

LOCATION ESTIMATION OF BROADBAND SOURCE WITH ADAPTED MUSIC ALGORITHM

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ABSTRACT

This study deals with location estimation of the broadband acoustic source in the distributed sensor network. The proposed algorithm is based on the MUSIC (Multiple Signal Classification) algorithm, which solves the localization problem in time domain. The broadband source signal arrives at each sensor with different amplitudes and delay times. Therefore, the signals which are reached at the sensors are the noisy versions of the each other. The collected sensor data is called the observation set, and the cross correlation matrix is calculated by using the observation set. The correlation matrix provides a measure of the similarity between each sensor signal and the relative delay time between the received sensor signals which can be determined by using this data. A reference time point is chosen among the received signals and each time delays are eliminated according to this signal after calculating the cross correlation matrix. Consequently, the source location estimation task is performed by MUSIC algorithm in time domain. Moreover, the simulation results which demonstrate the performance of the method are given in the paper.

Keywords: *Broadband source, location estimation, MUSIC algorithm, sensor network*

INTRODUCTION

Recently, sensor networks are used widely in many applications including source location estimation. Therefore, the efficiency of the algorithms used for processing the collected sensor data and estimation of the source location via the gathered data. Especially acoustic and seismic source signals are unmodulated, real valued broadband, baseband signals. The inherited features of the broadband source signals should be taken into consideration in the proposed solution.

Uptodate there have been several studies for the source location or direction of arrival (DOA) estimation resulting in some suggested approaches. Some of the suggested algorithms are acoustic source position estimation with the particle filter method, the algorithms based on the signal energy measurements and efficient ML (Maximum Likelihood) estimator [1], [2], [3], [4], [5]. Least squares (LS) signal subspace method is another estimation algorithm [6]. In addition, some suboptimal subspace estimation methods such as ESPRIT (Estimation of Parameters by Rotationally Invariant Techniques) and MUSIC (Multiple Signal Classification) are used for the source location or DOA

estimation [7], [8], [9]. However, these methods mainly have been developed for narrowband signals and these methods have been adapted for the broadband signals. Generally, the source signal is processed in frequency domain in order to estimate the location. In some approaches, the frequency spectrum of the signal is divided into subframes do not overlap. Each resulting narrowband signal is processed separately to yield the solution [6]. In another approach, the signal is sampled into a set of narrowband signals. Then, the transformation of the correlation matrix of the observation data into a definite subspace is fulfilled by using the focusing matrix. Finally, the solution is implemented by using the subspace techniques.

MUSIC algorithm is important one of these methods. MUSIC algorithm can not be used directly for the broadband signals effectively. The observation data should be suitable to use the method estimating source location. First of all, one of the sensors is chosen as the reference point. The cross correlation matrix are calculated by using the reference sensor data and the other sensor readings respectively. The resulted observation delays are compensated considering the correlation between each sensor signal. The final solution is made by using MUSIC algorithm.

In this study, firstly the signal model is constructed. Then the cross correlation matrix of the observation data is calculated. After compensating the delay times between the sensor readings the solution of the source location estimation problem is made by using formed data employing MUSIC algorithm in time domain. the source signal is assumed to be deterministic and unknown. There is no need for the signal propagation speed for the solution. The location of the source is estimated effectively with the proposed method.

2. SIGNAL MODEL

Let's suppose that we have a distributed sensor network of M elements (sensors). The source signal is broadband baseband signal and the source is inside or in the close range of the

sensor field. The signal is received by randomly deployed P sensors. The received sensor data is unmodulated, real valued, deterministic and unknown signal. As in many applications we suppose $P > M$, and the sensors are identical and receive signal equally. The propagation distortion and multipath effects are not taken into consideration. We suppose the source signal is propagates omnidirectionally. The sensor outputs are collected during a time frame $t = 0, \dots, L-1$ and therefore the observation data is consisted of the collected data by each sensor. The observation data is given as follows,

$$\mathbf{x}(t) = [x_1(t), \dots, x_p(t)]^T. \quad (1)$$

m th real values source signal, $x_p(t) \in \mathfrak{R}$, received by the p th sensor at time instant t is expressed by the following form;

$$x_p(t) = \sum_{m=1}^M \frac{s_m(t - t_{pm})}{|\mathbf{r}_m - \mathbf{r}_p|} + w_p(t) \quad (2)$$

$t = 0, \dots, L-1$, $p = 1, \dots, P$ and $m = 1, \dots, M$

where

s_m ; propagated m the source signal,

t_{pm} ; time delay between the p th sensor and the m th source,

w_p ; independent, identical distributed (i.i.d.), zero mean, with σ^2 variance white Gaussian noise,

\mathbf{r}_m ; m th source location,

\mathbf{r}_p ; p th sensor location.

