Improved Smart Antenna Design Using Displaced Sensor Array Configuration

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***Abstract*-**In this paper we propose a new sensor array configuration for improved smart antenna design. The new configuration involves two parallely-displaced sensor arrays in the vertical plane. The proposed sensor array configuration avoids the problem of spatial aliasing encountered in largely spaced sensor arrays and reduces the effects of inter-element mutual coupling for closely spaced arrays. Moreover, the proposed sensor array configuration allows for doubling the number of array elements and, hence, increasing the system capacity, without significantly increasing the array aperture. This allows for a more accurate beam pattern to be generated especially in a radio environment with a large number of interference signals. Numerical results are presented to demonstrate the improved performance of direction-of-arrival estimation and adaptive beamforming algorithms when the proposed array configuration is used.

***Index Terms*-**Smart antennas, adaptive array processing, direction-of-arrival estimation, adaptive beamforming.

I. INTRODUCTION

The main impairments in wireless communication are multipath fading, co-channel interference, and delay spread. Smart antenna systems overcome these impairments providing a wider coverage and a greater capacity. This promising technology has been incorporated in 3G and 4G wireless systems allowing for high data rate applications [1], [2].

A smart antenna system at the base station of a cellular mobile system is depicted in Fig. 1. It consists of a uniform linear antenna array for which the currents are adjusted by a set of complex weights using an adaptive beamforming algorithm. The adaptive beamforming algorithm optimizes the array output beam pattern such that maximum radiated power is produced in the directions of desired mobile users and deep nulls are generated in the directions of undesired signals representing co-channel interference from mobile users in adjacent cells [1]-[4]. Prior to adaptive beamforming, the directions of users and interferes must be obtained using a direction-of-arrival estimation algorithm [5], [6], shown in Fig. 1.

Most smart antenna systems utilize a uniform linear array (ULA) of N elements that are spaced apart by half-wavelength ($d = \lambda/2$). The inter-element spacing in a ULA is chosen to be $\lambda/2$ in order to reduce mutual coupling effects which deteriorate the performance of the direction-of-arrival estimation and adaptive beamforming algorithms as demonstrated in [2], [6]. If the inter-element spacing is chosen to be smaller than $\lambda/2$ the mutual coupling effects cannot be ignored and the beamforming algorithm fails to produce the actual beam pattern. On the other hand, increasing the inter-element spacing beyond $\lambda/2$ results in spatial aliasing which takes the form of unwanted peaks in the output beam pattern. It is therefore concluded that $d = \lambda/2$ represents the optimum value for the inter-element spacing in a ULA.

Due to practical considerations it is desirable to design smart antenna systems with smaller size. This can be done by employing a ULA with smaller aperture. Since the inter-element spacing in a ULA is maintained to $\lambda/2$ the array aperture can be reduced by decreasing the number of antennas in the array. This will, however, limit the array capability of handling more desired and interfering signals resulting in a reduction in the system capacity.

It is therefore desirable to design a smart antenna system which can handle more signals without increasing the array aperture significantly. This paper presents a new array configuration which consists of two parallely-displaced arrays aligned in the vertical plane. The proposed array configuration has several advantages. First, it maintains almost the same radiation aperture as the conventional uniform linear array yet it can handle more signals from users and interferers. Second, the horizontal displacement between the two arrays in the proposed array configuration allows for resolving correlated signals encountered in multipath propagation environment without having to apply spatial smoothing techniques [5]. Moreover, the vertical displacement between the arrays allows for resolving signals arriving at grazing incidence (endfire direction) in the vertical plane.

