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Smart Antennas

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Abstract

Smart antennas have recently received increasing interest for improving the performance of wireless radio systems. These systems of antennas include a large number of techniques that attempt to enhance the received signal, suppress all interfering signals, and increase capacity, in general. The main purpose of this article is to provide an overview of the current state of research in the area of smart antennas, and to describe how they can be used in wireless systems. Thus, this article provides a basic model for determining the angle of arrival for incoming signals, the appropriate antenna beamforming, and the adaptive algorithms that are currently used for array processing. Moreover, it is shown how smart antennas, with spatial processing, can provide substantial additional improvement when used with TDMA and CDMA digital-communication systems. The material presented is tutorial in nature, leaving the details for further study from the papers appearing in the reference list.

Keywords: Smart antennas; adaptive arrays; switched beam antennas; TDMA; CDMA; wireless communications; land mobile radio cellular systems

1. Why Smart Antennas?

A smart antenna consists of an antenna array, combined with signal processing in both space and time. Spatial processing leads to more degrees of freedom in the system design, which can help improve the overall performance of the system. The concept of using antenna arrays and innovative signal processing is not new to radar and aerospace technology [1]. Until recent years, cost effectiveness has prevented their use in commercial systems. The advent of very fast and low-cost digital signal processors has made smart antennas practical for cellular land- or satellite-mobile communications systems.

Recently, the application of smart-antenna arrays has been suggested for mobile-communications systems, to overcome the problem of limited channel bandwidth, satisfying a growing demand for a large number of mobiles on communications channels. Smart antennas, when used appropriately, help in improving the system performance by increasing channel capacity and spectrum efficiency, extending range coverage, steering multiple beams to track many mobiles, and compensating electronically for aperture distortion. They also reduce delay spread, multipath fading, co-channel interference, system complexity, bit error rate (BER), and outage probability. Delay spread occurs in multipath propagation environments when a desired signal, arriving from different directions, becomes delayed due to different travel distances. Delay spread and multipath fading can be reduced with an antenna array that is capable of forming beams in certain directions and nulls in others, thereby canceling some of the delayed arrivals. Usually, in the transmitting mode, the array focuses energy in the required direction, which helps to reduce multipath reflections and the delay spread. In the receiving mode, however, the array provides compensation in multipath fading by adding the signals emanating from other clusters after compensating for delays, as well as by canceling delayed signals emanating from directions other than that of the desired signal. System complexity and cost is decreased by the use of a smaller number of base stations. The increase in the spectrum efficiency– which is defined as the amount of traffic a given system with certain spectrum allocation could manage–is a result of the capability of the antenna array to provide virtual channels in an angle domain. This is referred to as Spatial-Division Multiple Access (SDMA), which means that it is possible to multiplex channels in the spatial dimension, just as in the frequency and time dimensions [2]. The increase is achieved by using spatially selective reception and spatially selective transmission. Finally, the reduction in outage probability, which is the probability that a mobile user would lose the ability to communicate, is achieved from the increase in the available channels.

Perhaps the most important feature of a smart-antenna system is its capability to cancel co-channel interference. Co-channel interference is caused by radiation from cells that use the same set of channel frequencies. Thus, co-channel interference in the transmitting mode is reduced by focusing a directive beam in the direction of a desired signal, and nulls in the directions of other receivers. In the receiving mode, co-channel interference can be reduced by knowing the directional location of the signal's source, and utilizing interference cancellation.

The antenna system needs to differentiate the desired signal from the co-channel interference, and normally requires either "a priori" knowledge of a reference signal, or the direction of the desired signal source, to achieve its desired objectives. There exist a variety of methods to estimate the direction of sources, with conflicting demands of accuracy and processing power. Similarly, there are many algorithms to update the array weights, each with its speed of convergence and required processing time [3, 4]. Algorithms also exist that exploit properties of signals to eliminate the need of training signals, in some circumstances.

Today, adaptive antennas and the algorithms to control them are vital to high-capacity communications-system development.

2. Types of Smart antennas

Today, several terms are used to refer to the various aspects of smart-antenna system technology, including intelligent antennas, phased arrays, SDMA, spatial processing, digital beamforming, adaptive antenna systems, and others [5]. Smart-antenna systems, however, are usually categorized as either switched-beam or adaptive-array systems. Although both systems attempt to increase gain in the direction of the user, only the adaptive-array system offers optimal gain, while simultaneously identifying, tracking, and minimizing interfering signals [6]. It is the adaptive system's active interference capability that offers substantial performance advantages and flexibility over the more-passive switched-beam approach. Smart antennas communicate directionally by forming specific antenna-beam patterns. They direct their main lobe, with increased gain, in the direction of the user, and they direct nulls in directions away from the main lobe. Different switched-beam and adaptive smart antennas control the lobes and the nulls with varying degrees of accuracy and flexibility.

