



Audio Editing

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Audio Editing

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Unit one

Introduction of Sound and Listening

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INTRODUCTION

Dear students to understand the audio recording and its editing first you need to have the basic know how of sound and listening. The following discussion will give you the brief idea about sound and its fundamentals. when we are discussing about the fundamentals we are basically explaining the basic qualities of sounds that how different sounds can be produced and it arrives in to the ears and how the sound waves travel in the form of periodic variations and their depiction in the form of graphical representation.

When we make a recording, in effect we're actually capturing and storing sound into a memory media so that an original event can be re-created at a later date. But to follow this regeneration of event we need to get help from the graphical wave forms and their characteristics will allow you to get help and distinguished between one another. The two basic fundamentals of a graphically presented wave form is its frequency and the amplitude. The content in this unit will explain you the basic range of human hearing in the form of frequency and various range to measure the amplitude of basic wave form cycle and its repetition within the positive and negative amplitudes.

This unit will enable you to get a better idea of how the frequency range has a direct relationship over the period of time for determining the difference between low and high frequency ranges. The unit also explains you to get a better idea about the behavior of sound while striking at different surfaces and the relationship of mixing the wave form at different sound angles, the addition of sound at different phases and their results.

After having clear background knowledge of a sinusoidal wave form and its characteristics this unit will lead you to more advance level or the actual form of sounds produced by different musical instruments along with their original sound frequencies. In addition to their even and odd multiples called the harmonics which enables you to have a better idea of differentiating between different musical instrumental voicing and the simple and complex waveform mechanism. The unit will also explain the characteristic variation in different musical levels.

After having a detailed discussion on sound and its characteristics the chapter will lead you to the next part that is hearing. If we start with the idea that sound is actually a concept that describes the brain's perception and interpretation of a physical auditory stimulus, the examination of sound can be explained by the characteristics of the ear and how the ear is stimulated by sound.

By understanding the physical nature of sound and the basics of how the ears change a physical phenomenon into a sensory one, we can discover how to best convey this science into the subjective art forms of music, sound recording and production.

OBJECTIVES:

After studying this unit you will be able to:

1. Know about the fundamental of sound
2. Identify the characteristics of a sinusoidal waveform.
3. You can differentiate between a simple and a complex waveform.
4. Know about the musical waveform and its qualities
5. Know about calibration of sound
6. Basics of logarithm.
7. Function of human ear as a transducer

1 THE BASICS OF SOUND

Sound is a vibration, or wave, that travels through matter (solid, liquid, or gas) and can be heard. It's a vibration that propagates as a typically audible mechanical wave of pressure and displacement, through a medium such as air or water. In physiology and psychology, sound is the reception of such waves and their perception by the brain.

All sounds are vibrations traveling through the air as sound waves. Sound waves are caused by the vibrations of objects and radiate outward from their source in all directions. A vibrating object compresses the surrounding air molecules (squeezing them closer together) and then rarefies them (pulling them farther apart). Although the fluctuations in air pressure travel outward from the object, the air molecules themselves stay in the same average position. As sound travels, it reflects off objects in its path, creating further disturbances in the surrounding air. When these changes in air pressure vibrate your eardrum, nerve signals are sent to your brain and are interpreted as sound.

Sound arrives at the ear in the form of periodic variations in atmospheric pressure called sound-pressure waves. This is the same atmospheric pressure that's measured by the weather service with a barometer; however, the changes in pressure heard by the ear are too small in magnitude and fluctuate too rapidly to be observed on a barometer. An analogy of how sound waves travel in air can be demonstrated by bursting a balloon in a silent room. Before we stick it with a pin, the molecular motion of the room's atmosphere is at a normal resting pressure. The pressure inside the balloon is much higher, though, and the molecules are compressed much more tightly together—like people packed into a crowded subway car (Figure 1.1a). When the balloon is popped... "POW!" (Figure 1.1b), the tightly compressed molecules under high pressure begin to exert an outward force on their neighbors in an effort to move toward areas of lower pressure. When the neighboring set of molecules has been compressed, they will continue to exert an outward force on the next set of lower-pressured neighbors (Figure 1.1c) in an ongoing outward motion that continues until the molecules have used up their energy in the form of heat.

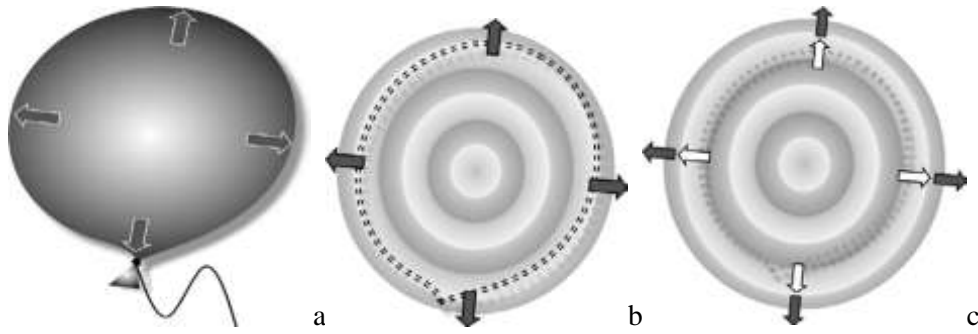


Figure 1.1 Wave movement in air as it moves away from its point of origin.

(a) An intact balloon contains pressurized air. (b) When the balloon is popped, the compressed molecules exert a force on outer neighbors in an effort to move to areas of lower pressure. (c) The outer neighbors then exert a force on the next set of molecules in an effort to move to areas of lower pressure ... and the process continues.

Likewise, as a vibrating mass (such as a guitar string, a person’s vocal chords or a loudspeaker) moves outward from its normal resting state, it squeezes air molecules into a compressed area, away from the sound source. This causes the area being acted on to have a greater than normal atmospheric pressure, a process called compression (Figure 1.2a). As the vibrating mass moves inward from its normal resting state, an area with a lower-than-normal atmospheric pressure will be created, in a process called rarefaction (Figure 1.2b). As the vibrating body cycles through its inward and outward motions, areas of higher and lower compression states are generated. These areas of high pressure will cause the wave to move outward from the sound source in the same way waves moved outward from the burst balloon. It’s interesting (and important) to note that the molecules themselves don’t move through air at the velocity of sound—only the sound wave itself moves through the atmosphere in the form of high-pressure compression waves that continue to push against areas of lower pressure (in an outward direction). This outward pressure motion is known as wave propagation.

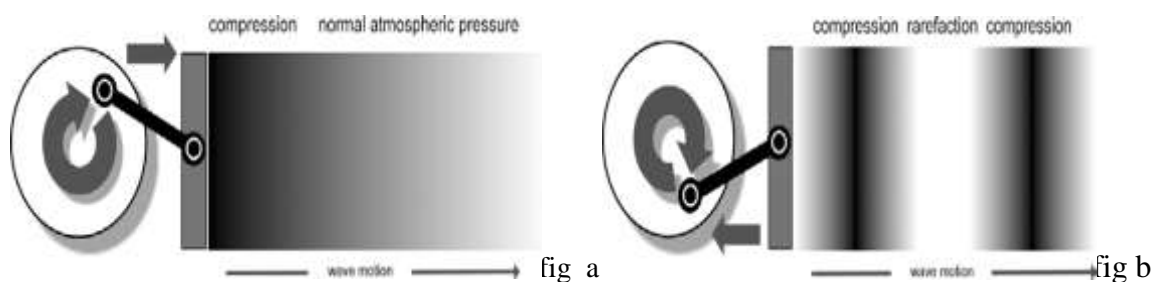
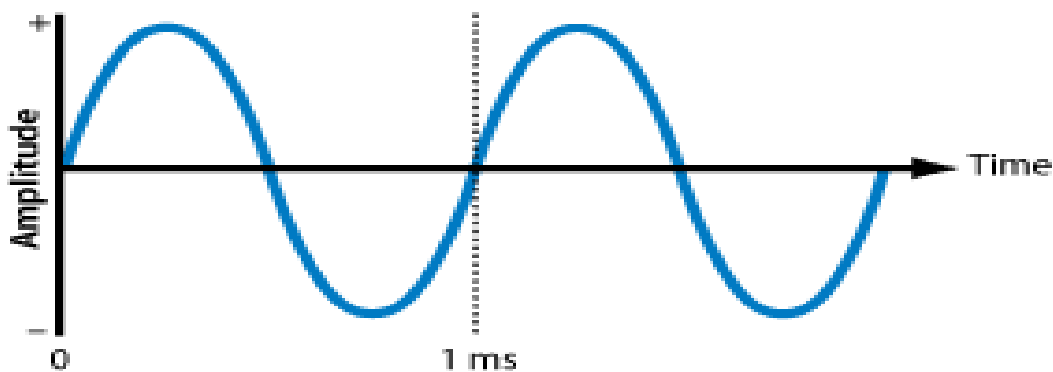


Figure 1.2: Effects of a vibrating mass on air molecules and their propagation.

(a) *Compression*—air molecules are forced together to form a compression wave (b) *Rarefaction*—as the vibrating mass moves inward, an area of lower atmospheric pressure is created.

2 WAVEFORM CHARACTERISTICS

A waveform is essentially the graphic representation of a sound-pressure level or voltage level as it moves through a medium over time. The simplest kind of sound wave is a sine wave. Pure sine waves rarely exist in the natural world, but they are a useful place to start because all other sounds can be broken down into combinations of sine waves. A sine wave clearly demonstrates the three fundamental characteristics of a sound wave: frequency, amplitude, and phase.



In short, a waveform lets us see and explain the actual phenomenon of wave propagation in our physical environment and will generally have the following fundamental characteristics:

- **Amplitude**
- **Frequency**
- **Velocity**
- **Wavelength**
- **Phase**
- **Harmonic content**
- **Envelope.**

These characteristics allow one waveform to be distinguished from another. The most fundamental of these are amplitude and frequency (Figure 1.3). The following sections describe each of these characteristics. Although several math formulas have been included, it is by no means important that you memorize or worry about them. It's far more important that you grasp the basic principles of acoustics rather than fret over the underlying math.

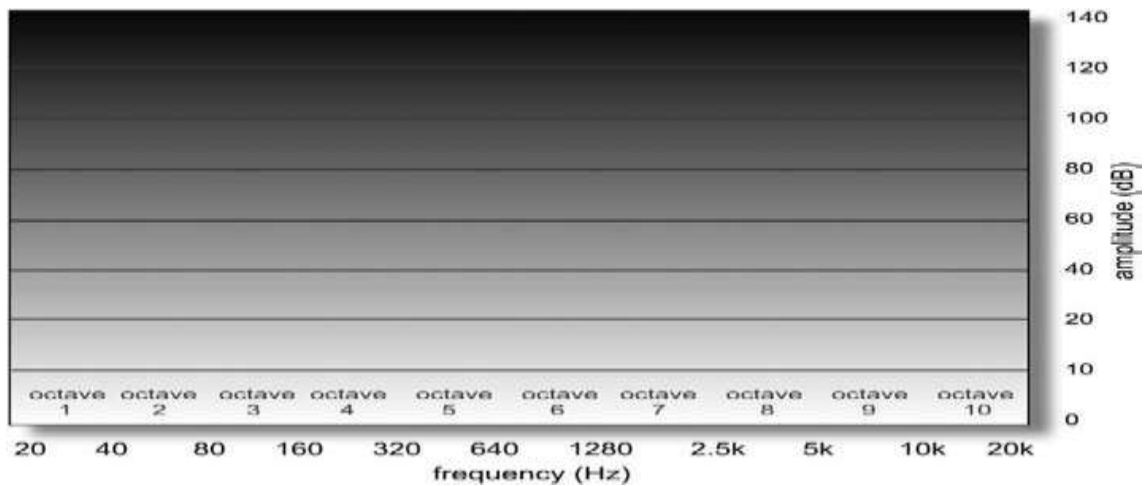


Figure 1.3: Amplitude and frequency ranges of human hearing.

2.1 Amplitude

Amplitude (or intensity) refers to the strength of a sound wave, which the human ear interprets as volume or loudness. People can detect a very wide range of volumes, from the sound of a pin dropping in a quiet room to a loud rock concert. Because the range of human hearing is so large, audio meters use a logarithmic scale (decibels) to make the units of measurement more manageable.

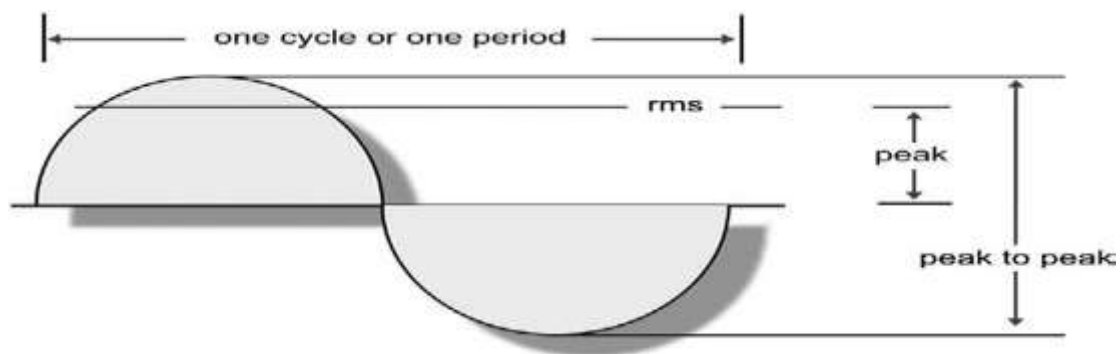


Figure 1.4: Graph of a sine wave showing the various ways to measure amplitude.

The distance above or below the centerline of a waveform (such as a pure sine wave) represents the amplitude level of that signal. The greater the distance or displacement from that centerline, the more intense the pressure variation, electrical signal level, or physical displacement will be within a medium. Wave-form amplitudes can be measured in several ways (Figure 1.4). For example, the measurement of either the maximum positive or negative signal level of a wave is called its peak amplitude value (or peak value). The total measurement of the positive and

negative peak signal levels is called the peak-to-peak value. The root-mean-square (rms) value was developed to determine a meaningful average level of a waveform over time (one that more closely approximates the level that's perceived by our ears and gives a better real-world measurement of overall signal amplitudes).

