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LOW COST BEAM FORMING SOLUTION FOR A DEVELOPING COUNTRY

Zair Asrar Bin Ahmad* Raja Ishak Bin Raja Hamzah†

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ABSTRACT

Beamforming is an effective method for sound localization problem. In developing countries like Malaysia, most of the acoustic and beamforming equipment is typically supplied by established United States or European manufacturers. They are very expensive equipment with cost comparable to our houses. Only universities aided with special grants from the government can afford it. Even the basic building block of a beamforming system, the microphone is very expensive, more than our monthly salary. Armed with internet, a low cost microphone and beamforming system is being developed in the Faculty of Mechanical Engineering, Universiti Teknologi Malaysia. This is our attempt to start the open hardware movement here which should follow the open source (software) movement that is being widely embraced worldwide. Our microphones and beamforming system are developed using the available resources in our university.

1 INTRODUCTION

The detection of sound sources was usually done by hand with level measuring instruments or sound intensity probes. This manual scanning can nowadays be replaced by stationary arrays of microphones that allow the examination of complete structures within one measurement. The output of such an array usually results in a noise map, which can be overlaid on a digital photo of the test object. The noise measurement can be directly compared with a visual image of the object, thus significantly facilitating the interpretation of the acoustic data. Unfortunately, commercially available microphone arrays are rather costly and hardly affordable, especially for third world universities like ours. For this reason we started to consider building such a microphone array by ourselves.

*Faculty of Mechanical Engineering, Universiti Teknologi Malaysia, 81310 Johor Bahru, Johor, Malaysia, zair@fkm.utm.my

†Faculty of Mechanical Engineering, Universiti Teknologi Malaysia, 81310 Johor Bahru, Johor, Malaysia, rishak@fkm.utm.my

The lab equipment in Malaysian universities are sources from well known equipment manufactures coming from western companies i.e United States and European countries. These include Brüel & Kjær, National Instruments, Dewetron, GRAS and PCB Piezotronics. The same can be said for our Noise and Vibration Lab at Universiti Teknologi Malaysia. These equipment are nonetheless good and robust. However, they are very expensive which require special budget from the government that is hard to get. Furthermore, the situation get worse as the equipment cannot be bought directly from the manufacturer overseas. The procurement process need to be made through multiple international and local distributors that markup the price considerably.

In our lab, there is only a few 1/2 inch GRAS microphones. These are our prized possession such that students rarely had a chance to use them. Fearing that the students might damage them, handling is done mostly by our technicians. As the number of microphones are very limited, having a beam forming array is almost out of our mind. Recent visits by Brüel & Kjær local distributor indicates that a beam forming array system cost at least more than half a million Malaysian Ringgit (RM) equivalent to a couple of hundred thousand Euros. Even the impedance tube (with only two microphones) cost about RM400,000. Therefore, alternative cheaper and affordable solutions are badly needed for us.

2 IDEAS FOR THE PROJECT

We have been inspired by Bischof [2] that has developed a beam forming array system through junior year student projects in a university in Graz, Austria. The high cost of the commercial beam forming system is one of his main motivation for the project which is similar to ours. They have used microphones from ROGA Instruments while the beam forming algorithm is done using Matlab. We are adopting the same approach but we need to do it by using our own microphones instead. Even the low cost ROGA Instruments microphones is still expensive for us, as a large quantity of microphone is needed in a beam forming array.

An interesting statement in Bischof's work is that the ROGA Instruments microphone is actually an electret microphone. After doing some market research in Malaysia, we can get an electret microphone component from electronic shops for just RM1 (roughly equivalent to €0.25). The challenge for us is how to get these cheap electret microphones to work as good as our expensive GRAS microphone.

They are many previous researchers who are also attracted by the idea of using cheap electret microphone element in beamforming array especially for aeroacoustic applications [4–6]. However, their electret microphone element is powered using external power source [1, 6]. In this contribution, we are using electret microphone element with ICP circuit. These microphones would then be connected to available DAQ cards in our lab to form a beamforming array system. The beamforming algorithm is made using Matlab. Finally, the acoustic imaging result of our proof of concept low cost beamforming system is presented.

3 IMPLEMENTATION

3.1 Hardware

The electret microphone component requires power to operate. Connecting them directly to a sound card is one of the option as the sound card can power the microphone component as well as boost the sound output. However, the SNR is still low. Another alternative for us is to use the available DAQ card in our lab which is the NI-PCI 4472 card. This card has a feature to turn on IEPE/ICP (constant current source) at the input channel. This will power IEPE microphone attached to the card. Therefore, we need to make an IEPE microphone using the electret microphone component. A way to do this has been suggested in [3]. The constructed microphone is shown in Figure 1.

