INNOVATION

Reduction of heart sound interference from lung sound signals using empirical mode decomposition technique

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During the recording time of lung sound (LS) signals from the chest wall of a subject, there is always heart sound (HS) signal interfering with it. This obscures the features of lung sound signals and creates confusion on pathological states, if any, of the lungs. A novel method based on empirical mode decomposition (EMD) technique is proposed in this paper for reducing the undesired heart sound interference from the desired lung sound signals. In this, the mixed signal is split into several components. Some of these components contain larger proportions of interfering signals like heart sound, environmental noise etc. and are filtered out. Experiments have been conducted on simulated and real-time recorded mixed signals of heart sound and lung sound. The proposed method is found to be superior in terms of time domain, frequency domain, and time–frequency domain representations and also in listening test performed by pulmonologist.

Keywords: Heart sound (HS), Lung sound (LS), Empirical mode decomposition (EMD), Power spectral density (PSD), Spectrogram

Introduction

Lung sounds are produced by vertical and disruptive flow [1] within lung airways during inspiration and expiration of air [2]. Heart sounds are created by the flow of blood into and out of the heart and by the movement of structures involved in the control of this flow [3] and the dominant frequency components are in the range of 20–150 Hz [4]. Heart sounds are clearly audible in lung sounds recorded on the anterior chest and may be heard to a lesser extent in lung sounds recorded over posterior lung lobes.

After the invention of stethoscope by Lannec [5] in 1816, it is employed as a primary tool to detect lung diseases, but traditional auscultation with a stethoscope constraints a diagnostic test mainly due to the interference of the heart sound with lung sounds. The heart sound signal interference obscures the interpretation of lung sounds and leads to an inaccurate diagnostic result of the lung diseases. It is highly desirable, for accurate assessment of the lung diseases based on lung sound information, to remove the heart sound interference. The secondary drawback of the auscultation technique is the limitation of the human auditory system [6]. A less experienced physician may find it more difficult to interpret LS in presence of HS. However, this problem may be solved by the skilled and experienced physician. Apart from the Heart sound interference, LS signals are affected by noises generated from surrounding environment and friction rubs. The environment and friction rubs noise are reduced by using a sound proof room and with proper placement of microphone. The methods used for HS separation from LS signal remained an area of interest among researchers. Methods like adaptive filtering [4],[7]–[9], wavelet-based filtering [10–12], and short time Fourier transform (STFT) based approach [13] have been tried out with some success. In this paper, we propose a new method based on EMD technique [14], for separation of HS signals from LS signals. The EMD technique decomposes the signal into a number of different time scales or intrinsic mode functions (IMFs). This allows us to filter signal components individually instead of filtering the original signal. It is found that some components of the signal highlight the presence of interference or noise present in the LS signal. Thus it becomes easier to remove them from the signal using the EMD technique [15]. The proposed method has the ability to remove efficiently the HS interference from LS signals without any requirement of the reference signal unlike [7]. The idea introduced by S. Ari et al. [16] was employed in our experiment for detection of the HS peaks within the LS signals. This is followed by removal of HS signal and estimation of corresponding sample values.

This paper is organized as follows: A brief description of the empirical mode decomposition (EMD) technique is presented in EMD technique section. Data acquisition section discusses about the database used in this study, Methodology section presents the proposed method. Experimental outcomes and its discussion is presented in Results and discussion section followed by conclusions.

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EMD technique

EMD technique has been introduced by Huang et al. (1998), for nonlinear and nonstationary data analysis. This technique offers a great advantage that is the basis functions are derived directly from the signal itself. Hence, the functions are varying in nature in contrast to fixed basis functions of conventional methods. The purpose of EMD technique is to represent a signal as a sum of oscillatory functions, called IMFs, starting from high-frequency mode IMFs to low-frequency mode IMFs. According to Huang et al. [17], an IMF satisfies two conditions: (1) the functions should be symmetric in time, and the number of extrema and zero-crossing must be equal, or differ at most by one; (2) the mean value of the envelope, defined by the local maxima and envelope defined by the local minima must be zero at any function point. This means that the IMF is obtained by locally eliminating the superposition of different frequency and amplitude waves, and eliminating signal asymmetries with respect to the zero level. This is done by using the EMD technique, which decomposes the signal into IMFs with an iterative procedure consisting of extrema identification and sifting steps. The procedure of computing IMFs from a given signal $x(t)$ is as follows:

