On the LMS Algorithm Performance for Interference Elimination in Smart Antennas Array

WilanderTestonePereira da Silva, João Viana da Fonseca Neto

Abstract—The efficient use of limited radio frequency spectrum is possible due to the smart antenna arrays. These antennas arrays incorporate adaptive algorithms, such as: Least Mean Square (LMS) algorithm, which finds the spatial temporal filter gains or weights according to the signal environment behavior. In terms of the mean error and mean squared error convergences of the LMS algorithm, the performance evaluation of the algorithm is oriented by its convergence properties and the improvements in the mobile communication systems. In this paper is presented the LMS algorithm to solve the beam forming problem and antenna array concepts, as well as, it is presented general performance analysis, in terms of the LMS beam former to eliminate interference in antennas array. The potentialities of adaptive design are verified in models of smart linear antenna arrays. These antenna arrays models are connected to the beam former model. The integration of these models allows to design the adaptive beam former. The results obtained from simulations of the models shows that the LMS algorithm is a good alternative for smart antenna design in mobile communication environment, due to the directivity improvement promoted in the antenna array.

Index Terms—Smart Antenna Array, Adaptive Filter, LMS Algorithm, Algorithm Convergence, Beam forming, Interference Elimination, Mobile Communication, Wireless communications.

I. INTRODUCTION

The research and development of antennas arrays (sets) to receive and transmit signals is now considered a feature that brings a number of benefits, control of interference and noises signals that are present in the middle of communications. Due to offer a higher quality, overcomes the limitations related to bandwidth and interference with a single element [4]. Currently, the arrays together with adaptive algorithms of the same class are subject to a comprehensive study, besides the advantage of high-performance computing power and processing signal so as to achieve more precise goals. The basic idea is to filter the adaptive processing signals through sensors, thus creating a main lobe on direction of interest; that is, updating the weights become adaptive, that is, in nutshell the general concept of smart antenna. The aim of this paper is to present a performance analysis of standard LMS algorithm that is oriented to smart antenna design. The LMS algorithm performance is evaluated in terms of the convergence speed and optimal gain via the mean error and mean square error. The impact in elimination of the interference signal are evaluated in terms of the elements quantity in the antenna array. Theoretical concepts of the antenna array and beam forming technic, as well as, LMS method concepts, formulation and the statements that are applied in the convergence analysis.

Revised Version Manuscript Received on December 12, 2015.

WilanderTestone P. da Silva, Department of Electrical Engineering, Federal University of Maranhão (UFMA), São Luís, Brazil.

Dr. JoãoViana da F. Neto, Professor of Department of Electrical Engineering, Federal University of Maranhão (UFMA), São Luís, Brazil.

Smart antennas are systems (sets) of antennas that react to environmental changes dynamically, in order to provide a higher quality signal and generate a better use of frequency bands in wireless communications. Figure 1 represents the adaptation of the weights respective for the desired signal using the LMS algorithm. During the input signal acquisition, occurs adaptation of the weights that are updated via LMS algorithm. In this Figure, in our context, it represents a linear smart antenna array; the weights are adjusted by LMS algorithm.

