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Accurate acoustic ranging system using android smartphones

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A thesis submitted in partial fulfillment of the requirements for the degree of

Master of Science at Virginia Commonwealth University

By

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List of Contests

I. ABSTRACT

ACCURATE ACOUSTIC RANGING SYSTEM USING ANDROID SMARTPHONES

By Mohammadbagher Fotouhi, Master of Science

A thesis submitted in partial fulfillment of the requirements for the degree of Master of Science at Virginia Commonwealth University

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Major Director: Dr. Ruixin Niu, Associate Professor of Department of Electrical and Computer Engineering

In this thesis, we present the design, implementation, and evaluation of an android ranging system, a high-accuracy acoustic-based ranging system which allows two android mobile phones to learn their physical distance from each other.

In this system we propose a practical solution for accurate ranging based on acoustic communication between speakers and microphones on two smartphones. Using the audible-band acoustic signal with the Wi-Fi assistance without the sound disturbance is promising for large deployment. Our method is a pure software-based solution and uses only the most basic set of

commodity hardware: a speaker, a microphone, and Wi-Fi communication. So it is readily applicable to many commercial-off-the-shelf mobile devices like cell phones.

Our system is the result of several design goals, including user privacy, decentralized administration, and low cost. Rather than relying on any centralized management which tracks the user's location to help them find their distance, our system helps devices learn their distance from each other without advertising their location information with any centralized management. Compared to alternatives that require special-purpose hardware (such as [20,29]) or pre-existence of precision location infrastructure [14], our system is applicable on most of off-the-shelf components so it is a commodity-based solution will obviously have wider applications and is cost effective.

Currently, two smartphones are used to estimate the distance between them through Wi-Fi and audio communications. The basic idea is estimating the distance between two phones by estimating the traveling time of audio signal from one phone to the other as the speed of sound is known. The preliminary results of ranging demonstrate that our algorithm could achieve high accuracy, and stable and reliable results for real time smartphone-based indoor ranging.

II. INTRODUCTION

These days smartphones are becoming the most popular personal computing companions. Since most of part of our daily life has been spent indoor, the idea of indoor mobile ranging and positioning systems on Android or IOS devices seems like an attractive topic for both customers and industries. The evidence is the major investment of most IT companies on mobile indoor ranging systems. Using the built-in capabilities of smart devices such as voice user interface, wireless communication hardware commodities, speaker and microphone and Wi-Fi. Accordingly, various acoustic localization techniques have been introduced to utilize the voice user interface.

In this thesis, we aim to achieve high accuracy ranging using only these most basic hardware commodities, because it will be widely applicable in many sensing and mobile applications and it will be feasible and less costly. Also, we can claim that our goal will result in a user friendly system, because the set of the mentioned hardware capabilities are the basic hardware part of many sensor platforms and component off-the-shelf (COTS) devices compared to alternative systems that require special-purpose hardware such as self-calibrating acoustic platform [1] or Pinpoint system [2]. Therefore, we can claim that using acoustic wave to create a high accuracy ranging system will be the best answer to our goal.

Even though environment localization and ranging using acoustic wave has its advantages, the design and deployment of a high accurate system for indoor environment is a challenging task. The challenges that should be addressed in our design include, the preservation of user privacy, administration and management overheads, system scalability, and the harsh

nature of indoor wireless channels. The degree of privacy offered by the system is an important deployment consideration, since people often value their privacy highly. By not tracking users and services, user-privacy concerns are adequately met. In this section we describe our design goals, then we will have a review with previous ranging and localization systems (Chapter III), and we will go over system methodology and challenges (Chapter IV), system design and architecture (Chapters V andVI), implementation (Chapter VII), experimental results and applications (Chapters VIII and IX).

The design of system was driven by the following specific goals:

- Indoor application. Due to the blockage of the global positioning system (GPS) signals in indoor environments, alternative approaches over the years have been proposed to address the problem of indoor automatic location sensing. These approaches vary in many aspects, such as the positioning method, signal type, cost of infrastructure, power requirement, and resolution in time and space. Our approach in this project is using acoustic signals and estimating the distance between two phones by estimating the traveling time of audio signal between them.
- Cost and applicability. Achieving building-wide deployment requires cost effective components. Our proposed indoor ranging system makes the high-accurate indoor localization possible by using smartphones without additional hardware. Our method is a pure software-based solution so that it is readily applicable to many commercial-off-the-shelf mobile devices like cell phones.

- User privacy. Whenever a system for providing location or ranging information to clients has been deployed in the past, the issue of user privacy has arisen. This is because many

previous systems were location tracking systems, where a database kept track of the locations of all the entities, including users in the system. To address this concern, we designed a ranging support system, which allows different clients to learn their distance from each other without centralized tracking in order to construct location-specific queries for resources. Also to differentiate the audio signals sent by different phones, direct sequence spread spectrum (DSSS) is adopted to encrypt audio signals.

III. BACKGROUND

Indoor positioning/ranging systems are the systems which use wireless concepts, optical tracking, or sonic/ultrasonic techniques to locate and track objects within the buildings and closed environments. Among many solutions for position/distance estimation of objects [1,2], the most common methods provide positional information, triangulation and multi-lateration methods using light [3,4], sound/ultrasound [5,6], radio signals, [7,8,10,11], and other techniques such as inertial methods [12,13] which provide relative positioning, accumulate errors in time, and require periodic recalibration.

A few applications of indoor positioning systems can range from finding required objects to locating people in closed environments. A few application examples that motivate indoor ranging systems are locating patients in the hospital, finding people trapped in a burning building, or finding workers in a large office block. Different ranging/localization systems with different architectures to determine the location of objects have been proposed. Each model has different configurations, accuracies, and reliabilities.

Some outstanding positioning systems are GPS [14], AT&T Cambridge Ultrasonic Bats [21], Microsoft Research's Wave-LAN system [15], Active Badges [16], Smart Floor from Georgia Tech [17], Radio tags, Computer vision systems [18], and cellular phone based systems.

In this chapter, first, we give a general overview of GPS system. Then we go over some of the most popular indoor positioning systems, using infrared, ultrasonic, and RSSI techniques together with computer vision, cell phone, and Integrated radio frequency identification (RFID) positioning systems.

A. The Global Positioning System (GPS)

GPS is the most popular system to find the location and the position of the objects. The worldwide satellite network is used to measure the distances with a great accuracy. Object locations can be estimated to be within 1 to 5 meters of the true locations. But it generally does not function when the receivers are indoors. GPS is an outdoor positioning system. It receives signals from multiple satellites and employs a triangulation process to determine physical locations. Although the signals are heavily attenuated and reflected by the building materials, it was observed that high sensitivity GPS receivers can track people through 3 layers of brick walls. But the positioning accuracy is very low. GPS is made up of three parts: Space, Control and User. The Space part is composed of 24 to 32 satellites in medium earth orbit. The Control part is composed of a Master Control Station and many shared Ground Antennas and Monitor Stations. The User part is composed of hundreds of thousands of military, civil, commercial, and scientific users. GPS satellites broadcast signals from space to GPS receivers to provide 3-D location information (latitude, longitude, and altitude) and precise time. GPS cannot provide location information for indoor use. This is because the electromagnetic waves will be scattered and attenuated by the buildings and outdoor obstacles.

B. Infrared Positioning system

Active badges is the first indoor location sensing system developed by AT&T Cambridge [3]. A miniture infrared beacon, worn by every person, emits a unique code identifier every 15 seconds. Each location in a building covered with a network of IR sensors which detect these transmissions. A central server collects data from fixed IR sensors around the building, and

gathers them into a central data bank, and the location of the badge (hence its wearer) can thus be determined.

C. Ultrasonic positioning systems

a) Active bats: AT&T Cambridge has developed an ultrasonic tracking technology which provides more accurate indoor positioning results than the previous Active Badges. Users and objects are tagged with ultrasonic tags identified as “bats”. The system was described in [21]. These bats emit periodic ultrasonic signals to receivers mounted across the ceiling. This system produces basic position data and additional orientation information. The problems of using this ultrasonic technique are the requirement of large number of receivers across the ceilings and their placements across the ceiling which need quite accurate alignments.

b) Crickets: It is an ultrasonic 3D positioning system with a claimed positioning accuracy of 1–2 cm in an indoor environment of 10 m^3. The Cricket unit can be programmed either as a beacon or listener. Real-time tracking can also be done with an update-rate of 1 Hz. The system details are given in [22].

c) Dolphin: It is an ultrasonic positioning system. “Distributed Object Locating System for Physical space Inter networking” (DOLPHIN) was presented in [23] and [24]. The DOLPIN system consists of distributed wireless sensor nodes which send and receive RF and ultrasonic signals. These nodes are attached to various indoor objects. Using a distributed positioning algorithm in the nodes, DOLPHIN enables positioning of the objects with minimal manual configuration. The system claims an accuracy of 2 cm in a room of 3mx3m in size.

