

Painting with Music

Weijian Zhou
University of Toronto
Toronto, Canada

Parham Aarabi
University of Toronto
Toronto, Canada

Abstract—In this paper, we proposed and built an inexpensive, portable yet reasonably accurate local area sound localization system with two microphone arrays. We evaluated different array architectures and demonstrated that a two array system outperforms a single array system of similar size. The proposed system localizes in 80 millisecond and achieved an average error of less than 3 cm for both point localization and movement tracking in a local one meter by one meter region.

I. INTRODUCTION

Accurate indoor localization allows creation of novel applications with surrounding awareness that uses position and movement information as input. One application is to allow users to draw with music, without physically touching the computer. Another example is to build AI games with physical pieces such as toy car racing where the computer controls some of the pieces. In this work, we aim to build such a source localization system that is portable, inexpensive, yet reasonably accurate for localization in a small area.

Global Positioning System (GPS) is the prevailing technology used for outdoor localization. Commercial grade GPS has an average error of a few meters, depending on the size and quality of the receiver [1]. While accuracy in this range is good for many applications including driving navigation and vehicle tracking, it does not provide enough precision for local movement tracking. Ultrasound based indoor localization has achieved sub-centimeter accuracy [4]. However, ultrasound systems require use of expensive transducers.

Bluetooth and Wi-Fi based technologies have gained popularity in indoor positioning recently, mainly due to the widespread deployment of bluetooth tags and Wi-Fi stations in public spaces. In these systems, signal strength received from different base stations are used for the estimation of the device location. However, their reported accuracy are in the range of 1 to 5 meters [2], [3], which is not enough for local movement tracking.

In this project, we have built a local area localization system using microphone array that localizes normal audio source. The system is built with inexpensive electret microphones. People can interact with the system using any device that has audio output such as a mobile phone.

In this paper, we discuss prior relevant research and the theory in sound localization in Section II. In Section III, we evaluated different array architectures and their impact on accuracy. We demonstrated that a two array system outperforms a single array system of similar size. In Section IV, we present the chosen architecture along with hardware details. Finally, experiment details and results are presented in Section V.

II. BACKGROUND

Acoustic localization has been researched extensively in the literature. Localization techniques can be broadly categorized into Location Template Matching (LTM) based approaches and Time Difference of Arrival (TDOA) based approaches.

A. LTM

In LTM based approaches, acoustic templates of locations are first stored in the system during a “training” phase. Then, incoming acoustic waveform is compared with stored templates. Localization result is the location that has the best matching template. Different ways of extracting templates from raw acoustic source and different similarity measures have been investigated in the past.

[5] and [6] investigated using max value from cross-correlation as similarity measure to localize user tap on interactive surface. [7] used L2 distance in Linear Predictive Coding coefficient space as similarity measure to localize taps on surfaces. [8] further explored accuracy improvement by using multiple templates for each location and speed improvement by merging multiple templates into one representative templates.

The requirement of having a template for each location to be detected makes this approach too restrictive for our project, since we want the drawing to be continuous in the 2D region. Moreover, the need to recalibrate all locations during setup is too cumbersome for end users in a portable localization system. Therefore, our main focus is on TDOA approaches.

B. TDOA

Points with the same TDOA to two fixed points on a plane form a hyperbola. With more than two microphones, intersection of hyperbola curves generated by each pair of microphones give the source location. Localization relies on accurate estimate of delay differences between microphones.

In [9], authors used eight microphones mounted on corners of a ping pong table to localize the point where the ping pong ball hits the table. They used a threshold to determine the arrival time of acoustic signal. This approach works well in noise free environment but performance degrades with background noise. It also suffers from dispersive deflections that arrives before the main wavefront. To make it more robust, authors in [13] and [14] extracted descriptive parameters for each significant peak(e.g., peak height, width, mean arrival time). The algorithm then used extracted parameters to predict arrival time with a second order polynomial, the parameters of which were fitted during calibration at fixed locations.

Cross-correlation has also been used in measuring signal delays[10], [11], [12]. Cross-correlation with different prefilters (known as *generalized cross correlation (GCC)*) have also been investigated to improve delay estimation [15], [16], [17].

GCC provides a framework to estimate delay differences t_0 between two signals $x_1(t)$ and $x_2(t)$:

$$t_0 = \arg \max_{\tau} R_{x_1 x_2}(\tau) \quad (1)$$

$$R_{x_1 x_2}(\tau) = \int_{-\infty}^{\infty} W(\omega) X_1(\omega) X_2^*(\omega) e^{j\omega\tau} d\omega \quad (2)$$

, where $X_1(\omega)$ and $X_2(\omega)$ are Fourier Transform of $x_1(t)$ and $x_2(t)$. $W(\omega)$ provides a way to prefilter signals passed to cross correlation estimator. We focused on three ways of prefiltering the signal:

GCC

$W(\omega) = 1$. No prefiltering is done. This is unfiltered normal cross correlation.

GCC_PHAT

$W(\omega) = \frac{1}{|X_1(\omega)X_2^*(\omega)|}$. Each frequency is divided by its magnitude. Only phase information contributes to delay estimation.

GCC_PHAT_SQRT

$W(\omega) = \frac{1}{|X_1(\omega)X_2^*(\omega)|^{0.5}}$. This is somewhere between GCC and GCC_PHAT. part of magnitude information is included in delay estimation.

