Performance comparison of wavelet denoising based fast DOA estimation of MIMO OFDM system over Rayleigh Fading Channel

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Abstract

This paper presents a tool for the analysis, and simulation of direction-of-arrival estimation for MIMO OFDM signal over the Rayleigh fading channel. The performance of the proposed technique is tested for wavelet denoising based CYCLIC MUSIC algorithm. Simulation results demonstrate that the proposed system not only has good ability of suppressing interference, but also significantly improves the DOA estimation of the system. In this paper, it is proposed to find DOA of the received MIMO OFDM signal, and the performances are analyzed using matlab simulation by the Monte Carlo computer iteration. This paper provides a fairly complete image of the performance and statistical efficiency with QPSK signal model for coherent system at a lower SNR(18dB) and interference environment.

Keywords

MUSIC; QPSK; DOA; MIMO, OFDM, SNR, MLM..

INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) has been adopted as a standard for various applications such as digital audio/video broadcasting (DAB/DVB) and wireless LANs etc. Conventional OFDM systems transform information symbol blocks into N sub blocks and then insert redundant bits in the form of either cyclic prefix (CP) or zero padding (ZP). Inter block interference (IBI) is avoided only if the length of the cyclic prefix or zero padding should be longer than the channel delay spread in the frequency-selective fading nature of the channel [16-18]. Symbol recovery is assured in zero padding even when channel nulls occur on some subcarriers, which is not possible with the use of Cyclic Prefix. Hence the code rate of the OFDM system is reduced due to the CP/ZP redundant bits. OFDM enhances the code rate without bandwidth expansion and without increasing the number of subcarriers but with a moderate increase in computational complexity and delay.

The goal of direction-of-arrival (DOA) estimation is to use the OFDM received signal using QPSK on the downlink at the base-station sensor array to estimate the directions of the signals from the desired mobile users as well as the directions of interference signals. The results of DOA estimation are then used by to adjust the steering weights of the adaptive beam former. So that

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the radiated received power is maximized towards the desired users, and radiation nulls are placed in the directions of the interference signals. Array signal processing has found important applications in diverse fields such as Radar, Sonar, Mobile Communications and Seismic explorations. The problem of estimating the DOA of narrow band signals with noisy environment using antenna arrays has been analyzed intensively over fast few years.[1]-[9]. The wavelet denoising is a useful tool for various applications of image processing, radar signal processing, speech signal processing and acoustic signal processing for noise reduction. There are some trials for DOA estimation by applying the wavelet transform method to CYCLIC MUSIC scenarios [6-8]. In this paper, the DOA estimation algorithm using a time-frequency conversion pre-processing method was proposed for CYCLIC MUSIC and the effectiveness was verified through the simulation results using matlab. This is focused on the improvement of DOA estimation performance at lower SNR and interference environment than MUSIC and MLM.

This paper is organized as follows. Section 1 presents the MIMO OFDM signal model over fading channel environment for the coherent system. Section 2. 2.1, 2.2 and 2.3 briefly describes the algorithms what we have described. Section [a],[b] deal with MUSIC and CYCLIC MUSIC and [c] proposed CYCLIC MUSIC with wavelet denoising method for MIMO OFDM system. MUSIC procedures are computationally much simpler than the MLM but they provide less accurate estimate [2]. The popular methods of Direction finding algorithm such as MUSIC suffer from various drawbacks such as 1.The total number of signals impinges on the antenna array is less than the total number of receiving antenna array. 2. Inability to resolve the closely spaced signals 3. Need for the knowledge of the existing characteristics such as noise characteristics. Section 3 describes the simulation results and performance comparison of exponential pulse with QPSK signal for the OFDM received signal. Finally Section 4 concludes the paper.

1. MIMO OFDM SIGNAL MODEL

The algorithm starts by constructing a real-life fading signal model for the noisy environment. Consider a number of plane waves from M narrow-band sources impinging from different angles θi, $i = 1, 2, \ldots, M$, impinging into a uniform linear array (ULA) of N equi-spaced sensors, as shown in Figure 1.

