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Acoustic Source Localisation by Direction of Arrival Estimation and Microphone Arrays

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ABSTRACT: The primary goal of the project is to prove that passive location fixing is feasible over a distributed wireless network which can form basis for 'low cost' solution for security applications where any acoustic activity can be detected by multiple transducers deployed along the area of interest. The microphone grid serves as a multi-transducer network to detect sound disturbance in the area of interest. The outputs from the various transducers are sent to a central processing unit which post processing sends accurate location of the source. The presented work encompasses the design and implementation of the generic platform hardware (nodes), a simple but effective way to achieve time synchronized acoustic data from various nodes connected over RF transceivers and to achieve passive localization by visual representation of multi transducer nodal information.

KEYWORDS: Acoustic source, source localization, Dominant Frequency Selection, Direction of arrival, time difference of arrival

I.INTRODUCTION

The main aim is to locate the origin of disturbance or sound by passive localization which consists of microphone grid array and a wireless sensor network. The sound disturbance is detected by the sensor network which can be 2D or 3D depending on the application. The outputs from the various transducers are sent to a central processing unit which after post processing sends accurate location of the source.
A 2D grid[1] consisting of five nodes (four at four corners and one at the centre of the square) is designed and placed in the area under surveillance. The node consists of the condenser microphone, amplifier circuit and transmitter module. The transceiver module used herein is NRF24L01[2].The transmitted signal is sent to processing unit which compares and acquires a specific range of frequencies categorized under the disturbing sources. When the frequency crosses a certain threshold the data is transferred to the main localization software. DFSE[3][6] (Dominant Frequency Selection) algorithm is used to detect the disturbance. Direction of arrival (DOA) estimation using microphone arrays is to use the phase information present in signals from microphones that are spatially separated. DFSE uses the phase difference between the Fourier transformed signals to estimate the direction of arrival.[3][4] (DOA). This method is based on simply locating the maximum amplitude from each of the Fourier transformed signals and thereby deriving the source location by solving the set of non-linear least squares equations. For any pair of microphones, the surface on which the time difference of arrival (TDOA) is constant is a hyperboloid of two sheets. Acoustic source localization algorithms typically exploit this fact by grouping all microphones into pairs, estimating the TDOA of each pair, then finding the point where all associated hyperboloids most nearly intersect. The coordinates of this point is given as our desired location to the concerned officials where they can view the entire mapped area under surveillance on a visual platform and take necessary action. Use of both closed-form solutions and iterative techniques is done to solve for the source location.

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II.RELATED WORK

The technical paper referred has listed their motivation as to implementing a safe and precise method to help the soldiers at the border. The idea is completely remarkable and caters to all our required needs. This is a first project of its kind specially designed to solve the problem faced by officers to locate the cause of a disturbance and take immediate action. The research paper has described in accurate detail all the steps required to follow for the implementation of the design. Being a first step in finding a solution for the localization issue, it is very basic and simple and does not contain any advanced technology in its field. Here the aim is to simply study the research and implement it on a small scale basis for demonstration purposes. This process can help us learn sound localization and study the system in detail so that we can apply the same principle to other domains for e.g., RF or light.

III.SYSTEM IMPLEMENTATION

Hardware Implementation:
The spatial location of a sound source can be determined based onmultiple observations of the emitted sound signal. This can be realised using multiple sensors, which detect a signal emitted by the source that is to be localized, whose signals are sensed by the sensor. The data obtained from the sensors is analog data. For further processing it is converted into digital format using ADC of microcontroller. The digital data is wirelessly transmitted to the master node using transmitter on the slave nodes. The data undergoes signal processing at the master node.[8][9]

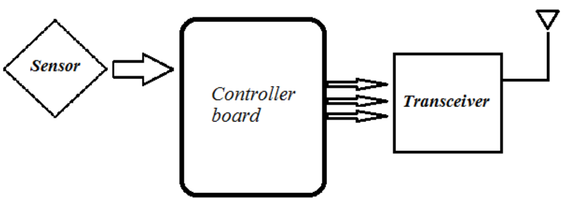

Fig: Sensor node[4]

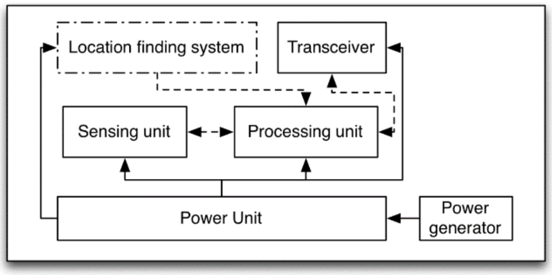

Fig: Control unit[5]

The transceiver forms a link between two microcontrollers located at the two nodes, here between each sensor and control unit. The analog data from sensor is mapped to digital values when fed to the Tx-Rx at the node, then data is easily transmitted to another receiver in vicinity. The addressing of each node is necessary for successful communication.[6]

