PROJECT REPORT:

Embedded Systems II - AOA of Multidirectional Audio System

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1. Introduction

This Embedded Systems II project delves into developing a cost-effective and energy-efficient solution tailored for environments where resources are limited. By harnessing a multidirectional array of microphones, the system is designed to pinpoint the origins of sound through sophisticated yet resource-aware techniques.

Central to our implementation is the TM4C123GH6PM microcontroller, a device chosen for its great performance in handling multiple inputs and outputs while maintaining low power consumption. This microcontroller processes audio signals captured by the microphones, utilizing cross-correlation techniques to analyze the time difference of arrival (TDoA) of sound waves at each sensor. The determination of these time differences is crucial for accurately calculating the direction from which the sound originates.

Our system is uniquely designed to identify and process valid acoustic events—those instances of sound that meet predefined thresholds of intensity and clarity, thus ensuring that only significant audio cues are considered for AoA computation. This approach not only enhances the accuracy of our system but also optimizes the computational load, making it an ideal choice for applications within resource-constrained environments.

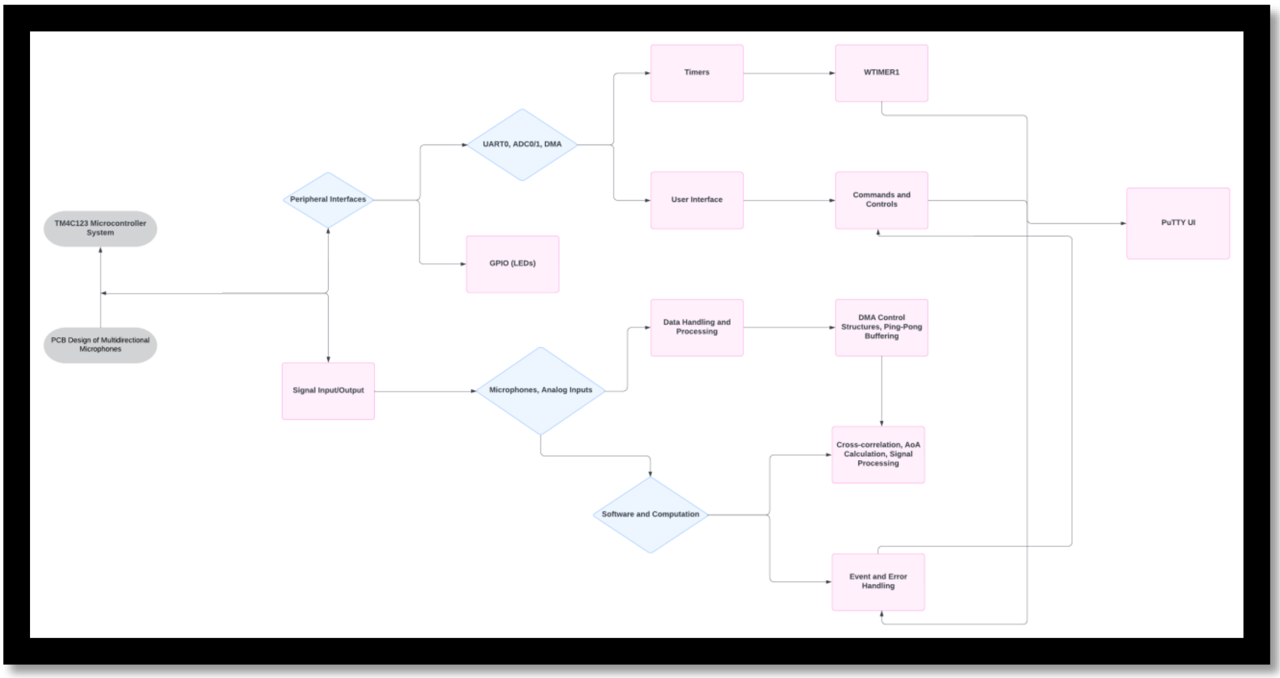

Fig.1 Project Flowchart

2. Project Objectives

The primary objectives include:

- Developing an embedded system capable of calculating AoA with high accuracy.
- Ensuring the system operates efficiently under low power and computational constraints.
- Creating a robust user interface for real-time data monitoring and system configuration.

3. Hardware Architecture

Microphones and Conditioning Circuits:

- Four precision microphones with a sensitivity of -44 dBV/Pa are used. Each microphone's output is amplified using a 40 dB gain circuit to ensure signal integrity over environmental noise.

Microcontroller:

- The ARM Cortex-M4F core of the TM4C123GH6PM is use for it robust handling of digital signal processing tasks and multiple I/O interfaces, facilitating real-time audio data acquisition and processing.

