

A 3D Localization Method of Multiple Acoustic Sources Using Microphone Array

KwangMyong Kim & NamChol Yu

Faculty of Electronics, Kim Chaek University of Technology, Pyongyang, DPR Korea

Received: 30.08.2025 | Accepted: 02.10.2025 | Published: 22.10.2025

*Corresponding Author: NamChol Yu

DOI: [10.5281/zenodo.17418052](https://doi.org/10.5281/zenodo.17418052)

Abstract

Original Research Article

Localization technology of acoustic source is widely used in detection and wireless communications.

In some references, many researchers have addressed the methods of 1D and 2D estimating the DOA far field multiple acoustic sources. However, 3D localization method that includes distance information is poorly introduced. That's why because in near field, the distance can be directly calculated but in far field, it can never. And also the distance information can be indirectly calculated relative easily in case that acoustic source is one or they are narrowband acoustic sources. However, it is difficult to obtain the distance information of wideband multiple sources. This paper presented a new method to estimate distance information of narrowband multiple acoustic sources. The simulation examples indicated the suggesting method can be accurately estimated locations of multiple sound sources.

Keywords: multiple acoustic sources, 3D localization, DOA, array.

Copyright © 2025 The Author(s). This is an open-access article distributed under the terms of the Creative Commons Attribution-NonCommercial 4.0 International License (CC BY-NC 4.0).

1. Introduction

The localization technology of using sensor array is widely used to all sorts of detection of target and wireless communication on the ground, sea and space.

In general, the acoustic source localization techniques can be classified into three categories^[5,6], namely, steered beamformer, super-resolution spectral estimation, and time-delay estimation.

The methods based on subspace used the subspace of signal and noise and they are separated from MUSIC, ESPRIT and etc.^[1,7] These methods are being used widely because estimation performance is relatively high and calculating magnitude is lower than beamforming method.

This paper presented a method to estimate the position including distance information of long distance multiple acoustic source correctly. In the literatures, DOA of near and far field acoustic source

based on subspace and frequency joint estimation are widely introduced.^[2,3,4]

Now many researchers have addressed the methods of 1D and 2D estimating the DOA far field wideband multiple acoustic sources. However, 3D localization method that includes distance information is poorly introduced.

In near field, the distance can be directly calculated but in far field, it can never.^[8]

And also the distance information can be indirectly calculated relative easily in case that acoustic source is one or they are narrowband acoustic sources. However, it is difficult to obtain the distance information of wideband multiple sources.^[9,10]

In order to estimate the location of far field wide band multiple acoustic sources, this paper used two L-shape arrays.

And first, in each array, using DOA and frequency



joint estimation, 2D DOA and frequency of signals were estimated.

Next, locations of 3D acoustic sources were estimated based on frequency and DOA information.

The result of simulation examples on Matlab shows that high estimation capacity can be obtained when acoustic source noise is relatively high and the space between 2 arrays are settled reasonably.

The suggested method can be used widely in detection of target and wireless communication such as radar and sonar.

2. STRUCTURE OF L-TYPE ARRAY AND SIGNAL RECEIVING MODEL

When the distance between microphone array

and sound source is further, amplitude of signal that is received between microphones is very low and it can be omitted and the signal model can be regarded as plane model. Under the ideal situation of far field, signals that microphone received doesn't contain information for distance, but contain information for DOA and frequency. L-type microphone array model is as following.

In Fig.1 L-type array is consisted with unique linear array X, Y the array number of which is M for each on X axe and Y axe. Microphone on original point $(0, 0)$ is used in both arrays, is reference microphone. The frequency, pitching angle and direction angle of acoustic source are f, θ, φ for each and the distance neighboring each is d .

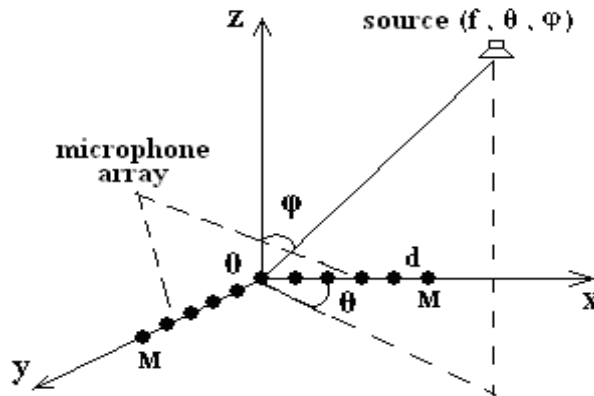


Fig.1 The signal model of L-shaped microphone array

Let's say distance d between array elements of linear array X, Y is shorter than half wavelength, the average value of output noise of array is 0, and it is white Gaussian noise of statistic independence the

variance of which is σ^2 and there is no other relationship with signal source. Suppose k signal source of statistic independence reach at this array, the receiving signal of array is as following.

