

UNIT-IV

Basics of Audio production techniques - mono - stereo - multi-channel - characteristics - types - directional features - different recording media - recording equipment accessories - mixing consoles - talk-back units - monitoring sound - live mixing - AM - FM - satellite radio-Dubbing - re-recording.

BASICS OF AUDIO PRODUCTION TECHNIQUES

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 20kHz. It's not a bad idea to memorise those numbers — 20Hz is the lowest-pitched (bassiest) sound we can hear, 20kHz 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:

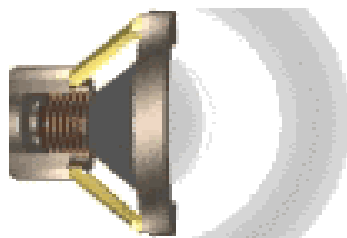
1. **Sound Waves:** What they are, how they are produced and how we hear them.
2. **Sound Equipment:** What the different components are, what they do, how to choose the correct equipment and use it properly.

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.

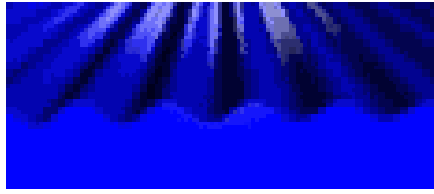
How Sound Waves Work

Sound waves exist as variations of pressure in a medium such as air. They are created by the vibration of an object, which causes the air surrounding it to vibrate. The vibrating air then causes the human eardrum to vibrate, which the brain interprets as sound.

The illustration on the left shows a speaker creating sound waves

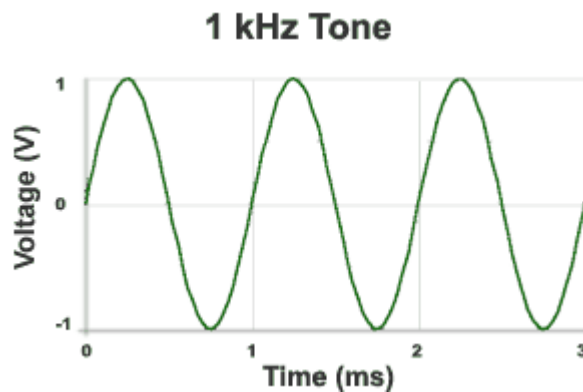


Sound waves travel through air in much the same way as water waves travel through water. In fact, since water waves are easy to see and understand, they are often used as an analogy to illustrate how sound waves behave.



Sound waves can also be shown in a standard x vs y graph, as shown here. This allows us to visualise and work with waves from a mathematical point of view. The resulting curves are known as the "waveform" (i.e. the form of the wave.)

The wave shown here represents a constant tone at a set frequency. You will have heard this noise being used as a test or identification signal. This "test tone" creates a nice smooth wave which is ideal for technical purposes. Other sounds create far more erratic waves.



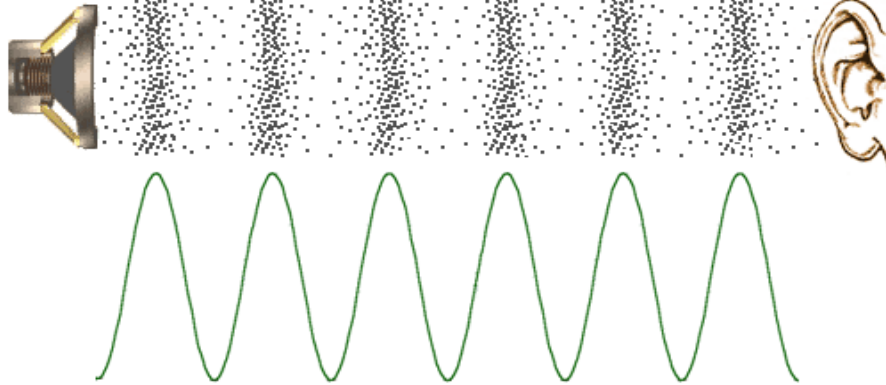
Note that a waveform graph is two-dimensional but in the real world sound waves are three-dimensional. The graph indicates a wave traveling along a path from left to right, but real sound waves travel in an expanding sphere from the source. However the 2-dimensional model works fairly well when thinking about how sound travels from one place to another.

