



PAPER REPORT

2023

HLBSIB006
KRNNEN001
LNTZUH001

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CONTRIBUTIONS

Individual contributions

Sibusisiwe Hlabangana	<ul style="list-style-type: none">• Triangulation (all relevant sections)• User interface• Document formatting• UML Diagram• Feasibility analysis• Signal of interest• Microphone array
Nene Karinigi	<ul style="list-style-type: none">• Time delay estimation• Signal of interest• Microphone array• Noise reduction• Development timeline• Referencing• Feasibility analysis
Zuhayr Loonat	<ul style="list-style-type: none">• Data acquisition• Pi synchronization• Inter-subsystem and inter-Sub-subsystem Interactions• Feasibility analysis• Project management page• Feasibility analysis• Referencing

REQUIREMENTS ANALYSIS

Interpretation of requirements

The overarching system requirements are:

- The system must have four microphones arranged around a measured geometry and these microphones must capture a acoustic sound signal
- The system must use two synchronized Raspberry Pis to process the signal using a time delay estimation algorithm
- The TDoA values must be processed using a triangulation algorithm to locate the position of the acoustic sound signal on the grid
- The system must display the predicted results on a GUI interface

PI SYNCHRONIZATION

Having the Raspberry Pis synchronised is extremely important in a TDoA system as it affects the calculations. If the Pis are not synchronised, then the time measurements will yield inaccurate data.

COMPARISON OF POSSIBLE IMPLEMENTATION

Network Time Protocol

NTP (Network Time Protocol) is a protocol designed to synchronise the clocks of computers on a network. We have chosen this protocol for the following reasons:



01 — Scalability

NTP's architecture can support multiple devices over large-scale networks. In this project we need to connect multiple Raspberry Pis to each other.



02 — Accuracy

By using symmetrical network communication NTP synchronises time more accurately. It also allows for extremely fast synchronisation (within milliseconds).



03 — Easy Configuration:

NTP servers are easy to install and configure. It allows one to control the working terms of the servers. The Raspberry Pi systems are also configured to support this protocol.



04 — Reliability

NTP servers have mechanisms to prevent inaccuracies. In the case of network disconnection, it can still maintain an accurate time by using estimation.

It also has other features that allow for the prevention of packet loss, jitters, and other things that may impede the system.

	Advantages	Disadvantages
NTP	<ul style="list-style-type: none">- Scalability- Accuracy- Easy configuration- Reliability	<ul style="list-style-type: none">- Malware- Hacking- Synchronization is only in order of milliseconds so could cause serious inaccuracies for other calculations

CHOSEN IMPLEMENTATION

There is only one reasonably viable Pi synchronization as Pi only supports NTP.

POSSIBLE BOTTLENECKS

- Some of the possible bottlenecks include setting up the server as it will be the first time any of the team will be attempting to do so.
- Ensuring accurate Pi Synchronization as the calculations are off by 1ms has a huge effect on the overall results

REQUIREMENTS

- Must be able to connect multiple Pis.
- Must be able to keep the time between Pis synchronised.

SPECIFICATIONS

- Must be accurate to the closest millisecond.

SIGNAL ACQUISITION

The process of signal acquisition involves the signal of choice, the geometry of the microphone array, as well as data acquisition, synchronisation, and noise reduction of the target acoustic signal. A flowchart of the signal acquisition subsystem is shown below in figure 1.



Figure 1: Signal acquisition flowchart

1. SIGNAL OF INTREST

COMPARISON OF POSSIBLE IMPLEMENTATIONS

1. Pure sinusoid

A pure sinusoid sound is one where the sound is a plain and flat note. It has one frequency and is a sound that cannot be found in nature.

2. Chirp signal

The signal of interest is a chirp signal, whose time domain representation can be seen in Figure 2. The chirp signal is a sinusoid which has a constant amplitude but has a varying frequency with respect to time.

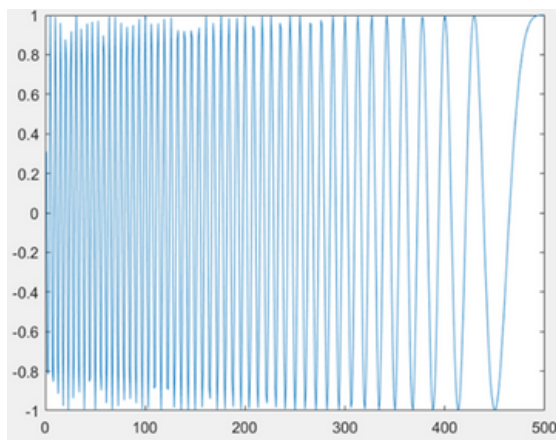


Figure 2: A chirp signal

CHOSEN IMPLEMENTATIONS

The chirp signal was chosen due to its robustness to noise. The chirp signal spreads the energy over many frequency components in its frequency domain, thus reducing the effects of noise, which in turn will increase the accuracy of the TDoA and in turn the localisation.

2. MICROPHONE ARRAY

The number of microphones increases the level of accuracy of the TDoA, and in turn the locating the source of the signal. The downside of increasing the number of microphones is that it increases the complexity due to having to synchronise all the microphones. As well, it also increases the budget of the project which is limited.

When the number of microphones is increased, the geometry that the microphones can be arranged in increases. There is a need to have a fixed geometry for the TDoA to be calculated.

COMPARISON OF POSSIBLE IMPLEMENTATIONS

1. Corner geometry

This microphone array would consist of placing the 4 microphones on each corner to form a square.

