UNIVERSITY OF SOUTHERN QUEENSLAND

Noise reduction method for the heart sound records from digital stethoscope

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Abstract

In recent years, digital instruments have been widely used in the medical area with the rapid development of digital technology. The digital stethoscope, which converts the acoustic sound waves in to electrical signals and then amplifies them, is gradually replacing the conventional acoustic stethoscope with the advantage of additional usage such as restoring, replaying and processing the signals for optimal listening. As the sounds are transmitted in to electrical form, they can be recorded for further signal processing. One of the major problems with recording heart sounds is noise corruption. Although there are many solutions available to noise reduction problems, it was found that most of them are based on the assumption that the noise is an additive white noise [1]. More research is required to find different de-noising techniques based on the specific noise present. Therefore, this study is motivated to answer the research question: **'How might the noise be reduced from the heart sound records collected from digital stethoscope with suitable noise reduction method'.**

This research question is divided into three sub-questions, including the identification of the noise spectrum, the design of noise reduction method and the assessment of the method. In the identification stage, five main kinds of noise were chosen and their characteristics and spectrums were discussed. Compared with different kinds of adaptive filters, the suitable noise reduction filter for this study was confirmed. To assess the effect of the method, 68 pieces of sound resources were collected for the experiment. These sounds were selected based on the noise they contain. A special noise reduction method was developed for the noise. This method was tested and assessed with those sound samples by two factors: the noise level and the noise kind.

The results of the experiment showed the effect of the noise reduction method for each kind of noise. The outcomes indicated that this method was suitable for heart sound noise reduction. The findings of this study, including the analysis of noise level and noise kind, indicated and concluded that the chosen method for heart sound noise reduction performed well.

This is perhaps the first attempt to understand and assess the noise reduction method with classified heart sound signals which are collected from the real healthcare environment. This noise reduction method may provide a de-noising solution for the specific noise present in heart sound.

Key words: digital stethoscope, noise reduction, Otsu's method, noise classification

[1] White noise: a random signal with a flat (constant) power spectral density

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Publications related to this study

Chen, H & Gururajan, R 2011, 'A de-noising method for heart sound signal using Otsu's threshold selection'. IET International Communication Conference on Wireless Mobile and Computing (CCWMC 2011), Shanghai, China

Chen, H & Gururajan, R 2012, 'Otsu’s Threshold Selection Method Applied in De-noising Heart Sound of the Digital Stethoscope Record', in Advances in Information Technology and Industry Applications, Springer, pp. 239-44.

Gururajan, Raj and Tsai, Heng-Sheng and Chen, Haoyu (2011) Using digital stethoscopes in remote patient assessment via wireless networks: the user’s perspective. International Journal of Advanced Networking and Applications, 3 (3). pp. 1140-1146.

Wang, Gengkun and Xiang, Wei and Chen, Haoyu and Wen, Peng (2011) Wireless sensor home automation networks based upon Sun SPOT. In: ICSSC 2011: IET International Conference on Smart and Sustainable City (IET ICSSC2011), 6-8 Jul 2011, Shanghai, China.

Chapter 1 Introduction

The world is experiencing a digital era (Young 2006). Currently, the advanced digital technology has wide practical applications in commercial, financial, military, medical and other fields. In the medical field, particularly, the technological advances improve almost all the medical facilities, including healthcare, imaging, auscultation and patient documentation.

The stethoscope, for example, is a commonly used acoustic device for auscultation, which was invented by Doctor Rene Lanennec, a French physician, in 1816. At that time, the stethoscope was used for listening to the sounds of a human body (Laennec 1819). Nowadays, it is not only used to listen to heart and lung sounds, but also to listen to intestines and blood flow in arteries and veins in combination with sphygmomanometer (Brusco & Nazeran 2004). Stethoscope is often considered as a symbol of all the physicians, as it can always be found on the doctor's neck. Therefore, the stethoscope has been perceived as 'the highest positive impact on the perceived trustworthiness of the practitioner' (Jiwa et al. 2012).

There are several types of stethoscopes. The conventional acoustic stethoscopes are familiar to most people for their common usage. The acoustic stethoscopes operate on the transmission of sound from chest piece via air-filled hollow tubes (Mangion 2007). The chest piece usually consists of two sides (diaphragm and bell) for sensing sound. The bell transmits low frequency sounds while the diaphragm transmits higher frequency sounds. One problem with acoustic stethoscopes is that the sound level is extremely low (Lasky 1977). Thus the sound needs to be amplified before it is sent to be heard.

The electronic stethoscope was developed in late twentieth century, which overcomes the shortcomings of the conventional stethoscope by electronically amplifying body sounds. The amplification of stethoscope only focuses on the mid-range frequency of the sound. This part of sound is converted to electrical signals and then amplified and processed for optimal listening. This conversion suffers from ambient noise interference during amplification (Durand, J. et al. 1997).

The noise corruption could be solved by the equipped analogue or digital filters. The filtering is important and useful not only in reducing the background noise, but also in enhancing diagnosis by amplifying a particular frequency band of sounds relative to the requirement (Zhang, Y. T. et al. 2006).

The emergence of the electronic stethoscope has also provided a convenient approach to record heart sound or innocent heart murmurs for clinical diagnosis, teaching and research. The electronic stethoscopes are also used with computer-aided auscultation programs to analyse the recorded heart sounds. The direct audio output can be restored and used with an external recording device, such as a PDA or PC. This function allows further study for general research, noise reduction as well as evaluation and consultation of a particular patient's condition and telemedicine, or even remote diagnosis (Palaniappan et al. 2013). The function of recording and restoring the output sound in the external device is useful as it makes the noise removal processing and simulation possible with the computer supported software. As one of the major problems with the recording heart sounds is noise corruption, some the solutions to noise reduction can be found in literature review. Various signal processing methods have been designed and implemented by software to remove the unwanted noise, these methods including averaging (Messer, S. R. et al. 2001), adaptive filtering (Jatupaiboon et al. 2010; Patel, S. B. et al. 1998), and wavelet decomposition (Gavrovska et al. 2013; Mishra et al. ; Varady 2001).

Based on the literature review, most of the methods listed above are implemented on the assumption that the noise is an additive white noise. Although the methods listed above have been proved effective, more research is required to determine the exact kind of noise corrupting the recorded heart sounds. Studies focused on both the noise classified and the noise reduced for heart sound record is limited. Thus previous studies indicated that a system which could employ suitable de-noising techniques based on the specific noise present is required (Zhang, Y. T. et al. 2006).

This study focuses on the noise reduction for the heart sound signal from the digital stethoscope. The heart sound signals are collected from the digital stethoscope, and then saved in the computer for further processing. This study analyses those signals, investigates the suitable noise reduction methods, and answers the research problem: **How might the noise be reduced from the heart sound records collected from digital stethoscope with suitable noise reduction method.**

The objective of this research is to provide suitable noise reduction method for the heart sound output from the digital stethoscope. Based on the literature and the experiment, the finding of this study can lead to robust and suitable solution to solve the noise corruption in the heart sound records. To approach the research objective, those research questions can be addressed and the further discussion and details will be explained in the following chapters.

To address this problem and to concentrate on the experiment and analysis, three research sub-questions were designed:

RQ 1. What kinds of the possible noise are in the output signals and what are their spectrums respectively?

A comprehensive knowledge of the noises mixed with the heart sound is necessary before the noise can be reduced. There are generally two kinds of noise: background noise and body noise. The background noise includes all kinds of noise generated from noisy environment while the body noise is coupled through the patient's body. Five main kinds of noise were chosen for this study. Their spectrum and characteristics are identified in the Experiment Chapter.

RQ 2. How to develop suitable noise reduction method for the noises mentioned in the above question?

The design of the noise reduction method depends on the different spectrums and frequency between the heart sound and the noise. The premise of assumption is that all kinds of signal are in different statistical properties and each of them is time and frequency shifted. In order to filter these different kinds of noises, all kinds of noise are classified into several types depending on their own statistical properties. Then the chosen noise reduction method is designed for the suitable digital signal processing. The details about how to design a suitable method for digital heart sound signals would be provided in the Methodology Chapter.

RQ 3. How to validate that the heartbeat signal or the other useful information would not be distorted after processing?

When the noise is removed from the heart sound signal, other important detailed information in the original signal should not to be eliminated as well. The processed signal would be compared with the original signal by their images and sound. Through the images, it can be identified that the effect of the de-noising method and the remaining of the two main parts of the heart sound (S1 and S2). Through the replay of sound, it can be confirmed whether the useful information for auscultation has been kept. The details can be found in the Experiment Chapter.

In addition to answering these research questions, there is a requirement for the assessment of designed noise reduction method in this study. Therefore, the effect of the noise reduction method is examined and assessed by two main factors: the noise kind and the noise level.

Noise kind is the classification of the noise and sound resources. As the sources and characteristics of noise vary, it is necessary to classify the noises before the noise reduction method is investigated. To assess the effectiveness of the noise reduction method, five main kinds of noise have been considered. The sound resources considered for this study have been divided and put into one of those groups according to the noise it contains.

On the other hand, noise level, within the scope of this study, refers to the extent that the heart sound is corrupted by the noise. The sound samples in this study were collected from hospitals. Thus the heart sounds were corrupted by various noises with different noise level. These corrupted heart sounds were then considered for the experiment using adaptive filtering techniques. The function of the adaptive filtering method can then be assessed, and the results of the noise reduction method have been provided in the Finding Chapter.

The scope of this study is restricted to designing the noise reduction method for the heart sound recordings. The focus of this study differs from other research studies as it not only designs the noise reduction method, but assesses the method with the real heart sound recordings that have been selected and classified into different groups. Therefore, the result of this study is more robust and practical.

Chapter 2 Literature Review

Electronic systems, with the context of audio communication, perform collection, recording, playback, transmission, analysis or synthesis of audio signals. Noise corruption must be considered carefully when a system is designed for any of these functions (Davis 2002). Thus noise reduction has become a long-term and popular research area (Benesty et al. 2009). To characterize different types of noise and reduce their effect, a number of signal processing methods can be introduced and thus can enhance the quality or clarity of the audio signal. A number of digital signal processing (DSP) tools or methods can be applicable to specific noise corruption depending on the different environments and different electronic systems. De-noising for heart sound records is one of the cases since the digital stethoscope has been invented. The scope of this chapter is to provide a simple and generally critical review of the DSP fundamentals and some noise reduction methods based on them. Some relevant definitions are attached as well.

In this chapter, some of the relevant concepts in noise reduction algorithms and the de-noising algorithm methods applied for digital stethoscope are presented. The content of each section is listed as follows. In the first section, a brief review of several commonly used noise reduction methods is introduced. The difference between these methods, including their strength, weakness and efficiency for noise reduction are described respectively. Then these are summarised in a table. The following section reviews the importance and role of noise reduction in the context of digital stethoscopes. In the next Section, an introduction is provided to a special technique called the Otsu's method and its current application in signal processing. The probability and advantage of applying this method to noise reduction for digital stethoscope is also discussed. The final part of this chapter provides a short review of some relevant knowledge for this research study including the development of stethoscope and the heart sound record analysis.

The subtitles of this chapter are listed below:

- 2.1 Common noise reduction algorithms
- 2.2 Heart sound record analysis and development of stethoscope
- 2.3 Noise reduction in digital stethoscopes
- 2.4 Research objectives

2.1 Common noise reduction algorithms

Noise reduction is the process of removing noise from a signal (Davis 2002). All recording devices, both analogue and digital, when collecting useful signals, have a chance to be susceptible to noise.

The signal noise reduction can be divided into two parts, the analogue style noise reduction and the digital style one. This research focuses on the digital signal noise reduction only because analogue style noise reduction systems are no longer necessary as the improvement of modern digital sound recordings. Generally, a class of noise reduction algorithms cannot work in the time-domain directly. The original digital signals need to be transformed to decomposed signals, time-frequency domain for example, before the noise reduction process. Then the de-noising algorithms, such as time-frequency filters or threshold filtering methods can be applied on the decomposed signals. Fig 2.1 below shows the three main steps of de-noising a digital signal, and the details of each step is given in the following contents.

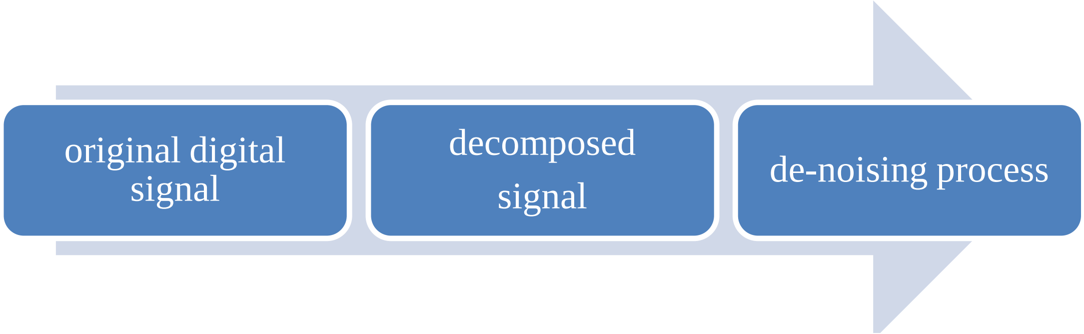


Fig 2.1 de-noising signal process

2.1.1 Original digital signal and digital signal processing Digital signal

A digital signal is a physical signal that is a representation of a sequence of discrete values (a quantified discrete-time signal), for example of an arbitrary bit stream, or of a digitized (sampled and analog-to-digital converted) analog signal(Mitra & Kuo 2006). The term digital signal can refer to two parts:

1. a continuous-time waveform signal used in any form of digital communication.

2. a pulse train signal that switches between a discrete number of voltage levels or levels of light intensity, also known as a line coded signal, for example a signal found in digital electronics or in serial communications using digital baseband transmission, or a pulse code modulation (PCM) representation of a digitized analog signal.

A signal that is generated by means of a digital modulation method (digital passband transmission), produced by a modem, is in the first case considered as a digital signal, and in the second case as converted to an analog signal.

Digital signal processing (DSP) is concerned with the representation of discrete time, discrete frequency, or other discrete domain signals by a sequence of numbers or symbols and the processing of these signals. Digital signal processing and analog signal processing are subfields of signal processing.

DSP includes subfields like: audio and speech signal processing, sonar and radar signal processing, sensor array processing, spectral estimation, statistical signal processing, digital image processing, signal processing for communications, control of systems, biomedical signal processing, seismic data processing, etc.

The goal of DSP is normally to measure, filter and/or compress continuous real-world analog signals. The first step is usually to convert the signal from an analog to a digital form, by sampling and then digitizing it using an analog-to-digital converter (ADC), which turns the analog signal into a stream of numbers. However, often, the required output signal is another analog output signal, which requires a digital-to-analog converter (DAC). Even if this process is more complex than analog processing and has a discrete value range, the application of computational power to digital signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression(Shuai & Renyi 2011).

DSP algorithms have long been run on standard computers, on specialized processors called digital signal processor on purpose-built hardware such as application-specific integrated circuit (ASICs). Today there are additional technologies used for digital signal processing including more powerful general purpose microprocessors, field-programmable gate arrays (FPGAs), digital signal controllers (mostly for industrial apps such as motor control), and stream processors, among others(Stranneby & Walker 2004).

DSP domains

In DSP, researchers normally study digital signals in one of the following domains: time domain (one-dimensional signals), spatial domain (multidimensional signals), frequency domain, and wavelet domains. They choose the domain to process a signal in by making an informed guess (or by trying different possibilities) as to which domain best represents the essential characteristics of the signal. A sequence of samples from a measuring device produces a time or spatial domain representation, where a discrete Fourier transform produces the frequency domain information, is the frequency spectrum. Autocorrelation is defined as the cross-correlation of the signal with itself over varying intervals of time or space.

Time and space domains

The most common processing approach in the time or space domain is enhancement of the input signal through a method called filtering. Digital filtering generally consists of some linear transformation of a number of surrounding samples around the current sample of the input or output signal. There are various ways to characterize filters; for example:

- A "linear" filter is a linear transformation of input samples; other filters are "non-linear". Linear filters satisfy the superposition condition, i.e. if an input is a weighted linear combination of different signals, the output is an equally weighted linear combination of the corresponding output signals.
- A "causal" filter uses only previous samples of the input or output signals; while a "non-causal" filter uses future input samples. A non-causal filter can usually be changed into a causal filter by adding a delay to it.
- A "time-invariant" filter has constant properties over time; other filters such as adaptive filters change in time.
- A "stable" filter produces an output that converges to a constant value with time, or remains bounded within a finite interval. An "unstable" filter can produce an output that grows without bounds, with bounded or even zero input.
- A "finite impulse response" (FIR) filter uses only the input signals, while an "infinite impulse response" filter (IIR) uses both the input signal and previous samples of the output signal. FIR filters are always stable, while IIR filters may be unstable.

Filters can be represented by block diagrams, which can then be used to derive a sample processing algorithm to implement the filter with hardware instructions. A filter may also be described as a difference equation, a collection of zeroes and poles or, if it is an FIR filter, an impulse response or step response.

The output of a digital filter to any given input may be calculated by convolving the input signal with the impulse response.

Frequency domain

Frequency domain analysis, also called spectrum- or spectral analysis, is a commonly used analysis in signal processing. Signals are converted from time or space domain to the frequency domain usually through the Fourier transform. The Fourier transform converts the signal information to a magnitude and phase component of each frequency. Often the Fourier transform is converted to the power spectrum, which is the magnitude of each frequency component squared.

The most common purpose for analysis of signals in the frequency domain is analysis of signal properties. The researchers can study the spectrum to determine which frequencies are present in the input signal and which are missing.

In addition to frequency information, phase information is often needed in signal processing. This can be obtained from the Fourier transform. With some applications, how the phase varies with frequency can be a significant consideration.

Filtering, particularly in non-realtime work, can also be achieved by converting to the frequency domain, applying the filter and then converting back to the time domain. This is a fast operation, and can give essentially any filter shape including excellent approximations to brickwall filters.

There are some commonly used frequency domain transformations. For example, the spectrum converts a signal to the frequency domain through Fourier transform, takes the logarithm, then applies another Fourier transform. This emphasizes the frequency components with smaller magnitude while retaining the order of magnitudes of frequency components.

2.1.2 General techniques for decomposing a signal

The decomposition methods include Fourier transform and wavelet decomposition. In this section, four decomposed methods are introduced and their strength and weakness are compared. The theory of several common signal processing methods based on the software--short-time Fourier transform (STFT), wavelet transform (WT) and wavelet packet method to digital signals are presented in this chapter. Then, a comparison of those four methods will be given to show the resolution differences among them.

There are four kinds of techniques listed: Fourier Transform (FT), Short-Time Fourier Transform (STFT), Wavelet Transform (WT) and Discrete Wavelet Transform (DWT). To determine which method is appropriate for sound de-nosing, it is necessary to review their decomposition methods of original signal. The theory of these common time-frequency transforms is presented in this chapter. Then, a comparison of the difference of those three methods was shown.

Fourier Transform (FT)

Fourier Transform (FT) is a widely used method to process and analysis signals. Fourier Transform converts a signal expressed in the time domain to a signal expressed in the frequency domain. Normally, FT is implemented in the form of a Fast Fourier Transform (FFT) algorithm. FFT, computing the Discrete Fourier Transform (DFT) speedily, has gained a wide acceptance in both academia and industry (Frigo 1999).

The mathematical definition of the FT is described below. The FT $X(\omega)$ of a signal $x(t)$ is defined as:

$$X(\omega) = \int x(t)e^{-j\omega t}dt$$

where t and ω are the time and frequency parameters respectively. It defines the spectrum of x(t) which consists of components at all frequencies over the range for which it is non-zero. For many signals, Fourier analysis is extremely useful because the signal's frequency content is often the key to analyzing signal. Therefore, the FT, especially the FFT is widely used even in some unsuitable areas (Hubbard & Meyer 1998). But Fourier analysis has a serious drawback. As the signal is transformed to the frequency domain over time, information in time domain is lost because FT does not provide frequency content information localized in time. Moreover, most signals like heart sound contains numerous non-stationary characteristics which are often the most important part of the signal and Fourier analysis is not adequate to detect it (Lee, J. et al. 2002). This will introduce difficulty for heart sound analysis for any longitudinal analysis.

Short-Time Fourier Transform (STFT)

In an effort to make up this insufficiency, Short-Time Fourier Transform (STFT) was developed in 1946 by Denis Gabor (Hubbard & Meyer 1998). The STFT differs from the usual FT by windowing and analyzing one section of the signal at a time. Thus the time signal x(t) multiplies a suitable sliding time window w(t). The additive time window introduces a time dimension and obtains a time-varying frequency analysis.
The STFT $X(t, \omega)$ of a signal x(t) is defined as:

$$X(t, \omega) = \int x(t)w(\tau - t)e^{-j\omega t}dt$$

where w(t) is the time window applied to the signal.

STFT does offer some information about time and frequencies simultaneous of the signal determined by the size of the window.

However, it still has limitations on time-frequency resolution. If more detailed frequency resolution is required, it can be attained only at the expense of temporal representation (Abbas & Bassam 2009). In brief, The STFT is just a tradeoff between the time and frequency resolution of a signal providing information on the frequency content and this negotiation is determined by the window size. The smaller the window size is, the more sharply changing component are seized, but those low frequency details are not found well. If a larger window is used, lower frequencies may be detected accurately, but localization in time domain becomes fuzzy (Obaidat 1993).
Most signals in real life, like the heart sounds, are non-stationary signals which vary greatly according to time. These signals require an adaptive method- the window size should be alterable to reveal more exact information in either time or frequency domain. As a result, the STFT analysis with a constant window size still does not meet the requirement of heart sound processing.

The Wavelet Transform (WT)

The Wavelet Transform (WT) is also used to analyze the heart sound in time and frequency domains. The wavelet transform was designed as a technique to map the signal into a frequency-time domain while achieving high time and frequency resolution at the same time (Misiti et al. 2001). The term “wavelet” was first mentioned in 1909 in a thesis by Alfred Haar. However, the breakthrough in this field has not been made until 1980’s (Hubbard & Meyer 1998).

By applying a variable sized window (the wavelets), instead of a constant window size, the WT shows a great improvement over the STFT because it can obtain both time and frequency resolution simultaneously (Rioul & Vetterli 2002). As the wavelet may be stretched or compressed, different features of the signal are extracted. The high frequency components of the signal are extracted by a narrow wavelet, while the low frequency components are picked up on a broadened wavelet (Messer, S. et al. 2001).
A wavelet is a signal of limited duration with a zero average value. A continuous wavelet transform (CWT) is defined as the convolution between the original signal x(t) and a wavelet (Vasios et al. 2001). The mathematical description of the Continuous Wavelet Transform (CWT) of a signal x(t) is defined as:

$$\mathrm{WT(t)} = \frac{1}{\sqrt{a}} \int x(t)\, g^{*}\left(\frac{t-b}{a}\right) dt = \sqrt{a} \int X(\omega)\, G^{*}(a\omega) e^{j\omega t} d\omega$$

The mother wavelet g(t) is defined as: the * denotes a complex conjugate, g(t) is the transforming function, called mother wavelet. $X(\omega)$ and $G(\omega)$ are the Fourier transforms of x(t) and g(t) respectively. The scale parameter 'a' of the wavelet is considered inversely proportional to frequency (Khadra et al. 1991);'b' is the translation parameter.

The wavelet function is given by

$$g_{a,b}(t) = \frac{1}{\sqrt{a}} g\left(\frac{t-b}{a}\right)$$

The analyzing wavelet g(t) should satisfy a number of properties. The importance is integrability and square integrability. Furthermore, the wavelet has to be concentrated in the time and frequency as much as possible (Lee, J. et al. 2002).

The process of calculating the CWT is very similar to that of the STFT. The wavelet is compared to a section at the beginning of a signal. Then a number is calculated showing how closely correlated the wavelet and original signal section are. The wavelet is moved right and the process is repeated until the wavelet covered the whole signal. The wavelet is scaled (stretched or compressed) and the previous process is repeated for all scales (Messer et al. 2000).

The Discrete Wavelet Transform (DWT)

Compared with the STFT, the CWT reveals much more details about a signal, but since all scales of the signal are used to compute the WT, the required computation time seems to be enormous. Therefore, the Discrete Wavelet Transform (DWT) is widely accepted instead. Similar to the DFT, DWT coefficients are sampled from CWT on a so-called dyadic grid, thus these coefficients are at discrete intervals of time and scale. The DWT chooses parameters of translation $b = n * 2^{m}$ and $a = 2^{m}$. The wavelet function in DWT is defined as:

$$g_{a,b}(t) = \frac{1}{\sqrt{2^m}} g\left(\frac{t - n * 2^m}{2^m}\right)$$

The DWT of a signal x[n] is calculated by passing it through a series of half-band filters and decompose the signal into approximation (from the low-pass filter) and detail (from the high-pass filter) coefficients. It is important that the two filters are related to each other and they are known as a quadrature mirror filter. This is a technique developed by Mallat (1999).

The procedure can be seen in Fig 2.2(a) and described on the basis of the following equation:

$$y_{high}(k) = \sum_n x(n) \cdot h(2k - n)$$
$$y_{low}(k) = \sum_n x(n) \cdot g(2k - n)$$

Where $y_{high}(k)$ and $y_{low}(k)$ are the output of high-pass and low-pass filtering operation, h[n] is half-band high pass filter and g[n] is half-band low pass filter. However, since each filter reduces half the bandwidth of the signal, half the samples can be abandoned according to Nyquist's rule. Thus the output signal is then sub-sampled by two (Fig2.2 (a)). After decomposition the signal at the first level, the subsequent DWT decomposition is implemented on the approximation coefficients only, showed in Fig2.2 (b).

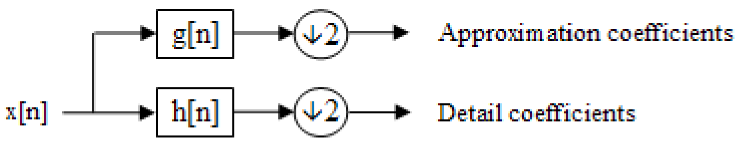


Fig 2.2 (a) First level decomposition

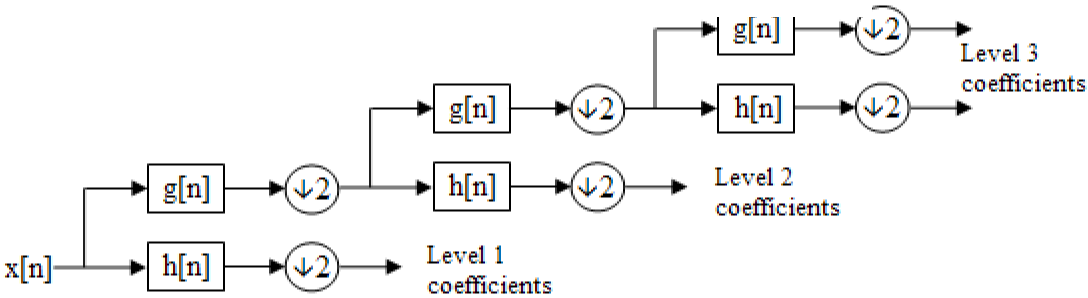


Fig 2.2 (b) Multi-level decomposition

Fig 2.3(Lei 2008) is th l signal through the DWT. The testing signal, showed on the first row, is a frequency-varying signal according to the time axis. The wavelet analysis of this signal is performed with five levels of decomposition. From each row below the first row, the left image shows the approximation output while the right image maps the detail output. Through the decomposition result, it is obvious that the testing signal is segmented into several sections due to its different frequency components by DWT, using the stretched or compressed wavelets.

With the application of DWT, the signal is decomposed into low and high frequencies at each level; the time resolution becomes fairly good at high frequencies, while the frequency resolution becomes fairly good at low frequencies (Chourasia 2009), which meets the requirement of the reality.

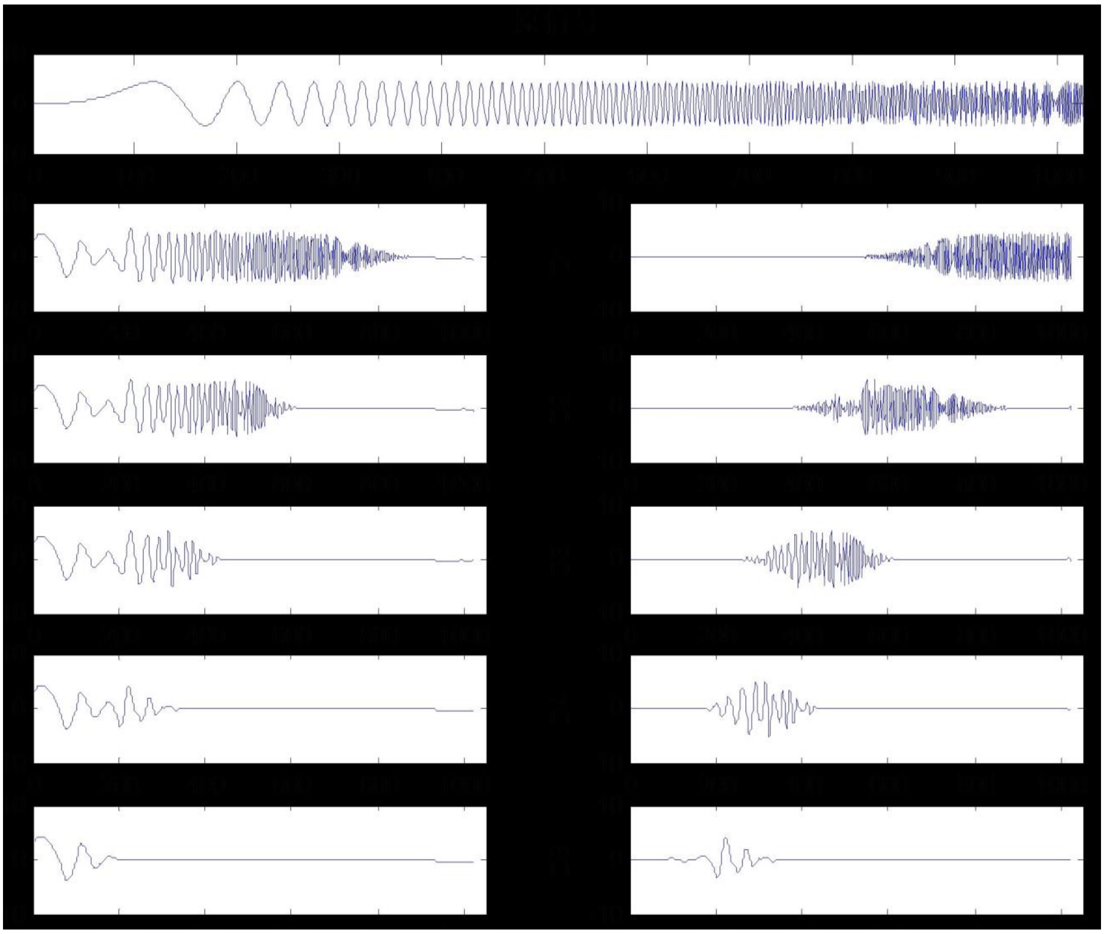


Fig 2.3 wavelet decomposition of an empirical signal

Inverse Discrete Wavelet Transform (IDWT) is the opposite direction of decomposition, which reconstruct the original signal from the wavelet coefficients. For reconstruction purposes, at each level, after up-sampling, the approximation coefficients are convolved with a low-pass reconstruction filter and the detail coefficients are convolved with a high-pass reconstruction filter. Then the upper level approximation coefficient is the sum of the outputs of the low- and high-pass reconstruction filters.
With the DWT, a fast algorithm is possible with keeping the same accuracy as other methods (Gopinath & Burrus 2002). Because of holding a huge advantage of revealing time and frequency information, DWT method is widely used in the current study.

2.1.3 Popular techniques for noise reduction

Currently, various signal processing methods can be designed and implemented by hardware or software to remove the noise (Zhang, Y. et al. 2006). Those signal processing methods include averaging (Berouti et al. 2003; Marro et al. 2002), adaptive filtering (Glover Jr 2003; Goodwin & Sin 2009; Kaneda & Ohga 2003), and wavelet decomposition (Fang & Huang 2004; Pizurica et al. 2003).

A class of algorithms work in the time-frequency domain using some linear or non-linear filters that have local characteristics and are often called time-frequency filters. Noise can therefore be removed by use of spectral editing tools, which work in this time-frequency domain, allowing local modifications without affecting nearby signal energy. This can be done manually by using a mouse with a pen that has a defined time-frequency shape. This is done much like in a paint program drawing pictures. Another way is to define a dynamic threshold for filtering noise, which is derived from the local signal, again with respect to a local time-frequency region. Everything below the threshold will be filtered, everything above the threshold, like partials of a voice or "wanted noise", will be untouched. The region is typically defined by the location of the signal Instantaneous Frequency, as most of the signal energy to be preserved is concentrated about it.

Although adaptive filtering, based on Fourier-based analyzing tools, has some limitations concerning frequency and time resolutions, it is still the most commonly used method in signal processing(Chourasia & Mittra 2009). Wavelet transform, which addresses these limitations, still requires selecting suitable de-noising algorithm(Rosas-Orea et al. 2005).

Thresholding method is one of the commonly used noise reduction method. The whole de-noising progress consists of three ordinal steps: decomposition, thresholding and reconstruction. The decomposition and the reconstruction are performed with the selection of right wavelet family and mother wavelet to transform the empirical signal into a set of coefficients and then reconstruction with the adjusted coefficients, described in Section 2.4.3. The thresholding step is the selection of threshold level for de-noising of the signals. So the whole presented de-noising methods are mainly different in the way coefficients are treated by de-noising algorithms(Hess et al. 1997). Three different kinds of de-noising algorithms (thresholds) are always applied for the approach: the Universal threshold, the Rigorous SURE threshold and the Minimax threshold (Donoho & Johnstone 1994).

The Minimax threshold

The Minimax consists in an optimal threshold that is derived from minimizing the constant term in an upper bound of the risk involved in the estimation of the signal (Donoho & Johnstone 1994). The optimal threshold is defined as

$$\lambda_M = \sigma\lambda_{\mathrm{n}}^*$$

Where σ is the standard deviation and λ_{n}^* is determined by minimax rule such as the maximum risk of estimation error(Taswell 1995) and the minimum of the maximum mean square error(Jansen & Bultheel 2002).

Universal threshold

The Universal threshold de-noising algorithm, also called VisuShrink (Donoho & Johnstone 1994) or Sqtwolog in Matlab, uses a fixed threshold form, which is defined as

$$\lambda_{UNI} = \sigma\sqrt{2\log(n)}$$

where ‘n’ indicate the length of the signal and σ is the standard deviation.

The implementation of this threshold does not need the development of lookup tables. Nevertheless, the universal threshold is depended on the data size and substantially larger than the Minimax threshold (Antoniadis et al. 2001).

Rigorous SURE threshold
Both of the previously described de-noising algorithms use global thresholds. That means the computed threshold is applied to all wavelet coefficients. The Rigorous SURE threshold (Sardy et al. 2004), named Rigrsure in Matlab, describes a scheme that uses a varying threshold - at each resolution level 'j' of the wavelet coefficients a threshold value λ_j would be used. The Rigorous SURE threshold de-noising algorithm, also called SureShrink, achieves an unbiased estimate through the Stein's Unbiased Risk Estimate criterion.

Soft and hard threshold
After the efficient decomposing of signals, the wavelet coefficients due to useful signal (heart sound) tend to be larger than those coefficients due to noise. The noise-based coefficients below a certain threshold are then filtered out by de-noising algorithm. The de-noising algorithms can be divided into linear and non-linear methods.
The linear method is implemented regardless of the wavelet coefficient size. This method assumes that the noise can be detected not in coarse scales coefficients but mainly in fine scale ones (Hess et al. 1997). On the other hand, Non-linear de-noising method is based on the assumption that noise can be found in every coefficient and is distributed over all scales. This method can be applied in two ways: hard thresholding and soft thresholding. In both thresholding, the coefficients below a certain threshold are set to zero; the other coefficients are maintain in hard thresholding, but also reduced by the value of threshold in soft thresholding to some extent. They are:

$$\text{Hard thresholding:} \quad \mathrm{s(x)} = \begin{cases} \mathrm{s(x)}, |\mathrm{x}| > \lambda \\ 0 \;, |\mathrm{x}| < \lambda \end{cases}$$

$$\text{Soft thresholding:} \quad \mathrm{s(x)} = \begin{cases} \mathrm{sign(x)}(|\mathrm{x}| - \lambda), |\mathrm{x}| > \lambda \\ 0 \qquad\qquad\quad , |\mathrm{x}| < \lambda \end{cases}$$

where $\mathrm{s(x)}$is the empirical signal, and λ is selected threshold.
There appears to be limited complete de-noising system designed for the newly developed electronic stethoscopes which was just developed in the end of twentieth century. Wavelet-based de-noising method has applied to heart sound de-noising (Bing-lian & Qian 2006; Huiying et al. 2002) and viewpoints in the literature promote the de-noising algorithms for phonocardiographic (PCG) output signals (Chourasia 2009; Khadra et al. 1991). However, limited evidence is found to show that those methods are also suitable for electronic stethoscopes outputs. Furthermore, the Daubechies, Coiflet and Symmlet wavelets, are frequently-used in the wavelet transform, but no one wavelet seems to consistently give better results than another in de-noising (Messer, S., Agzarian, J. & Abbott, D. 2001). The threshold value λ is an empirical value and varies in different environments. There is little principle about identifying the threshold value and the thresholding function particular for sound de-noising.

2.2 Heart sound record analysis and development of stethoscope

In this section, the signal collected by the current digital stethoscopes is introduced. In this study, the noise of the original heart sounds need to be reduced. Thus it is necessary to introduce the heart sound and heart sound analysis. Then a brief review of the current digital stethoscopes which used for collected heart sound is described. How signals are collected by digital stethoscopes and problems or issues in the collecting process is discussed as follows.

2.2.1 The heart sound analysis and heart sound signal

Heart sound is highly a non-stationary signal. Currently, the whole processing of heart sound auscultation is implemented into the following general steps: Preprocessing, Segmentation, Feature Extraction and Classification. The first step, Preprocessing, is to sample and filter the raw heart sound so as to remove the noise and to prepare it for further analysis. During filtering, in order to limit the impact on the useful signal which also overlaps in the filtered frequency band, it is important to study the frequency and time information of the heard sound.

The origins of heart sounds

Heart sounds are the acoustic vibrations emerged during the cardiac cycle. Usually, the whole process of cardiac cycle is divided into two parts, the systole and diastole phases due to fast accelerations and retardation of the blood in the chambers and arteries (Reed, T. et al. 2004). During those two phases of the cardiac cycle, audible sounds are made from the opening and the closing of the heart valves, and the blood flowing in the heart (Zhang, Y. T. et al. 2006).

The most widely accepted theory on the genesis of the heart sound is described by Rusher and states that heart sound consists of four components (Durand, L. et al. 2002), S1 to S4. Normal heart sound has two major components, the first heart sound (S1) and the second heart sound (S2), which can be heard clearly in each heart cycle.

S1 is generated at the onset of ventricular contraction (the end of arterial contraction), while S2 occurs during ventricular diastole. Yuenyong states that systole is the period between S1 and S2 and diastole is the period between S2 and S1 (Yuenyong et al. 2009). The normal third heart sound (S3) is always audible in children and youth but not in most adults while the fourth heart sound is seldom audible in normal individuals, and pathological if occurred in older adults. The example of a normal heart sound is showed on Fig 2.4 (Woywodt et al. 2004).

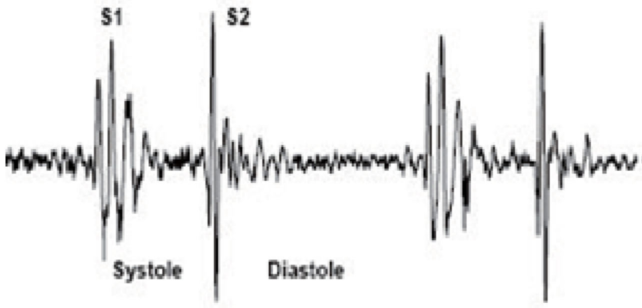


Fig 2.4 the records of a normal heart sound

Heart sound analysis

Heart sound provides clinicians with valuable diagnostic clues and crucial prognostic information with acoustical and mechanical phenomena of the cardiac cycle. As many heart diseases are associated with the characteristic changes in the intensities of or the time relation between the S1 and S2 (Lee, J. et al. 2002), it is crucial to analysis the frequency range of each heart component to conduct the initial diagnostics.

The whole frequency of heart sounds and murmurs is a wide range from 0.1Hz to 2000Hz (Webster 2009). However, most of the information carried by the heart signal is too weak to be recognized by human ear. Thus the audible range of the heart sounds above the audible level is about 40–500 Hz, which possessed only a narrow audible range (Leatham 1970).

The first heart sound (S1) is characterized by higher amplitude, low tone and longer duration in comparison with other heart sounds. S1 has two major high-frequency components and its frequency components are mainly in the range of 10–200 Hz. The second heart sound (S2) usually has a more extended spectral activity compared with the first heart sound (S1). Specifically, S2 spectra have greater amplitude than S1 spectra above 150Hz (Arnott et al. 1984). It occupies frequencies between 50Hz and 300Hz. Because the frequency range of heart sound is generally certain to some degree, removing the noises outside this range is as easy as introducing the suitable digital filters. However for those noises contained in the pass band, another method is necessary.

2.2.2 The development of stethoscope

The development of the conventional stethoscope

The stethoscope, used to transmit heart and lung sounds from those organs, through the chest wall to human ears, was the first diagnostic instrument and has become the badge of the physician. Although the modern stethoscope is quite different from the original one in appearance, function and application, it is always regarded as an essential medical instrument. The first model of stethoscope was created by R.T.H. Laennec in 1816. Then the next innovation was the flexible, monaural stethoscope, which was invented in 1828-29, and the binaural stethoscope which was invented in the early 1950s (Bishop 1980). Both inventions provided a model structure to the modern medical stethoscope. Currently, five basic components are believed to be found in the modern mechanical stethoscope: the earpieces, binaural, metal brace, tubing, and the chest piece (or the stethoscope head). Once the sound is recorded by the chest piece at the skin surface, it transmitted via the tubing and the binaural to the earpiece (Zhang, Y. T. et al. 2006).

Auscultation, the process of listening to sounds emanating from the body, is important and easy to use such tool in clinical diagnostic routines. Conventional medical practice uses a mechanical stethoscope for auscultation. Often, the practitioners would need to rely on their hearing ability and their subjective judgment on the interpretation of the sounds (Messer, S., Agzarian, J. & Abbott, D. 2001).

Another major problem with the conventional stethoscope is noise corruption. Many sources of noise may pollute the heart sound including fetal breath sounds if the subject is pregnant, lung and breath sounds, environment noise from contact between the recording device and the skin. Moreover, the manner the stethoscope is used can greatly affect the sounds perceived. Therefore, a need existed for a better instrument that could convey sound more accurate and loudly to make the auscultation process more comfortable and convenient for both physicians and patients (Zhang, Y. T. et al. 2006).

The emergence of electronic stethoscope (digital stethoscope)

In late twentieth century, electronic stethoscopes were developed to overcome the shortcomings of the conventional mechanical stethoscopes. With the introduction of electronic stethoscopes, people are now hoping to measure and to analyze heart sounds in a more objective manner. As the development of the electronic stethoscope, a computer-based system for recording and analyzing of body sounds is introduced. Though the system cannot replace the human ear, it can complement diagnose (Schuttler et al. 1996). However, the stethoscope is highly prone to interference from ambient noise and thus becomes clinically useless in high ambient noise environments (Patel, S. et al. 1998).

The newly designed electronic stethoscope allows heart sound to be digitally recorded and downloaded to a computer for analysis. It gives chance to the introduction of various signal processing methods. In order to auscultate in noisy condition, the application of noise reduction technique is essential (Müller & Kompis 2002). Most of the reduction techniques are based on an adaptive noise canceller and use a least mean square algorithm (LMS). However, in practice, most DSP algorithms can only remove periodic noise which is much easier to cancel than broadband random noise (Harley 1997). Nevertheless, there are many situations in which random noise cancellation is required. Therefore, a lot of the computer-based heart sound analysis techniques adopted by researchers, such as wavelet transform and neural network, have already provided new insight into the de-noising techniques. In this research, it is schemed to design an improved and synthetical wavelet-based noise cancellation system which can control the noises in a better way.

Similar devices: PCG and ECG

There are two different devices which are similar to electronic stethoscope, namely phonocardiograph (PCG) and electrocardiograph (ECG).
Electrocardiography is a transthoracic interpretation of the electrical activity of the heart. The wave is over time captured and externally recorded by skin electrodes (Alvarez 1922). It is a noninvasive recording produced by an electrocardiographic device. The ECG works mostly by detecting and amplifying the electrical changes on the skin that are caused when the heart muscle "depolarises" during each heartbeat. During each heartbeat a healthy heart will have an orderly progression of a wave of depolarisation. This is detected as tiny rises and falls in the voltage between two electrodes placed either side of the heart which is displayed as a wavy line either on a screen or on paper. This display indicates the overall rhythm of the heart and weaknesses in different parts of the heart muscle. The weakness of the ECG is that when the sound is recorded by ECG, it is impossible to replay and heard by human ears.

Phonocardiogram or PCG is a high fidelity technique for registering sounds and murmurs made by the heart during a cardiac cycle with the help of phonocardiograph. The sounds are thought to result from vibrations of the heart valves. It allows the detection of the timing and relative intensities of faint sound and murmur, and make a permanent record of these events. In contrast, the conventional stethoscope cannot detect such sounds or murmurs, and provides no record of their occurrence. The ability to quantify the heart sounds provides vital information about the effects of certain cardiac changes in wave shape and timing parameters upon the heart. It is also an effective method for tracking the progress of the patient's disease(Walker et al. 1990).

Although Phonocardiography can record and store auscultator findings accurately, its usage as a diagnostic tool is uncommon because of critical procedures and complicated instrumentation. A standard procedure to record requires a specially designed, acoustically quiet room. Further, the phonocardiographic devices were typically large, noisy, and inconvenient to use (Zhang, Y. et al. 2006).

2.3 Noise reduction in digital stethoscopes

Auscultation now uses digital stethoscopes and concerns with the automated acoustic recording and processing of medical signals. But these medical signals can potentially be corrupted by noise in a variety of ways (Messer, Agzarian & Abbott 2000). The sequence of corruption demonstrates several key areas where external interferences could cause degradation of the original signal. When dealing with critical medical signals, such as heartbeats, it is important if the data does become corrupted by noise and these alterations can be eliminated in an accurate and effective manner (Danahy et al. 2005).

2.3.1 Noise analysis

In reality, heart sound records are often disturbed by various factors, which can prohibit the accuracy of the original sound (Varady 2001). Most of these factors are noises from sources such as breath sounds, contact of the stethoscope with the skin, fetal heart sounds if the subject is pregnant, and ambient noise that may corrupt the heart sound signals.

To make it easier, these factors can be categorized as two aspects in the mass: external factors and internal factors.

External factors:

- Small movement of the stethoscope ("shear noises" or friction noises)
- Ambient noise
- Instrument noises
- Human voices
- Patient movements

Internal factors:

- Respiration sounds (lung mechanics) or breathing noise.
- Acoustic damping through the bones and tissues

Currently, there is no way of knowing a priori what the particular noise component is, or of determining the noise component once the measurement has been recorded. In every case and situation, the noise will be different (Messer, Agzarian & Abbott 2000).

The electronic stethoscope will become a much more useful diagnostic tool if unwanted noises are removed revealing the heartbeat sound clearly and integrated. This research attempts to find the suitable way to reduce the unwanted noise and improve the quality of the heart sound.

2.3.2 Noise reduction for heart sound

Based on the literature review, there appears to be several gaps. There appears to be limited complete de-noising system or method designed for the newly developed digital stethoscopes which was just invented less than twenty years. Adaptive threshold selection de-noising method has applied to heart sound de-noising (Bing-lian & Qian 2006; Huiying, Sakari & Iiro 2002) and viewpoints in the literature promote the de-noising algorithms for phonocardiographic (PCG) output signals (Chourasia 2009; Khadra et al. 1991). However, limited evidence is found to show that those methods are also suitable for electronic stethoscope outputs. Furthermore, those adaptive threshold selection methods, which are frequently-used in other areas, not seem to consistently give better results in de-noising (Messer, S., Agzarian, J. & Abbott, D. 2001). In the adaptive threshold selection methods, the threshold value λ is an empirical value and varies in different environments. There is little principle about identifying the threshold value and the thresholding function particular for heart sound de-noising.

To fill these gaps, this study has developed a new de-noising method particularly applied in electronic stethoscope outputs quoted from Otsu's method. In this study, it would be detected which decomposition levels and thresholding methods can be best used for removing the noise in a heartbeat sound from the electronic stethoscope without great loss of useful information.

2.4 Research objectives

The main objective of this research is to use a set of suitable digital processing techniques to reduce the noises in the heart sound from the digital stethoscope. This involves properly recognizing various sound attributes and their respective frequencies, including the frequencies of unwanted noises, and then developing the suitable algorithms to control these unwanted sections. These lead to the following sub-questions:

1) What kinds of the possible noises are in the output signals and what are their spectrums respectively?

To reduce the noise contained in the heart sound, it is necessary to comprehend the content of the noises at first. The noise in the output signal of the digital stethoscope is generally made up of two different kinds-- background noise and body noise. The background noise includes all kinds of noise generated from noisy environment while the body noise is coupled through the patient's body (section 2.2). To condition the digital stethoscope data, it is necessary to know the difference between these noises as each of them has distinct spectrum and statistical properties.

2) How to develop suitable noise reduction method for the noises mentioned in the above question?

Noise reduction is not a simple problem and cannot be solved by just subtracting a part of the signal from another. The reason of this assumption is that all kinds of the signal are in different statistical properties and each of them is time and frequency shifted.

In order to filter these different kinds of noises, it is essential to classify these noises into several types depended on their own statistical properties and then design the suitable digital signal processing. The details about design the suitable method for digital heart sound signals will be introduced in "Methodology" chapter.

3) How to validate that the heartbeat signal or the other useful information would not be distorted after processing?

The aim of the de-noising process is to remove noise by correctly identifying the corrupted data parts and reduce this segment. However, one has to take into account throughout the signal process is not to eliminate other important detailed information that both the structure and details of the original clean signal contains. The mean squared error (MSE) and zero difference count (ZDC) are widely applied in most research (Danahy, Agaian & Panetta 2005). Both methods will be calculated between the processed outputs and clean signal to demonstrate results.

