Costs and Quantity

3.4 Project Milestones:

Knowing the do’s and don'ts of a senior design project is very hefty at times. In order to have a successful project development and design, the team needs to have cohesive and strict schedule in terms of determining when specific milestones have to be done and what kind of requirements are needed that would eventually result in a hassle free environment wherein each member of the team is able to work on their own sectors of the project as smooth as possible. If it wasn’t for specific milestones being implemented in our project design, our team will be in complete disarray and chaos due to not knowing what the real task in hand is Having no milestones will produce confusion amongst the members of the team that will possibly lead to more problems in terms of creating the software or hardware because procrastination has the major effect of affecting a person’s well being and ability to progress in the project plus being able to overcome all the unexpected obstacles that may arise at the time of creation of the Minuteman system.

The main point of these milestones are to be responsible for the duties possible and to be aware of the circumstances whether it may be good or bad depending on the situation of the person. The question is: how are project milestones decided upon? These milestones are decided upon critical events that the team might resolve as something that relays good progression and not disorganized information that can potentially disturb the team’s growth and rate of progress. The correct use of project milestones is at the start of a critical phase of work and at the end of the critical phase of the work as well. Another time to use project milestones are in the likes of deadlines. Milestones basically act as deadlines in the sense that it is referenced in by the team whenever there is confusion between members. By definition, a deadline is states as the latest time or date that a task should be completed” With this definition, it is imperative that all of these priorities be finished at the specified date. In our team, however, we will try and implement our deadlines that have cushion in between the real deadline date and a personal deadline date for the team as whole. In this way, there will be a leeway in between these two end points just in case there may be changes that are needed to be displaced for better results in the design.

Another question is: how frequently do we really need to put milestones in our project development? The answer is For every planning/ developing cycle it is important that a milestone is laid out before the team so the team understands properly what duties are aligned for them each time. With how humongous this project is, there should be consistent inclusion of these milestones at least 2 a week especially in regards to senior design 2 where the main purpose is to actually create the Minuteman system. It doesn’t apply to Senior Design I where it is purely research. There will be mandatory meetings each week to discuss potential changes to the plan or if there are changes that arise that is worth noting for future use. It isn't guaranteed that there are changes that will affect the flow of the development cycles, but there should be critical review for each of the next priorities so that the whole team is unified in what to do. Table 3-3 displays the full listing of our project milestones, as known currently. As shown, some dates are to be determined.

Milestone	**Time Duration**	
Task	**Start**	**End**
Research	1/10/2019	2/15/2019
Group Formed	1/10/2019	1/10/2019
Divide and Conquer	1/10/2019	1/30/2019
Task Division	1/10/2019	2/1/2019
Project Meeting	2/7/2019	
Update Divide and Conquer	1/31/2019	2/13/2019
FPGA/Microcotroller	1/21/2019	2/15/2019
Power	2/1/2019	2/15/2019
Communication	2/1/2019	2/15/2019
Sensor	2/1/2019	2/15/2019
Application	2/1/2019	2/15/2019
GPS	2/1/2019	2/15/2019
Software-Hardware Integration	2/1/2019	2/15/2019
First Draft Senior Design	2/15/2019	3/27/2019
Design	2/15/2019	4/20/2019
FPGA/Microcotroller	2/15/2019	4/10/2019
Power	2/15/2019	4/10/2019
Communication	2/15/2019	4/10/2019
Sensor	2/15/2019	4/10/2019
Application	2/15/2019	4/10/2019
GPS	2/15/2019	4/10/2019
Second Draft Senior Design	3/29/2019	4/10/2019
First Design Revise	4/10/2019	4/15/2019
Second/Final Design Revise	4/15/2019	4/20/2019
Final Draft Senior Design	4/12/2019	4/20/2019
Verification/Implementation	4/25/2019	TBD
FPGA/Microcotroller	4/25/2019	TBD
Power	4/25/2019	TBD
Communication	4/25/2019	TBD
Sensor	4/25/2019	TBD
Application	4/25/2019	TBD
GPS	4/25/2019	TBD
First Test Revision	5/20/2019	5/30/2019
Second/Final Test Revision	5/30/2019	6/30/2019
Final Project Presentation	TBD	

Table 3-3: The Project Milestone Listing

3.5 Decision Matrix:

The project was decided through meetings that took place after class and at certain meeting times. After the first three meetings, we created a priority project list shown below. We weighted our individual desires and what resources the project would cost. We ranked what we wanted with one being the best and 5 being the worst. The score is added up in the Decision Matrix below in the Group Ranked tab. After this, each project was analyzed with how much time it would consume. The cost of the project and capacity of knowledge that we would want to expand. Table 3-4 shows the decision matrix of the group, including the ranking each member of the group gave to each of the project ideas presented between us.

Project Idea	Nathan	Nathaniel	Daniel	Eric	Added Lowest Score	Group Ranked
Master/Slave RC Car	3	4	2	4	13	3
Warning Bracelet	2	1	3	2	8	2
Gunshot Detection	1	2	1	1	5	1
Laser Guided Drone	4	3	4	3	14	4
Mini Surveillance	5	5	5	5	20	5
Time	**Cost**	**Knowledge**	**Resource Cost**	**Total**	**Total Rank**	
3	3	4	10	23	3	
1	2	3	6	14	2	
2	1	2	5	10	1	
5	4	1	10	24	4	
4	5	5	14	34	5	

Table 3-4: The Decision Matrix

The ranking show with the Group Ranked and the Total that the Gunshot Detection project was the best option for our group. This project was taken well by all members in our group with the desire to help each other in time of need. This project will not only cost the least but it will also help expand the knowledge of the group.

3.6 Methodology for Design:

Developing the necessary code for the Minuteman project, our crew will be focusing on the usage of agile software development paradigm. This kind of methodology adapts unique way of planning how to start developing code in different stages. As our device is being created, it is expected that there will be new constraints or requirements that will arise as the deadline gets closer and closer or when new information is gathered. There may be times when the team is starting to build the device but would encounter some roadblocks along the way whether it may be a piece that needs to be replaced because of unforeseen circumstances that will pop up and we find out that a part is not compatible with the system as a whole or it is not robust enough when the software application is running. With agile movement, this is a very perseverant movement that forces users to readjust with their developmental tactics or plan when obstacles appear along the way of development. Continuous improvement and early delivery are key in making this system work without any dead ends so that each developers knows the main plan of attack.

This is compared to the traditional linear system of planning, designing, implementing, careful testing and eventual release. With an agile system, it will follow a regular cyclic schedule where testing passes through the planning phase and back again cycling every single stage until it has a definite signal of a thumbs up for a release. What this kind of scheduling does is that it will constantly accompany any new changes or ideas to the system whether it's a concrete change or a minor change that doesn't need much thought put into it.

The kind of framework that the team will be using is Scrum. Using the Scrum process, the development of the Minuteman into various levels of sprints throughout all methods of project development, that varies within the amount of difficulty that a task desires or needs. A scrum framework is led by scrum master to monitor each stage of development and decide on whether progress has been made and appeal any modifications that may be needed for the system from everyone in the team. Since our whole team has experience is software development,half of us will be targeting microcontroller programming and the other half will be handling the task of programming the UI and the arduino boards.

4. Research and Theory

4.1 Existing Projects

There are a number of pre-existing projects to Minuteman that have similar goals. They are typically gunshot detectors designed for large cityscapes or smaller facilities such as school campuses. These projects include a previous Senior Design project, several products for the private marketplace, and several products for military/government use. These are typically made with various more specific applications in mind, but they generally follow the same idea of using audio sensors to detect gunfire and respond accordingly.

4.1.1 Project GLASS

Project GLASS was a senior design project created in 2014. GLASS was created with similar circumstances in mind to Minuteman, attempting to breach the political divide created by mass shootings in the American public by creating a technological solution to the issue. Although GLASS is more focused on outdoor applications, such as in cities, it is relatively similar in that it detects gunshots, determines the type of firearm used, and locates where the gunshot originated, and then outputs this data to an application that can then inform authorities.

GLASS determines a gunshot from an event by looking for specific conditions to confirm that a gun was fired; a decibel level exceeding 140 decibels, peak frequency in the event indicating the gunpowder discharge in the barrel, and the subsonic frequency made by the projectile flying through the air. The former filters out most events not involving a firearm (although GLASS admits that suppressed weapons could have a lower decibel level), the second condition allows GLASS to also determine the caliber of the projectile, and the latter allows GLASS to confirm if it was definitely a firearm and not something similar like a recording.

GLASS ultimately didn't produce their theoretical product, and instead had to create a prototype to demonstrate it's abilities using the limited resources their team had available. The theoretical design involved four microphones with individual processors to filter out events that don't meet the first condition. Once an event does, it passes it on to a processing unit to analyze the event and filter it through the two remaining conditions. Then it passes this information on to an android application. Their prototype ended up using a simpler audio sensor that instead communicated with a normal computer, but demonstrated what the project attempted to do. [3]

4.1.2 Shotspotter

Shotspotter is a company that specializes in providing outdoor gunshot detector products to law enforcement, as well as public and private facilities. Although they have several products currently, they are all based off of their Shotspotter system. Over a large area, such as a neighborhood or campus, the system detects gunshots and timestamps when they detected it. Knowing the speed of sound, and the direction of the sound, the system is able to use these timestamps to determine the distance, and pinpoint where the gunshot occured [4].

Shotspotter offers four products utilizing their technology, although we will focus on two of these products since they are more relevant to Minuteman.

4.1.2.1 SiteSecure

SiteSecure is a general product provided for facilities, whether they be businesses, hospitals or school campuses. It's unclear how it would differentiate from SecureCampus on the latter. The detector is able to determine the number of shooters and shots fired, whether the weapon is automatic, and the direction of travel. It can accurately pinpoint the location of a gunshot to within 25 meters Then it sends this information to authorities and on-site personnel through alerts. It can also be integrated with other on-site security systems. It has a successful detection rate of about 90%, and can send out alerts within a minute. Audio samples are verified by a ShotSpotter Incident Verification Center [4].

SiteSecure is also advertised to energy infrastructure sites, such as power stations or transformer sites. This describes them as fence or pole mounted systems. It also denotes that ShotSpotter can detect gunshots from up to 500 meters away. It also claims that it is able to detect subsonic projectiles, and can locate them even if the gunshot is not toward the direction of the device [4].

4.1.3 Raytheon Boomerang III

Boomerang is a military shot detection system designed to detect ballistic projectiles heading toward it, determine the direction they came from, then deliver this information to its users. It filters out non-ballistic sounds, as well as gunfire going out away from the device. It can be mounted onto protected buildings or facilities, or on vehicles such as a HUMVEE.

The system works through a multi-directional microphone array, as shown in Figure 4-1. It passively detects acoustic events and filters them through signal processing. Boomerang audibly calls out that a shot was detected, and states the rough clockwise direction and distance, while presenting more accurate data to an LED display. It is also able to broadcast the data to an outside facility or center through an Ethernet connection.

The system is able to operate on vehicles moving up to 60 miles-per-hour in any outdoor terrain. It has a detection rate of greater than 95%, and is able to pinpoint the direction to within 2.5 degrees and distance with about a 10% margin of error. It requires no calibration, just activation. It can be easily incorporated with a separate system or third-party systems through an integration kit and interface. [6]

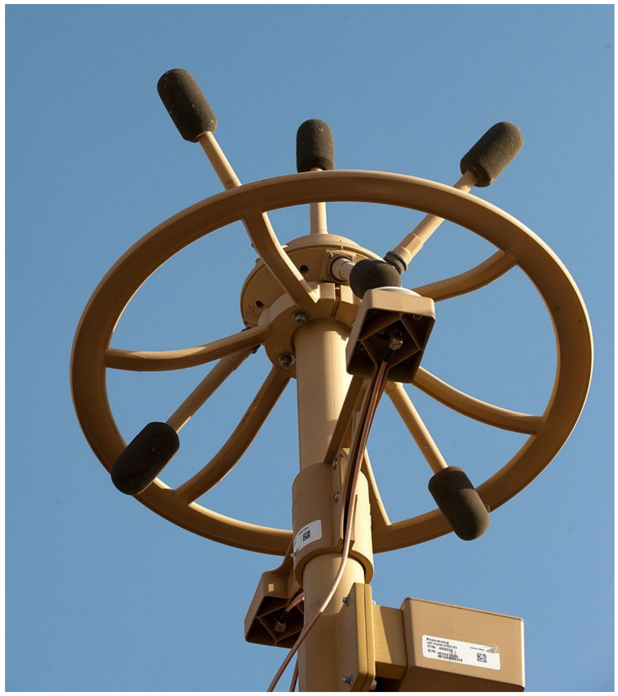

Figure 4-1: Boomerang III microphone cluster. Open Government License. Obtained from wikipedia.

4.1.3.1 Raytheon Boomerang Warrior-X

The Boomerang Warrior-X is a derivative product of the Boomerang III system. It is designed to be worn over the shoulder of soldiers in active combat areas. It features a separate display and like the Boomerang III, it detects incoming gunfire and returns where it is coming from. It is then able to relay this information through audio played aloud or through an earpiece.

Like the Boomerang III it has a detection rate of greater than 95%, although the margin or error is greater for directions, at less than 7.5 degrees. The margin of error for range detection is also greater at 20% rather than 10%. No information is available about it’s power supply but it is able to operate for about 12 hours. [7]

4.1.4 Guardian Indoor Active Shooter Detection System

The Guardian Indoor Active Shooter Detection System is created by a company called Shooter Detection Systems. As the name describes, the system is designed to detect indoor shootings within a building or facility. It detects a gunshot by both picking up the infrared light emitted by the muzzle flash of the weapon and the audio generated by the gunshot. It is able to use both to locate the shooter within a provided floorplan of the building. Upon detection, it is able to alert security through its own program or application, and is able to integrate with third-party systems.

The devices themselves take the appearance of a normal wall-mount, similar to a power outlet. Due to it using both infrared and audio detection methods, it is able to confirm gunshots with a high degree of accuracy. Although no statistics were provided, it is stated that the system has a range of a 40m radius from the mount. It draws about 1 watt of power through a power over ethernet relay, which connects to a computer containing the software needed to run the system. [8] [9]

4.1.5 SenseShot

SenseShot is a series of products provided by Information Systems Technologies Incorporated that offers both indoor and outdoor gunshot detection systems. Specifically they are designed to be low cost, and low power solutions. For their indoor application, it consists of a network of MEMS microphone sensors similar in appearance to a fire-alarm system. Much like Guardian it utilizes a Power-over-Ethernet connection to a relay that connects to a base station that is hosting the software for the system. At this system, the software filters out real events from false alarms through analysis, like GLASS. It is also able to estimate the shooter’s location and the trajectory of the gunshots detected.

The SenseShot indoor detection system can pick up low-powered firearms to an estimated 100 meters, while being able to pick up high-powered firearms at 200 meters. It is recommended they are spaced out every 200 meters, and the network itself can handle up to 252 sensors. It can be set up to take 120 volt A/C power, and has a wireless wifi option.

Their outdoor detection system operates through a network of audio sensors mounted onto municipal street lights. This network is managed through a server operating their SensorAgent software. Multiple sensors are used to both confirm and triangulate the shooting, and report it to authorities, even directly to on-duty officers through dispatch. The system requires at least 2 arrays. It is able to detect a low-powered weapon within about 500 meters, and a high-powered weapon within 1km of the sensor. It can utilize either ethernet, fiber optic, or wireless signals for each sensor. [10]

4.1.6 Observations

In conclusion, we have decided that it might be cheaper to integrate an outdoor audio system. While an indoor system would be greatly useful, due to reverberation, relying on audio confirmation can make the results inaccurate. While incorporating a secondary detection system, such as infrared, can make the results far more accurate, the implications is that the range and visibility of the sensors requires that more sensors are provided. Since our goal is to provide an affordable platform, more sensors would increase the costs of our system.

The Boomerang III and Warrior-X provide interesting examples of just how accurate an audio detection system can be in the outdoors, even while on a moving platform such as a vehicle or individual. It is likely that our system will be similar, but will use triangulation between multiple sensors to more accurately locate the shooter.

ShotSpotter offers an interesting take on facility-based shot detection systems, even if technical specifications aren't all that available. However, SenseShot and GLASS are able to provide more technical angles on such things, especially with ideas such as Power-over-Ethernet connections, multiple condition filters, such as decibel level and muzzle-blast shockwaves. These are likely to influence the design of Minuteman, in that it will be an outdoor gunshot audio detection system. Likely for facilities, rather than large scale environments.

4.2 Gunshots

Detecting acoustic events relating to gunfire requires understanding the nature of these events. When a gun is fired, it is propelling a led projectile (sometimes with other materials, such as copper) out through a barrel, by means of gaseous expansion of ignited gunpowder. Upon leaving the barrel, the gun generates a muzzle blast as the gases escape after the bullet. The bullet itself, due to the velocities it is propelled at, usually creates a supersonic shockwave behind it as well. [11] The only exceptions to these outcomes is whether the firearm is equipped with a suppressor and if the ammunition itself is designed to propel the bullet at subsonic speeds.

Excluding those exceptions, and despite other varying characteristic which will be explored, a gunshot will always generate a sound wave above 130 dB within a certain vicinity of the gunshot (depending on the caliber and intensity of the muzzle blast). The muzzle blast will generate a spherical sound wave from the location of the blast, whereas the shockwave of the bullet will follow it in a conical pattern, following the doppler effect, expanding from the trajectory of the bullet as it soars through the atmosphere. [11]

The rest can vary depending on the firearm and ammunition. Factors such as gas system, barrel length and rifling, to projectile shape and consistency of the primer and gunpowder can alter these characteristics in varying ways. This is why, although a gunshot can be fairly distinct from other sound events, it is important to be able to roughly make a distinction between the characteristics of these events for the purposes of confirmation.

When the shockwave is picked up by a sensor, it will appear as a thin “N” shaped waveform. This is the result of the shockwave compressing air particles at a high-velocity (represented by the upper amplitude) with a trailing rarefaction wave of low-velocity particles (represented by the lower amplitude). This is distinct to ballistic projectiles. This can be used to confirm that a gun is being fired by determining if the sensor detects a shockwave following the projectile. [12]

The amplitude of the N-wave is weakly dependent on the Mach velocity of the projectile [12]. Thus, so long as the bullet is supersonic, it will emit a very clear shockwave that can be detected by a sensor. However, the amount of time it takes to receive the shockwave after the muzzle-blast depends on the position of the sensor, due to how the shockwave follows the bullet in a conical pattern, rather than just releasing in a circular pattern from the muzzle-blast. Thus any method to detect the shockwave will first need to respond to the muzzle-blast, and need to be able to pick up the shockwave within a set amount of time.

4.3 Triangulation

For our project, triangulation is one of the options for determining where the sound of a discharged weapon is coming from. Triangulation uses trigonometry in order to determine the position of an object by using angles to get the direction where it’s coming from.

2D Triangulation - For using 2D triangulation, two arrays of three microphones would be needed, which would be shaped in the form of an equilateral triangle. Using two arrays, the angle obtained from each array can be used to find a point of intersection, which is where the source where the sound is coming from.

In order for this approach to work, the microphones would need to be close together, and the sound be far enough so it can be considered a line that is perpendicular to the origin. Sound has to be picked up far away anyways, as the microphones that will be used have a decibel peak of 130 dB for the standard pressure level. If a firearm is discharged near the array, this would lead to distortion as most firearms have a decibel peak of around 140 dB to 150 dB, so the sound picked up from the microphone would not be able to be compared to other firearm sounds due to this distortion.

When the array of microphones picks up sound from a discharged firearm, the time difference between the first microphone and the second microphone to pick up the sound can be used. This is represented as (t_B - t_A), where t_A is the time of the first microphone, and t_B is the time of the second microphone. Knowing the speed of sound, which is represented by C, the distance can be calculated by multiplying the speed of sound by the time difference, which is represented as Δx in the following equation:

$$\Delta x_{BA} = C \times (t_B - t_A)$$

The equation for speed of sound used in our project is the ideal speed of sound equation related to temperature in a dry environment. This equation was used, as the pressure and humidity did not have much of a significant impact of the speed of the sound. A website [16] with a handy calculator was used to perform the calculation of the speed of sound using data from a local weather report, where the temperature (T) was equal to 28℃, pressure (p) was equal to 101.8 kPa, and relative humidity (φ) was equal to 36%, which resulted in a speed of sound of 348.74 m/s. The calculation for the ideal speed of sound with the temperature (T) equal to 28℃ is 348.1 m/s, which uses the equation shown down below. Using this comparison, only temperature is really necessary to be measured for our project, so no barometer or humidity sensor is needed for our project, which helps save cost on components.

$$C = 331.3\sqrt{1 + \frac{T}{273.15\ K}}$$

Knowing the calculated distance Δx and each side of the equilateral triangle is the same value, which can be represented as S, trigonometry can be used to calculate the angle from these values. Since the sound is considered perpendicular, a right triangle is formed on the side of the equilateral triangle, where the side of the equilateral triangle is considered the hypotenuse of the right triangle, and the distance Δx is considered the adjacent side of the right triangle, so the cosine function of the adjacent over the hypotenuse is used, which is then inverted to solve for the angle, which is represented as θ. After θ is calculated, the angle α can be calculated by adding the angle θ to the 60° angle of the equilateral triangle next to it. Figure 4-2 shows a visual representation of this.

$$\theta = \mathrm{Cos}^{-1}(\Delta x \div S) \qquad \alpha = \theta + 60°$$

Even if solving for the angle α is more than enough to get the angle the sound source is coming from, error in the degree of the angle calculated can still occur. This amount of error can be determined by using two equations. The first equation is solved by getting the time difference of the two microphones that picked up the sound after the first microphone, which for this equation, can be assumed to be $(t_B - t_C)$. The difference is then multiplied by the speed of sound, which is represented by C. This distance can be represented as Δy_{BC}, which is the side of another right triangle that is formed. Knowing the angle α is the addition of θ and the equilateral triangle angle of 60°, a straight line can be formed if another angle is added so it is 180°. This can be used for the second equation of Δy, which uses the trigonometric function where the cosine of (180° - α) is set equal to the adjacent side of the triangle (Δy) divided by the hypotenuse of the side of the equilateral triangle (S). The equation is then rearranged in which the value for Δy is solved for. If both equations of Δy are not equal to each other, then that means there is an error in the angle calculated. The equations are shown below. Figure 4-2 also shows the visual representation of sound moving directly to the front microphone A and sound coming from the right side of the microphone array.

$$\Delta y_{BC} = S \times \cos(180° - \alpha) \qquad \Delta y_{BC} = C \times (t_B - t_C)$$

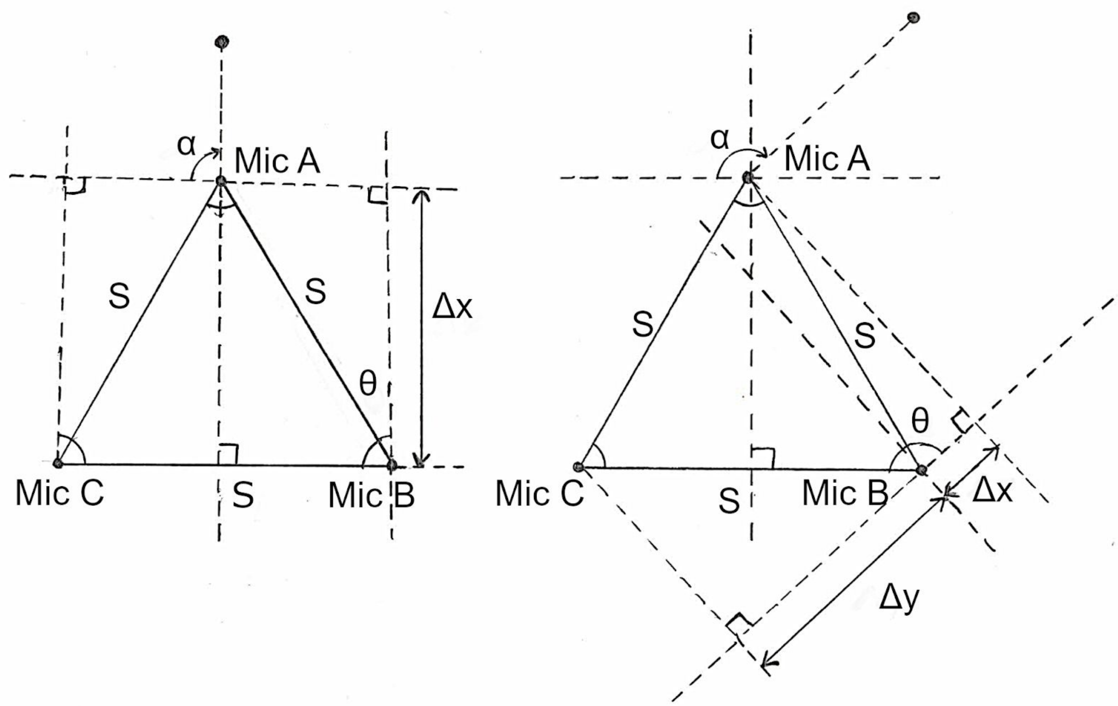

Figure 4-2: Shows how the direction of sound is located from the mic array

Since the angle of the direction of the sound can be located, the distance can be determined to the source if a secondary microphone array picks up the sound of the discharged firearm. Since the arrays would be a distance apart, the length between the the microphones of both arrays that picked up the sound first would be known. This would be done by using a GPS module that would give the coordinates of where the array is. Since a GPS module is used, a compass would also be needed to get the orientation of the microphone array, so the values of θ and α can be adjusted accordingly. With these adjustments, the position of the sound source should be able to be obtained. The direction angles of α_1 and α_2 of the two microphone arrays are greater than 90°, so two new angles of β_1 and β_2 would be calculated by having the angle α subtracted from 180°. Given both of these are less than 180°, the angle of β_3 can be calculated by having (β_1 + β_2) subtracted from 180°. Since the distance between both microphone arrays are known, which can be represented as D_3, and all angles of the triangle formed between both microphone arrays are known, the law of sines can be used to solve the distance of one of the microphone arrays from the source. The sine of β_3 divided by the distance D_3 can be set equal to either the sine of β_1 divided by the distance D_1 and solve for D_1 or the sine of β_2 divided by the distance D_2 and solve for D_2. The equations are shown below and Figure 4-3 represents the 2D triangulation.

$$\beta_1 = 180° - \alpha_1 \qquad \beta_2 = 180° - \alpha_2 \qquad \beta_3 = 180° - (\beta_1 + \beta_2)$$

$$D_1 = [\text{Sin}(\beta_2) \times D_3] \div \text{Sin}(\beta_3) \qquad D_2 = [\text{Sin}(\beta_1) \times D_3] \div \text{Sin}(\beta_3)$$

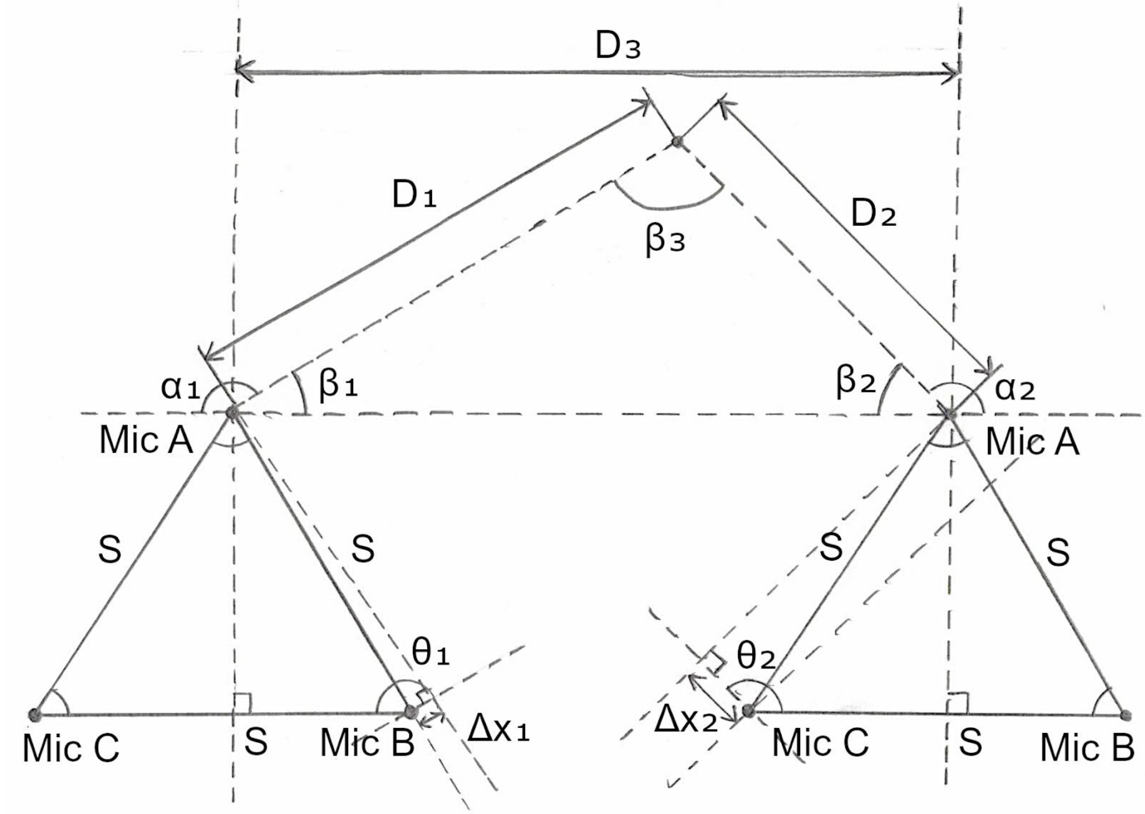

Figure 4-3: Diagram showing two microphone arrays locating sound source

As the distance can now be calculated using the law of sines, the position of the sound source can be located, but the distance needs to be split into x and y components. To get the x component of the distance D_1, the cosine of β_2 is multiplied by D_1, and to get the y component, the sine of β_2 is multiplied by D_1. The same procedure is used to split the components of D_2, but D_{x2} will be negative as it is moving to the left as shown in Figure 4-3. The equations to split the components are shown below:

$$D_{x1} = Cos(\beta_1) \times D_1 \qquad D_{y1} = Sin(\beta_1) \times D_1$$
$$D_{x2} = -Cos(\beta_2) \times D_2 \qquad D_{y2} = Sin(\beta_2) \times D_2$$

3D Triangulation - GPS utilitizes latitude and longitude as it’s coordinate system, so the microphone array in this project would need to be 3D, so an additional microphone would need to be added. This would then form an equilateral triangular pyramid for its shape, as shown in Figure 4-4 for the side view and top view. The fourth microphone in the array would use the same calculations as done for the 2D coordinates. When observing the figures below, it is found that the coordinates can become spherical, where α_2 in the Figure 4-4 side view can be represented as θ for spherical coordinates if α_2 is subtracted from 90°, as θ starts from the z-axis and moves clockwise to a maximum of 180°.