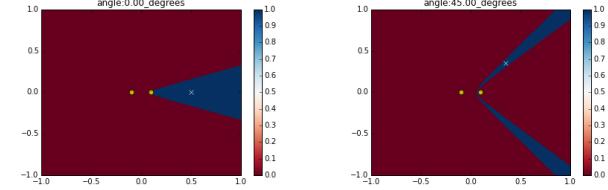
III. ARRAY ARCHITECTURE

As was mentioned in the previous section, points with the same TDOA to two fixed locations form a hyperbola in a 2D plane. In practical systems, we can only measure TDOA up to a precision. Therefore we look at all points with difference of distance close to some target value within measurement error ϵ . This ϵ represents accuracy on measurement of difference of distances, and in practice it is related to sampling rate and TDOA methods. In this section we evaluate delay estimation's impact on localization accuracy.

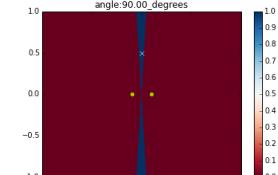
To see how precision affects localization accuracy, we simulated two microphones placed at: $M_1 : (x = -10 \text{ cm}, y = 0 \text{ cm})$ and $M_2 : (x = 10 \text{ cm}, y = 0 \text{ cm})$. A test sound source is emitted at point P which is 50 centimeters away from $(0, 0)$. Fig 1 shows the region where all points \hat{P} satisfy:

$$|(\hat{P}M_1 - \hat{P}M_2) - (PM_1 - PM_2)| < 1 \text{ cm}$$

Intuitively, points in the region have difference of distance very similar to each other. From fig 1, the region still has the shape of a hyperbola, but with an uncertainty region around the curve. The uncertainty region is not uniform around the curve, the farther away the point is, the larger the uncertainty region becomes. It indicates that the same delta distance movement will generate smaller difference of distance when the source is farther away from the array. The size of the uncertainty region is also angle dependent: points closer to the line of microphones have larger region compared to points close to the line perpendicular to microphones.



(a) source at $(r = 50 \text{ cm}, \theta = 0$ degrees) (b) source at $(r = 50 \text{ cm}, \theta = 45$ degrees)



(c) source at $(r = 50 \text{ cm}, \theta = 90$ degrees)

Fig. 1: Uncertainty region

This can also be seen analytically. Assuming two microphones are placed at $M_1 : (-c, 0)$ and $M_2 : (c, 0)$. All points $P : (x, y)$ with difference of distance $|PM_1 - PM_2| = 2a$ satisfies:

$$\frac{x^2}{a^2} - \frac{y^2}{c^2 - a^2} = 1 \quad (3)$$

To see how difference of distance changes with respect to distance, we can expand the equation and find partial differential $\frac{\partial a}{\partial x}$:

$$\frac{\partial a}{\partial x} = \frac{x(c^2 - a^2)}{a(x^2 + y^2 + c^2) - 2a^3} \quad (4)$$

Since all points in equation 4 must lie on the hyperbola, we can substitute 3 into 4:

$$\frac{\partial a}{\partial x} = \frac{c^2 - a^2}{\frac{c^2}{a}x - \frac{a^3}{x}} \quad (5)$$

The denominator of equation 5 increases monotonically as $|x|$ increases, which indicates $\frac{\partial a}{\partial x}$ decreases as we move farther away along the hyperbola. The same distance move δx would generate smaller change in difference of distance a when source is farther away from the microphones.

With more than two multiple microphones, each pair of microphones generates a hyperbolic region and localization becomes finding the intersection of hyperbolic regions. The smaller the intersection region, the better the localization accuracy. To see how accuracy changes with array placement and sound source location, three microphones are placed at three vertices of an 20 cm equilateral triangle. An audio source is placed at 20 cm away from the center of the array. Fig 2 shows the intersection of regions for 5 different placement of the sound source. It can be seen that accuracy is worse when sound source is close to the line of any two microphones. This observation is consistent with two microphone case, since