Figure 1 illustrates the beam patterns that each system might choose, in a scenario involving one desired signal and two co-channel interferers. The switched-beam system is depicted on the left, while the adaptive system is on the right. Both systems direct their major lobe in the general direction of the signal of interest, but the adaptive system chooses a more-accurate placement, providing greater signal enhancement. Similarly, the interfering signals arrive at places of lower gain outside the main lobe, but, again, in the adaptive system, the interfering signals receive maximum suppression.

The traditional switched-beam method is considered as an extension of the current cellular sectorization scheme, in which a typical sectorized cell site is composed of three 120-degree macro-sectors. The switched-beam approach further subdivides the macro-sectors into several micro-sectors. Each micro-sector contains a predetermined fixed beam pattern, with the greatest gain placed in the center of the beam. Typically, the switched-beam system establishes certain choices of beam patterns before deployment, and selects from one of several choices during operation. When a mobile user is in the vicinity of a macro-sector, the switched-beam system selects the micro-sector containing the strongest signal. During the call, the system monitors the signal strength, and switches to other fixed micro-sectors, if required. All switched-beam systems offer similar benefits, even though the different systems utilize different hardware and software designs. Compared to conventional sectored cells, switched-beam systems can increase the range of a base station from 20% to 200%, depending on the circumstances of operation. The additional coverage means that an operator can achieve a substantial reduction in infrastructure costs.

There are, however, limitations to switched-beam systems. Since the beams are predetermined, the signal strength varies as

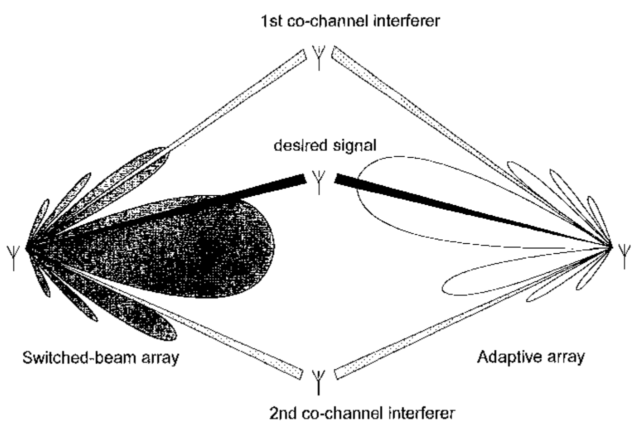

Figure 1. The beamforming lobes and nulls that switched-beam (left) and adaptive-array (right) systems might choose for identical user signals and co-channel interferers.

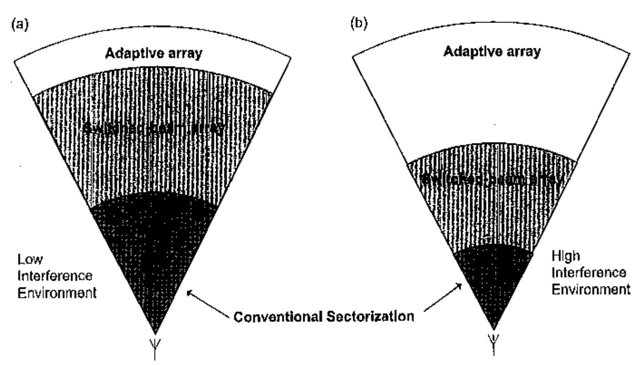

Figure 2. The coverage patterns for (a) switched-beam and (b) adaptive-array antennas.

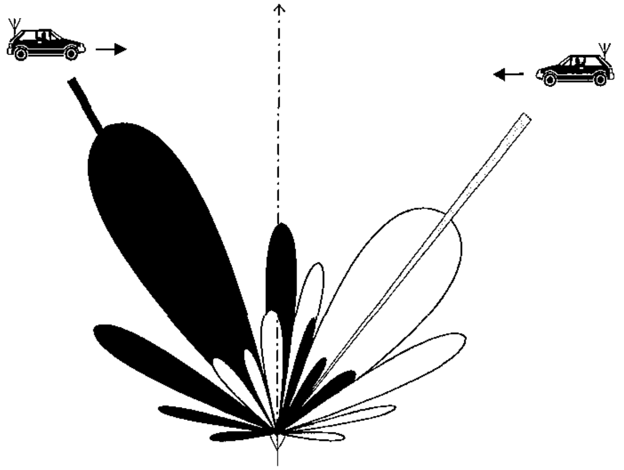

Figure 3a. An illustration of fully adaptive spatial processing, supporting two users on the same conventional channel simultaneously in the same cell. The dark beam pattern is used to communicate with the user on the left, while the light beam pattern is used to communicate with the user on the right.

