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Antenna Array Signal Direction of Arrival Estimation on Digital Signal Processor (DSP)

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Abstract

Antenna arrays are used in many digital signal processing applications due to their ability to locate signal sources. Direction of Arrival (DOA) estimation is a key task of array signal processing. Although various algorithms have been developed for DOA estimation, their high complexity prevents their use in real-time applications. In this paper, we design and develop an efficient parallel implementation of DOA on DSP which is the most widely used processor in embedded system. Due to the potential parallelism in MUSIC algorithm, it is selected for 2-D DOA estimation. Two computational cores in MUSIC are identified and parallelized. Vectorization of multiple single precision floating point operations is proposed to make good use of the 128-bit vectors on DSP C6678. Then, the parallel DOA estimation algorithm is implemented on one core of DSP C6678 which is the latest version up to now. Experiments are conducted on both 1-D and 2-D antenna array signals. Considerable performance improvement is obtained.

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Keywords: parallel computing, signal processing, DSP, DOA estimation

1. Introduction

Antenna array signal direction of arrival (DOA) estimation is a key task of array signal processing. Array signal processing emerged in the last few decades as an active area and was centred on the ability of using and combining data from different antennas to deal with specific spatial and temporal estimation task. Array signal processing is used in radar, sonar, seismic exploration, anti-jamming and wireless communication. As array signals are everywhere in modern times, it is very important to make good use of them to serve both civil and military applications. Research problems associated with array processing include the number of sources used, their direction of arrivals, and their signal waveforms.

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An antenna array is a configuration of multiple antenna elements arranged and interconnected in space to obtain a directional radiation pattern. Arrays built using small antenna elements achieve the same level of performance as that of a single large antenna, by trading the electrical problems of feeding for mechanical simplicity. Arrays can be constructed in various types of geometric configurations. A linear array is the most elementary form in which the centres of the elements of the array are aligned along a straight line. Planar arrays could be circular, rectangular or arbitrarily shaped. Arrays whose element locations conform to a given non-planar surface are called conformal arrays.

Direction of arrival (DOA) estimation or direction finding has been an active area of research for a while. The goal in DOA detection and estimation is to accurately determine the number of sources producing the waveforms and their corresponding locations. Historically, DOA techniques have found application in radar, sonar, electronic surveillance and seismic exploration fields. In radar applications, they are useful for air traffic control and target acquisition [1]. Intelligence agencies use them for covert location of transmitters and signal interception. DOA also finds applications in position location and tracking systems. More recently, direction of arrival estimation has become important in mobile radio communications, for example, determining the multipath structure of radio channels.

Although both 1-D and 2-D DOA estimation have been implemented in different platforms by various methods, its efficiency and accuracy needs to be improved. DOA estimation of 2-D antenna arrays is to estimate elevation and azimuth angles of interesting signal sources. As 2-D DOA estimation is much more complex than 1-D's, a fast implementation of DOA estimation of 2-D antenna arrays is in strong demand. DSP, widely used in embedded systems, is a specialized microprocessor with optimized architecture. As more and more signal processing applications are in portable devices, or embedded systems, DOA estimation on DSP is valuable.

The main contribution of our work is the parallel implementation of MUSIC (Multiple Signal Classification, a widely used algorithm for 2-D DOA estimation) on DSP. Two computational cores in MUSIC are identified and parallelized. Single Instruction Multiple Data (SIMD) technique is adopted on DSP C6678 which is the latest version. In the experiment, we used the sampled signal data of a simulative 2-D antenna array in L-shape which is a widely used shape in 2-D DOA estimating, and a 1-D array for the purpose of comparison. The exact direction of each antenna element of the array was obtained with greater performance than ever before.

In Section 2, we present the mainly used methods for the DOAs in array signal processing. In Section 3, the subspace-based algorithm, MUSIC, is presented in 1-D and 2-D DOA estimation. In Section 4, we provide a detailed description of our implementation on DSP C6678. In Section 5, we use antenna array signals to conduct performance evaluation on DSP C6678 and present the results. In the last section, we summarize the work.