2. Circle geometry

This microphone consists of placing the microphones to form a circle and has the sound source in the centre of the circle.

3. Mid-point geometry

This microphone array consists of placing the midpoint between two corners of the square grid

	ADVANTAGES	DISADVANTAGES
Corner geomtry	<ul style="list-style-type: none">• Simplified triangulation that is highly accurate• Most common array• Covers the entire grids	<ul style="list-style-type: none">• Triangulation less accurate near the edges of the grid
Circle geometry		<ul style="list-style-type: none">• -More complex triangulation• Complex to set up
Mid point geometry	<ul style="list-style-type: none">• Accurate tracking of signal when in the centre	<ul style="list-style-type: none">• More complex triangulation depending on algorithm.• More inaccuracies when signal is at the corner.

CHOSEN IMPLEMENTATIONS

To strike a balance between budget, complexity and accuracy of the results, four microphones are to be used. In terms of the geometry, the microphones are to be arranged at the vertices of a square, as shown in figure 3 below. This is because a square is a geometry that is easy to work with.

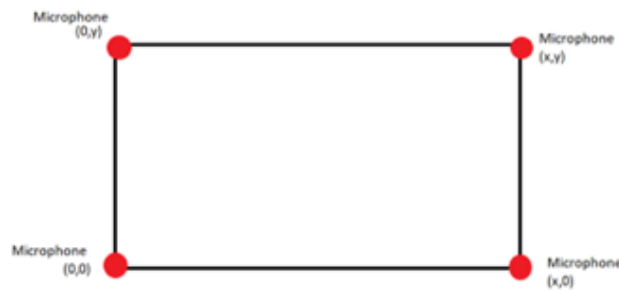


Figure 3 : Microphone array arrangement in Square

POSSIBLE BOTTLENECKS

- Inaccuracy in the placement of the microphones could cause inaccurate readings

SUBSYSTEM REQUIREMENTS

- The microphone array is in a sensible arrangement.
- Microphone array strikes a balance between budget, complexity and accuracy
- Microphones should detect signals of varying frequencies.

SUBSYSTEM SPECIFICATIONS

- The microphone array uses four microphones.
- Microphones must have fixed positions on the grid at each corner of the grid.
- Microphones must be placed on the vertices of a square of length 0.5 metres

3) DATA ACQUISITION

The most important part of this project is the acquisition of the data. Two main data acquisitions will take place. First, the sound data needs to be acquired, and second, the data needs to be sent to the parent PC to be processed.

3.1) SOUND ACQUISITION

For the sound acquisition, 4 MEMs microphones will be used. These microphones use a protocol called I2S, which is a protocol used to transmit audio data through a three-line serial bus which includes: SCK (Continuous Serial Clock), WS (Word Select) & SD (Serial Data).

	Advantages	Disadvantages
I2C	<ul style="list-style-type: none"> • I2S uses separate CLK & serial data lines. So it has very simple receiver designs as compared to asynchronous systems. • Single master device makes synchronisation easier • Complete digital connection 	<ul style="list-style-type: none"> • Not supported for high-level applications. • This protocol has a synchronization problem between three signal lines which is noticed at a high bit rate & sampling frequency. • Does not include error detection

CHOSEN IMPLEMENTATIONS

I2S was mainly chosen because it is easy to use with a Raspberry Pi and the MEMS microphones that we were provided with supported I2S.

Since I2S does not support sound compression, the sound will need to be stored in a “.wav” file.

The microphones will need to be configured in certain aspects as well. These include:

- Stereo or Mono Channels
- Audio format
- Volume Control
- File Type

The sound will also need to be normalised; this will be focused more on filtering.

3.2) DATA ACQUISITION

After the data from the mics has been acquired by the Pis, it needs to be sent to the parent to be processed so that the relevant calculations can be done. Here the data will be sent either to MATLAB, a parent Pi, or a parent PC.

COMPARISON OF POSSIBLE IMPLEMENTATIONS

1. MATLAB

For the data acquisition in MATLAB the child Pis will send data over a network to MATLAB on a parent PC, thereafter the processing will take place. MATLAB has built-in functionality for Raspberry Pies and is relatively easy to connect to from a Pi over the same network.

2. Parent PC/Pi

If we are to go the GUI route where a graphical interface is created on a parent PC or Pi, the data will be sent via SSH to the parent.

CHOSEN IMPLEMENTATIONS

Either of these options is feasible and the one chosen depends on the GUI that we choose. The chosen option is MATLAB.

3.3) SOUND SYNCHRONISATION

1. Calibration signal

The sound acquisition from the microphones can be tested and synchronised by emitting a sound at a point where the sound would take equal time to reach all the microphones. This would be the middle of the grid. When a sound is played from such a point, every mic should receive the data at the same time. If this doesn't happen, we then know we need to make changes to fix the problem.

2.NTP

The microphones that were provided allow for a stereo connection. A stereo connection is when two microphones work in sync as a "left and right". Two sets of a stereo configuration can be set up and the Raspberry Pis that each is connected to two will be synchronized using an NTP.

CHOSEN IMPLEMENTATIONS

The NTP method is chosen as the project relies more on accuracy compared to the calibration signal however NTP implementation is more complex

3) NOISE REDUCTION

1. Faraday Cage

A Faraday cage is an enclosure made of electrically conductive material. The Faraday cage reduces the noise that can be present in the wires by absorbing electromagnetic radiation on the surface of the cage.

