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WAVELET-BASED ADAPTIVE DENOISING OF PHONOCARDIOGRAPHIC RECORDS

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Abstract-**The various noise components make the diagnostic evaluation of phonocardiographic records difficult or in some cases even impossible. This paper presents a novel wavelet-based denoising method using two-channel signal recording and an adaptive cross-channel coefficient thresholding technique. The qualitative evaluation of the denoising performance has shown that the proposed method cancels noises more effectively than the other examined techniques. The introduced method can be used as preprocessor stage in all fields of phonocardiography, including the recording of fetal heart sounds on the maternal abdominal surface.**
***Keywords* - phonocardiography, wavelets, denoising, signal processing**

I. INTRODUCTION

Auscultation is an ancient medical diagnostic tool. Its modern form, called phonocardiography, uses electronic stethoscopes and concerns with the automated acoustic recording and processing of heart sounds. The proper analysis of heart sounds allows the non-invasive detection of cardiac failures (eg. heart murmurs caused by valve disorders, basic cardiac arrhytmias) [1]. The processing of phonocardiographic records aims the automated detection of these abnormalities. In the field of fetal diagnosis the primary role of phonocardiography is the non-invasive long-term fetal heart rate monitoring [2], [7], [8].

Unfortunately, heart sound records are very often disturbed by various factors, which can prohibit their automated analysis. These factors can be categorized as:

- respiration sounds (lung mechanics),
- patient movements,
- small movements of the stethoscope (“shear noises”),
- acoustic damping through the bones and tissues, and
- external noises from the environment.

There are some other typical disturbances in the case of fetal phonocardiography [2]:

- acoustic damping of forewaters and maternal tissues,
- acoustic noises produced by the fetal movements,
- noises of the maternal digestive system, and
- sounds of maternal heart.

Most of the existing phonocardiographic processing methods concern only with the diagnostic analysis of heart sounds without an adequate emphasize on the denoising of the records. Existing methods usually apply digital band-pass filters (most commonly IIR-filters of FFT-based filtering) as a simple denoising method. The cut-off frequencies of the filters are determined by empirical observations, and commonly the passed band lies between 30 and 200 Hz [3], [4], [5], [6], [7].

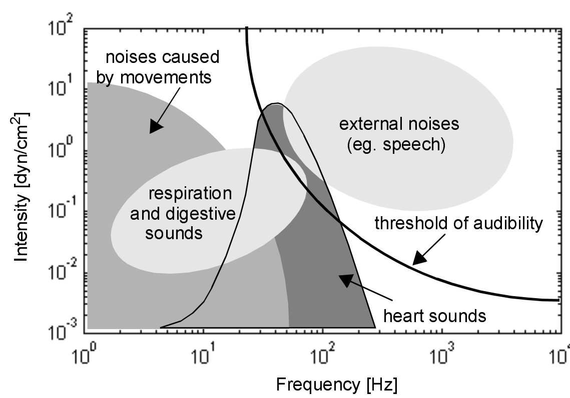

Fig. 1. Spectral intensity map of phonocardiograph records [8].

Although band-pass filtering removes a relevant part of the disturbances, it has also impact on the useful signal which also overlaps in the filtered frequency bands. Another problem is that the passed band can still contain noises (see Fig. 1.)

This paper presents a novel method which allows a more effective cancellation of noises and can be used as an advanced preprocessing stage in phonocardiographic analyzer systems.

II. METHODS

A. Signal Recording

For the recording of the heart sounds a battery powered hand-held prototype electronic stethoscope was developed which is shown in Fig. 2.

A novelty of the recording technique is that two signal channels were used which provides the effective external noise cancellation. Two identical high-sensitivity electret microphones recorded the acoustic signals. The first microphone was inserted into the focus of a stethoscope and it recorded the acoustic signals of the heart (heart channel, *A*). The second microphone was directed into the open-air and it measured the external noises (external channel, *B*). The microphone signals were amplified using ultra-low noise

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Abstract

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high-precision instrumentation amplifier stages. The obtained line level audio signals were sampled by a personal computer in stereo format at 11025Hz, 16 bits resolution.

