A SIMPLE DIGITAL ADAPTIVE ANTENNA BASED ON UNDERSAMPLING

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Abstract

A simple adaptive antenna based on the pilot signal method is described in the presented paper. The system consists of coherent mixers, which shift RF signals into IF band, of sampling amplifiers and A/D converters, which digitalize signals, and of PC, which performs the adaptive control. Functionality of the system is verified by a simple experiment.

Keywords

adaptive antenna, pilot signal, undersampling, LMS, simplified Kalman filter

1. Introduction

An adaptive antenna is a system, which automatically sets minims of its directivity pattern to directions of the interferences' arrival on one hand and which does not change its properties in the direction of the desired signal's arrival on the other hand. By this way, signal to interference ratio can be maximized at the antenna output.

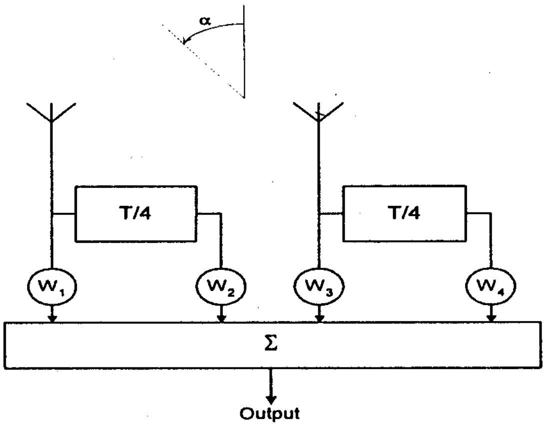

Fig. 1 A simple adaptive antenna

The simplest adaptive antenna (fig.1) consist of two antenna elements. The output of each element is divided into two branches - direct and quadrature ones. In both the branches, signals are weighted (amplified or attenuated) and by this way, both the amplitude and the phase can be steered at the output of the antenna element. By the convenient setting of weights, minims of the directivity pattern can be set to directions from which the most powerful interference come and hence, power of interference signals can be minimized at the antenna output. At the same time, desired signal can be received authentically.

In most cases, adaptive antennas are based on the pilot signal or the steering vector methods [1], [2].

The pilot signal method comes from an exactly defined signal (pilot), which is known both on transmitting and receiving sides. If the pilot is transmitted then the difference between the pilot and the signal at the output of the adaptive antenna is proportional to the power of received interferences. Changing weights, the difference signal can be minimized, and hence, minims of the directivity pattern can be set to directions from which the most powerful interferences come [1].

The need of transmitting pilot signal is disadvantage of this method. On the other hand, the method exhibits very low complexity of algorithms for computing proper weights.

The steering vector method is based on the minimization of power at the antenna output. The minimization is constrained by the condition that the directivity pattern is not changed in the main lobe direction from which the desired signal comes [2].

The steering vector method has removed the need of transmitting the pilot signal but the computational complexity of adaptive algorithms has significantly increased.

If a simple system for experimental purposes is going to be realized then the pilot method seems to be a better choice because of its simplicity.

More, as much as possible functional blocks of the antenna are suitable to be shifted from the hardware to the software part of the system because of simpler *debugging* the system.

Hence, a simple digital adaptive antenna array has been proposed. The system consists of coherent mixers which shift RF signals into IF band without the RF phase distortion, of sampling amplifiers and AD converters, which digitalize signals, and of PC, which performs adaptive weighting of signals and computing the output signal of the antenna. More detailed description of the system can be found in section 2 of this paper.

Section 3 describes the simplification of the proposed adaptive antenna so as the efficient experimental verification of used principles can be done. In the last section, experimental results are discussed.

2. System Description

An antenna array is the basic block of the adaptive antenna. The antenna array has to be design so as it can have good directivity pattern (narrow main lobe, low level of sidelobes) and so as it can provide enough degrees of freedom for the adaptive control (the antenna can eliminate interferences coming from N - 1 directions where N is number of controllable antenna elements). In the real world, half-wavelength or less spaced dipole, horn or microstrip arrays are usually used [3],[4].

Each antenna element is completed by a coherent mixer at its output to convert RF signals to IF ones preserving information about the RF phase of signals. As proven in [5], the desired conversion can be implemented by the use of the narrow-band amplitude modulation for the transmission of information and by the use of a two-gate FET mixer [6] for the coherent mixing. Mixers should be identical in the ideal case because differences in their phase characteristics introduce errors to the information about the directions of interferences' arrival.