1. Identify all extrema of $x(t)$ based on the sign changes across the derivative of $x(t)$,

2. Compute the lower and upper envelopes. The lower and upper envelopes are produced using the local minima and maxima with the help of cubic spline interpolation method, and they designated as $e_{\text{min}}(t)$ and $e_{\text{max}}(t)$, respectively,

3. Compute the average of upper and lower envelopes, $m(t)$

$$ m(t) = \frac{e_{\text{max}}(t) + e_{\text{min}}(t)}{2} $$

4. Calculate the difference between the data $x(t)$ and the mean $m(t)$ and designate it as $h_1(t)$,

$$ h_1(t) = x(t) - m(t) $$

5. $h_1(t)$ is probably not an IMF. To correct this, the sifting process has to be repeated as many times as is required to reduce the extracted signal as an IMF,

6. Once the first set of “siftings” results in an IMF, define

$$ c_1(t) = h_k(t) $$

7. where $k$ is the number of sifts.

8. Obtain the residue, $r_1(t)$

$$ r_1(t) = x(t) - c_1(t) $$

9. Treat $r_1(t)$ as a new set of data and repeat steps 1–7 up to $N$ times until the residue $r_N(t)$ becomes a function from which no more IMFs can be extracted,

![Figure 1](image.png)

Figure 1. Signal $x(t)$ on the top and its fourteen IMF components (IMF$_1$ to IMF$_{14}$) and the residual (IMF$_{15}$ on the bottom) obtained by EMD method from signal $x(t)$. 
Reduction of heart sound interference

Finally, EMD technique may be understood as a step-by-step taking out of the locally highest frequency oscillation of the signal, increasingly forming low-pass intrinsic mode components (figure 1).

Data acquisition

The impure LS signals were recorded from the chest wall of 15 individuals at Audio and Biosignal Processing Laboratory, IIT Kharagpur and 27 subjects from the Institute of Pulmocare and Research center, Kolkata, India. The LS data were recorded by using a data acquisition system, as shown in figure 2. The environmental and friction rub noises are reduced by choosing a proper recording condition, selecting a suitable microphone and properly placing the stethoscope over the body surface of the subject. Hence, HS interference is present in the LS recordings. The recorded LS signals were stored as *.wav files in 16 bit, PCM, Mono audio format at sampling frequency of 8 kHz. The HS signals were collected from Maulana Azad medical Institute, Delhi, India and the LS signals without HS interference were obtained from the R.A.L.E. data sets available at: www.rate.cal. The synthesized data employed in our experiment are produced by mixing up the HS signal with the pure LS signals at different ratios.

Methodology

The basic principle of the proposed HS interference reduction method presented here is the extraction of HS corrupted segments from the IMFs which are generated from the impure LS signal $x(n)$ with the help of EMD technique. The biomedical domain feature based method has been proposed by S. Ari et al. [16] is employed in this paper for the detection of the location of HS peaks within the noisy LS signals. The HS peaks are bounded and extracted from every IMF. The proposed method comprises of five parts: an amplitude normalization scheme, an IMF generation scheme, a HS localization scheme, a boundary estimation scheme, and a LS reconstruction scheme. The whole process is depicted in figure 4. The HS peaks are localized and bounded from the LS signals contaminated with HS signal first by employing a suitable HS localization scheme and also a boundary estimation scheme. After the boundary estimation of HS peaks, the bounded HS corrupted segments are extracted from the all IMFs of the mixed signal of LS and HS. As a result, the IMFs of the mixed signal are divided into two parts: HS-contaminated portion of IMF and HS-free portion of IMF that contains only LS signal information. The high pass filter then removes the HS frequency components from the HS corrupted segments of IMF. Finally, the unaltered HS-free portions of IMF are added to the output of the high pass filter to reconstruct the HS-free LS signals.

