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CSAVT-01
Sound Technology

Certificate in Sound, Audio and Video Technology (CSAVT)

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Sound Technology

Dr. Babasaheb Ambedkar Open University

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Editor

Dr. Himanshu Patel
Assistant Professor, School of Computer Science
Dr. Babasaheb Ambedkar Open University, Ahmedabad

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Unit-1 Evolution of Sound Technology

Introduction

The growth of the live music industry and movies pushed audio technology from its very humble beginnings into quite technically advanced systems that were utilized in the event industry. And certainly audio systems have evolved far from being a simple PA at today's events – they are engaging, immersive and highly refined masterpieces that are played by audio artists around the world.

For the electric PA, or audio system, to come into being, the component parts first had to be invented and then combined to fulfill a need. The three basic building blocks of all PA systems are:

- A device to capture sound vibrations and convert them into an electrical signal
- A way of increasing and controlling the electrical signal
- A device to convert the electrical signal back into vibrations and propagate them.
- Or, in simple terms, we need a microphone, an amplifier, and a loudspeaker.

Outcomes

Upon completion of this unit you will be able to:

- Describe evolution of sound technology

The Acoustic Era To 1925

Most historians go with David Edward Hughes, who seems to have beaten Alexander Graham Bell, Emile Berliner, and Thomas Edison by a few years to demonstrate his rudimentary carbon microphone in 1875. He never sought a patent for his invention, as he wanted it to be a gift to mankind but he did coin the word "microphone" by thinking of it as the audio equivalent of the microscope.

Throughout the 1870s and 80s, various loudspeaker-like devices existed, most notably on Alexander Graham Bell's telephone (1876) and Edison's phonograph (1877), but the true moving coil loudspeaker, the forebear of all loudspeakers since, was invented by Oliver Lodge in 1898.

The final piece of the PA puzzle came along in 1906 when Lee DeForest invented the Audion, the first device capable of amplifying an electrical signal.

The first documented example of an electric PA system being used to amplify speech and music at a public event was on December 24, 1915, at San Francisco City Hall, when Jensen and Pridhams' Magnavox system was publicly demonstrated (Magnavox being Latin for "great voice").

The Electrical Era To 1945

Improvement and refinement of PA technology continued, with systems for motion pictures being a primary driver. Things really started to get interesting during the 1950's as the invention of the electric guitar and the growing popularity of rock 'n' roll increased the need for greater amplification.

The first operational amplifiers were used in professional audio equipment, notably as summing devices for multichannel consoles.

In 1967, the Monterey International Pop Festival becomes the first large rock music festival and the Broadway musical "Hair" opened with a high-powered sound system. These large format demands for more power continued to drive evolution forward with technology being invented to fill the need.

The Magnetic Era To 1975

Once the amplification concern was addressed, focus turned to the quality and shape of the sound that was now coming (loudly) from the system. EMT produced the first digital reverberation unit as its Model 250 in 1975. The ability to affect the quality of the sound coming out of an audio system charged the well-known 1970's commercial, "Is it Live or is it Memorex?" In the TV commercial, the "Queen of Jazz," Ella Fitzgerald, would sing a note that shattered a wine glass while a Memorex tape recorded her voice. When playing back the recording from the tape, the commercial would question the audience, "Is it live or is it Memorex?" The idea being that with new technology, live and recorded sound couldn't be differentiated.

The Digital Era To Current

In 1990, Dolby proposed a five-channel surround-sound scheme for home theater systems. The digital and experiential era had begun and digital audio systems were rapidly replacing analog systems.

Digital audio is an audio signal or more simply, a sound signal that has been converted into digital form, where the sound wave of the audio signal is encoded as numerical samples in continuous sequence.

Digital audio systems may include such applications and are used to manage compression, storage, processing and transmission components. Conversion to a digital format allows for convenient manipulation, storage, transmission and retrieval of an audio signal. Unlike analog audio, in which making copies of a recording may degrade the original recording quality when using digital audio, an infinite number of copies can be made without any degradation of signal quality.

Where Audio Technology Is Today

Electronics Are Getting More Efficient

Gone are the days of seeing racks on racks of amplifiers on stage. In the past, an Audio Engineer required many racks of amplifiers for a show, as each one was capable of only producing so much power. Audio Engineers can now fit the same amount of power into a smaller footprint, as amplifiers have got lighter and more powerful. This effectively reduces the quantity or size of the electronics needed to reproduce an amplified signal.

Audio Consoles Are Smaller

As seen in consumer electronics, technology has advanced to needing smaller components, are made with lighter materials and have immensely more capabilities compared to analog counterparts.

Analog consoles did not have all the onboard processing, effects units. The advent of digital consoles brought on the ability to put much more into a single device.

Powered Speakers Are More Popular

There is a more prolific use of powered speakers, as opposed to a separate amplifier and speaker combination. In a traditional audio setup, you have amplifiers on the ground and speakers in the air. Now those two components are combined into one. This saves both on the quantity of equipment and the footprint required. Audio Engineers no longer need to have five to six racks of amplifiers to power their system – they only need an electrical distribution rack. They are effectively running power directly to the speakers instead of power to the amplifiers and then cable to the speakers.

Effects Units Rock!

With the desire for more control, more flexibility and more creativity, effects units have become ubiquitous in the audio setup. In the days of analog equipment, an Audio Engineer may only have 3 or 4 effects units for the entire show. Now they may have three or four effects units available on a single input. This proliferation of effects units provides Audio Engineers with increased artistic choice and control.

Digital Audio-Transport Technology Continues To Grow

Digital audio can now be carried over a network using audio over Ethernet, audio over IP or other streaming media standards and systems. Digital audio can be carried over digital audio interfaces, such as Dante (delivering ultra-low latency and near-perfect synchronization), AES3 (also known as AES/EBU, this was jointly developed by the Audio Engineering Society (AES) and the European Broadcasting Union EBU) or MADI (Multichannel Audio Digital Interface).

Several interfaces are engineered to carry digital video and audio together, including HDMI and DisplayPort.

Network Control In The Palm Of Your Hand

Modern technology now allows everyone to have a computer in their pocket. As we see in Smartphones, technology is getting smaller but the screens are getting bigger. There is a demand for them to do more and more with the ease of just adding an app. This provides the Audio Engineer with the capability to tune the PA and make adjustments right from a Smartphone or tablet, giving them the flexibility to assess the sound quality anywhere in the room and not have to run back to the console to make adjustments.

Technology Supporting Design

Computer-based acoustic PA modeling allows Audio Engineers to “design” their audio system rather than “just hang a bunch of speakers.” Systems are now strategically designed, as opposed to just assembled. These CAD programs allow you to input room models and architectural CAD drawings to accurately plan and project how speaker systems will work in a venue before you even get on site. This design work improves PA design and can reduce setup time on site.

Where Audio Technology Is Going And What We'd Like To See

While audio technology hardware advancement appears to have plateaued, the growth will be in the software and programming that manipulates the systems to make them more flexible and allow for greater control and creativity.

While Technology Never Seems To Stand Still, What We Would Like To See In The Next Evolution Of Audio, Would Be:

Continuing wireless technology improvements and reliability to reduce the amount of cabling and connectivity that is currently required.

Lighter, smaller and more powerful components that reduce the costs associated with the storage, shipping and setup time for large audio systems.

Cost reductions as new technologies are developed so that the advancements can be put into action on the stages, floors, trusses, and towers of the events of tomorrow.

Unit summary

The audio industry has come along way since that first microphone was created in 1875 and will certainly continue to think “outside the box” to drive future innovation of technology. While basic physics rule the law of sound and how it is carried or transmitted, the Audio Engineers of today and tomorrow will keep on using their creative and the hardware available to meet the demands of an ever-evolving industry.

Unit-2 Audio Fundamentals

Introduction

With the expansion of multimedia in all aspects of our life, inclusion of audio in that equation is increasingly important. Audio technology has expanded dramatically over the last few years. In our daily life, we use all components of multimedia presentation i.e. Sound, video, animation, text and images etc. One can easily say that speech is often the most preferred and used tool of interaction. That is why; in multimedia presentation sound is one of the most important elements of communication. A presentation can be based on sound alone or sound may be used in supporting role in the form of music and sound effects. In the form of speech, the sound becomes the main tool of communication but its supporting role is equally important as it influences the emotions of the audience. In order to utilise sound to its maximum potential in our multimedia presentations, it is very important to understand the very nature of the sound itself and the devices used to create, process and record the sound. In this unit, we will explore the basics of audio and sound, as well as some of its fundamental frequency, format, equipments, other tools, etc.

Outcomes

Upon completion of this unit you will be able to:

- Describe how sound is produced.
- Identify factors influencing quality of sound.
- Judiciously use recording devices for quality recording.
- State techniques of noise suppression and utilisation.
- Evaluate and select right kind of microphones.
- Explain studio system
- Differentiate between mono, stereo and surround sound systems

Terminology

Sound:	Disturbance in the air that can be heard.
Audio:	Generally refers to the sound in electrical form.
Frequency:	Cycles/vibrations per second.
Amplitude:	Height of Sound wave.
Hertz:	Unit of frequency measurement.
Polar:	Related to direction of sound
Reverberation:	Persistence of sound in space.
Equalisation:	Adjusting relative levels of different frequencies.

What is Audio?

Audio means "of sound" or "of the reproduction of sound". Specifically, it refers to the range of frequencies detectable by the human ear — approximately 20Hz to 20 kHz. It's not a bad idea to memorise those numbers — 20Hz is the lowest-pitched (bassist) sound we can hear, 20 kHz is the highest pitch we can hear.

Audio work involves the production, recording, manipulation and reproduction of sound waves. To understand audio you must have a grasp of two things:

Sound Waves: What they are, how they are produced and how we hear them.

Sound Equipment: What the different components are, what they do, how to choose the correct equipment and use it properly. Fortunately it's not particularly difficult. Audio theory is simpler than video theory and once you understand the basic path from the sound source through the sound equipment to the ear, it all starts to make sense.

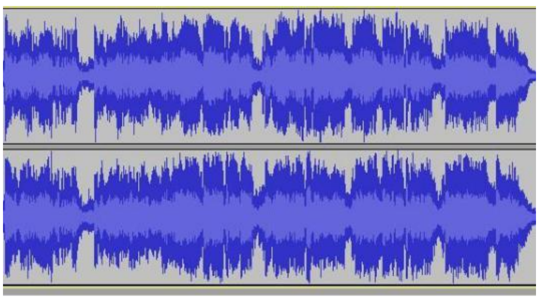
Note: In physics, sound is a form of energy known as *acoustical energy*.

What is Sound?

Sound is any disturbance that travels through an elastic medium such as air and it is heard by the human ear. We are saying that sound is a disturbance not because it is useless, but actually it refers to the mechanism of sound production/generation. Anything that vibrates would disturb the air around it and it is this disturbance which travels in the air and reaches our ears. The disturbance in air produces a sensation of hearing in us by vibrating our ear drum. You may observe the phenomenon of sound generation by placing your fingers on your throat when you are speaking. Same way, when you hit a table or any other solid object, it vibrates and generates the sound.

Nature of sound

A vibrating (oscillating) body causes a periodic (rhythmic) disturbance in the surrounding air and generates sound waves (also called pressure wave) which on reaching us produce a sensation of hearing. In real life, the vibrating objects could be of any shape and sizes vibrating in a very complex manner, therefore the sound generated are very complex. A typical sound wave represented graphically, as shown below.

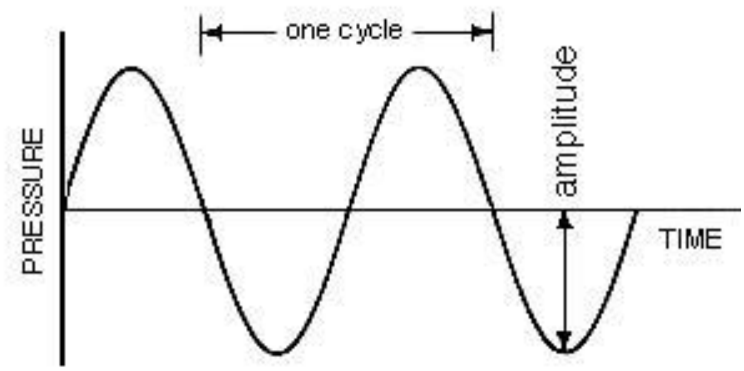


Title- Fig. 1.1 Graphical Representation of Sound Wave

This kind of waves you would be able to see while working on audio workstation. For the sake of understanding, the wave is usually shown as simple wave, in the form of sinusoidal wave (sine wave), as depicted below

Title- Fig. 1.2 Simple / Sinusoidal Sound Wave

Let us understand how a vibration converts into a waveform. In the diagram below, the body is vibrating along the vertical axis. The body starts from its initial position at the centre and goes to one side of the centre (say above) and after reaching an extreme position comes back to the centre. Continuing its journey, it moves to another extreme point on the other side (lower side) and then come back to the centre. This completes one cycle of the vibration. You may see that the wave is simply imitating this movement but expanded along X-axis with time. The image below indicates how a single vibration is translated into waveform.



Title- Fig. 1.3 Diagram showing a vibration converting into a waveform

Attribution-

Source- www.sfu.ca

Link-https://www.sfu.ca/sonic-studio/handbook/Sine_Wave.html

The number of vibrations occurring per a second is the *frequency* of the vibration and therefore of the sound being generated due to this vibration. The frequency is measured in *Hertz*, named after the scientist Heinrich Hertz and is denoted as Hz. The human ear can listen to the sounds only in the frequency range from 20Hz to 20 kilo hertz (KHz.). All the sounds of frequency below 20Hz are called *subsonic* and the sounds above 20 KHz. are called *ultrasonic*.

The word "*Sonic*" refers to sounds within the audible range. The frequency range from 20 Hz. to 20 kHz is called the *audiblerange*.

The graph is merely a representation of the sound wave and it helps us in understanding the quality of sound being generated, processed and recorded by various audio devices. In the image above, the amplitude of the wave represents the strength of the wave which in the case of sound represents the power/, intensity of that sound wave. More is the sound amplitude (level), louder is the sound.

The terms “sound” and “audio” may be used interchangeably but the convention is to use word “sound” in the physical space while “audio” is more often used to represent sound travelling through the devices in the form of electric current. How sound is converted into electrical current we shall understand later in this unit.

The intensity/level of the sound is measured in “*decibels*” denoted as “db”.

Title-Fig. 1.4 A VU Metre

Attribution- Iainf

Source- wikimedia.org

Link- https://commons.wikimedia.org/wiki/File:VU_Meter.jpg

The audio level in audio devices is usually measured by a metering device known as VU metre, as shown above.

Perceiving Sound & Listening

Our brain separates the desired sounds from the unwanted sounds through a very complex psycho acoustical process. When we are listening to a conversation or a piece of music, our ears keep adjusting to small changes in the sound levels.

The quality of sound that our ear will accept and prefer shall depend on what we are used to listening in our daily life. If we listen to good quality sounds in routine e.g. sounds of good quality TV and music systems, then we are not satisfied with an inferior quality sound as we are able to differentiate between good quality sound and the bad one. At the same time, any person deprived of good quality sound may remain happy with an inferior quality. It means quality of sound may become a subjective issue. Different quality may be acceptable to the same person listening to different types of programmes e.g. telephonic sound is much inferior to the sound in a cinema hall. Another thing to keep in mind is that our ear sensitivity is highest at 1 KHz. and the loudness of sound is a subjective quality.

The Audio Quality

As you know the basic taste of food depends upon factors like its saltiness, sweetness, sourness or the bitterness. The additional parameters which are colour flavour, hardness, temperature and its crispness (sound in food!!!) also influence the taste. Therefore an expert in food tasting would consider all above parameters while evaluating food. Similarly, the major factors which need utmost attention to decide the quality of the sound are:

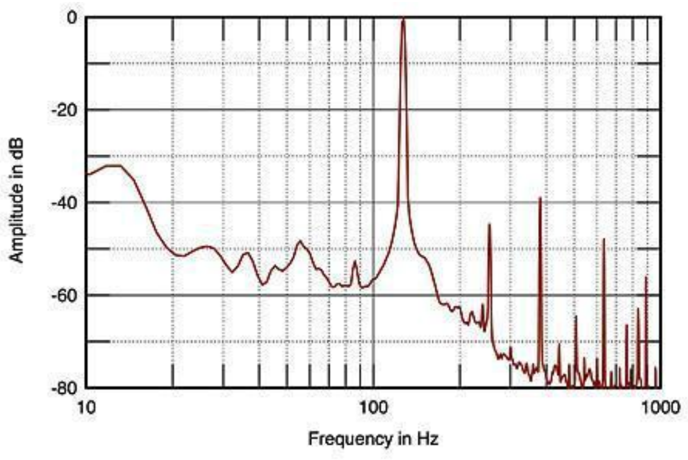
- Frequency response
- Distortion
- Noise
- Reverberation

These four factors influence the quality of the sound to a great extent therefore we need to understand them in detail as discussed below.

