PHONOCARDIOGRAM SIGNAL COMPRESSION

Project report submitted in partial fulfilment of the requirement for the degree of

BACHELOR OF TECHNOLOGY
IN
COMPUTER SCIENCE AND ENGINEERING

By

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To

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Declaration by the Scholar

I hereby declare that the work presented in this report entitled **"PHONOCARDIOGRAM SIGNAL COMPRESSION"** in partial fulfilment of the requirements for the award of the degree of **Bachelor of Technology** in **Computer Science and Engineering** submitted in the department of Computer Science & Engineering**,** Jaypee University of Information Technology Waknaghat, Solan is an authentic record of my own carried out under the supervision of **Dr. Rajinder Sandhu**. We have not submitted this work elsewhere for any other degree or diploma.

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Certificate

This is to certify that the work reported in the B.tech project report entitled **"Phonocardiogram Signal Compression"** which is being submitted by **Abhimanyu Kapil (141265)** in fulfilment for the award of Bachelor of Technology in Computer science and Engineering by the Jaypee University of Information Technology, is the record of candidate's own work carried out by him/her under my supervision. This work is original and has not been submitted partially or fully anywhere else for any other degree or diploma.

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Acknowledgement

Behind every achievement lies an unfathomable sea of gratitude of those who energized us, without whom it would never have come into existence.

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Abhimanyu Kapil (141265)

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List of Abbreviations:

HSS – Heart sound signal

HS - Heart sound.

RLE – Run Length Encoding

CR - Compression Ratio

MSE- Mean Square Error

Chapter 1 Introduction

1.1 Introduction

In some recent years, there is some increase in interest shown towards auscultation of human heart which was lost in some recent decades favouring some the expensive and accurate techniques which give a better result. The analysis of PCG (phonocardiogram) is discovered again as a very fast, useful and also is a low-priced technique through which we can diagnose any kind valvar disease. Like in any of the application of telemedicine, an uninterrupted monitoring requires an efficient transmission of data.

Since, Phonocardiogram is graphical illustration of sounds and the murmurs in heart, more common techniques of audio compression can be used. However, these techniques focus more on speech or much, and because of this these techniques have a low performance on PCG compression.

1.1.1 Phonocardiography - Definition

Phonocardiography (PCG) is a Diagnostic technique which creates a graphical record (also known as phonocardiogram) of sounds and murmurs which are produced in the heart, these includes sounds produced by its valves and great vessels.

1.1.2 Cardiac Anatomy

The human heart which is a very strong muscle and whose function is to pump blood throughout the circulatory system in the human body. An electric pulse is generated by the sinoatrial node which is the reason for the contraction of the heart. When the human heart contracts, blood in the body firstly flows through valves of atria and then to ventricles and lastly it is supplied to rest of the body. There are four chambers in each heart, two are upper and other two are lower. Left and right atria are situated in the upper chamber of the heart, and left ventricle and right ventricle are situated in the lower chamber of the heart. There are phases in each heart which are systolic phase and diastolic phase. Systole phase includes the contraction of heart and due to this contraction, blood in the heart leaves the heart through arteries. And the diastole phase includes the relaxation of heart and due to this relaxation blood enters the ventricles of the heart. Through the ventricles the blood is passed to the lungs (to oxygenate the blood). In other words, during systole blood is transferred to the other parts of body. And during diastole blood is transferred from auricles of the heart to ventricles. There are four valves in every human heart. These valves provide a passage between the two chambers of the heart. These four valves can be broadly divided in two categories. The categories are atrioventricular valves and semilunar valves. The atrioventricular valves include tricuspid valve and mitral valve. The semilunar valves include pulmonary valve and aortic valve. The main purpose of these valves is to prevents the backward flow of blood. The tricuspid valve is present between the right atrium and right ventricle and passes deoxygenated blood between them. The pulmonic valve is present between ventricle and lungs and passes the blood to the lungs to oxygenate. The mitral valve is between the left atrium and left ventricle and it passes oxygenated blood between them. The tricuspid valve passes the oxygenated blood into aorta.

1.1.3 Generation of heart sound

The heart sounds are sounds or noises which are generated because of the flow of blood between the valves within the heart. Also, when the heart valve closes suddenly, these noises shows the disturbances created. At every location of auscultation, it is very important to listen to each of the component generated by the cardiac cycle. A healthy person has two types of heart sounds in each heartbeat. These two heart sounds are known as lub (in scientific terminology known as S1) and dub (in scientific terminology known as S2). S1 and S2 are first and second heart sounds respectively. As discussed above these sounds are the result of the closure or opening of the heart valves. But in some of the abnormal cases, the heartbeat may contain some extra sounds which are murmurs, gallops and S3, S4.

The murmurs in the heart are some of the extra sounds which are caused because of disturbances which are created in the heart. These disturbances are caused because of some kind of abnormal activities of the valves. Some murmurs are known as Pathological murmurs and these occur because of narrowed valves (stenosed) or back flow caused by some non-functioning valve (regurgitant) or because of some septal defect. Usually a normal heart sound will consist of only two thuds(sounds) which are S1 and S2 also known as primary components. The systole is the time interval from S1 to S2. The period between the systole is known as systolic period. While, diastole is the time interval from S2 to the next S1. The period between a diastole is known as diastolic period.

S1 occurs when both the tricuspid and mitral valve closes at the end of each diastole. And S2 occurs when aortic and pulmonary valve closes at end of each systole. Beside these primary components, there are two sounds that also occurs in some kind of special situations in addition to S1 and S2. These extra components are known as S3 and S4. S3 heart sound is an innocent heart murmur while S4 occurs because of some pathology. Due to some of the pathologies, there occurs some of the additional sound in the heart sounds signal. These sounds occur because of some problems within the valvar heart. Many of the valvar heart disease occur because of both regurgitation and stenosis. In stenosis the opening of a valve becomes small and due to this the pressure in the chamber connected to the valve is increased hugely. Because if this increased pressure in the chamber the valve becomes more and more stiffer and the tissues of the valve becomes slow and are not able to move as quick as before. This results in the reduction of the amount of the blood that flow through that valve. In regurgitation, the tissues of the valve are not able to close fully. This leads for blood to move in the backward direction. This backward flow of blood because of regurgitant is known as regurgitant flow.

S1 (First heart sound):

The lub sound which is produced when the Mitral and Tricuspid valve closes. This sound occurs when there is a change from diastole phase to systole phase.

i. S2 sound is weaker than S1.
ii. Mitral valve is a major component when determining in how intense is S1.

So, at the Apex region the S1 sound is best heart.

S2 (Second heart sound):

The dub sound which is produce when the Aortic and the Pulmonic valve closes. This sound occurs when there is a change from systole phase to diastole phase.

i. The pitch of this sound is very high.
ii. Because of the contraction of the time difference between the aortic valve and the pulmonary valve there is a split in this sound.

So, at the upper-sternal border S2 sound is best heard.

The below Figure 1-1 shows S1 and S2 signals

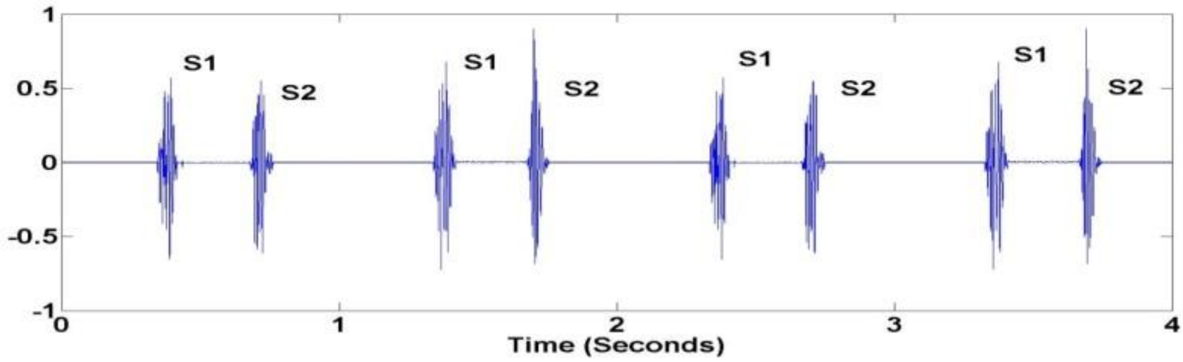

Figure 1-1 Healty heart PCG signal

S3 (Third heart sound):

The main cause for S3 sound is because of unexpected deacceleration of the blood it makes some attempts to fill the ventricle. This leads in walls of ventricle to vibrate which sounds like a disease. Usually S3 is a very low-pitched sound. This occurs at the start of diastolic phase. The S3 sound are usually innocent but these sounds can be due to a disease. The S3 sound which occurs in young people, in some women in pregnancy and also in some athletes is known as innocent murmur.

Auscultation finding:

These sounds can be heard very clearly at apex region while the patient is lying in the left lateral position and also in decubitus position.

Below Figure 1-2 shows PCG signal of S3

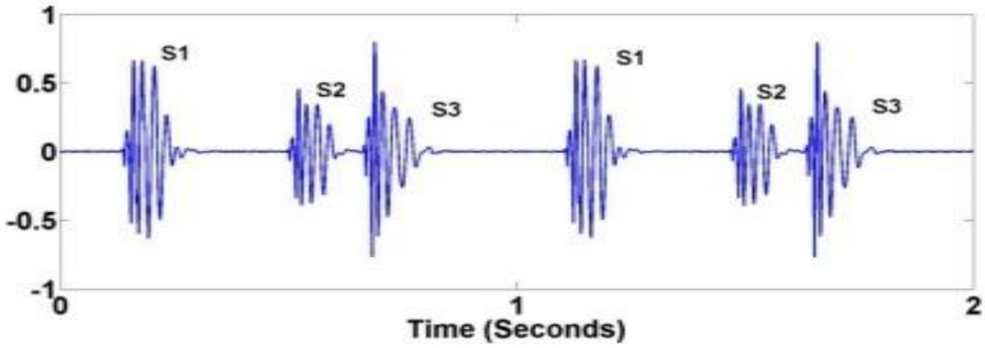

Figure 1-2 S3 PCG Signal

S4 (Fourth heart sound):

S4 sound is a low-pitched sound that occurs due to the swollen left ventricle. When blood strikes this swollen left ventricle, it produces a sound which is S4. It usually occurs at the end diastolic phase or earlier to systolic phase. It is always because of a disease. This occurs because of the atrial contractions which leads to a rigid and swollen ventricle.

