CSE – 5342

Embedded Systems - 2

Project

REPORT

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INTRODUCTION

The Angle of Arrival (AoA) problem is a challenging computational problem with various practical applications, including determining the direction of a gunshot or a speaker for law enforcement and robotics industries, respectively. However, solving this problem requires significant computational resources, which makes it difficult to implement in resource-constrained devices. In this project, our goal is to design a low-cost, low-power device capable of solving the AoA problem with limited computational capabilities. The device will have a command-line interface that enables users to configure the system, read its status, and retrieve angle information. By providing this data, the device can augment a hosted OS device, extending its functionality and enabling new applications in various industries. This project will enable us to explore the limits of computational performance in resource-constrained devices, creating new possibilities for solving complex problems in a cost-effective and energy-efficient manner.

OVERVIEW

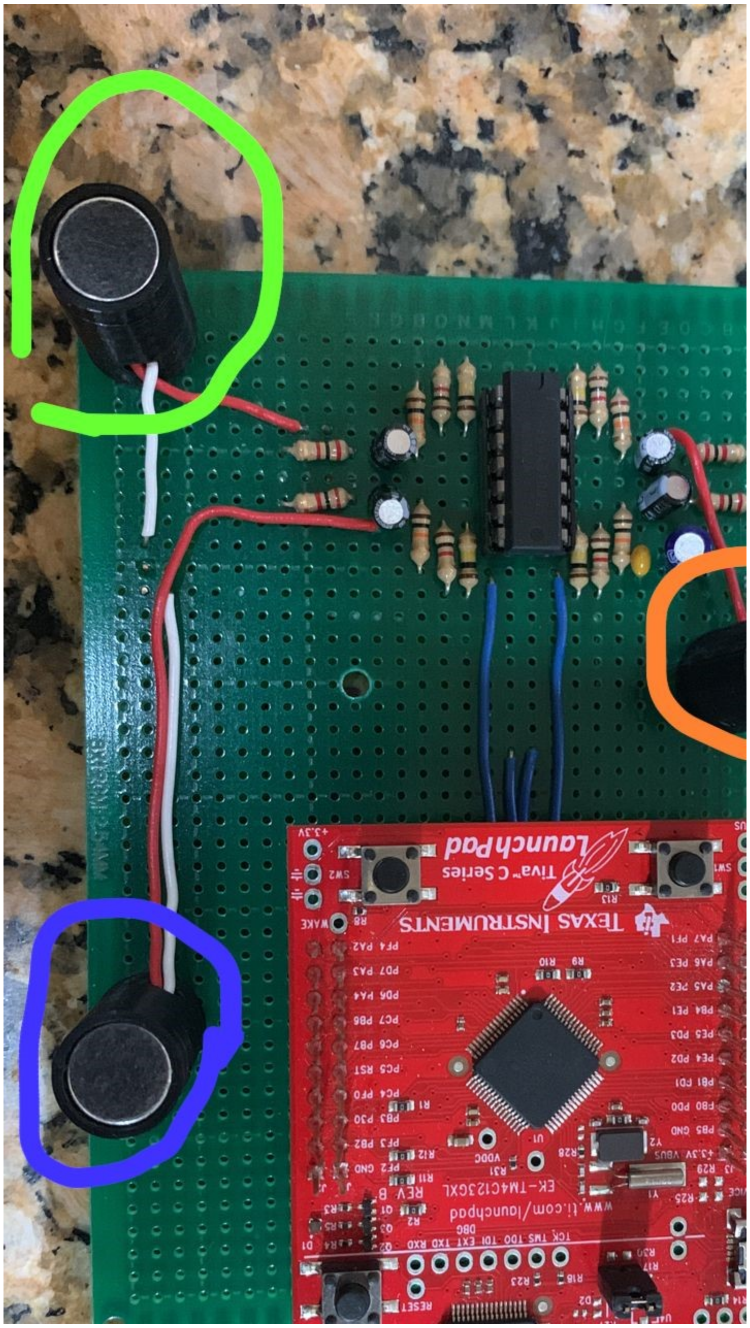
The purpose of this report is to detail the system used to determine the angle of arrival of a noise source in 2 dimensions using three microphones. The microphones are biased using a voltage to activate the internal FET device. The resulting output is then AC coupled to an active half-wave filter before passing through an amplifier circuit. The outputs of the three amplifiers are then connected to the three analog inputs of a microcontroller. This setup allows for the precise measurement of the angle of arrival of the noise source using data from all three microphones. The system's design aims to achieve accurate measurements while keeping costs and power consumption low. Overall, this report provides an overview of the design and implementation of the system for determining the angle of arrival of a noise source, highlighting its components and how they work together to achieve accurate and cost-effective measurements.

