Gazi University

2023-2024 Spring Term

Microprocessors Interdisciplinary Project

Preliminary Design Report

Software Requirements Specification (SRS) +

Hardware Requirements Document (HRD)

Smart Microphone: A Microphone array equipped with sound source localization capability

Computer Engineering Students

201180003 - Muhammet Buğra Güler

201180036 - Emrehan Çelik

C181152024 - Göktuğ Karaca

23118080904 - Diego de Luis Ballesteros

21118080742 - Furkan Tayyip Arfat

Electrical Electronics Engineering Students

191112008 - Seniha Aydın

201112035 - Muhammed Deha Kuramaz

151112007 - Mesut Furkan Baran

181110058 - Zikrullah Özdemir

28.04.2024

1. Introduction

This preliminary report presents the introduction and objectives of the "A Microphone array equipped with sound source localization capability" project. Today, sound detection and location technologies play an important role in many sectors. In this context, the smart microphone system to be developed aims to offer an alternative solution to traditional methods in determining the location of sound sources. The main aim of this project is to design and implement a sensitive and effective system to locate the sound source. This preliminary report includes the general purpose, scope, usage areas of the project and important concepts. In the following sections, topics such as system features, conceptual design, project team and project tasks will be discussed.

1.1. Objective

The main objective of this project is to provide an alternative solution to traditional methods used for locating sound sources. The intelligent microphone system to be developed aims to improve the accuracy of locating sound sources by offering higher resolution and sensitivity. In accordance with this, the specific objectives of the project are as follows:

- Overcome the limitations of current methods used to locate sound sources.
- Design and implement a system that determines the location of sound sources.
- Evaluate the effectiveness and accuracy of the designed system and make improvements to enhance its performance.

By achieving these objectives, the successful completion of the project will offer a more precise and reliable solution in the process of locating sound sources.

1.2. Scope

This project involves the design and implementation of an intelligent microphone system to detect the location of a sound source. Some critical components to be used in the project have been highlighted:

Choice of Arduino Microcontroller: The Arduino microcontroller has been chosen as the central control unit of the project. Arduino stands out for its solid platform, extensive developer community, and low cost. Therefore, Arduino will provide an economically efficient and cost-effective implementation.

Microphone Selection: Three microphones have been chosen to detect the sound source. These microphones offer cost advantages. The reasons for using three microphones in the project are based on various factors. First, theoretically, using at least two microphones to locate the sound source may be sufficient. However, in practical applications, increasing the number of microphones may be preferred to achieve higher accuracy and reliability. Using three microphones allows the detection of incoming sound waves at three different points using triangular geometry, enabling more precise location determination. Additionally, the use of three microphones provides a backup for identifying and correcting possible errors in the system. Finally, using three microphones represents an ideal balance between keeping costs low and providing high accuracy and performance. It should provide sufficient precision for the system to perform its basic functions effectively.

Choice of Bluetooth Module: The use of a Bluetooth module in this project has many advantages. First, it provides wireless communication, eliminating the use of complex cables in the system and facilitating installation. Additionally, its universal compatibility allows easy integration with other Bluetooth-compatible devices, providing access to a wide user base. Its low

power consumption feature extends battery life and allows for long-term use in portable devices. A wide coverage area increases the flexibility of system use and offers a wide range of motion for remote control or data exchange.
Use of LCD Screen: An LCD screen will be connected to the Arduino to provide a user interface. This screen will visually present the estimated location of the sound source to the users, facilitating interaction with the system.
Use of Assembly as Software Language: Assembly programming language will be used in the software development process of the project. Assembly is a low-level language that directly handles microcontroller commands. This will enhance system performance and efficiency while providing low-level hardware control.
Mobile Application: Unity has been chosen for its physics engine capabilities and ability to process data easily on mobile devices. A mobile application will be developed using Unity in C#.

The selection of these components is crucial for the successful completion of the project. These choices contribute to achieving desired outcomes by ensuring cost-effectiveness and technical optimization in the project.