Auscultation finding:

These sounds can be best heard very clearly at the apex region while the patient is lying in the left lateral position or in decubitus position while he holds his breath.

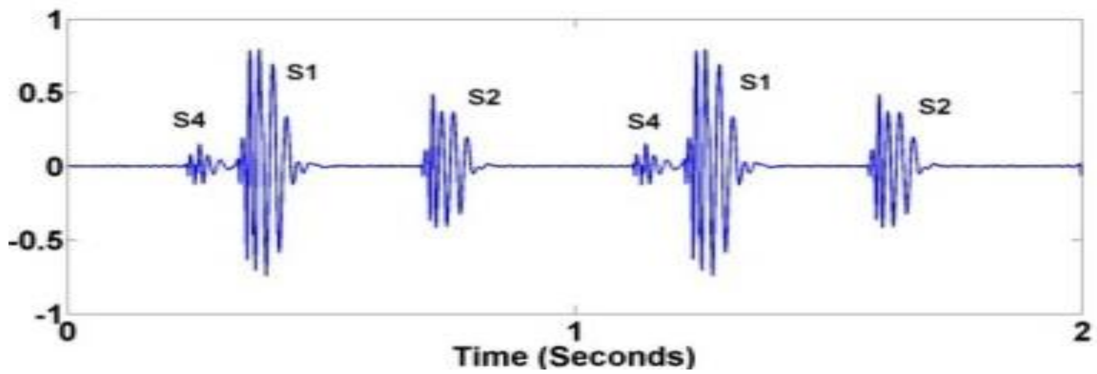

Figure 1-3 S4 PCG signal

Murmurs: -

a) Aortic Stenosis: -

This occurs because of the narrowing of the aortic valve. And because of this narrowing the muscles of the left ventricle become broader and broader. Because of this thickening there is a increase in the pressure in the ventricle. Aortic Stenosis can leads to angina which are followed by many other diseases like syncope, dyspnoea and pulses tardus.

Causes:

Unicom or the bicuspid valve instead of the tricuspid valve, Calcification, S2 occurs because of the reverse split.

Auscultation findings:

Crescendo-decrescendo murmur is the type of murmur which is heard. The best place to hear this sound is 2nd Right Intercostal Space. Apex region is the best area to hear this sound. In some situations, due to the stiffness of left ventricle S4 sound can also be heard. But this S4 is of very low intensity. This sound is heard only when the Atrial Fibrillation does not occur.

b) Mitral Stenosis

This occurs because of the narrowing of the mitral valve which increases the pressure on the pulmonary capillary. Mitral value is placed in-between the atrium and ventricle on the left side of the heart. Due to gradual closing of the valve and the greater pressure of the left artery, the pressure on the wedge of the pulmonary capillary increases. Mitral Stenosis is a fatal disease, average survival time of this disease is only three years.

Causes:

One of the major cause of this disease is inflammation of the tissues.

Some other causes include amyloidosis, hereditary, carcinoid and myxoma of left artery.

Auscultation findings:

A sound while the transfer of blood to pulmonary capillary. A murmur in the middle of diastolic phase. S1 in this in very loud. Lastly a S3 sound at the right ventricular.

c) Mitral Regurgitation

Mitral regurgitation occurs because of the non-functioning or slowing of the mitral valve. Mitral valve passes blood from atrium to the ventricle on the left side of the heart. If mitral valve stops functioning or becomes slow the blood will start flowing backward. This backward flow is known as Regurgitation.

i. Chronic MR
ii. Acute MR

i. Chronic Mitral Regurgitation:

Chronic MR does not show any of symptoms for a long time except for palpitation. It is the stage of development of disease. Patient with this disease will get some time for adapt.

Causes:

Inflammation of tissues, bulging of mitral valve and cleft mitral valve which is a present from birth.

ii. Acute Mitral Regurgitation:

Acute MR gives no time for the body to adapt. A sudden huge amount of blood is dumped in Left Atrium. There occurs an unexpected fall in the amount of blood pumped to the left ventricle.

Causes:

Ischemic Heart Disease and backward muscle of left ventricle.

Auscultation findings:

There is a pansystolic murmur at Apex. Murmur in the middle of systolic phase at the pulmonary area. Sound at the ejection of blood to the pulmonary capillaries. S3 sound may be present. S1 is very soft.

d) Tetralogy of Fallot:

Tetralogy is a combination of four things which can happen inside a human heart. Right Ventricular Hypertrophy, Ventricular Septal Defect, Overriding of Aorta and Pulmonary stenosis

Overriding of Aorta

The overriding of aorta is a problem in which a very large volume of blood has to pass by the aortic valve, but because of the size of aortic valve, it is unable to allow such a large volume.

Pulmonic stenosis

This narrows pulmonic valve space. This occurs when some kind of septum is formed in-between of both the ventricles. This septum permits the flow of deoxygenated blood in the RV to LV.

Ventricular Septal Defect

In pulmonic valve stenosis, a hole is formed between RV and LV. Because of this hole de-oxygenated blood present in RV mixes with oxygenated blood present in LV which is known as Ventricular Septal Defect.

Right Ventricular Hypertrophy

Because of pulmonic stenosis, the valve space becomes less and less. So, RV has to work more to supply blood through this pulmonic valve. This all makes the ventricular muscle to enlarge (hypertrophy). This is known as right ventricular hypertrophy.

Cause:

The cause for this is only when this disease is present since birth.

Auscultation finding:

The Pulmonic stenosis can be detected. We just have to look at the border of upper left sternal for the murmur during ejection in the systolic phase.

e) Aortic Regurgitation

It is a problem in which some of the blood starts to flow in the backward direction. This is caused because the valves not closed firmly. The blood flow back in the LV.

Causes:

Heart defect which are present by birth. The collapse of valve as one ages, fever caused by inflammation of fibres.

f) Tricuspid Stenosis

This problem occurs in the tricuspid region when the tricuspid valve is not able to open completely and which permits backward movement of blood. Mitral stenosis and tricuspid stenosis can occur concurrently.

Auscultation finding:

A wide split is present in the S1 sound. It is best heard at left sternal boarder. It has an opening sound and then a continuous resonant murmur in the diastolic phase.

1.2 Why do we need compression?

We can use a phonocardiogram signal (PCG), if we want a long-term record of heart monitoring. If we record for a long time like days or weeks huge amount of data is produce. So, it becomes very important task to compress a PCG signals. So, we are able to lower the storage in the play or record systems. One more situation can be that the PCG signal is sent to a distant health care centre for analysis in telemedicine. So, in these cases the compression of phonocardiogram signal is important so that we can reduce the data to be sent. Since heartbeats are repeated over an interval of time, so compression algorithm can use the similarity between these cycles in the adjoining cycles and eliminate reoccurring elements.

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1.3 Objective

For a given wavelet the compression features basis is usually linked to the relative scarceness of wavelets representation of domain for signal. The idea behind compression of a signal is that regular signal components of a signal can be approximated accurately if we use the following elements: some approximation coefficients whose levels are appropriately chosen and some detail coefficients

1.4 Methodology

1.4.1 Types of compression techniques

- Lossless: Lossless data compression is the type of algorithm which usually exploits the redundancies present in the statics so that data can be represented without losing any of the information. This type of compression is possible because there is always some statistical redundancy in real-world data.
- Lossy: Lossy data compression algorithms are opposite to lossless compression. In these schemes, loss of some information is acceptable. If we drop some of the details which are not essential from the source of data, we can save some storage space. These schemes are mainly designed for the purpose of research mainly as how the people recognise data in question.

1.4.2 Pre-processing

1.4.2.1 Filtering: Throughout the process of acquisition, there can be some of noise interference. These noise interferences can occur because on many reasons which include the environmental disturbances, noise formed by human respiratory tracts, or it can be noises by the measuring equipment. The frequencies of all these interferences will be less than 40Hz. For good detection of S1 and S2 signals we need to eliminate all these extra noises. So, to eliminate all these low frequency noises we can use some kind of filtering techniques like HPF (high pass filter) by using 40Hz as cut-off frequency.

1.4.2.2 Decimation: It has been shown by some researchers that the primary components of almost all the heart signals is between the range of 40 and 200Hz. These are due to the disturbance in the flow of the blood in valves of heart. All the murmurs which are caused because of vulvar dysfunctions are in the range of 600Hz. So, for analysis and diagnose purposes of heart signal it is not necessary to consider the frequency which is does not have a significance. We have to process the signal some more. So that only those frequencies are present which are of clinical significance. This process is implemented in decimation.

Decimation uses eighth-order low-pass Chebyshev type I filter which has a cut-off frequency equal to 0.8*(Fs/2)/r, where Fs is sampling frequency and r is decimation factor. Here 'r' is chosen is such a way to obtain sample rate of 700Hz.

It is an evident that all the important events of the PCG signal lies below 700Hz frequencies and also it is observed that no data is available at very low frequency area which depicts no important information is present in that frequency range. So, it is inevitable to filter and decimate the PCG signal before processing.

1.4.3 Decomposition of wavelets.

Decomposition of wavelets is explained in the chapter 3.

1.4.4 De-noising Signals:

The Wavelet and the wavelet packet de-noising allow us to retain features in our data that are often smoothed out or removed by some other de-noising techniques. We can compress data by setting some perceptually not important wavelets and wavelet packet coefficient equal to zero and then reconstructing the data. The procedure of de-noising generally used usually has three steps which are explained below:

- Decompose: Firstly, select a wavelet, choose level N. Calculate wavelet decomposition at level N.
- Threshold detail coefficients: We have to choose threshold and then perform thresholding (soft) to the detail coefficients, for every level between 1 and N.
- Reconstruct: Lastly, with the help of original level N approximation coefficients and level 1 to N detail coefficients which are modified we compute wavelet reconstruction.

The points which should be think about are:

- Choosing of a appropriate threshold.
- Performing the thresholding.