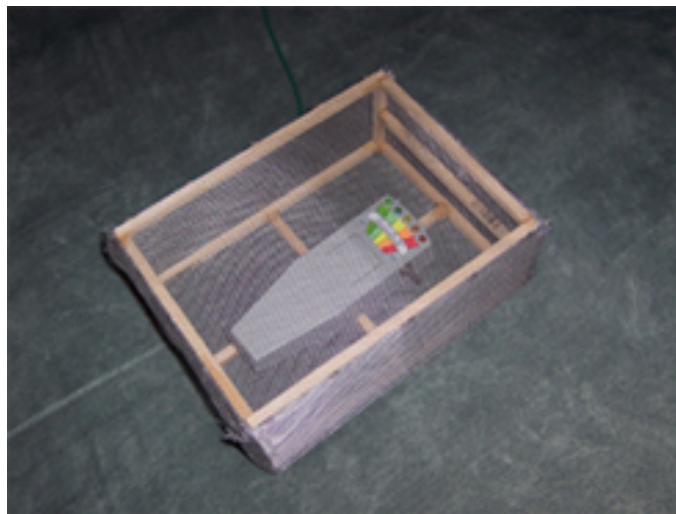


Figure 4: A Homemade Faraday Cage

2. Shielding cables

This technique involves covering all the wires in the system with an electrically conductive material. Shielding the cables has the same working principle as the Faraday cage as it absorbs external electromagnetic radiation.

3. Physical Filtering

A bandpass filter is a type of filter that only allows a specific range of signals through it. It selects signals that fall within a certain range, known as the passband, and attenuates signals outside this range. The block diagram of the bandpass filter is shown in Figure 5.

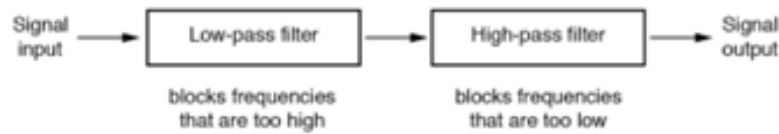


Figure 5: Block Diagram of a bandpass filter

A physical active bandpass filter is constructed by a combination of capacitors, resistors and op-amps. The cutoff frequencies of the bandpass filter depend on the values of each resistor and capacitor. A circuit diagram of a bandpass filter is shown below in figure 6.

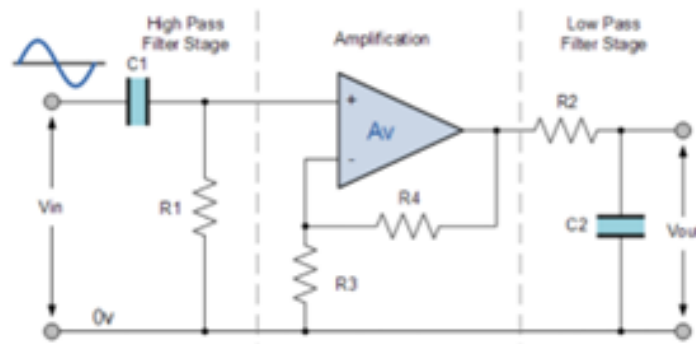


Figure 6: Bandpass filter circuit diagram

Physical bandpass filters, such as the one shown above are limited by physical limitations. This means that the frequencies outside the passband, while attenuated are still present in the filtered signal. Additionally, some frequency components of the noise present in the passband of the bandpass filter will not get attenuated. The frequency response of a bandpass filter is shown in Figure 7.

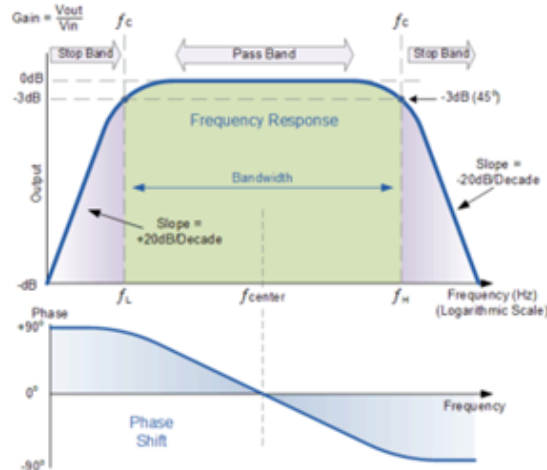


Figure 7: Frequency Response of a Bandpass filter

4. Simulated Filtering

Mathematical software such as MATLAB has signal processing tools which can simulate an ideal bandpass filter. An ideal bandpass filter completely attenuates all frequencies outside the passband region. The frequency response of a bandpass filter is shown below in Figure 8.

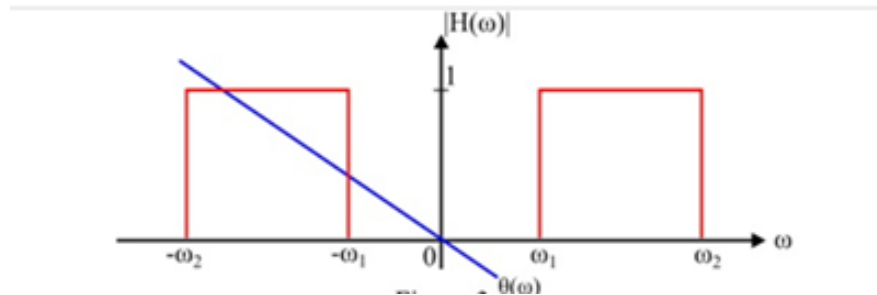


Figure 8: Ideal bandpass filter frequency domain

	Advantages	Disadvantages
Faraday Cage	<ul style="list-style-type: none">• Blocks electromagnetic noise	<ul style="list-style-type: none">• Difficult to construct.• Expensive• Does not account for acoustic noise.
Shielding Cables	<ul style="list-style-type: none">• Cheap• Simple to construct	<ul style="list-style-type: none">• Does not account for acoustic noise
Physical Filtering	<ul style="list-style-type: none">• Caters for all types of noise.• Simple to construct.• No computational power used	<ul style="list-style-type: none">• Hard to vary.• Limited bandwidth
Simulated Filtering	<ul style="list-style-type: none">• Ideal filter• No hardware is involved.• Easily varied.• Can be used at high frequencies	<ul style="list-style-type: none">• Uses computational power

