Implementation and Performance Evaluation of TMSC6711 DSP-based Digital Beamformer

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Abstract

This paper discusses the implementation and performance evaluation of a DSP-based digital beamformer using the Texas Instrument TMSC6711 DSP processor for smart antenna applications. Two adaptive beamforming algorithms which served as the brain for the beamformer, the Normalized Least-Mean-Square (NLMS) and the Constant Modulus Algorithms (CMA) were embedded into the processor and evaluated. Result shows that the NLMS-based digital beamformer outperforms the CMA-based digital beamformer: 1)For NLMS algorithm, the antenna steers to the direction of the desired user even at low iteration value and the suppression level towards the interferer increases as the number of iteration increase. For CMA algorithm, the beam radiation pattern slowly steers to the desired user as the number of iteration increased, but at a rate slower than NLMS algorithm and the sidelobe level is shown to increases as the number of iteration increase. 2) The NLMS algorithm has faster convergence than CMA algorithm and the error convergence for CMA algorithm sometimes is subject to misadjustment.

Keywords: Smart antenna, beamforming algorithm, TMS320C6711 DSK, NLMS, CMA

I. INTRODUCTION

The demand for mobile communication services is increasing at a rapid pace throughout the globe. The need for better coverage, improved capacity, higher transmission quality, and broadband services become major challenges to the wireless mobile system and service providers. Thus, a more efficient use of the radio spectrum is required. To meet these requirements, ITU proposed the IMT2000 system, formerly known as Future Public Land Mobile Telecommunications Service (FPLMTS), which exploit the used of smart antennas at the mobile base station as one of the technological solution.

Smart antenna systems consist of multiple antenna elements at the transmitting and/or receiving end of the communication link, in which the radio link signals are processed adaptively in order to exploit the spatial dimension of the mobile radio channel. Spatial processing capability of smart antenna enables it to locate many users, creating a different sector for each user. This means that more than one user can be allocated simultaneously to the same physical communication channel in the same cell, with only an angular separation. This technology dramatically improves the interference-suppression capability and greatly increases frequency reuse, resulting in increased capacity. Smart antenna with its beamforming capability optimized the signal-to-noise performance or power consumption at the both end of the links. Advancement in powerful low cost digital signal processor (DSP), general-purpose processors, application-specific integrated circuits (ASIC) as well as innovative software-based signal processing techniques (algorithms) or software radio (SDR) has allowed the development of beamforming system to progress rapidly and make the smart antennas practical for cellular communications systems [1].

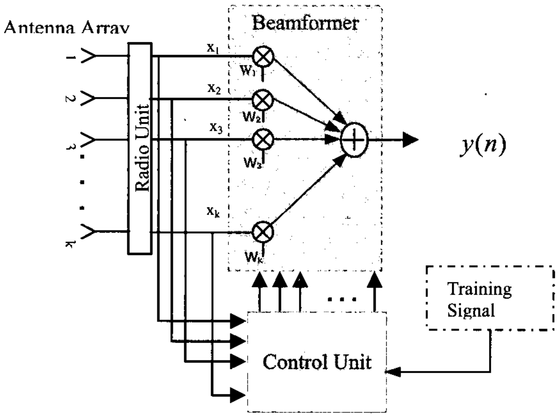
On the signal processing aspects the smart antenna system research is concentrated on the development of efficient algorithms for direction-of-arrival (DOA) estimation and adaptive beamforming. Recent works on DOA algorithms were reported in [2, 5]. Recently, neural networks have also been employed to detect the direction of arrival. In this method, once the neural network is trained offline, multiple signals can be tracked in real time. Adaptive beamforming algorithms have move from the classical least mean squares (LMS) type of beamformers [6] towards advance beamforming algorithms such as constant modulus algorithms (CMA) [7] and "eigen-projection algorithms" [8].

This paper discuss the development and performance evaluation of a normalized least-mean-square (NLMS) and CMA adaptive beamforming algorithms implemented on TMS320C6711 evaluation board (DSK). The TMS320C6711 DSK board is to serve as a digital beamforming unit, the hardware engine for smart antenna system design, which computes the weight vectors based on the embedded beamforming algorithms. Section II of this paper discusses the structure of a general smart antenna system. The adaptive beamforming algorithms will be introduced in Section III, in particular the NLMS and CMA algorithms. Section IV discusses the TMS320C6711 evaluation board which served as the processor for the computation of the beamforming algorithms and Section V describes the development of software. Simulation results presented in Section VI and finally Section VII concluded the paper.