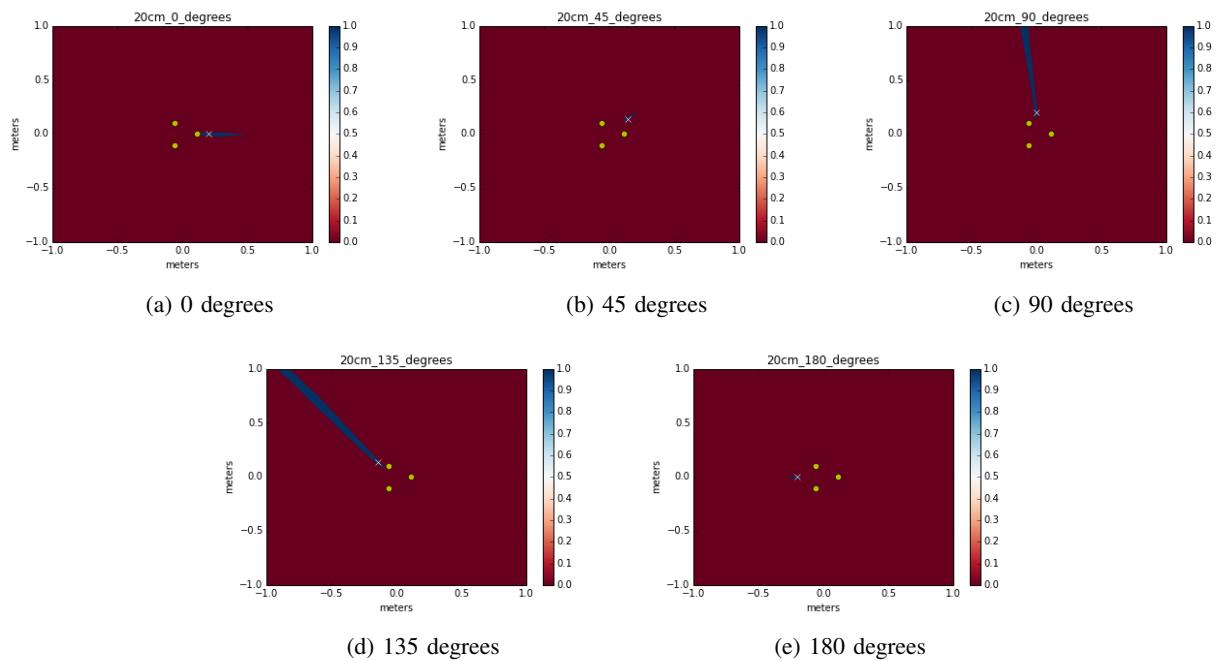


Fig. 2: 20cm equilateral triangle array. Source is 20cm away from the array

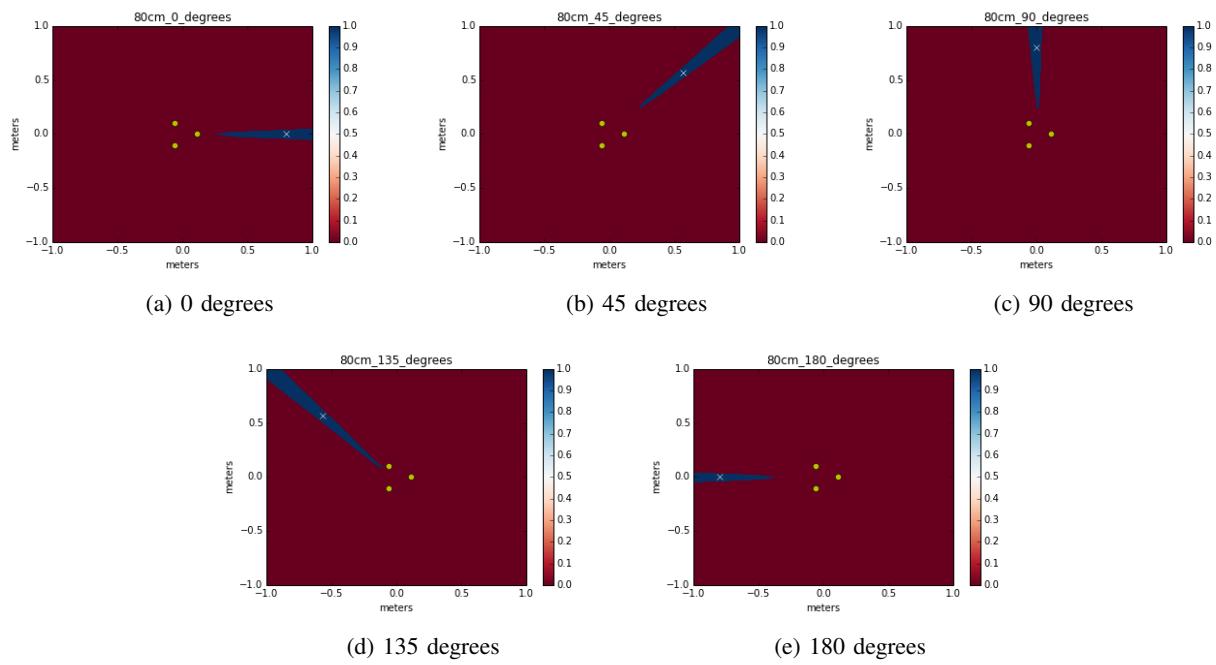


Fig. 3: 20 cm equilateral triangle array. Source is 80 cm away from the array

points close to line of microphones have a larger uncertainty region.

To see how sound source distance affects localization accuracy, the same simulation is carried out with the sound source moved from 20 cm to 80 cm away from the center of the array. Results are presented in fig 3. Comparing with fig 2, accuracy decreases with distance to the array. This is also consistent with our observation in 2 microphone case where source farther away would result in larger uncertainty region.

Intersection area is a measure of the localization accuracy. To evaluate an array's accuracy in a region, we can place sound source at predetermined grid points in the region and look at the intersection area for each tested point in the grid. The center location of intersection region can be used as localization estimate to calculate localization error. Results for a few different microphone array configurations are presented in fig 4.

Fig 4a shows the accuracy when microphones are placed at three vertices of a 20 cm equilateral triangle. The region inside the array has good accuracy. However, for regions along line of any two microphones, the accuracy drops significantly. Average error across the region is 18.6 cm.

To evaluate how adding one microphone (without increasing array size) improves accuracy, another microphone is added to the array at (0, 0). Result is presented in fig 4b. Addition of the new microphone only slightly improved the accuracy around the array region. Average error dropped from 18.6 cm to 17.1 cm. Regions near lines of microphones still have significantly larger uncertainty region.

To evaluate array size's impact on accuracy, the size of original array from fig 4a is increased by a factor of 2. The result is presented in fig 4c. The overall uncertainty area decreased across the region. Average error improved to 10.04 cm.

In fig 4d, three microphones are placed 10 cm apart from each other on x-axis. Error heatmap shows high uncertainty on the x axis, and the overall accuracy is not as good as that with three microphones placed in a triangle. The average error is 55.05 cm.

To further increase the distance between microphones, we placed four microphones at four corners of the region. Fig 4e shows the result. With this configuration, accuracy is consistently good across the region. The average error is 0.05 cm. However, placing microphones far apart at corners of the region requires accurate placement of all four individual microphones. The system is less portable compared to small arrays with microphones near each other. Placing microphones far apart from each other also causes problems in TDOA estimation, because sampling of microphones in the same array requires synchronized clock.