CHOSEN IMPLEMENTATIONS

The preferred implementation is a combination of the shielded cables and the simulated low-pass filter. This is because the simulated filter completely attenuates all the frequencies outside the passband of the filter, therefore, increasing the SNR of the filter. On top of that, it is versatile as changing the cutoff frequencies does not require changing any components in a circuit. The shielded cables will reduce the presence of noise which is within the passband of the filters.

POSSIBLE BOTTLENECKS

- Working in a loud environment can make it impossible for the noise to be reduced.
- Having a big bandwidth means that the frequency components in the passband will still be present, giving a lower SNR than desired.

REQUIREMENTS

- Should produce a higher SNR after noise reduction.

SPECIFICATIONS

- The SNR of the output signal should be higher than the input signal.
- The system should operate well for an SNR of 80dB

TIME DELAY ESTIMATION

Cross-correlation, Generalized Cross-Correlation (GCC), and General Cross-Correlation with Phase Transform are extensively employed methodologies within the domain of signal processing for the purpose of time delay estimation. While these methodologies share similarities, it is important to recognize that each possesses distinct merits and limitations relative to one another.

The methods are used to calculate the TDoA, which in turn is used in the triangulation process to locate the signal. The TDoA is the time delay of one signal relative to another identical signal.

COMPARISON OF POSSIBLE IMPLEMENTATION

1. Cross-correlation

Correlation is a measure of the similarity between two signals. In the case of time delay estimation, the term cross-correlation is used. Figure 9 shows the result from the correlation between a signal $u(t)$ and $v(t)$ which is delayed and has additive noise to give $w(t)$ whose peak time is at the time delay.

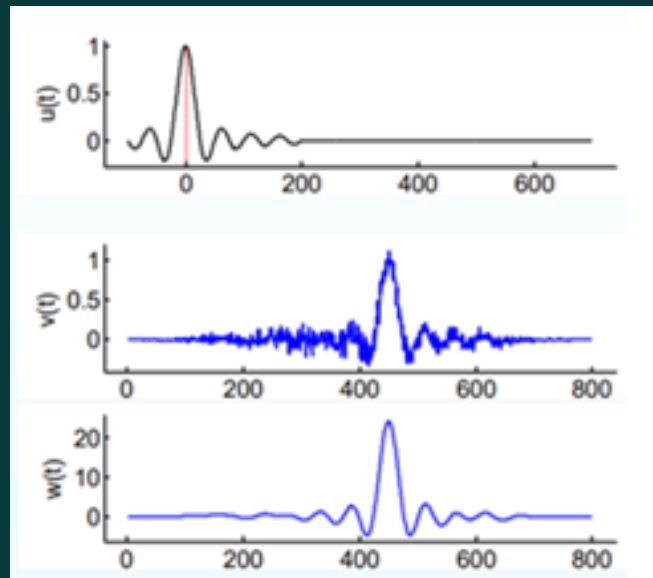


Figure 9: Cross-correlation between two signals.

The mathematical expression for cross-correlation is provided below:

$$w(t) = \int_{-\infty}^{\infty} u^*(\tau - t)v(\tau)d\tau$$

2. Generalised Cross-Correlation (GCC)

The Generalised Cross-Correlation is an improved implementation of the cross-correlation method. It is used to improve the accuracy of the time delay estimation through the incorporation of a diverse range of techniques, encompassing but not restricted to windowing, frequency domain analysis, and additional statistical methodologies. These measures collectively contribute to heightening the accuracy of estimations, particularly when confronted with signals characterized by noise.

The general formula for the GCC is given below:

$$GCC(t) = F^{-1} \left\{ \frac{U^*(\omega)V(\omega)}{|U(\omega)V(\omega)|} \right\}$$

This is a simplified general formula and may be modified to incorporate additional functionalities depending on the setting.

3. Generalised Cross-Correlation with Phase Transform (GCC-PHAT)

The Generalized Cross-Correlation with Phase Transform (GCC-PHAT) is one of the many GCC specialisations. Distinguished by its particular emphasis on refining time delay estimations within contexts of signal attenuation and noise, GCC-PHAT emerges as an improved implementation. This technique not only provides information on both phase and amplitude but also incorporates the normalization of cross-correlation using the magnitudes of the Fourier transforms of the involved signals.

	Advantages	Disadvantages
Cross-Correlation	<ul style="list-style-type: none">• Easy to implement.• Works well when sound source is near microphone array	<ul style="list-style-type: none">• Sensitive to noise• Does not distinguish between positive and negative time delays.• Limited bandwidth• No information on phase
GCC	<ul style="list-style-type: none">• Provides phase information.• Distinguishes between positive and negative time delays.• Less sensitive to noise	<ul style="list-style-type: none">• Increased complexity• Accuracy is dependent on the signal quality and precision
GCC-PHAT	<ul style="list-style-type: none">• Distinguishes between positive and negative time delays• Not affected by attenuation• Less sensitive to noise	<ul style="list-style-type: none">• Limited bandwidth

CHOSEN IMPLEMENTATION

GCC-PHAT is the most suitable method to use to calculate the TDoA. This is because it is less sensitive to noise and attenuation of the signal, as well as provides information on whether the time delay between two signals is positive or negative.