1.3. Use Cases

The microphone array equipped with sound source localization capabilities has versatile applications. A microphone array can provide a mobile robot with the capability of localizing, tracking and separating distant sound sources in 2D, i.e., estimating their relative elevation and azimuth [1.3.1] If a robot can detect the direction of a person through his voice, the interactivity between them will be improved a lot [1.3.2]. Another use case is technical applications and production machines. They are perceived as disturbing noise. When trying to solve noise problems or refine the acoustic design of a product, knowledge of the position and distribution of sound sources is necessary [1.3.3]. A microphone array can be used to identify the areas of a device that produce significant acoustic emission. Additionally microphone array serves security and surveillance needs by detecting and localizing suspicious sound. At present, many countries have developed various types of equipment that can detect explosion site or location of shooting with the operations of counter-terrorism [1.3.4]. In search and rescue activities, unmanned aerial vehicles (UAV) should exploit sound information to compensate for poor visual information. A UAV-embedded microphone array system can be effective for the detection of people needing assistance in disaster-stricken areas [1.3.5].

1.4 Description and Abbreviation

Mbps : Mega bit per second.

ADC : Analog to digital converter. It is a system that converts an analog signal, such as a sound picked up by a microphone into a digital signal.

LCD: Liquid crystal Display. It is a type of flat panel display which uses liquid crystals in its primary form of operation.

DSP: Digital signal processing. Herein it stands for amplifier circuit.

TTL: Transistor - transistor logic.

IDE : Integrated Development Environment. A software application that helps programmers develop software code efficiently.

I/O: Input / Output

1.5 Preliminary Information

1.5.1 Microphone

A microphone is a transducer that converts sound signals (acoustic energy) into electrical signals. Also referred to as transducers, these components detect sound signals and convert them into electrical signals. Regardless of their structure, characteristics, or operating principles, microphones are primarily defined by their key component, the diaphragm. The diaphragm, a flexible membrane, is crucial as it vibrates in response to sound waves traveling through the air. Microphones are classified based on the structures behind the diaphragm.

1.5.1.1. Types of Microphones

Dynamic Microphones: Dynamic microphones are types of microphones that convert sound waves into mechanical vibrations and then into electrical signals. They operate by having sound waves vibrate a diaphragm. A coil surrounds this vibrating diaphragm and moves within a magnetic field. This movement creates electromagnetic induction, converting sound waves into electrical signals.

Dynamic microphones are popular due to their durability, simple designs, and wide range of applications. They are preferred for many applications such as stage performances, recording studios, radio broadcasting, and live events. They are also generally more affordable and known for their ability to handle high sound levels. However, they may experience some sensitivity loss at high frequencies and typically have lower frequency responses.

Electret Microphones: Electret microphones are one of the most commonly used microphone types and can be found in many consumer electronic devices. They are made using a special material layer called electret material. This electret material possesses a static electric charge, which enables the microphone to convert sound signals.

The working principle of electret microphones begins with sound waves striking the electret material. This causes changes in the charge within the electret material. Subsequently, these charge variations are detected by an electrode or capacitor inside the microphone and converted into an electrical signal. This electrical signal corresponds to the original sound waves.

1.5.1.2. Microphone Arrays

A microphone array is a cluster of multiple microphones placed at geometrically different points and operating simultaneously. Microphone arrays have been designed and developed for various purposes in audio processing systems. These purposes include:

• Separating the sound signal from ambient noise in noisy environments,

• Directing the array in multi-speaker environments,

• Enhancing the performance of hearing aids,

• Locating the position of sound sources,

• Automatically tracking speakers,

• Improving the performance of speech recognition systems.

1.5.1.3 Beamforming with Microphone Arrays

Beamforming, generally defined as a processor used in conjunction with an array of sensors to perform some form of spatial filtering. Spatial samples of waves propagating in space, captured by sensor arrays, are then processed by the beamformer. The aim here is to obtain only the desired signal coming from a specific direction in an environment with noise and interference. The beamformer performs spatial filtering to separate signals coming from different directions but having overlapping frequencies.

The foundation of beamforming lies in the delay of signals during propagation in space. The primary challenge for systems designed to receive signals propagating spatially is interference signals. If the desired signal and interference signal propagate at the same operating frequency, frequency filtering cannot be used to separate the desired frequency from the interference frequency. However, interference signals typically propagate from different spatial points. Leveraging this spatial difference, the spatial filtering method allows the separation of desired signals from interfering signals. Additionally, the beamforming process compensates for phase differences between microphones.

1.5.2 Arduino

Arduino is an open-source development platform that allows for the easy design of systems that can interact with their surroundings. This enables users to make modifications as they desire. Thanks to Arduino libraries, programming becomes straightforward. It can process analog and digital data through its analog and digital inputs. It can work with sensors, allowing the utilization of data from various sensors. Output to the external world such as sound, light, motion, text, images, etc., can be generated. Arduino facilitates the implementation of robotic and electronic applications. Moreover, the programming language used in Arduino is simple, making it easy to find a multitude of resources.

