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Acoustic Sensor Array for Sonic Imaging in Air

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***Abstract*—Sonic imaging in air is a field of study with growing applications such as objects location, tracking and identification. It is also used in NDE applications to detect defects or hidden objects inside the structure. The objective of this work is to present a sonic system capable of locating and imaging objects in air with high quality and good resolution. The acoustic transducer array is a novel 52 MEMS sensors array, embedded in a network enabled, data acquisition and signal processing architecture. The system operates in the 20-100 kHz frequency range and makes use of phased array techniques in order to accomplish its objectives. It operates actively where one transducer is employed to generate the scanning beam and the sensor array will capture the echoes. The system is capable of acquiring and processing with up to 800 Mbits of raw data generated by the acoustic sensor array. Beamforming, matched filtering, spectral processing, and independent component analysis are the main techniques utilized in this work for imaging, tracking and identification of objects.**

Keywords –source localization; ultrasound; acoustic arrays; MEMS; sound imaging; sound tracking; real-time.

I. INTRODUCTION

Acoustic emissions are generated by a large variety of structures and machines during operation. These signals can represent the normal operation of the system or indicate an abnormal situation and be used as an early warning, indicating that the system is about to fail. For example, it can be used to monitor the structural health of bridges and locate failure points.

Object detection and recognition system proposed in this work integrates an advanced 52-element MEMS acoustic array with source localization and signal processing techniques. Source localization is used to provide the direction in which acoustic sources are located in the acoustic horizon of the array when the sources are within the array near field. Two different techniques can be used to perform source localization with this system; when the source is farther than ten centimeters, beamforming technique using phase shifts is the preferred technique. When the source is located closer than ten centimeters, the sound pressure magnitude information is the technique of the choice. Beamforming is implemented to give the direction of the incoming acoustic pressure wave and is based on time delay and sum operations. Techniques mentioned above can be used in conjunction, improving the efficiency of this system. Furthermore, the waveforms acquired by the system are digitally stored and can be used for source identification. This identification is done by comparing the digitized waveform against a database of interest such as the waveform signature of a bridge with structural failures.

II. ACOUSTIC ARRAY

This study uses a specially designed acoustic array called Acoustic MEMS Array or AMA [1]. The design of the acoustic array board was based on three basic requirements; good spatial resolution, high signal to noise rate (SNR) and good acoustic aperture. Spatial resolution is governed by the number of MEMS microphones, inter-microphone distance, and microphone sensitivity. 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 center-to-center in the horizontal and vertical.

It is possible to observe the assembled printed circuit board of the sonic array on *Figure 1*. In the figure, the array is shown stacked on the top of the Node Processing and Control board (NPCB) [2], this is the board that manages the array. The spatial sampling rate of this array can be approximated by dividing the sound speed by the 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 any 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 have in general a sensitivity of 1V/Pa at 1 kHz [4]. These microphones are Omni-directional and when combined in the array provide a very good acoustic aperture.

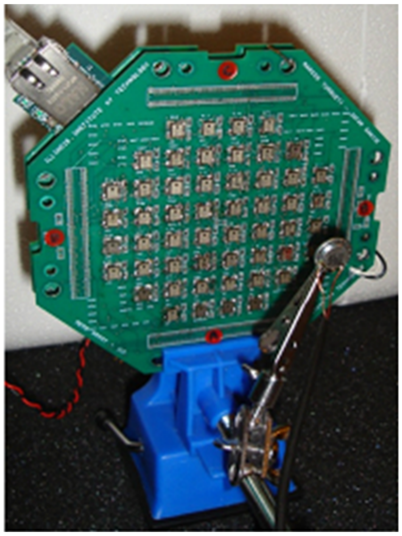

Figure 1. Top view of the acoustic array. The MEMS elements are clearly visible on the top of the printed circuit board (PCB). In this picture the array is going through a calibration process.

The MEMS microphone is a fundamental piece of the array due to the small size, high sensitivity, and low reverberation. This array can be equipped with two distinct types of MEMS microphone, the first one with frequency response from 12 kHz to 65 kHz and the second one with a flat response from 10 Hz to 10 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 surface mounted with silver epoxy glue. 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 a calibration element, the loudspeaker emits a set of tones that are then captured by the microphones and used for its calibration, taking in account the geometry of the array. When the central loudspeaker is used for active sonar applications, a series of pre-programmed pulses is generated and captured back by the array.

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 and one in the back of the PCB board. Individual ADCs are provided for every channel; each ADC has 12 bits resolution, 2.0 Volts dynamic range and a maximum sampling rate of 400 ksps, which allows the sampling of ultrasonic frequencies up to 200 kHz.

