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An Adaptive Eigen-value Based Diagonal Loading Technique to Improve Wideband Direction of Arrival Estimation (DOA) Accuracy for Smart Antenna System.

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Abstract: The need for mobile communications is continually expanding, which drives up the demand for greater coverage, more capacity, and enhanced transmission quality. As a result, the radio spectrum needs to be used more effectively. The sort of antenna array that is adaptive Smart antenna systems hold out hope for a viable remedy to the issues with current wireless systems while attaining dependable and robust high-speed high-data-rate transmission. Smart antenna systems are capable of effectively exploiting the radio spectrum. Radars may effectively employ direction of arrival estimation (DOA) algorithms and interference cancellation to rebuild the original received signals and aid in the position determination of those signals. military surveillance, sonars, seismic exploration, and communications systems. Wideband signals are more challenging since they require more data and computer power to solve the same problem. Both fixed diagonal loading and diagonal loading based on Eigenvalues employ fixed diagonal loading factors that are independent of direction inaccuracy. The approaches, known as FDL and EDL, estimate the loading factor by adequately adapting already-presented narrowband beamforming techniques, and they demand perfect knowledge of the steering vector error, just like narrowband techniques do.

Wideband DOA estimation techniques like Incoherent Subspace Processing (ISSM) and Coherent Signal Subspace Processing (CSSM) with Eigen-value Based Diagonal Loading Technique are presented in this study to improve wideband DOA accuracy. A uniform linear array antenna was used to test the mathematical adjustment for various correlated and uncorrelated wideband signals entering at various incoming angles. with knowledge of or an estimate of the incoming signal direction. This document will be processed entirely on a computer using MATLAB.

Keywords: Coherent Signal Subspace Processing (CSSM), Incoherent Subspace Processing (ISSM).

Introduction

The development of adaptive antenna arrays is based on digital signal processing techniques. In order to improve reception in the direction of signals that are of interest while reducing interference in the direction of signals that are of no interest, the adaptive antenna array system gains the ability to find and track signals from both users and interference sources. The performance of an adaptive antenna array system is significantly influenced by the efficiency of digital signal processing techniques. Several DOA estimation approaches are used in adaptive antenna arrays to locate the target signal. The number of plane wave occurrences and their angles on the antenna array are calculated using the DOA algorithms [1].

Wideband sources can also be located using array processing techniques. As was previously indicated, compared to the center frequency, the frequency bandwidth for wideband transmissions is rather big. Compared to narrowband signals, which had an indefinite length and just one frequency, wideband signals had a bounded duration. These presumptions can be loosened to accommodate signals with narrow bandwidths compared to carrier frequencies and lengthy durations compared to array sizes. Wideband signals are more challenging since they require more data and computer power to solve the same problem. While the phase delays of the narrowband signals may be used to estimate the time delays, wideband signals require further signal processing before the detection and estimation issues can be resolved using the current techniques.

1. Wideband Processing Beamformer

Based on time-domain processing and frequency-domain processing, there are two primary methods for wideband beamforming [2]. These methods can create beam patterns that are frequency-invariant over a wide range of signal bandwidths. However, for signals with large bandwidths, the frequency-domain approach offers computational benefit compared to the time-domain method [2]. Figure (1) depicts the structural layout of a beamformer for frequency-domain processing. The fast Fourier transform (FFT) is used in this beamformer to convert wideband signals from each element into frequency domain, and a narrowband processor processes each frequency bin. the time-domain technique has a computational benefit [2].

Figure (1) depicts a beamformer's structural layout for frequency-domain processing. In this beamformer, the fast Fourier transform (FFT) is used to convert wideband signals from each element into frequency domain, and a narrowband processor processes each frequency bin. Additionally, the frequency-domain beamformer's employment results in inexpensive hardware costs because it does not call for a high A/D conversion sample rate. In contrast to the frequency-domain approach, which only needs a sampling rate equal to the Nyquist frequency, the time domain method often requires five to ten times the Nyquist rate in order to provide accurate beamforming. When a large number of antenna components are employed particularly in high-frequency bands, the cost associated with high sampling rates will be more noticeable. However, the data storage needs and computing effort (for example, for FFT, inverse FFT) are higher for frequency-domain beamformers.

