ANALYSIS OF HEART SOUND SIGNAL IN REAL-LIFE SCENARIOS

Project report submitted in partial fulfillment of the requirement for
the degree of Bachelor of Technology

in

Computer Science and Engineering/Information Technology

By

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CANDIDATE'S DECLARATION

I hereby declare that the work presented in this report entitled **"Analysis of heart sound signals in real-life scenarios"** in partial fulfillment of the requirements for the award of the degree of **Bachelor of Technology** in **Computer Science and Engineering/Information Technology** submitted in the department of Computer Science & Engineering and Information Technology, Jaypee University of Information Technology Waknaghat is an authentic record of my own work carried out over a period from August 2017 to May 2018 under the supervision of **Dr. Vivek Sehgal,** Associate Professor, Computer Science & Engineering and Information Technology.

The matter embodied in the report has not been submitted for the award of any other degree or diploma.

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This is to certify that the above statement made by the candidate is true to the best of my knowledge.

Dr. Vivek Sehgal
Associate Professor
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Dated:

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LIST OF ABBREVIATIONS

ANC-Adaptive Noise Cancellation

CHD- Coronary Heart Disease

CT- Computed Tomography

CVDs - Cardiovascular Diseases

CWT- Continuous Wavelet Transform

DWT- Discrete Wavelet Transform

ECG- Electrocardiography

FHS- Fundamental Heart Sounds

HP- High Pass

HS- Heart Sound

LMS- Least Mean Square

LP- Low-pass

MRI- Magnetic Resonance Imaging

NRMSE- Normalized Root Mean Square Error

PCG- Phonocardiogram

PSNR- Peak Signal-to-noise ratio

RS- Respiratory Sound

SNR- Signal-to-noise ratio

STFT- Short-Time Fourier Transform

WHO-World Health organization

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ABSTRACT

Worldwide, Cardiovascular Diseases (CVDs) have been and continue to be the largest cause of death and heart valvular disease is one of the fast growing CVDs and these diseases are manifested in heart sound signals. The major problem with CVDs is that their symptoms occur at a later stage of the disease when it becomes difficult to cure. For early stage detection, a system should be there which would be convenient for the users to have frequent monitoring of the heart without a need to visit hospital and can be used at home or at workplace. So this project focuses on the processing of the heart sound signal to monitor the cardiac functions in real-life scenarios.

In this we will use Phonocardiography (PCG) for the analysis of heart sound signal. PCG uses a sensor called stethoscope to measure the acoustic sounds produced by the movement of the heart valves. However, the signals are vulnerable to environmental noises as well as noises generated due to the motion of the subject and noise generated due to movement of other organs such as lungs. In literature, various de-noising methods have been proposed to address such issues with the PCG systems. For this, a Discrete Wavelet Transform (DWT) based de-noising algorithm is proposed that thresholds the wavelet coefficients adaptively, according to the level of noise. The level of noise present in the signal is measured with a new statistical parameter, *med*75 that is based on the fact that the summation of the length of the fundamental heart sounds, S1 and S2, remains less than 25 % of the length of a cardiac cycle.

In conclusion, the thesis work has focused to propose a system that is robust against real-life noises and convenient to subject for its long-term use. The system with such features would make it possible to monitor the heart in ambulatory subjects and hence would minimise the unnecessary visits to hospitals. The work can be extended to classify the heart sound signal for specific diseases. Moreover, extreme noise conditions, such as running and jumping conditions of the subject can be considered in noise removal algorithm.

Chapter-1

INTRODUCTION

Introduction

One of the top causes of deaths in the world is Cardio-vascular diseases (CVDs) and these are responsible for more than one-fourth of the deaths all around the globe. Early stage detection of CVDs is very important. With early stage diagnosis, proper medication can be started timely and hence mortality rate can be reduced. Technological advances have been facilitating the research into both the creation of new areas and the development of existing methods of monitoring of physiological signals. Several methods are available in medical field that provide accurate evidences of CVDs. Some of the methods are:

- Magnetic Resonance Imaging i.e. MRI
- Electrocardiography i.e. ECG
- Computed Tomography i.e. CT

However these methods are expensive and are not used unless some problem is detected while listening to the heart sound using stethoscope i.e. cardiac auscultation. It usually depends on the physician’s ability and experience. However it is possible to get plots of the recorded heart sound signal using electronic stethoscope. Pathological conditions of the heart produce sounds which are different from those of the normal heart and hence the plot obtained is different from the normal heart sound plot. The graphical recording of cardiac sound is called phonocardiography. The plot obtained from recording heart sound signal is called phonocardiogram (PCG). PCG can be interpreted using different technologies and methods with the aid of computer systems. However the signal could get affected due to noises present in the vicinity. More substantial analysis of heart sounds and their processing using some improved fundamental methods may lead to cost efficient and accurate interpretation of the state of the heart. Hence, it is crucial to analyze heart sound signals properly and eliminate the noise and other interfering signal correctly during signal’s pre-processing [1].

Physiology of heart

The knowledge of physiology of the heart and its functioning is important for understanding the cardiography systems. The functioning of heart, cardiac anatomy and cardiac cycle are described in following section.

The human heart (Fig 2.1) is a muscular organ of about the size of a fist. Anatomy of heart reveals four cavities or chambers which are:

1. left atrium & left ventricle
2. right atrium & right ventricle

Its function is to pump deoxygenated blood collected from the body cells and tissues at different locations of the human body, to the lungs. The red blood cells then release carbon dioxide and absorb oxygen in the lungs. After this, it delivers the oxygenated blood back to the cells and tissues in the body. Pumping of blood is done via a complex system of blood vessels distributed in the human body, which is connected to heart. The lower two chambers (heart’s ventricles) are the main transport chambers of the heart. The upper chambers (heart’s atria) deliver blood to their respective ventricles.

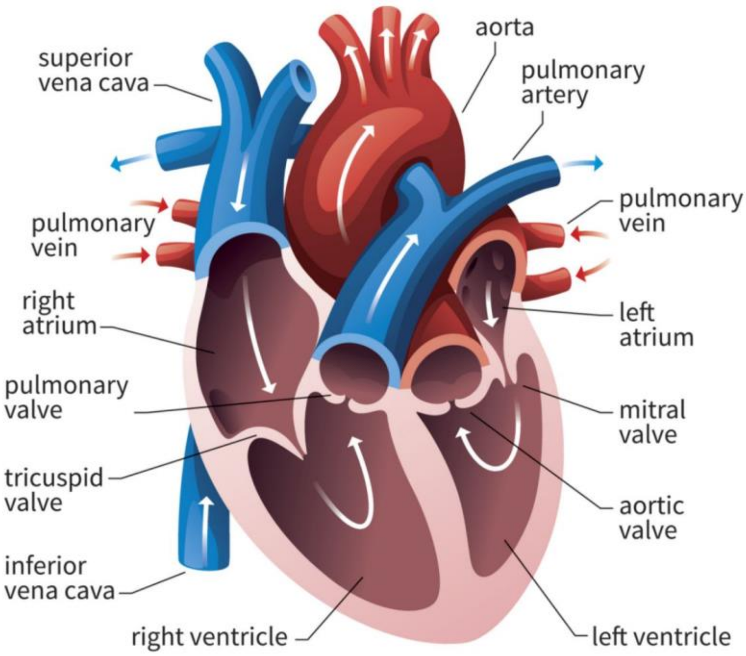

Fig 1.1: Internal structure of the human heart [20]

The human heart consists of four major valves to facilitate proper blood transport:

- Valve between left atrium and ventricle - mitral valve
- Valve between right atrium and ventricle - tricuspid valve
- Valve between left ventricle and aorta - aortic valve
- Valve between right ventricle and pulmonary artery- pulmonary valve

These valves ensure proper blood flow in forward direction by opening in proper time. And they also avert backflow of the blood when they get closed after the blood passes them. Thus, their major function in heart is the flow control. The mitral and tricuspid valves are called atrioventricular valves. Mitral valve is located between the left atrium and the left ventricle & tricuspid valve is present between the right atrium and the right ventricle. Aortic and pulmonary valves are present between ventricles and main blood vessels that transport blood. Aortic valve separates the left ventricles from the aorta and pulmonary valve separates right ventricle from pulmonary artery. All of the four heart valves are attached to a fibrous cardiac skeleton which is formed of dense connective tissues. It also act as a form of attachment for the heart valves and also for the ventricular and atrial muscles of the heart[2].

Path of Blood Flow in the Heart

The direction of arrows in figure 1.1 shows the direction of blood flow in the human heart. There is a superior vena cava and an inferior vena cava. The former mentioned vena cavae deliver deoxygenated blood collected from the body cells and tissues back to the upper right heart chamber. The right atrium then narrows and shrinks which makes the blood enter the lower right chamber or ventricle. Then the right ventricle shrinks which transports the blood across the pulmonary valve and eventually entering pulmonary artery. Pulmonary artery delivers the deoxygenated blood to the lungs for oxygenation and removal of carbon dioxide. The oxygenated blood is transported to the left upper chamber or left atrium of the heart via pulmonary veins. Left atrium then narrows & shrinks resulting in transport of blood across the opening of mitral valve and eventually entering into the left ventricle. Then left ventricle narrows and pumps the blood across the aortic valve and then entering into the aorta. The aorta then delivers the oxygenated blood to all body cells via a network of smaller blood vessels in the body.