The rms value of a sine wave can be calculated by squaring the amplitudes at points along the waveform and then taking the mathematical average of the combined results. The math isn't as important as the concept that the rms value of a perfect sine wave is equal to 0.707 times its instantaneous peak amplitude level. Because the square of a positive or negative value is always positive, the rms value will always be positive. The following simple equations show the relationship between a waveform's peak and rms values:

$$\text{rms voltage} = \text{peak voltage} \times 0.707 \qquad \text{peak voltage} = \text{rms voltage} \times 1.414$$

2.2 FREQUENCY

Frequency is the rate, or number of times per second, that a sound wave cycles from positive to negative to positive again. Frequency is measured in cycles per second, or hertz (Hz). Humans have a range of hearing from 20 Hz (low) to 20,000 Hz (high). Frequencies beyond this range exist, but they are inaudible to humans.

The rate at which an acoustic generator, electrical signal or vibrating mass repeats within a cycle of positive and negative amplitude is known as the frequency of that signal. As the rate of repeated vibration increases within a given time period, the frequency (and thus the perceived pitch) will likewise increase ... and vice versa. One completed excursion of a wave (which is plotted over the 360° axis of a circle) is known as a cycle (Figure 1.5). The number of cycles that occur within a second (frequency) is measured in hertz (Hz). The diagram in Figure 1.6 shows the value of a waveform as starting at zero (0°). At time $t=0$, this value increases to a positive maximum value and then decreases back through zero, where the process begins all over again in a repetitive fashion. A cycle can begin at any angular degree point on the waveform; however, to be complete, it must pass through a single 360° rotation and end at the same point as its starting value. For example, the waveform that starts at $t=0$ and ends at $t=2$ constitutes a cycle, as does the waveform that begins at $t=1$ and ends at $t=3$.

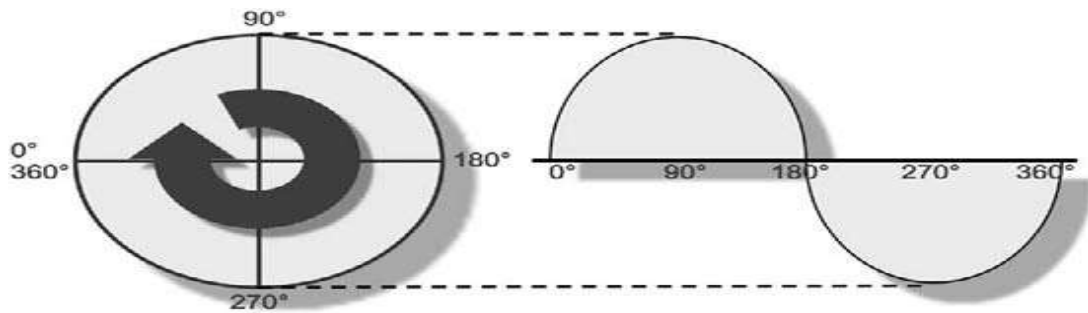


Figure 1.5: Cycle divided into the 360° of a circle.

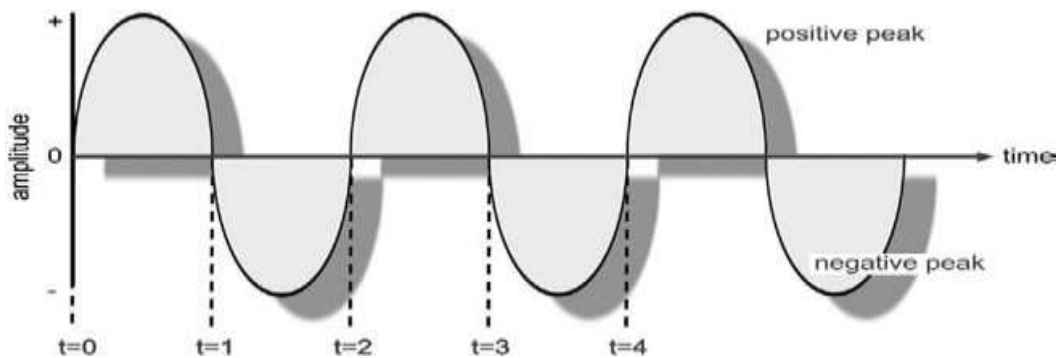
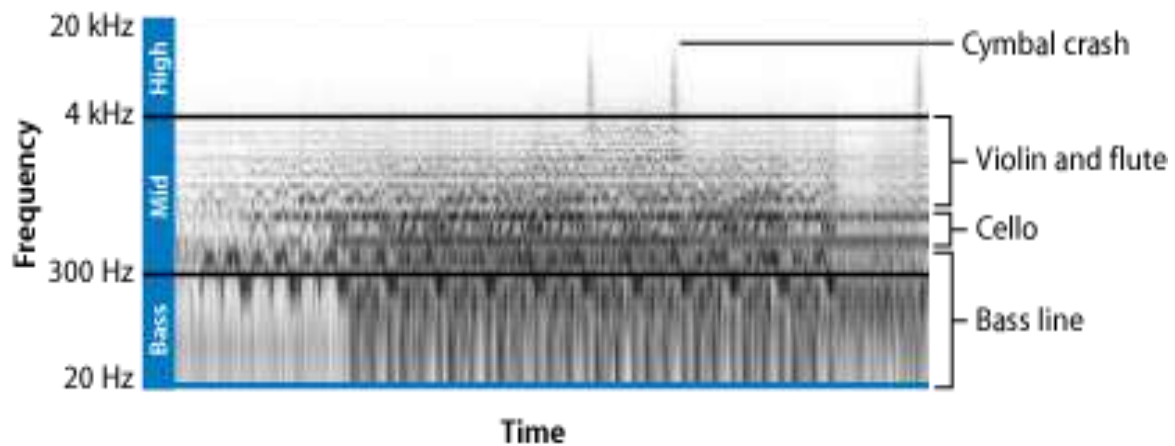


Figure 1.6: Graph of waveform amplitude over time.

Frequency Spectrum of Sounds

With the exception of pure sine waves, sounds are made up of many different frequency components vibrating at the same time. The particular characteristics of a sound are the result of the unique combination of frequencies it contains.

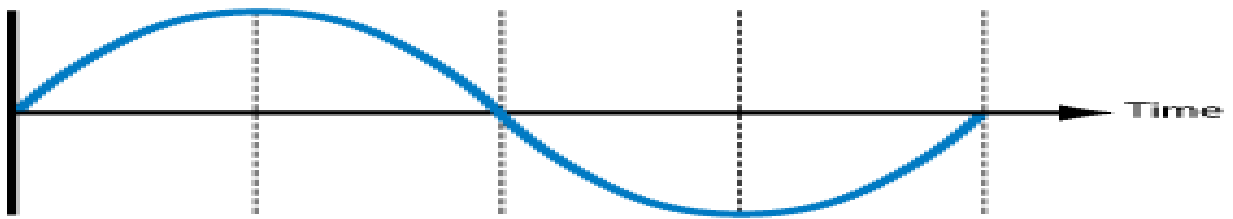
Sounds contain energy in different frequency ranges, or bands. If a sound has a lot of low-frequency energy, it has a lot of bass. The 250–4000 Hz frequency band, where humans hear best, is described as midrange. High-frequency energy beyond the midrange is called treble, and this adds crispness or brilliance to a sound. The graph below shows how the sounds of different musical instruments fall within particular frequency bands.



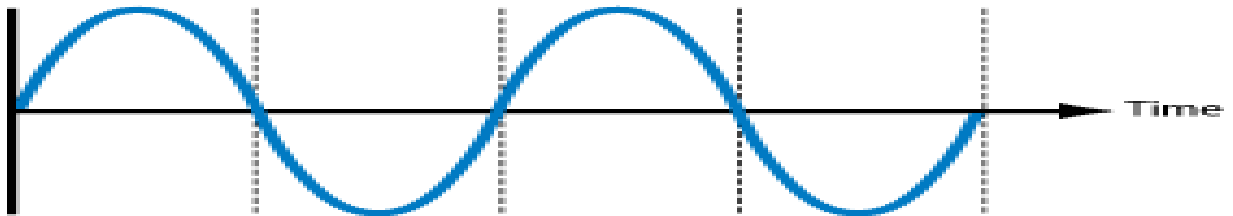
Note: Different manufacturers and mixing engineers define the ranges of these frequency bands differently, so the numbers described above are approximate.

The human voice produces sounds that are mostly in the 250–4000 Hz range, which likely explains why people’s ears are also the most sensitive to this range. If the dialogue in your movie is harder to hear when you add music and sound effects, try reducing the midrange frequencies of the non-dialogue tracks using an equalizer filter. Reducing the midrange creates a “sonic space” in which the dialogue can be heard more easily.

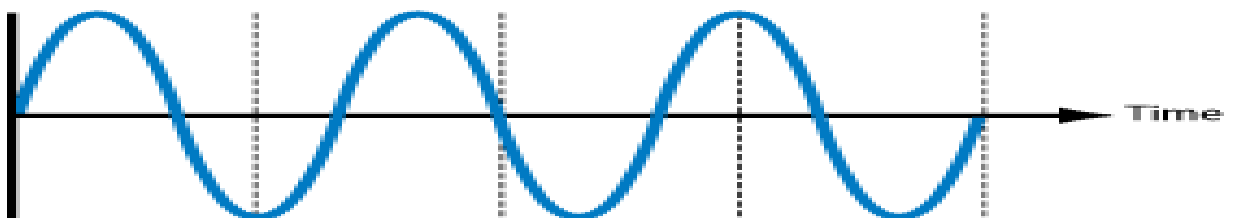
Musical sounds typically have a regular frequency, which the human ear hears as the sound’s pitch. Pitch is expressed using musical notes, such as C, E flat, and F sharp. The pitch is usually only the lowest, strongest part of the sound wave, called the fundamental frequency. Every musical sound also has higher, softer parts called overtones or harmonics, which occur at regular multiples of the fundamental frequency. The human ear doesn’t hear the harmonics as distinct pitches, but rather as the tone color (also called the timbre) of the sound, which allows the ear to distinguish one instrument or voice from another, even when both are playing the same pitch.



Fundamental

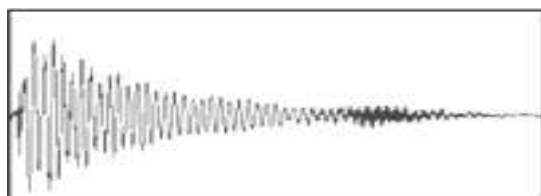


First harmonic

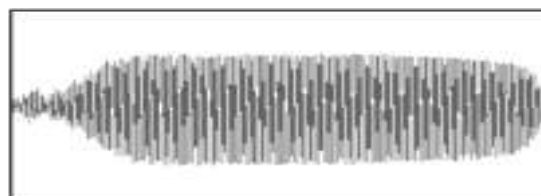


Second harmonic

Musical sounds also typically have a volume envelope. Every note played on a musical instrument has a distinct curve of rising and falling volume over time. Sounds produced by some instruments, particularly drums and other percussion instruments, start at a high volume level but quickly decrease to a much lower level and die away to silence. Sounds produced by other instruments, for example, a violin or a trumpet, can be sustained at the same volume level and can be raised or lowered in volume while being sustained. This volume curve is called the sound's envelope and acts like a signature to help the ear recognize what instrument is producing the sound.



Percussive volume envelope



Sustained volume envelope

2.3 VELOCITY

The velocity of an object is the rate of change of its position with respect to a frame of reference, and is a function of time. Velocity is equivalent to a specification of its speed and direction of motion (e.g. 60 km/h to the north). Velocity is an important concept in kinematics, the branch of classical mechanics that describes the motion of bodies.

Velocity is a physical vector quantity; both magnitude and direction are needed to define it. The scalar absolute value(magnitude) of velocity is called "speed", being a coherent derived unit whose quantity is measured in the SI (metric) system as meters per second (m/s) or as the SI base unit of (m.s⁻¹). For example, "5 meters per second" is a scalar (not a vector), whereas "5 meters per second east" is a vector.

If there is a change in speed, direction, or both, then the object has a changing velocity and is said to be undergoing an acceleration.

2.4 WAVE LENGTH

The wavelength of a waveform (frequently represented by the Greek letter lambda, λ) is the physical distance in a medium between the beginning and the end of a cycle. The physical length of a wave can be calculated using:

$$\lambda = Vf$$

Where λ is the wavelength in the medium V is the velocity in the medium f is the frequency (in hertz).

The time it takes to complete 1 cycle is called the period of the wave. To illustrate, a 30-Hz sound wave completes 30 cycles each second or 1 cycle every 1/30th of a second. The period of the wave is expressed using the symbol T:

$$T = 1/F$$

Where T is the number of seconds per cycle.

Assuming that sound propagates at the rate of 1130ft/sec, all that's needed is to divide this figure by the desired frequency. For example, the simple math for calculating the wavelength of a 30-Hz waveform would be $1130/30 = 37.6$ feet long, whereas a waveform having a frequency of 300Hz would be $1130/300 = 3.76$ feet long (Figure 1.7). Likewise, a 1000-Hz

waveform would work out as being $1130/1000 = 1.13$ feet long, and a 10,000-Hz waveform would be $1130/10,000 = 0.113$ feet long. From these calculations, you can see that when-ever the frequency is increased, the wavelength decreases.

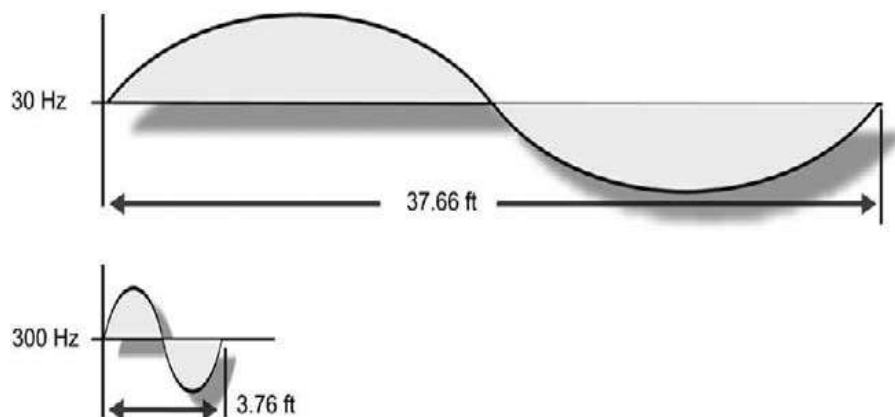


Figure 1.7: Wavelengths decrease in length as frequency increases (and vice versa).

2.5 REFLECTION OF SOUND

Much like a light wave, sound reflects off a surface boundary at an angle that is equal to (and in an opposite direction of) its initial angle of incidence. This basic property is one of the cornerstones of the complex study of acoustics. For example, Figure 1.8a shows how a sound wave reflects off a solid smooth surface in a simple and straightforward manner (at an equal and opposite angle). Figure 1.8b shows how a convex surface will play the sound outward from its surface, radiating the sound outward in a wide dispersion pattern. In Figure 1.8-c, a concave surface is used to focus a sound inward toward a single point, while a 90° corner (as shown in Figure 1.8d) reflects patterns back at angles that are equal to their original incident direction. This holds true both for the 90° corners of a wall and for intersections where the wall and floor meet. The secorner reflections help to provide insights into how volume levels often build up in the corners of a room (particularly at wall-to-floor corner intersections).corner reflections help to provide insights into how volume levels often build up in the corners of a room (particularly at wall-to-floor corner intersections).

