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TRACKING THE POSITION OF AN ULTRASONIC TRANSMITTER IN A ROOM

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Abstract:

Determination of the position of an acoustic transmitter in a room in three dimensions has multiple applications. We are designing a system that will track the position of a person in a room in two or three dimensions using techniques that could be expandable to multiple rooms. The transmitter emits an acoustic pulse that is detected by four or five receivers in the room. The receiver array sends the analog signal as it is received with noise to a signal conditioner. The signal conditioner consists of an active band-pass filter, an envelope detector and a comparator and its output is a digital signal corresponding to a time-delayed version of the original pulse. The rising edge of each receiver's digital signal will trigger a microcontroller to capture the time of arrival of that signal. One of two location algorithms will be used. Either the initial start time of the emitted pulse and the Time of Arrival (TOA) of each rising edge pulse will yield the Time of Flight (TOF) for each received signal or the TOA of each signal will be used to calculate a series of Time Differences of Arrival (TDOA) of the signals. A matrix-based location algorithm will use either the TOF or TDOA information to determine the two or three dimensional position of the ultrasonic transmitter in the room.

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Introduction:

The purpose of this project is to track the position of a person in a room by designing a small mobile device to be worn or held by a person or to be supported by a moving object or vehicle. The mobile device will be an acoustic emitter and will have a companion processing system that will provide the location of the emitter in a room in 2 or 3 dimensions. The hand-held transmitter will emit an ultrasonic pulse which will be detected by multiple fixed receivers in the room. The receivers detect the acoustic signal and send an analog version of the received signal to a signal conditioning unit. The signal conditioning unit converts the analog signals to digital signals and sends the digital signals to the microcontroller. The microcontroller records the time of arrival of each signal and sends the data to the location algorithm within which the position of the emitter is determined. We are currently working on tracking the emitter in a single room with the potential to expand our results to a multiple-room case in the future.

Tracking the position of a mobile entity in a confined space has a number of potential applications. For example, many adults are continuously challenged to keep track of the whereabouts of children. With mobile tracking devices, teachers and parents could keep up with the location of a child in a building or play area. Hospital safety issues present additional applications. Mobile transmitters could be used to track the movement of disoriented Alzheimer's patients in the hectic surroundings of a busy hospital. The system could also potentially aid a parole officer in monitoring the location of criminal parolees confined to a house or apartment complex.

Development:

There are a number of algorithms and techniques to explore, evaluate and compare in the design process of our project. Our analysis includes research into the following measurement techniques: Time of Arrival (TOA), Time Difference of Arrival (TDOA), Angle of Arrival (AOA) and Time Gated Angle of Arrival (TGAOA). In researching these measurement techniques, advantages and disadvantages were identified for each. A thorough evaluation of position measurement techniques is necessary to determine the combination of techniques that will best fulfill our design requirements and constraints.

TOA techniques require synchronization of the transmitter clock and the clock associated with the receiver array. With TOA, the system captures the time the signal is transmitted, and the time at which each receiver senses the rising edge of a pulse. The time of transmission is subtracted from the time of arrival for each receiver producing the time of flight. The time of flight of the signal to each receiver is sent to the processing algorithm. A visualization of the processing algorithm presents three circles: each with one of the receivers as its center. The radius of each circle is determined by the time of flight of the emitted pulse to its respective receiver. The point at which the circles intersect is the point of transmission of the signal.

TDOA is similar to TOA except that the actual time of flight of each signal is not known. Instead, the difference between the signal arrival times is used to calculate the position of the transmitter. With three receivers R1, R2, and R3, there will be three time differences, ΔT_{12}, ΔT_{13}, and ΔT_{23}. These time differences are sent to the process algorithm where they are used to calculate the transmitter position. Whereas TOA techniques yield a set of circles with radii based on time of flight of the signal, TDOA techniques yield a set of hyperbola with fixed differences based on the time difference of arrival of the signal between any two receivers. A fixed time difference defines a hyperbola of possible locations, so three receivers will generate three distinct hyperbolas. The point at which all three hyperbolas intersect is the estimated location of the transmitter.

TDOA can be implemented mathematically more easily using four receivers than it can be using three receivers. The fourth receiver is used as a reference point in determining the time differences. With the fourth receiver, R4, there are still four time differences, but they are ΔT_{12}, ΔT_{13}, and ΔT_{14} instead of ΔT_{12}, ΔT_{13}, and ΔT_{23}.

AOA and TGAOA are very similar. In AOA, we would need to measure the angle at which the acoustic signal is received. This angle is used to draw a straight line in the direction of wave propagation. Two of these lines will intersect at a point, determining the location of the transmitter. TGAOA is based on the same concept with an added feature. In TGAOA, each receiver is programmed to detect only signals received within a predetermined time interval. This timing feature aids in the prevention of interference due to multi-path.