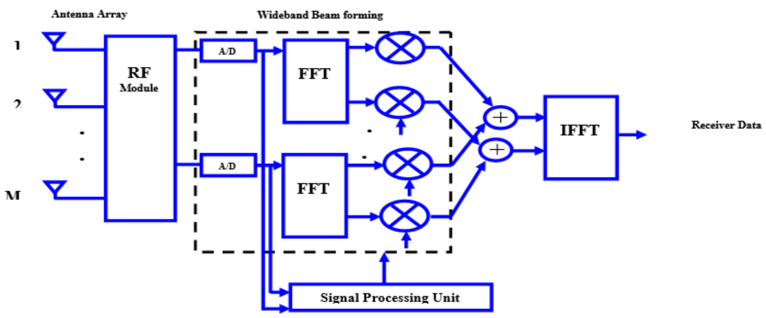


Fig. (1). Frequency-Domain Processing Beamformer.

The FFT is used in the FDFIB to convert wideband time-domain signals to frequency domain, and each frequency bin is subsequently weighted by a suitably set complex factor. A FDFIB's structure is seen in Figure (1). Any antenna array can be used in conjunction with the FDFIB. Additionally, the beam pattern's frequency-invariant property is only dependent on the phases of the beamforming weights. Since the amplitudes of the beamforming weights may be changed, the FDFIB has an advantage in manipulating beam shape.

2. Incoherent Frequency Combining Methods

These techniques are wideband adaptive DOAs. The typical method for processing wideband signals involves taking a spectrum sample of the incoming signals from each sensor to create an array of narrowband signals. The narrowband signals at each frequency are treated independently in the so-called incoherent signal-subspace processing approach, and the outcomes from all frequency bins are then merged to provide the final result. However, when the sources are coupled and the SNR is low, the performance of this technique suffers. The subspace fitting methods, specifically MUSIC [3], are among the techniques for incoherent wideband DOA estimation that are now accessible.

3.1 Wideband MUSIC Beamformer

The observation space is often divided into signal and noise subspaces using subspace decomposition techniques. Through the decomposition of the array correlation matrix into its Eigen-structure form, the signal and noise subspaces are estimated in the first stage of these methods. The phrase "signal subspace" refers to the region of space that is covered by the covariance matrix eigenvectors that correspond to the dominating eigenvalues. The fact that signals' Eigen-values are greater than those of noise is used by the detection techniques. The approach for calculating the specific frequencies of many time-harmonic signals, known as MUSIC (Multiple Signal Classification) [3], is one of the most widely used techniques. By performing the Fourier transform of the aforementioned equation, one may derive the vector X(t)'s representation in the frequency domain.

$$X(\omega,\theta) = A(\omega,\theta)S(\omega) + N(\omega) \qquad (1)$$

Where, $X(\omega,\theta) = [X_1(\omega,\theta), X_2(\omega,\theta),.......X_M(\omega,\theta)]^T$ is the M vector of array output.

$S(\omega,\theta) = [S_1(\theta), S_2(\theta),......S_d(\theta)]^T$ Is the d vector of source signals.

$N(\omega) = [N_1(\omega), N_2(\omega),.......N_M(\omega)]^T$ is the N vector of noise.

The array steering vector takes into account the far-field and assumes that the arrays are similar. A(ω,θ) can be written as $A(\omega,\theta) = [a(\omega,\theta_1),................a(\omega,\theta_D)]$.

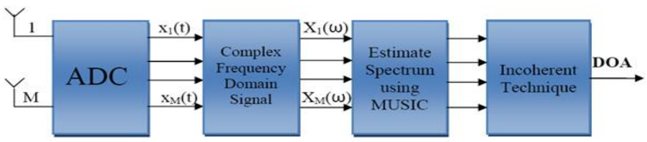


Fig. (2). Basic Principle of Incoherent Method.

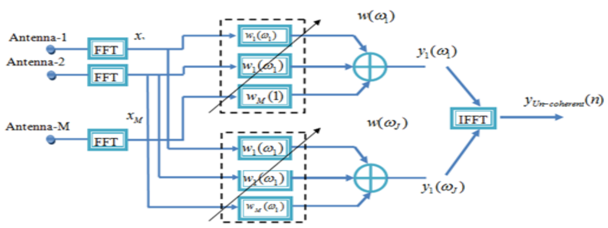


Fig. (3). Block Diagram of the Incoherent Wideband Adaptive Beamformer.

