Audiobeam loudspeaker system based on TMS320VC5509A

ZHAO Hong-liang(赵洪亮), TIAN Xiang-e(田祥娥), FENG Guo-jin(冯国金), SUN Ke(孙 珂)

(*College of Information and Electrical Engineering, Shandong University of Science & Technology, Qingdao 266510, China*)

Abstract: An audiobeam loudspeaker system based on the fixed-point digital signal processor (DSP) TMS320VC5509A (VC5509A for short) is developed to emit audible sound with high directivity along a selected path. This system is composed of preamplifier, anti-aliasing filter, ADC (analog to digital converter), VC5509A basic system, DAC (digital to analog converter), smoothing filter, ultrasonic power amplifier and ultrasonic transducer array. VC5509A is used to implement the key signal preprocessing algorithms, such as signal preprocessing, amplitude modulation, carrier signal generator, etc. The experiments show that this system has the advantages of high accuracy and good stability.

Key words: parametric loudspeaker; audiobeam; digital signal processor(DSP)

CLD number: TN643 **Document code:** A

Article ID: 1674-8042(2012)01-0057-05 **doi:** 10.3969/j.issn.1674-8042.2012.01.012

Audiobeam loudspeaker, which is different from traditional loudspeaker emitting sound waves in all directions, can generate audible sound wave with high directivity in a specified direction just like a beam of light. It is also called soundbeam or parametric array system.

Research about audiobeam loudspeaker has a long history. In 1962, the concept of "parametric array" was first proposed by Westervelt, providing theoretical basis for high directivity audiobeam[1]. In 1965, Berktay gave out the simplified propagation formula of audiobeam which was called "Bertay's far-field solution"[2]. In 1983, the feasibility of audiobeam loudspeaker was verified by Yoneyama et al. with experiment. A directivity loudspeaker system, using a transducer array which consisted of 547 bimorph transducers, was developed by Yoneyama and put into practical use[3]. Subsequently, several problems such as the selection of optimal carrier signal, signal distortion preprocessing methods and ultrasonic exposure issues were discussed[4-7]. In 1999, Pompei proposed a preprocessing approach which integrating the modulation signal twice and taking the square root to remove distortion[8]. In 2002, Pompei proposed a design to suppress grating lobes in an array antenna with element spacing greater than a half wave-length[9,10]. In 2003, Tan et al. in Nanyang Technological University proposed an algorithm to control the sidelobe level of the difference frequency directivity[11]. In 2004, Karnapi et al. in Nanyang Technological University designed and implemented the digital signal processing subsystem in FPGA platform[12].

In recent years, studies related to audiobeam loudspeaker have been carried out by domestic researchers. Much progress has been made in studies on audiobeam algorithm, acoustic field calculation and production development by some research institutions, such as Shandong University of Science and Technology[13-15], Institute of Acoustics, Chinese Academy of Sciences and University of Electronic Science and Technology of China[16-18].

Signal processing algorithm is realized by analog circuit in conventional audiobeam loudspeaker system. There are considerable difficulties in designing analog circuit due to the complexity of signal processing algorithm. Disadvantages of inconvenience to debug, bad stability and long development cycle also restrict the study of audiobeam loudspeaker system.

In Shandong University of Science and Technology, studies about audiobeam loudspeaker system have been carried out since 2004. JI Pei-feng studied preprocessing algorithm by theoretical analysis and computation simulations[13]. ZHU Hai-sheng and ZHAO Hong-liang designed and implemented audiobeam loudspeaker based on TMS320VC33 chip[14,15].

A design scheme of audiobeam loudspeaker system based on VC5509A is described in this paper. Key signal processing algorithms are implemented on DSP chip.