1.5.3 Directional Sound Localization

A multi-microphone array, commonly known as a microphone array, is a technology often used to determine the direction from which a sound originates. This system is created by combining multiple microphones and is used to perceive sound waves as emanating from a specific location.

Multiple microphones are arranged in a specific configuration.

The microphones detect and digitally record ambient sounds.

Time and phase differences between microphones determine the location of the sound source.

The calculated data is analyzed using mathematical operations or algorithms.

Based on the calculated data, the position of the sound source is determined.

1.5.4 Bluetooth

HC-05 is a popular Bluetooth serial communication module that facilitates wireless serial communication between Bluetooth-enabled devices. Widely utilized in microcontroller projects like Arduino, HC-05 is known for its low cost, ease of use, and large user base. The module features a set of I/O pins and typically employs RX (receiver) and TX (transmitter) pins for serial data transmission at TTL levels. HC-05 is commonly employed to meet wireless communication requirements in various projects.

1.5.5 Data Processing

Data Reading: Data from sensors, user inputs, or other devices is read by the Arduino. This can be done through various methods such as reading data from analog or digital inputs, receiving data via serial communication (UART, I2C, SPI), or receiving data wirelessly.

Data Processing: The read data is processed according to programmed algorithms. These processes may include mathematical calculations, decision structures, loops, and data conversions (e.g., converting analog data to digital values).

Control and Decision: Based on the processed data, the Arduino program controls various outputs or performs specific actions depending on certain conditions. For example, it can turn on a fan or an LED when a certain temperature threshold is exceeded.

Output Generation: The results of the processed data can be used as output to provide information to the user or control another device. This can include functions such as turning on LEDs, rotating motors, displaying information on screens, or sending data to another device wirelessly.

2. System Characteristics

2.1. Functional Requirements

The microphone array used to determine the location of a sound source is being designed on a mobile platform. A minimum of 2 microphones is required to determine the location of the sound source; however, 3 microphones are most ideal in terms of price performance, so we have decided to use 3 microphones. Among the hardware components to be used in the project are a microcontroller, microphones, microphone amplification circuits, a Bluetooth module, and an LCD screen. Since the audio signals from the microphones are generally weak, they need to be amplified through electronic circuits or hardware. Subsequently, the audio signals can be processed using appropriate algorithms, and the location of the sound source can be estimated.

Software requirements include reading and processing data from sensors, software for printing data to the LCD screen, control via a mobile application over Bluetooth, and visualizing the location information of the sound source through a user interface. The mobile application should enable the user to visually see the determined location of the sound source. These functional requirements enable the project to operate on a mobile platform capable of transmitting data via Bluetooth. The hardware and software components provide a user-friendly interface, allowing the accurate detection and visualization of the sound source, thereby ensuring the success of the project.

2.2. Performance Requirements

Sensitivity: The system is expected to detect the location of the sound source with a sensitivity of 0.1 degrees or better. This indicates how close the detected location of the sound source is to its actual position. High sensitivity is critical for accurate location detection and provides users with reliable results.

Accuracy: The difference between the estimated position of the sound source and its actual position should not exceed ±1 degree. This determines the reliability of the information provided by the system. Accurate location detection ensures that the system provides users with correct information and increases the level of reliability.

Reliability: The system should produce accurate results with a reliability level of 95%. This determines the reliability of the information provided by the system. Reliable results enable users to trust the system and enhance the success of the application.

Latency: The latency between detecting the sound source and visualizing the results should be a maximum of 100 milliseconds. This enables users to receive real-time feedback. Low latency enhances the user experience and ensures the effectiveness of the system.

Data Transfer Rate: A data transfer rate of at least 2 Mbps should be provided via Bluetooth or WiFi. Fast data transfer enables the quick processing and presentation of audio data from the microphones to the user. This enhances the performance of the system and prevents users from waiting.

Energy Efficiency: The system should maintain a daily average battery life of at least 10 minutes during active usage. This is important for preserving battery life in mobile devices. Energy efficiency allows users to use their devices for longer periods and enhances the user experience.

2.3.1. Software and Interface Requirements: (For Software Components)

Operating System Compatibility: The software should operate seamlessly on the Android operating system.

