MASTER REPORT

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Localization and tracking of moving targets by microphones.

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Abstract

The aim of this thesis is to develop a sound localization and tracking system using a four-microphone array to determine the direction and approximate distance of a sound source. Sound localization involves identifying the position of a sound-emitting object, which has diverse applications in fields such as robotics, surveillance, and automated systems. In this project, four microphones are strategically placed at the corners of a 17cm x 17cm square to capture sound waves. The Time Difference of Arrival (TDOA) method is employed to calculate the time it takes for sound waves to reach each microphone, thereby estimating the direction and distance of the sound source. The system uses the calculated time differences and sound intensities to activate an LED corresponding to the nearest microphone, providing a visual indication of the sound's direction.

A threshold-based approach ensures that only significant sound events are detected, reducing the influence of background noise. The estimated distance to the sound source is calculated using the speed of sound and the time differences, providing real-time feedback. The system has been tested in various controlled environments, demonstrating its ability to consistently identify the direction of the sound source and provide reasonable distance estimates.

This research outlines the design and implementation of the sound localization system and discusses its performance, highlighting its potential applications in enhancing the capabilities of autonomous systems. Future work could involve refining the distance estimation technique and exploring the system's adaptability to different environmental conditions and sound types.

Keywords: *acoustics; sound source localization; robotics; artificial intelligence; microphone arrays*

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Abbreviations

2D	Two-Dimensional
3D	Three-Dimensional
DOA	Direction of Arrival
TDOA	Time Difference of Arrival
TOA	Time of Arrival
FDOA	Frequency Difference of Arrival
ML	Machine Learning
GPS	Global Positioning System
TFT	Thin Film Transistor
SPI	Serial Peripheral Interface
AUV	Autonomous Underwater Vehicles

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1. Introduction

Sound localization is the process of determining the origin of a sound source within an environment, a capability that is crucial for a wide range of applications such as robotics, surveillance, human-computer interaction, and autonomous navigation. Understanding where a sound originates can significantly enhance the situational awareness of autonomous systems, enabling them to respond dynamically to changes in their surroundings. For example, in robotics, knowing the location of a sound can help a robot navigate towards a sound-emitting object, like a human calling for assistance. In surveillance, it can help pinpoint the exact location of unexpected sounds, such as breaking glass or unauthorized access (1).

This thesis focuses on developing a sound localization and tracking system using a square array of four microphones (2). Each microphone is positioned at the corners of a 17cm x 17cm square, capturing sound waves and using the Time Difference of Arrival (TDOA) method to determine the direction and distance of the sound source. TDOA is widely recognized for its effectiveness in localizing sound sources by measuring the slight differences in arrival times of sound waves at each microphone (3; 4). By comparing these time differences and the intensity of the sound signals, the system can identify the microphone closest to the sound source, providing real-time feedback via LEDs. This approach not only localizes the sound source but also estimates its distance, offering a comprehensive solution for tracking moving targets using sound. The potential applications of this system span from enhancing the capabilities of service robots to improving the security of automated monitoring systems.

1.1. Importance and applications of sound localization

Sound localization is a critical capability in both natural and engineered systems, providing valuable information about the environment by identifying the origin of sound sources. In humans and animals, sound localization is essential for survival, enabling the detection of predators, prey, or other environmental cues. For engineered systems, sound localization plays a significant role in enhancing the functionality and interaction of technologies such as robotics, surveillance, and virtual reality.

One of the primary applications of sound localization is in robotics, where it helps robots navigate and interact with their environment more effectively. Robots can use sound localization to locate and respond to sound sources, such as human voices, which is particularly useful in-service robots, autonomous vehicles, and search and rescue operations. In surveillance systems, sound localization enables the detection and identification of unauthorized intrusions, gunshots, or other unusual sounds, enhancing security measures and response times.

Another important application is in hearing aids and auditory prosthetics, where sound localization helps improve the quality of life for individuals with hearing impairments by enabling them to identify the direction of sound sources. This capability is crucial for safe navigation and social interaction. In virtual reality (VR) and augmented reality (AR), sound localization adds realism by allowing users to perceive sound sources accurately within immersive environments, enhancing the overall user experience.

In the military and defense sectors, sound localization is used to detect and track enemy movements or locate the source of gunfire, providing tactical advantages in combat situations. Additionally, environmental monitoring employs sound localization to track wildlife, monitor urban noise pollution, and study natural phenomena, contributing to conservation and public health efforts.

These applications highlight the broad utility and importance of sound localization in various fields, underscoring its value in enhancing situational awareness, safety, and interaction in both every day and specialized contexts.

1.2. Objectives of the research

The primary objective of this research is to develop and implement a sound localization and tracking system using a four-microphone array to accurately determine the direction and distance of a sound source. The specific goals of the research are outlined as follows:

- **Design and develop a microphone array system.**
  To create a hardware setup consisting of four microphones arranged in a square configuration, each positioned at the corners of a 17cm x 17cm square, to capture sound waves from various directions.

- **Implement the Time Difference of Arrival (TDOA) method.**
  To utilize the TDOA technique to calculate the differences in sound wave arrival times at each microphone, enabling precise estimation of the sound source's direction.

- **Visual indication of sound direction**.
  To integrate LEDs into the system that will be activated based on the nearest microphone, providing a real-time visual indication of the direction from which the sound is originating.

- **Estimate the distance of the sound source**.
  To develop algorithms that calculate the approximate distance of the sound source from the microphone array using the measured time differences and the known speed of sound.

- **Evaluate system performance**.
  To conduct experiments in various controlled environments to test the accuracy and reliability of the system in localizing sound sources. This includes assessing the system's ability to detect and respond to different types of sound events.

- **Explore potential applications**.
  To identify and discuss potential applications of the developed sound localization system in fields such as robotics, surveillance, human-computer interaction, and assistive technologies.

By achieving these objectives, the research aims to contribute to the field of sound localization by demonstrating a practical, accessible, and effective solution for real-time sound source tracking and localization.

2. Literature Review

2.1. Overview of existing sound localization techniques

Sound localization refers to the process of determining the origin of a sound in a given environment. Various techniques have been developed to achieve accurate sound localization (5), each with its own strengths, limitations, and applications. The most commonly employed methods are Time Difference of Arrival (TDOA), Frequency Difference of Arrival (FDOA), and beamforming.

