DIRECTION OF ARRIVAL ESTIMATION BASED ON SMOOTH MUSIC ALGORITHM COMBINED SPECTRUM SELECTION TECHNIQUES OF NOISE CHARACTERISTICS OF MARINE TARGETS EQUIPPED PROPELLER

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Abstract

This article demonstrates a proposal for a direction of arrival (DOA) estimation algorithm for passive sonar systems based on the combination of the smooth multiple signal classification (Smooth MUSIC) algorithm and frequency bin selection (BS) of the sound of marine propeller-equipped targets, so-called BS-SMUSIC. The harmonics of propeller shaft and blade frequencies are considered useful information for detecting targets and estimating their parameters, including DOA. Low frequency analysis recording (LOFAR) is utilized to implement BS. By analyzing MUSIC-based algorithm together BS, passive DOA estimation algorithms have been significantly improved. Simulation results have been given in both cases: uncorrelated and correlated sources. Outcomes demonstrate that the BS-SMUSIC algorithm has the best performance in comparison to three MUSIC-based algorithms (MUSSIC, Modified MUSIC and Root MUSIC) in all investigation cases: narrow band signals at 5 dB of signal to noise (SNR); wideband signals at -5 dB of SNR. Besides, BS-SMUSIC algorithm also archives the lowest root mean square error (RMSE) in SNR range -5 dB to 20 dB. In the SNR range, the RMSE of BS-SMUSIC algorithm remains stable and is about 0.3° at -5 dB of SNR. This performance reflects its ability to ensure stable operation, super resolution and high accuracy of BS-SMUSIC algorithm for the passive sonar systems.

Index terms

Underwater surveillance, passive sonar, marine targets, DOA, MUSIC algorithm, line spectrum selection, propeller.

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1. Introduction

The MUSIC is a DOA estimation algorithm based on subspace techniques, proposed in the last decade 90s of the twentieth century [1]. This algorithm was initially designed to address the limitations suffering from poor resolution capabilities found in traditional DOA algorithms like Bartlett and Capon [2]. MUSIC-based algorithm earns the classification of "super resolution" algorithm owing to its exceptional resolution capabilities. Despite its numerous advantages, the MUSIC algorithm still has several limitations. One of those limitations is its performance in scenarios with closely spaced sources, or sources that are correlated, or both [3].

Several variations of the MUSIC algorithm have been proposed to overcome such limitations, including the Modified MUSIC (MMUSIC), Root MUSIC (RMUSIC), and Smooth MUSIC (SMUSIC) algorithms [2]. The RMUSIC algorithm is proposed for a unified linear antenna array (ULA) only, instead of examining the spatial spectrum function like the MUSIC algorithm, RMUSIC finds the solutions of characteristic polynomials. In terms of computation, RMUSIC basically does not reduce the time and complexity compared to MUSIC, but it offers more advantages in conditions of low signal-to-noise (SNR) ratio and close sources. Multipath propagation is a common phenomenon for underwater acoustic waves, occurring in both shallow water and deep water regions. When this phenomenon occurs, signals from direct paths and indirect paths will correlate with each other. The SMUSIC and MMUSIC algorithms are proposed to solve the direction finding problem in this situation. The two ones have used different methods to process the covariance matrix, **R**, in order to reduce the effects of correlated signals. As mentioned above, there are varying degrees of improvement, it is clear that each modified algorithm will be more suitable and effective than the original one.

Besides, for implementing most DOA algorithms in general, including the MUSIC algorithm specifically, it is necessary to know the number of signal sources, but determining the number of signal sources is also a big challenge. Many studies have used the minimum description length (MDL) technique and the Akaike information criterion (AIC) to determine the number of signal sources [4]. However, to count the number of sources according to [4], it is necessary to estimate the covariance matrix and make the eigenvalue decomposition of the matrix, **R**. In case of correlated sources, it will suffer large errors, leading to wrong results when solving the DOA estimation problem. To overcome this limitation, adaptive methods based on the spatial time frequency distribution have been considered for frequency-modulated sources [5] or using centralized methods to estimate multipath before orientation [6]. However, these studies still have several limitations, as follows. Firstly, if only processing wide-band signals in the frequency domain, it is necessary to take additional steps to estimate the number of sources, which affects the performance of the algorithm. Secondly, it will take more time and the computational costs when considering the time-frequency distribution of the signal.