Suppose the sensor locations are known and the sensors are static. Hence, the distance between the p th sensor location and m th source location is defined as follows,

$$\mathbf{r}_{pm} = \mathbf{r}_p - \mathbf{r}_m \quad (3)$$

$$r_{pm} = \sqrt{(x_m - x_p)^2 + (y_m - y_p)^2} \quad (4)$$

r_p is not a function of time due to the sensors are static and in the equations it is not dependent on the time. The time delay between the p th sensor and the m th source is described by the equation

$$t_{pm} = \frac{|\mathbf{r}_{pm}|}{V_s} = \frac{|\mathbf{r}_p - \mathbf{r}_m|}{V_s} = \frac{\sqrt{(x_m - x_p)^2 + (y_m - y_p)^2}}{V_s} \quad (5)$$

(x_p, y_p) is the true location of p th sensor, $p = 1, \dots, P$, and (x_m, y_m) is the true position of the m th source, $m = 1, \dots, M$, in cartesian coordinate system in the formulations. V_s is the constant and known propagation speed of the signal (speed of sound in the air is 345 m/s). The source signal arrives at each sensor with different observation delay time. Equation (2) can be rewritten more clearly in the following form:

$$x_p(t) = \sum_{m=1}^M \frac{s_m(t - t_{pm})}{|\mathbf{r}_{pm}|} + w_p(t) \quad (6)$$

where d_{pm} is an element of the position manifold vector and it is a function of spatial position. The elements of the position manifold matrix is written as

$$d_{pm} = \frac{1}{|\mathbf{r}_{pm}|} \quad (7)$$

Position manifold vectors are given as

$$\mathbf{d}_m(x_m, y_m) = [d_{1m}, d_{2m}, \dots, d_{pm}]^T \quad (8)$$

3. METHOD: DELAY TIME ADAPTED MUSIC ALGORITHM

The sensors are deployed in the field randomly and the source is in the near field of the sensor network in the estimation problem at hand. The source emits broadband unmodulated acoustic signal. The signal is received by a sensor with an amplitude and phase delay due to the distance between the source and sensor, and the same source signal is received by another sensor with a different amplitude and phase delay because of the different distance. The relative time delays between the sensors are different from each other. A simple demonstration is given in the figure 1.

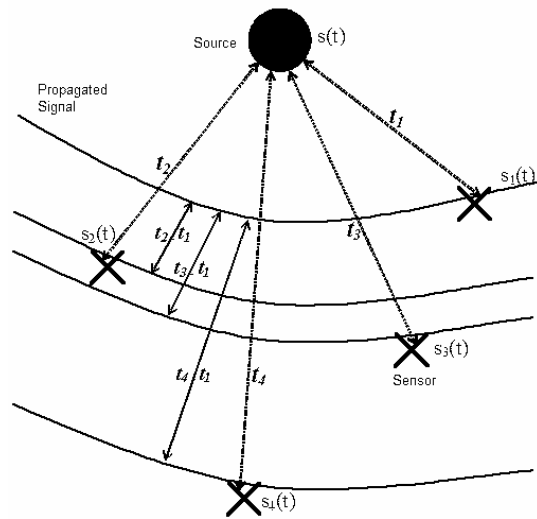


Figure 1. The propagation of the source signal at the distributed sensors field and the relative time delays between the different sensor.

In broadband signal case, different time delays can not be neglected due the specific wideband characteristics of the source signal. The location estimation procedure is difficult in time domain because of these phase differences. In addition, in some cases, the estimation algorithms is used by taking Fourier Transform of the sensor data in order to study in frequency domain.

The sensors collect observation data along the length of a certain time frame before the proposed algorithm is run. The length of the time frame is chosen with such a sufficient length so the source signal reaches at the farthest sensor. The time delays between the sensors can be useful for estimation process due to the source location knowledge is included at the observation time delays. The cross correlation is calculated for each sensor reading in the time domain. For example, one of the sensors is selected as the reference sensor, then the correlation matrix is calculated between the reference sensor reading and the second sensor reading. Then, the cross correlation is calculated between the selected reference sensor observation data and the third sensor data. This process is performed simultaneously for all the sensor data.

A measure of similarity between two signals which are different from each other in amplitude and phase is acquired by calculating the cross

correlation. Two noisy observation data received by different sensors are emitted from the same source. The signals are similar but some differences are occurred due to the different distances between the source and the sensors. Thus, the time delay information is obtained via the correlation matrix. Therefore, the time delays of each signal received by the sensors are compensated as if the signals arrived at each sensors at the same time instant with the reference sensor considering with the information at hand obtained via correlation matrix. We have a new set of data after this step. It is not important that which sensor is selected as the reference sensor point. Then, the solution of the problem is performed with the MUSIC algorithm which is an efficient method for such location estimation problems by using the new data at hand.