The next thing to consider is what the graph represents; that is, what it means when the wave hits a high or low point. The following explanation is a simplified way of looking at how sound waves work and how they are represented as a waveform. Don't take it too literally — treat it as a useful way to visualise what's going on.

In an electronic signal, high values represent high positive voltage. When this signal is converted to a sound wave, you can think of high values as representing areas of increased air pressure. When the waveform hits a high point, this corresponds to molecules of air being packed together densely. When the wave hits a low point the air molecules are spread more thinly.

In the diagram below, the black dots represent air molecules. As the loudspeaker vibrates, it causes the surrounding molecules to vibrate in a particular pattern represented by the waveform. The vibrating air then causes the listener's eardrum to vibrate in the same pattern. Viola — Sound!

Variations in Air Pressure and Corresponding Waveform



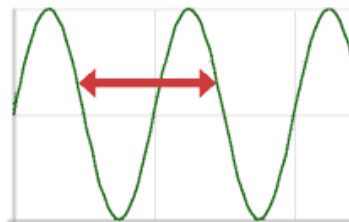
Note that air molecules do not actually travel from the loudspeaker to the ear (that would be wind). Each individual molecule only moves a small distance as it vibrates, but it causes the adjacent molecules to vibrate in a rippling effect all the way to the ear.

Sound Wave Properties

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

Wavelength: The distance between any point on a wave and the equivalent point on the next phase. Literally, the length of the wave.

Wavelength



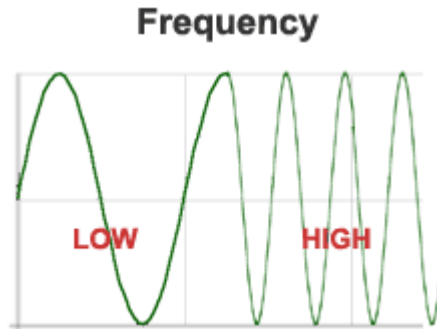
Amplitude: The strength or power of a wave signal. The "height" of a wave when viewed as a graph. Higher amplitudes are interpreted as a higher volume, hence the name "amplifier" for a device which increases amplitude.

Amplitude



Frequency: The number of times the wavelength occurs in one second. Measured in kilohertz (Khz), or cycles per second. The faster the sound source vibrates, the higher the frequency.

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.



How Sound Waves Interact with Each Other

When different waves collide (e.g. sound from different sources) they interfere with each other. This is called, unsurprisingly, *wave interference*.

The following table illustrates how sound waves (or any other waves) interfere with each other depending on their phase relationship:

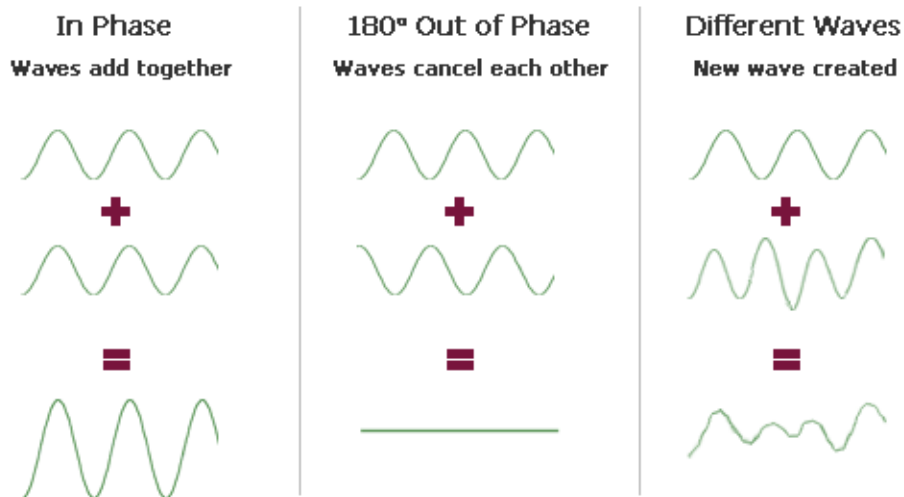


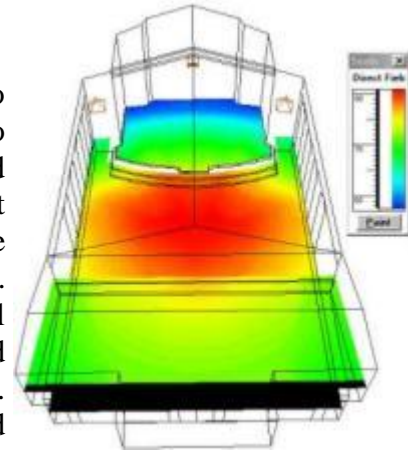
Image courtesy of www.mediacollege.com

- Sound waves which are exactly in phase add together to produce a stronger wave.
- Sound waves which are exactly inverted, or 180 degrees out of phase, cancel each other out and produce silence. This is how many noise-cancellation devices work.

What is the difference between mono and stereo?

Mono

Mono or monophonic describes a system where all the audio signals are mixed together and routed through a single audio channel. Mono systems can have multiple loudspeakers, and even multiple widely separated loudspeakers. The key is that the signal contains no level and arrival time/phase information that would replicate or simulate directional cues. Common types of mono systems include single channel centre clusters, mono split cluster systems, and distributed loudspeaker systems with and without architectural delays. Mono systems can still be full-bandwidth and full-fidelity and are able to reinforce both voice and music effectively. The big advantage to mono is that everyone hears the very same signal, and, in properly designed systems, all listeners would hear the system at essentially the same sound level. This makes well-designed mono systems very well suited for speech reinforcement as they can provide excellent speech intelligibility.



Stereo

True stereophonic sound systems have two independent audio signal channels, and the signals that are reproduced have a specific level and phase relationship to each other so that when played back through a suitable reproduction system, there will be an apparent image of the original sound source. Stereo would be a requirement if there is a need to replicate the aural perspective and localization of instruments on a stage or platform, a very common requirement in performing arts centres.

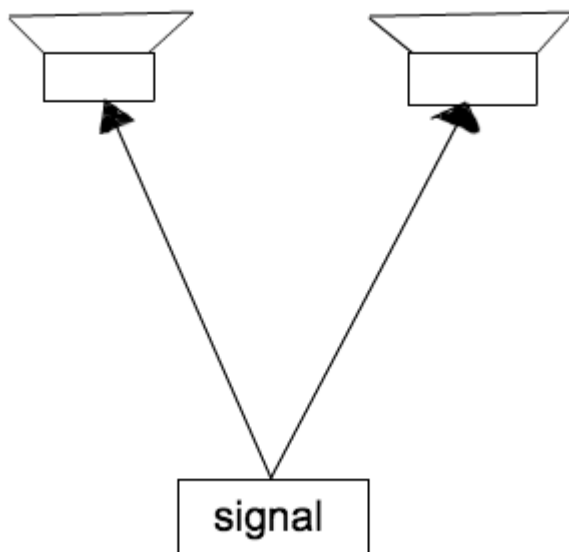
This also means that a mono signal that is panned somewhere between the two channels does *not* have the requisite phase information to be a true stereophonic signal, although there can be a level difference between the two channels that simulates a position difference, this is a simulation only. That's a discussion that could warrant a couple of web pages all by itself.

An additional requirement of the stereo playback system is that the *entire* listening area must have equal coverage of both the left and right channels, at essentially equal levels. This is why your home stereo system has a "sweet spot" between the two loudspeakers, where the level differences and arrival time differences are small enough that the stereo image and localization are both maintained. This sweet spot is limited to a fairly small area between the two loudspeakers and when a listener is outside that area, the image collapses and only one or the other channel is heard. Living with this sweet spot in your living room may be OK, since you can put your couch there, but in a larger venue, like a church sanctuary or theatre auditorium, that sweet spot might only include 1/3 the audience, leaving 2/3 of the audience wondering why they only hear half the program.