D. RSSI positioning systems

Received Signal Strength Information (RSSI) was employed to estimate the distances between transmitters and receivers. Usually RF signals are used [1]. The locations of the objects are determined by calculating the distance of the object from the transmitters using triangulation or tri-lateration techniques. Initially a test run can be accomplished in an indoor environment to determine the RSSI database for various transmitters [25]. The unknown RSSI data set was then compared with the test database and the best match was obtained. One weakness is that the radio signals attenuate through the walls and the receivers perform poorly in the indoor environment. Typical indoor environments have many walls and obstacles which are made of various materials. As a result, RSSI values change and become unreliable. Many sensors were developed to measure signal strengths and angles of signal orientations. Many algorithms were also developed for better signal acquisition and tracking. There is a trend for integrating various sensors and data sources. They also use the triangulation, trilateration and data matching techniques. RF signal based systems can be categorized into Wave-LAN, Ultra-Wide Band, and RFID.

a) Wave-LAN wireless networking technology: a tracking system in the buildings was developed by using Wave-LAN wireless networking technology [15]. This system uses the signal strength and signal to noise ratio available from the Wave-LAN network interface card (NIC). The system triangulates 2D position of an object within a building by using either empirical data or a mathematical model of indoor radio propagation. Advantages of this system are such that it requires few base stations and it uses the same general wireless networking in the buildings. The disadvantage is that the tracked object must support a Wave-LAN Network Interface Card (NIC). Hence it is difficult to use this system in multi-floored buildings. The system claims an accuracy of finding objects to be within 3 meters of their actual positions.

b) Ultra-Wide Band technology Ultra-Wide band (UWB): impulse radio signals are employed for indoor location and tracking [26]. The system platform was constructed using standard electronic components. The system allows fast performance evaluation and estimates the time of arrivals (TOA) of received pulse signals. These signals were transferred to a server computer where the location of the transmitter was calculated. In this system an indoor object was equipped with an active tag and provided accurate position information even in a multipath propagation environment [28]. The advantages of the UWB signals are to have high temporal resolution and to provide accurate TOA measurements in multi path environment. Transmitted signal has a sequence of short pulses. They propagate in the media and received by the four receivers placed in know positions. Two network interconnect the receivers. Clock network provides reference clock to all receivers. Data network is used for data communication between receivers and the PC. Data received are processed in the PC to analyze the TOA of the receiver pulses and the position of the transmitter on the object is determined.

c) RFID technology: Non-contact and non-line-of-sight characteristics are the advantages of this technology .They can work in high speed and their RF tags can be read in any environment. They are also very cost effective. Some popular RFID location finding sytems are called SPOT ON [8] and LAND-MARC [26]. They manage 3D location sensing based on RSSI. Tags are developed and they measured RSSI to calculate inter distance between the tags. An RFID system contains RFID readers and tags and a communication media between them. The signal strength drops with the square of the distance between the tag and the antenna ıf there is free space around them. In an indoor environment the signal level drops drastically. LANDMARC is a successful RFID positioning system [26] where an RFID active tag is preprogrammed with an ID to be identified by the readers. RFID reader has 8 power levels with level 1 is the shortest and

level 8 is the longest range. Each reader has a predetermined power level which corresponds to a certain range where it can detect RFID tags. The readers are placed in known positions dividing the region into sub-regions. As they travel in these sub-regions, the tags can be associated with the sub-regions. The accuracy of the system depends on the number of these readers and their placements. LANDMARC increased the accuracy without placing more readers by employing extra fixed location reference tags for location calibration. These reference tags become the reference points in the system. Another indoor positioning system is called SPOT ON [8]. It is a new tagging technology for 3D location detection based on RSS. An embedded hardware system was developed named Hydra Micro server. Hydra has both Ethernet and RS232 port and it is used for internetworking task .The interconnecting mixture of multiple base stations are used in the system to provide RSS measurements. These measurements are sent through RS232 port via internet and stored in the server. The server processor maps the RSS values and uses triangulation technique [30] to determine the precise position of the object.

Once the position is known, a virtual 3D display of the indoor environment was constructed to show the locations of the tags on the objects. This application was built with OpenGL. RADAR is another RF based popular system used for locating and tracking objects or people in indoors [11]. The system records and processes the signal strength information received from base stations. These stations are positioned to provide overlapping coverage of the area of interest. It uses signal propagation models to estimate the object location with a great accuracy. Signal strength information collected at multiple receiver locations were triangulated to find the user coordinates. This triangulation was made using empirical approaches and computations of signal strength information. RADAR estimated the user’s location within a few meters of his/her actual location.

E. Computer vision systems

There are visual systems which track the people with multi-cameras [18] and generate an intelligent environment. The system uses two sets of stereo color cameras to track multi persons in living rooms. Stereo images are used to locate the people and color images are used to maintain their identities. The system claims a location measurement accuracy of around 10 cm. A disadvantage of the system is that ıt uses multiple cameras to cover all the corners and occlusions of indoors. Hence it is expensive. Another vision based localization system by two cameras is given in [19]. New feature initialization and feature matching techniques are used with two cameras to locate people. Experimental results show that the 3D positioning of the objects is more accurate than single camera cases. The work was aimed for the development of intelligent robots to increase their ability to recognize their environment and their position.

F. Cell phone positioning system

(E- 911) Enhanced 911 is a North America based system which links the telephone calls with the callers' positions. The caller's address and information is displayed for the call taker upon the arrival of the call. E-911 technology is used by wireless telephones. It will allow emergency dispatch centers to process emergency 911 calls and provide the number and geographic information for public safety service providers. Location of the wireless users are found by location pattern matching (LPM) and time difference of arrival (TDOA). When a person makes a 911 call using a telephone via landlines, the service provider routes the call to the nearest public safety answering point (PSAP), which then distributes the emergency call to the appropriate service(s) with the geographical information.

G. Integrated positioning systems

Recently the positional accuracy was greatly improved by using integrated systems. For example, inertial technologies [13], where low cost gyros are used for position determination and error correction, were included in positioning systems. Inertial navigation system (INS) and RFID positioning methods have been used together to calculate the position of the objects. User's position information is obtained by using RFID technique and RSS measurements. This position information is integrated with the information obtained from INS [30]. Integration of RFID and INS improves position determination since the inertial sensors are not affected by the signal propagation limitations such as obstructions and multipath. This system is collect more RSS information so it can produce a higher resolution of positioning. Hence a probabilistic method was used to determine the tag positions. Another technique called finger printing method [7] (FPM) is integrated with RFID technique. Best positioning was achieved in two stages. In first stage, identified as off–line stage. RSS values and physical coordinates are collected from RF transmitters at a reference point and stored in a database which is called fingerprint. In second stage, identified as On-line stage, the mobile user samples the RSS pattern and searches for similar pattern in off-line database to find the best possible position [32]. The accuracy with this method is around 5m. It calculates the user position according to conditional probabilities of the location under certain RSS. The technique is quite accurate to deal with instabilities of RSS. Finally, INS/RFID and the finger printing RFID methods were integrated to improve the position accuracy further. Kalman-Filtering algorithm was used [27] to integrate these two systems. The accuracy recorded was under 1m.

If the accuracy and the cost are the most important parameters, then the systems using ultrasonic techniques are the desirable systems according to what we discussed. Ultrasonic and infrared

techniques used TOA (time of arrival) techniques while the techniques using RF signals employed RSS and triangulation/lateration. All the systems were real time systems and the position information was produced in real time. The accuracies of the systems varied between 2 cm and a few meters.

Finally, for the years to come in the near future, we are forecasting the following trends:

I. Indoor/Outdoor IPS. Outdoor positioning systems will merge with Indoor Positioning System (IPS) in a seamless way to locate a person with a smartphone anywhere. This means that while current IPS systems involve specialized equipment and applications, future IPS systems will be part of the smartphone operating system and leverage its sensors so any location-sensitive smartphone application will use indoor or outdoor location services as they are available.

II. Consideration of Privacy and Security Issues in the Development of IPS. From the analysis of IPS, we noticed that the privacy and security issues regarding the user's location are only addressed in very few projects. Nevertheless, some authors provide evidence that these factors may influence the adoption and use of the IPS or argue that the system must give the users the possibility of deciding whether they want to share their locations with others. Though privacy has been a concern since the very beginning of the development of IPS systems, in the future, this will become one of the main considerations for the adoption or choice of specific IPS systems.

III. SYSTEM MTHODOLOGY AND CHALLENGES

The mentioned approaches vary in many aspects, such as the positioning method, signal type, cost of infrastructure, power requirement, and resolution in time and space. As we mentioned we prefer ranging-based method because it has low complexity and high scalability. Two common ranging techniques are the Time-of-Arrival (TOA) and received signal-strength (RSS) estimation. Due to low accuracy of RSS approach it needs the radio attenuation model as a prior on the other hand TOA estimation scheme is often preferred in systems with high-accuracy requirement. Our goal is to design a system with high accuracy therefore our preferred method is TOA. The ranging accuracy directly depends on the bandwidth and transmission speed of the operating signal. Two types of signals are suitable for the ranging purpose,

(1) Impulse-Radio Ultra-wideband (IR-UWB) signal with its sharp pulse and wideband properties.

(2) Ultrasound or acoustic signal because of its lower transmission speed.

UWB devices are very expensive and not available in consumer market. Therefore, we use acoustic signals for localization, and in order to deal with the limited bandwidth and strong attenuation drawback we can using radio signals for assistance to compensate the mentioned drawbacks.

High accuracy ranging is typically achieved through measuring time-of-arrival (TOA) information of acoustic or radio signals. The distance is thus the product of the signal speed and the time of flight of the signal traveling between two devices. Obviously, the ranging accuracy depends on the signal speed and the precision of TOA measurement. To elevate the accuracy,

acoustic signals are chosen because of their relative slow speed. But the precision of TOA measurement remains a big challenge in any system implementation. In practice, TOA measurement is often done with both sides taking a timestamp of their respective local clock at the moment the signal is emitted or received. There are three intrinsic uncertainties in this process that can contribute to the ranging inaccuracy: the possible clock skew and drift between devices, the possible misalignment between the sender timestamp and the actual signal emission, and the possible delay of a sound signal arrival being recognized at receiver. In general, many factors can cause the latter two uncertainties in a real system, such as the lack of real-time control, software delay, interrupt handling delay, system loads, etc. These uncertainties, if not controlled, can seriously affect the ranging accuracy. For example, our tests on two COTS mobile devices reveal that these two delays can easily add up to several milliseconds on average, which translates to several meters of ranging error. It is therefore extremely challenging to provide high accuracy ranging in a software-only and device-only solution using only the minimum commodity hardware set we specified earlier. For the solution to be applicable to COTS mobile devices, there are additional constraints. We cannot assume that we have a real-time operating system or be able to change kernel or driver. In fact, many COTS devices like cell phones are built on closed platforms and many often have operator imposed locks that prevent changing OS. We will have to implement the entire ranging system in user-space.

