Audio Visual Production
Block – I: Audio Production

Odisha State Open University
Audio Visual Production

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## Contents

<table>
<thead>
<tr>
<th>Course Overview</th>
<th>5</th>
</tr>
</thead>
<tbody>
<tr>
<td>Welcome to Audio-Production</td>
<td>5</td>
</tr>
<tr>
<td>Concept of Sound</td>
<td>5</td>
</tr>
<tr>
<td>Audio Equipments</td>
<td>5</td>
</tr>
<tr>
<td>Sound Recording</td>
<td>5</td>
</tr>
<tr>
<td>Audio Editing</td>
<td>6</td>
</tr>
<tr>
<td>Course outcomes</td>
<td>6</td>
</tr>
<tr>
<td>Timeframe</td>
<td>7</td>
</tr>
<tr>
<td>Study skills</td>
<td>7</td>
</tr>
<tr>
<td>Need help?</td>
<td>8</td>
</tr>
<tr>
<td>Assignments</td>
<td>8</td>
</tr>
<tr>
<td>Assessments</td>
<td>8</td>
</tr>
<tr>
<td>Video Resources</td>
<td>9</td>
</tr>
</tbody>
</table>

### Getting around this Course material

<table>
<thead>
<tr>
<th>10</th>
</tr>
</thead>
<tbody>
<tr>
<td>Margin icons</td>
</tr>
</tbody>
</table>

### Unit 1

<table>
<thead>
<tr>
<th>11</th>
</tr>
</thead>
<tbody>
<tr>
<td>Concept of Sound</td>
</tr>
<tr>
<td>Introduction</td>
</tr>
<tr>
<td>Outcomes</td>
</tr>
<tr>
<td>Terminology</td>
</tr>
<tr>
<td>Concept of Sound</td>
</tr>
<tr>
<td>Types of Sound</td>
</tr>
<tr>
<td>Characteristics of a Wave Form</td>
</tr>
<tr>
<td>Velocity</td>
</tr>
<tr>
<td>Phase</td>
</tr>
<tr>
<td>Frequency Response</td>
</tr>
<tr>
<td>Pitch</td>
</tr>
<tr>
<td>Unit Summary</td>
</tr>
<tr>
<td>Assignments</td>
</tr>
<tr>
<td>Resources</td>
</tr>
</tbody>
</table>

### Unit 2

<table>
<thead>
<tr>
<th>21</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Equipments</td>
</tr>
<tr>
<td>Introduction</td>
</tr>
<tr>
<td>Outcomes</td>
</tr>
<tr>
<td>Terminology</td>
</tr>
</tbody>
</table>
Unit 3

Sound Recording .................................................................................................................. 42
Introduction .......................................................................................................................... 42
Outcomes .............................................................................................................................. 42
Terminology ........................................................................................................................ 42
Recording Process ............................................................................................................... 43
Magnetic Tape ...................................................................................................................... 44
The Function of Magnetic tape Heads ................................................................................ 45
The Signal flow and recording Process .............................................................................. 45
Signal Flow in a recording setup ......................................................................................... 46
Digital Recording ................................................................................................................ 47
Sampling: ............................................................................................................................. 48
Quantization: ......................................................................................................................... 48
The Digital Recording process ............................................................................................ 48
Digital Audio tape (DAT) System ....................................................................................... 49
Mixing or Mastering: ........................................................................................................... 50
Sound processing ................................................................................................................ 51
Recording level ..................................................................................................................... 52
Digital Audio Workstation ................................................................................................. 53
Amplitude Modulation (AM) ............................................................................................... 53
Frequency Modulation (FM) ............................................................................................... 54
Unit summary ....................................................................................................................... 56
Assessment .......................................................................................................................... 56
Resources .............................................................................................................................. 56

Unit 4

Audio Editing ........................................................................................................................ 57
Introduction .......................................................................................................................... 57
Outcomes ............................................................................................................................. 57
Terminology ........................................................................................................................ 57

Contents
Course Overview

Welcome to Audio-Production

In this block, you will learn about two parts of film making like hypothesis and practical. You will find out about the means utilized as a part of making preparations for different mediums like T.V., Film and so on. Other than that you will learn the skill to write a screenplay, shooting script, storyboard, and making of a budget, designs a production and so on.

Concept of Sound

Sound plays a greater role in the field of communication, entertainment, understanding, sharing of the information. It is one of the most important fields for electronic media like Television, Radio, Cinema and Public Broadcasting. The world of modern music is becoming so popular that the audio technology has become simpler.

Audio Equipments

Audio equipments are used for different purpose and for various applications. Audio engineer uses outdoor portable recorder for an outdoor recording and other equipments for studio recording.

Sound Recording

Sound recording is a process that involves both the skill of art and science. The purpose of recording is to restore the information for future use and it may be stored for years. Sound recording process takes multiple steps.
Audio Editing

Audio Editing is done in many stages of audio production. In the post production many things are to be done on the mixing console desk. Noise is an important factor to be remembered and taken care of during track laying and editing. Noise is removed in this stage of audio production.

This video will provide a brief overview of this course.

<table>
<thead>
<tr>
<th>Topic</th>
<th>YouTube link</th>
<th>QR Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video 1 – Audio fundamental and its equipments</td>
<td><a href="https://youtu.be/AqN6_SFoAaw">https://youtu.be/AqN6_SFoAaw</a></td>
<td><img src="https://youtu.be/AqN6_SFoAaw" alt="QR Code" /></td>
</tr>
<tr>
<td>Video 2 – Audio editing process using Audacity</td>
<td><a href="https://youtu.be/aXspSADKwsY">https://youtu.be/aXspSADKwsY</a></td>
<td><img src="https://youtu/be/aXspSADKwsY" alt="QR Code" /></td>
</tr>
</tbody>
</table>

Course outcomes

Upon completion of Audio-Production you will be able to:

- Describe the Concept of Sound and its phenomena.
- Name the audio equipments that are used in audio productions.
- Differentiate between the types of Microphones.
- Explain the Process of Sound Recording
- Describe Voice Dubbing process.
Timeframe

This course will be completed within “2” classes.

This course is of “1” credits.

4 Hours of study time is required to complete this unit.

Study skills

Learning about the audio visual productions are a part of Multimedia Coaching. But, when it comes to creation of an output, there is no specific ABC formula for doing so. It is just like a painting in which the artists choose the colour and proportion according to the need, not as per a written plan.

In creating outputs for TV, Films etc. we have to utilize the software tools from one corner to another randomly. So for every project, the formula is different. There is no fixed recipe for all kind of output. So, the more and more you learn about the options, more variety you will get. Each and every option is explained step by step in the course material.

Apart from this course material, the learner has to adopt the tendency of learning from multiple sources i.e.,

- Internet tutorials
- Video tutorials on youtube
- Collaboration with people working in the industry etc.

Only classroom study will not make you a professional. You have to be active to grab the opportunity of learning wherever you get a chance.
Need help?

In case of any help needed you can browse the internet sites like youtube.com for video tutorials about the subject.

Assignments

There will be some assignments at the end of each unit.

These assignments are mostly practical based and should be submitted in CD or DVD. Theoretical assignments are to be submitted neatly written on A4 size sheet.

All assignments will be submitted to respective study centre of Odisha State Open University or as directed by Co-ordinator.

All assignment should be unit wise on separate CD/DVDs clearly mentioning course title and unit on Top. Theoretical Assignment will be neatly filed or spiral bind with cover clearly mentioning necessary information of course.

Assessments

There will be “1” assessment for each unit.

All practical assessment will be submitted to OSOU.

Assessment will take place once at the end of each unit.

Learner will be allowed to complete the assessment within stipulated time frame given by the university.
Video Resources

This study material comes with additional online resources in the form of videos. As videos put in human element to e-learning at the same time demonstrating the concepts visually also improves the overall learning experience.

You can download any QR code reader from Google Play to view the videos embedded in the course or type the URL on a web browser.
Getting around this Course material

Margin icons

While working through this Course material you will notice the frequent use of margin icons. These icons serve to “signpost” a particular piece of text, a new task or change in activity; they have been included to help you to find your way around this Course material.

A complete icon set is shown below. We suggest that you familiarize yourself with the icons and their meaning before starting your study.

