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Denoising Processing of Heart Sound Signal Based on Wavelet Transform

Yang Yong
School of Humanities and Information Management
Chengdu Medical College
Chengdu 610500 China

***Abstract*—When the heart sounds reach the chest wall surface through mediated tissue, it is prone to generate noise, which can reduce the accuracy of pathological diagnosis. A new denoising method for heart sound signal based on wavelet transform is proposed. First of all, the information signal is transformed by multi-scale wavelet. The wavelet coefficients of each scale indicate a distribution sequence of probability according to the corresponding wavelet entropy threshold to find the maximum entropy of wavelet on certain interval, and the interval is recognized as the leading range of noise. And then, a fixed threshold denoising method is used to adaptively enhance the judgment about absolute value with larger attenuation wavelet coefficients, which can reduce the high frequency sound signal loss and improve the heart sound signal to noise ratio. The denoising simulation experiment is carried out to test the performance. The result shows that the proposed method can improve the output signal to noise ratio of heart sound signal, reducing the influence of noise on the extraction of heart sound signal, and therefore, the noise elimination algorithm has stronger anti-interference ability and superior performance.**

Keywords—Wavelet transform, Heart sound signal, Denoising processing, Signal to noise ratio

I. Introductions

Heart sound refers to the sound produced by the contraction of heart, the closing of valves and the shock of blood to the walls of ventricles and large arteries. It can be heard with a stethoscope in certain parts of chest wall. And mechanical vibration sound recorded by transducer instrument is called phonocardiograph [1] (PCG), which can be divided into the first heart sound (S1) and the second sound (S2). (Under normal circumstances it can be heard. As for the third heart sound (S3), usually only children and adolescents can hear, the fourth heart sound (S4) normal rarely heard.) The frequency of normal heart sound is 1~800Hz [2] while human hearing is sensitive to the frequency band of 40 ~ 400Hz. And the relationship between frequency and sound can reflect the heart function and myocardial blood flow within the heart. The heart sound signal in the transmission process will inevitably be interfered by the surrounding environment and the transmission medium as well as other factors. The noise will lead to the performance deterioration of the sound processing system, so the heart sound denoising technology is an important research direction in heart sound signal processing. It can improve the sound quality, enhance sound intelligibility, playing an important role in reducing noise pollution and other important aspects [3].

At present, heart sound signal denoising is studied deeply, and there have been various sound denoising methods [4, 5], such as expert system, fuzzy logic, neural network and wavelet transform technology. The wavelet transform is a time-frequency analysis method with multi-resolution. Wavelet threshold denoising method is simple and easy to be implemented and the threshold detection method based on wavelet transform has become the most widely used denoising method in reference [6]. The above presents several commonly used thresholds, but the threshold with speculation factors makes the denoised results not accurate enough. The threshold function includes soft threshold and hard threshold. And if hard threshold function is not continuous, it is prone to be the reconstructed signal with oscillation. The soft threshold function and signal processing of the real contraction signal value has certain deviation; the filtering algorithm of heart sound proposed in reference [7] can be flood signals with no spectrum in the background noise to achieve information enhancement and noise cancellation, which will improve the output SNR. The digital filter, namely FIR filter, is a type of denoising method. Its drawback is the large amount of computation and thus the real-time signal processing is not good. Adaptive wavelet transform filter and threshold method in a certain extent overcome the fixed threshold, so that useful signal with noise can be processed to improve sound quality. However, only relying on the improved threshold is far from enough to achieve a good denoising effect, because the threshold function is also very important. The traditional threshold function denoising method can be divided into the soft, hard threshold method, and both the two methods have some defects. The hard threshold function may make noise elimination not clean enough [8], which is easy to cause discontinuity (Pseudo-Gibbs) pseudo Gibbs phenomenon caused by the waveform distortion; the soft threshold function while ensuring the continuity of function at the threshold, but the constant bias has made the loss of high frequency information, which will directly affect reconstruction of heart sounds and true heart sound approximation degree. In order to overcome the defects of the two traditional threshold function, the improvement of the threshold function and empirical mode decomposition (EMD) denoising method are proposed in reference [9], but the threshold function does not have continuity in view of the above. In order to solve the problem, this paper proposes a wavelet transform denoising processing