SOFTWARE IMPLEMENTATION

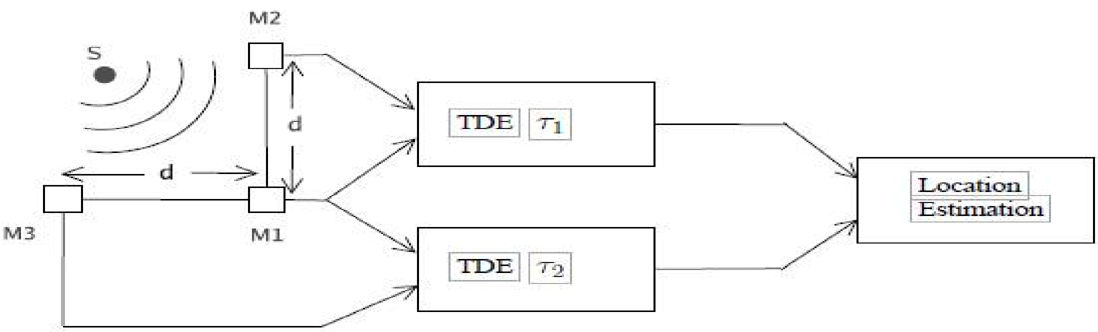

Fig 1: A two stage algorithm for sound source localization[6]

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Sound source localization is a two-step problem:

1. First the signal received by several microphones is processed to obtain information about the time-delay between pairs of microphones. We use the DFSE method for estimating the time-delay which is based on dominant frequencies of received signals.
2. The estimated time-delays for pairs of microphones can be used for getting the location of the sound source.[3][7]

STEP I

The measurement system is constructed by three microphones arranged in 'L' fashion. Three band pass filters are used to extract the signals in the desired frequency range. A personal computer is used to capture the filtered signals, then for algorithm implementation for source location estimation.

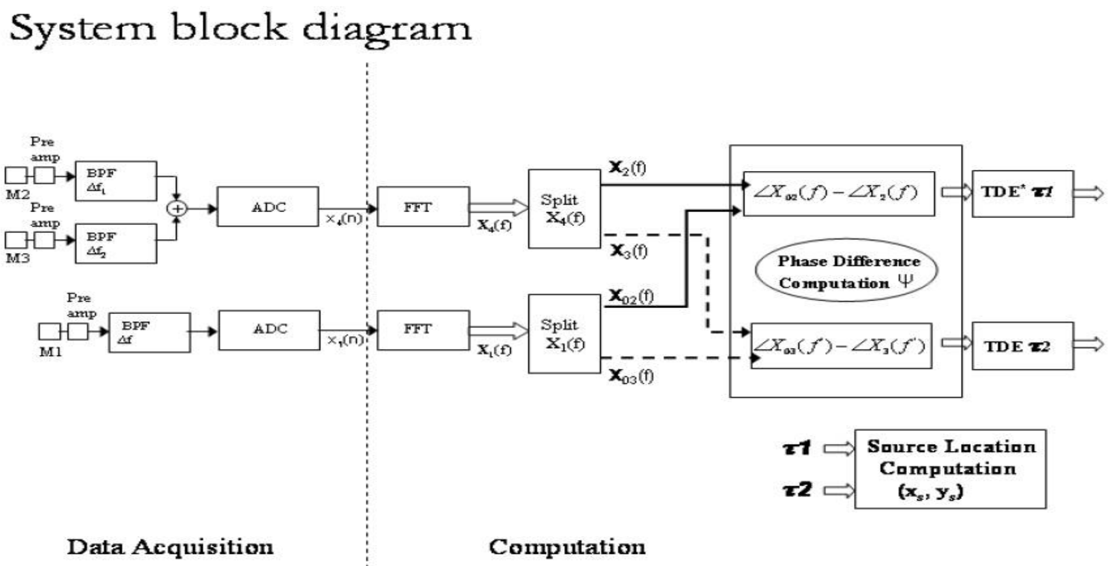

Fig 2: TDOA Estimation[3]

A system block diagram of the proposed method is shown in the Fig. The output signals x1 (n), x2 (n) and x3 (n) of the microphones M1, M2 and M3 are band pass filtered. Signal x2 from M2 is band pass filtered in frequency range Δf1 and the signal x3 from M3 is band pass filtered in the frequency range Δf2. Denote these band pass filtered signals as bx2 and bx3.

