1. INTRODUCTION

Auscultation is a technique used to measure sounds within the body. [1] From 1819 to the present, sounds within the body continue to be measured using a stethoscope to examine the cardiovascular [2], respiratory [3], and gastrointestinal systems. [4]

The term stethoscope literally translates to “chest-seeing device” and is used for a quick and noninvasive means of diagnosis by relating biological sounds heard to patient physiology. Invented in 1816 by R.T.H. Laennec, the stethoscope has grown in sophistication substantially [5].

In September 1816, during a cool morning, while walking in the courtyard of the Le Louvre Palace in Paris, Dr. Rene Theophile Hyacinthe Laennec, a 35-year-old French physician, observed two children sending signals to each other using a long piece of solid wood and a pin. With an ear to one end, the child received an amplified sound of the pin scratching the opposite end of the wood. Later that year, Laënnec was called to a young woman with “general symptoms of a diseased heart.”[6, 7] Both application of his hand to the chest and percussion offered little diagnostic assistance. Laënnec was reluctant to start immediate auscultation (placing the doctor’s ear on the patient’s chest) because of the age, sex and plumpness of the patient. In this moment of embarrassment, Laënnec recalled his observation of the children’s wood borne signaling. It was this observation that inspired Laennec’s invention of the stethoscope. [8]

Fig. 1: Early monaural wooden stethoscope

Laennec proceeded to experiment with various wooden cylinders and rods, finding to his satisfaction that a solid rod placed between his ear and the patient's chest significantly improved sound transmission, and that a rod pierced with a narrow bore was an even more effective sound conductor. [9]

While the monaural stethoscope, in allowing doctors to practice mediate auscultation rather than employing the ear-to-chest method, was a very useful invention, it had its drawbacks. Because doctors still had one ear open to their surroundings, they had to learn to concentrate hard in order to focus on hearing what they needed to through their instrument. Not all doctors had the ability to do this as well as they should, which contributed to the development of the binaural stethoscope. (10)

Irishman Nicholas Comins designed the first binaural stethoscope in 1829. It was made of brass tubes connected by movable joints with coiled silk inside the joints to give an airtight seal. Commercial binaural stethoscopes first became available in the 1850s. Dr. George Cammann designed a binaural stethoscope that became widely used. He never patented his design, believing it should be available to everyone. His original model, designed in 1855, had ivory earpieces that were connected to two curved metal tubes joined by a metal hinge joint. An elastic band was put on the earpieces to hold them on the doctor's head. A spiral tube covered with wound silk connected the two tubes, which then joined into a hollow ball, which in turn was attached to a conical-shaped bell chest-piece. Because the two short flexible tubes connected directly to the bell, physicians using these early binaural stethoscopes had to lean close to their patients. (11)

In 1832, a monaural stethoscope with a longer tube of cloth-covered spiral wire was developed, but stethoscopes with long flexible rubber rubes were not available until the 1890s. In 1885, the "Ford's Bell" chest piece was invented. It is a slim cone that is usually composed of metal with a base of wood, ivory, ebony, or gutta-percha to avoid chilling the patient's skin. Two tubes connect the apex of the cone to the two long rubber tubes and earpieces. (11)

All of these stethoscopes chest pieces with a funnel or trumpet shape, known as bell stethoscopes. No significant modifications were made to this basic style until the invention of the diaphragm chest piece, which had a rigid covering over the end. In 1851 in Cincinnati, Dr. Marsh patented a stethoscope that had a flexible membrane stretched over the end. (12)

In 1889 James Murray described the rubber ring which could be fitted to the bell-end, and claimed that it fitted better to the chest-wall, quite apart from being warmer for the patient. The old monaural forms continued to go through many modifications. Sometimes the straight stem was graduated in centimeters to measure Kronig's area, and sometimes the ear-piece was grooved and rubber-shod to allow it to be used as a percussion hammer, plessor, or knee-jerk hammer; and this type was known as the 'Burrows pattern'. Stethoscopes were even made with special bell-ends, to facilitate intercostal auscultation of emaciated consumptives. (13)

The simple diaphragm chest-piece was not illustrated before the development of the 'phonendoscope', and the first such illustration of a vulcanized diaphragm fitted to a bell-end is in a catalogue of 1900; although James Edward Pollock, of the Brompton Hospital, had mentioned an American attempt to introduce a 'chest-piece membrane' as early as 1850. Sahli referred to the early use of a rubber membrane in 1906; but Samuel Jones Gee, of St Bartholomew's Hospital, who produced six editions of a standard work on auscultation between 1870 and 1907, said that the 'microphone', as he called the membrane, had so far proved of little value in auscultation. In 1894, R C M Bowles, an engineer of Brookline, Mass., patented the modern form of diaphragmatic chest-piece. This was the first, or one of the first models to have a metal or celluloid diaphragm. It was sometimes described as a 'phonendoscope', or 'resonating stethoscope'. At first these diaphragms were shaped like flatirons, and then later they became rounded. (13)

Bianchi and Bazzi, in 1894, developed the first stethoscope with a rigid diaphragm. The rigid diaphragm attenuates low-pitched sounds, accentuating faint, high-pitched murmurs present with slight aortic regurgitation. The combination of bell and diaphragm chest pieces can adequately discern the acoustic ranges needed in examining the heart, but electronic amplification can offer even more clarity when using a stethoscope. At the turn of the century, scientists attempted to improve auscultation with acoustic and electroacoustic devices. The acoustic attempts failed, but an electroacoustic stethoscope was developed which was like a telephone with a carbon-granule microphone that modulated an electric current which then excited a telephone receiver. However, excessive distortion made this instrument impractical and useless. (12)

Fig. 2: Cammann binaural stethoscope

Some modifications were designed for teaching, and as early as 1850 Landouzy suggested a multiple stethoscope. It had a number of articulated wooden tubes, forty-eight inches long, and enabled several examiners to listen at once; or, alternatively, one or more could use two tubes and so have a binaural stethoscope. He called this a 'stethograph', but it was never a practicable thing, even with rubber tubes, and was eventually discarded. Some time before 1907 Dr. Aitchison Robertson connected a number of binaural tubes to one collector, enabling ten or twelve people to listen at one time. This was said to be useful in cases of aortic aneurysm, where frequent examination was inadvisable. In 1926 Dr Jenner Hoskin described the various models designed for multiple uses, including the 'multiple electrical stethoscope'. Probably the first of these was devised by Einthoven of Leyden in 1907. In 1923 Robert Cabot used a 600 foot cable to transmit through an ordinary ear-piece the heart sounds of a patient in a ward to an audience in a lecture room. (13)

Frederick and Dodge (14) first recognized that the stethoscope was deserving of acoustical study in 1924. They studied the intact stethoscope, but their data understandably reflected the limitations of acoustical test instruments of the time.

In 1926 Sato and Nukiyama described their 'magnoscope', an electrical stethoscope with a three-stage triode amplifier. The modern form of 'Bowles-Sprague' instrument, with a chest piece combining a bell and diaphragm, was described by Howard Sprague in 1926, and this was destined to have a long vogue. In 1937 William J Kerr and his colleagues described their 'symballophone', a modified stethoscope for the lateralization and comparison of sounds. (13)

In 1940, Johnston and Kline (15) made an objective acoustical study of stethoscope components. Their test method was physiologically oriented and employed a sound source implanted within a cadaver heart. They concluded that the design of the chest piece was an important determinant in shaping the response of a stethoscope.

Rappaport and Sprague (16, 17) studied stethoscope tubing. They interpreted their data to indicate that the physical properties of tubing had considerable influence on stethoscope efficiency.

Groom (18, 19) investigated stethoscope performances through well-executed subjective studies. He called attention to the importance of well-fitting earpieces and cited the impairment of stethoscope performances caused by air leaks and ambient noise levels.

There continued to be modifications of pre-existing designs as well as introductions of new designs, some of which were considered impractical while others proving to be useful. However, the invention of the binaural stethoscope in the 1850's has left its mark in history, as this ingenious invention is more or less the modern day form that is used today.(20)

The first types of sensors used to record lung sounds were merely amplified stethoscopes, but high levels of background noise and frequency band distortion soon led to their rejection. Experiments with carbon microphones, cord galvanometers and various optical devices also proved unsatisfactory. (21)

Most researchers in the 1970s chose to work with phonocardiographic microphones(22, 23) although it rapidly became clear that they were ill-adapted to the low sound levels and relatively broad spectrum of frequencies generated by breathing.

By the mid-1970s, other types of microphone were being adopted; condenser and externally polarized microphones, or electret microphone. A few research teams experimented with dynamic microphones, (30) accelerometers, and various piezoelectric contact microphones, but the most suitable type was generally considered to be the condenser microphone. (21- 23)

In 1955, MCKUSICK et al. (24) employed condenser-type microphones and recorded the resulting electrical signals on a magnetic disc, which could be read several times successively, each time scanning at a different frequency with a variable filter. The corresponding signal strength of each frequency band was then translated into a proportionate level of light intensity, and the data thus recorded on light-sensitive paper.