Due to the nature of the array, this circuit board requires extensive use of decoupling capacitors. There are a total of 52 10 uF capacitors on the back of the printed circuit board, one for every channel. This turned to be fundamental, 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 set at 400 Hz.

The first stage amplifier of this circuit 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 Node Processing and Control board, stacked underneath the array.

III. Detection and Localization Algorithm

The development of the detection and localization algorithms demonstrated on this paper was performed with frequencies within the sonic range for simplification of the setup. In order to test the phase discerning and beamforming capabilities of the system, an experimental setup was assembled with a source emitting a 0.5 seconds chirp waveform ranging from 2 kHz to 5 kHz. The source was placed at five different localizations as illustrated on *Figure 2*.

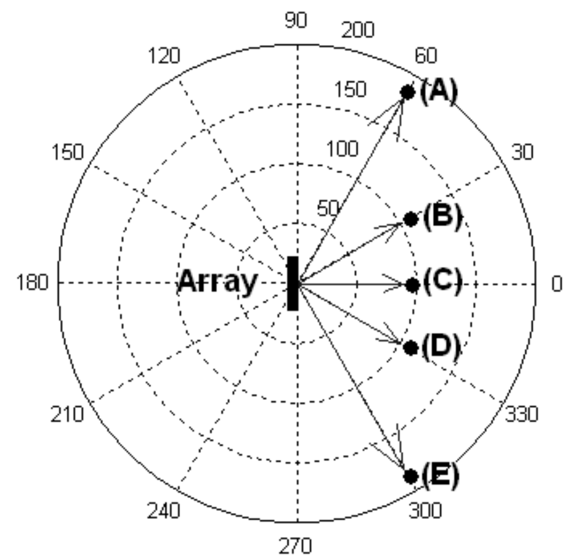

Figure 2. Test setup. Position relative to the array where the chirp source was introduced (A) located at 60 degrees left and 180 cm distant, (B) located at 30 degrees left and 100 cm distant, (C) located straight ahead and 90 cm distant,(D) located at 30 degrees right and 100 cm distant and (E) located at 60 degrees right and 180 cm distant.

The array was initially set to acquire 200 samples of data and it was triggered by the acoustic source initially set at position (A). The array captured data with all its sensors activated. *Figure 3* illustrates the signals captured by two microphones on opposite sides of the array. The phase difference between the signals can be clearly observed, and it was measured in six samples with the ADC digitizing at a sample frequency of 36 ksps. This is the basic principle on which phase-based source localization is realized on this work.

The algorithm initially implements peak detection, where it detects the position of the first peak and then fits a fifth order polynomial to it. After that, the first derivative of the fitting is calculated yielding a more precise peak identification. The sample where this peak occurred is then stored and this process is repeated for the whole acoustic array and stored in a matrix called the phase matrix.

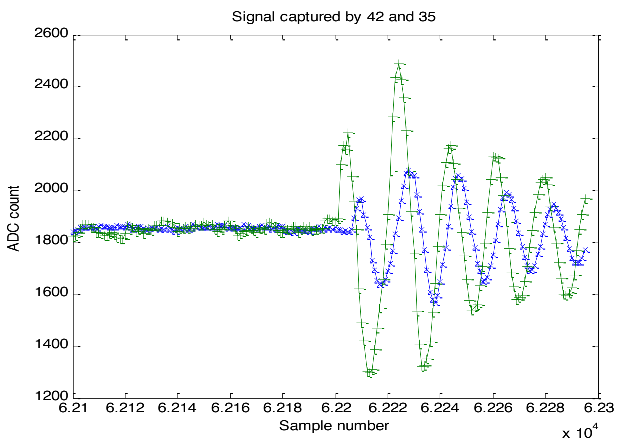

Figure 3 Phase difference from a signal captured by two different microphones on the sonic array originated from a source at 60 degrees from the array normal and 180 centimetres far. In green, microphone 42 and in blue microphone 35 are shown.

The phase matrix basically provides all the information that the system needs to perform beamforming, and source localization. *Figure 4* displays the phase matrix. It's possible to observe on this picture that raw direction from where the sound originated can be obtained by the histograms of the rows and columns of the matrix. A further improvement on source localization can be achived by performing interpolation of the phase matrix and then calculating the center of mass of the system, which can be observed on *Figure 5*.

With the phase matrix, the system adjusts all the channels to have the same phase and then sums and accumulates all the signals. This will result in a single signal that will be called the template vector. Figure 6 illustrates this concept with two microphones.

All the signals coming from the sonic array can now be compared with the template vector, making it possible to immediately realize if the source is moving and in what direction by comparing any of the sources with the template vector. This is the simplest form of beamforming that can be performed with the sonic array. The array can also have multiple templates. For example, the columns can have one template and the rows another, in this way a more precise vertical and horizontal tracking and localization is possible.

