SIGNAL PROCESSING IV: Theories and Applications
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Elsevier Science Publishers B.V. (North-Holland)

MULTIBEAM ADAPTIVE ARRAY

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To achieve adaptive multibeamforming to allow reconfiguration in SSW-TDMA satellite communications, a system for narrow band adaptive multibeamforming is described in this work. The signal processor allows an unconstrained array to form independent beams for the impinging directions without any prior knowledge of them. The simultaneous use of constant modulus and minimum variance (least-mean square loops) is done by an especial policy of master-slave for subarrays or local beams which acts both in acquisition and tracking. The results of simulations confirm the theoretical work.

I INTRODUCTION

In satellite digital communication systems, arrays of sensor elements are often used in reception doing a beamforming that steers towards the direction of signal arrival and attenuates others directions.

In movil communications the directions of arrival at the reception systems varies, so it's necessary an array tracking the produced changes continuously. in this work a digital processing system is developed and used to receive simultaneously signals proceeding from different directions. A multibeam antenna is continuously adapted to detect CPM /1,2/, MPSK signals using a subarray divided-system. Since these signals are constant envelope modulated, the adaptive constant-modulus algorithm (CMA) /4,5/, provides the framework to the receiver array digital beamforming. It avoids the necessity of reference or pilot signal to catch and track the possible signals coming from desired sources, as happen with other methods /3/.

Since every subarray steers a CM-signal if any other restrictions are imposed, all the subarrays will probably steer at the same source. To avoid this problema master-slave policy is introduced between subarrays, that makes that every subarray catches a different signal. This is possible orthogonalitating the outputs of the subarrays. This is made using a minimum square algorithm (LMS) /6/.

The resultant beamforming system has two adaptive filtering algorithms working with two objetives; The CMA catch CM-signals and the LMS algorithm does a master-slave policy for subarrays.

In section II a theoretical description of all the system is presented, explaining every algorithm used (CMA and LMS). Furthermore, it's described the technic used for spatial filtering, optionally introduced to discriminate coherent sources spatially localitated in different places /7/. Section III presents the results obtained with the simulation of the system acting in real scenes. The behaviour of the system is as the expected from the theoretical analysis. Finally in section IV the main conclusions extracted of this work, are exposed.

This work is supported by CAYCIT grant number 21096/84 (Spain).

II ADAPTIVE BEAMFORMING BY MEANS OF A SUBARRAY-SYSTEM

The system we will describe, appears as the solution to the following problem: it's desired to detect and discriminate different spatial directions which usually are unknown and varying with time. The first developed step is the division of an array into some subarrays; the object is to obtain the radiation diagram of any one of them, with his main lobe steering to different signals. If there are more subarrays than CM-signals, the subarrays in excess have andirective diagrams. The weights of every subarray are adapted by the CMA, first to catch and then to track all the possible CM-signals.

II.1 Constant modulus algorithm (CMA)

Since the desired signals in the system are exponentially modulated (MPSK, CPM, TCM..), and so, they have constant envelope, the CMA presented by Treichler /4/, to adapt the weights of sensors is the best one to be used in such scenes. Here, this algorithm is applied to the weights of every subarray, minimizing the envelope error between the received signal at the output of a subarray and the desired one. The output of the ith subarray in the kth sampling instant is

$$y_i(k) = \underline{X}^T(k).\underline{W}_i(k) \quad (1)$$

where $\underline{X}^T(k)$ is the signal vector and $\underline{W}(k)$ the weight vector of the subarray.

The envelope error to be minimized is

$$J = 1/4.E\{(|y(k)|^2 - M^2)^2\} \quad (2)$$

"M" is the modulus of the desired signal.