POSSIBLE BOTTLENECKS

- Limited bandwidth, therefore, may be unable to process the high-frequency components of the chirp signal.
- The presence of excess noise can cause inaccurate TDoA results.
- Limited information on the full mathematical computation of the algorithm
- Requires a lot of computation power.

REQUIREMENTS

- Should estimate the time delay value to within an acceptable margin of error.
- This should account for time delays and other errors that may occur.
- Should produce results in a format that is understandable.
- Should minimise latency during processing.

SPECIFICATIONS

- The TDoA should have an accuracy of $\pm 5 \mu\text{s}$.
- The TDoA of all microphones should be 0 when the signal source is in the middle of the grid.
- The TDoA should be produced using MATLAB and the output should be a numerical value.

TRIANGULATION

The microphones act as sensors which capture the sound (the signal) to get the Time Difference of Arrival of the signal. The location of the system in an ideal situation can be calculated using the intersections of hyperbolas however most practical situations have noise. The hyperbolas would not intersect at one point and will deviate from the source of the signal. Factoring noise makes the calculations' non-linear localization equations inconsistent. Below different triangulation methods will be discussed, and the advantages and disadvantages will be weighted for each method of triangulation.

COMPARISON OF POSSIBLE IMPLEMENTATION

1. Basic triangulation

The triangulation uses the difference ToA if the propagated signal is captured by each sensor to calculate the position of the signal within the grid. The TDoA denotes to the idea that each sensor will receive the signal at different times depending on the signal's distance from the sensor. The method forms hyperbolas based on the TDoA of one of the sensors and the points on the hyperbola are representative of all the possible points where the signal could have originated from. The point where all the hyperbolas intersect is the location of the signal. A huge issue is that hyperbolas are symmetrical functions reflected in their asymptotes thus there will be two intersections which cause ambiguity.

2. Total Least square estimation

The algorithm denotes one sensor as the reference sensor at which the other sensors' TDoA is calculated. The algorithm starts by using linear closed-form solutions. It uses an equation that relates the TDoA data to the coordinates of the sensor, signal and reference sensor as well as the distance from the signal to the sensor and distance to the reference sensor. The equation is squared, grouped into like terms and rearranged to give the linear model for solving the coordinates of the signal and the distance of the reference signal. A matrix $Ax = b$ is formed from the 3 equations that arise from calculating an equation for each sensor (excluding the reference sensor).

The ordinary least square method only factors in zero-mean Gaussian noise for the b matrix while the total least square method factors in noise for both the A and b variables. The TLS method offers better approximation.

3. Approximate maximum likelihood estimation

The algorithm starts with the maximum likelihood equation and creates a simplified approximate solution to the maximum likelihood (ML) equations. The ML equations are transformed into two linear equations with unknowns (x,y) , and the equations have coefficients that are dependent on the (x,y) . Thus, utilizing certain initial values for (x, y) , the approximate maximum likelihood algorithm solves for a new value of (x,y) and updates the coefficients to the new value of (x,y) . Following a series of at least five iterations, the cost function of ML is checked by the AML using the (x, y) values generated during each iteration and the minimum is selected as the solution.

4. Look-up Table (LUT)

The lookup table is made up of 8 columns. The first six columns are the TDoA combinations of the 4 sensors. The last two columns of the LUT correspond to the X and Y coordinates of the signal. Those coordinates are for the signal that produced that produced that TDoA values. The LUT essentially works by providing the coordinates for the sound signal that most closely resembles the TDoA values that were taken from the recording. The correct coordinate is chosen by finding the minimum sum of the square of the absolute difference between the TDoAs received from the recording and the TDoA from the LUT.

	Advantages	Disadvantages
Basic Triangulation	<ul style="list-style-type: none">• Simple to compute	<ul style="list-style-type: none">• High margin of error• Only accurate for high SNR• Relies heavily on precise synchronization
Least square estimation	<ul style="list-style-type: none">• -Easy TDoA localization method• Less computationally intensive	<ul style="list-style-type: none">• Significant margin of error
Maximum likelihood estimation	<ul style="list-style-type: none">• Highly accurate even for lower SNR	<ul style="list-style-type: none">• Computationally expensive• Need to have a close guess to get convergence.• Does not allow real time processing.

	Advantages	Disadvantages
LUT	<ul style="list-style-type: none"> • Simple to compute • Simple implementation 	<ul style="list-style-type: none"> • Requires minimal distortion of sound and very little noise. • Devices may have sampling errors thus wrong value is selected from table. • High margin of error

CHOSEN IMPLEMENTATION

Total least square algorithm is the best fit as there are several resources that use it as it is a popular method. The method also factors in noise in its calculation which means it is more accurate than the LUT and basic triangulation. It is not very computationally expensive and thus will work well with the Raspberry Pi capabilities.

POSSIBLE BOTTLENECKS

- Increased complexity of the algorithm due to the need to decrease the error of the system.
- Limited information on the full mathematical computation of the algorithm
- Limited computation power as most PC has 4 cores.

REQUIREMENTS

- The triangulation algorithm finds the grid point location of the signal.
- The triangulation algorithm transforms the TDoA data into equations that compute a grid location.
- The triangulation algorithm equations should factor in noise.
- Triangulation algorithm must be simple to implement.