Amplitude normalization

Let the input digital LS signal interfered by HS signal is denoted by $x(n)$ and let its sampling frequency is $f_s$. If the duration of the signal is $t$ seconds, the total number of
samples \((N)\) in the signal is given by \((t \times f_s + 1)\). The peak-to-peak amplitude of the mixed signal of LS and HS, varies with the amplifier gain, physiology, age and sex of the subject. So, the signal is first normalized to the absolute maximum of the signal as:

\[
[x(n)]_{\text{Normalized}} = \frac{x(n)}{[x(n)]_{\text{max}}}
\]  

(1)

Where, \(n = 1, 2, \ldots, N\) and \([x(n)]_{\text{Normalized}}\) is the normalized signal. This normalized signal lying in the \(\pm 1\) range in amplitude, is presented as input to the next block, IMF generation unit.

**Generation of IMFs**

The EMD technique can decompose the signal into a number of different time scales or IMFs. This means that it is also possible to examine signal components individually instead of the original signal. The steps are required for generation of IMFs have been discussed in EMD technique section. The EMD technique is employed for decomposing the input mixed signal of LS and HS into IMFs, IMF\(_1\), IMF\(_2\), \ldots, IMF\(_m\), where \(m\) is the total number of IMFs as shown in figure 1.

**Detection of peak locations of HS signal**

The peak location identification of HS signal is one of the important steps for reducing HS interference from LS signals. There are many algorithms available for the localization of HS peaks in the noisy LS signals [18]. In our work, a biomedical domain feature based method proposed by S. Ari et al. [16] is used for detection of HS peak position within the LS signals.

Summary of the HS peak detection algorithm.

**Algorithm 1: Peak detection algorithm**

- Step 1: Normalize the mixed signal of LS and HS.
- Step 2: Calculate HS cycle duration.
- Step 3: Filter the mixed signal using a low-pass filter with cut-off frequency 150 Hz.
- Step 4: Calculate energy.
- Step 5: Picking up the significant energy peaks.
- Step 6: Locating S1 and S2 (first and second HS peaks).

Summary of the boundary estimation algorithm.

Figure 3. Boundary estimation of primary heart sounds.

Figure 4. Block diagram of the proposed method showing the sequence of steps required for the reconstruction of the pure LS signal.
Algorithm 2: Boundary estimation algorithm

Step 1: Take the mixed signal that contains both the HS and LS signals.

Step 2: Detect the peaks of HS signal.

Step 3: Calculate the number of samples both in forward and backward directions for each peak
   (a) Calculate the number of samples for $T = 50$ ms time in the backward direction ($SN_b$).
   \[ SN_b = T \times f_s \]  

   (b) Calculate the number of samples for $T_1 = 100$ ms time in the forward direction ($SN_f$).
   \[ SN_f = T_1 \times f_s \]  

Step 4: Estimate the boundary (BB) of each peak

   \[ BB = \left\{ P(k) - SN_b \right\} + \left\{ P(k) + SN_f \right\} \]  

where $f_s$ is the sampling frequency of the mixed signal.

4.4. Boundary Estimation of HS peaks

The next step of the proposed method is to estimate the boundary of the HS corrupted segment which is undesirable for diagnosing of LS signals. The HS signals are produced due to the blood circulation through the heart. Hence, the HS signal is extended to a certain range in both directions of its peak. Basically HS peak is extended in forward and backward directions by 100 ms and 50 ms [19], respectively as shown in figure 3. The boundary of HS peak is calculated under the consideration of its extensions in both directions.