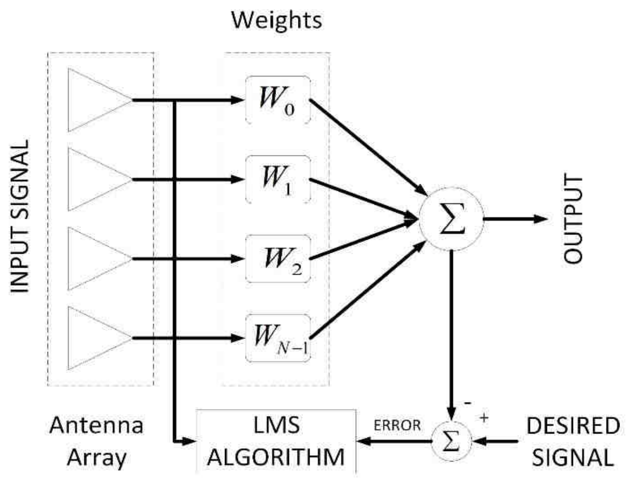


Figure 1: Smart antenna array system

The weight vectors of LMS algorithm are adapted to eliminate the antenna interferences. The mean square error (MSE) is directly linked to weight vector, where it is fundamental in the process of convergence of the algorithm LMS.From the factor of array, it is built the irradiation diagrams, which in turn in this paper, where is that the greater number of elements in array, more directive the radiation lobes will stay to capture desired signal. To achieve this goal, based on adaptive algorithms is used mean least squares method (LMS). It is through the minimum mean square error, the arrays (sets) are able to adapt the disposition [2], [10]. The use of more efficient algorithms, combined with significant improvements of hardware, has made the use of Smart Antennas to have increased considerably in the last decade.Wireless communications have become the most promising area for application of antennas due to the quality of research being done, and the demand for this service by the population.The paper is organizedin three parts:a) Theory that supports the design method and the analysis of its performance in our application context. 2) Computational experiments and analysis of its results. 3) Conclude remarks.Consequently, the parts are distributed in sections. Smart antennas array and beamforming concepts and technology are presented in Section II. The main LMS algorithmformulation,based on the steepest descent method and Wiener filter, is presented in Section III. The

*Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.*

computational experiments to evaluate the convergence of LMS and interference elimination in the smart antennas design is presented in Section IV. The conclusion andcomments, about the performance of smart antenna design via standard LMS algorithm and its impact in the antenna array, are presented in Section V.

II. SMART ANTENNAS ARRAY AND BEAMFORMING

In terms of smart antenna array design, beam forming and antenna array concepts and technics are presented in this section. Before addressing these topics, certain concepts are highlighted, such as: a) Array of uniform antennas is a set of elements (antennas) of identical characteristics that are separated by a fixed distance. b) Array Factor is defined as a parameter of the effect that results from the combination of elements of radiation.

A. Smart Antennas Array Modeling

The mathematical representation of antenna array encompasses the array factor, output and input signals of Adaptive filtering system. If every element of the array is isotropic punctual source, then the radiation pattern of the antenna set will depend on the geometry of the array and changes of the amplitude and phase of the incident wave. The array factor of antennas set that are configure in a linear array the geometry is given by

$$AF(\theta) = \sum_{n=0}^{N-1} w_n e^{jnkd\cos(\theta)}\ , \qquad (1)$$

Where w_n represents an element of the complex array of weights, N is number of antennas of the array, $\theta$is the angle of incidence of an electromagnetic wave of the plan of array shaft $k$that is equal to $\left(\frac{2\pi}{\lambda}\right)$ and $\lambda$is the wavelength. When the array factor is adaptable, the array is known as array of smart or adaptive antennas. According [10], the mathematical relation of $s(t)e^{jw_c t}$ signal from a distant electromagnetic wave source, when reaches the antenna 1 on the n-thinstant of time $t$and$s(t)$has a slowly-varying envelope, such that$s(t) \approx s(t + \Delta t)$,assumes the following form:

$$\mathrm{s_{nc}}(t)(t) = s(t)e^{jw_c t}e^{j\frac{2\pi n}{\lambda}d\cos\theta}, n = 0,1,2,\dots, M-1\ , (2)$$

Where $\omega_c$is the carrier frequency, $s(t)$ it is baseband signal, θ is the angle at which the incident signal arrives and d is the spacing between elements. The incident signals are processed in the baseband. Taking into account this fact, the antenna receives signals without the carrier component that is given by

$$\mathrm{s_n(t) = s(t)e^{j\frac{2\pi n}{\lambda}d\cos\theta}}, \qquad \mathrm{n = 0,1,2,\dots, M-1},$$

According to [4] [6] and [10], thenoisy snapshot of the baseband signal at the M antennas is given by

$$x = s(t)\begin{bmatrix} 1 \\ e^{j2\pi d\cos\theta/\lambda} \\ e^{j4\pi d\cos\theta/\lambda} \\ \vdots \\ e^{(M-1)\cos\theta/\lambda} \end{bmatrix} + \begin{bmatrix} v_0(t) \\ v_1(t) \\ v_2(t) \\ \vdots \\ v_{M-1}(t) \end{bmatrix}$$

Where $v_n(t)$ is the noise in the nth element of the array. In terms of x vector components, the incident signal at n-*th*antenna of the array is given by

$$x_n(t) = s(t)e^{j2\pi d\cos\theta/\lambda} + v_n(t) \quad n = 0,1,2,\dots, M-1\ , (3)$$

In Equation (3), the interference signal is added to the signal of interest (desired). In Figure 2 is shown the linear array of uniformly spaced, where S_1 is the surround signal and S_M refers to the array in the directions of the incident signals θ_1, θ_2andθ_M.

