Development of Signal Monitoring Platform for Sound Source Localization System

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Abstract

The sound source localization system is used to some area such as robotic system, object localization system, guarding system and medicine. So time delay estimation and angle estimation of sound direction are studied until now. These days time delay estimation is described in LabVIEW which is used to create innovative computer-based product and deploy measurement and control systems.

In this paper, the development of signal monitoring platform is presented for sound source localization. This platform is designed in virtual instrument program and implemented in two stages. In first stage, data acquisition system is proposed and designed to analyze time delay estimation using cross correlation. In second stage, data obtaining system which is applied and designed to monitor analog signal processing is proposed.

Key words : signal monitoring, data acquisition, time delay, LabVIEW

1. Introduction

In general, sound source localization system is necessary to analyze at least more than two microphone systems. Time delay estimation is important parameter. If time delay estimation can`t measure exactly, it is compensated by direction angle of analog signal. This issue requires array microphone. But array microphone system is large and high cost when sound source system considers designing reduced model. A compensation algorithm is studied to decide a problem that measure precision of time delay estimation from noise and reverberation environment.[1]

Nowadays national instrument data acquisition card and module are applied to sound source localization system widely because LabVIEW language is flexible to describe any systems. This language is simpler and better than other basic programming language.[2]-[3]

In medicine, data acquisition card is used to estimate and depict the sound source localization to reveal lung disease when detecting sound components in the lung.[4] Time delay signal process is developed and assisted by virtual instrument.[5] Ultrasonic range measurement system is designed and analyzed in LabVIEW to estimate for time delay [6]. Therefore LabVIEW can be applied to develop integration sensor module and measure sensor raw data in real time for Unmanned Aerial Vehicle. [7]

In this paper, the signal monitoring platform is designed and implemented in virtual instrument graphical program to be aimed analysis for sound source localization system. First raw data of analog signal acquired through data acquisition module is filtered and validated by cross correlation for time delay estimation. Then raw data obtained by atmega 128 is designed and described in LabVIEW language to analyze for sound source localization.

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2. Signal Monitoring Platform Design

2.1 Design Consideration

Signal monitoring platform for sound source localization considers at least two microphones devices. And then four microphone devices are used to precision measurement of time delay. The processing time of signal monitoring platform can execute in real time because analyzing analog signals of each microphone devices.

2.2 Platform Architecture

2.2.1 Overview

In Figure 1, the signal monitoring platform consists of two kind systems: data acquisition system and data obtaining system. Human can hear and detect sound source location and direction using his two ears. So data acquisition system considers human hearing system. Data acquisition system is designed to receive two microphone signals for measuring time delay. Data acquisition system has multiplex and A/D converter. It can process each analog signal.

The environment noise and reverberation impact to detect sound source. In that reason data obtaining system consider four microphone devices. Data obtaining system has not multiplex function. It sends data to LabVIEW in RS-232 communication.

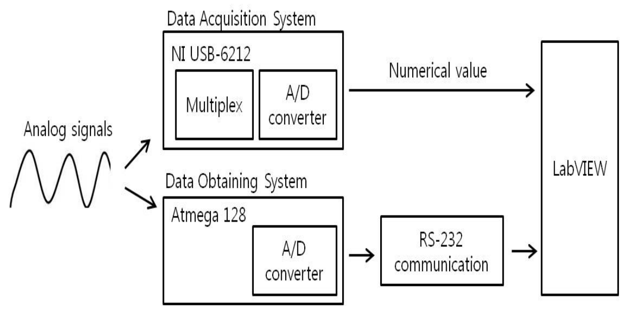

Figure 1. View of whole platform architecture

Those systems have same function that receiving analog signals of microphones are collected to convert analog to digital and detect time delay through data acquisition module and atmega128 processor.

2.2.2 Data Acquisition System

Data acquisition system integrates some functions and nodes to design data acquisition system. Those are DAQmx nodes, low pass filtering, graphical waveform monitoring, peak point detector, cross correlation, calculating statistics value for time and arccosine. A composition of two microphone original signals is viewed in Figure 2. This system needs to set up synchronization time for sending two microphone original signals to LabVIEW in real time. So A0/StartTrigger is connected to ground. Cut off value of low pass filter function set 100.

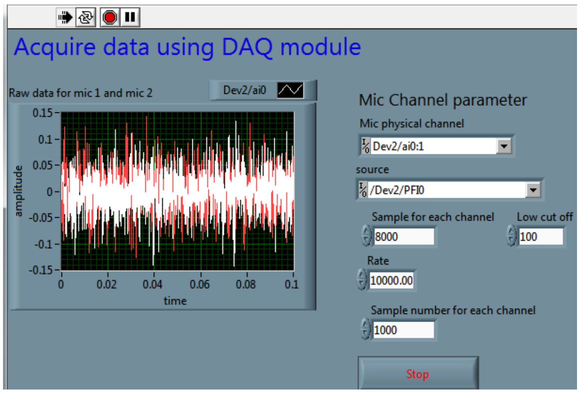

Figure 2. View of composition signal

Each microphone device data can be acquired by its channel in data acquisition system, and each channel of two microphone signals displayed in the virtual waveform graphic in Figure 3 separately.