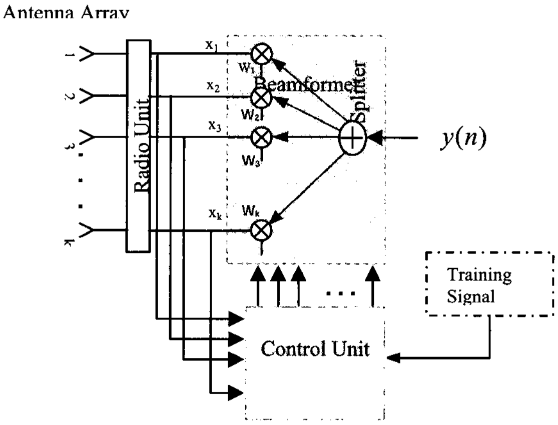
II. SMART ANTENNA SYSTEM DESCRIPTION

Figure 1 shows the generic system diagram of a smart antenna system. It consists of an antenna array, a radio unit, a beamformer, and a control unit. The hardware component for a smart antenna system is similar both for the transmitter and the receiver. The radiating elements of the antenna array of the smart antenna system can be constructed from dipoles, slots, TEM horns, reflectors antenna, etc. Microstrip antenna array has now being received greater attention due to its low profile, lightweight, conformability, compatibility with microwave integrated circuit, and ease of fabrication using PCB technique.

In the radio unit of the receiver, the received RF signal from each antenna element is amplified and downconverted to IF. Each signal from the IF band is



(a) Reception part of a smart antenna



(b) Transmission part of a smart antenna

Figure 1. Schematic diagram of smart antenna

downconverted to baseband, producing a complex I and Q signal for the beamformer inputs ($x_1, x_2.....x_k$). In the radio unit of the transmitter, the complex baseband signals (I and Q) produced by the beamformer is amplified and upconverted to produce the required RF signals, which are then transmitted using antenna element.

The beamformer section first estimate the direction-of-arrival (DOA) of the intended signal before calculating the required weight vectors (phase and amplitude for each array element) using the adopted beamforming algorithm. These signals which contain both the desired signal and the interfering signals are appropriately scaled by the weight vectors ($w_1, w_2.....w_k$) and combined to generate the array output in the beamformer section. The output for the beamformer is forming a weighted combination of signal from k element of antenna array [9].

$$y(n) = \sum_{q=1}^{K} w_q^* x_q(n) \tag{1}$$

or by matrix vector

$$y(n) = w^H x(n) \tag{2}$$

In the control unit section the array output is compared with reference signal to generate the error signal which then adaptively minimizes it using an adaptive algorithm. The adaptation process involves changing the weight vector according to certain minimization criteria.

For the receiver section of a smart antenna, a combiner is used to combine the array output for increased signal enhancement through space time processing whereas in the transmitter section a splitter is used to split up the signal in n number of branches, which are weighted by complex weights $w_1, w_2.....w_k$ in the beamforming unit.

The beamforming unit and the control unit can be designed using a microcontroller, DSP, FPGA, or ASIC processor depending on the design requirements and development costs.

III. BEAMFORMING ALGORITHM

There are many types of adaptive beamforming algorithms exist in the open literature. Most of the adaptive beamforming algorithms can be categorized under two classes according to whether training signal is used or not. These two classes are non-blind adaptive algorithm and blind adaptive algorithm. Non-blind adaptive beamforming algorithm used a training signal, $d(n)$ to update its complex weight vector. This training signal is sent by the transmitter to the receiver during the training period. Beamformer in the receiver used this information to compute new complex weight. LMS, N-LMS, and RLS algorithms are categorized as non-blind algorithm. Blind algorithms on the other hand do not need any training sequence to update its complex vector. CMA, spectral self-coherence restoral (SCORE), and decision directed (DD) algorithms are examples of blind adaptive beamforming algorithms. These algorithms use some of the known properties of the desired signal. In blind algorithm, the goal is to retrieve the input signal by using output signal and possibly the statistical information for the input. In this paper, only the NLMS algorithm and CMA algorithm will be discussed and implemented in this work. These algorithms are chosen because of its simplicity. They do not need any matrix inversion and complex computations, and only involves multiplication, division, addition and subtraction.