<table>
<thead>
<tr>
<th>Activity</th>
<th>Assessment</th>
<th>Assignment</th>
<th>Case study</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discussion</td>
<td>Group activity</td>
<td>Help</td>
<td>Note it!</td>
</tr>
<tr>
<td>Outcomes</td>
<td>Reading</td>
<td>Reflection</td>
<td>Study skills</td>
</tr>
<tr>
<td>Summary</td>
<td>Terminology</td>
<td>Time</td>
<td>Tip</td>
</tr>
</tbody>
</table>
Unit- 1

Concept of Sound

Introduction

Sound is an important concept in our life. It plays a greater role in the field of communication, entertainment, understanding, sharing of the information. It is one of the most important fields for electronic media like Television, Radio, Cinema and Public Broadcasting. The world of modern music is becoming so popular that the audio technology has become simpler. The sound production abides with different aspects, this is a field where both art and science work together and creates the wonders in the form of music that has a healing power. Imagination, artistic support and technical expertise can give birth to a creative production. Sound recording technology changes with the time.

Now we are living in a digital world, where most of us are spending so much time with the social media like facebook, twitter, whatsapp and dealing with voice calling, voice sms, video calling, chatting etc. Knowingly or unknowingly you are using a camera, video recorder, audio recorder, transmitter and a receiver in the form of a smart phone. The reason is the advancement in the technology, vis-à-vis the convergence of audio-video technology. If you want to pursue a career in the media industry, then you have to learn the expertise of media technology. It may be an audio recording, a video shooting or a digital promotion. Sound technology is one of the widely used in the electronic media and film production.
Unit-1  Concept of Sound

Outcomes

Upon completion of this unit you will be able to:

- Describe the Concept of Sound and its phenomena
- Differentiate between different types of Sound.
- Describe the construction of human ear and its function.
- Explain the characteristics of Wave Form

Terminology

Wave: A Graphic representation that describe sound.
Loudness: It is the Gain of sound pressure level.
Propagation: Travel of sound in air medium.
Decibel: Unit of Loudness.
Amplitude: Distance above or below the centre line of wave.
Stimulus: The vibrating sensation.

Concept of Sound

In our surrounding, many physical events occur. You can see them through the eyes and hear the sound by ears. Sound is a phenomenon that describes the brain’s perception and interpretation of a physical stimulus that arrives to the ears. It is a both physical and psychological phenomenon. Sound is generated from a source attached to the atmosphere which travels through wave propagation in the air (Figure 01). Not only in the air, it can also travel through the water and metal also. It takes place through periodic vibration in the air molecules and reaches to our
ear. When it passes into our ear through the external object many things tend to happen and our brains responds to it accordingly.

Now let us learn how the ear acts. The Sound arrives at our ear in the form of a periodic vibration in the atmospheric pressure known as sound pressure level (SPL). Sound Pressure Level is the acoustic pressure that is built up within a specific atmospheric area. The greater the sound pressure level, the louder the sound. Sound pressure level can be measured in decibel (dB). Our ear can receive the sound vibrations from 360 Degree angle.

![Wave propagation](https://commons.wikimedia.org/wiki/File:CPT-sound-physical-manifestation.svg)

**Title** - Fig-1 Wave propagation  
**Attribution** - Pluke  
**Link** - [https://commons.wikimedia.org/wiki/File:CPT-sound-physical-manifestation.svg](https://commons.wikimedia.org/wiki/File:CPT-sound-physical-manifestation.svg)

The vibrating waves travel through the air, medium are collected at the outer ear and then passes through the aural canal of inner ear and hits the stretched drum like membrane called eardrum which is attached to the ear drum, a snail like organ called cochlea that contains so many tinny hairs. The sound waves after reaching the inner ear are then changed into mechanical vibrations, which are transferred to the inner ear of three bones; these three bones are the hammer, anvil and stirrup (Figure 02). These bones act like an amplifier and a limiter, means they help the weak signals to boost and limit the louder sound too (we will discuss on amplifier and limiter in the next unit). The vibrations are then applied to the cochlea-a tubular, snail like organ that
contains two fluid filled chambers. Within these chambers there are tiny hair receptors. These tiny hairs can respond to different frequencies and are lined in a row along the length of the cochlea. Then the mechanical signals are sent to the brain and this neural stimulation gives us the sense of hearing.

A convenient pressure level that produces the phenomenon of hearing is called threshold of hearing. The minimum Sound Pressure Level (SPL) that is required for hearing in most of the people is equal to 0.0002 microbar. One microbar is equal to one million atmospheric pressure. The SPL that causes discomfort in a listener 50 percent of the time is called the threshold of feeling. It occurs at a level of about 228 decibel. The SPL that causes pain in the listener 50 percent of the time is called threshold of pain and it occurs at a level of 140 dB.

**Types of Sound**

We are dealing with so many types of sounds. Some are artificial and some are natural. The sound we hear may be a combination of multiple frequencies, and having different magnitude. The wave forms are in the shape of a triangle wave,
sine wave, square wave and saw-tooth wave (Figure 03). Our ear can receive frequency of a single cycle per unit time and a maximum of 20,000 cycles per second.

Frequencies are categorised into low, low-mid, high-mid and high band. There are two kinds of waves they are transverse and longitudinal. Longitudinal waves move parallel to the direction the wave is travelling. Whereas the transverse waves on the other hand is where the vibration is at 90 degrees to the motion of the wave.

When we speak, the sound usually generated at the vocal cord of the throat. Human beings have a definite frequency band in our vocals. Similarly there are different tones and timber in animals and musical instruments.

**Characteristics of a Wave Form**

A wave form is the graphical symbol that represent of a signal’s sound pressure level as it moves through a medium over time. A wave form is helpful to see & understand the actual
phenomenon that takes place in the physical environment. Followings are the characteristics of wave form.

- Amplitude
- Frequency
- Wave length
- Velocity
- Phase
- Harmonic content
- Envelope

**Amplitude** - Amplitude is the distance above (+ve) or below (-ve) the centre-line of a wave form. The greater the distance from the centre line, the more louder the sound will be.

**Frequency**: Frequency is termed as the wave vibration per unit time. The number of waves that passes through fixed place in a given time. In other words, it is the number of cycle per second and is measured in *Hertz* (Hz). The vibrating mass repeats a cycle of positive and negative amplitude. One completed journey on
both positive and negative side of the centre line is known as a cycle.

**Wave Length:** It is the actual distance in a medium between the beginning and the end of a cycle, and is measured in \( \text{Lambda} (\lambda) \). The wave length changes according to the frequency. The lower frequencies have a greater wave length and the higher frequencies have the shorter wave length. Low Frequencies travel more distance than high frequencies.

The physical length of a wave can be calculated using the formula.

\[
\lambda = \frac{v}{f}
\]

Where \( \lambda \) is the wave length in the medium?

\( V \) is the velocity in the medium

\( F \) is the frequency in hertz

**Velocity**

In physics the term ‘velocity’ is a rate of change of speed. Like the light, the sound which travels in the air. The speed of sound when it travels through air medium at 68\(^0\) F or 20\(^0\) C temperature is approximately 344 meter per second. The speed may vary according to the variations in atmospheric temperature.

**Phase:**

A cycle can begin at any point on a wave form. It is the position of a point in time on a wave form. When two sine waves combine together, it produces a single sound; their relative amplitude is different at any point of any time.
Phase is measured in degree. The sine wave is usually consider to begin at 0° with zero amplitude and it can increase to a maximum of 90° and then decrease to having a zero amplitude at 180° and then increase to a maximum of 270° (in the negative direction) & finally come back to its original level at 360°.

In-Phase: When two waves form and having the same frequency shape & peak amplitude are added, the resulting wave form will have the same frequency, phase &shape but double in amplitude. This wave forms will be called as in-phase wave.

Out-of-Phase: When two waves of different frequency, shape and phase are combined together, it results in a straight line of zero amplitude. That means both the positive phase and negative phase will cancel each other.

Harmonic Contents: A Sine wave is a single frequency that produces a pure tone. Compound sound waves have a combination of multiple frequencies. A piano can produce the tones of different frequencies and at different pitch too. The factors that help us to differentiate between instruments are called partials that exist in addition to the pitch that’s being played which is called fundamental. Partial that are higher than the fundamental frequency are called upper partial or over tone. The over tone frequencies that are whole numbers of multiple of the fundamental frequency are called harmonies.

Acoustic envelope: Every instrument has its unique timber. Timber differentiates one instrument to others. Envelope of a wave form is the characteristic variations in level that occurs over the duration of a played note.

Frequency Response

Frequency response is simply defined as the response to frequency range. Our ears can response to a too low frequency and a too high frequency. This frequency range is known as dynamic range of hearing. And the dynamic range of human ear is 20Hz to 20 kHz.
For example, we can hear the sound of a bass guitar, which carries low and low mid frequencies. And we can also hear the sound of a mosquito which creates extreme high frequencies.