2. Related work

2.1. Overview of direction of arrival estimation algorithms

This section provides a brief review of the methods available for estimation of angle of arrival of a radio signal using an antenna. The array-based DOA estimation techniques considered here can be broadly divided into four different types: conventional techniques, subspace based techniques, maximum likelihood techniques and the integrated techniques which combine property restoral techniques with subspace based techniques. Conventional methods are based on classical beamforming techniques and require a large number of elements to achieve high resolution. Subspace based methods are high resolution sub-optimal techniques which exploit the eigen structure of the input data matrix. Maximum likelihood techniques are optimal techniques which can perform well even under low signal-to-noise ratio conditions, but are in general computationally very intensive. The integrated approach use property restoral based techniques to separate multiple signals and estimate their spatial signatures where their directions of arrival can be determined by using subspace techniques. As subspace based methods exploit the eigen structure of the input data matrix and can obtain high resolution, they are mainly discussed in this paper.

2.2. Subspace methods for DOA estimation

Though many of the classical beamforming based methods, such as the Capon's minimum variance method [6], are successful and widely used, these methods still have some fundamental limitations in resolution. Most of these limitations arise due to the fact that they do not exploit the structure of the narrowband input data model of the measurements. Schmidt [2] was the first to exploit the structure of a more accurate data model for the case of sensor arrays of arbitrary form. Schmidt derived a complete geometric solution to the DOA estimation problem in the absence of noise, and extended the geometric concepts to obtain a reasonable approximation to the solution in the presence of noise. The technique proposed by Schmidt is called the Multiple Signal Classification (MUSIC) algorithm, and has been thoroughly investigated since its emergence. The geometric concepts upon which MUSIC is founded form the basis for a much broader class of subspace-based algorithms. Apart from MUSIC, the primary contributions to the subspace-based algorithms include the Estimation of Signal Parameters via Rotational Invariance Technique (ESPRIT) proposed by Roy et. al. [10], and the minimum-norm method proposed by Kumaresan and Tufus [3].

2.3. DOA estimation on DSPs

As signal's DOA estimation requires fast calculation and high precision, specialized microprocessors with optimized architecture for digital signal processing are required, and thus, DSP becomes a good choice. In recent years, DOA estimation has been implemented on different versions of DSPs. Zhu [11] gives an example of multi-DSP chips' parallel processing system which consists of 20 TMS320C31 chips. Li [12] presents an implementation of MUSIC algorithm on ADSP21160. However, it used assembly language for programming which is difficult in programming and understanding and tends to make mistakes. Dong [13] used an improved super-resolution adaptive arrays to produce sharper direction map and reduce error judgment on TMS320C6701 DSP.

3. DOA estimation algorithm MUSIC

MUSIC and ESPRIT are the mostly used subspace-based algorithms in DOA estimating. Both methods are high-resolution, much better spatial resolution than beamforming and other methods mentioned above. And both methods are able to detect multiple sources. Though ESPRIT needs less computation, it is suitable only if there are plenty of array sensors compared with the number of sources to detect. As large numbers of sensors are often too expensive to access, MUSIC [2,5] has broader application prospects. In addition, MUSIC algorithm is suitable for parallel because its computation focuses on matrix operations.

Based on the particular structure of the covariance matrix of the signal, MUSIC uses the concept of subspace, which contains the information on the model of signal propagation. The principle is to decompose the data space into subspaces: signal subspace and noise subspace. So, the first step in MUSIC is to decompose the covariance matrix of observation vectors [4] into orthogonal subspaces.

