ANALYSIS OF THE SECOND HEART SOUND FOR MEASUREMENT OF SPLIT

A Thesis Submitted in Partial Fulfillment
of the Requirements for the Award of the Degree of

Master of Technology

in

Electronics and Instrumentation Engineering

by

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May 2013

Declaration

I hereby declare that the work presented in the thesis entitled *"Analysis of The Second Heart Sound For Measurement of Split"* is a bonafide record of the research work done by me under the supervision of Prof. Samit Ari, Department of Electronics & Communication Engineering, National Institute of Technology, Rourkela, India and that no part thereof has been presented any other University/Institute for the award of any Degree or Diploma.

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CERTIFICATE

This is to certify that the Thesis Report entitled "**ANALYSIS OF THE SECOND HEART SOUND FOR MEASUREMENT OF SPLIT** " submitted by **RIA PRAMANIK** bearing roll no. **210EC31866** in partial fulfillment of the requirements for the award of Master of Technology in Electronics and Communication Engineering with specialization in **"Electronics and Instrumentation Engineering"** during session 2010-2013 at National Institute of Technology, Rourkela is an authentic work carried out by her under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other University / Institute for the award of any Degree or Diploma.

Place:

Date:

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Dedicated

to

My family

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Abstract

A2 and P2 are the two components of the second heart sound S2. A2 is the sound produced by the closing of the aortic valve and P2 is due to the closing of the pulmonary valve. Usually the pulmonary valve closes after the aortic valve. The closure of the two valves introduces a time delay. This delay is known as split. Discrete wavelet transform (DWT) and continuous wavelet transform (CWT) are used to measure the split between the two components of A2 and P2 of second heart sound of normal and pathological case of Phonocardiogram (PCG) signal. To recognize the split, A2 and P2 are identified and delay between the A2 and P2 are estimated. DWT is used to find the split between the two components and continuous wavelet transform is used to find the number of frequency components of S1 and S2. Split between the two components is measured by the continuous wavelet transform. Also normalized split Z_n can be calculated. If Z_n is less than one, the split is the normal otherwise, it is pathological split.

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LIST OF ABBREVIATIONS

FT   Fourier transform

STFT   Short Time Fourier Transform

WT   Wavelet transform

DWT   Discrete wavelet transform

CWT   Continuous   wavelet   transform   c

1

Introduction

Introduction

The analysis of the sound produced by the heartbeat by auscultation is insufficient to diagnose some heart disease. The analyst cannot estimate qualitative and quantitative characteristics of the phonocardiogram signal [1, 2]. Study of physical characteristics of heart sound by human ear is incapable of diagnosing cardiac condition [3]. In this paper we will concentrate on analysis of the second heart sound and its two major components A2 and P2. First, the aortic valve closes then after some time delay the pulmonary valve closes. This delay is known as split [4-6].

S2 is the composite sound originating from the closure of aortic and pulmonary valve. The aortic component (A2) is louder than the pulmonic components (P2) of the second heart sound (S2) and is discernible at all the auscultation sites. It is heard from the right base with the diaphragm of the chest piece firmly pressed. The pulmonary component of S2 is softer than A2.

Second heart sound is generated when aortic and pulmonary valve close. A split S2 is heard in both physiological and pathological cases. In those cases the two valves do not close together. In physiological S2 splitting, the aortic valve closes before the pulmonary valve.At the time of inspiration both the components of S2 are heard. The split between A2 and P2 narrows at the time of expiration. So the sound S2 is heard as a single sound during expiration.

Delay in the activation of A2 is the main cause of reverse splitting. The result is that closure sound sequence is reversed. A2 closes after P2. Complete left bundle branch block (LBBB) causing delayed activation of the left ventricle is the common cause of reverse splitting.

About 50ms is the duration of wide split whereas normal split has a duration of about less than equal to 30ms.If the split is less than equal to 30ms then the split is normal split and if the split is greater than 30ms then the split is pathological split.

For normal and pathological cases, wavelet transform is a useful tool for analysing the sound S2. Wavelet transform has a very good time resolution for high frequency components. This time resolution increases with the increase in frequency and the frequency resolution decreases with the decrease in frequency [9,10].

STFT cannot analyses the signal which is very sensitive to sudden change in time direction. To analyses those type of signal it is very important to keep the length of the time window very

small. But if we keep the length of the time window small, there will be reduction of resolution of frequency in the time frequency plane. So there is a trade-off problem between the time and frequency plane [11].

However Wigner Distribution (WD) and corresponding Wigner Ville Distribution (WVD) can analyses the non-stationary signal properly. This ability comes from the fact that WD can separate the signal in both time and frequency directions. The advantage of WD over STFT is that it has no time frequency trade-off problem, but its disadvantage is that in its response it has a cross-term. Nonlinear behavior of the WD is the main cause of the cross-term. To remove the cross-term it is necessary to smooth the time frequency plane but it decreases the time frequency resolution [12].

WD cannot separate or display the signal components of the heart sound in time and frequency direction. In mono component signal analysis, it provides high time and frequency resolution.

The difficulties encountered in the non-stationary signal analysis with WD and STFT can be overcome using wavelet transform.

Most biomedical signals are non-stationary. On the other hand they have highly complex time frequency characteristics. They have high frequency components closely spaced in time accompanied by long lasting low frequency components closely spaced in frequency. WTs are compatible with non-stationary random signal so it is useful tool for biomedical signal analysis.

In case of stationary signal where frequency changes with time Fourier transform is powerful tool to track the signal magnitude, frequency or phase . In non-stationary signal statistical property changes with time and Fourier transform is inadequate track the change in signal magnitude, frequency or phase . Non-stationary signal can be analyzed by the use of Short Time Fourier Transform (STFT) and wavelet transform. Non-stationary signal analysis can be done using STFT and wavelet transform. Wavelet transform can be a useful tool for overcoming this disadvantage.

1.2 Mechanism of heart sound production

The human heart has four chambers i.e. left atrium left ventricles and right atrium and right ventricles. Inside the heart, blood flows from Atrium to Ventricles and from ventricles it is pump

out from the heart through pulmonary Artery and Aorta [15] .By the two stage i.e. Systole and Diastole the pumping of the blood happens.

Diastolic period

In this period the ventricle relaxes. So inside the ventricle pressure will drop. When the pressure falls below the pressure in the atrium the Mitral and Tricuspid valves are open and blood flows from the atrium into the ventricles.

Systolic period

In this period the ventricles contract. So inside the ventricles the pressure suddenly increase. This will open M itral and Tricuspid valves and open Aortic and pulmonary valves and pump out the blood.

