Heart Sound Reduction in Lung Sounds by Spectrogram

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Abstract: Adaptive Filtering is an accepted method to intelligently remove the heart sound from the lung sounds. However the drawback of the adaptive noise canceling scheme is the need of a reference signal that is exactly in the same time alignment as the interference signal in the primary signal. In this work an effective and easy method based on Spectrogram is presented to automatically generate a reference signal from the lung sound signal. Adaptive Noise Cancelation with Recursive Least Square (RLS-ANC) method is used to filter out the heart sound from lung sound.

Introduction

Although different techniques exist to detect the pulmonary disorders, lung sound analysis has drawn great attention because it does not require maximal breathing effort and can therefore be used with uncooperative patients. However, recorded lung sounds do not only comprise the sounds from the respiratory system but also from the heart and the inspiratory muscles [1]. Environmental noise and electrical noise can be eliminated to some extent by using a sound-proof room and/or an adequate sensor placement and inspiratory muscle noise is generally of a low frequency (< 20 Hz) and weak intensity [2]. Heart sound is the unavoidable source of noise that overlaps with lung sound components within the frequency band. The main frequency components of the heart sound are in the range of 20-150 Hz [3]. Adaptive-based approaches may be the most suitable methods to reduce intelligently the undesired heart sound [4, 5]. However, they have been partially successful, as their performance generally depends on accurate time alignment of the reference and primary signal.

Here, we propose a method to generate reference signal for adaptive noise canceler based on spectrogram of the noisy lung sounds. Proposed heart sound interference cancelation is accomplished in two steps: First step deals with the localization of the heart sounds by Spectrogram whereas second step deals with the interference cancelation by Recursive Least Squares Adaptive Noise Cancelation (RLC-ANC) adaptive filter.

Materials and Methods

In this work lung sound is used from the database [6]. Sound data were acquired with a piezoelectric contact accelerometer (Siemens EMT25C) at the trachea. The signal was filtered to remove DC with 8^{th} order Butterworth band-pass filter with pass band 7.5 Hz - 2500 Hz then digitized with 16 bit analog-to-digital converted (ADC). The original sampling rate was 11025 Hz.

A. Heart Sound Localization by Spectrogram

Time-frequency (TF) representations such as the spectrogram are specifically designed to process non-stationary signals as they jointly display time and frequency information demonstrating which frequencies occur at a certain time, or, at which times a certain frequency occurs. Spectrogram is computed by the windowed discrete-time Fourier transform of a signal using a sliding window and therefore sometimes called Short Time Fourier Transform (STFT)[3]. The function of the window is to extract a portion of the signal by ensuring that the extracted section is approximately stationary. The decrease of the window length increases the time resolution property of the spectrogram (wideband spectrogram) whereas the frequency resolution increases with an increase in window length (narrowband spectrogram). In this study short window length (25 ms-256 sample Hanning window) with 50 % overlap was used to calculate the Spectrogram. The reason of using short time period for spectral analysis is that high time resolution is beneficial to detect the heart sounds in the spectrogram.

Lung sounds are non-stationary signals in which properties (frequency, content, etc.) evolve over time. If the lung sound is examined with 3D spectrogram showing time, frequency, and amplitude, heart sound can be easily detected by its distinctive spiky energy spectrum at low frequencies (see Figure 1). High energy spikes of the heart sound signals can be localized by applying a threshold. In this work the threshold was chosen to be 1100 after the examining the spectrum of the signal at 50 Hz.

After the localization of the heart sound spikes, the input signal was then masked with the resulted signal which is composed of 127 data samples before and 128 data samples after the localized signal points, being one window length of the signal. Reference signal was constructed using these portions only with zeros in between. To subtract the lung sound frequency content from the reference signal, it was then filtered with 2^{nd} order Butterworth low-pass filter with cut-off frequency of 150 Hz.

