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Hybrid Nelder-Mead search based optimal Least Mean Square algorithms for heart and lung sound separation

Ruban Nersisson *, Mathew M. Noel

School of Electrical Engineering, VIT University, Vellore, India

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Algorithms for separation of heart sounds from background lung sound noises are vital for accurate diagnosis of heart diseases. In this paper, an improved adaptive noise cancellation technique based on the Least Mean Square (LMS) algorithm is used to separate heart sounds from lung sounds. The step size parameter in the LMS algorithm is optimally chosen using a hybrid Nelder-Mead (H-NM) optimization algorithm. The NM algorithm is initialized with a good initial solution by using computationally cheap random search to compute a rough estimate of the global minimum. Initialization of the NM algorithm with a good initial guess avoided convergence to shallow local minima and improved the quality of the final solution. The effects of using two state-of-the-arts biologically inspired heuristic optimization algorithms instead of the H-NM algorithm and three variants of the standard LMS algorithm are investigated. The correlation coefficient between the ideal and filtered heart sound signal and running time-complexity of different algorithms are taken as the metric for comparison of different heart sound separation approaches. Simulation results indicate that the approach presented in this paper performs significantly better than a variety of alternate approaches on heart sound separation problems.

1. Introduction

Methods for effective separation of the Heart Sound Signal (HSS) from background lung sound noise are of great importance in the diagnosis of cardiac diseases. The separated noise free HSS is used in real time diagnostic applications like feature segmentation and analysis, the study of a second heart sound (S2) split [1] and sleep parameter assessment [2]. The separated lung sound is also used as an indicative tool for anesthetic management during surgical procedures [3,4]. In this paper, a novel optimal Least Mean Square (LMS) algorithm based approach for accurate separation of the heart sound is proposed and compared with a variety of existing approaches.

Auscultation refers to the action of listening to the sounds produced by internal organs traditionally with a stethoscope [5]. Physicians use auscultation as a non-invasive method to get functional information relating to internal organs like the heart, the lung, and the gastrointestinal system. In auscultation of the heart, besides the sounds produced from the flow of blood into and out of the heart, and the breath sounds, there are artifacts in the form of murmurs, gallops, and environmental noises. The HSS is the sound produced by the flow of blood, in and out of the cardiac structure and the movement of the cardiac structure itself. The HSS is basically composed of two major sounds S1 and S2. S1 is caused by ventricular contraction during the closure of the atrioventricular valves. S1 is the longest and loudest of the heart sounds. S2 is due to the closure of the semilunar valves at the end of ventricular systole. Lung Sound Signal (LSS) is produced by turbulent air flow during respiration. Major frequency components of the LSS lie in the range of 20 to 100 Hz [5,6]. This is also the range in which the HSS has its main frequency components [7]. The spectral overlap of the heart and Lung sounds makes the HSS separation problem challenging. Moreover, HSS and LSS are random signals and can suffer unexpected fluctuations, and also due to the spectral overlap, the separation of the two signals cannot be performed using any non-adaptive or time invariant linear filter. So, the filter used should be able to adapt with such inconsistencies. The word “adapt” means to adjust the filter coefficients to cope with the fluctuations of the input signals [8]. Adaptive filters have a self-learning ability where as traditional digital filters do not have [9].

Yang-sheng Lu et al. [7] used adaptive filters for accurate separation of heart and lung sounds. Hans pasterkamp et al. [10] discussed the problem of recording LSS using a stethoscope and also

* Corresponding author.
E-mail address: nruban@vit.ac.in (R. Nersisson).

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suggested active noise cancellation techniques for operation in noisy environments. Yip et al. [11] proved the reduction of heart sound artifacts using the adaptive noise cancellation algorithms with automatic gain control technique using experimental results. Thato Tsalaile et al. [12] considered the separation of HSS from colored noise using the Adaptive Line Enhancement (ALE) LMS algorithm. This paper also addresses the issues relevant to the optimal selection of ALE algorithm parameters. M. T Pourazad et al. [13] proposed a time-frequency filtering technique for separation of HSS and LSS. January Gnitecki et al. [14] presented a detailed study of various adaptive noise cancellation algorithms for HSS and LSS separation and indicated the importance of properly selecting adaptive filter parameters such as filter order and the convergence rates. Foad Ghaderi et al. [15] proposed a separation approach based on the singular spectral analysis. Muhammad Sukrisno Mardiyanto et al. [16] analyzed the frequency spectrum of the LSS for diagnostic applications. Ruban et al. [17] reviewed a variety of algorithms for HSS separation and concluded that adaptive filters with some modification in the step size could improve the quality of the separated signals. Mostafa Guda et al. [18] explored a variety of LMS algorithm improvements for denoising Electro Cardio Graph (ECG) signals. An adaptive noise cancellation technique, where the step size is updating based on the power of the input signal is reported by Yüksel Özbay et al. [19].

