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SCHOOL OF SCIENCE AND HUMANITIES

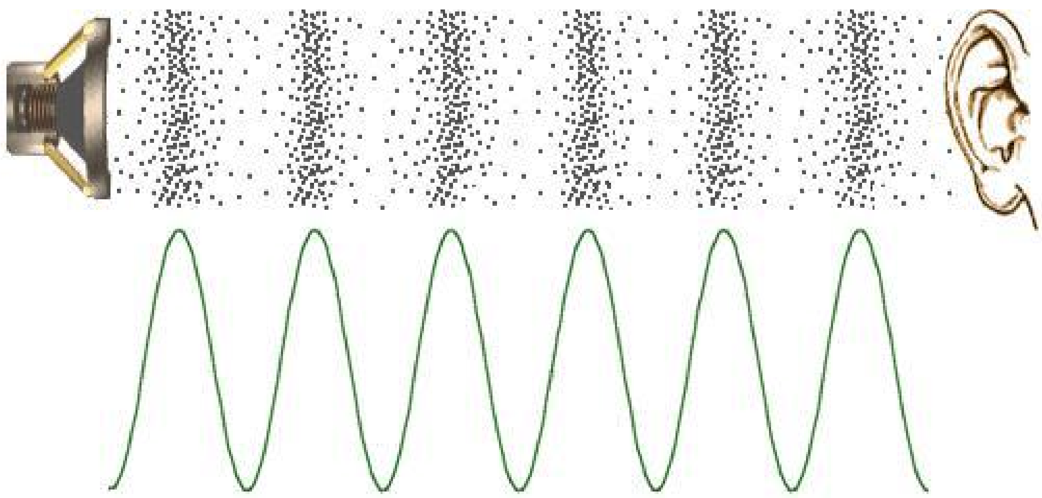
DEPARTMENT OF VISUAL COMMUNICATION

UNIT – I - Audio Production Theory – SVCA1301

What is sound? Sound is a form of energy, just like electricity and light. Sound is made when air molecules vibrate and move in a pattern called waves, or sound waves. Think of when you clap your hands, or when you slam the car door shut. That action produces sound waves, which travel to your ears and then to your brain, which says, "I recognize that sound." Sound is defined as a disturbance in an elastic medium that can be detected by the human ear. The medium can be gas, liquid or solid

Sounds are produced by mechanical vibrations. Vibrating object disturbs the molecules of air surround it causing periodic variations in the air pressure. As the object vibrates back and forth. the pressure becomes alternately more. and then less dense.

Variations in Air Pressure and Corresponding Waveform



The image shows a speaker creating sound waves to reaches an ear

The loudspeaker moves and pushes the air particles on its right (phase a) causing compression to take place. These particles then push their adjacent particles and transfer the energy they have received from the loudspeaker to them. The loudspeaker then moves back again and carries out a compression in the opposite direction, in other words dilation towards the left takes place (phase b) and in doing so forms an air gap that gets filled up by the air particles that are immediately close by. These particles in turn create other gaps to their right, and so on and so forth. This process allows the particles to transfer their energy to each other by oscillating without physically moving in the sound's direction.

Compression (C= compression) and dilation (R = rarefaction) of particles in the air

The normal range of frequencies audible to humans is **20 to 20,000 Hz** (the number of vibration per second). A range of **200 to 2000 Hz** is required to understand speech. the speed of sound is approximately **343 m/s**

(Sound) Db meter

History of decibel

Alexander Graham Bell. Before Mr. Bell started to experiment with audio transmission and reproduction, there was no scale to measure acoustic energy, so Bell came up with his own scale to measure how loud a given sound was. He called his scale the Bell scale. His instruments were crude and his 1-10 scale was not very fine or accurate. We still use his scale today, but we have increased the sensitivity tenfold (that's where the deci part of decibel comes from); in other words, what Bell would have measured to be 7 bells, we now call 70 decibels. In fact, Bell would have measured any sound between about 65 and 75 decibels at 7 bells, Definition for **decibel** the measuring of sound level ..

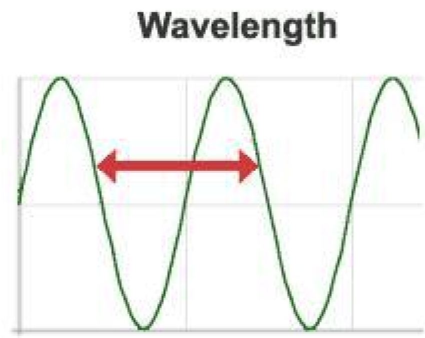
NdB =10 log (P/Pr)
NdB = Number of decibels
P = The power being measured
Pr = A reference power level

A decibel is a unit of measurement which is used to indicate how loud a sound is. Continuous exposure to sound above 80 decibels could be harmful. a unit for measuring the loudness of sound (deci + bel unit of sound power (20-21 centuries), from Alexander Graham Bell (1847-1922), US inventor). Unit for measuring the relative intensities of sounds or the relative amounts of acoustic or electric power.

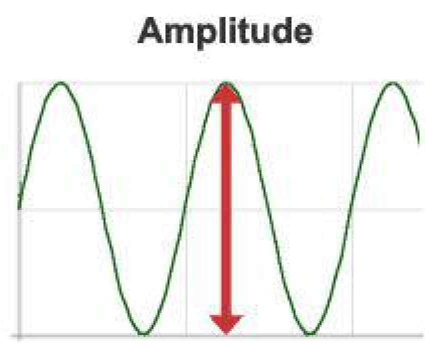
FUNDAMENTALS OF ACOUSTICS

Sound Wave Properties

All waves have certain properties. The three most important ones for audio work are shown here:



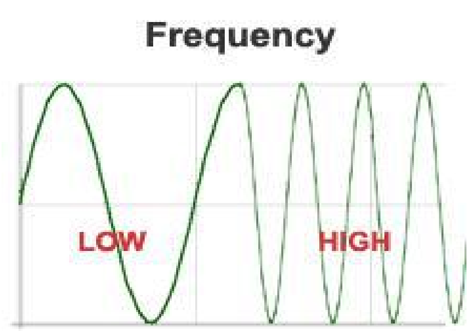
Wavelength: Wavelength is the distance between two corresponding points (for example two successive maxima) along the waveform. Literally, the length of the one wave.



Amplitude: Amplitude is the unit that measures the distance between the equilibrium point and the maximum point of the waveform. Greater amplitudes correspond to higher volumes..

The strength or power of a wave signal. The "height" of a wave when viewed as a graph.

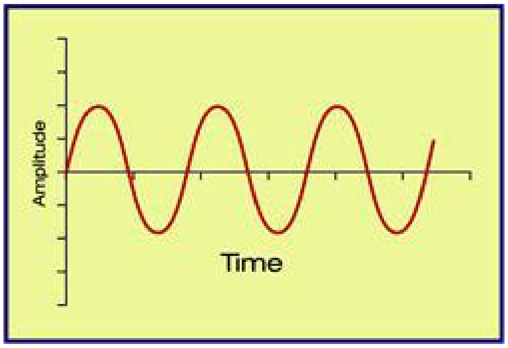
Frequency:



Frequency is literally the number of cycles made by a wave in one second. A cycle is composed of a positive half wave and a negative half wave. It is measured in Hz (1/sec). A 1Hz frequency wave completes one cycle every second.

Higher frequencies are interpreted as a higher pitch. For example, when you sing in a high-pitched voice you are forcing your vocal chords to vibrate quickly.

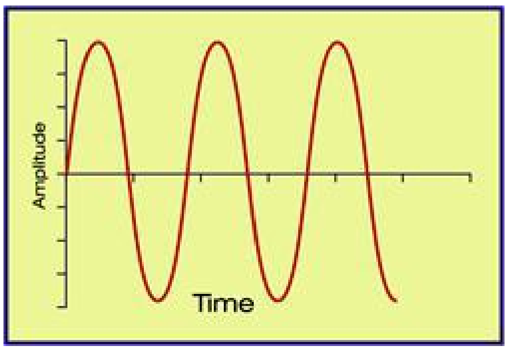
LOW-FREQUENCY



The range of low frequency 30 Hz to 800 Hz

Long wave form

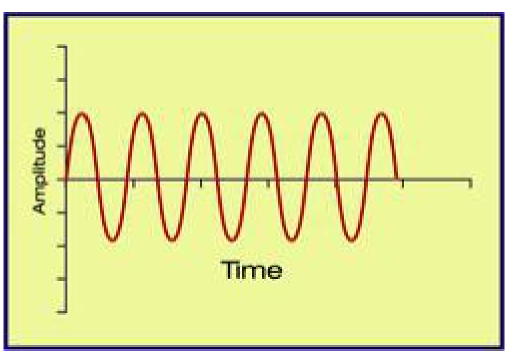
MID-FREQUENCY



The range of low frequency 800 Hz to 2200 Hz

Medium wave form

HIGH-FREQUENCY



The range of low frequency 2200 Hz to 8000Hz

Short wave form

<u>Very high frequency range is 8000 Hz to 18000 Hz</u>

Threshold of Hearing

The measured threshold of hearing curve shows that the sound intensity required to be heard is quite different for different frequencies. The standard threshold of hearing at 1000 Hz is nominally taken to be 0 dB, but the actual curves show the measured threshold at 1000 Hz to be about 4 dB. There is marked discrimination against low frequencies so that about 60 dB is required to be heard at 30 Hz.

