MULTISENSOR STETHOSCOPE FOR CHEST SOUND LOCALIZATION

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Abstract— **This paper addresses the feasibility of using a multisensor system consisting of several stethoscope chest pieces for the localization of chest sounds. A signal processing algorithm called beamforming can combine multiple inputs to amplify sounds coming from a particular region inside the chest and attenuate all others. Beamforming can also be used for source localization; that is, determining from where a particular sound originated. These methods can provide a non-invasive and inexpensive acoustic image of the chest cavity for diagnostic and therapeutic purposes.**

This paper reviews methods for localizing a sound source using a microphone array. It describes a method comparing the delay between signals received at known microphone locations to determine the coordinates of the sound source. To illustrate this concept, we use a circular array of microphones in a free-field homogeneous medium and use delay-and-sum beamforming to combine the received signals. The challenges of applying this method to a non-homogeneous medium will also be discussed.

Keywords— **localization, microphone array, multisensor stethoscope, near-field beamforming**

INTRODUCTION

The traditional stethoscope used in practice today has changed relatively little in the past 200 years when compared to other technological and medical advances that have taken place in the same period of time. Although the stethoscope is a useful tool for non-invasive preliminary diagnosis, it is very limited in its capabilities. It is only able to detect sounds on the surface of the body and diagnoses are highly subjective and can vary greatly between physicians [1].

A multisensor system, consisting of several stethoscope chest pieces, enhances the usefulness of the traditional stethoscope and adds some new features. The physician will be able to listen to each channel separately, as with a traditional stethoscope, but since these sounds are recorded, the physician can listen to them as many times as necessary. It is also possible to slow down the playback, which can be helpful when listening to chest sounds of infants who have faster heart beats and respiration. Additionally, the physician can change stethoscope listening positions while listening to the same breath or heart beat, enabling him or her to hear the same sound at the same point in time, but from a different position. The physician can also amplify certain portions of the signal which may contain valuable diagnostic information but would often be missed with the traditional stethoscope.

In addition to sequential playback, signal processing algorithms, such as beamforming, can be used to combine the multi-sensor inputs. This allows physicians to hear signals below the surface of the skin by focusing the beam in a particular location. A possibility for future development is that the computer can analyze the sounds in the time and frequency domains and suggest a diagnosis based on a database of previous results.

Delay-and-Sum Beamforming

Beamforming is an algorithm applied to the signals from multiple microphones. Its goal is to amplify sounds coming from a selected location and attenuate sounds coming from all other locations. This is useful for noise cancellation and signal enhancement. One well known method is "delay-and-sum" in which the input signals are delayed to set a specified focus location and then summed. In free-field, the delay is based on time of arrival, whereas in a more realistic situation of a nonhomogeneous medium or with reflections, the delay would be the conjugate of the transfer function between the source and the microphones [2].

In near-field, the source is close to the microphones and the wave fronts are curved. In far-field, the source is far from the microphones and the wave fronts are planar (as seen in Figures 1 and 2) [2, 3]. Near-field beamforming yields a sound signal focused on a particular point in space whereas far-field beamforming can focus on a particular look-direction but not depth [2].

Far-Field Source Localization

Source localization is somewhat the reverse of the process for beamforming. Based on the differences in delays of the incoming signal at different microphones, one is able to determine from where the sound originated [2]. This is also how the human ears locate a sound source. Far-field localization assumes plane-wave sound propagation and therefore one can only determine the direction of the source but cannot discriminate depth (see Figure 1). The angle of the direction from which the sound originates is calculated as in Equation 1, below.

$$a = \cos^{-1}\left(\frac{delay}{separation}\right) \qquad (1)$$

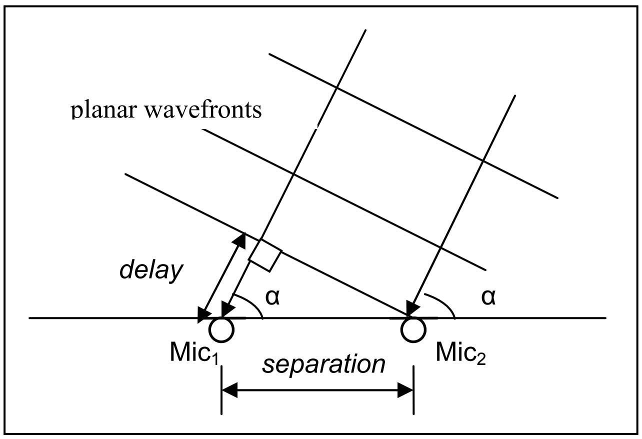

Figure 1. Far-field source localization

Near-Field Source Localization

Using the near-field assumption permits us to determine the direction as well as the depth of the sound source relative to the microphone positions (see Figure 2). Near-field is also more appropriate for measurements on the scale of the human chest. The estimated source location is calculated based on the difference in sound arrival time at each microphone [4, 5]. The delay is determined from the cross-correlation between the signals received at two microphones. Then, using the speed of sound, the delay is converted from sampled time units to distance units. Equation (2) uses the standard distance equation to represent $delta_{1\text{-}2}$, the difference between the distance from microphone 1 to the source and from microphone 2 to the source. Then, a system of two equations (Equations (2) and (3)) and two unknown (the coordinates of the source (x_s, y_s)) is constructed. In the equations below, (x_1, y_1), (x_2, y_2), and (x_3, y_3) are the coordinates of microphones 1, 2, and 3, respectively.

$$delta_{1\_2} = \sqrt{(x_1 - x_s)^2 + (y_1 - y_s)^2} - \sqrt{(x_2 - x_s)^2 + (y_2 - y_s)^2} \qquad (2)$$

$$delta_{1\_3} = \sqrt{(x_1 - x_s)^2 + (y_1 - y_s)^2} - \sqrt{(x_3 - x_s)^2 + (y_3 - y_s)^2} \qquad (3)$$

This procedure requires at least three microphones to find the source location. If more microphones are used, the average of all solutions is returned as the estimated source location.

