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Acoustic Source Localization Based on Time Difference of Arrivals using Generalize Cross-Correlation Method

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ABSTRACT

This paper describes how to detect the location of a sound source using a microphone array by determining the Time Delay of Arrivals (TDOA), which is a powerful technique to obtain the position of the sound source depending on the position of the microphone array. The mathematical model represents the sound propagation, which affects the sensors. To estimate the location of the sound, a digital signal processor-educational kit DSK6713-was used to collect the sound signals from the microphones through four channels audio Daughter Card, the Code composer studio software is used for the configuration of the DSK6713 and Daughter Card, and MATLAB is used for analyzing the previously collected sound signals. The estimated position was coming properly and the error minimized to less than two centimetres.

Keywords:- **Microcontroller DSK6713, Audio Daughter Card, Code Composer Studio.**

I. INTRODUCTION

Sound source localization is a system used for distinguishing and detecting the sound source position in space. This device is used for multiple applications, such as an intelligent robot, security and monitoring system [1].

The time difference of arrivals is one of the methods used for detecting the sound source. There are several methods that can be used to determine the delay of time between the two signals such as, spiking neural network, pulse neural network, Multichannel acoustic localization and Generalized cross-correlation(GCC) [2-5**]**.

Correspondingly, to get a correct result from the sensors (microphones), the position of the microphones should always be fixed. In general, the microphone array is arranged according to a regular structure such as linear, rectangular, circular, and spherical [6].

The sensors positions are constant and having different time of arrival, the mathematical algorithm can be used to estimate sound source position [7].

II. METHODOLOGY:

The generalized cross-correlation [GCC] method is used to find the time delay between two signals, by sampling procedures convert the sound signal into a sequence [9]**.** The Microphone array consists of four sound sensors; the output of each microphone is an input for a DSK kit. The inputs are stored in the memory as a data [buffer]. The real-time algorithm uses a simple filter, the input data should go through the filter [8]**.**

Furthermore, convolving the predetermined filter with sensor sequence for each microphone sensor, the output data from the filter is compared with a predetermined threshold by selecting the maximum value from the data, if the value reaches the threshold that means the system has detected the desired sound. Then the detection process will stop and the delay of time arrival will be calculated using (GCC) method for any two microphone waveforms, so as to find the delay between every two sensors, the convolution is done in this way:

$$(x * h)[n] = \sum_{m=1-L}^{n} x(n-m)h(n) \quad ----- (1)$$

The algorithm can be divided into two main stages, the first stage defines the time difference, and the second stage uses the time difference and the microphones locations to estimate the Acoustic Source Localization (ASL). Figure 1 describes the flow chart of the operation.

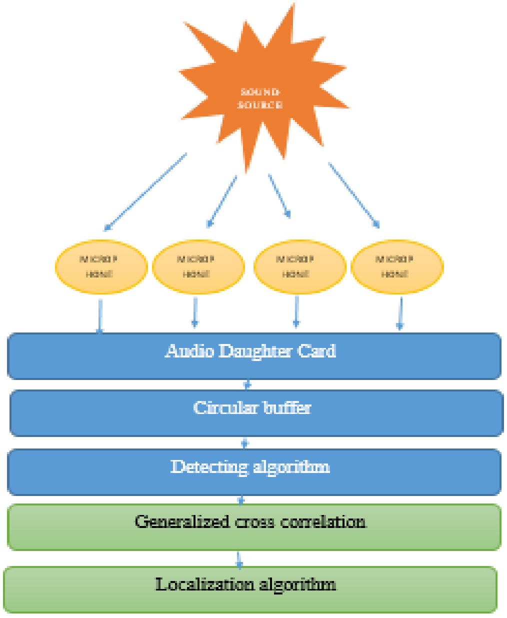

Figure 1. Declare Stages of Operation

Generalized cross-correlation (GCC) algorithm is used to compute the delay of time arrival (dt) between a pair of sound signals. The delay time of arrival system is used to solve the equation: Velocity*time =distance

$$v * dt = dL - - - - - (2)$$

The distance can be represented as a magnitude of vectors, as like below:

$$f(x,y,z,t) = \sqrt{(x-x_1)^2+(y-y_n)^2+(z-z_n)^2} - C(t-t_n) = 0 - - - - - (3)$$

