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Automatic segmentation of Phonocardiogram using the occurrence of the cardiac events

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ABSTRACT

Objective: This paper presents automatic method of segmentation of heart sound using the occurrence of the cardiac rhythmic events.
Methods: Noisy heart sound is filtered using the 6th order Chebyshev type I low pass filter to remove the redundant noise. Bark Spectrogram is calculated from the cardiac signal by converting spectrogram to the bark scale. The bark spectrogram is smoothened and the loudness index is calculated by averaging the amplitude across all frequency bands. The loudness index is smoothened and differentiated to obtain the event detection function. The smoothened event detection function gives the occurrence of the cardiac events namely the first and the second heart sounds.
Conclusion: This method is highly effective in identifying peaks S1 and S2 with the segmentation accuracy of 96.98% giving an F1 measure of 97.09%.
Significance: This method does not require the setting up of any type of threshold. So it is a highly effective type of segmentation under noisy conditions.

1. Introduction

[1]Phonocardiography is the study of human cardiac sounds. Phonocardiograms (cardiac sounds) as they are called, represent the most vital physiological and pathological information about the human body. The two vital sounds in it are first cardiac sound (S1) and the second cardiac sound (S2). Cardiac sounds different from normal sounds are called as abnormal sounds. The whole process of dividing the cardiac sounds into S1-S2 and S2-S1 cycles is called segmentation. Segmentation helps to assess the occurrence of cardiac events. Ever since the inception of phonocardiography and the evolution of digital stethoscope signal processing based on cardiac sounds have taken giant strides, may be segmentation or classification of these sounds. Segmentation has its own history. The most popular method of cardiac sounds is the wavelet transform [2,4]. Dinesh Kumar et al. performed segmentation of the cardiac sounds by using the wavelet decomposition simplicity filter [2,4]. Liang et al. segmented the HSs based on heart sound envelogram [5]. Normalized average Shannon Energy was used for this purpose. As per his method cardiac sounds were divided into four different categories namely the cardiac sound (S1), systolic period, cardiac sound (S2) and diastolic period. Most of the algorithms use standard reference such as an electrocardiogram (ECG) signal and carotid pulse data obtained simultaneously along with cardiac sound. Later, Groch presented the new method of cardiac sound segmentation. According to him, segmentation of cardiac sounds could be done effectively based on the time domain features extracted from the signal [9]. [2]Strunic used a different procedure for cardiac sound segmentation. Signals were now extracted from certain bands which reduced anomalies. Amplitude threshold were set to pick out those spikes [6]. All these methods are based on the threshold set for noise peaks. All peaks with energy more than the threshold are only considered while others are ignored. This causes loss of information and energy. To solve this problem heart sounds are segmented by a novel method by identifying the occurrence of the cardiac events. This method does not require the necessity of setting up any threshold value for the noise peaks, but instead relies on the onset and the offset of the cardiac events.

[3]This work focusses on implementing a segmentation algorithm free of any noise threshold and gives high performance under low SNR

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[1] Related work included here.
[2] Copied areas corrected.
[3] Novelty addressed.

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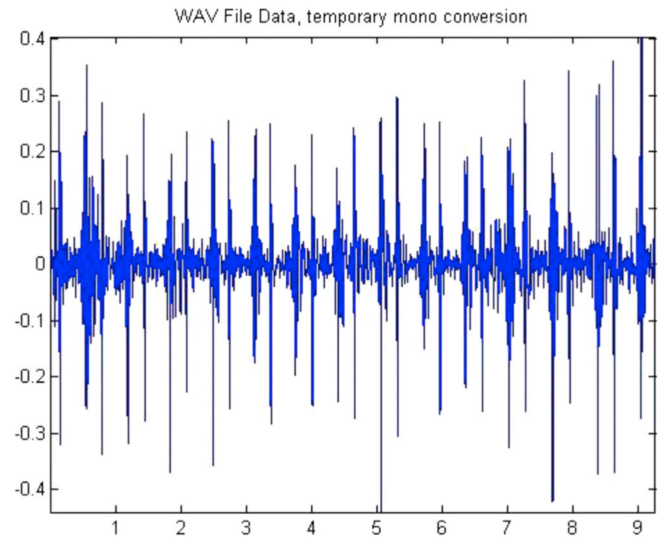

Fig. 1. Signal plot of filtered normal sound.

