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Design and Performance Analysis of Sound Source Localization using Time Difference of Arrival Estimation

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ABSTRACT

Sound source localization (SSL) is a process of processing sound signals received from sound sensors and locating the sound origin. In many applications, precise localization of these sound sources is crucial. The accuracy of the localization of the sound source depends on several factors, and one of the factors is the number of sound sensors. This work aims to develop a microcontroller-based sound sensor localization system using the Time Difference of Arrival (TDOA) method, as well as to analyze SSL by comparing the time difference received by two and three sound sensors with respect to the distance of sound, and to compare the angle accuracy of the sound localization through TDOA with the real sound location. A few experimental tests were conducted using a constant sound of 90db and a frequency range of sound of 50Hz to 10kHz. The mean error (inaccuracy) of the experimental test of the angle acquired within 5cm of the source is 34.66°, while the angle obtained within 10cm of the source is 34.29°. The difference in angle inaccuracy between two and three sensors has a reduction of 27.64%. By adding more sound sensors, angle accuracy can be increased through the TDOA method.

1. Introduction

Sound covers every area of our daily lives, and its presence or absence has a direct influence on our health and well-being. As a result, devices for monitoring sound volume, kind, and source exist in the environmental, industrial, and military sectors. Sound source localization is in demand in these sectors for various applications that include detecting illegal deforestation [1], gunshot localization [2], robots [3], drones [4], autonomous vehicles [4] and indoor surveillance systems [5].

Sound source localization (SSL) is the process of assessing sound signals received from a microphone array to locate the location of the sound. Sound source direction identification [6] gives important information that may be applied in a variety of ways. As a result, several works [7,8] have

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previously been done using different methods for developing sound source localization systems in noisy reverberant environments [9–11].

The type of microphones used, as well as the number of microphones [12] and their geometry, or location, contribute a significant effect on the signal processing performance of SSL systems. Binaural, tri-aural, and tetra-aural configurations, clustered arrays, and non-co-planar array structures are only a few of the array configurations that may be employed, each with a different number of microphones and positions [13,14].

Acoustic source localization [15-17] methods are becoming more widely used as a result of recent advances in machine learning, cloud computing, and on-chip electronics. Acoustic source localization in a vehicle and ship identification has been used to lead product noise control [18], target selection and interference rejection for communication devices or speech recognition processing, and condition monitoring for mechanical systems. Due to their ability to estimate source strengths and the resultant sound field, localization techniques have been commonly employed in the acoustical design of audio equipment and theatre systems, noncontact vibration measurements, and audio virtual reality systems, among other uses.

The time difference of arrival (TDOA) [19-21] approach makes advantage of the signal's timing difference. The TDOA was calculated on the assumption that the sound source signal is narrowband. The TDOA method is designed to determine the sound source using known nodes. To put these ideas into action, a reference node must be chosen to remove noise and simplify synchronization needs. As a result, sound source localization accuracy [22,23] is highly dependent on the reference utilized. Because of this dependency, when the reference is chosen poorly, the performance of such systems typically decreases.

The source angle accuracy of SSL has been identified as a concern through study. This is due to the lower number of microphones, which will affect the reading, such as inaccurate time readings as well as the distance between each microphone and the distance of the sound source. Therefore, this work aims to develop a microcontroller-based sound source localization system using the TDOA method and to compare the time difference received by two sound sensors with respect to the distance of sound as well as to compare the angle accuracy of the sound localization through TDOA with the real sound location.

2. Methodology

This section explains the framework and procedure of this study, beginning with the main focus, which is sound detection, which comprises SSL implementation and motor control based on source location angle. The following stage is hardware implementation, which in this case uses an Arduino Uno as a microcontroller, a sound sensor as an input, and a stepper motor as an output operated by a motor shield. Following hardware implementation, an experiment was carried out, and the data acquired were analyzed. The time difference of arrival (TDOA) technique was utilized for SSL and the microcontroller used as a hardware implementation in this study is an Arduino Uno board and Motor Shield board for the stepper motor driver.