The paper is organized as followsSection II develops the signal model for the proposed array configuration. Sections III and IV present the theory of the direction-of-arrival estimation and adaptive beamforming algorithms, respectively. Section V presents results that demonstrate the improved performance of

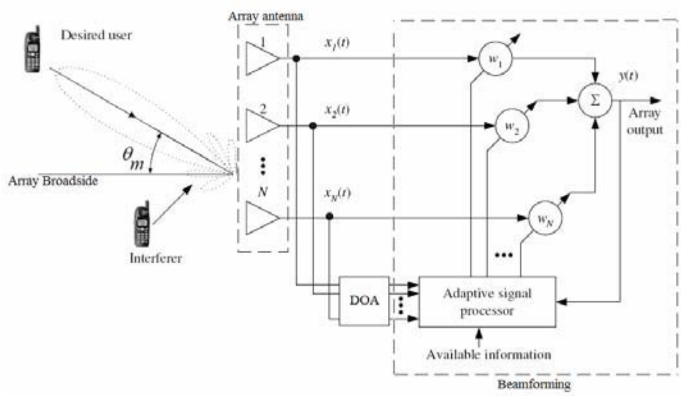

Fig. 1. A functional bolock diagram of a smart antenna system.

the smart antenna system employing the proposed array configuration when compared to the conventional ULA. Finally, conclusions are given in Section VI.

II. SIGNAL MODEL

Consider two parallel uniform linear arrays displaced by a horizontal distance $d = \lambda/4$ and vertical separation $s = \lambda/2$, as shown in Fig. 2. Each array consists of N linear equispaced omni-directional sensors with inter-element spacing $d = \lambda/2$ receiving M narrowband signals $s_m(t)$ incident with azimuth angles of arrival $\theta_m, 1 \leq m \leq M$. The two arrays are assumed to be located parallel to the x-axis and the azimuth angle θ_m is measured with respectto the z-axis. Following this coordinate system, the received data vector $\mathbf{x}(t)$ is given by

$$\mathbf{x}(t) = \sum_{m=1}^{M} [\mathbf{a}_1(\theta_m) + \mathbf{a}_2(\theta_m)]\, s_m(t) + \mathbf{n}(t) \tag{1}$$

where $\mathbf{n}(t)$ is a noise vector modeled as temporally white and zero-mean complex Gaussian process, $\mathbf{a}_1(\theta_m)$ and $\mathrm{a}_2(\theta_m)$ are the steering (or response) vectors for the two parallel arrays, corresponding to the DOA of the m^{th}signal. The steering vectors for the two subarrays are defined as

$$\mathbf{a}_1(\theta_m) = \left[e^{-j(n-1)2\pi\left(\frac{d}{\lambda}\right)\sin\theta_m}\right]^T, \qquad 1 \leq n \leq N \tag{2}$$

$$\mathbf{a}_2(\theta_m) = \mathbf{a}_1(\theta_m)\ .\ e^{-j2\pi\left(\frac{\Delta}{\lambda}\right)\sin\theta_m}\ .\ e^{-j2\pi\left(\frac{s}{\lambda}\right)\cos\theta_m} \tag{3}$$

where $[.]^T$ is the transpose operator, λ is the wavelength of the incident signals, and Δ is the horizontal displacement between the two arrays and is equal to $\lambda/4$ as illustrated in Fig. 2. The combination of all possible steering vectors forms the array manifold (or steering vector) matrices $\mathbf{A}_1$ and $\mathbf{A}_2$. The received signal vector $\mathbf{x}(t)$ of (1) can then be re-written as

$$\mathbf{x}(t) = [\mathbf{A}_1 + \mathbf{A}_2]\, s(t) + \mathbf{n}(t) = \mathbf{A}\mathbf{s}(t) + \mathbf{n}(t) \tag{4}$$

where the overall array manifold matrix $\mathbf{A} = \mathbf{A}_1 + \mathbf{A}_2$.

III. DIRECTION-OF-ARRIVAL ESTIMATION

A common direction-of-arrival estimation algorithm is MUSIC (Multiple Signal Classification) [5]. It starts by applying temporal averaging over K snapshots (or samples) taken from the signals incident on the sensor array. This averaging process leads to forming a spatial correlation (or covariance) matrix $\mathbf{R}$ defined as

$$\mathbf{R} = \frac{1}{K}\sum_{t=1}^{K} \mathbf{x}(t)\mathbf{x}(t)^H \tag{5}$$

where $[.]^H$ denotes the Hermitian operator. Substituting $\mathbf{x}(t)$ from (4) into (5) results in

$$\mathbf{R} = \frac{1}{K}\sum_{t=1}^{K} \mathbf{A}\mathbf{s}(t)\mathbf{s}(t)^H\mathbf{A}^H + \mathbf{n}(t)\ \mathbf{n}(t)^H \tag{6}$$

which can be expressed as

$$\mathbf{R} = \mathbf{A}\mathbf{R}_{ss}\mathbf{A}^H + \boldsymbol{\sigma}_n^2\mathbf{I} \tag{7}$$

where $\mathbf{R}_{ss}$ is the signal covariance matrix, $\boldsymbol{\sigma}_n^2$ is the noise variance, and $\mathbf{I}$ is an identity matrix of size $N \times N$.