Passive sonar systems are used to detect, locate, classify, and identify marine targets (surface targets and underwater targets). Although these targets have different characteristics, but most marine targets are equipped with a propulsion system for movement, including a propeller component. The propeller noise spectrum has a complex structure, a wide frequency range stretching from infrasound to several thousand hertz, and acoustic emission characteristics that depend on the carrier speed as well as operating conditions. However, they share the common feature of having characteristic discrete spectral components (line spectrum) in the infrasound and acoustic regions [7]. In addition, most base protection sonar and submarine surveillance systems are deployed in shallow sea areas, where multipath propagation occurs, leading to the need for accurate signal source DOA estimation.

This article proposes a novel approach based on the combination of the SMUSIC algorithm and the line spectrum selection, called bin selection, to maximize the useful information of marine targets, improving the performance of the system. This proposal utilizes techniques to select the low-frequency spectral region (below 100 Hz), followed by spectral line components, which are the harmonics of shaft and blade frequency. Then, the filtered signal will be processed to simultaneously estimate the number of signal sources and perform direction finding using the SMUSIC algorithm. As a result, the SNR is significantly improved. This solution not only enhances operational capabilities and accuracy of the direction-finding algorithm but also addresses the issue in scenarios with correlated sources.

The remainder of this article is organized as follows: Section 2 describes the signal model, including the mathematical model of propeller and the array receiver signal model; Section 3 presents proposed method; Section 4 shows experiment results and discussion; and finally, conclusions are made in Section 5.

2. Signal model

2.1. Signal model of noise generated from marine targets

Assuming the $r-th$ target is located in the direction θ_r ($r \in [1 \div \Gamma]$), which N_r is the number of propeller blades, and the rotation frequency of the propeller shaft is f_r. Then, the sound spectrum generated by this target has components that are harmonics of the shaft rotation frequency and blades rotation frequency $f_r \times N_r$. The assumption is that if the $r-th$ target has a total of L_r harmonics, then line spectrum components of the target can be represented [7]–[9]:

$$f_{dr} = lf_r \tag{1}$$

in which, l is the harmonic index, $l \in [1 \div L_r]$.

These frequencies have different characteristics for each type of target and change according to the target's operating conditions. Hereafter, they will be called characteristic frequencies (or characteristic line spectrum). Based on the above analysis, the mathematical model of the sound signal ($r-th$ target) can be considered to have the form [7], [9] :

$$s_r(t) = s_{rd} + s_{rc} = \sum_{l=1}^{L_r} A_l \mathrm{e}^{-j2\pi.(l.f_r).t} + ((1 + a_r(t)).G_{rc}(t)) \tag{2}$$

where: s_{rd} is the time domain representation of the signal component containing characteristic frequencies; A_l is the corresponding amplitude of $l-th$ spectral line; s_{rc} is the time domain representation of the continuous spectrum of the sound signal, where $a_r(t)$ is the amplitude modulation function and $G_{rc}(t)$ is the time domain representation of the component containing the continuous spectrum of the signal.

In real-world scenarios, when dealing with marine target signals, which often exhibit multiple spectral components, it becomes necessary to apply broadband DOA techniques. Broadband DOA problems is naturally synthetic results of narrowband problems in the particular method [10].

2.2. Antenna array model, parameters of model and targets

Assumptions: ULA antenna has M hydrophones with the same receiving characteristics ($Hyd\ No.\ 0,\ Hyd\ No.\ 1, ...,\ Hyd\ No.\ M-1$). The elements are spaced evenly apart at a distance of D (constrained to not exceed half a wavelength). The first element ($Hyd\ No.\ 0$) is taken as the referent element. There are Γ targets in the observed water area. The speed of sound in water is c. The passive sonar system illustration diagram is shown in figure 1.