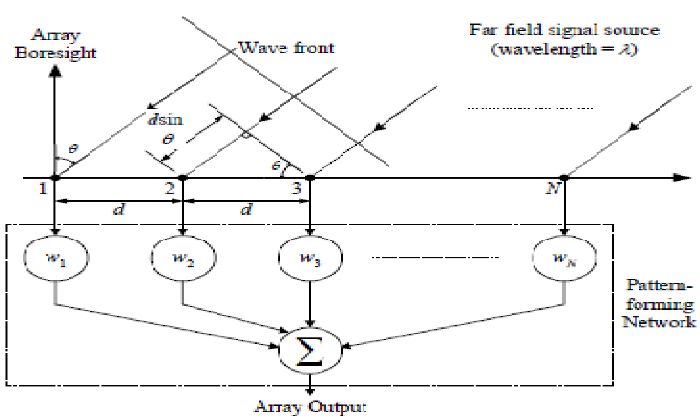

Figure1.Uniform linear array antenna

The MIMO OFDM received signal data model is given by

$$y_1(t) = \sum_{k=1}^{K} \alpha_1(k)\ x_{mk}(t) + n_1(t) \qquad (1)$$

Where $\alpha_1(k) = \alpha(k) a_k(\Phi)$; $a_k(\Phi)$ is the antenna response vector. Where $x_{mk}(t)$ is the signal transmitted by k^{th} user of m^{th} signal for an OFDM system. Figure 2 shows the general OFDM system and the OFDM signal is given by

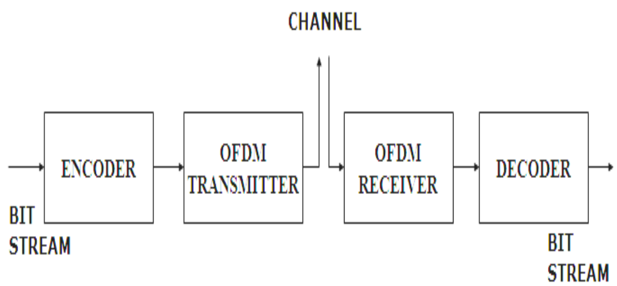

Figure2.OFDM system

$x_{mk}(t)=1/N \sum S_{mk}(t)e^{-2\pi K/N}$

$\alpha_1(k)$ is the fading coefficient for the path connecting user k to the l^{th} antenna, $n_l(t)$ is circularly symmetric complex Gaussian noise. In coherent wave front fading channel the fading parameters for each user is modeled as $\alpha_1(k) = \alpha(k) a_k(\Phi)$, where $\alpha(k)$ is a constant complex fading parameter across the array , Φ_k is the DOA of the k^{th} user's signal relative to the array geometry, and $a_k(\Phi)$ is the response of the l^{th} antenna element to a narrow band signal arriving from Φ_k. $a(\Phi)$ is an $N\times1$ vector referred to as the array response to that source or array steering vector for that direction. It is given by:

$$\mathbf{a}(\Phi) = [1\ e-j\varphi \ldots\ldots e-j(N-1)\varphi\ ]\ T \qquad (2)$$

where T is the transposition operator, and φ represents the electrical phase shift from element to element along the array. This can be defined by:

$$\varphi = (2\pi/\lambda)d\cos\theta \qquad (3)$$

where d is the elemental spacing between the two element and λ is the wavelength of the received signal.

Due to the mixture of the signals at each antenna, the elements of the $m\times1$ data vector y(t) are multicomponent signals. Whereas each source signal x(t) of the $n\times1$ signal vector, x(t) is often a monocomponent signal. N(t) is an additive noise vector whose elements are modeled as stationary, spatially and temporally white Gaussian, zero mean complex random processes that are independent of the source signals. That is

$$E[n(t+\Gamma)\,n^H(t)]=\sigma\delta(\tau)I$$
$$E[n(t+\Gamma)\,n^T(t)]=0, \quad \text{for any } \tau \qquad (4)$$

Where $\delta(\tau)$ is the delta function, I denotes the identity matrix, σ is the noise power at each antenna element, superscripts H and T, respectively, denote conjugate transpose and transpose and E(.) is the statistical expectation operator.