It can be shown [6] that the covariance matrix $\mathbf{R}$ has M signal eigenvalues with corresponding eigenvectors $\mathbf{e}_1,\ \mathbf{e}_2\ , ..., .\mathbf{e}_M$. The remaining $N - M$ eigenvalues of the covariance matrix $\mathbf{R}$ represent noise eigenvalues with corresponding eigenvectors $\mathbf{e}_{M+1},\ \mathbf{e}_{M+2}\ , ..., .\mathbf{e}_N$. We form the signal subspace matrix $\mathbf{E}_s$ for which the columns are the M signal eigenvectors of the covariance matrix $\mathbf{R}$, i.e., $\mathbf{E}_s = [\mathbf{e}_1\ \mathbf{e}_2\ ....\ \mathbf{e}_M]^T$. Similarly, we form the noise subspace matrix $\mathbf{E}_n$ for which the columns are the remaining $N - M$ noise eigenvectors of the covariance matrix $\mathbf{R}$, i.e., $\mathbf{E}_n = [\mathbf{e}_{M+1}\ \mathbf{e}_{M+2}\ ....\ \mathbf{e}_N]^T$. The normalized MUSIC angular spectrum is then given by [1],

$$P(\theta) = \frac{\mathbf{A}^H\mathbf{A}}{\mathbf{A}^H\mathbf{E}_n\mathbf{E}_n^H\mathbf{A}}. \tag{8}$$

By examining the denominator in (8), it is evident that peaks in the MUSIC angular spectrum occur whenever the overall array manifold (or steering vector) matrix $\mathbf{A}$ is orthogonal to the noise subspace matrix $\mathbf{E}_n$. These angles θ at which the peaks occur define the directions-of-arrival of the of the signals impinging on the sensor array. The number of signals that can be detected is restricted by the number of elements in the sensor array. In [5] it was verified that an N-element sensor array can detect up to $N - 1$ uncorrelated signals. This number reduces to $N/2$ signals if they are correlated. A comprehensive performance evaluation of the MUSIC algorithm for DOA estimation can be found in [6] and [7].

IV. ADAPTIVE BEAMFORMING

An adaptive beamformer, shown in Fig. 3, consists of multiple antennas; complex weights, the function of which is to amplify (or attenuate) and delay the signals from each antenna element; and a summer to add all of the processed signals, in order to tune out the signals not of interest, while enhancing the signal of interest. Hence, beamforming is

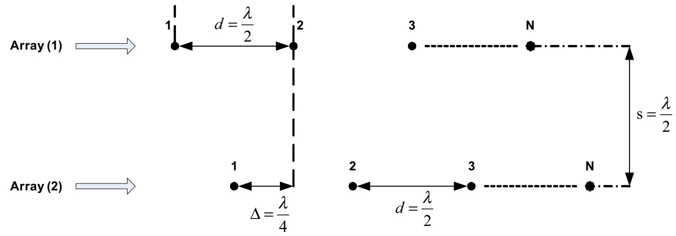

Fig. 2. Proposed array configuration consists of two parallely-displaced arrays in the vertical plane displaced horizontally by a distance $\Delta = \lambda/4$ and separated vertically by a distance $s = \lambda/2$.

sometimes referred to as spatial filtering, since some incoming signals from certain spatial directions are filtered out, while others are amplified. The output response of the uniform linear array shown in (3) is given by

$$y(n) = \mathbf{w}^H \mathbf{x}(n), \tag{9}$$

where $\mathbf{w}$ is the array weights vector and $\mathbf{x}$ is the received signal vector given in (4).

