

Evaluation of MUSIC algorithm for DOA estimation in Smart antenna

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Abstract: This paper presents a simulation study of a smart antenna system on basis of direction-of-arrival estimation for identifying the direction of desired signal using Multiple Signal Classification (MUSIC) algorithm. In recent times smart antennas are gaining more popularity by increasing user capacity, achieved by effectively reducing interference. In this paper, performance and resolution analysis of the algorithm is made. Its sensitivity to various perturbations and the effect of parameters such as the number of array elements and the spacing between them which are related to the design of the array sensor are investigated. The analysis is based on uniform linear array antenna and the calculation of the pseudo spectrum function of the algorithm. MATLAB is used for simulating the algorithm.

Keywords: Smart antenna, DOA, MUSIC, Pseudo spectrum, MATLAB.

I. INTRODUCTION

The requirement of mobile communication resources has been increased rapidly over few years. The smart techniques or adaptive beamforming techniques have been emerged as a main key way to achieve the enthusiastic requirements introduced for present and various types of beamforming schemes, Smart antenna is a array of antenna elements combined with digital signal processing either to transmit or receive in an adaptive manner. The word adaptive means, automatically adjusts the directionality of its radiation pattern in response to the signal, these can also be called as adaptive array antennas, hence smart antennas can be enhancing the capacity of the channel, broadens range coverage, steer multiple beams to track users, compensates aperture distortion or reduce multipath fading and co-channel interference. The smart antenna system can be divided mainly into three sections.

- The first section performs the estimation of DOA and finds the number of entering signals.
- The second section performs the classification of DOA which in turn finds out which signals start from the user and which one forms the interferes.
- The final section is the beamforming algorithms which forms the antenna pattern with a main beam steered in the direction of the user, while minimizing the noise.

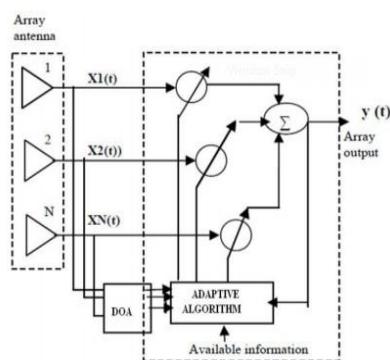


Figure1: Block diagram of smart antenna system.

1.1 DOA ESTIMATION ALGORITHM

DOA is a research area in array signal processing and many engineering application such as radar, radio astronomy, sonar and other emergency assistance devices that need to be supported by estimation of DOA[2]. Mainly DOA estimation is used as a part of research in the field of array processing, many works highlight radio direction finding. From the past few years, Wireless Local Area Network(WLANs) have increased rapidly due to its flexibility and convenience, high data rate is required in order to fulfill the requirements of advance frequency bands.

With the higher user density, frequency and data rate, multipath fading and cross interference becomes the main problem. In order to have higher communication capacity and the main problems are enhanced by smart antenna system which is used to be very effective in suppression of the interference [1][3].

The digital signal processing in smart antennas concentrates on the improving the efficient use of algorithms in direction of arrival estimation and beamforming techniques. There are many limitations in fixed antenna while used for DOA estimation. The main lobe beam of an antenna is inversely proportional to its shape, practically it is not an option to improve the accuracy of angle measurement with respective to an increase in the physical aperture of the incoming signals. The example like missile seeker or aircraft antenna can have a limited physical size, hence they are wide in beam width of the main lobe.

As they do not have a good resolute it's difficult to distinguish between incoming signal and interfere signal. There are different types of algorithms such as spectral estimation, ESPRIT, Min-norm and MUSIC [4]. The most popular and widely used subspace-based techniques to estimate the DOA of multiple signal sources is the MUSIC algorithm.

