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A SMART ANTENNA BASED TECHNIQUE FOR DIRECTION OF ARRIVAL FOR MUSIC, MVDR, LMS – MUSIC, LMS –MVDR AND ERADICATE THE CHALLENGES BEING FACED WITH THE TRADITIONAL OMNI DIRECTIONAL ANTENNA – SIMULATION AND RESULT

Nida Khan and Kamal Niwaria

Department of Electronics & Communication Engineering, SRK University, Bhopal (MP) India

e-mail: nidaimrozmalik@gmail.com, kamalniwaria@gmail.com,

**Corresponding Author: nidaimrozmalik@gmail.com*

ABSTRACT
The article is going to establish that Smart antenna systems are capable of efficiently utilizing the radio spectrum and are a promise for an effective solution to the present wireless systems problems while achieving reliable and robust high-speed, high-data-rate transmission. In fact, smart antenna systems comprise several critical areas such as individual antenna array design, signal processing algorithms, space-time processing, wireless channel modeling and coding, and network performance.

INTRODUCTION
A device able to receive or transmit electromagnetic energy is called an antenna. Antennas have become ubiquitous devices and occupy a salient position in wireless system experienced the largest growth among industry systems Antennas couple electromagnet energy from one medium (space) to another medium as wire, coaxial cable, or wave guide. Physical designs can vary greatly. Antenna produces complex electromagnetic fields both near to and far from antennas. Not all of electromagnetic fields generated actually radiated into space. Some of the fields remain in the vicinity of antenna and are viewed as reactive near fields; much the same way as inductor or capacitor is a reactive storage element in lumped element circuits.

BEAMFORMING
Beamforming is a general signal processing technique used to control the directionality of the reception or transmission of a signal on a transducer array. Beam forming creates the radiation pattern of the antenna array by adding the phases of the signals in the desired direction and by nulling the pattern in the unwanted direction. The phases and amplitudes are adjusted to optimize the received signal, as shown in figure 4.1. A standard tool for analyzing the performance of a beam-former is the response for a given N-by-1 weight vector W (k) as function of□, known as the beam response. This angular response is computed for all possible angles.

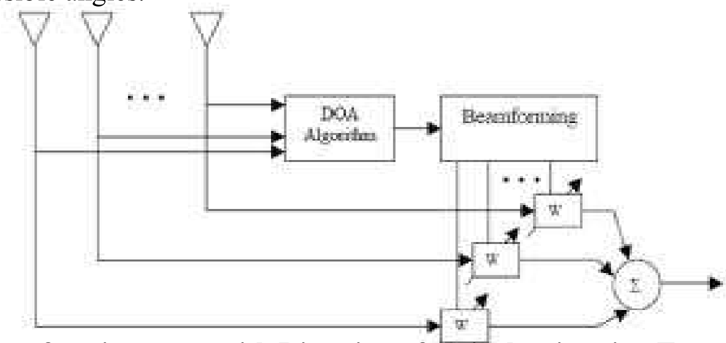

Figure 4.1: Beamforming setup with Direction of arrival estimation Type of beamforming algorithm

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i) Fixed Weight Beamforming

ii) Adaptive Beamforming

Fixed Weight Beamforming -

A Fixed weight beam-former is a smart antenna in which fixed weight is used to study the signal arriving from a specific direction. Since it optimize the signal arriving from specific direction while attenuating signals from other directions, thus it is called the spatial matched filter.
In the fixed weight beamforming approach the arrival angles does not change with time, so the optimum weight would not need to be adjusted

Different techniques of Fixed Weight Beamforming –

1. Maximum Signal-to-Interference Ratio

2. Minimum Mean-Square Error Method

3. Maximum Likelihood Method

4. Minimum Variance Distortion less Response Method

Adaptive Beamforming Algorithms –
Adaptive Beamforming is a technique in which an array of antennas is exploited to achieve maximum reception in a specified direction by estimating the signal arrival from a desired direction (in the presence of noise) while signals of the same frequency from other directions are rejected. This is achieved by varying the weights of each of the sensors (antennas) used in the array. It basically uses the idea that, though the signals emanating from different transmitters occupy the same frequency channel, they still arrive from different directions. This spatial separation is exploited to separate the desired signal from the interfering signals. In adaptive beamforming the optimum weights are iteratively computed using complex algorithms based upon different criteria. As shown in figure 4.4.
Beamforming is generally accomplished by phasing the feed to each element of an array so that signals received or transmitted from all elements will be in phase in a particular direction. The phases (the inter element phase) and usually amplitudes are adjusted to optimize the received signal. The array factor for an N-element equally spaced linear array is given,

$$\Phi 0 = (2\pi/\lambda 0)d \cos \theta \qquad \text{………….(4.2 9)}$$

Φ 0 is the desired beam direction. At wavelength $\lambda 0$ is the phase shift corresponds to a time delay that will steer the beam to Φ 0

