Manonmaniam Sundaranar University,

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Tirunelveli - 627 012 Tamilnadu, India

OPEN AND DISTANCE LEARNING (ODL) PROGRAMMES

(FOR THOSE WHO JOINED THE PROGRAMMES FROM THE ACADEMIC YEAR 2023–2024)

B.Sc. Physics

Course Material

PHYSICS OF MUSIC

JSPH22

Prepared

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Tirunelveli - 12

PHYSICS OF MUSIC

JSPH22

SCIENTIFIC STUDY OF MUSIC:

Vibrations of atoms of matter - vibrations coupling to air — propagation of sound waves in air, other media, fluids and solids — velocity, frequency, wavelength, time period, intensity: definition and units — classification of sound on frequency and velocity— human and animal sound perception— mechanism of ear and hearing — psychoacoustics.

SIMPLE VIBRATING SYSTEMS:

Simple harmonic motion -tuning fork -amplitude, phase, energy, energy loss/damping/ dissipation — power — travelling waves and standing waves— laws of vibration in stretched strings— one dimensional medium — open and closed organ pipes — over tones, harmonics — quality of sound: pitch, timber, loudness — octaves, musical notes.

MUSICAL TONE:

Pure / simple tones — sine / cosine waves— well- defined frequencies, wavelengths, amplitudes and phases — partial tones — assembly of pure tones— mix of different frequencies and amplitudes— complex tone - superposition of simple tones — complex waveform — periodic complex wave form—formants—resonances— sound envelope.

PRODUCTION OF MUSICAL SOUNDS:

Human voice, mechanism of vocal sound production — larynx (sound box) — stringed Instruments: plucked and bowed, guitar, mandolin, violin, piano, etc.— wind instruments: whistles, flute, saxophone, pipe organ, bagpipes, etc.— percussion instruments: plates, membranes, drums, cymbals, xylophone etc. — electronic instruments: keyboards, electric guitars, rhythm pads, etc.—analog and digital sound synthesizers, —MIDI instrument— computer generated music.

RECORDING OF MUSIC AND SOUND:

Edison phonograph - cylinder and disk records — magnetic wire and tape recorders — digital recording (e.g. to CD, DVD, etc.)— analog transducers, condenser, dynamic microphones, loudspeaker complex sound fields — near and far fields of acoustic —spectral analysis techniques—continuous and discrete Fourier transforms, digital signal processing — digital filtering — specifications of recording studios.

Recommended text

i Physics and Music: The Science of Musical Sound by Harvey White (2014)

ii Good Vibrations: The Physics of Music by Barry Parker, (2009)

iii The History of Musical Instruments: B Curt Sachs, (2006)

iv Physics and Music: Essential Connections and Illuminating Excursions by Kinko Tsuji and Stefan C. Müller (2021)

UNIT - I

SCIENTIFIC STUDY OF MUSIC

vibrations of atoms of matter

The vibration of an object moves the molecules of air around it. The molecules nearby vibrate as a result of these molecules colliding with one another. They collide with more adjacent air molecules as a result. Sound waves are this "chain reaction" activity that continues until the molecules run out of energy.

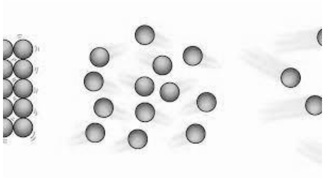


Fig 1.1

vibrations coupling to air

vibration of an item brought on by matter and atoms vibrating. The vibrations generated by the vibrating object are coupled with the surrounding air molecules. which causes the molecules of air to move in a certain way. When an object vibrates in conjunction with air, the air molecules create a wave that can propagate far beyond the source of sound.

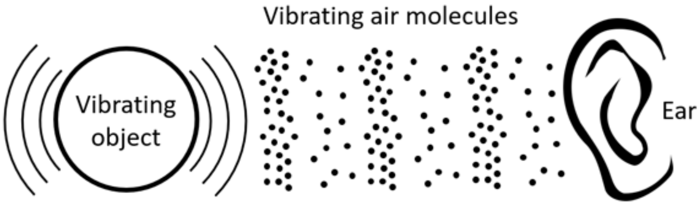


Fig 1.2

Propagation of sound waves in air, other media, fluids and solids

Production of sound waves

A tuning fork serves as a useful illustration of how a vibrating object can produce sound. The fork consists of a handle and two tines. When the tuning fork is hit with a rubber hammer, the tines begin to vibrate. The back and forth vibration of the tines produce disturbances of surrounding air molecules.

Propagation of sound waves in air

The simple experiment to understand the progationn of sound waves in air medium.

The setup consists of a tuning fork a metallic ball attached to a thread stinted to a solid board. The ball flipped to hit the tuning fork. The tuning fork starts oscillating back and forth. This produces the waves in the air as shown in the figure

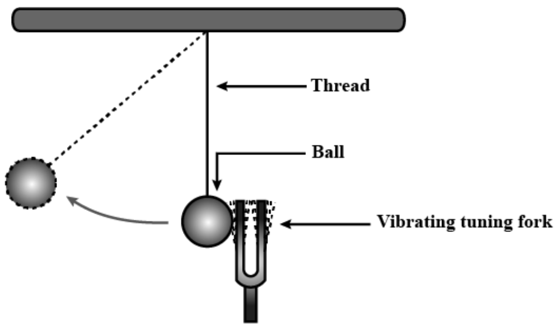


Fig 1.3

Fig 1.4

Propagation of sound waves in fluid

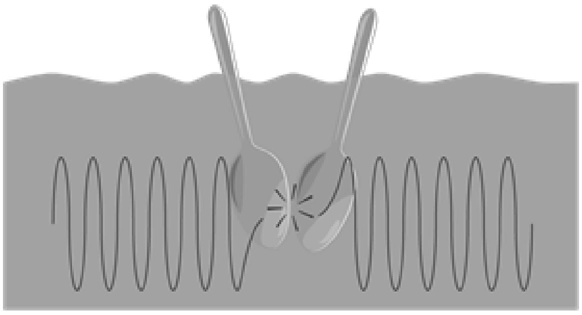


Fig 1.5

The sound waves move through solid, liquid and gas by vibrating the molecules in the matter.

In water, sound travels four times faster than it does in the air. Sound moves faster in water than air because water particles are packed very tightly compared to air.

Thus, the energy the sound waves carry is transported faster.

Propagation of sound waves in solids

The sound waves move through different mediums by vibrating the molecules in it.

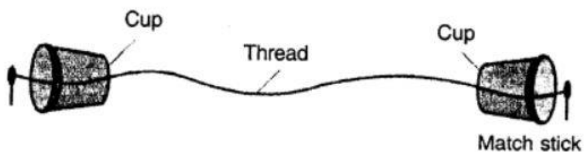


Fig 1.6

The molecules in the solids are packed very tightly when compared to liquid as well as gas. Thus, it enables sound to travel much faster through a solid than gas and liquid.

Sound travels 13 times faster in solid than through the air.

Propagation of sound waves in vacuum

Sound cannot travel in a vacuum because sound waves always need molecules of some kind to travel in a given space.

Sound produced by a source travels from one point to another by transmitting the energy through the molecules of the medium around it. This is why sound does not travel through a vacuum, where there are no molecules in it.

Velocity

The rate of change of an object's position with respect to time. It is a vector quantity, meaning it has both magnitude and direction. Mathematically, velocity (v) is defined as the displacement (Δx) of an object divided by the time (Δt) it takes to undergo that displacement:

$$v = \frac{\Delta x}{\Delta t}$$

Where:

- v represents velocity,
- Δx is the change in position (displacement),
- Δt is the change in time.

The SI unit of velocity is metre per second (m/s).

Frequency

Frequency (f) of a wave is inversely proportional to its period (T), where the period is the time it takes for one complete cycle of the wave.

$$f = \frac{1}{T}$$

where;

f is the frequency in hertz (Hz)

T is the period of the wave

SI unit of frequency is Hertz.

Wavelength

The wavelength is the distance between two consecutive compressions or rarefactions of the wave. It represents the physical distance between two points in the wave where the air pressure is at its maximum (compression) or minimum (rarefaction).

Wavelength is inversely related to frequency, according to the formula

$$\lambda = \frac{c}{f}$$

Where;

- ❖ λ is the wavelength,
- ❖ c is the speed of the wave
- ❖ f is the frequency of the wave.

The SI unit of wavelength is metre (m)

Time period

Time period (T) is the reciprocal of the frequency (f), and it can be expressed as:

$$T = \frac{1}{f}$$

Where;

- ❖ T is the time period,
- ❖ f is the frequency of the wave or oscillation.

SI unit of time period is second (s)

Intensity

The intensity of sound refers to the amount of energy transmitted by sound waves per unit area perpendicular to the direction of wave propagation. In simpler terms, it measures how much sound energy is passing through a particular region in space.

$$I = \frac{P}{A}$$

Where;

- I is the sound intensity,
- P is the power of the sound wave, typically measured in watts (W),
- A is the area through which the sound wave is passing, typically measured in square meters (m^2).

SI unit of sound intensity is Decibel

Classification of sound on frequency

Sound waves are classified into three categories on the basis of frequency.

- Infrasonic sound (below 20 Hz)
- Audible sound (between 20 Hz to 20,000 Hz)
- Ultra sound (above 20,000 Hz)

Infrasonic sound

The term "infrasonic" applied to sound refers to sound waves below the frequencies of audible sound, and nominally includes anything under 20 Hz.

Sources of infrasound in nature include volcanoes, avalanches, earthquakes and meteorites. The eruption of the Fuego volcano in Guatemala produced infrasonic sound in excess of 120 decibels in the range below 10Hz.

Ultra sound

The term "ultrasonic" applied to sound refers to anything above the frequencies of audible sound, and nominally includes anything over 20,000 Hz. Frequencies used for medical diagnostic ultrasound scans extend to 10 MHz and beyond.

Bats use ultrasonic sound for navigation. Their ability to catch flying insects while flying full speed in pitch darkness is astounding. Their sophisticated echolocation permits them to distinguish between a moth (supper) and a falling leaf.

Audible sound

A reasonably standard definition of audible sound is that it is a pressure wave with frequency between 20 Hz and 20,000 Hz and with an intensity above the standard threshold of hearing. Since the ear is surrounded by air, or perhaps under water, the sound waves are constrained to be longitudinal waves. Normal ranges of sound pressure and sound intensity may also be specified.

Classification of sound on velocity

Sound waves are classified into four categories on the basis of velocity

- Supersonic Sound
- Hypersonic Sound
- Subsonic Sound
- Transonic Sound

Supersonic Sound:

Supersonic refers to speeds faster than the speed of sound in a particular medium. In air at sea level and room temperature, the speed of sound is approximately 343 meters per second (m/s) or 1235 kilometres per hour (km/h). Sounds traveling faster than this speed are considered supersonic. Examples include the shockwaves produced by aircraft traveling faster than the speed of sound, creating a sonic boom.

Hypersonic Sound:

Hypersonic refers to speeds significantly faster than the speed of sound. While hypersonic typically refers to velocities exceeding Mach 5 (five times the speed of sound), in the context of sound, it can refer to any sound waves traveling at such extreme velocities. These velocities are often encountered in aerospace engineering and high-speed research.

Subsonic Sound:

Subsonic refers to speeds slower than the speed of sound. In air, this means velocities below approximately 343 meters per second. Most everyday sounds, such as conversations, music, and ambient noise, are subsonic.

Transonic Sound:

Transonic refers to speeds close to the speed of sound. This region often involves a mix of subsonic and supersonic flow and can lead to phenomena such as shockwaves and sonic booms.

Human and animal sound perception

The perception of sound is a major form of communication in humans and it's the key in the development of the civilisation. Before the civilized era sound is vital to know the awareness of the surrounding for the self-defence and hunting.