In (1), it is assumed that the number of receiving antenna element is larger than the number of sources, i.e., m>n. Further, matrix A is full column rank, which implies that the steering vectors corresponding to n different angles of arrival are linearly independent. We further assume that the correlation matrix

$$R_{yy} = E\left[y(t)y^H(t)\right] \qquad (5)$$

is nonsingular and that the observation period consists of N snapshots with N>m.

Under the above assumptions, the correlation matrix is given by

$$R_{yy} = E\left[\left(y(t)y^H(t)\right)\right] = AR_{xx}A^H + \sigma I \qquad (6)$$

Where $R_{xx} = E[(x(t)x^H(t))]$ is the source correlation matrix.

Let $\lambda_1 > \lambda_2 > \lambda_3 .... \lambda_n = \lambda_{n+1} = ... \lambda_m = \sigma$ denote the eigen values of R_{yy}. It is assumed that Λ_i, i=1, 2, 3…….n are distinct. The unit norm Eigen vectors associated with the columns of matrix S= [s_1 s_2 …..s_n], and those corresponding to $\lambda_{n+1} \ldots\lambda_m$ make up matrix G=[g_1 …….g_{m-n}] . Since the columns of matrix **A** and **S** span the same subspace, then $A^HG=0$;

In practice R_{yy} is unknown and, therefore, should be estimated from the available data samples y(i), i= 1 2 3………N. The estimated correlation matrix is given by

$$R_{yy} = 1/N \sum_{n=1}^{N} (y(t)y^H(t)) \qquad (7)$$

Let { $s_1, s_2,$ .... $s_n,$ ....g_{m-n} } denote the unit norm eigen vectors of R_{yy} that are arranged in descending order of the associated eigen values respectively. The statistical properties of the eigen vectors of the sample covariance matrix R_{yy} for the signals modeled as independent processes with additive white Gaussian noise are given in [9].

2. ALGORITHMS USED

2.1 MUSIC

MUSIC is a method for estimating the individual frequencies of multiple times – harmonic signals. MUSIC is now applied to estimate the arrival angle of the particular user from the noisy environment [1],[2].The structure of the covariance matrix with the spatial white gaussion noise assumption implies that its spectral decomposition matrix is expressed as

$$R = APA^H = U_S A U^H{}_S + \sigma^2 U_n U^H{}_n \qquad (8)$$

Where assuming APA^H to be the full rank matrix, the diagonal matrix Us contains the M largest required signal Eigen values. Since the Eigen vectors in Un (the noise Eigen vectors) are orthogonal to A.

$$U_n a(\phi) = 0, \text{ where } \phi \in \{\varphi_1, \phi_2, ...\phi_m\} \quad (9)$$

To allow for unique DOA estimates, the array is usually assumed to be unambiguous; that is, any collection of N steering vectors corresponding to distinct DOAs Φm forms a linearly independent set { $a_{\Phi 1}, \ldots, a_{\Phi m}$ } . If a(.) satisfies these conditions and P has full rank, then APA^H is also full rank. The above equations (8) and (9) are very helpful to locate the DOAs in accurate manner. Let $\{s_1 \ldots s_n, g_1 \ldots g_{m-n}\}$ denote a unit norm eigenvectors of R, arranged in the descending order of the associated Eigen values, and let Š and Ĝ denote the matrices S and G made of $\{s_I\}$ and $\{g_I\}$ respectively. The Eigen vectors are separated in to the signal and noise Eigen vectors. The orthogonal projector onto the noise subspace is estimated. And the MUSIC 'spatial spectrum' is then defined as

$$f(\phi) = \left[a^*(\phi)\widehat{G}\widehat{G}^* a(\phi)\right] \quad (10)$$

$$f(\phi) = \left[a^*(\phi)\left[I - SS^*\right]a(\phi)\right] \quad (11)$$

The MUSIC estimates of $\{\Phi_i\}$ are obtained by picking the n values of Φ for which f(Φ) is minimized. To conclude, for uncorrelated signals, MUSIC estimator has an excellent performance for reasonably large values of N, m and SNR. If the signals are highly correlated, then the MUSIC estimator may be very inefficient even for large values of N, m, and SNR.