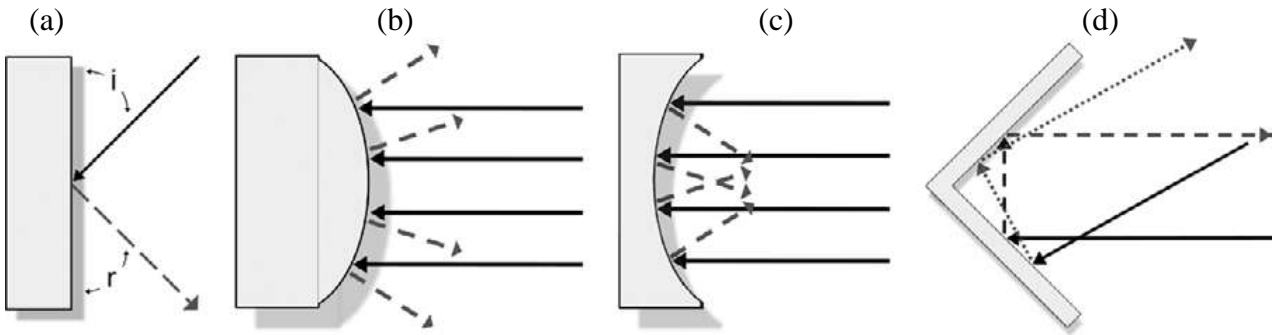


Figure 1.8: Incident sound waves striking surfaces with varying shapes: (a) single-planed, solid, smooth surface; (b) convex surface; (c) concave surface; (d) 90° corner reflection.

2.6 DIFFRACTION OF SOUND

Sound has the inherent ability to diffract around or through a physical acoustic barrier. In other words, sound can bend around an object in a manner that reconstructs the signal back to its original form in both frequency and amplitude. For example, in Figure 1.9a, we can see how a small obstacle will scarcely impede a larger acoustic waveform. Figure 1.9b shows how a larger obstacle can obstruct a larger portion of the waveform; however, past the obstruction, the signal bends around the area in the barrier's wake and begins to reconstruct itself. Figure 1.9c shows how the signal is able to radiate through an opening in a large barrier. Although the signal is greatly impeded (relative to the size of the opening), it nevertheless begins to reconstruct itself in wavelength and relative amplitude and begins to radiate outward as though it were a new point of origin. Finally, Figure 1.9d shows how a large opening in a barrier lets much of the waveform pass through relatively unimpeded.

2.7 FREQUENCY RESPONSE

The charted output of an audio device is known as its frequency response curve (when supplied with a reference input of equal level over the 20- to 20,000-Hz range of human hearing).

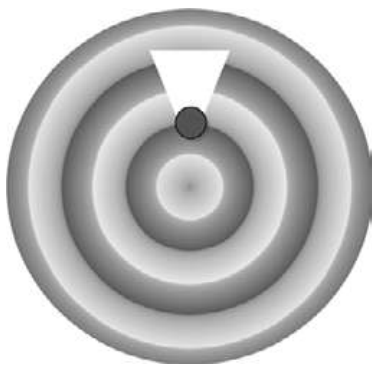


Fig a

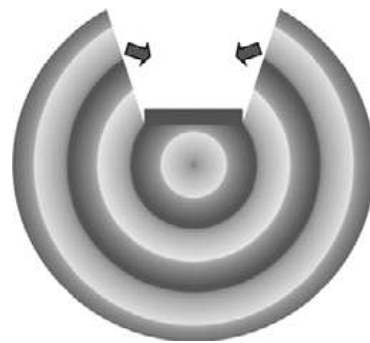


Fig b

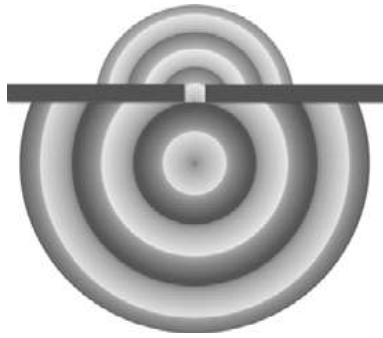


Fig c

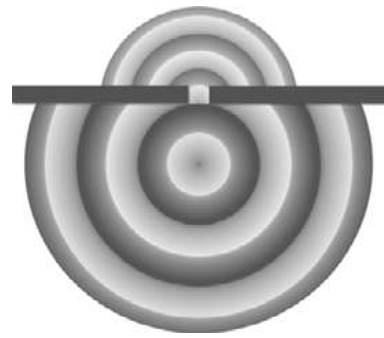


Fig d

Figure 1.9: The effects of obstacles on sound radiation and diffraction. (a) A small obstacle will scarcely impede a longer wavelength signal. (b) A larger obstacle will obstruct the signal to a greater extent; the waveform will also reconstruct itself in the barrier's wake. (c) A small opening in a barrier will greatly impede a signal; the waveform will emanate from the opening and reconstruct itself as a new source point. (d) A larger opening allows sound to pass unimpeded, allowing it to quickly diffract back into its original shape.

This curve is used to graphically represent how a device will respond to the audio spectrum and, thus, how it will affect a signal's overall sound. As an example, Figure 1.10 shows the frequency response of several unidentified devices. In these and all cases, the x-axis represents the signal's measured frequency, while the y-axis represents the device's measured output signal. These curves are created by feeding the input of an acoustic or electrical device with a constant-amplitude reference signal that sweeps over the entire frequency spectrum. The results are then charted on an amplitude versus frequency graph that can be easily read at a glance. If the measured signal is the same level at all frequencies, the curve will be drawn as a flat, straight line from left to right (known as a flat frequency response curve). This indicates that the device passes all frequencies equally (with no frequency being emphasized or de-emphasized). If the output lowers or increases at certain frequencies, these changes will easily show up as dips or peaks in the chart.

2.8 PHASE

Because we know that a cycle can begin at any point on a waveform, it follows that whenever two or more waveforms are involved in producing a sound, their relative amplitudes can (and often will) be different at any one point in time. For simplicity's sake, let's limit our example to two pure tone waveforms (sine waves) that have equal amplitudes and frequency but start their cyclic periods at different times. Such waveforms are said to be out of phase with respect to each other. Variations in phase, which are measured in degrees ($^{\circ}$), can be described as a time delay

between two or more waveforms. These delays are often said to have differences in relative phase degree angles (over the full rotation of a cycle, e.g., 90°, 180°, or any angle between 0° and 360°). The sine wave (so named because its amplitude follows a trigonometric sine function) is usually considered to begin at 0° with an amplitude of zero; the waveform then increases to a positive maximum at 90°, decreases back to a zero amplitude at 180°, increases to a negative maximum value at 270°, and finally returns back to its original level at 360°, simply to begin all over again.

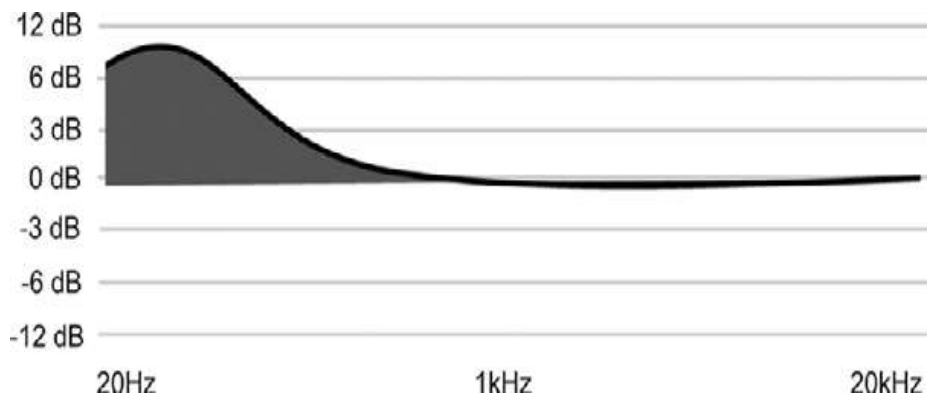


Fig a

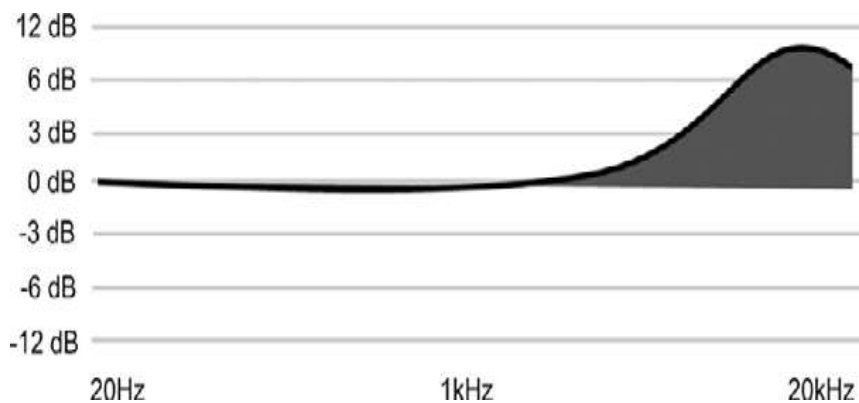


Fig b

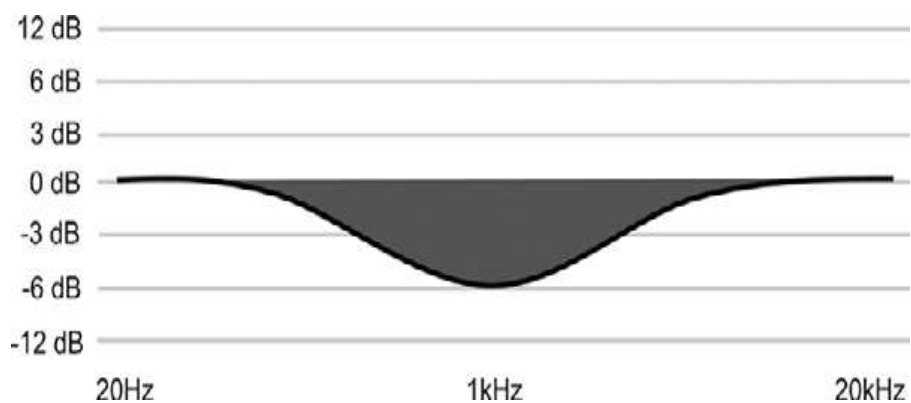


Fig c

Figure 1.10: Frequency response curves: (a) curve showing a bass boost; (b) curve showing a boost at the upper end; (c) curve showing a dip in the midrange.

Whenever two or more waveforms arrive at a single location out of phase, their relative signal levels will be added together to create a combined amplitude level at that one point in time. Whenever two waveforms having the same frequency, shape and peak amplitude are completely in phase (meaning that they have no relative time difference), the newly combined waveform will have the same frequency, phase and shape ... but will be double in amplitude (Figure 1.11a). If the same two waves are combined completely out of phase (having a phase difference of 180°), they will cancel each other out when added, which results in a straight line of zero amplitude (Figure 1.11b). If the second wave is only partially out of phase (by a degree other than 180°), the levels will be added at points where the combined amplitudes are positive and reduced in level where the combined result is negative (Figure 1.11c).

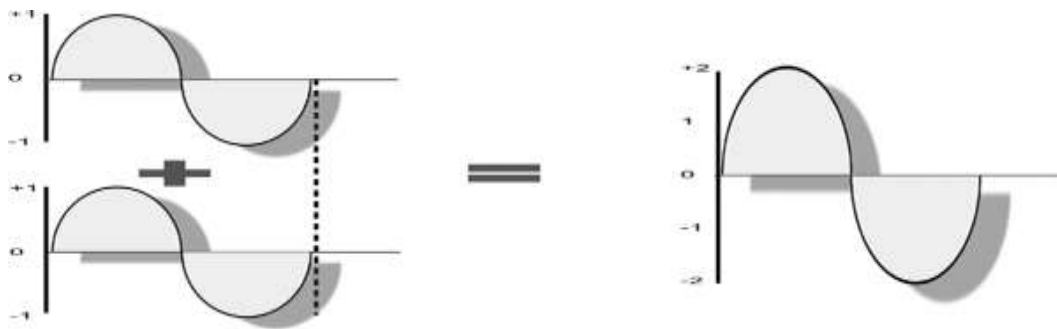


Fig 1.11 a

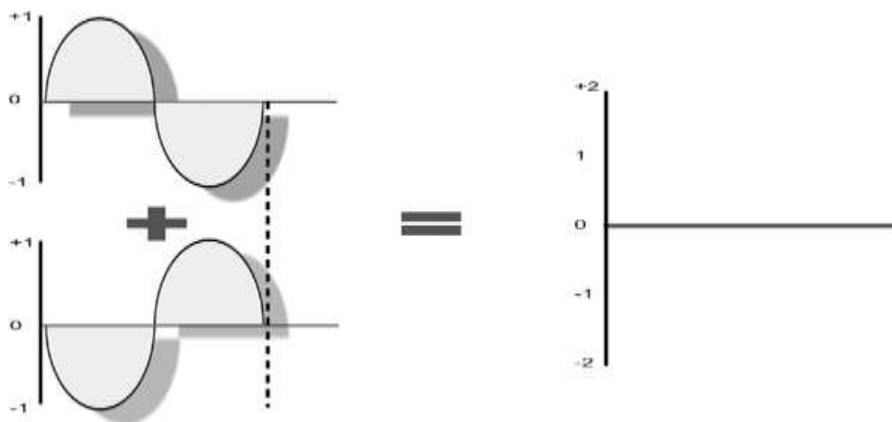


Fig1.11 b

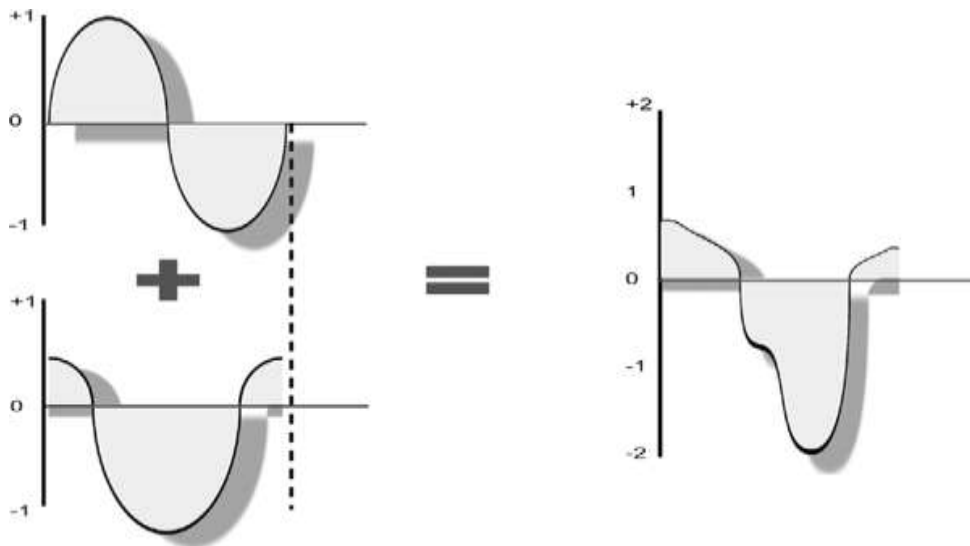


Fig1.11c

PHASE SHIFT

Phase shift is a term that describes one wave-form's lead or lag in time with respect to another. Basically, it results from a time delay between two (or more) waveforms (with differences in acoustic distance being the most common source of this type of delay). For example, a 500-Hz wave completes one cycle every 0.002 sec. If you start with two in phase, 500-Hz waves and delay one of them by 0.001sec (half the wave's period), the delayed wave will lag the other by one-half a cycle, or 180°. Another example might include a single source that's being picked up by two microphones that have been placed at different distances (Figure 1.12), thereby creating a corresponding time delay when the mics are mixed together. Such a delay can also occur when a single microphone picks up direct sounds as well as those that are reflected off of a nearby boundary. These signals will be in phase at frequencies where the path-length difference is equal to the signal's wavelength and out of phase at those frequencies where the multiples fall at or near the half-wavelength distance. In all the above situations, these boosts and cancellations combine to alter the signal's overall frequency response at the pickup. For this and other reasons, acoustic leakage between microphones and reflections from nearby boundaries should be kept to a minimum whenever possible.

