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Noise Reduction in Breath Sound Files Using Wavelet Transform Based Filter

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Abstract. The development of science and technology in the field of healthcare increasingly provides convenience in diagnosing respiratory system problem. Recording the breath sounds is one example of these developments. Breath sounds are recorded using a digital stethoscope, and then stored in a file with sound format. This breath sounds will be analyzed by health practitioners to diagnose the symptoms of disease or illness. However, the breath sounds is not free from interference signals. Therefore, noise filter or signal interference reduction system is required so that breath sounds component which contains information signal can be clarified. In this study, we designed a filter called a wavelet transform based filter. The filter that is designed in this study is using Daubechies wavelet with four wavelet transform coefficients. Based on the testing of the ten types of breath sounds data, the data is obtained in the largest SNRdB bronchial for 74.3685 decibels.

1. Introduction

To make diagnosis of a human respiratory system, health practitioner or doctor uses a device called a stethoscope. With it, medical practitioners can hear the sound of the lungs to diagnose a person's physical condition and health. Breath sounds heard through the stethoscope, is still mixed with other noises like the sound of the heart, skin friction with the stethoscope, and chest movements. It is difficult for health practitioners to diagnose a person's physical condition and health. In signal processing, other voices or extraneous noise is commonly called signal interference or noise.

The development of science and technology in the field of healthcare increasingly provide convenience in diagnosing respiratory system problem. Breath sounds recording is one example of these developments. Breath Sounds are recorded using a digital stethoscope, and then stored in a file with sound format. As with conventional methods, respiratory sounds are not free from interference signals. Therefore, filter noise or signal interference reduction is required so that breath sounds components that contain information signal can be clarified. Research on reducing signal interference had been done since the last few years, but has not been able to give a satisfying result.

Previous studies related to respiratory noise filter, including the separation of heart and lung sounds from the breath sounds with modified spectro-temporal representation [1], and reduction of heart sounds from lung sounds recordings using adaptive filter [2] have been carried out.

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Furthermore, researches on the wavelet transform including separation of discontinuous adventitious sounds from vesicular sounds using wavelet transform stationary-non-stationary filters [3], the separation splits the non-stationary input signals (heart sounds) from the stationary part (lung sounds) using a stationary-non-stationary wavelet transform filters [4], and the reduction of heart sounds from lung sounds have been recorded at low and medium flow rate using wavelet transform based filter [5] have also being done.

In this research, we applied wavelet transform based filter to reduce interference signals and white noise in the breath sounds recording, so we get a clearer signal without losing important information contained in the signal. With breath sounds recording filtered, health practitioner or doctor will get more detailed information for a better diagnosing method.The difficulty of diagnosing the disease through the breath sounds recording is that it is mixed with other noises which makes it difficult to diagnose a person's physical condition and health. A wavelet transform based filter is used with wavelet daubechies and global threshold in the breath sounds recording to produce a better breath sounds signal for diagnosis of diseases in the respiratory. In short, this work attempts to reduce noise so that the breath sounds recording can be used to provide a better diagnosis.

2. Methodology

2.1 Respiratory sounds

Respiratory sounds are all sounds related to respiration including breath sounds, adventitious sounds, cough sounds, snoring sounds, sneezing sounds, and sounds from the respiratory muscles. Voiced sounds during breathing are not included in respiratory sounds. [6].

Respiratory sounds can be classified into two groups: breath sounds and adventitious sound (abnormal) [7]. Breath sounds produced from healthy subject’s chest called normal breath sounds. Normal breath sounds include both the inspiration and expiration. Both occur when the air moves in and out during regular breathing cycle. Adventitious sounds are additional respiratory sounds in the breath sounds. This sound occurs unexpectedly during regular breathing cycle.