The general formulation and the steps of the proposed method to solve the estimation problem are discussed in the following rows. In the case of the first sensor is selected as the reference sensor, the collected sensor data (observations) of P sensors and the subsequent steps are given as below,

$$\begin{aligned} x_1(t) &= \frac{1}{r_{11}} s(t-t_1) + n_1(t) \\ x_2(t) &= \frac{1}{r_{21}} s(t-t_2) + n_2(t) \\ &\vdots \\ x_p(t) &= \frac{1}{r_{p1}} s(t-t_p) + n_p(t) \end{aligned} \Rightarrow$$

by continuing here

$$\begin{aligned} x_1(t+t_1-t_1) &= \frac{1}{r_{11}} s(t-t_1) + n_1(t) \\ x_2(t+t_2-t_1) &= \frac{1}{r_{21}} s(t-t_1) + n_2(t) \\ &\vdots \\ x_p(t+t_p-t_1) &= \frac{1}{r_{p1}} s(t-t_1) + n_p(t) \end{aligned}$$

$n_p(t)$ and $n_p(t+t_p)$ are both Gaussian distributed, and from this point forward $n_p(t)$ is

written instead of $n_p(t+t_p)$ for the simplification. The equations are rearranged according to the data collected by the reference sensor and written in the matrix form as given follows

$$\begin{bmatrix} x_1(t) \\ x_2(t+t_{21}) \\ \vdots \\ x_p(t+t_{p1}) \end{bmatrix} = \begin{bmatrix} 1/r_{11} \\ 1/r_{21} \\ \vdots \\ 1/r_{p1} \end{bmatrix} s^1(t) + \begin{bmatrix} n_1(t) \\ n_2(t) \\ \vdots \\ n_p(t) \end{bmatrix} \quad (9)$$

where $s^1(t) = s(t-t_1)$ and $t_{21} = t_2 - t_1$. $s^1(t)$ is the signal received by the first sensor. The cross correlation between the observation data is described as

$$\mathbf{R}_{x_p x_q}(t_{pm}) = E \left[x_p(t) \cdot x_q^*(t-t_{pm}) \right]. \quad (10)$$

Each sensor signal is supplied by means of the cross correlation matrix as if the signal arrived at the sensor at the same instant with the reference sensor signal. The covariance matrix is calculated to be used for the problem solution by employing new observation data set.

The MUSIC algorithm is one of the efficient suboptimal techniques. It provides high resolution estimates by utilizing the orthogonality between the signal and noise subspaces.

\mathbf{U} unitary matrix and $\Lambda = \text{diag} \{ \lambda_1, \lambda_2, \dots, \lambda_p \}$ is a diagonal matrix of real eigenvalues ordered such that $\lambda_1 \geq \lambda_2 \geq \dots \geq \lambda_p > 0$. The spectral decomposition of the structure of the covariance matrix with the spatial white noise can be expressed as

$$\mathbf{R}_{x_p x_q} = \mathbf{d} \mathbf{R}_s \mathbf{d}^T + \sigma^2 \mathbf{I} = \mathbf{U}_s \Lambda_s \mathbf{U}_s^T + \sigma^2 \mathbf{U}_n \mathbf{U}_n^T \quad (11)$$

where, assuming $\mathbf{d} \mathbf{R}_s \mathbf{d}^T$ to be of full rank, the diagonal matrix Λ_s contains the M largest eigenvalues. Since the eigenvectors in \mathbf{U} , (the noise eigenvectors) are orthogonal to \mathbf{d} , we have

$$U_n^T d(x, y) = 0, (x, y) \in \{(x_1, y_1), \dots, (x_m, y_m)\} \tag{12}$$

To allow for unique location estimations, any collection of P steering vectors corresponding to location parameters forms a linearly independent set ($M < P$). If $d(\cdot)$ satisfies these conditions and R_s has full rank, then $dR_s d^T$ is also of full rank. $(x_1, y_1), \dots, (x_m, y_m)$ are the possible solutions to the relation in equation (12). In practice, an estimate of the covariance matrix is obtained, and its eigenvectors are separated into the signal and noise eigenvectors as follows.

$$\hat{R} = \hat{U}_s \hat{\Lambda}_s \hat{U}_s^T + \hat{U}_n \hat{\Lambda}_n \hat{U}_n^T \tag{13}$$

The orthogonal projector onto the noise subspace is described as $\hat{\Gamma}^\perp = \hat{U}_n \hat{U}_n^T$ and $I = \Pi + \Pi^\perp$. The MUSIC spatial spectrum is defined as

$$Q(x, y) = \frac{d^T(x, y)d(x, y)}{d^T(x, y)\hat{\Gamma}^\perp d(x, y)} \tag{14}$$

The location estimates are found by maximizing the objection function expression. The maximum point of equation (14) is searched beginning from an initial point in the field. A graphical representation of the peak value of the function and the vicinity of the point also can be drawn. The maximum point is the estimation value.

4. THE SIMULATIONS AND RESULTS FOR THE PROPOSED METHOD

There are nine sensors in the first computer simulation run for testing the method. A broadband signal emitting source is in the sensor field and the sensor locations are given as (-10,20), (-15,8), (-5,-8), (4,-10), (7,7), (0,10), (5,15), (15,-13), (-8,1) meter. The true source location is (2,-2). The placement of the sensor network and the source in the field is shown in the figure 2. It is assumed that the signal is unknown in the simulations. The sensor field is searched point by point when the algorithm is run. The sampling frequency of the signal is

$f_s = 1000 \text{ Hz}$. The sensor field is limited as $-25m \leq r_{\max} \leq 25m$ so that the maximum distance between the sensors and the source can be calculated.