3.1. Decomposition of the covariance matrix in MUSIC

Consider K narrow-band plane wave signals from directions $\theta_1, \theta_2, \ldots, \theta_k, \ldots, \theta_K$ centred at frequency

f_0 and received by a linear array composed of M identical elements ($M>K$) spaced by a distance d. It is supposed that the signals and noises are stationary and uncorrelated, and the reference element is the first one. Using complex signal representation, the complex envelop representation of the received signals can be expressed as

$$x(t) = \sum_{k=1}^{K} a(\theta_k)s_k(t) + n(t) \tag{1}$$

where $a(\theta_k) = [1, e^{-j\frac{2\pi d}{\lambda}\sin\theta_k}, \ldots, e^{-j\frac{2\pi d}{\lambda}(M-1)\sin\theta_k}]$ is the $M\times1$ steering vector for the kth signal and λ is the wavelength, $s_k(t)$ is the kth transmitted signal, and $n(t) = [n_1(t), n_2(t), \ldots, n_m(t), \ldots, n_M(t)]^T$ is the $M\times1$ vector of the noise.

In matrix notation, (1) becomes

$$x(t) = A(\theta)s(t) + n(t) \tag{2}$$

where $x(t) = [x_1(t), x_2(t), \ldots, x_m(t), \ldots, x_M(t)]^T$ is the envelope representation of the K received signals of the array, $A(\theta) = [a(\theta_1), a(\theta_2), \ldots, a(\theta_k), \ldots, a(\theta_K)]^T$ is the $M\times K$ matrix of the array response vector and $s(t) = [s_1(t), s_2(t), \ldots, s_k(t), , , , .s_K(t)]^T$ is the $K\times1$ signal vector.

The covariance matrix for the data vector $x(t)$ can be written as

$$R_{xx} = E\{x(t)x^H(t)\} = AR_{ss}A^H + \sigma^2 I \tag{3}$$

where R_{ss} is the covariance matrix for the signal vector, σ^2 is the noise power, and I is the $K\times K$ identify matrix.

Assuming the initial hypothesis is that the covariance matrix R_{xx} is not singular. This assumption physically means that sources are totally uncorrelated between them. MUSIC algorithm assumes that signal and noise subspaces are orthogonal[9]. Then R_{xx} can be written as

$$R_{xx} = E \sum E^H = \sum_{i=1}^{M} \lambda_i \varepsilon_i \varepsilon_i^H \tag{4}$$

where $\sum = diag(\lambda_1, \lambda_2 \ldots, \lambda_M)$ is the diagonal matrix of eigenvalues. And we can prove that the eigenvalues can be arranged as $\lambda_1 \geq \lambda_1 \geq \ldots \geq \lambda_K > \lambda_{K+1} = \ldots = \lambda_M = \sigma^2$. That is to say, eigenvectors corresponding to the first K eigenvalue construct the signal subspace E_s, and the rest form the noise subspace E_N. Accordingly, $\sum$ can be decomposed as $\sum_s = diag(\lambda_1, \lambda_1, \ldots, \lambda_K)$ and $\sum_N = diag(\lambda_{K+1}, \lambda_{K+1}, \ldots, \lambda_M)$. Finally,

$$R_{xx} = E \sum E^H = E_s \sum_s E_s^H + E_N \sum_N E_N^H \tag{5}$$

3.2. Peak searching of the pseudo-spectrum

MUSIC is based on the properties of signal and noise subspaces, vectors derived from E_s generate a signal subspace collinear with steering vectors of sources $a(\theta_k)$ and vectors derived from E_N generate a noise subspace orthogonal to the steering vectors of these sources. It follows that

$$E_N^H a(\theta_k) = 0 \text{ for } k = 1,2,\ldots,K. \tag{6}$$

To determine DOA, it is necessary to project all the possible steering vectors into the noise subspace and retain those minimizing projection, resulting in the following discriminant function:

$$a(\theta)E_N E_N^H a(\theta) = 0 \qquad (E_N = [e_1, e_2, \ldots, e_{M-K}]) \tag{7}$$

whose zeros are the DOAs.

