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ONE SENSOR MICROPHONE ARRAY APPLICATION IN SOURCE LOCALIZATION

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Abstract

The conventional digital signal processing dictates that a signal must be sampled at Nyquist rate or higher (over-sampling) in order to be represented without error. For multi-channel microphone arrays, it calls for the same bandwidth for each of the channels. Not only good sensors are needed, but many channels of A/D converters are required. This is wasteful of sensing and data acquisition resources.
Over the past years, a new theory of Compressive Sensing Processing (CSP) has begun to emerge [1] [4] [5], in which the signal is sampled and compressed simultaneously at a greatly reduced rate. Exciting new theory and applications are popping up in analog-to-digital conversion, image processing and other areas. This paper will explore the theoretical foundations and emerging applications of Compressive Sensing. We are especially interested in low cost, high channel-count applications in acoustic applications.
In this paper, an innovative design is proposed for noise source identification. The hardware required is a single microphone sensor that utilizes mechanical multi-gate array with deterministic or random opening sequences. The analog to digital conversion is accomplished by a high speed single-channel A/D converter. Since only one sensor and one A/D channel are needed, the electronic circuit and hence the associated cost can be reduced. The computational results are checked with known sources to make sure the algorithm is working.

1. INTRODUCTION

A new concept is formed in dealing with complicated design and high cost of multiple sensor array system. The main characteristics of the new system are described in the following:

- Simplifying the multiple sensor array system design to be modular design, in the case of upgrade or downsizing the array, only front end of the system (gate elements) and software algorithm need to be changed

- The system comprises of the front end (gate elements), passive or active control of gate elements state, and single sensor and single channel of data acquisition with high bandwidth.

- Since the semi-active or passive gate element has replaced actual active sensor, the total system cost can be drastically reduced, it is not so apparent in a 10 channel array, but could make a lage difference in a 100 channel array, comparing the cost of 100 microphones with the cost of one microphone plus 100 gate elements. Also the cost of cabling is a concern, with one microphone instead of 100 microphones, the cost saving in cabling along is 100 folds.

- Another advantage is that when special physical quantity needs to be measured, for example, ultrasound or subsonic low frequency pressure wave, only one special sensor is needed, not multiple sensors.

- With the advent of electronics, it is fairly easy to find a single channel data acquisition system that is capable of the total bandwidth needed. Since there is no polling between data channels, a sample-and-hold circuit is not needed to preserve phase between channels. All signals go through the gate simultaneously.

The concept of this sensor array application is not limited to microphone sensors, it can be used as vibrational, seismic, gas, optical sensors, etc. For example, it can be used for medical instruments [3]. The processing algorithm used is statistically based, and is concentrated on source localization. The same design and the theorem will be illustrated by a simple 4x4 array to start with, and then the potential of this new design will be discussed.

Figure 1 shows the basic diagram, with one source, source weights relationship between source and gate, receiving weights between gate and the sensor, along with the DAQ unit and computer interface. The red elements in the gate represent the closed element positions, i.e., the obstructed positions, and is defined as the binary position "0".