The output covariance matrix is used in all traditional beamforming methods to identify the source's direction of arrival. The output power at each of the sensors can be seen using the covariance matrix, and the matrix may simply be changed to achieve improvements at all of the sensors. [3] provides the symmetric covariance matrix of X in Eq. 2.

$$R_{XX}(\omega,\theta) = E\left[X.X^{H}\right] = \frac{1}{M}\sum_{n=1}^{M} X(\omega,\theta)X(\omega,\theta)^{H} \qquad (2)$$

Where, E[·] is the expectation operator and $[.]^{H}$ denotes complex conjugate transpose.

$$R_{XX}(\omega,\theta) == \frac{1}{M}\sum_{n=1}^{M}(A(\omega,\theta)S(\omega)+N(\omega)).$$
$$(A(\omega,\theta)S(\omega)+N(\omega))^{H} \qquad (3)$$

The Eq. 3.11 is further calculated as Hence the Covariance matrix is given by.

$$R_{XX}(\omega,\theta) = A(\omega,\theta)R_{ss}A^{H}(\omega,\theta)+\sigma^{2}I \qquad (4)$$

Where I is the M x M identity matrix. σ2 is the noise variance in each channel if the noise is white, $\sigma^{2}I$ is the MxM covariance matrix of the noises. R_{ss} is the d x d Auto-covariance matrix of input signal. For MUSIC to be applicable the input signals are assumed to be uncorrelated, so the covariance matrix R_{ss} will be a diagonal matrix having full rank d [4].

$$R_{ss} = diag\{P_{1},.....P_{d}\} \qquad (5)$$

Where $P_{d} = dE\left[\left|S_{d}(\omega)\right|^{2}\right]$ is the spectral power density of the d^{th} signal. R_{ss} will be positive-definite if and only if the d signal vectors are linearly independent. Under these assumptions, $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ is a positive semi definite M x M matrix of rank d with $\mathrm{rank}\ (\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}) = span[a(\theta_{1}),........a(\theta_{D})] < M$ Let $\lambda_{1} \geq \lambda_{2} \geq ...... \geq \lambda_{M}$ denote the Eigen-values of R_{xx} and $\eta_{1} \geq \eta_{2} \geq ...... \geq \eta_{M}$ denote the Eigen-values of matrix $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ respectively. From Eq. (5) we can easily see the following relation.

$$\lambda_{i} = \eta_{i} + \sigma^{2}I \qquad (6)$$

Since the rank of R_{SS} is d and the number of sources d is smaller than the number of arrays M, the matrix $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ is singular, i.e.

$$\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}} = 0 \qquad (6)$$

Eq. 3.15 implies that the d columns of $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ span a D-dimensional subspace of M-dimensional complex space. This subspace is referred as Signal Subspace. The smallest (M-d) Eigen-values of $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ are zero [4], i.e.

$$\eta_{D+1} = ...= \eta_{M} = 0 \qquad (7)$$

Finding the D distinct elements of A(ω ,θ) that cross the subspace will yield the DOA for the no-noise. But if we take into account the matrix's inclusion of a noise component $\mathrm{A}(\omega,\theta).\mathrm{R}_{ss}.\mathrm{A}(\omega,\theta)^{\mathrm{H}}$ Given that the eigenvectors of the MxM matrix R_{XXx} are all linearly independent, the Eigen-value decomposition (EVD) can be carried out as follows:

$$R_{xx} = \begin{bmatrix} E & Q & E^H \end{bmatrix} (E_S E_R) \begin{bmatrix} Q_S & 0 \\ 0 & Q_R \end{bmatrix} = \begin{bmatrix} E_S^H \\ E_R^H \end{bmatrix}$$

$$= E_S.Q_S.E_S^H + E_R.Q_R.E_R^H \qquad (8)$$

$$R_{xx} = \sum_{i=1}^{M} \lambda_i e_i e_i^H = \sum_{i=1}^{D} (\eta_i + \sigma^2)_i e_i e_i^H + \sum_{i=D+1}^{M} \sigma^2{}_i e_i e_i^H \qquad (9)$$

Because is a Hermitian matrix R_{XX} of full rank, each of the matrix's eigenvectors e_i is mutually orthogonal to one another, i.e.

$$e_i^H e_j = \delta_{ij} \quad where \quad \delta_{ij} = \begin{cases} 1 & i = j \\ 0 & i \neq j \end{cases}$$

Because λ_i denotes the Eigen-value and e_i denotes the eigenvector of matrix R_{xx}, the implies

$$R_{xx} e_i = \sigma^2 e_i \qquad i = D+1, \ldots\ldots M \qquad (10)$$

or

$$(R_{xx} - \sigma^2 I) e_i = 0 \qquad i = D+1, \ldots\ldots M \qquad (11)$$

Hence Eq (6) can be rewritten as.

$$\mathrm{A}(\omega, \theta).\mathrm{R}_{ss}.\mathrm{A}(\omega, \theta)^{\mathrm{H}} e_i = 0 \qquad i = D+1, \ldots\ldots M \qquad (12)$$

Since covariance matrix R_{XX} is real, positive, full rank diagonal matrix, it follows that

$$.\mathrm{A}(\omega, \theta)^{\mathrm{H}} e_i = 0 \qquad i = D+1, \ldots\ldots M \qquad (13)$$