A comparison of 6 different stethoscope models in 1992 examined the transfer function of the instruments for sound frequencies between 37.5 and 1000 HZ and demonstrated that, in most cases, low frequency sound (37.5-112.5 Hz) was amplified by bells and attenuated by diaphragms. (25) However, there were only small differences in sound transmission between stethoscopes, and at least one model had good similarity between bell and diaphragm. More recently, another group examined the performance of the bell and diaphragm of a Littmann Classic II SE stethoscope and found that sound transmission in the frequency range of 20 – 400 Hz (where most lung and cardiac sounds are found) was superior for the diaphragm. [26]

The electrical stethoscope was produced by the 3M(TM) Littmann Company in 1999, after which the audio files could be saved. In 2000, the vibration response imaging system was introduced into medicine. In 2006, a noise-free active stethoscope was displayed in the Acoustics Conference held in Honolulu, USA. This kind of stethoscope could be used in noisy or bumpy circumstances. These new pieces of equipment made the stethoscope's application field even broader. [27]

Several forms of electronic stethoscopes have been developed to replace the conventional acoustic stethoscope. Basically, the goal of the electronic stethoscope is to improve sound resolution, allow variable amplification of the sound, minimize interference noise, and also provide data for visualization and storage. [28]

2. BASIC CONSIDERATION

2.1 Anatomy and physiology of the lungs:

The lungs are soft, spongy, cone-shaped organs located in the thoracic cavity. The right and left lungs are separated medially by the heart and the mediastinum, and they are enclosed by the diaphragm and the thoracic cage. [29]

Each lung occupies most of the thoracic space on its side and is suspended in the cavity by a bronchus and some large blood vessels. These tubular structures enter the lung on its medial surface through a region called the hilum. A layer of serous membrane, the visceral pleura, is firmly attached to the surface of each lung, and this membrane folds back at the hilus to become the parietal pleura. The parietal pleura, in turn, forms part of the mediastinum and line the inner wall of the thoracic cavity. There is no significant space between the visceral and parietal pleurae, since they are essentially in contact with each other. The potential space between them, the pleural cavity, contains only a thin film of serous fluid that lubricates the adjacent pleural surfaces, reducing friction as they move against one another during breathing. This fluid also helps hold the pleural membranes together. [29]

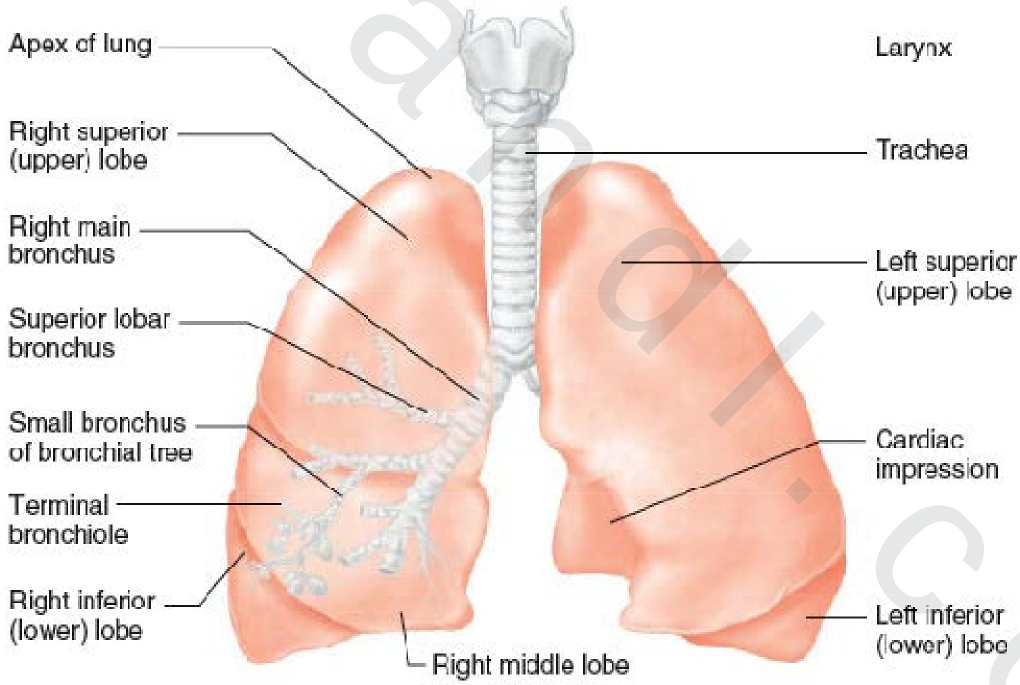


Fig.3: anatomy of the human lungs

The right lung is larger than the left lung, and it is divided by fissures into three parts, called the superior, middle, and inferior lobes. The left lung is similarly divided and consists of two parts, a superior and an inferior lobe. A lobar bronchus of the bronchial tree supplies each lobe. A lobe also has connections to blood and lymphatic vessels and is enclosed by connective tissues. Connective tissue further subdivides a lobe into lobules, each of which contains terminal bronchioles together with their alveolar ducts, alveolar sacs, alveoli, nerves, and associated blood and lymphatic vessels. [29]

2.2 Lung volumes:

There are five volumes and four capacities used to define the lung space. They are:

- Tidal Volume (TV) is the amount of volume that is inspired and exhaled during normal quiet breathing. 7–9 mL/kg of ideal body weight ~8–10 % of TLC
- Inspiratory Reserve Volume (IRV) is the maximum volume that can be inhaled above the tidal volume.
- Expiratory Reserve Volume (ERV) is the maximum volume that can be expired after the expiration of a tidal volume.
- Residual Volume (RV) is the volume left in the lungs after maximum expiration.
- Functional Residual Capacity (FRC) is the volume of air left in the lungs that can be exhaled after normal expiration.
- Inspiratory Capacity (IC) is the volume of maximum inhalation.
- Vital Capacity (VC) is the volume of maximum inhalation and exhalation.
- Total Lung Capacity (TLC) is the maximum volume in the lungs. [30]

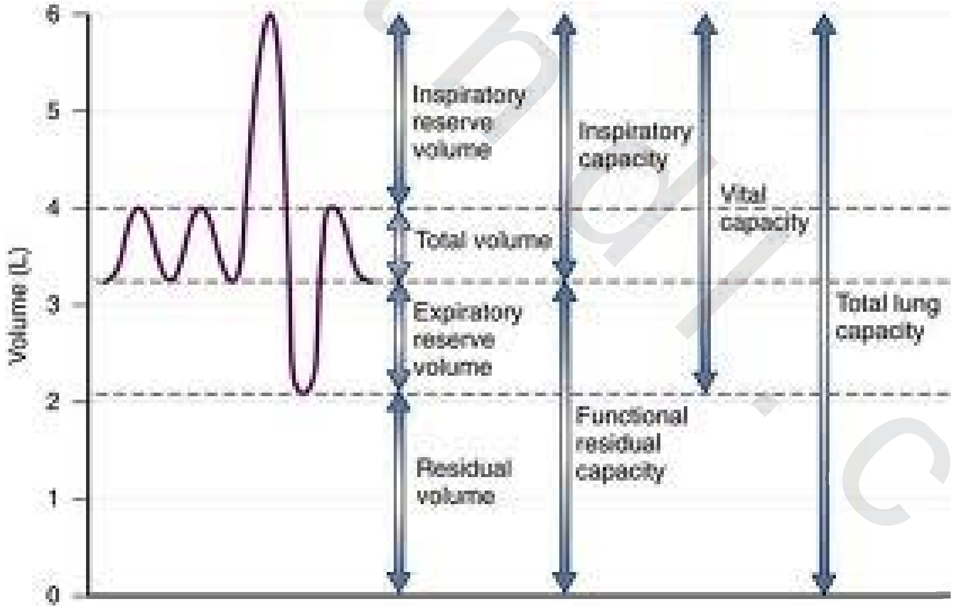


Fig.4: lung volumes

2.3 Sound transmission through the pulmonary system:

Mechanical compression waves (sound) travel in the lung much more slowly than in the air and soft tissue of which it is comprised; sound speed in the human lung [31] and animal lung have been studied by several groups. In human studies, sound was usually introduced into the mouth; but, the sound speed range from each group has been very different. In animal studies, sound was usually applied to the lung surface. Sound speed ranges from all groups were fairly consistent even though the measurements were taken on different animals. Also, all these animal studies concluded that the sound speed depended on the lung volume as the lung volume change leads to the change of air volume fraction in the lung, thus affecting the sound speed. (31)

2.4 Lung sound:

The sound of normal breathing heard over the surface of the chest is markedly influenced by the anatomical structures between the site of sound generation and the site of auscultation. Characteristically, normal lung sounds are heard clearly during inspiration but only in the early phase of expiration. In sound analysis, the frequency range of normal lung sounds appears to be narrower than that of tracheal sounds, extending from below 100 Hz to 1000 Hz; with a sharp drop at approximately 100 to 200 Hz. (32) The idea that "vesicular" sound is produced by air entering the alveoli ("vesicles") is incorrect. Indeed, modern concepts of physiology indicate that in the lung periphery gas molecules migrate by means of diffusion from parts of the lung reached through bulk flow, a silent process. Most important, studies support the idea of a double origin, with the inspiratory component generated within the lobar and segmental airways and the expiratory component coming from more central sources. (32)

Several mechanisms of vesicular sounds have been suggested, including turbulent flow, vortexes, and other, hitherto unknown mechanisms. (33-35) Clinically, a decrease in sound intensity is the most common abnormality. Mechanistically, this loss of intensity can be due to a decrease in the amount of sound energy at the site of generation, impaired transmission, or both. (36)

Sound generation can be decreased when there is a drop in inspiratory airflow, which can result from several conditions, ranging from poor cooperation (e.g., a patient's unwillingness to take a deep breath) to depression of the central nervous system (e.g., drug overdose). Airway conditions include blockage (e.g., by a foreign body or tumor) and the narrowing that occurs in obstructive airway diseases (e.g., asthma and chronic obstructive pulmonary disease [COPD]). The decrease in the intensity of breath sounds may be permanent, as in cases of pure emphysema, or reversible, as in asthma (e.g., during a bronchial provocation test or asthma attack). (36)

Sound transmission can be impaired by intrapulmonary or extrapulmonary factors. The latter include conditions such as obesity, chest deformities (e.g., kyphoscoliosis), and abdominal distention due to ascites. Intrapulmonary factors, which can be harder to recognize, include disruption of the mechanical properties of the lung parenchyma (e.g., a combination of hyperdistention and parenchymal destruction in emphysema) or the interposition of a medium between the source of sound generation and the stethoscope that has a different acoustic impedance from that of the normal parenchyma (e.g., collections of gas or liquid in the pleural space — pneumothorax, hemothorax, and intrapulmonary masses). Breath sounds have been analyzed and modeled electronically using computer algorithms to detect respiratory rate and classify breath sounds. (37- 39)

2.5 Abnormal lung sounds:

Lung sounds have naturally non-stationary signals. This property can be observed both in healthy normal and abnormal subjects. But this non-stationarity is more severe in cases of abnormal lung sounds. Therefore, significant diagnostic information can be obtained from the frequency distribution of lung sounds, with the selection of the signal processing technique used to extract this information being very important to maximize the efficiency of extraction [(40)]. This task has motivated many studies on the classification of lung sounds using frequency analysis. [(40-44)]

The spectrograms, which are also called sonograms [(45)] or respirosonograms, when applied to respiratory sounds [(46)], have been widely used for auscultation teaching, lung sound researching and evaluation of techniques for the processing of respiratory sounds [(41-44,46)]. However, their applications are usually restricted to the visualization of the spectral information of lung sounds.