The array weights vector w in (9) in are obtained using an adaptive beamforming algorithm. Adaptive beamforming algorithms are classified as either DOA-based, temporal-reference-based, or signal-structure-based. In DOA-based beamforming, the direction-of-arrival algorithm passes the DOA information to the beamformer, as illustrated in Fig. 1. The beamforming algorithm is used to design a radiation pattern with the main beam directed towards the signal of interest, and with nulls in the directions of the interferers.

On the other hand, temporal-reference beamformers use a known training sequence to adjust the weights, and to form a radiation pattern with a maximum towards the signal of interest and nulls towards the signals not of interest. Specifically, if $d(n)$ denotes the sequence of reference or training symbols known a priori at the receiver at time n, an error, $\epsilon(n)$ is formed as

$$\epsilon(n) = d(n) - \mathbf{w}^H x(n). \tag{10}$$

This error signal ϵ is used by the beamformer to adaptively adjust the complex weights vector $\mathbf{w}$ so that the mean-squared error (MSE) is minimized. The choice of weights that minimize the MSE is such that the radiation pattern has a beam in the direction of the source that is transmitting the reference signal, and that there are nulls in the radiation pattern in the directions of the interferers. Once the beamformer has locked onto the reference signal, then the complex weights are maintained as fixed, and transmission of the data packet begins. The array weights vector using the Least Mean Squares (LMS) algorithm can be expressed as [7],

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{x}(n) \epsilon^*(n) \tag{11}$$

where $\mathbf{w}(n+1)$ denotes the weights vector to be computed at iteration $n+1$ and μ is the LMS step size which is related to the rate of convergencein other words, how fast the LMS algorithm reaches steady state. The smaller the step size the longer it takes the LMS algorithm to converge. This means that a longer reference or training sequence is needed, which

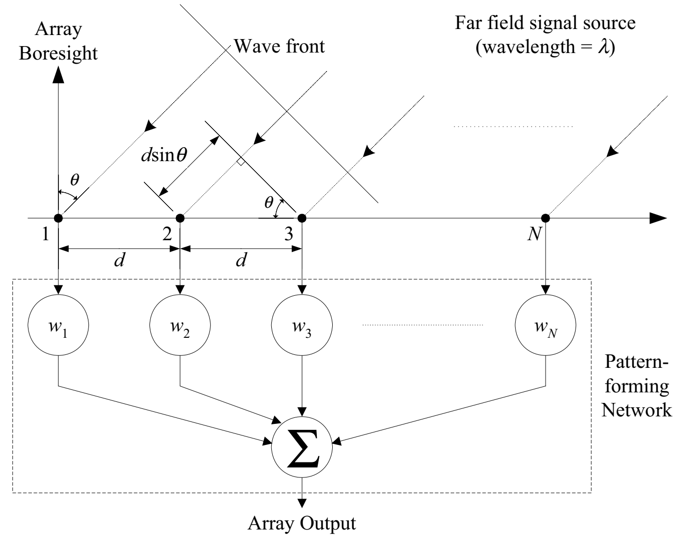

Fig. 3. Top view of a smart antenna system utilizing a uniform linear array (ULA) of N sensors.

would reduce the payload and, hence, the bandwidth available for transmitting data.. In order to ensure the stability and convergence of the algorithm, the adaptive step size should be chosen within the range specified as

$$0 < \mu < \lambda_{\max}^{-1} \tag{12}$$

where λ is the maximum eigenvalue of the input covariance matrix $\mathbf{R}$ obtained in (7).

V. RESULTS AND DISCUSSION

We have obtained the results for MUSIC direction-of-arrival estimation and LMS adaptive beamforming algorithms using both the conventional uniform linear array configuration with $N = 4$ elements in the array, and the proposed two parallely-displaced array configuration with $N = 4$ elements in each array so that total number of elements used is $2N = 8$. Inter-element spacing of $d = \lambda/2$ is maintained in both configurations.