Frequency response

You know that human ear can listen to sounds ranging from 20Hz. to 20 kHz but all the sources of sound do not produce or receive the sound in full audio range.

The *frequencyrange* a device is capable to generate, process, record or playback is known as frequency range of that device. A device having a 'flat' frequency response would mean it will not change the weighting i.e., intensity of the audio signal across the specified frequency range while processing. A typical frequency response is shown graphically in the diagram



below.

Title-Fig. 1.5 A typical frequency response shown graphically

Attribution- John Atkinson

Source- stereophile.com

Link- https://www.stereophile.com/content/audioengine-2-powered-loudspeaker-measurements

You may observe that in graph the signal from 20 Hz to 20 KHz is at the same level of about 0 db. If the gain (Level control) of the system is turned down, the graph would also come to a lower level but maintaining the horizontal line across the frequency range.

Any sound in nature is usually not generated as a single frequency. The lowest frequency in a sound is called *fundamentalfrequency*. Along with fundamental frequency some high frequencies are also produced called *overtones*. These overtones add special character and richness to the sound. The low frequency sounds are called "*Bass*" and they attribute heaviness and richness to the sound e.g. Sound of a drum has a heavy bass. On the other side, the high frequencies give brilliance to the sound and are generally known as "*treble*". The chirping sound of the birds belongs to high frequency.

In music, the range of frequencies generated by a musical instrument is known as the range of that musical instrument. The range may be understood as the distance between the lowest pitch (frequency) and the highest pitch produced by the instrument.

A single frequency sound, in the music reference, is known as "*note*". For example, Sa, Re, Ga , Ma, Pa, Dha, Ni sounds are called notes where Sa is the lowest frequency and Ni is the highest frequency (double of Sa) in an octave. The octave is the range of sounds in which frequency doubles i.e. the frequency of the last note is two times of the frequency of starting note. Range from "Sa" to "Ni" is one octave.

A musical composition comprises of combinations of these notes at different intervals and intensities. Therefore to preserve the originality of the music any sound recording device should neither alter the frequency of the notes and their relative intensities. If the device is unable to record different frequencies at their relative levels then such deviation would mean that the device has poor frequency response (non-flat or limited frequency response). The concept of frequency response can be understood by listening to a piece of music on a portable device e.g. mobile phone through its speaker and then listening to the same piece of music on a home theatre system or a cinema hall. You will notice that the music on two situations gives you different experience. It sounds quite differently in cinema hall as compared to mobile phone. It is so because of difference in frequency response of the sound systems in mobile phone and cinema hall.

In the same way, the deficiency in the frequency response of a device would also affect the speech quality. Therefore you may conclude that good frequency response is one the essential features of good quality audio device. Closer is the frequency response to 20-20KHz., better is the device.

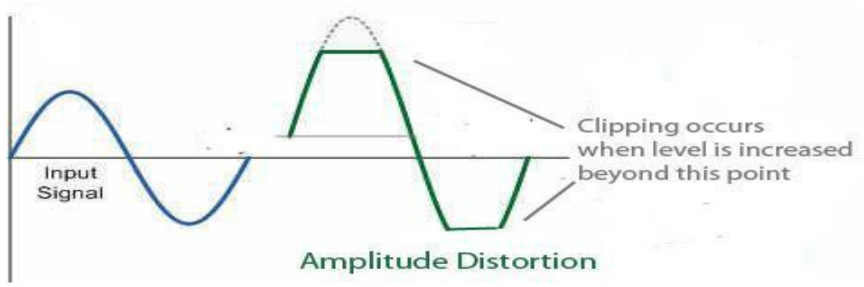
In a marriage or other party, you might have observed that parties having, each speaker box/enclosure of the music system comprises of two or more speakers. It is because; no single speaker can reproduce full audio range from 20 Hz. to 20 KHz. The largest speaker, known as woofer, handles low frequency spectrum whereas the smallest speaker, known as Tweeter, playback all the middle and high frequencies. In superior speaker systems a third speaker is also added, known as Mid-Range/Squawker, which exclusively handles the mid-range of the audio spectrum.

Distortion

The distortion in simple understanding is the deformity in the shape or character of an object. Usually a distortion occurs because of overloading of a system. For example, a sheet metal table capable of taking a 50 kg is loaded with more than 50 kg then it would bend under the excess weight. The table's shape would distort, which is a deviation from the original shape. Similarly, the electronic devices such as microphones, recorders, amplifier etc, can handle audio signal up to certain level (threshold) but if the signal level exceeds this limit then the audio signal being recorded/playback would get deformed i.e. distorted.

The “*VU metre*” as discussed before, is used to control the audio level to avoid any distortion in the sound being recorded. If, the pointer (needle) moves into the red zone i.e. beyond 0 dB level, then signal would start distorting. Therefore, VU metre is a very useful device to control the quality of an audio recording. At the time of recording one needs to adjust the recording level so that the pointer only occasionally enters into the red zone. The two kinds of distortions are introduced by the audio devices known as “*amplitudedistortion*” and “*frequencydistortion*”.

The **amplitude distortion** occurs due to overloading of the electronics because of high level of input signal as described above.



Title-Fig. 1.6 Tkop;phe Amplitude Distortion

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Source-

Link- http://www.qooljaq.com/AmplifierDistortion.htm

A typical amplitude distortion can be seen as a clipping (cutting/chopping) of the peaks of the sound waves as depicted above. Beyond a certain threshold level, the audio device is unable to handle (amplify) the input signal and therefore distorts its shape.

The **frequency distortion** happens due to imperfect/limited frequency response of the audio device. Different frequencies in an audio signal are amplified unequally thus leading to change in the tonal quality of the sound (also known as colouring of sound).

In professional quality audio recordings, the distortion level is maintained below 1%. Though quite often subtle distortions may not be easily comprehendible to general listener but high level distortions may make communication unclear and unpleasant. Therefore, to have an effective communication through multimedia content, the distortion in audio must be kept to the minimum possible.

Noise

Any unwanted sound may be categorised as *noise*. It means, irrespective of the quality of the sound, the question is whether a sound is desired or not. For example, if two persons are in conversation and some other sounds are also present in area disturbing their conversation, then these disturbing sounds may be referred to as noise. One would like to keep the noise at minimum by adopting different mechanisms to do so. For example, one may simply move away from the source of noise.

In another case, if a presenter standing in the market is describing a story related to the market then the noise around him in the market is desired to have a feeling of the location being shown. In this case, this noise in the market becomes a desired one and this sound effect is known as *ambience*. However you need to avoid too much of ambient sound that may disturb the main voice making it difficult to comprehend.

The electronic devices used for recording and playback of the sound exhibit an inherent electronic noise are known as "*mush*". This noise is usually represented as "*noisefigure*" or "*signaltonoiseratio*" (S/N). The unit of noise measurement is *decibel* (db.). Higher the signal to noise ratio batter is the equipment.

Similarly, the magnetic tape/cassette recording systems generate a noise known as "hiss". This noise originates from the tape material. With the introduction of the digital recording systems, this kind of noise from recording media (tape, CD etc.) has been nearly eliminated.

At the time of recording, the noise is kept to minimum by keeping the recording level around 0 decibels. If the recording level is quite low then the noise of the system would become audible and interfere with the communication capability of the recorded sound. Best noise rejection should be achieved at the time of recording else it would be very difficult to deal with during post-production.

The louder sound tends to suppress weaker sound and this phenomenon is called *masking*. The masking technique is often used during post-production to suppress the noise. For example, if there is some noise in the recording then, as a rescue measure, it may be masked by mixing background music at a level higher than noise.

Reverberation

The *reverberation*, also called "*reverb*" in short, is the quality of the space in which a sound is being produced, captured or playback. It is basically the persistence of sound after the sound source has ceased to generate. Reverberation occurs due to multiple reflections of sound in a closed space (e.g. room, hall). To understand it better let us take an example of clapping in a large size hall. If you clap once and listen to the sound of the clap, you will notice that the sound does not vanish (fade away) immediately. It is because the sound keeps reflecting from the hard surfaces of the walls and it takes some definite time to completely fade away.

The time period that sound takes to completely fade out is known as *reverberation time (RT)*. The r**everberation time** is defined as the *length of time* required for sound to decay 60 decibels from its initial level. Larger the room size higher is the reverberation time. You might have noticed that it is difficult to hold a conversation in a big empty hall. The large reverberation time of hall makes the speech difficult to understand. It implies that a short reverberation time is desired for the speech and generally a RT of around 0.4 seconds is preferred.

Another phenomenon we observe in our daily life is our tendency to sing in a bathroom. Even though the size of the bathroom is quite small but because of the hard tiles (highly reflective surface) on the walls of the bathroom, the sound keep reflecting without getting absorbed by the walls. Longer reflection sustenance results in longer reservation time which is most desired for singing and music. Quite often, reverberation times of longer than 0.5 seconds are preferred for song and music recording.

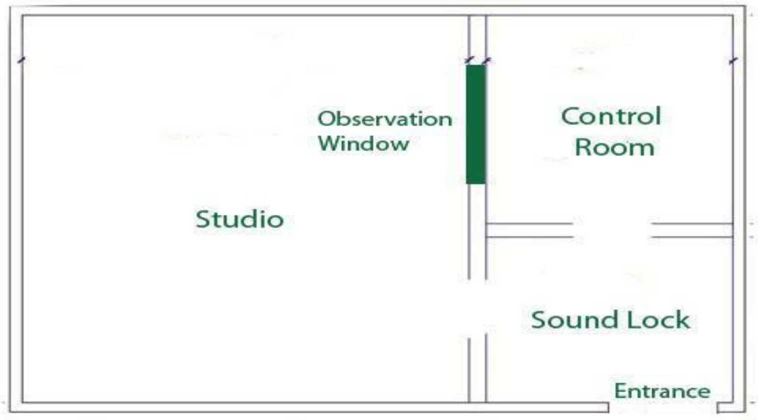
Therefore it may be concluded that to have clear verbal communication a short reverberation time is preferred while a longer reverberation is more suitable for song and music recording. The reverberation quality of a space can be controlled to certain extent by introducing soft materials in the recording space for higher absorption of sound. For example, the curtains, carpets, sofas, furniture and the humans help in reducing the reverberation time by way of more sound absorption. The sound studios are generally designed keeping in mind the intended purpose of the studio i.e. whether studio would be used more often for the speech recording or the music recording

In case of reverberation the reflected waves reach our ears so quickly that we are mostly not able to hear it as a separate sound from the direct sound. The reflected sound mixes well with the original (direct) sound. But, if the sound takes very long time to reflect due to large distance it has to travel e.g. sound in a valley of hills, then the reflected sound can be heard as a separate sound. If you shout hello then valley would reflect the same hello after some time. This effect of space on sound is called "*echo*".

Note: During post production/editing the reverberation time can be increased or echo can be added electronically.

The Audio Studio

Audio studio not only offers a space suitable for the performers but it is also equipped with lot of audio equipment used for sound capturing, processing, monitoring and recording. A typical sound studio would comprise of at least two major areas where one area called "*studio*" is an acoustically treated area created for the artists to perform and the other area, adjacent to the studio, is known as the "*control room*". Both areas are generally acoustically treated for controlling the reverberation time. The common brick wall between the two rooms usually has an observation window from where the audio professionals and the performers can see and signal each other. The diagram below shows a simple audio Studio setup.



Title-Fig. 1.7 A Typical Sound Studio Layout Diagram

The Audio Equipment

The studio area is well isolated from the outside world and the control room through sound proof doors and observation window to keep it noise free. The control room is installed with a range of audio equipment as listed below. Generally Control room is the place where most of the equipment along with the technical and production professionals is accommodated

- Microphones
- Audio mixer
- Audio processors
- Audio recorders and Audio workstations

These are the basic building units of an audio studio. The list is not exhaustive and varies from studio to studio. In order to make use of the equipment to our advantage an understanding of them would help us in making their optimum use. Let us understand them one by one.

Microphone

In the simplest form, the *microphone* is a device that converts the sound into electrical signal known as *audiosignal*. Even though some of you may not have visited an audio studio or a video studio but still you might have used the microphone fitted into your mobile phone. Whatever you speak into the mobile is converted into electrical signals by the microphone and after required processing it is sent to the receiving phone of the distant person where the audio signal is converted back into the sound through a speaker or an earphone. Similarly, in the sound studio, an artist/performer speaks in front of a microphone and his voice is converted into audio signal and recorded.

All microphones are not the same. They come in different shapes and sizes for different kind of applications. They can be classified into different categories as below

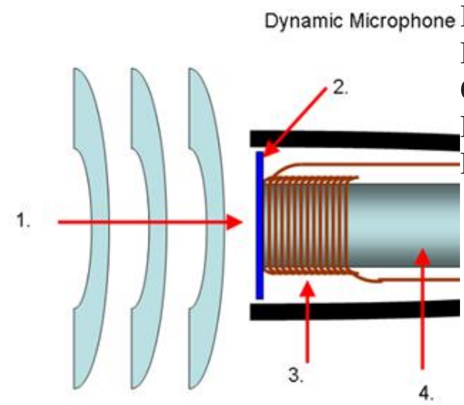
- The technology used i.e. their electrical characteristics,
- The way they pick sound around them i.e. Polar characteristics and
- According to their application i.e. the way they are used.
- The same microphone may get different names in different situations and applications.

Microphones - Electrical Characteristics

There are many microphone design technologies but for professional applications primarily two types of microphones would find use in our multimedia productions. These microphones are – the *Dynamicmicrophone* and the *Condensermicrophone*.

Dynamic Microphone

Any microphone would have a diaphragm which vibrates when sound waves fall on it. Microphones differ from each other in the way the movement of the diaphragm is converted into electrical signal. Dynamic microphone uses the same dynamic principle as in a loudspeaker, only reversed. A small movable coil made of thin conducting wire positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters the microphone, the sound waves move/vibrate the diaphragm.



Title-Fig. 1.8 A Dynamic Microphone

Attribution- Banco~commonswiki

Source- wikimedia.org

Link- https://commons.wikimedia.org/wiki/File:Mic-dynamic.PNG

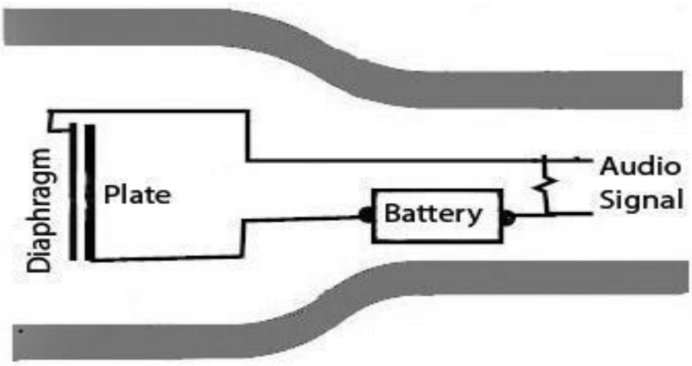
When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction.

The current generated varies in strength in accordance to the wave shape of the sound falling on the microphone diaphragm. You might have noticed that this kind of microphone generates audio signal without requiring any power supply or the batteries. The dynamic microphone has the following primary features:

- It is quite robust in operation.
- Does not require any power supply to operate
- Low sensitivity – captures loud sounds without distortion.
- The output level ranges from - 60 dB to -70 dB.

Condenser Microphone

While the dynamic microphone is preferred for the stage performances due to its ruggedness, the condenser microphone is preferred in the studio due to higher sensitivity and audio quality. The condenser microphone is also called a *capacitormicrophone* or the *electrostaticmicrophone*. The name condenser or capacitor comes from a component called condenser/capacitor which comprises of two parallel metallic plates separated by a medium e.g. air to store electric charge. The capacitance of these plates is inversely proportional to the distance between them.



Title- Fig. 1.9 A Condenser Microphone

Attribution- *Harumphy*

Source- Harumphy at en.wikipedia

Link- https://commons.wikimedia.org/wiki/File:AKG_C451B.jpg

In case of condenser microphone, when sound waves strike the diaphragm they change the distance between the diaphragm and the plate causing a current to flow through a battery powered circuit in proportion to the sound signal. The current variations become an audio signal. Some of the primary features of a condenser microphone are:

- Superior audio quality
- Higher sensitivity
- Low noise
- Must be handled carefully
- Need power supply to operate

As you know the condenser microphone needs a power supply for it to function, there are two possibilities of powering it. A good number of condenser microphones have a provision for installing small battery in the microphone body itself. The battery is usually an AA size cell also known as *pencilcell*. This kind of internal battery is quite useful when the microphone has to be used in external environment directly connecting to the camera or recorder. There is a possibility for connecting an external power source to the microphone. Such a power supply is known as "*Phantompower*". In the case of studio recording, the microphone is usually connected to an audio mixer which may have a provision for powering the microphone. Audio mixers with provision for phantom power obviate the need for battery replacement of the condenser microphone. The phantom power supply is usually of +48 volts.

