Implementation of a Real-Time System to Detect the Position of Underwater Objects

Shun-Hsyung Chang, Chin-Pin Chou, Tony Lu, Fu-Sheng Lu

Dept. of Electrical Engineering, National Taiwan Ocean University, Keelung, Taiwan, R.O.C.

Tel:886-2-2462-5180, Fax:886-2-2503-4675, Email:q0046@mail.ntou.edu.tw

Abstract - In this paper, a real-time DOA （Direction-Of-Arrival）estimation system is proposed. The algorithm we adopted is Least Squares algorithm which has better performance in azimuth estimation of signal source. A TMS320C32 DSP chip manufactured by TEXAS INSTRUMENTS is used as core of this system. And the data acquisition component is an 8 channel 12 bits A/D converter with sample rate 454ksps. The hardware circuit is accomplished using four layers printed circuit board. In the simulation, the proposed real-time system could determine the azimuth of signal source correctly.

I. INTRODUCTION

The chief technology used to detect a signal source is array signal processing. This method requires plentiful sets of A/D channels to receive the data from an array sensitive unit. Many researches used the Least Square algorithm to solve this problem such as [1-5]. Therefore, we decided to fabricate hardware with efficiency the same as the simulation. A complete set of multi-channel, high sample rate, fast-processing information systems were developed to work with the proper calculations.

Therefore, manufacturing a system of underwater acoustic detector with prompt reading ability is the main purpose of this paper.

This study used a communication receiver to detect the position of an underwater sound signal. The purpose of this system is to detect the position of a coherent signal and to apply it to a multiplied environment. This system can work independently and respond to prompt demand.

First, the Least Squares algorithm used to implement the system is introduced. Following that, each unit of this system and some experiment results are discussed.

II. LEAST-SQUARES ALGORITHM

In a linear system, the linear algebra operation is explained, if it exists, as a $z_i\,(i=1,2,\cdots,N)$ [6] pattern. The solution with the least error is acquired using the $\mathbf{X}=(\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H\mathbf{b}$ equation. This method can solve the polynomial coefficient if there is a polynomial

$$r_0+r_1x+r_2x^2+\cdots+r_mx^m=y \tag{1.1}$$

it can be expressed as following.

$$\begin{bmatrix} 1 & x_1 & x_1^2 \cdots & x_1^m \\ 1 & x_2 & x_2^2 \cdots & x_2^m \\ \vdots & \vdots & \vdots \ \ddots & \vdots \\ 1 & x_n & x_n^2 \cdots & x_n^m \end{bmatrix} \begin{bmatrix} r_0 \\ r_1 \\ \vdots \\ r_m \end{bmatrix} = \begin{bmatrix} y_1 \\ y_2 \\ \vdots \\ y_m \end{bmatrix} \tag{1.2}$$
$$\Rightarrow A \qquad\qquad X = b$$

The equation $\mathbf{X}=(\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H\mathbf{b}$ will solve the coefficient $[r_0, r_1, r_2, \cdots, r_m]$ with the least error. A new method [7] is used in this section that modulates the information received from the detection units to (1.2), solves the coefficient with the Least-Squares Algorithm and finds the incidence angle of the coherent signal with this coefficient.

The number of array detection units is M; the number of coherent signals is N; the output of the i^{th} detection unit during the k^{th} period is $y(k)=[y_1(k), y_2(k), \cdots, y_M(k)]^T$

Thus,

$$\mathbf{y}(k) = \mathbf{AS}(k) \qquad k = 1,2,\cdots \tag{1.3}$$

where $\mathbf{A}=[a_j^{i-1}]_{N\times M}$ is Vandermonde Matrix.