TOA systems are inherently problematic when it comes to the synchronization of the transmitter and receivers. The TOA algorithm must have information containing the exact time at which the acoustic signal is transmitted and received. In order to satisfy the synchronization requirement, we would be forced to tether the mobile emitter. Transmitter mobility is a key factor in our project expectations so we will not use TOA techniques. AOA techniques also have potential problems. For example, it is difficult to measure the angle of arrival of a signal and AOA is most useful when working with RF sources.

Finally, the location algorithms associated with TDOA techniques are more complicated than those associated with TOA techniques. However, the fact that TOA methods require the transmitter to be tethered to the system makes the more complicated TDOA process a more attractive method for location determination.

In addition to choosing an appropriate location algorithm, we needed to evaluate various wave types in order to choose between RF (radio frequency) and ultrasonic (acoustic) signals. Acoustic waves travel at approximately 1100 ft/sec, while RF waves travel at the speed of light.

As a transmission wave type, RF signals have a few key advantages to acoustic signals. Signal loss to the air during transmission is very low for RF signals because of the high speed at which they propagate. Also, the electromagnetic waves have very low loss factors. In addition, for RF signals, interference from objects or walls in a room is minimal. RF signals at the proper frequency would propagate through walls and any other objects, thus eliminating the problems created by reflection or refraction. Implementation of RF would also eliminate the need to distinguish between signals received directly from the transmitter and signals that are reflected off of other objects in the room before reaching the receivers.

The problem with RF signals is that they require extremely fast processing. RF signal processing is executed with expensive equipment that is not readily available. RF processing also requires specific knowledge that is beyond the scope of our courses. Despite the advantages of RF described in the previous paragraph, the significant drawbacks associated with expense and our limited experience with RF are impossible to ignore.

Acoustic signal propagation is driven by pressure differences in a medium. The pressure differences are created by a mechanical device such as a speaker cone. The transmitter mechanically displaces the medium, in this case air, transmitting a signal that propagates omni-directionally into the medium and eventually, to our receiver array.

The problems with acoustic signals are not the same as those associated with RF signals. The transmitted acoustic signal reflects off of objects in the room and the room walls and these reflections cause false arrival signals at the receivers. Signal loss is also a big problem when dealing with acoustic signals. As the signal moves through the air, it loses strength, which requires a large transmitter amplitude to get the signal to the receivers.

The advantage of using acoustic signals stems from our level of experience with many of the hardware and software techniques we will need to use. With acoustic signals, we can utilize circuits similar to those presented in our introductory electronics class and we can utilize microcontroller programming techniques that we learned in our microcontroller class. The background we have in the academic areas relevant to acoustic signal processing as opposed to RF signal processing makes the decision to use acoustic signals an easy one.

Design requirements for the transmitter were derived from the possible applications of our design. We would like the transmitter to be hand held, possibly the size of a pager or garage door opener. Power restrictions were set to allow the apparatus to operate for one day, preferably from batteries. The signal should propagate omni-directionally from as close to a point source as possible and the signal strength needs to be large enough to ensure reliable detection at the receivers.

Receiver design requirements will be derived from the chosen process algorithm and the desired tracking type. For 2-D and 3-D applications, the number and location of the receivers will vary. The fact that we will use over-determined arrays, more receivers then the required minimum, will also effect the position of the receivers. The optimal geometric array for signal reception will be determined and implemented for our application. Each receiver will be physically attached to the signal conditioning system with wires.

Because our problem statement requires that we track the position of a person in a room, we chose to try to track the position of the emitter within a five foot radius. We chose five feet as a reasonable distance because the transmitter could be at a person's feet, head, left hand extreme, or right hand extreme which covers an average range of approximately five feet. We also decided that ten seconds would be a reasonable time update rate. Ten seconds allows the system time to process the signals and prevent overlapping and bottle-necking in the data acquisition process. Ten seconds was chosen as a maximum update time and one goal of the final design is to significantly reduce this processing time.

Preliminary Design:

The objective of our project is to track the position of a person in a room in two or three dimensions using techniques that can be expandable to multiple rooms. This is done with a transmitter that sends a 50 kHz omnidirectional ultrasonic pulse. This pulse propagates to a receiver array where each receiver in the array receives the pulse at a different time. Signals from each receiver are conditioned and arrive times are recorded. These times are sent to the position algorithm to find the transmitter location. Figure 1 displays a block diagram of the process.