Chapter 3 Otsu's Method

3.1 The Otsu's method theory

The theory of the Otsu's noise reduction method is quoted from Threshold Selection Method from Gray-Level Histograms (Otsu, N 1975) which also called Otsu's method. It is a nonparametric and unsupervised method of automatic threshold selection applied for many areas. In picture segmentation, for example, it selects the threshold at the gray level with the maximal amount of difference. The benefit of this theory is threshold selection without other priori knowledge.

Otsu's method (Liao et al. 2001) is a very popular global automatic thresholding technique, which has been applied to a wide range of applications (Tian et al. 2003). Those applications are not limited to such area as noise reduction for human action recognition (Arseneau & Cooperstock 1999), adaptive progressive thresholding to segment lumen regions from endoscopic images (Asari et al. 1999), document segmentation (Sund & Eilertsen 2003), pre-processing of a neural-network classifier for hardwood lo inspection using CT images (Li et al. 1996), low cost in process gauging system in removing illumination dependencies well(Miller et al. 1998), real-time segmentation of images with complex background environment and segmentation of moving lips for speech recognition (Broun et al. 2002).

According to the theory, it lets the data points of a given signal be represented in L different levels [1, 2, …, L]. The number of data points at level i is denoted by n_i and the total number of data points by $N = n_1 + n_2 + \cdots n_L$. And the probability distribution of the signal is:

$$p_i = \frac{n_i}{N}, \quad p_i \geq 0, \sum_{i=1}^{L} p_i = 1.$$

Then it supposes that the signal is dichotomized into two classes C_0 and C_1 (noise and heartbeat) by a threshold at value K; C_0 denotes data with levels [1, …, k], while C_1 denotes data with levels [k+1, …, L]. The probabilities of class occurrence and class mean levels are:

$$\omega_0 = \Pr(C_0) = \sum_{i=1}^{k} p_i = \omega(k)$$
$$\omega_1 = \Pr(C_1) = \textstyle\sum_{i=k+1}^{L} p_i = 1 - \omega(k)$$

And

$$\mu_0 = \sum_{i=1}^{k} i\Pr(i/C_0) = \sum_{i=1}^{k} ip_i/\omega_0 = \mu(k)/\,\omega(k)$$
$$\mu_1 = \sum_{i=k+1}^{L} i\Pr(i/C_1) = \sum_{i=k+1}^{L} ip_i/\omega_1 = \frac{\mu_T - \mu(k)}{1 - \omega(k)}$$

Where the zeroth- and first- order cumulative moments to the k level respectively:

$$\omega(k) = \sum_{i=1}^{k} p_i \ , \mu(k) = \sum_{i=1}^{k} ip_i \text{ and } \mu_T = \mu(L) = \sum_{i=1}^{L} ip_i$$

μ_T is total mean level of the original picture. The following relationship is existed for any k:

$$\mu_0\omega_0 + \mu_1\omega_1 = \mu_T, \qquad \omega_0 + \omega_1 = 1$$

The class variances require second-order cumulative moments, are given by

$$\sigma_0^2 = \sum_{i=1}^{k}(i-\mu_0)^2 Pr(i/C_0) = \sum_{i=1}^{k}(i-\mu_0)^2 p_i/\omega_0$$

$$\sigma_1^2 = \sum_{i=k+1}^{L}(i-\mu_1)^2 Pr(i/C_1) = \sum_{i=k+1}^{L}(i-\mu_1)^2 p_i/\omega_1$$

Then the within-class variance, the between-class variance, and the total variance of levels are described as:

$$\sigma_W^2 = \sigma_0^2\omega_0 + \sigma_1^2\omega_1$$

$$\sigma_B^2 = \omega_0(\mu_0-\mu_T)^2 + \omega_1(\mu_1-\mu_T)^2 = \omega_0\omega_1(\mu_1-\mu_0)^2$$

$$\text{And} \quad \sigma_T^2 = \textstyle\sum_{i=1}^{L}(i-\mu_T)^2 p_i$$

And the following basic relation always holds:

$$\sigma_W^2 + \sigma_B^2 = \sigma_T^2$$

Which σ_W^2 and σ_B^2 are functions of threshold level k, but σ_T^2 is independent of k; and σ_W^2 is based on the second-order statistics (class variance), while σ_B^2 is based on the first-order statistics (class means).

In order to evaluate the following discriminant criterion measures (or measures of class separability) used in the discriminant analysis:

$$\lambda = \frac{\sigma_B^2}{\sigma_W^2}, \qquad \kappa = \frac{\sigma_T^2}{\sigma_W^2}, \qquad \eta = \frac{\sigma_B^2}{\sigma_T^2}$$

The discriminant criteria maximizes λ, κ, and η respectively, and equivalent to one another. Because $\kappa = \lambda + 1$ and $\eta = \frac{\sigma_B^2}{\sigma_W^2}$. Therefore, η is the simplest measure with respect to k. Thus η is regarded as the criterion measure to evaluate the separability of the threshold at level k. The optimal threshold k that maximizes η, or equivalently maximizes σ_B^2.

$$\sigma_B^2 = \frac{[\mu_T \omega(k) - \mu(k)]^2}{\omega(k)[1-\omega(k)]}$$

And the optimal threshold k is:

$$\sigma_B^2 = \max_{1 \le k \prec L} \sigma_B^2(k)$$

A method to select a suitable automatically from a certain level of heart sound record in frequency domain comes from the concept of discriminant analysis. The issue of evaluating the appropriation of the threshold is the key point. The optimal threshold(or multiple thresholds) is selected by the discriminant criterion and maximizing the between class variance(Otsu, N. 1979). Thus the BCVC (Between Class Variance Computation) of Otsu's method is seemed to meet the high-speed requirements.

3.2 Otsu's methods' advantage and applications

This proposed method is named by its nonparametric and unsupervised nature of threshold selection and has the following desirable advantages.

1) The procedure is very simple; only the zeroth and the first order cumulative moments are utilized. That means the higher order statistics calculation is avoided.
2) A straightforward extension to multi-thresholding problems is feasible by virtue of the criterion of the based method.
3) An optimal threshold (or set of thresholds) is selected automatically and stably, not based on the differentiation, but on the integration of the sound level.
4) Further important aspects can also be analyzed (evaluation of class separability, class mean levels or the computation of between class variance)
5) The method is quite general and covers a wide range of unsupervised decision procedure.

Taking the above benefit points into account, the Otsu's method suggested in this stage may be recommended as the most simple and standard one for the adaptive threshold level selection method that can be applied to various practical problems.

Chapter 4 Research Methodology and Design

In this Chapter, an appropriate research methodology for the study will be discussed. First, it reviews the research paradigms in Information System (IS) and compares their difference. Then the research approach and methodology applied for this study, which attempts to explain the research question posed in previous chapter, would be selected and determined. The research problem focusing on the noising reduction of heart sound record from digital stethoscopes in the context of tele-health, would be explored by three sub- questions. The most suitable research strategy and approach for this study would be identified in this chapter though the comparison and selection from the research paradigms in IS.

In section 4.1, a brief summary of some of the relevant concepts in research philosophy and paradigms are presented. In the next section of this chapter the research paradigm, as well as the research epistemologies and ontologies in this research study, is introduced. After that, the positivist and interpretivist research paradigms are also discussed in this section and an appropriate philosophy is selected as a basis for this study. Qualitative quantitative and experiment research methods are discussed against each other in section 4.3 and the most appropriate research approach to answer the research question is presented in section 4.3.1.

4.1 Introduction to research

Research is regarded as something very abstract and complicated. While the different parts or phases of a research project or research study is confirmed then fitted together, it's not nearly as complicated as it seems to be at first glance. A research study has a relatively-fixed structure -- a beginning, middle and end. The major components or parts of a research study are guided by different research philosophies. The way in which research is conducted may be conceived of in terms of the research philosophy subscribed to, the research strategy employed and the research instruments utilised to fulfil a goal, or the research objectives, and the quest for the solution of a problem -- the research question.

When undertaking research of this nature, it is important to consider different research paradigms and matters of ontology and epistemology. Since these parameters describe perceptions, beliefs, assumptions and the nature of reality and truth (knowledge of that reality), they can influence the way in which the research is undertaken, from design through to conclusions, and it is therefore important to understand and discuss these aspects in order to approaches the nature and aims of the particular inquiry are adopted, and to ensure that researcher biases are understood, exposed, and minimised.

As the research philosophy is considered about the way in which data about a phenomenon should be gathered, analysed and used, the term epistemology (what is known to be true) as opposed to ontology (what is believed to be true) are introduced to the various philosophies of research approach. The purpose of science, then, is the process of transforming things believed into things known. Two major research philosophies have been identified in the Western tradition of science, namely positivist (sometimes called scientific) and interpretivist (also known as antipositivist) (Galliers & Sutherland 1991).

There are different research paradigms operating in information systems (IS) research. One well-known differentiation is made by (Orlikowski & Robey 1991). They describe three different “research epistemologies (what is known to be true)” in IS research which have been identified in the western tradition of science, named positivist, interpretive and critical approaches. They follow an earlier division made by (Fluhr et al. 1986). This division of IS research approach has been acknowledged by several other scholars (Myers & Avison 2002) in their introduction to an anthology of qualitative IS research. Several scholars have written about interpretive research and made this in contrast to positivist research (Myers & Avison 1997; Walsham 2006). The main competing research paradigms in IS seem to be positivism and interpretivism.

4.2 Critical theory, positivism and interpretivism

4.2.1 Interpretivism

Interpretivists contend that only through the subjective interpretation of and intervention in reality can that reality be fully understood (Walsham 1995). The study of phenomena in their natural environment is the key to the interpretivist philosophy, together with the acknowledgement that scientists cannot avoid affecting those phenomena they study. They admit that there may be many interpretations of reality, but maintain that these interpretations are in themselves a part of the scientific knowledge they are pursuing. Interpretivism has a tradition that is no less glorious than that of positivism, nor is it shorter.

Wilhelm Dilthey in the mid-twentieth century was influential in the interpretivist paradigm or hermeneutic approach as he highlighted that the subject matter investigated by the natural sciences is different to the social sciences, where human beings as opposed to inanimate objects can interpret the environment and themselves (Rhodes et al. 2010). In contemporary research practice, this means that there is an acknowledgement that facts and values cannot be separated and that understanding is inevitably prejudiced because it is situated in terms of the individual and the event (Cousin 2005; Elliott & LUKEŠ 2008). Researchers recognise that all participants involved, including the researcher, bring their own unique interpretations of the world or construction of the situation to the research and the researcher needs to be open to the attitudes and values of the participants or, more actively, suspend prior cultural assumptions (Conway 2009). These principles are particularly important in ethnographic methodology (Elliott & LUKEŠ 2008; Somekh & Lewin 2005). Some interpretivist researchers also take a social constructivist approach, initiated by Lev Vygotzky (also around the mid-twentieth century), and focus on the social, collaborative process of bringing about meaning and knowledge (Kell & Oliver 2004). The case study research methodology is suited to this approach (Elliott & LUKEŠ 2008; Somekh & Lewin 2005). Interpretivist research methods include focus groups, interviews, and research diaries

One of the criticisms of interpretivism is that it does not allow for generalisations because it encourages the study of a small number of cases that do not apply to the whole population (Arnold 1970). However, others have argued that the detail and effort involved in interpretive inquiry allows researchers to gain insight into particular events as well as a range of perspectives that may not have come to light without that scrutiny (Macdonald et al. 2002; McMurray et al. 2004). More detailed information of interpretivism is provided in the following content.

4.2.2 Positivism

In positivists' mind, reality does exist, stable and can be observed and described from an objective viewpoint (Lin 1998; Smith et al. 1996) without any influence on the phenomena being studied. They suggested that phenomena should be isolated and that observations should be repeatable. This often involves manipulation of reality with variations in only a single independent variable so as to identify regularities in, and to form relationships between, some of the constituent elements of the social world.

In the nineteenth century, positivism began with Auguste Comte (Lather 2006) and insisted that a deterministic and empiricist philosophy, which generates determine effects, and aims to directly observe, quantitatively measure and objectively predict relationships between variables (Conway 2009). It assumes that social phenomena, like objects in natural science, can be treated in the same way. The viewpoint of this presentation is that positivism should be regarded as one of the tools in the arsenal of a researcher. Arguably, recognizing science as a problem-solving, positivism itself (if it could be separated from the people who articulate or follow this position) would probably support this position.

The positivist paradigm has the following positions with regard to the three dimensions:

1. An objective reality is assumed which can be systematically and rationally investigated through empirical investigation, and is driven by general causal laws that apply to social behaviour. This is sometimes called naïve realism (the ontological position) (Guba and Lincoln 1994).

2. The researcher and the phenomena being investigated are assumed to be independent, and the researcher remains detached, neutral and objective (Shanks 2007).

3. General theories are used to generate propositions that are operationalised as hypotheses and subjected to empirical testing that is replicable. Hypotheses should be testable and provide the opportunity for confirmation and falsification. This is the essence of the scientific method (Rubinstein 1975).

One major criticism of positivism is the issue of separating the researcher from what is being researched. The expectation that a researcher can observe without allowing values or interests interfering is arguably impossible (Somekh & Lewin 2005). As a result, positivism today, also known as post-positivism, acknowledges that, even though absolute truth cannot be established, there are knowledge claims that are still valid in that they can be logically inferred; we should not resort to epistemological theory or relativism (Hammersley, n.d.). Positivist research methods include experiments and tests, that is, particularly those methods that can be controlled, measured and used to support a hypothesis.

4.2.3 Critical research

Critical researchers assume that social reality is historically constituted (Boland & Hirschheim 1992) and that it is produced and reproduced by people. Although people can consciously pretend or perform to change their social and economic circumstances, critical researchers recognize that their ability to do so is constrained by various forms of social, cultural and political domination. The main task of critical research is seen as being one of social criticism, which includes the restrictive and separated conditions. Critical research focuses on the oppositions, conflicts and contradictions in society (Parker 2002) from the same period, and devotes to be patulous. It should help to eliminate the causes of separation and domination.

A growing number of information systems (IS) researchers lay claims to adopting a critical research perspective. They state this on their web pages, gather at conferences dedicated to critical research and publish work where their declared aim is to challenge dominant discourses. These are key epistemological and methodological questions for any research perspective and, unsurprisingly, the answers that researchers in information systems have offered to them have changed over time. Specifically, IS researchers have declared what they mean by a critical perspective (Walsham 2005), identified particular critical theories and concepts as promising ones for the discipline (Avgerou 2002). Furthermore, numerous calls since the 1970s for research that adopts a critical, reflexive stance on the interrelationships among information systems, organizations and society (Baskerville & Wood-Harper 1996; Mingers 2003; Walsham 2005).

4.3 Research strategy

It has often been observed (Benbasat et al. 1987) that no single research methodology is intrinsically better than any others, many authors calling for a combination of research methods in order to improve the quality of research (Kaplan & Duchon 1988). Equally, some institutions have tended to adopt a certain specific methodology (Galliers & Sutherland 1991); this seems to be almost ignored the fact that, given the richness and complexity of the real world, a methodology best suited to the problem under certain consideration, as well as the objectives of the researcher, should be chosen(Pervan 2007). In this research, we have tried to avoid the insistence on using a single research method. This is due to the ability to decide between the various merits and demerits of the various methods. Instead, we believe that all methods are valuable if used appropriately, that research can include elements of both the positivist and interpretivist approaches, if applied carefully.

A number of research methodologies have been identified in Table 4.1 below. The list of the methodologies were identified by Galliers (1991), indicating whether they typically conform to the positivist or interpretivist paradigms has been shown the methodologies applied in this research. The key features of the key methodologies have been summarised in the table, identifying their respective strengths and weaknesses. In the following sections, the methodology for this study will be selected and explained how they both operate and interoperate in this research.

Table 4.1 A list of Research Methodologies

Positivist	Interpretivist
Laboratory Experiments	Case Studies
Action Research	Interview
Surveys	Focus group
Case Studies	Research diary
Simulation	

Case study is an intensive analysis of an individual unit (e.g., a person, group, or event) stressing developmental factors in relation to context. The case study is common in social sciences and life sciences. Case studies may be descriptive or explanatory. The latter type is used to explore causation in order to find underlying principles. They may be prospective (in which criteria are established and cases fitting the criteria are included as they become available) or retrospective (in which criteria are established for selecting cases from historical records for inclusion in the study).

Focus group is a form of qualitative research in which a group of people are asked about their perceptions, opinions, beliefs, and attitudes towards a product, service, concept, advertisement, idea, or packaging. Questions are asked in an interactive group setting where participants are free to talk with other group members. The first focus groups were created at the Bureau of Applied Social Research in the USA, by associate director, sociologist Robert K. Merton. The term itself was coined by psychologist and marketing expert Ernest Dichter.

An interview is a conversation between two people (the interviewer and the interviewee) where questions are asked by the interviewer to obtain information from the interviewee. The qualitative research interview seeks to describe and the meanings of central themes in the life world of the subjects. The main task in interviewing is to understand the meaning of what the interviewees say(Dilley 2004).

Research diary is a written record of the researcher's activities, thoughts and feelings throughout the research process from design, through data collection and analysis to writing and presenting the study. The research diary is many things to many people(Lee, R. & Fielding 1996). Some researchers may use a diary to record factual items such as contact numbers of key informants or reasons for changes to the research protocol. Others use it more prolifically to record analytical, conceptual or methodological ideas. Others still will be more inclined to use their research diary to express emotions, perhaps their concerns or delights throughout the study.

One of the criticisms of interpretivism is that it does not allow for generalisations because it encourages the study of a small number of cases that do not apply to the whole population (Hammersley 2012). However, others have argued that the detail and effort involved in interpretive inquiry allows researchers to gain insight into particular events as well as a range of perspectives that may not have come to light without that supervision (Mackrell & Nielsen 2007; McGregor & Murnane 2010; Medcalf 2010). In this research, the noise reduction need to be applied to all kinds of the stethoscopes and should be effective in most cases of the environment. The function of the de-noising method should not be verified much according to the changeable surroundings. Thus the interpretivism, encourages the study of a small number of cases while not apply to the whole population is not suitable for this study.

Quantitative research methods were originally developed in the natural sciences to study natural phenomena. Quantitative research is a research method that relies less on interviews, observations, small number of questionnaires, focus groups, subjective reports and case studies but is much more focused on the collection and analysis of numerical data and statistics. Examples of quantitative methods now well accepted in the social sciences include survey methods, laboratory experiments, formal methods (e.g. econometrics) and numerical methods such as mathematical modelling (Straub et al. 2005).

Qualitative research methods are various as various philosophical perspectives can inform qualitative research. A research method is a strategy of inquiry which moves from the underlying philosophical assumptions to research design and data collection. The choice of research method influences the way in which the researcher collects data. Specific research methods also imply different skills, assumptions and research practices. The four research methods that will be discussed later.

Laboratory experiments permits the researcher to identify precise relationships between a small number of variable (Benbasat 1988). These relationships are studied intensively through a designed laboratory environment and using quantitative analytical techniques with a view to making general statements which can be applied to real-life situations. The key weakness of laboratory experiments is the "limited extent to which identified relationships exist in the real world due to oversimplification of the experimental situation and the isolation of such situations from most of the variables that are found in the real world" (Galliers, 1991, p.150).

The ways in which the experiments presented above is different from the setting in which they are conducted(Latham et al. 1988). Some are laboratory experiments that take place in a setting created by researchers, and others such as field experiments are conducted in participants' natural setting. Additional ways for communication researchers to conduct their studies would be research questionnaires which ask participants to write their answers to questions researchers pose and panel studies which are surveys in which responses from the same people are obtained to learn how their beliefs, attitudes, and/or behaviours change. There are particular strengths and weaknesses of each type of experiment done.

Laboratory experiments are perceived as the most 'scientific' method of obtaining data within Psychology(Maxwell & Delaney 2004). This is due to the fact that generally, most lab experiments are in a controlled setting, which means that the researcher can easily manipulate the independent variable and record the dependent variable (Casebeer & Verhoef 1997). This is the strength of Lab experiments, due to the fact that if something is classified as 'scientific' it is taken more seriously. Therefore, this method is an easy, relatively quick, and scientific way of researchers to collect data.

On the other hand, although there are many strengths of using a laboratory experiment, there are also weaknesses. The ecological validity of laboratory experiments is a huge criticism(Benton et al. 2007). Lab experiments are extremely artificial settings, which leads to the problem of the lack of ecological validity. Due to the fact that participants are not in a real life situation, this may encourage artificial behavior. An example of where lack of ecological validity may affect participant's behavior is in Milgram's obedience study. As many of you will know, in this experiment, he got participants to believe they were potentially harming a confederate by giving them electric shocks. The majority of participants carried on inflicting the shocks even when they were lead to believe that the confederate was suffering a lot from the shocks or may have even died. This suggests that the participants may have acted differently to how they would in this situation in a real life scenario, due to the fact that the setting was extremely artificial. This makes us question whether the results gained from laboratory experiments are in fact reliable, as the behaviour being observed isn't natural behavior due to the artificial setting in the laboratory.

Simulation involves copying the behaviour of a system. Simulation is used in situations where it would be difficult normally to solve problems analytically and typically involves the introduction of random variables. As with experimental forms of research, it is difficult to make a simulation sufficiently realistic so that it resembles real world events(Swope et al. 1982).

Simulation is a very flexible modelling approach, which makes it one of the most widely used Operational Research techniques. The approach taken is to model the behaviour of individual elements within the system, often using random sampling to generate realistic variability. The overall behaviour of the system emerges from the interactions between the elements.

The simulation process consists of problem definition, conceptual modelling, model coding, model verification and validation, experimentation and analysis of results, and solution implementation. Application areas for simulation in industry include manufacturing, call centres, business processes, service operations, military, transport, health care, IT, and environment. There are also applications in many other areas of science.

Action research is a form of applied research where the researcher attempts to develop results or a solution that is of practical value to the people with whom the research is working, and at the same time developing theoretical knowledge. Through direct intervention in problems, the researcher aims to create practical, often emancipatory, outcomes while also aiming to reinform existing theory in the domain studied. As with case studies, action research is usually restricted to a single organisation making it difficult to generalise findings, while different researchers may interpret events differently. The personal ethics of the researcher are critical, since the opportunity for direct researcher intervention is always present(Argyris et al. 1985). The main weakness of action research is such method often lends itself to small-scale studies and is time-consuming. That means it is not suitable for the real-time research or research that need the instant results. The value of this a methodology is that it provides a powerful means of improving and enhancing practice.

It is important to realize the different philosophies of research because that would enable the researchers to conduct the correct decision about research design. Actually, the philosophy is the overall configuration of research, for example, what kind of evidence to be collected and where the evidence can be gathered, how such evidence is interpreted in order to provide good answers to the research questions. Furthermore, it would help you to consider whether the above research approaches are effective. For example, if one is interested in knowing why something is happening then interpretivism is more appropriate than positivism. Knowledge of different research traditions enables you to adapt your research design to cater for the constraints as well. These could be practical (e.g., not be able to the access to interviews) or they could arise from a lack of prior knowledge of the subject. One cannot be in a position to construct a hypothesis if the realization of the topic is not sufficient.

Positivism emphasizes the importance of an objective scientific method. These researchers prove the theories through collecting facts and then studying the relationship between those facts. They analyse quantitative data using statistically valid techniques and produce quantifiable and generalized conclusions. Positivism stresses the importance of studying social and organizational realities in a scientific way that mirrors the research processes used in the natural sciences. However, interpretivism is concerned to understand human's perceptions of the world. Interpretivists see facts as the product of human interactions – they are the product of shared understandings and meanings and are not always predictable. The less quantifiable and the subjective interpretations, reasoning, and feelings of humans are seen as a more relevant line of enquiry in order to understand and explain the phenomena. Therefore, the focus of interpretivism is not numbers but on words.

4.4 Research model and instrument

In this research, the research objectives focus on finding the appropriate noise reduction method for the sound record of stethoscope. To achieve the research target, a lot of data about the sound resources would be collected, analysed and compared before the final noise reduction solution can be achieved. Thus, interpretivism, the study of subjective interpretation of reality and phenomena in their natural environment, seems not suitable for this study.

In this study, when the research data, the sound resources from stethoscopes, has been collected, some factors are required to be considered. The volume and quality of the sound resources differ from each specific medical device, the styles of noise change according to the environment, the speed and strength of heartbeat is depended on individual's body conditions. Each factor is variable during the collection of the sound resources and able to influence the final sound quality dependently. To study how those issues affected the sound quality and then design the probable de-noising resolution, the measurement and control of each variety is essential. Since it will be concerned the affection from each variable and a certain designed situation would be applied to explore the relationship between each variable and the result. Then the possible solution, the laboratory experiment research technique will be provided.

The laboratory experiment which has been identified above is the study involving intervention by the researcher beyond that required for measurement. The usual intervention is to manipulate some variable in a setting and observe how it affects the subjects being studied. In this research, the variation of each issue (e.g. the volume of the sound, the kind of background noises, the change of heart rate) is considered to be a variable. As the foremost advantage of experiment is the researcher's ability to manipulate the independent variable, each variable is regarded as one independent variable (IV) and their function of influence the whole quality of sound quality will be researched. As all the co-factors affect the sound quality together, we need to evaluate the function of each one and design the probable solution. The contamination from extraneous variables can be controlled more effectively than that in other designs. This helps us isolate experimental variables and then evaluate their impact over time. Further, after finish designing the noise reduction method, the lab experiment technique helps us to repeat the experimental results with different subject groups and conditions, for example using extra sound resources.

The artificiality of the laboratory is arguably the primary disadvantage of the experiment method, while many subject's perception of a contrived environment can be improved by investment the facility. In this research, the co-affection of all the factors will be considered and tested when the final noise reduction method is determined. Thus the limitation of contrived environment is minimized.

In this study, the influence of each variable needs to be confirmed. While in the reality, some certain variable always cooperate together and it is quite hard to collect the sound resource which is affected by only one variable. The result of affection from certain factor needs to be generated and form one system. Then the result/behaviour of each system will be put together to simulate the corruption of sound quality in real life. Thus the simulation research technique is considered to be applied in this research as well. The simulation is hardly to make a factor sufficiently realistic while it can resemble some variables of the real world events which cannot occur independently. As a very flexible modelling approach, simulation makes it easier to be applied with the laboratory experiment research.

4.5 Research facilitation software

MATLAB 2010b is employed in this research as a signal processing and simulation software. Compared with other numerical or data analysis software, MATLAB proves by a number of considerations. MATLAB is, arguably, the most widely used program for performing numerical calculations. It comes with its own programming language, in which numerical algorithms can be implemented.

MATLAB (matrix laboratory) is a numerical computing environment and fourth-generation programming language. Developed by MathWorks, MATLAB allows matrix manipulations, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs written in other languages, including C, C++, Java, and Fortran.

The selected software has been widely used in signal processing areas. Although MATLAB is intended primarily for numerical computing, an optional toolbox uses the Mu PAD symbolic engine, allowing access to symbolic computing capabilities. An additional package, Simulink, adds graphical multi-domain simulation and Model-Based Design for dynamic and embedded systems. In 2004, MATLAB had around one million users across industry and academia. MATLAB users come from various backgrounds of engineering, science, and economics. MATLAB is widely used in academic and research institutions as well as industrial enterprises.

In this research, all the sound resources will be stored in hard disk with WAV form. MATLAB is used to conduct the following tasks, to read the digital signal record stored in the hard disk, to cut the original signal into certain fragments, to analyse the noise corruption and to apply the de-nosing signal processing with the appropriate noise reduction method we provide, and to output the processed signal with WAV form again.

4.6 Design of laboratory experiments

Experimental flow graph:

1. • **Load sound files into matlab**
2. • **Noise reduction for different environments?**
3. • **Discuss about possible threshold option**
4. • **The plan to run a pilot sound track**

1. Load sound files into Matlab

The heart sound examples of electronic stethoscope will be acquired from existing and other available electronic stethoscopes. During the measurement, system parameters of these heart sounds are set at the following conditions: 44.1 KHz sampling frequency (sanple rate), high sensitivity and about one minute length for each piece of recording data. These heart sounds are stored in PC in the form of MP3. MP3 audio format should be converted into format such as WAV which can be recognized by MATLAB software.

The easiest way to read audio data from a file is to use the Import Wizard from Matlab, a graphical user interface. The Import Wizard can read WAV, AU, or SND files, thus the MP3 files from the electronic stethoscope need to be transferred in to WAV files through file-convert software first. While to import WAV files without invoking a graphical user interface, ‘wavread’ is recommended.

MATLAB audio functions can read and store single-channel audio data in an m-by-1 column vector, and stereo data in an m-by-2 matrix, m is the number of samples. For stereo data, the first column contains the left channel, and the second column contains the right channel.

Typically, each sample is a double-precision value between -1 and 1. In some cases, particularly when the audio hardware does not support high bit depths, audio files store the values as 8-bit or 16-bit integers. The range of the sample values depends on the available number of bits. The MATLAB sound functions (‘wavread’) support only single-or double-precision values between -1 and 1. Other audio functions support multiple data types, as indicated on the function reference pages.

In this experiment, the sound files are two-channel audio data, thus both the left and right channel should be imported and processed in Matlab. After the noise reduction process, experiment, the result of each sound channel will be shoed and the complete processed double-channel sound file will be given.

Before the processing of the noise reduction, the WAV form sound files need to be cut into certain length for the ease of comparison after experiment. In this experiment, the sound length is dependent on the sample rate. The audio signal in a file represents a series of samples that capture the amplitude of the sound over time. The sample rate is the number of discrete samples taken per second and given in hertz. The precision of the samples, measured by the number of bits per sample, depends on the available audio hardware. In this experiment, the sample rate is 44.1 KHz. And the sound files need to be cut into two, four and eight seconds for the processing.

2. Noise reduction for different environments

The medical signals can be corrupted by noise in a variety of ways. The sequence of corruption demonstrates several key areas where external interferences could cause degradation of the original signal. In reality, heart sound records are very often disturbed by various factors, which can prohibit the accuracy of the original sound and most of these factors are noises from sources such as breath sounds, contact of the stethoscope with the skin, fetal heart sounds if the subject is pregnant, and ambient noise that may corrupt the heart sound signals.

To make it easier, these factors can be categorized as two aspects in the mass: external factors and internal factors.

The external factors include: Small movement the stethoscope, ambient noise, instrument noises, human voices and patient movements; while the internal factors include: respiration sounds or breathing noise and acoustic damping through the bones and tissues.

Currently, there is no way of knowing a priori what the particular noise component is, or of determining the noise component once the measurement has been recorded. In every case and situation, the noise will be different. Thus one of the probable ways to make the de-noising method suitable for different environments is to utilize the adaptive noise reduction method which can be adjust automatic due to the different or changeable noise background. The details of choosing reduction method will be discussed in the next segment.

3. The possible threshold option

There are four threshold selection rules available in MATLAB, Rigrsure, Sqtwolog, Heursure and Minimaxi. The third threshold is just constituted by the first two according to the signal-to-noise ratio (SNR) of the signal. The introduction of these thresholds has been provided in Literature Review Chapter.
As these thresholds are designed for all kinds of signals, none of them performs well in heart sound processing particularly because they are all fixed thresholds while the heart

sound and noises are mutable or sometimes random. In addition, the real environment where the sound is collected is always changeable. In this research, it is supposed to design a new adaptive threshold method for the noise reduction for heart sound. This threshold can be used to adjust the signal due to the SNR of the original heart sound. When the SNR is large (means little noise contained in the heart sound signal), the threshold value will be low so that more useful information will be maintained; while the threshold value would be relatively high to remove more noise when the SNR seems to be small. The probable threshold is seemed to be the Otsu's method which can be seen in literature review chapter.

4. The plan to run a pilot sound track

The new de-noising method will be test by two different ways by using the empirical signal and real heart sound signal respectively.

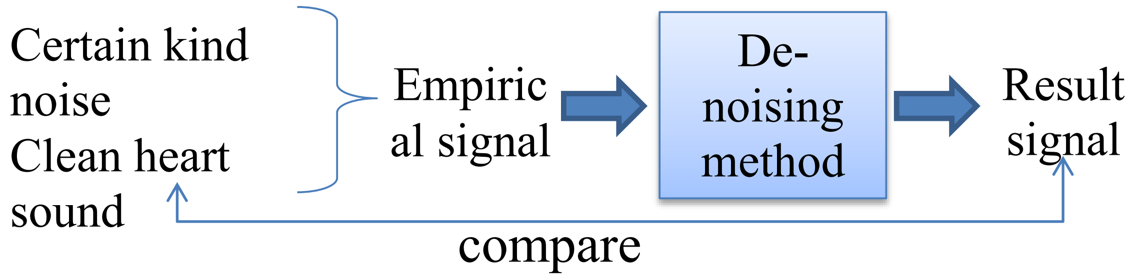


Fig 4.1 de-noising method tested with empirical signal

The method will be first tested by the empirical signal, which is consisted of certain kind of noise (Gaussian noise, Friction noise or breathing noise) and clean heart sound with limited noises. The output signal from the de-noising method will be compared with the clean heart sound. The evaluation criterion called 'mean square error' (MSE) would be introduced in this step and compare the similarity between them. MSE is a risk function, corresponding to the expected value of the squared error loss or quadratic loss. MSE calculates the average of the square of the "error", which is the amount of the difference between the estimator and the estimated quantity(Lehmann & Casella 1998). The MSE of the signal estimator can be obtained by the following expression:

$$\mathrm{MSE}(s) = \frac{\sum_{i=1}^{n}(s - s_e)_i^2}{n}$$

Where 'n' means the length of the signal, 's' denoted the original signal and 's_e' is the estimated signal achieved from the de-noising coefficients.

Both the new de-noising algorithms and the common ones will be simulated in MATLAB for de-noising the heart sound signals. MSE is used to evaluate the performance of both the approaches. The de-noising algorithm will prove to be better if the MSE value is smaller.

The final step of the research will test the new de-noising methods with the real heart sound output from the electronic stethoscopes.

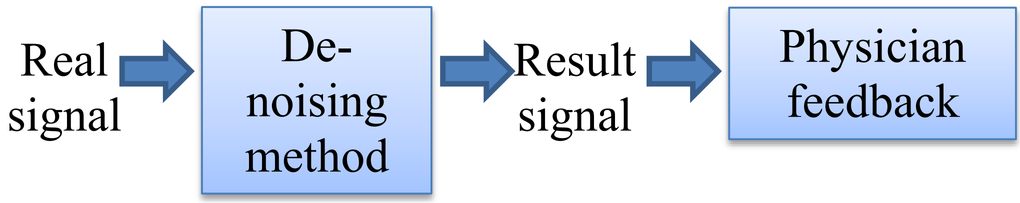


Fig 4.2 De-noising method tested with real heart sound

The raw heart sound will be inputted into the new de-noising method and the output signal will be obtained. The result signal after de-noising as well as the raw heart sound signal will be sent to the doctors and physicians to seek the feedback of the acceptance of the recommended method.

4.7 Conclusion

In this chapter we have presented a detailed account of the research philosophy, strategy and methodology according to which we shall conduct this research. We place our research in the positivist camps and explain why interpreticist theory seems not suitable to this research. We utilise a mixture of laboratory experiment and simulation research approaches. We explain how we propose to interoperate the two positivist methods so as to achieve our research objectives. This includes a substantial literature review, the early research models and the development of an instrument. These are explained in much greater details elsewhere in the thesis and in other published papers. Finally we detail our broad procedures for data analysis and software intervention. More details about the data collection and analysis will be given in 5.4.1 ‘The integration process of the experiment’.

Chapter 5 Data Collection and Analysis

In this chapter, the approaches employed for data collection and data analysis will be provided along with a description of how the data was analysed will be discussed. The process of data collection and analysis for the research is introduced and explained in the following sections. Section 1 defines the format of the data required for the experiment. In Section 2, the real data which has been collected is described. Section 3 and 4 discuss the application of the algorithm applied in the experiment and the adjustment of the coefficient. In section 5, the data is tested with different threshold levels, and the results are showed then.

5.1 Data assumption discussion

In this research, the main objective was to understand various noises mixed with the heart sound and how these can be reduced with a noise reduction method so that a clear heart sound can be presented to physicians. The raw records are restored in the form of MP3 before the conducting experiment. Thus all heart sound records (MP3 form) are stored and saved as digital signals, in other words, in the form of numbers. This helps to analyse the sound in a quantitative manner, which is essential for the experiment method chosen for this study.

Conducting the quantitative data collection, especially for sound samples, is not an easy task as there are certain challenges to be overcome by the researcher. In the following paragraph, these challenges are briefly discussed.

In the Methodology Chapter, the general view of the target data has been described. The data in this research is the real heart sound files which are recorded from existing and other available electronic stethoscopes.

The sound specification needs to be defined when the sound is recorded. This is essential so that clear and precise audible sounds can be extracted. The sample rate of the heart sound record is 44.1 kHz, which is a highly sensitive, widely used and commonly found sampling frequency for analog audio sampling, for example, applied in Compact Discs. The interpretation of this is that the analog audio is recorded by sampling it 44,100 times per second, and then these samples are used to reconstruct the audio signal when playing the recording back.

For the convenience of auscultation, the sound needs to reach both ears during the diagnostic process. This is quite important so that a balanced sound is heard when the auscultation process occurs. Otherwise, people will encounter perceived discomfort. Thus the raw sound records are two channel audio and the sound in each channel is exactly same when it reaches the ears. In this research, the sound from one of the channels is used to process noise reduction and after the experiment, the other channel is copied with the processed sound so that both ears receive the same sound and the computation time is half of the usual processed time.

Each heart sound record should be about one minute long for the sake of the experiment. In this research, the sound file was cut into certain different lengths (2 seconds, 8 seconds etc.) prior to the noise reduction process, and these samples are used to examine the effect of the de-noising algorithm.

The required heart sound records, collected from the real environment for this experiment, consist of two key parts: the pure heart sound and the noise components. The pure heart sound is the heart sound with limited noise corruption which can be used as a reference after the noise reduction process. The heart sound with the noise is the sound which needs to be de-noised because of external noises other than the heart sound. Thus both sounds need to be collected and restored during the data collection.

The heart sounds vary for the people from different age or gender (Barber et al. 1950). For example, the third heart sound (S_3) may be normal in people under 40 years of age and some trained athletes but should disappear before middle age (Drazner et al. 2001). Thus a range of heart sounds from people of different ages and gender should be considered to ensure the reliability of the sound.

The noise is another key issue in the collected data. Noises occur in real life and come with heart sound as described in chapter 2 need to be collected in different samples to understand how heart sound are combined with a range of noises so that these unwanted noises can be de-noised.

The causes of different kinds of noise which may affect the quality of heart sound have been discussed in the Literature Review and Methodology chapter. Both internal and external factors need to be considered in this research. To make it easier, five main kinds of noise have been chosen through this data collection. These noises are: (1) small movement of human or stethoscope, (2) ambient noise, (3) human voices, (4) respiration sounds or breathing noise, and (5) acoustic damping through the bones and tissues.

These noises are briefly introduced below.

(1) Small movement of human or stethoscope: when the stethoscope is utilized on the patient body, the physician need to move the chest piece on the human body to get the heart sound from different part of the body.
(2) Ambient noise: ambient noise or background noise is any sound other than the sound being monitored (primary sound). Ambient noise is a form of noise pollution or interference. For example, if a trolley or equipment is operated in the surroundings, stethoscopes will pick up these noises.
(3) Human voices: there is interaction between the patient and the physician during auscultation procedures. A sensitive digital stethoscope picks this conversation and some noise will be transmitted through the stethoscope.
(4) Respiration sounds or breathing noise: the patients need to take the breath during the recording of the sound. These noises will also travel through the device.
(5) Acoustic damping through the bones and tissues: sound will attenuate when it transmit out from the bones tissues and clothes.

Each noise kind includes different participants. The heart sound collection from each of them will be added with one specific kind of noise. That means, when the data (heart sound record) is collected, pure heart sound from each of them will be recorded first. Then one piece of the designed noise mixed with the heart sound will be recorded. Both the clean and noise-polluted heart sound recorded will be collected for further comparison.

5.2 The collection of data

To perform data collection is a complicated job in a health type research environment. The researcher encountered certain challenges when the data is collected. For example, recruitment of patient was an issue as many ethical approvals need to be obtained. In the next section, the whole process of data collection, both in India and Australia, is described.

The concept electronic stethoscope was introduced in chapter two of this thesis. The data collection process includes three components: the researcher, the physicians and the patients (participants). The data need to be collected in a quiet and isolated location to assure sound quality. The sound quality should consist of clear and pure heart sound and should be contained of specific noise. The research set-up consisted of one electronic stethoscope and one laptop. The heart sounds were collected by the stethoscope operated by an experienced physician. The sound was also transmitted to a recording device in a format that is suitable for analysis. The laptop was used to document the entire session in a spreadsheet file to comply with the ethical approvals.

The number of samples and the participants' details for the experiment will be provided in the description of the experiment in healthcare institutes in India and Australia. All the participants would attend the data collection with each gender. The collection would cover different age groups. In this research the majority of participants are patients from hospitals and health institutes.

The supervisor undertook the role of an observer in this research of data collection exercise. The data were collected in India and in Australia. In India, PSG Hospital was the main data collection centre and the heart signals were collected in the cardiology ward. In Australia, The Prince Charles Hospital was used for this purpose, and the cardiology ward was used for the data collection. In both centres, trained cardiologists used a USQ developed Digital Stethoscope and recorded the data. The supervisor helped with technical aspects, and observed the procedures.

The data collection in India was conducted in PSG hospital located in Coimbatore. PSG hospital Coimbatore is a general hospital. This is a multi-specialty hospital, with 1000 beds. The hospital is a recognised teaching institution. The data collection occurred in the cardiology ward, and facilitated by the medical director, who is a trained cardiologist. The protocol for data collection was developed in advance for the research team to be used in Indian data collection.

Two trained medical professionals, one cardiologist and a nurse participated in the data collection activities. The USQ-designed stethoscope which was introduced in Literature Chapter was used by these two people to collect heart sounds. A total of 42 sound samples were collected and these included heart and lung sounds. The data collection activities were conducted over for days, where the first day was used to test the device. The recruitment strategy included channelling patients from day care units into the cardiology unit. A consent from was given to patients, and the purpose was explained.

The data collection in Australia was conducted in the Prince Charles Hospital. The Prince Charles Hospital provides health services to residents living in the northern suburbs of Brisbane and specialist services to the broader Queensland and northern New South

Wales population with approximately 3500 staff. The USQ-designed stethoscope was given to the users.

It was planned to have one remote telehealth system setting as this experiment and data collection is not only designed for this research but also for the remote area sound transmission. Then the research team prepared two separate rooms at the hospital to build the telehealth platform. The sound records were collected in on room and transmitted to another room through the computer-to-computer network for diagnosis. Participants' observation was fully documented by the main researcher.

The participants from this hospital tested the stethoscope for about half an hour in separate rooms and the sound records from the stethoscope have been restored. The participants were two medical doctors, one nurse and one medical student from Queensland Health. 26 different pieces of sound included the heart and lung sound has been recorded in this experiment.

5.3 Documenting and recording of data

The following flow chart, showed in Fig 5.1, represents the procedure of documenting, processing and analysis of data, and the paragraphs following the flow chat give the details of each step in order.

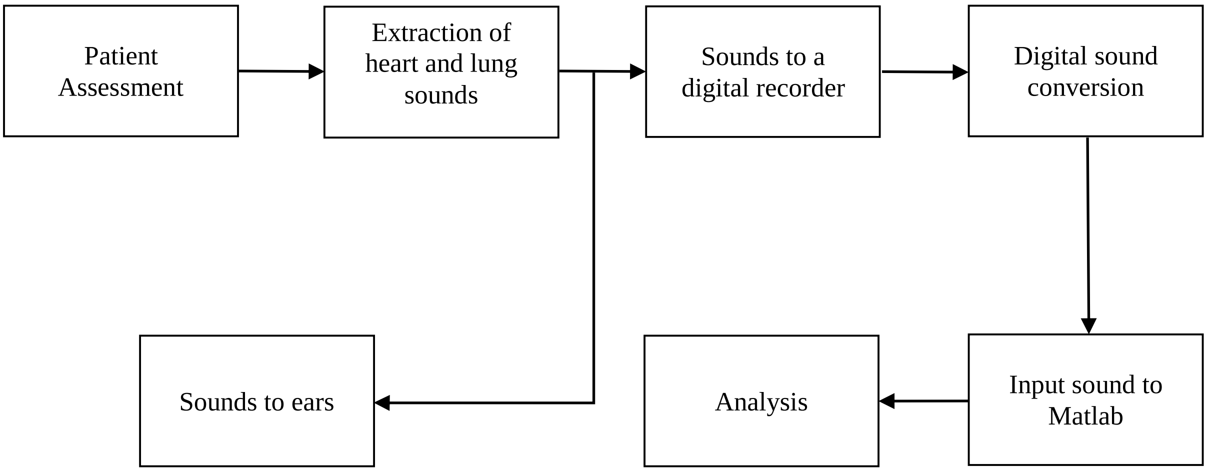


Fig 5.1 Flow diagram of data analysis

The data documenting began with patient assessment, which included the appropriate way to collect the required and qualified sound files through digital stethoscope. Then heart and lung sounds were extracted separately, and these sounds were sent back to the physicians' feedback directly and recorded by digital recorder at the same time. The sound files were converted into suitable format before the signal processing routines were employed in Matlab. All the noise reduction process was accomplished in Matlab and the result would be discussed then.

Patient Assessment

Digital stethoscope technology is fast becoming an accepted popular option, aiding in the diagnostic process of patients. It has a very versatile range of features and applications(ZIN 2011). In this experiment, all the sound files would be collected as target data through the digital stethoscope. Thus before the experiment begins, it is necessary to figure out the patient assessment. For example, how can we use the digital stethoscopes correctly on the chest and back, or how can we overcome the difficulties when auscultate the female's chest and male's hair.

General background information:

1. The stethoscope feature allows digital recording to be made by using a digital stethoscope. The recordings are saved in the memory space of stethoscope and then be transmitted and played back at another location with certain form (e.g. MP3).
2. Digital stethoscope in this experiment requires the use of a USQ—designed stethoscope and equivalent software.
3. The clinical experience in using stethoscope is assumed.
4. The researcher needs to inform the required sound file to the doctors before it is collected.

Stethoscope Operation
Fig 5.2 shows the terminology of a stethoscope and the procedures to use the digital stethoscopes appropriately. Clean off the earpieces before putting the stethoscope into your ears. Then hold the chest piece between your palms to warm it before placing it on a patient's chest. Then place the stethoscope into your ears. Then hold the chest piece in your hand. With the other hand, tap a finger against the chest piece and listen. Grip the chest piece between your middle and index fingers to provide firm contact with the skin. To minimize extraneous noises, avoid touching or rubbing the tubing or chest piece against clothing is necessary. Place the chest piece onto the part of the body you want to listen to. For the auscultation of heart sound, this is a few inches above the left nipple. A steady "lub dub" sound can be detected. This is known as the apical pulse(Werblud 2001).

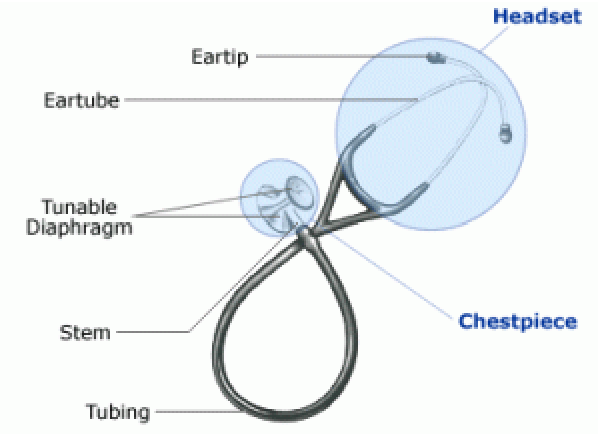


Fig 5.2 Terminology of a stethoscope
(Source: http://www.mystethoscope.com/images/scope_anatomy.gif)

A patient's size and weight can affect the transmission of sound—decreased breath sounds, for instance, may be a result of obesity. The technique may be helpful if the patient has a lot of muscle or adipose tissue that muffles heart and lung sounds. For example, when we were examining a female patient with large breasts, use the free hand to hold the breast up and position the stethoscope head as close to the chest wall as possible; if that isn't helpful, move her onto her left side would work(Markel 2006).

Extraction of heart and lung sounds

Auscultating heart sounds is a fundamental component of physical assessment. When heart sound is obtained through a stethoscope, a physician listens for rate, type, and rhythm of it, as well as any sounds that shouldn't be there (adventitious sounds), such as gallops, murmurs or clicks gallops, murmurs or clicks (Coombs & Moorse 2002).

All hearts noises sound the same at first. But after listening to many hearts, eventually sounds will seem to jump out. For heart sounds, we listen to the four primary areas: left and right of the sternum at the level of the second rib, left of the sternum at the forth rib, and on the left nipple line at the level of the 5th rib. Remember these with the mnemonic "2-2-4-5." The names of the valves that are heard in these locations are: (2 right) aortic, (2 left) pulmonic, (4) tricuspid, (5) mitral.

Assessing lung sounds allows one to identify the rate, rhythm and quality of breathing, any obstructions of the airways, as well as rubs that indicate inflammation of the pleura. Remember that for lung sounds (according to the Bates "Bible,") were listened in six paired areas on the chest, and seven paired areas on the back. Physicians listen to left and right sides at the same level before moving down to the next level – this way they get a side-by-side comparison, and any differences will be more apparent (Traver 1973).

Sound to digital records:

Some electronic stethoscopes, called recording stethoscope, have the feature that their direct audio output can be used and stored with a recording device, such as a laptop or MP3 recorder. The USQ—designed digital stethoscope is one of them. The same connection can be used to listen to the previously-recorded auscultation through the stethoscope headphones or the speaker from laptop or MP3 player. This feature allows more detailed study for general research as well as evaluation and consultation regarding a particular patient's condition and telemedicine, or remote diagnosis.

Digital sound conversion/suitable form

Sound files are collected and then restored in laptop with MP3 form. MP3 form cannot be read by Matlab (the software help to process the noise reduction) directly, those sound files need to be converted in to WAV form in stand. A lot of softwares applications can accomplish this conversion. In this research study, the specific software in sound processing area, named Cool Edit Pro 2.0, would be employed help to finish the conversion. Through this software the shape of sound files can be seen and compared visually. Every piece of MP3 sound file would be import into this software and then save as WAV form before noise reduction. Fig 5.3 shows the interface of Cool Edit Pro 2.0.