2.1.1. Time Difference of Arrival (TDOA)

TDOA calculates time differences in sound wave arrivals at multiple microphones to estimate the sound source's direction and distance. It's simple and effective in open or controlled environments, with accuracy depending on microphone spacing, sound speed, and precise timing.

2.1.2. Frequency Difference of Arrival (FDOA)

FDOA uses frequency shifts from the Doppler effect to track moving sound sources, offering direction and velocity information. It requires complex signal processing and is sensitive to noise and reflections, impacting accuracy.

2.1.3. Beamforming

Beamforming uses an array of microphones to focus on sound from a specific direction, reducing noise and interference. It's widely used in applications like conference calls and sonar, offering high accuracy but requiring significant computational resources.

2.1.4. Machine learning approaches

Machine learning techniques use neural networks trained on sound data for localization, adapting to complex environments. These methods provide high accuracy but need extensive training data and computational power, suitable for advanced applications like autonomous vehicles. However, these methods often require extensive training data and computational power, which can be a limiting factor (4).

2.1.5. Hybrid techniques

Hybrid techniques combine multiple methods, such as TDOA with beamforming, to improve localization accuracy in noisy or obstacle-rich environments. While offering enhanced reliability, these approaches increase system complexity and computational demands (3).

2.2. Comparison of existing methods

Sound localization techniques are crucial in various applications, from robotics to surveillance, and different methods have been developed to achieve accurate localization.

1. Time Difference of Arrival (TDOA)

TDOA is a method for sound localization that estimates the direction of a sound source by measuring the time differences in sound wave arrival at multiple microphones, requiring precise synchronization and accurate timing.

Strengths

- **Simplicity:** Cost-effective and easy to implement with basic hardware.
- **Effective in controlled environments**: Accurate where sound travels without interference.
- **Real-time processing**: Provides immediate localization, suitable for responsive applications.

Limitations

- **Sensitivity to noise**: Prone to errors from environmental noise and echoes.
- **Requires multiple sensors**: Needs several well-placed microphones for accuracy.

2. Frequency Difference of Arrival (FDOA)

FDOA uses the Doppler effect to detect frequency shifts in sound waves, estimating the direction and speed of a moving sound source.

Strengths

- **Tracks movement:** Ideal for identifying moving sound sources.
- **Noise resistant:** Less affected by background noise.

Limitation

- **Complex signal processing**: Needs advanced algorithms.
- **Limited by stationary sources**: Less effective if sound source isn't moving.

3. Beamforming

Beamforming uses a microphone array to focus on sound from specific directions by adjusting signal phase and amplitude, enabling spatial filtering.

Strengths

- **High precision**: Accurately determines sound direction.
- **Noise reduction**: Effectively isolates sound and reduces background noise.

Limitations

- **Computationally intensive**: Requires high computational resources.
- **Array dependency**: Effectiveness depends on microphone placement and configuration.

4. Machine learning-based localization

Machine learning uses neural networks trained on large datasets to recognize patterns for sound localization, adapting to dynamic environments.

Strengths

- **Adaptable**: Learns and improves with data, suited for diverse environments.
- **Highly accurate**: Capable of precise localization in complex scenarios.

Limitations

- **Data-intensive**: Needs extensive training data.
- **Computationally complexity**: Requires significant processing power, limiting use in real-time or low-power setups.

3. Methodology

The methodology for this research involves the design, implementation, and testing of a sound localization system using a four-microphone array. The system utilizes the Time Difference of Arrival (TDOA) method to estimate the direction and distance of a sound source (6) by capturing sound waves and analyzing the time differences at which they reach each microphone. This approach requires a well-structured setup to ensure accurate data collection and processing (7). To achieve this, the methodology is structured into three main components: System Design, Algorithm Development, and Signal Processing. Each component plays a crucial role in developing a robust and effective sound localization system, enabling accurate and reliable localization and tracking of sound sources.

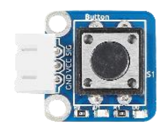
3.1. System Design

The system's design involves setting up a square configuration of four microphones, each placed at the corners of a 17cm x 17cm square. This arrangement allows the system to capture sound waves from different directions efficiently. The hardware setup includes an Arduino microcontroller for processing, four microphones for capturing sound signals, LEDs to indicate the direction of the sound, and a buzzer for audible alerts. Each component is strategically selected and configured to ensure reliable performance and integration.

3.1.1. Description of the hardware setup

The hardware setup for the sound localization system is designed to facilitate the accurate detection, processing, and localization of sound sources using a simple yet effective configuration. The setup comprises various components, each playing a critical role in the system's functionality. Below is a description of the hardware used:

Hardware Name	Description	Quantity	Image
Arduino Uno	Acts as the system's brain, processing microphone inputs, calculating time differences, and controlling LEDs and buzzer.	1	
USB A – USB B Cable	Connects Arduino to a computer for power and programming.	1	
LM393 Sound Detection Sensor Modules	Detects sound intensity and converts it to analog signals, arranged to measure time differences.	4	

LEDs with Resistors	Provides visual feedback by lighting up to indicate the direction of the detected sound.	4	
Breadboard for Prototyping	Allows for easy assembly and modification of the circuit without soldering.	1	
Active Buzzer Module	Provides an audible alert when sound is detected.	1	
Button Switch	Enables manual control for resetting or activating/deactivating the system.	1	
Jumper Wires	Connects components to the Arduino on the breadboard.	30	
Computer for Programming	Used to program and debug the Arduino using the Arduino IDE.	1	

3.2. Algorithm Development

The core functionality of the system is enabled through algorithm development, focusing on the implementation of the Time Difference of Arrival (TDOA) method. The algorithm is designed to measure the slight differences in sound wave arrival times at each microphone. By analyzing these time differences, the system can determine the closest microphone to the sound source. Additionally, the algorithm includes steps for calculating the estimated distance of the sound source based on the TDOA measurements, using the known speed of sound for accurate estimations (8).

