Adv. Radio Sci., 4, 185–188, 2006
www.adv-radio-sci.net/4/185/2006/

Advances in
Radio Science

Hardware implementation of smart antenna systems

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Abstract. Smart antenna systems attract a lot attentions now and believably more in the future, as it can increase the capacity of mobile communication systems dramatically. Design of smart antenna systems combines the technologies of antenna design, signal processing, and hardware implementation. In this paper, a propose of smart antenna structure, as well as some function blocks that have been already implemented in hardware will be presented.

1 Introduction

Traditional base station antennas are omni-directional, this is actually a waste of power because most of it will be transmitted in other directions than toward the desired user. In addition, other users will experience the power radiated in other directions as interference. A promising technique to increase the spectrum efficiency is using smart antennas. This technique adds a new way of separating users on one base station by space, so called SDMA (Spatial Division Multiple Access), as well as provides increased range, improved link quality, higher level of security, etc. (Litva, 1996) (Appelbaum, 1976) Several projects and field trials have emerged reinforcing the benefits of smart antenna systems. (Bellofiore and Balanis, 2002)

This paper will be presented in this way: Sect. 2 gives readers an overview of how a smart antenna system works, Sect. 3 concentrates on the DOA algorithms, Sect. 4 introduces adaptive beamforming algorithms, Sect. 5 shows some hardware implementation items, and there is a conclusion at the end.

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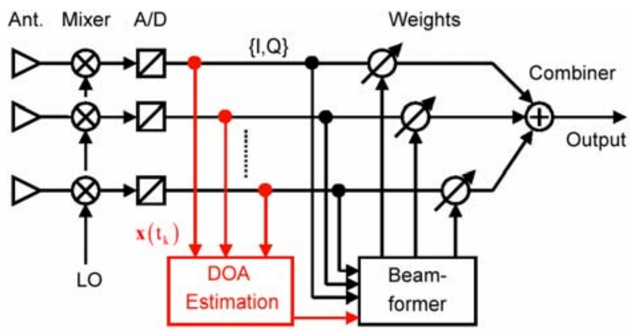

Fig. 1. Functional block diagram of a Smart Antenna System (Jacob, 2003).

2 System overview

Electrical smart antenna systems work in the following way: After the digital signal processor receives signals collected from each antenna element, it computes the direction-of-arrival (DOA) of the signal of interest (SOI). It then uses adaptive beamforming algorithms to produce a radiation pattern that focuses on the SOI, while tuning out any signal not of interest (SNOI). Figure 1 shows the functional block diagram of a smart antenna system.

3 DOA estimation algorithms

As shown in Fig. 1, the first objective of DSP block is to estimate the DOAs of all impinging signals from the time delays of each antenna element.

Before we introduce the details of algorithms, let's do some assumptions and formulation to simplify our deduction.

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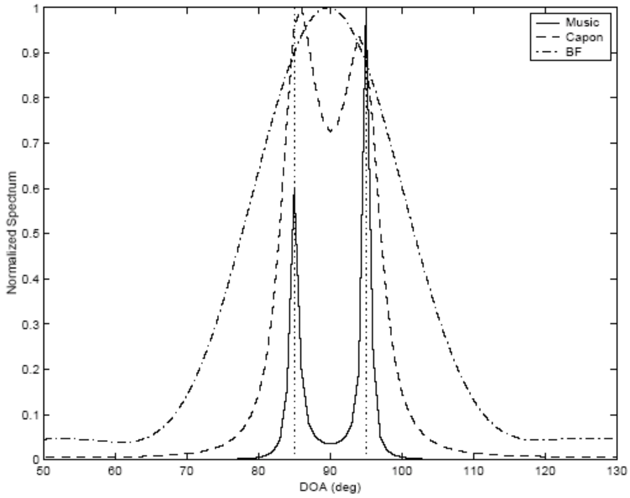

Fig. 2. Normalized spectra of MUSIC (solid), Capon (dashed), and beamforming (dash-dotted) methods. The true DOAs are indicated by dotted vertical lines. An ULA of 5 elements is used with half-wavelength spacing; N = 100 samples; SNR is 15 dB.

In narrowband condition, If M signals impinge on an L-dimensional array from distinct DOAs $\theta_1, \theta_2, \ldots, \theta_M$, the array model is given by

$$x(t) = A(\theta)s(t) + n(t) \tag{1}$$

Where $\mathrm{A}(\theta) = [a(\theta_1), \mathrm{a}(\theta_2), \ldots, a(\theta_M)]$ is the steering matrix; $s(t) = [s_1(t), s_2(t), \ldots, s_M(t)]^T$ denotes the baseband signal waveforms, and $n(t)$ is spatially white noise. All the methods in this paper require $M < L$. Then we can get the spatial covariance matrix:

$$\begin{aligned} R &= E\{x(t)x^H(t)\} \\ &= AE\{s(t)s^H(t)\}A^H + E\{n(t)n^H(t)\} \\ &= \mathrm{ASA}^H + \sigma^2 I \end{aligned} \tag{2}$$

where $S = E\{s(t)s^H(t)\}$ is the source covariance matrix; $E\{n(t)n^H(t)\} = \sigma^2 I$ is the noise covariance matrix that is a reflection of the noise having a common variance σ^2 at all sensors and being uncorrelated among all sensors.