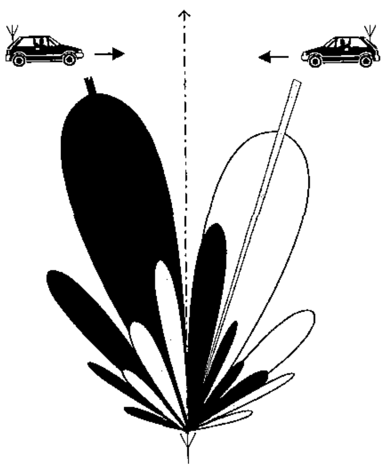

Figure 3b. An illustration of how the beam patterns for the adaptive system of Figure 3a are updated to accommodate the motion of the users.

the user moves through the sector. As a mobile unit approaches the far azimuth edges of a beam, the signal strength degrades rapidly before the user is switched to another micro-sector. Moreover, a switched-beam system does not distinguish between a desired signal and interfering signals. If the interfering signal is around the center of the selected beam and the user is away from the center, the quality of the signal is degraded for the mobile user.

Adaptive antennas take a very different approach. By adjusting to an RF environment as it changes, adaptive-antenna technology can dynamically alter the signal patterns to optimize the performance of the wireless system. The adaptive antenna utilizes sophisticated signal-processing algorithms [6] to continuously distinguish among desired signals, multipath, and interfering signals, as well as to calculate their directions of arrival. The adaptive approach continuously updates its beam pattern, based on changes in both the desired and interfering signal locations. The ability to smoothly track users with main lobes, and interferers with nulls, insures that the link budget is constantly maximized. This effect is similar to a person's hearing. When one person listens to another, the brain of the listener collects the sound in both ears, combines it to hear better, and determines the direction from which the speaker is talking. If the speaker is moving, the listener, even if his eyes are closed, can continue to update the angular position, based solely on what he hears. The listener also has the ability to tune out unwanted noise and interference, and to focus on the conversation at hand.

Figure 2 shows a comparison, in terms of relative coverage area, of conventional sectorized, switched-beam, and adaptive antennas. In the presence of a low level of interference (Figure 2a), both types of smart antennas provide significant gains over the conventional sectored systems. When a high level of interference is present (Figure 2b), the interference-rejection capability of the adaptive system provides significantly more coverage that either the conventional or switched-beam system.

Another important advantage of the next generation of adaptive-antenna systems is its capability to "create" spectrum. Because of the accurate tracking and robust interference-rejection capabilities, multiple users can share the same conventional channel within the same cell. System capacity increases through inter-cell frequency re-use patterns, as well as intra-cell frequency re-use.

Figure 3a shows how this technology can be used to accommodate two users on the same conventional channel, simultaneously, in the same cell. The dark beam pattern is used to communicate with the user on the left, while the light beam pattern is used to communicate with the user on the right. It should be noted that every pattern has nulls in the direction of the other user. As the users move, the beam patterns are constantly updated, to insure these positions (Figure 3b). It is this ability to continuously modify the beam pattern with respect to both lobes and nulls that separates the adaptive approach from the switched type. As interfering signals move throughout the sector, the switched beam pattern is not altered, because it only responds to movements of the signal of interest. Unlike the switched-beam approach, the adaptive system is able to continue to distinguish between the signal and the interferer, and to allow them to get substantially closer than in the switched-beam system, while maintaining enhanced carrier-to-interference-ratio levels. The most-sophisticated adaptive smart antennas will hand off any two co-channel users, whether they are inter-cell or intra-cell, before they get too close, and begin to interfere with each other.

3. Signal-Model and Beamforming Schemes of Smart Antennas

Array signal processing involves the manipulation of signals induced onto the elements of an array. A signal model useful for array processing is presented here, along with various beamforming schemes, adaptive algorithms to adjust the required weighting on antennas, and direction-of-arrival (DoA) estimation methods [4, 7]. Detailed information and an extended reference list can be found in [4], in a very-well-written paper by L. Godara.