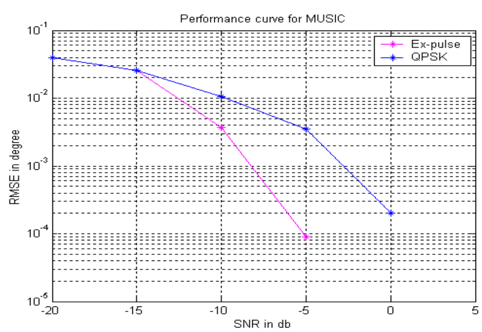

Figure 3. Performance comparison of MUSIC

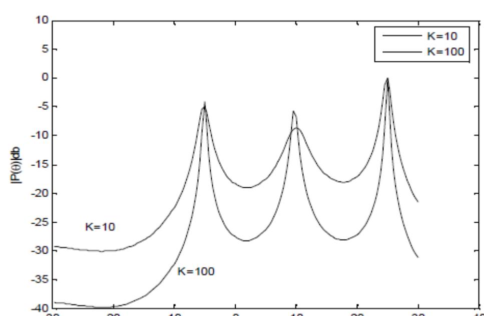

Figure4. Spectrum of MUSIC for two snapshots

2.2 Cyclic MUSIC

We assume that m_α sources emit cyclostationary signals with cycle frequency α ($m_\alpha \leq m$). In the following , we consider that x(t) contains only the m_α signals that exhibit cycle frequency α, and all of the remaining $m-m_\alpha$ signals that have not the cycle frequency α.

Cyclic autocorrelation matrix and cyclic conjugate autocorrelation matrix at cycle frequency α for some lag parameter τ are then nonzero and can be estimated by

$$R_{yy}^{\alpha}(\tau) = \sum_{n=1}^{N} y(t_n+\tau/2)\, y^H(t_n-\tau/2)\, e^{-j2\pi\alpha t_n} \qquad (12)$$

$$R_{yy^*}^{\alpha}(\tau) = \sum_{n=1}^{N} y(t_n+\tau/2)\, y^T(t_n-\tau/2)\, e^{-j2\pi\alpha t_n} \qquad (13)$$

where N is the number of samples.

Contrary to the covariance matrix exploited by the MUSIC algorithm [1], the Cyclic MUSIC method [8] is generally not hermitian. Then, instead of using the Eigen Value decomposition (EVD), Cyclic MUSIC uses the Singular value decomposition (SVD) of the cyclic correlation matrix. For finite number of time samples, the algorithm can be implemented as follows:

- Estimate the matrix $R_{yy}^{\alpha}(\tau)$ by using (12) or $R_{yy^*}^{\alpha}(\tau)$ by using (13).

- Compute SVD.

$$\begin{bmatrix} U_s & U_n \end{bmatrix} \begin{bmatrix} \Sigma_s & 0 \\ 0 & \Sigma_n \end{bmatrix} \begin{bmatrix} V_s & V_n \end{bmatrix}^H \qquad (14)$$

Where [Us Un] and [Vs Vn] are unitary, and the diagonal elements of the diagonal matrices ∑s and ∑n are arranged in the decreasing order. ∑n tends to zero as the number of samples becomes large.

- Find the minima of $\| Un^H a(\Phi) \|^2$ or the max of $\| Us^H a(\Phi) \|^2$

2.3 NEW APPROACH OF DOA ESTIMATION

A signal subspace based DOA estimation performance is affected by the two factors of an accurate array manifold modeling and a spatial covariance matrix of a received array signal. A higher SNR signal for a target source is required for an accurate estimation from finite received signal samples. But the DOA estimation performance is limited by the lower SNR from interference signals such as fading and environmental noise. For the performance improvement of DOA estimation, this paper proposed a pre-processing technique of time-frequency conversion methodology for signal filtering. This method includes a time-frequency conversion technique with a signal OBW (Occupied Bandwidth) measurement based on wavelet de-noising method as shown in Fig. 5 and 6. This is a DOA method for SNR improvement based on time-frequency conversion approach. The improvement of a DOA estimation performance is verified by using the matlab simulation.

