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A NOVEL AND EFFICIENT METHOD FOR ANTENNA RADIATION PATTERN USING ADAPTIVE FILTERING TECHNIQUES

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ABSTRACT – Smart antenna is recognized as promising technologies for higher user capacity in 3G wireless networks by effectively reducing multipath and co-channel interference. Advent of powerful, low-cost, digital processing components and the development of software-based techniques has made smart antenna systems a practical reality for both base station and mobile station of a cellular communications systems in the next generation. The core of smart antenna is the selection of smart algorithms in adaptive array. Using beam forming algorithms the weight of antenna arrays can be adjusted to form certain amount of adaptive beam to track corresponding users automatically and at the same time to minimize interference arising from other users by introducing nulls in their directions. Thus interferences can be suppressed and the desired signals can be extracted. Here we concentrate on elimination of noise energy in received signals by using variable step size algorithms called NLMS and VSSLMS which are adaptive filtering methods for error reduction. Although we have good performance in operation we extend our work by using LSCMA algorithm for speed convergence in error reduction.

KEYWORDS: NLMS, VSSLMS, LSCMA, WSN, Convergence, Cellular communication, Hybrid Technology, Noise energy.

INTRODUCTION:

Conventional base station antennas in existing operational systems are either omnidirectional or sectorised. There is a waste of resources since the vast majority of transmitted signal power radiates in directions other than toward the desired user. In addition, signal power radiated throught the cell area will be experienced as interference by any other user than the desired one. Concurrently the base station receives interference emanating from the individual users within the system Smart Antennas offer a relief by transmitting/receiving the power only to/from the desired directions. .Smart Antennas can be used to achieve different benefits. The most important is higher network capacity. It increase network capacity by precise control of signal nulls quality and mitigation of interference combine to frequency reuse reduce distance (or cluster size), improving capacity. It provides better range or coverage by focusing the energy sent out into the cell, multi-path rejection by minimizing fading and other undesirable effects of multi-path propagation. The smart antenna is a new technology and has been applied to the mobile communication system such as GSM and CDMA .It will be used in 3G mobile communication system or IMT 2000 also. Smart antenna can be used to achieve different benefits. By providing higher network capacity, it increases revenues of network operators anfd gives customers less propability of blocked or dropped calls. A smart antenna consists of number of elements (referred to as antenna array), whose signals are processed adaptively in order to exploit the spatial dimension of the mobile radio channel . All elements of the adaptive antenna array have to be combined (weighted) in order to adapt to the current channel and user characteristics . This weight adaptation is the “smart” part of the smart antenna , which should hence be called “adaptive antenna”. The adaptive antenna systems approach communication between a user and base station in a different way, in effect adding a dimension of space. By adjusting to an RF environment as it changes, adaptive antenna technology can dynamically alter the signal patterns to near infinity to optimize the performance of the wireless system. Adaptive arrays utilize sophisticated signal processing algorithms to continuously distinguish between desired signals, multipath, and interfering signals as well as calculate their directions of arrival. This approach continuously updates its transmit strategy based on changes in both the desired and interfering signal locations. 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.

ADAPTIVE BEAM-FORMING ALGORITHMS

The digital signal processor interprets the incoming data information, determines the complex weights (amplification and phase information) and multiplie the weights to each element output to optimize the array pattern. The optimization is based on a particular criterion, which minimizes the contribution from noise and interference while producing maximum beam gain at the desired direction. There are several Adaptive beam-forming algorithms varying in complexity based on different criteria for updating and computing the optimum weights. Block implementation of the adaptive beam-former uses a block of data to estimate the adaptive beam-forming weight vector and is known as “sample matrix inversion (SMI)”. The sample-by-sample method updates the adaptive beam-forming weight vector with each sample.

The SMI adaptive beam-former uses a block of code to get optimum weight; therefore it is not suitable for non stationary environment. Since the sample-by-sample adaptive beam-former alters its weight with each new sample, it can dynamically update its response for such a changing scenario. Another important distinction between sample and block adaptive method is the inclusion of the signal of interest in each sample and thus in the correlation matrix. Therefore, for sample adaptive method, we can not use signal free version of the correlation matrix, that is, the interference plus noise correlation matrix, but rather must use the whole correlation matrix .The inclusion of the signal in the correlation matrix has profound effect on the robustness of the adaptive beam-former in the case of signal mismatch.