The performance of the developed ICP microphone across the frequency spectra is assessed using white noise. A normal PC speaker (effective range up to 10kHz) has been used to generate the noise. Figure 2 shows that our developed ICP microphone is comparable in term of performance to our GRAS microphone (Type 26AH).

In terms of connection, the signal from the GRAS microphone is send to the DAQ card via an external filter and amplifier device which also powered the microphone itself. However, since our developed microphone is a constant Ampere microphone which draws power directly from the DAQ card, the signal is send directly to the DAQ card without any signal processing. Thus, for our developed ICP microphone, any signal processing need to be done digitally.

The microphones are then mounted and arranged on a fabricated aluminum frame to form a microphone array as shown in (Figure 3). BNC cables are fabricated using RG58 coaxial cable and suitable BNC connectors are used to connect the microphones to the DAQ cards. A webcam is placed at the center of the array for taking snapshot of the measured area. This snapshot will later be superimposed with the noise map to indicate the sound source location.

In order to prevent spatial aliasing, the spacing s between any pair of microphones in the array should not exceed the smallest wavelength λ_{min} present in the signal

$$s \leq \frac{\lambda_{\text{min}}}{2}. \tag{1}$$

In out implementation, a rectangular array is used for simplicity. The array has 9 electret microphones with spacing of $s = 0.3$m. Thus, the effective frequency of our array is 560Hz. The supply current of 4mA for the microphones is delivered by the DAQ cards.

3.2 Software

Beamforming theory is already well known [7]. A delay-and-sum beamformer enhances signals from a particular direction and attenuates signals from other directions. In most beamforming applications two assumptions simplify the analysis: the signal sources are located far enough away from the array that the wave fronts impinging on the array can be regarded as plane waves (far field assumption), and the signals incident on the array are narrow-banded (narrow-band assumption).

At the moment, the development process of this beamforming array system is focusing on the hardware side. Thus, only a simple delay and sum algorithm has been implemented in Matlab. Far field sound source is considered. Consider an array consisting a total of N microphones at

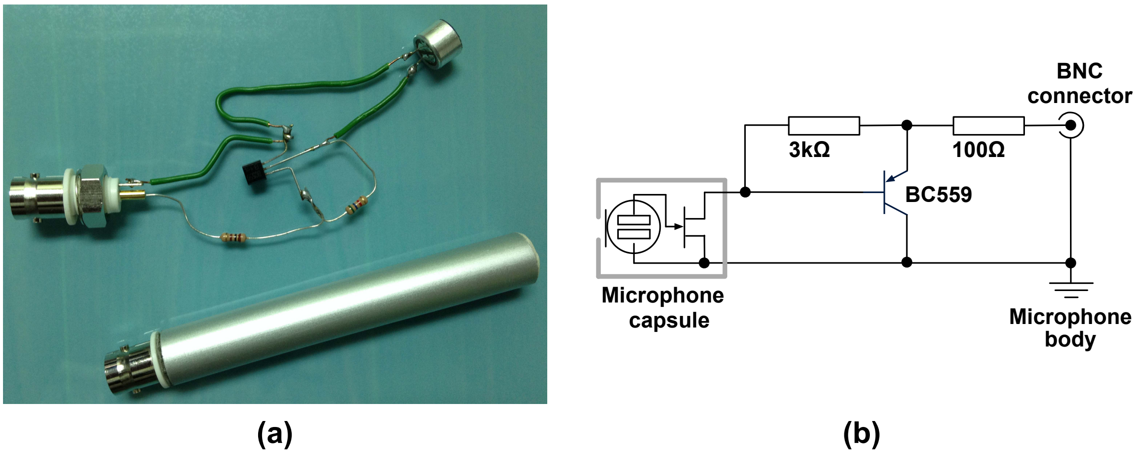


Figure 1: (a) The inner components of the ICP microphone shown next to the assembled microphone in the aluminum tube, (b) Schematics of the ICP microphone circuit.