Psycho-acoustics

1. The frequency range of the voice is typically only from about **500 Hz to 4 kHz**

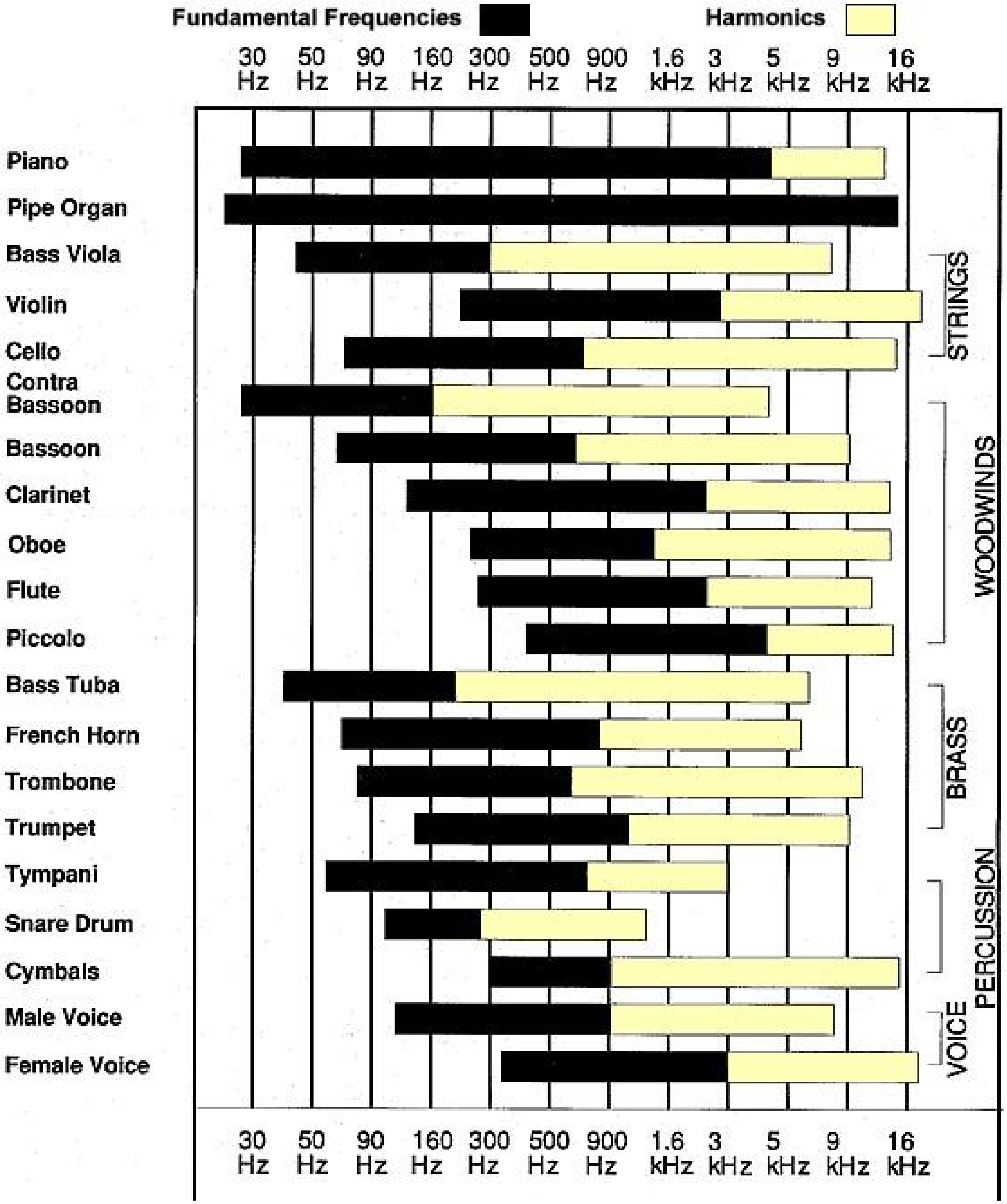
2. The normal range of frequencies audible to humans is **20 to 20,000 Hz.**

Threshold of hearing range chart

Source	Intensity	Intensity Level	# of Times Greater Than TOH
Threshold of Hearing (TOH)	$1*10^{-12}$ W/m^2	0 dB	10^0
Rustling Leaves	$1*10^{-11}$ W/m^2	10 dB	10^1
Whisper	$1*10^{-10}$ W/m^2	20 dB	10^2
Normal Conversation	$1*10^{-6}$ W/m^2	60 dB	10^6
Busy Street Traffic	$1*10^{-5}$ W/m^2	70 dB	10^7
Vacuum Cleaner	$1*10^{-4}$ W/m^2	80 dB	10^8
Large Orchestra	$6.3*10^{-3}$ W/m^2	98 dB	$10^{9.8}$
Walkman at Maximum Level	$1*10^{-2}$ W/m^2	100 dB	10^{10}
Front Rows of Rock Concert	$1*10^{-1}$ W/m^2	110 dB	10^{11}
Threshold of Pain	$1*10^{1}$ W/m^2	130 dB	10^{13}
Military Jet Takeoff	$1*10^{2}$ W/m^2	140 dB	10^{14}
Instant Perforation of Eardrum	$1*10^{4}$ W/m^2	160 dB	10^{16}

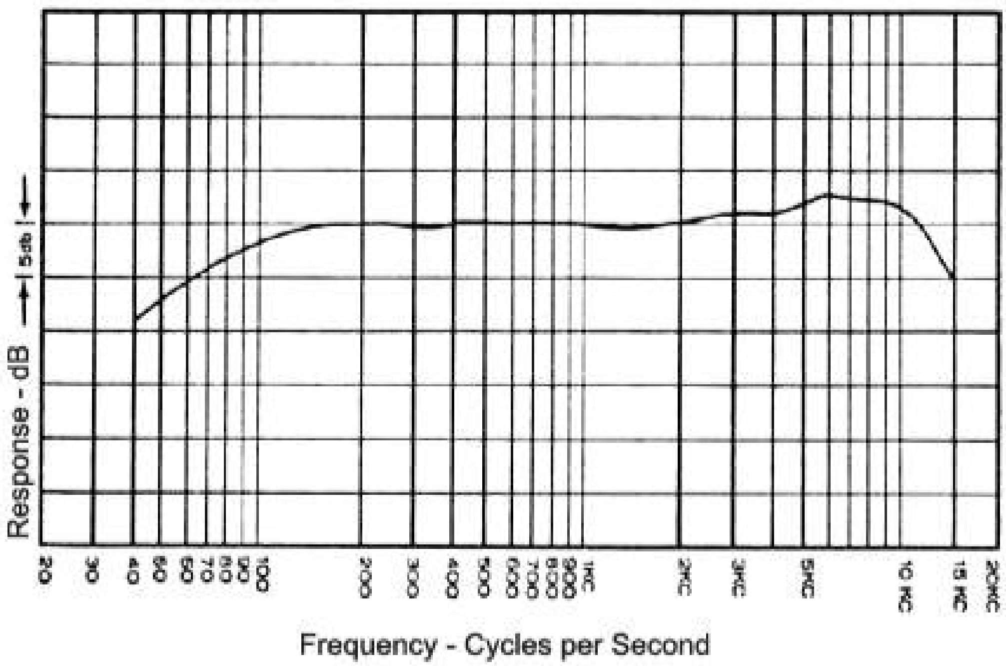
Every microphone has a signature and part of that signature is its Frequency Response. Frequency response determines the basic “sound” of the microphone. It is determined by the range of the sound (from lowest to highest frequency) that a microphone can reproduce and how that sound varies at different frequencies.The most common response curves you are likely to see are flat and tailored. When you look through catalogs or web pages, you’re probably going to see icons that look something like these.

Frequency Range of Various Musical Instruments



Frequency Response Charts

A microphone's frequency response pattern is shown using a chart like the one below and referred to as a frequency response curve. The *x* axis shows frequency in Hertz, the *y* axis shows response in decibels. A higher value means that frequency will be exaggerated; a lower value means the frequency is attenuated. In this example, frequencies around 5 - kHz are boosted while frequencies above 10kHz and below 100Hz are attenuated. This is a typical response curve for a vocal microphone.



Frequency range of various musical instruments

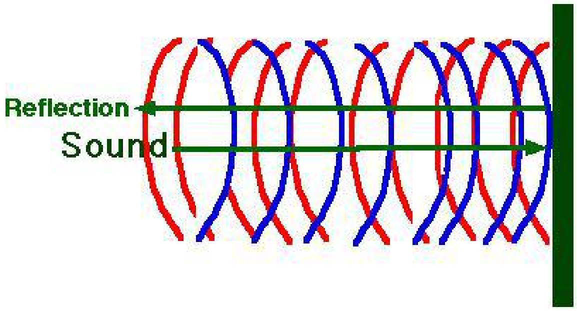
The Approximate Frequency Ranges chart below displays the frequencies generated by some familiar musical instruments (including our voices) with BOTH the numerical fundamental and harmonic approximate frequencies shown.
Given the often-talked-about musical range of 20hz-20khz, it is surprising to see just how low the musical fundamental frequencies actually are (almost all are under 3,500khz)

My chart above shows the approximate frequency ranges of various musical instruments and the human voice. The black boxes represent their fundamental frequencies and the yellow boxes represent their harmonic frequencies. It's much easier to understand and enhance the instruments you want to hear when you know what frequencies they cover.

Echo

(A repetition of sound) **Echo's,** one or a few at most repetitions of an audio signal. Not all sound that hits matter is absorbed. Some of it is reflected. That means sound bounces off the solid matter the way a tennis ball bounces off a wall. Sound reflected back to its source is an echo.

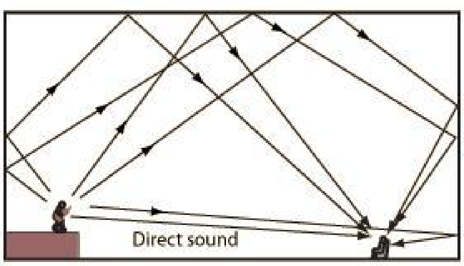
1.One repetitions of an audio signals called as echo



Reverberation

Reverberation is the collection of reflected sounds from the surfaces in an enclosure like an auditorium. It is a desirable property of auditoriums to the extent that it helps to overcome the inverse square law drop-off of sound intensity in the enclosure. However, if it is excessive, it makes the sounds run together with loss of articulation - the sound becomes muddy, garbled. To quantitatively characterize the reverberation, the parameter called the reverberation time is used.

2.Many repetitions becoming more closely speed (denser) with time



(**Reverberation sound is the collection of all the reflected sounds in an auditorium)**

Delay

3.The time interval between a direct signal and its echoes (to postpone to a later date)

Decay

4. The time it takes for the echoes and reverberation to die away (Progressive decline.

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UNIT – II - Audio Production Theory – SVCA1301

Microphones are a type of **transducer** or convector - a device which converts energy from one form to another. Microphones convert acoustical energy (sound waves) into electrical energy (the audio signal). A Basic Look at Microphones

Different types of microphone have different ways of converting energy but they all share one thing in common: The *diaphragm*. This is a thin piece of material (such as paper, plastic or aluminium which vibrates when it is struck by sound waves. In a typical hand-held mic like the one below, the diaphragm is located in the head of the microphone.

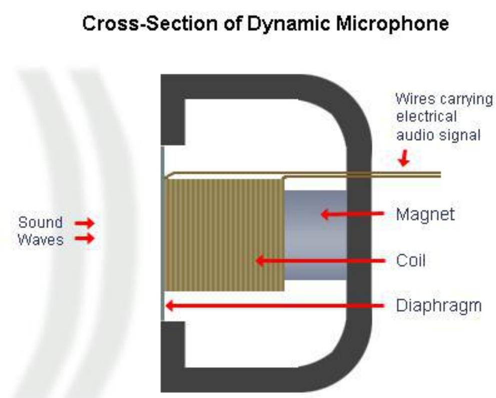
There are three common type of microphones

1. Dynamic microphone

2. Ribbon microphone

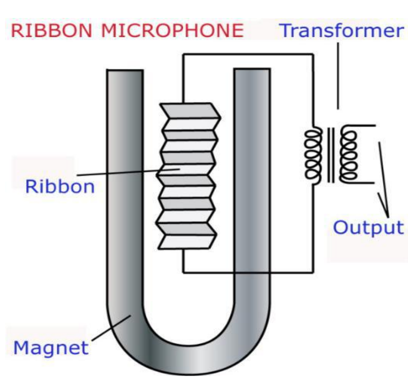
3. Condenser microphone…

1. Dynamic Microphone



Dynamic microphone, a very thin diaphragm of Mylar or other material is attached to a coil of hair-thin copper wire. The coil is suspended in a magnetic field and, when sound vibrates the diaphragm, the coil moves up and down, creating a very small electrical current.