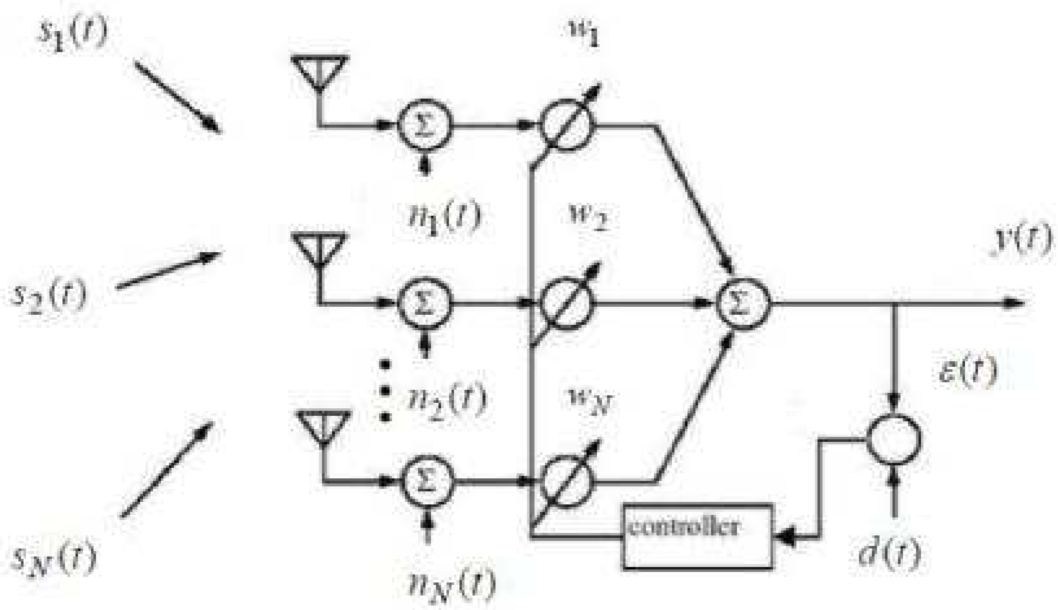

Figure 4.4: An Adaptive array system

Temporal Reference Algorithms –
Temporal reference algorithms (TR) are based on prior knowledge of the time structure of the received signals. Usually a training sequence is used as a temporal reference. The receiver aims to adjust or choose antenna weights in a way so that the deviation of the combined signal at the output and in the known training sequence is minimized. The calculated weights are then used to form a beam pattern.

Least Mean Squares Algorithm -

This algorithm was first developed by Widrow and Hoff in 1960. The design of this algorithm was stimulated by the Wiener-Hopf equation. By modifying the set of Wiener-Hopf equations with the stochastic gradient approach, a simple adaptive algorithm that can be updated recursively wa developed. This algorithm was later on known as the least-mean-square (LMS) algorithm. The algorithm contains three steps in each recursion: the computation of the Processed signal with the current set of weights, the generation of the error between the processed signal and the desired signal, and the adjustment of the weights with the new error information by the gradient method. This algorithm like the preceding one requires a reference signal and it computes the weight vector using the equation:
w(n+1)=w(n)+μx(n)[d*(n)-xH(n)w(n)]
…………(4.3
0)

Where μ is the gain constant that controls the rate of adaptation, i.e. how fast and how close the estimated weights approach the optimal weights. The convergence of the algorithm depends upon eigenvalues (the array correlation matrix), the array correlation matrix. In a digital system, the reference signal is obtained by periodically transmitting a training sequence that is known to a receiver, or using the spread code in the case of a direct-sequence CDMA system. The LMS algorithm described here a basic structure for most dynamic adaptive algorithms. This method requires information about a reference signal. The LMS beamformer configuration is shown in Figure 4.5.LMS beam former The output of the array is given by –

y(t) = wHx(t)

The reference signal d(t) generated at the receiver is usually assumed to have similar statistical properties as the transmitted signal.

For the purpose of simulation, we will simply assume that the reference signal is identical to the incoming signal. The error signal e(t) = d(t) − y(t) is fed into the weight updating algorithm. The criterion for determining the weights is based on minimizing the mean squared error (MSE) between the beamformer output and the reference signal:

E[e2] = E[(d-y)2]

= E[(d-wHx)2]

= E[d2]-2E[d-wHx]E[(wHx)2]
= = E[d2]-
2wHr+wHRxxw
…………(4.3
1)
=
= Where Rxx= E[xxH] is the autocorrelation matrix of the received signal x and r = E[dx] is the cross-correlation between the reference signal and the received signal. The MSE surface is a quadratic function of w and is minimized by setting its gradient with respect to w to zero. Yielding the well-known Wiener-Hopf solution-
=
Wopt = Rxxr ……….(4.32) The LMS algorithm is a stochastic gradient optimization algorithm that converges to this solution.

It is based on a traditional optimization technique called the Method of Steepest Descent. The weight vector is made to evolve in the direction of the negative gradient which points towards the minimum.