Many animals, including dogs, can hear frequencies in the ultrasonic range (above 20,000 Hz). Dog whistles used to train dogs have frequencies between about 23,000 Hz and 54,000 Hz so dogs (and many other animals) can hear them but humans cannot. Birds and fish generally hear a much smaller range of frequencies than mammals and also do not hear softer sounds as well. Some insects hear very specific bands of frequencies, for example crickets have a hearing range for signals generated by other crickets for communication purposes but they have a different, unconnected range of hearing to detect the echolocation signals coming from bats, which are a predator.

Animals	Hearing range in hertz
Humans	20–20,000
Bats	2000–110,000
Elephant	16–12,000
Fur Seal	800–50,000
Beluga Whale	1000–123,000
Sea Lion	450–50,000
Harp Seal	950–65,000
Harbor Porpoise	550–105,000
Killer Whale	800–13,500
Bottlenose Dolphin	90–105,000
Porpoise	75–150,000
Dog	67–45,000
Cat	45–64,000
Rat	200–76,000
Opossum	500–64,000
Chicken	125–2,000

Parakeet	200–8,500
Horse	55–33,500

Mechanism of ear and hearing

Hearing depends on a series of complex steps that change sound waves in the air into electrical signals. Our auditory nerve then carries these signals to the brain.

Sound waves enter the outer ear and travel through a narrow passageway called the ear canal, which leads to the eardrum.

The eardrum vibrates from the incoming sound waves and sends these vibrations to three tiny bones in the middle ear. These bones are called the malleus, incus, and stapes.

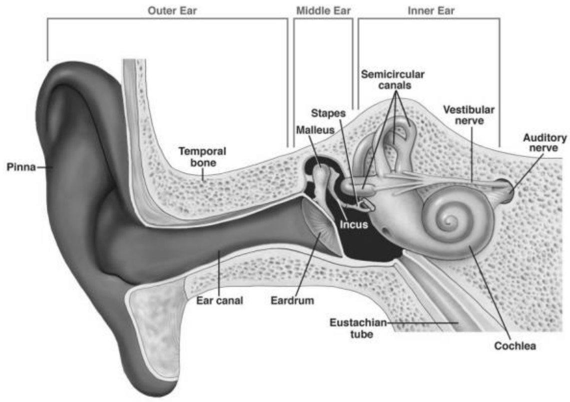


Fig 1.7

The bones in the middle ear amplify, or increase, the sound vibrations and send them to the cochlea, a snail-shaped structure filled with fluid, in the inner ear. An elastic partition runs from the beginning to the end of the cochlea, splitting it into an upper and lower part. This partition is called the basilar membrane because it serves as the base, or ground floor, on which key hearing structures sit.

Once the vibrations cause the fluid inside the cochlea to ripple, a traveling wave forms along the basilar membrane. Hair cells sensory cells sitting on top of the basilar membrane ride the wave. Hair cells near the wide end of the snail-shaped cochlea detect higher-pitched sounds, such as an infant crying. Those closer to the centre detect lower-pitched sounds, such as a large dog barking.

As the hair cells move up and down, microscopic hair-like projections (known as stereocilia) that perch on top of the hair cells bump against an overlying structure and bend. Bending causes pore-like channels, which are at the tips of the stereocilia, to open up. When that happens, chemicals rush into the cells, creating an electrical signal.

The auditory nerve carries this electrical signal to the brain, which turns it into a sound that we recognize and understand.

The human ear can respond to minute pressure variations in the air if they are in the audible frequency range, roughly 20 Hz - 20 kHz.

It is capable of detecting pressure variations of less than one billionth of atmospheric pressure. The threshold of hearing corresponds to air vibrations on the order of a tenth of an atomic diameter. This incredible sensitivity is enhanced by an effective amplification of the sound signal by the outer and middle ear structures. Contributing to the wide dynamic range of human hearing are protective mechanisms that reduce the ear's response to very loud sounds. Sound intensities over this wide range are usually expressed in decibels.

Psychoacoustics

Psychoacoustics is the scientific study of how humans perceive and interpret sound. It combines principles from psychology and acoustics to understand how the brain processes and interprets auditory information. Psychoacoustics explores various aspects of human hearing, including perception of pitch, loudness, timbre, sound localization, and masking effects.

Key areas of study in psychoacoustics include:

Frequency Perception:

Psychoacoustics investigates how humans perceive different frequencies of sound and how they interpret pitch. This includes phenomena such as the critical bands, which are ranges of frequencies within which sounds will interfere with each other and affect perception.

Intensity Perception:

Understanding how humans perceive the intensity or loudness of sound is another focus of psychoacoustics. This includes studying the relationship between physical measures of sound intensity (such as decibels) and perceived loudness.

Temporal Perception:

Psychoacoustics examines how humans perceive the timing and duration of sounds. This includes studies on auditory temporal resolution, which refers to the ability to distinguish between brief auditory events.

Spatial Perception:

Investigating how humans localize sound in space is another aspect of psychoacoustics. This involves understanding how the brain processes cues such as interaural time differences and interaural level differences to determine the location of a sound source.

Masking and Auditory Perception:

Psychoacoustics explores the phenomenon of masking, where the perception of one sound is affected by the presence of another sound. This includes studies on simultaneous masking, where one sound makes another sound inaudible or less audible.

Psychoacoustics has applications in various fields, including audio engineering, music production, telecommunications, and hearing aid design. By understanding how humans perceive sound, researchers and practitioners can develop technologies and techniques to enhance sound quality, improve communication systems, and address hearing-related issues.

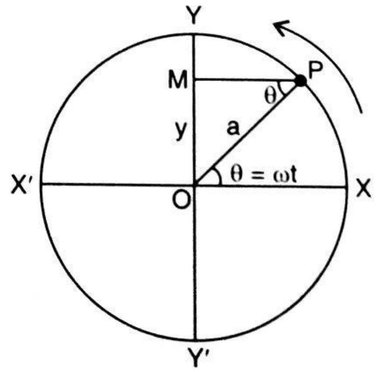
UNIT - II

SIMPLE VIBRATING SYSTEMS

Simple harmonic motion

Let *P* be a particle moving on the circumference of a circle of radius a with a uniform angular velocity ω. *O* is the centre of the circle.

A perpendicular *PM* is drawn from the particle on the diameter *YY'* of the circle. As the particle *P* moves round the circle, the foot of the perpendicular *M* vibrates along the diameter *YY'*. Since the motion of *P* is uniform, the motion of *M* is periodic. As the particle P completes one revolution, the foot of the perpendicular *M* completes one vertical oscillation. The distance *OM* is called the displacement and is denoted by *y*



The particle moves from X to P in time t.

$\angle POX = \angle MPO = \theta = \omega$ t

Fig 2.1

From the ΔMPO,

$\sin\theta = \sin\omega\, t = \frac{OM}{a}$

$OM = y = a \sin\omega\, t$

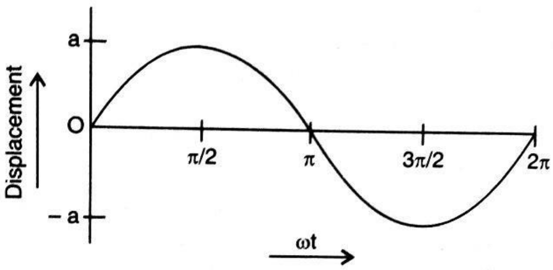


Fig 2.2

OM is called the displacement of the vibrating particle. The displacement of a vibrating particle at any instant can be defined as its distance from the mean position of rest. The maximum displacement of a vibrating particle is called amplitude.

$$\text{Displacement} = y = a \sin\omega\, t$$

The figure 2.1 shows the changes in the displacement of a vibrating particle in one complete vibration.

$$Velocity = v = \frac{dy}{dt} a\omega\, cos\omega t = \omega\sqrt{(a^2 - y^2)}$$

$$Acceleration = \frac{d^2y}{dt^2} - a\omega\, sin\omega t = \omega t - \omega^2 y \qquad (1)$$

acceleration is directly proportional to displacement and directed towards a fixed point. This type of motion is called Simple harmonic motion.

Tuning fork

XY is a rod which vibrates transversely with nodes at N and N. (Fig. 2.3).When this rod is bent gradually as shown in Fig. 2.3 (ii), (iii) and (iv), the nodes come closer. As the curvature is gradually increased, the nodes shift towards each other more and more. When the rod becomes U- shaped, the positions of nodes and antinodes are as shown in Fig. 2.3(iv). When this U- shaped rod is attached to a metal stem, they come still closer and this arrangement is called a tuning fork. The tuning fork vibrates in three portions as shown in Fig. 2.3(v). The free ends of the tuning fork behave as antinodes. The position where the stem is attached also behaves as an antinode.

A tuning fork is of great use as a source of standard frequency. It is made of an alloy of nickel and steel. A tuning fork maintains its pitch for a number of years and it can be set into vibration when one of the prongs is struck against a hard rubber pad. Tuning forks of frequencies 256, 288, 320, 341. 33, 384, 426.66, 480 and 512 are commonly manufactured.

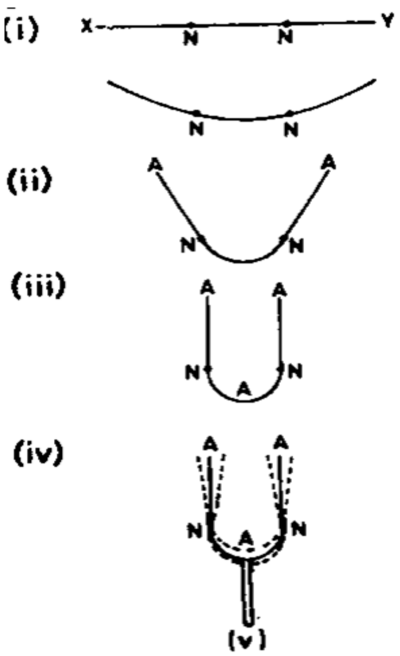


Fig 2.3

Amplitude

Amplitude is the character of the sound wave which determines the loudness. Amplitude is the maximum displacement of the particle from its mean position. So if we are going to increase the amplitude of the tuning fork the louder the sound produced.

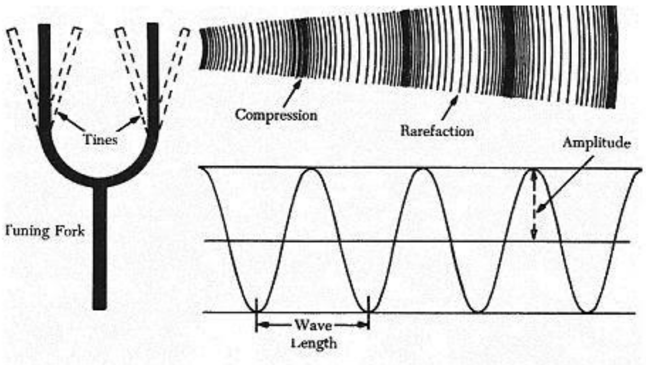


Fig 2.4

Phase

There are two types of phases in a tuning fork. They are

- In Phase
- Anti phase

In Phase: In this mode the two prongs of the tuning fork move in the same direction.

Anti phase: In this mode the two prongs of the tuning fork move in opposite direction to one another

The In phase and Anti phase were shown in the fig2.5

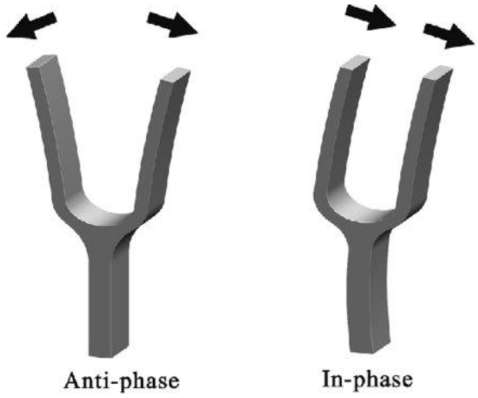


Fig 2.5

Energy:

When a tuning fork is struck, energy is imparted to it, causing it to vibrate. This energy is initially in the form of mechanical energy provided by the striking force. As the tuning fork vibrates, this mechanical energy is converted into vibrational kinetic energy.