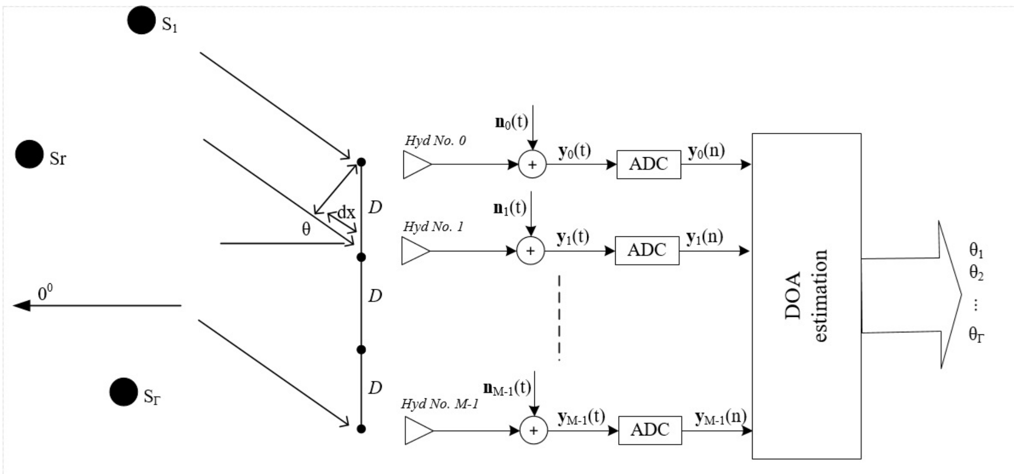

Fig. 1. Passive sonar model with ULA antenna.

Sound generated from targets is received by the ULA hydrophones. It is believed that the hydrophones' thermal noise and ambient noise have an additive effect on the system, $\mathbf{n}_m(t),\ m \in [0 \div M-1]$. The received signals, $\mathbf{y}_m(t),\ m \in [0 \div M-1]$, are digitized by analog-to-digital converters (ADCs) into digital signals, $\mathbf{y}_m(n),\ m \in [0 \div M-1]$.

The DOA algorithm is implemented on a digital computing device and provides the desired angles of the targets to be determined. The model of the received signal at the output of the antenna elements has form:

$$\mathbf{Y} = \mathbf{A}.\mathbf{S} + \mathbf{e} \tag{3}$$

where: $\mathbf{Y} = [\mathbf{y}_0(t),\ \mathbf{y}_1(t),\ ...,\ \mathbf{y}_{M-1}(t)]^T$ is the received signal matrix (with noise added), size of $M \times N$, where N is the number of snapshots); $\mathbf{y}_m(t)$ is received signal vector on $m - th$ hydrophone, size of $1 \times N$; $\mathbf{S} = [\mathbf{s}_1(t),\ \mathbf{s}_2(t),\ ...,\ \mathbf{s}_\Gamma(t)]^T$ is the sources signal matrix, size of $\Gamma \times N$; $\mathbf{e} = [\mathbf{n}_0(t),\ \mathbf{n}_2(t),\ ...,\ \mathbf{n}_{M-1}(t)]^T$ is the additive noise, size of $M \times N$. In this article, assuming that the noise follows a Gaussian distribution with zero mean, and variance $\sigma_0^2 = \sigma_1^2 = ... = \sigma_{M-1}^2 = \sigma^2$; and $[.]^T$ is the transpose operator; $\mathbf{A} = [\mathbf{a}(\theta_1), \mathbf{a}(\theta_2),\ ..., \mathbf{a}(\theta_\Gamma)]$ is the steering matrix, size of $M \times \Gamma$ where $\mathbf{a}(\theta_r)$ is the steering vector of $r - th$ source. For a narrowband source, central frequency f_c, located at direction θ_r, steering vector can be represented:

$$\mathbf{a}(\theta_r) = \left[1, e^{-j\phi_r}, ..., e^{-jm\phi_r}, ..., e^{-j(M-1).\phi_r}\right]^T, \tag{4}$$

where, $\phi_r = (2\pi f_c dx)/c = (2\pi f_c D \sin(\theta_r))/c$.

3. Proposed methods

The MUSIC algorithm, introduced by Schmidt in 1986 [1], relies on subspace decomposition. The covariance matrix, **R**, is calculated by:

$$\mathbf{R} = E\left[\mathbf{Y}\mathbf{Y}^Y\right] \tag{5}$$

where $E[.]$ is the expectation operator, and **R** is a $M \times M$ matrix with rank Γ, and it is subjected to eigenvalue analysis:

$$\mathbf{R} = \mathbf{U}\Sigma\mathbf{U}^{-1} \tag{6}$$

where, orthogonal matrix, $\mathbf{U} = [u_1, u_2, ..., u_M]$, contains eigenvectors, and eigenvalues are the diagonal elements of diagonal matrix, $\Sigma = diadag(\lambda_1, \lambda_2, .., \lambda_M)$. Without loss of generality, the assumption that the eigenvalues are arranged in descending order, corresponds to it: first Γ eigenvectors of U are the vectors contained signal information, forming signal subspace $\mathbf{U}_S = [u_1, u_2, ..., u_\Gamma]$, the remainder of **U** is $\mathbf{U}_N = [u_{\Gamma+1}, u_{\Gamma+2}, ..., u_M]$ is considered noise subspace. Here, **U** is presented as a form of $\mathbf{U} = [\mathbf{U}_S\ \mathbf{U}_N]$. Then, the spatial spectrum function (there are some documents called pseudo spectrum) for the MUSIC algorithm is established:

$$P_{MUSIC}(\theta) = \frac{1}{\mathbf{a}^H(\theta).\mathbf{U}_N\mathbf{U}_N^H.\mathbf{a}(\theta)} = \frac{1}{\left\|\mathbf{U}_N^H.\mathbf{a}(\theta)\right\|^2} \tag{7}$$

where, $[.]^H$ is the complex conjugate transpose operator.

In several cases, especially in low SNR, or in resolution improvement situations, the improvements are applied to the MUSIC algorithm by examining the polynomial instead of directly examining the spectral function. This algorithm is named RMUSIC. And the algorithm was proposed in 1983 [1] and only applied to ULA antenna. Here, polynomial $J(z) = P_{MUSIC}^{-1}(\theta)$ is considered instead of spectral function (7), this polynomial has $2(M-1)$ roots, those are $z = \exp(j.(2\pi f D/c).\sin(\theta))$, but only roots lying within the unit circle are considered. Within those points, the Γ points closest to the unit circle will be the points reflecting the direction angles of Γ signal sources: $\theta_r = \arcsin\left[(c/2\pi f_c D).\arg(z_r)\right],\ r = 1 \div \Gamma$. In cases of correlated sources, the MUSIC and RMUSIC algorithms have the inherent problem of being inapplicable to correlated source cases. At that case, $\mathbf{R}$ does not have rank Γ. As a result, the calculation results of (7) and (8) are no longer accurate. To overcome that limitation, the MMUSIC and SMUSIC [2] algorithms have been proposed. Accordingly, for MMUSIC, the covariance matrix $\mathbf{R}$ is added with another matrix $\mathbf{R_1}$ to form the modified covariance matrix $\mathbf{R}_\Sigma$, and the algorithm steps are performed similarity to the MUSIC algorithm; the spatial spectrum function is calculated with new noise subspace. With $\mathbf{J}$ being the $M \times M$ exchange matrix (with anti-diagonal elements equal to 1, remaining elements equal to 0), $\mathbf{R_1}$ is formed:

$$\mathbf{R_1} = E\left[\left(\mathbf{J}\mathbf{Y}^*\right)\left(\mathbf{J}\mathbf{Y}^*\right)^H\right] = \mathbf{J}\mathbf{R}\mathbf{Y}^*\mathbf{J} \tag{8}$$

where, $\mathbf{Y}^*$ is the complex conjugate matrix of $\mathbf{Y}$ and then:

$$\mathbf{R}_\Sigma = \mathbf{R} + \mathbf{R_1} = \mathbf{A}\mathbf{R}_s\mathbf{A}^H + \mathbf{J}\left[\mathbf{A}\mathbf{R}_s\mathbf{A}^H\right]^*\mathbf{J} + 2\sigma^2\mathbf{I}. \tag{9}$$

According to matrix theory, $\mathbf{R}_\Sigma, \mathbf{R}, \mathbf{R_1}$ matrices have the same noise subspace. Then, $\mathbf{R}_\Sigma$ will be taken to eigenvalue analysis for the same step like MUSIC algorithm.

Also solving the problem of the correlated sources, the SMUSIC algorithm divides the M hydrophone array into K sub-arrays, with overlap; each sub-array has Q hydrophones. The covariance matrix is created by taking a smooth average of the covariance matrices of the sub-array, accordingly:

$$\mathbf{R}_{SL} = \frac{1}{K}\sum_{k=1}^{K}\mathbf{R}_k = \frac{1}{K}\frac{1}{N}\sum_{k=1}^{K}\sum_{j=1}^{N}\mathbf{Y}_k(j)\mathbf{Y}_k^H(j). \tag{10}$$

Up to $K-1$ correlation signals can be DOA estimated. This is possible because the new correlation matrix is a rank Γ matrix.

SMUSIC algorithm is chosen for the proposal to ensure DOA estimation in both uncorrelated and correlated sources. A system applying the algorithm BS-SMUSIC is illustrated on figure 2.