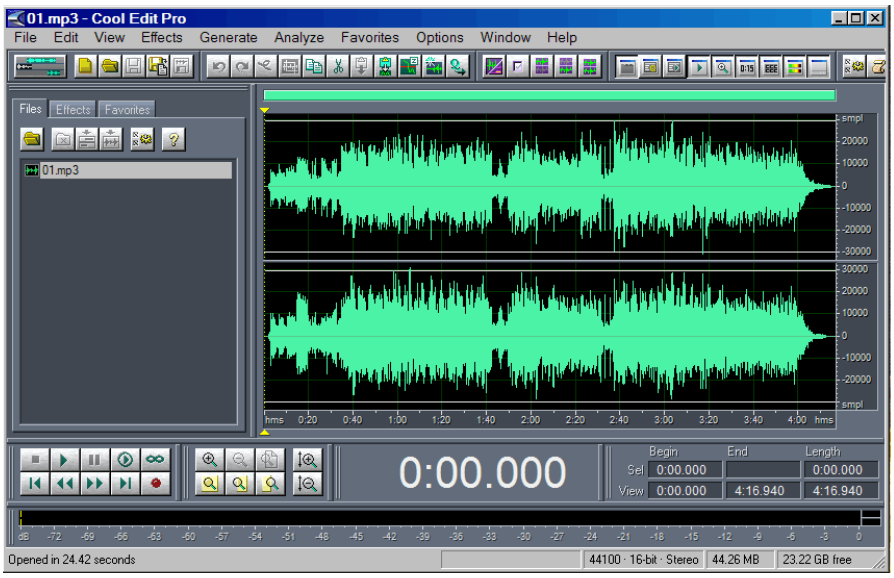


Fig 5.3 The interface of Cool Edit Pro 2.0

Cool Edit Pro 2, current version Adobe Audition CS6, is one of the digital audio workstation from Adobe Systems which offers high-performance, intuitive tools for audio editing, mixing, restoration, and effects. From the Cool Edit Pro interface, MP3 sound files were imported into Cool Edit and the file name showed on the voice list (on the left side of the interface). The waveform view of the double channel sound is displayed in the middle of the window (both left and right channel). The general information of this piece of sound is listed on the right bottom of the interface. The file can be easily converted from MP3 form to WAV form by saving the file in different format, and the software can accomplish the conversion automatically.

Input sound to Matlab (sound look like) analysis

Before the sound files are input into Matlab, they need to be cut into certain length first, thus the processing aspects can be controlled. The length of heart sound depends on how many heart beats included in the sample file. At least two heart beats should be contained in one piece of sound so that the processing can be properly accomplished. As the heart rate of human being is about 60-90Hz, two seconds is the minimal time length. For the sake of the experiment, heart sound with 2s, 4s and 8s are selected to put in to Matlab de-noise processing.

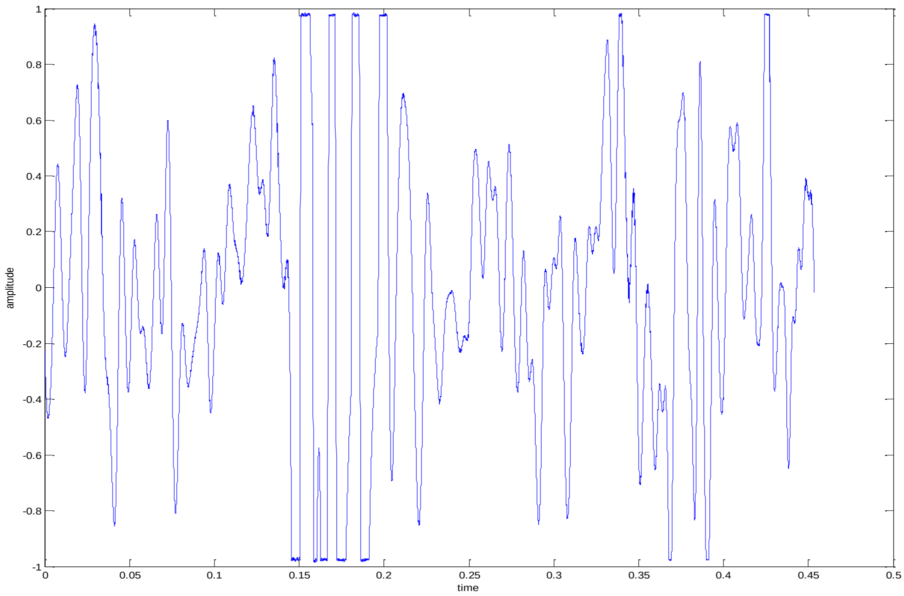
5.4 Data analysis

5.4.1 The integration process of the experiment

In electronics, signal processing, and statistics, both time domain and frequency domain are the domains for analysis of mathematical functions or physical signals with respect to time and frequency respectively (Ferreira 1999). In brief, a time-domain graph shows how a signal changes with time, whereas frequency-domain graph shows how the signal distributes in the given frequency band. In the time domain, the signal's value is real numbers, for the case of continuous time or discrete time. A frequency domain can include information of the frequency components and phase shift that can recover the original time signal. Fig 5.3 and Fig 5.4 are the time and frequency form of a section of heartbeat signal extracted from an existing digital stethoscope. The noise, mixed with the real heart sound, covers the whole time domain in Fig 5.4, thus it is quite improbable to tell the heartbeat from the noise in time domain.

A given signal can be converted between the time and frequency domains with a pair of mathematical operations which named as transforms. For example, the Fourier Transform (FT) which decomposes a function or signal into the sum of a number of sine wave frequency components. The sine wave frequency component, also called the 'spectrum' of frequency component is the way the frequency domain represents the signal. In Fig 5.5, FFT (Fast Fourier Transformation) formula is utilized to transform the time-domain signal into frequency domain. It is detected from Fig 5.5 that the main frequency component of the whole signal is below 1000Hz.

Amplitude



Hz

Fig 5.4 Time domain of a heartbeat signal

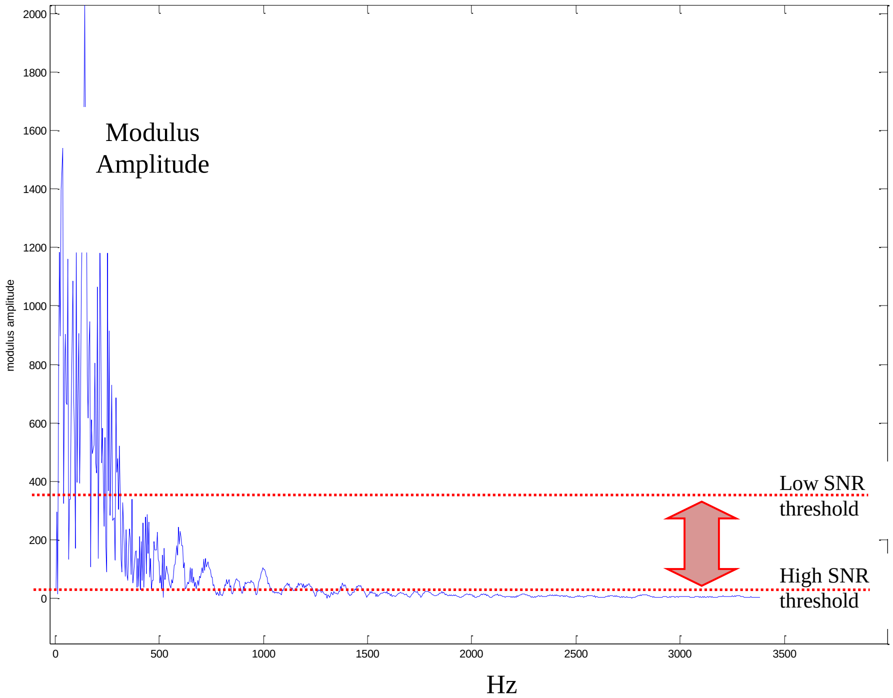


Fig 5.5 Frequency domain of a heartbeat

The frequency components of heart sound are almost below 1000Hz, and the amplitude is relatively higher than that of noise which covers the whole frequency domain. In chapter 2, Otsu's method has been introduced and the self-adaptive threshold separates the heat sound into two parts. The data points of a given signal is represent in L different levels [1,2,…,L]. It assumes that each heart sound signal is dichotomized into two classes, C_0 and C_1. C_0 (denotes data with levels[1,…,k]) represents the noise part and C_1 (denotes data with levels[k+1,…,L]) represents the heartbeat sound part with limited noise in frequency domain.

From the Otsu's theory, the issue of evaluation of the appropriation of the threshold is the key point. From the theory, the optimal threshold is to select the discriminant criterion through maximizing the between class variance σ_B^2, which has been given in details in Chapter 2.

$$\sigma_B^2 = \frac{[\mu_T\omega(k) - \mu(k)]^2}{\omega(k)[1-\omega(k)]}$$

Where $\omega(k)$ is the zeroth- and first- order cumulative moments to the k level (C_0), μ_T is total mean level of the original signal:

$$\omega(k) = \sum_{i=1}^{k} p_i \text{ , } \mu(k) = \sum_{i=1}^{k} ip_i \text{ and } \mu_T = \mu(L) = \sum_{i=1}^{L} ip_i$$

This threshold value can depart the useful heart sound (C_1) from the noise (C_0) in frequency domain. In Fig 5.5, the dotted line depicts the value of the threshold. The amplitude values of frequency components above the threshold are regarded as heart sound part and would be kept; while the values below the threshold are noise part and would be adjusted. The threshold is fluctuated adaptively according to the SNR of the input signal. If the SNR is large which means the heart sound contains limited noise, the threshold value will be altered low to maintain more useful information (Low SNR threshold); while the threshold value would be relatively high so as to remove more noise when the SNR seems to be small(High SNR threshold). The fluctuation of the threshold is shown in Fig 5.5 as well.

The real heart sound records collected from the hospitals are import into the experiment and validate the de-noising effect of the noise reduction method.

5.4.2 Introduction of the process of experiment

In order to validate the performance of the new thresholding function, several sections of the heart sound are required from the output of a digital stethoscope. Those sound sections are extracted from the heart sound collected from the hospitals. One of them is shown in Fig 5.6. These heart sounds are stored in PC in the form of MP3 and converted into format of WAV which can be read by MATLAB software. The whole noise reduction progress was conducted in MATLAB.

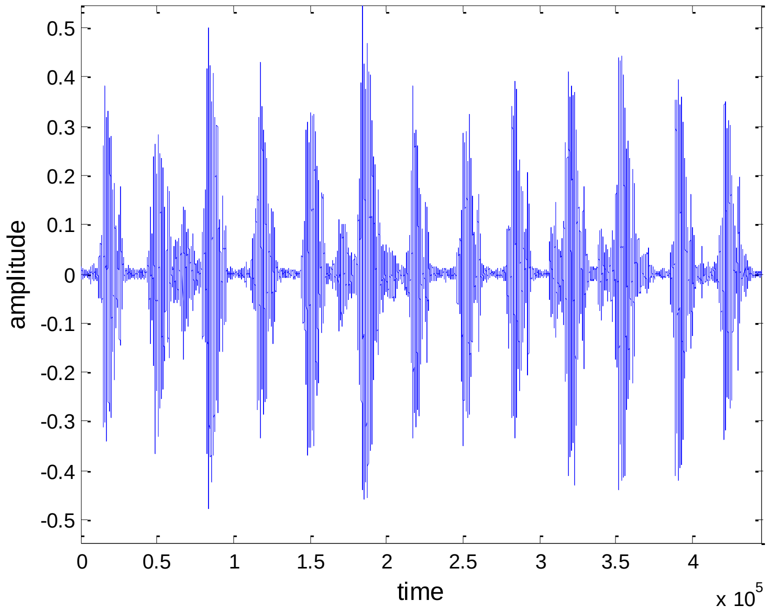


Fig 5.6 One section of the heart sound

A piece of heart sound with two whole heart beats, is extracted and shown in Fig 5.7. In Fig 5.8, the frequency domain of the heartbeat through FFT is depicted. The main frequency component of heart sounds and murmurs is in a range from 0.1Hz to 1000Hz, and some limited frequency component (include part of the murmur and noise) lies above 1000Hz. Then the frequency band over a range of 0 to 2000Hz is drawn in Fig 5.8. Thus the overall distribution of the frequency component of heart sound can be seen in Fig 5.8.

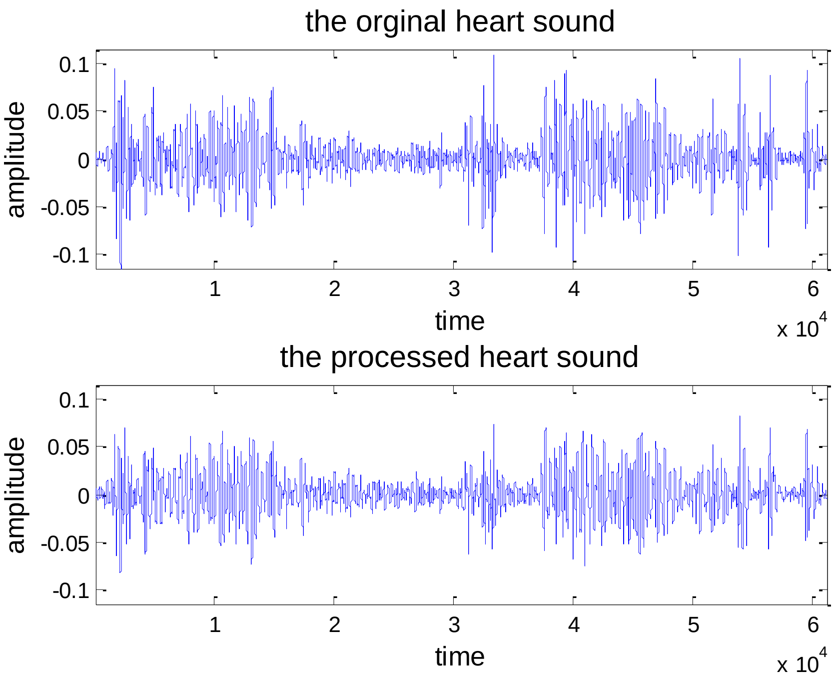


Fig 5.7 Comparison between original heart sound and processed heart sound

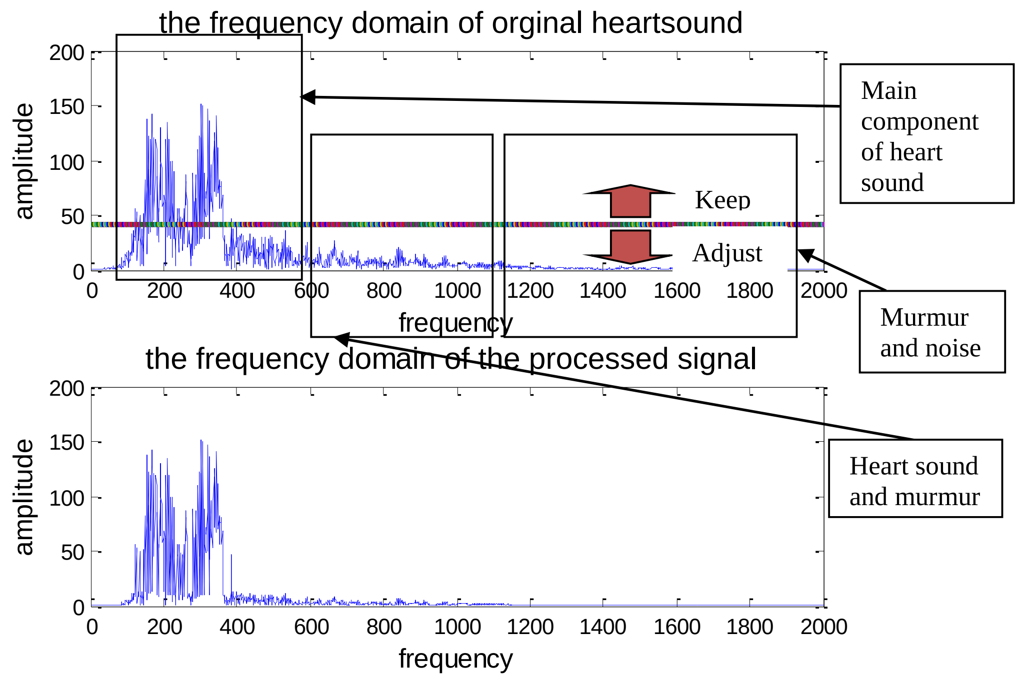


Fig 5.8 Comparison between the frequency of the heart sounds

Then the optimal threshold of Otsu's method, which can be adjusted according to the input data, is applied in the frequency domain of the signal. In this case, the maximum value of the data in frequency domain is 150.78. The threshold value is set as a positive integer, the data value range is from 1 to 151 and all the data are re-distributed and accounted to their nearest round values. As the threshold value is set to be an integer, 150 possible threshold values (from 1 to 150) have been calculated to find the maximum σ_B^2.

In Fig 5.9, each possible threshold value is calculated to get the corresponding value σ_B^2. The algorithm to calculate σ_B^2 is listed above. Every value of σ_B^2 corresponding to the different threshold value is marked in Fig 5.9. Thus it is easy to find that the value σ_B^2 reaches the top value at 391 when threshold value is 42. The vertical and horizontal values are shown in Fig 5.9. Thus the value 42 is set as the optimal threshold value of this piece of signal, drawn as a straight line in full colour in Fig 5.9. Then all the data values below 42 are regarded as noise and would be attenuated or reduced.

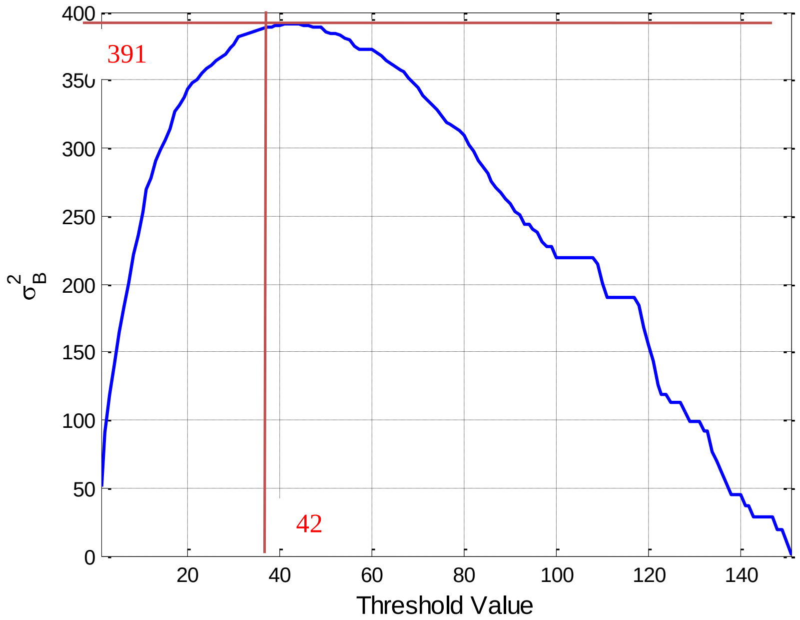


Fig 5.9 Inter-Class variance σ_B^2

Fig 5.8 also shows the comparison between the frequency of the signal before and after signal processing. The de-noise parameter is set as 3 which means all the data values below this optimal threshold (42) would be divided by 3. In Fig 5.8 It is obvious to find that the frequency components higher than 500 are all reduced while the main frequency of the heart sound, from 50 to 500 Hz are mostly kept.

5.4.3 The result and analysis of the experiment

In this research, the original heart sounds are the heart sounds mixed with different kinds of noise. Those heart sounds are collected and de-noised through the Otsu's noise reduction method. The heart sounds, collected in hospital, are cut into four-full heart sound samples for the sake of calculation. The whole procedure of the experiment are recorded and shown in the following content.

The original and processed heart sound signals (the heart sound after noise reduction), are drawn in one figure, copied from the Matlab simulation result in figure. These figures show the original and processed heart sounds by applying the sound data to Otsu's method in Matlab. Heart sound samples, mixed with different kinds of noises, are selected and classified into several groups. In each group, to explain one certain kind of noise corruption and the effect of noise reduction method, a figure contains the original and processed heart sound is attached and the noise reduction effect of our method would be discussed then.

The whole process of the noise reduction experiment is described first with a sample. To identify the effect of the noise reduction method, several steps in this experiment will be mentioned respectively to discuss the realization of the de-noising method. Several steps are included in each heart sound sample, stating the feature of the heart sound, importing the heart sound to Matlab simulation programme, explaining the programme of Otsu's method and discussing the experiment result.

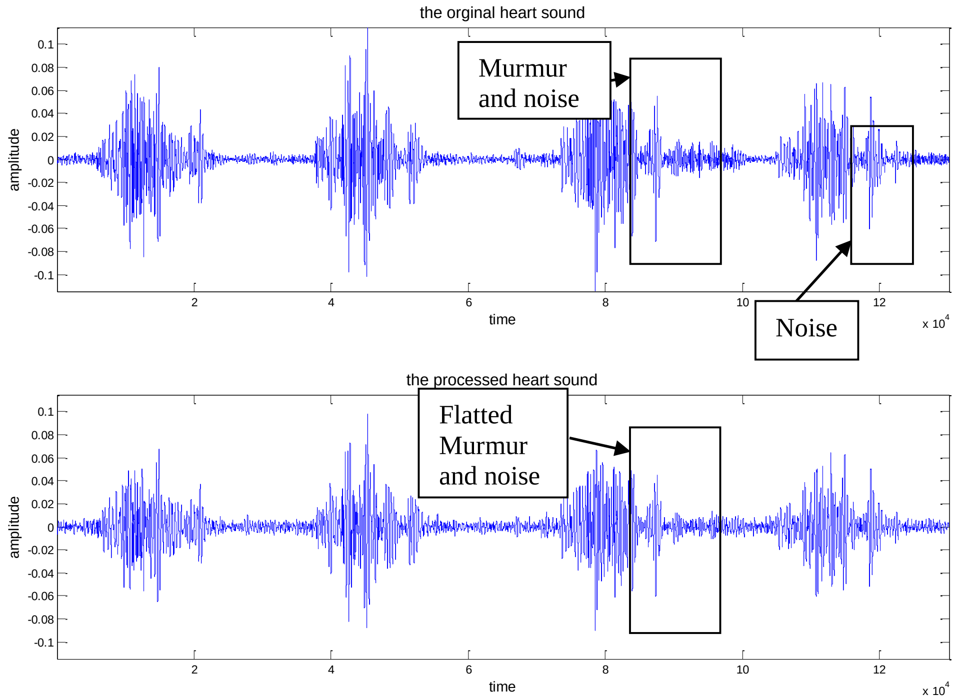


Fig 5.10 Comparison between original heart sound and processed heart sound

1. Statement of the original heart beat

This sample piece of heart sound, shown in Fig 5.10, consists of four full heartbeats. The heartbeat signal is restored in PC by WAV form. The sampling frequency of this signal is 44.1 KHz. In this figure, the first two heartbeats contain little noise, while the third heart beat contains the murmur and noise which has been flagged on the figure. The fourth heart beat is mixed with some noise as well. The heart sound is then inputted in Matlab for the noise reduction process. The heart sound after the noise reduction is shown in Fig 5.10 as well. By comparing the difference between those two graphics, the effect of the noise reduction method can be described and confirmed.

2. Importing the heart sound to Matlab

The simulation of noise reduction is conducted in Matlab. Both the original and processed heart sound signals are recorded and restored. Each step of the noise reduction is explained with the programme command. The programme of Matlab simulation is attached in the appendix.

The original heart sound signal is inputted into matlab by the command `[y, Fs, nbits] = wavread(filename)`, which loads a WAVE file specified by the string *filename*, returning the sampled data in *y*, the sample rate (*Fs*) in Hertz used to encode the data in the file and the number of bits per sample (*nbits*). Fig 5.10 shows the original heart sound with it's sampled data, sample rate and the number of bits. The command `'Plot(Y)'` in Matlab can plot the columns of `Y` versus the index of each value to visual the heart sound. That shows the original heart sound in the Fig 5.10.

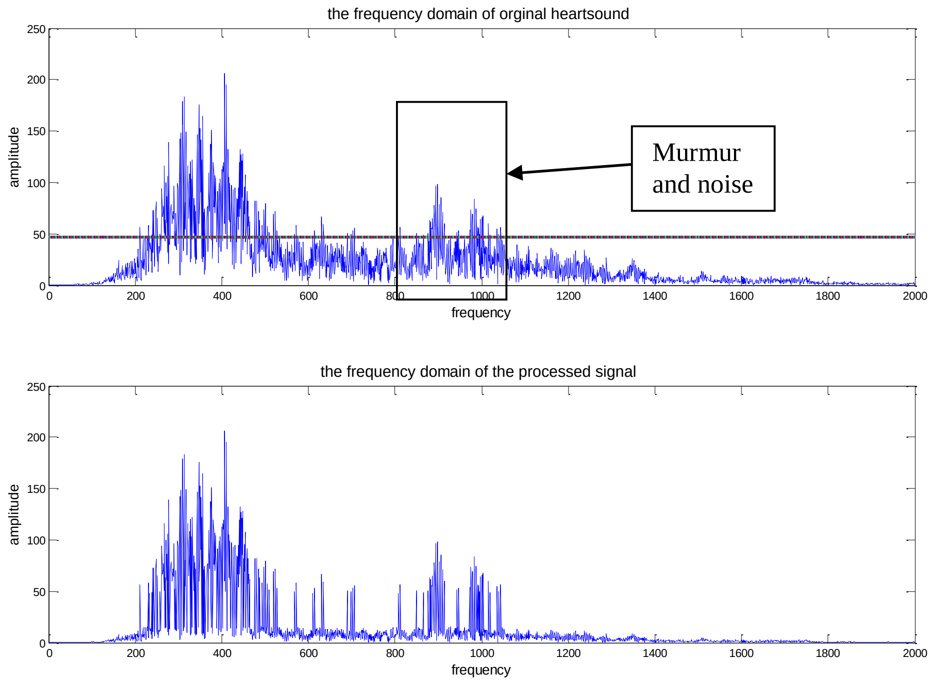


Fig 5.11 Comparison between the frequency of the heart sounds

The original heart sound signal then is transformed into frequency domain for further process. Fig 5.11 shows the result of the Fast Fourier Transform (FFT) of the original signal. The frequency component of the processed signal is shown in the same figure for comparison. The function `Y = fft(X)` returns the discrete Fourier transform (DFT) of vector `X`, computed with a fast Fourier transform (FFT) algorithm. The fast discrete fourier transform and inverse transform pair can be implemented by the functions `Y=fft(x)` and `y=ifft(X)`, x is the sampled data which obtained from command `wavread`. All the value of the data is turned into absolute value for requirement of Otsu's method. `Y=abs(x)` returns an array `Y` that each element of `Y` is the absolute value of the corresponding element of `X`.

The main frequency component of heart sound is under 500 Hz. From Fig 5.11, it can be inferred that the frequency component is not attenuated from 500 Hz in this piece of heart sound. The murmur and noise component can be clearly seen during 800~1000Hz. Some part of noise contains in 0~500Hz as well.

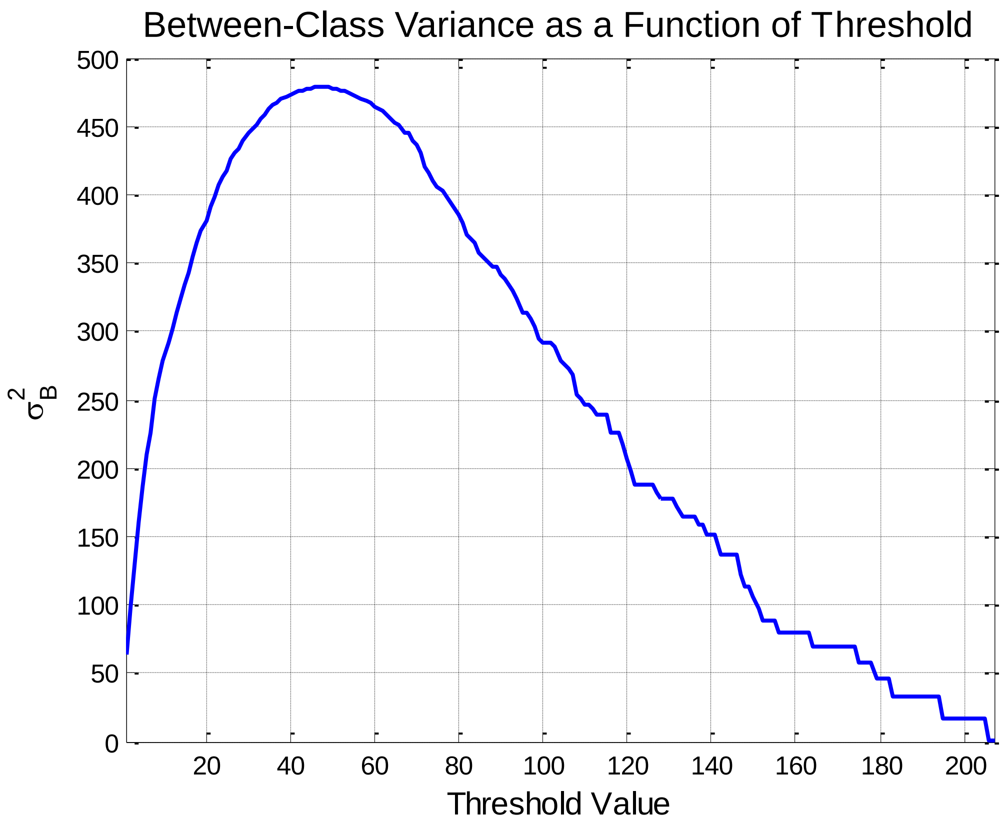


Fig 5.12 Inter-Class variance σ_B^2

3. Explanation of the Otsu's method programme

The `'for'` statement execute certain times of statement in a loop. The Otsu's method is conducted with `'for'` statement. From 1 to the highest value of the signal in frequency domain, each integer value is selected in the `'for'` statement and the threshold value 'k' is obtained when the σ_B^2 number 'vb' is maximum. The whole procedure is shown in Fig 5.12. In this sample, the highest value of the signal is 207, thus the possible threshold value is from 1 to 207. Each integer values between 1 and 207 are selected as assumed threshold value and the corresponding number 'vb' is recorded. The max 'vb' will be found and the value 'k' is regarded as the optimal threshold value when the 'vb' reaches maximum.

For this piece of heart sound sample, the threshold value is 'k=47'. After the threshold value is obtained, all the values lower than 47 will be adjusted. The adjusted number is set as '3' for this sample which means all the value below 47 needs to be divided by three. The adjusted number is changeable and will be tested according to the original signal. The result is shown in Fig 5.11 as the frequency domain of processed signal. The component between 800 to 1000 Hz has been reduced obviously. From the processed signal in Fig 5.10, the murmur and noise part is flatted then.

4. Discussion of the function of the experiment

In this research experiment, about 20 pieces of the heart sound records mixed with different kind of noises have been inputted into the programme and the results and analysis will be shown below. Both the original and processed heart sound figure will be shown together in a figure for the ease of comparing, and the signals in time and frequency domain will be displayed in the following analysis.

This experiment is supposed to test the function of the noise reduction method with heart sound samples in two ways: whether the original heart sound signal with limited noises is corrupted by addictive noise after noise reduction; whether original heart sound signal polluted by noise can be de-noised effectively by the noise reduction method. The advantage, as well as the limitation of this algorithm for different kind of noises will be discussed based on the experiment result.

5.4.4 The experiment records

In the Literature Review Chapter, several main kinds of noise have been identified as main noises to affect the quality of heart sound records from DS. In this experiment, all the sound resources are classified into five groups by the different main noises they mixed with. Several pieces of sound samples for each kind of noises have been selected in this experiment. These noises include:

(1) The respiration sounds or breathing noise
(2) The ambient noise
(3) The noise burst
(4) The human voices
(5) The friction-induced noise

1. The pure heart sound

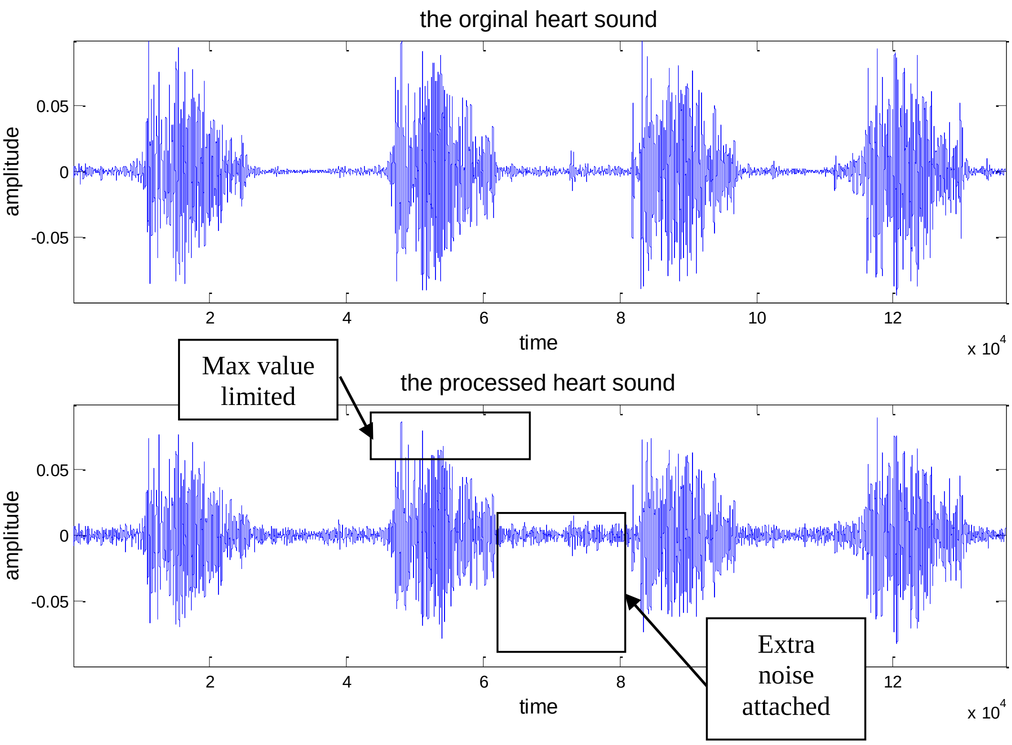


Fig 5.13 The pure heart sound and processed heart sound

This piece of heart sound, shown in Fig 5.13, consists of four whole heartbeats. It is the pure heart sound with limited noise. The original heart sound and processed heart sound after noise reduction are both shown in the same figure for the sake of comparison.

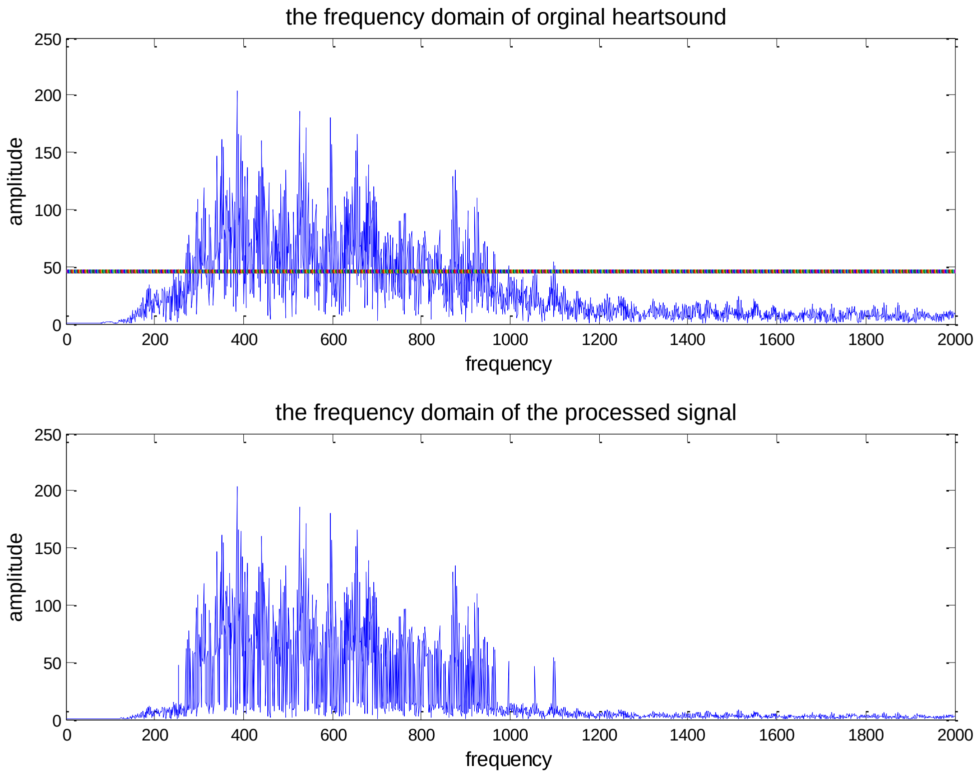


Fig 5.14 Comparison between the frequency of the heart sounds

The main frequency component of this piece of heart sound is between 200Hz and 1000 Hz. From Fig 5.14, the highest value in frequency domain is 207 and the threshold value is 47. It is showed that the frequency component between 300Hz and 950 Hz has been kept, while the rest of the frequency component has been adjusted and limited. Due to the lack of this part of frequency component, in Fig 5.13, the maximum amplitude of the heart sound diminishes, and a little extra noise appears between the heartbeats.

Five pieces of the pure heart sound samples have been selected in this experiment. The experimental data is recorded in the Table 5.1.

Table 5.1 Experiment report of the pure heart sounds

Signal No.	1	2	3	4	5
Max value in frequency domain	207	201	383	255	178
Threshold value	47	45	68	55	44
Main frequency (Hz)	250~1000	300~1100	200~800	200~700	250~1050
Kept frequency (Hz)	300~950	300~900	200~750	300~700	300~800

The noise reduction method is conducted in frequency domain of the heart sound signal. Some key experiment data in frequency domain is recorded for further discussion. Those numbers include: the maximum value in frequency domain, the threshold value, the main frequency (Hz) and the kept frequency component of the signal after noise reduction (Hz).

The full noise reduction process of the first heart sound sample has been described above with the figures. Each of the five samples includes four full heart beats. From the table, the threshold value of the samples varies according to the maximum value of the signal. The 'Main frequency' shows the main frequency component of the samples in frequency domain. For this experiment, 'Main frequency' shows the frequency component of the heartbeat. It is varied (low frequency between 200 and 300Hz, high frequency between 700 to 1100Hz) according to the different sound resources. The 'Kept frequency' shows the unchanged frequency component after noise reduction. Comparing with the 'Main frequency', the range of 'Kept frequency' is quite similar or just the same, which means the main frequency component of the pure heartbeat is well remained after noise reduction in the experiment.

2. The respiration sounds or breathing noise

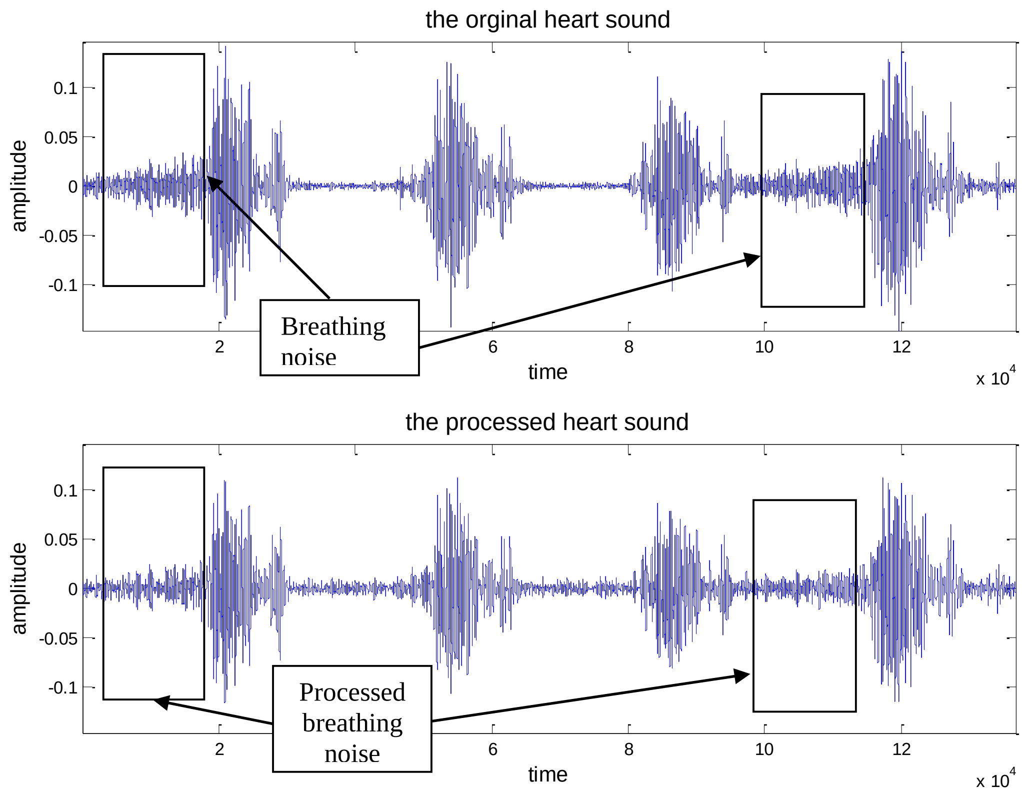


Fig 5.15 heart sound and processed heart sound with breathing noise

This piece of heart sound, shown in Fig 5.15, consists of four whole heartbeats. From the Fig 5.15, the first and the fourth heartbeats contain noises which are validated as breathing sound, while the second and third heartbeats contain little noise. In the processed heart sound, the noise part has been flagged and been limited, and little noise has been added between the second and third heartbeat.

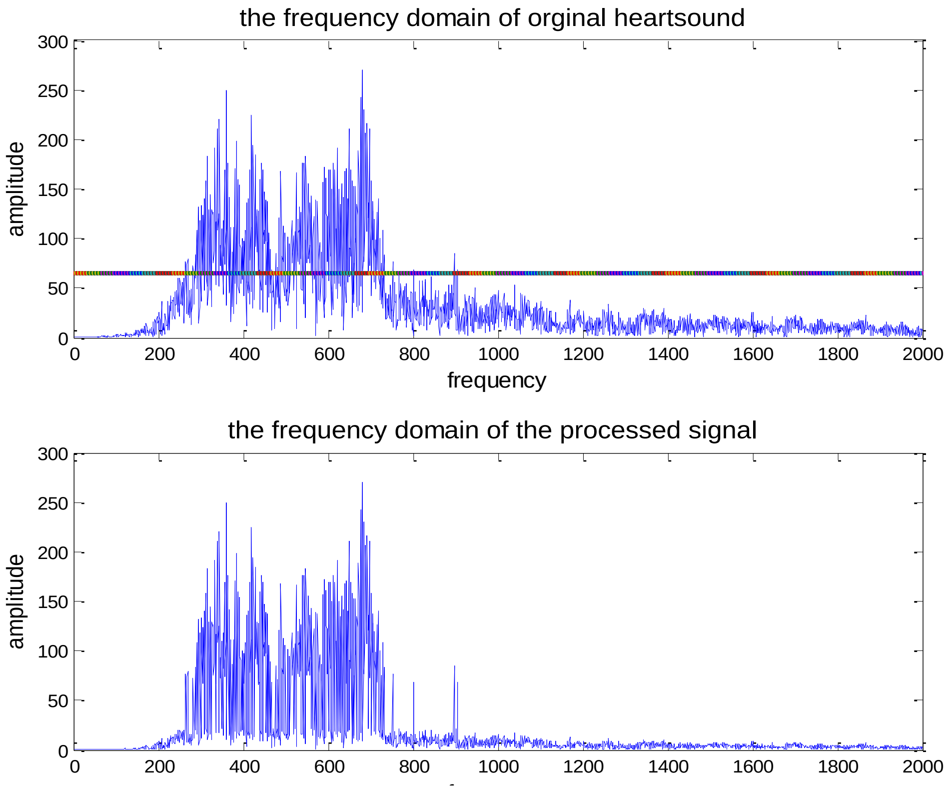


Fig 5.16 Comparison between the frequency of the heart sounds

The main frequency component of this piece of heart sound is below 700 Hz. The breathing noise covers the whole frequency domain and it can be clearly detected in 600~900Hz. From Fig 5.16, it is showed that the frequency component between 300Hz and 700 Hz has been kept, while the rest of the frequency component has been limited. The breathing noise component can be clearly seen after 700Hz.

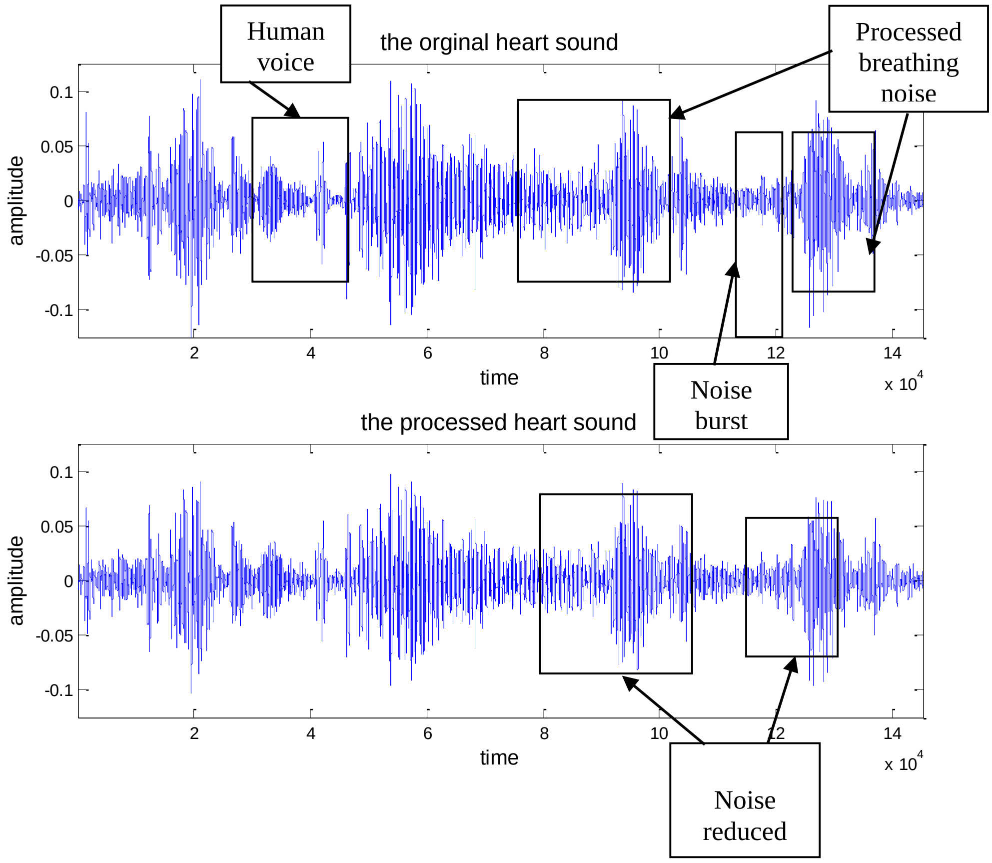


Fig 5.17 Heart sound and processed heart sound with breathing noise

This piece of heart sound, shown in Fig 5.17, consists of four whole heartbeats. In Fig 5.17, the whole heart sound is corrupted by breathing noises, the second heartbeat contains human voice and third heartbeat is mixed with extra noise burst. In the processed heart sound, the noise part has been lighted limited while the noise reduction effect is not obvious. The breathing noise contains in the second heartbeat is reduced to a certain extent, and the human voice in third heartbeat is flatted. But the human voice mixed in the first heart beat is almost unchanged.

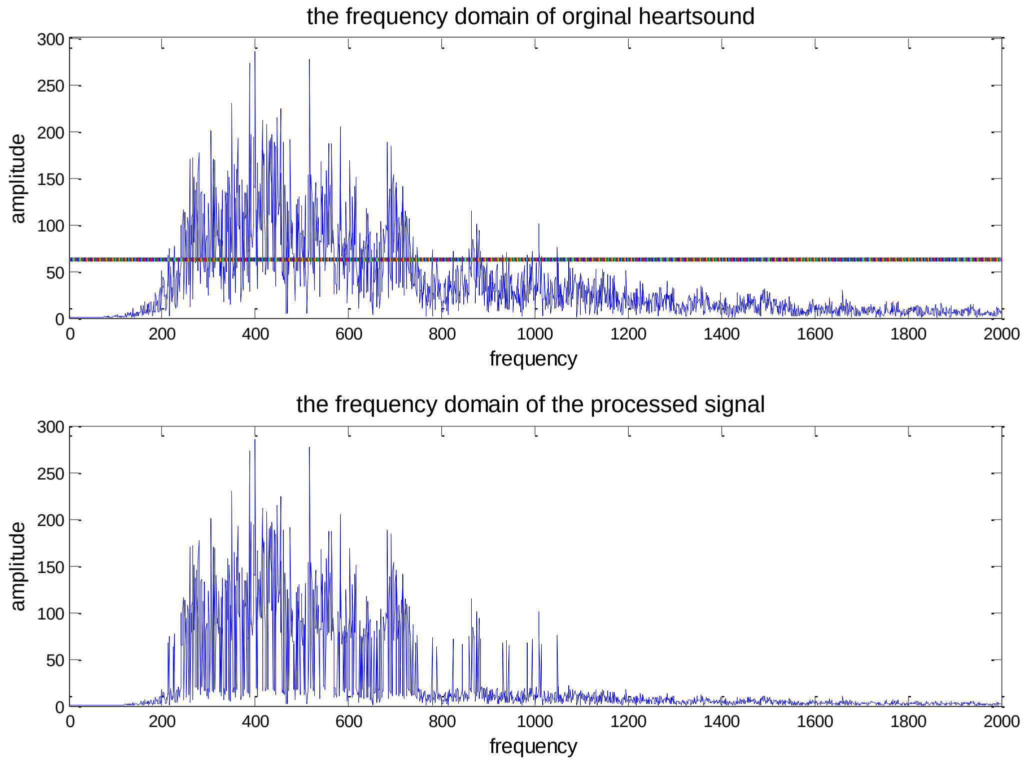


Fig 5.18 Comparison between the frequency of the heart sounds

The main frequency component of this piece of heart sound is 200~1100Hz. The breathing noise covers the whole frequency domain, and it is obviously seen between 600~800Hz. Human voice and noise burst also affect the frequency component from 300 to 500Hz. From Fig 5.18, it is showed that the frequency component between 300Hz and 900 Hz has been kept, while the rest of the frequency component has been limited.