1.4.5 Compression

1.4.5.1 Run length coding

Run length coding is explained in chapter 3.

1.4.5.2 Huffman coding

Huffman coding is explained in chapter 3.

1.4.6 Decompression of wavelets

1.4.6.1 Run length decoding

Run length decoding is explained in chapter 3.

1.4.6.2 Huffman decoding

Huffman decoding is explained in chapter 3.

1.4.7 Reconstruction of wavelets

Reconstruction of wavelets is explained in chapter 3.

1.4.8 Generation of the original signal

Huffman decoding is explained in chapter 3.

Chapter 2 Literary Survey

NAME OF THE RESEARCH PAPER	AUTHORS AND DATE	WORK
Phonocardiogram signal compression using sound repetition and vector quantization	Hong Tang, JinhuiZhang, Jian Sun, TianshuangQiu, YongwanPark, Accepted 14 January 2016. journal homepage: *www.elsevier.com/locate/cbm*	Compression of heart sound signal using sound repetition technique.
Wavelet threshold based ECG compression using USZZQ and Huffman coding of DSM.	M. Sabarimalai Manikandan, S. Dandapat. Accepted 27 November 2006. journal homepage: *www.elsevier.com/locate/cbm*	Compression of ECG signal using wavelet thresholding and Huffman coding.
Wavelet-based electrocardiogram signal compression methods and their performances: A prospective review	M. Sabarimalai Manikandana, S. Dandapatb Accepted 6 July 2014. journal homepage: *www.elsevier.com/locate/bspc*	Wavelet based ECG compression. Review of various techniques.
Wavelet and wavelet packet compression of phonocardiograms.	J. Martı´nez-Alajarı´n and R. Ruiz-Merino. 19 May 2004. Electronics Letters online no: 20045476 doi: 10.1049/el:20045476	We have presented the first reported specific method to compress the PCG signal, based on lossy compression
ASEPTIC: Aided System for Event-Based Phonocardiographic Telediagnosis with Integrated Compression	J Mart´_nez-Alajar´_n, J L´opez-Candel, R Ruiz-Merino. ISSN 0276−6547 Computers in Cardiology 2006; 33:537−540.	Signals derived from the PCG provides enough information to discriminate and identify the events.
Compression system for the phonocardigraphic signal.	F J. Toledo-Moreo, A. Legaz- Cano, J. J. Martlinez-Alvarez J. Martı'nez-Alajarı'n, R. Ruiz-Merino. 1-4244-1060-6/07/$25.00 (C2007 IEEE.	An FPGA device has been used as hardware platform to design a Micro Blaze based solution which can compress and transmit in real time the PCG signal.
A Remote Heart Sound Monitoring System Based on LZSS Lossless Compression Algorithm.	Wei Qin and Ping Wang 978-1-4673-4933-8/13/$31.00 ©2013 IEEE	This paper discusses the heart sounds remote monitoring system based on LZSS compression algorithm.

Table 2-1 Literary Survey.

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NAME OF THE RESEARCH PAPER	AUTHORS AND DATE	WORK
Wavelet based Signal Processing for Compression a Methodology for on-line Tele Cardiology	M.SeshaGiri Rao, Dr. V S Chauhan. IOSR Journal of VLSI and Signal Processing (IOSR-JVSP) Volume 5, Issue 6, Ver. I (Nov -Dec. 2015), PP 46- 51 E-ISSN: 2319 – 4200, p-ISSN No. : 2319 – 4197 *www.iosrjournals.org*	There can be a saving of more than 200% in time in transmission and may result a compression of the order of 200% for ECG recordings of Tele Cardiology.
Book- Cardiac Physical Examination	Author-Joseph perloff	Principles of sound formation in the heart and their clinical assessment.
Book-A. Jensen and A. la Cour-Harbo. An Animated Introduction to the Discrete Wavelet Transform.	Arne Jensen Aalborg University, 2001.	This is a tutorial introduction to the discrete wavelet transform. *Ripples in Mathematics* *The Discrete Wavelet Transform*
Wavelets and signal processing.	Olivier Rioul and Martin vetterli 1053-5888/91/1000-0014$1.00©IEEE October 1991 IEEE magazine	Information about wavelets and how and why it is used in signal and image processing.
LECTURE notes: Heart Sounds. Phonocardiography	Practical Notes of the Physiology Department II, Carol Davila Univ. of Medicine and Pharmacy, Bucharest Dr. Ana-Maria Zagrean, Coordinator, 2nd Year English Module	Phonocardiography – Definition, What produces the heart sounds, Where to listen for the heart sounds, How to record a phonocardiogram, Normal heart sounds
BOOK: Advances in Wavelet Theory and Their Applications in Engineering, Physics and Technology.	Edited by Dr. Dumitru Baleanu. Publisher InTech Published online 04, April, 2012	All about wavelet transform and signal de-noising.

Table 2-2 Literary Survey continued

Chapter 3 System Development

3.1 Wavelets

- Because to the behaviour of the Heart sound signals is not stable. So, signals for many components (both normal and abnormal components of heart sounds) cannot be shown by using only one decomposition level. This is because the wavelets have the tendency to capture even change in rapid transient signals. The translation and scaling properties help the wavelet in determining these rapid changes.
- **Mother wavelet:** It is very to select the best wavelet present, because these wavelets help in analysing the coefficients structure. The wavelet we select is known as mother wavelet. We can use Daubechies or Meyer wavelet as mother wavelets

3.2 Wavelet decomposition

- A specific frequency range is carried by a particular detail signal.
- So, breaking of signal in will different ranges of frequency.

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3.3 Wavelet synthesis

- To obtain the best temporal resolution, it is very important to use each of the coefficient vectors to synthesize original heart sound signal.
- This helps us get some featured signals whose temporal resolution is high.

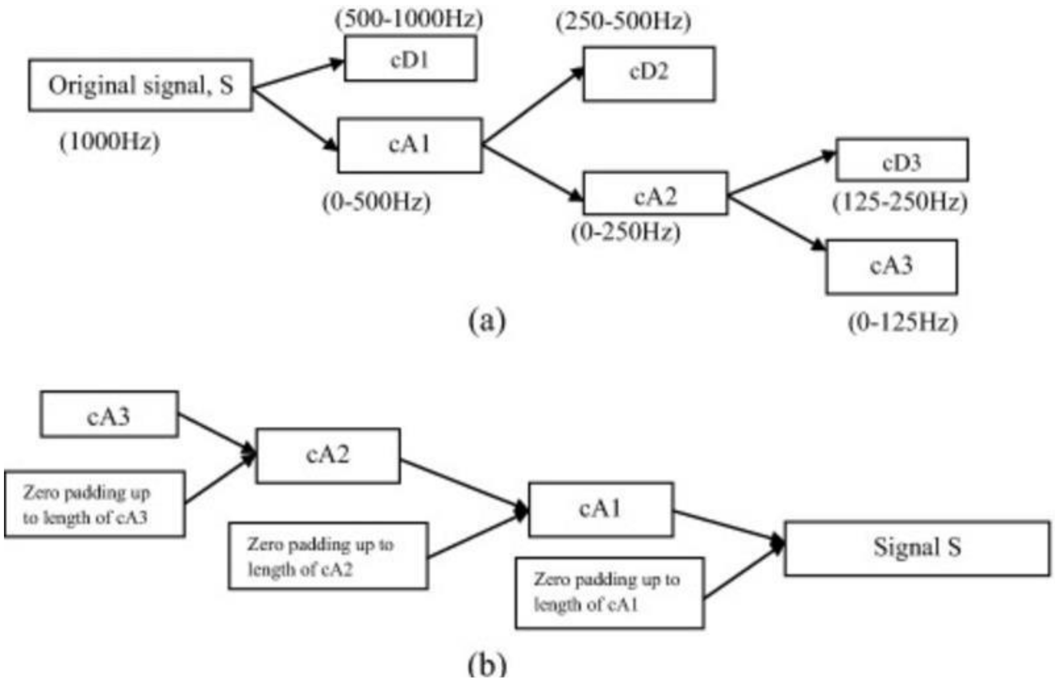

Figure 3-1 a) Decomposition b)Reconstruction

3.4 Wavelet families:

a) Haar wavelet: These resembles a step function and are not continuous. These are similar as Daubechies db1. These wavelets usually use the input vales then make pairs of them and then compute the difference of the pairs and also these passes the sum of the pairs. Finally, after many repetitions, it results in a single sum and difference values.

Advantages:

a) These wavelets are the only ones to be supported compactly, symmetrically and orthogonally.
b) Because of compact support, time localization of haar wavelet is good.

Disadvantages: a) These have jump discontinuities which is the reason for poor decay of the coefficients of haar.

b) Symlets: Theses wavelets are short form of “Symmetrical Wavelets”.

Advantages:

a) These wavelets are symmetrical in nature.
b) These are made in a way that they have very less asymmetry.

Disadvantages:

a) These wavelets are not fully symmetrical.

c) Daubechies Wavelets: These were invented by Ingrid Daubechies. These are also known as compact support orthonormal wavelet. These are one of most important wavelets in research. These make the analysis of discrete wavelet practical. These waves are denoted as dbN, N denotes the order, db is the name of the family to which the wave belongs.

Advantages:

a) These are orthogonal which means that these wavelets preserve energy.
b) These are supported compactly.

d) Coiflets Wavelets: Function for this wavelet is 2N moments which is equal to 0. The function for scaling is a 2N-1 moment which is equal to 0. Both these functions also have a support of length which is equal to 6N-1

Advantages:

a) These wavelets have symmetric graphs.
b) These wavelets are somewhat same as daubechies wavelet.

Disadvantages:

a) The arbitrary genus of Coiflets has no formula, and also there is no proof of its existence.

e) Biorthogonal Wavelet: These are compactly supported wavelets.

Advantage:

a) The current compression system instead of using orthogonal wavelets, use biorthogonal wavelet.

Disadvantage:

a. These wavelets are not energy preserving.

f) Meyer wavelet: These wavelets are orthogonal wavelet. These wavelets are described in the frequency domain. Also, these wavelets are differentiable indefinitely and has infinite support.

Advantage:

a) These are compactly supported.