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method based on information signal, the wavelet coefficients can transform into probability distribution according wavelet entropy threshold for each scale, find the interval wavelet maximum entropy based on wavelet entropy theory, the interval is recognized as the leading range of noise, using fixed threshold adaptive denoising method to reduce the judgment of the absolute value of attenuation larger wavelet coefficients, which can reduce the loss of high frequency signal and improve the sound signal SNR, after inverse wavelet transform, enhanced the sound, realization of heart sound signal denoising improved design. The simulation experiment is carried out, showing the superiority of this method in improving the heart sound signal denoising performance.

II. Construction of Signal Model and Preprocessing of Notch Filter Structure

A. Principle of heart sound signal diagnosis and signal modeling

In order to realize the denoising processing of heart sounds, signal model is analyzed, heart sound signals is collected by heart sound sensors, in normal heart, heart rate can produce physiological factors which accelerate ejection murmurs. In cardiac and vascular disease, changes of myocardial contractility can cause cardiac contraction in vibration. Amplitude or frequency change which changes the normal sound intensity can also produce abnormal heart sounds or heart murmurs. These pathological changes contribute to the diagnosis of cardiovascular disease, and so on. Signal distribution is denoted by $v(t,\theta)$, i.e.:

$$v(t,\theta)=\sum_{m=1}^{M}\omega_i^*(\theta)x_i(t)=\sum_{m=1}^{M}x_i^*(t)\omega_i(\theta) \quad (1)$$

In the formula, "*" denotes the complex conjugate operator. For the rapidly changing signals, the decorrelation processing is performed according to the Heisenberg uncertainty principle:

$$v(t,\theta)=\omega^H(\theta)X(t)=X^H(t)\omega(\theta) \quad (2)$$

In the formula, " H " represents complex conjugate transpose; the coefficient vector $X(t)$ and $\omega(\theta)$ respectively, the frequency resolution, can be expressed as:

$$X(t)=\left[x_1(t) \quad x_2(t) \quad \cdots \quad x_M(t)\right]^T \quad (3)$$

$$\omega(\theta)=[\omega_1(\theta) \quad \omega_2(\theta) \quad \cdots \quad \omega_M(\theta)]^T \quad (4)$$

To find out the frequency component of jumping in heart sound signal, and to get the signal cross correlation information component. The i order components using ARMA (2P, 2q) model for high frequency fitting, get:

$$c_i(n)+\sum_{j=1}^{2p}\Phi_{ij}(n)c_i(n-j)=\sum_{k=1}^{2q}\Theta_{ik}(n)u_i(n-k)+u_i(n) \quad (5)$$

In this case, the heart sound signal is represented as a group of non-steady state signals, and the output of the linear FM component is:

$$\tau_m(\theta_i)=(m-1)\tau_0(\theta_i)=(m-1)\frac{\Delta}{c}\sin\theta_i \quad (m=1,2,\cdots,M) \quad (6)$$

Here, $\tau_0(\theta_i)=\frac{\Delta}{c}\sin\theta_i$ is the frequency dispersion (frequency standard deviation), so the time-frequency characteristics of the signal can be represented by the following array model:

$$\begin{bmatrix} x_1(t) \\ x_2(t) \\ \vdots \\ x_M(t) \end{bmatrix}=\begin{bmatrix} \sum_{i=1}^{d} g_1(\theta_i)s_i(t) \\ \sum_{i=1}^{d} g_2(\theta_i)s_i(t-\frac{\Delta}{c}\sin\theta_i) \\ \vdots \\ \sum_{i=1}^{d} g_M(\theta_i)s_i(t-(M-1)\frac{\Delta}{c}\sin\theta_i) \end{bmatrix}+\begin{bmatrix} n_1(t) \\ n_2(t) \\ \vdots \\ n_M(t) \end{bmatrix} \quad (7)$$

With the above signal model construction, to provide the source of signal input for the noise elimination of heart sound signal.