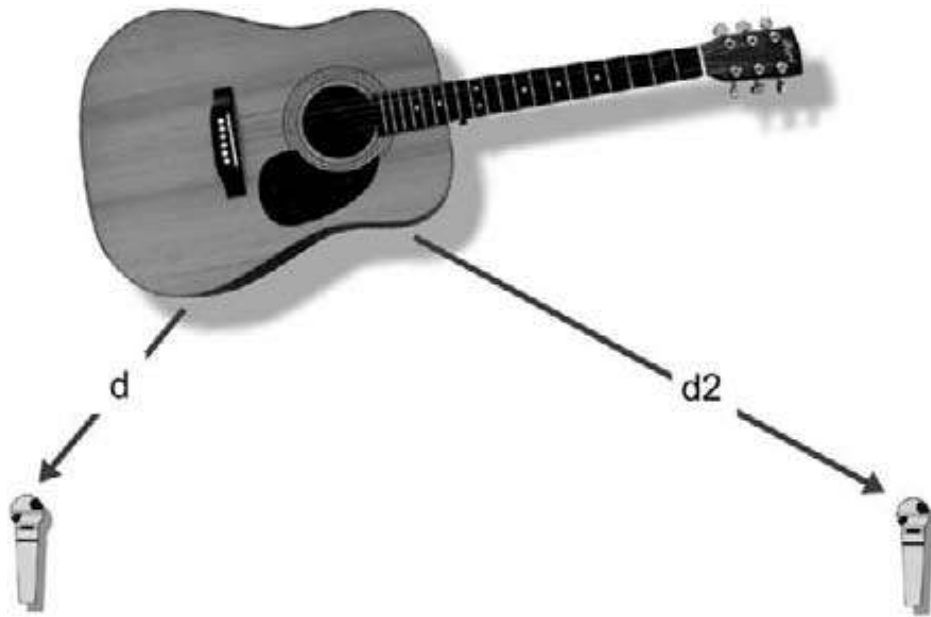


Figure 1.12: Cancellations can occur when a single source is picked up by two microphones.

2.9 HARMONIC CONTENT

Up to this point, the discussion has centered on the sine wave, which is composed of a single frequency that produces a pure sound at a specific pitch. Fortunately, musical instruments rarely produce pure sine waves. If they did, all of the instruments would basically sound the same, and music would be pretty boring. The factor that helps us differentiate between instrumental “voicings” is the presence of frequencies (called partials) that exist in addition to the fundamental pitch that’s being played. Partials that are higher than the fundamental frequency are called upper partials or overtones. Overtone frequencies that are whole-number multiples of the fundamental frequency are called harmonics. For example, the frequency that corresponds to concert A is 440Hz (Figure 1.13a). An 880-Hz wave is a harmonic of the 440-Hz fundamental because it is twice the frequency (Figure 1.13b). In this case, the 440-Hz fundamental is technically the first harmonic because it is 1 times the fundamental frequency, and the 880-Hz wave is called the second harmonic because it is 2 times the fundamental. The third harmonic would be 3 times 440Hz, or 1320Hz (Figure 1.13c). Some instruments, such as bells, xylophones and other percussion instruments, will often contain overtone partials that aren’t harmonically related to the fundamental at all.

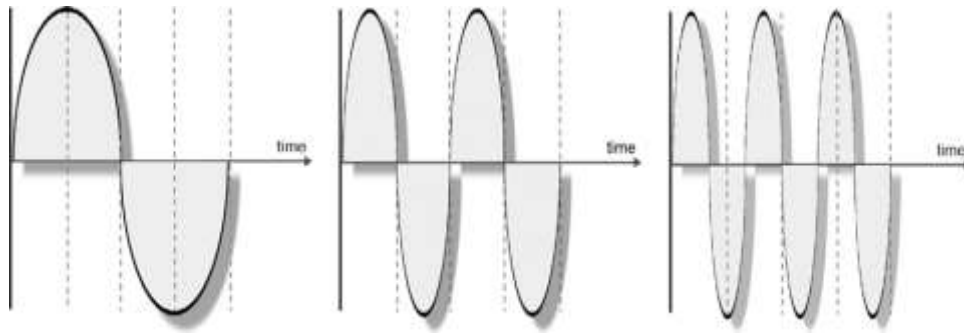


Figure 1.13: An illustration of harmonics: (a) first harmonic “fundamental waveform”; (b) second harmonic; (c) third harmonic.

The ear perceives frequencies that are whole, doubled multiples of the fundamental as being related in a special way (a phenomenon known as the musical octave). For example, as concert A is 440Hz (A3), the ear hears 880Hz (A4) as being the next highest frequency that sounds most like concert A. The next related octave above that will be 1760Hz (A5). Therefore, 880Hz is said to be one octave above 440Hz, and 1760Hz is said to be two octaves above 440Hz, etc. Because these frequencies are even multiples of the fundamental, they’re known as even harmonics. Not surprisingly, frequencies that are odd multiples of the fundamental are called odd harmonics. In general, even harmonics are perceived as creating a sound that is pleasing to the ear, while odd harmonics will create a dissonant, harsher tone.

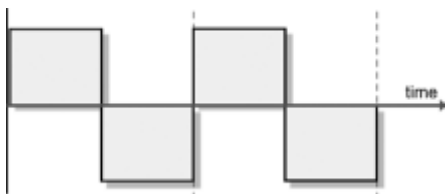


Fig a

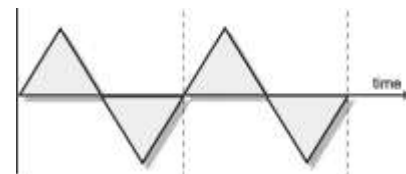


Fig b

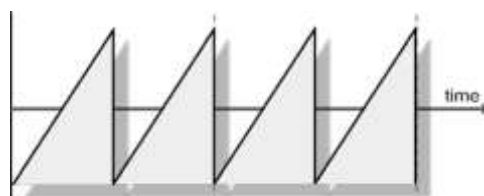


Fig c

Figure 1.14: Simple waveforms: (a) square waves; (b) triangle waves; (c) saw tooth waves.

Because musical instruments produce sound waves that contain harmonics with various amplitude and phase relationships, the resulting waveforms bear little resemblance to the shape of the single-frequency sine wave. Therefore, musical waveforms can be divided into two

categories: simple and complex. Square waves, triangle waves and saw tooth waves are examples of simple waves that contain a consistent harmonic structure (Figure 1.14). They are said to be simple because they're continuous and repetitive in nature. One cycle of a square wave looks exactly like the next, and they are symmetrical about the zero line.

Complex waves, on the other hand, don't necessarily repeat and often are not symmetrical about the zero line. An example of a complex waveform (Figure 1.15) is one that's created by any naturally occurring sound (such as music or speech). Although complex waves are rarely repetitive in nature, all sounds can be mathematically broken down as being an ever-changing combination of individual sine waves.

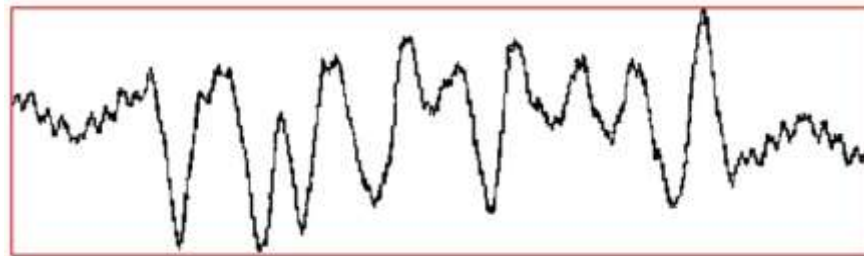


Figure 1.15: Example of a complex waveform.

Regardless of the shape or complexity of a wave-form that reaches the eardrum, the inner ear is able to perceive these component waveforms and transmit the stimulus to the brain. This can be illustrated by passing a square wave through a band pass filter that's set to pass only a narrow band of frequencies at any one time. Doing this would show that the square wave is composed of a fundamental frequency plus a number of harmonics that are made up of odd number multiple frequencies (whose amplitudes decrease as the frequency increases). In Figure 1.16, we see how individual sine-wave harmonics can be combined to form a square wave.

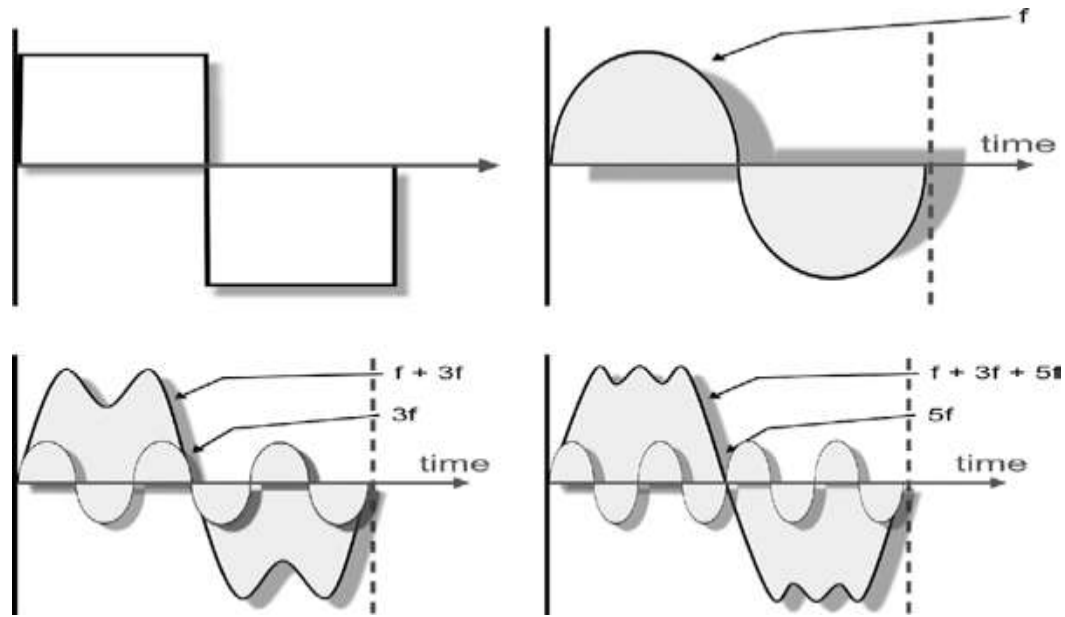


Figure 1.16: Breaking a square wave down into its odd-harmonic components: (a) square wave with frequency f ; (b) sine wave with frequency f ; (c) sum of a sine wave with frequency f and a lower amplitude sine wave of frequency $3f$; (d) sum of a sine wave of frequency f and lower amplitude sine waves of $3f$ and $5f$, which is beginning to resemble a square wave.

If we were to analyze the harmonic content of sound waves that are produced by a violin and compare them to the content of the waves that are produced by a viola (with both playing concert A, 440Hz), we would come up with results like those shown in Figure 1.17. Notice that the violin's harmonics differ in both degree and intensity from those of the viola. The harmonics and their relative intensities (which determine an instrument's characteristic sound) are called the timbre of an instrument. If we changed an instrument's harmonic balance, the sonic character of the instrument would also be changed. For example, if the violin's upper harmonics were reduced, the violin would sound a lot like the viola.



Figure 1.17: Harmonic structure of concert A-440: (a) played on a viola; (b) played on a violin.

Because the relative harmonic balance is so important to an instrument's sound, the frequency response of a microphone, amplifier, speaker and all other elements in the signal path can have

an effect on the timbre (tonal balance) of a sound. If the frequency response isn't flat, the timbre of the sound will be changed. For example, if the high frequencies are amplified less than the low and middle frequencies, then the sound will be duller than it should be. For this reason, a specific mic, mic placement or an equalizer can be used as tools to vary the timbre of an instrument, thereby changing its subjective sound.

In addition to the variations in harmonic balance that can exist between instruments and their families, it is common for the harmonic balance to vary with respect to direction as sound waves radiate from an instrument. Figure 1.18 shows the principal radiation patterns as they emanate from a cello (as seen from both the side and top views).