The granularity of our TOA measurement is limited only to the sound sampling rate. Under today's prevailing hardware standard of 44.1 KHz, our mechanism can have a ranging accuracy of 8cm. As far as we know, 40 meter is the best ever achieved using only commodity hardware (speaker and microphone) on COTS mobile devices. Time-of-arrival based systems estimate the distance D between the sender and the receiver to be the product of the time-of-flight, i.e., the

time (Δt) it takes a signal such as sound, radio wave, or light to reach the receiver, and the propagation speed c of the signal, which is usually assumed to be a constant known a priori.

$$D = c \cdot \Delta t \tag{1}$$

Given the requirement on the desired precision, acoustic signal is usually chosen because the speed of radio or light signal is so fast that a small timing error would lead to an unacceptably large ranging error. But even if the relatively slower acoustic signal is chosen, the precision requirement on TOA estimation is still very stringent. For example, one millisecond error in TOA estimation will translate to more than 30 centimeters error in the ranging result. TOA measurement is done with both sides taking a timestamp of their respective local clock at the moment the signal is emitted or received. There are several intrinsic uncertainties in this process that will contribute to the TOA measurement error. The first one is clock synchronization uncertainty (μc): the possible clock skew and rifting between the two devices. To address this problem, many solutions have been proposed in the literature. Some rely on GPS for time synchronization and some others chose to work around by using round-trip time measurement (assuming symmetric propagation path) so that all time readings refer to the same clock. Yet most solutions have resorted to dedicated mechanisms. The second uncertainties is the sending uncertainty (μs): the possible misalignment between the timestamp and the actual signal emission. For example, there is often a small yet arbitrary delay after an output command is issued till the sound actually comes out from the speaker. Similarly, the third uncertainty is the receiving one (μr): the possible delay of a sound signal arrival being recognized. In general, many factors can cause these two uncertainties in a real system, such as the lack of real-time control, software delay, interrupt handling delay, system loads, etc. There has been little work in addressing the sending and receiving uncertainties in software. Most previous work managed to

minimize them by resorting to customized hardware design so that the system can precisely control and obtain the exact instant when a signal is sent or received [3, 4]. This is clearly inapplicable if we desire a software solution and only use commodity hardware.

The results indicate that μs+μr appears to be very random and affected heavily by the CPU load [9]. Both the average and the standard deviation increases when the load becomes heavy, such as playing a video, even if we give the test program the highest priority. In any case, this study shows that the uncertainties easily add up to several milliseconds and translate to tens of centimeters of ranging error when the TOA measurement is done in software. In a typical computer system, obtaining the exact time instance when the signal arrives is difficult due to the random latency introduced by hardware and software (receiving uncertainty). In our design, this issue is resolved by using the Beep-beep ranging technique.

Beep-beep ranging mechanism achieves high accuracy through a combination of three techniques: two-way sensing, self-recording, and sample counting. The basic idea is the following. To estimate the range between two devices, each will emit a specially-designed sound signal ("Beep") and collect a simultaneous recording from its microphone. Each recording should contain two such beeps, one from its own speaker and the other from its peer. By counting the number of samples between these two beeps and exchanging the time duration information with its peer, each device can derive the two-way time of flight of the beeps at the granularity of sound sampling rate. This technique cleverly avoids many sources of inaccuracy found in other typical time-of-arrival schemes, such as clock synchronization, non-real-time handling, software delays, etc.

V. SYSTEM ARCHITECHTURE AND TRANSCEICER DESIGN

A. Direct sequence spread spectrum

To differentiate the audio signals sent by different phones, direct sequence spread spectrum (DSSS) is adopted to encrypt audio signals.

Direct sequence spread spectrum, is one of the approaches to spread spectrum modulation for digital signal transmission which are used for reducing overall signal interference over the airwaves. In direct sequence spread spectrum, the stream of information to be transmitted is divided into small pieces, each of which is allocated across to a frequency channel across the spectrum. A data signal at the point of transmission is combined with a higher data-rate bit sequence (also known as a chipping code) that divides the data according to a spreading ratio. The redundant chipping code helps the signal resist interference and also enables the original data to be recovered if data bits are damaged during transmission. Direct-sequence spread-spectrum transmissions multiply the data being transmitted by a "noise" signal. This noise signal is a pseudorandom sequence of 1 and −1 values; at a frequency much higher than that of the original signal. However, this noise-like signal is used to exactly reconstruct the original data at the receiving end, by multiplying it by the same pseudorandom sequence (because $1 \times 1 = 1$, and $-1 \times -1 = 1$).

There are many benefits to spread-spectrum technology. Resistance to interference is the most important advantage. Intentional or unintentional interference and jamming signals are rejected because they do not contain the spread-spectrum key. Only the desired signal, which has the key, will be seen at the receiver when the dispreading operation is exercised. We can practically

ignore the interference, narrowband or wideband, if it does not include the key used in the dispreading operation. That rejection also applies to other spread-spectrum signals that do not have the right key. Resistance to interception is the second advantage provided by spread-spectrum techniques. Because non-authorized listeners do not have the key used to spread the original signal, those listeners cannot decode it. Without the right key, the spread-spectrum signal appears as noise or as interference. Resistance to fading (Multipath Effects) is the other advantage of spread-spectrum technique. Wireless channels often include multiple-path propagation in which the signal has more than one path from the transmitter to the receiver. Such multipath can be caused by atmospheric reflection or refraction, and by reflection from the ground or from objects such as buildings. The reflected path can interfere with the direct path in a phenomenon called fading. Because the dispreading process synchronizes to main signal, the reflected signal is rejected even though it contains the same key.

In order to use DSSS the sender system generates a 5-digit random seed and then generates pseudorandom +1 and -1 sequence using the 5-digit random seed and plays it through speaker. The length of the pseudorandom +1 and -1 is fixed.

The audio signals are played and recorded through the speaker and microphone using Port Audio APIs. Port-Audio provides a very simple API for recording and playing sound using a simple callback function or a blocking read or write interface.

B. User datagram protocol (UDP)

Since the pseudo random code needs to be known by both phones, it is sent through Wi-Fi just before audio signal is sent so the receiver has the key. The matched filter is adopted to decrypt

audio signals and estimate the time when the smartphone receives the desired audio signal. We use Socket to send and receive signals through Wi-Fi. The user datagram protocol (UDP) is used to broadcast packets in the Internet.

UDP is suitable for purposes where error checking and correction are either not necessary or are performed in the application. Time-sensitive applications often use UDP because dropping packets is preferable to waiting for delayed packets, which may not be an option in a real-time system. The user datagram protocol (UDP) is used to broadcast packets in the Internet. UDP provides two services not provided by the IP layer. It provides port numbers to help distinguish different user requests and, optionally, a checksum capability to verify that the data arrived intact.

C. Signal design

Our goal of not using any external hardware in designing this system makes accurate measuring possible only by choosing an appropriate transmitter acoustic signal band which match the capabilities of a user's smartphone. The acoustic signal band of microphones is very limited. The typical band of microphone is in the audible range, 200Hz-20 KHz. Since these microphones are designed to record the speech voice, to reduce the interference of voice signals and daily environment noise the output signal of the microphones usually are filtered using the bandpass filter with the passing range of 200 Hz to 2 KHz. From Figure 1, we can see that the usable voice frequency band ranges from approximately 250 Hz to 2KHz. Based on the output frequency band of the phone's microphone and also our experiment we decided to play our audio sequence in the frequency of 1.2 KHz.

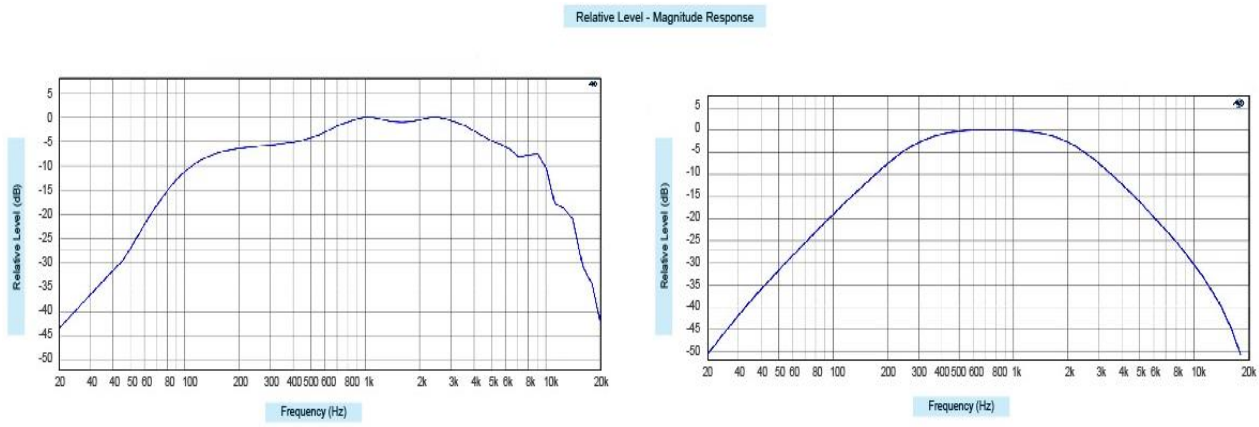

Figure 1. Frequency responses of two android phone microphones

VI. SYSTEM IMPLEMENTATION

The main purpose of this acoustic ranging system is to estimate the traveling time of audio signal from one phone to the other. If we obtain the traveling time by calculating the difference between two time stamps, one is recorded when the acoustic signal is sent and the other is recorded when the acoustic signal is received, then the clocks of the two computers are required to be synchronized. To avoid clock synchronization, we use beep-beep ranging mechanism to calculate the traveling time of the acoustic signal. In this project, we call the phone which first send wi-fi and acoustic signals as client and call the other phone as server. Fig. 1 illustrates the way we implement the beep-beep ranging mechanism in this project.