III. **REAL-TIME IMPLEMENTATION RESULTS:**

The final configuration for this system is developed as the following:
Install four microphones at coordinate axes:

$$Mic\_pisL1=(0,28,10), Mic\_pisR1=(28,0,5), Mic\_pisR0=(0,-28,10), Mic\_pisL0=(-28,0,5)$$

In addition, to connecting each microphone with the audio Daughter Card in DSK6713 Kit, and save its signals as data in the circular buffers, the MATLAB uses the sound data to apply the algorithm and find the estimated position.

HARDWARE TEST: For the hardware implementation four microphone sensors were used, installed at coordinate axis mic1 in the point (0.00, 0.28, 0.10) , mic2 at (0.28, 0.00, 0.05), mic3 at (0.00, - 0.28, 0.10) and finally mic4 at point (- 0.28, 0.00, 0.05), the sound source was put at an arbitrary point inside the coordinate axis as in the figure below:

Figure 2. Microphones Installation Sound Source

The microphones were connected with DSP processor through the Audio Daughter Card with two codecs, left and right channels, used to collect the audio signals and save the data into a circular buffer in the memory.

Figure 3. Microphones with DSK 6713

MATLAB results: The estimated position was coming properly and the error was minimized to less than two centimeters, the figure below shows the difference between the actual position and the estimated position.

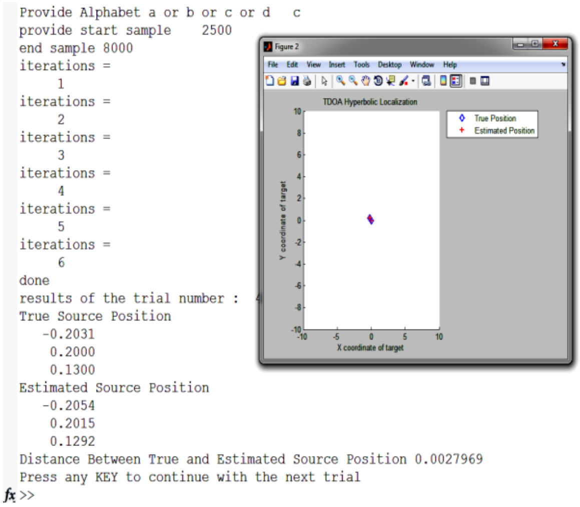

Figure 4. Estimation Position Compere Actual

The sound data collected from four microphones can be represented as shown.

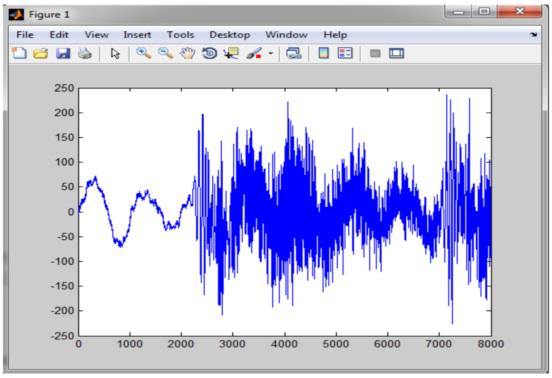

Figure 5: The Sound Data

Collected from four microphones.

IV. Conclusion

The time delay of arrival technique is a powerful technique for acoustic source localization and it gets good results in estimating the source location. So overall, the project was successful because it was possible to be built and tested. Moreover, the project is low cost, small and has the potentials to be a very good Acoustic Source Localization (ASL) that could be implemented in the real-life scenarios. The implementation and testing of the ASL are not far from being completed because there is a number of aspects of the system which still needs implementation, however, the project is very near to be completed with the software nearing full functionality and the hardware is ready to test. Improving localization accuracy has become clear and the focus on this accuracy for the future works. It is preferred to recommend two methods of doing this, improving the system accuracy by increasing the number of microphones; and numerous areas of the system can be improved. The detection process certainly deserves more work and filtering techniques, and then the system will be more sophisticated**.** label, present them in parentheses. Do not label axes only with units.

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