There are several types of abnormal breath sounds. The four most common types are rales, rhonchi, wheezing, and stridor [8]. Rales are small clicking, bubbling, or rattling sounds in the lungs. They are heard when a person breathes in (inhales). And are believed to occur when air opens in closed air spaces. Rales can be further described as moist, dry, fine, and course. Rhonchi are sounds that resemble snoring. They occur when air is blocked or air flow becomes rough through the large airways. Wheezing are high-pitched sounds produced by narrowed airways. They are most often heard when a person breathes out (exhales). Wheezing and other abnormal sounds can sometimes be heard without a stethoscope. Stridor is wheeze-like sound heard when a person breathes. Usually it is due to a blockage of airflow in the windpipe (trachea) or in the back of the throat. Fort the characteristics of Lung Sound and Noise, the peak sound of normal lung is usually found at frequencies below 100 Hz, where lung sound energy decreases sharply between 100-200 Hz, but can still be detected at or above 800 Hz with a sensitive tool [2]. According to Earis & Cheetham [9], noises such as the sound of the respiratory muscles, chest motion sounds, heart sounds, and other low-frequency noise, have a frequency between 50 to 150 Hz [9].

2.2 Wavelet transform

Fourier method specifies only the spectral content of a signal in the frequency domain. The disappearance of time information during Fourier transformation for preservation during the incident is not considered. This condition can be ignored if the signal is stationary. However, for stationary signals such as speech, time and frequency data is important to avoid significant loss of information in the signal. Wavelet analysis can be used as an alternative method to solve the problem in Fourier method [10]. By using the concept of multi resolution wavelet analysis (for example, the representation of time and frequency scaling) to produce precise decomposition of the signal to obtain an accurate representation, detailed characteristics such as small discontinuities, similarity, and even higher order derivation hidden by conventional Fourier analysis can be revealed.

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Wavelet is a family of functions $\psi_{a,b}(t)$ derived from a base wavelet $\psi(t)$, called the "mother wavelet", by dilation and translation [11], as shown in (1) as an example.

$$\psi_{a,b}(t) = \frac{1}{\sqrt{a}}\psi\left(\frac{t-b}{a}\right), a > 0, b \in \Re \quad (1)$$

Wavelet analysis is basically shifting and scaling a limited form of energy called the *mother wavelet* $\psi(t)$ of the desired signal. So, that the discrete wavelet transform can be written as in (2).

$$\psi_{j,k}(t) = 2^{j/2}\psi(2^j t - k) \quad (2)$$

2.3 Signal-to-noise ratio

Signal-to-noise ratio can be generally defined as the dimensionless ratio of signal power to the noise power contained in a record (3) [12].

$$SNR = \frac{P_{signal}}{P_{noise}} = \left(\frac{A_{signal}}{A_{noise}}\right)^2 \quad (3)$$

where

P_{signal} = mean of signal power
P_{noise} = mean of noise power
A_{signal}= *root mean square* (RMS) of signal amplitude
A_{noise} = *root mean square* (RMS) of noise amplitude

2.4 General architecture of the proposed method

Explanation of the components contained in the general architecture as shown in figure 2 is as follows:

a. Respiratory voice recording file is used as input. Respiratory sound is a combination of sound lungs and signal interference (noise).
b. Sounds can be played and printed out in the form of a signal.
c. Then respiratory sound is read by system and decomposed into an array of type byte and stored in the data [].
d. Array Data [] is converted into the form of an array of type double.
e. Array Data [] repeatedly decomposed according to the specified level of decomposition, produces two arrays, each array having half of the length of the data array []. The first array called a low pass filter and a second array called a high pass filter.
f. In each array, apply wavelet transform to the coefficients.
g. Both arrays are reassembled in the data array [] with a low pass filter placed in the first half and the high pass filter is placed at the end of the half.
h. Array Data [] is passed through a threshold, resulting in two arrays, array and the array of respiratory sound signal noise.
i. Perform repeatedly reconstruction as many as the level of reconstruction that has been assigned to each array.
j. Change the order in the data array [] of the previous half and half low pass filter high pass filter, be alternating low pass filter high-pass filter to each array.
k. Reapply the wavelet transform coefficients of each array.
l. Array Data [] and is then converted from an array of type double into an array of type byte. Audio formats and file name that has been set, is applied to the data [].
m. An array of data [signal] breathing sounds is rebuilt into a breathing sound file and the data array [] noise rebuilt into a beam of noise.