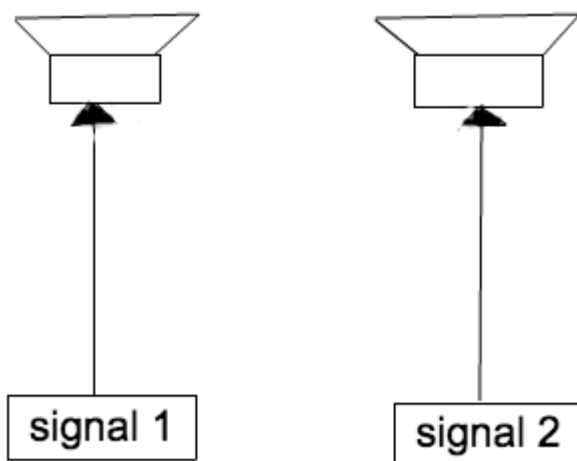
In addition a stereo playback system must have the correct *absolute phase response* input to output for both channels. This means that a signal with a positive pressure waveform at the input to the system must have the same positive pressure waveform at the output of the system. So a drum, for instance, when struck produces a positive pressure waveform at the

microphone and should produce a positive pressure waveform in the listening room. If you don't believe that this makes a tremendous difference, try reversing the polarity of both your hifi loudspeakers some day and listening to a source that has a strong centre sound image like a solo voice. When the absolute polarity is flipped the wrong way, you won't find a stable centre channel image, it will wander around away from the centre, localizing out at both the loudspeakers.

Mono

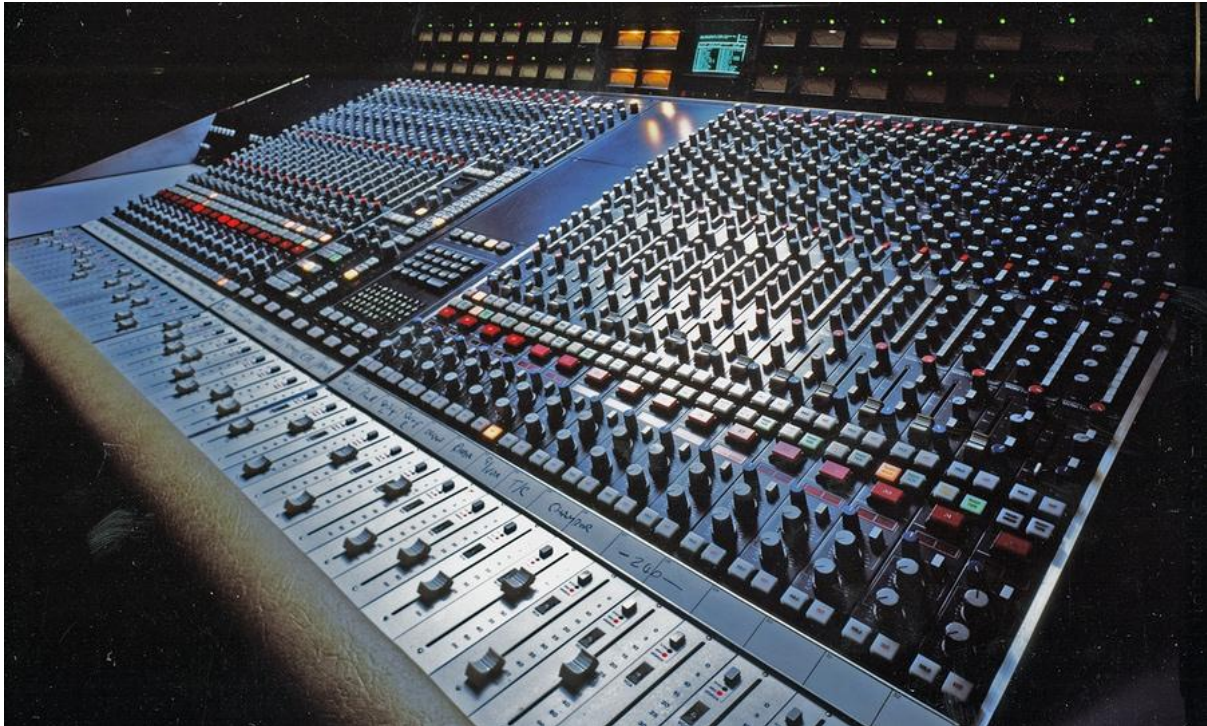


Stereo



Mixing console

In sound recording and reproduction, and sound reinforcement systems, a mixing console is an electronic device for combining sounds of many different audio signals. Inputs to the console include microphones being used by singers and for picking up acoustic instruments, signals from electric or electronic instruments, or recorded music. Depending on the type, a mixer is able to control analog or digital signals. The modified signals are summed to produce the combined output signals, which can then be broadcast, amplified through a sound reinforcement system or recorded.



