6. DISCUSSION

Passive listening (auscultation) has been used qualitatively by physicians for hundreds of years to aid in the monitoring and diagnosis of a wide range of medical conditions, including those involving the pulmonary system (breath sounds), the cardiovascular system (e.g., heart sounds and bruits caused by partially occluded arteries and arteriovenous grafts), and the gastrointestinal system. There may be unique and diagnostically important information in audible frequency sound since characteristic times for many physiological processes and anatomical structural resonances are in that range. This approach offers several potential advantages including noninvasiveness, safety, availability, prompt results, and low cost, making it suitable for in-office checkups, outpatient home monitoring, the emergency room (ER), and field operations following natural or man-made catastrophes. Simple stethoscopic use is skill dependent, provides qualitative rather than quantitative information at only a single location, and suffers from inherent limitations of human ability to discern certain acoustic differences.

The development of computer-based diagnostic decision support systems for the analysis and interpretation of heart and lung sounds depends critically on the electro-acoustic properties of the sensors, typically embodied in electronic stethoscopes, used to acquire the signals of interest. [(107)]

Electronic stethoscopes function in a similar way, but the sound is converted to an electronic signal which is transmitted to the listener by wire. Functionalities often included in electronic stethoscopes are amplification of the signal, filters imitating the function of the diaphragm and the bell and in some cases recording abilities to allow storage of data.[(108)]

The acoustic stethoscope operates on the transmission of sound from the chest piece, via air filled hollow tubes, to the listener ears. A typical stethoscope chest piece usually consists of two sides that can be placed against the patient for sensing sound – a diaphragm (Plastic disk) or bell (Hollow cup). The diaphragm and the bell work as two filters, transmitting higher frequency sounds and lower frequency sounds, respectively. If the diaphragm is placed on the patient, body sounds vibrate the diaphragm, creating acoustic pressure waves which travel up the tubing to the listener's ears. If the bell is placed on the patient, the vibrations of the skin directly produce acoustic pressure waves traveling up to the listener's ears. The bell transmits low frequency sounds, while the diaphragm transmits higher frequency sounds. This 2-sided stethoscope was invented by Rappaport and Sprague in the early part of the 20th century. The drawback of the acoustic stethoscope is that the diagnosis accuracy is limited by the listening capability of human ears, which is between 20Hz – 20 KHz. And heart murmur detection using a stethoscope is very subjective and success rates vary enormously from physician to physician e.g. from 20% for a trainee to 80% for an expert cardiologist. [(109)]

In the present work, a type of stethoscope that is very advantageous over traditional acoustic stethoscopes. Digital stethoscope electronically amplifies body sounds. Digital stethoscope uses sound waves, which are converted from analog to electrical signals that are amplified and provide much lauder and clearer sound. This makes identifying problems quicker and easier, and it allows the recording and playback of sounds picked up, making it useful for references or in the teaching for students.

There are multiple types of sensors that can be used in the chest piece of an electronic stethoscope to convert body sounds into an electronic signal. Microphones and accelerometers are the common choice of sensor for sound recording. These sensors have a high-frequency response that is quite adequate for body sounds. Rather, it is the low-frequency region that might cause problems. The microphone is an air coupled sensor that measure pressure waves induced by chest-wall movements while accelerometers are contact sensors which directly measures chest-wall movements. For recording of body sounds, both kinds can be used. More precisely, condenser microphones and piezoelectric accelerometers have been recommended. (110)

In the present work, an electret condenser microphone combines a condenser microphone with a Field Effect Transistor (FET) were used for the digital stethoscope system. Because of its low cost, availability, and small weight and volume, it amplifies the signal and transforms the impedance to a more useful level.

The large variety of segmentation algorithms for heart sounds available in literature makes the complete description of the existing methods impractical.