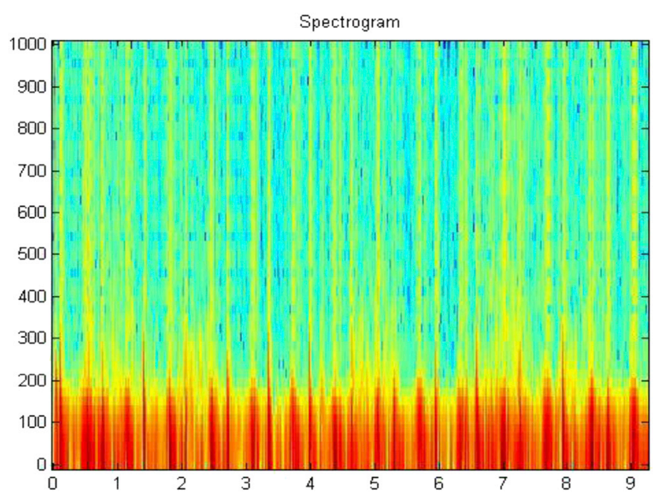

Fig. 2. Spectrogram of the filtered normal sound.

conditions.

2. State of the art

2.1. Segmentation procedure

[4]The segmentation procedure for the event synchronous segmentation of cardiac sound is divided into five steps namely Spectrogram analysis, Identification of masker and masked cardiac sounds, evaluation of Loudness index, Detection of cardiac events and Identification of S1-S2. Each of them are discussed one after the other.

2.2. Spectrogram analysis

As a first step, Spectrogram, is obtained from the cardiac sound signal, at roughly 3 ms window. The window is zero padded to allow for a broader frequency range. The following Fig. 1 shows the signal plot. [5]However, splitting up of frequencies in the audible range by means of a spectrogram does not vital information about these sounds. Audibility of sounds stretch wider with rising frequencies in the sound, from the low frequencies to the high frequencies, in terms of loudness and pitch. Thus, the spectrogram is converted into the bark scale form. The power spectrum of the sound is given in Eq. (1) [9]

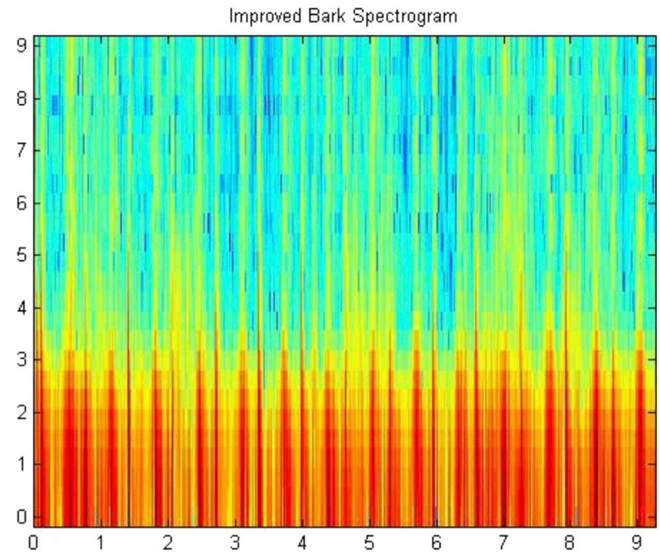

Fig. 3. Bark Spectrogram of the filtered normal sound.

$$I_i(dB) = 20\log_{10}\left(\frac{I_i}{I_0}\right), \quad i > 0 \tag{1}$$

Here, i is the instantaneous value of the power-spectrum bin of intensity I_i. I_0 is the hearing intensity threshold of the cardiac sound. To obtain a reasonable trade-off between dynamic range and resolution, $I_0 = 60$ is chosen. Sound pressure levels below −60 dB are completely clipped. The threshold of hearing is dependent on frequency and the response of these sounds on audibility. The frequency f relates to the Bark scale z(f) as per (2) [6].

$$z(f) = 13\arctan(0.00076f) + 3.5\arctan\left(\frac{f}{7500^2}\right) \tag{2}$$

2.3. Identification of masker and masked cardiac sounds

[6]A Phonocardiogram has got both low frequencies and high frequencies sounds in it. For a small time-gap of less than 150 ms it is not possible to hear a low frequency sounds such as first heart sound (20 Hz–140 Hz) and second heart sound (20 Hz–250 Hz) and a high frequency murmur sound (frequencies greater than 250 Hz) simultaneously. The high frequency sounds always have the tendency to mask low frequency sounds because of high energy content in them. This is called the simultaneous masking. Apart from simultaneous masking, temporal masking is classified into two more categories namely: pre-masking and post-masking. Pre-masking is not yet researched yet, but studies from noise bursts reveal that the duration of pre-masking is only about 20 ms duration [9]. The sounds with frequencies lower than the masker sounds lies in that period. Those sounds are not audible. It is not implemented since signal-windowing artefacts used for smoothing show a similar effect. Post-masking is different from pre-masking. It is a ringing type of temporal masking. Duration of post-masking phenomena last for about 200 ms. To smoothen the spectrogram shown in Fig. 2, the envelope of each frequency band of the cardiac sound is now convolved with a

4 Theoretical part enhanced.
5 Use of Bark Scale clarified.
6 Use of soft and loud sounds clarified.