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A novel envelope extraction method for multichannel heart sounds signal detection

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Abstract. In order to analyze the feature of multichannel heart sounds accurately and effectively, a novel envelope extraction approach based on homomorphic filtering was investigated for real-time analysis. A synchronous acquisition system built on BIOPAC physiological signal recorder was used for multichannel heart sounds acquisition at different positions on the chest, and a cutoff frequency alterable lowpass filter based on autoregressive power spectral density (AR-PSD) was established for homomorphic filtering .The performance of the algorithm was not only validated by both clinical data and 3M-littmann database, but also compared with the Normalized Average Shannon Energy and Hilbert Transform algorithm. The results of the study demonstrated that the envelope curves, extracted by the proposed Frequency Alterable Homomorphic Filtering (FAHF)method, were more smoother to read and made the adaptive segmentation of S1-S2 sounds between normal and certain abnormal heart sounds more exactly than the other envelope extraction methods.

Keywords: multichannel heart sounds; envelope extraction algorithm; Frequency Alterable Homomorphic Filtering(FAHF)

1. Introduction

According to the report of Chinese Cardiovascular Disease in 2009[1]: For the moment, there have been more than 0.23 billion people suffered from the cardiovascular disease and approximately 3 million people died of the disease each year; the fee-for-service is even high to 130 billion Yuan. With the rapid improvement of people's living standard, the morbidity and mortality of valvular disease are increasing year by year. Thus, how to find the symptom and know the status of the disease in time is extremely significant for the prevention and diagnosis.

Heart sounds signal carries important physiological and pathological information, which is about the general state of contractile activity of the cardiovascular system. The heart murmurs caused by turbulent blood flow and the incomplete opening or closing of the valves, could be heard clearly sounding like whistling, swishing or humming [2]. With the evolvement of computer science and digital signal processing, the heart sounds auscultation has been widely used to screen patients with cardiac diseases as a noninvasive medical diagnostic tool. Since most valvular diseases are multivalve lesions, and the pathologic murmurs appear with different noise patterns and intensity, conventional methods only recording heart sounds data in a single position at a time, can't provide enough pathologic information for the cardiac disease diagnoses [3]. Multichannel synchronous acquisition and analysis system can not only describe the status of the heart function according to the action of each heart sounds component at the same site, but also analyze the time duration, intensity, split and other pathological information in different sites synchronously, and evaluate the relative changes of the load status and contraction ability between the left and the right ventricle efficiently.

In this study, a multichannel synchronous acquisition system was established and an improved envelope extraction method based on homomorphic filtering [4] was proposed for multichannel signal analysis. The

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comparison between the proposed algorithm and the other representation envelope extractions, like the Normalized Average Shannon Energy [5] and Hilbert Transform [6], was also investigated. Both normal and abnormal heart sounds, which were recorded directly by the acquisition system synchronously or asynchronously, or included in 3M-littmann database, were used to test the three envelope extraction algorithms.

The overall aim of this work is to demonstrate the potential of the proposed system to facilitate screening for cardiac pathologies. The simple and useful Frequency Alterable Homomorphic Filtering envelope extraction algorithm for multichannel heart sounds signal, not only has a robust and adaptive performance, but also improves the accuracy of heart sounds segmentation efficiently. The successful heart sounds analysis lays the foundation for the following classification and identification in the future work. It is envisaged that the four-channel heart sounds acquisition and analysis system can eventually be used by a lesser skilled healthcare worker to record all the heart sounds at the different positions simultaneously, which can be saved for doctor review.

2. Data acquisition and pre-processing

2.1. Multichannel heart sounds acquisition

As shown in Fig.1, the multichannel heart sounds acquisition system(Fig.1 (c)) is built on the traditional stethoscopes(Fig.1 (a)) and the digital stethoscopes Fig.1 (b), and composed of a computer, an audio amplifier, a BIOPAC physiological signal recorder and four auscultation chest pieces.

Before acquisition, let the research object lie in the supine position, breathing normally, and put the four auscultation chest pieces on his or her four clinical auscultatory sites(These are: Aortic area, Pulmonic area, Tricuspid area, Mitral Area) to collect weak heart signals. Then, the weak signals will pass the audio amplifier to be amplified. Finally, signals will be recorded in BIOPAC physiological signal recorder and transmitted to a computer via Ethernet interface. The system breaks the limit of single channel auscultation. You can select and store any segment of the signal in a visualization environment for real-time analysis, as shown in the left column of Fig.2.