1.2 BASIC PRINCIPLE OF DOA ESTIMATION

DOA is radio signal for the direction of array antennas. if the received radio signal meets the condition of the narrowband far field, it can take the radio signal at the front as a plane. Array normal and the direction vector of the plane wave angles is the direction of arrival (DOA). The target estimated of DOA gives N snapshots $X(1) X(N)$, using an algorithm to estimate the value of multiple signals' DOA (θ).

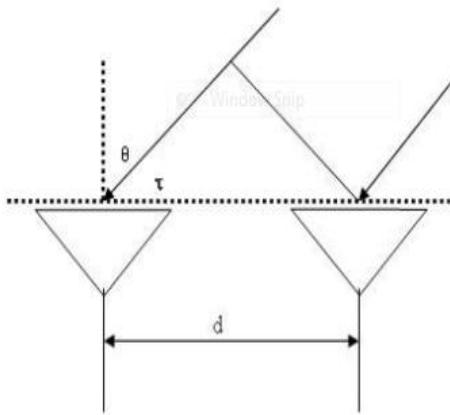


Figure2: The principle of DOA estimation

For generally far and wide signals, a wave-way difference exists when the same signal reaches different array elements. This wave-way difference leads to a phase difference between the arrival array elements. Using the phase difference between the array elements of the signal one can estimate the signal azimuth, which is the basic principle of DOA estimation [20].

For instance, Fig. 2 considers two array elements, d is the distance between the array elements, c is the speed of light,

θ is the incident angle of the far field signal,
 τ is the time delay of the array element.

The signal received by the antenna due to the path difference is

$$\tau = (d \sin \theta)/c$$

thus one can obtain the phase difference between the array elements as

$$\varphi = e^{-j\omega\tau} = e^{-j\omega(d \sin \theta)/c} = e^{-j2\pi f(d \sin \theta)/\lambda f_0}$$

where f_0 is the centre frequency. For narrow band signals, the phase difference is

$$\varphi = e^{-j2\pi(d \sin \theta)/\lambda}$$

where λ is the wavelength of the signal. Therefore, if the time delay of the signal is known, the direction of the signal can be gained according to formula, which is the basic principle of spatial spectrum estimation techniques.

In this paper, the following assumptions are used:

- 1) Point source assumption. Assume that the signal source is a point source, when looking from the array signal source, the opening angle is zero, and thus the signal source relative to the direction of the array is determined uniquely.

- 2) Narrowband signal hypothesis. That means that the signal bandwidth is far less than the reciprocal of the signal wave propagation across the largest diameter time. Meeting the narrowband assumption is to ensure that all array elements in the array can capture a signal at the same time.
- 3) Array assumptions. Assuming the array is located in the far field region of the source, the wave is projected to the plane wave. Assuming each element is the same lattice element and the position is accurate, the array element channel and amplitude and phase are consistent. This assumption guarantees that the array elements and their channel have no error.
- 4) Noise assumptions. Assuming the noise between each array element is zero, variance σ^2 is Gaussian white noise, statistical independently between each array noise and statistically independent between signal and noise.

II. MUSIC ALGORITHM

Multiple Signal Classification (MUSIC) algorithm was proposed by Schmidt. It has created a new era for spatial spectrum estimation algorithms. The promotion of the structure algorithm characterized rise and development, and it has become a crucial algorithm for theoretical system of spatial spectrum. Before this algorithm was presented, some relevant algorithms directly processed data received from array covariance matrices.

The basic idea of MUSIC algorithm is to conduct characteristic decomposition for the covariance matrix of any array output data, resulting in a signal subspace orthogonal with a noise subspace corresponding to the signal components. Then these two orthogonal subspaces are used to constitute a spectrum function, be got though by spectral peak search and detect DOA signals.

It is because MUSIC algorithm has a high resolution, accuracy and stability under certain conditions that it attracts a large number of scholars to conduct in-depth research and analysis.