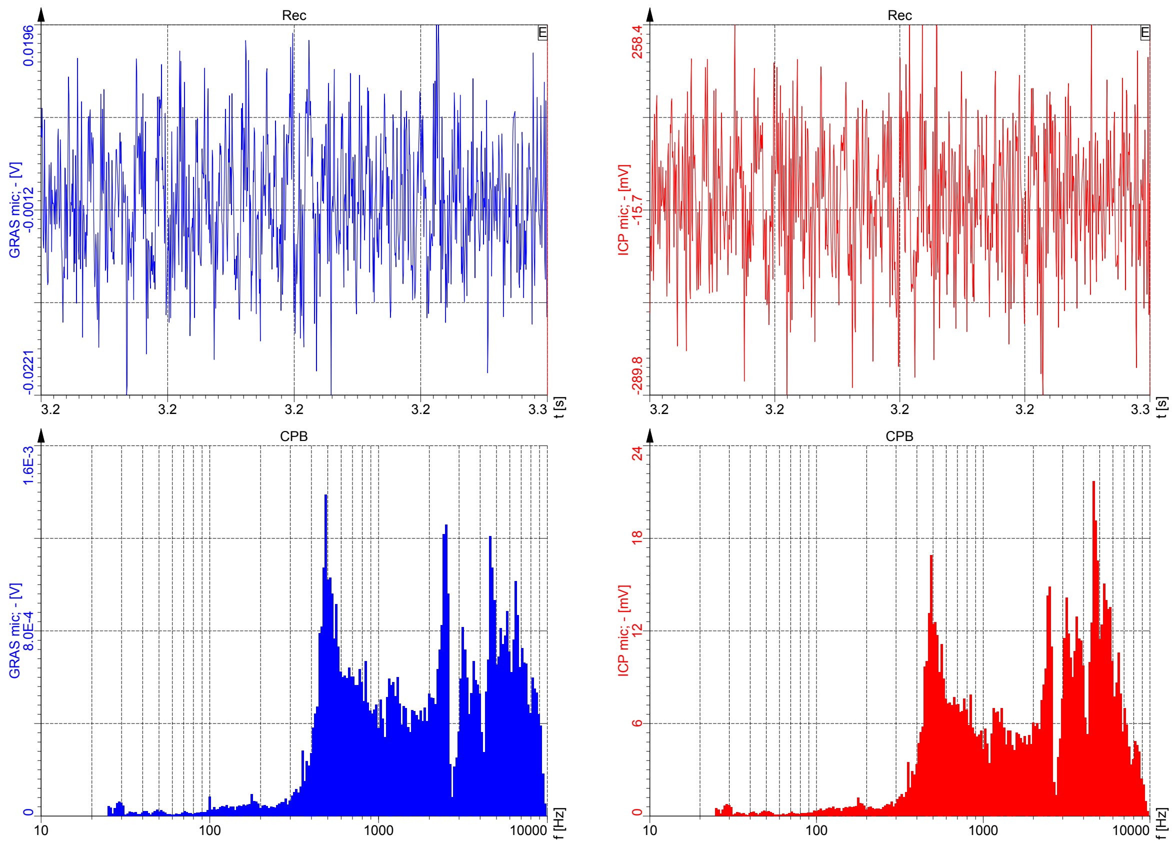
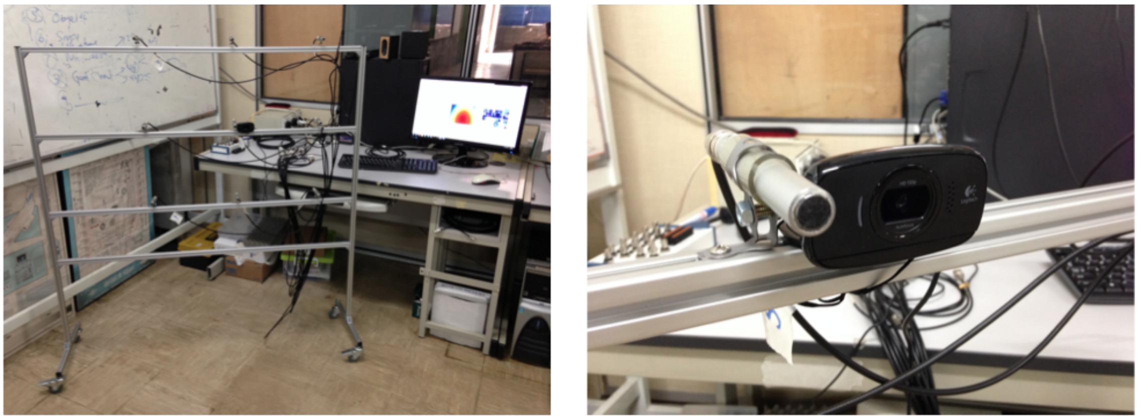


Figure 2: Comparison between GRAS microphone and our ICP microphone in time domain and 1/24 octave band for white noise.



