Jimma University
School of graduate studies
Jimma Institute of Technology
School of Biomedical Engineering
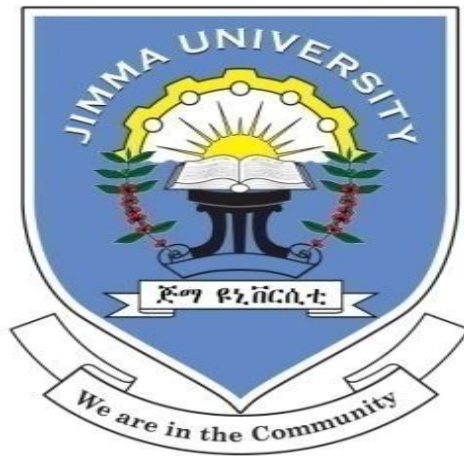# Bioinstrumentation Engineering Stream

IDENTIFICATION OF ABNORMAL HEART SOUNDS USING LabVIEW

A Thesis submitted to the School of Graduate Studies of Jimma University in Partial Fulfillment for the Degree of Master of Science in Biomedical (Bioinstrumentation) Engineering.

BY: Abreham Haddis

July, 2020

Jimma, Ethiopia

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BY: Abreham Haddis

Main advisor: Dr. T. Bheema Lingaiah

Co-advisor: Dr. Timothy Kwa

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Declaration

I, Abreham Haddis, declare that this research work titled, **"Identification of Abnormal Heart Sounds Using LabVIEW"** is my original work and has not previously been submitted for a degree or any other qualification at this University or any other institution

Abreham Haddis   _____   _____/_____/_____

Name   Signature   Date

On behalf of the School of Biomedical Engineering at Jimma Institute of Technology, we the advisors of this research with the title of **"Identification of Abnormal Heart Sounds Using LabVIEW"** and I, the evaluator, confirm that this research is approved as MSc. Thesis for the student

DR-T. BHEEMA LINGAIAH   _____   _____/_____/_____

Main Advisor   Signature   Date

DR-TIMOTHY KWA   _____   _____/_____/_____

Co-Advisor   Signature   Date

Examination Approval:

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Internal Examiner   Signature   Date

Dr- Dawit Anely___   _____   _____/_____/_____

External Examiner   Signature   Date

Abstract

Effective interpretation and diagnosis of heart sounds is vital part of daily life to Medical practitioners and health care professionals working in cardiac clinics requires ensuring that the information gathered from the patient is properly interpreted and categorized to its respective categories of Heart sounds(Normal and abnormal). The existing analysis techniques lacks the most important part needed in daily examination of electrocardiography, (Technique of measurement of heart electrical activity) where the rhythms obtained lively from the patient are displayed or recorded on paper and are not properly categorized which will determine the characteristic of cardiac muscle. The problem is how much effective the interpretation will be on real time detection and display. The expertise required is the time the patient's gets diagnosed and results for treatment plan, also at the same time and mostly how best junior medical practitioners can interpret the rhythms' independently. These gaps exist and increases patient long waiting time for results, sometimes Interpretation errors and reduce the confidence of junior practitioner.

This thesis work is typically done using virtual instrumentation LabVIEW (Laboratory virtual Instrument Engineering Work Bench) to improve the detection of abnormal Heart sounds by all medical practitioners regardless of their specialty level which reduces the interpretation error and time delay in routine outpatient service by creating cost effective abnormal Heart sound analysis technique. As the heart is divided in four chambers of atrium and ventricle in which mechanical activity produces sound signals which constitute the hearts Sound and in a wave form on a Phonocardiograph. This thesis work is done using Online Abnormal Heart sounds from Thinklabs, Physionet and where LabVIEW software is used for analysis.

Key words*: Abnormal Heart Sounds, Heart Chamber, Phonocardiograph, LabVIEW.*

Acknowledgement

First and foremost, I would like to thank almighty God for helping and guiding me throughout my research work and to reach this level. Secondly I would like to thank my advisor Dr T. Bheema Lingaiah for his continues and tireless support in the entire work and guidance especially on LabVIEW.

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Acronym

ADC	Analogue to digital converter
AI/AO	Analog Input output
AP	Aortic Pressure
CHF	Congestive Heart Failure
CT	Computed Tomography
CVD	Cardiovascular disease
DAQ	Data Acquisition Device
DAQmx	DAQ Measuring System
DDK	Driver Development Kit
DIO	Digital Input/output
DSP	Digital Signal Processing
ECG	Electrocardiogram
FIFO	First in First Out
FFTs	Fast Fourier Transform
GND	Ground
HP	High Pass Filter
HR	Heart Rate
HS	Heart Sound
IEEE	Institute of Electrical & Electronics Engineers
JU	Jimma University
JIT	Jimma Institute of Technology
JUMC	Jimma University Medical Center
LabVIEW	Laboratory Virtual Instrument Engineering Work bench
LEDs	Light Emitting Diodes
LP	Low Pass Filter
MAX	Measurement and Automation explorer
MRA	Multi Resolution Analysis
MRI	Magnetic Resonance Imaging
NI	National Instrumenation

PCG …………………..……………………………….……………..Phonocardiogram

PLCs ……………………………..………...………..….… Programmable Logic Controller

PDA………………………………………………..………… Patent ductus Arteriosus

PFI ……………………………………...……..……Programmable Functional Interface

S1 …………………………………….……..……...…..………. First Heart Sound

S2 ………………………………………….....….………..…… Second Heart Sound

S3 ………………………………….……………...…..…..…..…Third Heart Sound

S4 ………………………………………...…..………..……..…… FourthHeart Sound

STFT ………………………………….…….….………Short Time Fourier Transform

VI ……………………………………………………...……… Virtual Instrument

WT………………………………………….….…………….Wavelet Transform

WVD………………………………………………...…Discrete Wavelet Transform

Chapter 1

Introduction

1.1 Back Ground of the Study

1.1.1 Anatomy of heart

The heart is the muscular organ of the circulatory system that constantly pumps blood through the body. Approximately the size of a clenched fist, the heart is composed of cardiac muscle tissue that is very strong and able to contract and relax rhythmically through a person's life time. The heart has four compartments or chambers. The upper chamber on each side is atrium and the lower part is called Ventricle. The heart collect deoxygenated blood from all parts of the body and oxygenated blood back to the body from the lung.

The heart is located between the lungs in the middle of the chest, behind and slightly to the left of the breast bone (sternum). A double-layered membrane called the pericardium surrounds the heart like a sack. The outer layer of the pericardium surrounds the roots of the heart's major blood vessels and is attached by ligaments to the spinal column, diaphragm, and other parts of the body. The inner layer of the pericardium is attached to the heart muscle. A coating of fluid separates the two layers of membrane, letting the heart move as it beats, yet still be attached to the body.

Because the heart is composed primarily of cardiac muscle tissue that continuously contracts and relaxes, it must have a constant supply of oxygen and nutrients. The blood leaving the left ventricle exits through the aorta, the body's main artery. The two coronary arteries, referred to as the "left" and "right" coronary arteries, emerge from the beginning of the aorta, near the top of the heart.

The coronary arteries are the network of blood vessels that carry oxygen and nutrient rich blood to the cardiac muscle tissue [1].Figure 1 shows anatomy of the heart major blood vessels, structures and its chambers.
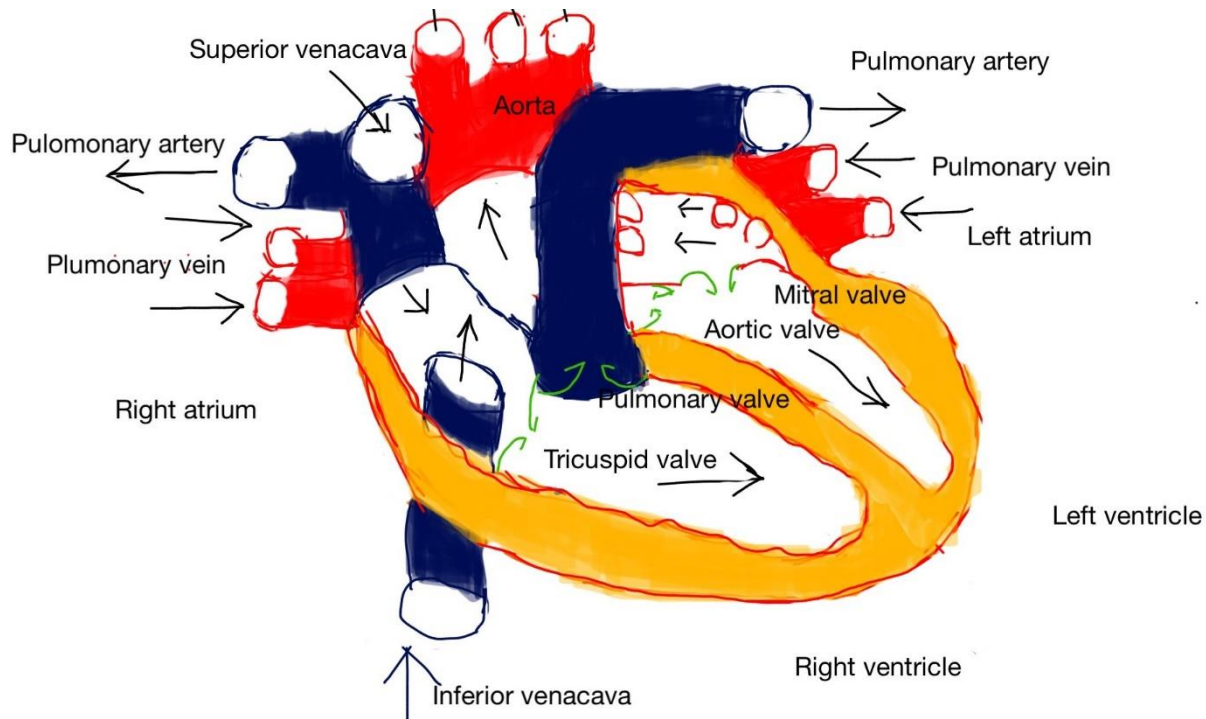
Figure1: Anatomy oftheheart

The initial segment of the left coronary artery is called the left main coronary. This blood vessel is approximately the width of a soda straw and is less than an inch long. It branches into two slightly smaller arteries: the left anterior descending coronary artery and the left circumflex coronary artery. The left anterior descending coronary artery is embedded in the surface of the front side of the heart. The left circumflex coronary artery circles around the left side of the heart and is embedded in the surface of the back of the heart.

Just like branches on a tree, the coronary arteries branch into progressively smaller vessels. The larger vessels travel along the surface of the heart; however, the smaller branches penetrate the heart muscle. The smallest branches, called capillaries, are so narrow that the red blood cells must travel in single file. In the capillaries, the red blood cells provide oxygen and nutrients to the cardiac muscle tissue and bond with carbon dioxide and other metabolic waste products, taking them away from the heart for disposal through the lungs, kidneys and liver.

When cholesterol plaque accumulates to the level of blocking the flow of blood through a coronary artery, the cardiac muscle tissue fed by the coronary artery beyond the point of the blockage is deprived of oxygen and nutrients.

This area of cardiac muscle tissue ceases to function properly. The condition when a coronary artery becomes blocked causing damage to the cardiac muscle tissue is called a myocardial infarction or heart attack.

A. Right Atrium

The right atrium receives de-oxygenated blood from the body through the superior vena cava (head and upper body) and inferior vena cava (legs and lower torso). The Sinoatrial node sends an impulse that causes the cardiac muscle tissue of the atrium to contract in a coordinated, wave-like manner. The tricuspid valve, which separates the right atrium from the right ventricle, opens to allow the de-oxygenated blood collected in the right atrium to flow into the right ventricle.

B. Right Ventricle

The right ventricle receives de-oxygenated blood as the right atrium contracts. The pulmonary valve leading into the pulmonary artery is closed, allowing the ventricle to fill with blood. Once the ventricles are full, they contract. As the right ventricle contracts, the tricuspid valve closes and the pulmonary valve opens. The closure of the tricuspid valve prevents blood from backing into the right atrium and the opening of the pulmonary valve allows the blood to flow into the pulmonary artery toward the lungs.

C. Left Atrium

The left atrium receives oxygenated blood from the lungs through the pulmonary vein. As the contraction triggered by the Sinoatrial node progresses through the atria, the blood passes through the mitral valve into the left ventricle.

D. Left Ventricle

The left ventricle receives oxygenated blood as the left atrium contracts. The blood passes through the mitral valve into the left ventricle. The aortic valve leading into the aorta is closed, allowing the ventricle to fill with blood. Once the ventricles are full, they contract. As the left ventricle contracts, the mitral valve closes and the aortic valve opens. The closure of the mitral valve prevents blood from backing into the left atrium and the opening of the aortic valve allows the blood to flow into the aorta and flow throughout the body.

E. Papillary Muscles

The papillary muscles attach to the lower portion of the interior wall of the ventricles. They connect to the chordae tendineae, which attach to the tricuspid valve in the right ventricle and the mitral valve in the left ventricle. The contraction and relaxation of the papillary muscles opens and closes these valves respectively.

F. Chordae Tendineae

The chordae tendineae are tendons linking the papillary muscles to the tricuspid valve in the right ventricle and the mitral valve in the left ventricle. As the papillary muscles contract and relax, the chordae tendineae transmit the resulting increase and decrease in tension to the respective valves, causing them to open and close. The chordae tendineae are string-like in appearance and are sometimes referred to as "heart strings."

G. Tricuspid Valve

The tricuspid valve separates the right atrium from the right ventricle. It opens to allow the de-oxygenated blood collected in the right atrium to flow into the right ventricle. It closes as the right ventricle contracts, preventing blood from returning to the right atrium; thereby, forcing it to exit through the pulmonary valve into the pulmonary artery.

H. Mitral Value

The mitral valve separates the left atrium from the left ventricle. It opens to allow the oxygenated blood collected in the left atrium to flow into the left ventricle. It closes as the left ventricle contracts, preventing blood from returning to the left atrium; thereby, forcing it to exit through the aortic valve into the aorta.

I. Pulmonary Valve

The pulmonary valve separates the right ventricle from the pulmonary artery. As the ventricles contract, it opens to allow the de-oxygenated blood collected in the right ventricle to flow to the lungs. It closes as the ventricles relax, preventing blood from returning to the heart.

J.   Aortic Valve

The aortic valve separates the left ventricle from the aorta. As the ventricles contract, it opens to allow the oxygenated blood collected in the left ventricle to flow throughout the body. It closes as the ventricles relax, preventing blood from returning to the heart.

1.1.2 Heart Sound Production

The mechanical activities of the heart during each cardiac cycle cause the production of some sounds, which are called heart sounds. Heartsounds area normal and involuntary process. In cardiac auscultation, they can be easily heard by placing a stethoscope on a patient's chest. The 'Lub' is the first heart sound or the (S1), while 'Dub' is second heart sound or the (S2). Basically these sounds are produced by the rhythmic contraction and relaxation or opening and closure of valves such as bicuspid/mitral, tricuspid, the aortic and the pulmonic valves as well as blood flow and its viscosity. When there are any changes in these sounds, they are termed as heart murmurs and adventitious sounds; S3 and S4 respectively. These are also called gallop rhythms. These sounds can be divided into two types - normal heart sounds andabnormal heart sounds.

1.1.2.1 Primary Heart Sound

Normal heart sounds are associated with heart valves closing, causing changes in blood flow.

S1 (First Heart Sound)

The first heart tone, or S1, forms the "Lub" of "Lub-Dub" and is composed of components M1 and T1. Normally M1 precedes T1 slightly. It is caused by the sudden block of reverse blood flow due to closure of the atrioventricular valves, i.e. tricuspid and mitral (bicuspid), at the beginning of ventricular contraction, or systole. When the ventricles begin to contract, so do the papillary muscles in each ventricle. The papillary muscles attached to the tricuspid and mitral valves via chordae tendineae, bring the cusps or leaflets of the valve closed (chordae tendineae also prevent the valves from blowing into the atria as ventricular pressure rises due to contraction).
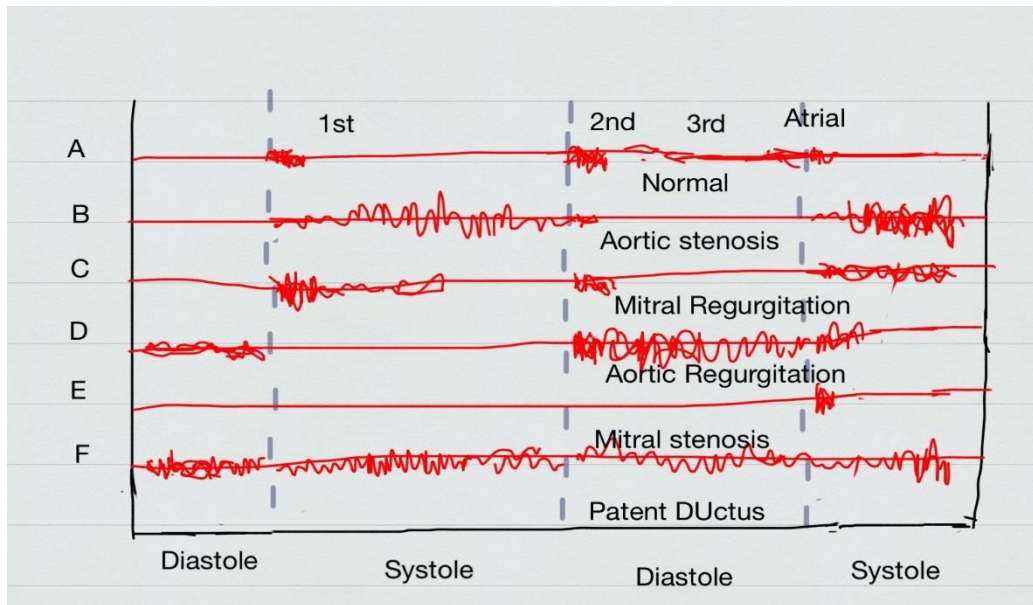
Figure 2: Phonocardiography from normal and abnormal heart sound

Phonocardiograph in figure 2 above shows graphical representation of heart sounds. The closing of the inlet valves prevents regurgitation of blood from the ventricles back into the atria. The S1 sound results from reverberation within the blood associated with the sudden block of flow reversal by the valves [2]. First heart Sound lasts long but it does have low frequency ranging from 20 Hz to 140 Hz.

S2 (Second Heart Sound)

The second heart tone, or S2, forms the "dub" of "Lub-dub" and is composed of components A2 and P2. Normally A2 precedes P2 especially during inspiration when a split of S2 can be heard. It is caused by the sudden block of reversing blood flow due to closure of the aortic valve and pulmonary valve at the end of ventricular systole, i.e. beginning of ventricular diastole. As the left ventricle empties, its pressure falls below the pressure in the aorta and aortic blood flow quickly reverses back toward the left ventricle, catching the aortic valve pocket like cusps and is stopped by aortic (outlet) valve closure. Similarly, as the pressure in the right ventricle falls below the pressure in the pulmonary artery, the pulmonary (outlet) valve closes. The S2 sound results from reverberation within the blood associated with the sudden block of flow reversal.

Splitting of S2, also known as physiological split, normally occurs during inspiration because the decrease in Intrathoracic pressure increases the time needed for pulmonary pressure to exceed that

of the right ventricular pressure. A widely split S2 can be associated with several different cardiovascular conditions, including right bundle branch block and pulmonary stenosis.

1.1.2.2 Extra Heart Sound

The rarer extra heart sounds form gallop rhythms and is heard in both normal and abnormal situations.

S3 (Third Heart Sound)

Rarely, there may be a third heart sound also called a protodiastolic gallop, ventricular gallop. It occurs at the beginning of diastole after S2 and is lower in pitch than S1 or S2, as it is not of valvular origin. The third heart sound is benign in youth, some trained athletes, and sometimes in pregnancy but if it re-emerges later in life it may signal cardiac problems like a failing left ventricle as in dilated congestive heart failure (CHF). S3 is thought to be caused by the oscillation of blood back and forth between the walls of the ventricles initiated by inrushing blood from the atria. The reason the third heart sound does not occur until the middle third of diastole is probably because during the early part of diastole, the ventricles are not filled sufficiently to create enough tension for reverberation.

