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Direction of Arrival Estimation Using Underwater Acoustic Vector Sensor Array Towards Coastal Surveillance Applications

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ABSTRACT

The objective of this paper is to present the performance of Direction of Arrival(DoA) estimation algorithms for underwater sound source localization using an acoustic Vector Sensor Array (VSA) that is developed by the National Institute of Ocean Technology, Chennai. Algorithms such as conventional beam forming, Multiple Signal Classification (MUSIC) with Eigen value decomposition, and MUSIC with Singular Value Decomposition (SVD) are used for estimation of DoA and performance study. An experiment has been conducted with the VSA at the Acoustic Test Facility of NIOT with the source transmission of 1 kHz to 5 kHz for different azimuth angles. The estimation of DoA using the above three algorithms and the comparison of the results on resolution and accuracy have been studied in detail in terms of the number of vector elements. Results reveal that the MUSIC method gives results with higher accuracy and resolution than the conventional method. The maximum deviation from the true angle in the conventional method is 4°; in MUSIC, it is 2°, whereas in MUSIC with SVD, it is 1°. While the standard MUSIC algorithm involves computing the eigenvectors of the covariance matrix, which can be computationally expensive, MUSIC with SVD provides a more efficient way to achieve better results. SVD enables straightforward computation of the signal subspace, making it more practical for real-time applications like coastal surveillance. Further to the laboratory experiment, the vector sensor system has been deployed in an open sea environment near the harbor and a known source experiment is carried out. The DoA estimated using MUSIC with SVD for the field data reveals that the results are in good comparison with the measured azimuth and elevation positions. The deviations in the field results are due to dynamic conditions of the ocean ,and more sea trials need to be carried out for further study.

Keywords: Vector sensor array; Azimuth angle; Beam forming; Direction of arrival; Source localization

NOMENCLATURE

k	: Wave number
θ	: Azimuth angle
∅	: Elevation angles
ω	: Angular frequency
p	: Pressure
v	: Particle velocity
ρ	: Density of medium
c	: Speed of light
k	: Wave number

1. INTRODUCTION

A vector sensor measures acoustic pressures and particle velocity components in three orthogonal directions. The acoustic Vector Sensor Array (VSA) has become more popular and extensively used in many applications like coastal surveillance, harbour defence, underwater acoustic communication, marine mammal localization studies, bottom parameters estimation, etc. The Ocean Acoustics team of the National Institute of Ocean Technology, Chennai, has developed an underwater acoustic vector sensor array consisting of three elements for underwater source localization towards coastal surveillance applications. The direction of arrival (DoA) in the underwater environment can be determined using VSA, where the measurements of DoA from a sensor reveal the direction from which the sound propagates at each instant. The acoustic pressure in the underwater environment is often measured with hydrophones, and a very long array of hydrophones is required to obtain the DoA of the source. VSA provides more directivity than an array of conventional hydrophones while having a smaller aperture and enabling accurate estimation of the range and DoA of the source[1]. Each element of the vector sensor array consists of a hydrophone and a tri-axial accelerometer to measure the acceleration in three orthogonal directions (i.e., x, y, and z). A beam forming technique often estimates the signal coming from a specific direction[2]. It is possible to use vector sensors in underwater acoustic applications such as Direction of Arrival (DoA) estimation and acoustic inversion for high-frequency signals[3]. Different measurement models are used for a single source and single vector sensor and multiple sources multi-vector sensors for source localization and derived the Cramer-Rao bond on the estimation errors[4]. There are many underwater source localization techniques and challenges for

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estimating accurate DoA, and one can perform a comparison based on application and efficiency[5].

Santos Paulo[6], *et al.* analyzed the field data from the Makai experiment using the Bartlett beamforming technique of source localization. Results demonstrated that reliable DoA estimations could be obtained using high-frequency signals with a four-element array. Bereketli Alper[7], *et al.* have presented the DoA estimation results from the Arc Tan, intensity-based, and time-frequency domain beamforming methods in shallow water. It is shown that the Arc Tan-based technique provides satisfactory performance for the practical approach.

Zhao Anbang[8], *et al.* presented an estimation of the azimuth angle in an open lack experiment using five methods, namely, the Complex Acoustic Intensity Measurement (CAIM) method, Weighted Complex Acoustic Intensity Measurement (WCAIM) method, Conventional Beam Forming (CBF) method, Minimum Variance Distortion-Less Response (MVDR) method, and Multiple Signal Characteristic (MUSIC) methods. It is shown that no previous information is needed to estimate the azimuth angle for the weighted complex acoustics intensity measurement system. It is also shown that the complex acoustics intensity measurement and Multiple Signal Classification (MUSIC) give better performance in spatial resolution than conventional beam forming systems.

Barat[9], *et al.* proposed DoA estimation in vector-sensor arrays using second-order statistics and compared matrix-based and tensor-based models for studying second-order MUSIC methods. The tensor-MUSIC algorithm utilizes Higher Order Singular Value Decomposition (HOSVD) to decompose the covariance tensor, while the matrix-MUSIC method relies on the Singular Value Decomposition (SVD) of the covariance matrix. The comparison evaluates the performance and accuracy of these methods in Direction of Arrival (DoA) estimation for vector-sensor arrays. Simulation results show Root Mean Square Error (RMSE) in DoA estimation for different methods and the number of snapshots. The RMSE values are analyzed with the number of snapshots used in the estimation process, providing insights into the performance and accuracy of the different algorithms in estimating DoAs in vector-sensor arrays.

Dakulagi[10] proposed a method that uses a modified symmetric sensor array to estimate the DoA of signals. By reconstructing modified toeplitz matrices, the rank of these matrices corresponds only to the DoA unaffected by the coherency between signals. This approach aims to improve the accuracy of DoA estimation for both uncorrelated and coherent sources. A method called Covariance Matrix Focusing Fitting (CMFF) is proposed[11] to estimate the DoA of signals in the presence of fluctuating misorientation in acoustic vector sensor arrays. The method transforms measured data into a reference segment to improve accuracy and reduce errors. Simulation results show the robustness and superiority of the proposed method compared to existing methods. Experimental results using an AVSA system demonstrate the effectiveness of the proposed method. Various methods like Particle-Velocity Field (PVF) smoothing, PVF difference smoothing, and compressed sensing technology have been developed[12] to improve DoA estimation with VSA. Sparse signal reconstruction methods like ℓ1-norm based singular value decomposition (ℓ1-SVD), Sparse Bayesian Learning (SBL), and iterative adaptive approach (IAA) offer better performance under low SNR and limited snapshots. Challenges arise in DoA estimation due to axial deviation and non-uniform noise in acoustic VSA, leading to the development of modification methods for accurate estimation. Existing methods like augmented subspace MUSIC and Alternating Iterative Adaptive Approach (AIAA) show limitations in handling non-uniform noise or axial deviation, while the two-step iterative minimization (TSIM) method provides superior DoA estimation performance. The ambient noise covariance matrix for underwater acoustic vector sensors (AVSs) is not equal to a scaled identity matrix. This contradicts the requirement of subspace-based direction-of-arrival estimation methods like the conventional Multiple Signal Classification (MUSIC) method. To address this, a MUSIC-based DoA estimation method (ANE MUSIC method) is proposed[13]. The method transforms the array covariance matrix, concentrates the noise in the real part, and eliminates ambient noise using a real-valued Singular Value Decomposition (SVD). The ANE MUSIC method is asymptotically independent of ambient noise and reduces computational complexity by 75 %. Experimental results verify the practical effectiveness of this method.

VSAs are commercially available primarily for air applications, and very few countries have developed underwater VSAs for their own naval purposes. Hence, NIOT developed a Vector Sensor and Array in-house for underwater source localization purpose in the open ocean. Sensor arrays deployed in shallow waters need to account the challenges for the effects of wave action, sea surface and seafloor reflections, and other shallow water phenomena. The novelty of this work is the development of a VSA with three elements to localize the underwater acoustic source with more accuracy in shallow waters towards coastal surveillance applications. The performance is assessed by applying different DoA estimation algorithms, such as conventional, MUSIC (Multiple signal classification), and MUSIC with SVD, by conducting laboratory and open sea experiment.