SPECIFICATIONS

- The triangulation algorithm locates the grid point with an error rate of $\pm 10\text{cm}$.
- Triangulation can run on a basic Raspberry Pi.
- Triangulation algorithm has a runtime of less than 5 seconds.

USER INTERFACE

The user interface consists of a grid that will visually show the location of the signal and display the coordinates of the signal. The location of the signal is computed using the triangulation algorithms and the coordinates of the signal are displayed on a GUI with the 2D grid.

COMPARISON OF POSSIBLE IMPLEMENTATION

1. MATLAB

Matlab has an easy-to-use tool that enables one to create a great GUI (also called Apps). Laying out the aesthetic components and programming the app's behaviour are the two main duties of creating an app, and App Designer is an interactive environment that merges both processes. It enables swift transitions between writing code in the MATLAB editor and visual design on the canvas.

2. GuiZero

A Python 3 library that allows one to easily create a GUI. It is highly accessible and beginner friendly and does not require any additional libraries.

	advantages	disadvantages
MATLAB	<ul style="list-style-type: none">• Easy to use• MATLAB is platform-independent so can be used on any operating system.	<ul style="list-style-type: none">• MATLAB is interpreted language thus takes more time compiling code which would be inefficient for computationally expensive triangulation algorithms.• Difficult to develop real time applications using MATLAB
GuiZero	<ul style="list-style-type: none">• Easy to use.• uses simple syntax.• no additional libraries	<ul style="list-style-type: none">• Sometimes difficult to debug.• GUI is not as presentable as other ones

CHOSEN IMPLEMENTATION

MATLAB will be used as it is simple to implement and the team members have exposure to using the software.

POSSIBLE BOTTLENECKS

- Language used to code GUI needs more power than the Raspberry Pi can provide.

REQUIREMENTS

- Grid layout should mirror the actual grid.
- The grid must have a reasonable resolution to allow for accurate representation of signal location.
- The position of the grid point should be indicated on the GUI grid.

SPECIFICATIONS

- Grid is divided into 1cm x 1cm blocks thus GUI has a grid size of 50 x 50 grid blocks with labelled axis numbered the outer grid points.
- The grid point computed is rounded off to the nearest cm and plotted in the exact grid point.
- GUI displays the numerical value coordinates of the grid point.
- The GUI plots the grid point at the exact location that was computed by the triangulation algorithm.

FEASIBILITY ANALYSIS

Strengths

- Easy to debug due to the different subsystems.
- Easily adaptable to other geometries and signals
- Fast communication
- Good time accuracy

Weakness

- Limited computational capability
- Small grid so minimal room for error
- Complex setup for the Pi
- Highly dependent on NTP and I2C
- There are places for inaccuracies to occur