Five pieces of the heart sound samples with breathing noise have been selected in this experiment. The experimental data is recorded in Table 5.2 and the experiment analysis is concluded in Table 5.3.

Table 5.2 Experiment report of the heart sounds with breathing noise

Signal No.	1	2	3	4	5
Max value in frequency domain	277	286	347	305	281
Threshold value	61	63	71	59	70
Main frequency (Hz)	250~1000	200~1150	300~1100	200~1100	200~800
Kept frequency (Hz)	300~750	300~800	300~850	300~750	250~650

Table 5.3 Experiment analysis of the heart sounds with breathing noise

No.	Explanation of the sample	Frequency component	Algorism effect	Comment
1	A four-heartbeats sound, first and fourth heart beat are corrupted by the breathing noise	The frequency range of breathing noise dominates 600~1000Hz, mixed with heart sound from 600 to 700 Hz	Noise from 700 to 1000Hz has been reduced, while 600~700Hz has been kept	Most noise has been reduced, the noise component stoke up with heart sound is higher than threshold and failed to be eliminated
2	A four-heartbeats sound corrupted with breathing noise, and extra human voice and noise burst	The frequency range of breathing noise dominates 600~800Hz, human voice and noise burst appear from 300 to 500Hz	Noise higher than 700Hz has been obvious limited, noise from 400 to 600Hz has been kept	Noise frequency component same as heart sound is not elimited
3	A four-heartbeats sound corrupted by breathing noise in fourth heartbeat	The frequency range of breathing noise is 700~1100Hz	Noise from 800 to 1100Hz has been reduced, the processed signal is similar to the pure heart sound	noise higher than 800Hz has been reduced accurate, the noise component mixed with heart sound 700~800Hz is still remain
4	A four-heartbeats sound corrupted by breathing noise in third and fourth heartbeat	The frequency range of breathing noise is 700~1100Hz, mostly shown in 700~1100Hz	Noise from 750 to 1100Hz has been reduced, the processed signal is obviously noise reduced	The breathing noise corrupted between third and fourth heart sound has been reduced effectively
5	A four-heartbeats sound corrupted by breathing noise in first and second heartbeats	The frequency range of breathing noise is 600~800Hz	Noise from 650 to 800Hz has been reduced, some heart sound frequency is affected	The breathing noise between first and second heart sound has been reduced, while little part of the heart sound is reduced

3. The ambient noise

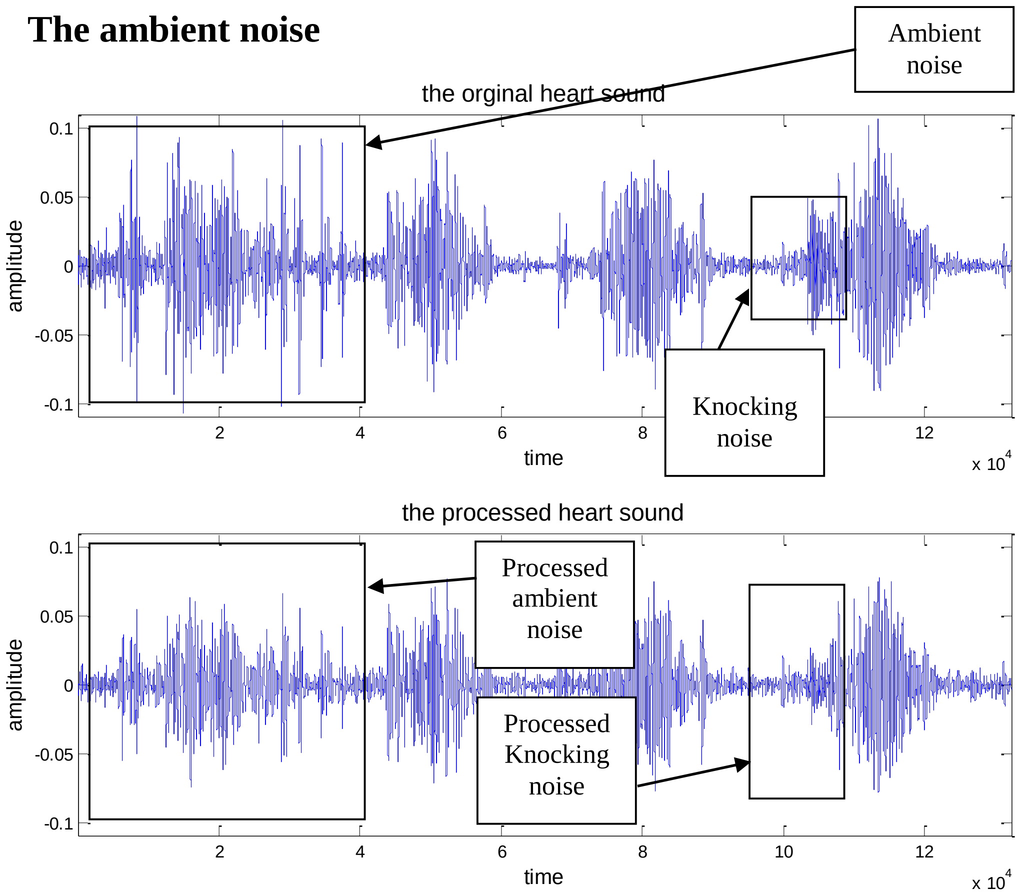


Fig 5.19 Heart sound and processed heart sound with ambient and knocking noise

This piece of heart sound, shown in Fig 5.19, consists of four whole heartbeats. From the Fig 5.19, the first heartbeat is mixed with much ambient noise, and the noise covers the whole time domain of it. The fourth heartbeat contains a piece of knocking noise, sound like something dropping on the table. The second and third heartbeats contain relatively less noise. In the processed heart sound, the first heart sound has been limited entirely, especially the burst of sound during the time domain. The noise part of the fourth heart sound is reduced obviously, and the shape of the heart beat is with little change. Little noise has been added between the second and third heartbeat.

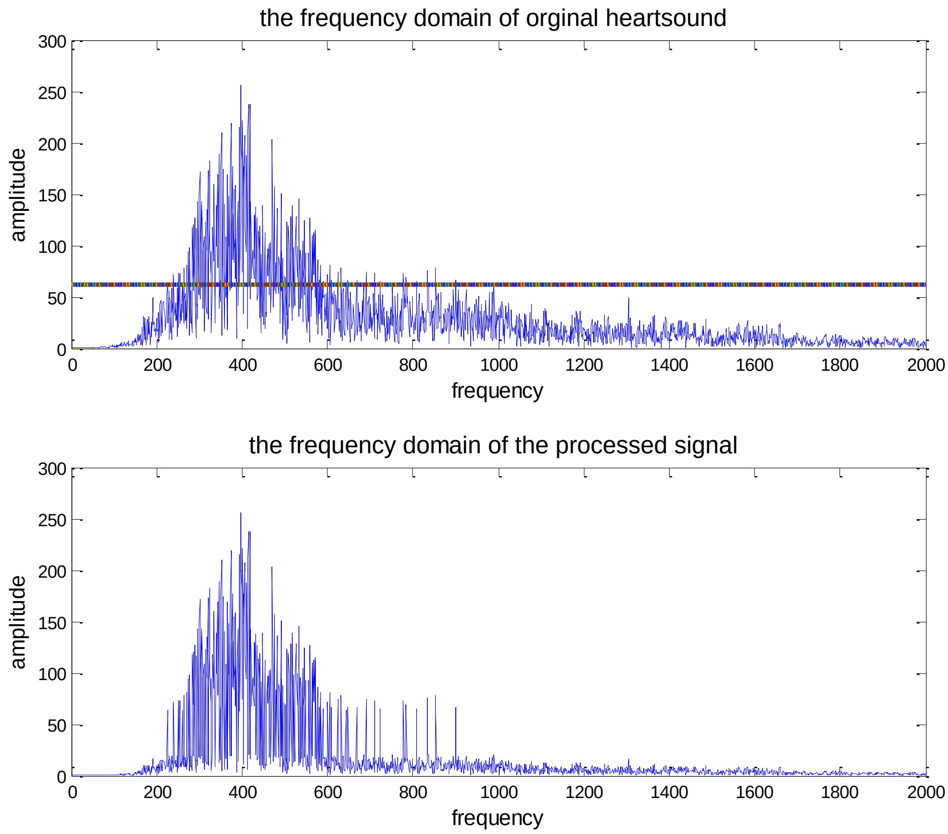


Fig 5.20 Comparison between the frequency of the heart sounds

The frequency component of this piece of heart sound reaches maximum around 400Hz, and then the component dies down from 400Hz to 600Hz. And the frequency component between 600Hz and 1000Hz which is flat and higher than the rest of component are the noise part. The noise also covers the heart sound frequency component, form 300Hz to 700Hz. From Fig 5.20, it is showed that the frequency component between 300Hz and 600 Hz has been almost kept, while the rest of the frequency component has been limited.

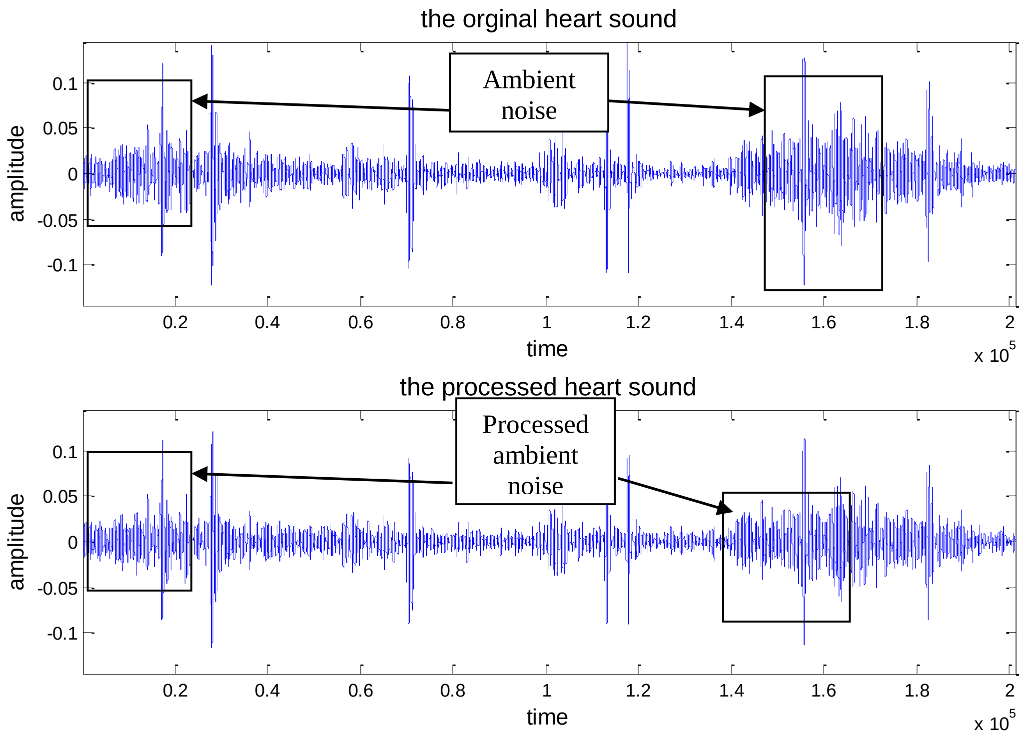


Fig 5.21 Heart sound and processed heart sound with ambient noise

This piece of heart sound, shown in Fig 5.21, consists of four whole heartbeats. From the Fig 5.21, the first and fourth heartbeats are mixed with ambient noise, and the noise covers the whole time domain of the heart beats. The second and third heartbeats contain relatively less noise.

In the processed heart sound, the noise part of both first and fourth heart sound has been reduced, and the shape of the heart beat is with little change. Little noise has been added between the second and third heartbeat.

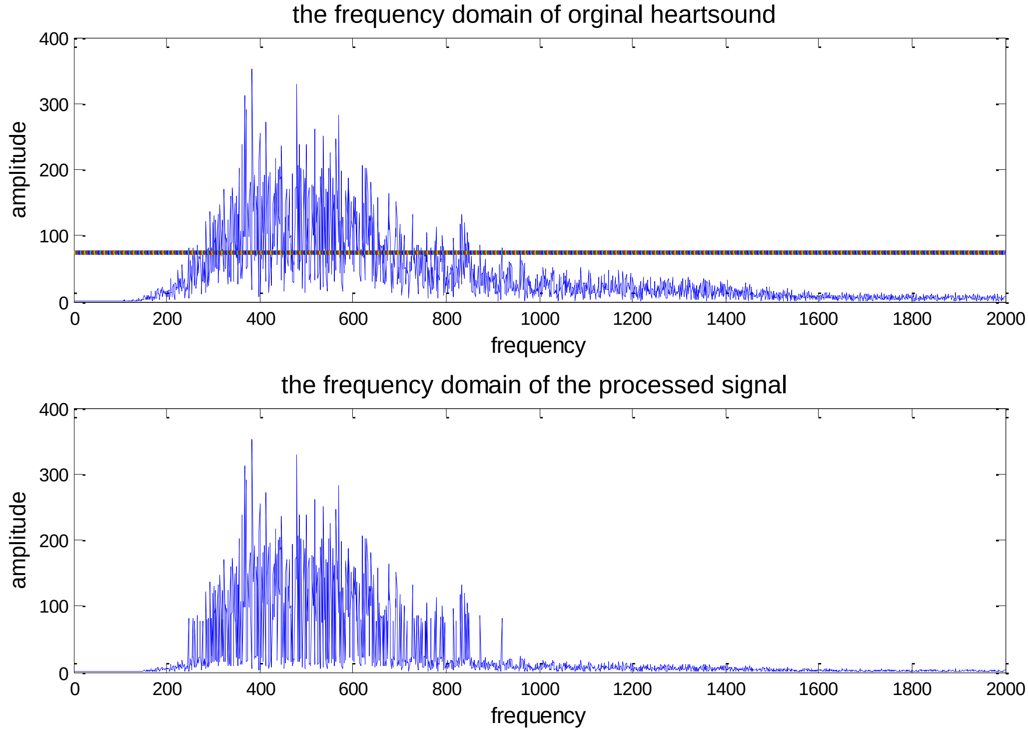


Fig 5.22 Comparison between the frequency of the heart sounds

The frequency component of this piece of heart sound reaches maximum around 350Hz, then the amplitude of component descends from 300Hz to 800Hz and keeps from 900Hz to 1200Hz. According to the characteristic of ambient noise, the noise component tiles the whole frequency domain. The noise also covers the heart sound frequency component from 300Hz to 800Hz. From Fig 5.22, it is showed that the frequency component between 400Hz and 800 Hz has been almost kept, while the rest of the frequency component has been limited.

Table 5.4 Experiment report of the heart sounds with ambient noise

Signal No.	1	2	3	4	5
Max value in frequency domain	264	353	347	315	376
Threshold value	54	75	62	53	68
Main frequency (Hz)	300~1000	250~1150	300~1100	200~1000	200~850
Kept frequency (Hz)	300~600	300~800	300~850	300~750	250~650

Table 5.5 Experiment analysis of the heart sounds with ambient noise

No.	Explanation of the sample	Frequency component	Algorism effect	Comment
1	A four-heartbeats sound, first heart beat corrupted by the ambient noise, fourth heart beat with knocking noise	The frequency range of ambient noise covers 300~1000Hz, mixed with heart sound from 300 to 700 Hz	Noise from 600 to 1000Hz has been reduced, while 300~600Hz has been kept	Most noise has been reduced, the noise component mixed with heart sound is failed to be eliminated
2	A four-heartbeats sound corrupted with ambient noise in first and fourth heart beats	The frequency range of ambient noise covers 250~1150Hz,	Noise higher than 800 has been obvious limited, noise from 300 to 800 has been kept	Noise frequency component same as heart sound is not reduced
3	A four-heartbeats sound corrupted by ambient noise in fourth heartbeat	The frequency range of ambient noise is 300~1100Hz	Noise from 850 to 1100Hz has been reduced, the processed signal is restored as the pure heart sound	noise higher than 850Hz has been reduced accurately, the noise component mixed with heart sound 300~850Hz is still remain
4	A four-heartbeats sound corrupted by ambient noise in third and fourth heartbeats	The frequency range of ambient noise is 200~1000Hz	Noise from 750 to 1000Hz has been reduced, the processed signal is obviously noise reduced	The breathing noise corrupted between third and fourth heart sound has been reduced effectively
5	A four-heartbeats sound corrupted by breathing noise in second heartbeats	The frequency range of breathing noise is 200~850Hz	Noise from 650 to 850Hz has been reduced, some heart sound frequency is affected	The breathing noise in second heart sound has been reduced, while little part of the heart sound is reduced

4. The noise burst

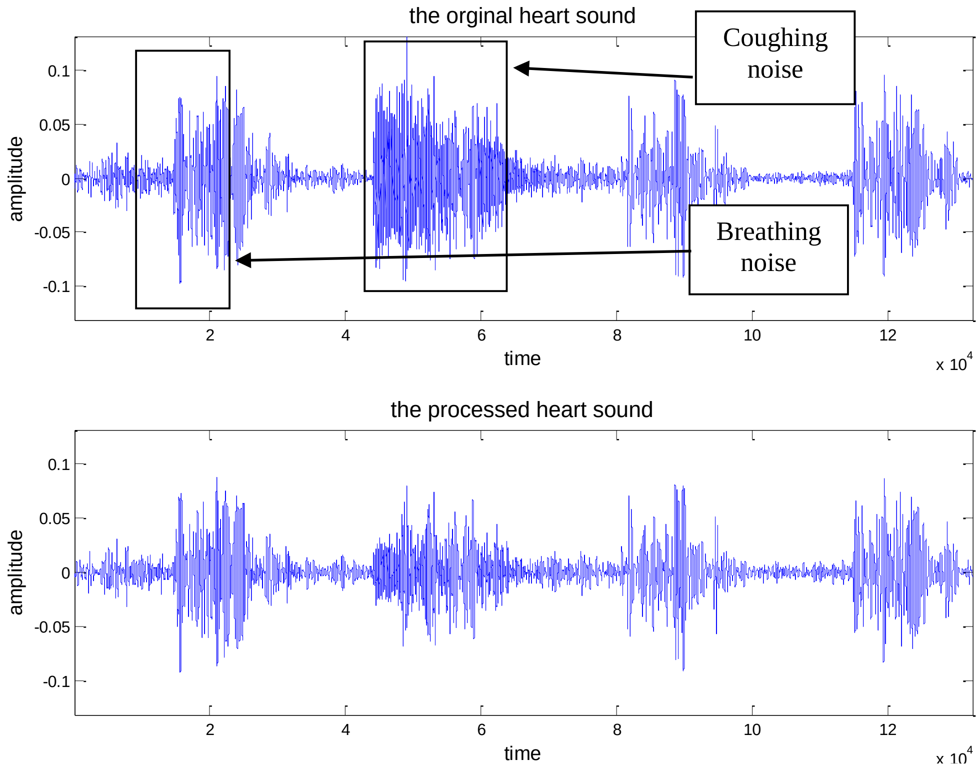


Fig 5.23 Heart sound and processed heart sound with breathing and coughing noise

This piece of heart sound, shown in Fig 5.21, consists of four whole heartbeats. From the Fig 5.23, the first heartbeat contains breathing noise. The second heartbeat is mixed with a piece of coughing noise, and the heartbeat is corrupted seriously and can hardly be identified in the time domain. The third and fourth heartbeats contain relatively less noise.

In the processed heart sound, the first heart sound has limited changed, and the playback sound shows that the breathing noise is not much reduced. The second heart sound, in the contrast, is changed to an obvious extent, and the playback sound shows that the coughing noise has been aggressively removed. Little noise has been added between the third and fourth heartbeat, but the shape of the whole heartbeat is with little deformation.

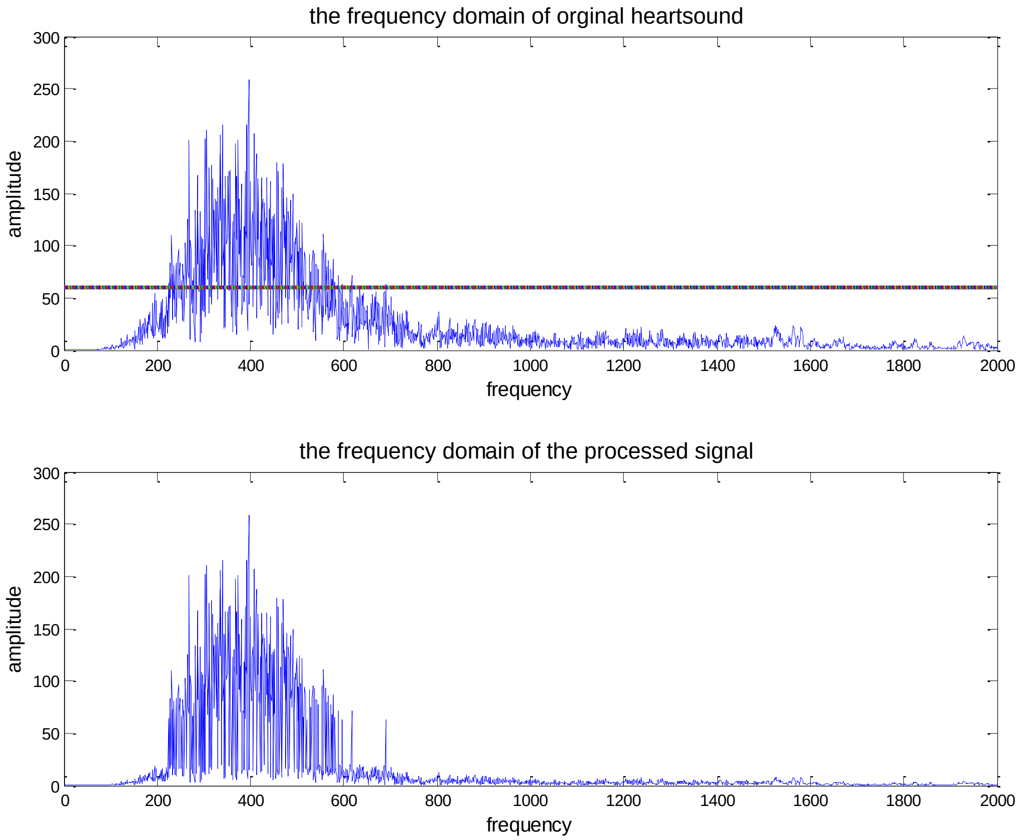


Fig 5.24 Comparison between the frequency of the heart sounds

The frequency component of this piece of heart sound reaches maximum around 400Hz, and then the component dies down until 800Hz. As the frequency component of heart sound is below 500Hz, the frequency between 600Hz and 800Hz is regarded as pure noise. From Fig 5.24, the processed signal frequency figure shows that the frequency component between 250Hz and 600 Hz has been almost kept, while the rest of the frequency component has been limited.

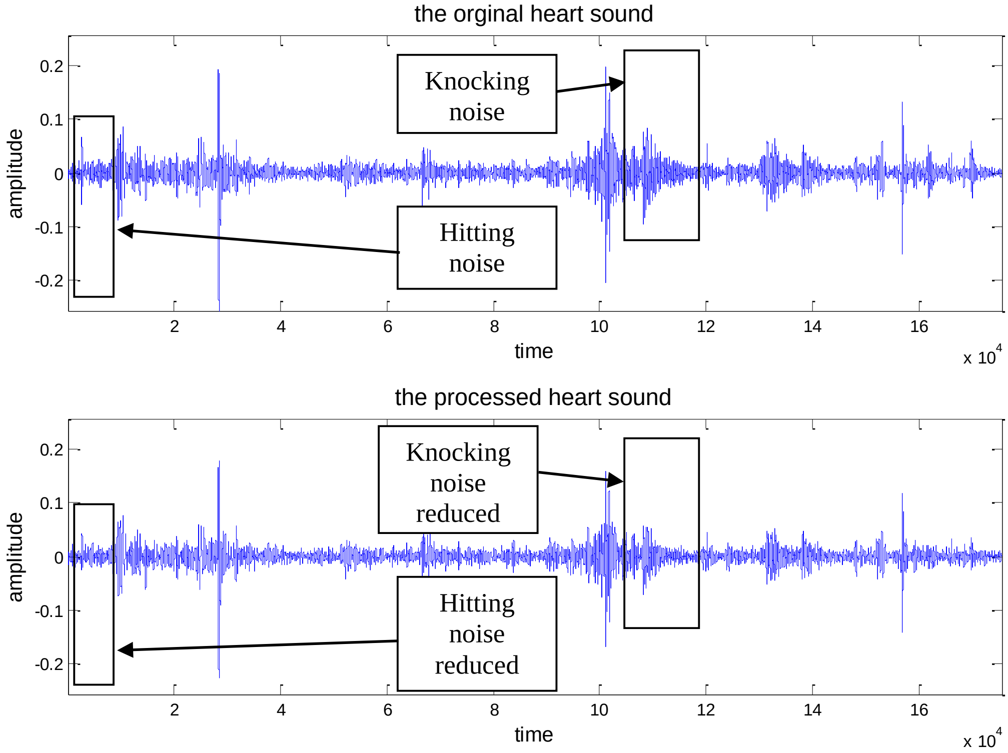


Fig 5.25 Heart sound and processed heart sound with knocking noise

This piece of heart sound, shown in Fig 5.25, consists of four whole heartbeats. From the Fig 5.25, the first and third heartbeats contain noise burst. The noise burst of the first heart beat is the chest piece hits the clothes buttons. The noise burst of the third heart beat sounds as a piece of knocking noise nearby.

In the processed heart sound, the first heart sound has limited changed, and the playback sound shows that the hitting noise is partial reduced. The third heart sound, in the contrast, is changed obviously, and it shows on the figure that the knocking noise has been mostly removed. Little noise has been added between the second and fourth heartbeat, and the shapes of them are with little deformation.

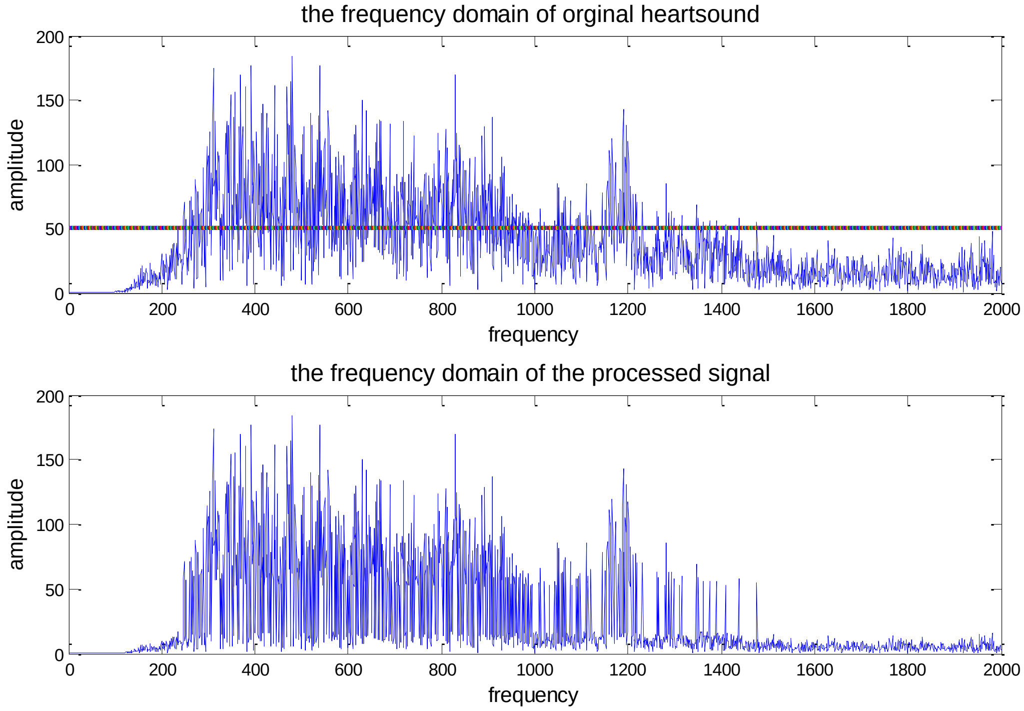


Fig 5.26 Comparison between the frequency of the heart sounds

The frequency component of this piece of heart sound is between 250Hz to 1400Hz. The component amplitude reaches maximum on 500Hz and keeps in a high level until about 800Hz. Then it floats from 800Hz to 1400Hz. As the main frequency component of heart sound is below 700Hz, the frequency higher than 800Hz is regarded as the noise burst. From Fig 5.26, the processed signal frequency figure shows that the frequency component between 250Hz and 1200Hz has been almost kept, while the rest of the frequency component has been limited.

Table 5.6 Experiment report of the heart sounds with noise burst

Signal No.	1	2	3	4	5
Max value in frequency domain	285	276	343	184	263
Threshold value	58	62	56	51	67
Main frequency (Hz)	200~700	200~1050	300~1200	200~1400	200~950
Kept frequency (Hz)	200~600	300~750	300~850	250~1200	250~750

Table 5.7 Experiment analysis of the heart sounds with noise burst

No.	Explanation of the sample	Frequency component	Algorism effect	Comment
1	A four-heartbeats sound, first heart beat corrupted by a piece of light breathing noise, second heart beat with coughing noise	The frequency range of knocking noise dominates 300~700Hz, mixed with heart sound 200 ~ 600 Hz	Noise from 600 to 700Hz has been reduced, while 300~700Hz has been kept	Most of the breathing noise has failed to be eliminated, part of the coughing noise component is reduced,
2	A four-heartbeats sound corrupted by noise burst of bumping noise	The frequency range of noise burst covers 700~10500Hz,	Noise higher than 750Hz has been obvious limited, noise from 600 to 750Hz has been kept	Noise frequency component same as heart sound is not eliminated
3	A four-heartbeats sound corrupted by noise burst in second heartbeat	The frequency range of noise burst is 600~1200Hz	Noise from 1000 to 1200Hz has been reduce	noise higher than 1000Hz has been reduced, the noise component mixed with heart sound 600~1000Hz is still remain
4	A four-heartbeats sound corrupted by noise burst in first and fourth heartbeats	The frequency range of noise burst is 700~1400Hz	Noise from 1000 to 1400Hz has been reduced, the knocking noise is obviously reduced	The noise burst corrupted the third heart sound has been reduced effectively, while the hitting noise is not eliminated well
5	A four-heartbeats sound corrupted by noise burst in first and second heartbeats	The frequency range of noise burst is 600~950Hz	Noise from 750 to 950Hz has been reduced, some heart sound frequency is affected as well	The noise burst between first and second heart sound has been reduced, while little part of the heart sound is reduced

5. The friction-induced noise (small movement of human or stethoscope)

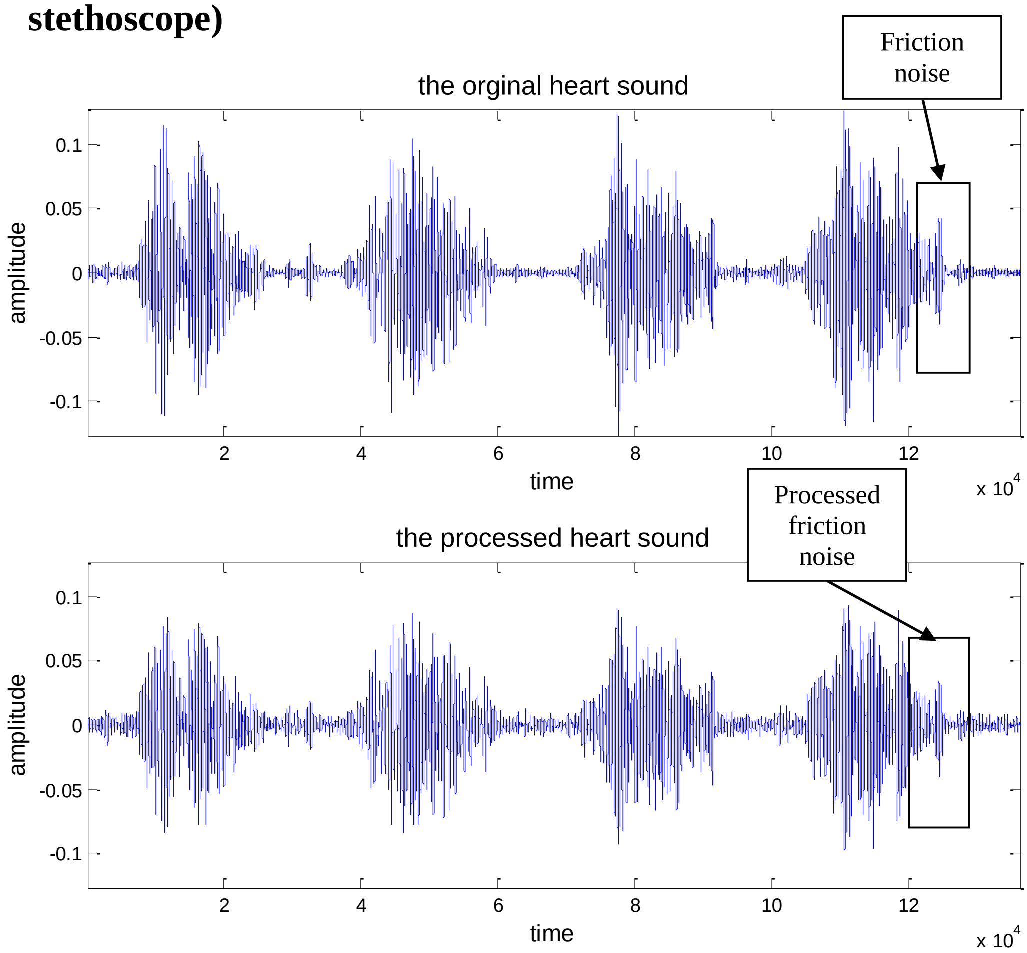


Fig 5.27 Heart sound and processed heart sound with friction noise

This piece of heart sound, shown in Fig 5.27, consists of four whole heartbeats. From the Fig 5.27, the forth heartbeat contains friction noise. The rest three heartbeats contain relatively less noise.

In the processed heart sound, the whole heart sound is not much changed. The fourth heart sound with friction noise has been slightly reduced, while the whole amplitude of the fourth heart is decreased as well.

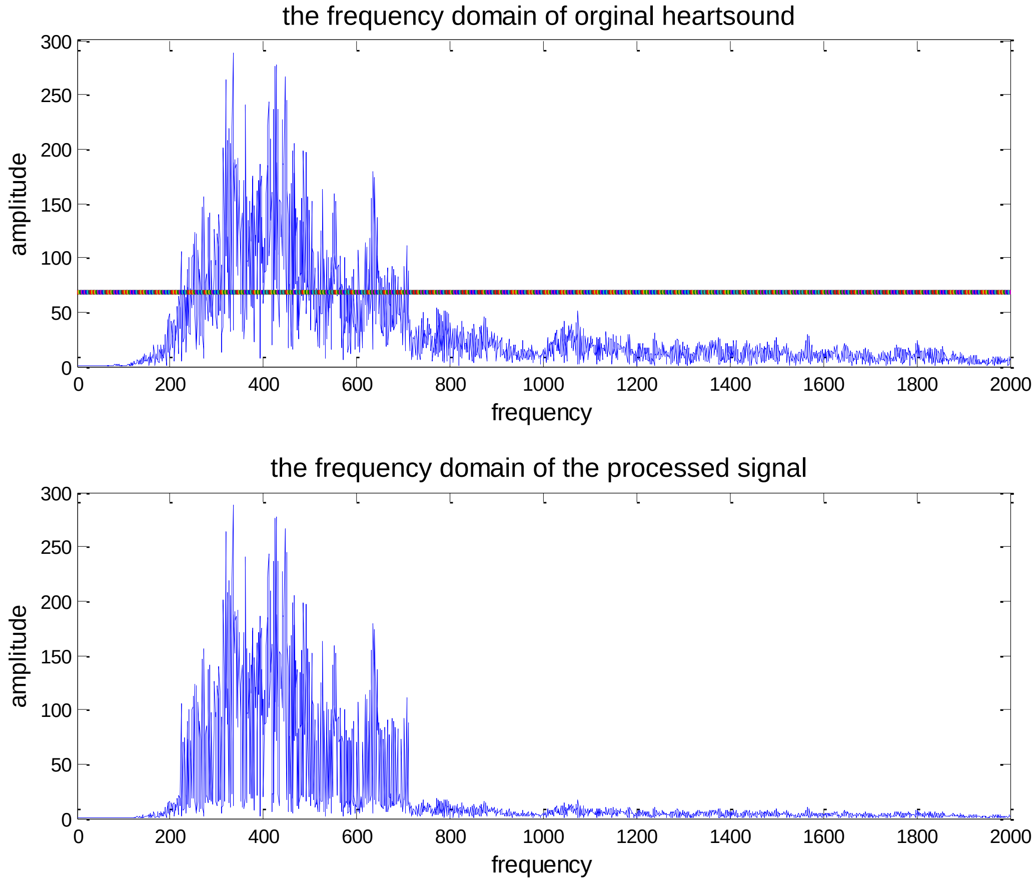


Fig 5.28 Comparison between the frequency of the heart sounds

The main frequency component of this piece of heart sound is 200~750Hz. As the friction noise does not severely affect the whole heart sound, the component of the friction noise is not obvious on the frequency domain. From Fig 5.28, the optimal threshold of this piece of heart sound is 69. It is showed that the frequency component between 250Hz and 750 Hz has been kept, while the rest of the frequency component has been limited. With relatively less noise, the main heart sound component can be defined and kept accurately through this algorithm.

6. The human voice noise

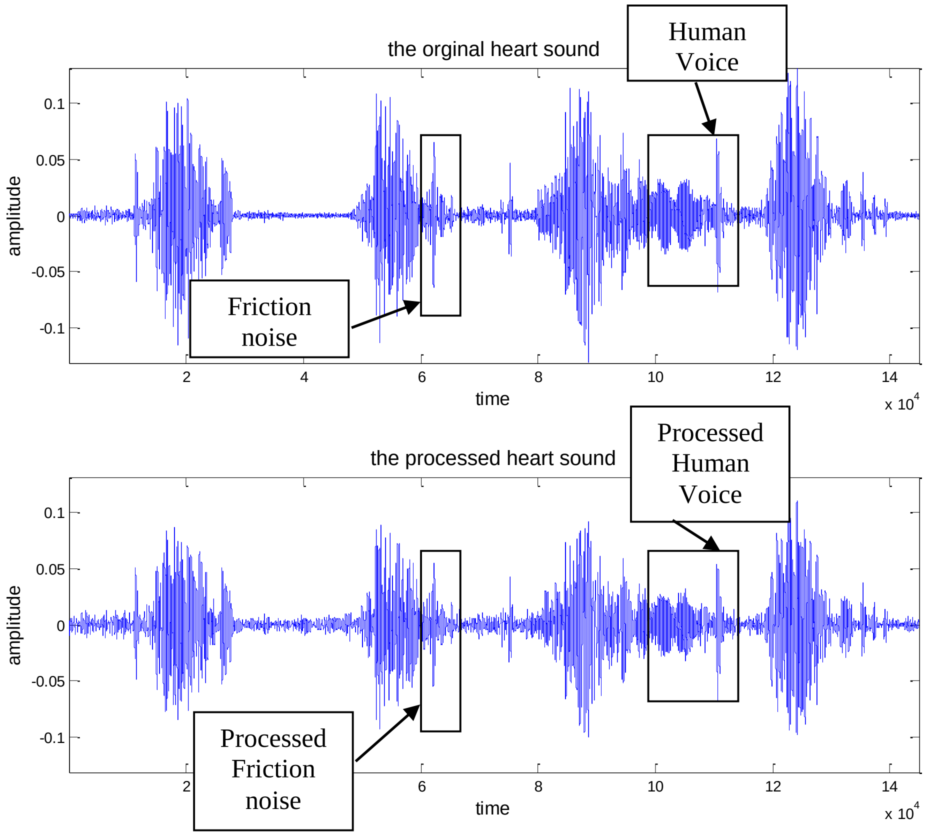


Fig 5.29 Heart sound and processed heart sound with human voice

This piece of heart sound, shown in Fig 5.29, consists of four whole heartbeats. From the Fig 5.29, the third heartbeat is cover by physician's voice. The second heartbeat is mixed with little friction noise as the movement of the chest piece on the patient's body. The first and fourth heartbeats contain relatively less noise.

In the processed heart sound, the whole heart sound has limited changes. From the processed heart sound, both the second and third heart beats are mostly kept and the shape of the heart sound is not much changed. The playback sound shows that the most part of human voice is not much reduced. Little noise has been added into the processed heart sound, but the human voice is not properly defined as noise in this algorithm.

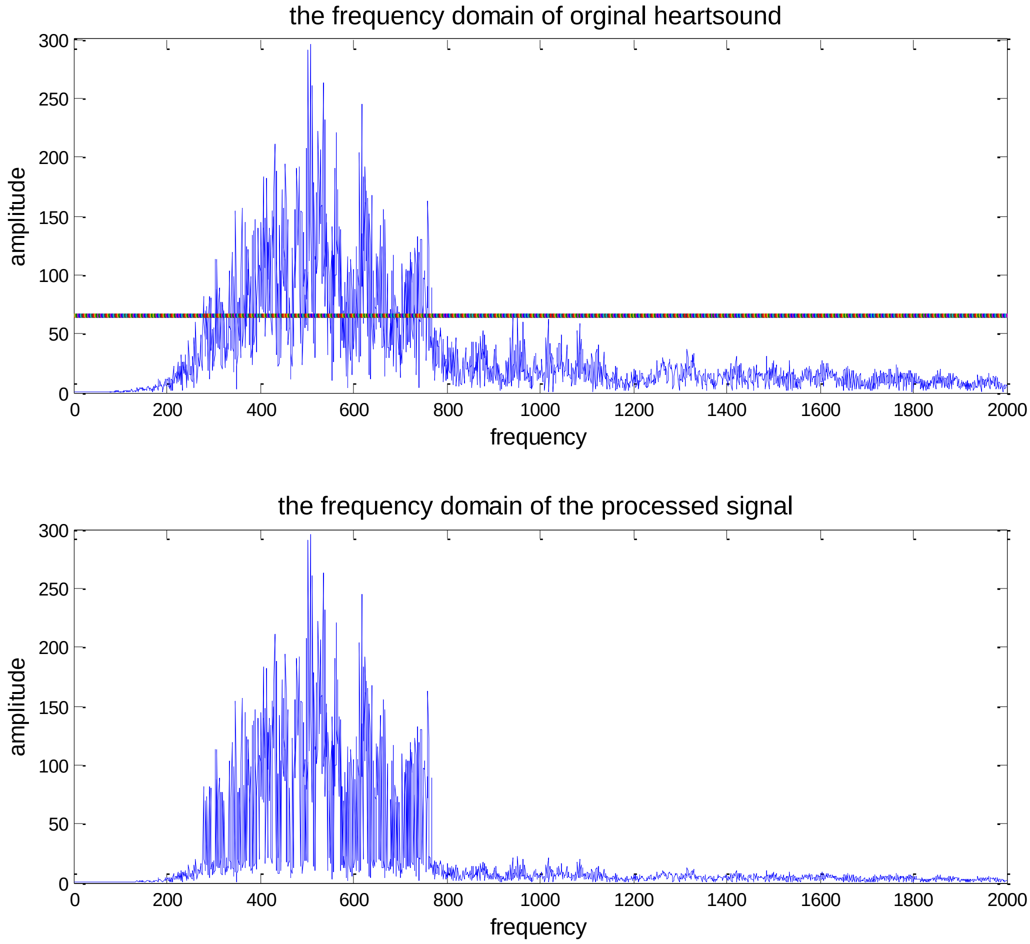


Fig 5.30 Comparison between the frequency of the heart sounds

The main frequency component of this piece of heart sound is between 300Hz to 750Hz. The component amplitude reaches maximum on about 500Hz and declines sharply on 750Hz. The optimal threshold of this piece of heart sound is 66. As the main frequency component of heart sound is below 750Hz, the frequency higher than 750Hz is regarded as the noise part. From Fig 5.30, the processed signal frequency figure shows that the frequency component between 250Hz and 750Hz has been almost kept, while the rest of the frequency component has been limited.

5.5 Summary of the findings

This experiment utilized 68 heart sound samples (42 collected in India and 26 collected in Australia) to gain the efficiency of the noise reduction method. All heart sound samples are segmented in to small pieces (four heart beats section). Those pieces with one or more kinds of certain noise have been selected in the experiment. With these sound samples, the results show that the noise reduction method is effective in all the noises, but the method has different effects on noises of different kind.

The experiment starts with the pure heart sound signal. Five pieces of heart sound from different patients have been selected in this part of experiment. These heart sounds are various in amplitude and frequency. The Max value, threshold value, main and kept frequency domain have been recorded then. With the results, the range of kept frequency domain is similar to that of real main frequency domain of the heartbeats. The closeness of low frequency is 86.87%, and the closeness of high frequency is 89.35%. The result proves that most part of the heart sound component has been kept. From the comparison between the original and processed signal, the heart sound shape is also almost the same. Thus this noise reduction method can catch the pure heart sound component accurately with little addictive changes.

For the respiration and breathing noise, the method acts good as well. The closeness of low frequency is 79.33%, while the closeness of high frequency is 74.25%. The frequency range of breathing noise various from 600~800Hz, which mixed with high frequency component of the heart sound samples. From the results, when more noise is introduced, the rate between threshold value and max value would increase (the rate of sample 5 is 24.91%), and less frequency component would be kept (66.7% in sample 5); while with less noise, the rate of value is relatively low (20.46% in sample 3) and more frequency component would be kept (75% in sample 3). Thus the results prove that the threshold is fluctuated due to the quality of input signal. From the playback of the processed signal, the breathing noise has been reduced obviously as well.

The ambient noise is one of the most common noises for the DS records. In the part of experiment for ambient noise, the closeness of low frequency is 86%, while the closeness of high frequency is 72.55%. As the background environment and the reason that the noise creates are various, each ambient noise can be different. Thus the efficiency of the method differs in each piece of sound sample. For the knocking noise, the de-noising method can identify the noise part properly, and the noise would be limited obviously; for the continuous background noise, the method doesn't work very well and the noise is slighted changed.

The noise burst is the noise occurs suddenly. In the part of experiment for noise burst, the closeness of low frequency is 85.33%, while the closeness of high frequency is 78.53%. In the experiment, the noise burst signal is always mixed with other noise, but the noise burst (coughing noise or hitting noise) can be eliminated effectively.
The friction-induced noise always exists in auscultation. It happens when the chest piece moves on patient's body. Compared with the heart sound, this noise is not obvious. The results of the experiment show that the whole heart sound component has been kept and little part of the friction noise has been eliminated. The result proves that with little input noise, the threshold value would be increased and most of the heart sound component would be kept well.

The human voice noise happens when patient or physician speaks during the auscultation. The record of the experiment shows that the human voice covers the same frequency domain of as heartbeat, thus the method can hardly identified and depart the human voice from the heart sound. Most of the human voice has been kept in the processed signal. Further research of the de-noising method can be conducted in this area.

5.6 Conclusion of this chapter

This chapter reports the whole process and results of the experiment study, which tested the noise reduction method, which designed in the Methodology Chapter, with 68 heart sound samples collected in Australia and India.

The descriptive data analysis include the introduction of the whole process of the experiment, the classification of the sound samples, and the recorded results of the sound samples mixed with each kind of noises. Those data and description, concluded in tables, are useful in understanding the progress and efficiency of the designed noise reduction method.

The data analysis covers the benefits and drawbacks of the noise reduction method for each kind of noise. Through the experiment result, pure heart sound can be accurately identified without noise; the breathing noise can be reduced properly; the result of ambient noise reduction is various in different backgrounds; part of the noise burst can be identified and eliminated; the friction-induced noise doesn't affect the whole heart sound quality thus it is not reduced much; and this method failed to depart the human voice with the heart sound.

There are some limitations in this study. First limitation is in the data collection process——the original sound resources are collected and restored in MP3 form. It causes the loss of the high frequency component during the transformation of the signal. Sometimes there is more than one kind of noises mixed with the heart sound, and it is hard to compare the different noise reduction efficiency on the same heart sound sample. The heart sounds are collected from different heart patient, so the result for different heart disease may differ.

A further discussion about the analysis of method is provided in the next chapter, Discussion Chapter, for comparing the relationship between the findings of this experiment and relevant studies in the literature review.

Chapter 6 Findings

In Chapter 4, the research approach and methodology for noise reduction study were identified. In Chapter 5, five kinds of noises were examined and analysed. The experiment findings were discussed in the context of literature review chapter. The newly designed noise reduction method was also discussed in the previous chapter, and the effect of this method is concluded in this chapter.

6.1 Otsu’s noise reduction method

Otsu’s method was introduced in the Literature Review chapter as a self-adaptive threshold method. In the Otsu’s algorithm, the input signal can be separated into two parts by an optimal threshold of this method. The threshold is selected by the discriminant criterion of the input signal, which maximizes the separability of the two parts of the signal. The details of the algorithm can be found in Literature Review Chapter.

Otsu’s method has been used to investigate image thresholding (Sezgin 2004), or the reduction of a gray level image. In this study, Otsu’s method was applied in signal processing area in the domain of heart sounds. The outcomes indicate that this method is also suitable for heart sound noise reduction. This study has established the following specific aspects in regards to heart sound analysis using Otsu’s method.

- The selection of the optimal threshold

To apply the Otsu’s method into heart sound signal processing, it was assumed that the input signal (in frequency domain) contained two parts of signals (useful sound and noise). The calculated optimum threshold separated those two parts so that their combined spread (intra-class variance) was minimal. At the same time, the algorithm kept the between class variance σ_B^2 to a maximum. In the signal processing, the optimal threshold was to select the discriminant criterion through maximizing the between class variance σ_B^2 of those two parts of signals. It has been identified that the spectral characteristics of heart sound was transient and like periodical, while the characteristic of the noises was varied and sometimes random. Thus when the between class σ_B^2 reach maximum, the optimal division between heart sound and mixed noise was obtained. As the threshold has been set automatically after the calculation of maximum σ_B^2, it directly solved the problem of evaluating the goodness of thresholds.