2.1 Sound Sensing

Figure 1 shows the circuit of the sound sensor that consists of the omnidirectional microphone with a frequency range of 50 Hz to 10kHz. LM393 comparator is used as a voltage comparator in the sound sense which is connected to the digital out pin. When the sound is produced, the comparator will compare the amplitude of the produced sound. If the amplitude is equal to or more than half of the voltage supply, the signal will be sent as 'HIGH' at the digital pinout. For this project, the voltage supply is 5V, so when the sound produced is 2.5V or more, the digital output will turn 'HIGH'.

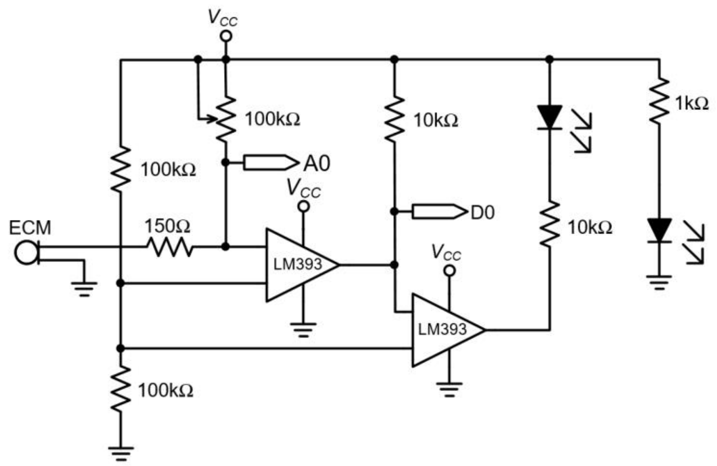


Fig. 1. Circuit of the sound sensor

Figure 2 shows the amplitude of sound from the analog pinout that is produced by clapping using an oscilloscope. The first channel and second channel are presented for 'Mic 1' and 'Mic 2' respectively. For this part, the clapping sound was made 5cm to Mic 1, so Mic 1 will detect the sound first and as shown above, it appeared to be more than 2.5V. When the amplitude is reached 2.5V, the digital out will read as HIGH and the time difference that is recorded is 522µs.

Fig. 2. The amplitude of sound when it reached the threshold voltage

2.2 Sound Source Localization through Time Difference of Arrival

For source localization, the time of arrival (TOA), time difference of arrival (TDOA), and direction of arrival (DOA) of the emitted signal is commonly used. TOAs and TDOAs in general, offer distance information between the source and sensors, whereas DOAs represent source bearings relative to the receivers. Finding the source position, on the other hand, is not an easy operation since these measures have nonlinear connections with the source position.

TDOA method has the advantage of less computation time due to its distributed processing in comparison to TOA but has a disadvantage of mitigating noise and reverberation interference and is less sensitive on synchronization between sound sensors [8]. TDOA method uses the time difference between the signals. This may be accomplished by measuring the time difference between zero-level crossings or the onset timings of both signals. The basic idea of TDOA is to exploit the time difference between N-pairs of sensor nodes. Assuming the sensor nodes have multiple sound sensors, the TDOA computations can be performed separately by each node. The TDOA algorithm is meant to pinpoint the sound source using nodes whose coordinates are known. To implement these approaches, a reference node must be selected to eliminate noise and simplify synchronization requirements. As a result, the accuracy of sound source localization is highly dependent on the reference used [24]. Because of this dependency, the performance of such systems frequently decreases when the reference is chosen incorrectly.

$$\Delta t\ (\mu s) = \frac{D}{c} \tag{1}$$

The angle of arrival the of sound source, θ, is based on the times difference of the two sound sensors and can be derived as

$$\theta(°) = \arcsin\left(\frac{\Delta t \times c}{D}\right) \tag{2}$$

Figure 3 shows the illustration of the angle of source with respect to the time difference and distance between nodes (sound sensors). Using the speed of sound, it is simple to determine how long the sound will take to travel from one sound sensor to another. With c as the speed of sound which is $343\ m/s$, and D as the distance between sensors, the time difference given as Δt