In addition to the dynamic and the condenser microphones described above there are number of other kind of microphones using different technologies in their design. Since these microphones may not be often used for our purpose, we only list them here for reference.

Ribbon Microphone – It usually has a corrugated metal ribbon suspended in a magnetic field. Ribbon microphone doesn't require power supply but is quite fragile.

Carbon Microphone- It uses a capsule or button containing carbon granules pressed between two metal plates. Have extremely low-quality sound reproduction and a very limited frequency response range but is very robust.

Piezoelectric Microphone - Uses the phenomenon of Piezoelectricity, which is the ability of some materials. This Microphone utilises the sound pressure waves to produce an audio signal.

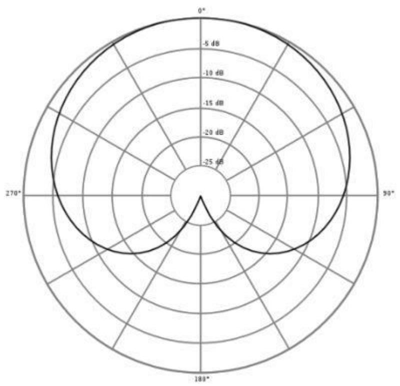
Fibre optic Microphone - A Fibre optic microphone converts acoustic waves into electrical signals by sensing changes in light intensity. Fibre optic microphones are robust, resistant to environmental changes in heat and moisture,

MEMS – Micro Electrical-Mechanical System (MEMS) microphone is also called a microphone chip or silicon microphone. Most MEMS microphones are variants of the condenser microphone design.

Loud Speaker as a microphone - Since a conventional loud speaker is constructed much like a dynamic microphone (with a diaphragm, coil and a magnet), the speaker can actually be used as microphone if instead of sending current to it, it picks sound and current is taken from it. Speakers are sometimes used as microphones in applications where high quality and sensitivity are not needed such as stage shows in villages.

Microphones - Polar Characteristics

So far we have studied that microphones are classified on the basis of technology employed in their design. The technology greatly influences the quality of the microphone but the design also takes care of the way a microphone would capture the sound from its surrounding. A microphone may be able to pick up sounds only from a particular direction and reject other sounds coming from other directions. The microphone directionality helps in achieving a good quality sound recording by rejecting the undesired sound coming from other directions. It helps in controlling the ambient noise and at the same time the revelation that adds to the voice.



Title- Fig. 1.10 A Cardioid Microphone

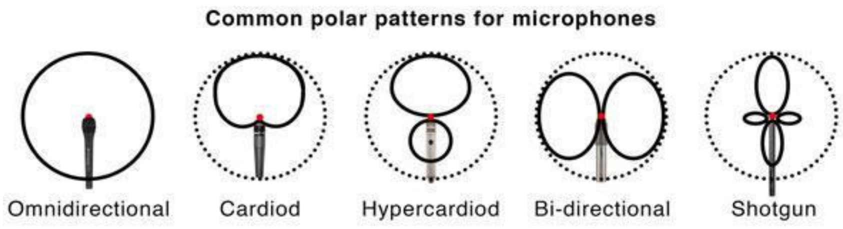
Attribution- Nicoguaro

Source- wikimedia.org

Link-https://commons.wikimedia.org/wiki/File:Polar_pattern_cardioid.svg

The microphone directionality is also known as its *polarpattern*. The polar pattern of a microphone shows the sensitivity of the microphone relative to the direction or angle from which the sound arrives.

The most common types of directionality are omni directional, bidirectional and Unidirectional. The diagram below illustrates a number of polar patterns. The microphone faces upwards in each diagram.



Title- Fig. 1.11 Microphone Polar Patterns

Attribution- jay goodman

Source- adorama.com

Link- https://www.adorama.com/alc/8502/article/recording-interview-audio-all-budgets

Omni Directional Microphone

An *omnidirectional* (or non-directional) microphone's response is generally considered to be a perfect sphere in three dimensions as shown. But in the real world, this is not the case as the body of microphone obstructs sound from rear side from reaching its diaphragm.

Bi-directional Microphone

"Figure of 8 (eight)" or *bi-directional* microphone receives sound equally from both the front and back of the element. Most ribbon microphones are of this pattern.

Unidirectional microphone

Unidirectional microphones can further be subdivided into

- Cardioid microphone.
- Hyper cardioid microphone.
- Super cardioid microphone.

All these microphones have highly directional characteristics with a difference in their area of coverage in a particular direction. The Super cardioid microphones, also called *gunmicrophones*, are mainly used for picking up the sound from the distance. Such microphones find good application in television program production where the microphone is used with a boom rod and kept out of the camera frame.

Microphones - as per application

Irrespective of the technology used in the making of microphone and its directional pattern, the microphones are also named according to their use. The microphone held by hand is known as *handheldmicrophone* and the same microphone can be fitted on a stand, through a microphone adaptor, to make it a stand mike. Same microphone when attached to a boom rod is called a *boommike*.

There are some small microphones which can be clipped on to the clothing near the neck for good pickup of sound and even giving the freedom of face movement. Such microphone is called *liveliermicrophone* or *lapelmicrophone*.

Title-Fig. 1.12 A Livelier / Lapel Microphone

Attribution- jay goodman

Source- adorama.com

Link- https://www.adorama.com/alc/8502/article/recording-interview-audio-all-budgets

Any microphone which has the capability to connect through a radio frequency link instead of a cable/wire is known as *wireless* or *RF* microphone. Such microphones usually have a small transmitter attached to the microphone. In some cases the transmitter is built in the body of the microphone. A distant receiver receives the radio signal that carries sound and after separating audio from radio signal, sends audio signal to the recorder. These microphones are very suitable in the situations where are cable is either to be avoided or cannot be laid. RF microphones provide complete mobility to the person carrying it and particularly the lapel microphones are easy to hide.

How to use microphone?

In our discussion, it is apparent that condenser microphones are highly sensitive and superior in quality of sound as compared to dynamic microphone. That is why condenser microphones are expensive and mostly used in the studios. You may be tempted to use condenser mike for good quality sound but actually the choice of the microphone is dictated by the situation in which a recording is to be performed.

Due to their robustness and relatively low sensitivity, the dynamic microphones are preferred in the situations where harsh environmental conditions exist. These microphones can be easily placed close to the sound sources without the fear of overloading or failure.

Dynamic microphones are good at feedback rejection in the public address applications. The feedback usually occurs when sound from nearby loudspeakers reach back to the microphone and generate a whistling sound. The feedback can be eliminated either by turning down the volume control or by increasing the distance between loudspeakers and the microphone. Another technique to keep the feedback low is by holding microphone too close to the lips so that volume control need not be turned up too much.

You might have witnessed singing competition TV shows having large studio audience and the judges using public address system for monitoring. The singers keeping the microphone too close to their lips to avoid feedback but this is possible only if the microphone has a low sensitivity. In such a situation, the dynamic microphone becomes the first choice. If condenser mike is to be used then enough damping (cutting sound level) needs to be added. Dynamic microphones are more frequently used for audio only programs where there is no need to hide the microphone and it can be used conveniently in close proximities.

Generally, dynamic microphone will not be the right choice if it cannot be placed within 12 inches from the sound source. Then condenser microphone becomes the choice as it is capable of picking up sounds from a distance due to its high sensitivity. The shotgun microphone, because of their narrow pickup pattern (high directivity), is a good choice for noise rejection. Condenser microphone (Shotgun) quite frequently finds application as a boom microphone.

In outdoor recording, the wind noise becomes an issue in sound recording. When the blowing wind strikes the diaphragm of the microphone, it creates a low frequency noise. In such situations the windscreen, a piece of moulded foam, is placed on the microphone.

Title-Fig. 1.13 A Windshield Microphone

Attribution- jay goodman

Source- adorama.com

Link- https://www.adorama.com/alc/8502/article/recording-interview-audio-all-budgets

The windshield breaks the speed of the air reaching the diaphragm but at the same time, due to its perforations, it permits the sound to reach the diaphragm without much resistance.

If you don't find a windshield available at the location then you may wrap a handkerchief over the microphone for similar effect.

The wind noise can also be minimised by changing the microphone direction to avoid facing the direct air.

The wind screen also helps in reducing the “pop sound” generating due to the air gushing out of the mouth of the presenter when a microphone is placed too close to the lips.

Multi microphone recording

When many people are to be recorded together, in some situations, an omni directional microphone may become the choice as it receives sounds from all the directions. Inside a studio the persons can be placed around and near to the Omni microphone and a reasonably good audio can be recorded. If the

Omni microphone have to be placed at a large distance from the persons it may start sounding hollow, The hollowness is the degradation in the sound quality due to excessive reverberation (high reflected sound) reaching the microphone in comparison to the direct sound from the source. The richness in the audio is lost due to hollowness and it sounds to be coming from a distance.

A bi-directional microphone would allow dividing all the persons into two groups on both sides of the microphone. The dead sides of the microphone would not pick up any reflected sound thus reducing the hollowness.

An alternative choice would be to use many unidirectional microphones and provide separate microphone for each individual. The situation would allow each microphone to be placed near to the resource person avoiding the hollowness in sound. However, it is not as simple as it appears to be. For example, if all the persons are on one side of the table and sitting close to each other. Then each microphone would also receive the sound from the adjacent resource persons. This delayed sound coming from a distant person would again introduce hollowness. The best technique would be to arrange the seating arrangement so that each microphone would receive only the sound from its main source and all other persons are either away from it or they are on the dead side (behind the mike) of the microphone. Use of shotgun mike is another possibility.

Audio mixer

The simplest kind of audio recording system would consist of a microphone connected directly to a recorder. This kind of system works best for simple outdoor recordings when either single presenter is speaking or two persons are in conversation. In the studio, usually the system is so arrange that several persons can participate together in a performance. You need to have several microphones for different persons and instruments. Also you may like to take some piece of music or speech from previously recorded source. An audio device known as *audiomixer* accepts inputs from all the microphones and audio devices and mixes them down for feeding to a recorder. The mixer has several controls on its surface as shown in the

image.

Title- Fig. 1.14 An Audio Mixer

Attribution- Biggerbyfar~commonswiki

Source- wikimedia.org

Link- https://commons.wikimedia.org/wiki/File:Audio_mixer_faders.jpg

The audio mixers are capable of receiving mainly three types of input signals namely Line, Mic (Microphone) and Aux (Auxiliary). The input signal handling capacity of a typical mixer may be as:

Mic	:	-40 db. to -70 db.
Line	:	0 db. And +4 db.
Aux	:	-10 db. to -30 db.

The professional audio devices provide line output signal and consumer grade audio devices more often are able to offer an Aux audio output. The microphones are connected to the mixer through the XLR connectors marked as mic inputs. The number of inputs an audio mixer can handle will depend upon the number of channels available on the mixer. Each channel consists of at least one input connector, gain control and the channel fader for controlling the audio level of that channel. In the image you can see there are 8 white faders which mean it is an 8 channel audio mixer. The output of all the channels goes to the master faders who control the final mixed audio level of the program and this mixed output of the mixer is connected to a recorder through the XLR or RCA output connectors.

Generally, the mixers are also equipped with equalizers for changing the frequency response of the mixer for that particular channel and hence the tonal quality of the sound being mixed.

Generally three controls termed as “Bass”, “Mid” and “Treble” allow cutting down or boosting the related frequencies by about 10db to 15 db.

Some mixers may also have a Phantom power supply provisions at the inputs for powering the condenser microphones. The supply is marked as +48V and is often associated with a corresponding on/off switch.

Audio processors

Audio signal processing or *audio processing* is the intentional alteration of audio signals often through an audio effect unit. The processers are used to introduce echo in recording. Echo is used to simulate the effect of reverberation in a large hall. In this process, one or several delayed signals are added to the original signal. To be perceived as echo, the delay has to be of the order of 35 milliseconds or above.

In equalisers, different frequency bands are attenuated or boosted to produce desired frequency response. Moderate use of equalization (often abbreviated as "EQ") can be used to fine-tune the tone quality of a recording; extreme use of equalization, such as heavily cutting certain frequency can create more unusual effects.

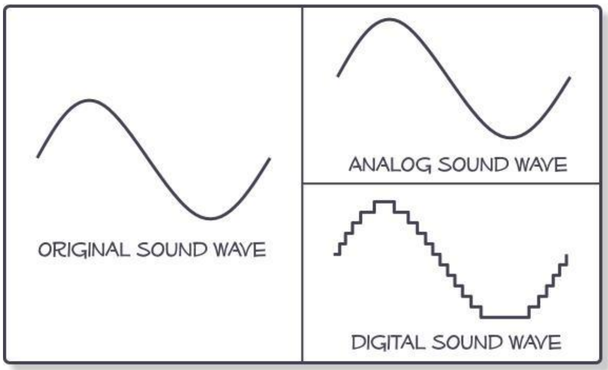
Some of other roles performed by the processes are chorus, phaser, flanger, compressors, filtering, overdrive, robotic voice, pitch shift, time stretching and modulation etc.

Audio Recording Device

Audio recording Technology has evolved greatly. It has made a transition from earlier magnetic tape/cassette recorders to today's Flash Memory based portable recorders. For more Complex audio works the computer based audio workstations are being used. Even today's smart phones have the capability to make a quality audio recording if attached to a good quality external microphone. The modern devices are based on digital technology and the earlier recorders used to be of analogue technology. In the case of analogue technology, the quality of the sound heavily depends upon the medium of recording e.g. Magnetic tape.

Digital Audio

Analogue (Analog) *sound* is the sound we hear and these sound waves are continuous in time and are analogous to another time varying signal. The digital signals are abruptly/sharply changing signals. In the case of digital technology, an analogue signal is taken as input and is converted into digital through a process of sampling and quantization. The digital signal is basically a kind of mathematical number which any computer can understand where the numbers are made up of only 0 and 1 digits. These mathematical numbers are unaffected by the medium of recording and can be easily manipulated using the computers which are good at processing the digital signals.



Title- Fig. 1.15 Analog Sound Wave

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Link- http://www.centerpointaudio.com/Analog-VS-Digital.aspx

We are accustomed to do mathematics using "decimal number system" comprising of 10 digits from 0 to 9. The prefix "deci" comes from the Latin decimus, meaning "tenth". With these 10 digits we are able to count any number and also do all sorts of mathematical computations. Similarly the digital electronics uses "Binary number system" consisting of only two digits i.e. 0 and 1. As we do with decimal system, the mathematical calculations can be done using only 0 and 1 using Boolean (scientist's name) algebra.

Most digital recorders accept analog audio signal and do the conversion to digital internally before recording in digital audio file format. The analogue to digital conversion (A/D conversion) is done through a process of "sampling" and "quantization". Instead of taking the whole waveform a number of samples are taken from the audio wave and at the time of playback these samples; (through approximation) form the basis for recreating the audio signal back to analogue signal.

The number of samples obtained per second is known as *samplingfrequency*. Higher the frequency better would be the signal quality. These samples are then converted into binary numbers (Quantisation) using a set of binary digits/numbers. The set of digits could be 8 bit or more. Higher the bit size better is the quality of audio. However, recording higher quality audio either by raising the sampling frequency or by using more bits, results in larger file size requiring more storage space to record.

Digital Audio File Formats

The digital audio is stored in various file formats which includes WAV and MP3 file formats.

Wav File Format

This format was developed to reduce the file sizes by compromising quality but in an efficient manner. *Wav* is the extension used for files containing digitized sound recorded in a professional sound recording system. Wav files are normally large in size depending on length, whether it's recorded in stereo or mono and the sampling rate used. Most of good quality recordings are done at sampling rate of 44.1 kHz or 48 KHz. using 16 bit or higher. Wave file name has a three digit extension represented as .wav.

MP3 file format

It is an open compression standard, designed for storing digital sound's suitable for the Internet. The *MP3* compression algorithm (formulas) analyses the digital file to eliminate sounds inaudible to the human ear. This reduces the file size by up to 90%. The filters can be adjusted to increase the compression with a decrease in both file size and quality. A good quality MP3 should be recorded at a bit rate of 192 Kbits/second or more. Most internet compatible audio content in mp3 format is available at 128 Kbps or less and is not really suitable for good quality music jobs. MP3 file name caries a three digit extension represented as .mp3.