To avoid the need to accurately place four microphones at far distances (as required by fig 4e), we explored configuration with two arrays. Two 3 microphone array are placed 1 meter apart and the result is presented in fig 4f. The result indicates that this configuration has good accuracy when source is close to the arrays. Accuracy decreases as sound source moves outside the one meter by one meter region. The average error is 2.60 cm.

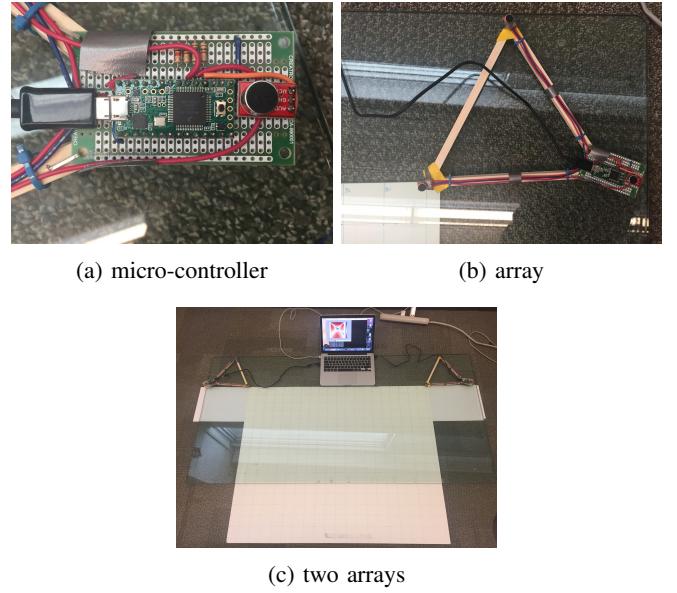


Fig. 5: Localization arrays

With simulation results, we decided to build the two array system as described in fig 4f. The setup is reasonably portable (compared to fig 4e), while at the same time having significantly better accuracy compared to one array systems.

IV. SYSTEM SETUP

The end system has two arrays, each with three microphones mounted on vertices of a 20 cm equilateral triangle. A micro-controller is also attached to one of the vertices. Fig 5 shows a picture of the array setup. Micro-controller used in this project is *teensy 3.1*. It has 64k RAM memory and the ADC is capable of sampling at 500kHz. In this project, the micro-controller collects microphone data on all three channels for 12 millisecond and then send the recorded data to a computer through USB port for localization.

To handle uncertainty in TDOA estimation, instead of using point estimate that maximizes equation 1, we take the cross-correlation function 2 as a mapping from delay to likelihood. For each microphone array, we build a likelihood heatmap of each point in the grid being the sound source. For each point in the grid, the theoretical TDOA to each microphone pair can be precomputed. Then the heatmap can be generated by going through points in the grid and perform a lookup using equation 2. With three microphones m_1, m_2, m_3 , there are three microphones pairs: m_1m_2, m_1m_3, m_2m_3 . Theoretical TDOA from each location (x, y) to each microphone pair is precomputed and stored in $D_{m_1, m_2}(x, y)$, $D_{m_1, m_3}(x, y)$, and $D_{m_2, m_3}(x, y)$. Then the likelihood map $L(x, y)$ can be built as:

$$L(x, y) = R_{m_1, m_2}(D_{m_1, m_2}(x, y)) + R_{m_1, m_3}(D_{m_1, m_3}(x, y)) \\ + R_{m_2, m_3}(D_{m_2, m_3}(x, y))$$

, where $R_{m_1, m_2}(\tau), R_{m_1, m_3}(\tau)$, and $R_{m_2, m_3}(\tau)$ denote GCC output from microphone pairs m_1m_2, m_1m_3 , and m_2m_3 .