According to the equation above, the steering vector's $[a(\omega,\theta_1), \ldots\ldots\ldots\ldots\ldots\ldots a(\omega,\theta_D)]$ subspace is the orthogonal complement of the subspace spanned by the eigenvectors. $\{e_{D+1}, e_{D+2}, \ldots\ldots, e_M\}$ This is shown as The eigenvectors of the covariance matrix, as was previously noted, are orthogonal to each other, so E_S and E_R are orthogonal complement. This can be expressed as $span[e_{D+1}, e_{D+2}, \ldots\ldots, e_M] \perp span[a(\omega,\theta_1), \ldots\ldots\ldots\ldots\ldots\ldots a(\omega,\theta_D)]$ As mentioned above that the eigenvectors of the covariance matrix R_{XX} are orthogonal to each other, so E_S and E_R are orthogonal complement. This can be expressed as

$$E_S = [e \ ,, \ldots\ldots, e_D]^T \perp E_n [e_{D+}, , \ldots\ldots, e_M] \qquad (14)$$

As a result, we can observe that the columns of E_S span the M-dimensional complex space's d-dimensional signal subspace similarly to the column vectors of matrix A(ω ,θ) ,that is. $span[e_1, e_2, \ldots\ldots, e_D]^T = span[a(\omega,\theta_1), \ldots\ldots\ldots\ldots\ldots\ldots a(\omega,\theta_D)]$ the subspace covered by the d greatest eigenvalues of R_{xx} the D eigenvectors, has the name "signal subspace." The subspace covered by the M-d eigenvectors of R_{xx} the M-D lowest Eigen-values is known as the noise space. The orthogonal complement of the signal subspace and the noise subspace is one another. Finding steering vectors on the array manifold that have zero paperion in the noise subspace allows one to identify the direction of arrival. These steering vectors must be orthogonal to the noise subspace. The real direction of arrival is the direction that the best steering vector steers in. The DOAs are calculated by the MUSIC algorithm as the peaks of the MUSIC spectrum as [4].

$$P_{music}(\omega,\theta)=\frac{1}{a^{H}(\omega,\theta)E_{n}E_{n}^{H}a(\omega,\theta)} \qquad (15)$$

The second stage is transforming a real Time domain signal to a complex Frequency domain since the approach acts in the Frequency domain. This is accomplished for each data block using FFT.

$$X(\omega_{k},\theta)=A(\omega_{k},\theta)S(\omega_{k})+N(\omega_{k}) \qquad (16)$$

$$P_{music}(\omega,\theta)=\frac{a^{H}(\omega,\theta)a(\omega,\theta}{a^{H}(\omega,\theta)E_{n}E_{n}^{H}a(\omega,\theta)} \qquad (17)$$

All beam-patterns or pseudo spectrum are incoherently averaged in this final stage, as illustrated in Fig. (3), to produce the resulting spectrum.

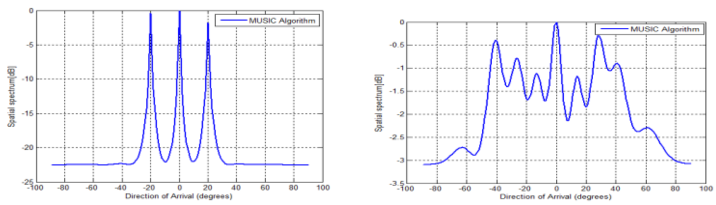
$$P_{Incoherent-music}(\omega_{k},\theta)=\sum_{k=1}^{M}\frac{1}{a^{H}(\omega_{k},\theta)E_{n}E_{n}^{H}a(\omega_{k},\theta)} \qquad (18)$$

The orthogonality of and Un will decrease it to the minimum, which will grow, as can be seen from the denominator.

$P_{Incoherent-music}(\omega_{k},\theta)$. Thus, the highest peaks of the MUSIC spectrum are corresponding to the DOAs of the d signals impinging on the array.

3.2 Simulation Results

To assess the performance of the MUSIC method, a computer program is created using MATLAB. With arrival angles of [-20° 0° 20°] and uncorrelated, respectively, the array is lighted. When compared to the white Gaussian noise in the background, the three sources are considered to have identical power, which is 0 dB. There are 128 pictures captured. then used to identify the DOA estimate on the MUSIC Beamformer. The outcomes shown in Figure (5) demonstrate that, when incident signals are expected to be uncorrelated, the MUSIC Beamformer can accurately determine the direction of arrival. Figure (5) displays the spatial spectral function produced by the multipath version of the signal at -20°, which is the situation in another scenario.