The range of heart sound is 15-200 Hz. S1 or lubb and S2 or dubb are the two major heart sound. At the beginning of the ventricular contraction i.e at the time of systole S1 is produce due to the sudden blockage of reverse blood flow due to the closer of the Atrioventricular valves i.e. Tricuspid and Mitral valve.At the end of ventricular systole, i.e at the time of ventricular diastole S2 is produced by the sudden blockage of reverse blood flow due to the closer of the Semilunar valves i.e. Aortic valve and Pulmonary valve. Figure 1.3 shows a pictorial form of heart sounds in time domain and shows the location of S1,S2 and Murmur.
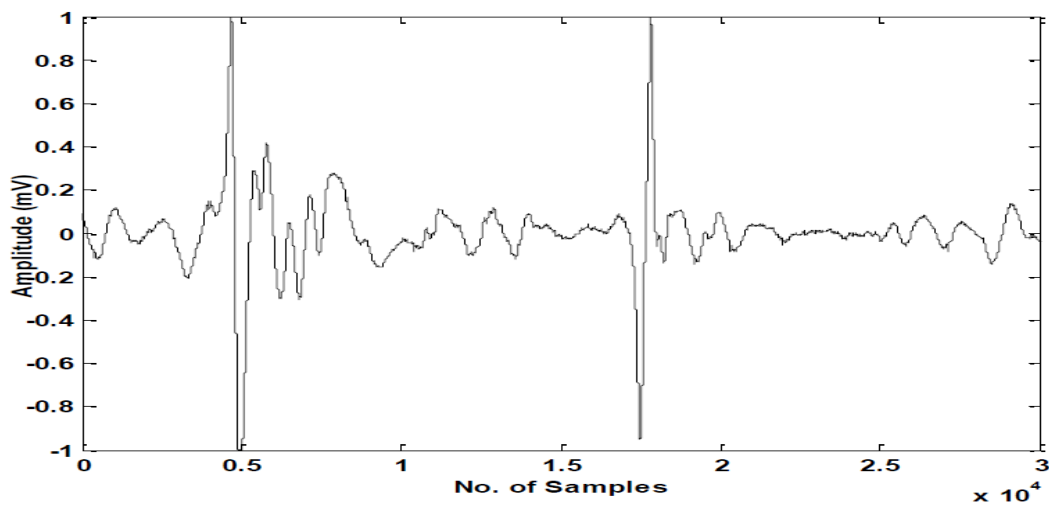

Figure 1.3: Normal phonocardiogram signal

1.3 Auscultation process of heart sound

By the use of stethoscope auscultation is the process of hearing the sound produced in body.there are four auscultation sites in the chest region . These are Aortic, pulmonary, lower Sternal Border and Mitral.

1.2 Thesis objective

Measurement of the delay known as the split in medical terminology between the aortic component A2 originating from the closure of aortic valve and pulmonary component P2 originating from the closure of pulmonary valve of the heart sound. This split gives the important information about the human heart. For a human being accurate measurement of the split by using stethoscope is impossible. By the use of two method i.e. wavelet transform and STFT we can easily measure the split. We also determine the normalized split. If it is less than one it is normal split otherwise pathological split.

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2

Spectral Analysis

2.1 Introduction

In case of a stationary time series the statistical property does not change with time. Fourier transform is adequate to measure the signal magnitude, frequency and phase. STFT has a localized time frequency. Here window function is used to localize the time and Fourier transform is used to localize the frequency.STFT has fixed width of window function. So it has poor time frequency resolution. So it has poor time frequency resolution. Basis function that is used by the wavelet transform dilates and contracts with frequency. The wavelet transform does not retain the absolute phase information and the visual analysis of the time scale plots that are produced by the WT is intricate. Time frequency region analysis is not done by the WT rather it does time scale region analysis. It can analyze the localized area of a larger signal.

2.2 Fourier Transform

Time domain signal is converted into frequency domain signal by the Fourier transform. Frequency domain signal is an alternate form of representation of a time domain signal. Fourier transform is a most common spectral analysis technique. In most cases, certain features of the signal that are visible in the frequency domain were invisible in the time domain. Delta basis function in the time domain is converted into infinite long sinusoidal basis function in the frequency domain by the Fourier transform. The sinusoidal basis functions are the solution of the mathematical equation describing a small perturbation of a physical system about a stable equilibrium point [2].

Fourier transform X(f) of a time series x(t) is given by

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-2\pi ft}\, dt \qquad (2.1)$$

Where t denotes time, f denotes frequency, x denotes differentiable, piecewise smooth, square integral function of time, X denotes Fourier transform of x. The inverse relationship is denoted by

$$x(t) = \int_{-\infty}^{\infty} X(f)e^{i2\pi ft}\, df \qquad\qquad (2.2)$$

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The Discrete Fourier Transform (DFT) of a time series of length N is given by

$$X[\frac{n}{NT}] = \frac{1}{NT} \sum_{K=0}^{N-1} x(kT) e^{-(\frac{i2\pi nk}{N})} \tag{2.3}$$

Where T is the sampling interval of the time series. The inverse relationship is given by

$$X(kT) = \sum_{n=0}^{N-1} x[\frac{n}{NT}] e^{(\frac{i2\pi nk)}{N}} \tag{2.4}$$

All integer frequency can be estimated to a certain extent by the Fourier Transform. But it has some disadvantages.

1. Fractional frequency cannot be estimated by the Fourier Transform. If we do Fourier analysis of a signal which has a fractional frequency the result is spreading of spectrum to other frequencies.which donot belong to the original signal.

2. In non-stationary signal statistical property changes with time and Fourier transform is inadequate to track the change in signal magnitude, frequency or phase. No information can be hidden in the Fourier amplitude spectrum. The time information is induced in the phase spectrum

3.If a short duration frequency signal exist in a long time series the short duration frequency may not be noticeable in the Fourier amplitude spectrum.

12.3 Short Time Fourier Transform (STFT)

STFT was one of the first time frequency representation techniques. It multiplies the time series with a series of shifted time windows and calculates the Fourier transform of the multiplied signal [3]. For a signal x(t), STFT is express as:

$$\text{STFT}(\tau, f) = \int_{-\infty}^{\infty} x(t) w(t - \tau) e^{-i2\pi ft} \, dt \tag{2.5}$$

Where arbitrary chosen window function is w(t). The size of the window is chosen in such a way that signals segment of the window is assumed to be stationary. The windowed signal is localization in time and thus the spectrum obtained is called the local spectrum.

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Time resolution is defined as how well a transform can resolve rapid variations in the time domain and frequency resolution refers to how well the change in frequencyies of a signal can be tracked. The time and frequency resolution are dependent directly on the width of the window used in the time frequency analysis [21]. Bandwidth of the window is proportional to the frequency resolution on the other hand length of the window function is proportional to the time resolution. So wide window is needed for good frequency resolution and shorter window offers good time resolution.