3.2.1. Explanation of the code for detecting and localizing sound

The core of the algorithm involves continuously reading the analog signals from the four microphones, which are arranged in a square configuration. The Arduino continuously monitors these signals to detect sound events that exceed a predefined threshold, indicating a significant sound source. When a sound event is detected, the algorithm records the time at which each microphone picks up the sound. This allows the system to determine which microphone detected the sound first and helps in localizing the sound source (9). The LED corresponding to the nearest microphone is activated to provide visual feedback, while the buzzer provides an audible alert.

3.2.2. Formula for use of TDOA for localization

To localize the sound source, TDOA measures the time difference between sound wave arrivals at each pair of microphones. Using these time differences, the system can infer the direction from which the sound originated.

TDOA Formula

$$\Delta t_{ij} = t_i - t_j$$

The symbols in the above equation represent:

- Δt_{ij} : Time difference of arrival between microphones i and j.
- t_i : Time of arrival of the sound at microphone i.
- t_j : Time of arrival of the sound at microphone j.

The above formula calculates the time difference between each pair of microphones, which is crucial for determining the direction. By comparing Δt_{ij} values for different pairs, the system can estimate the direction of the sound source relative to the microphone array.

3.2.3. Steps to Calculate Distance Using TDOA

The distance estimation process involves calculating the time it takes for sound to travel from the source to the microphone and using the known speed of sound. The following steps outline how to calculate the distance using TDOA:

1. **Detect Sound Arrival Time**: Measure the arrival time t_i of the sound wave at each microphone. This is done by recording the timestamp when each microphone detects the sound.

2. **Calculate Minimum Detection Time**: Identify the microphone with the earliest detection time (t_{min}) . This microphone is the closest to the sound source.

$$t_{min} = min(t_1, t_2, t_3, t_4)$$

3. **Compute Time Differences**: Calculate the time differences between the earliest detection time and the detection times of the other microphones.

$$\Delta t_i = t_i - t_{min}$$

   These time differences (Δt_i) help to determine the relative position of the sound source.

4. **Estimate Distance Using TDOA**: Use the speed of sound (v) to convert the minimum detection time (t_{min}) into distance. Assuming a direct path from the sound source to the microphone, the distance (d) to the sound source can be estimated as:

$$d = t_{min} \times v$$

The symbols in the above equation represent:

- v: Speed of sound in $\text{cm}/\mu s$ (typically 0.0343 $\text{cm}/\mu s$ or 343 m/s).

3.2.4. Application in code

In the provided code, these principles are applied to estimate the distance:

```
// Function to calculate the estimated distance to the sound source
float calculateDistance(unsigned long time1, unsigned long time2, unsigned long time3,
unsigned long time4) {
  unsigned long minTime = min(time1, min(time2, min(time3, time4)));
  float distance = minTime * speedOfSound; // Distance in cm based on TDOA and speed of sound
  return distance;
}
```

- **Detecting time**
  The times (t_1, t_2, t_3, t_4) are recorded when the sound is detected by each microphone.

- **Finding minimum time**
  The smallest time (t_{min}) is identified as the first detection time.

- **Distance calculation**
  This time is multiplied by the speed of sound to estimate the distance to the sound source.

3.3. Signal Processing

Effective signal processing is essential for accurate sound localization. Techniques used include averaging the readings from each microphone to reduce noise and improve reliability. A thresholding approach is applied to distinguish significant sound events from background noise, ensuring that only relevant signals are processed for localization (10). This step is crucial in minimizing false positives and enhancing the accuracy of the system's response to sound events.

3.3.1. Techniques used to process the microphone signals.

3.3.1.1. Averaging of Readings

- To minimize the impact of random noise and fluctuations in the environment, the system takes multiple readings from each microphone and calculates the average value. Averaging helps smooth out any irregularities in the sound data and provides a more stable input for further analysis.
- This process involves capturing a predefined number of samples (e.g., 100 samples) and calculating the mean value for each microphone. By doing so, the system reduces the impact of transient noise spikes that might otherwise lead to incorrect localization.

3.3.1.2. Thresholding for Event Detection

- To differentiate between meaningful sound events and background noise, a threshold value is defined. Only sound readings that exceed this threshold are considered significant and processed for localization. This technique effectively filters out low-intensity background noise and focuses on the more substantial sound events.
- The choice of threshold value is crucial; it must be high enough to ignore irrelevant noise but low enough to detect genuine sound sources. This value can be adjusted based on the ambient noise level of the environment where the system is deployed.

3.3.1.3. Time Difference of Arrival (TDOA)

- As a core component of the signal processing strategy, TDOA is used to calculate the time it takes for sound to reach each microphone. By measuring these time differences, the system can estimate the location of the sound source. TDOA requires precise time synchronization and accurate signal processing to ensure reliable localization.

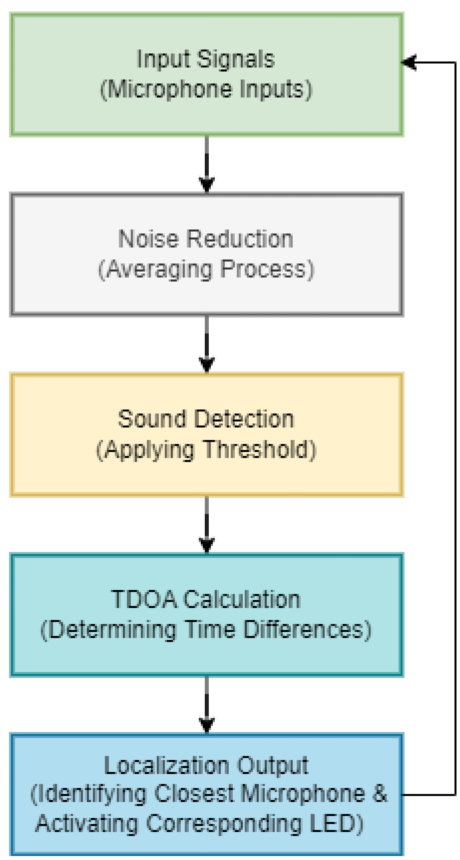


Figure 1. Techniques used to process the microphone signals.

4. Implementation

4.1. Hardware Implementation

4.1.1. Schematic diagram of the hardware connections.

The hardware implementation of the sound localization system begins with setting up the connections between the components, including the microphones, Arduino, LEDs, buzzer, and other peripherals.