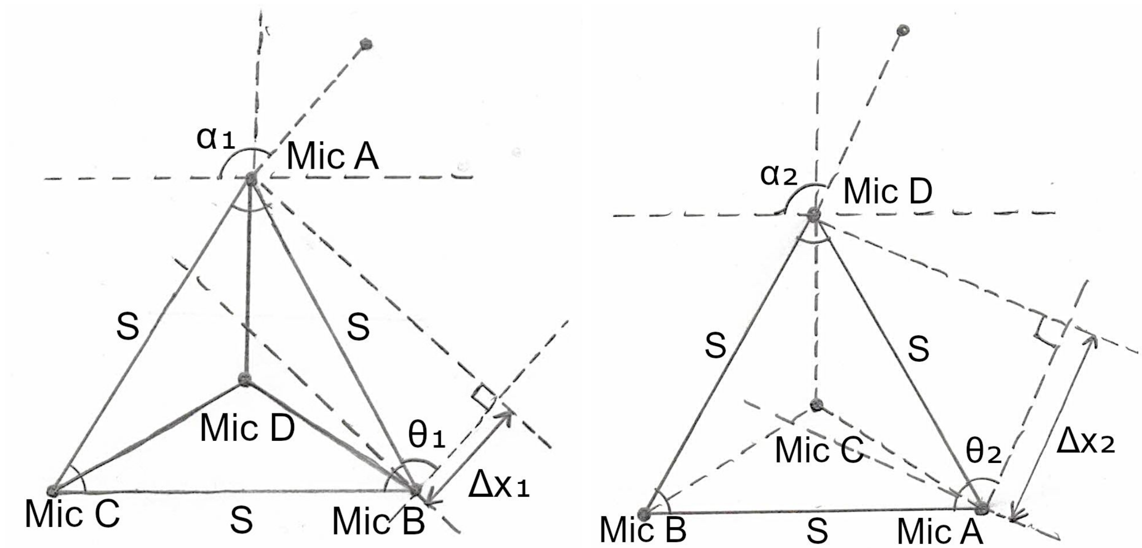

Figure 4-4: Diagram of the top (left) and side (right) view of the 3D microphone array

In the Figure 4-4 top view, α_1 can be considered φ, but would need the have the value of α_1 subtracted from 270°, as φ starts from the x-axis and rotates to a maximum of 360° counter-clockwise. The radius of spherical coordinates would then be the distance of the source of the sound to the first microphone that picked up the sound, which is the length of D. Knowing this, the coordinates calculated should be able to be easily used with the GPS coordinates of the microphone array.

With triangulation, having the requirements of two or more microphone arrays can be problematic, as if only two microphone arrays are used, and there is a scenario in which one breaks down, then this can lead to the failure of detecting a discharged firearm's location, but the usage of using one microphone array can still be used to report it to authorities. When multiple firearms of the same type are being discharged, each microphone array could potentially get sound from two different firearms. Having this occur, the location of the sound source calculated would be wrong, as both sounds would be from two different locations, compared to one location using both arrays from two different sounds.

4.4 Multilateration

Multilateration is another known method that can be used to locate a transmission of a source. A real life application of this is known as WAM (wide area multilateration). For WAM to work, an aircraft would send a transmitted signal, which is received from several antennas at known positions from each other. Each antenna would be synchronized at the same clock and would receive the transmitted signal from the aircraft, which would be at a different time from each other. The antennas would then compare their TDOA (time distance of arrival) and would use this to form hyperboloids which are then intersected with each other. When the hyperboloids intersect, they would give the position of the aircraft. [17]

2D Multilateration - For WAM, the transmitted signal would have the speed approximately of the speed of light, which is approximately 3 x 10^8 m/s. This method can also be applied for sound as well, if the speed of sound is used instead of speed constant of a signal sent from an emitter. Knowing this, each microphone can be used instead of antennas and the same principle of the intersection of hyperboloids can be used to determine the location of a discharged firearm for our project. Figure 4-5 shows a visual representation of how hyperboloids are used to determine the location of a source. The left side of the figure shows how the Loran-C hyperbolic navigation works, in which a flat line of a hyperbola is determines the possible location of the source of the signal. The right side of the figure shows how intersection of hyperbolas between multiple receivers can be used used to determine the location of a source.

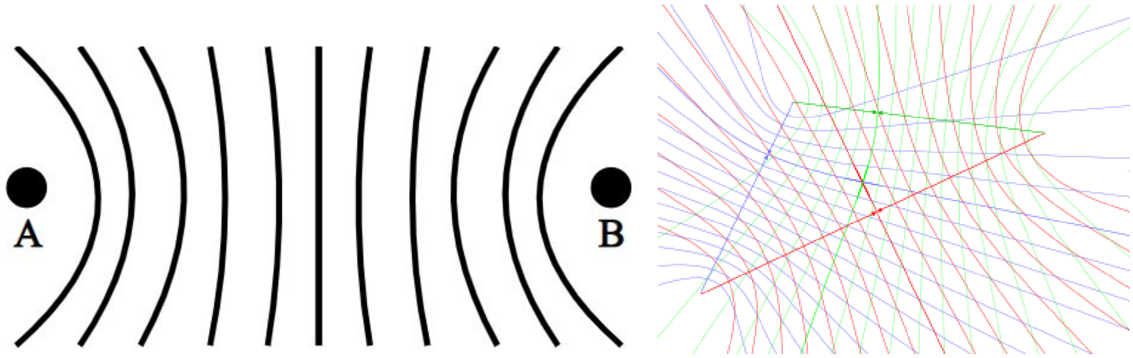

Figure 4-5: (left) Loran-C hyperbolic navigation [18], (right) 2D Multilateration [19]

Multilateration would still require the same components as used for the triangulation method. The GPS module would provide the origin of the microphone array, so the coordinates of each microphone relative to the GPS module are known. The compass would also provide the orientation of the array, so adjustments can be made when the location of the source is obtained.

2D Multilateration would use the same three microphones used for 2D triangulation, with the exception that it does not need to be in the form of an equilateral triangle. Knowing this, the distance equation can be utilized for multilateration, which uses the pythagorean theorem. Using the distance equation, the axis values of x and y are both the power of 2, which are both added, and set equal to the distance of D which is also the power of 2. For a microphone, represented by 'i' that receives the sound, the values of x and y are replaced by $(x - x_i)$ and $(y - y_i)$, where x_i is the distance of microphone 'i' from the origin of the x axis, and y_i is the distance of microphone 'i' from the origin of the y axis. The value of D is also replaced with D_i for the microphone that receives the sound input.

$$(x - x_i)^2 + (y - y_i)^2 = D_i^2$$

The microphone array would pick up the sound of the discharged firearm the same way as done using triangulation. The value of D_i is calculated using the distance formula, in which the speed of sound C is multiplied by the time of t_i, when it detected the sound of the discharged firearm. The microphone array would also use the temperature probe that would be used for the triangulation method, which would then be plugged into the equation for the speed of sound, which can then be applied to the distance equation.

$$D_i = C \times t_i \qquad C = 331.3\sqrt{1 + \frac{T}{273.15\,K}}$$

Knowing the value for D_i can be calculated, the distance equation using the pythagorean theorem can be squared for the values of $(x - x_i)^2$ added to $(y - y_i)^2$

and D_i^2. The value of D_i will also be replaced by the speed of sound C multiplied by t_i.

$$\sqrt{(x-x_i)^2+(y-y_i)^2} = D_i \quad \rightarrow \quad \sqrt{(x-x_i)^2+(y-y_i)^2} = C\ x\ t_i$$

Knowing there are three microphones for the microphone array in 2D, then there are three equations that can be written, which would be represented as A, B, and C.

$$\begin{aligned}\sqrt{(x-x_A)^2+(y-y_A)^2} &= C\ x\ t_A \\ \sqrt{(x-x_B)^2+(y-y_B)^2} &= C\ x\ t_B \\ \sqrt{(x-x_C)^2+(y-y_C)^2} &= C\ x\ t_C\end{aligned}$$

In the current form, all equations have the values for x and y being unsolvable. However, by using the difference of time from the first microphone that picks up the sound from the discharged firearm and the other two microphones, two equations can be written. Knowing there are two unknowns, and two equations, the values for x and y can be solved for.

$$\begin{aligned}\sqrt{(x-x_B)^2+(y-y_B)^2} - \sqrt{(x-x_A)^2+(y-y_A)^2} &= C\ x\ (t_B-t_A) \\ \sqrt{(x-x_C)^2+(y-y_C)^2} - \sqrt{(x-x_A)^2+(y-y_A)^2} &= C\ x\ (t_C-t_A)\end{aligned}$$

This equation can also be simplified further by having the origin changed as the first microphone that receives the sound from the discharged firearm. The relative distance values of the other microphones, represented as 'i', has x_i and y_i changed to x_i' and y_i', which are set equal to the value of $(x_i - x_0)$ and $(y_i - y_0)$, where x_0 and y_0 are the relative position of the microphone that first received the sound.

$$\begin{aligned}\sqrt{(x-x_B')^2+(y-y_B')^2} - \sqrt{x^2+y^2} &= C\ x\ (t_B-t_A) \\ \sqrt{(x-x_C')^2+(y-y_C')^2} - \sqrt{x^2+y^2} &= C\ x\ (t_C-t_A)\end{aligned}$$

3D Multilateration - Knowing three hyperbolas are utilized by 2D multilateration to solve the x-axis and y-axis location of the discharged firearm, then three hyperbolas would be utilized for 3D multilateration, as shown in Figure 4-6. This would add a fourth microphone similarly to how 3D triangulation is performed, but differs in which the fourth microphone can be located anywhere on the z-axis instead of needing to form the shape of a pyramidal triangle. Using the same four equations, the time difference between the first microphone that received the sound and the other three microphones can be used to solve for the unknown values of x, y, and z.

$$\sqrt{(x-x'_B)^2 + (y-y'_B)^2 + (z-z'_B)^2} - \sqrt{x^2+y^2+z^2} = C\ x\ (t_B - t_A)$$
$$\sqrt{(x-x'_C)^2 + (y-y'_C)^2 + (z-z'_C)^2} - \sqrt{x^2+y^2+z^2} = C\ x\ (t_C - t_A)$$
$$\sqrt{(x-x'_D)^2 + (y-y'_D)^2 + (z-z'_D)^2} - \sqrt{x^2+y^2+z^2} = C\ x\ (t_D - t_A)$$

Knowing these equations are nonlinear, they are considered too computationally expensive to solve if a cheap microcontroller is used for our project. However, given there are three values of x, y, and z, it is possible to linearize this equation and remove any square roots. This linearized equation can then be used with a 3x3 matrix, and the method of gaussian elimination will solve the values of x, y, and z.

The equation would be linearized by first using a value to represent the square root of the distance equation, where R_i can represent the square root of $(x - x_i)^2 + (y - y_i)^2 + (z - z_i)^2$. Using the TDOA between the microphone of 'i' and the first microphone '0' to pick up the sound, the square of R_i to the power of 2 can be set equal to $[(D_i - D_0) + R_0]^2$. The equation can then be further simplified and be set equal to 0, as shown below.

$$[(D_i - D_0) + R_0]^2 = R^2_i \rightarrow \qquad (D_i - D_0) + 2R_0 + [(R_o^2 - R_i^2) \div (D_i - D_0)] = 0$$

Since $2R_0$ is the square root of the problem that needs to be eliminated, then another equation can be set equal to the simplified equation above. This can be done using any of the other microphones in the array that is not the first microphone. The microphone of 1 will be used to be set equal to this, which will then remove the value of $2R_0$.

$$(D_i - D_0) + 2R_0 + [(R_o^2 - R_i^2) \div (D_i - D_0)] = (D_1 - D_0) + 2R_0 + [(R_o^2 - R_1^2) \div (D_1 - D_0)]$$
$$(D_i - D_0) - (D_1 - D_0) + [(R_o^2 - R_i^2) \div (D_i - D_0)] + [(R_o^2 - R_1^2) \div (D_1 - D_0)] = 0$$

After the square root is removed, R^2_i, R^2_1, and R^2_0 can have their x, y, and z values foiled. These would then be substituted to replace $(R_o^2 - R_i^2)$ and $(R_o^2 - R_1^2)$.

$$-x_i^2 - y_i^2 - z_i^2 + 2[(x)(x_i) + (y)(y_i) + (z)(z_i)] = (R_o^2 - R_i^2)$$
$$-x_1^2 - y_1^2 - z_1^2 + 2[(x)(x_1) + (y)(y_1) + (z)(z_1)] = (R_o^2 - R_1^2)$$

After the equations above are substituted, the only values of x, y, and z are unknown, while the rest are constant values. Knowing this, the equation can be written in the format of Ax + By + Cz + D = 0, where each constant of A to D can be calculated as shown below. The constant of D is not to be confused with the distance of D_i. D_i is substituted with the distance equation as shown below.

$$A_i = \frac{2}{C}\left(\frac{x_i}{t_i - t_0} - \frac{x_1}{t_1 - t_0}\right), \quad B_i = \frac{2}{C}\left(\frac{y_i}{t_i - t_0} - \frac{y_1}{t_1 - t_0}\right), \quad C_i = \frac{2}{C}\left(\frac{z_i}{t_i - t_0} - \frac{z_1}{t_1 - t_0}\right),$$

$$D_i = C(t_i - t_1) + \frac{1}{C}\left(\frac{-x_i^2 - y_i^2 - z_i^2}{t_i - t_0} + \frac{x_1^2 + y_1^2 + z_1^2}{t_i - t_1}\right), \quad A_i x + B_i y + C_i z + D_i = 0$$

Since two microphones are already used in these equations, then in order to solve for three unknowns, a fifth microphone would be needed to be added to the microphone array. With the fifth microphone added, three equations can be solved. The values of x, y, and z can then be solved using gaussian elimination, using the equation of Ax = B. B is a 3 x 1 matrix that contains the constants of D_2 to D_4. A is a 3 x 3 matrix where the first row is A_2, B_2, C_2, second row is A_3, B_3, C_3, and the third row is A_4, B_4, C_4. The 3 x 1 matrix of 'x' contains the values to solve for of x, y, and z. In order to solve for x, y, and z, the matrix of A is inverted, which is then multiplied by the matrix of B, which then solves the values of x, y, and z. Given these, the distance of a discharged firearm with respect to the microphone array is known, and the GPS and compass can be used to get the correct position of the firearm used.

$$\begin{bmatrix} x \\ y \\ z \end{bmatrix} = \begin{bmatrix} A_2 & B_2 & C_2 \\ A_3 & B_3 & C_3 \\ A_4 & B_4 & C_4 \end{bmatrix}^{-1} \begin{bmatrix} D_2 \\ D_3 \\ D_4 \end{bmatrix}$$

Multilateration, compared to triangulation, offers advantages that make it easier to utilize for our project. This includes reducing the amount of materials needed for this project, as only one array of microphones would be needed instead of using two microphone arrays. This can also allow it to become mobile, as it would be possible to mount it on a vehicle given only one microphone array is needed. However, if the microphone array is mobile, this can affect the results of the location of the sound, as the vehicle moving can cause differences in the time of arrival between microphones that receive the sound, which would reduce the accuracy of finding the location of a discharged firearm.

Sample rate is also a design concern, as having a lower sample rate would reduce the time difference between microphones, which will affect the accuracy of the location of a discharged firearm. This can be alleviated by spacing microphones further apart, but this can eventually lead to issues with the stability of the structure. Having microphones that are spaced further away can also eventually lead to the point where it would not be able to be easily transported by hand, and would also go over the weight restriction of 15 to 10 lbs.

If the microphone array is designed to be mobile, the compass utilized would require an accelerometer and gyroscope added with the magnetometer. A new compass with a magnetometer, accelerometer, and a gyroscope together would be expensive. However, if only one microphone array is needed, then only one GPS module and compass module is needed, which from this offset, would be around the same cost using just one GPS module and a more advanced compass. Factoring the cost of the structure for the microphone array, using only one would save cost as well. From these observations, utilizing multilateration is more beneficial than triangulation, and will allow more features for how the sensor module can operate.

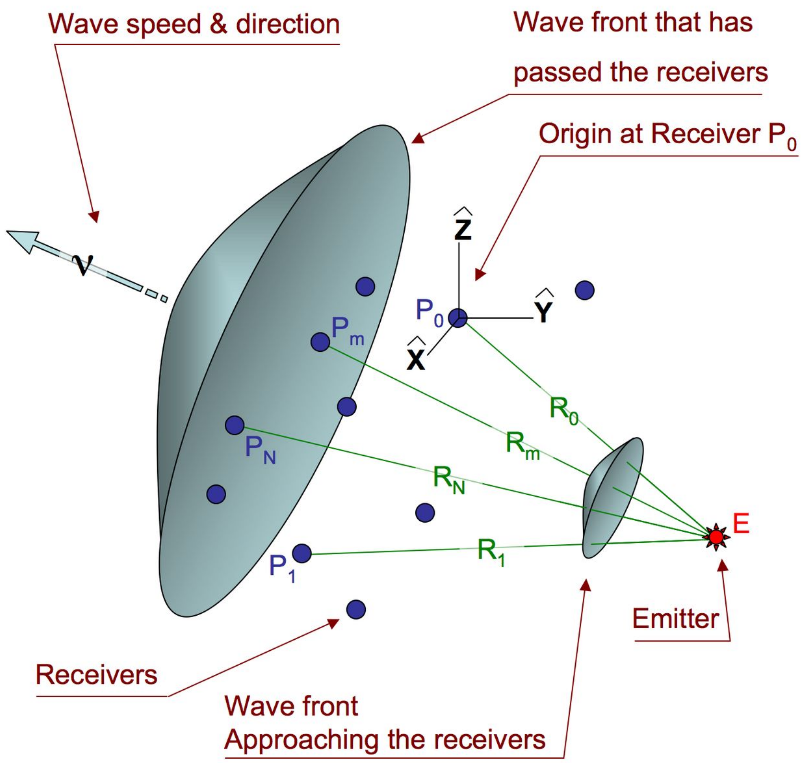

Figure 4-6: Diagram representing 3D hyperboloids [20]

4.5 Machine Learning:

Machine learning is one of the core learning points that we decided to insert into our project. It is vital that we master this aspect of our project because this is the basis of everything that will have to work in every component that we will be releasing for our final product. By definition, Machine Learning is “scientific study of algorithms and statistical models that computer systems use to effectively perform a certain task”. What this is saying that computers will learn stage by stage until it gets used to, most importantly, be as efficient of a system as it possibly can. This process is used worldwide. For instances, it is being utilized for filtering emails, computer vision, where usually it is very inconvenient to be able to learn algorithms properly for performing a specific task that the computer is being asked to do.

One crucial aspect of machine learning is the mathematical optimization which transmits methods or application domains and gets spread all throughout the field. There are subsystems of machine learning. One of which is supervised learning.

Supervised learning consists of a broad library of terms that makeup or explain what machine learning is there to have purpose for. Two types of supervised learning are classification and regression algorithms. For classification algorithms, it is considered as where outputs are restricted to a set of values that vary to be limited. For example, when filtering emails, the input would take the role of the incoming email from one client to another, wherein the output will be the folder where the email is classified into like the Inbox or the Sent Items. Meanwhile, Regression algorithms are named for the continuous outputs it produces. This matters in the fact that may have an arbitrary value within a random range. For example, we measure the temperature, a length of an object, or a price.

Another is unsupervised learning. Unlike supervised learning, unsupervised learning tends to build a model from a set such data that only comprises of inputs whereas the output is not needed. The main purpose of the unsupervised learning is create a structure of a data. It can also discover patterns in a set of data which can group inputs in a set of classification.

Next we have Active learning. These algorithms access outputs which can be desired that is related to a input set that can be limited in size. When used, this type of algorithm is given to a user for labeling. A type of learning algorithm that closely sewed in active learning algorithms are Reinforcement learning algorithms. These algorithms are feedback based in a moving environment that presents feedback in the form of positive and sometimes negative. These algorithms are mainly used for autonomous vehicles or sometimes in playing a game with a human.

Machine learning as a whole is in good relations to optimization or the varying set of ideas that algorithm optimization explains for software developers. This is due to the different kinds of problems arising in creating these machines which can result into loss function to a set of examples. While this is the case, generalization is what separates the two from being in the like. With optimization, it is only concerned mainly to the minimize the loss on a set while machine learning is concerned with minimizing loss on samples that haven't been discovered yet.

Machines are usually modeled as artificial neural networks. Artificial neural networks are the computing of machine systems that are designed and inspired by the neural networks of brains. This is not considered as an algorithm but a framework for multiple and multiple of machine learning algorithms to have the maximum cohesion needed for efficiency and speed in execution. These systems learn by considering unsurmountable examples which is, most of the case, not programmed to specific rules. A neural network is primarily a model based on a set of nodes called "artificial neurons" that is a close representation of neurons in our own brains. An artificial neuron receives a signal and then processes it which signals all the connected artificial neurons near it.

Implementing such processes involve a number which serves as the signal between different artificial neurons in a network. The connections that these neurons are connected into are called edges. All the connections and these neurons require a certain weight which gets more accustomed to the system as it learns more and more. These neurons are aggregated into layers. These different layers will perform different tasks and would transform based on their inputs. Usually, the signals transfer from the first layer all the way to the deepest layer,in consideration that these signals traverse layers more than one time.

Set goals for a artificial neural network was to mainly solve problems in a similar way a human brain would work. It would divert into different things later on and would be used primarily in speech recognition, machine translation, medical diagnosis especially in our technologically advanced society that craves for these helpful tools that can assists in daily duties by the human people.

5.4.1 Convolutional Neural Networks

A convolutional neural network is a series of layers, consisting of an input layer, a series of convolutional and pooling layers, one or more hidden fully connected layers, and an output layer [23]. The convolutional layers organize the data, while a pooling layer simplifies the result. The rough architecture of a convolutional neural network is shown in Figure 4-7.

A convolutional layer works by taking the input only in smaller subsets of data called a receptive field, rather than the whole thing at once. This receptive field filters data through the weights of its inputs. This is like standard convolution filtering, except the filter itself can be adjusted through the optimization of the network.

This receptive field is then scanned over the entire input, reusing the weights that were useful locally on other areas of the data set. Unlike a fully connected layer, there are a lot less neurons and connections to adjust, and repeated or similar weights are reduced significantly. The final output of this receptive field once it has passed over the entire input is called a feature map. A convolutional layer will generate multiple feature maps.

Pooling downsamples the data received in the feature maps generated by merging neighboring cells in blocks. Typically this is either done by taking the average of the data, or the maximum value. This is important as, along with receptive fields, it increases the robustness of the analysis against data translation (or the perspective or change of perspective of the data of the sensor) [23].

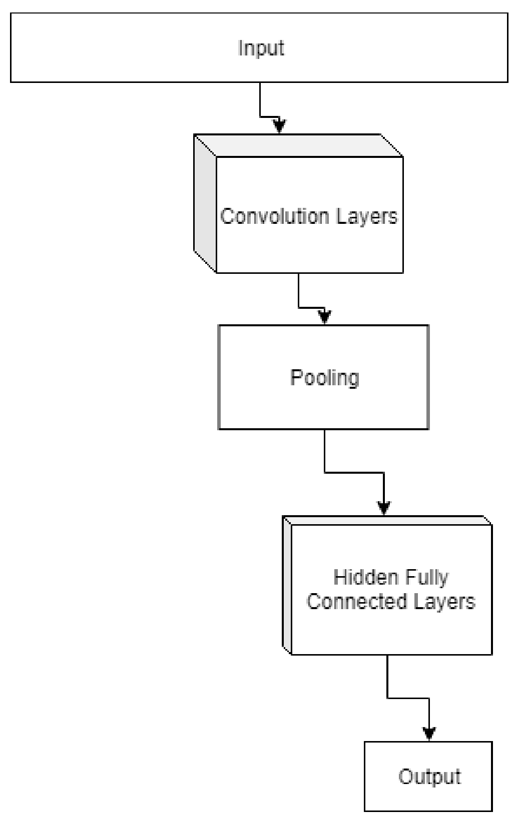

Figure 4-7: A simplified view of a convolutional neural network.

4.5.2 Activation Function

An artificial neuron is a combination of weighted inputs that are summed together and passed through an activation function in order to produce an output. There are a lot of ways of doing this, including step functions and piecewise functions, although these tend to limit the number of potential outputs to just zero or on or a limited function in between. Logistic sigmoid or hyperbolic tangent functions offer a softer output, more variables, and were thus were more commonly used ANNs in the past [23].

However, more recently deep neural networks have started to replace such models with alternatives that offer greater efficiency. One such common alternative is the Rectified Linear Unit (ReLU) and Leaky ReLU activation functions. They offer faster computation and resistance to saturation, unlike sigmoid activation functions do. It offers this simplicity while still avoiding providing insufficient discriminatory abilities, like older step functions. Leaky ReLUs offer greater resistance to "dead zones" where the weights of the inputs will consistently force the output to produce zeroes, by providing a softer positive gradient to negative output instead [23]. A standard and leaky ReLU are shown in two equations below respectively.

$$f(x) = max(0, x)$$
$$f(x) = x\ (if\ x > 0);\ 0.01 * |x|\ (otherwise)$$

4.5.3 Stochastic Gradient Descent

Stochastic Gradient Descent is an algorithm for optimizing a network after it has tested itself against the training batch. The "gradient descent" part explains that the algorithm is attempting to find an error gradient, or slope, then "descend" down that gradient until the model reaches the lowest point.

Stochastic implies something that is randomly determined. In normal Batch Gradient Descent, the optimization takes the error gradient of all samples and sums them together to form a final gradient descent. This can be problematic as it means collecting potentially millions of gradients and then summing them all together. Instead, stochastic takes the gradient of one random example.

4.5.4 Dropout Learning

While deep neural networks have proven to be more efficient than shallow neural networks, especially in classification, there is a notable drawback to this approach. As data is passed through several layers, the output becomes more complicated than it needs to be which creates inaccuracies [23].

Dropout learning is an attempt to address this by randomly eliminating neurons within a layer. These random disruptions (of a predefined probability) keep the network focused by cutting out unnecessary connections [23]. They also make it simpler over time.

4.6 Digital Signal Processing

Our project has to be able to analyze the audio files of potential gunshots detected by the sensors. Specifically, as we have a neural network that will be analyzing the data, we need to be able to present these audio files as graphs or images. This will allow the network to observe the features, and will save resources by allowing use of a computer's GPU.

In a normal format, an audio file presents a signal in the time domain. It is a series of samples taken over time, measuring the amplitude of the audio recorded. The number of these samples is determined by the sample rate, or the number of samples taken per second. While this is useful, especially in determining when a gunshot occurs in the recording, especially with regards to when the N-wave is detected, it is not complete.

More information can be gathered from the frequency representation of the signal. This details how often a particular amplitude is measured over the time recorded. We can get more information by translating it into polar coordinates, featuring a magnitude and phase, which more directly correlates the relationship between the amplitude and frequency. This can inform us on patterns within the signal.

The goal is to use all of these sources, by combining them into a single image. Then it can be passed through a neural network where it will have enough features to use in classifying the recording.

4.6.1 Fourier Transform

The Fourier transform is a method of either obtaining frequency domain information from a time domain source, or time domain information from a frequency domain source. The former is called analysis, while the latter is called synthesis. Since we already have audio files in time domain format, we will mostly be looking into analysis [26].

There are two analysis functions, since the frequency domain is represented with both real and imaginary graphs. That is because frequency is represented as the sum of a real number and an imaginary number.

The real frequency analysis function (depicted as $Real\ of\ X()$) is the sum of the product of the time domain amplitudes times the cosine wave of the frequency index b [26]:

$$Real\ of\ X(a) = \sum_{a=0}^{N-1} x(a) * cos[(2 * \pi * b * a)/N]$$

Likewise, the imaginary frequency analysis function (represented as $Imaginary\ of\ X()$) is the sum of the product of the time domain amplitudes times the sine wave of the frequency index b [26]:

$$Imaginary\ of\ X(b) = -\sum_{a=0}^{N-1} x(a) * sin[(2 * \pi * b * a)/N]$$

Since a frequency cannot be more than half the total number of samples, the frequency index b runs from 0 to half of the total number of samples.