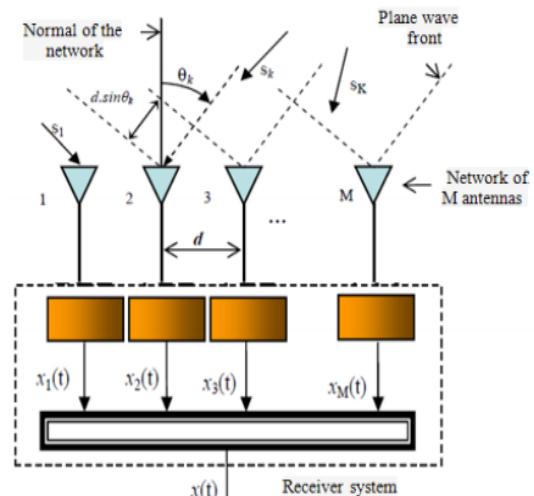


Figure3: Model of MUSIC estimation.

Let the number of signal sources k ($k=1,2,\dots,D$) to the antenna array, the wave front signal be $S_k(t)$, as previously assumed, $S_k(t)$ is a narrowband signal, and $S_k(t)$ can be expressed in the following form

$$S_k(t) = s_k(t)\exp\{j\omega_k(t)\}$$

Where $s_k(t)$ is the complex envelope of $S_k(t)$ and $\omega_k(t)$ is the angular frequency of $S_k(t)$.

As assumed before, all signals have the same centre frequency. So

$$\omega_k = \omega_0 = 2\pi c/\lambda$$

where c is electromagnetic wave velocity, λ is wave length.

The output signal of the m^{th} element is

$$x_m(t) = \sum_{k=1}^D s_k(t) \exp\left[-j(m-1)\frac{2\pi d \sin \theta_k}{\lambda}\right] + n_m(t),$$

where $n_m(t)$ is measurement noise; all quantities of labelled m belong to the m -th array element; all quantities of label k belong to the signal source k . Let

$$a_m(\theta_k) = \exp\left[-j(m-1)\frac{2\pi d \sin \theta_k}{\lambda}\right]$$

be the response function of array element m to signal source k .

This expression can be described by matrices

$$X = AS + N$$

Where,

$$X = [x_1(t), x_2(t), \dots, x_M(t)]^T,$$

$$S = [S_1(t), S_2(t), \dots, S_D(t)]^T,$$

$$A = [a(\theta_1), a(\theta_2), \dots, a(\theta_D)]^T$$

$$= \begin{bmatrix} 1 & 1 & \dots & 1 \\ e^{-j\varphi_1} & e^{-j\varphi_2} & \dots & e^{-j\varphi_D} \\ \dots & \dots & \dots & \dots \\ e^{-j(M-1)\varphi_1} & e^{-j(M-1)\varphi_2} & \dots & e^{-j(M-1)\varphi_D} \end{bmatrix},$$

$$\text{with } \varphi_k = \frac{2\pi d}{\lambda} \sin \theta_k,$$

$$N = [n_1(t), n_2(t), \dots, n_M(t)]^T.$$

The implementation steps of MUSIC algorithm are shown below. Obtain the following estimation of the covariance matrix based on the N received signal vector:

$$R_x = \frac{1}{N} \sum_{i=1}^N X(i)X^H(i)$$

to eigenvalue decompose the covariance matrix above

$$R_x = AR_s A^H + \sigma^2 I$$

According to the order of eigenvalues, take eigenvalue and eigenvector which are equal

to the number of signal D as signal part of space; take the rest, $M-D$ eigenvalues and eigenvectors, as noise part of space. Get the noise matrix E_n

$$A^H v_i = 0 \quad i=D+1, D+2, \dots, M,$$

$$E_n = [V_{D+1}, V_{D+2}, \dots, V_M],$$

vary θ ; according to the formula

$$P_{mu}(\theta) = \frac{1}{a^H(\theta) E_n F_n^H a(\theta)}$$

Calculate the spectrum function; then obtain the estimated value of DOA by searching the peak.