Mono, Stereo and Surround Sound

Monaural or *monophonic* sound reproduction also called *monosound* is intended to be heard as if it were a single channel of sound coming from one direction. Any music system with single speaker may be termed as mono sound system. In the case of Mono sound, the sound reaches listener from a single direction.

In real life, we find sounds coming from all different directions, and to achieve that effect stereo system was developed. The stereo system comprises of two speakers as left and right source of sound. Stereo uses two channels to convey the impression of sound coming from different places i.e. from left, middle, and right.

The stereo system was unable to offer sounds coming from all directions i.e. from the front as well as the rear. To achieve more realism, the surround sound system was developed.

The *Surround System* comprises of minimum 4 sound sources (speakers), out of which 2 are placed in front and two at the rear position in comparison to the sitting position of the listener. Since two front speakers placed at two corners of the room leave a sound gap in the middle, a speaker was added in the centre making a total speakers count of 5. Because the channel speakers are of small to medium dimensions and are not able to handle very low frequency sounds in the range of 20 to 100 Hertz, a special speaker known as *subwoofer* was added to the system.

The sub-woofer does not carry any individual channel information but is used to provide the very low frequency content of all channels combined. The Sub-woofer is termed as 0.1 of the system thus upgrading a5 channel surround to 5.1 surround sound system. Further development in the technologies introduced more channel surround systems to bring higher level of realism in the sound reproduction. Various technologies like "*Dolby*" were developed to record multi-channel surround sounds on audio recorders capable of recording only two channels/tracks.

The recording techniques of stereo and surround sound are quite complex. In today's film industry, the sound recording technique has evolved to the level of sound designing where the films are made keeping in mind the special theatres (e.g. Dolby/DTS audio based theatres) capable of reproducing surround sound.

Unit summary

In this unit you learn about the basic nature of sound in terms of frequency and amplitude. The audio ingredients like frequency response, distortion, noise and reverberation primarily influence the quality of sound. Moreover the quality becomes a subjective matter because it depends on the quality of sound that a person is accustomed to hear in his routine life. Different methods of attaining the good audio quality recording are discussed. Understanding the technologies behind the microphone design helps in making a right selection of microphone in different recording situations. Various microphone placement techniques are also learnt. An overview of audio mixer and studio is also discussed. A basic knowledge about mono, stereo and surround system is obtained for generating interest in further learning.

Assignment

Can you use a sound recorded over a telephone call for your multimedia content? Find out the issues involved and methods to tackle them.

The audio can be recorded using the microphone mounted on a video camera but is that recorded sound suitable for professional works like your multimedia presentation? Find the reasons for the quality issues and how to overcome them?

Make a recording using sound coming from a loud speaker. What are the quality concerns?

Assessment

1. What is Audio?
2. What is Sound?
3. What is the importance of audio / sound quality in a multimedia work?
4. Explain the various types of microphones.
5. What is noise & distortion?
6. When and how you can use the "Proximity Effect" to your advantage?

7. You are recording in a big empty hall. Describe the expected problems and write how you will overcome these problems?

References and Further Readings

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Unit- 3 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.

Outcomes

- 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

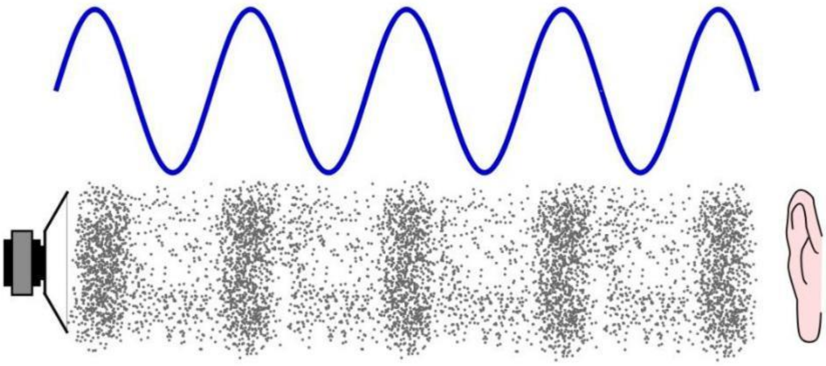
Terminology

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.



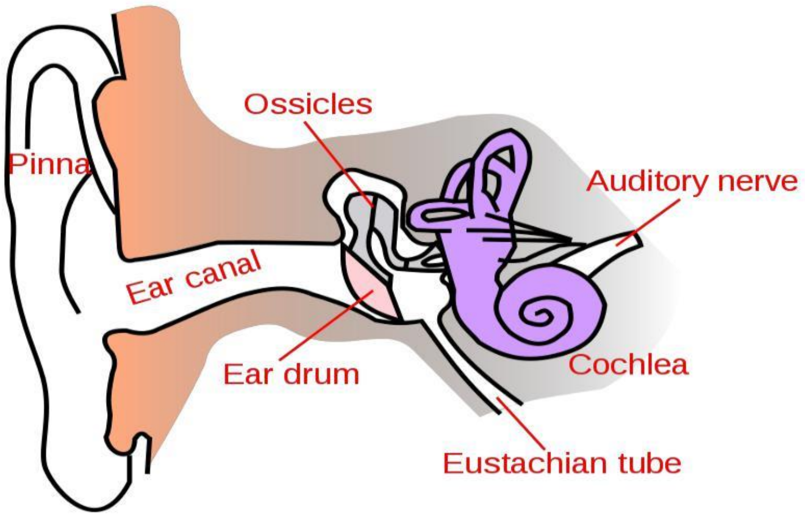
Title- Fig-1 Wave propagation

Attribution- Pluke

Source-

Link- 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 *hamme*r, *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 response 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.



Title- Figure-02: **Diagram showing the outer, middle and inner ear.**

Attribution- Iain at English Wikipedia, **SVG** conversion by User:Surachit

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Link- https://commons.wikimedia.org/wiki/File:Ear-anatomy-text-small-en.svg

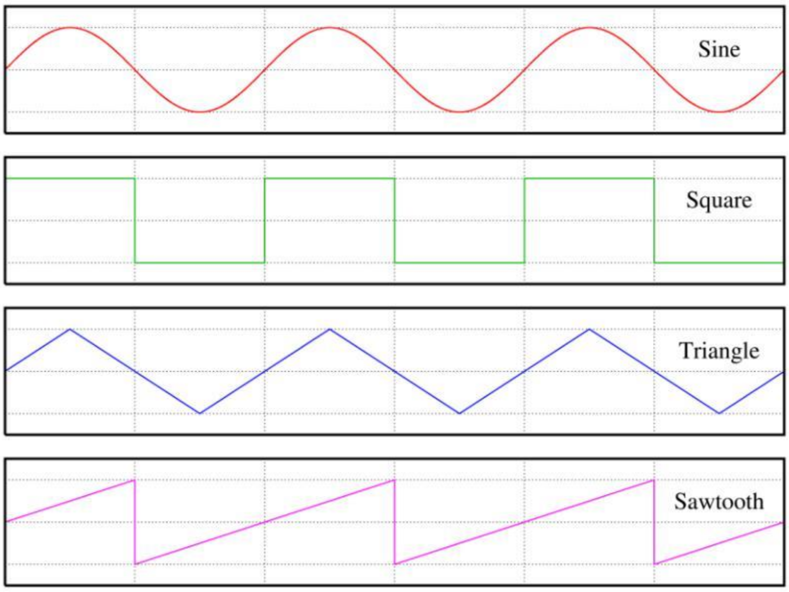
A convenient pressure level that produces the phenomenon of hearing is called threshold of hearing. The minimum Sound Pressure Level (SPL) that is required for hearing in most of the people is equal to 0.0002 microbar. One microbar is equal to one million atmospheric pressure. The SPL that causes discomfort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

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.



Title- Figure-03: Different wave forms

Attribution- Omegatron

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Link- https://commons.wikimedia.org/wiki/File:Waveforms.png

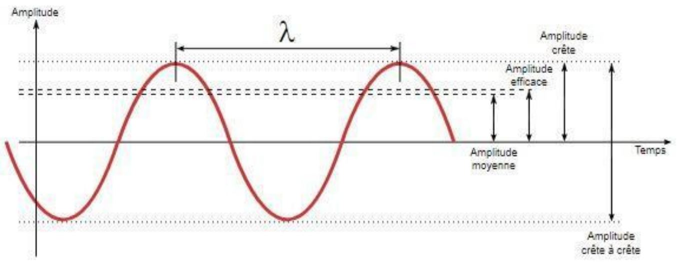
Characteristics of a Wave Form

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.



Title- Figure-04: Wave form

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https://upload.wikimedia.org/wikipedia/commons/e/ea/Sinus_amplitude.svg

(Courtesy: dba.med.sc.edu)

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 *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.

$$\lambda = v/f$$

Where λ 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 0^0 with zero amplitude and it can increase to a maximum of 90^0 and then decrease to having a zero amplitude at 180^0 and then increase to a maximum of 270^0 (in the negative direction) & finally come back to its original level at 360^0.

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? Arrange 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 andfunction 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. Name the unit of loudness
6. Frequency is measured in_______.
7. We measure the wave length in______.
8. What is the dynamic range of human ear?

Resources

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

Unit 4 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 toelectrical 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 micro phone is essential for everyone working in sound industry (Figure 07).

Title- Figure-07: Microphone

Attribution- Didgeman

Source-

Link- https://pixabay.com/en/microphone-mixer-cable-626032/

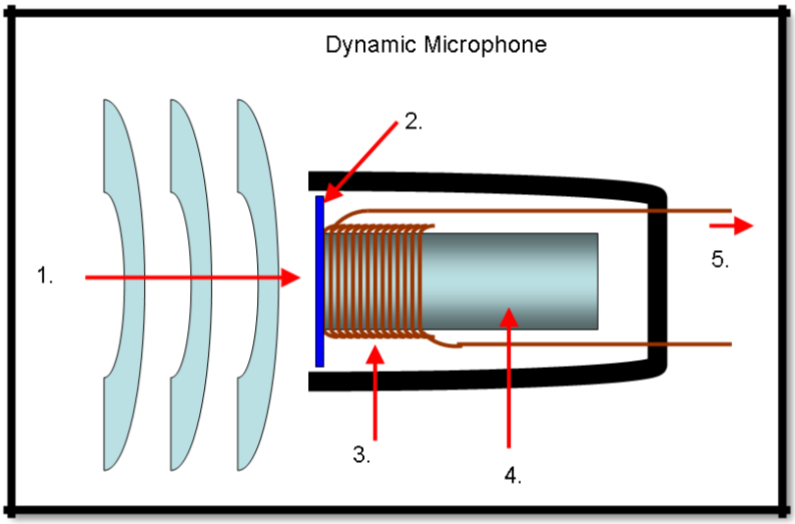
In order to deal with the wide range of musical, acoustic and situational circumstances, a large number of microphone types, styles and designs are available for a purposeful use. Before go for a recording let us know the microphone.

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.



1. incoming sound
2. diaphragm
3. Coil
4. permanent magnet
5. resulting signal

Title- Figure-08: Microphone Construction

Attribution- Banco

Source-

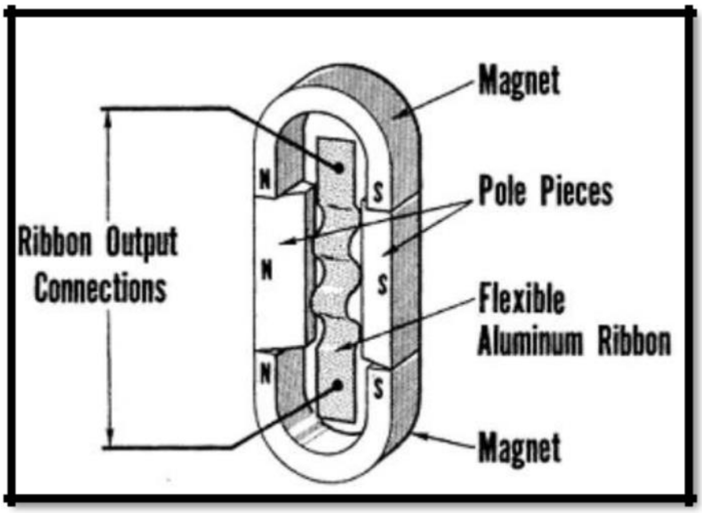
Link- https://commons.wikimedia.org/wiki/File:Mic-dynamic.PNG

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.

Title- Figure-09: Cut away details of ribbon microphone



Attribution- N.H. Crowhurst

Source-

Link- http://www.vias.org/crowhurstba/crowhurst_basic_audio_vol1_032.html

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

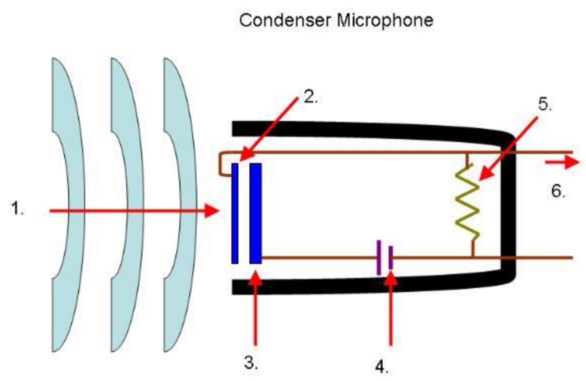
$$Q=CV$$

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.



1. Sound Waves
2. Diaphragm
3. Back Plate
4. Battery
5. Resistor
6. Audio Signal

Title- Figure-10: Cutaway of Condenser Microphone

Attribution- Banco

Source-

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 resister 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=Q/C. When the voltage across the capacitor changes, the voltage across the resistor will also changes. 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- Terodaktil

Source-

Link- https://commons.wikimedia.org/wiki/File:Lavalier_mikrofon.jpg

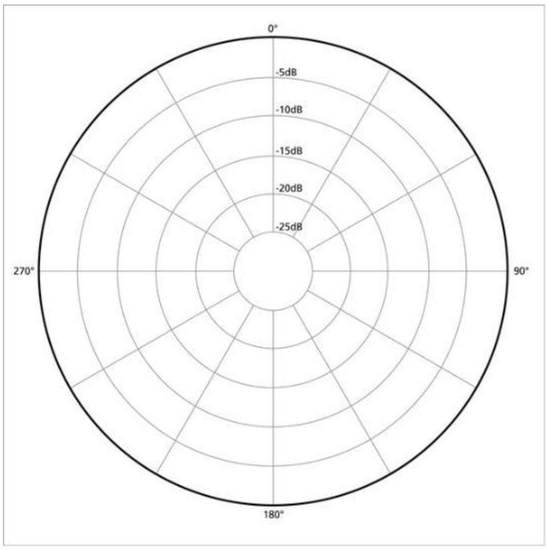
Characteristics of Microphone:

The microphones are of different types, styles and designs to fulfil 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 in to two categories.

- Omni directional polar pattern
- Unidirectional polar Pattern



Title- Figure 12: Directive response of Mic.

Attribution-Galak76

Source-

Link-

https://commons.wikimedia.org/wiki/File:Polar_pattern_omnidirectional.png

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 pattern 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

Attribution-akd835

Source-

Link- https://pixabay.com/en/singer-sillouette-concert-mic-1595864/

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)

Title- Figure 14: Stage show recording

Attribution-Thibault Trillet

Source-

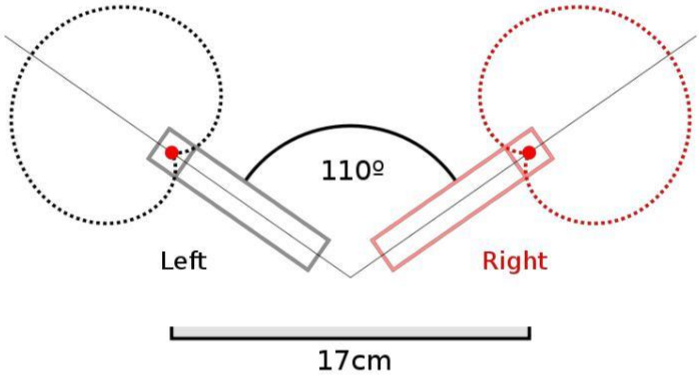
Link- https://www.pexels.com/photo/audience-band-concert-crowd-167636/

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.



Title- Figure 15: Stereo Mic placement

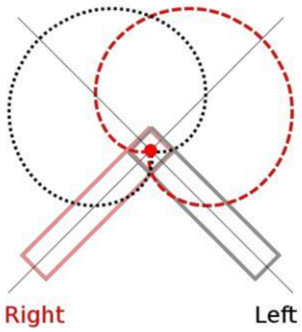
Attribution-Lainf

Source-

Link- 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.