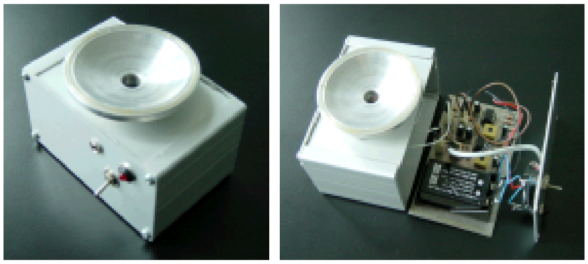

Fig. 2. Photos of the prototype recording device
Dimensions of the case: 110 x 60 x 40 mm
Diameter of the stethoscope cone is 70 mm.

B. Recorded Signals and Noises

The entire signal flow is depicted in Fig. 3. The n_i internal noises (eg. respiration sounds, movement noises, digestive sounds) coupled into the h useful heart sounds through the tissues. The n_e external noises originating from the environment are coupled into the signal via the recording surface (chest or maternal abdomen in the case of fetal phonocardiography) and via the air gap between the microphone and the surface.

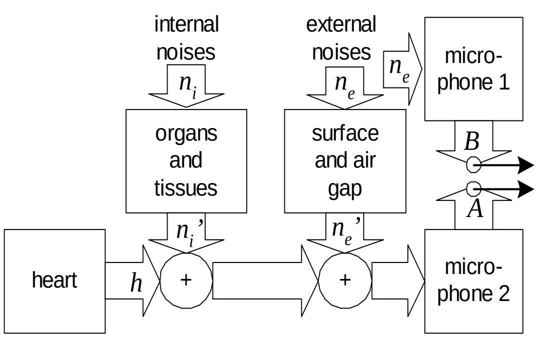

Fig. 3. The signal flow of phonocardiography.

According to Fig. 3. the recorded A and B channels can be written as:

$$A = h + n_i' + n_e'$$
$$B = n_e$$

Please note that the B external channel contains the undamped version of the environmental noises (i.e. $n_e \neq n_e'$).

C. Overview of Noise Cancellation

Fig. 4. illustrates the structure of the proposed wavelet-based denoising method consisting of four major steps:

1. wavelet-based decomposition of both recorded signal channels A and B,
2. canceling of external noises using the adaptive cross-channel thresholding of the signal coefficients,
3. adaptive thresholding of the coefficients resulted in Step 2 in order to eliminate the internal noises, and
4. reconstruction of the D denoised signal from the thresholded coefficients of Step 3.

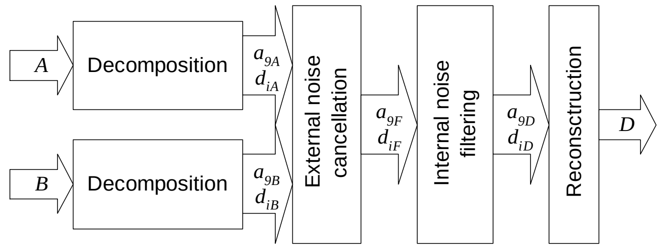

Fig. 4. Block diagram of the wavelet-based denoising system.

D. Wavelet-Based Decomposition and Reconstruction

Wavelet transform is a frequently used tool for the denoising of audio signals. The cancellation of the external noises in our case is a two-channel problem. A novel method was developed which utilizes the adaptive cross-channel thresholding of the wavelet transformed signal coefficients.