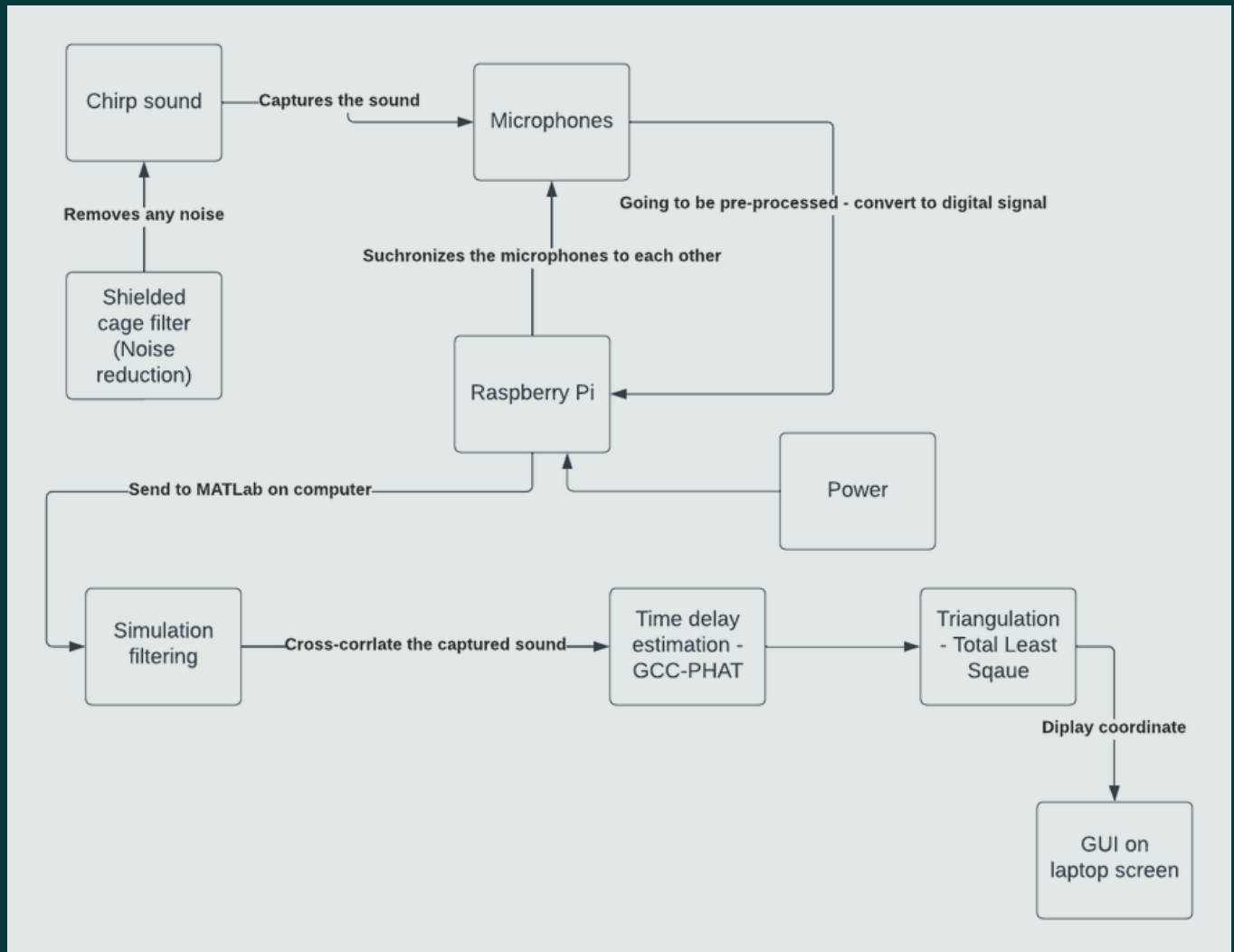
Opportunities

- Many different software available for simulations
- Raspberry Pi is a popular product so many resources are available.
-

Threats

- Loadshedding
- Damage to the system during transit
- Extremely loud environment
-

UML DIAGRAM



INTER-SUBSYSTEM AND INTER-SUB-SUBSYSTEM INTERACTIONS

The main interactions between the sub-systems and sub-subsystems have already been discussed. To recap these interactions were between the Pis and microphones, the Pis to each other, and the Pis to the parent PC. In summary:

Sound Acquisition System

- Uses I2S to connect MEMs microphones to Raspberry Pi.
- Sound is captured, filtered and other preprocessing is done to it.
- Sound is saved as a ".wav" file and then sent for processing.

Processing

- Sound is sent to the parent for processing in specified software.
- Sound is processed using triangulation and coordinates are located.

Output/Display

- Coordinates are plotted using a GUI. The GUI shows a graphical representation of the grid and the position of the sound.

Communication

- SSH server for data transmission
- NTP for time synchronisation
- I2S for communication between the Raspberry Pis and the MEMs mics.

ACCEPTABLE TESTS

PI SYNCHRONIZATION

A) TEST PROCEDURES

Test the time clock and compare times.

B) EXPECTED RESULTS.

Pis should be synced to the same time.

C) RECORD OF ACTUAL RESULTS

	Checklist
Synced Clocks	

SIGNAL ACQUISITION

A) TEST PROCEDURES

- Microphone Array
 - Produce a sound.
- Noise Reduction
 - Calculate SNR before and after signal pre-processing.

B) EXPECTED RESULTS.

- Microphone Array
 - All microphones should detect the sound.
- Noise Reduction
 - SNR should be higher after the pre-processing of the signal.

C) RECORD OF ACTUAL RESULTS

	Checklist
Simultaneous Recording	
Improved SNR	

DATA ACQUISITION

A) TEST PROCEDURES

- Check if data is transferred through I2S
- Check if data is sent to the parent

B) EXPECTED RESULTS.

- Pis receive data through I2S
- Parent PC receives data over SSH

C) RECORD OF ACTUAL RESULTS

	Checklist
Parent PC receives data	
Pis Receive data	

SIGNAL SYNCHRONIZATION

A) TEST PROCEDURES

- Place sound in the middle of the grid

B) EXPECTED RESULTS.

- Sound should reach each mic at the same time

C) RECORD OF ACTUAL RESULTS

	Checklist
Synchronised Sound detection	

TIME DELAY ESTIMATION

A) TEST PROCEDURES

- Using a point with known TDoA values, compare the calculated value to the expected result
- Place the microphone in the exact centre of the grid
- Check if numerical values are produced by MATLAB

B) EXPECTED RESULTS.

- The calculated TDoA should be within $\pm 5 \mu\text{s}$ of the expected value
- All the TDoA values should be 0
- MATLAB should produce a numerical value for the TDoAs

C) RECORD OF ACTUAL RESULTS

	Expected value	Actual value	Error
Accuracy of TDoA			
Center point of TDoA			
Numerical Value			

TRIANGULATION

A) TEST PROCEDURES

- To check the output of the algorithm gives a grid value of $\pm 10\text{cm}$.
- To check the algorithm takes less than 5 seconds to compute output grid values.

B) EXPECTED RESULTS.

- The grid value computed is $\pm 10\text{cm}$ within the actual grid point.
- The algorithm computes grid points in less than 1 second

C) RECORD OF ACTUAL RESULTS

	Expected value	Actual value	Error
Runtime of algorithm			
Grid point computed			

USER INTERFACE

A) TEST PROCEDURES

- GUI displays a grid.
- GUI displays the coordinates of the sound source.
- GUI displays a dot on the grid where the sound source is located.

B) EXPECTED RESULTS.

- GUI displays a simple 50×50 grid with a labelled axis.
- GUI displays the written (x,y) coordinates on the laptop screen
- GUI displays a dot that represents a grid point where the sound source is located.

C) RECORD OF ACTUAL RESULTS

	Checklist
GUI labelled grid display	
GUI coordinate display	
GUI dot on grid matching coordinates	

INTER-SUBSYSTEM AND INTER-SUB-SUBSYSTEM INTERACTIONS

- A) TEST PROCEDURES**
- Check if all subsystems are communicating through relevant protocols.
- B) EXPECTED RESULTS.**
- I2S, NTP and SSH are working.