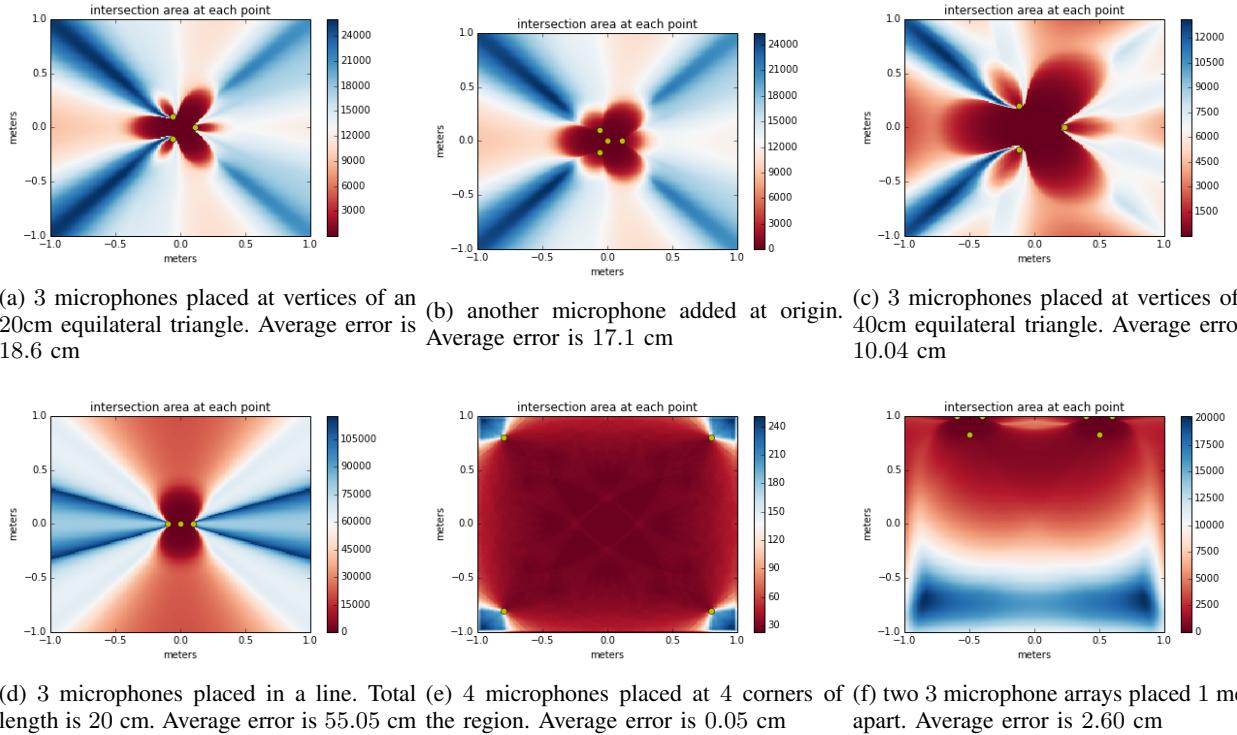


Fig. 4: Accuracy for different array configurations

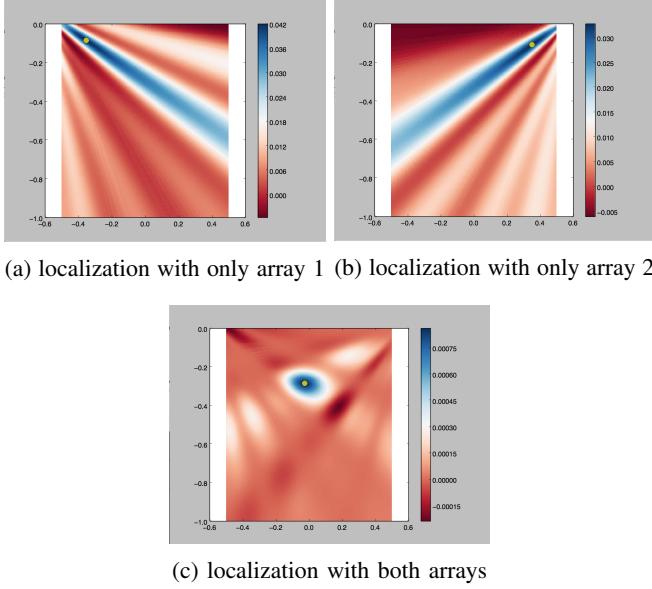


Fig. 6: Likelihood maps for localizing point at $(0.0, -0.3)$ m

Likelihood maps from two arrays can be combined into final likelihood map:

$$L(x, y) = L_1(x, y)L_2(x, y) \quad (6)$$

, where $L_1(x, y)$ and $L_2(x, y)$ represents the likelihood map from array 1 and array 2.

To see the effect of accuracy improvement using multiple arrays, fig 6 shows a real life localization where the source is placed at $(0 \text{ cm}, -30 \text{ cm})$. It shows the individual likelihood map from each array and also the combined likelihood map according to equation 6. Individual array gives accurate angle estimate, but has high uncertainty in source distance. It demonstrated that by combining estimates from two arrays the system is able to perform more accurate localization.

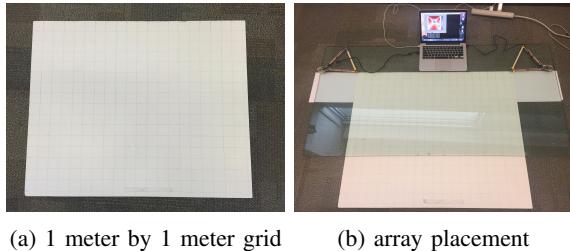
From a timing point of view, the micro-controller spends 12 millisecond on sampling microphone data before sending the data to the computer for processing. Sending data through USB port takes another 18 millisecond, and processing on computer takes around 50 millisecond. Therefore, the total time lag between sound source and localization is around 80 millisecond.

V. EXPERIMENT

A. Setup

1) Point localization: To test localization accuracy, an one meter by one meter grid was set up and the arrays are placed at the top left and top right corners of the grid. Fig 7 shows a picture of the setup. A total of 32 positions are chosen uniformly in this region where microphone data is recorded.

2) Movement tracking: To test how well the arrays track movement, we mounted a rotating disk 40 centimeter in diameter onto the grid at $(x = 0 \text{ cm}, y = -30 \text{ cm})$. Fig 11 shows a picture of the setup. A sound source is placed on the edge of rotating disk and the arrays localize the sound



(a) 1 meter by 1 meter grid (b) array placement

Fig. 7: Setup for localization accuracy testing

source as it rotates in a circle. In this experiment, we tested how accuracy changes with:

- different window sizes
- different audio sources
- different movement tracking filters
- different movement speeds

To test how different sound sources impact localization quality, we conducted three experiments on the same movement track with three different sound sources:

White Noise	A recording of white noise.
Music A	A music that has normal audio amplitude throughout experiment period was chosen. “Honest Eyes” by Black Tide was the music used.
Music B	A music with intermittent low amplitude sections was chosen. “Canon” was the music used.

To test how sound source movement speed affects localization quality, each of three experiments were conducted at two different speeds:

Normal	An angular speed of 0.5 rad/s was maintained, which translates to linear speed of 10 cm/s.
Fast	An angular speed of 1.0 rad/s was maintained, which translates to linear speed of 20 cm/s.