The signal power and the noise level is set as $\sigma_s = 10^6$ and $\sigma^2 = (10^5; 10^4; 10^3; 10^2; 10^1; 10^0)$ in the simulation. The real signal power level depends on the distance between the source and the sensors. For example; even the signal power is 60 dB at the source point, the true signal level is $10 \log_{10}(1000000/10^2) = 40 \text{ dB}$ if there is a 10 m distance between them. In this case there is a path loss of 20 dB. The signal power decreases until reaching the sensors considerably because of the distance. In the simulations, 100 independent trials are executed for each σ^2 and Root Mean Square Errors (RMSE) are calculated for the estimated location parameters (x, y) . The RMSE is plotted versus the noise variance and CRB (Cramer Rao Bound), in the plottings noise variance decreasing as SNR (Signal to Noise Ratio) increasing [10]. The plottings are given the figure 3. The method yields a great estimation performance as can be seen the related figures. The source signal arrives at the sensors by decreasing in power reasonably due to the distance.

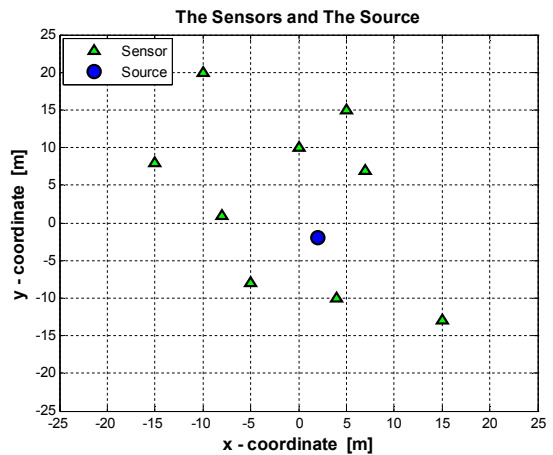


Figure 2. The placement of the sensors and the source

Hence, the signal power is less than that of noise when the variance is between $10^5 - 10^4$ values in the figures presenting the error. Therefore, the

increase of the error is innate in this range of variance. The bias of the method is given in the figure 4. The estimates are plotted versus changing values of noise variances are given by the figure 5.

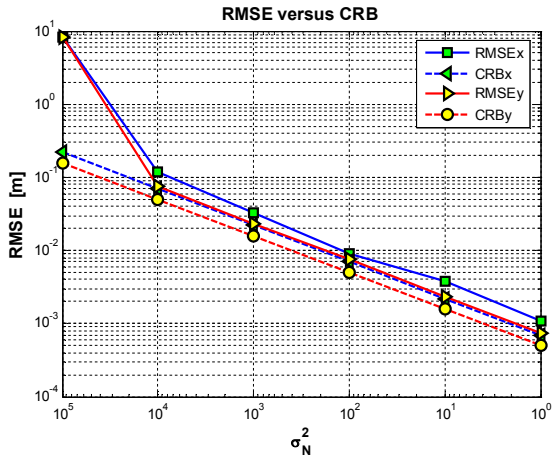


Figure 3. RMSE plotting for x and y coordinate

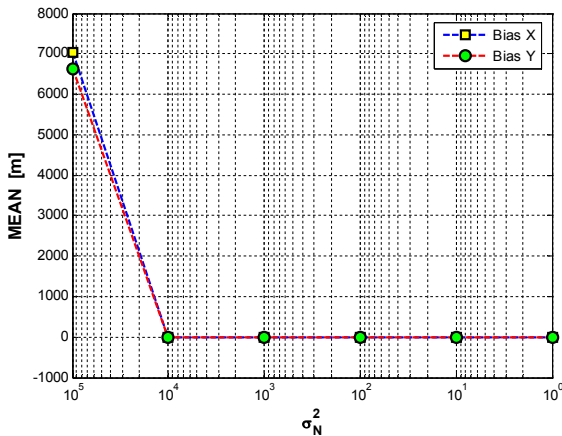
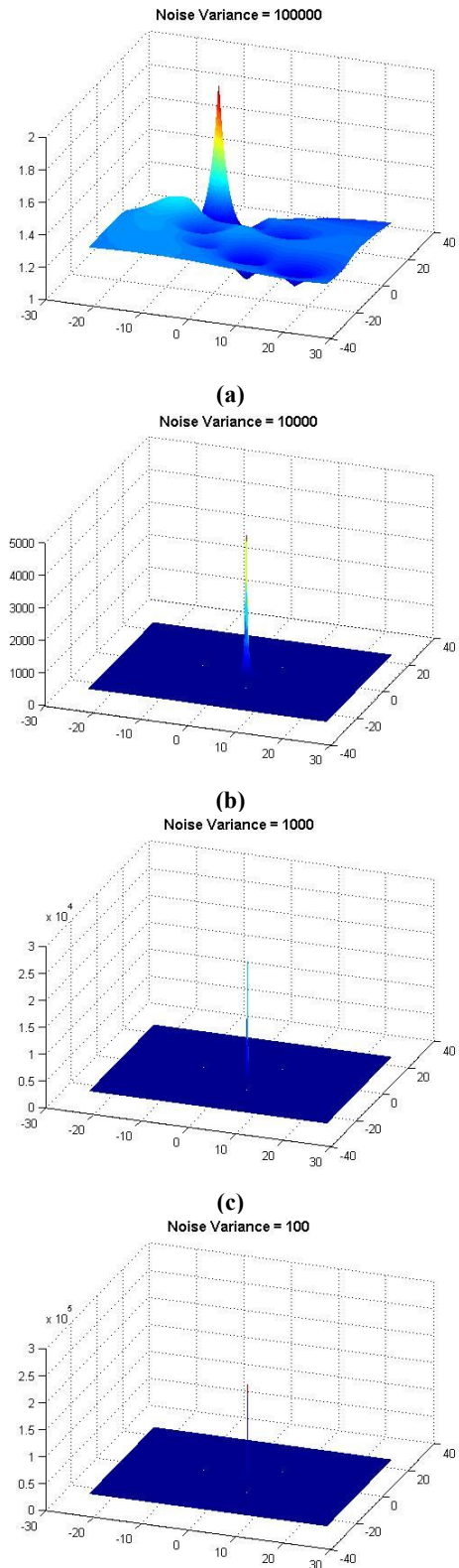