HARDWARE

Parts Lists:

Part	Quantity
TM4C123G evaluation board (ARM M4F)	1
LM2902 (quad op amp)	1
2.2kΩ, 5% resistor (mic bias)	3 or 4
1kΩ, 5% resistor (inverting input)	3 or 4
10kΩ, 5% resistor (shunt input)	3 or 4
100kΩ, 5% resistor (amplifiers)	3 or 4
0.1µF capacitor (supply bypassing)	2 or more
1µF capacitor (microphone AC coupler)	3 or 4
10µF capacitor (supply)	1
14pin 300mil socket (op amp)	1
2x10 double-row header, unshrouded	2
CMC-9745-44P microphone	3 or 4
Microphone holder (short)	3
Microphone holder (tall)	1 optional
#4 x ¼" sheet metal screw	3 or 4
80x120cm FR4 PC board	1

Whole Hardware Setup:



MIC 1 => Orange circled

MIC 2 => Green Circled

MIC 3=> Blue circled

User Interface Commands

average: If “average” is received, the average value (in DAC units and SPL) of each microphone will be displayed.

Reset: if “reset” is received, the hardware will get reset.

Backoff: If “backoff B” is received, the backoff between the first and the subsequent microphone signal threshold levels can be set. If the argument B is missing, return the current setting

Holdoff: If “holdoff H” is received, set the minimum time before the next event can be detected. If the argument H is missing, return the current setting.

Angle of Arrival: If “aoa” is received, return the most current angle of arrival (theta and optionally theta and phi).

AoA always: If “aoa always” is received, display the AoA information of each “event” as it is detected.

Time Difference of Arrival : If “tdoa ON|OFF” is received, enable or disable the display of the TDoA data for qualified events (all sensors see the signal within the possible time window) when AoA data is shown.

Fail Command: If “fail ON|OFF” is received, enable or disable the display of the partial data set of data from the sensors when you do not have a qualified event (the TDoA is greater than possible or is incomplete).

Theory of Operation

The objective of the project is to design an Angle of Arrival (AoA) system for sound localization using three microphones. The microphones are connected to the ADC0 of the microcontroller. However, the ADC0 runs at 1 MSPS while the human voice frequencies are much smaller, so the ADC0 is used only for sampling all the three microphones.

Sample Sequencer 0 of ADC0 is used and it is configured to generate an interrupt for every 6 samples. The samples in the FIFO are in the format: FIFO SAMPLES = {MIC3, MIC2, MIC1, MIC3, MIC2, MIC1}. As the ADC runs at 1 MSPS, the time taken to fill the FIFO is 6 microseconds. Therefore, the ADC ISR will trigger for every 6 microseconds.

To provide more time to perform the AoA math, down-sampling is performed, which is user-configurable. In this case, down-sampling by 10 is used, which means that samples are collected only once for every 10 ISR hits. A time counter runs in the ISR to calculate the Time Difference of Arrival (TDOA).

When any one of the three microphones receives the sound, the time counter will be set to 0, and the AoA_calc() function is called until the other two microphones receive the sound or until the timeout. During the second peak and the third peak, the time counter value will be stored in two separate variables, which will be used to calculate the angle.

In addition, there is a filter in between the ADC samples and the peak detection logic. Specifically, a Finite Impulse Response (FIR) filter is used to filter the input data. The FIR filter is used to remove noise and improve the signal-to-noise ratio. The filter takes 16 input samples and produces a single output sample. The FIR filter is implemented using a circular buffer and a pointer to the next input sample.

Overall, the system provides an efficient way to calculate the Angle of Arrival of sound using three microphones and a microcontroller.

Calculations

```
if (plx_flag == EVENT_MIC_1)
{
    angle = 60 * (int) ((t3x - t2x)) * K;

    if (tdoa_flag == 1 || aoa_always_flag == 1)
    {
        sprintf(str, "t1x:           %4i\n", t1
        putsUart0(str);
    }
}
else if (plx_flag == EVENT_MIC_2)
{
    angle = 60 * (int) (t1x - t3x) * K;

    if (tdoa_flag == 1 || aoa_always_flag == 1)
    {
        sprintf(str, "t2x:           %4i\n", t2
        putsUart0(str);
    }
}
else if (plx_flag == EVENT_MIC_3)
{
    angle = 60 * (int) (t2x - t1x) * K;
```

The angle of arrival is presented relative to the position of the microphones. The circle of 360 degrees is divided into three 120-degree arcs, each of which centers on one of the microphones. The right side of the microphone contributes to positive 60 degrees, while the left side contributes to negative 60 degrees.

The mathematical equations are based on the microphone that received the first peak. If microphone 1 received the first peak, the angle is calculated **as 60 * (t3-t2)*K**, where K is a constant set to 0.022, and t1, t2, and t3 represent the time counter values when the peak was detected on microphone 1, microphone 2, and microphone 3, respectively. If microphone 2 received the first peak, the angle is calculated as **60 * (t1-t3)*K**. If microphone 3 received the first peak, the angle is calculated as **60 * (t2-t1)*K.**