In the next step, the IF signal is assumed being of such a low frequency so as a sampling amplifier can be directly used for discretizing signals. E.g., the ultra high-speed monolithic track-and-hold amplifier AD9100 (Analog Devices), which has been designed for the direct IF sampling, exhibits the acquisition time about 20 ns and hence, signals on frequencies about 1 MHz can be directly sampled in our application [7].

The sampling amplifiers at all the IF outputs of antenna elements have to work synchronously so as the IF signals were not distorted by the additional phase error due to time delays in sampling. As discussed in [8], small phase errors can be eliminated by the adaptive process but the higher phase errors make parameters of the system significantly worse.

If the bandwidth of the modulated signal $B_s=F_m$ - F_d (F_d is the lowest and F_m is the highest frequency of the modulated signal) meets the condition

$$B_s = F_m - F_d < F_d, \tag{1}$$

then it can be sampled with the sampling period longer than this given by the Shannon's law without appearing the aliasing (so called undersampling). The sampling frequency is then given by the relation [9]

$$\frac{2F_m}{k_c+1} \le F_S \le \frac{2F_d}{k_c} \tag{2}$$

where

$$k_c = \left[\frac{F_d}{B_s}\right] \tag{3}$$

The undersampling produces a signal, in the spectrum of which the spectrum of the original signal periodically repeats without any overlapping. Hence, there is no problem to select the spectrum of the original signal from such a spectrum.

The outputs of the sampling amplifiers are connected to a PC through an A/D converter. The undersampled digital data are then used for the adaptive computing of optimal weights and for weighting signals of the antenna elements. In the next step, the weighted signals are summed and processed by a digital low-pass FIR filter in the PC to obtaining the desired signal.

Functionality of the described system has been verified through the PSPICE model during the development stage. Then, we have tried to arrange a very cheap experiment to do these verifications in the laboratory.

3. Experimental Verifications

Arranging the entire experiment, antennas and generators for transmitting the desired signal and interferences have to be considered to model the electromagnetic environment to which the adaptive antenna is placed. Dealing with the adaptive system, a receiving antenna array, coherent mixers with identical phase characteristics completed by high-speed sampling amplifiers and A/D converters and a powerful computer for the adaptive signal processing have to be at the disposal. Hence, the system would be rather complex and expensive.

That is why the experiment has been modified to keep costs of the system as low as possible. On the other hand, simplifications should not influence the function of the system so as all the thesis from the above section can be proven.

Hence, an antenna array consisting of two omnidirectional antenna elements spaced d has been simulated. The desired signal has been modelled to come from the main lobe direction and the interference from the direction declining of the angle α. It means, if plane waves are supposed then the desired signal is of the same phase on both the elements and the interference exhibits a phase shift $+\Delta\varphi$ on the right antenna element and $-\Delta\varphi$ on the left one. The phase shift $\Delta\varphi$ is given by

$$\Delta\varphi = \frac{1}{2} kd \sin\alpha \tag{4}$$

where k is wave-number of the free-space surrounding the receiving antenna, d denotes the distance between antenna elements and α is the angle of the interference's arrival.

The described antenna array including the electromagnetic environment and coherent mixers is modelled by analog circuits. Outputs of these circuits go through the A/D converters to the PC where the regular adaptive processing and filtering the desired signal is performed (fig. 2).

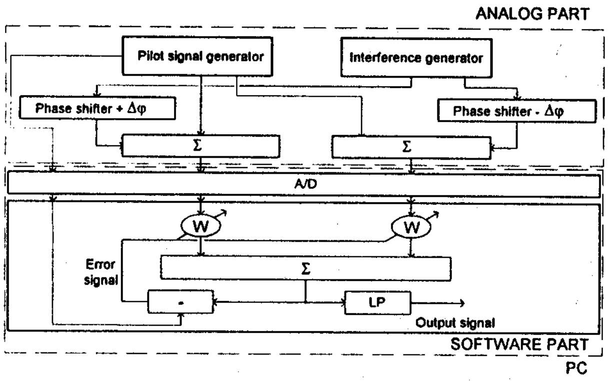

Fig. 2 Block diagram of the experiment

The analog part of the experiment has been designed for the carrier 30 kHz and for the modulation frequency 300 Hz so as the existing laboratory equipment can be used. The AM signal consisting of the above described carrier and modulation signal models outputs of coherent mixers. The desired signal has the same phase at both the outputs and phase of the interference is computed by (4) and set by opamp-based phase shifters [10].