Pitch

Pitch is word that defines the position of a note in the musical scale. Then what is a musical scale? Arrangement of notes in different pitch. We can say that the pitch refers to the highness and lowness of a note. Pitch is related to frequency. Two notes are very closely related. In fact it is convenient to give an exact frequency to a particular musical note, or rather the fundamental frequency of that note but we ought to note that the pitch of sound can be affected by its loudness.
Unit Summary

In this unit you learned about the definition of sound, wave propagation, role of human ear in receiving the sound, how the sound travels and reaches to our ear drum, the construction and function of the human ear, various characteristics of a wave form such as frequency, wavelength, amplitude, velocity, harmonic content, phase and acoustic envelope, pitch etc.

Assignments

1. Discuss the characteristics of wave form.
2. List the main obstacles for hearing?
3. Write the various medium through which sound can travel.
4. Describe the construction of ear with diagram.
5. Describe the pitch.
6. Name the unit of loudness
7. Frequency is measured in______.
8. We measure the wave length in______.
9. What is the dynamic range of human ear?

Resources

- Modern Recording Techniques, David Miles Hurber and Robert E.Runstein, Focal Press
- https://www.makeuseof.com/tag/7-free-resources-learn-sound-engineering/
- https://www.recordingconnection.com/courses/audio-engineering/audio-lesson-01/
Unit 2

Audio Equipments

Introduction

Sound recording is a process of converting one form of signal to another corresponding form of energy using so many equipments. In this unit you will learn about the various kind of equipments related to sound recording and processing. These equipments are used for different purpose and for various applications. For an outdoor recording audio engineer they use different kinds of outdoor portable recorder whereas for studio recording they use required equipment as applicable.

Outcomes

Upon completion of this unit you will be able to:
- Name the audio equipments that are used in audio productions.
- Differentiate between the types of Microphones.
- Identify the use of different microphone at different places
- Locate various sections of an audio mixing console.

Terminology

Acoustic Wave: A longitudinal wave that consists of a sequence of pressure pulses or elastic displacements of the material, whether gas, liquid, or solid, in which the wave propagates.

Microphone: A device that convert acoustic signal to electrical signal.

Diaphragm: A thin sheet of material forming a partition.

Capacitor: An electricity storing device.
Omni directional: Response to all direction.

Unidirectional: Response to a particular direction only.

Learners, it is very important that you should know the different types of audio equipments that are used in the recording setup.

Acoustics

The term acoustic is an interdisciplinary science deals with the study of all mechanical waves and concerned with the production, effects, and transmission of sound waves. The transmission of sound waves through different mediums, we often hear people talking about good or bad acoustics. Good acoustics may improve the sound quality. To judge the sound, we need good audible conditions. That means the venue should be good and free from any sound reflection, diffraction, refraction, absorption and interference. You might have visited any auditorium, theatre or music studio, which are treated acoustically to produce a good sound outputs. Proper acoustic design of a music recording studio is not so easy. It may be cost effective and time taking.

Microphone

Microphone is a transducer that changes one form of energy (sound waves) to another corresponding form of energy (electrical signal). It is often the first device in a recording chain. The overall sound quality of a microphone will depend upon its operating type, sensitivity and polar pattern. The other factors that influence the sound quality are placement, distance, and the acoustic environment. Knowledge of the behaviour of microphone is essential for everyone working in sound industry (Figure 07).
In order to deal with the wide range of musical, acoustic and situational circumstances, a large number of microphone types, styles and designs are available for a purposeful use. Before go for a recording let us know the microphone.

Types of microphone

There are three types of transducer used: Dynamic microphones, Ribbon microphones and Condenser microphones. All of them work on different principles.

The Dynamic Microphone

Dynamic microphone operates by using the principle of electromagnetic induction. The theory of electromagnetic induction states that whenever an electrically conducted metal cuts across the flux lines of a magnetic field, a current of a specific magnitude and direction will be generated within that metal.
Dynamic microphone consists of a stiff Mylar diaphragm of about 0.35 mil thickness. Attached to the diaphragm is a finely wrapped core of wire, also called as moving coil or voice coil. That is precisely suspended within a high level magnetic field. Whenever an acoustic pressure wave hit the diaphragm, the attached voice coil is displaced in proportion to the amplitude and frequency of this wave. Causing the coil to cut across the lines of magnetic flux supplied by a permanent magnet. Hence an analogues electrical signal is induced in to the coil and across the output lead (Figure 08).

**The Ribbon Microphone**

Ribbon microphone operates on the same principle the dynamic microphone operates.
It consists of two extremely thin aluminium ribbons. Often this diaphragm is corrugated along its width and is suspended within a strong field of magnetic flux. Sound pressure variations between the front and the back of the diaphragm causes it to move and cut across these flux lines, inducing a current into the ribbon that is proportional to the amplitude & frequency of the account wave form.

**The Condenser Microphone:**

The third type of microphone is condenser microphone, which operates on the principle of electro-static. The condenser microphone consists of two very thin plates. One immovable and one fixed. These two plates form a capacitor. And the capacitors are capable to store an electrical charge. The amount of charge that a capacitor can store is determined by its capacitance value and the voltage that’s applied to it. According to the formula

\[ Q = CV \]

Where Q is the charge, in Coulomb
C is the capacitance, in Farads.

V is the Voltage, in volt

The capacitance is determined by the distance between the plates, the substance between the plates and surface area of the plates.

Title- Figure-10: Cutaway of Condenser Microphone
Attribution- Banco
Link- https://commons.wikimedia.org/wiki/File:Mic-condenser.PNG

The plates are commuted to opposite sides of a DC power supply, which provides a polarizing voltage to the capacitor. The sound pressure wave hits the diaphragm, its capacitance charges and the distance between the two plates decreases with the increase of capacitance and vice-versa. In the formula Q=CV, R.C and V are interrelated. So if the charge (Q) is constant and the capacitance (C) changes, the voltage (V) must change in inverse proportion. Along with the variable capacitor a high value resistor is placed into the circuit that produce a circuit time that’s longer than a single audio cycle. The resistor prevents the capacitor’s
charge from varying with rapid changes in capacitance due to the applied sound pressure, the voltage across the capacitor changes according to \( V=\frac{Q}{C} \). When the voltage across the capacitor changes, the voltage across the resistor will also change. But the direction is opposite. This voltage across the resistor will become the microphone’s output signal.

**Electrets-Condenser Microphone:**

This microphones work on the same operating principles as an externally polarized condenser microphones except that a static polarizing charge is permanently stored within the diaphragm or on the back plate of Microphone. Due to this electrostatic charge no external powering is required to charge the diaphragm.

**Title-** Figure 11: Lapel Microphones  
**Attribution-** Terodaktit  
**Link-** [https://commons.wikimedia.org/wiki/File:Lavalier_mikrofon.jpg](https://commons.wikimedia.org/wiki/File:Lavalier_mikrofon.jpg)
Characteristics of Microphone:

The microphones are of different types, styles and designs to fulfill a wide range of applications. Its physical and electrical characteristics also differ. To get the best result one need to understand its directional response, sensitivity, frequency response & transient response.

Microphone’s directional Response:

All microphones have different directional responses, means responding to the sound wave at various angles of incidence. This is known as polar pattern of a microphone. Microphone directionality can be classified into two categories.

- Omni directional polar pattern
- Unidirectional polar Pattern

![Polar pattern omni directional](https://commons.wikimedia.org/wiki/File:Polar_pattern_omnidirectional.png)

**Title**- Figure 12: Directive response of Mic.
**Attribution**- Galak76

The Omni directional microphone’s diaphragm reacts equally to all sounds from all the direction. In other word the Omni directional microphone can catch the sound wave from 360 degree angle.
When the diaphragm picks up the sound pressure from the front side (on-axis) and backside (off-axis) of the microphone, we called it a bidirectional polar pattern. It means responding the sound wave from both the directions. Here the microphone can receive the sound from the opposite directions only. Various polar patterns used in the sound engineering technology are unidirectional, cardioids, super cardioids, hyper cardioids, bidirectional etc. (Figure 12).

**Microphone Placement**

Placing the microphone at a suitable distance and appropriate direction from the sound source, can give a good recording output. Microphone placement is very important. The sound intensity may vary according to the distance of the microphone from the source. Don’t hesitate to experiment on the placement of microphone. Hear the sound first. Place the microphone at the source and hear it through monitor. A microphone can give its best result when the placement will be accurate. The more the distance of the microphone from the sound source, the more it will add the ambience sound with the main signal. That means the microphone allow the room’s acoustic environment to be picked up with the direct sound signal.