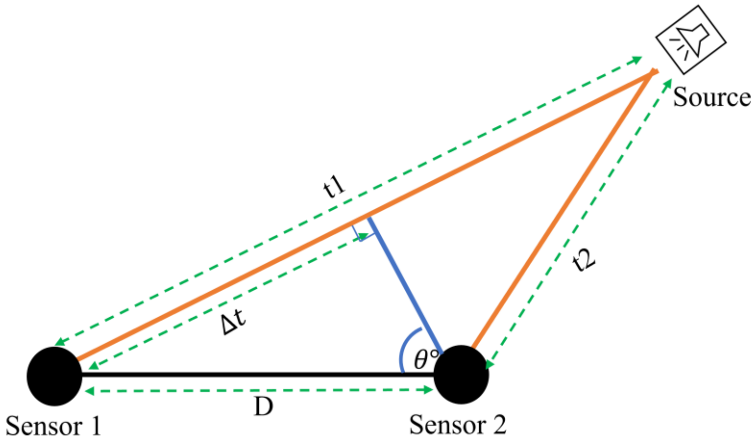


Fig. 3. 2-dimensional TDOA estimation between two sound sensors

Eq. (3), Eq. (4), Eq. (6), and Eq. (7) show the angle of arrival in TDOA estimation for three sound sensors setup. The accuracy of the source angle can be improved due to the average estimation as in Eq. (5). Hence more sound sensors will achieve accurate angle arrival calculations. The angle arrival calculation is separated into three different set of angles, first as shown in Figure 4, the range is between 0° to 89° with the distance between the sensors is depicted as $D_{S1\text{-}S3}$ and $D_{S2\text{-}S3}$.

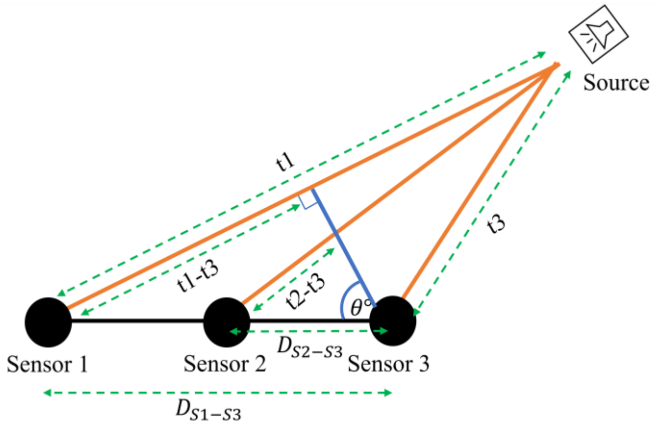


Fig. 4. Illustration of TDOA when the sound is at the range of 0° to 89°

$$\theta_1 = \arcsin\left(\frac{(t1-t3)\times c}{D_{S1-S3}}\right) \tag{3}$$

$$\theta_2 = \arcsin\left(\frac{(t2-t3)\times c}{D_{S2-S3}}\right) \tag{4}$$

$$\theta = \frac{\theta_1+\theta_2}{2} \tag{5}$$

Second as in Figure 5, specifically for 90° as t1 is equal to t3 and the third angle is between 91° to 180° as in Figure 6 with the distance between the sensors is depicted as D_{S3-S1} and D_{S2-S1}.