Simulation results for the MUSIC direction-of-arrival estimation algorithm are presented in Figs. 4 and 5 We have assumed a signal-to-noise ratio $SNR = 20$ dB and the number of snapshots $K = 1000$. The MUSIC algorithm is used to detect two incoming signals arriving at incidence angles $\theta_{m=1}$=-60^{o}and $\theta_{m=1}$=60^o. Figs. 4 and 5 show the MUSIC angular spectrum for the conventional uniform linear array configuration and proposed parallely-displaced array configuration, respectively. It is evident from Fig. 5 that using the proposed array configuration results in a MUSIC spectrum with sharper peaks and lower noise floor. Hence, the MUSIC algorithm can better resolve incoming signals when the proposed array configuration is used.

Simulation results for the LMS adaptive beamforming algorithm are presented in Fig. 6. In this case, both the desired and interfering signals take the form of a simple complex sinusoidal-phase modulated signal. By doing so it can be shown in the simulations how interfering signals of the same

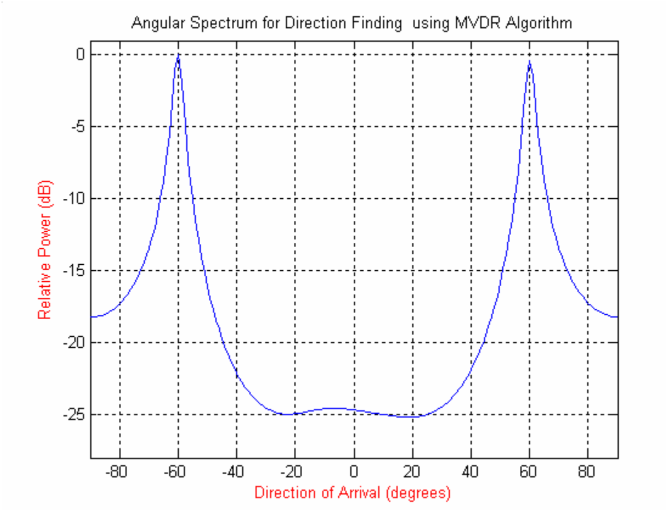

Fig. 4. MUSIC angular spectrum using conventional uniform linear array configuration with $N = 4$.

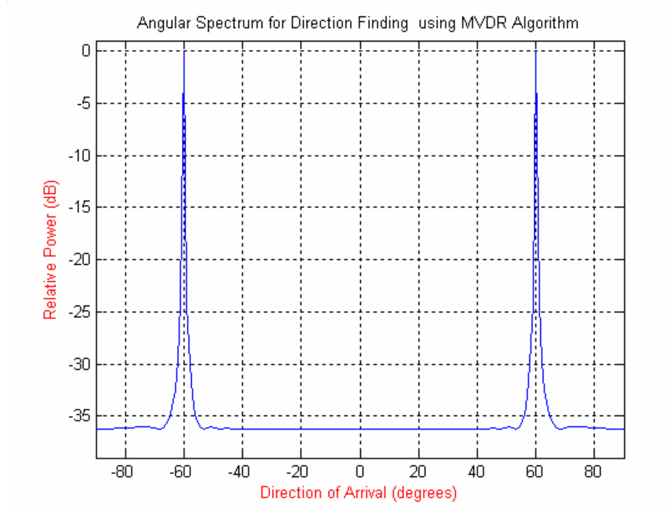

Fig. 5. MUSIC angular spectrum using proposed two parallely-displaced array configuration with $2N = 8$.

frequency as the desired signal can be separated to achieve rejection of co-channel interference. In the example provided here, the desired signal arrives at an angle $\theta_S = 20^o$ and interference signal arrives at an angle $\theta_I = -40^o$. The signal-to-noise ratio $SNR = 20$ dB and signal-to-interference ratio $SIR = -10$ dB. The LMS adaptive step size is $\mu = 0.001$ and the number of iterations is 1000. Fig. 6 shows the LMS beam pattern using both conventional and proposed array configurations. It is evident from Fig. 6 that using the proposed array configuration (solid line), the LMS beamformer is able to iteratively update the array weights to force a deep null in the direction of the interference signal $\theta_I = -40^o$. The null is -60 dB deep below the maximum. The proposed array configuration (solid line in Fig. 6) also allows the LMS beamformer to produce a maximum in the right direction toward the desired signal at θ_S=20^o. Hence, the performance of the LMS beamformer is superior when the proposed paralley-displaced array configuration is used.