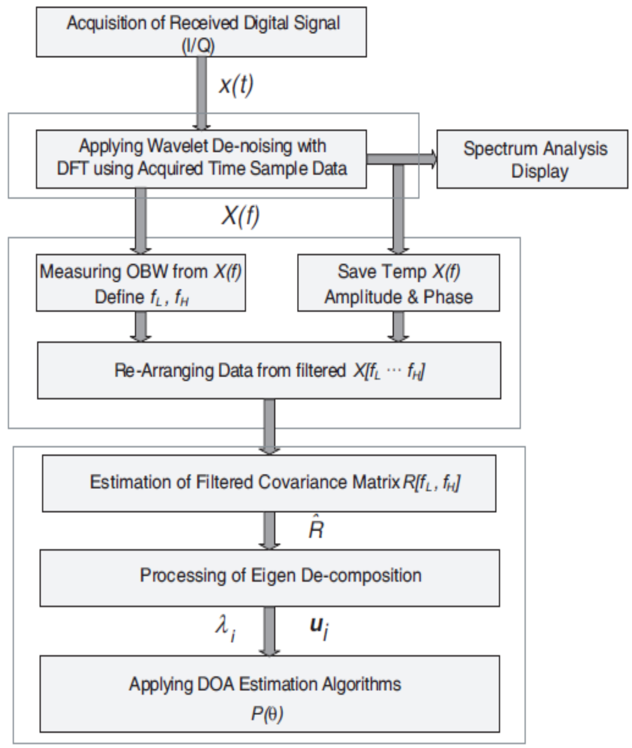

Figure 5. MUSIC Method

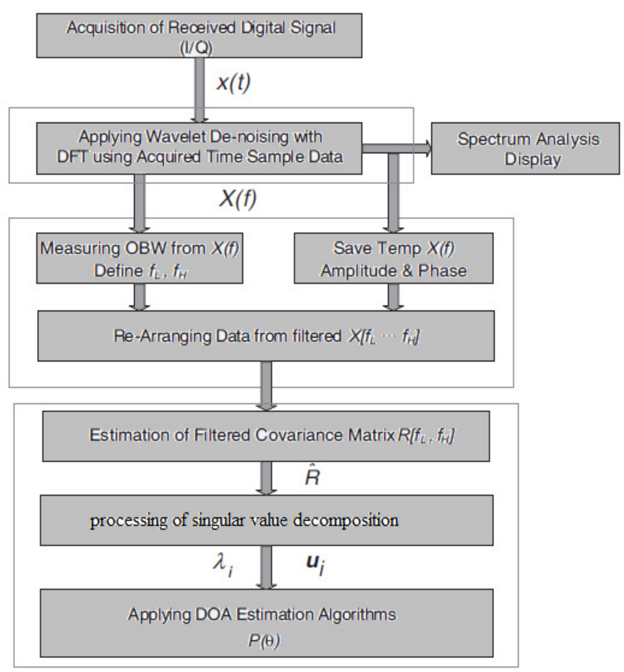

Figure6. Proposed Model for CYCLIC MUSIC

This is proposed to overcome the limitation of existing DOA estimation techniques based on only time domain analysis such as MUSIC. The more effective estimation is expected by the improvement of SNR from the proposed pre-processing techniques of frequency domain analysis. The proposed method collects a time sampled signal y(*t*) from an array antenna as shown in Figure 1. The upper and lower limits *fL* and *fH* of a signal are determined from y(*f*), which is the DFT or FFT result of a received signal y(*t*). The filtered covariance matrix *R*[*fL; fH*] can be obtained from the estimated from the received OFDM signal energy, y[*fL; fL*+1*; : : : ; fH*] with an improved SNR. This process can effectively eliminate the interference noises from the target received signal streams by the frequency domain analysis. Where *P* is a power of each spectrum frequency elements { *f*1,….., *fN*}. The 99% OBW is calculated from the *fH* and fL each spectrum boundary.