Interactions Between Systems:

- Interaction with Microcontroller: The software should communicate with the microcontroller and be able to receive data from the microphones. This enables the provision of necessary data for processing audio data to determine the location of the sound source.
- Communication with Bluetooth Module: The mobile application should be able to exchange data with the microcontroller via the Bluetooth module. This enables users to control the microphone array from their mobile device and receive processing results.

User interface:

- User Experience: The interface should enable users to easily control the microphone array and understand the processing results.

2.3.2. Hardware Requirements

2.3.2.1. Arduino UNO: In order to store and access data obtained from sensor and interpret data, a microcontroller is needed. Based on its ease of use, ease of accessibility and variety of information source, the use of Arduino Uno has been decided. The Arduino UNO is a widely used microcontroller board based on the ATmega328P microcontroller. It offers a user-friendly environment for prototyping and developing various electronic projects. Key features of the Arduino UNO include:

2.3.2.1. 1. Microcontroller: The ATmega328P microcontroller provides 14 digital input/output pins, 6 analog input pins, a 16 MHz quartz crystal, and programmable flash memory for storing code.

2.3.2.1. 2. Programming: Arduino UNO is programmed using the Arduino IDE (Integrated Development Environment) , which simplifies coding with its user-friendly interface and extensive library support. It will be useful for software part of this project.

2.3.2.1.3. I/O Interfaces: The Arduino UNO includes multiple I/O interfaces, such as UART, I2C, and SPI, allowing easy communication with peripheral devices.

2.3.2.1.4. Power Supply: The board can be powered via USB or an external power supply, providing flexibility for different applications

2.3.2.2. Energy : For proper operation of the Arduino Uno, a suitable power source such as a 9-12 volt battery or adapter can be used.

2.3.2.3. Bluetooth Module: The hardware requirements of the HC-05 Bluetooth module are quite low and it typically consumes a modest amount of resources suitable for microcontroller-based projects. The basic hardware requirements include:

2.3.2.4. Power Supply Voltage: The HC-05 module is usually powered with 3.3V or 5V. This may vary depending on the power requirements of your project, but it's generally recommended to operate it at 3.3V.

2.3.2.5. UART Connection: The HC-05 module utilizes the UART (Universal Asynchronous Receiver/Transmitter) protocol for serial communication. Therefore, your microcontroller or other device should have a UART connection to establish serial communication with the HC-05 module.

2.3.2.5. Antenna: The HC-05 module has an antenna for transmitting and receiving Bluetooth signals. The antenna needs to be connected for the module to operate efficiently.

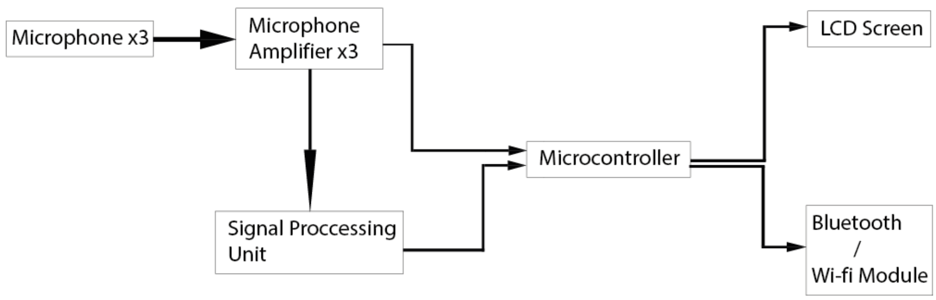
2.3.2.6. Connection Pins: The HC-05 module typically has standard connection pins such as VCC, GND, TX (Transmit), and RX (Receive) to establish connections with your microcontroller or other devices.

2.3.2.7. Electret microphones, consist of a capsule containing an electret capacitor. The capsule converts sound waves into mechanical vibrations, which are then converted into varying capacitance on the electret capacitor. Electret microphones typically require a pre-polarization voltage between 1.5 - 10 V due to the permanent polarization of the electret layer. The output signal of electret microphones converts the vibrations of sound waves in the microphone capsule into changes in voltage. This voltage typically varies between 10 mV/Pa to 100 mV/Pa depending on the sensitivity of the microphone. They are characterized by low power consumption. Typically, they consume power in the range of microamps depending on the operating voltage and sound levels. For example, an electret microphone may typically consume a current between 0.5 mA to 2 mA.

