

Application Report SPRA532

Implementation of an Adaptive Antenna Array Using the TMS320C541

Kim Phillips, Zhong Hu, Keith Blankenship, Zeeshan Siddiqi, Neiyer Correal Virginia Polytechnic Institute

Abstract

Adaptive antenna arrays are expected to play a key role in meeting the demands of the wireless communication systems of the future. This application report focuses on the implementation of a DSP based three-element adaptive antenna array for narrowband wireless communications.

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Introduction

Adaptive arrays are one of the key technologies expected to dramatically improve the performance of future wireless communications systems because they have the potential to expand coverage, increase capacity, and improve signal quality.

An antenna array consists of N identical antenna elements arranged in a particular geometry, where the geometry of the array determines the amount of coverage in a given spatial region. A very widely used array type is the uniform linear array.

For a given array geometry, the phases and amplitudes of the currents exciting the elements determine the gain of the array in a certain direction. In order to better estimate a signal arriving from a particular direction, the phases and amplitudes of the currents on the antenna array elements can be electronically adjusted such that received signals from this direction add in phase, and maximum gain is achieved in that direction. Due to the reciprocal nature of antennas, this approach is also applicable to focus the array beam for transmission. Figure 1 shows a block diagram representation of an adaptive antenna array.





To adjust the amplitude and phases of the individual currents, complex weights are placed in the signal path of each element. The weighted signals are combined and the output is fed to a control unit that operates on the individual signals and their combined output to update the weights. Weight updating is usually accomplished adaptively to satisfy an optimization criterion.



There are several adaptive algorithms available for updating the weights. These include gradient-based algorithms, recursive methods, and others such as the constant modulus algorithm (CMA). The CMA is particularly suitable for constant envelope signals such as frequency or phase modulated signals.

Basic Principles of Beamforming

Figure 2 shows a uniformly-spaced linear array with an incident plane wave impinging on the array from an angle θ . The incident wave is described by:

$$E(t,r) = E_0 \exp[j(\omega t - \vec{k} \bullet \vec{r})]$$

where $\vec{k} = \frac{2\pi}{\lambda} (-\sin\theta \,\vec{e}_x - \cos\theta \,\vec{e}_y)$ points in the direction of propagation and $\vec{k} = \frac{2\pi}{\lambda} (-\sin\theta \,\vec{e}_x - \cos\theta \,\vec{e}_y)$, defines the observation point.





Taking into account the time delay due to plane wave propagation, the signals at the *n*-th array element is given by:

$$\begin{aligned} x(0) &= E(t, \vec{r} = 0) = u(t) \exp[j(\omega t)], \\ x(1) &= E(t, \vec{r} = d\vec{e}_x) = u(t) \exp[j(\omega t - \frac{2\pi d}{\lambda}\sin\theta)] = x(1) \exp[-j\frac{2\pi d}{\lambda}\sin\theta], \\ x(2) &= E(t, \vec{r} = 2d\vec{e}_x) = u(t) \exp[j(\omega t - \frac{4\pi d}{\lambda}\sin\theta)] = x(1) \exp[-2j\frac{2\pi d}{\lambda}\sin\theta], \\ \dots \\ x(n-1) &= E(t, \vec{r} = (n-1)d\vec{e}_x) = u(t) \exp[j(\omega t - \frac{2(n-1)\pi d}{\lambda}\sin\theta)] = x(1) \exp[-(n-1)j\frac{2\pi d}{\lambda}\sin\theta] \end{aligned}$$

where *d* is the inter-element spacing. Thus, when $d = \lambda/2$, the phase difference between the outputs of adjacent elements produced by a plane wave arriving from a direction θ is $\pi \sin \theta$.

The fundamental function performed by an adaptive array is to focus a beam in a particular direction for either reception or transmission. The operation of an antenna array may be viewed as the spatial counterpart of FIR filtering of a discrete signal in the time domain.

The most common adaptive arrays form a linear combination of the sampled signal at the antenna elements. The set of weights is referred to as the steering vector. The steering vector acts as the weights of a spatial filter and determines the gain of the array in a particular direction. In situations where the direction of the sources of interference are known, interference mitigation is possible by placing the nulls in the antenna pattern in these directions while simultaneously steering the main beam in the direction of the desired signal.

Constant Modulus Algorithm

Many communication signals have a constant envelope. For these types of signals, you can take advantage of the prior knowledge of this characteristic and specify the adaptation algorithm to achieve a desired steady state response from the array. For the implementation of the adaptive array, the constant modulus algorithm (CMA) was selected as the performance criterion that governs the process of adaptively updating the weight vector.

The constant modulus algorithm adjusts the weight vector of the adaptive array so as to minimize the variation of the envelope of the desired signal at the array [3]. After the algorithm converges, a beam is steered in the direction of the signal of interest and nulls are placed in the direction of the interference.

The CMA tries to minimize the cost function:

$$J(k) = E\left\{\left\|y(k)\right\|^{p} - 1\right\|^{q}\right\},\$$

where *E* is the expectation operator and y(k) is the linear combination of the samples of the array element x(k):





$$y(k) = \sum_{i=1}^{n} w_i^*(k) x_i(k)$$

The convergence of the algorithm depends on the coefficients p and q in the above equation. This implementation uses p = 1 and q = 2.

