Adaptive Based Beamforming for Direction-of-Arrival (DOA) Estimation

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***Abstract*— Nowadays mobile users require increased coverage, quality of service and capacity in wireless communications. By using adaptive antennas at both transmitter and receiver in wireless communication, we can easily reach the required improvement. In this paper Adaptive Antenna plays a major role to improve the performance of wireless communication systems. This antenna uses advanced signal processor that modifies self generated beam pattern to strengthen the desired signal and reduce the unwanted signal. This type of antenna is called as smart antenna. This paper presents the adaptive array antenna fundamentals, direction of arrival and several digital beamforming algorithms. Performance analysis of the MUSIC algorithm with LMS is studied because these results are useful for the smart antenna system design. All the simulations are carried out using MATLAB.**

***IndexTerms:*ArrayAntenna, Beamforming, DOA, LMS, MUSIC.**

I. Introduction

Smart antenna can be used for finding angle of arrivals accurately by accurate direction finding technique. The best spectral estimation isolates the angles of arrivals and gives better resolution. This estimation is used for tracking moving object accurately. Smart antenna is classified into two systems, switch beam and adaptive beam formed system. Switch beam system has several fixed beam patterns, whereas in second system, beam is guided by the antenna towards desired direction. To find the exact location of the desired user adaptive antenna direction finding capabilities are used. Previously this adaptive array antenna technology was used in military communication, which needs narrow beams to avoid noise signals. Nowadays this technology is used for the personal communication. This paper presents direction of beam towards desired target and rejecting all other targets towards an unwanted direction by using Multiple Signal Classification (MUSIC) algorithm[1] with the digital beamforming algorithm LMS.

A Adaptive Array Systems

This system reduces the interference that occurs from different users by introducing nulls in the same direction and at the same time follows the user with its movement. Adaptive array antenna system working is truly smart in the real time applications. In this system, digital signal processing element provides the intelligence.

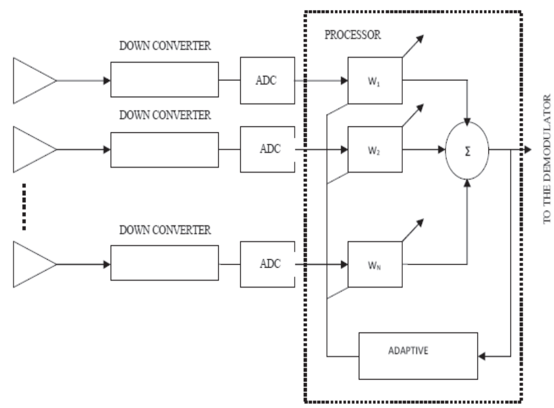

Fig.1: Adaptive Array Antenna

The Adaptive Array Antenna system is shown in Figure 1. Here individual signals are combined to get single desired digital beamforming output. Baseband frequencies are brought down by using down converter and then weighted by adaptive antenna array systems. Down-converted signal is processed by A to D converter. The digital signal processor is the key element of a system. This processor changes incoming information by updating the weights to reduce the noise. Adaptive technology reduces co-channel interference and handoff, increases capacity and range of communication, and reduces transmitting power etc.

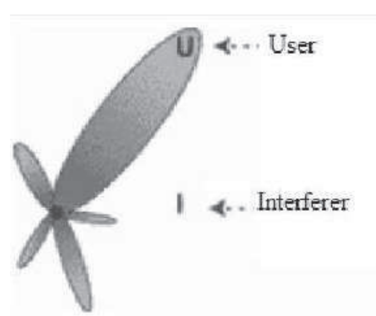

Fig.2: Beam Pattern In Adaptive Array [4]

Adaptive Array Beam pattern is shown in figure 2.This antenna is truly smart so it updates the information with the varying situation. This system uses signal processing element with antenna array. It switches the radiation in the direction of wanted user. It too reduces the interference that occurs from different users [4].

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II. Direction Of Arrival Estimation

Direction finding (DF) is a process for determining the transmitting source by observing the direction of arrival along with signal properties. Figure 3 shows the Direction Finding system.