III. SIMULATION RESULTS

3.1 MUSIC ALGORITHM

The MUSIC algorithm of DOA estimation is simulated using MATLAB. In these simulations, 8 elements of ULA is consider which are equally separated by the distance of $\lambda/2$. The noise is ideal Gaussian white noise, SNR=20dB and number of snapshots is 200. The simulation has been run for two independent narrow band signals with equal amplitude, angle of arrival is 20° and 60° . The performance has been analyzed for different value of snapshots, array elements, spacing, SNR.

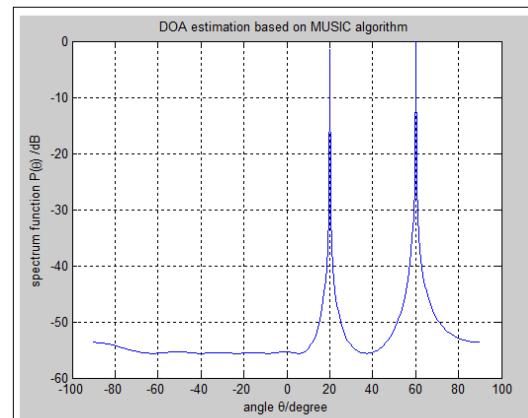


Figure 4: Basic simulation for MUSIC algorithm

CASE 1: The relationship between DOA estimation and the number of array elements

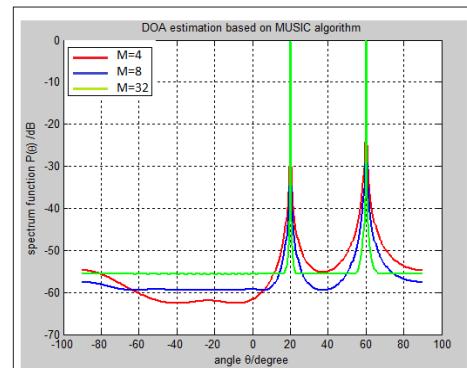


Figure 5: Simulation for the relationship between MUSIC algorithm and the number of array elements.

The three different values of the number of array elements are 4,8 and 32, also other condition remain as it is. Figure 5, shows that as the value of array elements increases, the spectral beam width becomes narrower.

CASE 2: The relationship between DOA estimation and the array element spacing: The three different values of the array spacing are $\lambda/6$, $\lambda/2$ and λ , also other condition remain as it is. Figure 6, shows that as the spacing between the array elements increases in terms of wavelength, the spectral beam width becomes narrower. But, when the spacing of the array elements is larger than half the wavelength, the estimated spectrum, except for the signal source direction, shows false peaks, so it has lost the estimation accuracy.

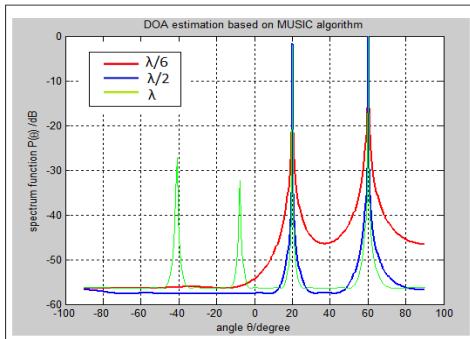


Figure6: Simulation for the relationship between MUSIC algorithm and array element spacing.

CASE 3: The relationship between DOA estimation and the number of snapshots.

The three different values of the number of snapshots are 5,50 and 200 without changing the other condition. As the number of snapshot increases the beam width of the spectrum becomes narrow while accuracy increases. Hence, the number of snapshots can be increased for accurate DOA estimation of signal.

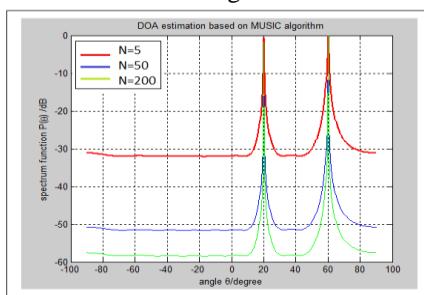


Figure7: Simulation for the relationship between MUSIC algorithm and the number of snapshots.