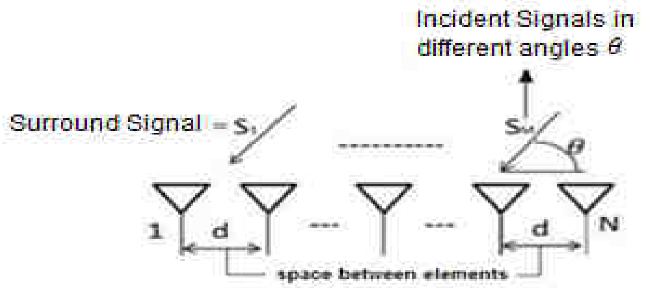


Figure 2: Array geometry (set) of Antennas

Source: AdaptedfromRao,M;Sarma,N;“AdaptiveBeamformingAlgorithmsforSmartAntennaSystems”.2014.

The total output of the array is given by

$$Y(t) = \boldsymbol{W^H}(t)\boldsymbol{X}(t)\ , \qquad (4)$$

where$\boldsymbol{W^H}(t)$is conjugate and transpose of the weight the matrix and $\boldsymbol{X}(t)$ the input vector.

B. Beamformer Models

The beamforming is a signal processing technique used in sensor arrays for directional signal transmission or reception. In this section is described the mathematical representation of several beamformers. In Figure 3 is presented a beamformer that is commonly used for processing of narrowband signals; the same is shown in the wave field in the space, which is given by

$$y(n) = \sum_{l=1}^{L} w_l^* x_l(n), \qquad (5)$$

where * refers to the complex conjugate value of the weights.

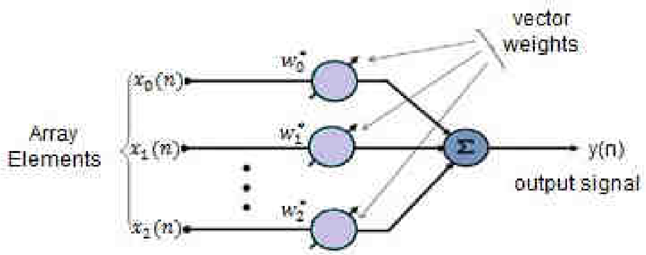


Figure 3: Linear combinationof the output of the sensors"Narrowband".

In Figure 4 is presented a beamformer for broadband signals, samples of the wave field propagating in time and space. It is generally used when there is interest in signals with high frequency [7].

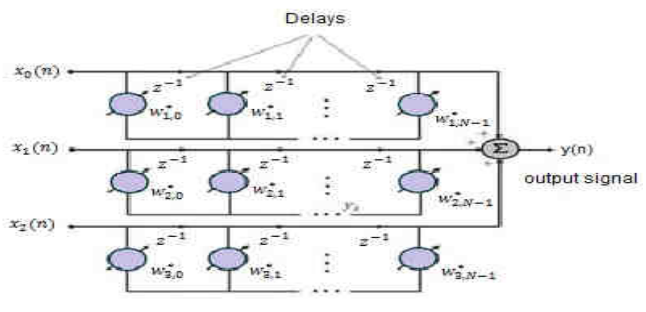


Figure 4: Linear combination of the output of the sensors "Broadband".

Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.

It is represented by the Equation:

$$\sum_{l=1}^{L}\sum_{p=0}^{K-1} w_{l,p}^{*}x_l(n-p)\ , \qquad (6)$$

where$K-1$is the number of delays in each of the sensors. Note that the beamformer is the representation of multiple entries with a single output. The response of a beamformer is basically the amplitude and phase of a complex waveform, depending on the location and frequency, respectively. Often, the focus is on the estimation of direction of arrival(DOA) (θ) and frequency (ω).

C. Performance of an Array in a Beamformer

An antenna array together with a beamformer (signal processor), provide the capability to perform several tasks at once, and improve the performance of a particular process of receiving, at the same time achieves also suppress unwanted interferences and also maintains a signal of interest. As for the antenna array there are some factors to consider, among them are: resolution capability, the angular coverage, the number of elements of the array and the level of side lobes [8]. Although the number of array elements must be more substantial to avoid degradation of the desired angular coverage. Also, it is known that the existence of certain compensation between the parameters mentioned above goes beyond a directly proportional relationship between improvement and cost. To achieve the goal of improvement in one direction, reference [8] presents a compensation principle by selection of the complex value of the weight, in the desired angular coverage and interference suppression. The authors show the improvement of desired the performance and noise it rejects while preserving the signal of interest.

III. LMS STANDARD ALGORITHM

In this section is presented the development of the LMS algorithm, which is based on the same principles of the steepest descent method. As well as, it is presented the procedure to evaluate the convergence of LMS method, when this is applied in the of smart antenna array.

