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Estimating the Angle of Arrival of the Sound Source using Microphone-Array Beamforming

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Abstract

In a speech recognition device, to track and locate the sound source, the device needs calculate the direction and angle of arrival of sound. This is done with the use of a microphone-array. A microphone-array is a MIMO device that can track, locate the sound source and can also help to identify the type of sound source from the surrounding environment. Different array types can be used depending on the equipment available. Various types of array geometry exist for different applications. In this project, we are designing a Linear Microphone-Array using the concept of Delay-and-Sum Beamforming. Beamforming is a signal processing technique used in audio conferencing systems to increase the Signal-to-Noise Ratio (SNR) of the source sound signal. After using the concept of beamforming, we will be able to estimate the angle of arrival of sound. In this project, we are estimating the angle of arrival using MATLAB and proving the concept of DAS beamforming using LabVIEW.

Keywords: *Home-Automation, Microphone-Array, Beamforming, Sound-Source localization.*

I. INTRODUCTION

A microphone array is any number of microphones operating in tandem. It is one of the advanced devices in improving speech quality. A single microphone can only provide so much directivity and thus so much reduction in noise and reverberation without a post-processing solution. A microphone array effectively does quality enhancement implicitly by focusing a receiving radiation pattern in the direction of a desired signal, thereby reducing interference and improving the quality of the captured sound.A technique called Delay-and-Sum beam-forming is used to estimate the angle of arrival of sound. This an old technique which was implemented during WW2 to track stealth planes.

Array processing involves the use of multiple sensors to receive or transmit a signal carried by propagatingwaves. Sensor arrays have application in a diversity of fields, such as sonar, radar, seismology, radioastronomy and tomography. While the use of sensor arrays for speech processingis a relatively new area of research, the fundamental theory is well established as it is common to allsensor arrays, being based on the theory of wave propagation.

Distant Speech Recognition (DSR) occurs when speech is acquired with one or many microphone(s) moved away from the mouth of the speaker, making recognition difficult because of background noise, overlapping speech from other speakers, and reverberation. DSR is necessary for enabling verbal interactions without the necessity of using intrusive body- or head mounted devices. But still, recognizing distant speech robustly remains a challenge. Microphone arrays make it possible to capture sounds for DSR in human–robot interaction (HRI). This requires the installation of multiple microphones on the robot platform, and process distant speech perceived by filtering out noise from fans and actuators on the robot and non-stationary background sound sources in reverberant environments, fast enough to support live interactions. This process usually relies first on localizing and tracking the perceived sound sources, to then be able to separate them for specific processing such as speech recognition. Using sound source localization and tracking methods robust to noise and low computational cost is important, as it is usually the first step to engage speech-based human–robot interaction (HRI)

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II. Literature survey

[1] Jacob Benesty, JingdongChen, Yiteng Huang and Jacek Dmochowski in their paper "On Microphone-Array Beamforming From aMIMO Acoustic Signal Processing Perspective" , discusses about different types of Microphone-Arrays, the signal-enhancement performance of the Microphone-Arrays, and develops a general beamforming framework. It also explains the basics of beamforming and its working.

[2] François Grondin and François Michaud in their paper "Lightweight and Optimized Sound Source Localization and TrackingMethods for Open and Closed Microphone Array Configurations", discusses about the concepts of sound source localization and its various techniques.

[3] The presentation by Ulf Michel at the 2006 Berlin Beamforming Conference discusses about the history of acoustic beamforming. The survey paper talks about the applications of microphone array used during World War II

[4] The referred PhD thesis by Ian McCowan on "Robust Speech Recognition using Microphone Arrays", discusses key concepts of fundamental array processing and beamforming theory relevant to microphone array speech processing. The thesis was referred to understand the basic fundamentals of the signal acquisition, filtering and processing. The drawback is that it doesn't has information on designing the array. The author uses it custom designed software to implement manufacturing of the array on a large scale.

[5] The book referred,"Array SignalProcessing" by Unnikrishnan Pillai discusses about the core mathematical equations required to implement the arrays and sound source calculations. The book was referred test various array geometries on paper using different kind of algorithms. The book requires prerequisites of computational as well as number theory to design the algorithms and at times can be difficult.

[6] The book referred, "Study and Design of Differential Microphone Arrays" byJingdong Chen, Jacob Benesty, focuses heavily on array designing fundamentals and spacing the array correctly in order to maximise the sound tracking efficiency. This book was referred to calculate the spacing between the array. Also the book used a custom c library called as Open embedded Audition System(ODAS) to run low-cost hardware. The drawback is that the hardware has to be run on specific DSPs and these DSPs are not available in the Indian Market.