Cardiac cycle

The human cardiac cycle consists of rhythmic atrial and ventricular contractions and relaxations that deliver blood into pulmonary and systemic circulations. The contraction of lower heart chambers or ventricular contraction is called systole and the ventricular relaxation is called diastole. Referring to figure 1.1, during diastole, blood moves from the atria, through tricuspid and mitral valve openings into the ventricles. The following event called systole which is caused by ventricular contraction. The ventricular pressure will become larger than the atrial pressure which will result in the closure of tricuspid and mitral valves that will prevent backflow of blood. This leads to the generation of the first heart sound S1. S1 consists of two components:

- mitral component, occurring slightly earlier than tricuspid component
- tricuspid component

The ventricular pressures will exceed the diastolic pressures inside the pulmonary artery and aorta, forcing the opening of the pulmonary and aortic valves and the blood will be able to flow through for pulmonary and systemic circulations. The ventricular pressures will decrease as compared to the pressures of the major blood vessels of the heart i.e. pressure of pulmonary artery and aortic pressure, closing the semilunar valves to prevent backflow of blood into ventricles. The shutting down of the valves leads to the generation of the second heart sound S2. S2 is made up of 2 components:

- aortic (A_2) component
- pulmonic (P_2) component

P_2 usually occurs after A_2 as the pressure gradient between the aorta & the left ventricle is larger as compared to that between the pulmonary artery & the right ventricle. The ventricular pressures will then decrease as compared to the pressures in the two atria, leading to the opening of the atrioventricular valves. Diastolic ventricular filling will occur again and then the cycle repeats itself. Systole can be determined by the time span between S1 & S2, and diastole determined by the time span between S2 & the next S1. The main heart sounds S1 and S2, form a framework from which all other heart sounds and murmurs can be identified.

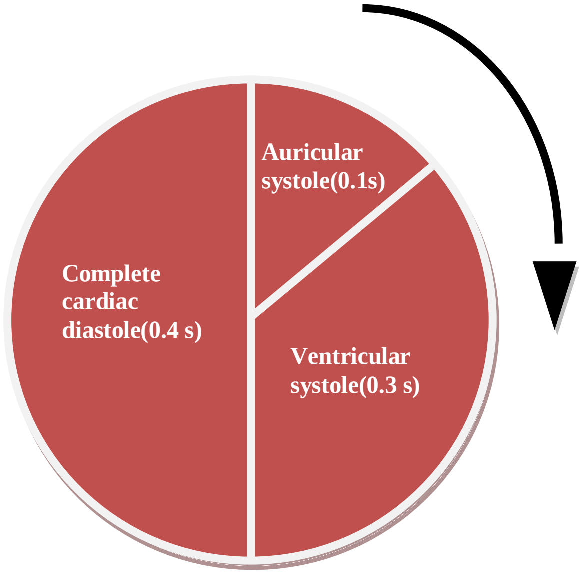

Fig 1.2: Cardiac cycle [21]

Phonocardiography (PCG)

Cardiac sound signals carry the information the physiological and pathological traits of the heart. Cardiac cycle is of very short time span (0.8 seconds). Each recorded sound is complex. The frequency of heart sound signal is found between 10Hz to 250Hz. PCG records the heart sounds using a simple set-up. The noise in the surrounding is also caught in the samples [3]. PCG is most commonly used tool in clinical detection and verification of CVDs because of its various features which includes easy to use set-up and classic process. The set-up consists of a sensing device called stethoscope. Because of stethoscope, PCG becomes a very portable, cheap & non-invasive method of recording heart sounds. As shown in Figure 1.5, PCG signal consists of two Fundamental Heart Sounds(FHS) i.e.

- S1, also called Lub – closing of the mitral and tricuspid valves
- S2, also called Dub – closing of the aortic and pulmonary valves

Due to the presence of the electrical impulse at ventricles, they simultaneously contract. This leads to the closing of the atrioventricular valves which prevents backflow of blood into the atria. This results in production of the first heart sound i.e. lub. The ventricular pressure then drops below the pressures of the pulmonary artery as well as the aorta. It leads to the closing of the mitral and tricuspid valves. This results in generation of the second heart sound i.e. dub.

S1 (First heart sound):

- This component is loudest
- Energy lies between 20-120 Hz
- Lasting period is 110-150 ms
- Consist of 2 elements: one occur when mitral valve is closed and the other one when the tricuspid valve is closed
- The mitral sound part occurs before tricuspid part since the left ventricle contracts earlier compared to right one.

S2 (Second heart sound):

- The closure of aortic and pulmonary valves generates S2.
- S2 consists of 2parts: the aortic (A2) and pulmonic (P2) elements
- P2 occurs typically 30ms after A2.

The interval between Lub and Dub is known as systole, while the interval between second and first heart sound is known as diastole.

There are rarely occurring S3 and S4 components other than the fundamental heart sounds.

S3 and S4 heart sound components

- S3 takes place at the beginning of the diastole
- frequency range is between 70-90 Hz.
- S4 occurs at late diastole.

There are additional sounds known as murmurs. The frequency range of murmurs is 100-700 Hz. They stipulate cardiac diseases like pulmonary stenosis, mitral regurgitation i.e. failure of mitral valve, aortic stenosis i.e. aortic valve narrowing, and mitral stenosis.

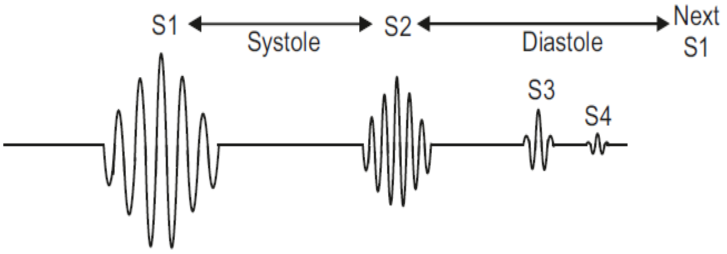

Fig 1.3: Single cardiac cycle represented by PCG [9]

Sound	Origin
S1	when semilunar valves closes
S2	when atrioventricular valves closes
S3	filling of ventricles at early diastole
S4	filling of ventricles at late diastole

Table 1.1:Classification of PCG signal [11]

Heart Sounds

Lub - First Heart Sound (S1)

S1 or Lub is the first heart sound. It has high frequency and is loudest. Heart's mitral and tricuspid elements are separated by nearly 0.01s. The factors determining intensity of S1 [2]:

- The distance between open flaps of the atrioventricular valves at ventricular contraction.
- The ability of atrioventricular valves leaflets to move.
- Rising rate of pressure in ventricles.

The contraction of ventricles causes the closure of tricuspid and mitral valves to close, when the pressure in ventricles exceed the atrial pressure. S1 is prominent when the valve pamphlets don't have enough time to float back together or when they are constrained closed when they are wide separated. This happens in mellow mitral stenosis where a delayed diastolic weight inclination happens between the left chamber and ventricle, keeping the two valves more extensive separated than ordinary amid diastole [2]. S1 may likewise be constricted when the heart rate is quicker than ordinary (i.e. tachycardia), since diastole is abbreviated and the two valves have lacking time to float together. They are consequently constrained closed from a wide distance separated.

In patients with mitral regurgitation or reverse backflow of blood, S1 is decreased in power in light of the fact that the mitral handouts may not come into full contact with each other as they close. In serious mitral stenosis (calcium stores on valve), the pamphlets are relatively settled in position and less versatile all through the heart cycle. They can't close completely and S1 is reduced. In patients with solid left ventricles (hypertrophied or scarred chamber), atrial constriction causes a higher than ordinary ventricular diastolic weight [2]. The higher weight accelerates the floating together of the mitral pamphlets. In this way the pamphlets are constrained closed from a shorter distance separated amid ventricular constriction, lessening the force of S1.

Dub - Second Heart Sound (S2)

Difference between S1 and S2 is that S1 is heard as a solitary sound typically while S2 has 2 parts A2 and P2. These are consolidated during expiration and separated during inspiration. As indicated by specialists, variations from the norm of S2 incorporate changes in its force and changes in the example of its part [2]. The power of S2 is dictated by three elements [2]:

- Separation between the flaps of the opened semilunar valves at the event of ventricular unwinding.
- Versatility of the semilunar valves' flaps.
- The effect of backflow of blood on the shut semilunar valves.

The manners in which the initial two elements influence the force of S2 are like that of the initial two components influencing the power of S1. S2's force likewise relies upon the speed of blood streaming back towards the valves from the aorta and the pulmonary artery route toward the finish of ventricular compression and the suddenness with which the blood reverse is countered by the end valves. In fundamental hypertension or pulmonary arterial hypertension, the diastolic weights in the aorta and aspiratory supply route are more prominent than typical which builds the speed of blood stream back towards the valves [2]. This reinforces the force of S2. However in extreme aortic or pulmonic stenosis, the valve pamphlets are less versatile, diminishing the power of S2.