The CMA is an steepest descent algorithm that minimizes the gradient of (2). This gradient is:

$$\underline{V}_w J = E\{(|y(k)|^2 - M^2)y(k).\underline{X}^*(k)\} \quad (3)$$

In the adaptive algorithm, (3) is substituted by an estimation of $V_w J$, where the statistic operator is replaced with the instantaneous value of the variables. Then the adaptation equation for the vector W(k) is

$$\underline{W}(k+1)=\underline{W}(k) - \mu(|y(k)|^2 - M^2)y(k)\underline{X}^*(k) \quad (4)$$

"μ" is a parameter to control convergence and the minim envelope error achieved.

The characteristics of CMA can be seen in /4,-5/. It's not necessary to know previously, the localization of the sources (arrival directions of CM-signals). A subarray is initialized with a steering vector (for instance, an omnidirectional vector) and by means of the CMA, it steers a CM-signal.

But with the described system of an array subdivided into subarrays adapted by CMA, there doesn't exist any control which CM-signal is catched by every subarray. In the next section, a master-slave policy is introduced to avoid that a signal may be catched by more than one subarray.

II.2 LMS algorithm in a master-slave policy

The LMS steepest descent algorithm is introduced in the preceeding system. It hierarchically orthogonalizes the outputs of all the subarrays. The first subarray of the system steers a CM-signal source. Its output is properly weighted and subtracted to the second subarray output. If this subarray steers the same signal, the gain after the subtraction must have a null in the direction of the source. With the LMS algorithm, is obtained the weighting factor of the first subarray (MASTER) to be subtracted to the output of the second subarray (SLAVE). Figure 1 shows a two subarray-system. "G" is the pondering factor of the first subarray mastering the second.

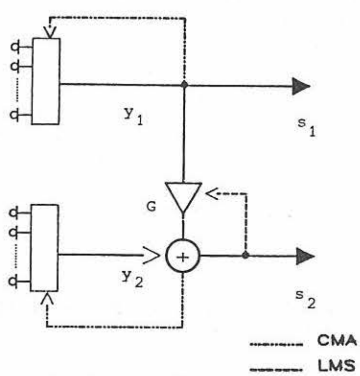

Figure 1. Two-subarray system.

To get the proposed objectives there is a minimum power loop (LMS) managed by "G". $y_i(k)$ is the output of the i-th subarray and $s_i(k)$ is the resultant output of the i-th subarray after the application of the minimum power loop. Then

$$s_1(k) = y_1(k) \quad (5\text{-a})$$

$$s_2(k) = y_2(k) - G.s_1(k) \quad (5\text{-b})$$

To study better the system, we suppose it receiving two signals in directions of polar coordinates (θ_1, ϕ_1) and (θ_2, ϕ_2) at frequencies w_1 and w_2 respectively. With $F_i(\theta_i, \phi_i)$ we design the response of i-th subarray to direction (θ_i, ϕ_i). Then the first subarray has an output $s_1(k)$.

$$s_1(k)=F_1(\theta_1,\phi_1)\exp(jw_1k) + F_1(\theta_2,\phi_2)\exp(jw_2k) \quad (6)$$

the steering vector of the first subarray is adapted by CMA. If it steers the first signal, it cancels the second CM-signal.

$$F_1(\theta_2,\phi_2) \cong 0 \quad (7)$$

Considering (7), the second subarray has a resultant output $s_2(k)$

$$s_2(k) = F_2(\theta_1,\phi_1)\exp(jw_1k) + F_2(\theta_2,\phi_2)\exp(jw_2k) - G(k).F_1(\theta_1,\phi_1)\exp(jw_1k) \quad (8)$$

and it must cancel the first signal steered by the master subarray in direction (θ_1, ϕ_1).

$$F_2(\theta_1,\phi_1) - G(k).F_1(\theta_1,\phi_1) = 0 \quad (9)$$

Then to the signal $y_2(k)$ in (eq.5-b), we are subtracting the components that it receives and which are also present in the output master. Joining (9) and (10)

$$s_2(k) = F_2(\theta_2,\phi_2)\exp(jw_2k) \quad (10)$$

The resultant outputs of the master subarray and the slave subarray are orthogonalized. $s_2(k)$ is going to be minimized by the loop G(k). This only has sense if the first subarray has converged and is steering a CM-signal. To obtain G(k), the output power of the slave subarray is minimized.

