Modified Spectral Subtraction for de-noising heart sounds: Interference Suppression via Spectral Comparison

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Abstract— **Heart disease remains the leading cause of fatalities in the western world. To address this many long-term systems have been developed including heart sound monitors or phonocardiograms (PCG). In daily life ambient acoustic noise interferes with the heart sound signal. This noise must be suppressed to allow effective monitoring of heart function. This paper proposes a new 2-channel signal processing technique, termed Interference Suppression via Spectral Comparison (ISSC) which is based on the 1-channel spectral subtraction algorithm, and aims at improving the quality of the recorded heart sound or PCG data. The proposed technique provides the best overall quality improvement when applied to six different heart sounds when compared to other noise cancellation algorithms namely: simple subtraction, adaptive least mean squares (LMS) and recursive least squares (RLS) filter and Wavelet Thresholding.**

I. INTRODUCTION

Heart disease is the leading cause of death in the developed world. The majority of this has been brought about by coronary heart disease (CHD). It is estimated that the economic cost of CHD alone is £7.9 billion in the UK (2004 data) and $142.5 billion in the US (2006 data) [1,2]. Many research groups have recognized the health and economic impact of heart disease. Thus, they have started to introduce long-term monitoring platforms as a mean to monitor heart function and diagnose heart disease. One such system is non-invasive, wireless phonocardiography (PCG) [3,4].

Since PCG systems operate acoustically by recording and analyzing heart sounds [5], they are highly susceptible to acoustic interference i.e. ambient noise. Thus for effective monitoring, this ambient acoustic noise must be suppressed. This often presents a complex signal processing task, due to the variation in the nature of ambient noise from one environment to another. Recent publications have concentrated on using both 1-channel [6] and 2-channel [4] Wavelet Thresholding techniques, based on the Wavelet Transform (WT), to de-noise PCG data.

The introduction of dual or multi-channel systems has proved to be advantageous in hearing aids applications [7,8]. This is because it allows the source to be localized and separated from noise using a technique called beamforming [8,9]. Recently, a dual-channel PCG monitoring system was introduced by Varady [4].

For PCG signals, source localization through beamforming is not practical due to sensor size restrictions. However, when an additional (or reference) signal is configured to not pick-up any heart sound, it can provide invaluable information about the ambient noise environment which can be exploited to facilitate the de-noising of the recorded heart sound.

The aim of this paper is to propose a novel 2-channel signal processing technique, termed Interference Suppression via Spectral Comparison (ISSC), for PCG data and to investigate its de-noising performance relative to other noise cancellation algorithms, including: simple subtraction, Adaptive Least Mean Squares (LMS), Adaptive Recursive Least Squares (RLS), and Wavelet Transform (WT)-based techniques.

II. THE RECORDING SYSTEM

A new recording system was developed for this study and is composed of five main parts: the sensor, the amplifier and filter unit, recording via a PC, the heart sound generator, and the noise generator. In what follows we provide the details of these five components.

A. Sensor Prototype

Two surface-mount microphones, manufactured by Knowles Acoustics, were used (see Table I for the specifications). The sensor is a printed circuit board (PCB), measuring 45 x 40 x 2 mm (length x width x thickness), mounted with one microphone on each side – front and back.

TABLE I. SPECIFICATIONS OF THE MICROPHONE USED

	SPM0103ND3
Directionality	Omni-directional
Sensitivity (0dB= 1V/Pa)	-22dB @ 1kHz
Output Impedance	100 Ω @ 1kHz
SNR	59 dB (nominal)
Current Consumption	0.1-0.35 mA
Power Supply	1.5-5.5 V

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B. *Amplifier and Band-Pass (BP) Filter Unit*

This 2-channel unit amplifies and filters the front and back microphone signals. Two instrumentation amplifiers (Burr-Brown INA-118) provide 26 dBs of gain. Subsequently, the signals are filtered using two 4th order, active, Bessel band-pass (BP) filters with a lower cut-off frequency, f_{min}, of 25 Hz and a higher cut-off frequency, f_{max}, of 1 kHz. The signals are then combined to form a line level stereo signal (see Fig. 1). The power supply of this unit is 3 V.

C. *Recording Unit (PC)*

The line level stereo signal is converted at 16-bit resolution with a sampling frequency, f_s, of 44100 Hz, using open-source software 'Audacity'.

These recordings were decimated 22 times to have an effective sampling frequency of about 2 kHz. This was done with the aim to minimize computational time when running the de-noising algorithms, while obeying the Nyquist condition – $f_s \geq 2f_{max}$.