There are three main stages on the diagram: LOFAR (Low Frequency Array), FBS&SNE (Frequency Bins Selection and Source Number Estimation), and the SMUSIC. In which, LOFAR will perform sampling and convert signal into digital one (S/H&A/D), followed by low-pass filtering (FIR low pass filter), and finally, fast Fourier transform (FFT) to provide signal in frequency domain for the next stage,

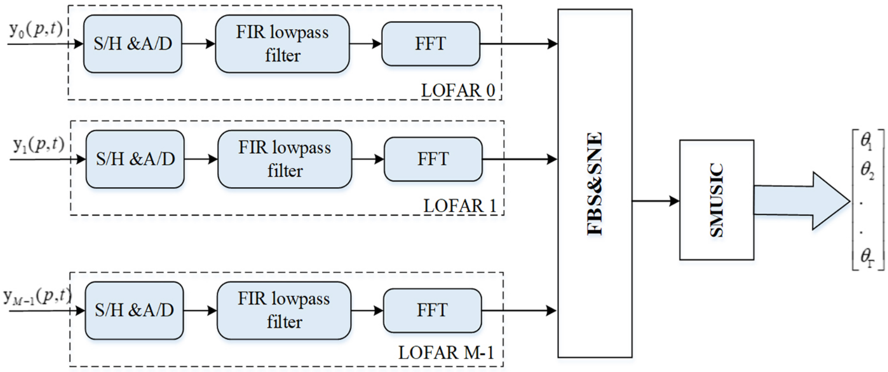

Fig. 2. System functional block diagram applying the BS-SMUSIC algorithm.

details about LOFAR are presented in [8] and SMUSIC in [11], respectively. The BS-SMUSIC algorithm is implemented in two stages: FBS&SNE and SMUSIC.

Depending on the number of FFT points (N), the considered frequencies are taken at values $f_i = i.\left(f_s/N\right)$, where f_s is sampling frequency and $i = 1 \div N$, each of these frequencies is called bin. With the BS-SMUSIC algorithm, SMUSIC is not applied for all frequency bins but to chosen bins. FBS&SNE has two functions: Select characteristic bin (BS) and estimate the number of signal sources. In particular, the signal at the FBS&SNE input has the form:

$$\mathbf{Y}(p,f) = \left[\mathbf{y}_0(p,f),\ \mathbf{y}_1(p,f), ..., \mathbf{y}_{M-1}(p,f)\right]^T \tag{11}$$

where: $\mathbf{y}_m(p,f) = \sum\limits_{n=0}^{N-1} \mathbf{y}_m(p,n).e^{-j.2\pi.f_s.n/N} = \left|Y_m(p,f)\right|.e^{j\varphi(p,f)}$ is the signal vector at the output of $m-th$ LOFAR. In wideband processing, P data frames will be processed instead of one data frame in narrowband processing, and p is the index of data frame.

In several publications, the line spectrum of a ship's propeller sound was extracted by several different methods [12]–[15]. However, these methods are computationally complex and time-consuming. Furthermore, in the case of a stationary signal, the spectrum of the received signal is quite stable in terms of frequency, so the uncorrelated accumulation method can be used to calculate the average spectrum:

$$\mathbf{y}_\Sigma(f) = \frac{1}{M.P}\sum_{p=1}^{P}\sum_{m=0}^{M-1}\left|\mathbf{Y}_m(p,f)\right|. \tag{12}$$

Assuming that the sources have the same total harmonic L_r, equal to L, then $\mathbf{y}_\Sigma(f)$ is a $1 \times N$ vector and contains $\Gamma \times L$ characteristics of line spectrum. The task of the FBS&SNE stage is to select all characteristic spectral lines and estimate the number of signal sources. The flow chart of the FBS&SNE process is shown in figure 3. The input of FBS&SNE is the average spectrum $\mathbf{y}_\Sigma(f)$ (alternatively, it can be represented using frequency indices, $\mathbf{y}_\Sigma(g),\ g \in [1 \div N]$), the threshold level P_0, and the potential frequency range $[f_{\min} \div f_{\max}]$ containing the frequencies of the shaft. This frequency range is selected based on prior information and depends on the development of ship hull technology. Typically, $[f_{\min} \div f_{\max}]$ is chosen to be $0.2 \div 10\ Hz$.