Figure 4. The bias of the estimation algorithm

The source is outside of the sensor field in second simulation scenario. The true source location point is (-15,-5) at that time. The sensor network placement is shown in figure 6. When the algorithm is run under the same conditions, the resulted error is given in figure 7 and the bias graphic is given in figure 8. At this time the estimates are plotted versus changing values of noise variances are given by the figure 9.



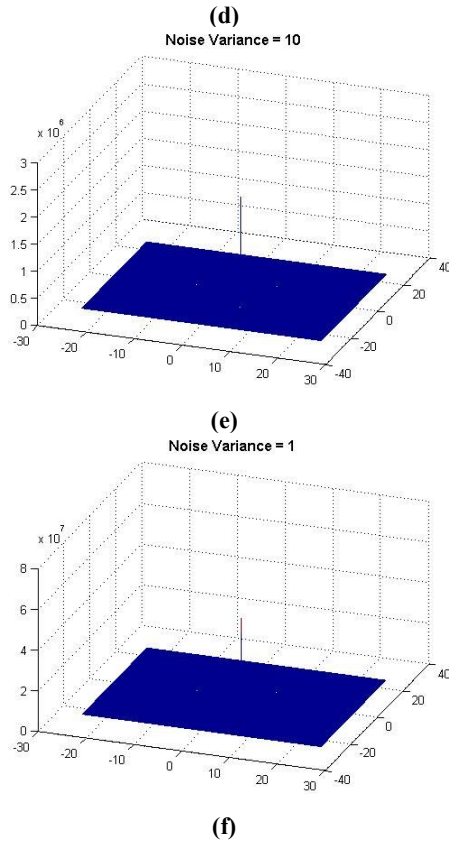


Figure 5. The estimation points versus changing noise variance values

The sensors are placed symmetrically in the sensor field for the third simulation. The thirteen sensor locations and the true source location are given as (-20,20), (20,20), (-20,-20), (20, -20), (0,0), (-12,12), (12,12), (-12,-12), (12,-12), (0,35), (0,-35), (35,0), (-35,0) and (5,15) respectively. The related figures are given as follows and also performance of the method has been examined. There is no performance degradation in this case.