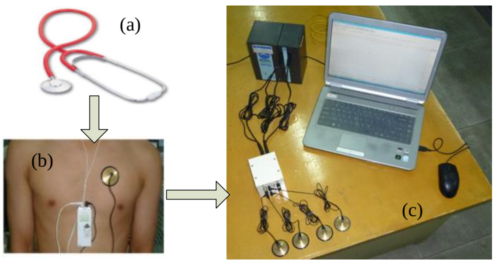

Fig.1.Multichannel heart sounds acquisition system.

2.2. Pre-processing

Due to the complexity and instability of heart sounds signal, denoising is necessary before analysis. At first, suppose the original multichannel heart sounds signal recorded using our system with 16 bit-depth and 40,000Hz sampling frequency, is defined as $x(t,i)$. Then, $x(t,i)$ is decimated by factor 20 from 40,000Hz to 2,000Hz, and filtered by a ten-order Butterworth-type bandpass filter with cutoff frequency of 20 and 700Hz. Next, as the heart sounds usually intermixed with some white gauss noise, the wavelet threshold shrinkage denoising method was applied.

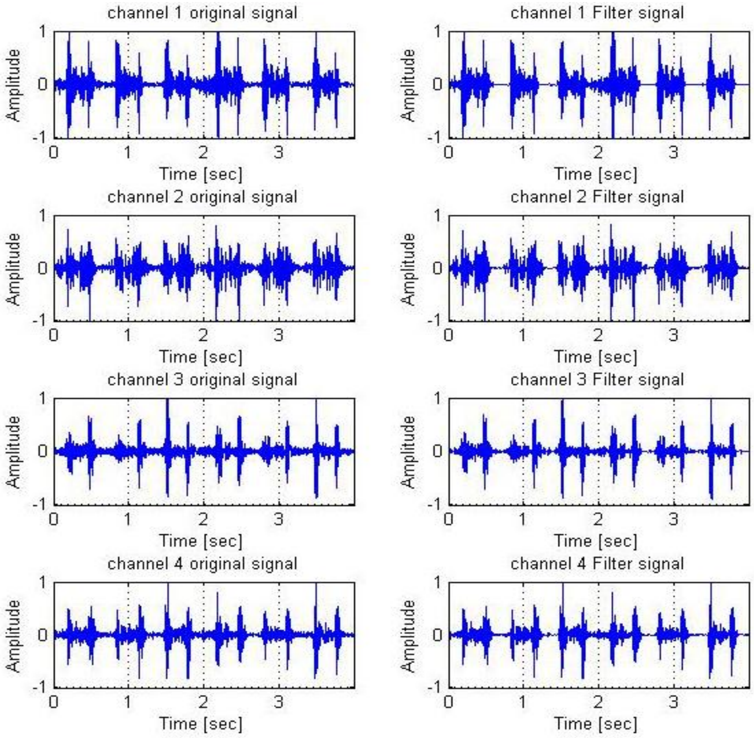

Fig.2. Multichannel signal pre-processing. The raw signals are in the left column and their pre-processed signals are in the right column.

After pre-processing the original signal, we tried normalizing the signal by setting the variance of the signal to the value of 1.0. The resulting signal can be expressed by:

$$x_{norm}(t,i) = \frac{x_{2000}(t,i)}{\max(|\ x_{2000}(t,i)\,|)} \tag{1}$$

Wherein i is the channel number, changing from 1 to 4, corresponding to the Aortic area, Pulmonic area, Tricuspid area and Mitral area respectively. Fig.2 shows the effect of the multichannel signal pre-processing.

3. Heart sounds signal analysis

The envelope curves of heart sounds signals reflect the amplitude deflections and durations from heart mechanical vibration, which possess valuable clinical diagnosis information. Thus, envelope extraction becomes one of the most important parts in heart sounds study. The most representative envelope extraction techniques like the Normalized Average Shannon Energy, Hilbert Transform and Homomorphic Filtering, which are popularly used in digital signal processing field for the moment, were discussed for multichannel signal analysis in this study.

3.1. Hilbert transform(HT)

Ville presented Hilbert Transform named of the famous mathematician David Hilbert summarized the work which Carson, Fry and Gabor did together in the year 1948. HT expresses the real signal as a complex analytic one and changes the phase of the signal but not the power and energy. The definition of HT is:

$$H[x(t,i)] = \frac{1}{\pi}\int_{-\infty}^{\infty} \frac{x(\tau,i)}{\tau - t} d\tau = x(t,i) * \frac{1}{\pi t} \tag{2}$$

Wherein '*' indicates convolution operator.