3.5 Compression algorithm:

The procedure for compression contains the following steps:

a. Decompose:

   Firstly, select wavelet, select a level N. then calculate decomposition of wavelet for signal s which is at level of N.

b. Threshold detail coefficients

   We have to choose threshold and then perform thresholding (hard) to the detail coefficients, for every level between 1 and N.

c. Applying the compression technique

   The compression technique involves both Huffman and Run length encoding technique for data compression.

d. Calculation

   After the compression of the signal we measure the amount of compression achieved.

e. Decompression

   After the calculation step we move in the reverse direction in order the regenerate the original signal, so we apply Run length decoding as well as Huffman decoding to the compressed signal.

f. Reconstruct

   Lastly, with the help of original level N approximation coefficients and level 1 to N detail coefficients which are modified we compute wavelet reconstruction.

The proposed algorithm consists of lossy compression method as well as lossless compression algorithm that makes the overall algorithm lossy. So, we create a sample of N blocks of the PCG signal then we apply an algorithm to the independent block which are not overlapping. The procedure for compression of these blocks is:

- Decompose the PCG signal using the wavelet
- Thresholding the coefficients of the wavelet dynamically

- Compress the coefficient vector of the wavelet by the use of zero removal
- Use Run-Length or Huffman encoding to compress significance map.

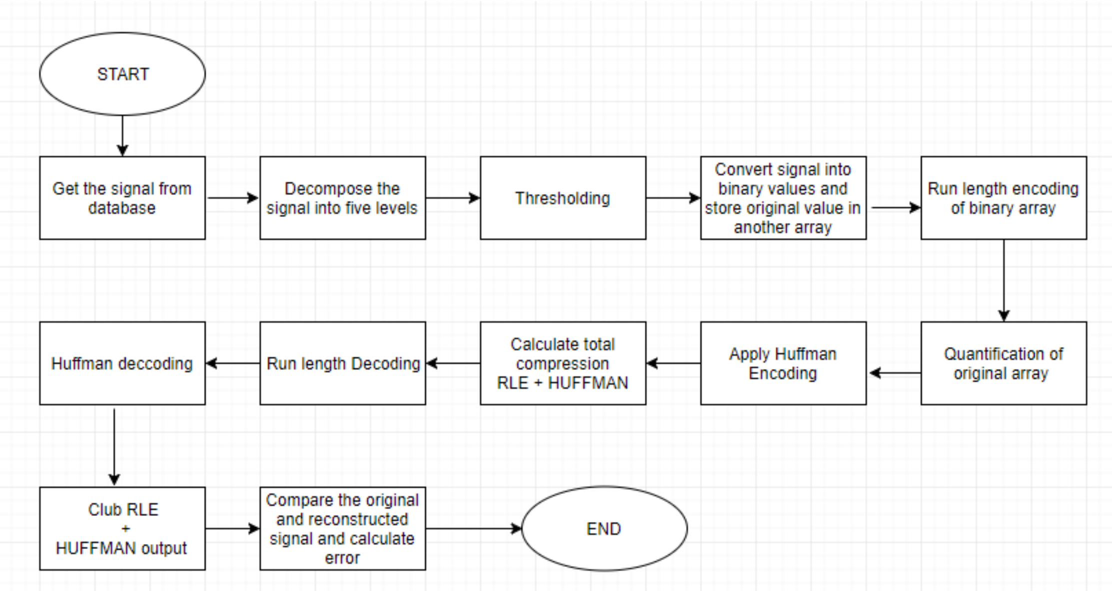

Figure 3-2 Algorithm Flowchart

3.5.1 Huffman Code

This algorithm was given by David Huffman. It forms a binary tree by assigning every symbol to leaf node. Every node has some weight assigned. This weight is frequency or number of times a symbol occurs which is also known as cost.

This tree structure is a result because of the combination of all the leaf nodes one by one until all nodes are embedded in one single root tree. This algorithm adds the frequency of the two nodes with the lowest frequency or cost. The combination procedure is bottom up. The newly formed interior nodes (which is the parent of combined nodes) gets the value which is equal to the sum of the frequency of both the children nodes.

ASCII Encoding

The example string **"happy hip hop"** will be used throughout the encoding procedure. If we use standard ASCII encoding, this string which has 13 characters requires 104 (13*8) bits in total. The below table shows subset of the standard ASCII table which will help us encode our string.

Char	ASCII	bit pattern (binary)
A	97	01100001

H	104	01101000
I	105	01101001
O	111	01101111
P	112	01110000
Y	121	01111001
Space	32	00100000

Table 3-1 ASCII coding char and patter

We will encode our example string **"happy hip hop"** using ASCII encoding as **104 - 97 - 112 - 112 - 121 - 32 - 104 - 105 - 112 - 32 - 104 - 111 - 112**. Though humans cannot read or understand this, this is the bit stream: 01101000 - 01100001 - 01110000 - 01110000 - 01111001 - 00100000 - 01101000 - 01101001 - 01110000 - 00100000 - 01101000 - 01101111 - 01110000

For decoding this string (i.e. to change binary code to original characters), for this we will create 8-bit bytes divisions of the encoded bit strea, and then we have to change each division back to character using table of ASCII encoding. For example, the starting 8 bits of the bit stream are **01101000**, which can be converted to 104 (decimal), and in ASCII set, the lowercase **'h'** is mapped at position 104. A file which is to be encode in this encoding does not require any kind of additional data.

A more compact encoding

Using ASCII encoding is excessively generous because it uses 8 bits for each character. Though using ASCII encodings allows us to represent 256 characters, but we only had 7 characters in our example. So, we can use less characters. We can use 3 bits for representing each character. Here is an example that uses 3 bits:

Char	ASCII	bit pattern (binary)
H	0	000
A	1	001
P	2	010
Y	3	011
I	4	100
O	5	101
Space	6	110

Table 3-2 3 bit char and bit pattern

Using this table, we can encode **"happy hip hop"** as **0-1-2-2-3-6-0-4-2-6-0-5-2**

When converted to binary: 000 – 001 – 010 – 010 – 011 – 110 – 000 – 100 – 010 – 110 – 000 – 101 - 010

If we use only 3 bits for each character, our example which is encodes will require 39 bits in place of using ASCII coding which was 104 bits, which is about 38% compression of the original size.

But for decoding we must know mapping table which can cut on our compression.

A variable-length encoding

In this encoding, we drop the need of all the characters take the same bits. For more frequently occurring characters like **'p'**, **'h'** and space, we can use fewer bits and more bits to encode characters less frequently occurring characters like **'y'** and **'o'**. By using this encoding, we can compress even further. The table below is optimal Huffman encoding (which is a type of variable length encoding) for string **"happy hip hop"**:

char	**bit pattern**
A	000
H	01
I	001
O	1110
P	10
Y	1111
Space	110

Table 3-3 Variable length encoding char and bit pattern

There is a unique bit pattern for every encoded character, but the bit length is not a constant.

when encoded our example **"happy hip hop"** by the use of above table is:

01 - 000 - 10 - 10 - 1111 - 110 - 01 - 001 - 10 - 110 - 01 - 1110 – 10

The total bits required by the encoded phrases are 34 bits, which is fewer bits than the fixed-length version.

The only problem is that we can no longer determine the start or end of a character in binary form. We won’t know if a character is 0 or 01 or 010. But if we look there are no two characters that encode to 01 and other to 0100 or 010.

Like fixed-length encoding, a file encoded by Huffman encoded will attach a header containing the table which is used. So, by using the table file can be decoded. Table for every file is different because for every file the table is constructed explicitly so that it will be optimal.

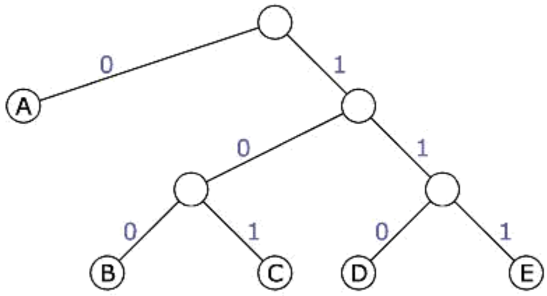

Figure 3-3 Huffman Tree

The tree branches represent binary values of 0 or 1 which is a rule for the code trees which are prefix-free. A particular code word is defined by the path between the root to corresponding leaf node.

Example of Huffman code:

This example is based on a data with only five symbols.

The frequency for every symbols is:

Symbol	**Frequency**
A	24
B	12
C	10
D	8
E	8

186 total bits

(if we us 3 bit for every code word)

Table 3-4 Huffman code Input symbol with frequencies

Firstly, the two of the rarest symbols which are E and D, are added, C and D are connected next. The parent nodes of the previously connected nodes have frequency of 16 and 22 respectively and these new nodes will be connected next.

Remaining nodes which is 'A' node and the node formed by adding 16 and 22 are connected to create root node.

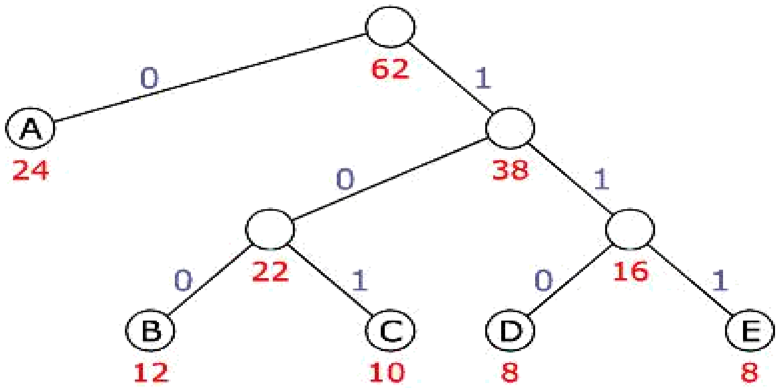

Figure 3-4 Huffman Tree Example

Symbol	Frequency	Code	Code Length	Total Length
A	24	0	1	24
B	12	100	3	36
C	10	101	3	30
D	8	110	3	24
E	8	111	3	24

Total bits required 138 bit

Table 3-5 Huffman table for construction of Huffman tree

Huffman Codes Characteristics

These codes are binary code trees which are prefix-free, therefore all considerations for prefix-free trees applies.

