Design and Implementation of Universal Acoustic Signal Processing Platform

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Abstract. Acoustic signal processing has been widely used in many fields. In order to meet the demand of engineering applications, a universal acoustic signal processing platform based on DSP+ARM has been proposed to realize acquisition, processing and output of two-channel acoustic signals. The hardware and software realization methods have been discussed. At the same time, the acoustic signal acquisition performance and impulse response of acoustic signal output module have been measured. The results show that the acquisition sensitivity is 86 mV/Pa, the output response is changed by different loudspeakers.

1. Introduction
Acoustic signal is one of the common signals in nature, relevant applications and analysis have penetrated into almost all of important natural science and technology fields [1-3]. The acoustic signal acquisition, processing and output are the most basic and crucial in acoustic-related applications. The traditional platforms for processing acoustic signal are mainly based on PC or MCU. When PC is used, the platform functional renewal is convenient, but the cost, size and power consumption are relatively higher [2]. While platforms based on MCU is simple in structure, but difficult to meet the needs of human-computer interaction and complex computing [4].

In this paper, a universal acoustic signal processing platform is designed based on DSP+ARM. DSP is responsible for acoustic signal acquisition, processing and output in the structure. The human-computer interaction and the monitoring and control of DSP are realized by ARM. Also, the drive circuits of microphone and loudspeaker are implemented. A standard acoustic source (94 dB SPL, 1 kHz) is used to test the acoustic signal acquisition performance, the impulse response of acoustic signal output channel is measured by the log-sine chirp signal. The universal platform proposed in this paper can fulfill the requirements of engineering applications and is capable to handle the acoustic signals in noise measurement, active noise control and electro-acoustic product testing.

The rest of the paper is organized as follows. In Section II, the structure of universal platform and its hardware are introduced. Section III describes the platform’s software design. Section IV shows the methods and results to measure the acquisition performance and impulse response of acoustic signal output channel. Finally, the conclusion is drawn in Section V.
2. The structure design of universal platform

The capabilities of acoustic signal processing platform proposed in this paper not only include signal acquisition, processing and output, but also data visualization to let users manage the process easily. The overall architecture of platform is shown in figure 1.

![Figure 1. The overall architecture of platform](image)

The platform is constituted by signal acquisition module, signal processing module, signal output module and human interface module. Acoustic signals picked up by microphone are amplified by signal conditioning circuit in acquisition module, then converted to the digital signal by A/D. The digital signal processing algorithms are realized by DSP in processing module. Processed signals are transformed into the analog signal through D/A and drove the loudspeaker by power amplification circuit. This whole process can be monitored and controlled through the human interface. At the same time, lithium batteries are used to supply electricity and improve the platform’s practicality.

With improvements of the platform performance, OMAP-L138 dual-core CPU is selected, which contains an ARM9 core and a C6748 floating-point DSP core. Compared with the traditional ARM+DSP dual-chip structure, OMAP-L138 is smaller, lower in power consumption and higher in integration. At the same time, TLV320AIC23 which contains dual A/D and D/A inside is used to complete the conversion between analog signals and digital signals, its sampling frequency can be controlled by software. The selecting not only meets the requirements of acoustic signal acquisition and output, but also can simplify the hardware structure well.

DSP core in the platform communicates with TLV320AIC23 through McBSP interface. ARM core controls human-computer interaction interface, including USB interface, Ethernet interface, STAT3 interface and LCD. DSP communicates with ARM through interrupts and shared memory. CRY333 microphone and OPA1688 are used to realize acoustic signal acquisition and conditioning, TDA7294 is used to build power amplifier circuit. The overall hardware block diagram is shown in figure 2.

![Figure 2. The overall hardware block diagram](image)

2.1 Signal conditioning circuit

CRY333 is a pre-polarized free-filed measurement microphone with type CRY506 preamplifier, its sensitivity is 50 mV/Pa. The microphone conditioning circuit including power supply and amplification circuit is shown in figure 3. The constant current diodes D1 and D2 are adopted to supply 4mA constant current power for CRY333. The AC signal is coupled out through a capacitor C1 and then amplified by OPA1688 which is an audio operational amplifier. The magnified signal is connected to the LINEIN of TLV320AIC23 and converted by its internal A/D. OPA1688 offers low offset, drift, and quiescent current to meet the needs of acoustic signal amplifier applications. The voltage gain is described as:

\[ A_v = 20 \log\frac{R_p + R_i}{R_i} \]  

(1)
2.2 Power amplifier circuit

In signal output module, the power of signal outputted from the LINEOUT of TLV320AIC23 is amplified by TDA7294. Used as audio class AB amplifier, TDA7294 has the wide voltage range and the high out current capability for both 4 Ω and 8 Ω loudspeakers. The power amplifier circuit is shown in figure 4. The gain can be controlled as:

\[
A_g = 20 \log \left( \frac{R_o + R_s}{R_s} \right)
\]

(2)

2.3 Human interface circuit

The human-computer interaction interface includes USB interface, STAT3 interface, Ethernet interface and LCD. USB and STAT3 interface are used to store data and working status to USB flash disk and hard disk respectively. The platform can exchange data and commands with computer via Ethernet interface. LCD can display the platform status and data. Owing to the rich peripheral interfaces of OMAP-L138, the human interface circuit is simplified. The USB, STAT3 and LCD interface are no need of external chips. The Ethernet interface uses KS8001 for the physical layer transceiver.