Note: At the other end of the audio chain, the loudspeaker is also a transducer - it converts the electrical energy back into acoustical energy.



Principle :

A ribbon microphone is a unique type of dynamic microphone that is based around a thin, corrugated strip of metal (often aluminum) or film suspended between two magnetic poles. Unlike traditional moving-coil dynamic mics, the ribbon element responds to variations in the *velocity* of air particles, rather than the *pressure*. As the ribbon vibrates within its magnetic field, it generates a tiny voltage that corresponds to these changes in velocity. In classic ribbon designs, this level is very low compared to typical dynamic mics, and a step-up transformer boosts both the output voltage and impedance. Pre-amp choice is very important when using ribbon mics.

Because a ribbon mic has an extremely thin, delicate element, it is capable of capturing fast transients. Ribbons mics have a wide dynamic range, and are capable of handling high SPL s at high frequencies.

Ribbon mics are very sensitive, but they are often quite fragile; delicate older models can be broken by strong gusts of air, voltage spikes or even by being stored on their side.

Condenser Microphones

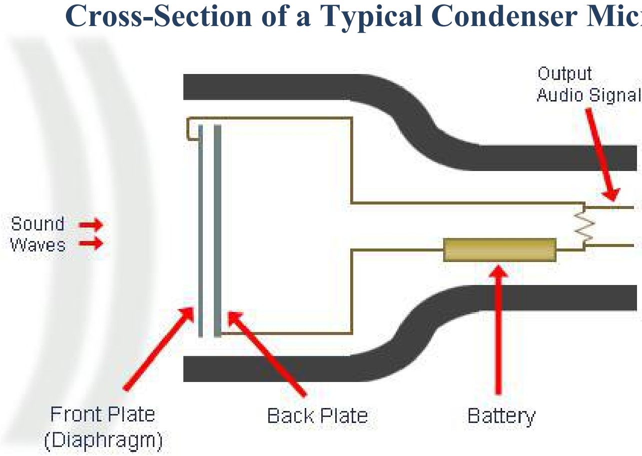
Condenser means *capacitor*, an electronic component which stores energy in the form of an electrostatic field. The term *condenser* is actually obsolete but has stuck as the name for this type of microphone, which uses a capacitor to convert acoustical energy into electrical energy.

How Condenser Microphones Work

A capacitor has two plates with a voltage between them. In the condenser mic, one of these plates is made of very light material and acts as the diaphragm. The diaphragm vibrates when struck by sound waves, changing the distance between the two plates and therefore changing the capacitance. Specifically, when the plates are closer together, capacitance increases and a charge current occurs. When the plates are further apart, capacitance decreases and a discharge current occurs.

A voltage is required across the capacitor for this to work. This voltage is supplied either by a battery in the mic or by external **phantom power (48v dc).**

Cross-Section of a Typical Condenser Microphone



Microphone Directional Characteristics

Microphones: Polar pattern / Graphical representation of microphone's pick-up pattern or (Polar pattern)

The pickup, or polar pattern, of a microphone is the shape of the area around it where it picks up sounds with maximum fidelity and volume. Nearly all microphones can pick up sounds from areas outside the ideal pattern, but their quality is not as good. For best results, the sound source should be within the pickup pattern, generating enough volume to allow the audio switcher to keep the volume control pot at a minimal level.

Microphones are classified according to the following three basic polar patterns.

There are three common type of polar pattern

1. Uni-Directional

2. Bi- Directional

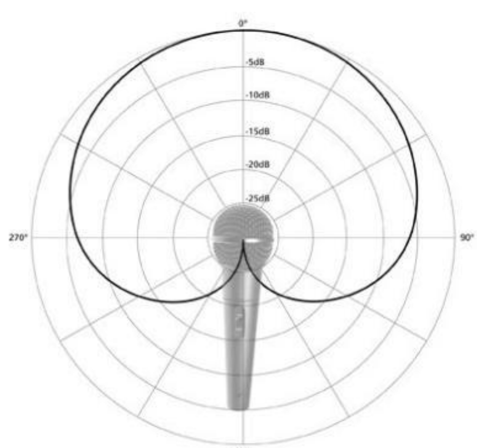
3. Omni- Directional

Uni- Directional

Cardioid Microphone

The unidirectional microphone picks up sound from only one direction. Because of this characteristic, the unidirectional microphone is used most frequently for television work. It is used by aiming it in the direction of the sound source being recorded. One advantage to the unidirectional microphone is its ability to reject unwanted sounds at the side and rear of the direction the microphone is aimed.

Front side (pick up)

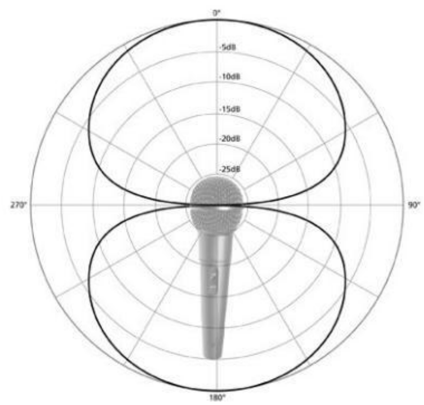


Bi-directional

As the name implies, the bidirectional microphone picks up sound in two directions. This type of microphone is used primarily in the broadcast or recording studio. It is also used for critical sound reinforcement applications in which front and rear.

Figure of Eight (8)

Front side (pick up)

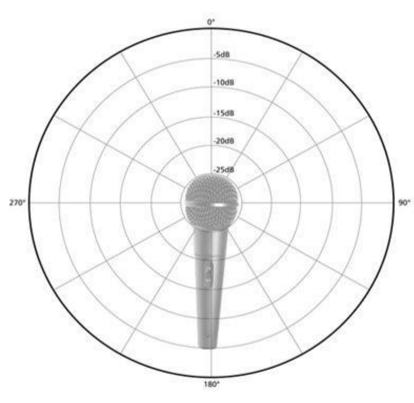


Rear side (pick up)

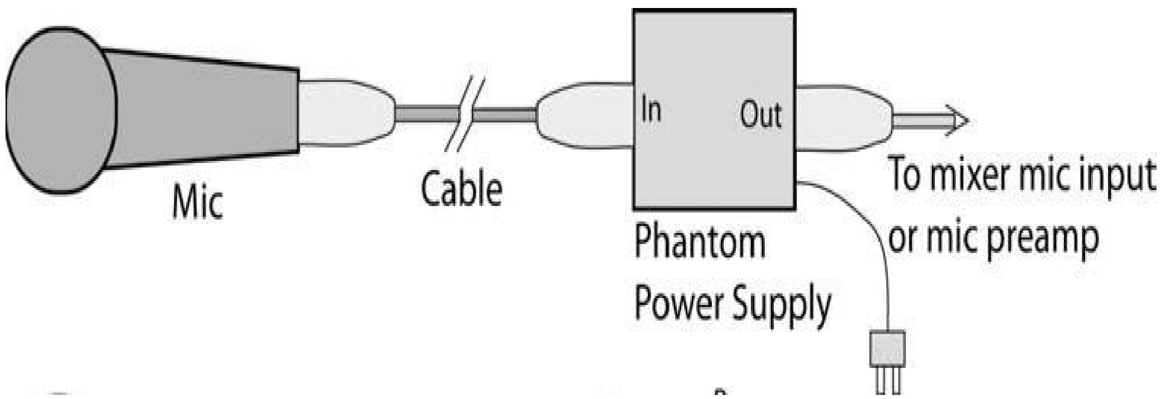
Omni- directional

The Omnidirectional (or all directional) micro- phone is live in all directions. This type of microphone has sensitivity characteristics in which sound is picked up in a 360-degree radius. The use of this microphone in television production is limited; however, in certain situations, you may use it to create a specific sound presence. One example is recording crowd noise for a sports production.

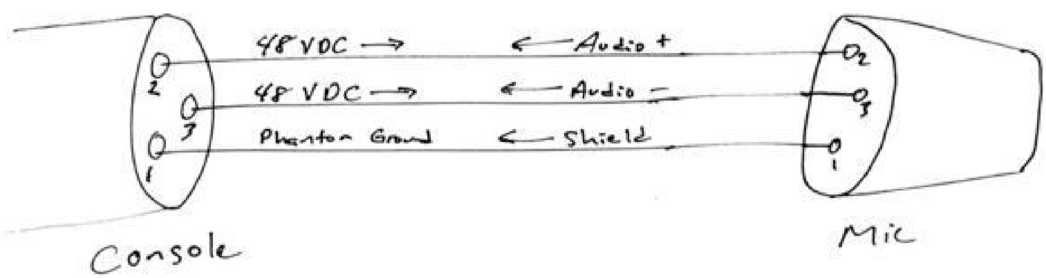
360 degree (pick up)



PHANTOM POWER SUPPLY



Condenser microphones need phantom power to operate their internal circuitry. Phantom power is a DC voltage—typically 48 volts— that is applied to both signaling lines of a balanced connector. In practical terms, 48 volts are applied to both pin 2 and pin 3 of an XLR. As both pins are getting the same voltage, in the same polarity, it's cancelled out at the input transformer.



Audio signals are generated based on the differences between the "high" or "hot" (pin 2) and the "low" or "cold" (pin 3) signals; thus if the same voltage (48) is appearing on them at the same time, it is effectively 0 volts.

That is to say, there is no delta or difference between them, so to the input amplifier in the mixer, there is no signal.Since phantom power is direct current, it can coexist on the audio lines with no problem. A condenser mic is free to pull power from either pin 2 or pin 3, and the current returns to ground over the shield

How is Phantom Power Generated?
Phantom power can be generated from sound equipment such as mixing consoles and Pre -amplifiers. Special phantom power supplies are also available

Microphone placements for various music instruments

Getting great live sound is the result of many factors. Not the least of these is mic placement. So if you're not playing arenas with a full sound crew well versed in these techniques, we can help with some useful tips and guidelines.

In this issue, we'll look at a typical rock line-up - guitar, keyboards, sax and drums. A good place to start is by listening to the sound of the instrument or the amplifier you are miking. How does the sound radiate? Listen up close, and then a few feet away. The closer a microphone is placed to the instrument (to maximize signal level and reduce pick-up of unwanted sounds), the more important placement becomes. Every instrument and microphone has its own characteristics. Every musician has his or her own idea of what sounds good. Experiment and listen.