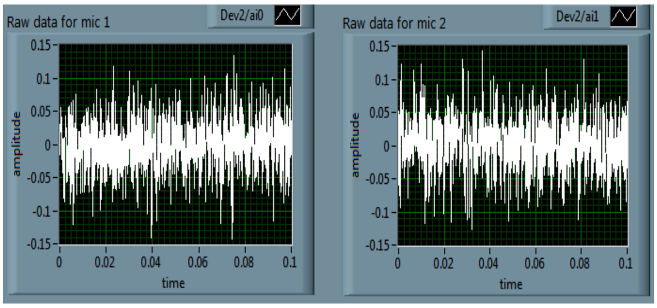

Figure 3. View of microphone original signal

2.2.3 Data Obtaining System

Data obtaining system consists of VISA nodes, file input/output nodes and using functions such as string data convertor, low pass filter, peak point detector and cross correlation. Four microphone data are controlled by one processor. Therefore each microphone signal can send sound source to LabVIEW in looping processing time. It means that data obtaining system uses one channel sending digitalized data of each microphone in regular sequence. Therefore all microphones obtained data is represented in waveform in non-real time as showing Figure 4.

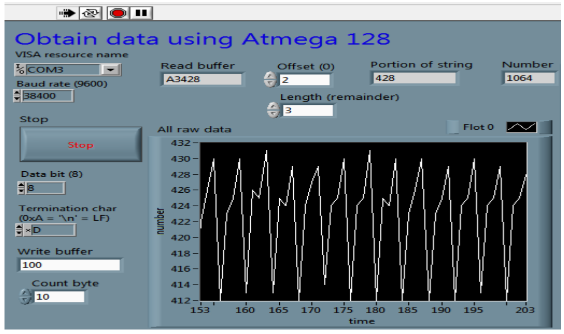

Figure 4. View of data obtaining system

3. Signal Monitoring Platform Implementation

3.1 Data Acquisition System

Data acquisition system consists of amplified two microphone signals, sixteen channel data acquisition NI USB-6212 module and LabVIEW. It is viewed on Figure 5.

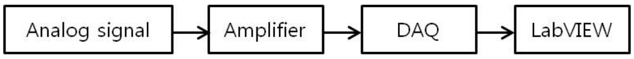

Figure 5. Component of analog signal acquisition system

3.2 Data Obtaining System

In Figure, data obtaining system consists of amplified four microphone devices, expressing analog signal block, amplifier, atmega 128 processor and LabVIEW.

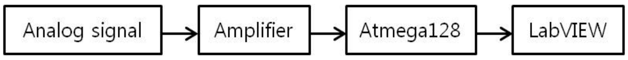

Figure 6. Component of analog signal obtaining system

3.3 Express to LabVIEW

Generally data acquired from data acquisition module is sent to LabVIEW by numerical value. But analog signal is converted to digital and passes string type data to LabVIEW through RS-232 communication using atmega 128. Thus string type data should be converted to numerical data to process and describe a graphic signal in LabVIEW.

3.4 Message Structure for Analog Signal Obtaining System

A frame is defined in ASCII data communication protocol between the data obtaining system and LabVIEW. Table 1 shows the frame dividing fields.

Table 1. ASCII Communication Frame

Start	CPU	Mic number	Data		End
			MSB	LSB	

Information of each field contains:

- Start: ASCII value of the 'Start' is '3A' hex and set one in LabVIEW.
- CPU: 'CPU' indicates the CPU number.
- Mic number: 'Mic number' indicates the microphone number noticed between atmega 128 and LabVIEW.
- Data: 'Data' means the composited three bytes.
- MSB: 'MSB' means the most significant byte.
- LSB: 'LSB' means the least significant byte.
- End: 'End' is 0x0D in hexadecimal.

4. Experiments

Figure 7 and 8 shows the experiment result of data acquisition system. In Figure 7, each amplified analog signal is filtered by low pass filter to passes low-frequency signal. But low pass filter attenuates (reduces the amplitude of) signals with frequencies higher than the cutoff frequency.