**Title**- Figure 13: Placement of Microphone for vocal artist
Close microphone placement will give a tight and present sound quality and excluded the acoustic environment. The position of microphone for a singer may be different for male singer and female singer. Similarly the placement for musical instruments is also varying from source to source according to the timber, intensity and quality of the musical instrument.

**Microphone stands and cabling**

We learnt about the microphones and its characteristics. Now we will study about its supporting stands. Microphone stands should be stable and not movable and must have the ability to tolerate the weight of the microphone. There are different types, designs of microphone stands are available in the market.

Cabling is the wired communication between two devices. For example-

- from microphone to audio mixer,
- from mixer to recorder and
- from recorder to amplifier to audio monitor.

The quality of sound depends upon a neat and clear cabling. The untidy cables can create a bad impression in the minds of performers and public. A neat layout makes it easier to track down. Don’t cross the line cable over audio cable to avoid electrical fluctuations.

**Ambient Microphone Placement**

The microphone should be placed in such a distance that the room ambience is more prominent than the direct signal. The ambient pickup is often a stereo cardioids pair. This type of placement is preferred for a live concert recording, in a hall to restore the natural reverberation. It is also applicable in a live concert or musical stage show to pick up the reaction and applause of the audience. (Figure 14)
Microphone Applications

There are so many ways microphones can be used. Omni directional microphones are less affected by wind than other microphones and are therefore very suitable for out of doors interviews. It should normally be possible to hold the microphone still and between the two people (interviewee and interviewer). Most of the lapel microphones clipped to the clothing of the artist or anchor is often Omni-directional. Hand held vocal microphones are usually cardioids but good results can be obtained with Omni-directional mikes. Cardioids microphones are the most commonly used microphones and they tend to be prone to the effects of wind because there are slots behind the diaphragm to allow some sound in turbulence around these slots can be troublesome. However a good wind shield can greatly reduce the problem. Hyper cardioids microphones have the same drawbacks as cardioids microphones.
Stereo Microphone Placement

Stereo miking techniques are used to obtain a coherent stereo image. In this technique two microphones are used. These techniques can be used in either the close or distance miking of background vocals, large or small ensembles, and single instrument, on location or studio applications. There are three types of miking techniques used such as space pair, XY-Technique and the M-S Method.

![Stereo Mic placement](https://commons.wikimedia.org/wiki/File:XY_stereo.svg)

**Title** - Figure 15: Stereo Mic placement

**Attribution** - Lainf

**Link** - [https://commons.wikimedia.org/wiki/File:XY_stereo.svg](https://commons.wikimedia.org/wiki/File:XY_stereo.svg)

**Spaced Microphone** In this type of placement, the mic can be placed in front of an instrument. This techniques place the two mics anywhere from only a few feet to more than 30 feet apart and to create a stereo image, where the time and amplitude cue are used. This technique has a drawback of phase discrepancies.

**The XY Technique** The XY Technique is an intensity dependant system that uses only the cue of amplitude to discriminate direction. In this technique two directional microphones of same type, manufacture and model are placed at their face close together and facing at angle to each other. The microphone outputs are equally panned left and right. In this technique the stereo image is excellent. The generally accepted polar pattern for this technique is cardioids. They are effective for string instruments and acoustic instruments.
M-S Method

In this configuration, one of the microphone capsules is designed to be the mid position pickup. The side capsule is generally chosen to be a figure 8 patterns that’s oriented sideways. In this way the direct sound is picked up by the mid capsule, where the ambient and reverberant sound is picked up by the side figure 8 capsules.
Things to remember

Microphones are sensitive devices. So handle them with care. Never try to repair them yourself. Because microphones are complicated and delicate things and it’s all too easy unless we really know what we are doing to get things out of alignment, over tighten a screw or in some other way to make things worse. Some of the faulty symptoms with remedial actions are given below.

1. If there is no mic output: The possible causes are faulty cable and connector, plugged into wrong socket, not faded up on mixing desk. In case of an electrostatic microphone, the power may not be on.
2. If the output is very low: There may be incorrect setting on mixer.
3. If the output is too high and distorting: There may be incorrect setting on mixer.
4. Thin sound with no bass: This may be due to faulty cable.
5. Excessive bass: This may be due to faulty mic.

So whenever you face any problem due to microphone, do check plugging, mixer setting, powering, cable, connectors. If problem still persist, then take the microphone to a mechanic.

Recorder

A recorder is a device which can record or write the information and reproduce the data on demand. Recorders are of various shape, size and designs. In the field of reporting mobile phone is now becoming a portable audio visual recorder. We can record things instantly anytime and anywhere. Professional recorders have some specific features and specifications. And a layman may not operate it. In the professional recording fields people use magnetic recorders, and digital recorders for their audio project.
Title- Fig-18 Movement of Tape through 3 magnetic heads.

Link- http://artsites.ucsc.edu/EMS/music/equipment/analog_recorders/analog_recorders.html

The analog audio tape recorder (ATR) is a sound recording device that has the capacity to store audio information, on request it may play this information back using Magnetic medium. An analog ATR is called analog because of its ability to transform an electrical input signal into corresponding magnetic energy that can be store on to the magnetic tape (Figure 20)

The professional Analog Tape recorder (ATR)

The professional ATRs can be found in 2-, 4-, 8-, 16-, 24-track formats. Multi track ATRs are used to record so many individual tracks of a time layer by layer. The 8-, 16- and 24 track machines are generally used for multi track recording whereas the 2 track recorder is used to recording the final output of an audio project. Most of Professional ATRS uses three magnetic tape heads, each of which performs different tasks.

- Record head- can record the information on the tape.
- Reproduce / play back head can reproduce the information.
- Erase head can erase the data previously recorded.
Magnetic recording medium was the most popular format among the audio Engineers for decades. Technology changes rapidly and now we are in a digital world. Digitalization has become the slogan for the today’s generation. Recording has taken its place at a new level in the digital medium. The magnetic tape recorders became outdated due to some draw backs such as the machine noises, breakdown of tapes & maintenance costs. To overcome these problems engineers developed this modern recording technology called digital recording. The digital technology makes things so easier. In a simple meaning it is process of data encoding and reproducing numeric representations of analog’s original levels over time through the use of the binary number system. Recording, editing, voice dubbing and mixing become more accurate and perfect due to digital recording medium. Now a day’s potable digital recording device are used for outdoor recording and shooting also.

**Audio Mixer**

A mixer is a device which mixes up so many individual signals together in a proper ratio and can give a balanced and
processed final signal in two tracks. The basic purpose of an audio console is to give us full control over volume, tone, blending and frequencies of all signals that are applied to its input, from a sound source through microphones or electronic device, effect devices & other audio devices. It helps us to route the signals quickly and reliably. Individual signals come to the mixing console, mix together at one time during a live performance. The console has so many features to facilitate the operator, a wide range of opportunities. It provides amplification for the weak signals from the microphones and other sources, allow the operators to control, mix and balance them. Provide monitoring so that the sources and outputs can be checked and controlled. Provide communication facilities with other staffs, artists, technicians in the studio, allow audio signal to be processed and add artificial effects in the signal. The consoles are of different type, various styles and designs. On the recording industry people use various types of console. There are 2 channel, 4 channel, 8 channel, 16 channel, 24 channel, and 32 channel mixing consoles available for different applications.

![Audio Mixer](https://pixabay.com/en/sound-mixer-mixing-board-1503092/)

**Title** - Fig-20 Audio Mixer  
**Attribution** - JohnDILiberto  
The mixing console (Figure 21) has different sections for different applications. There are input section, equalization section, auxiliary section, insert point, dynamic section, monitor section, fader section, group faders, output fader, patch bay, metering, channel assignment, pan pot, aux return and master output.

**Input Section:**

The channel input section provides to optimize the signal gain levels at the input can be an I/O module before being further processed and routed. Either mic or line input can be selected. Gain trims can increase the weak signal from microphones.

**Auxiliary Section:**

The auxiliary section is used to route the signal to various effect processor and the processed signal can be returned back through the auxiliary return. Effects like delay, echo, reverberation, chorus etc can be mixed with the signal via auxiliary section.

**Equalization:**

This section is used to compensate for variations or discrepancies in frequencies that are present in the audio signal. It is having a variable bandwidth and a boost or cut control. It helps the engineer to differentiate frequencies at the time of editing, processing and mixing.