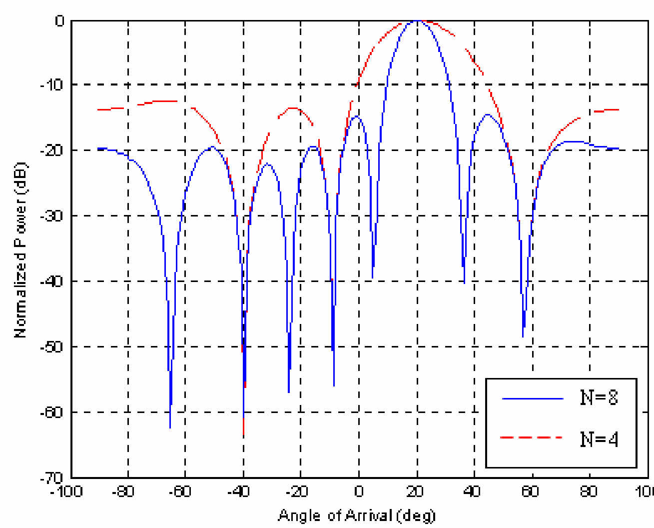

Fig. 6. LMS beam pattern using conventional uniform linear array configuration with $N = 4$ (dashed line), and proposed two parallely-displaced array configuration with $2N = 8$ (solid line).

VI. CONCLUSIONS

We have proposed a new sensor array configuration which improves the performance of smart antenna systems. The proposed configuration consists of two parallely-displaced sensor arrays aligned in the vertical plane. Performance results for both MUSIC direction-of-arrival estimation and LMS adaptive beamforming algorithms show significant improvement when the proposed array configuration is used. Besides the improved performance of the direction-of-arrival and adaptive beamforming algorithms, the proposed array configuration has several other advantages. First, it maintains almost the same radiation aperture as the conventional uniform linear array yet it can handle more signals from users and interferers because it has more array sensors when compared to the conventional uniform linear array. Second, the horizontal displacement in the proposed configuration between the two parallel arrays allows for resolving correlated signals encountered in multipath propagation environment without having to apply spatial smoothing techniques. Moreover, the vertical separation between the two parallel arrays allows for resolving signals arriving at grazing incidence (endfire direction) in the vertical plane.

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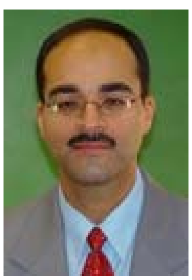

Raed M. Shubair received his B.A.Sc. degree from Kuwait University, Kuwait, in June 1989 and his Ph.D. degree from the University of Waterloo, Canada, in February 1993, both in Electrical Engineering. From March 1993 to August 1993 he was a Postdoctoral Fellow at the Department of Electrical and Computer Engineering, University of Waterloo, Canada. In September 1993 he joined Etisalat University College, UAE, where he is currently an Associate Professor at the Communication Engineering Department and leader of the Communication & Information Systems (CIS) research group. His current research interests include adaptive array processing, smart antennas and MIMO systems, as well as applied and computational electromagnetic modeling of RF and microwave circuits for wireless communications. Dr. Shubair has authored over 80 papers in refereed technical journals and international conferences. He has been a member of the technical program, organizing, and steering committees of numerous international conferences and workshops. Dr. Shubair organized and chaired a number of technical sessions in international conferences including IEEE Symposium on Antenna and Propagation (AP-S), IEEE Symposium on Electronics, Circuits and Systems (ICECS), Progress in Electromagnetics Research Symposium (PIERS), and Applied Computational Electromagnetics Symposium (ACES). Dr. Shubair is a Senior Member of the IEEE. He was also elected to the MIT Electromagnetics Academy in 2001. He has supervised a large number of research projects including *Adaptive Beamforming for Next-Generation Wireless Communications*, recipient of the 2004 IEE Award, as well as *Design of Optimum SMI Beamformers for Spatial Interference Rejection*, recipient of the 2005 IEE Award. Dr. Shubair is a founding member of the IEEE UAE Signal Processing and Communications Joint Societies Chapter. He is listed in *Who's Who in Electromagnetics* and in several editions of *Who's Who in Science and Engineering*.