$$p_{rel} = \sum_{i=f_1}^{f_N} Py(i) \qquad (15)$$

$$\Delta P_{\beta/2} = P_{rel} \times \beta / 2[\%] \qquad (16)$$

(β = 1 for 99% OBW analysis)

$$f_L = \arg\min_{f_L} \left\| \sum_{f_L=f_1}^{f_N} Py(f_L) - \Delta P_{\beta/2} \right\| \qquad (17)$$

$$f_H = \arg\min_{f_H} \left\| \sum_{f_H=f_1}^{f_N} Py(f_H) - \Delta P_{\beta/2} \right\| \qquad (18)$$

An improved DOA estimation is expected from the filtered covariance matrix and Eigen-decomposition processing and singular value decomposition for cyclic music at particularly low SNR signal conditions. By the proposed pre-processing, it can effectively reject adjacent interferences at low SNR conditions. Moreover, it can acquire the signal spectrum with an improved DOA estimation spectrum simultaneously without additional computation.

3. SIMULATION AND PERFORMANCE COMPARISON

Data Specification

OFDM specification:		
Bandwidth	100MHz	
Number of subcarriers	1024	
Packet length	0.6 ms (48 × 12.5 µs):	48 OFDM symbols
Symbol duration	2.5 µs (10.24 + 2.26: effective symbol + guard interval)	
Modulation	QPSK and FM signal	
Channel model	AWGN and Rayleigh fading channel	

Antenna Array Model:

Type: Uniform Linear array antenna

No. of array Elements	N	8
Free space velocity	c	$3*10^8$
Centre frequency	fc	2.4GHz
Wavelength	λ	c / fc
Inter element Spacing	d	$\lambda/2$
Angle of arrival in degrees θ		-5 to 100