A. NORMALIZED LMS ALGORITHM

Theoretically, LMS method is the most basic method for calculating the weight vectors. However, in practice, an improved LMS method, the Normalized-LMS is used to achieve stable calculation and faster convergence [10]. The NLMS algorithm can be formulated as a natural modification of the ordinary LMS algorithm and based on stochastic gradient algorithm [11]. From LMS algorithm, we know that

$$y(n) = w^H x(n) \tag{3}$$

$$e(n) = d(n) - y(n) \tag{4}$$

$$w(n+1) = w(n) + \mu x(n) e^*(n) \tag{5}$$

where, $e(n)$ is the error signal, $x(n)$ is the sampled signal, $d(n)$ is the desired signal, and $y(n)$ is the output signal. Gradient noise amplification problem may be occurring in the standard form of LMS algorithm. This is because the correction $x(n)e^*(n)$ at iteration n applied to the weight vector $w(n)$ is directly proportional to the input vector $x(n)$. This can be solved by normalization of the correction at iteration $n+1$ with the square Euclidean norm of the input vector $u(n)$ at iteration n [8]. NLMS algorithm can be summarized as

$$y(n) = w^H x(n) \tag{6}$$

$$e(n) = d(n) - y(n) \tag{7}$$

$$w(n+1) = w(n) + \frac{\mu}{\|u(n)\|^2} x(n) e^*(n)$$

(8)

Figure 2 shows the flowchart for the NLMS beamforming algorithms.

B. CONSTANT MODULUS ALGORITHM

CMA is a gradient-based algorithm that works on the premise that the desired signals have constant modulus. This algorithm adjusts its complex weight vector to minimize the fluctuation in amplitude of output signal. The advantage of this method is that no training signal is needed. CMA tries to update the weights by minimizing the cost function $J(n)$ [12] given as

$$J(n) = E\left[(|y(n)|^{p} - R_{p})^{q} \right] \tag{9}$$

Finally the CMA can be summarized as

$$y(n) = w^{H}x(n) \tag{10}$$

$$e(n) = \frac{y(n)}{|y(n)|} - y(n) \tag{11}$$

$$w(n+1) = w(n) + \mu x(n)e^{*}(n) \tag{12}$$

Figure 3 shows the flowchart for the CMA beamforming algorithm.

IV. TMS320C6711 DSP STARTER KIT (DSK) BOARD

The Texas Instrument, TMS320C6711 (C6711) is a floating-point digital signal processor (DSP) suitable for numerically intensive computation such as the computation of beamforming algorithms. This floating point DSP processor can operate up to 900 million floating-point operations per second (MFLOPS) at clock speed of 150 MHz. The C6711 has the following features.

- Multichannel buffered serial port (McBSP)
- 16-bit host port interface (HPI)
- 32-bit general-purpose timer
- 32-bit external memory interface (EMIF)
- Enhanced direct memory access (EDMA) – for C6x11

The DSP Starter Kit (DSK) evaluation board offers good programming facility which enables fast implementation of the algorithm. Figure 4 shows the system diagram of the TMS320C6711 DSK. The board is designed to be as a standalone unit controlled by the built-in XDS510 emulator through the PC's parallel port. The DSK board has the following features [13].

- TMS320C6711 DSP processor as a CPU
- Clock: 150 MHz.
- Memory: 16MB, 100 MHz, 10ns SDRAM. 128 KB flash ROM.
- Expension memory and peripheral.
- PC's parallel port and JTAG emulator interface.
- Host port interface (HPI) access all DSP memory
- 16-bit codec interface: 8 kHz sampling rate, mono input with two speaker drivers.
- Six LED indicators.