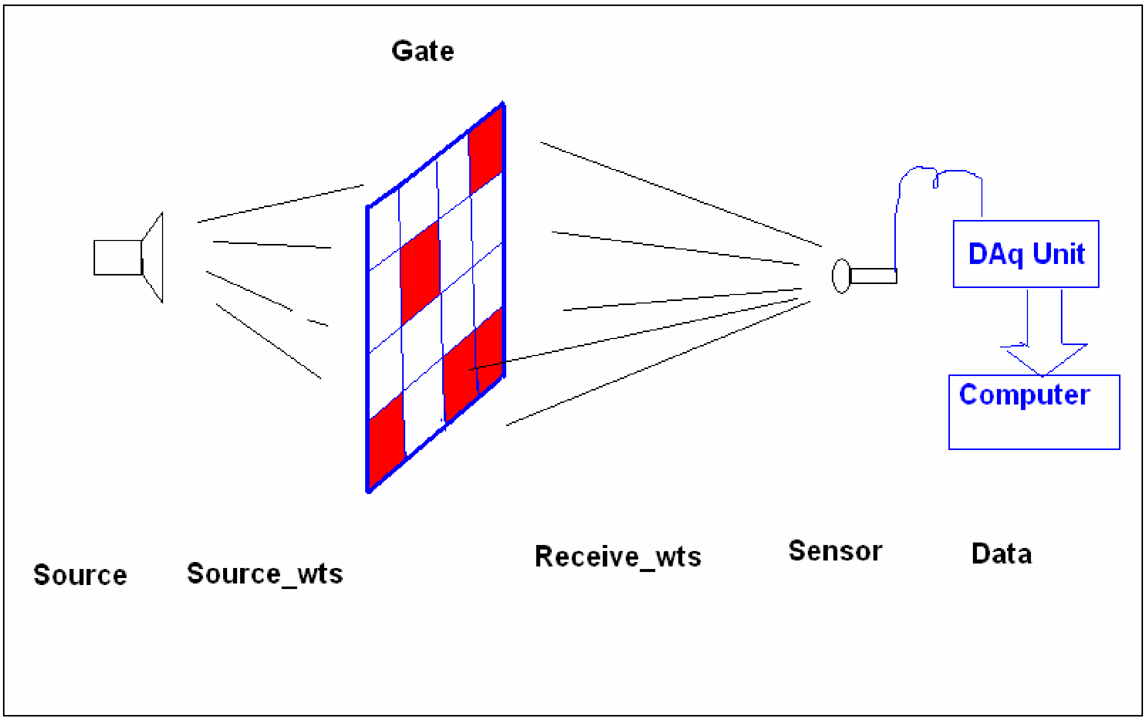

Figure 1. One microphone array setup for gathering noise data through gate array

2. DESIGN CONSIDERATIONS

These are many possible types of single microphone array configurations can be considered, below, two of these types are described in detail.

2.1 Blockage Type

As shown in Figure 1, the source and microphone are on the opposite sides of the gate. The gate is composed of many array elements. In this type, the array element is in the "blocked" position, when the small grid is closed by the grid door (like rotating door), we call this binary position "0", when there is no sound transmitted through that grid and reaches the sensor. When the grid door is open, this is called binary position "1". Assuming the gate plane is fairly large and there is minimum leakage, it is fair to say the only path for sound to reach the microphone is via the gate elements.

5.2 Reflective Type

In the Reflective Type setup, the source and sensor are on the same side of the gate array. In this configuration, the gate array element can be more complicated, because it needs to reflect sound from source to aim at the sensor or elsewhere. A parabolic reflector geometry should be able to collect sound wave, and re-direct them to the sensor effectively. When reflector is focused at the sensor, it is said to be in binary "1" position, where the sound is collected at the sensor location. When at binary "0" position, then reflector at the array elements directs sound to elsewhere than the sensor location.

The authors decided to use blockage type in this paper for demonstration because it is easier for hardware realization and simpler to set up for simulation and demo. In theory, both types should work very similarly. In the following, most simulations are run with 4x4 (16 elements) gate array.

3. METHODOLOGY

The beauty of compressive sensing used in this application is that it considers all frequency contents, without having to worry about phase difference. It considers constructive and destructive signal not in a sense of narrow band sinusoidal phase, but in a grand scheme, how sensor can be exposed to source in randomized nature, and how these information can be combined together for signal reconstruction and estimation of source location.

The array elements are controlled by a random number to close or open, if the number received by the controller is 1, then it opens the elements, if the number is "0", it blocks the array elements. The random numbers decide how the source is presented to the sensor, and are generated by pseudo-random number generator. This is close to the random array concept as presented in [2], but here uses passive array to replace active array.

Let's assume the gate array is fairly close to the source plane, and there is no reflection from the grid back to the source. Since we have no way of knowing the source spatial distribution, we set source weights to all one's. The source plane is assumed to be a

2-dimensional plane, and is parallel to the gate plane.

$$Mic(runs) = \sum S * Wt_{source} * Gate_{binary} * Wt_{receiving} \qquad (1)$$

$$\mathrm{SPM} = Xcorr(Mic(Runs)), \qquad (2)$$

where Mic(runs) are the received value at the microphone sensor, and SPM is defined as Source Probability Map, and is the accumulated results after several runs, a cross correlation functions is used to find the most likely source distribution.

Figure 2 shows the assignment of different weighting between gate array and the sensor. In a 4x4 array, the 4 corners are assigned with weight x, the outer edges are assigned with weight 2, and the 4 elements close to the center are assigned with equal weight of c. The sensor are presumably place in alignment with the center of the array, so all the elements closest to center should receive equal weights. The weights describes the distance relationship between the elements to the sensor, the farther the distance, the smaller the weight.