The modelled outputs of coherent mixers are connected to the A/D converter, where signals are both sampled and converted to the digital form. Both the sampling and the A/D conversion has been performed by the A/D card PCA-1208 (Tedia Ltd.) consisting of the multiplexer associating 8 inputs (two of them have been used for two antenna elements, one for the pilot and the other have not been explored) and of the 12-bit A/D converter with the conversion time 12 μs.

Afterwards, obtained samples have been proceeded by the adaptive algorithm in the PC. For this purpose, the classical LMS algorithm [1]

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mu e(n)\mathbf{X}(n) \tag{5}$$

and the simplified Kalman filter [11]

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mathbf{K}(n)e(n) \tag{6a}$$

$$\mathbf{K}(n) = \frac{\mathbf{P}(n)\cdot\mathbf{X}(n)}{\mathbf{X}^{\mathrm{T}}(n)\cdot\mathbf{P}(n)\cdot\mathbf{X}(n) + R} \tag{6b}$$

$$p_i(n+1) = p_i(n)\cdot\left[1 - k_i(n)x_i(n)\right] \tag{6c}$$

have been used. In (5) and (6), $\mathbf{W}$ is column vector of recursively estimated optimal weights, e denotes the error signal (difference between the pilot and the signal at the antenna output), $\mathbf{X}$ is column vector of signals at the elements' outputs, μ is the learning constant of the LMS algorithm (influences rate of convergence and stability of the algorithm), $\mathbf{K}$ is column vector of Kalman gains, R denotes residual error, $\mathbf{P}$ is the diagonal matrix of auto-correlations of the predicted-weights' error and p_i, k_i and x_i are ith elements of $\mathbf{P}$, $\mathbf{K}$ and $\mathbf{X}$.

All the control and monitoring tools of the system are integrated into the computer program running under Windows '95 which has been developed in Delphi 2.0 (Borland International). The program enables to control sampling and A/D conversion, gives the possibility to set adaptation parameters of algorithms, visualizes the time courses of the error, output and pilot signals etc.

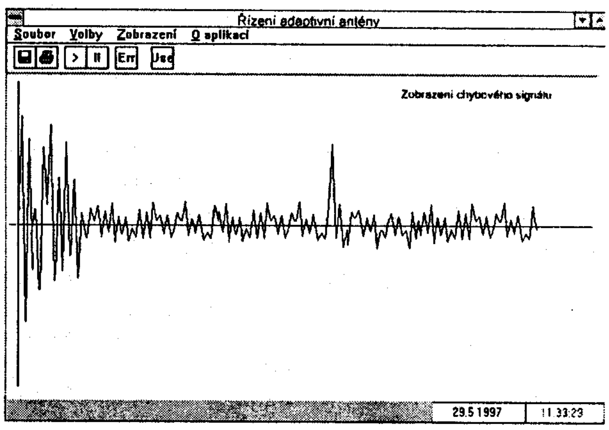

Fig. 3 Time course of the error signal (256 samples). *Antenna has been controlled by the simplified Kalman filter with the residual error set to 10^{-4} and with the initial values of the auto-correlations of the predicted-weights' error set to 10^{-1}. Angle of interference's arrival was 30° and signal to interference ratio has been considered 1.*

The designed experimental system has been tested for various angles of arrival of the interference, for various signal to interference ratios and for various parameters of adaptation algorithms. In all the cases, the system has worked properly and behaved by the same way as classical systems, i.e. systems working in IF band without undersampling [1], [2], [11].

4. Conclusion

The presented paper deals with the design of the simple digital adaptive antenna array which comes from the pilot signal method and which uses undersampling for its work. Next, a simple experimental verification of basic principles is described and obtained results are discussed.

Results of the experiment have proven functionality of the simple digital adaptive antenna based on undersampling. Thanks to the simplicity, low hardware demands and low costs of the system on one hand and thanks to the relatively good parameters on the other hand, there is a wide range of possible applications in which such a system could be used.

In the paper, influence of mutual coupling between antenna elements, function of the system in the presence of the interference correlated with the desired signal and problems of the RF-systems' design have not been considered.

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