3.1 Conventional beamformer

Conventional beamformer algorithm is a classic DOA method, which scans the beam to evaluate the received power in each direction and to find the signal DOAs from the maxima of the array output.

3.2 Capon beamformer

In order to alleviate the limitations of conventional beamformer, such as its resolving power of two sources spaced closer than a beam width, researchers have proposed numerous modifications. A well-known method was proposed by Capon (1969). The key idea is to steer array toward a particular DOA θ and reject the signals at all remaining directions.

3.3 Subspace-based algorithms

Subspace-based algorithms, like MUSIC, ESPRIT, Root-Music, Min-Norm, etc., are often used today. They are all high-resolution DOA algorithms, where the eigen-structure of the covariance matrix is explicitly invoked. Here we focus on MUltiple SIgnal Classification (MUSIC) algorithm, as later we will choose it for hardware implementation because of its simplicity. MUSIC is the first subspace-based DOA estimation approach (Schmit, 1986). It can be summarized in 5 Steps:

Step 1. Collect data and form the spatial covariance matrix R;
Step 2. Calculate eigenstructure of R;
Step 3. Evaluate $\mathrm{f_{MUSIC}}(\theta)$ versus θ;
Step 4. Pick M peaks of $\mathrm{f_{MUSIC}}(\theta)$;
Step 5. Calculate remaining parameters.

Recalling Eq. (2), the positivity guarantees the following representation

$$R = \mathrm{ASA}^H + \sigma^2 I = U\Lambda U^H \tag{3}$$

With U unitary and $\Lambda = diag\{\lambda_1, \lambda_2, \ldots, \lambda_L\}$, a diagonal matrix of real eigenvalues ordered such that $\lambda_1 \geq \lambda_2 \geq \ldots \geq \lambda_L > 0$. Any vector orthogonal to A is an eigenvector of R with the eigenvalue σ^2. There are L–M such linearly independent vectors. Since the remaining eigenvalues are all larger than σ^2, we can partition the eigenvalue-vector pairs into noise eigenvectors (corresponding to eigenvalues $\lambda_{M+1} = \ldots = \lambda_L = \sigma^2$) and signal eigenvectors (corresponding to eigenvalues $\lambda_1 \not< \ldots \not< \lambda_M \not< \sigma^2$). Hence, we can write

$$R = U_s \Lambda_s U_s^H + U_n \Lambda_n U_n^H \tag{4}$$

Where $\Lambda_n = \sigma^2\mathrm{I}$. Assuming ASA^H to be of full rank, the diagonal matrix Λ_s contains the M largest eigenvalues. Since the eigenvectors in U_n are orthogonal to A,

$$U_n^H a(\theta_i) = 0, \theta_i \in \{\theta_1, \ldots, \theta_M\} \tag{5}$$

The MUSIC "spatial spectrum" is then defined as

$$f_{\mathrm{MUSIC}} = \frac{1}{a^H(\theta)\,\hat{U}_n\,\hat{U}_n^H\,a(\theta)} \tag{6}$$

Figure 2 shows some simulation results by comparing different DOA algorithms (Krim and Viberg, 1996).

4 Adaptive beamforming

In adaptive beamforming, the goal is to adapt the beam by adjusting the amplitudes and phases of signals such that a desirable pattern is formed. Algorithms for adaptive beamforming have departed from classical Least mean square (LMS) toward more sophisticated constant modulus algorithms and eigen-projection algorithms. But in this paper we still focus on the LMS algorithm, as it's easy to understand and implement later.

Most popular criteria of adaptive beamforming are either Minimum of the Mean Square Error (MSE), or Maximum of Signal-to-Interference-plus-Noise-Ratio (SINR). Aiming to Minimize the MSE, we can get famous Wiener-Hopf equation. $\boldsymbol{x}(k)$ is a vector from each antenna array element at time step k,

$$\boldsymbol{x}(k) = \left[\, x(k)_1 \; x(k)_2 \; ... \; x(k)_L \,\right]^T$$

respectively, we also get a weight vector $\boldsymbol{w}$

$$\boldsymbol{w} = \left[\, w_1 \; w_2 \; ... \; w_L \,\right]$$

Then the output $y(k)$ for time step k is

$$y(k) = w^H x(k) \tag{7}$$

By comparing the desired response $d(k)$ with the output $y(k)$, we produce an estimation error denoted by