$$a_p^{i-1}=\exp\left[-j\phi_{i-1}(\theta_p)\right]$$
$$\phi_i=k\omega_0\frac{d}{c}\sin\theta_i \qquad k=1,2,\cdots$$
$$\mathbf{S}=[\alpha_1 s_1,\alpha_2 s_1,\cdots,\alpha_N s_1]^T=\alpha s_1$$

d is the distance among receivers, c is the signal transmission speed, α is plural, representing the amplitude of the signal and the delay angle, under the $\alpha_1=1$ condition. If noise is taken into consideration, $\mathbf{X}(k)=\mathbf{AS}(k)+\mathbf{N}(k)$. $\mathbf{N}(k)$ stands for white Gaussian noise. In the beginning, the autocorrelation matrix $\mathbf{R}_{yy}=\{r_{i,j}\}_{N\times N}$ is solved, which is the signal received with no noise. That is:

$$\mathbf{R}_{yy}=\mathbf{AR}_{ss}\mathbf{A}^H \tag{1.4}$$

$$\mathbf{R}_{ss}=E[\mathbf{SS}^H]=\sigma_s^2\alpha\ \alpha^H \tag{1.5}$$

Obviously, the rank of R_{yy} is 1; therefore, the method in the last section cannot be applied. From (1.4) and (1.5), $r_{i,j}$ can be written as

$$r_{i,j} = \sum_{l_1=1}^{N}\sum_{l_2=1}^{N} a_{l_1}^{i-1} a_{l_2}^{(j-1)*} \alpha_{l_1} \alpha_{l_2}^{*} \sigma_s^2 \tag{1.6}$$

To define $\mathbf{h}_i = [a_1^i, a_2^i, \cdots, a_N^i]$, we can get

$$r_{i,j} = \sigma^2 \mathbf{h}_{i-1} \alpha \alpha^H \mathbf{h}_{j-1}^H \tag{1.7}$$

From the array of $\mathbf{R}_{yy}$ and the element of the i^{th} row, we can define the following matrix and vector.

$$\mathrm{Q}_{(i)} = \begin{bmatrix} r_{i,1} & r_{i,2} & \cdots & r_{i,N} \\ r_{i,2} & r_{i,3} & \cdots & r_{i,N+1} \\ \vdots & \vdots & \ddots & \vdots \\ r_{i,M-N} & r_{i,M-N+1} & \cdots & r_{i,M-1} \end{bmatrix} \tag{1.8}$$

$$\mathrm{C}_{(i)} = \begin{bmatrix} r_{i,N+1} \\ r_{i,N+2} \\ \vdots \\ r_{i,M} \end{bmatrix} \tag{1.9}$$

Apparently, each row of $\mathbf{Q}_{(i)}$ corresponds to each element of $\mathbf{C}_{(i)}$. The following element is the increased -power arrangement for the previous element. Take (1.2) for example. The system represents a polynomial with a a_i($i = 1,2,\cdots,N$) root. Thus, we can produce a coefficient $\mathbf{Y}_N$ whose corresponding root is a_i ($\mathrm{i} = 1,2,\cdots,N$). It follows that

$$\mathbf{Q}_{(i)} \mathbf{Y}_N = \mathbf{C}_{\mathbf{(i)}} \tag{1.10}$$

To increase the precision and resolution of the algorithm, every row of $\mathbf{R}_{yy}$ is adjusted to $\mathbf{Q}_{(i)}$ and $\mathbf{C}_{(i)}$ and spread $\mathbf{Q}_{(M)}$ to be $\mathbf{C}_{(M)}$.

$$\mathbf{Q}_{(M)} = [\mathbf{Q}_{(1)}; \mathbf{Q}_{(2)}; \cdots; \mathbf{Q}_{(N)}] \tag{1.11}$$

$$\mathbf{C}_{(N)} = [\mathbf{C}_{(1)}; \mathbf{C}_{(2)}; \cdots; \mathbf{C}_{(N)}] \tag{1.12}$$

Therefore,

$$\mathbf{Q}_{(M)} \mathbf{Y_N} = \mathbf{C_{(M)}} \tag{1.13}$$

where the rank of $\mathbf{Q}_{(N)}$ is min (N, M-N).

This algorithm is also a robust method. We assumed that the number of actual signals is N and the number of preset signals is N_p.