Reconstruction of LS signal

This is the final step of the proposed method. In this step, a number of functions are executed. The output of this stage is a pure reconstructed LS signal without HS interference. The input of this block is the output of the boundary estimation scheme. The subsequent steps of the LS reconstruction scheme are following:

Step 1: Find the HS-free (HF) segments of all the IMFs by extracting the HS-included (HI) segments from them. Each IMF having several HS-free and HS-included segments, respectively. The HS-included segments contain both the HS signal as well as LS signal information. However HS-free segments contain only the LS signal information. Hence, the differences between the IMFs and HS-included segments give heart sound free segments associated with several gaps created for the extraction of HS-included segments.
\[ HF_i(n) = IMF_i(n) - HI_i(n) \]  

where \( i \) indicates the particular IMF, \( i = 1, 2, \ldots, m \), HI\(_i\)\((n)\) stands for HS-included portion and HF\(_i\)\((n)\) for HS-free portion of IMF\(_i\)\((n)\).

Step 2: Compute the resultant HS corrupted portion (H) by summing up linearly the HS-included segments in all IMFs.

\[ H(n) = \sum_{i=1}^{m} HI_i(n) \]  

Step 3: Filter the output H of step 2 using a 10th order high pass Butterworth filter having a cut-off frequency (\( f_c \)) of 150 Hz.

Step 4: Find the resultant HS-free LS portion (L) for all IMFs by combining linearly the outputs of step 1.

\[ L(n) = \sum_{i=1}^{m} HF_i(n) \]  

Step 5: Add up the output, \( H \), of high pass filter to the output, \( L \), of step 4, and as a result obtaining a pure reconstructed LS signal without HS interference. The reconstructed LS signals contain information that help the doctors or machine intelligence system for taking a correct decision regarding the lung diseases.

**Results and discussion**

**Qualitative analysis of the results**

**Auditory test**

A subjective test by experienced and skilled pulmonologist has been performed on the reconstructed LS signals without HS interference. Listening to the reconstructed LS signals confirmed that the HS interference is completely removed. Hence, the proposed method is able to produce HS-free LS signals with high-sound quality.

**Visual inspection of the mixed and reconstructed lung sound signals**

(a) Time domain comparison of the results

Figure 5(a,b) depicts the LS signals free from HS interference and HS signal. These signals were employed in producing the simulated mixed signal at different ratios of LS and HS. Experimental results of the proposed method are shown in

![Figure 6](image-url)
Figures 6(a–e) and 11(a–c), where mixed signals, separated HS interference, and reconstructed pure LS signals are illustrated, for simulated data and real time recorded data, respectively. These figures show that the heart peak locations in mixed signal and in separated heart sound are synchronized. By comparing between the reconstructed LS signals and original LS signals, it is evident that both are same in structure and amplitude level.

(b) Frequency domain comparison of the results

Besides the time domain inspection, the performance of our proposed method was verified by comparing the power spectral densities (PSDs) of the LS signals with and without HS interference with that of the reconstructed LS signal. For an effective HS removal method, the PSD of the reconstructed LS signal should be coincide or be close to that of the original HS-free LS signal. Figures 8–10 show the PSDs of the mixed signals, reconstructed...
LS signals obtained by the proposed method and original LS signal, respectively for three different cases. These figures show that PSD of the reconstructed LS signal closer to the PSD of the original LS signal free of HS interference. The PSD comparisons corroborate the proposed method successfully remove the HS interference from LS signal.

(c) Time-frequency domain or spectrogram comparison of the results:
The performance of our proposed method was also evaluated by comparing the spectrogram of noisy LS signals and reconstructed pure LS signals. figure 7(a–e) illustrates spectrogram of mixed signal, reconstructed LS signals and original LS signals for simulated data. figure 11(d,e) depicts the spectrogram of mixed signal and reconstructed LS signals for real-time recorded data. By visually inspecting the spectrograms of the signals, it is evident that our proposed method can efficiently reduces the HS interference from the desired LS signals.