2.4 Power circuit

The power circuit block diagram is shown in figure 5. The signal conditioning circuit and power amplifier circuit need the positive and negative voltage source simultaneously. Most of DC-DC voltage converters provide a small negative supply output power while the available input is a positive supply. For this situation, the four lithium batteries with protection modules in series are used to output the ±14.8 VDC. The voltage of each battery is +7.4 VDC. At the same time, other two lithium batteries in series provide +5 VDC for the OMAP-L138 minimum system board by the voltage regulator LM7805.
3. The software design of universal platform

The platform software development is composed of OMAP-L138 and host computer. The acoustic signal acquisition and output are the main software functions. At the same time, the communication between host computer and OMAP-L138 should be implemented by software.

3.1 The OMAP-L138 software

The EDMA module on OMAP-L138 is used to complete signal acquisition and output without CPU intervention. So DSP core can process data in real time. The PING-PONG buffers are designed to simplify software development. When the PING is buffered, the PONG is processed. If there is a monitoring requirement, the ARM core can be interrupted and transmit PONG data to host computer when the processing is completed.

The ARM core runs an embedded Linux operating system. Based on memory mapping, monitoring and controlling DSP core’s running state can be realized. Mapping the interior memorizer address space to the user space, the DSP core registers and PING-PONG buffers can be accessed in the embedded Linux operating system. After system starts, the ARM core realizes different functions according to the different commands received from the host computer.

3.2 The host computer software

The host computer exchanges message with OMAP-L138 via the Ethernet interface. The software is implemented based on LabVIEW programming. The software interface is designed as figure 6.

The user can start and stop the DSP’s running by buttons. Also the loudspeaker data can be controlled by the file selected in the host computer software. The acquisition and output data can be displayed in the graphs and saved to files.
4. The performance test method and results
After the platform is built, the signal acquisition module is calibrated and the response curve of the
signal output module is measured. Data-transfer word length of TLV320AIC23 is set at 16 bits with a
sampling rate of 96 kHz.

4.1 The calibration of the signal acquisition module
The standard acoustic source (94 dB SPL, 1 kHz) is used to calibrate the signal acquisition module.
The voltage gain $A_V$ of signal conditioning circuit is 4.7 dB. The field measurement and signal in
time domain are shown in figure 7 and 8 respectively.

$$K_a = \frac{\sqrt{2} X}{2^{16}} V_r,$$

where $X$ is the sampling peak-to-peak value, the TLV320AIC23 full-scale range $V_r$ is 1.0
V RMS. So $K_a$ can be obtained as 86 mV/Pa. The maximal sound pressure level of the acoustic input
is 118 dB SPL.

4.2 The measuring of the signal output module
The principle block diagram measuring the response of the signal output module is shown in figure 9.

Log-sine chirp signal whose frequency increases exponentially with time is used for the stimulant
signal. It is encapsulated by [5]
\[ x(t) = a \sin \left( \frac{2\pi f_T}{\ln(f_2/f_1)} \left[ \ln\left( \frac{f_2}{f_1} \right) \exp\left( -\frac{f_2}{T} \right) - 1 \right] \right) \] \tag{4}

Where, \( a \) is the amplitude, \( T \) is the duration, \( f_1 \) and \( f_2 \) are the starting and ending frequency respectively. This signal is generated by DSP core and amplified to 1 mW to drive the loudspeaker. At the same time, \( y(t) \) is picked up by the microphone inside the CRY318 artificial ear \([6]\). The amplitude \( a \) can be determined by

\[ a = \frac{\sqrt{2R}}{A_g} \] \tag{5}

Where, \( R \) is the loudspeaker’s resistance. The response of signal output module is analyzed as \([7]\)

\[ H(f) = \frac{Y(f)}{X(f)} \] \tag{6}

Where, \( X(f) \) and \( Y(f) \) are obtained by Fourier transform of \( x(t) \) and \( y(t) \) respectively.

In this paper, two sets of loudspeakers are used to test the response of signal output module, and both groups are \( R = 4 \, \Omega, \, A_g = 14 \, \text{dB}, \, T = 100 \, \text{ms}, \, f_1 = 20 \, \text{Hz} \) and \( f_2 = 20 \, \text{kHz} \), figure 10 respectively show the frequency response and phase response of the acoustic output module. It can be seen from the pictures of the two sets of tests that, with the same sound output module test, the sound pressure level of the No. 1 loudspeaker is higher than that of the No. 2 loudspeaker in the frequency response within 20 Hz to 2 kHz. In the phase response test, No. 1 loudspeaker perform better.

5. Conclusion

The universal acoustic signal processing platform based on OMAP-L138 dual-core CPU is described in this paper. The hardware and software realization methods have been discussed. The acoustic signal acquisition performance has been tested by a standard acoustic source (94 dB SPL, 1 kHz), and the impulse response of acoustic signal output channel has been measured by the log-sine chirp signal. The results show that the universal platform can realize acquisition, processing and output of...
two-channel acoustic signals. The structural optimization design makes the platform more flexible to meet the needs of engineering applications.

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