When $N_p > N$, the incident angle of the N signal sources can be solved only if the number of detection unit is greater than twice the number of signal sources. From linear algebra, when the problem conforms to the limitation of $M - N > N$ that can only decide $\mathbf{Y}_{(N)}$. If we want to detect M signals with this algorithm, the number of detection units must be twice the number (M) of signal sources. This means $M \geq 2N$.

The algorithm procedure follows:

A. Estimate the array detection unit. Output the covariance matrix $\hat{\mathbf{R}}_{xx}$. Then Estimate $\hat{\mathbf{R}}_{yy}$ from $\hat{\mathbf{R}}_{xx}$.

B. Get $\mathbf{Q}_{(N)}$ and $\mathbf{C}_{(N)}$ from $\hat{\mathbf{R}}_{yy}$ matrix.

C. With LS to get $\hat{\mathbf{Y}}_N$, then

$$\hat{\mathbf{Y}}_N = (\hat{\mathbf{Q}}_N^H \hat{\mathbf{Q}}_N)^{-1} \hat{\mathbf{Q}}_N^H \hat{\mathbf{C}}_N \tag{1.14}$$

D. We can apply the coefficient $\hat{\mathbf{Y}}_N = [-y_N, -y_{N-1}, \cdots, 1]$ to the polynomial

$$w^N - y_1 w^{N-1} - \cdots - y_N w - y_N = 0 \tag{1.15}$$

By solving the root of the polynomial or from the pseudo-spectrum function $P(\theta)$, we can find the peak value and get the signal azimuth.

$$P(\theta) = \frac{1}{\left|\mathbf{w}^T(\theta)\hat{\mathbf{Y}}\right|^2} \tag{1.16}$$

where $\mathbf{w}(\theta) = [1, e^{j\omega_0\tau}, \cdots e^{j\omega_0 N\tau}]^T$, $\tau = \frac{d}{c}\sin\theta$, ω_0 is the central frequency of the signal source. c is the speed of signal transmission. And $\hat{\mathbf{Y}} = [-\hat{\mathbf{Y}}_N^*; 1]$

III. COMPUTER HARDWARE CIRCUIT DESIGN

A. TMS320C32 Processor

The core of this designed system is the TMS320C32-50 Digital Signal Processing (DSP) Chip, which is a 32-bit floating point DSP chips produced by Taxis Instruments (TI). The chip advantages are high efficiency, strong exterior peripherals, and more memory flexible design. The entire circuit structure is shown in Fig 1. The following will discuss each individual square function in Fig 1.

B. A/D Converter Circuit

The A/D converter is divided into three types: dual slope, charge balancing converter and flash converter. The flash converter is the fastest. The A/D converter used in this system is the AD 7891, a flash converter, produced by the Analog Device Company. It has 8 analog signal input channels. The voltage range is ± 10 and resolution is 12-bits. That means that the individually corresponding voltage ranges from 000H to FFFH is –10V and 10V.

C. Sampling and Holding Circuit

Each resembling input channel of the system adds a sampling and holding circuit and adopts the number LF398 IC.

D. Decoder Circuit

For flexible and convenient design, GAL (General Array Logic) was used to design the decoder. The number of GALs used by this system was 22V10 and 16V8. PALASM software is used, produced by the AMD Company, to design the decoder.

E. RAM

This system uses two GLT710016 RAM, which contain 64K × 16 bytes memory, produced by the G-LINK Company. The purpose for using two SRAM is to make up the 32 byte width in order to accelerate the access time.

F. ROM

AM29F010-70 Flash ROM, produced by the AMD Company, was used as the external memory to access the code. Its capacity is 128K × 8 bytes and its main function is to enable the system to operate independently. Figure 2 shows the external ROM wiring.

G. Emulator

C32 adopts the MPSD technique to achieve emulator XDS510 connection. Thus, when designing the circuit, it is necessary to connect the pins (EMU1, EMU2, EMU3, H3) of chip C32 and adjust them to the MPSD 12 PIN standard to develop the program and debug using the XDS510 emulator, produced by the TI Company.

H. LCD

The LCD can be divided into two molds: words display and picture display. The 16×2 words system was used in this process. The LCD shows the angle that it estimates.