Quantitative analysis of the results:
The performance of the proposed method was analyzed quantitatively. The quantitative analysis of the two types of data was done by means of local heart noise or interference reduction percentage (LH-NRP) and overall heart noise or interference reduction percentage (O-HNRP) calculation as suggested by Hadjileontiadis and Panas [4]. The heart noise reduction percentage (HNR) unit is defined by

$$\text{HNRP} = \frac{E[M^2(n)] - E[F^2(n)]}{E[M^2(n)]} \times 100$$

where $M(n)$ stands for mixed signal and $F(n)$ stands for reconstructed LS signals. In order to perform quantitative comparison between the proposed method and wavelet-based method [10], it was implemented and applied to our database. Results obtained using the proposed method
and wavelet-based method are presented in Table 1 and Table 2 respectively. The results of the proposed method and existing wavelet-based method are similar for simulated data containing 20% LS and 80% HS signals. On the other hand existing method gives worst performance for mixed signal in which 20% HS and 80% LS are present. The proposed method gives better result than existing method for all cases. The wavelet-based method considers some assumptions firstly: HS peak has large amplitude over many wavelet scales while LS signals die out swiftly with increasing scale, secondly: HS signals are non-stationary and LS signals are stationary in nature. These considerations are not valid for all cases. Our

Table 1. Experimental results for simulated signal.

<table>
<thead>
<tr>
<th>Data</th>
<th>Wavelet-based method</th>
<th>Proposed-EMD based method</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>L-HNRP</td>
<td>O-HNRP</td>
</tr>
<tr>
<td>Simulated signals</td>
<td></td>
<td></td>
</tr>
<tr>
<td>80% + 20%</td>
<td>78.52%</td>
<td>37.10%</td>
</tr>
<tr>
<td>50% + 50%</td>
<td>98.65%</td>
<td>88.97%</td>
</tr>
<tr>
<td>20% + 80%</td>
<td>99.70%</td>
<td>98.90%</td>
</tr>
</tbody>
</table>

EMD, empirical mode decomposition; HS, heart sound; LS, lung sound; L-HNRP, local heart noise or interference reduction percentage; O-HNRP, overall heart noise or interference reduction percentage.

Table 2. Experimental results for real-time recorded data.

<table>
<thead>
<tr>
<th>Data</th>
<th>Wavelet-based method</th>
<th>Proposed-EMD based method</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>L-HNRP</td>
<td>O-HNRP</td>
</tr>
<tr>
<td>Real-time recorded</td>
<td></td>
<td></td>
</tr>
<tr>
<td>signals</td>
<td></td>
<td></td>
</tr>
<tr>
<td>S1(M)</td>
<td>83.74%</td>
<td>73.76%</td>
</tr>
<tr>
<td>S2(M)</td>
<td>70.84%</td>
<td>67.27%</td>
</tr>
<tr>
<td>S3(M)</td>
<td>70.33%</td>
<td>54.33%</td>
</tr>
<tr>
<td>S4(M)</td>
<td>88.00%</td>
<td>83.19%</td>
</tr>
<tr>
<td>S5(M)</td>
<td>89.38%</td>
<td>78.69%</td>
</tr>
</tbody>
</table>

EMD, empirical mode decomposition; HS, heart sound; LS, lung sound; L-HNRP, local heart noise or interference reduction percentage; O-HNRP, overall heart noise or interference reduction percentage.
method does not make any assumption like wavelet-based method. In addition, wavelet-based method uses a fixed type of basis function, but EMD technique uses basis functions which are generated from signal and adaptive in nature.

**Conclusions**

In this paper, we have proposed a novel method for reducing HS interference from LS signal based on the EMD technique. This method provides a number of advantages such as: it does not depend on the prior information about the data to be analyzed. It is possible to analyze in more detail about the signal components in terms of IMFs instead of the original signal. It is found that some components of signal highlight the noise or HS interference thus making it easy to remove their presence from the desired LS signal. The performance of our proposed method was verified by both the subjective listening test by the skilled and experienced pulmonologist, as well as the quantitative analysis of the reconstructed LS signal. The results of the analysis of the synthetic and real-time recorded mixed signal show that the proposed method has ability to remove efficiently the HS interference, without any degradation of the quality of the reconstructed LS signal. This proposed method may be incorporated into a lung sound analysis system to enhance quality of lung sound for diagnosing the respiratory diseases.

**References**