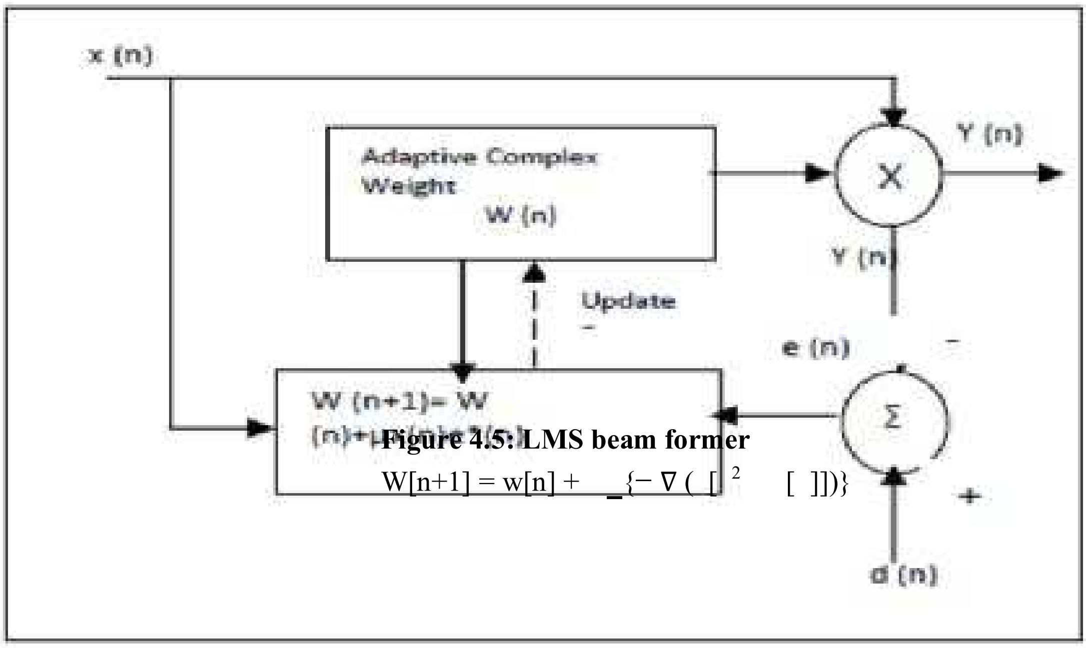

Figure 4.5: LMS beam former

W[n+1] = w[n] + _{− ∇ ([ ] 2 [ ]])}

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$$W[n+1] = w[n] + \frac{2}{2}\{2[\ ] - 2\ \ [\ ]\ [\ ]\} \quad ....(4.33)$$

Where Rxx and rare given by -

$Rrr(n) = x(n)x^H(n)$

$r(n) = d(n)x(n)$

This gives us a simple expression for weight updating
=

$W(n+1) = w(n)\ \mu x(n)[d(n)-x^H(n)w(n)]$ ..
...........(4.34)

The larger the value of μ, the faster the convergence but the lower the stability around the minimum value. On the other hand, the smaller the value of μ, the slower the convergence but the higher the stability around the optimum value. The speed of convergence depends also on the Eigen structure of Rxx.

The LMS algorithm is initiated with an arbitrary value w0) for the weight vector at n=0. The successive of the weight vector eventually leads to the minimum value of the mean squared error. LMS algorithm can be summarized in following equation

Output: $y(n) = w^H(n)$

Error: $e(n) = d(n)-y(n)$

Weight :$w(n+1) = w(n)+\mu x(n)e(n)$

Recursive Least Squares Algorithm -

As discussed above, the convergence of the LMS algorithm depends upon the eigenvalues of R. If R leads to a large spread, the algorithm converges slowly. This problem is solved here by replacing the step gradient size μ with a gain matrix R-1(n) at the nth iteration, producing the following weight equation

$$w(n) = w(n-1)-R^{-1}(n)x(n)\square*(w(n-1)).................. \quad (4.35)$$

Where R(n) is given by

$R(n) = \delta 0R(n-1)+x(n)x^H(n)$

Where, δ0 a real scalar smaller than but close to 1, is used for exponential weighting of past data. ε is the error signal. The RLS algorithm does not require any matrix inversion computations as the inverse correlation matrix is computed directly. It requires reference signal and correlation matrix information. It is almost ten times faster compared to LMS.

Spatial Reference Algorithm -

In Spatial reference algorithms (SR) the antenna weights are chosen based on knowledge of the array structure. These algorithms estimate the direction of arrival (DOA) of both the desired and interfering signals. The DOAs can be determined by applying different methods to the sampled data from the antenna array. The simplest way of extracting the DOAs is to use spatial Fourier transform on the signal vector. This method is limited by its resolution (size of antenna array) and has therefore limited usages. In cases where good resolution is necessary, so called high
resolution could be used. High-resolution methods are limited only by the modeling errors and noise and not by the size of the antenna array. Common high-resolution algorithms include:

Multiple Signal Classification (MUSIC) -

In 1979 Schmidt proposed the MUSIC algorithm. MUSIC algorithm is a high resolution Multiples Signal Classification technique based on exploiting the eigen structure of the input covariance matrix. It provides information about the number of signals, DOA of each signal, strengths and cross correlations between incident signals, noise power etc. Consider a N-element linear array that detects M signals impinging on it whose directions of arrival need to be known. From the previous discussion we know that the received signals at the output of the array have the following form:
$x(t)=A(\square)s(t)+n(t)$ ……………(4.36)

or in matrix notation it can be represented as

$x=As+n$

Where S is the signal vector, A is the array propagation vector and n is the noise vector with zero mean and variance σ^{2}.

The N N covariance matrix is given by:

$$Rx = E(XX^{*})=AE(SS^{*})A^{*}+E(nn^{*})$$
$$=AP^{*}A^{*}+\square^{2}I \quad ………(4.37)$$

Where $P = E(SS^{*})$ $AP^{*}A^{*} = Rx-\square^{2}I$

when the number of signals M is less than N then AP∗A is singular and has a rank less than N. The eigenvalues of Rx can be found by,

$$\square AP^{*}A^{*}\square=\square Rx-\square^{2}I\square=0 \quad ……………(4.38)$$

The eigenvectors of must satisfy

$Rx\ ei = \square^{2}Iei$

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(Rx-□^{2}I) ei =0

where ei is the eigenvector and i varies from 1 to N-M.