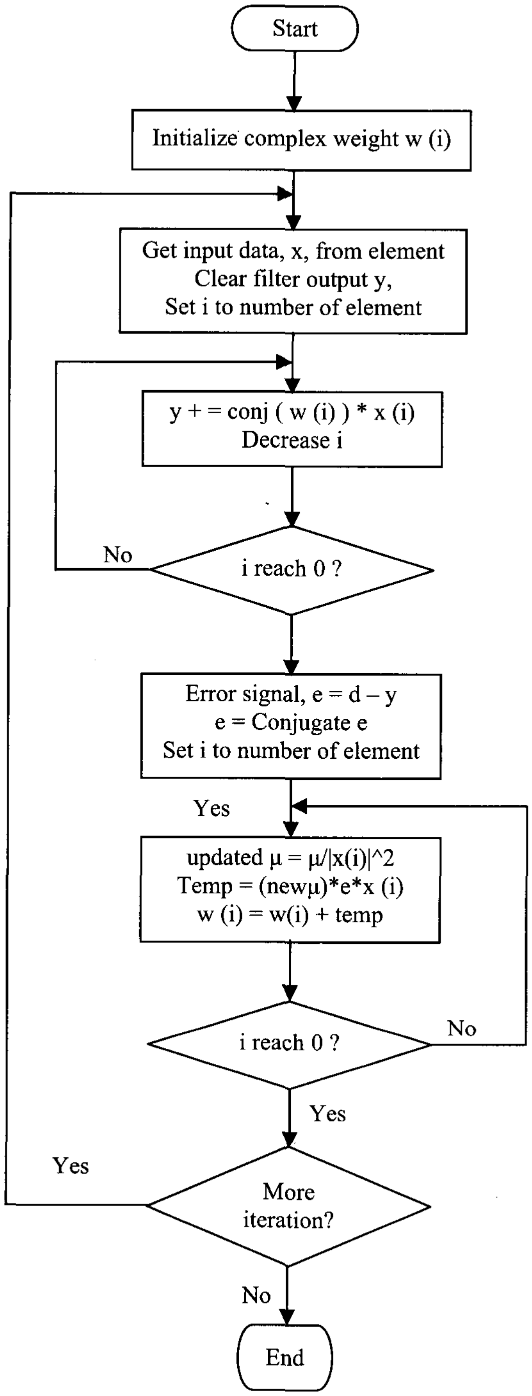


Figure 2. NLMS algorithm flow chart

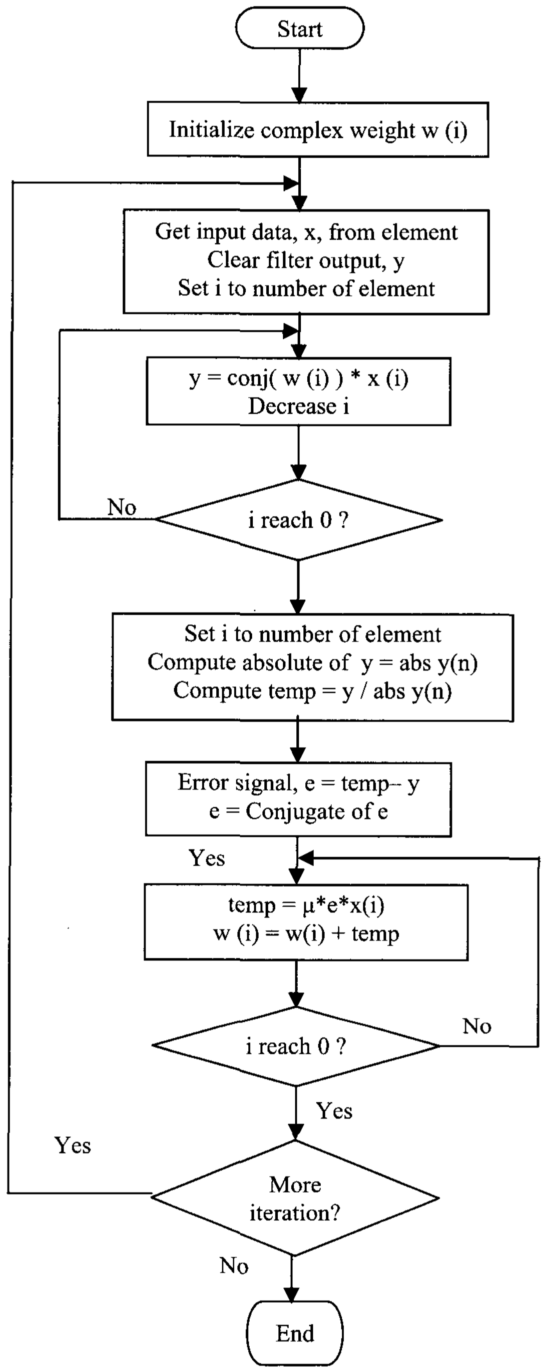


Figure 3. CMA algorithm flowchart

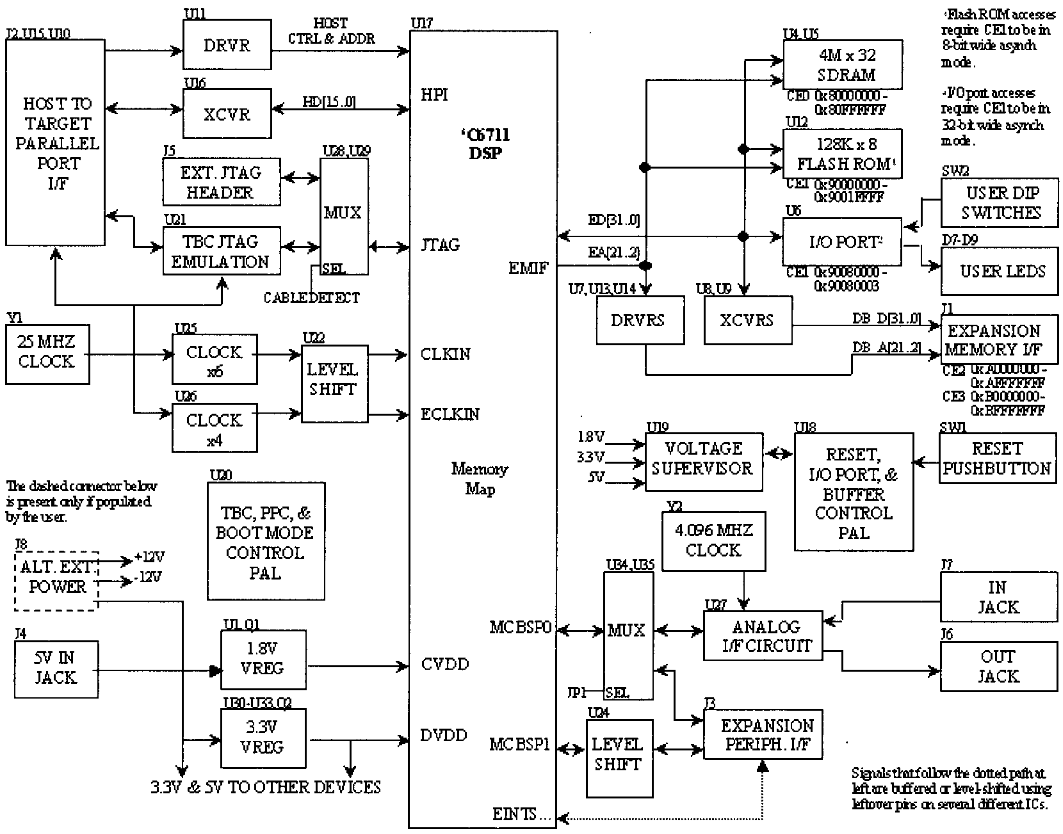


Figure 4. TMS320C6x DSK block diagram

V. SOFWARE DEVELOPMENT

The adaptive beamforming algorithms for this work are developed using the TI's code composer studio (CCS) and supported library included in the DSK board. The CCS provides an integrated development environment to incorporate tools for code generation, such as a C compiler, assembler, and a linker. It also has an interface with Matlab which allows the programs to be written in Matlab and downloaded onto the DSK.

Figure 5 shows the development set-up for the evaluation of the beamforming algorithms and Figure 6 gives the DSP code for the NLMS algorithm. A minimum shift keying (MSK) which has constant envelope properties is used to generate I and Q streams. This complex baseband data streams which are the