D. *Heart Sound Generator*

A 12-inch bass speaker, connected to a Sony stereo hi-fi system (Model No. DHC-MD313), was used to play back the heart sounds. A thin silicone sheet was also used to mimic the skin and to provide the transmission surface of the heart sound to the microphone.

The six heart sounds used in this study were obtained from www.cardiosource.com and correspond to: Normal heart sound (NORMAL), Aortic Regurgitation (AR), Aortic Stenosis (AS), Mitral Regurgitation (MR), Innocent Systolic Murmur (ISM), and Mitral Valve Prolapse (MVP).

E. *Noise Generator*

A Sony stereo speaker (Model No. SS-MD313) was connected to an identical stereo hi-fi system to play back the recorded radio program, a mixture of speech and music. It was placed facing the back microphone and 40 cm away from the heart sound generator.

F. *Overall Set-up and Recording*

For each heart sound, a recording was taken at two distances – first with the front microphone in contact with the transmission surface and second with the front microphone one centimeter away from the transmission surface (see Fig. 2). A total of 12 recordings were taken inside an anechoic chamber.

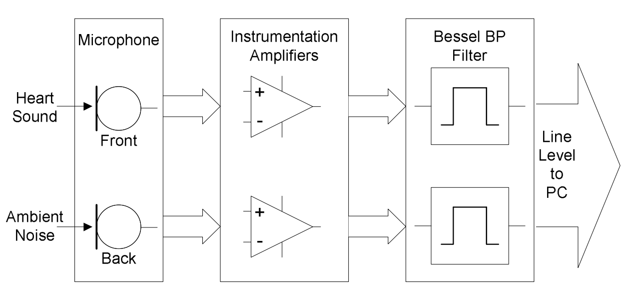


Figure 1. Block diagram of the amplifier and band-pass filter unit.

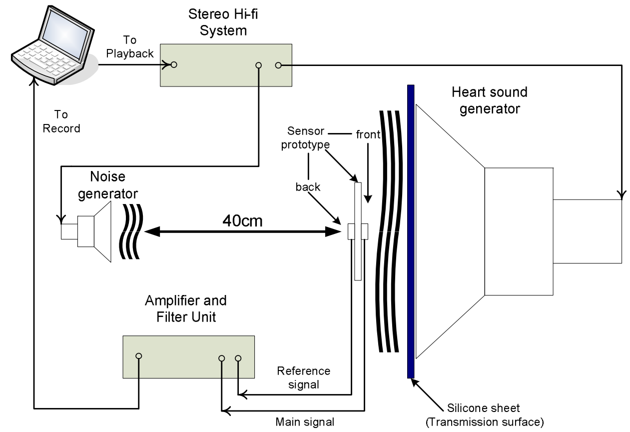


Figure 2. Overall set-up of the experiment.

III. DUAL MICROPHONE SYSTEMS

In any dual-microphone system, there will always be two signals that can be exploited to try and obtain the desired or wanted signal *s(t)*. The first is the main signal *d(t)*, which contains the wanted signal *s(t)* corrupted by a version of the ambient noise *n(t)*. The second is the reference signal *x(t)*, which contains the ambient noise *n(t)* (see Fig. 3). Normally, the required complexity of the noise cancellation operation is determined by the transmission path *h(t)*, between the front and the back or the main and the reference microphones.

Assuming, the microphones are ideal: (* denotes convolution)

$$d(t) = s(t) + n(t) * h(t) \tag{1}$$

$$x(t) = n(t) \tag{2}$$

IV. INTERFERENCE SUPPRESSION VIA SPECTRAL COMPARISON (ISSC)

Traditionally, noise cancellation algorithms have been classified in two categories. For single-channel systems, the approach is to find some aspects of the desired signal (e.g. wave shape - Wavelet Thresholding [4,6], statistical properties - Spectral Subtraction [10]) or the noise so that they can be distinguished from one another. The second approach requires a system with multi-channels to allow for more effective noise estimation which can be applied to a wider range of situations e.g. simple subtraction, Adaptive Least Mean Squares (LMS) [11], and Adaptive Recursive Least Squares (RLS).

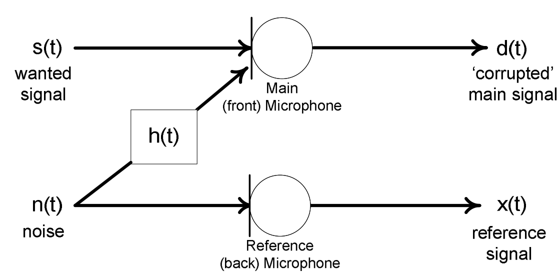


Figure 3. Input model for a dual-microphone system.