Let the noise eigenvector be defined as such that,

(s-□^{2}I)EN = 0

or

APA*EN = 0 ………….(4.39)

Based upon this approach, the pseudo spectrum P(θ) is given by –

P(□)=1/[A(□)*ENEN*A(□)] …………..(4.4
0)

When the pseudo spectrum P(θ) is plotted, peaks appear at the angles of arrival of the incident signals.

Drawback:

1. It required very precise and accurate calibration

2. If the impinging signals are highly correlated, it fails because the covariance matrix of the received signal becomes singular.

3. It is computationally intensive.

Estimation of Signal Parameters via Rotational Invariant Techniques –

ESPRIT is one of the most efficient and robust methods for DOA estimation. It uses two arrays in the sense that the second element of each pair is displaced by the same distance in the same direction relative to the first element. It is not required to have two separate arrays but can be realized using a single array by being able to select a subset of elements. Let the array signals received by the two arrays be denoted by x(t) and y(t)such that –

x(t)=As(t)+nx(t) …………..(4.4
1)

y(t)=As(t)+ny(t) …………..(4.4
2)

A is K×M matrix; where M is the number of steering vectors produced by N elements of the array nx(t) and ny(t) denotes the noise induced at the elements of the two arrays. Now, by using the available methods, the numbers of directional sources are estimated based on principles such as Akaike's information criterion (AIC) and Minimum description length (MDL). Two matrices Ux and Uy are

formed which denote the M eigenvectors corresponding to the largest eigenvalues of the two array correlation matrices Rxx and Ryy (Array correlation matrices).
The eigenvectors of the following 2M by 2M matrix are obtained and are denoted by –

$$V = \begin{bmatrix} U_x^H \\ U_y^H \end{bmatrix} \left[U_x U_y \right]$$

……….. (4.43)

Once the eigenvector V is obtained its eigenvalues λm ,m=1,…..M are computed
Now the DOA is given

$$\theta_m = \cos^{-1}\{\frac{Arg(\lambda_m)}{2\pi\Delta_0}\}, m = 1,......., M$$

………….. (4.44)

□m is the element separation in terms of wavelengths.

The DOA estimation technique is found to be more robust and faster when compared to MUSIC technique. The computation is also less complex comparatively. However, ESPRIT cannot handle correlated sources

ESPRIT has several advantage over MUSIC such as that is :

1. Is less computationally intensive.

2. Required much less storage

3. Does not involve an exhaustive search through all possible steering vectors to estimate the

DOA

4. Does not required the calibration of the array.

Blind Algorithms -
The third class of algorithms is termed blind algorithms (BA). These algorithms are based on prior knowledge of the signal properties of the transmitted signal. Depending on which statistical properties of the transmitted signal are exploited, we are able to apply different algorithms to determine the signal matrix from the received sample data.
Constant Modulus Algorithm The configuration of CMA adaptive beamforming is the same as that of the SMI system discussed above except that it requires no reference signal. It is a gradient-based algorithm that works on the theory that the existence of interference causes changes in the amplitude of the transmitted signal, which otherwise has a constant envelope (modulus). The weight updates are obtained by minimizing the positive mean cost function Many adaptive beamforming algorithms are based on minimizing the error between reference signal and array output. The reference signal is typically a

training sequence used to train the adaptive array or a desired signal based upon a priori knowledge of nature of the arriving signals. In the case where a reference signal is not available one must resort to an assortment of optimization techniques that are blind to exact content of the incoming signals. The Constant Modulus algorithm is blind algorithm where a reference signals is not available. It is a gradient-based algorithm that has a constant amplitude or modulus. Godard was the first to propose a family of constant modulus blind equalization algorithms.

SIMULATION RESULTS

The result of the proposed method that is Performance analysis of estimating algorithm MUSIC,LMS (MUSIC),MVDR,LMS (MVDR) using MATLAB shown in this section. For simulation of proposed method we have to use MATLAB R2012b (8.0.0.783) software. Basic configuration of our system where we have checked our proposed method is Manufacturer: Hewlett-Packard HP 4540s Processor: Intel(R) Core(TM) i3-3110M CPU @ 2.40 GHz 2.40 GHz with 4.00 GB (2.64 GB usable) RAM: System type: 32-bit Operating System.