B. Wavelet transform of heart sound signal processing

Wavelet transform is the signal conversion from time domain to frequency domain, signal processing in frequency domain is easy to implement, the majority of single channel sound enhancement algorithm is based on short time Fourier transform (STFT) in the frequency domain . Compared with STFT, wavelet transform overcomes the disadvantages which window size does not change with wavelet frequency [10]. Heart sound signal is adjusted by Q-factor in wavelet transform processing, processing of each layer is adjusted by coefficient threshold $T_j=\sigma_j\sqrt{2\ln(N_j)}$, σ is determined through wavelet transform and its sampling rate makes the parameter.

Assume $\mathbf{x}=[x(0),...,x(N-1)]$, where $x(n)$ is a finite length discrete-time signal, $0\le n\le N-1$, then the discrete Fourier transform (DWT) of $\mathbf{x}$ is defined as follows:

$$X(k)=\sum_{n=0}^{N-1}x(n)\exp(-j\frac{2\pi}{N}nk) \quad (8)$$

Where $0\le k\le N-1$.

$\mathbf{X} = DFT\{\mathbf{x}\}$ is the finite length signal, the information signal by wavelet multi-scale transform, the wavelet coefficients is transformed into probability distribution sequence according to the wavelet entropy threshold for each scale, i.e.:

$$\mathbf{X} = [X(0),...,X(N-1)] \tag{9}$$

A signal $x(n)$ is taken by discrete orthogonal wavelet transform, j decomposition in the high frequency coefficient scale at $d_{j,k}$, low-frequency coefficient is $a_{j,k}$ the sampling frequency is f_s. Detailed signal energy resolution of $j = 0,1,...,M$ is $E_j = \sum_k |C_j(k)|^2$, the wavelet coefficient is $C_j(k) = [x(t), \varphi_{j,k}(t)]$.

The total energy of the signal can be expressed as $E = \|x(t)\|^2 = \sum_j \sum_k |C_j(k)|^2 = \sum_j E_j$. Normalized relative wavelet energy is $P_j = E_j / E$. Due to the relative wavelet energy $\{P_1, P_2, ..., P_j\}$ covering the entire frequency range signal, each sub signal P_j describes the signal in this sub-band energy distribution probability. So if the $\sum_j P_j = 1$. j decribesr $C_j(k)$ signal which is decomposed into n cells, as:

$$E_{j,k} = \sum_k^{m/n} |C_j(k)|^2 \tag{10}$$

In the formula, m is the number of sampling points, the wavelet energy k of the corresponding $E_{j,k}$ sub interval is calculated, and the wavelet energy E_j is compared with the total energy $P_{j,k}$ of the wavelet coefficients of the layer:

$$E_j = \sum_{k=1}^{n} E_{j,k} \tag{11}$$

$$P_{j,k} = E_{j,k} / E_j \tag{12}$$

Then the wavelet entropy k definition in the WE_k interval:

$$WE_k = -\sum_j P_{j,k} \ln(P_{j,k}) \tag{13}$$

According to the different characteristics of the signal and noise energy distribution on different decomposition scales, the adaptive threshold denoising method based on wavelet entropy is used to denoise the heart sound.

III. Improved Noise Elimination Algorithm of the Heart Sound Signal

A. Wavelet entropy adaptive threshold denoising

This paper presents a signal processing denoising method based on wavelet transform which the wavelet coefficients of signal are transformed into probability distribution sequence, while the denoising threshold and threshold function are two key factors. The denoising threshold and threshold function determine directly the denoising effect of heart sound signals. The formula is adaptively determined the noise threshold of high-frequency coefficients after wavelet decomposition:

$$\lambda_j = \sigma\sqrt{2\ln(N)/\ln(j+1)} \tag{14}$$

λ_j is the threshold of each high frequency wavelet coefficients, j is decomposition level, N is the length of the signal, σ is the estimation of noise variance. $\sigma = median\dfrac{(|d_{1,\max WE}|)}{0.6745}$, in which $median(|d_{1,\max WE}|)$ is the median of lowest wavelet coefficients.

According to the wavelet entropy, the threshold λ_j of each scale is calculated, and the noise of each layer is denoised by the compromise index threshold function:

$$d_{j,k} = \begin{cases} sign(d_{j,k})(|d_{j,k}| - \dfrac{\partial\lambda_j}{1+e^{\frac{(|d_{j,k}|-\lambda_j)}{\beta}}}), |d_{j,k}| \ge \lambda_j \\ 0, |d_{j,k}| < \lambda_j \end{cases} \tag{15}$$

Among them, ∂ and β are regulatory factors, which can change the denoising effect by adjusting their values.