$$e(k) = d(k) - y(k) \tag{8}$$

Then the MSE can be defined as

$$\begin{aligned} \mathrm{MSE} &= E\{|e|^2\} = E\{|d(k) - w^H x(k)|^2\} \\ &= E\{(d(k) - w^H x(k))(d^*(k) - x^H(k)w)\} \\ &= E\{|d|^2\} - w^H p - p^H w + w^H R w \end{aligned} \tag{9}$$

where $\boldsymbol{p} = E\{x(k)d^*(k)\}$ is the cross correlation vector; $\mathbf{R}$ is the spatial variance matrix. By solving

$$\frac{\partial \mathrm{MSE}}{\partial w} = 0 \tag{10}$$

we obtain the optimal weight vector

$$\boldsymbol{w}_{opt} = R^{-1} p \tag{11}$$

If transfer function of the channel is static, we can calculate the optimal weight vector and set the weight vector in advance. But most of the channel transfer functions vary with time and are therefore nonstatic. For such nonstatic channels, the calculation of the optimal weight vector must be repeated after a time that has a lot disadvantages. The method of steepest descent is then used in practice. It's a recursive method for finding the minimum point of the error-performance surface without knowledge of the error-performance surface itself. At first, we start with an arbitrary value of the weight vector $\boldsymbol{w}(0)$, typically the zero-vector. Second, we compute the gradient vector with respect to the actual weight vector $\boldsymbol{w}(k)$. Third, we change the values of the actual weight vector by a constant parameter μ in negative direction of the gradient vector. The mathematical representation is

$$w(k+1) = w(k) + \mu \left(-\frac{\partial \mathrm{MSE}}{\partial w} \right) \tag{12}$$

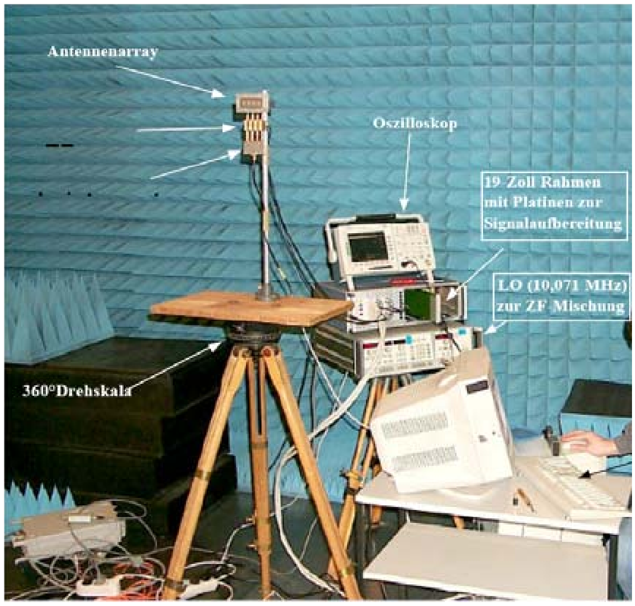

Fig. 3. Structure of the Receiver.

Combining Eqs. (9) and (12) leads to the LMS algorithm

$$w(k+1) = w(k) + 2\mu e^*(k) x(k) \tag{13}$$

(Van Veen and Buckley, 1988)

5 Hardware implementation

5.1 Receiver for the measurement

A substantial component presented here is the structure of hardware of a receiver.

The system (shown in Fig. 3) starts with a 4-element linear antenna array. The received signals are converted from 10 GHz high-frequency into the intermediate frequency 71 MHz and transferred finally through I/Q demodulation to the baseband. After A/D conversion, the measured data can be used for the offline DOA analysis through a PC.

5.2 FPGA-based MUSIC Estimator

One big drawback of using smart antenna systems is the high computing cost. So a powerful processor is necessary. The researchers now focus on either using DSP or FPGA. We compared the advantages and disadvantages of both, and chose FPGA for our implementation, mainly because the consuming time of FPGA-based design is much shorter. It is important for a real time control.

The whole implementation can be divided into 4 steps: (1) Estimation of correlation matrix R including unitary transform. (2) Eigenvalue Decomposition of R, by using Cyclic Jacobi processor based on CORDIC (COordinate Rotation Digital Computer). (3) Computation of MUSIC spectrum, where FFT for spatial DFT is used. (4) 1-dimensional peak search.(Kim et al., 2004)

We get input data from an offline PC: e.g. 4-element ULA antenna array, 2 incoming signals at -15 and 20 degrees with same power, SNR=10db; FPGA device: Xilinx XC2VP30. The estimation result is -14.97 and 19.15 degree. The total consuming time is less than 30 μs.

6 Conclusions

In this paper, the background of smart antenna systems is introduced, including the overview of the whole system, DOA algorithms, and adaptive beamforming algorithms. A hardware structure of a receiver was developed, as well as a FPGA-based MUSIC DOA Estimator.

The future work will focus on replacing the offline PC in receiver measurement system with the FPGA-based MUSIC DOA Estimator, adding the adaptive beamforer, and completing the whole real time controlling test bed of a smart antenna system.

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