Sound Source Location Enhancement with Dynamic Microphone Array Elements

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Abstract

This project sought to evaluate the benefit of using a dynamic microphone array for sound source detection and location in contrast to a static array. The goal is to give robotic platforms superior sound source detection location abilities by shifting microphone positions to enhance performance. This hypothesis was inspired by human behavior of head twisting and turning to locate an active sound source. Monte Carlo simulations were run to determine the relationship between platform orientation and the peak power Steered Response Coherent Power (SRCP) source-localization algorithm. Variables of interest during simulation were number of microphones per platform, number of platforms, distance to source (equi-distance setup), and pitch angle of platform. A non-equidistance setup was also simulated. In these simulations, a maximum improvement in SNR of 15 dB was observed as the array orientation was rotated between end-fire and broadside orientations. From the simulation results a physical system was built to test/verify these results experimentally.

Methods

The Audio Array Toolbox[1] was used to simulate the SRCP algorithm with the dynamic microphone arrays. A Platform object was defined in MATLAB code to define the set of microphones, equally spaced around a circle of constant radius. The Platform object has methods for rotating and setting the orientation by Euler angles, as well as quaternions. To simulate a source, a chirp signal was generated that lasted 0.2361 seconds and contained frequencies equally in a band from 100 to 3000 Hz.

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Materials

To complete this research, the following hardware was used: Raspberry Pi 3 Model B, PCA9685 16ch PWM driver, Audio Injector Octo 6in,8out Soundcard, 2x Pan-Tilt dual servo platform, 6x Electret Microphone with MAX4866 IC, 3D printers in the Engineering Innovation Lab, JST-3 pin connectors and 24-AWG wire. The following software was used: MATLAB, Autodesk Inventor, CMAKE was used to compile code on RPi, C++ libraries rtaudio and PCA9685.

Hardware Design

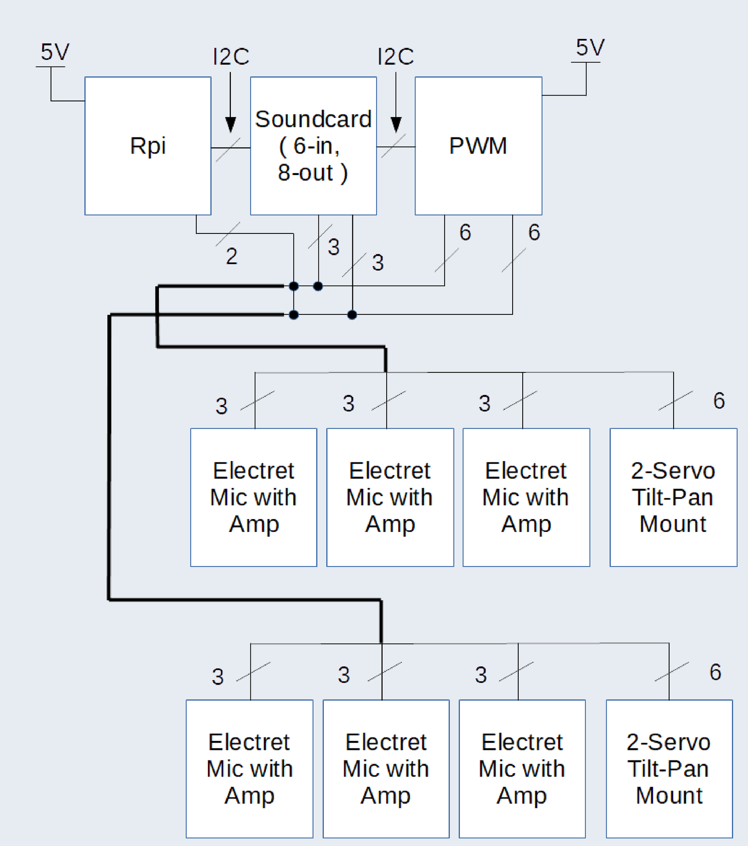

Figure 1: Hardware Design

CAD

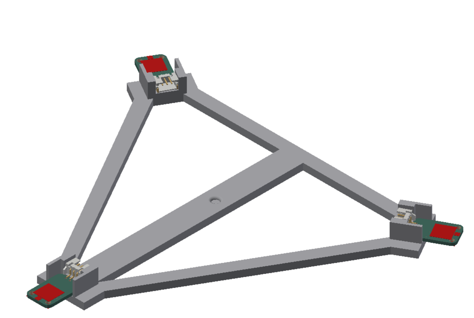

Figure 2: CAD assembly of 3-microphone array

The microphone to adjacent microphone separation used in the above apparatus and the simulations was 17.3205 cm (10 cm radius). This distance is the wavelength of a 2 kHz sound wave.