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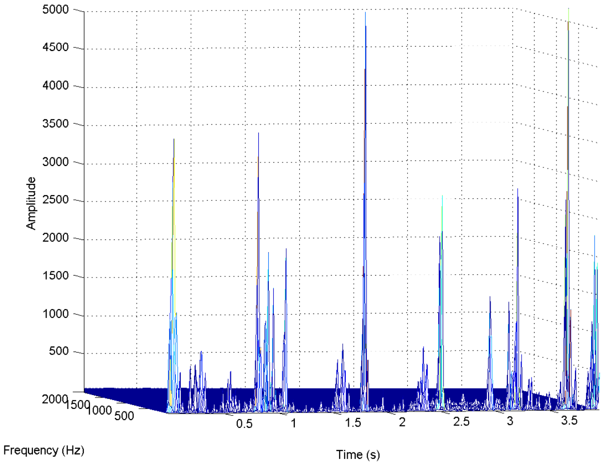

Figure 1: 3D Spectrogram of the original lung sound signal.

B. Heart Sound Filtering by RLS-ANC Adaptive Filter

In this study RLS-ANC has been applied for heart sound reduction. The standard RLS adaptive filtering scheme consists of a finite-duration impulse response transversal filter and RLS algorithm, which upgrades the tap weights w_k of the transversal filter in a recursive manner so that the cost function is minimized [7]. The RLS algorithm accepts two input vectors: a reference and a primary input. The primary signal, x(n), contains interference, m(n), alongside information bearing signal component, b(n). Reference signal, r(n), must be highly correlated with interference signal and is arranged by covariance method for data windowing. (U(n) covariance matrix is formed.)

The output of the RLS Filter, y(n), is as close to the interference component of the primary signal as possible.

$$y(n) = \sum_{k=0}^{M-1} w_k^* r(n-k) = \underline{w}^H(n)\, \underline{u}(n) \qquad (1)$$

where M is filter order, $\underline{w}^H(n)$ is the Hermitian transposition of the w_k, tap-weight vector of the FIR filter calculated for the current iteration n, and $\underline{u}(n)$ is the n^{th} column of the $U(n)$ covariance matrix of the reference input $r(n)$.

Output of the RLS-ANC Filter, $e(n)$, is the error term which is to be minimized in every iteration and also defined by:

$$e(n) = x(n) - y(n) = [b(n) + m(n)] - y(n) \qquad (2)$$

Substituting (1) in the (2) and by exploiting a relation in matrix algebra known as the matrix inversion lemma, algorithms of RLS-ANC is developed. In this study standard RLS-ANC algorithm was used. Output, e(n), provided the lung sound with a little or no portion of heart sound.

Table 1 shows the complete proposed algorithm including heart sound localization by Spectrogram and heart sound filtering by RLS-ANC adaptive filter. A *Matlab* code was developed to implement the algorithm.

As the RLS adaptive filter is highly sensitive to numerical instability [7], filter order severely affects the performance of the filter. In order to keep computational time as low as possible, RLS-ANC filter order was chosen to be eight on trial and error basis and λ was set to one to be infinite memory.

Table 1: PROPOSED ALGORITHM INCLUDING HEART SOUND LOCALIZATION BY SPECTROGRAM AND FILTERING BY RLS-ANC

Step	Description
1	Read the primary input lung sound data from wav file.
2	Calculate the PSD of the for lung sound and lung sound segments that are free of heart sound, using the last 30% of each ECG R-R interval and when the inspiratory flow is below 20% of its max level.
3	Calculate the spectrogram of the primary input.
4	At a low frequency band (i.e. at 50 Hz) of the spectrogram, automatically locate the first and second heart sounds.
5	Compose the signal by taking 127 data samples before and 128 data samples after the localized signal points.
6	Mask the data between these portions.
7	Filter by 2^{nd} order Butterworth filter with cut-off frequency of 150 Hz to remove remaining lung sound frequency from reference signal.
8	Apply standard RLS-ANC Algorithm with reference heart sound signal. $M = 8$, $\lambda = 1$.
9	Calculate the PSD of the output and compare the one calculated in step 2.

C. Power Spectral Density Calculations

Power Spectral Density (PSD) is estimated for the segments that are free of heart sound, using the last 30% of each ECG R-R interval and when the inspiratory flow is below 20% of its max level, for the original lung sound signal, and for the RLS-ANC filter output. Welch's method [8, 9] (or the periodogram method) for estimating PSD is carried out by dividing the time signal into blocks, and averaging squared-magnitude Discrete Fourier Transform (DFT) of the signal blocks. 1024 point Hamming window is applied to signal blocks before Fourier Transform. Equation of the PDS is expressed as below:

$$PSD(w) = \frac{1}{M_w} \sum_{m=0}^{M_w-1} |DFT_k(x_m)|^2 \qquad (3)$$

where M_w is the number of the windows constituting the whole signal and x_m is the windowed signal block.