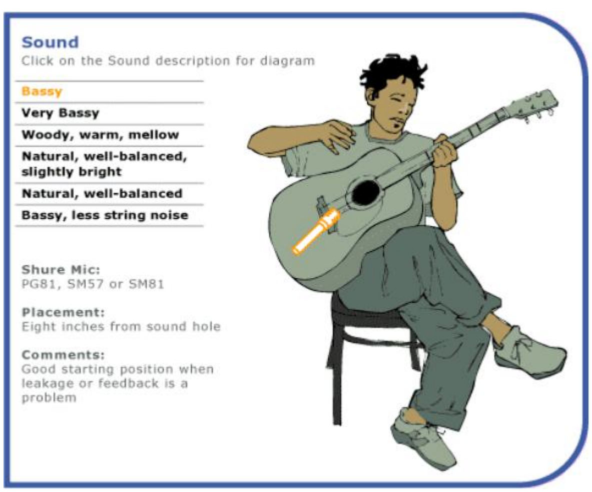
General Rules for Placement

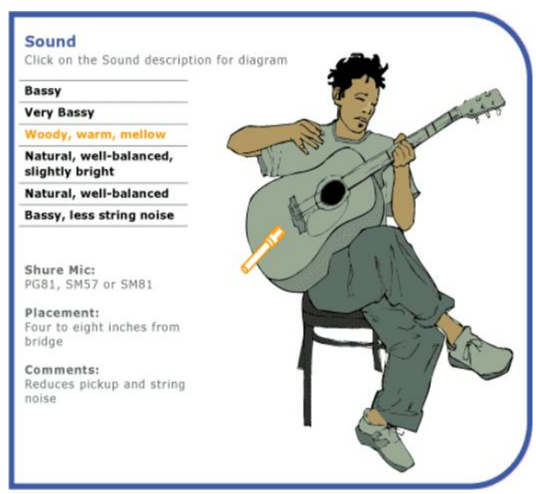
A major difference between miking for live sound versus recording is proximity to the sound source. The goal in live sound is to get the mic as close as possible to the source for two reasons:

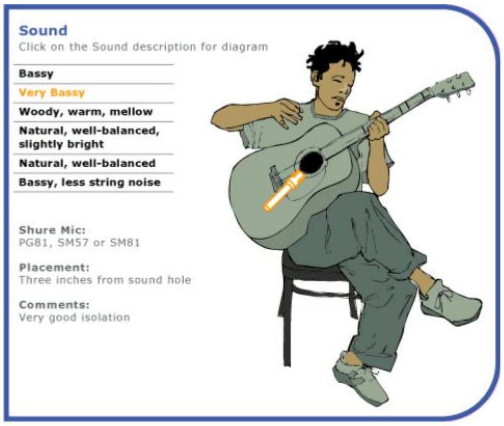
1. Placing microphones as close as possible achieves maximum sound level before feedback occurs.

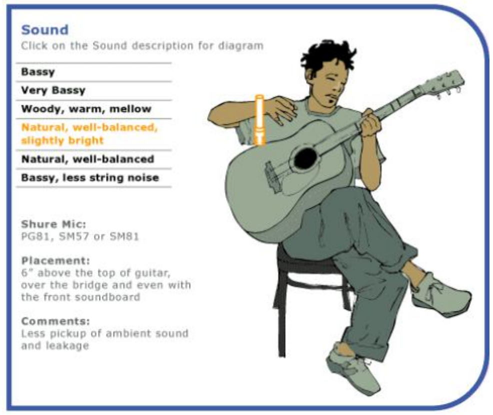
2. Close-miking reduces leakage and pick-up of unwanted sources.

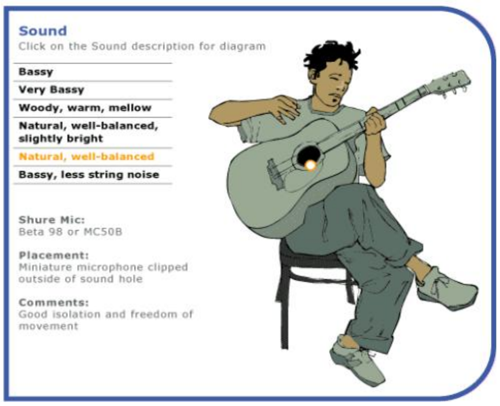
1. Various types of Microphone placement for Acoustic guitar

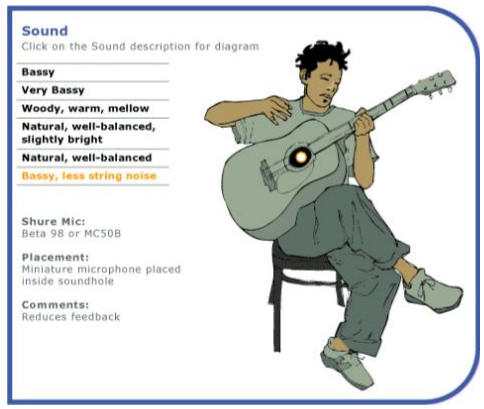




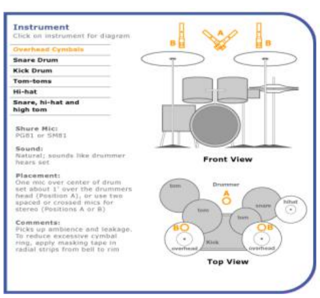


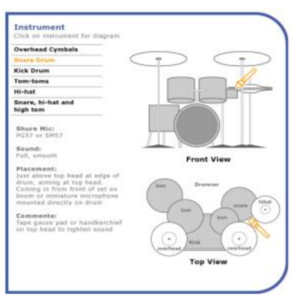


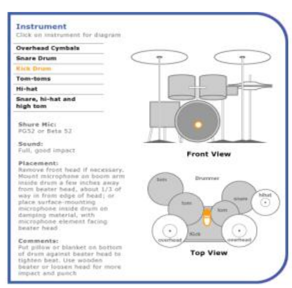


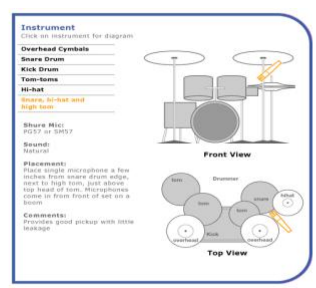


2. Various types of Microphone placement for drum kit (Percussion Instruments)









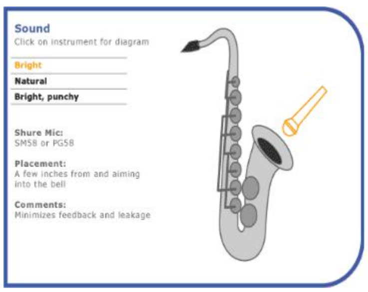
Drum Kit

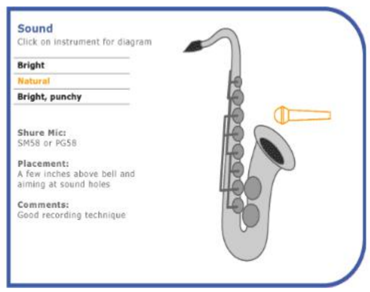
In most live sound systems, the drum set is miked with each drum having its own mic. Using microphones with tight polar patterns on toms helps to isolate the sound from each drum. It's possible to share one mic with two toms, but then, a microphone with a wider polar pattern should be used. The snare requires a mic that can handle a very high SPL (sound pressure level) so a dynamic mic is usually the choice. To avoid picking up the hi-hat in the snare mic, aim the null of the snare mic toward the hi-hat. The brilliance and high frequencies of cymbals are picked up best by a flat-response condenser mic.

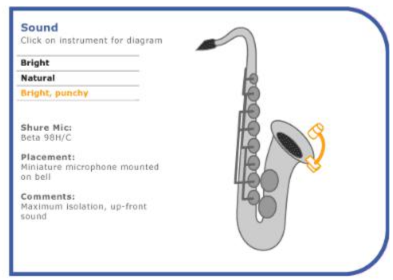
Saxophone

Like the electric guitar, the sax has sound characteristics similar to the human voice. And that's why the shaped response of a dynamic microphone is generally preferred. However, a miniature condenser microphone mounted on the bell often does the trick. The sound is fairly well distributed between the finger holes and the bell. Miking close to the finger holes produces key noises, so generally mics are placed toward the middle of the instrument. (Note: this technique does not apply to the soprano sax, since its bell does not curve upward - therefore, miking in the middle of the instrument won't pick up sounds from both the key holes and the bell.)

3. Various types of Microphone placement for Saxophone







CORDLESS MICROPHONE

A wireless microphone that is connected to a high-power transmitter in a fixed location. The transmitted voice is picked up by an FM receiver and heard through a speaker or headset .In a wireless microphone system, the components are miniaturized but the same principles apply. The transmitter is small enough to fit into the microphone handle or into a small pocket-sized case. Since the microphone and transmitter are battery powered, the user is free to move around while speaking or singing into the mic. The transmitted voice is picked up by a receiver that is wired to a speaker.

1. Handled cordless microphone
2. Lavaliere microphone or lapel mic

Two types of microphones are available with wireless mic systems: **Handheld mic,** with a transmitter in its handle and another **lavaliere mic or collar mic** which is small enough to be concealed as a lapel pin or hung around the neck. Lavaliere mics are wired to miniature body-pack transmitters, which fit into a pocket or clip onto a belt.

Wireless Microphones Transmitters Hand held wireless microphones (Fig1) have conventional microphone elements mounted to a handle into which a miniature radio transmitter and mic preamp are built. Several very good vocal performance microphones elements (and a lot more mediocre ones) are available on wireless transmitters from at least a dozen manufacturers

Lavaliere microphones (Fig 2) are also known as *lav*, *lapel* or *lap* microphones. A lavaliere mic is a very small condenser mic designed to pick up speech from a single person. Lavaliere mics are usually attached to the subject's clothing with a specific clip. Obviously the preferred position is on the lapel or thereabouts. This provides consistent close-range sound pickup and is ideal for interview situations in which each participant has their own mic.

<u>Usage of Cordless microphone</u>

Wireless microphones are widely used today in television and video Production. They eliminate the need for stage personnel to feed cables around cameras, props, etc. For location film production, as well as **ENG** (Electronic News Gathering) and **EFP** (Electronic Field Production)

<u>Contact microphone</u>

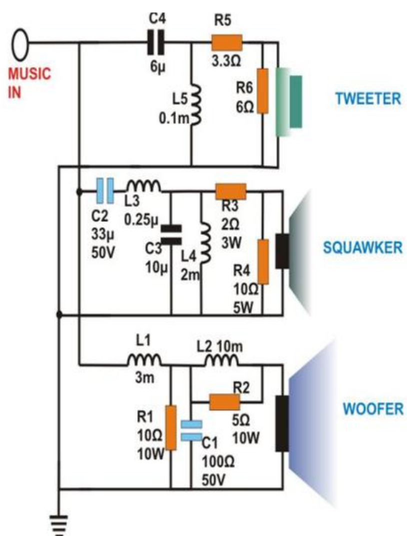
A contact microphone functions differently than dynamic, ribbon, and condenser microphones. Though they do function similarly to the microphones mentioned above in that they convert sound pressure waves into electrical energy. Contact mics pick up vibrations from the surface that they are placed on. They then convert the vibrations into a voltage that is passed into an appropriate preamp to boost the signal so that it is at a volume level that is suitable for recording. It is important to use a preamp designed for use with a contact microphone to attain the full frequency spectrum capable of the device

Speakers

A loudspeaker is simply a device that converts electrical energy into sound that is amplified so that it can be heard from a greater distance than the original sound would allow. There is no difference in usage of the terms *speaker* and *loudspeaker* and both are often used interchangeably. Some loudspeakers are capable of producing sounds over a wide range of frequencies and some are only made to reproduce certain frequencies.