- No need for priori noise characteristics

OTUS’s method is a nonparametric and unsupervised method of automatic threshold selection. From the literature and data analysis, the noise could influence the quality of heart sound in many ways (Zhang, Y. T. et al. 2006). In addition, the noise was always changeable when capturing the sound under real environment. Thus it was quite difficult to predict the characteristics of different kinds of noise before the sound was collected. Through the noise reduction of Otsu’s method, the spectral characteristics of heart sound and noise were computed in real-time, thus eliminating the need for any knowledge of noise in advance. Thus the procedure of noise reduction became very simple because the optimal threshold was only based on the integration of signal itself. This feature, thus, supported the method for different kind or random noises.

- The fluctuating de-noising threshold

The optimal threshold of Otsu's method, which applied in the frequency domain of the input signal, could be adjusted according to the quality of input data. Through the experiment data and analysis, this threshold could fluctuate due to the SNR of the input heart sound. In particular, when the SNR was large (means little noise contained in the heart sound signal), the threshold value would be low so that more useful information will be maintained, while the threshold value would be relatively high to remove more noise when the SNR seemed to be small. The fluctuating of the threshold made the balance between the noise reduction and the useful sound remaining, thus providing better output sound quality.

- The changeable de-noising parameter

After the threshold value was confirmed in frequency domain, all the data values above the threshold value were regarded as useful heart sound and would be kept, while the data values below the threshold were regarded as noise part and needed to be adjusted or reduced. Those data needed to be reduced would be adjusted according to the 'de-noise parameter'. The default de-noising parameter was set as 3, which means all the data values below this optimal threshold would be divided by 3. However, when the noise was increasing, the de-noising parameter should rise up thus more noise would be eliminated during the noise reduction procedure. This feature together with the fluctuating threshold provided better noise reduction result.

6.2 Noise and sound quality

It should be noted that designing a suitable noise reduction method for managing heart sounds is the core content of this study. Thus, the noise contained in the recorded heart sound and its influence to the diagnosis becomes necessary knowledge of this study. From the finding of this study, 'noise' was the most important factor, which influenced the quality of heart sound. Thus it appeared to have a high impact on physicians' diagnosis. This issue has been identified from the literature. For example, "the contribution of each noise source may vary significantly depending on the noise kind, the technical characteristics of the recording instrumentation, the recording environment, and the physiological status of the subject" (Zhang, Y. T. et al. 2006). Belloni mentioned that high levels of environmental noise invalidated the auscultations and suitable electronic cancellation was not normally implemented for this reason (Belloni et al. 2010). Further, ambient noise and disturbances could corrupt the recorded heart sound signal and affect the accuracy of data collection (Tang et al. 2010). The finding of this study proved that most of the recorded heart sounds were combined with noises, and the impact of sound quality differed from the noises of various kinds. For example, during the heart sound capturing process, the background noise (ambient noise, noise burst) and friction noise between the digital stethoscope and the skin would always appear to corrupt the original heart sound. The background noise was unpredictable and uncontrollable, thus it needed to be eliminated or minimised. The friction noise was unpredictable as well, but it could be controlled by the physicians, by adjusting the movement of the chest piece of the stethoscope. Some kind of noises, such as human voice between the patients and physicians, could be predicted and controlled by the physicians during sound capturing.

Noise reduction is necessary to help the physicians to obtain the correct diagnostic information from the original heart sound combined with noise. As different noises impact the sound quality with different degree, the noise reduction would require analysis of heart sound signal and the noises it contains (Reed, T. R. et al. 2004). The analysis of the original heart sound has been identified in the literature, and four components of heart sound as well as their time-frequency domain characteristics were described in the literature review chapter. The analysis concluded that the heart sound signals are transient and periodical, while the characteristic of the noises is varied. For example, the ambient noise and disturbances usually have high amplitude and last for a short period of time (Tang et al. 2010), while other noises, like human breathing or voice, may be continues (Bai & Lu 2005).

The main research target of this study is to eliminate or minimise the noises introduced during the auscultation using a digital stethoscope. Thus the designed noise reduction method should be effective for different kinds of noise. As the characteristic of each noise is different, the designed noise reduction method should be tested for the heart sound mixed with each kind of the noise. Thus the classification of the noises and test of the method for different noises is necessary.

6.3 Noise reduction method

In this study, the original heart sound samples mixed with different kinds of noise are collected and restored in digital form. In the experiment, to investigate the effect of the noise reduction method, this method would be used for the heart sound signal mixed with five selected kinds of noise. These selected heart sound sample would be mixed with one certain kind of noise, thus the effect of the method for each kind of noise could be recorded during the experiment. To intuitively and rapidly examine the effect of the noise reduction method, two main factors have been explained in the experiment design: the kind of noise and the noise level.

6.3.1 Noise classification

In reality, heart sound signals are often corrupted by various noises, which can prohibit the accuracy of the original sound (Varady 2001). Noise exited in the real world has different types, which would affect the sound quality in different ways. The ambient noise, disturbances, sliding movements and surround speech sounds can all affect the accuracy of data being acquired.

As the sources and characteristics of noises vary, it is necessary to classify the noises before the noise reduction method is investigated. Varady categorized the noises by two aspects: external and internal factors (2001). The external noise included the sliding movements of the stethoscope diaphragm, ambient noise, instrument noise, human voice and the patient's movement; and the internal noise included the breathing noise and the damping through bones and tissues. Compared with the internal noise, the external noise is a main reason of noise corruption and body sounds are difficult to hear in complete statement. Furthermore, the physicians had different requirements for different kind of noise (Jatupaiboon, Pan-ngum & Israsena 2010). For example, the physicians were more sensitive to the long-time ambient noise and required these noises to be mostly eliminated; while they paid less attention to the friction-induced noise, because this noise accrued when they moved the chest piece on the patient body, the auscultation results during these time periods were not applicable.

In the literature review, most of the previous research studied the noise reduction method with certain kind of noise or the manual noise. Although some studies indicated the classification of the noises, less of them test the noise reduction method with each kind of noises or mixed noises collected from real life. Additionally, few studies identified and summarised the effectiveness of the noise reduction method with different noises as this study.

The finding of the study which study suggested that in order to test the efficiency of the noise reduction method, the noise which most commonly appeared, should be chosen to test the noise reduction method.

In this study, all the noise resources were selected and classified from the real heart recorded collected in India and Australia. From the 68 heart sound samples we collected, five most common noises were selected from all kinds of the noises. Four of them, the ambient noise, the noise burst, the human voices and the friction noise were external noises as the external factors were the main reason of the noise corruption. One internal noise, the respiration sounds, was selected as well.

In the experiment, each selected heart sound sample contained one (or two) kind of noise. All the samples were classified into five different groups by the noise it contained. Thus it would be easy to test the efficiency of the newly designed noise reduction for certain kind of noise through the experiment by comparing the difference between the processed and original sound sample.

For each kind of noise, five pieces of the sound samples (contained the same kind of noise) were investigated with the noise reduction method (Otsu's noise reduction method). The method would judge and eliminate the noise part of the signal automatically. During the noise reduction progress, the de-noising parameter was fixed as a constant (defaulted set as 3). This ensured that the effect of noise reduction for each kind of noise would be the same. The result of each kind of the noise reduced signal was recorded in the Experiment Chapter. Compared with the original signal, the effect of the noise reduced method for each kind of noise is described in the table below.

The effect of the noise reduction for each kind of the noise is listed in Table 6.1:

Table 6.1 The effect of the noise reduction method

Kind of noise	Noise description	Noise reduction result
Respiration sounds or breathing noise	• Patient's breaths noise • Noise frequency component is certain • Need eliminate	• Most of the breathing noise has been reduced • Noise component mixed with heart sound signal is partly remained • Main heart sound component is kept
Ambient noise	• Common noise • Noise frequency component is vary • Need eliminate urgently	• Effect differ from each other • Effect better in shot-time noise, not obviously in continuous noise
Noise burst	• No-priori noise • Mixed with other noises • Noise frequency component is wide • Need eliminate urgently	• Eliminate effectively • Main heart sound component is identified and kept
Human voices	• Speak during auscultation • Noise component mixed with heart sound • No need to eliminate	• Frequency domain of the noise covers that of heart sound • Hardly identified or reduced
Friction-induced noise	• Chest piece moves on body • Noise component is wide • Need to eliminate	• Hardly identify the noise • Both noise and heart sound signal are kept

The result of the experiment showed that the effect of de-noising method differed by noises. For the respiration noise and noise burst, this method could eliminate the noise properly. The effect for ambient noise depended on the kind of input noise, and the method did not work properly with human voice and friction-induced noise. In consequence, it could be concluded that this method effected better in short term noise but did not work properly with some kinds of continue noise.

6.3.2 Noise level

It showed in the literature review that most of the papers test their noise reduction method with certain kind of artificial noise. Jatupaiboon used white, pink, babble and factory noises (2010), Belloni et al. (2010) only used the white noise. Belloni et al. (2007) used heart sound signals in both empirical and realistic conditions to estimate the coefficient of the digital filter.

In this study, the real heart sound signals were collected form Australian and Indian health centres. The experiment made use of these signals with different noise levels, to test the adaptive filter in more realistic experimental conditions. As the original heart sound signals were mixed with noises, it was hard to provide the SNR for each piece of the sample. Thus the ratio of the threshold value and max value (RV), the ratio of kept frequency component and the whole frequency (RF) were selected as the factors to describe the efficiency of the adaptive filter.

The results reported in chapter 5 showed the function of the noise reduction method with different noise levels. For the pure heart sound signals group, the results proved that most part of the heart sound component had been kept. The results also showed that little addictive noises appeared in the processed noise. For each kind of the noise, the floating threshold method was effective. Three kinds of noises, with which the noise reduction method worked properly, were examined in this study. The ratio of threshold value and max value represent the floating threshold (RV), and the ratio of kept frequency component and the whole frequency (RF) represent showed the affection of the noise reduction method.

The noise level and results of noise reduction are listed in the Table 6.2:

Table 6.2 The effect of the floating threshold

Kind of noise	RV	RF
Pure heart sound (little noise)	22.7% 22.38% 17.75% 21.57% 24.72%	86.67% 75% 91.67% 80% 71.5%
Respiration sounds or breathing noise	22.02% 22.02% 20.46% 25.90% 19.57%	60% 58.8% 68.75% 50% 66.67%
Ambient noise	24.24% 21.25% 17.86% 16.8% 18.08%	42.8% 55.5% 68% 56.25% 61.5%
Noise burst	20.35% 22.46% 17.57% 16.85% 25.47%	80% 52.9% 61.3% 79% 58.82%

From the table 6.2, the floating threshold fluctuated due to the input signal. When the input signal was the pure heart sound (contain little noise), most frequency component of the signal would be kept (70%~90%). While when the signal mixed with noise comes, the ratio reduced sharply. For certain kind of noise, the threshold floated with the SNR of the input signal. For example the heart sound mixed with the breathing noise, when the rate of value (RV) rose up (25.9%), which meant more noise contained in this piece of noise, the rate of frequency (RF) fell down (50%) and less part of the frequency component was kept in the processed signal; while when the threshold level fell down (19.57%), more frequency component was kept (66.67%). The other two kind of noise showed the similar results. Thus it proved that the floating threshold method worked for those three certain noises. As this method did not work properly with the human voices or friction-induced noise, the result for them had not been listed in the table.

Chapter 7 Conclusion

With the development of digital technology in recent years, the digital stethoscope, with the advantage of restoring and replaying function, is gradually replacing the acoustic stethoscope. However, the sound corruption in digital stethoscope is still a problem to the users. The problem motivated this research to analyse and discuss the noise elements of the digital stethoscope, and then to find a suitable solution to reduce unwanted noise in the digital signal processing area.

Thus, the research question of this research is identified as: **How might the noise be reduced from the heart sound records collected from digital stethoscope with suitable noise reduction method.**

To address this problem and to concentrate on the experiment and analysis, three research sub-questions were designed.
RQ 1 What kind of the possible noises is in the output signals and what are their spectrums respectively?
RQ 2 How to develop suitable noise reduction method for the noises mentioned in the above question?
RQ 3 How to validate that the heartbeat signal or the other useful information would not be distorted after processing?

7.1 Structure of this research

The objective of this research is to find a better solution of noise reduction method for the heart sound recordings from a digital stethoscope. The overarching aim of the study is to improve the quality of heart sounds for auscultation and to help the physicians' diagnose.

A multiple-stage research design was developed to answer the research question posited above and achieve the objectives of this study. These stages included initiation study, method design, experiment and the result.

In the initiation stage, the idea of this research was formed and identified through the initial review of the literature research. The output of this stage was the research proposal which was approved and supported by the Faculty and University. In this research proposal, several essential parts were concluded, such as the review of related literature, the research question and research objectives, the research method introduction and framework, and the possible contribution. The literature review part offered necessary background knowledge for this study and the synthesis of noise reduction. Then the research question and objectives were identified. The methodology part reviewed the research paradigms in Information System (IS), compared their difference and then designed the appropriate experiment method for this study. The experiment described the process of the experiment and tested the collected data with the designed noise reduction method. The results were presented in appropriate formats for easy reading.

In the literature review, the necessary knowledge for this study, included heart sound, stethoscope, relevant concepts of noise reduction and the de-noising algorithm methods applied for heart sound. Heart sound and heart sound analysis offer the characteristics and forms of heart sound. The introduction of current digital stethoscope describes the sources of sound samples. The review of several commonly used noise reduction methods provides the scope and possible noise reduction solutions. The difference between these methods, including their strength, weakness and efficiency for noise reduction are described respectively and then summarized in a table. The probability and advantage of applying those methods to noise reduction for heart sound is also discussed. Then the introduction of a special technique called the Otsu's method is provided and its current application in signal processing. The research question and objectives are provided at the end. The result showed the advantage of the Otsu's method, and the improvement of the sound quality after noise reduction.

In the research methodology, the research paradigms were reviewed, and research strategies in Information System (IS) and compared their differences. Then the research approach and methodology applied for this study was provided. Laboratory experiment was selected as the suitable research method as its scientific and obtaining data feature. In research model and instrument, the details and steps of the experiment was designed which was focused on answering the three sub-questions.

In the data collection and analysis part, the approaches employed for data was provided along with a description of how the data was analysed will be discussed. The process of data collection and analysis for the research was introduced and explained in several sections: the definition of the format of the data required for the experiment, the real collected data discussion of the application of the algorithm applied in the experiment and the adjustment of the coefficient, the data which is tested with different threshold levels, and the results in figures.

In the finding and results of the research, the benefit and advantage of the new designed reduction method was described with the results of the experiment. The Otsu's method proved to be better in heart sound noise reduction because of its optimal division, no priori characteristics, fluctuation threshold and changeable de-noising parameter. The comparison of sound quality between the original and processed heart sound support the research objectives. The finding also offered the appropriate classification of the noise contained in the heart sound records.

7.2 The solution of research question

This study attempts to answer the research question,' **How might the noise be reduced from the heart sound records collected from digital stethoscope with suitable noise reduction method.'** To address the research target, the main work contains three major components: identifying noises, designing noise reduction techniques and evaluating the proposed method. Three sub-questions have been discussed correspond to each component.

RQ 1 What kind of the possible noises is in the output signals and what are their spectrums respectively?
In this research, five main kinds of noise have been identified as main noises to corrupt the quality of heart sound records from DS. These noises include: The respiration sounds or breathing noise, the ambient noise, the noise burst, the human voices and the friction-induced noise. The spectrum and features of each noise is recorded in chapter 5.4.4 'The experiment records'. The noises which corrupted the heart sound signals were collected

in real-life environments. The heart sound samples were classified into five sound groups with each kind of noises.

RQ 2 How to develop suitable noise reduction method for the noises mentioned in the above question?

After classifying and investigating the input signals, several features of the main noises are confirmed. The noise can corrupt the heart sound signal in many ways and always be changeable. It is impossible to predict the noise kind, noise level or the time when noise appears. Thus the designed noise reduction method should include the following features: no need for priori noise characteristics; a fluctuating noise reduction threshold to adapt different SNR noise; a changeable reduction parameter to adjust the de-noising value with different noise.

In the literature review, little principle proves that the current thresholding function is designed particular for heart sound de-noising (2.3.2 Noise reduction for heart sound). Otsu's method, however, provides all the above features and has been applied in picture segmentation before. Thus Otsu's method has been chosen as the probable suitable noise reduction method in this research (6.1 Otsu's noise reduction method).

RQ 3 How to validate that the heartbeat signal or the other useful information would not be distorted after processing?

In the experiment, except the five noise sound group, there is a separate sound group named 'pure heart sound', which includes the heart sound samples with limited noise corrupted. Before the noise reduction step, the effectiveness of the designed method has been tested with pure heart sound first (5.4.4 the experiment records). Result shows the pure heart sound has little change with Otsu's method. In Literature Review chapter, the main frequency component of heart sound is confirmed as 40 – 500 Hz (2.2.1 the heart sound analysis and heart sound signal). Through the experiment of noise corrupted heart sound, those part of frequency component have been mostly kept after signal processing (5.4.3 the result and analysis of the experiment). The result of the experiment also proves that the heart sound signal would not be distorted by the noise reduction process.

7.3 Theoretical and practical implication

Otsu's method has been successfully applied into a wide range of area, such as human action recognition and picture segmentation. The details of Otsu's method application is in 'Chapter 3 Otsu's Method'. In this research, Otsu's method is applied to the noise reduction for heart sound signals. In the experiment, the Otsu's method separated the input signal to two parts and found the minimal value of combined spread (intra- class variance) between heart sound signal and noise in frequency domain. With the advantage of Otsu's method (optimal threshold, no priori noise characteristics and fluctuating threshold), the experiment proves that the noises with different kinds and degrees, mixed in the heart sound signal can be de-noised automatically. The experiment also proves that the de-noising threshold is adaptive according to the quality of input heart sound and the de-noising parameter could be adjusted for different kinds of noise. In brief, Otsu's method is firstly utilized in noise reduction area in this research. That method proved to be effective to heart sound signal through the experiment. This is the main theoretical implication of this research.

In this research, the input testing signals are real heart sound signals which collected from health centre in both Australia and India. Five kinds of noises are determined as main noise corruption for heart sound. The collected heart sounds are classified in to 5 groups with certain kind of noise (details in '5.4.4 the experiment records'). Then the effectiveness of the Otsu's method is demonstrated by each group of noise. The effect of the floating threshold is recorded and compared by different degree of noise corruption in '6.3.2 Noise level'. Thus the designed method could be downloaded into a hard chip in digital stethoscope for real auscultation. It is the main practical implication in this research.

7.4 Limitation of this research

There are certain limitations in this study. In the stage of experiment study, considered the time, accessibility and cost, all the data (sound samples) are second hand data. All the sound samples were collected by the physician and their assistants in the hospital in both Australia and India. The data were collected from the real environment, it was impossible to obtain the data with a completely silent room. Thus the reference signal is with limited noise.

With the limitation of the digital stethoscope, the heart sound samples were stored as MP3 form before the research. Thus these sound samples lost some high frequency component before the signal processing

Another limitation was that the development of the noise reduction is still in initial stages. This increases the difficulty of identifying a suitable noise threshold for each kind of noise. Then appropriate threshold parameter levels have to be determined to fit the main noises in the heart sound. Moreover, the relationship between the SNR, the fluctuating threshold and the adjustment of de-noising parameter need to be researched more with more sound samples or in practical use.

7.5 Future improvement

The objective of this study was to provide a suitable noise reduction method for the digital stethoscope sounds. Thus the designed method needs to be compiled as a software algorithm and then downloaded into a hard chip to achieve the usage of the noise reduction in real life.

The Otsu's noise reduction method is conducted in frequency domain. The transformation between time and frequency domain is time cost. It can prove the transforming of data (from time domain to frequency domain) from FFT transform to the wavelet transform.

In this stage, the research objective was to focus on the noise reduction for heart sound records. As the development of the different digital stethoscope, different models for certain aim can be set for stethoscope. Thus this noise reduction method can be spread and applied to other sounds (lung sound for example).

References

Abbas, A & Bassam, R 2009, 'Phonocardiography Signal Processing', *Synthesis Lectures on Biomedical Engineering*, vol. 4, no. 1, pp. 1-194.

Alvarez, W 1922, 'The electrogastrogram and what it shows', *JAMA*, vol. 78, no. 15, p. 1116.

Antoniadis, A, Bigot, J & Sapatinas, T 2001, 'Wavelet estimators in nonparametric regression: a comparative simulation study', *Journal of Statistical Software*, vol. 6, no. 6, pp. 1-83.

Argyris, C, Putnam, R & Smith, DML 1985, 'Action science: Concepts, methods, and skills for research and intervention'.

Arnold, DO 1970, 'Dimensional sampling: An approach for studying a small number of cases', *The American Sociologist*, vol. 5, no. 2, pp. 147-50.

Arnott, P, Pfeiffer, G & Tavel, M 1984, 'Spectral analysis of heart sounds: relationships between some physical characteristics and frequency spectra of first and second heart sounds in normals and hypertensives', *Journal of biomedical engineering*, vol. 6, no. 2, pp. 121-8.

Arseneau, S & Cooperstock, JR 1999, 'Real-time image segmentation for action recognition'.

Asari, K, Srikanthan, T, Kumar, S & Radhakrishnan, D 1999, 'A pipelined architecture for image segmentation by adaptive progressive thresholding', *Microprocessors and Microsystems*, vol. 23, no. 8, pp. 493-9.

Avgerou, C 2002, *Information systems and global diversity*, Oxford University Press, USA.

Bai, Y-W & Lu, C-L 2005, 'The embedded digital stethoscope uses the adaptive noise cancellation filter and the type I Chebyshev IIR bandpass filter to reduce the noise of the heart sound', paper presented to Enterprise networking and Computing in Healthcare Industry, 2005. HEALTHCOM 2005. Proceedings of 7th International Workshop on.

Barber, J, Magidson, O & Wood, P 1950, 'Atrial septal defect with special reference to the electrocardiogram, the pulmonary artery pressure and the second heart sound', *British heart journal*, vol. 12, no. 3, pp. 277-92.

Baskerville, RL & Wood-Harper, AT 1996, 'A critical perspective on action research as a method for information systems research', *Journal of Information Technology*, vol. 11, no. 3, pp. 235-46.

Belloni, F, Della Giustina, D, Malcangi, M & Riva, M 2010, 'A new digital stethoscope with environmental noise cancellation', paper presented to In: Advances in Mathematical and Computational Methods. Editors: L. Rogozea, Stevens Point, WSEAS International

Conference on Mathematical and Computational Methods in Science and Engineering (MACMESE'10), Faro, Portugal.

Belloni, F, Della Giustina, D, Riboldi, S, Riva, M, Spoletini, E & Bertossi, L 2007, 'Towards a computer-aided diagnosis by means of phonocardiogram signals', paper presented to Industrial Electronics, 2007. ISIE 2007. IEEE International Symposium on.

Benbasat, I 1988, 'Laboratory experiments in information systems with a focus on individuals: A critical appraisal', *The Information Systems Research Challenge Experimental Research Methods*, no. 88-MIS-023, pp. 33-47.

Benbasat, I, Goldstein, DK & Mead, M 1987, 'The case research strategy in studies of information systems', *MIS quarterly*, pp. 369-86.

Benesty, J, Chen, J, Huang, Y & Cohen, I 2009, *Noise reduction in speech processing*, vol. 2, Springer Verlag.

Benton, TG, Solan, M, Travis, JMJ & Sait, SM 2007, 'Microcosm experiments can inform global ecological problems', *Trends in ecology & evolution*, vol. 22, no. 10, pp. 516-21.

Berouti, M, Schwartz, R & Makhoul, J 2003, 'Enhancement of speech corrupted by acoustic noise'.

Bing-lian, Z & Qian, L 2006, 'Denoising of Heart Sound Signal Based on Adaptive Wavelet', *Computer*, p. 10.

Bishop, P 1980, 'Evolution of the stethoscope', *Journal of the Royal Society of Medicine*, vol. 73, no. 6, p. 448.

Boland, RJ & Hirschheim, RA 1992, *Critical issues in information systems research*, John Wiley & Sons, Inc.

Broun, CC, Zhang, X, Mersereau, RM & Clements, M 2002, 'Automatic speechreading with application to speaker verification'.

Brusco, M & Nazeran, H 2004, 'Digital phonocardiography: a PDA-based approach', paper presented to Engineering in Medicine and Biology Society, 2004. IEMBS'04. 26th Annual International Conference of the IEEE.

Casebeer, AL & Verhoef, MJ 1997, 'Combining qualitative and quantitative research methods: considering the possibilities for enhancing the study of chronic diseases', *Chronic Diseases in Canada*, vol. 18, no. 3, pp. 18-3.

Chourasia, V & Mittra, A 2009, 'A Comparative Analysis of De-Noising Algorithms for Fetal Phonocardiographic Signals', *IETE Journal of Research*, vol. 55, no. 1, p. 10.

Chourasia, V 2009, 'A Comparative Analysis of De-Noising Algorithms for Fetal Phonocardiographic Signals'.

Conway, A 2009, 'Giving adult learners a voice after the inquest: A review of a mixed method approach to educational research'.

Coombs, MA & Moorse, SE 2002, 'Physical assessment skills: a developing dimension of clinical nursing practice', *Intensive and Critical Care Nursing*, vol. 18, no. 4, pp. 200-10.

Cousin, G 2005, 'Case study research', *Journal of geography in higher education*, vol. 29, no. 3, pp. 421-7.

Danahy, E, Agaian, S & Panetta, K 2005, 'Non-linear algorithms for noise removal from medical signals using the logical transform'.

Davis, GM 2002, *Noise reduction in speech applications*, CRC.

Dilley, P 2004, *Interviews and the philosophy of qualitative research*, 1, JSTOR, 0022-1546.

Donoho, D & Johnstone, J 1994, 'Ideal spatial adaptation by wavelet shrinkage', *Biometrika*, vol. 81, no. 3, p. 425.

Drazner, MH, Rame, JE, Stevenson, LW & Dries, DL 2001, 'Prognostic importance of elevated jugular venous pressure and a third heart sound in patients with heart failure', *New England Journal of Medicine*, vol. 345, no. 8, pp. 574-81.

Durand, J, Durand, L-G & Grenien, M-C 1997, *Electronic stethescope*, Google Patents.

Durand, L, Blanchard, M, Cloutier, G, Sabbah, H & Stein, P 2002, 'Comparison of pattern recognition methods for computer-assisted classification of spectra of heart sounds in patients with a porcine bioprosthetic valve implanted in the mitral position', *Biomedical Engineering, IEEE Transactions on*, vol. 37, no. 12, pp. 1121-9.

Elliott, J & LUKEŠ, D 2008, 'Epistemology as ethics in research and policy: The use of case studies', *Journal of Philosophy of Education*, vol. 42, pp. 87-119.

Fang, H & Huang, D 2004, 'Noise reduction in lidar signal based on discrete wavelet transform', *Optics Communications*, vol. 233, no. 1-3, pp. 67-76.

Ferreira, P 1999, 'Mathematics for multimedia signal processing II: Discrete finite frames and signal reconstruction', *NATO ASI SERIES F COMPUTER AND SYSTEMS SCIENCES*, vol. 174, pp. 35-54.

Fluhr, R, Moses, P, Morelli, G, Coruzzi, G & Chua, NH 1986, 'Expression dynamics of the pea rbcS multigene family and organ distribution of the transcripts', *The EMBO journal*, vol. 5, no. 9, p. 2063.

Frigo, M 1999, 'A fast Fourier transform compiler', *ACM SIGPLAN Notices*, vol. 34, no. 5, pp. 169-80.

Galliers, R & Sutherland, A 1991, 'Information systems management and strategy formulation: the ‘stages of growth’model revisited', *Information Systems Journal*, vol. 1, no. 2, pp. 89-114.

Gavrovska, A, Slavkovic, M, Reljin, I & Reljin, B 2013, 'Application of wavelet and EMD-based denoising to phonocardiograms', paper presented to Signals, Circuits and Systems (ISSCS), 2013 International Symposium on.

Glover Jr, J 2003, 'Adaptive noise canceling applied to sinusoidal interferences', *Acoustics, Speech and Signal Processing, IEEE Transactions on*, vol. 25, no. 6, pp. 484-91.

Goodwin, G & Sin, K 2009, *Adaptive filtering prediction and control*, Dover Publications, Inc. New York, NY, USA.

Gopinath, R & Burrus, C 2002, 'Efficient computation of the wavelet transforms'.

Hammersley, M 2012, 'Troubling theory in case study research', *Higher Education Research & Development*, vol. 31, no. 3, pp. 393-405.

Harley, T 1997, *Active noise control stethoscope*, Google Patents.

Hess, F, Kraft, M, Richter, M, Bockhorn, H & Dcnoising, A 1997, 'Comparison and assessment of various wavelet and wavelet packet based denoising algorithms for noisy data', *Universit‰ t Kaiserslautern*.

Hubbard, B & Meyer, F 1998, *The world according to wavelets: the story of a mathematical technique in the making*, AK Peters Wellesley.

Huiying, L, Sakari, L & Iiro, H 2002, 'A heart sound segmentation algorithm using wavelet decomposition and reconstruction'.

Jansen, M & Bultheel, A 2002, 'Asymptotic behavior of the minimum mean squared error threshold for noisy wavelet coefficients of piecewise smooth signals', *Signal Processing, IEEE Transactions on*, vol. 49, no. 6, pp. 1113-8.

Jatupaiboon, N, Pan-ngum, S & Israsena, P 2010, 'Electronic stethoscope prototype with adaptive noise cancellation', paper presented to Knowledge Engineering, 2010 8th International Conference on ICT and.

Jiwa, M, Millett, S, Meng, X & Hewitt, VM 2012, 'Impact of the Presence of Medical Equipment in Images on Viewers’ Perceptions of the Trustworthiness of an Individual On-Screen', *Journal of Medical Internet Research*, vol. 14, no. 4.

Kaneda, Y & Ohga, J 2003, 'Adaptive microphone-array system for noise reduction', *Acoustics, Speech and Signal Processing, IEEE Transactions on*, vol. 34, no. 6, pp. 1391-400.

Kaplan, B & Duchon, D 1988, 'Combining qualitative and quantitative methods in information systems research: a case study', *MIS quarterly*, pp. 571-86.

Kell, DB & Oliver, SG 2004, 'Here is the evidence, now what is the hypothesis? The complementary roles of inductive and hypothesis - driven science in the post - genomic era', *Bioessays*, vol. 26, no. 1, pp. 99-105.

Khadra, L, Matalgah, M, El-Asir, B & Mawagdeh, S 1991, 'The wavelet transform and its applications to phonocardiogram signal analysis', *Informatics for Health and Social Care*, vol. 16, no. 3, pp. 271-7.

Laennec, RTH 1819, *De l'auscultation médiate: ou, Traité du diagnostic des maladies des poumons et du coeur; fondé principalement sur ce nouveau moyen d'exploration*, vol. 2, Culture et civilisation.

Lasky, M 1977, 'Review of undersea acoustics to 1950', *The Journal of the Acoustical Society of America*, vol. 61, p. 283.

Latham, GP, Erez, M & Locke, EA 1988, 'Resolving scientific disputes by the joint design of crucial experiments by the antagonists: Application to the Erez–Latham dispute regarding participation in goal setting', *Journal of Applied Psychology*, vol. 73, no. 4, p. 753.

Lather, P 2006, 'Paradigm proliferation as a good thing to think with: Teaching research in education as a wild profusion', *International journal of qualitative studies in education*, vol. 19, no. 1, pp. 35-57.

Leatham, A 1970, *Auscultation of the Heart and Phonocardiography*, Churchill, London.

Lee, J, Lee, S, Kim, I, Min, H & Hong, S 2002, 'Comparison between short time Fourier and wavelet transform for feature extraction of heart sound'.

Lee, R & Fielding, N 1996, 'Qualitative data analysis: representations of a technology: a comment on Coffey, Holbrook and Atkinson'.

Lehmann, E & Casella, G 1998, *Theory of point estimation*, Springer Verlag.

Lei, L 2008, *http://www.ilovematlab.cn/thread-88080-1-1.html###*.

Li, P, Abbott, AL & Schmoldt, DL 1996, 'Automated analysis of CT images for the inspection of hardwood logs'.

Liao, PS, Chen, TS & Chung, PC 2001, 'A fast algorithm for multilevel thresholding', *Journal of information science and engineering*, vol. 17, no. 5, pp. 713-28.

Lin, AC 1998, 'Bridging positivist and interpretivist approaches to qualitative methods', *Policy Studies Journal*, vol. 26, no. 1, pp. 162-80.

Macdonald, D, Kirk, D, Metzler, M, Nilges, L, Schempp, P & Wright, J 2002, 'It's all very well, in theory: Theoretical perspectives and their applications in contemporary pedagogical research', *Quest*, vol. 54, no. 2, pp. 133-56.

Mackrell, D & Nielsen, S 2007, 'Structuration Theory in Information Systems Research: Relevance and Rigour from a Pluralist Research Approach'.

Mallat, S 1999, *A wavelet tour of signal processing*, Academic Pr.

Mangion, K 2007, 'The stethoscope', *Malta Med J*, vol. 19, pp. 41-4.

Markel, H 2006, 'The stethoscope and the art of listening', *New England Journal of Medicine*, vol. 354, no. 6, pp. 551-3.

Marro, C, Mahieux, Y & Simmer, K 2002, 'Analysis of noise reduction and dereverberation techniques based on microphone arrays with postfiltering', *Speech and Audio Processing, IEEE Transactions on*, vol. 6, no. 3, pp. 240-59.

Maxwell, SE & Delaney, HD 2004, *Designing experiments and analyzing data: A model comparison perspective*, vol. 1, Lawrence Erlbaum.

McGregor, SLT & Murnane, JA 2010, 'Paradigm, methodology and method: intellectual integrity in consumer scholarship', *International Journal of Consumer Studies*, vol. 34, no. 4, pp. 419-27.

McMurray, AJ, Scott, DR & Pace, RW 2004, 'The relationship between organizational commitment and organizational climate in manufacturing', *Human Resource Development Quarterly*, vol. 15, no. 4, pp. 473-88.

Medcalf, R 2010, 'The Experiences of Children and Young People with Social Emotional and Behavioural Difficulties in Physical Education', Coventry University in collaboration with the University of Worcester.

Messer, S, Agzarian, J & Abbott, D 2000, 'Optimal wavelet denoising for smart bio-monitor system s'.

Messer, SR, Agzarian, J & Abbott, D 2001, 'Optimal wavelet denoising for phonocardiograms', *Microelectronics Journal*, vol. 32, no. 12, pp. 931-41.

Miller, JWV, Shridhar, V, Wicke, E & Griffin, C 1998, 'Very low-cost in-process gauging system', *Machine Vision Systems for Inspection and Metrology VII, SPIE*, pp. 14-9.

Mingers, J 2003, 'The paucity of multimethod research: a review of the information systems literature', *Information Systems Journal*, vol. 13, no. 3, pp. 233-49.

Mishra, G, Biswal, K & Mishra, AK 'DENOISING OF HEART SOUND SIGNAL USING WAVELET TRANSFORM'.

Misiti, M, Misiti, Y, Oppenheim, G & Poggi, J 2001, 'Wavelet Toolbox, for use with MATLAB®, User's guide, version 2.1', *Online document: Mathwork*, pp. 1–18.

Mitra, SK & Kuo, Y 2006, *Digital signal processing: a computer-based approach*, vol. 2, McGraw-Hill New York.

Müller, P & Kompis, M 2002, 'Evaluation of a noise reduction system for the assessment of click-evoked otoacoustic emissions', *The Journal of the Acoustical Society of America*, vol. 112, p. 164.

Myers, MD & Avison, D 1997, 'Qualitative research in information systems', *Management Information Systems Quarterly*, vol. 21, pp. 241-2.

Obaidat, M 1993, 'Phonocardiogram signal analysis: techniques and performance comparison', *Journal of medical engineering & technology*, vol. 17, no. 6, pp. 221-7.

Orlikowski, WJ & Robey, D 1991, 'Information technology and the structuring of organizations', *Information systems research*, vol. 2, no. 2, pp. 143-69.

Otsu, N 1975, 'A threshold selection method from gray-level histograms', *Automatica*, vol. 11, pp. 285-96.

Palaniappan, R, Sundaraj, K, Ahamed, N, Arjunan, A & Sundaraj, S 2013, 'Computer-based Respiratory Sound Analysis: A Systematic Review', *IETE Technical Review*, vol. 30, no. 3, p. 248.

Parker, I 2002, *Critical discursive psychology [Electronic book]*, Palgrave Macmillan.

Patel, S, Callahan, T, Callahan, M, Jones, J, Graber, G, Foster, K, Glifort, K & Wodicka, G 1998, 'An adaptive noise reduction stethoscope for auscultation in high noise environments', *The Journal of the Acoustical Society of America*, vol. 103, p. 2483.

Pervan, G 2007, 'Information Systems Management: an Australasian view of key issues-1996', *Australasian Journal of Information Systems*, vol. 5, no. 1.

Pizurica, A, Philips, W, Lemahieu, I & Acheroy, M 2003, 'A versatile wavelet domain noise filtration technique for medical imaging', *Medical Imaging, IEEE Transactions on*, vol. 22, no. 3, pp. 323-31.

Reed, TR, Reed, NE & Fritzson, P 2004, 'Heart sound analysis for symptom detection and computer-aided diagnosis', *Simulation Modelling Practice and Theory*, vol. 12, no. 2, pp. 129-46.

Rhodes, T, Stimson, GV, Moore, D & Bourgois, P 2010, 'Qualitative social research in addictions publishing: Creating an enabling journal environment', *The International journal on drug policy*, vol. 21, no. 6, p. 441.

Rioul, O & Vetterli, M 2002, 'Wavelets and signal processing', *Signal Processing Magazine, IEEE*, vol. 8, no. 4, pp. 14-38.

Rosas-Orea, M, Hernandez-Diaz, M, Alarcon-Aquino, V & Guerrero-Ojeda, L 2005, 'A comparative simulation study of wavelet based denoising algorithms'.

Rubinstein, BB 1975, 'On the clinical psychoanalytic theory and its role in the inference and confirmation of particular clinical hypotheses', *Psychoanalysis and Contemporary Science*, vol. 4, no. 1, pp. 3-57.

Sardy, S, Antoniadis, A & Tseng, P 2004, 'Automatic smoothing with wavelets for a wide class of distributions', *Journal of Computational and Graphical Statistics*, vol. 13, no. 2, pp. 399-421.

Schuttler, F, Penzel, T & von Wichert, P 1996, 'Digital recording and computer-based analysis of lung sounds', paper presented to Engineering in Medicine and Biology Society, 1996. Bridging Disciplines for Biomedicine. Proceedings of the 18th Annual International Conference of the IEEE, 31 Oct-3 Nov 1996.

Sezgin, M 2004, 'Survey over image thresholding techniques and quantitative performance evaluation', *Journal of Electronic imaging*, vol. 13, no. 1, pp. 146-68.

Shanks, G 2007, 'Guidelines for conducting positivist case study research in information systems', *Australasian Journal of Information Systems*, vol. 10, no. 1.

Shuai, C & Renyi, S 2011, 'New Idea to Prolong Period of Sequence'.

Smith, S, Booth, K & Zalewski, M 1996, *International theory: positivism and beyond*, Cambridge Univ Pr.

Somekh, B & Lewin, C 2005, *Research methods in the social sciences*, Sage Publications Ltd.

Stranneby, D & Walker, W 2004, *Digital signal processing and applications*, Elsevier.

Straub, DW, Gefen, D & Boudreau, MC 2005, *Quantitative research*, Amsterdam: Elsevier.

Sund, T & Eilertsen, K 2003, 'An algorithm for fast adaptive image binarization with applications in radiotherapy imaging', *Medical Imaging, IEEE Transactions on*, vol. 22, no. 1, pp. 22-8.

Swope, WC, Andersen, HC, Berens, PH & Wilson, KR 1982, 'A computer simulation method for the calculation of equilibrium constants for the formation of physical clusters of molecules: Application to small water clusters', *The Journal of Chemical Physics*, vol. 76, p. 637.

Tang, H, Li, T, Park, Y & Qiu, T 2010, 'Separation of heart sound signal from noise in joint cycle Frequency–Time–Frequency domains based on fuzzy detection', *Biomedical Engineering, IEEE Transactions on*, vol. 57, no. 10, pp. 2438-47.

Taswell, C 1995, 'WavBox 4: A software toolbox for wavelet transforms and adaptive wavelet packet decompositions', *LECTURE NOTES IN STATISTICS-NEW YORK-SPRINGER VERLAG-*, pp. 361-.

Tian, H, Lam, S & Srikanthan, T 2003, 'Implementing Otsu's thresholding process using area-time efficient logarithmic approximation unit'.

Traver, GA 1973, 'Assessment of thorax and lungs', *The American Journal of Nursing*, vol. 73, no. 3, pp. 466-71.

Varady, P 2001, 'Wavelet-based adaptive denoising of phonocardiographic records', paper presented to Engineering in Medicine and Biology Society, 2001. Proceedings of the 23rd Annual International Conference of the IEEE.

Varady, P, TECHNOLOGY, BUO, ENGINEERING, EDOC & INFORMATION 2001, 'Wavelet-based adaptive denoising of phonocardiographic records'.

Vasios, G, Prentza, A, Blana, D, Salamalekis, E, Thomopoulos, P, ELECTRICAL, NTUADO & ENGINEERING, C 2001, 'Classification of Fetal Heart Rate Tracings Based on Wavelet-Transform & Self-Organizing-Map Neural Networks'.

Walker, H, Hall, W & Hurst, J 1990, *Clinical methods: the history, physical and laboratory examinations*, Butterworths Stoneham, MA.

Walsham, G 1995, 'The emergence of interpretivism in IS research', *Information systems research*, vol. 6, no. 4, pp. 376-94.

Webster, J 2009, *Medical instrumentation: application and design*, Wiley-India.

Werblud, MS 2001, *Earpiece, for use on stethoscope, having a harder, load-bearing portion and a softer, acoustic sealing portion*, Google Patents.

Woywodt, A, Herrmann, A, Kielstein, J, Haller, H, Haubitz, M & Purnhagen, H 2004, 'A novel multimedia tool to improve bedside teaching of cardiac auscultation', *British Medical Journal*, vol. 80, no. 944, p. 355.

Young, JR 2006, 'Scribes of the Digital Era', *Chronicle of Higher Education*, vol. 52, no. 21.

Yuenyong, S, Kongpravechnon, W, Tungpimolrut, K & Nishihara, A 2009, 'Automatic heart sound analysis for tele-cardiac auscultation', paper presented to ICCAS-SICE, 2009.

Zhang, Y, Chan, G, Zhang, X & Yip, L 'Heart Sounds and Stethoscopes'.---- 2006, 'Heart Sounds and Stethoscopes'.

Zhang, YT, Chan, G, Zhang, Xy & Yip, L 2006, 'Heart sounds and stethoscopes', *Wiley Encyclopedia of Biomedical Engineering*.

ZIN, RABMAT 2011, 'ELECTRONICS DIAGNOSTIC STETHOSCOPE'.

Appendix A: Programme of the Otsu's Method

```
clear all
close all
%-------------------------read signal and do FFT---------------------
--%
[Y,FS,NBITS]=wavread('05_fourbeats.wav');
[N,M]=size(Y);

YY=(Y(:,1)+Y(:,2))/2;
figure(3)
plot(YY);

axis([0 N -max(abs(YY)) max(abs(YY))]);
title('the orginal heart sound', 'fontsize', 13);
xlabel('time', 'fontsize', 12);
ylabel('amplitude', 'fontsize', 12);

fk=fft(YY);
absfk=abs(fk);


%-------------------------------------------------------------------
--%
% number of the signal is N, so N/2 is the sample

   NN=2000;
v=absfk(1:NN,:);

%--------------------------------------------------------------%
% assume L is the maximum value of the FFT

L = ceil(max(v));

%--------------------------------------------------------------

%--------------------------------------------------------------
% general bimodal distribution

%--------------------------------------------------------------

%--------------------------------------------------------------
x = 1:L;
n = hist(v, x);
%plot(n)
figure(1);
bar(x, n, 'b', 'linewidth', 2);
title('Distribution of Source Data', 'fontsize', 13);
grid('on');
set(gca, 'xlim', [1 L]);
set(gca, 'xtick');
xlabel('Data Value', 'fontsize', 12);
ylabel('Count', 'fontsize', 12);
hold('on');

sum(n)
p = n/NN;
```

```
p = p(:);
%----------------------------------------------------------------

%----------------------------------------------------------------
vbmax = 0;
kstar = 0;
vbplot = [];
kplot = [];

for k = 1:L
    kplot = [kplot k];
    w0 = sum(p(1:k));
    w1 = 1 - w0;

    if( (w0 < eps) || (w1 < eps ) )
        vbplot = [vbplot 0];
        continue;
    end
    i = [1:k]';
    mu0 = sum(i.*p(i))/w0;

    i = [k+1:L]';
    mu1 = sum(i.*p(i))/w1;

    vb = w0*w1*(mu1 - mu0)^2;
    vbplot = [vbplot vb];
    %w0
    %w1
    %vb

    if( vb > vbmax )
        vbmax = vb;
        kstar = k;
    end
    %pause
end
%----------------------------------------------------------------

fprintf(1, 'Optimal threshold kstar=%d\n', kstar);
figure(2);
clf
plot(kplot, vbplot, 'linewidth', 2);
title('Between-Class Variance as a Function of Threshold', 'fontsize',
13);
grid('on');
set(gca, 'xlim', [1 L]);
set(gca, 'xtick');
xlabel('Threshold Value', 'fontsize', 12);
ylabel('\sigma_B^2', 'fontsize', 12);
%----------------------------------------------------------------
% add threshold to histogram plot
figure(1);
ylimhist = get(gca, 'ylim');
line([kstar kstar], ylimhist, 'linestyle', ':', 'linewidth', 2)
buf = sprintf('\\leftarrow Optimal threshold \n        k = %.0d',
kstar);
text(kstar+2, ylimhist(2)*0.9, buf, 'BackgroundColor', 'w', 'Color',
'r', 'EdgeColor', 'w', 'FontSize', 12, 'FontAngle', 'italic');
%----------------------------------------------------------------
% add threshold value to FFT frequency figure
```

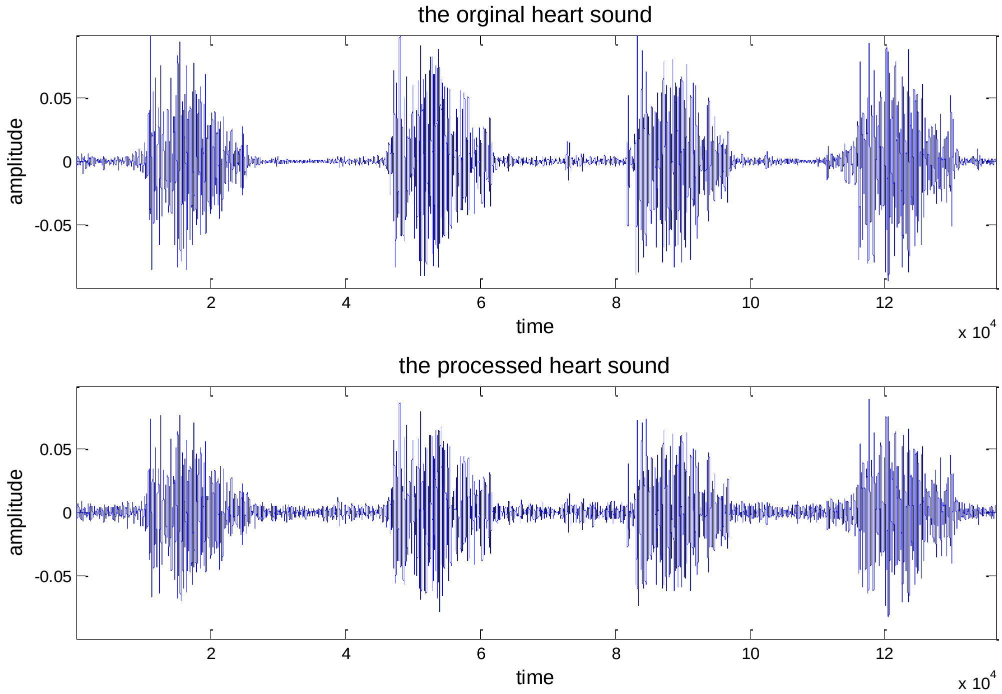
```
figure(4)
plot(v);
hold on
plot(1:NN,kstar);
title('the frequency domain of orginal heartsound ', 'fontsize', 13);
xlabel('frequency', 'fontsize', 12);
ylabel('amplitude', 'fontsize', 12);

%--------------------------------------------------------------
% delete the noise value under the threshold
% D is the adjust parameter to denoise the noise
fkabsclean=[];
fkclean=[];

de=3;
for i = 1 : N
    if absfk(i) < kstar
       fkclean=[fkclean fk(i)./de];
       fkabsclean=[fkabsclean absfk(i)./de];
    else
         fkclean=[fkclean fk(i)];
       fkabsclean=[fkabsclean absfk(i)];

    end
end

fkclean=fkclean';
fkabsclean=fkabsclean';


Yclean = ifft(fkclean) ;
Yclean = flipud(Yclean);
```

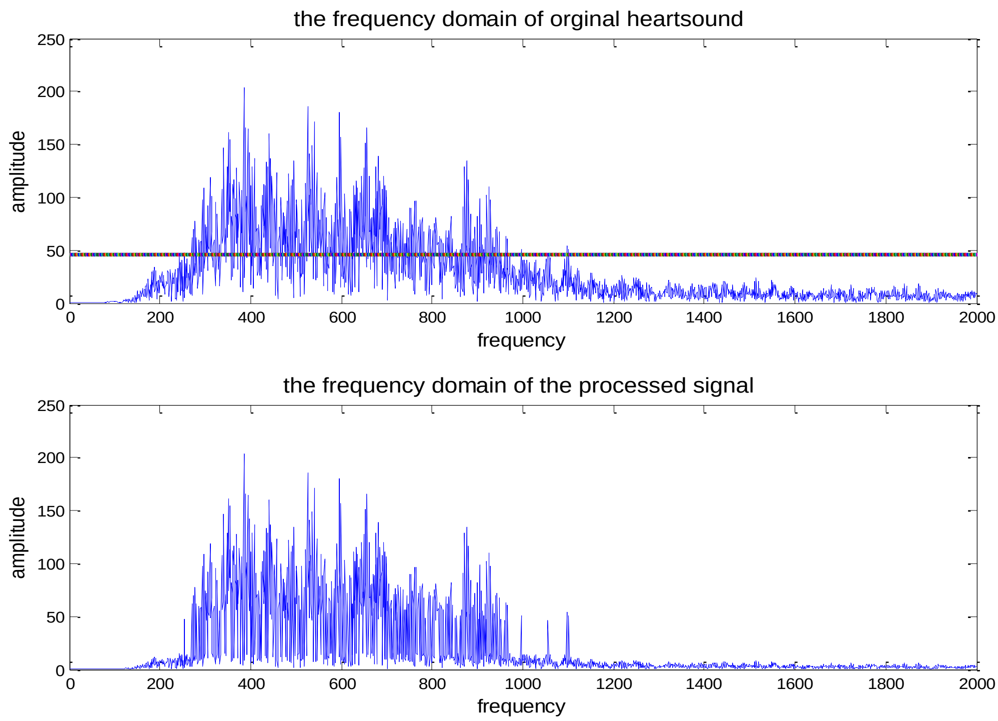
Appendix B: Experiment Results

In this appendix, the sound samples and of noise reduced form of these samples would be recorded by figures in both time and frequency domain.