Energy Loss:

While the tuning fork vibrates, it inevitably experiences energy loss due to various factors such as air resistance, friction at the point of attachment, and material damping. These losses gradually diminish the amplitude of vibration over time, leading to a decrease in the loudness and duration of the sound produced.

Dissipation:

Dissipation refers to the conversion of mechanical energy into other forms, such as heat, during the damping process. As the tuning fork vibrates and experiences damping, a portion of the mechanical energy is dissipated as heat, sound radiation, and other forms of energy, leading to a gradual decay in vibration.

Power:

The power of a tuning fork refers to the rate at which it converts mechanical energy into sound energy. It is related to both the amplitude and frequency of vibration. A tuning fork with a higher amplitude and frequency will typically produce sound waves with greater power, resulting in a louder and more sustained tone.

Damping:

Damping refers to the process by which the amplitude of the tuning fork's vibrations is reduced over time due to the dissipation of energy. Damping can be classified into several types, including viscous damping (related to friction within the material of the tuning fork) and structural damping (related to energy dissipation within the material itself).

Damped vibration

In actual practice, when the pendulum vibrates in air medium there are frictional forces and consequently energy s dissipated in each vibration. The amplitude of swing decreases continuously with time and finally the oscillations die out. Such vibrations are called free damped vibrations. The dissipated energy appears as heat either within the system itself or in the surrounding medium. The dissipative force due to friction etc. (resistance in LCR Circuit) is proportional to the velocity of the particle at that instant. Let μ $\frac{dy}{dt}$ be the dissipative force due to friction etc.

Therefore, the differential equation in the case of free damped vibrations is

$$m\frac{d^2y}{dt^2} + Ky + \mu\left(\frac{dy}{dt}\right) = 0 \qquad (1)$$

$$\frac{d^2y}{dt^2} + \left(\frac{\mu}{m}\right)\frac{dy}{dt} + \left(\frac{K}{m}\right)y = 0 \qquad (2)$$

This equation is similar to a general differential equation,

$$\frac{d^2y}{dt^2} + 2b\frac{dy}{dt} + k^2y = 0 \qquad (3)$$

The solution of this equation is

$$y = ae^{-bt}\sin(\omega t - \alpha) \qquad (4)$$

The general solution of equation (2) is also given by

$$y = Ae^{\left(-b+\sqrt{b^2-k^2}\right)}t + \ -Be^{\left(-b+\sqrt{b^2-k^2}\right)}t$$

Here,

$$b = \frac{\mu}{2m} \quad and \quad k^2 = \frac{K}{m}$$

$$\omega = \sqrt{k^2 - b^2}$$

$$\omega = \sqrt{\frac{K}{m} - \frac{\mu^2}{4m^2}}$$

$$n = \frac{\omega}{2\pi} = \frac{1}{2\pi}\sqrt{k^2 - b^2}$$

Laws of vibration in stretched strings

Velocity of a transverse wave travelling in stretched string is given by:

$$v = \sqrt{\frac{T}{\mu}}$$

Where T is the tension in the string and μ is the mass per unit length. Now, the fundamental frequency of the stretched string is given by:

$$f = \frac{1}{2l}\sqrt{\frac{T}{\mu}}$$

l is the resonating length.

From the above expression, there are three laws of transverse vibration of string.

Law of length: The fundamental frequency is directly proportional to the resonating length (L) of the string.

From the above expression, there are three laws of transverse vibration of string.

Law of length:

The fundamental frequency is directly proportional to the resonating length (L) of the string.

$$f \propto \frac{1}{L}$$

Verification:

When we take different tuning forks of different frequencies, and measure the resonating length for each of them keeping tension applied and the material of the wire as constant. The product of frequency and resonating length of one tuning fork was found equal to the frequency and resonating length of another tuning fork. This verifies the law of length.

Law of Tension:

The fundamental frequency is directly proportional to the square root of the tension.

$$f \propto \sqrt{T}$$

Verification:

The resonance of different tuning forks of different frequencies was observed by varying tension keeping resonating length and wire as constant. When a graph for f vs $\sqrt{T}$ is plotted, a straight line is obtained. This verifies the law of tension.

Law of Mass:

The fundamental frequency is inversely proportional to the square root of the mass per unit length.

$$f \propto \frac{1}{\mu}$$

Verification:

The resonance is observed by taking different tuning forks of different frequencies and different wires with separate mass per unit length (μ) keeping resonating length and tension applied as constant. When the graph for f vs $1\sqrt{\mu}$ is plotted and the graph is found to be a straight line. So, law of mass is verified.

Organ pipe

It is a wind instrument. It consists of the mouthpiece M, a slot A and a lip (Fig.2.5). If the other end of the pipe is closed it is called a closed end organ pipe. If theother end is open it is called an open end organ pipe. Air is blown through themouthpiece M and is ejected out through the slot A. The lip L vibrates. The aircolumn acts as a resonator. The air column resonates with a particular fundamental frequency depending on its length. The open end behaves as an antinode and theclosed end as a node.

Closed end organ pipe

Suppose the length of the pipe is l. For the fundamental frequencyof vibration of the air column

$$l = \frac{\lambda_1}{4} \ldots \text{open end}$$

Frequency,

$$n_1 = \frac{V}{4l}$$

For the first overtone,

$$l = \frac{3\lambda_2}{4} \ldots \text{closed end}$$

Frequency,

$$n_2 = \frac{3V}{4l} = 3n_1$$

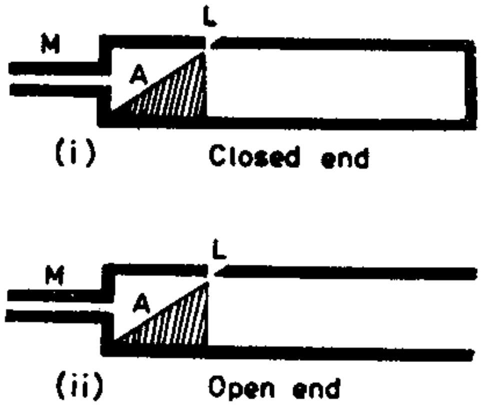


Fig 2.5

For the second overtone,

$$l = \frac{5\lambda_3}{4}$$

Frequency,

$$n_3 = \frac{5V}{4l} = 5n_1$$

Therefore, in the case of a closed end organ pipe , only the odd harmonics are produced and the even harmonics are missing. The fundamental frequency = V/4l.

Open end organ pipe

In an open end pipe antinodes are present at both the ends. Suppose the length of the pipe is l. For the fundamental frequency of vibration of the air column

$$l = \frac{\lambda_1}{2} \ldots \text{open end}$$

$$n_1 = \frac{V}{2l}$$

For the first overtone,

$$l = \lambda_2$$

Frequency,

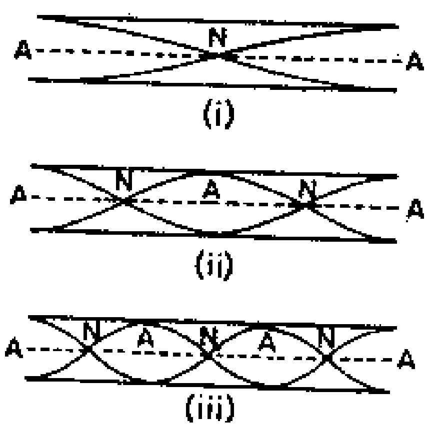


Fig 2.6

$$n_2 = \frac{V}{l} = 2n_1$$

For the second overtone,

$$l = \frac{3\lambda_3}{2}$$

Frequency,

$$n_2 = \frac{3V}{22l} = 3n_1$$

Therefore the harmonics having frquencies $n_1, 2n_1, 3n_1 \ldots etc.$, are produced. Hence all the harmonics are present.

Harmonics

A harmonic is a wave or signal whose frequency is an integral (whole number) multiple of the frequency of the same reference signal or wave. As part of the harmonic series, the term can also refer to the ratio of the frequency of such a signal or wave to the frequency of the reference signal or wave.

Quality of sound:

Quality of sound was determined by the following factors

Pitch:

Pitch refers to the perceived frequency of a sound wave, which determines how high or low a sound is. Higher frequencies result in higher pitches, while lower frequencies produce lower pitches. Pitch is measured in hertz (Hz), where one hertz equals one cycle per second. In music, pitch is crucial for creating melodies and harmonies, and it is typically represented by musical notes on a scale.

Timbre:

Timbre, also known as tone colour or quality, refers to the unique characteristic of a sound that allows us to distinguish between different sources of sound, even when they have the same pitch and loudness. It is influenced by various factors, including the shape, material, and resonance properties of the sound-producing object or instrument. Timbre is what allows us to differentiate between, for example, the sound of a piano and a guitar playing the same note.

Loudness:

Loudness is the subjective perception of the intensity or amplitude of a sound wave, which determines how loud or soft a sound is. It is measured in decibels (dB) and is influenced by the physical properties of the sound wave, such as its amplitude and energy. The human ear perceives loudness logarithmically, meaning that a doubling of sound intensity results in an increase of about 10 dB in perceived loudness. In music and everyday life, loudness plays a crucial role in conveying emotion, emphasis, and dynamics.

Octaves and Musical Notes:

In music theory, an octave is the interval between one musical pitch and another with double or half its frequency. This means that notes an octave apart sound similar but have different pitches. For example, middle C (C4) on a piano is one octave higher than C3 and has twice the frequency. Musical notes are named using letters from A to G, with variations such as sharps and flats denoted by symbols. Octaves allow for the organization of musical scales and provide a framework for understanding the relationships between different pitches in music.

UNIT - III

MUSICAL TONE

Pure Tones

Pure tones are sound waves that consist of only a single frequency. They have a sinusoidal waveform, meaning they follow a smooth, regular pattern without any distortion or complexity. Pure tones are often used in scientific experiments and mathematical models to study the behavior of sound waves in idealized conditions. While pure tones are rare in natural environments, they provide a useful reference point for understanding the properties of more complex sounds.

Simple Tones

Simple tones are sound waves that consist of a fundamental frequency and may also contain a series of harmonics. The fundamental frequency is the lowest frequency component of the sound wave, while harmonics are integer multiples of the fundamental frequency. Simple

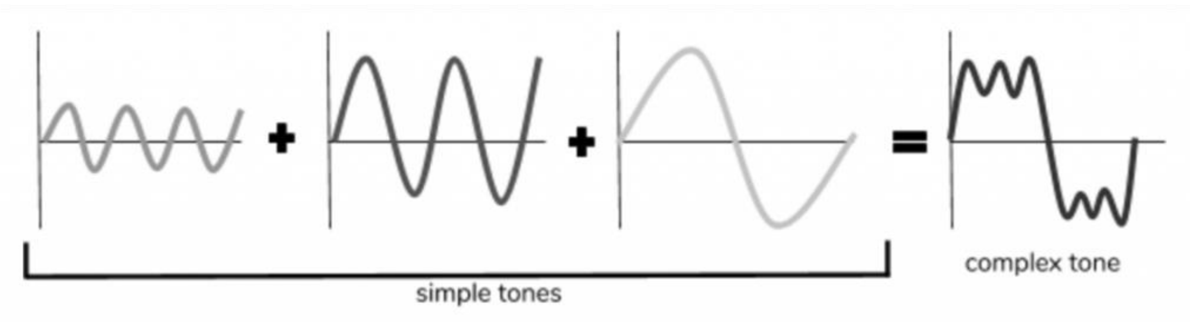


Fig 3.1

tones are characteristic of many musical instruments and voice production. They have a more complex waveform compared to pure tones but still exhibit a regular, repetitive pattern. Simple tones contribute to the timbre or quality of a sound and are essential for creating rich, dynamic musical textures.