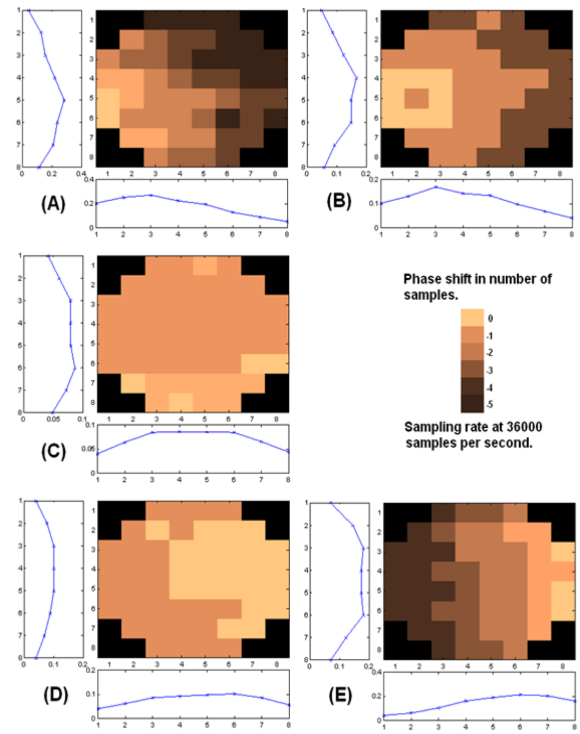

Figure 4. The phase matrix for the five positions wish the test source was placed in, (A) 60 degrees left,(B) 30 degrees left, (C) perpendicular, (D) 30 degrees right and (E) 60 degrees right.

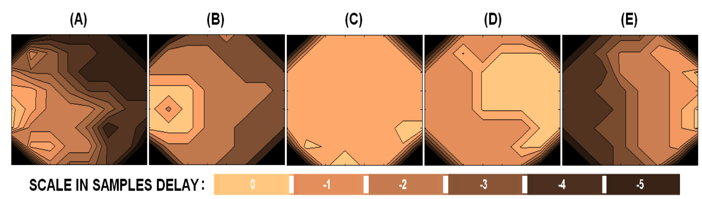

Figure 5. The interpolated phase matrix for the five positions wish the test source was placed in, (A) 60 degrees left,(B) 30 degrees left, (C) perpendicular, (D) 30 degrees right and (E) 60 degrees right.

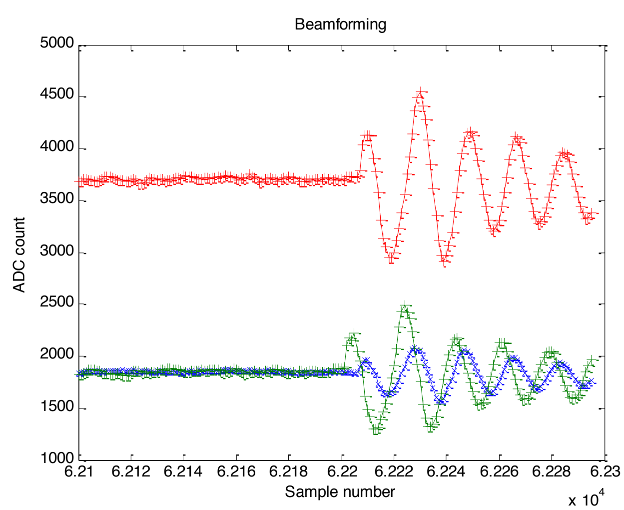

Figure 6. Beam forming with two microphones. Microphone 42 in green, microphone 35 in blue and on the top the template vector in red.

Source localization using the sound pressure magnitude requires the gain of all channels in the system to be equalized. After gain equalization, the source localization becomes trivial. The calibration is performed by placing a calibrated loudspeaker at 20 cm orthogonal to the array and then collecting data. After calibration, the system can be deployed for source localization. Basically, the signals are collected and interpolated. The center of mass of the matrix is calculated, thus indicating the position of the source or sources. *Figure 7* shows the results obtained by utilizing this method. Two sound sources were placed a distance of 2 centimeters from the acoustic array, the first one right in front of microphone number 16 while the other one in between microphones 44 and 37. It is pretty clear from the picture that the array is capable of locating both sources just by measuring the sound intensity difference.

