MEMS Acoustic Array Embedded in an FPGA Based Data Acquisition and Signal Processing System

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Abstract—Acoustic arrays are currently utilized in many different applications, ranging from consumer electronics to military systems. Active research on sensors, hardware, algorithms, and system integration specific to acoustic arrays are ongoing on a wide range of engineering fields. Applications such as sound source separation, sound navigation, sound imaging, and speech recognition are few of the many possible applications that benefit acoustic sensor array. Using multiple sensors in arrays has many advantages; however it is also more challenging. As the number of signals increases, the complexity of the electronics to acquire and process the data will grow as well. Such challenge can be quite formidable depending on the number of sensors, processing speed, and complexity of the target application. This paper describes the design and implementation of a 52 microphone MEMS array, embedded in an FPGA platform with real-time processing capabilities. The paper also provides the first results on the use of acoustic array as a source separation system utilizing Independent Component Analysis (ICA) technique.

Index Terms— Source separation, acoustic arrays, MEMS, sound imaging, sound tracking, scalable, real-time, ICA.

I. INTRODUCTION

This paper describes the design of a networkable broadband acoustic MEMS array embedded in a compact readout and data processing system. This acoustic array has been used for acoustic source separation and localization. The present work will provide experimental results obtained with this array for acoustic source separation utilizing Independent Component Analysis algorithm [1].

In practice, the implementation of systems utilizing acoustic arrays is challenging, and its complexity will depend on the application for which the system is designed to be engaged. For source separation and localization there are several important characteristics that the system must have including array spatial resolution, low reverberation, and real-time data acquisition and processing. Therefore, the design of this system is composed of two key components: i) the acoustic MEMS array, and ii) the data acquisition and processing module.

The acoustic MEMS array design takes into account mechanical factors such as array geometry, sensors disposition, microphone reverberation, and form factor. The electronics of the array, on the other hand, have demanding issues to deal with, such as electronic noise, power decoupling, cross-talk, and connectivity. Furthermore, the system needs to integrate the acoustic array with real-time data acquisition, signal processing, and network communication capabilities. In this type of system, characteristics such as speed, scalability, and real-time signal processing are paramount and require highly advanced data acquisition and processing modules. Therefore, integrating acoustic arrays with generic data acquisition systems is impractical and challenging due to the wide variety of requirements regarding speed, signal processing capabilities, and the scale that acoustic arrays can present. The objective of this work is to present an acoustic imaging system using the CAPTAN (Compact And Programmable daTa Acquisition Node) architecture [2] that can deliver these requirements while closely integrated with a specially designed sound imaging array.

II. ACOUSTIC ARRAY

The design of the acoustic array board was based on three basic requirements; good spatial resolution, high signal-to-noise ratio (SNR) and user selectable unidirectional/omni-directional acoustic aperture (i.e., beam steering). Spatial resolution is governed by the number of MEMS microphones, inter-microphone distance, and microphone sensitivity. This acoustic array is made of 52 microphones; this number was chosen in order to obtain a highly flexible system with good spatial resolution and sensitivity, and high SNR. The microphones are distributed in an octagonal grid with the inter-microphone distance of 10.0 mm centre to centre in the horizontal and vertical. The spatial sampling rate of this array can be approximated by dividing the sound speed by the

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inter-microphone distance and then further dividing the result by two in order to satisfy the Nyquist-Shannon theorem [3]. Spatial sampling rate of frequencies up to 18 kHz without spatial aliasing can be achieved with this spacing.

The sensitivity of the array increases monotonically with the number of sensors, and the MEMs microphone chosen for this array (SPM0208) has a sensitivity of 1V/Pa at 1 kHz [4]. These microphones are Omni-directional and when combined in the array they provide a very good acoustic aperture.

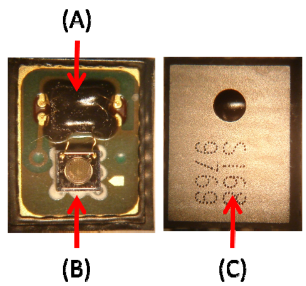

Figure 1. MEMS microphone, (A) amplifier, (B) microphone and (C) aluminium cover.