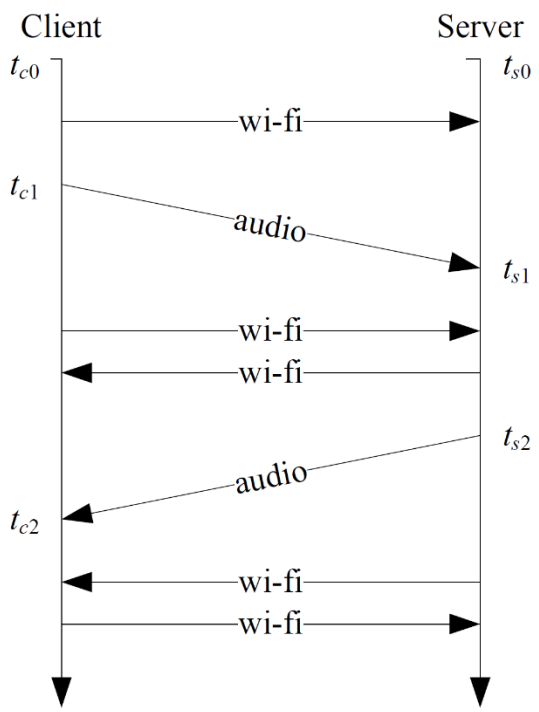

Figure2. Implementation of Beep-beep ranging mechanism

We use Socket to send and receive signals through Wi-Fi. The user datagram protocol (UDP) is used to broadcast packets in the Internet. The audio signals are played and recorded using Port-Audio APIs. The beep-beep ranging is implemented as follows.

First, the client generates one 5 digits random seed and sends it to server through Wi-Fi. Second, the client generates pseudorandom +1 and -1 sequence using the 5 digits random seed and plays it through speaker. The length of the pseudorandom +1 and -1 sequence is fixed. The client records the time when it sends audio signal as t_{c1} and sends t_{c1} to server. The server receives audio signal through microphone, generates pseudorandom +1 and -1 sequence using the 5 digits random seed, and estimates the receiving time ts1 using matched filter. Third, server sends its time stamp t_{s1} to client through Wi-Fi. Fourth, the server uses the first 4 digits of the previous random seed as a new random seed to generate a new pseudorandom +1 and -1 sequence. Then, the server plays this new rectangular wave through speaker, records the time as t_{s2}, and sends it to client through Wi-Fi. The client receives the new audio signal, generates pseudorandom +1 and -1 sequence using the 4 digits random seed, and estimates the receiving time t_{c2} using matched filter. Finally, the client sends its time stamp t_{c2} to the server through Wi-Fi. Now, both client and server collect all the time stamps. Since client and server will finish the beep-beep ranging within a short time, we assume the distance between client and server is fixed within this time. Denote the traveling time of audio signal from one user to the other by Γ. Assume the time difference between the clock of server and the clock of client is Δt. Then, t_{s1} and t_{c1} have the following relationship

$$t_{s1} = t_{c1} + \Gamma + \Delta t \tag{2}$$

t_{c2} and t_{s2} have the following relationship

$$t_{c2} = t_{s2} + \Gamma - \Delta t \tag{3}$$

From (1) and (2), we can calculate the traveling time with the time stamps as follows

$$\Gamma = \frac{(ts1 - tc1) + (tc2 - ts2\ )}{2} \tag{4}$$

A. The Protocol Between Client and Server

Since we need to implement multiple tasks at the same time, multithreading is used to realize them. If a thread gets a lot of cache misses, the other threads can continue taking advantage of the unused computing resources, which may lead to faster overall execution as these resources would have been idle if only a single thread were executed. Also, if a thread cannot use all the computing resources of the CPU (because instructions depend on each other's result), running another thread may prevent those resources from becoming idle.

Since the delays of processing is important to us (1ms delay will cause 34 cm of error) multithreading can help us avoid extra processing delays. If several threads work on the same set of data, they can actually share their cache, leading to better cache usage or synchronization on its values.

TABLE I

THREADS OF CLIENT AND SERVER

	Client	Server
Thread 1	send/receive wi-fi signal	receive/send wi-fi signal
Thread 2	record audio	record audio
Thread 3	play audio	play audio
Thread 4	matched filter	matched filter

Table1. Threads of client and server

B. Estimate of Recorded Audio Signal Beginning Time

Note that the matched filter is used to estimate the time when audio signal is received. In this procedure, we need to know when each computer begins to record audio signals which is crucial for the accuracy of estimating traveling time. Since the exact beginning recording time cannot be obtained directly, we need to estimate it. Since each computer is designed to estimate the beginning recording time in the same way, we take client for example here.

The client estimate the beginning recording time by sending an audio signal to itself. When the client sends the audio signal, it records the sending time too. Since the speaker and microphone of client are close to each other, we assume the client receives the audio signal as soon as it plays it. Hence, the client knows the receiving time which is also t_0. The number of samples between the first sample of the audio signal and the first recorded sample can be estimated via matched filter. Let it be N. Since both the sample rate of audio signal and that of recording are 44100, the

time difference between receiving audio signal and beginning to record is *N/44100*. Therefore, the estimated beginning recording time of client is

$$tc0 = t0 - \frac{N}{44100} \tag{5}$$

This idea is realized in the following method. The client implements thread 2 (record audio) from the first beginning and this thread will not be stopped until the end of beep-beep ranging. The thread 3 (play audio) waits until thread 2 begins. In thread 3, a pseudorandom +1 and -1 sequence is generated, saved in a file, and sent using the speaker.. The thread 4 (matched filter) waits until thread 3 ends. In thread 4, the client estimates *N* by using the audio recorded in thread 2 and pseudorandom sequence saved in thread 3. Then, the client calculates t_{c0}, which is used in beep-beep ranging.

C. Beep-beep Ranging Implementation

After both client and server obtain their beginning recording time t_{c0} and t_{s0} respectively, the beep-beep ranging is implemented. Note that, for both client and server, the thread 2 is started in the first beginning and keeps being implemented until the end of beep-beep ranging. For both client and server, thread 1 is also implemented all the time.

In the client end, the beep-beep ranging is implemented in the following steps. After t_{c0} is obtained, the client first broadcasts 5 digits random seed in thread 1. As soon as the 5 digits random seed is broadcasted, the client implements thread 3 to play audio and obtain t_{c1}. Then, the client implements thread 1 to broadcasts t_{c1} and receive t_{s1} and t_{s2} from the server. As long as t_{s2} is received, the client knows the second audio signal is also received. Then, the client

implements thread 4 to obtain t_{c2}, in which the 4-digit random seed is used. Finally, the t_{c2} is broadcasted in thread 1.

In the server end, the beep-beep ranging is implemented as follows. After t_{s0} is obtained, the server first implement thread 1 to receive 5 digits random seed and ts1. As long as the ts1 is received, the server obtains ts1 in thread 4 and broadcasts ts1 in thread 1. Then, the server implements thread 3 to play the second audio and obtain t_{s2}. As soon as the audio is finished, the server implements thread 1 to broadcast t_{s2} and receive t_{c2} . To run the beep-beep ranging multiple times automatically, the server needs to send a "ready" sign to the client each time when the beep-beep ranging is finished.

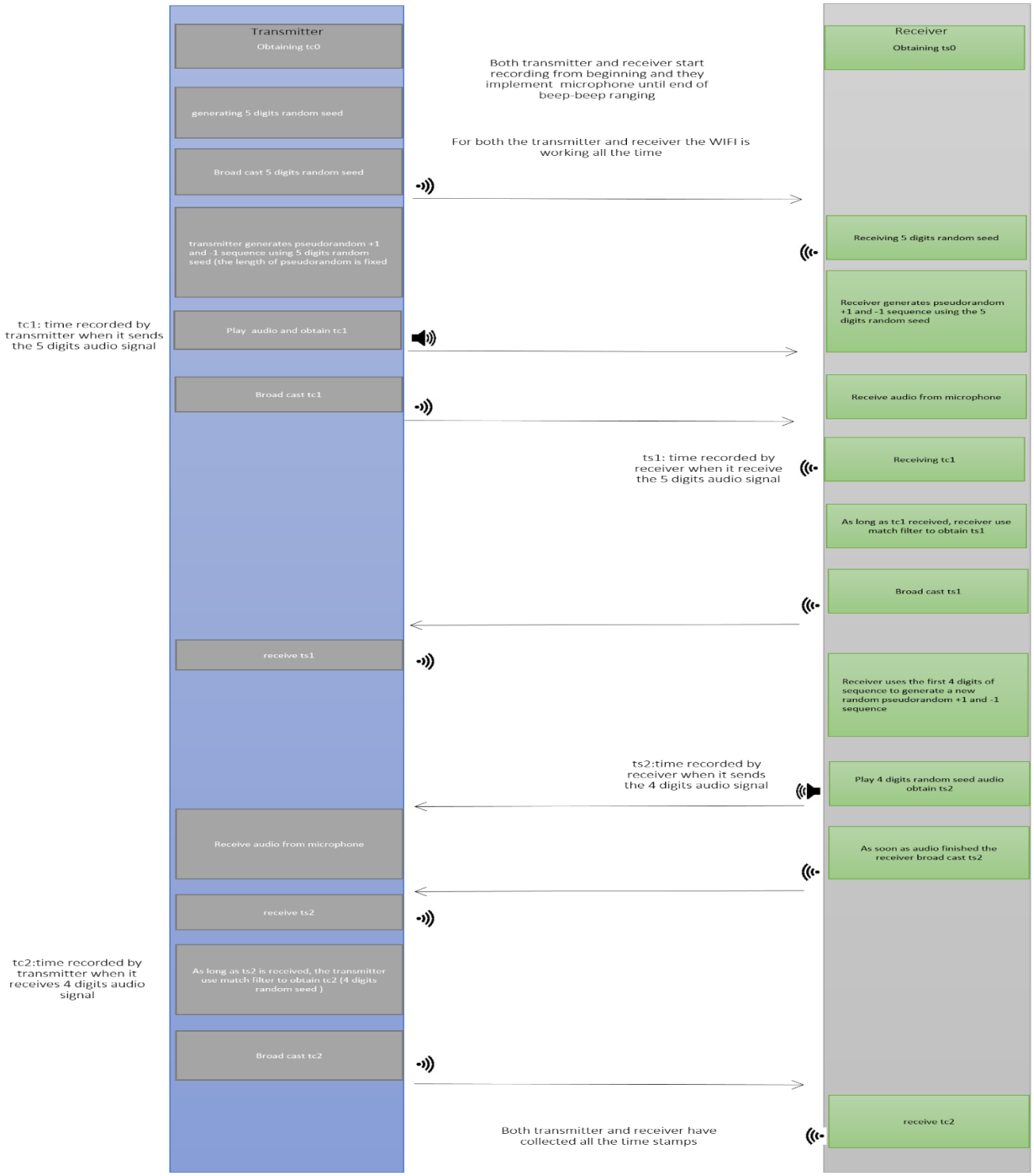

Figure 3. Implementation of Beep-beep ranging

D. *Results by Using a PC*

To evaluate the system performance of ranging, in the first step we implemented the algorithm in two Mac computers and we conducted the experiments in a typical office environment showed in Figure 4 to test the ranging accuracy. The distance between two computers is changed from 0.5m to 6m. At each distance, the beep-beep ranging is run 100 times.