※ **Received data:** 2011-08-18
Foundation item: Science and Technology Development Plan of Shandong Province (No. 2010GSF10809)
Corresponding author: ZHAO Hong-liang (zhl6401@126.com)

1 Principle of audiobeam

In traditional loudspeaker, sound frequency electric signal is directly converted to audible sound in the air. Principle of audiobeam loudspeaker is quite different from traditional loudspeaker in that sound frequency electric signal is modulated with carrier signal before being radiated into air by ultrasonic transducer array. As shown in Fig. 1, sound frequency signal $x(t)$ is modulated by ultrasonic frequency carrier signal $x_c(t)$, and then the modulated signal $y(t) = x(t)x_c(t)$ with high directivity is emitted by ultrasonic transducer array. Next, the sound signal modulated onto carrier signal is demodulated into audible frequency $p(t)$ with high directivity in the air. But serious distortion has been introduced to the represented audible sound $p(t)$ due to nonlinearity of air demodulation. The signal preprocessing module in Fig. 1 is designed to preprocess sound signal before modulation process to reduce distortion. Amplitude-adjusted module is used to control the modulation depth in a proper range according to input audio signal amplitude, and ensure that the system has high power conversion efficiency.

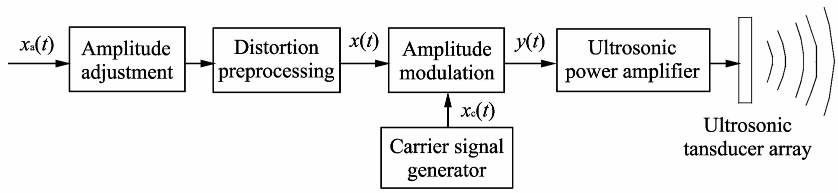

Fig. 1 Audiobeam loudspeaker system

2 Hardware design of audiobeam

2.1 Overall structure

As shown in Fig. 2, audiobeam loudspeaker system hardware consists of eight modules: preamplifier, anti-aliasing filter, ADC, VC5509A basic system, DAC, smoothing filter, ultrasonic power amplifier and ultrasonic transducer array.

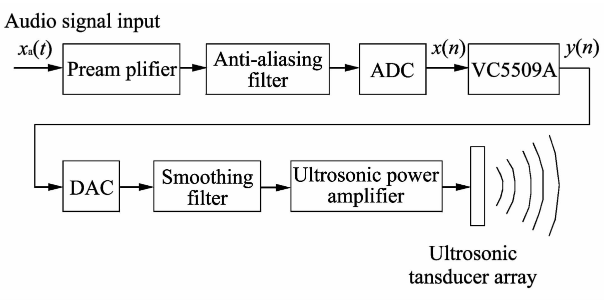

Fig. 2 Overall block diagram of the system hardware

First, the voltage of audible sound signal $x_a(t)$ is amplified from a few tenths of a volt to several volts and then processed by anti-aliasing filter and ADC, and converted to digital signal $x(n)$. Next, it is processed by VC5509A, such as amplitude adjustment, distortion precompensation, amplitude modulation; and the output is digital signal $y(n)$. After it goes through digital-to-analog converter and smoothing filter, the modulated digital signal is sent to ultrasonic power amplifier and then emitted by the ultrasonic transducer array with centre frequency at 40 kHz.

2.2 VC5509A basic system

VC5509A basic system is shown in Fig. 3 including power circuit, clock circuit, reset circuit, expanding SDRAM, Flash memory, JTAG emulator interface and CPLD logic control unit.

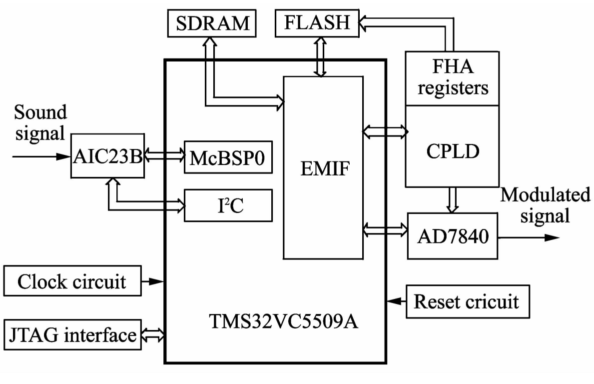

Fig. 3 Block diagram of VC5509A basic system

In the system, we adopt the Hynix company's 16-bit SDRAM HY57V641620 for temporary storage of preliminary information. Its capacity is 64 Mb. Flash memory used in the system is the 8 Mb, 3.0 volt-only boot sector flash memory S29AL008D produced by Spansion for permanent storage of important data and program.