2. METHODOLOGY

2.1 Beam Forming with VSA

In a Cartesian coordinate system, it is assumed that the impinging signals are plane waves, and a planar wave moves in the direction of the origin. Figure 1 represents the vector sensor array coordinates in all three directions. The maximum wavelength should be much smaller than the distance between the sources to the receiver. As a result of transmission, the plane-wave acoustic pressure can be expressed as[15-16]:

$$p(r,t) = p_0 e^{i(\omega t - kr)} \tag{1}$$

where, p_0 is the amplitude, $\omega=2\pi f$ represents the angular frequency, k= ω/c is the wave number of the acoustic wave and r is a position vector where the sound wave is estimated, and c represents the speed of sound in acoustic propagation. The vector $r=[r_x r_y r_z]$ can be used to represent any point in a three-dimensional space where, r_x, r_y and r_z are the coordinates in the Cartesian coordinate system. As a function of its location in spherical coordinates, each point in a rectangular Cartesian coordinate system can be represented as:

$$x = r\cos\theta\cos\varnothing \tag{2}$$

$$y = r\sin\theta\cos\varnothing \tag{3}$$

$$z = rsin\varnothing \tag{4}$$

Nehorai Arye[4], *et al.* presented the measurement model for a single vector sensor with a single source and a vector sensor array with multiple sources. Let u be the unit vector pointing towards the source for a single vector sensor with a single source. The response of a vector sensor will be described by this unit vector such that $r = |u| = 1$. It can be expressed mathematically as a function of azimuth and elevation angles as:

$$u = \begin{bmatrix} cos\theta cos\varnothing \\ sin\theta cos\varnothing \\ sin\varnothing \end{bmatrix} \tag{5}$$

where, θ and ∅ denote the azimuth and elevation angles of the unit vector thus, θ ranges from (0, 2π) and elevation ranges $-\pi/2 \le |\varnothing| \le \pi/2$. Hence, Eqn. (1) has the following form[17]

$$p(r,t) = p_0 exp[j(kr_x\cos\theta\cos\varnothing + kr_y\sin\theta\cos\varnothing + kr_z\sin\varnothing - \omega t) \tag{6}$$

Euler's conservation of momentum equation describes the relation between acoustic pressure and particle velocity[18],

$$\frac{\partial \boldsymbol{v}}{\partial t} + \frac{1}{\rho}\nabla p = 0 \tag{7}$$

where, v is the acoustic particle velocity, ρ is the density of medium, and p is the acoustic pressure.

Spatial derivatives of the acoustic pressure are used to calculate the acoustic particle velocity in the x, y and z directions as[15]

$$v = -\frac{1}{j\omega\rho}\nabla p = \frac{1}{j\omega\rho}\begin{bmatrix} \frac{\partial}{\partial x}p \\ \frac{\partial}{\partial y}p \\ \frac{\partial}{\partial z}p \end{bmatrix} \tag{8}$$

Time derivative changes to $j\omega v = \frac{1}{\rho}\nabla p$ in the frequency domain, where $j = \sqrt{-1}$ and ω is the angular frequency. Since $a = \frac{\partial v}{\partial t}$, Eqn. (7) can be written as[15]:

$$a = -\frac{1}{\rho}\nabla p = \frac{1}{\rho}\begin{bmatrix} \frac{\partial}{\partial x}p \\ \frac{\partial}{\partial y}p \\ \frac{\partial}{\partial z}p \end{bmatrix} \tag{9}$$

It is necessary to process each channel with similar units when combining the particle motion and hydrophone channels. The acoustic impedance is used to scale the pressure measurements. The acoustic impedance for a three dimensions plane wave is given as Z=ρc[12]. By scaling v by Z, it is possible to process the pressure and particle speed in the same unit.

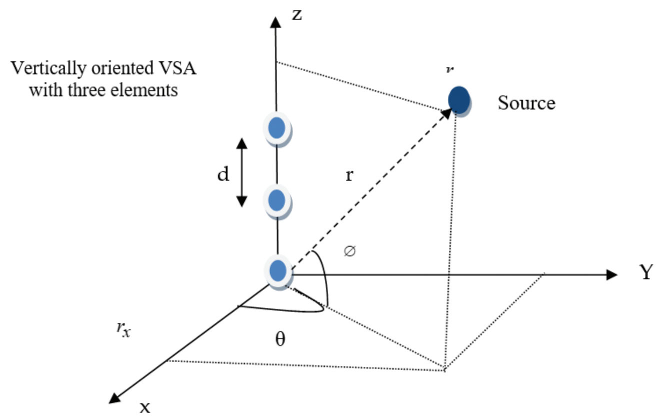


Figure 1. Array coordinates and the propagation of planar wave front with azimuth θ and elevation ∅.

The acoustic impedance for a three dimensions plane wave is Z=ρc. Therefore, for plane-wave beam formation, the pressure readings might be scaled by -1/ρc before being linearly combined with the components of particle velocities.

Let $v(r,t)$ represent the acoustic particle velocity at a position r and time t, and $p(r,t)$ denote the acoustic pressure. The relation between acoustic particle velocity and acoustic pressure can be shown as[20]:

$$v(r,t) = \frac{-p(r,t)}{\rho c}u \tag{10}$$

where, c represents the speed of sound in the medium, ρ represents the medium density. The collection of phase delays encountered by a plane wave can be represented by the array manifold vector or weighting vector, which is assessed at a collection of array elements. The array manifold vector for a vector sensor in Cartesian coordinate in response to an acoustic wave can be written as[15]:

$$w(\theta,\varnothing) = [1\ \cos\theta\cos\varnothing\ \ \cos\theta\sin\varnothing\ \ \sin\theta] \tag{11}$$

The first value in array manifold vector represents the acoustic pressure and the remaining three are direction cosines for particle velocity at x, y and z directions respectively. In order to beamform, signals from different sensors were delayed and weighed to create an output whose maximum value gives the estimate of source direction. Direction cosines for particle velocity components and unity for pressure component is used as weighing vector.

The weighing vector for the i^{th} element of L element VSA is given by[15]:

$$W(\theta,\varnothing) = [1\ \cos\theta\cos\varnothing\ \ \cos\theta\sin\varnothing\ \ \sin\theta].exp(jk.r) \tag{12}$$

where, i=1…L is the number of elements in array. Consider M number of sources emits signal towards L elements uniformly, towards VSA then the dimension of manifold vector will be $M\times L$ and the sensor array's acquired signal vector has the following form

$$Y(t) = \sum_{s=1}^{M} W(\theta_s,\varnothing_s)S_s(t) + n(t) \tag{13}$$

W is the steering vector towards the direction $(\theta_s, \varnothing_s)$, $s_s(t)$ is the signal emitted by the s^{th} source and $n(t)=[n_1(t)\ldots\ldots n_L(t)]^T$ is the noise vector.

2.2 Direction of Arrival Estimation Algorithms

2.2.1 Conventional Beam Forming Method

The concept of the conventional beam forming method is that the best estimate of power arriving in a certain direction is obtained by aiming the strongest beam in that particular direction. The Bartlett parameter estimate is given by[15] :

$$P_B = E\left\{W(\theta,\varnothing)^H . Y(t) . Y(t)^H . W(\theta,\varnothing)\right\} \tag{14}$$

The power spectrum of the conventional beam forming method is given as:

$$B_B(\theta,\varnothing) = W(\theta,\varnothing).RW(\theta,\varnothing)^H \tag{15}$$

where, ^{H} represents the complex transposition conjugation operator, *E{}* denotes statistical expectation. The covariance matrix R with dimension $L \times L$ is given as $E=\{Y(t).Y(t)^H\}$. The position corresponding to the maximum value of $B_B(\theta,\varnothing)$ will be the value of direction of arrival.

2.2.2 Multiple Signal Classification (Music) Algorithm With Eigen Value Decomposition

The MUSIC technique is a subspace-based approach for estimating DoA that takes advantage of the Eigen structure of the correlation matrix of array output. The correlation matrix's Eigen value decomposition will provide signal and noise subspaces. This noise subspace is employed in the spectrum function to determine the peak, which will reveal the signal's direction of arrival. The MUSIC algorithm implementation can be given as follows [21-22].

First step is to find out the input covariance matrix from the received signal vector as:

$$R_X = E\left|Y(t)Y(t)^H\right| \tag{16}$$

where, Y(t) is the received signal vector from sensor array. The dimension of covariance matrix will be $L \times L$. The second important step is to find Eigen values using covariance matrix. Assume that signal and noise are uncorrelated. Eqn. (13) can be substituted into Eqn. (16). Thus, the following can be obtained:

$$\begin{aligned} R_x &= WE\left[SS^H\right]W^H + E[nn^H] \\ R_x &= E\left[(WS+n)\right]\left[(WS+n)^H\right] \\ R_x &= WR_sW^H + R_n \end{aligned} \tag{17}$$

where, R_s is the signal correlation matrix with dimension $M \times M$ and $R_n=\sigma^2 I$ is the noise correlation matrix with dimension $L \times L$. σ^2 is the power of noise and I is the unit matrix. The Eigen values of the matrix R_x are sorted by size, which is:

$$\lambda_1 \geq \lambda_2 \geq \ldots \geq \lambda_L > 0$$

where, larger Eigen values M corresponds to signal while L-M smaller Eigen values corresponds to noise. The noise subspace matrix (with dimension $L \times (L-M)$ is given as:

$$E_n = W^H V_i = 0 \tag{18}$$

where, V_i is the eigenvector corresponding to λ_i. The power density spectrum for the MUSIC algorithm is:

$$P_{MUSIC}(\theta,\varnothing) = \frac{1}{W(\theta,\varnothing)' E_n W(\theta,\varnothing)} \tag{19}$$

The estimated value of DoA can be found by looking at the peak values after the spectrum function has been calculated in the final phase.

2.2.3 MUSIC Algorithm with Singular Value Decomposition

A modified approach is presented for recovering signal parameters from noisy observations using Singular Value Decomposition (SVD) in the existing MUSIC algorithm to improve the results. Singular value decomposition involves factorizing a matrix into a number of linear approximations. These approximations will reveal the fundamental structure of the matrix. The SVD transforms a matrix of correlated variables into an uncorrelated matrix to better understand the relationship between all of the data points, which may have needed to be clarified at their initial generation[23-24].