I. RS232

Serial transmission was adopted within the C32 and adjusts the serial data using software. The TTL C32 data standard must be added to ICL232 IC as the standard voltage transformation to fit the logically high potential –12v and logically low potential +12v.

IV. SOFTWARE DESIGN

A. Program developing procedure

Figure 3 shows the program development procedure. The software in the system, develop in this study, partly adopts assembly language. The development process was researched and produced using CODE computer, cooperating with the XDS510 Emulator produced by TI. Among them, the CODE Composer is developing software that can operated and developed easier in the Windows than in DOS, and is easier to debug.

B. Software Program Plan

In this system, the program plan involved dividing the program into several sub-programs and saving them in different files. The program files included main programs, hardware programs and DOA algorithm program. We adopted the LS method and matched it with the multinomial solution method to carry out direction of arrival (DOA) for the source. In DOS circumstances in the PC, C language was used to achieve a program that could communicate with C32 and receive the information and calculation result transmitted from the DOA circuit. The pseudo-spectrum was shown on the monitor. Figure 4 shows the complete software structure.

C. The main program procedure.

Figure 5 shows the complete main program procedure. In the beginning, the program will set up initially for the circuit, including the serial baud rate, the A/D converter sampling rate, and the break off control set up for outside and inside parts. INT1 and INT2 are enabled to break off to start the A/D converter and receive 500 pieces of data. INT1 and INT2 are then disabled to break off to shut down the A/D converter. After that, the program will judge whether to transmit the data to the PC or not. It will enable XINT to break off starting serial transmission and transmit data. If not, it will directly transform the data from the A/D converter in a floating point. The DOA algorithm is operated to calculate the signal source azimuth, and shows the angle on the LCD.

D. DOA Algorithm Program

This system focuses mainly on estimating the sound source azimuth in the water. This is inevitable for the multi-path effect in the water. Therefore, it is necessary to adopt DOA algorithm that can be used in the coherent signal. It is easy to adopt assembly language. We choose the LS method to achieve the source signal arrival direction. The judgment for the last angle involves solving a multinomial to calculate the signal incidence angle to increase the rate of accuracy and save the time on scanning pseudo-spectrum. Figure 6 is the DOA algorithm procedure.

In the DOA algorithm, sub-program is necessary to transform the received signal into the Hilbert transform to get an analytical signal to estimate the correlation matrix **R**,

$$\overline{\mathbf{R}} = \frac{1}{N}\sum_{1}^{N} \mathbf{X}\mathbf{X}^{H} \tag{4.1}$$

further use (1.8)(1.9)(1.11)(1.12) to get $\mathbf{Q}_N$ and $\mathbf{C}_N$. Then (1.14) is used for a vector $\mathbf{Y}_N$ to get $\mathbf{Y} = [-\mathbf{Y}_N^*;1]$. This vector is used as a coefficient to solve the root and incidence angle and save the angle into the planned memory.

V. THE EXPERIMENTAL RESULTS AND ANALYSIS

The actual test was divided into two parts. The first part is the communication mode simulation test. The second is the actual test in a trough. The test environment and its set up are illustrated in detail and the results are presented in the following.

A. TIMS (Telecommunications Instructional Modeling System) simulation

In this experiment, the signal was produced from a waveform production that is a 5kHz-frequency and 5V-amplitude sine-wave, directly transmitted into four A/D channels to stimulate the situation when source incidence from 0 degree. The DOA system that we accomplished is shown in Fig.7. Figure 8 is the setting for the entire simulation. The signal was received using four A/D channels to process the DOA algorithm calculation and transmit its result to the PC and display the pseudo-spectrum on the screen. The result is shown in Fig.9. The computer screen displayed the peak value existing when in 0 degree.