For each experiment conducted, two different movement filters are tested:

Averaging filter	localization for past 0.5 seconds are averaged and outputted as current estimate.
Kalman filter	A 2nd order Kalman filtering is used.

B. Results

1) *Point localization*: To test how accuracy varies with window size, the algorithm is fed with recorded microphone data with different segment length. Fig 8 shows how accuracy changes with window length for three GCC algorithms. The error decreases as window size increases and plateaus after window size exceeds around 10 millisecond. The lowest error achieved is 2.53 centimeters. It is achieved when window size is set to 12 millisecond and GCC_PHAT is used for TDOA estimation.

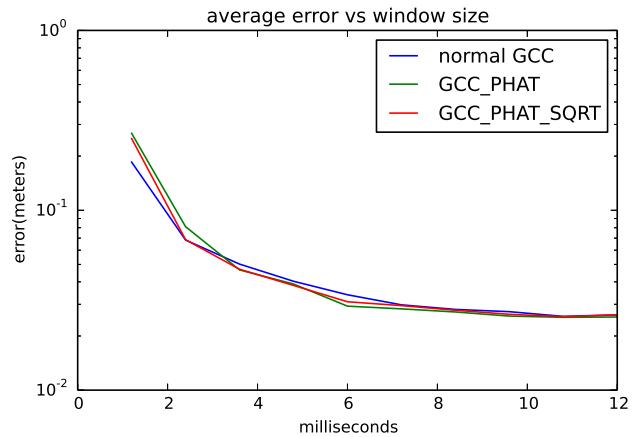


Fig. 8: accuracy versus window size

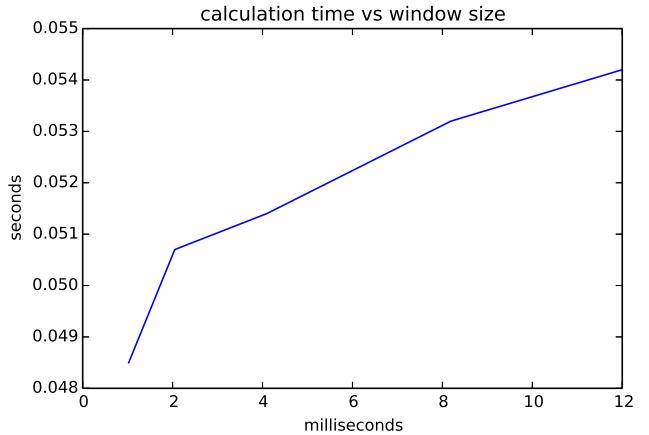


Fig. 9: speed versus window size

Although accuracy improves with window length, the calculation time also increases with window length. The part of calculation that depends on window length is using cross correlation to estimate TDOA. cross correlation can be calculated with FFT and the runtime is of order $O(N \log N)$. We measured how the computation time varies with window length and Figure 9 shows the result. The runtime increases approximately linearly in the window size region of interest.

We also calculated the localization error for each tested point in the region. Figure 10 shows a heatmap of the error distribution inside the grid. The error is below 3 cm for most areas inside the region. There is one error spike in the mid-left region and we contribute this to audio source misplacement because the error is fairly low and consistent around that spike region.

2) *Movement tracking*: Fig 13 gives an intuitive representation of how accuracy changes with window size. When window size is small(1.02 millisecond), the audio does not contain enough information to reliably estimate TDOA. The localization is noisy. As window size increases, the localization converges to the shape of ground truth circle. Fig 13 shows