processed signal at the input of the beamformer represent the composite signals due to intended user and interferers. These signals are stored into the DSK data buffer through CCS link. Once the buffer has been filled, the DSP then processes each samples and updates the weight vectors by the DSP code based on the selected beamforming algorithm. The generated complex weight vectors are automatically stored in the DSK memory location defined by the DSP code. The calculated weight vectors are then extracted from the DSK to the PC to evaluate the performance of the algorithms in terms of their convergence factor and radiation properties.

In this simulation, a uniform linear array (ULA) of 1 x 4 antenna element is considered. For the array, the antenna factor, AF is given [14] as

$$AF = \sum_{n=1}^{N} e^{j(n-1)(kd\cos\theta + \beta)} \qquad (13)$$

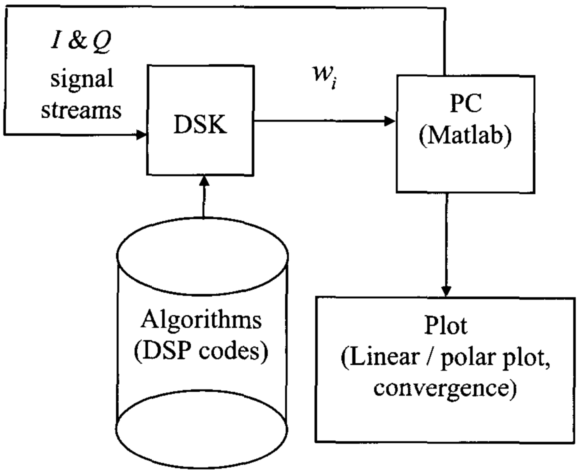


Figure 5. Development set-up for the evaluation of the beamforming algorithms.