(a) (b)

Fig. (5). Spatial Spectrum of MUSIC Estimations for a- Un-Coherent Sources b- Coherent Sources.

As can be seen, the MUSIC algorithm is unable to determine the corresponding DOA of either of the linked signals at -20° and -20° since the direction of both signals could not be determined with sufficient accuracy. When it comes to identifying linked input signals, MUSIC fails. Figure (6b) shows that the response of the music is not sharp at the peaks as opposed to being sharp in the case of an input signal that is uncorrelated. The source covariance matrix does not meet the complete rank constraint required by the MUSIC for Eigen decomposition $E_{n}(\omega_{i})$, which leads to this issue. Any narrowband approach may be used to calculate the degrees of freedom (DOAs) at each frequency where is the noise subspace at frequency (ω_i) rather than MUSIC.

The incoherent method's conclusions and operation are straightforward, but it is susceptible to inaccurate wideband DOA estimations. Utilizing the narrowband into Line DOA computed using the Coherent Signal Subspace Method (CSSM) for each narrowband signal's Covariance matrix of the focus to the reference frequency. Wideband signal information may be used more effectively, and DOA estimation performance is superior than CSSM algorithm's low signal-to-noise ratio. The DOA estimated starting value [5], high signal to noise, and the ISSM method's performance below it make the CSSM algorithm susceptible.

3. Coherent Frequency Combining

Wideband non-adaptive DOAs are the coherent technique. introduced the CSSM in this section. The first approach for estimating wideband DOA is based on splitting the frequency band into non-overlapping narrowbands and estimating narrowband DOA in each band. To get the final DOA estimate, the DOA estimates for the various bands are then added together. However, this method cannot handle coherent signal sources. The CSSM [20] is a different approach for handling coherent signal sources. This approach begins by using FFT to break down the wideband array data into a number of narrowband components. The narrowband array manifold matrices are then converted into focusing matrices in order to create matrices matching to a chosen reference frequency. The use of narrowband DOA estimation techniques, like the spectral estimation approach [5], is then applied to determine the directions of arrival. The spectrum estimating techniques are favourable since they need less computing effort than other techniques (such those based on least squares, maximum likelihood, etc.). Preliminary DOA estimations close to the real directions of arrival are needed for the CSSM method's focusing matrix design.

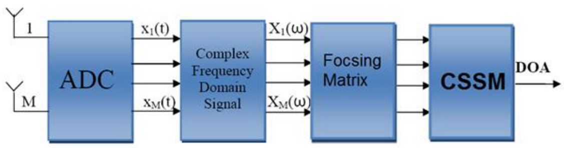


Fig. (6). Basic Principle of Coherent Signal Subspace Method.

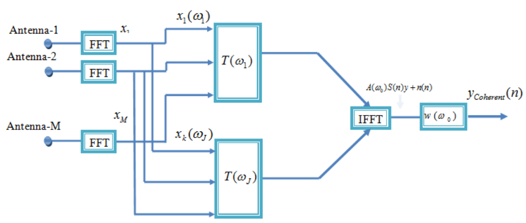


Fig. (7). Block Diagram of the CSSM Wideband Adaptive Beamformer.

The first method to coherently sum the correlation matrices of different frequency bins was CSSM, which Wang and Kaveh developed [5]. Using a transformation matrix (focusing matrix) that is dependent on the frequency bin, it merges the correlation matrices at various frequency bins into a single general correlation matrix at a single focusing frequency. This process is known as concentrating. Coherent techniques' general process is depicted in Figure (6-7).

A generic block architecture for the focused wideband adaptive beamformer is shown in Figure (7). In this method, a single time domain narrowband beamformer is used to concentrate the signal subspaces at various frequencies to a single frequency using a pre-processor that is implemented as a frequency dependent linear transformation matrix $T(\omega_j)$. Low computing cost, the ability to address the signal cancellation issue, and enhanced convergence qualities are the key advantages of the coherent technique. Theoretically, the CSSM should utilize a focusing matrix $T(\omega_j)$ fulfilling [5].

$$T(\omega_j)A(\omega_j,\theta) \cong A(\omega_0,\theta) \qquad (19)$$

Where (ω_j) are the frequencies within the bandwidth of the signals and w0 is the focused frequency, i.e. $T(\omega_j)$ focuses the signal subspaces $T(\omega_j)A(\omega_j,\theta)$ at frequencies (ω_j) onto the signal subspace $T(\omega_j)A(\omega_0,\theta)$. The matrices $T(\omega_j)A(\omega_j,\theta)$ and $T(\omega_j)A(\omega_0,\theta)$ contain the steering vectors of the sources in their columns at frequencies (ω_j) and (ω_0), respectively, where (θ) is the DOAs vector.

