Direction of Arrival Estimation using a Dynamic-MUSIC Algorithm

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Abstract: **The quality of mobile communication should be improved, however it is difficult to maintain the quality of mobile communication at certain locations because of multipath fading, delay spreading and co-channel interference. In order to cope with these problems, transmitters of the interference waves are estimated and it is desirable to improve the area design. An array antenna system with innovative signal processing can enhance the resolution of a signal direction of arrival (DOA) estimation. Super resolution algorithms take advantage of array antenna structures to better process the incoming signals. They also have the ability to identify multiple targets. This paper explores the eigen-analysis category of super resolution algorithm. A class of MUltiple Signal Classification (MUSIC) algorithms known as a Dynamic-MUSIC algorithm is presented in this paper. We focus on the dynamic environment of user i.e. user moves from his initial position to particular location. And by using Dynamic-MUSIC algorithm estimate their correct position location (PL) to provide the services to the desired user using extended version of MUSIC.**

Key words – **direction-of-arrival (DOA),** ***Dynamic MUltiple SIgnal Classification (DMUSIC),*** **Position Location (PL), Smart Antenna, Signal Processing.**

I. INTRODUCTION:

Wireless communication systems are limited in performance and capacity by three major impairments. These impairments are Rayleigh fading, delay spread and co-channel interference. Rayleigh fading causes due to multipath reception fig.1. [1]

Apart from these impairments, position location (PL) finding of the desired user is major aspect. If the elements of antenna are fixed and users are static in nature, accurate estimation of a Direction of Arrival of incoming signal is possible. In many commercial applications, it is important to find the position location of the desired user or to find the accurate possible direction of threat. Also it is helpful to dispatch the rescue team to the proper location. Multiple antenna arrays are more helpful as compare with single antenna for finding the accurate location of desired user.

Instead of using a single antenna, an array antenna system with innovative signal processing can enhance the resolution of signal DOA. An array sensor system has multiple sensors distributed in space. This array configuration provides spatial samplings of the received waveform. A sensor array has better performance than the single sensor in signal reception and parameter estimation. Its superior spatial resolution provides a means to estimate the direction of arrival of multiple signals.

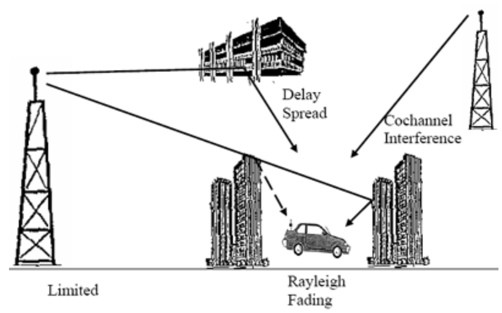

Fig. 1 Wireless system impairments

Basically many algorithms work on to find accurate position location (PL) of desired user i.e. MUSIC, ESPRIT & Propagator Method (PM) which includes spectral estimation, and eigen analysis [2], [3]. Also there are many different super resolution algorithms including spectral estimation, model based and eigen-analysis to name a few [4], [5], [6].

In this paper, we focus on the finding the position location of the desired user and extend the discussion, when the user is in dynamic nature. Extension of MUSIC i.e. Dynamic MUltiple SIgnal Classification (DMUSIC) and analyze detailed MATLAB simulation results for this algorithm [10], [11], [12].

II. BASICS OF SMART ANTENNA SYSTEM FOR DOA ESTIMATION:

Since most DOA estimation algorithm have reached a mature state, accurate estimation of the angle of arrival of signals impinging an array of antenna becomes the most important parameters regarding the performance of an adaptive array. Conventional methods, linear prediction methods, eigen structure methods and estimation of signal parameters via rotational invariance techniques etc. are the most powerful tool for DOA estimation. [7] All these methods are based on the digital beamforming antenna array. Incoming signals are received by the antenna elements and down converted to base band signal and fed into a digital signal processor chip where the algorithm can execute and processed on the incoming data, DOA is to be estimated. Till all this theories are analyzed for the static users i.e. users are fixed at their initial position (angle) and radiate the radiation pattern towards the antenna element. We focus the users are moves from one position to another and find out the correct position location (PL) of the desired user. We use here the extended version i.e. Dynamic-MUSIC algorithm for correct DOA estimation in the dynamic environment.