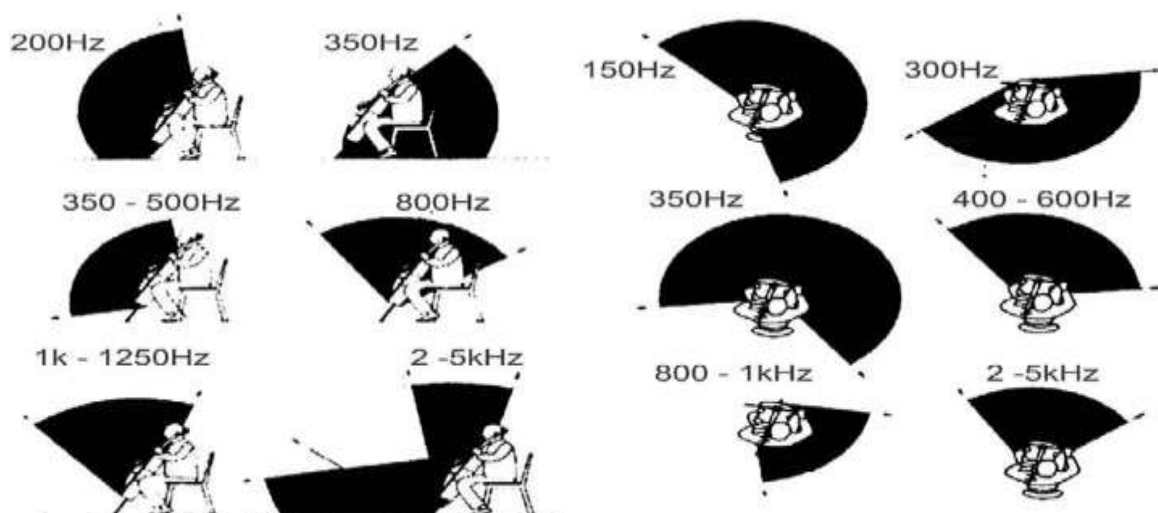


Figure 1.18: Radiation patterns of a cello as viewed from the side (left) and top (right).

2.10 ENVELOPE

Timbre isn't the only characteristic that lets us differentiate between instruments. Each one produces a sonic envelope that works in combination with timbre to determine its unique and subjective sound. The envelope of a wave-form can be described as characteristic variations in level that occur in time over the duration of a played note. The envelope of an acoustic or electronically generated signal is composed of four sections that vary in amplitude over time:

- Attack refers to the time taken for a sound to build up to its full volume when a note is initially sounded.
- Decay refers to how quickly the sound levels off to a sustain level after the initial attack peak.
- Sustain refers to the duration of the ongoing sound that's generated following the initial attack decay.
- Release relates to how quickly the sound will decay once the note is released.

Figure 1.19a illustrates the envelope of a trombone note. The attack, decay times and internal dynamics produce a smooth, sustaining sound. A cymbal crash (Figure 1.19b) combines a high-level, fast attack with a longer sustain and decay that creates a smooth, lingering shimmer. Figure 1.19c illustrates the envelope of a snare drum. Notice that the initial attack is much louder than the internal dynamics ... while the final decay trails off very quickly, resulting in a sharp, percussive sound.

It's important to note that the concept of an envelope relies on peak waveform values, while the human perception of loudness is proportional to the average wave intensity over a period of time (rms value). Therefore, high-amplitude portions of the envelope won't make an instrument sound loud unless the amplitude is maintained for a sustained period. Short high-amplitude sections tend to contribute to a sound's overall character, rather than to its loudness. By using a compressor or limiter, an instrument's character can often be modified by changing the dynamics of its envelope without changing its timbre.

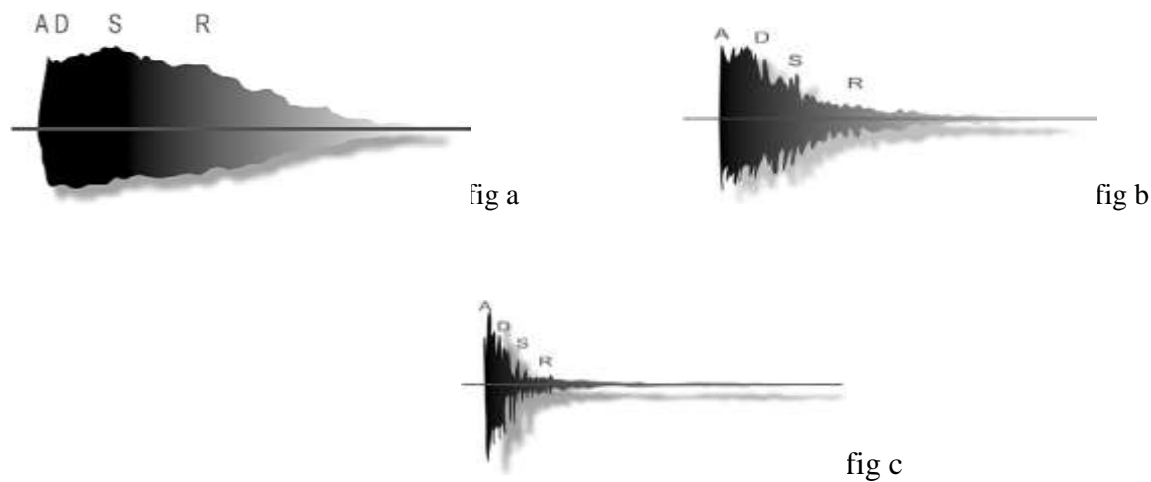


Figure 1.19: Various musical waveform envelopes: (a) trombone, (b) cymbal crash, and (c) snare drum, where A = attack, D = decay, S = sustain, and R = release.

3 Sound measurement and its unit

The ear operates over an energy range of approximately $10^{13}:1$ (10,000,000,000,000:1), which is an extremely wide range. Since it's difficult for us humans to conceptualize number ranges that are this large, a logarithmic scale has been adopted to compress the measurements into figures that are more manageable. The unit used for measuring sound-pressure level

(SPL), signal level and relative changes in signal level is the decibel (dB), a term that literally means 1/10th of a Bell ... a telephone transmission measurement unit that was named after Alexander Graham Bell, inventor of the telephone. In order to develop an understanding of the decibel, we first need to examine logarithms and the logarithmic scale (Figure 1.20). The logarithm (log) is a mathematical function that reduces large numeric values into smaller, more manageable numbers. Because logarithmic numbers increase exponentially in a way that's similar to how we perceive loudness (e.g., 1, 2, 4, 16, 128, 256, 65,536), it expresses our perceived sense of volume more precisely than a linear curve can.

Before we delve into a deeper study of this important concept and how it deals with our perceptual senses, let's take a moment to understand the basic concepts and building block ideas behind the log scale, so as to get a better understanding of what examples such as "+3dB at 10,000Hz" really mean. Be patient with yourself! Over time, the concept of the decibel will become as much a part of your working vocabulary as ounces, gallons and miles per hour.

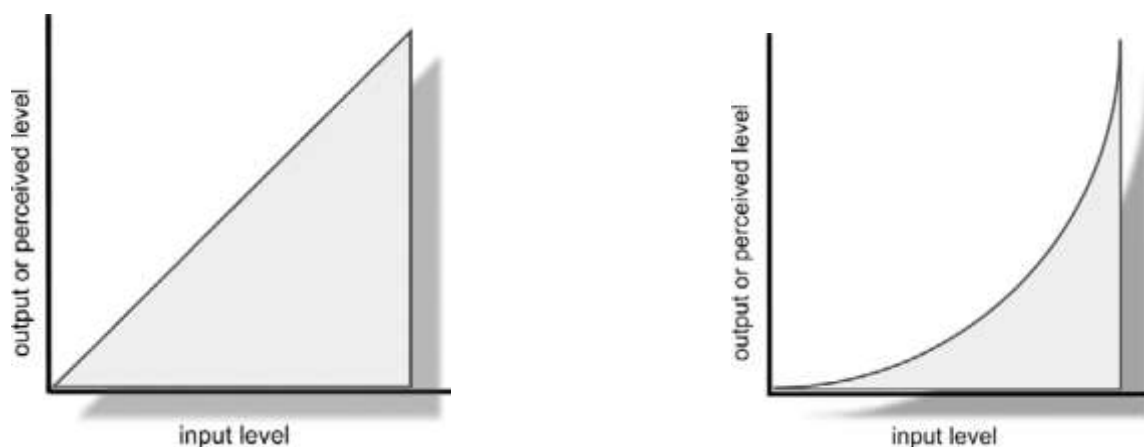


Figure 1.20: Linear and logarithmic curves: (a) linear; (b) logarithmic.

3.1 LOGARITHM BASICS

In audio, we use logarithmic values to express the differences in intensities between two levels (often, but not always, comparing a measured level to a standard reference level). Because the differences between these two levels can be really, really big, a simpler system would make use of expressed values that are mathematical exponents of 10. To begin, finding the log of a number

such as 17,386 without a calculator is not only difficult ... it's unnecessary! All that's really important to help you along are three simple guidelines:

- The log of the number 2 is 0.3.
- When a number is an integral power of 10 (e.g., 100, 1000, 10,000), the log can be found simply by adding up the number of zeros.
- Numbers that are greater than 1 will have a positive log value, while those less than 1 will have a negative log value.

The first one is an easy fact to remember: The log of 2 is 0.3 ... this will make sense shortly. The second one is even easier: The logs of numbers such as 100, 1000 or 10,000,000,000,000 can be arrived at by simply counting up the zeros. The last guideline relates to the fact that if the measured value is less than the reference value, the resulting log value will be negative. For example:

$$\log 2 = 0.3$$

$$\log 1/2 = \log 0.5 = -0.3$$

$$\log 10,000,000,000,000 = 13$$

$$\log 1000 = 3$$

$$\log 100 = 2$$

$$\log 10 = 1$$

$$\log 1 = 0$$

$$\log 0.1 = -1$$

$$\log 0.01 = -2$$

$$\log 0.001 = -3$$

All other numbers can be arrived at by using a scientific calculator (most computers and many cell phones have one built in); however, it's unlikely that you will ever need to know any log values beyond understanding the basic concepts that are listed above.

3.2 THE DECIBEL

Now that we've gotten past the absolute bare basics, I'd like to break with tradition again and attempt an explanation of the decibel in a way that's less complex and relates more to our day-to-day needs in the sound biz. First off, the decibel is a logarithmic value that "expresses differences in intensities between two levels." From this, we can infer that these levels are expressed by several units of measure, the most common being sound-pressure level (SPL), voltage (V) and power (wattage, or W). Now, let's look at the basic math behind these three measurements.

3.3 SOUND-PRESSURE LEVEL

Sound-pressure level is the acoustic pressure that's built up within a defined atmospheric area (usually a square centimeter, or cm²). Quite simply, the higher the SPL, the louder the sound (Figure 1.21).

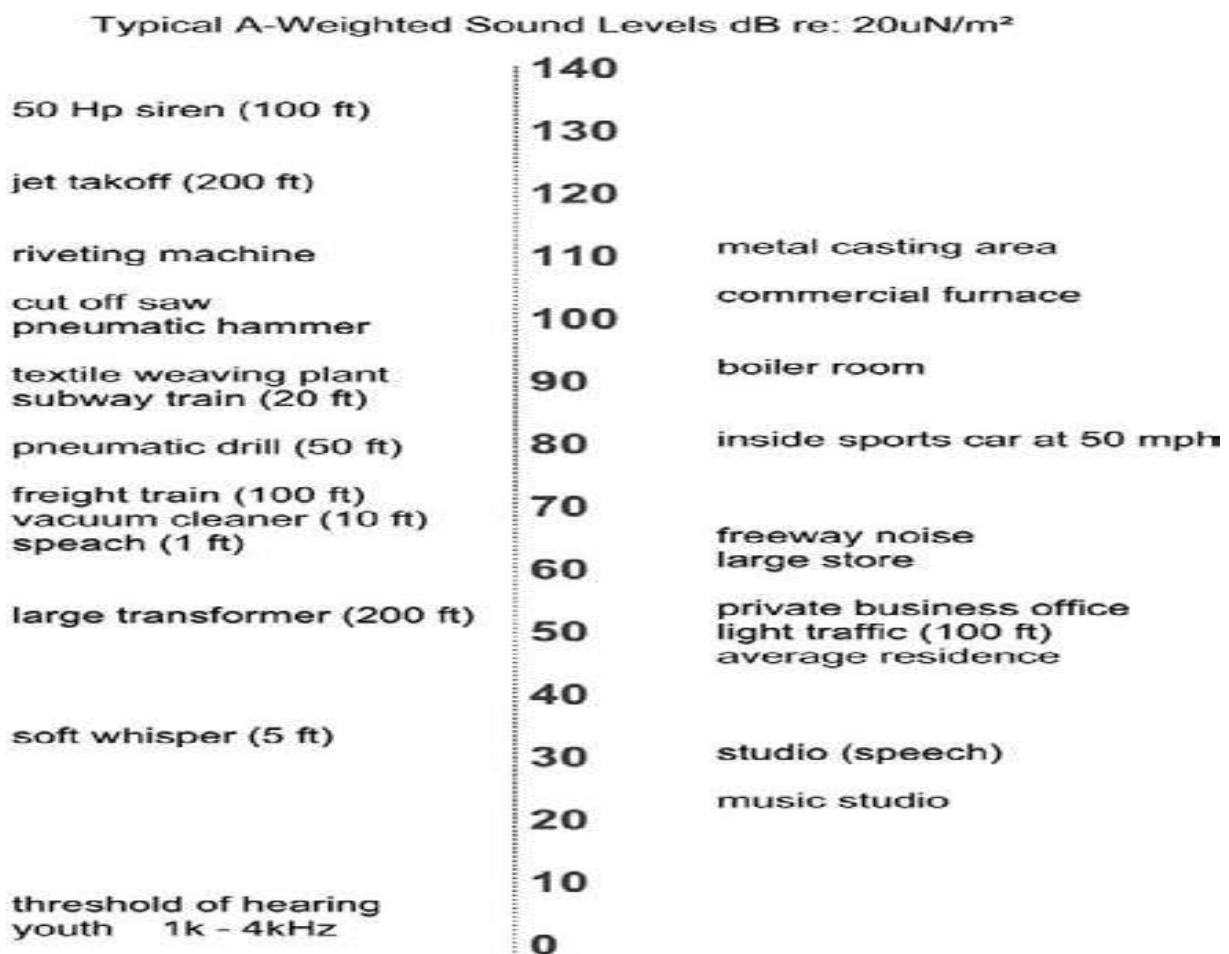


Figure 1.21: Chart of sound-pressure levels. (Courtesy of General Radio Company.)

In this instance, our measured reference (SPL_{ref}) is the threshold of hearing, which is defined as being the softest sound that an average person can hear. Most conversations will have a SPL of about 70dB, while average home stereos are played at volumes ranging between 80 and 90dB SPL. Sounds that are so loud as to be painful have SPLs of about 130 to 140dB (10,000,000,000,000 or more times louder than the 0-dB reference). We can arrive at an SPL rating by using the formula:

$$dB\ SPL = 20 \log SPL/SPL_{ref}$$

where SPL is measured sound pressure (in $dyne/cm^2$). SPL_{ref} is a reference sound pressure (the threshold limit of human hearing, 0.02 millipascals = 2 ten-billionths of our atmosphere).