Murmurs

A murmur is a sort of heart sound that occurs because of turbulent blood stream. Blood stream is laminar and noiseless under typical conditions. Notwithstanding, it can be aggravated because of haemodynamic or basic changes in the heart or vasculature to deliver a capable of being heard clamor. Murmurs can be delivered in the accompanying circumstances as recommended by specialists [2]:

1) Blood stream over an impeded opening (e.g. in case of aortic stenosis).
2) Quickened blood stream across ordinary structures
3) Launch of blood into a dilated chamber (e.g. in case of aneurysmal expansion of the aorta).
4) Reverse flow over a clumsy valve (e.g. in case of mitral regurgitation).
5) Strange shunting of blood starting with one vascular chamber then onto the next low-weight chamber (e.g. ventricular septal imperfection).

Murmurs are described by their planning, power, pitch, shape, area, radiation and reaction to moves.

Diastolic Mumbles

- **Early Diastolic Murmurs** result from semilunar valve reverse and start with the second stable (beginning of diastole) and stretch out for variable length into or through all of diastole (holodiastolic); a shorter mumble for the most part suggests a more extreme injury.
- **Mid Diastolic Murmurs** result from inflow, which starts around 0.12 second after S2.
- **Presystolic Murmurs** reach out into to the primary sound.

Systolic Murmurs

- **Early Systolic Murmurs** start with the first heart sound and don't stretch out to the second.

- **Mid Systolic Murmurs** are isolated from the first and second sounds and are for the most part, however not generally, related with discharge through a surge tract.
- **Holosystolic Murmurs** by definition begin with the first stable and stretch out to or through second solid. The term is desirable over "pansystolic" which implies each systole, though holosystolic implies all of systole.
- **Late Systolic Murmurs** start after the first solid, and reach out to or through the second.

Continuous Murmurs

- **Continuous Murmurs** reach out from systole during that time sound into diastole and ought not be mistaken for the to-and–fro murmurs of aortic disgorging in which there is an unmistakable interruption between the systolic (anterograde) and early diastole (retrograde) stream. Conversely, constant murmurs result from unidirectional spill out of a high weight source to a lower weight beneficiary.

Problem Statement

One of the biggest problems with cardiovascular diseases is that they become difficult to cure because their symptoms occur at later stage of the disease.

Motivation: Increasing mortality rate due to CVDs

- According to the survey report of WHO of September 2016 more than 17 million people die from cardiovascular diseases every year which comprises 31% of all deaths worldwide. It's one of the top causes of deaths in the world. [WHO survey].
- According to "Cardiovascular Disease (CVD) surveillance and health promotion in Industrial settings: A module for CVD surveillance and health promotion" report submitted to WHO in 2007, more than 2.5 million people in India are estimated to die because of coronary heart diseases which constitutes 54.1% of all the death caused due to cardiovascular diseases in India by 2020.

- According to a 2016 survey in India, there are more than 40 million heart patients in the country. Among these, 19 million people belong to urban areas & 21 million belong to rural areas.

The main motivation behind this project is to put an effort to decrease the mortality rate due to CVDs which is increasing rapidly. We cannot eliminate the disease but we can decrease the chances of deaths by detecting the disease at early stages before it becomes life threatening. By early diagnosis people get aware of their heart conditions and they can take precautions in early stages and hence can cure themselves on time.

Scope:

We can produce better de-noising results using appropriate thresholding techniques and de-noising methods. As a result, the mortality rate due to CVDs can be reduced.

Objectives

One of the issues with CVDs is that their symptoms occur, some of the time, at a later stage when the conditions get exacerbate. Absence of beginning period identification and subsequently medication delay makes cardiac diseases worse to the extent when it becomes difficult to cure. PCG is most broadly utilized technique in center as screening device for conclusion of CVDs in light of its highlights including simple to work and timeless set-up process. It utilizes a sensor called stethoscope. Stethoscope makes PCG a profoundly compact, minimal effort, and non-obtrusive cardiography method. PCG motion, as appeared in Figure 1.5, comprises of two Fundamental Heart Sounds (FHS) known as S1 (Lub) & S2 (Dub). Our task's primary goal centers around enhancing existing techniques for de-noising, division and examination of PCG signal.

Methodology

Analysis of Wavelet

A wavelet can be defined as a type of waveform of successfully constrained term that has a zero average value. Here we are contrasting wavelets & basic sine waves from each other, which are basically the proposition of Fourier examination of the signals. Sinusoids don't have restricted term. They stretch out in continuity from minus infinity to plus infinity & sinusoids are smooth & foreseeable. On the other hand, wavelets have a tendency to be unpredictable & asymmetric.

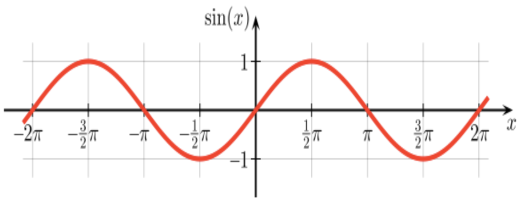

Fig. 1.4: Sine wave (Source: [22])

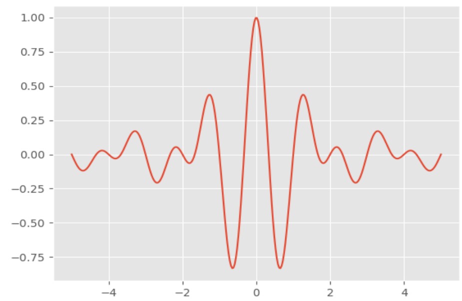

Fig. 1.5: Wavelet (Source: [23])

The series of wavelet is only a sampled version of Continuous Wavelet Transform & its calculation may devour critical measure of time & resources, and also upon the resolution requirement. The Discrete Wavelet Transform i.e. DWT, which is dependent on the sub-band encoding, is found to give quick results of Wavelet Change. It is anything but difficult to execute and reduces the calculation time and resources required [20].

Mallat's algorithm

On account of DWT, expecting that the length of the signal fulfils N = 2J for some positive J, the transform can be figured proficiently, utilizing Mallat's algorithm[4], which has a complexity of O(N). Basically the calculation is a quick progressive plan for determining the required internal items utilizing an arrangement of successive low and high pass filers, trailed by a demolition. This outcomes in a decay of the signal into various scales which can be considered as various frequency groups.

The low-pass (LP) and high-pass (HP) channels utilized as a part of this calculation are determined by the mother wavelet being used. The yields of the LP channels are alluded to as approximation coefficients and the yields of the HP channels are alluded to as detail coefficients.

At every decomposition level, the filters deliver signals crossing just a half portion of the frequency band. This twofold the frequency determination as the vulnerability in frequency is decreased to 50%. In this manner, the yield of each filter can be down-inspected (decomated) by a factor of 2 as per Nyquist's criteria. Utilizing this approach, the time determination turns out to be great at high frequencies, while the frequency determination turns out to be subjectively great at low frequencies. The quantity of decomposition levels ought to be dictated by the user (as indicated by the characters of the signal to be broke down). Typically signal is decomposed up to $3-5$ steps. The Discrete Wavelet Transform of the first signal is gotten by connecting every one of the coefficients beginning from the last decomposition level of the signal. The detail coefficients have a tendency to have high an incentive in the noise containing parts of the signal.

DWT process

The implemented work focuses on analysis of heart sound signals and applying different de-noising techniques and compares them. The main focus is on de-noising of phonocardiogram signals which is done using various families of discrete wavelet transforms, threshold calculation, thresholding methods and techniques & signal decomposition levels. Wavelet de-noising involves various steps.

1. Multilevel decomposition of signal – First step is to obtain approximation and detailed coefficients. This is done by performing multilevel decomposition of the signal. Discrete wavelet transform splits up a signal into two bands i.e.
    - low pass sub-band which is also called ‘approximation level’
    - high pass sub-band which is also called ‘detailed level’

   The approximation level can be decomposed into multiple levels for fine scale analysis. The first level detailed coefficients capture the high frequencies of the signal. Our aim is to retain the original heart signal while eliminating the disturbances. Hence we scale the detail level coefficients using the calculated threshold value. We here decomposed the signal into 5 detail levels and one approximation level.

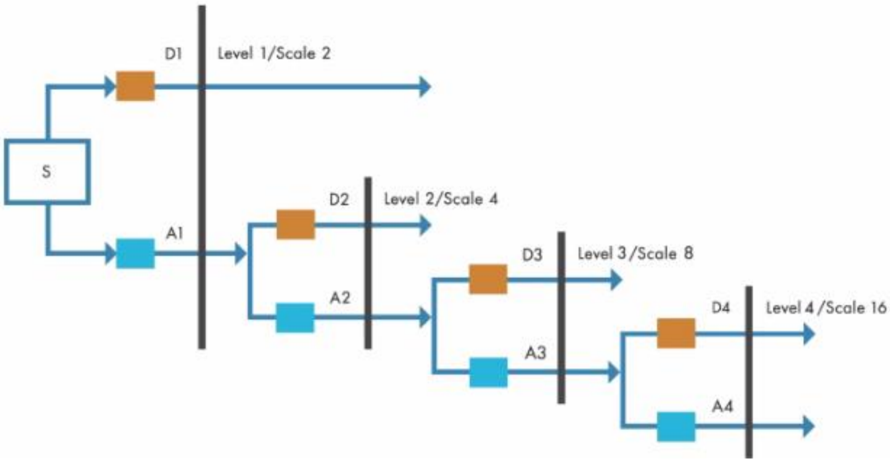

Fig 1.6: DWT decomposition level (source [7])

2. Identify a thresholding technique –The work implemented here introduces a new thresholding function. Standard thresholding techniques are soft thresholding and hard thresholding. The threshold calculated is compared with the threshold calculated using other basic techniques and is found more efficient.
3. Threshold the detailed coefficients and reconstruct the signal – The coefficients are thresholded using the calculated threshold. And then the signal is reconstructed. Reconstruction of the signal included the use of fourth and fifth detail levels. Other levels are reduced to zero.