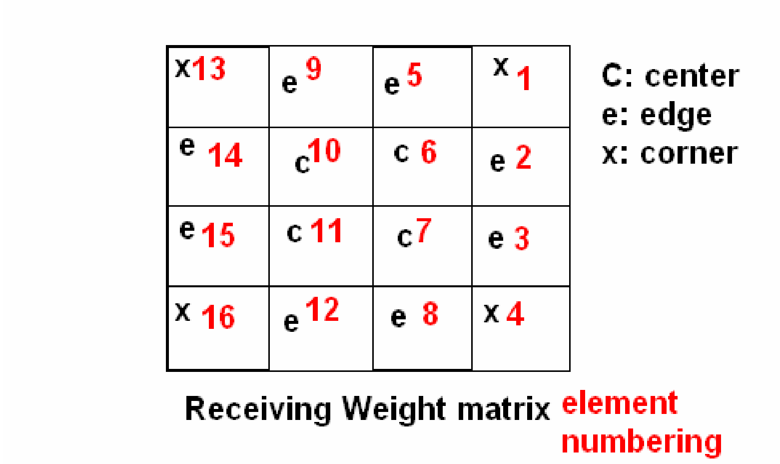

Figure 2. Receiving weight (Wt: receiving) matrix between gate and sensor..

4. COMPUTER SIMULATION AND RESULTS

In the simulation that follows, this set of weighting x=0.6, e=0.8 and c=0.9 were used. The results of computer simulation done by MATLAB code are presented here in different case scenarios.

4.1 Single Source (Case1)

Assume the source is located at around element 6 on the source plane (see Figure 2 for the assignment of element numbering in red color), following algorithms in Equations (1) and (2). The Source Probability Map result is shown in Figure 3. For simplicity of plotting, the plots are presented as contour plot instead of color maps. In Table 1 is shown the tablet version of the value results of the SPM. The result predicted the source location very well.

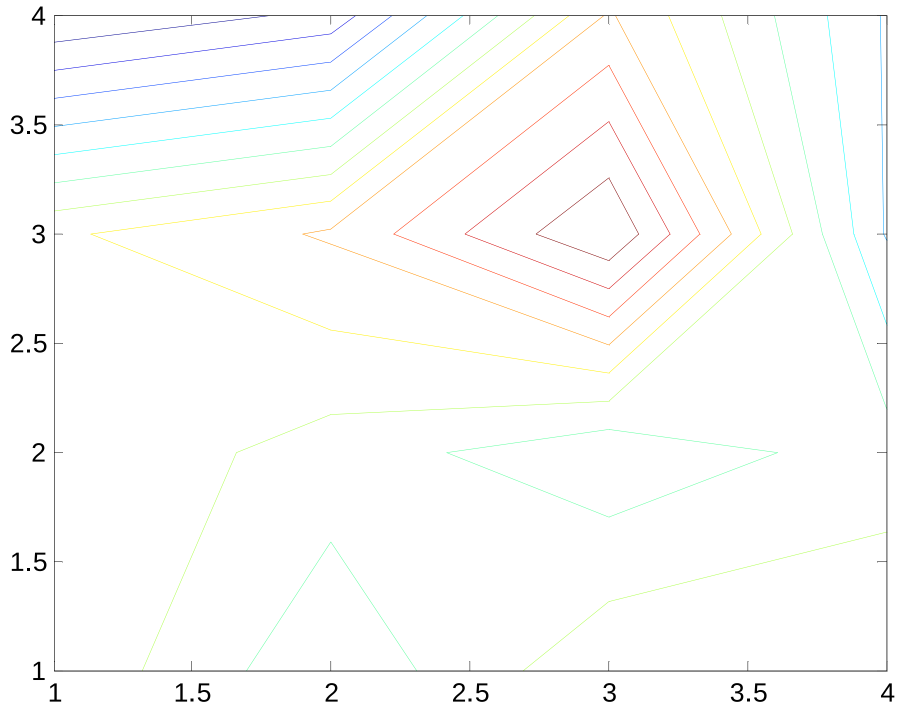

Figure 3 SPM results of case1, source at location 6, 20 runs, and minimal 0.2% noise floor

Table 1. SPM result of Case 1

4.2710	8.2600	32.4970	16.3590
28.5040	32.5610	44.5270	16.4010
28.5740	24.5900	20.4430	24.4100
28.6020	20.3510	28.4510	28.5680

4.2 Introduction of Noise (Case2)

In case 2, noise floor is elevated to 2% instead of previous 0.2% (location 6 is the upper right central element). The noise floor is elevated, and result is somewhat distorted, but as one can see, the peak of the map is still located at the right place (location 6).