Time frequency is limited due to the Heisenberg-Garbor inequality that states that

$$\Delta t . \Delta f \geq K$$

Where Δt is the time resolution, Δf is the frequency resolution, K is the constant depends upon the type of the window used. So a pair of STFT is needed for good time resolution and good frequency resolution . Therefore narrow window is needed for good time resolution and wider window is needed for good frequency resolution.
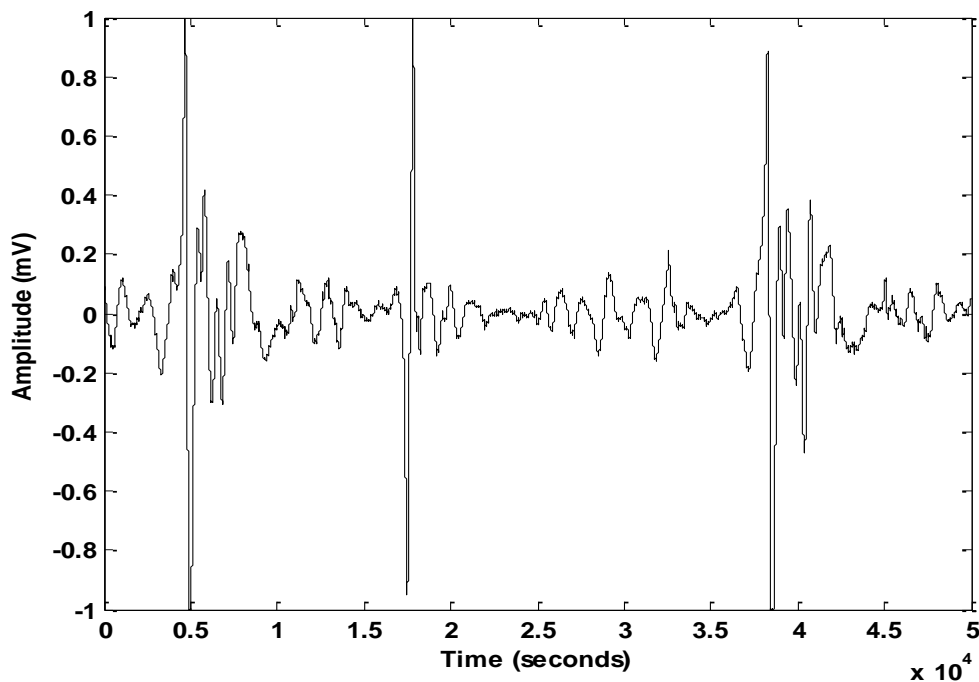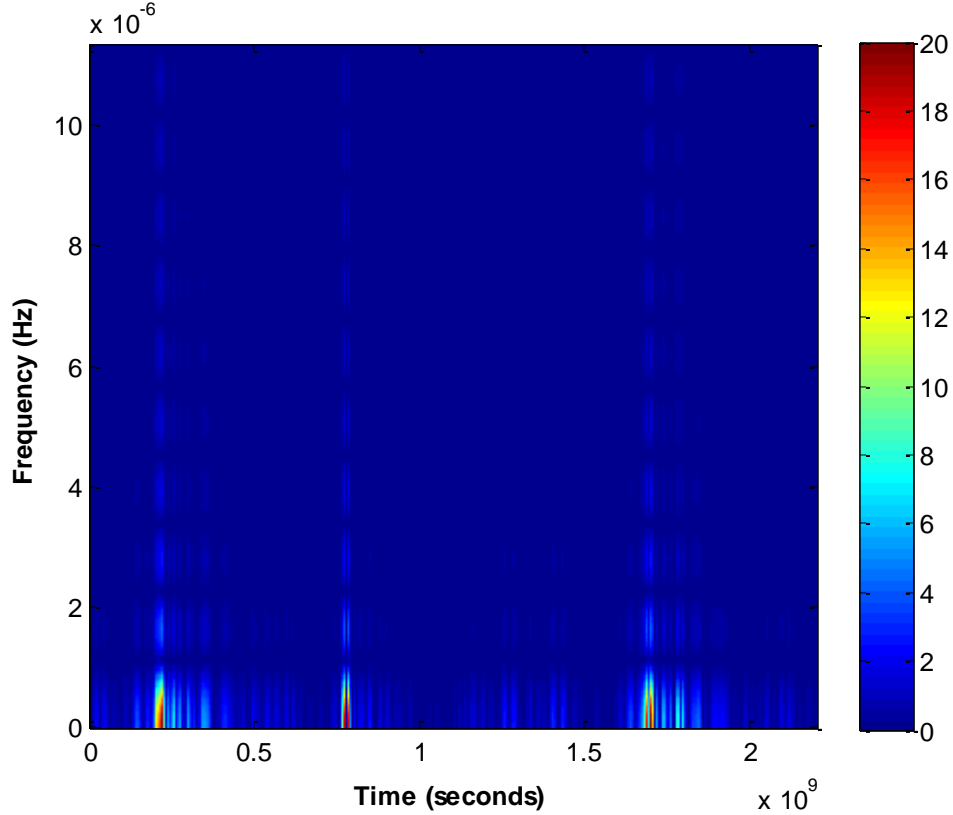
Figure 2.1: Short Time Fourier Transform

2.4 Wigner distribution (WD)

The Wigner distribution (WD) was discovered in 1932 by Wigner in the field of quantum mechanics . It usefulness to problem of communication theory was introduced by Ville in1948. So it is called the Wigner-Ville distribution. In order to differentiate from the higher order WD to the second order WD , we will refer the conventional WD as the Wigner spectrum (WS).

In contrast STFT has a limited resolution either in the time or frequency direction and suffer from the smearing and side lobe leakage. On the other hand STFT offer good resolution in both time and frequency direction. The Wigner cross spectrum (WCS)of two signals, x(t) and y(t) is defined as

$$\text{Wxy}(t,\omega) = \int_{-\infty}^{\infty} x(t+\tfrac{\tau}{2})y^*(t-\tfrac{\tau}{2})e^{-j\omega\tau}\,d\tau = \frac{1}{2\pi}\int_{-\infty}^{\infty} X\left(\omega+\tfrac{\varepsilon}{2}\right)y^*(\omega-\tfrac{\varepsilon}{2})e^{j\varepsilon t}\,d\varepsilon \quad (2.7)$$

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Where X(w) and x(t) are FT pair. The auto WS can be derived when x(t)=y(t),and is a bilinear function of the signal x(t). the WS can be express as

$$AF(\theta, \tau) = \int_{-\infty}^{\infty} x^*(t - \frac{\tau}{2})x(t + \frac{\tau}{2})e^{j2\pi\theta t} dt \qquad (2.8)$$

The WSis a quadratic transform. The WS is the sum of the two signal is not the sum of the individual WS. The WS has a cross-term. These cross-term can also be eleminated by appropriate filtering in the ambiguity function domain. That can be achieved by appropriate choice of the kernel. The effect of cross term can be reduced by using the kernel $\varphi(\theta, t)$.

$$\varphi(\theta, \tau) = \exp(- \theta^2 \tau^2 / \sigma^2) \qquad (2.9)$$

The amount of attenuation is controlled by the parameter σ (cross-term amplitude is proportional to σ).