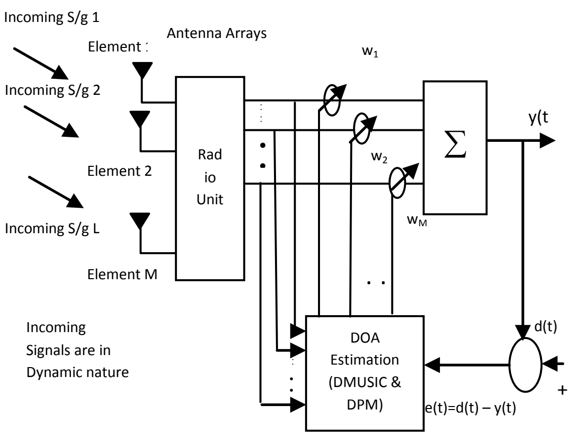

Fig. 2 Model for DOA Estimation

Fig. 2 shows, the basic schematic for DOA estimation here we use 6 antenna array elements (M) and 5 incoming signals (L) and the antenna elements are arranged with 0.5 spacing for better result. It also consists of radio unit for converting the signal in to base band signal. Then they are digitized and fed into a digital signal processor (DSP) chip. Here the Dynamic-MUSIC algorithm is executed to calculate the correct DOA for desire users.

III. DYNAMIC-MUSIC SYSTEM:

In this paper we use uniform linear array (ULA) structure with 6 elements and 5 incoming signals (L). Fig. 3 shows the general configuration for a ULA antenna having M elements arranged along a straight line with the distance between elements, d=λ/2, where λ is the incoming signal wavelength. Incoming signals are 5 out of these, one user is in dynamic condition and rest is static in nature.

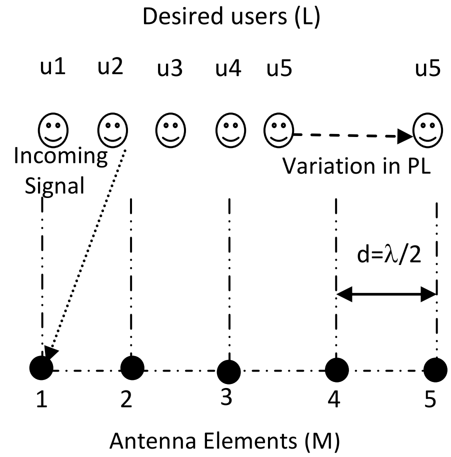

Fig.3 ULA Antenna & Dynamic user configuration

A signal impeding onto an antenna array over a linear communication channel with additive and temporally white Gaussian noise, the received time continuous and complex valued signal can, usually be expressed as

$$\tilde{x}(t) = \sum_{k=0}^{M} s(k)\tilde{h}(t-kT) + \tilde{n}(t), \qquad (1)$$

Here, $\tilde{h}(t)$ is the overall channel impulse response, $s(k)$ are the transmitted data symbols and $\tilde{n}(t)$ is the additive noise. The order of the communication channel is equal to *M* and the symbol period is *T*.
The signal, which is time continuous, received at the array that is sampled at the symbol rate or faster is turned into a time discrete signal, which is expressed as

$$x(t) = \sum_{k=0}^{M} h(k)s(t-k) + n(t), \quad t = 1........k \qquad (2)$$

Where *t* denotes the discrete time, $x(t)$ denotes the sampled received signal, $h(k)$ is the time discrete impulse response and $n(t)$ is the noise. In short, equation (2) is just a vector counterpart of the signal model in equation (1).

The input signal scalar valued quantities can be expressed as vector in

$x(t) = [x_1(t) \;\; x_2(t) \;\; \cdots x_L(t)]^T$; Received signal (3)

$h(k) = [h_1(k) \;\; h_2(k) \;\; \cdots h_L(k)]^T$; Channel impulse Response (4)

$n(t) = [n_1(t) \;\; n_2(t) \;\; \cdots n_L(t)]^T$; Additive noise (5)

The *L* signal equation received at *M* array can be described by

$$x(t) = \sum_{k=0}^{M} h(k)s(t-k) + n(t), \quad t = 1,......,k \qquad (6)$$

Since the number of signals/parallel channels is *L*, the vectors $x(t)$, $h(k)$ and $n(t)$ will all be *L*-dimensional column vectors. The number of antenna elements/order of the channel is *M* and the number of received vectors snapshots is *K*.

The received vector valued signal descript in equation (6) is a linear combination of the vectors *h(0),…..,h(M)*. Since subspace based methods exploit algebraic description, it is possible to formulate equation (6) using a matrix description with the following definitions.