2.5.1 Musical sounds:

i. Stridors:

Stridor is a high-pitched, musical sound produced as turbulent flow passes through a narrowed segment of the upper respiratory tract. [(47)] It is often intense, being clearly heard without the aid of a stethoscope. In sound analysis it is characterized by regular, sinusoidal oscillations with a fundamental frequency of approximately 500 Hz, often accompanied by several harmonics. Evaluating stridor is especially useful in patients in the intensive care unit who have undergone extubation, when its appearance can be a sign of extrathoracic airway obstruction requiring prompt intervention. In cases of such obstruction, stridor can be distinguished from wheeze because it is more clearly heard on inspiration than on expiration and is more prominent over the neck than over the chest. [(47)]

Although stridor is usually inspiratory, it can also be expiratory or biphasic. Other causes of stridor in adults include acute epiglottitis, airway edema after device removal, anaphylaxis, and vocal-cord dysfunction, inhalation of a foreign body, laryngeal tumors, thyroiditis, and tracheal carcinoma. The stridorous sound of vocal-cord dysfunction deserves special mention because it is often confused with asthma and is responsible for numerous visits to the emergency department and hospitalizations. (Vocal-cord dysfunction, also called paradoxical vocal-cord motion, is a respiratory condition characterized by the inappropriate adduction of the vocal cord with resultant airflow limitation at the level of the larynx, accompanied by stridorous breathing). [(48, 49)]

ii. Wheezes:

The wheeze is probably the most easily recognized adventitious sound. [(50)] Its long duration, typically more than 100 ms, allows its musical quality to be discerned by the human ear. In sound analysis the wheeze appears as sinusoidal oscillations with sound energy in the range of 100 to 1000 Hz and with harmonics that exceed 1000 Hz on occasion. [(50)] It is probably incorrect to credit high-pitched wheezes to the narrowing of peripheral airways and low-pitched wheezes to the narrowing of central airways. Purportedly, wheezes are formed in the branches between the second and seventh generations of the airway tree by the coupled oscillation of gas and airway walls that have been narrowed to the point of opposition by a variety of mechanical forces. [51]

In addition, the model incorporates two principles: first, that although wheezes are always associated with airflow limitation, airflow can be limited in the absence of wheezes, and second, that the pitch of an individual wheeze is determined not by the diameter of the airway but by the thickness of the airway wall, bending stiffness, and longitudinal tension. [51]

Wheezes can be inspiratory, expiratory, or biphasic. Although typically present in obstructive airway diseases, especially asthma, they are not pathognomonic of any particular disease. In asthma and COPD, wheezes can be heard all over the chest, making their number difficult to estimate. Localized wheeze is often related to a local phenomenon, usually an obstruction by a foreign body, mucous plug, or tumor. Failure to recognize this type of wheeze can have serious consequences for patients, who often receive a misdiagnosis of “difficult-to-treat asthma” and are not referred to appropriate specialists for months or even years after the initial evaluation. [52]

Wheezes may be absent in patients with severe airway obstruction. In fact, the model cited above predicts that the more severe the obstruction, the lower the likelihood of wheeze. The typical example is a severe asthma attack, a condition in which the low respiratory flows cannot provide the energy necessary to generate wheezes (or any sounds). As a consequence, the accompanying normal breath sound is also severely reduced or even absent, creating a clinical picture known as “silent lung.” As the obstruction is relieved and airflow increases, both the wheeze and normal breath sounds reappear. [52]

Finally, a word must be said about the rhonchus. This sound is considered to be a variant of the wheeze, differing from the wheeze in its lower pitch typically near 150 Hz which is responsible for its resemblance to the sound of snoring on auscultation. The rhonchus and the wheeze probably share the same mechanism of generation, but the rhonchus, unlike the wheeze, may disappear after coughing, which suggests that secretions play a role. Although many physicians still use the term rhonchus, some prefer to refer to the characteristic musical sounds simply as high-pitched or low-pitched wheezes. [50]

2.5.2 Nonmusical sound:

i. Crackles:

Crackles are short, explosive, nonmusical sounds heard on inspiration and sometimes during expiration. [50] Two categories of crackles have been described: fine crackles and coarse crackles. On auscultation, fine crackles are usually heard during mid-to-late inspiration, are well perceived in dependent lung regions, and are not transmitted to the mouth. Uninfluenced by cough, fine crackles are altered by gravity, changing or disappearing with changes in body position (e.g., bending forward). Coarse crackles tend to appear early during inspiration and throughout expiration and have a “popping” quality. They may be heard over any lung region, are usually transmitted to the mouth, can change or disappear with coughing, and are not influenced by changes in body position. In sound analysis, crackles appear as rapidly dampened wave deflections with a repetitive pattern.

As compared with coarse crackles, fine crackles have a shorter duration (5 ms vs. 15 ms) and higher frequency (650 Hz vs. 350 Hz). [53]

The most likely mechanism for the generation of fine crackles is the sudden inspiratory opening of small airways held closed by surface forces during the previous expiration. [54] Coarse crackles are probably produced by boluses of gas passing through airways as they open and close intermittently. With the exception of the crackling sounds heard in moribund patients or in patients with abundant secretions, crackles are probably not produced by secretions. [50]

Evaluation for crackles is important because it can help with the differential diagnosis. Because fine crackles have a distinctive sound that is similar to the sound heard when joined strips of Velcro are gently separated, they have been called Velcro rales. Typically, fine crackles are prominent in idiopathic pulmonary fibrosis, appearing first in the basal areas of the lungs and progressing to the upper zones with disease progression.[55] However, fine crackles are not pathognomonic of idiopathic pulmonary fibrosis; they are also found in other interstitial diseases, such as asbestosis, nonspecific interstitial pneumonitis, and interstitial fibrosis associated with connective-tissue disorders. Notably, fine crackles tend to be minimal or even absent in sarcoidosis, probably because sarcoidosis primarily affects the central lung zones not abutting the pleura. Among patients with similar levels of scarring on chest films, those with few crackles are more likely to have sarcoidosis, whereas those who have many crackles are more likely to have idiopathic pulmonary fibrosis. Advanced computerized acoustic analysis, which involves the use of a multichannel sound-detection device, has made it possible to diagnose idiopathic pulmonary fibrosis and congestive heart failure, in addition to other cardiopulmonary disorders, with good sensitivity and specificity. [56]

In idiopathic pulmonary fibrosis and asbestosis, fine crackles can be discerned before radiologic abnormalities are detected and are thus considered to be an early sign of pulmonary impairment. [57, 58]

Although the presence of Velcro rales as heard on auscultation has not been formally accepted as diagnostic of idiopathic pulmonary fibrosis, auscultation is considered to be the only realistic means of detection early in the course of the disease. [55] In asbestosis, the use of computerized detection of crackles has appeared to be as accurate as CT in locating disease that is not radiologically apparent. [58] In a study of 386 workers exposed to asbestos, crackle detection by means of auscultation performed by a trained technician correctly identified all the cases of asbestosis, suggesting that auscultation may have a role to play as a noninvasive method of screening in these populations. [59]

Coarse crackles are commonly heard in patients with obstructive lung diseases, including COPD, bronchiectasis, and asthma, usually in association with wheezes. They are also often heard in patients with pneumonia and those with congestive heart failure. In pneumonia, the characteristics of the crackles may vary markedly during the disease: the coarse, midinspiratory crackles heard in the early phase give way to shorter, end-inspiratory crackles in the recovery phase. [60] Fine and coarse crackles may also coexist. Finally, although crackles can be heard in healthy persons, the crackles tend to disappear after a few deep breaths. The presence of persistent crackles in both lungs in older persons with dyspnea should prompt an investigation for interstitial lung disease. [61]

2.5.3 Mixed sound:

i. The squake:

Also called "short wheeze" or "squeak," the squawk is a mixed sound, containing musical and nonmusical components. The mechanism underlying the production of squawks is not entirely known, but according to one theory, they are produced by the oscillation of peripheral airways (in deflated lung zones) whose walls remain in apposition long enough to oscillate under the action of the inspiratory airflow. Squawks are typically heard from the middle to the end of inspiration in patients with interstitial diseases, especially hypersensitivity pneumonitis. [62] However, they are not pathognomonic of this condition, having also been documented in diseases such as bronchiectasis and pneumonia.[63] In a patient with squawk and no evidence of interstitial disease, pneumonia should be suspected, because it is the next most likely cause. [63]

2.6 Anatomy of the heart:

The heart is located in the chest, directly above the diaphragm in the region of the thorax called mediastinum, specifically the middle mediastinum. The normal human heart varies with height and weight. [64] The tip (apex) of the heart is pointed forward, downward, and toward the left. The (inferior) diaphragmatic surface lies directly on the diaphragm. The heart lies in a double walled fibroserous sac called the pericardial sac, which is divided into (a) fibrous pericardium, and (b) serous pericardium. The fibrous pericardium envelops the heart and attaches onto the great vessels. [65] The serous pericardium is a closed sac consisting of two layers – a visceral layer or epicardium forming the outer lining of the great vessels and the heart, and a parietal layer forming an inner lining of the fibrous pericardium. [65-67] The two layers of the serous pericardium contain the pericardial fluid, which prevents friction between the heart and the pericardium. [65-67]