3.1 A Signal Model

Consider an array antenna of L isotropic elements, shown in Figure 4. Assume the array to be receiving M uncorrelated sinusoidal point sources, of frequency f_0. A plane wave, radiated from the

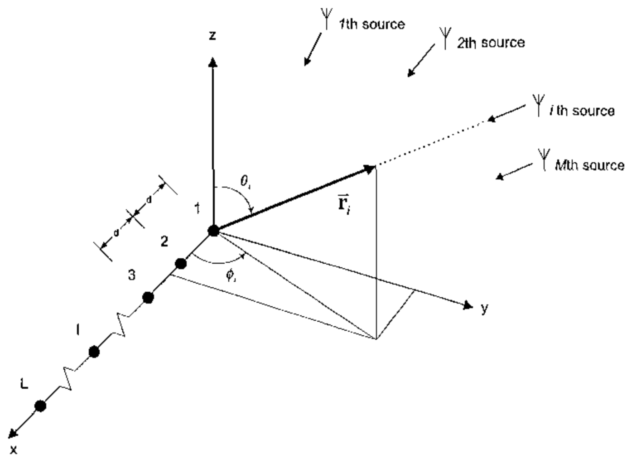

Figure 4. The coordinate system for the signal model.

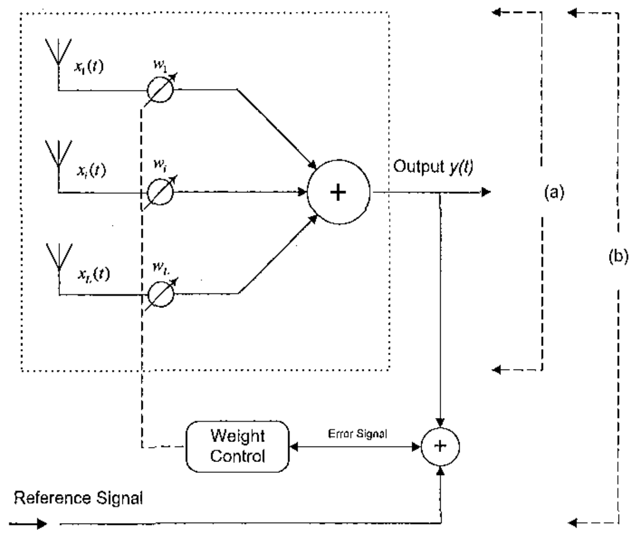

Figure 5. The structure of a narrow-band beamformer (a) without and (b) with a reference signal.

ith source in direction (ϕ_i,θ_i), arrives at the lth element of the array after time

$$\tau_l(\phi_i,\theta_i)=\frac{\vec{\mathbf{r}}_l\bullet\hat{\mathbf{a}}(\phi_i,\theta_i)}{c}, \tag{1}$$

where $\vec{\mathbf{r}}_l$ is the position vector of the lth element, $\hat{\mathbf{a}}(\phi_i,\theta_i)$ is the unit vector in direction (ϕ_i,θ_i), and c is the speed of propagation of the plane wavefront. For a linear equi-spaced array, aligned with the x-axis such that the first element is situated at the origin (Figure 4), this becomes

$$\tau_l(\theta_i)=\frac{d}{c}(l-1)\cos\theta_i, \tag{2}$$

where d is the inter-element distance. The signal induced on the reference element due to the ith source is normally expressed in complex notation as

$$m_i(t)\exp(j2\pi f_0 t), \tag{3}$$

where $m_i(t)$ denotes the complex modulating function.

Assuming that the plane wave on the lth element arrives $\tau_l(\phi_i,\theta_i)$ seconds before it arrives at the reference element, the signal induced on the lth element due to the ith source can be expressed as

$$m_i(t)\exp\{j2\pi f_0[t+\tau_l(\phi_i,\theta_i)]\}. \tag{4}$$

This expression is based upon the narrow-band assumption for array signal processing, which assumes that the bandwidth of the signal is narrow enough for the modulating function to stay almost constant during $\tau_l(\phi_i,\theta_i)$ seconds, that is, the approximation $m_i(t)\cong m_i[t+\tau_l(\phi_i,\theta_i)]$ holds.

Let x_l denote the total signal induced due to all M directional sources and background noise on the lth element. Then, it is given by

$$x_l=\sum_{i=1}^{M}m_i(t)\exp\{j2\pi f_0[t+\tau_l(\phi_i,\theta_i)]\}+n_l(t), \tag{5}$$

where $n_l(t)$ is a random-noise component on the lth element, which includes background noise and electronic noise generated in the lth channel. It is assumed to be temporally white, with zero mean and variance equal to σ_n^2.