SNR Metric

To determine the performance of the array, an SNR metric was devised from the resulting planar image of the SRCP algorithm. For the numerator, an 16x16 grid (box) around the source location was used and the max SRCP value in that box was found. For the denominator, the same box of values was removed from the image, then the mean value was determined. This SNR is stated in the equation below.

$$SNR = \frac{\text{max(Box around Source Values)}}{\text{mean(SRCP image without Source Box)}}$$

$$SNRdB = 20 \cdot log_{10}(SNR)$$

SRCP Images

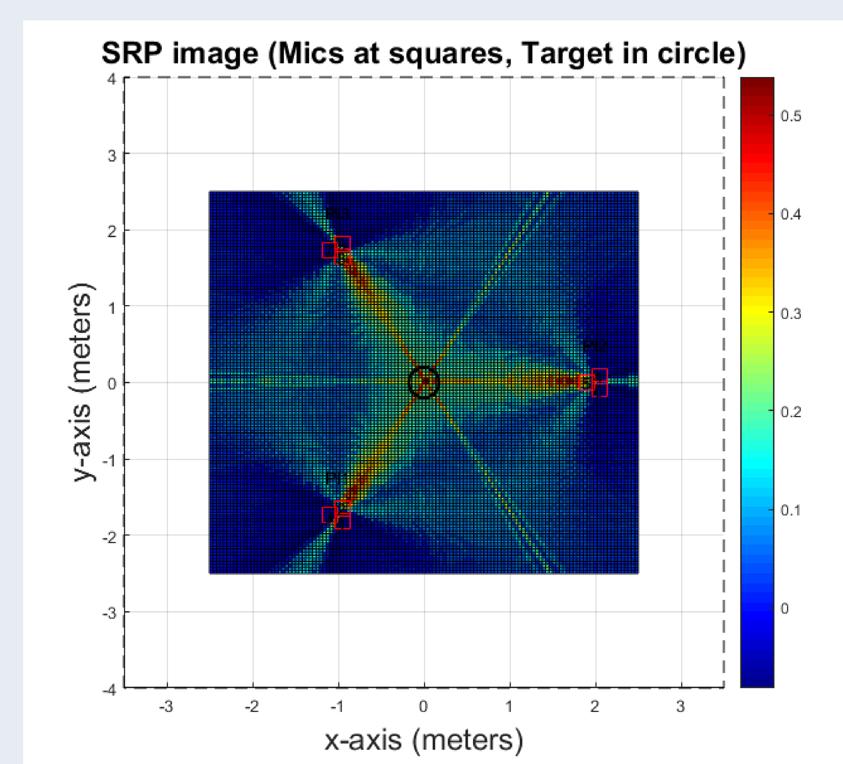

Figure 3: SRCP Image

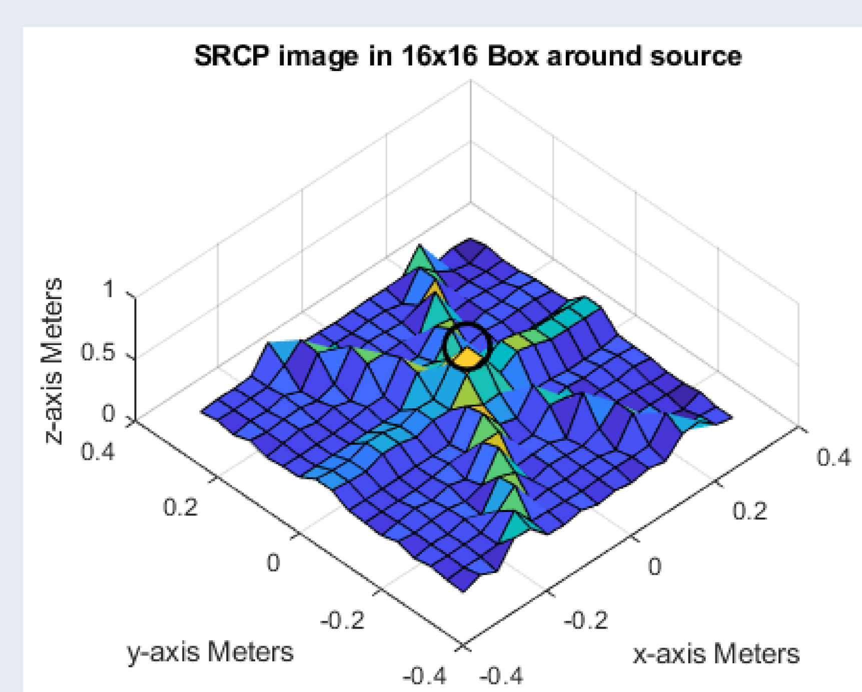

Figure 4: Box around Source

Results

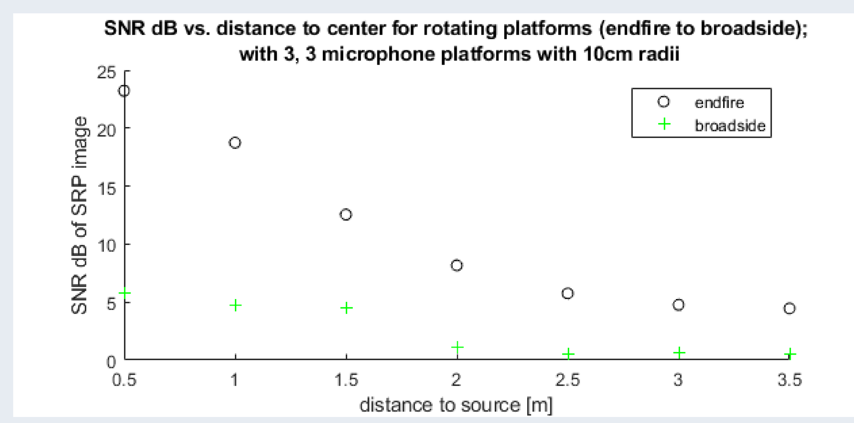

Figure 5: SNRdB results for 3 platforms with 3 microphones