Estimating DOA of signals is equivalent to look for maximum values of the MUSIC pseudo-spectrum $P(\theta)$:

$$P_{music}(\theta) = \frac{1}{a(\theta)^H E_N E_N^H a(\theta)}. \tag{8}$$

This is for the case of 1-D antenna array. Similarly, assume that an antenna array in L shape receives signals from K sources, and (θ_k, ϕ_k) $(k = 1,2,\ldots,K)$ represents the 2-D direction of arrival (θ_k and ϕ_k represent elevation angel and azimuth angle respectively). And there are N antenna elements in X axis and M antenna elements in Y axis. Then two auxiliary figures are shown below.

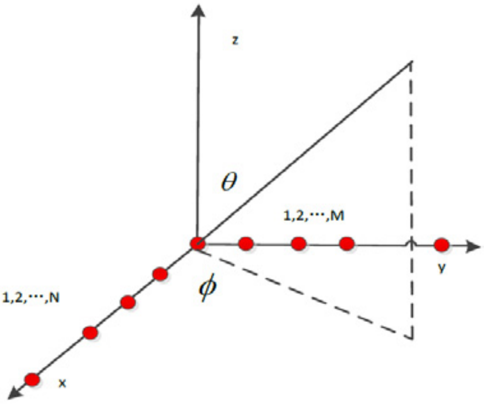

Fig.1. A sketch map of an L-shape antenna array

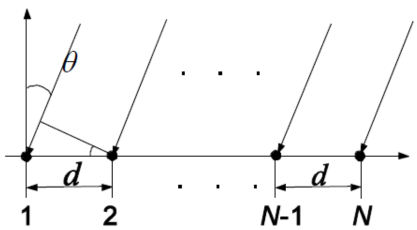

Fig.2. A narrow band wave incident on a uniform linear array of N equispaced elements[6]

Thus, estimating DOA of 2-D antenna array signals is equivalent to look for maximum values of MUSIC pseudo-spectrum $P(\theta, \phi)$:

$$P_{MUSIC}(\theta, \phi) = \frac{1}{a^H(\theta, \phi) E_N E_N^H a(\theta, \phi)} \tag{9}$$

4. Parallel implementation of MUSIC on DSP C6678

4.1. Architecture of C6678 DSP

DSP is a specialized microprocessor, widely used in embedded systems such as radar, sonar, seismic exploration, anti-jamming and wireless communication. As mentioned above, as DOA estimation is of much importance in signal processing, DSP is a good choice for it. In addition, as more and more signal processing applications are in portable devices, or embedded systems, DSP has become a dominant choice for DOA estimation.

C6678 DSP is a high-performance fixed/floating-point DSP based on TI's KeyStone multicore architecture [6]. Incorporating eight C66x DSP cores, the device runs at a core speed of 1 GHz. TI's KeyStone architecture provides a programmable platform integrating various subsystems, including up to eight C66x cores, memory subsystem, peripherals, and accelerators. Figure 3 shows a functional diagram of the C6678 DSP. For fixed-point use, the C66x core has 4x the multiply accumulate (MAC) capability of previous DSP generations. As a novelty, it integrates floating point capabilities, with a per-core raw peak performance of 32 MACs/cycle and 16 flops/cycle. The integrated floating-point capabilities makes the C66x core a suitable platform for general-purpose high performance codes.