Sine Wave

A sine wave is a pure tone that oscillates smoothly between a minimum and maximum amplitude, following a sinusoidal waveform. In the context of music, sine waves represent the simplest form of sound, containing only a single frequency without any harmonics or overtones. While pure sine waves are rarely encountered in musical instruments, they serve as fundamental building blocks in electronic synthesis and signal processing. Sine waves are often used as reference tones in tuning instruments and testing audio equipment due to their pure, unambiguous sound.

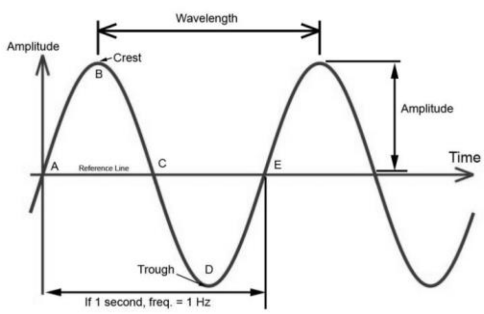


Fig 3.2

Cosine Wave

A cosine wave is closely related to a sine wave and shares the same mathematical properties but is shifted in phase by 90 degrees. In musical contexts, cosine waves are less commonly used directly as sound sources compared to sine waves. However, cosine waves are integral to Fourier analysis, a mathematical technique used to analyze complex sounds by decomposing them into their constituent sine and cosine components. By representing musical tones as combinations of sine and cosine waves at different frequencies, Fourier analysis provides insights into the timbral characteristics and harmonic structure of sound.

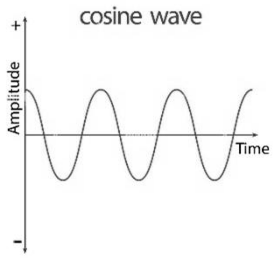


Fig 3.2

Frequencies

Frequency refers to the number of cycles of a wave that occur in a unit of time, usually one second. It is measured in hertz (Hz). In the context of well-defined frequencies, it means that the wave oscillates at a specific, constant rate, producing a consistent pitch. For example, in the case of musical tones, each note corresponds to a specific frequency, resulting in a well-defined pitch.

Wavelengths

Wavelength is the physical distance between two consecutive points on a wave that are in phase with each other, such as two peaks or two troughs. It is denoted by the symbol lambda (λ) and is related to frequency by the equation: $\lambda = c / f$, where c is the speed of the wave and f is the frequency. In waves with well-defined wavelengths, the distance between successive crests (or troughs) remains constant.

Amplitudes

Amplitude refers to the maximum displacement or intensity of a wave from its equilibrium position. It represents the strength or magnitude of the wave and is often measured from the equilibrium position to the crest (or trough) of the wave. In waves with well-defined amplitudes, the maximum displacement remains constant over time, resulting in a consistent loudness or brightness.

Phases

Phase refers to the position of a point on a wave relative to a reference point, often measured in degrees or radians. It indicates the fraction of a wave cycle that has elapsed since a specific reference point. In waves with well-defined phases, the relationship between different points on the wave remains constant over time, allowing for precise synchronization and interference patterns.

Partial tones

A time graph of one of the prongs of a tuning fork, the waves transmitted through the air to an observer, and the vibration the waves impose upon the eardrum serve as good examples of this. See Figure 3.3. Any vibrating body that rapidl executes simple harmonic motion in air emits a sinusoidal sound wave. This sound wave is referred to as a pure tone, although the aural perception of even a pure tone is impure . Actually, nearly all tone produced by musical instruments, and other sources in general, are not pure tones but mixtures of pure-tone frequencies called partials. The lowest such frequency is called the fundamental. All partials higher in frequency than the fundamental are referred to as upper partials, or overtones. In special cases, the frequencies of these overtones are exact multiples of the fundamental and are called harmonics. If we designate the frequency of any fundamental by f, all higher harmonics are designated by 2f, 3f, 4f 5f and so on. If, for example, we select a fundamental frequency of 200 Hz and call it the first harmonic, it and its higher harmonics are given by

First harmonic: 1f= 200 Hz

Second harmonic: 2f= 400 Hz

Third harmonic: 3f= 600 Hz

Fourth harmonic: 4f= 800 Hz

and so forth.

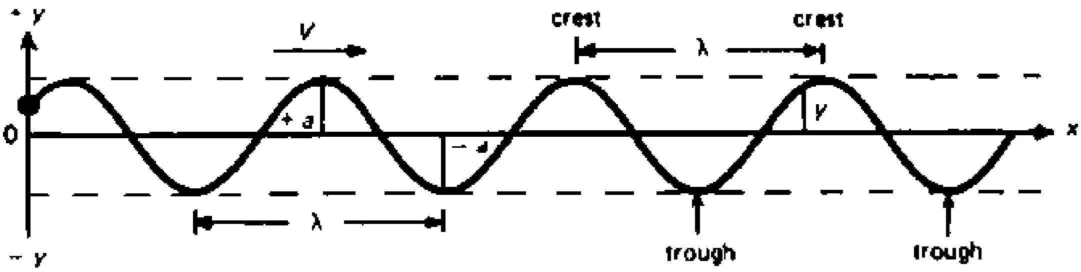


Fig 3.3

If singing voices, or different musical instruments, sound notes of the same pitch and loudness, we recognize the pitch as that of the fundamental, but the timbre or quality of each note differs from the others by virtue of the relative amplitudes of its partials. In most cases, particularly with the percussion instruments, the upper partials (overtones) are not exact multiples of the fundamental frequency. Such an overtone is called an inharmonic partial, and the combined tone is often unpleasant.

Assembly of pure tones

An assembly of pure tones refers to a combination of multiple sound waves, each consisting of a single frequency and amplitude. When these pure tones are combined, they create a complex waveform with a unique timbre or tone color. Here's a brief explanation of the assembly of pure tones

Combination of Frequencies

In an assembly of pure tones, each individual tone has its own frequency, which may be distinct or related to the frequencies of other tones present. The combination of different frequencies results in a complex waveform with varying patterns of constructive and destructive interference, leading to a rich and diverse sound quality.

Variation in Amplitudes

The amplitudes of the pure tones in the assembly may vary, meaning that some tones may be louder or softer than others. The relative amplitudes of the tones influence the overall volume and intensity of the combined sound. By adjusting the amplitudes of individual tones, the overall balance and character of the assembly can be controlled.

Complex tones

A complex tone is a sound wave that repeats with a given pattern BUT the pattern is not a sine wave. As we will see the complex tone consists not only of one frequency sine wave but rather a number of different frequency sine waves superimposed on top of one another. In the time domain the repeat time of the pattern gives the period of the lowest frequency sine wave that is contained within the complex tone.

Superposition of simple tones

The superposition of simple tones refers to the process by which multiple individual sound waves, each consisting of a single frequency and amplitude, are combined to create a complex waveform. This principle is based on the idea that when two or more waves occupy the same space, their amplitudes add together at every point in time, resulting in a combined waveform with characteristics determined by the individual waves. Here's a brief explanation of the superposition of simple tones and the resulting complex waveform:

Superposition Principle

The superposition principle states that when two or more waves overlap in space, the displacement of the resulting waveform at any given point is equal to the sum of the displacements of the individual waves at that point. In other words, the amplitude of the combined waveform is the sum of the amplitudes of the individual waves.

Combination of Frequencies

When simple tones with different frequencies are superimposed, the resulting waveform contains a combination of these frequencies. The specific combination of frequencies depends on the frequencies of the individual tones and their relative amplitudes. As a result, the combined waveform exhibits a complex pattern of oscillation with multiple frequency components.

Creation of Complex Waveform

The superposition of simple tones results in the creation of a complex waveform with characteristics determined by the frequencies, amplitudes, and phase relationships of the constituent tones. The interaction between the individual waves leads to patterns of constructive and destructive interference, which influence the overall shape and behavior of the combined waveform.

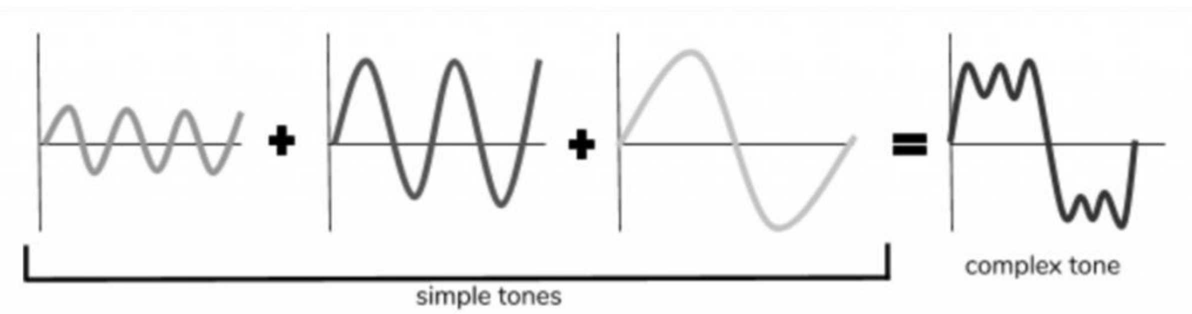
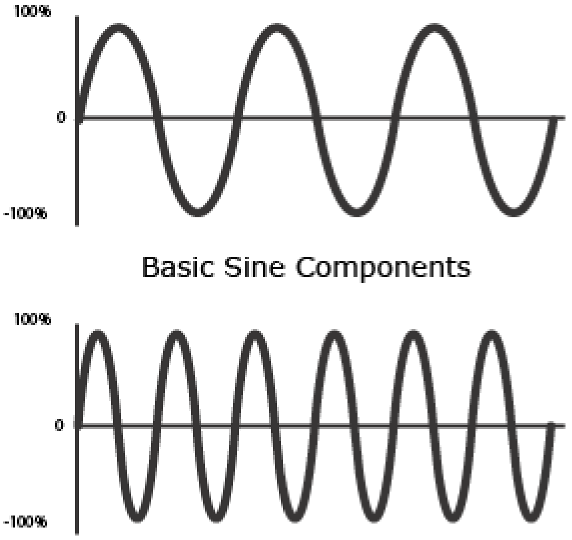


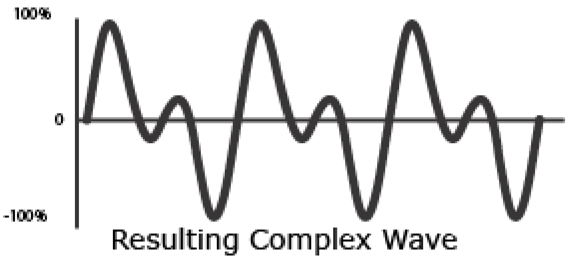
Fig 3.4

Periodic complex wave form

Fourier Analysis, named after the nineteenth century French mathematician Jean Baptiste Fourier, enables one to break down complex periodic waveforms into their basic components, which happen to be sine waves of various frequencies, amplitudes, and phases. The opposite method, combining sine waves of various frequencies, amplitude, and phase to create complex periodic waveforms, is Fourier Synthesis.

A complex waveform is the result of combining the instantaneous amplitudes of two (or more) sine waves.





Formants

Formants are spectral peaks in the frequency spectrum of a sound wave, particularly in the case of speech sounds. They are produced by resonances in the vocal tract and are characterized by their frequencies, amplitudes, and bandwidths. Each vowel sound in human speech is associated with specific formant frequencies, which are determined by the shape and configuration of the vocal tract. Formants play a crucial role in speech perception and recognition, as they contribute to the distinctiveness of different vowels and consonants.

Resonances

Resonances refer to the natural frequencies at which a system vibrates most strongly when excited by an external force or input. In the context of sound, resonances occur when the frequency of an external sound wave matches the natural frequency of a vibrating object or medium. This causes the object or medium to absorb energy from the sound wave and vibrate with increased amplitude. Resonances are responsible for the production of formants in speech sounds, as well as the characteristic timbre of musical instruments. They also play a role in sound amplification and filtering in acoustic systems.