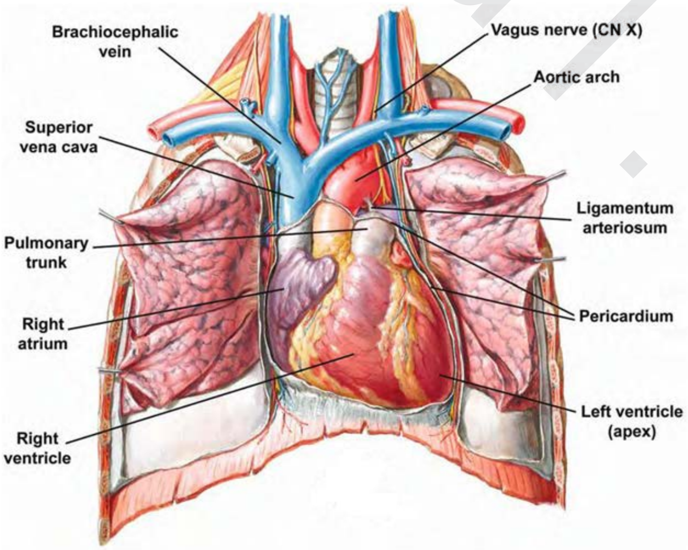


Fig.5: anatomy of the heart

The wall of the heart is composed of three layers: (a) epicardium; (b) myocardium; and (c) endocardium. [68, 69] The epicardium is the outer lining of the cardiac chambers and is formed by the visceral layer of the serous pericardium. The myocardium is the intermediate layer of the heart and is composed of three discernable layers of muscle that are seen predominantly in the left ventricle and inter-ventricular septum alone and includes a subepicardial layer, a middle concentric layer and a subendocardial layer. The rest of the heart is composed mainly of the subepicardial and subendocardial layers. The myocardium also contains important structures such as excitable nodal tissue and the conducting system. The endocardium the innermost layer of the heart is formed of the endothelium and subendothelial connective tissue. [68, 69]

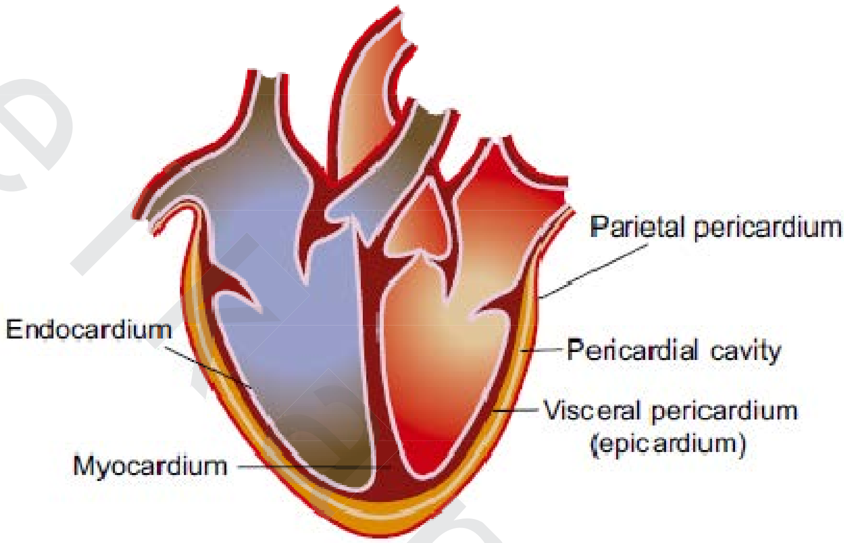


Fig.6: the heart wall layers

2.6.1 Chambers and valves:

The heart is divided into four distinct chambers with muscular walls of different thickness. [70-72] The left atrium (LA) and right atrium (RA) are small, thin-walled chambers located just above the left ventricle (LV) and right ventricle (RV), respectively. The ventricles are larger thick-walled chambers that perform most of the work. [70- 72] The atria receive blood from the venous system and lungs and then contract and eject the blood into the ventricles. The ventricles then pump the blood throughout the body or into the lungs. The heart contains four valves which are two atrioventricular (AV) valves, the mitral valve (bicuspid valve), and the tricuspid valve, which are between the upper atria and the lower ventricles.The two semilunar (SL) valves, the aortic valve and the pulmonary valve, which are in the arteries leaving the heart and the fibrous skeleton of the heart contains the annuli of the four valves, membranous septum, aortic intervalvular, right, and left fibrous trigones. [71, 73] The right trigone and the membranous septum together form the central fibrous body, which is penetrated by the bundle of His. [71, 73] The fibrous skeleton functions not only to provide an electrophysiological dissociation of atria and the ventricles but also provides structural support to the heart. [74] Each of the four valves has a distinctive role in maintaining physiological stability. [73]

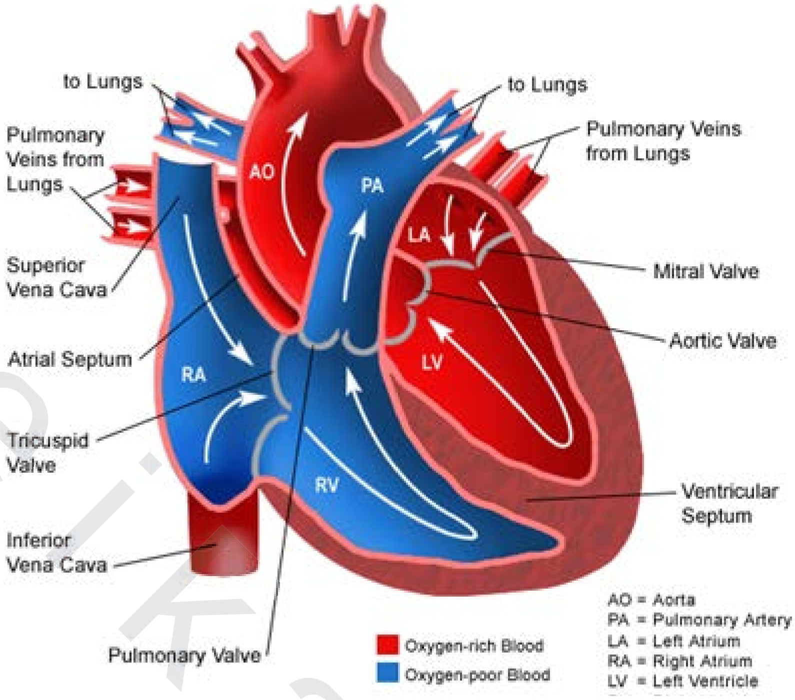


Fig.7: chambers and valves of the heart

2.6.2 Cardiac cell and cardiac muscle:

The cardiac cell contains bundles of protein strands called myofibrils. These myofibrils are surrounded by sarcoplasmic reticulum, which contains cysternae[69, 75- 77]. The sarcomeres are the contractile unit of myofibrils and the T tubules are continuations of the cell membrane located near the Z-lines, which conduct the action potential (AP) to the interior of the cell. [73, 75] The T tubules connect the sarcolemma to the sarcoplasmic reticulum in the skeletal muscle and the cardiac muscle. [75, 76]

Cardiac muscle is an involuntary striated muscle, which is mononucleated and has cross-striations formed by alternate segments of thick and thin protein filaments, which are anchored by segments called Z-lines. Cardiac muscle is relatively shorter than skeletal muscle and actin and myosin are the primary structural proteins. When the cardiac muscle is observed by a light microscope, the thinner actin filaments appear as lighter bands, while thicker myosin filaments appear as darker bands. The dark bands are actually the region of overlap between the actin and myosin filaments and the light bands are the region of actin filaments. The thinner actin flaments contain two others proteins called troponin and tropomyosin, which play an important role in contraction. [73, 75, 76]

Cardiac muscle also contains dense bands (specialized cell junctions) called intercalated discs that separate individual cells from one another at their ends and these discs consist of a transverse and a lateral portion. The transverse portion of the disc acts as a zone of firm adhesion and a route of transmission of contractile force and the lateral portion of the disc acts as a gap junction across which propagation of electrical impulses between the adjacent cardiac cells occurs. This in effect allows the individual cells of the heart to act as a syncytium. (74, 76)

2.7 Heart sounds:

The heart sounds heard as \lub, dub" occur because of the closing of the heart valves. \Lub" is known as the first heart sound (S1) and \dub" is known as the second heart sound (S2). The third heart sound (S3) occurs immediately after the S2, and it is of lower energy than the second one. The fourth heart sound (S4) occurs before the S1 and it has a lower scale of amplitude than the other sounds. In addition, the sounds due to the flow of blood in the vessels and in the heart are components of the heart sounds. However, in fact, how they occur is still a subject of discussion. (78)

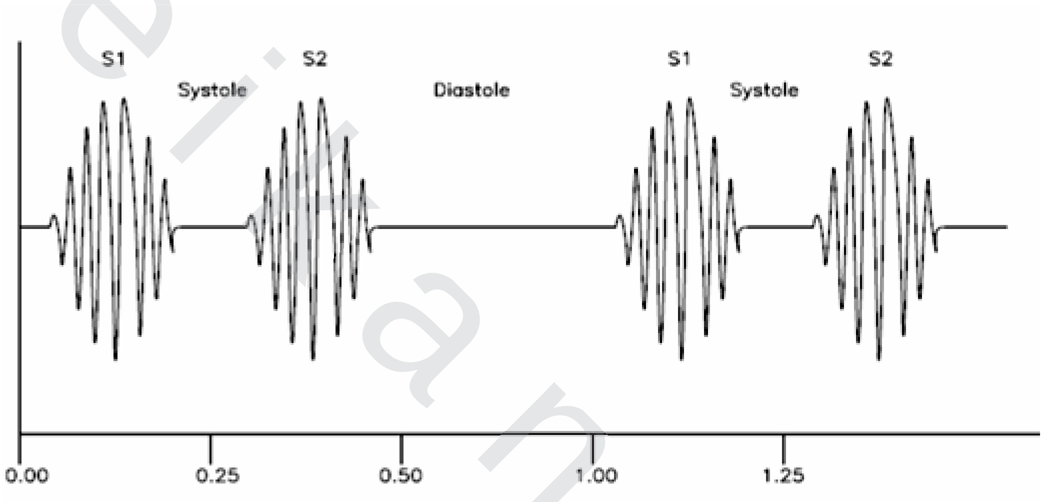


Fig.8: normal heart sound

Abnormal heart sounds such as murmurs are extra heart sounds heard among real heart sounds and they are caused by damaged valves. (78) The murmurs are the first signs of pathologic changes occurring in the heart valves, and they can be detected by the auscultation method in primary health organizations. (79) The common heart diseases result from heart valve disorders. Heart valve disorders are important heart diseases. Thus, in heart valve disorders, early diagnosis is one of the chief areas of study in the field of medicine. (80)

In distinguishing normal and abnormal heart sounds, the auscultation method is one of the primary ones used by the physicians. (81) These sounds are listened to and interpreted by the doctors using stethoscopes, and they help to detect whether the patient has any heart disorder. Doppler, echocardiography and magnetic resonance imaging techniques are effective methods used nowadays in determining the anomalies of heart valves. With the use of these methods, the importance of auscultation and phonocardiography, both conventional methods, has started to decrease according to expert doctors. The auscultation method is a major and substantial diagnosis method for physicians but it is relatively more expensive, inaccessible and time-consuming than the other methods. However, in primary health organizations, auscultation still has an important place in determining whether patients need expert interference. (79)