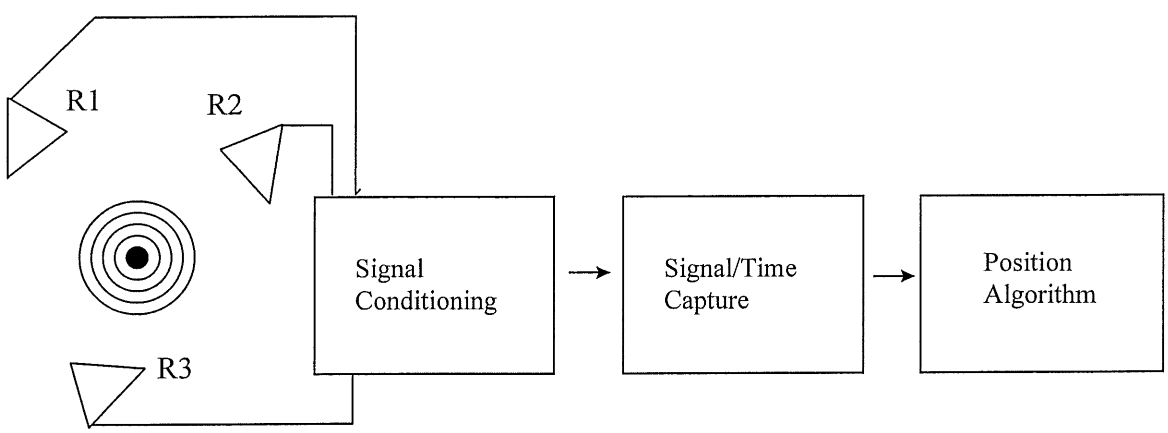

Figure 1: System Modularization

The symbols, R1,R2, and R3 in Figure 1, represent the receiver array, while the black dot with concentric rings represents our transmitter and its acoustic waves. Each box represents a subsection, or module, of our project. The preliminary design consists of three modules: the transmitter/ receiver circuitry, the signal and time capture or, data acquisition, and the position algorithm. Modularization of a project allows for specialization and efficiency within the group. The transmitter/receiver circuitry consists of a working transmitter, a receiver array, and the conditioning circuitry in the signal conditioner. The data acquisition module consists of identifying and programming an appropriate data acquisition interface, capturing the time of arrival of each signal, and sending the timer data to the position algorithm. The position algorithm module consists of understanding the different methods of position calculation, and developing unique equations for our project.

Technical Approach

Transmitter/Receivers

The first module in the system is the transmitter/receiver circuitry. The purpose of this module is to initialize an emitted acoustic pulse, receive the pulse, and transform the pulse into a digital signal. There are three basic sections to this module; the transmitter, the receiver array, and the signal conditioning system. These three sub-sections must be inter-linked in order to produce an effective transmission of the signal.

The first goal of this module was to design a mobile ultrasonic transmitter to emit an omni-directional pulse. The first milestone in this quest was to locate and purchase transducers that could be tested for our application. During our research we found an ultrasonic ranging system built by Polaroid. Polaroid's system is a sonar system which has a single transducer that acts both as a transmitter and a receiver. The single transducer measures the time of flight of its own signal in order to calculate the distance to a target. We found that we could use the transducer and its circuitry as a transmitter alone by operating it as it is designed to operate, but by using separate receivers to complete the system.

An array of receivers is required to capture the data we need for calculating the location of the transmitter. We needed to find transducers that would work in the frequency range of the transmitter. From our research, we found that the transducer we chose to use as a stand-alone transmitter can also be used as a stand-alone receiver. In order to test the transmitter/receiver relationship, the next milestone in this circuitry module is to test the signal reception using the transmitter and only a single receiver. Once we identified the Polaroid transducers, we ordered two of the ultrasonic ranging systems in order to use one of the transducers as a transmitter and one as a receiver. The single transmitter and receiver case is illustrated in Figure 2.

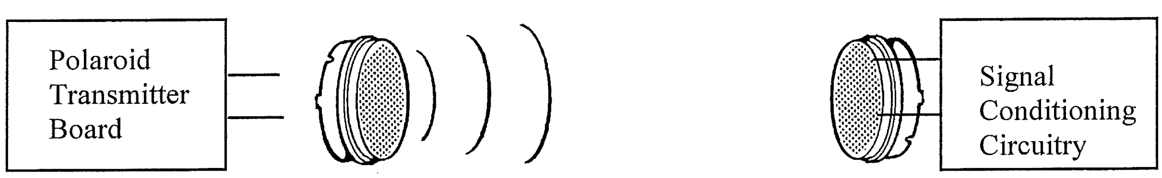

Figure 2. Transmitter and Receiver

The transducer transforms a digital electrical pulse into an ultrasonic pulse so that it can be transmitted through the air. A thin foil on the transmitting transducer mechanically moves and transforms the electrical energy from the power source into sound waves. The opposite is also true in that, at the receiving end, the transducer converts these sound waves back into electrical energy for signal processing.