Sound Envelope:

The sound envelope describes the temporal evolution of the intensity or amplitude of a sound wave over time. It typically consists of four main stages: attack, decay, sustain, and release (ADSR). The attack phase represents the initial increase in amplitude when the sound begins, the decay phase represents the subsequent decrease in amplitude, the sustain phase represents the stable level of amplitude during the main part of the sound, and the release phase represents the final decrease in amplitude as the sound fades away. The sound envelope is an important perceptual cue for identifying and characterizing sounds, as it provides information about their temporal dynamics and structure.

UNIT – IV

PRODUCTION OF MUSICAL SOUNDS

Human voice

Most sounds of the human voice, Originate in the vibrations of the vocal cords in the larynx. See Figure 4.1. As the air from the lungs passes these stringlike membranes, the Bernoulli effect sets them vibrating. The frequency of vibration depends mainly on the muscular tension applied to the cords. When singing, these tensions are altered, just as stringed instruments are tuned by changing the string tension. A good singing voice can normally span about two octaves by varying vocal cord tension.

Larynx (sound box)

Each time the cords part, a sharp gust of air is emitted through the opening (glottis) into the oral cavity. A series of such sharp gusts, as shown in Figure 4.2 (a), is characterized by a series of harmonics of only slowly decreasing amplitude. Such an overtone-rich sound produces a bright tone. If the air from the lungs is passed through the larynx cords more gently, the vocal cords do not completely close, resulting in a train of waves more closely resembling a sine wave, as shown in Figure 4.2 (b). Such a wave is dominated by the fundamental frequency, with a rapidly decreasing set of overtones. The resulting sound is generally described as a dark tone.

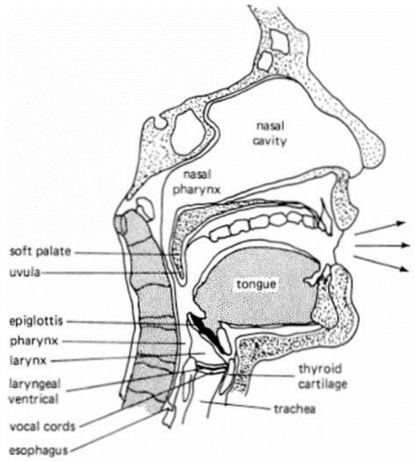
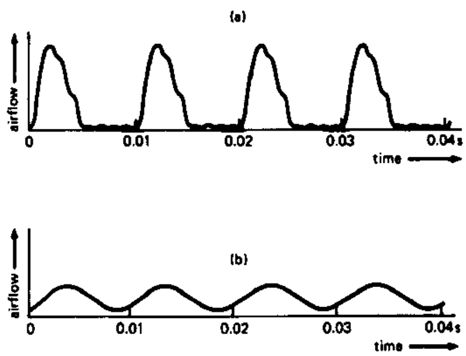


Fig 4.1

The vocal tract acts as a resonator; it is similar in many ways to a reed organ pipe in that it acts as if it were closed at the larynx and open at the mouth. If it were a cylinder of length L it would resonate at frequencies whose wavelengths are, and so on. However, the soft walls of the vocal cavities absorb the sound waves much more than do the hard walls of the reed instruments. This damping of the sound wave broadens the resonant frequencies, so that resonance occurs over several broad frequency ranges instead of at narrowly defined frequencies. See Figure 4.3(b). These broad resonances are called formants



Typical wave forms for the larynx at (a) high intensity and (b) low intensity.

Fig 4.2

Mechanism of vocal sound production

A whisper is caused by allowing air to move past stationary vocal cords, causing a hiss rather than a definite tone. Since a hiss is basically noise a random combination of all frequencies the resulting audible sound will be caused by those generated frequencies that resonate in the vocal tract at the formant frequencies. The rather indefinite pitch of the whisper is therefore characteristic of the formants. If a tone is vocalized by causing the vocal cords to vibrate, any partials of the vocal cord frequency [Figure 4.3 (a) that fall within the formants will produce a strong resonant response, as shown in Figure 4.3 (c). If different frequencies are generated by the vocal cords, but the vocal cavity remains unchanged, different partials will be accentuated. The quality of a voiced sound is determined by the size and shape of the larynx, throat (pharynx), mouth, and nasal cavities. The singer has only partial control over these variables, although voice training can extend the degree of control. Studies indicate that the jaw angle and lips control mainly the first formant, whereas the position and size of the constriction between the tongue and mouth roof determines mainly the second formant.5 Control of the velum (soft palate) can open the nasal cavity, adding another resonance at around 1000 Hz. The adult male generally can vary his first formant between 200–700 Hz and the second formant between 700–2500 Hz. The third formant will be higher yet.

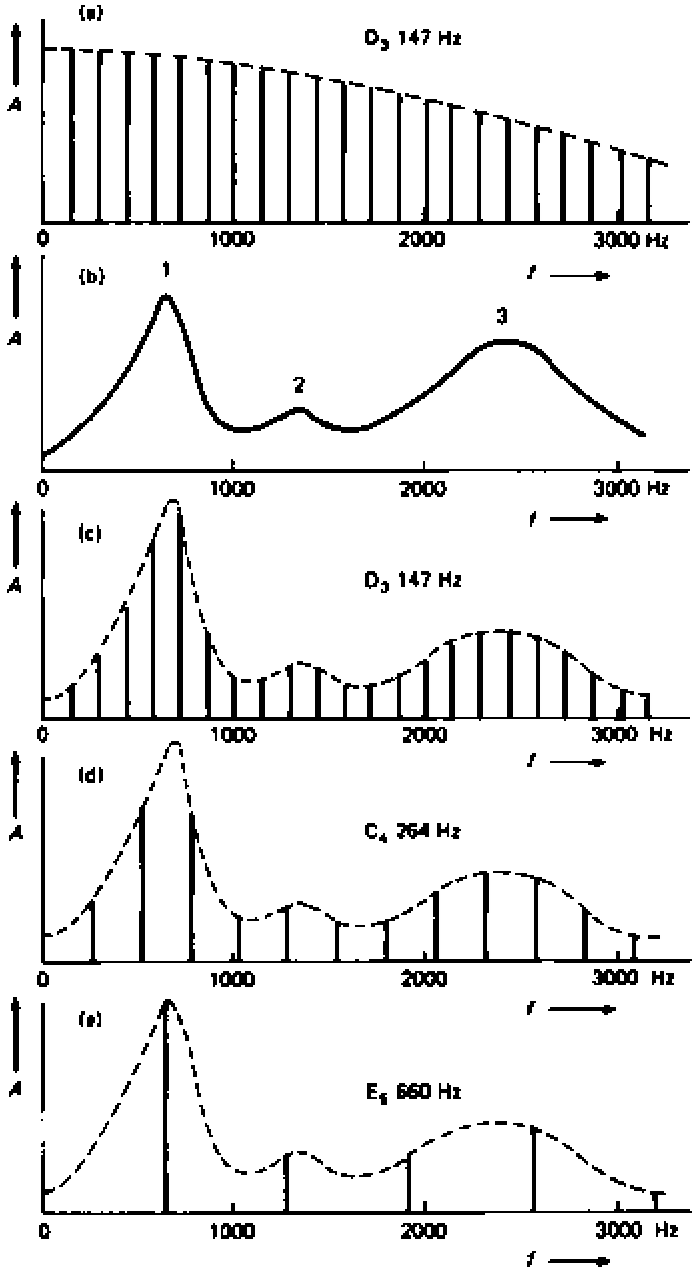


Fig 4.3

Male opera singers often develop an extra singing formant in the range 3000–3500 Hz by reshaping the larynx-pharynx transition so that the larynx cavity resonates somewhat independently of the rest of the vocal tract.The female voice generally runs 15–20 percent higher than the male, and the child's voice is higher yet. In the high ranges, the vocalized partials in the region of the formants. If these partials fall in the regions of low response, the singer's voice will be subdued.

Stringed Instruments

Stringed instruments are musical instruments that produce sound by vibrating strings. They can be categorized into two main types based on how the strings are set into motion: plucked and bowed.

Plucked String Instruments

Plucked string instruments produce sound when the strings are plucked or strummed by hand or with a plectrum. Some examples of plucked string instruments include

Guitar: The guitar is one of the most popular and versatile plucked string instruments. It typically has six strings arranged in pairs, with each pair tuned to the same pitch but one octave apart. Guitars are used in a wide range of musical genres, from classical and folk to rock and pop.

Fig 4.4

Mandolin: The mandolin is a small, pear-shaped instrument with eight strings arranged in pairs. It is commonly used in bluegrass, folk, and traditional music styles, known for its bright and distinctive sound.

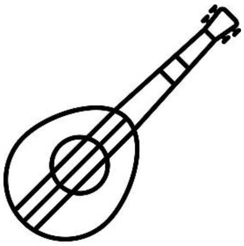


Fig 4.5

Banjo: The banjo is a plucked string instrument with a round body and a long neck. It typically has five strings and is widely used in bluegrass, country, and folk music.

Fig 4.5

Bowed String Instruments

Bowed string instruments produce sound when the strings are set into vibration by drawing a bow across them. Some examples of bowed string instruments include

Violin: The violin is a small, four-stringed instrument with a hollow wooden body and a long neck. It is known for its expressive and versatile sound and is widely used in classical music, as well as in various other genres such as jazz, folk, and contemporary music.

Viola: The viola is similar to the violin but slightly larger and with a deeper tone. It is often used as a middle voice in string ensembles and orchestras.

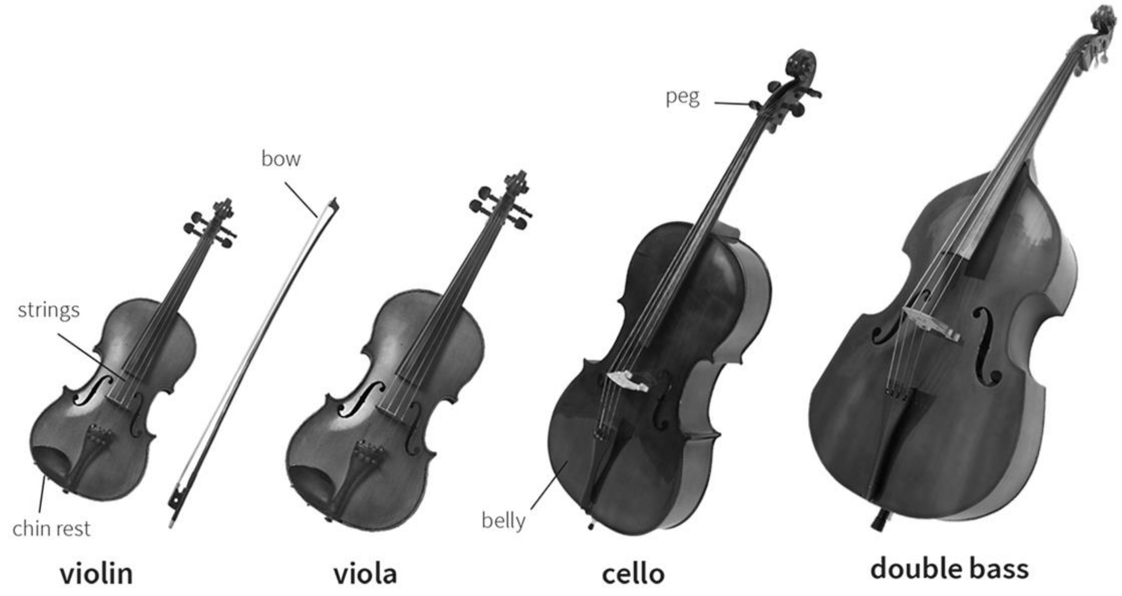


Fig 4.6

Cello: The cello, or violoncello, is a larger bowed string instrument with four strings and a deep, rich tone. It is commonly used in classical music as a solo instrument, as well as in chamber music and orchestras.