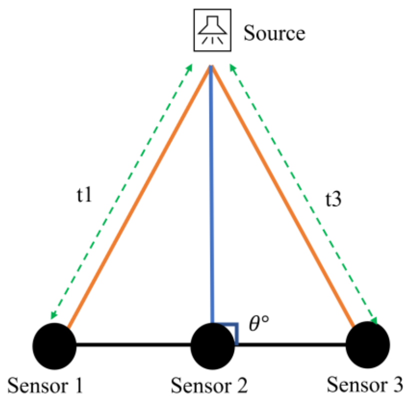


Fig. 5. Illustration of TDOA when the sound is at 90°

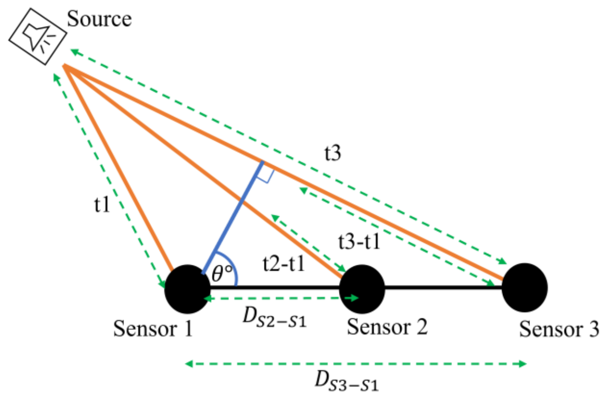


Fig. 6. Illustration of TDOA when the sound is at the range of 91° to 180°

$$\theta_1 = \arcsin\left(\frac{(t3-t1)\times c}{D_{S3-S1}}\right) \tag{6}$$

$$\theta_2 = \arcsin\left(\frac{(t2-t1)\times c}{D_{S2-S1}}\right) \tag{7}$$

2.3 Motor Control by the Calculated Source Angle

A stepper motor is used for this work for its maximum torque at low speeds and it is suitable for low-speed rotation with high precision. In this work, the stepper will rotate according to the angle of arrival had been calculated after the time difference is calculated as Eq. (2). As for this work, the stepper motor's whole rotation has been set as 200 steps. As a result, 200 rotation steps are equivalent to 360° with 1.8° per step. For the experimental test, it tested within 100 steps which is the 180° range. In this work, if the sound is coming from 0°, 30°, and 60°, the motor will rotate 90°, 60°, and 30° respectively towards sensor 3. If the sound is coming exactly 90°, the motor will not rotate. While if the sound is coming from 120°, 150°, and 180°, the motor will rotate 30°, 60°, and 90° respectively towards sensor 1.

2.4 Hardware Implementation and Experimental Setup

Figure 7 shows the schematic diagram of a hardware implementation for this work. Two sound sensors act as input, Arduino Uno acts as the microcontroller, the output is the stepper motor, and the motor shield is a driver for the stepper motor.

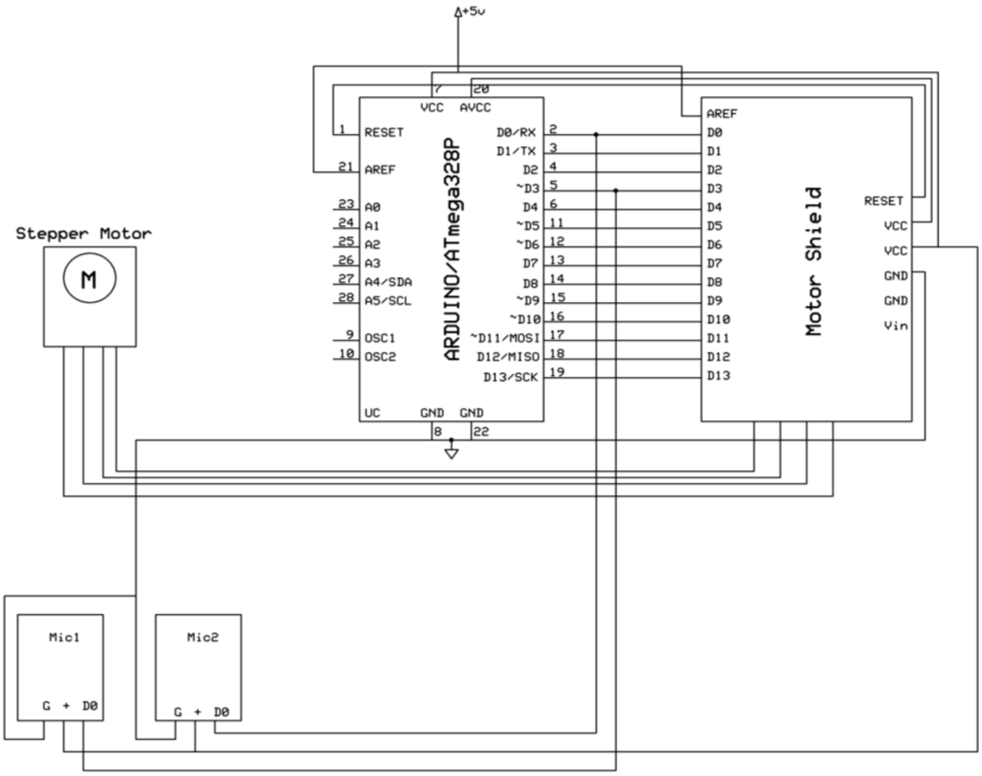


Fig. 7. Schematic diagram

ATmega328P-based Arduino UNO is a microcontroller board that contains 14 digital I/O pins (of which 6 may be used as PWM outputs), 6 analogue inputs, a ceramic resonator operating at 16 MHz, a USB connection, a power connector, an ICSP header, and a reset button. It includes everything required to support the microcontroller; simply connect it to a computer through a USB connection to power it on. For this work, all of the pins are used to attach to the motor shield excluding the analogue pin.