$$F(t) = \begin{bmatrix} f_1(t) \\ \vdots \\ f_{M+N-1}(t) \end{bmatrix} = \begin{bmatrix} 1 & \dots & 1 \\ e^{j2\pi f_1 \tau_{21}} & \dots & e^{j2\pi f_k \tau_{2k}} \\ \vdots & \dots & \vdots \\ e^{j2\pi f_1 \tau_{M+N-1,1}} & \dots & e^{j2\pi f_k \tau_{M+N-1,k}} \end{bmatrix} \begin{bmatrix} S_1(t) \\ S_2(t) \\ \vdots \\ S_K(t) \end{bmatrix} + \begin{bmatrix} n_1(t) \\ n_2(t) \\ \vdots \\ n_{M+N-1}(t) \end{bmatrix} \quad (1)$$

$$= [a(f_1, \theta_1, \varphi_1) \dots a(f_k, \theta_k, \varphi_k) \dots a(f_K, \theta_K, \varphi_K)] [S_1(t), \dots, S_k(t), \dots, S_K(t)]^T + [n_1(t) n_2(t) \dots n_{M+N-1}(t)]^T = AS + N$$

f_k is k 'th central frequency of narrowband source, τ_{ik} is relative delay-time when the k 'th signal source is radiated to i 'th array element and n_i is noise that is received by array element. Among them signal vector received by array X and Y is as following.

$$\begin{aligned} X(t) &= A_x S(t) + N_x(t) \\ Y(t) &= A_y S(t) + N_y(t) \end{aligned} \quad (2)$$

where $S(t)=[s_1(t), \dots, s_k(t), \dots, s_K(t)]^T$ is signal phasor; $A_x=[a_x(f_1, \theta_1, \varphi_1), \dots, a_x(f_K, \theta_K, \varphi_K)]$; $A_y=[a_y(f_1, \theta_1, \varphi_1), \dots, a_y(f_K, \theta_K, \varphi_K)]$

3. JOINT ANGLE-FREQUENCY ESTIMATION

Nowadays joint estimate of multiparameter is being concerned and the study about this is being taken deeply and in paper [7, 8] suggests some methods for joint estimate. For the 2-dimensional joint angle-frequency estimation, in this paper based on ESPRT rule a method for joint estimate is suggested.

In the above L-type microphone array, from the mathematical model of array, we can see that receiving signal vector of array element of sub array X of which the first part is $M-1$ and $M-1$ at the back is $X_1(t)$ and $X_2(t)$ as for each and the if the receiving signal vector of array element of sub array Y of which the first part is $M-1$ and $M-1$ at the back, is $Y_1(t)$ and $Y_2(t)$ as for each, the formula is formed as following

$$X_1(t) = A_1(f, \theta, \varphi) S(t) + N_X(t) \quad (3)$$

$$X_2(t) = A_1(f, \theta, \varphi) \Phi_X S(t) + N_X(t) \quad (4)$$

$$Y_1(t) = A_1(f, \theta, \varphi) \Phi_Y S(t) + N_Y(t) \quad (5)$$

$$Y_2(t) = A_1(f, \theta, \varphi) \Phi S(t) + N_Y(t) \quad (6)$$

where

$$\Phi_X = \text{diag}\{e^{-j2\pi f_1 d \cos \theta_1 \sin \varphi_1 / c}, \dots, e^{-j2\pi f_N d \cos \theta_N \sin \varphi_N / c}\} \quad (7)$$

$$\Phi_Y = \text{diag}\{e^{-j2\pi f_1 d \sin \theta_1 \sin \varphi_1 / c}, \dots, e^{-j2\pi f_N d \sin \theta_N \sin \varphi_N / c}\} \quad (8)$$

$$\Phi = \text{diag}\{e^{-j2\pi f_1 d \tau_1}, \dots, e^{-j2\pi f_N d \tau_N}\} = \Phi_X \Phi_Y \quad (9)$$

Where $\tau_i = \frac{1}{c}(\cos \theta_i \sin \varphi_i + \sin \theta_i \sin \varphi_i)$, A_1 is directional matrix of X_1 submatrix.