3.2. Normalized average shannon energy(NASE)

The Normalized Average Shannon Energy named as Shannon envelope is known as the popular technique on envelope extraction, which emphasizes the medium intensity signal and attenuates the low and the high intensity signal, making the envelope smoother to read [7].

The average Shannon energy is computed for each segmented signal in continuous 0.02s segments throughout the pre-processed signal with 0.01s segment overlapping using the following formula:

$$E_s = -\frac{1}{N}\sum_{j=1}^{N} x_{norm}^2(j,i)\log x_{norm}^2(j,i) \tag{3}$$

Wherein $x_{norm}(j,i)$ is the pre-processed signal, and N is signal length in 0.02s segment. So the NASE of the entire signal was obtained via normalizing the energy by the maximum of E_S.

3.3. Homomorphic filtering

3.3.1. AR-PSD

Heart sounds spectrum analysis method was usually to extract recognizable parameters indexes [8]. In this research, AR-PSD estimation method was used to determine the cutoff frequency of the lowpass filter for homomorphic filtering.

In AR model, the AR-PSD estimation can be given by:

$$P_{AR}(f,i) = T\sigma_{wi}^2 / |1+\sum_{k=1}^{p} a_{ki}e^{-j2\pi fkT}|^2 = T\sum_{m=1}^{C-1} r_{xxi}e^{-j2\pi mkT} \tag{4}$$

Wherein a_{ki} is the AR coefficient, σ_{wi}^2 is the variance of the driving noise input in channel *i*, *T* is the sampling period, *C* is selected as a power of 2 and r_{xxi} is an AR-based extrapolation of the biased estimate of the autocorrelation series derived from the channel *i* data series. The linear prediction extrapolation is:

$$r_{xxi} = -\sum\nolimits_{k=1}^{p} a_{ki} r_{xxi}(t-k) \tag{5}$$

Therefore the results of the AR-PSD estimation could be calculated from a_{ki} and σ_{wi}^2. Fig.3 shows the AR-PSD of a heart sound signal, where Fig.3(b) is the AR-PSD of the original signal in Fig.3(a) .

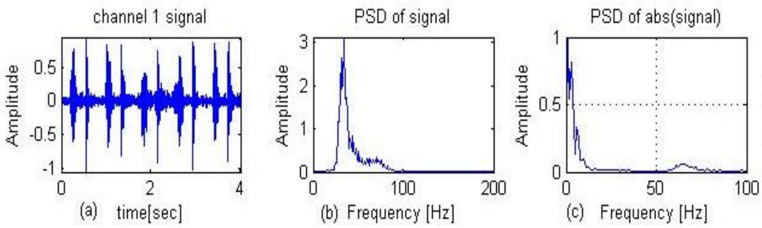

Fig.3. AR-PSD of the heart sounds signal

3.3.2. Traditional homomorphic filtering

As for heart sounds signals, the ranges of the frequency bear huge individual differences and are widely distributed. However, after square or absolute operation for the signal, it is discovered that two wave crests were appeared in frequency domain with a main peak concentrate in the low band and a smaller peak in the medium band, as shown in Fig.3(c). Table1 shows the frequency band of some heart sounds signals selected from 3M-littmann database, where FW is the frequency band of the pre-processed signal, while Fbw1 and Fbw2 are the first and second frequency band of the signal after absolute value operation respectively. According to the results of the experiments in table1, we can discover that though there was no fixed cutoff point between the two ‘peaks’, an obvious regional boundary was existed.

Homomorphic Filtering technique, which has been widely used in speech signal processing, converts a non-linear combination of signals into a linear combination by employing logarithmic transformation. The resulting spectrum can be viewed as a combination of slowly varying and fast varying parts, wherein the high frequency content corresponding to the unwanted murmurs would be removed using a lowpass filter[9].

Since only positive signal can apply logarithmic transformation to separate signal from noise during homomorphic filtering processing, the absolute value of heart sounds signal was computed for the reason that:

squared value will bury the low intensity signals under the high intensity ones by enlarging the intensity ratio, while absolute value will give the same weight to all signals without murmur weakening, especially in early disease.