The sound sensor is used as a digital signal that sends a signal when detecting any sound, for this work is 90 dB which is the amplitude is more than 2.5V. This is confirmed by experimenting while monitoring the value of threshold voltage that can give the signal for the sound sensor to receive it. The sound sensor detected the signal and sent it to the microcontroller to calculate the time difference for both sensors as well as calculate the angle of arrival.

The Motor Shield is a motor driver module that allows using Arduino to control the motor's speed and direction. It can drive two DC motors or a step motor thanks to the Dual Full-Bridge Drive Chip L298.

A stepper motor is an electric motor whose fundamental feature is that its shaft rotates in steps, or by progressing by a fixed number of degrees. This property is gained due to the internal construction of the motor, and it allows one to know the precise angular position of the shaft by just counting the number of steps taken, without the use of a sensor.

Figure 8 shows the flowchart of the hardware implementation for this work. The sound source localization using the TDOA method will start when a clapping sound is applied at a 5cm and 10cm distance to the sensor. If the sensor receives sound with an amplitude greater than 2.5V, the amplitude and time for each sensor will be recorded. By that, the time difference will be calculated as well as the angle of arrival. Then, with the value of the angle, the stepper motor will rotate. If the motor rotates according to the actual sound source angle, the angle will record but if not, it will return to the process of applying clapping sound.

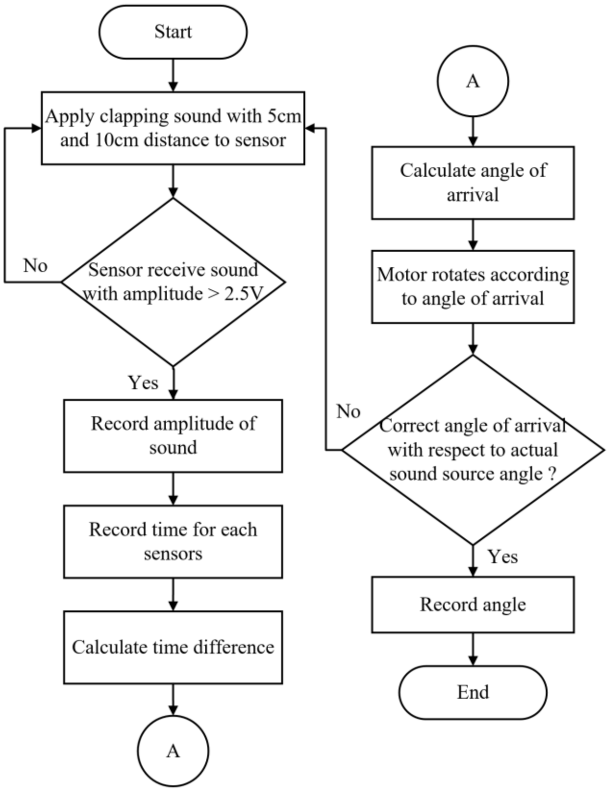
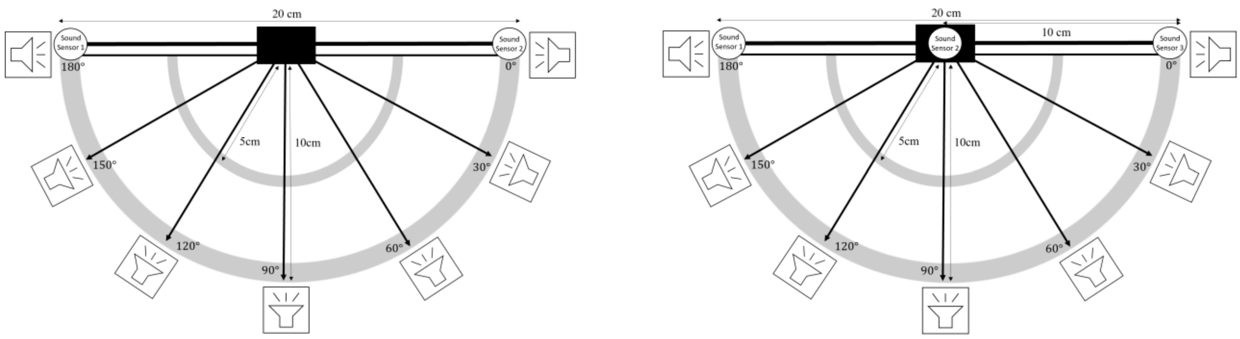