Ideal code length is achieved when Huffman algorithm generates the code. These have a deviation of which is smaller than 1 bit.

Example:

Symbol	P(x)	I(x)	Code Length	H(x)
A	0.387	1.369	1	0.530
B	0.194	2.369	3	0.459
C	0.161	2.632	3	0.425
D	0.129	2.954	3	0.381
E	0.129	2.954	3	0.381

theoretical minimum: 2.176 bit

code length Huffman: 2.226 bit

Table 3-6 Optimality of Huffman coding

Using the assumptions of distribution mentioned, The computation of entropy will on an average result in a 2.176 code length bit for every symbol. On the other hand, Huffman code will attain an average code length of 2.226 bits for every symbol. So, the Huffman coding may approach to the optimum on 97.74%.

Variants

The Huffman coding code tree is constructed by using a certain probability distribution. There are three variants for determining this distribution:

- Static probability distribution
- Dynamic probability distribution
- Adaptive probability distribution

Static Distribution

The Static Huffman's coding procedure operate with the previously defined code tree. For all types of data code is defined. Also, this is not dependent on the contents decoded or encoded. Usually, this type of trees is based on some kind of analysis which is standard, for e.g. English texts, we take the frequencies of all the symbol which are found in it.

An efficiency which can be accepted can be achieved, if the source data can correspond to frequency distribution which is used. It is not important to send Huffman tree and frequencies of each symbol in encoded data. We just have to make it available in both the encoding and decoding software. Plus, this tables for coding need not be created at the runtime.

The main problem for the static and predefined code tree occurs, only if distribution probability of the encoded file is very different from assumption. In that case, the rate of compression will decrease drastically.

Dynamic Distribution

Instead of using a static tree for all types of data, we can do dynamic analysis of probability distribution. The codes which are generated from such type of code trees match real conditions better than that of the standard distributions.

One of the major disadvantage of the dynamic procedure is, that the Huffman tree has to be stored in the compressed files. A code table or frequencies of the symbols should be a part of header data.

If we apply dynamic Huffman code to the entire data set, all the internal variations will not be considered. A practical solution to this problem is that we divide data into segments and periodically update the code tree. But the problem with this is that the size of header would also increase because of these additional data.

Adaptive Distribution

The procedure for adaptive coding has a code tree which is adapted permanently to the data which is encoded or decoded previously. In this we start with an empty tree or some kind of standard distribution. We will refine the code tree after we encode each symbol. By using this way, we can continuous adapt the code tree. Local variations can be done at run-time.

The adaptive Huffman codes starts with empty trees which are operating with a special control character which identify new symbols which are currently not a part of the tree. This type of variant is used because of its minimum requirement of the header data. But the compression rate attained at the beginning of compression (and for small files) is unfavourable.

Construction of the Tree

At a given frequency distribution, the binary code tree generated by the Huffman algorithm are very efficient. The only requirement is a table which contains all the symbols with their frequency. A leaf node can be represented by any of the symbol within the tree.

The general procedure for Huffman coding is:

- Firstly, we have to search for two nodes which has the lowest frequencies and has no parent node

- Combine both of the nodes to form newly created interior node
- Then add the frequency of both the nodes and assign the sum to new interior node formed

This procedure will be repeated till all of the nodes are connected to form a root node.

For example: "abracadabra"

Symbol	Frequency
a	5
b	2
c	2
d	1
e	1

Table 3-7 Dynamic Huffman code input (symbol and frequency)

According to Huffman coding, firstly "d" and "c" are combined. The new interior node formed will have a frequency of 2.

First step:

Symbol	Frequency	Symbol	Frequency
a	5	A	5
b	2	B	2
r	2	R	2
c	1	--------------->	12
d	1		

Table 3-8 First step of Huffman tree construction

Code tree generated after the 1st step:

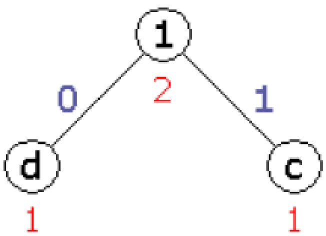

Figure 3-5 Tree after first step

Second step**:**

```
Symbol    Frequency     Symbol  Frequency
   a         5            a        5
   b         2            b        2
   r         2 ----------- ---> 2   4
   1         2
```

Table 3-9 Second step of Huffman tree construction

Code tree generated after the 2nd step:

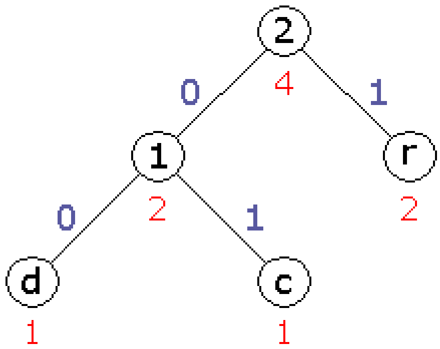

Figure 3-6 Tree after second step

Third Step**:**

```
Symbol    Frequency     Symbol  Frequency
   a         5            a        5
   2         4 ------------ ---> 3   6
   b         2
```

Table 3-10 Third step of Huffman tree construction

Code tree after the 3rd step:

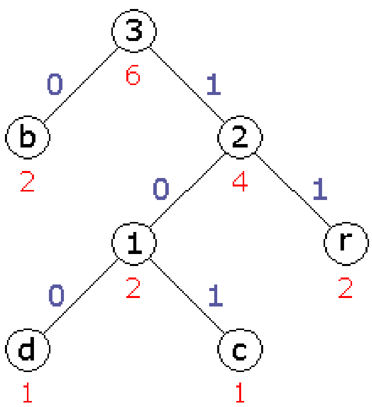

Figure 3-6 Tree after third step

Fourth Step:

Symbol	Frequency	Symbol	Frequency
3	6-------------→	4	11
a	5		

Table 3-11 Fourth step of Huffman tree construction

Code tree after the 4th step:

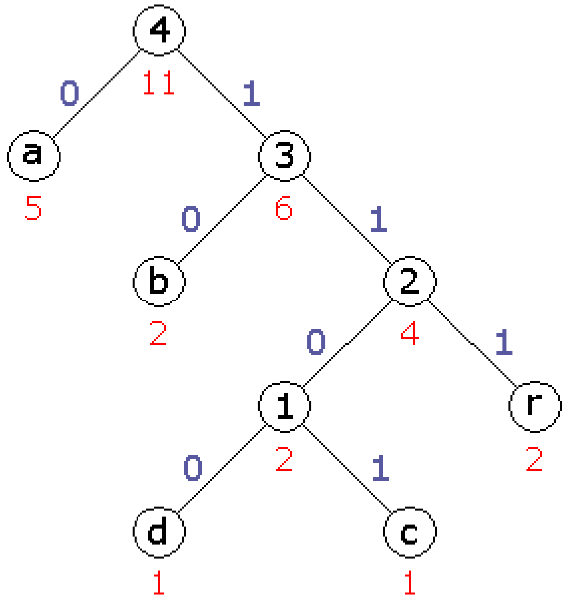

Figure 3-7 Code tree after 4th step.

Code Table

If there remains one node in table, we will form root of code tree. The path from root to a particular leaf node will make a codeword for that particular symbol.

Symbol	Frequency	Code Word
a	5	0
b	2	10
r	2	111
c	1	1101
d	1	1100

Table 3-12 Fourth step of Huffman tree construction

Complete Huffman Tree:

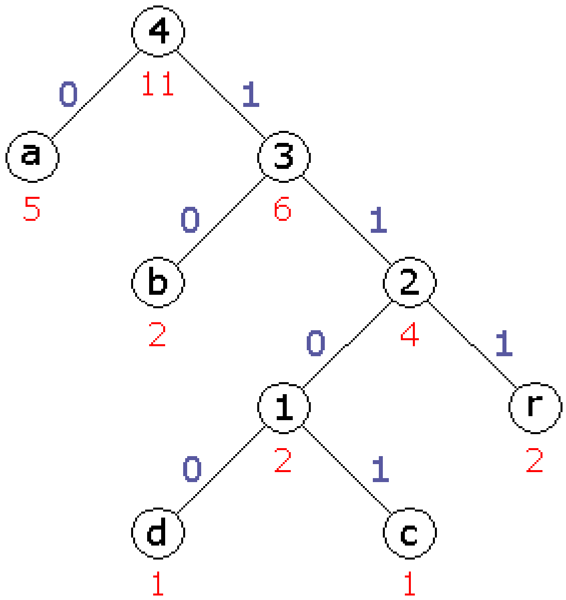

Figure 3-8 Complete Huffman tree

Encoding

The code table is shown below:

Symbol	Frequency	Code Word
a	5	0
b	2	10
r	2	111
c	1	1101
d	1	1100

Table 3-13 Original data with encoded code

a	b	r	a	c	a	d	a	b	r	a
0	10	111	0	1101	0	1100	0	10	11	0

Encoded bits: 23 Bits

Original bits: 33 Bits

Decoding

The decoding for Huffman tree is done by passing the Huffman tree step by step to encoded data. When a node which has no successor is reached, the symbol of the node is written in the decoded data.