Title- Figure 16: XY Placement of Mic.

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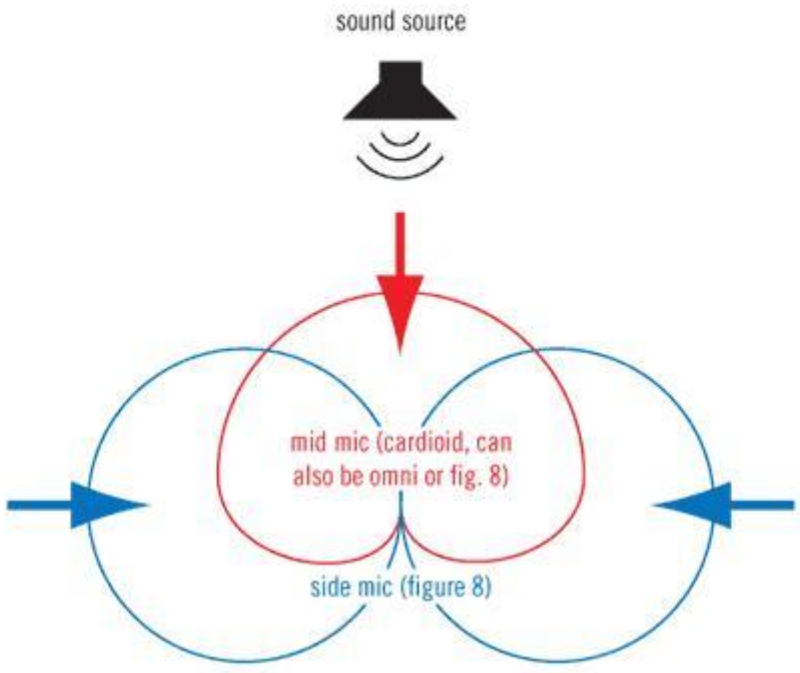
Source-

Link- https://commons.wikimedia.org/wiki/File:XY_stereo.svg

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.



Title- Fig-17 M-S Method mic placement.

Attribution-Daniel Keller

Source-

Link- https://www.uaudio.com/blog/mid-side-mic-recording/

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.

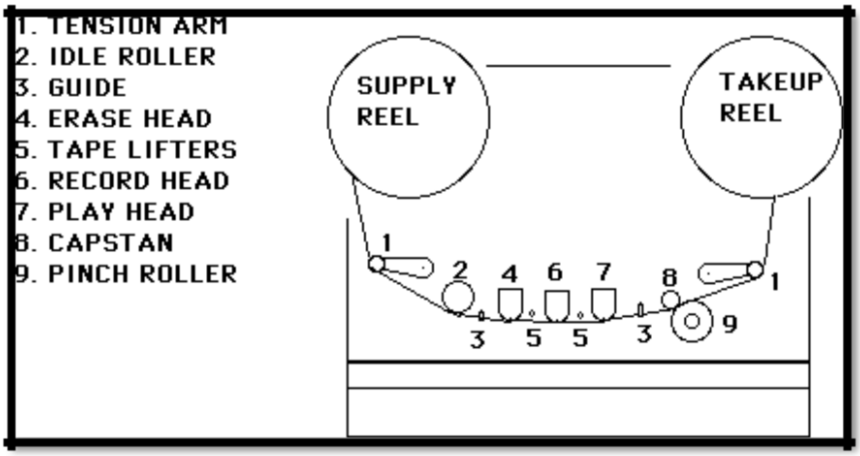
1. *If there is no mic output:* The possible causes are faulty cable and connector, plugged into wrong socket, not faded up on mixing desk. In case of an electrostatic microphone, the power may not be on.
2. *If the output is very low*: There may be incorrect setting on mixer.
3. *If the output is too 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.

Attribution-

Source-

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

Title- Fig-19 Analog Tape Recorder

Attribution- Erkaha

Source-

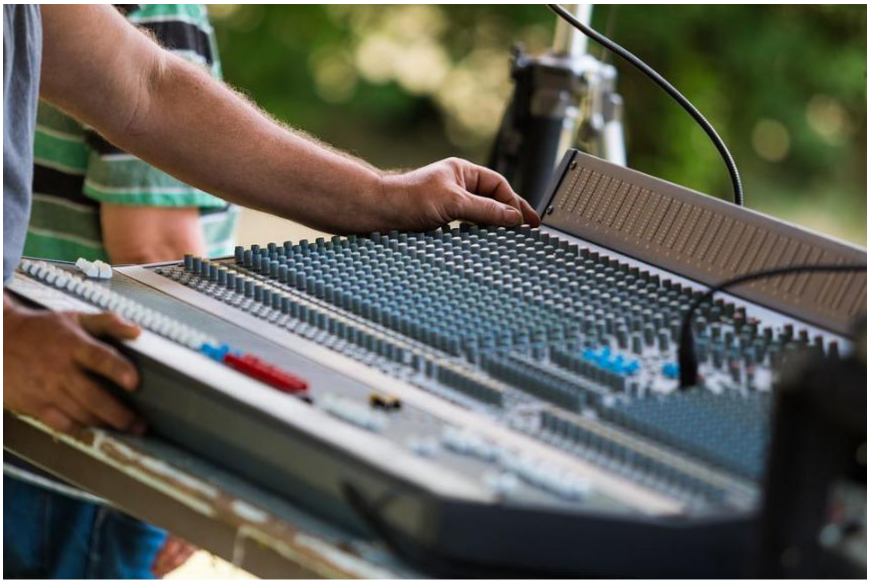
Link- https://commons.wikimedia.org/wiki/File:Tape_recorder_GX-6300.jpg

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.



Title- Fig-20 Audio Mixer

Attribution- JohnDILiberto

Source-

Link- https://pixabay.com/en/sound-mixer-mixing-board-1503092/

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.

Title- Fig-21 Audio Cable

Attribution- byrev

Source-

Link- https://pixabay.com/en/audio-black-box-cables-ebn-music-88202/

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.

Title- Fig-22 Audio Connectors

Attribution- Trude Bergheim Mikkelsen

Source-

Link- https://commons.wikimedia.org/wiki/File:Jack_mono_jack_stereo_xlr_m ale_and_female.JPG

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
- http://cemca.org.in/ckfinder/userfiles/files/7_Lesson-06_MICROPHONES.pdf
- http://artsites.ucsc.edu/ems/music/tech_background/te-20/teces_20.html
- https://en.wikipedia.org/wiki/Mixing_console
- http://downloads.izotope.com/guides/iZotope-Mixing-Guide-Principles-Tips-Techniques.pdf

Unit 5 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:

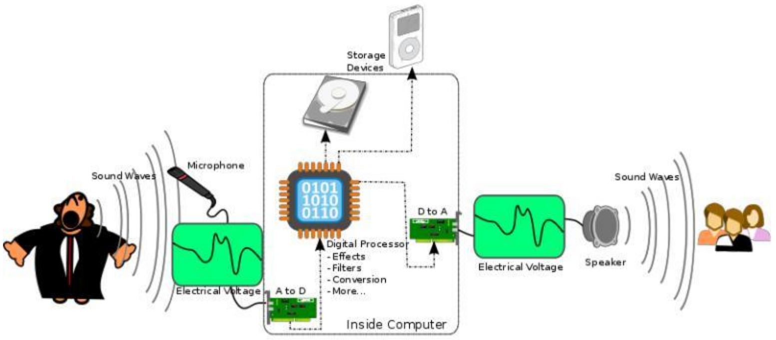
- Explain the Process of Sound Recording
- Layout the diagram of an analog recording setup.
- Describe the signal flow in a digital recording system.

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.



Title- Fig-1 Signal flow from mic to recorder

Attribution- Teeks99

Source-

Link- https://commons.wikimedia.org/wiki/File:A-D-A_Flow.svg

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 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 1/44100th 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 32bit.

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

Source-

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.

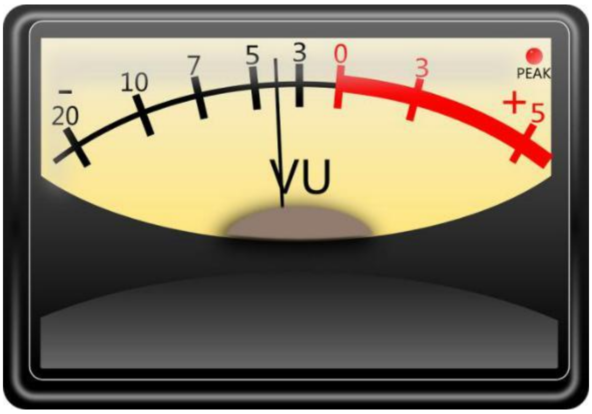
Mixing keep going on until the engineer gets the level of satisfaction. Balancing the live audio during a concert is a big task. Experiment on mixing can be done in a postproduction studio where as there is no chance for an engineer to do experiment in a live concert. So technical expertise, skill and 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.



Title- VU meter

Attribution-Openclipart

Source-

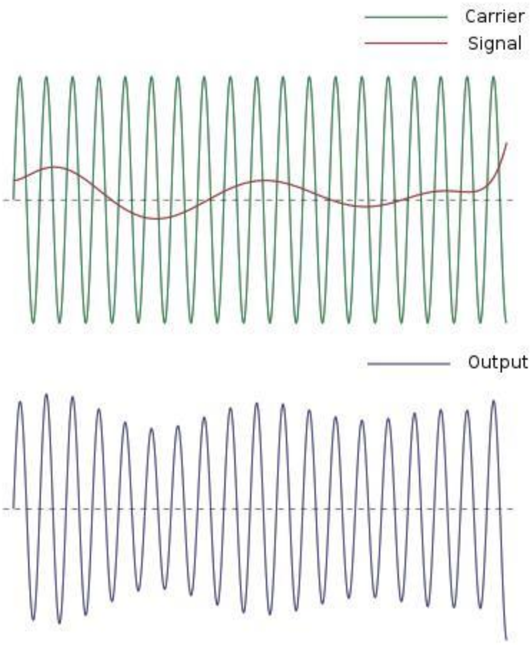
Link- https://pixabay.com/en/metering-electronic-measure-meter-156796/

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.



Title- **Amplitude Modulation**

Attribution- The.ever.kid

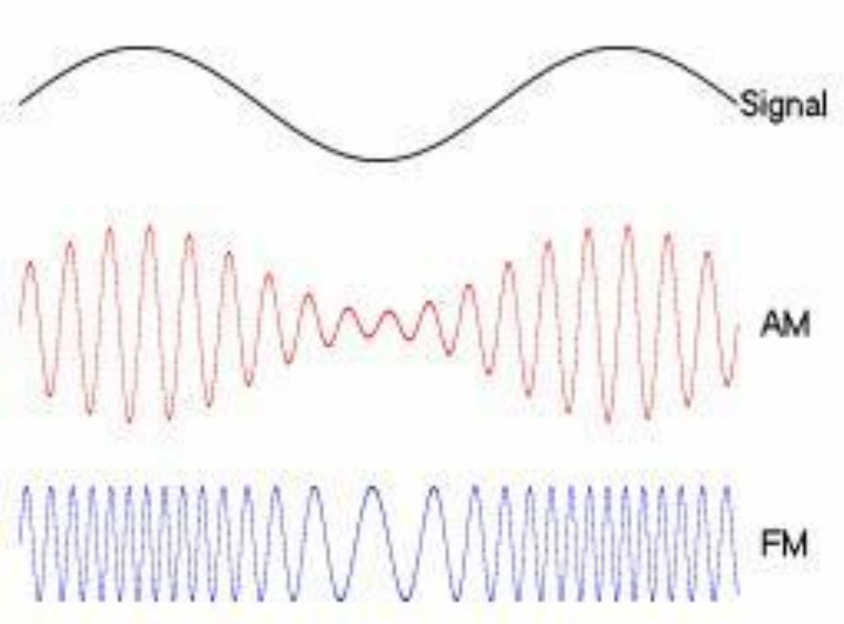
Source-

Link- https://commons.wikimedia.org/wiki/File:AM-Final-Wiki-TEK.svg

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.

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.



Title- Frequency Modulation

Attribution- Berserkerus

Source-

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

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

- Modern Recording Techniques , David Miles Hurber and Robert E.Runstein, 7th Edition, Focal Press
- 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/Digital_recording
- https://en.wikipedia.org/wiki/AM_broadcasting
- https://en.wikipedia.org/wiki/Frequency_modulation

Unit 6 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 A device to filter the noise.
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 ahead 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.

Title- Voice dubbing by vocal artist

Attribution- (Photograph taken by Author)

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

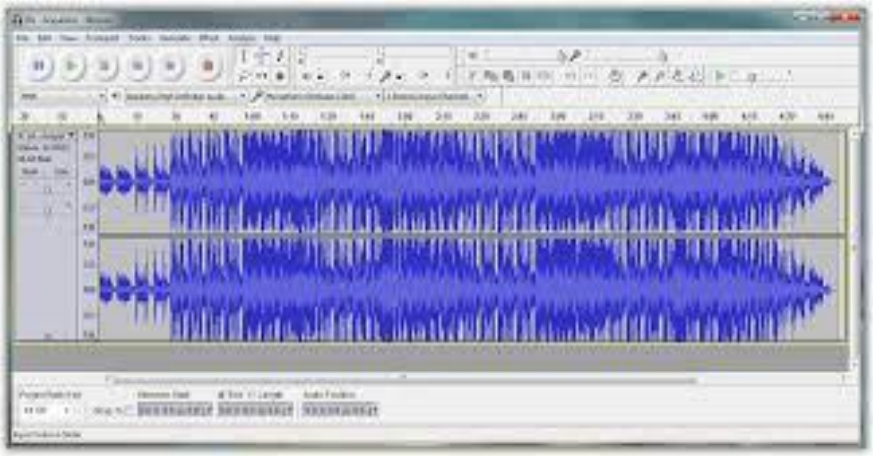
Editing in analog medium needs more attention. It needs both theoretical and practical knowledge. During this process, the engineer edit the original music track out from their reels and begins the process of splicing them together into a final sequence on a master reel set. Once this is done, the mix 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 cumbersomeprocess 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

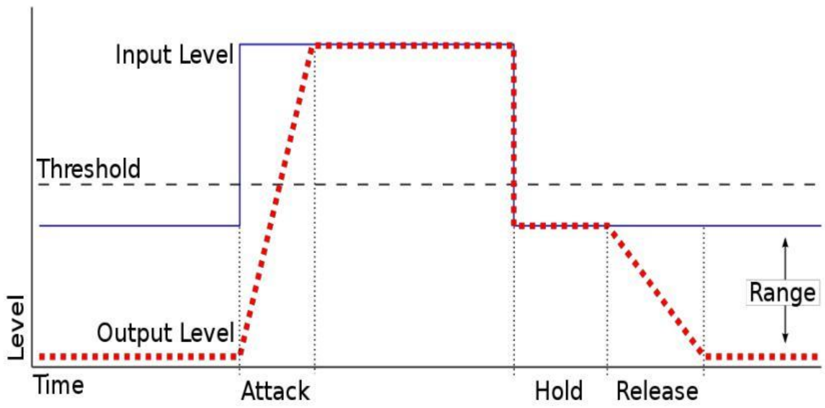
In digital signal processing, the noise can be reduced easily by the use of various noise filter plug-inns. The noises like tape hissing, hum, obtrusive background ambience, needle ticks, pop and certain types of distortion that are present in the original recording can be removed by using noise gate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



Title- Diagram of a Noise Gate

Attribution- Lainf

Source-

Link-https://commons.wikimedia.org/wiki/File:Noise_Gate_Attack_Hold_Rel ease.svg

Monitoring

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.

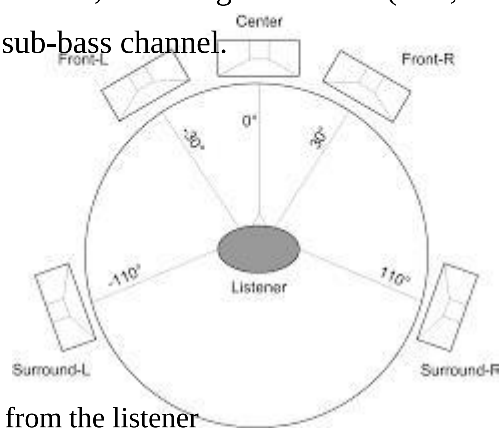
Stereo

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.



Title- Placement of speaker from the listener

Attribution- Kamina

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Link- https://commons.wikimedia.org/wiki/File:5-1-surround-sound.svg

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 refection 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.