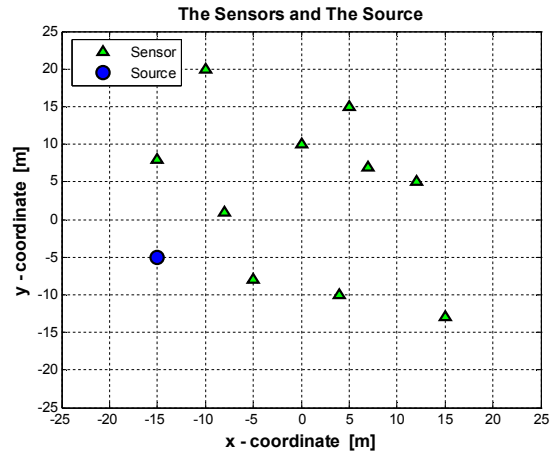


Figure 6. The placement of the sensors and the source

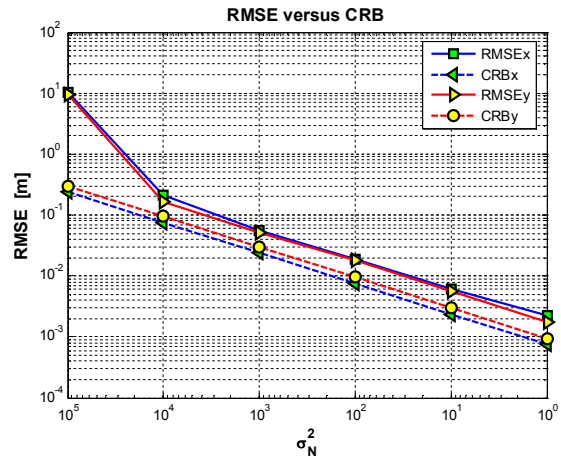


Figure 7. RMSE plotting for x and y coordinate

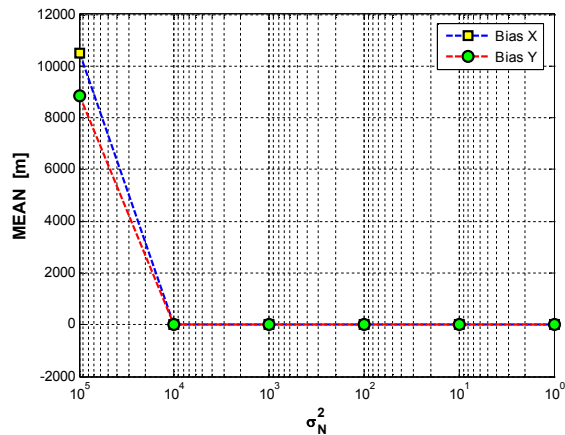


Figure 8. The bias of the estimation algorithm

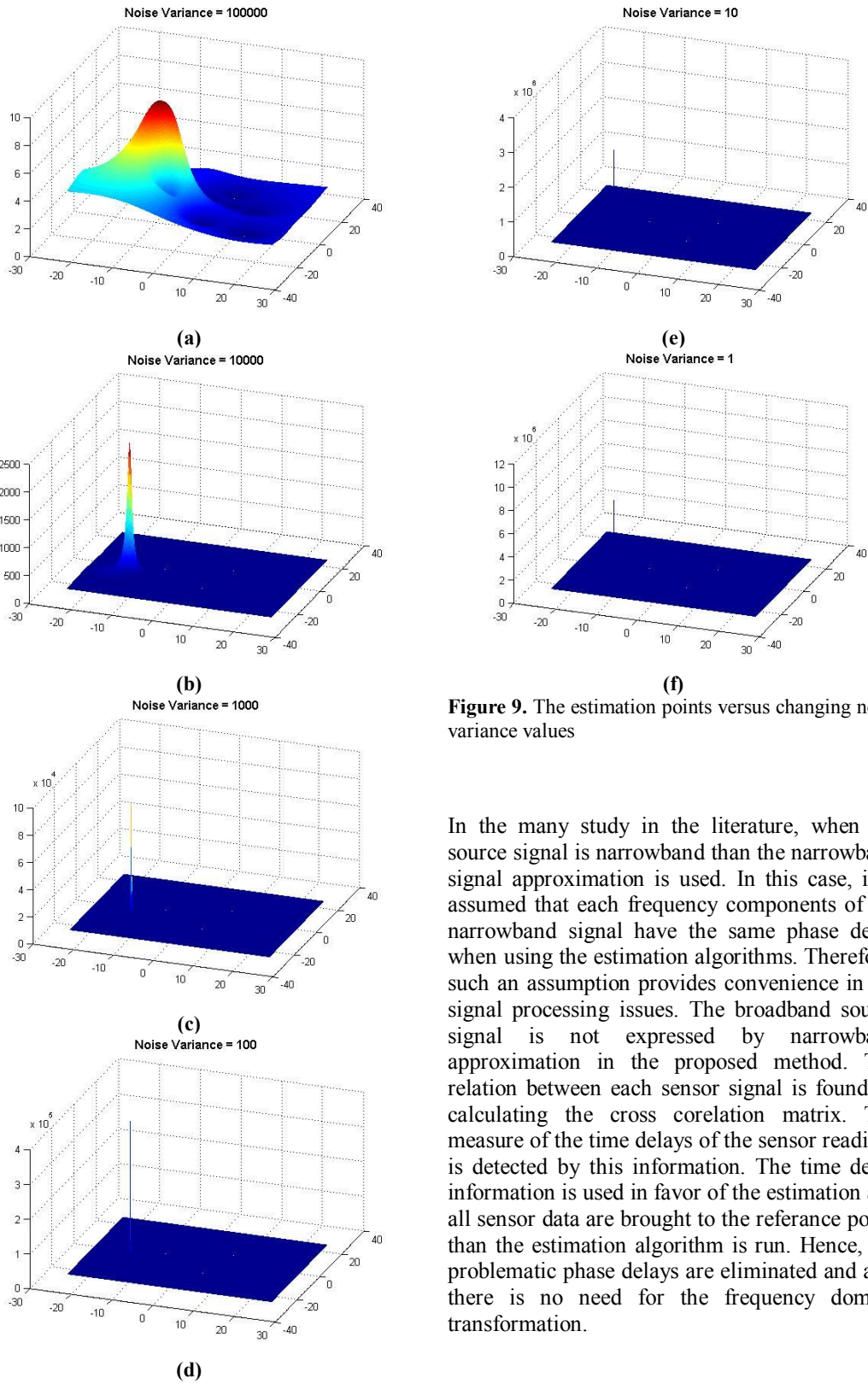


Figure 9. The estimation points versus changing noise variance values

In the many study in the literature, when the source signal is narrowband than the narrowband signal approximation is used. In this case, it is assumed that each frequency components of the narrowband signal have the same phase delay when using the estimation algorithms. Therefore, such an assumption provides convenience in the signal processing issues. The broadband source signal is not expressed by narrowband approximation in the proposed method. The