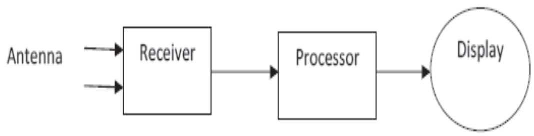

Fig.3: DF System

Direction of Arrival estimation using a single antenna has many disadvantages such as its resolution is very poor. To improve the resolution, an array of antenna system with signal processing can be used [4]. DOA estimation depends on number of users, spacing and number of signals. In this paper, we concentrate on estimation of DOA using MUSIC (Multiple signal classification) algorithm[1]. To explain the behaviour of larger array linear array can be used and it has maximum two elements. This fundamental array helps to study the phase relation of nearby array elements and the electric field of two elements. Pattern multiplication is the product of array factor and antenna pattern. If we know an array factor, pattern multiplication is derived from antenna pattern. Array factor plot shows number of side lobes along with main lobe. We need to suppress side lobes to reduce the interference by weighting the array components. This is very important in signal processing.

For any array, the array factor is given by

$$AF = \sum_{n=1}^{N} w_n e^{jk(Xn\sin\theta\cos\phi + Yn\sin\theta\sin\phi + Zn\cos\theta)} \qquad (1)$$

Where N is number of elements

w_n is complex weight for element n

$k = 2\pi / wavelength$

(Xn, Yn, Zn) is location of element n

(θ, ϕ) is direction in space

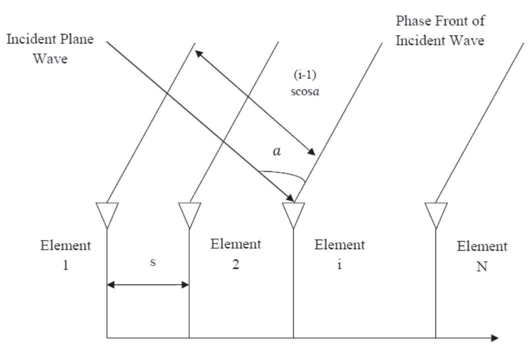

Fig.4: Geometry Of Linear Array

Figure 4 shows the construction of linear array. It contains number of array elements. The array radiation pattern is the multiplication of the radiation patterns of the entity constituents and array factor in the adaptive array and is given by

$$R(W,a,\varphi e) = r(W,a,\varphi e) * a(W,a,\varphi e) \qquad (2)$$

The array factor depends upon array amplitude level and the overall pattern adopted by updating current at each element. The limited RF spectrum can be efficiently used with the help of an adaptive array antenna in mobile communication. Antenna pattern is directed by updating the array weights using signal processing element. This process is known as adaptive array antenna system. Signal processing along with mobile communication improves capacity and quality of service, and it gives narrow beam towards desired user and null in the direction of interference.

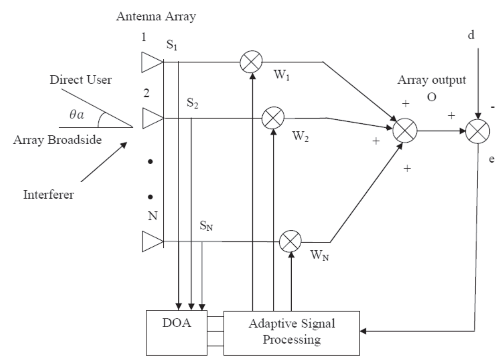

Fig.5: Adaptive Array Antenna [19]

Fig.5 shows that N number of elements are equally spaced by a distance S. The desired user signal arrives from desired angle θ_a with interference signal. These N no of signals feed to the direction of arrival estimation technique and process with the error signal e. These signals are generated by comparing the array outputs with the desired signal d. This paper describes the adaptive array antenna with uniformly placed linear array elements along with LMS algorithm[2][4]. In this beamforming technique, weights are updated to attain the desire signal with an angle θ_a and optimizing the signal to noise ratio through the above algorithms [3].

The array factor for N elements is given by

$$a(\theta_a) = \sum_{m=0}^{N-1} A_m e^{(im(\frac{2\pi}{Y}\cos\theta_a + B))} \qquad (3)$$

where B indicates the phase shift among the internal elements of the array and it is given by the following equation

$$B = \frac{-2\pi S}{Y_0}\cos\theta_a \qquad (4)$$

where θ_a gives the angle between the desired user direction and array broadside.