Title- Headphones

Attribution- Photographs taken by Author

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 semi-conductor transistor design.

Title- Audio Amplifier

Attribution- Inspiredimages

Source-

Link- https://pixabay.com/en/stereo-vintage-audio-music-sound-883186/

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

- https://en.wikipedia.org/wiki/Voice-over_translation
- https://en.wikipedia.org/wiki/Sound_editor_(filmmaking)
- https://en.wikipedia.org/wiki/Noise_reduction

Unit 7 Sound Equipment

Introduction

This unit introduces you to the functionality of clapboards, various audio equipments and field monitors. After camera and lights these are the necessities for proper and systematic production. These devices have evolved during course of time and change of technology.

Outcomes

Upon completion of this unit you will be able to:

- Identify different sound/audio equipments.
- Plan for audio recording.
- Choose different kind of field monitors

Terminology

Clapboard	Clapboard is a hinged slate used for proper marking and description of the visuals.
Time Code	Time Code is the digital marking of the tape or celluloid depending upon the length of time it is shot.
Microphone	Microphone is an electromagnetic device used to convert sound energy to electrical energy.

Sound Equipment

In the field of video production or filmmaking the role of sound or audio is as important as video or visuals. There are different kinds of sound around us which we come across in our daily life. Like the camera is used to video images, special instruments called microphone are used to record these sounds. Though innovation changes every single year, one thing is always constant: the sound is similarly as vital as the visual. Regardless of how inventive and highly executed the video part of the production might be, and wherever it may be posted on YouTube, copied to a DVD or shows up on a major, small or portable screen, the watcher's experience can be absolutely destroyed by dull sound. A more noteworthy degree than a great many people acknowledge, audio could "make or break" any video or movie venture.

In the realm of expert sound for-video and sound for-movie, The first thing, is the catch, recording, and playback procedures are normally taken care of by various bits of hardware. Secondly, there is an extra transitional stage in which the sound is altered to dispose of errors or fit a specific time limitation, handled to improve quality of the audio, and organized to match a specific deployment medium, for example, online video or DVD.

In the following section we will come across the equipment which are used for recording and reproduction of sound.

Microphones

Microphones or mic are the device used to catch sound waves and amplify them.An exceptionally fundamental need is filled by the microphones i.e.: conversion of acoustic energy (sound) to electrical energy by the electromagnetic process. The sound waves are converted into an audio signal by them such that the product can be recorded, edited, deployed, and amplified for playback. As the microphone's function is that general, that one mayinquire as to why there are such a large number of various types of microphonesThis is on the grounds that a few sorts of amplifiers are more qualified to specific uses than others, similarly as a few cameras are more qualified for use on a tripod in a sufficiently bright studio whereas others are better for handheld use with accessible light. Some microphones are used to record voice while some are used to capture the ambient sounds. If one is familiar with the different types of microphones then one can use them to get the best effect, the productions will start sounding more professional. The microphones are mainly categorized upon their directionality i.e. in which the sound waves they can capture.

In order to speak to larger groups of people, a need arose to increase the volume of the human voice. The earliest devices used to achieve this were acoustic megaphones. Some of the first examples, from fifth century BC Greece, were theatre masks with horn-shaped mouth openings that acoustically amplified the voice of actors in amphitheatres. In 1665, the English physicist Robert Hooke was the first to experiment with a medium other than air with the invention of the "lovers' telephone" made of stretched wire with a cup attached at each end.

German inventor Johann Philipp Reis designed an early sound transmitter that used a metallic strip attached to a vibrating membrane that would produce intermittent current. Better results were achieved with the "liquid transmitter" design in Scottish-American Alexander Graham Bell's telephone of 1876 – the diaphragm was attached to a conductive rod in an acid solution.These systems, however, gave a very poor sound quality. David Edward Hughes invented a carbon microphone in the 1870s.

The first microphone that enabled proper voice telephony was the (loose-contact) carbon microphone. This was independently developed by David Edward Hughes in England and Emile Berliner and Thomas Edison in the US. Thomas Edison refined the carbon microphone into his carbon-button transmitter of 1886. This microphone was employed at the first ever radio broadcast, a performance at the New York Metropolitan Opera House in 1910.

In 1923, the ribbon microphone was introduced which was another type of electromagnetic microphone, believed to have been developed by Harry F. Olson. It was a kind of reverse-engineered version of a ribbon speaker. These microphones were further developed by several companies but they were kind of raw microphones which would catch all the sound near them.RCA made advancements in pattern control of the diaphragm giving directionality to the microphones. With television and film technology booming there was demand for high fidelity microphones and greater directionality. Electro-Voice responded with their Academy Award-winning shotgun microphone in 1963.

Primarily microphones can be categorised as per the target directions used while recording or covering.

Omni-directional

By "Omni" we means "all", this suggests that a microphone with Omnidirectional feature so it can grab sound from all directions equally. An Omni mic receives a 360 degree COVERAGE sphere, this implies it can grab sound from above, below, in front of, behind, and to the side of the mic. The polar example for an Omni, likewise, is generally spherical. It could have an additive advantage, as a single Omni-directional microphone could be utilized to grab voices from various directions, till every individual voice is roughly in a similar pitch and a similar separation from the microphone.

Largely, an Omni-directional mic grabs a portion of the mood of the circumstance, which can strengthen the visual setting. On the off chance that the scene is occurring on a road corner, and we require the sounds of the considerable number of exercises going around, the Omni-directional mic could be utilized. The handheld mouthpieces utilized by the reporters in news from field and games columnists or reporters are typically Omni directional, enabling the journalist and interviewee to be grabbed by one microphone receiver held amongst them, and conveying a specific measure of surrounding sound. But few disadvantages also exists in utilizing Omni-directional mic.Since they pick up sounds from every corner they cannot be used for those scenes where the dialogue or the sound of the main subject is more necessary. Hence the background noises can create disturbance.They also tend to get more noteworthy measures of room resonation when utilized as a part of rooms that have hard-surfaced dividers and floors. This can once in a while result in a diffuse, empty, "inside a barrel" sound.

Bi-directional

Such microphones are generally used, in case of two persons, facing each other in an interview, live – interaction, or alternative bi-presentation, of a content to match a recording or live telecast.

Unidirectional

A microphone with unidirectional feature discards sound originating from back side of the mic whereascontinue tograb up sound originating from the front. In other words it captures sound signals from only that direction in which they are faced. Hence, unidirectional microphones could grab only little noise of the background and room resonation and on utilization with the loudspeaker systems they show lesser sensitisation towards 'feedback'. The 'feedback' is a situation of loop sound which occurs when the sound which comes out of the speaker is again caught by the microphone. As the same sound is caught between the mic and speaker it creates a noise like sound. There are different kinds of unidirectional, microphones every one possess a somewhat unique polar pattern and its own particular arrangement of points of strength and weaknesses. Till now cardioid is the highly common sort of unidirectional, it is named as cardioid, on the grounds that it's polar pattern somehow look alike a heart-moulded figure. Many cardioid mic will grab lesser to as much as half as much sound from the sides, than from the front, and short of what one tenth as much sound from the back than from the front. In this way, the cardioid mic have a tendency to grab most of desired sound (where you are pointing the mic) and little of the undesired sound (where you are not pointing the mic).

Transducer

Again the microphones can be divided in two categories depending on the transducers used. The transducer is the mechanism which actually converts sound energy into electrical energy. The two types of transducer are dynamic and condenser.

a) Dynamic Microphones

Dynamic mouthpieces simply utilizes a magnet and loop of wire to change over sound waves into a flag or signal. At the point if a thin diaphragm is appended with a loop of fine wire it starts vibrating if got struck via sound waves. In turn it makes the loop of wire move forward and backward around a magnet, generating a little electricity, which streams out of the microphone’s connector and via the cable of microphone. Dynamic mics of great quality produces a great quality of sound; they can deliver sound with less measure of commotion, are exceptionally tough, and will as a rule endure unpleasant taking care of or extraordinary temperatures and stickiness exposure. Since the Dynamic amplifiers can't be made in little size, numerous handheld and voiceover mics are the dynamic sort, as here the size of the mic isn't a matter.

b) Condenser Microphones

In the Condenser microphones (once in a while called electrets condenser mouthpieces) does not use a dynamic diaphragm rather utilizes a significantly more slender diaphragm extended tight simply over a piece of flat metal or ceramic i.e. metal-coated, also known as back plate. At the point if an electrical charge is fixed electrical charge thereafter it is placed on the diaphragm/back plate assembly, now its electrical yield fluctuates relying upon the diaphragm movements, that in corresponding to sound waves will vibrate. The output signal is to a great degree feeble and reactionary to outside electrical obstruction, be that as it may, it must be altered and additionally amplified and opened up by a circuit called a preamplifier. The location of preamplifier could be the the handle of the mic or in a little outboard or detachable electronic tube or pack. Many advantages are offered by Condenser mouthpieces. Their most vital feature is that they can be built smaller, therefore all mini livelier mics are of condenser variety. Condensers have a tendency to be exceptionally touchy to the extraordinary low and high frequencies, and generally have an extremely fresh, clean sound that improves dialogue intelligibility and also numerous melodic instruments. Their implicit preamplifiers permit condenser microphones to give higher yield than dynamic mics, implying that for a given sound level, resulting a output electrical signal of greater strength. This might be useful when you are endeavoring to record somebody who talks delicately, or who is more distant far from the mic.

Handheld

The most common kind of microphone for general use is the handheld type. While it can be held by the user, mounted on a floor or desk stand, or attached to a flexible “gooseneck” on a lectern, these options result in the mic being very visible, which is not practical in all video productions. A decent quality handheld mic ought to have an inward stun mount which will limit taking care of commotion or noise (pounding sounds transmitted through the handle and got by the mouthpiece cartridge), and it ought to be roughly built to withstand physical manhandle.

Title-Handheld Microphone

Attribution-Ousk

Source-

Link-https://commons.wikimedia.org/wiki/File:Svart_mikrofon.jpg

Models at the upper end of the value scale will typically produce clearer, more extensive sound range, better mounting of shock, and more strengthened. Sennheiser MD 42 is the most regularly utilized handheld mic.

Lavalier

It’s always suggestive to have a Lavalier type of microphone if you have only one microphone in your audio kit. This type of microphone can be attached to a user’s cloth, could be laid on a podium or in a pinch could be clip to a mic stand. Lavalier mics frees the orator’s hands to signal or exhibit an item, and in light of the fact that they are little they have a tendency to vanish on camera. Likewise, utilizing a lavalier will keep the separation from the mic to the orator’s mouth genuinely consistent, diminishing the requirement for visit change once the levels have been set. In circumstances where the receiver can't be noticeable, it's typically conceivable to hide a lavalier mic under a shirt neckline or even below a thin layer of garments.

Title-Lapel Microphone

Attribution-Terodaktil

Source-

Link-https://commons.wikimedia.org/wiki/File:Lavalier_mikrofon.jpg

Besides, a miniature type, lapel microphones having cords can be very easily concealed in the artist's dress. In addition cordless lapel microphones have been the latest trend giving free movement to the artists – inside or outside the studio. This facilitates, a cameraman, to take shorts from various angles, as per the desirable movement of the artist.

Head worn

A head worn microphone is of great need in situations where the orator's hands should be free and essential. Head worn mouthpieces can be situated nearer to the orator's mouth and keep up a steady separation and sound quality even during the movement of orators head while talking.

While headworn receivers are winding up perpetually undercover and are accessible in different skin tones (search for them in Broadway plays and musicals), they will at present be noticeable on camera.

Surface Mounted

Surface mounted mics are developed to grab the sound at a fixed or levelled surface. Surface mount mics are typically physically moulded to look less nosy on a meeting table or work area. The mic component is found near (yet not touching) the surface, with the goal that sound waves reflected from the surface land at the mic component in the meantime as the immediate sound. This viably duplicates the sensitization of the mic contrasted with an unsupported handheld write at a similar separation. (This affectability support accept that the surface is adequately substantial to reflect even sound waves of lower frequencies.)

Shotgun Microphone

The shotgun microphone is so named because the long, slotted tube in front ofthe microphone cartridge makes it resemble a shotgun.

The presence of “interference tube” helps it to be lesser sensitize for sounds originating from back and sides, compared to other directional mics. A shotgun mic is amazingly directional pickup design (called a line/inclination design) making these well known for news gathering, outside games scope and TV/movie creation.

Shotgun mouthpieces are not zooming focal points for sound or enhancers. They don't enable you to zoom in on a discussion from

100 feet away. Here's a considerably more precise similarity: envision looking through a long tube at a man standing 20 feet away. The individual's picture does not have all the earmarks of being any bigger or closer, but rather is to some degree easy to see, in light of the fact that the eye isn't diverted by things occurring off to either side. This is precisely what shotgun mics do best: screen out sounds originating from the sides, making the sound originating from the front easily audible.

Wireless Components

In short a wireless mic can be called as a mini radio station. The working starts with a conversion process of input sound waves in to audio signal by the microphone cartridge. Now the signal is conveyed by a low-control transmitter, and after that grabbed by a receiver present close-by, that changes over the radiofrequency signal once more into sound. This transmitter could be placed in the receivers handle, in a little pack such that one can wear it in the body, or in a piece or tube which could be connected specifically to any standard mouthpiece with a XLR connector. A table top unit, a rack mount unit, or a convenient battery-worked type that can mount on top or in the camera could be a receiver. A remote framework comprises of the mix of the mouthpiece, transmitter, and recipient. Than a link or wire associates the sound output of the receiver of the camera or sound recorder.

Portable and Camera Mounted Receivers

The battery powered portable mics can be accessed and used only with the presence of receiver and transmitter in and around. Such units size wise are very small about the extent of a deck of cards and could be worn on the body or mounted specifically to a camera. Connection is done via a short cable from the receiver’s portable output to the cameras or recorders audio input. But today we have advanced models that provides a separate output for headphone or an earpiece with the goal that the camera administrator can screen the sound through earphones or an ear piece. A microphone having wireless system is an extremely helpful thing for a video shooting, as here both the camera and the subject might move. For bigger projects where various orators are involved, numerous portable receivers could be connected to an audio recorder that could be carried in a bag.

The other application for a wireless receiver that is portable is feeding sound from a blender to a camera situated over the room. If we take example of an extensive gathering room, for instance, the sound blender is frequently situated at a side of the stage, whereas the camera is at the back of the room. During situation like this, the wireless transmitters input needs to be connected with the output of the mixer or the blender, and with the portable receiver attached to the camera. Thus the requirement to depend on the internal microphone of the camera or to put extra mics particularly for recording video is eliminated.

Audio Mixers, Interface and Recorders

If you view the videos streaming online you could sense that the sound quality is not up to the mark,this is because the video recording takes place via the inbuilt microphone of the camera. Generally the inbuilt cameras microphone are not of superior quality and moreover the mic is placed far from the orator and very close to the cameras autofocus and image stabilization system. What's more, the sound hardware in the camera may have excessively murmur and insufficient capacity to oversee changing sign levels for an expert sounding creation. On the off chance that you need better audio sound, you should utilize an external audio, and perhaps an outer sound interface or recorder.

Audio Interface

A sound interface is a little box that goes about as a middle of the road arrange between your mouthpiece and the audio input of the camera. One or two adjusted XLR mouthpiece inputs are there in many interfaces (sometimes equipped with phantom power for condenser mic), and a means of adjusting the audio imbalanced level.

Title- Audio Interface

Attribution- Nicolas Esposito

Source- Interface audio M-Audio FireWire Solo

Link- https://commons.wikimedia.org/wiki/File:Interface_audio_M-Audio_FireWire_Solo.jpg

The interface output can be an connection that is unbalanced that works with a DSLR, or a USB connection that enables you to record specifically to a PC. A sound interface is helpful in light of the fact that it gives you better control over sound levels and makes it easy to utilize proficient mics with XLR connectors.

Audio Mixers

On the off chance that we are utilizing a few receivers immediately, for instance to record a gathering or board discourse, prior to start recording it may be important to consolidate the signal of the mic altogether. The audio blender here provides a single output that incorporates the consolidated output of all the mics and it also enables adjustment at individual level for every mic.

Title-Audio Mixer

Attribution- Evan-Amos

Source-

Link- https://commons.wikimedia.org/wiki/File:Behringer-Xenyx-1002FX.jpg

The downside of utilizing a blender is that it makes it hard to segregate the voice of one orator while editing.. There are different blenders relying upon their ability of tracks like 8, 16, 24 tracks. They additionally include some sound impacts like reverberate, reverb, bass and treble. These days advanced blenders are likewise accessible in type of PC applications

Audio Recorder

Nearly all cameras have sound recorder inbuilt in it however one might find that audio recording from the camera directly is not satisfactory, because of inordinate murmur, bending, or absence of control over sound levels. For this situation, video makers utilize a "double framework" in which the video is recorded on the camera, and the sound is recorded on an outside sound recorder. Utilizing an outside sound recorder enables you to have a powerful control over sound levels and less murmurs than generally cameras. Furthermore, a great sound recorder as a rule has adjusted mic input with XLR connectors, and frequent phantom control for condenser mics.

