A Texas Instruments DSP-based Acoustic Source Direction Finder

G. Miramontes de L., I. de la Rosa V., M. Araiza E.

Abstract—**This document describes the implementation of a DSP-based acoustic source direction finder and its application to control the direction of a steerable camera. For real-time implementation a TMS320F2812 digital signal controller was used. The system can be used for different applications like in teleconferencing rooms or surveillance systems. This report presents a description of the developed system in three phases of the project: direction of arrival estimation, analog interfase, and servomotor control. The application was developed by undergraduate students at the Autonomous University of Zacatecas as a final project in the course "DSPs programming", which is one of the DSP courses thought in the Communications and Electronics Engineering program, with a major in DSP.**

***Index Terms*— Acoustic source localization, Direction of Arrival, , Teleconferencing rooms.**

I. Introduction

WITH modern digital signal processors, and basic digital signal processing theory, many different applications can be implemented in real-time easily. One interesting application is the use of Direction of Arrival (DoA) estimation techniques. DoA estimation has been applied in the localization of a moving source in different areas like communications, robotics, and many others. For DoA estimation an array of sensors is used. When the source is an acoustic signal, DoA estimation can be performed using an array of microphones. The objective is to steer an active device, usually a video camera, to follow an active speaker in a conference room [1].

Other operations than can be performed with a microphone array, among others, are: enhancement and source separation. When a single source is present, the change of phase and sum of all the microphone signals can give an improvement in the signal to noise ratio (SNR) [2]. When the interest is the cancellation of interference signals, the improvement depends on the number of microphones used in the array [3]. With the same approach, an array can be used for source localization problems, where the arrival time difference (time-delay) for each pair of microphones is used to make an estimate of the angle direction of the source.

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II. Theoretical foundations of DoA estimation

The basic configuration setup for a DoA estimation is shown in Figure 1. An array of two microphones and only one acoustic source are used. The angle of arrival from the first microphone M_0 is denoted by θ_0 and a distance ρ_0, and *d* is the distance between microphones in meters.

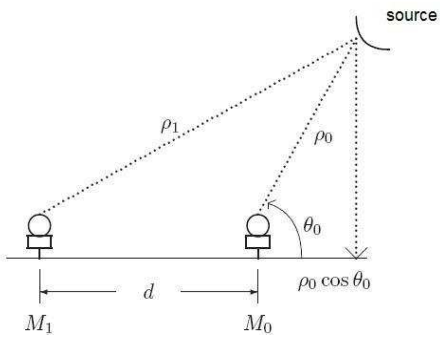

Fig. 1. Linear microphone array.

The separation between microphones should not be longer than $\lambda/2$, [4]. For a signal of 1 kHz, λ=0.342, assuming a constant sound velocity of 342 m/s. In practice, the time-delay of the signals between both microphones has to be measured in some way. Assuming far field conditions, the time-delay is given by

$$\tau = (d \cos\theta)/C,$$

where C is the sound velocity, and *d* is the distance separation of the microphones.

If the time-delay is measured, then is possible to estimate the angle of arrival, solving for θ, where $-\pi/2 < \theta < \pi/2$. This angle is better known as DoA.

A. Time-delay estimation

Several methods for time-delay calculation can be used. The most commonly used is the generalized cross-correlation (GCC) [6]. The simplicity of this method has made it the preferred method for DoA estimation in microphone arrays.

Let $m_0(t)$, and $m_1(t)$ be the signals from microphone M_0 and M_1, respectively.

So,

$$m_0(t) = s(t) + n_0(t)$$
$$m_1(t) = s(t+\tau) + n_1(t),$$

where *s(t)* is the source signal, and $n_0(t)$ and $n_1(t)$ are additive noise at each microphone.

The time-delay can be estimated by

$$\hat{\tau} = \arg\max_\beta \int_{\omega=-\infty}^{\infty} M_0(\omega)\overline{M_1(\omega)}e^{j\,\omega\,\beta}d\omega,$$

where $M_0(\omega)$ and $M_1(\omega)$ are Fourier transforms, and the upper line denotes complex conjugate. A time domain implementation was used in this project.

III. Design Issues

Figure 2 shows the block diagram of the system. It has three main sub-blocks: analog front-end, DSP controller, and a PWM controlled servo motor, which holds the steerable camera.

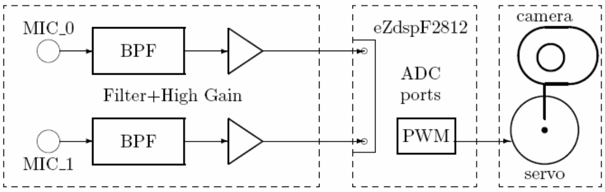

Fig. 2. Blocks diagram of the system.

A. The Analog front-end

For the DoA estimation, a minimum of two-microphone array is necessary. Two electrect microphones were used, together with analog band-pass filters (BPF) and a high gain amplification stage, delivering a very sensitive analog front-end. In this way, the system is capable of detecting acoustic signals from several meters away of the microphones.

The electronic design comprises high quality audio operational amplifiers, the Texas Instruments OPA2337 and OPA337. The signal conditioning circuit is a two channel filter/amplifier. The first stage forms a speech band pass filter, covering the frequency range from 300 Hz to 3 kHz [8]. The second stage is a high gain amplifier, which adds high sensitivity to the analog inputs. Figure shows a 3D view of one of the two channels.