The foil on the transducer is plastic (Kapton) with a conductive gold coating on the front side. This foil is stretched over an aluminum backplate as shown in Figure 3. The backplate and foil represent an electrical capacitor, which, when charged, produces an electrostatic force on the foil. An AC voltage of given frequency forces the foil to move at the same frequency, producing sound waves. A steel retainer transfers the voltage from the backplate and holds the foil in tension. The housing protects the foil from damage with minimal signal strength loss.

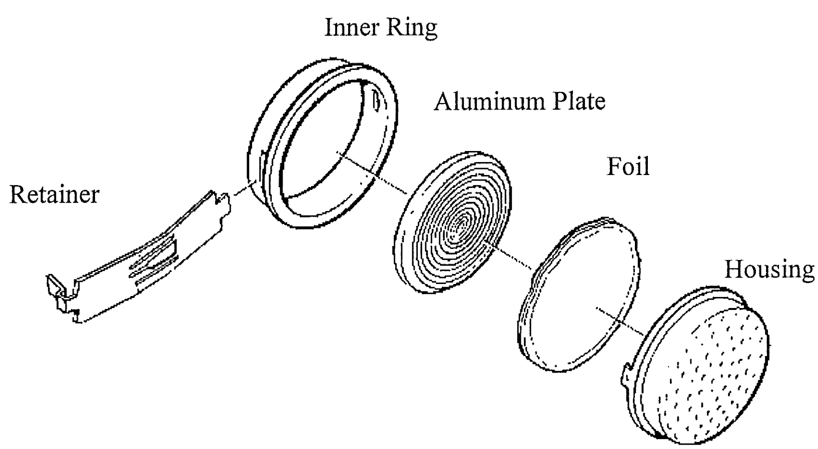

Figure 3. Transducer

As a transmitter, the transducer circuitry emits a simulated chirp consisting of four frequencies: 60, 56, 52.5, and 49.41 kHz. The electrical signal, consisting of these frequencies, is transformed by the transducer into an ultrasonic pulse which propagates to the receiver array. The receivers work like the transmitter, but in reverse. They convert the ultrasonic pulse into an electrical pulse that can be sent through wires to the signal processing unit.

Placement of the receivers in the array is very important. By changing their arrangement we can vary the reliability of the results from our system. The receiver location will be used in the algorithms that we will develop for calculating the position of our transmitter. The first milestone in the receiver array subsection is the use of time of arrival equations to find an optimal receiver array configuration. For two dimensional location determination, the optimal configuration is an equilateral triangle as shown in Figure 4.

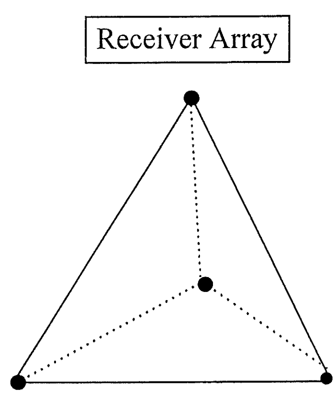

Figure 4. Receiver Array

The signals that are sent from the receiver transducers are small in amplitude and have noise that is picked up during transmission and reception. The signal conditioning unit is required to process these signals and convert them into signals that are more representative of the original transmitted pulse.

This circuitry has to amplify the signal, filter out the noise, and transform the analog signal into a digital signal.

In order to eliminate the effects of noise, we will use a band-pass filter. The band-pass filter will allow us to set the bandwidth such that the circuit allows the original four transmitted frequencies to pass to the output, while filtering out all other frequencies. Since there is minimal noise in the range of 50-60 kHz, a simple band-pass filter will be adequate, but in order to compensate for signal loss associated with transmission of an acoustic signal, the band-pass filter will also need to incorporate signal amplification. We will combine the filtering and amplification steps to build an active band-pass filter as shown in Figure 5.

