Acoustic Imaging of Heart Using Microphone Arrays

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Abstract — **In this paper a novel method for acoustic imaging of heart is proposed. Heart is assumed to be a spread acoustic source which generates sound from its different locations. A simultaneous multisensor recording is used to record the heart sound from human chest wall. By using beamforming techniques the recorded data are combined to form a two or three dimensional representation of heart sound distribution. This map shows the acoustic energy of heart in different locations. Simulations with different array arrangements are done to verify the performance of this system. The heart sound is segmented without any need for other biosignals using an unsupervised method. This procedure is used to separate first and second heart sound. The segmented signal is used in beamforming algorithm. This results in a time-space map of heart sound for different intervals of heart activity. A setup is designed and implemented to record heart sound from human chest. Data are saved on a personal computer for signal processing in order to form a two or three dimensional image. This noninvasive and inexpensive imaging method offers new aspects, which with further studies, can help physicians in more precise diagnosing of heart diseases.**

Keywords — **Microphone array, acoustic imaging, near-field beamforming, heart sounds.**

I. Introduction

Initial diagnosis of heart diseases by listening to heart sound has been a very common method for a long time [1]. Physicians usually listen to the patient's heart sound in different auscultatory areas. Heart sound has different characteristics in different places on the body. This fact recommends an idea that if an electronically recorded heart sound or phonocardiograph (PCG) from different locations in front and at the back of the human chest is used, by combining these data, we can have an acoustic map of the heart.

For this purpose, we need an array of microphones which records the sound simultaneously and sends it to a processing unit. By using an array processing method we can have a spatial filter which helps us to localize the sound source. The configuration of the array and the location of microphones play an important role in array processing. In this research, we investigate different microphone arrangements to find which one is more efficient for heart sound localization.

II. Simulations

Sound is a mechanical wave which propagates through a medium with a speed which depends mainly on the material of medium and ambient conditions. It is generated by a source, travels through a medium and we can sense it with a sensor which is usually a microphone. Heart is a spread sound source which generates sound from its different parts. This source can be estimated with some point sources which generate sound separately [2]. If we estimate each point's signal energy by combining these data, we will have a map which represents the distribution of heart sound through the thorax. As a result, an image is formed that can be named as an acoustic image of heart.

A. Delay-and-sum beamformer

Beamforming is one the most common solutions for sound localization. In this research a delay-and-sum beamformer is used to localize the source. It has been shown that an imaging resolution for human chest cannot be better than approximately 2 cm [3]. So the beamformer scans the total area of the chest by focusing on each point in 1 cm steps. It estimates each point's signal energy and assigns a number between 0 and 1 to that point due to the maximum and minimum of energy in that region.

B. Sound propagation model in human chest

The acoustic properties of human chest are partly known because of its complexity. The speed of sound is varying from 23 m/s to 1500 m/s in human chest [4]. According to some literatures, sound travels as slow as 10m/s in the thorax [5]. Sound speed in chest wall and heart tissue is about 1500 m/s. The large air ways transmit sound with 222 to 312 m/s. Parenchymal sound speed is estimated to be between 23 and 60 m/s depending on air content. Damping is just about 0.5 to 1.0 decibel per centimeter at 400 Hz [3, 4].

Normal heart sound ranges in frequency from 20 to 150 Hz. Other sounds caused by structural abnormality or pathology, have the frequency up to 1200 Hz [4].

In order to verify our algorithm we conducted the simulations on the basis of these assumptions for different sound speeds and different array arrangements. A delay-and-damp model is used in these simulations [6]. In this model the

Chwee Teck Lim, James C.H. Goh (Eds.): ICBME 2008, Proceedings 23, pp. 738–741, 2009
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propagating wave is delayed and damped in its way from source to sensors. The delays are according to different times of arrival and are basis criteria in array processing. As mentioned above, damping factor in human chest is relatively low, but by considering it, we can make the model more precise. This model is presented in equation (1) [6]:

$$s\left(t-|x-y|/c\right)=d.|x-y|.r(y,t)/|x-y|^2 \qquad (1)$$

In this equation s is the recorded signal at each microphone, x is the location of each microphone, y is the location of source in any coordinate system and $r(y,t)$ is source signal. Assuming constant characteristic through the medium, c and d represent speed of sound and damping factor respectively.

C. Array arrangement

Array arrangement plays an important role in array processing. Here, we have a spread sound source which is in the near-field of the array. Our aim is to find a proper arrangement to fit the geometry of human body and at the same time localize the source with least error possible. The simulations are done in two dimensional and three dimensional arrangements. The 2-D arrangement is used for 2-D maps and the 3-D arrangement for the 3-D maps. Several arrangements are tested in each category. Figure 1 shows three typical array arrangements.

A single source of sound is located in a medium with a sound propagation speed of 10, 40 and 344 m/s. 10 m/s is the least propagation sound speed, 40 m/s is an average speed in human body [3] and 344 m/s is the speed of sound in air at room temperature.