These functions represent the rectangular form of the frequency domain. A cartesian coordinate graph, where the x-axis is frequency and the y-axis is amplitude. We can convert this into polar coordinates. First, we find the magnitude, which is the square-root of the sum of the squares of the real and imaginary analysis functions [26]:

$$Magnitude(b) = \sqrt{ReX(b)^2 + ImX(b)^2}$$

And the phase analysis function is the arctangent of the real analysis function divided by the imaginary analysis function [26]:

$$Phase(b) = arctan[ReX(b)/ImX(b)]$$

A big drawback to using fourier transform is that it relies on the signal being stationary. As in, a certain frequency is expected to be equally present across the entire signal recording. This is not true in reality.

4.6.1.1 Short-time Fourier Transform

A simple approach to this problem is short-time Fourier transform. It divides the signal recording into “windows” of samples, and applies fourier transform to each window instead of across all of the samples at once. This information can then be depicted as an image. The value of each pixel is the amplitude, but the location refers to time in the x-axis and frequency in the y-axis. This allows the frequency information to be depicted over time.

However, this approach produces a drawback of its own in the form of data resolution (the amount of data to work with). The size of these windows affects both the frequency and time resolution. The larger the windows, the higher the frequency resolution but the lower the time resolution and vice versa [25]. This information is still useable, but the window size is fixed, and cannot adjust to the changes throughout the signal, where more frequency or time resolution would be needed.

4.6.2 Wavelet Transform

The wavelet transform is another approach to signal analysis. It attempts to resolve the drawbacks of short-time fourier transform by providing a time-frequency analysis that can adapt based on the resolution needed. If there is a high-frequency part, the window shifts for high time resolution and low frequency resolution. If there is a low-frequency part, the window shifts for low time resolution and high frequency resolution. Thus, this addresses the dynamic/non-stationary signal issue, while also addressing the resolution problem [25]. This is shown in Figure 4-7.

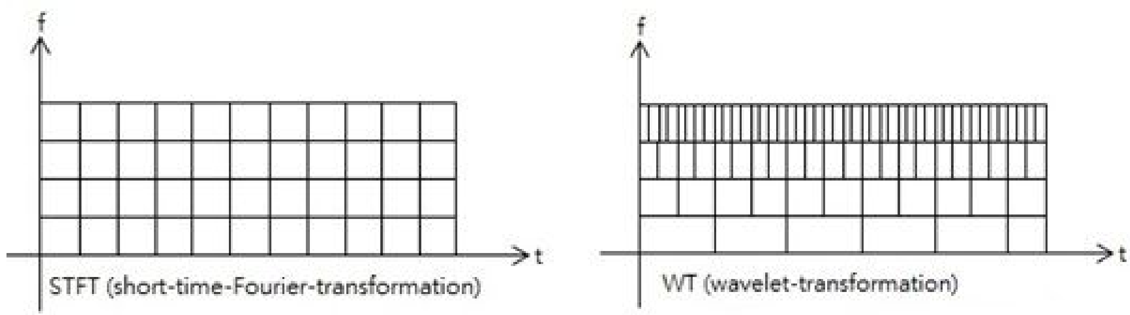

Figure 4-8: An example displaying the differences in implementation of frequency windows over time. Obtained from wikipedia, free to share.

The continuous wavelet transform is a function, $W(a, b)$, with parameters defining the expansion factor, a, and the shift factor, b. It is equal to the integral over negative and positive infinity, of the signal function, $f(t)$, times the complex conjugate of the wavelet sequence $\psi_{a,b}(t)$. This is shown in the following equation [25]:

$$W[a, b; f(t), \psi_{a,b}(t)] = \int_{-\infty}^{+\infty} f(t) * \overline{\psi_{a,b}(t)}\, dt$$

The wavelet sequence is the result of the expansion and shift of the mother wavelet function $\psi(t)$. It is the product of one over the square root of the expansion factor times the mother wavelet function, with parameters of time minus the shift factor divided by the expansion factor. This is shown below [25]:

$$\psi_{a,b}(t) = 1/\sqrt{a} * \psi(\tfrac{t-b}{a})$$

As these are examples of continuous wavelet transform, the formula must be discretized in order to be used in our program. In discrete form, the wavelet sequence changes. Although it is still a result of the shift and expansion factors applied to the mother wavelet function, the process is somewhat different. It is now the product of two to the negative expansion factor divided by two, times the mother wavelet function with parameters two to the negative expansion factor, multiplied by time minus the shift factor. This is shown in the equation below [25]:

$$\psi_{a,b}(t) = 2^{-a/2} * \psi(2^{-a} * t - b)$$

This results in a final wavelet transform analysis function by bringing in the new discretized coefficients. Where the function is now equal to two to the expansion factor divided by two, times the integral of the signal function times the mother wavelet function with parameters two to the negative expansion function, multiplied by time minus the shift factor. This is shown in the equation below [25]:

$$W[a, b; f(t), \psi_{a,b}(t)] = 2^{a/2} \int_{-\infty}^{+\infty} f(t) * \psi(2^{-a} * t - b)\, dt$$

4.7 Database Design:

For software development to work, we decided on the programming languages that would best suit our project and for them to be languages that we already are familiar about. Outside of the usual C programming, Java, or Javascript, C++, there are a lot more convenient programming languages that can entail success for software developers and engineers alike.

One of the programming languages that is thought of by our team at this very moment is using Python for the creation of the database. The purpose of the database is to store all vital information needed to show in recording these timed events when trying to test our final product(prototypes). With this database, it will be the source of the data storage that the sensors may produce like the time of the event, the date in which the test has happened, and the location of the gunshot sounds recorded from the sensors in which they transmit these signals through the bridge into the user interface. With Python and its already ready system that provides valuable resources and benefits in especially creating a database that would be large enough to store these test values and signals. This is in relation to the UI module that will show said information after the test finishes.

Python is an incredible and viable programming language in the world of computer science. It has a huge amount of benefits that surpasses most of the languages used by most developers. With Python, the various number of libraries that it has already implemented in their system, it will be fairly straightforward in the creation a simple database.

Python has its similarities and differences to the other programming languages. There are many disadvantages Python entices to the users but there are also plenty of advantages that levels Python up more than the other languages typically used by software developers.

First we have Java. In comparison to Python, Java programs take much faster to run than the programs developed in Python but they have the benefit of having a faster time to develop. Python programs usually take 3-5 times shorter than Java programs so you
can tell that it is way more applicable to a simple database for our design to use a simpler language in Python. One of the best parts about Python compared to Java is of its powerful and polymorphic list and a array of dictionary types that is also pretty straightforward to handle for even the newest users of this language. Java, on the hand, is much more efficient in executing functions like a simple add function but the main warning for using Java in variable execution is declaring these variables or the machine won't detect the variables setup by the programmer with also the point of no overloading an operator such as the addition in case of the user-defined classes. Python is considered as a better "glue language" while Java is more looked at as a low-level implementation language. Both these programming language actually work properly. What is efficient is that you can actually create components in Java which can be combined to eventually form a program in Python. In Python, we can use the prototype components in building code eventually to be passed to a java implementation so it is more defined and more polished which can ran in vice versa between the two languages that defines unidirectional access. In relation to implementation, the source code in Python can be translated into a Java bytecode which helps the dynamic semantics of a Python program.

Next programming language we can compare with Python is Javascript. The biggest similarities with Python and Javascript is that it is focused on being object-based. Python supports a programming style that prioritizes a simple function that doesn't involve class definitions. Another benefit of Python is that it primarily supports larger programs wherein classes and instance of inheritance play a big role.

The last programming language involved in this design is C++ which is mainly used in the implementation of the triangulation and gunshot recognition algorithms. C++ is a more complicated language in the sense that it requires more attention to detail compared to Python whereas Python is a mere simpler option. Java and C++ are almost identical in requirements and details. If Python is 3-5 times shorter to write compared to Java, it is 5-10 times even shorter to write compared to C++ which says that C++ is complex and needs a higher learning curve but will eventually pay off because of how direct it is in regards to the output. A good analogy for this is that, in average, it takes 3-5 months for a python programmer to finish his/her code whereas a C++ programmer won’t even be able to finish it in a year timeframe.

All in all, it is with definite assurance that Python works so perfectly in the making of a pretty simple aspect of a user interface such as a database. We could’ve used any other programming language like C programming but the nature of Python is approachable compared to the others listed above. There may be changes in this aspect of our project but we most likely will have to stick with Python as our main language in the database modeling.

5. Design

From the research gathered, it is now possible for the Minuteman to be designed. This process will include a multitude of design tasks that must be completed. Each individual will be responsible for their own subject while designing the Minuteman. This will help in the design flow and will allow those individuals to fully understand its application. If each individual is becomes an expert in their own part it will help in integration and testing. With possible failures and troubles later in this program, troubleshooting will become easier if individuals take responsibility for their part.

It is important to maximize the quality of our initial design. Detail shall be taken when designing the Minuteman. Then when the integration of the Minuteman's modules comes, it will have either minor errors or none at all. This is to avoid redesign in the integration section of this project. The responsibility of each individual depends on the designation of their roles. If something fails in the system, then it is the group's responsibility to fix the issue. Like stated above, it is important that the members know their design so they can provide detailed support in the troubleshooting effort.

5.1 Standards

Standards are a means of a providing common guidelines for products and services across an industry, nation, or globally. They are utilized so that new products or services can be introduced into a market in a way that it can be made compatible with other products/services or to keep these products/services safe. Standards can either be self-imposed/regulated, or enforced by law depending on the nation or supranational entity.

5.1.1 802.11 Standards

With 802.11, it is a group of standards that consists wireless techniques that include the same protocols that indicate Layer 2 protocols. The first standard we have is the 802.11a which utilizes a 5 GHz frequency band while the next standards like the IEEE 802.11b and 802.11g use the 2.4 GHz frequency band. The 802.11b and the 802.11g can operate in an unstable and unregulated frequency band, there has a chance of interference or noise from phones and other devices. The IEEE standards only sets specs and does not particularly test certain equipment for compatibility. Just like in Wi-Fi, it can mean 802.11a, 802.11b, or 802.11g, and also does not include the security standard protected access or the WPA or even the WPA2. The Wi-Fi products should note the frequency in which a device runs. The maximum rate of data varies enormously between the different set of standards.

The table of wireless standards that describe each of the choices are shown below.

IEEE standard	**802.11a**	**802.11b**	**802.11g**	**802.11n**	**802.11ac**
Frequency	5 GHz	2.4 GHz	2.4 GHz	2.4/(5) GHz	5 GHz
Data Rate	54 Mbps	11 Mbps	54 Mbps	600 Mbps	1 Gbps
Typical Range(Out)	100 ft	100 ft	125 ft	225 ft	90 ft
Typical Range (In)	400 ft	450 ft	450 ft	825 ft	1000 ft

Table 5-1: Table of Wireless Standards

5.1.2 Bluetooth

Bluetooth is often looked at as a wireless technology used for transmission of files in today's society that uses short-wavelength ultra high frequency waves that consists in the ISM band with a range of 2.4 to 2.485 GHz to purposely transmit over short distances. A bluetooth system manages frequency-hopping spread spectrum that is determined as a radio based technology. With a bluetooth system, data is divided into various tiny packets that are sent to at least one of the different 79 designated Bluetooth channels. Each of the said channels usually has a measurable bandwidth of 1 MHz along with an Adaptive Frequency-Hopping activated, where it typically runs 800 hops/s. With Bluetooth Low Energy uses a frequency of 2 MHz spacing, but instead of 79 bluetooth channels, it only has 40 available. With this Low energy version of Bluetooth, it conserves energy by clustering channels into an arsenal of smaller choices hence the name. Bluetooth grasps a master-slave process and that only one master can communicate with 7 slaves. What the master does is that it directs the clock which usually operates at 313 microsecond intervals.

The following standard table below shows the different standard specifications for both the Classic Bluetooth device and the low energy version of it.

Table 5-2 shows what the disadvantages and advantages of Bluetooth and the Low Energy are. As you can see, the classic version has a higher latency, while the current consumption of the low energy requires less compared to the classic. The range of the classic bluetooth is way greater compared to the low energy version.

	C Bluetooth	Bluetooth Low Energy
Standards	Bluetooth SIG	Bluetooth SIG
Network Standard	IEEE 802.15.1	IEEE 802.15.1
Range	100 m	50 m
Frequency	2.4-2.5 GHz	2.4-2.5 GHz
Over the air	2.1 Mbps	1 Mbps
Throughput	0.7-2.1 Mbps	0.27 Mbps
Latency	100 ms	6 ms
Peak current consumption	< 30mA (varies)	<15 mA

Table 5-2: Table of Bluetooth Standards

5.1.3 Bluetooth SIG standards

To have the bluetooth trademark or to brand a certain bluetooth product, you have to be a member of the Bluetooth Special Interest Group or SIG, in short. What this group does is that they monitor the licensing and the development of the standards of the Bluetooth device. The Bluetooth Qualification Process lists steps that are required to have in the usage or modification of such devices. These steps are the radio qualification step, software qualification step, and the end product. Listed below are the following steps and their respective descriptions that make up the due process for SIG standards:

- RQ (Radio Qualification) - The qualification process starts with the radio qualification which consists of the the process of the radio conforming to the required specifications of the device. What this process entails is that this step is done by requiring a testing facility that tests if the radio specifications is running properly with the scrupulous data of frequencies related to the radio and, at the same time, more in depth testing that are structured hand in hand with the main bluetooth specifications for any version of the developed system.

- SQ (Software Qualification) - These bluetooth devices include stacks and certain protocols that are also required to be gradually tested and a qualified design identification that is eventually issued by the group themselves. Typically, the protocols that are tested for a Bluetooth v4.1 device include GATT, ATT, SMP,L2CAP, RF PHY.
- End Product - The last step for the Qualification process is the listing. A bluetooth product can be listed in the end product that can be found on the bluetooth website. These cost up to $8000 for adopters and $4000 for promoters.

The SIG standard is mainly used for when a group is utilizing trademarks, images, logos, etc that is in relation to Bluetooth. With this, if a certain product is advertised as a compliant, there will be strict following of the qualification process or else trademarks and patents will have to come in play. Violations will be the consequences for these mistakes and could result to legal action later on.

5.1.4 I^{2}C

The I2C or as it's called I squared C, is a "de facto world standard" that is implemented in over 1000 ICs all around the globe that is manufactured by 50 plus companies. The I2C-bus is used in various designs such as a System Management Bus, Power Management Bus, Intelligent Platform Management Interface, Display Data Channel, etc.

When it comes to designing I2C compatible chips, all specifications and proper standards must be met in order to ensure a successful creation of these chips. There should be careful review with the right protocols before making these and always refer to component data sheets.

Wires like the serial date (SDA) and the serial clock (SCL) transmit relative information between devices that are connected to this I2C bus. Every device is noticed by a unique address which can act as a transmitter or a receiver depending on the function. Devices can be applied as masters or slaves when transferring data between devices. The master is the device that initiates a data transfer to the bus which also generate clock signals. The one addressing this transmission is the slave.

The generation of certain clock signals on the bus is assigned to the master devices, the master devices generates its own clock signals whenever it transfers data on the bus itself. The clock signals coming from the bus can be modified when they are stretched by a slow slave that is holding down the line or when another master device when an arbitration occurs.

Table 5-3 shows the features of a I2C bus protocol.

Feature	**Single Master**	**Multi-Master**	**Slave**
Start condition	Mandatory	Mandatory	Mandatory
Stop condition	Mandatory	Mandatory	Mandatory
Acknowledge	Mandatory	Mandatory	Mandatory
Synchronization	Not applicable	Mandatory	Not applicable
Arbitration	Not applicable	Mandatory	Not applicable
Clock Stretching	Optional	Optional	Optional
7-bit slave address	Mandatory	Mandatory	Mandatory
10-bit slave address	Optional	Optional	Optional
General Call Address	Optional	Optional	Optional
Software reset	Optional	Optional	Optional
START byte	Not applicable	Optional	Not applicable
Device ID	Not applicable	Not applicable	Optional

Table 5-3: I2C protocol features

Table 5-3 illustrates the different characteristics of the I2C protocol that is required to be looked at by people who are willing to be implementing these buses into their projects. This table serves as a foundation or mini manual for developers and what the required standards are like for the I2C bus.

5.1.5 C/C++ programming

The goal in terms of programming in C or C++ language is the point of consistency, induce the ease of portability and to reduce the number of errors when it comes to developing and writing code. One of the aspects that follows the standards of C programming is to be able for the user to have a better experience in terms of reading specific lines of code when written by a programmer. Without the clarity in code, the reader will have a harder time in analyzing someone else's source code and would lead in an eventual confusion and maybe the risk of misunderstanding and destroying the flow of a project because of it. There are many priorities pertaining to the development in C/C++ programming. While these two are somewhat different in terms of the language being built, they are similar when it comes to standardization and pure specifications in coding using this language.

First, we have naming conventions. In this aspect of c programming it is considered good practice to name the variables in c programming to something that can be easy to understand. It is also good practice to name the files properly with formal naming so it comes off with clear and concise intentions. The priority is with the owner of the c file to abide the rules of the the naming conventions which can be interpreted by the programmer and/or others as well. It can be seen as very inefficient to traverse through various files without having the trouble of finding the right files you're looking for. The naming of the source code should be highly consistent because it makes the code look cleaner in looks and very easy for people to seek for the right pieces of code needed for the situation.

The next priority standard is commenting. It seems like commenting is the #1 best friend for developers all over the world even with a simpler and more direct language like C. Without comments along the source code, the reader will not be able to understand properly what the owner of the file is trying to convey in the first place. Each of the code blocks in the source code have to be well commented to help out the people in dire need of instruction or knowledge with a certain file. With consistent comments along the source, whoever gets to modify the source will be able to fully understand how the previous user's code is behaving like. Most likely the ones who are in charge of the coding in our group will be the computer engineers, this doesn't stop from adhering to this standard as though it will assist the electrical engineers who are not as experienced with coding as the CpE students are. Having well commented code means that the electrical engineers in the group to assist the already more experienced computer engineering students with the overall scope of the Minuteman system. Comments such as putting a TODO above code that needs work is also essential in a source code. Usually a TODO above a piece of code indicates the name of the programmer, their credentials, and whatever the problem may be at the current time.

The next standard for C programming is formatting. Formatting is extensively essential because without formatting, the code wouldn't look neat at all. Formatting such as well indented code and a well structured code shows a portrayal to be accurate in the whole block scope of the programming code. The variables in the source code has to have the narrowest scope as possible because it is perfectly put at the start of the block of the code. Functions in c programming should be very concise. A functions should not be larger than 40 lines as something bigger than that will be too clustered and hard to understand for the next reader of the code. There has to be an allowance of space, the size of the indentation tabs should be at least two spaces wide. A file of c code should consist of sections separated by a couple of blank lines so it's easier to determine which functions are which. Knowing from the basic knowledge about programming, there is no limit when it comes to the maximum number of lines a code is willing to take up. Although there is no certain limit when it comes to length, files shouldn't be more than 1000 line because these are a pain to deal with. A good practice also deals with the use of constants in uppercase or even names that vary in upper and lower-case alphabet because of the strain to the eye trying to read variables names that way.

The last main standard of C programming is the header. Every C file has an associated header file. The header files have to be self-contained. If a certain header file is not in containment, it has to have a define statement and should include the other headers in the same file. A concrete example of a Define statement is:

```
#ifndef code_h_
#define code_h_
---
#endif //example
```

5.1.6 Java Programming

Outside of other programming languages that are being used around the world, we can consider the fact that some languages are easier to understand than the others. While others can be more direct, like doing Python or C programming, we have a more object oriented approach in JAVA. Java is known for having to set classes and variables that are considered to be related to one another by the sense of Inheritance and hierarchy that manages all java code to be glued to each other in terms of all the different imports and definitions of classes determining if they're the parent class or the child class along with knowing if they are abstract or not. There are various standards listed over the internet that show how programmers can use java programming properly without any hassle to any future users.

The first coding standard for Java is the standards of components. There's a way to write components name in java and its by the purpose. When it comes to writing java code, it is required to know how structure the code properly so as it is easier for the next users to get the handle of off it smoothly. In this way, this improves the clarity and the maintainability of the code.

The next coding standard is the standard for classes. With classes, their names and things related to variables have to be nouns that are always used with an uppercase letter. These class names can never be started with lowercase because variables typically start with lowercase letters and it will be confusing for others because they might mistake it for the class names and eventually lose track of their progress. Some of examples of good coding practice with class names are like “StringBuffer”, “Apple”, “Cat”, etc. It is required to know this practice very well so it also accounts for the programmer to know what is right and what is wrong.

Following, we have the coding standard for Interfaces. The interface names should be an adjective related to Java programming that also starts with an uppercase letter. It only should be one word so having multiple words as a name for interfaces is prohibited. Names such as ‘Runnable, Serializable, and Comparable’ work for this kind of situation. Usually these words end with the suffix ‘ble’ to fully express the necessary intentions when structuring interfaces. This is absolutely not gonna work when someone tries to go around it because it will eventually run as an error using an implemented IDE debugger when you run the code.

Next we have the standards for Methods. Methods necessarily start with a lowercase letter this time along with required parentheses next to it. If the methods contain multiple words, it is needed that the next word has to start with an uppercase instead so it doesn’t look like one big chunk of a word and the names of the methods are easily readable.

5.1.7 USB 2.0

5.1.7.1 USB Electrical, Mechanical and Environmental Compliance Standards

The USB is one the most prominently used devices in the computer world nowadays being used for data storage by people connecting them to their laptops or desktops to be able to store valuable information that they can bring around wherever they want to in a small holdable device that they can put in their pockets for convenience. Like all produced apparatus in the technological society today, there has to be implemented standards that pave way to how a device like the USB is supposed to be worked on and how the safety procedures relay in information for the users to fully capture the essence of this instrument wherein there will be no mere consequences that arise when these things are used by the consumers. The following table below illustrates the different criterias in determining the standards for USB cables, how assemble these cables and the connectors that are connected in the sources.

Test Description	**Test Procedure**	**Performance Required**
Visual and Dimensional Inspection	Visual,dimensional and functional inspection in accordance with the USB quality inspection plans	Must Meet or exceed the requirements specified by the most current version of the USB specs
Insulation Resistance	This test procedure is to detail a standard method to assess the insulation resistance of USB connectors. To determine the resistance offered by the insulation materials and various seals of a connector to a DC potential	1000 MegOhms minimum
Dielectric Withstanding Voltage	Detail a test method to prove that a USB connector can operate safely at its rated voltage and withstand momentary over-potentials due to switching, surges	The Dielectric must withstand 500 V AC for one minute at sea level
Random Vibration	A USB connector has to function during vibration or to survive conditions that of vibration should be clearly stated by the detailed product	No discontinuities of 1 microseconds or longer duration when mated USB connectors are subjected to 5.35 Gs RMS.

	specifications.	
Thermal Shock	To determine the resistance of a USB connector to exposure at extremes of high and low temperatures and to the shock of alternate exposures to these extremes, simulation worst case conditions.	10 cycles -55 C and +85 C. The USB connectors under test must be mated
Solderability	To detail the uniform test method for determining USB connector solderability. The test procedure contained utilizes the solder dip technique.	USB contact solder tails must pass 95% coverage after one hour steam aging as specified in Category 2.

Table 5-4 : Standards for USB 2.0

Table 5-4 displays some of the important safety measures laid out by the people who created Universal Serial Bus'. If it wasn't for these standards, there will be major consequences in the end of it all. No one knows how severe these complications will be and that's why there should be applicable solutions to each and every part of the traits a USB possesses. There are many more standards to apply to USB's but we have implemented some of the really important and most well-noted ones for our report.

5.1.8 American Wall Outlets

In the world, there are many different kinds of outlets that show users where they can connect their devices into to gain power such as appliances like the television, stoves, smartphone chargers, etc. Without these sockets/outlets there wouldn't be any source of power to make these tools work. There are numerous amount of outlets that are offered all around the world which offer a unique design and structure for each of them. Some of the different outlets offer 2 pins while there are sockets in which there are 3. As for this project, we will only focus on the american made outlets because that is what we will be using to provide power for our Minuteman system which enable our audio sensors as a whole and power up all the necessary inputs to be able for our project to work properly. The the standards for these outlets are the following:

TYPE A	TYPE B
Mainly used in the USA, Mexico, Canada, Japan	Mainly used in USA, Mexico, Canada, Japan
2 pins	3 pins
Not grounded	grounded
15 amps	15 amps
Almost always 100 to 127 V	Almost always 100 to 127 V
Socket compatible with plug type A	Socket compatible to plug type A and B

Table 5-5 : Standards for American Outlets

Table 5-5 above illustrates a few of the details that incorporate outlets in america in specific. To go beyond further what these offer, Type A plugs and outlets are as known as NEMA 1-15. The Type A plug has two flat 1.5 mm thick blades, which are polarized and can be only inserted in a particular way because the two prongs don't have the same shape. The blades are neutral to 7.9 mm wide. In contrast, Type B systems are designated as standard NEMA 5-15. Like the Type A, it also has 1.5 mm thick blades wherein the device is grounded before it even it connected. In addition, it has a 4.8 mm diameter round or u-shaped pin. Both Type A and B plugs are not insulated.

5.1.8.1 Electrical Safety Standards

For our project, it is critically required for our system to be safe for use by others later on even after the team graduates so that we have a good reputation from numerous consumers out there. Safety mechanisms such as enclosures for systems that can prevent any electrical contacts from being shorted and such. Other important Occupational Safety & Health Administration (OHSA) standards when it comes to safety for specific electrical related subsystems in terms of interactions with numerous users serve as the following:

- Electrical Power Generation, Distribution - what this piece of informations relays is that it regulates operation and the keen care for electrical power generation, control, transmission, and distribution lines.
- Electrical Protective Devices - This regulates the design requirements in relation to different types of equipment for electrical systems that include such as rubber insulating covers, rubber insulating gloves, and sleeves

which have to meet the certain project requirements listed in by the team in this report

- Wiring Design and the means of Protection - This standard illustrates the labeling of device wiring and the aspect of protection in the design of the Minuteman system. This also allows proper grounding terminal connections are perfectly labeled all throughout the project.
- Hazardous locations - This standard relays in requirements for hazard equipment that are particularly classified whether it may be flammable vapors, liquids, gasses, combustible dusts and fibers that may show the possibility of a combustible concentration quantity
- Use of equipment - regulates the use of plug and cords connected in major equipment used for the project. This includes flexible cord like extension cords.
- Personal Protection - regulates the use of personal equipment needed for ample safety in case there is a troublesome event that may happen during the design stages of the project. Anyone working possible electrical hazards have to be very careful in which parts can go with others and not potentially short circuit and burn needed parts. Everyone in the team has to use protective equipment that is appropriate for parts of the body that needs to be protected.

In summary, safety has to be the number one priority for our team from start to finish. We cannot let ourselves get too loose with the equipment that we are working with because someone can easily just accidentally burn or destroy a part that may be too expensive for student funds that will be hard to replace later on. It is also imperative that we look out for each other in case something unfortunate happens during the implementation stage that may affect our states mentally and physically. This only cannot harm our education as a whole but it can also potentially destroy our self being and put our bodies in harm as well. Dealing with these kinds of tools are no joke. That’s why only experienced professionals usually handle these devices. There will be strict embodiment of rules between the team to always make sure every piece of equipment works properly and that nothing is connected incorrectly.

5.2 Hardware

The design of the hardware in the project will have constraints from the cost and the components available for the project. The parts can be exchanged depending on further research done in the project. There is careful consideration and debate where the group understood each person’s opinions on what the right parts to get will be. There is mere possibility that added devices will be implemented later on because of the complex nature of the Minuteman project. Table 5-6 further shows the features and specifications of the hardware.

Feature	Hardware Specifications
Development Board	Arduino Due
Microcontroller	AT91SAM3X8E
Power Adaptor	12V DC 1 Amp Power Adapter Supply
Switching Regulator	Analog Device LTM4622A Dual Ultrathin 2A or Single 4A Step-Down DC/DC μModule Regulator
Voltage Converter	MAX1044 CMOS Switched-Capacitor Voltage Converters
Microphone	CMC-6035-130T, CUI, Mic Omni-Directional 2200Ohm -42dB 2VDC Round Solder Pad
Gyroscope, Accelerometer, Geomagnetic Sensor	BNO055 9 DOF Absolute Orientation IMU Fusion Breakout Board
Global Positioning System (GPS)	NEO-6 u-blox 6 GPS Modules
Temperature Probe	Maxim Integrated DS18B20 Programmable Resolution 1-Wire Digital Thermometer
Amplifier	Texas Instruments TL072

Table 5-6: Hardware Specifications

5.2.1 Central Module

The Central Modules purpose is to be the connector between the Sensor Module and the software program on the computer. In the initial design of the Minuteman, it was planned to use triangulation. There would be two sensor modules with 5 microphones each, both connecting to the Central Module. This changed when multilateration became feasible. With this development the Central Module and Sensor Module could be compartmentalized into one hub.

This type of compartmentalization could provide a high rate of reliability, since large functions of the Minuteman were modulated.