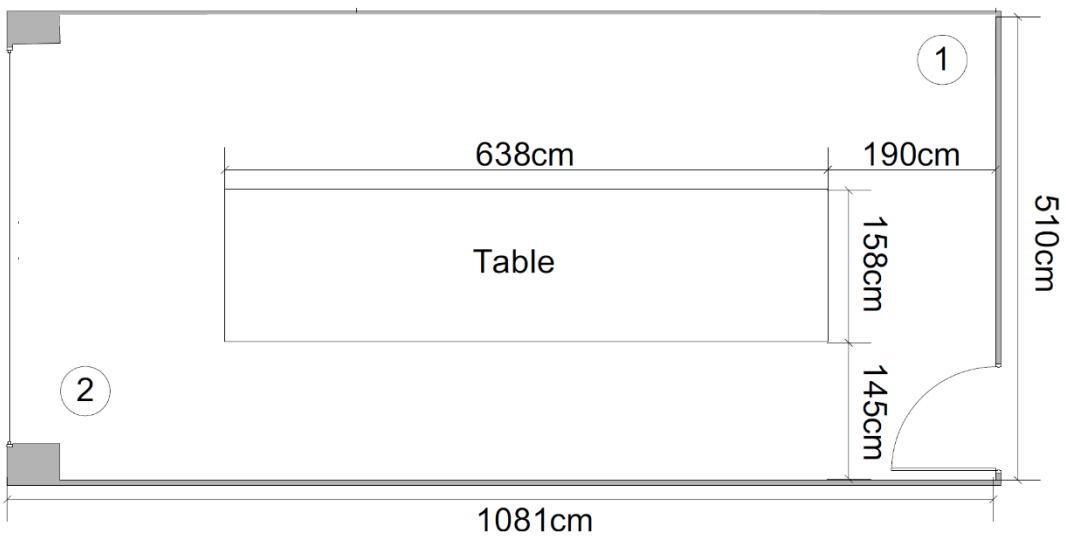

Figure 4. Experiment environment plan

The mean of estimated distance is shown in Figure 5. The mean of estimation error in distance is shown in Figure 6. From Figure 6, we know that the accuracy of this method is between 1m and 4.5m. The accuracy is varying when the true distance between two computers is changed. One reason is that the reverberation scenario is changed when the computers are moved. The standard

deviation of estimated distance is shown in Figure 7. Obviously, the standard deviation is smaller than 1.1m at all distances. That means the results are stable and reliable.

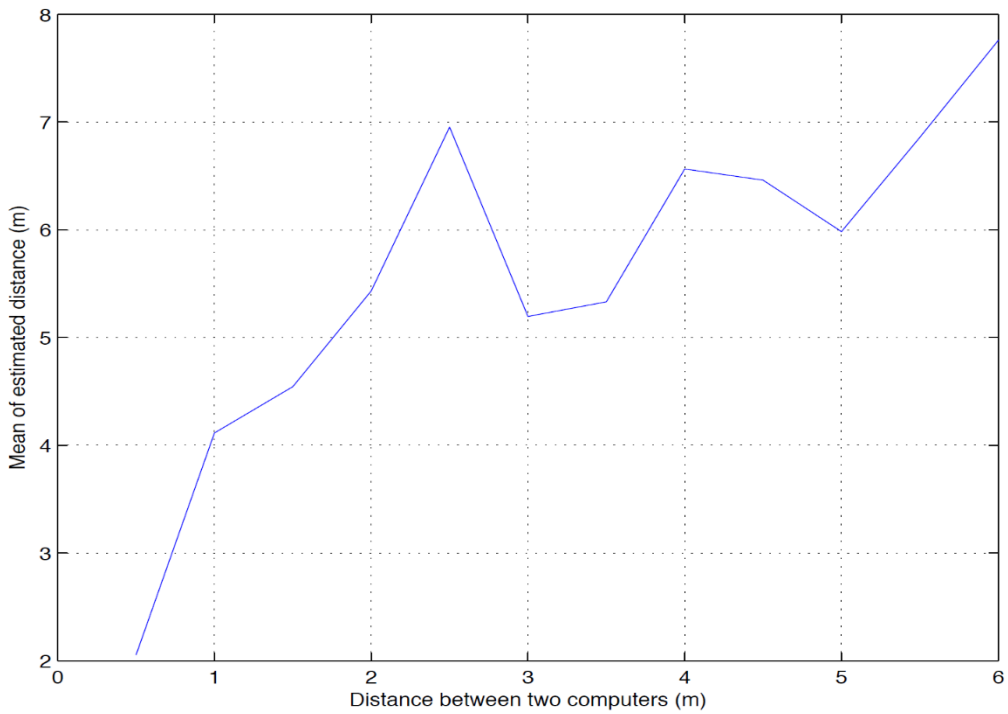

Figure 5. Mean of estimated distance

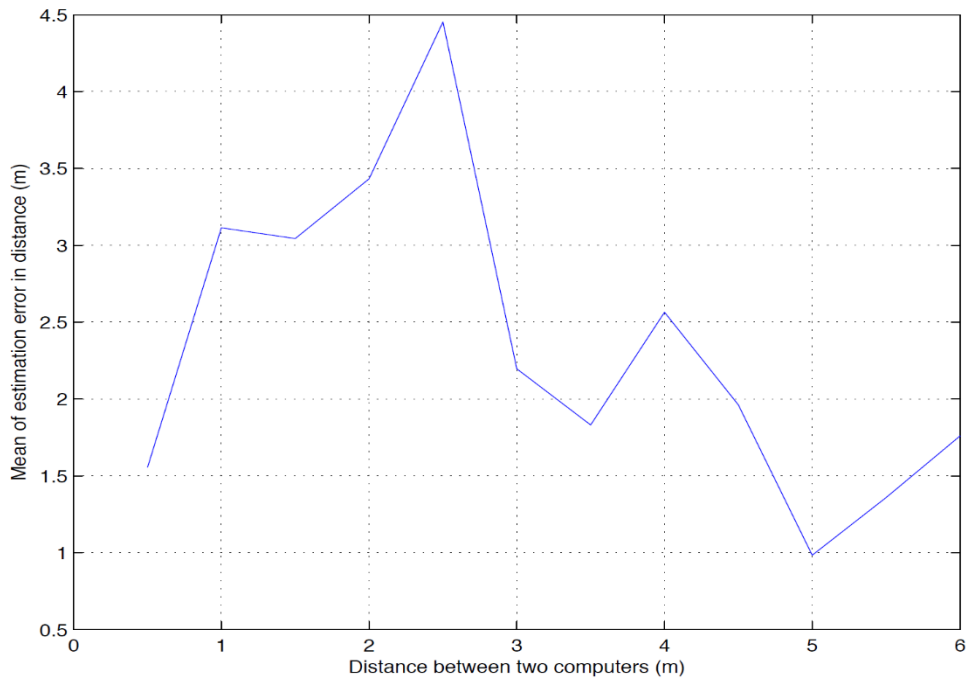

Figure 6. Mean of estimation error in distance

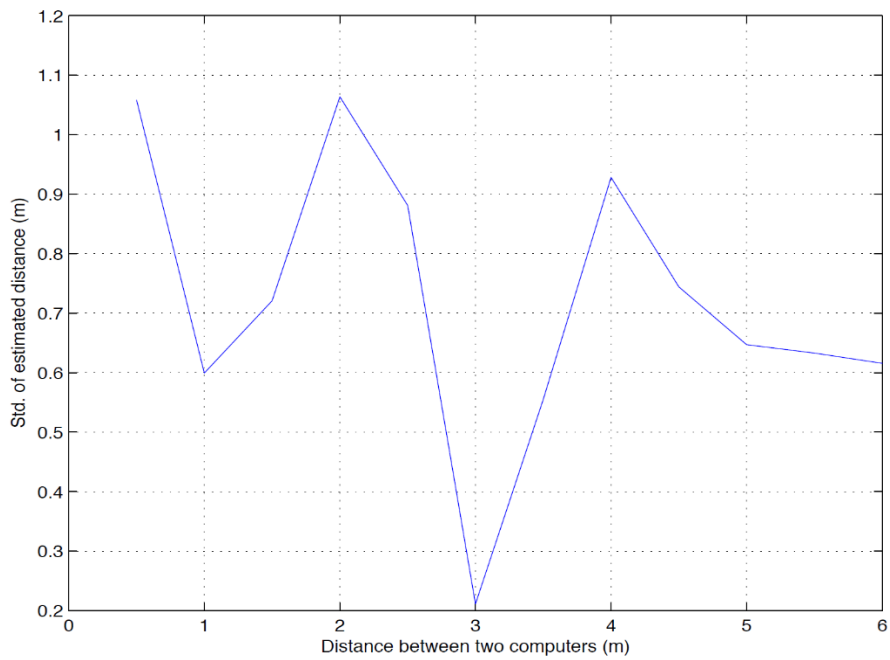

Figure 7. Standard deviation of estimated distance error

E. Results by Using Android Components

Using the Android Studio, our model has been implemented in Android phones and run as a ranging application, but the result that we got from two Android phones we were using was not satisfying. The problem was with the task multithreading in android based components. Unfortunately Android has a very big thread scheduling problem while using multithreading. Android multithreading problem will cause a random delay in our application. This random delay completely depends on the version of the Android OS installed in the component and also the advancement of the component hardware.