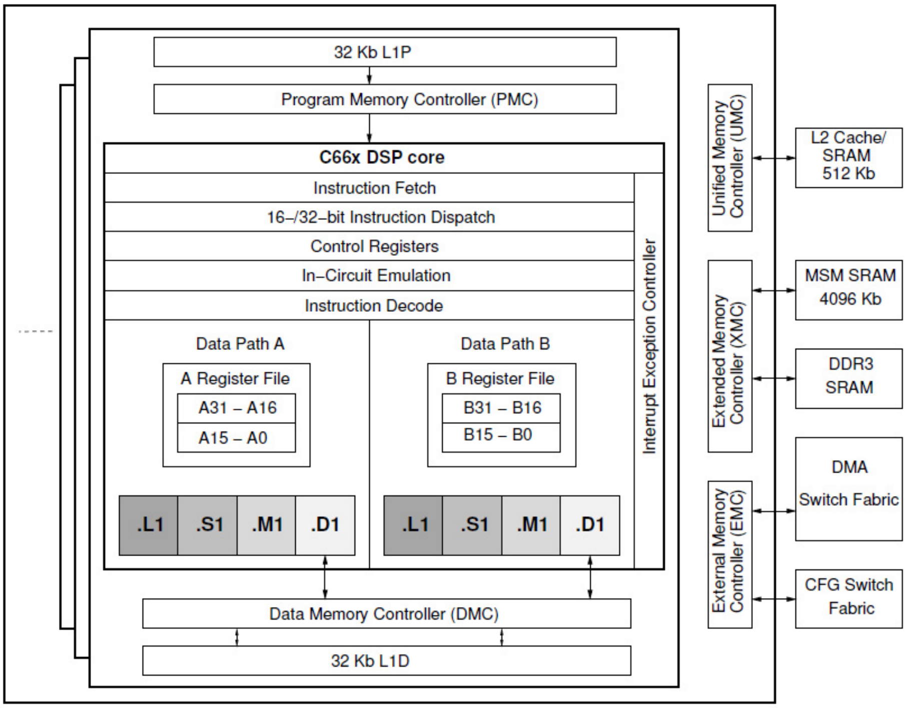

Fig.3. Functional diagram of TI C6678 DSP

New features presented in C66x DSP promote the performance of vector processing. The vector processing capability has extended the width of the SIMD instructions. C66x DSPs can execute instructions that operate on 128-bit vectors. The C66x DSP also supports SIMD operations using floating-point arithmetics. The improved vector processing capability combined with the natural instruction level parallelism of C6000 architecture (e.g execution of up to 8 instructions per cycle) results in a very high level of parallelism that can be exploited through the use of TI's C/C++ compiler.

Compared with GPUs, programming on DSP is much simpler. Meanwhile, it is fast, green and cheap. It has been proved that this potential can be applied to general-purpose high performance computing, more specifically to dense matrix computations, without major change in the exiting codes and methodologies, and with excellent performance and power consumption.

4.2. Parallel implementation of MUSIC on DSP C6678

As mentioned in Section 3, MUSIC consists of two steps: covariance decomposition and peak searching of the pseudo-spectrum.

- Covariance decomposition: we use Householder and QR factorization algorithms [7, 8] to calculate the eigenvalues and eigenvectors, because it is parallelizable and the widely used in signal processing. Compared with iterative methods, such as Krylow subspace and Jacobi decomposition method, QR factorization algorithm is based on similarity transformations. Parallel QR factorization algorithm has been implemented in ScaLAPACK. As explained in Section 3, finding the eigen of the covariance matrix is required first of all to construct the signal and noise subspaces.
- Peak searching: search the pseudo-spectrum $P(\theta, \phi)$ for the peaks which means we have to calculate the value of P at each point (θ, ϕ). That is to say, if values of θ and ϕ range from 0^0 to 90^0 and -90^0 to 90^0 with interval 1^0 respectively, we will need to calculate the value of P for (91*181) times. In each iteration of P calculation, it incurs three complex matrix multiplications where each matrix multiplication can be regarded as independent MACs. The estimating process incurs a large number of MACs. As C66x DSPs can execute instructions that operate on 128-bit vectors, in order to maximize the performance, we perform four single precision floating point MACs in one row simultaneously in a SIMD manner. Part of the code is shown below:

```
inline __x128_t mpy_four_way_example(__x128_t s, float a, float b, float c, float d)
{
__x128_t t = _fto128(a, b, c, d); // Pack values into a __x128_t
__x128_t r = _qmpysp(s, t); // Perform a four-way SIMD multiply
return r;
}
… …
inline float vec_mat(float *ar, float *ai,float *br,float *bi) {
… …
__x128_t t1 = _fto128(ar[j], ai[j], ar[j + 1], ai[j + 1]);
r1 = mpy_four_way_example(t1, br[j * M + i], bi[j * M + i], br[(j + 1) * M + i], bi[(j + 1) * M + i]);
cr[i] += (_get32f_128(r1, 3) + _get32f_128(r1, 2)+ _get32f_128(r1, 1) + _get32f_128(r1, 0));
… …
}
```