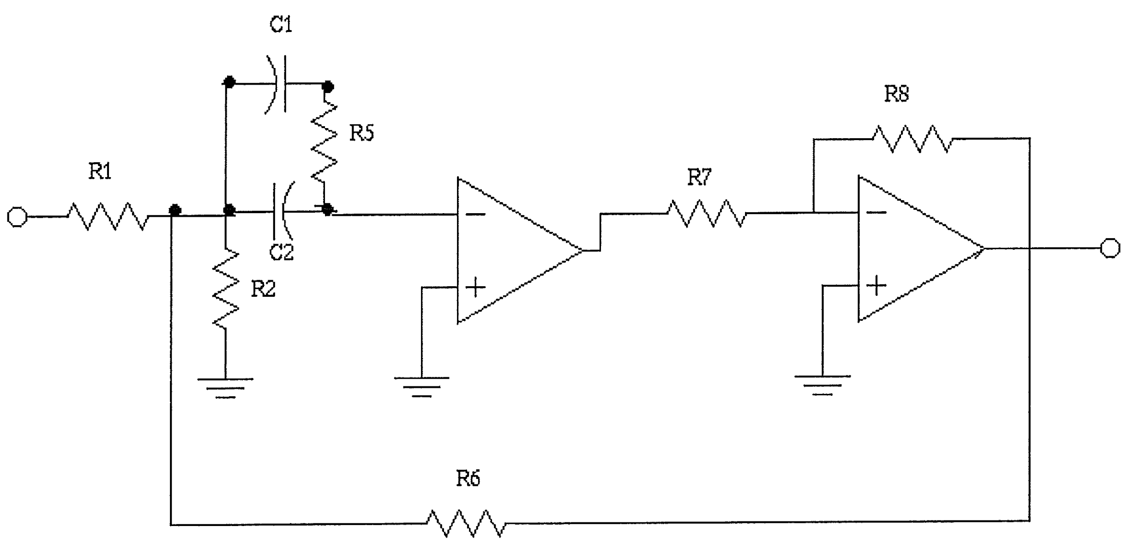

Figure 5. Active Band-Pass Filter

By changing the values of C1 and C2 as shown in Figure 5, we can vary the center frequency of the filter. To change the amplification, we can vary the value of resistor, R8. This filter will take a received signal with noise, amplify it and filter out frequency that are not in its bandwidth. A relatively narrow bandwidth and a high inherent amplification are expected from this active band-pass filter. Because of the narrow bandwidth most of the noise should be filtered out, and the amplification will aid in converting the signal from analog to digital in the next step.

The input and output of the active band-pass filter are analog signals, but we need to convert them to digital signals using the two-step process illustrated in Figure 6.

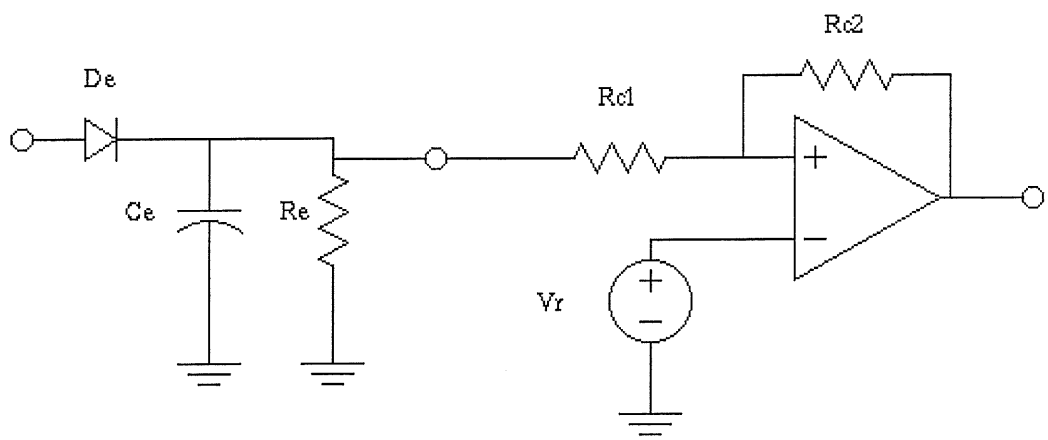

Figure 6. Envelope Detector/ Comparator

An envelope detector was designed to trace the top of the analog signal, creating a rough square wave. Controlling the rise time of the envelope will improve its recreation of the square wave by varying the values of Ce and Re as shown in Figure 6. The resulting square wave will have a high value above the desired 5 volts and a non-zero low value. A comparator compares the voltage of a signal to a set voltage. When the voltage of a digital signal rises above the set voltage, the output is set to 5 Vots. When the voltage falls below the set voltage, the output is set to 0 Volts. Vr is the set voltage for this comparator and varying

Rc1 and Rc2 effectively set the high and low values to 5V and 0V respectively. The output of the circuit is the desired digital signal.

Each receiver in the array will be fitted with the signal conditioning circuitry described above. A signal from each of the receivers in the array is passed through a conditioning circuit and is then passed on to the data acquisition system.

Data Acquisition

The output from the signal conditioning module is a set of digital signals varying between 0 Volts and 5 Volts corresponding to the signal detected at each receiver. This section explores the options available for acquiring the data signals and determining from them the information necessary to calculate the position of the emitter in the room. The information we need will depend a little on the location algorithm we choose to use, but for both the TOA and TDOA methods, we need a way of finding either the absolute or relative time at which each signal is received. The data acquisition problem becomes one of rising edge detection and timing.

