

TRIAD - Final Project Report

Course: APSC 140

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Date: December 9th, 2025

Abstract

Spatial sound localization presents a complex engineering challenge, obstructed by environmental factors such as reflection, absorption, and background noise. To address this problem, an autonomous acoustic tracking device was designed, fabricated, and tested. The primary objective of this system was to accurately determine the azimuthal direction angle of an incoming sound source, with a secondary objective of estimating the source distance. Three design concepts were evaluated with a weighted decision matrix, leading to the selection of a three-microphone omnidirectional array using a TDoA approach. The final prototype incorporated a Raspberry Pi Pico microcontroller, analog microphone amplifiers, and a short-term energy onset detector to identify transient sounds and solve for their direction. Testing showed that the system reliably produced directional estimates for transient acoustic events across repeated trials, though distance estimation and continuous-source tracking remained limited due to the small array aperture and analog front end. This report details the complete design lifecycle, from problem definition and conceptual analysis to the fabrication and testing of the final prototype.

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1. Introduction

The localization of a sound source within three-dimensional space is a non-trivial task, complicated by acoustic phenomena including multipath propagation, absorption, and signal noise. In response to these challenges, this project focuses on the development of an acoustic tracking device capable of identifying the direction of an incoming sound.

The goal is to create a device which can, as a primary objective, determine the direction to a sound source, and as a secondary objective, determine the distance to the sound source.

1.1 Client's Problem Statement

The client requires the development of an acoustic tracking device capable of detecting and indicating the direction of an incoming sound. The device must utilize at least one type of sensor to capture audio signals and process this data to determine the sound's direction, either through a mechanical or digital output system. The complete system must fit within a portable enclosure measuring 55cm x 40cm x 23cm, be easily assembled and disassembled with a maximum setup time of five minutes, and adhere to a maximum budget of 200\$. The device must be complete by the end of November.

2. Problem Definition and Objectives

2.1 Clarify Objectives

In order to define more specific parameters to guide the design process, we identified 5 distinct objectives for the device, and ranked them in descending order of importance by means of an aggregated PCC chart. The objectives the team aligned on are as follows:

- **Direction Detection**
 - The device needs to be able to detect the direction of impulse and stationary noises relative to a fixed reference point (i.e. the centre of the device).
- **Consistency**
 - The device needs to produce consistent results given identical sound inputs.
- **Durability**
 - The device needs to be durable such that it can withstand multiple assembly/disassembly cycles.
- **Distance Detection**
 - The device needs to be able to identify the distance of the sound source to a fixed reference point.
- **Cost**

- The device needs to be as inexpensive as possible while adequately achieving other objectives.

The order of importance of these objectives was determined by means of an aggregated PCC chart based on the inputs of the three team members. The three individual PCC charts are provided below:

Shygaan PCC Chart - *Shygaan's individual evaluation of the relative importance of each objective*

	Direction Detection	Consistency	Durability	Cost	Distance Detection	TOTALS
Direction Detection	***	1	1	1	1	4
Consistency	0	***	1	1	1	3
Durability	0	0	***	1	1	2
Cost	0	0	0	***	0	0
Distance Detection	0	0	0	1	***	1

Luca PCC Chart - *Luca's individual evaluation of the relative importance of each objective*

	Direction Detection	Consistency	Durability	Cost	Distance Detection	TOTALS
Direction Detection	***	1	1	1	1	4
Consistency	0	***	1	1	1	3
Durability	0	0	***	1	1	2
Cost	0	0	0	***	1	1
Distance Detection	0	0	0	0	***	0

Devon PCC Chart - *Devon's individual evaluation of the relative importance of each objective*

	Direction Detection	Consistency	Durability	Cost	Distance Detection	TOTALS
Direction	***	1	1	1	1	4

Detection						
Consistency	0	***	1	1	1	3
Durability	0	0	***	1	1	2
Cost	0	0	0	***	0	0
Distance Detection	0	0	0	1	***	1

Aggregated PCC Chart - the sum of the PCC charts of each individual member

	Direction Detection	Consistency	Durability	Cost	Distance Detection	TOTALS
Direction Detection	***	3	3	3	3	12
Consistency	0	***	3	3	3	9
Durability	0	0	***	3	3	6
Cost	0	0	0	***	1	1
Distance Detection	0	0	0	2	***	2

The aggregated PCC chart was created by adding together the values for each cell in the individual PCC charts. The totals for each objective represent the hierarchy of importance for each objective; the team importance of each objective is therefore direction detection, consistency, durability, distance detection, and cost, in descending order.

2.2 Establish Metrics for Objectives

The success of the design was evaluated according to several measurable performance criteria. Given below are our baseline criteria for success.

For direction detection, the system should achieve an angular deviation of less than 20% from the true direction of the sound source. In terms of consistency, it should demonstrate repeatability such that identical test inputs yield direction results within 20% of one another in at least four out of five trials. Additionally, the device should provide valid distance estimates within the specified tolerances for at least 70% of test cases.

The distance detection capability should provide source distance estimates within 15% accuracy for targets at ranges of 5–10 m, within 20% accuracy for ranges of 10–25 m, and within 30%

accuracy for ranges of 25–50 m. Baseline success here is defined as meeting two of the above three targets. Furthermore, distance estimation must not interfere with the accuracy or responsiveness of direction detection, with direction detection taking precedence where trade-offs arise.

In terms of durability, the device should reach full operational readiness within 5 minutes when assembled by the entire team and should withstand at least 20 complete assembly and disassembly cycles. It should also maintain reliable operation for at least 25 hours of cumulative uptime across both indoor and outdoor environments under normal weather conditions.

Finally, with respect to cost, the total material expenditure must not exceed \$200. Approximately 20% of the total budget should remain unused or be allocated to auxiliary quality-of-life improvements, such as enhanced cabling or more robust mounting structures.