(a) (b)

Figure 3: (a) The array of microphones attached to an aluminum frame, (b) A webcam attached at the middle of the array.

the positions $\mathbf{p}_n(x,y,z)$ with $n = 1 \ldots N$, and produces a set of signals denoted by the vector

$$\mathbf{f}(t,\mathbf{p}) = \begin{pmatrix} f_1(t,\mathbf{p}_1) \\ f_2(t,\mathbf{p}_2) \\ \vdots \\ f_N(t,\mathbf{p}_N) \end{pmatrix} \tag{2}$$

The direction of an incident plane wave approaching the array is represented by the position vector $\mathbf{a}(x,y,z)$. The origin of the cartesian coordinate system is at the centre of the microphone array.

If the distance between the acoustic source and the array is large and the microphones are identical, then the gains of all microphones must be equal. The incident plane wave hits the microphones at different times depending on its location, which leads to a delay with respect to a reference microphone. The individual time-delays τ_n are chosen with the aim of achieving selective directional sensitivity in a specific direction, characterized by the unit vector $\mathbf{a}(x,y,z)$.

This objective is achieved by adjusting the time delays in such a way that signals associated with a plane wave, incident from the direction $\mathbf{a}$, will be aligned in time before they are summed. This can be obtained by choosing

$$\tau_n = \frac{\mathbf{a} \cdot \mathbf{p}_n}{c} \tag{3}$$

where c is the speed of sound. Thus the sum of the microphone signals divided by the number of microphones, the array output $y(t)$, represents the sound pressure of the acoustic signal

$$y(t) = \frac{1}{N} \sum_{n=1}^{N} f(t - \tau_n). \tag{4}$$

Every other set of time delays leads to an attenuated array output. The fundamental principle behind the direction of arrival estimation is the use of the phase information present in the signals picked up by the spatially separated microphones. For the processing of phase information the preceding transformation of time signals into frequency domain is most suitable. With the introduction of the wavenumber $\mathbf{k} = \frac{\omega}{c}\mathbf{a}$ and the far-field assumption taken into account the Fourier transform of the microphone signals (2) gives

$$\mathbf{F}(\mathbf{k},\omega) = F(\omega)\mathbf{u_k} \tag{5}$$

with

$$\mathbf{u_k} = \begin{pmatrix} e^{-i\omega\tau_1} \\ e^{-i\omega\tau_2} \\ \vdots \\ e^{-i\omega\tau_N} \end{pmatrix} = \begin{pmatrix} e^{-i\mathbf{k}\cdot\mathbf{p}_1} \\ e^{-i\mathbf{k}\cdot\mathbf{p}_2} \\ \vdots \\ e^{-i\mathbf{k}\cdot\mathbf{p}_N} \end{pmatrix} \tag{6}$$

being the array manifold vector, which incorporates all of the spatial characteristics of the array. These microphone signals are multiplied by appropriate and in general complex weights and then summed up to get the frequency domain representation of the array output

$$Y(\omega) = \mathbf{W}^*\mathbf{F}(\omega). \tag{7}$$

The weighting vector used in this work is simply $W_n = \frac{1}{N}$ for all n (uniform shading). The beamformer output in the frequency domain, the array output power spectral density, can be written as

$$P(\omega) = Y(\omega)^2 = \left(\mathbf{W}^*\mathbf{F}(\omega)\right)\left(\mathbf{W}^*\mathbf{F}(\omega)\right)^* = \mathbf{W}^*\mathbf{R}(\omega)\mathbf{W} \tag{8}$$

where $\mathbf{R}(\omega) = \mathbf{F}(\omega)\mathbf{F}(\omega)^*$ represents the $N \times N$ cross power spectral density matrix of the channel input signals. The asterisk (*) represents the complex conjugate transpose (the Hermitian). For the localization of the sound sources the array response function is introduced. It represents the response of a microphone array to a plane wave of frequency ω incident on the array at an arbitrary direction of arrival $\mathbf{a}(x,y,z)$. With the weighted array manifold vector (or steering vector).