Consider a narrow-band beamformer, shown in Figure 5a, where signals from each element are multiplied by a complex weight, and summed to form the array output. It follows from the figure that an expression for the array output can be given by

$$y(t)=\sum_{l=1}^{L}w_l^* x_l(t), \tag{6}$$

where * denotes the complex conjugate.

Denoting the weights of the beamformer as

$$\bar{w}=[w_1,w_2,\cdots,w_L]^T, \tag{7}$$

and the signals induced on all elements as

$$\bar{x}(t)=[x_1(t),x_2(t),\ldots,x_L(t)]^T, \tag{8}$$

the output of the beamformer becomes

$$y(t)=\bar{w}^H\bar{x}(t), \tag{9}$$

where superscripts T and H, respectively, denote the transpose and complex-conjugate transpose of a vector or matrix. $\bar{w}$ and $\bar{x}(t)$ are referred to as the *array weight vector* and *the array signal vector*, respectively.

If the components of $\bar{x}(t)$ can be modeled as zero-mean stationary processes, then for a given $\bar{w}$, the mean output power of the processor is given by

$$P(\bar{w}) = E\left[y(t) \quad y^*(t)\right] = \bar{w}^H R\bar{w}, \tag{10}$$

where $E[\cdot]$ denotes the expectation operator, and R is the array correlation matrix, defined by

$$R = E\left[\bar{x}(t) \quad \bar{x}^H(t)\right]. \tag{11}$$

Elements of this matrix denote the correlation between various elements. For example, R_{ij} denotes the correlation between the ith and the jth elements of the array. Denote the steering vector, associated with the direction (ϕ_i, θ_i) or the ith source by an L-dimensional complex vector $\bar{s}_i$, by

$$\bar{s}_i = \left\{\exp\left[j2\pi f_0\tau_1(\phi_i,\theta_i),\ldots,j2\pi f_0\tau_L(\phi_i,\theta_i)\right]\right\}^T. \tag{12}$$

Algebraic manipulation using Equations (5), (8), and (11) leads to the following expression for R:

$$R = \sum_{i=1}^{M} p_i \bar{s}_i \bar{s}_i^H + \sigma_n^2 I, \tag{13}$$

where I is an identity matrix and p_i denotes the power of the ith source, measured at one of the elements of the array. Using matrix notation, the correlation matrix, R, may be expressed in the following compact form:

$$R = ASA^H + \sigma_n^2 I, \tag{14}$$

where columns of the L by M matrix A are made up of steering vectors, i.e.,

$$A = \left[\bar{s}_1, \bar{s}_2, \ldots, \bar{s}_M\right], \tag{15}$$

and the M by M matrix S denotes the source correlation. For uncorrelated sources, it is a diagonal matrix with

$$S_{ij} = \begin{cases} p_i, & i = j \\ 0, & i \neq j \end{cases}. \tag{16}$$

Sometimes, it is useful to express R in terms of its eigenvalues and their associated eigenvectors. The eigenvalues of R can be divided into two sets, when the environment consists of uncorrelated directional sources and uncorrelated white noise.

Thus, the R matrix of an array of L elements immersed in M directional sources and white noise has M signal eigenvalues and $L-M$ noise eigenvalues.

Denoting the L eigenvalues of R in descending order by λ_l, $l = 1, L$, and their corresponding unit-norm eigenvectors by $\bar{U}_l$, $l = 1, L$, the matrix takes the following form:

$$R = \Sigma\Lambda\Sigma^H, \tag{17}$$

with a diagonal matrix

$$\Lambda = \begin{bmatrix} \lambda_1 & & & & \\ & \cdot & & 0 & \\ & & \lambda_l & & \\ & 0 & & \cdot & \\ & & & & \lambda_L \end{bmatrix}, \tag{18}$$

and

$$\Sigma = \left[\bar{U}_1 \quad \cdots \quad \bar{U}_L\right]. \tag{19}$$

This representation sometimes is referred to as the *spectral decomposition* of R. Using the fact that the eigenvectors form an orthonormal set, this leads to the following expression for R:

$$R = \sum_{l=1}^{M} \lambda_l \bar{U}_l \bar{U}_l^H + \sigma_n^2 I. \tag{20}$$

There are many schemes that can be used to select the weights of the beamformer, as depicted in Figure 5a. Some of these schemes are discussed next.