3. Concept Design

The Conceptual Design section outlines the technical composition of the "Smart Microphone" system, focusing on the microphone array setup, signal amplification and filtration, digital conversion through ADC, and sound processing with DSP. It also details the integration of a Bluetooth module, chosen for energy efficiency and connectivity, supported by visual diagrams for clarity.

3.1. Level 0 Design



System Input: The system utilizes three microphones to capture environmental sound signals. These microphones are configured to detect sound sources and are equipped with spatial sound collection capabilities.

Signal Amplification: The output from each microphone is amplified by microphone amplifiers for analog signal processing. These amplifiers enhance the quality of the signal and provide an appropriate level of input to the processing unit.

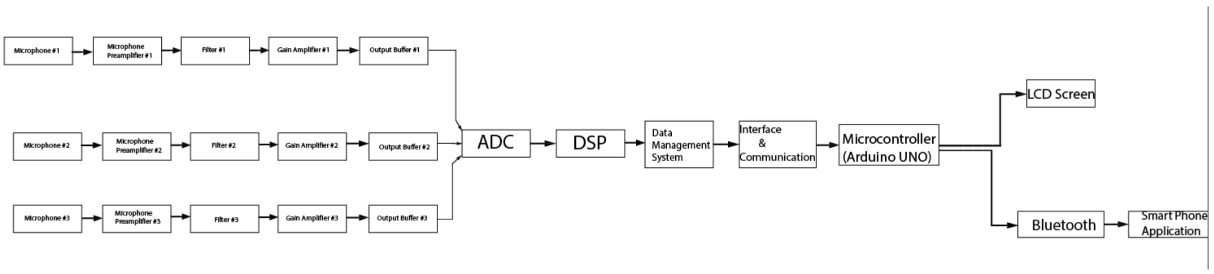
Signal processing unit: The amplified signals are then transmitted to the Signal Processing Unit for complex operations. This unit applies various algorithms aimed at improving the accuracy and utility of the sound signals.

Microcontroller: The processed signals are managed by a microcontroller to coordinate among various components within the system. The microcontroller takes charge of hardware control and visualizes the data necessary for user interaction.

User Interface: The microcontroller is connected to an LCD display to provide feedback about the system status and processing results. The user can see the location of the sound source and other important information on this screen.

Wireless : Finally, the microcontroller is connected to a Bluetooth/Wi-Fi module, enabling the system to adjust parameters remotely. This module adds the capability for wireless communication with smartphones and other devices to the system.

3.2. Level 1 Design



Level 1 design offers a detailed preliminary layout for the subsystems of the "Smart Microphone" project. It presents a modular structure that encompasses the processes of capturing acoustic signals, processing them, and interfacing with the user. Below is an explanation of the functionality of each module and how they integrate into the overall system.

Microphones and Pre-Amplifiers: Each microphone channel captures sound signals and processes them initially through a pre-amplifier. These pre-amplifiers enhance the signal while reducing noise.

Filters: Filters in each channel eliminate unwanted frequencies and emphasize specific frequency bands to improve the quality of the signal.

Gain Amplifiers and Output Buffers: Gain amplifiers further amplify the filtered signals. Following this, output buffers stabilize the signal to ensure it is transmitted smoothly to the ADC.

ADC (Analog-Digital Converter): The ADC module converts the analog signals into digital data. This conversion facilitates the processing of digital signals in the DSP module.

DSP (Digital Signal Processor): The DSP processes the digital signals using sound processing algorithms to determine the location of the sound source.

Data Management and Communication: The processed data are then managed and organized by the data management system before being presented to the user. The interface and communication module transfer this data to the microcontroller.

Microcontroller (ATmega328p): The microcontroller receives the processed data and utilizes it to display information on the LCD screen. The system benefits from the flexibility and accessibility advantages by using an Atmega328p.

Bluetooth Module:

Choosing the Bluetooth module over Wi-Fi offers several advantages:

- **Energy Efficiency:** Bluetooth typically consumes less power than Wi-Fi, which can extend the battery life for mobile devices.
- **Simple Pairing and Connection:** Bluetooth devices often have a more straightforward and user-friendly process for pairing and establishing connections.
- **Broad Compatibility:** Bluetooth enjoys wide compatibility across a variety of devices and platforms**.**
- **Prevalence in Devices:** Since most smartphones and tablets come with Bluetooth modules, they can easily be integrated with these devices.