Due to the new development of using multilateration, the Central Module will house almost all of the hardware except for the microphones. These microphones will be spread out from the Central Module. This means the central housing unit must be large enough for our design to be able to contain the PCB and assortment of sensors. To be able to fully contain everything while being waterproof, this device must be made of plastic with large enough dimensions to fit everything inside. The Central Module will be a rectangular box with a enough room for a sizeable PCB. The module will had input and output port for the microphones, the USB, and the wall adapting power supply.

The Central Module contains the power supply with either the Microcontroller Unite (MCU) or the Field Programmable Gate Array (FPGA) depending on which is used. The power supply inside the Central Module will power the Central Module and the Sensor Module. The Central Module will use analog to digital converters (ADC) with a culmination of other functions to provide accurate digital signal representation of the audio sound recording.

5.2.2 Microcontroller

The Microcontroller unit (MCU) is a less sophisticated computer. It usually contains RAM, Program Memory, Microprocessor, Analog to Digital Converter, Input/Output Pins, and a clock generator. All these function are combined on a integrated circuit in various ways to make many different microcontrollers. The microcontroller is a very compact device able to electrically manipulate analog and digital informations. These devices all have different uses depending on their configuration/design. To understand the use of these devices you must read the datasheet and know which configuration you need. For this project the group researched to pick the best microcontroller that would fit our requirements.

5.2.3 Development board

The boards below are a selection of development boards. These development boards have microcontrollers integrated inside of them. With other functions integrated within those microcontrollers. The development boards make it easier for designing, because there is a large support from the development board designers. One of these boards will be used for easy integration for our project. This will also help in the testing and design. When it is required to create our own PCB it will be easy because, whatever development board that was chosen, the microcontroller that is inside that development board will be used in the Printed Circuit Board (PCB).

5.2.3.1 Arduino Due

The Arduino series of boards are one of the most favored of development boards. One such reason is the integrated development environment (IDE) that Arduino has. Their IDE is very similar to the C language and so it is very easy to pick up if you have previous coding experience, including having learned the syntax and implementation. During the initial glimpse, this board was very impressive with a 3.3VDC operating voltage and 54 Digital Input/Output Pins and 12 Pins with ADC capabilities providing PWM output. This is exactly what was needed for the design. Though some of the specification may not be needed, they might be used later on if the initial design is expanded upon later in the development of this project. This board is also fiscally responsible, costing $38.50 and including the development information that Aduino provides.

Microcontroller	AT91SAM3X8E
Operating Voltage	3.3VDC
Input Voltage (recommended)	7VDC to 12VDC
Digital Input/Output Pins	42
Digital Input/Output Pins capable of Pulse Width Modulation	12
Analog Input Pins	12
Analog Output Pins using a Digital to Analog Converter (DAC)	2
Flash Memory	512 KB
SRAM	96 KB
Clock speed	84 MHZ
Universal Asynchronous Receiver/Transmitter (UART) communication	Yes
USB Protocol	Yes
Reset Protocol	Yes
Overvoltage Protection	Yes

Overcurrent Protection	Yes

Table 5-7: ARDUINO DUE Ideal Specifications

5.2.3.2 ARDUINO MEGA 2560 REV3

Similar to the Arduino Due, the Arduino MEGA 2650 REV3 is a capable board for this project. During the initial inspection, it has no issue with the amount of Input/Output Pins needed. This board provides extra protection as well with more pins for PWM. These two board are very similar and contain all necessary aspects for this project. This is a comparable board to the Arduino Due. The Arduino MEGA 2650 trades off some hardware for different configurations like more coding pins, but less memory for coding compared to the Arduino Due. This board also has a smaller clock speed than the Arduino Due. Costing the same as the Arduino Due at $38.50, the Arduino MEGA 2650 REV3 is a different board with similar capabilities with minor changes for different applications. Just like the Arduino Due, the Arduino MEGA 2650 REV3 comes with extra hardware that might not be needed. This hardware could be used later on if extra or unseen qualities are added to the project.

Microcontroller	ATmega2560
Operating Voltage	5VDC
Input Voltage (recommended)	7VDC to 12VDC
Digital Input/Output Pins	38
Digital Input/Output Pins capable of Pulse Width Modulation	16
Analog Input Pins	16
Flash Memory	256 KB
SRAM	8 KB
Clock speed	16 MHZ
Universal Asynchronous Receiver/Transmitter (UART) communication	Yes
USB Protocol	Yes

Reset Protocol	Yes
Automatic Reset (Software capable to reset Arduino MEGA in code)	Yes
Overvoltage Protection	Yes
Overcurrent Protection	Yes

Table 5-8: ARDUINO MEGA 2560 REV3 Ideal Specifications

ARDUINO UNO REV3

Looking at the Arduino Uno REV3 it is noticeable the difference in its capabilities. The Arduino Uno has a noticeable drop in hardware capabilities. With only 14 Input/Output Pins, this board has considerably less capable than the previous two. The hardware in this board is half the specs of the Arduino MEGA and Arduino Due. It does not have enough Pins for this project and would require multiple boards if used. Each individual sensor module would need its own board. Though this board is less capable, its price is cheaper than both the Arduino Due and Arduino MEGA costing $22.00, a little over half the cost of the other two boards.

Microcontroller	ATmega328P
Operating Voltage	5VDC
Input Voltage (recommended)	7VDC to 12VDC
Digital Input/Output Pins	8
Digital Input/Output Pins capable of Pulse Width Modulation	6
Analog Input Pins	6
Flash Memory	32 KB
SRAM	2 KB
Clock speed	16 MHZ
Universal Asynchronous Receiver/Transmitter (UART) communication	Yes

USB Protocol	Yes
Reset Protocol	Yes
Overcurrent Protection	Yes

Table 5-9: ARDUINO UNO REV3 REV3 Ideal Specifications

5.2.3.3 BeagleBone Black

The BeagleBone Black is just or more capable than any of the other Arduino boards. Beagleboard gives out a large amount of information about their project, which helps in our development process. This is the same as the Arduino boards. The BeagleBone Black might be more than what is needed for this project. With 92 Input/Output Pins this board is not only able to do the job required, it might be to much even if the project were to be added on to. With having a AM335x 1GHz ARM Cortex-A8 processor. This board is made for a more complex project with video and other functions. When comparing its price, it is cost effective for what you get in functionality, but it is much more than what this project requires. Costing $62.38, this could be a great test board for our project.

Microprocessor	Sitara AM3358BZCZ100
512MB DRR3L	512MB DRR3L
Operating Voltage	1.5VDC, 3.3VDC, and 5VDC
Video Out HDMI (Graphic capability)	Video Out HDMI (Graphic capability)
Digital Input/Output Pins	92
Flash Memory	2 GB
Clock speed	24.576 MHZ
Universal Asynchronous Receiver/Transmitter (UART) communication	Yes
USB Protocol	Yes
Reset Protocol	Yes
Overcurrent Protection	Yes

Reliability	Yes

Table 5-10: BeagleBone Black Ideal Specifications

5.2.3.4 Field Programmable Gate Array (FPGA)

The Field Programmable Gate Array (FPGA) is a simple product to understand, but a complex device to implement. The reason why it is easy to understand is that it is a device with no hardware until it is coded. This mean that the FPGA has no physical hardware until someone codes the hardware onto it. If you look at (Figure 5-1) you can see the top down representation of what a FPGA looks like.

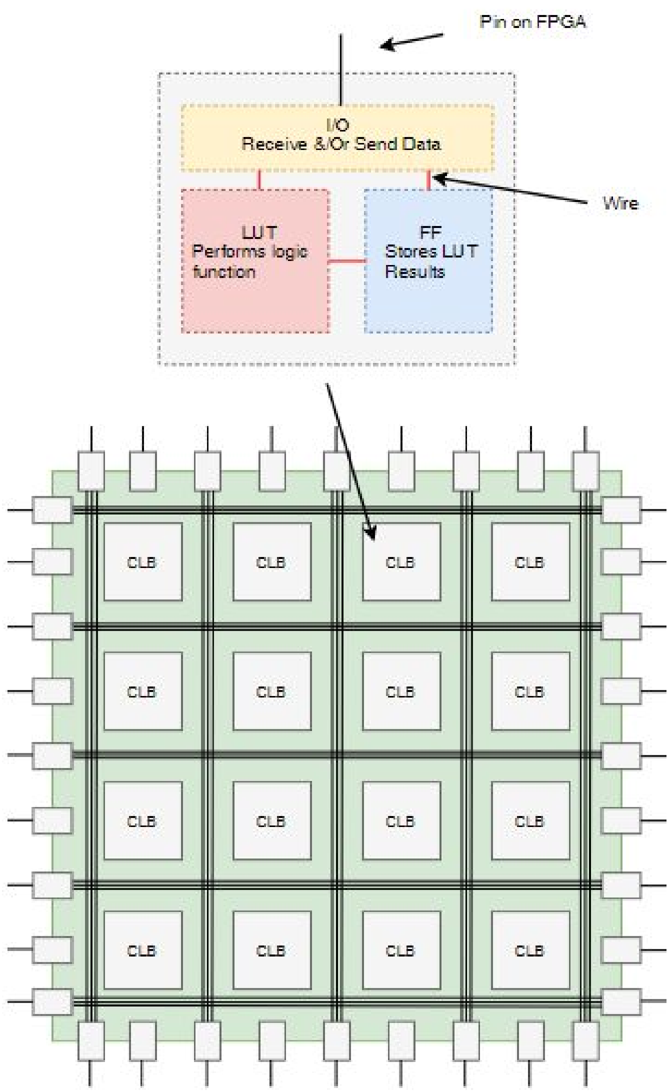

Figure 5-1: FPGA Diagram

The board is an array of blocks that are coded to form many different tasks. These blocks are called Configure Logic Blocks (CLB). Each one of these blocks has a assortment of internal components. The internals of these block are what is programmed. The CLB can perform Analog to Digital Conversion, Digital Signal Processing, and other assortments of logic configurations. The Internal of the CLB are also present on (Figure 5-1). The LookUp Table (LUT) is a truth table where for every set of inputs you have outputs. The LUT is what is programmed to have different functions and results. Those results are sent to the Flip Flops to be stored. The stored value are either transferred to other CLB or sent to an output pin at the borders of the FPGA diagram.

These devices are very powerful and can come in a large range of configurable specs. With the more complex FPGA needing more Input/Output Pins. This plays a large part of how much the FPGA will cost and how much it is able to do. Because the FPGA is programmable hardware, it is able to reconfigure many times. This mean in one configuration a group of CLB could be an ADC and another configuration it could work as a switch. That is why it has 'Field Programmable' at the start of it name. Some provide a limited amount of reconfigurations and others provide an unlimited amount of reconfigurations.

The FPGA has a steep learning curve, even with enough sources online. It requires you not only to understand the hardware you are creating, but you must also understand the process of creating it as well. The user is required to learn one of the Hardware Descriptive Languages. The two most common languages are VHDL and SystemVerilog. For both languages, they have a process of designing and creating the hardware in the system as well as a step for verification and testing. These two section are broken up into design and verification. For this project, SystemVerilog is being studied for application.

Just like the Microcontroller the choice of FPGA are depended on our need and what the complexity its job will be. The best choice will be dependent on the need for the project and complexity that the designer wants.

5.2.3.5 Mpressions Odyssey MAX 10 Evaluation Kit

The choice of using a MAX 10 Intel FPGA was because of the easy of access to the development application provided by Intel. In this project using, a Evaluation Kit for the FPGA integration would save time and reduce complexity. The MAX 10 Intel FPGA was the a deciding factor when looking at different boards, with a large amount of user support from Intel that will help in the integration part of this project.

This board is capable of providing a lot of utility with an audio monitor able to connect straight to a microphone to process the data. The Mpressions Odyssey is also able to connect with the Arduino boards for integration. The Mpression has an extensive user guide as well.

Microcontroller	EFM32 (USB Device)
FPGA	Intel MAX 10
Arduino Compatibility	Yes
Accelerometer	Yes
Light Sensor	Yes
Temp/Humidity Sensor	Yes
Audio Input	Yes
Operating Voltage	3.0VDC
Digital Input/Output Pins	14
SRAM	512 KB
Flash Memory	2 MB
Clock speed	50 MHZ
USB Protocol	Yes
Bluetooth Protocol	Yes
Overcurrent Protection	Yes
Reliability (Test Ports)	Yes

Table 5-11: Mpressions Odyssey MAX 10 Evaluation Kit Ideal Specifications

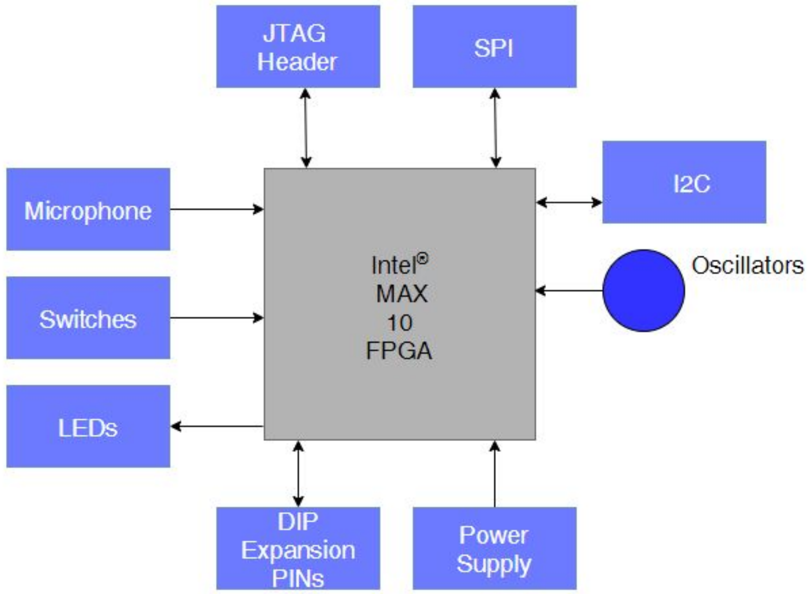

Figure 5-2: Mpressions Odyssey MAX 10 Evaluation Kit Diagram

5.2.3.6 MAX® 10 FPGA Evaluation Kit

The MAX 10 FPGA Evaluation Kit is a board made by Intel using their MAX 10 FPGA. This kit was made so that the FPGA could be tested with many different application. Intel provides an extensive amount of information in the user guide with videos and different design application for this board.

When comparing Mpressions Odyssey to this board, the MAX 10 FPGA Evaluation Kit has more connection ports. It allows for the user to manipulate it better, since it has less already integrated hardware. This will be better for our project since in these stages it might change in the design. The Mpressions Odyssey is initially more capable than the Intel Kit, because it has hardware already interfaced to it. While the intel kit has very little, this means that the Intel Kit is more flexibility for new hardware integration.

FPGA	Intel MAX 10
Arduino UNO Rev3 Compatibility	Yes
Potential Potentiometer	Potential Potentiometer
Operating Voltage	5VDC
Digital Input/Output Pins	40

Arduino Input/Output Pins	14
Analog Input/Output Pins	8
Flash Memory	172 KB
Clock speed	50 MHZ
Bucked Power to Filtered	3.3VDC
Mini USB Protocol	Yes
USB Protocol	Yes
Reset Protocol	Yes
Overcurrent Protection	Yes
Reliability (Test Ports)	Yes

Table 5-12: MAX® 10 FPGA Evaluation Kit Ideal Specifications

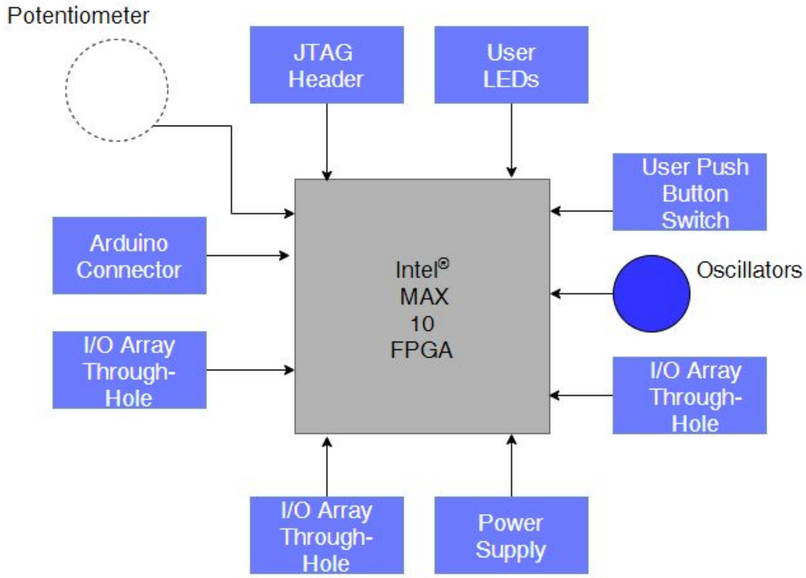

Figure 5-3: MAX® 10 FPGA Evaluation Kit Diagram

The results of the search for the right board has come to a decision. One is to take the Arduino Due by itself to do the majority of the signal analysis. Other designs were to take the Arduino UNO REV3 and combine that with the MAX® 10 FPGA Evaluation Kit. Both these decisions do not change the results of the power supply, but it does change a lot of the hardware design and how the modules work. The choice of this project depends on the requirements of the sensors and the function that the development board must complete. The Arduino Due is the best development board for this project. This also means that with later development of our PCB the AT91SAM3X8E microcontroller will be used for the project.

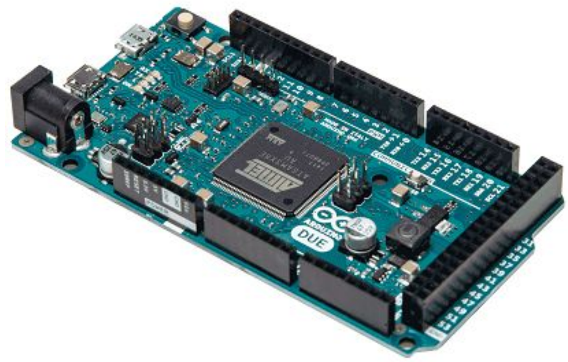

Figure 5-4: ARDUINO DUE [34]

5.2.3.7 AT91SAM3X8E

The AT91SAM3X8E microcontroller designed by Microchip Technology was chosen from the Arduino Due. The AT91SAM3X8E microcontroller shall provide the best results of the development boards that were screened. One of the main functions that was required for our design was a ADC with 12 bits or great resolution with at least 5 channels. This would allow our microphones to operate properly. The microcontroller also has USB 2.0 protocol so it is able to bidirectional interface with the computer easily. This will help with our hardware to software integration.

Processor	ARM Cortex-M3
Operating Voltage	3.3VDC & 1.8VDC (self regulated)
Voltage Regulator	1.8VDC
Programmed input/output (PIOs)	103
Flash Memory	256 to 512 KB
SRAM	32 to 100 KB
ROM	16 KB (bootloader routine/startup)

Clock speed	84 MHZ
ADC	12-bit, 16 channel
DAC	12-bit, 2 channel
Timer	32-bit, 9 channel
I2C	2
Serial Peripheral Interface (SPIs)	6
USART/UART	3/2
Low-Power Mode	2.5 μA
Power-on Reset (PoR)	Yes
USB Protocol	Yes
Ethernet Protocol	Yes
HSMCI Interface	Yes
Dimensions (L x W x H) mm | in	22 x 22 x 1.4 | 0.866 x 0.866 x 0.055

Table 5-13: AT91SAM3X8E Ideal Specifications

With the size of this device at 484 mm^2 it is the largest device that will be on the PCB. The placement of this device will be important because it take up so much space. When designing the PCB this microcontroller should be place in the most optimal station so the traces will be short and it can be sectioned for functional purpose. Inside of the PCB I2C interfacing will be used for communication between the ICs. The USB will provide communication between the Central Module and the computer to interface with the software application.

5.2.4 Printed Circuit Board

The Printed Circuit Board (PCB) was to be designed in the Central Module of this project. The hardware requirements that the Minuteman needs to function will determine the size of the PCB itself. With the initial change from triangulation to multilateration the size of the PCB has increased. With more components being added to the PCB, it is a beneficial to separate the PCB into functions. Just like how the Minuteman was designed with modules, it would be an asset to the project design if the PCB was sectioned as well. The three separate sections are audio & sensors, power, and microcontroller. This will allow for easier assurance testing and development of design. When designing a PCB, there are certain standards and choices that need to be made. To help with the design, a simplified diagram was created.

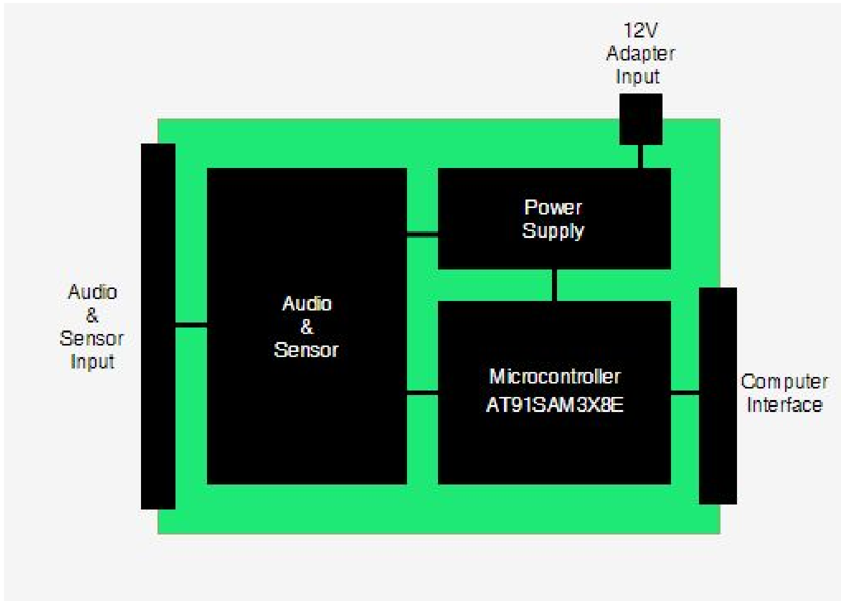

Figure 5-5: PCB diagram

The diagram above (Figure 5-5), shows a simplified representation of how our PCB will be designed. The configurations chosen for our design is the Star Configuration to minimize the loss of voltage from the power section. This will help optimize our design, as well as to section our test points. These test point will provide the capability for the PCB to be externally tested for the initial accuracy and troubleshooting.

Depending on the complexity of our PCB, there may be a need to be more than a single layer. For example, if it requires 2 layers then it will have a top and a bottom layer like what is shown below in Figure 5-6.

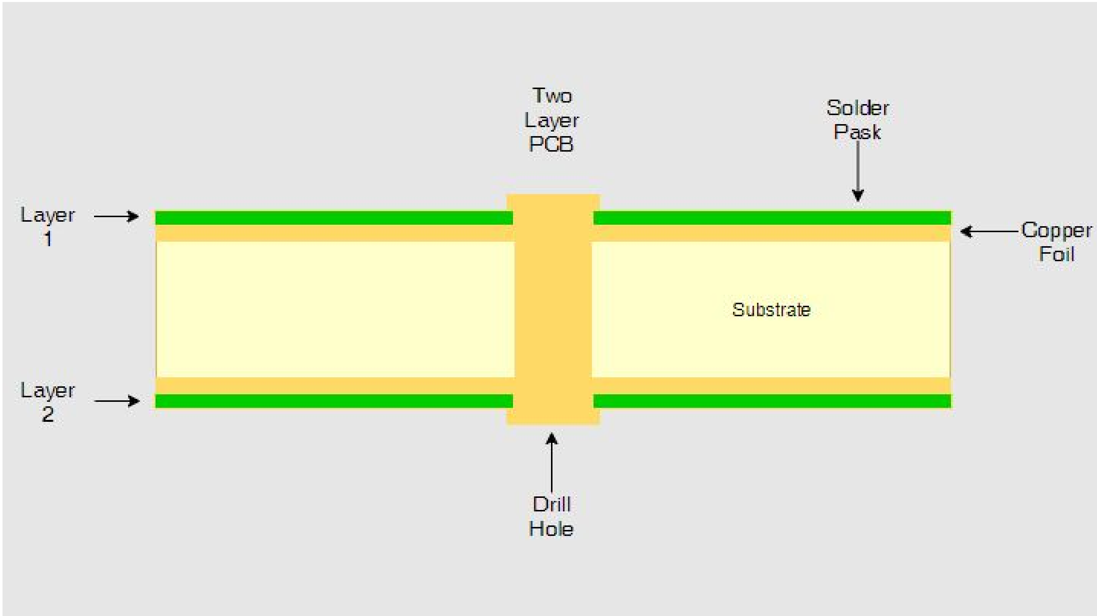

Figure 5-6: PCB Two Layer Diagram

Both of these layers would be separated if it did not have the Drill Hole. The Drill Hole provides a connection between both layer 1 and 2. With more layers added, if fills in where the substrate is located. The amount of layers can depend on the PCB size and the trace width. It will be cost beneficial to limit the layers to two, so that it will not increase the manufacturing cost of the PCB to much. Like different aspect of the circuit design, there are several options that can decrease the overall cost.

Another quality that will limit our room in the PCB will be the trace width. This is the width of the copper interconnects between the IC. This is comparable to a wire in non-PCB form design. Just like in non-PCB design, wires are able to sustain a certain amount of current depending on the material, diameter and purity. In the PCB the trace width is dependent on the current going through the layer and weight of the copper. One of the biggest reasons why it is important to prevent too much current going through the layer, is that the traces could heat up to much a melt. So for preemptive measures a online calculator will be used for modeling the trace width. There are many sources out there and the one chosen for this project was ADVANCED CIRCUITS 4PCB Trace Width Calculator. This is the most reliable prediction model.

When reading the Eagle guide and other sources, it is important to have short traces between the ICs in the PCB. This is for resistive loss, where external thermal temperature affect the traces. This is a big factor when setting up the PCB. The locations of each part need to be maximized so thermal dispersion can also occur in our PCB. If internal as well as external temperatures in the PCB exceed the limit of any integrated circuit then catastrophic failure shall occur. The section that will need to be most concerned about thermoregulation is the power supply, because this section will have a large amount of current consumption within a condensed area. It might be good for internal protection to have extra thermal regulation with thermistor resistors or spread out a certain to isolate it from other section. Another reason to have short traces was for thermal noise, but this can be mitigated if necessary.

With these cautions in mind, it will be important to keep other standards like taking 45° angles edges show below in (Figure 5-7).

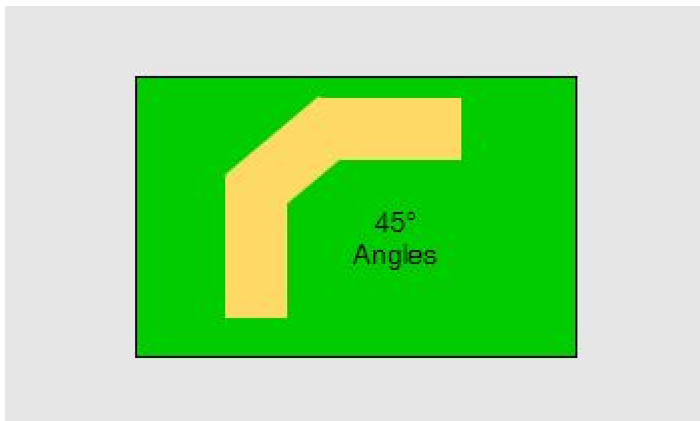

Figure 5-7: PCB 45° angles

The reason this is done with copper tracing, is a 45° angles is shorter than and 90° angles and interference with high speed logic. The reflections in high speed logic can cause a small amount of capacitance to occur, and this could affect our PCB design if precaution is not taken.

If enough room permits, it will be important to add a ground plane. Since our PCB will be 2 layers it could be possible if enough room is on the first layer to have a grounding plane connected to the second layer. This will can boost our signal integrity and will make our circuit more resistant to electromagnetic interference. The ground plane can also help shorten traces and prevent ground loops by providing easier direct connections from traces and drill holes.

To design the Minuteman PCB it was decided to use the EasyEDA software. This program was chosen over the EAGLE software because of the application interface. Just like NI Multisim and LTspice the interface and software of EasyEDA seemed to have a easier interface.
(design of PCB)

The ability to test and troubleshoot the PCB will be important after it is created. If any issues arise it will be too complicated to try and test the PCB without certain PCB design requirements. If precaution is take when initially designing the PCB it will be important to create test points. These test point could shorten any time reworking the PCB later on. Creating the test points would allow for easy access to section on the board. The sectioned circuit blocks shall provide accurate testing procedures that will give predictable results. Using test points take room on our board. There is a balance with how many test points should be on the board. It will be important to maximize the space that the test ports take and their location on the PCB. It would not be wise to over lose to much space for redundant test.

Space on the PCB is very important for this project. The more space needed and the more component required can increase the price drastically. It will be important to compare prices and manufacturing time of different suppliers. These choices will be further researched at a later period, but before the finished PCB design.

5.2.5 Power Supply

The design of a power supply was base of the requirements that the circuit need to operate. This process took time to research. One could take the power supply design and use a simple regulator, but for this project we want to pay attention to the development of power supply. The use of Texas Instruments Webench Power Designer program was initially used to help in the development of our board. It helped simulate the different power sources. In this project we will compare the use of different sources and the power consumption that we will need.

When the design was considered for this project it was to take the American standard 120VAC at 60 Hz outlet power to a 12VDC adapters. The 120VAC 60 Hz would convert from a wall outlet to a filtered 12VDC. This will be our supply voltage for the DC to DC buck down voltage regulator. This would create a voltage that would power our sensor module. In the design there will be one sensor module and each sensor module will hold 5-8 microphones. It was essential to ensure that the voltage delivered was consistent and with limited noise for good reliability. Through overvoltage and overcurrent sensing, our project can meet that goal. So in the search for the best applicable power design it is important for our integrated circuit to have certain reliability function other than just an regulated voltage.