$$\text{MIN: } P = |s_2(k)|^2 = y_2(k) - G(k).s_1(k)^2 \quad (11)$$

the gradient of the object of minimization is

$$V(k) = -2s_1^*(k)(y_2(k) - G(k).s_1(k)) \quad (12\text{-a})$$

$$V(k) = -2s_1^*(k)s_2(k) \quad (12\text{-b})$$

Equalling this gradient to zero, implies that the two subarray outputs are spatially orthogonalized.

Since the directions of arrival are unknowns and possibly time varying, an adaptive solution for G(k) is proposed.

$$G(k+1) = G(k) - \mu V(P) \qquad (13\text{-a})$$

$$G(k+1) = G(k) + \mu s_1^*(k) \cdot s_2(k) \qquad (13\text{-b})$$

This solution minimizes (11) and follows the possible time variations. When the system reaches stability, the first CM-signal catched by the master subarray, doesn't appear in the signal $s_2(k)$; the CMA in the slave subarray gets a null for the first signal producing (9)

$$F_2(\theta_1, \phi_1) = 0 \quad \text{--->} \quad G(k)=0 \qquad (14)$$

The factor "G" when the solution converges, tends to zero and the slave subarray has a null where the master one has the maximum.

A complete study of a two subarray system has been presented. The generalization to a larger number of subarrays is obtained considering that the output of every subarray is mastered by all the previous ones, to be applied the minimum power objective. The maximum number of CM-signal that can be steered is the number of subarrays. With the introduction of minimum power loops between subarrays and a master-slave policy, the system converges towards a solution where every subarray steers a different CM-signal. If there are more subarrays than CM-signals the extra subarrays only receive omnidirectional noise. The ambiguity of II.1 has been solved.

II.3 Discrimination of coherent signals

If the system studied in II.2 doesn't receive completely coherent and spatially different signals, it will be necessary a correction in the method. Only if surprisingly the system receives two or more completely coherent signals, the system cannot discriminate them /7/. This is because the received vectors of coherent sources in a subarray, produce constant delays between snapshots and the CMA steers them as a single source. /7,8,9/.

A solution to this problem is presented by Shan an Kailath in /7/ by means of the introduction of a spatial filtering in every snapshot. Every signal-vector is subdivided in some consecutive subvectors with a sample or sensor-signal different in every new subvector. Then in a subarray of N sensors, taking subvectors of k samples, N-k+1 adaptations are done with every snapshot.

Including this spatial filtering to the subarray system presented in II.2, as a lateral option, an adaptive multibeam forming is achieved which operates optimally in a general and real scene.

III SIMULATIONS

To examine the performance of the suggested adaptive beamformer, several computer simulations in real scenes have been carried out with results that support the theoretical predictions.

In the following, we present two proves developed with narrow band-systems centered at 100 MHz. In the first example, a system with three linear subarrays has been simulated. Every subarray has six sensor-elements. In the scene there were two sources at directions 15 gr. and -5gr. respect to the broadside direction, at frequencies 96 and 98 MHz. respectively and with the first one being 1 dB above the second one. There always was additive noise with a power of 10 dB below the signals. Figure 2 shows the radiation diagrams of the three subarrays. The first and the second steer a signal and crossly cancel the other and the third subarray cancels the two signals. Figure 3 shows the learning curves of CMA for the three subarrays. The third subarray has the highest error, since it doesn't catch any CM-signal. The minimum error achieved an the convergence parameters can be varied always controlled by the " " parameter.

In the second example, it has been created a scene with two coherent signals at frequency 100 MHz. arriving from directions of angles 15 gr. and -5 gr. respect the broadside. The preceeding system has been proved with and without spatial filtering. If the spatial filtering isn't used, the master subarray doesn't discriminate the two signals. With the spatial filtering, the proposed objectives are reached since the coherency between signals is broken. Now the conditions to the second subarray are harder, nevertheless it cancels the signal catched by the master subarray and steers the other.