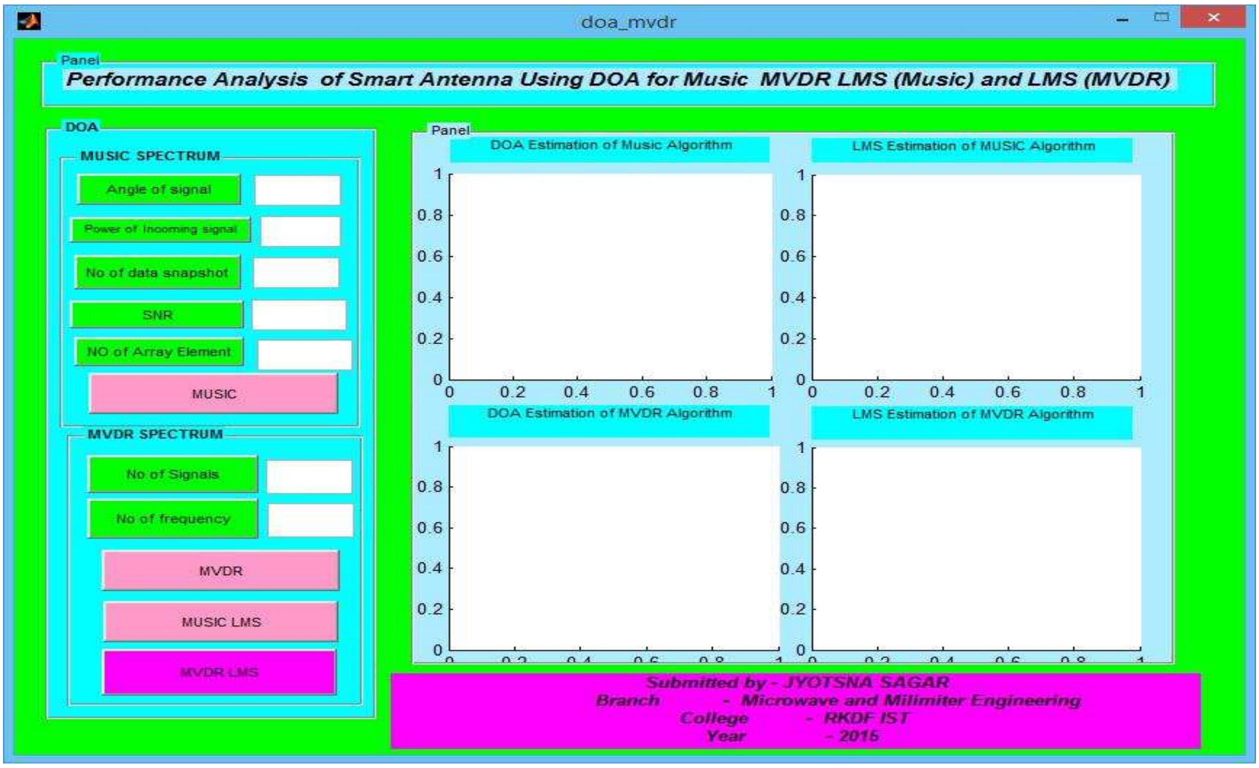

Figure 5.1: Display of DOA estimation for different algorithm MUSIC detection DOAs

In this figure we display the simulation result of smart antenna using direction of arrival (DOA) estimation and different adaptive beamforming (MUSIC, MVDR algorithm). Here we have compare the DOA and LMS for different algorithms and we perform the different calculation on different parameter like angle of signal, power of incoming signal, number of data snapshot, signal to noise ratio and number of array element.