It may also be a result of tensing of the chordae tendineae during rapid filling and expansion of the ventricle. In other words, an S3 heart sound indicates increased volume of blood within the ventricle. An S3 heart sound is best heard with the bell-side of the stethoscope (used for lower frequency sounds). A left-sided S3 is best heard in the left lateral decubitus position and at the apex of the heart, which is normally located in the 5th left intercostal space at the mid-clavicular line. A right-sided S3 is best heard at the lower-left sternal border. The way to distinguish between a left and right-sided S3 is to observe whether it increases in intensity with inspiration or expiration. A right-sided S3 will increase on inspiration whereas a left-sided S3 will increase on expiration.

S4 (Fourth Heart sound)

The rare fourth heart sound when audible in an adult is called a presystolic gallop or atrial gallop. This gallop is produced by the sound of blood being forced into a stiff/hypertrophic ventricle. It is a sign of a pathologic state, usually a failing left ventricle, but can also be heard in other conditions such as restrictive cardiomyopathy. The sound occurs just after atrial contraction ("atrial kick") at the end of diastole and immediately before S1, producing a rhythm sometimes referred to as the "Tennessee" gallop where S4 represents the "Ten-" syllable. It is best heard at the cardiac apex with the patient in the left lateral decubitus position and holding his breath. The combined presence of S3 and S4 is a quadruple gallop, also known as the "Hello-Goodbye" gallop. At rapid heart rates, S3 and S4 may merge to produce a summation gallop (sometimes referred to as S7).

1.1.2.3 Murmurs

Heart murmurs are produced as a result of turbulent or irregular flow of blood, turbulence sufficient to produce audible noisy signals. They are usually heard as a whooshing sound. The term murmur only refers to a sound believed to originate within blood flow through or near the heart; rapid blood velocity is necessary to produce a murmur. Yet most heart problems do not produce any murmur and most valve problems also do not produce an audible murmur.

The following paragraphs overview the murmurs most commonly heard in adults who do not have major congenital heart abnormalities, the research utilized Aortic and Mitral stenosis as well as regurgitation.

A. Aortic Regurgitation

- Characterized by the presence of murmur during the diastole between S2 & S1 sound to the beginning point of the first sound.
- Aortic Regurgitation is the back flow of blood from the aorta into the left ventricle. Regurgitation is due to incompetence of the aortic valve resulting in diastolic flow of blood into the left ventricular chamber.
- It is often described as a relatively high pitched murmur of blowing character.
- May be loud or soft depending on the severity of the regurgitation, with more severe aortic regurgitation the murmur becomes longer, and will usually decrease in intensity

B. Mitral Stenosis

- Occurrence of murmur during diastole after completion of the second sound in which prolonged murmurs are observed. Sometimes these may be mixed with third sound where the third sound is impossible to detect.
- Depending on the stenosis condition the amplitude varies.
- The complex noise is observed for the long interval about 100 milliseconds after the end of the second sound.
- No pulses are observed between the first and second sounds.

C. Aortic stenosis

- Aortic Stenosis is a condition where the valve fails to open properly. The opening of the valve is narrow and causes a restriction in the flow of blood from left ventricle to the aorta.
- Characterized by the presence of pure systolic murmur
- Start shortly after the first heart sound and ends before the second heart sound. The murmur is mid systolic ejection murmur, heard best over the "aortic area".
- The amplitude at the starting point is high and it is very difficult to distinguish between starting of the first sound and starting of the murmur. The murmur is often described as harsh.

D. Mitral Regurgitation

- Also called Mitral insufficiency or Mitral incompetence.
- In mitral regurgitation the valve does not closely properly
- The amplitude of the murmur is found to be high, immediately after the first sound and decreases at the beginning of the second sound.
- Causes blood to leak back (regurgitate) into the left atrium when the left ventricles contracts.
- Basically, the more open the valve remains, the more blood regurgitate, the more severe the problem will be.
- This increases the pressure in the left atrium and in the blood vessels coming from the lungs
- Regurgitation through the mitral valve is by far the most commonly heard murmur, producing a pansystolic murmur which is sometimes fairly loud to a practiced ear, even though the volume of regurgitate blood flow may be quite small. Yet, though obvious using

echocardiography visualization, probably about 20% of cases of mitral regurgitation does not produce an audible murmur.

- Stenosis of the aortic valve is typically the next most common heart murmur, a systolic ejection murmur. This is more common in older adults or in those individuals having a two, not a three leaflet aortic valve.

- Regurgitation through the aortic valve, if marked, is sometimes audible to a practiced ear with a high quality, especially electronically amplified, stethoscope. Generally, this is a very rarely heard murmur, even though aortic valve regurgitation is not so rare. Aortic regurgitation, though obvious using echocardiography visualization, usually does not produce an audible murmur.

- Stenosis of the mitral valve, if severe, also rarely produces an audible, low frequency soft rumbling murmur, best recognized by a practiced ear using a high quality, especially electronically amplified, stethoscope.

- Either regurgitation through, or stenosis of, the tricuspid or pulmonary valves essentially never produces audible murmurs.

- Other audible murmurs are associated with abnormal openings between the left ventricle and right heart or from the aortic or pulmonary arteries back into a lower pressure heart chamber.

1.1.3 Phonocardiograph

Phonocardiograph is an instrument used for recording the heart sounds. These sounds provide an indication of the heart rate and its rhythmicity. They also give useful information regarding effectiveness of blood pumping and valve action.

A graphical representation of the waveform of cardiac sounds is called a phonocardiogram, and the technique used to capture the sound is referred to as phonocardiography. This technique allows a visual interpretation of the cardiac sounds, thus allowing thorough investigation of temporal dependencies between mechanical processes of the heart sounds produced. A number of other heart related variables may also be recorded with the phonocardiogram, including the ECG, carotid arterial pulse, jugular venous pulse, and apex cardiogram simultaneously. Such information allows clinicians to evaluate the heart sounds of a patient with respect to the electrical and mechanical events of the cardiac cycle.

1.1.3.1 Problems associated with heart sound

Most of the heart sound information lies in the frequency range of 20-150 Hz [3], but murmur sounds have a higher frequency range of 600 Hz [4]. Abnormal functionality of the heart must be diagnosed in early stage as it is very critical, it can be done by examination using auscultation technique. It allows listening and analyzes heart sound. Phonocardiograph is an electronic device which provides digital recording of heart sounds. Phonocardiogram carries useful information about heart status and functionality. Cardiovascular diseases are censorious and must be detected with no time delay [5]. The difference in the pattern can be found between a normal heart sound signal and abnormal heart sound signal as their PCG signal varies from each other with respect to time, amplitude, intensity, homogeneity, spectral content, etc.[6]. The duration, pitch and shape of the heart sound show us details about the different conditions the heart [7]. The closure of Tricuspid valve ranges 20 to 30ms. If the duration is greater than 200ms it is called split with frequency ranging 40 to 200 Hz and it is considered medically fatal to the person.

Heart sound has bigger potential expressing the underlying problem that could threaten our life in which valvular narrowing, prolapse, murmurs, congestive heart failure, patient Ductus arterioles, regurgitations, splits, aneurism and ejections can be easily isolated from one another by visual aid during examination by their characteristics pitch frequency time and shape. As they have different source of origin and reference it is not easy to categorize merely by auscultation using stethoscope as well as phonocardiogram.

The recorded signals analysis has to be fast and reliable to identify abnormal rhythms. Signal processing has contributed significantly to a new understanding of ECG and its dynamic properties as expressed by changes in rhythm and beat morphology.

1.1.3.2 Algorithm and Equipment's used to diagnose Heart Sounds

Heart sounds are basically diagnosed by auscultation techniques; Stethoscope is the most commonly used device. The essential technology for listening ("auscultation") of heart sounds is still the basic stethoscope. Electronic stethoscopes are available, but they have had little impact. Heart-sound auscultation is both a science and an art. The science of the sounds that relate to a specific heart problem has been well established. The ability to discriminate the sounds or to recognize the sounds that are being heard is perhaps the art, although time-frequency techniques have been employed to analyze the recorded heart sounds [8].

The conventional stethoscope based auscultation technique is a cost effective and non- invasive diagnostic procedure. This technique is very popular among the physicians and commonly used by them since 1816[9]. However, the performance of this diagnostic technique degrades due to the presence of noise in lung sound signals. Modern electronic stethoscope can reduce the ambient noises from lung sounds, but they are inefficient to avoid heart sound noise.

1.2 Statement of the Problem

Heart sounds are known to be non-stationary and have very wide varying characteristics and it is hard to differentiate just only by auscultation using ordinary stethoscope. The varying characteristics of the heart produce abnormal heart sounds which must be identified at the earliest. Globally 31% of all death each year is related to heart diseases. In addition to this percentage abnormal heart itself is a big problem on its own. Since in depth skill and proper hearing capacity is required by each health professional identification becomes a question to diagnose the patient on his immediate visit to health care facility as detection is also another problem. It is seen that the non-stationary heart sounds either normal or abnormal do have their own characteristics which is closely related to one another with time, pitch, shape frequency and occurrence but under visual aid or display they can be easily identified from one another. The skill required to identify and analyze the heart sounds both normal and abnormal by visualization is much easier than the audio as there are many different types of abnormal heart sounds in addition to the normal with a slight variation in return each of them have their own impact on the patient life.

1.3 Objective and Hypothesis of the study

1.3.1 General Objective
➤ The main objective of the study is to identify abnormal heart sounds using LabVIEW in a graphical wave form.

1.3.2 Specific Objectives
➤ Identification of abnormal heart sounds
➤ Classifying the detected abnormal heart sounds in relation to normal heart sound and to compare Auscultation with graphical display
➤ To design a Virtual instrumentation system using LabVIEW

1.3.3 Research Hypothesis

It is possible to distinguish and identify abnormal heart sounds using LabVIEW better than other existing techniques.

1.4 Significance of the study

Health is the most important and basic primary needs for human being, Productivity will be secured in the presence of adequate healthcare coverage institution with well-equipped and trained manpower, which will assure improved quality and quantity service to the community in turn it will let the citizens get engaged on production. Health institutions at various levels have different professionals and equipment ownership for diagnostic and therapeutic purpose. As technology advances the life style gradually changing and different disease conditions occurs in all age groups including young teenager's diseases like cancer, diabetic and Heart disease are spreading due to eating bad habit, limited sporting activity and environmental changes. Problems related to heart disease need early detection and follow-ups due to the fact that they are the soul of existence for most of the problem encountered in daily lives of human being. Abnormal heart rhythm need Cardiologist in each and every health institution to counter act and prevent further disease and complication from undiagnosed abnormal cardiac rhythms. It is obvious that all the patients will not have full access to be seen by cardiologist on prompt visit to health institution as this is expensive and less likely to avail a sub-specialized personnel or Cardiologists in every facility. On emergency visit to health institution the frontline examiners will be Nurses and junior physicians who are limited to basic Auscultation, ECG machine and shallow knowledge to interpret rhythms appropriately to provide service timely and quality wise to treat patients on their arrival at the health institution.

These heart diseases are disorders of the heart rhythms and blood vessels, and they can be detected from ECG bio signals. The bio signals are signals from human bodies and various researches has been carried out for performing analysis on the measured signals for diagnosis and treatment purpose.

The thesis work what is being done is to categorize the heart beats detected and help the medical professional to identify the rhythms of the heart easily as there are different types of abnormal heart sound difficult to recall and remember clinically. This work is implemented using virtual

instrumentation system using LabVIEW to facilitate smooth work flow, timely and quality health service delivery by Emergency and cardiac unit as well it will make analysis of abnormal cardiac rhythm much easier for Junior physician and Nurse. Additionally, utilization of the software will save the life of patients who visit and wait at emergency and cardiac unit by prompt diagnostic result output on immediate patient visit.

1.5 Scope of the Study

The study was particularly aimed to identify abnormal heart sound using LabVIEW. The data used is online heart sound data from a clinical database Thinklabs.

1.6 Organization of Study

The study is segregated into five chapters where chapter one deals with the heart disease in general related to heart sound, chapter two different Literature review, chapter three Materials and methodology of the research, chapter four result and discussion and whereas chapter five conclusion and recommendation.

Chapter Two

Literature Review

There are advanced signal processing technologies that hold promise for developing computer-aided auscultation solutions that are intuitive, efficient, informative and accurate. Computer-aided auscultation [10]offers an objective, quantitative and cost-effective tool for acquiring and analyzing heart sounds, providing archival records that support the patient evaluation and referral decision as well as serial comparisons for patient monitoring. There is further promise of new quantitative acoustic measures and auscultatory findings that have more precise correlation with underlying physiological parameters. These solutions are being developed with the benefits of a rich literature of clinical studies in phonocardiography, the added insights derived from echocardiography, and advances in signal processing technology.

Reza, developed Phonocardiogram method to analyze [11]. They used color spectrographic technique. It is a technique recommended for amplifying non-invasive cardiac monitoring.

Wavelet-based segmentation [12]algorithm is quite effective in localizing the important components of both normal and abnormal heart sounds. They demonstrated that wavelet-based analysis, extracting vectors which are clearly differentiable for automatic classification of heart sounds.

Obaidat MS.et al [13]presented the applications of the spectrogram; Wigner distribution and wavelet transform analysis methods to the PCG signals. A comparison between these three methods has shown the resolution differences between them. It is found that the wavelet transform is capable of detecting the two components, the aortic valve component A2 and pulmonary valve component P2, of the second sound S2 of a normal PCG signal. Furthermore, the wavelet transform provides more features and characteristics of the PCG signals that will help physicians to obtain qualitative and quantitative measurements of the time-frequency characteristics.

Durand LG, Pibarot P., et al [14] presented application of spectral analysis and the potential of new time-frequency representations and cardiac acoustic mapping to resolve the controversies and better understand the genesis and transmission of heart sounds and murmurs within the heart-thorax acoustic system are reviewed. The most recent developments in the application of linear predictive coding, spectral analysis, time-frequency representation techniques, and pattern

recognition for the detection and follow-up of native and prosthetic valve degeneration and dysfunction are also presented in detail. New areas of research and clinical applications and areas of potential future developments are then highlighted.

L.Khadra, M. Matalgah, B. El-Asir and S. Managed et al [15] used wavelet transform, which is the decomposition of a signal into a set of independent frequency channels, is shown to be a useful diagnostic tool in the analysis of heartbeat sounds. In particular, the wavelet transform enables the experimentalist to obtain qualitative and quantitative measurements of time-frequency characteristics of phonocardiogram (PCG) signals.

Tranulis C, Durand LG, Senhadji L, Pibarot P. et al [16] Developed a method to conduct a study on non-invasive method for the estimation of pulmonary arterial pressure (PAP) using a neural network (NN) and features extracted from the second heart sound (S2).

[17]Developed a method to map a one-dimensional function of time (or frequency), the time-frequency representation can localize the signal energy in both the time and frequency directions. It has been shown that many biomedical signal problems may benefit from time-frequency analysis. The objective of this paper is to review the advances in time-frequency analysis of biomedical signals. Relevant theoretical methodologies and practical considerations are introduced, and five application areas are reviewed: electroencephalography (EEG), electrocardiography, phonocardiography, electrogastrography, and electromyography.

Recently [18] studied the phono-spectrographic features of heart murmurs to investigate and to distinguish innocent murmurs from pathological murmurs in children.

[19]Developed an approach based on spectrograms to present an intensity/frequency image of heart sound properties over a period of time, which they called heart energy signature spectrogram.

2.1 Review of LabVIEW based abnormal heart sound analysis

In 2006, Cardiac Auscultation by Travel ME since its invention 200 years ago, the stethoscope has been an invaluable bedside tool for auscultating heart sounds and over the years has undergone very few changes to both itself and the way in which it is used in health facility [20].

In 2007, Heart Energy signature spectrogram for cardiovascular Diagnosis by Vladmire K Library online heart sound from authors used and found that a visual map of Heart sound energy established with time frequency was found reduced disadvantage of PCG and it didn't need special filter there was a high resolution of generated heart sound image in time and pitch with improved quality than spectrogram image [21].

In 2008, Software develop for the analysis of Heart beat sounds with LabVIEW in diagnosis of CVD by Taner the data acquisition & preprocessing signals are analyzed using the time frequency scale spectral analysis technique STFT, WT to visualize frequency band activity depending on time & signal are processed with STFT, WVD. Real heart sound from 35 individuals compared STFT, WVD & WT frequency band activity mapped on frequency time axis using STFT sampling frequency & recording interval are conveniently adjustable 8 kHz sufficient to digitize heart sound. It overcomes the deficiency of expert doctors. It was found WVD, PWVD applied provides high time frequency resolution & performance in spectral analysis STFT was more applicable for real time processing of heart sound signals it also stated the powerfulness of LabVIEW and its flexibility for future fast easy manipulation process of heart sound signals[22].

In 2010, Examination of formant frequencies for further classification of heart murmurs designed by Molcer PS et al. The electrocardiogram (ECG) is a powerful and common screening tool for heart diseases. ECG technique is relatively inexpensive, non-invasive, and easy to use. However, one of the limitations is the difficulty in detecting structural abnormalities in heart valves and defects characterized by heart murmurs [23]. Heart disease is the leading cause of death in most countries in the world. Cardiovascular diseases killed 17.5 million people, i.e., three in every ten deaths [24].

In 2012, Heart sound Analysis using LabVIEW by N Ruban et al, online recording using ultrasound probe with a.wav format audio file and a separation coefficient to divide the iteration values where the quotient compared with a fixed value in which the final output will be in a wave

form for analysis, the algorithm was successful and it requires less computational speed and user friendliness, as a result it separated heart sound in its objective to improve in monitoring and evaluation of cardiovascular diseases[25]. Furthermore, study of HS analysis instrument by using Wireless Heart sound receiver sensor PC and LabVIEW incorporated with time and frequency domain subsystem for real time recording to calculate the heart beat frequency and interval of heart sound without aliasing to calculate the energy spectrum using layering technique normalized energy envelops then feed this value back to heart sound to obtain time. As a result, the sub system calculates sampling signals heart beat frequency and heart sound intervals correctly [26].

In 2014, Automatic localization of the left ventricle from cardiac cine magnetic resonance imaging by Zhong L et al a computed tomography (CT) scan of the heart, on the other hand, is an imaging method that uses x-rays to create detailed pictures of the heart and its blood vessels. Above all recently, many advanced geometry processing techniques have been developed for the MRI and CT based 3D and 4D heart model reconstruction, to further understand and visualize what we are unable to obtain from static 2D images [27].

In 2015, the electronic stethoscope was designed by Shuang L. et al. The system provides an informative and illustrative presentation of the electronic stethoscope, which is valuable and beneficial to academics, researchers and engineers in the technical field, as well as to medical professionals to facilitate its use clinically. In particular, the emergence of the electronic stethoscope has paved the way for a new field of computer-aided auscultation as well technological and medical basis for the development and commercialization of a real time integrated heart sound detection, acquisition and quantification system.

Likewise, electrocardiogram (ECG) is a powerful and common screening tool for heart diseases. It is relatively inexpensive, non-invasive, and easy to use. However, it does have some limitations, one of which is the difficulty in detecting structural abnormalities in heart valves and defects characterized by heart murmurs [28]. It is very important to have a cost effective and accurate method for the early detection of cardiac illnesses. Heart auscultation, defined as listening and interpretation of the heart sound (HS), has been a very important method for the early diagnosis of heart diseases [29]. Despite the advantages of low cost and easy operation, auscultation of the heart has been traditionally limited by three factors. Firstly, as the HS contains a mixture of high

frequency (HF) and low frequency (LF) acoustic signals with low amplitude, it is highly required for the stethoscope (or the sensor used in the stethoscope) to have a high selective sensitivity. Secondly, the HS data recorded with the stethoscope is often corrupted with noise, which can prohibit accurate and effective diagnosis of heart diseases. Thirdly, the interpretation of the HS is very subjective, and it depends largely on the experience, skills, and hearing ability of the physician. Therefore, the need for advancement of heart auscultation is highly evident. The acoustic based automatic diagnosis of cardiac dysfunction by electronic stethoscope has attracted much attention in recent years.

The smart phone to be connected with an external stethoscope, while some other stethoscope apps use the sensitive enough microphones embedded in the smart phone to acquire and process hearts sounds. Future healthcare have pointed out that the use of ultrasound at the point of care can truly enhance patient care and outcomes by expediting diagnoses and treatment [30].

There is a tremendous scope for the design of the very portably adoptable electronic stethoscope, so that each and every doctor and nurse can employ it for automated diagnosis in the clinical setting.