The MEMS microphone is a fundamental piece of the array due to the small size, high sensitivity, and low reverberation. Its frequency response is essentially flat from 1 to 10 KHz and it has a low limit of 100Hz and a high limit of 25 KHz. The array geometry was chosen in order to allow beam steering on the horizontal and vertical planes [5].

In order to reduce reverberation on the system the microphones are glued with silver epoxy to the surface mount copper pads in the board. The array also provides a central loudspeaker that has dual use: it can be used for calibration purposes, or for sonar like applications. When used as calibration element, the microphone emits a set of pure frequencies that then are captured by the microphones and used for its calibration taking in account the geometry of the array. When the central microphone is used for active sonar applications a series of pre-programmed pulses is generated and captured back by the array.

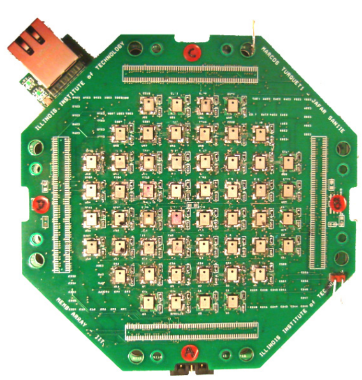

Figure 2. Top view of the acoustic array. The MEMS elements are clearly visible on the top of the printed circuit board (PCB).

In support of the microphones the board also provides amplification and analog-to-digital conversion (ADC). Every single channel has two operational amplifiers, one embedded in the MEMS package as illustrated by Figure 1, and one in the back of the PCB board. An individual ADC is provided for every channel, each ADC has 12 bits resolution, 2.0 Volts dynamic range and a maximum sampling rate of 5Mbps.

Due to the nature of the array, this circuit board requires extensive use of decoupling capacitors, there is a total of 54 10uF capacitors on the back of the printed circuit board, one for every channel, this proves to be critical since all the channels are parallel and will essentially have a peak demand of power at the same time. The second stage amplifier provides a first order high-pass filter with corner frequency at 400Hz.

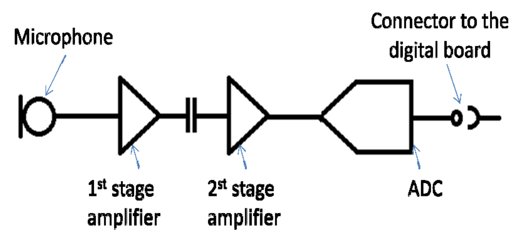

Figure 3. Electronics circuit supporting each individual channel.

Figure 3 provides a high level schematic of the analog supporting circuitry for each channel. The first stage amplifier gives a 20dB gain on the signal and it is AC coupled to the second stage that gives additional 20dB of gain. After the signal is digitized it is passed to the digital board called Node Processing and Control board (NPCB) [2]. The NPCB is stacked underneath the array.

III. READOUT AND PROCESSING SYSTEM

The digital part of the sonic array system is implemented in the NPCB. The NPCB board is part of the CAPTAN system. The CAPTAN system is a distributed architecture based on core elements known as system nodes. A node is a stack of boards connected together by the Vertical Bus in which every board in the same node has access to the Vertical Bus and is therefore accessible to each other.

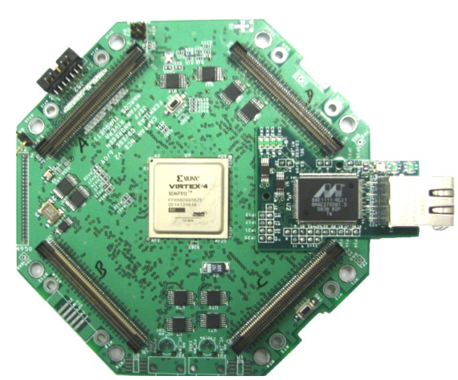

Figure 4. Top view of the NPCB board. In the centre the FPGA, surrounding it the four vertical bus connectors and in the right the Gigabit Ethernet card.

Basically, the NPCB board contains a VIRTEX-4 XC4VFX12 FPGA [6], an Ethernet Gigabit link and an

EPROM. All data from the acoustic array is stored and processed on this board. The FPGA chip is booted by the 32MB EPROM that contains a specially designed configware for managing the data coming from the sonic array.