Fig. 8. Flowchart of the system

In this experiment, two sensors are used because to do a TDOA method, at least two sensors are needed. The experimental setup for two sound sensors had been illustrated in Figure 9(a). The distance between each sensor is 20cm because if the sensor is placed too closed, the possibility for the sensor to receive almost the same signal is high and it will make almost no difference in time among the sensors. Meanwhile, for the distance from the sound source to the sensor is 5cm and 10cm. The range of 5cm to 10cm is used because if the clapping sound produced more than 10cm, the amplitude of the sound signal received is less than 2.5V, and if this happened, the sensor will not able to come out with ‘HIGH’ digital out unless if the sound produced is beside than clapping, it will be different. In this case, as long as the sound produced is more than 2.5V, the sensor will able to come out with ‘HIGH’ digital out.

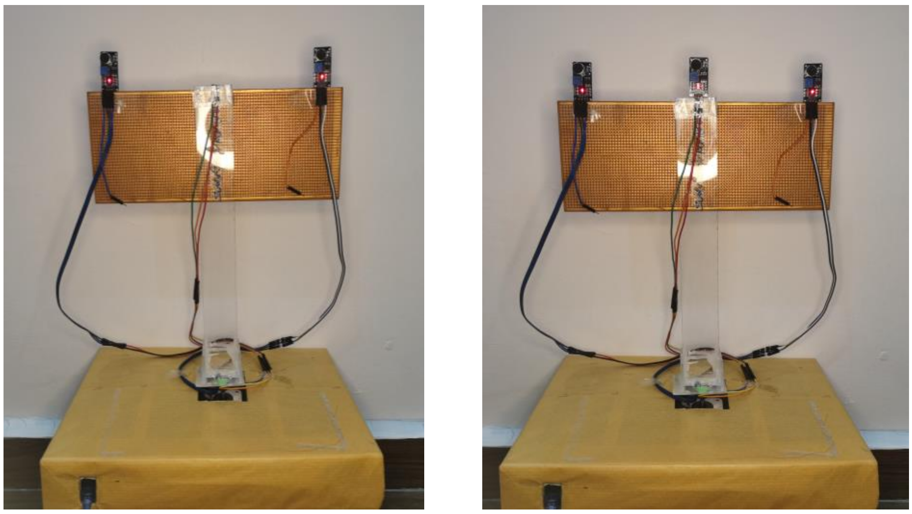
In this experiment, three sensors are used because, with the constant distance of the sound source which is 5cm and 10cm, it will be too close if more than three sensors are used. As mentioned, if the sensor is placed too closed, the possibility for the sensor to receive almost same signal is high and it will make almost no time difference among the sensors. Figure 9(b) shows the experimental setup for three sensors. This setup had been held to compare the time difference of arrival for three sensors. The distance between each of the sensors is 10cm. In this experiment, stepper motor control is not included, instead, the angle had been calculated manually based on Eq. (1) and Eq. (2).



(a) (b)

Fig. 9. Experimental setup for (a) two sound sensors and (b) three sound sensors

Figure 10 shows the prototype that had been designed for this work. There are a few things that had been considered while designing the prototype including the distance between each sensor as well as the sound source, the angle from the sound source, the amplitude of sound, and the number of microphones. As for the distance, the distance between each sensor is set to 20cm.



(a) (b)

Fig. 10. Prototype of the project (a) 2 sound sensors (b) 3 sound sensors

This is to prevent overlapping of the sound since the sensor that has been used is an Electret Condenser Microphone (ECM) type of microphone, and the microphone is the omnidirectional type which will obtain sound from all sides or directions of the microphone as shown in Figure 11. As for the distance between sensor and source is 5cm and 10cm.