The actual circuit design for this project will be done on LTspice. Using this program will help us in development, since it was developed by Analog Devices.

This means that LTspice already has the many related integrated circuits that we may be simulating in the software.

5.2.5.1 Power Requirements

The power requirement are strict and dependent on the load of the circuit design. Each individual integrated circuit will consume a different amount of current and need a specific voltage to work. If these values are not provided then failure will occur in the system. The to meet the requirement it must first be understood what is required. Table 5-14 below is the power requirements for the system.

Component/Function	Operating Voltage	Current Consumption	Quantity
Arduino Due	12VDC	800 mA	1
AT91SAM3X8E	3.3VDC & 1.8VDC	200 mA	1
CMC-6035-130T	3.3VDC	0.5 mA	5 - 8
DFR0198	3.3VDC	0.7 mA	1
BNO055 9 DOF Absolute Orientation IMU Fusion Breakout Board	3.3VDC	12.3 mA	1
NEO-6 u-blox 6 GPS Modules	3.3VDC	67 mA	1
Maxim Integrated DS18B20	3.3VDC	1.5 mA	1
Texas Instruments TL072	+/-5VDC to +/-15VDC	2.5 mA each and 5 mA for the IC	5-8
	Total With development board	909 mA	
	Total With PCB	309 mA	

Table 5-14: Power Supply Current Consumption

It will be very important to meet high efficiency with the load current delivered. With the development board it sums to 909 mA. This is not very accurate because the Arduino Due will be used to only test the other functions and will not be a part of the power supply in the end. While the load current of 309 mA is what could truly be drawn from the power supply. This is important to how efficient our supply could be. To effectively gadge how much current is needed the whole circuit design would need to be complete. The range from what can be assumed for the Minuteman current consumption is between 250 mA and 500mA.

5.2.5.2 12V DC Power Adapter Supply

Like stated this is the power supply adaptor that will take the Unite States standard wall outlet of 120VAC at 60 Hz to acquire a steady 12VDC at 1A. This will proceed to the power supply and bucked down with the voltage regulator chosen below. This product was chosen to because of it capabilities and the prices. It was the cheapest selection, which meant buying multiple would not be a problem. For reliability and test assurance three of these devices were bought in case the faults with the initial product or failure later in the Minuteman design cycle.

Input Voltage Range	120VAC 60 Hz
Output Voltage	12VDC -/+5%
Output Voltage	Single
Current Range	1 A
Output Port	2.1 mm
Cost	$4.59

Table 5-15: 12V DC Power Adapter Supply Ideal Specifications

5.2.5.3 TPS62125

This Texas Instrument TPS62125 Buck DC to DC converter is optimized for low power conversion. Still applicable for our project taking in a 12VDC supply and outputting to a range of 1.7VDC to 10VDC. This product is much smaller than the Analog Devices LTM4622A, which would be better for our printed circuit board. This also mean it is a much more simpler design, but with little to no sensing capabilities compared to the LTM4622A. Another issue that this product has is the low output current. This can be a problem since this product is optimized for small integrated circuit design and not for multiple sensor modules with many integrated circuits drawing power from it.

Input Voltage Range	3VDC - 17VDC

Output Voltage Range	1.2VDC - 10VDC.
Output Voltage	Single
Current Range	0 mA - 300 mA
Output Ripple Voltage	Low
Power efficiency	High
Reliability	Yes

Table 5-16: TPS62125 Ideal Specifications

5.2.5.4 LTM4622A

The LTM4622A Switching Regulator is able to step down a 12VDC supply to a desired 0.3VDC to 6VDC for our microphones and the microcontroller. This DC to DC regulator is a small enough size for our PCB and is able to connect using a Land Grid Array (LGA). Looking through the datasheet, this product has reliability capabilities with overvoltage protection, overcurrent protection, and overtemperature protection. This is a big difference from TPS62125 where it ops for a smaller device size, but with less capabilities. The utility that the LTM4622A is greater and more diverse. The LTM4622A device is also able to output a larger current range from 0 Amps to 2.0 Amps. This will allow us to supply our sensor module and microcontroller with as much current as needed. Not only can it supply a large current, but it is able to regulate two different voltages with a more better filtering capabilities so that the regulated voltage is consistent and within the desired voltage range. This will ultimately fill our requirements for the design of the project.

Input Voltage Range	3.6VDC - 20VDC
Output Voltage Range	0.3VDC - 10VDC.
Current Range	0 A - 2.0 A Dual to single 4 A
Output Voltages	Dual
Output Ripple Voltage	Low
Power efficiency	High
Overvoltage Protection	Yes
Overtemperature Protection	Yes

Power Good Protection	Yes

Table 5-17: LTM4622A Ideal Specifications

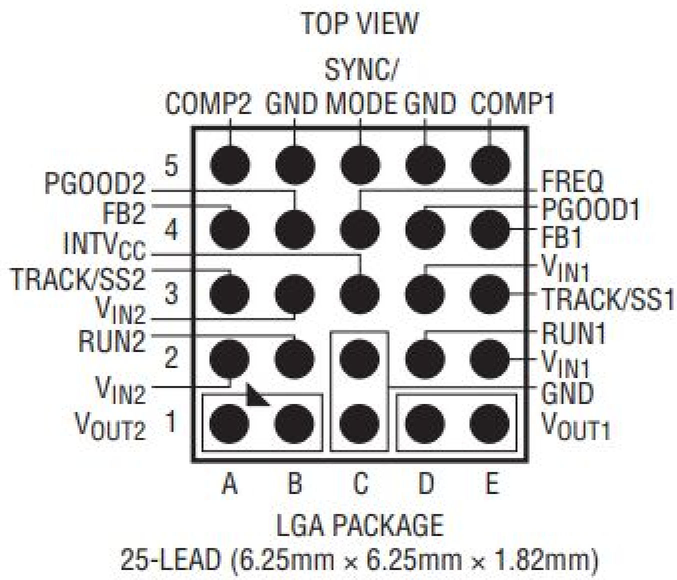

Figure 5-8: Analog Device LTM4622A Pin Layout [30]

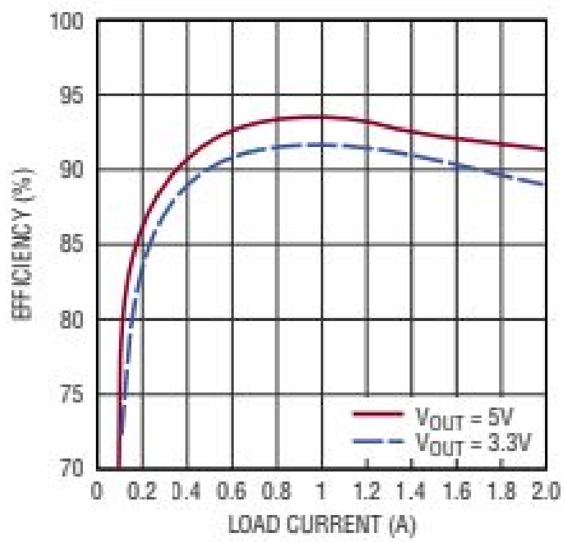

Figure 5-9: Output Efficiency vs Load Current LTM4622A [30]

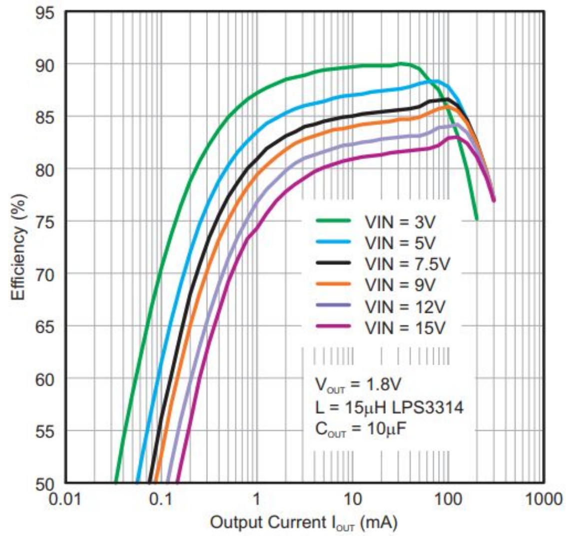

Figure 5-10: Output Efficiency vs Load Current TPS62125 [33]

When comparing these two devices we see that they have many different functions, but some functions are the same. Both having the range of output voltages that we want, but the LTM4622A is able to have dual output voltages. In this case our sensor module will have more than just the microphone so the LTM4622A can supply two different output voltages for different sensors in our sensor module. When comparing the current supplied from both integrated circuits, it is noticeable that the current from the TPS62125 is much smaller than the current supplied by the LTM4622A. The efficiency of the load current is also distinctly different, with LTM4622A peak efficiency at 93% with 12VDC input at 1 A (Figure 5-9) when compared to TPS62125 peak efficiency at 84% with 12VDC input at 130 mA (Figure 5-10).

This means that the LTM4622A can output a higher current with more power efficiency. With the LTM4622A higher current, more power efficiency, and reliability utility it is a much better option for our design. With the great amount of utility that the LTM4622A has when comparing it to the TPS62125, the LTM4622A costs over ten times the price. LTM4622A costs $17.31 in contrast to $1.60 for the TPS62125. The LTM4622A still outshines the TPS62125, even with the cost in mind, which makes the LTM4622A the choice for this project.

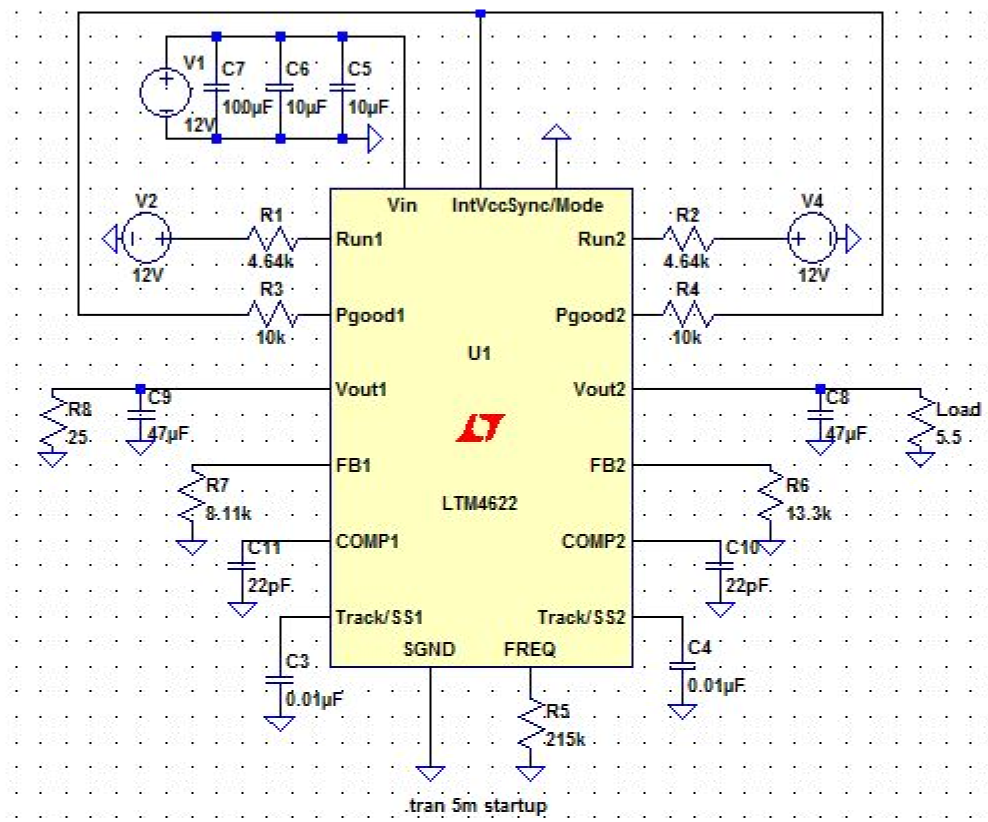

Figure 5-11: Power Supply Design LTM4622A

In the design of the power supply you can see that there are two outputs voltages. With Vout1 being 5VDC and 200 mA and Vout2 with 3.3VDC and 600 mA. The values can be changed with the FB1 and FB2 pins. By tying a 8.11 kΩ and 13.3 kΩ resistor to ground it create the 5V and 3.3V. The values for the resistors were found in the application section of the datasheet with the table below giving direction of the chosen resistors values.

V_{OUT} (V)	1.5	1.8	2.5	3.3	5.0	8.0	10.0	12.0
R_{FB} (k)	40.2	30.1	19.1	13.3	8.25	4.87	3.83	3.16

Figure 5-12: Power Supply Design LTM4622A FB1 & FB2 Resistor Values [LTM]

To simulate the current consumption, a load of 25 Ω and 5.5 Ω were use simulate the maximum current draw of 200 mA and 600 mA. Both the voltage output and current draw can be seen in Figure 5-13 and Figure 5-14. Because this is a switching regulator the voltage will switch on and off until it reach the set voltage by the feedback pins. The simulation was stopped at 5 ms so that the startup sequence could be shown as well as steady state at normal operation. From the transient response shown it can be seen that at 3.5 ms the operation changes from startup to normal operation.

An important part of this project was to get a high efficiency for both of our loads. It is important to do this for our power design. Specifically the efficiency of the power supply is dependent on the regulator. With the LTM4622A our efficiency with a 12VDC input is 87% for our 5VDC supply and 91% for our 3.3VDC supply. For this project these numbers suffice for our needs. For more power efficiency, burst mode can be used. This allows higher efficiency with lower currents. For this project burst mode was not used.

The use of coupling capacitors help with filtering out AC voltage from the DC line. This mean that if you have a capacitor tied to ground on a DC line, then as it should any small amount of AC voltage that might interfere with the supplies will be shorted to ground. Nullifying the AC aspect of the voltage and providing noise protection to the voltage supplies. Using multiple capacitors with different Farad values so that the circuit could be coupled at different frequencies. This provides even better immunity to noise, which could cause critical failures in downstream circuit.

This is a switching regulator. So it must switch on and off for the desired voltage. The FREQ port is the design port for the frequency that the regulator switches on and off. To implement the desired frequency a resistor tied to ground is used. The equation below is used to get the desired frequency by setting the resistor value.

$$f(\mathrm{Hz}) = \frac{3.2e11}{324k \| R_{FSET}(\Omega)}$$

The resistor value for R_{FSET} for our project is 215K □, which gives 2.5 MHz. This was chosen in the midrange of the power supplies capabilities of 800 KHz to 4 MHz. With suggestions from the datasheet a 2 MHz frequency should be used to reduce switching current ripple when the voltage is above 3.3VDC.

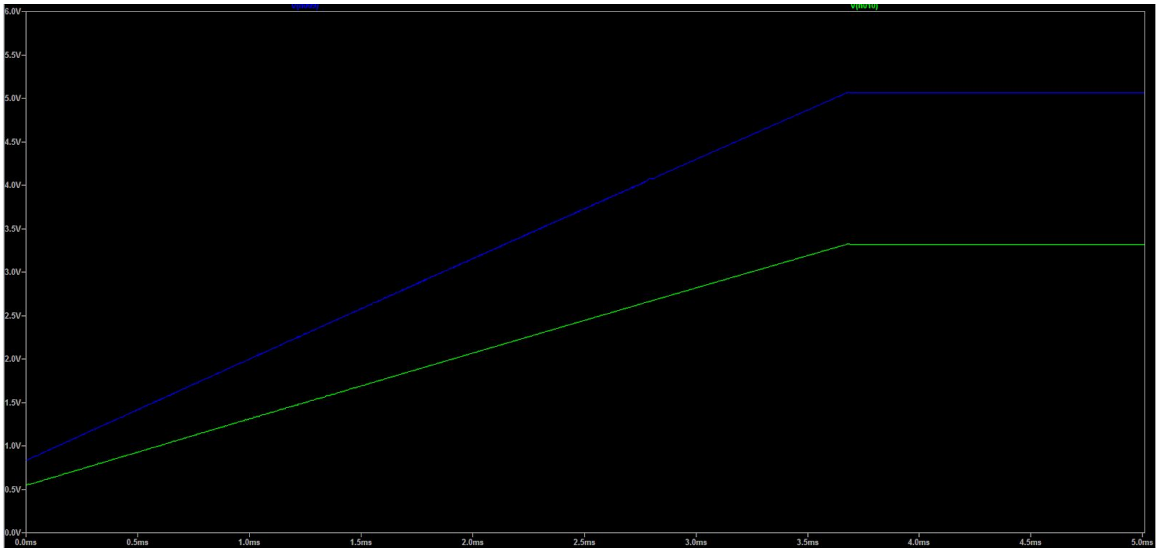

Figure 5-13: Power Supply Design LTM4622A Voltage vs Time

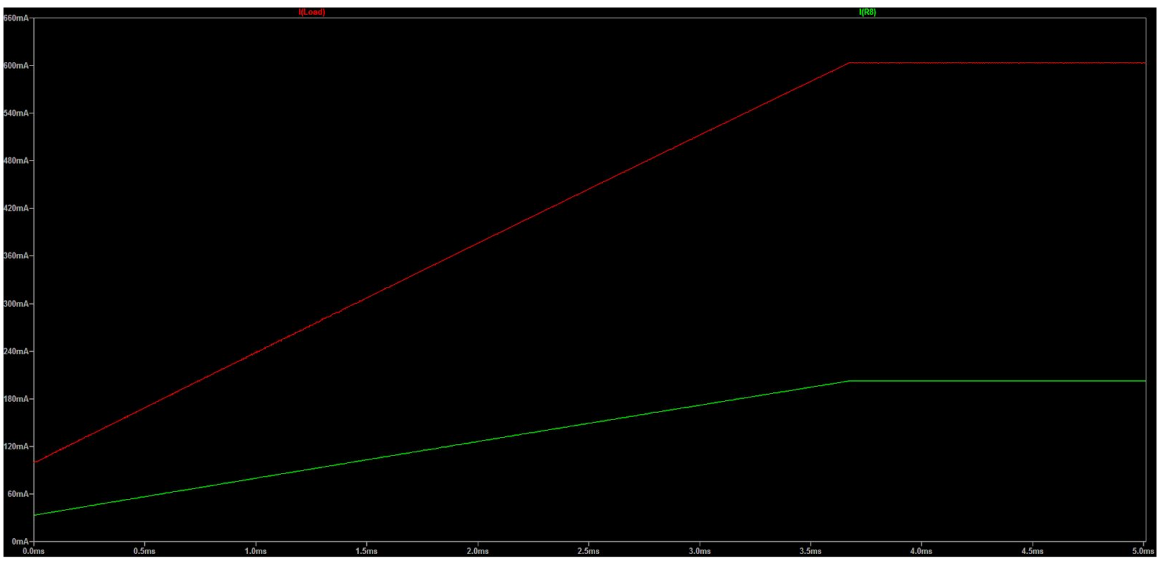

Figure 5-14: Power Supply Design LTM4622A Current vs Time

5.2.5.5 MAX1044/ICL7660

The MAX1044/ICL7660 is a CMOS switched-capacitor voltage converter. It will be used to invert the 5VDC power supply. This would give a -5VDC supply, which will be used for the rails on the Texas Instruments TL072. There were different methods of doing this, but through searching for the best method this was chosen. Previously this was not needed because a single rail operational amplifier was used, but with new designs this component became necessary. For this project the MAX1044/ICL7660 will be used as the basic negative voltage converter as shown in the datasheet. The specification for why this device was chosen can be found in the table below.

Operating Voltage (LV pin open)	3VDC to 10VDC
Output Voltage	-5VDC
Current Consumption (LV pin open)	200 uA
Power Efficiency	98%
Output Efficiency	99.9%
Dimensions (L x W x H) mm	1.930 x 1.930 x .630

Table 5-18: MAX1044 Ideal Specifications

With a high output efficiency of 99.9% this product will provide a stable rail for the Texas Instruments TL072. The design of this product is simple and is shown below.

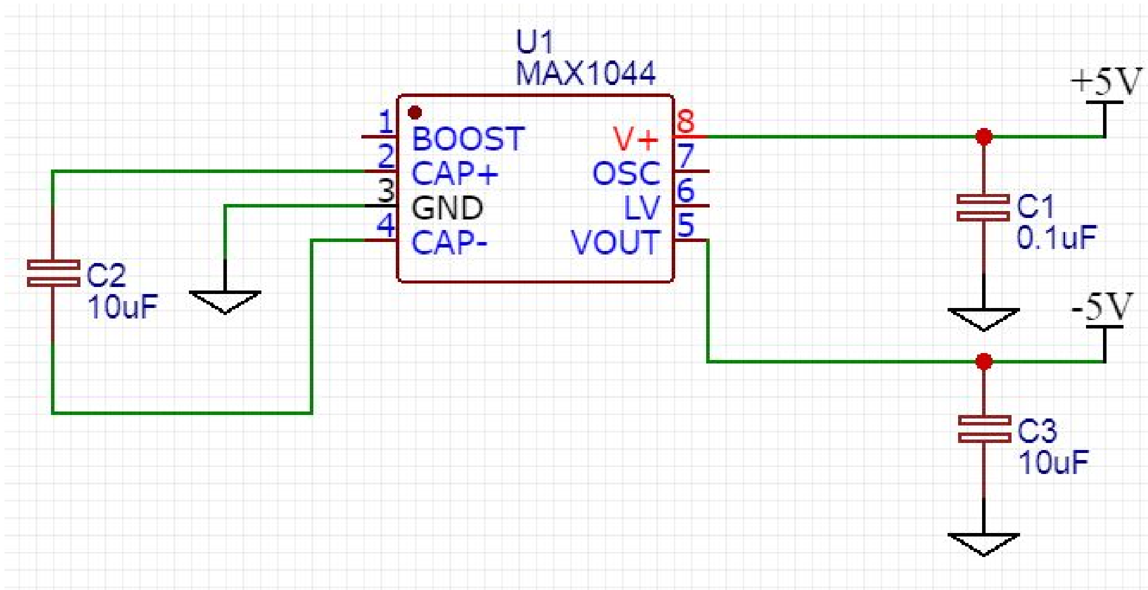

Figure 5-15: MAX1044 CMOS Voltage Converter Design

As stated in the datasheet the LV pin should be open for voltages above 3.5VDC as well as the Boost pin and the OSC pin. These pins should be used for lower voltages. The capacitance values were taken from the application section and are used for coupling. This product will be implemented into the PCB board within the power section as shown in the diagram for the PCB layout.

5.2.6 Overvoltage and Overcurrent protection

To make sure the power supply has the best reliability, it will be important to add circuit that protects the downstream circuit so minor fluctuations do not cause catastrophic effects to the circuit as a whole. One way this can be done is adding overvoltage and overcurrent protection to our 12VDC 1 A wall outlet power supply. This can be done in many ways with very complex circuits depending on the results that are required. For this project the design of the overvoltage and overcurrent protection will be simple, since it will be used only for the main supply.

To design our overvoltage and overcurrent protection circuit a zener diode and a fuse will be used. This design will be straight forward, since it only contains two components. The overvoltage and overcurrent protections depends on the ratings chosen for the zener diode and the fuse. Because the overvoltage protection for the main power supply is 12VDC at 1 Amp our design will use a zener diode with a turn on voltage of 12VDC. This means that any overvoltage that goes across the zener will be dissipated, leaving nearly 12VDC on the line. For this design any current above 1 Amp will be considered overcurrent and our fuse should be rated equal or over this. The circuit below will be the design of the overcurrent and overvoltage protection.

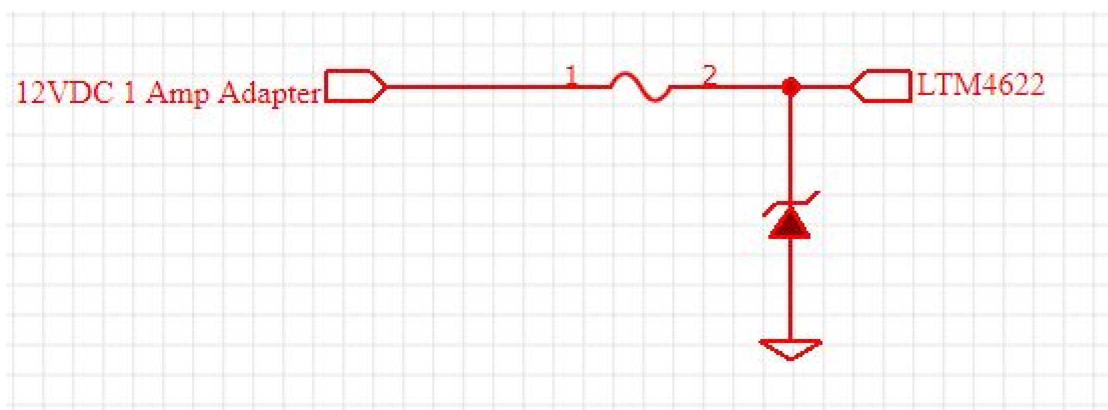

Figure 5-16: Overcurrent and Overvoltage Protection Design

The fuse is in series with the main power source so that if current exceeds 1 Amp, then the fuse can blow to protect downstream circuitry. The zener diode is in parallel to the fuse, so that the circuit can still get the 12VDC and dissipate any voltage that will trigger the turn on voltage of the zener diode. It is also important that the current of the zener diode is within range of the fuse. The fuse that was chosen for the design is SF-0603F125 and the zener diode is the MMSZ5242B_H2.

5.2.6.1 SF-0603F100

The SF-0603F125 is a fuse for the overcurrent protection of the power supply. This table below shows the specs for why this component was chosen.

Rate Current	1.00 Amp
Rated Voltage	32VDC
Fast Acting	Yes

Table 5-19: SF-0603F100 Ideal Specifications

5.2.6.2 MMSZ5242B_H2

Zener diode is used for overvoltage protection for power supply circuit. The table below are the specification of why the part was chosen.

Zener Voltage Range	11.40VDC to 12.60VDC
Power Dissipation	500mW

Table 5-20: MMSZ5242B_H2 Ideal Specifications

5.2.7 Sensor Module

The sensor module the project is the main component that is used to receive sound input signals, and will have the signals sent to the central module. The sensor module will utilize multiple hardware components, such as microphones, GPS, a compass, a thermometer, and amplifiers to power the microphones. The microphones require to be of decent quality so distinct sounds of gunshots can be compared, but also cheap so the project does not go over budget. The research done for finding the right microphone to use for this sensor module is discussed in the hardware initial design section of this project. Two methods of finding the location of the sound from the microphones are discussed in research section of this project, which is triangulation and multilateration.

When this module was initially proposed, we suggested that it should be wireless for this project. This seemed ideal at first and would allow it to be set up without dealing with the hassle of cable management, but since this module is designed as security device to protect people, transmitting any signal wirelessly can be compromised. By using Wi-Fi or any other wireless communication method, this can lead to an exploit where the signal could be jammed by the attacker, or either congestion of the network can cause the transmitted signal sent to the receiver to be significantly delayed. Due to this, the design of the sensor module was changed so the signals would be sent using wires to the central module, which reduces the risk of the wireless transmitter being exploited by an attacker.

The sensor module for this project would be designed such that it would be mounted on a tripod, so it can be set up in a short time. The sensor module mounted on the top of the tripod would also be able to fold up like a tripod stand, where each arm that extends out contains one microphone. When folded out, the microphones would have the same distance between each other by around 50 centimeters, and would have one microphone at the top of the main pole the arms are attached to. This forms the shape of an equilateral triangular pyramid, which is necessary in order to use triangulation as a location method. With multilateration, the shape is not necessary, but having the sensor module designed as a pyramidal form can allow the flexibility of choosing either method to use. The sensor module would also be designed so it is at least less than 15 to 10 lbs, as using light material for the tripod will reduce cost and make it easier to transport.

A GPS module would be used for each sensor module that is designed. This would allow ease of use when setting up the sensor module, as having no GPS module used would mean manual calculations would need to be done to get the location, which would then be set as constants for the program to run the location algorithm. This means the sensor module can be more portable and be set up in around a minute. A compass is also used alongside the GPS module to give the orientation of the array, so necessary adjustments can be made for the location algorithm. Only one temperature probe would be installed on one of the sensor modules, as they would not be too far apart for this project that it will make a difference.