With recent technological advancements on signal processing algorithms and machine learning techniques, the computer-aided diagnosis of cardiac dysfunction using electronic stethoscope will become a reality, and represent another tremendous step forward toward finding a truly acoustic based diagnostic tool in all medical clinics as well as in bedside use. The new intelligent computer-based stethoscope system can contribute to a new auscultatory semiology, based on reliable methods of signal analysis and automatic interpretation, and will be the preliminary point of care to be followed by other points of care modality like portable ultrasound. Hence, we are advocating that this new intelligent computer-based stethoscope system be made available for universal use in clinics and hospitals [31].

To date, however, the computer based HS analysis has not yet been developed to a level that can be used for automated reliable diagnosis in the clinical setting. Almost all the available electronic stethoscopes and smart phone stethoscope apps in the market are only for general auscultation, i.e., hearing, recording and transferring the HS data. They are still far away from being called diagnostic devices. Some electronic stethoscopes can become portable by establishing connections

to other handheld devices like Mobile phone or wirelessly transmitting the sound signals to a remote processing unit through Bluetooth and apps on mobile device.

Furthermore, nowadays, the magnetic and ultrasound scanner can take detailed and even moving images of the heart. Echocardiogram (echo) uses the principle of rebounding waves to create a moving picture of the heart. It provides information about the size, shape, structure and function of the heart, the MRI test produces both still and moving pictures of the heart and major blood vessels accordingly [32].

2.2 Research Gaps

The major gaps in the above five designed systems for the identification of abnormal heart sounds include:

> ➢ The system designed most focus on normal hearts sound analysis
> ➢ Concentration on signal input sensor is more of the focus area rather than proper display for interpretation.
> ➢ The major disadvantages of echo, MRI and CT are their high cost and the need of specialized personnel to operate the complex machines. This equipment's are usually only affordable in large hospitals in the big cities. The availability of the medical imaging equipment's in these countries is quite low.
> ➢ Lack the most important part in the display of amplitude, time, duration and shape of the irregularity of abnormal heart sound so called murmurs
> ➢ Need for algorithm improvement is high, sample collection in clinical setup.
> ➢ Some techniques are limited by insignificant reception by clinician.
> ➢ Skills are needed to visualize energy level
> ➢ Back and forth needed where continues processed data are not transmitted at one time

2.3 Summary

Taking in to accounts the gaps on the literatures reviewed positively for the methodology development and considering the weight of existing problem in the current identification of abnormal heart sound. Abnormal heart sound embedded together with normal heart sound are not more dealt and considered in their research taking advantage of existing technique of phonocardiography as a potential source for the next step on display of abnormal heart sound

visualization. LabVIEW based virtual instrumentation system was developed which can be used by health care professionals working with patients at the health care facility on immediate patient arrival for diagnosis and treatment with affordable additional support of design to existing technology.

<div align="center">

Chapter 3

Material and Methodology

</div>

3.1 Study Area

The study was conducted in Jimma University Institute of Technology school of Biomedical Engineering Bioinstrumentation Laboratory. Heart sounds data used was from online Thinklab database.

3.2 Study Design

This research used online audiovisual heart sound from certified clinical database Thinklab. Accessed data was downloaded as an input signal to the data acquisition device in to the designed virtual instrumentation system, these data are analyzed by the block of instrumentation system which forms unique graphical wave envelops as desired by the research.

3.3 List of Material

The materials used in this work are listed in table 1.

SN	Material	Function	Category
1	*DAQ 6008/9*	Data Acquisition device	Hardware
2	*LabVIEW 2019*	Virtual Instrumentation software plate form	Software
3	*USB Cable*	Communication Cable between DAQ & LabVIEW	Hardware
4	*Laptop Computer*	To acquire, process and display wave forms	Hardware

<div align="center">Table 1: List of Material</div>

3.4 LabVIEW

A. Encoding

A key benefit of LabVIEW over other development environments is the extensive support for accessing instrumentation hardware. Drivers and abstraction layers for many different types of

instruments and buses are included and are available for inclusion. These present themselves as graphical nodes. The abstraction layers offer standard software interfaces to communicate with hardware devices. The provided driver interfaces save program development time. The sales pitch of National Instruments is so popular, that even people with limited coding experience can write programs and deploy test solutions in a reduced time frame when compared to more conventional or competing systems. A new hardware driver topology (DAQmx Base), which consists mainly of G-coded components with only a few register calls through NI Measurement Hardware DDK (Driver Development Kit) functions, provides platform independent hardware access to numerous data acquisition and instrumentation devices. The DAQmx Base driver is available for LabVIEW to control the device on Windows, Mac OS X and Linux platforms.

B. Code Compilation

In terms of performance, LabVIEW includes a compiler that produces native code for the CPU platform. The graphical code is translated into executable machine code by interpreting the syntax and by compilation of codes. The LabVIEW syntax is strictly enforced during the editing process and compiled into the executable machine code when requested to run or upon saving. In the latter case, the executable and the source code are merged into a single file. The executable runs with the help of the LabVIEW run-time engine, which contains some precompiled code to perform common tasks that are defined by the G language. The run-time engine reduces compile time and also provides a consistent interface to various operating systems, graphic systems, hardware components, etc. The run-time environment makes the code portable across platforms. Generally, LabVIEW code can be slower than equivalent compiled C code, although the differences often lie more with program optimization than inherent execution speed.

C. Large Libraries

Many libraries with a large number of functions for data acquisition, signal generation, mathematics, statistics, signal conditioning, analysis, etc., along with numerous graphical interface elements are provided in several LabVIEW package options. The number of advanced mathematic blocks for functions such as integration, filters, and other specialized capabilities usually associated with data capture from hardware sensors is immense.
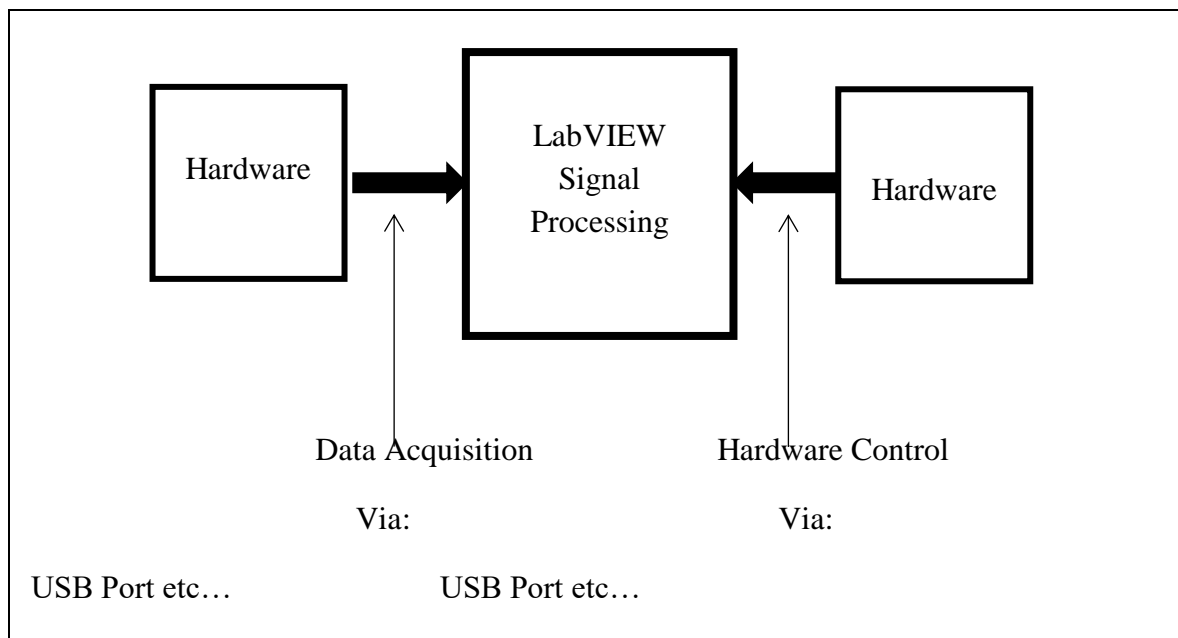
Figure 3: Schematic diagram of an Instrument system based on LabVIEW

In addition, LabVIEW includes a text-based programming component called MathScript with additional functionality for signal processing, analysis and mathematics. MathScript can be integrated with graphical programming using "script nodes" and uses a syntax that is generally compatible with MATLAB.

D. Code Reuse

LabVIEW allows code reuse without modification as long as the data types of input and output are consistent, two sub VIs are interchangeable. The LabVIEW Professional Development System allows creating stand-alone executable and the resultant executable can be distributed a number of times. The run-time engine and its libraries can be provided freely along with the executable.

A benefit of the LabVIEW environment is the platform independent nature of the G code, which is (with the exception of a few platform-specific functions) portable between the different LabVIEW systems for different operating systems (Windows, Mac OS X and Linux). National Instruments is increasingly focusing on the capability of deploying LabVIEW code onto an increasing number of targets including devices like Phar Lap or VxWorks OS based LabVIEW Real-Time controllers, FPGAs, Pocket PCs, PDAs, and Wireless sensor network nodes.

E. **Educational Objectives**

It is difficult to design laboratory exercises using complex scientific equipment and still have the student learn the concepts being demonstrated. Using LabVIEW to facilitate the data generation and collection changes the focus from learning how to use the equipment to learning the physiological concepts being presented in the lab. Because LabVIEW allows designing simple, easy-to-use interfaces, however, care must be taken not to make the virtual instruments (VIs) so simple to use that students may not see the details of the measurement process. Hence instructors must ensure that students do not leave the lab thinking that measuring physiologic phenomena means you click once on a button and the data magically appears.

F. **Countless Applications**

LabVIEW applications are implemented in many industries worldwide including automotive, telecommunications, aerospace, semiconductor, electronic design and production, process control, biomedical, and many others. Applications cover all phases of product development from research to design to production and to service. By leveraging LabVIEW throughout an organization the user can save time and money by sharing information and software. Below are some of the features.

- ➢ Graphical user Interface
- ➢ Drag-and-drop built-in functions
- ➢ Modular design and hierarchical design
- ➢ Multiple high level development tools
- ➢ Professional Development Tools
- ➢ Multi platforms
- ➢ Reduces cost and preserves investment
- ➢ Flexibility and Scalability
- ➢ Connectivity and instrument
- ➢ Open Environment
- ➢ Distributed environment
- ➢ Visualization capabilities
- ➢ Rapid development with express technology
- ➢ Complied language for fast execution
- ➢ Simple application distribution

➢ Target Management

➢ Objects-oriented designs

➢ Algorithm Design

National instrumentation, develop algorithms using math-oriented textual programming and interactively debug.m file script syntax with LabVIEW Math Script [33].

3.4.1 LabVIEW Programming Process and Techniques

LabVIEW also referred to as G, is different from most other general-purpose programming languages in two major ways. First, programming is performed by wiring together graphical icons on a diagram, which is then compiled directly to machine code so the computer processors can execute it. The second main differentiator is that G code developed with LabVIEW executes according to the rules of data flow instead of the more traditional procedural approach found in most text-based programming languages like C and C++. Dataflow languages like G promote data as the main concept behind any program. Dataflow execution is data-driven, or data-dependent. The flow of data between nodes in the program, not sequential lines of text, determines the execution order. LabVIEW provides a variety of features and tools ranging from interactive assistants to configurable user-defined interfaces; it is differentiated by its graphical, general-purpose programming language (known as G) along with an associated integrated compiler, a linker, and debugging tools. LabVIEW programs/subroutines are called virtual instruments (VIs). Each VI has three components: a block diagram, a front panel and a connector pane. The last is used to represent the VI in the block diagrams of other, calling VIs. Controls and indicators on the front panel allow an operator to input data into or extract data from a running virtual instrument. However, the front panel can also serve as a programmatic interface. Thus a virtual instrument can either be run as a program, with the front panel serving as a user interface, or, when dropped as a node onto the block diagram, the front panel defines the inputs and outputs for the given node through the connector panel. This implies each VI can be easily tested before being embedded as a subroutine into a larger program.

3.4.1.1 Block Diagram

The block diagram is a pictorial representation of the program code. It corresponds to the lines of text found in text based programming language, and is, the actual executable code. The block diagram is built by wiring together objects that perform specific functions. The block diagram

contains this graphical source code. Front panel objects appear as terminals on the block diagram. Figure 4 shows LabVIEW block diagram.

The components of block diagram belong to one of three classes of objects.

Nodes: - program execution elements.

Terminals: - parts through which data passes between block diagram

Wires: - data paths between terminals

Nodes are analogous to statements, functions and subroutines in text based programming languages. There are three types of Nodes Function, subVI Nodes and Structures Node.

Functions are the built in nodes for performing elementary operations such as File I/O, adding numbers or string formatting. Functions are the fundamental operating element of a block diagram. Sub VI node are the VI that a user can design and later call from the diagram of another VI. Structures are the icons that are used to control the program flow (such as FOR loop and WHILE loop).
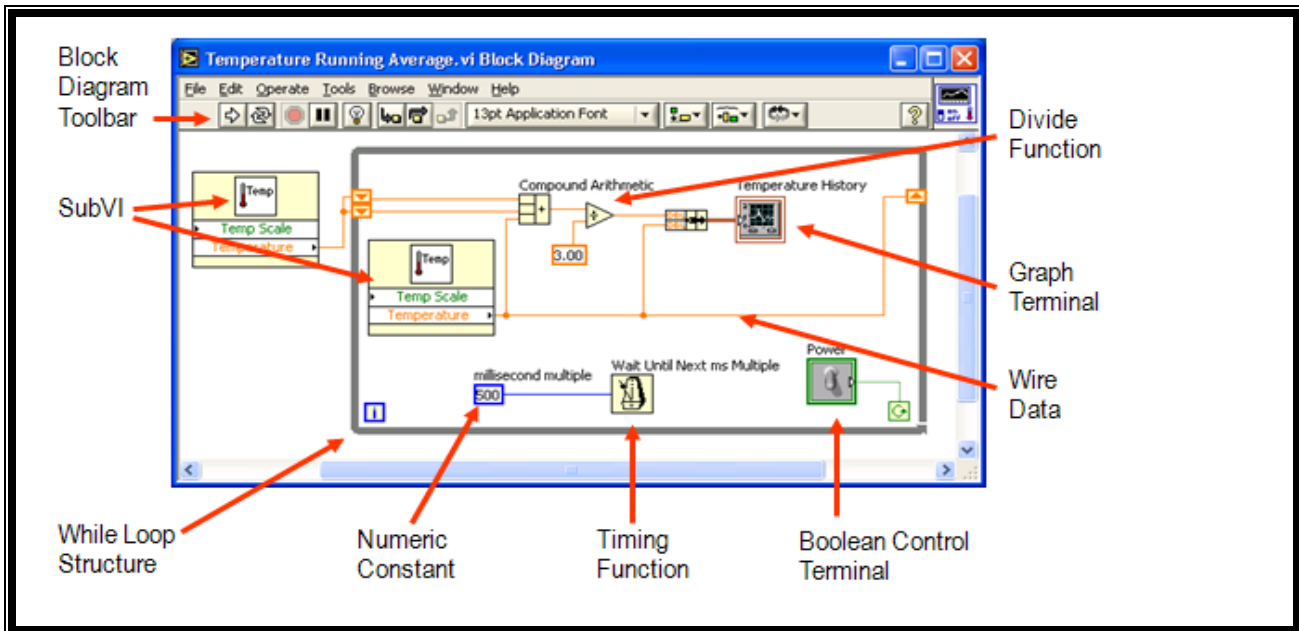

Figure4: LabVIEW Block Diagram

3.4.1.2 Front Panel

The front panel shown on figure 5 is the interactive interface of the VI. It contains the graphical interface with which a user interacts. The front panel can house various graphical objects ranging from simple buttons to complex graphs.
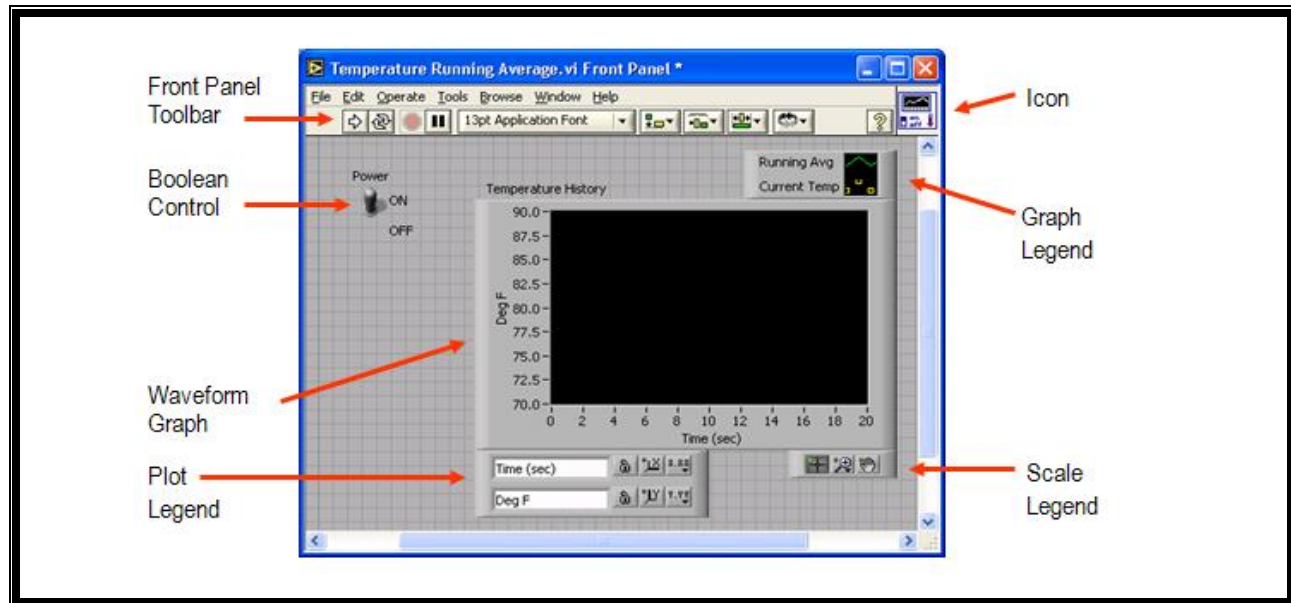
Figure5: LabVIEW Front Panel

3.4.1.3 Connector Pane

The connector pane on figure 6 is a set of terminals that correspond to the controls and indicators of that VI, similar to the parameter list of a function call in text-based programming languages. The connector pane defines the inputs and outputs a user can wire to the VI so that he can use it as a Sub VI. A connector pane receives data at its input terminals and passes the data to the block diagram code through the front panel controls and receives the results at its output terminals from the front panel indicators [34].

The user can designate which inputs and outputs are required, recommended, and optional to prevent users from forgetting to wire subVI terminals. The block diagram on which a user placed the subVI will be broken if he does not wire the required Terminal inputs. For terminal inputs and outputs, recommended and optional means is that wiring of the terminals is sometimes optional and sometimes compulsory. If user does not wire the terminals, the block diagram on which he placed the subVI does not generate any warnings.

Inputs and outputs of VIs in vi.lib are already marked as Required, Recommended, or Optional. LabVIEW sets inputs and outputs of VIs a user creates to Recommended by default. A user sets a terminal setting to required execution only if the VI has the input or output to run properly.