The location of the source is changed in 1 cm step size and the error in detecting the location of source is measured as the distance between the detected source and the real source. The total error of each arrangement is defined as the average of these errors. The total error and the standard deviation of the error of each arrangement are shown in tables 1, 2 and 3. The region of interest is set to match the typical human body size. Signal source is a 500 Hz sine as well as a recorded heart sound.

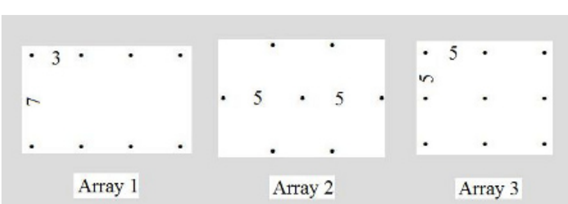

Fig. 1 Array arrangements (distances between microphones are in cm)

Table 1 Total error and standard deviation of the error (speed of sound is 10 m/s)

Array arrangement	Total error (cm)	Error standard deviation (cm)
1	0.02	0.27
2	0.002	0.05
3	0.0	0.0

Table 2 Total error and standard deviation of the error (speed of sound is 40 m/s)

Array arrangement	Total error (cm)	Error standard deviation (cm)
1	0.36	1.57
2	0.35	0.36
3	0.0	0.0

Table 3 Total error and standard deviation of the error (speed of sound is 344 m/s)

Array arrangement	Total error (cm)	Error standard deviation (cm)
1	3.65	2.34
2	4.57	2.60
3	2.31	1.15

The result of simulations for 3-by-3 array at the speed of 40 m/s is presented in figure 2. This array has the least total error and standard deviation of the error among other simulated arrays. It can be seen that the algorithm separated two sources distinctly. In these images points with higher energy are darker and points with lower energy are brighter. Figure 3 shows a three dimensional image. A three dimensional array is used in this case. The array is consist of two 3-by-3 arrays, one in front and one at the back of the chest. So, 18 microphones are used in this array. Because of overlapped voxels, one percent of the points are presented in this image. A side lob can be seen in the image, but it not in the region of interest, so it can be neglected.

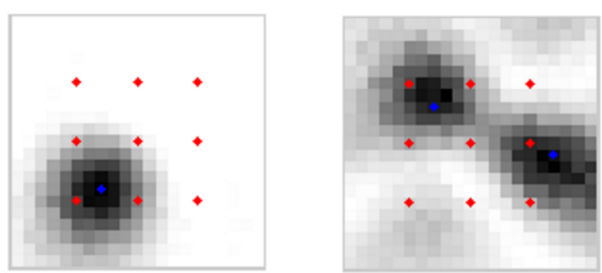

Fig. 2 Localization of one (left) and two (right) acoustic sources (red points indicate microphones location and blue points indicate sources)

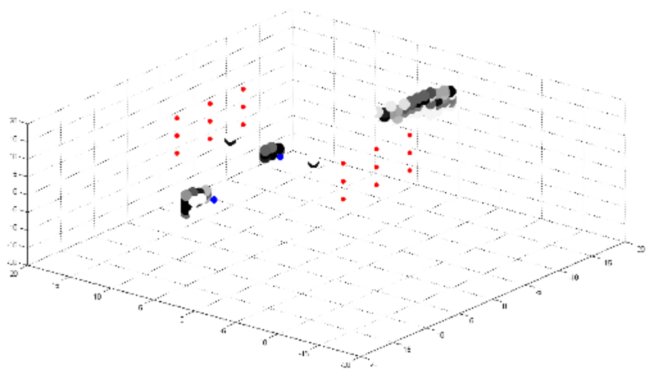

Fig. 3 Three dimensional localization of two acoustic sources (red points indicate microphones location and blue points indicate sources)

III. Segmentation

Heart sound is generated from different places in heart in different intervals [2]. The place of source is different during first heart sound (S1) and second heart sound (S2) so if we separate the sources in time domain, it can help a lot in the problem of source separation and generating the map. We used a heart sound segmentation method presented in [7]. This method doesn't need any extra biological signal (e.g. ECG) as a time guide. More over it overcomes the problem of asynchrony between these extra signals and the phonocardiograph as sometimes occur in certain diseases.

In this method a Morlet bank is used because of its similarity to heart sound. The energy profile of the signal is calculated using this wavelet. Peaks in the energy profile derived from the time-scale representation, are identified and are used to obtain segments containing any activity. Then the singular value decomposition (SVD) of the signal is calculated to determine which segment has a fundamental activity. Due to the following characteristics of PCG the segmented signal is labeled either as S1 or S2: (1) sequential repetition of fundamental activities, (2) the duration of time interval between S1 and S2 (systole) is longer than the duration of time interval between S2 and S1 (diastole), and (3) the heart rate is less than or equal to 160 beat per second. Figure 4 illustrates a PCG signal and its segmented energy profile. These labeled signals are used separately in the beamformer to form time-space maps. More details can be read in [7].