Particularly, our purpose is to analyze the effect of the selected mother wavelet and number of wavelet decomposition levels on the accuracy of different de-noising algorithms. De-noised signals are observed with the actual recorded Phonocardiogram signal to find out the most satisfactory values for parameters i.e. wavelet family, number of decomposition levels, and the type of thresholding technique used for the de-noising process. The evaluation of different techniques and values is done using signal-to-noise ratio (SNR value). The analysis shows that the most important parameters are: levels of decomposition & thresholding type. These factors affect the accuracy of the de-noising method used. Finally, the results are compared with those from other studies to get clear idea of the techniques.

Flowchart for Discrete wavelet transform (DWT)

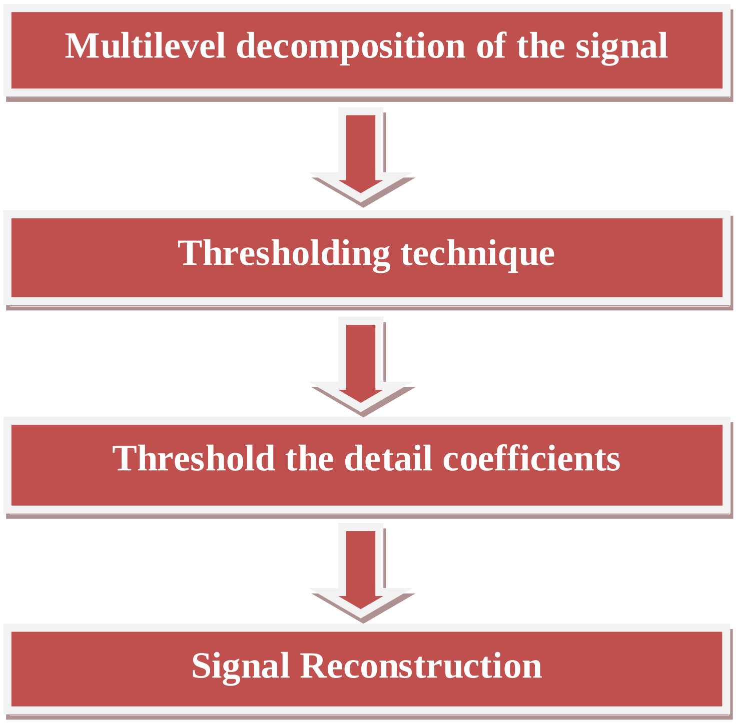

Fig 1.7:DWT flowchart

Organization

Chapter 1: Introduction.

It contains physiology of heart, path of blood flow in the heart, cardiac cycle & phonocardiography. It also provides information about lub and dub, types of murmurs- diastolic murmurs, systolic murmurs, holosystolic murmurs, continuous murmurs etc. Problem statement, scope, objective and methodology are also included.

Chapter 2: Literature Survey

It contains all the details of the papers that have been reviewed during our understanding towards the project.

Chapter 3: System Development

It contains the whole methodology that has been used i.e. processing of the heart sound, signal decomposition using wavelet transformation.

Chapter 4: Performance Analysis

It contains comparison of different methods of de-noising which include Rigrsure, Minimax, Sqtwolog and adaptive thresholding technique for de-noising.

Chapter 5: Conclusion

It contains the conclusion and the future scope of the project work.

Chapter-2

LITERATURE SURVEY

Literature Survey

De-noising of signals stays to be a fundamental issue in the field of signal analysis and their processing. Many research works and papers detailed in the writing on de-noising utilizing different techniques & wavelet transform. S. Djebbari et al. (2000) introduced after-effects of the PCG signal examination utilizing the Short Time Fourier Transform i.e. STFT in one of the publication. Messer et al. (2001) proposed the benefits of PCG over customary auscultation. They might be replayed and broke down for knowledge and information about frequency. Several de-noising algorithms for the heart sound signals in time & frequency domain have been proposed in literature. Anindya S. Paul et al. (2006) created technique for the isolated channel lessening of the noise present in cardiac signals that have been recorded. F. Jin et al. (2008) assessed the issue of cardiac sound confinement from isolated channel respiratory sounds chronicles by applying restriction method based on wavelets. Vikhe et al. (2009) worried about the investigation of lub and dub sound in the PCG by utilizing Discrete Wavelet Transform (DWT) & Continuous Wavelet Transform (CWT).

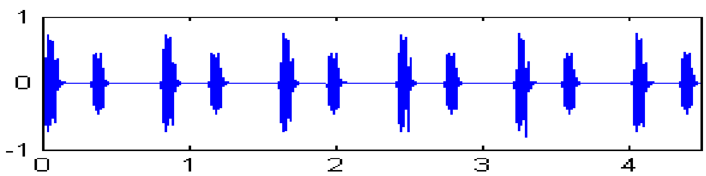

Fig 2.1: Normal heart sound signal [24]

2.1 "Comparison of wavelet transforms for de-noising and analysis of PCG signal"

In this paper, PCG (Phonocardiogram) signals and their de-noising techniques are discussed. PCG is graph obtained by recording the heart sound signals. PCG is prone to noise interference since it is comparatively weak. Because of its vulnerability, it is important to de-noise the PCG signal to get better and accurate diagnosis results. Several methods are available for de-noising of PCG. We are still looking for the best result producing and most effective method for de-noising. In this paper, 4 types of PCG signals are de-noised and different parameters are checked and compared. It is concluded in the result that which wavelet gives the better and accurate result for all four types PCG signals.

In this research work, a new technique is proposed by the authors which generates the entire graph for cardiac sound analysis that helps the doctors in diagnosis of the heart disease or other problems related to the heart. Different types of noises are added to the PCG signal and the results are compared using various parameters. In the standard heart sound signal, 3 common types of noises are added which include white Gaussian noise, random noise and pink noise. Various parameters and ratios are calculated like SNR, PSNR and NRMSE. These values are compared for each PCG signal. A comparison of the different wavelet shows the resolution differences among them. It is found in the work that the Daubechies wavelet filter (db2) and level 4 it gives the maximum value of SNR, PSNR and minimum RMSE for standard the sample heart sound and for other three heart sound, coiflet wavelet gives maximum values of SNR and minimum NRMSE. In the de-noising process of PCG signal, this wavelet (coiflet) is used and also spectrogram is provided for analysis of PCG signal. [5].

2.2 "Denoising of pcg signal by using wavelet transforms"

In this research work, a technique for isolated channel reduction of noise present in cardiac signal recordings is proposed and implemented. Various sources of noise can contaminate the heart sound signal which makes it difficult to analyze. These sources include sounds in vicinity, lung sounds due to breathing, contraction of muscles etc. A 'decision-directed' approach is used in this technique which helps in estimation of noise in the signal without changing or separating the original signal. De-noising can be achieved using different thresholding functions. In this paper different functions are used and compared and the proposed function is compared to the standard thresholding functions:

- Traditional Soft thresholding function
- Traditional Hard thresholding function
- New upgraded 'novel thresholding function'

This new function which consists of double alterable parameters is 'novel'. One of the crucial decisions in wavelet-based de-noising of heart sound signal is the selection of threshold and the type of wavelet. Different wavelets can give different results. The new thresholding function overcomes the shortcomings of traditional functions. This function de-noises the signal effectively by selecting appropriate and efficient wavelet.

To illuminate the accuracy of the proposed novel thresholding function introduced in the paper, a part of the basic heart sound signal in the basic Phonocardiogram is selected and tested two times. Firstly, by adding Gaussian noise of low strength to the basic signal as the first sample signal. Secondly, by adding Gaussian noise of high strength to the signal as the second sample signal. Then different parameters such as SNR, PSNR and NRMSE are calculated and the values are compared with the values of other methods.

It was observed that in case the signal of more than 0.2 volts & the signal in which less noise is present, '*sym6*'produces the best results and for less than 0.2 volts amplitude and noisier signal the '*coif5*'produces the best results. We can use coif5 wavelet for people suffering from the cardiac diseases or having possibility or suspicion about some heart condition. And for people like players and young people, we can use the sym6 wavelet for analysis[6].

2.3 “An Adaptive Method for Shrinking of Wavelet Coefficients for Phonocardiogram De-noising”

In this paper, again an attempt is made to improve de-noising methods of PCG signals. A new adaptive method is introduced in the paper to estimate the threshold value. This threshold value is used for convenient reduction of the wavelet coefficients obtained from the Phonocardiogram signal. This work has been implemented and compared here in our project. A new statistical measurement factor is introduced and calculated by assimilating the medical understanding of the Phonocardiogram signal recorded to analyze in the system. The value of threshold is calculated on the basis of the following:

- Study of the coefficients obtained from the wavelet
- Current noise level

Moreover, to beat the problems present in currently available functions used for thresholding, new threshold functions are introduced. Soft and hard thresholding are traditionally used techniques. ‘Mid function’ and ‘non-linear mid function’ techniques are introduced. The introduced strategy is experimented on the Phonocardiogram signal tainted with different noises e.g. white Gaussian noise, red noise & pink noise. The acquired consequences of the introduced strategy are contrasted with the aftereffects of best in class strategies and they demonstrate the prevalence of the proposed technique. Further, another function named ‘non-linear mid function’ is additionally used to address the limitations of the current techniques, hard threshold and soft threshold [7].