From this, I feel that the major concept that needs to be understood is the idea that SPL levels change with the square of the distance (hence, the 20 log part of the equation). This means that whenever a source/pickup distance is doubled, the SPL level will be reduced by 6dB ($20 \log 0.5/1 = 20 \times -0.3 = -6dB\ SPL$); as the distance is halved, it will increase by 6dB ($20 \log 2/1 = 20 \times 0.3 = 6dB\ SPL$), as shown in Figure 1.22.

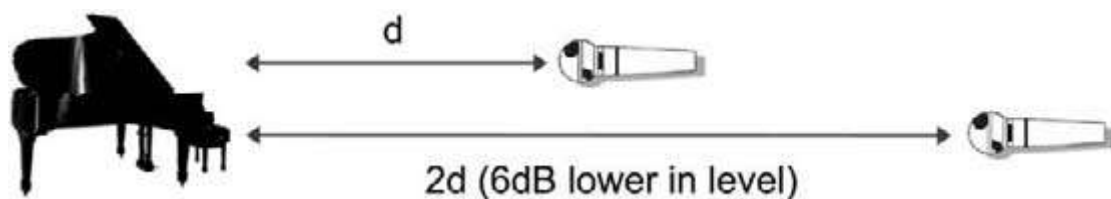


Figure 1.22: Doubling the distance of a pickup will lower the perceived direct signal level by 6dB SPL.

$$dBm = 10 \log P/P_{ref}$$

where P is the measured wattage, and P_{ref} is referenced to 1 milliwatt (0.001 watt).

The simple heart of the matter

I am going to stick my neck out and state that, when dealing with decibels, it's far more common for working professionals to deal with the concept of power. The dBm equation expresses the spirit of the decibel term when dealing with the markings on an audio device or the numeric values in a computer dialog box. This is due to the fact that power is the unit of measure that's most often expressed when dealing with audio equipment controls; therefore, it's my personal opinion that the average working stiff only needs to grasp the following basic concepts:

- A 1-dB change is barely noticeable to most ears.
- Turning something up by 3dB will double the signal's level (believe it or not, doubling the signal level won't increase the perceived loudness as much as you might think).
- Turning something down by 3dB will halve the signal's level (likewise, halving the signal level won't decrease the perceived loudness as much as you might think).
- The log of an exponent of 10 can be easily figured by simply counting the zeros (e.g., the log of 1000 is 3). Given that this figure is multiplied by 10 ($10 \log P/P_{ref}$), turning something up by 10dB will increase the signal's level 10-fold, 20dB will yield a 100-fold increase, 30dB will yield a 1000-fold increase, etc.

Most pros know that turning a level fader up by 3dB will effectively double its energy output (and vice versa). Beyond this, it's unlikely that anyone will ever ask, "Would you please turn that up a thousand times?" It just won't happen! However, when a pro asks his or her assistant to turn the gain up by 20dB, that assistant will often instinctively know what 20dB is ... and what it sounds like. I guess I'm saying that the math really isn't nearly as important as the ongoing process of getting an instinctive feel for the decibel and how it relates to relative levels within audio production.

4 THE EAR

The ear is the organ of hearing and, balance. A sound source produces acoustic waves by alternately compressing and rarefying the air molecules between it and the listener, causing fluctuations that fall above and below normal atmospheric pressure. The human ear is a sensitive transducer that responds to these pressure variations by way of a series of related processes that occur within the auditory organs ... our ears. When these variations arrive at the listener, sound-pressure waves are collected in the aural canal by way of the outer ear's pinna. These are then directed to the eardrum, a stretched drum-like membrane (Figure 1.23), where the sound waves are changed into mechanical vibrations, which are transferred to the inner ear by way of three bones known as the hammer, anvil and stirrup. These bones act both as an amplifier (by significantly increasing the vibrations that are transmitted from the eardrum) and as a limiting protection device (by reducing the level of loud, transient sounds such as thunder or fireworks explosions). The vibrations are then applied to the inner ear (cochlea)—a tubular, snail-like organ that contains two fluid-filled chambers. Within these chambers are tiny hair receptors that are lined up in a row along the length of the cochlea. These hairs respond to certain frequencies depending on their placement along the organ, which results in the neural stimulation that gives us the sensation of hearing. Permanent hearing loss generally occurs when these hair/nerve combinations are damaged or as they deteriorate with age.

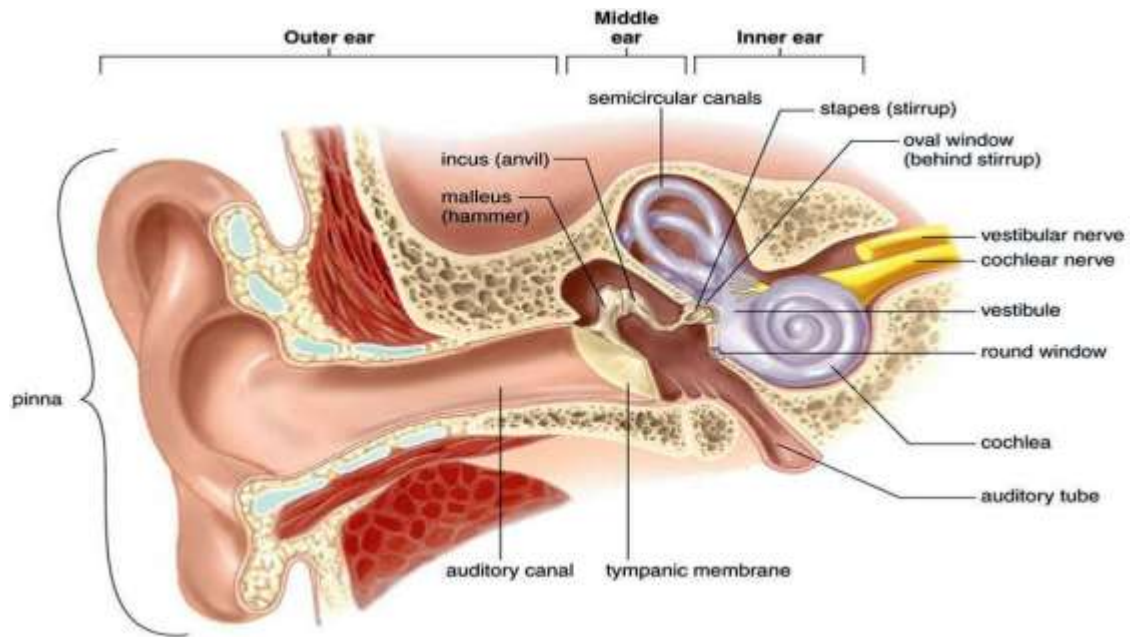


Figure 1.23: Outer, middle, and inner ear.

Hearing

Sound waves travel through the outer ear, are modulated by the middle ear, and are transmitted to the vestibule cochlear nerve in the inner ear. This nerve transmits information to the temporal lobe of the brain, where it is registered as sound.

Sound that travels through the outer ear impacts on the eardrum, and causes it to vibrate. The three ossicles bones transmit this sound to a second window (the oval window) which protects the fluid-filled inner ear. In detail, the pinna of the outer ear helps to focus a sound, which impacts on the eardrum. The malleus rests on the membrane, and receives the vibration. This vibration is transmitted along the incus and stapes to the oval window. Two small muscles, the tensor tympani and stapedius, also help modulate noise. The two muscles reflexively contract to dampen excessive vibrations. Vibration of the oval window causes vibration of the endolymph within the vestibule and the cochlea. (Hall, 2005)

The inner ear houses the apparatus necessary to change the vibrations transmitted from the outside world via the middle ear into signals passed along the vestibulo cochlear nerve to the brain. The hollow channels of the inner ear are filled with liquid, and contain a sensory epithelium that is studded with hair cells. The microscopic "hairs" of these cells are structural protein filaments that project out into the fluid. The hair cells are mechanoreceptors that release a chemical neurotransmitter when stimulated. Sound waves

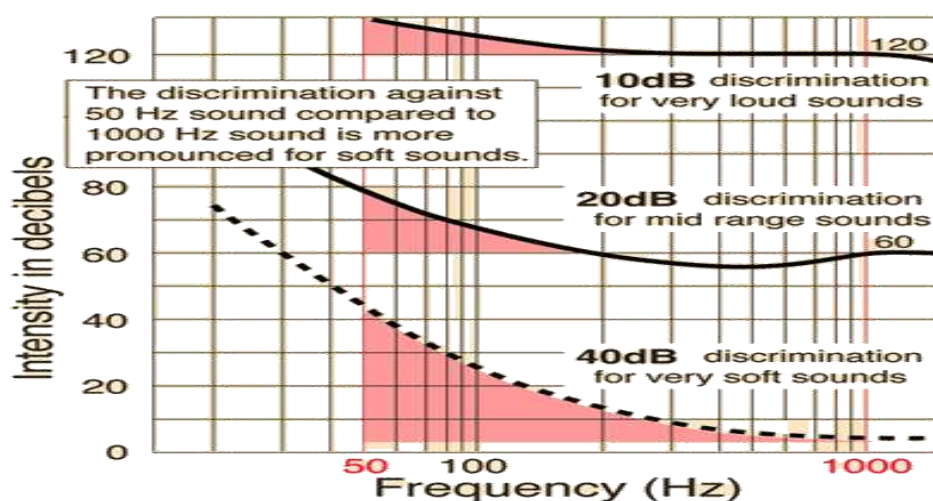
moving through fluid flows against the receptor cells of the organ of Corti. The fluid pushes the filaments of individual cells; movement of the filaments causes receptor cells to become open to receive the potassium-rich endolymph. This causes the cell to depolarise, and creates an action potential that is transmitted along the spiral ganglion, which sends information through the auditory portion of the vestibulo cochlear nerve to the temporal lobe of the brain.

The human ear can generally hear sounds with frequencies between 20 Hz and 20 kHz (the audio range). Sounds outside this range are considered infrasound (below 20 Hz) (Greinwald, 2002) or ultrasound (above 20 kHz) (Collins English Dictionary, 2016). Although hearing requires an intact and functioning auditory portion of the central nervous system as well as a working ear, human deafness (extreme insensitivity to sound) most commonly occurs because of abnormalities of the inner ear, rather than in the nerves or tracts of the central auditory system.

4.1 Threshold of hearing

The hearing threshold is defined as the lowest threshold of acoustic pressure sensation, possible to perceive by an organism. It is a subjective value which may differ individually. The hearing threshold forms the lowest limit of the hearing range (the highest limit is formed by the threshold of pain).

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.



In the case of SPL, a convenient pressure-level reference is the threshold of hearing, which is the minimum sound pressure that produces the phenomenon of hearing in most people and is equal to 0.0002 microbar. One microbar is equal to 1 millionth of normal atmospheric pressure, so it's apparent that the ear is an amazingly sensitive instrument. In fact, if the ear were any more sensitive, the thermal motion of molecules in the air would be audible! When referencing SPLs to 0.0002 microbar, this threshold level usually is denoted as 0dB SPL, which is defined as the level at which an average person can hear a specific frequency only 50% of the time.

4.2 Threshold of feeling

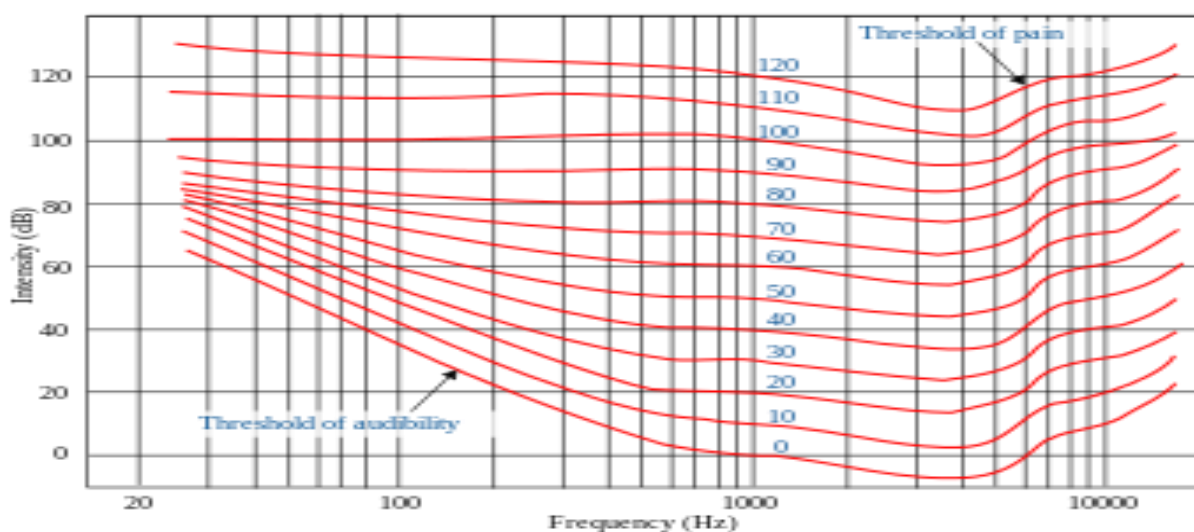
The threshold of 'feeling' is the sound pressure level at which people feel discomfort 50 per cent of the time. Approximately 118 dB SPL at 1 KHz. The threshold of 'pain' is the sound pressure level at which people feel actual pain 50 per cent of the time. Approximately 140 dB SPL at 1 KHz.

An SPL that causes discomfort in a listener 50% of the time is called the threshold of feeling. It occurs at a level of about 118dB SPL between the frequencies of 200Hz and 10kHz.

4.3 Threshold of pain

The threshold of pain or pain threshold is the point along a curve of increasing perception of a stimulus at which pain begins to be felt. It is an entirely subjective phenomenon.

The SPL that causes pain in a listener 50% of the time is called the threshold of pain and corresponds to an SPL of 140dB in the frequency range between 200Hz and 10kHz.



4.4 Taking care of your hearing

During the 1970s and early 1980s, recording studio monitoring levels were often turned so high as to be truly painful. In the mid-1990s, a small band of powerful producers and record executives banded together to successfully reduce these average volumes down to tolerable levels (85 to 95dB) ... a practice that continues to this day. Live sound venues and acts often continue the practice of raising house and stage volumes to chest-thumping levels. Although these levels are exciting, long-term exposure can lead to temporary or permanent hearing loss. So what types of hearing loss are there?