2.3 Design of ADC and DAC

TLV320AIC23B which has signal up to 90 dB to noise ratio (SNR) at audio sampling rates up to 96 kHz is applied in this system to implement ADC. AIC23B is a high-performance stereo audio codec with highly integrated analog functionality. It can be controlled through software by TI McBSP-compatible multiprotocol serial port. The audio data is input and output via TI McBSP-compatible programmable audio interface, and supports 16, 20, 24 and 32 bits data word length with sample rates 8 – 96 kHz.

DAC is used for the digital-to-analog conversion of 40 kHz amplitude modulated signal, so its sample rate should be no less than 80 kHz. We choose 14-bit high performance ADC AD7840 which is capable of 14-bit parallel and serial interfacing. It is adequate to use 200 kHz sampling rate for the DAC.

2.4 Design of anti-aliasing filter and smoothing filter

Traditional filters are implemented with operational amplifiers or RC circuits. There are several drawbacks in the method which needs many electronic components and complex parameters adjusting, and filter characteristics are affected by stray capacitances. To overcome these disadvantages, we implement anti-aliasing filter and smoothing filter with monolithic integrated active filter chip MAX274.

MAX274 is an 8th-order continuous-time active filter containing four identical 2nd-order filter sections. Each section can implement bandpass or lowpass filter response, such as Butterworth, Bessel, Chebyshev and Elliptic by selecting four proper external resistors. Filter design software has been published by Maxim Integrated Products to assist filter design and simulation. With the help of the software, we can get corresponding filter design parameters by inputting filter specifications.

Anti-aliasing filter and smoothing filter are lowpass filters. The specifications are shown in Table 1 and Table 2.

Table 1 Anti-aliasing filter specifications

Passband edge frequency	$f_{1p} = 4$ kHz
Stopband edge frequency	$f_{1s} = 8$ kHz
Maximum passband attenuation	$\alpha_{1p} = 3$ dB
Minimum stopband attenuation	$\alpha_{1s} = 20$ dB

Table 2 Smoothing filter specifications

Passband edge frequency	$f_{1p} = 44$ kHz
Stopband edge frequency	$f_{1s} = 48$ kHz
Maximum passband attenuation	$\alpha_{1p} = 3$ dB
Minimum stopband attenuation	$\alpha_{1p} = 20$ dB

3 Signal processing algorithm

3.1 Signal preprocessing algorithm in audiobeam loudspeaker

In 1965, Berktay proposed far-field solution[2] of parametric array as

$$p(t) \propto \frac{d^2}{dt^2}[E(t)]^2, \qquad (1)$$

where $E(t)$ can be termed the modulation envelope signal. The demodulation resulting audible sound $p(t)$ is proportional to the second derivative square of the modulation envelope square.

Early audiobeam loudspeaker used simple AM modulation to generate the audible signal as

$$S_{AM}(t) = [1 + mg(t)]\cos(w_c t), \qquad (2)$$

where $g(t)$ is the audible signal to be reproduced (assumed to be normalized to unity magnitude), and $m C$ is the modulation depth, usually taken as one or slightly than one. And $\cos(w_c t)$ is the carrier signal.

A major shortcoming of this method is that when no audible sound is intended to be reproduced ($g(t)=0$), the system still outputs high levels of ultrasound. The output signal is pure ultrasound carrier as

$$p(t) = \cos(w_c t). \qquad (3)$$

To alleviate this shortcoming, it has been proposed to use a modulation envelope which contains an envelope follower. Adding the envelope signal to the audio signal provides a suitable offset to keep the signal positive before modulation. The resulting sound generated is

$$E(t) = e(t) = g(t), \qquad (4)$$

$$p(t) \propto \frac{d^2}{dt^2}[e(t) + g(t)]^2. \qquad (5)$$

As pompei proposed in American patent US7596228[19], the frequency of envelope $e(t)$ should be lower than 100 Hz in this signal preprocessing algorithm, and a lowpass filter is used to restrict the frequency of $e(t)$.