2.5 The Wavelet Transform

A continuous time signal is wavelet that satisfies the following property

$$\int_{-\infty}^{+\infty} \varphi(t)dt = 0$$

$$\int_{-\infty}^{+\infty} |\varphi(t)|^2 dt < \infty$$

Where $\varphi(t)$ is defineds as the mother wavelet [4]. The continuous wavelet transform

$$W(a,b)=\int_{-\infty}^{\infty} y(t)$$

Y(t) is a square integrable function. a is the dilation parameter and b is a translation parameter and is the translation dilation of the mother wavelet defined as
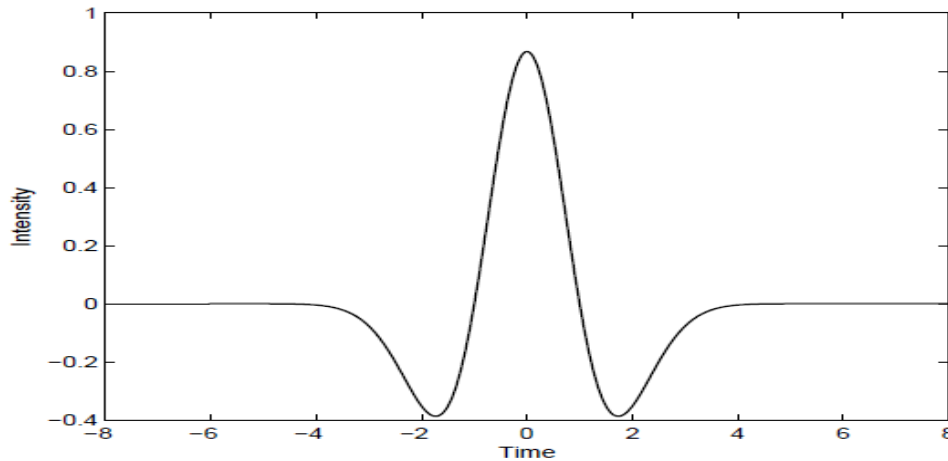
$$=\frac{1}{\sqrt{|a|}} \varphi(\frac{t-b}{a})$$

Figure 2.2: Mexican hat wavelet

From the continuous wavelet transform y(t) can be reconstructed provided the mother wavelet. The admissibility condition is defined as

$$C=\int_{-\infty}^{\infty} \frac{|\varphi(\omega)|^2}{|\omega|} d\omega < \infty$$

Where $\varphi( )$ is the Fourier transform of $\varphi(t)$. The reconstructed signal y(t) is given as

$$Y(t)=\frac{1}{c}\int_{a=\infty}^{\infty}\int_{b=-\infty}^{\infty}\frac{1}{|a|^2}w(a,b)\varphi_{a,b}(t)dadb$$

The wavelet transform is a two dimensional. Wavelet transform can be obtained by dilation and translation of the mother wavelet and signal's inner product . CWT can be represented by the time scale plot. In the time scale plot inverse of frequency is the scale. CWT offers high time resolution at high frequency (at low scale). CWT offers high time resolution at a lower frequency (at a high scale).

2.6 continuous wavelet transform

An alternate approach to the STFT is the continuous wavelet transform [15] . CWT is best suited for signal analysis. Wavelet transform consist of family of wavelet and computing coefficients that are inner product of signal x(t).

Continuous wavelet transform can be defined as

$$\varphi_x^\varphi(\tau, s) = \int x(t)\varphi_{s,\tau}^*(t)dt$$

Where$\varphi_{(s,\tau)}$(t) is called wavelet function.

The variables τ and s are called translation and scale respectively.

The wavelets are generated from a single basic wavelet $\varphi(t)$ (that satisfy the following properties), the so called mother wavelet, by scaling and translation [15] :

$$\varphi_{(s,\tau)}(t) = \frac{1}{\sqrt{|s|}}\varphi(\frac{t-\tau}{s})$$

The factor $s^{-1/2}$ is used for energy normalization across the different scales

Wavelet function in CWT should satisfy the following properties:

Admissibility condition:

$$C_\varphi = \int_0^\infty |\frac{\varphi(\omega)}{\omega}|d(\omega) < \infty$$

Where $\varphi(\omega)$ is the FT of $\varphi(t)$. This condition ensure that $\varphi(\omega)$ goes to zero as quickly as $\omega \to 0$

Zero average: It is normalize which means $\|\varphi\| = 1$ and is centered in the neighborhood of t=0 i.e.

$$\int_{-\infty}^\infty \varphi(t)dt=0$$

Unit energy: wavelet function should have unit energy

$$\int_{-\infty}^\infty |\varphi(t)|^2dt=1$$

In short wavelet zero average, fast decay and have unit energy. This means that $\varphi(t)$ is wave for short duration and hence the name wavelet. If $\varphi(t)$ is the real wavelet, then the resulting $\varphi_x(s,\tau)$ is called a real WT, measure the variation of x(t) in a neighborhood of τ whose size is proportional to scale s. A real wavelet transform maintains a energy conservation principle, as long as the wavelet satisfies admissibility condition [15].

2.7 Discrete wavelet transform

CWT has a redundancy. CWT is calculated by continuously shifting scalable function and calculating the correlation between them. This scalable function is orthonormal basic and the obtain wavelet is highly redundant. The problem of redundancy can be overcome by use of discrete wavelet transform. In discrete wavelet transform the scale and translation parameter are chosen in such a way that the resulting wavelet set forms a orthogonal set i.e. the inner product of the individual wavelets $\varphi_{s,\tau}$ are equal to zero. Discrete wavelet transform are not continuously scalable and translatable but can only be scaled and translated in discrete step [14]. Modifying the wavelet representation as

$$\varphi_{j,k} = \frac{1}{\sqrt{s_0^j}} \varphi\left(\frac{t - k\tau_0 s_0^j}{s_0^j}\right)$$

Where j and k are the integers and $s_0 > 1$ is a fixed dilation step and τ_0 depends on the dilation step. Discretizing wavelet has time scale space which is sample at a discrete interval. We generally choose $s_0 = 2$ so that sampling frequency is a ddyadic sampling. We generally choose $\tau_0 = 1$ for translation factor. In such a case the equation becomes

$$\varphi_{j,k}(t) = \frac{1}{\sqrt{2^j}} \varphi\left(\frac{t - k2^j}{2^j}\right)$$

One of the efficient ways to construct the DWT is to iterate a two –channel perfect reconstruction filter bank over the low pass scaling function branch. This approach is called the Mallat algorithm [15]. DWT has two sets of related function called scaling function and wavelet function and are given by

$$\emptyset(t) = \sum_{n=0}^{N-1} h[n]\sqrt{2}\,\emptyset(2t - n)$$

$$\varphi(t) = \sum_{n=0}^{N-1} g[n]\sqrt{2}\,\emptyset(2t - n)$$

Where the function $\varphi(t)$ is called the wavelet function and $\emptyset(t)$ is called the scale function, g[n] is the impulse response of a high pass filter and h[n] is an impulse response of a low pass filter.

2.8 Conclusion

The information obtained from the frequency domain analysis of the signal , can not be obtained from the time domain analysis of the signal. Fourier transform is adequete to track the change in signal magnitude, frequency or phase for non stationary signal where the statistical property does not change with time. But where the statistical property changes with time that is in case of non stationary signal fourier transform is inadequate to track the signal. There we use STFT and Wavelet transform for time frequency analysis. STFT has fixed time resolution.