A second experiment involved taking a 5kHz-frequency and 5V- amplitude sine-wave, through three phase shift mode to individually delay the signal for 90 degree, 180 degree, and 270 degree to produce four sets of incident signals. When the delay occurred in simulating a signal for a 30 degree incident array unit azimuth direction, it was found that there a peak value occurred when the signal was 30 degrees. Figure 10 is its result

B. Water Trough Simulation

In this experiment, the signal was produced from a waveform production from a 5kHz-frequency and 5V-amplitude sine wave. A power amplifier was used to strengthen the power. The signal was sent to a hydrophone 8105 in the water. The delivered signal was received by the four hydrophones, through the power amplifier to amplify the weak signal. The four signals were sent into the system to determine the direction of arrival. The data was saved into the memory. The developed algorithm was used to calculate the signal source azimuth and display it on the LCD monitor. Figure 11 displays the trough equipment.

The setup for the test parameters is as follows:

1. The number of sensing unit: 4 hydrophones.
2. The distance for sensing unit: 15 cm
3. Measuring distance: 2m
4. Frequency of signal source: 5kHz sine wave
5. Amplitude of signal source: 5V
6. The angle of incidence: 0 degree

Figure 12 shows a picture from the actual experiment. The waveforms of the signal received from the four A/D channels are shown in Fig.13. The direction of arrival calculation by the algorithm is shown on the LCD, as shown in Fig.14. The display value is –1.9 degrees.

VI. CONCLUSION

Underwater communications are important. Seabed resources must be developed for the future. Owing to the complex environment under the sea, it is still a big challenge to achieve correct information transference and increase the quality of communication using only sound waves carriers. This paper researched source arrival directions for underwater sounds. A complete hardware circuit and software processing system was designed. Using Direction of Arrival (DOA), the azimuth of a coherent signal was detected. The proposed method can be applied in multi-path environments and it can achieve accuracy using a hardware circuit in cooperation with software. The azimuth of a signal was detected using TIMS simulation. However, the effect was worse if the test was conducted in a water trough. The main reason is that the water trough environment had a smaller area that produced serious reflection. We used only four hydrophones, making the analysis worse. In the simulation, we proved that if the number of sensitive units was enough, the signal source azimuth could be correctly estimated.

This paper contributes an algorithm and a real-time system. We are currently waiting for initial underwater communication technique development stage completion. IEEE Ocean Engineering Society also thinks highly of this underwater communication technique development.

References

[1]James A. Cadzow, "Methods of Direction-Of -Arrival," *IEEE Pacific Rim Conference on Communications, Computers and Signal Processing*, May 9-10, 1991, pp25-28

[2]Karim Abed-Meraim, Yingbo Hua, and Adel Belouchrani, "Second-Order Near-Field Source Localization: Algorithm and Performance Analysis," *Signals, Systems and Computers, 1996. 1996 Conference Record of the Thirtieth Asilomar Conference on*, vol. 1, 3-6 Nov 1996 pp. 723 -727 vol.1

[3]R.K. Mallik, K.V. Rangarao and U.M.S. Murthy, "DOA Estimation by Least Squares Approach," *Electronics Letters* vol.34, No.12,11th June 1998, pp. 1187-1189

[4]Ming Jian, Alex C. Kot, Meng H. Er, "DOA Estimation of Speech Source with Microphone Arrays," *Circuits and Systems, 1998. ISCAS '98. Proceedings of the 1998 IEEE International Symposium on* , vol. 5 , 31 May-3 Jun 1998 pp. 293 -296 vol.5

[5]Youngmo Han and F.C. Park, "Least Squares Tracking on the Euclidean Group," *IEEE Transactions on Automatic Control*, vol.46,No.7, July 2001, pp.1127-1132

[6]W. K. Nicholson, "Linear algebra with application," 3rd ed. ,PWS Publishing Boston 1995.

[7]W. K. Lai and P. C. Ching, " A New approach for coherent direction-of-arrival estimation, " *International Symposium of Circuits and Systems*, 1998. ISCAS '98, vol. 5, pp. 9-12, Mar 1998.