There are three common type of Speakers

1. **Woofer – Sub woofer** **(Re-produce the low and very low frequency)**
2. **Squawker** **(Re-produce the low mid and mid frequency)**
3. **Tweeter** **(Re-produce the high and very high frequency**



A three-way speaker system can give a fuller range of sound compared to a two-way system. In a two-way system, the bass ranges go to the woofer, and the treble goes to the tweeter. Three-way speakers include a mid-range driver that allows the frequencies between bass and treble to be heard more clearly than without the mid-range driver. Since three-way speakers have more drivers than a two-way, they are larger and commonly found as floor models. One of the benefits of using floor, or tower, speakers is that they provide big, room-filling sound.

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Fundamental recording techniques

Recording Technique Tips :

1. Make sure the singer is well rehearsed, physically comfortable, and under no psychological pressure. Most singers perform best standing up in a room that has a comfortable but not over-warm temperature

2. Take time to get the vocalist's headphone mix right, and give them a little reverb to help them sing more confidently. If you can rig up a system which allows vocalists to adjust their own monitor level, it will make life a lot easier. A good headphone mix really helps to encourage a good performance.

3. Always use a pop shield between the singer and the microphone. Failure to do so will almost certainly result in unnatural 'pops' on plosive 'b' and 'p' sounds that can't be fixed afterwards.

4. Use a good microphone: it doesn't have to be anything too special, but you should avoid low-cost 'bargain' models or those designed for use with home stereos or portable cassette recorders. Professional studios generally use capacitor microphones, but in the project studio a good back-electret mic or even a good dynamic vocal mic can produce excellent results.

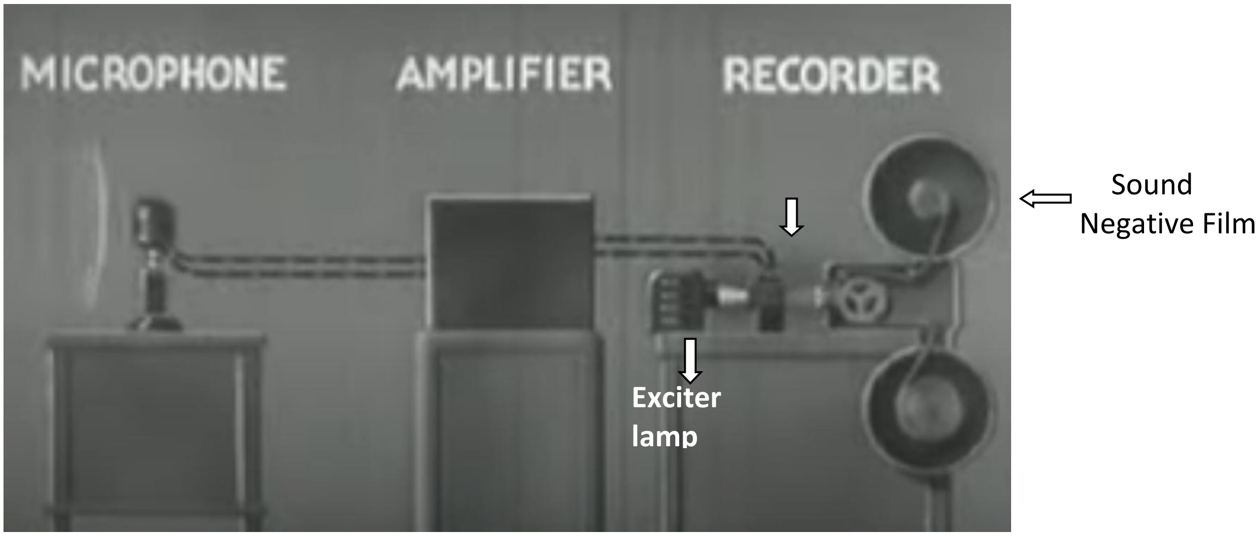
5. Pick a mic to suit the singer. Singers with thin or excessively bright voices may actually sound better with a dynamic mic, such as the ubiquitous Shure SM58

6. Use the right mic pickup pattern: most project studio vocal recordings are made using a cardioid or unidirectional mic, as these pick up less sound from the sides and rear. However, an Omni mic of a similar quality generally imparts a more natural,. If you work a couple of inches closer to an Omni mic, you'll get close to the same 'direct sound to room sound' ratio you'd achieve with a cardioid.

7. Put the mic at the right distance, because if you get too close to it you'll increase the risk of popping and the level will change noticeably every time the singer moves slightly. As a rule, a mic distance of around six to nine inches (15-24 centimeters) is ideal.

8. Minimize the room's influence on your sound. The mic picks up both direct sound from the singer and reflected sound from the room. Reduce the room's contribution by keeping away from the walls and by improvising screens using sleeping bags or duvets behind and to the sides of the singer.

Optical Recording



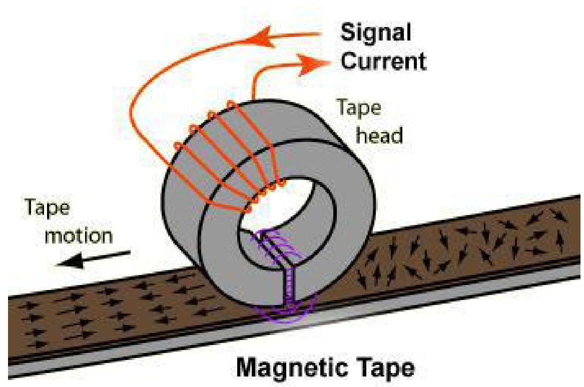
Methods of Optical Recording of Sound on Film :

In this method,sound is picked up by microphone,and converted into electrical signals which are amplified.Audio o/p of the amplifier is fed to the anode of special type of vacuum tube,called an AEO lamp.The lamp contains a little quantity of helium gas.The anode gets high dc voltage in series with the audio voltage.The filament of the lamp is connected to a low dc voltage The intensity of light coming out from the lamp varies in accordance with the audio signal.This varying light passes through a slit and a focusing lens.The focused light falls on moving photographic film where the image is recorded.

Sound For optical recording on film there are two methods utilized. Variable density recording uses changes in the darkness of the soundtrack side of the film to represent the sound wave. Variable area recording uses changes in the width of a dark strip to represent the sound wave.

Electrical signal has been converted as light Energy by photoelectric cell and light energy which is going to stored in sound negative or film

Magnetic sound Recording



High fidelity tape recording requires a high frequency biasing signal to be applied to the tape head along with the signal to "stir" the magnetization of the tape and make sure each part of the signal has the same magnetic starting conditions for recording.

Sound signal can be stored on tape in the form of magnetized iron oxide or chromium dioxide granules in a magnetic emulsion. The tiny granules are fixed on a polyester film base, but the direction and extent of their magnetization can be changed to record an input signal from a tape head. Sound energy has been converted as electrical energy by microphone. An electrical energy which is going to stored or recorded in magnetic tape acordding to the ac bias current.

Digital recording

The most common digital sound recording method is pulse code modulation OR (PCM)

Process of Digital recording

1. The sound signal from your audio mixer is run through a low pass filter which removes all frequencies above 20 kHz

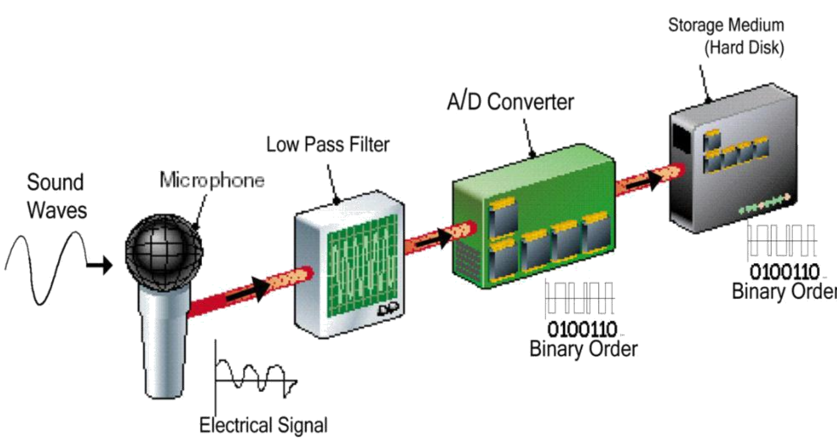
2. Next, the filtered signal passes through an analog-to-digital (A/D) Converter. An Analogue to Digital (A/D) converter converts analogue signal into a digital number. This converter measures sample the voltage of the audio wave form several thousand time

3. Each time wave form is measured, a binary number made (1, 0.s) is generated that represents the voltage of the wave form at the instants its measured this process is called quantization. Each 1 and 0 is called a bit, which stands for digit.

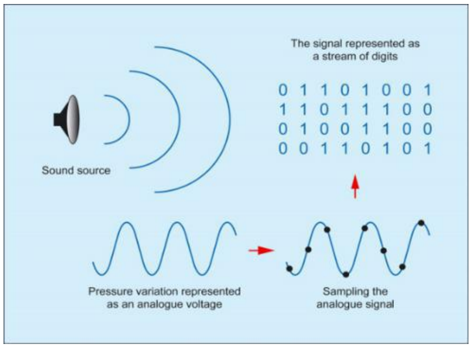
4. The binary numbers are stored on the recording medium as a modulated square wave recorded at maximum level.

5. Storage of digital data can be performed on magnetic tape, optical disk, magnetic disk, or RAM (Random Access Memory

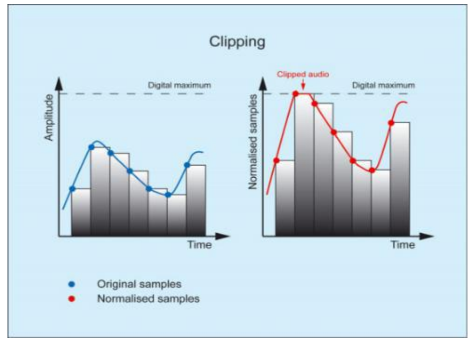
Block Diagram Of Digital Recording



Sound signal with binary order Conversion



Digital signal after A/D

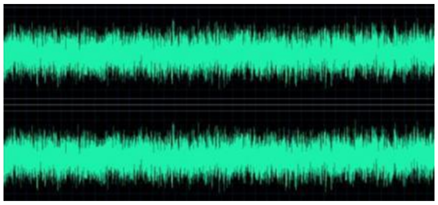


Mono recording is recording that is done on one single channel. This is in contrast to stereo recording, which is recording done on two separate channels composing of left and right sound inputs. Mono mixing is useful when only one source has been recorded a recording with a single microphone.

Stereo Recording

Stereo recording is recording onto two separate channels, one channel for the left sound input and the other channel for the right sound input. With stereo, recording on the two channels are independent of each other, and, thus, the channels can record completely different signals at a given time. This makes stereo recording dynamic, since it can produce different distinct sounds on the left channel and right channel.