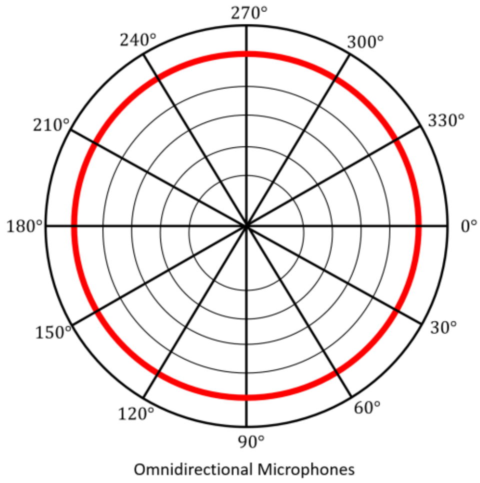


Fig. 11. Polar charts illustrating pick-up sounds from omnidirectional microphones

The angle of the source that had been chosen is from 0 to 180°. The sound will be made at 0°, 30°, 60°, 90°, 120°, 150° and 180°. For the amplitude of sound, since the voltage supply is 5V, the amplitude will be half of the voltage supply which is 2.5V. The number of microphones that had been used for sound localization is two since sound source localization is impossible using a single sensor. Considering a fixed sensor structure, it is a natural conclusion that there should be at least two sensors.

3. Results and Discussion

In this part, the result and analysis that had been obtained are discussed.

3.1 Error of Angle

Table 1 shows the result of the error of the angle obtained with respect to the actual angle of the sound source. The experiment had been held twice. Based on the result, when the clapping sound is produced at whichever angle from 0° to 180°, the first sensor that detects the sound will record the time of arrival then the second sensor will do so too. Then, the time difference for both sensors is calculated. After the angle of arrival had been calculated by the time difference obtained, the stepper motor is rotated according to the calculated angle. As for the clapping sound produced at 0°,30°, and 60°, sensor 2 will receive a signal before sensor 1. The time of arrival for both sensors will record, and the difference will be calculated. Then the angle of arrival will be calculated then the stepper motor will rotate to where the sound is produced which is at 'Right'. Same as when sensor 1 receives a signal before sensor 2 at 120°,150°, and 180°. As for the 90°, both of the sensors received a signal at the same time. Hence, there is no time difference between sensor 1 and sensor 2. The conclusion that can make from the experimental test is within a 5cm distance from the source, the average error is 34.66° while for a 10cm distance from the source is 34.29°. This high inaccuracy occurs due to the type of sound source, the sound duration, and the relative movements of the sound source.

Table 1
The error of angle obtained in comparison to the actual source angle

			Sound source distance of 5cm		Sound source distance of 10cm	
No. of the test	Source angle (°)	Motor rotation angle(°)	Calculated angle with respect to TDOA (°)	|Error of angle| (°)	Calculated angle with respect to TDOA (°)	|Error of angle| (°)
1	0	90	21.84	68.16	43.67	46.33
	30	60	21.84	38.16	43.67	16.33
	60	30	21.84	8.16	87.36	57.36
	90	0	43.67	43.67	43.67	43.67
	120	30	21.84	8.16	43.67	13.67
	150	60	43.67	16.33	43.67	16.33
	180	90	43.67	46.33	43.67	46.33
2	0	90	21.84	68.16	21.84	68.16
	30	60	21.84	38.16	21.84	38.16
	60	30	21.84	8.16	43.67	13.67
	90	0	21.84	21.84	43.67	43.67
	120	30	43.67	13.67	43.67	13.67
	150	60	21.84	38.16	43.67	16.33
	180	90	21.84	68.16	43.67	46.33

3.2 Time Difference of Arrival

Table 2 shows the time difference of arrival for three microphones. For 0°, 30°, and 60°, the sound produced is near to sensor 3 as illustrated in Figure 11. Based on this, sensor 1 takes longer time to receive the sound signal compare to sensor 2 and sensor 2 takes time to receive the sound signal compare to sensor 3. For 90°, the sound produces is at sensor 2 as in Figure 12. Hence, both sensor 1 and sensor 3 need time to receive the sound signal. For 120°, 150°, and 180°, the sound produced is near sensor 1. Based on this, sensor 3 takes a longer time to receive the sound signal compare to sensor 2 and sensor 2 take time to receive the sound signal compare to sensor 1.