To designate which inputs and outputs are Required or Recommended, and Optional, right-click a terminal on the connector pane and select This Connection Is from the shortcut menu. A user can configure LabVIEW to set all new terminals to configure on the connector pane required instead of recommended. Select Tools»Options»Front Panel and place a checkmark in the Connector pane terminals default to required checkbox. This option also applies to connections made using the Edit» Create SubVI menu item.
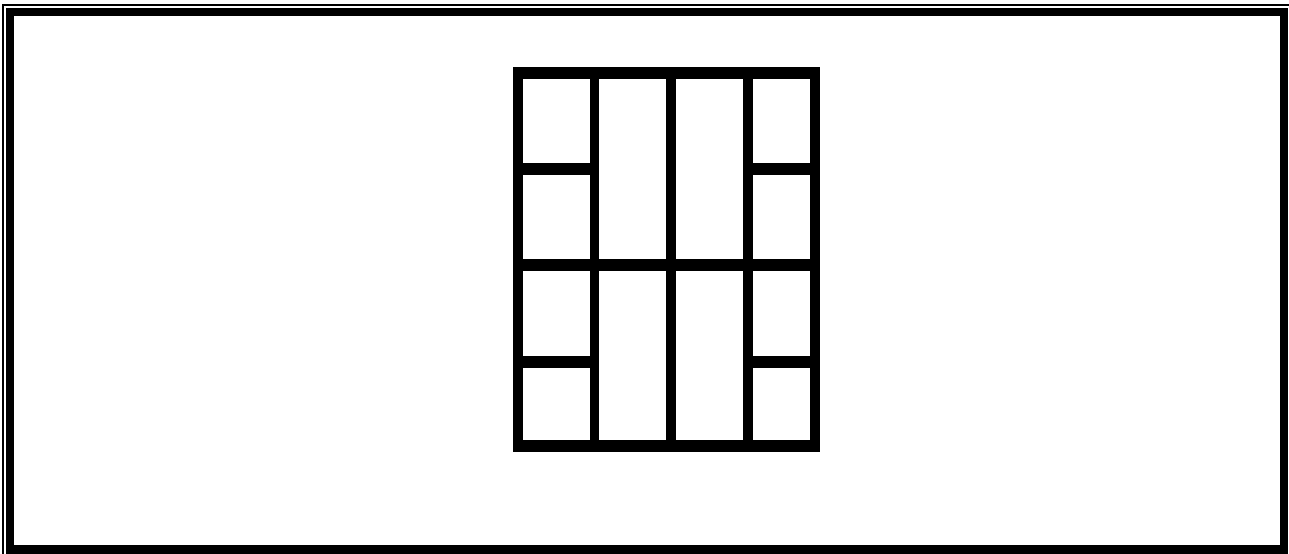

Figure6: Connector Pane

3.4.1.4 Controls and Indicators

Controls and indicators are the interactive input and output terminals of the VI, respectively. Controls are knobs, push buttons, dials, and other input devices. Indicators are graphs, LEDs, and other displays. Controls simulate instrument input devices and supply data to the block diagram of the VI. Indicators simulate instrument output devices and display data the block diagram acquires or generates. Objects for the Front Panel are dragged over from the Controls Palette (this palette can be opened from the View menu as long as the Front Panel widow is active and it disappears when the Block Diagram is selected). Controls supply the VI with data and Indicators display data from the VI. When a control or indicator is dragged on to the Front Panel, an object corresponding to it automatically appears on the Block Diagram. Objects for the Block Diagram are dragged over from the Functions palette (which becomes available when the Block Diagram is active). Once all the required objects have been dragged on to the Front Panel and Block Diagram next is wiring

them together. Each object on the Block Diagram has one or more input or output terminals, and it is good programming practice to make sure each is wired properly, even if the program runs with a terminal left unwired. Wiring is done by dragging and clicking the wiring tool from terminal to terminal. Dragging the tool creates the wire, while clicking tacks down the wire at that point and lets a user to draw the tool in another direction. Once the wiring is completed, the VI is saved in the usual way (with .vi as the extension).

One complication is that there are many different types of data, and each object produces and expects certain ones.

LabVIEW uses labels and colors to make this easier, but it is a part of the programming that needs careful attention. The types you will be using at first are 16 bit signed integers (I16), single and double precision real floating point numbers (SGL and DBL), Boolean states (e.g., true or false), and I/O name controls (Channel Types, which users have LabVIEW generate for them). A user must remember to save the developed VI.

3.4.2 Design of Virtual Instrument for processing of signals using LabVIEW

National Instruments hardware and software products are used in a broad variety of biomedical applications such as biophysics cellular physiology, tissue bath recordings, exercise physiology, biomechanics, and lab automation. This series of Webcasts explores a variety of topics around the challenges in designing and validating these types of medical applications

The instrument which is developed using LabVIEW is called the virtual instrument [35]. The designed instrument makes the best of active computer resource, matches with oneness of constructive instrument hardware and proprietary software, and achieves all the functions of the traditional medical instrument as well as some fantastic specific functions that cannot be carry out on the traditional medicine instrument.

A virtual instrument designed for medical application includes set of software and hardware on the general-purpose computer, while a user operates this computer as if operating special traditional medicine equipment designed by user.

The appearance of the virtual instrument technology breaks through the mode, that traditional medicine instrument is defined by manufacturer and a user cannot change it. Given the adequate space to exert ability and imagination, a user can design personal medical instrument system to satisfy various applications.

The virtual medical instrument is the testing platform based on the computer soft and hardware; it can replace the traditional medical instrument, such as electrocardiograph [36]. It also can be used in auto control and data processing system, to build up special medical instrument system. A virtual medical instrument is composed of a computer, application software and medical instrument hardware. No matter which kind it is, the virtual medical instrument system is sure to be made up of these three aspects that the application software and the medical instrument hardware are installed into a platform based on a notebook computer, a table computer or a workstation (even a PDA). The virtual medical instrument puts the computer hardware resources and medical instrument hardware together organically through software, and then links the powerful count processing ability of the computer with the measuring and controlling ability of the medical instrument hardware, thereby the cost and volume of the medical instrument hardware is reduced and the data can be displayed.

The biomedical signals acquired from the human body are frequently very small, often in the millivolts range, and each has its own signal processing needs. We are designing the VI for processing of heart sounds and this biomedical signal require signal processing before analysis LabVIEW contains the tools, from Fast Fourier transforms (FFTs) to digital filters, to do the job [37]. In order to do frequency analysis, a complex signal must first be broken down into its frequency components. One of the most common ways to do this is with an FFT. In order to facilitate this type of analysis, LabVIEW comes with built in FFTs that make the process of component separation quickly and easily. In addition, biomedical signals, being extremely small in amplitude, are prone to being overwhelmed by noise. To combat this, it is necessary to run the acquired signal through a set of filters [38]. This can be done externally to the computer using standard hardware filtering devices. However, after the signal reaches the computer, it can still contain noise. Another way to solve the noise problem is to use the digital filters provided with LabVIEW. LabVIEW offers the choice of Butterworth, Bessel, Chebyshev, and Chebyshev II digital filters [39]. With a few adjustments, these filters can be configured for almost any design that is needed.

Analysis of the biomedical signals can also be easily done in LabVIEW. The graphical nature of LabVIEW allows even the programmer to easily write programs to analyze data without having to worry about the syntax problems associated with most programming languages. Once again, the user just drags subprograms onto the diagram screen to set up a working program. These modules

can be configured to do any kind of manipulation required with just a few clicks of a mouse button. In addition to the analysis modules that the programmer writes, National Instruments offers a wide variety of analysis libraries, including ones for biomedical signals.

LabVIEW can be an efficient alternative to stand alone medical instrument, and as the speed and reliability of the system increases, there will be more and more of virtual medical instrument systems available.
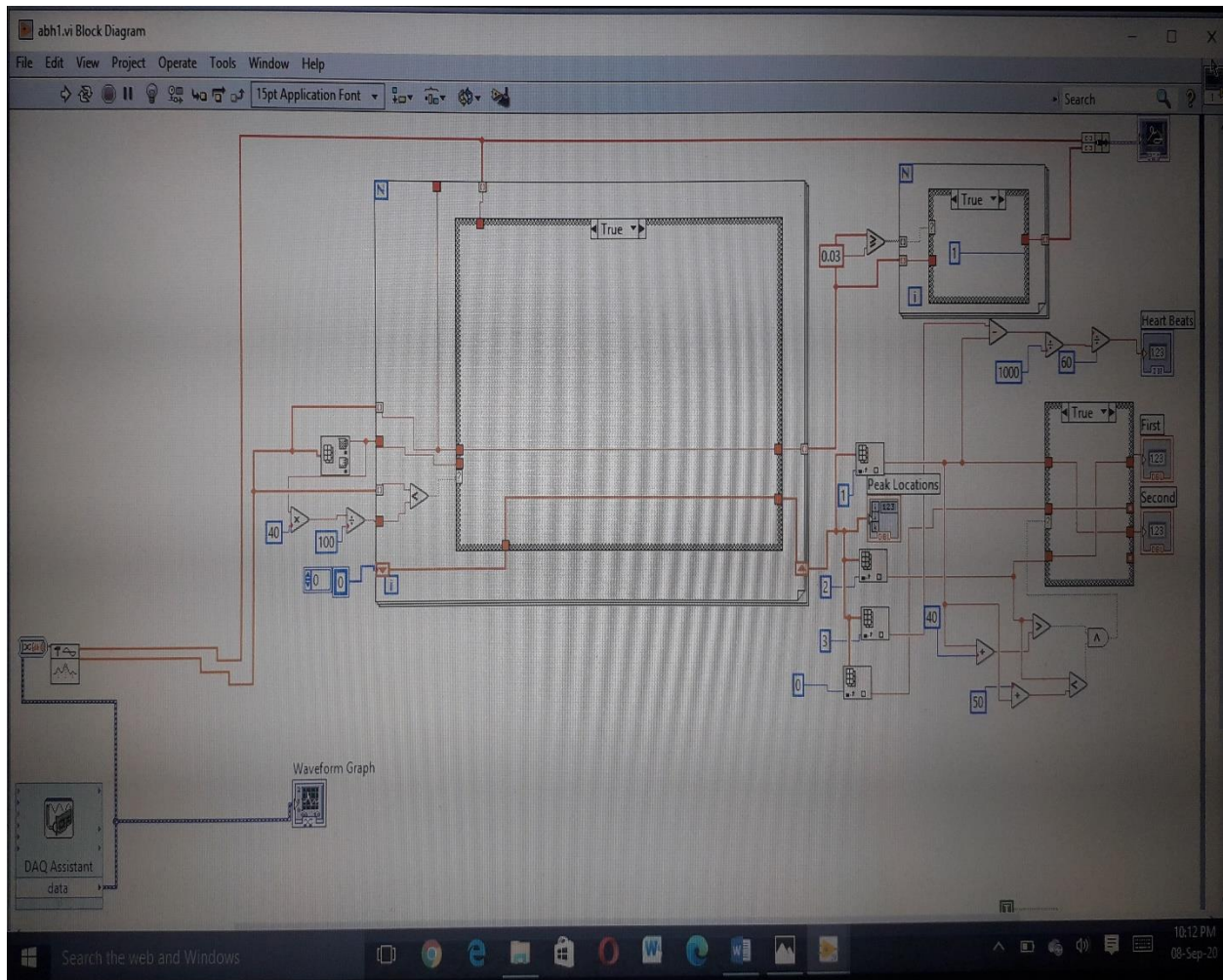

Figure7: LabVIEW block diagram for analysis of heart sounds designed

The components necessary for a LabVIEW based acquisition and analysis system are inexpensive and readily available. The experience necessary to program this type of LabVIEW is small, and the number of libraries available from National Instruments is growing.

LabVIEW (Laboratory Virtual instrumentation work bench) used as a plate form for designing, programming, acquire and process the input heart signals through DAQ (Data acquisition device). The DAQ 6008/9 receives the input signal from the second computer through wires connected one end by standard audio jack and the other end electronic circuit jack in to the referenced analogue signal input zero AI0 of DAQ and common ground.Below is figure 7 is the designed VI for this work.

3.5 Limitation of LabVIEW and DAQ

3.5.1 Limitation 1

The software package of updated version is not compatible with window 64 bit 2019 and insisted to change three different personal laptop computers it gave a trial period of the software and failed in the middle once it is registered on National instrumentation.

3.5.2 Limitation 2

Device purchased from national instrumentation, DAQ 6008/9 was not accessible on the system device manager and failed to be updated. This initiated full version hardware and software compatible to specific window needs to be repurchased.

3.6 Additional Design to Overcome the Limitation

To overcome the limitation two different window s on a smart laptop computer core I 5 windows 10 installed.

   I.   Window 32-bit core i5 and LabVIEW software version 2014
  II.   Window 64-bit core i5 and LabVIEW software version 2019

In both windows DAQ 6008/9 was installed but in the first case with 32 bit unable to update DAQ in device manger and not able to see it on the LabVIEW block diagram express function pallet input device. Whereas on the 64-bit window the DAQ was registered by the company and updated the software in device manger and accessible on the LabVIEW block diagram function pallet express VI as input for ready to use and calibration for the intended design.

3.7 Conceptual Diagram of proposed System

As indicated on the figure 7above the block diagram with system design to acquire, analyze the signal and display abnormal heart sounds. Whereas figure 8 below is the conceptual diagram of the proposed system in general which is the back bone of the virtual instrument in this work.
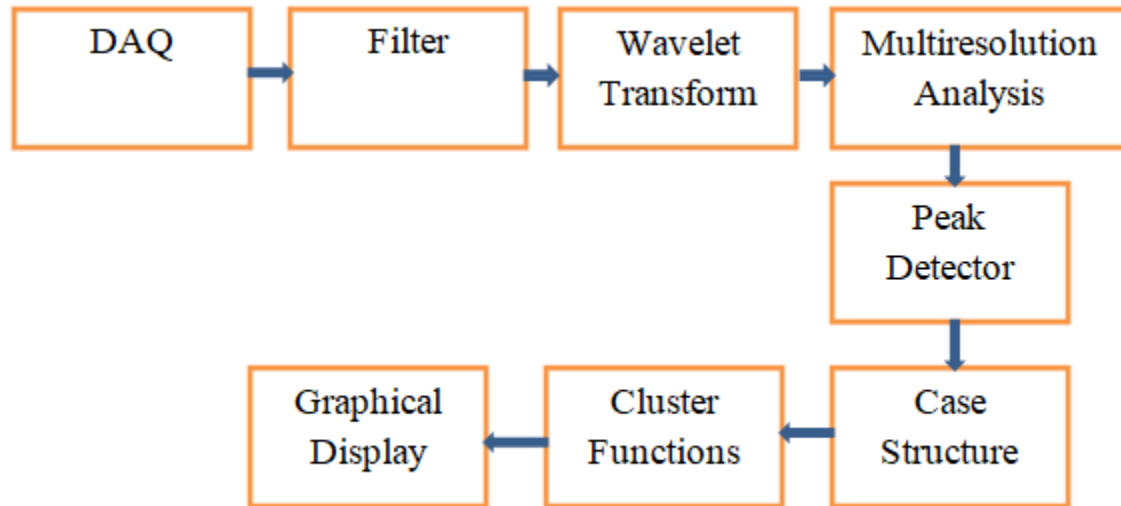

Figure8: Conceptual Diagram of Proposed System

3.7.1 Data Acquisition unit

The data acquisition device used in the present study shown on figure 9 is National Instruments LabVIEW capability USB-6008/6009 DAQ which has eight channels, 12 bit DAC, 12/14-bit ADC and can configure the Analog input channels on the device to take single-ended or differential measurements during the process. The USB-6008/6009 can perform both single and multiple ADC of a fixed or infinite number of samples. A first-in-first-out (FIFO) buffer holds data during AI acquisitions to ensure that no data is lost.
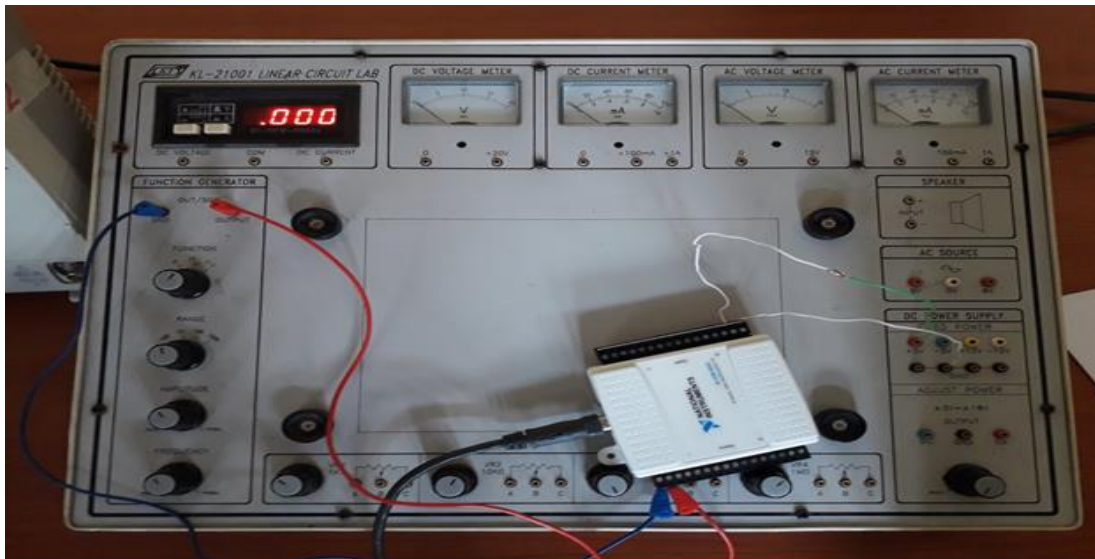
Figure 9: NI DAQ 6008/9

3.7.2 Filters

Filtering of digital signals is a fundamental concept in digital signal processing. Filters are used to reject unwanted noise within a certain frequency range. Many systems will exhibit 60Hz periodic noise components from sources such as power supplies or machinery. Low pass filters on our signal conditioning circuitry can eliminate unwanted high-frequency components. However, be sure to select the filter bandwidth carefully so that the time response of the signals is not affected. Although many signal conditioners include low pass noise filters to remove unwanted noise, an extra precaution is to use software averaging to remove additional noise.

Software averaging is a simple and effective technique of digitally filtering acquired readings; for every data point that is required, the DAQ system acquires and averages many voltage readings [40].

To get rid of aliasing problems, you need to use a low-pass filter, known in this case as an antialiasing filter. Analog filters are mandatory, regardless of the sampling rate, unless you know the signal's frequency characteristics and can live with the aliased noise. There has to be something to limit the bandwidth of the raw signal to fs/2. The analog filter can be in the transducer, in the signal conditioner, on the ADC board. If the desired signal is fairly close to the Nyquist limit, the filter needs to cut off very quickly, implying lots of stages (this is more formally known as the order of the filter's transfer function).

Digital filters can augment, but not replace, analog filters. Digital filters VIs are included with the LabVIEW analysis library, and they are functionally equivalent to analog filters. The simplest type of digital filter is a moving Averager (examples of which are available with LabVIEW) which has the advantage of being usable in real time on a sample-by-sample basis. One way to simplify the antialiasing filter problem is to oversample the input. If ADC hardware is fast enough, just turn the sampling rate way up, and then use a digital filter to eliminate the higher frequencies that are of no interest. This makes the analog filtering problem much simpler because the Nyquist frequency has been raised much higher, so the analog filter doesn't have to be so sharp. A compromise is always necessary: You need to sample at a rate high enough to avoid significant aliasing with a modest analog filter; but sampling at too high a rate may not be practical because the hardware is too expensive and/or the flood of extra data may overload your poor CPU.

If you are attempting to average recurrent waveforms to reduce noise, remember that the arrays of data that you acquire must be perfectly in phase. If a phase shift occurs during acquisition, then your waveforms will partially cancel one another or cause distortion. Triggered data acquisition (discussed later) is the normal solution because it helps to guarantee that each buffer of data is acquired at the same part of the signal's cycle.

Some other aspects of filtering that may be important for some of your applications are impulse response and phase response. For ordinary data logging, these factors are generally ignored [41]. But if you are doing dynamic testing such as in vibration analysis, acoustics, or seismology, then impulse and phase response can be very important. As a rule of thumb, when filters become very complex (high-order), they cut off more sharply, have more radical phase shifts around the cutoff frequency, and, depending on the filter type, exhibit greater ringing on transients.

Use a dedicated spectrum analyzer instrument (very expensive), a digital oscilloscope with FFT capability (or let LabVIEW do the power spectrum), or even a multifunction I/O board running as fast as possible with LabVIEW doing the power spectrum. Spectrum analyzers are included in the data acquisition examples distributed with LabVIEW, and they work quite well. The labVIEW have icon that selects IIR filter which will allow to select Butterworth filter.

3.7.3 Wavelet Transform

Wavelets are functions that you can use to decompose signals. Just as the Fourier transform decomposes a signal into a family of complex sinusoids, the wavelet transform decomposes a signal into a family of wavelets. Unlike sinusoids, which are symmetric, smooth, and regular, wavelets can be symmetric or asymmetric, sharp or smooth, regular or irregular. For different types of signals, you can select different types of wavelets that best match the features of the signal you want to analyze. Therefore, you can perform wavelet signal processing and generate reliable results about the underlying information of a signal.