One of the previous works in this field include “Analysis of Heart Sounds and Murmurs by Digital Signal Manipulation” by David Theodor Kerr Gretzingera(1996). An arrangement of virtual instruments was customized to perform errands identified with the sounds of the heart. Modules were made to recreate heart sounds, to gather heart sound signal of the patient and ECG information, to analyze the heart sound signal, to perform some calculations on the heart sounds, and to perform murmur isolation. Use of the gadget with simulated and real analyzes data uncovered that the numerous functions programmed into the instrument functioned up to the mark. Exposure to the clinical conditions assisted the analysis of the points of interest of program operation. The aftereffects of these tests uncovered agreeable execution, particularly in the zone of manipulation and show of transduced physiological information. This was aimed to help in the assessment of the source & type of the murmurs present in heart [8].

A ‘novel’ de-noising strategy was introduced in 2009 for cardiac sound accumulation by utilizing enhanced thresholding technique in the basic wavelet. The proposed thresholding technique can produce distinctive outcomes just by modifying the estimations of changeable parameters. H. Sava et al. (1997) showed the practicality of a versatile time-frequency examination, the coordinating interest strategy, to identify heart cycles of the phonocardiogram.

De-noising algorithms have been introduced in time-domain also. These are based on ordinary filters, for example,

- Chebyshev IIR filters
- Active noise control i.e. ANC
- Autocorrelation technique

The standard filters are constrained to stifle the noise that is not in the frequency band of the acquired signal elements. Then again, algorithms based on ANC, for example, LMS, stifle the noise in a versatile way and, subsequently, suppress the noise in the band too. A significant disadvantage of the ANC based techniques is that they require a signal to refer, which isn't accessible in case of ideal situations. Manikandan and Soman [10] worked on a productive de-noising method which was based on lag-1 autocorrelation technique. In any case, execution of these calculations essentially debases as the strength of noise in the acquired signal increases.

For the algorithms which are frequency based, the acquired signal needs to be converted into frequency domain first from the given time domain signal. This work is done using a particular transform functions like Fourier transform (FFT) & wavelet transform (WT). After conversion, the resulted transformed signal is further analyzed and processed. Noise removal can be more efficient in case of frequency domain as compared to time domain because spectral characteristics of the elements of signal can be analyzed in frequency domain. Saneiet proposed a method to discard murmurs from the heart sound signals using singular spectrum analysis. Patidar and Pachori introduced an algorithm to discard murmurs with the help of constrained tunable-Q wavelet transform (TQWT). However, computational time of both the proposed algorithms is very high.

The most commonly used technique for de-noising the heart sound is based on discrete wavelet transform i.e. DWT. This is because in case of DWT, the obtained coefficients of the heart sound signal elements have larger value and they are confined to a particular frequency band, whereas the coefficients for the elements of noise has lower amplitude value & confined in separate frequency band. Hence, de-noising can be attained by repressing the lower value coefficients of noise.

Still, the implementation of Discrete Wavelet Transform based de-noising algorithms rely greatly on the selection of different parameter values:

1) The type of mother wavelet
2) The number of levels in which signal is decomposed
3) The calculated Threshold
4) The function used for Thresholding.

To achieve effective signal de-noising, orthogonal mother wavelet is good choice. Orthogonal wavelet allows ideal reconstruction of the heart sound signal. The second parameter is the number of levels in which the acquired signal is being decomposed. The selection of decay levels should be accurate so that the useful elements of the signal are retained & useless elements can be discarded. Therefore these elements must be present in separate levels. After the decomposition of heart sound signal, the levels to be processed should be selected suitably.

The next parameter, which is the threshold value, performs a crucial role in DWT de-noising. A small value of threshold will be ineffective to subdue the unwanted noise elements while a large threshold value influences the important signal elements. For the analysis of heart sound signal, most commonly used threshold estimation techniques are ‘rigrsure’, ‘heursure’, ‘sqtwolog’, and ‘minimaxi’. The method 'sqtwolog' is a fixed type of technique & it does not consider the substance of signal and solely rely upon the length of the acquired signal to be processed. It gives a threshold value bigger as compared to available strategies & henceforth it might leads to overthresholding. 'minimaxi' is likewise a fixed form technique, in which the threshold value is calculated with the end goal that the most extreme risk of estimation error is limited. 'rigrsure' technique calculates a threshold value that satisfy the ‘Stein's Unbiased Risk Estimation’ i.e. SURE. The remaining two functions i.e. 'rigrsure' & 'minimaxi' strategies calculates the value of threshold to reduce the risk estimation. Hence they produces comparatively smaller values of threshold[30]. 'heursure' technique chooses one of the techniques from the 'sqtwolog' and 'rigrsure' techniques, on the basis of correlation among SURE and SNR. Naseri and Homaeinez had [11] proposed a method for threshold calculation which is based on the change of level of noise in the signal while Kumar and Saha [12] estimated the value of threshold value as the 20% of the greatest vitality of the coefficient value.

The last factor is the threshold function used. Its value characterizes the best approach to manage the coefficients utilizing the calculated wavelet threshold value. The standard current thresholding functions are soft & hard thresholding functions. These are utilized widely for the de-noising of heart sound signals. The coefficients having lower value than the threshold are reduced to zero & higher coefficients are reduced by the calculated value of threshold, in case of soft thresholding function. In case of hard thresholding technique, the coefficients with lower value than the value of threshold are supplanted by zeros, while higher coefficients stay unaltered.

In order to resolve the issues involved in above mentioned methods which are related to threshold calculation and thresholding function, a new Discrete Wavelet Transform based de-noising algorithm has been implemented for the analysis of heart sound signals. The implemented work uses 75th percentile value of the wavelet coefficients which are first sorted and then the value is extracted. 75th value is taken in place of standard 50th percentile value i.e. the median. Threshold functions are also defined. The signal is reconstructed using 4th and 5th detail levels. Other levels are reduced to zero. Mother wavelet used in the method is coif5.

Moreover, to resolve the shortcomings of thresholding function, a method called as ‘non-linear mid’ function is used for the analysis of heart sound signal. Further, its parameters are enhanced to refine its execution for heart sound signals. Thus, this method adaptively reduces the wavelet coefficients obtained from the acquired signal.

Chapter-3

SYSTEM DEVELOPMENT

Heart Sound Signal Processing

Wavelet Transform: Wavelets enable the user to perform multi-resolution investigation, which accomplishes both time as well as frequency localization. Wavelet based algorithms can process information at various scales or resolutions. When the signal is analyzed in a large window, we would see net features[33]. We would see point by point features if the signal is viewed in small window. Therefore, by utilizing changing resolution, the strategy not just acquires the great qualities of the fourier transform but compensates for its shortcomings. The way of wavelet's interpretation & companding potential empowers the wavelet to have adaptable and changeable time-frequency windows which limit or reduces at high frequencies & widen at lower values of frequencies, making it accessible to center on any values of object parameters and flawlessly appropriate unstable heart sounds. Therefore, the multi-resolution investigation property has great qualities & benefits in both space and frequency domains. These days, wavelet examination has fruitful uses in biomedical research and engineering, signal analysis, image processing, sound & picture coding, speech analysis & synthesis, multi-scale edge discovery & recreation, digital TV & also other different fields.

Signal Decomposition Using Wavelet Transform

A pair of basic high pass filter & low pass filter is required for the execution of the scaling function & the wavelet function. If the response of filters are characterized as 'h(n)' in case of LPF & 'g(n)' in case of HPF, then the signal processing using DWT can be given as in Fig (2.1). 'Dyadic decomposition' is another name for this type of decomposition. The frequency spectrum is divided into 2 equal parts (LP & HP) in first stage or level. After that, the low pass band is again divided into one low pass & one high pass band in level two or stage two of decomposition. Level two therefore divides the lower half of the signal into quarter and so on[13].

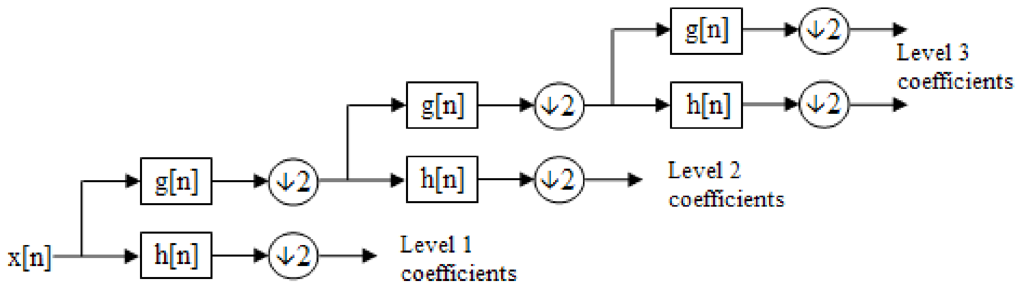

Fig 3.1: Signal decomposition using DWT [25]

Here, x[n]= signal to be decomposed or de-noised

h(n) = the impulse output of LPF

g(n) = the impulse output of HPF

'2' indicates down sampling by factor 2.