Talkback (television production)

Talkback, or in-ear talkback, is a device used by directors and producers to talk directly to the anchor or the host of the show. This device enables the show directors to send out commands, instructions, content information and even the complete script to the anchors or hosts of the show. Talkback consists of an earpiece made of silicon or foam bud that sits just inside the left or right ear, attached to a curly or straight acoustic tube that goes around and behind the ear, then down the back of the neck to a wireless receiver. There is an optional collar clip to hold the cable in place, and hair/clothing can be used to hide the kit as much as possible. The volume of the speech coming through the earpiece can be adjusted. The wireless receiver gets signals from a transmitter. An audio mixer placed in production control rooms (PCR) would be wired to the transmitter. The input for the audio mixer would come from a microphone in which the director speaks and sends out information. This information is carried to the talkback and the host follows the instruction.

AM Radio:

□ Amplitude Modulation (AM)

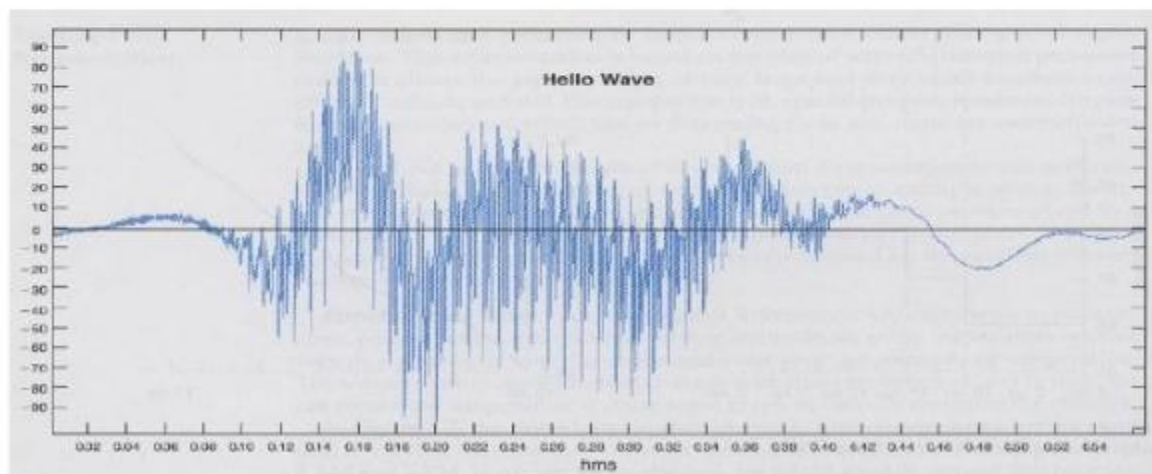
- **Amplitude modulation is the process of varying the amplitude of a carrier wave in proportion to the amplitude of a baseband signal. The frequency of the carrier remains constant.**
- *Modulation is the process of impressing a low-frequency information signal (baseband signal) onto a higher frequency carrier signal.*



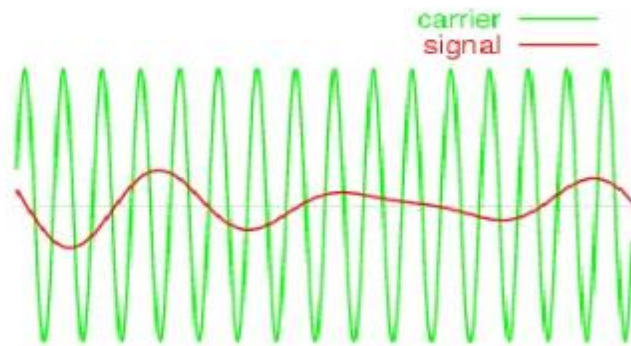
The term **baseband** is used to designate the band of frequencies representing the original signal as delivered by the input transducer