The family of wavelets contains the dilated and translated versions of a prototype function. Traditionally, the prototype function is called a mother wavelet. The scale and shift of wavelets determine how the mother wavelet dilates and translates along the time or space axis as indicated on figure 10. A scale factor greater than one corresponds to a dilation of the mother wavelet along the horizontal axis and a positive shift corresponds to a translation to the right of the scaled wavelet along the horizontal axis.
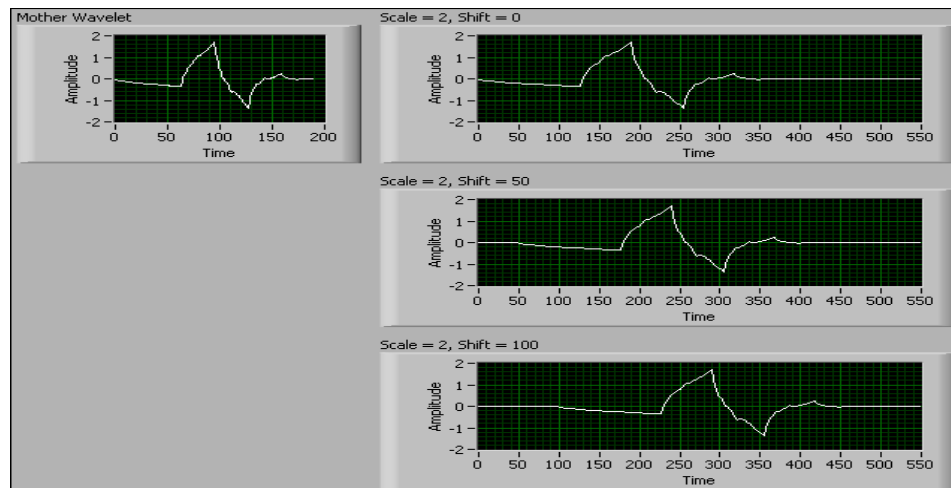

Figure 10: Wave form graph

The wavelet transform computes the inner products of a signal with a family of wavelets. The wavelet transform tools are categorized into continuous wavelet tools and discrete wavelet tools. Usually, you use the continuous wavelet tools for signal analysis, such as self-similarity analysis

and time-frequency analysis. You use the discrete wavelet tools for both signal analysis and signal processing, such as noise reduction, data compression, peak detection and so on.

In contrast with sinusoids, wavelets are localized in both the time and frequency domains, so wavelet signal processing is suitable for non-stationary signals, whose spectral content changes over time. The adaptive time-frequency resolution of wavelet signal processing enables you to perform Multiresolution analysis on non-stationary signals. The properties of wavelets and the flexibility to select wavelets make wavelet signal processing a beneficial tool for feature extraction applications.

By analyzing output signals in the frequency domain, one can see that these signals are composed of several frequency components. For many signals, frequency domain analysis using the Fourier transform (FT) is extremely useful because the signal's frequency content is of great importance for understanding the nature of the signal and any noise that contaminates it. However, Fourier analysis has a serious drawback: Transforming to the frequency domain causes time information to be lost. When looking at a Fourier transform of a signal, it is impossible to tell when a particular event took place.

In an effort to overcome this deficiency, the same transform was adapted to analyze only a small window of the signal at a time. This technique, presently known as the short-time Fourier transforms (STFT), maps a signal into a two-dimensional function of time and frequency.

STFT's only drawback is that once the user chooses a particular size for the time window, that window is the same for all frequencies. Many signals require a more flexible approach (for example, one in which the window size is varied to determine more accurately either time or frequency).Wavelet analysis represents the next logical step, which is based on a windowing technique with variable-sized regions. The wavelet transform (WT) allows the use of long time intervals for more-precise low- frequency information and shorter regions for high-frequency information. In general, the major advantage of wavelets is the ability to perform local analysis that is, to analyze a localized area of a larger signal. The mother wavelets used in this work is signal

A. Benefits of Wavelet Signal Processing

Wavelet signal processing is different from other signal processing methods because of the unique properties of wavelets. For example, wavelets are irregular in shape and finite in length shown on figure 11. Wavelet signal processing can represent signals sparsely, capture the transient features of signals, and enable signal analysis at multiple resolutions.
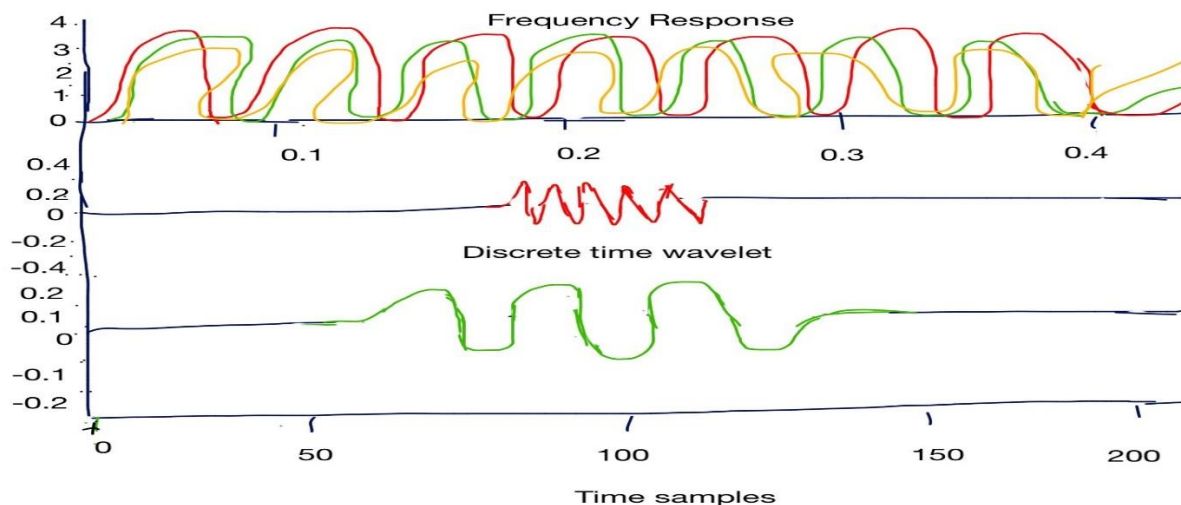

Figure 11: Wavelet Analysis

B. Sparse Representation

Wavelets are localized in both the time and frequency domains because wavelets have limited time duration and frequency bandwidth. The wavelet transform can represent a signal with a few coefficients because of the localization property of wavelets. The following figure shows the waveform of the Doppler signal.

The time duration and frequency bandwidth determine the time and frequency resolutions of a wavelet, respectively. Long time duration means coarse time resolution. A wide frequency bandwidth means coarse frequency resolution

3.7.4 Wavelet Multiresolution Analysis

Multiresolution analysis (MRA) shown on figure 12 determines the general WT [42]. It allows the decomposition of signals into various resolution levels. The data with coarse resolution contain

information about lower-frequency components and retain the main features of the original signal. The data with finer resolution retain information about the higher-frequency components. The filtering approach to Multiresolution WT is to form a series of half-band filters that divide a spectrum into a high-frequency band and a low- frequency band. It is formulated on a scaling function or low-pass filter (LP) and a wavelet function or high-pass filter (HP). MRA builds a pyramidal structure that requires an iterative application of scaling and wavelet functions to low-pass and high-pass filters, respectively. These filters initially act on the entire signal band at the high frequencies (small-scale) first and gradually reduce the signal band at each stage.
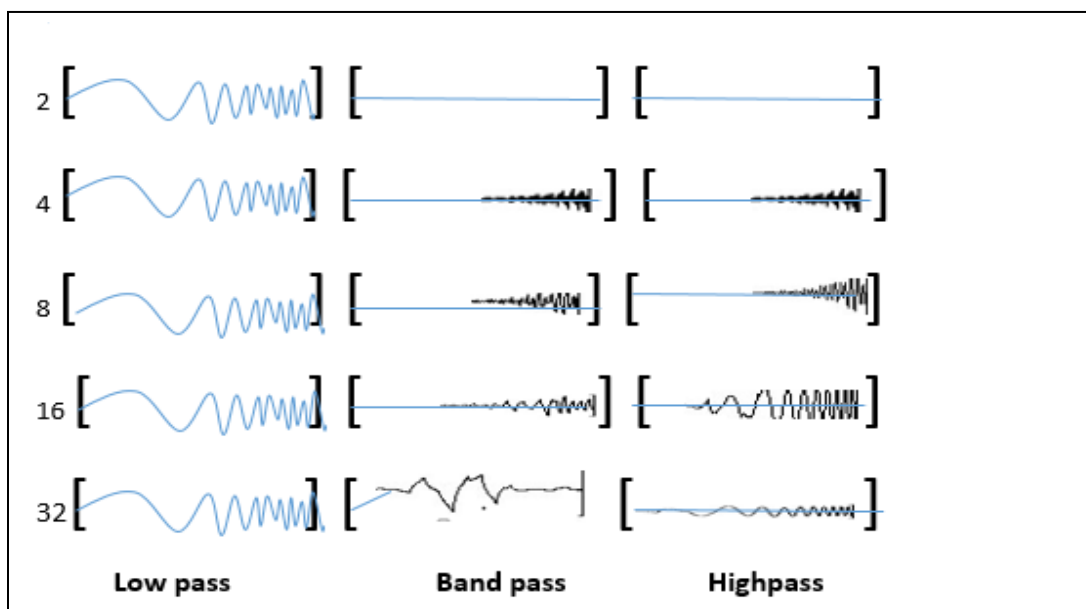

Figure 12: Multiresolution wavelet

Digital signal processing uses exclusively orthogonal wavelets to separate effectively the low- and high-frequency signal contents at each resolution level. The non-redundant representation and perfect reconstruction of the original signal can be realized only through compactly supported wavelets. Un- like the continuous wavelet transform, the discrete wavelet transform is supported by two filters, low pass and high pass, which have a certain number of coefficients that can effectively separate the low-pass and the high-pass frequencies of the signal. The wavelets that are frequently used for signal processing are Daubechies (named after their discoverer, Ingrid Daubechies), biorthogonal (involving two sets of low- and high- pass filters), coiflets (variations on Daubechies' wavelets), and symlets (nearly symmetrical wavelets also proposed by

Daubechies). These wavelets exhibit different attributes and performance criteria when applied to specific applications such as detecting signal transients, signal compression, and de-noising

Choosing an appropriate wavelet filter is important in identifying the characteristics of transient analysis. Because the convolution sum between the signal and the wavelet- system filters measures the similarity between the signal and the wavelet basis function, the simplest method for choosing a wavelet to observe the signal's time-frequency behavior is to copy the form of the transient signal. However, most transient signals, which are exponentially damped sinusoids, cannot be wavelet basis functions because of their nonzero mean.

3.7.5 Waveform Peak detector

Waveform Peak detector as shown on figure 13 below function finds the location, magnitude of amplitude and the Second derivatives of the peaks in the detector signal. The threshold and width, input parameters, serve as separation tools of the true detector pulses from the noise. The threshold determinates the minimum value of the peak amplitude and the width determinates the minimum peak width according to the number of samples over the given threshold. Waveform Peak detector is a software equivalent of the electronic pulse height analyzer; hence, the optimal treatment of the detector signal from the various types of detectors is achieved by DSP.

The Peak Detector VI takes an array of points representing a signal and finds the locations, amplitudes, and second derivatives of any peaks or valleys in the signal. The VI is based on an algorithm that fits a quadratic polynomial to sequential groups of data points.

The width parameter determines the number of consecutive points to fit the data using a quadratic least squares fit. As such, the width roughly controls smoothing of the data when searching for peaks. If the data is noisy, you would want more smoothing (specifically, a larger width), and vice versa. The disadvantage of larger widths is that the position (and value) of the detected peak may turn out to be less accurate. For fast moving signals and better tracking, use the minimum width of three. If a narrow width leads to errors due to noise, increasing the width may help, but a blurring of the detection could occur. For slower and noisier signals, you can expand the width, but do not increase it past the half-width of the peak or valley you are expecting. The VI performs a quadratic

fit on a window consisting of the number of data points specified in the width. It slides this window, performs the fit, slides the window, and performs the fit, and so on. While doing the fit, the VI calculates the parameters of the quadratic fit. For the fit, it finds the second derivative, and checks to see if the slope changes, to determine if there is a peak or a valley.
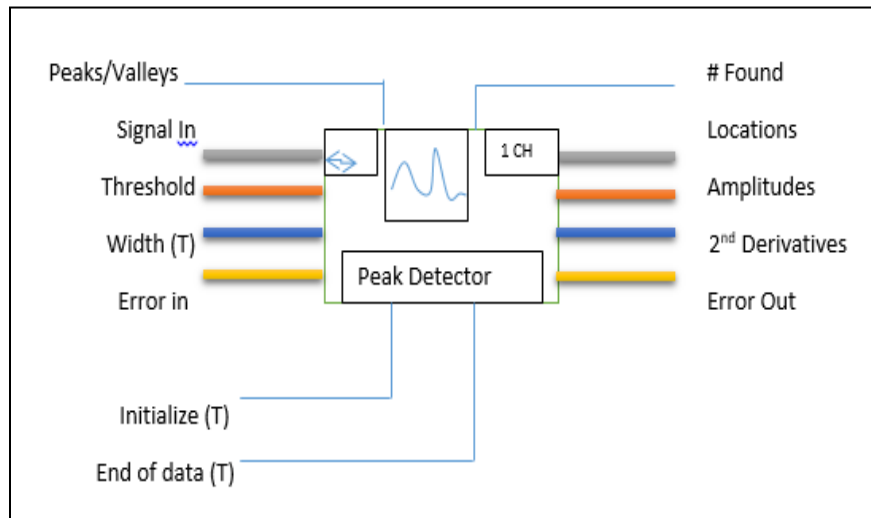

Figure13: Peak detectors VI

This Peak Detector VI is based on an algorithm that fits a quadratic polynomial to sequential groups of data points. The number of data points used in the fit is specified by width. For each peak or valley, the quadratic fit is tested against the threshold. Peaks with heights lower than the threshold or valleys with troughs higher than the threshold are ignored. Peaks and valleys are detected only after the VI processes approximately width/2 data points beyond the location of the peak or valley. This delay has implications only for real-time processing. The Peak Detector VI must be notified when the first and last blocks are passed into the VI, so that the VI can initialize and then release data internal to the peak detection algorithm.

3.7.6 Case Structure

The *Case Structure* is a method of executing conditional statements. The Case Structure is similar to *if...then... else* statements in conventional programming languages. Below is the pseudo code:

If (Condition = *Something*) then

Do This

Else

Do That

Endif
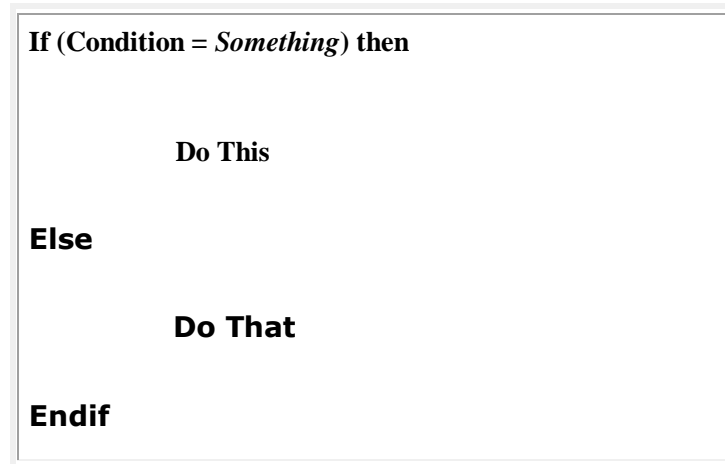
The LabVIEW Case Structure is configured like a deck of cards. Each "card" is a case in the case structure and each case contains a sub diagram. You only see one case and sub diagram at a time. LabVIEW executes only one at a time and case execute is determined by the value wired to the selector terminal (⍰) on the left side of the Case Structure shown on figure 14.
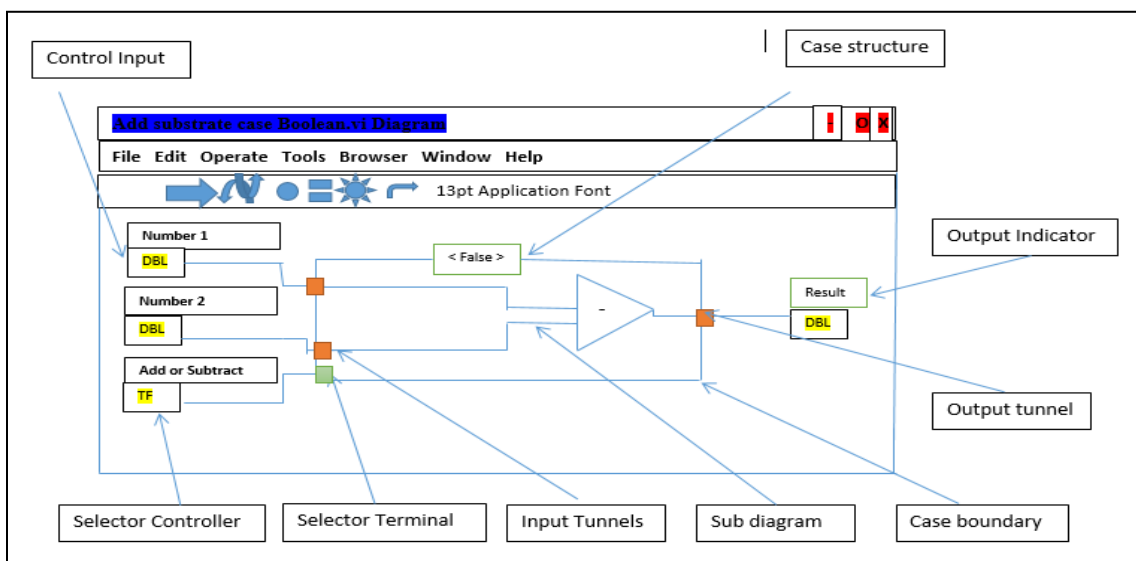

Figure14: Labeled example of a Case Structure defining the key terms

A. Case Structure Data Types

The data type wired to the selector terminal can be Numeric, Boolean, or String as shown similarly on figure 15. By default, Case Structures are set to Boolean but automatically adapt to the data type wired to the selector terminal. Below are examples of three Case Structures with the three different data types wired to the selector terminal. The cases have been separated in this image to better show their structure.
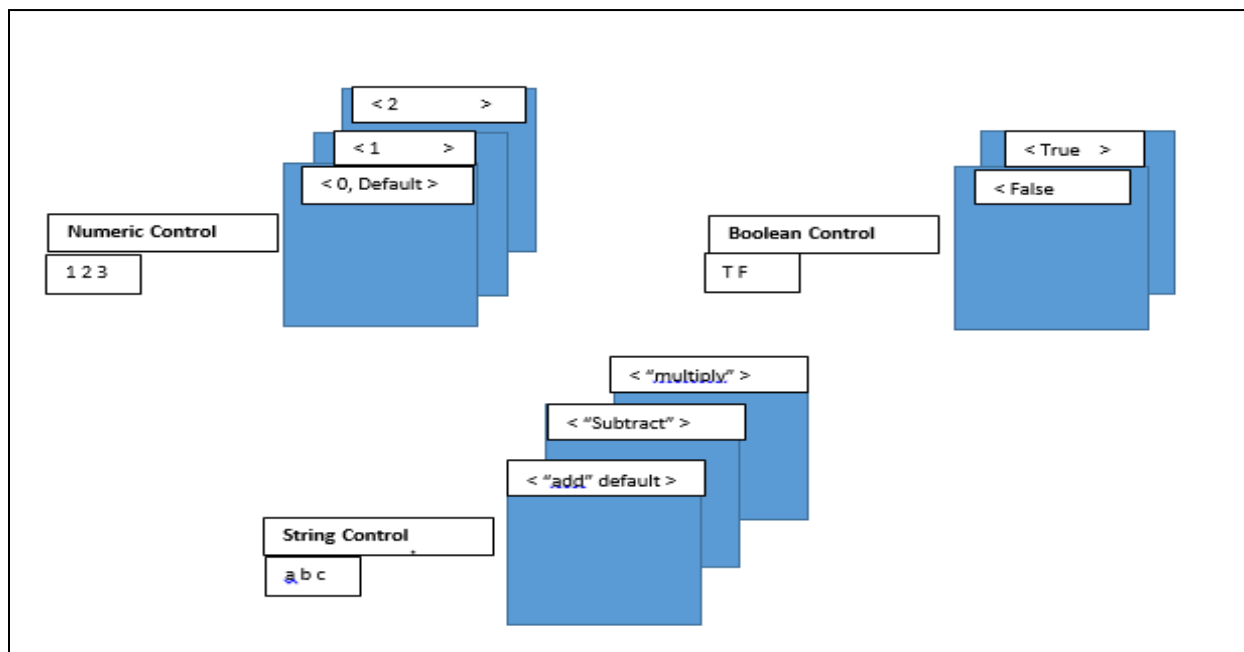

Figure15: Case structures and data types

A few things to remember about Case Structures and their data types:

➢ If the data type is Boolean, the structure can only have two cases: True or False.
➢ If the data type is Numeric or String, the structure can have up to 231 - 1 cases.
➢ If the data type is Numeric or String, the top or first case will have "default" in the menu at the top of the Case Structure.
➢ If the data type is Numeric, use integer values. All floating-point numbers are converted to integers and negative numbers are converted to zero.
➢ If the data type is String, the data values must be in quotes.
➢ Each case must have a sub diagram of some kind.
➢ If any case outputs data, *all* cases must output data through the output tunnel.