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III. Beam forming Algorithms

A. LMS Algorithm

In this algorithm, the gradient vector uses number of iteration on the available data and it updates the weight vectors to reduce the minimum square error. The optimum weight vector depends on the input covariance matrix C and 1 input filter matrix Cc. By using this technique, the array output is given by

$$O = W^T(n) * S \qquad (5)$$

$$e = d - O$$

$$w(m+1) = w(m) + \varsigma e\ S \qquad (6)$$

Equation (6) gives the updated weight vector for m iterations, where ς is known as the rate of adaptation limited by array antenna processing gain [5]. Convergence condition is given by equation (7).

$$0 < \varsigma < \frac{1}{Y_{\max}} \qquad (7)$$

where,

$Y_{\max}$ is Largest value of eigen in autocorrelation matrixCa. Convergence becomes slow when ς is kept very small, so that the rate of adaptation ς is selected within the limits. The autocorrelation matrix of input vector is given by

$$Ca = SS^T \qquad (8)$$

The cross correlation matrix is given by

$$Cc = E[S\ d] \qquad (9)$$

while,

d is the desired reference signal.Optimum weight vector is obtained by using the following equation

$$w_0 = CcCa^{-1} \qquad (10)$$

The computations to be carried out at various parameters like the number of elements and spacing . The required signal which is to be used phase modulated and the SNR is assumed to be 20dB. It is given by an equation[5]

$$S(t) = e^{isin(W*t)} \qquad (11)$$

where,

W is the signal frequency in radian.

B. Music Algorithm

MUSIC stands for Multiple Signal Classification and it is used to separate the signal from noise using eigen decomposition of the correlation matrix of the received signal based on orthogonality property. It gives the information about number of incident signals, direction of arrival (DOA) of each signal strength[5]. The input signal correlation matrix is given by

$$C_c = A * C_s * A^H + \sigma_{n2} * I \qquad (12)$$

where

A is N x P array steering matrix equal to

$$[\,a(\theta a1)\ \ a(\theta a2)\ \ a(\theta a3)\ \ldots\ldots\ldots .a(\theta aD\ ))]$$

and Cs is P x P source correlation matrix equal to $[s_1\ \ s_2\ \ s_3\ \ldots\ldots\ldots\ ..s_D\,]^T$ .The Pseudospectrum of MUSIC is given by an equation

$$P_{MUSIC(\theta US} = 1/\,abs((a(\theta bs^H * a(\theta(\theta * S_{vN} * S_{vN^H}\ (13)$$

The resolution of MUSIC diminishes when noise variances varies or signal sources are coherent. The correlation matrix is estimated by time averaging given by an equation as follows:

$$C_c = \frac{1}{m}\sum_{1}^{m} S * S^H \qquad (14)$$

$$C_c = A * C_s * A^H + A * C_{sn} + C_{ns} * A^H + C_n (15)$$

Now high angular resolution is achieved by the above expression.

IV. Results And Performance Analysis

This paper describes the simulation results of LMS algorithm and the effects of number of elements, spacing between them on array factor. Similarly, subsequent sections give the effect of elements and spacing distance on the mean square error (MSE). The simulations are carried by changing various parameters like N number elements used in array, each element spaced by distance S and several number of iterations used for computations. These are used to analyze the features of digital beamforming.

In this system all the elements are equally spaced to form a linear array and the distance between the array element S = λ/ 2, the optimal weight vector is calculated for the number of elements N=4,8 and 12. The normalized array factor is found by considering the angle of arrival in degree.