The above baseline criteria are reproduced below in addition to criteria for failure, marginal success, and excellent success:

Metrics for Objective Success - *conditions for failure, marginal, baseline, and excellent success of objective metrics*

Metric	Failure	Marginal Success	Baseline Success (Target)	Excellent Success
Direction Detection	> 30% angular deviation	20% - 30% deviation	10% - 20% deviation	< 10% deviation
Consistency	< 3/5 trials repeatable or < 50% valid distance estimates	3/5 trials repeatable or 50-70% valid distance estimates	4/5 trials repeatable and 70-85% valid distance estimates	5/5 trials repeatable and > 85% valid distance estimates
Distance Detection	Fails to meet accuracy target at any range	Meets accuracy target for one of the three distance ranges	Meets accuracy targets for two of the three distance ranges	Meets or exceeds accuracy targets for all three ranges
Durability	Meets one or fewer criteria	Meets two of the three criteria	Meets all three criteria (<5 min assembly, >20 cycles, >25 hr uptime)	Exceeds all three criteria (<3 min assembly, >30 cycles, >40 hr uptime)
Cost	> \$200 budget	\$180 - \$200 budget (little to no reserve)	\$160 - \$180 budget (~10-20% reserve)	< \$160 budget (>20% reserve)

2.3 Identify Constraints

The client has identified a number of specific constraints for the project, which are given below:

1. The project has a maximum budget of \$200, not including the microcontroller.
2. The project needs to be completed by late November 2025.
3. The device needs to have at least one type of sensor, and the output needs to be processed to provide the direction information either through a mechanical system or a digital system.
4. The device should be able to fit into/disassembled to fit into a box with dimensions of 55cm x 40cm x 23 cm.
5. The maximum assembly/set up time for the device is 5 minutes.

2.4 Revise Client's Problem Statement

The localization of a sound source in space is a non-trivial task, complicated by factors including sound reflection, absorption, and background noise. In response to this challenge, the client has requested an acoustic tracking device that can identify the direction of an incoming sound.

In the face of these obstacles, the client is requesting a device capable of detecting the source of both continuous and discrete sounds in a variety of indoor and outdoor environments. The device must cost a maximum of \$200 and assembly should take no longer than 5 minutes. It should be capable of detecting sound sources up to 50 metres away from the center of the device. It should primarily be able to identify the direction to the sound source, and secondarily the distance to the sound source. It should be able to be disassembled into a box of dimensions 55cm x 40cm x 23 cm. It should be completed and ready to use by late November. It will need to use at least one sensor and output direction and distance information through a mechanical or digital system.

The device must produce consistent results given identical inputs. It must be durable such that it can withstand multiple assembly/disassembly cycles. It must also be as inexpensive as possible while adequately achieving the other objectives.

3. Conceptual Design

3.1 Functions

1. Determine Sound Direction

2. Determine Sound Distance
3. Ensure Durability
4. Ensure Consistency
5. Manage Cost

3.2 Requirements (Function Specs)

To quantify the success of the design objectives and translate them into actionable engineering requirements, each system function was assigned specific performance criteria reflecting the client's need, the practical constraints of the project, and the inherent limits of acoustic physics.

Foremost, for direction detection, the device needed to provide an angular estimate of an incoming sound relative to the geometric centre of the array. This required an aperture large enough to produce measurable time delays, a sufficiently high sampling speed to resolve those delays, and an algorithm capable of identifying the onset of a transient sound. The system therefore had to sample all three microphones synchronously at a minimum of tens of kilohertz and detect events consistently across channels.

Secondly, the metric for consistency required the device to produce repeatable directional estimates under identical test conditions. This constrained sensor stability, mechanical alignment, and the reliability of the detection algorithm, particular in the presence of noise.

Thirdly, to satisfy the durability objective. The mechanical structure needed to withstand repeated assembly and disassembly without connections loosening or drifting out of alignment. This meant rigid mounting for the microphones, reliable press-fit or threaded connections, and a frame that would not deform under standard transport and use conditions.

Furthermore, for distance detection, the system was required to estimate the relative distance of the source wherever feasible. This demanded that the array resolve small variations in waveform and that the processing pipeline had to incorporate delay-based or amplitude-based cues. Although the distance objective was secondary, the design had to, at a minimum, support the possibility of estimating distance at shorter ranges.

Finally, the cost objective required simply that all expenses remain within the project's \$200 budget. Components were selected in accordance with this constraint while ensuring that the abovementioned technical requirements could still be met.

3.3 Establish Means for Functions

Functions-Means Morphological Chart - a map of possible means corresponding to each function that the design must fulfill

Function	Means 1	Means 2	Means 3
1. Determine Sound Direction	TDoA Algorithm (spatially distributed mics)	RSS Algorithm (static omnidirectional mic array or directional mics)	RSS Algorithm (mechanical scanning)
2. Determine Sound Distance	TDoA (uncalibrated)	RSS (with calibration)	
3. Ensure Durability	Static Design (no moving parts)	Robust Materials (3D prints, threaded inserts)	Mechanical Design (actuator-based)
4. Ensure Consistency	Signal Pre-Processing (sound filtering)	Static Hardware Array	High-Precision Actuator
5. Manage Cost	Use Low-Cost Components	Utilize Provided Microcontroller	

An explanation of technical means is provided below:

Microphone Array Geometry

A spatially distributed array of microphones allows for the use of a time difference of arrival (TDoA) algorithm. By placing the three microphones at the vertices of an equilateral triangle whose side lengths are known, we can use the TDoA algorithm which relies on the delay in the arrival time of a sound at each microphone to calculate the direction and distance to a sound source.

Dual-Algorithm Approach

The system can use two distinct algorithms to handle each type of sound source (i.e. transient and continuous):

TDoA: For transient sounds like a clap, a TDoA algorithm can be used. TDoA is capable of precisely locating sounds with a clear start time, and the algorithm allows for distance estimation without pre-calibration.

RSS: For continuous sounds like a drone, a Received Signal Strength (RSS) algorithm is simpler to implement. This method determines direction by identifying which microphone is receiving the loudest signal. With calibration of the system to the sound source in question ahead of time, distance could also be determined.