The singular value decomposition of an $n \times m$ complex matrix M is a factorization of the form $M=USV^*$ where, U is an $n \times n$ complex unitary matrix, S is an $n \times m$ rectangular diagonal matrix , V is an $m \times m$ complex unitary matrix, and V* is the conjugate transpose of V. Column of U and V gives the eigen vectors and diagonal of S gives the square root of eigen values of the input matrix which is used to find out the signal and noise subspace.

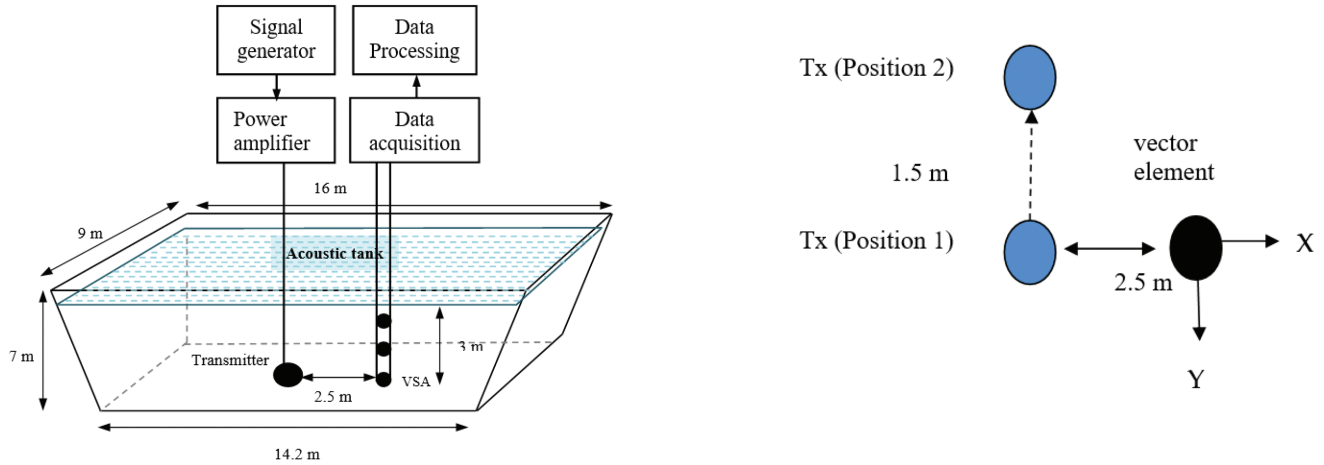
3. EXPERIMENTAL DETAILS

3.1 Laboratory Experiment

The lab experiment is conducted at the Acoustics Test Facility (ATF) of the National Institute of Ocean Technology (NIOT), Chennai. The VSA is positioned in the middle of the water column of the acoustic tank and has the dimensions of 16 (length) x 9 (breadth) x 7 (depth) at a depth of 3 meters from the water surface. Fig. 2(a) shows the schematic diagram of the experimental setup. The VSA has three elements, and the distance between the elements is 125 mm. A single frequency burst of 2 ms from 1 kHz to 5 kHz at 0.5 kHz intervals is generated using a waveform generator and amplified and fed to the acoustic transmitter. The distance of the transmitter to the receiver is 2.5 m. A moving platform is used to mount the transmitter, and its depth varies to take the different datasets from different positions. Fig. 2 (b) shows the top view of two different positions of the transmitter for azimuth angles, i.e., 180° for Position 1 and 215° for position 2 with the elevation angle of 0°. Initially, the transmitter was mounted at position 1, where the acoustic center was aligned with the acoustic vector sensor, and the signal was transmitted for this position. Then, the transmitter moved in a 1.5 m upward direction, representing position 2 of the transmitter. The VSA signals are acquired through 12-channel data acquisition systems where twelve outputs from three elements are received with a sampling frequency of 25 kHz for an acquisition time of 20 sec. The received signal consists of the direct signal received from the transmitter, along with boundary reflections. Only the direct signal portion is considered for DoA estimation.

3.2 Known Source Experiment Off Chennai Harbour

A known source transmission experiment was conducted on 10 October 2019 inside Chennai harbour. Here, the VSA system consists of three elements with a spacing of 230 mm. Figure 3 shows the experimental setup for VSA field testing.



(a) (b)

Figure 2. (a) Schematic of VSA laboratory experiment in ATF tank; and (b) Different transmitter positions for different azimuth (top view).

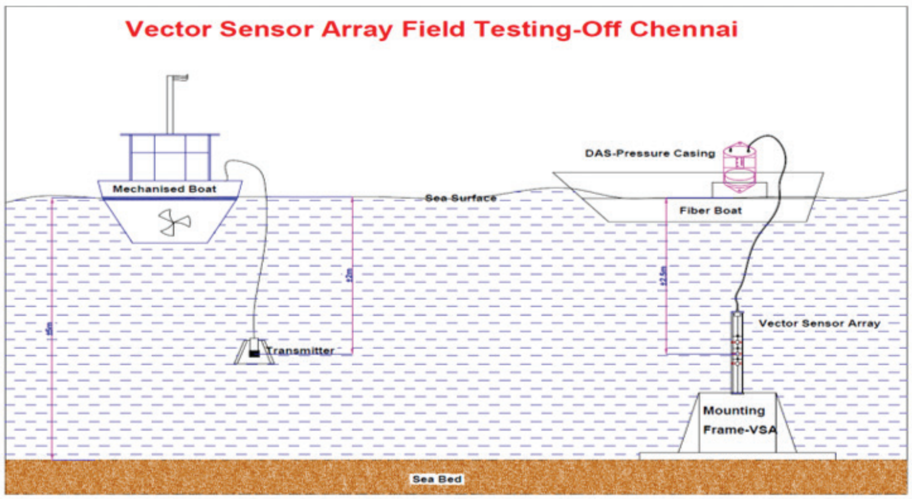


Figure 3. The experimental setup.

In this experiment, the acoustic source and VSA are lowered at a depth of 2.5 m, where the water column depth is 5 m. The transmitter was lowered from one boat, and VSA was lowered from another, as shown in Fig. 3(a). Single-tone frequencies of 0.5 to 6 kHz have been transmitted periodically from transmitter position 1 (P1) and transmitter position 2 (P2). The distance between P1 and VSA is 56 m, and for P2 and VSA, it is 327 m. VSA received the transmitted signal with a sampling frequency of 25 kHz and an acquisition time of 240 sec.

4. EXPERIMENTAL RESULTS

4.1 Results From the Lab Experiment

The hydrophone output voltage is converted into pressure using receiving sensitivity, and the accelerometer output is converted into particle velocity. Pressure and particle velocity are processed in the same unit using acoustic impedance, as described in section 2. The Fast Fourier Transform (FFT) converts the time series acoustic pressure data to the frequency domain. The data received from positions one and two are analyzed and used for comparison. The experiment is conducted for a frequency range of 1-5 kHz. The output signal acquired for transmitter Position 1 from all three elements of the array for 3 kHz is shown in Fig. 4(a), 4(b) and 4(c). Signal strength is more in the x direction because it faces the transmitter. The presence of reflections can be observed in the received signal from pressure and x, y, and z components of the particle acceleration due to the boundaries of the acoustic tank. Only direct signals are windowed for further processing to obtain the appropriate beam forming results as shown in Fig. 4(d). This direct signal is converted into a frequency domain, as shown in Fig. 4(e). All three algorithms, i.e., conventional, MUSIC with Eigen value decomposition, and MUSIC with singular value decomposition, are applied for the DoA estimation based on the number of elements. DoA is estimated for a single vector sensor element, two elements, and three elements for the 1 to 5 kHz frequencies for two positions.

Figure 5(a) shows the estimated azimuth and elevation for one element (I), two elements (II), and three elements (III), respectively, for each of all three methods at a frequency of 3 kHz for position 1. Figure 5 shows the estimated azimuth and elevation for one element (I), two elements (II), and three elements (III), respectively, for each of all three methods at a frequency of 3 kHz for position 1. In Fig. 5, the X-axis represents the azimuth angle, and the Y-axis represents the elevation angle in degrees.