2.8 Obstructive Lung Diseases

Chronic obstructive pulmonary disease (COPD) denotes either a combination of emphysema, chronic bronchitis, and asthma or any one of these disease entities alone. Emphysema is classified as panlobular or centrilobular. In panlobular emphysema, the alveoli and alveolar ducts are destroyed. Patients with panlobular emphysema typically have a hyperinflated chest. In centrilobular emphysema, the respiratory bronchioles are destroyed. A common auscultatory finding in pulmonary emphysema (panlobular or centrilobular) is a reduction of lung sounds, which is predominantly due to airflow limitation. In chronic bronchitis, there is excessive bronchial mucus production most often caused by cigarette smoking; rhonchi are commonly auscultated. Also, early inspiratory crackles are characteristic of chronic bronchitis, whereas mid-inspiratory and expiratory crackles are characteristic of bronchiectasis. Chronic bronchitis causes wheezes, as well.[82]

Asthma and chronic obstructive pulmonary disease (COPD) are both very common and their incidence is increasing globally, placing an increasing burden on health services in industrialized and developing countries. [83- 85] Both diseases are characterized by airway obstruction, which is variable and reversible in asthma but is progressive and largely irreversible in COPD. In both diseases, there is chronic inflammation of the respiratory tract, which is mediated by the increased expression of multiple inflammatory proteins, including cytokines, chemokines, adhesion molecules, inflammatory enzymes and receptors. In both diseases there are acute episodes or exacerbations, when the intensity of this inflammation increases. The similarity between these airway diseases prompted the suggestion in the 1960s that asthma and COPD are different forms of a common disease (chronic obstructive lung disease), and this came to be known as the 'Dutch hypothesis'. This was countered by the 'British hypothesis', which maintained that these diseases were separate entities; the debate continues today, with evidence both for and against these two views. [86, 87]

Despite the similarity of some clinical features of asthma and COPD, there are marked differences in the pattern of inflammation that occurs in the respiratory tract, with different inflammatory cells recruited, different mediators produced, distinct consequences of inflammation and differing responses to therapy. In addition, the inflammation seen in asthma is mainly located in the larger conducting airways, and although small airways can also be affected in more severe forms of the disease, the lung parenchyma is not affected. By contrast, COPD predominantly affects the small airways and the lung parenchyma, although similar inflammatory changes can also be found in larger airways. [88, 89]

These differences in disease distribution may partly reflect the distribution of inhaled inciting agents, such as allergens in asthma and tobacco smoke in COPD. In both diseases, there are different clinical phenotypes recognized. Most patients with asthma are atopic (extrinsic asthma), but a few patients are non-atopic (intrinsic asthma), and these patients often have a more severe form of the disease. [90] There is a range of asthma severity, which tends to be maintained throughout life. [91]

Exacerbation of congestive heart failure and asthma are two common causes of acute dyspnea. On pulmonary auscultation, physicians most often record fine inspiratory crackles for patients with congestive heart failure and wheezes for patients with asthma. [92] (However, medium and coarse crackles are also encountered in cases of congestive heart failure.) The crackles of congestive heart failure occur at all times during inspiration (paninspiratory) but usually occur late in inspiration. [93] However, crackles associated with congestive heart failure and wheezes associated with asthma are not specific findings for the diseases. [94] For example, in a study by Epler et al, 60% of patients with interstitial lung disease had fine crackles on lung auscultation, and 10% to 12% of patients with COPD had fine crackles on lung auscultation. [95]

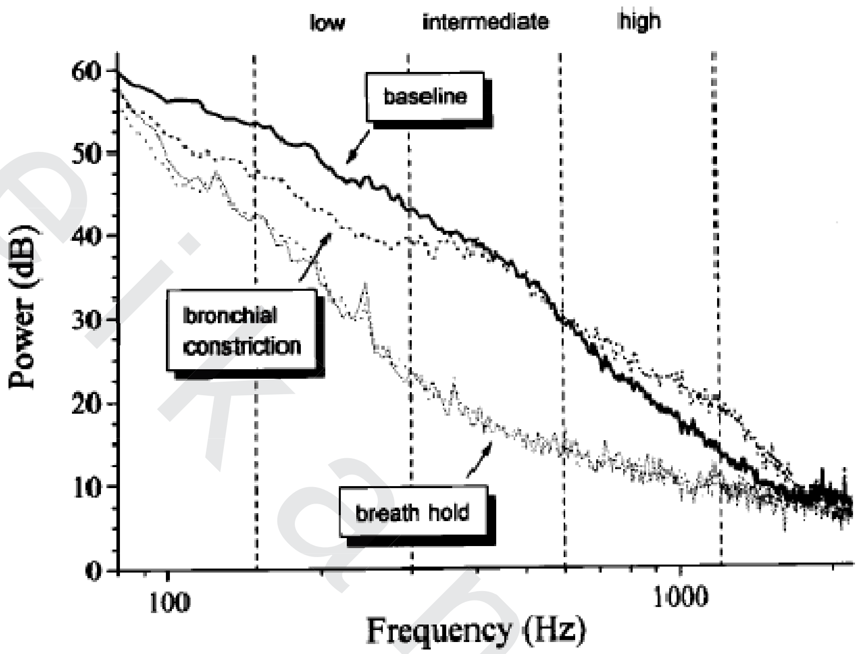


Fig.9: Power spectra of normal lung sounds in a boy with asthma

2.9 The stethoscope: [96]

Stethoscopes are used to transmit sounds from the surface of the body to the human ear. There is much variability in interpretation of sounds received from stethoscopes due to auditory acuity and training. Furthermore, depending on the placement of the stethoscope to the desired surface of the body, some sounds may be attenuated or perceived differently. The frequency response of the mechanical stethoscope, illustrated in (fig. 10), was determined by applying a known audio frequency signal to the end of the stethoscope via a coupling mechanism. It can be seen from (fig. 10) that the frequency response of the mechanical stethoscope is uneven and appears to have a low pass filtering effect. Furthermore, critical information exists from the stethoscope near the lower threshold of human hearing. This information can often be missed if a physician does not have perfect hearing in the lower frequency ranges. Therefore, physicians using a mechanical stethoscope may not detect important low frequency sounds. Also, the mechanical stethoscope is only capable of detecting low frequency sounds under around 1000 Hz.

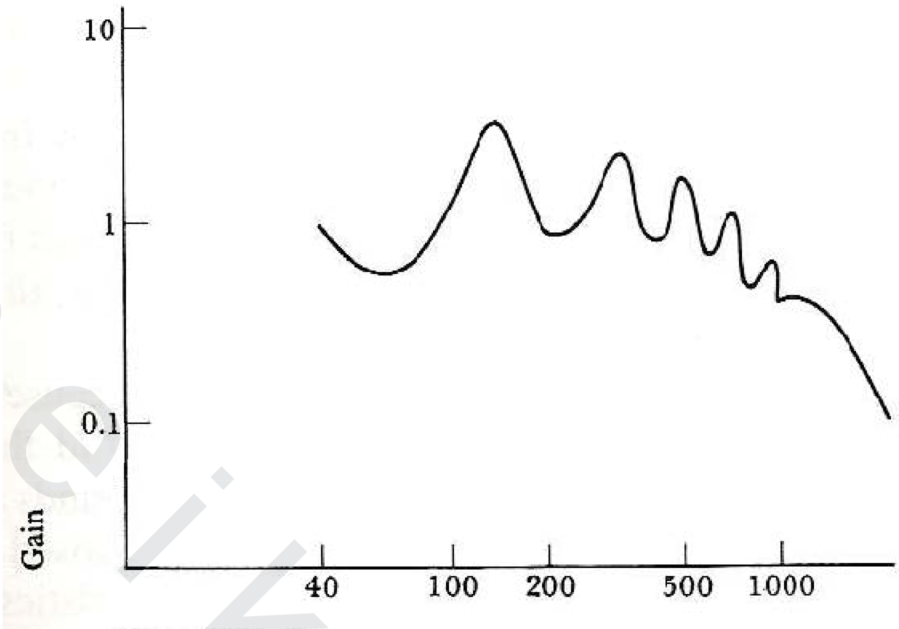


Fig.10: The frequency response of the mechanical stethoscope

Also, if the stethoscope malfunctions and attenuates the captured sound signals more than 3 dB, the sound signal will be completely lost to the physician using the stethoscope due to a low signal to noise ratio (SNR). The attenuation of sound varies between stethoscopes and therefore a physician may hear sounds from one stethoscope and not another.

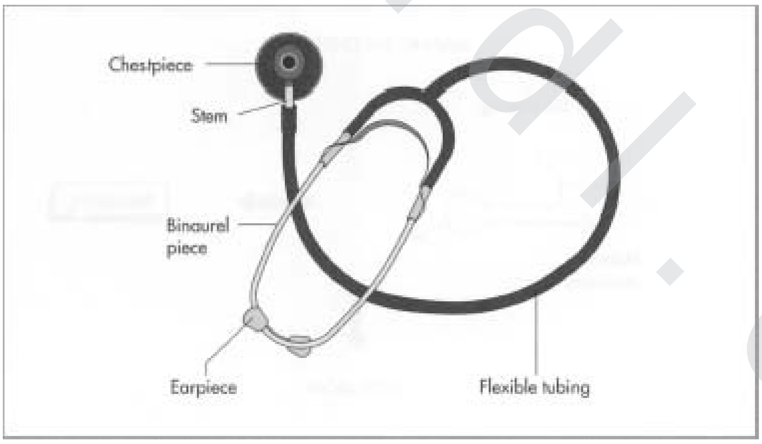


Fig.11: acoustic stethoscope

The acoustic stethoscope is the traditional and most commonly used technology for auscultation and is based on the principle of a changing air pressure that produces sound waves. An acoustic stethoscope can be seen in (fig. 11) and consists of three main components: (1) Chest Piece: consists of a diaphragm and bell. On one side is the diaphragm which consists of a plastic disc that is used to listen to high frequency sounds such as sounds produced by the lungs and the heart during normal function. On the other side of the chest piece is the bell which is used to listen to low frequency sounds such as potential heart problems such as heart murmurs and bowel sounds. (2) Tube: consists of a hollow air filled tube which facilitates the transmittal of sound waves from the chest piece to the ear piece for listening. (3) Ear Piece: inserted into the ear for listening to the sounds produced by the body.