In huge projects, four or eight tracks recorders are accessible, enabling independent recording of every orator's mic to empower more exact control of signal levels and enabling easy editing. An essential thing to be consider is the way to synchronize the recorders audio with the cameras video. The arrangement is to start each shot with a hand applaud that is picked by the mic and visualise by the camera. While editing, the sound track is adjusted by moving backward or forward such that the sound of the applaud matches perfectly with the visual. The sound recorder enables you to definitely modify for signals of various levels from different sources and safeguard them as isolated sound tracks that can be altered into the program as proper. A few recorders have an assortment of input connectors, developed to assort sounds of different levels and types.

Cables and Connectors

The most important link that connects the recorder or receiver and the microphone are the cables and connectors. These are likely the most ignored connection in the sound chain, but low quality cables or potentially damaged connectors are as often as possible the reason for real audio issues. Basically among audio devices for connection two types of connections are used:

- Balanced &
- Imbalanced

In a balanced connections there is a requirement of cable consisting of two wires (one for the "hot" signal and one for the "return") this wires needs to be shielded with a mesh, braid, or metal foil. From different sources the random electrical signals are bombarded to the cable that is intercepted by the shield and then drain it in to the ground. Both the shield and the wires works together to avoid interference of the various audio signals. The quality of the audio depends on the types of connectors and cables utilized.

In an imbalanced connection a shielded single wire is used by a cable, however here the shield has to perform dual task i.e. it has to carry back the returned audio signal along with providing protection to the wire from electrical interference. Balanced audio connections are more stable than the imbalanced audio connection as these cables are prone to be affected via florescent light fixtures, some types of dimmer switches, and other audio or electrical cables present nearer to it.

So according to rule balanced connections are more reliable in terms of clean, free of noise output. In present era all the well-known connectors for proficient mics and audio devices trust and favour XLR and USB connectors

XLR Connector

There are two types of XLR Connectors 1) Male XLR connectors 2)

Female XLR Connectors.

Male XLR Connectors consist of three pins, these are utilized for providing signal output; whereas female XLR connectors comprises of three sockets, utilized for provision of signal input.

The XLR connector is strong, it generally don't break or bend when connected, and many varieties have secure interlocks to make it free from accidental unplugging. A cable that has an XLR connector at both ends almost certainly indicates a balanced connection.

Title- XLR Connector

Attribution- Photographer: Michael Piotrowski (2005-06-04)

Source-

Link- https://commons.wikimedia.org/wiki/File:Xlr-connectors.jpg

Both low- and high-impedance microphones contains XLR connectors. High calibre and expert mics supports the XLR connector over the 1/4-inch telephone plug.

USB Connector

Universal Serial Bus (USB) connectors and links, which have turned out to be indistinguishable with PC peripherals, are obviously winding up more typical for mics, since more sound and video recording is occurring specifically on PCs. This has made the requirement for XLR-to-USB connectors, which allows you to utilize your XLR cables and XLR mics with gadgets which contains USB ports.

Quarter inch phono plug

Another to some degree normal sound connector is the male 1/4-inch telephone plug, which mates with the female 1/4-inch telephone jack. The origin of the name is from utilization of this connector on early phone switchboards. These can be found on cables utilized with a sound hardware: earphones, amplifiers, loudspeakers, signal processing gear, and mics. As a rule, 1/4-inch telephone plugs are utilized on mics of lower ends.

Two-conductor types (at times referred as "TS" or "tip-sleeve", which alludes to the region of the connector utilized for each wire) comprises of two different portions and are utilized for imbalanced mono connections. Three-conductor types (at times referred as "TRS" or "tipring-sleeve") can be arranged to convey an adjusted mono signal or an unbalanced stereo signal Microphones and microphone inputs on blenders utilizing 1/4-inch telephone connectors are quite often of the unbalanced high-impedance compose.

RCA Plug

The last sort of connector you'll likely keep running into is the male RCA plug or phono plug, which mates with the female phono jack. The prefix "phono" originates from the way that these are the standard for associating phonograph turntables (and also cassette players, CD players, et cetera) to home stereo hardware

Miniplug or EP Jack

The Mini plug is available in two sizes: 3.5 millimetres (1/8 inch) and 2.5 millimetres. Mini plug of 3.5 millimeter form is similar to the normal attachment generally found on earphones and ear buds. Even though famously delicate for microphone applications, because of their little size, mini plug connectors are every now and again utilized on consumer and even semi-proficient video hardware, even in DSLR cameras. Quite often they show an unbalanced stereo sound association. Most mics that come furnished with mini plugs are affordable units of low cost. Even if your equipment has a smaller than normal attachment port or 1/4-inch microphone input, you can at present utilize a decent quality expert mic. You simply need to acquire a cable with the proper connectors, or in certain instances, an impedance transformer.

Field Monitors

A field monitor is an external, portable, battery-powered display that replicates the picture being recorded to camera. We most often attach our monitor directly to the camera or camera rig, and sometimes we’ll attach it to a light stand as a stand-alone monitor for clients to view. Most field monitors available today range from five to nine inches in screen size, making them two to three times larger than most camera displays. When we’re moving quickly in the field, the extra pixels are extremely helpful for maintaining a focused image and seeing all the details within a frame.
It is very much difficult to monitor a picture while conducting an interview for a solo camera operator interviewing someone. As the interviewer, you’re often sitting beside the camera, making it difficult to keep an eye what’s being recorded. An external field monitor acts as a crucial piece of gear in these types of situations, allowing you to observe whether your subject is drifting out of frame or focus.

When shooting high-perspective shots above crowds with the camera is mounted much higher, it can be difficult or sometimes impossible to see our camera’s screen.So an external field monitor can potentially be the only option for framing a shot. We also regularly use jibs, sliders and stabilizers that require the use of an external monitor to direct camera movement.

Necessity of Field Monitors

The majority of field monitors use traditional LCD technology for their screens. However some of the newer, higher-end LCD panels integrate IPS (In-Plane Switching) technology, resulting in a higher contrast ratio and better, more accurate overall colour and better image quality. Lately, the monitor trend has been moving towards even newer, OLED (Organic Light-Emitting Diode) displays, which according to Small HD, offer the richest colours and extremely high contrast ratios. The extra brightness of OLED monitors makes them a great choice for outdoor production.

The present day field monitors also coming with recorders. That means you don't have to carry an extra video or audio recorder known as back pack such equipments help the spot-reporter or the artist to monitor the programme on air, locate his proper position and link the production control room with matching presentation, as per requirement.

Unit summary

In this unit we came across some necessary filmmaking and video shooting devices such as clapboard, various kinds of microphones and other audio equipments used from shooting to post production. We also came across the use of clapboard, sound equipment and field monitors. We learnt about the use of microphones according to the situation.

Assessment

1. What is a microphone?
2. Differentiate between Traditional and Digital Clapboards
3. Classify various types of microphones. Clipboards.
4. Write the functions of Field Monitors
5. How many columns are there in clapboard and name them.
6. Who developed electromagnetic type of microphone?
7. How many types of microphones are there depending on direction?
8. What is the other name for RCA plug connector?
9. What is a transducer in a microphone?

Resources

- Fundamentals of Digital Audio Alan P. Kefauver and David Patschke
- Visual Studio 2013 Cookbook Bruce Johnson
- How to shoot Video that doesn't suck Stockman, Steve, 1958
- The Book of Audacity : record, edit, mix and master with the free audio editor Schroder, Carla

Unit-8 Audio Video Program Production

Introduction

In this unit you will learn the basics of audio-video production techniques. It is a preface to the skills that are essential to know the operations of audio and video equipment in studio settings and a foundation an establishment to enhance the development of visual and aural education. You will learn how to operate camera, audio control, basic directing, lighting, and editing, and more and will get opportunity for hands-on experience.

Outcomes

Upon completion of this unit you will be able to:

- Get acquainted with the basics of audio and video production.
- Exhibit knowledge of audio video production process.
- Utilize the steps of production process in creating audio-visual files.

Terminology

Ambient Sound	A kind of sound recorded on location while shooting.
POV	Meant to show the character’s perspective.
Panning	Camera movement from left to right or vice versa.
Key Light	Main source of Light

Video Production

Video production is the process of creating video by capturing moving images (videography), and creating combinations and reductions of parts of this video in live production and post-production (video editing). In most cases the captured video is recorded on the most current electronic media such as SD cards. Earlier the footage was captured on *video tape,* hard *disk,* or *solid state storage.* Video tape capture is now obsolete and solid state storage is reserved for just storage. It is now distributed in digital formats such as the **Moving Picture Experts Group format (.mpeg, .mpg, .m4p), QuickTime (.mov), Audio Video Interleave (.avi), Windows Media Video (.wmv), and DivX (.avi, .divx)**. It isthe equivalent of filmmaking, but the images recorded digitally instead of on film stock.

Practically, video production is the art and service of creating content and delivering a finished video product. This can include production of television programs, television commercials, corporate videos, event videos, wedding videos and special-interest home videos. A video production can range in size. Examples include:

- A family making home movies with a prosumer camcorder,
- A solo camera operator with a professional video camera in
- a single-camera setup (aka a "one-man band"),
- A videographer with a sound person,
- A multiple-camera setup shoot in a television studio
- A production truck requiring a television crew for an electronic field production (EFP) with a production company using set construction on the backlog of a movie studio.

Audio Production

Audio production is the general term used for all stages of production happening between the actual recording in a studio and the completion of a master recording. It involves, sound design, sound editing, audio mixing, and the addition of effects.

Title- Fig 3.1 The Audio Visual Production Process

Attribution- U. S. Fish and Wildlife Service - Northeast Region

Source- https://www.flickr.com/photos/usfwsnortheast/9444972202/

Link-
https://commons.wikimedia.org/wiki/File:Working_in_audiovisual_and_broadcast_production_for_the_U.S._Fish_and_Wildlife_Service_(94449 72202).jpg

What is the Production Process?

The *process of production is basically concerned with the* stages (phases) needed to finish the production of a media product, beginning from the idea till the finalization of the master copy.

This process can be applied to any kind of the media production incorporatingmovie, featured film, video, television and audio recording.
The phases in every medium change; for instance, there is clearly no storyboard in an audio recording. However a similar general ideas work for any medium.

The phases of production

Pre-production – Planning, Scripting & Storyboarding, etc.
Production – The actual shooting/recording.
Post-Production – Every aspect amongst production and making the final master copy.

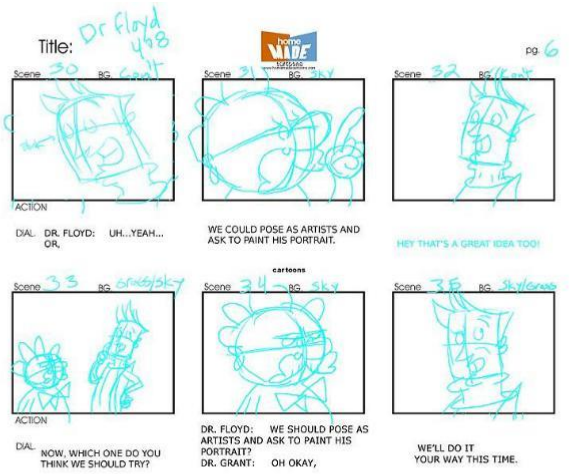
Basics of Audio-Video Production Shots

The simplest element in video and film
Is an image resulting from a single continuous running of a camera.
A continuous piece of video or film footage.
It’s everything you get between pressing “record” and “stop”

Scene

Scene comprises of all the *action/shots* which takes place at a certain time and location and consist of a segment of a program.

Storyboard



Title- Fig 3.2 A Storyboard

Attribution- http://www.flickr.com/photos/tmray02/

Source- http://www.flickr.com/photos/tmray02/1440415101/

Link
https://commons.wikimedia.org/wiki/File:Storyboard_for_The_Radio_Adventures_o f_D r._Floyd.jpg

A movie maker draws simple schematics of frames.

They use the frames to plan how they want to tell a story.

The frames show the correct order of significant objects or actors and the camera’s position.

Camera

Title- Fig 3.3 Video Camera

Attribution- Jeremy C. Schultz

Source- wikimedia.org

Link- https://commons.wikimedia.org/wiki/File:JVC_KYD291.JPG

The production stage requires producing a video utilizing a video camera. The Video may vary from video to video. It will depend on the style and content of the video being made and the amount of time, effort and money that is being put into production but… However large or small your video may be… This is the proven production process that successful video producers use…It works out into three main phases.

Camera Shots

Different types of shots are captured according to the requirement of the storyboard/demand of the director. These are

- Close-ups

- Wide shots
- High angle
- Low angle
- Point-of-view

The pre-production phase:

The first step of the video production process is planning. Before starting a new project you first conduct research, identify the problems & also the solutions and perform other organizational duties.

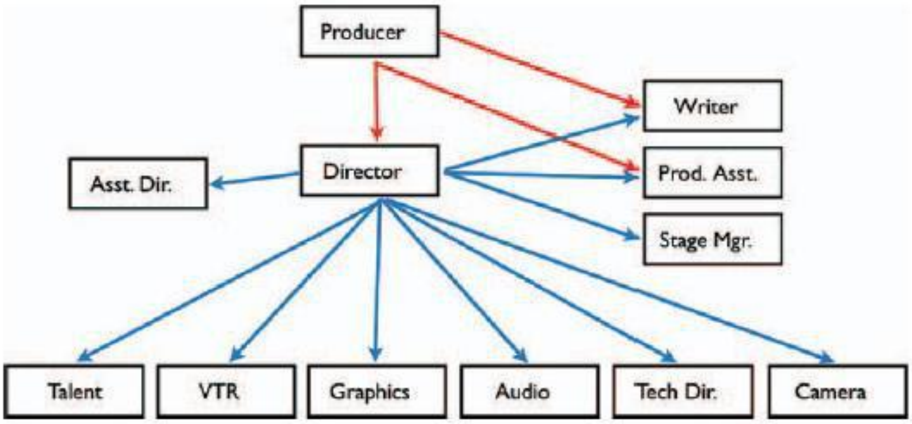
By this stage the team has been formulated including *Producer, Director, Production Designer, Director of Photography (DOP), Sound and Editor* (in this production the roles of sound and editor were taken on by the same person).

The **Producer** is responsible for the overall organisation of the production including working with the Director and Production Designer to come up with locations and co-ordinating with the actors which would enact in the film and convey the story as written down in the script.

The **Director** is responsible for the creative visualisation of the script or event. He would take on overall 'creative' responsibility of the production, the style of shooting, types of shots, selection of location and final editing. He directs the actors how to act to convey the emotions to its best. He brings out the essence of the characters and adds life to the story. The DOP in conjunction with the Director is responsible for the look of the film, the style of the filming; again achieving what would have been suggested by the writer in the script.

It's the job of the **Camera operators**toset up their cameras and after thatusing the camera for capturing the video as desired or directed by the director.

Audio mixer/sound mixer will be in charge of provision of quality sound and they manage the sound balance and also responsible for the technical and artistic quality of the sound.



Title- Fig 3.4 The Structure of a Video Production Crew

Attribution- Gerald Millerson

Source- *Video Production Handbook-4th edition*

Link-

http://home.fa.utl.pt/~cfig/Anima%E7%E3o%20e%20Cinema/Realiza%E7%E3o%20Cinematogr%E1fica/Video%20Production%20Handbook,%20Fourth%20Edition.pdf

Production Methods

Having brilliant ideas are not sufficient. Ideas generated have to be turned into action to convert it in to out in realistic, practical terms. They can be viewed or heard as images and sounds. At the last it will be the decision of the director about the camera that is to be used to shoot and what will be the liking of the audience to view and hear.

There are two very unique techniques for moving towardvideo production:

The *empirical*technique is the place where intuition and opportunity are guides.

The *planned*technique, which composes and assembles a program in precautiously, arranged steps.
The directed has to opt for one method before getting started.

The video's concept

In the planning stage director should map out what exactly he wants. This helps him in accomplishing him aim while recording, editing, sound mixing, captioning, etc.

Script

The script forms the basis for the planning of any film project. For certain types of production, such as drama, the script generally begins the production process. The director reads the draft script, which contains general information on characters, location, stage directions, and dialog. He/she then visualizes the script or envisions the scenes and assesses possible treatment. The director must anticipate the script’s possibilities and potential problems. At this stage, changes may be made to improve the script or make it more practical. The next director prepares the camera treatment. Scripts do the following:

- Help the director to classify ideas and to develop a project that works.
- Help to coordinate the production team.
- Help the director to assess the resources needed for the production.