CASE 4: The relationship between DOA estimation and SNR.

The three different values of the signal to noise ratio are -20 dB, 0 and 20dB, again other condition remains constant. It is clear from Figure8 that as the value of SNR increases, the spectral beam width becomes narrower the direction of the signal becomes clearer. The accuracy of DOA estimation can be increased by increasing SNR. This implies that the performance of the MUSIC algorithm is affected by the value of SNR.

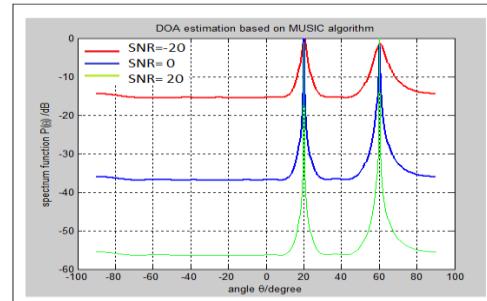


Figure8: Simulation for the relationship between MUSIC algorithm and SNR.

CASE 5: The MUSIC algorithm for coherent signals.

When the signals are coherent, let the incident angle is 20° and 60° respectively, those two signals are not correlated, the noise is ideal Gaussian white noise, the SNR is 20dB, the element spacing is half of the input signal wavelength, array element number is 10, and the number of snapshots is 200.

IV. CONCLUSION

This paper presents the MUSIC method based on the eigenvector of the sensor array correlation matrix to estimate angle of incoming signals. We give extensive simulation results to demonstrate the performance of the algorithms, which enhance the DOA estimation. Results obtained from the simulation of MUSIC algorithm include higher resolution for more number of array elements, more number of snapshots and larger SNR. When the array element spacing is not more than half the wavelength, the resolution of MUSIC algorithm increases correspondingly with the increase of array element spacing; however, if the array element spacing is greater than half the wavelength, the spatial spectrum causes false peaks in other direction except the direction of signal source. Hence, the MUSIC algorithm still has much room for development, and it is also worth further study.

REFERENCES

- [1]. Pascal Vallet,, Performance Analysis of an Improved MUSIC DoA Estimator, IEEE transactions on signal processing, VOL. 63, NO. 23, december 1, 2015.
- [2]. Z. Chen, G. Gokeda, and Y. Q. Yu. Introduction to Direction-of-arrival Estimation. Artech House, (2010).
- [3]. Y.J Huang, Y.W Wang, F.J Meng, G.L Wang. A Spatial Spectrum Estimation Algorithm based on Adaptive Beamforming Nulling. Piscataway, NJ USA. Jun, 2013. Pp 220-4.
- [4]. RK Jain, Sumit Katiyar and NK Agrawal," Smart Antenna for Cellular Mobile Communication", VSRD-IJEECE, Vol. 1 (9), 2011, 530-541.
- [5]. Raed M. Shubair, Mahmoud A. Al-Qutayri, and Jassim M. Samhan," A Setup for the Evaluation of MUSIC and LMS Algorithms for a Smart Antenna System", journal of communications, vol. 2, no. 4, june 2007.
- [6]. Adil Majdoubi, Mohamed Essaïdi," The Estimation of DOA in Smart Antenna Systems ", (IJITEE), ISSN: 2278-3075, Volume-1, Issue-6, November 2012.
- [7]. Haykin S, Reilly J P, Vertachitsch E. Some Aspects of Array Signal Processing. IEEE Proc. F, 1992, 139; p1~26.
- [8]. Haykin S. Array Signal Processing. Prentice Hall. 1985.
- [9]. S.Unnikernna Pillai. Array Signal Processing, Springer verlag, 1989.