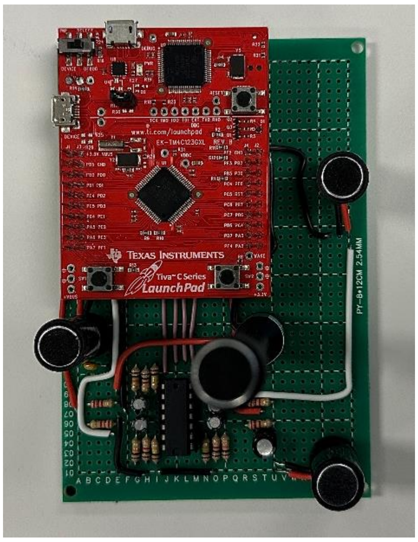

Fig.2 Hardware Architecture

4. Software Development

Audio Processing Requirements:
The software is designed to execute cross-correlation algorithms efficiently, enabling precise TDoA calculations necessary for accurate AoA determination.

Cross-correlation for TDoA Calculation:

- The implementation involves calculating the cross-correlation between signals received at different microphones to derive the TDoA, using optimized algorithms to handle computational limits of the hardware.

User Interface Commands and Control:

- A simple command-line interface over UART allows users to configure system parameters, calibrate the device, and view real-time processing results.

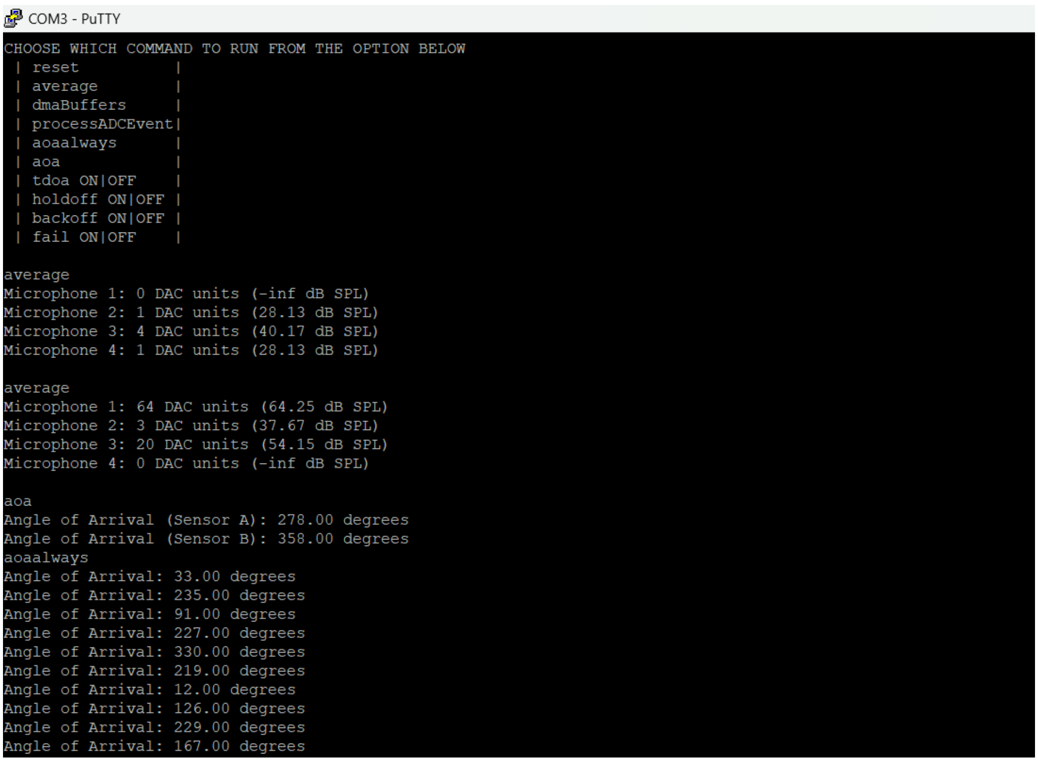

Fig.4 PuTTY

5. Approach for Design

Choice of Digital over Analog Comparators:

The use of digital comparators was selected over analog to achieve higher accuracy and repeatability essential for signal processing. Digital comparators provide the capability for precise threshold adjustments, which are critical for consistent detection of sound events across varying ambient conditions. These thresholds were determined through empirical testing, ensuring that the system not only reliably detects true sound events but also effectively minimizes the influence of background noise.

DMA Buffers

The DMA buffers, often called ping-pong buffers in this context, are used to facilitate efficient data transfer between the peripheral (in this case, the ADC) and the system memory without continuous CPU intervention. This setup is particularly useful in applications like audio signal processing where data must be continuously sampled and processed in real time.

Overview of Ping-Pong Buffering Technique:

- **Dual Buffers**: Ping-pong buffering employs two separate memory regions (buffers). While one buffer (ping) is actively receiving data from the ADC, the other buffer (pong) can be processed by the CPU. This alternation continues cyclically to ensure a seamless flow of data.
- **Purpose**: This method is crucial for maintaining data integrity and flow, especially in systems with high data throughput requirements. It prevents data loss that could occur if the CPU is busy processing and cannot handle incoming data in real time.

Description of the Displayed DMA Buffers:

- **Array Representation**: In the CCS IDE, these buffers are visualized as arrays of unsigned integers. Each entry in the array represents a sample of data collected from the ADC.
- **Sample Values**: The values seen in the buffers are ADC sample outputs, which have been directly stored in memory by the DMA controller. The indices of the array represent sequential sample points.
- **Memory Address**: The starting memory address of the buffer (**0x20002800**) indicates where in the system's memory the buffer is located. Each subsequent index in the buffer array is typically incremented by the size of the data type (in this case, four bytes for an unsigned integer).