For each spectral component within the selected frequency range, $\mathbf{y}_\Sigma(f)$, $f \in [f_{\min} \div f_{\max}]$, equivalent to $\mathbf{y}_\Sigma(g),\ g \in [g_{\min} \div g_{\max}]$, the cumulative spectral power is calculated with a total of L components:

$$P_{pow}(g) = \sum_{l=1}^{L} \mathbf{y}_\Sigma(l.g). \tag{13}$$

If $P_{pow}(g) \geq P_0$ then an additional target is recorded, and the characteristic frequency domain is supplemented with spectral lines corresponding to that target. At the same time, a set of covariance matrices $\mathbf{R}_\mathbf{\Omega}$ for those frequencies is also added. As a result, after the FBS&SNE stage, the SMUSIC algorithm is provided: Number of targets, set of covariance matrices $\mathbf{R}_\mathbf{\Omega}$ corresponding to frequency bins in the set $\mathbf{\Omega}$.

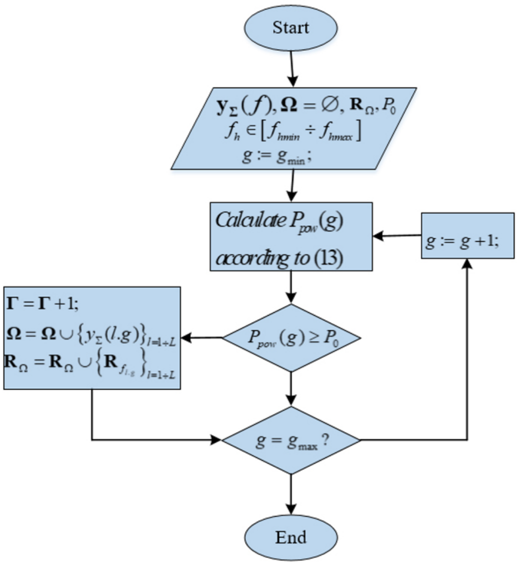

Fig. 3. Flow chart of the FBS&SNE process.

Spectrum function of BS-SMUSIC algorithm:

$$\mathbf{P}_{BS-SMUSIC}(\theta) = \frac{1}{n(\Omega)} \sum_{i \in \Omega} \mathbf{P}_{SMUSIC}(f_i, \theta). \tag{14}$$

As a results, angles of sources, $\boldsymbol{\theta} = [\theta_1, \theta_2, ..., \theta_\Gamma]^T$, are estimated.

4. Simulation results and discussion

4.1. General parameters and conventions

The common symbols and parameters used for simulation are chosen as follows: a ULA antenna array consisting of M hydrophones with inter-distance between hydrophones is $D = 10$ m; $c = 1540$ m/s is the underwater sound speed and $f_s = 1000$ Hz is sampling frequency; hydrophone number in sub-array is L_s, and ρ is attenuation coefficient of the correlated source.

4.2. Scenario 1:

Narrow-band sources are used to analyze the performance of the MUSIC-based algorithm in both correlated and uncorrelated cases.

4.2.1. Input: The input parameters for uncorrelated sources are shown in table 1.

Table 1. The input parameters for uncorrelated sources

Parameter	Γ	Kind	θ_1	θ_2	f_{r1}	f_{r2}	SNR	M	L_s	N
Value	2	sin	$-10°$	$20°$	35 Hz	31 Hz	5 dB	24	12	512

In case of correlated sources: $f_{r1} = f_{r2} = 35$ Hz, and $\rho = 45\%$.

4.2.2. Results: The results are shown in figure 4.

It is clearly seen that the correct results are shown by all four algorithms in case of uncorrelated sources (a), the two spectral peaks are found in angles $-10°$ and $+20°$. SMUSIC algorithm presents the best performance with the gap between spectral peak and background level (GPBL) is 37 dB, follows by 27 dB for MMUSIC algorithm and 21 dB for standard MUSIC. Also, the RMUSIC algorithm produces accurate results. Moreover, there is a significant difference results for the case of two correlated sources (b) compared to the previous one. For this case, the first source at $-10°$ has parameters similar to the first source in the uncorrelated case, while the second one is assumed to be the indirect first source, after multiple reflections due to multipath propagation, reaching the receiver at $+20°$ with an attenuation coefficient of ρ. The MUSIC and RMUSIC algorithms have reduced functionality. Meanwhile, the SMUSIC and MMUSIC algorithms still yield accurate results. And the GPBL of the SMUSIC algorithm still remains at its highest level, which is 15 dB higher than that of the MMUSIC algorithm.