Results

Fig. 2 shows amplitude of the spectrogram of the original lung sound signal and filtered lung sound signal at

50 Hz. 3D spectrogram of the filtered lung sound signal is shown in Fig. 3. By comparing Fig. 1 and Fig. 3 significant reduction of the heart sound spikes can be easily seen. PSD's of the original lung sound, original lung sound without heart sound, and RLS-ANC adaptive filter output are shown in Fig. 4. It is noticeable that PSD of the filtered lung sound signal is very closed to the PSD of the original lung sound signal without heart sound signal at low frequencies ($<$ 100 Hz) and becomes almost same as PSD of the original lung sound signal at high frequencies ($>$ 150 Hz).

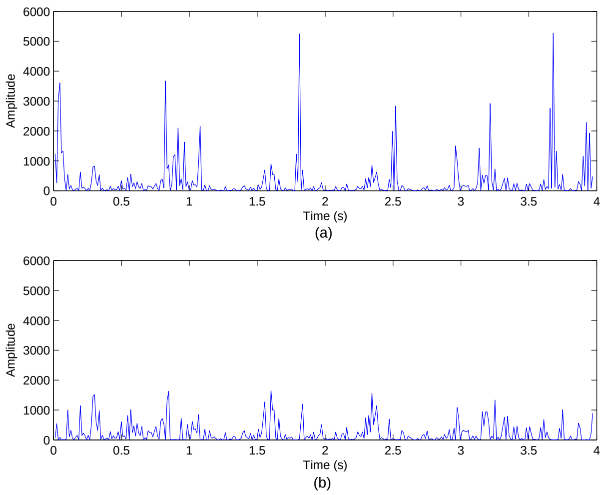

Figure 2: Spectrogram comparison of the original lung sound signal and filtered lung sound signal at 50 Hz.

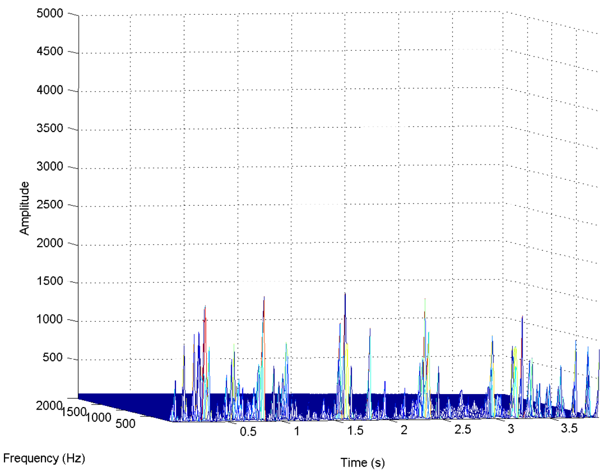

Figure 3: 3D Spectrogram of the filtered lung sound signal.

Discussion

Variety of methods including Spectrogram and RLC-ANC filtering have been proposed to identify the interferences in the lung sound and filter out the heart sound from lung sound without any loss of sound quality and spectral

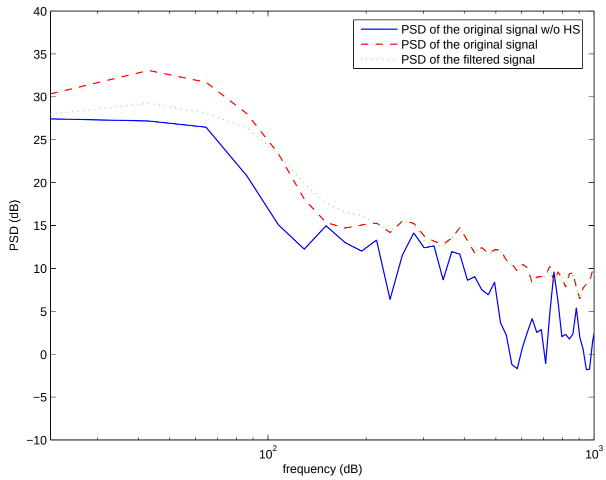

Figure 4: PSD comparison of the lung sound signals.

information [9]. In Reference [3] Adaptive Segmentation method was proposed in conjunction with RLS-ANC. The method is based on the idea of dividing the signal into locally stationary segments. However, nonstationary of the lung sound may not be resulted from only the heart sound interference and that assumption may cause the over estimation of the heart sound signal.