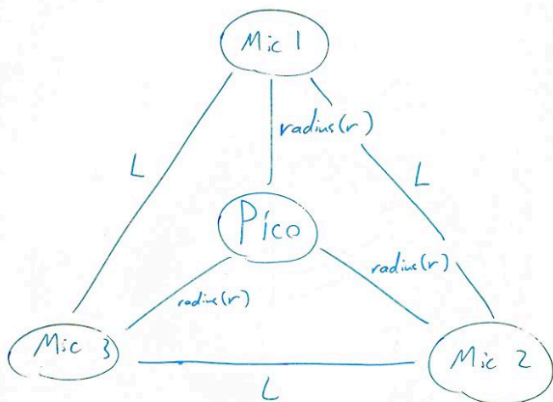
Signal Pre-Processing

Raw audio signals should be pre-processed before analysis to improve precision and accuracy of the system. The signal to noise ratio can be improved by implementing a band-pass filter to filter out ambient high and low frequencies. This filter will remove irrelevant background noise and electrical hum, creating a cleaner signal and allowing the TDoA algorithm in particular to more reliably cross-correlate the sound signals from each microphone.

3.4 Design Alternatives Considered

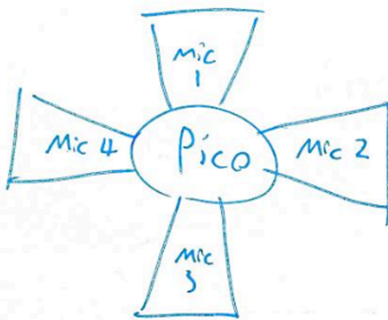
1) Omnidirectional Microphone Array

The first design considered was an omnidirectional microphone array, consisting of three microphones arranged in an equilateral triangle around the system centre, which houses the microcontroller. Microphones would be placed 0.5–3 m from the centre, depending on testing. This configuration was chosen because the spatial distribution allows the use of a TDoA algorithm

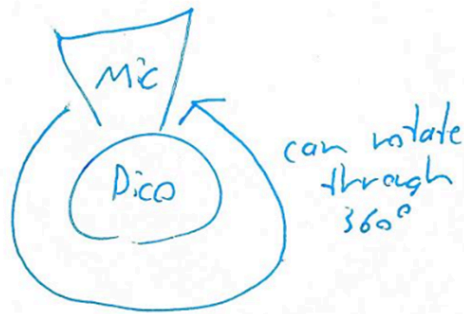


for transient sounds (e.g., clapping), providing higher precision than RSS-based methods, while still enabling RSS for continuous sources (e.g., drones). The combination of these features offered a flexible and accurate approach for detecting both transient and continuous sound sources.

(2) Directional Array



An alternative design considered made use of 3 or 4 directional microphones arranged radially at the centre of the system, with equal angles between them. Direction was estimated from the relative intensity of each microphone, with higher readings microphones pointed closer to the source. While this design could provide a reasonable direction estimate, it carried its own challenges; for example, the TDoA algorithm could not be used for this design, as the roughly coincident arrangement would not have enough distance between microphones for any measurable delay in arrival.



(3) Rotating Microphone

The final design alternative considered featured a single directional microphone mounted to a continuous rotary actuator. The device would repeatedly rotate in search of the highest sound intensity and follow it accordingly until a more intense sound source is detected. While this design would likely be capable of determining the acoustic direction, it would require a complex control algorithm, which was deemed unnecessarily complex relative to the other design alternatives.

3.5 Refine and Apply Metrics to Design Alternatives

We aligned on the following evaluation scheme for the three designs considered. The raw scores for each objective were given on a scale from 1 to 5, where 1 represented absolute failure to achieve the considered objective and 5 represents the best possible success. Note that each objective is weighted according to its score in the aggregated PCC chart.

Numerical Evaluation Chart - *each design received a raw score which was then weighted according to the aggregated PCC scores; these were summed to give a total score for each design.*

Objective	Weight	Design 1: 3-Mic Array	Design 2: Directional Array	Design 3: Rotating Mic
		Score / Weighted	Score / Weighted	Score / Weighted
Direction Detection	12	5 / 60	3 / 36	1 / 12
Consistency	9	4 / 36	4 / 36	2 / 18
Durability	6	4 / 24	4 / 24	2 / 12
Distance Detection	2	5 / 10	2 / 4	2 / 4
Cost	1	4 / 4	4 / 4	3 / 3
TOTALS		134	104	49

3.6 Basis for Design Selection

Design 1, the three-microphone omnidirectional array, was selected as the final design, achieving the highest total score of 134. Its primary advantage stemmed from robust directional detection using the Time Difference of Arrival (TDoA) approach. TDoA relies on the precise timing of sound arrival between spatially separated microphones and the known speed of sound, rather than amplitude differences, which would allow the system to produce consistent results for transient sounds such as claps. This method proved largely independent of environmental factors like reflections and background noise, making it effective in both indoor and outdoor testing.

The other alternatives failed to satisfy the metrics and were eliminated. Design 3, the rotating directional microphone, was fundamentally limited by its reliance on continuous sound sampling to locate the angle of maximum amplitude. As a result, it could not capture instantaneous sounds, making it unsuitable for transient events. Design 2, the static directional RSS array, was able to detect direction through amplitude differences, but it was incompatible with TDoA and required pre-calibration to measure distance accurately. It was also more susceptible to multipath reflections and environmental interference, which would have reduced the device's consistency and accuracy.

Design 1 not only supported TDoA for transient sounds but could also implement RSS for continuous sounds if required, providing flexibility without compromising primary objectives. Its superior directional accuracy, ability to operate without pre-calibration, and compatibility with spatial and budget constraints made it the clearly optimal choice. Based on these considerations, the three-microphone omnidirectional array was implemented as the prototype for testing and competition.

4. Preliminary Design

4.1 Model and Analyze Chosen Design

As outlined above, we selected the three-microphone omnidirectional array (Design Alternative 1) based on it being the strongest alternative across our weighted objectives. However, an important technical consideration that proved to be relevant at this point in the design process is the fundamental difference in array geometry requirements between TDoA and Received Signal Strength (RSS) algorithms.