There are several approaches to design the focusing matrix $T(\omega_j)$ in the following sections, we describe several representative focusing methods. They propose Rotational signal subspace focusing transformation (RSS).In [6] a quantities measure for The ratio of the array's SNR following and before the focusing operation is used to determine the focusing loss. The benefit of employing unitary focusing matrices is emphasized by the authors since they have no focusing loss. In order to solve the following constraint minimization issue (Hung and Kaveh [6] devised a unitary focusing matrix), they suggest unitary transformations as follows:

$$\min_{T(\omega_j)} \left\| A(\omega_0,\theta) - T(\omega_j)A(\omega_i,\theta) \right\|_F \quad j=1,2,........J \qquad (20)$$

Subject to $T^H(\omega_j)T(\omega_j) = I$

Where $\|.\|_F$ is the Frobenious matrix norm [20]. The solution to (22) is given by.

$$T_{RSS}(\omega_j) = V(\omega_j)U(\omega_j)^H \qquad (21)$$

Where the columns of $U(\omega_j)^H$ and $V(\omega_j)$ are the left and right singular vectors of $A(\omega_j,\theta)A(\omega_0,\theta)^H$ They named this focusing matrix the Rotational Signal Sub-space (RSS) focusing matrix. then the general focusing correlation matrix as.

$$R_{focuing} = \sum_{j=1}^{J} T^H(\omega_j)R_{XX}T(\omega_j) \qquad (22)$$

Therefore, the issue is how to identify focusing angles that should match the signal's DOA but are not accessible. Finding focusing angles $A(\omega_0,\theta)$ that ought to be sufficiently near to the genuine DOAs typically involves employing low-resolution DOA estimating techniques. Some closely spaced DOAs, however, cannot be addressed using this method. Furthermore, focusing angles may not always result in a solution with the right value, unlike beginning values in other iterative approaches. For accurate estimates, focusing angles should be near to the genuine DOA; however, if they are not the same as the true DOA and the focusing frequency is not at the center of the signal's band, the bias will never reach zero. For high resolution techniques, the bias could be absolutely crucial.

5- Eigen-value based Diagonal Loading

The EDL technique attempts to overcome the adaptive approach of the FDL technique by using the following equations to estimate [128]

$$R_{dl-eig}(k) = \mathbf{R}_{XX} + \alpha(k)\mathbf{I}_M \tag{23}$$

Where α is a positive diagonal loading factor. I represents a unit matrix. with RXX replaced by the signal free correlation matrix and the loading factor (7).

$$\alpha(k) = -\lambda_{\min} + \sqrt{(\lambda_{\min} - \lambda_M) - (\lambda_{M-1} - \lambda_{\min})} \tag{24}$$

with $\lambda_1 > \lambda_2 >$........ $>\lambda_M$ denoting the ordered eigenvalues of signal-free R_{XX} with M the dimension of interference subspace [7].

The algorithm can improve the presence of random array manifold vector error Poor adaptive beamforming algorithm. However, this method the primary The question is how to choose the amount of discretion to load. Usually selected based on experience special Given value, the most typical is 10σ2, where said single sensor σ^2 The noise power, the paper loaded in the selected amount of sediment it was $10\sigma^2$.The concept of diagonal loading is straightforward. We use,

$$R_{xx}(\omega,\theta)_L = \frac{1}{M}\sum_{n=1}^{M} X(\omega,\theta)X(\omega,\theta)^H + \sigma_L^2 I \tag{25}$$

In place of the estimated spectral matrix in order to design w. We use $R_{xx}(\omega,\theta)_L$ instead of $R_{xx}(\omega,\theta)$ because it is not used as an estimate of $R_{xx}(\omega,\theta)$.

based on eigen values Using diagonal loading factors, which are independent of direction inaccuracy, is the preferred method. The ways to estimate the loading factor using EDL need precise knowledge of the steering vector error, just as narrowband techniques, and adapt current techniques for wideband beamforming to appropriately estimate the loading factor. then using Eq. (18) to represent the new correlation matrix in beam-patterns or a pseudo spectrum.

6. Simulation Results

Under these circumstances, the result of the CSSM method is shown in the last assumption, where it is expected that the signal at -20° is a multipath variant of the signal at 20°. For each signal, the SNR is adjusted to -10 and 10 dB.