5.2.7.1 Microphone

A microphone is a device that converts a sound input into an electrical signal output. Microphones can generate electrical signals using multiple types of principles, but utilize one thing in common: a diaphragm. The diaphragm is a thin piece of material that vibrates when sound waves collide with it. When the diaphragm vibrates, a small electrical current is generated, which is then amplified by the microphone. The types of microphones we considered for this project were dynamic and condenser microphones, as other types were too expensive or not needed for our project. [13]

Dynamic microphones operate by utilizing a coil wrapped around a permanent magnet. When the diaphragm vibrates, the coil would then vibrate over the magnet and generate an electromagnetic field, which would then produce a current output through the wires connected to the coil. Due to its simple design, dynamic microphones are more durable and can handle sound waves with high dB compared to condenser microphones. [13]

Condenser microphones operate by utilizing the diaphragm as the front plate of the capacitor. When the diaphragm vibrates, the distance between the front and back plate changes, which would change the value for its capacitance. When the plates of the capacitor are close, a charge occurs, and when the when they are apart, the charge would then be discharged as current. In order for this microphone to operate, a voltage would need to be supplied to the capacitor. Since condenser microphones utilize the diaphragm as the front plate of the capacitor, they are more sensitive to sound compared to dynamic microphones. [13]

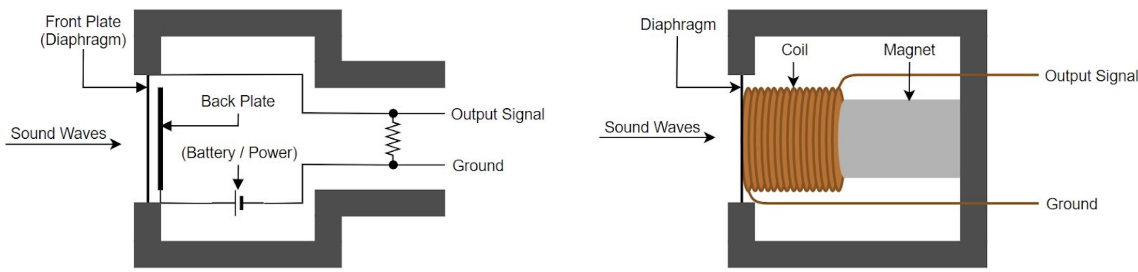

Figure 5-17: Diagram of dynamic (left) and condenser (right) microphone

Microphones are designed with certain patterns for how they pick up sound around them. Cardioid patterns are shaped like a heart and pick up sound around the front of the microphone, but sound outside the heart shape is lower in amplitude. Unidirectional patterns only pick up sound from the front of the microphone, and Omnidirectional patterns pick up sound all around the microphone. For our project, any microphone we chose had to be omnidirectional, since collecting sound from all direction would increase the accuracy of pinpointing the location of a discharged firearm.

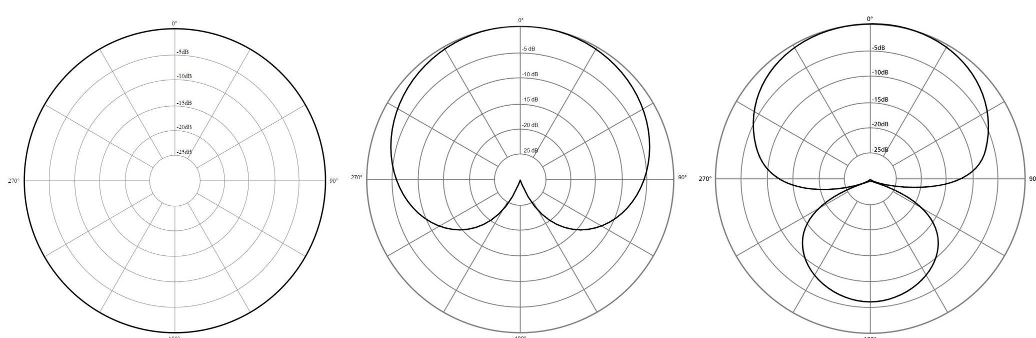

Figure 5-18: Mic polar patterns (Omnidirectional, Cardioid, Supercardioid) [14]

For our project, we needed a microphone that would meet the following requirements, such that it has high sensitivity, is omnidirectional, has a good signal to noise ratio, has a flat frequency response, and is also cheap. The following microphones were considered for our project:

Knowles SPW0690LM4H-1 - This was the first microphone we considered, which uses MEMs technology. This uses an etched diaphragm on a silicon wafer to generate an electrical signal output, which works similarly to a condenser microphone. This microphone was considered initially as it had the following characteristics as shown in Table 5-21.

Directionality (Polar Pattern)	Omnidirectional
SNR (Signal to Noise Ratio)	66.5 dB
Sensitivity	-41 dB
AOP (Acoustic Overhead Point)	135 dB SPL (Sound Pressure Level)
Frequency Range	20 Hz to 10 kHz
Output Type	Digital PDM (Pulse Density Modulation)

Table 5-21: Knowles SPW0690LM4H-1 Specifications

We chose this initially as it already had a built-in analog to digital converter, a high resolution, and a high AOP of 135 dB SPL, which is good since firearms have a decibel level around at least 140 dB, so the microphone should not distort as much as the sound is picked up from a distance. However, we later decided not to use this microphone due to the way it needs to be soldered, and also because we realized its built-in ADC is a constraint. If the ADC ended up needing to be redesigned later in the project, this meant the entire mic would need to be replaced because of it, so we decided to only use an analog mic instead.

CUI CMC-6035-130T - This is an electrostatic condenser microphone that is ideal for our project. The reasons we chose this is because of its following characteristics as shown in Table 5-22.

Directionality (Polar Pattern)	Omnidirectional
SNR (Signal to Noise Ratio)	70 dB
Sensitivity	-42 dB

High Maximum Input	130 dB SPL
Frequency Range	100 Hz to 20 kHz
Output Type	Analog

Table 5-22: CUI CMC-6035-130T Specifications

The microphone is also designed to be waterproof, which is ideal for our project as the microphones would be exposed to the outside environment, where it will be exposed to rain and moisture in the atmosphere. The frequency response is also ideal for our project, as we will be only using lower frequency ranges, which are within the flat line as shown in the Figure 5-19. Figure 5-20 also shows the diagram of how the power module interfaces with the microphone. A 2.2 kΩ load resistor is added to match the impedance of the microphone, which is 2.2 kΩ. This ensures maximum power transfer for the microphone. A coupling capacitor is added between the microphone and the amplifier, so DC noise from the power module is removed.

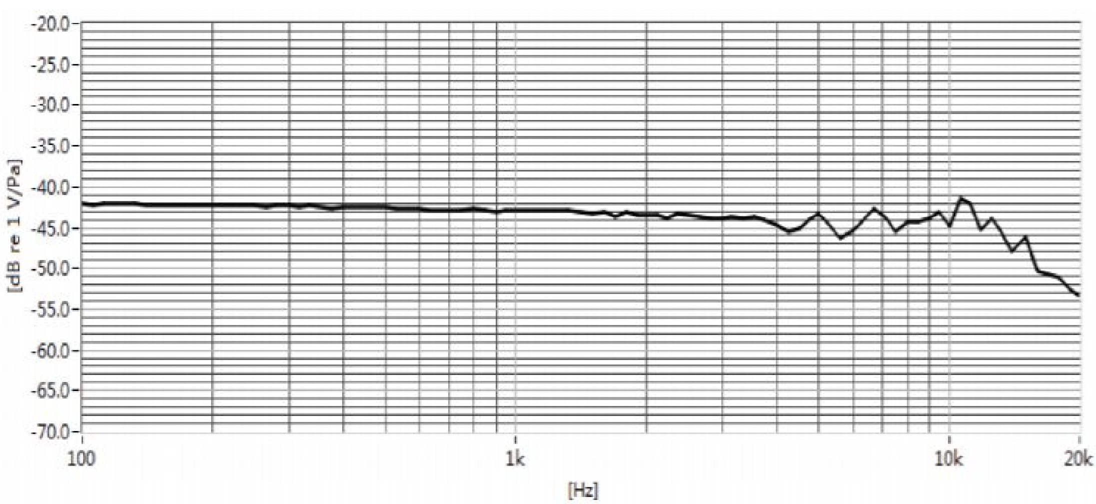

Figure 5-19: Frequency response of CUI CMC-6035-130T [15]

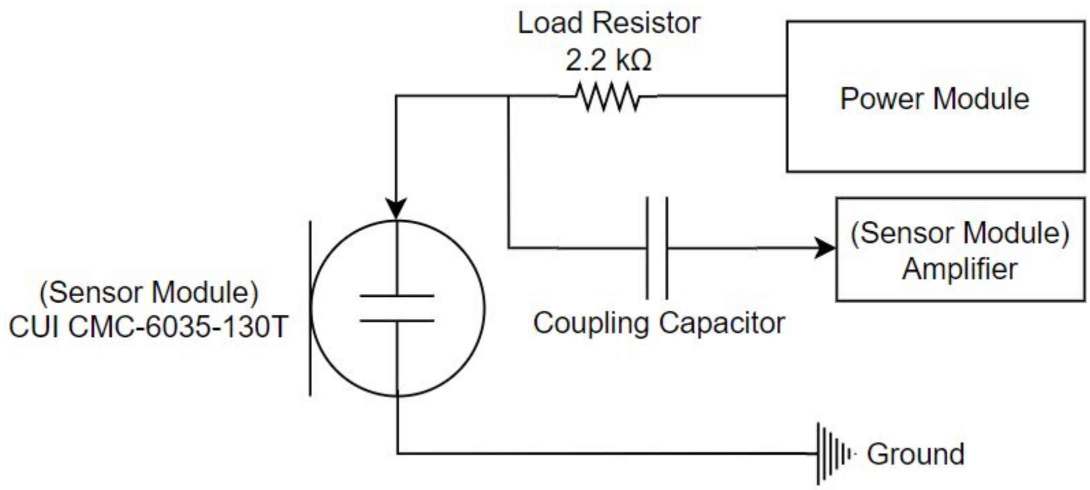

Figure 5-20: Microphone Interface Diagram

5.2.7.2 Thermometer

Knowing the speed of sound changes with the respect of temperature, based on the equation of $C(m/s) = 331.3\sqrt{1 + \frac{T}{273.15\,K}}$, where T is in degrees Celsius, a thermometer is needed to get the temperature of the environment where the microphones are located. This will ensure that the speed of sound calculated would be close to the real value as possible instead of just using the ideal speed of sound of 343 m/s, which could affect the accuracy of pinpointing the location of a discharged firearm. The following temperature probe will be used for our project:

Maxim Integrated DS18B20 - The microphone arrays used in this project would be also be exposed to the environment, which contains moisture and other elements that can damage electrical components that aren’t protected. Due to these conditions, the DFRobot DFR0198 thermocouple thermometer was chosen, as it provides a waterproof enclosure for the DS18B20 temperature sensor. This thermocouple was also chosen as it has the following specifications that are ideal for our project as shown in Table 5-23.

Temperature Accuracy	±0.5℃ for -10℃ to 85℃
Digital Output Resolution Range	9 bits to 12 bits
Temperature Range	-55℃ to 125℃
Query Time	Less Than 750 ms

Voltage Range	3V to 5.5 V

Table 5-23: DS18B20 Specifications

Since the thermometer has a voltage range of 3V to 5.5V, this allows any microcontroller with a 3.3 V and 5 V voltage output to power this thermocouple. Knowing the temperature accuracy is around ±0.5°C, the speed of sound calculated would also be accurate as possible in the environment it will be used in. The query time being less than 750 ms means that it will be able to update it's current temperature more than 1 second, which is more than enough to ensure the speed of sound calculation will be accurate. Figure 5-21 shows how the temperature probe interfaces with the other modules, where the power module supplies 3 V to the sensor, and the data from the sensor is sent to an I^2C port of the microcontroller. A 4.7 kΩ resistor is added between the voltage input and data output as suggested from the specifications of the temperature probe.

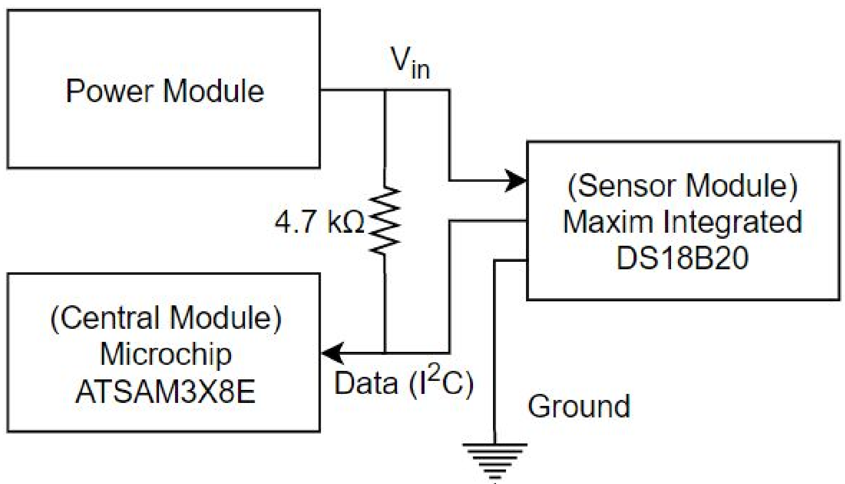

Figure 5-21: Temperature Probe Interface Diagram

5.2.7.3 Compass

Knowing the microphones in this project are used to pick up the location of a discharged firearm, that means they will be deployed as an array on a stand, which could have it's orientation at any direction where it's set up. Due to this, a compass would need to be used, as it faces towards the strongest magnetic field. This would face towards the Earth's north magnetic pole, as it will not be exposed to any other strong magnetic fields in our project. Using the compass, it will be able to provide a reference of the orientation of the microphone array which would then be used with the GPS coordinates for triangulating the location of a discharged firearm.

QMC5883L 3-Axis Magnetic Sensor - This magnetic sensor module was chosen for our project, as it is compact, provides reasonable accuracy, and has a low power consumption. The QMC5883L magnetic sensor has the following specifications that are ideal for our project as shown in Table 5-24.

Update Rate Range	10 Hz to 200 Hz
Operating Temperature	-40℃ to 85℃
Voltage Range	2.16 V to 3.6 V
Heading Accuracy	1°
Output Type	I^2C Digital Output

Table 5-24: QMC5883L Specifications

When the microphone array is deployed, it will remain stationary until it is either relocated somewhere else. The update rate selection of 10Hz of this sensor is ideal since it is unnecessary to continuously update when the microphone array isn't going to move at all when it is deployed. The voltage range of 2.16 V to 3.6 V would allow this to be used with a microcontroller that provides 3.3 V and also the I^2C digital output provides an easy interface to work with. The heading accuracy of 1° would provide a reasonable offset orientation of the microphone array that is used for pinpointing the location of a discharged firearm.

Bosch BNO055 9-Axis Absolute Orientation Sensor - This sensor is the fusion of an accelerometer, gyroscope, and a magnetometer. This sensor is more expensive than the QMC5883L, however it can provide better orientation if the microphone array is designed to be mobile. The Bosch BNO055 has the following specifications that are ideal for our project as shown in Table 5-25.

Operating Temperature	-40℃ to 85℃
Voltage Range	1.7 V to 3.6 V
Gyroscopic Range	±125 °/s to ±2,000 °/s
Acceleration Range	±2 g or ±4 g or ±8 g or ±16 g
Compass Heading Accuracy	±2.5°
Max Update Frequency	400 Hz

Output Type	I^2C Digital Output

Table 5-25: BNO055 Specifications

This sensor is ideal as it uses euler angles to provide the pitch, yaw, and roll, which makes the orientation easy to comprehend. The sensor also separates the acceleration due to gravity and acceleration due to movement. The maximum voltage range allows this to easily be used with a microcontroller that provides 3.3 V, the digital output allows easy interface with the microcontroller, and the clock frequency of 400 Hz is useful as having a higher update rate allows the results to be more accurate if the microphone array is in motion. The ±2 g range of the accelerometer is also more than enough to be suitable for a vehicle in motion, if the microphone array is mounted on top. Figure 5-22 shows a diagram of how the compass interacts with other components. The power module supplies around 3 V for the compass, while the data output of the compass is sent to an I^2C port of the microcontroller.

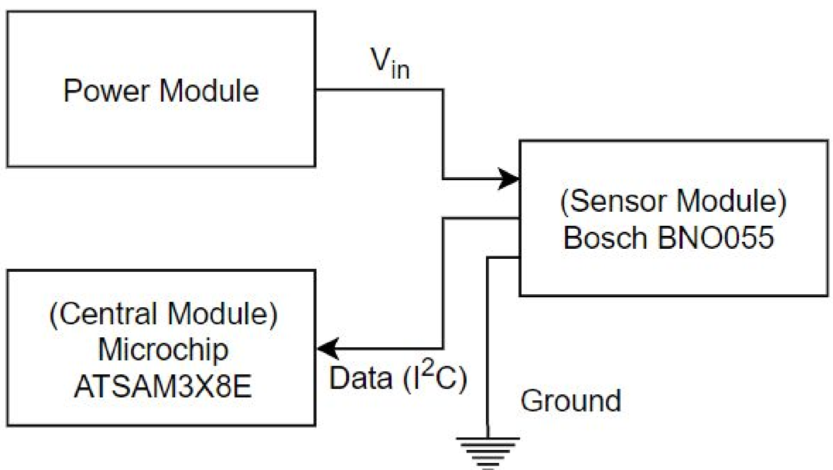

Figure 5-22: Compass Interface Diagram

5.2.7.4 GPS

GPS (Global Positioning System) is necessary for our project, as a reference is needed for the location. A GPS module would be attached to each microphone array, which would also be combined with a compass to provide a reference frame. Using the coordinates from the GPS module for each array, the coordinates can be used to perform the calculations necessary to determine the location of a discharged firearm.

u-blox NEO-6M GPS Module - The DIYmall 6M GPS Module was chosen for this project and uses the u-blox NEO-6M, as it is cheap. The GPS module has the following specifications that are ideal for our project as shown in Table 5-26.

Maximum Voltage	3.6 V
Highest Sensitivity	-161 dBM
Cold Start Time	27 seconds
Accuracy	2.5 m

Table 5-26: NEO-6M Specifications

Since the module has a cold start of 27 seconds, this would allow the microphone array to be deployed quickly and have it’s position accurate less than 1 minute. The maximum voltage of 3.6 V would allow this to be used with the power module that would supply 3 V of power, and the 2.5 m horizontal accuracy of the GPS module would also allow the location of the discharged firearm to be in a tolerable position. Figure 5-23 shows an interface diagram of how the GPS module interacts with the other modules in the project.

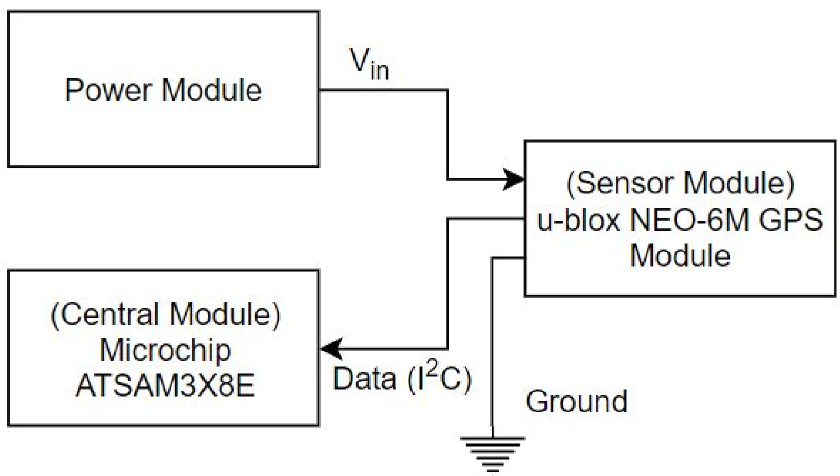

Figure 5-23: GPS Interface Diagram

5.2.7 Analog to Digital Converter

The analog to digital converter, also known as ADC, is a critical component required for this project. The analog to digital converter is to convert an analog input with a continuous time and continuous voltage level to a discrete time and discrete voltage level digital output. The discrete time of the analog to digital converter is known as the sampling period, which is the time difference between two consecutive samples. The frequency of the analog to digital converter would be the inverse of the sampling period, which is also known as the sampling rate.

The performance of an analog to digital converter is determined mainly by the bandwidth of the signal it's converting and also the signal to noise ratio (SNR). The bandwidth is the frequency range of the analog input, which then determines the sampling rate that is needed. This is done by using the Nyquist Shannon sampling theorem. The Nyquist Shannon sampling theorem requires the sampling rate to be greater than twice the bandwidth of the analog input in order for a wave to be reconstructed. For instance, if a frequency is a range of 500 Hz to 4000 Hz, then a sampling rate greater than 7000 samples per second is needed for the minimum sampling rate.

The signal to noise ratio of an ideal analog to digital converter can be determined by the quantization error. The quantization error is the difference between the digital signal level that is closest to analog signal level. The digital signal level is determined by the resolution the analog to digital converter uses. The resolution is the value of 2^N, where N is the amount of bits the analog to digital converter can encode. The quantization error then uses the equation as shown below:

$$\text{Quantization Error} = -20\log(2^N)\ \text{dB}$$

Given this equation, the ideal signal to noise ratio can be determined knowing the amount of bits used. For example, a 16-bit analog to digital converter has an ideal signal to noise ratio of -96.3 dB using this equation, while an 8-bit analog to digital converter has a signal to noise ratio of -48.2 dB. The example here shows that having a higher resolution is ideal for our project, as having more noise would be less suitable for comparing gunshot samples.

In order to determine the right amount of sampling to use for an analog to digital converter, the time difference between microphones are needed. This was done using a scenario when a gun was discharged 500 meters in front of the first microphone, as shown i. The array for this scenario utilizes multilateration, has five microphones, and each microphone is separated by 50 centimeters from the origin point of the microphone array, as shown in Figure 5-24.

Knowing the microphone A picks up the sound first, the distance from the sound source to it would be 500 meters. The second microphone, microphone D, is chosen, which is 50 centimeters behind from microphone A in the x-axis, and 50 centimeters above microphone A in the y-axis. Knowing the x-axis length, and the y-axis length, the hypotenuse of the triangle can be calculated using the x-axis distance of 500 meters added by 50 centimeters and the y-axis distance of 50 centimeters, which is shown below.

$$Length = \sqrt{500.5^2 + 0.5^2} \cong 500.50025\ m$$

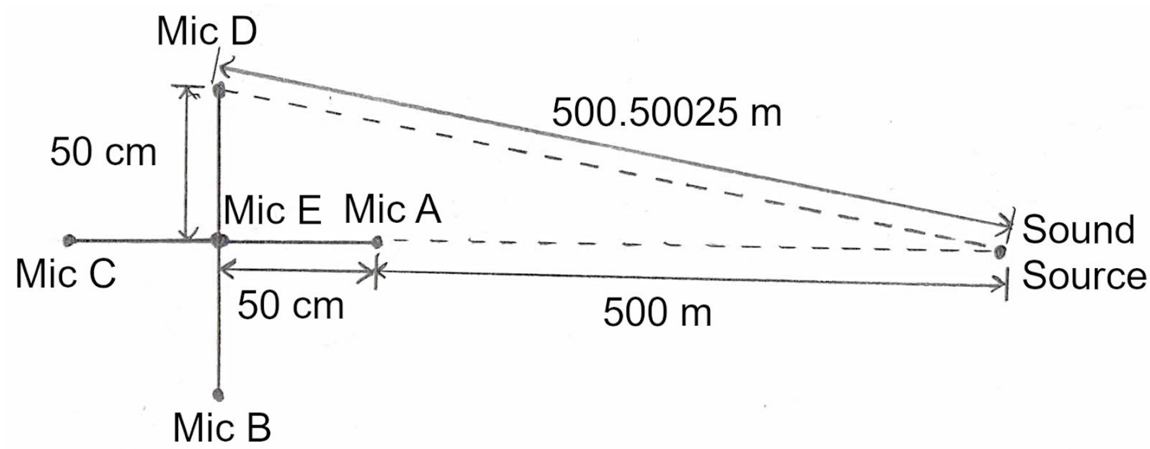

Figure 5-24: Microphone array with a sound source 500 meters in front of mic A

Knowing both distances between microphone A and microphone D, the time can the be calculated by having both lengths divided by the ideal speed of sound constant. The distance of 500 meters would be subtracted from 500.50025 meters, and would then be divided by 343 meters per second, which would then equal 1.458 milliseconds. Knowing the time difference, the sampling period can be obtained, which would be the inverse of the time. This would be equal to 686 samples per second

Time Difference = [(500.50025 - 500) m] / 343 (m/s) = 1.458 ms

Sampling Rate = (1 / 1.458 ms) = 686 samples/second

Given the Nyquist Shannon Theorem, the sample rate is the twice frequency range of the analog input in order for it to be faithfully reproduced. The microphones in the microphone array do not require the entire frequency spectrum, so only a frequency range of around 300 Hz to 5000 Hz would be needed for this project, as only lower frequency sounds are required for comparing gunshots. A high pass filter at 300 Hz is used so low frequency sounds, such as the human voice spectrum can be removed, where voice of men are normally at 85 Hz to 180 Hz and women are normally at 165 Hz to 255 Hz. The low pass filter is located at 5 kHz as when viewing the frequency response of the microphone, the frequency response after 5 kHz becomes more unstable and starts drops off after 10 kHz, where 20 kHz is the half cutoff of the microphone.

Knowing the bandwidth chosen is 4.7 kHz, then the sampling rate must be greater than 9400 samples per second. Knowing the sampling rate is greater than the frequency for the time difference between microphones A and D, a sampling rate of 9500 samples per second would be more than enough to ensure the microphone array calculates the position of sound accurately.

The Arduino Due is a development board that has a microcontroller known as the ATSAM3X8E, which is made by Microchip. This microcontroller has a built-in analog to digital converter that has a 12-bit resolution, which has 4096 voltage levels. This gives a maximum ideal quantization error of -20log(4096), which is -72.3 dB for the signal to noise ratio. However, when looking through the data sheet of the microcontroller, the signal to noise ratio has a range that goes from -60 dB to -73 dB. This is expected as the noise would be higher, especially if noise is generated from other components used as well.

The ATSAM3X8E microcontroller has a theoretical maximum analog to digital sample rate of 1 MHz. This is only for one input, as the total amount of inputs the analog to digital converter of the microcontroller is 16. Given the microphone array has a minimum amount of five microphones, this sample rate would be limited by the time it takes for the microcontroller to have the samples read.

5.2.8 Amplifier

Since an electrostatic microphone was chosen for the project, an amplifier would be required to increase the gain. The amount of gain needed is determined by the voltage required for the analog to digital converter. For the ATSAM3X8E microcontroller, the voltage range for the conversion from analog to digital is 0 V to 3.3 V or 5 V, depending upon the design chosen. Given a range starting from 0 V for the analog to digital converter, the amplifier would require a DC bias offset in order to keep the input signal from being clipped.

Knowing the sensitivity of the microphone used, the voltage output can be determined. The decibel equation related to voltage is equal to -20log(V). Taking the inverse of the equation, the voltage would be equal to $10^{(-db/20)}$. Knowing the microphone chosen is the CMC-6035-130T, the maximum sensitivity of -39 dB can be plugged into the equation as shown below:

$$V_{out} = 10^{(-39/20)} = \pm 11.2\ mV$$

Knowing the maximum output voltage, the gain can be determined by having the voltage needed for the analog to digital converter divided by the voltage of the microphone. The range chosen for the analog to digital converter is 0 V to 3 V with DC bias, then the waveform voltage range without the bias would be ±1.5 V. Knowing the voltage needed, the gain would need to be around 133.7, which can be rounded up to 140, to make designing the amplifier easier.

Texas Instruments TL084 - When choosing an amplifier, the TL084 is a consideration. The TL084 has been used in labs from previous classes, so given the familiarity with it, it will be easy to use for the project. The TL084 has the following specifications as shown in Table 5-27.

Supply Voltage Range (V_{CC})	±5 V to ±15 V
Maximum Differential Input Voltage (V_{IN})	±30 V
Maximum Input Voltage (V_I)	±15 V
Common Mode Rejection Ratio (dB)	86 dB
Total Harmonic Distortion (%)	Average of 0.003%
Input Noise Voltage Density (nV/√Hz)	Average of 18 nV/√Hz
Slew Rate (V/µs)	Average of 13 V/µs
Input Resistance (Ω)	10^{12} Ω
Amount of Amplifiers	4

Table 5-27: TL084 Specifications

The TL084 has a total of four amplifiers, which is useful for this project as it can reduce the space taken on the PCB. Others specs listed, such as the common mode rejection ratio, low noise voltage, and a small harmonic distortion percentage is useful for our project. To simulate the amplifier, National Instruments Multisim application was used. A non-inverting feedback amplifier design was chosen for the simulation, so the general equation is the following:

$$A_V = [(R_2 \div R_1) + 1]$$

Given the equation, the resistors can be picked so a gain of 140 is obtained, such as R_2 being set to 139 kΩ, and R_1 being 1 kΩ. However, by using only one amplifier, noise and distortion would be amplified as well, so the design can be made such that two amplifiers are used to obtain a gain of 140. This can be done by using an amplifier that has a gain of 10, and a second amplifier that has a gain of 14. The equations for solving the gain of both amplifiers are shown below:

First Stage ($A_v = 10$)

$$R_1 = 1\ k\Omega, \quad R_2 = 9\ k\Omega, \quad A_v = [(9\ k\Omega \div 1\ k\Omega) + 1] \rightarrow A_v = 10$$

Second Stage ($A_v = 14$)

$$R_1 = 1\ k\Omega, \quad R_2 = 13\ k\Omega, \quad A_v = [(13\ k\Omega \div 1\ k\Omega) + 1] \rightarrow A_v = 14$$

Given the fact that the analog to digital converter requires a range of 0 V to 3 V, a DC offset bias is also needed so the AC signal from the microphone is shifted. If this offset is not used, then half of the waveform from the microphone would be cut off when it goes through the analog to digital converter. This can be supplied by a 3 V, which has it's voltage divided by half, which is 1.5 V for the offset. Figure 5-25 shows the simulated circuit of the microphone amplifier using Multisim. Figure 5-26 also shows the gain of the amplifier by comparing an input voltage of ±10 mV and the output voltage, which is around 2.8 V with an offset as shown.