Placement of the stethoscopes chest piece may also affect captured sounds. Placing the chest piece too firmly will severely low-pass filter the received signals. This low-pass filtering effect occurs because the skin forms a diaphragm at the chest piece rim. The diaphragm pressurizes and thus attenuates low frequencies. For reasons outlined above, engineers have developed electronic stethoscopes. Modern electronics allows for adjustable frequency responses and therefore more valuable acoustical information may be extracted from the thorax.

2.10 Basic principles of digital stethoscope:

Electronic stethoscopes are designed to overcome the disadvantages of the acoustic stethoscopes. They are designed to have a uniform frequency response and to amplify the sound level. In general, an 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]

2.11 Sensor:

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 (fig. 12). Both transducers are popular in sound recording. However, accelerometers are typically more expensive than microphones, are often fragile, and may exhibit internal resonances. Thus, this concludes that the microphone is perfect for the application. [98]

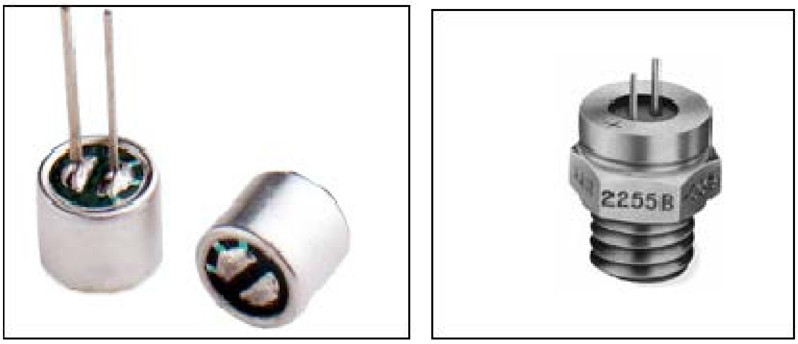


Fig.12: condenser microphones and piezoelectric accelerometers

2.12 Operational amplifier: [99]

An operational amplifier, or op-amp, is a very high gain differential amplifier with high input impedance and low output impedance. Typical uses of the operational amplifier are to provide voltage amplitude changes (amplitude and polarity), oscillators, filter circuits, and many types of instrumentation circuits. An op-amp contains a number of differential amplifier stages to achieve a very high voltage gain.

Fig. 13 shows a basic op-amp with two inputs and one output as would result using a differential amplifier input stage. Each input results in either the same or an opposite polarity (or phase) output, depending on whether the signal is applied to the plus (+) or the minus (-) input.

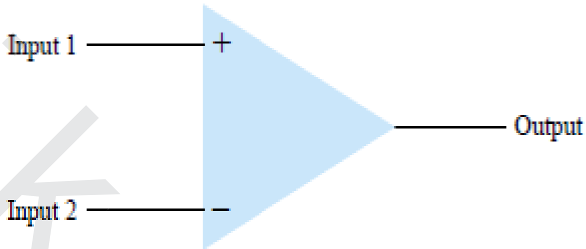


Fig. 13: basic operational amplifier

2.12.1 Double-Ended (Differential) Input:

In addition to using only one input, it is possible to apply signals at each input—this being a double-ended operation. Fig.14.shows an input, V_d, applied between the two input terminals (recall that neither input is at ground), with the resulting amplified output in phase with that applied between the plus and minus inputs. Figure 14.3b shows the same action resulting when two separate signals are applied to the inputs, the difference signal being Vi_1 - Vi_2.

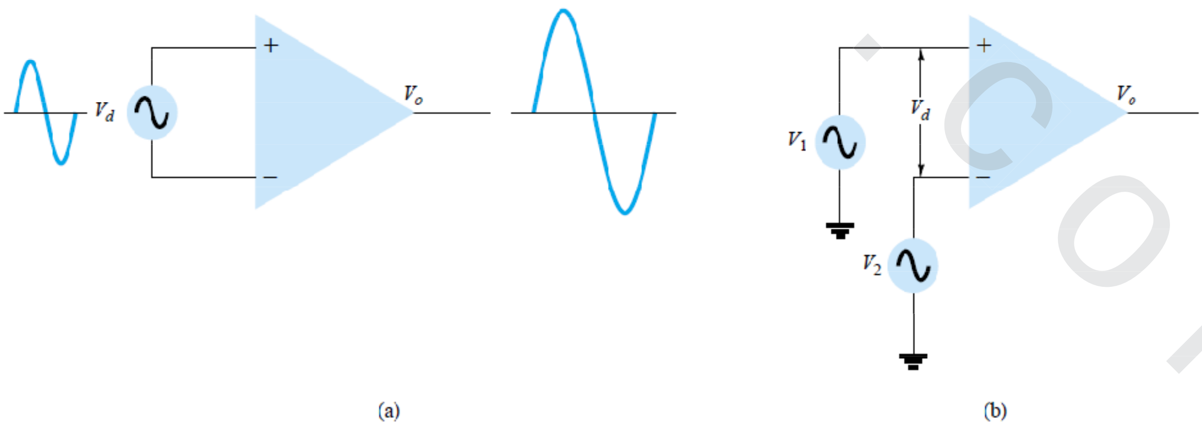


Fig.14: (a) input V_d, applied between the two input terminals (b) two separate signals are applied to the inputs

2.12.2 Double-Ended Output:

While the operation discussed so far had a single output, the op-amp can also be operated with opposite outputs, as shown in Fig. 14.4. An input applied to either input will result in outputs from both output terminals, these outputs always being opposite in polarity.

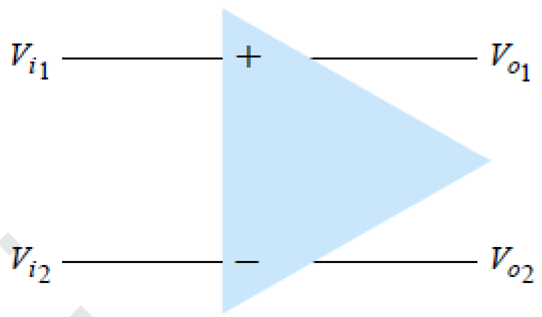
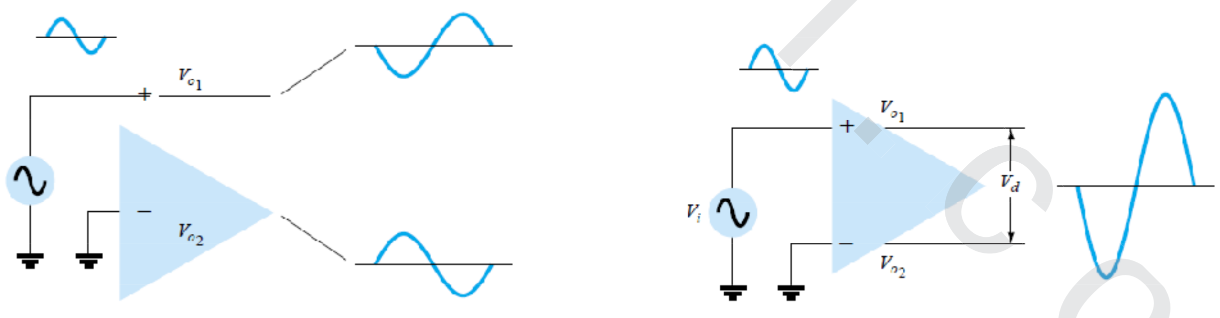


Fig. 15: double ended output

Fig. 15 shows a single-ended input with a double-ended output. As shown, the signal applied to the plus input results in two amplified outputs of opposite polarity. (Fig. 16(a)) shows the same operation with a single output measured between output terminals (not with respect to ground). This difference output signal is V_o1 - Vo2. The difference output is also referred to as a floating signal since neither output terminal is the ground (reference) terminal. Notice that the difference output is twice as large as either Vo1 or Vo2 since they are of opposite polarity and subtracting them results in twice their amplitude [i.e., 10 V - (-10 V) - 20 V]. (Fig. 16(b)) shows a differential input, differential output operation. The input is applied between the two input terminals and the output taken from between the two output terminals. This is fully differential operation.



(a) (b)

Fig. 16: (a) Double-ended output with single-ended input. (b) Double-ended output.

2.12.3 Basic practical operational amplifier circuits:

i. Inverting amplifier:

The most widely used constant-gain amplifier circuit is the inverting amplifier, as shown in (Fig. 17). The output is obtained by multiplying the input by a fixed or constant gain, set by the input resistor (R_1) and feedback resistor (R_f)—this output also being inverted from the input.

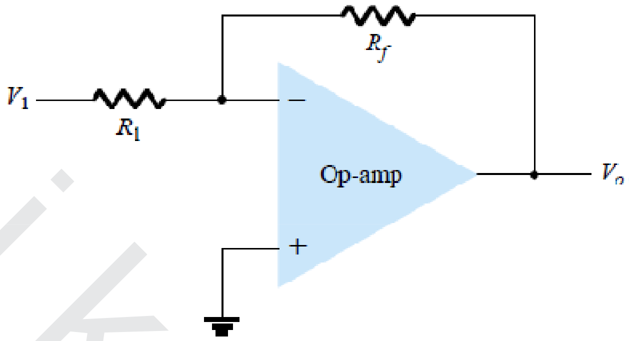


Fig. 17: inverting amplifier

ii. Non-Inverting amplifier:

The connection of (Fig. 18) shows an op-amp circuit that works as a non-inverting amplifier or constant-gain multiplier. It should be noted that the inverting amplifier connection is more widely used because it has better frequency stability.