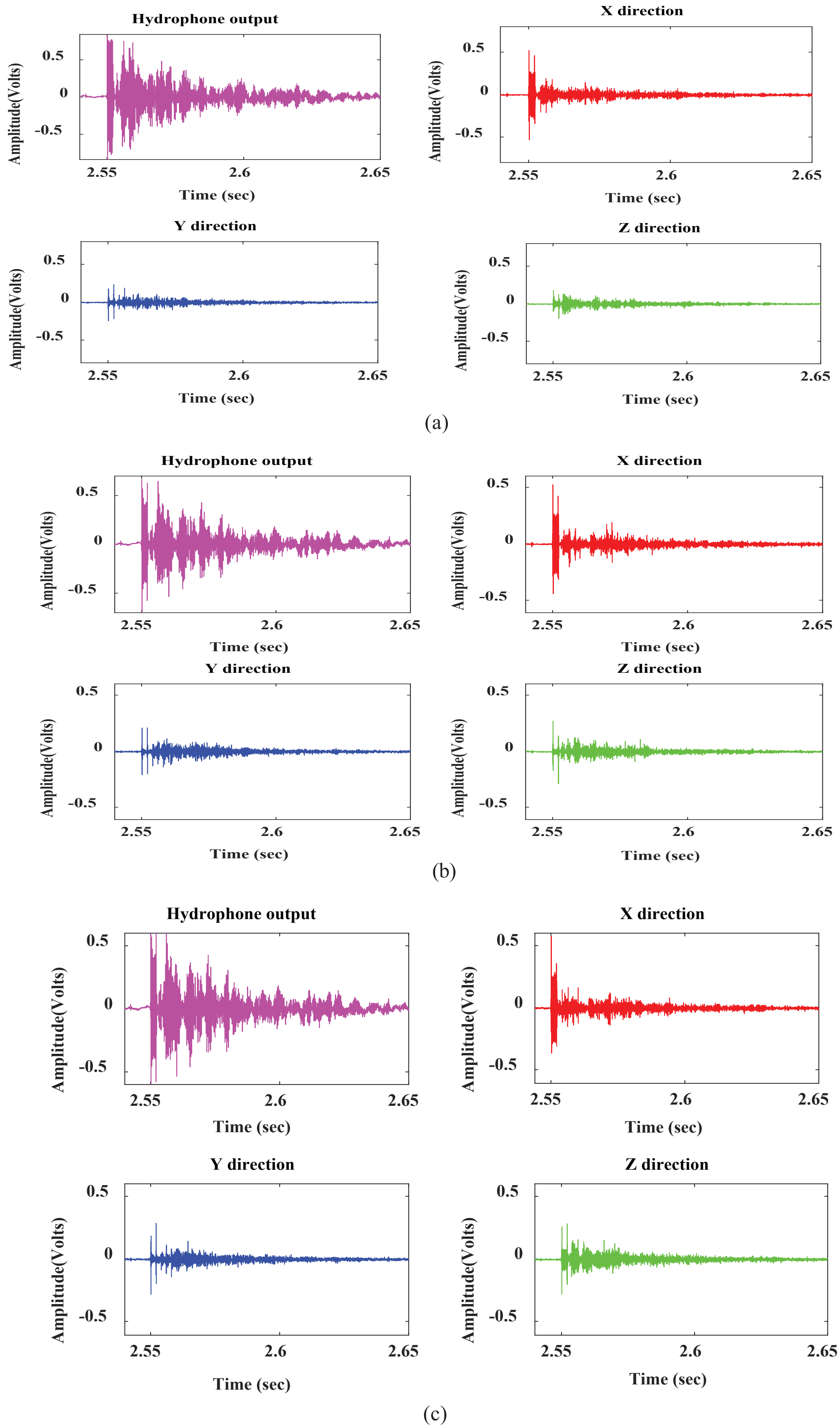
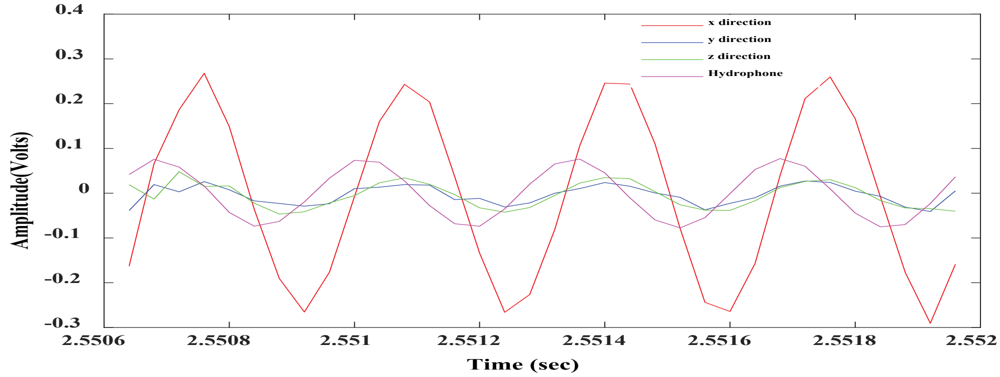
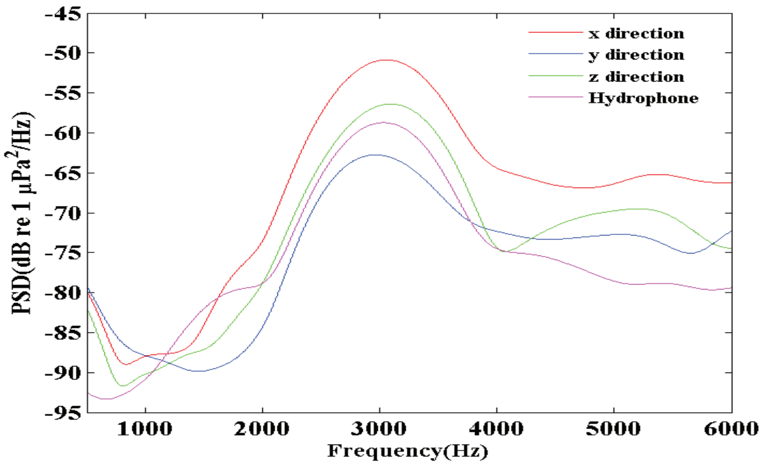


Figure 4. Raw Signal acquired by VSA at 3 kHz for position 1 (a) First element (b) Second element (c) Third element.



(d)



(e)

Figure 4. Raw Signal acquired by VSA at 3 kHz for position 1 (d) Direct signal from one channel (e) Frequency spectrum of the first element of the vector sensor array.

Table 1. Different algorithms estimated azimuth (Azi) and elevation (Elev) angles with spread comparison for one, two and three element array for 3 kHz (actual angle 180°)

No of elements	One element		Two element		Three element	
Method	Azi±X axis spread (deg)	Elev±Y axis spread (deg)	Azi±X axis spread (deg)	Elev±Y axis spread(deg)	Azi±X axis spread (deg)	Elev±Y axis spread(deg)
Conventional	184±21	-1±20	183±20	-1±10	182± 18	-1±6
MUSIC	183±12	0±11	183±11	0 ±5	182± 11	0 ±3
SVD-MUSIC	183±12	0±10	183± 10	0 ±5	181± 9	0 ±3

The actual azimuth angle is 180°, and the elevation angle is 0° for position 1 because the transmitter and receiver were kept at the same depth and facing - x direction. Table 1 quantifies these results and presents accuracy and resolution comparison of the algorithms. At a frequency of 3 kHz, the conventional method for one element has a variation of 4°, two elements have a variation of 3°, and three elements show a variation of 2° from the actual azimuth. One element and two elements, MUSIC and SVD MUSIC have a variation of 3°, three elements of MUSIC have a 2° variation, and three elements of SVD MUSIC have a 1° variation from the actual azimuth angle for 3 kHz frequency.

All three algorithms show -1° variation in elevation for one element and 0° variation for two and three-element arrays. The estimated DoA angles matches well with known azimuth and elevation angles for all algorithms. Fig. 6(a) and 6(b) show the estimated azimuth and elevation angles with the spread along the X-axis and Y-axis, respectively, for the three algorithms and one element, two elements, and three elements array, respectively, at a frequency of 4 kHz. Results imply that the spread angle is higher for the conventional method, and it is decreasing for MUSIC with EVD and MUSIC with the SVD algorithm. Also, in terms of the number of elements, one element has more spread in the y direction and the x direction, and it is comparably less for two elements and three elements. An increasing number of elements can increase the resolution capacity of the algorithm because larger number of elements may cover a broader range of scenarios or conditions, allowing the algorithm to better generalize and perform well across a wider spectrum of inputs.

Similarly, the above process is carried out for position 2, where the actual azimuth angle is 215° with zero elevation. Figure 7 shows the results of beam former outputs for three