RSS algorithms perform better with larger microphone arrays. Large spacing between microphones leads to bigger differences in signal strength between microphones, improving directional accuracy. In contrast, TDoA algorithms require small arrays for continuous sound sources. To implement GCC-PHAT (Generalized Cross-Correlation with Phase Transform) for continuous sounds like drone noise, microphone spacing must be small enough to avoid spatial aliasing. Based on this, we chose to go with a 20 cm side-length equilateral triangle, which also has the additional benefit of being printable as a single unit.

This fundamental constraint was the primary consideration for our pivot away from RSS. While RSS would work well for transient sounds with a larger array, the need to also handle continuous sounds through GCC-PHAT requires a smaller array. The selected 20 cm triangle side length allows us to use onset detection for transient sounds (claps), use GCC-PHAT cross-correlation for continuous sounds (drone), and maintain sufficient time delay resolution at a 48 kHz sample rate.

4.1.1 Mathematical Modelling

The mathematical model for angle and distance calculation is detailed in Appendix A. In summary, the system uses an algebraic closed-form solution to compute the angle to the sound source based on measured time differences, with distance estimation performed via Levenberg-Marquardt optimization (note: LM optimization for distance was removed from the final design due to real-world limitations of the solver; see Section 4.2.1: Testing Results).

4.2 Prototype, Test, and Evaluate the Chosen Design

The system consists of the following key components:

Hardware Components

The system consists of three MAX4466 microphone amplifiers arranged in an equilateral triangle with 20 cm sides. The microphones contain potentiometers for gain adjustment, so no external resistors are needed. A Raspberry Pi Pico 2 (RP2350) was used as the microcontroller for analog-to-digital conversion and signal processing. The Pico is connected to a laptop via USB for development, debugging, and result output. These components are mounted on a 3D-printed triangular frame with printed pegs at each corner where microphones slot directly onto the frame. The frame includes a tripod on its underside for additional stability.

The design prioritizes simplicity and durability. By using microphones with integrated potentiometers and a direct peg-mounting system, we minimize points of failure and physical complexity.

Software Architecture

At this stage in the design process, we implemented clap detection, which fed data to a TDoA solver.

The clap detection system is comprised of two parts:

Stage 1: Time Delay Estimation

The system continuously monitors audio from all three microphones. When the energy level from any microphone crosses a given threshold, the system flags an onset detection event. After detection, the system notes the onset time at the first microphone and monitors for onsets from the other two microphones. These times are then converted into distance differences (Δd_{12} and Δd_{13}) by multiplying by the speed of sound in air.

Stage 2: TDoA Localization

The distance differences (Δd_{12} and Δd_{13}) from Stage 1 are fed into the TDoA solver, which uses geometric calculation of the angle to the sound source (per the mathematical model detailed above in section 4.1.1) and Levenberg-Marquardt iterative optimization for distance estimation.

4.2.1 Testing Results

Specific synthetic data testing results (for the TDoA solver) and hardware prototype testing results are included in Appendix E.

In synthetic data testing, the TDoA solver was tested with known source positions across various ranges and angles. For each test position (x, y), we calculate the true distance differences, then run the solver to see how well its solution corresponds to the true distance differences.

Synthetic data testing covered the following cases:

Close range (1-3 m): Excellent angle and distance accuracy.

Medium range (3-10 m): Good angle accuracy, acceptable distance accuracy.

Long range (10-25 m): Accurate angles, distance unreliable.

360° coverage: Consistent performance across all angular directions.

Testing showed that angle detection works reliably at all tested ranges, while distance estimation predictably degrades as range increases.

With this information we proceeded to hardware testing. Initial tests showed catastrophic failure of the distance solver using real-world data, which can likely be attributed to the relative noisiness of real data and the extreme sensitivity of the distance solver to minor inaccuracies in the distance deltas from the onset detector. As such, at this point in the design process we decided to move forward with angle detection only. The results of angle-only hardware testing at this stage yielded satisfactory results, with over 50% of actual claps in testing generating angle values from the device, and the majority of calculated angles being quite accurate (sub 10 degree deviation). The primary failure mode was missed triggers (i.e. the system not detecting a sound when a valid sound was present).

5. Detailed Design

5.1 Refine and Optimize the Chosen Design

This project implements an acoustic localization system using a Raspberry Pi Pico 2 (RP2350) and three MAX4466 microphone amplifiers to determine the direction to sound sources. The design prioritizes durability and user usability, featuring a 3D-printed equilateral triangular frame (20 cm side length) with a screwless peg mounting system to create a tripod. To assist with immediate validation of the system's output, angle tick marks were printed directly onto the frame, allowing users to visually cross-reference the estimated direction with the physical source.

The microphones are directly integrated with gain potentiometers, eliminating the need for external components.

The processing pipeline converts raw audio to direction through two stages. While Generalized Cross-Correlation Phase Transform (GCC-PHAT) was initially explored for robust time-delay estimation, it was cut from the final build to reduce complexity of the final build. Instead, Stage 1 utilizes a TDoA method to calculate time delays. Attempts were made to implement dynamic noise floor adjustment to improve this detection, but these failed to produce reliable results in varying environments. Consequently, the system relies on flagging energy thresholds crossing a fixed floor. Stage 2 computes the source angle algebraically using a closed-form vector solution. This approach operates under tight physical constraints, where a 48 kHz sample rate provides $\sim 20.8 \mu\text{s}$ resolution to resolve a maximum theoretical time difference of just 0.583 ms across the array. Distance calculation, while marginally successful in synthetic data testing, was dropped for the hardware implementation due to insufficient precision in real-world time delays for meaningful results.

5.2 Assign and Fix Design Details

The final design uses a Raspberry Pi Pico 2 as the microcontroller, sampling three MAX4466 microphones at 48 kHz per channel. Audio is captured via DMA into a ring buffer, while the CPU calculates STE over 64-sample windows to detect events. Detected onset times feed into the algebraic TDoA solver to compute the source direction. The PLA frame features press-fit mounting for microphones, allowing easy replacement and ensuring alignment.