where, $k = \frac{2\pi}{\lambda}$, and $\beta = 0$

$$AF = \sum_{n=1}^{N} e^{j(n-1)(\frac{2\pi}{\lambda} d \cos\theta)} \quad (14)$$

In the simulation, three different signals are assumed to arrive at the antenna, among them one is the desired signal and the other two are interferers. The desired signal arrived at an angle of 20°, and the interferers arrived at an angle of -10° and -30° with respect to the antenna main beam. The carrier frequency f_c is set to 2 GHz and step size (μ) is set to 0.005. In [15] we have shown that the speed of convergence is greatly depends on step size parameter. In this simulation, it is assumed that the radio channel only introduced the Additive White Gaussian Noise (AWGN) with no multipath.

```
/
/ NLMS code written for C6711 DSP
#ifndef CFLOAT
#define MUI 0x001d
#else
#define MUI (float) 0.005
#endif
#ifndef CFLOAT
short xI_1[N],xI_2[N],xI_3[N],xI_4[N];
short xQ_1[N],xQ_2[N],xQ_3[N],xQ_4[N];
short dinr[N],dini[N];
int w_NLMS_r[SIZES], w_NLMS_i[SIZES];
#else
far float xI_1[N],xI_2[N],xI_3[N],xI_4[N];
far float xQ_1[N],xQ_2[N],xQ_3[N],xQ_4[N];
float dinr[N],dini[N];
float w_NLMS_r[SIZES], w_NLMS_i[SIZES];
#endif
void NLMS_Algorithm (complex wo[SIZES],
     const complex xo[SIZES],  complex di)
{complex y,e,temp;
  float norm, mu; int i;
  y.real = 0;   y.imag = 0;
  for (i=SIZES-1; i>=0;i--)
  {temp = conj(wo[i]); temp = cmult(temp,xo[i]); // y = w'x
     y  = cadd(y,temp);}
     e = csub(di, y);                 // e = d - y
     e = conj(e);                     // e = conj(e)
     for(i = SIZES-1; i>=0; i--)
  {norm = xo[i].real * xo[i].real + xo[i].imag * xo[i].imag;
     mu   = MUI/norm;                       // new mui
     temp = mult(mu,xo[i]);
     #ifndef CFLOAT
     temp = cscale(temp,15);
     #endif
     temp = cmult(e,temp);    // w(n+1) = w(n) +
u*x(n)*conj(e(n))
     wo[i] = cadd(wo[i],temp);  }}
int main()
{complex x[SIZES], w[SIZES], d;
  int i;
  for(i = SIZES-1; i >= 0; i--)
  {  w[i].real = 0;      // start initial w[0..3] = 0
     w[i].imag = 0; }
  for(i=0;i<N;i++)
  { # ifndef FINALP
     x[0].real = xQ_1[i];  x[1].real = xQ_2[i];  x[2].real =
xQ_3[i];
     x[3].real = xQ_4[i];  x[0].imag = xI_1[i];  x[1].imag =
xI_2[i];
     x[2].imag = xI_3[i];  x[3].imag = xI_4[i];
     d.real = dinr[i];  d.imag  = dini[i];
    #endif
     NLMS_Algorithm(w, x, d); }
  #ifndef  FINALP
  for(i=SIZES-1;i>=0;i--)
  {  w_NLMS_r[i] = w[i].real;
     w_NLMS_i[i] = w[i].imag; }
  #endif
```

Figure 6. DSP code for the NLMS algorithm

VI. RESULT AND DISCUSSION

Figure 7 shows the normalized antenna response in linear form for the NLMS algorithm. As shown in this figure, at iteration i=100, the algorithm cannot cancel the interferer effectively, the suppression is about around -40 dB and -55 dB for the interferer 1 and interferer 2, respectively. At i=800, the NLMS algorithm give a better result with suppression level less than -80 dB for both interferers. The polar plot of the beampattern for the NLMS algorithm is shown in Figure 8. For the NLMS algorithms, the antenna steers to the direction of desired signal even at low iteration value and at higher iteration the suppression level towards the interferer increases.