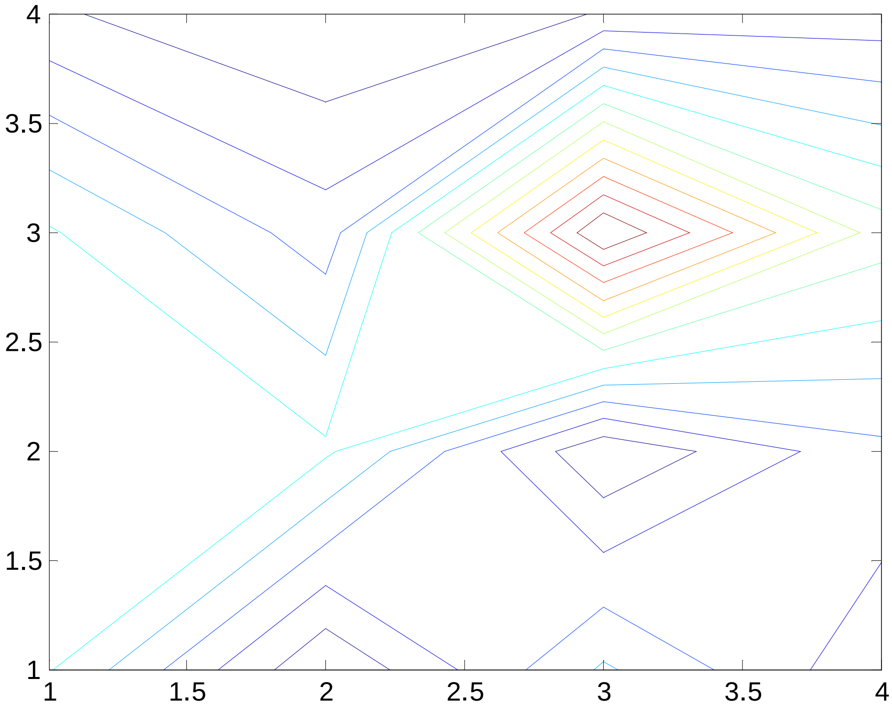

Figure 4 SPM results of case2, source at location 6, 20 runs, with noise floor 2%

4.3 Multiple Sources (Case3)

In case 3, other than the source at location 6, another discrete source of same amplitude is placed at location 10, which is to the left of location 6. The SPM result in Figure 5 shows the identification of the 2nd source, although a little bit skew on the contour map. The sources as indicated seems to be at the right position.

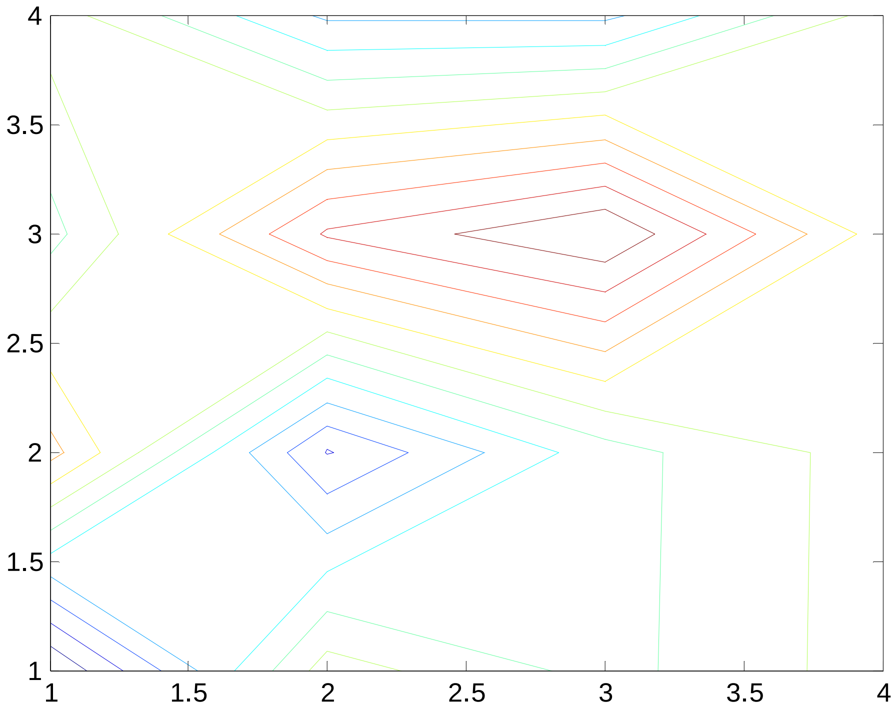

Figure 5 SPM results of case3, source at locations 6, with an additional source of equal strength at location 10

6. CONCLUSIONS

The algorithm used here is an approximation of compressive sensing, which is a mathematical tool for getting data out of limited information. The initial results suggest this method can be expanded to use in large array applications, especially when the source is spatially distinctive. The same algorithm can also be used for simpler application, for example, in a 4 zone area, a speaker moves around, you just want to freeze time and find the location of the speaker.

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