The first data acquisition interface we considered was LabVIEW® software. LabVIEW interacts with a National Instruments Multifunction I/O board. The board in the electronics lab is an AT-MIO-16E-2 board. The AT-MIO-16E-2 board has a 1 byte, 8 bit digital I/O port which means that the card can read up to eight different signals. Each bit corresponds to a separate channel of input on the DAQCard and our applications would use only four of the input channels, one for each receiver. The card has a maximum sampling rate of 500,000 samples/second.

With whatever data acquisition technique we use, we need to detect the rising edge of each signal and trigger some sort of time capture. The National Instruments DAQCard has only 2 up/down counter/timers which will not be adequate for the type of individual signal processing we need to do. The realization that clocking presents a problem with using LabVIEW and the DAQCard in the Electronics Lab led us to explore some other options.

The data acquisition methods we are currently exploring involve the use of microcontrollers. Right now, we are considering three different microcontrollers: the Z-World BL1100, the Z-World BL1400, and the Motorola MC68HC11.

Z-World is one of the leading producers of programmable controllers and two of the microcontrollers they market are called the BL1100 and the BL1400. The BL1100 is a general purpose controller with analog inputs and outputs, user configurable digital I/O, serial I/O and more. It measures 5.6" by 4.8" by 1.2" and it has 4 serial ports which interface directly with a host of serial I/O equipment, such as printers, modems and PCs. The BL1100 has a clock speed of 9.216 MHz and has 6 general-purpose counter/timers that adapt to applications requiring either periodic outputs or counting of external events. The BL1100 costs between $350 and $570.

The Z-World BL1400 has 12 user configurable digital I/O lines, resistive measurement input, and serial communications all on a board the size of a credit card. The BL1400 offers a high level of control in a very small package. It measures just 3.2" by 2.0" by 0.56" and can support up to 12 channels of logic-level, digital inputs and outputs. The BL1400 has 12 versatile, programmable I/O lines complemented by a real-time clock. Each I/O line may be set up individually as either an input or an output, providing a great deal of flexibility. The clock speed of the BL1400 is 6.144 MHz and costs between $70 and $120.

The third microcontroller we are looking at is the Motorola MC68HC11. The MC68HC11 has a high performance timer which is provides some flexibility and ease of use. The system is based on a free-running 16-bit counter with a programmable prescaler, overflow interrupt, and separate function interrupts. The MC68HC11 timer has multiple input capture functions and provides a selection of timer sub systems geared towards timing-intensive applications. Some of the supported subsystems include input captures, output compares, real-time interrupts and pulse accumulation. The Motorola MC68HC11 is available in the microcontroller lab and is, therefore, available for free.

Because we have the MC68HC11 microcontrollers in the lab, information about them is more readily available at this time. In the MC68HC11, input-capture functions are used to record the time at which some external event occurred. Input-captures are accomplished by latching the contents of the free-running counter when a selected edge is detected at the related timer input pin. The edge-detection logic includes control bits so that user software can select the edge polarity that will be recognized. Each of the input-capture functions can be set to detect rising edges only, falling edges only, or any edge (rising or falling). The time at which the event occurred is saved in the input capture register. Even though the software may take some small amount of time to respond to the event, it can tell exactly when the event occurred.

We will use the microcontroller to record the time at which each signal's rising edge is detected. For now, we are planning to use appropriate software designed to interface the microcontroller with the C programming language. Within C, we will use the timer data and a location algorithm to determine the position of the emitter.

If we decide to use the TOF methods described in the Development section, we will need a digital version of the acoustic emitter signal in time because the position algorithm for TOF is based on the absolute time of flight of each signal. In the case of TOF, the output of the data acquisition module will be the time of arrival of each of three signals from the receivers as well as the initial start time of the emitted signal. The absolute time of flight for each received signal is determined within C by subtracting the time of arrival of the received signals from the initial start time of the emitted signal.

On the other hand, if we decide to use the TDOA methods described in the Development section, we will have four captured receiver times as opposed to three receiver times and an initial start time. The output of the data acquisition module will be the time of arrival of each of four signals from the microcontroller.

Position Algorithm

The position algorithms discussed in this section TOF and TDOA measurement techniques as basic milestones in solving the location problem.

The TOF absolute ranging equations function as a preliminary step in understanding the basic process of the position algorithm. Through research of both measurement methods, we plan to develop our own, tailored, TDOA algorithm that will require only three fixed receivers.