- Acoustic trauma: This happens when the ear is exposed to a sudden, loud noise in excess of 140 dB. Such a shock could lead to permanent hearing loss.
- Temporary threshold shift: The ear can experience temporary hearing loss when exposed to long-term, loud noise.
- Permanent threshold shift: Extended exposure to loud noises in a specific or broad hearing range can lead to permanent hearing loss in that range. In short, the ear becomes less sensitive to sounds in the damaged frequency range leading to a reduction in perceived volume. ... What?

Here are a few hearing conservation tips (courtesy of the House Ear Institute, www.hei.org) that can help reduce hearing loss due to long-term exposure of sounds over 115dB:

- Avoid hazardous sound environments; if they are not avoidable, wear hearing protection devices, such as foam earplugs, custom-molded ear-plugs, or in-ear monitors.
- Monitor sound-pressure levels at or around 85dB. The general rule to follow is if you're in an environment where you must raise your voice to be heard, then you're monitoring too loudly and should limit your exposure times.
- Take 15-minute "quiet breaks" every few hours if you're being exposed to levels above 85dB.
- Musicians and other live entertainment professionals should avoid practicing at concert-hall levels whenever possible.
- Have your hearing checked by a licensed audiologist.

Self Assessment Questions

- Q 1 Name the basic characteristics of a wave form?
- Q2 Define and explain the following
- A amplitude frequency
 - B frequency
 - C wavelength
- Q3 fill in the blanks
- 1 The number of cycles that occur within a second is called its _____
 - 2 The measurement of either the maximum positive or negative signal level of a wave is called its _____
 - 3 rms voltage = _____ x _____
 - 4 peak voltage = _____ x _____
 - 5 The wavelength of a waveform frequently represented by the _____

Answers

- 1) Frequency
 - 2) Peak amplitude value (or peak value)
 - 3) Peak voltage $\times 0.707$
 - 4) rms voltage $\times 1.414$
 - 5) Greek letter lambda, λ
- Q4 Draw the following wave forms
- A sine wave
 - B saw tooth wave
 - C square wave
- Q5 Describe the envelope of a waveform. Explain its four sections that vary in amplitude over time?
- Q6 Explain the working of human ear as a transducer?
- Q. 7 Explain the primary function of the human ear.
- Q. 8 What do you understand by frequency, amplitude, wavelength, velocity, phase and intensity of sound?

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Web links for short videos on different topics of the unit.

- <https://www.youtube.com/watch?v=oWdQLTxwaDI>
- https://www.youtube.com/watch?v=R7-v-MaC_GI
- https://www.youtube.com/watch?v=qrZTgt9_0pE&index=3&list=PLD038BF29A4EB7A39
- <https://www.youtube.com/watch?v=jveKIYyafaQ&index=9&list=PLD038BF29A4EB7A39>
- <https://www.youtube.com/watch?v=qgdqp-oPb1Q>
- <https://www.youtube.com/watch?v=L4F4zaRqQdk>

Unit Two

Acoustics

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Reviewer: Zahid Majeed

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INTRODUCTION

Dear student this chapter will introduce acoustics which is the science of sound, that is, wave motion in different medium, and the effects of such wave motion. Thus the scope of acoustics ranges from fundamental physical acoustics to, say, psychoacoustics and music, and includes technical fields such as transducer technology, sound recording and reproduction, and noise control. The purpose of this unit is to give an introduction to fundamental acoustic concepts, to the physical principles of acoustic wave motion, and to acoustic measurements. By introducing you the basics, give you the better understanding of the properties of sound

In this unit you will also learn the detailed behavior of a sinusoidal waveform producing oscillations and the variation of these oscillations producing the effects in standard terminologies. You will get a better idea that how the sound waves propagate and you will be able to distinguish between types of waves used to transmit energy through a medium or substance, and the relationship between the speed of sound and its effect with the change in frequency. You will also get an idea of another cause in the deviation of sound which is the humidity factor in air, you will learn that how the humidity effects sound propagation.

This unit will cover another major area in acoustics that is psychoacoustics which is a multidisciplinary field that deals with the physical (e.g. vibrations, wave theory), physiological (e.g. construction of the ear), and perceptual (e.g. auditory sensations) correlates of sound production, transmission, and reception. More specifically, it forges a link among the physical, physiological, and perceptual frames of reference. It deals with how and why the brain interprets a particular sound stimulus in a certain way. As the ear is a nonlinear device means that hearing can be changed by many factors and the behavior of ear at different frequency response changes with the loudness of the perceived signal. As a result of the nonlinearities in the ear's response, tones will often interact with each other rather than being perceived as being separate. The type of interaction that effect different tones will also be studied and student will able to differentiate between Beats, different Combination of tones and Masking.

As soon the sound passes through ear, it stops being a physical phenomenon and becomes a matter of perception. Next thing that you will learn from this chapter is the sound perception which is mostly dependent on the two major factors out of these two first is the direction and the other one is space. The study of this topic will allow you to perceive the direction of a sound's origin which is either from the left , right, front, behind or below. And the study will also allow

you to have a better idea of perceiving the direct sound, early reflection, reverberation and doubling of sound field types that are generated within an enclosed space.

During the study of this unit you will learn the behavior of sound by applying the electronically controlled circuit variations in the frequency range. The unit will provide you a complete idea of the four major types of filters and their frequency response curves which shows that how filters are used for stopping high frequencies and allows the low frequency range to pass through them and vice versa. You will also study that how a range of frequency technically called bandwidth can easily be pass or stop while using the filters. The study of the chapter also helps you in finding the working of all the filters in a combine formation which we normally called equalizers and a brief idea about the working of graphic equalizers

While using equalizers you may get a better response or required response but you have to be aware of the damage that you're inflicting on some other parts of the signal. The chapter will also cover the study of major noises and the ways through which we can efficiently control them.

OBJECTIVES:

After studying this unit you will be able to:

1. explain about the acoustic and its types i.e. psycho acoustics and electro acoustics
2. Identify the harmonics and overtones.
3. Know about the speed of sound and its deviation with speed and humidity
4. Differentiate between beats combinational tones and masking
5. Know about the perception of sound with space.
6. Differentiate between direct sound, reflections and reverberation of sound due to direction.
7. Explain the function types and the purpose of filters and equalizers.
8. Explain different types of noises and their causes.

1. PRESSURE

If you listen to the radio in the mornings, they'll give you the news, the sports, the traffic and the weather. Part of the weather report is to tell you that the barometric pressure is something around 100 kilopascals (abbreviated kPa). What does this mean? Well, the air particles around you are all under pressure due to things like gravity and the weight of the air particles above them and other meteorological things that are outside the scope of this unit. That pressure determines the amount of physical space between molecules in the air. When there's a higher barometric pressure, there's less space between the molecules than there is on a day with a lower barometric pressure.

We call this the *stasis pressure* and abbreviate it p_0 .

When all of the particles in a gaseous medium (like air) in a given volume (like a room) are at normal pressure, then the gas is said to be at its volume density (also known as the constant equilibrium density), abbreviated ρ_0 , and measured in kg/m^3 . Remember that this is actually kilograms of air per cubic meter – if you were able to trap a cubic meter and weigh it, you'd find out that it's about 1.3 kg.

These molecules like to stay at the same pressure all over, so if you bunch them up in one place in a room somehow, they'll move around to try and equalize the difference. This is kind of like when you pour a glass of water into a bucket, the water level of the entire bucket equalizes and therefore rises, rather than the water from the glass all bunching up in a little mound of water where you poured it in...

Let's think of this as a practical example. We'll hang the piece of paper in front of a fan. If we turn on the fan, we're essentially increasing the pressure of the air particles in front of the blades. The fan does this by removing air particles from the space behind it, thus reducing the pressure of the particles behind the blades, and putting them in front. Since the pressure in front of the fan is greater than any other place in the room, we have a situation where there is a greater air pressure on one side of the piece of paper than the other. The obvious result is that the paper moves away from the fan.

This is a large-scale example of how you hear sound. Let's say hypothetically for a moment, that you are sitting alone in a sealed room on a day when the barometric pressure is 100 kPa. Let's

also say that you have a clarinet with you and that you play a concert A. What physically happens to convert air coming out of your mouth into a concert A coming in your ears?

To begin with, let's pretend that a clarinet is just a tube with a hole in each end. One of the holes has a springy piece of wood next to it which, if you press on it, will close up the hole.

1. When you blow into the hole, you bunch up the air particles and create a little area of high pressure inside the mouthpiece.
2. Blowing into the hole with the reed on it also has the effect of pushing the reed against the hole and sealing it so that no more air can enter the clarinet.
3. At that point the little high pressure area moves down the clarinet and leaves a low pressure behind it.
4. Remember that the reed is springy, and it doesn't like being pushed up against the hole in the mouthpiece, so it bounces back and lets more air in.
5. Now the cycle repeats and goes back to step 1 all over again.
6. In the meantime, all of those high and low pressure areas move down the clarinet and radiate out the bell into the room like ripples on a lake when you throw in a rock.
7. From there, they get to your ear and push your eardrum in and out (high pressure pushes in, low pressure pulls out)

Those little fluctuations in the air pressure are small variations in the stasis pressure p_0 . They're usually very small, never more than about ± 1 Pa (though we'll elaborate on that later...). At any given moment at a specific location, we can measure the the *instantaneous pressure*, p , which will be close to the stasis pressure, but slightly different because there's a sound source causing it to change.

Once we know the stasis pressure and the instantaneous pressure, we can use these to figure out the *instantaneous amplitude* of the sound level, (also called the *acoustic pressure* or the *excess pressure*) abbreviated p , using Equation 2.1.

$$p - p_0 \dots\dots\dots(2.1)$$

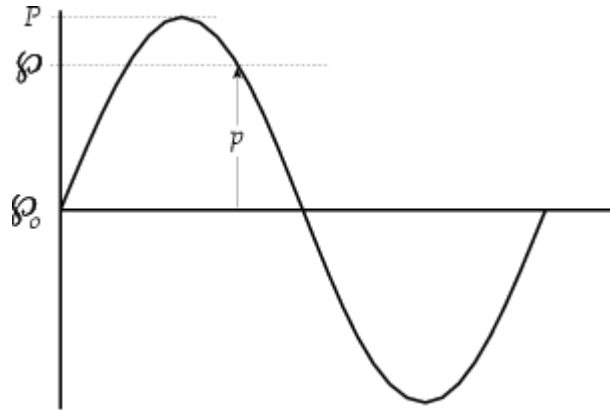


Figure 2.1: A graphic representation of the values p_0 (the stasis pressure), p (instantaneous amplitude), p_0 (instantaneous pressure), and P (maximum peak pressure).

To see an animation of what this looks like, check out www.gmi.edu/~drussell/Demos/waves/wavemotion.html.

A sinusoidal oscillation of this pressure reaches a maximum peak pressure P which is used to determine the *sound pressure level* or *SPL*. In air, this level is typically expressed in decibels as a logarithmic ratio of the *effective pressure* P_e referenced to the *threshold of hearing*, the commonly-accepted lowest sound pressure level audible by humans at 1 kHz, 20 microPascals, using Equation 2.2 [Woram, 1989].

$$= 20 \log_{10} \left[\frac{P_e}{20 * 10^{-6} Pa} \right] \dots\dots\dots (2.2)$$

Note that, for sinusoidal waveforms, the effective pressure can be calculated from the peak pressure using Equation 2.3

$$P_e = \frac{P}{\sqrt{2}} \dots\dots\dots (2.3)$$

2. SIMPLE HARMONIC MOTION

Take a weight (a little one...) and hang it on the end of a Slinky which is attached to the ceiling and wait for it to stop bouncing.

Measure the length of the Slinky. This length is determined by the weight and the strength of the Slinky. If you use a bigger weight, the Slinky will be longer – if the slinky is stronger, it will be better able to support the weight and therefore be shorter.

This is the point where the “system” is at rest or stasis.

Pull down on the weight a little bit and let go. The Slinky will pull the weight up to the stasis point and pass it.

By the time the whole thing slows down, the weight will be too high and will want to come back down to the stasis point, which it will do, stopping at the point where we let it go in the first place (or almost anyway...)

If we attached a pen to the weight and ran piece of paper along by it as it sat there bobbing up and down, the line it would draw a sinusoidal waveform. The picture the weight would draw is a graph of the vertical position of the weight (the y-axis) as it relates to time (the x-axis).

If the graph is a perfect sinusoidal shape, then we call the system (the Slinky and the weight on the end) a simple harmonic oscillator.

3 DAMPING

Let’s look at that system I just described. We’ll put a weight hung on a spring as is shown in Figure 2.2

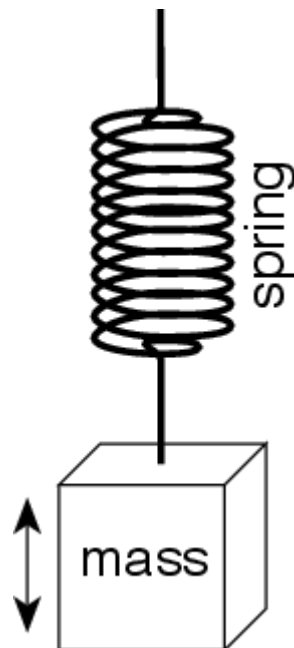


Figure 2.2: A mass supported by a spring.

If there was no such thing as air friction, and if the spring was perfect, then, if you started the mass bobbing up and down, then it would continue doing that forever. Since, as we saw in the previous section, that this is a simple harmonic oscillator, if we graph its vertical displacement over time, then we get a perfect sinusoidal waveform as shown in Figure 2.3

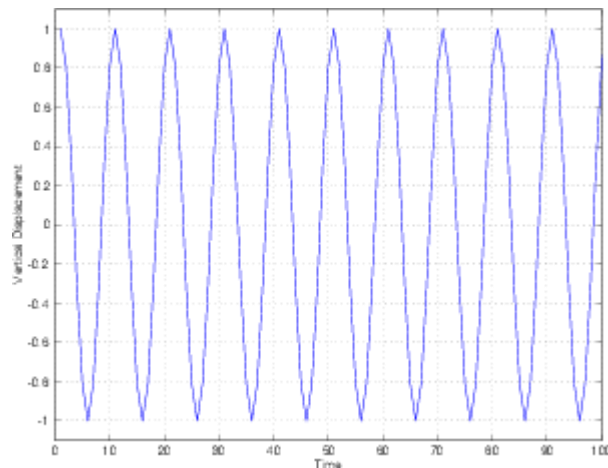


Figure 2.3: The vertical displacement of the mass versus time if there is no loss of energy due to friction. Notice that the frequency and amplitude of the oscillation never change. The mass will bob up and down exactly the same, forever.