The idea of Discrete Wavelet Transform based de-noising technique relies on the detail that the elements of the signal intersect into few bigger value coefficients, and the noise remains or reside in small coefficients[14]. Therefore, the de-noising can be applied by repressing the coefficients with smaller value compared to a calculated threshold value[14]. Discrete wavelet transform is implemented in steps:

1. Multilevel signal decomposition

First step in this process is to obtain approximation and detailed coefficients. This is done by performing multilevel decomposition of the signal. Discrete wavelet transform (DWT) splits up the acquired signal into a LP sub-band, also known as 'approximation level' & a HP sub-band, also known as 'detailed level'. The approximation level can be decomposed into multiple levels for fine scale analysis. The first level detailed coefficients capture the high frequencies of the signal. If a signal with 1 KHz sampling frequency is decomposed into 5 levels using discrete wavelet transform, the frequency bands of the detailed levels (1 to 5) will be given by: 250- 50 Hz, 125-250 Hz, 62.5-125 Hz, 31.25-62.5 Hz & 15.56-31.25 Hz. The approximation level has the frequency band of 0-15.56 Hz. The coefficients can be obtained by using following formulae:

$$A_j(k) = \sum_n A_{j-1}(n)\, h(n - 2k) \quad (1)$$

$$D_j(k) = \sum_n A_{j-1}(n)\, g(n - 2k) \quad (2)$$

When j=1 i.e. level one, the approximation ($A0$) is the input signal. After that, the coefficients of further levels can be calculated by applying the filters on the down-sampled coefficients, as given in Fig. 3.1. Here, in this technique, the PCG is fragmented up to 5 levels using *'coif5'* as mother wavelet. 'coif5' gives better analytical results for the cardiac sound processing[14]. Standard frequency range of the basic heart sound elements, lub and dub, lie in between 30-100 Hz range[15]. Hence, 4*th* & 5^{th}detailed levels have the most of the power of the acquired PCG signal in common real-life cases. Thus, in the implemented technique, only fourth and fifth levels are used & rest are reduced to zero.

2. Thresholding technique[7]

In this step, as proposed by P. K. Jain and A. K. Tiwari in "An Adaptive Method for Shrinking of Wavelet Coefficients for Phonocardiogram De-noising", 2016; the coefficients of the chosen fourth and fifth detailed levels are threshold. The technique for the calculation of value of threshold & the thresholding function are:

(a) Estimating Threshold: In order to calculate threshold value, *Med*75 is calculated. *Med*75 is the 75*th* percentile value of the wavelet coefficients sorted in order. This value is taken in place of usual 50*th* percentile value i.e. median value. This is done because of the fact that the addition of time span duration of S1 (lasts for 110ms) & S2 (lasts for 90ms) exists for less than one-fourth of the total time for which the cardiac cycle exists (which is approximately 850ms) [15], [16]. Therefore, the coefficient at 75*th* percentile position will give the strength of disturbance or noise in the signal. The value of threshold is estimated using:

$$T = \begin{cases} Med_{75} & \text{if } (Med_{75} < m) \\ Med_{75} + 2\,(Med_{75} - m) & \text{otherwise} \end{cases} \quad (3)$$

where, $m = mean|Dj|$, and $j = 4, 5$. In case the *Med*75 is less than *m*, it means the PCG is less noisy & thus the value of threshold should be low. Hence, it is scaled to *Med*75 coefficient value that increases with the rise of noise level in the signal. For another case, when level of noise is high, the *Med*75 value becomes higher than *m* & the gap between m and Med75 increases as the noise level rises. Hence, the value of threshold is estimated as the summation of *Med*75 & two times the difference between the value of *Med*75 & *m*. Therefore, this technique successfully finds out the value of threshold[7]

(b) Thresholding function: Thresholding function provides a method in which the coefficients are treated or manipulated according to the calculated threshold value in previous step. Two functions are proposed in the paper:

- Mid function: This function utilizes two different values of threshold which is calculated as $T1 = T$, & $T2 = 2\times T$.

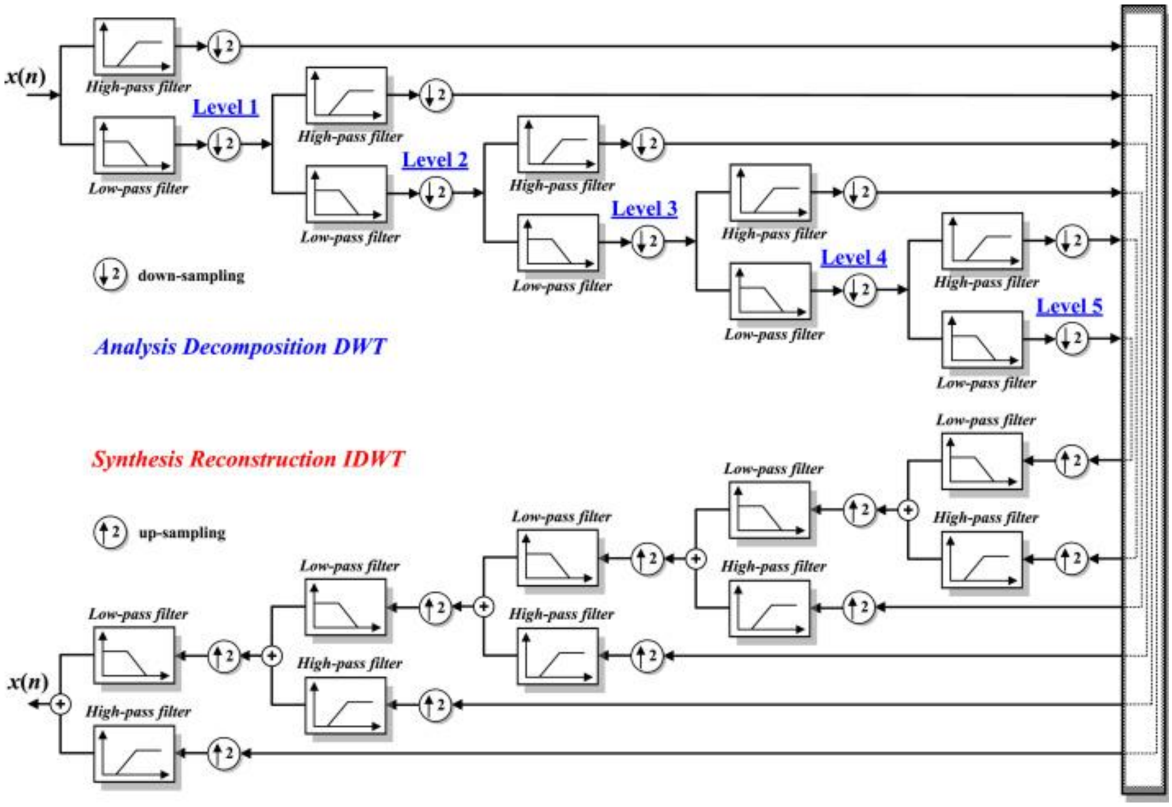

Fig 3.2: The implemented DWT based de-noising algorithm [25]

After that, the coefficients are handled in three ways:

1) preserve the coefficients with large values
2) discard the coefficients with small values
3) linear shrinking of middle coefficients as given:[7]

$$D_j^T[n] = \begin{cases} D_j[n] & \text{if } |D_j[n]| > T_2 \\ \text{sign}\,(D_j[n])\,(|D_j[n] - T_1) & \text{if } T_1 \leq |D_j[k]| \leq T_2 \\ 0 & \text{if } |D_j[n]| < T_1 \end{cases} \tag{4}$$

where, *Dj* is detailed level coefficient vector of the wavelet. Therefore this method overcomes the shortcomings of soft threshold function by preserving larger coefficients. And also overcomes the issue of discontinuities of hard threshold method.

- Non-linear mid function: In order to improve the continuity of ‘mid’ function, non-linear behaviour is included in the method.

$$D_j^T[n] = \begin{cases} D_j[n] & \text{if } |D_j[n]| < T_2 \\ D_j[n]^{po} / \; T_2^{po-1} & \text{if } T_1 \leq |D_j[n]| \leq T_2 \\ 0 & \text{if } |D_j[n]| < T_1 \end{cases} \tag{5}$$

Where power po is:

$$po = \begin{cases} 17 & \text{if } (Med_{75} < m) \\ 9 & \text{otherwise} \end{cases} \tag{6}$$

3. Signal Reconstruction

Here in this final step, the thresholded values of the coefficients are used to reconstruct the PCG signal. The reconstructed signal is the required de-noised result, as shown in Fig. 2. The final resultant signal is obtained using the approximation and detailed coefficients as given here[18]:

$$x(n) = \sum_k A_j(k)\, \tilde{H}\,(k - 2n) + \sum_{j=1}^{J} \sum_k D_j(k)\, \tilde{G}\,(k - 2n) \tag{7}$$

here, $\tilde{H}$ and $\tilde{G}$ are LPF & HPF respectively & they are known as ‘synthesis filters’. Synthesis filters are formed using ‘analysis filters’[19][7].