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3

Measurement of split in the second cardiac sound

3.1 Introduction

The second heart sound (S2) [1-2] has two components A2 and P2. A2 is due to the closer of the aortic valve and P2 is due to the closer of the pulmonary valve. Closure of the aortic valve usually occurs before the closure of pulmonary valve. This introduces a time delay known as split [1]. A discrete wavelet transform and continuous wavelet transform are used to measure the split of the second heart sound for normal and pathological case. To quantify the split the two components in S2 are identified and delay between two components can be estimated. Most common cause of the reverse split is the delay in the closure of the aortic component A2. There result is that the sequence of the closure sound is reversed. P2 precedes A2. Reverse splitting in most of the cases is due to the left bundle branch block and, consequent delayed activation of the left ventricle. The duration of the wide split is about 50ms compared to the normal split of about less than equal to 30ms
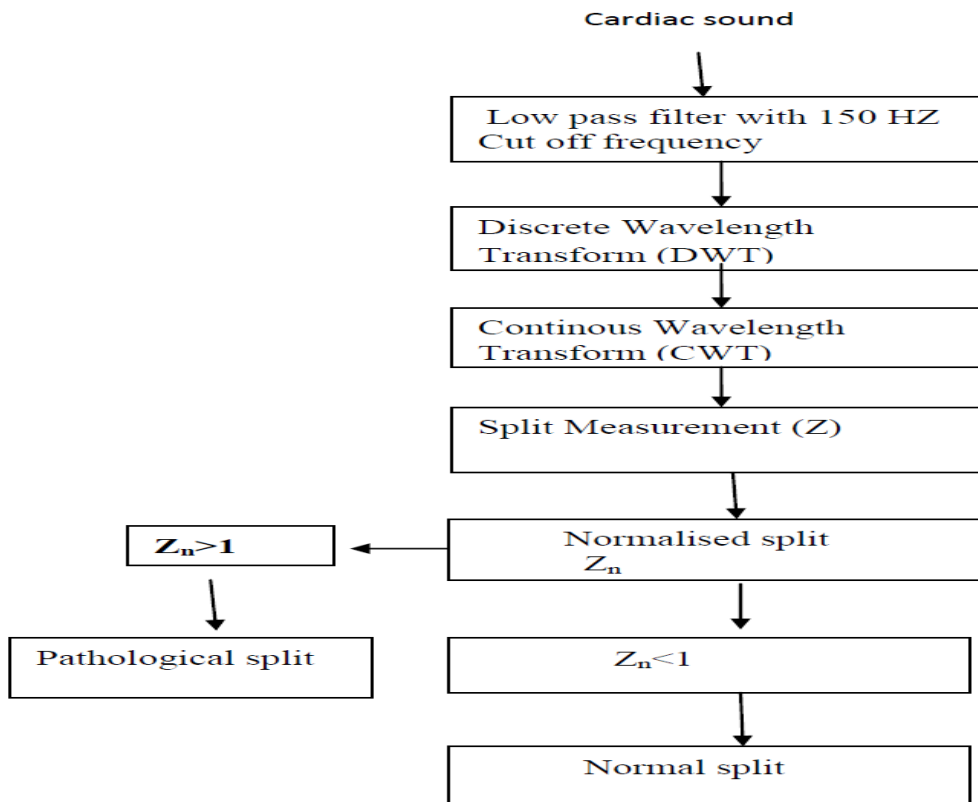
3.2 Algorithm

Cardiac sound

Low pass filter with 150 HZ Cut off frequency

Discrete Wavelength Transform (DWT)

Continous Wavelength Transform (CWT)

Split Measurement (Z)

Normalised split Z_n

$Z_n > 1$

Pathological split

$Z_n < 1$

Normal split

Fig. 3.1: Block diagram of the split measurement using wavelet based technique [1]

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(a) The discrete wavelet transform is used as a filtering tool for the cardiac sound [1].

(b) The result of stage a is used by the continuous wavelet transform. Continuous wavelet transform is used for measurement of the split S2 [1].

(c) Time delay between the components A2 and P2 is the split S.

(d) Relating the value of split to that of the normal split we can measure the normalized value Zn.

Zn is given as

Zn=measured split/30ms

Zn> 1 the split is the pathological split

Zn<1 the split is the normal split

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3.3 Wavelet transforms

Wavelet transforms are very useful tools in various signal processing application. For signal analysis continuous wavelet transform is most suitable technique [3].

Its fully discrete version (the DWT) and semi-discrete one (wavelet series (WS)) are used for various task for computer vision [4], image compression[5] and signal coding.

A wavelet allows multiresolution analysis in both time and frequency direction. At different resolution or scale wavelet process data. If we consider a signal with a large window we will consider gross (or average) feature. Similarly if we consider a signal with a small window we will consider detail feature. The short time Fourier has a fixed window. Thus by using varying resolution wavelet transform can solve the problem of short time Fourier transform[6]. Wavelet transform is two type continuous wavelet transform and discrete wavelet transform.

Fast wavelet transform shows very practical filtering algorithm. For many signal most important part is the low frequency content. It gives the signal identity. On the other hand high frequency content imparts flavor or nuance [1].

Let us consider human voice as an example , the voice sounds different if we remove the high frequency components still we can get what is being said. But we hear the gibberish if we

remove the low frequency frequency components. This is the reason that in wavelet analysis there is a approximation and details[1] .

 The detail is the high frequency components and low scale of the signal .The approximation is the high scale low frequency components of the signal [1].

The original signal s passes form low pass and high pass filter and two signals emerge: signal approximation "A" and signal details "D" [1].

The decomposition process can be repeated with successive approximation so many lower resolution components one signal can be broken down. This is called the "wavelet decomposition tree".

3.6 Wavelet analysis of the normal phonocardiogram

Stage-1

Figure 3.3 shows the application of the one cycle of the normal PCG

- Level d2 and d1 show the high frequency variation of the base line of PCG signal [1].
- Level d4 and d3 show the side of high frequency content of the sound S2compare to sound S1 [1].
- The component A2 of sound S2 appear better on level d6 and d5.
- Level d7 represents the two principal components M1 and T1 of sound S1 and two principal components A2 and P2 of sound S2 and allow one to measure the split in S2.

Stage2

Fig3.4a and b show that the sound S2 has high frequency content than the sound S1 [11]. Analysis of the sound S1 and S2 by the application of CWT after their separation or identification [9] are respectively displayed in fig 3.4 c and e. the contour plot of these figure are shown in fig 3.4d and f. These contour plot gives the information of internal components of the sound [10] .

S1 has two major components M1 and T1 due to the closure of mitral and tricuspid valves is shown in fig3.4 d. Similarly S2 has two major components A2 and P2 due to the closer of aortic and pulmonary valves is shown in fig3.4 f [10].For the normal heart sound A2 closes before P2 but in case of pathological sound P2 precedes A2 [2]. The time delay between the component A2 and P2 gives the split S. This split is useful to distinguish between the pathological and normal sound. In case of normal heart sound the split is less than 30ms [8]. Here we measure the split between A2 and P2 for normal PCG. The split is 6ms (table3.1).

We can conclude for the sound S2 that

1. A2 precedes P2 in time.
2. A2 have high frequency content than P2 [1].
3. The sound S2 have high frequency content than the sound S1 [1].