Double Bass: The double bass is the largest and lowest-pitched member of the bowed string instrument family. It typically has four strings and is used in classical, jazz, and popular music settings.

Piano:

The piano is a keyboard instrument with strings that are struck by hammers when keys are pressed. While it is not strictly a plucked or bowed string instrument, the piano is often included in discussions of stringed instruments due to the presence of strings as the primary sound-producing element.

Fig 4.7

Wind instruments

Wind instruments are musical instruments that produce sound by vibrating air. They are categorized based on the method of producing the vibrations, which can involve blowing air into a mouthpiece, causing a reed to vibrate, or using a vibrating column of air. Here are some examples of wind instruments:

Whistle: Whistles are simple wind instruments consisting of a hollow tube with a mouthpiece at one end and a small opening at the other. When air is blown into the mouthpiece, it passes over the edge of the opening, causing vibrations and producing sound. Whistles are commonly used in folk music, sports events, and as signalling devices.

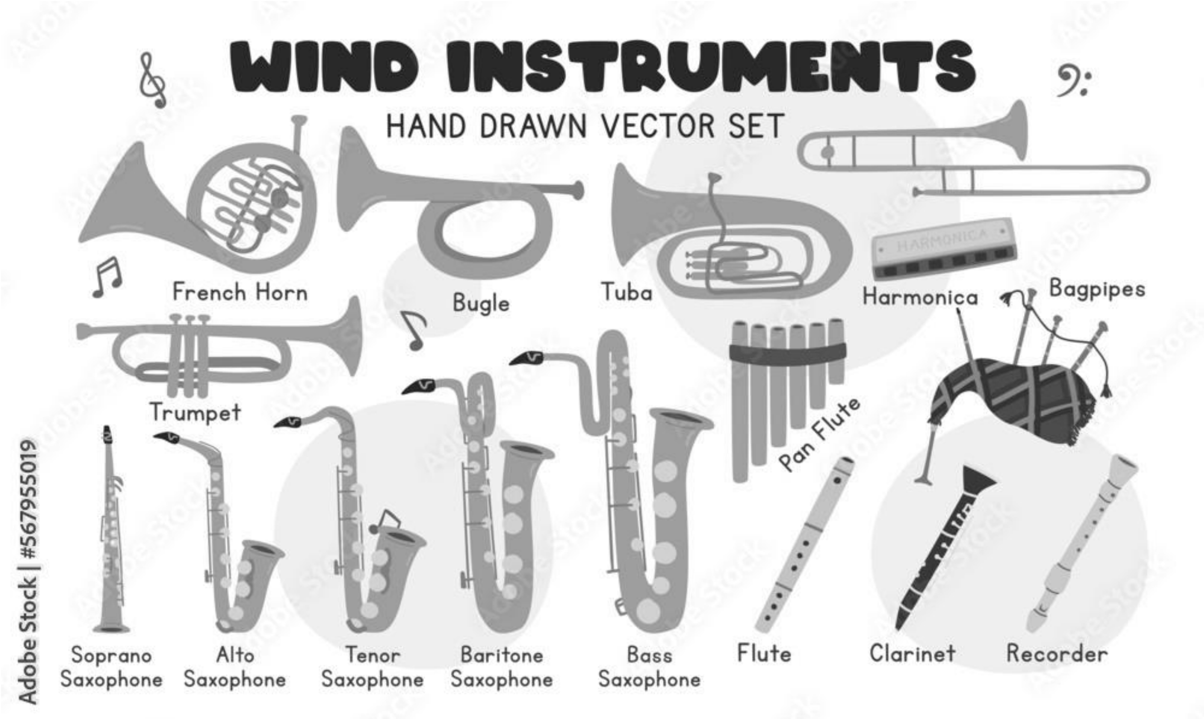


Fig 4.7

Flute: The flute is a woodwind instrument that produces sound by blowing air across a hole in the side of the instrument. The air stream causes the air inside the flute to vibrate, producing sound. Flutes can be made of various materials, including metal, wood, and plastic, and they are commonly used in classical, jazz, and folk music.

Saxophone: The saxophone is a brass instrument with a single-reed mouthpiece similar to that of a clarinet. It features a conical metal tube with keys and produces sound when air is blown into the mouthpiece, causing the reed to vibrate. Saxophones come in various sizes and are used in a wide range of musical genres, including jazz, classical, and popular music.

Pipe Organ: The pipe organ is a keyboard instrument that produces sound by passing air through pipes of different lengths and shapes. The pipes are arranged in groups called ranks and are controlled by a complex system of valves and keys. Pipe organs are commonly found in churches and concert halls and are capable of producing a wide range of dynamic and expressive sounds.

Bagpipes: Bagpipes are a class of wind instrument consisting of a bag, chanter, and drones. The bag acts as a reservoir of air, which is squeezed by the player to force air through the chanter and drones, producing sound. Bagpipes are traditionally associated with Scottish and Irish music but are also used in other cultures around the world.

Percussion instruments

Percussion instruments are musical instruments that produce sound by being struck, shaken, or scraped. They are classified into different categories based on the materials they are made of and how they produce sound. Here are some examples of percussion instruments:

Membranophones: Membranophones produce sound by striking or vibrating a stretched membrane. Examples include:

Drums: Drums consist of a hollow shell with a drumhead stretched over one or both ends. When struck with hands, sticks, or mallets, the drumhead vibrates, producing sound. Drums come in various shapes and sizes, including bass drums, snare drums, and tom-toms, and are used in a wide range of musical genres.

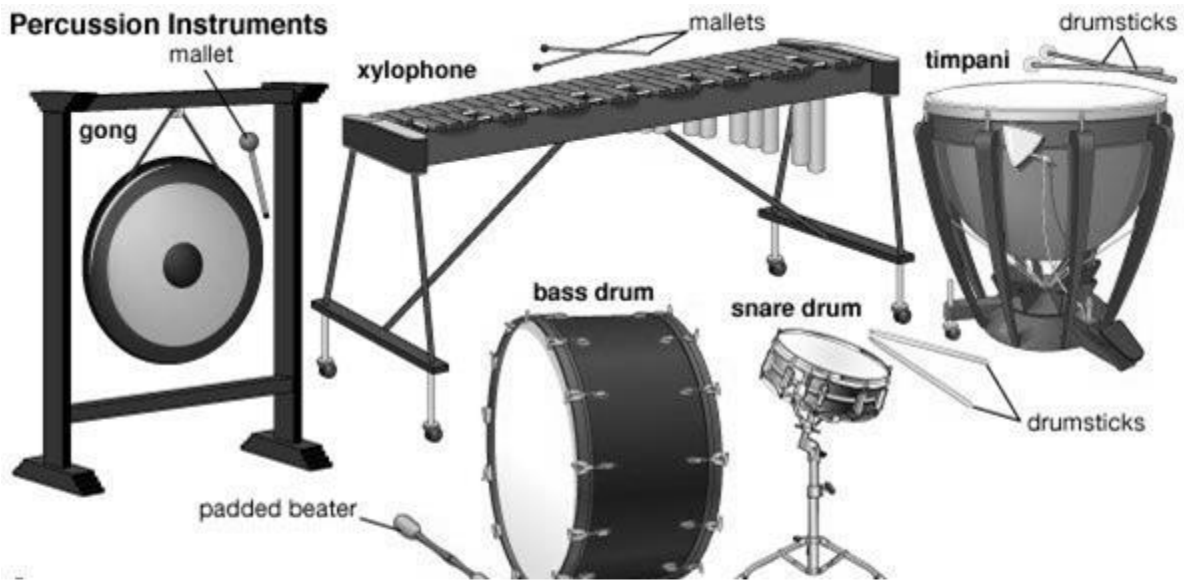


Fig 4.8

Timpani: Timpani, or kettle drums, are large, bowl-shaped drums with a calfskin or synthetic drumhead stretched over the top. They are played with mallets and are commonly used in orchestral music to provide rhythm and colour.

Idiophones: Idiophones produce sound by the vibration of the instrument itself, without the need for a membrane. Examples include:

Cymbals: Cymbals are metal discs that produce sound when struck together or with a drumstick. They come in various sizes and thicknesses and are commonly used in orchestras, bands, and various forms of popular music.

Xylophone: The xylophone consists of wooden bars of varying lengths suspended over resonators. When struck with mallets, the bars vibrate and produce sound. Xylophones are often used in orchestral and ensemble settings, as well as in marching bands and jazz.

Chordophones: Some percussion instruments produce sound by the vibration of strings stretched over a resonating body. Examples include:

Glockenspiel: The glockenspiel is similar to the xylophone but features metal bars instead of wooden ones. It is played with mallets and produces a bright, bell-like tone.

Other Percussion Instruments:

Tambourine: The tambourine consists of a circular frame with metal jingles attached to it. It is played by shaking or striking the instrument, producing a jingling sound. Tambourines are commonly used in folk, rock, and pop music.

Maracas: Maracas are percussion instruments consisting of hollow gourds filled with seeds or beads. They are played by shaking the instrument, causing the seeds or beads to strike the walls of the gourd and produce a rattling sound. Maracas are commonly used in Latin American and Caribbean music.

Electronic instruments

Electronic instruments are musical instruments that produce sound using electronic circuits or digital technology rather than traditional acoustic mechanisms. They offer a wide range of possibilities for sound creation, manipulation, and synthesis. Here are some examples of electronic instruments:

Keyboards Synthesizers

Synthesizers are electronic keyboards that generate sound using oscillators, filters, and amplifiers. They allow musicians to create and manipulate a wide variety of sounds, including traditional instrument sounds, synthesized tones, and complex textures.

Digital Pianos: Digital pianos replicate the sound and feel of acoustic pianos using electronic sensors and sound samples. They often feature weighted keys and high-quality sound engines to provide a realistic playing experience.

Electric Organs: Electric organs use electronic circuits to simulate the sound of pipe organs. They typically feature multiple presets and effects, allowing musicians to customize their sound.

Electric Guitars

Electric guitars use electromagnetic pickups to convert the vibrations of the strings into electrical signals. These signals are then amplified and processed by electronic circuits to produce the final sound. Electric guitars come in various styles and configurations, each offering its own unique tone and character.

Effects Pedals: Effects pedals are electronic devices that modify the sound of electric guitars by adding effects such as distortion, delay, reverb, and modulation. They are commonly used by guitarists to enhance their sound and create unique sonic textures.

Rhythm Pad

Electronic Drum Kits: Electronic drum kits use electronic pads or triggers to produce drum sounds. These pads are typically connected to a sound module or computer, which generates the sounds based on the player's input. Electronic drum kits offer versatility and flexibility in terms of sound customization and performance.

Sample Pads: Sample pads are electronic controllers that trigger pre-recorded sounds or samples. They are often used by percussionists and electronic musicians to incorporate loops, samples, and sound effects into their performances.

Other Electronic Instruments

Digital Audio Workstations (DAWs): DAWs are software applications used for recording, editing, and producing music on computers. They offer a wide range of virtual instruments, effects, and production tools for musicians and producers.

MIDI Controllers: MIDI controllers are electronic devices used to control MIDI-compatible instruments and software. They typically feature keys, pads, knobs, and sliders that can be mapped to different parameters for performance and production purposes.

Fig 4.9

Analog Synthesizers

Technology: Analog synthesizers generate sound using analog electronic circuits, such as oscillators, filters, and amplifiers. These circuits manipulate electrical voltages to create and shape sound waves in real-time.

Sound Generation: Analog synthesizers produce continuous waveforms that closely resemble traditional acoustic sounds, such as sine, sawtooth, square, and triangle waves. They can create a wide range of timbres and textures through voltage-controlled modulation of parameters like frequency, amplitude, and filter cutoff.

Characteristics: Analog synthesizers are known for their warm, rich, and often gritty sound quality. They are prized for their organic and unpredictable nature, as the analog circuits can produce subtle variations and imperfections that add character to the sound.