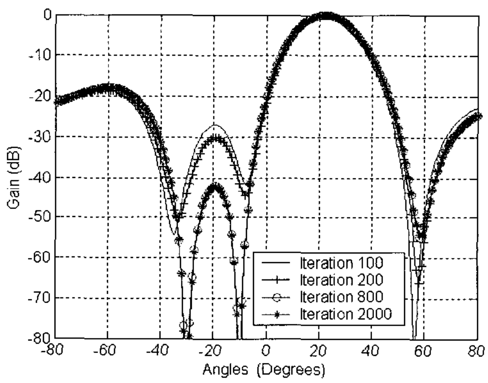


Figure 7. Antenna response for NLMS algorithm at different iteration

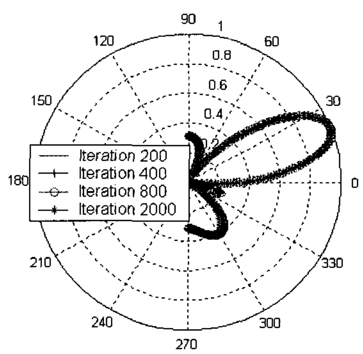


Figure 8. Polar plot for NLMS algorithm at different iteration

Figure 9 shows the normalized antenna response in linear form for the CMA. As shown in this figure, at iteration i=100 there is no significant attenuation at the interferer and the beam does not point out to the direction of desired signal even at i=800, the algorithm still does not give the required performance and the beam is still pointing away from the intended direction of the desired signal. At i=2000, CMA gives a similar performance to that of NLMS but with the maximum gain pointing at 18° close to the desired direction. From Figure 10, it can be observed that the beam radiation pattern for the CMA algorithm slowly steers to the desired direction as is expected as the number of iteration is increased, but at a rate slower than NLMS algorithm. For CMA, the sidelobe level of the beam pattern increases with increase in iteration.

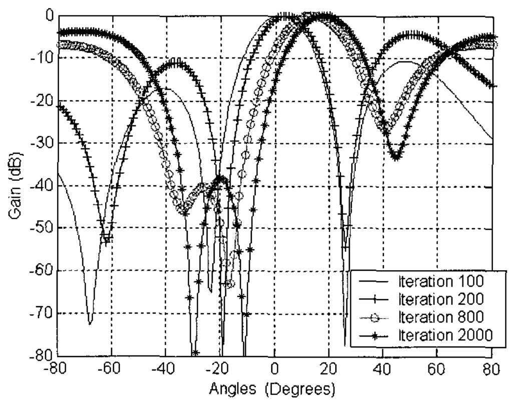


Figure 9. Antenna response for CMA at different iteration

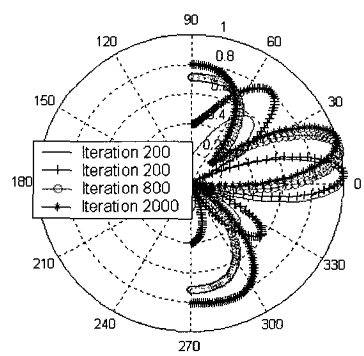


Figure 10. Polar plot for CMA at different iteration

Figure 11 and Figure 12 show the rate of error convergence for the NLMS and CMA. It can be clearly seen from the figure that the NLMS algorithm has faster convergence than CMA. This is also shown in Figure 9, where for CMA, the convergences of the complex weight vectors are slower than the NLMS algorithm. From these results, it can be said that the NLMS algorithm has better efficiency of canceling the interferer. However, the NLMS needs an extra bandwidth to transmit its training signal. This reduces the spectral efficiency of the system. For CMA, although it is not efficient to cancel the interferer at low iteration, it does not waste the bandwidth for the training signal. But it converges more slowly than NLMS and sometimes subject to miss adjustment.