This paper is organized as follows: firstly HSS, LSS, mixed signal and adaptive noise cancellation based schemes for separation are discussed, secondly, an improved adaptive noise cancellation scheme where the step size is optimally chosen using a hybrid Nelder-Mead algorithm is proposed and finally the proposed approach is compared with a variety of alternate approaches.

2. Heart and lung sounds

2.1. Recording sounds produced by internal organs

HSS is recorded with electronic stethoscopes and suitable data acquisition systems. HSS is usually digitally stored in .mp3 or .wav formats [10]. The prime location for the HSS recording is right and left sternal margin between second and fifth intercostal spaces.

The lung sound auscultation is mostly done on upper anterior region of the chest, mid axillary region and on the posterior basal side [20]. The HSS is recorded near the mid-axillary line to minimize LSS noise.

2.2. Heart sounds

The heart sound has multiple components such as first heart sound (S1), second heart sound (S2), third heart sound (S3) and murmurs Fig. 1.

2.3. Lung sounds

Breathing consists of two phases – inspiration and expiration. Lung sounds are created when air moves through the airways (trachea and bronchi). The nature of heart and lung sounds is determined by the movement of the body structures. These sounds can be classified as tracheal, bronchial, broncho-vesicular, vesicular and adventitious sounds [20]. The types of lung sounds considered in this paper are described in Fig. 2.

a. **Bronchial sounds:** The bronchial sounds are mainly present and detected over the large airways in the anterior chest near the second and third intercostal spaces and are thus heard above the sternum. So these sounds mostly overlap with the HSS. These sounds are not as harsh and coarse as tracheal breath sounds but are loud and high in pitch.
b. **Vesicular breath sounds:** These sounds are heard over most of the lung region. There is a significant overlap between the vesicular breath sound and HSS. These sounds are high pitched in the inspiration cycle and low pitched in the expiration cycle without a gap between inspiration and expiration cycles [20].
c. **Adventitious lung sounds:** These sounds include crackles, pleural sounds and wheezes. Wheezing is the major sound present with lung sounds of patients suffering from breathing related problems, so the breath sound recorded with wheezing is also considered as one of the noise signal.

In this paper four different corrupted signals are used to test the performance of different HSS separation algorithms. Fig. 3 shows the heart sounds contaminated with different lung sound noises.

3. Methodology

In the following section, the standard LMS algorithm and its popular improved variants are reviewed. The design procedure for filter parameters and the values are presented.

3.1. Adaptive algorithms

The process of active noise cancellation uses an adaptive Finite Impulse Response (FIR) filter. The filtering is performed in two parts – the adaptive algorithm and the digital filter.

3.1.1. The LMS algorithm

The LMS algorithm [21,22] is used to adapt the coefficients of a FIR (Finite Impulse Response) filter based on a suitably defined error signal to achieve noise separation from the input signals. To obtain the pure heart sound output y(n) from the noisy input, an estimate of the noise (lung sound) is computed using an adaptive FIR filter. The LMS filter coefficient update rule is given in Eq. (1).

$$w(n+1) = w(n) + 2\mu e(n)x(n) \tag{1}$$

$$y(n) = w(n).x^T(n) \tag{2}$$

The error signal e(n) is defined as follows:

$$e(n) = d(n) - y(n) \tag{3}$$

where

x(n) → contaminated heart sound signal
w(n) → vector of filter coefficients
y(n) → filtered heart sound
d(n) → desired signal
μ → step size

The convergence rate of the LMS algorithm depends critically on the step size parameter μ [23]. The overall scheme for separation of the heart sound signal using the LMS algorithm is shown in Fig. 4.

3.1.2. Normalized LMS

In the conventional LMS algorithm, the noise level varies based on the value of the step size (μ), since the step size is calculated by the Eigen value of the input vector. To solve this problem, another approach is used in which the step size is calculated by the autocorrelation of the input vector [24]. The filter coefficient vector w(n) is normalized [24–27] based on the input vector in each iteration.

The step size is given by [24];

$$\mu(n) = \mu_0 \left[\frac{1}{N} \sum_{i=0}^{N-1} x^2(n-i) \right]^{-1} = \frac{N\mu_0}{X^T(n).X(n)} \tag{4}$$

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