In order to calculate frequency of signal, delay time τ has been added to receiving signal of array as fig 2 shows and it's presumed as $0 < 2\tau < 1/\max(f_i)$.

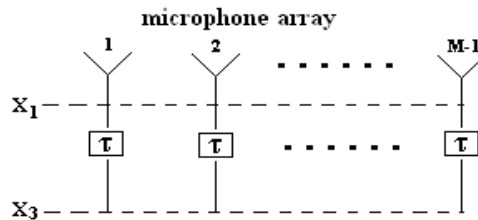


Fig.2 The received signal mode with delay

If one time delay is added to output of $M-1$ first part of array element of x sub array and the data that is calculated is X_3 , according the knowledge of space spectrum we can get the formula as following.

$$X_3(t) = A_1(f, \theta, \varphi) \Phi_\tau S(t) + N_X(t) \quad (10)$$

Where

$$\Phi_\tau = \text{diag}\{e^{-j2\pi f_i \tau}, \dots, e^{-j2\pi f_N \tau}\} \quad (11)$$

From the principle of rotation-invariant subspace, if space noise is white noise, auto-covariance matrix R_{X11} of $X_1(t)$, cross-covariance matrix R_{Y11} , of $X_1(t)$ and $X_2(t)$ auto-covariance matrix R_{Y11} of $Y_1(t)$ and cross-covariance matrix R_{Y21} of $Y_2(t)$ can be formed and as for 4 covariance matrixes that are mentioned above, if the noise is removed, the result will be as following.

$$R_{X11} - \sigma^2 I = A_1(f, \theta, \varphi) R_S A_1^H(f, \theta, \varphi) = C_{X11} \quad (12)$$

$$R_{X21} - \sigma^2 I = A_1(f, \theta, \varphi) \Phi_\tau R_S A_1^H(f, \theta, \varphi) = C_{X21} \quad (13)$$

$$R_{YX} - \sigma^2 I = A_1(f, \theta, \varphi) \Phi_Y R_S A_1^H(f, \theta, \varphi) = C_{YX} \quad (14)$$

Also cross-covariance matrix R_{X31} of sub array X_1 and X_3 can be formed as following.

$$R_{X31} = A_1(f, \theta, \varphi) \Phi_\tau R_S A_1^H(f, \theta, \varphi) + \sigma^2 I \quad (15)$$

If space noise is presumed as white Gaussian noise, the array after noise removal will be as following

$$R_{X31} - \sigma^2 I = A_1(f, \theta, \varphi) \Phi_\tau R_S A_1^H(f, \theta, \varphi) = C_{X31} \quad (16)$$

So we can get 3 arrays' generalized eigenvalue of $\{C_{X11}, C_{X31}\}, \{C_{X11}, C_{X21}\}, \{C_{X11}, C_{YX}\}$.

$$v_{f_i} = \exp(-j2\pi f_i t)$$

$$v_{X_i} = \exp(-j2\pi f_i d \cos \theta_i \sin \varphi_i / c)$$

$$v_{Y_i} = \exp(-j2\pi f_i d \sin \theta_i \sin \varphi_i / c)$$

Where $i=1, 2, \dots, K$, K is the number of signal source.

If we connect 3 formulas above, we can get frequency of signal, direction angle and pitching angle.

$$f_i = \left| \frac{\text{angle}(v_{f_i})}{2\pi\tau} \right| \quad (17)$$

$$\theta_i = \arctan \left\{ \frac{\text{angle}(v_{Y_i})}{\text{angle}(v_{X_i})} \right\} \quad (18)$$

$$\varphi_i = \arcsin \left\{ \frac{c}{2\pi f_i d} \sqrt{(\text{angle}(v_{Y_i}))^2 + (\text{angle}(v_{X_i}))^2} \right\} \quad (19)$$

4. THE METHOD FOR CALCULATING MULTIPLE ACOUSTIC SOURCE LOCALIZATION

In this paper model for calculating multiple acoustic source localization is as following.

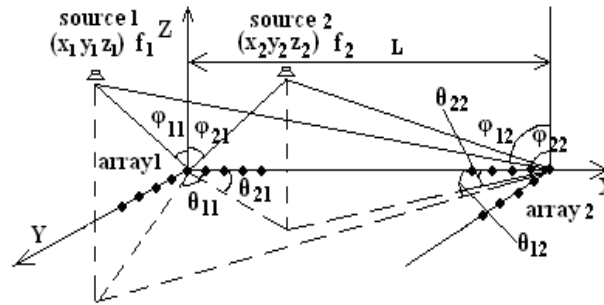


Fig.3 The proposed 3-D distance measurement model

As fig 3 shows, 2 L-type microphone arrays are located on X axis of measurement space and distance between two microphones is L , direction angle and pitching angle for array 1 of i 'th signal source are θ_{i1} , ϕ_{i1} and direction angle and pitching angle for array 2 of i 'th signal source are θ_{i2} , ϕ_{i2} .