01011101101011000101110

Encoded	0	10	111	0	1101	0	1100	0	10	111	0
decoded	a	b	r	a	c	a	d	a	b	r	a

Table 3-14 Encoded and Decoded table

Huffman Code Tree:

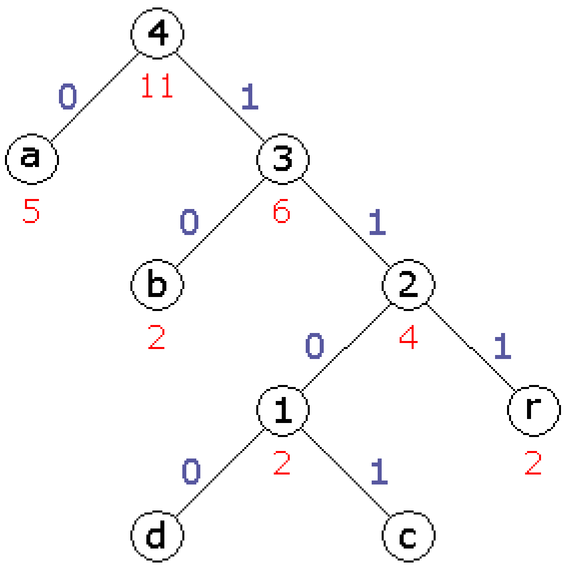

Figure 3-9 Huffman code tree

Alternative Construction

In scheme of coding previously described, newly created nodes at the interior are added at end of table given that the frequency remains same. Rule can be modified so that we can treat the nodes at the leaf better. If the nodes have the same frequencies the nodes at the interior can be added at top of leaf nodes. This creates the following table:

```
a(5)      a(5)        a(5)   --    3(6)   -+--> 4(11)
b(2) --> 1(2)    --> 2(4)   -+    a(5)   -+
r(2) |    b(2)    -+  1(2)   -+
c(1) -+   r(2)   -+
d(1) -+
```

Code Tree:

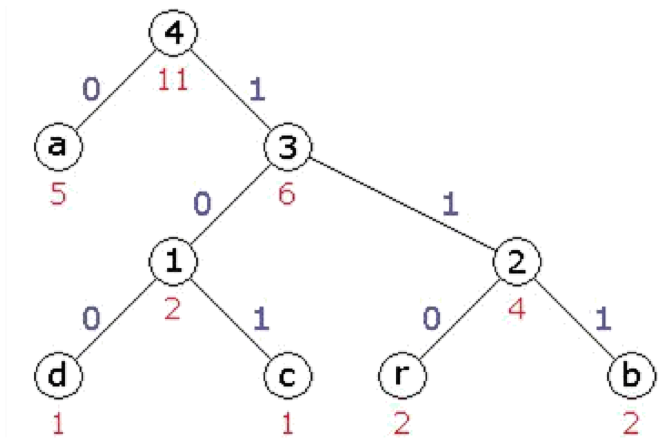

Figure 3-10 Huffman code tree

Symbol	Frequency	Code Word
a	5	0
b	2	111
r	2	110
k	1	101
d	1	100

Table 3-15 Symbol with frequencies and code word

Comparison

	Version	
Symbol	1	2
a (5)	0	0
b (2)	10	111
r (2)	111	110
k (1)	1101	101
d (1)	1100	100

Table 3-16 Comparison table

Coding Version 1

```
a    B    r    a    k    a    d    a    b    r    a
0    10   111  0    1101 0    1100 0    10   111  0
-> 23 bit
```

Coding Version 1

```
a    b    r    a    k    a    d    a    b    r    a
0    111  110  0    101  0    100  0    111  110  0
-> 23 bit
```

Both of these methods could be equally used for to the rate of compression.

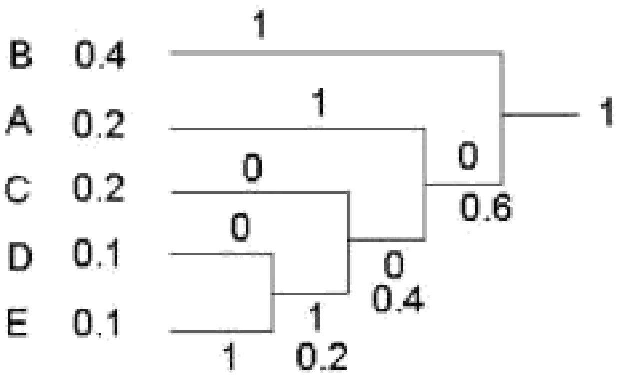

Figure 3-11 Huffman Tree

The following table illustrates the codes.

Symbol	Probability	Code
A	**0.2**	**01**
B	**0.4**	**1**
C	**0.2**	**000**
D	**0.1**	**0010**
E	**0.1**	**0011**

Table 3-17 Symbol, Probability and Huffman code

3.5.2 Run Length Encoding (RLE)

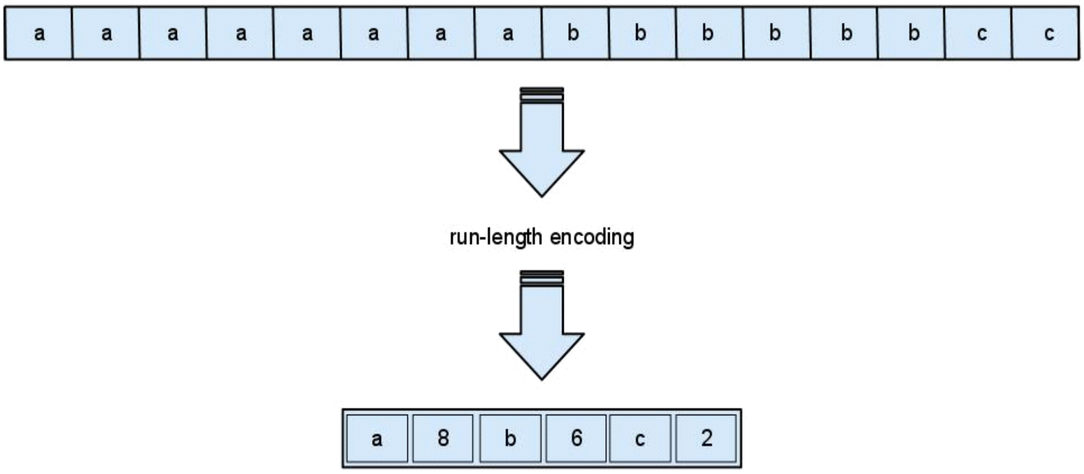

Figure 3-12 Runlength example

The Run Length Encoding (RLE) takes advantage of fact that there are certain sequences of recurring and identical symbols in a data set. These repetitions in the data set would be replaced when we declare the length of sequences.

This compression method has one of its most important application in simple graphic data which is characterized by having large parts of same colour. The simple graphic data will include sketches, technical drawings or diagrams. The procedures of RLE are a part of TIFF, PCX or BMP sub-formats.

The RLE compression procedures are one of simplest data compressions. These require only a small amount of resources (hardware and software). Therefore, RLE being a simple data compression was introduced early and since then a large number of its derivatives have been developed.

RLE General Principle:

In place of storing the data, we will store the so called runs. Run in a general form is sequence of a certain length which contains only one symbol. The symbol is known as run value while length of the sequence is known as run count.

```
data: aaaabbc
  Run        "aaaa"  "bb"  "c"
  Count        4       2    1
  Value        a       b    c
data: aaaabbc
RLE coded: 4a2b1c
```

In a case which is ideal, the RLE can change a sequence containing 256 same symbols to just 2 Bytes, supposing a counter of 1 Byte. But RLE will also be applied to only one symbol, this will lead to the doubling of data volume This is a unfavourable case.

8 Bit RLE

The general procedures of RLE are based on a standard unit byte, which is in accordance to configurations of the usual computer systems and the appropriate data formats. Thus, both the size of counter and set of symbols are limited to 256.

8 Bit Coding Scheme:

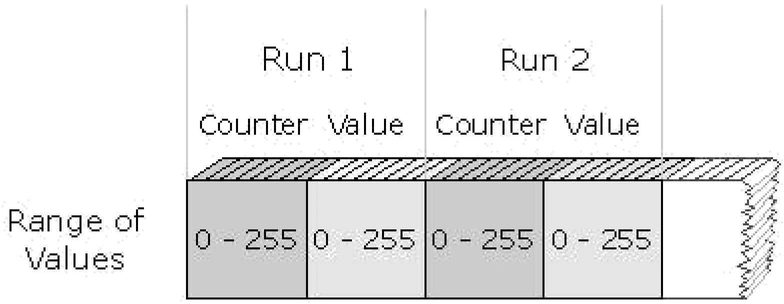

Figure 3-13 8 bit coding scheme

If the data which does not have any repetitions is coded by using this scheme, the volume of the data will be doubled. These should be a more practicable approach which should provide a mechanism which can process such kind of data segments.

8 Bit Coding schemes:

In regard with the counter used as a byte variable, it will provide a range of values which will be between 0 and 127 indicating a run which will be between 1 and 128 repetitions and the symbol range will be between 128 and 255 un-coded data which is indicated between 1 and 128 symbols.

If we use the following coding scheme, we can hold the loss limits. If even the original data, does not have any kind of repetitions, we only have to insert an additional counter every 128 symbol.

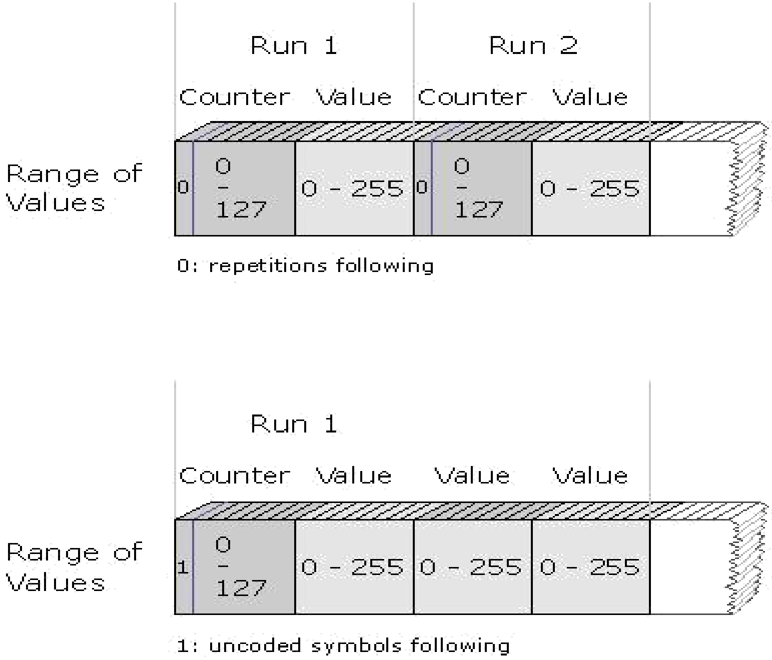

Figure 3-14 8 Bit Coding Scheme a) 0 repetitions following b) 1 Unicoded symbols following