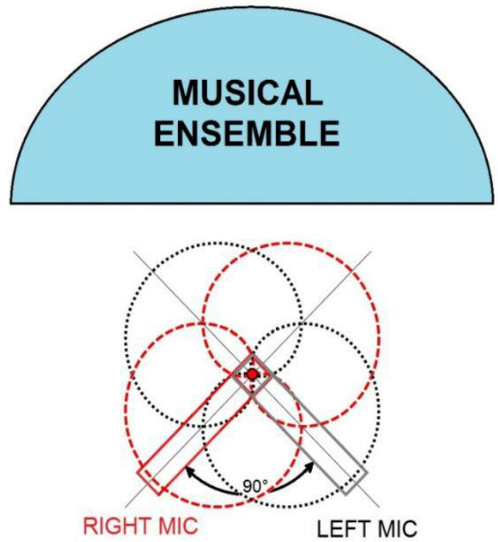
How can stereo recording be achieved? In order to record in stereo, a recording device must be used that has two microphones. Why? Because in order to record onto the two channels, two separate microphones are needed, one microphone for the left sound input channel and the other microphone for the right sound input channel. Speaking into the left microphone yields the left side input and speaking into the right microphone yields the right side input. The recording device that has two microphones that can achieve stereo recording is a 2-microphone_array device.



Left track (L)

Right track (R)

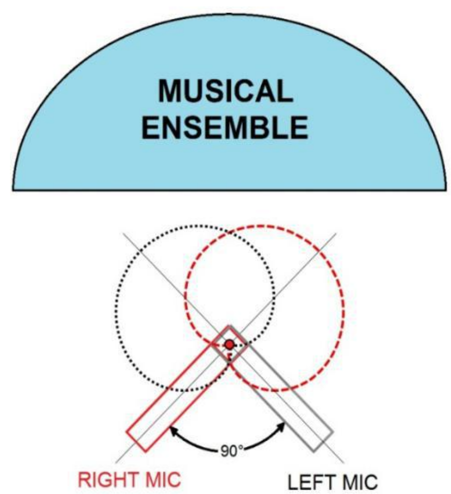
Stereo sonic Recording



Blumlein Pair

The stereo sonic technique) was invented by Alan Blumlein this method consists of a coincident pair of bi-directional (figure-eight) microphones placed at an angle of 90 degrees from one another, with the center line bisecting that angle pointing at the music source. This configuration provides a high degree of stereo separation along with a large amount of room ambience. The Blumlein technique produces a good, natural-sounding stereo image, but the sound quality is greatly influenced by the room acoustics and the size of the sound source. Since it is coincident, it provides excellent mono compatibility. Both condenser and ribbon microphones have been used with this technique.This technique should not be used if room acoustics or audience noise will be a problem.

X-Y Recording

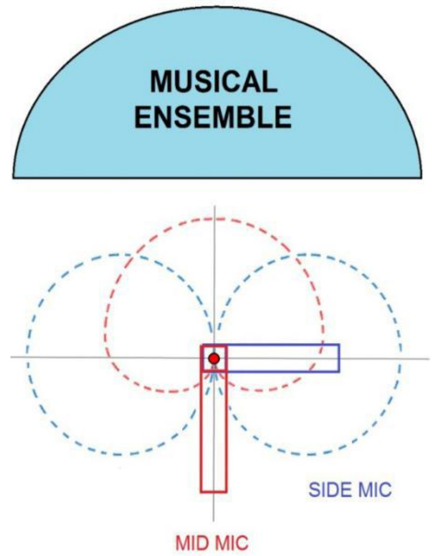


X-Y Pair

The X-Y technique uses two uni-directional microphones in a coincident configuration angled typically at 90 degrees apart (45 degrees to either side of a center line that faces the sound source). Angles from 90 to 135 degrees can be used and even as high as 180 degrees have been reported.6 The X-Y technique is sometimes used in the near coincident configuration with the two mics spaced about 12 inches apart.7 In the coincident mode phase cancellation problems are essentially nil since the capsules are so close together resulting in good mono compatibility

Mid-Side Recording

This coincident technique employs a bidirectional microphone facing sideways and a cardioid (generally a variety of cardioid, although Alan Blumlein described the usage of an omnidirectional transducer in his original patent) at an angle of 90° facing the sound source. One mic is physically inverted over the other, so they share the same distance.



Mid-Side Technique

The left and right channels are produced through a simple matrix: Left = Mid + Side, Right = Mid − Side (the polarity-reversed side-signal). This configuration produces a completely mono-compatible signal and, if the Mid and Side signals are recorded (rather than the matrix's Left and Right), the stereo width can be manipulated after the recording has taken place. This makes it especially useful for film-based projects.

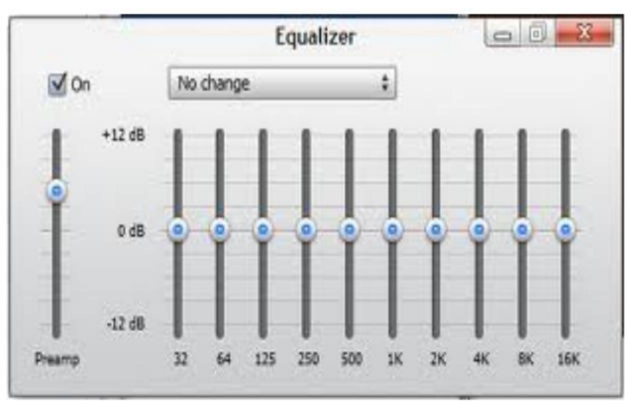
Audio Equalizer

Equalization, or ***EQ*** for short, means boosting or reducing (attenuating) the levels of different frequencies in a signal. Equalizer which is balancing or equalizing the frequency like low, mid and high.

The most basic type of equalization familiar to most people is the treble/bass control on home audio equipment. The treble control adjusts high frequencies; the bass control adjusts low frequencies. There are two types of equalizer in audio mixer

Parametric Equalizer

Graphical Equalizer



Parametric EQ

Parametric equalizers use bell equalization, usually with knobs for different frequencies, but have the significant advantage of being able to select which frequency is being adjusted.

Parametric are found on sound mixing consoles and some amplifier units (guitar amps, small PA amps, etc).

Graphic EQ

Graphic equalizers provide a very intuitive way to work — separate slider controls for different frequencies are laid out in a way which represents the frequency spectrum. Each slider adjusts one frequency band so the more sliders you have, the more control and we can adjust the particular frequencies.

Audio compression is a method of reducing the dynamic range of a signal. All signal levels above the specified threshold are reduced by the specified ratio.

Limiters are used as a safeguard against signal peaking (clipping). They prevent occasional signal peaks which would be too loud or distorted. Limiters are used in audio mixer board and other public address system

Difference between compressors and limiters

Compressor : An amplifier,whose gain decreases as its input level is increased.

Limiters : A compressor,whose output level remains constant regardless of its inputs level.

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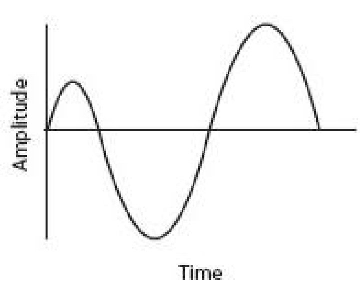
Sample Rates

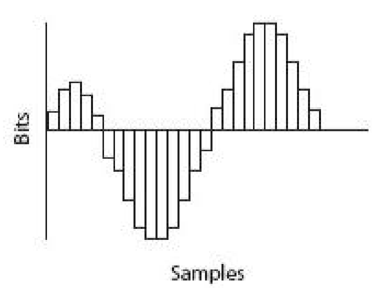
Sample rate is quality of audio and sampling frequency is the rate at which the A/D converter samples or measure the analog signal while recording Sample Rate refers to how many times per second the audio information is captured.

Sample rates are measured in Kilohertz (KHz). 1 KHz equals 1000 samples per second. So, audio recorded at 44.1 KHz captures data at a rate of 44,100 samples per second. (For example a rate of 48 kHz is 48000 samples per second that is 48000 measurements are generated for each second of sound signal.

Most audio interfaces support sample rates of 44.1, 48, 88.2, and 96 KHz. Many higher-end devices also support 176.4 and 192 KHz sample rates as well.

1. Analog sound signal **2.Digital sample audio**





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S.M.P.T.E. Time Code :

SMPTE Time code, is an acronym than stands for Society of Motion Picture Television Engineers. This group created a clock or time based standard first used in television for synchronizing video tape machines for editing purposes. The time code will be recorded on one audio track. Then, at playback time, the time code audio is read and decoded by a SMPTE time code reader device.

A S.M.P.T.E. display reads time as **00.00.00.00**. From left to right this hours, 0 minutes, 0 seconds, 0 frames, 0 sub frames. By comparing S.M.P.T.E tracks on two different machines a synchronizer can Lock or synchronize the two machines. This is done by establishing a Master machine whose timing is imposed on a Slave machine by the synchronization.

Audio Mix down :

Mixing down is the process of recording the output from multiple tracks to a stereo or mufti-channel format. This process is also often referred to as bouncing, which traditionally has been done to combine several tracks together to free up resources or reduce track count. Mix down is often the last phase of music production, but in Pro Tools mix down can be done any time you want to bounce tracks or create a completed mix for use outside of your session.

Mono mix down The number of audio track mix as a single track called **(Mono**) mix

down mono literally means "ONE"

one track. Down the middle. no left. no right. no panning.

Stereo mix down The number of audio tracks mix as two track called **(Stereo)** mix down

stereo has two tracks. One left, one right.

Once all your tracks are recorded and edited, it’s time for mix down here is the general

procedure;

1. Balance the all audio tracks and set the level 0db.
2. Adding EQ, Effects and Panning to audio tracks.
3. Set up the automation (Mixing the Dialogue, Effects, BGM)
4. Once your mix is perfected, Export it to a mono or stereo file.

Audio file formats

The Audio Files category includes compressed and uncompressed audio formats, which contain waveform data that can be played with audio playback software. This category also includes MIDI files, musical scores, and audio project files, which typically do not contain audio data.

Common audio file extensions include **.WAV, .AIFF, MP3, and MIDI.**

1. **Wave** (wav)

2. **AIFF** (Audio interchange file format)

3. **MIDI** (Musical Instruments digital Interface)

4. **MP3** (MPEG Level-1 Layer-3- Motion picture expert group)

5. **MP3** PRO (An improvement over MP3.Songs encoded at 64 kbps

6. **WMA** (Windows Media Audio)

7. **Real Audio** (Format is used for music downloads)

8. **AAC (**Advanced Audio coding)
   (MPEG) AAC offers better sound quality than MP.3 at same bit rate.

Blu-ray Disc (BD)

Blu-ray Disc (official abbreviation **BD**) is an optical disc storage medium designed to supersede the DVD format. The disc diameter is 120 mm and disc thickness 1.2 mm plastic optical disc, the same size as DVDs and CDs. Blu-ray Discs contain 25 GB (23.31 GB) per layer, with dual layer discs (50 GB) being the norm for feature-length video discs. Triple layer discs (100 GB) and quadruple layers (128 GB) are available for BD-XL Blu-ray re-writer drives. Currently movie production companies have not utilized the triple or quadruple layer discs; most consumer owned Blu-ray players will not be able to read the additional layers, while newer Blu-ray players may require a firmware update to play the triple and quadruple sized discs.
The name *Blu-ray Disc* refers to the blue laser used to read the disc, which allows information to be stored at a greater density than is possible with the longer-wavelength red laser used for DVDs.