That is x4 (n) = bx2 (n)+bx3 (n). Signal x1 (n) from M1 is band pass filtered in the range Δf. We keep Δf as sum of Δf1 and Δf2. The captured signals were post-processed inside the PC.[7]

Fourier transforms of the signals can be represented as

$$x1\,(n) \longleftrightarrow X1\,(f) = |X1\,(f)|\, e^{j\arg[X1(f)]} \quad \text{..................eqn.(1)}$$

and

$$x4\,(n) \longleftrightarrow X4\,(f) = |X4\,(f)|e^{j\arg[X4(f)]} \quad \text{.................eqn.(2)}$$

In the frequency domain X4 (f) is de-multiplexed to obtain X2 (f) and X3 (f).From the amplitude spectrum |X2 (f)| and the phase spectrum ∠X2 (f), the maximumamplitude frequency component |X2 (fm)| is determined.

|X2(fm)| = max{|X2 (f1)|, ..., |X2 (fN1)|}…..where f1, ..., fN1 belong to frequency range Δf1.

Similarly, the maximum amplitude frequency component |X3(fm)| is determined.

Set a threshold power and pick up the frequency components that are above the threshold.

In the frequency domain X2 (f), X02 (f) and X3 (f), X03 (f) are all complex.

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The phase differences between the microphone pairs are calculated by

$$\Psi 1 = \angle X2\ (f) - \angle X02\ (f) \quad \ldots\ldots\ldots\ldots\text{eqn.(3)}$$
$$\Psi 2 = \angle X3\ (f) - \angle X03\ (f) \quad \ldots\ldots\ldots\ldots\text{eqn.(4)}$$

The time-delay of the sound arrival gives the path difference that defines a hyperbola on one branch of which the emitter must be located.
The standard equation of a hyperbola centred at the origin is given by:

$$\frac{x^2}{a^2} - \frac{y^2}{b^2} = 1 \ldots\ldots\ldots\ldots\text{eqn.(5)}$$

Let (x, y) be the source location and (Xi, Yi) be the known location of the ithmicrophone. The squared range difference between the source 'S' and the ithmicrophone is given as:

$$R_i = \sqrt{(X_i - x)^2 + (Y_i - y)^2} \ldots\ldots\ldots\ldots\text{eqn.(6)}$$

Chan's method [8] for a three microphone system (M = 3), producing two TDOA's,x and y can be solved in terms of R1 from the above equation for i = 1.
The solution isin the form of

$$\begin{bmatrix} X \\ Y \end{bmatrix} = - \begin{bmatrix} x_{2,1} & y_{2,1} \\ x_{3,1} & y_{3,1} \end{bmatrix}^{-1} \text{X} \left\{ \begin{bmatrix} r_{2,1} \\ r_{3,1} \end{bmatrix} r1 + 0.5 \begin{bmatrix} r^2{}_{2,1} - K_2 + K_1 \\ r^2{}_{3,1} - K_3 + K_1 \end{bmatrix} \right\} \ldots\ldots\ldots\ldots\text{eqn.(7)}$$

Where, $K_1 = X_1{}^2 + Y_1{}^2$
$K_2 = X_2{}^2 + Y_2{}^2$
$K_1 = X_3{}^2 + Y_3{}^2$

When the equation (6) is inserted into equation above, with i = 1, a quadraticequation in terms of r1 is produced. Substituting the positive root back into the equation (7) results in the final solution.

IV.RESULTS AND DISCUSSION

A very simple experiment was done with a set of four microphones arranged in a square grid fashion. Then examining the performance, compare the amplitude with the look-up table values.

Sensor response:
The sensor value is read and serially plotted on the software. With a disturbance in its range the value jumps to high peaks from the noise floor.

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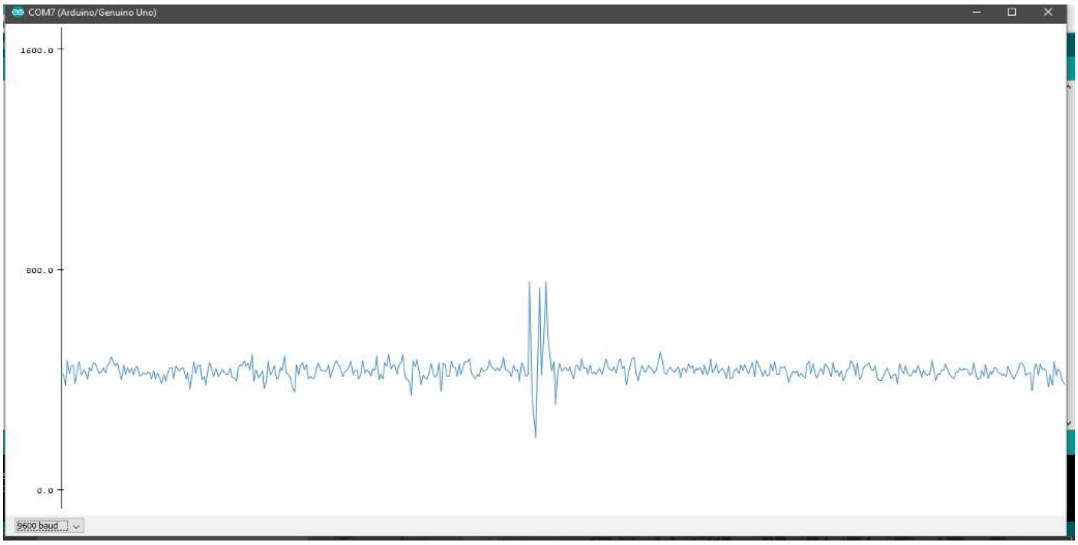