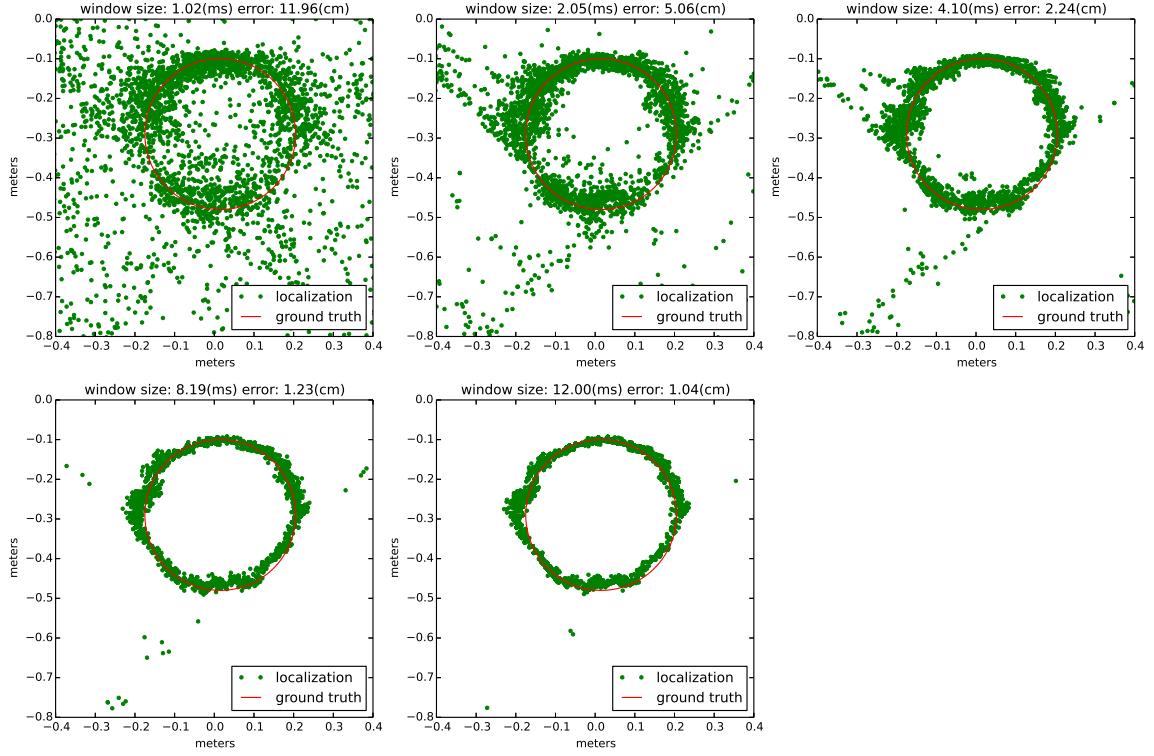


Fig. 12: Localization quality versus window size

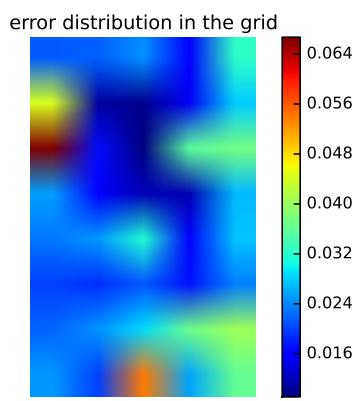


Fig. 10: error distribution in the grid

how the error changes with window size. The general trend is similar to that in point localization case. The error decreases as window size increases and plateaus after window size exceeds around 10 milliseconds.

Fig 14 shows results for experiments at normal speed, and Fig 15 shows results at fast speed. By comparing localization

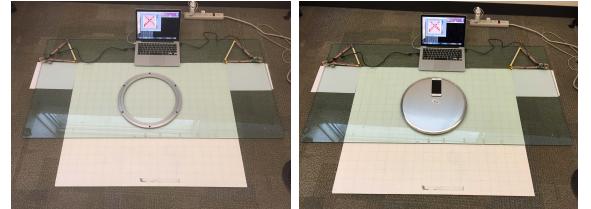
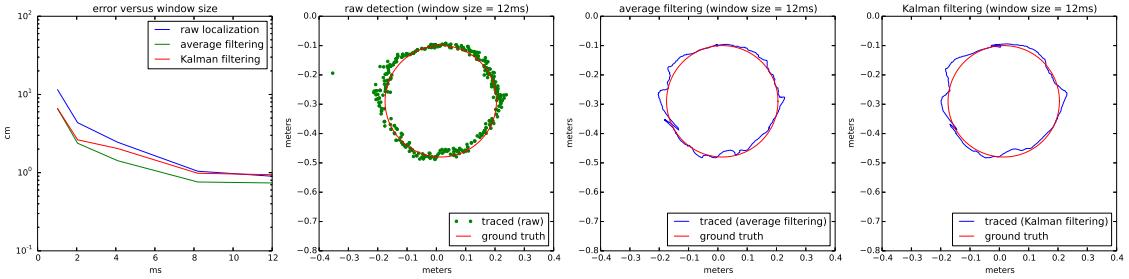


Fig. 11: Setup for circle movement localization

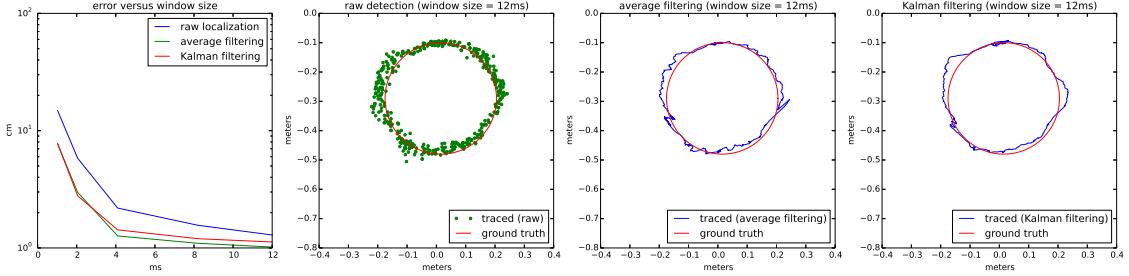
error for each audio source between normal movement speed and fast movement speed, we find the localization error does not depend on how fast the sound source is moving. For example fig 14b and fig 15b shows that localization error is 1.289 cm at normal movement speed and 1.291 cm at fast movement speed.

From fig 14, localization error is 0.9 cm for white noise, 1.29 cm for Music A, and 2.9 cm for Music B. Localization accuracy is the best for white noise source, and worst for Music B. This is consistent with our expectation because low amplitude regions in Music B would cause arrays to lose track of where the source is. This can also be seen from the "blank" regions in fig 14c.

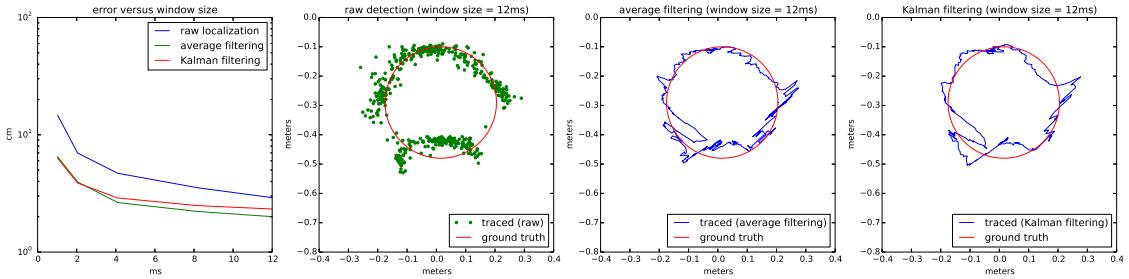
Fig 14 also shows that raw detection has the most amount