3.2 Beamforming Schemes

The simplest beamformer has all the weights of equal magnitudes, and is called a *conventional beamformer* or a *delay-and-sum beamformer*. This array has unity response in the look direction, which means that the mean output power of the processor, due to a source in the look direction, is the same as the source power. To steer the array in a particular direction, the phases are selected appropriately. This beamformer provides the maximum output SNR for the case that no directional jammer operating at the same frequency exists, but it is not effective in the presence of directional jammers, intentional or unintentional. A *null-steering beamformer* can cancel a plane wave arriving from a known direction, producing a null in the response pattern in this direction. The process works well for canceling strong interference, and could be repeated for multiple-interference cancellation. But although it is easy to implement for single interference, it becomes cumbersome as the number of interferers grows. Although the beam pattern produced by this beamformer has nulls in the directions of interference, it is not designed to minimize the uncorrelated noise at the array output. This can be achieved by selecting weights that minimize the mean output power, subject to the above constraints.

The *optimal beamformer*, referred also as the *optimal combiner* or *minimum-variance distortionless response beamformer* (MVDR), does not require knowledge of the directions and power levels of the interferers, nor the level of the background noise power, to maximize the output SNR. In this case, the weights are computed assuming all sources as interference, and the processor is referred to as a noise-alone matrix-inverse (NAMI) or maximum-likelihood (ML) filter, as it finds the ML estimate of the power of the signal source with the above assumption. Minimizing the total output noise, while keeping the output signal constant, is the same as maximizing the output SNR. This method requires the number of interferers to be less than or equal to $L-2$, as an array with L elements has $L-1$ degrees of freedom, and one has been utilized by the constraint in the look direction. This may not be true in a mobile-communications environment with multipath arrivals, and the array beamformer may not be able to achieve the maximization of the output SNR by suppressing every interference. However, the beamformer does not have to fully suppress interference, since an increase of a few decibels in the output SNR can make a large increase in the channel capacity.

In the *optimization using reference signal* method, the processor requires a reference signal instead of the desired signal direction (Figure 5). The array output is subtracted from an available reference signal to generate an error signal, which is used to control the weights. Weights are adjusted such that the mean-squared error (MSE) between the array output and the reference signal is minimized. Arrays which use zero reference signals are referred to as *power-inversion adaptive arrays*. The MSE minimization scheme is a closed-loop method, compared to the open-loop scheme of MVDR (the ML filter), and the increased SNR is achieved at the cost of some signal distortion, caused by the filter.

Beam-space processing is a two-stage scheme, where the first stage takes the array signals in input and produces a set of multiple outputs, which are then weighted and combined to produce the array output. The processing done at the first stage is by fixed weighting of the array signals, and amounts to producing multiple beams, steered in different directions. The weights applied to different beam outputs to produce the array outputs are optimized to meet specific optimization criteria, and are adjusted during the adaptation cycle. In general, for an L-element array, a beam-space processor consists of a main beam, steered in the signal direction, and a set of not more than $L-1$ secondary beams. The secondary beams, also known as auxiliary beams, are designed such that they do not contain the desired signal from the look direction, to avoid signal cancellation in the subtraction process. Beam-space processors have been studied under many different names, including the Howells-Applebaum array, generalized sidelobe canceler (GSC), partitioned processor, partially adaptive arrays, post-beamformer interference cancelled (PIC), adaptive-adaptive arrays, and multiple-beam antennas (see references in [4]). These arrays are useful in situations where the number of interferers is much less than the number of elements.

As the signal bandwidth increases, the narrow-band beamformer does not operate well. For processing broadband signals, a tap delay line (TDL) structure, after the steering delay on each channel, is used, forming a finite-impulse-response (FIR) filter. This *broad-band beamforming* scheme has real weights, and is useful when the actual signal direction and the known direction of the signal are not precisely the same. Generally, the performance of broadband arrays depends on a number of various parameters, such as the number of taps, the tap spacing, the array geometry, the array aperture, and the signal bandwidth.

In the *frequency-domain beamforming* scheme, the broadband signals from each element are transformed into the frequency domain using the FFT, and each frequency bin is processed by a narrowband processor structure. The weighted signals from all elements are summed to produce an output at each bin. The weights are selected by independently minimizing the mean output power at each frequency bin, subject to steering-direction constraints. Thus, the weights required for each frequency bin are selected independently, and this selection may be performed in parallel, leading to a faster weight update.

In *digital beamforming*, the weighted signals from each element are sampled and stored, and beams are formed by summing the appropriate samples such that the required delay is incorporated by this process. These exact beams are normally referred to as *synchronous or natural beams*, and it is possible to form a number of these beams simultaneously, using a separate summing network for each beam. The practical requirement of an adequate set of directions where simultaneous beams need to be pointed implies that the array signals must be sampled at much higher rates than required by the Nyquist criterion to reconstruct the waveform back from the samples. The high sampling rate means a large storage requirement, along with high-speed input-output devices, analog-to-digital converters, and large-bandwidth cables.