Fig.4. Part of the code for vectorization of multiple single precision MACs

In addition, function *DSPF_sp_mat_mul_cplx* in *dsplib* package is adopted in our implementation as it is specially designed for fast calculation of single precision floating point complex matrix multiplication. It vectorizes the complex matrices firstly and then uses SIMD technique to execute one signal instruction on multiple data in parallel.

5. Experiments

5.1. Data set

In the experiment, we used the sampled signal data of a simulative 2-D antenna array in L-shape, and a 1-D one in comparison.

5.2. Platform

We conducted the performance evaluation of MUSIC on a C66x DSP board using 1-D and 2-D antenna arrays.
Hardware platform: TMS320C6678 DSP with XDS560v2 Emulator.
Software: CCS5.3.

5.3. Experiment results

In 1-D DOA estimating, we set the parameters as follows:
$\lambda = 30$, $d = 0.5018 \times \lambda = 15.054$, where λ donates the wavelength and d donates the distance between the two elements.

The outcome of DOA estimating and beamforming are shown in figure 5 & 6. Two signal sources are observed in the space and their angles of arrivals are 1^o and 16^o which are consistent with the real directions of the input signals. As we can see in Figure 6, the two beams are in different forms. The execution time is 2.058ms. In the experiment, we search the DOAs from a range [-90^0, 90^0] with scanning interval 1^0.

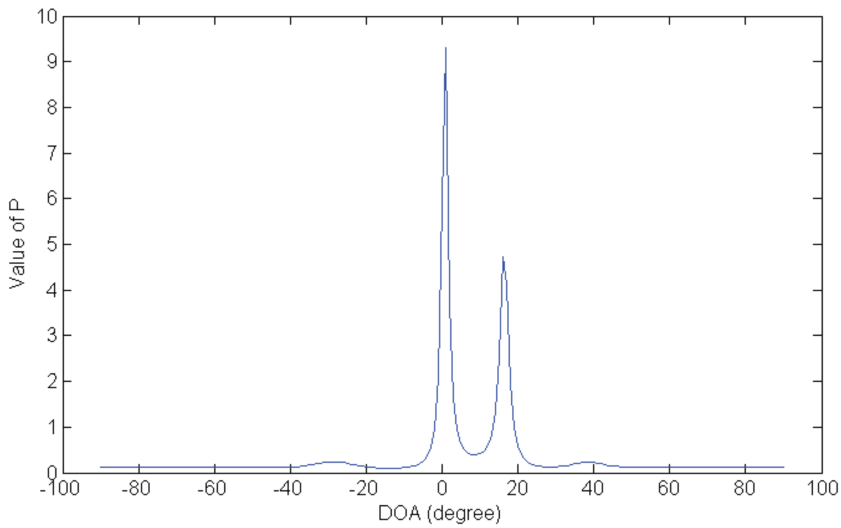

Fig.5. 1-D DOA estimation

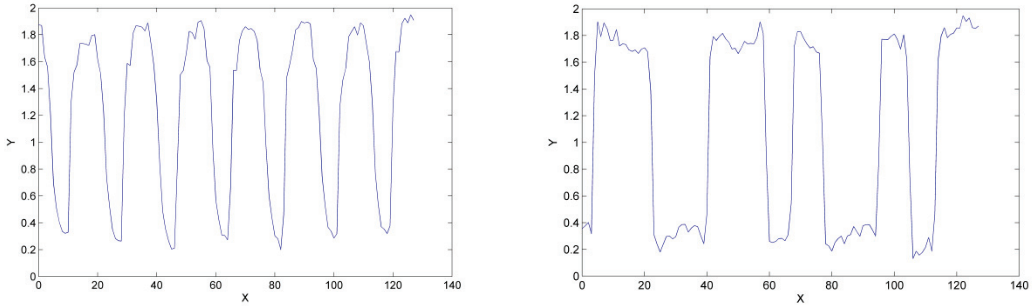

Fig.6. 1-D antenna array beaming forming

In 2-D DOA estimating, we set the parameters as follows:

$\lambda = 30, \qquad d = 0.5 \times \lambda = 15$

The outcome of DOA estimating is shown in Figure 7. In the figure, six signal sources are observed in the space and their angles of arrivals are $(50^o,-30^o)$, $(20^o,20^o)$, $(10^o,10^o)$, $(60^o,60^o)$, $(45^o,45^o)$ and $(45^o,-45^o)$ which are consistent with the real directions of the input signals. The execution time is 190.39ms. In the experiment, we search the DOAs from a range $[0^0, 90^0]$ in X axis and $[-90^0, 90^0]$ in Y axis with scanning interval 1^0 in each axis.

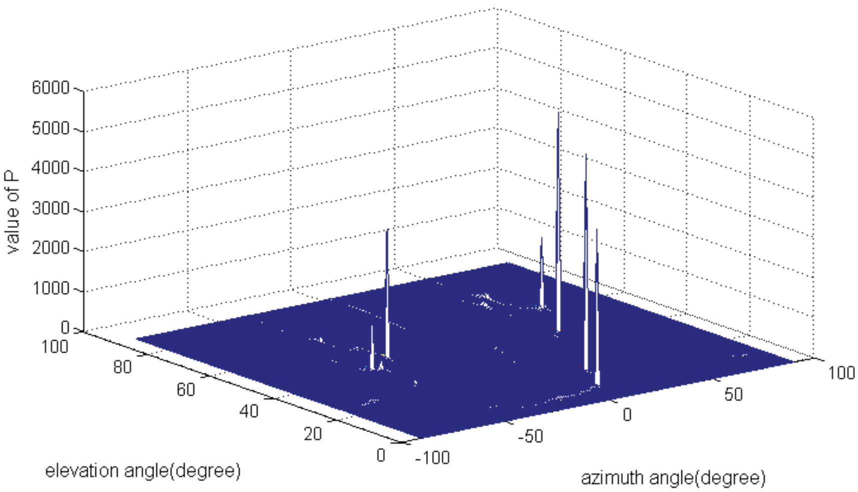

Fig.7. 2-D DOA estimation

The same 2-D DOA estimation algorithm in C were also conducted on a PC with a 3.30 GHz CPU and 8.00 GB RAM. The execution time is 63.00ms. As C6678 DSP is 1.00 GHz and with only about 2MB RAM, it is evident that our parallel implementation on DSP C6678 is very efficient and optimized. Comparing with implementations of DOA estimation on other DSPs [11, 12, 13], which cost more than 200ms or even 1s on problems with much smaller scale, our implementation showed considerable performance improvement.

6. Conclusion and future work

In this paper, we present a parallel implementation of DOA estimating of 2-D antenna arrays based on a subspace based algorithm MUSIC on DSP C6678. As DSP C6678 is the latest version up to now, it is for the first time a parallel MUSIC is implemented on it. Two main steps of MUSIC algorithm are studied and massive parallelism has been observed. Covariance matrix decomposition is parallelized by using parallel QR decomposition. Parallel peak searching is implemented by vectorizating multiple single precision floating point MACs into one row simultaneously in a SIMD manner. Finally, the exact direction of each antenna element of the array is obtained in the experiments with acceptable efficiency and higher performance than ever before.

In the future, we will explore how to use multiple DSP cores simultaneously in the DOA estimating to further improve the performance.

Acknowledgements

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