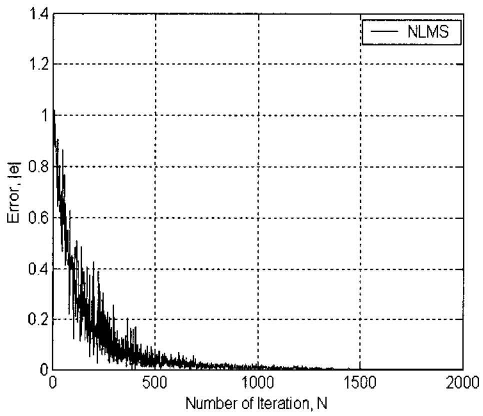


Figure 11. Error convergence for NLMS algorithm

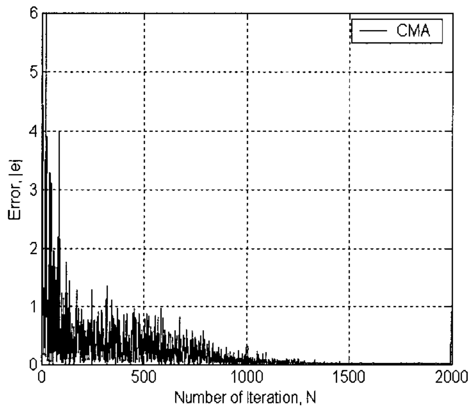


Figure 12. Error convergence for CMA algorithm

VII. CONCLUSION

A DSP based digital beamformer unit based on the TMS320C6711DSK for smart antenna system application has been successfully developed and tested. The beamformer is embedded with adaptive beamforming algorithm for calculating the optimal weight vector for steering the beam radiation pattern in an adaptive manner. Two adaptive beamforming algorithms, the NLMS and CMA were evaluated based on this hardware to determine their convergence rates and their ability to steer the beam efficiently in the desired user and to suppress the interferers in a multipath free radio channel with only AWGN is present. The beamforming algorithms were also tested for their convergence rate. In the analysis, the carrier frequency f_c is set to 2 GHz and the step size (μ) is set to 0.005. Our results show that: 1) The NLMS performed better than the CMA. For the NLMS algorithm, the antenna steers to the direction of the desired signal even at low iteration value and at higher iteration, the suppression level towards the interferer increases. For the CMA, the beam radiation pattern slowly steers to the desired direction as the number of iteration increased, but at a rate slower than NLMS algorithm and the sidelobe level is shown to increase with increase in iteration. 2) The NLMS has faster convergence rate than the CMA, i.e., the error convergence rate for CMA algorithm is higher than that of the NLMS and sometimes subject to misadjustment. The beamformer unit developed provided the capability to test various beamforming algorithms independently of the radio unit and antenna system and hence can speed up the development process of a smart antenna system. The system should further be tested for different simulated radio condition in the future to determine the robustness of the beamforming algorithms.

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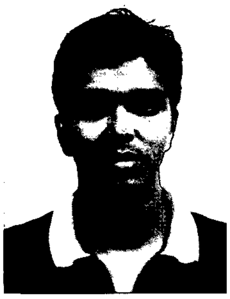
Biography

Zainol Abidin Abdul Rashid is a lecturer at the Department of Electrical, Electronic and System Engineering, Faculty of Engineering, Universiti Kebangsaan Malaysia, Malaysia where he joined in 1989. He received his B.Sc. degree in Electronics from the same university in 1985 and obtained his M.Sc. degree in Microprocessor Engineering and PhD degree in Electrical Engineering from University of Bradford, UK in 1987 and 1997 respectively. He research into ice crystal crosspolarization effect on earth-satellite link at 20 GHz from the ESA Olympus satellite during his PhD work. After finishing his PhD work, he was appointed as a team leader for the first Malaysian microsatellite, TiungSAT-1. He is currently pursuing research in 3G air interface and smart antenna system, and leading the Malaysian team for the polar ionospheric and water vapour research at Scott Base, Antarctica. He is also involved in the development of aircraft collision avoidance system.



Mohammad Tariqul Islam was born in Dhaka, Bangladesh in 1975. He received his B.Sc. and M. Sc. Degree in Applied Physics and Electronics from University of Dhaka, Dhaka, Bangladesh in 1998 and 2000, respectively. He worked as a lecturer at International Islamic University Chittagong (IIUC), Dhaka. He is currently pursuing his PhD at Electrical, Electronics and System Engineering department at University Kebangsaan Malaysia with the supervision Dr. Zainol Abidin Abdul Rashid. His research interest includes Smart antenna system.

Liew Chang Sheng (no picture available) was born in Penang, Malaysia in 1980. He is pursuing his Bachelor Degree at Electrical, Electronics and System Engineering department at University Kebangsaan Malaysia with the supervision Dr. Zainol Abidin Abdul Rashid. He received his B.E. Degree in Communication and Computer in 2005. He now currently worked as a Design Engineer at Intel, Penang.