A. LMS Formulation

This method is applied to deduce the optimal equation of the Wiener that is given by $w = R^{-1}d$. If the statistics of the measured signals are known, the steepest descent algorithm is a good alternative for adaptive filtering. In most applications of real world, these statistics are unknown. To overcome the problem of the unknown statistics, the LMS implements the strategy of estimating the parameter during the acquisition of input signal. The problem is to minimize the error signal$E\{|e(n)|^2\}$, applying adaptive algorithms and stochastic gradient methods. According to [10], the estimative of the autocorrelation matrix **R** and cross-correlation vector **P** are given by

$$\hat{R}(n) = x(n)x^H(n) \qquad (7)$$

$$\hat{P}(n) = x(n)d^*(n) \qquad (8)$$

The LMS algorithm is based on a stochastic approach gradient vector $\nabla\hat{J}(n)$, obtained from instantaneous data available in the system. In this way, the gradient vector is given by

$$\nabla\hat{J}(n) = \frac{\partial\hat{J}(n)}{\partial w^*(n)}\ , \qquad (9)$$

where $\hat{J}(n) = |e(n)|^2$. Developing the Equation (9) and using the chain rule in the derivative, the gradient vector for a complex signal is given by

$$\nabla\hat{J}(n) = e(n)\frac{\partial e^*(n)}{\partial w^*(n)} + e^*(n)\frac{\partial e(n)}{\partial w^*(n)} \qquad (10)$$

Now considering: $e(n) = d(n) - \mathbf{w}^H(n)\mathbf{x}(n)$,the partial derivatives present in Equation (10) are given by

$$\frac{\partial e^*(n)}{\partial w^*(n)} = \frac{\partial}{\partial w^*(n)}[d(n) - \mathbf{w}^H(n)\mathbf{x}(n)] = 0 \qquad (11)$$

and

$$\frac{\partial e(n)}{\partial w^*(n)} = \frac{\partial}{\partial w^*(n)}[d(n) - \mathbf{w}^H(n)\mathbf{x}(n)] = -\mathbf{x}(n) \qquad (12)$$

Therefore, substituting (11) and (12) to (10), the estimate of the gradient vector is given by

$$\nabla\hat{J}(n) = -e^*(n)\mathbf{x}(n)\,. \qquad (13)$$

Finally, from the equation of steepest descent, $w(n+1) = w(n) - \mu\nabla\hat{J}(n)$and Equation (13), the equation of the LMS algorithm to update the coefficients, considering the complex shape can be seen as [1], [3], [9], [11], is given by

$$w(n+1) = w(n) + \mu\mathbf{x}(n)e^*(n) \qquad (14)$$

B. Convergence Analysis

Taking into account parameter variations of η of Equation (14), the convergence analysis is performed in terms of mean error and the mean square error. The time evolution of weights process characterizes the mean error that is represented by

$$E[\widehat{w}(n)] \to w_0 \quad n\to\infty\,, \qquad (15)$$

where$\widehat{w}(n)$ is the estimated weight and w_0 is the optimal weight of Wiener filter. The following relation evaluates the convergence performance of LMS algorithm via the mean square error:

$$E[e(n)^2] \to k \;\; n\to\infty\,, \qquad (16)$$

where the error $e(n)$ is given by $e(n) = d(n) - \mathbf{w}^H(n)\mathbf{x}(n)$, is error and $k$is a constant value.

IV. COMPUTATIONAL EXPERIMENTS

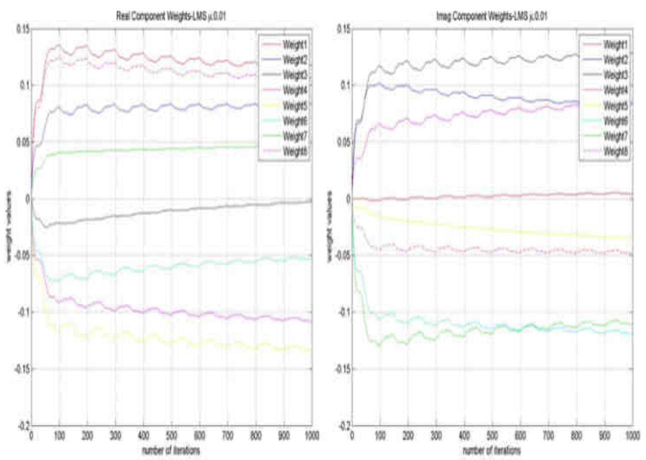
In this section is presented the computational experiments. The behavior of the interference elimination process, antenna array gains or LMS-weight vector and mean square error of the adaptive filter are evaluated in terms of relations (15) and (16), as well as, in terms of radiation diagrams. The antenna incident signal and environment are modeled according to Equations (1-5) and beamformer main equation is given by Eq.(14). The LMS beamformer algorithm is coded in MATLAB scripts. Applying the knowledge previously described in the last sections and based on the standard LMS algorithm, it is presented the behavior of the design method to improve antenna array performance. Our goal is to achieve a given SNR (Signal Noise Rate) with small value, this parameter is fundamental to the algorithm, in terms of its adaption to the scenario. The problems is due to the

Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.

proximity of the signal that difficulties the noise attenuation and the improvement of quality of the incident signal in antenna. To solve this problem is proposed to increase the number of elements in the array, because how much more elements in the array, the lobes become narrower for better capture of the desired signal. It is intended to prevent unwanted signal capture and get better directivity. For our case we work with large angles of incidence, they make easy the separation of desired signals and interference. The results obtained in this smart antenna design. In all of them, they were regarded with two signals, one desired and the other, interference. For the proposed design is considered a uniform linear antenna array, in addition the weight vectors initialized in zero and experiments numbers equal to 120. In terms of noise, we have an input SNR value of 10dB, this means that we have 0.03 contamination, that means that we taking into account a minimum amount of noise.

A. Elimination of Interferences

The interference elimination process of antenna shappens in two ways: deterministic and automatic. The deterministic form consists of functions or their weights are formed by functions series, such as: series of Blackman, Hamming, Binomial, etc, this type of interference elimination will not be addressed in this paper. In the automatic way, the weights involve algorithms beamforming and adaptive, so, they adapts to new conditions of the entries, ie, when the scenario in which the system is housed to changes behavior so that the same conditions are matched. It can be said that systems with this feature are considered to have a high performance, since they aim to use: less power, higher data rate, lower vulnerability to interference, among others. The LMS algorithm, as is known, converges to the Wiener solution in average with step size μ, which controls the rate of convergence and stability of the algorithm. The convergence of the LMS algorithm to the problem in question, shows that is directly associated with the step of adaptation. In Figure 5 shows the performance of the optimal weight vector. It is observed that the smaller the step size, the slower the convergence. However, the step size must satisfy the condition$0 < \mu < \frac{2}{\lambda_{max}}$ for weight vector, in order to have a stable system, where λ_{max} it is the largest eigenvalue of the autocorrelation matrix. In this case, when $\mu_1 = 0.01$, the system needs at least 100 iterations in order to get the optimal weight vector, when $\mu_3 = 0.0001$, over 1000 iterations are required.



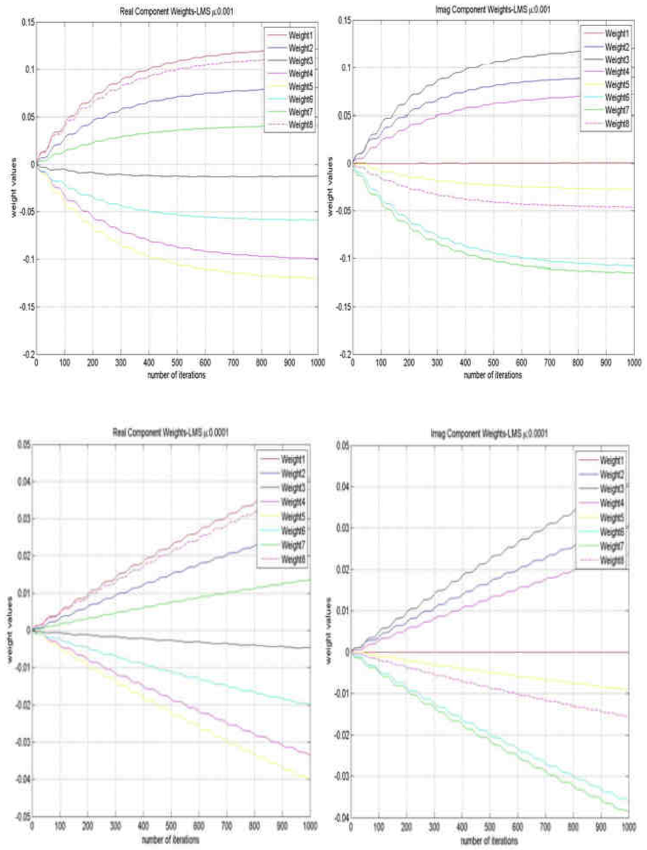


Figure 5: weight vector for LMS algorithm in an array M=8 antennas with $\mu_1 = 0.01$, $\mu_2 = 0.001$ and $\mu_3 = 0.0001$, SNR input=10dB; d/λ=0.5 and θ=π/6.