Hardware Specifications:

- **Microcontroller:** Raspberry Pi Pico 2 (RP2350)
- **Sensors:** 3x MAX4466 Electric Microphone Amplifiers (Gain adjusted to $73.6 \pm 0.2 \text{ k}\Omega$)
- **ADC Config.:** Sampling on GPIO pins 26, 27, and 28 at 144 kHz total clock rate (48 kHz effective rate per channel)
- **Power:** 5V via USB bus
- **Physical Dimensions:** 20 cm side-length equilateral triangle

Code Structure:

- Direct memory access is used to capture 6144 samples (approx. 42ms) of audio into a buffer without CPU intervention
- The CPU computes STE over 64-sample windows. If the energy exceeds the set threshold (default factor of 6x the calculated noise floor) on all three channels simultaneously, a “clap” event is flagged

- The time indices of the clap onset are converted to time differences and fed into the algebraic angle solver

6. Conclusion

We initially set out to design and implement a comprehensive acoustic system capable of full 2D localization, with the specific objective of resolving both the angle and the radial distance for a variety of audio signals, including both transient pulses and continuous waveforms. The original vision was to create a highly versatile sensor array that could pinpoint a sound source's exact coordinates in a two-dimensional plane.

As the project evolved, it became clear that reliable distance estimation and the detection of continuous sound sources presented signal processing challenges that exceeded initially anticipated. Specifically, resolving distance with high precision using a small-aperture array requires extremely tight timing tolerances and high level analysis of the sounds that were difficult to isolate from environmental noise. Consequently, to adhere to our quality standards and project timeline, the scope was adjusted. We chose to prioritize the reliability of the system's core function, direction finding, rather than delivering a product with many features that don't fully work.

Despite the reduction in scope, the project is a success. The final product is a robust directional finder that excels at localizing transient sound sources. By focusing our resources on the Time Distance of Arrival (TDoA) algorithm as applied to transient sound sources, we achieved a system that provides consistent and accurate directional data, effectively filtering out noise that troubled earlier iterations. The shift in focus allowed us to refine the system's responsiveness and stability, ensuring that the primary objective of acoustic localization was met with a high degree of confidence.

Ultimately, this project stands as a successful proof of concept for array-based sound localization. While distance estimation and continuous source detection remain complex stretch goals, treating them as future enhancements rather than current requirements allowed us to deliver a functional, high-performance unit. The current system lays a solid architectural foundation, providing a modular platform upon which future iterations can build to solve the more advanced challenges of wavefront curvature and continuous signal processing.

7. Further Work

We have identified three primary avenues for improvement, each of which would significantly enhance the performance and reliability of future iterations of the system: a more capable detection algorithm, the use of digital microphones, and additional signal processing measures.

The implementation of the Generalized Cross Correlation with Phase Transform (GCC-PHAT) algorithm would enable the system to move beyond simple transient detection and accurately localise continuous sound sources, such as drones. By estimating time delay in the frequency domain, GCC-PHAT is substantially more robust to noise and reverberation than direct time-domain thresholding.

In addition, the current analog microphones could be replaced with digital I²S microphones. An I²S-based front end would place all channels on a common clock, ensuring synchronization across channels, and the fully digital signal would eliminate the timing skew caused by the Raspberry Pi's multiplexed ADC as well as any noise in the analog signal prior to conversion.

To further attenuate noise, frequency-domain or statistical noise filtering methods could be used to suppress interference and improve the data's signal-to-noise ratio prior to correlation and localisation. In particular, a Wiener filter could be applied to reduce broadband noise, while a Kalman filter or probabilistic Bayesian filter can be used to stabilise delay estimates and reduce factitious peaks.

8. Appendix

A. Supporting Calculation or Modeling Results

Physical Constraints

Given the equilateral triangular mic array with 20 cm sides:

- Maximum time difference: $0.2 \text{ m} / 343 \text{ m/s} = 0.583 \text{ ms}$
- Sample resolution at 48 kHz: $\sim 20.8 \text{ } \mu\text{s}$ per sample
- Maximum sample delay: ~ 28 samples for sound traveling the full 20 cm

Angle Calculation

The angle to the sound source is calculated algebraically based on the microphone positions and measured distance differences. Given position vectors V_{21} (from mic 1 to mic 2) and V_{31} (from mic 1 to mic 3), the direction unit vector components are:

$$\cos(\theta) = (V_{31}^y \times \Delta d_{12} - V_{21}^y \times \Delta d_{13}) / \det$$

$$\sin(\theta) = (V_{21}^x \times \Delta d_{13} - V_{31}^x \times \Delta d_{12}) / \det$$