The dyadic wavelet transformed of a signal *f(t)* on the *i*-th dyadic scale can be written as:

$$W\{f(u, 2^i)\} = \frac{1}{\sqrt{2^i}}\int_{-\infty}^{\infty} f(t)\psi\left(\frac{t-u}{2^i}\right)dt = f * \bar{\psi}_{2^i}(u)$$

where ψ is the orthogonal mother wavelet, and:

$$\bar{\psi}_{2^i}(t) = \psi_{2^i}(-t) = \frac{1}{\sqrt{2^i}}\psi\left(-\frac{t}{2^i}\right)$$

The discrete version of the dyadic wavelet transform can be based on digital filter banks [9]. Let us define:

$$d_i[n] = W\{f(n, 2^i)\} = \langle f(t), \psi_{2^i}(t-n)\rangle$$
$$a_i[n] = \langle f(t), \bar{\psi}_{2^i}(t-n)\rangle$$

where $i \geq 0$. For a given filter $x[n]$, $x_i[n]$ denotes the filter obtained by inserting 2^i*-1* zeros between every filter coefficient ("holes algorithm"). $a_0[n]$ is the original signal, and its wavelet-based decomposition at the *i+1*-th scale is:

$$a_{i+1}[n] = a_i * h_i[-n], \quad \text{and} \quad d_{i+1}[n] = d_i * g_i[-n]$$

where a defines the approximated signal, and d the signal details at the given scale; h and g are biorthogonal filters, corresponding to the selected mother wavelet.

The reconstruction of the signal is done by the inverse wavelet transform, which is at the *i*-th scale:

$$a_i[n] = \frac{1}{2}(a_{i+1} * \hat{h}_i[-n] + d_{i+1} * \hat{g}_i[-n])$$

where $\hat{h}$ and $\hat{g}$ are the dual filters of h and g, respectively. In case of biorthogonal filters: $\hat{g} = h$ and $\hat{h} = g$.

TABLE I

BASE COEFFICIENTS OF THE EQUIVALENT FILTER BANK

h_0	g_0
-0.0007, -0.0018, 0.0056, 0.0237, -0.0594, -0.0765, 0.4170, 0.8127, 0.3861, -0.0674, -0.0415 0.0164	-0.0164, -0.0415, 0.0674 0.3861, -0.8127, 0.4170 0.0765, -0.0594, -0.0237 0.0056, 0.0018, -0.0007

We used the Coiflet-wavelet with 4 moments equal to zero (Coiflet-2) as the mother wavelet. The two channels were wavelet decomposed up to the 9th order ($i = 0, 1 ... 8$), resulting in two sets of coefficients: a_{9A}, d_{9A}, d_{8A}, ..., d_{1A}, and similarly: a_{9B}, d_{9B}, ... d_{1B}, where the A and B letters in the indices correspond to the recorded channels.

E. Adaptive Cross-Channel Cancellation of External Noises

The task of the external noise cancellation is to construct a new set of coefficients: a_{9F}, d_{9F}, ... d_{1F}, which can be reconstructed (inverse wavelet transformed) as the heart sound signal free of external noises. The following algorithm is proposed to obtain these coefficients:

- $a_{9F} = a_{9A}$,
- if $| d_{iA}[n] | > \max(\mu, \lambda)$ then $d_{iF}[n] = d_{iA}[n] - d_{iB}[n]$,
  else $d_{iF}[n] = 0$
  where $i = 1, 2, ... 9$; $n = 1, 2, ..., \dim(d_{iA})$, and
  $\lambda = 0.5 \max(| d_{iA} |)$, $\mu = 2 \max(| d_{iB}|)$.

Please note, that only the detail coefficients d_{iA} of the abdominal channel are adaptively thresholded (μ and λ), whereby the details d_{iB} of the external channel are adaptively subtracted from d_{iA}.

F. Internal Noise Cancellation

Another algorithm was designed which works in the frequency band $35\ Hz < f < 200\ Hz$ and adaptively thresholds the previously resulted set of coefficients a_{9F}, d_{9F}, ..., d_{1F}:

- $a_{9D} = 0$, $d_{9D} = 0$
- if $| d_{iF}[n] | > \lambda$ then $d_{iD}[n] = d_{iF}[n]$,
  else $d_{iD}[n] = 0$
  where $i = 1, 2, ... 8$; $n = 1, 2, ..., \dim(d_{iF})$, and
  $\lambda = 0.5\ \mu$, where μ is the average of the a_{9F}.