Three different parameters μ were considered in order to study the rate of convergence of a set of antennas. It is presented weights in the real part and imaginary. According to relation (15) is shown in Figure 5 that the LMS has its convergence by the mean error. In Figure 6 shows the mean square error for the step size $\mu_1 = 0.01$. The mean square error is directly related to weight vector; when the system converges to the values of the optimal weight, the mean square error converges asymptotically to the minimum error, as can be seen in Figure 6. In this way, The MSE behavior attends relation (16). From figure 6, it is observed that the MSE decreases at each iteration and converges after 100 iterations.

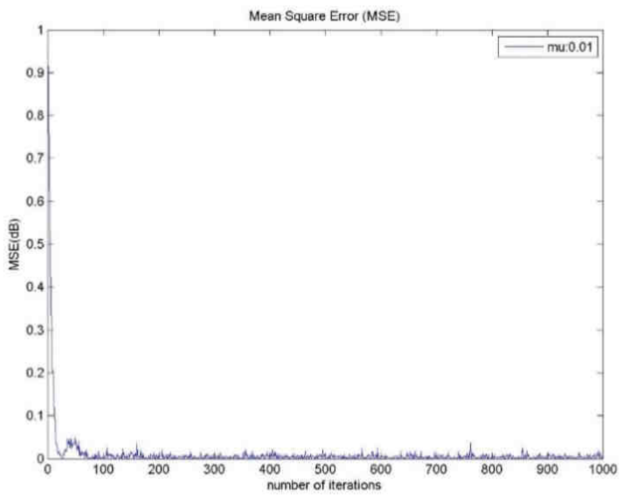


Figure 6: MSE for LMS algorithm in an array M=8 antennas with $\mu_1 = 0.01$, SNR input=10dB; d/λ=0.5 and θ=π/6.

Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.

Thus, we can conclude smaller that step size, slower will be convergence. If we increase the step size, we have a fast convergence of the LMS algorithm. Moreover, if increase the number of elements in the antenna array, will also have a slow or fast convergence of the algorithm, dependent on the adaptation step.

B. Radiation Lobes

Beamformer is known is a lobe former; its furnish information to plot irradiation diagrams. In the conception of electronic circuits, the beamformer can seen as network, this network can simultaneously feed the various elements of an antenna array with different signals that produces lobes in different directions. Basically, its function isto receive the signals from the linear array and deliver them to the signal processing network for proper management, ie, to obtain the signals for the desired communication. The beamforming algorithm is alternative technique to obtain communication signal.

C. SNR Control with Linear Arrays and Great Difference in Incidence Angles

There is a relationship between the number of antenna elements in the array and the proximity between the signals that affects to achieve the desired SNR, this fact can verified through simulations of the adaptive antenna arrays. Initially, one should realize that the closer the interference signal and the desired signal, harder it is for the system to mitigate the unwanted signal and increase the desired signal. One way to can be overcome this difficulty, it is increasing the number *M* of antenna elements in the array. Consequently, the widths of the main lobe are reduced and are created more secondary lobes with very low levels. Our proposal will be dealt major differences that are good for the system, because in this manner, making easy, the separation of desired signals and interference. It is important to mention that the irradiation diagrams is more directive as the main lobe becomes narrower. Thus, it is possible to get rid of unwanted interference signals at the antenna input. There is a great relationship between the discrepancy of angle of arrival (AOA) of the desired signal and the interfering signal with the number of antenna elements in the array. The radiation diagrams for *M*=8, 16 and 24 elements array are presented in Figures 7, 8 and 9. These arrays has d spacing 0.25λ, receives a arriving signal at an angle $\theta_1 = 60°$, an interference at the angle $\theta_2 = 45°$ and input signal. $s_1 = A_1 \cos(2\pi R T_s)$, where A_1 is amplitude of signal, R is iteration n and T_s is sample time. For this, one can show the irradiation diagrams with a difference of 15° between the AOA desired and unwanted signal, and this gap considered as optimum angle to eliminate the interference. Therefore, the desired signal has an incidence angle of 30° and interference signal at 45°. Observing the irradiation diagrams for *M* = 8, 16 and 24 elements. It is verified the narrowing of the main lobe and directivity to 30° and null 45°. Considering a difference of 15° between the signals desired and interferer is shown in Figures 7, 8 and 9 the irradiation diagrams, for array with various sizes of *M*. In the irradiation diagram of Figure 7for an array with *M* = 8 elements, the angle of incidence of the desired signal in 30° and the angle of incidence of the interference signal at 45°. It is observed that a relative spacing does not allows a catchment of interference, the large differences are good for the system, since making easy the separation of interferences to obtain the desired signals.