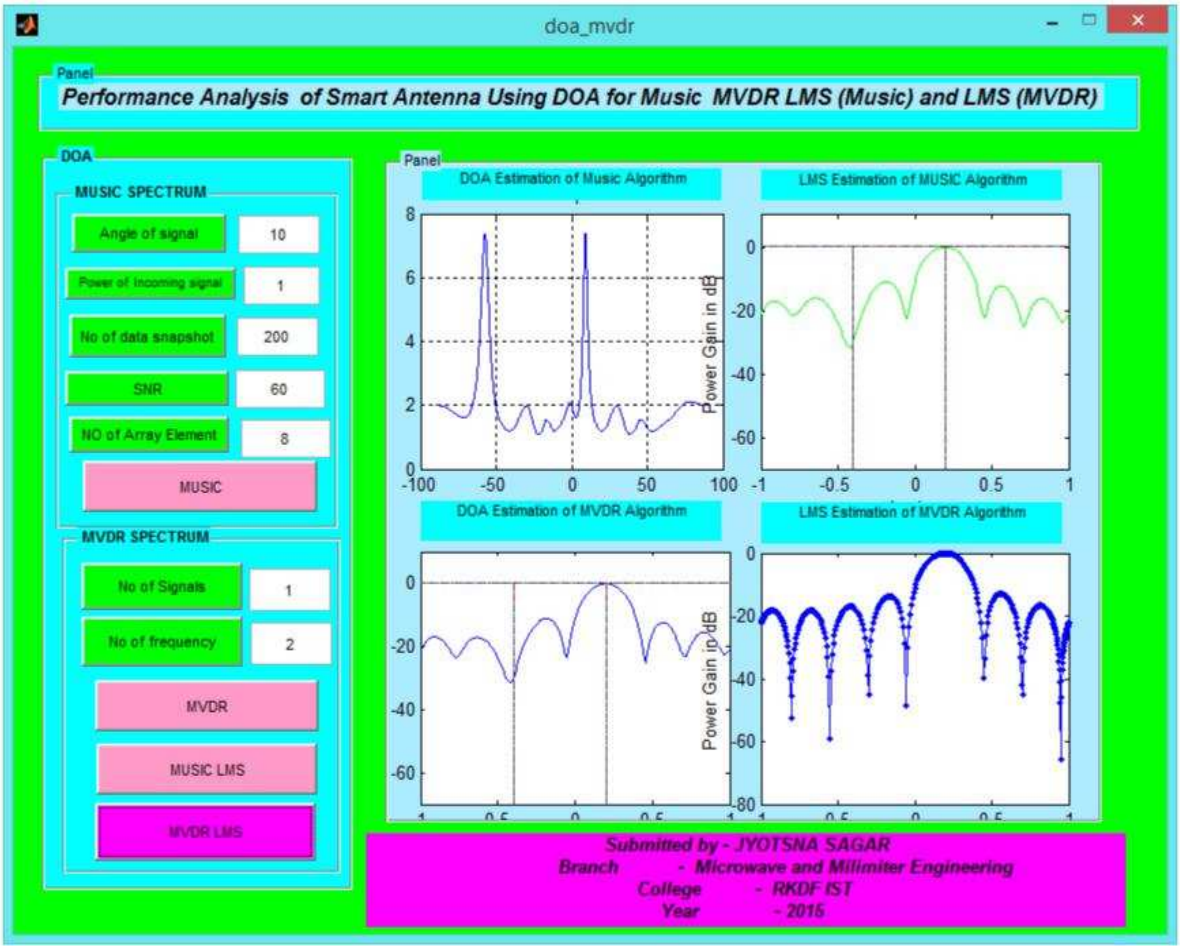

Figure 5.2: DOA for MUSIC Algorithms if N=8, with different angle & power

In this figure we display the simulation result of smart antenna using direction of arrival (DOA) estimation and MUSIC algorithm for different angle and different power with SNR=60 and N=8elements.

Table No, 1 – Power of gain compare for DOA and LMS

No.	Parameter	Values	Data Snap shot	DOA of Music	LMS of Music G(dB)
1	Angle of Signal	10	-100	2 dB	-20 dB
2	Power of Incoming	1	-50	7.689 dB	-30 dB
3	No. of data Snapshot	200	0	2 dB	0 dB
4	Signal to noise ratio	60	50	1 dB	-10 db
5	No. of Array Element	8	100	2 dB	-20 dB

We clearly see that the result for 8 array element analysis for both DOA music and LMS music both power gain in dB compare in this figure 5.2 and also compare the result in the table no 1.

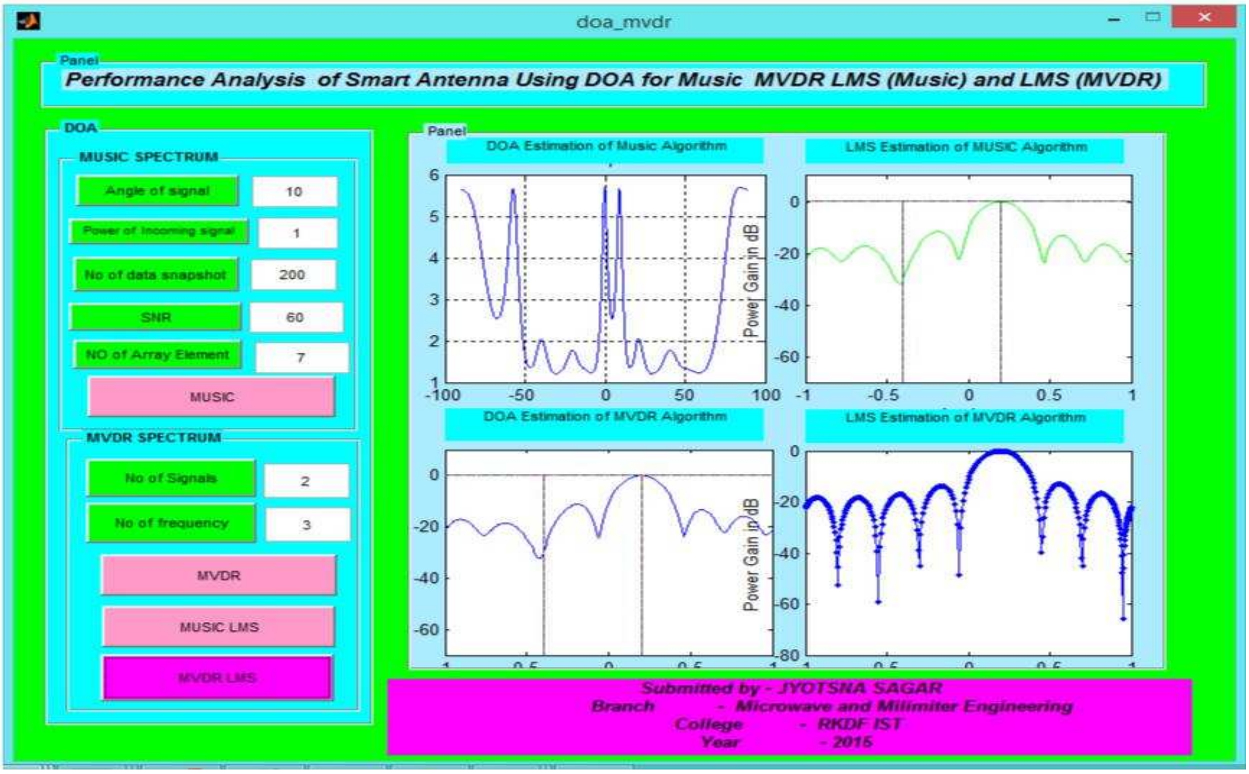

Figure 5.3: DOA for MVDR Algorithms if N=7, with same angle

Table no. 2: Shows Power of gain compare for DOA and LMS for 7 array element

No.	Parameter	Values	Data Snap shot	DOA of Music	LMS of Music G(dB)
1	Angle of Signal	10	-100	6 dB	-18 dB
2	Power of Incoming	1	-50	2dB	-25 dB
3	No. of data Snapshot	200	0	6 dB	0 dB
4	Signal to noise ratio	60	50	1.5 dB	-8 dB
5	No. of Array Element	7	100	6 dB	-20 dB