Other methods involved

Thresholding technique	Explaination
rigrsure	Uses SURE principle
minimaxi	Utilizes a fixed value of threshold
sqtwolog	sqrt(2*log(length(x)))
Adaptive thresholding method	Specific formulae used

Table 3.1: thresholding methods

There are other available de-noising methods and functions which are involved in the project for comparisons. These are:

- **rigrsure** - This method uses the principle of Stein's Unbiased Risk Estimate i.e. SURE and obtain an adaptive threshold value. Rigrsure uses a rule for threshold selection that uses Stein's Unbiased Estimate of Risk basis for soft threshold value estimator which is basically a quadratic loss function. The estimation of risk is done for the possible threshold value. Reducing the risks in possible threshold value results in selection of the threshold value.
- **heursure** - The first option 'heursure' has a variant which is 'heuristic variant'. It is a combination of previous two options. Therefore, if the value of SNR is very small, the SURE estimate is quite noisy. Fixed form threshold is used in case of such conditions.
- **sqtwolog** – The value of threshold is calculated using: sqrt(2*log(length(x))). 'sqtwolog' method also uses a fixed form threshold generating results as in minimax multiplied by a small factor corresponding to 'log(length(X))'.
- **minimaxi** – The 'minimaxi' method utilizes a fixed value of threshold selected to obtain minimax results for MSE against an ideal procedure. Statistics are used to find the estimators.

'sln' - rescaling done using a single approximation. Estimation is based on level one coefficients.

'mln' - rescaling done using approximation of level noise which is level-dependent

Hard & soft threshold

Hard threshold	Soft threshold
value less than or equal to that of threshold value reduced to zero	value less than or equal to that of threshold value reduced to zero
higher coefficients stay unaltered	threshold value is subtracted from higher coefficient value

Table 3.2: threshold functions

Chapter-4

PERFORMANCE ANALYSIS

The method is implemented using the threshold functions introduced the paper. The results are obtained using MATLAB software. The signal is decomposed using discrete wavelet transform and the detail level coefficients and approximation level coefficients are obtained. Decomposition is done till 5 detail levels and one approximation. Then threshold value is obtained for each level to be used in reconstruction. The levels to be used in reconstruction are level 4 and level 5. Others are forced to zero. The threshold value is obtained as explained above. The coefficients are then threshold using threshold function. After this, the signal is reconstructed.

The adaptive threshold method gives better results as compared to other built in MATLAB functions. The results will also be obtained for samples captured in real-life scenarios. Non-linear mid function is used in the experiment which is found better as compared to soft and hard thresholding techniques. The results obtained using various methods are observed and compared. Methods that are involved are rigrsure, minimaxi, sqtwolog and the adaptive wavelet coefficient shrinking method. Most commonly used techniques for de-noising are rigrsure, minimaxi, sqtwolog, heursure etc. The plots are obtained for these techniques using MATLAB.

De-noising results using the method

The signal is Decomposed into 5 detail levels and one approximation level. The first level detailed coefficients capture the high frequencies of the signal. For instance, If a signal with 1 KHz sampling frequency is decomposed into 5 levels using discrete wavelet transform, the frequency bands of the detailed levels (1 to 5) will be given by: 250- 50 Hz, 125-250 Hz, 62.5-125 Hz, 31.25-62.5 Hz & 15.56-31.25 Hz. The approximation level has the frequency band of 0-15.56 Hz. Most of the noise is of high frequency and hence is captured in first, second and third levels. The de-noised resultant signal is obtained after reconstruction.

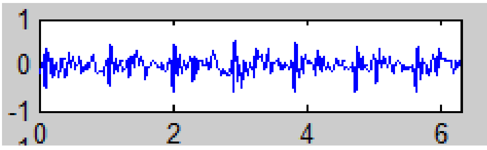

Fig 4.1: Noisy signal

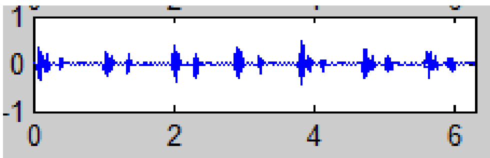

Fig 4.2: de-noised signal

The experiments are performed using MATLAB. Figure shows plot for noisy signal and figure shows the de-noised result of the signal. The technique is applied on PCG signal contaminated with different type of noises. The obtained results are observed and compared. The noise has been eradicated significantly. The SNR ratios are also obtained and compared. The SNR values are higher for the adaptive thresholding method as compared to other methods. Better values are obtained in case of 'sqtwolog' with hard thresholding as compared to other two. The results obtained in all the cases are given in figure 4.7.

Discrete wavelet transform de-noise the input signal by decomposing it into number of levels and then reconstructing it by combining suitable levels that produce best results. Many methods are present to de-noise the signal using DWT. All levels in detail are given in following figure:

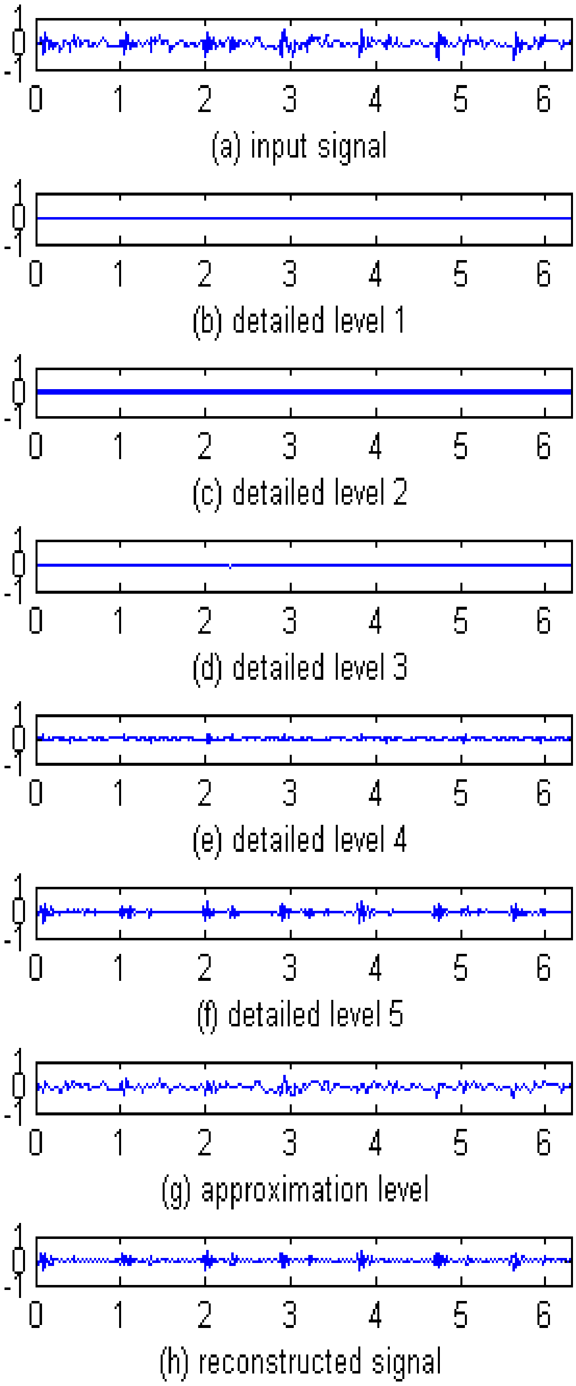

Fig. 4.3 different DWT levels in detail

De-noising results when signal is contaminated with red noise

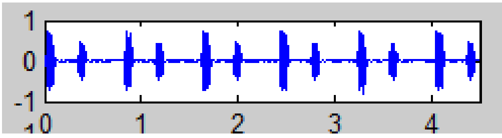

Fig 4.4: Signal

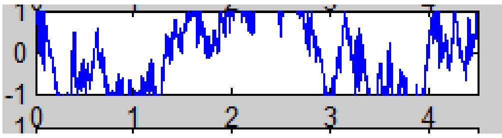

Fig 4.5: Contaminated signal

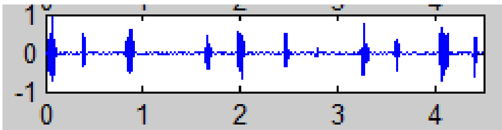

Fig 4.6:De-noised signal

Fig 4.4 shows heart sound signal which was then contaminated with noise. The signal contaminated with the noise is plotted which is given in figure 4.5. The signal is contaminated with red noise. The obtained result is shown in figure. The obtained results are obtained using the adaptive thresholding method.

- The signal is first decomposed and then the threshold value is obtained for each level.
- The threshold function that is used is given in equation (5) that will decide how the coefficients are treated according the threshold values.
- Fourth and fifth detail levels are used to reconstruct the signal. Other levels are reduced to zero.