In this section, we propose a new de-noising algorithm for PCG data which combines the two approaches: Interference Suppression via Spectral Comparison (ISSC) based on Spectral Subtraction.

If both the main and the reference microphones have the same sensitivity, the following can be deduced:

1) The magnitude of the PCG signal is *higher* at the main (front) microphone than the reference (back) microphone, since it is closer to heart sound.

2) When the main microphone is in contact with the skin, which has an attenuation effect on acoustic signals, the magnitude of the noise signal is *lower* at the main microphone than at the reference microphone.

3) Therefore when the spectra of the main and the reference microphones are compared, it should be easy to distinguish which frequency components belong to the desired signal and which to noise.

The above logic allows us to discriminate the desired PCG signal components from noise. The ISSC algorithm, in Fig. 4, first applies the Short-time Fourier Transform (STFT) on the time-domain main and reference signals ($d(t)$ and $x(t)$) to obtain the frequency-domain signals ($D(f)$ and $X(f)$). The algorithm then iterates through each frequency component f and compares the magnitude of the main signal $|D(f)|$ with a threshold(α)-scaled magnitude of the reference signal $|X(f)|$. If $|D(f)|$ is greater (i.e. PCG data), $D(f)$ is passed to the output $O(f)$. However if $|D(f)|$ is less (i.e. noise), then $\beta D(f)$ is stored with $\beta \ll 1$. After this is repeated for all frequency components, Inverse-STFT (ISTFT) is applied to $O(f)$ to obtain the time-domain output $o(t)$.

An additional algorithm 'PeakRemove' was also incorporated into the algorithm to remove any high magnitude spikes present in the spectra of the main microphone due to the power supply's frequency, its harmonics, and other. This was achieved by suppressing any transient spikes in the frequency range 55-70 Hz and above 97 Hz, which has an average peak-to-base ratio above a variable threshold, thr.

V. Algorithm Comparison

Initially, only 6 recordings were taken when the front microphone is in contact with the transmission surface. However to further investigate the performance the proposed algorithm, another set of recordings was performed with the front microphone 1 centimeter away from the transmission surface. All of the recordings were subjected to 5 different de-noising techniques in MATLAB:

a) Simple subtraction (SUB) between the front and the back microphone,

b) Adaptive LMS filtering (LMS) with 256 filter coefficients,

c) Adaptive RLS filtering (RLS) with 128 filter coefficients,

d) 1-channel Wavelet Thresholding (WLET) with Coiflet-2 as the mother wavelet and soft thresholding using 8 levels of decomposition (1-8) with a fixed form threshold for un-scaled white noise, and

e) Interference Suppression via Spectral Comparison (ISSC) (see Fig. 4) with $\alpha = 0.7$, $\beta = 0.001$, frame size = 2048, and thr =7.5.

In this paper, two measures of signal quality are used. The first is derived from the mean square error (MSE) between the processed signals and the original 'clean' template signal in the time-domain. In practice, this is a measure of the signal-to-(noise plus) distortion ratio (SDR) because any shift in phase, relative wave shape and amplitude differences affects the distortion levels. To derive the SDR values, a correlation detector was used to locate the heart beats.

The second measure is derived from a comparison between the signal power (in the regions where the heart sound signal is present) to the noise power (in the regions where there is no heart sound signal) in the time domain, which is a signal-to-noise ratio indicator (SNRi). The SNRi improvements have been calculated for normal (NORMAL) heart sound only (see Table II).

VI. Results

A. Comparing De-noising Algorithms

Figs. 5a and 5b illustrate the SDR improvement obtained when using the five different de-noising algorithms on the six heart sounds. Overall, the Interference Suppression via Spectral Comparison (ISSC) and the Wavelet Thresholding (WLET) techniques provide the best de-noising performance. The simple subtraction (SUB) comes third, with the two adaptive filters giving the worst result. In Fig. 5a, the WLET technique is the best overall, while in Fig. 5b it is the proposed ISSC technique which is the best.