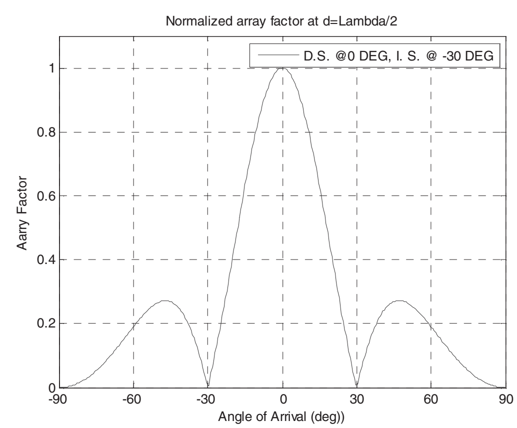

Fig.6: Plot between AOA and AF for N=4

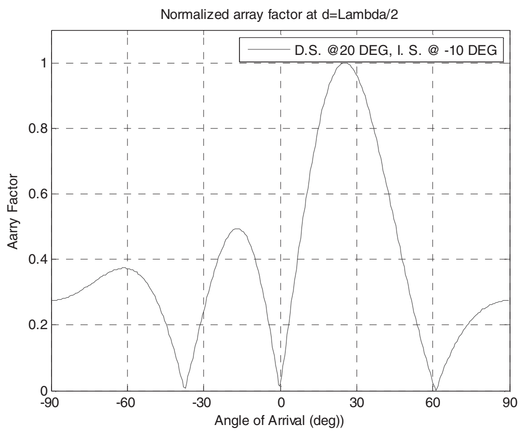

Fig. 7: Plot between AOA and AF for N=4

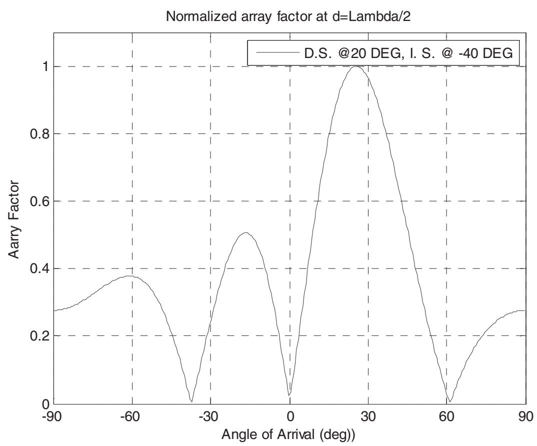

Fig. 8: Plot between AOA and AF for N=4

Figure 6,7 and 8 gives the plot between Angle of Arrival and Array Factor for different users 0 and 20 degrees on -30,-10 and -40 degrees interferer for number of elements N=4. Beam width becomes broader due to the decrease of number of antenna elements.

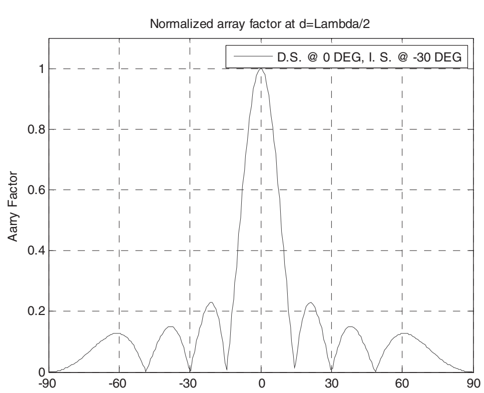

Fig 9: Plot between AOA and AF for N=8

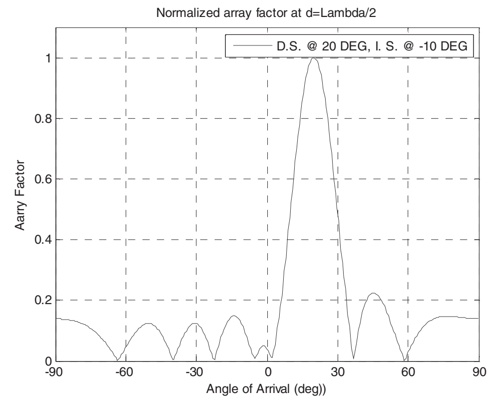

Fig.10: Plot between AOA and AF for N=8

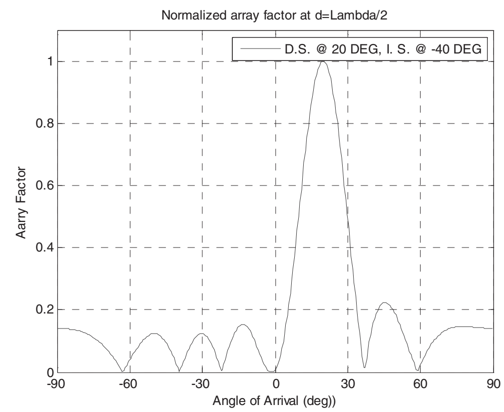

Fig.11: Plot between AOA and AF for N=8

Figure 9,10 and 11 gives the plot between Angle of Arrival and Array Factor for different users and interferer for number of elements N=8.