The results show that the method suppresses the noise in the signal significantly. The SNR values and the plots shows the efficiency of various methods.[7]

Comparison of various de-noising methods

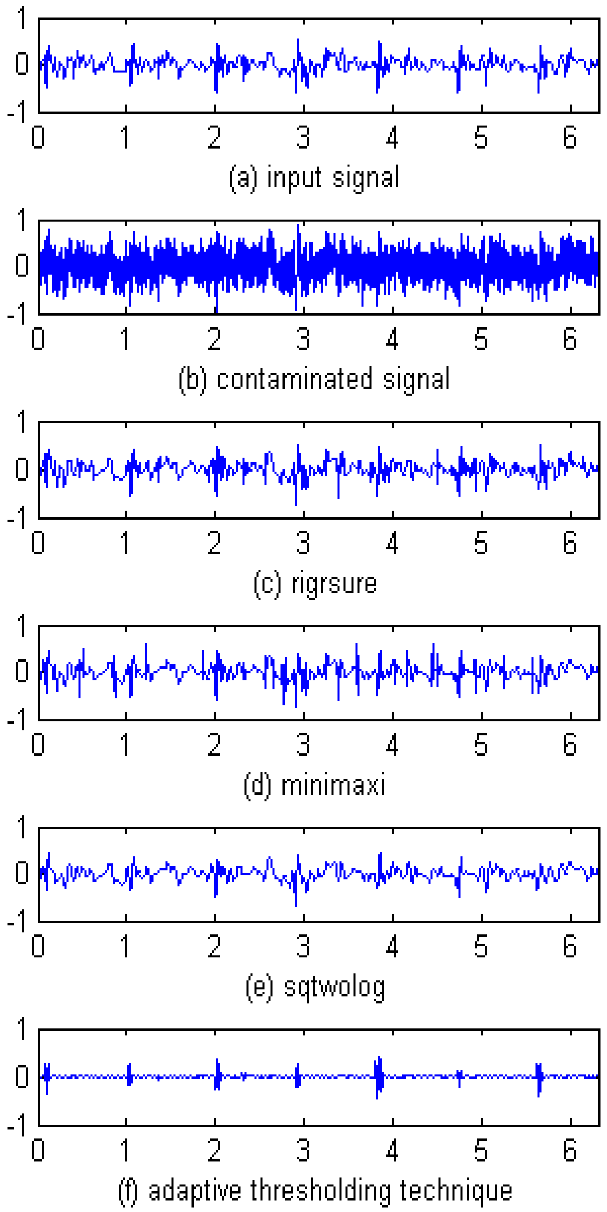

Fig 4.7: signal to be processed & denoising techniques (a) signal; (b) signal plus noise (c) rigrsure; (d) minimaxi; (e) sqtwolog; (f) adaptive method for threshold shrinking

With another input signal, the results obtained with different techniques

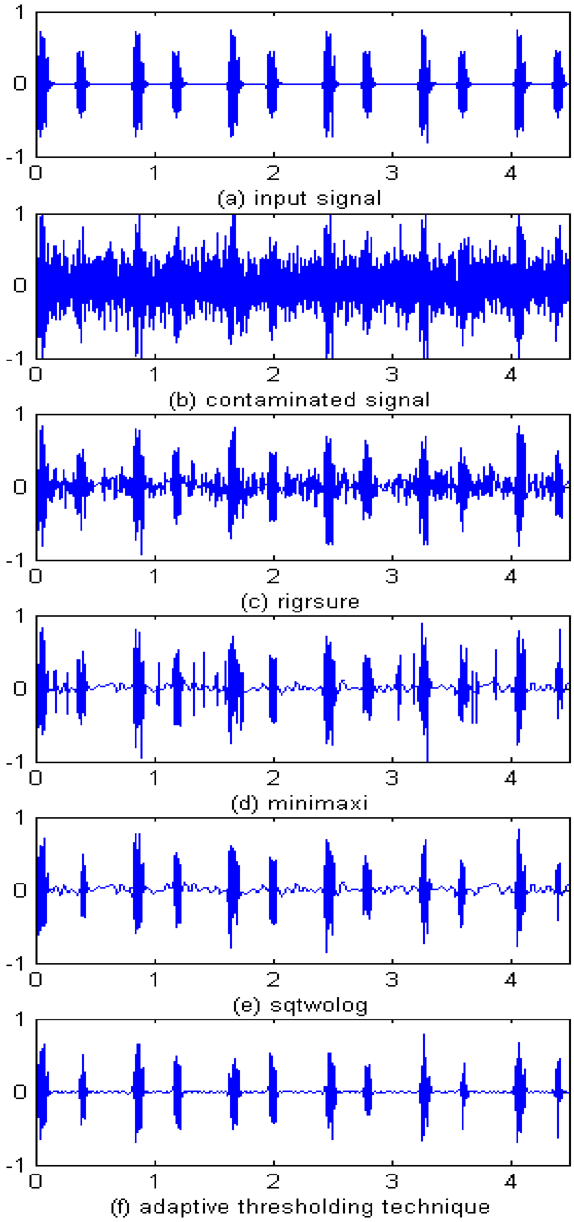

Fig 4.8: signal to be processed & denoising techniques (a) signal; (b) signal plus noise (c) rigrsure; (d) minimaxi; (e) sqtwolog; (f) adaptive method for threshold shrinking

The performance of the adaptive thresholding technique is compared with the performance of traditional methods. In literature, various techniques and algorithms for threshold estimation have been introduced which are used for the PCG signal analysis. The methods to which it is compared are rigrsure, minimaxi and sqtwolog. These methods are most widely used methods for signal de-noising. The threshold value has been obtained using these methods and the signal is then processed using hard thresholding in each case. Rescaling is done using ‘mln’ in each case. Different plots are obtained in each case and then compared. The signal is contaminated with noise and then is de-noised using each of the methods. The original signal & contaminated signal are as presented in Fig 4.6 and 4.7 for different input signals. The contaminated signal is then processed using the methods and the plots are obtained for the same. The plots obtained in each case are given in Fig 4.6 & 4.7 for different inputs.

Comparing the techniques with each other, the best results are obtained using adaptive thresholding and sqtwolog. The noise has been eradicated significantly from the contaminated signal using these techniques producing the best results among all the methods.

Chapter-5

CONCLUSIONS

In this report an attempt is made to study and analyze the characteristic features of PCG for detection of various heart diseases. The algorithms in this report are time efficient, simple and require only PCG as input signal. The following points are concluded during this study:

- For murmur detection SNR values for different techniques are observed and compared.
- The algorithm for murmur detection is useful to detect mainly the valve-related diseases and other congenital abnormalities.
- The algorithm can also detect arrhythmia and can further classify it into Tachycardia and Bradycardia.
- Our experiments show the performance of different DWT methods used for PCG de-noising with different values of function parameters. Adaptive thresholding technique gives best results. In case of built-in traditional functions, sqtwolog with hard threshold produces best results.

Future scope,

- The technique can also be executed using the smart mobile phones with the applications which can work as electronic stethoscope or phonocardiogram.
- The method can be very easily executed using electronic stethoscope in very cost effective way by interfacing it with the present embedded technology.
- The accuracy of the presented algorithm can be further increased by incorporating Artificial Intelligence techniques or other hybrid classifiers on a larger dataset.

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APPENDIX

Thresholding of coefficients in case of wavelet

Application of thresholding algorithm to the coefficients uses different thresholding methods. Two methods are currently used:

1. **Hard threshold** – Any value less than or equal to that of threshold value is forced to zero.

```
if (coef_value[i] <= thresh)
coef_value[i] = 0.0;
```

2. **Soft thresholding -** Any value less than or equal to that of threshold value is forced to zero. The threshold value is subtracted otherwise from the coefficient value.

```
if (coef_vakue[i] <= thresh)
  coef_value[i] = 0;
 else
  coef_value[i] = coef_value[i] - thresh;
```

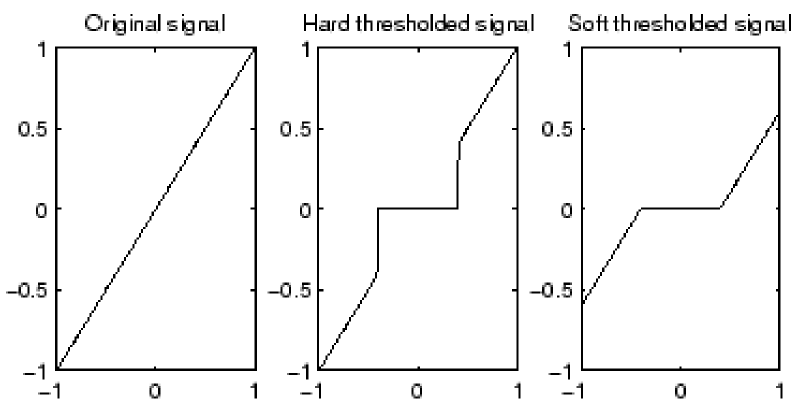

Figure: Threshold functions' working[27]

Threshold according to selected method

Working of thselect: let's assume w is a white noise added to a signal $N(0,1)$. The values are obtained as follows fir different methods:

w = randn(1,1000); (random white noise)

thresh = thselect(w,'rigrsure') / * hselect using rigrsure */

thresh =

2.073

thresh = thselect(w,'sqtwolog') / * hselect using sqtwolog */

thresh =

3.716

thresh = thselect(w,'heursure') / * hselect using heursure */

thresh =

3.716

thresh = thselect(w,'minimaxi') / * hselect using minimaxi */

thresh =

2.216

Universal Threshold: It's calculated as $\boldsymbol{T = \sigma\sqrt{(2\log 2\ n)}}$ where n = size of the sample. The threshold give is an ideal context along with arbitrary Gaussian disturbance for soft thresholding. This plan is not hard to execute, but it provides a threshold level greater than that of other decision criteria, and hence provides results as smoother renovated data.

Minimax Threshold: It is determined as $T = \sigma\ Tn$ and here, Tn is calculated by a minimax rule, in such a way that the maximum negative outcome of estimation failure across all points of the data is minimized.