(a) white noise



(b) music A



(c) music B

Fig. 14: Localization of circle movement with different sound sources. Sound source is moving at 10 cm per second

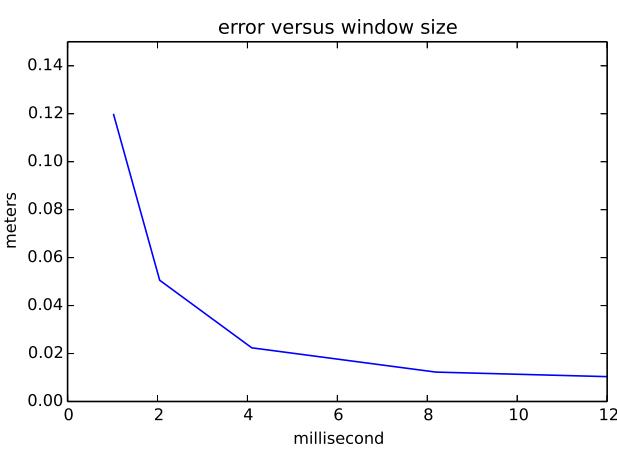


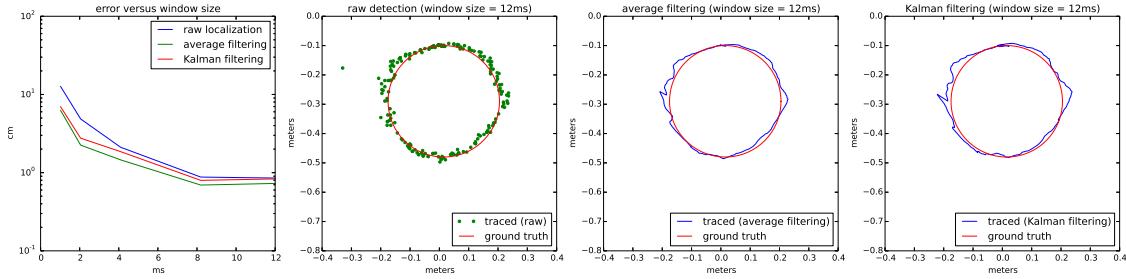
Fig. 13: Localization error versus window size

of jiggling. Kalman filter reduces the amount of jiggling from raw detection. Averaging filter has the least amount of jiggling. However, averaging filter averages detection outputs from past 0.5 seconds, which makes the filtered output lag the real movement.

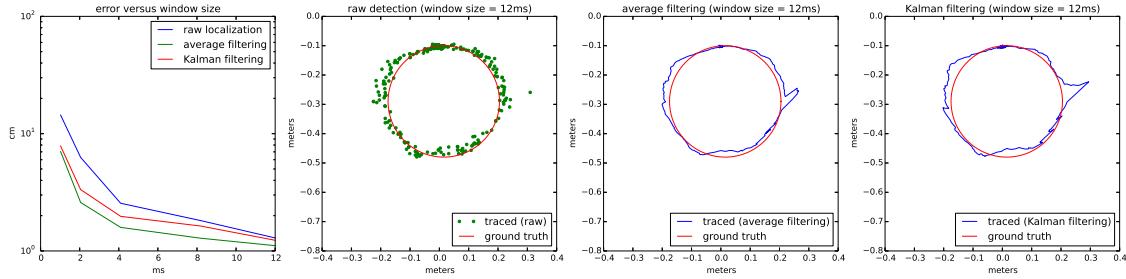
VI. CONCLUSION

In this paper, we proposed and built an inexpensive, portable yet reasonably accurate sound localization system with two microphone arrays. We analyzed different array architectures and demonstrated that a two array architecture achieves better accuracy compared to single array system of similar size. Using cross-correlation output as likelihood for each input delay, each array outputs a heatmap of likelihood for each possible location in the area. We demonstrated that by merging two heatmaps from two arrays, we were able to achieve good localization accuracy (less than 3 cm) in a local one meter by one meter region.

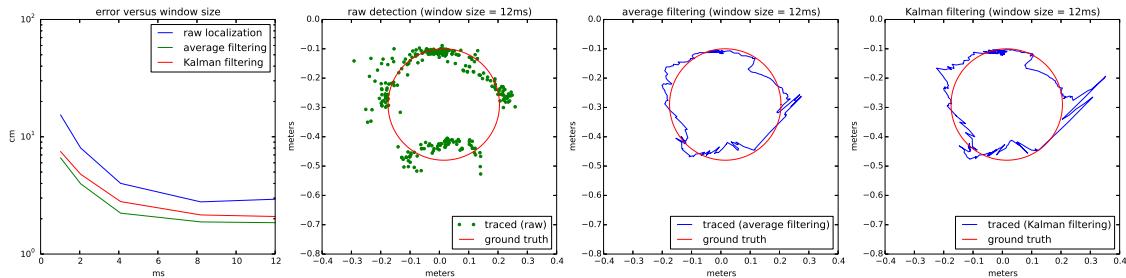
This system can be used in HCI applications that uses



(a) white noise



(b) music A



(c) music B

Fig. 15: Localization of circle movement with different sound sources. Sound source is moving at 20 cm per second

sound position and movement as input, such as drawing with music. Another application is to build AI games with physical pieces where the computer controls some of the pieces. This system can also be used to build Augmented Reality (AR) applications that allow user computer interaction with music tags.

For future directions, this system can be extended with another microphone on each array to perform 3D localization. Another possible direction is to investigate multisource localization.

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