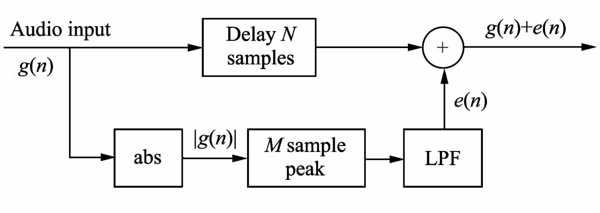

Fig. 4 Block diagram of signal preprocessing algorithm

Block diagram of signal preprocessing algorithm we used in the system is illustrated in Fig. 4. The delay time N corresponds to the group delay of the low pass filter.

3.2 Carrier signal generation algorithm

Carrier signal is 40 kHz sinusoidal generated by recursive algorithm illustrated in Fig. 5.

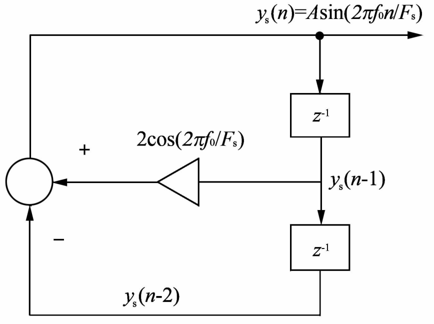

Fig. 5 Block diagram of recursive algorithm

It is the most accurate and efficient method of generating a sinusoidal waveform.

The output equation is

$$y_s(n) = A\sin(2\pi f_0 n/F_s), \tag{6}$$

where f_0 is carrier frequency, F_s is system sampling rate, y_s is system internal variable updated as

$$y_s(n) = 2\cos(2\pi f_0/F_s) \cdot y_s(n-1) - y_s(n-2), \tag{7}$$

with initial conditions

$$y_s(1) = A\sin(2\pi f_0/f_s), \tag{8}$$

and

$$y_s(0) = 1. \tag{9}$$

3.3 Amplitude modulation algorithm

Signal $y(n)$ sent to DAC from VC5509A is the result of signal $E(n)$ modulated by carrier $x_c(n)$ within ultrasonc frequency band. The amplitude modulation algorithm in DSP is expressed by

$$y(n) = E(n)x_c(n). \tag{10}$$

3.4 System sampling rate selection

In this system, sampling rate is divided into two parts (low frequency and high frequency) according to work frequency. Low frequency portion comprises ADC, gain adjustment and distortion preprocessing, in which the highest signal frequency is 8 kHz, and sampling rate is 32 kHz. The high frequency portion consists of carrier signal generation, amplitude modulation and DAC. The highest signal frequency is 40 kHz + 8 kHz = 48 kHz, and sampling rate is 200 kHz.

4 Experimental results

The ultrasonic transducer array consists of 90 ultrasonic sensors produced by Shanghai Nicera Sensor Co. Ltd. with resonant frequency at 40 kHz. A front view of the array appears in Fig. 6.

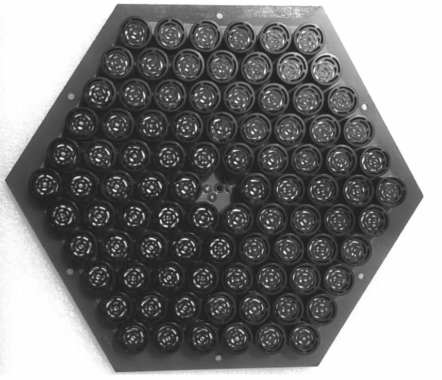

Fig. 6 Ultrasonic transducer array

Fig. 7 shows the sound pressure picked up by condenser microphone at a point 2.5 m from the array while the angle between microphone and central axis changes from − 80° to 80°. The input signal of system is sinusoidal signal of 2 kHz and modulation depth m is 0.8. The microphone is on the central axis when the angle is 0°. Fig. 7 reflects the directivity of audiobeam loudspeaker system. The points in the figure show the angles when the sound pressure decays 3 dB and the angle is half-power angle. From Fig. 7, we can see that the half-power angle of the system is smaller than 5°.