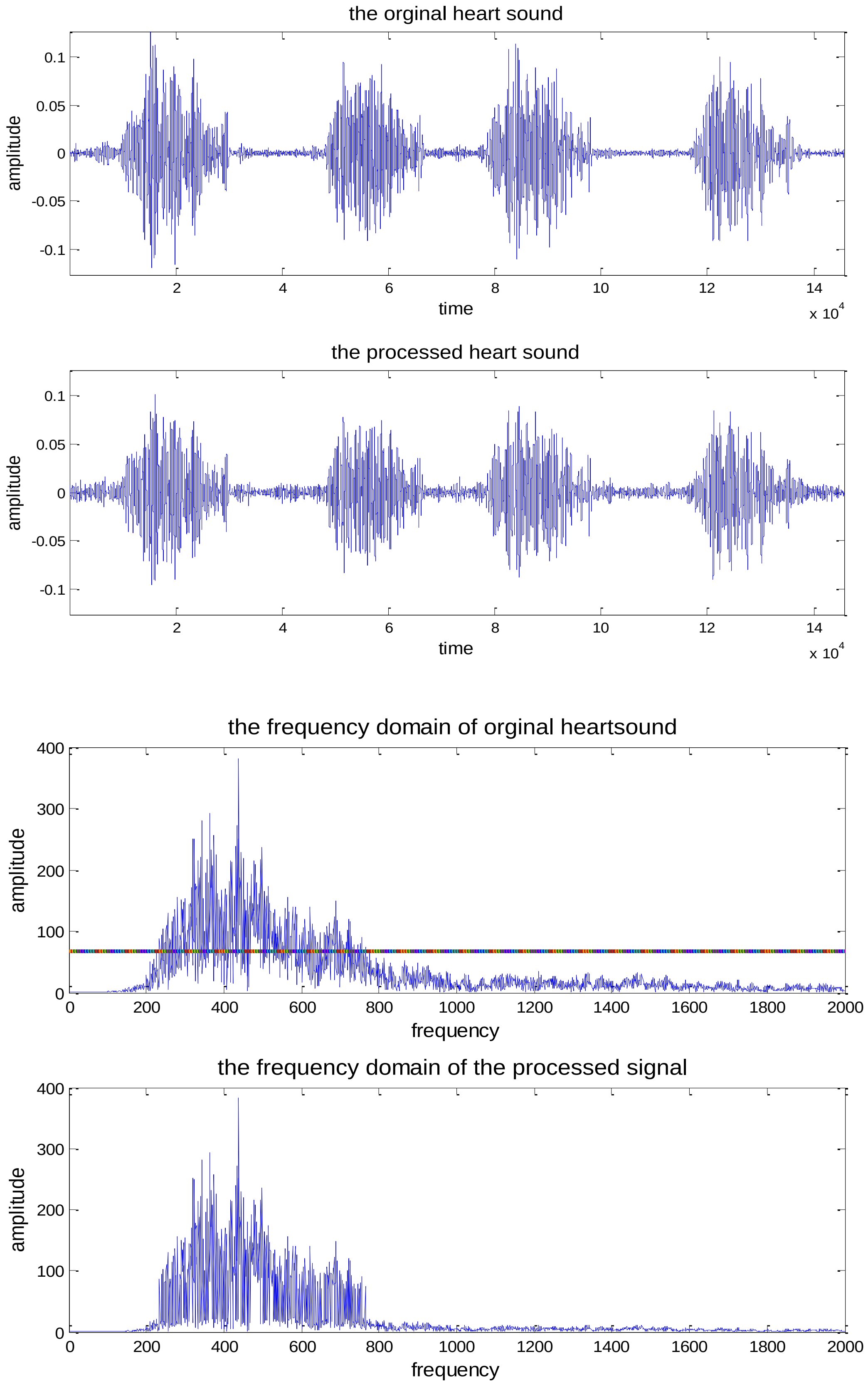
1. The pure heart sound

Sample 1:

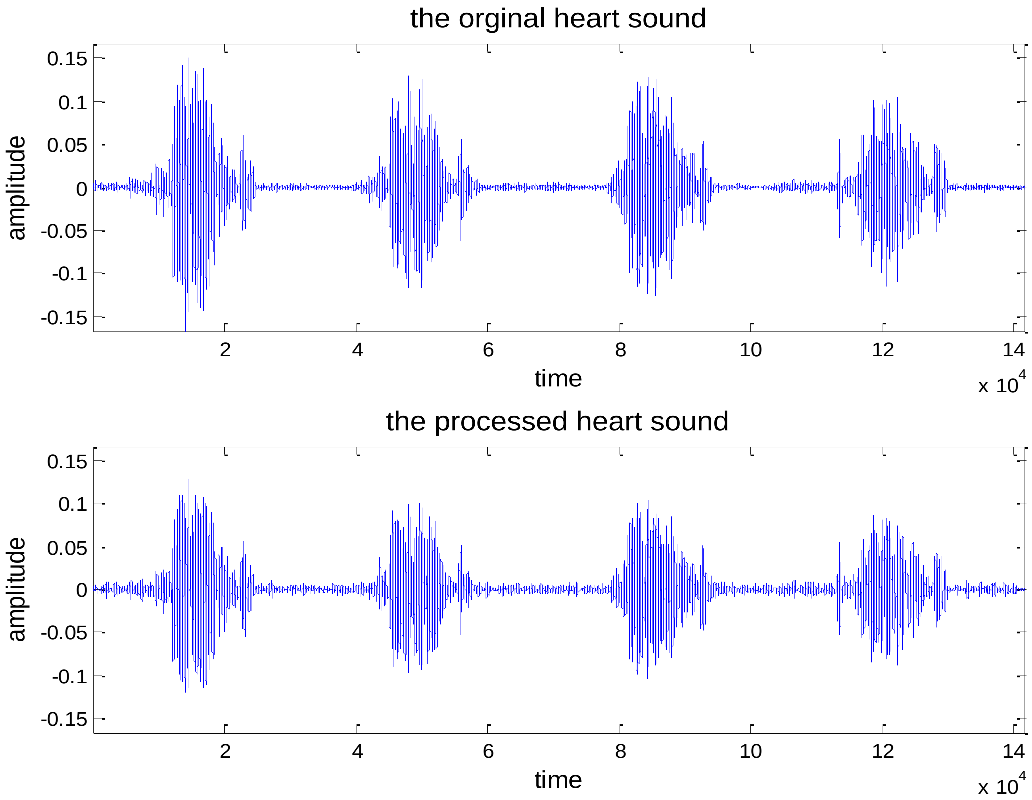




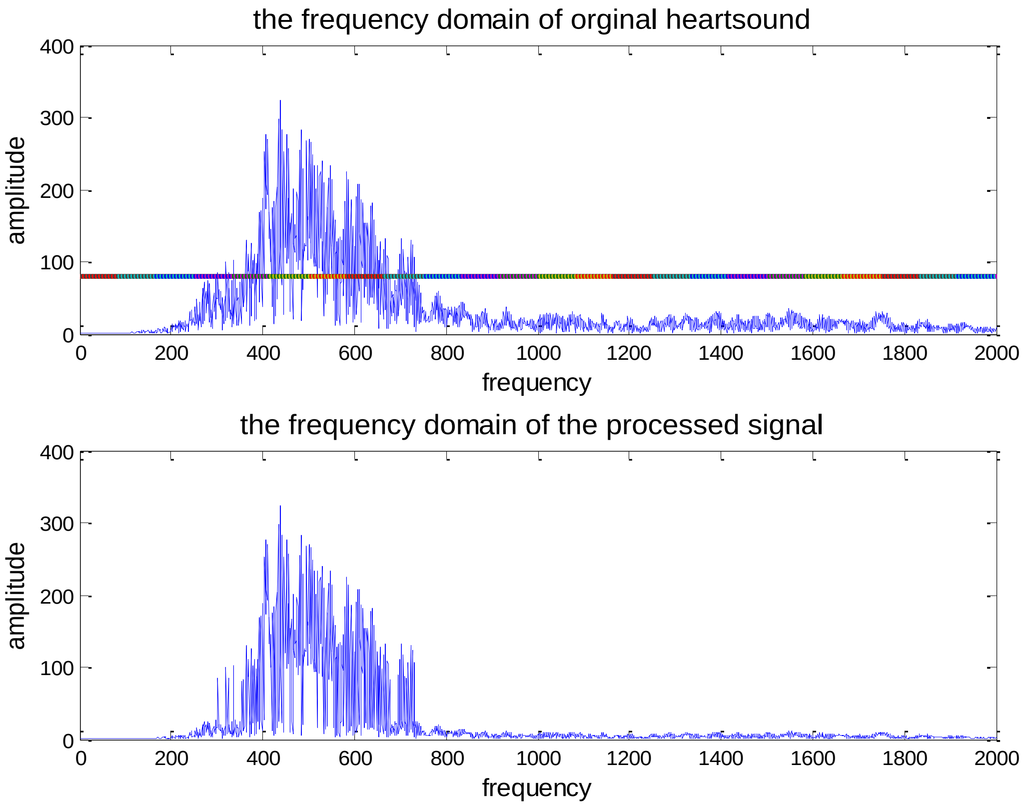
Sample 2:


the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4
the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

Sample 3:

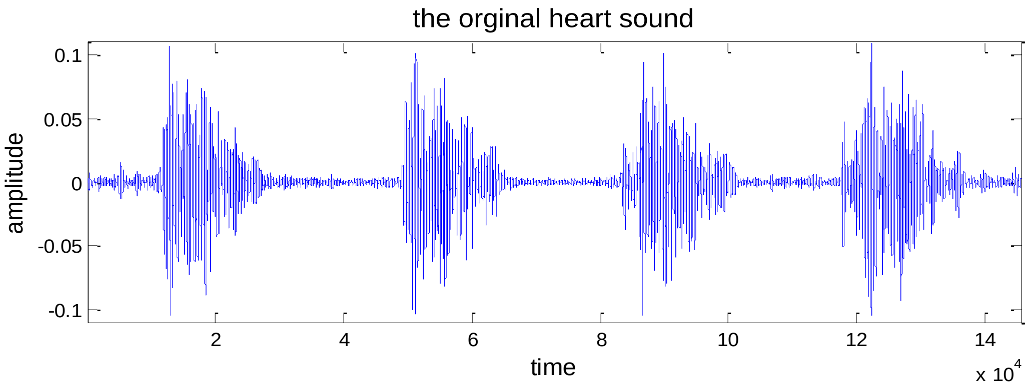


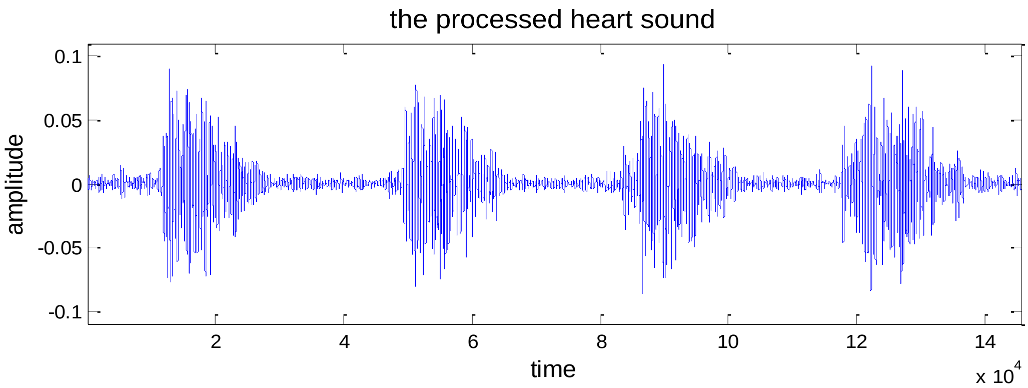
the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4

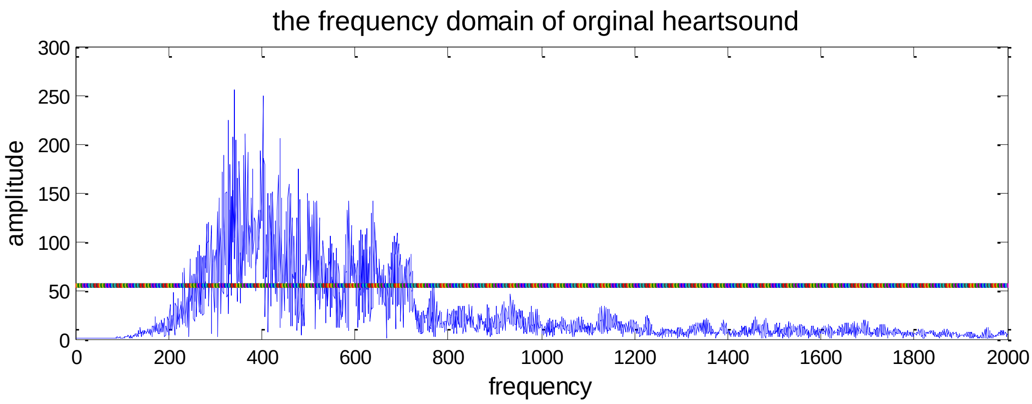


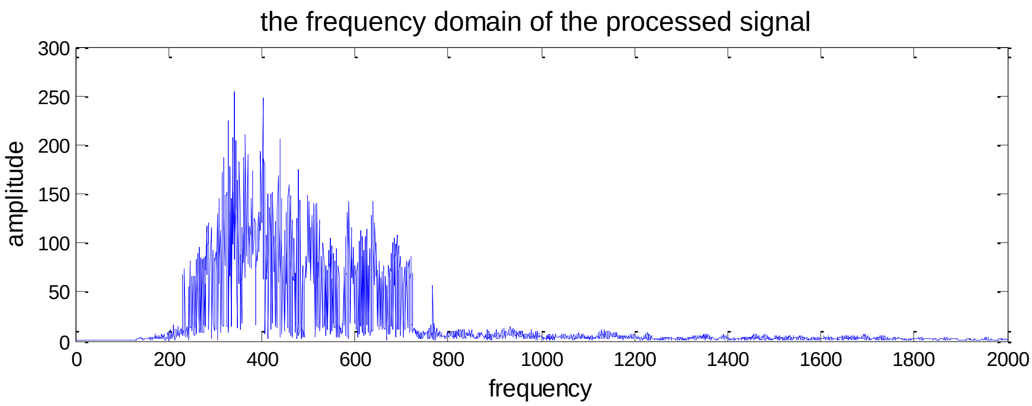
the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

Sample 4:

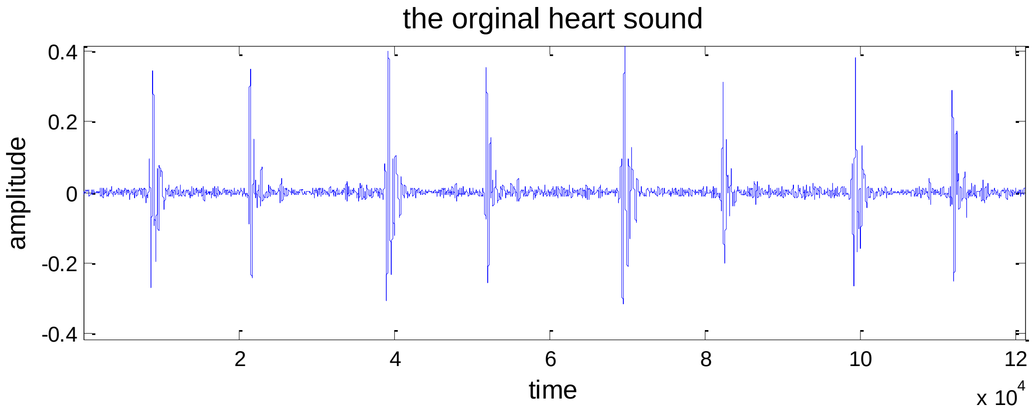

the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4

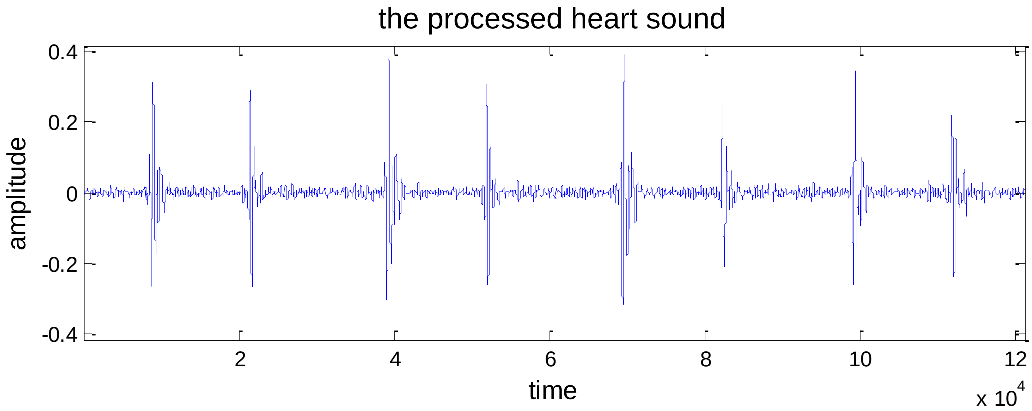

the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4

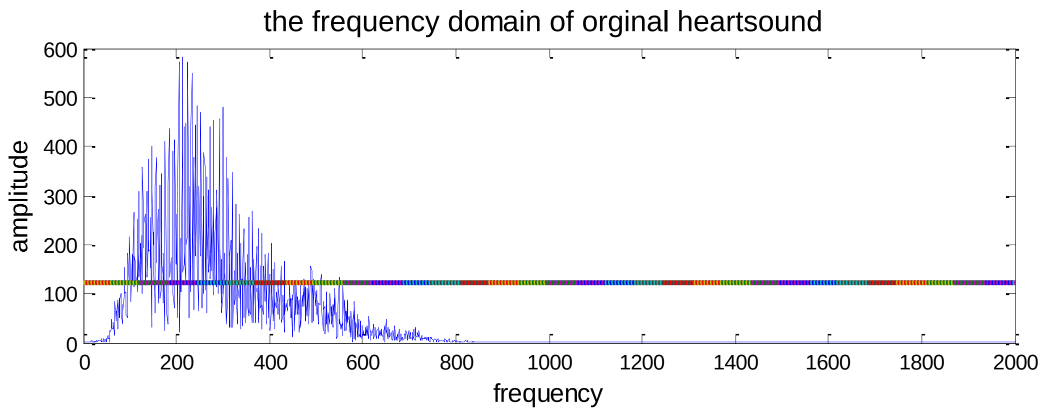

the frequency domain of orginal heartsound
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

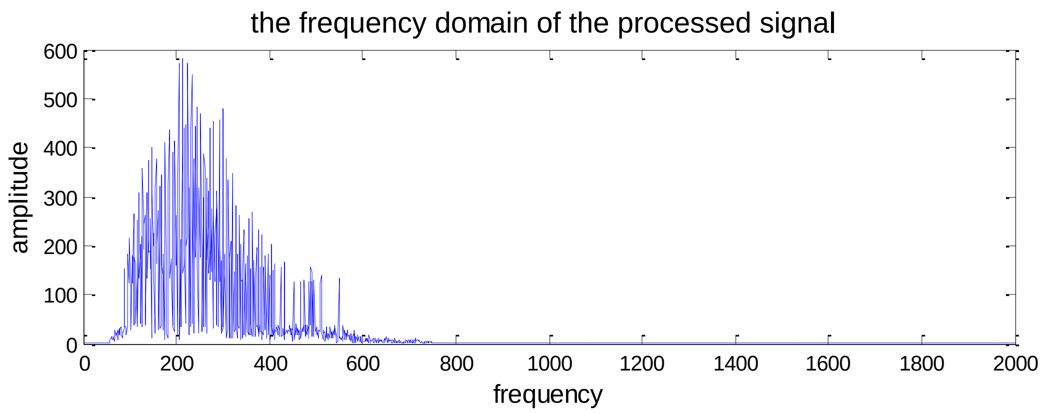

the frequency domain of the processed signal
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

Sample 5:


the orginal heart sound
amplitude
0.4
0.2
0
-0.2
-0.4
2
4
6
8
10
12
time
x 10^4

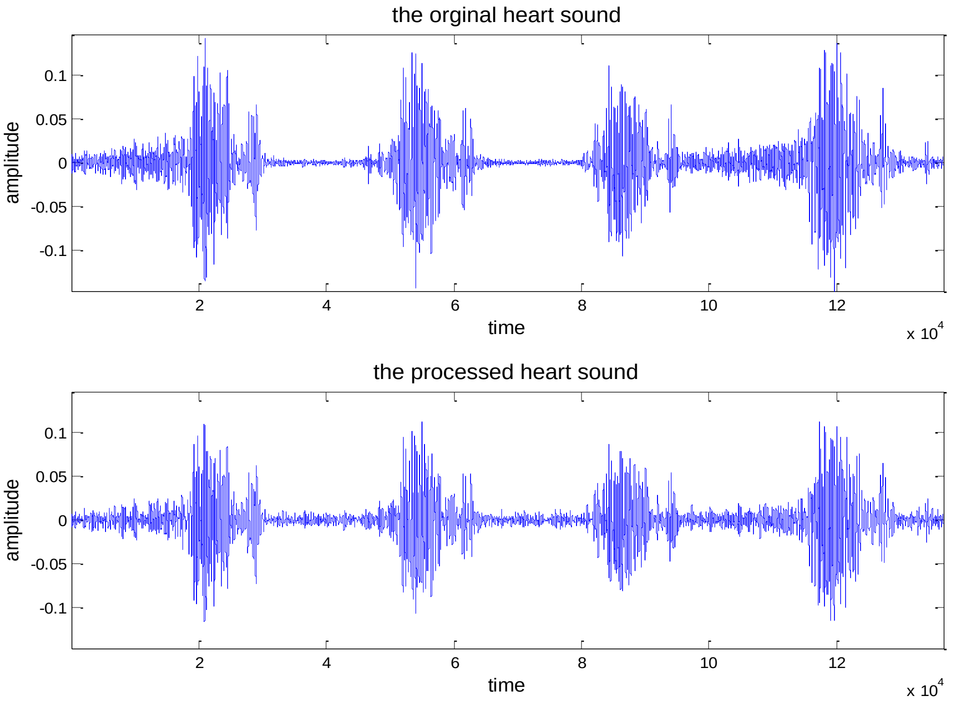

the processed heart sound
amplitude
0.4
0.2
0
-0.2
-0.4
2
4
6
8
10
12
time
x 10^4

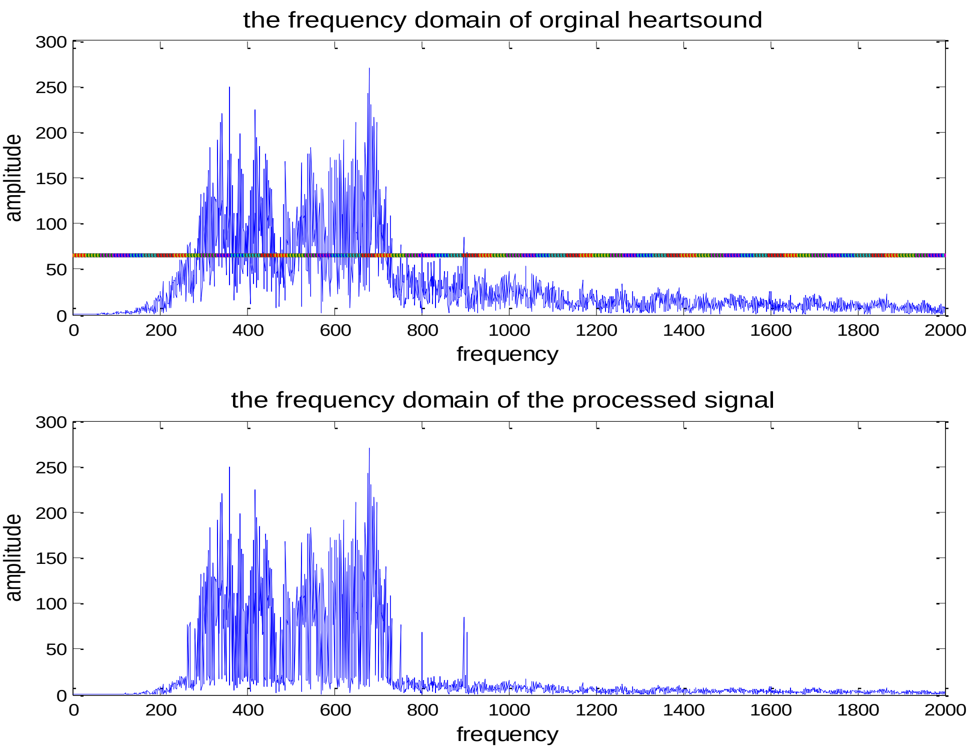

the frequency domain of orginal heartsound
amplitude
600
500
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency


the frequency domain of the processed signal
amplitude
600
500
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

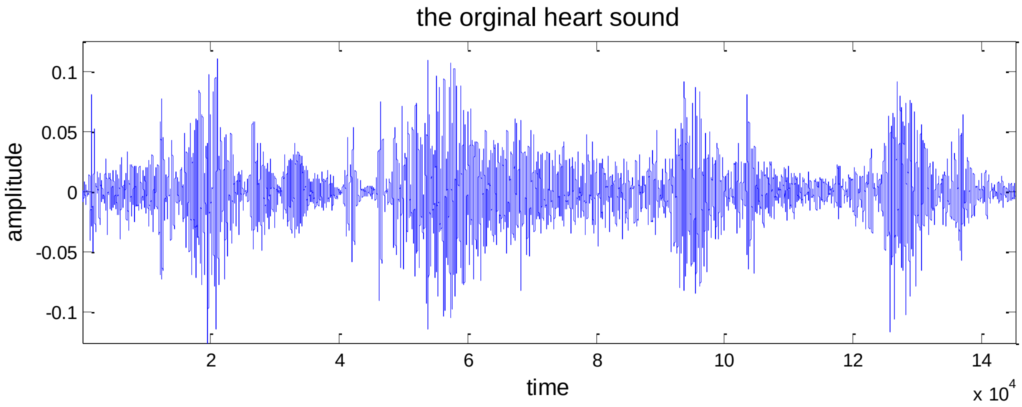
2. **The respiration sounds or breathing noise**

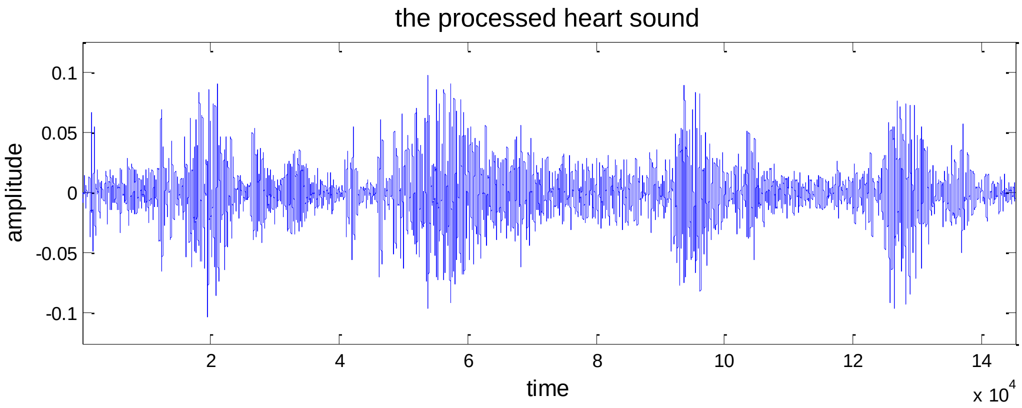
Sample 1:

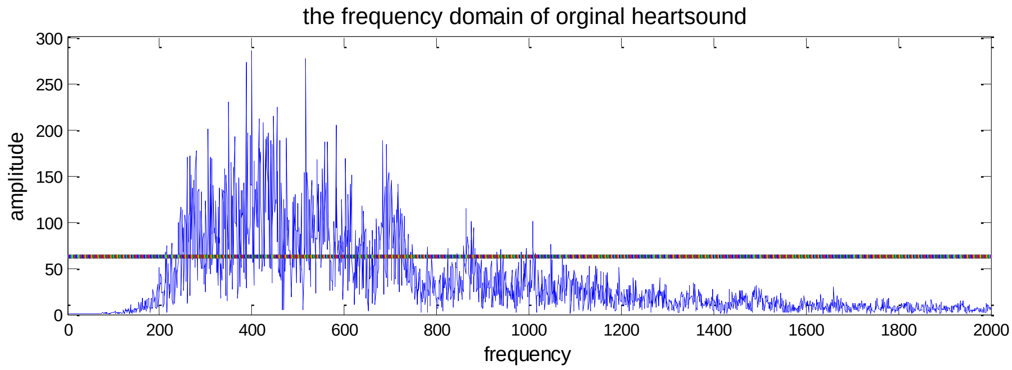


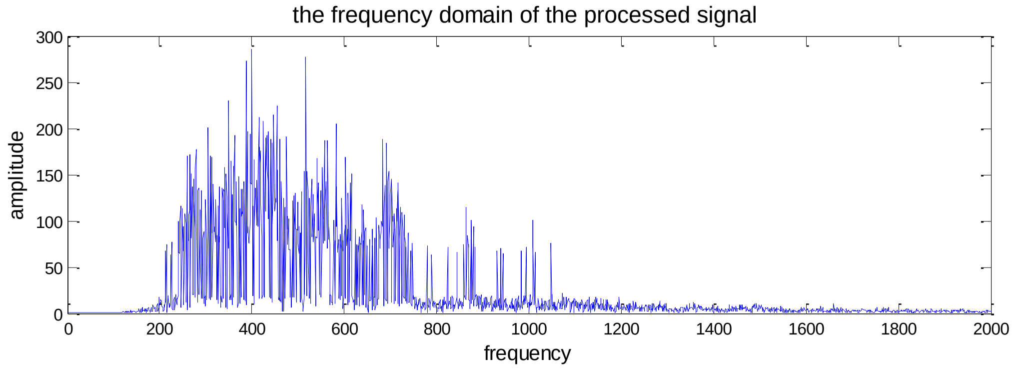


Sample 2:

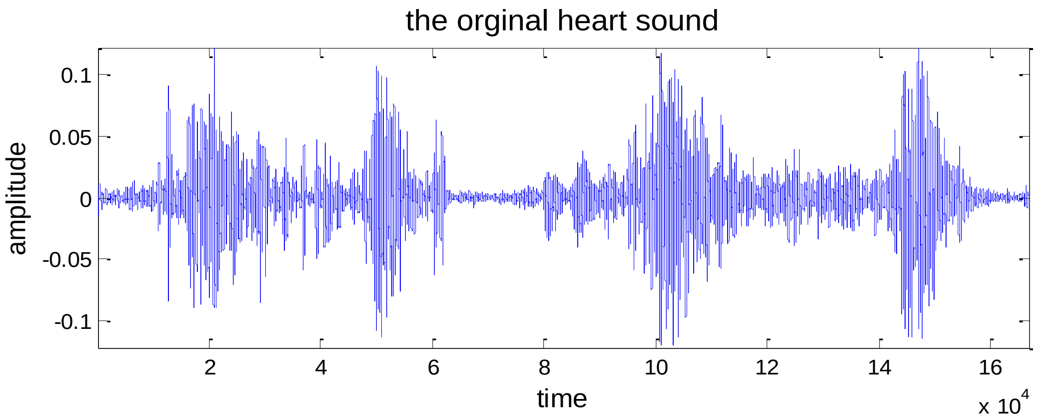

the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4

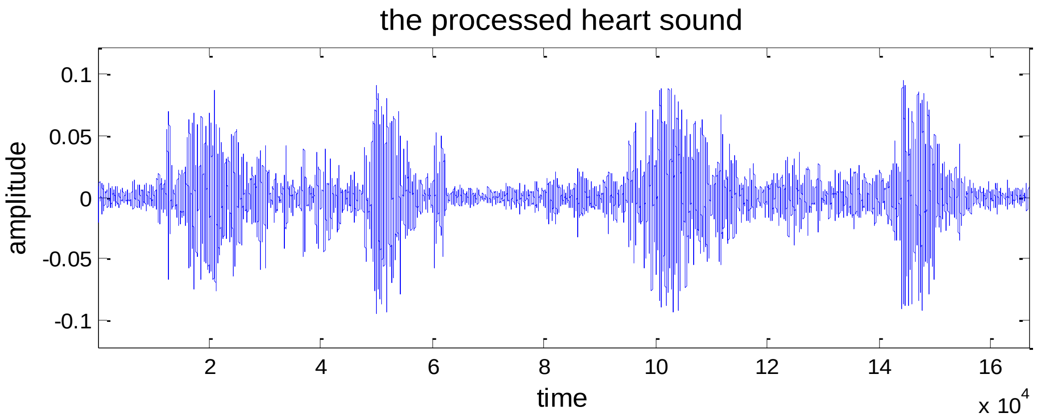

the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4

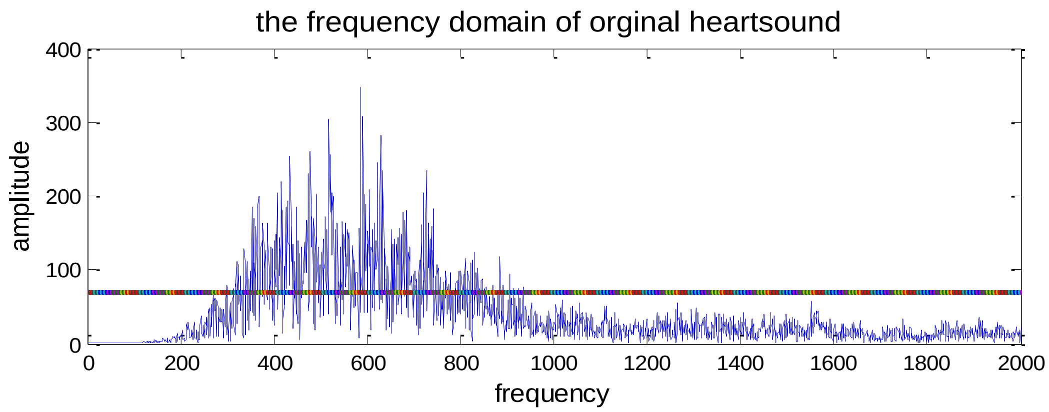

the frequency domain of orginal heartsound
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

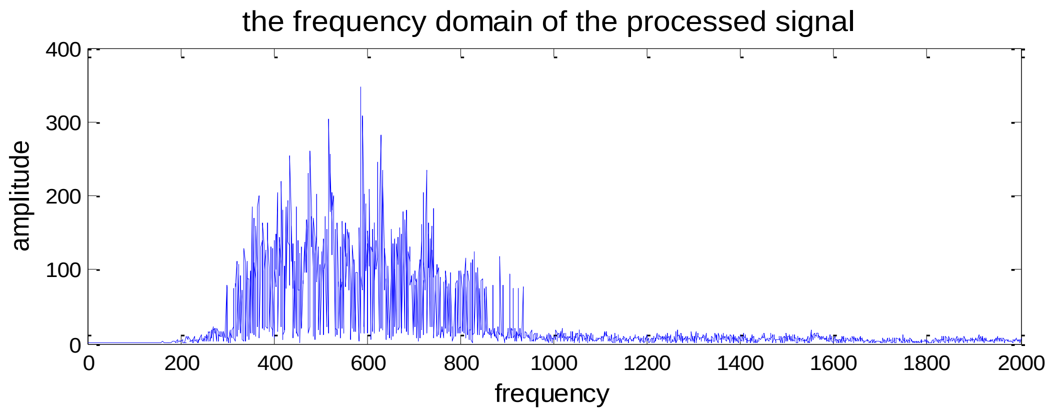

the frequency domain of the processed signal
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

Sample 3:

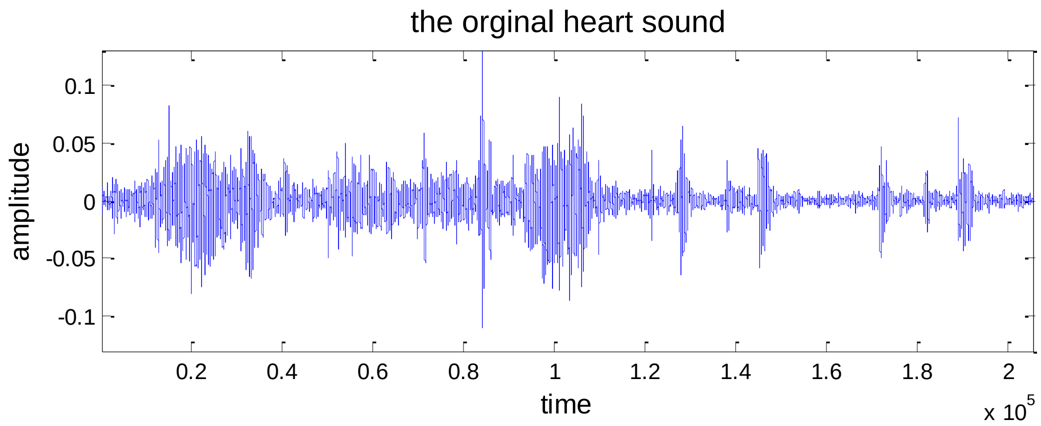

the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
16
time
x 10^4


the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
16
time
x 10^4

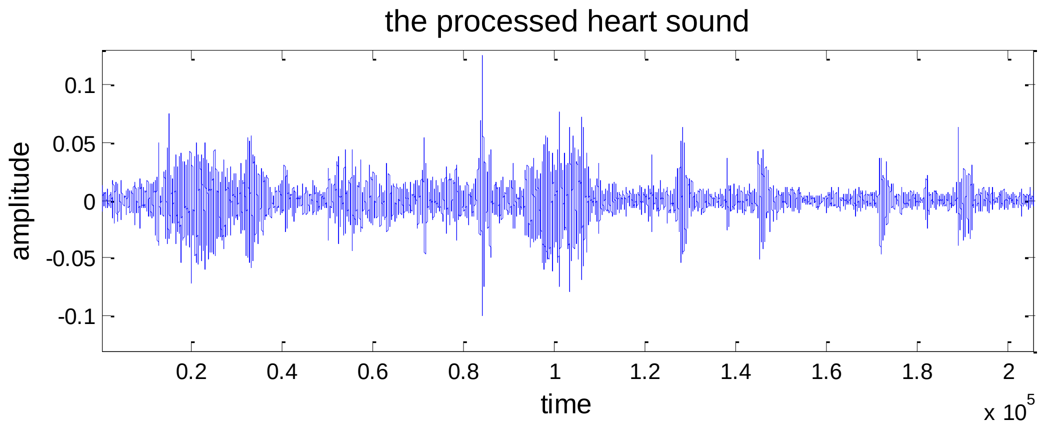

the frequency domain of orginal heartsound
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency


the frequency domain of the processed signal
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

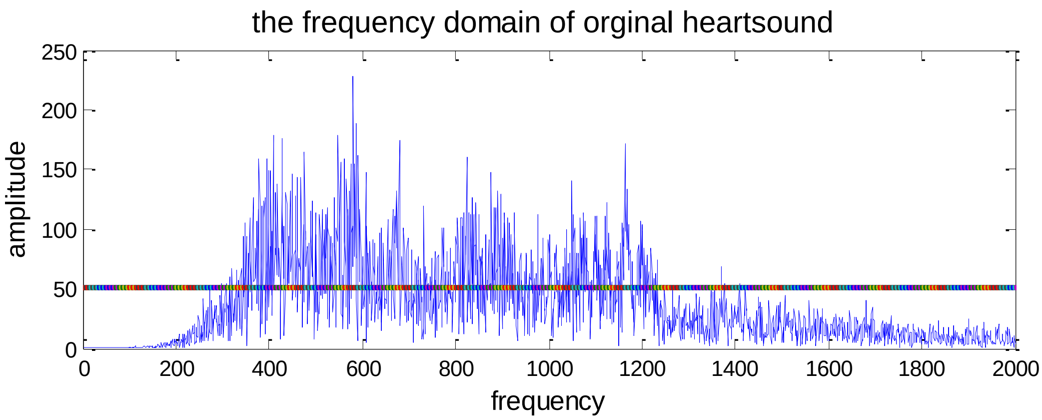
Sample 4:



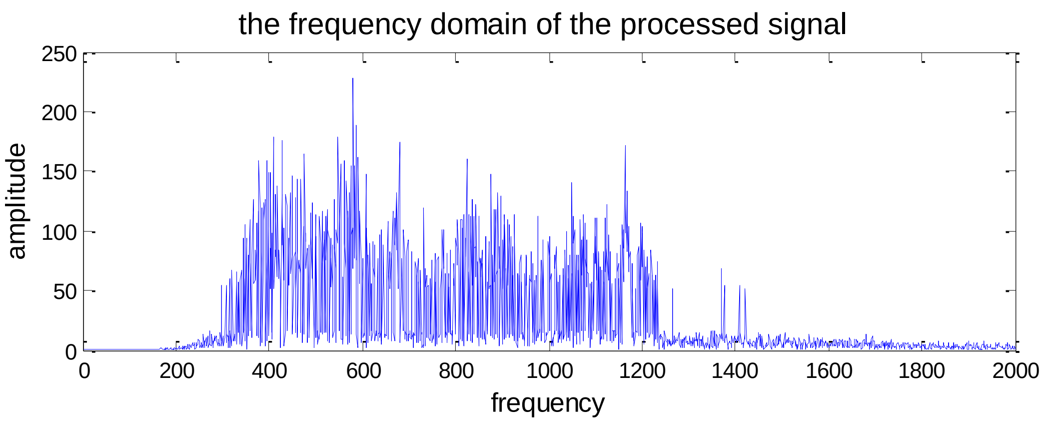
the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
time
x 10^5



the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
time
x 10^5

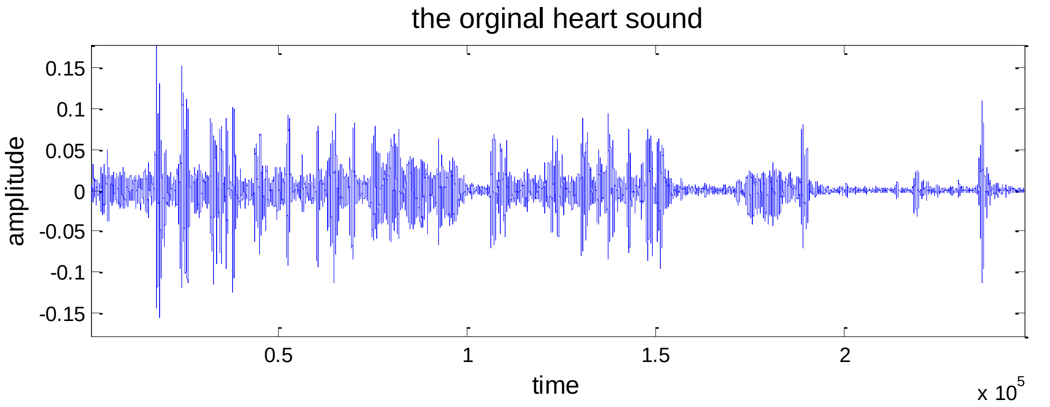


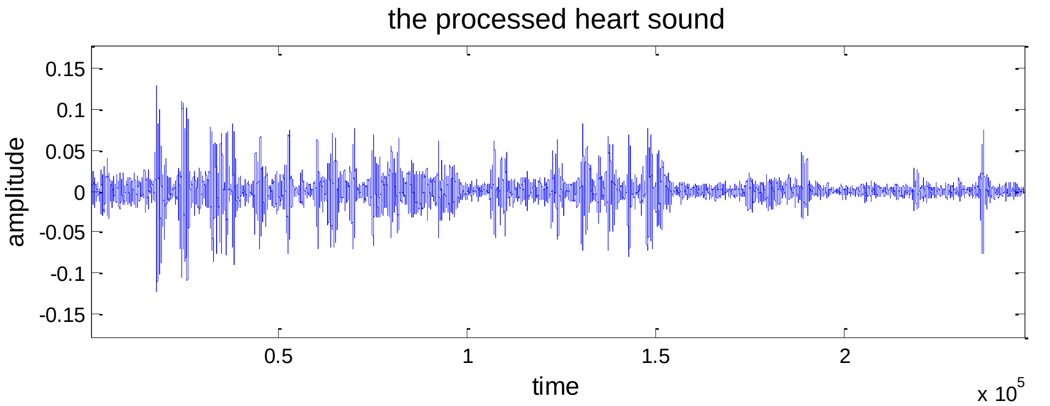
the frequency domain of orginal heartsound
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

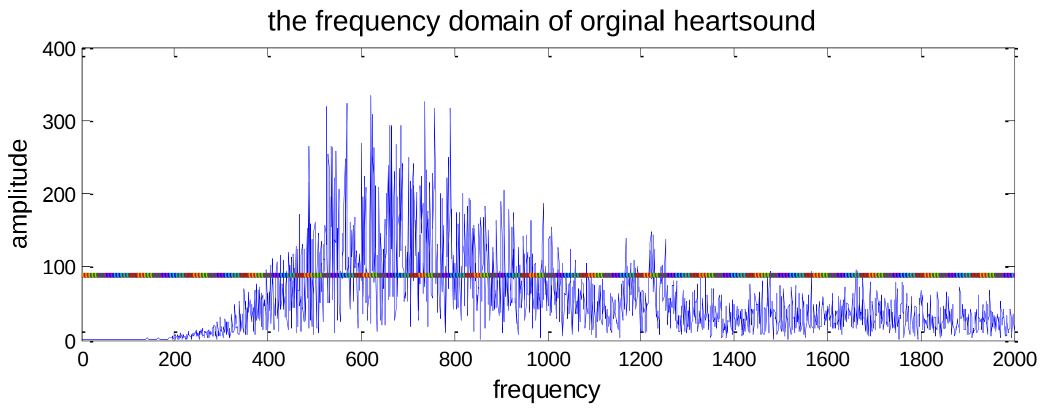


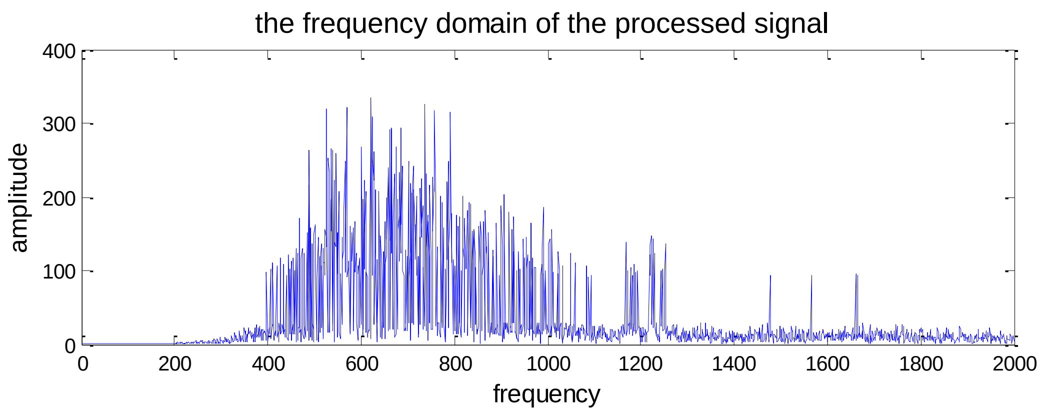
the frequency domain of the processed signal
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

Sample 5:


the orginal heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
0.5
1
1.5
2
time
x 10^5

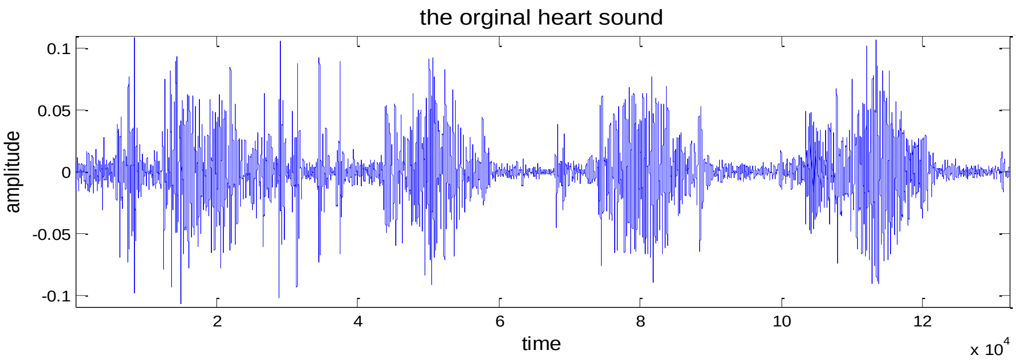

the processed heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
0.5
1
1.5
2
time
x 10^5

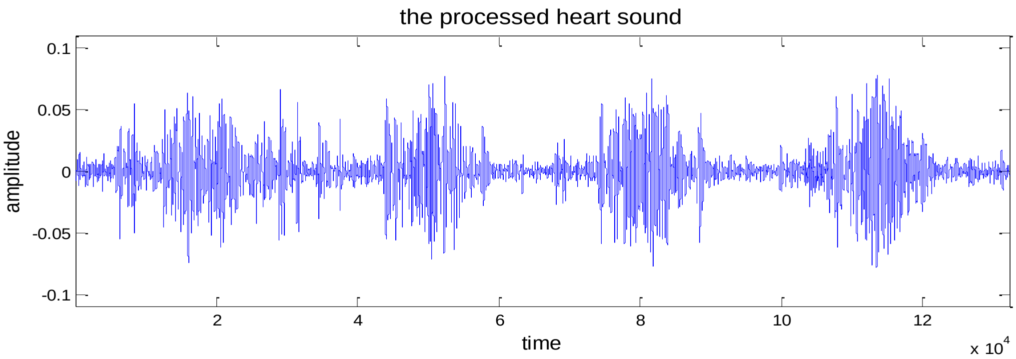

the frequency domain of orginal heartsound
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