Finally, the *eigenstructure method* uses the eigenvalues of R. The largest M eigenvalues, which correspond to M directional sources, give the signal eigenvectors, which define the signal subspace. The L-M smallest eigenvalues, which are equal to the background noise power, produce the noise eigenvectors, which define the noise subspace. Since these subspaces are orthogonal to each other, they may be thought of as spanning an L-dimensional space. M steering vectors, associated with M directional sources, also span the signal subspace. An array using a weight vector contained in the signal space, such that it is orthogonal to the interference-direction steering vector, is able to cancel the interference.

3.3 Adaptive Beamforming Algorithms

Since in practical situations, the R and R_N are not available, the estimation of the optimal weights uses the available information, derived from the array output and the array signals. There are many such schemes, which are normally referred to as *adaptive algorithms*.

The *Sample Matrix Inversion (SMI)* Algorithm estimates the array weights by replacing R with its estimate. The estimate of R may be updated when new samples arrive, resulting in a new estimate of the weights. As the number of samples grows, the matrix update approaches its true value, and thus the estimated weights approach the optimal weights [8].

The *Least Mean Square (LMS) Algorithm* is a commonly used algorithm, referred to as the *constrained* or *unconstrained* LMS algorithm, according to whether the weights are subjected to or not subjected to constraints at each iteration, respectively. The unconstrained case is mostly applicable when the update of the weights is done using a reference signal, and no knowledge of the direction of the signal is utilized, as in the constrained case.

The *Constant Modulus Algorithm (CMA)* is a gradient-based algorithm, which is based on the fact that the existence of an interference causes fluctuation in the amplitude of the array output, which otherwise has a constant modulus [9]. CMA is useful for eliminating correlated arrivals, and is effective for constant-modulated envelope signals, such as GMSK and QPSK, which are used in digital communications.

The *Conjugate Gradient Method* converges to the minimum of an error surface within at most L iterations, for an L-rank matrix equation, and provides the fastest convergence of all the iterative methods. The use of this method to eliminate multipath fading in mobile-communications situations shows that the BER performance of the system is better than that using the RLS algorithm [10].

In the *Neural Network* approach [11], a special algorithm referred to as the Madaline Rule III (MRIII) can be used, which is applicable when a reference signal is available. This algorithm minimizes the MSE between the reference signal and the modified array output, rather than the MSE between the reference signal and the array output, as other algorithms do. Despite the fact that the algorithm does not converge in some cases, where local minima exist on the MSE surface of the error signal, the algorithm can be characterized as robust, suitable for analog implementation, and giving fast weight updates.

3.4 Direction-of-Arrival Algorithms

As was shown for the previous beamforming and adaptive-beamforming algorithms, the calculation of the direction-of-arrival (DoA) of a signal or interference is of great importance. There are various DoA-estimation methods [4]. According to some of them, the direction can be estimated from the spatial-frequency spectrum, which can be obtained by using the discrete Fourier transform (DFT) [12], or the Maximum Entropy Method (MEM) [13] for spatially sampled signals. The weight coefficients are updated by the Wiener solution, derived form the estimated spatial spectrum. Moreover, the Multiple Signal Classification (MUSIC) [14] Algorithm estimates DoAs in the noise subspace, which is defined by the eigenvectors of a covariance matrix of spatially sampled signals, while the DFT and MEM do it in the signal subspace. MUSIC has better estimation performance than MEM if the noise subspace is larger for uncorrelated signals than is the signal subspace. These spatial-spectral estimation algorithms can be used to obtain optimum or sub-optimum weights by using spatial samples at one time instant, i.e. one snapshot. Therefore, if the processing speed is fast enough to track the time variation of channels, these algorithms can be more attractive for a fast-fading channel than the temporal updating algorithms. Most recently, neural networks have also been employed to detect the direction of arrival. Once the neural network is trained offline, multiple signals can be tracked in real time [15].

4. Applications in Wireless Systems

The major digital wireless cellular systems in operation today are the Pan-European Global System for Mobile Communications (GSM), and its extension, DCS-1800; the Japanese PDC system, which uses time-division multiple access (TDMA); and the North American IS-95 system, with code-division multiple access (CDMA) [16]. These digital systems offer significant performance and capacity improvements over first-generation mobile systems, which are analog. In all these systems, antenna arrays with spatial processing can provide substantial additional improvement.