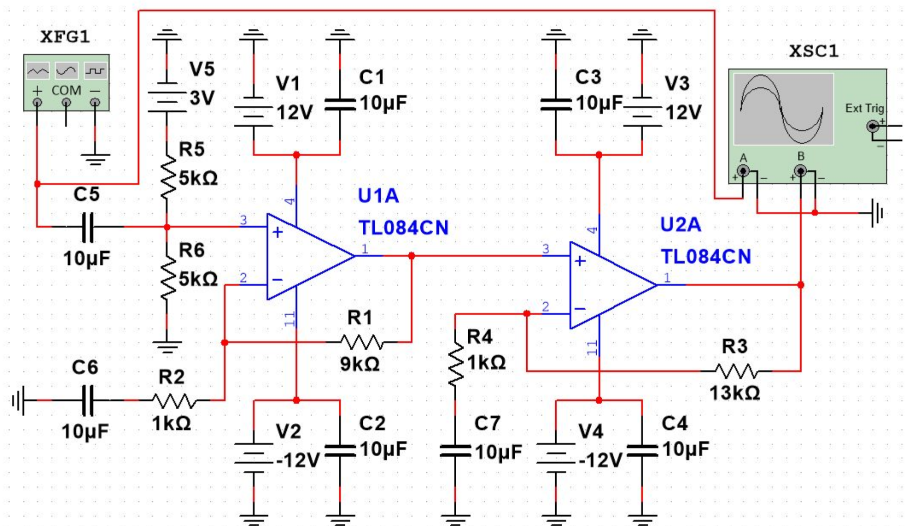

Figure 5-25: Microphone Amplifier Circuit

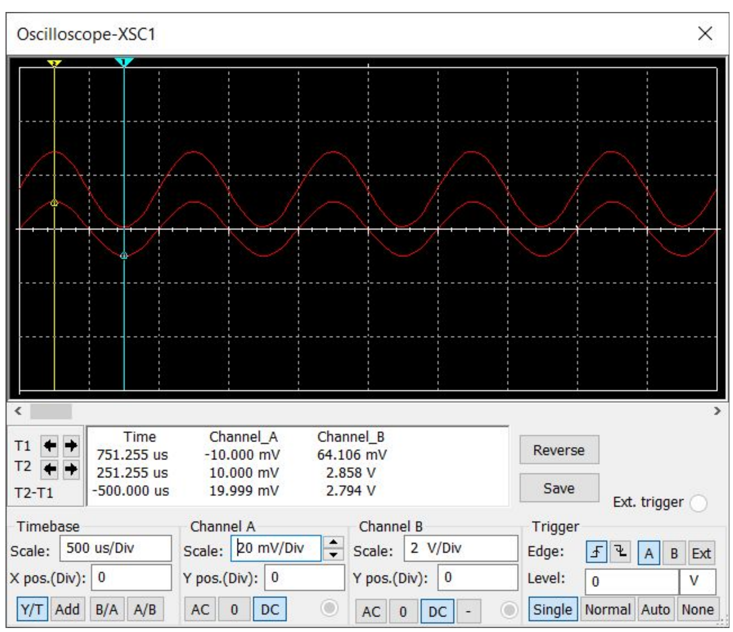

Figure 5-26: Gain of Microphone Amplifier

In Figure 5-25, the capacitor C5 is used to block any DC voltage from the microphone source. Since only two terminals exist for the microphone, the DC voltage from the power module is connected to the output of the microphone. This will cause noise and also add an unwanted DC offset if a capacitor is not used. This capacitor is known as a coupling capacitor, which forms a short for AC signals, but is an open loop for DC signals, so the DC signal from the power module is blocked.

Coupling capacitors can also be used with the voltage supply for the amplifiers, as AC noise can be picked up as well. By forming a short as shown for capacitors C1, C2, C3, and C4, the AC signal picked up from the voltage supply would be sent to ground instead of the amplifiers. Figure 5-27 shows the frequency response of the amplifier and the phase as well, which was done using the AC sweep simulation in Multisim. The dB value of the flat line is around 43 dB, so the half power frequency would be 40 dB, which is at 25 Hz. The response starts to smooth out around at 100 Hz.

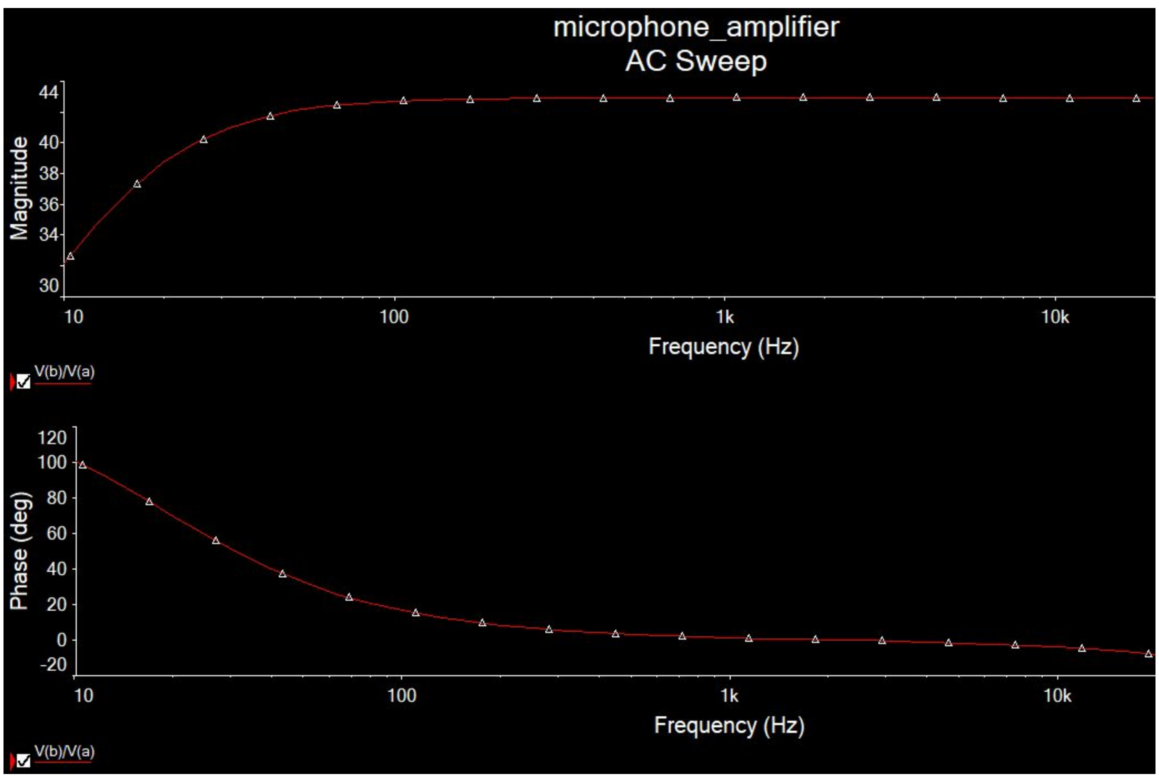

Figure 5-27: Microphone Amplifier Frequency Response

After looking through other amplifiers, the TL084CN might not be ideal. One reason is due to the fact that it has 4 amplifiers within it. By having multiple microphones outputs connected, this can cause potential noise between them, which is not ideal for our project. Another reason another amplifier is needed is due to the power module design. By selecting an amplifier that uses only a positive voltage input, this would make designing the rails to supply the voltage for the microphones easier, as other devices used do not require negative voltage.

Multiple alternative amplifiers were searched, such as the LM386 operation amplifier made by Texas Instruments. This amplifier was found to have an adjustable gain of 20 to 200 by using a resistor and capacitor in series to adjust it. However, when looking through other specifications, it was found the have a total harmonic distortion of 0.3%, which is much worse than the total harmonic distortion of the TL084 amplifier. The amplifier's design is also old, and dates back from the 1980's, so more suitable amplifiers could be used today that are the same price level and are of higher quality.

Maximum Integrated MAX4466 - The maximum integrated MAX4466 is an audio amplifier designed specifically for electrostatic microphones. This amplifier is used in breakout boards, which amplify the the audio of the electrostatic microphone embedded in it, which provides a 3 V to 5 V output depending on how much voltage is added to the input port. This amplifier also has a rail to rail design, and only requires a positive voltage input, which is ideal for the power module design. Table 5-28 shows specifications that are ideal for the project.

Supply Voltage Range (V_{CC})	+2.4 V to +5.5 V
Common Mode Rejection Ratio (dB)	126 dB
Total Harmonic Distortion (%)	Average of 0.03%
Input Noise Voltage Density (nV/√Hz)	Average of 80 nV/√Hz
Slew Rate (V/µs)	Average of 300 mV/µs
Open-Loop Gain (dB)	95 dB (R_L = 10 kΩ)

Table 5-28: MAX4466 Specifications

The amplifier might suitable for our project as it has an ideal gain, which means one stage can be used for the design. Knowing this is used with breakout board designs, this makes it easier to use other designs as a reference for designing the amplifier to be used for the PCB. The MAX4466 is also designed to be used only for one microphone each, so this also removes the issue where the noise between multiple microphones are picked up from each other.

Texas Instruments TL072 - After reviewing the MAX4466, the amplifier maximum gain is around 125, which is less than the gain of 140 if only one of these amplifiers is used. So, after looking through multiple amplifiers available, the TL072 is considered, as it has good specifications like the TL084, as shown in Table 5-29. This amplifier is a dual low-noise JFET operation amplifier, which has a high slew rate, high input impedance, low harmonic distortion percentage of 0.003%, and also has a low noise voltage density.

Supply Voltage Range (V_{CC})	±5 V to ±15 V
Maximum Input Voltage (V_I)	±15 V
Common Mode Rejection Ratio (dB)	Average of 100 dB
Total Harmonic Distortion (%)	Average of 0.003%
Input Noise Voltage Density (nV/√Hz)	Average of 18 nV/√Hz
Slew Rate (V/μs)	Average of 13 V/μs
Input Resistance (Ω)	10^{12} Ω
Amount of Amplifiers	2

Table 5-29: TL072 Specifications

Given this is also a dual operational amplifier, this means that for one microphone, both amplifiers on each TL072 can be used for one microphone, instead of two microphone like the TL084, which could cause interference with both of them potentially. The operational amplifier however still requires a positive and negative supply voltage, but this is not an issue as much as it thought it would be, as a virtual ground setup can be used to supply voltage, as shown in Figure 5-28.

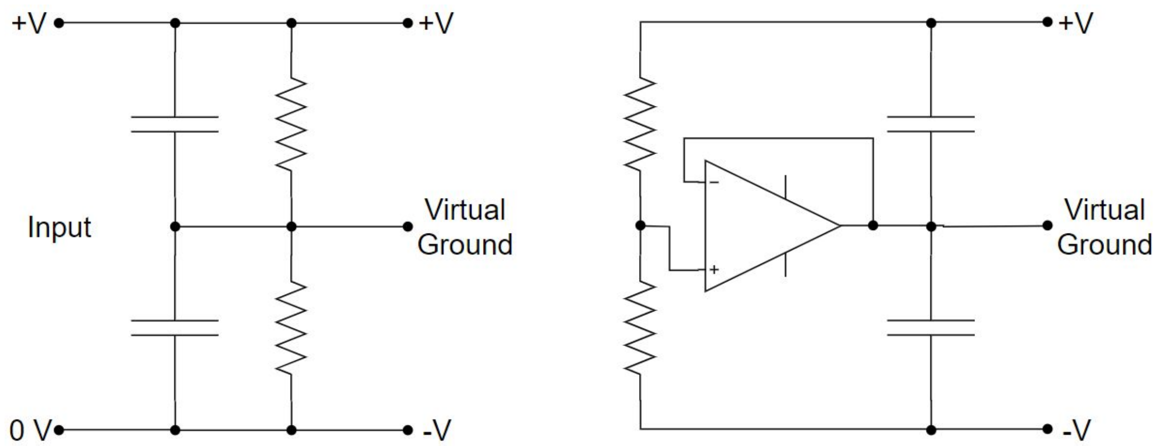

Figure 5-28: Passive (left) and Active (right) single rail supply

In Figure 5-28, passive design is a simple circuit that uses a voltage divider with resistors of equal resistance, which split the voltage supplied in half for the virtual ground. Using the virtual ground as the reference, +V would be half the positive voltage of the supply voltage, while -V is the negative half. The passive design however has drawbacks, in which DC load can change the offset of the virtual ground, so the value of +V and -V would not be the same opposite voltages from each other. So, using an active design, a unity gain buffer is used, which reduces the DC current load, and keeps the values of +V and -V the same when other devices are used along with it.

5.2.8.1 Filter

Since Nyquist's sampling theorem needs to be satisfied, a filter is needed so the desired bandwidth is achieved. Initially, this would be a filter of a range of 300 Hz to 5 kHz, however designing a bandpass is not really necessary, so only a low pass filter is needed. This filter is also known as an anti-aliasing filter, which is designed to reduce aliasing in analog to digital conversion. An example of aliasing is when a bandwidth of 5 kHz is used, but the frequency is 6 kHz. This frequency would be sampled to 1 kHz, which then causes aliasing, in which high frequencies cannot be distinguished from low frequencies. By using an anti-aliasing filter, the frequencies higher than the cutoff frequency would be attenuated, which reduces the gain and prevents aliasing.

The options for using a low pass filter involve using a passive design or an active design. A passive design just uses a resistor connected to the voltage output and a capacitor after the resistor connected to ground. This has the high frequencies go to ground, so only low frequency signals pass through. For a first order low pass, the equation of $f_c = 2\pi RC$ is used, where given the known cutoff and the known value of the capacitor or resistor chosen, the value of the other part can be determined. However, this is not the case, as when designing a passive low pass filter, input impedance,and output impedance need to be considered, which makes designing passive filters more complicated the higher the order is. By using an active filter, this utilizes an operational amplifier which can be used to provide high input impedance and low output impedance. There are also calculators available online which make designing active low pass filters easier.

Since designing active low pass filters are easier, the filter would be designed this way with an order that is desirable for our project. By using a fourth or fifth order filter, two operational amplifiers can be used, which can also provide gain as well. If fifth order is used, then a first order low pass can be added before the first operational amplifier. Given this, it is possible to design the amplifier and also have the frequency cutoff designed on the same PCB used for each microphone. In order for the gunshots to be compared easily, it will be desirable to have the frequency response flat as possible, so a Butterworth filter is needed for our design, as it fulfills this criteria.

The circuit would be designed using a Sallen-Key configuration, where each stage is a second order low pass filter, which combine would become a fourth order filter. Figure 5-29 shown below shows the frequency response of low pass filters with different orders. A first order low pass has a drop-off of 20 dB/decade, while a fourth order has a drop-off of 80 dB/decade and a fifth order has a drop-off of 100 dB/decade. Given the higher the order, the drop-off becomes sharper and allows more low frequencies to be passed through the ADC compared to a first order filter. A fourth order filter is desirable, since having the drop-off being 80 dB/decade is more than enough to filter out unwanted high frequency waves and allows fewer parts to be used, which simplifies the PCB design compared to a sixth order filter that would take up more space.

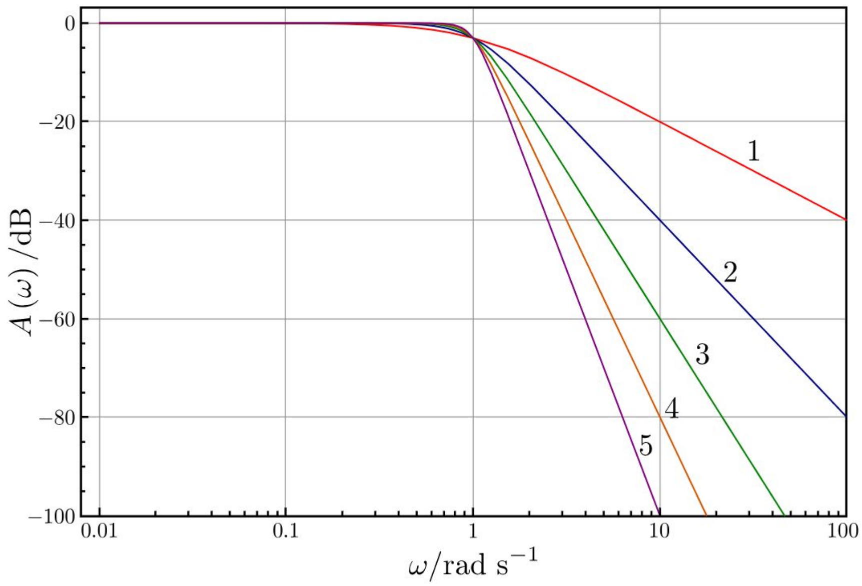

Figure 5-29: Frequency response for low pass filter with different orders

In order to make the filter design easier, an online calculator was used to determine the resistor and capacitor values used [27]. The circuit was designed to be a Butterworth filter such that the passband is flat for the frequency response. The cutoff frequency was chosen to be 5 kHz, which is the new bandwidth of the microphone, instead of 4.7 kHz, as a bandpass filter is not necessary. The gain of each stage was then set equal to 12, so a total gain of 144 is obtained for the output. The capacitors for the calculator are chosen based on the tolerance set, such as E6 capacitors have a tolerance of 20%, which E12 capacitors has a tolerance of 10%. For the calculator, E6 capacitors were chosen, as they're the most abundant available and would work well enough for the filter design.

Using the given values from the calculator, the filter was then simulated using National Instrument's MultiSim circuit simulator. The circuit is designed using a non-inverting amplifier for each stage. The gain uses the equation of $A_V = 1 + (R_1 \div R_2)$, in which the value of R_1 from the calculator is 110 kΩ and the value of R_2 from the calculator is 10 kΩ, which gives a gain of 12 for each stage. Figure 5-30 shows the fourth order Sallen-Key Butterworth filter, and Figure 5-31 shows the amplification of the output compared to the input. The left side of Figure 5-31 has an input frequency of 2 kHz with a peak amplitude of 10 mV. The right side of Figure 5-31 has an input frequency of 15 kHz, which has an attenuated amplitude which is close to the original input amplitude, showing the filter is working as intended.

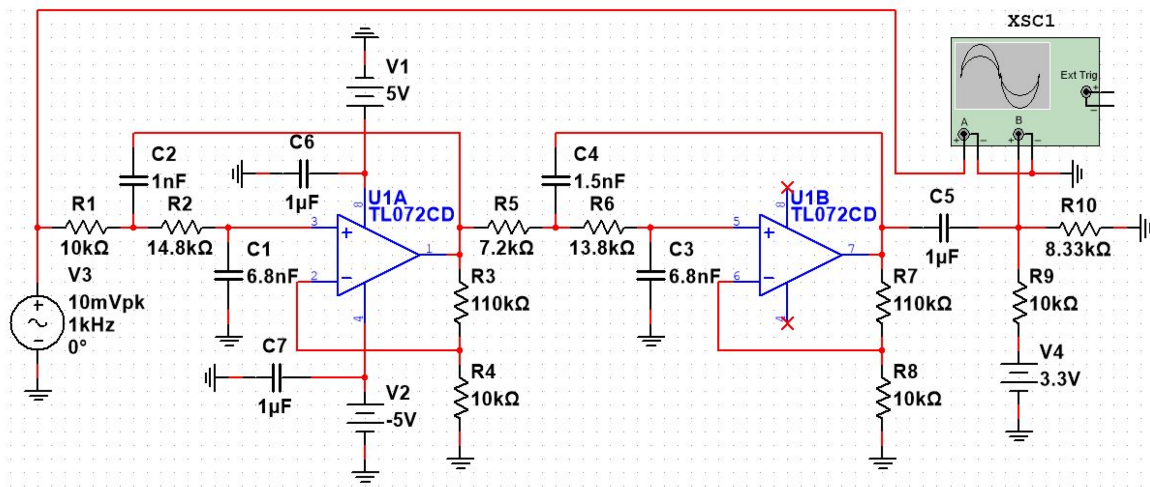

Figure 5-30: Microphone 4th order filter

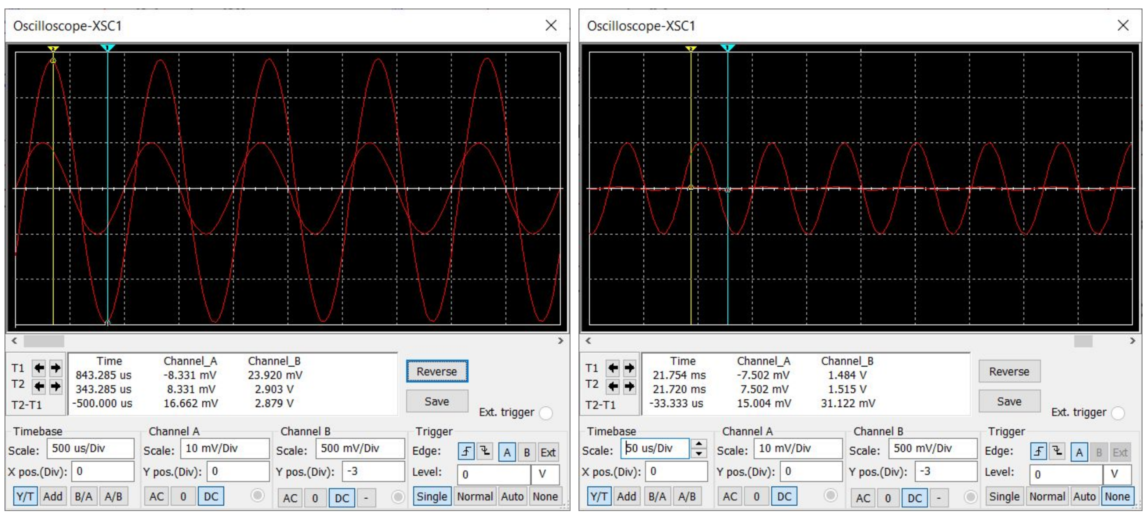

Figure 5-31: Microphone filter amplitude 2 kHz input (left) and 15 kHz input (right)

The filter would be designed to utilize a single supply voltage source that uses a virtual ground. Another either use another design where the input is offset to work only in positive voltage, while the negative supply of the amplifier will be attached to ground. The amplifiers also utilize coupling capacitors similar to the older circuit design for the amplifier, which is C6 and C7. Resistors of R3, R4, R7, and R8 are used to for the gain of each stage, which supplies 12 for each one, while the remaining resistors and capacitors are used for the second order low pass filter of each stage, which when combined together would form the fourth order filter. Figure 5-29 also shows the frequency response and phase response of the fourth order filter.

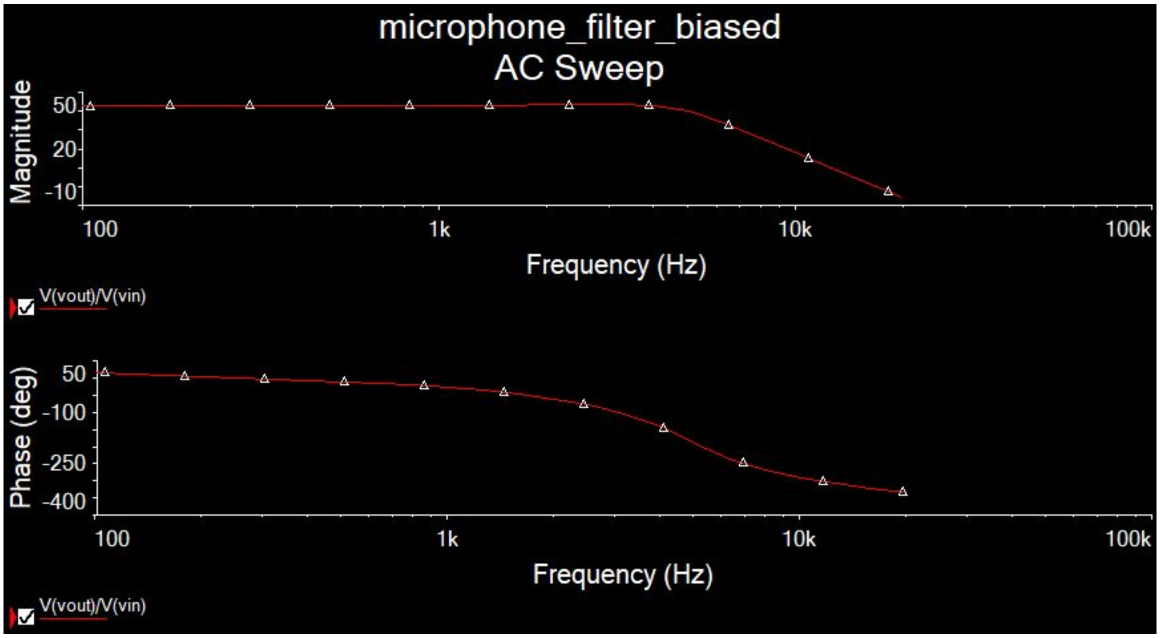

Figure 5-32: Microphone 4th order filter frequency response

5.3 Software

For this project, there are many varying programming languages used such as C++, JAVA, Verilog, embedded Linux which will be very important in designing the android app, GPS creation, etc. Table 5-30 shows the software specifications of the features.

Feature	**Specifications**
Peak Detection	This feature will be conducted with C programming language
Triangulation & Sound Recognition	C++ language will be used along with a Linux influenced system(if not a Windows OS)
Operating System	Linux will be the used Operating System that runs the main processing unit
Wifi Transfer	Preferably using C code that is used from Arduino

Table 5-30: Software Specifications

The software aspect may not be included in the physical creation of our project, it still steps in as an integral part to be able to make each piece of the puzzle work or function properly because, without software implementation, the device is basically paper weight. The software and hardware sections for the project vary in difficulty whether it comes to creating the necessary power modules or finding out how to properly manipulate to varying parts or whether it may be figuring out the suitable program to run using Arduino based language into an Arduino Uno board which is also very hefty in detail because setting up and creating a feasible and efficient piece of program takes time to mold into a smooth working process for our audio system to work properly.

While the first set of prototypes or even the initial hardware design is being made, it is imperative, there must be a set in stone decided on algorithms that coincide with the hardware being made and has to be tested in its compatibility with the architecture being built in parallel. The constant modification of the source code goes well with the subsequent constant change in design externally. Something like having an operating system as intimidating as Linux, with software implementation, it is easy for the compiler to transmit some language like c programming into assembly code, Linux will have a hard time figuring out where to place in the memory these custom designs will reside.

It is with intense priority that the software testing must be done at each stage of development for all the hardware parts that is being planned at a specific span of time when implementing the blueprint. One of the most critical points in making software code is its reliability. Without reliability, everything that has been done by the team will be for nothing because if a program crashes and ruins the flow of all the implementation that’s been inserted into the sensors, all code must be redone in a careful manner. In the way of testing the code, there will be pre-recorded input samples that will be inserted to the already made algorithms in which we will know if the source code holds up to its known standards. With the built in User Interface created for the ease of the user, it has to be handled with care and also be tested to see if it can coexist with all the other factors as a union. If one aspect fails, there will be much forethought consequences that, not only the team will face, but will have to resort into restarting a specific field of the creation process. One stage of the development cannot be finalized unless all important source code and software is implemented concretely.

When it comes to transmitting every software fully tested, there will be expected alterations in which there might be required changes by the customer for a better flow of events in regards to the transition of each stage of the process. Our primary objective in this project is to optimize well thought code into our Minuteman architecture, without any errors that may arise. Faster access times is crucial in getting the most optimized final design as possible when the time comes. There will be multiple testing modes when attempting to get ranges and locations for the final product which will take into account environmental variables such as surrounding noise and temperature peaks in a specific setting.

5.3.1 Development Tools

Regarding writing software, it's very critical to take the advantages that every tool offers. We will be using applications such as an IDE or VCS to build our source code for the software implementation part of our project.

An Integrated Development Environment is an application that consists of a simple text/code editor, a code compiler and a handy debugger to debug all the flaws of the source code. Applications like the Arduino software that will be used for the project is built in with Arduino related products. This type of IDE is open source which means that everyone gets a good grasp of what is going on when the code is being inserted into the application and the software development community is free to add in add-ons that can certainly aid in making the process easier for developers.

VCS of Version Control System is a tool that is made by Git. It is a system that manages changes to files over a span of time in that way it is possible to recall previous versions of the project later. With this mechanism, it speeds up the runtime just in case there is a mistake in producing source code where users can backtrack into a version that wasn't stained yet and refresh because of the notion that there are backups available for use even if the previous versions were used months ago. The use of the version control system improves team collaboration in the sense that people sharing ideas to make the code run more efficiently will come in very handy. The repositories are stored in a cloud system where files can be cloned to their local hub so that the original files will not get touched and will be available to be accessed by other members of the team.

One useful tool when it comes to developing software code is an application called Visual Studio Code. It is a Windows accompanied software that has many benefits in terms of ease and accessibility for the developers. With Visual Studio Code it can illustrate a specific folder a developer is managing and bring to life by listing down all the various directories and files that may be embedded into a certain project. There is a side tab that shows exactly all the files you need to access when creating your code and you have access into creating new folders and new files that may be necessary in building your program. Another aspect of VSC is the embedded command terminal where you can run JS, Python, C programming codes straight from the application. This is one of the main tools we aspire to use for our Minuteman system because of its simple nature and uncomplicated terms of use. One major factor that separates the VSC program is the git command being included already in the system. With the git command, you can easily push commits into your repository somewhere like Github which is the best in the world when it comes to a stable version control system wherein it detects who is doing what so as to avoid stepping over each other. The user has the choice of staging the code where he/she can set up a new version of the file or revert to the older version. This makes accessibility very easy for our team because everyone knows exactly which aspects of the project gets updates and which still need work.