RF power output has been a major planning aspect for engineers since the start of radio

transmission. Undoubtedly important, RF Power level is only one of many factors that determine a successful wireless network. To effectively evaluate and differentiate between various microwave systems and link performance, operators should be able to examine several key aspects of RF power output. Propagation and antennas set aside, parameters such as receiver threshold, modulation type and RF power level are the most important to consider. However, deciphering the diverse range of often incomparable radio system data sheets may prove a challenge. Adaptive Modulation schemes and ATPC (Automatic Transmit Power Control) already provide point-to-point microwave systems with a high degree of flexibility, ensuring better efficiency under changing weather conditions. This paper discusses how RF output power can now also be controlled dynamically so as to ensure the highest power efficiency under changing modulations. We will also present the various methods in which RF output power control is implemented by Ceragon’s FibeAir IP-10 point-to-point microwave solution.

The goal of Adaptive Modulation is to improve the operational efficiency of Microwave links by increasing network capacity over the existing infrastructure - while reducing sensitivity to environmental interferences. Adaptive Modulation means dynamically varying the modulation in an errorless manner in order to maximize the throughput under momentary propagation conditions. In other words, a system can operate at its maximum throughput under clear sky conditions, and decrease it gradually under rain fade.

Our proposed scheme is based on the beam forming approach to excluding interference emanating from a direction other than that of the desired signal. For MIMO system, there are a total of three different types of interference, which are Co-Cannel

Interference (CCI), Interference Inter Symbol ISI and Multiple Antenna Interference (MAI). The MAI defined here is the interference from different transmits antennas in the MIMO system. We are targeting on the three

interferences sources and aiming to provide a solution to the problems of resolving all ISI and MAI and canceling CCI effectively by employing LMS algorithm by adjusting both transmitter and receiver weights in the MIMO system

Standard LMS

The standard LMS algorithm performs the following operations to update the coefficients of an adaptive filter:

1. Calculates the output signal y(n) from the adaptive filter.
2. Calculates the error signal $e(n)$ by using the following equation: $e(n) = d(n)–y(n)$.
3. Updates the filter coefficients by using the following equation:

   $$\vec{w}(n+1) = \vec{w}(n) + \mu \cdot e(n) \cdot \vec{u}(n)$$

   where μ is the step size of the adaptive filter, $\vec{w}(n)$ is the filter coefficients vector, and $\vec{u}(n)$ is the filter input vector.

Use the AFT Create FIR LMS VI to create an adaptive filter with the standard LMS algorithm.

Normalized LMS

The normalized LMS (NLMS) algorithm is a modified form of the standard LMS algorithm. The NLMS algorithm updates the coefficients of an adaptive filter by using the following equation:

$$\vec{w}(n+1) = \vec{w}(n) + \mu \cdot e(n) \cdot \frac{\vec{u}(n)}{\|\vec{u}(n)\|^2}$$

You also can rewrite the above equation to the following equation:

$$\vec{w}(n+1) = \vec{w}(n) + \mu(n) \cdot e(n) \cdot \vec{u}(n)$$

VARIABLE STEP SIZE LMS ALGORITHMS:

Many variable step size LMS algorithms, that is to adjust the step size dynamically during the process of convergence. Document [1] proposed a variable step size LMS algorithm (n) is theμbased on sigmoid function, the step size sigmoid function of e(n) :

$$\mu(n) = \beta[\frac{1}{1+e^{-\alpha|e(n)|}} - 0.5]$$

It solved the contradiction of the convergence rate, tracking capabilities and the steady-state error, but when the changes too much, that leadμerror close to zero, step size to greater step in steady state. Documents [2] and [3] give the improved algorithms based on document [1], and make the error changes smaller in steady state. Document [4] proposed a variable step size LMS adaptive filter algorithm based on Lorentzian function it uses the Lorentzian function as the (n):μformula of step size

$$\mu(n) = \alpha \log[1 + \frac{1}{2}(\frac{e(n)}{\delta})^2]$$

It improved the convergence speed and tracking speed further. In this paper, a new adaptive variable step size algorithm is proposed based on the LMS filter algorithm of Lorentzian function, it eliminates the influence of irrelevant noise, and has good anti-interference ability, the formula of (n) is:μimproved step factor

$$\mu(n) = \alpha \log[1 + \frac{1}{2}\frac{e(n)e(n-1)}{\delta^2}]$$

THE LS-CMA-ES ALGORITHM :

The goal of this section is to introduce an algorithm that will try to use an approximation of the Hessian matrix of the target function as advocated in section 3: This approximation will be deterministically built using the available sample points – and later used within the Gaussian mutation 1. Whereas such strategy is optimal for elliptic functions, it can fail on functions that are “more flat” (e.g. the famous Rosenbrook function), because the quadratic approximation then becomes very inaccurate. A criterion needs to be designed to detect such cases and to alternatively use the CMA “steepest descent update rule” for the covariance matrix. a new approach is proposed, in which the update of the covariance matrix is based on a quadratic approximation of the target function, obtained by some Least-Square minimization. A dynamic criterion is