$$\mathbf{w_k}(x,y,z) = \frac{1}{N}\begin{pmatrix} e^{-i\mathbf{k}\cdot\mathbf{p}_1} \\ e^{-i\mathbf{k}\cdot\mathbf{p}_2} \\ \vdots \\ e^{-i\mathbf{k}\cdot\mathbf{p}_N} \end{pmatrix} \tag{9}$$

the array output power spectral density can be defined as

$$P_B(\omega,x,y,z) = \mathbf{w_k}(x,y,z)^*\mathbf{R}(\omega)\mathbf{w_k}(x,y,z), \tag{10}$$

where $P_B(\omega,x,y,z)$ represents the squared intensity of sound for a narrowband input signal of frequency ω with the direction of arrival $\mathbf{a}(x,y,z)$.

3.3 Application

Currently, we have no access to an anechoic chamber. Thus, a real measurement is taken in our lab whereby a PC speaker was used as the sound source. We are aware this would definitely affect the results. Furthermore, the number of microphone currently fabricated is still small for a beamforming array (9 units). However, just for proving the concept, our microphone array could localize the speaker as a major noise source as shown in Figure 4. The digital picture taken by the webcam is superimposed with a contour plot of the beamformer algorithm's output.

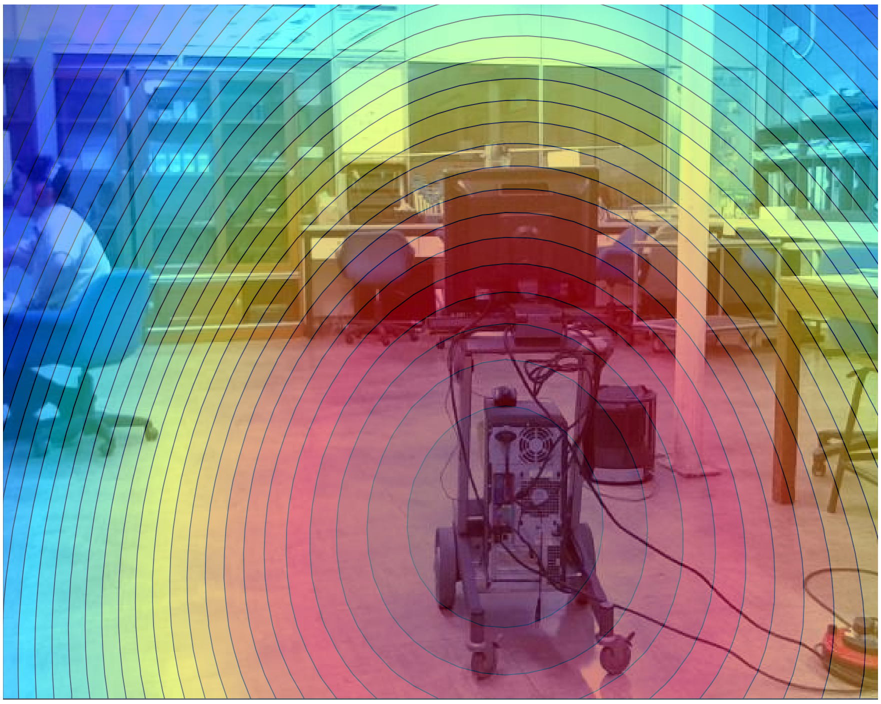


Figure 4: Acoustic imaging of sound source using the beam forming array.

4 CONCLUSIONS

The developed beam forming system is suitable for sound source localization even by using low cost microphones. The performance of our microphones is also comparable the more expensive GRAS microphones. The beamforming algorithm has also been made using Matlab platform which can be ported into open source software like Octave. However, in our research work, the link between the microphones and the beamforming code is still made using rather costly National Instrument DAQ cards. Our next research direction will be on replacing these cards with cheaper alternatives. There are researchers who use affordable on the shelve PCI sound card and USB sound card to capture the data from microphones. But, there are still no research that proves these sound cards can be use for the scale of a beam forming array system which requires a lot of sound source input. Another alternative worth exploring would be on using open source electronics such as Arduino board as the DAQ card. Apart from the hardware development, improvement on our beamforming algorithm and the considerations for near field localization and multiple sound source would be our next goals.

The developed beamforming system proves that third world universities does not need to be left behind even if they can not afford high cost equipment. We need to always bear in mind that some of these equipment can be made ourselves with the fraction of the cost. In this age, through internet and curiosity, one may find ideas that could open up for new possibilities.

ACKNOWLEDGMENT

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