3.7.7 Cluster Functions

Two of the more important cluster functions are the Bundle and Unbundle functions [43].

3.7.7.1 The Bundle Function

The Bundle function is used to assemble individual elements into a single new cluster or to replace elements in an existing cluster. When placed on the block diagram, the Bundle function appears with two element input terminals on the left side. You can increase the number of inputs by enlarging the function icon (with the resizing handles) vertically to create as many terminals as you need. You can also increase the number of inputs by right-clicking on the left side of the function and choosing Add Input from the menu. Since you must wire all the inputs you create, only enlarge the icon to show the exact desired number of element inputs. When you wire to each input terminal, a symbol representing the data type of the wired element appears in the originally empty terminal. The order of the elements in the cluster is equal to the order of inputs to the Bundle function. The order of elements as shown on figure 16 is defined top to bottom, which means that the element you wire to the top terminal becomes element 0, the element you wire to the second terminal becomes element 1, and so on.
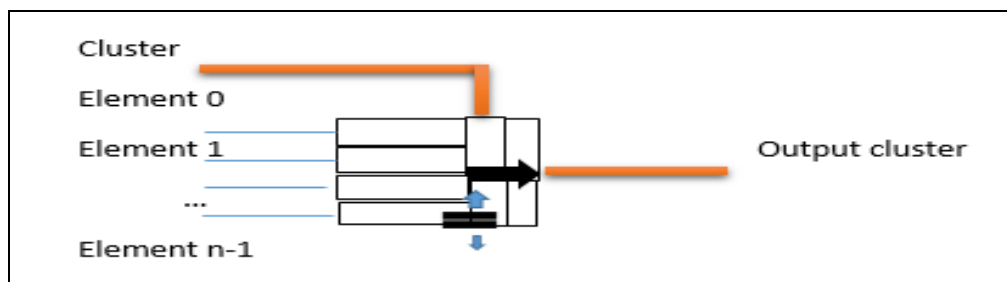

Figure16: Orders of Elements in Cluster

In addition to input terminals for elements on the left side, the Bundle function also has an input terminal for clusters in the middle, as shown in Figure above. Sometimes you want to replace or change the value of one or two elements of a cluster without affecting the others. Figure 16 shows a convenient way to change the value of two elements of a cluster, because the cluster input terminal (that is, the middle terminal) of the Bundle function is wired to an existing cluster named Cluster, the only element input terminals that must be wired are those that are associated with cluster elements that you want to replace in the illustration in figure 16 the second and fourth

element values are being replaced. To replace an element in the cluster using the Bundle function, you first place the Bundle function on the block diagram and then resize the function to show exactly the same number of input terminals as there are elements in the existing cluster that you want to modify. The next step is to wire the existing cluster to the middle input terminal of the Bundle function. Afterwards, any other input terminals on the left side that are wired will replace the corresponding elements in the existing cluster. Remember that if the objective is to create a new cluster rather than modify an existing one, you do not need to wire an input to the center cluster input of the Bundle function.

3.7.7.2 Unbundle Function

The Unbundle function as shown below on figure 17 extracts the individual components of a cluster. The output components are arranged from top to bottom in the same order as in the cluster. When placed on the block diagram, the Unbundle function appears with two element output terminals on the right side. You adjust the number of terminals with the resizing handles following the same method as with the Bundle function, or you can right-click on the right side of the function and choose Add Output from the menu. Element 0 in the cluster order passes to the top output terminal, element 1 passes to the second terminal, and one down the line. The cluster wire remains broken until you create the correct number of output terminals, at which point the wire becomes solid. When you wire an input cluster to the correctly sized Unbundle, the previously blank output terminals will assume the symbols of the data types in the cluster.
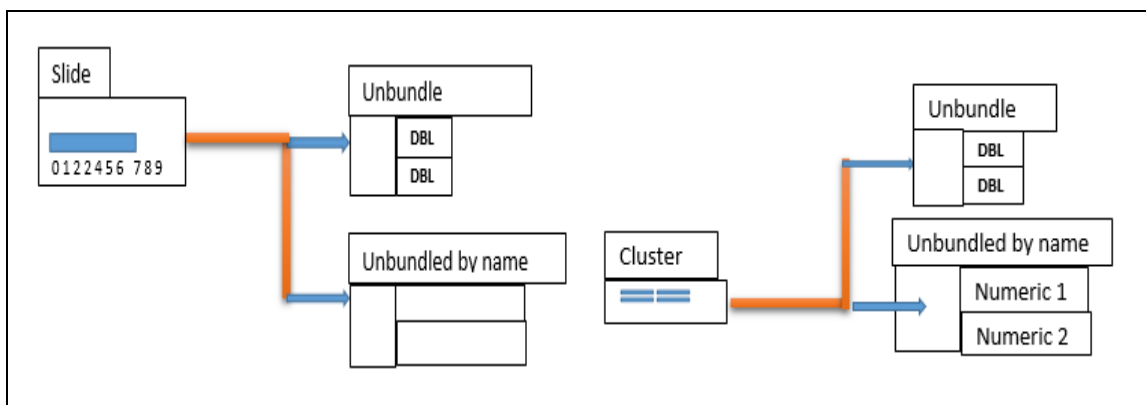

Figure17: Terminal wire connections

3.7.8 Graphical display

Graph displays one or more plots of evenly sampled measurements. Graph indicators are 2D displays of one or more data arrays which are called plots. There are two types of graphs: Waveform Graphs shown on figure 18 and XY Graphs on figure 19. Their Front Panel display looks the same, but they are functionally very different.

Both graphs often utilize Cluster data structures to control the incoming data display, so that Bundle function is frequently used. For Waveform Graphs, the bundled components include the initial X value, the delta X value (increment or spacing between X values), and the Y array data.
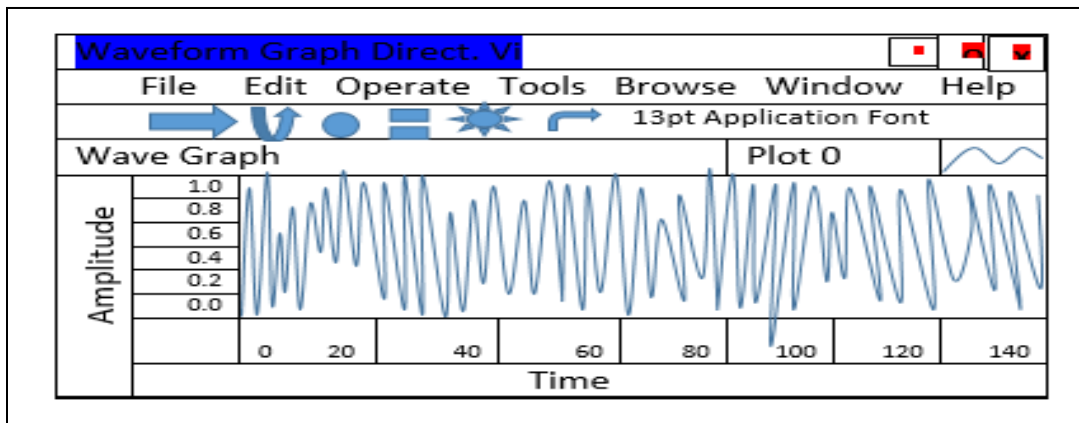
Figure18: Single-Plot Waveform Graph

3.7.8.1 XY Graphs

The XY Graph Indicator is a Cartesian graphing indicator used to plot multi-valued functions like circular shapes, mathematical functions or waveforms with varying time bases. Waveform graphs are designed to plot evenly-sampled waveforms. XY Graphs specify actual points' using their (X, Y) coordinates. The XY graph expects an input of bundled X and Y arrays, below is an example of an XY graph displaying a one-cycle Sine array versus one-cycle Cosine array on an XY graph, which is producing a circle.
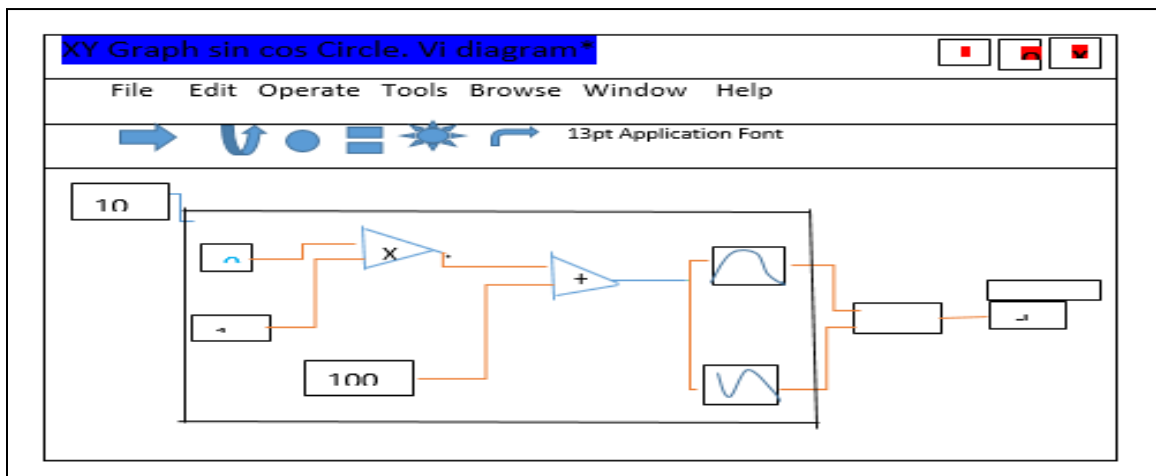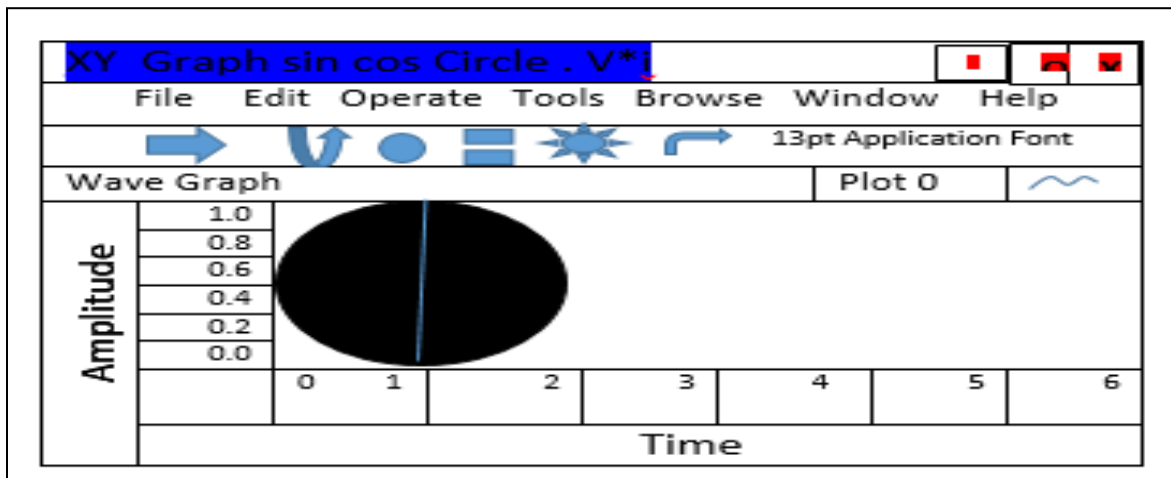
Figure19: X-Y Waveform Graph

3.8 System Calibration DAQ

The Virtual instrument designed to process the Heart sound input signals, before acquisition takes place as shown on figure 20 DAQ must be calibrated by launching the designed Virtual instrument LabVIEW on the computer and connected +5v powered NI DAQ6008/9 hard ware one end and USB cable on the other end to the computer directly. On the block diagram select function and express VI and input device select DAQ and drag over the block diagram to be visible as part of the design. Once it is accessible double click and choose voltage as input physical parameter and select Analogue input channel as an input and other parameters listed as per the design finally click ok the device will calibrate on its self automatically and ready to be used.
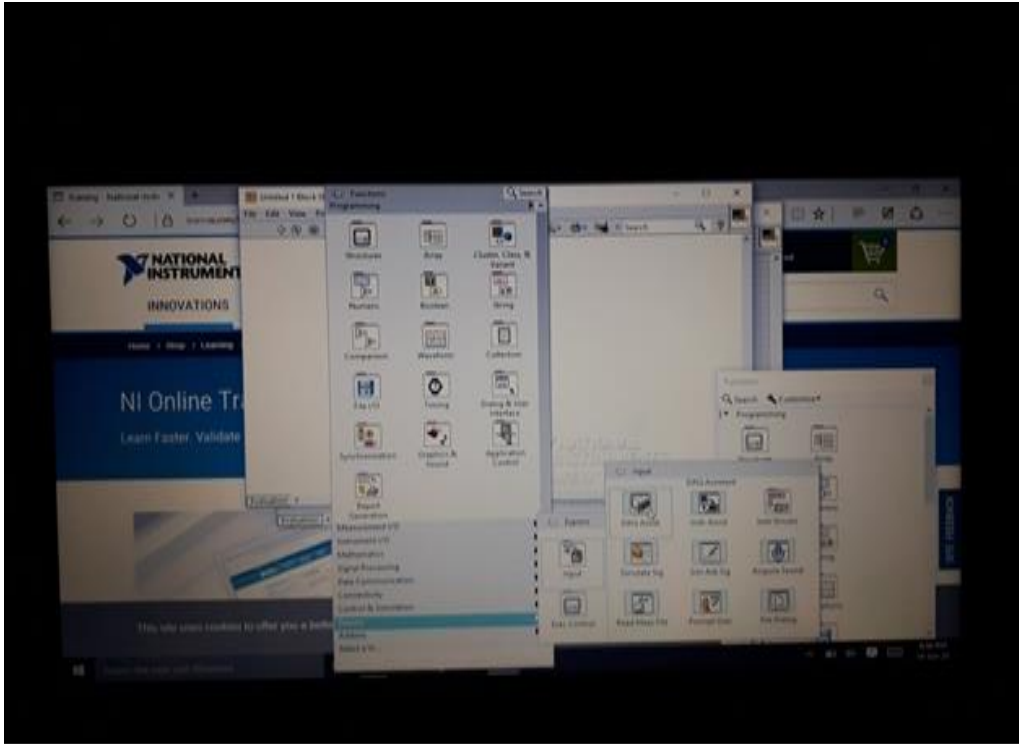
Figure 20: NI DAQ Calibration

The DAQ assistant shown on figure 21 will allow the user to configure the device on the virtual instrument directly by a drag and drop method once the DAQ is connected to the computer for signal acquisition and processing.
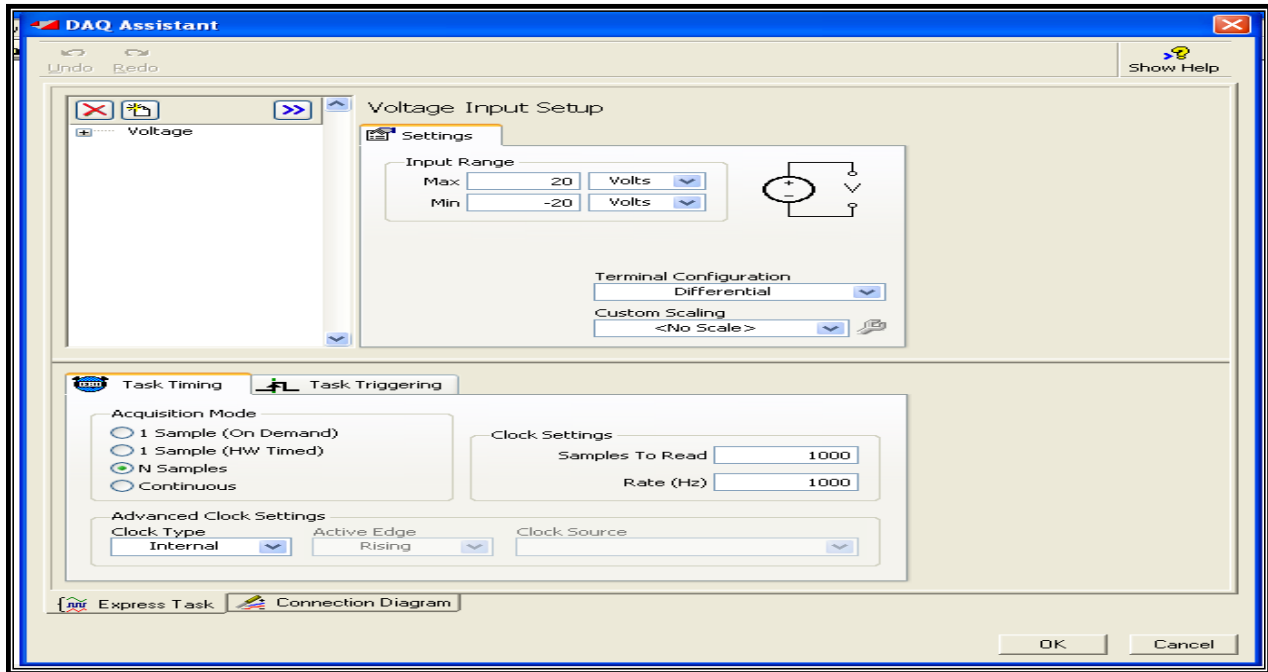
Figure 21: DAQ Assistant configuring the task

3.9 Description of Study Participants

The study used licensed data from Thinklabs online. Among 115 heart sounds used in our study , the abnormal heart sounds identified have been categorized in to the following as shown in the table 2 below,

Sl no	Subject Code	Characteristics	Pathologic Condition
1	001	Normal Pitch	Normal Heart Sound
2	002	High Pitch Systolic	Aortic Stenosis
3	003	High Pitch Systolic	Mitral Regurgitation
4	004	High Pitch Diastolic	Aortic Regurgitation
5	005	Low Pitch Diastolic	Mitral Stenosis

Table 2: AbnormalHeart sounds data

3.9.1 Historical Description of Murmurs

In addition to these Heart sounds, numerous heart murmurs may arise mainly from heart problems or diseases. The murmurs are extra or unusual sound heard during a heartbeat and broadly classified as systolic, diastolic and continuous [44] as well table 3 shows some common murmurs. During cardiac auscultation, the physicians are particularly interested in abnormal sounds, and various types of murmurs, which may suggest the presence of a cardiac pathology and also provide diagnostic information.The characteristics of murmurs could vary for different patients with the same cardiac dysfunction, or even change for a single patient during different cardiac cycles. Hence, advanced feature extraction and classification techniques with robustness to the murmur variations are highly required to make the system more reliable. The number and choice of features is one of the most critical steps to the success of a classifier [45].Table 3 shows the various types of Murmurs and their characteristics in comparison to one another.

Murmurs Type	Location	Timing	Pitch	Quality
Mitral stenosis (MS)	Apex	Diastolic	Low	Rumbling
Mitral regurgitation (MR)	Apex	Systolic	High	Blowing
Aortic stenosis (AS)	Apex/right upper sternal border	Systolic	High	Harsh
Aortic regurgitation (AR)	Right upper sternal border/left third/ fourth intercostal space	Diastolic	High	Blowing
Tricuspid stenosis (TS)	Lower right and left sternal borders	Diastolic	High	Rumbling
Tricuspid Regurgitation (TR)	Left fourth ICS	Diastolic	High	Blowing
Pulmonary stenosis (PS)	Left second ICS	Systolic	High	Blowing
Pulmonary Regurgitation (PR)	Second/third ICS	Diastolic	High	Blowing
Mitral valve prolapse (MVP)	Apex	Mid-late systolic	High	Blowing
Ventricular septal defect (VSD)	Left lower sternal border	Systolic	High	Harsh
Patent Ductus Arteriosus (PDA)	Left upper sternal border	Continuous	High	Harsh
Hypertrophic cardiomyopathy (HOCM)	Left lower sternal border	Mid-late systolic	High	Harsh

Table 3: Characteristics of Some Common Murmurs [ ]

3.10 Heart Sound Acquiring process

Heart sounds which consist of both normal and abnormal sounds acquired through a protocol from the patients are stored in a licensed Thinklabs. The stored data was used for testing the designed system to categorize the heart sounds into normal and abnormal sounds, and the abnormal sounds into different categories based on the disease. This raw data sources used in our thesis work as input signal in the absence of a clinical collaborator setup.