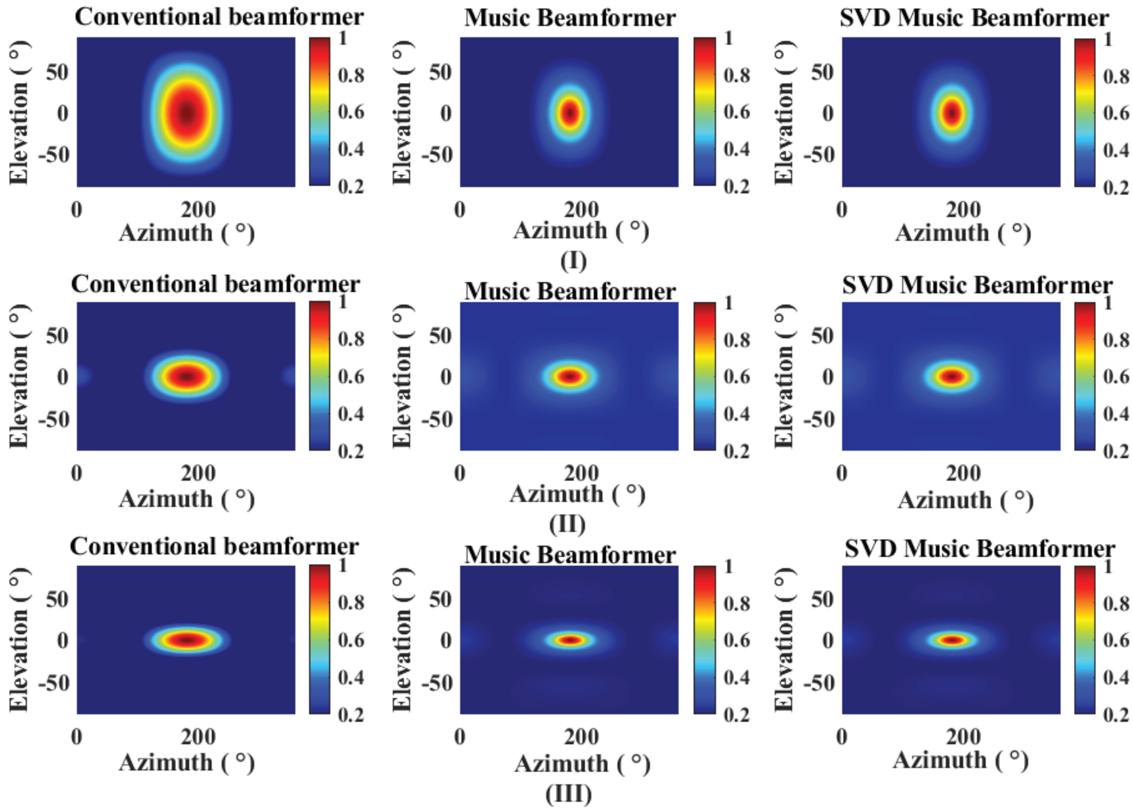


Figure 5. Experimental beam forming results from conventional, MUSIC and MUSIC with SVD algorithms for position 1 at 3 kHz frequency for (I) one element (II) Two elements (III) Three elements.

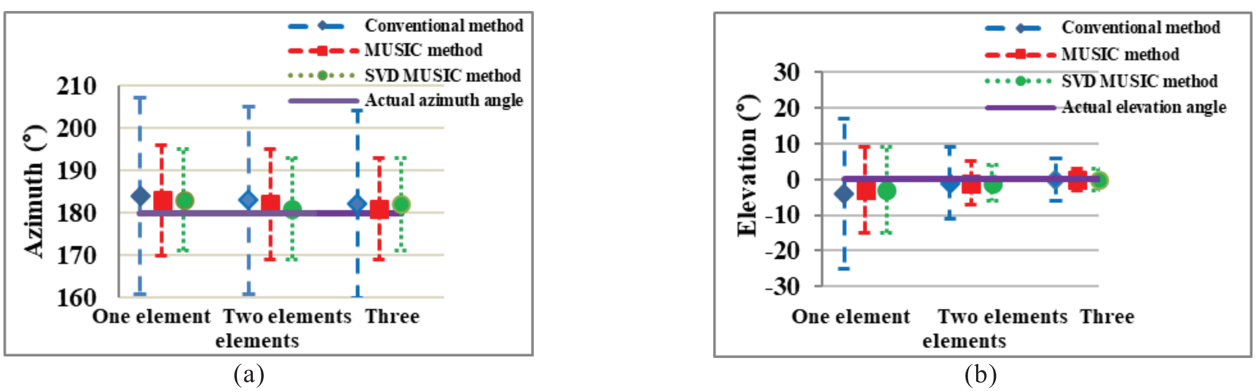


Figure 6. Estimated azimuth and elevation angles for One, two and three elements array for a frequency of 4 kHz (a) Spread in azimuth (b) Spread in elevation for position 1.

methods, where each method is compared based on one, two, and three elements at a frequency of 3 kHz for position 2 and Table 2 quantifies the results. The estimated azimuth angle for one, two, and three elements in the conventional method is 218°, 212°and 213°; the MUSIC algorithm shows estimated angles 217°, 213°, and 214° and SVD MUSIC shows 216°, 213°, and 214° estimated azimuth angle. The elevation angle is -7°,-2°, and -1° for one, two, and three-element arrays, respectively, for all the algorithms. Though the one vector element also has good accuracy, the three elements array shows better resolution capacity than the other two combinations. Figure 8(a) and 8(b) show the estimated azimuth and elevation with the spread for one, two, and three elements for all three methods. Results show that accuracy increases from the conventional algorithm to the SVD MUSIC algorithm, and the resolution increases from one element array to three elements array. The performance of the conventional method may degrade in scenarios with correlated signals or noise or in situations where the array geometry is not well-suited for DoA estimation. The conventional method offers simplicity and lower computational cost, but the MUSIC method provides higher resolution and better performance especially when dealing with closely spaced sources or correlated signal and noise.

This laboratory experiment demonstrates the performance of DoA estimation algorithms with the VSA system, which will be very useful while conducting experiments in an open ocean environment. Laboratory experiment data results reveal that MUSIC with SVD has good performance in terms of

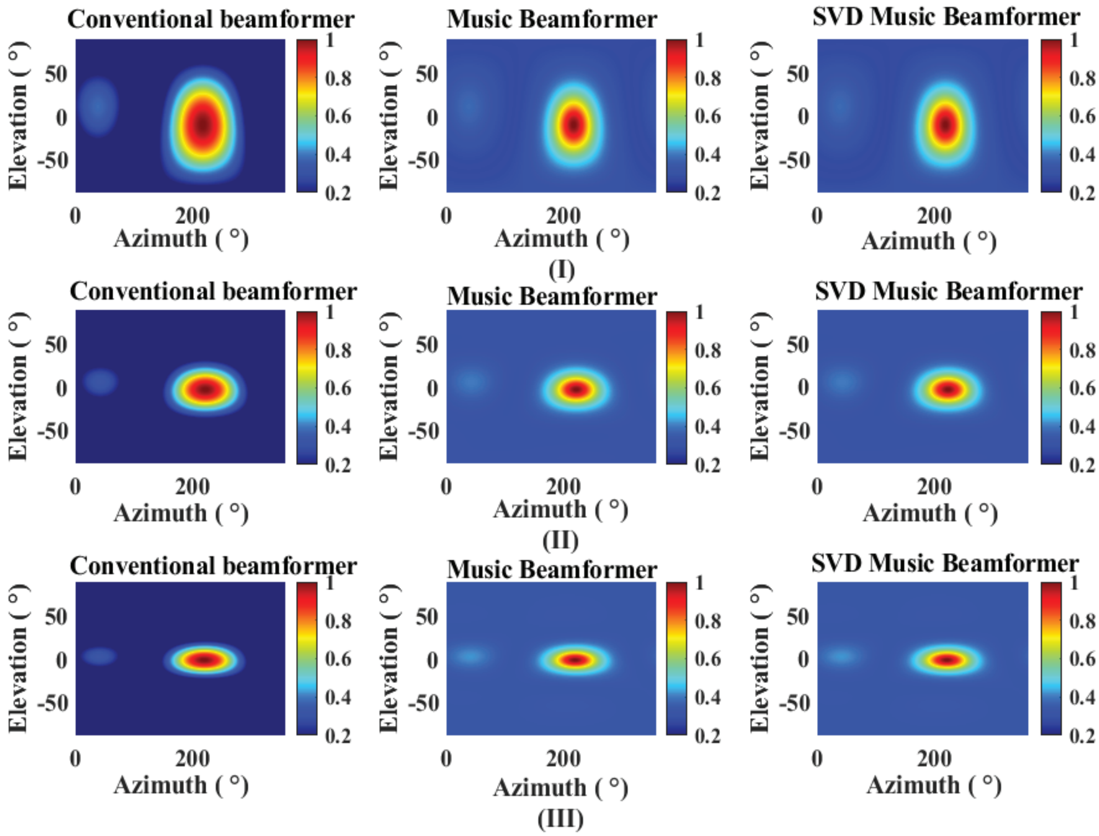
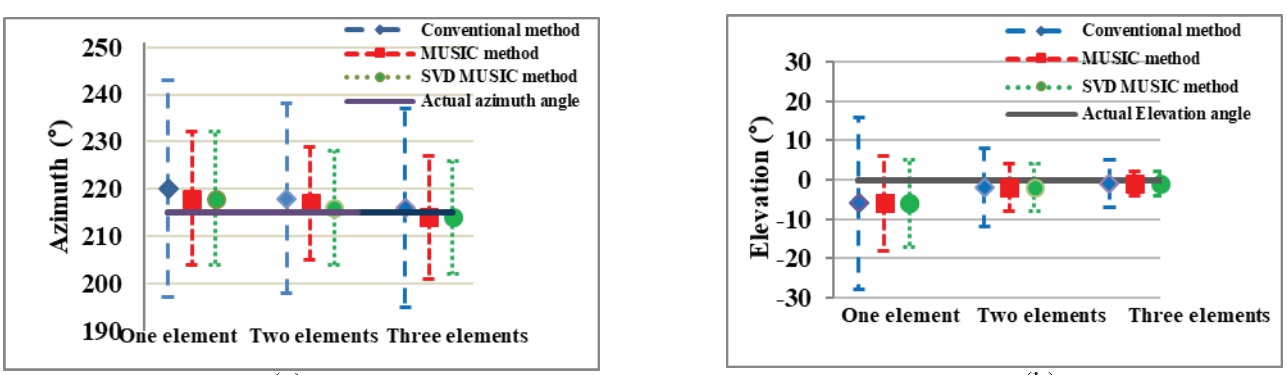


Figure 7. Experimental beam forming results from conventional, MUSIC and MUSIC with SVD algorithms at 3 kHz (I) one element (II) Two elements (III) Three elements for position 2.