Table 1. Frequency Band of 3M-Littmann Database

File Names	FW(Hz)	Fbw1(Hz)	Fbw2(Hz)
a_sclero	20-300	0-100	——
ai	20-110	0-66	110-175
ifm	20-150	0-40	115-200
ductus	70-210	0-20	——
m_regs	20-170	0-70	130-170
m_steno	20-130	0-45	70-150
normal	20-125	0-60	100-175
ourflow	20-180	0-30	120-200
p_regurg	20-150	0-75	90-170
prost	20-100	0-40	50-150
septal	20-300	0-45	60-200
t_regurg	20-80	0-36	58-130
t_steno	20-110	0-40	65-165

The flow chart of multiplication homomorphic filtering system referred in this study is as follow, in Fig.4.

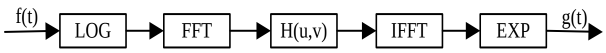

Fig.4. Homomorphic filtering system

3.3.3. Frequency alterable homomorphic filtering (FAHF)

The cutoff frequency of lowpass filter in traditional Homomorphic Filtering is invariable. As the frequency range of heart sounds is different from each other, an exorbitant cutoff frequency for lowpass filter may cause denoising incompleteness and an underestimated one may discard some necessary information. Thus, this paper proposed an improved envelope extraction method based on Frequency Alterable Homomorphic Filtering with cutoff frequency varying automatically. The procedures of the FAHF are as follows:

a) Let $x(t)$ represent the absolute value of heart sounds signal, expressed by:

$$x(t) = |s(t) * n(t)| = |s(t)| * |n(t)| \tag{6}$$

wherein $s(t)$ is the slow varying part mainly constituted by S1and S2, and $n(t)$ is the fast varying part related to heart murmurs.

b) Convert multiplication operation to addition via applying the logarithmic transformation,

$$z(t) = \log x(t) = \log[|s(t)| * |n(t)|] \tag{7}$$

In cases where $x(t) = 0$ or/and $z(t) < 0$, let:

$$z(t) = \log(x(t) + 1)) \tag{8}$$

Since:

$$|s(t)| * |n(t)| + 1 \approx |s(t) + 1| * |n(t)| \tag{9}$$

We obtain:

$$z(t) = \log(x(t)+1) = \log(|s(t)|*|n(t)|+1) \approx \log[(s(t)+1)*n(t)] = \log(s(t)+1) + \log(n(t)) \qquad (10)$$

c) Using the lowpass filter L to filter the unwanted components, having:

$$zl(t) = L[z(t)] = L[\log(x(t)+1)] \approx L[\log|s(t)+1|] + L[\log|n(t)|] \approx L[\log|s(t)+1|] \qquad (11)$$

The cutoff frequency of the lowpass filter was obtained by selecting a threshold value to seek the suitable boundary between the two peaks after depicting the AR-PSD of the pre-processed signal.

d) By exponentiation and subtracting operation, we have:

$$\exp[zl(t)] - 1 \approx \exp[\log(s(t)+1)] - 1 = s(t) \qquad (12)$$

3.3.4. Comparison of the three envelope extraction methods

Hilbert Transform expresses the real signal as a complex analytic one with its spectrum only in positive frequency domain, which is useful for the study of the instantaneous envelope. But it extracts both the detail envelope and the principal component, which resulted in a poor anti-interference performance.

The NASE emphasizes the medium intensity signal and attenuates the effect of the low and the high intensity signal, enlarging the energy gap between the signal and the low-frequency noise, which enables an easy identification of S1-S2. However, it calculates the signal in a subsection way and shows the result with the segment number as the time variable, which is equivalent to shorten the original signal in a certain proportion. The fuzzy segmentation approach may result in certain error for characteristic parameter extraction. Thus, it is more suitable for qualitative analysis of the heart sound signal.

Homomorphic Filtering technique converts a non-linear combination of signal into a linear combination and separates signal from noise, which obtains a smooth envelope curve, enabling the detection of the events suspected to S1, S2 or others easier. Fig.5 shows the comparison of the three envelope extraction methods.