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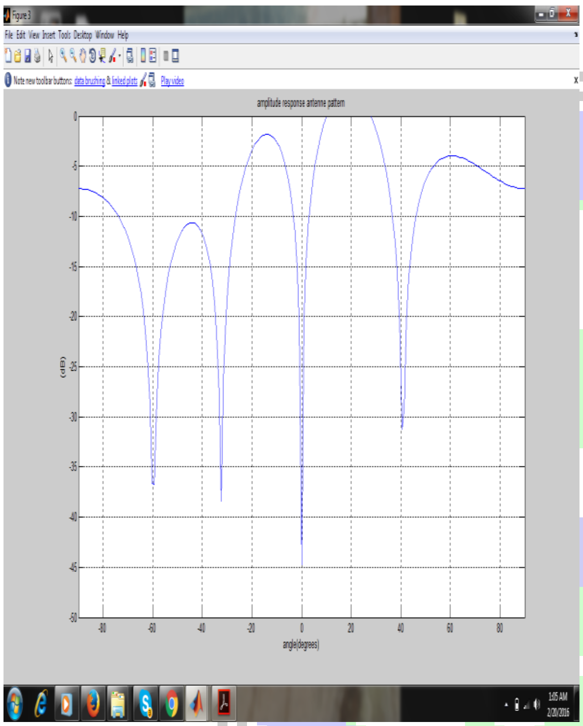
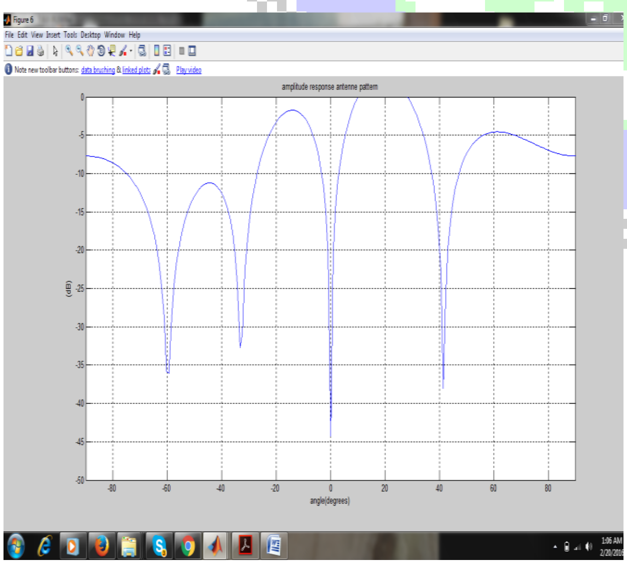
designed to detect situations where the approximation is not accurate enough,

RESULT:

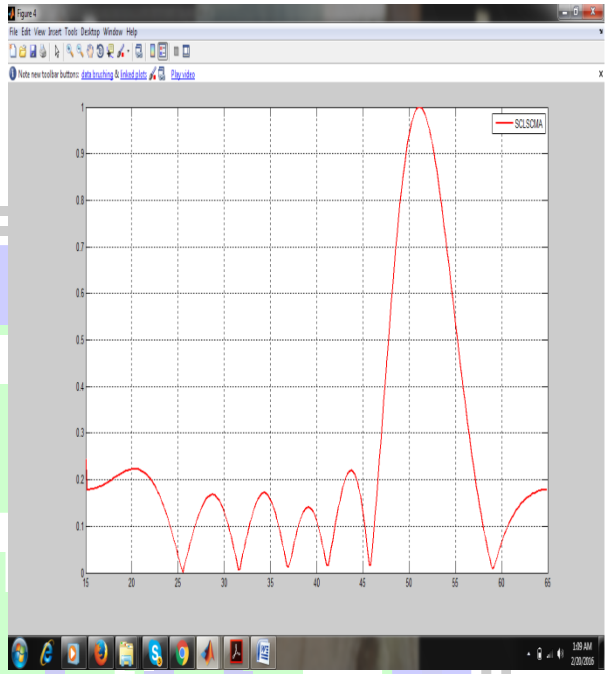
NLMS OUTPUT:



VSSLMS:



SCLMCMA



CONCLUSION:

Finally, this paper implemented and analyzed the use of a new variable step size normalized least mean square VSSNLMS and LS-CMA algorithm in the design of adaptive beam forming smart antenna system for interference suppression. The used VSSNLMS algorithm enhances the performance of the smart antenna by reducing the mean square error (MSE) which results in faster convergence rate when compared to NLMS algorithm. The robust performance of the developed LS-CMA algorithm takes the form of better interference suppression due to deeper nulls produced in the directions of interfering signals.

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