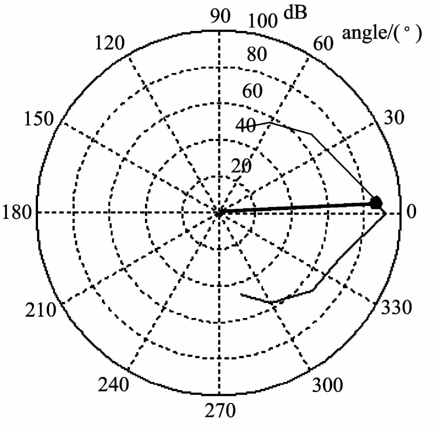

Fig. 7 Directivity of audiobeam loudspeaker system

5 Conclusion

An audiobeam loudspeaker system is designed based on TMS320VC5509A, in which the key signal processing algorithms are implemented on one VC5509A chip, such as signal preprocessing, carrier signal generator and signal amplitude modulation, etc. It has high precision and well stability.

References

[1] Westervelt P J. Parametric acoustic array. Journal of the

Acoustical Society of America, 1963,35(4):535-537.
[2] Berktay H O. Possible exploitation of nonlinear acoustics in underwater transmitting applications. Jounal of Sound and Vibration, 1965,2(4):435-461.
[3] Yoneyama M, Fijimoto J, Sasabe S. The audio spotlight: an application of nonlinear interaction of sound waves to a new type of loudspeaker design. Jounal of Acoustical Socity of American, 1983,73 (5): 1532-1536.
[4] Kamakura T, Tasahide Y, Ikegaya K. Studies for the realization of parametric loudspeaker. Journal of the Acoustical Society of American, 1985, 41(6):378-385.
[5] Kamakura T, Aoki K, Kumamoto Y. Suitable modulation of the carrier ultrasound for a parametric loudspeaker. Acustica, 1991, 73: 215-217.
[6] Aoki K, Kamakura T, Kumamoto Y. Parametric loudspeaker applied examples. Electronics and Communications in Japan, Part A, 1993, 76(8): 1127-1137.
[7] Kamakura T, Tani M, Kumamoto Y, et al. Parametric sound radiation f rom a rectangular aperture source. Acustica, 1994, 80: 332-338.
[8] Pompei F J. The use of airborne ultrasonics for generating audible sound beams. Journal of Audio Engineering Society, 1999, 47(9): 726-731.
[9] Pompei F J. Sound from ultrasound : the parametric array as an audible sound source. Cambridge, Mass: Massachusettes Institute of Technology, 2002.
[10] Pompei F J,. Wooh S C. Phased array element shapes for suppressing grating lobes. Journal of the Acoustical Society of America, 2002, 111(5I): 2040-2048.
[11] Tan K S, Gan W S, Yang Jun, et al. Constant beamwidth beamformer for difference frequency in parametric array. Proc. of 2003 International Conference on Acoustics, Speech and Signal Processing, IEEE Computer Society, 2003: 593-596.
[12] Karnapi F. A., Gan W S, Chong Y. K. FPGA impleme- ntation of parametric loudspeaker system. Microprocessors and Microsystems, 2004, 28 (526): 261-272.
[13] JI Pei-feng. The research on the theory and the experiment of the directional speaker. Qingdao: Shandong University of Science and Technology, 2005.
[14] ZHU Hai-sheng. Design and implementation of directional speaker based on dsp chip. Qingdao: Shandong University of Science and Technology, 2007.
[15] ZHAO Hong-liang, ZHU Hai-sheng, SUN Xi-rui. Design of directional loudspeaker based on TMS320VC33. Journal of Shandong University of Science and Technology, 2008: 72-75.
[16] CHEN Min, XU Li-mei, HUANG Da-gui, et al. Progress on audio directional loudspeaker research. Audio Engineering, 2006,11: 17-22.
[17] CHEN Min, XU Li-mei, HUANG Da-gui, et al. Square transducer array design of audio beam loudspeaker. Audio Engineering, 2006, 1: 26-29.
[18] CHEN Min. Research on the theory and key technologies of audio directional system. Chengdu: University of Electronic Science and Technology of China, 2008.
[19] Pompei F J. Parametric array modulation and processing method, USA Patent: 7596228. 2009-09-29.