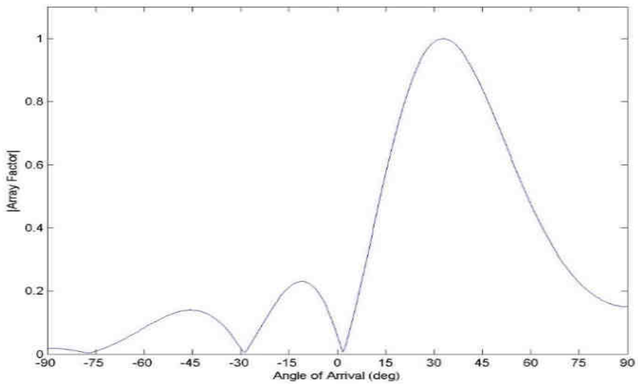


Figure 7 – irradiation diagram M=8 elements with incidence the desired signal in 30° and interference at 45°.

Similarly, there is the irradiation diagram of an array with M = 16 elements with an incidence angle of the desired signal in 30° and the incidence angle interference signal at 45°, as can be seen in Figure 8.Note in this figure that the increase of *M* implies at higher directivity, but also occur more side lobes. The presence of more lobes implies greater ability to capture unwanted signals. However, this fact represents an important function of smart array, because is possible to suppress interference.

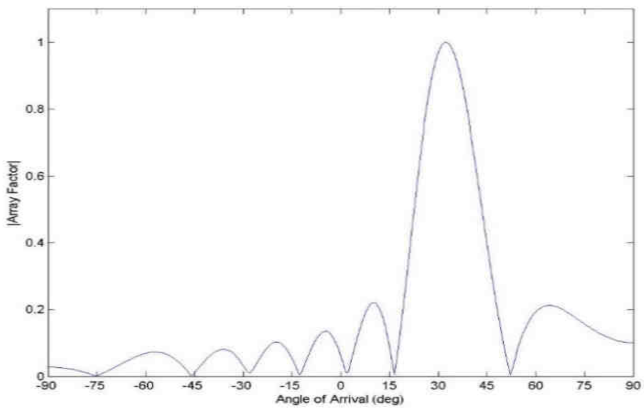


Figure 8 – irradiation diagram M=16 elements with incidence the desired signal in 30° and interference at 45°.

Finally in Figure 9 is shown the irradiation diagram with *M* = 24 elements. Note that the central beam is quite directive and the interference signal in 45° is more attenuated, always remembering that the technique of elimination of unwanted signals are accomplished by limiting the beam angle of incidence of the signal.

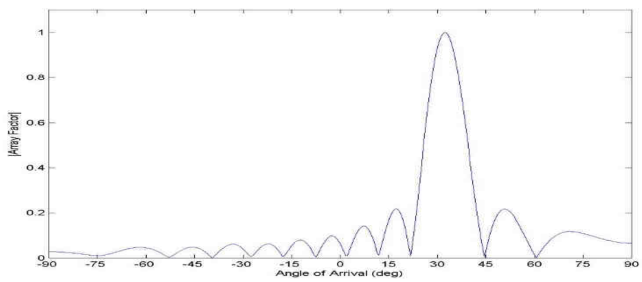


Figure 9 – irradiation diagram M=24 elements with incidence the desired signal in 30° and interference at 45°.

Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.

As general analysis, it can be checked that the amount of side lobes and the interference reduction, in this case, occurs as the number of elements in the antenna array increases. The AOA difference, of the signals reaching the array remains large in this case. Large differences are good for the system, since in this way make it easy the separation of the desired signals and interference signals. It also turns, narrowing relative signal of the antenna and a higher directivity compared to 8 and 24 elements.