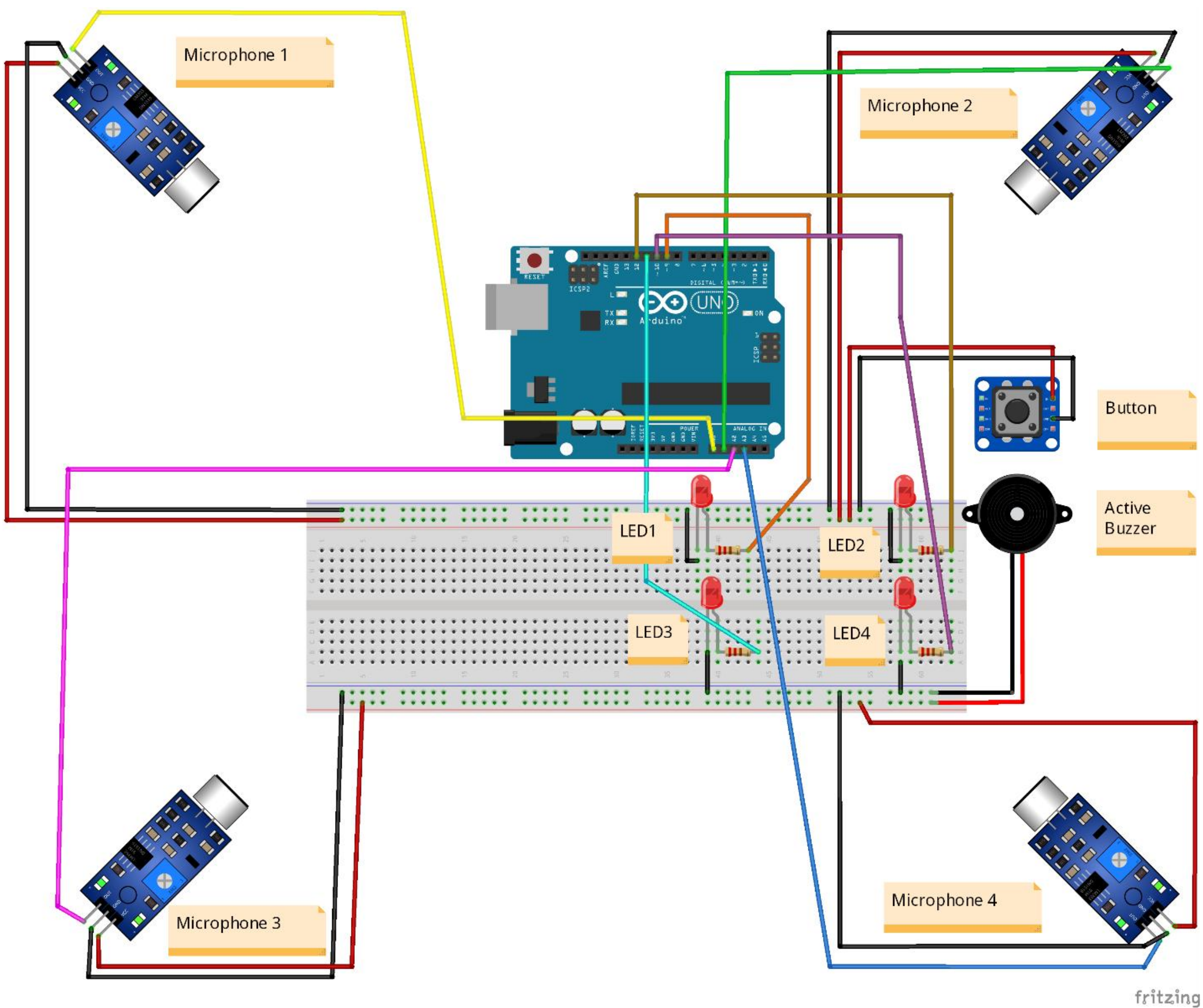


Figure 2. Schematic diagram of the hardware connections.

The diagram illustrates the following key components and their interconnections:

- **Arduino Uno**: Acts as the central controller, receiving input signals from the microphones and managing outputs such as LEDs and the buzzer.
- **Microphones**: Four LM393 sound detection sensor modules are connected to the analog input pins (A0, A1, A2, A3) of the Arduino. Each microphone is placed at a specific point to form a square configuration.
- **LEDs and Resistors**: Four LEDs, each paired with a resistor, are connected to digital output pins (9, 10, 11, 12) on the Arduino. These LEDs provide visual feedback corresponding to the microphone detecting the sound source.
- **Buzzer**: An active buzzer module is connected to a digital output pin (pin 8) to provide an audible alert when sound is detected.
- **Button Switch**: A button switch connected to pin 13 allows manual control over certain functions, such as resetting the system or silencing the buzzer.

4.1.2. Placement of microphones in the square configuration.

For effective sound localization, the microphones are arranged in a square configuration, with each microphone positioned at the corners of a 17 cm x 17 cm square. This geometric arrangement ensures that sound waves reaching the microphones from different directions will result in varying time delays, which are critical for the Time Difference of Arrival (TDOA) calculations.

- **Microphone 1 (Top-Right Corner)** : Placed at coordinates (0, 0).
- **Microphone 2 (Top-Left Corner)** : Positioned at (0, 17).
- **Microphone 3 (Bottom-Left Corner)** : Located at (17, 17).
- **Microphone 4 (Bottom-Right Corner)** : Situated at (17, 0).

This arrangement allows the system to effectively capture sound waves from any direction within the 2D plane defined by the square.