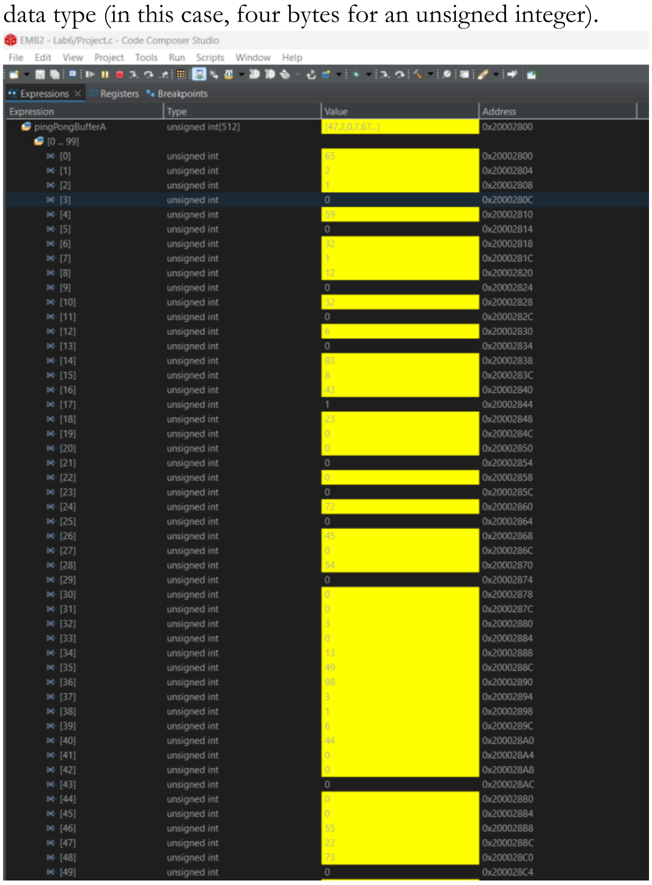

Fig.5 DMA Buffers

Detailed Microphone Calibration and Angle Adjustment:

- Hands-On Calibration Process: The calibration of microphones was performed manually to ensure precise angle detection. This involved physically adjusting each microphone's position to various angles and testing the response to sound from different directions. By playing a standardized sound source from multiple angles, I recorded the microphones' outputs to establish a baseline response at each position.

- Angle-Specific Adjustments: For each angle, adjustments were made to the microphone settings to optimize the pickup pattern and sensitivity. This meticulous calibration was critical in setting up the system to accurately interpret the sound's direction based on the differential response of the microphones. These adjustments were particularly important in environments with complex acoustic characteristics, where reflections and diffractions could cause anomalies in sound perception.

Design Challenges and Failings:

- Digital Comparator Setup: Setting up the digital comparator presented several challenges. Initially, the digital thresholds were not effectively discriminating between noise and actual sound events. This was mainly due to variability in microphone sensitivity and the environmental noise levels, which were more dynamic than anticipated. Several iterations were required to fine-tune these thresholds, involving empirical testing and adjustments based on observed outputs. Enhancing the robustness of this setup could involve integrating adaptive threshold algorithms that adjust in real-time based on ambient noise characteristics.

- Cross-Correlation Implementation: Implementing the cross-correlation for determining the time difference of arrival (TDoA) between microphones was technically demanding. The primary issue was the computational intensity of cross-correlation, which strained the processing capabilities of the TM4C123GH6PM microcontroller. The initial attempts led to significant system lag, impacting real-time processing requirements. To mitigate this, the cross-correlation computations were optimized by reducing the sample rate and simplifying the algorithm, which unfortunately compromised some measurement accuracy. Further improvements could include exploring hardware acceleration options or dedicated DSP chips that can handle more complex algorithms without sacrificing system responsiveness.

Potential Improvements:

- **DMA Buffer Management:** The management of DMA buffers under high-intensity sound events needs improvement. Implementing a more sophisticated buffer management strategy that includes dynamic allocation and real-time monitoring could prevent data loss and enhance system reliability.

- **Advanced ADCs**: Upgrading to advanced ADCs with higher sampling rates could significantly improve the accuracy and responsiveness of the system. This would allow for more precise and faster data processing, critical for real-time audio analysis.

Technical Insights:

- **Threshold Calibration:** Continuous recalibration of thresholds based on environmental data helped in achieving better stability in event detection. This approach, however, requires a balance between responsiveness and computational demand.

- **Algorithm Optimization:** Simplifying the cross-correlation calculations demonstrated a trade-off between system performance and accuracy. Exploring alternative algorithms or parallel processing techniques could provide a path forward for handling intensive computations without degrading system performance.

Conclusion

The project demonstrates a feasible approach to solving the AoA problem using digital signal processing techniques on a low-cost microcontroller platform. With further refinements, this system has the potential to be applied in various practical applications, such as automated surveillance systems or interactive sound installations.

References

- TM4C123GH6PM datasheet
- online resources