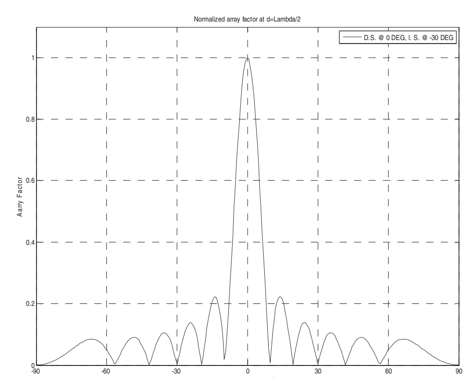

Fig.12: Plot between AoA and AF for N=12

Figure 12 shows the plot between the AOA and AF for 0^0 degrees desired user for -30^0 degrees interferer and give deep nulls on the -30 degrees .

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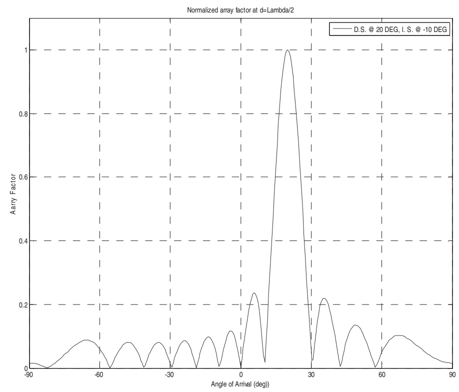


Fig.13: Plot between AoA and AF for N=12

Figure 13 shows that deep null is achieved at -10 degrees interferer and 20 degrees desired user . Here the interference signal is blocked and the desired signal is produced respectively.

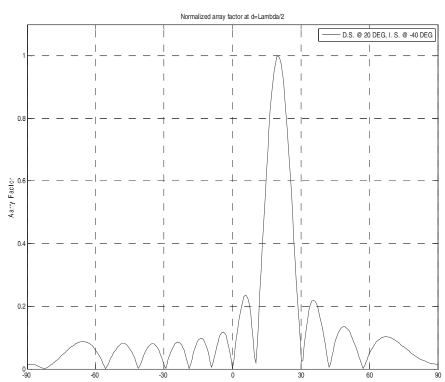
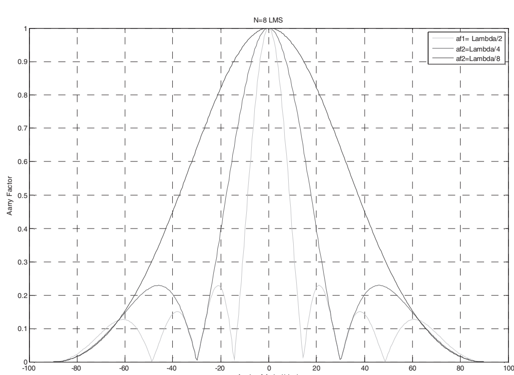


Fig.14: Plot between AoA and AF for N=12

Figure14 clearly shows that the beamwidth of the signal becomes more broadened by changing number of array elements. Here the no of elements are N=12.



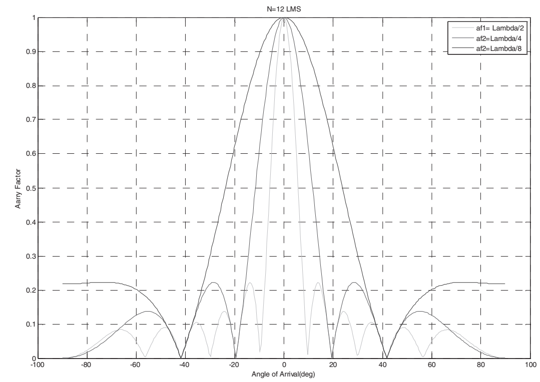


Fig.15: Plot between AOA and AF at spacing λ /2 , λ /4 and λ /8 (N = 8 and N=12)

Figure 15 shows that the spacing between the array element for different values λ / 2, λ/ 4 and λ/ 8 for N = 12 and N = 8. We can clearly observe that the false echoes and beam spread out are caused due to change in separation gap. Hence the spacing is a critical issue. Narrow beamwidth is achieved due to the decrease of no of elements.