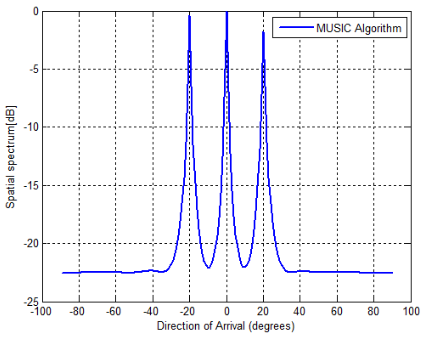


Fig. (8). Spatial Spectrum of the CSSM Algorithm in the Coherent Signals.

Figure (8) makes it evident that the spatial spectral function for the case of correlated signals was produced using the CSSM algorithm.

7. Behaviour Analysis for Different Algorithms

Methods for estimating DOA rely on either the array manifold's parametric structure or the signals' non-Gaussianity or cyclo-stationarity. These approaches include multiplying a weight matrix by the incoming data matrix in order to estimate the waveform of the signals. variables that have an impact on the DOA.

1. The array's element count.
2. Signal-to-Noise Ratio, second.

3. Quantity and the distance between array items.

4. The number of signals event sampled.

1- Effect number of elements of the antenna array on DOA's estimation

The number of array elements is assumed to be a varying variable during the simulation where it varies from 5 to 11 elements.

The other settings are SNR = 0dB and 128 snapshots. Figure (9) illustrates the relationship between the number of array elements and the DOA estimation. It is obvious that as the number of array elements rises, the MUSIC Method becomes more accurate in estimating the DOA of incident signals. This is due to the fact that adding more array components will result in a narrower beam around the directions of incident signals.

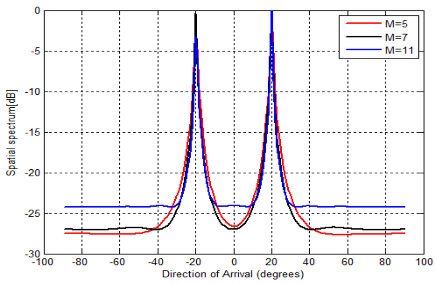


Fig. (9). The effect of the number of elements on DOA's on MUSIC algorithm.

2- Effect of The Signal to Noise Ratio (SNR)on DOA's Estimation

During the simulation, it is assumed that the signal to noise ratio is a variable that changes from -10 dB to 10 dB for each signal. The 11 elements in the array, which are spaced apart by half a wavelength, are regarded as making up a linear array. 128 snapshots are used to evaluate the array covariance matrix and the spatial spectrum of the MUSIC technique. In the figures below, the SNR's impact on DOA estimate is depicted. Figure (10), which shows that MUSIC Methods can precisely estimate the DOA of incoming signals as SNR increases, makes this point quite evident.

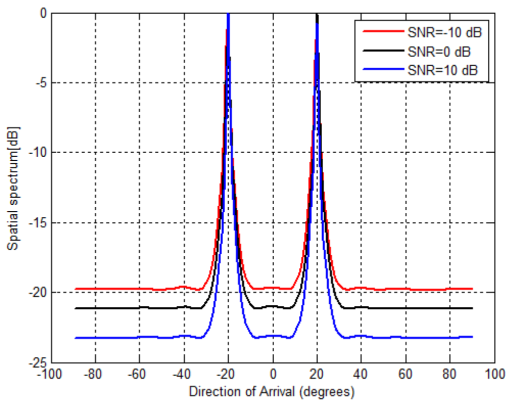


Fig. (10). The effect of the signal to noise ratio on DOA's MUSIC algorithm.

This is due to the fact that higher SNR will result in a narrower beam around incident signal directions. The peaks are sharper when the noise is lower. The peaks spread out as a result of the additional noise. As the SNR increased, the performance improved, indicating that the SNR had an impact on the resolution of the MUSIC technique. It is evident that the ratio of the output peaks to the noise level at the array's output grows correspondingly as SNR rises.

3- Effect the Number of Snapshots of The Signal on DOA's Estimation

To test the method algorithm estimator's capabilities with various signal snapshots, a simulation program was run. The number of snapshots is assumed to be a variable during the simulation with a range of 50 to 1000 snapshots for each signal. This differs from the assumptions we made in the earlier analysis of the influence SNR. Figure (11), which shows how the MUSIC algorithm can see that as the number of pictures increases, the peaks in the MUSIC spectrum get sharper, is abundantly

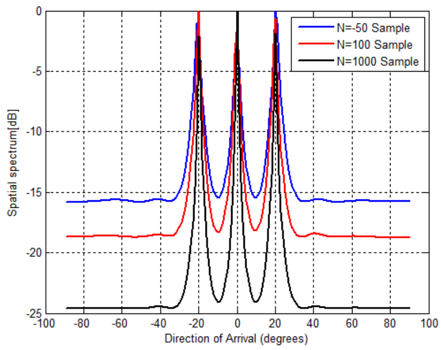


Fig. (11). The effect of number of snapshots signal on DOA's MUSIC algorithm.