M. El-Segaier et al. developed a method based on ECG gating. This method uses the simultaneous acquisition of PCG and ECG signals. Using an envelope-based detection algorithm, the R-waves of the ECG were detected and the distance R-R computed. The T-waves were also computed. The Short-Time Fourier Transform was then used to obtain the spectrum of the PCG. Using the temporal relations between the PCG and ECG, intervals of search for S1 and S2 were defined and the maximum in the spectrum in each of those intervals was defined as S1 or S2. An additional tool was also developed to determine if the maximum obtain formed a well-defined peak in the time domain. (111)

H. Liang et al. designed a segmentation algorithm dependent on the Shannon energy envelope. A threshold is set to select the peaks from the Shannon energy envelope. Time relationships between the obtained peaks are evaluated to reject extra peaks and to recover low-amplitude heart sounds that were not obtained due to the threshold used. The S1s and S2s are then separated by comparing the lengths of systoles and diastoles. (112)

J. Martínez-Alajarín et al. developed a segmentation method dependent mostly on time domain analysis. Using the amplitude, energy and frequency envelopes, and the heart rate is found through the ACF, a simple function specialized in finding the periodic elements of a signal. The amplitude envelope and a series of empirically defined rules is used to find the events of interest, the heart sounds. (113)

H. Naseri et al. developed a method based both on time domain and time-frequency analysis. A specific function was designed to be sensitive to high amplitudes and the specific frequencies of the main heart sounds. This was done using the Fast Fourier Transform. The peaks of the envelope function obtained were then considered as candidates for heart sounds. Their shape and duration were also evaluated and, if validated according to these parameters, the events could be classified into S1 or S2. This process was done iteratively along the PCG signal. (114)

In the present work, heart and lungs sound signals were amplified and filtered by the constructed digital stethoscope system. Signals were recorded and analyzed in time domain to show that robust and reliable features of normal and abnormal heart sounds can be extracted. And frequency domain to gain more information for the audio signal measurements: spectral information, or knowledge about the frequency content and behavior of the audio signals. There exists a defined technique for converting, or transforming data from the time domain into the frequency domain, where information exists about the spectral content of signals. The Fourier transform allows us to convert a time signal to the frequency domain, meaning the spectral data contains information about both the amplitude and phase of the sinusoidal components that make up the signal.

Arathy R, et al. showed that how to classify the real heart sound audio which is also known as "beat classification" into one of four categories as Normal, Murmur, Extra Heart Sound and Artifact. A dataset, which contains many files in .wav format for murmur, extra systole, artifact and normal, was chosen. The audio files are of varying lengths, between 1 second and 30 seconds and some have been clipped to reduce excessive noise and provide the salient fragment of the sound. When an FFT study of these where conducted it has been observed that most information in heart sounds is contained in the low frequency components, with noise in the higher frequencies. Hence it is common to apply a low-pass filter at 195 Hz. Fast Fourier transforms are also likely to provide useful information about volume and frequency over time of each classes. The database consists of heart sounds of four different classes they are murmurs, extra systole, artifact, and normal. Initially these values of each files are then stored separately. FFT's of each file are also stored separately. Then plotted FFT's and find that plot is different for murmur, extra systole, and artifact and normal. (115)

In the present work, the heart sounds were collected from the university of Michigan website that was digitized in a .wav format. The collected heart sounds are of length of 30 seconds and were auscultated from four auscultation points which are:

- Apex area – supine
- Apex area – left ducubitus
- Aortic area
- Pulmonic area

After amplification and filtration from the constructed system, the information regarding the acquired signals is:

- Normal heart sound S1 and S2 from different auscultation points.
- All types of systolic murmur (early systolic murmur, mid systolic murmur, late systolic murmur, holo systolic murmur.
- Extra heart sounds S3 and S4.
- Split of heart sounds S1, and S2.
- Diastolic murmur.
- Opening snap.

