AIN SHAMS UNIVERSITY
FACULTY OF ENGINEERING
Mechanical Design and Production Department

360 Degree Sound Source Localization in Free Field

A Thesis submitted in partial fulfillment of the requirements of the

Master of Science in Mechanical Engineering

By

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B.Sc., Mechanical Engineering, Design and Production Section
Ain Shams University, 2013

Supervised by

Prof. Dr. Tamer Elnady

Prof. Dr. Adel Elsabbagh

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Date: 31/7/2018

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STATEMENT

This thesis is submitted as a partial fulfillment of M.Sc. degree in Mechanical engineering, Faculty of Engineering, Ain Shams University.

The author carried out the work included in this thesis and no part of it has been submitted for a degree or qualification at any other scientific entity.

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Abstract

Sound can be used to determine the location of sound sources. This can be achieved using microphones which can pick up sound signals and by processing these signals we know the position of the sound source in space.

Time Difference Of Arrival (TDOA) technique was used to determine the location of a broad band sound source. It was critical to make the calculations as fast as possible to make a real time system.

The environment at which the source and the microphones are located is assumed to be free field.

LabVIEW was used to create a Virtual Instrument (VI) that could run heedlessly on a RIO system which would acquire the data through Field Programmable Gate Array (FPGA), send the data to real time processor, make the necessary calculations, and take an action to point at the location of the sound source and show the location on a small LCD screen.

Keywords:

Field programmable gate array (FPGA), Signal processing, Sound source localization, Labview, Comapct DAQ (cDAQ), Compact RIO (cRIO), Cross correlation, microphone array, Data acquisition, Free field sound source, stand-alone system, trilateration.

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Contents

Chapter 1: Introduction

Humans are constantly seeking ways to copy nature; copying the senses of humans such as vision or hearing are one of the most sought-after goals. Human hearing is very complicated such that using our two ears we not only detect the location of a sound source, but also, we can extract sounds coming from a certain source in noisy conditions (such as focusing on one speaker in a room with multiple speakers), we can even recognize the origin of different sources such as a phone ringing or different voices of people. That means that human hearing system can localize, extract and recognize different sound sources. In this thesis the work will be focused on just one aspect which is localization.

Accurate determination of different sound sources in a free field could prove beneficial in many situations such as determining the source of sound in a desert where there are no buildings, only open spaces of land. This has many uses in surveillance and monitoring; such that a camera could be directed at the sound source and film it. The range of such an array could detect a sound source from a large distance, provided that the sound signal is loud enough.

This project will focus on locating a sound source in free field using a microphone array arranged such that they can detect a sound source in plane in 360 degrees around the array.

Two previous projects were completed at Ain Shams university:

1. The first had an acoustic array of microphones that listens to a machine and record what it listens to, post processing is then made to determine the location of different abnormal noise sources and identify their cause such as a bad gear or a faulty bearing [1].
2. The second, used 3 microphones installed on a moving car that can track a sound source moving in line, the car moved automatically to follow the moving source [2].

Both projects used frequency analysis to identify the sound source; in the first project the recorded microphone readings were analyzed in the frequency domain to find the peak frequencies and by linking each frequency to the different elements, the faulty element could be determined.

The second project had a buzzer as the moving source which emits a single frequency, such that all the calculations were made in the frequency domain.

1.1. Sound Source Localization techniques

Sound source localization is done using many techniques that can be divided into two main categories [3].

- Time domain techniques
- Frequency domain techniques

There are also some research papers on how to combine both time and frequency domain techniques in order to achieve better performance [4].

1.1.1. Time Domain Techniques

Time domain techniques work on the acquired signal in the time domain. Calculations can be divided into two steps: 1) Estimating the Time Difference Of Arrival (TDOA). 2) Calculations based on this estimation to pinpoint the source location.[3] The estimation of TDOA is based on using the Cross-Correlation function to determine the shift between signals at different microphones. Microphones are usually spatially organized in an array with known locations (such as circle, a line or a sphere). The accuracy of determining the microphone locations is very important as it is considered a factor in the calculations to determine the source location.

The calculations for localization are well established and are based on simple geometry, the main problem occurs with the estimation of the TDOA which depends on many different factors such as the signal to noise ratio (SNR) and reverberation [3]. Cross correlation is the method used to determine the TDOA, but due to its nature it can only calculate the delay in integer number of samples. This means that the delay is always a multiple of the inverse of the sampling frequency. Increasing the sampling frequency would increase the accuracy of the estimation but this would come on the cost of computation complexity and eventually making the application not suitable for real time applications due to the long computation time. A number of papers focused on improving the TDOA estimation such as in [5], where T. May etal. used different interpolation schemes in order to minimize the errors occurring due to cross-correlation function to determine the delay in a fraction of a sample instead of integer samples. Another factor that affects TDOA estimation is the SNR, such that the noisy background affects the accuracy of the detection. In [6] L. Chen etal. used different weighting techniques in order to decrease the effect of noise. This proposed weighting technique also suppresses the echoes which makes it more difficult to differentiate between the actual sound source and the direction of its echoes.

Simultaneous acquisition of signals is of great importance when estimating TDOA because all the calculations are based in the time domain, so all microphones need to share 1 common clock if possible.

Another TDOA estimation technique is presented in [7] where each microphone is connected to a rising edge trigger, when an impulse sound is detected, the absolute time stamp for each microphone is recorded and sent to the processing unit which measures the time delay between each of the microphones and from this data, the position is calculated.

1.1.2. Factors affecting TDOA estimation

The main factors that affect the TDOA estimation are: [3,8]

1. Signal to Noise Ratio (SNR)
2. Reverberation and echoes
3. Calculation window (number of samples on which cross-correlation is calculated)
4. Sampling rate (determines the least time difference calculated)

1.1.2.1 Generalized cross correlation

Generalized cross correlation (GCC) is the most basic technique used in sound source localization, it uses the cross-correlation function to estimate TDOA