4- Array antenna Spacing & resolution

The performance of the beamformer depends on the array's element spacing and number, as shown by an examination of the MUSIC algorithm equation from the Beam Pattern section.

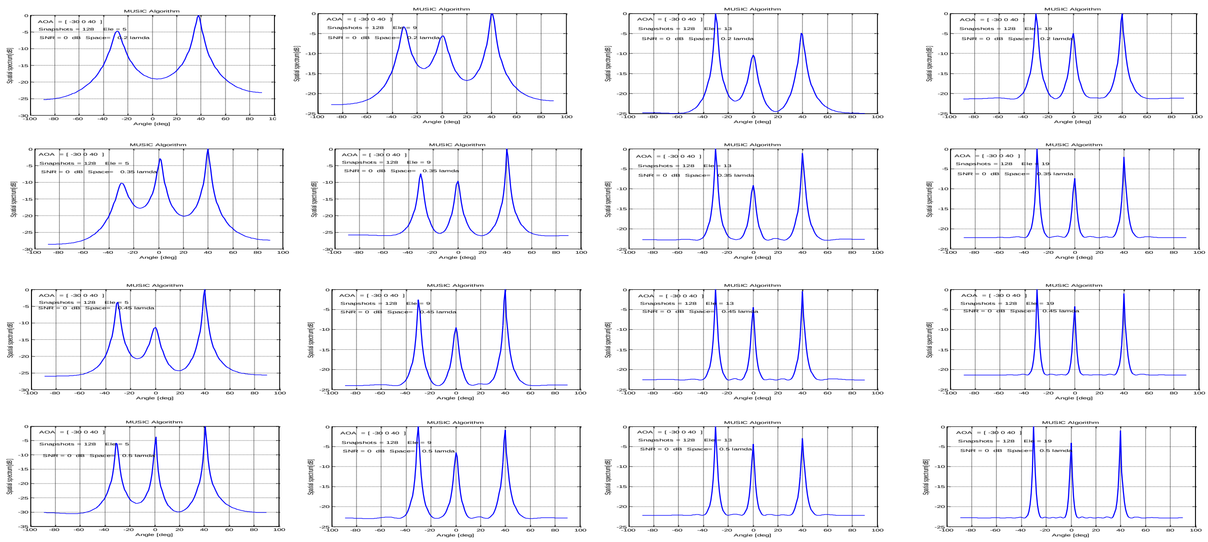


Fig. (12). Performance quantity music dependent on the spacing and array's elements.

Figure (12) above shows how altering these settings impacts the spatial filtering capability of the beamformer. Figures are influence the SNR and element count for DOA separation. The development of an adaptive antenna system based on direction of arrival benefits from this approach.

4. Conclusion

This article discusses the configurations of smart antenna systems as well as the reasons to use adaptive antenna array for DOA. It also describes the fundamental functions and parts of adaptive antenna array for DOA and interference cancellation for wideband signal. Wideband DOA performance was examined utilizing theoretical framework and simulation with MATLAB software.

The simulation results displayed in this study indicate that the uniform linear array-based DOA estimate approaches are effective. Computer simulation is used to study and examine two DOA estimation techniques for wideband signals. These techniques are incoherent signal subspace methods (ISSM), which are wideband DOA estimation techniques for high resolution estimate of angles of arrival of several wideband plane waves as part of the MUSIC algorithm. The CSSM, or second method, combines a solid, almost ideal data-adaptive statistic. To assure a statistically sound pre-processing of wideband data, this approach is employed in conjunction with improved focusing matrix design. The response of the MUSIC is not sharp at the peaks whereas it was sharp in the case of uncorrelated input signal condition, which is why incoherent approaches like IMUSIC fail when it comes to identifying correlated input signals. The source covariance matrix does not meet the complete rank constraint required by the MUSIC for Eigen decomposition, which leads to this issue. CSSM with TCT performs better, however it is not a fair estimate method. to lessen or eliminate bias and increase resolution. When array defects of any type are taken into consideration, Eigenvalue EDL is resilient. The higher performance of the suggested approaches on beam-pattern control, output SINR augmentation, and robustness against have been demonstrated by extensive numerical testing. When diagonal loading is used Wideband signals were used to study the situation of DOA estimating, and the impact of bandwidth on estimator accuracy was examined using computer simulations, the Incoherent method, and coherent subspace method-based techniques like MUSIC. It is obvious that the beamformer may be used to load correlated signals using the spatial spectral function produced by the CSSM method.

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