[7] The referred paper , "Flexible and Optimal Design of Spherical Microphone Arrays for Beamforming" by Zhiyun Li and Ramani Duraiswamidiscusses about designing flexible array geometries that can be placed in remote and difficult conditions without reducing the performance of the array. The paper was referred to understand more about array design and its calculations.

[8] The book referred,"Fundamentals of Differential Beamforming" by Jacob Benesty, discusses about techniques in Adaptive beamforming. Adaptive beamforming, is a better way to implement instead of using the traditional beamforming methods. This paper was used to implement the LCMV-"Linearly Constrained Minimum Variance" algorithm, but the algorithm failed severely because it was very resource heavy.

[9] The book referred "Microphone Arrays" by Michael Brandstein, talks about the development and design of microphone arrays for specific applications like theatre, near field source communication inside astronaut helmets, inside vehicle infotainment system. This paper was used to select application for the purpose of showing the output in a feasible way.

[10] The book referred, "Fundamentals of Spherical Array Processing" by Boaz Rafaely, discusses about design and development of 3D arrays, spiral and helical arrays. The 3D geometry is vey advanced form of array geometry and its implementation is generally done in the frequency domain.

Lastly multiple websites[11],[12],[13],[14] were referred throughout the design and implementation phase to solve and remove the bugs and problems.

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III. Methodology

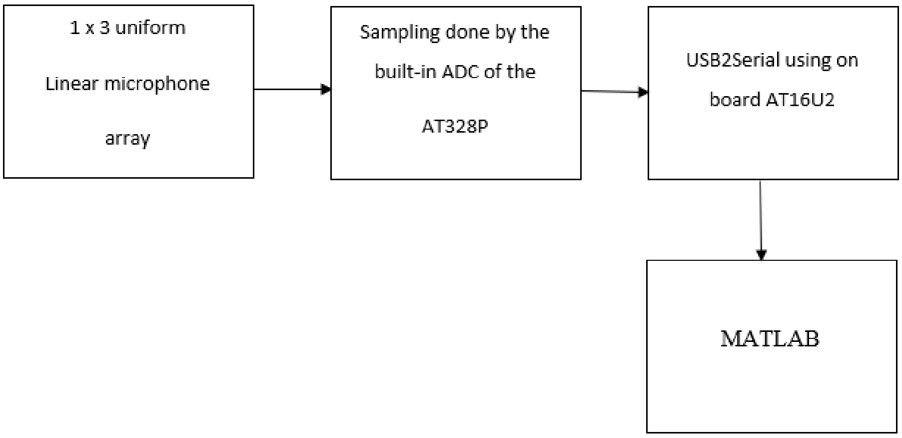

Fig1 : Block Diagram

The Fig 1 is the block diagram of the project. The signal will be sampled by a 3 microphones whicj are spaced at a distanceof 7~7.5 cms. The microphones have a built-in preset to remove the noise from the signal.
The signals are then sent to MATLAB at a maximum possible baud rate of 1152000. MATLAB decides if the data is valid
or not by checking the data at COM port 3 times. If the data doesn't make sense, the data will be discarded, and a fresh set
of data will be updated over the existing data.
While designing the microphone array it is necessary to maintain a trade off between the time taken to process a single instance of data versus the sampling frequency versus the tracked signal. The processing time has to be kept as minimum as possible. The processing function was clocked at an average of 532.637ms using the tic-toc built-in function in MATLAB. That means one iteration on an average took 532.67ms to process. This is the time taken by the signal to enter the channel when it first appears on the COM port up till it is displayed on the polar plot generated using MATLAB. What can happen during this is that by the time one iteration is completed, the sound source can change its location with respect to the array. So in some cases like these, the result can be difficult to interpret. So all of these tradeoffs have to be considered during the designs and implementation stage
Initially, a basic uniform linear array is designed to test the working and the feasibility of the project.The array is tested in different conditions to see the response to a sound source. Multiple designs of the array are tested at this point using different equations.A linear array showed the best possible result. It is easier to experiment with this kind of an array because the mathematical equations doesn't require changes even when the array geometry is changed.The array works best and most efficient on the principle of Delay-and-Sum beamforming.
Next after selecting this algorithm, the task is to select the correct number of microphones to acquire the signal. Initially 8 microphones were selected, but the sampling was not correct so the mics were reduced to 5. These mics showed distortions in the signal and the time required to process the signal was high using a 4 or 5 element array. That is why only 3element array is used.
Considering λ as the wavelength of far field sound source, Φ as the phase shift between consecutive microphones and L as the spacing between consecutive microphones, we can represent the angle of arrival Θ as follows: -

$$\Theta=\sin^{-1}((\lambda *\Phi)/ (2 *\pi*L)) \text{ ………radians} \qquad \text{……….(1)}$$