4. References

[1.3.1] 3D Localization of a Sound Source Using Mobile Microphone Arrays Referenced by SLAM Simon Michaud, Samuel Faucher, François Grondin

[1.3.2] A Sound Source Localization System Using Three-Michrophone Array and Crosspower spectrum phase Hung-Yan Gu, Shan-Siang Yang

[1.3.3] Sound source localization-state of the art and new inverse scheme Stefan Gombots, Jonathan Nowak & Manfred Kaltenbacher

[1.3.4] Localization Estimation of Sound Source by Microphones Array Jing Fan , Qian Luo, Ding Ma

[1.3.5] Design of UAV-Embedded Microphone Array System for Sound Source Localization in Outdoor Environments, Kotaro Hoshiba , Kai Washizaki , Mizuho Wakabayashi

[1.3.5] IMPLEMENTATION OF SPEECH RECOGNITION HOME CONTROL SYSTEM USING ARDUINO Nurul Fadzilah Hasan , Mohd Ruzaimi Mat Rejab and Nurul Hidayah Sapar

[1.3.6] Sensing through signal amplification Paolo Scrimin and Leonard J. Prins

[1.3.7] A Low-Power Reconfigurable Analog-to-Digital Converter ,Kush Gulati

Arduino. "Arduino Official Website." [https://www.arduino.cc/]

Smith, John. (Jan 2011). "Advanced Techniques in Sound Localization." Journal of Acoustic Engineering.

5. Project Team and Tasks

5.1 Team Members and Task Allocation

Seniha Aydın → Meeting records, use cases (6-1.3)

Muhammed Deha Kuramaz → Level 0 and level 1 design (3-3.1-3.2)

Engin Ergül → Objectives, scope (1.1-1.2)

Zikrullah Özdemir → Preliminary information, hardware requirements (1.5 - 2.3.2)

Mesut Furkan Baran → Timeline, preliminary information (1.5-5.2)

Muhammet Buğra Güler → Mobile software (2.1)

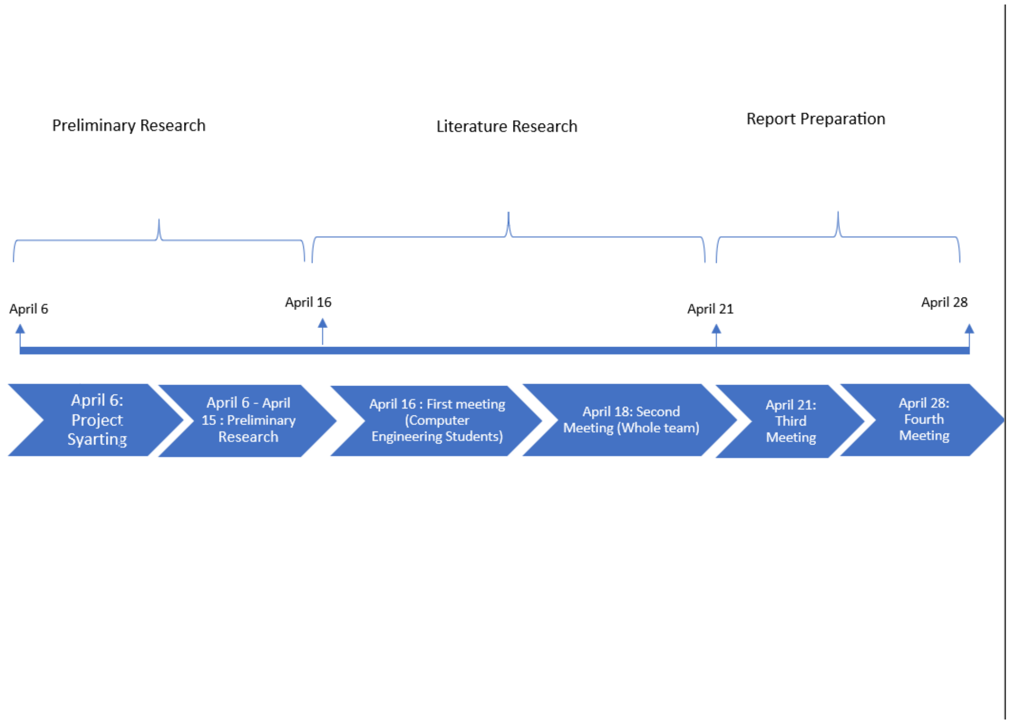
Göktuğ Karaca → Location detection using microphones (2.2)