Using the method of steepest descent and replacing the gradient vector with its instantaneous estimate, the weight vector is updated using the following recursive equation:

$$\vec{w}(k+1) = \vec{w}(k) - \mu \vec{\nabla}(J(k)) = \vec{w}(k) - \mu \vec{x}(k) \left(y(k) - \frac{y(k)}{|y(k)|} \right)^{T}$$
 where the adaptive step

constant has been denoted by μ .

By defining the error term as:

$$\boldsymbol{e}(k) = \frac{\boldsymbol{y}(k)}{\|\boldsymbol{y}(k)\|} - \boldsymbol{y}(k)$$

and substituting, the weight update equation can be expressed as:

$$\vec{w}(k+1) = \vec{w}(k) + \mu e^{\hat{x}}(k) \vec{x}(k)$$

Computation of the error term requires the square root operation; it is beneficial to seek less computationally intensive methods of approximating the desired error term.

Simulations

Before implementing the adaptive array in a DSP on a hardware platform, a simulated implementation was performed on a workstation. This was useful for rapid testing of implementation alternatives.

The simulation was programmed in the C language. The two independent variables in the simulation were the algorithm update step size, μ , and the angle of arrival of the received signal, θ . The transmitted signal was assumed to be a continuous wave (CW) signal, a static line-of-sight radio channel (that is, no multipath) was used, and a perfectly locked receiver (that is, no frequency offset) was assumed.

The array elements were assumed to be one-half wavelength apart at a frequency of 2050 MHz since that was consistent with the hardware setup. Figure 3 shows the results of a simulation where the signal was assumed to be incident from an azimuthal angle ϕ =30°. As illustrated, the antenna array was initially steered to an azimuthal angle of ϕ =90° (broadside pattern) and after about 5000 iterations the CMA converged to the correct beam pattern. Some noise had to be added to avoid ill-conditioned matrix situations. In the hardware implementation, the noise at the RF front-end and A/D quantization helps alleviate any potential convergence problems.

The simulations also provide the opportunity to examine the trade-off between CMA step size, μ , and the rate of convergence. In general, small step sizes result in smaller final mean-squared error (MSE) at the expense of larger convergence time.

The assembly code generated from the C program became the foundation for the optimized system implementation on the TMS320C541.

Figure 3. Evolution of the Array Response



System Description

A block diagram of the adaptive array system is shown in Figure 4. The antenna segment of the multi-sensor testbed consists of a custom made linear array of three quarter-wavelength monopoles spaced half a wavelength apart. The elements drive a set of identical analog front ends. Rather than using direct sampling at RF, the analog tuning segment performs down-conversion to IF for bandpass digitization. The analog tuner downconverts signals at an RF of 2050 MHz. It has a noise figure of 5 dB, and provides 75 dB of spurious free dynamic range in a 2 MHz IF bandwidth with a minimum detectable signal of -111.4 dB.

The RF bandpass filter provides rejection of out of band interference. The RF low noise amplifier (LNA) increases the level of the signal before it reaches the mixer and basically determines the noise figure of the receiver.

Following RF amplification, the signal is mixed down to IF. A common clean LO-signal is used to drive the mixers of all the branches of the analog tuners. Post-mixer bandpass filtering provides rejection of undesired out-of-band signals, mixer spurious products, and RF and LO mixer-leakage.





After amplification of the analog IF signal, IF digitization using harmonic sampling takes place. Once the IF signal has been digitized, the HSP50016 digital down-converter is used to demodulate the desired signal to its complex baseband in-phase and quadrature components. By relieving the DSP from the processing burden associated with the downconversion functions, more computational power becomes available for the tasks required for array processing. Digital downconversion not only eliminates the need for another IF stage but it also overcomes many of the problems related to analog downconversion and lowpass digitization.

Thus, after proper signal conditioning, the IF signal is downconverted, downsampled, and filtered for baseband processing with the adaptive array algorithm in the TMS320C541 EVM evaluation board. The input to the signal-processing segment is now a complex-baseband data stream. Data acquisition from the digital downconversion (DDC) boards is interrupt driven. The DDCs trigger an interrupt service routine at regular intervals every time down-converted in-phase and quadrature samples from the antenna elements become available for transfer to the EVM. Figure 5 shows the schematic diagram of the digital interface used for communications between the EVM and the digital down-conversion boards.

The EVM access each one of the DDC boards and proceeds to store the baseband samples in a data buffer. Once the buffer has been filled, interrupts are disabled. The DSP then processes every *n*-th set of samples in the received block, updates the steering vector according to CMA, and transfers the steering vector to the host PC for display of the array patterns. When the end of the data buffer is reached, interrupts are enabled and the processor returns to data acquisition mode.

As a result, the DSP is able to manipulate through signal processing the shape of the array beam pattern to optimize system performance.

The EVM successfully collects data from the DDCs, fills in the data buffer, carries out CMA and updates the steering vector. Although we have implemented a PC interface for testing, the graphical antenna pattern display has been developed but has not been tested. In order to test the data, we sent the raw antenna data to the host PC, saved the antenna data in a file, and plotted the data with Matlab. The raw antenna data was successfully downsampled and acquired by the EVM and sent to the PC.

Figure 5. Schematic Layout of Digital Interface





Summary

A DSP-based three-element narrowband adaptive array has been implemented based on the TMS320C541 EVM board. The algorithm used for adaptation of the adaptive array is the constant modulus algorithm. The system consists of an analog tuner, a digital down-conversion section, and a DSP core implemented on the EVM.

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