In normal scenario science state stethoscope as the tool to acquire Heart sounds with the added skill of the health care professional, environment, and the quality of the stethoscope which matters highly. In this work since the online abnormal heart sounds are recorded with scientific method data collection and acquiring signals. Hence, this insisted the thesis work to proceed without the need for patient physical interaction for normal and abnormal heart sound signal acquisition which will require more number of volunteer normal and sick patients, space, time, environment with different life style including exercise habits as some of the murmurs are none pathological and extra heart sounds .The data source utilized gave a chance to minimize the necessity of additional high technology hardware design for acquiring of data which is beyond the scope of this thesis work.

3.11 Experimental Setup

Experiment was done at Jimma University Institute of technology school of biomedical engineering bioinstrumentation laboratory as shown on figure 22. One Laboratory bench was assigned at the electronic lab which has functional generator, digital millimeter with power supply. There were two laptops in which one is used as source of input signal as well displays the online data visually as raw and the second one as data acquisitions and signal processing using NI DAQ 6008/9 connected by audio jack plugged at the audio hole of first laptop with analog in put AI0 DAQ and USB cable between the DAQ and second laptop. DAQ is powered by Functional generator +5v DC and ground as a source. The acquired input audio signals are passed to the second laptop for analysis and processing on the virtual instrument designed using LabVIEW. The sampled signal was used repeatedly five times each and the input signal was the same on the virtual instrument second laptop used for analysis compared to the first laptop used as a source of input signals for the thesis work for next step of signal processing for accuracy purpose.
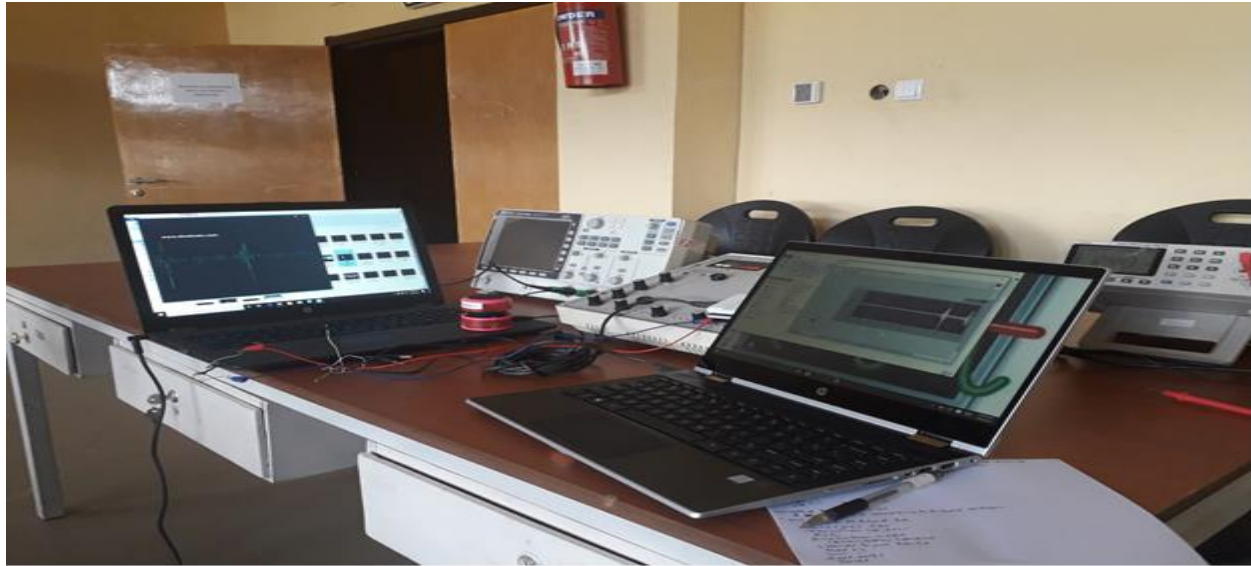
Figure 22: Interfacing DAQ & Heart Sound on LabVIEW Software

3.12 Data Analysis Techniques

LabVIEW designed virtual instrument processes the signal acquired first through DAQ then the input signal filtered in the preprocessing stage, once filtering is complete sampling technique used and enveloping occurs if the enveloped signals is greater than threshold value peak detection occurs which will pass the peaks at the limit set and this values will be displayed as wave form graph envelops between S1 and S2 with time and amplitude showing their characteristics difference.

3.12.1 Heart sounds signal Processing analysis and display using the virtual instrument

The acquired Heart sounds is often contaminated by noise and artifacts that can be within the frequency band of heart sounds and with similar character of heart sounds. In order to extract useful information from the noisy heart sounds, preprocessing of raw heart sound signals is needed. Heart sound signal processing can be roughly divided into two stages pre-processing and feature extraction [46]. The pre-processing stage removes or suppresses noise from raw heart sound signal and the feature extraction stage extracts the diagnostic information from the heart sounds signal.

3.12.1.1Preprocessing of heart sounds

Preprocessing of heart sound signals is to remove unwanted signals called interference from the Heart sounds [47]. The most common interference is the power line interferences and the base line wandering that can significantly affect the heart sounds. The power line interference is narrow-band noise centered at 50 Hz with a band width of less than 1 Hz. This type of power line interference is eliminated by using a programmable digital Butterworth band reject filter.

The baseline wondering is suppressed by wavelet transform approach method. The LabVIEW advanced signal processing tool kit (ASPT) provides WA Detrend Virtual Instrument (VI) which can be used to remove the low frequency baseline wandering (or trend) of a signal. An internal parameter called trend level is required for base line removal. The trend level is calculated as follows

$$\text{LEVEL}_{\text{Trend}} = \left\lceil \frac{\log_2 2t}{\log_2 N} \right\rceil \qquad (1)$$

Where t is the sampling duration and N is the number of sampling points in t time.

After the baseline wandering is suppressed, the heart sounds becomes more stationary and explicit than the raw heart sound signals. The remaining noise is complex stochastic process with wideband nature. To suppress the wide band noise, the wavelet Denoise VI module from LabVIEW is used.

This VI first decomposes the heart sounds signal data into several sub bands by applying the wavelet transform, and then modifies each wavelet coefficients by applying a threshold or shrinkage function, and finally reconstructs the denoised signal. In the present study, we have used undecimated wavelet transform (UWT) Syms with single level and soft thresholding for the wavelet denoising VI block setup.
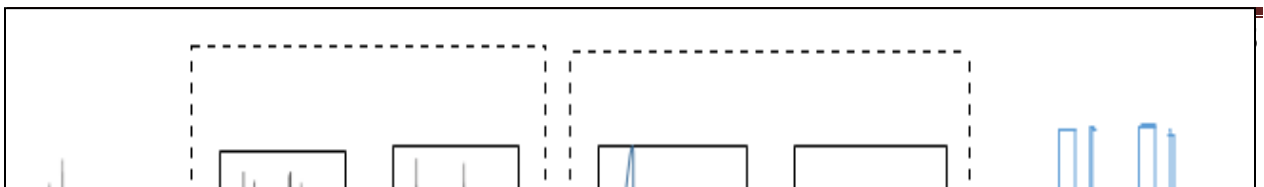
Figure 23: Schematic description of heart sound localization

These settings perform perfect denoising on the original heart sound signal and smoothing the signal without suppressing heart sounds features such as peaks of the first, second and so on. For effective feature extraction, we apply a wavelet Denoise VI with UWT sym5 and multiple level on the detrended signal to make only the peaks of first and second if possible third and fourth signal more distinct.

3.12.1.2 Performing Feature Extraction on heart sounds Signals

For the purpose of diagnosis, we often need to extract various features from the preprocessed heart signal data, including the peaks of first and second heart beats. The multiresolution analysis Express VI used decompose the heart sounds signal by 6 level, Daubechies 6 (db6) wavelets and then reconstructs the signal with D4,D5 & D6 sub bands. D4, D5 and D6 sub bands will be kept for reconstruction of heart signals because it makes more accurate for detection of heart sounds (beats)

3.12.1.3 Peak Detection

After detrending the heart signals and applying denoising VI, the resulting signal is applied to the wave peak detection VI. Peak detection is one of the most important time domain function performed in signal monitoring. Peak detection is the process of finding the locations and amplitudes of local maxima and minima in a signal that satisfies certain threshold value. In the

present study the waveform detector used records the locations of the signal peak above some threshold value in an output array in terms of indices.
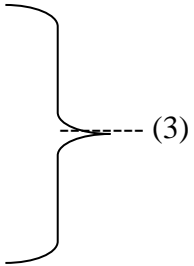
The peak detector function used has some important features, such as

➢ The function can process many blocks of data that are part of the same signal.

➢ This function retains the internal states and history information from one call to the next. This VI internally allocates the structure that contains the information on the first block of data and destroys it on the last block of data.

➢ The peak location function gives the peak locations at fractional indices. It uses the quadratic fit algorithm and returns the peak locations as floating point numbers, not as integer's index values.

➢ The function allows implicit noise reduction while finding the peaks. Using the width parameter effectively reduces the noise in the input signal when finding the peaks.

➢ The function performs a quadratic curve fitting to find the peaks. The core of the peak finding algorithm consists of fitting a parabola to successive groups of points, equal in number to width. The function uses the coefficients from the fit to determine whether a peak or valley is present.

➢ For each set of points, the algorithm performs the least-square quadratic fit, and then performs a series of tests on the coefficients to see whether they meet the criteria for a peak. The function checks whether each parabola is at local maxima, determines the sign of the quadratic coefficients, which indicate the parabola's concavity, and finally checks that the peak is above the designated threshold.

At the output of waveform detector both amplitude and time locations are taken into considerations for the construction of envelope. The envelope will be constructed for a sampled value of the input signal.

To develop the envelope, the following notations are considered

$$MPEAK = \text{Maximum Amplitude of the sampled signal}$$

$$MT1 \quad = \text{Time of the maximum amplitude} \quad \text{------ (3)}$$

$$THR \quad = \text{Threshold of the signal}$$

$$TSEC = \text{limits of the signal where peaks are measured}$$

The technique used for heart signal feature extraction is an approach which employs peak or valley detection and to construct envelopes based on the peak values detected which satisfies certain threshold conditions. The developed envelopes of the heart signals give an approximate representation of instantaneous magnitudes.

This is done because the envelopes are easier to process the heart signals while they retain enough information to detect the cardiac events.

1. Determine the MPEAK and MT1
2. Merge the peaks that are above THR that are very close to MT1 about (20 msec apart) to the MPEAK value
3. If the peaks are lower than the threshold ignore that and search for the new local maxima
4. Assign systole to the envelope that has short intervals which do not vary much over time
5. Assign diastole to the remaining intervals.
6. Assign S1 or S2 labels to each peak so that S1 signals the start of systole and S2 the start of diastole.

Chapter 4

Results and Discussion

4.1 Study Result

The following are the results of the selected five different online abnormal heart sound sample signals that were used in this work from figure 24 to 29 explained below.

4.1.1Result One

The waveform graph shows the normal acquired heart sound through the system and processed heart sound signal in an envelope form clearly marked on the front panel for easy diagnosis and interpretation. It also displays the occurrence of the firsts and second Heart sound as well heart beat is calculated and displayed in fig 24.

Figure 24: Normal Heart Sounds

From the waveform graph table 4 summarized and indicates the time and occurrence of heart sounds with murmur as analyzed from the waveform graph from figure 24 indicated above, this will allow the health professional to understand the cardiac abnormality easily as shown both on the figure and table with in time and properly visually rather than depending on audio signal using Stethoscope alone as well as Phonocardiograph.

The algorithm is able to detect and separate S1 and S2 components of the signal. The time duration of each sound is clearly seen in comparison to the other if any variation either long or short will tell the presence of abnormality in the occurrence of the normal heart sound.

SN	Parameter	Time in Seconds Figure 24
1	Duration of the first Sound	0.07
2	Duration of the Second Sound	0.05
3	Duration of the third Sound	---
4	Time Interval between two First Sounds	0.797
5	Time Interval between First and Second Sounds	0.240
6	Number of beats per minute	75
7	Occurrence of murmur	Nil

Table 4: Time Characteristics of Normal Heart sounds

4.1.2 Result Two

Recorded online abnormal heart sounds are acquired using the NI DAQ and feed in to the designed system and the resulting output is displayed as shown in figure 25 below. The result output when abnormal heart sound signal used from sources between the two normal hearts sounds a different peak and amplitude and noise seen with different duration visually which indicated as an abnormal heart sound called Murmurs compared to figure 24above.
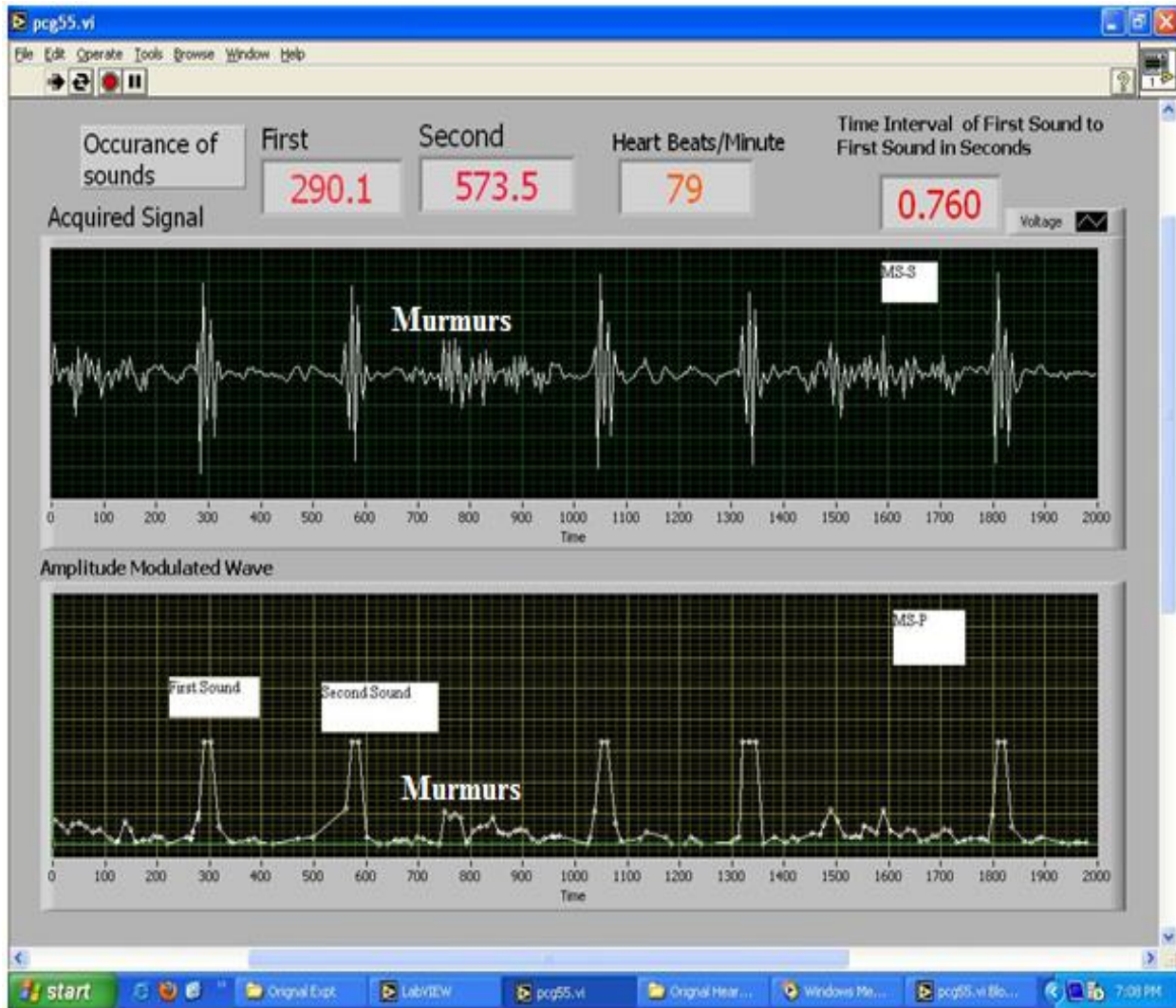

Figure 25: Mitral Stenosis

The results shown in table 5 are the analysis from fig 25, the heart murmurs mitral stenosis is displayed and clearly marked just after the second heart sound with different amplitude, and duration of frequency as the noise shows while the normal heart sounds are seen bigger and regular and unaltered shape with time.

SN	Parameter	Time in Second Figure25
1	Duration of the first Sound	0.06
2	Duration of the Second Sound	0.045
3	Duration of the third Sound	---
4	Time Interval between two First Sounds	0.760
5	Time Interval between First and Second Sounds	0.230
6	Number of beats per minute	79
7	Occurrence of murmur	Diastolic

Table 5: Timing characteristics of Mitral Stenosis

4.1.3 Result Three

The second abnormal heart sound signal used processed and analyzed by the system designed and displayed on figure 26, the noise an disturbance of the Murmur between the normal heart sound visualized clearly with peak and frequency as marked and labeled which will help the health care professional to analyze and make diagnostic intervention on the characteristic of the heart sound easily. The murmur has irregular shape and frequency pattern than the normal heart sound in the second part of the front panel shown envelop for decision making.
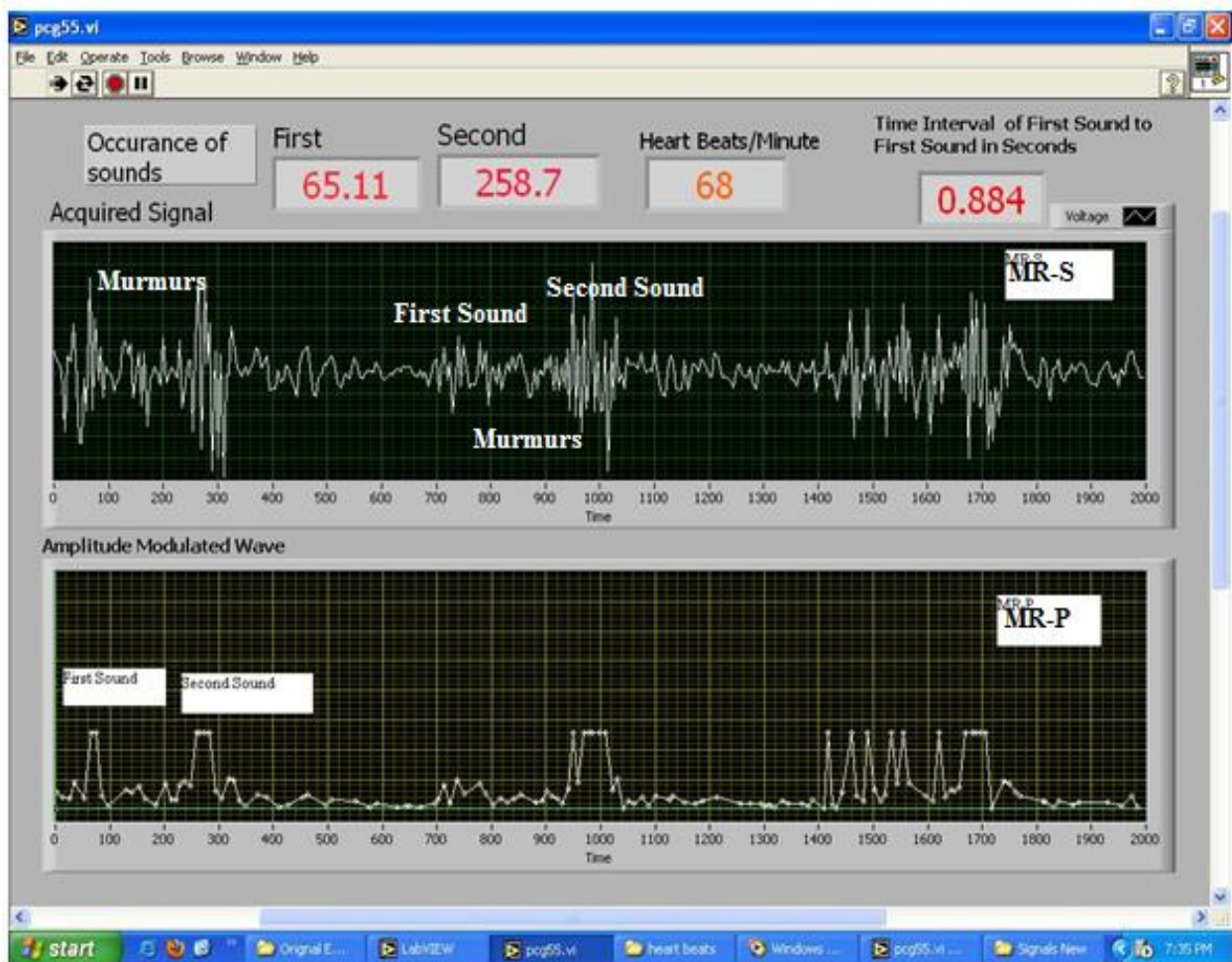

Figure 26: Mitral Regurgitation

In table 6 summarized the results of figure 26 the duration and occurrence of the heart sound and murmurs including beat rate displayed in the presence of Mitral Regurgitation which will allow decision making easily with the comparison made to phonocardiography.