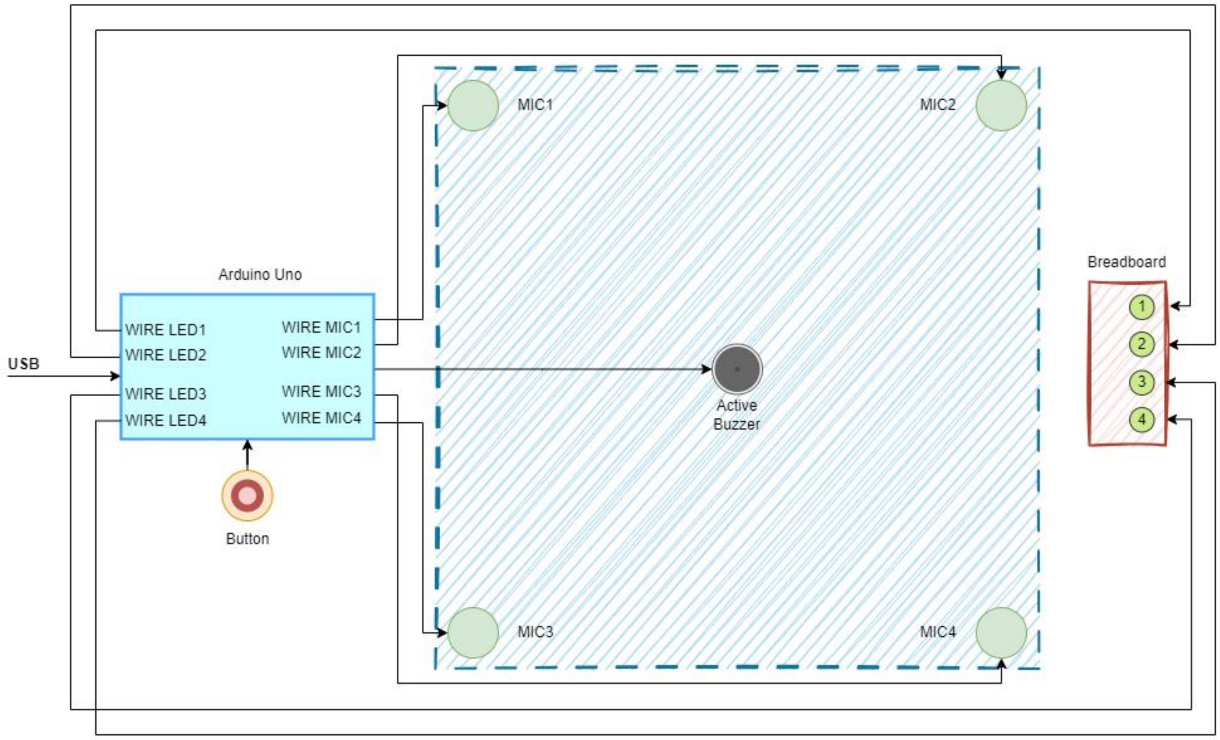


Figure 3. Microphones in the 17cm x 17cm square configuration.

4.2. Software Implementation

The software implementation involves setting up the development environment, installing necessary tools, writing the Arduino code, and running the program to achieve sound localization. This section covers the initial setup required to manipulate the hardware, provides an overview of the software components, and explains the specific role of the button and buzzer in the system.

4.2.1. Setting up the development environment.

To get started with the sound localization system, certain software installations and setups are required:

1. **Arduino IDE**: The primary tool used for coding and uploading programs to the Arduino board is the Arduino Integrated Development Environment (IDE). This software can be downloaded from the official Arduino website and is compatible with Windows, macOS, and Linux.

2. **Required libraries**: Although the project primarily uses standard functions provided by the Arduino environment, users should ensure they have the latest version of the Arduino libraries installed. These libraries handle input/output operations, serial communication, and timing functions. No additional external libraries are required for this project.

3. **Hardware connections**: Before uploading the code to the Arduino, ensure all components are connected as per the schematic diagram:
   - Microphones connected to the analog input pins (A0, A1, A2, A3).
   - LEDs connected to digital output pins (9, 10, 11, 12).
   - Buzzer connected to digital output pin 8.
   - Button switch connected to digital input pin 13 with a pull-up configuration.

4.2.2. How to start manipulating the hardware.

Once the software environment is set up and the hardware is connected:

1. **Connect Arduino to computer**: Use the USB A – USB B cable to connect the Arduino Uno to your computer. This connection will power the Arduino and allow code to be uploaded from the IDE.
2. **Uploading the code**: Open the Arduino IDE, load the code into the editor, and select the appropriate board (Arduino Uno) and port under the 'Tools' menu. Click on the 'Upload' button to compile and upload the code to the Arduino.
3. **Running the program**: After uploading the code, the system will start monitoring for sound events. The Arduino reads inputs from the microphones continuously. The program is designed to remain active, continually scanning for sound events and processing the data.
4. **Button and Buzzer functionality**: In this implementation, the button plays a critical role in initiating the sound detection process. When the button is pressed, it triggers the active buzzer to emit a sound. This sound serves as a controlled input that the microphones will detect. The purpose of this setup is to allow for easy testing and calibration of the system. By pressing the button, users can generate a sound source, which the microphones can then identify. The Arduino will measure the time it takes for each microphone to detect the buzzer's sound, perform the TDOA calculations, and activate the corresponding LED to indicate the closest microphone to the sound source.

4.3. Program Logic

The program logic outlines the core functionality of the sound localization system using pseudocode. This pseudocode provides a high-level overview of the steps the system takes to detect, localize, and respond to sound events.

Pseudocode for Sound Localization System

```
START
SET threshold = 150
SET speedOfSound = 0.0343 // cm/us
DEFINE micPositions for 4 microphones
INITIALIZE pins for sensor inputs, LED outputs, buzzer, and button

LOOP FOREVER
    IF button is pressed THEN
        TURN ON buzzer
    ELSE
        TURN OFF buzzer
    ENDIF

    SET time1, time2, time3, time4 TO -1
    SET detectedMic TO NONE

    FOR each microphone
        READ microphone values
        ADD readings to sum
        IF reading > threshold AND not yet detected THEN
            RECORD time of detection
            SET detectedMic to current microphone
        ENDIF
    END FOR

    FIND minimum detection time among all microphones
    SET distance = minimum detection time * speedOfSound
    ACTIVATE corresponding LED
    PRINT detected microphone and estimated distance
    WAIT 100 milliseconds
END LOOP

END
```