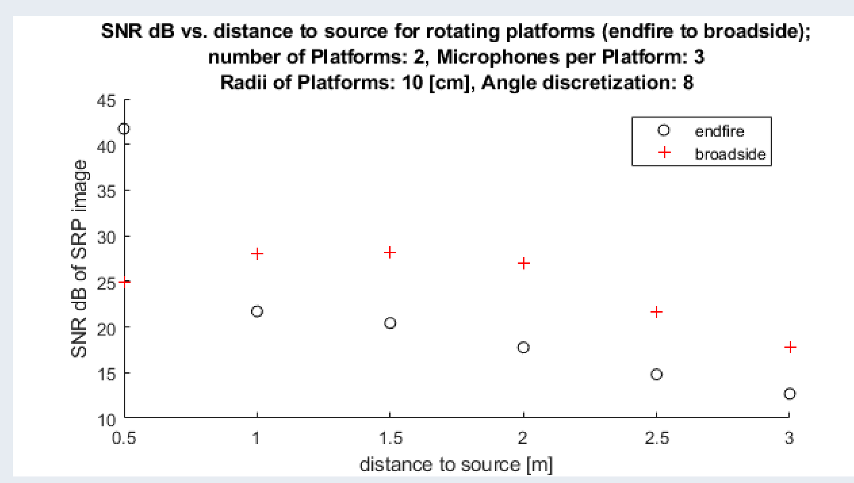

Figure 6: SNRdB results for 2 platforms with 3 microphones

Conclusion

The most noticeable improvement in SNR of up to 15 dB was observed when simulating 3 platforms with 3 microphones each for the near-field placement (1m to source). At all distances the end-fire orientation performed better. For 2 platforms, 3 microphones each, the broadside showed higher SNR values (up to 10dB difference) than the end-fire.

Future Research

- Simulate and test the robustness of the array to noise sources
- Automate the orientation to improve SNR

References

[1] K. Donohue, PhD, “Audio array toolbox.” [Online]. Available: http://vis.uky.edu/distributed-audio-lab/about/

https://github.com/michaelsikora/EE959sims