Acoustic Localization Using Mobile Platforms

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Abstract

In this research, we develop and implement a Labview project to accurately locate a sound source using acoustic sensors mounted on mobile platforms. In the Labview project we created, a centralized program communicates with the two independent mobile platforms, navigating them to the source's location. On each platform, we implement a cross-correlation algorithm to obtain information about the source location. We then employ the location information from the two platforms on the centralized program. The program uses a triangulation algorithm we developed to find the source location, and then relays the proper location to the mobile platforms.

Overview

Goal

The goal of this project is to accurately locate a sound source using acoustic sensors mounted on mobile platforms.

Approach

Develop and implement a Labview protocol under which:

-Mobile platforms use cross-correlation algorithm to compute information regarding the sound source

-Centralized program obtains the location information from the mobile platforms

-Centralized program uses triangulation algorithm to determine its location

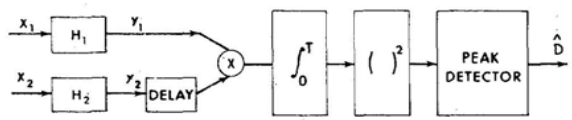
-Automated path decision routine allows robot to actively improve the sound source location estimate through multiple movement iterations

Measurement Model

$$x_1(t) = s_1(t) + n_1(t)$$
$$x_2(t) = \alpha\, s_1(t+D) + n_2(t)$$

- $s_1(t)$ is the incoming stationary random process
- $n_1(t)$, $n_2(t)$ are all real, zero-mean, jointly stationary random processes representing random noise
- $s_1(t)$ is assumed uncorrelated with $n_1(t)$ and $n_2(t)$
- α is the attenuation constant between the two microphones
- D is the delay between the two microphones

Generalized Cross-Correlation



The peak of $R^{(g)}_{y_1 y_2}(\tau) = \int_{-\infty}^{\infty} \psi_g(f)\, G_{x_1 x_2}(f)\, e^{j2\pi f\tau}\, df$, provides an estimate of the time delay, D, where:

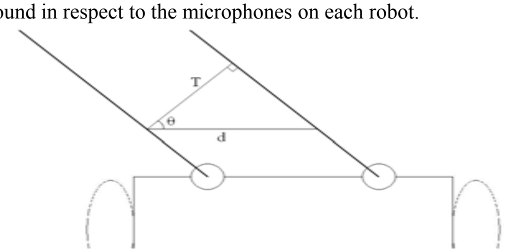
- $R_{x_1 x_2}(\tau) = E[x_1(t)\, x_2(t-\tau)]$ is the cross-correlation of $x_1(t)$, $x_2(t)$
- $G_{x_1 x_2}(f)$ is the Fourier transform of the cross-correlation such that $R_{x_1 x_2}(\tau) = \int_{-\infty}^{\infty} G_{x_1 x_2}(f)\, e^{j2\pi f\tau}\, df$ is true.
- $\psi_g(f) = H_1(f)\, H_2^*(f)$ is the general frequency weighting.

In general, H_1(f) and H_2(f) can be chosen to best suit the application. For example, to accentuate the signals at the frequencies in which the signal-to-noise ratio is highest. In our research, we used the simple cross correlation, such that: $H_1(f) = H_2(f) = 1, \forall f$

As a possible extension to our research, we will consider different choices for H_1(f) and H_2(f), such as the Roth processor or HT processor.

Angle of Sound Source for Each Mobile Platform

-Once each mobile platform calculates a time delay between its microphones, the centralized program calculates the angle of the sound in respect to the microphones on each robot.



T = Time Delay

d = Distance between the two microphones

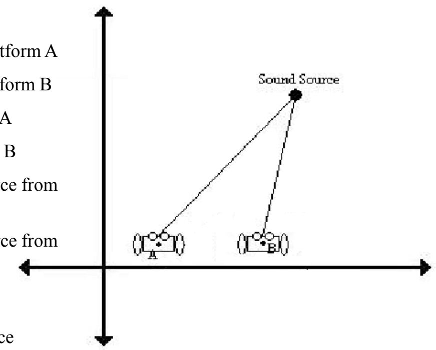
v = Speed of sound

θ = Angle of Sound Source with respect to the microphone pair

Estimation of Sound Source Location

-Once the angle of the sound source relative to each microphone array is calculated, the centralized program takes the position of each mobile platform and its rotation, and uses this information to create a line representing the direction of the sound source.

-The estimated location of the sound source is the intersection of these two lines.



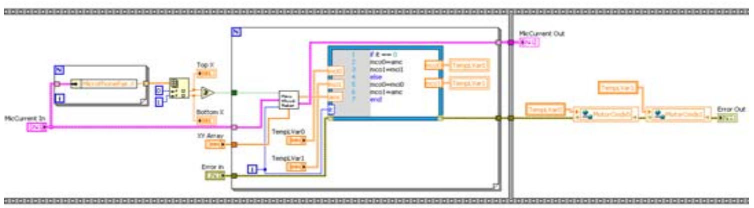
Inputs

- (x_A, y_A) = Coordinates of Mobile Platform A
- (x_B, y_B) = Coordinates of Mobile Platform B
- r_A = Initial Angle of Mobile Platform A
- r_B = Initial Angle of Mobile Platform B
- a_A = Calculated Angle of Sound Source from Mobile Platform A
- a_B = Calculated Angle of Sound Source from Mobile Platform B

Output

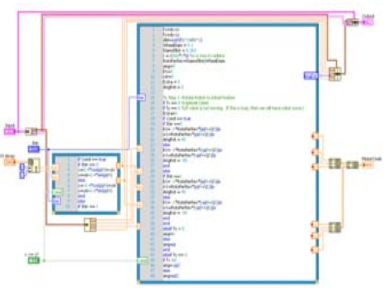
- (x_L, y_L) = Coordinates of Sound Source
- $x_L = (\tan(r_A+a_A)*x_A - y_A - \tan(r_B+a_B)*x_B + y_B)/(\tan(r_A+a_A) - \tan(r_B+a_B))$
- $y_L = \tan(r_A+a_A)*(x_L - x_A) + y_A$

Automation



Automation Goal

-The goal is to automatically move the platforms to improve the source position estimate.



Logic

-The automation algorithm has been designed to move the mobile platforms from their initial coordinates until a preset threshold is successfully reached.

References

- http://classes.engineering.wustl.edu/ese497/images/0/05/Presentation_Robotic_Microphone_Array.pdf
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