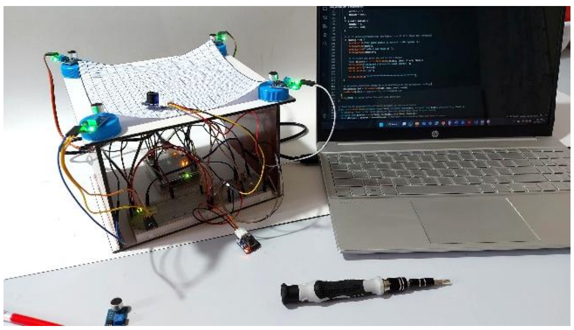
Figure 4. Program logic pseudocode

5. Experimental Setup

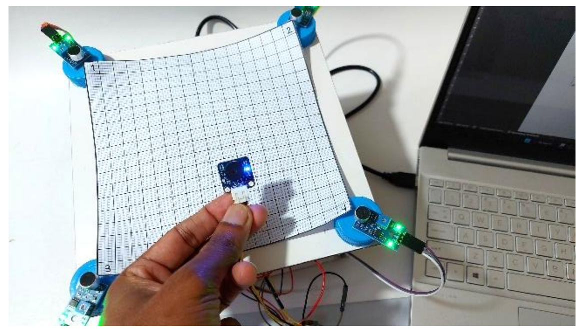
The experimental setup is designed to validate the functionality and accuracy of the sound localization system. This section outlines the steps involved in setting up the testing environment, the calibration process, and the various scenarios used to evaluate the performance of the system.

5.1. Environment setup for testing.

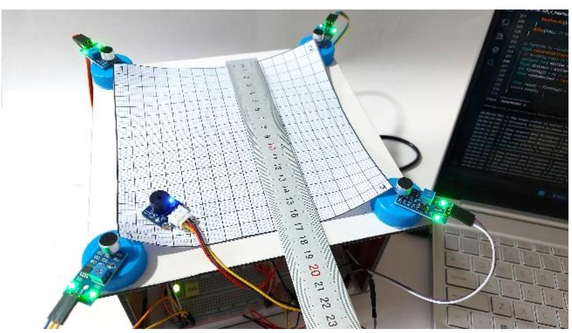
To ensure accurate and consistent results, the testing environment must be carefully controlled. The following considerations are made during the environment setup:



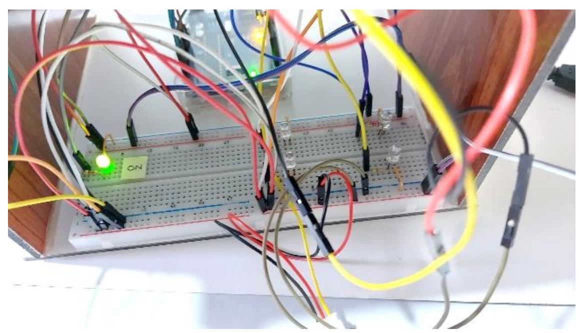
Testing is conducted in a quiet room to minimize background noise and avoid interference. This allows the system to focus on the primary sound source without distractions.



A controlled sound source, such as the active buzzer connected to the Arduino, is used to generate consistent sound events. This ensures that each test has a repeatable sound input, making it easier to compare results.



The four microphones are positioned in a square configuration, each at the corners of a 17cm x 17cm square. This placement is critical for accurately calculating the Time Difference of Arrival (TDOA) and localizing the sound source.



Adequate lighting is provided to clearly observe the LED indicators corresponding to each microphone, which light up to show the direction of the detected sound.

6. Preliminary testing with two microphones

6.1. Initial results with a simpler two-microphone setup.

The initial phase of this research involved testing sound localization using a basic setup consisting of only two microphones. These microphones were positioned at a fixed distance from each other, allowing for the detection of sound waves from a specific direction. The primary aim was to determine if a minimal configuration could provide reliable localization data. Initial results showed that the two-microphone setup could detect the direction of sound sources to some extent by calculating the Time Difference of Arrival (TDOA). However, this configuration was limited in its ability to accurately pinpoint the exact location of a sound source, especially when the source was located at angles equidistant from both microphones.

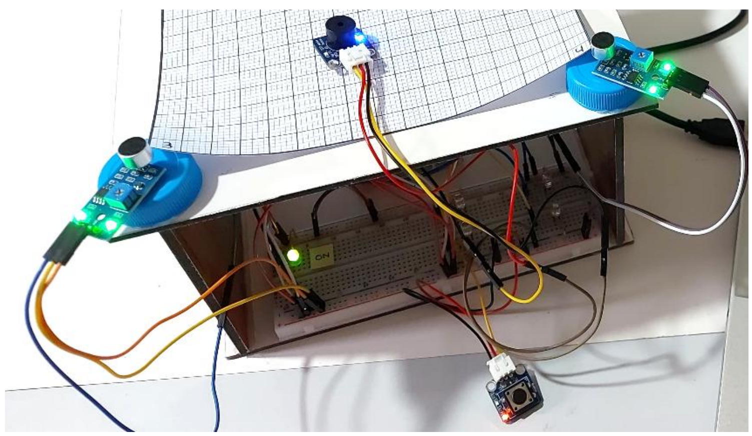


Figure 5. A simpler two-microphone setup.

6.2. Observations on the limitations and accuracy.

The two-microphone configuration showed significant limitations in accurately and reliably localizing sound. The limited spatial resolution of two microphones made it difficult to differentiate between sound sources from similar directions. Environmental noise and reflections further affected the TDOA calculations, introducing errors and inconsistencies. This setup struggled to determine sound source distance precisely, especially in acoustically complex environments, due to the insufficient directional information captured by just two microphones.

6.3. Rationale for expanding to a four microphone configuration.

Given the limitations of the two-microphone setup, expanding to a four-microphone configuration significantly improved localization accuracy. The square arrangement allowed for capturing sound from multiple angles, enhancing TDOA calculations. This approach provided better spatial resolution, reduced noise interference, and offered more reliable direction and distance estimation for practical applications.

7. Results

The experimental results provided valuable insights into the performance of the sound localization system. Data was collected from multiple experiments, focusing on the system's ability to accurately detect and localize sound sources using the four-microphone array setup. Key observations are summarized below:

7.1. Data collected from experiments.

During testing, various sound sources were introduced at different locations around the microphone array. The system recorded the time of arrival for sound waves at each microphone, allowing for the analysis of time differences and their impact on localization accuracy. Data was gathered over several trials to ensure consistency and reliability.