It can self adaptively reduce decision on the absolute value of attenuation larger wavelet coefficients, which can reduce the high frequency signal loss, improve the heart sound signal to noise ratio.

B. Wavelet denoising implementation

Assume $\mathbf{x}$ be an adjustable Q-factor SD with finite length as follows:

$$\mathbf{C}^{(0)} \leftarrow DFT\{\mathbf{x}\}$$
$$\{\mathbf{C}^{(j)}, \mathbf{W}^{(j)}\} \leftarrow AFB(\mathbf{C}^{(j-1)}, N_0^{(j)}, N_1^{(j)})$$

$$\mathbf{w}^{(j)} \leftarrow DFT^{-1}\{\mathbf{W}^{(j)}\}$$
$$\mathbf{c}^{(J)} \leftarrow DFT^{-1}\{\mathbf{C}^{(J)}\}$$

The $1 \le j \le J$, AFB is decomposition filter group, $\mathbf{c}^{(j)}$, $\mathbf{w}^{(j)}$ respectively are filter low-pass and high pass subbands from the j layer.

The inverse is as follows:

$$\mathbf{C}^{(J)} \leftarrow DFT\{\mathbf{c}^{(J)}\}$$
$$\mathbf{W}^{(j)} \leftarrow DFT\{\mathbf{w}^{(j)}\}$$
$$\mathbf{C}^{(j-1)} \leftarrow SFB(\mathbf{C}^{(j)}, \mathbf{W}^{(j)}, N^{(j)})$$
$$\mathbf{y} \leftarrow DFT^{-1}\{\mathbf{C}^{(0)}\}$$

Among them, $1 \le j \le J$ and SFB are the reconstruction filter banks

$$w(t) = s(t) + n(t),\ \ t = 1,2,...,N \qquad (16)$$

In the formula, $w(t)$ is a noisy heart sound signal, $s(t)$ is pure heart sound, $n(t)$ is noise signal. Suppose $n(t)$ is zero mean and obey $N(0,\sigma^2)$ Gauss white noise.

The wavelet basis function is selected to determine the decomposition scale j of wavelet transform, and the heart sound signal is decomposed by j scale discrete wavelet transform, and the wavelet coefficients of each scale J of the noisy signal are obtained.

The wavelet entropy theory is used to find the largest interval of wavelet entropy, and the interval is defined as the dominant range of noise, and the interval σ is set as the variance of the noise estimation.

IV. Simulation Experiment and Result Analysis

In order to test the method proposed in this paper to the application performance of noise in realization of heart sound signal simulation to heart sound signal "pathological" as an example, respectively using the minimax threshold denoising (minimax), fixed threshold (sqtwolog), and adaptive threshold based on wavelet entropy denoising, and compare these three kinds of denoising methods in denoising effect.

Figure 1 is the pathological heart sound signal, in which the sampling point of pathological heart sound signal is 3672.

Figure 2 is a pathological signal with noise, it can be seen from Fig. 2 due to noise added, voltage amplitude pathological heart sound signal has changed significantly, part of heart sound signal has been submerged in the noise and the noise added to the extraction and analysis of heart sound signal to bring some difficulties.

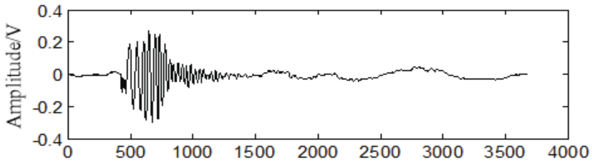

Fig. 1 pathological heart sound signal

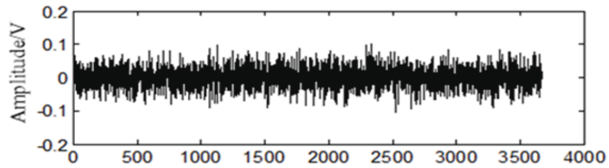

Fig. 2 pathological heart sound signal with noise

In order to test the accuracy of threshold algorithm uses the db3 wavelet decomposition of noisy signal, decomposition of 5 layers, each layer of wavelet coefficients are divided into 10 small sections. Figure 3 is the wavelet decomposition of noisy pathological signal wavelet coefficients of each layer, can be seen from the figure that the noise mainly concentrated in layers of high-frequency wavelet coefficients.