When reference microphone of array 1 is located on the original point(0, 0, 0), if a pair of direction angles(θ_{i1} , θ_{i2}) and a pair of pitching angle(ϕ_{i1} , ϕ_{i2}) are calculated, coordinate of sound source (x_i , y_i , z_i) can be calculated.

In order to do this, if it is presumed that the central frequency of each signal sources is constant and different from each other, angle of arrival and

frequency can be calculated through joint angle-frequency estimation from each arrays.

By comparing frequency, signal sources are recognized and we can get a pair of pitching angle for two L-type array of each signal source.

At this point, frequency value won't be the same due to the deviation in joint estimation, but the sound source, the frequency value of which is minimum between two arrays, can regarded as the unique one and incidence angle of those are defined as a pair of direction angle(θ_{i1} , θ_{i2}) and a pair of pitching angle(ϕ_{i1} , ϕ_{i2}). According to fig 3 if i 'th acoustic source's coordinate is presumed as (x_i , y_i , z_i), a formula as following is formed.

$$\tan q_{i1} = \frac{y_i}{x_i} \quad (20)$$

$$\tan q_{i2} = \frac{y_i}{L - x_i} \quad (21)$$

From (20),(21) the coordinate of signal on XOY surface is as following.

$$\begin{cases} x_i = \frac{L \tan \theta_{i2}}{\tan \theta_{i1} + \tan \theta_{i2}} \\ y_i = x_i \tan \theta_{i1} \end{cases} \quad (22)$$

Next, Z coordinate of signal source must be calculated, but due to the deviation in calculation of two arrays, Z coordinate value, calculated from a pair of pitching angle(ϕ_{i1} , ϕ_{i2}), won't be the same. So the average value of Z coordinate value, calculated from two arrays is defined as Z coordinate of i 'th sound source.

$$z_{i1} = \frac{\sqrt{x_i^2 + y_i^2}}{\tan j_{i1}} \quad (23)$$

$$z_{i2} = \frac{\sqrt{(L - x_i)^2 + y_i^2}}{\tan j_{i2}} \quad (24)$$

$$z_i = (z_{i1} + z_{i2})/2 \quad (25)$$

Here if the distance between two arrays is longer or shorter than measured distance, deviation might be getting more and more. So distance between arrays should be in measured distance range and should be selected in suitable way.

5. SIMULATION RESULTS

(1) Experiment of joint angle-frequency estimation

Like fig 3, microphone array is unique L-type array that is formed with 8*2-1 microphones and central frequency of two narrowband signal source that is about to be measured is 3400 Hz, 3200Hz, for each, direction angles are 30°, 50°, pitching angles

are 25°, 40° and numbers of samples are 512.

Distance between array elements is $C/2f_{\max}$ which means 4cm and the speed of sound wave is 340m/s. In order to see the performance of joint angle-frequency estimation, MATLAB simulation has been taken in various SNR conditions, standard deviation which we get through experiment is average value from 10 times' calculation.

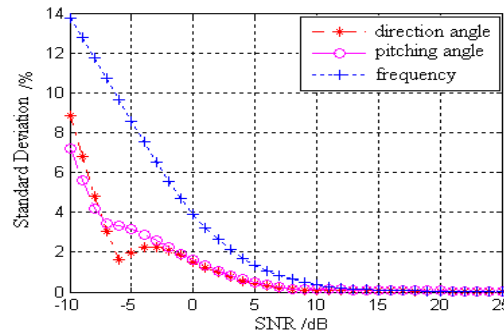


Fig.4 The relationship between estimation deviation of frequency、DOA and SNR

Simulation result value from Fig 4 shows that if SNR is 0dB, standard deviation in calculation of frequency and DOA is lowered to 5%

(2) Experiment of multiple acoustic source localization measurement.

In order to analyze the influence of SNR of the method of multiple acoustic source localization measurement that is suggested in this paper, 2

narrowband signal source are used and central frequency is 3400、3200Hz for each, XYZ coordinate (40m 90m 70m)、(80m 70m 50m) for each, and distance between two arrays is 2m.