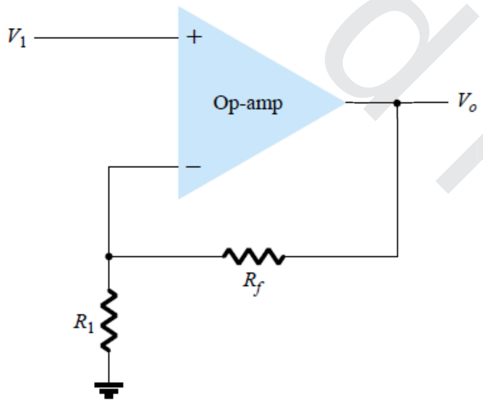


Fig. 18: non-inverting amplifier

iii. Unity follower:

The unity-follower circuit, as shown in Fig. 19, provides a gain of unity (1) with no polarity or phase reversal. From the equivalent circuit it is clear that:

$$V_{out} = V_s$$

and that the output is the same polarity and magnitude as the input. The circuit operates like an emitter- or source-follower circuit except that the gain is exactly unity.

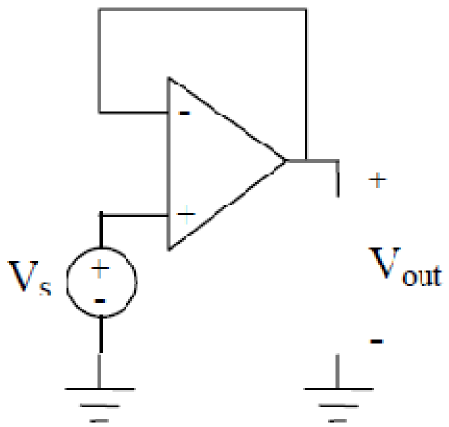


Fig. 19: unity follower

iv. Summing amplifier:

Probably the most used of the op-amp circuits is the summing amplifier circuit shown in fig. 20. The circuit shows a three-input summing amplifier circuit, which provides a means of algebraically summing (adding) three voltages, each multiplied by a constant-gain factor.

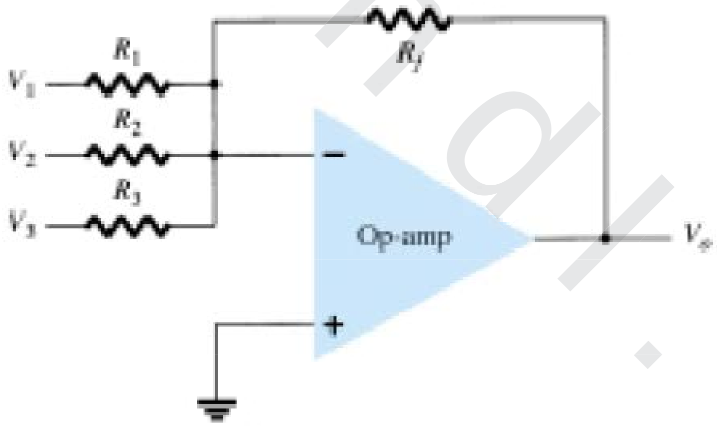


Fig. 20: summing amplifier

2.13 Physics of sound [100]

2.13.1 Types of waves:

A disturbance that propagates from one place to another is referred to as a wave. Waves propagate with well-defined speeds determined by the properties of the material through which they travel. In addition waves carry energy. It is important to distinguish between the motions of the wave itself and the motion of the individual particles that make up the wave.

i. Transverse waves:

The easiest type of wave to visualize is a wave on a spring (fig. 21). The displacement of particles in a string is at right angles to the direction of propagation of the wave. A wave with this property is called a transverse wave. In a transverse wave, the displacement of individual particles is at right angles to the direction of propagation of the wave.

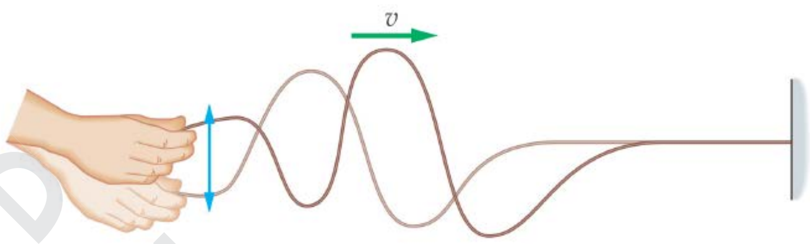


Fig. 21: a wave on a string

ii. Longitudinal waves:

Longitudinal waves differ from transverse waves in the way that particles in the wave move. In particular, a longitudinal wave is defined as the displacement of individual particles is parallel to the direction of propagation of the wave.

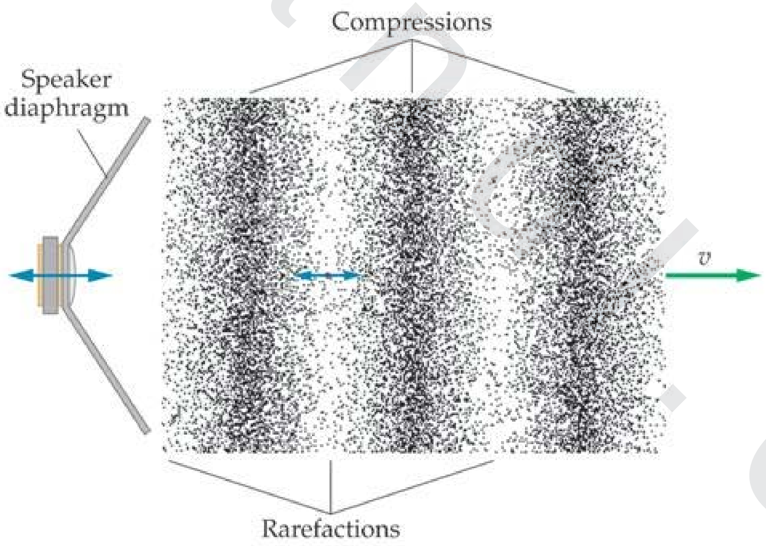


Fig. 22: sound produced by a speaker

The classic example of a longitudinal wave is sound. When you speak, for example, the vibration in your vocal cords creates a series of compressions and expansions in the air. The same kind of situation occurs with a loud speaker (fig. 22), the speaker diaphragm vibrating horizontally with simple harmonic motion. As it moves to the right it compresses the air momentarily; as it moves to the left it rarefies the air. A series of compressions and rarefactions then travel horizontally away from the loudspeaker with the speed of sound.

2.13.2 The frequency of a sound wave:

When we hear a sound, its frequency makes a great impression on us; the frequency determines the pitch of a sound. For example, the keys on a piano produce sound with frequencies ranging from 55 Hz for the key farthest to the left to 4187 Hz for the rightmost key. The frequency range of human hearing extends well beyond the range of a piano. Humans can hear sounds between 20 Hz on the low frequency end and 20000 Hz on the right frequency end. Sound with frequencies above this range are referred to as ultrasonic, while those with frequencies lower than 20 Hz are classified as infrasonic. Though we are unable to hear ultrasound and infrasound, these frequencies occur commonly in nature, and are used in many technological applications as well.

2.13.3 Sound intensity:

The loudness of a sound is determined by its intensity; that is by the amount of energy that passes through a given area in a given time. If the energy E passes through the area A in the time t, the intensity I, of the wave carrying the energy is

$$I = E/At \quad (3)$$

Recalling that power is energy per time, $P = E/t$, we can express the intensity as follows:

$$I = P/A \quad (4)$$

The units are those of power (watt, W) divided by area (meters squared, m^2).

2.13.4 Intensity level and Decibels:

In the study of sound, loudness is measured by the intensity level of a wave. Designed by the symbol (β), the intensity level is designed as follows:

$$\beta = (10\ \text{dB}) \log (I / I_o) \quad (5)$$

In this expression, log indicates the logarithm to the base 10 and I_o is the intensity of the faintest sounds that can be heard.

2.14 Audio Preamplifiers: (101)

Audio signal preamplifiers (preamps) represent the low-level end of the dynamic range of practical audio circuits using modern IC devices. In general, amplifying stages with input signal levels of 10 mV or fewer falls into the preamp category. Basic types of audio preamps, which are: (1) Microphone including preamps for dynamic, electrets, and phantom-powered microphones, using transformer input circuits, operating from dual and single supplies. (2) Phonograph including preamps for moving magnet and moving coil phono cartridges in various topologies, with detailed response analysis and discussion.

In general, when working signals drop to a level of ≈1 mV, the input noise generated by the first system amplifying stage becomes critical for wide dynamic range and good signal-to-noise ratio. For example, if internally generated noise of an input stage is 1 μV and the input signal voltage 1 mV, the best signal-to noise ratio possible is just 60 dB.

In a given application, both the input voltage level and impedance of a source are usually fixed. Thus, for best signal-to-noise ratio, the input noise generated by the first amplifying stage must be minimized when operated from the intended source. This factor has definite implications to the preamp designer, as a “low noise” circuit for low impedances is quite different from one with low noise operating from high impedance. Successfully minimizing the input noise of an amplifier requires a full understanding of all the various factors that contribute to total noise. This includes the amplifier itself as well as the external circuit in which it is used; in fact, the total circuit environment must be considered both to minimize noise and maximize dynamic range and signal fidelity.

A further design complication is the fact that not only is a basic gain or signal scaling function to be accomplished, but signal frequency response may also need to be altered in a predictable manner. Microphone preamps are an example of wideband, flat frequency response, and low noise amplifiers. In contrast to this, phonograph preamp circuits not only scale the signal, they also impart a specific frequency response characteristic to it. A major part of the design for the Recording Industry Association of America RIAA phono preamps of this section is a systematic analysis process, which can be used to predictably select components for optimum performance in frequency response terms. This leads to very precise functioning, and excellent correlation between a computer-based design and measured lab operation.

2.15 Fourier transforms: [102]

The Fourier transform is a mathematical procedure that was discovered by a French mathematician named Jean-Baptiste-Joseph Fourier in the early 1800's. It has been used very successfully through the years to solve many types of engineering, physics, and mathematics problems. The Fourier transform is defined for continuous (or analog) functions, and is usually applied in situations where the functions are assumed to be continuous. More recently however, it has been implemented in digital form in various types of analyzers. These analyzers compute digital (or sampled) forms of power spectrums, frequency response functions, and other types of frequency domain functions from measured (sampled) time domain signals.