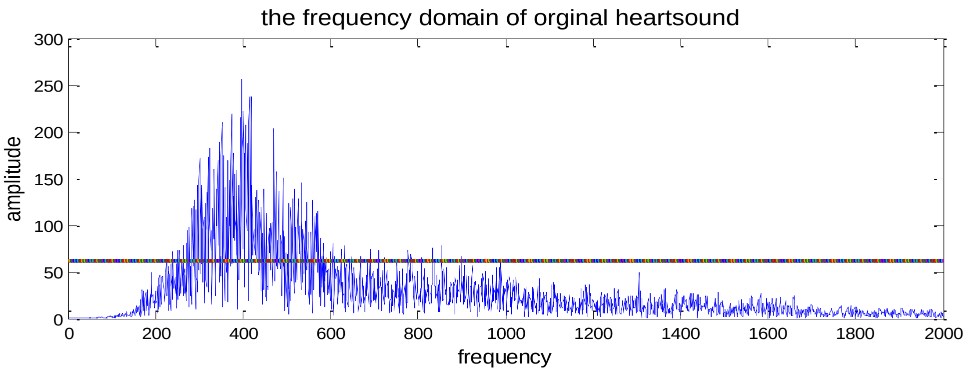

the frequency domain of the processed signal
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

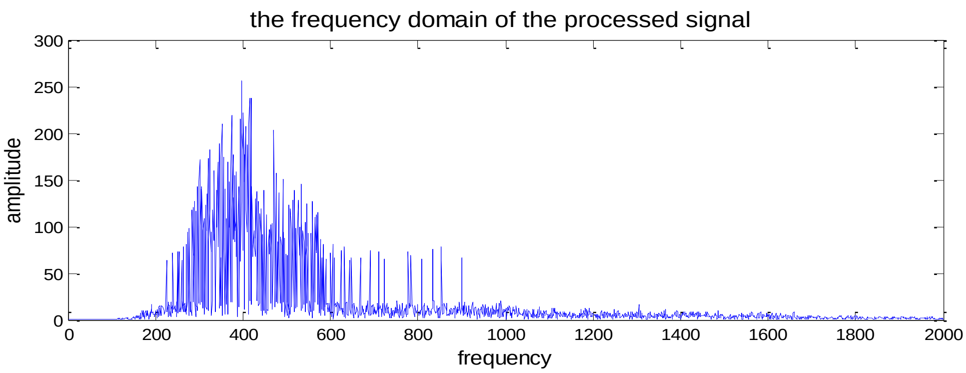
3. **The ambient noise**

Sample 1:

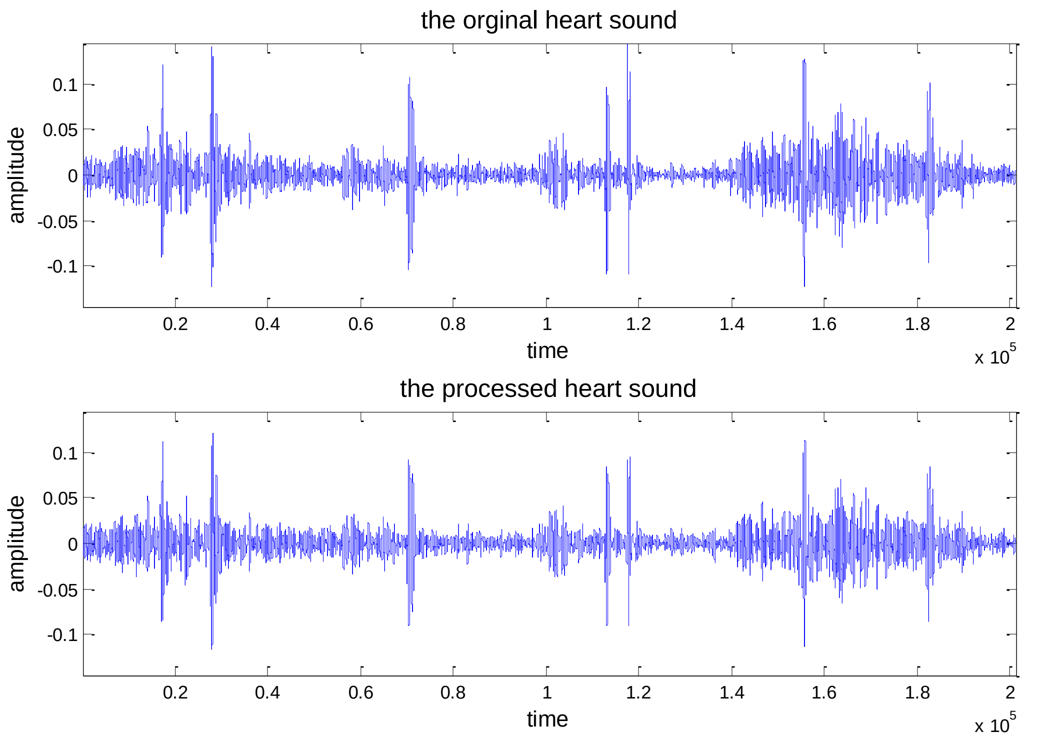


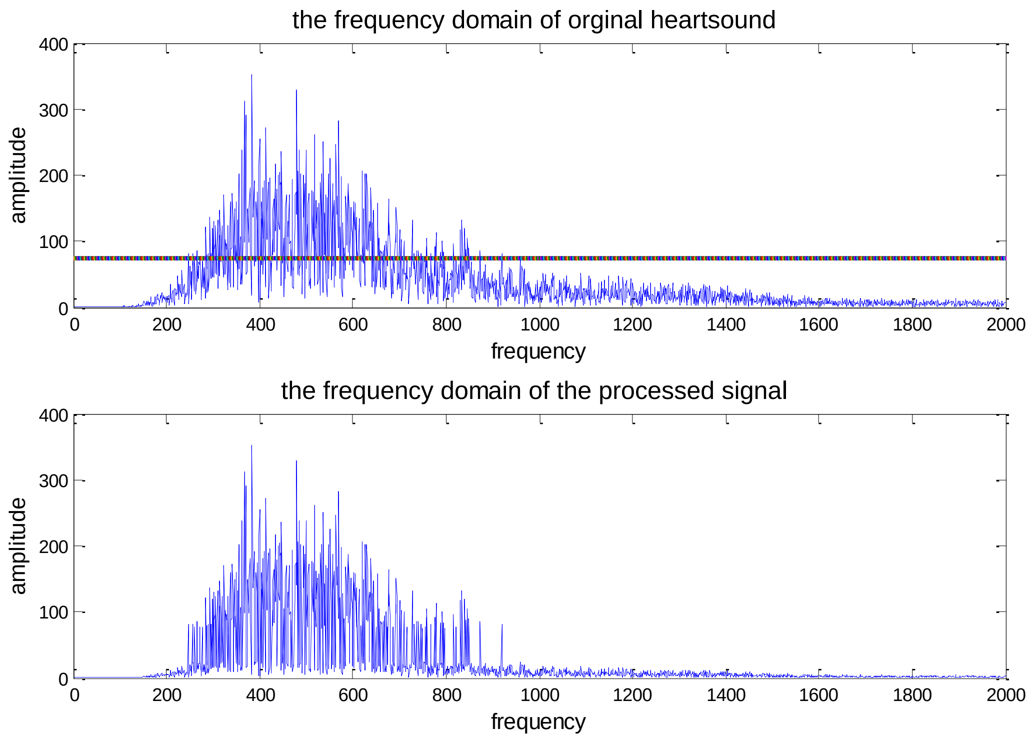




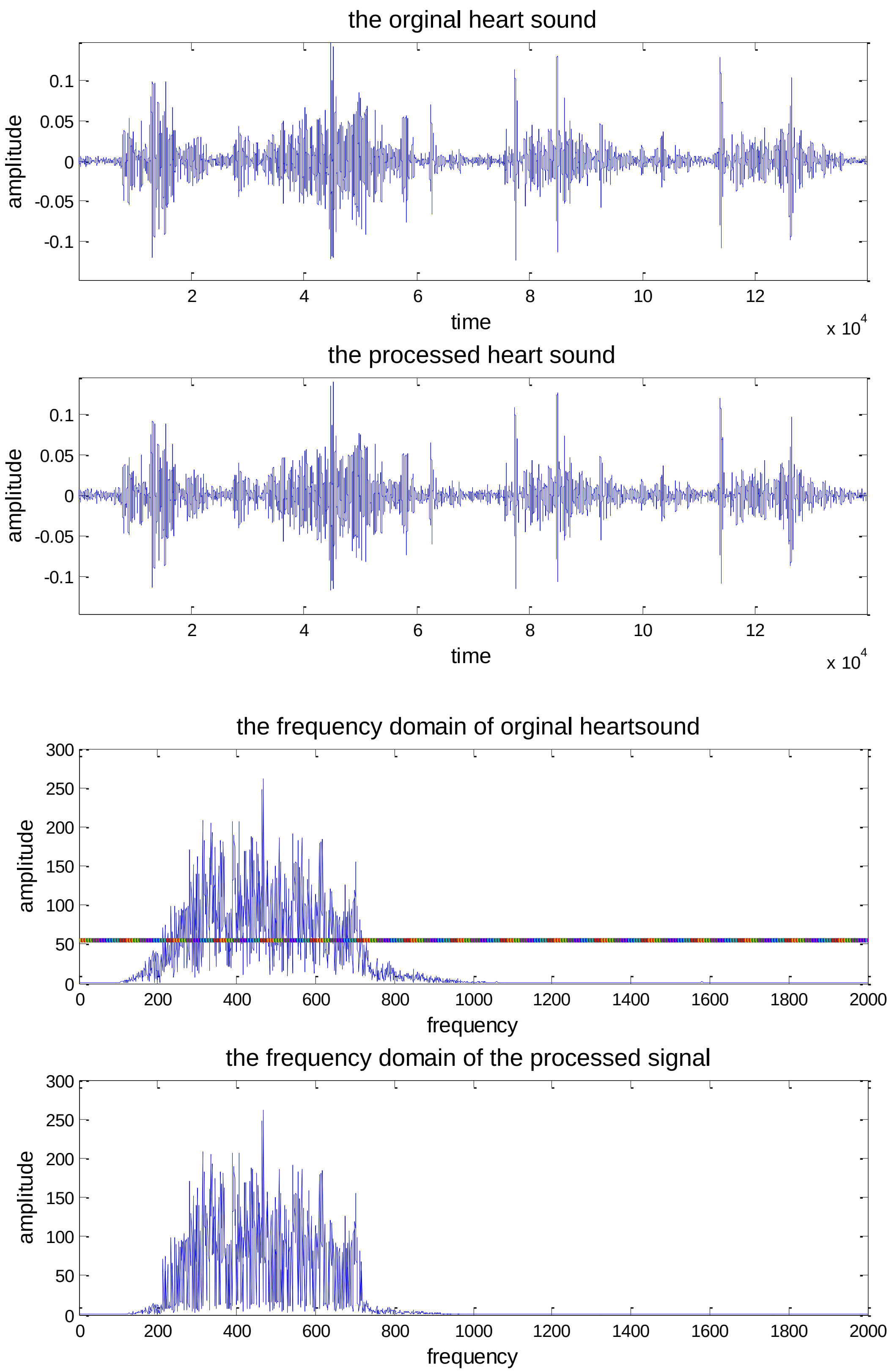


Sample 2:

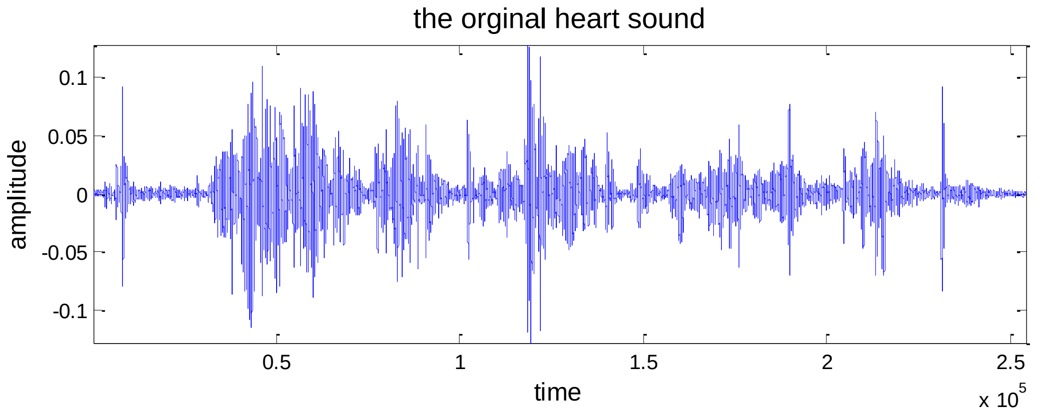

the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
time
x 10^5
the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
time
x 10^5


the frequency domain of orginal heartsound
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency
the frequency domain of the processed signal
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

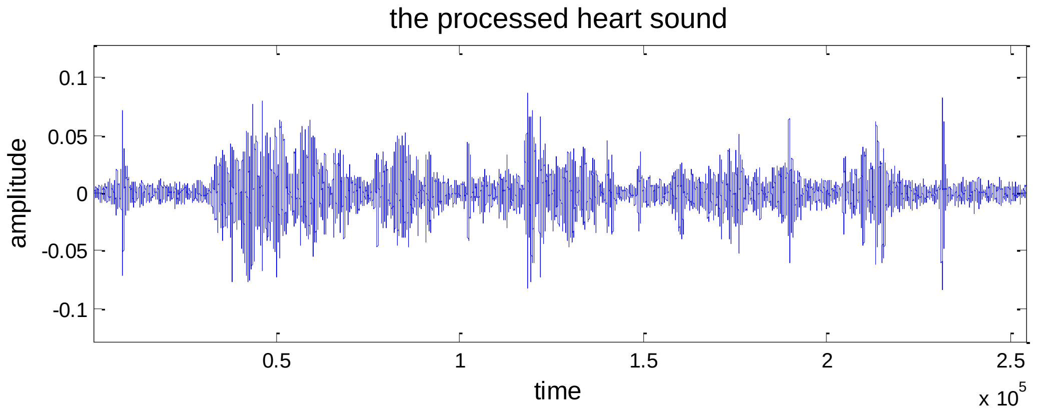
Sample 3:


the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4
the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

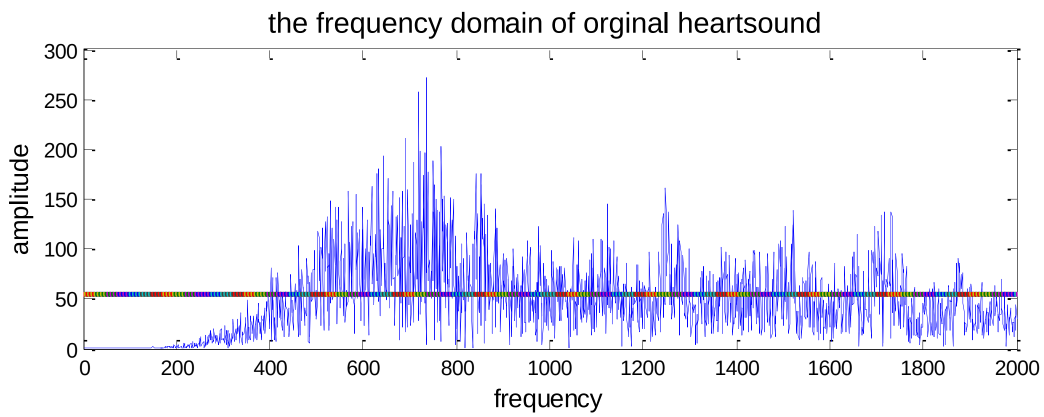
Sample 4:



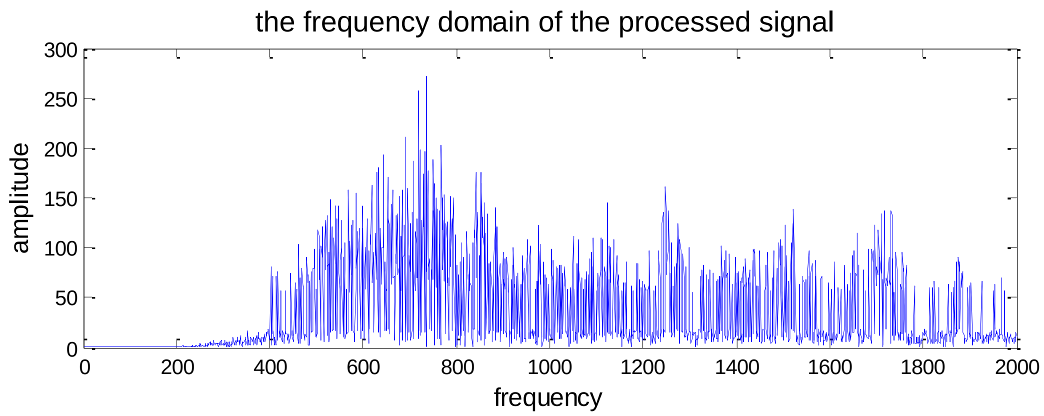
the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.5
1
1.5
2
2.5
time
x 10^5



the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
0.5
1
1.5
2
2.5
time
x 10^5

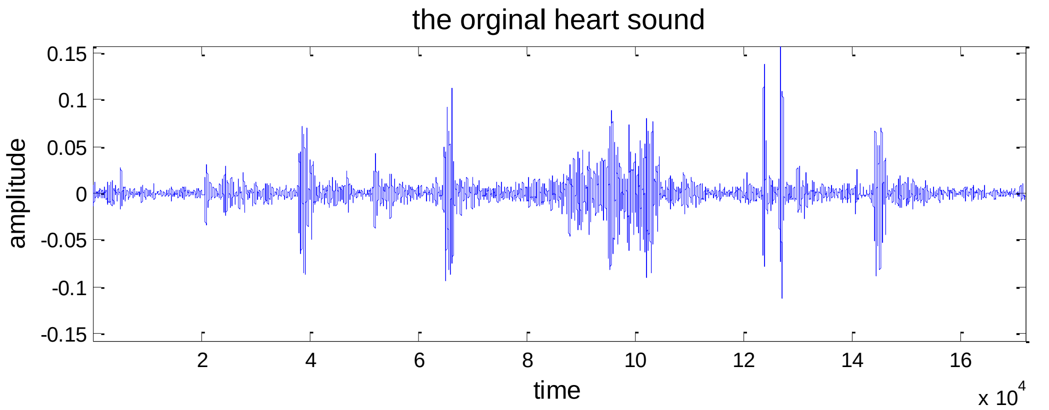


the frequency domain of orginal heartsound
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

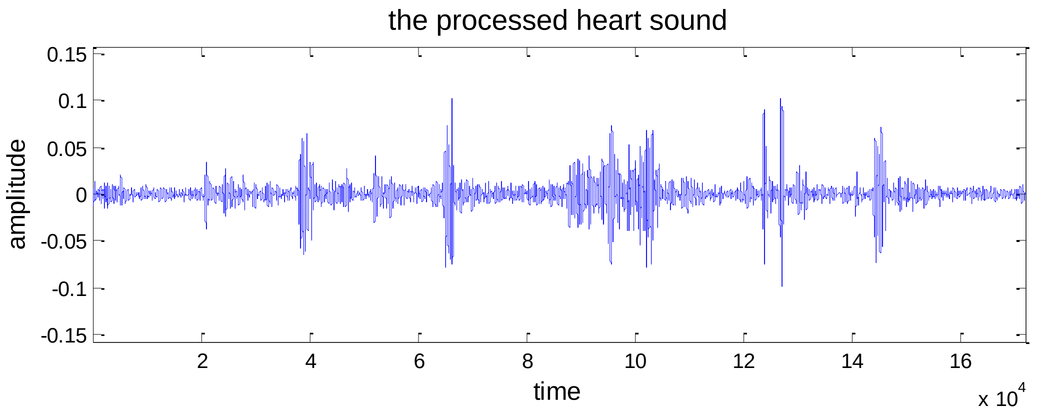


the frequency domain of the processed signal
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

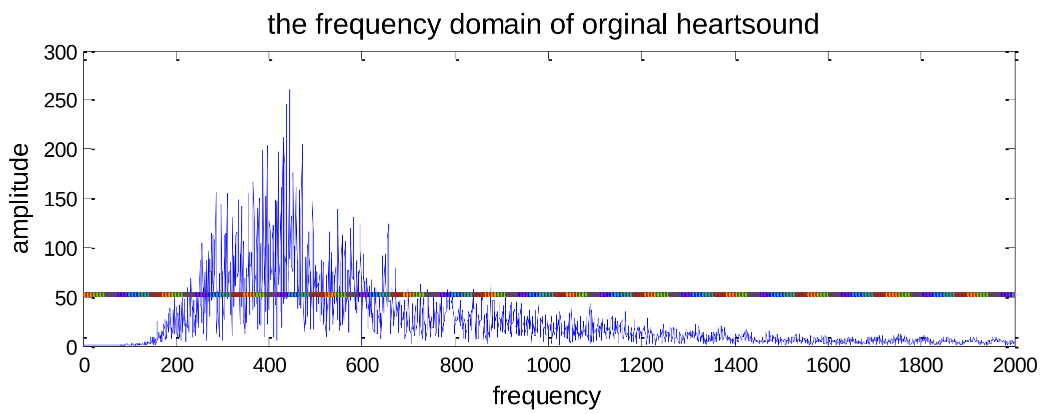
Sample 5:



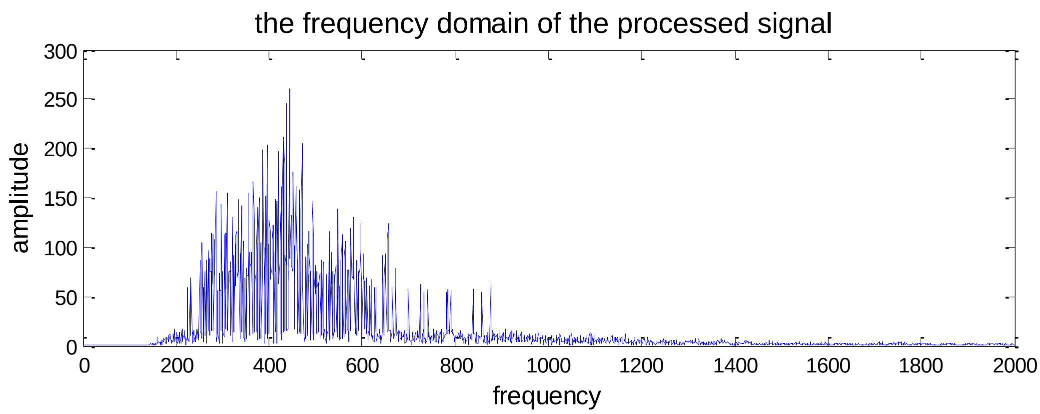
the orginal heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
16
time
x 10^4



the processed heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
16
time
x 10^4



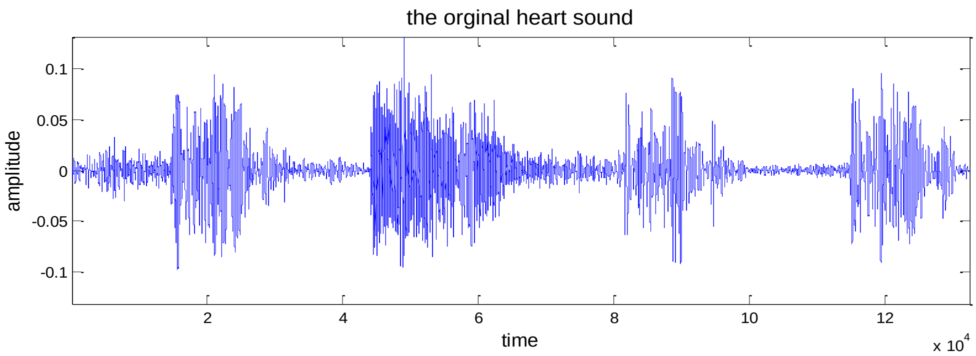
the frequency domain of orginal heartsound
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

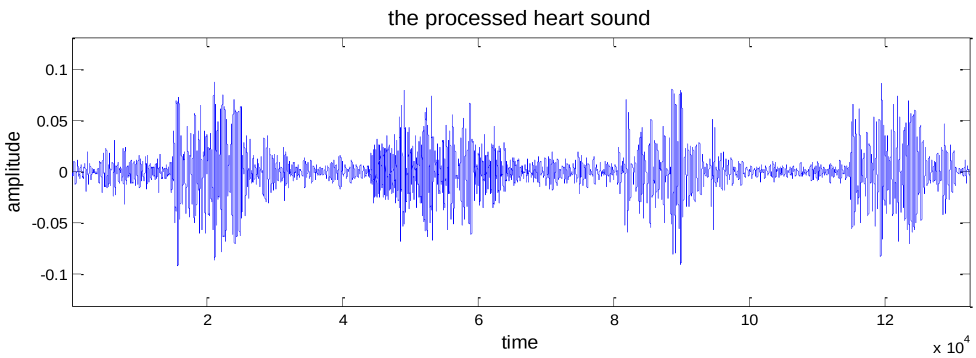


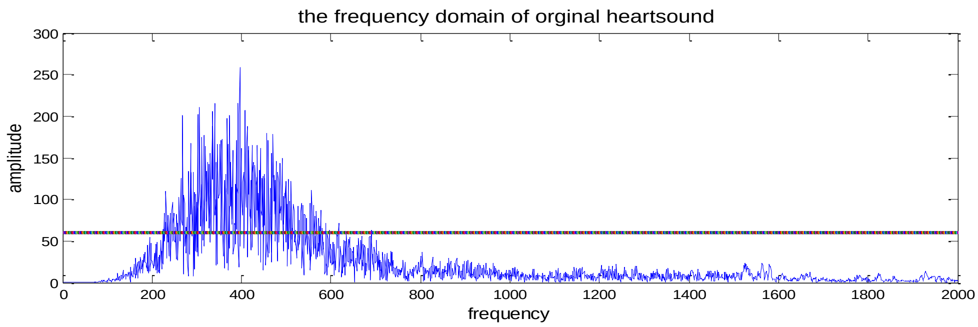
the frequency domain of the processed signal
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

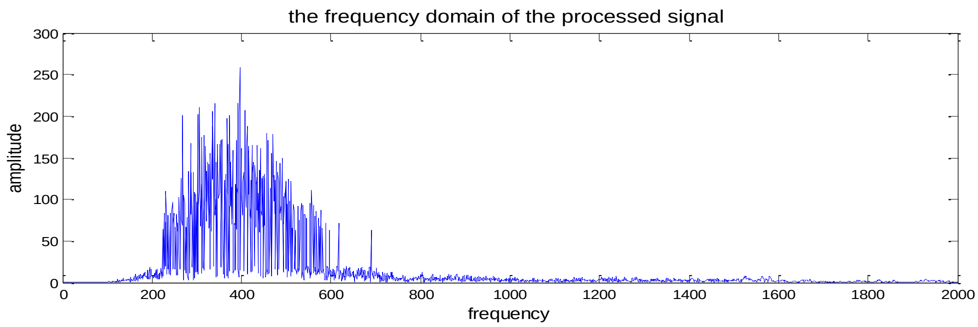
4. **The noise burst**

Sample 1:

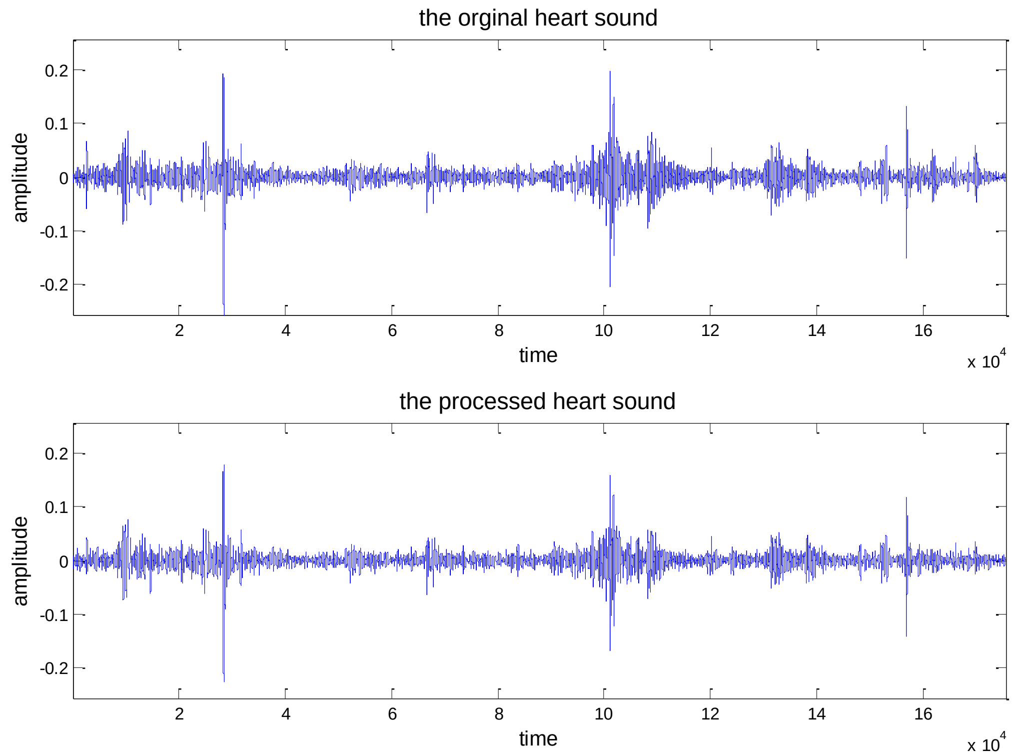


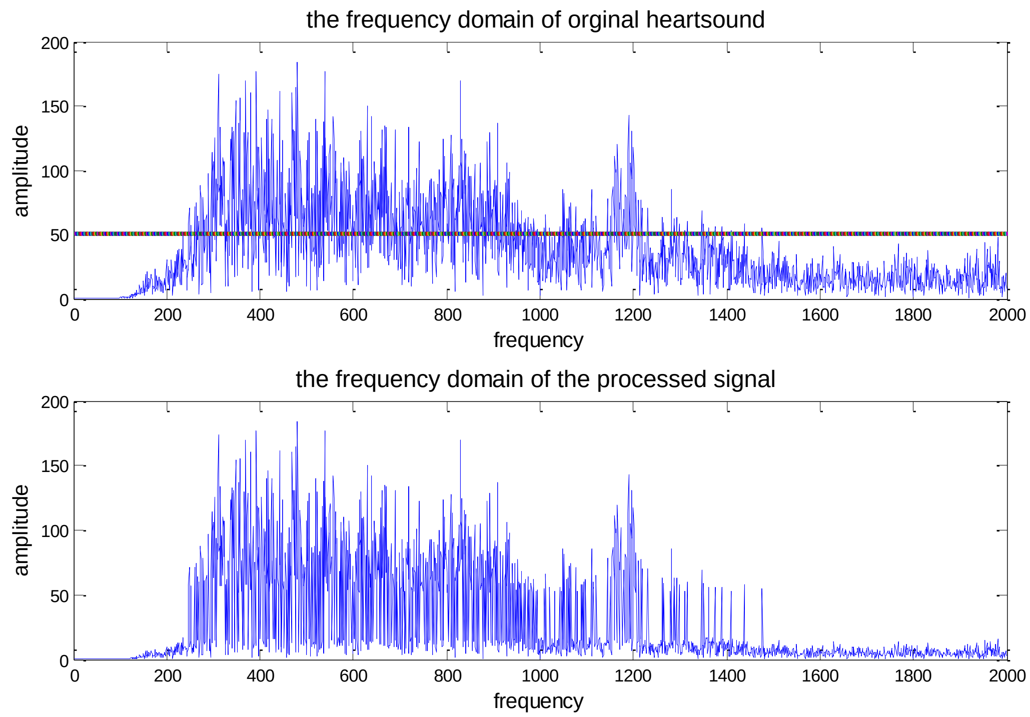




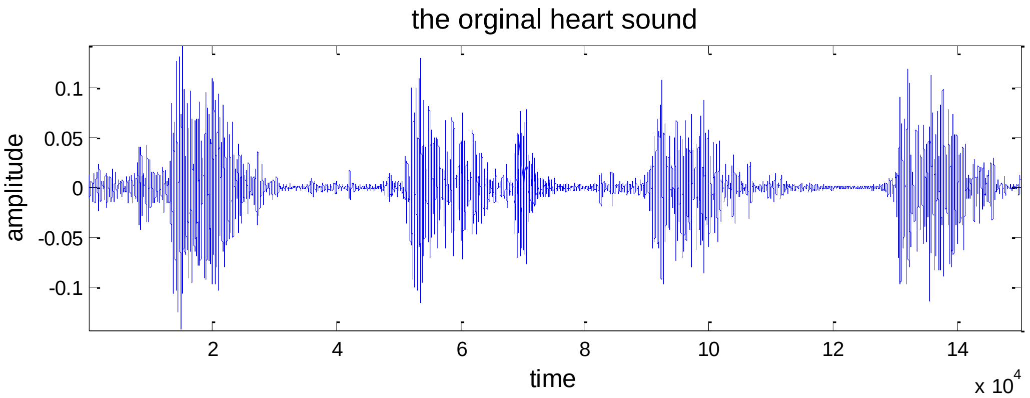


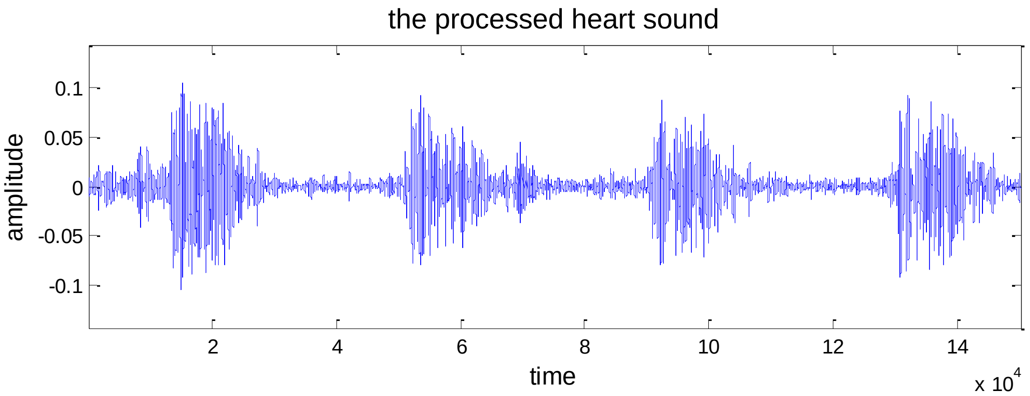
Sample 2:

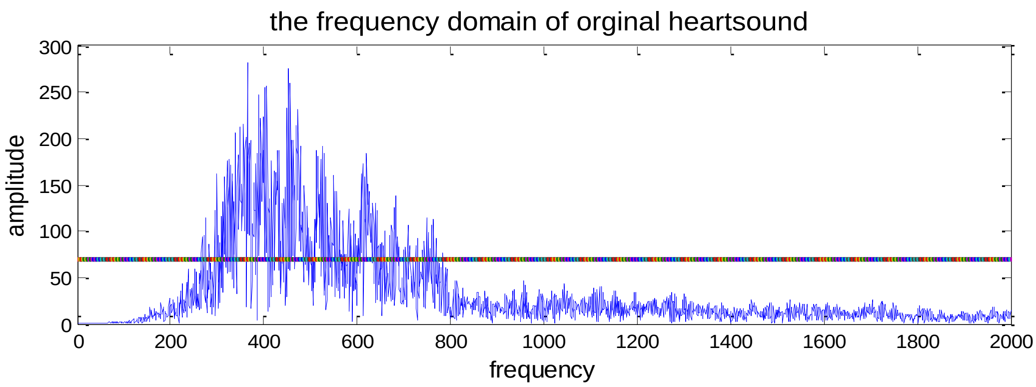

the orginal heart sound
amplitude
0.2
0.1
0
-0.1
-0.2
2
4
6
8
10
12
14
16
time
x 10^4
the processed heart sound
amplitude
0.2
0.1
0
-0.1
-0.2
2
4
6
8
10
12
14
16
time
x 10^4

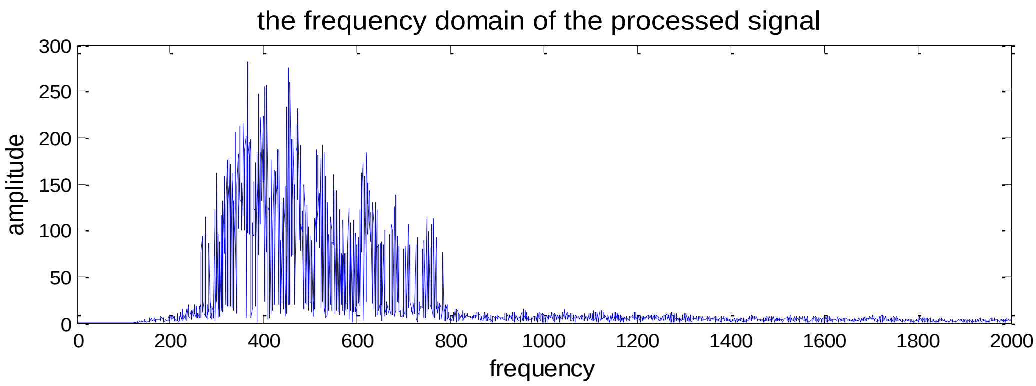

the frequency domain of orginal heartsound
amplitude
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency
the frequency domain of the processed signal
amplitude
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

Sample 3:

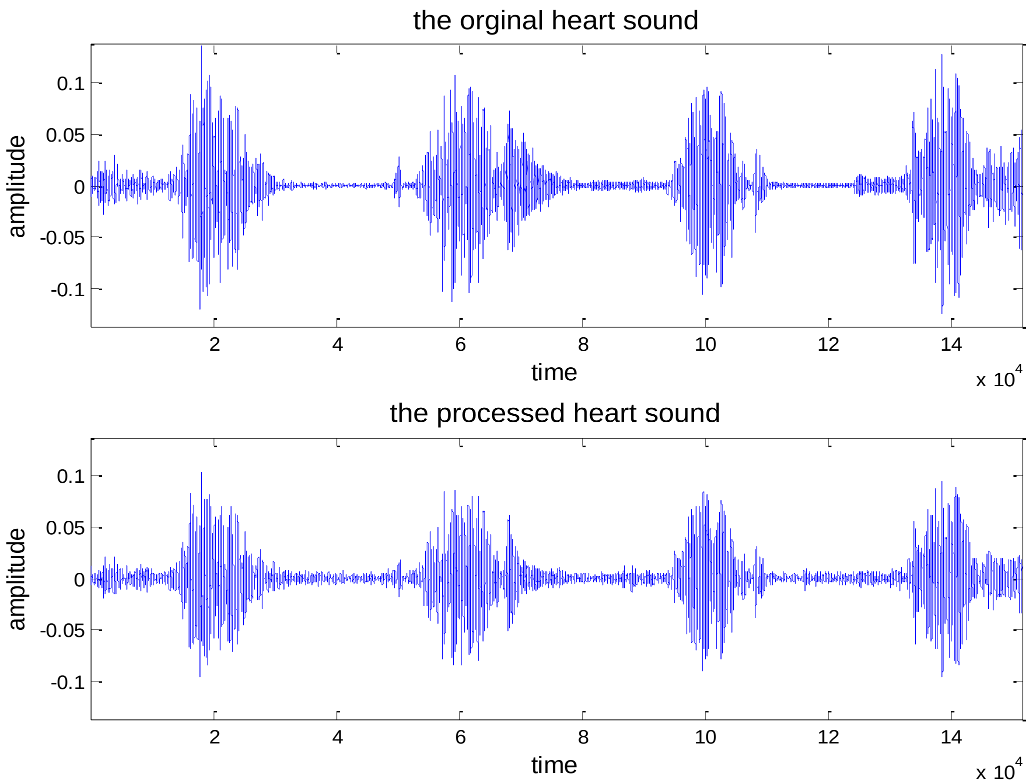

the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4

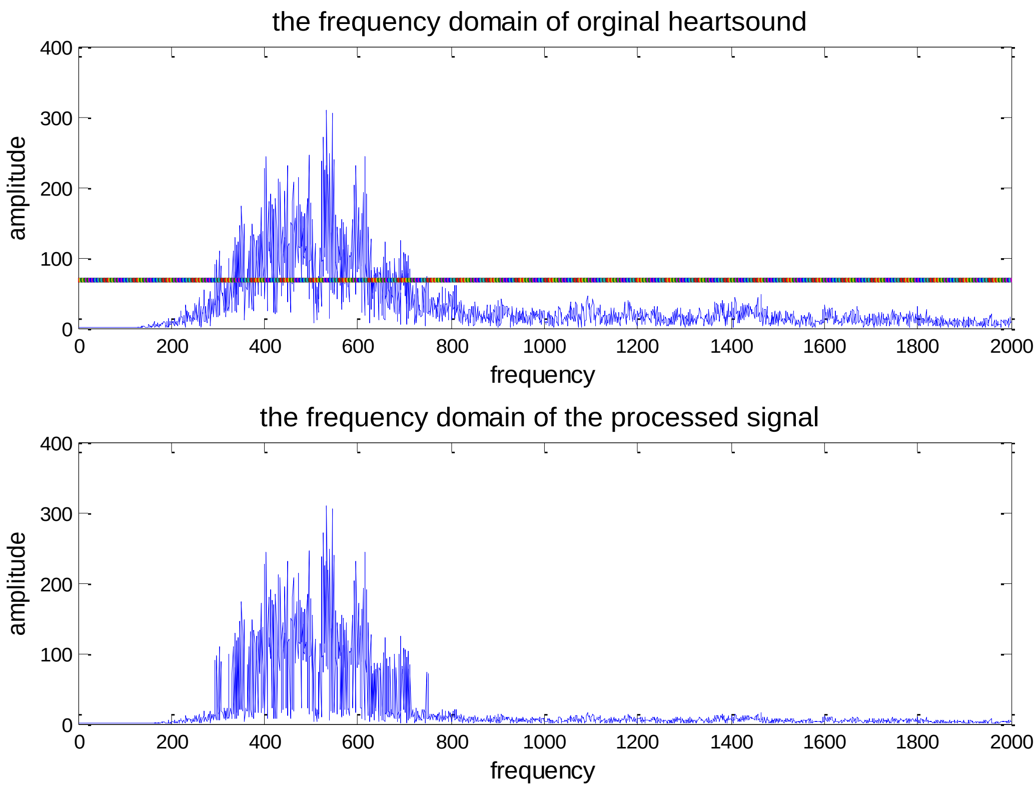

the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4


the frequency domain of orginal heartsound
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

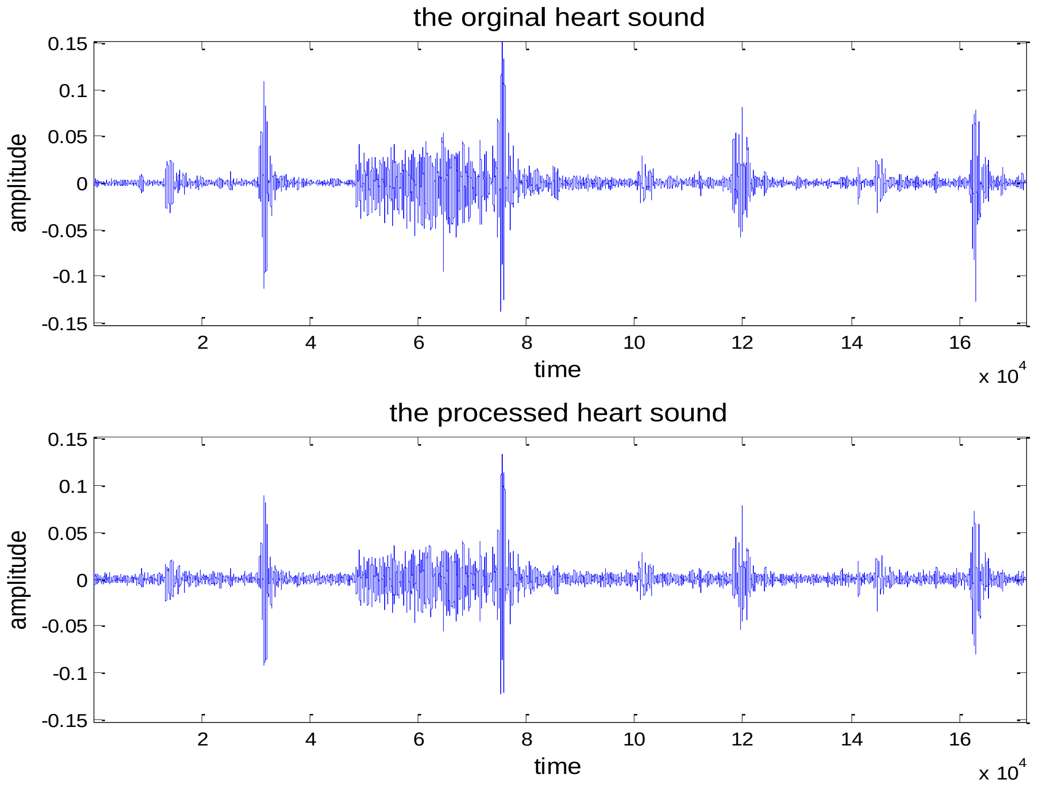

the frequency domain of the processed signal
amplitude
300
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

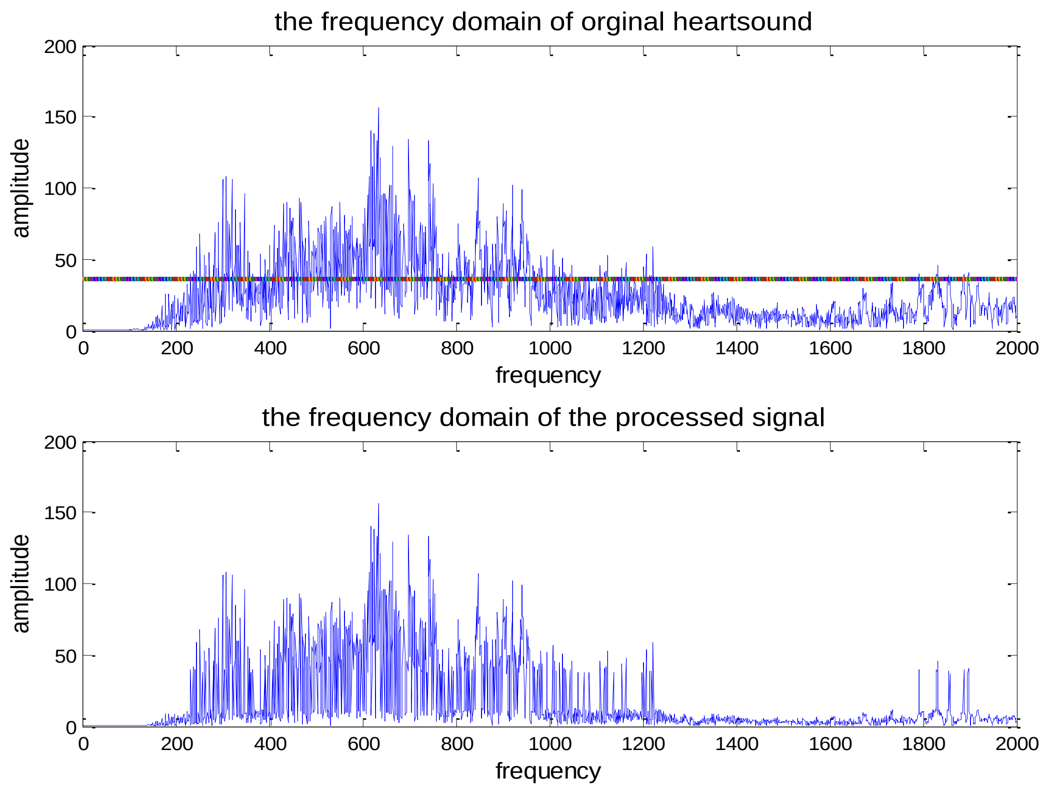
Sample 4:


the orginal heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4
the processed heart sound
amplitude
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
time
x 10^4


the frequency domain of orginal heartsound
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency
the frequency domain of the processed signal
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

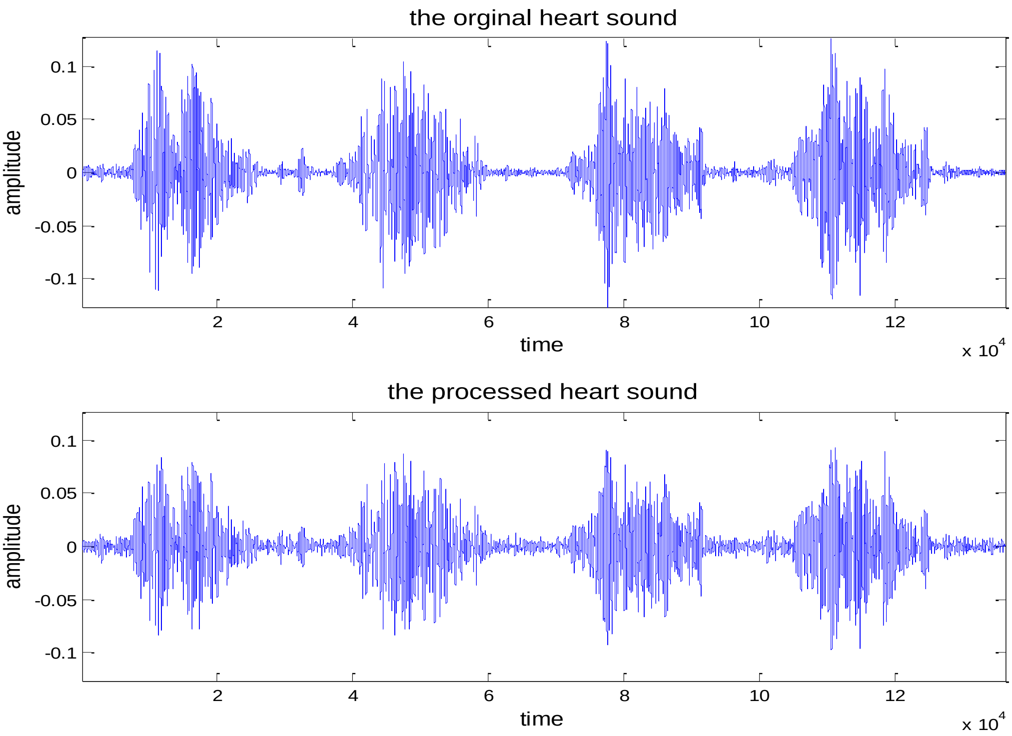
Sample 5:

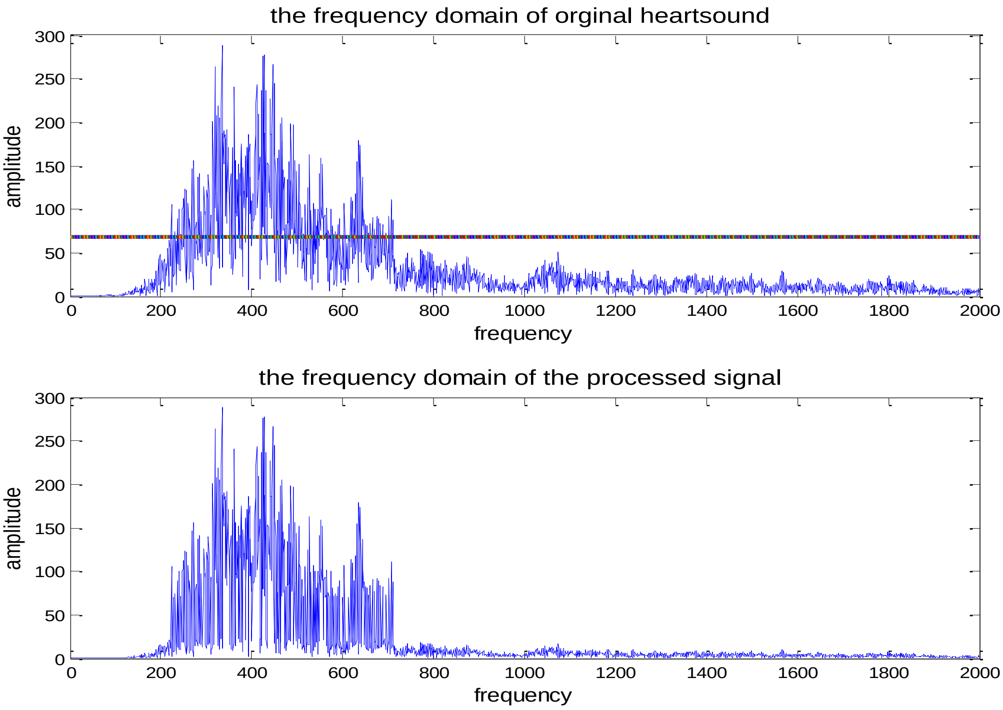

the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4


the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

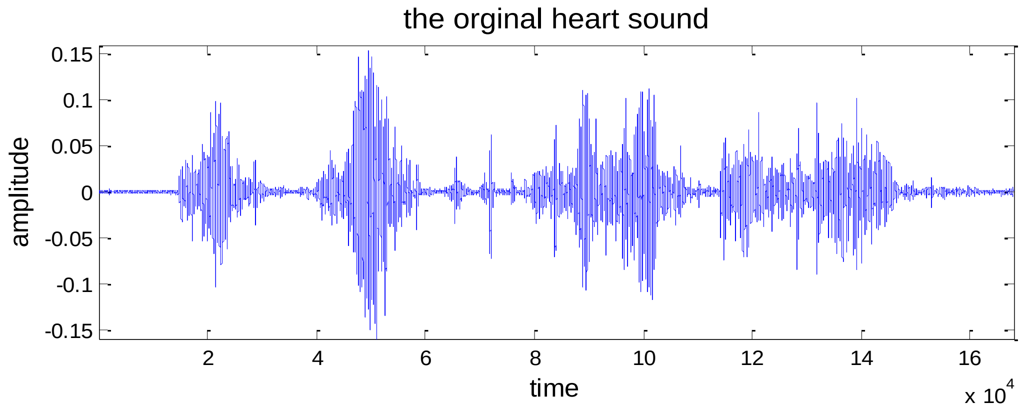
5. The friction-induced noise (small movement of human or stethoscope)

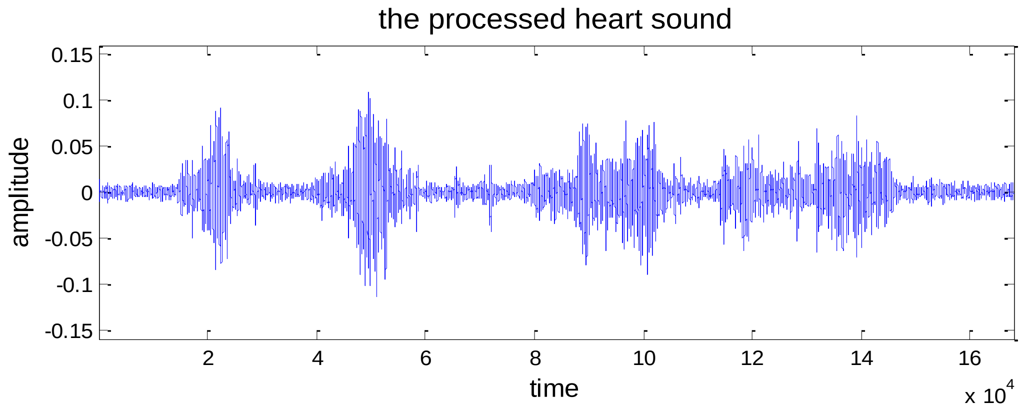
Sample 1:

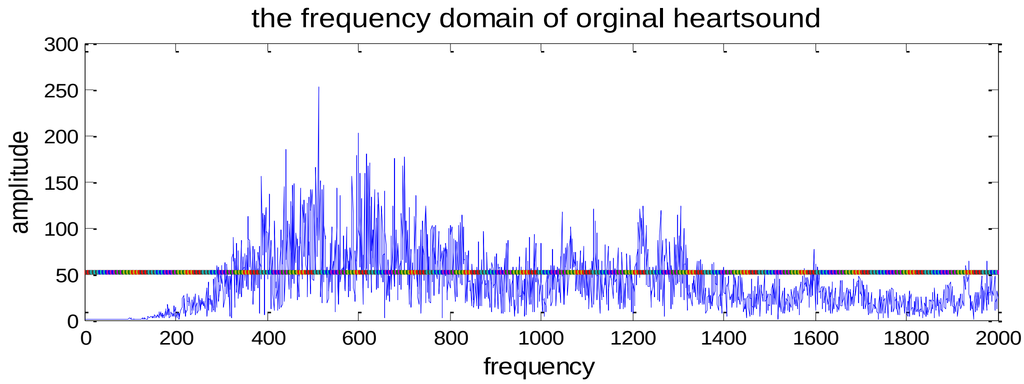


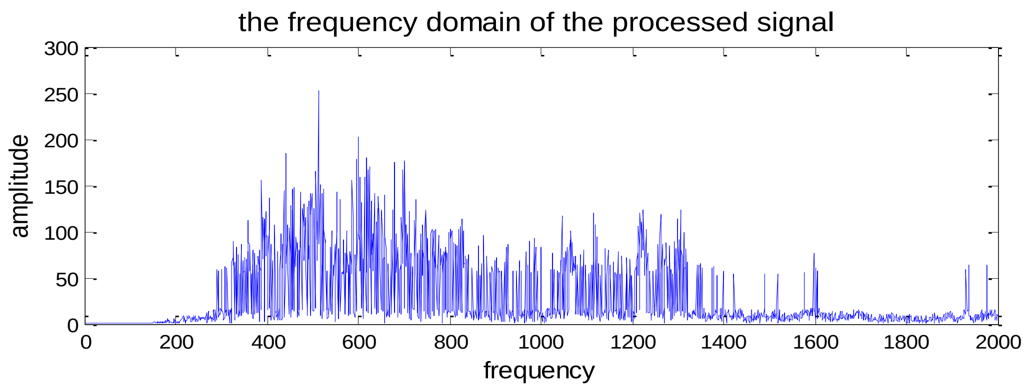


Sample 2:

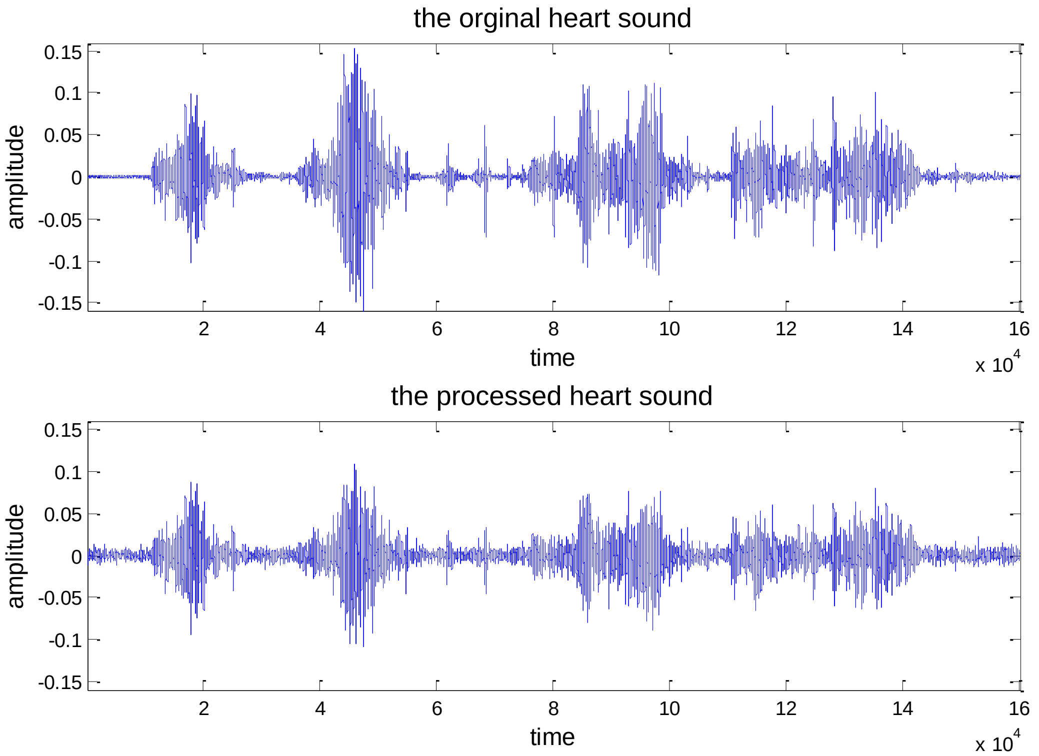

the orginal heart sound
amplitude
time
x 10^4

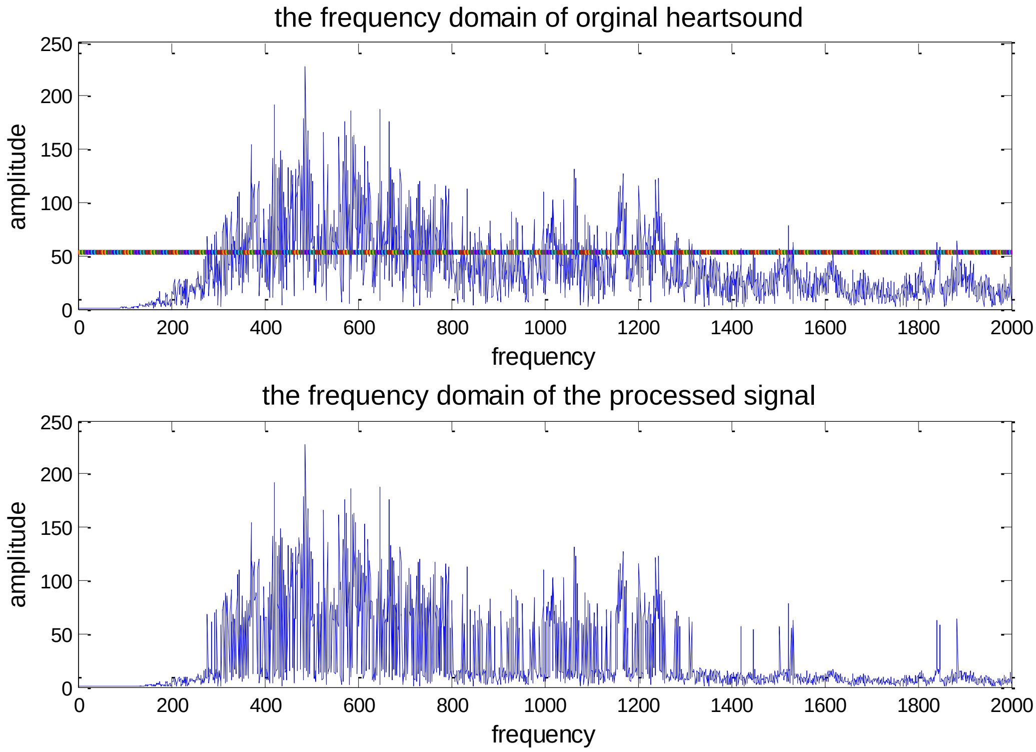

the processed heart sound
amplitude
time
x 10^4


the frequency domain of orginal heartsound
amplitude
frequency

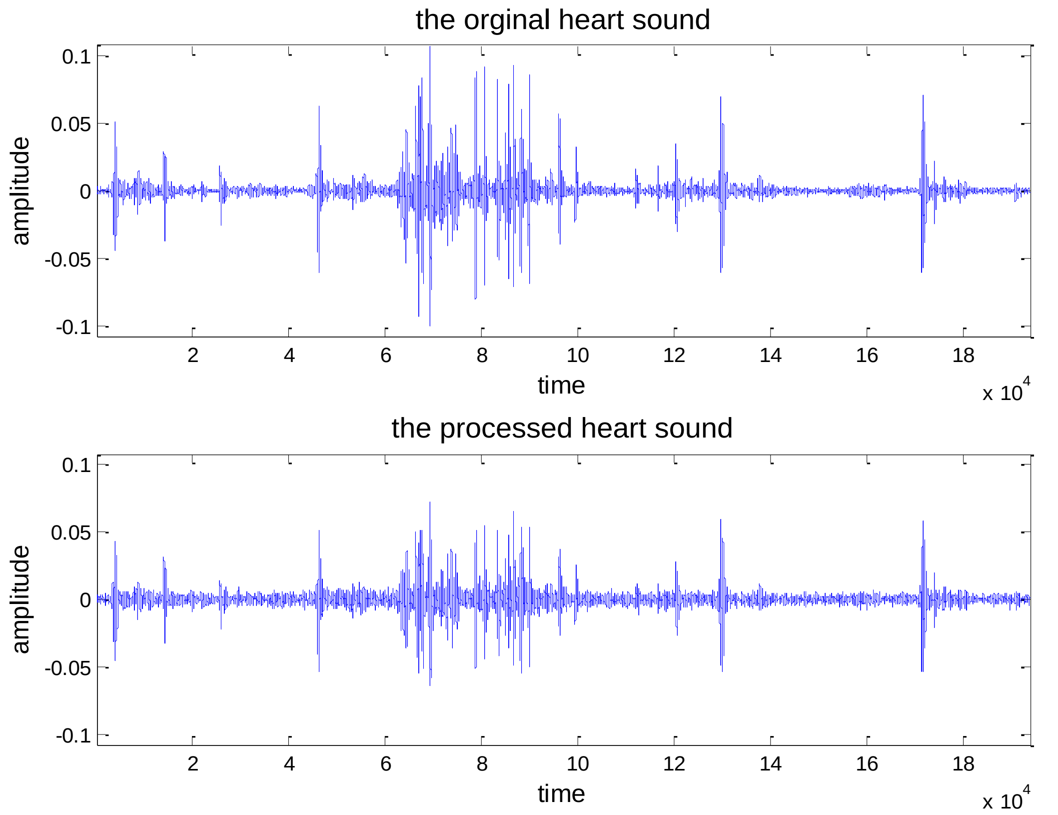

the frequency domain of the processed signal
amplitude
frequency

Sample 3:

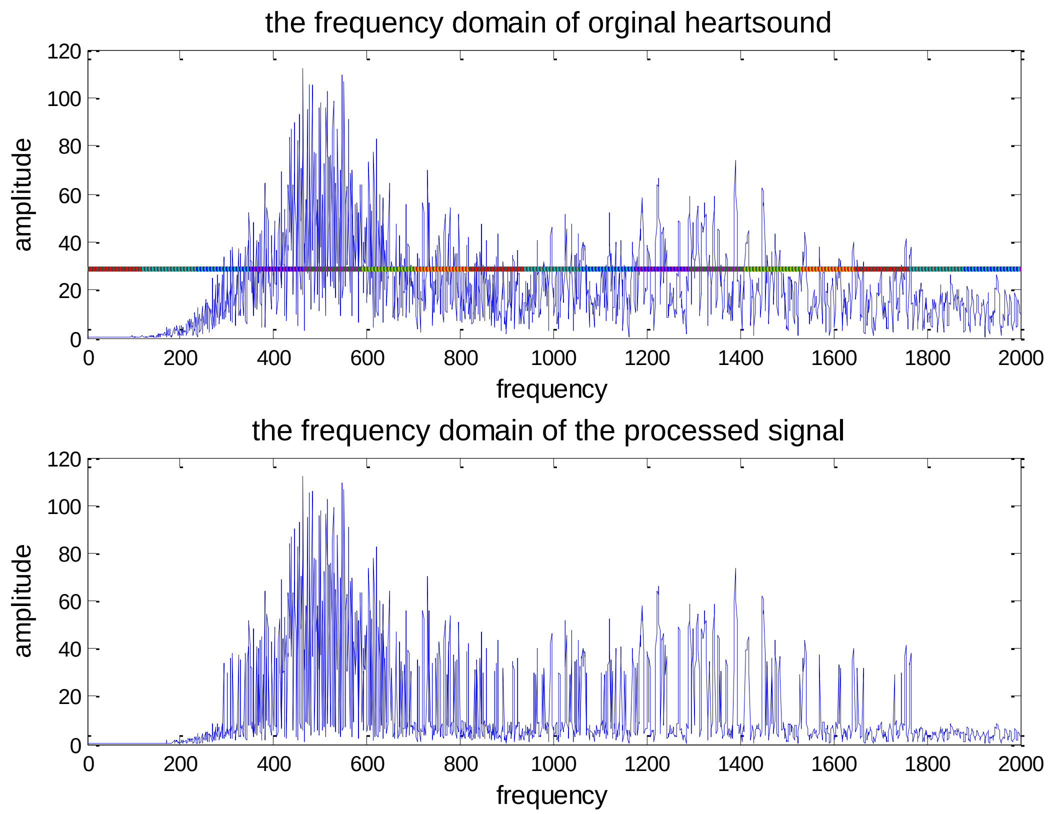

the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4


the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

Sample 4:

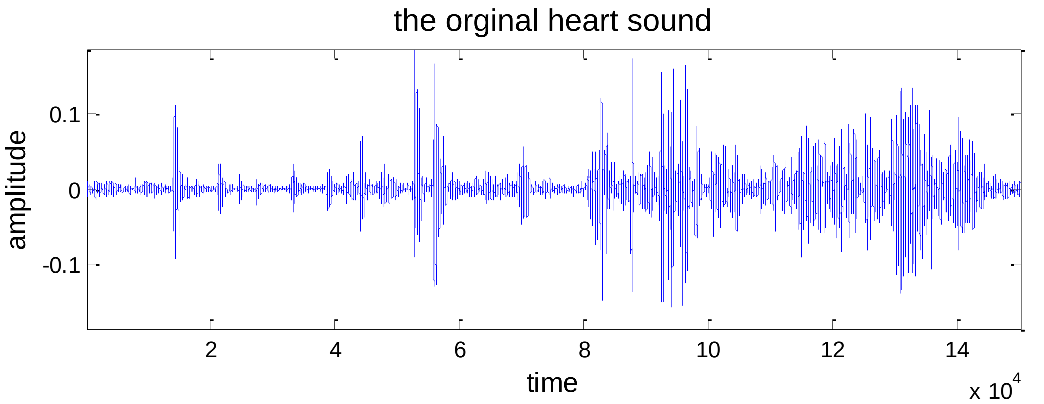


the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4

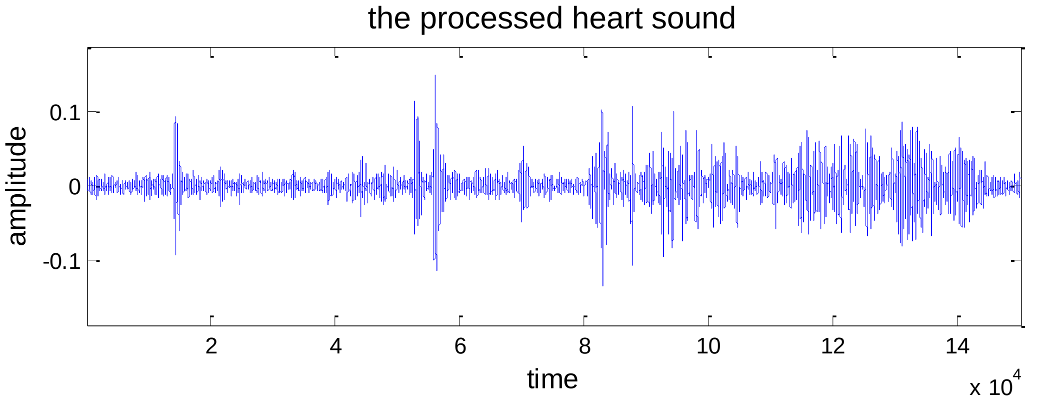


the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

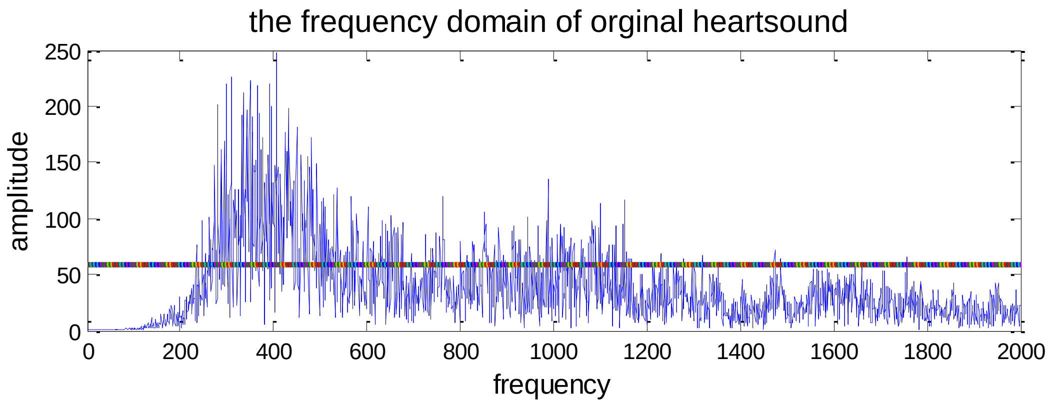
Sample 5:



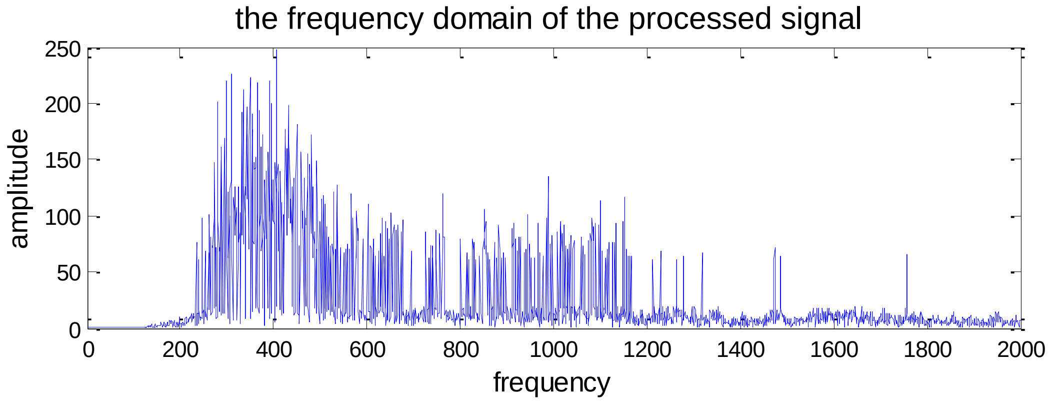
the orginal heart sound
amplitude
0.1
0
-0.1
2
4
6
8
10
12
14
time
x 10^4



the processed heart sound
amplitude
0.1
0
-0.1
2
4
6
8
10
12
14
time
x 10^4



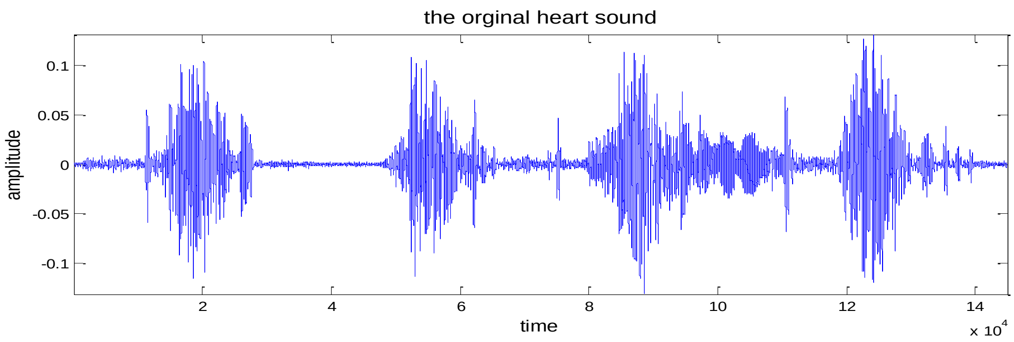
the frequency domain of orginal heartsound
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

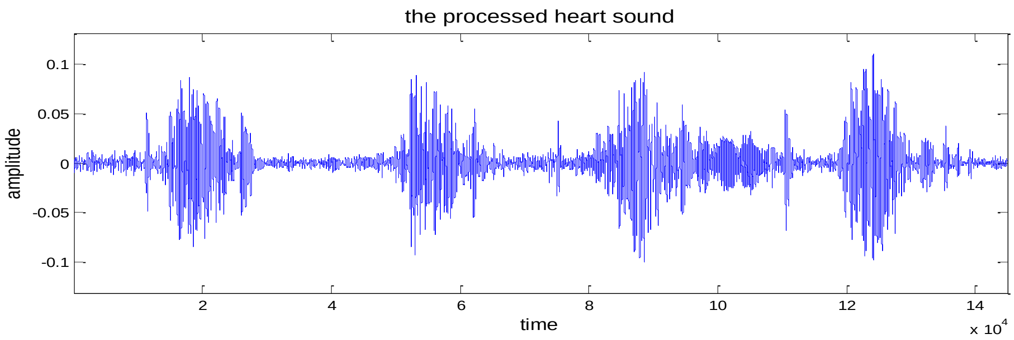


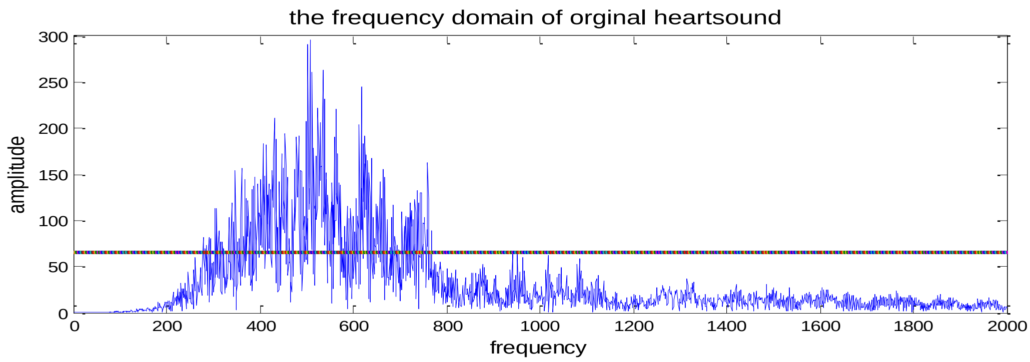
the frequency domain of the processed signal
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

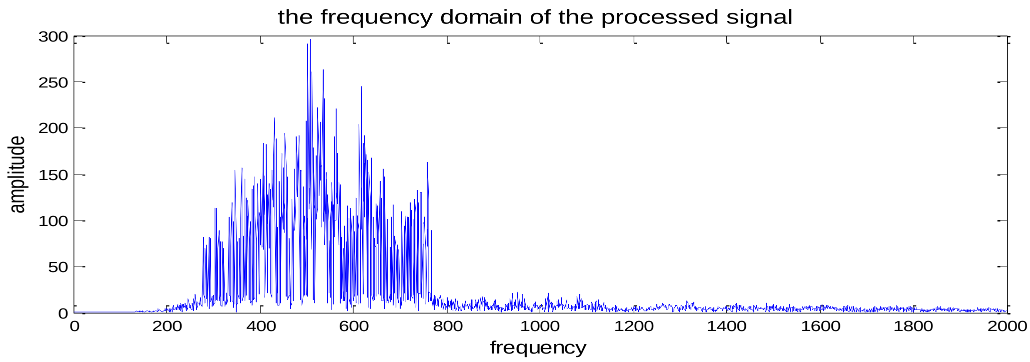
6. **The human voice noise**

Sample 1:

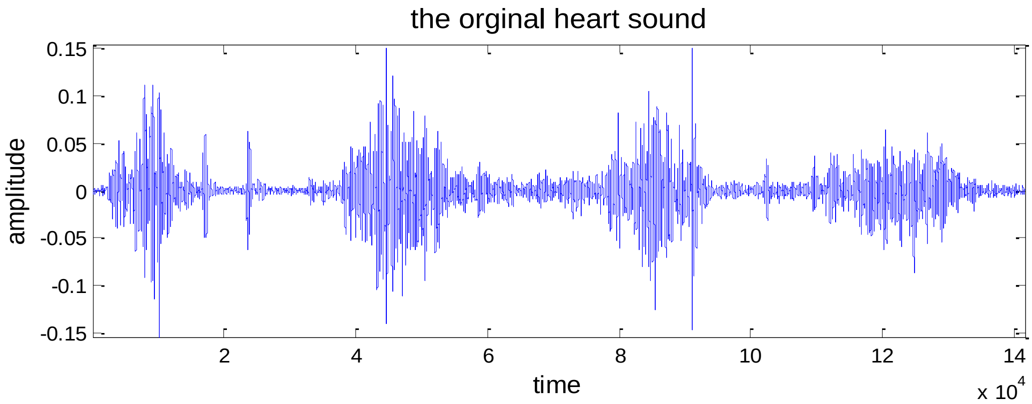


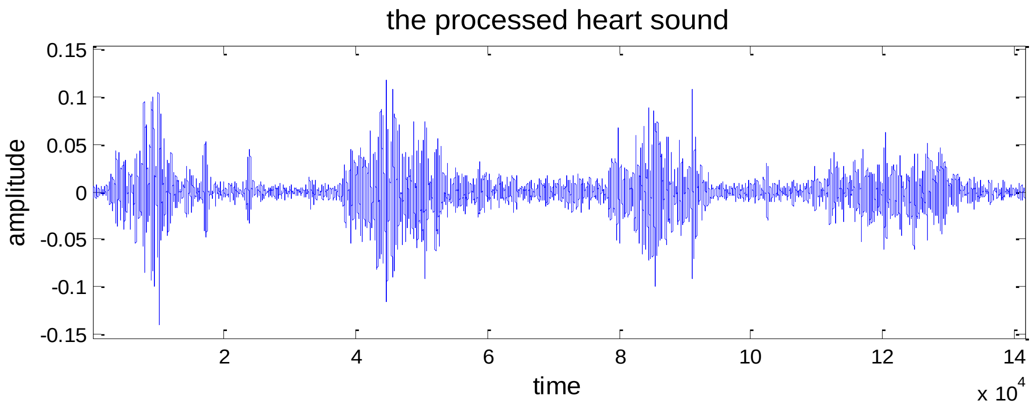


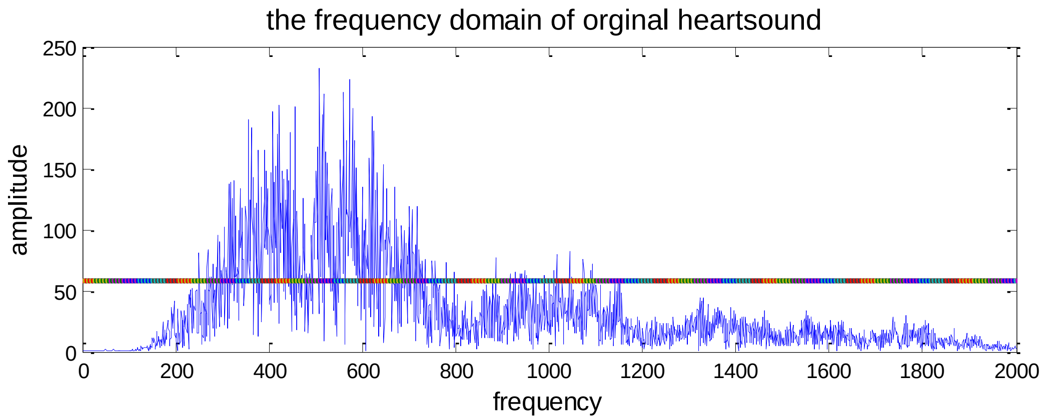


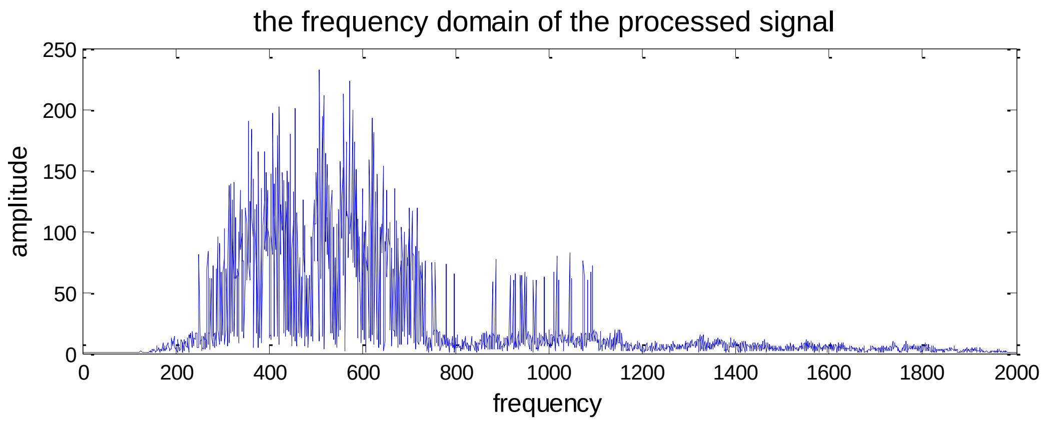


Sample 2:

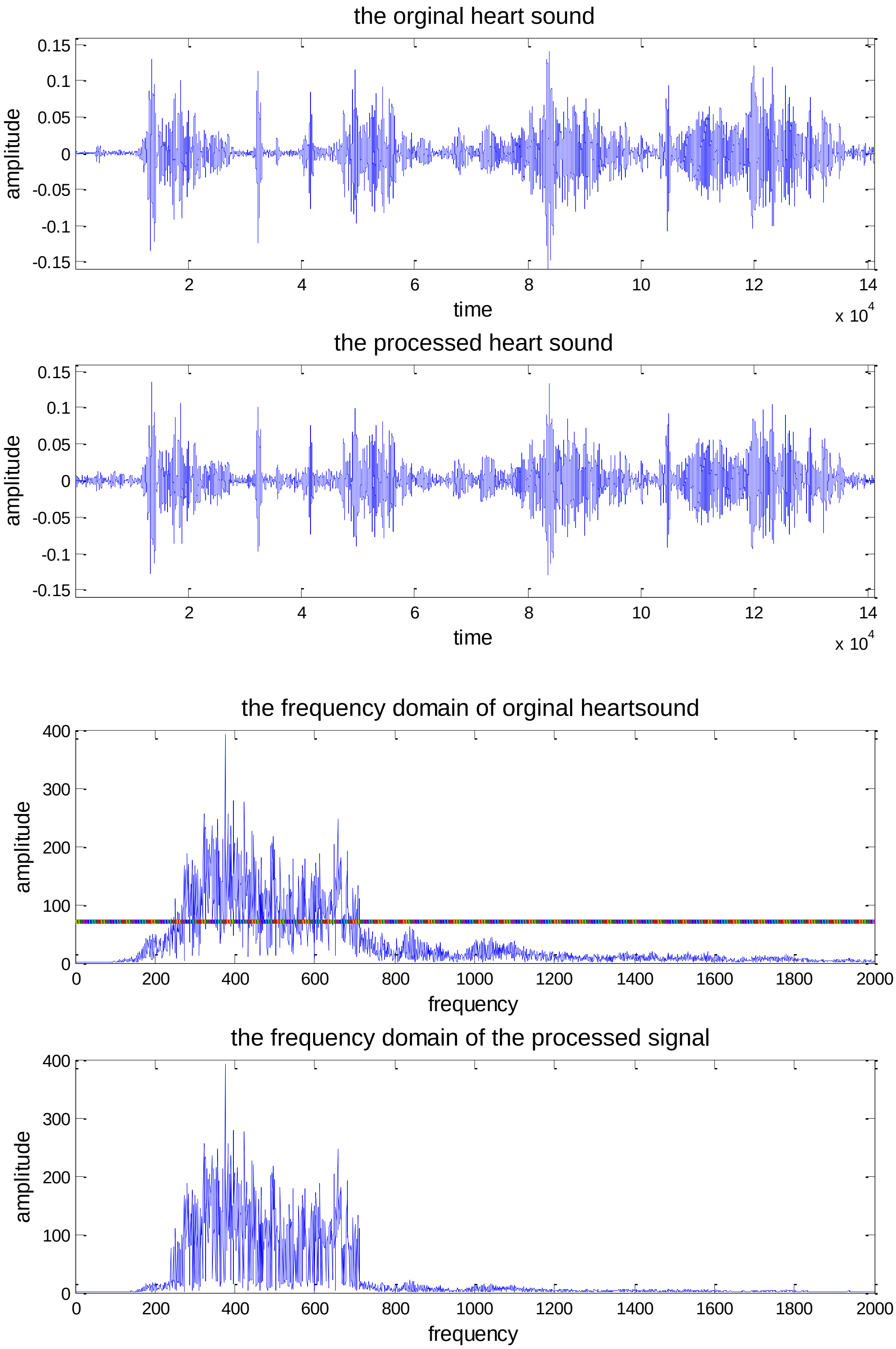

the orginal heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
time
x 10^4


the processed heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
time
x 10^4


the frequency domain of orginal heartsound
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

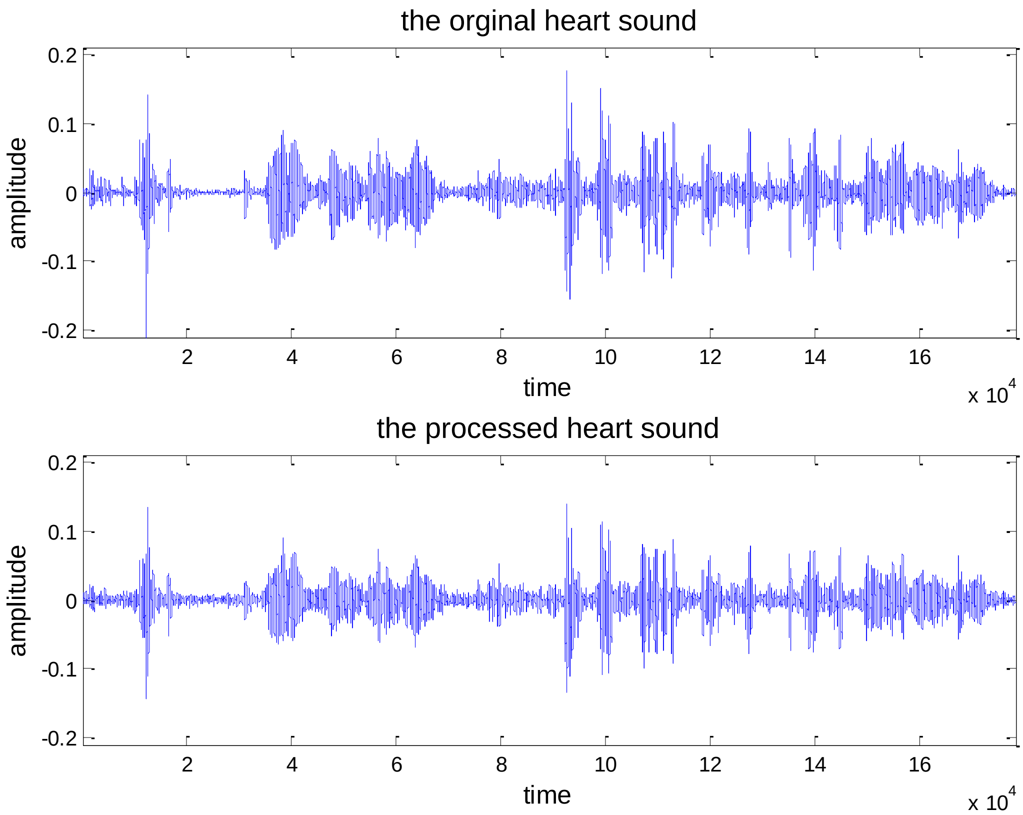

the frequency domain of the processed signal
amplitude
250
200
150
100
50
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

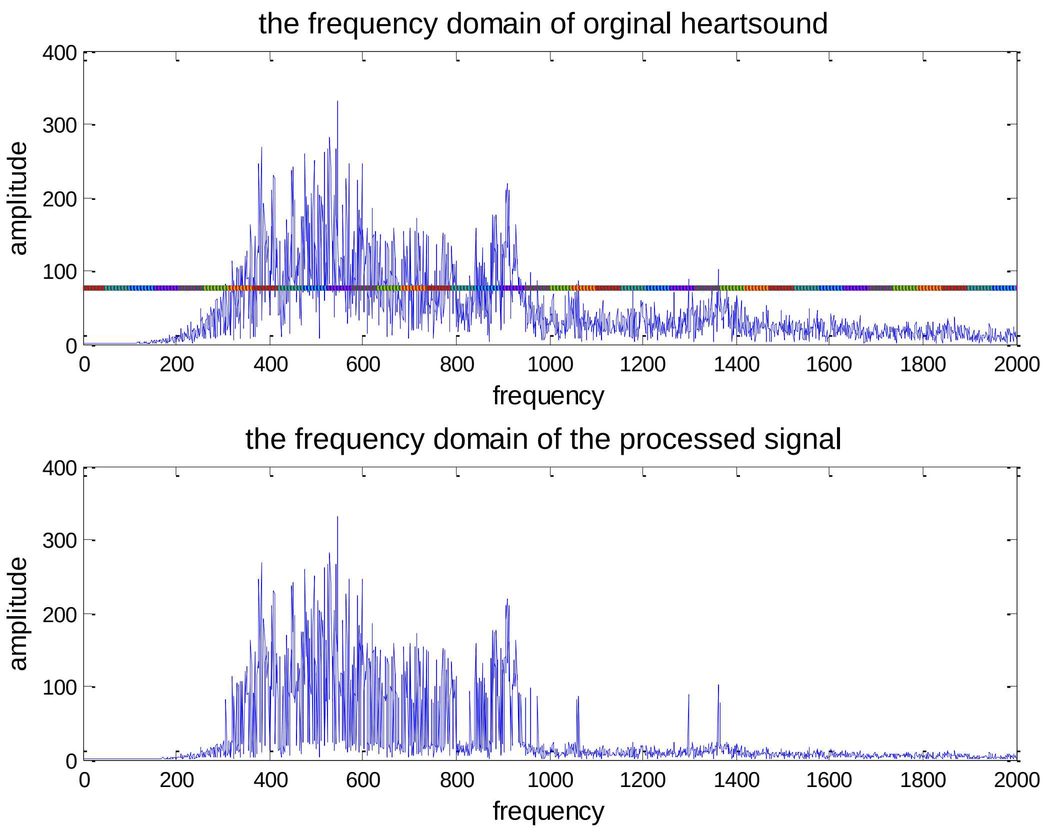
Sample 3:



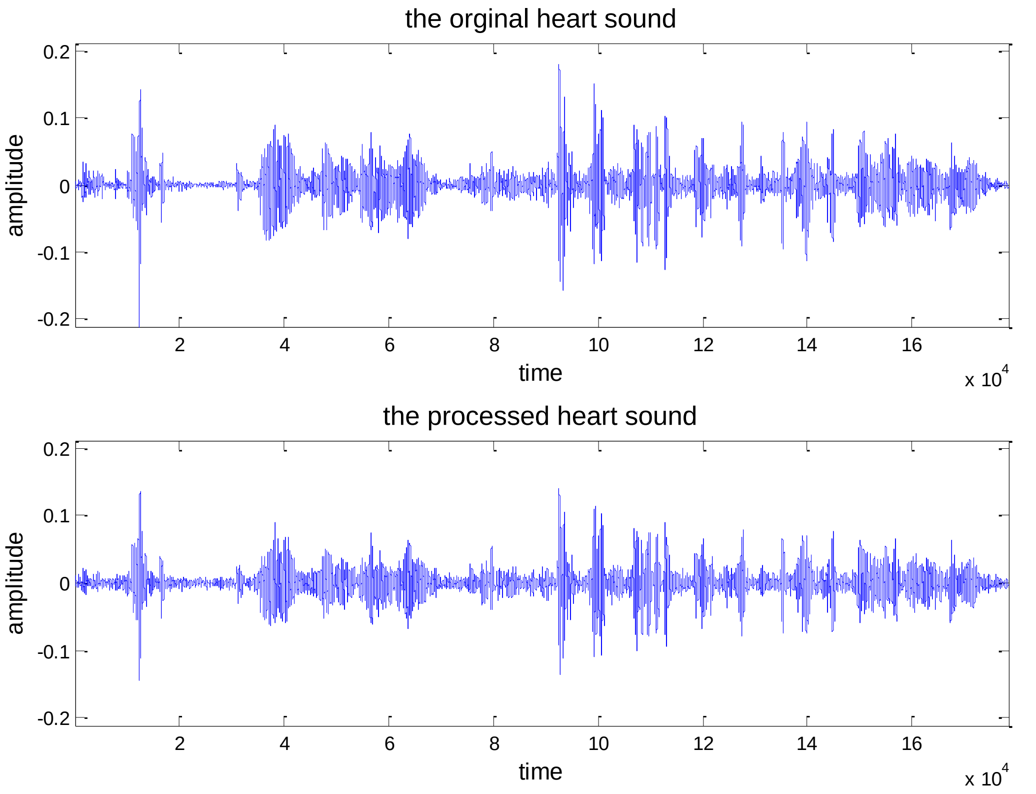
the orginal heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
time
x 10^4
the processed heart sound
amplitude
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
2
4
6
8
10
12
14
time
x 10^4
the frequency domain of orginal heartsound
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency
the frequency domain of the processed signal
amplitude
400
300
200
100
0
0
200
400
600
800
1000
1200
1400
1600
1800
2000
frequency

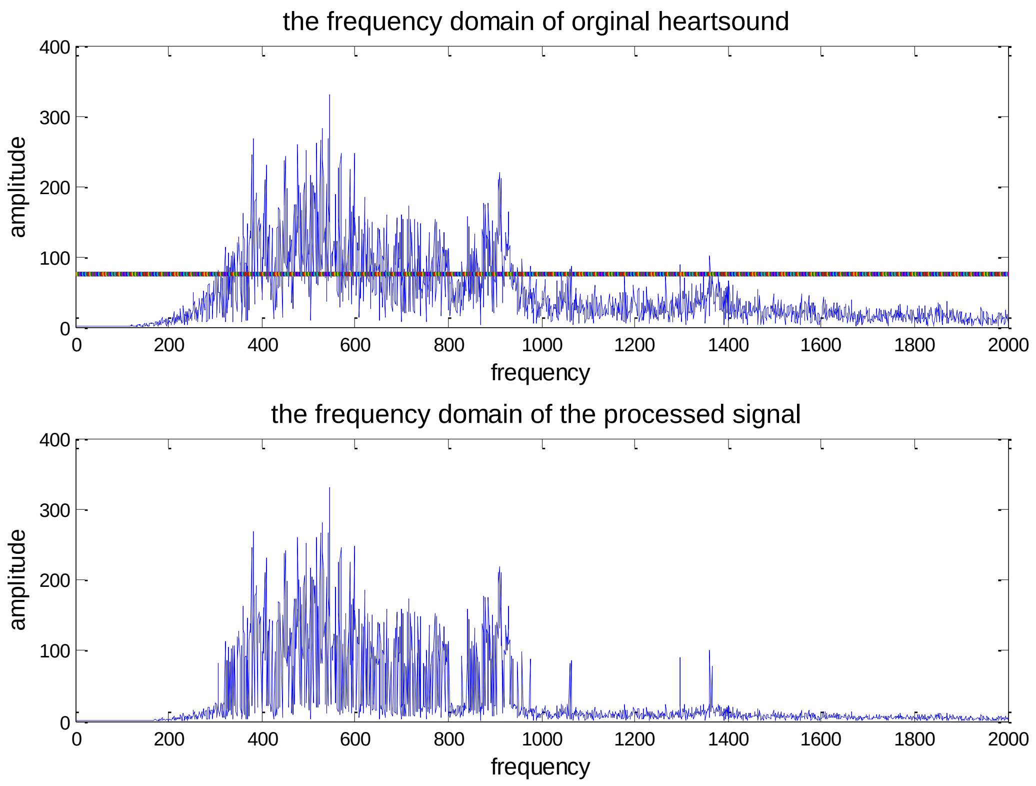
Sample 4:


the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4


the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency

Sample 5:


the orginal heart sound
amplitude
time
x 10^4
the processed heart sound
amplitude
time
x 10^4


the frequency domain of orginal heartsound
amplitude
frequency
the frequency domain of the processed signal
amplitude
frequency