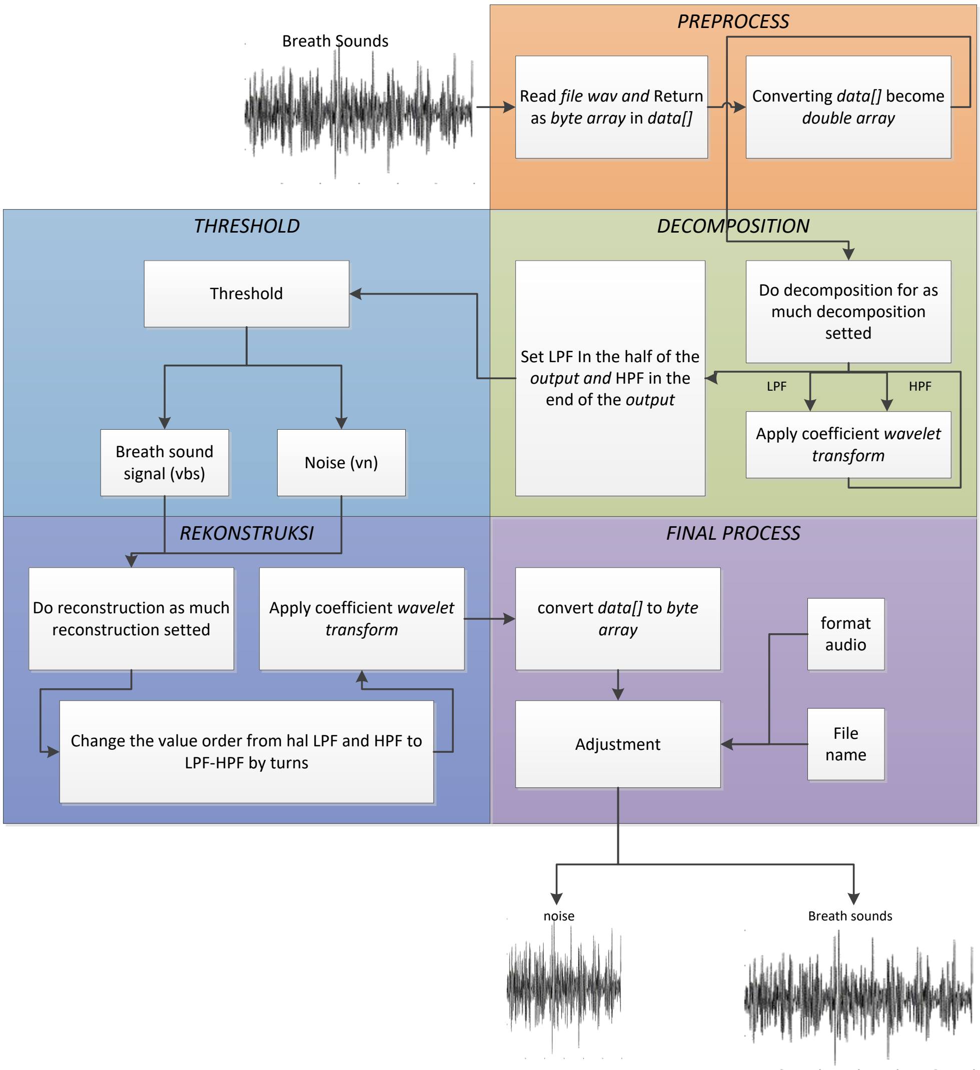

Figure 1. *General Architecture*

3 Experiments, Results and Discussion

3.1 Data

The data used in this research is breath sounds records, available from Littmann company. The existing data can be viewed as 10 different sounds with 10 different type of sounds. The data is described as: (*type, source*) = (bronchial, left lower lobe), (coarse crackles, right lower lobe), (fine crackles with deciduous bronchial, right middle lobe), (fine crackles, lung basis), (inspiratory stridor, tracea), (normal tracheal, tracea interscapular), (normal vesicular, right and left lower lobe), (pleural friction, right middle lobe), (rhonchus, right lower lobe), (wheezing, left lower lobe).

3.2 System analysis

Systems analysis aims to identify the system's development. Analysis is needed as a basis for system design. In this study, there is a five preprocess stage which is decomposition of the signal, the threshold stage, reconstruction stage, and the final stage process.

To test the designed system, a scenario is prepared to measure the sound of respiratory filter and the impact of the system with the objective and subjective criteria. Objective criteria is the signal-to-noise ratio (SNR) and signal display. While subjective criteria is the human auditory signal and the noise which is being observed.

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A breathing sound recording of data, ie, bronchial signal used in this work is shown in Figure 2.