Table 2. Different algorithms estimated azimuth (Azi) and elevation (Elev) angles with spread comparison for one, two and three element array for 3 kHz (actual angle 215°).

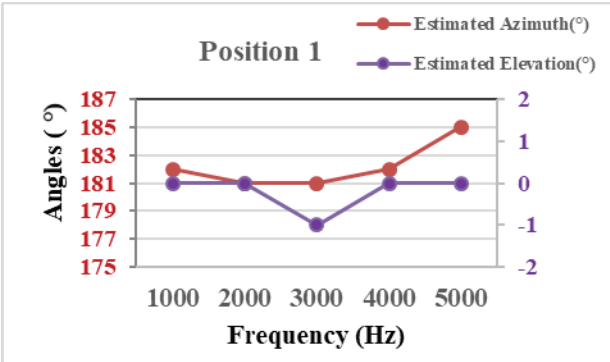
No of elements	One element		Two element		Three element	
Method	**Azi±X axis spread(deg)**	**Elev±Yaxis spread(deg)**	**Azi±X axis spread (deg)**	**Elev±Y axis spread(deg)**	**Azi±X axis spread(deg)**	**Elev±Y axis spread(deg)**
Conventional	218±18	-7 ±17	212±11	-7±13	213±10	-7±11
MUSIC	217±15	-2 ±8	213±4	-2 ±6	214±4	-2 ±5
SVD-MUSIC	216± 14	-1 ±4	213± 2	-1 ±5	214± 3	-1 ±5



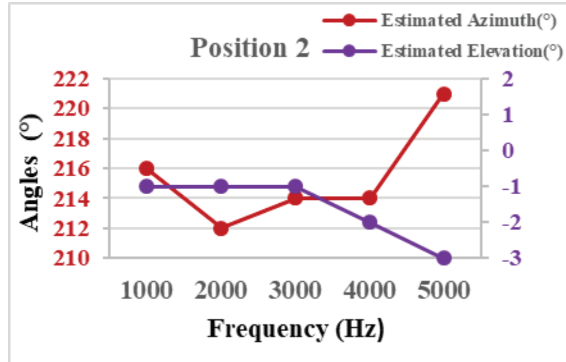
(a) (b)

Figure 8. Estimated Azimuth and elevation angles for one, two and three elements array for a frequency of 4 kHz (a) Spread in Azimuth (b) Spread in elevation for position 2.

resolution and accuracy, and can be used for DoA estimation for harbour experiments. Figure 9 shows estimated azimuth and elevation angles from the SVD MUSIC algorithm for the 1-5 kHz frequency range and for both position 1(azimuth 180°, elevation 0°) and position 2 (azimuth 215°, elevation 0°). 3 kHz frequency is performing well for position 1, and



(a)



(b)

Figure 9. Estimated Azimuth and elevation angles for SVD MUSIC algorithm for frequency range 1- 5 kHz (a) position 1 and (b) position 2.

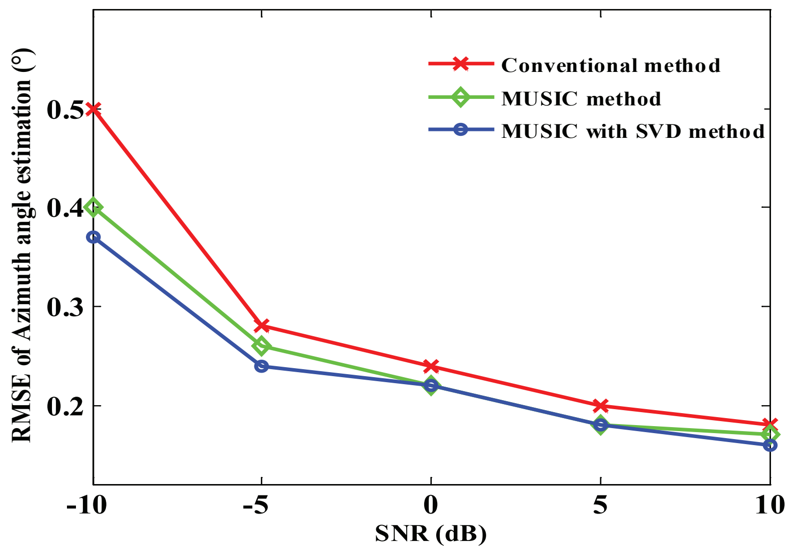
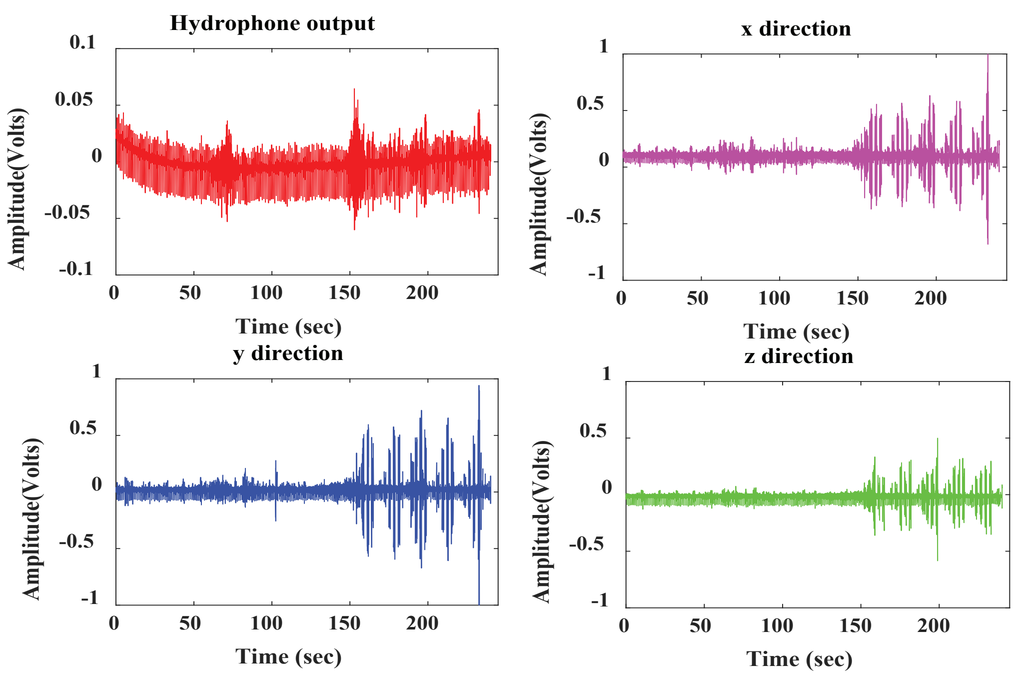
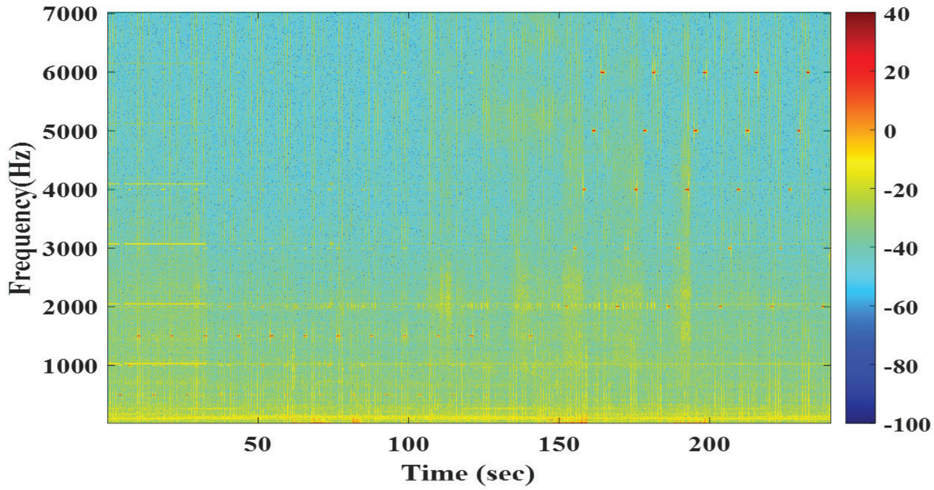


Figure 10. RMSE versus SNR.