In this figure we display the simulation result of smart antenna using direction of arrival (DOA) estimation and MUSIC algorithm for different angle and different power with SNR=60 and N= 7 array element. If we compare the N in previous result N = 8 and this result N = 7 dynamic changes in DOA Music and very low changes in LMS Music. All the parameters are same but we change the array element and see the changes in in figure also. Figure 5.3 also the graphical changes in power gain in dB for both DOA of Music as well as LMS of Music. Here we also perform this for two more these are the DOA for MVDR and LMS for MVDR but in our proposed thesis mostly focus on the DOA and LMS of Music.

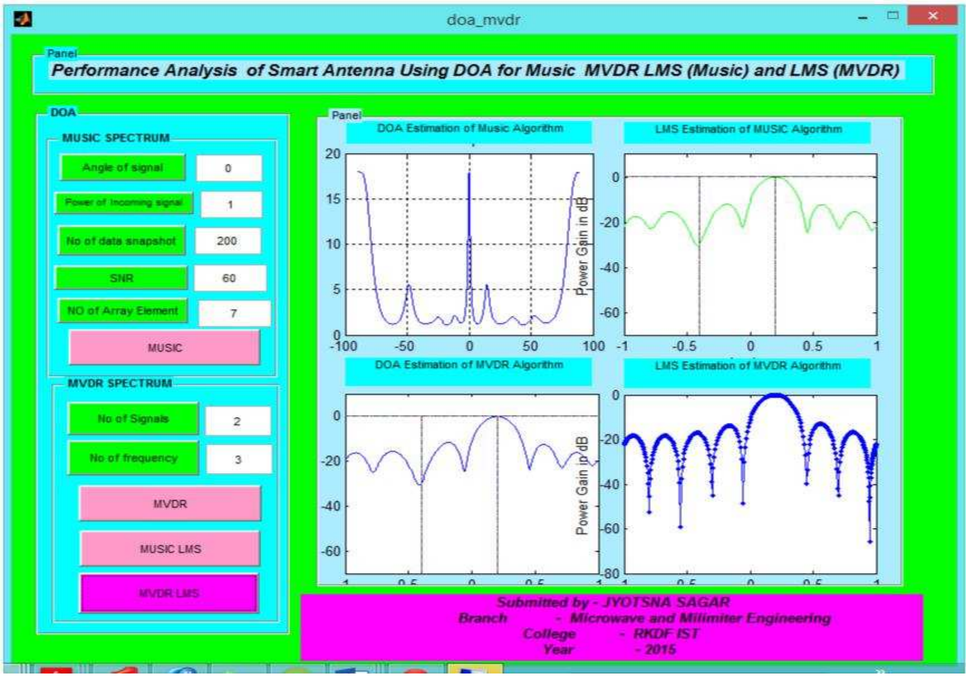

Figure 5.4: DOA for MUSIC Algorithms if N=7, with different angle & power

Table no. 3: Shows Power of gain compare for DOA and LMS for 7 array element

No.	Parameter	Values	Data Snap shot	DOA of Music	LMS of Music G(dB)
1	Angle of Signal	0	-100	20 dB	-22 dB
2	Power of Incoming	1	-50	5 dB	-28 dB
3	No. of data Snapshot	200	0	15 dB	0 dB
4	Signal to noise ratio	60	50	2 dB	-10 dB
5	No. of Array Element	7	100	15 dB	-23 dB

In this figure we display the simulation result of smart antenna using direction of arrival (DOA) estimation and MUSIC algorithm for 0* angle and different power with SNR=60 and N= 7 array element. If we compare the angle of signal in previous result = 10 and this result = 0 dynamic changes in DOA Music and again very low changes in LMS Music. All the parameters are same but we change the array element and see the changes in in figure also. Figure 5.4 also the graphical changes in power gain in dB for both DOA of Music as well as LMS of Music. Here we also perform this for two more these are the DOA for MVDR and LMS for MVDR but in our proposed thesis mostly focus on the DOA and LMS of Music.