3.7 Six level decomposition

Heart sound signal is affected by various kind of noise which has temporal and spectral characteristics. Wavelet transform is applied to convert the corrupted signal into several levels. The purpose of decomposition process is to remove the decompose level which is seriously corrupted by the noise so as to improve the signal to noise ratio. In my project there are six level decomposition is use .figure 3 shows the graph of de-noise standard normal heart sound by db (1) wavelet from one to six level .
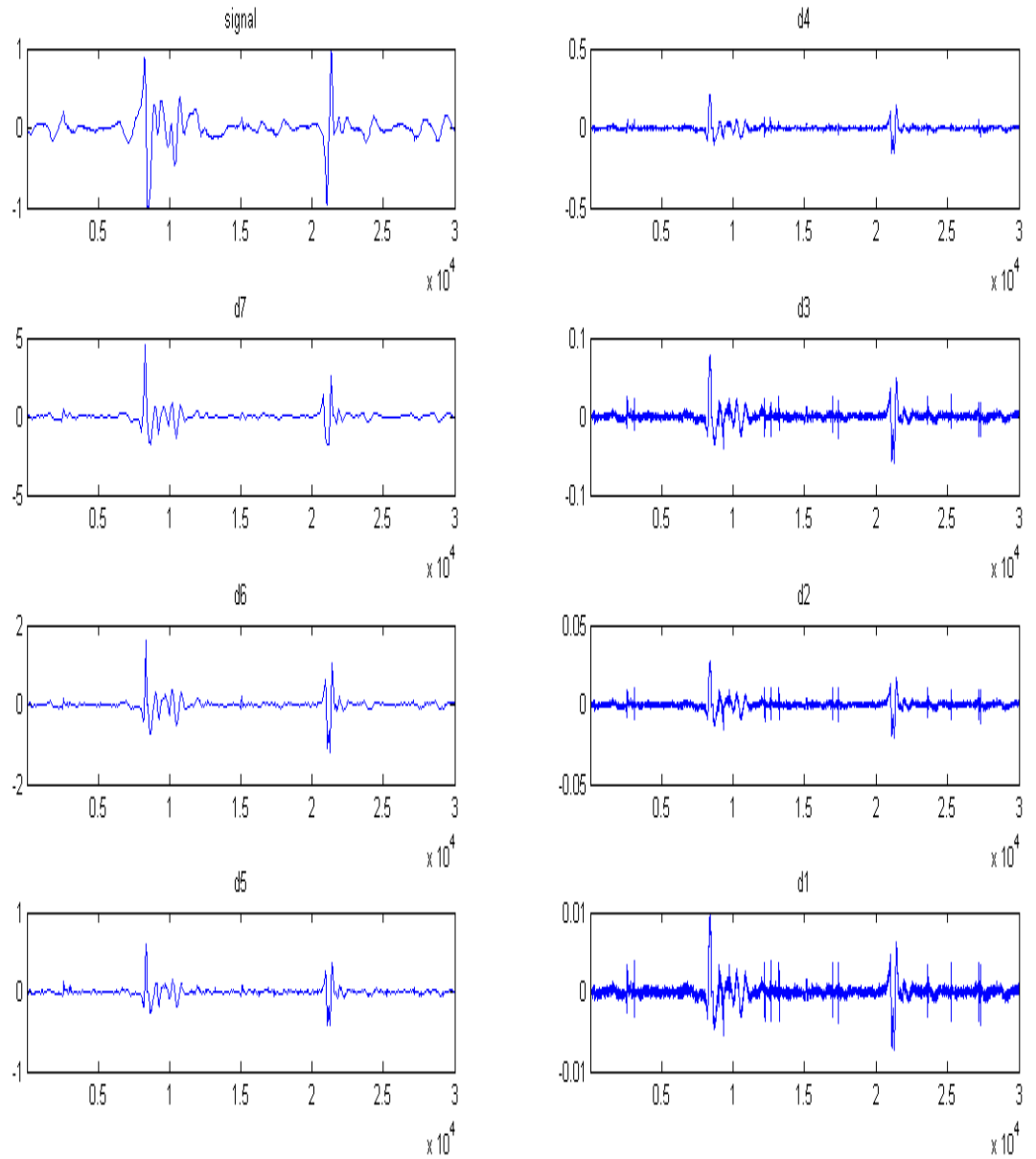
Figure3.1: wavelet decomposition of normal cardiac sound

3.8 Measurement of the split

Both S1 and S2 are transient signal containing very close frequency components [2]. Multisresolution analysis based on the discrete version of the wavelet transform is applied on

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this. Discrete wavelet transform with a multiresolution analysis which is used as a filtering technique in this work in order to locate and identify the components A2 ans P2 and measurement the split between them [1].

The phonocardiogram signals were normalized in energy to accommodate the disparityin magnitude arising out of diffirent amplification used at the time of signal collection and variation induced by the lead sites. DWT decompose the PCG signal in a different energy partition for resolution levels of the bit under study [1]. The split S2 appears in a finest level on the other hand high frequency information are localize on a coarser level. Detail 7 gives the best information of the split S2.

DWT which has a multiresolutiond analysis is a power ful tool in seperating , filtering and identification of the internal components and murmur of the various analysed signals [1].

In the representation of the coefficient C (of the WT) Y axis represent the scale (represent the frequency) and X-axis represent the position (represent the time) and colour at each x-y point represent the magnitude of the wavelet coefficient [1].

We use the sampling frequency 44100sample/sec. both the scale along x and y axis are a linear scale.

CWT is used in this work in order to measure the split between A2 and P2.Using the algorithm based on the WT the sound S2 is decomposed into number of component. The split S2 is measured and the component A2 and P2 are identified and selected. The normalised split Sn also calculated.
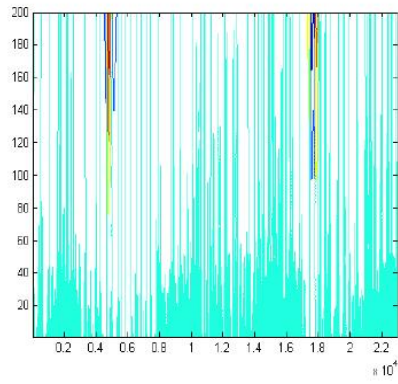
Zn>1 indicates the pathological split and

Zn<1 indicates the normal split

Zn is given as

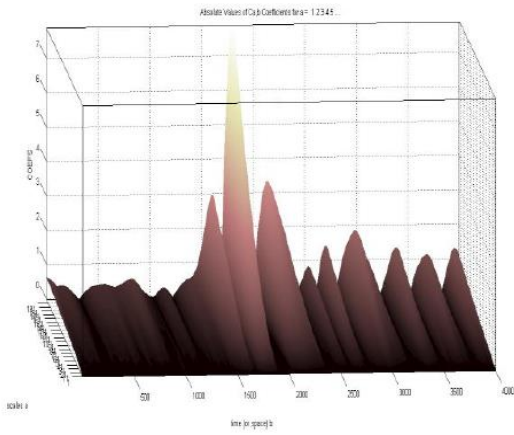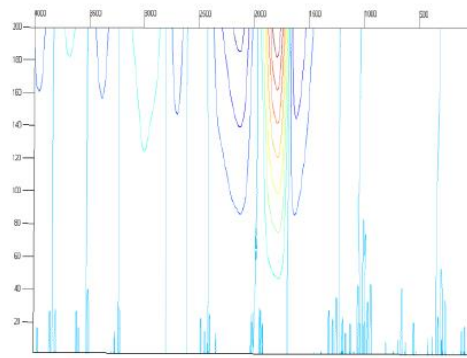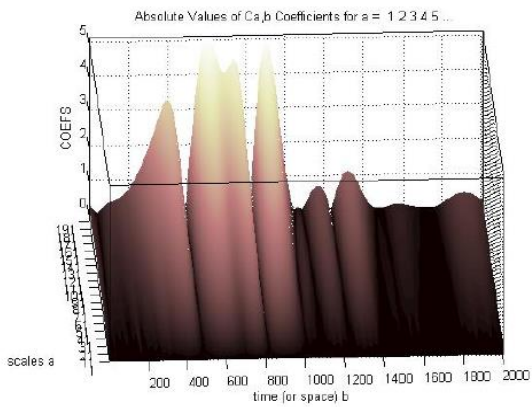Zn=measured split/30m [1]
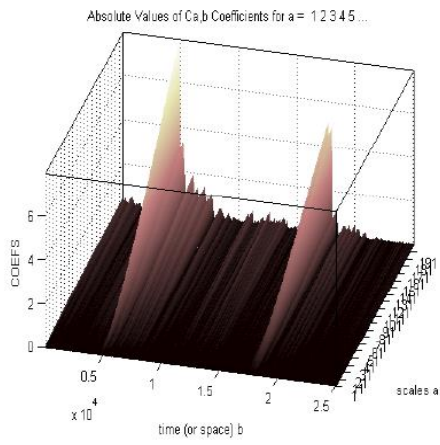
(a)