Controls: Analog synthesizers typically feature physical knobs, sliders, and switches for hands-on control over sound parameters. This tactile interface allows musicians to intuitively shape and manipulate the sound in real-time.

Examples: Classic analog synthesizers include the Minimoog, Roland Jupiter-8, ARP Odyssey, and Moog Modular synthesizers. While vintage analog synthesizers are highly sought after, modern analog synthesizers continue to be developed and produced by various manufacturers.

Digital Synthesizers

Technology: Digital synthesizers use digital signal processing (DSP) technology to generate and manipulate sound. They employ algorithms and mathematical calculations to simulate the behaviour of analog circuits and generate complex waveforms and timbres.

Sound Generation: Digital synthesizers can produce a wide variety of sounds, from realistic instrument emulations to futuristic and abstract textures. They often feature sample-based synthesis, wavetable synthesis, FM synthesis, and physical modeling synthesis techniques.

Characteristics: Digital synthesizers offer precise and consistent sound generation, with the ability to recall and store presets digitally. They can produce clean, precise, and highly detailed sounds, but some musicians argue that they lack the warmth and character of analog synthesizers.

Controls: Digital synthesizers typically feature digital interfaces, such as LCD screens, buttons, and menus, for parameter adjustment and preset management. Some digital synthesizers also offer MIDI and USB connectivity for integration with computers and external controllers.

Examples: Popular digital synthesizers include the Yamaha DX7, Roland D-50, Korg M1, and Native Instruments Massive. Modern digital synthesizers continue to push the boundaries of sound synthesis with advanced DSP algorithms and software integration.

Computer-generated music

Computer-generated music refers to music that is composed, performed, or manipulated using computer software and digital technology. With the advancement of digital audio technology and music software, computers have become powerful tools for creating, recording, editing, and producing music across various genres and styles.

Computer-generated music encompasses a wide range of techniques and approaches, including:

Algorithmic Composition: Algorithms are used to generate musical sequences, melodies, harmonies, and rhythms. These algorithms can be based on mathematical principles, rules, randomness, or machine learning algorithms.

Digital Audio Workstations (DAWs): DAW software allows musicians and producers to record, edit, arrange, and mix music using virtual instruments, effects, and audio samples. DAWs provide a comprehensive environment for music production and composition.

Virtual Instruments: Virtual instruments are software-based emulations of traditional acoustic instruments, synthesizers, and samplers. They allow musicians to play and manipulate realistic or synthesized sounds using MIDI controllers or computer keyboards.

Synthesis Techniques: Various synthesis techniques, such as subtractive synthesis, additive synthesis, FM synthesis, and wavetable synthesis, are used to create and shape sounds digitally. These techniques offer a wide range of sonic possibilities for creating original and expressive music.

Generative Music: Generative music systems generate music autonomously based on predefined rules, algorithms, or input parameters. These systems can create endless variations of musical compositions in real-time, providing inspiration and new ideas for musicians and composers.

UNIT - V

MUSICAL TONE

Edison phonograph - cylinder and disk records

The original basic technique of sound recording involved cutting a spiral groove into a disc such that the sound vibrations were transformed into horizontal undulations of the groove. See Figure 5.1. Playback of early discs was accomplished by allowing the record to turn at the proper speed while a needle attempted to follow the undulations of the groove. See Figure 5.2. The needle was attached to a lever arm that caused a flexible diaphragm to vibrate. This acted as the driver of a horn, allowing the vibrations to be transferred to the surrounding air as sound. This all-mechanical system has been modified over the years to the modern turntable and cartridge pickup.. The record is placed on a heavy metal turntable that turns at a steady angular speed of 33i rpm. A diamond-tipped stylus rests in the record groove and follows the undulations

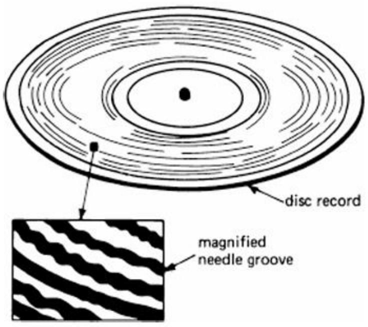


Fig 5.1

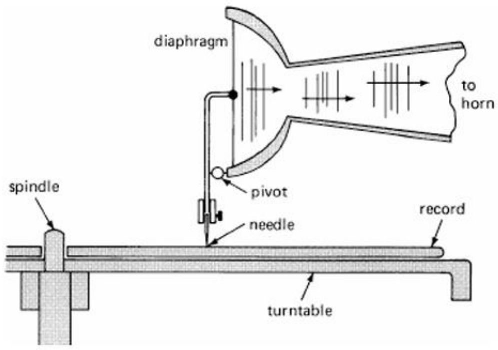


Fig 5.2

The cartridge that holds the stylus is capable of turning the vibrations into electric signals. Of the many different ways in which this is done (moving magnet, moving coil, piezoelectric effect, etc.), the most common is the moving magnet method. The end of the stylus is attached to a small magnet that moves back and forth through a small coil in the cartridge, generating electric current in the coil. This low-level current is then routed to the preamplifier for amplification, using well-shielded wires to avoid noise pickup from the surroundings. The cartridge is mounted at the end of a tone arm, which is pivoted at the other end, allowing for both vertical and horizontal motion. Ideally, the axis of the cartridge should be parallel to the grooves across the entire playing track. Deviations from this alignment, called tracking error, can be minimized by proper tone arm length, placement of the pivot, angling of the cartridge, and other more elaborate additions and modifications. The stylus is generally tipped with diamond to ensure a long life, and it is shaped according to various formulas to follow the high-frequency track variations. The weight of the stylus on the record is usually adjustable and should not be too heavy if wear is to be minimized. However, if the weight is too light, the stylus will be thrown about, causing shatter in the reproduced sound and chipping of the sides of the groove, causing scratch.

Magnetic wire and tape recorders

The most commonly used recording medium today is magnetic tape. This is an outgrowth of the wire recorders developed from 1920 to 1950. Relying on the magnetic properties of iron, it was found that sound can be recorded as varying states of magnetization of atoms or clusters of atoms (domains) of iron in the wire, each acting as a small magnet with north-south pole alignment. These microscopic magnets are oriented back and forth in larger regions in accordance with the sound wave. By the 1950s, the iron wire was replaced by plastic tape with a coating of iron oxide and, more recently, with a coating of chromium oxide. See Figure 5.3. Since magnetic tape can be made in any width, several parallel tracks can be recorded at the same time, making it simple to use for stereo or quadraphonic recording.

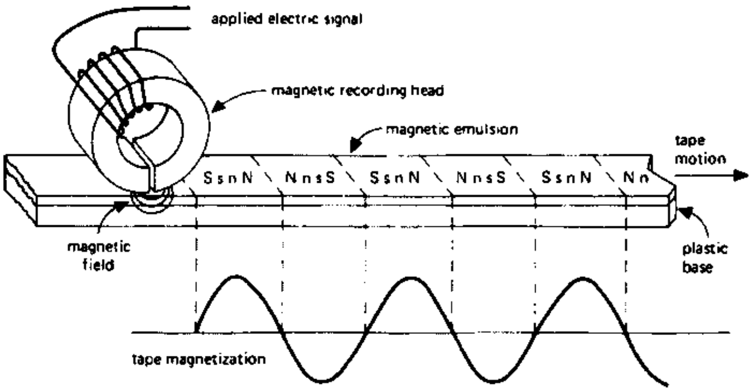


Fig 5.3

A typical high-fidelity tape deck, designed to be used in conjunction with an amplifier, is shown in Figure 5.4. The tape is fed from one open reel to another at constant speed by the capstan and pinch roller. Three magnetic heads are positioned close to the tape. The first is an erase head E, consisting of an electromagnet driven by a high-frequency electric current. This causes ultrasonic magnetic variations to occur, which realign the magnetic domains on the tape, thereby erasing the previous recording. The recording head R consists of another electromagnet, which applies the desired signal to the tape. The third head is for playback P, allowing the passing tape to induce electric signals in the small coil in the head, much as an electric generator generates electric current. This current is amplified and routed to the preamplifier for further amplification. The playback head also makes it possible to monitor a recording by listening to the recorded signal immediately after it is recorded

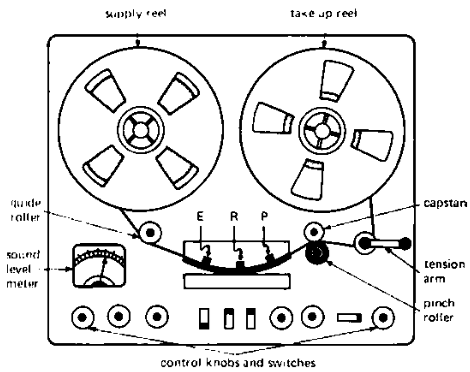


Fig 5.4

Many tape decks have replaced the open reels with small packaged reels or cassettes. For use in the automobile, endless-loop cartridges are commonly used. Tapes generally consist of the magnetic oxide on a base of cellulose triacetate or polyester. Unfortunately, the cellulose triacetate tends to age if not properly cared for. The polyester tape does not age, but it stretches easily. For best fidelity in the highs, the tape should be run at relatively high speed. This means or $7\frac{1}{2}$ or 15 inches per second for music, although a lower speed of $3\frac{1}{2}$ inches per second is used for long-playing tapes. The higher speed causes the tape to be used more rapidly. Thinner tape will allow more tape to be wound on the reel, but thin tape is more susceptible to stretch, as well as print-through —that is, the tendency for magnetization of one layer to induce magnetization of the adjacent layer. Another problem with magnetic tape is that unevenness in motor and roller speed, as well as tape stretch, can cause pitch variations called drift (long term), wow (moderate), and flutter (rapid).

Digital recording

Digital recording refers to the process of capturing, storing, and reproducing audio or video data in a digital format. Unlike analog recording, which involves capturing and storing information in continuous, analog signals, digital recording converts the analog signals into discrete, digital data that can be stored, manipulated, and reproduced with high fidelity. Here are some key points about digital recording:

In digital recording, analog audio or video signals are converted into digital data using analog-to-digital converters (ADCs). These converters sample the analog signals at regular intervals and measure their amplitude, converting them into binary data (0s and 1s) that represent the audio or video waveform.

The digital data captured by the ADCs is stored in digital files on various types of storage media, such as hard drives, solid-state drives (SSDs), optical discs (CDs, DVDs, Blu-ray discs), or flash memory cards. These digital files can be easily copied, transferred, and archived without degradation of quality.

Examples

CD/DVD comes under the category of a permanent storage device, used in computers. CD/DVD uses the principle of writing the datas using a high power laser beam and reading the datas using a low power laser beam, by forming pits (0's) and lands (1's) over the CD.

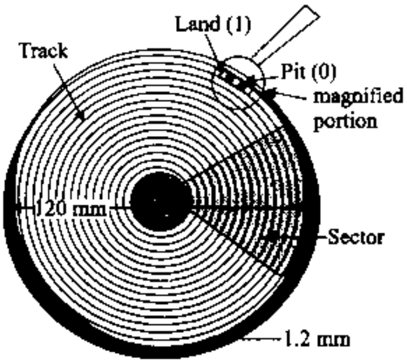


Fig 5.5

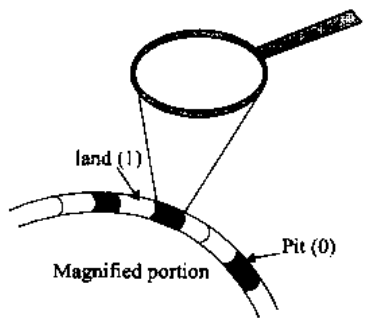


Fig 5.6

CD/DVD is a plastic (or) photo-polymer disk madeup of poly-carbonate, grown over the substrate. It has a diameter of 120 mm and thickness of 1.2 mm, with a circular hole of diameter 15 mm as shown in Fig. 5.5. In this disk the datas can be stored in tracks and sectors, from 46 mm to 116 mm of diameter. The magnified portion of data storage is indicated in Fig. 5.6.