**Monitor Section:**

This section helps to monitor the signal which provides control over each inputs level, pan & effects.

**Channel Assignment:**

This section helps to distribute the signal to any or all tracks of a multi track recorder. Pressing any or all assignment
bottoms will route the input strips main signal to the corresponding track output buses.

**Fader:**

Fader provides control on the volume or gain of the balanced or imbalanced signals. The master fader serves as a convenient point for the controlling overall group output levels that are being sent to recording device.

**Monitor level Section:**

This section helps to compare and judge the input signal and output signal. Also controls levels for the various monitoring functions like control room level, head phone levels & talk back.

**Cables and connectors:**

Cables and connectors (Figure 22) are considered as the transportation system between the sound source and the recorder, from the microphone to the speaker. Cables helps in carrying the signal and the connectors help the cable and the devices to be connected for signal flow. An audio cable is used to carry the audio signal from the microphone to the mixer, from mixer to the recorder and from recorder to the audio monitor. Microphones are available with different impedances. Each impedance range has its advantages.
Cables are of two types- Balanced cable & unbalanced cable. The cable contains three wires specifics, positive (+), Negative (-) and shield or common is termed as balanced cable. The cable in which the negative, shield are combined together and positive (+) is kept separate is called unbalanced cable.

Balanced lines operate on the principle that the alternating current of an audio signal will be presented in opposite polarity potential between the two conductors.

Any electrostatic or electromagnetic pick up will be simultaneously induced into both leads at equal polarities and level. The input transformer or amplifier of the receiving device will only responds to the difference in voltage between the two leads. As a result, the unwanted noise signal will cancel and the audio signal will be unaffected. The various connector used in audio recording studio are XLR (Male and female), ¼” Jack pin, RCA, EP Jack pin. The connectors have 3 pin termed as 1, 2, 3. The pin 2 is used as positive (+) or hot, pin 3 used as negative (-) or neutral and pin 1 is used as shield or common.
Unit summary

In this unit we have described about the acoustic, different types of microphones, their operating principle, polar pattern, applications, microphone techniques and its placement, cable and connectors, analog tape recorder, record head, erase head, reproduce head, audio mixer, different segments of audio mixer.

Assessment

- Discuss the role of a sound engineer in an audio studio.
- Write how to check the connectivity of an audio cable?
- How the signal flows from Microphone to the monitor.
- Describe various polar patterns of Microphone.
- Briefly discuss the connectors used in the studio.
- Describe the principle and function of Dynamic microphone.
- Explain the construction and function of condenser microphone.

Resources

- [https://en.wikipedia.org/wiki/Microphone](https://en.wikipedia.org/wiki/Microphone)
- [http://cemca.org.in/ckfinder/userfiles/files/7_Lesson-06_MICROPHONES.pdf](http://cemca.org.in/ckfinder/userfiles/files/7_Lesson-06_MICROPHONES.pdf)
- [http://artsites.ucsc.edu/ems/music/tech_background/teces_20.html](http://artsites.ucsc.edu/ems/music/tech_background/teces_20.html)
- [https://en.wikipedia.org/wiki/Mixing_console](https://en.wikipedia.org/wiki/Mixing_console)
Unit 3

Sound Recording

Introduction

Sound recording is a process that involves both the skill of art and science. The purpose of recording is to restore the information for future use and it may be stored for years. Sound recording process takes multiple steps. In this unit we will discuss on the process of recording, analog and digital recording format, Analog Tape Recorder (ATR), DAT recorder, Analog to digital conversion (ADC), Digital to analog conversion (DAC), AM and FM radio wave transmission, audio production, editing, processing, editing and mixing. In this unit we will learn all these process.

Outcomes

Upon completion of this unit you will be able to:

- Explain the Process of Sound Recording
- Layout the diagram of an analog recording setup.
- Describe the signal flow in a digital recording system.

Terminology

ATR: Analog Tape Recorder.
ADC: Analog to digital converter
DAC: Digital to analog converter
PVC: Polyvinyl chloride
Equalization: Blending of frequencies
AM: Amplitude modulation
FM: Frequency Modulation
Recording Process

Sound recording is the technique of writing the information and storing the data in a medium such as magnetic tape, CDs, DVDs or hard disk. These are the storing devices. The recorder can record the signal and the recorded signal may be stored in a storing device for future reproduction or playback. There are two types of recording method we are using in professional field. One is Analog and the other is digital. The analog tape recording medium depends on magnetic induction theory and the digital recording medium depends on data encoding and decoding. The recording technology differs from one medium to another. Each recording format has its own distinct type of sound and application in audio and music production. Recording in analog medium is cost effective than the digital medium. We will learn the recording process in both analog and digital medium. In the music industries analog recording process was most popular among the sound engineers, musician, producers and directors for so many decades. In some places analog recording is still playing a key role in multi-track music recording. An analog ATR is called “analog” because of its ability to transform an electrical input signal into corresponding magnetic energy that can be stored on to tape in the form of magnetic remnants. On playback this magnetic energy can be reconverted back into corresponding electrical signals that can be amplified, mixed and processed. In the analog tape recording (ATR), electromagnetic induction theory plays the key role.
Magnetic Tape

The magnetic tape is the storing device in analog ATR. The tape itself is composed of several layers of materials, each serving a specific function. The base material is composed of polyester or poly-vinyl chloride (PVC), which is a durable polymer. And it can withstand a great deal of stress before being damaged. Bonded to the PVC base is the all important layer of magnetic oxide. The molecules of the oxide works together to create some of the smallest known permanent magnets which are known as domains. On an un-magnetized tape, these domains are oriented randomly over the entire surface of the tape. The net result of this random magnetization is a general cancellation of the north and south magnetic poles of each domain at the reproduced head, resulting in a signal at the recorder’s output. The speed of motor that rotate the spool that contains the magnetic tape should be constant at the time of recording and reproducing of the programming. The process of recording audio into magnetic tape depends on the transport’s capability to pass the tape across the head path at a constant speed and within uniform tension. During playback the
same time relationship is maintained by replaying the tape across the heads at the same speed, there by obtaining the original pitch, rhythm and duration.

The Function of Magnetic tape Heads

In a magnetic tape recorder, the magnetic tape head write the information on the tape. In most professional recorder, there are three magnetic tape heads, the record head, the erase head and the playback head. All the three heads perform different task during recording process.

Record Head:

The record head is responsible to record or write the data which electromagnetically translate the analog input signal supplied to it into the corresponding magnetic fields that can be permanently stored into magnetic tape.

The Signal flow and recording Process

The input signal (current) flows through coils of wire, which are wrapped around the head of magnetic pole pieces. The magnetic pole has two gaps, one at the front side called front gap and one is at the back side called rear gap. The input current causes the magnetic forces to flow through the pole pieces and across the head gap. The head gaps between poles create an insulator or breaks in the magnetic field, and create a physical resistance to the magnetic force that’s been set up. Since the gap is indirect contact with the moving magnetic tape, the tape’s magnetic oxide offers a lower resistance path to the field than does the magnetic gap. Thus, the flux path travels from one pole piece through the tape to the other pole. So the actual recorded signal occurs at the
trailing edge of the record head and the magnetic domains retain the same polarity and magnetic intensity that they had on leaving the gap. The recorded signals are stored in the tape and can be played back on demand. The playback head operates in a way that is just opposite to that of the record head.

**Erase Head:**

The head which is helping to erase the information previously recorded in a magnetic tape. The function of the erase head is to reduce the average magnetization level of a recorded tape track to zero, thereby allowing the tape to be re-recorded and reused. The professional ATRs can be found in 2, 4, 8 or 16 tracks format. Tracks are the different lines of a magnetic tape. The multi-track recorder performs a specific production and post production task. Generally two track recorders are used to record the final, stereo mix output of a project and the 8-, 16- and 24-track recorders are used for multi track laying.

**Signal Flow in a recording setup**

In a recording studio there are so many equipment, machines, cable and connectors. The recording process begins with the microphone and the acoustic signal after being converted to electrical signal by the microphones flows through the audio cables to the mixer inputs. The audio mixer has specific role to blend or modify the signal and the processed signal then given to the individual tracks of a recorder input. For example, if we are going to record the voice of a singer, then we have to set the input level of the vocal mic and then correct the errors and equalize the voice signal in the mixer and finally assign the mixer output to the recorder. After the voice being recorded in the recorder we play it
back to monitor the recording output. Like this method we can record some other instruments such as Violin, Guitar, Flute, Tabla, Drums, and Percussions etc. After the microphone, the second important device is the audio mixer. The audio mixer is a device that can receive so many input signals at a time. Then it blends, modify, split and process the individual signal channels. As the signal being processed it can be assigned to the recorder. The ATRs record the program according to the amplitude and frequencies of the corresponding signals. As soon as the recording process over the tracks are played back for reviewing purpose. This is how the signal travels from the sound source to the audio monitors through many types of machinery.