Lung sounds provide vital information about respiratory health and disease. Lung sounds can be classified as normal sounds or adventitious sounds. The presence of adventitious sounds usually indicates a pulmonary disorder. The adventitious sounds are of two types: continuous and discontinuous. Continuous adventitious sounds are wheezes and rhonchi. Wheezes have dominant frequency, usually .100 Hz, and duration.100 milliseconds. Wheezes are due to airways obstruction in the lungs. Rhonchi are .250 milliseconds in duration, low-pitched; with dominant frequency of #200 Hz. Crackles are short, explosive discontinuous sounds with duration, 100 milliseconds. Crackles are produced either by pressure equalization or by change in elastic stress resulting from sudden opening of closed airways in the lungs. [116]

Sahgal et al. designed a system that monitors these respiratory sounds using LabView software. The program developed calculates the respiratory rate, displays the time expanded waveform of the lung sound, and computes the fast Fourier transform and short-time Fourier transform to present the power spectrum and spectrogram respectively.[117]

In the present work, lung sounds were downloaded from 3M™ Littmann® Stethoscopes website and recorded by the constructed digital stethoscope system. The lung sounds that were used in the study are:

- Normal vesicular sound
- Coarse crackles
- Inspiratory stridor
- Pleural friction
- Wheezing

These lung sounds were recorded, analyzed and stored using Audacity software for displaying the signal in time domain and computes the fast Fourier transform to present the frequency spectrum of the sound signal.

Grenier et al. designed an electronic stethoscope to overcome the disadvantages of the acoustic stethoscopes. It was designed to have a uniform frequency response and to amplify the sound level. The electronic stethoscope is comprised of a chest piece, sound transducer, adjustable gain amplifier, frequency filters, mini-speaker/head phones, and a dry cell or battery. The chest piece consists of a sound transducer (microphone) that converts the sound to an electrical signal, and this converted electrical signal is transmitted to the conditioning circuit, which may consist of an amplifier and a frequency filter. This conditioned signal then is transmitted through an electrical cable to a headset. [97]

In the present work, digital stethoscope system was designed and constructed to overcome the limitations of traditional stethoscopes. It consists of a signal generator circuit for generating a sound signal with appropriate frequency and waveform to the constructed model by a mini-speaker. Electrets condenser microphone receives the sound signal and converts it into an electric signal. The signal then was amplified and filtered with a suitable gain and transmitted through an electrical cable to the sound card of a personal computer (PC) for further monitoring and analysis of the output sound signal.

Mulligan et al. presents the design and prototype testing of a novel medical instrument designed to measure changes in to acoustic transmission properties of lung tissue. Since tissue acoustic transmission is largely determined by the distribution of lung fluid and lung tissue density, this instrument has potential applications for monitoring and diagnosis of patients with such obstructive lung diseases which are associated with accumulation of lung fluid and collapse of lung tissue. The apparatus consists of an array of 4 electronic stethoscopes linked together via a fully adjustable harness. A White Gaussian Noise (WGN) input sound is injected into the mouth via a modified speaker and measured on the surface of the chest using the array of stethoscopes. Data were analyzed using the Normalized Least Mean Squares (NLMS) adaptive filtering algorithm to develop a transfer function based on the propagation characteristics of the injected signal. This transfer function is then analyzed to determine the frequency response and the propagation delay at each stethoscope. [118]

In the present work, lung model was developed to simulate the human lung and trachea. An array of two chest pieces of an acoustic stethoscope connected to two electrets condenser microphones; one was placed under the lung model, and the other placed at one side of the model. The input sound signal was injected into the simulated trachea by a mini-speaker and received it by the stethoscope array. The optimum input frequency was 700 Hz in sawtooth waveform. Two mediums were used in the designed model to simulate fluids that may cause obstructive lung diseases to the human lungs which are water and ultrasound gel to simulate the mucous. Different solutions (sodium chloride, glucose) with different concentrations were also used. As the quantity of water or the ultrasound gel increases, the amplitude of the output signal increases, and therefore the attenuation of the signal decreases. There's no response with the change in solution concentrations.

The output signals were analyzed using multi track sound editor (Audacity) software to the fast Fourier transform analysis for each medium and quantity. It is a free, open source audio program for recording, editing and analyzing sound by displaying its waveforms and computes their discrete Fourier transforms (DFT) by the fast Fourier transform (FFT) analysis to reliably reveal frequency components.