relation between each sensor signal is found by calculating the cross correlation matrix. The measure of the time delays of the sensor readings is detected by this information. The time delay information is used in favor of the estimation and all sensor data are brought to the reference point, than the estimation algorithm is run. Hence, the problematic phase delays are eliminated and also there is no need for the frequency domain transformation.

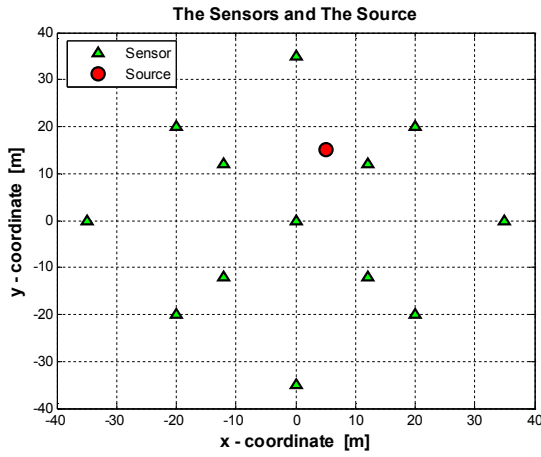


Figure 10. The placement of the sensors and the source

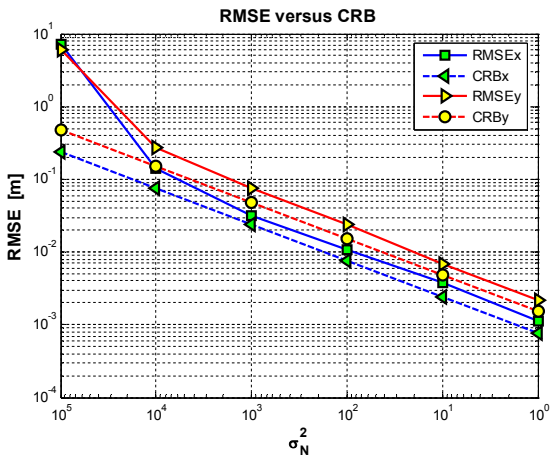


Figure 11. RMSE plotting for x and y coordinate

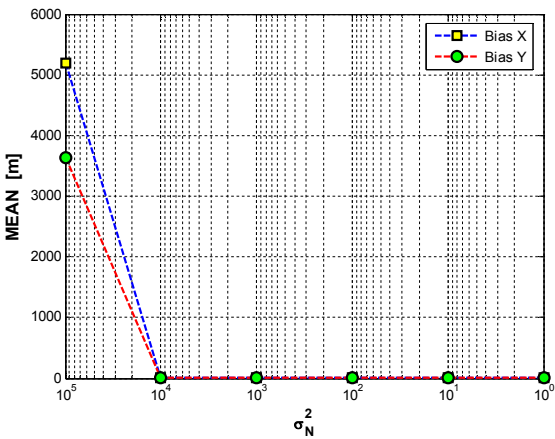
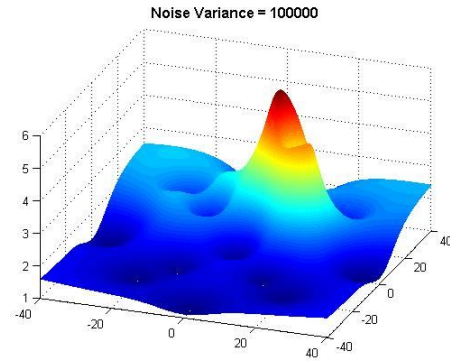
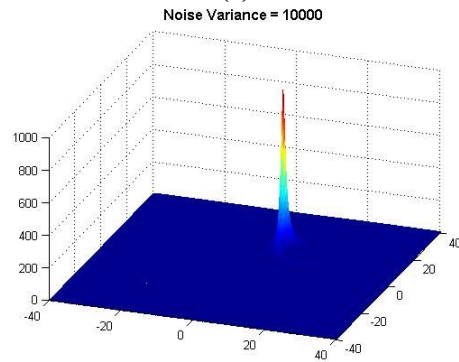


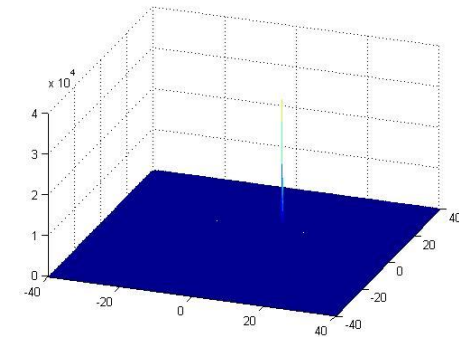
Figure 12. The bias of the estimation algorithm