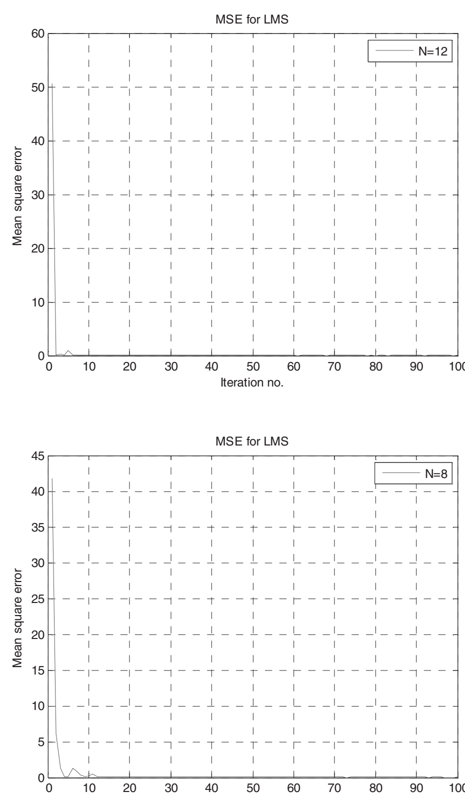


Fig.16: Plot of Mean square error at S = λ / 2

Figure 16 shows that how the mean square error get reduced by changing the no of elements N=8 and N=12 for constant spacing S= λ/2. MSE is minimum at N = 12 compared to the N = 8.The gradient vector is an estimation

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technique in LMS algorithm and it uses no of iteration on the available data for beamforming. MSE is reduced by changing the weight vectors in the gradient with a negative direction.

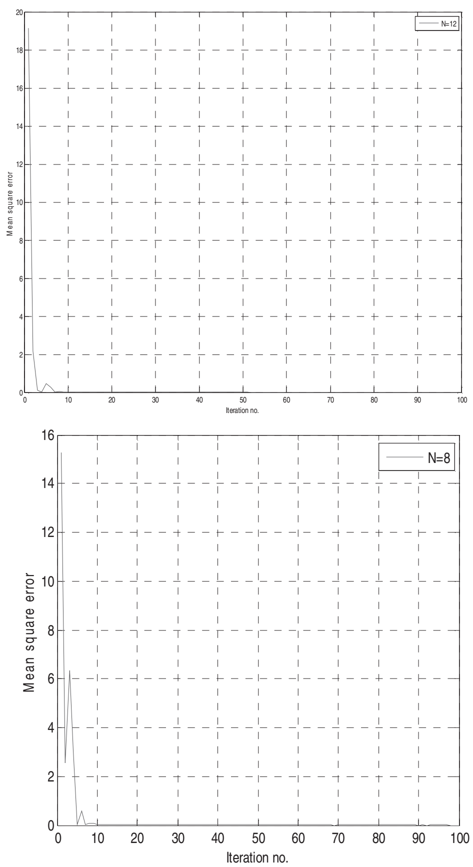

Fig.17: Plot of Mean square error at S = λ / 4

Figure 17 shows that the MSE is minimum at N = 12 compared to the N = 8 with constant spacing S= λ/ 4. MSE is minimum at N = 12 compared to the N = 8. We conclude that MSE is minimum when the array elements and spacing distance is more.

PERFORMANCE ANALYSIS OF LMS ALGORITHM

TABLE-1

EVALUATION OF BEAMWIDTH AND NO OF SIDE LOBES FOR LMS

(Ds=0(or) 20 degrees ,IS=-30 (or) -10 degrees, S= λ/ 2)		
No.Array	No of sidelobes	Beamwidth
N=4	less	Broad
N=8	Medium	Medium
N=12	More	Narrow
(Ds=0(or) 20 degrees ,IS=-30 (or) -10 degrees, N=12)		
Spacing	No of sidelobes	Beamwidth
λ/ 2	Optimum	Narrow
λ/ 4	Medium	Medium
λ/ 8	Less	Broad

From Table1 we can clearly observe that LMS algorithm provides narrow beamwidth with more number of side lobes and it increases resolution with more number of array elements.