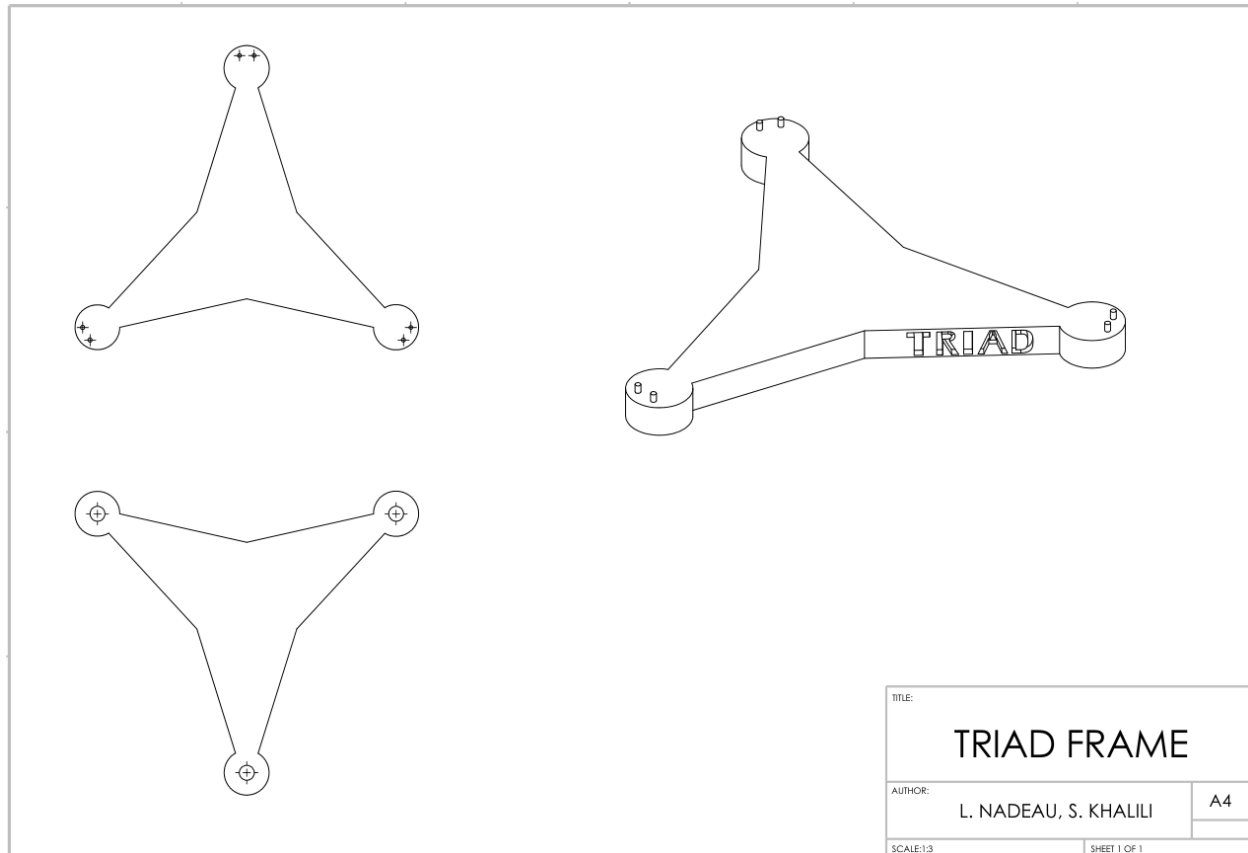
where $\det = V_{21}^x \times V_{31}^y - V_{21}^y \times V_{31}^x$. A 180° correction is applied to account for the systematic offset in our coordinate system. This method provides angle estimates within $0\text{-}5^\circ$ of the true direction across all ranges tested.

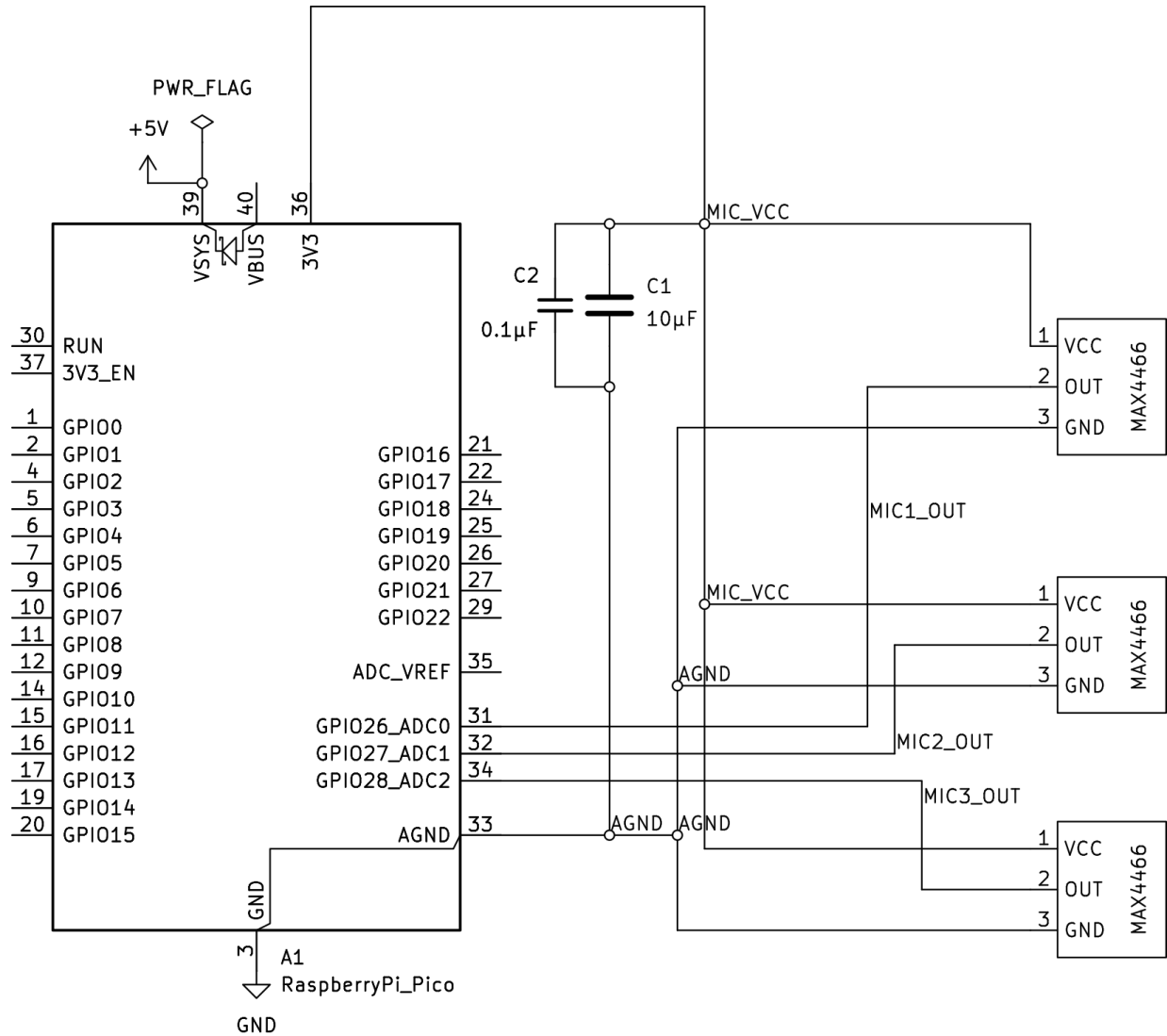
Distance Estimation

We used Levenberg-Marquardt optimization to minimize the residual error between measured and calculated distance differences. The system tries multiple starting points (1m, 3m, 5m, 8m, 12m, 18m) along the calculated angle direction and selects the solution with the lowest error.