Research on adaptive antenna arrays for cellular systems dates from the early- to mid-1980s, but research and development of smart and adaptive antennas for these systems has intensified only in the last few years. In 1995, Nortel introduced smart-antenna technology for PCS-1900 systems [17]. Other companies, such as Metawave [18] and ArrayComm [5], have introduced similar technology, and the European Advanced Communications Technologies and Services (ACTS) TSUNAMI project is considering adaptive antennas for third-generation wireless systems [19]. Commercial products include a four-beam smart antenna, incorporated into a GSM base station, produced by Nortel; adaptive array processing using two base-station antennas, incorporated into an IS-136 base station, produced by Ericsson; adaptive-array antenna technology in enhanced base stations, used by DDI-Pocket Telephone (DDI-P), the world' s largest Personal Handyphone System (PHS) network in Japan, by ArrayComm; etc.

In an IS-136 TDMA system [20], the 14-symbol synchronization sequence, which is present in each time slot for equalizer training, can also be used to determine the adaptive-array weights. However, because with rapid fading the channel can change significantly across a time slot, the adaptive-array weights must be adjusted across the time slot, with recalculation of the weights for each symbol. Since the equalizer is relatively simple, however, joint spatial-temporal processing (i.e., adaptive-array combining with equalization) is practical. This processing is more difficult in GSM/DCS 1800, the other TDMA system, because the equalizer is more complex. Because of the higher data rate in this system, the equalizer must operate with a delay spread over several symbols, and thus it is more complicated than that for IS-136. Fortunately, at typical mobile-radio fading rates, the channel does not change significantly over a time slot, and the equalizer and adaptive-array weights need only be calculated once per frame (a 26-symbol synchronization sequence is present in each time slot). Things are more simple in an IS-95 CDMA system, where a RAKE receiver–which combines delayed versions of the CDMA signal–overcomes the delay-spread problem, and provides diversity gain. The CDMA spreading codes can provide the reference signal for adaptive-array weight calculation.

It has been shown, in [21], that adaptive arrays provide a better range increase than switched-beam (multi-beam) antennas. Since switched-beam antennas require less complexity, particularly with respect to weight/beam tracking, they appear to be preferable for CDMA. In contrast, adaptive arrays are more suitable for TDMA applications, especially in environments with large angular spread.

In CDMA systems [21, 22], the capacity (defined as the bits per second per hertz per base station) depends on the spreading gain, and the corresponding number of equal-power co-channel interferers. Although, adaptive arrays can provide additional interference suppression–by using nulls in the direction of interferers–they do not perform very well when the number of interferers is larger than the number of antennas. Thus, switched-beam antennas are generally preferred in CDMA systems. On the other hand, in TDMA systems, since there are fewer interferers, adaptive arrays can cancel the dominant interferers with just a few antennas. An M-element array has the potential to permit greater than an M-fold increase in capacity (independent of the angular spread). Computer-simulation results indicate that a four-element adaptive array can permit frequency reuse in every cell (in a three-sector system), for a sevenfold increase in capacity over current systems. A four-beam (switched-beam) antenna can permit a reuse of three or/ four, for a doubling of capacity. The above adaptive-array results apply to the uplink, only. For the downlink, switched-beam antennas can be used at the base station, in combination with adaptive arrays on the uplink.

The problem with switched-beam arrays is even worse in IS-136, since the handsets require a continuous downlink, and therefore the same beam pattern must be used for all three users in a channel. This fact further reduces the effectiveness of switched-beam antennas against interference. Therefore, TDMA systems may require multiple antennas on the handset to achieve high frequency reuse. Interference, however, is generally worse on the uplink than on the downlink, for two reasons. First, it is possible that the signal from an interfering mobile could be stronger than that from the desired mobile at the base station. In contrast, at the mobile, the signal from an interfering base station should not be stronger, since the mobile chooses the base station with the strongest signal. Second, base stations are more uniformly spaced near the center of cells than are the mobiles. Thus, more interference suppression on the uplink than on the downlink may be desirable.

5. Conclusions

The advantages and disadvantages of the two main categories of smart antennas–switched-beams antennas and adaptive arrays–were presented and discussed. A basic model for determining the angle of arrival for incoming signals, the appropriate antenna beamforming, and the adaptive algorithms that are currently used for array processing, were included, as well. Moreover, it was shown how smart antennas, with spatial processing, can provide substantial additional improvement, when used with the TDMA and CDMA digital-communication systems.

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Introducing the Author

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