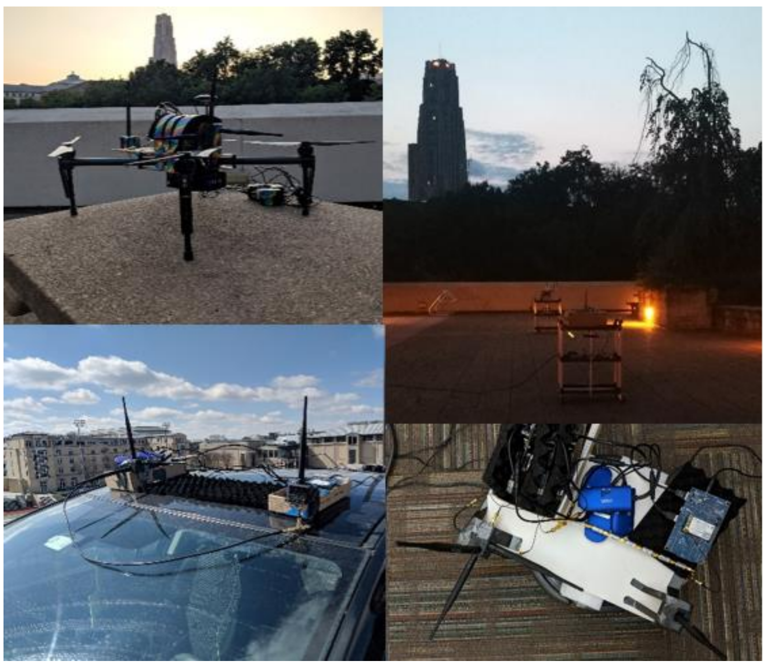
PROJECT REPORT



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ACKNOWLEDGEMENT

With immense pleasure I, Miss **Diksha Agarwal** presents project report as a part of the curriculum of Carnegie Mellon School of Engineering (ECE Department). I would like to thank all the people for providing me unintended support.

I would like to express my profound thanks to **Professor James Bain** for providing me with various resources and support during my course work. I would also like to extend my gratitude to my faculty advisor **Professor Tamal Mukherjee** for their constant support and motivation.

CONTENTS

1. Scope:

Context:

I worked under the guidance of Professor James Bain. The project involves localization of sound sources using four microphone arrays (we have considered a square geometry).

Motivation:

The primary motivation behind this project was the localization of sound sources in a conference meeting room. So, basically, we intend to zoom or focus our camera on the person who is speaking. Also, we aimed at removing background noises when the speaker is speaking in the zoom calls etc.

2. Professional and Personal Development:

During the course, I got a chance to network with people from Quadric. It was a great learning opportunity as I got in contact with founders of Quadric. I got to know more about their processors and basically the whole idea of startup came up from robotics. Also, I learnt how to analyze a problem statement. Apart, from that I got a chance to network with people working on various projects like how to use LIDAR, drone projects etc.

3. Skills and Implementation:

For the localization, my algorithm is as follows:

1. Consider a microphone array. (Considered a 4 micro phone array, square is the use case)
2. Consider speaker source at SX, SY (typically inside square). This is ground truth value.
3. Compute transit time to each microphone (4 values). Using Euclidean distance formulae.
4. Generate a sound signal sampled at 20 ksamples/sec i.e. (0.25 s) for example approx. 20 pts)
5. Create associated time vector to correspond to sampling times
6. Add delay to the four sound signals using circshift in MATLAB. Basically, the delay will be added as number of points shifted. To calculate the number of points to shift we do (transit time calculated in Step 3)/ time interval. Here time interval is 1/sampling freq.
7. Now we will use xcorr function in MATLAB.
8. After doing co-relation we will have the number of points each signal is shifted with respect to each other. So, we can calculate delay using (number of points) * time interval.
9. Now, basically we know time difference of arrival. So, it will be a hyperbola trajectory. So, we can calculate the estimated source location.
10. After finding estimated source location we can calculate the error with ground truth value.
11. I have attached a video demo.