Multiple threads can interfere with each other when sharing hardware resources such as caches or translation lookaside buffers (TLBs). As a result, execution times of a single thread are not improved but can be degraded, even when only one thread is executing, due to lower frequencies or additional pipeline stages that are necessary to accommodate thread-switching hardware. This might not be a problem for most of the android applications as it cause just a very short delay when executing different tasks of an app, but in our application this random delay will cause a very large range error. As an example it can cause a random delay up to 150 millisecond while running an app in Galaxy Nexus (with Android 4.3). For comparison, the average reaction time of a human to an audio stimulus is around 170 millisecond. But in our application this much delay may cause an error with the range of 50 meters. This problem becomes even worse when there is another application running in the background. As an example, we have tried playing a

video on the background while using our ranging application in the Galaxy Nexus, and the delay became up to 190 milliseconds.

By using more powerful and advanced Android phones, we achieved a better result. This time we implemented the algorithm in two Samsung Galaxy S7 phone (Android 7.0) and we conducted the experiments in a typical office environment showed in Figure 8 to test the ranging accuracy. The distance between two phones is changed from 0.5m to 15m. At each distance, the algorithm is run 100 times.

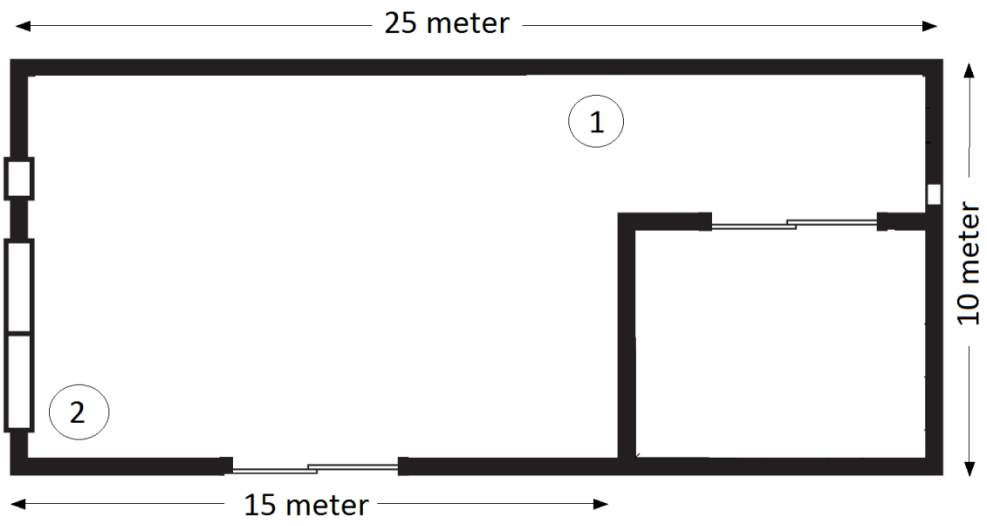

Figure 8. Experiment environment plan

The mean of estimated distance is shown in Figure 9. The mean of estimation error in distance is shown in Figure 10. From Figure 10, we know that the accuracy of this method is between 0.5 m and 5.3 m. The accuracy is varying when the true distance between two phones is changed. One reason is that the reverberation scenario is changed when the phones are moved. The standard

deviation of estimated distance is shown in Figure 11. Also, the standard deviation is smaller than 1.8m at all distances which shows stable and reliable results.

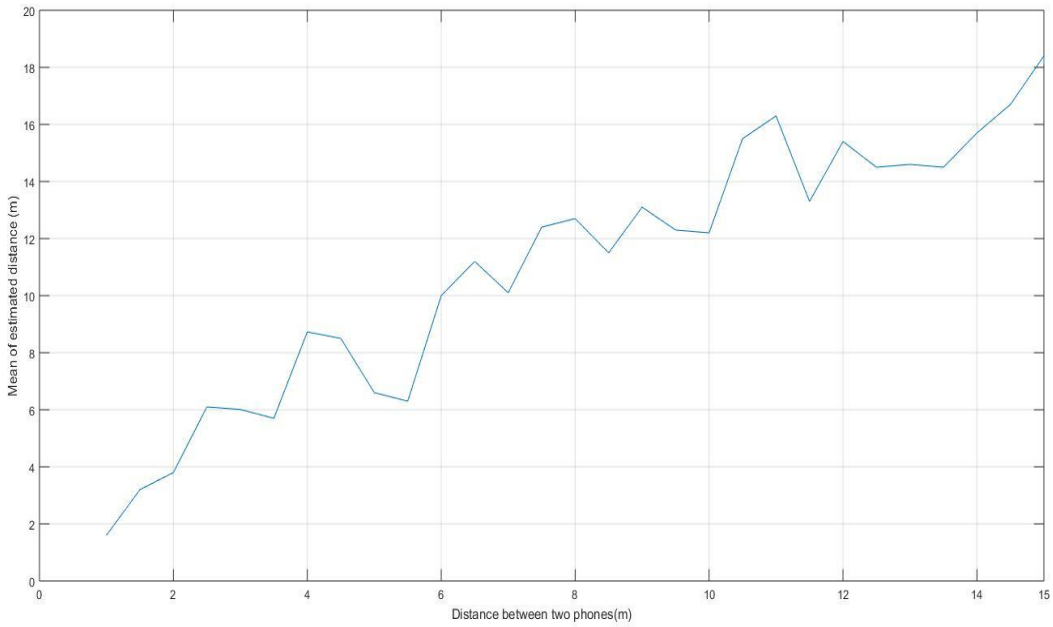

Figure 9. Means of estimated distance

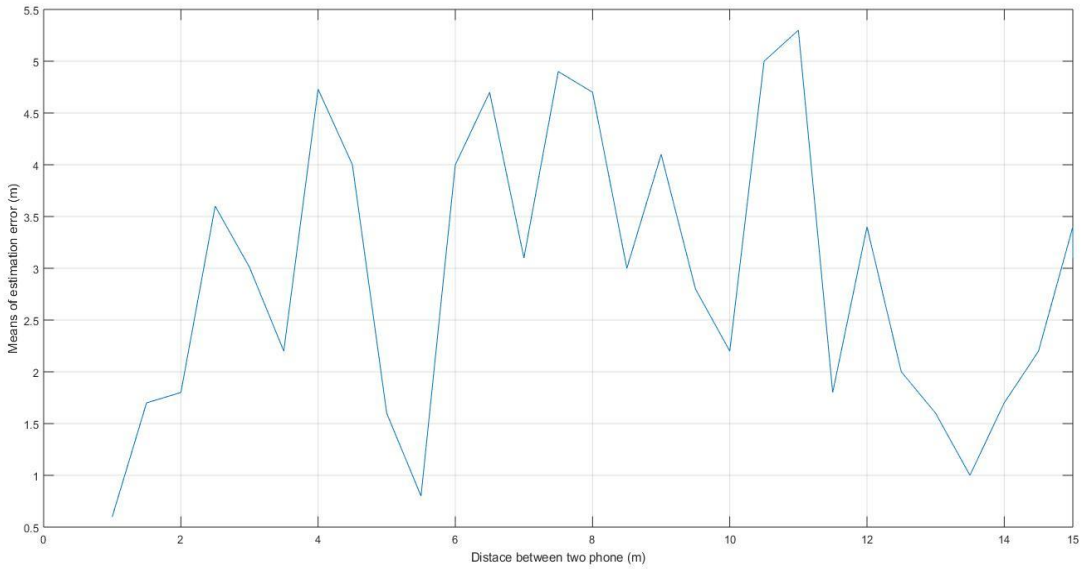

Figure 10. Mean of estimation error in distance

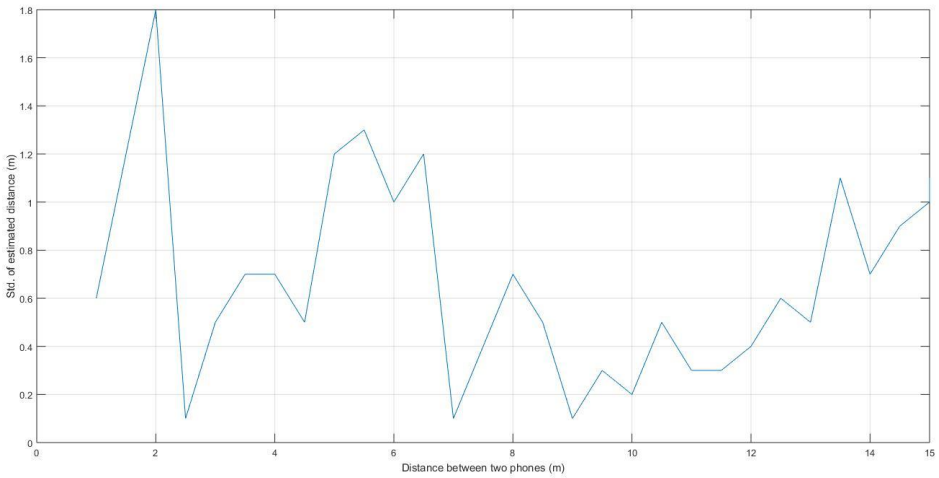

Figure 11. Standard deviation of estimated distance

As we said Android multithreading problem will bring a random delay in our application, which may cause up to several meters of error. However, our application can achieve a better result if we use one of the new versions of Android based components and prevent any other application to be run in the background. But our aim was to develop a system which is widely applicable in many low-cost sensor platforms and to most commercial-off-the-shelf mobile.

In the newer version of our algorithm we tried to simplify the program and avoid the multithreading.