Table 2
TDOA for three sound sensors

The angle of Source (°)	Time difference (µs)				
	Left		Middle	Right	
	t3-t1	t2-t1	t1-t3	t1-t3	t2-t3
0	-	-	-	1068	528
30	-	-	-	1152	576
60	-	-	-	1152	568
90	-	-	-584	-	-
120	1064	528	-	-	-
150	1072	532	-	-	-
180	1156	572	-	-	-

Table 3 shows the result of the time difference of arrival for 3 sound sensors. For 0°,30°, and 60°, the clapping sound that was produced had been detected by sensor 3 then the time of arrival for each sensor was recorded and the time difference of arrival was calculated. The same goes for when the clapping sound was produced at 120°, 150°, and 180°, the sensor 1 detected it first, then sensor 2 and sensor 3 respectively. This experiment aims to compare the error of angle between using two sensors and using three sensors. Based on the result, the mean error that had been obtained is 25.08°. The percentage difference error of angle between two sensors and three sensors shows a reduction of 27.64%. It can be concluded that the more sensor used for sound source localization, the least error of angle that will be obtained.

Table 3
The error of angle between source angle and calculated angle for three sound sensors

Source angle (°)	Motor Rotation angle(°)	Calculated angle TDOA, θ_1 (°)	$\lvert$Error of angle $\theta_1\rvert$ (°)	Calculated angle TDOA, θ_2 (°)	$\lvert$Error of angle $\theta_2\rvert$ (°)	Average θ (°)	Average $\lvert$Error of angle $\theta\rvert$ (°)
0	90	66.32	23.68	64.89	25.11	65.61	24.39
30	60	81.06	21.06	81.06	21.06	81.06	21.06
60	30	81.06	51.06	76.94	46.94	79.00	49.00
90	0	-30.05	30.05	0	0	-30.05	30.05
120	30	65.84	35.84	64.89	34.89	65.37	35.37
150	60	66.82	6.82	65.84	5.84	66.33	6.33
180	90	82.42	7.58	78.81	11.19	80.62	9.385

3.3 Stepper Motor Rotation

Table 4 shows the result for the angle motor rotation throughout the experiment. The number of tests is two and the angle that had been chosen are from 0° to 180° and the distances between the sound source and sensors are 5cm and 10cm. From the result, it can be concluded that the motor will rotate closer to where the sound was produced even though the rotation is not the same as the angle of the sound source.

Table 4
The angle of stepper motor rotation

No. of tests	Angle of Source	Angle motor rotation		
		The direction of motor rotation	5cm	10cm
1	0°	Towards Sensor 3	30°	90°
	30°	Towards Sensor 3	30°	90°
	60°	Towards Sensor 3	30°	120°
	90°	-	90°	90°
	120°	Towards Sensor 1	30°	90°
	150°	Towards Sensor 1	90°	90°
	180°	Towards Sensor 1	90°	90°
2	0°	Towards Sensor 3	30°	30°
	30°	Towards Sensor 3	30°	30°
	60°	Towards Sensor 3	30°	90°
	90°	-	30°	90°
	120°	Towards Sensor 1	90°	90°
	150°	Towards Sensor 1	30°	90°
	180°	Towards Sensor 1	30°	90°

4. Conclusion

In conclusion, a microcontroller-based sound source localization system using the TDOA method has been developed and is capable of comparing the time difference received by two and three sound sensors with respect to the distance of sound as well as the angle accuracy of the sound localization through TDOA with the real sound location. The mean error of the angle obtained within a 5cm distance from the source is 34.66° while for a 10cm distance from the source is 34.29°. The percentage difference in angle inaccuracy between two and three sensors achieves a reduction of 27.64%. It may be stated that the more sensors used for sound source localization, the less error is achieved. For future work, instead of using the sound sensor with Electret Condenser Microphone (ECM), Micro-Electromechanical System (MEMS) can be used as a signal receiver because compared to ECM, MEMS microphones are less sensitive to temperature, vibrations, and mechanical shocks. It may reduce the inaccuracy of the sound source localization

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