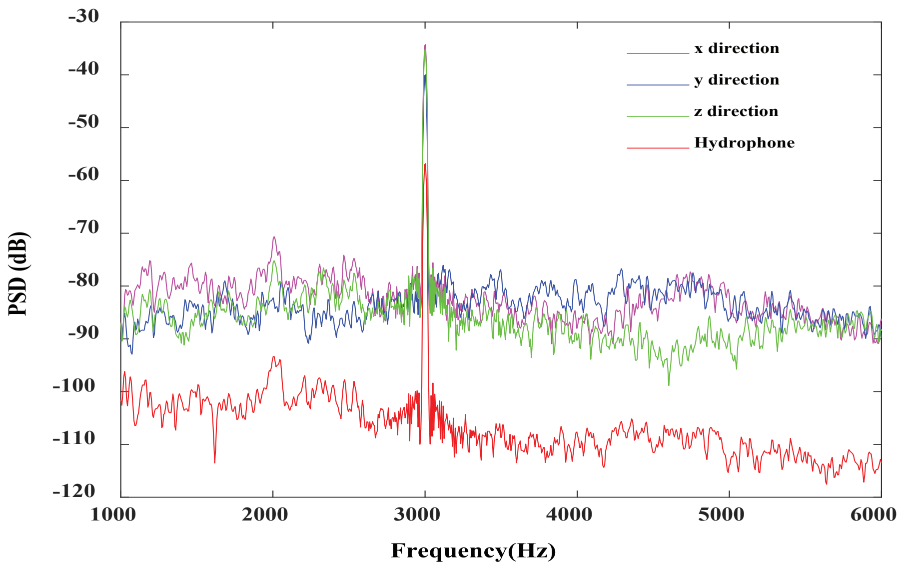


(a)

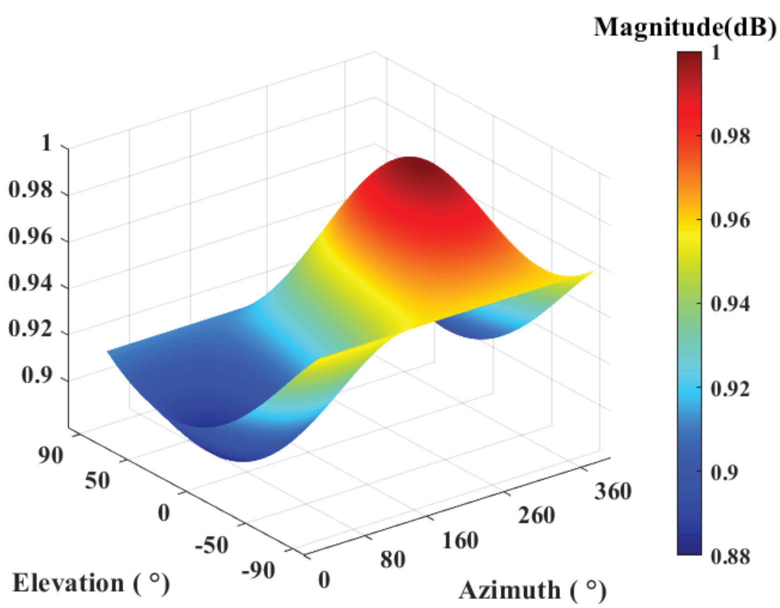
Figure 11. (a) VSA element 1 output for position 1.



(b)



(c)



(d)

Figure 11. (b) Spectrogram of hydrophone output, (c) Frequency spectrum, and (d) Beam former output for 3 kHz.

for position 2, 3 kHz and 4 kHz have a good correlation with actual angle compared to other frequencies.

4.1.1 Statistical Performance Analysis

To investigate the performance of the above-mentioned algorithms from a statistical point of view, Root Mean Square Error (RMSE) of DoA is estimated with varying SNRs. The RMSE is defined as[25]

$$RMSE = \sqrt{\frac{1}{n}\sum_{n-1}^{n}\left(\theta^{(n)} - \theta_a\right)^2} \tag{20}$$

where, n is the number of Monte Carlo trials, $\theta^{(n)}$ is the estimated DoA value in the n^{th} Monte Carlo simulation and θ_a is the true DoA value. The performance of algorithms is measured by calculating the RMSE under varying levels of additive white noise and number of Monte Carlo trials is 100. It is clearly shown in Fig.10 that the RMSE decrease as the SNR increases and MUSIC with SVD performs better for low SNR than the other algorithms.

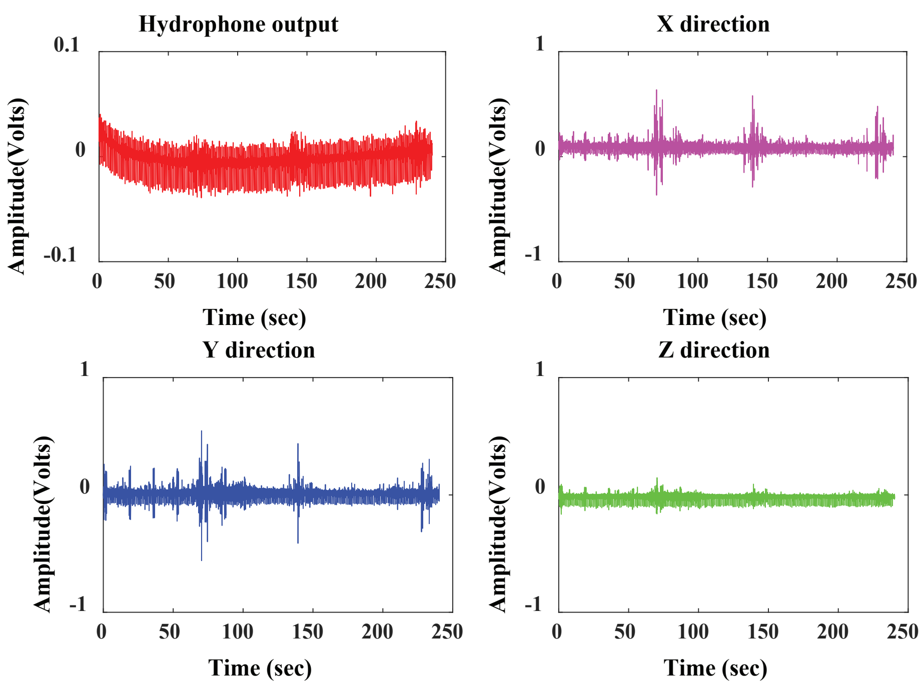
4.2 Results from Harbor Data

The acquired known source harbor data is analyzed, and beam forming is performed with MUSIC with the SVD algorithm. The estimated azimuth angle from the algorithm is corrected from compass data. Figure 11 (a) shows the hydrophone output and x, y, and z components of particle accelerations from the first element of the vector sensor array, and Fig. 11 (b) presents the spectrogram of the hydrophone output, which shows the transmitted signals from 0.5 kHz to 6 kHz frequency for position 1(P1) of the transmitter. Initially, low-frequency signals (>2 kHz) are transmitted from 1 sec to 150 sec, and then 2 kHz-6 kHz frequencies are transmitted from 150 sec to 240 sec.

Figure 11(c) presents the frequency spectrum of the transmitted signal, and Fig. 11(d) shows the beam former output from the SVD MUSIC algorithm. The actual angle for this position is 264°, and the estimated angle is 270° with an elevation angle of -5 °. Here, the depth of the water column is only 5 m; hence, the signal interference is greater, which results in a noisy signal with the desired signal output, which affects the direction of arrival estimation results.

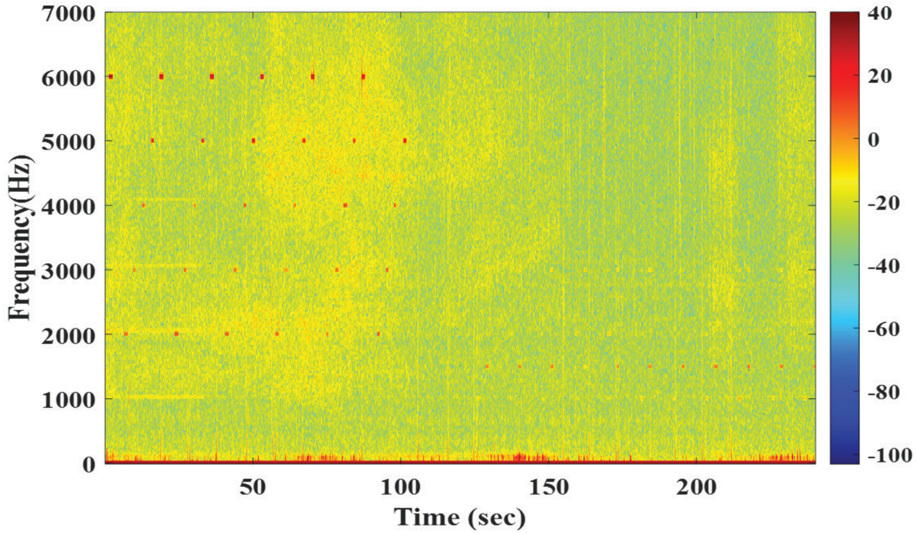
Similarly, the results are obtained from transmitter position 2 (P2) and shown in Fig. 12(a), 12(b), 12(c), and 12(d). The actual angle for this position is 202°, and the estimated angle is 207° with an elevation of -3°. The beam-forming algorithm is performed for all the Fig. 13(a) and 13(b) present estimated and actual azimuth angles for the SVD MUSIC algorithm for frequency 2 kHz to 6 kHz for transmitter position 1 and transmitter position 2, respectively.