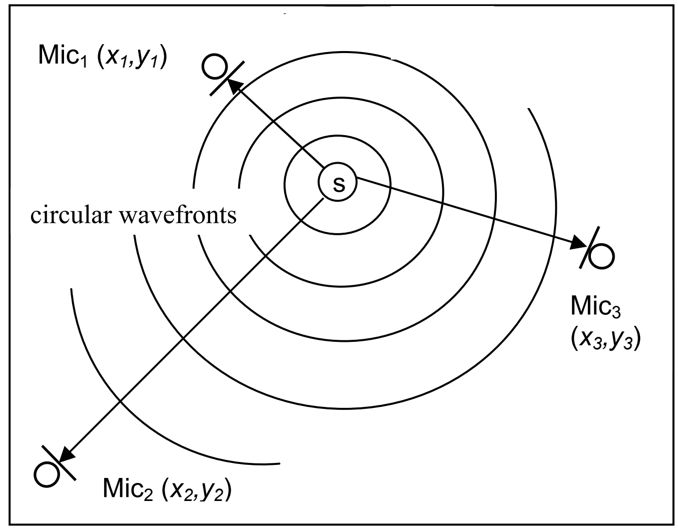

Figure 2. Near-field source localization, 's' indicates the sound source

METHODOLOGY

We are considering a two-dimensional circular microphone array. All of the microphones are arranged in one plane and are evenly spaced with the source located inside the array. This method of sound capture was also used in [6], but for the purposes of a meeting system.

The goal of the experiments was to determine how accurately we can calculate the coordinates of the sound source given the coordinates of the microphones, the signals received at the microphones, and the speed of sound in the transmission medium. We assumed a non-directional point source with near-field transmission in a homogeneous medium with no reflections.

Simulation

Matlab was used to conduct the simulations. The sound source used was a recording of normal heart sounds. Three microphones and the source were assigned specific coordinates. The microphone inputs were simulated by delaying the source signal as much as it would have been delayed had it been coming from the assigned position. Matlab performed near-field source localization, as described earlier, and returned the estimated source location (which should be the same as the location initially assigned to it).

In-Vitro Verification

A model was built to verify the simulation results. A sound source was placed in a plastic container lined with foam to reduce reflections. Four Littmann Master Classic II stethoscope chest pieces were placed on the outside of the container. The chest pieces were attached to Audio-Technica microphones using a short piece of rubber tubing. These were connected to an M-Audio Delta 1010 amplifier and digitizer. Data was collected and analyzed by Matlab and Simulink. Near-field source localization was performed using the actual microphone inputs (not in real-time) and the measured microphone locations.

RESULTS

Source localization error was considered to be the distance between the true source location and the estimated source location. In simulation, the localization error was due only to the sampling frequency reducing the resolution of the delay. Thus, in simulation, all the assumptions held and the results were excellent. However, in vitro, the sources of error were more numerous. There was error in the measurement of coordinates of the microphones and the source. Background noise contaminated the received signals and many of the assumptions did not allow sufficient modeling of the system. In verification, the average of the solutions became more useful as the systems of equations returned slightly different results. Results were satisfactory. The source localization was effective in certain locations but impossible in others. In those cases, the systems of equations were not solvable.

DISCUSSION

The proposed multi-sensor stethoscope for chest sound localization presents many challenges. The first is that the chest cavity is nonhomogeneous. It is composed of bone, air, blood, lung parenchyma, muscles, etc. all of which have different acoustic properties [4]. At the boundaries between tissues we will see a change in the speed of sound, signal distortion, dispersion, scattering, and refraction.

The speed of sound transmission in the chest is quite diverse. The chest wall and the heart transmit sound at about 1500 m/s. The large airways transmit at 222 to 312 m/s. The lung parenchyma transmits at between 23 and 60 m/s [4]. Variations in the speed of sound will affect the arrival time of the sounds and will also prevent the sound signals from traveling in a straight line from the source to the skin surface.

Normal heart and lung sounds are in the ranges of 20 to 150 Hz and 40 to 600 Hz, respectively. Adventitious sounds are present at frequencies up to 1200 Hz for the heart and 2000 Hz for the lungs [7, 8, 9]. Low frequencies pose a challenge for delay-and-sum beamformers which focus better at higher frequencies. It is also difficult to obtain sound capture systems with a flat response over this frequency range.

Another challenge is that it is very difficult to form a sound transmission model for the human body. There is great variation from person to person and even over time. A different model would be required for different sizes of males, females, adults, and children. This would also be affected by percentage body fat and disease affecting function or structure.

CONCLUSION

Future Work

The use of microphone arrays in nonhomogeneous media will be investigated in simulation and in vitro. Reflections, delays, interference, and asymmetry must be considered. Next, in vivo trials will be conducted with healthy and diseased persons. In the future, this kind of application could allow a physician to listen to sounds emanating from a particular area of the heart, such as a heart valve, to improve medical diagnosis.

Summary

This paper has presented a multisensor stethoscope for localization of chest sounds. It has reviewed some methods for localization in far-field and in near-field. Simulation and verification methods and results were summarized. Some challenges experienced in applying these methods to human trials were explained.

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