Results:

```
Enter the x-location of source 3
Enter the y-location of source 4
Enter number of mics 4
Enter distance between mics 6
Enter the co-ordinate of microphone array 1 2
Enter the co-ordinate of microphone array 2 0
Enter the sampling frequency 20000
>> error

error =

    2.2361

>> sol

sol =

  struct with fields:

    X_unknown: 5
    Y_unknown: 3
```

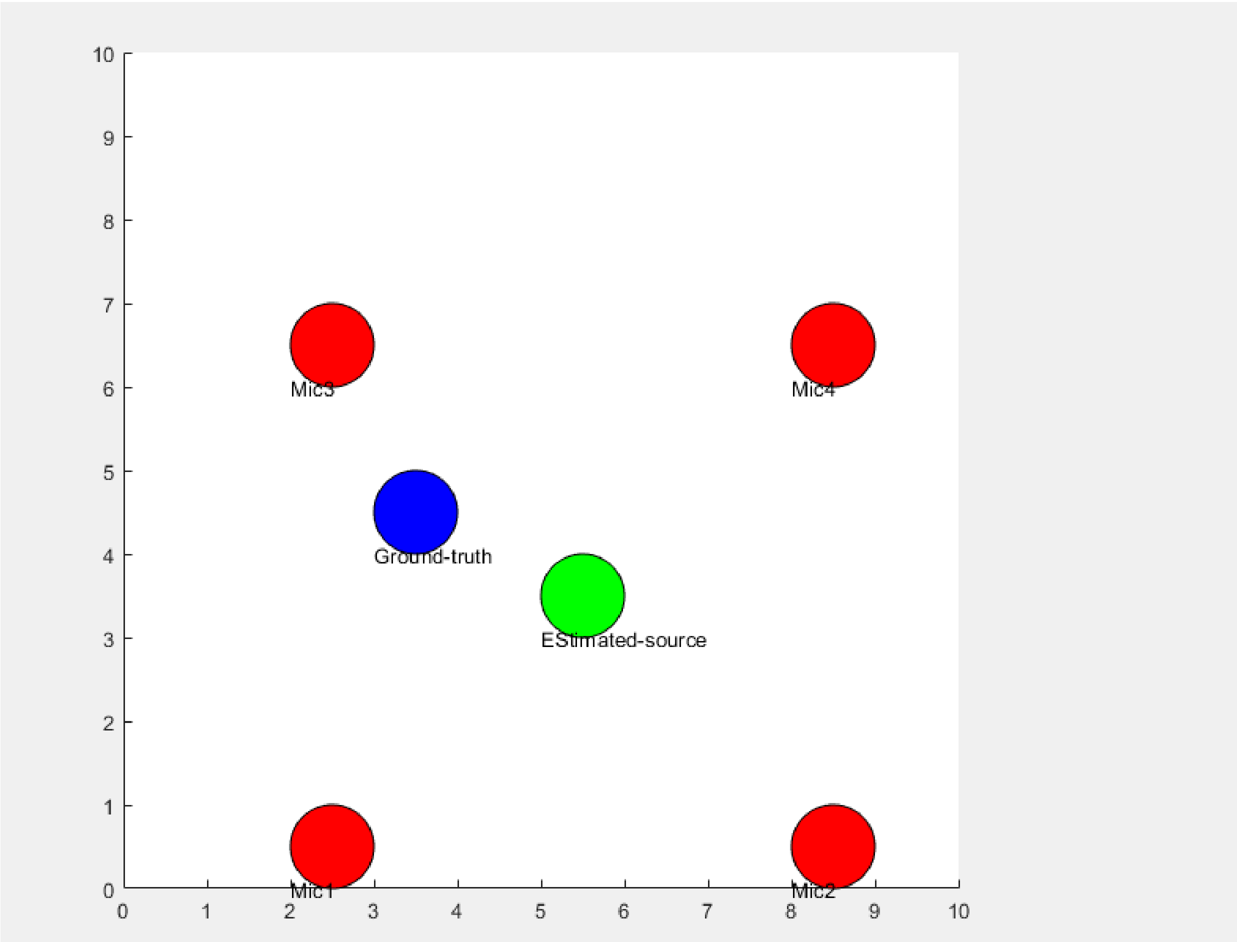

Figure 1: Basically, the figure shows the ground truth and estimated value

Future Work:

We plan to come up with a startup idea with this project. For now, it is just a basic implementation in MATLAB. We plan to develop it fully on Quadric process next semester.