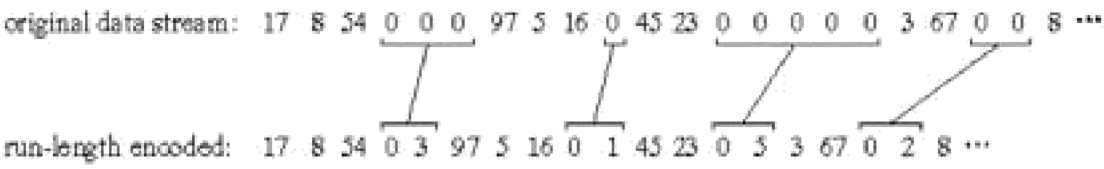

Figure 3-15 Run Length Encoding Example

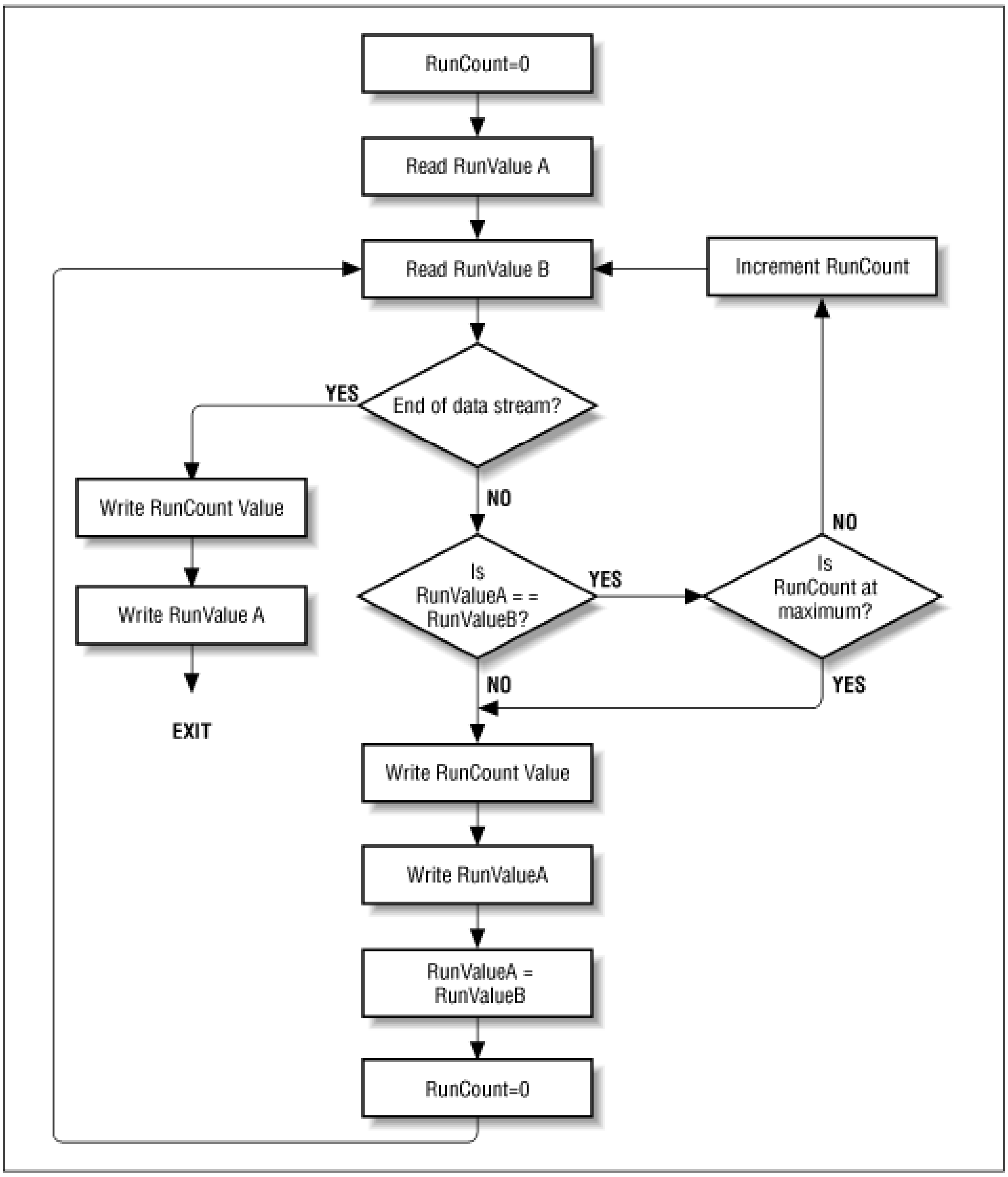

Figure 3-16 Flow chart of RLE algorithm

Chapter 4 Performance Analysis

Decomposition of normal heart sound signal and reconstruction graph compression using different wavelets.

a) Coiflets

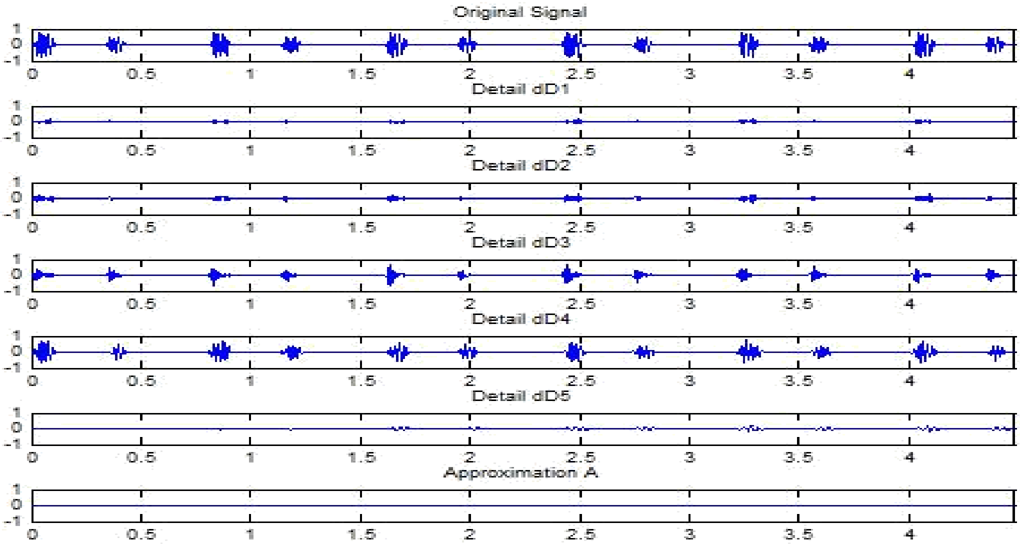

Figure 4-1 level-5 decomposition of normal heart sound signal for coiflet Mother Wavelet

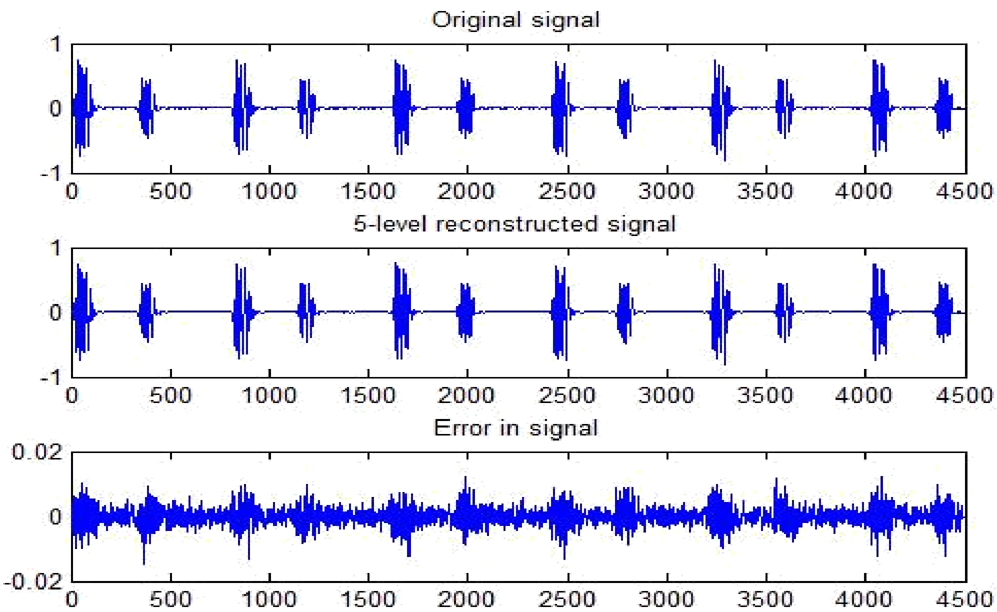

Figure 4-2 Comparison between original signal and reconstructed signal and error occurred

b) Haar wavelets

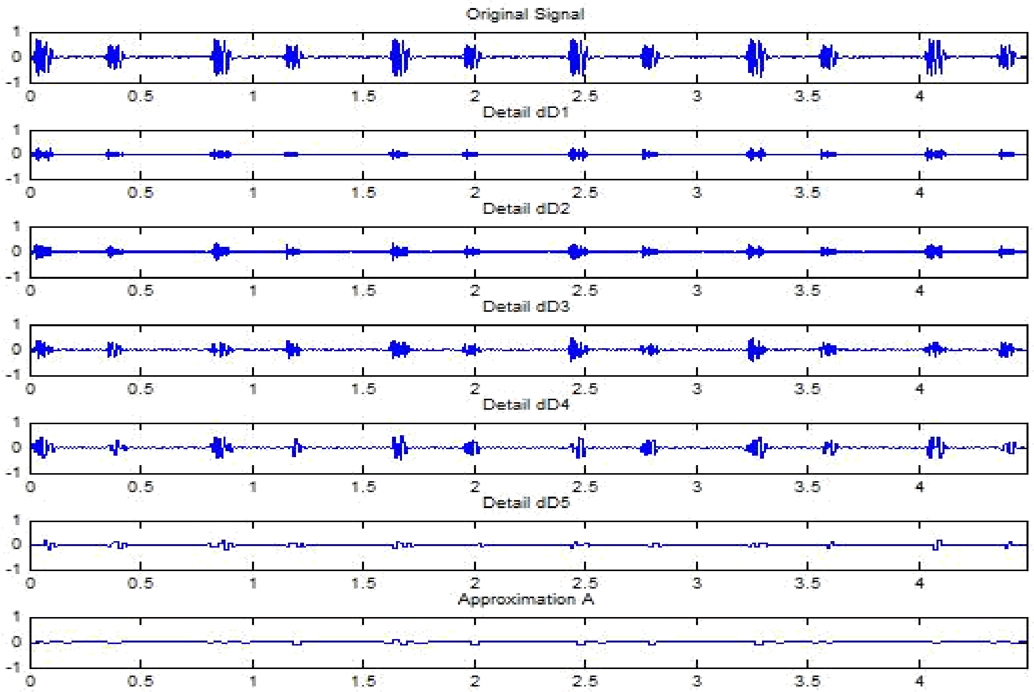

Figure 4-3 level-5 decomposition of normal heart sound signal for Haar Mother Wavelet