By calculating the phase difference in between the microphones, the angle of arrival is calculated.The phase difference is plotted on a polar plot.Now, with this information of phase angle and gain, the position of the sound source with respect to the array is estimate

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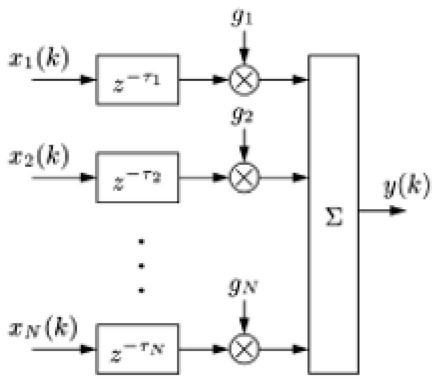

Fig 2 :Structure of a delay-and-sum beamformer

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IV. Results

The result of the microphone array is interpreted using a polar plot which plots the directivity versus gain.

Fig. 3 is the ideal result for 1 by 3 array.

The result shows that the sound will be perfectly recorded by the array at 0^0, 90^0, 180^0,270^0 which surrounds the whole environment.

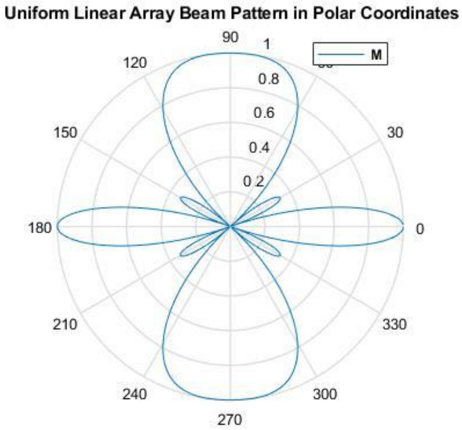

Fig 3: Ideal beam output for a linear array

Fig 4 is the result taken by recording a 1 kHz signal.

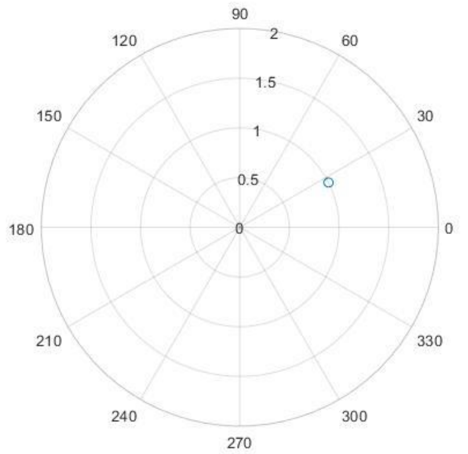

Fig 4: Actual output as displayed on MATLAB

The dot is the direction of the signal. It shows the direction from which the signal is coming from.
To verify if the beamforming is working around or not, we implemented the algorithm on a software called LabVIEW from National Instruments.
A GUI panel was designed to see the results of the array, and, the array was tested by directing sound on it from different dimensions of the array.

Following results confirmed that beamforming worked correctly.

Case 1: Testing Results for the mic placed at middle position of the array
Type of Sound Source: - Far field
Angle of Incidence: 0 degrees
Source Frequency 400Hz