The resulted coefficient set a_{9D}, d_{9D}, ..., d_{1D} is used at the reconstruction of the phonocardiographic signal free of noises.

It should be noted that this second thresholding cannot remove all internal noises. Disturbances still remain in the reconstructed signal, however the overall noise cancellation performance is far better than in the case of simple band-pass filtering.

III. RESULTS

A prototype phonocardiographic device was designed and used for the recording. The line level input signal channels were digitized by a personal computer and the records were stored as WAV audio files.

Altogether 6 phonocardiographic records were taken and processed based on the described wavelet-based noise cancellation method. Three of the records were measured on the chest of adults and the other 3 were fetal phonocardiograms recorded on the maternal abdominal surface. The average length of the records was 5,1 minutes.

Fig. 5 shows the same 2-second sections of a fetal phonocardiogram after simple band-pass filtering ($35\ Hz < f < 200\ Hz$) and denoised by the introduced method.

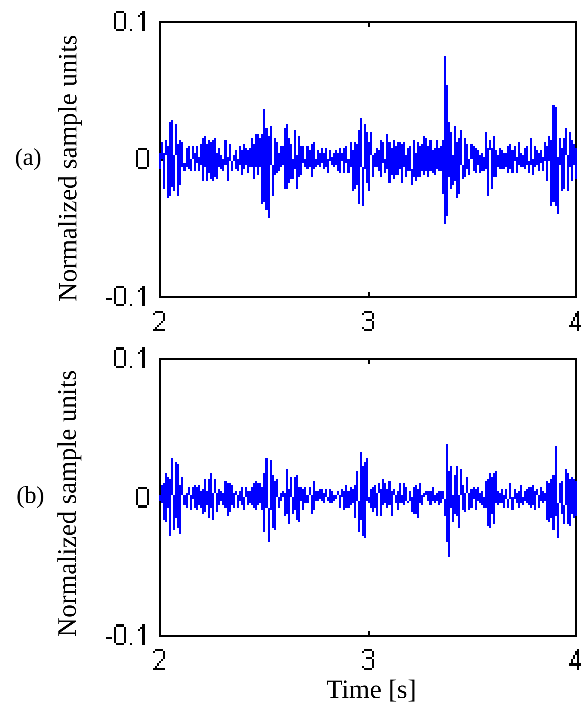

Fig. 5. The same excerpt of a fetal phonocardiogram (a) with band-pass filtering and (b) denoised by the proposed method.

In the first step we evaluated the performance of the proposed noise cancellation method by acoustic replaying of the D denoised signal. The signals were heard and listened to by a cardiologist who realized a significant improvement in signal quality (qualitative evaluation).

The records were also preprocessed by some other standard methods (adaptive filtering, blind source separation and model based filtering) which are detailed in [8]. Another comprehensive qualitative evaluation was also done using these signals. This comparison have shown that the proposed wavelet-based method provided better performance than any of the other ones.

The objective evaluation of the method is currently ongoing. A real-time phonocardiographic analysis software is currently under development which will allow the quantitative assessment of the denoising performance respecting to reference recordings (eg. measuring the heart rate using ECG or fetal CTG).

IV. Conclusion

This paper introduced a novel wavelet-based denoising method which can be applied as a preprocessing stage of phonocardiographic analyzer systems. The applied wavelet-based signal analysis and adaptive coefficient thresholding methods produce a denoised heart sound signal which is more suitable for further diagnostic analysis.

The two-channel recording technique allows an improved denoising performance. The presented method was evaluated using qualitative assessment and provided significantly better performance than any of the other examined denoising techniques. The quantitative evaluation is still ongoing.

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