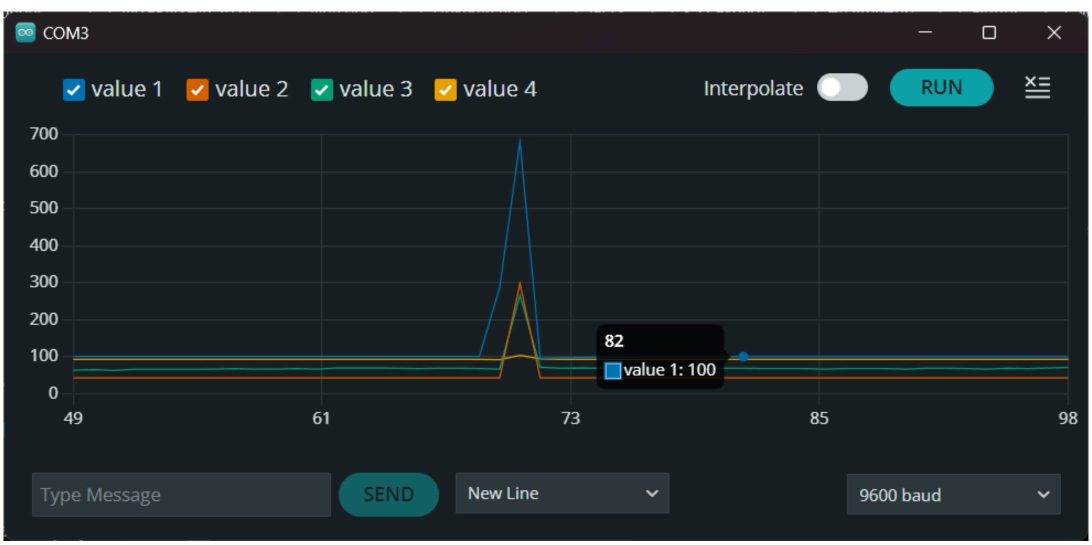


Figure 6. Sound intensity measured by the sensors (1)

Figure 7. Sound intensity measured by the sensors (2)

7.2. Data collection and frequency analysis in MATLAB.

Data was collected by recording the time of arrival and sound intensity at each of the four microphones. This information was crucial for determining the direction of the sound source using the Time Difference of Arrival (TDOA) method. MATLAB was used to analyze the frequency response of each microphone, offering insights into how they reacted to different sound frequencies.

While the frequency data analysis provided valuable information, it was not possible to plot the exact origin of the sound source from this data. Further development is needed to accurately map the sound's location, highlighting a direction for future research.

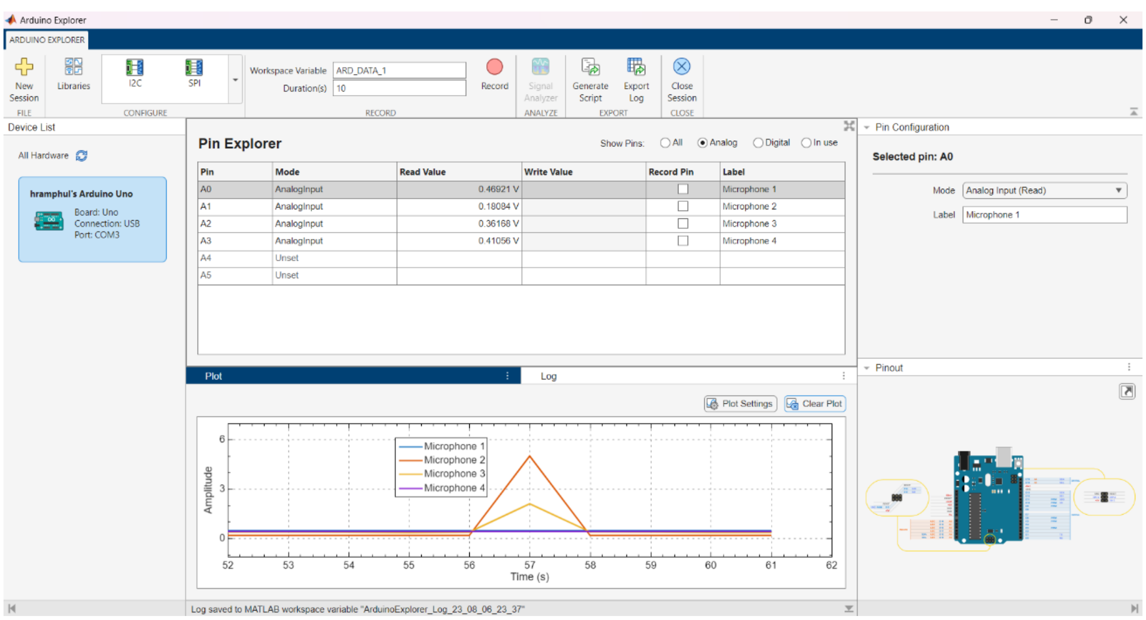


Figure 8. Data collection and frequency analysis in MATLAB.

7.3. Observations on the Time of Arrival Differences (TDOA)

The Time Difference of Arrival (TDOA) measurements consistently showed discernible differences between the microphones based on the direction of the sound source. These differences were critical in determining which microphone detected the sound first, thereby aiding in localizing the source. The timing accuracy proved sufficient for effective localization in most scenarios, demonstrating the effectiveness of the TDOA method.

```
Output    Serial Monitor ×
Message (Enter to send message to 'Arduino Uno' on 'COM3')
00:00:41.276 -> Time elapsed when Microphone 1 detected the sound: 464
00:00:41.341 -> Time elapsed when Microphone 3 detected the sound: 468
00:00:41.407 -> Time elapsed when Microphone 4 detected the sound: 476
00:00:41.440 -> The sound source is nearest to Microphone 4 with a sum value of 278
00:00:41.538 -> Estimated distance to sound source: 15.92 cm
00:00:41.571 -> ****************************************
00:00:43.390 -> Time elapsed when Microphone 1 detected the sound: 460
00:00:43.455 -> Time elapsed when Microphone 3 detected the sound: 464
00:00:43.520 -> Time elapsed when Microphone 4 detected the sound: 472
00:00:43.586 -> The sound source is nearest to Microphone 4 with a sum value of 249
00:00:43.652 -> Estimated distance to sound source: 15.78 cm
00:00:43.684 -> ****************************************
00:00:44.811 -> Time elapsed when Microphone 1 detected the sound: 456
00:00:44.843 -> Time elapsed when Microphone 3 detected the sound: 464
00:00:44.909 -> Time elapsed when Microphone 4 detected the sound: 468
00:00:44.974 -> The sound source is nearest to Microphone 4 with a sum value of 265
00:00:45.040 -> Estimated distance to sound source: 15.64 cm
00:00:45.106 -> ****************************************
```