Fig 5 shows the relationship between standard deviation in calculation of the suggested method and SNR when SNR changes.

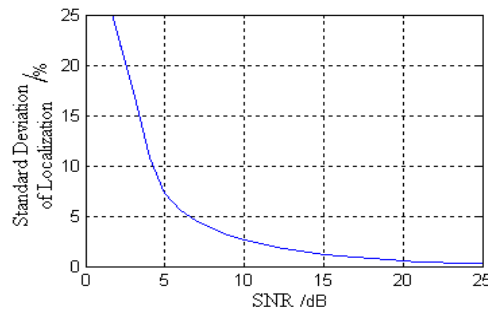


Fig.5 The relationship between estimation deviation of proposed method and SNR

Simulation result shows that when SNR is 7dB, the standard deviation of suggested method is under 5%. In order to analyze the distance relationship between the deviation in coordinate calculation and two arrays, when SNR is 20dB under the same condition

above, 10 experiments are taken by changing the distance between two arrays and standard deviation of calculated coordinates is calculated and the result is as following.

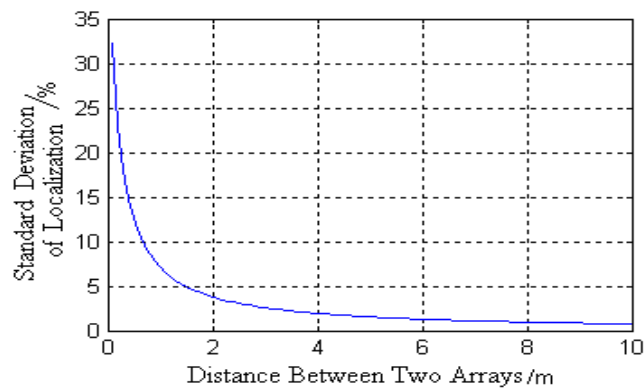


Fig.6 The relationship between estimation difference of proposed method and distance between two arrays

Fig 6 shows that if the distance between two L-type arrays is more than 1.5m, standard deviation in coordinate calculation is lower under 5%. If measurement distance is R and distance between arrays is L, we can get an experience formula as following.

$$R/10 < L < 10R$$

Simulation result above shows that distance between two L-type arrays should be selected suitably to get highly accurate localization calculation value.

localization measurement of far field of multiple acoustic source. Simulation result shows that multiple acoustic source localization can be measured with the standard deviation of which is under 5% by controlling the distance between two L-type arrays.

This method because it recognize the signal source using frequency, can be applied in multiple acoustic source localization. Also if SNR is 7dB, standard deviation is under 5% and it's possible to apply.

6. Conclusion

We have here described the method of

REFERENCES

- [1] N. Kanaya, Y. Iiguni, H. Maeda, 2-D DOA estimation method using Zernike moments, *Signal Process*, 82 (2002) 521 – 526
- [2] A. Olfat, S. Nade, A new signal subspace processing for DOA estimation, *Signal Processing*, 84 (2004) 721 – 728
- [3] J. Verhaevert, E. V. Lil, A. V. Capelle, Direction of arrival (DOA) parameter estimation with the SAGE algorithm, *Signal Processing* 84 (2004) 619 – 629
- [4] J. Huang, N. Ohnishi, Spatial localization of sound sources: Azimuth and elevation estimation, *IEEE Instrumentation and Measurement Technology Conference*, 1998, pp. 330–333.
- [5] M. Wax, T. Kailath, Optimum localization of multiple sources by passive arrays. *IEEE Transactions on Acoustics, Speech and Signal Processing*, 1983, 31(5), pp. 1210–1217.
- [6] T. I. Carlos, C. Olivier, et al, Evaluation of a MUSIC-based Real-time Sound Localization of Multiple Sound Sources in Real Noisy Environments. *IEEE Intelligent Robots and Systems*, 2009. 10, pp. 2027–2032.
- [7] Y. Desen, S. Jie, L. Bosheng, A new near-field source localization algorithm based on generalized ESPRIT, *IEEE Conference on Digital Object Identifier*, 2009, pp. 1115–1120.
- [8] L. Jackson, H. Chien, Frequency and bearing estimation by two-dimensional linear prediction. *IEEE International Conference on ICASSP '79*, 1979, pp. 665–668.
- Q. Xin, G. Lu, et al, Joint frequency and 2-D DOA estimation using pseudocovariance matrices, *International Conference on Digital Object Identifier*, 2009, pp. 398–401.