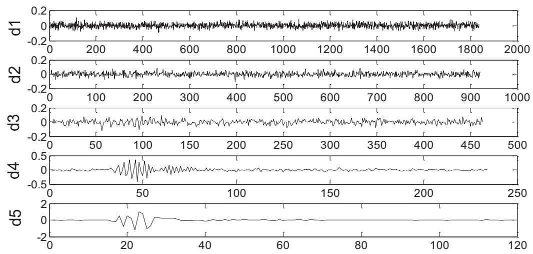

Fig.3 pathological signal layers of high-frequency wavelet coefficients

After the denoised heart sound signal, the signal-to- noise ratio SNR is selected as the performance index to evaluate the effect of the heart sound signal denoising. The pathological heart sound signal is denoised by three thredhold methods, as shown in Figure 4~6.

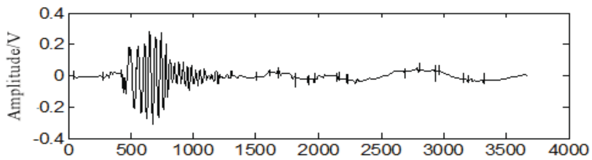

Fig 4. Minimax threshold denoising effect

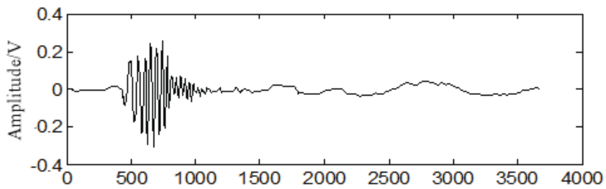

Fig. 5 Fixed threshold denoising effect

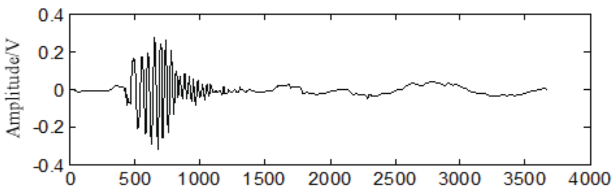

Fig. 6 Denoising effect of wavelet entropy adaptive threshold

It can be seen in Figure 4~6, the wavelet entropy adaptive threshold denoising effect is the best, the minimax threshold and fixed threshold of heart sound signal denoising after the waveform is mixed with some noise, distortion phenomenon is greater. The fixed threshold denoised waveform is relatively smooth, but in the heart sound feature sampling 500-1000 has a large error in the estimation of SNR. Table 1 is signal-to-noise ratio of pathological heart sound signal after denoising.

TAB.1 SIGNAL-TO-NOISE RATIO OF PATHOLOGICAL HEART SOUND SIGNAL AFTER DENOISING

method	SNR
minimax threshold	11.412
fixed threshold method	10.248
The proposed method	13.900

As shown in Table 1, the signal to noise ratio of wavelet entropy adaptive threshold method is increased sharply. Compared with other threshold denoising methods, the denoising performance of the proposed method is better.

V. CONCLUSIONS

This paper presents a signal denoising processing method based on wavelet transform, the wavelet coefficients are transform into probability distribution sequence according to the wavelet entropy threshold for each scale, find the interval wavelet maximum entropy by wavelet entropy theory, and the interval is recognized as the leading range of noise, using fixed threshold adaptive denoising method to reduce the judgment of the absolute value of attenuation larger wavelet coefficients, which can reduce the frequency of heart sound signal loss, improve the heart sound signal to noise ratio, after the inverse wavelet transform, enhanced the sound. The simulation result shows that using the method of signal denoising processing, can improve the heart sound signal output SNR, reduce the noise of the heart sound signal, the algorithm has strong anti-interference ability and excellent performance, and it has good application value in heart sound signal analysis and diagnosis.

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