In order to begin the development of the position algorithms, the given variables must be assessed. Since we are only concerned with two-dimensions and we assumed that the speed of sound is a known constant value, the system will require three receivers i=1, 2, 3. The coordinates of each receiver (x_i , y_i) in the array will be known based on the initial position calibration of the system. In order to proceed with time of flight calculations, the transmitting signal start time, t_s., and each of the four receiver's capture time, t_{ri}, will need to be determined. From these two times, the time of flight of each signal from transmitter to receiver, t_i *can be calculated,* $t_i = t_{ri} - t_s$. The main variables that need to be determined are the transmitter coordinates, (u , v), and the time delay of the network: t_d. The time delay is a constant time offset value that will be determined during system calibration.

The distance between two points (transmitter and receiver), l_i, can be found using the Pythagorean theorem: $l_i^2 = (x_i - u)^2 + (y_i - v)^2$.

When the squared terms are expanded, the equation is:

$l_i^2 = x_i^2 + y_i^2 + u^2 + v^2 - 2x_i u - 2y_i v$. To proceed, we must introduce a few new variables and expand upon the distance term. A term denoted by p^2 will be equal to the sum of the squares of the transmitter coordinates, $p^2=u^2+v^2$. The sum of the squares of the coordinates of each of the receivers is denoted as

$r_i^2=x_i^2+y_i^2$. The distance between the transmitter and receiver is equal to the time of flight of the signal including the time delay factor multiplied by the speed of sound in the medium, in this case room temperature air. The distance squared can be noted as: $l^2=c^2(t_i - t_d)$. If these new terms are placed in the expanded distance equation, the following equality is produced.

$$\left(t_i - t_d\right)^2 = \frac{r_i^2}{c^2} + \frac{p^2}{c^2} - \frac{2ux_i}{c^2} - \frac{2vy_i}{c^2}$$

Expanding this equation into a matrix equation containing information from the three receivers. The linear system of equations, $m = A\mu$, is represented as follows:

$$\begin{Bmatrix} \left(t_1 - t_d\right)^2 - \dfrac{r_1^2}{c^2} \\ \left(t_2 - t_d\right)^2 - \dfrac{r_2^2}{c^2} \\ \left(t_3 - t_d\right)^2 - \dfrac{r_3^2}{c^2} \end{Bmatrix} = \begin{bmatrix} 1 & x_1 & y_1 \\ 1 & x_2 & y_2 \\ 1 & x_3 & y_3 \end{bmatrix} \begin{Bmatrix} \dfrac{p^2}{c^2} \\ \dfrac{-2u}{c^2} \\ \dfrac{-2v}{c^2} \end{Bmatrix}.$$

Where m is the time measurement vector which includes a constant term, $\frac{r_i^2}{c^2}$, A is the receiver coordinate matrix, and μ, is the vector containing the unknown transmitter coordinates. The unknown vector can easily be solved for using basic linear algebra techniques. Thus the coordinates of the transmitter (u, v) can be determined and displayed.

Although the aforementioned time of flight method works, the accompanying electronic circuitry is rather difficult to accurately implement. Two main problems may arise from this implementation. The first deals with synchronizing the receiver and transmitter pairs. Originally we wished to have an autonomous un-tethered transmitter, retaining this transmitter requirement would cause a great deal of difficulty in the synchronization of the system. Secondly, once a time of flight measurement is attained, the time delay of the system must be identified and quantified. This can be very difficult due to delay caused by the signal conditioning circuitry, inherent delay in the signal detection method to acknowledge reception of the pulses at the receivers, and acousto-electro-mechanical delay associated with the transducers.

In order to avoid the timing problems mentioned above, a method using the differences in the time of flights to the receivers or TDOA should be used. After much research, a position algorithm based on differences in time of flight was found. This algorithm allows the transmitter to remain un-tethered and filters the time delay that is inherent in the time of flight method.

As in the TOF method, the speed of sound in the TDOA method is assumed to be a constant value. The two-dimensional system calls for a transmitter and four randomly located receivers. The only requirement for the receivers is that they are all on different circles, with no circle having more than one receiver on it as shown in Figure 7.

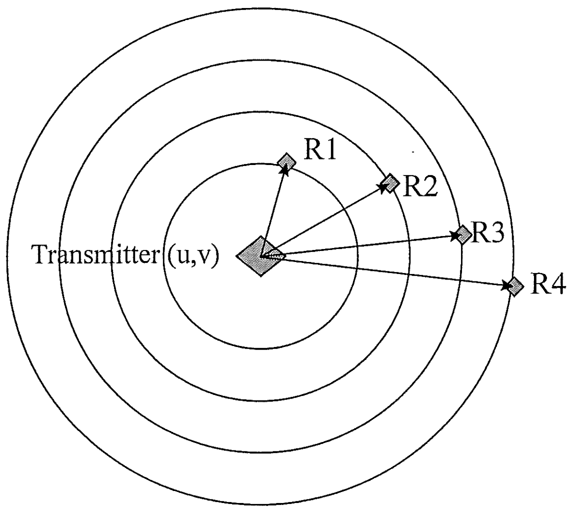

Figure 7. Physical Interpretation of Algorithm

With the TDOA method, the time-of-flight between the transmitter and any receiver is unknown, but the difference in times when the receivers detect the signals can be measured.