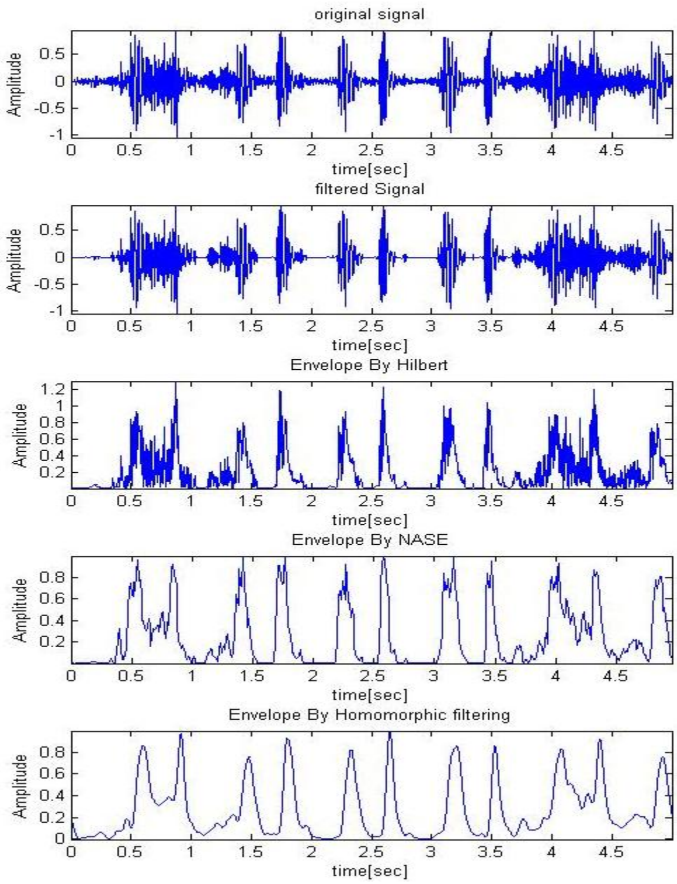

Fig.5. Envelope comparison. From the top down, they are original signal, filtered signal, HT envelope, NASE envelope and FAHF envelope

3.3.5. Peak Condition and Detection for S1-S2 Heart Sounds

The actual heart sounds recordings are very complicated and patterns of heart sounds vary significantly from recording to recording. There are some problems preventing us from using a simple threshold to pick up all the S1s and S2s. There might be 'extra peaks' due to the second part of the split S2 or other events, like systolic or diastolic clicks caused by dysfunction of the heart. In addition, some peaks may be too weak compared with other peaks to be marked. In order to solve these problems, adaptive dual-threshold method was used to mark all the peak locations and seek boundaries of segments, wherein the high threshold (T1) was used for the useful peak detection and the lower one (T2) for back noise removing. Moreover, peak condition has been performed to remove extra peaks and find lost weaker peaks, which brings easy cycle detection. The procedures are as follow:

- a) Calculate the mean value of the envelope curve to obtain the T2, and make T1 by multiplying T2 and a coefficient.
- b) Find all the peaks whose levels exceed T1.
- c) Remove the extra peaks with the time interval between two adjacent peaks less than 0.13s. If the time interval is more than 0.8s, seek for the lost weaker peaks with the peaks greater than T2.
- d) The boundaries of the S1s and S2s were decided by T2, and modified by confining the duration within 0.03s to 0.15s.
- e) Identify S1s, S2s, the systolic periods and the diastolic periods. By comparing the adjacent time intervals of the peaks, the longer one can be marked as the diastolic periods, and the start and the end of the interval can be set as S2 and S1 respectively.

In Fig.6, a multichannel heart sounds synchronous analysis system was shown, and the original signals, AR-PSDs and segmentations for each channel were also shown in the figure. Moreover, lots of features used for heart sounds classification can be extracted after envelope extraction and segmentation.