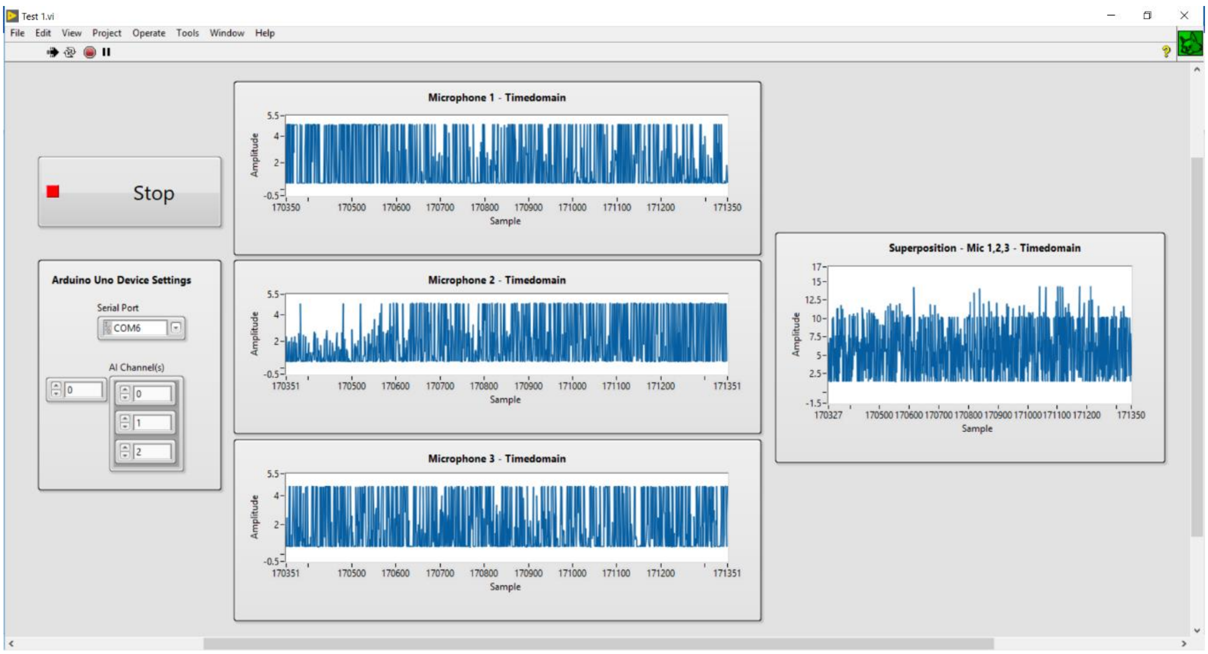

Fig5 : Mic placed at middle position of the array

Case 2: Testing results for mic placed at the left end of the array

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Type of Sound Source: - Far field source
Angle of Incidence: 90 degrees
Source Frequency 400Hz

Fig 6 : Second Mic

Case 3: Testing the third Mic
Type ofSound Source: - Far field
Angle of Incidence: 270 degrees
Source Frequency 400Hz

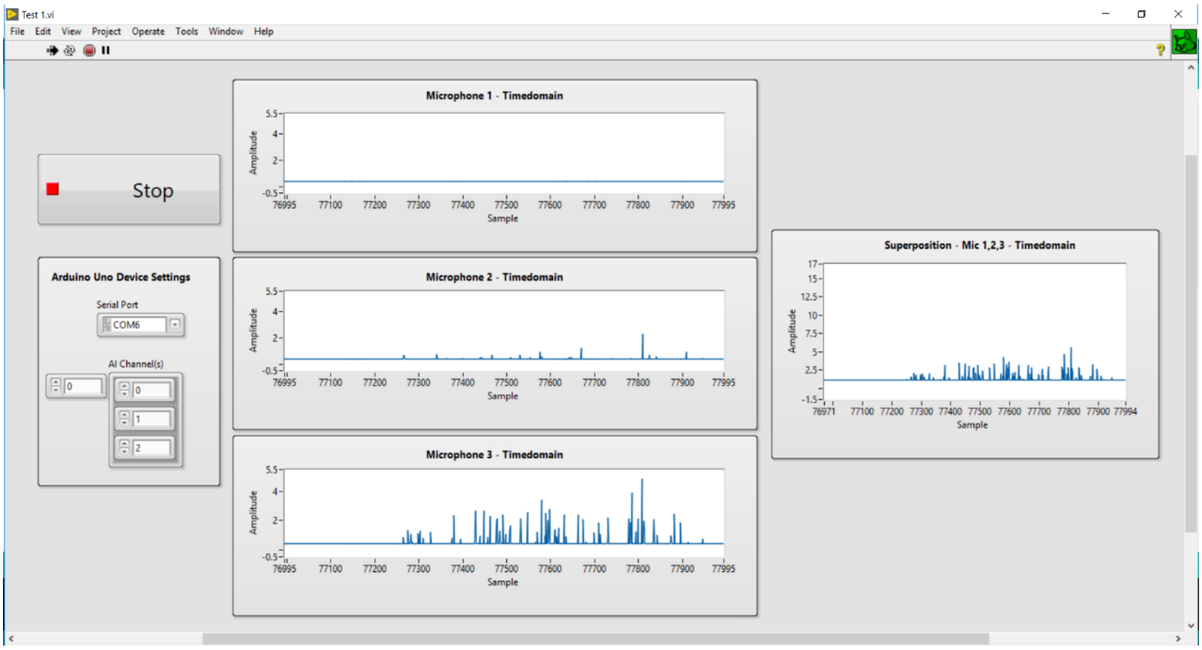

Fig 7 : Third Mic

V. CONCLUSION

Microphone-Array is an emerging field in the hands-free technology world. But there are so many important parameters to be considered while designing the array. The array can only work for a near-field source or a far field source. But having the location of the sound source is very important to automate or design systems efficiently.

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