(b)

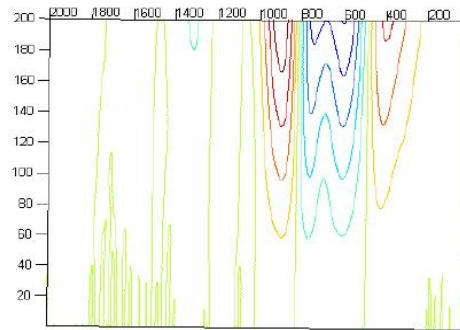(c)

(d)

(e)

(f)

Figure 3.2:wavelet transform for the normal phonocardiogram: (a) CWT for one cycle of the normal PCG; (b) contour plot of figure 4a; (c) CWT for the sound S1; (d) contour plot of figure 4c;(e) CWT for the sound S2; and (f) contour plot of fig 4e

Table 1

Temporal measurement of the component A2 and P2 of the sound S2

	Location(ms)	Delay	mormalised split=measured split/30ms	comment
A2	19ms	6ms	0.2	normal PCG signal
P2	13ms			

(a)

(b)

(c)

(d)

(e)

(f)
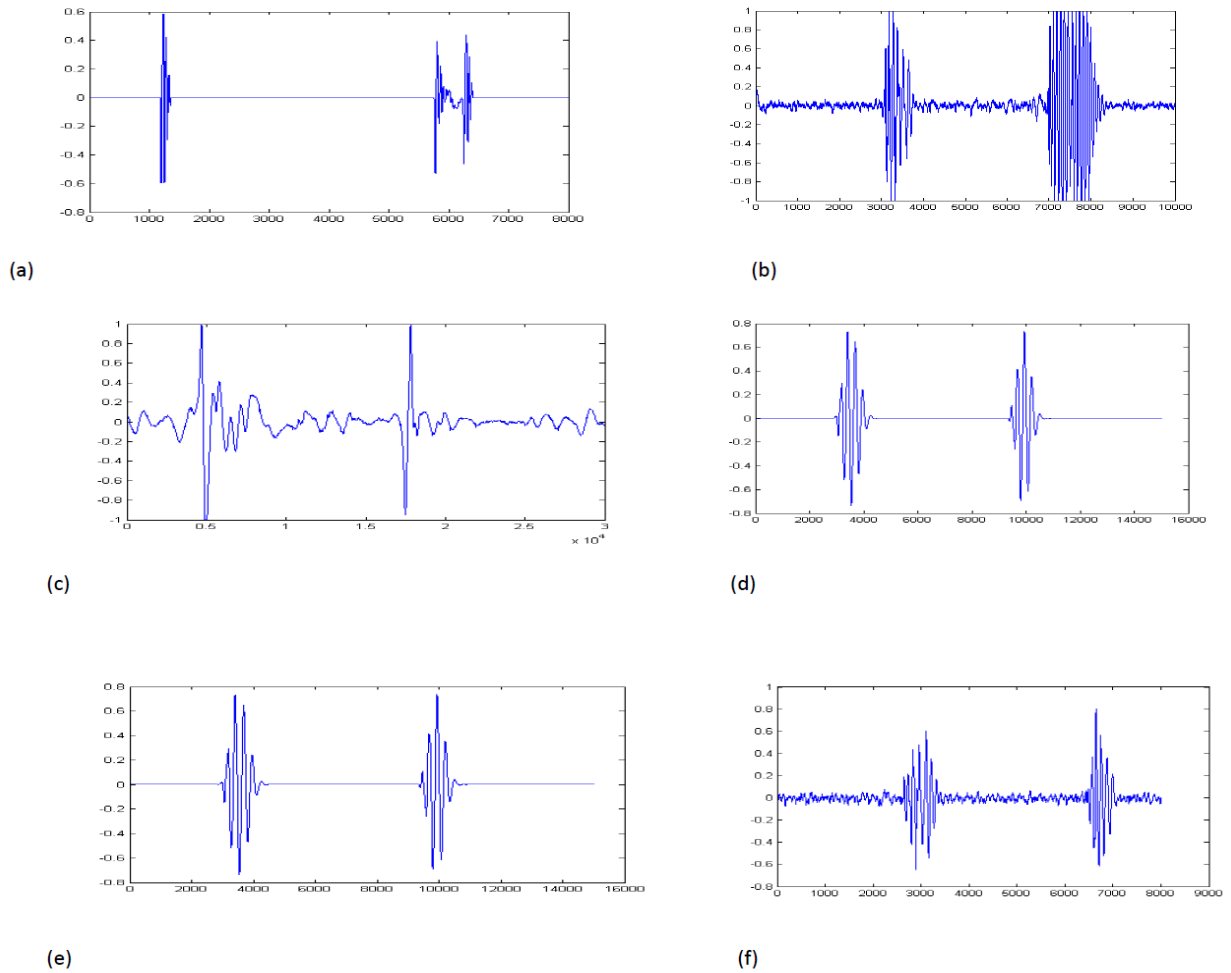
Figure 3.3: some phonocardiogram signal