The maximum variation in elevation is -2° for all frequencies. The signal level is poor for 0.5 kHz and 1 kHz, and the number of cycles in the received signal for these two frequencies is insufficient to perform a beam-forming operation, so the results are not present for these frequencies. The results reveal that 3 kHz and 6 kHz frequencies have a good correlation between the estimated and actual angles for positions 1 and 5 kHz, working well with the actual azimuth for transmitter position 2, compared to other frequencies. Actual/harbor environments are often characterized by high noise levels, including ambient noise from wind, waves, and vessel traffic. Analyzing the performance of the MUSIC method in such noisy conditions provides insights into its robustness and ability to suppress noise while accurately estimating DoAs. Maintaining accuracy in noisy environments is essential for reliable target detection and tracking in maritime surveillance applications. We can infer essential signal propagation characteristics in marine environments by deploying sensors or transducers in

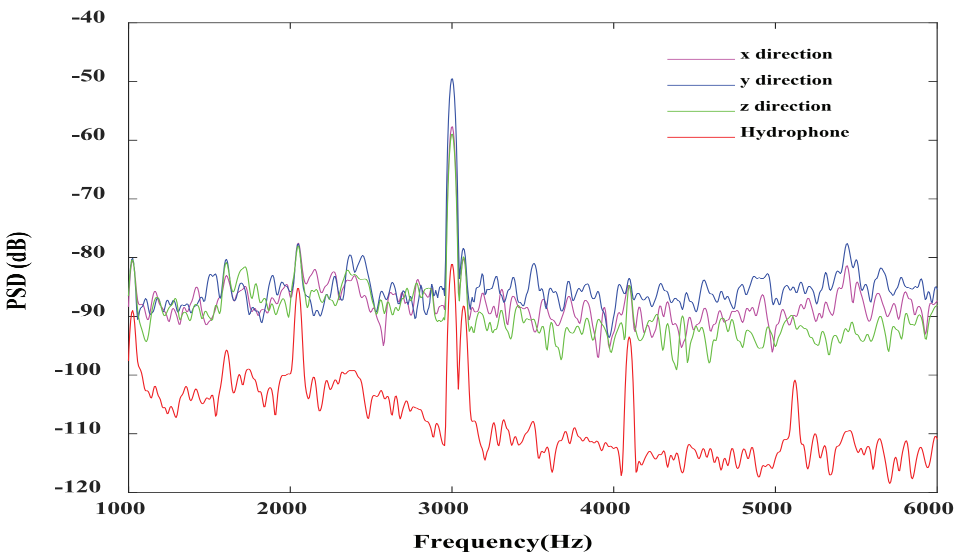


(a)

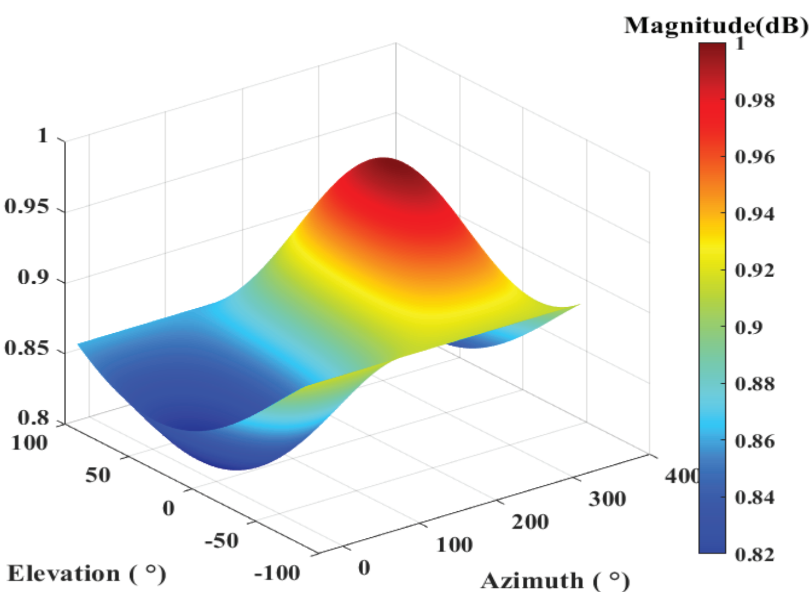
Figure 12. (a) The hydrophone output and x, y, and z component from the first element of the vector sensor array from position 2.



(b)

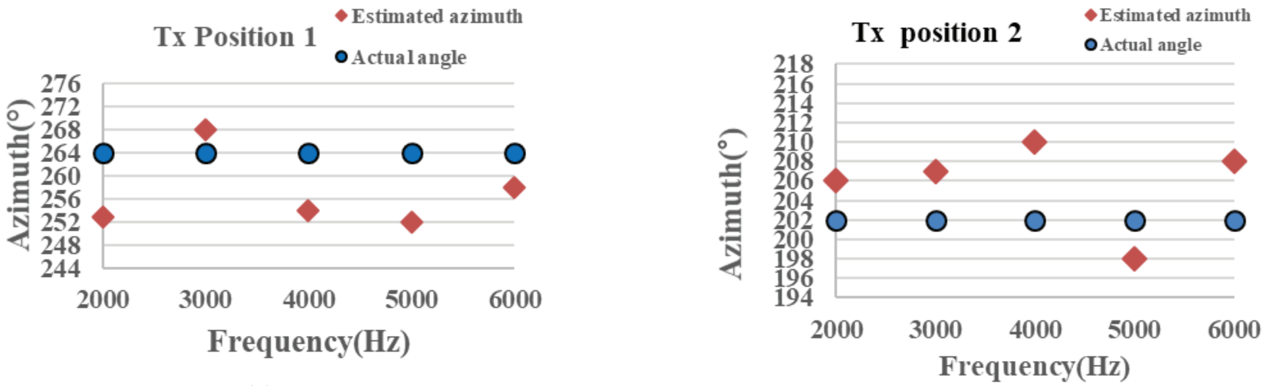


(c)



(d)

Figure 12. (b) Spectrogram of hydrophone output, (c) Frequency spectrum, and (d) Beam former output for 3 kHz.



(a) (b)

Figure 13. Estimated and actual Azimuth angles for SVD MUSIC algorithm for frequency 2 kHz- 6 kHz for (a) Transmitter position 1 and (b) Transmitter position 2.

a harbor environment and measuring acoustic signals. This includes understanding factors such as attenuation, dispersion, reflection, and refraction, which are crucial for designing and optimizing communication, navigation, and sensing systems in maritime settings.

5. DISCUSSION

The work carried out to study the performance of Direction of Arrival (DoA) estimation algorithms for underwater sound source localization using an acoustic Vector Sensor Array (VSA) gives the confidence to perform more field trials in an open ocean environment and use it for surveillance applications. The VSA system has three elements, and each element is built with a triaxial accelerometer and cylindrical hydrophone with a preamplifier. The system has a working frequency of 6 kHz and is neutrally buoyant. The performance of the VSA with existing sensor arrays is better in terms of accuracy, reliability, and efficiency. Najeem, *et al.*[15] used conventional beamforming technique to estimate DoA using VSA. Agni Mantouka, *et al.*[26] describe a Dual Accelerometer Vector Sensor (DAVS) manufacturing and testing. Tests were conducted in a calibration tank to measure the sensitivity, and directionality and the experimental results suggest that without accounting for the accelerometer response difference and with no correction for the AUV motion, the acoustic source could be tracked using accelerometer. Fabrication and underwater testing of a 3D vector hydrophone with three accelerometers operating in x, y, and z directions is presented by Roh, *et al.*[27].

Many simulation and tank-based literature is present where a comparison of various algorithms has been performed[28-32] and the results showed that the MUSIC algorithm has a higher resolution. In the current work, for the estimation of DoA, MUSIC with SVD is implemented since this technique increases the accuracy of the results and is also computationally efficient. Table 3 shows the comparison of current work with existing work.

Table 3. Comparison with existing work

Reference for existing work	Maximum variation from actual azimuth (in degree)
Gou, *et al.*[33]	17.5
Latha, *et al.*[16]	20
Najeem, *et al.*[15]	15
Current work	4 to 12

6. CONCLUSION

The objective of this work is to analyze the performance of the various algorithms, i.e., conventional beam forming, Multiple Signal Classification (MUSIC) with Eigen value decomposition, and MUSIC with Singular Value Decomposition (SVD) for estimating the Direction of Arrival of an underwater acoustic source, from a laboratory experiment initially and then apply the technique that gives more accurate results to open sea experimental data using a VSA. In the laboratory experiment, a comparison of the three different algorithms for resolution and accuracy has been carried out in terms of the number of array elements. The results show that at 3 kHz and 4 kHz, the techniques perform well in laboratory experiments. It is seen from this experiment that reliable estimates and good resolution can be obtained even with a small aperture array.

Laboratory experiment results reveal that Music with SVD provided a good result with a three-element VSA array, and the same has been applied while conducting experiments at the harbor. A low signal-to-noise ratio directly impacts the direction of arrival estimation. All three algorithms perform well in high SNR conditions, and performance degrades at low SNR conditions. The results of the known source transmission experiment conducted in the harbor are encouraging. The difference between the actual and estimated DoA for the harbor experiment is due to low SNR and reflections from the sea surface and seabed in the open environment. This work establishes the good performance of the vector sensor array system for the direction of arrival estimation in an open sea dynamic environment. More field tests need to be carried out to enhance the direction of arrival estimation towards coastal surveillance applications.

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