**Digital Recording**

Digital recording is the process of data conversion and storage of information in the form of binary numbers. The process involves some extra device such as ADC & DAC to convert analog to digital and digital to analog signal. This method of recording makes thing easier and simpler.

Digital audio recording process has become so familiar now a day. Like the analog recording process, this medium also has the same steps like signal routing, balancing, levelling, encoding, processing, recording, decoding and reproducing numeric representations of analog signal levels over time through the use of the binary number system. Digital audio can be likewise broken down into two analogous components: Sampling and Quantization. The sampling represents the time and the quantization represents level.
Sampling:

Sampling is the breakdown of the analog wave in time direction. In digital audio system, the sampling rate is defined as the number of instrument (samples) that are taken of an analog signal in one second. Its reciprocal sampling time is the elapsed time between each sampling period. For example, a sample rate of 44.1 kHz corresponds to a sample time of $1/44100^{th}$ of second.

During the sampling process an incoming analog signal is sampled at discrete and precisely timed intervals. The universal sample rate for an audio CD is 44.1 kHz.

Quantization:

Quantization is the breakdown of the amplitude of an analog wave signal. Quantization represents the amplitude component of the digital sampling process. The amplitude of the incoming signal is broken down into a series of discrete voltage steps. Each step is then assigned an analogous set of binary numbers that are arranged together to form binary word. The representative word encodes the signal level with as high as degree of accuracy as can be permitted by the word’s bit length and system’s overall design. The most common binary word length for professional audio is 16 bit. However this word length can be increased to 24 bit or 32 bit.

The Digital Recording process

The digital recording chain include a low pass filter a sample and hold circuit, an analog to digital converter and the circuit for signal coding and error correction. The sampling rate to
be chosen that is higher than twice the highest frequency to be recorded.

For example, a system with a frequency ranges that reaches in to 20 KHz range is often sampled at a rate of 44.1k or 48k samples/second. After the signal has been converted into a digital bit form, the data must be conditioned for further data processing and storage. This conditioning includes data coding modulation and error correction. In most of the cases, the digital media encodes data onto magnetic media in the form of highly saturated on/off transition states, the reproduced signal must be reconditioned so as to restore the digital bit stream back to its originally modulated binary state. Once this is done, the data is then de-interleaved back into its original form, where it can be easily converted back into PCM data. After the signal has been reconstructed back in to its original PCM form, the process of digital to analog (D/A) conversion can take place. We can hear the sound when it will be converted to analog form. The digital recording formats facilitate so many advance features for the audio editing, processing & mixing purpose. Digital recording can be done by using the digital recorder, a sound work station, audio recording software, Analog to Digital Converter (ADC) and Digital to Analog Converter (DAC) etc.

**Digital Audio tape (DAT) System**

The digital audio tape or DAT format is used for the creation of a compact, dedicated PCM digital audio recorder that display a wide dynamic range, low distortion and low noise ratio. A DAT is an enclosed compact cassette that is even smaller than a compact audio cassette. It has the ability with both analog and digital input.
/ outputs. And it can record and play back at the sample rate of 32 KHz, 44.1 KHz and 48 KHz.

Title- Portable DAT Recorder
Attribution- JPRoche
Link- https://commons.wikimedia.org/wiki/File:Kenwood_DAT-140218-0002WP.jpg

Mixing or Mastering:

Mixing is the final stage of a recording process. Mixing is a process of audio levelling, balancing the individual channels, blending the frequencies, modifying bandwidth, effect processing. And all these tasks are done during the post production, once the recording process is over. At this point, the multi-track’s play back outputs channels are assigned to the console input. The recorded tape is then repeatedly played while adjustments in the level, panning, EQ, effects, etc. are made for individual lack. In this artistic process the individually recorded signals are blended into a composite surrounding, stereo or mono signal that is fed from the console outputs to the master mix-down. Mixing is a key role for an engineer. It may take hours or even days to finalize a mixing.
Mixing keep going on until the engineer gets the level of satisfaction. Balancing the live audio during a concert is a big task. Experiment on mixing can be done in a postproduction studio where as there is no chance for an engineer to do experiment in a live concert. So technical expertise, skill and creativity are must required for mixing or mastering the tracks.

**Sound processing**

The processing of signal means deliberately modifying or altering the characters of audio signal. There are many processes available like frequency correction, dynamic range manipulation, addition of effects into the main signal etc. In frequency correction we usually modify the frequencies according to the timber or tonal quality of the instruments by judging through our ears and monitor. We may increase or decrease the low bandwidth, mid or high bandwidth as per the demand of the signal. It is wise not to blend or modify the original signal if it is not required. Taking the tracks carefully is much more important than modifying the signal at post production. Levelling of each channels, filtering of the noise, use of SFX or effects, normalize the audio, fade-in, fade-out, cross fade, trimming, panning etc. are the editing tools used for the signal process. Effects like eco, delay, reverberation chorus are applied to specific instruments for a live feeling. Before recording the final audio project all the necessary editing, processing, rectifications need to be done during audio post production.
Recording level

Recording level is the reference of the amount of input signal that arrives at the recorder’s input. It is necessary to check the record input level to avoid the over gain signals beyond the saturation point of the recording medium and audio distortion. A proper level of input signal has to be maintained throughout the recording process. The channel input section serves to optimize the signal gain levels at the input of an I/O module before being further processed to levels that cause the preamp’s output to be driven above +28 dBm, serve clipping distortion will almost certainly occur. In order to avoid such overloads, the input gain must be reduced. On the other hand, signals that are too low in level will unnecessarily add noise into signal path. Finding the right levels is often a matter of knowing your equipment, watching the meters and/or overload lights and using year experience.
Digital Audio Workstation

A Digital audio workstation is a computer based hard disk recording system that offers advanced multi track recording, editing, processing and integrated peripheral features in one desk. It can perform a wide range of audio related task. It has a great ability to integrate a wide range of applications and devices into a single, connected audio production environment. These systems are usually suited to perform audio, video, MIDI and hardware peripherals together under a single multifunctional umbrella that can freely communicate data and perform tasks related to sequencing, sampling, editing, signal processing and mastering etc.

Amplitude Modulation (AM)

This is a kind of technology that is used for radio wave transmission. Radio waves carry the information, whether it may be speech, music, picture or any other form of communication. This involves causing some characteristics to vary in a controlled way. We call this a modulation. There are several methods of modulating radio waves. In Amplitude Modulation (AM), the amplitude of the wave is varied to follow the shape of the modulating signal. The original wave is called the carrier frequency, usually denoted by ‘f’. An important aspect of AM and one which is perhaps not obvious at first sight, is that the modulation process creates other frequencies are called side bands.
Frequency Modulation (FM)

All the commercial and community radio stations use this technology to broadcast their signal. In FM, it is not the amplitude of the carrier which is caused to vary, it is the frequency. The amplitude remains constant. The big advantages of FM over AM is that it is a system which is much less prone to interference, whether man made or natural, for the reason that the effects tend to be amplitude affects, these might occur and affect the FM wave but receivers are designed to disregard such things being connected only with the frequency. While AM is still an important broadcast medium.
Title- Frequency Modulation

Attribution- Berserkerus

Link- https://commons.wikimedia.org/wiki/File:Amfm3-en-de.gif

It has largely been supplanted by FM for high quality music transmission. In this transmission method the carrier is modulated up and down in frequency by the program source. Since the frequency spectrum of FM is considerably wider than that if AM, an extra sub carrier can be added, enabling the medium to accommodate stereo. In this mode of transmission the stereo channels are summed to form a compatible base band or monophonic signal reception on non-stereo receiver and tuners. The subcarrier is modulated with a difference of signal which is recovered, enabling the left and right to be recovered independently.

FM transmission takes place between 88 MHz to 108 MHz, with stations located at odd intervals of 0.2MHz. Normal frequency deviation of the carrier is limited to 75 KHz in order to avoid interference with adjacent channel.
Unit summary

In this unit we discussed about the process of recording. The signal flow from the Microphone to the recorder, different medium of recording, the function of erasing, record and playback heads, analog to digital converter and digital to analog converter, DAT recorder, amplitude modulation and frequency modulation.