Fig 3: Sensor response

Localisation:

Each sensor node comprises of its own transceiver. The base station transmitter broadcasts its availability to the transceivers on the sensor nodes. It then establishes a connection with any one of the sensor node at a time. The analog data at the sensor's output is transmitted using NRF24l01 transceiver to the base station receiver. After the base station receives data from the sensor node, it then sends an acknowledgement to the corresponding sensor node and the connection with that sensor node is terminated. Further it communicates with other sensor nodes and the process repeats. After the base station receives data from all the sensor nodes, the source was positioned.

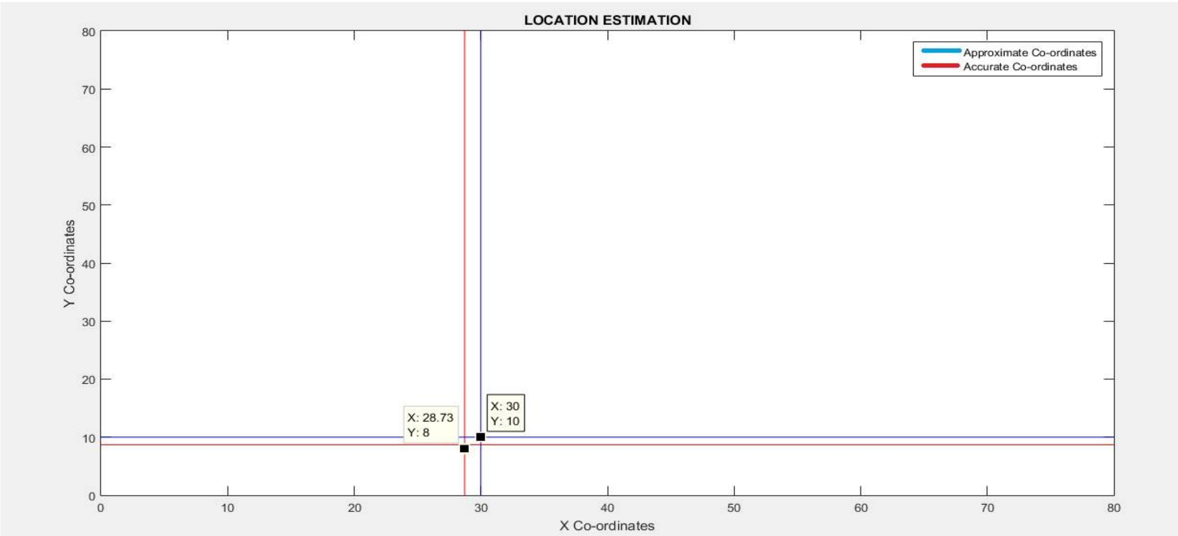

Fig 4: Location of the sound source

The co-ordinates of the sound source (disturbance) is shown in Fig 4. There are two points which are formed by the intersection of the red and blue lines. The blue line intersection gives the approximate point and the red lines intersection gives the actual point. The error can be found by the mean square method.

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Sr. No.	True Position	Estimated Position	% Error
1	(30,10)	(30,12.39)	2.62
2	(60,20)	(57.67,19.69)	3.6
3	(40,30)	(43.45,29.7)	5.26
4	(20,40)	(18.18,39.01)	3.75
5	(40,40)	(37.30,44.35)	2.47
6	(70,50)	(73.53,50.85)	3.92
7	(30,50)	(31.99,53.66)	7.15
8	(60,70)	(58.6,72.29)	0.093
9	(60,50)	(60.78,52.78)	2.13
10	(40,70)	(40.94,72.61)	3.26

Table 1: Calculation of mean square error

The source is positioned at various places in 2-D space. The table 1 produces the true, estimated source locations of the sound source. Nonlinear optimization method is utilized for solving the nonlinear equations. The mean square error is calculated for each set of readings. Fig. 4 shows clearly the true and estimated source positions.

V. CONCLUSION

It has been proved that location fixing over acoustic domain can be achieved over wireless medium with reasonable degree of accuracy. Therefore, the viability of such a product with intended applications is now very clear. The basic limitation of excessive requirement of human input has to be removed to realize as a good prototype. The prototype is fully functional, but the functionality is limited due to generic nature and many design compromises due to cost factor. It is proved that passive acoustic localization is feasible approach for military and security applications like border monitoring.

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