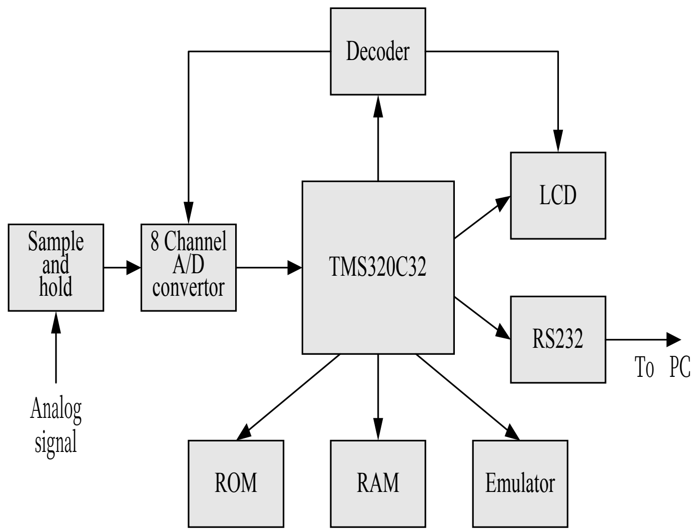

Fig. 1: The complete hardware structure

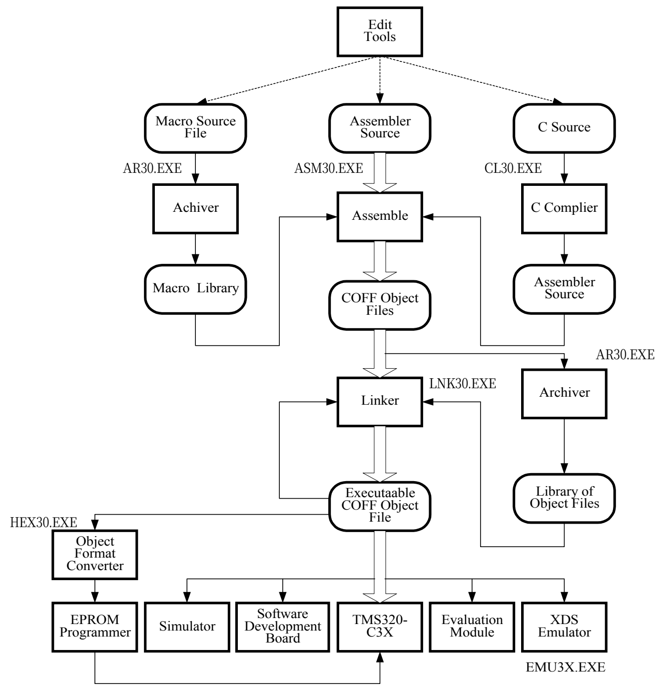

Fig 3: Structure of software development

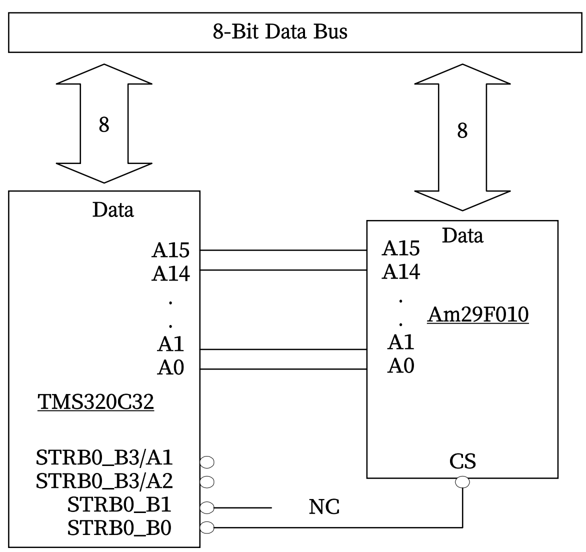

Fig 2: Wiring figure of external ROM

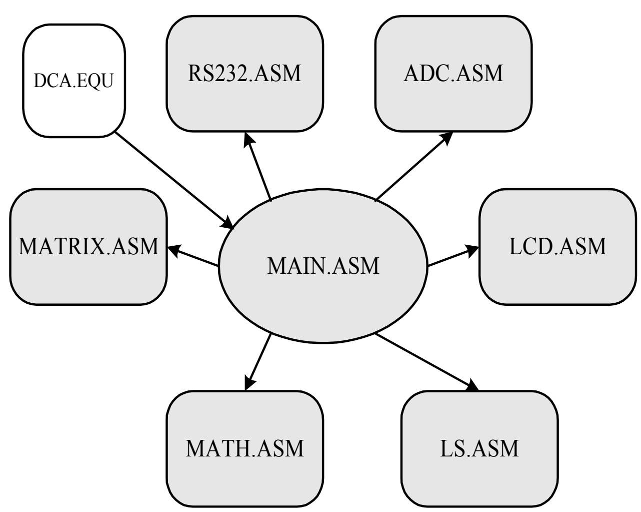

Fig. 4: Software structure