Our synthetic data testing included flags when the system detects substantial error in the distance estimation (>1 metre error); see Appendix E.

B. Drawings





C. Fabrication Specifications

Frame

- **Manufacturing Process:** FDM 3D Printing.
- **Material:** PLA
- **Infill:** 15% gyroid pattern for structural rigidity without excessive weight.
- **Assembly:**
 - The legs friction fit into the holes on the underside of the mount.
 - The frame features printed pegs at the vertices sized for a friction fit with the MAX4466 modules.
 - The Raspberry Pi Pico is mounted to a central breadboard, situated between the mount's legs.

D. Lists of Material and Costs

Description	Cost*	Count	Acquired?	Details
Project				
Pi Pico 2	CA\$17.91	1	Amazon	https://www.amazon.ca/dp/B0DGGMVTWV
MAX4466 Microphone Modules (6pc)	CA\$16.79	1	Amazon	https://www.amazon.ca/dp/B0C6K4G6Z4
Frame	-	1	3D Printed	
Jumper Wires	-	-	Lab	
Breadboard	-	1	Lab	
F/M DuPont Wires	-	-	Lab	
Project Cost*	CA\$34.70			
Remaining budget	CA\$165.30			
% Budget spent on project	21.0%			
Other				
Poster	-	1	Lab	
Poster Board	CA\$70.54	1	Staples	
Other Cost*	CA\$70.54			
Total Cost*	CA\$105.24			
Budget Remaining	CA\$94.76			
Total % Budget spent	52.6%			
* All costs inclusive of tax				

E. Prototype Testing

TDoA Solver Synthetic Data Testing Results

===== TDoA Angle + Distance Solver =====

Output: ANGLE, DISTANCE with flag for substantial error (>1m error)

+ Close: 2m @ 27° Angle: 26.4° (err: -0.1°) + Dist: 2.2m (err: 0cm)
 + Close: 2.1m @ 135° Angle: 134.4° (err: -0.6°) + Dist: 2.1m (err: 0cm)
 + Close: 2.5m @ 79° Angle: 79.0° (err: +0.4°) + Dist: 2.5m (err: 0cm)
 + Med: 5.8m @ 31° Angle: 31.0° (err: +0.0°) + Dist: 5.8m (err: 0cm)
 + Med: 8.1m @ 150° Angle: 150.3° (err: +0.0°) + Dist: 8.0m (err: 6cm)
 + Med: 8.5m @ 291° Angle: 290.7° (err: +0.2°) + Dist: 8.5m (err: 7cm)
 + Long: 18m @ 34° Angle: 33.7° (err: +0.0°) + Dist: 18.0m (err: 3cm)

+ Long: 21.5m @ 158° Angle: 158.2° (err: +0.0°) Dist: 18.0m [UNRELIABLE, err: 355cm]
 + Long: 19.2m @ 309° Angle: 308.7° (err: +0.1°) Dist: 18.0m [UNRELIABLE, err: 120cm]
 + Long: 25m @ 90° Angle: 90.0° (err: -0.0°) Dist: 21.9m [UNRELIABLE, err: 308cm]
 + Long: 25m @ 0° Angle: 359.9° (err: -0.1°) Dist: 26.7m [UNRELIABLE, err: 169cm]
 + 360: 20m @ 0° Angle: 359.9° (err: -0.1°) Dist: 18.7m [UNRELIABLE, err: 131cm]
 + 360: 20m @ 45° Angle: 45.1° (err: +0.1°) Dist: 18.0m [UNRELIABLE, err: 200cm]
 + 360: 20m @ 90° Angle: 90.0° (err: -0.0°) - Dist: 19.1m (err: 93cm)
 + 360: 20m @ 135° Angle: 134.9° (err: -0.1°) Dist: 18.0m [UNRELIABLE, err: 200cm]
 + 360: 20m @ 180° Angle: 180.1° (err: +0.1°) Dist: 18.7m [UNRELIABLE, err: 131cm]
 + 360: 20m @ 225° Angle: 224.9° (err: -0.1°) Dist: 18.0m [UNRELIABLE, err: 200cm]
 + 360: 20m @ 270° Angle: 270.0° (err: +0.0°) - Dist: 19.1m (err: 94cm)
 + 360: 20m @ 315° Angle: 315.1° (err: +0.1°) Dist: 18.0m [UNRELIABLE, err: 200cm]

===== SUMMARY =====

Angle accuracy: 19/19 correct (100%) - within 5°
 Distance accuracy: 9/19 cases below error threshold (47%)

Hardware Testing Results

Range Category	Test Cases	Success Rate	Failure to Trigger	Catastrophic Failure	Outcome
Close (1m)	15	9	4	2	Marginal Success
Medium (5m)	15	8	6	1	Marginal Success

Note: "Catastrophic Failure" indicates trials that returned nonsensical values, such as distance deltas of 0 for both pairs of microphones.

F. Failure Analysis

While the system successfully achieved directional localization, physical and hardware constraints compromised distance estimation and continuous source tracking. Distance detection, while working in synthetic testing, broke down under real-world conditions, as the extreme sensitivity of the Levenberg-Marquardt solver to small errors in the distance deltas from the TDoA calculator rendered its outputs nonsensical. Additionally, we descope continuous source tracking via GCC-PHAT because the hardware configuration was un conducive to effective cross-correlation. The algorithm was compromised by the high noise floor of the analog front-end and the irreducible timing skew introduced by the microcontroller's multiplexed ADC. Lastly, after dynamic noise floor algorithms proved unstable, we adopted fixed energy thresholds, leading to occasional false negatives when processing fainter signals.