Spectrogram was also used for both in the detection and the cancelation of heart sound [10]. Although graphical representation of the Spectrogram provides the useful tool to localize the heart sound in the lung sound, graphically substraction of the heart sound signal from the lung sound signal may have other effects on the spectral content and therefore needs more attention. In our method the spectrogram was only used for the heart sound localization whereas lung sound signal processing was achieved with the nonstationary signal processing tool that is adaptive filtering. The nature of the lung sound is well suited for the adaptive filtering scheme and it was demonstrated and shown in Fig. 2 that RLS-ANC adaptive filter reduces the interference component significantly. Another advantage of the RLS-ANC filter is that it has no spectral effect on the signal. We observed that the RLS-ANC filter has no effect on the high frequencies of the lung sound signal. Signal change as in the Fig. 2 only occurred at intended frequencies that are between 50 Hz and 150 Hz. For the frequencies higher than 150 Hz. the signal remained the same.

Low-pass filter was used to deduce the high frequency lung sound signal from the low frequency heart sound signal to obtain reference signal. We observed significant signal corruption with the absent of the low-pass filter. The reason of unavoidable corruption can be explained with the RLS-ANC adaptive filter principles. The loss of the low-pass filter results in the reference signal containing both heart sound signal and lung sound signal. Therefore RLS-ANC adaptive filter also filters out the lung sound signal portions at the time segments where heart sounds present and considerable signal corruption

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is observed.

RLS-ANC adaptive filter order severely affected the filtering especially at the frequencies below 100 Hz. Filter order of eight was found to be the optimum in terms of both computational complexity and filtering. To examine the computational complexity and execution time, Normalized Least Mean Square Adaptive Noise Cancelation (NLMS-ANC) was also used instead of RLS-ANC for adaptive filtering the lung sound signal. NLMS-ANC is derived as an approximation to a gradient descent on a quadratic error surface of e(n), output of the adaptive filter [7]. In the NLMS-ANC algorithm, the correction that is applied in updating the old estimate of the w_k tap-weight vector is based on the instantaneous sample value of the tap-input vector, μ, step size parameter and the error signal. Step size parameter governs the convergence rate of the algorithm and was selected in a adaptive manner for optimum convergence rate. Spectral corruption of the signal was observed when NLMS-ANC was used. Compared to the RLS-ANC algorithm, the NLMS-ANC algorithm had the advantage of decreased computational complexity but this came at the cost of decreasing the convergence rate, which also raised the computational time.

Future work may include the comparative study of the more numerically stable adaptive filter algorithm with the RLS-ANC adaptive filter. Also faster algorithms with high convergence rate must be searched to take a step for the real time applications.

Conclusions

Adaptive Noise Cancelation with Recursive Least Square (RLS-ANC) method is used to filter out the heart sound signal from lung sound signal successfully and shown to be of better performance than the NLMS-ANC algorithm in terms of faster convergence and less spectral corruption. Superior performance of RLS-ANC adaptive filter, in terms of better signal spectral quality compared to NLMS-ANC was attained.

The need of the reference signal to the adaptive filter was overcome by the Spectrogram successfully. 3D visualization of the spectrogram reveals the heart sound signal locations, which can be easily derived from the whole original signal.

Low-pass filter, applied before the RLS-ANC adaptive filter to filter out the lung sound information on the reference signal was found to be necessary as the lung sound component on the reference signal may result in the decrease of spectral quality.

Finally, we proposed an efficient and easily implemented method to localize the hearth sound signals, to generate a reference signal and filter out the hearth sounds from the lung sounds using RLS-ANC adaptive filter. It was also demonstrated in this work that time-frequency representation is a powerful tool for the analysis and processing of the lung sounds.

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ISSN: 1727-1983