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Table 2

Measurement of the split S2 of the PCG signal

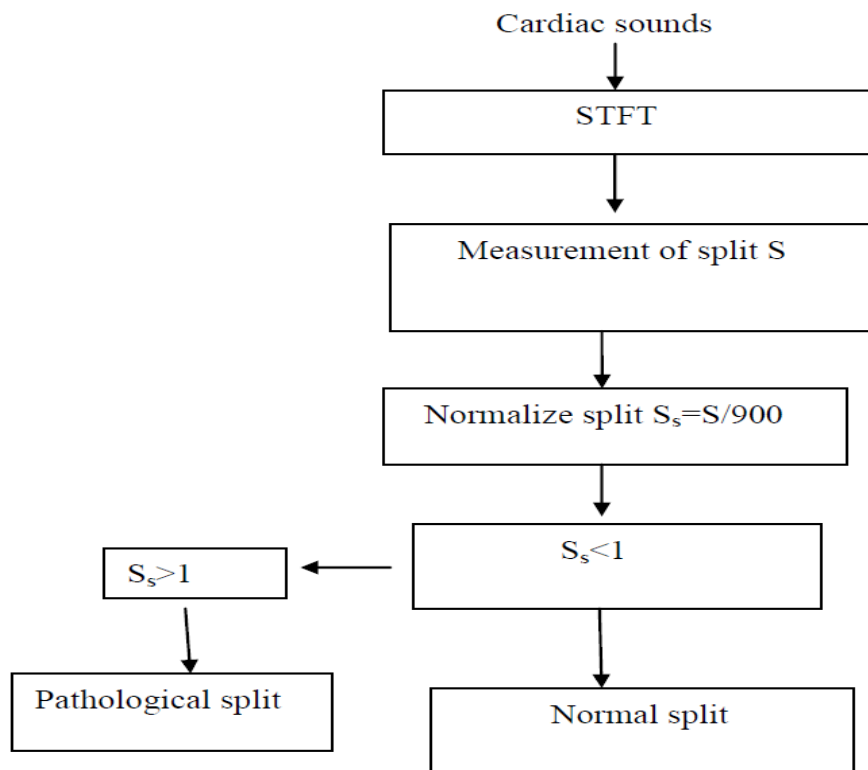| Type of PCG	a	b	c	d	e	f
Split (ms) S	50	55	20	5	8	7
Normalized split Zn	1.667	1.833	0.667	0.1667	0..267	0.233

3.9 Short time Fourier transform

Algorithm



Figure 3.4: Block diagram of measurement of split using STFT

(a) STFT is applied to the cardiac sound.

(b) After seperating and identification of the components A2and P2 we measure the split.

(c) The split is the time delay between the components A2 and P2.

(d) Relating the value of measured split to that value of the limit of the normal split (900s) we get the normalized value S_s
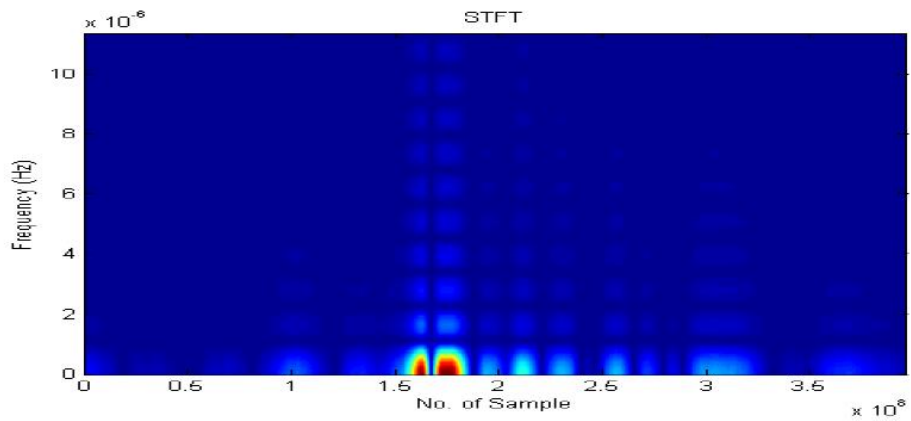
Ss is given by

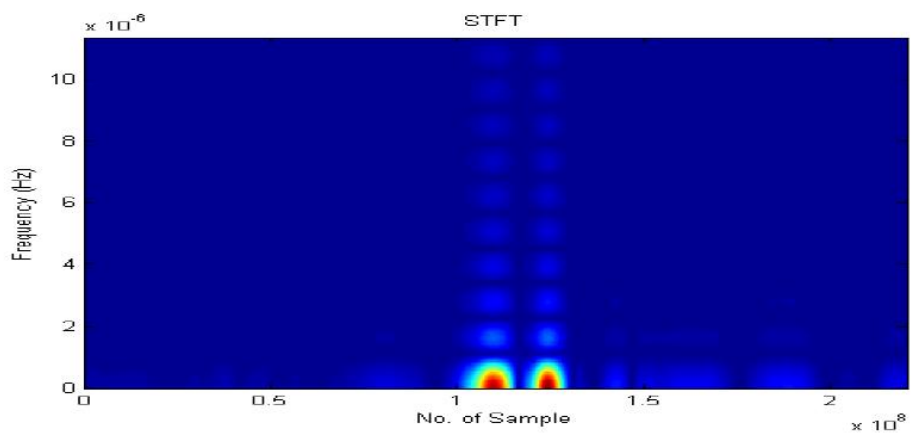Ss=measured split/900s

If Ss<1 the split is the normal split.

If Ss>1 the split is the pathological split
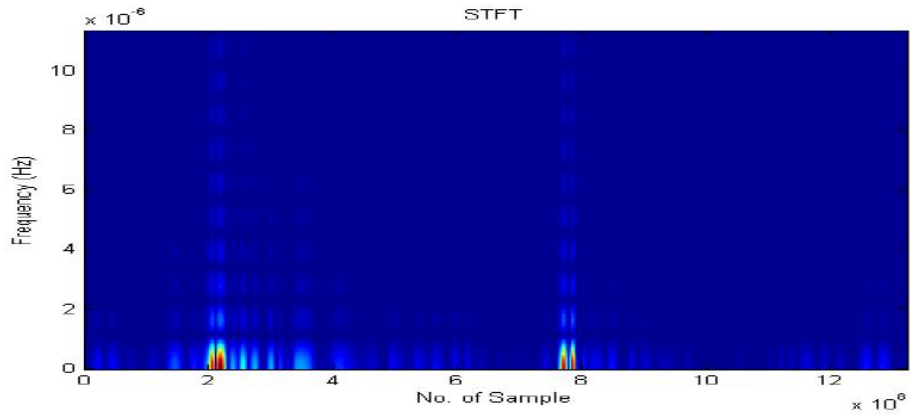
(a)

(b)

(c)

Figure 3.4: STFT of the normal phonocardiagram,(a)STFT of the one cycle of the normal

PCG,(b) STFT of the sound S1,(c) STFT of the sound S2

Table 3

Measurement of the split using STFT

	Location(sec)	delay	normalized split=measured split/900sec	comment
A2	2464	345	0.383	normal PCG signal
P2	2809			

Table 4

Measurement of the split in S2 of phonocardiagram using STFT

Type of signal	a	b	c	d	e	f
Split(sec) S	1896	2177	378	245	363	289
Normalized Split Ss	2.107	2.419	0.42	0.272	0.403	0.321

3.10 Conclusion

The cardiac cycle of the phonocardiogram is characterized by transient and fast change in frequency as the time progress [1]. To analyse such a non stationary signal WT is a suitable technique. The discrete wavelet transform which has a multiresolution analysis is used to decompose the signal into elementary building blocks and localize the split S2 between the components A2 and P2 [1].

After the localizition the split S2 continuous wavelet transform produce graphical representation that provides quantitive analysis in time and frequency direction [1].

We analysed the phonocardiogram signal by the use of STFT. STFT can distinguish the components A2 and P2. After seperation of A2 and P2 the split can be measured. Also the normalized split can be determined. If the normalized split is greater than one it is pathological split. If it is less than one it is normal split. We can see from the above results that wavelet transform gives better time frequency resolution than STFT.

3.11 Reference

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