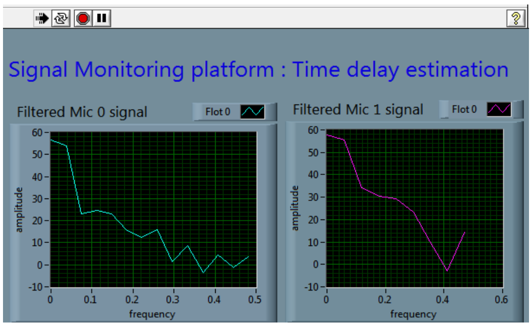

Figure 7. Two microphone filtered signal

The peak point function is detected each analog signal for cross correlation analysis. Though the purpose of cross correlation is to detect time delay using two microphones data. The execution of cross correlation function is shown in Figure 8. The two microphones signals are sampled large numbering sampling points. The experiment result of cross correlation processing is depended on setting sampling point.

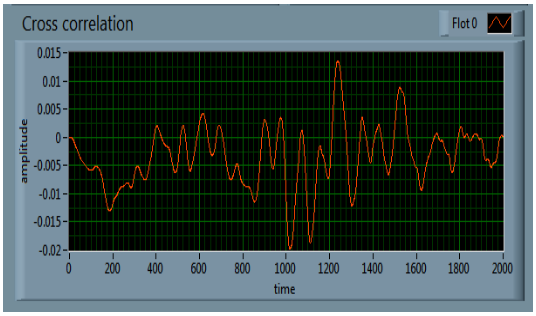

Figure 8. Cross correlation result

Figure 9 shows four microphone signals obtained from data obtaining system separately.

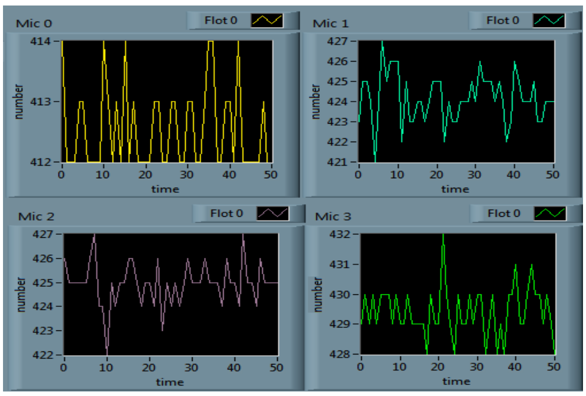

Figure 9. Experiment result of data obtaining system

Detecting time for each microphone data is slowly because there is delay processing time as data obtaining system uses looping time to send each analog signal to LabVIEW and to parse for depicting waveform. It means that each microphone has equal timing gap sending to atmega 128 from sound source and LabVIEW. In case each microphone signal can`t pass low pass filtering function and use cross correlation for detecting time delay. Just this system can check strength value of the each microphone. So the microphone being near sound source can send higher value than other microphones.

5. Conclusions and Future Works

In this paper is proposed signal monitoring platform for sound source localization system. Signal monitoring platform design is considered two stages, and each stage system is developed and tested for measuring time delay.

Data acquisition system can acquire analog signal in real time and send to desktop real time at almost. Accordingly data acquisition system result is better than analog signal obtaining system because analog signal obtaining system can`t obtain and process in real time. When sound source sends to LabVIEW, data acquisition system can separate each signal. Thus each microphone signal can use own channel to send. But data obtaining system applies one channel for four microphone devices to send data to LabVIEW.

Future work is scheduled to extend distance between microphones and amplified microphone circuit will be updated by filtering circuit to reduce original analog signal error. It will be studied band pass filter to apply microphone devices to send clear analog signal to LabVIEW. Data obtaining system should be updated to process analog signal in real time. Algorithm and some research will be done according data obtaining system for sound source localization.

References

[1] Jong Chan Kim, "Implementation of Sound source tracking Using Time Difference of Arrival," Hanbat university, Computer Science, Master degree Dissertation, 2009.

[2] http://www.ni.com/

[3] Hye Young Son, Standard basic of LabVIEW, INFINITY BOOKS, 2011.

[4] Ipek Sen, Murat Saraclar and Yasemin P. Kahya, "Acoustic Mapping of the Lung Based on Source Localization of Adventitious Respiratory Sound Components," The 32nd Annual International Conference of the IEEE EMBS, pp. 3670-3673, 2010.

[5] Li Ming, Jiang Chunlan, Yuan Zi and Wan Zaicheng, "Time Delay Signal Process Assisted by Virtual Instrument," IEEE Trans. on Sig Proc., The 2nd International Workshop in Database Technology and Applications (DBTA), pp. 1-4, 2010.

[6] Abdallah Hammad, Ashraf Hafez and Mohamed Tarek Elewa, "A LabVIEW Based Experimental Platform for Ultrasonic Range Measurements," DSP Journal, Vol.6, Issue 2, 2007.

[7] A.L. Rodriguez, L. M. Parrilla, A.Simon-Muela and M.M.Prats, "Real time sensor acquisition platform for experimental UAV research," IEEE/AIAA 28th Digital Avionics System Conference, pp. 1-10, 2009.