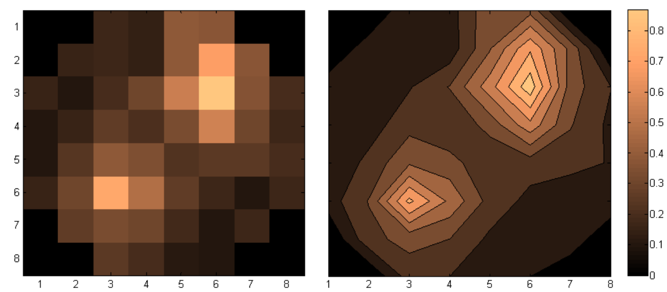

Figure 7. Two sources located, on the left the raw data and in the right the interpolated data pinpointing the sources.

IV. Recognition Algorithm

The recognition algorithm compares the input vectors X_n acquired by the acoustic array with the signals of interest on the user's digital pattern library. The patterns can be any kind of acoustic signal that the user desires to recognize.

The algorithm works by creating a window containing the search pattern and then running it through the incoming signal and continuously comparing it against a slice with same size of the incoming signal. Initially, the algorithm tries all combinations for correlation. This is achieved by applying the Pearson product-moment correlation coefficient [6], also known as population correlation coefficient and given by

$$\rho_{X_n,X_m} = \frac{cov(X_n X_m)}{\sigma_{X_n}\sigma_{X_m}}$$

where ρ_{X_n,X_m} is the population correlation coefficient, X_n is the input variable that we want to evaluate against the search pattern X_m, and σ_{X_n} and σ_{X_m} the respective standard deviations. The population coefficients are then compared with the correlation threshold variable set by the user. If the value is above the threshold, the signals are considered different, if bellow, the signals are considered the same. If the signals are considered the same, they are then logged using as reference the time stamp of the input signal.

This technique requires the incoming signal to be preprocessed. Preprocessing prepares the data for the efficient application of the algorithm and in this case it is composed of centering and normalization.

Normalization is done first and consists of the mapping of the data set from their Analog to Digital Converter range to the values between 0 and 1. Centering follows and consists of subtracting the mean vector for the incoming variable making it a zero-mean variable. This is done based on the window size of the search pattern.

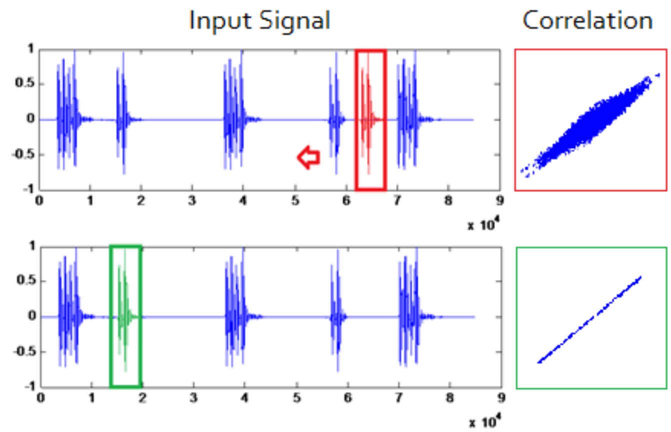

Figure 8. This figure illustrates how the process of pattern recognition happens. On the top, the search pattern goes through the signal looking for a match. On the bottom the correlation coefficient reached the correlation threshold indicating a match.

V. Conclusion

This work describes the development and capabilities of an object imaging detection and recognition system specially designed to work in conjunction with the AMA array. The algorithms have been applied to data collected during the test of the system and results were presented. Results have demonstrated that this system is promising. The next step for this work is to test the system with ultrasound sources in an anechoic chamber. After that the system will be tested in a non-controlled environment.

This research demonstrates that it is possible to integrate key technologies such as MEMS, high performance FPGA, and Gigabit Ethernet to produce a very compact, network based acoustic array that delivers high performance computing power for sonic imaging applications.

VI. References

[1] M. Turqueti, J. Saniie, and E. Oruklu, "MEMS Acoustic Array Embedded in an FPGA Based Data Acquisition and Signal Processing System", *IEEE I2MTC conference*, 3-6 May 2010.
[2] M. Turqueti, R. Rivera, A. Prosser, J. Andresen, and J. Chramowicz, "CAPTAN: A hardware architecture for integrated data acquisition, control and analysis for detector development", *IEEE Nuclear Science Symposium*, 3546-52, October 2008.
[3] E. Weinstein, K. Steele, A. Agarwal, and J. Glass, "A 1020-Node Modular Microphone Array and Beamformer for Intelligent Computing Spaces", *MIT/LCS Technical Memo*, MIT-LCS-TM-642, April, 2004.
[4] Knowles acoustics, SPM204 and SPM208, Datasheet, Rev C, 2006.
[5] J. Benesty, J. Chen, Y. Huang, "Microphone Array Signal Processing", Springer Topics in Signal Processing, first edition, Springer, Vol. 1, 2008.
[6] *P. J. Huber, Robust Statistics*, Wiley, 2004.