Storyboard

Directors need to think through each scene in their minds so that they can capture the images and turn them into a storyboard. The storyboard is simply a series of rough sketches; these sketches help the director to visualise and organise the camera treatment. It is the visual map of how to arrange the key shots for each scene or action sequence.

Talent Contract

At this stage director secures a signed Talent Release form from anyone involved in his audio, video, or slideshow projects. Customize the form for your own project(s).

Copyright

It is illegal to reuse copyrighted content that you do not have explicit, written permission to reuse. This means that you cannot reuse music, television clips, voiceovers, photos or other multimedia that you have not produced and do not have permission to use. Fair use does not typically allow the reuse of original or derivatives of creative, copyrighted works for public distribution.

The Production Phase

The *production phase* is the actual production (making) of the video. It's at this stage of the video production process that you are actually shooting the video.

Production techniques are the features used to make the text(s) interesting and unique. Techniques may include: music, dialogue, lighting, graphics, colour, special effects, soundtrack, camera work, layout, use of space, oral and visual production techniques, or use of links. By looking at the production techniques closely you will gain a better understanding of how the text has been produced in order to present the themes, characters, settings, and plot. The attitude of the director towards the character helps to set the mood or feeling of the text. Think about how the techniques and the mood of text work together to make the production convincing.

Key points to study

Structure – how the text and the ideas have been put together. Look at the overall structure of the text(s), the order of scenes, sequencing, and transitions.

Narrative point of view – who is telling the story and how this influence what the audience experiences and feels towards the text, does. The director will choose and/or change the point of view to control the relationship between the audience and the character to support their purpose. Changes in perspective can be shown by techniques such as voice-over and camera shots, like the point of view shot.

Narrative style refers to how the subject matter is presented to the audience.

Dialogue – identify repeated language patterns in a character's speech. Look at the types of words used and how they speak. What does this show you about their personality and background? Think about how the voice is used to show subtle changes in emotion, accent used to show social status and background, and gesture to show response to other characters.

Required for the project

Equipment

- Camera (video/film camera, Smartphone, tablet, digital camera).
- Tripod (to steady your shots and minimize camera movement).
- High quality microphone (for audio-only).

Lapel microphone (better quality than a camera's built-in microphone).

Lights.

You can also record audio directly at your desk via your computer. Audacity is a free program that you can download to record and edit sound.

Location

The locations of the video/films are selected according to the script keeping in mind the budget of the producer.

Audio

Audio is recorded in a quiet environment, free from any interruptions or ambient background noise (including: cell phones, pagers, co-workers, etc.).

Video

Videos are recorded in a well-lit environment. Be careful of reflections and glare from eyeglasses, windows, computer screens, etc.

Tips for Recording Audio:

- Read from a prepared script.
- Speak slowly and clearly; carefully pronounce words
- Pause frequently for the benefit of your listening audience — before starting your recording and between long sentences and paragraphs.
- Practice and review a short test recording.
- Preview your final recording before delivering.

Tips for Recording Video:

- Plan your recording session carefully; recordings often take longer than expected.
- Prepare your audio script ahead of time.
- Record with a specific purpose in mind to avoid recording excess footage.
- Record the shots that you've outlined on your storyboard.
- Ask subjects to wear solid, neutral colours.
- Practice and review a short test recording.
- Keep subjects within frame of view — record more of the subject matter and less of the surroundings.
- Preview your final recording before delivering to your intended audience.

While recording the video

- Set up your equipment before the shoot.
- Use a lapel microphone to cut down on background noise.
- Ask the person speaking to.
- Pronounce words carefully and to speak slowly and clearly. Practice saying any technical or scientific terms before recording.
- Be careful about using 'umm,' 'uhuh,' and other similar "filler words."
- Explain/describe what they are doing and why. This verbal description of activity will better clarify what is happening in the video. Intentional narrative helps people reading a transcript or listening to the audio to understand what is happening.
- Do not interrupt the person(s) you are interviewing; wait until they finish speaking to ask for clarification.
- Do retake if.
- The person coughs unless it is part of the video.

The Post-Production

Post-production literally is where the director bring together all of the different elements and material created in the production phase to form a finished product as envisioned in the pre-production stage.

Title- Fig 3.5 The Post Production Studio

Attribution- Autobahn Two

Source- pixabay.com

Link-https://commons.wikimedia.org/wiki/File:STEP_studio_pict2.J PG

Post Production includes editing, but it is much more than that. Post begins with the script and continues in the Pre-Production phase with the planning, scheduling and budgeting of finishing processes.

During Post-production the *Editor* is syncing dailies and assembling a rough cut for the Director to view as the shooting progresses. Finally, there are the **sound design, scoring, titles, visual effects, mix, colour correction** and **delivery** that comprise the finishing process.

Students usually experience little difficulty with the Pre-Production and Production phase of filmmaking, but once principle photography is complete their projects tend to lose momentum and unravel. The primary reason for this is a failure to think holistically about all of the work involved in making a film. A movie is like a cake – you add the flour, the eggs, the sugar, but until it's baked it's not a cake. Efficient Post Production requires serious multi-tasking.

Most importantly, Post Production has to be seen as an integrated part of the whole and approached with the same attention to planning and scheduling that is given to Production. No one plans to fail, but failure to plan can lead to disaster. Goals and deadlines have to be set and progress must be monitored continually if the film is to be finished – and after having spent crores of rupees in Production – what a waste not to have a finished film to show for all that effort. The areas include in post production are:

- Film Processing
- Adr And Foley
- EditinRoom
- Sound Mixing
- Equipment
- Playback
- Telecine Transfer
- Visual Effects
- Titling And Optical
- Negative Cutting
- Sound Editorial And
- Delivery Elements
- Design

The Lab

In addition to processing a film, printing film dailies and prepping your dailies for transfer to videotape, the lab is also where you go to procure bags, cans and cores which go to the production set. When picking up these items, the lab needs to know the film's gauge and what size "loads" you'll be using. The production manager can answer these and other questions. Be sure to meet with the laboratory contact prior to the start of production. This will help in avoiding expensive mistakes down the road. It will also insure that the lab is prepared to process your dailies when you need them.
The lab contact will need to know the details of your shoot. This will include the amount of film you expect shot on a daily basis, if you have any night shoots or weekend shoots scheduled, and if you are cutting on film or videotape (or both). Arrange a film lab tour for prior to starting the postproduction process. This will give a leg up on how film is processed and what information the lab needs to do the job correct and on time.

Have someone to show what to look for on a camera report. There is vital information the lab needs from those reports to even begin your job. Understanding this information will allow you to properly communicate should information be missing. Most film laboratories offer a variety of services. They develop film and prepare it for transfer to videotape, create prints, and repair damaged film. Some have optical departments where they create film effects and titles, blow-ups and repositions. To fully understand and appreciate the work that goes on at the film lab, take a tour.

Dailies and Telecine

In a film shoot, dailies, as the name implies, is the footage that is shot each day and rushed to the lab for processing. It then moves on to telecine or printing so that one can view them, usually the next morning. The dailies from a tape shoot are still the footage that is shot each day; it just does not require processing.

If you are having your film dailies transferred to videotape (telecine), you will need to speak with the transfer facility prior to the beginning of your job. As with the film lab, they will have a list of questions for you to answer before they can schedule your job. The information they will need includes details about what type of film and sound you are shooting, how you plan to complete your project once shooting is finished, and what your time schedule is for your project. How much film is budgeted for each day, and how many days you will be shooting will also be important.

Some information must be taken directly from film during the transfer process. Whether you plan to do a film and/or videotape finish will tell the facility what information they need to gather at the time of telecine. Not planning ahead and having to go back to get this information is extremely costly and time consuming.

Off-line editing

Off-line editing indicates an electronic cut. This means that processed negative shot each day will be transferred to videotape or to a hard drive. This videotape is then provided to the off-line editor to be recorded into electronic editing equipment for (non-linear) editing. It can also mean you have taken your digital raw files and compressed them to create smaller, more manageable file sizes for your editing workflow.

On-line editing

The *on-line* is where the final assembly of the project or the conforming of project by linking to highest definition or raw digital files after the edit is complete.

You may have to change your sequence settings and relink to highest resolution footage (2K or 4K for example) to conform the locked cut to the highest resolution files before you send them to colour correction.

Just like each earlier process (film processing and telecine dailies transfer), the on-line facility will have a list of details they will need from you before they can book your on-line session and complete this process. This will include questions about what videotape format your dailies are on, where the tapes will be coming from, what off-line system was used to create the *editing list* (called the edit decision list-EDL), and any instructions involving special effects. Sometimes the same facility that did your film processing and telecine will also be doing your on-line, sometimes not. Other steps that will take place as part of this process may be creation of special effects, titling and colour correcting your picture.

Sound

Sound for a project actually starts in dailies with the "*production sound*." This is sound recorded right on the set at the same time when the picture dailies are recorded. Whether shooting on film or videotape, you will probably have some production sound. The exception will be a project that relies solely on voiceovers or sound and effects that are recorded later.

Production sound elements are delivered to sound editors to be used to help "*sweeten*" the sound that was combined to the picture either in the film editing room or the off-line editing room. Once all of the sound edits have been agreed upon, production sound, along with any ancillary sound effects and music are mixed together. This is called *mixing* or "*dubbing*" (it is also called "audio sweetening in commercials and television). Mixing takes your production audio and finalizes it with enhancements, ADR, music, sound effects, and various clean-up procedures.

Completion

Once you have the picture and sound elements nailed down, your delivery requirements will determine how you complete your project.

A film finish means that all of your work toward delivery has been done on film. This does not preclude making a file-based or videotape master from your film elements, but the file-based or videotape master will only be struck once the film's picture and sound elements are completed. A completely finished film element must be created to satisfy your delivery requirements. The negative is cut once the show has been locked (final edits are approved) and optical (fades, dissolves and titles) are ordered. The film lab creates the colour-corrected print. The movie is colour-corrected prior to striking release prints and can also be colour-corrected for use as a telecine print master.

For a feature or movie-of-the-week, allow at least 10 days for a negative to be cut and spliced into a finished piece. Allow another week (or more) to arrive at the right colour-corrected film element.

If the file or videotape is to be the only delivery format, and then it will not require cutting negative prior to delivery, you have chosen what is referred to as a tape finish. A file or tape finish can also take place on a project that will ultimately be finished on film if materials for preview or advertising are required prior to the film finish being completed. A two-hour show can take at least one day to several days to complete. One-hour TV shows usually spend one to two days in colour correction.

The master is electronically colour-corrected scene-by-scene. Depending on the complexity of the look of the project and the evenness of the negative exposures, it can take from hours to days to colour correct a master. If finishing on videotape, formatting will either be incorporated into the EDL or done "tape-to-tape" near the end of the process. Formatting can include adding logos, bars and tone (videotape) and commercial blacks (videotape), and closed captioning (again, videotape).

When finishing on **film, titles, credits, locales, legends,** etc. are created optically. They are shot on film using the plain "*textless*" backgrounds. These backgrounds are matted together with titles creating a new piece of "texted" film which is then cut into the final-cut film negative. On videotape, these are done after the entire picture alterations are accomplished (such as special effects and colour correcting). As with film, the "textless" pictures are mixed with text, making a new "texted" picture.

Delivery

Delivery is completed successfully only when the film has fulfilled all of the delivery requirements and the distributor has accepted the elements. The only way to safeguard against missing delivery materials is to get, read and understand the delivery requirements. Delivery elements are best made along the way, at the steps where they are the easiest and most cost-effective to create. They often require paperwork and contracts drawn and signed. Collect delivery requirements at the start of your project. Make a checklist and keep it updated so you are not caught short and costing the producer unnecessary expenses.

Project Workflow

Workflow refers to the management of steps required to produce a program.

The First Step: Gather

Gathering: This may take place with one or more tools, and the primary tool is the video camera.

The Second Step: Capture

You must capture or transfer your video (or audio) from its source to a computer work.

The Third Step: Edit

Editing your video with a nonlinear editing program

The Fourth Step: Compress and Code

You must determine what target platform your video is to perform, and you must optimize or compress that video to play efficiently on that platform. It may CDROM or DVD. In any event, your computer's software and hardware have the tools to accomplish this task. You may save as a QuickTime file or a Real Media file.

The Fifth Step: Encode to CD-ROM

If you intend to archive or distribute your project to CD-ROM, you must learn the optimal compression strategy for the piece you have produced. This strategy's objective is to produce the highest quality and best performing video yet it must fit on a CD-ROM disk and play properly (without hesitation or distracting breaks in performance) on your computer platform.

The Sixth Step: Archive

It's important that you save an uncompressed version of your work for full-broadcast play at full screen resolution. This archive can reside on DV tape (least expensive) or it can be archived to a hard drive or disk array for easy retrieval. This is most expensive, but time is money, and many production houses are archiving entire projects this way so that they are handy. In certain cable operations, videos are archived in a disk array data base for retrieval for broadcast.

Unit summary

The focus of this Unit lies in the three central areas of video production: pre production, production, and post production. Students will develop a comprehensive idea about, shoot video, and edit both audio and video to produce a finished project.

Assessment

1. Identify video production equipment/components.
2. Explain the components of video production.
3. Identify and explain the operation, components and function of all major video equipment.
4. Define terminology related to video production.
5. Explain basic trouble shooting and safety procedures for video production.
6. List the various positions in a video production.
7. Use video camera functions such as zoom, pan and fade.
8. Describe in detail the process of editing.
9. Identify the basic requirement in video production i.e. equipment and software.
10. Evaluate the effectiveness and the process of a video production.
11. What are the three elements of a story?
12. What is the visual diary of your video?
13. When should all or most of your production decisions be made?
14. What should always be considered while framing a shot?
15. Define editing and describe the types of editing?
16. What does AVI stands for?
17. What does MPEG stands for?
18. Primary light that shines directly on subject is called?
19. Moving the camera side to side is called?
20. The narrator talking off screen is called?

Resources

- Wikipedia.com
- Video Production Handbook-4th edition
- https://en.wikipedia.org/wiki?curid=1553972
- http://communionmarketing.com/index.php/web-applications-other/introduction-to-web-applications
- www.medialit.org
- http://www.medialit.org/reading-room/video-basics-and-production-projects-classroom
- www.mediastudentbook.com
- www.docplayer.net
- http://www.moviemaker.com/archives/moviemaking/directing/ articles-directing/post-perfect-in-10-easy-steps-3341/
- http://docplayer.net/23069879-Post-production-handbook.html
- http://caes2.caes.uga.edu/unit/occs/resources/multimedia/record.html

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સ્વાધ્યાયઃ પરમં તપઃ
સ્વાધ્યાયઃ પરમં તપઃ
સ્વાધ્યાયઃ પરમં તપઃ

શિક્ષણ, સંસ્કૃતિ, સદ્ભાવ, દિવ્યબોધનું ધામ
ડૉ. બાબાસાહેબ આંબેડકર ઓપન યુનિવર્સિટી નામ;
સૌને સૌની પાંખ મળે, ને સૌને સૌનું આભ,
દશે દિશામાં સ્મિત વહે હો દશે દિશે શુભ-લાભ.

અભણ રહી અજ્ઞાનના શાને, અંધકારને પીવો ?
કહે બુદ્ધ આંબેડકર કહે, તું થા તારો દીવો;
શારદીય અજવાળા પહોંચ્યાં ગુર્જર ગામે ગામ
ધ્રુવ તારકની જેમ ઝળહળે એકલવ્યની શાન.

સરસ્વતીના મયૂર તમારે ફળિયે આવી ગહેકે
અંધકારને હડસેલીને ઉજાસના ફૂલ મહેંકે;
બંધન નહીં કો સ્થાન સમયના જવું ન ઘરથી દૂર
ઘર આવી મા હરે શારદા દૈન્ય તિમિરના પૂર.

સંસ્કારોની સુગંધ મહેંકે, મન મંદિરને ધામે
સુખની ટપાલ પહોંચે સૌને પોતાને સરનામે;
સમાજ કેરે દરિયે હાંકી શિક્ષણ કેરું વહાણ,
આવો કરીયે આપણ સૌ
ભવ્ય રાષ્ટ્ર નિર્માણ...
દિવ્ય રાષ્ટ્ર નિર્માણ...
ભવ્ય રાષ્ટ્ર નિર્માણ

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(Established by Government of Gujarat)
'Jyotirmay' Parisar,
Sarkhej-Gandhinagar Highway, Chharodi, Ahmedabad-382 481
Website : www.baou.edu.in

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