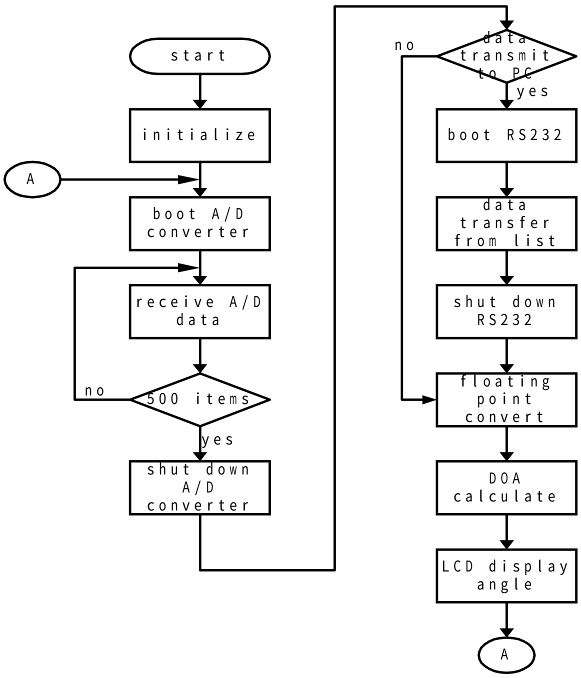

Fig. 5: The procedure for the main program

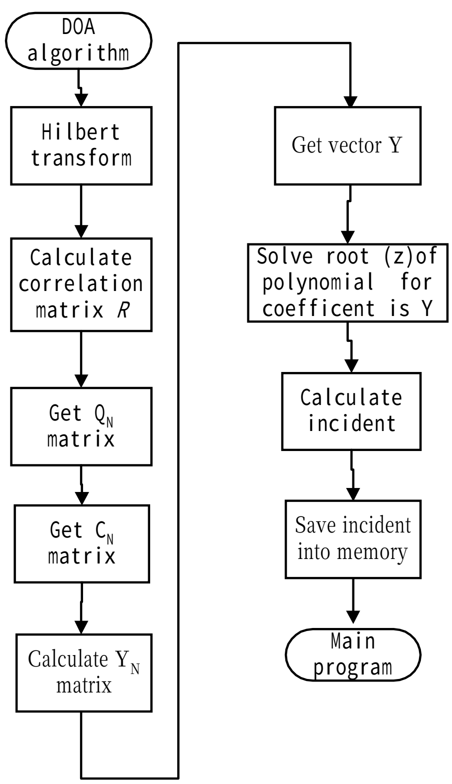

Fig. 6: The procedure of DOA algorithm in this program

Fig.7:The DOA system

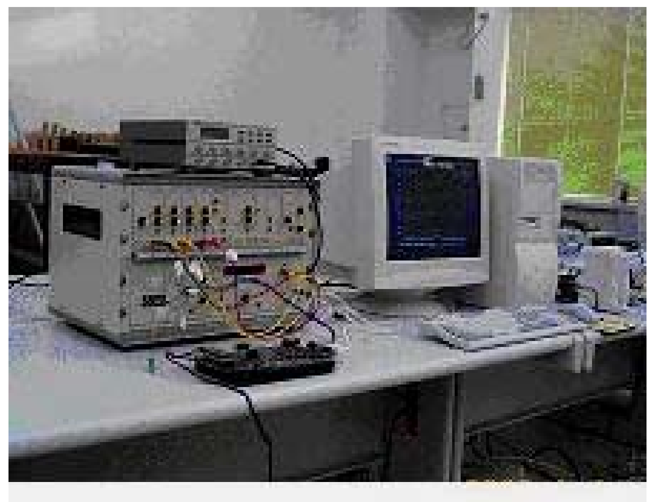

Fig. 8: The furnishing of TIMS simulation

Fig. 9: The result of TIMS simulation(0 degree)