There are in total 68 digital lines interconnecting the sonic array and the NPCB, 52 data lines from the channels, 8 clock lines, and 8 data valid. These lines are evenly distributed into the four board-to-board Vertical Bus connectors that the system provides, 17 LVCMOS lines per connector. The power is also provided through these connectors, and the system is powered in a single 3.3V connection to the NPCB board, regulated and then distributed to the sonic array.

All the lines on the NPCB board have the length matched to avoid timing issues and routed in alternated copper layers between power planes to reduce cross-talk. The clock distribution is in a star fashion to the ADCs on the sonic array; each clock line can support 7 or 6 ADCs depending on the quadrant of the board.

This board also provides a Gigabit Ethernet Link that provides gigabit communication between nodes, or between a node and a computer. This board is the main external interface of the system, and can communicate directly with any computer with 1000BASE-X network capabilities. The board is designed to work with Ethernet protocol 10/100/1000 and to use UDP/IP as the communication protocol.

Although the board is capable of connecting using the IEEE 802.3ab (1000BASE-X) [7] protocol, it cannot send pure user data at this speed due to the addition of several layers of protocol, maximum packet size limitations, and the particular hardware used. The board can, however, send pure user data up to 800 Mbps.

The NPCB currently is responsible for formatting the sonic array data, processing it and sending to a computer through the Ethernet link. Currently, the computer will further process the data using specially designed software [8] and apply the Independent Component Analysis algorithm in order to achieve source separation.

IV. Independent Component Analysis

Independent Component Analysis (ICA) [1] is an analytical tool that can recover independent sources mixed in a linear combination. This technique is based on the assumption that signals from different sources are statistically independent, and statistically independent signals can be extracted from mixture signals. ICA defines a model for the observed data that requires a large number of samples in order to establish the necessary statistics. The model assumes that the data variables are linear combination of unknown variables, the unknown variables are assumed to be non-Gaussian and independent. The goal then becomes to find a transformation in which the components are as statistically independent as possible from each other. This technique is related to methods such as principal component analysis and factor analysis. The main distinction between ICA and these techniques is that while ICA founds a set of independent sources, principal component analysis and factor analysis finds sets of signals which are uncorrelated. This means that ICA recovers the original sources while the other two methods only find sets of the signal that can be locally uncorrelated but not necessary globally independent. The ICA algorithm is computationally intensive since it must accumulate and go through the signal samples performing complex operations. Efficient versions of the algorithm have being already deployed using different techniques such as the FastICA[1] that can be implemented efficiently in hardware platforms such as DSP processors and FPGA's. The ICA algorithm being used for this application is a custom version of the FastICA and is implemented in Matlab. Because of the nature of the ICA algorithm, this system can only be used to separate non-Gaussian and independent sources.

V. Results

First results of the system operating for the target application of source separation are presented in this section.

The system was placed in a test stand composed of two loudspeakers L0 and L1, the speakers where equidistant from the sonic array with a 30 degrees aperture with respect to the array. The distance between the speakers and the array was set to 30cm. Figure 5 shows the disposition of the loudspeakers relative to sensor array, and the microphones chosen for the test.

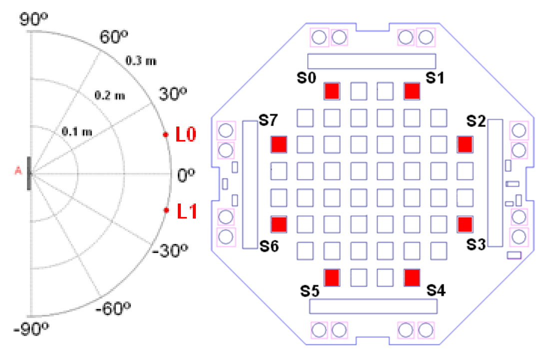

Figure 5. Disposition of the loudspeakers relative to the sonic array. Microphones chosen for the test and the microphones used for this test.

The sonic array system was set to two seconds capture and sampling frequency equals to 128 kHz. The data was captured, and immediately sent trough the Ethernet link to a computer where the fastICA software processes the data. The entire process takes 5 seconds for 8 sources. Using only eight microphones reduced the ICA processing time and data analysis. The microphones chosen for this test were the eight vertices of the octagonal array.