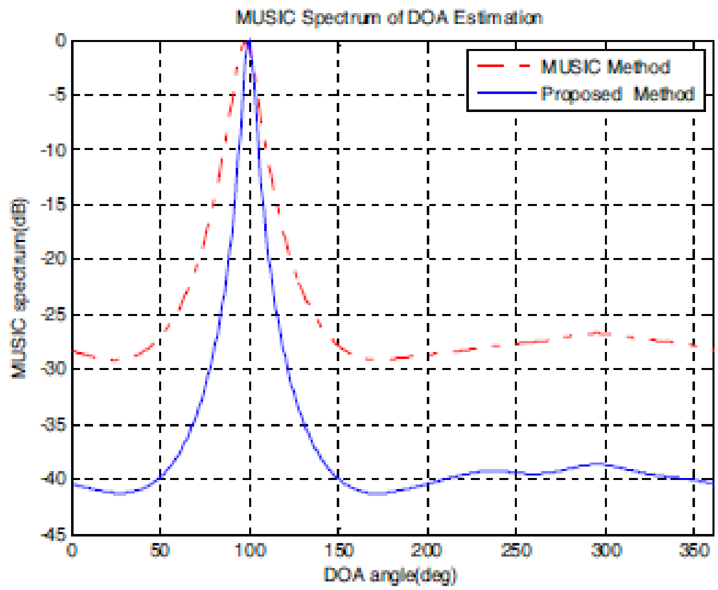

Figure 6. DOA estimation spectrum

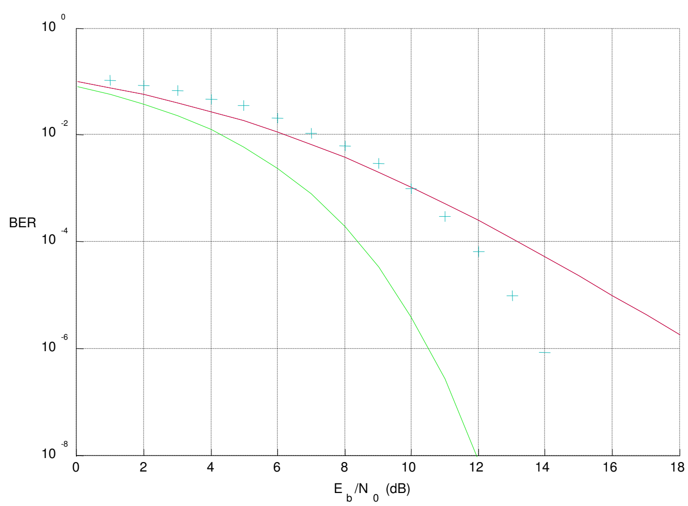

Figure 7. Performance comparison

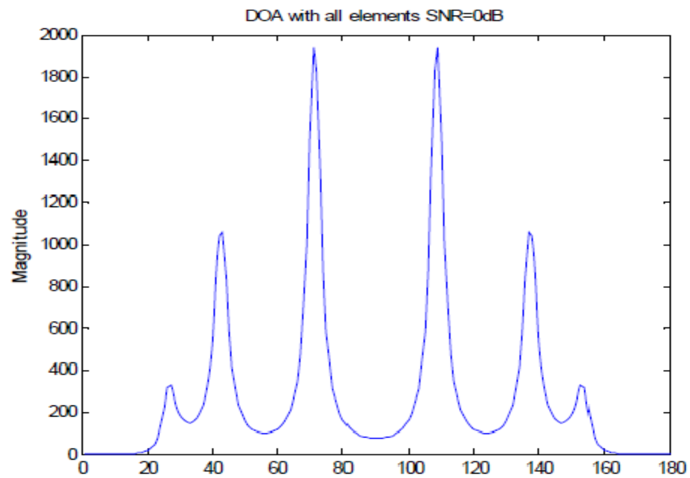

Figure8. Direction of Arrival of the proposed cyclic music

In this section, we present some simulation results, to show the behavior of the two methods and to compare the performance with the analytically obtained Mean Squared Error (MSE) Vs SNR. We consider here a linear uniformly spaced array with 16-antenna elements spaced λ/2 apart. Incoming QPSK signals are generated with additive white Gaussian noise in fading environment with signal to noise ratio 10dB, the bit rate of the QPSK signal of interest is 2Mb/s and other QPSK modulated signals with data rate 1Mb/s are considered as interference. MUSIC is simulated using the specified parameters. One QPSK signals arrived at 20 degree and an interferer at 5 degree DOA. BER Vs SNR plots and there corresponding spectrum for the two methods as shown in figure 3,4, 5 ,6,7and 8. This section presents Monte Carlo computer simulation results to illustrate the performance of the proposed algorithms for synchronous system. Each Monte Carlo experiment involved a total of 1000 runs, and each estimation algorithm is presented with exactly the same data intern. It is interesting to note that QPSK signal performs better than FM signal. So that bandwidth requirement is as low as possible for QPSK signals as compared to FM signals. From the above simulation results, the proposed method improves the DOA estimation performance of accuracy and spatial spectrum especially for lower SNR signals. Figure 7 shows the comparison results of DOA estimation performance for low SNR (18dB) signals for AWGN and Rayleigh fading environment. The proposed method improves the peak characteristic more than 10 dB at less than -10 dB SNR signal condition by applying CYCLIC MUSIC algorithm for DOA estimation.

4. CONCLUSION

Good signal selective capability and high resolution is achieved in wavelet denoising based CYCLIC MUSIC algorithm. It can be concluded that the proposed method not only has the good ability of suppressing interference, but significantly improves the DOA estimation and Bit error rate (BER) performance for MIMO-OFDM system. Therefore the proposed method shows an improved ability of DOA resolution and estimation error at the noise and interference conditions. These are the measurement limits at on-air environment. The good thing of MUSIC and CYCLIC MUSIC is that it works well in MIMO OFDM transmission.

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