Since we are going to use laser light for writing and reading the data, the reflectivity at the top surface should be increased. Therefore a layer of silver (or) aluminium is deposited as vapour over the poly-carbonate surface as shown in Fig. 5.7. A protective layer protects the CD from dust and moisture.

The storage capacity in CD's depends on the number of sectors and the number of bytes (8 Bits = 1 Byte) per sector. It is given by the formula, i.e.,

Storage capacity = Number of sectors × Number of bytes per sector

Fig. 5.7 shows the system used for recording/writing the datas over the CD and also for retrieving/reading the datas from it.

Initially all over the CD we will have lands (1's) of highly reflectivity. Now to change it as 0's and 1's a high power laser is directed towards the surface of the CD through the condensing lens C1, beam splitter (Bs) and condensing lens C2 as shown in Fig. 5.7.

This high power laser beam burns the surface of the CD and creates a small hole called pit, corresponding to 0's of 0.8 μm diameter.By similar method wherever we want 0's, laser should be focussed and pits are created and the remaining area will be as such as lands i.e. 1's.

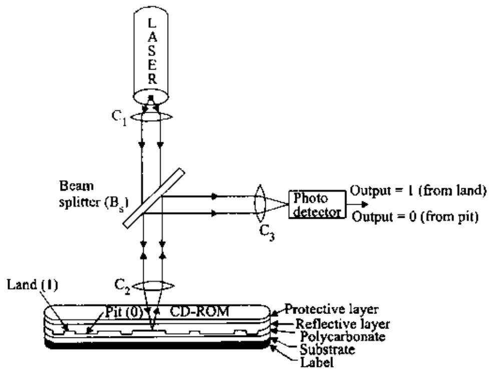


Fig 5.7

Therefore, the pit (0's) represents the burnt area and the land (1's) represents the unburnt area in CD. Thus, the datas are digitally recorded/written over the CD.

Analog transducer

The main function of the Analog transducer is to change the quantity of input to a constant function. The best examples of the analog transducer are LVDT, thermocouple, strain gauge & thermistor. Digital transducers are used to change the quantity of an input to a digital signal that works on low or high power.

Strain Gauge Transducer

The main function of the strain gauge transducer is to convert physical quantities electrically. They function through changing physical quantities into mechanical pressure within a component known as a sensing element & after that convert the stress electrically using a strain gauge.

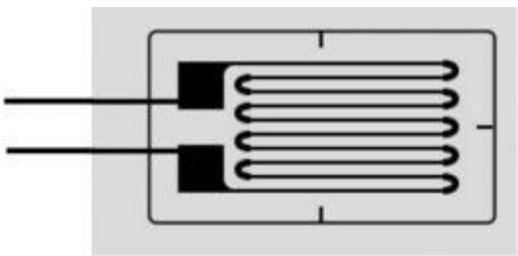


Fig 5.8

The structure of the sensing element, as well as the strain gauge, is designed optimally to give handling & superior accuracy products. These transducers are classified generally based on their application to construction/civil engineering types or general types. Some of the general type transducers are utilized in the construction or civil engineering field. The types of strain gauge transducers are Wire Strain Gauge, Foil Strain Gauge & Semiconductor Strain Gauge.

Condenser Microphones

Condenser microphones are known for their high sensitivity and wide frequency response. They are capable of capturing subtle nuances and details in sound, making them ideal for capturing vocals, acoustic instruments, and other critical audio sources. Most condenser microphones require external power, typically provided by phantom power, to operate. This polarizes the microphone's capacitor and amplifies the signal, allowing for high-quality audio capture. Condenser microphones offer precise and transparent sound reproduction, with low distortion and noise levels. They excel in capturing the full dynamic range of complex sound fields, providing accurate recordings with exceptional fidelity.

Dynamic Microphones

Microphones are rugged and durable, making them suitable for use in demanding environments such as live performances and outdoor settings. They can handle high sound pressure levels (SPL) without distortion, making them ideal for close-miking loud sound sources. Dynamic microphones have lower sensitivity compared to condenser microphones but are more resistant to feedback and handling noise. They are often used for capturing loud instruments, amplifiers, and drums in live sound and recording applications. Dynamic microphones typically have a narrower frequency response compared to condenser microphones, but they excel in rejecting off-axis sound and background noise, making them suitable for capturing sound in noisy environments.

Loudspeakers complex sound fields

Loudspeakers are transducers that convert electrical signals into sound waves. They play a critical role in reproducing complex sound fields accurately and faithfully to the original source. Loudspeakers vary in their frequency response and dispersion characteristics, which affect how they reproduce sound within a given space. Proper speaker placement and configuration are essential for achieving even coverage and optimal sound quality in complex environments. In live sound reinforcement settings, loudspeakers can contribute to feedback issues if not properly managed. Techniques such as speaker placement, equalization, and feedback suppression systems are employed to minimize feedback and maintain a clear and balanced sound.

In complex sound fields, the choice and placement of microphones and loudspeakers play a crucial role in capturing and reproducing sound accurately. A combination of condenser and dynamic microphones, along with well-designed loudspeaker systems, allows for precise and immersive audio experiences in various applications. Proper setup, calibration, and optimization are essential for achieving optimal performance and fidelity in complex sound environments.

Near and far fields of acoustics

The near and far fields are two distinct regions surrounding an acoustic source where sound behaves differently. Understanding these fields is crucial in various applications, including audio engineering, acoustics, and sound reinforcement. Here's a short note on each:

Near Field:

- ❖ The near field, also known as the reactive field or Fresnel region, is the region close to the acoustic source where sound waves are still developing and have not yet fully dispersed.
- ❖ In the near field, sound waves are characterized by complex interference patterns and significant changes in amplitude, phase, and directionality.
- ❖ Sound pressure levels in the near field may vary significantly depending on the distance from the source and the frequency of the sound.
- ❖ Near-field monitoring is commonly used in audio engineering and recording to assess the direct sound from studio monitors or speakers before it interacts with room acoustics.

Far Field:

- ❖ The far field, also known as the radiation field or Fraunhofer region, is the region farther away from the acoustic source where sound waves have fully developed and exhibit more predictable behaviour.
- ❖ In the far field, sound waves propagate as spherical waves, spreading outward uniformly in all directions.
- ❖ Sound pressure levels in the far field decrease with distance according to the inverse square law, where the intensity of sound decreases by 6 dB for every doubling of distance from the source.
- ❖ In sound reinforcement systems, loudspeakers are typically positioned in the far field to provide even coverage and uniform sound distribution throughout a venue.

Spectral analysis techniques

Spectral analysis techniques are fundamental tools used in various fields such as signal processing, audio engineering, telecommunications, and scientific research to analyze the frequency content of signals. Two primary methods for spectral analysis are continuous and discrete Fourier transforms, along with digital signal processing techniques. Here's a brief overview of each

Continuous Fourier Transform (CFT)

- ❖ The continuous Fourier transform is a mathematical tool used to decompose a continuous-time signal into its frequency components.
- ❖ It represents the signal in the frequency domain, showing the amplitude and phase of each frequency component present in the signal.
- ❖ The mathematical expression for the continuous Fourier transform involves integrating the signal over all time.
- ❖ While the continuous Fourier transform provides an exact representation of the signal's frequency content, it can only be applied to signals with continuous and infinite time domains.

Discrete Fourier Transform (DFT)

- ❖ The discrete Fourier transform is a digital implementation of the Fourier transform, used to analyze discrete-time signals sampled at discrete intervals.
- ❖ It converts a sequence of N time-domain samples into a corresponding sequence of N frequency-domain samples.
- ❖ The most common algorithm for computing the DFT is the Fast Fourier Transform (FFT), which significantly reduces the computational complexity of the calculation.

- The DFT provides a discrete spectrum of frequency components, which can be used for spectral analysis, filtering, and modulation.

Digital Signal Processing (DSP)

- Digital signal processing involves the manipulation and analysis of digital signals using algorithms implemented on digital computers or specialized DSP hardware.
- DSP techniques include filtering, convolution, correlation, modulation, and spectral analysis, among others.
- Spectral analysis using DSP involves applying various algorithms and techniques to analyze the frequency content of digital signals.
- Common DSP-based spectral analysis methods include windowing, power spectral density estimation, periodogram analysis, and spectrogram analysis.

Digital filtering

Digital filtering can be implemented either in hardware or software; in the first case, the numerical processor is either a special-purpose chip or it is assembled out of a set of digital integrated circuits which provide the essential building blocks of a digital filtering operation – storage, delay, addition/subtraction and multiplication by constants. On the other hand, a general-purpose mini-or micro-computer can also be programmed as a digital filter, in which case the numerical processor is the computer's CPU, GPU and memory. Filtering may be applied to signals which are then transformed back to the analogue domain using a digital to analogue convertor or worked with entirely in the digital domain.

The numerical processor can easily be (re-)programmed to implement a number of different filters. The accuracy of a digital filter is dependent only on the round-off error in the arithmetic. This has two advantages:

- The accuracy is predictable and hence the performance of the digital signal processing algorithm is known a priori.
- The round-off error can be minimized with appropriate design techniques and hence digital filters can meet very tight specifications on magnitude and phase characteristics (which would be almost impossible to achieve with analogue filters because of component tolerances and circuit noise).
- The widespread use of mini- and micro-computers in engineering has greatly increased the number of digital signals recorded and processed. Power supply and temperature variations have no effect on a programme stored in a computer.
- Digital circuits have a much higher noise immunity than analogue circuit

Specification of recording studios

Recording studios vary widely in terms of their specifications, equipment, and layout, depending on factors such as budget, intended use, and desired quality of recordings. However, there are some common specifications and features that are typically found in professional recording studios. Here are some key specifications of recording studios:

Acoustic Treatment:

Proper acoustic treatment is essential to minimize reflections, reverberations, and external noise, ensuring a controlled and accurate listening environment.

Acoustic treatment may include soundproofing, diffusers, absorbers, bass traps, and acoustic panels strategically placed throughout the studio.

Room Size and Shape:

Recording studios come in various sizes and shapes, ranging from small home studios to large commercial facilities.

The size and shape of the studio can affect the acoustics, resonance, and sound diffusion within the space.

Isolation:

Recording studios often feature isolated rooms and booths to prevent sound leakage and interference between recording sessions.

Isolation techniques may include double walls, floating floors, and soundproof doors and windows.

Monitoring System:

High-quality monitoring systems, including studio monitors (speakers) and headphones, are essential for accurate playback and mixing of audio recordings.

Studio monitors should have a flat frequency response and high resolution to provide an accurate representation of the recorded audio.

Recording Equipment:

Recording studios are equipped with professional-grade audio interfaces, preamplifiers, mixing consoles, and digital audio workstations (DAWs) for capturing and processing audio signals.

Microphones, including condenser, dynamic, and ribbon mics, are used to capture vocals, instruments, and other sound sources.

Outboard gear such as compressors, equalizers, and reverbs may be used for signal processing and effects.

Instrumentation:

Recording studios may have a variety of musical instruments and equipment available for use, including keyboards, synthesizers, guitars, basses, drums, and percussion instruments.

MIDI controllers and samplers may also be available for creating electronic music and integrating virtual instruments.

Control Room and Live Room:

Professional recording studios typically feature separate control rooms and live rooms to isolate the recording and mixing processes from the performance space.

The control room houses the recording equipment, mixing console, and monitoring system, while the live room is used for recording performances.

Power and Electrical Requirements:

Recording studios require stable power sources and adequate electrical wiring to support the operation of audio equipment and prevent ground loops and interference.

Environmental Factors:

Temperature and humidity control are important considerations to maintain stable conditions for sensitive audio equipment and instruments.

Overall, the specifications of recording studios can vary widely depending on the intended use and budget. Professional recording studios strive to create a comfortable, acoustically treated environment with high-quality equipment to facilitate the recording, mixing, and production of audio recordings.