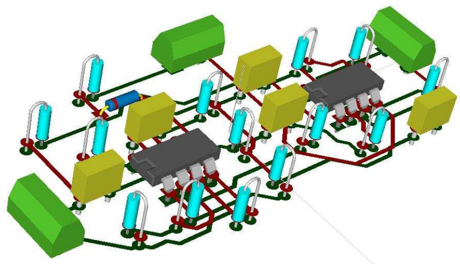

Fig.3. 3D view of the analog front-end, one channel.

Due to the polarity requirement of the analog inputs of the DSP board, the electronic design included provisions to limit the output of the analog front-end from 0 to 3.0 volts. The circuit uses the same 5 volts power supply of the eZdspF2812 board and by means of a zener diode the power supply voltage is regulated to 3.3 volts.

B. DSP programming

The system has to accomplish the following tasks:

1. Capture simultaneously two signals from MIC_0 and MIC_1.
2. Estimate the time-delay between the two signals.
3. Set a pulse width at the PWM output to control the position of a servo motor.
4. Return to 1.

The main component of the system is a digital signal processor, which performs all the necessary tasks to acquire the sound signals, perform the DoA estimation in real-time, and control the orientation angle of the camera. The eZdspF2812 from Spectrum Digital© is the heart of the whole system. It was decided to use this DSP board since it is a good alternative to develop and test prototypes for real-time control applications. For a comprehensive description of the functional blocks, memory maps, and functional overview of the TMS320C28x members, see [7].

The two main integrated peripherals of the F2812 used in this application are:

- Event-manager modules (EVA, EVB) for PWM waveform generation,
- Enhanced analog-to-digital converter (ADC) module.

The event manager (EV) module includes general-purpose timers, full-compare/PWM units, capture inputs (CAP) and quadrature-encoder pulse (QEP) circuits. The ADC block is a 12-bit converter, single ended, 16-channels. It contains two sample-and-hold units for simultaneous sampling, which is a necessary condition for this application.

A simplified functional block diagram of the ADC module is shown in Figure 4.

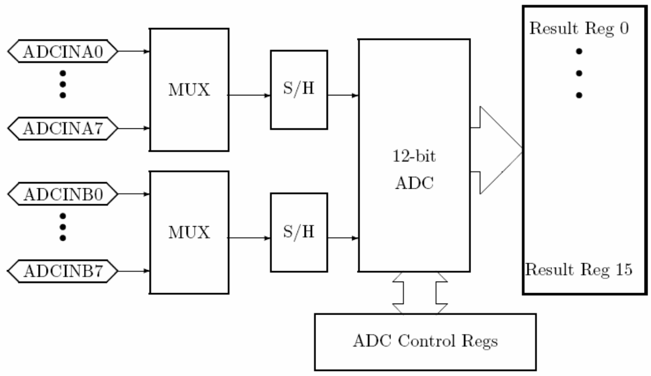

Fig. 4. Simplified ADC block diagram.

Functions of the ADC module include:

- 12-bit ADC core with built-in S/H circuits
- Analog input 0.0 to 3.0 volts
- Fast conversion rate: 80 ns at 25 MHZ ADC clock, or 12.5 Msamples/s
- 16 channel multiplexed inputs
- Autosequencing capabilities providing up to 16 auto conversions in a single session.
- Sixteen Result Registers
- Multiple event sources with EVA and EVB
- Many others

A template code was used as a starting point, then it was modified to fulfill the requirements of the project. The project files are borrowed from Lab14 Part I [9]. Some of these files deal with all initialization routines of timers, event manager, ADC, etc. The main changes were done to the configuration of EV managers to change the sampling frequency of the ADC, and the PWM output.

Once the time-delay has been estimated, its value is used as an index to read a look-up table who contains different values for θ. The following task is to set the duty cycle of the PWM output to adjust the position of the servomotor to the desired direction.

C. Motor selection and set up

To control the direction of the camera, different solutions have been proposed: stepermotor, DC motor, or servomotor. The last one was chosen because of its easy of position control by a PWM signal. The servomotor used is a HITEC HS-311, a commonly used servomotor for robotic experiments.

The main characteristics are:

- PWM control, 1.5 ms at center
- power supply 4.8 to 6 Volts
- velocity at 6 Volts 0.15 s/60 degrees with no load
- torque \@ 6 Volts 3.5 Kg cm
- steady state current 7.7 mA
- maximum current 800 mA\
- working range 1.1 ms to 1.900 ms

The next step was the calculation of the values that will give a pulse width related to a given position in degrees of the servomotor.

IV. Summary and Conclusions

A real-time acoustic source localization system was developed using a eZdspF2812 board and simple electronic hardware. The system was tested under different reverberant and noisy conditions. When the system was tested in a highly reverberant room, the performance failed to localize the true source. In an open environment the system worked very well. Even with the reverberant conditions the performance of the system was acceptable, and the objectives of the project were met. Further improvements can be done by the inclusion of a more sophisticated algorithm to reduce the effects of reverberant rooms. Figure 5 shows the final prototype, and Figure 6 shows the camera following the sound from a cellular phone.

Fig. 5. Final Prototype.

Fig. 6. Following an acoustic source.

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