Another development tool that is vital for our project is the MATLAB application. MATLAB is a useful resource that has the function to design in scientific and computing in engineering aspects. MATLAB can run with C, JAVA, C#, Python, SQL, and many more programming languages out there. MATLAB helps to prototype code into C code for simplicity which can work in real-time operating systems. In relation to our Minuteman project, this is mainly going be used in digital signal processing testing to test different signals for our audio sensors. It allows to be a simulation for adaptive filtering and also be able to act like noise algorithms that can be assured with efficient work and minimal latency. This kind of tool can assist in the process of testing and simulate processes needed by the user where it particularly analyzes the algorithm of noise cancellation sequences before it transmits to the processor itself.

The last candidate for a developmental tool is Eagle. The main function of Eagle is it is used to design a custom PCB that strives with a modular approach that insists in delineating every aspect of the PCB. All users are provided with various number of schematics and will check all the schematics for any possible errors that may pop up when testing. The Eagle user interface helps everyone in the team in regards to design the circuit board with smooth progress which can then be implemented to the project.

5.3.2 Embedded Systems Design

For our Minuteman design, it consists of connected components that work as a unit. We have the power module, the UI module, the sensor system, and the central module which consists of a set of microcontrollers.The custom board that we will be implementing serves as the bridge between the audio sensors and the UI module that our software developers will be working on. It will basically process the necessary data that is accepted through the Audio sensors which will transmit this critical information into the User Interface. There will be a clear signal where the location of the gunshot has been sourced from inside the UI that will alert the user specifically.

To be able to fully access the right data from its buffers, our software has to have clean processing while it is being read from the UI. From the board, it will read the time frame that a gunshot has occurred and, using a specified counting system that will access the max value that each of the microphones pick up during the event.

One of the best ways to reducing the memory time is using multithreading that accompanies the data retrieval and to also reduce the processing stage of the microphones picking up sound from the gunshots. There will be an alternate process in which the buffers be accessed. While there can only be a maximum of two arrays being accessed, one of the processes will wait for the other to retrieve the time frame needed for the data. We plan to halve the tasks between threads so while a specific thread is in the processing stage, the other one will be accessing the sample time frame for the other microphones that need data as well.

For the Audio sensor system, this will be where the sampling and then passed into a conversion function reside. All the microphones that are laid out in an arbitrary location has its output signal relayed into our Field Programmable Gate Array (FPGA) that is inserted into our custom board mechanism.

For the custom board, the microcontroller runs Linux that is put into a mode of low power, which is key to observing one of customer's requirements in power consumption. If the main system has unnecessary power usage, it won't be looked at as much of a profitable product for the consumers. Having low power consumption is much better because it will definitely help users in keeping their usage in check wherever they might be using the device. When the interrupt gets received by the Audio module, the Operating System (Linux) will then fork the threads for algorithms in our software such as the recognition of the gunshots and the algorithm of triangulation which picks up the max value of a signal produced by a weapon.

Next is the User Interface (UI), it is where the data is fed through that gives up an output alerting the user the specifics of the gunshot sounds being heard by the microphone or audio sensors. Bluetooth will come in handy for this module because our Minuteman design is designated for components to be in range of one another and not send long range transmitted signals to even farther distanced locations. The purpose of the User Interface is to filter out duplicate signals, which is then next forwarded to reliable connections afterwards. After that, there will be an alert displaying for the user to check the interface itself for the required data/ information. Upon entering the interface, there will be a designated map that will be seen coming from a node which sent the alert and the location where the gunshot occurred relative to the node. Every event recorded in the UI will have a date and time so it can be sorted out pretty easily later for future use. It is with imperative discussion that the UI be reliable and it be unidirectional wherein changes to the sensors will be made through the system directly.

5.3.3 Arduino Uno Programming

Our team will be utilizing the power of an Arduino Uno board that will serve as the bridge to our Minuteman system to the PC data. All the calculations being done (i.e the difference in peaks between the microphones) will be sent through Wifi that is then connected to a desktop PC using a USB drive. For our custom board, it will be using a C programming code compiled by the specific Arduino software that is already provided beforehand.

In this piece of code, it will be listening for audio signatures, which goes to the capture stage and then transmits that certain data over Wifi connection (Or Ethernet possibly) to another Arduino board and program connected to the PC. It will be utilizing the algorithm used for peak detection wherein the Wifi link is the transfer sequence that runs the procedure. While it is noted that Wifi will be used to transmit certain data, it is also possible that going with a wired approach might be a more suitable way to fully extenuate the data needed to be able to make the sensors work properly. With Wifi connection, there may be problems like delay in data transfer or the reliability of a wireless connection may be unknown and patchy at times unlike the safe nature of a wired network.

5.3.4 Operating System

For the creation of this project, we decided on using a Linux influenced system as the go to for development and we figured that it would be really beneficial because of its astute access and portability. With a Linux influenced system, it has the right benefits that can produce maximum effort from all of the pieces needed to create the necessary parts for our apparatus. With Linux, we can always modify components really easily or even have the ease of removing these components without any problems at all. There is a good chance that adding in the thought of Ubuntu will be in play in the future since Ubuntu is a popular mode of Linux.

In the boot process for our system, it will be sourced from a processor ROM. This will lead into the loading mechanism for the operating system which will then create the operation. When receiving a certain interrupt, the Linux process will begin to stay in Kernel to execute a wanted function to be placed into a slot in the internal memory which will exercised later on. Two main details will be accredited in running this operating system: going into mode of gunshot recognition and the detection of the location of the gunshots.

The user interface comes into play when getting sent information is ran by a sleeping thread until it is woken up by the conclusion of both the gun recognition and the detection location of the said gunshots. This process coincides with the communication in between alerts being given to the users and the interface itself. It then sends packets which is sent using bluetooth that contains the location of the gunshots, the time the gunshots were fired, and the type of gunfire that was used for people to see and determine what really cost the specific fire of a weapon.

We have decided on not using a mobile application as it will give extra duties for the group which will potentially hinder the performance of the group as a whole so we can just focus on getting the necessary information out to the user interface in the way of a simple alert mechanism just to let the users know the critical information they need. In that way, the person trying to use our device will fully know the cause of an unexpected event.

There were also different substitutes into which operating system we will be using for this matter. One of the substitutes that we had in mind was a Windows OS. But because of the notion of having Linux as our primary source, we decided to let go of the thought of a Windows OS because of its more complicated nature and stance as opposed to the
Linux' diverse arsenal of subsystems that provide more convenience to users that wish to get their work done in a more straightforward way.

5.3.5 Front-end and Back-end Web Development

There are many ways to be able to work a successful front end development system when it comes to a project and make sure it looks endearing and aesthetically beautiful in terms of functionality and use. Looking at the plethora of options being offered in the world of web development, there's definitely those options that stand out amongst the rest especially when it relates to conveniency and easy to access or to develop for that matter. Front end development encloses its system with a defined structure in which it comprises of different factors that make up the whole flow of creating a front end for a website. Before anything else, there has to be clear planning and careful decision making for how we have to build the front end of our team website. It is required to have an organized look that reflects how organized the team is working and how it functions as a group. A messy presentation wherein shows everyone's work and what the group is about presents a bad signal to the possible customers that may want to know about our team and possibly produce disappointment on their end. A clean, proper, and attractive domain for the team is one of the top goals that we are willing to fully exhibit to everyone who might want to discover this group.

Our planned design for the front end of our website consists of a simple yet upright look that is good to the eye and also easy to access different tabs that may also consists of necessary files that readers wish to access that won't be complicated to navigate into because that is a bad practice for making a stable domain for a certain agenda. There will be a planned color scheme that is both attractive to look at and that is not stress inducing to the user. The navigation component consists of various tabs that can bring the user to different parts of the website like the brief introduction of the team and what each member is all about. Another tab should be for the different documentations that the team is required to show for future reference by every user that may come across our website. It should be listed in a nice order and presentable to everyone that has good titling and make sure we don't attach the wrong link to the titles. There will be photos of each of the team members that show who we are and what we intend to do once we graduate and what our affiliated majors are. In this way, for the people who are reading our domain, they will know that we are serious about our craft in order to fulfill our goals as engineers whichever road we may enter in the future. Another tab will describe our project and our plan, design, and implementation stages. There will be sections that present images that show schematics and initial design about our Minuteman system and also information that may provide knowledge about the motives for our project and how it could benefit users that want to use our product. It should be shown in order from initial discussions to the development cycles and stages to the actual implementation of the different parts as a unit.

For our front end to work properly we need to know the right components that structures a fully functional web developed system. First, we have the makeup of HTML. HTML, or Hyper Text Markup Language, is the backbone of making one front end layout and development. HTML has the tendency to have text to consist links, some termed hyperlinks, that is usually embedded in the design. It has various hyperlinks in when the user clicks in the hyperlink it will navigate to another page. This markup language has different entry points for developers such as uploading source images, links, tables, and more access points that is relatively presented to the user. It can be modified depending on how complex the developer wants it to be. The main framework for this section of the front end is the code itself. HTML code is fairly easy to create and compile because of how simple the markup language really is. It doesn't take up much thought on using different tags that create different parts of the main look of the front end. Most of the thought process comes to the deciding how the front end will look from the initial framing and blueprints because that decides the inevitable fate of the finished product of the domain. There have been many versions of the HTML throughout history of the internet. The most recent version of the HTML is HTML5. The main update is the more feasible use of audio and video files in implementing for website creation.

Next, we have the assistant to the HTML application, CSS or Cascading Style Sheets. This consists of the different layout designs that incorporate with the HTML aspect of the front end. CSS is used mainly for different layouts in buttons, hyperlinks, fonts, tables, etc. This is utilized to make the website look better for the human eye. Every website on the internet has a unique look that separates each of these websites from another and that's why CSS is a vital aspect in making one. CSS serves as the focus for making a domain look good in terms of critical details that entail a domain whether it might be a text box, text input, screen size for the web browser used, resolution and smaller facets of the bigger picture such as perfect margin, alignments, and consistent theme throughout the whole domain. There are many possible combinations of designs that have been implemented when it comes to style sheets and it all depends on the creator on how the website is going to look like. These layouts are what the readers are going to see first. Almost none of the visitors visiting websites will carefully judge the functionality but always would see the website layout first because that is exactly what everyone sees firsts in these pages.

Next, we have the functionality side of the spectrum, Javascript. The main functionality tool used in creating front end development is the javascript implementation. Javascript is one of the most popular source languages in the world of software engineering. It is popular because of its utility and major source of reliability for developers around the world. A popular use for Javascript is the use of Document Object Model (DOM) that is provided by the HTML standard that usually manipulates website pages to traverse to the other pages which correlates to different events made up by the developer. Javascript code has the tendency to also retrieve various substance from the source web which can also react to events on the server side as well which creates a dynamic experience for both the developer and the user.

Lastly, we have the backbone of the backend process of the web development and that is the API or Application Programming Interface. The API is the main focus for backend developing as this is the connection of the server of the internet back and forth to the client's side of the spectrum. The API is a good source of inputting security into a website because of potential viruses and attacks that can possibly ruin one's domain. Basically a good analogy of the works of API is that when someone goes to a restaurant and orders certain kinds of food. The main point here is that the customer does not care about how the food is prepared as long as the food is able to be served in the end. This is similar to how API functions. API is a list of functions that one developer can utilize from time to time that provides a description of what they do. This helps the developer determine what functions to use particularly for the website and how it can relate to security that stabilizes the website entirely. What the API basically offers is the surmountable amount of ways a particular web development can be done. This kind of process saves time for the developers that takes advantage of the implementation of the program to do the special work. The best part is that this also helps reduce the code able to be created by the developers as they already have a plethora of resources presented to them.

For example, if you want the browser to show one or more consecutive pages, it doesn't necessarily require you to create your web browser program from the start as this creates a more time consuming environment. You can use the WKWebView API which is a type of API that embeds a browser object in your main application. Another example is capturing pictures or video for your project. Instead of creating your own interface for capturing these, you can always utilize the camera API that embeds a smartphone built-in camera system for the project. Every platform that is included in the world API, has a exclusive related API program for it. That's what makes API so useful because of the conveniency that it produces for the people who don't wanna get in a hassle trying to create their own programs which will eventually a longer time to do and would possibly irate developers. In this way, it's a community derived solution for long term problems that may arise for developers who are starting to create their own project in the interwebs.

One aspect is web development that strictly connects to the API is JSON or Javascript Object Notation which is a way of representing data which look like Javascript Objects. JSON can be seen in nearly all of the modern web applications. The best part is that it is very clear in what it intends to do and works really well with programs written in JavaScript code.

5.3.6 Determining Direction and Distance of Gunshots

After an audio sample is successfully classified, some additional information needs to be determined. For one, the direction and distance of the gunshot from the sensor. Distance is determined through the time difference between microphones on the sensor. This is done by determining the number of samples picked up between the microphones using the peak amplitude of the samples.

X and Y coordinates with respect to the sensor are then created by the multilateration equation derived in the research section. As the sensor has a compass, this can be used to adjust the coordinates with respect to due North. The final product is then converted into GPS coordinates by comparing the coordinates generated to the coordinates of the sensor when it recorded the sample.

5.3.7 Gunshot Classification through Machine Learning

Identifying gunshots is an important task for our software. Once a file is received from the sensor, it must be able to confirm the presence of a gunshot in the sample. While alternative software and hardware options are available for detecting gunshots are available, however, for this project we will seek a machine learning solution to this aspect of the project.

The noise level of firearm discharge is already a very apparent means of detecting a gunshot, as not many sounds generate such a high amplitude. While this isn't enough on it's own to detect a gunshot, by having the sensor filter out events with less amplitudes, it already eliminates many sounds that aren't gunshots.

However, additional confirmation is necessary for this process. As discovered in our research, supersonic projectiles produce a shockwave that follows in a conical pattern behind the projectile as passes through its trajectory. This produces a sharp, "N" shaped waveform that is depicted in the audio amplitude. There are a number of reasonable concerns in using this alone. Obstacles can obstruct the shockwave from reaching the microphone and distance will reduce the N-wave.

This means analysis of the muzzle blast is necessary, and it will be the primary thing to look for with our neural network. However, it will greatly add to the accuracy of the network if it can detect both the muzzle blast and supersonic N-wave.

Pytorch already offers a wealth of tools for machine learning, including fastAI, which builds on top of pytorch to easily produce high accuracy neural networks. In order to establish a working neural network, we will need to train one. This is done separately from the final product, as it is time consuming and resource intensive.

While training a learning algorithm is difficult and time consuming, the final model can be collected from the training as a file containing each layer and their respective weights. This can be extracted and established in the actual program when it is started on the host computer.

So long as this model has a sufficiently fast runtime and high classification accuracy for each sample, it will work sufficiently. This also means that the network can be easily updated later by simply replacing this file.

5.3.8 Gathering a Dataset

Given the nature of classification through artificial neural networks, and the required accuracy for this project, the ANN will need to be given supervised training before use. This requires a large dataset of samples that are accurately labeled. The ANN will be able to train by classifying each sample and comparing the results with the ground truth labels.

Freely available datasets for training are difficult to obtain. This is because the process of manually recording and labelling a sufficiently large enough dataset of samples for the ANN is expensive in both time and resources. Each sample has to be manually recorded and labelled, and done so correctly so as not to deceive the ANN.

There is at least one, publicly available dataset available on the internet. The Mivia Audio Events Dataset has 6,000 recorded events, consisting of glass breaking, screams, and gunshots. They are recorded at different translations (different angles, distances, elevation), and are recorded with background ambience and sounds that add variation [24].

While such a dataset would be perfect for this project, unfortunately differences in hardware and file specifications challenge the notion that this can be used to reliably train the ANN. Specifically, the dataset consists of PCM audio files with sample resolutions of 16-bits [24]. The sensors used in our project record 12-bit ADC samples.

Therefore, it is a necessity for this project to collect our own samples for use in ANN training. A safe gun range with an appropriate backdrop (a strong physical barrier such as a hill to stop the ammunition) and isolation will be selected to fire the weapons and collect the audio samples. A range of rifles, handguns, and shotguns are to be tested with appropriate ammunition. Preferably with numerous different guns and ammunition calibers between them. Samples of the gunshots are to be taken from the same microphone sensors used in our design.

Samples will be taken from a number of different positions, varying from angle and distance. At least two samples will be recorded from these positions. This is done for each weapon tested as well. Additional samples involving loud noises, such as glass breaking and screaming will also be collected and added to the dataset. After collecting the samples, each of them will be manually evaluated and labelled with a ground truth.

While the neural network is not needed to detect the exact angle and distance, it is required to classify the gunshots accurately, determine their type (handgun, rifle, shotgun), and filter out non-gunshot sounds. It also needs to be able to make these classifications regardless of translation. That is why so many samples need to be taken, with so much variation between them.

5.3.9 Dataloading

Like someone would for analyzing an image with a neural network, the information of the file needs to be manipulated into a form the network can use. The process of extracting and manipulating this data is called dataloading.

To extract the dataset, the audio files will be documented by a csv file, that will outline the file name and label for that file. Two of these are created for the purposes of training and testing. During either mode, the program will go through each CSV, identifying the file that needs to be loaded as well as the ground truth label which is used later. The file is then loaded by the program and passed through analysis.

Audio files consist of samples, taken over time, that provide a simple number representing the amplitude taken at that sample. The number of samples gathered is determined by the sample rate. This information can be translated into frequency information, and a format that allows for frequency-time analysis in the form of an image.

In this image, each pixel value, similar to a grayscale image, depicts the amplitude. These amplitudes are arranged in windows of frequency, each windowing corresponding to a certain set in time. As stated in the research, short-time fourier transform is limited in that it is unable to adapt window size to display the appropriate frequency or time resolution. Instead, however, this can be done with wavelet transform.

The wavelet transform only requires only two parameters and a matching wavelet function. A number of libraries already contain wavelet transform functions. All that would be required is adjusting the parameters.

5.3.10 Training and Testing the Model

A special program will create, train, and test the model we create, but the final program will only be using the final model generated by this program. Testing both programs later will be necessary to ensure accuracy is maintained. The training program will include a data loading file, that will extract and load the data for each batch. A file that will manage the training and testing of training batches, and a And a main file will train and test the model created using stochastic gradient descent.

The main file will start by extracting the necessary data from the audio samples stated earlier into a batch. It will then create the model by creating an object of the model using the class file mentioned earlier. It will take the audio data and pass it through the model, evaluating the accuracy until the entire batch is done. Finally, the clip is classified by a voting scheme. This voting scheme will most likely be done via probability voting. It will take the average accuracy of the whole batch. Then it will run the network through optimization. We will use stochastic gradient descent for optimization. This process repeats for as many epochs as is given.

6. Integration and Testing

6.1 Integration

Integration describes how the various components of the project comes together. For hardware, this describes how the various hardware components are connected. For software it describes how various functions and algorithms work together.

6.1.1 Software Integration

There will be various algorithms being implemented into the initial, current, and final design. One of the most important algorithms sought out for in our Minuteman device is the location and gunshot recognition. These algorithms will be entertained in a MATLAB software using sound that was entirely pre recorded. We have decided to use C++ language to fully extract each of these algorithms capabilities. There will be an SD card that will be used to boot the main system. The decided algorithms will need to be carefully tested and made sure that each of these work properly. Once they are determined to be working with no flaws, they will then be integrated into a linux machine. The purpose of this stage is to ramp up the process of the microcontroller being able to perform parallel data processing that is required on the definite hardware setup. For the Bluetooth and GPS aspects of the design, they will have respective drivers that will essentially be developed in parallel.

All this will be then transmitted into a custom PCB board that is specially customized just for this project. At this stage of the design, there should be expected bugs that may occur that will require fixing on the software end which is naturally meant to happen in an environment such as this. The user interface should be technically completed once all the previous steps are done and the only remaining step is to test the communication between the main Minuteman system and the user interface itself. This will include more beforehand tests that will make sure the user interface and the GPS proceeds through smoothly as though this will be an initial stage of these sets of testing.

With the use of microcontrollers, there will be a nuisance of slightly small bugs because of the difference in reading between these two. In the conclusion of this Minuteman design, there will be field testing in a goal of reproducing signals from a audio sensor to determine if the location origin is precise.

6.2 Testing

Testing each component of our project is important for ensuring that the entire system works. Due to the nature of the task put upon our design reliability is important. Failure could mean false alarms, misdirecting authorities to the source of a shooting, or worse, failing to identify and alert authorities of the incident at all. Hardware needs to be able to work at expected, and software must be accurate.

6.2.1 Hardware

The testing section is important to be detailed. Knowing that each individual component will have its own individual set of failure modes and failure rates can cause issues for the design of the system and any troubleshooting that will take place. This device is a prototype and proof of concept product. This means it is different in the how it is produced, since it will never reach a full product lifetime. Compared to how products are created in the business world, the proof of concept and design would be just one part of the initial processes in the development stage. Testing the system would be considered in the design, but not implemented until hardware integration and occurs when software can be integrated as well. The initial testing will look for hardware fidelity and individual component design. This mean each component or section would most likely be tested with the requirements that would be used in the design of the circuit.

6.2.2 Installation

Each individual component shall be tested to make sure that they work before any initial integration. The individual component will have a power test, where it will be powered with the desired voltage. This type of testing will mitigate any manufacturer faults that could occur in our project. This will also help in troubleshooting, since any initial issues would not be from the individual components but caused by faults in the design or integration.

After individual component testing of the hardware, it will be important to test the integrated hardware as well. This means, each individual section of the PCB should be tested before implementation. The power supply should have all of its parts on a breadboard and then tested at certain points to make sure that desired voltages and currents are supplied. It is important to test the sensor module as well, so that if any sensors are different from what is expected, it can be redesigned. The microcontroller section shall have similar implementation where after initial fidelity testing, it should be supplied with the voltage required and tested to see if any fault in the design occur. For the central module, it will be important that the other two modules are able to correctly provide the required functions. This is important, because the central module will process any information that comes from the sensor module and deliver that with the required specification to the software and application section. By separating this from the software testing, it would provide more knowledge of any faults if it lies within the hardware design.

6.2.2.1 Component

The individual component shall be tested for any manufacturing errors. These test will make sure that stated results either in the circuit design or datasheet are the results that are received. The table below shows each of the component test and the results.

Component/Function	Testing	Operating Result
Arduino Due	12VDC adaptor supplied to the Arduino Due input power port.	Operating correctly within datasheet direction and no deviation of desired results.
AT91SAM3X8E	Shall apply 3.3VDC & 1.8VDC.	Waiting for delivery.
CMC-6035-130T	Tested with 3.3VDC power supply with no fault. Shall test for integration.	No issues for initial component testing, waiting for further integration for more testing.
BNO055 9 DOF Absolute Orientation IMU Fusion Breakout Board	Shall apply 3.3VDC.	Waiting for delivery.

NEO-6 u-blox 6 GPS Modules	Tested with 3.3VDC power supply with no fault. Shall test for integration.	No issues for initial component testing, waiting for further integration for more testing.
Maxim Integrated DS18B20	Tested with 3.3VDC power supply with no fault. Shall test for integration.	No issues for initial component testing, waiting for further integration for more testing.
Power Adapter Supply	The value should be 12VDC	All three adaptors were within 12VDC +/-5%
Analog Device LTM4622A	Shall apply 12VDC	Waiting for delivery.
Maxim Integrated MAX1044	Shall apply 5VDC	Waiting for delivery.
Texas Instruments TL072	Tested with +/-5VDC power supply with no fault. Shall test for integration.	No issues for initial component testing, need further simulations.

Table 6-1: Testing table

The only device that still needs to be finished, is the design of the AT91SAM3X8E. This finished design includes integration of all the sensors and component. This is the only part that needs to be completed before the PCB design can be finalized and desgined.

6.2.2 Software

There are a number of tasks the software must complete in order to be satisfactory. It must be installable onto a computer that is connected to the sensor network. It must be able to communicate with the sensors and receive an audio sample. Then it must be able to filter sample from events that aren't gunshots. It must run in the background, presenting the user with a UI interface only to change settings or connect to new systems. All of this can be tested in real time with live samples in coordination with the hardware, and through direct inputs.

6.2.2.1 Installation

The user must be able to install the software onto a typical computer, or a computer intended for monitoring security. The installation should be simple and easy to do.

1. The user or individual in charge of setting it up should be able to easily download the software installer from the internet.
2. Upon execution, the installer will present the user with the option to choose the installation path, recommending a fast hard-drive or solid-state drive.
3. Upon clicking “Next”, the software should begin installation automatically.
4. Once installation is done, a final screen should notify the user that the program is installed and offer to open the program upon click “finish”.
5. Clicking “finish” should close the installer.

This is tested by simply installing the software on several computers and making sure it works from there.

6.2.2.2 Creating an account

To prevent unwanted individuals from accessing and tampering with the system, the user should be asked to create an account upon the first execution of the program. The user should be able to create a new account as well, for new personnel.

1. The user is directed to an account creation page on the program.
2. They enter a valid username and password, along with a re-typed password to confirm.
3. Upon entering, the program should validate that the account was created for the user, before proceeding to the control panel.

This is tested by seeing if the program asks the user to create an account upon first activation and doesn’t ask the user after they have already created an account upon the second activation.

6.2.2.3 Logging into an account

Once an account is created and the user has logged out of the system, they should be able to log back in through a login page. Authorities should also have a password granted to them to access the account in case of emergency.

1. The user is presented with a username and password field.
2. The user is redirected to the control panel if they enter the correct credentials.
3. The user is denied if they enter the wrong credentials.
4. The user is redirected to the control panel if they put in credentials from the authorities.

This is tested by determining if the login screen comes up upon second activation and asks the user to login. Then putting the created account username and password. A second test is done using a specially generated password for authorities.

6.2.2.4 Logging out

The program should automatically log out upon closing the control panel. The program will continue running in the background. To test this, the program can be reopened and require the user to log back in.

6.2.2.5 Connecting to the sensors

If the system is connected to the computer, the program should be able to connect to it while online. The program should be directed to connect to the sensor by the user through the control panel. To test this, the sensors will collect a random sample of sound and send it to the program.

6.2.2.6 Running in the background & detecting gunshots

The program should always run in the background, even if the control panel is not open. In the background the program should be readily accepting input from the system.

- The program should receive audio samples from sensors.
- Upon a sample the program should analyze the first sample to come to the program. Additional samples should be put into a buffer to be analyzed after.
- The program should first analyze the muzzle-blast confirming it with known samples of gunshots.
    - If it fails, it should reject the sample and test the next sample.
    - If it passes it should send an alert to authorities and listed contacts, as well as the user on the computer.
- If a sample detects a shockwave following a gunshot, use it to confirm the event and determine what kind of weapon is being used.

This is tested by closing the control panel, and providing the program with samples to simulate events arriving from the sensors. It should correctly identify each event by coming up with an alert or not coming up with an alert when there is no gunshot event.

7. Conclusion

Research into this project has drastically improved our understanding of the involved components and capabilities of the system we intend to create. Originally, we planned on creating an indoor gunshot detection system, utilizing triangularization with multiple sensors to locate the shooting. While this was realized with systems like SenseShot, it came to our understand that an indoor system was not realistically within the limitations of our capabilities. Such a system would need to compensate for the obstructions within the interior of the building, and would likely require a large number of sensors.

However, it is known that not all shootings occur indoors, and that any shooting occurring within a building can be easily detected from the outside, either from sensors mounted on the exterior of the building, or from sensors mounted onto nearby vehicles.

Our research into multilateration, and into the Raytheon Boomerang III system, also revealed that locating the shooter, and identifying what firearm was being used, did not require a large number of sensors. Instead, all that was required was one device within a specific area of operation. Like the Boomerang, this device can be mounted onto both buildings, and vehicles, such as police patrol cars. Such cars would also have a computer onboard.

Knowing that a computer in both instances can be used, led to the realization that machine learning through neural networks could be applied. While not conventional, this system could offer a high degree of accuracy in detecting gunshots. It can be readily adapted and modified, as more samples are collected, increasing accuracy. And with wavelet transform, the fidelity of the information being used by the network will be as high as it can be.

The end result is a detection system, using just one device—although several can be employed—that is not only affordable, but accurate, and can quickly be employed in situations that require it. Although a facility might not be equipped with such devices in the event that it might need it, a vehicle, possible a police patrol car, can be brought to such events.

Appendices

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Appendix B: Permissions

Arduino Due Permission

Sara Therner (Arduino)
Apr 9, 11:26 CEST

Dear Nathaniel,
Thank you for your email. Photos of boards and other Arduino references shall be used for explanatory or descriptive purposes only. Being a school project we understand that you need to be able to present your project and the knowledge you gathered during so you may use the image of DUE for your senior design project. Please note that other rules may apply in case you work with Arduino related products in the future (outside school) so it is always good to check with us first.

Best of luck with the design project.

Best Regards,

Sara Therner
Trademark & Licensing Manager
Arduino Customer Support

Nathanieldunn992
Apr 9, 02:10 CEST

Name: Nathaniel Dunn
Email: Nathanieldunn992@knights.ucf.edu
Subject: Permission

Message: Dear Arduino,

My name is Nathaniel Stence Dunn and I am in my senior year at UCF getting my Electrical Engineering degree. I am emailing you for permission to use the images below for my senior design project.

https://store.arduino.cc/usa/due
Use the first image of the Arduino Due.

Best regards,

Nathaniel Dunn

Nathanieldunn992@knights.ucf.edu