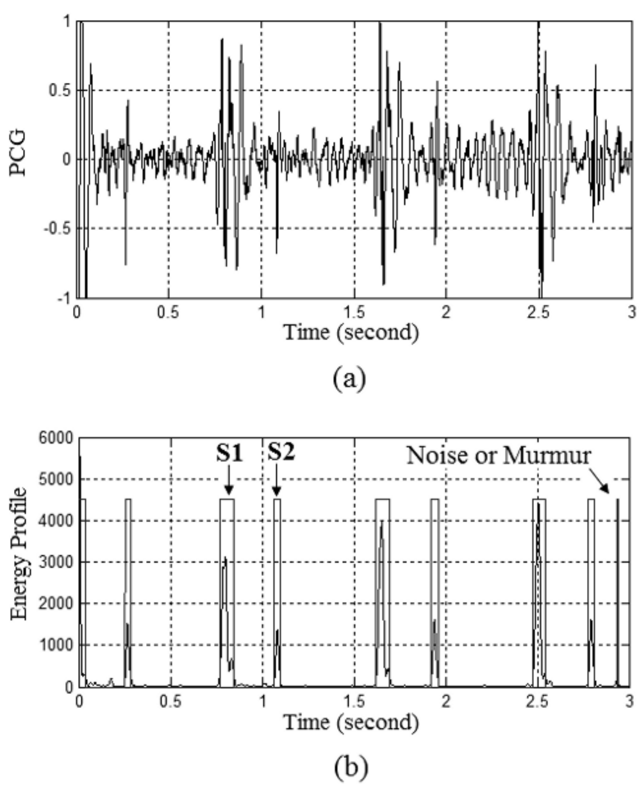

Fig. 4 An example of PCG and its corresponding energy profile. (a) PCG (b) Energy profile

IV. Experimental tests

An experimental setup is prepared for acoustic imaging of heart. It consists of software and hardware units. Software unit process the acquired data to form the acoustic map. Beamforming, segmentation and data presentation are performed in this unit.

Hardware has three major parts: microphone array is made up of Panasonic electret condenser microphones WM-52B, preprocessing unit which amplifies and filters microphone signal in order to enhance sound signal to noise ratio and data acquisition card (DAQ) which is an Advantech PCI-1713 card playing the role of interfacing with a Pentium IV platform PC.

Noise plays an essential role in experiments. In order to eliminate it, firstly we used a low pass filter in the preprocessing circuit with cut-off frequency of 1200 Hz. In software unit the frequencies below 20 Hz are damped with a high pass filter. Some noises such as electricity lines 50 Hz and its higher harmonics and environmental noises such as computer fan noise are inevitable. These noises occur in frequency band that heart sound has information in it and we cannot eliminate them with regular filters. In order to overcome this problem an extra microphone is used to collect noise and subtract it form the main signals adaptively. This method helps enhancing SNR and therefore signal's quality.

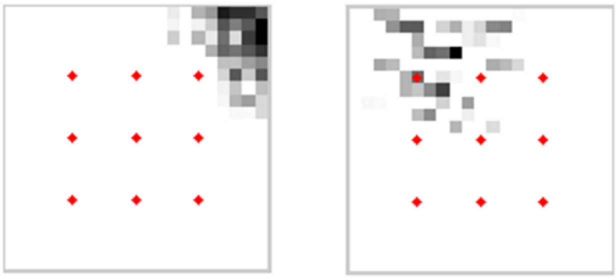

Fig. 5 Heart acoustic map for first (left) and second (right) heart sound (red points indicate microphone location)

Experiments are done on a human subject. He is asked to stop breathing at lung resting volume for 3 seconds in order to eliminate breathing sound and heart sound is recorded during this time. According to [3] sound speed in this physiologic condition is near 30 m/s and is closer to the propagation average speed which is 40 m/s. On the other hand overestimation of sound speed causes less artifact and deviation than underestimation [3]. So, experimental tests are done assuming 40 m/s as the propagation speed.

The signals are recorded and saved on a PC, they are segmented using the described method and beamforming is done. Finally a two or three dimensional image is presented on the screen. Figure 5 shows a two dimensional image of heart. It can be seen that heart sound is generated from different locations in different time intervals.

V. Conclusions

The results of this research show that a new imaging method can be introduced and might be helpful. Few clinical researches are done in this field, so there is a need for more investigations to indicate the abilities of this method in diagnosing heart diseases. The most important characteristic of this method is its noninvasive and nondestructive manner. So in future, we may see improvements in acoustic based instruments for heart disease diagnosis.

Some suggestions for improvement of this method are as follow: using more accurate microphones to improve the quality of recorded sound, a more precise estimation of the speed of sound in human chest and designing more complicated array arrangements to eliminate side lobes and to improve the focusing of the array.

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