4.3. Scenario 2:

The performance of the four algorithms, including MUSIC, BS-MUSIC, BS-MMUSIC and BS-SMUSIC, are assessed by simulated broadband signals in both correlated and uncorrelated cases.

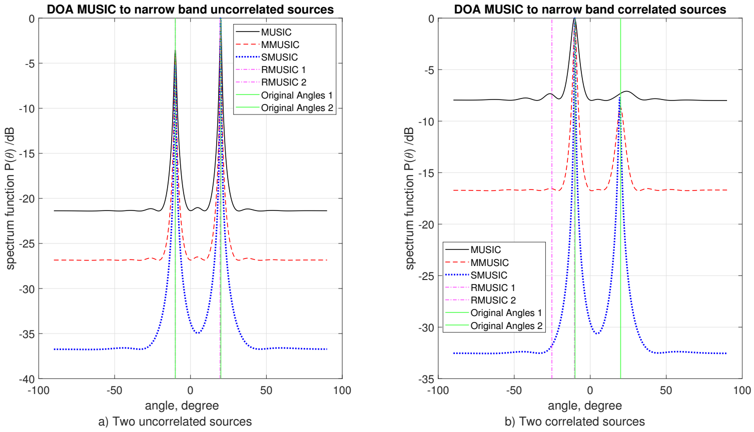

Fig. 4. Applying MUSIC-based algorithm to two narrow sources.

4.3.1. Input: The input parameters for uncorrelated sources are shown in table 2

Table 2. The input parameters for uncorrelated sources

Parameter	Γ	Kind	θ_1	θ_2	f_{r1}	f_{r2}	SNR	M	L_s	N	P
Value	2	Set of sin (L1=L2=8)	$-10°$	$20°$	7.5 Hz	8.7 Hz	-5 dB	24	12	512	256

In case of correlated sources: $f_{r1} = f_{r2} = 7.5$ Hz, and $\rho = 45\%$.

4.3.2. Results: The results are shown in figure 5.

In the case of uncorrelated sources (a), similar to the case of narrowband sources, all algorithms yield accurate directional results. And BS-MUSIC-based algorithms perform better than the original algorithm. The highest GPBL is above 35 dB, which belongs to BS-SMUSIC algorithm, follow by BS-MMUSIC, and BS-MUSIC and MUSIC algorithms at 21 dB, 17 dB and 9 dB, respectively. Besides, with the case of correlated sources (b), exactly as the mathematical foundation has determined, the MUSIC and BS-MUSIC algorithms have impaired functionality and do not produce correct results. The BS-SMUSIC algorithm is the best, still ensuring a narrow beam, and the GPBL is 32 dB. The BS-MMUSIC algorithm also gives correct results and has a GPBL of 16 dB and the beam is distorted at the second angle.

4.4. Scenario 3:

RMSE is assessed in the case of correlated sources based on the variation of SNR for SMUSIC-based and MMUSIC-based algorithms.

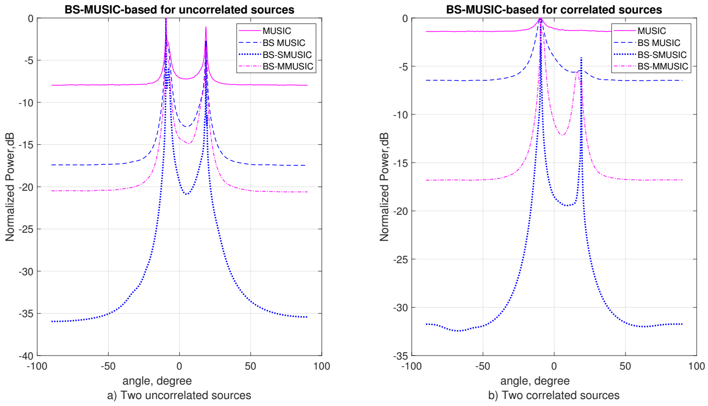

Fig. 5. Wideband DOA estimate by BS-MUSIC-based algorithms.

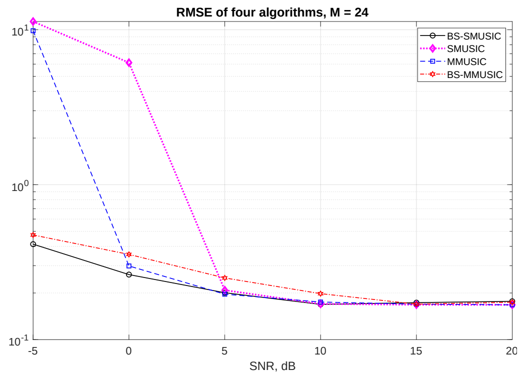

Fig. 6. Wideband DOA estimate by BS-MUSIC-based algorithms.