TABLE-2

EVALUATION OF MSE FOR LMS

(Iteration = 100, D. S. = 20 Deg., I. S. = -10 Deg.)

No.Array	Spacing	MSE
N=12	λ/ 2	Less
	λ/ 4	More
N=8	λ/ 2	More
	λ/ 4	Less

Table 2 clearly shows that the mean square error is reduced due to the increase of number of elements.

SIMULATION RESULTS OF LMS & MUSIC ALGORITHMS

All the simulation carried out through MATLAB .Here the required signal used is phase modulated with SNR 20 dB. Linear array is used for N number of elements and results are obtained by studying and changing various parameters like no of elements and Spacing that is clearly observed in Figure.18

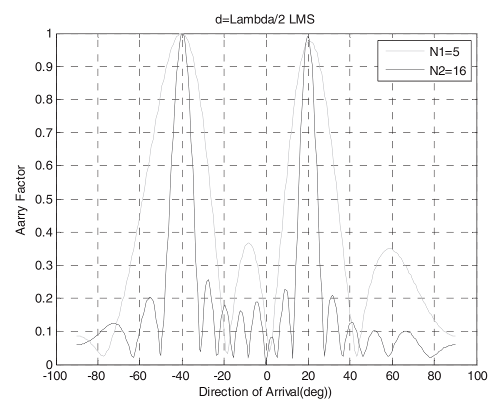

Fig.18: LMS, Plot Between DOA And AF At S= Λ/ 2 And Λ/ 4 For Different Elements

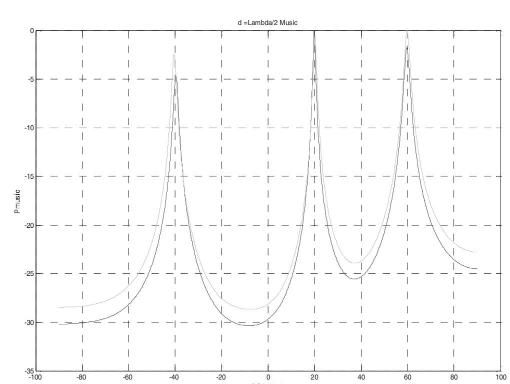

Fig. 19: MUSIC, Plot between DOA and AF at different Elements.

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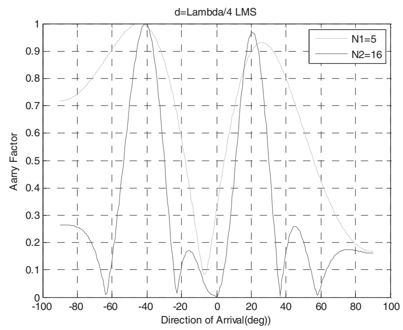
The Figure 19 shows that the spectrum MUSIC in dB for array spacing S= λ/ 2 for no of elements N=5,16 and SNR = 20 dB for direction of arrivals -40, 20, 60 degrees.In MUSIC spectrum sharper peaks are formed and resolution is improved by increasing no of elements.

V.CONCLUSION



In this paper Least Mean Square algorithm is used for adaptive beamforming. The performance of mobile communication is improved by using beamforming algorithm. This is proved with the help of simulation results. Simulation results of LMS show that the beamwidth of the signal becomes more broadened by changing number of array element. That is by decreasing the number of antenna elements, beam width becoming broader as compared to more number of elements with a reduced number of side lobes. By changing the spacing between the elements implies that at more number of antenna array element and the distance between them is decreasing, the beamwidth of the lobes increases. MSE is minimum at large number of elements compared to the fewer elements. LMS algorithm composed consecutive alteration in the weight vector within the path of the gradient vector that minimizes the mean square error |e2|. It is concluded that mean square error is minimum when the array elements and spacing distance is more.

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