V. Conclusion and Comments

In this paper was presented the performance convergence of Least Mean Squares (LMS) algorithm as the core of smart antenna array for interference elimination. I this work was verified that through the |LMS weights was possible determine the mean square error (MSE), which decreases at each iteration until you find point of convergence and through the same weights. We can calculate normalized matrix factor to plot the irradiation diagrams. Since these diagrams allows to determine the best directivity, the antenna will receive and transmit energy radiated in a particular direction. Furthermore, it is possible to get rid of unwanted interference signals at the antenna input. In this paper, we analyzed beamforming techniques with smart antennas and elimination of interference signals. Beamforming algorithms have been performed for various situations for variation of the numbers elements arrays. The plot of irradiation diagrams that we analyzed to verify the displacement of the main lobe, when there is distant interference of the desired signal. This occurs because the system tries to ensure that the relation SNR is the lowest possible value. With the generation of the radiation lobe, we concluded that a greater number of elements in the array improves representation graphics for radiation lobes; indicates that we are generating beamforming in the desired direction. The filter quality is remarkably high when the element of the array increases. The study and analysis of the various adaptive algorithms has allowed raising the basis for future work in the area of digital processingsignals. These have become the mainstay in the development of more robust algorithms; enabling an adaptive combination between algorithms.

ACKNOWLEDGEMENT

The authors are indebted to the Federal University of Maranhão(UFMA), the Graduate Program in Electrical Engineering(PPGEE), the Embedded Systems and Intelligent Control Laboratory (LABSECI),the Coordination for the Improvement of Higher Education Personnel(CAPES), and the National Council for Scientific and Technological Development (CNPq) for the development infrastructure and financial support.

References

[1] B. Farhang-Boroujeny, *Adaptive Filters: Theory and Applications*, 1st ed. Baffins Lane Chichester, U.K.: John Wiley& Sons, 1999.
[2] Diniz, P. *Adaptive Filtering Algorithms and Practical Implementation*, 4rd ed. New York, EE.UU: Springer, 2013.
[3] P. S. R. Diniz, *Adaptive Filtering: Algorithms and Practical Implementation*, 3rd ed. Boston, MA: Springer, 2008.
[4] Orozco, W., Nakano, M., and Pérez, H, (2013). *Adaptive Cascade Hybrid Configurations for Array Beamforming*,(I.J. Communications, Ed.) Retrieved 23 of March, 2015, from International Journal of Communications:http://www.naun.org/main/NAUN/communications/c012006-109.pdf
[5] Balanis, C. *Introduction to Smart Antennas*, 1rd ed. Arizona: Morgan and Playpool publishers, 2007.
[6] Rao, A., and Sarma, N. (2014). *Adaptive Beamforming Algorithms for Smart Antenna Systems. Índia. Retrieved 19 of January, 2015, from* http://www.wseas.org/multimedia/journals/communications/2014/b105704-189.pdf
[7] Van Veen, B., and Buckley, K., *"The Digital Signal Processing Handbook"*.1rd ed., Wisconsin of University, VillanovaUniversity, 1999.
[8] Mozingo, R., Haupt, R., and Miller, T. *"Introduction to AdaptiveArrays"*. 2rd ed., SciTech PublishingInc., 2011.
[9] A. D. Poularikas and Z. M. Ramadan, *Adaptive Filtering Primer with MATLAB*, 1st ed. Boca Raton, FL: Taylor & Francis Group, 2006.
[10] A. H. Sayed, *Adaptive Filters*, 1st ed. Hoboken, NJ: John Wiley & Sons, 2008.
[11] S. Haykin, *Adaptive Filter Theory*, 4th ed. Upper Saddle River, 2002.

Mr. Wilander Testone P. da Silva, Federal Universityof Maranhão, São Luís, Brazil.Graduated in Mathematics by the Federal Institute of Education, Science and Technology of Maranhão, IFMA, (2012). He has experience in the area of Mathematics, with emphasis on Applied Mathematics, acting on the following theme: Operational Research and Modeling. He is currently a Master's degree student in Electrical Engineering with emphasis on Automation and Control, in the Federal University of Maranhão, UFMA, acting on the following topics: Adaptive Filtering, Systems Modeling and Signal Processing.

Dr. João Viana da Fonseca Neto, Federal Universityof Maranhão, São Luís, Brazil. He is Professor of the Federal University of Maranhão. Graduated in Electrical Engineering from the Federal University of Paraíba, PB, Brazil (1982), master's degree in electrical engineering from the Federal University of Paraíba (1986), PB, Brazil and Doctorate degree in Electrical Engineering from the State University of Campinas, SP, Brazil(2000). Developed projects of R&Din Modeling and Control for Maranhão AluminumPlant (ALUMAR) and Intelligent Systems for Decision Making for Unloading and Storage of Minerals to VALE. He developed research in Computational Intelligence focused on neuro-fuzzy and genetic-algorithm to optimal control design. Currently, he is developing optimal structures for model free for state observer and indirect measurement systems to industrial applications.

Published By:
Blue Eyes Intelligence Engineering
& Sciences Publication Pvt. Ltd.