Distance Estimation Limitations

The most significant limitation encountered was the unreliability of distance estimation under real-world conditions. This failure is inherent to the geometry of the array: a 20 cm baseline is too short to effectively triangulate sources at long distances because the wavefront curvature becomes negligible (approximating a plane wave). Consequently, the time differences become almost exclusively dependent on the angle, and the sensitivity to distance drops below the noise floor of the sampling system.

Continuous Source Detection

The detection of continuous sound sources was descoped from the final deliverable. The planned GCC-PHAT (Generalized Cross-Correlation with Phase Transform) algorithm required cleaner signal inputs than the analog MAX4466 modules provided. Significant noise and timing skew from the multiplexed ADC on the Pico rendered any possibility of correlation unreliable and could not be implemented within the project timeline.

G. Project Charter

Project Objectives

The team has identified five core objectives, ranked in descending order of importance.

Direction Detection: The system must detect the azimuthal direction of both transient and continuous sound sources relative to the geometric centre of the device.

Consistency: The system must produce repeatable results given identical inputs across multiple trials.

Durability: The system must withstand multiple assembly and disassembly cycles without degradation in performance or alignment.

Distance Detection: The system should estimate the distance from the sound source to the centre of the device (secondary objective).

Cost: The system must be as inexpensive as possible while adequately achieving the above objectives.

Project Constraints

The project has a maximum budget of \$200 CAD, excluding the microcontroller (provided by the department). The project completion deadline is late November 2025. The device is required to incorporate at least one sensor, with output processed to provide directional information via a

mechanical or digital system. The device is required to fit into, or be disassembled to fit into, a box with dimensions 55 cm × 40 cm × 23 cm. Maximum assembly/setup time is 5 minutes.

Time Frames

The project completion target is the end of November 2025. Preliminary Report 1 is due in early October 2025, followed by Preliminary Report 2 in late October 2025. The final presentation and competition takes place in late November 2025, with the final report due in early December 2025. Detailed milestone timelines are provided in the team Gantt chart (Appendix H).

Team Members

Devon Streich, Shygaan Khalili, and Luca Nadeau.

Member Responsibilities

Devon is responsible for writing the prototype TDoA algorithm, synthetic data testing, and refining the TDoA algorithm and time delay estimator alongside Shygaan. He also co-owns firmware refinement with Shygaan, and was a primary contributor to the second preliminary report.

Shygaan is responsible for sourcing materials, aligning on the microcontroller, writing the prototype clap detection algorithm, and assembling the prototype alongside Luca. He also co-owns the TDoA algorithm refinement and firmware refinement with Devon.

Luca is responsible for creating the CAD mount files, printing the CAD mounts, and creating the final presentation and poster. He also assisted with prototype assembly and the second preliminary report.

All team members share responsibility for aligning on hardware configuration and signal processing, finalizing the design for competition, finalizing all code, testing the prototype and finalized design, writing the first preliminary report, and writing the final report.

Project Commitments

The team holds regular meetings on Tuesdays and Thursdays from 3:00–5:00 PM in the Engineering Lab. Asynchronous coordination is handled via group chat. Deadlines are treated as firm commitments, with deviations communicated in advance.

Criteria for Project Success

The primary success criterion is that the system reliably detects the direction to a transient sound source in a 2D plane centred on the device, with angular deviation under 20%. As a secondary criterion, the system should provide a distance estimate for close-range sources under 10 metres.

Stretch goals include 3D localization and continuous sound source detection via GCC-PHAT. The academic target is an A+ on the project component.

H. Gantt Chart

Triangular Ranging Instrument for Acoustic Detection (TRIAD) Project Gantt Chart

				PHASE 1								PHASE 2				PHASE 3									
WBS No.	Task Title	Owner(s)	% Complete	9/23	9/25	9/30	10/2	10/7	10/9	10/14	10/16	10/21	10/23	10/28	10/30	11/4	11/6	11/11	11/13	11/18	11/20	11/25	11/27	12/2	
1 Design																									
1.1	Align on TRIAD hardware configuration	Group	100%																						
1.2	Align on signal processing	Group	100%																						
2 Hardware																									
2.1	Source Materials	Shygaan	100%																						
2.2	Align on Microcontroller	Shygaan	100%																						
2.3	Create CAD Mount .stl Files	Luca	100%																						
2.4	Print CAD Mounts	Luca	100%																						
2.5	Assemble Prototype	Luca, Shygaan	100%																						
2.6	Finalize design for competition	Group	100%																						
3 Software																									
3.1	Write prototype TDoA algorithm	Devon	100%																						
3.2	Write prototype clap detection	Shygaan	100%																						
3.4	Refine TDoA algorithm & time delay estimator	Shygaan, Devon	100%																						
3.5	Refine other firmware	Shygaan, Devon	100%																						
3.6	Finalize all code	Group	100%																						
4 Testing																									
4.1	Synthetic Data Testing	Devon	100%																						
4.2	Test prototype	Group	100%																						
4.3	Test finalized design	Group	100%																						
5 Documentation																									
5.1	Write & Submit First Preliminary Report	Group	100%																						
5.2	Write & Submit Second Preliminary Report	Devon, Luca	100%																						
5.3	Create Final Presentation & Poster	Luca	100%																						
5.4	Write & Submit Final Report	Group	100%																						

I. Bibliography

Henri, G. (2024, May 5). *The Levenberg-Marquardt algorithm for nonlinear least squares curve-fitting problems*. <https://people.duke.edu/~hpgavin/lm.pdf>

Mathworks. (n.d.). *Object tracking using time difference of arrival (TDOA)*. MATLAB & Simulink. <https://www.mathworks.com/help/fusion/ug/object-tracking-using-time-difference-of-arrival.html>

Mathworks. (n.d.). *Gccphat*. Generalized cross-correlation - MATLAB.
<https://www.mathworks.com/help/phased/ref/gccphat.html>

Note: AI was used to structure and troubleshoot code.