Assessment

- How to check the connectivity of an audio cable?
- How does signal flows from mic to the monitor.
- Briefly discuss the connectors used in the studio.
- Differentiate AM and FM.

Resources


https://en.wikipedia.org/wiki/Sound_recording_and_reproduction
https://www.makeuseof.com/tag/7-free-resources-learn-sound-engineering/
https://www.recordingconnection.com/courses/audio-engineering/audio-lesson-01/
https://www.britannica.com/topic/digital-sound-recording
https://en.wikipedia.org/wiki/AM_broadcasting
https://en.wikipedia.org/wiki/Frequency_modulation
Unit 4

Audio Editing

Introduction

Recording takes place in three phases such as pre-production, production and post-production. In the production, tracks are recorded. After that the artist or singer’s voice is recorded on the same track. After the dubbing and editing final mixing/mastering is done. In the post production many things are to done on the mixing console desk. Noise is an important factor to be remembered and taken care of during track lying and editing.

In this unit we will discuss on various noise reduction process. We will also focus on audio monitor and its applications.

Outcomes

Upon completion of this unit you will be able to:

- Describe Voice Dubbing process.
- Describe the monitoring system.
- Explain different noise reduction system.

Terminology

Dubbing: Recording of voice or instrument on a pre-recorded track.

Gain: Volume control.

Noise control: Reducing of unwanted environmental sound

Noise gate: A device to filter the noise.
Dubbing

Once we have done with the track recording or shooting of a scene, we can proceed for dubbing. It may be an audio dubbing or a video dubbing. Audio dubbing can start when the track recording gets over. The artist or singer listen to the tracks carefully, remember the tune, scale and rhythm pattern of that song and then try to sing the song referring to the dummy-voice track. A dummy voice track is a reference track that represents the length of the lines, stanza, tempo, feelings and landing of the notes. The singer needs a phone to listen the music track. Along with the music track the singer must be able to hear his/her own voice loudly, so that he/she can sing in proper synchronization.

Dubbing is a process of track addition to the basic tracks previously recorded. These additional tracks are added by monitoring the previously recorded tape tracks. In an overdub session, the same procedure is followed for mic selection, placement, EQ and levels as occurs during the recording session. Dubbing can be done in a sound proof or acoustic studio. However, leakage can occur if the musician’s headphones are too loud or are not sealed properly on their head. Dubbing is done on proper synchronization with the original track. If the recorder to be used is analog it should be placed in the master sync mode. Dubbing takes place during post production work. Video dubbing can be done by referring the image or video on a big screen and record the voice in properly synchronization.
Editing

Editing is a process of assembling the data or images in a sequential order. In other word we can say the arrangement of clips, scenes, shots, images according to the flow of story. Editing is applicable to print medium, visual medium and audio medium. Editing in different medium is different. Different softwares are used to edit the image of video and audio. There are types of audio editing softwares are available like Audacity, Sound forge, Sony Vegas Pro, Nuendo, Protocols etc. Audio editing is an essential part of audio production. It helps us to remove the errors from a track to correct the errors, to process the signal, use different effects to the signal, bouncing the track, cut &remove the unused part, trimming, crossfade, fade-in, fade-out, overlapping, noise reduction, normalize the level etc. are the basic tools of sound editing.

Audacity Software

Audacity is user friendly software, compactable with windows 98, ME and 2000 and also with the XP, MAC OS,OS X, Linux, UNIX and
other operating systems. It provides with a full set of tools for recording and editing the audio files, adds on effect to the signal. It can both record and playback the tracks. It enables the user to schedule in and out of a recording. It can also import and export the different audio formats like WAV, MP3, AIFF, OggVorbis, WMA, AAC, AMR etc. It can be used for multi-track music recording, a large use of digital effects and plug-ins, VST plug-ins, Noise reduction, vocal reduction and isolation for the creation of karaoke tracks. It supports only 32-bit or 64-bit VST audio effect plug-ins. Whereas it lacks dynamic equalizer controls and real time effects while recording. Learner can download this software and start experimenting on this.

**Editing in Analog medium**

Editing in analog medium needs more attention. It needs both theoretical and practical knowledge. During this process, the engineer edit the original music track out from their reels and begins the process of splicing them together into a final sequence on a master reel set. Once this is done, the mix master in/out edits should be tightened. This is done by listening to the intro and outro at high volume levels, while the heads are in contact with the tape. The tape can then move back and forth to the exact point where the music begins and after it ends.

The length of time between the end of the song and the beginning of the next can be constant or the timings can vary according to the musical relationship between the songs. When the sequencing is complete, one or two analog or DAT back copies should be made to the final sequence master, before it leaves the studio.
Editing in Digital Medium

Editing in the digital workstation has wide applications because of the wide range of plug-ins and tools. With the advent of digital audio editing systems, the relatively cumbersome process of sequencing music tracks in the analog domain using magnetic tape has given way to the faster, easier and more flexible process of editing the final master from hard disk. When a computer based editor is used, the start and the end points can be located for each song and can define a region that can then be assembled into a final song version. Once this is done, each song can be individually processed using EQ. Overall level, dynamics, effects etc.

Title- Editing of Sound clip

Source- Screenshot

Noise Reduction

Any ambience sound that disturbs us in hearing to the main signal is considered as a noise. Noise is the unwanted sound or signal that disturbs our main program, that’s creating an obstacle in our communication. It is one of the most challenging tasks during recording. In audio recording process the noise needs to be reduced and should not overlap the main signal. Noise may
be created by so many sources in the room and outside also. In an analog ATR and VTR format, we usually notice so many noises like the tape noise, hissing, through print, humming etc. We will largely focus on reducing tape noise that’s a natural by product of the analog recording or playback process.

**Dolby and DBX noise reduction system**

We must have heard about Dolby digital sound system. The first widely used audio noise reduction technique was developed by Ray Dolby in the year 1966. Intended for professional use, Dolby Type A was an encode/decode system in which the amplitude of frequencies in four band was increased during recording (encoding), and then decreased proportionately during playback (decoding). The Dolby B system was a single band system designed for consumer products. In particular, when recording quiet parts of an audio signal, the frequencies above 1 kHz would be boosted. This had the effect of increasing the signal to noise ratio on tape up to 10 dB depending on the initial signal volume. When it was played back, the decoder reverses the process, in effect reducing the noise level by up to 10 dB. The Dolby B system, while not as effective as Dolby A, had the advantage of remaining listenable on playback systems without a decoder. DBX was the competing analog noise reduction system developed by D.E.BLACKMER.

**Analog noise reduction**

When recording is done in an analog medium, special care has to be taken at the time of recording. It is too difficult to reduce the noise from a recorded signal. Better to avoid the unwanted sound or noise at the time of recording.
Noise reduction in analog tape recording is very necessary in order to get a proper audio quality. Analog tape noise might not be a limiting factor when we are dealing with one or two tracks in an audio production, but the combined noise and other distortions that are brought about by combining 8,16,24 or 48 tracks can range from being bothersome to downright unacceptable. The different types of noise that create problems in analog recording formats are tape and amplifier noise, cross talk between tracks, print through and modulation noise.

**Digital Noise reduction**

In digital signal processing, the noise can be reduced easily by the use of various noise filter plug-ins. The noises like tape hissing, hum, obtrusive background ambience, needle ticks, pop and certain types of distortion that are present in the original recording can be removed by using noise gate and limiters.

**Source of noise**

Noise may generate from different sources. It may bear from the environment and travel through air is called external air borne noise. Noises that is being created by any giant machineries or constructions is called external structure borne noise, sometimes it may travel from other nearby rooms of the same building is called as internal noise transfer. Recording generally takes place in the indoor setting or at the outdoor location. Noise is everywhere in our surrounding. Our main goal is to avoid the unwanted sounds and record a clean and noiseless project. Noise can be reduced by taking some necessary steps like the proper microphone placement, using of windshield, controlling the environmental noise like crowd, traffic, switching off the electrical...
equipments such as fan, motor, Ac, generator, grinder or other noise producing accessories.

**Noise Gates**

To reduce the noise there are so many applications and tools available in a digital workstation. A noise gate is one of the best effective noise reduction devices which are used to reduce the background noise on certain program material. A noise gate effectively passes signals that fall above a user defined threshold at unity gain, while turning off signals that fall below this threshold. This is a useful tool for removing noise, leakage and other gremlins from a track within a mix.