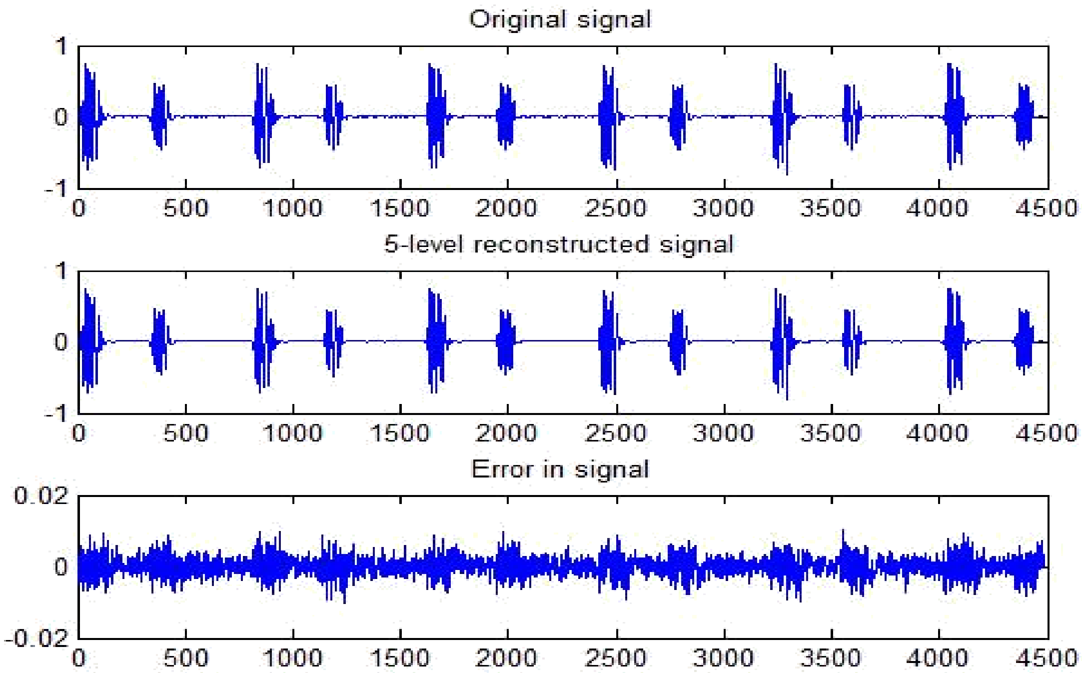

Figure 4-4 Comparison between original signal and reconstructed signal and error occurred

c) Daubechies wavelet

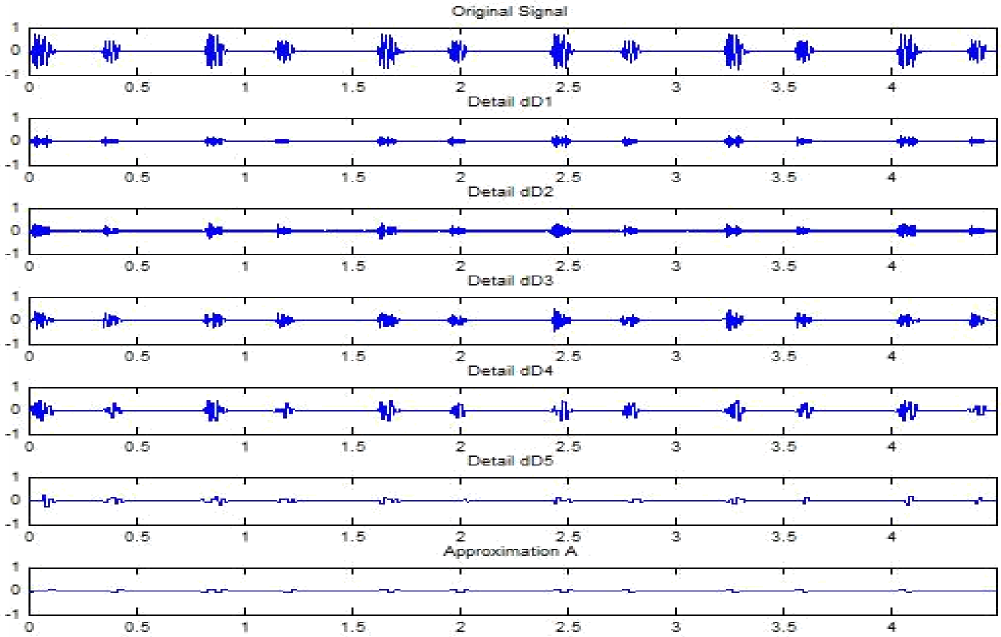

Fig 4.5 level-5 decomposition of normal heart sound signal for Daubechies Mother Wavelet

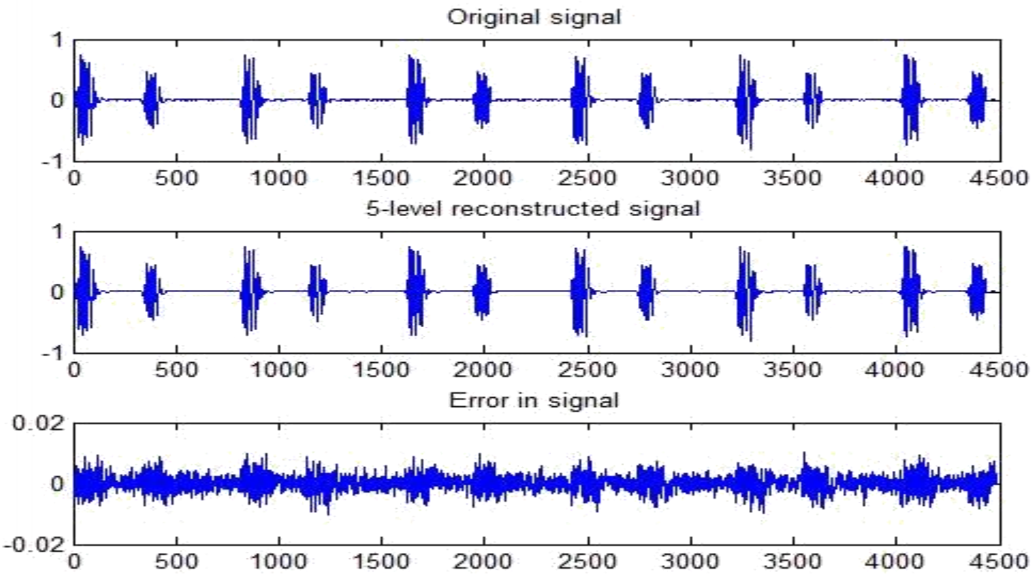

Fig 4.6 Comparison between original signal and reconstructed signal and error occurred

OUTPUT:

PCG SIGNAL COMPRESSION
===
Levels of decomposition = 5
Wavelet = coif5
Frequency = 1000

thr =

0.0012

thr =

0.0013

thr =

0.0050

thr =

0.0059

thr =

0.0059

=================COMPRESSION RESULTS===================
Total input bit required for Run length coding = 4626 Total output bit required for Run length coding = 2298

Approximate size of dictionary in number of bits = 2508
Total input bit required for HUFFMAN coding = 21450
Total output bit required for HUFFMAN coding = 10832
Final compression percentage = 49.6472
==
Compression ratio = 0.803808

==============Decompression starts============================
ErrCompareSignal =

6.9533e-06

========================END==============================

Table showing the effect of types of threshold on compression ratio, mean square error (MSE) on normal heart sound signal:

Serial no.	Threshold name	Compression Ratio	MSE
1	Rigrsure	0.803808	6.9533e-06
2	Heursure	0.803808	6.9621e-06
3	Sqtwolog	1.170456	1.5638e-05
4	Minimaxi	1.118601	9.7326e-06

Table 4-1 Dependency of Mean Square Error and compression ratio on threshold technique

Note: "Rigrsure" and "Heursure" produces same threshold value hence the compression ratio and mean square error is same in both the cases.

name	Level1	Level2	Level3	Level4	Level5
Rigrsure	0.0012	0.0013	0.0050	0.0059	0.0059
Heursure	0.0012	0.0013	0.0050	0.0059	0.0059
Sqtwolog	0.0063	0.0069	0.0243	0.0382	0.0587
Minimaxi	0.0040	0.0044	0.0155	0.0244	0.0375

Table 4-2 Table shows the threshold value of each method at different levels

Observation: From this table we can correlate the relation between threshold and MSE in case of PCG signal compression if threshold is more the mean square error is more.

Serial no.	Type of threshold	Effect on compression %	MSE
1	Soft	0.831678	1.1293e-05
2	Hard	0.803808	6.9533e-06

Table 4-3 Table shows effect of soft and hard thresholding on compression ratio and Mean Square Error

Chapter 5 Conclusion

5.1 Conclusions:

The signal can be compressed using the above proposed method very efficiently and hence can be stored and transmitted easily. We have shown the threshold technique that should be used for the compression of the PCG signal for better compression and the result shows that "Sqtwolog" Method gives better compression result on the basis of above experiment but Mean square error is more whereas "rigrsure" shows little less compression but mean square error is less. For such signal less, error compression technique is preferred in order to avoid loss of necessary information. And after this we check what the effect of soft and hard thresholding is over Mean Square Error and compression.

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