Robotic Microphone Sensing: Data Processing Architectures for Real-Time Acoustic Source Position Estimation

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Abstract

In a previous undergraduate research project on "Acoustic Source Location Using Cross-correlation Algorithms" we found that the performance of the 2D position estimation algorithms using two pairs of microphones depends on array variables such as the distances between the individual and pairs of microphones, and also the sampling frequency. Therefore, we propose to build a robotic microphone array with autonomous control of the array geometry and sampling rate for improving the localization performance of an acoustic source in 2D space. In particular, in this project we focus on developing data processing architectures for estimating in real-time the 2D locations of an acoustic source. We implemented our algorithms in LabVIEW combined with Matlab and developed a graphical user interface that allows for easy interaction with the experimental setup. Our system allows for tracking a fixed and moving wideband acoustic source.

Overview

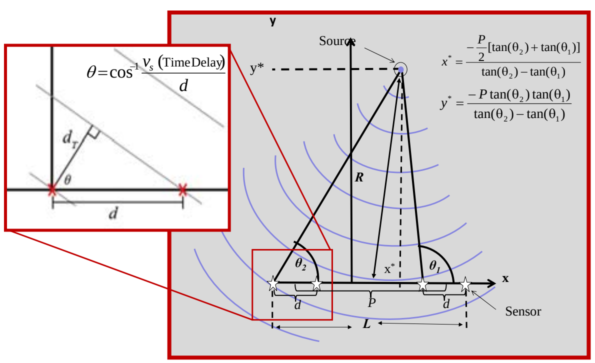
Goal:

• Design a system capable of acquiring measurements and estimating the acoustic source position in real time.

Approach:

• Use of data flow programming techniques for implementing signal processing architectures.

Background :



Variables

L is the length of the linear array

λ is the wavelength of the acoustic waveform

R is the radial distance from the source of the array

d if the distance between sensors pairs

Robotic Platform

Motivation

It can be shown that given a particular array geometry and sampling frequency, there are a finite number of possible locations which can be estimated using two pairs of microphones. This set of possible points is not uniformly distributed, as is shown in Figure 2.

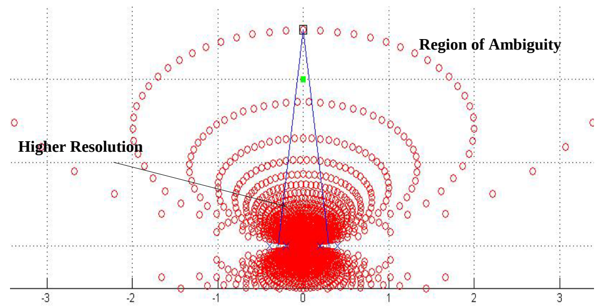


Figure 2

• Mounting the microphone array on robots allows us to change the physical parameters of the system in real time.

• Altering the distance between microphone pairs affects the spatial distribution of the possible estimation points.

• Shifting the array brings the source in-line with the array center.

• These movements increase the resolution near the source and improve the estimation.

Experimental Setup

Figure 5

Real-Time Data Processing Architecture

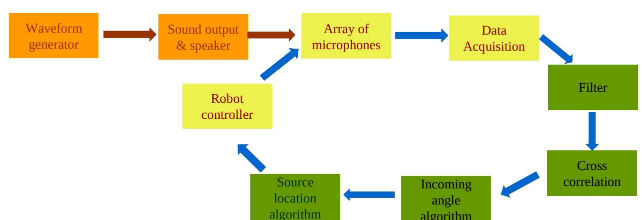


Figure 3

Data Flow Programming Implementation

The proposed system requires the synchronization of signal waveform generation, sound output functionality, data acquisition, signal processing, plotting of the source estimation, and constructive commands for robot movement, all occurring in parallel with one another. This system can be modeled as a constant flow of data, which can be considered to be a sequence of events, occurs as is shown in Figure 3.

In real time, however, these processes are occurring in parallel with respect to one another and can be more accurately represented by Figure 4. Given the parallel architecture that real-time processing demands, we opted to build our system in the dataflow programming environment, LabVIEW. Parts of these parallel functionalities are shown in Figure 6, in the top level abstraction of our LabVIEW code

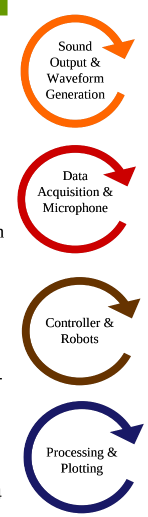


Figure 4

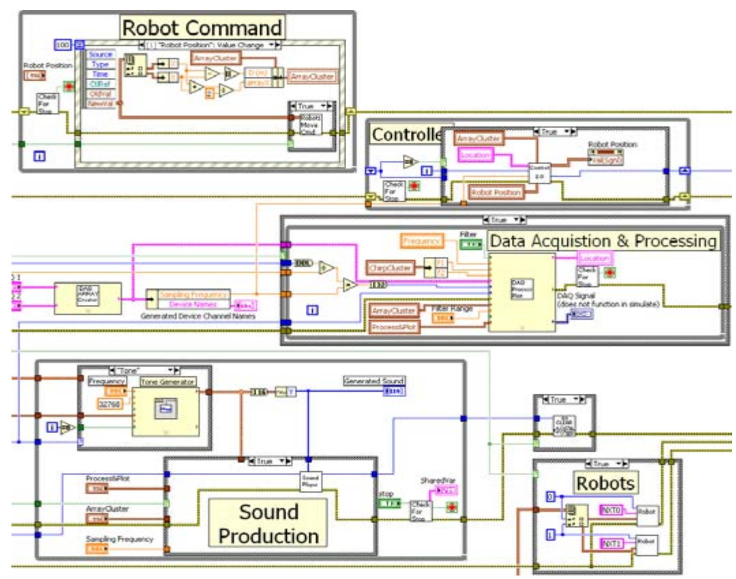


Figure 6

Acknowledgements

• Joshua York, "Acoustic Source Location Using Cross-correlation Algorithms", Fall 2008, http://ese.wustl.edu/~nehorai/josh/students.cec.wustl.edu/_jly1/

• Chase LaFont, "Robotic Microphone Sensing: Design of A Robotic Platform and Algorithms for Adaptive Control of Sensing Parameters", Fall 2009