![Diagram of a Noise Gate](https://commons.wikimedia.org/wiki/File:Noise_Gate_Attack_Hold_Release.svg)

**Title**- Diagram of a Noise Gate  
**Attribution**- Lainf

Monitor is required to view or listen to a program. It may be a TV, Computer, LCD, and LED Screen, a headphone, ear phone or speakers. Devices that allow us to judge a program is called a
monitor. In the recording process, we use professional audio monitors to listen a program. They should not produce colorful program with additional low and high frequency. In the professional studios good acoustic, active monitors are used. These monitors are used throughout the recording process such as track recording, editing and mixing or mastering. The placement of these monitors is also important. Monitors are placed on the front console board with a little angle to the ears. When mixing, it’s important that the engineer be seated as closely as possible to the center of the sound field and that all the speakers volumes are adjusted equally. The engineer should always make sure that an audible volume deferent between speakers is accompanied by a corresponding visual difference on the VU meters, which are monitoring the signal sent to tape.

Title- Field monitors at console desk
Attribution- Photographs by Author
Monitoring Configurations

Stereo monitoring and Mono monitoring are some terms used in audio technology. The programs are designed in such a manner that gives differentiation feelings in our hearing. There are mono, stereo, mono surround and stereo surround. A large percentage of people hear the music or program through various monitoring devices like speaker, earphones or head phones. The music seems good in stereo headphones. Stereo tracks have different musical treatment on both left and right side of the tracks. Whereas in mono, it contains a single and mix track that sounds equal to both the ears. Mono mastering is done for TV program, programs broadcasted at AM Radio channels, whereas stereo track mastering is done for songs, musical jingles that can be heard through ear phones, head phones or stereo speakers. The most commonly speaker configurations are mono, stereo and surround sound.

Mono

When all the individual sound tracks are bounced to a single track only, that gives a common listening to both the ears. For example, listening to the program of All India Radio, Watching the TV, listening to the music in an elevator, on the computer, we may experience a mix mono-aural sound. In this case the sound engineer assigns all the tracks to a single mono track. The sound on each ear will be same without any hearing variation.

Stereo

In stereo mastering, we have two different tracks, one is for left ear and another is for right ear. Stereo tracks give us different sound effects at both the side. To get the stereo feeling,
we need two speakers for both the ears and they should be placed in such a manner that the left ear can perceive the sound of left speaker and the right ear will perceive the sound of the right speaker. The mastering of a good and impactful stereo tracks are extremely important, with relation to L/R balance, overall frequency balance, dynamics and effects. When mixing in stereo, it is always a good idea to check for mono compatibility

**Surround Sound**

Our ears have the ability to receive the sounds from 360 degree. Hence to give a real surround effect, surround sound mixing is done for the theatres.

Surround sound configuration gives us a 360 degree hearing effects, depending on the design of the hall and the placement of the speakers. In the cinema theatre, we feel the Dolby digital surround sounds. Surround sound has grown into a major professional and consumer entertainment market. In the house and audio theatres, 5.1 surround playbacks are available. The 5.1 name refers to the five, full range channels (Left, center, right, surround left & surround right), plus a six sub-bass channel.

![Surround Sound Diagram](image)

**Title:** Placement of speaker from the listener  
**Attribution:** Kamina
Surround sound adds multi channels from loudspeakers behind the listener, thus is able to create the sensation of sound coming from any horizontal direction 360° about the listener. Surround sound is generally depending on the location of the listener and presents a fixed or forward perspective of the sound field to the listener at this location.

In a 5.1 surround system, the phantom images between the front speakers are quite accurate, with images towards the back and especially to the sides being unstable. Also 7.1 channel surround is another setup, commonly used in large cinema theatres which is compatible with 5.1 surround systems with two additional channels, center-left, and center-right to the 5.1 surround setup, with the speakers situated 15 degrees off center from the listener.

Another surround set up is 10.2. This format was developed by THX creator Tomlinson Holman of TMH Labs and University of Southern California. This is just the twice of 5.1 Setup.

Active Monitors for studio

A speaker is a device which convert electrical signal to acoustic signal. Each speaker is acoustically measured and adjusted using an instrument known as a spectrum analyzer, which is used to visually display the speaker’s frequency response as measured through a specially calibrated omnidirectional condenser microphone. More accurate readings of both a speaker’s frequency response and delay or reflection response can be measured by using a Time Delay Spectrometer (TDS). A variety of speaker types & design are available in the market for professional use.

Near-Field Speaker

Near field refers to the placement of small to medium sized speakers to each side of the working environment or on the production console. These speakers are usually placed at a closer
distance, allowing the engineer to hear more of the direct sound and less of the room’s overall reflection.

**Far-Field Speaker**

Far-field monitors often large loudspeaker systems that are capable of delivering relatively accurate sound at moderate to high volume levels. Because of their large size and basic design, the enclosures are generally built into the control room wall to reduce reflections around and behind the enclosure and to increase overall speaker efficiency.

**Speaker polarity**

Speaker Polarity is said to be electrically in phase whenever one signal is equally applied to both speakers, which will make their cones to move in the same direction.

When the speakers are wired out-of-phase, one speaker cone will move in one direction while the other moves in the opposite direction. Speaker polarities can be easily tested by applying a mono signal to both or all of the speakers at same level. If the signal images appear to originate from directly between the speakers, they have been properly wired in phase. An out of phase speaker condition can be easily corrected by checking the speaker wire polarities.

**Headphones**

Almost all of us use headphones or ear phones. Headphones are also an important monitoring tool. It helps us to get the actual sound and acoustic effects by avoiding the room’s acoustic environment. Open air and sealed headphone types have their advantages.
Amplifiers

An amplifier is a device that amplifies the signal level. Amplifiers have many applications. They can be designed to amplify, equalize, combine, distribute, or isolate a signal. They can even be used to match signal impedance between two devices. At the heart of any amplifier system is either a vacuum tube or semiconductor transistor design.
The operational Amplifier

The operational amplifier or op-amp is a stable, high bandwidth amp that has high input impedance and low output impedance. The qualities allow op-amps to be used as basic building blocks for a wide variety of audio and video applications, simply by tagging additional components into the basic circuit to fit the required design needs.

The Preamplifier

Every mixing console has a pre-amp section. The purpose is to amplify the weaker signals that come from the microphone. This amp type is often used in a wide range of applications, such as boosting a mic’s signal to line level, providing variable gain for line level signals, isolating signals from extraneous input interference or improper grounding or signal voltage conditions and equalization.

Distributing amplifiers

Distributing amplifiers are useful to increase power. It is necessary for audio signal to be distributed from one device to several other devices or signal paths within a recording console or music studio. Whenever increased power is needed, a distribution amp may be required. Under seen circumstances, a distribution amp might not provide gain, but instead will amplify the current that is delivered to one or more signal loads.
Power Amplifier

Power amplifiers are used to boost the current of a signal to a level that can drive one or more loud speakers at their rated volume levels.

Synchronization

Synchronization is a word used mostly at a multi setup studio. Whenever we use more than one audio visual machines at a setup for recording purpose, we have to run them in sync. In media production studios the sync refers to time relationship. This is a process that allows multiple audio visual media to maintain a direct time relationship. Synchronization is the occurrence of two or more events at precisely the same time. In the analog audio and video systems, Sync is achieved by interlocking the transport speed of two or more machines. All the devices of a setting must maintain the same relative speed at all points over the course of a program.

In digital systems, internal or external sync between compactable devices is often maintained by using a clocking pulse that is directly embedded within the digital data line itself. It is necessary for both analog and digital devices to be synchronized together. As a result a proper system communication and data translocation have been developed.

Time Code

The Method of interlocking audio, video, film making uses code that was developed by Society of Motion Picture and Television Engineers (SMPTE). The time code helps us to identify a specific position on a tape or within the media program by assigning a digital address to each specific length. The specified tape segments are called frames. Each audio or video frame is tagged with a unique identifying number. This number is known as time code address. The eight digit address is displayed in the form 00:00:00:00.
Unit summary

In this unit we learnt about the process of voice dubbing, voice over, editing, processing of signal, noise reduction, noise gate, amplifiers, synchronisation, time code, monitoring system, speaker polarity and many more.

Assessment

1. Discuss the role of a sound engineer in the audio studio.
2. How to check the connectivity of an audio cable?
3. How the signal flows from Mic to the monitor.
4. Describe various polar patterns of Microphone.
5. Briefly discuss the connectors used in the studio.
6. Discuss different noise reduction systems.
7. Discuss various types of amplifiers.
8. What is the need of time code?

Resources