Blu-ray Disc

Media type = High-density optical disc

Encoding = MPEG-2H.264/MPEG-4 AVCVC-1

Capacity = 25 GB (single-layer) 50 GB (dual-layer) 100/128 GB (BDXL)

Usage = Data storage High-definition video (1080p)
High-definition audio
Stereoscopic 3D
PlayStation 3 games

Developed by Blu-ray Disc Association

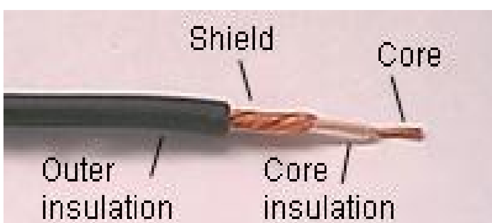
Audio Cables and Connectors :

Audio Cables

There are two main types of audio cable we will look at: Single core / shielded (unbalanced) and One pair / shielded (balanced).

Single Core / Shielded Cable

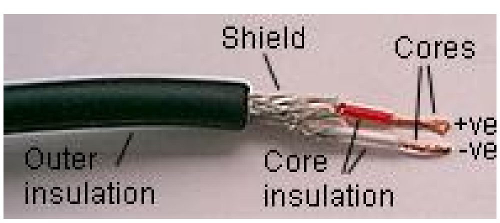
In a single core / shielded cable, the single core is used for the +ve, or 'hot', and the shield is used for the -ve, or 'cold'.
This type of cable is used for unbalanced audio signals.



Single core / shielded cable

One Pair / Shielded Cable

A one pair / shielded cable has one core as the +ve, and the other core is -ve. The shield is earthed.
This type of cable is used for balanced audio signals.



One pair / shielded cable

Audio Connectors :

There are a variety of different audio connectors available. The most common types are 3-pin XLR, RCA, and Phone jacks.

3-PIN XLR

3-pin XLR connectors are mainly used for microphone balanced audio signals. Using a balanced signal reduces the risk of inference.

- Pin 1 is the earth (or shield)
- Pin 2 is the +ve (or 'hot')
- Pin 3 is the -ve (or 'cold).

THERE ARE A NUMBER OF DIFFERENT XLR'S - 3-PIN, 4-PIN, 5-PIN ETC

3-PIN XLR FEMALE(LEFT) **3-PIN XLR MALE(RIGHT)**	

1. The three hole of blue (left) is female connector. For Output communication.

2. The three pin of black (Right) is male connector. For Input communication.

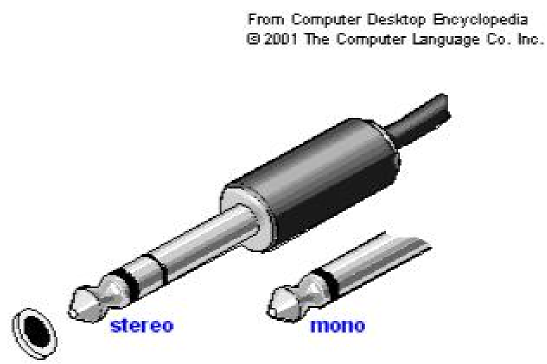
¼"Phono Jack / Connector (6.5mm Jack)

There are two types of phono Jacks: Mono and stereo. The mono jack has a tip and a sleeve, the stereo jack has ring, a tip and a sleeve.

- On the mono jack the tip is the +ve, and the sleeve is the -ve or shield.
- On a stereo jack being used for a balanced signal, the tip is the +ve, the ring is the -ve, and the sleeve is the shield.
- On a stereo jack being used for a stereo signal (left and right), the tip is the left, the ring is the right, and the sleeve is the shield.

Jacks also come in various sizes - phono (¼"), 3.5mm, 2.5mm. The wiring for all of them is the same.

1/4" Stereo Jack(left)
1/4" Mono Jack(Right)



RCA Connector :

RCA connectors are used a lot for home stereos, videos, DVD etc.

The RCA can carry either audio or video. It is wired the same way as a mono jack: The center pin is the +ve, and the outer ring is the -ve or shield.

- RCA Connector is commonly used to carry audio and video signals
- Yellow - Video
- White - Audio (L)
- Red - Audio (R)

Modern Recording studio console :

Mixing Consoles (Mixer Boards, or just Mixers) are the iconic symbol of the modern recording studio.

A Mixer can be divided into three sections

1. Input section

2. Output section

3. Monitor section

Input section

(Here are the main parts of each and what they do:

- Inputs connect to your mics, electric instruments, and recorder outputs.

- Faders are sliding volume controls that affect the loudness of each instrument.
- EQ knobs adjust the tone quality of each instrument (bass, treble, mid range)
- Aux knobs set the amount of reverb or other effects, and also can be used to set up a monitor mix or head phone mix.
- Pan pot can be used to set the left, centre, right.
- Channel assign buttons route each input signal to the desired recorder track.

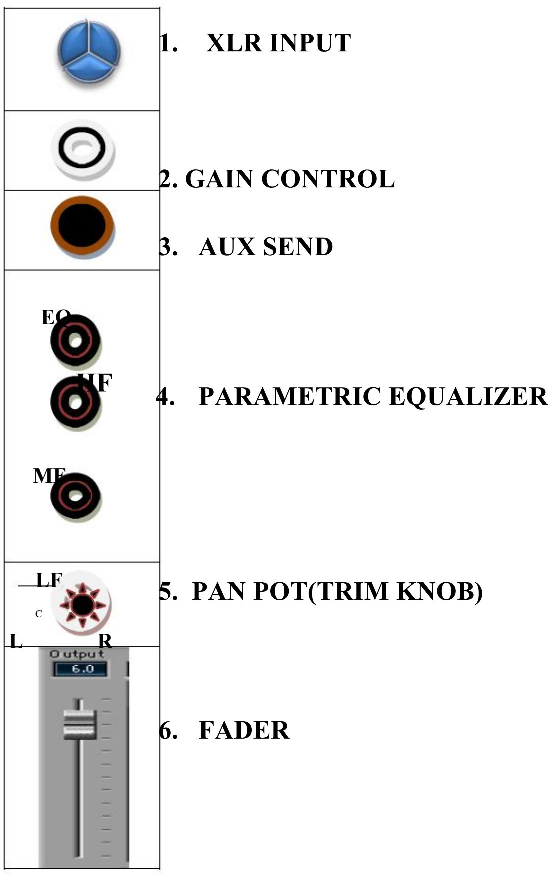
Output section

- Master fader set the overall of the entire stereo mix.
- Group faders set the overall of each group or sub mix.
- Output connects to your recorder inputs.
- Display Meters help you set the recording level (to prevent distortion and noise)
-

Monitor section

- Monitor controls select what you want to listen to.
- Aux knobs or channel faders set up the monitor mix.

Typical input module



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A **Guide track** or **pilot track**

In film making, guide tracks are commonly used as an aid in lip synchronization, timing actions such as dance, and when musicians appear on camera. These tracks may be completed recordings, and are used to aid the actors in creating the illusion of live performance. See also dubbing (film making).

Major reference :

Part or all of the dialogue in a scene may have to be added during post production. Production sound is used as a cue or guide track for replacing dialogue, a procedure commonly known as dubbing, or looping.

Dubbing :
in film-making, the process of adding new dialogue or other sounds to the sound track of a motion picture that has already been shot. Dubbing is most familiar to audiences as a means of translating foreign-language films into the audience's language. When a foreign language is dubbed, the translation of the original dialogue is carefully matched to the lip movements of the actors in the film. Dubbed sound tracks rarely equal the artistic quality of original foreign-language sound tracks, however, and hence subtitles may be preferred by viewers as a means of understanding the dialogue in foreign films.

Effects Recording :

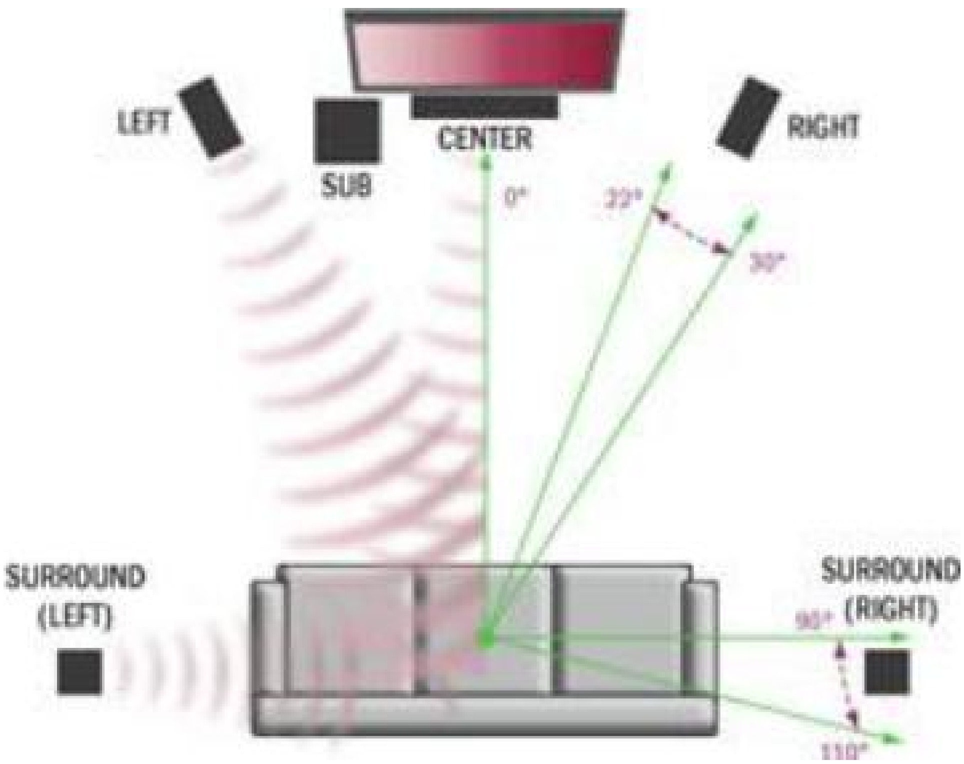
The most realistic sound effects originate from original sources; the closest sound to machine-gun fire that we can replay is an original recording of actual machine guns. Less realistic sound effects are digitally synthesized or sampled and sequenced (the same recording played repeatedly using a sequencer). When the producer or content creator demands high-fidelity sound effects, the sound editor usually must augment his available library with new sound effects recorded in the field.

When the required sound effect is of a small subject, such as scissors cutting, cloth ripping, or footsteps, the sound effect is best recorded in a studio, under controlled conditions. Such small sounds are often delegated to a Foley artist and Foley editor. Many sound effects cannot be recorded in a studio, such as explosions, gunfire, and automobile or aircraft maneuvers. These effects must be recorded by a sound effects editor or a professional sound effects recordis.