$$H_1 = \left[h_L(0) \ \cdots \ h_L(M) \right] \quad (7)$$

$$s_{M+1}(t) = \left[s(t) \ \cdots \ s(t-M) \right]^T \quad (8)$$

The snapshots of the outputs of L channels at time instant t are stored in $x(t)$. Together with the structure of $s_{M+1}(t)$ and $n(t)$, matrix H completely determines the communication system. Therefore, equation (6) can be rewritten as

$$x(t) = H_1 s_{M+1}(t) + n(t) \qquad t = 1, \cdots, K \quad (9)$$

To fully utilize the convolutive structure of the communication systems, *N* vectors samples is applied to the vector valued sequence $x(t)$. Equation (9) can be rewritten as

$$x_N(t) = H_N s_{N+M}(t) + n_N(t) \qquad t = N, \cdots, K \quad (10)$$

Where

$$x_N(t) = \left[x_1^T(t) \ \cdots \ x_1^T(t-N+1) \right]^T, \text{ Received matrix} \quad (11)$$

$$n_N(t) = \left[n_1^T(t) \ \cdots \ n_1^T(t-N+1) \right]^T, \text{ Noise matrix} \quad (12)$$

$$s_{M+N}(t) = \left[s(t) \ \cdots \ s(t-N+1) \right]^T, \text{ Data matrix} \quad (13)$$

$$H_N = \begin{bmatrix} H_1 & 0_{Lx1} & 0_{Lx(N-2)} \\ 0_{Lx1} & H_1 & 0_{Lx(N-2)} \\ \vdots & \vdots & \vdots \\ 0_{Lx1} & 0_{Lx(N-2)} & H_1 \end{bmatrix}, \text{ Channel matrix} \quad (14)$$

From equation (14), the $0_{i \times j}$ denotes a *i x j* matrix with zeros. The *NL* x (*N+M*) matrix H_N is block Toeplitz with *N* block rows and with the first block row equal to $[\ H_1 \ \ 0_{Lx(N-1)}\ ]$. The number of block rows, *N*, is called the length of the temporal window and H_N is known as the channel matrix. Using a simpler notation, equation (10) can be rewritten

$$x(t) = Hs(t) + n(t) \quad (15)$$

For the a conventional antenna, the main lobe beam width (MLBW) of an antenna manner is given by, in radians

$$MLBW = K \frac{D}{\lambda} \quad (16)$$

Where *D* is the diameter of the antenna array and k is proportionality constant, for most case $k \approx 1$.
The received signals can be models as with reference to the data model of equation (10)

$$x(t) = A(\theta)s(t) + n(t) \quad (17)$$

Here, *s(t)* denotes the signal vector and $n(t)$ represent the additive temporally and spatially white Gaussian noise. The array response matrix is for the special case of linear array given by

$$A(\theta) = \begin{bmatrix} 1 & 1 & \cdots & 1 \\ e^{-j\beta(\theta_1)} & e^{-j\beta(\theta_2)} & \cdots & e^{-j\beta(\theta_L)} \\ e^{-j2\beta(\theta_1)} & e^{-j2\beta(\theta_2)} & \cdots & e^{-j2\beta(\theta_L)} \\ \vdots & \vdots & \vdots & \vdots \\ e^{-j(M-1)\beta(\theta_1)} & e^{-j(M-1)\beta(\theta_2)} & \cdots & e^{-j(M-1)\beta(\theta_L)} \end{bmatrix} \quad (18)$$

where $\theta = [\theta_1 \quad \theta_2 \quad \cdots \quad \theta_M]^T$ denotes the vector containing the direction of arrivals of the *L* incoming signals. [8], [9]

IV. SIMULATION RESULTS:

Computer Simulation has been conducted to evaluate the DOA estimation using Dynamic-MUSIC. Fig.4, show that the incoming signals estimated angles are 25.3^0, 50^0, 80^0, 130.1^0 & 155.1^0 for the position location of the desire users are 25^0,50^0,80^0,130^0 and 155^0 respectively. And the real directions are estimated (Fig.5 & Table.1).

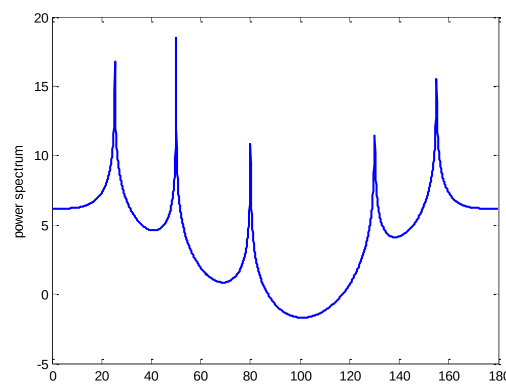

Fig. 4 Simulation result for static analysis (L=5,M=6)

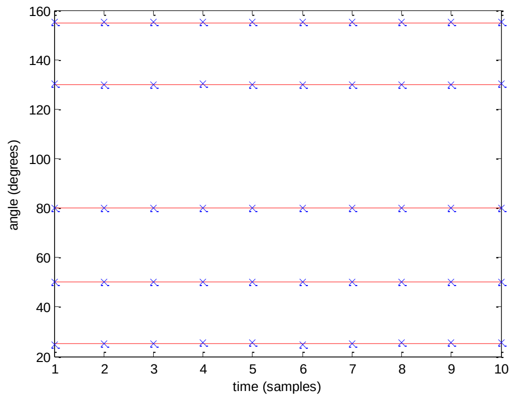

Fig. 5 Estimated directions for users (static analysis)