F. Client and Server communication protocol

We use Socket to send and receive signals through Wi-Fi. The user datagram protocol (UDP) is used to broadcast packets in the Internet. The audio signals are played and recorded using Port Audio APIs. The new algorithm is implemented as follows. First the client will make a 5 digit pseudorandom number and send it to the server through Wi-Fi. After receiving pseudorandom number server will sent a ready message to the client. As soon as the client receives the ready massage, it generates pseudorandom +1 and -1 sequence using the 5 digits random seed and plays it through speaker. The length of the pseudorandom +1 and -1 sequence is fixed. When the client start playing the sound it will record the time stamp as T_s and when it finished playing that it will record that time stamp as T_e. Immediately after sending the ready message the server will start recording for a period of time (for example 5 seconds) and it will save what it has recorded in a file or variable. The server will send another message “done” to client and wait for collecting the starting T_s and ending T_e time stamps from client so the client will wait to receive the “done” massage form server side and then it will send T_s and T_e to the server through Wi-Fi. As soon as

the server receives these time stamps it will implement the matched filter to find the R_s and R_e which is "starting time stamp of the sound at server side" and "ending time stamp of the sound at server side".

The duration of sound in client end is known and constant D_s. If we subtract T_s from T_e,

$$T_e - T_s = A \tag{6}$$

The difference of A and playing duration D_s is client system delay C_d.

$$A - D_s = C_d \tag{7}$$

The difference of $R_s - R_e = B$ and playing duration D_s will be server delay S_d as well.

$$B - D_s = S_d \tag{8}$$

We can calculate the traveling time Γ only with the time stamps as follows

$$\Gamma = [(R_s - T_s) - (C_d + S_d)] \tag{9}$$

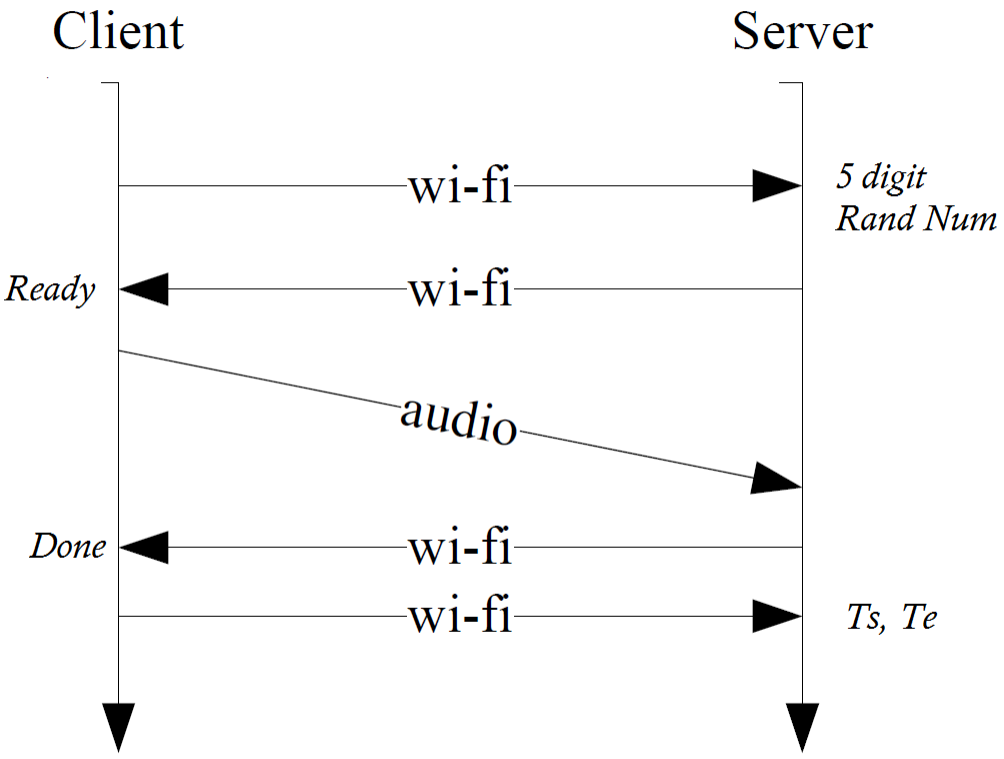

Figure 12. Client and server communication protocol

At the client end: the client will make a 5 digit pseudorandom number and send it to the server through Wi-Fi. Then it will wait for receiving a ready message from server. As soon as it receives the ready massage the client will start playing a sound based on this pseudorandom number for a constant duration. When the client starts playing the sound it will record the time stamp as T_s and when it finishes playing that it will record that time stamp as T_e. After that it will wait to receive the "done" massage form server side and then it will send T_s and T_e to the server.

At the server end: server will wait for the 5-digit pseudorandom number from the client side. As soon as it receives it, it will sent a ready message to the client , immediately after sending the

ready message the server will start recording for a fixed period of time and it will save what it has recorded in a file. The server will send another message "done" to client and wait for receiving the starting T_s and ending T_e time stamps from client. As soon as it receives these time stamps it will implement a matched filter to the recorded file to find R_s and R_e.

In order to achieve a more stable and reliable estimation we added a few steps to our model. Instead of sending just one audio package form client to server we decided to send several packages and by getting average of the most similar consecutive values, we will use the average of those values as the delay for time traveling period, and then we use the corresponding acoustic packet to estimate the range. We changed the model as the follow.

Client sends acoustic packet 1, time stamped T_{s1}; sends the acoustic packet 2 right after sending packet 1, time stamped T_{s2}. Server receives acoustic packet 1, time stamped T_{r1}; receives acoustic packet 2, time stamped T_{r2}. Then will have the sending interval as follow

$$T_{s21} = T_{s2} - T_{s1} \quad \textit{sending interval} \tag{10}$$

$$T_{r21} = T_{r2} - T_{r1} \quad \textit{receiving interval} \tag{11}$$

The delay caused by sender and receiver system processing can be estimated as

$$T_{D21} = T_{r21} - T_{s21} \tag{12}$$

Similarly, we can send packet 3 right after sending packet 2, and get

$$T_{D32} = T_{r32} - T_{s32} \tag{13}$$

Suppose we send 10 packets, then we can get T_{21}, T_{32} ... T_{109}, we find the most similar two consecutive values, use the average of these two as the delay for that period, and then we use the corresponding acoustic packets to estimate the range.

G. Implementation and results of new algorithm

In order to apply our new model we used a Galaxy Nexus phone (Android 4.3) and a Nexus 5 phone (Android 4.1). Experiment is carried out in one meeting room which is shown in Figure 8 So far we are sending 20 packages right after each other and recording the 20 delay times. The application run 50 times. There are no objects between two phones in this experiment.

The mean of estimated distance is shown in Figure 13. The mean of estimation error in distance is shown in Figure 14. From Figure 14, we know that the accuracy of this method is between 0.25 m and 4.8 m. The accuracy is varying when the true distance between two phones is changed. One reason is that the reverberation scenario is changed when the phones are moved. The standard deviation of estimated distance is shown in Figure 15. Obviously, the standard deviation is smaller than 1.8m at all distances. That means the results are stable and reliable.