C) RECORD OF ACTUAL RESULTS

	Checklist
I2S, NTP and SSH are working.	

DEVELOPMENT TIMELINE

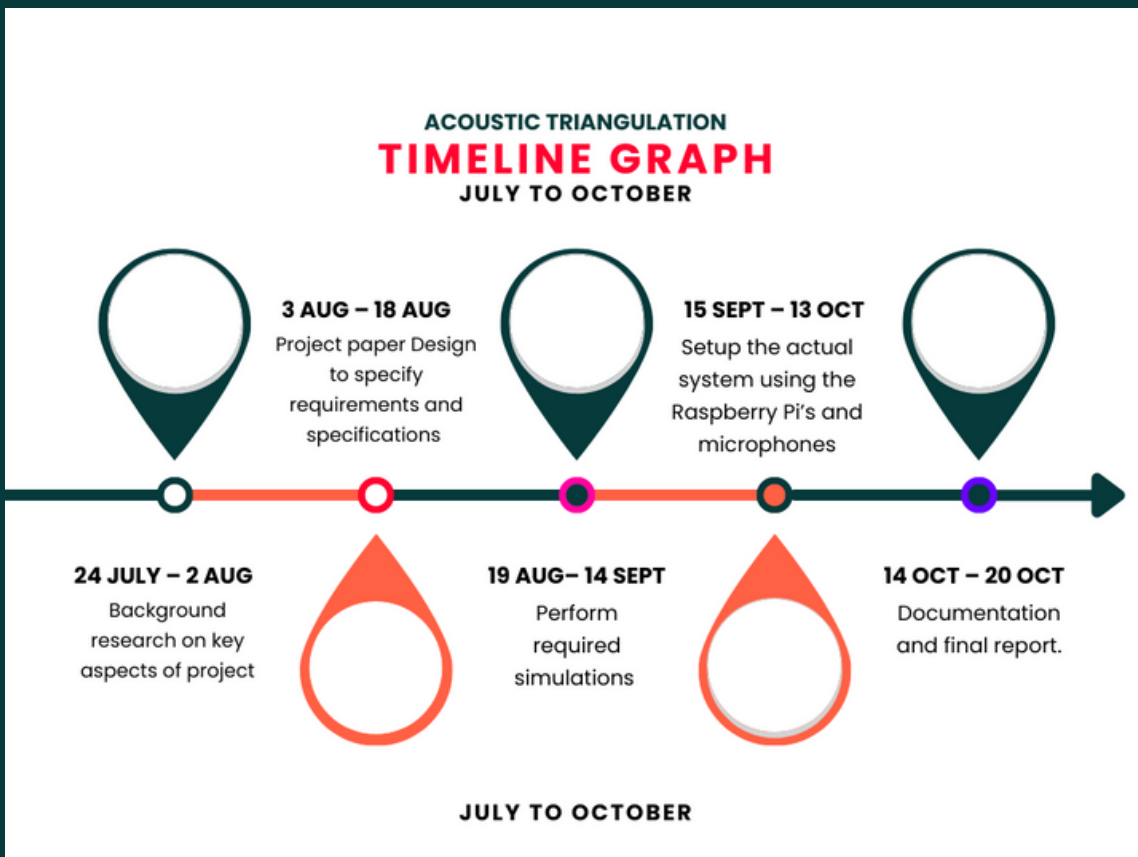
24 July – 2 August - Background Research: Perform research regarding the key aspects of the project, including the techniques and algorithms that can be implemented to achieve acoustic triangulation using Time Difference of Arrival and a distributed sensor network.

3 August – 18 August – Project Design: Perform a requirement analysis, develop the specifications, breakdown the system into various subsystems and sub-subsystems, provide the inter and intra subsystem interactions and develop acceptance test procedures.

19 August – 14 September – Simulation setup: Perform the required simulations on simulation software.

15 September – 13 October – System setup: Setup the actual system using the Raspberry Pi's, Ensure the microphones are communicating with the Raspberry Pi's.

14 October – 20 October – Documentation and final report: Document the entire project. Including the simulation results, hardware implementation as well as other important features of the project.



PROJECT MANAGEMENT PAGE

The screenshot displays the Monday work management interface. The top navigation bar includes the Monday logo, a 'See plans' button, and user profile icons. The left sidebar contains navigation links for 'Home', 'My work', 'Main workspace', a search bar, and a 'Timeline' tab. The main content area is titled 'Timeline' and features a 'Main Table' tab. Below the title, there are controls for 'New Task', search, and filters. The table lists tasks with columns for checkboxes, task names, owners, due dates, status, notes, and a plus icon. The tasks include individual assignments and milestones. A legend at the bottom indicates the color coding for task status: green for 'Done', orange for 'Working on it', and grey for 'Not Started'.

	Task	Owner	Due Date	Status	Notes	+
<input type="checkbox"/>	Zuhayr Matlab	ZL	Aug 14	Done		
<input type="checkbox"/>	Busi Matlab		Aug 14	Working on it		
<input type="checkbox"/>	Nene Matlab	NK	Aug 14	Done		
<input type="checkbox"/>	Milestone 1	+2	Aug 14	Working on it		
<input type="checkbox"/>	Milestone 2	+2	Sep 14	Not Started		
<input type="checkbox"/>	Milestone 3	+2	Oct 13	Not Started		
<input type="checkbox"/>	Milestone 4		Oct 20	Not Started		
<input type="checkbox"/>	+ Add task					

Legend: Aug 14 - Oct 20

- Done (Green)
- Working on it (Orange)
- Not Started (Grey)

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