The occurrence of the murmur location also signifies where the disturbance occurred for characteristic valvular problem to know exactly.

SN	Parameter	Time in Second Figure 26
1	Duration of the first Sound	0.06
2	Duration of the Second Sound	0.05
3	Duration of the third Sound	---
4	Time Interval between two First Sounds	0.884
5	Time Interval between First and Second Sounds	0.180
6	Number of beats per minute	68
7	Occurrence of murmur	Systolic

Table 6: Timing Characteristics of Mitral regurgitation

4.1.4 Result Four

In a similar pattern the fourth abnormal heart sound similarly to the others mentioned has been fed to the system processed and displayed the result in figure 27. The first upper part of the front panel showed the input raw abnormal heart sound signal in a wave form and the second lower part of the front panel showed the processed enveloped. Amplitude modulated wave form to visualize and analyze the waveforms clearly for characteristics identification with time and frequency and shape of the abnormality as well as the location in the cycle. It has shown that the waveform and the murmur occurrence between the two heart sound with smaller peak and irregular shape and duration relative to the normal heart sound.
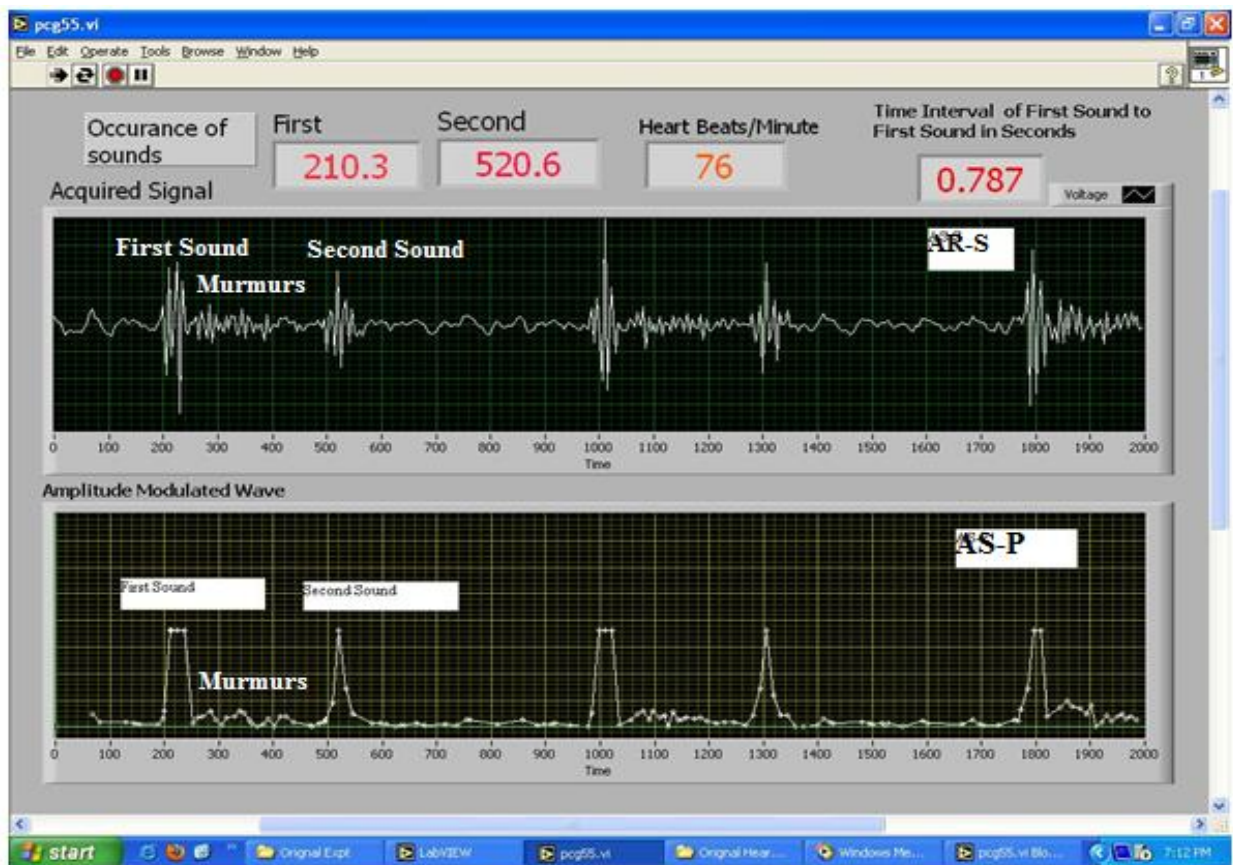

Figure 27: Aortic Stenosis

Aortic stenosis as one of the abnormal heart sound shown on figure27when analyzed it was found to be occurring in the systolic cycle of the heart sound in between first and second heart sounds as an important sign of where the problem occurred when location is of interest with time and shape.

SN	Parameter	Time in Second
		Figure 27
1	Duration of the first Sound	0.060
2	Duration of the Second Sound	0.040
3	Duration of the third Sound	---
4	Time Interval between two First Sounds	0.787
5	Time Interval between First and Second Sounds	0.250
6	Number of beats per minute	76
7	Occurrence of murmur	Systolic

Table 7: Timing characteristics of Aortic stenosis

4.1.5 Result Five

Aortic regurgitation, the fourth abnormal heart sound signal used in this work has been processed and analyzed in a similar way and visual display of the envelop and timing with heart beat has been visualized clearly by the method designed which was found helpful for investigation, diagnostic and treatment plan by the health care professional.
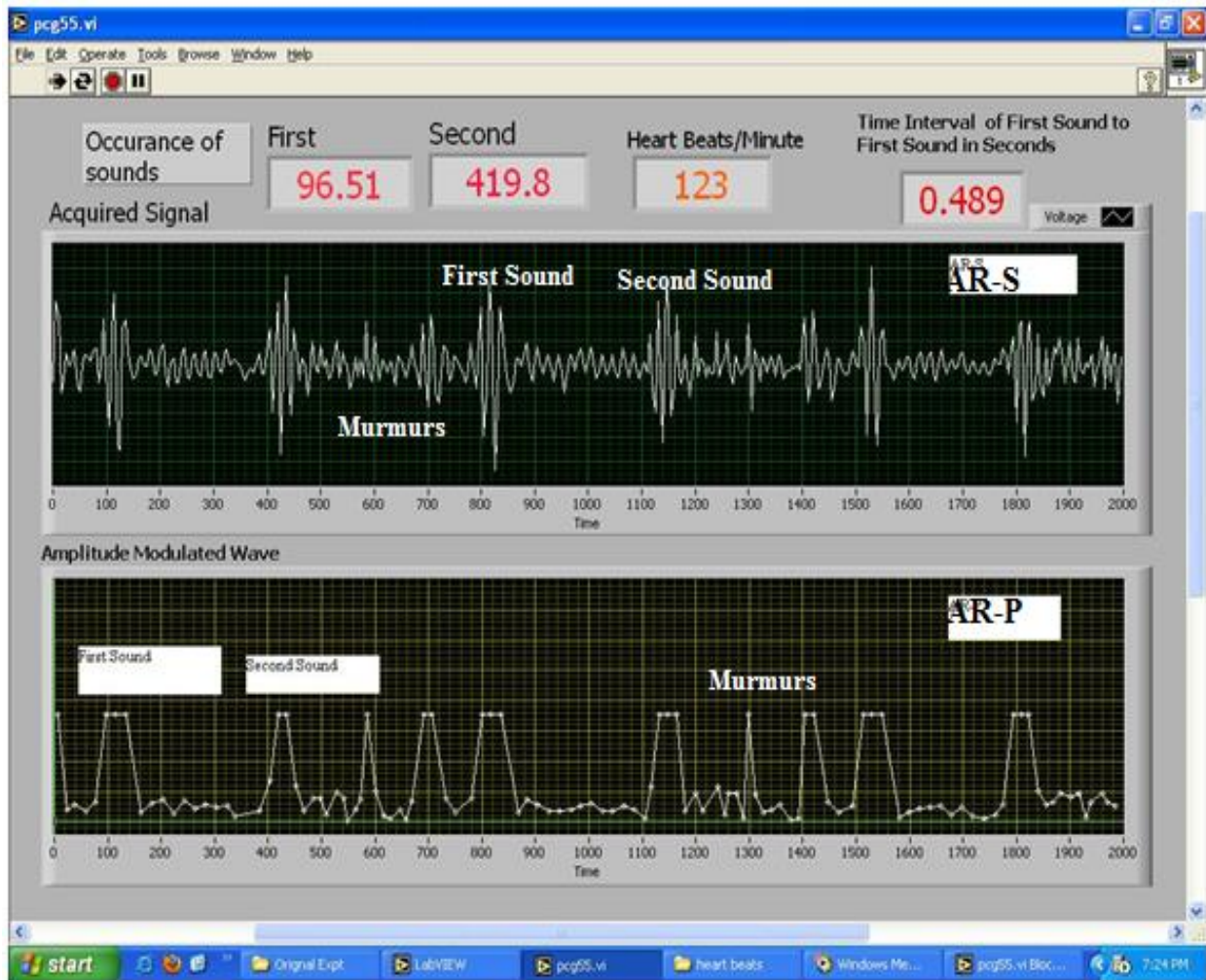

Figure 28: Aortic Regurgitation

In table 8 the results are summarized from figure 28 above and clearly indicated the time duration and occurrence of the murmur, Aortic Stenosis from the system.

SN	Parameter	Time in Second Figure 28
1	Duration of the first Sound	0.08
2	Duration of the Second Sound	0.08
3	Duration of the third Sound	---
4	Time Interval between two First Sounds	0.489
5	Time Interval between First and Second Sounds	0.240
6	Number of beats per minute	123
7	Occurrence of murmur	Diastolic

Table 8: Timing characteristics of Aortic Regurgitation

4.1.6 Result Six

Last but not least patent Ductus arteriole is the one of the most common abnormal heart sound focus on work among the other abnormities chosen. The input signal was fed and processed for analysis showed on figure 29 the waveform looks continuous and similar throughout in the upper part of the front panel as a raw input. The system analyzed the input signal and envelopes were made to clearly show the waveform displayed both the normal and abnormal heart sounds at a time all together the middle of big disturbance.
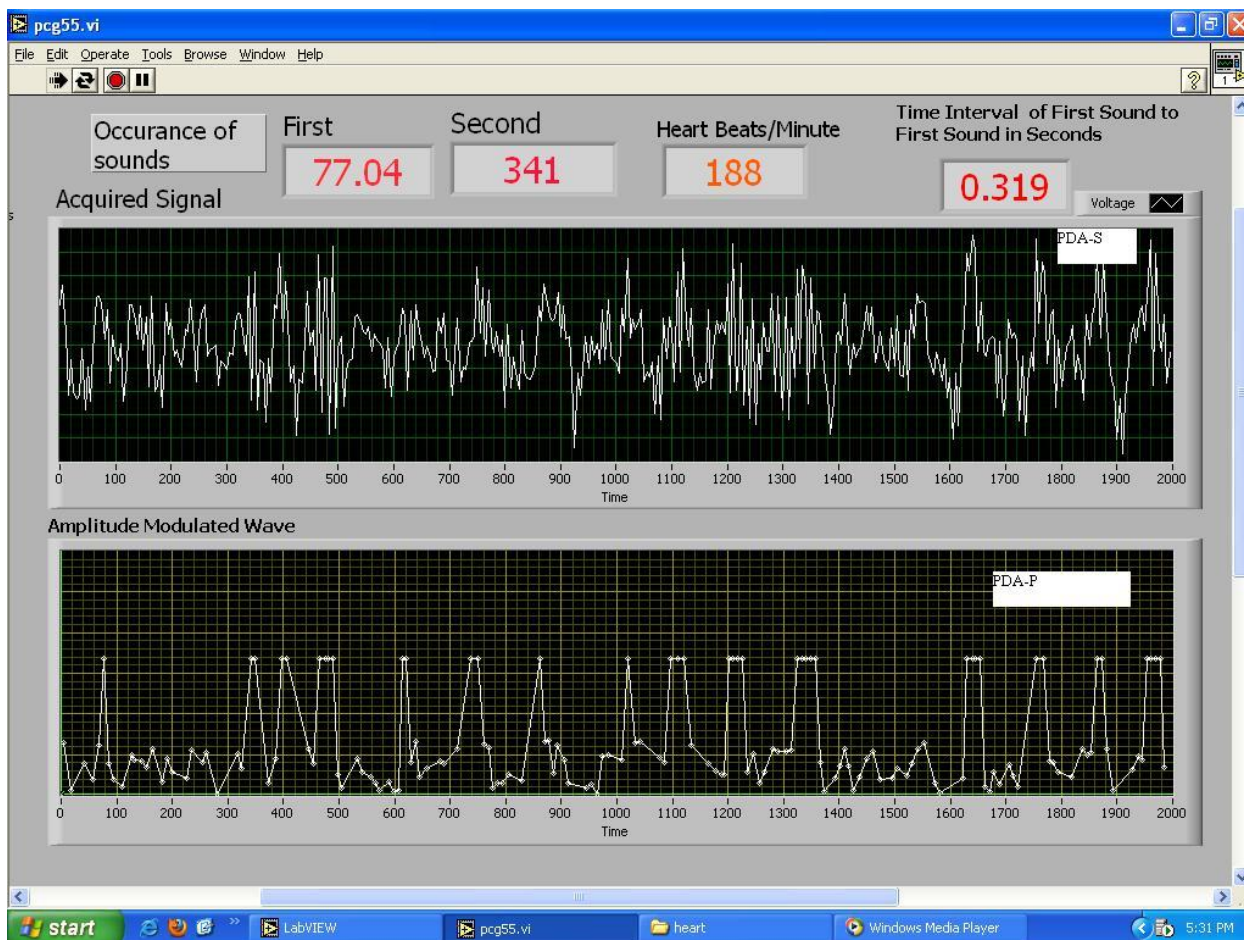

Figure 29: Patent Ductus Arteriosus

4.2 Comparison analysis with previous work

According to the method used in this work accuracy resolution precision is around 96%. The Virtual Instrument developed using LabVIEW for processing of the heart signals consists of tools which are configured using the settings available in each and every tool used. No calculation has been involved in processing and displaying the output of both normal and abnormal heart signals.

Sn	Author	Topic	Method	Efficiency
1	N Ruban et al, 2012	HS Analysis using LabVIEW	Separation coefficient used for a.wav form data	S1 S2 separated from signal No efficiency listed
2	Taner T 2008	Software Develop for the Analysis of Heart Beat sounds with LabVIEW in diagnosis of CVD	The time frequency scale spectral analysis technique STFT, WT to visualize F band activity	overcome the the deficiency of expert doctors
3	Fen Li 2019	Feature Extraction & classification of heart sound 1D convolution Network	Feature extraction by denoising auto encoder as an input to 1D CNN DWT STFT Hamming used to reshape in to small audio windows using FFT	High accuracy than back propagation model classification accuracy 99% , 87% accuracy 150 patients
4	Vladmire K, 2007	Heart Energy signature spectrogram for cardiovascular diagnosis	visual map HS energy established T-F transformation	reduce disadvantage of phonocardiograph doesn't need special filter

Table 9 : Comparison of Literature reviewed analysis of hearts sound

Most literature did not show the graphical representation of heart sound in an envelope like what this research work has performed in the entire course work.

Chapter 5

Conclusion and recommendation

5.1 Conclusion

Globally life style change has become inter related issues with the growth and frequently changing technology. Heart disease is the leading cause of death and hence needs more attention, early diagnosis and treatment by health care professional for patient during clinical examination.

Virtual instrumentation system using LabVIEW has made significant output and remarkable identification of abnormal Heart sound in relation to normal heart sound acquired at the same time from recorded input signals by converting the waveforms to envelops in which most researches didn't consider even though a lot has been said on the powerful benefit of LabVIEW in the analysis of heart sound.

As the main objective of this work is to identify abnormal heart sound using LabVIEW to help health care professional diagnose and treat cardiac problems for patients in time and proper interpretation since heart sounds are non-stationary and non-sinusoidal proper identification and interpretation is paramount and needs at most attention. The algorithm used is a user friendly and quick system which complies with the characteristic of heart sound signals acquired online as well from patients with proper clinical protocol. The method used has a promising output for the identification of abnormal heart sounds in clinical setups.

In this work DAQ, PC as hard ware instrument and LabVIEW based development platform were used. It has two parts data acquisition and pre-processing of the online recorded heart sound signal after filtering and followed by processing of abnormal heart sound in an envelope waveform graph form along with the normal heart sounds displayed for interpretation in which wavelet transforms were used for the analysis. The method employed for classifying the heart sound is sampling and Thresholding technique, if the amplitude of the sample value is greater than the threshold value the output of the sampled signal value is set to a positive pulse. If the sample value is less than the set Threshold value, then the output of the sample will be set to zero pulse. This process is repeated

for all the samples of the acquired heart signals which are processed by the designed bio VIRTUAL instrumentation system using LabVIEW.

All the signals were compared with the standard clinical heart sound recorded online clinically then processed to the work planned has proved to be efficient 96% which is open and easy to be used to all the different abnormal heart sounds obtained from real and online data recorded.

The strength of the work in comparison from other works is the utilization of software and hard ware to come up with the heart sounds being enveloped and displayed in a waveform for interpretation without advance mathematical techniques in an easy way to be understood. It has added value to the display of Phonocardiograph with time and frequency analysis of input signals compared from merely using audio auscultation. The opportunity to have access to DAQ and LabVIEW with experts support on the area of LabVIEW made and enabled this work easier. As a threat for inability, due to complexity and accuracy with limited scope unable to work on real heart sounds obtained from a patient in a clinical setup rather than on online internet clinically recorder heart sounds.

5.2 Recommendation

This thesis work would have considered to include input signal data obtaining from a patient online and processed in a clinical setup together with a clinical collaborator after designing a proper system/device to acquire heart sounds from both normal as well as abnormal patients.

Heart sounds are non-stationary, random and non-sinusoidal timely and efficiently diagnosis is most important thing and technologies available still lack a proper algorithm to support the health care professionals to utilize and help patients promptly without high technology and big investment. The designs need not consider underlying breath, movements and surrounding sounds characteristics and also the similarity of heart sounds on their own as there arevarious undistinguishable pathological heart sounds where signal loss accuracy of identification needs improvement.

In the future, scale up the work to PhD Level to record the heart sounds in a clinical setup involving real patients and with a clinical collaborator and process the signals online and display the result.

Designing and implementing efficient and accurate heart sound signal acquiring device at low cost which is independent of health care professional hearing capacity with better signal acquisition, processing and display capability is more feasible and important to invent.

In order to make the Virtual instrument designed to be practical, cost effective and functional Cardiologist, Biomedical Engineers and Industries needs to come together and perform clinical trials and avail the device for use. Finally, the system needs more in depth research improvement and participation to make the diagnostic technique more efficient with time and underlying diseases condition in away to easily understood by health care professionals in the field. The information provided in this thesis work will be valuable and beneficial to not only the researchers and engineers in the technical field, but also to medical professionals for clinical diagnostic and decision making in the era of COVID 19.

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