The experiment carried out was performed by first exciting the speaker (L0) with a sound track containing 2 seconds recording of a *chicken*. The signal was acquired and saved on the computer hard disk for subsequent comparison, and can be observed in Figure 6(a). The different amplitudes of the signals are caused mainly by variations on the gain of each channel, since this acoustic array was not trimmed.

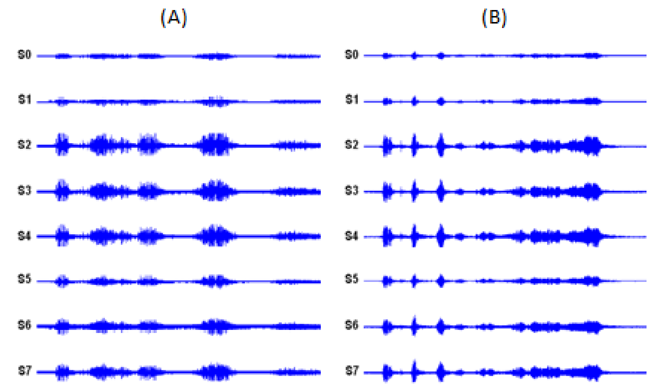

Figure 6. (a) Signal captured by the array from the source L0. (b) Signal captures by the array from the source L1.

Next, the same process was done but this time for a *pelican* with (L1) speaker. Again the recording of the *pelican* lasted 2 seconds and contained 256K samples as the first signal. These signals can be observed on Figure 6(b).

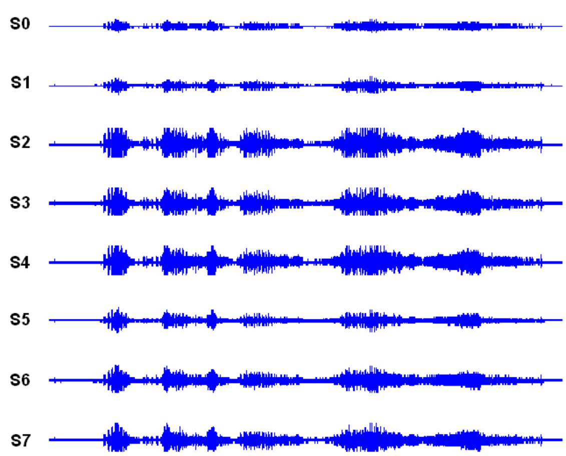

Figure 7. Signals captured when L0 and L1 where playing simultaneously.

Finally, source L0 and source L1 played simultaneously the *chicken* and *pelican* records and the results can be observed on Figure 7. The signal captured with both sources playing simultaneously was then fed to the fastICA algorithm that produced 8 results, and the two most uncorrelated were separated.

In order to compare the original signals captured individually with the extracted signal, Figure 8 shows a comparison between the solo L0 signal obtained by microphone S0 and the ICA extracted signal and the solo L1 signal from the single run with the extracted signal from the mixed run.

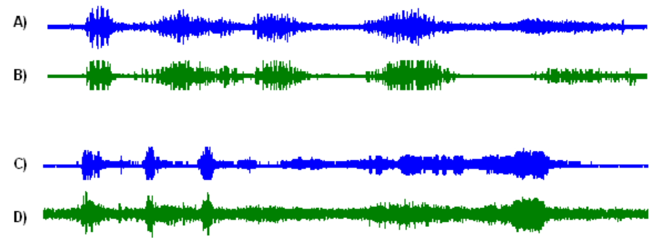

Figure 8. Results from the fastICA algorithm. (A) original chicken signal from L0, (B) the recovered signal of L0,(C) the original pelican signal from L1 and (D) the recovered signal of L1.

VI. CONCLUSION

This work shows that it is possible to integrate key technologies such as MEMS, high performance FPGA, and Gigabit Ethernet to produce a very compact, network enabled acoustic array that offers high performance. Experimental results for source separation applications were provided using this acoustic array. The results show that the array is capable of performing its designed task when integrated with the fastICA algorithm. The combination of the MEMS microphone arrays with the CAPTAN scalable architecture enables versatile configurations for other applications such as sound imaging.

VII. REFERENCES

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