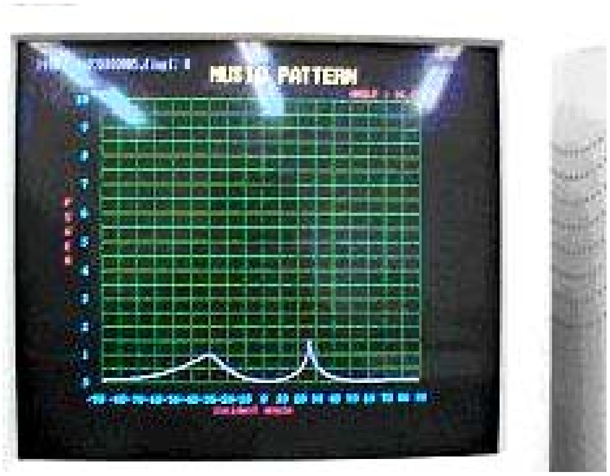

Fig.10: The result of TIMS simulation (30 degree)

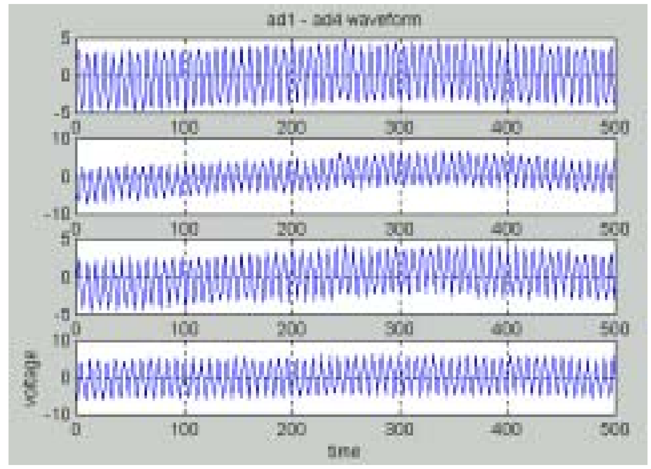

Fig. 13: Waveform of four AD channels

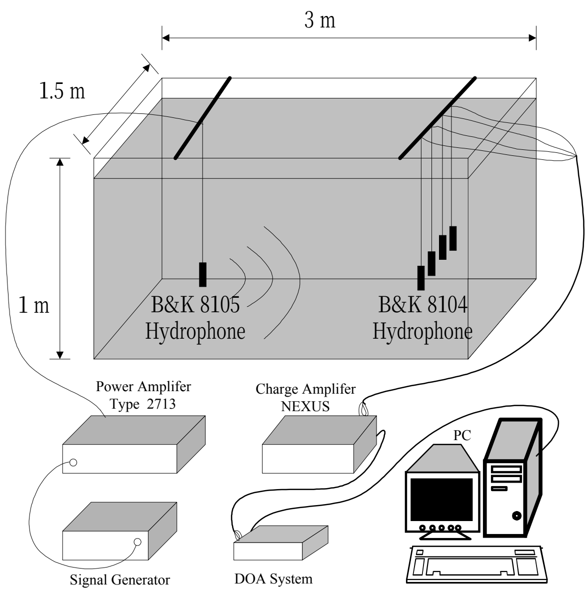

Fig.11: The equipment display of water trough

Fig. 14: The angle of LCD display

Fig. 12: The actual simulating environment in water trough.