(a)



(b)



(c)

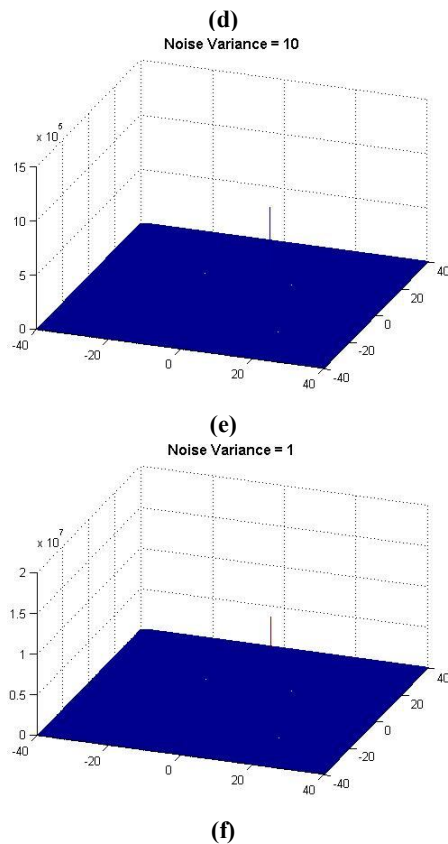


Figure 13. The estimation points versus changing noise variance values

5. CONCLUSION

In this study, a novel method is suggested to estimate the location of the broadband unmodulated signal emitting source in the close range of the randomly placed sensor network. The observation data collected by the randomly distributed sensors. Phase delay of the received sensor signals are eliminated by virtue of finding the correlation between the signals with different amplitudes and phases. Source location estimation has been performed via the MUSIC algorithm using the new observation data at hand. There is no need for the speed propagation of the signal and the algorithm is not dependent on the initial point. Utilizing the estimation algorithms for the baseband acoustic signals efficiently in the time domain, but the propose method provides great estimation performance without requiring transformation to the

frequency domain. Moreover, the computer simulation results are given supporting the method.

6. REFERENCES

- [1] E.A. Lehman, D.B. Ward, R.C. Williamson, "Experimental Comparison of Particle Filtering Algorithms for Acoustic Source Localization in a Reverberant Room", *Acoustic, Speech, and Signal Processing Proceedings (ICASSP), IEEE International Conference*, Vol. 5, Issue 6-10, April 2003 pp. V - 177-80.
- [2] X. Sheng, Y.H. Hu, "Maximum Likelihood Multiple-Source Localization Using Acoustic Energy Measurements with Wireless Sensor Networks", *Signal Processing, IEEE Trans.*, Vol. 53, Issue 1, Jan. 2005, pp. 44-53.
- [3] J.C. Chen, K. Yao, R.E. Hudson, "Source Localization And Beamforming", *Signal Processing Magazine IEEE*, Vol. 19, Issue 2, Mar., 2002, pp. 30-39.
- [4] X. Sheng, Y.H. Hu, "Maximum Likelihood Wireless Sensor Network Source Localization Using Acoustic Signal Energy Measurements", *IEEE Tran. Sig. Proc.*, (2003).
- [5] J.O. Chen, R.E. Hudson, K. Yao, "Maximum-Likelihood Source Localization and Unknown Sensor Location Estimation for Wideband Signals in the Near-Field", *IEEE Trans. Sig. Proc.*, Vol. 50, No. 8, Aug. (2002), pp. 1843-1854.
- [6] S. Valaee, B. Champagne, P. Kabal, "Localization Of Wideband Signals Using Least-Squares And Total Least-Squares Approaches", *IEEE Trans. Signal Proc.*, Vol. 47, No. 5, May, 1999, pp. 1213-1222.
- [7] N. Yuen, B. Friedlander, "Performance Analysis of Higher Order ESPRIT for Localization of Near-Field Sources", *IEEE Trans. Signal Proc.*, Vol. 46, No. 3, Mar., 1998, pp. 709-719.
- [8] R.O. Schmidt, "Multiple Emitter Location and Signal Parameter Estimation", *IEEE Trans.*

Antennas Propagat., Vol. AP-34, Mar., 1986, pp. 276-280.

[9] L. Zhou, Y.J. Zhao, H. Cui, "High Resolution Wideband DOA Estimation Based on Modified MUSIC Algorithm", *Pro. IEEE Int. Conf. Inf. and Auto.*, 2008, June 20-23, Zhangjiajie, China, pp. 20-22.

[10] P. Stoica, A. Nehorai, "MUSIC, Maximum Likelihood, And Cramer-Rao Bound: Further Results And Comparisons", *IEEE Transactions On Acoustic, Speech, And Signal Processing*, Vol. 37, No. 5, Dec., 1990, pp. 2140-2150.

[11] H. Krim, M. Viberg, "Two Decades Of Array Signal Processing Research", *IEEE Signal Processing Magazine*, July, 1996, pp. 67-94.

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