The preceeding results vary the system configuration (convergence parameters, number of sensors...) and with the scene (number of signal), but always support the theoretical study.

IV CONCLUSIONS

In this paper an adaptive beamforming array system has been presented with the objectives of discriminating and processing several signals coming from different spatial directions. The system resolves efficiently several CM-signals without prior knowledge of their directions of arrival, by means of subdivision in subarrays and a signal processing involving two adaptive steepest descent algorithms. It can catch and track as many signals as subarrays are used. The system is adequate to work in scenes of digital communications, where it is often needed to receive simultaneously several informations from unknown and possibly varying directions. With a spatial smoothing the system works efficiently even with coherent signals.

Many proves have been made in real scenes with noised characteristics and different system dimensions and the results obtained have always been the desired ones. Although the simulated proves had been made with narrow-band systems, the procedure can be easily extended to a wide-band systems.

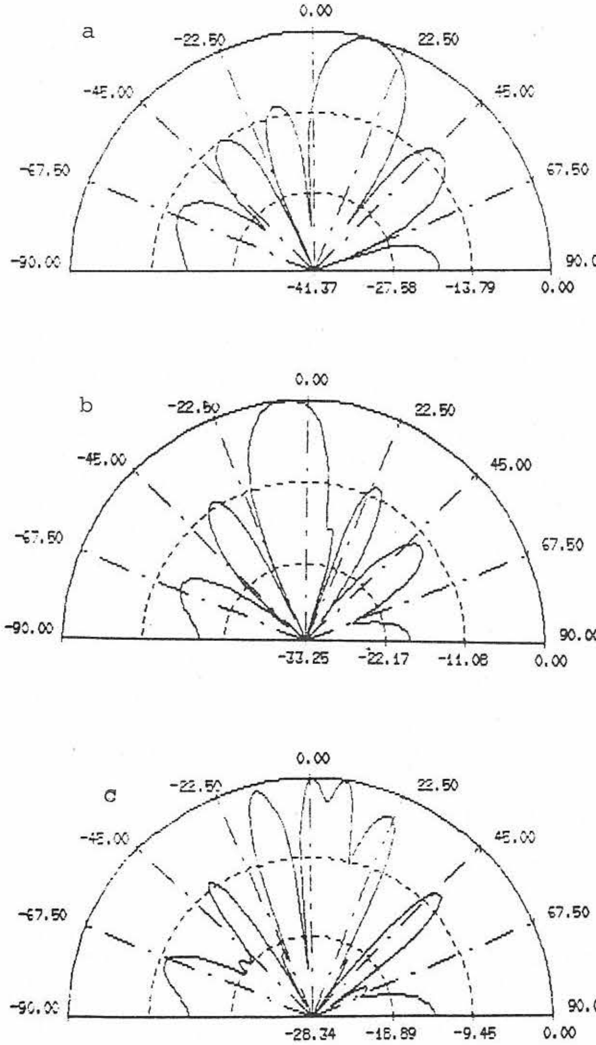

Figure 2. Prove 1, Radiation Patterns a,b,c Subarray 1,2,3.

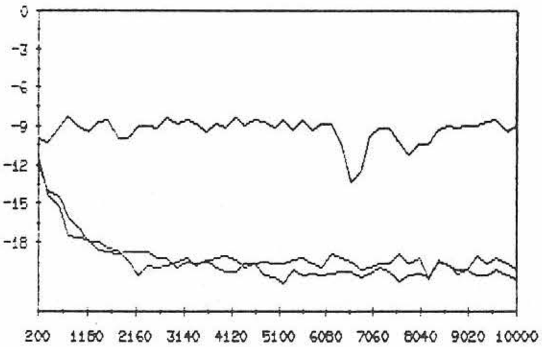

Figure 3. Prove 1. Learning curves.

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