For example, the voice signal from a microphone is a baseband signal, and contains frequencies in the range of 0-3000 Hz

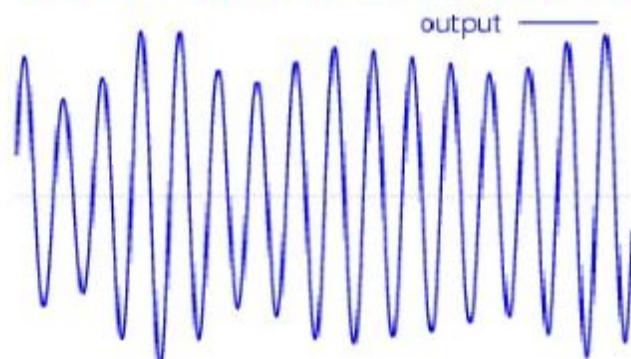
The “hello” wave is a baseband signal:



Amplitude Modulation (AM) uses changes in the signal strength to convey information



pressure modulation (sound)



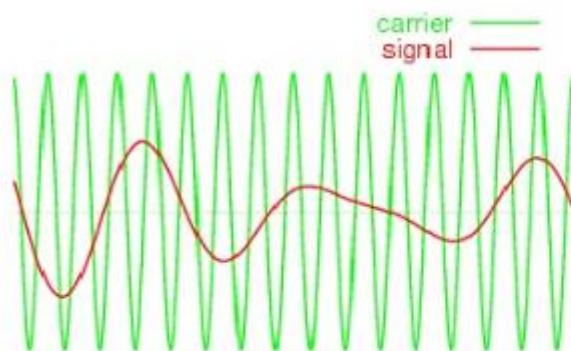
electromagnetic wave modulation

FM Radio

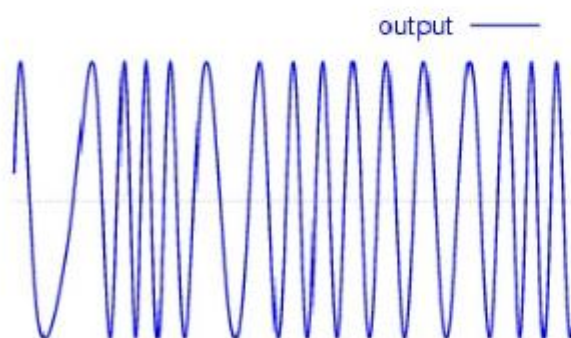
- *Frequency Modulation (FM)*
 - *Frequency modulation is the process of varying the frequency of a carrier wave in proportion to the amplitude of a baseband signal. The amplitude of the carrier remains constant*



Frequency Modulation (FM) uses changes in the wave's frequency to convey information



Pressure modulation (sound)



electromagnetic wave modulation

AM vs. FM

AM requires a simple circuit, and is very easy to generate. It is simple to tune, and is used in almost all short wave broadcasting.

The area of coverage of AM is greater than FM (longer wavelengths (lower frequencies) are utilized-remember property of HF waves?)

However, it is quite inefficient, and is susceptible to static and other forms of electrical noise.

The main advantage of FM is its audio quality and immunity to noise. Most forms of static and electrical noise are naturally AM, and an FM receiver will not respond to AM signals.

The audio quality of a FM signal increases as the frequency deviation increases (deviation from the center frequency), which is why FM broadcast stations use such large deviation.

The main disadvantage of FM is the larger bandwidth it requires

Satellite radio

Satellite radio is defined by the International Telecommunication Union (ITU)'s ITU Radio Regulations (RR) as a broadcasting-satellite service. The satellite's signals are broadcast nationwide, across a much wider geographical area than terrestrial radio stations, and the service is primarily intended for the occupants of motor vehicles. It is available by subscription, mostly commercial free, and offers subscribers more stations and a wider variety of programming options than terrestrial radio.