The input parameters for investigation in this scenario are similar to those in case of correlated sources, scenario 2. The difference is the variation of SNR, which change in range of $-5 \div 20$ dB, to assess the RMSE of the algorithms. And the results are shown in figure 6. According to these results, the algorithms combined with BS exhibit consistently increasing RMSE across the entire examined range, with acceptable errors when noise is high. In contrast, the non-combined BS algorithms show erratic increases in error as noise levels rise. Between $15 \div 20$ dB, all four algorithms demonstrate similar errors, with an RMSE of $0.1°$. Additionally, within the range of $-5 \div 15$ dB, the RMSE of the BS-MMUSIC algorithm steadily increases to $0.3°$. Meanwhile, the BS-SMUSIC algorithm follows a similar trend but consistently maintains a lower error, reaching about $0.3°$ at -5 dB.

5. Conclusions

The article has reviewed MUSIC-based DOA algorithms and the characteristics of the noise signal spectrum emitted from marine targets equipped with propellers to propose a method to increase the accuracy of DOA estimation. The article has assessed the performance of these DOA algorithms with and without combination BS. Simulations were performed for both cases: narrow band and wide band signals; correlated and uncorrelated sources. The results for narrowband DOA algorithms at $SNR = 5$ dB, and wideband DOA at $SNR = -5$ dB. Besides, RMSE is also surveyed at range of $SNR = -5 \div 20$ dB, the work prove that the BS-SMUSIC algorithm ensures the best accuracy and working efficiency in comparison to remain algorithms. In the case of strong noise, $SNR = -5$ dB, and correlated sources, this algorithm only produces an error of about $0.3°$. From the results achieved, it is possible to continue developing this algorithm to apply to passive underwater surveillance systems in sea areas with complex hydrological conditions.

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ĐỊNH HƯỚNG NGUỒN TÍN HIỆU TRÊN CƠ SỞ THUẬT TOÁN SMOOTH MUSIC KẾT HỢP KỸ THUẬT CHỌN LỌC PHỔ ĐẶC TRƯNG CỦA TIẾNG ỒN TỪ CÁC MỤC TIÊU BIỂN CÓ CHÂN VỊT

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Tóm tắt

Bài báo đề xuất giải pháp cho thuật toán định hướng (DOA) trong các hệ thống sonar thụ động trên cơ sở kết hợp thuật toán Smooth MUSIC và chọn lọc phổ đặc trưng (BS) của các mục tiêu biển có chân vịt, gọi là BS-SMUSIC. Các hài của tần số trục và tần số cánh được xem là các thông tin có ích để phát hiện mục tiêu và ước lượng các thông số của chúng, bao gồm bài toán định hướng. Ghi và phân tích tần số thấp (LOFAR) là kỹ thuật được chọn để thực thi BS. Bằng cách phân tích thuật toán trên cơ sở MUSIC kết hợp với BS, các thuật toán định hướng trong sonar thụ động được cải thiện đáng kể. Các kết quả mô phỏng được thực hiện với cả hai tình huống: các nguồn không tương quan và các nguồn tương quan. Các kết quả này cho thấy thuật toán BS-SMUSIC có hiệu năng tốt nhất so với ba thuật toán khác trên cơ sở thuật toán MUSIC (MUSIC, Modified MUSIC và Root MUSIC) trong tất cả các tình huống khảo sát: các tín hiệu băng hẹp có tỉ số tín/tạp (SNR) bằng 5 dB ; tín hiệu băng rộng có SNR bằng -5 dB. Hơn nữa, thuật toán BS-SMUSIC cũng đạt được sai số trung bình bình phương (RMSE) nhỏ nhất trong dải khảo sát của SNR (-5 dB đến +20 dB). Trong dải này, RMSE của BS-SMUSIC giữ ổn định và chỉ bằng 0.3° tại SNR bằng -5 dB. Hiệu năng này khẳng định khả năng làm việc ổn định, siêu phân giải và độ chính xác cao của BS-SMUSIC cho các hệ thống sonar thụ động.

Từ khóa

Giám sát ngầm, sonar thụ động, mục tiêu biển, định hướng, thuật toán MUSIC, chọn phổ rời rạc, chân vịt.