Emrehan Çelik → Mobile software (2.1)

Diego de Luis Ballesteros → Data transfer via Bluetooth (2.3.1)

Furkan Tayyip Arfat → Use of LCD screen (2.2)

5.2 Timeline



6. Meeting Records

Meeting No: 1

Project Subject: A Microphone array equipped with sound source localization capability

Meeting Date: 16.04.2024

Meeting Agenda: Task allocation

Participants:

Muhammet Buğra Güler, Göktuğ Karaca, Emrehan Çelik, Diego de Luis Ballesteros, Furkan Tayyip Arfat

Meeting No: 2

Project Subject: A Microphone array equipped with sound source localization capability

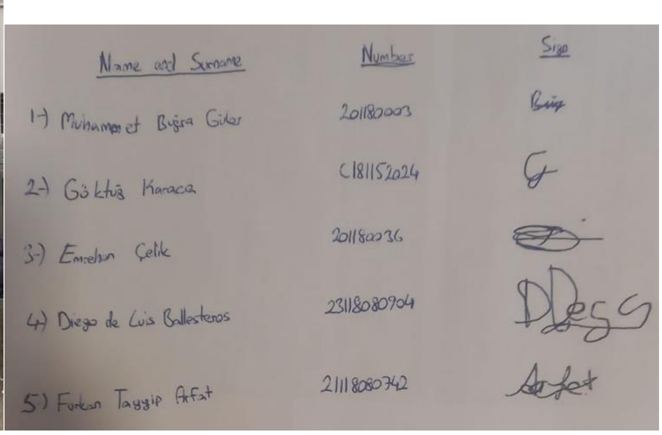
Meeting Date: 18.04.2024

Meeting Agenda: Hardware concept requirements, project timeline, task allocation

Participants

Muhammet Buğra Güler, Göktuğ Karaca, Emrehan Çelik, Diego de Luis Ballesteros, Furkan Tayyip Arfat (left early)

Evidence:



Name and Surname	Number	Sign
1-) Muhammet Buğra Güler	201180003	
2-) Göktuğ Karaca	C181152024	
3-) Emrehan Çelik	201180036	
4-) Diego de Luis Ballesteros	23118080904	
5) Furkan Tayyip Arfat	21118080742	

Meeting No: 3

Project Subject: A Microphone array equipped with sound source localization capability

Meeting Date: 21.04.2024

Meeting Agenda: Timeline, concept of the project

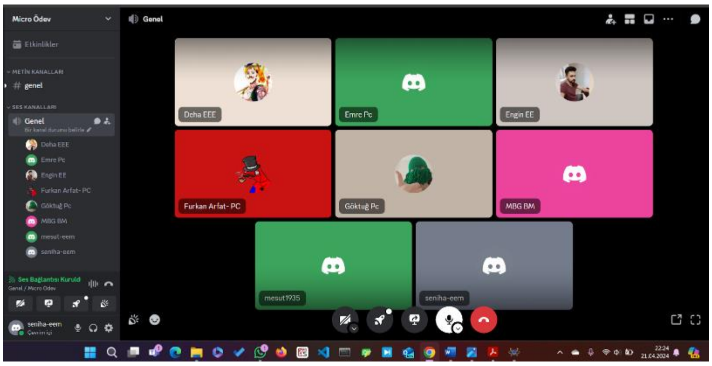
Decision Taken: 3 microphone, and Arduino Uno board will be used. For wireless communication functions a bluetooth module will be used. Task allocation decided as follows:

Participants

Muhammed Deha Kuramaz, Engin Ergül, Zikrullah Özdemir, Seniha Aydın

Muhammet Buğra Güler, Göktuğ Karaca, Emrehan Çelik, Diego de Luis Ballesteros, Furkan Tayyip Arfat

Evidence: The meeting was held on the Discord platform.



Meeting No: 4

Project Subject: A Microphone array equipped with sound source localization capability

Meeting Date: 28.04.2024

Meeting Agenda: The last revision of the report

Decision Taken: Concept design of the project completed. Level 2 design can be started.

Participants

Seniha Aydın, Muhammed Deha Kuramaz, Engin Ergül, Zikrullah Özdemir

Göktuğ Karaca

Evidence: The meeting was held on Discord platform.