If the transmitter sends the signal at time T=0, then each of the receivers senses the signal at unknown times T_1, T_2, T_3, and T_4. The differences in the times of flight are:

$$\Delta T_{12}=T_2-T_1$$
$$\Delta T_{13}=T_3-T_1$$
$$\Delta T_{14}=T_4-T_1.$$

Since sound travels in circular waves from a transmitter, four concentric circles can be drawn around the transmitter; one circle of radius d through the point R_1; another circle of radius $d+c\Delta T_{12}$ through the point R_2; a third circle of radius $d+c\Delta T_{13}$ through the point R_3; and a forth circle of radius $d+c\Delta T_{14}$ through the point R_4 as shown in Figure 7. The receiver that senses the signal first will always be named receiver R1 at a distance d from the transmitter. Another receiver will detect the signal second, and will be at a distance ($d+c\Delta T_{12}$), where c is the velocity of sound. The same is true for the third and forth receiver. The equations for each of these concentric circles are:

$$\begin{aligned}(x_1-u)^2+(y_1-v)^2&=d^2\\(x_2-u)^2+(y_2-v)^2&=(d+c\Delta T_{12})^2\\(x_3-u)^2+(y_3-v)^2&=(d+c\Delta T_{13})^2\\(x_4-u)^2+(y_4-v)^2&=(d+c\Delta T_{14})^2\end{aligned}$$

Multiplying out the equations and solving the first one for d^2 yields:

$$d^2=x_1^2-2x_1u+u^2+y_1^2-2y_1v+v^2$$

Substituting d^2 into the remaining three equations gives three linear equations with three unknowns, *u*, *v*, and *d*:

$$\begin{bmatrix}2x_1-2x_2 & 2y_1-2y_2 & -2c\Delta T_{12}\\2x_1-2x_3 & 2y_1-2y_3 & -2c\Delta T_{13}\\2x_1-2x_4 & 2y_1-2y_4 & -2c\Delta T_{14}\end{bmatrix}*\begin{bmatrix}u\\v\\d\end{bmatrix}=\begin{bmatrix}c^2\Delta T_{12}^2+r_1^2-r_2^2\\c^2\Delta T_{13}^2+r_1^2-r_3^2\\c^2\Delta T_{14}^2+r_1^2-r_4^2\end{bmatrix}$$

where r_i^2 is the sum of the squares of the coordinates of the i^{th} receiver, $r_i^2 = x_1^2 + y_i^2$. This matrix equation can be easily solved for the location of the transmitter (u, v) and the distance (d) from the transmitter to the first receiver (R_1). As mentioned before, the speed of sound, c, is assumed to be known, and is treated as a constant.

Using one of the above algorithms, the coordinates of the transmitter can be determined. From the resulting coordinates, we can graphically display the location of the emitter.

Future Objectives:

While we are pleased with the progress we have made thus far, we know that there is still quite a bit of work to be done. The transmitter/receiver system must be tested in order to determine the signal strength necessary to ensure adequate signal reception. We also need to design and build a custom transducer to function as an omni-directional transmitter. Currently, the transmitted signal is very linear and our design specifications are based on an omni-directional acoustic signal.

In terms of the signal conditioning unit, we have designed and built some basic circuits. The circuits work right now, but we need to continue to test them, ensure that they meet our minimum performance goals, optimize them, and produce them in a more permanent form.

We still have some design work to be done on the data acquisition system. We need to continue to evaluate the pros and cons of the different types of microcontrollers and be sure that it will interface successfully with the other aspects of our design so far. Once we decide on a microcontroller, we need to program it to capture the time of arrival of the signals and set up an interface between the microcontroller and the location algorithm in C.

We are currently considering the TOA and TDOA processing algorithms which essentially establish a set of circles to find the transmitter location. We would like to use the algorithms from previous studies to develop our own algorithm that will use TDOA and hyperbolic ranging to find the transmitter location. A custom algorithm will require fewer receivers to obtain the position of the emitter than the existing algorithms will. So far, we have had enormous difficulty trying to implement a TDOA system because of the complexity of working with hyperbolas.

The first milestone we hope to reach next semester will be to get our one-dimensional system running with the Polaroid transducers. The test system will give us an opportunity to test our circuits and preliminary algorithms. Once we have a fully-functional one-dimensional system, we plan to work on customizing the transmitter and expand the system to two and eventually, three-dimensions.

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