For the dynamic analysis, we consider the five incoming signals with different directions. Out of 5, four users at an angle of 25^0, 80^0, 130^0 and 155^0 and one user movement is starts at an angle of 50^0 and end at 89.99^0 with the time interval of 0.4000. Fig. 6 show the estimated angles 25.7^0, 80^0, 130^0, and one peak of an angle 155^0 is missing. Estimated angle of dynamic user is 88.6^0.

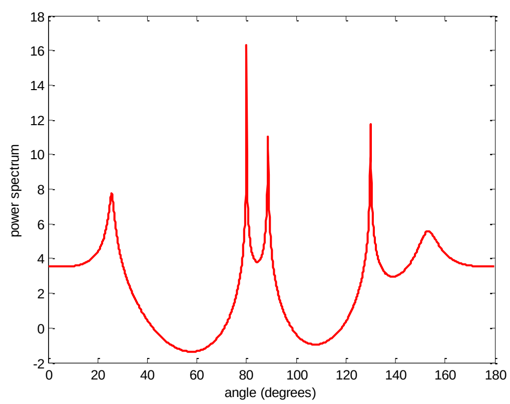

Fig. 6 Simulation result for dynamic analysis (L=5,M=6)

And the estimated real directions for the dynamic are shown in Fig. 7 and Table.2.

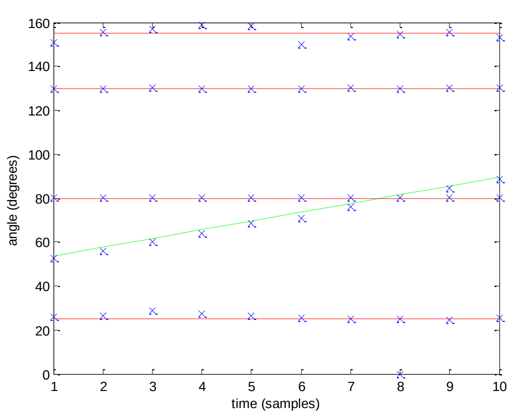

Fig. 7 Estimated directions for users (dynamic analysis)

V. CONCLUSION:

We have evaluated Dynamic-MUSIC algorithm based on the eigenvector decomposition to estimate angle of incoming signals. By studying this we come to following conclusion:

- In static condition the estimated DOA are quite better than Dynamic-MUSIC.
- All the estimated angles in MUSIC are much closer to the desired position location.
- In Dynamic-MUSIC one user is totally mission.
- When all incoming signals 20^0 & 30^0 apart then MUSIC algorithm estimate all DOA of incoming signals with small deviation in its angle.
- For proposed method i.e. Dynamic-MUSIC the angle below than 30^0 & above 140^0 are missing.
- But the user who moves from his initial position and rest at particular location, his DOA is measured accurately with high amplitude in Dynamic-MUSIC algorithm.

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TABLE 1: ESTIMATED ANGLES FOR EACH INCOMING SIGNALS [STATIC ANALYSIS]

Samples / Incoming Signal (IS)	10	20	30	40	50	60	70	80	90	100
IS 1	24.95	25.25	25.05	25.45	25.65	24.75	25.15	25.45	25.65	25.45
IS 2	50.15	50.15	50.15	50.05	50.05	50.15	50.15	50.05	50.15	50.05
IS 3	80.05	80.15	80.15	80.05	80.15	80.15	80.05	80.05	80.05	80.15
IS 4	130.15	130.05	129.95	130.15	129.95	129.95	129.85	129.95	130.05	130.25
IS 5	155.15	155.15	155.15	155.15	155.25	155.05	155.25	155.05	155.25	155.15

TABLE 2: ESTIMATED ANGLES FOR EACH INCOMING SIGNALS. [DYNAMIC ANALYSIS]

Samples / Incoming Signal (IS)	10	20	30	40	50	60	70	80	90	100
IS 1	25.85	26.65	28.75	27.35	26.35	25.35	25.25	25.25	24.75	25.65
IS 2	52.75	55.75	60.05	63.65	68.35	71.05	76.05	80.15	80.05	80.05
IS 3	80.05	80.05	80.05	80.05	80.15	80.05	80.05	130.05	84.45	88.75
IS 4	130.05	130.05	130.25	130.05	130.05	130.05	130.15	154.55	130.15	130.15
IS 5	150.75	155.75	156.75	158.95	158.45	149.95	153.65	0	155.75	153.35