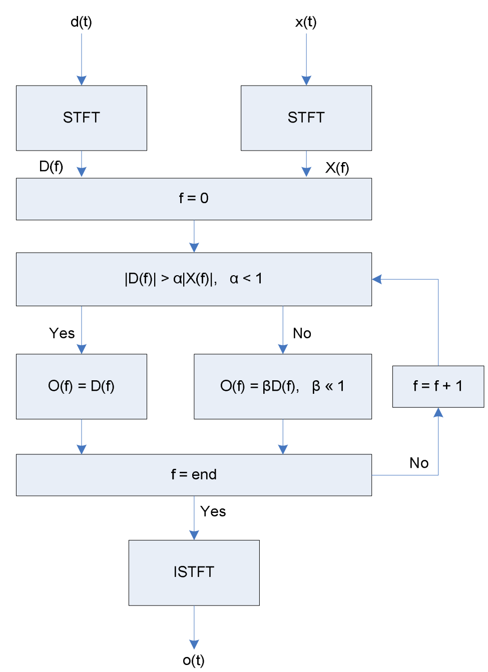
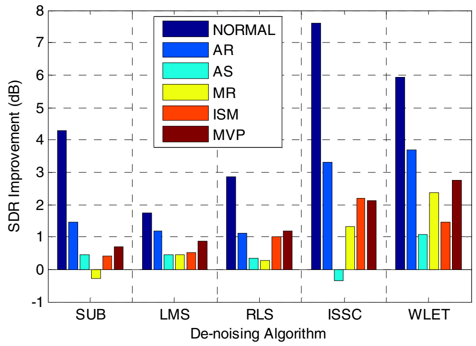
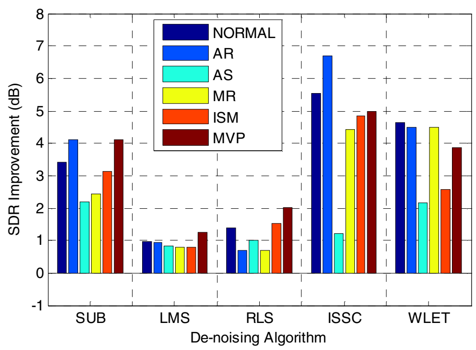


Figure 4. Flow chart of the Interference Suppression via Spectral Comparison (ISSC) technique.



(a)



(b)

Figure 5. The SDR improvement for the different de-noising algorithms and heart sounds, a) when the front microphone is in contact with the transmission surface and b) when the front microphone is 1 cm away from the transmission surface.

B. *Comparing Heart Sounds*

Again from Figs. 5a and 5b, the ISSC algorithm offers the best improvement for normal (NORMAL) and innocent systolic murmur (ISM) heart sounds. The result is mixed for aortic and mitral regurgitation (AR and MR) and mitral valve prolapse (MVP) but in Fig. 5b both the ISSC and WLET perform at a similar level. Lastly, WLET is the best algorithm for aortic stenosis (AS), while the ISSC algorithm struggles.

C. *Comparing the Recording Set-ups*

In this investigation, two recording set-ups were used. The first is when the front microphone was in contact with the transmission surface (Fig. 5a) while in the second the front microphone was 1 centimeter away from the transmission surface (Fig. 5b).

Usually, we would expect the first set-up to produce higher SDR improvements than the second set-up because of higher received signal-to-noise ratio. However, this is not the case from Figs. 5a and 5b. To confirm this we calculated the SNRi improvement and compare it to the SDR improvement as shown in Table II (for normal (NORMAL) heart sound).

By comparing the SNRi and SDR improvement values of the ISSC case, we deduce: the SDR is about 13 dBs lower than the SNRi in the first set-up and about 7 dBs lower in the second set-up. The reason(s) why this takes place are still under experimental investigation.

TABLE II. SNRI AND SDR IMPROVEMENT FOR BOTH RECORDING SET-UPS FOR NORMAL (NORMAL) HEART SOUND

Algorithm	1st Set-up (0 cm)		2nd Set-up (1 cm)	
	SNRi	***SDR***	***SNRi***	***SDR***
ISSC	20.29	7.59	12.26	5.55
WLET	18.94	5.91	13.79	4.63
SUB	15.12	4.30	9.33	3.43
LMS	4.56	1.75	2.22	0.97
RLS	8.33	2.87	3.64	1.40

VII. CONCLUSION

It has been shown experimentally that the novel Interference Suppression via Spectral Comparison (ISSC) algorithm proposed in this paper is, in general, the best de-noising algorithm for 4 out of 6 heart sounds when the front microphone is 1 centimeter away from the transmission surface. 1-channel Wavelet Thresholding (WLET) is very powerful and can outperform most 2-channel algorithms. In some cases, simple subtraction (SUB) can perform comparable to Wavelet Thresholding (WLET). Furthermore, the results suggest that the de-noising performance can be improved by *not* having the front microphone in contact with the skin. More detailed results will be presented elsewhere.

The next steps of this work are to determine whether microphone directionality plays a substantial role in the de-noising operation and to miniaturize the sensor front-end as an integrated circuit for long-term wireless monitoring.

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