The implementation of the Discrete Fourier Transform, or DFT, became practical in 1965 when Cooley and Tukey described an algorithm for computing the DFT very efficiently. Their algorithm (and others like it) has become known as the Fast Fourier Transform (FFT). Using the FFT algorithm, present day mini-computer based analyzers can compute a DFT in milliseconds where it used to take hours using standard computational procedures. Direct computation of the DFT on an N-point complex valued function requires N^2 operations; where an operation is defined as one multiplication plus an addition. The Cooley-Tukey algorithm takes approximately $N \log2 N$ operations; where N is a power of 2.

Table 1 indicates how much longer it takes to compute a DFT by direct computation compared to the Cooley-Tukey algorithm, for typical data record sizes.

Table 1: Direct vs. FFT computation of DFT.

N	**$N^2/N \log_2 N$**
256	32
512	57
1024	102
2048	186
4096	341
8192	630

Many other methods for efficiently computing the DFT have since been discovered. However, all methods which require on the order of N log N operations have become known as FFT's.

The properties of the Fourier transform, and its cousin the Laplace transform are quite extensively documented, and their use as mathematical tools is taught in most undergraduate engineering curriculums today. However the use of the DFT, and the problems encountered with its application to measured time domain signals are not generally understood.

In this section all the fundamental concepts associated with the use of the DFT are presented. We begin by examining the Fourier transform and some of its properties, and then show how a fundamental concept called "windowing" can be applied to the Fourier transform to derive the DFT and all of its properties. Using the convolution property, or as we will call it here, the windowing rule of Fourier transforms, we will define the concepts of sampling, aliasing, leakage and the wrap-around error. These are all important concepts which must be understood in order to avoid significant errors in the application of the DFT to measured data.

The Fourier Transform - The forward Fourier transform is defined as the integral

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-j2\pi ft}dt \qquad (1)$$

The inverse Fourier transform is defined as the integral

$$x(t) = \int_{-\infty}^{\infty} X(f)e^{j2\pi ft}df \qquad (2)$$

X (f) is the (complex) Fourier transform of x(t), where f and t are real variables. We will assume that t is the time variable (in seconds) and f is the frequency variable (in Hertz), although this transform can be used in many other applications where these variables have different meanings. Normally x(t) is a real valued function of time but this restriction is not at all necessary. X(f) represents its corresponding frequency domain function.

The two functions x(t) and X(f) are known as a Fourier transform pair. There is a unique Fourier transform X(f) corresponding to each function x(t). Thus knowing X(f) is equivalent to knowing x(t) and vice-versa. X(f) and x(t) are really two different representations of the same phenomenon. If the phenomenon is known in terms of x(t), then equation (1) shows how X(f) is represented in terms of x(t). Likewise if X(f) is known, equation (2) shows how x(t) is represented in terms of X(f).

Table 2 lists some commonly used Fourier transform pairs.

Table 2: Fourier transform pairs

Time Domain Function	Frequency Domain Function
Auto Correlation	Auto Power Spectrum
Cross Correlation	Cross Power Spectrum
Impulse Response	Frequency Response

2.16 Microcontroller: [103]

The microcontroller is simply a computer on a chip. It is one of the most important developments in electronics since the invention of the microprocessor itself. It is essential for the operation of devices such as mobile phones, DVD players, video cameras, and most self-contained electronic systems. The small LCD screen is a good clue to the presence of an MCU (Microcontroller Unit) it needs a programmed device to control it. Working sometimes with other chips, but often on its own, the MCU provides the key element in the vast range of small, programmed devices which are now commonplace.

216.1 Microcontroller Features:

Microcontrollers from different manufacturers have different architectures and different capabilities.

i. Supply Voltage:

Most microcontrollers operate with the standard logic voltage of 5V. Some microcontrollers can operate at as low as 2.7V, and some will tolerate 6V without any problem. The manufacturer's data sheet will have information about the allowed limits of the power supply voltage.

ii. The Clock:

All microcontrollers require a clock (or an oscillator) to operate, usually provided by external timing devices connected to the microcontroller. In most cases, these external timing devices are a crystal plus two small capacitors. In some cases they are resonators or an external resistor-capacitor pair. Some microcontrollers have built-in timing circuits and do not require external timing components. If an application is not time sensitive, external or internal (if available) resistor-capacitor timing components are the best option for their simplicity and low cost. An instruction is executed by fetching it from the memory and then decoding it. This usually takes several clock cycles and is known as the instruction cycle. In PIC microcontrollers, an instruction cycle takes four clock periods. Thus the microcontroller operates at a clock rate that is one-quarter of the actual oscillator frequency. The PIC18F series of microcontrollers can operate with clock frequencies up to 40MHz.

iii. Timers:

Timers are important parts of any microcontroller. A timer is basically a counter which is driven from either an external clock pulse or the microcontroller's internal oscillator. A timer can be 8 bits or 16 bits wide. Data can be loaded into a timer under program control, and the timer can be stopped or started by program control. Most timers can be configured to generate an interrupt when they reach a certain count (usually when they overflow). The user program can use an interrupt to carry out accurate timing-related operations inside the microcontroller.

iv. Watchdog:

Most microcontrollers have at least one watchdog facility. The watchdog is basically a timer that is refreshed by the user program. Whenever the program fails to refresh the watchdog, a reset occurs. The watchdog timer is used to detect a system problem, such as the program being in an endless loop. This safety feature prevents runaway software and stops the microcontroller from executing meaningless and unwanted code. Watchdog facilities are commonly used in real-time systems where the successful termination of one or more activities must be checked regularly.

v. Reset Input:

A reset input is used to reset a microcontroller externally. Resetting puts the microcontroller into a known state such that the program execution starts from address 0 of the program memory. An external reset action is usually achieved by connecting a push-button switch to the reset input. When the switch is pressed, the microcontroller is reset.

vi. Interrupts:

Interrupts are an important concept in microcontrollers. An interrupt causes the microcontroller to respond to external and internal (e.g., a timer) events very quickly. When an interrupt occurs, the microcontroller leaves its normal flow of program execution and jumps to a special part of the program known as the interrupt service routine (ISR). The program code inside the ISR is executed, and upon return from the ISR the program resumes its normal flow of execution. The ISR starts from a fixed address of the program memory sometimes known as the interrupt vector address. Some microcontrollers with multi-interrupt features have just one interrupt vector address, while others have unique interrupt vector addresses, one for each interrupt source. Interrupts can be nested such that a new interrupt can suspend the execution of another interrupt. Another important feature of multi-interrupt capability is that different interrupt sources can be assigned different levels of priority.

vii. Brown-out Detector:

Brown-out detectors, which are common in many microcontrollers, reset the microcontroller if the supply voltage falls below a nominal value. These safety features can be employed to prevent unpredictable operation at low voltages, especially to protect the contents of EEPROM-type memories.

viii. Analog-to-Digital Converter:

An analog-to-digital converter (A/D) is used to convert an analog signal, such as voltage, to digital form so a microcontroller can read and process it. Some microcontrollers have built-in A/D converters. External A/D converter can also be connected to any type of microcontroller. A/D converters are usually 8 to 10 bits, having 256 to 1024 quantization levels. Most PIC microcontrollers with A/D features have multiplexed A/D converters which provide more than one analog input channel. The A/D conversion process must be started by the user program and may take several hundred microseconds to complete. A/D converters usually generate interrupts when a conversion is complete so the user program can read the converted data quickly.

A/D converters are especially useful in control and monitoring applications, since most sensors (e.g., temperature sensors, pressure sensors, force sensors, etc.) produce analog output voltages.

ix. Serial Input-Output:

Serial communication (also called RS232 communication) enables a microcontroller to be connected to another microcontroller or to a PC using a serial cable. Some microcontrollers have built-in hardware called USART (universal synchronous asynchronous receiver-transmitter) to implement a serial communication interface. The user program can usually select the baud rate and data format. If no serial input-output hardware is provided, it is easy to develop software to implement serial data communication using any I/O pin of a microcontroller. The PIC18F series of microcontrollers has built-in USART modules.

x. EEPROM Data Memory:

EEPROM-type data memory is also very common in many microcontrollers. The advantage of an EEPROM memory is that the programmer can store nonvolatile data there and change this data whenever required. For example, in a temperature monitoring application, the maximum and minimum temperature readings can be stored in an EEPROM memory. If the power supply is removed for any reason, the values of the latest readings are available in the EEPROM memory. It can be erased by exposing it to an electrical charge.

xi. LCD Drivers:

LCD drivers enable a microcontroller to be connected to an external LCD display directly. These drivers are not common since most of the functions they provide can be implemented in software.

xii. Analog Comparator:

Analog comparators are used where two analog voltages need to be compared. Although these circuits are implemented in most high-end PIC microcontrollers, they are not common in other microcontrollers. The PIC18F series of microcontrollers has built-in analog comparator modules.

xiii USB Interface:

USB is currently a very popular computer interface specification used to connect various peripheral devices to computers and microcontrollers. Some PIC microcontrollers provide built-in USB modules.

2.16.2 Microcontroller Architectures:

Two types of architectures are conventional in microcontrollers. Von Neumann architecture, used by a large percentage of microcontrollers, places all memory space on the same bus; instruction and data also use the same bus. In Harvard architecture (used by PIC microcontrollers), code and data are on separate buses, which allows them to be fetched simultaneously, resulting in an improved performance.

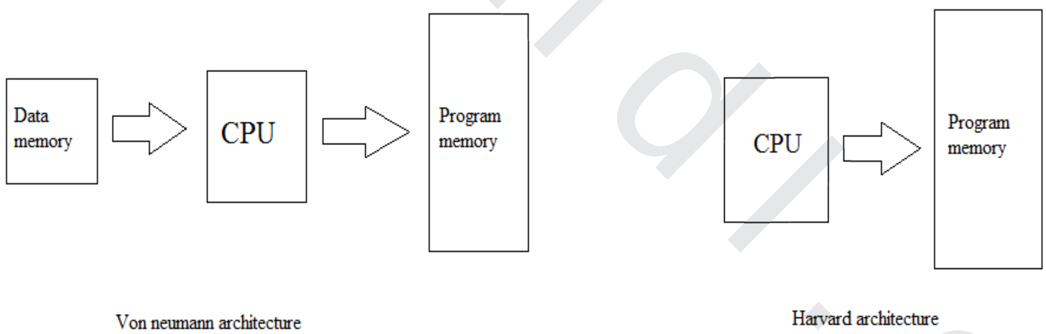


Fig. 23: Microcontroller architecture