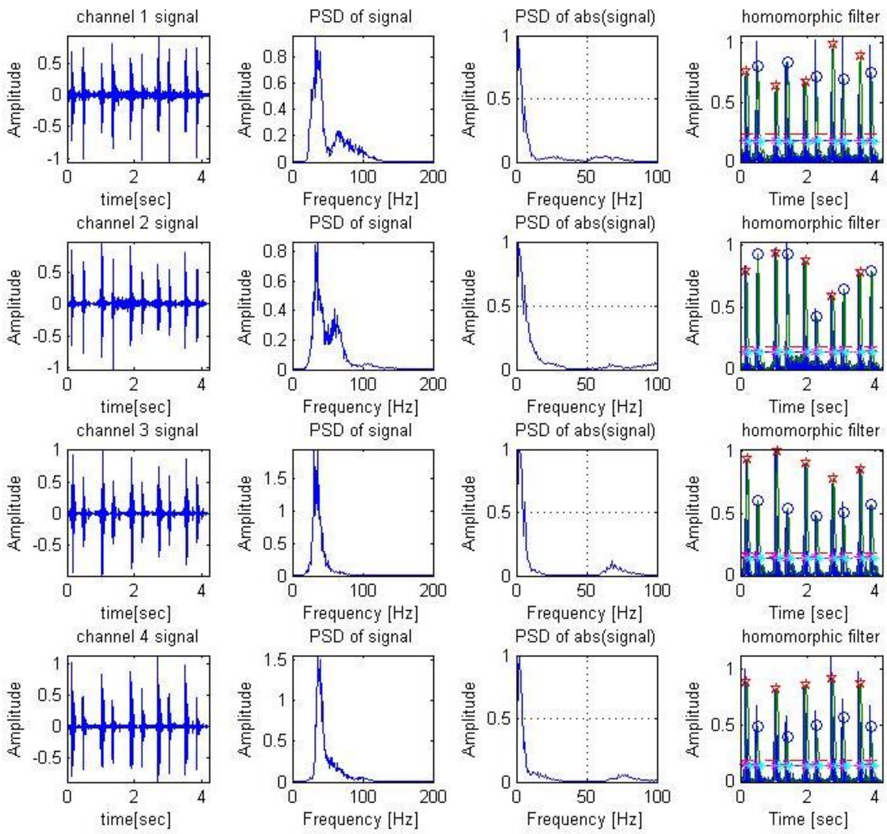

Fig. 6. Multichannel heart sounds synchronous analysis.

In the above figure, the red '☆' remarks S1, the blue 'o' remarks S2, the magenta and the cyan "*" remark the upper and the lower limits of the S1-S2 respectively, while the red and the blue line mark the adaptive dual-threshold, which are used for time parameters extraction in time domain.

4. Clinical Validation

In this section, heart sounds signal acquisition and analysis system is validated effectively by normal and abnormal cases using FAFH method. At present, 148 cases of multichannel synchronous heart sounds from 37 healthy volunteers were collected by this system (four clinical auscultatory sites of each person were tested respectively: Aortic area, Pulmonic area, Tricuspid area, Mitral area). And 86 normal cases and 88 abnormal cases were also acquired asynchronously for the experimental study. The normal heart sounds were collected from Xihua University, while the abnormal heart sounds were from Cardiothoracic Surgery, the Chengdu Military General Hospital of PLA. In order to check out the adaptability of the proposed algorithm, heart sounds from 3M-littmann database were also used for comparison. The experimental results are showed in Table 2. In this paper, we define the segmentation rate as:

$$\text{rate} = \frac{\text{The number of the cases segment correctly}}{\text{The total cases}}(\%) \tag{13}$$

Table 2. The segmentation rates according to the three envelope extraction algorithms compared in this work

Data Types	Total Cases	Segmentation rates (%)		
		HT	NASE	FAHF
Multichannel	148	85.1	92.6	97.3
Single normal	86	77.9	90.1	95.3
Single abnormal	88	77.3	79.5	87.5
3M-littmann	22	72.7	72.7	77.2

5. Discussion and Conclusion

Our study is consisted of two main steps: data acquiring and pro-processing, heart sounds analysis. In the first step, a multichannel heart sounds acquisition system was built for raw data acquisition, and some techniques such as bandpass filtering, the wavelet threshold shrinkage denoising and normalization a implemented for pro-processing. As the heart sounds analysis method, an improved envelope extraction algorithm based on AR-PSD, Frequency Alterable Homomorphic Filtering (FAHF), was developed for multichannel heart sounds signals analysis. The performance of the algorithm was compared with the Normalized Average Shannon Energy and Hilbert Transform, using both clinical data and 3M-littmann database.

With the multichannel acquisition system, the signals acquired in a visualization environment minified the background noises and were superior in quality, making the accuracy of the proposed three methods achieve 85.1%, 92.6% and 97.3%, respectively. According to the experimental results, it was shown that this simple and useful FAHF envelope curves not only had low noise and high precision as compared with NASE and HT envelope cures, but also can adaptively denoise the multiplicative noise and enable easy segmentation for non-stationary signal.

Depending on the circumstances, a heart sounds signal may be mixed with different background noises and pathological sounds. Though peak condition may eliminate some mistakable segmentation, high intensity murmurs overlapping the S1 and S2 could cause segmentation to fail, especially during the clinic abnormal signal analysis deeply buried in background noises. In the future work, we will acquire more multichannel abnormal heart sounds using our acquisition system to optimize our algorithm, and more features will be extracted in different points of view for heart sounds classification and recognition.

Overall, the multichannel heart sounds acquisition and analysis system presented in this paper is imperfect, but it shows great potential for such a device along with diagnostic software, which could be used as a fast and cost-effective screening tool for heart pathologies in the future.

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