Re -recording is the process by which the audio track of a film or video production is created. As sound elements are mixed and combined together the process necessitates "re-recording" all of the audio elements, such as dialogue, music, sound effects, by the sound re-recording mixer(s) to achieve the desired end result, which is the final soundtrack that the audience hears when the finished film is played.

5.1 dts digital surround alignments :

Surround-sound setup



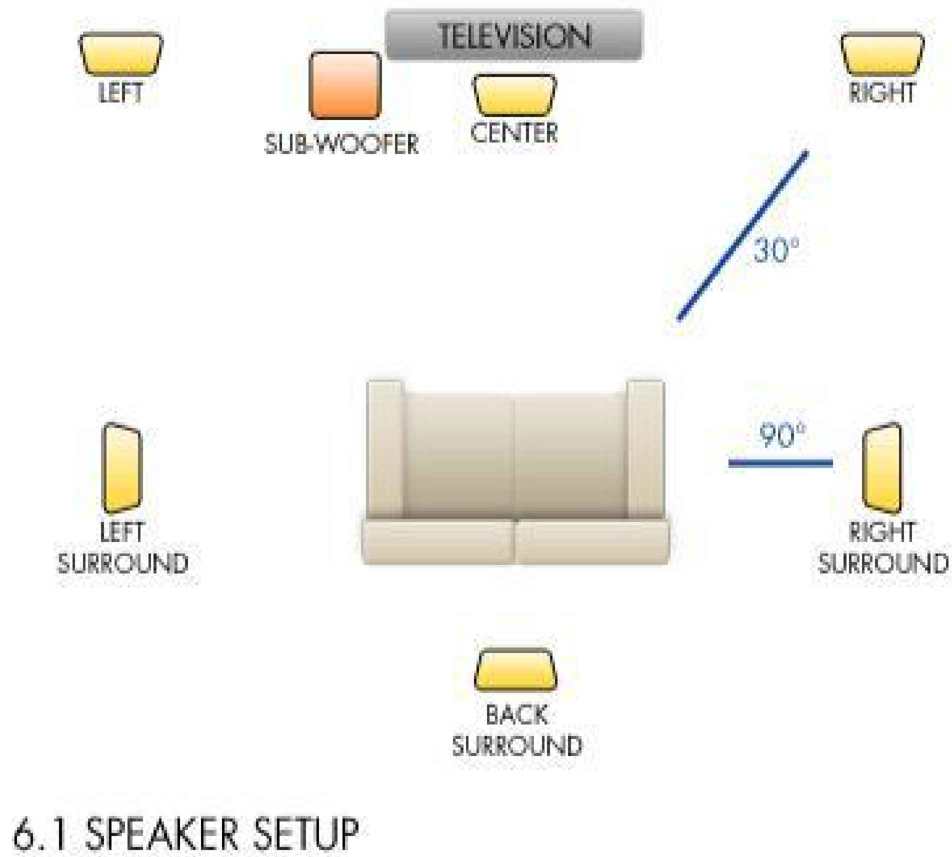
A 5.1 multichannel audio system locates the speakers as shown in Figure 1. The left (L) and right (R) channels drive the speaker pair in front of the listener and carry most of the music in the program. The center (C) channel primarily carries dialog, as producers usually want listeners to perceive this in the center of the video field.

The left surround (Ls) and right surround (Rs) channels drive the left and right speaker pair placed to the side or behind the listener. They typically handle the sound effects and ambient noises that create the aural illusion of a particular environment.

The Low Frequency Effect (LFE) channel delivers low-frequency non-localized special effects and creates the dramatic effects within the material (e.g., explosions). The LFE channel drives a high-power speaker (a subwoofer) that has a restricted frequency below 150Hz. The subwoofer is typically positioned in front of the listener. Although the speaker device is called a subwoofer, in a surround-sound system, it is often referred to as the LFE channel because it will have different responses depending on the size of the speaker system being used by the viewer.

Multiple sound tracks has been mixed as 6 Single tracks1.Left 2.Right 3.Centre 4.Left surround 5.Right surround .1Sub woofer (low frequency)

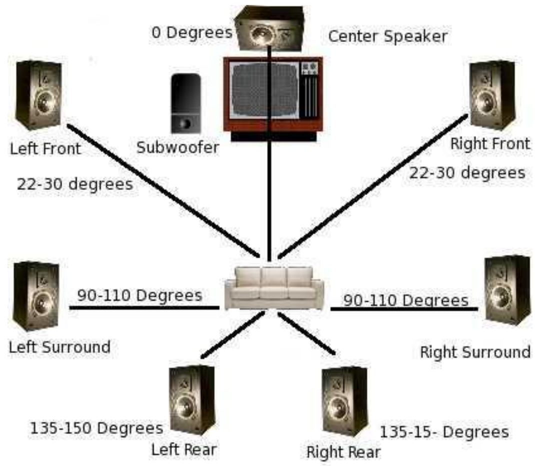
6.1 Dolby digital surround Mixing



- **Multiple sound tracks mix as 7 Single tracks**

- 1.Left
- 2.Right
- 3.Centre
- 4.Left surround
- 5.Right surround
- 6.Back surround
- .1Sub woofer (low freq)

7.1 SDDS Surround Mixing

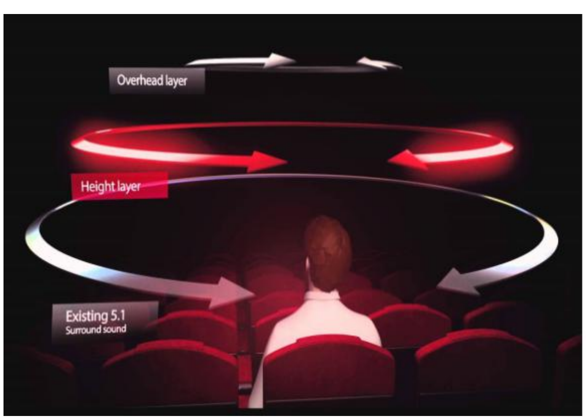


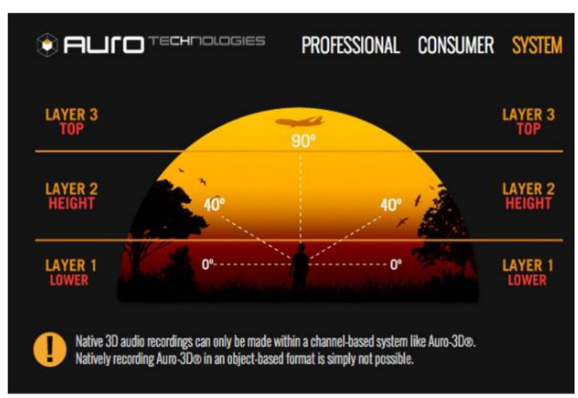
7.1 Channel Surround System

- 1.Left
- 2.Right
- 3.Centre
- 4.Left surround
- 5.Right surround
- 6.left rear surround
- 7.Right rear surround
- .1Sub woofer (low frequency)

Introduction to Auro-3D **Immersive sound system**

Auro-3D is the next generation three-dimensional audio standard. It provides a realistic sound experience unlike anything before. By fully immersing the listener in a cocoon of life-like sound, Auro-3D creates the sensation of actually ‘being there’. Thanks to a unique ‘Height’ channel configuration, acoustic reflections are generated and heard naturally due to the fact that sounds originate from around as well as above the listener.



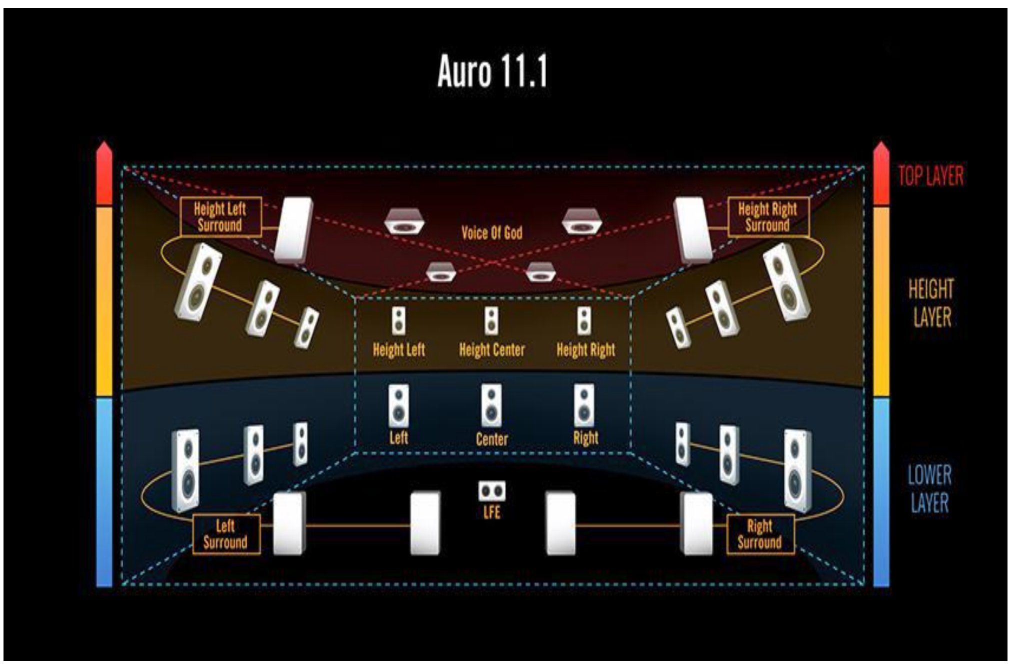


To achieve ‘true sound in 3D’, Auro-3D adds the crucial third and final dimension in the evolution of sound reproduction. While 5.1/7.1 Surround configurations fail to include height channels (z- axis), Auro-3D’s creates its outstanding effect with a HEIGHT-based sound hemisphere capable of thoroughly immersing the listener. Depending on the size of the room, either 1 or 2 additional layers (HEIGHT and TOP) are mounted above the existing Surround layer at ear-level to produce Auro-3D’s defining ‘vertical stereo field’ (see image). This field is the key to creating the most natural and immersive sound experience possible. The optional (third) TOP Layer placed overhead is a supplementary effect channel that is not critical for natural audio reproduction. As people are less sensitive to sounds originating from above, the TOP Layer is primarily used for ‘fly-overs’ and other special effects – most sound sources and their chief initial reflections are located between the Surround Layer and Auro-3D’s unique Height Layer.

11.1: Four height channels + Height Center + Top Channel

The Auro-3D 13.1 format is based on, and compatible with the 6.1 Standard. It includes 6.1’s standard channels plus the following additional channels.

Channel-based vs. Object-based sound mixing:



Auro-3D can reproduce immersive 3D audio for both films and music because Auro-3D is as well a native true 3D recording format. Auro Technologies is working closely with the industry to release music titles in Auro-3D. For this purpose it has developed a range of professional audio tools which are made available for Auro-3D production. Several labels will announce the release of Auro-3D content very soon, including classical, jazz, pop rock, as well as popular music genre.