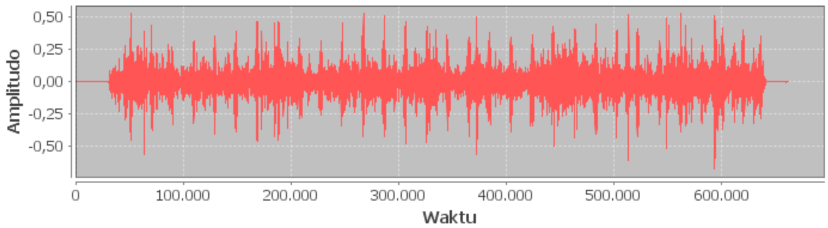

Figure 2. Bronchial sound signal before reduced

Measuring processes of noise reduction using wavelet transform based filter to the data in Figure 2 are as follow.

1. Change the data into a byte array type. After the array of type byte obtained, change again into the array of type double.
2. Perform second level decomposition of the data.
3. Perform the threshold stage using the following equation:

$$t = \sigma.\sqrt{2logN} \quad (4)$$

4. Perform level 2 reconstruction to the data.
5. Change the data back into the form of a byte array type.

Figure 3 and Figure 4 exhibits the filtered Bronchial signal and the noise of Bronchial signal of Figure 2, respectively.

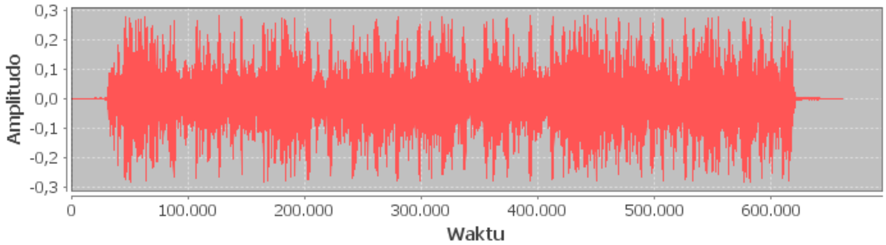

Figure 3. Bronchial signal after through filter

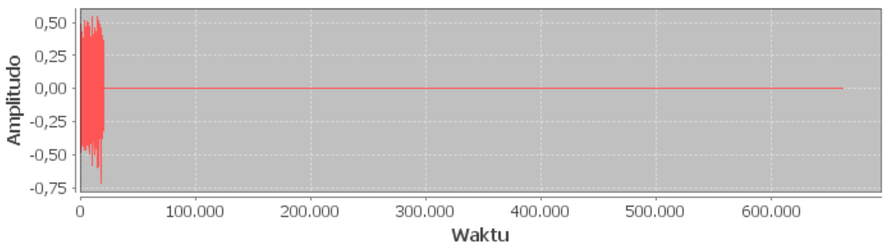

Figure 4. Noise signal after through filter

3.3 Results

Experiment is done to investigate the signal-to-noise ratio (SNR) of each data. The results are shown in Table 1.

Table 1. Noise Reduction Testing Results of Breath Sounds

No.	Type	SNR	SNR_{dB}
1.	Bronchial	2.7343069662486665E7	74.36847268923987
2.	Coarse Crackles	3703.999776899587	35.68670951851344
3.	Fine Crackles with Deciduous Bronchial Sound	1136888.1450540074	60.557177378843406
4.	Fine Crackles	333675.91665527434	55.23324862210189
5.	Inspiratory Stridor	223332.40934887607	53.48951751195278
6.	Normal Tracheal Sound	551447.7650258812	57.41504380900588
7.	Normal Vesicular Sound	15976.861637033	42.03491474128431
8.	Pleural Friction	4389520.347856363	66.42417066561016
9.	Rhonchus	294575.37479824974	54.69196438921298
10.	Wheezing	686282.2578168823	58.365027713927475

4 Conclusion

The experimental results showed that the biggest SNR_{dB} value is 74.36847268923987 decibel which is obtained from the reduction of bronchial. If the threshold value is too large, the noise cannot be reduced from the breath sounds signal. Conversely, if the threshold value is too small, breath sounds signal will loose a lot of important information.

As for our future work, we plan to utilize machine learning methods in order to increase the accuracy of the breath sounds analysis. In addition, we are not going to use the filters that have the level of decomposition and reconstruction greater than 2.

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