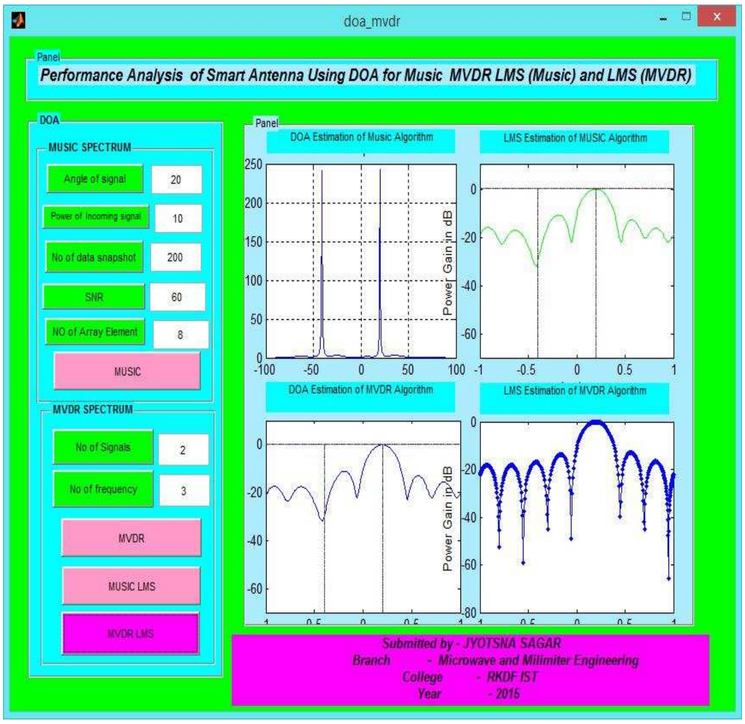

Table no. 3: Shows Power of gain compare for DOA and LMS for 7 array element

No.	Parameter	Values	Data Snap shot	DOA of Music	LMS of Music G(dB)
1	Angle of Signal	20	-100	0 dB	-22 dB
2	Power of Incoming	10	-50	250 dB	-28 dB
3	No. of data Snapshot	200	0	0 dB	0 dB
4	Signal to noise ratio	60	50	250 dB	-10 dB
5	No. of Array Element	8	100	0 dB	-23 dB

In this figure we display the simulation result of smart antenna using direction of arrival (DOA) estimation and MUSIC algorithm for angle and Power of incoming = 10 with SNR=60 and N= 8 array element. If we compare the angle of signal in previous result = 0 and this result = 20 and one more change that is power of incoming signal is 10. Then we see the dynamic changes in DOA Music and again very low changes in LMS Music. All the parameters are same but we change the array element and see the changes in in figure also. Figure 5.5 also the graphical changes in power gain in dB for both DOA of Music as well as LMS of Music. Here we also perform this for two more these are the DOA for MVDR and LMS for MVDR but in our proposed thesis mostly focus on the DOA and LMS of Music. In this performance analysis presented an analysis of smart antennas. After understanding the non-uniform model, several DOAs and adaptive beamforming algorithms were theoretically analyzed; such as MUSIC algorithms for direction of arrival, and the LMS, generalized normalized algorithms for the adaptive beamforming. These algorithms were implemented by using MATLAB. The first part of the experiments analyzes the DOA algorithms (MVDR and MUSIC) by considering two different scenarios. Both algorithms yield a good results for detecting DOAs; however, the MUSIC algorithm yields more accurately. It was shown that the performance of the adaptive beamforming algorithms is directly proportional to the step-size and forgetting factor. In the final result say that Sharper peaks in the MUSIC angular spectrum indicate locations of desired users. Peaks of LMS are formed in the same desired direction and deep null in the direction of the undesired interference. LMS Music algorithm has good response towards desired direction and has better capability to place null towards interferer. Peaks of LMS are formed in the same desired direction and deep null in the direction of the undesired interference. Results obtained verify the improved resolution when the number of elements and spacing between elements are more. This analysis is useful in implementation of direction of arrival based smart antenna system. This proposed results provides references to studies where array beam-forming and DOA schemes are cosidered for mobile communications systems. This type of analysis can be use full for implementation of smart antenna system based on DOA.

CONCLUSION

The Smart Antenna systems are the antennas with intelligence and the radiation pattern can be varied without being mechanically changed. With appropriate adaptive algorithms such as MUSIC and MVDR the beam forming can be obtained. DOAs and adaptive beamforming algorithms were theoretically analyzed; such as the Capon and MUSIC algorithms for direction of arrival, and the LMS implemented and also comparing the result of the both technique. As the system uses a DSP processor the signals can be processed digitally and the performance with a high data rate transmission and good reduction of mutual signal interference. The narrow beams get rid of interference, allowing many users to be connected with in the same cell at the same time using the same frequencies and can adapt the frequency allocation to where the most users are located. With adaptive beam forming, spectral efficiency of the cell could be multiplied at least ten times. Smart antennas effectively reduce the power consumption which in turn avoids RF pollution, minimize health hazard and save scarce resource (diesel & foreign exchange). Indeed it has been argued that performance requirement of a future cellular communication system cannot be made without the use of smart antennas.

FUTURE SCOPE

Lot of work is in progress on smart antenna. Once nanotechnology antenna arrays are developed, it will be possible to incorporate smart antenna at handheld system too. As a result the performance of cellular systems will be enhanced manifold. Neural network like Recurrent Neural.

Network (RNN) with reduced structural Complexity can be incorporated for adaptive beamforming. Also in future work includes new implementations of these beamforming and DOA approaches based on machine learning theory. In particular, it is interesting to change the well-known Least Squares (LS) approach used in all the algorithms presented in this work by other criteria. Indeed, LS is maximum Likely hood only in the presence of Gaussian statistics, but it will lead to suboptimal solutions when the statistics is non Gaussian. Also, in communications and related areas, only small data samples are often available for training the systems, which can lead to numerically ill-posed problems and, in general, produce over fitted solutions, a situation that produces solutions biased from the optimal ones.

In future work one more very important step for the next level development of this work that is Adaptive Neuro-Fuzzy Inference System (ANFIS) may be considered better robustness to the beamforming algorithms. We also implemented this algorithm in machine level language as well as also implemented in VHDL field programmable gate array (FPGA). FPGA is implementation is very helpful for chip level or IC level designing and testing in hardware.

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