Figure 9. Inaccuracy result with 4 microphones.

```
Output    Serial Monitor ×
Message (Enter to send message to 'Arduino Uno' on 'COM3')
16:03:34.961 -> The sound source is nearest to Microphone 3 with a sum value of 214
16:03:35.061 -> Estimated distance to sound source: 3.49 cm
16:03:35.093 -> ****************************************
16:05:57.393 -> Time elapsed when Microphone 1 detected the sound: 924
16:05:57.444 -> Time elapsed when Microphone 3 detected the sound: 460
16:05:57.491 -> Time elapsed when Microphone 4 detected the sound: 468
16:05:57.557 -> The sound source is nearest to Microphone 3 with a sum value of 202
16:05:57.623 -> Estimated distance to sound source: 3.73 cm
16:05:57.688 -> ****************************************
16:06:06.818 -> Time elapsed when Microphone 1 detected the sound: 924
16:06:06.862 -> Time elapsed when Microphone 3 detected the sound: 460
16:06:06.918 -> Time elapsed when Microphone 4 detected the sound: 468
16:06:06.982 -> The sound source is nearest to Microphone 3 with a sum value of 204
16:06:07.047 -> Estimated distance to sound source: 3.58 cm
16:06:07.113 -> ****************************************
```

Figure 10. Accurate result with 4 microphones.

8. Conclusion

The sound localization system developed in this research was successfully implemented and tested using low-cost materials. The project demonstrated the feasibility of using a four-microphone array and the Time Difference of Arrival (TDOA) method to localize and track sound sources. The system's ability to determine both the direction and distance of sound sources highlights its potential for practical applications in various fields. However, it is important to note that the accuracy of the results is not always optimal, suggesting the need for further refinement.

This project's success shows that accurate sound localization can be achieved without expensive or complex equipment, making it suitable for educational purposes, research initiatives, and hobbyist projects. The affordability and simplicity of the system enhance its appeal for environments where cost-effectiveness is a priority.

Future enhancements could focus on improving the system's robustness in noisy environments by integrating advanced noise reduction techniques. Further accuracy and adaptability might be achieved through more sophisticated signal processing algorithms or the use of machine learning. These improvements could address the limitations observed and help optimize the system's performance.

The versatility of this system makes it applicable in fields such as robotics, surveillance, and assistive technologies. Its interdisciplinary nature provides numerous opportunities for further exploration. As the project evolves, it has the potential to contribute significantly to advancements in sound localization and its practical applications.

8.1. Limitations and future work

While the system demonstrated effective sound localization, some limitations were identified. The accuracy of sound localization is sensitive to environmental noise, which may interfere with detection and timing measurements. Additionally, the accuracy of the results is not always optimal, highlighting the need for improvements. Future work could focus on enhancing noise reduction techniques and improving the robustness of the system against background noise. Expanding the system to include more microphones or integrating it with advanced signal processing algorithms could further improve localization accuracy and range. Exploring the use of machine learning models for sound localization could also offer new avenues for increasing system adaptability and precision in complex environments.

9. Bibliography references

1. **Wang, S.** *Sound Source Localization Using a Small Array of Microphones.* s.l. : Journal of Acoustics., 2019.

2. **Electronics, Learning about.** What is a Microphone Array? *http://www.learningaboutelectronics.com/Articles/What-is-an-array-microphone.* [Online] Learning about Electronics, 2023.

3. **Piotrowski, Gabriel Jekateryńczuk and Zbigniew.** *A Survey of Sound Source Localization and Detection Methods and Their Applications.* s.l. : MDPI, 2023.

4. **Purwins, Hendrik, et al.** *Deep Learning for Audio Signal Processing.* s.l. : IEEE Journal of Selected Topics in Signal Processing, 2019.

5. **Wikipedia.** ***https://en.wikipedia.org/wiki/Sound_localization.*** **[Online]**

6. Ying Liu, Yingmin Wang and Cheng Chen. *Efficient Underwater Acoustical Localization Method Based on TDOA with Sensor Position Errors.* s.l. : School of Marine Science and Technology, Northwestern Polytechnical University, Xi'an 710072, China**, 2023.**

7. Adam Wrobel, Michael Grisanti. ***Acoustic Impulse Marker.*** **s.l. : Cornell University, 2010.**

8. TÜRKMEN, Onur. *Automatic Camera Directing using Sound Source Localization.* **s.l. : EE491, 2023.**

9. Mantle, L., & Soroco, M. *Direction and Ranging of an Incoming Sound from an Arduino Sound Sensor System.* **s.l. : https://doi.org/10.14288/1.0398275, 2019.**

10. Wang, T., & Choy, Y. *An approach for sound sources localization and characterization using array of microphones.* **s.l. : International Conference on Noise and Fluctuations (ICNF)., 2015.**

11. GmbH, Fritzing. Fritzing. *https://fritzing.org/.* [Online] **Amtsgericht Berlin-Charlottenburg HRB 172704 B. https://fritzing.org/.**

10. Appendix

This section provides additional information to support a better understanding of this work.

10.1. The code & setup.

A small snippet of code has been written from scratch for the final setup. The complete source code, including both the four-microphone configuration ("localization_with_4_microphones.ino") and the initial two-microphone setup ("localization_with_2_microphones.ino"), is available on GitHub. These codes have been successfully tested and verified, demonstrating the progression from a simpler two-microphone system to the more robust four-microphone configuration. While the systems show potential, it's noted that the accuracy of the results is not always optimal, indicating areas for further refinement.

10.2. Demo video.

A comprehensive visual demonstration has been created to showcase the experimental setup, the functionality of the sound localization system, and the results. The demo video highlights the real-time detection and localization of sound sources, illustrating how the system activates the corresponding LED and provides distance estimations. This video serves as a valuable resource for understanding the practical implementation and effectiveness of the developed system.

- Demo 1
- Demo 2