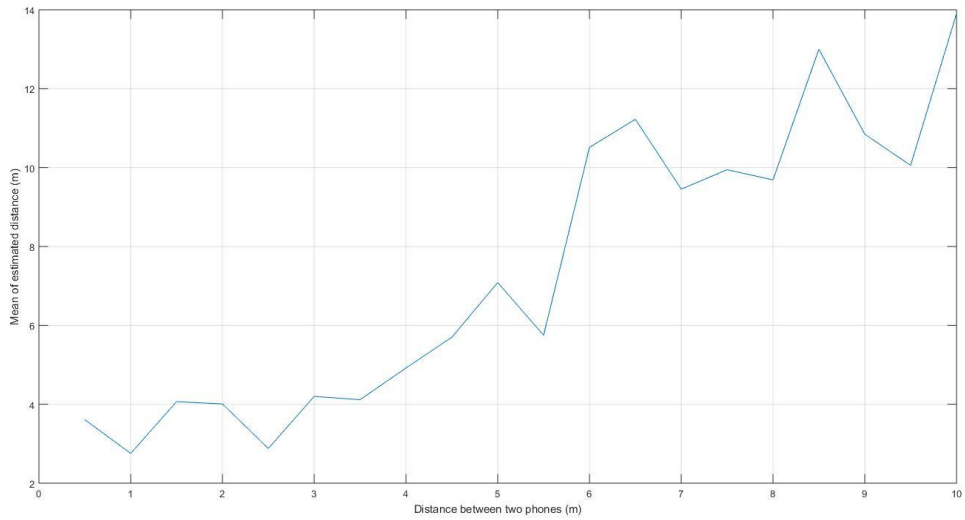

Figure 13. Mean of estimated distance

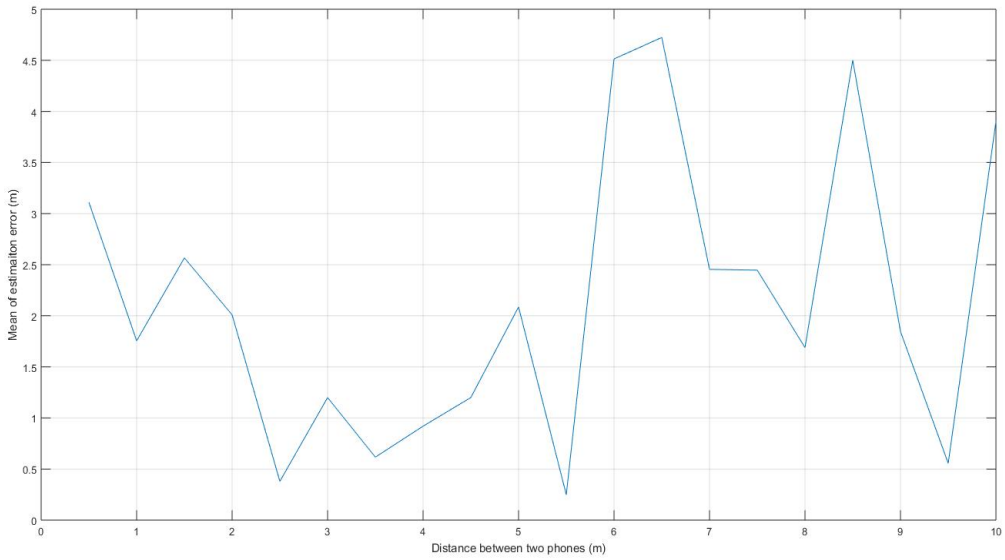

Figure 14. Mean of estimated distance error

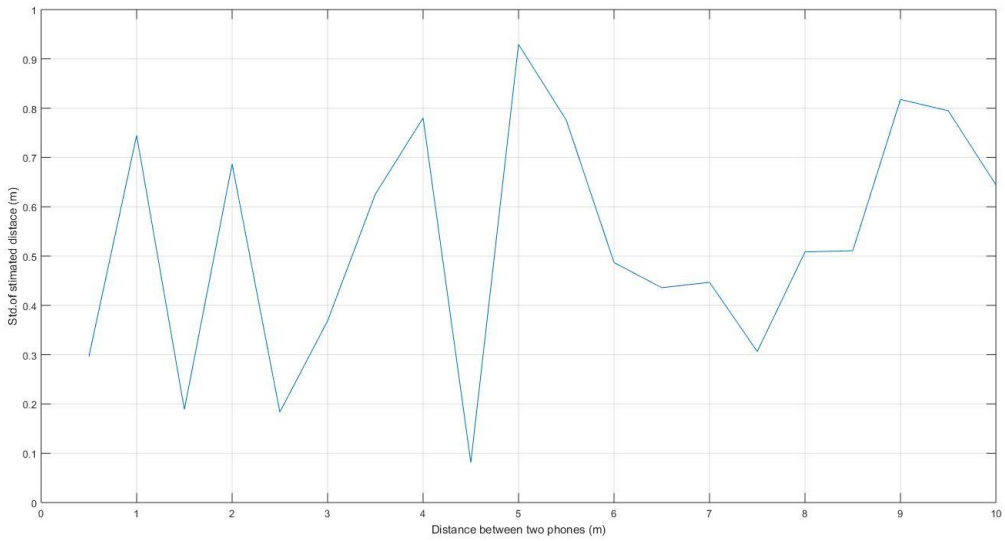

Figure 15. Standard deviation of estimated distance

VII. Conclusion

In this thesis, we have designed, implemented and evaluated the high-accuracy acoustic ranging system for Android Smartphones. It is a pure software-based solution and uses only the most basic set of commodity hardware, a speaker, a microphone, and some form of device-to-device communication. It operates in a spontaneous, ad-hoc, and device-to-device context without leveraging any pre-planned infrastructure. It is readily applicable to many sensor platforms and to most commercial-off-the-shelf mobile devices. We believe that it will have wide applications in low-cost sensor networks as well as in a compelling set of social related mobile applications that desire the proximity awareness and the fine-grained control over the spatial relationship.

Experimental results on handy cell phones demonstrate a good accuracy and excellent consistence and reliable results. So far it achieves about 4 meter ranging accuracy with less than 1.5 meter standard deviations for typical indoor and noisy outdoor environments, respectively. Because of the minimum hardware assumptions , we believe it is simple yet effective ranging mechanism can be directly incorporated into the design of other customized sensor platforms and will lead to significant cost reduction.

To summarize, we have made the following contribution. First, we identified the three major uncertainties common to any time-of-arrival based ranging system and evaluated them on COTS mobile devices. Secondly, we proposed different mechanism that cleverly overcomes all these uncertainties. Finally, we systematically evaluated the system and our design choices under several typical indoor environments using COTS mobile devices. There are still 3 problems that affect the accuracy of our estimation.

1. Audio latency is defined as the time delay that a signal experiences as it passes through a system. On a mobile device, this is deeply related to how long it takes between tapping a button on a screen and receiving audio feedback.
2. Clock unsynchronization: the possible clock skew and drift between devices.
3. The shadowing effect of acoustic signal especially in close distance.

In our future work we will focus on removing or reducing the effects of these problems. We are expecting to get much more accurate results by resolving them.

References

[1] B.g. Lee, Y.S. Lee, W.Y. Chung, ” 3D Navigation Real Time RSSI-based Indoor Tracking Application”, Journal of Ubiquitous Convergence Technology, November 2008

[2] C. Randell, H. Muller, ” Low Cost Indoor Positioning System” ,Department of Computer Science, University of Bristol, UK.

[3] R.Want , A.Hopper,V.Falcao and J.Gibbons, ” The active Badge location system” ,ACM Transactions on Information systems, January 1992 .

[4] R.Want, B.Schilit, N. Adams, R. Gold, D. Gold- berg, K.Petersen ,J.Ellis, M Weise, ” The Parctab Ubiquitous Computing Experiment”, Book Chapter: “Mobile Computing”, Chapter 2.

[5] A.Ward, A.Jones, A.Hoppe, ” A neq location technique for the active office” ,IEEE personal Communication Magazine, October 1997.

[6] A.Harter ,A.Hopper, P.Steggles, A.Ward, P. Webster,” The anatomy of a context aware application” , 5th annual ACM/IEEE ınternational conference on Mobile Computing and Networking ,August 1999.

[7] P. Bahl, V.N. Padmanabhan; RADAR, ” An in-building RFbased user location and tracking system” , IEEE INFOCOM 2000.

[8] J. Hightower, R. Want and G. Borriello; SpotON ,” An indoor 3D location sensing technology based on RF signal strength”, UW CSE, February 2000.

[9] Ch. Peng1, H. Guobin , Z. Zhang, Y. Li ,“BeepBeep: A High Accuracy Acoustic Ranging System” , Microsoft Research Asia, Beijing.

[10] K. Lorincz, M. Welsh, “MoteTrack: A Robust, Decentralized Approach to RF-Based Location Tracking”, International Workshop on Location and Context-Awareness, May 2005.

[11] P . Bahl, V.N. Padmanabhan, “RADAR: An In-Building RFbased User Location and Tracking System” , IEEE Infocom 2000.

[12] B. Kennedy, S. Hamilton, J. Martell,” Architecture and System Performance of SPAN NovAtel’s GPS/INS Solution”, IEEE/ION PLANS, San Diego, April 2006.

[13] J. Collin, O. Mezentsev, G. Lachapelle ,“Indoor positioning system using accelerometry and high accuracy heading sensors” , GPS/GNSS Conference, Portland, September 2003,

[14] A. Harter, A. Hopper, P. Steggles, A. Ward, and P. Webster, “The anatomy of a context-aware application,” MobiCom ’99: Proceedings of the 5th annual ACM/IEEE international conference on Mobile computing and networking. New York, 1999.

[15] P. Bahl, V. N. Padmanabhan, “User Location and Tracking in an In-Building Radio Network”, Microsoft Research Technical Report, February 1999.

[16] A. Harter and A. Hopper, “A Distributed Location System for the Active Office”, IEEE ,Network, January 1994

[17] R.J. Orr, G. D. Abowd, ” The Smart Floor: A Mechanism for Natural User Identification and Tracking,”, GVU Technical Report January 2000.

[18] J. Krumm, S. Harris, B. Meyers, B. Brumitt, M.l Hale, S. Shafer” Multi camera Multi-person tracking easy living”, Third IEEE International Workshop on Visual Surveillance, Dublin, Ireland, 2000.

[19] G.H. Kim, J.S. Kim, K.S. Hong” Vision-based Simultaneous Localization and mapping with Two Cameras” , RSJ Int conference on intelligent robots and systems.

[20] M. Youssef, A. Youssef, C. Rieger, U. Shankar, and A. Agrawala, “Pinpoint: An asynchronous time-based location determination system ” , 4th international conference on Mobile systems, applications and services. New York, 2006.

[21]M. Hazas , A. Hopper, " A Novel Broadband Ultrasonic Location System for Improved Indoor Positioning" IEEE Transactions on mobile Computing, May 2006.

[22] N.B. Priyantha,, "The cricket indoor location system", PhD Thesis, Massachusetts Institute of Technology, June 2005.

[23] Y. Fukuju, M. Minami, H. Morikawa, T.Aoyama," Dolphin: An autonomous indoor positioning system in ubiquitous computing environment", IEEE Workshop on Software Technologies for Future Embedded Systems, 2003 .

[24]M. Minami,. Y. Fukuju,.K. Hirasawa, S.Yokoyama,. M. Mizumachi, H.Morikawa, T. Aoyama, " Dolphin: A practical approach for implementing a fully distributed indoor ultrasonic positioning system", Ubicomp, 2004.

[25] E.E.L. Lau, W.Y. Chung, " Enhanced RSSI-based Real-Time User Location Tracking System for Indoor and Outdoor Environments", International Conference on Convergence Information Technology, November 2007.

[26] M.NI. Lionel, L. Yunhao, L. Iu Cho , A. P. Patil," LANDMARC: Indoor Location Sensing Using Active RFID", Ultra wide-band indoor positioning, Dec. 2004

[27] RG.Brown, PYC. Hwang, " Introduction to Random Signals and Applied Kalman Filtering" 1997, (3rd edition), John Wiley & Sons, New York.

[28] R. Merz, F. Chastellain, C. Botteron, A. Blatter, A. Farine, " An Experimental Platform for an Indoor Location and Tracking System.

[29] L. Girod, M. Lukac, V. Trifa, and D. Estrin, "The design and implementation of a self-calibrating distributed acoustic sensing platform" , the 4th international conference on Embedded networked sensor systems. New York, 2006,

[30] M. Zhu, " Novel Positioning Algorithms for RFID-Assisted 2D MEMS INS Systems" , Institute of Navigation GNSS 2008 conference, Savannah, Georgia, US 2008.

[31] G.Fu.Retscher, " Using Active RFID for Positioning in Navigation Systems", 4th International Symposium on Location Based Services and Telecartography, Hong Kong, 2007

[32]RG. Brown , PYC. Hwang, “ Introduction to Random Signals and Applied Kalman Filtering”, 1997, (3rd edition), John Wiley & Sons, New York, 484pp