In real life, however, there is friction. The mass pushes through the air and loses energy on each bob up and down. Eventually, it loses so much energy that it stops moving. An example of this behaviour is shown in Figure 2.4

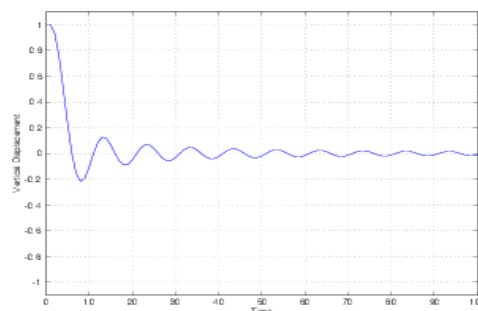


Figure 2.4: The vertical displacement of the mass versus time if there is loss of energy due to friction. Notice that the fundamental frequency of the oscillation never changes, but that the amplitude decays over time. Eventually, the mass will bob up and down so little that it can be considered to be stopped.

There is a technical term that describes the difference between these two situations. The system with friction, shown in Figure 2.4 is called a *damped oscillator*. Since the oscillator is damped,

then it loses energy over time. The higher the damping, the faster it loses energy. For example, if the same mass and spring were put in water, the system would be more highly damped than if it were in air. If they're put in oil, the system is more highly damped than it is in water.

Since a system with friction is said to be damped, then the system without friction is therefore called an un damped oscillator.

4. OVERTONES

Some people call the fundamental and its harmonics *overtones* but you have to be careful here. There is a common misconception that overtones are harmonics and vice versa. In fact, in some books, you'll see people saying that the first overtone is the second harmonic; the second overtone is the third harmonic and so on. This is not necessarily the case. A sound's overtones are the harmonics that it contains, which is not necessarily all harmonics. As we'll see later, not all instruments' sounds contain all harmonics of the fundamental. There are particular cases, for example, where an instrument's sound will only contain the odd harmonics of the fundamental. In this particular case, the first overtone is the third harmonic; the second overtone is the fifth harmonic and so on.

In other words, harmonics are a mathematical idea – frequencies that are related to a fundamental frequency whereas overtones are the frequencies that are produced by the sound source.

Another example showing that overtones are not harmonics occurs in many percussion instruments such as bells where the overtones have no harmonic relationship with the fundamental frequency – which is why these overtones are said to be enharmonically related.

5. LONGITUDINAL VS. TRANSVERSE WAVES

There are basically three types of waves used to transmit energy through a medium or substance.

1. Transverse
2. Longitudinal
3. Torsional

We're only really concerned with the first two.

Transverse waves are the kind we see every day in ropes and puddles. They're the kind where the motion of the particles is perpendicular to the direction of the wave propagation as can be seen in Figure 2.5. What does this mean? It's easy to see if we go fishing. A boat on the surface of the ocean will sit there bobbing up and down as the waves roll past it. The waves are traveling towards the shore along the surface of the water, but the water itself only moves up and down, not sideways (we know this because the boat would move sideways as well if the water was doing so...) So, as the water molecules move vertically, the wave propagates horizontally.



Figure 2.5: A snapshot of a transverse wave on a string. Think of the wave as moving from left to right – but remember that the string is really only moving up and down.

Longitudinal waves are a little tougher to see. They involve the compression (bunching together) and rarefaction (pulling apart) of the particles in the medium such that the motion of the particles is parallel with the direction of propagation of the wave. The easiest way to see a longitudinal wave is to stretch out a Slinky between two people, squeeze together a small section of it and let go. The compressed part will appear to move back and forth bouncing between the two ends of the spring. This is essentially the way sound travels through air particles.

Torsional waves don't apply to anything we're doing in this book, but they're waves in which the particles rotate around the axis along which the wave propagates (like a twisting rod). This type of wave can be seen on a Shive wave machine at physics demonstrations and science and technology museums.

6. DISPLACEMENT VS VELOCITY

Think back to our original discussions concerning sound. We said that there are really two things moving in a sound wave – the air molecules (which are compressing and expanding) and the pressure wave which propagates outwardly from the sound source. We compared this to a wave moving along a rope. The rope moves up and down, but the wave moves in another direction entirely.

Let's now think of this difference in terms of displacement and velocity – not of the sound wave itself (which is about 344 m/s at room temperature) but of the air molecules.

When a sound wave goes by a bunch of molecules, they compress and expand. In other words, they move closer together, then stop moving, then move further apart, then stop moving, then move closer together and so on. When the displacement is at its absolute maximum, the molecules are at the point where they're stopped and about to head back towards a low pressure. When the displacement is 0 (and therefore at whatever barometric pressure the radio said it was this morning) the molecules are moving as fast as they can. If the displacement is at a maximum in the opposite direction, the molecules are stopped again.

When pressure is 0, the particle velocity is at a maximum (or a minimum) whereas when pressure is at a maximum (or a minimum) the particle velocity is 0.

This is identical to swinging on a playground swing. When you're at the highest point off the ground, you're stopped and about to head in the direction from which you just came. Therefore at the point of maximum displacement, you have a velocity of 0. When you're at the point closest to the ground (where you started before you were moving) your velocity is highest.

So, in addition to measurements like instantaneous pressure, we can also talk about an instantaneous particle velocity, u . In addition, a sinusoidal oscillation results in a peak particle velocity, U .

Always remember that the particle velocity is dependent on the change in displacement, therefore it is equivalent to the instantaneous slope (or the partial derivative) of the displacement function. As a result, the velocity wave precedes the displacement wave by $\frac{\pi}{2}$ radians (or 90°) as is shown in Figure 2.6.

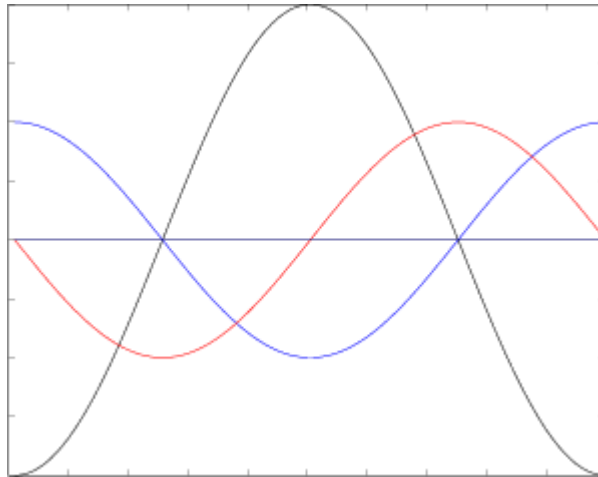


Figure 2.6: The relationship between the displacement (in blue), the velocity (in red) and the acceleration (in black) of a particle or a pendulum. Note that none of these is on any particular scale – the important things to notice are the relationships between the zero, maximum and minimum points on the two graphs as well as their relative instantaneous slopes.

One other important thing to note here is that the velocity is also related to frequency (which is discussed below). If we maintain the same peak pressure, the higher the frequency, the faster the particles have to move back and forth, therefore the higher the peak velocity. So, remember that particle velocity is proportional both to pressure (and therefore displacement) and frequency.

2.7 SPEED OF SOUND

Pay attention during any thunder and lightning storm and you'll be able to figure out that sound travels slower than light. Since the lightning and the thunder occur simultaneously and since the light flash arrives at you earlier than the clap of thunder (unless you're extremely unlucky...) then this must be true. In fact, the speed of sound, abbreviated c is around 344 m/s although it changes with temperature, pressure and humidity.

Note that we're talking about the speed of the wave front – not the velocity of the air molecules. This latter velocity is dependent on the waveform, as well as its frequency and the amplitude.

The equation we normally use for c in meters per second is

$$c = 332 + (0.6 * t)$$

where t is the temperature in $^{\circ}\text{C}$

There is a small deviation of c with frequency shown in Table 2.1, though this is small and therefore generally ignored

Frequency	Deviation
100 Hz	-30 ppm
200 Hz	-10 ppm
400 Hz	-3 ppm
1.25 kHz	0 ppm
4 kHz	+5 ppm
10 kHz	+10 ppm

Table 2.1: Deviation in the speed of sound with frequency. [Kuttruff, 1991]

Changes in humidity change the value of c as is seen in Table 2.2.

Humidity	Deviation
0%	0 ppm
20%	+415 ppm
40%	+1136 ppm
60%	+1860 ppm
80%	+2590 ppm
100%	+3320 ppm

Table 2.2: Deviation in the speed of sound with air humidity levels. [Kuttruff, 1991]

The difference at a humidity level of 100% of 0.33% is bordering on our ability to detect a pitch shift.

Also – in case you were wondering, “ppm” stands for “parts per million.” It’s just like “percent” really, except that you divide by 1000000 instead of 100 so it’s useful for really small numbers. Therefore 1000 ppm is $\frac{1000}{1000000} = 0.001 = 0.1\%$.

8. PSYCHO ACOUSTICS

The area of psychoacoustics deals with how and why the brain interprets a particular sound stimulus in a certain way. Although a great deal of study has been devoted to this subject, the primary device in psychoacoustics is the all-elusive brain ... which is still largely unknown to present-day science.

9. AUDITORY PERCEPTION

From the outset, it's important to realize that the ear is a nonlinear device (what's received at your ears isn't always what you'll hear). It's also important to note that the ear's frequency response (its perception of timbre) changes with the loudness of the perceived signal. The loudness compensation switch found on many hi-fi preamplifiers is an attempt to compensate for this decrease in the ear's sensitivity to low- and high-frequency sounds at low listening levels.

The Fletcher–Munson equal-loudness contour curves (Figure 2.7) indicate the ear's average sensitivity to different frequencies at various levels. These indicate the sound-pressure levels that are required for our ears to hear frequencies along the curve as being equal in level to a 1000-Hz reference level (measured in phons). Thus, to equal the loudness of a 1-kHz tone at 110dB SPL (a level typically created by a trumpet-type car horn at a distance of 3 feet), a 40-Hz tone has to be about 6dB louder, whereas a 10-kHz tone must be 4dB louder in order to be perceived as being equally loud. At 50dB SPL (the noise level present in the average private business office), the level of a 40-Hz tone must be 30dB louder and a 10-kHz tone 13dB louder than a 1-kHz tone to be perceived as having the same volume. Thus, if a piece of music is mixed to sound great at a level of 85 to 95dB, its bass and treble balance will actually be boosted when turned up (often a good thing). If the same piece were mixed at 110dB SPL, it would sound both bass and treble shy when played at lower levels ... because no compensation for the ear's response was added to the mix. Over the years, it has generally been found that changes in apparent frequency balance are less apparent when monitoring at levels of 85dB SPL.

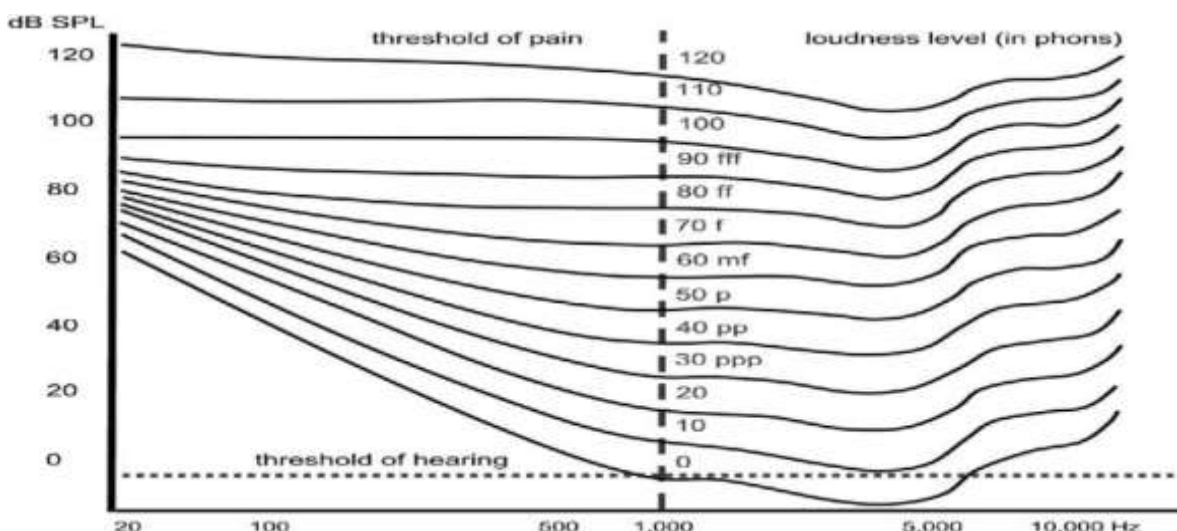


Figure 2.7 The Fletcher–Munson curve shows an equal loudness contour for pure tones as perceived by humans having an average hearing acuity. These perceived loudness levels are charted relative to sound-pressure levels at 1000Hz.

In addition to the above, whenever it is subjected to sound waves that are above a certain loudness level, the ear can produce harmonic distortion that doesn't exist in the original signal. For example, the ear can cause a loud 1-kHz sine wave to be perceived as being a combination of 1-, 2-, 3-kHz waves, and so on. Although the ear might hear the overtone structure of a violin (if the listening level is loud enough), it might also perceive additional harmonics (thus changing the timbre of the instrument). This is one of several factors that implies that sound monitored at very loud levels could sound quite different when played back at lower levels.

The loudness of a tone can also affect our ear's perception of pitch. For example, if the intensity of a 100-Hz tone is increased from 40 to 100dB SPL, the ear will hear a pitch decrease of about 10%. At 500Hz, the pitch will change about 2% for the same increase in sound-pressure level. This is one reason why musicians find it difficult to tune their instruments when listening through loud headphones.

As a result of the nonlinearities in the ear's response, tones will often interact with each other rather than being perceived as being separate. Three types of interaction effects can occur:

- **Beats**
- **Combination tones**
- **Masking.**

10. BEATS

Two tones that differ only slightly in frequency and have approximately the same amplitude will produce an effect known as beats. This effect sounds like repetitive volume surges that are equal in frequency to the difference between these two tones. The phenomenon is often used as an aid for tuning instruments, because the beats slow down as the two notes approach the same pitch and finally stop when the pitches match. In reality, beats are a result of the ear's inability to separate closely pitched notes. This results in a third frequency that's created from the phase sum and difference values between the two notes.

11. COMBINATION TONES

Combination tones result when two loud tones differ by more than 50Hz. In this case, the ear perceives an additional set of tones that are equal to both the sum and the difference between the

two original tones ... as well as being equal to the sum and difference between their harmonics. The simple formulas for computing the fundamental tones are:

$$\text{sum tone} = f_1 + f_2$$

$$\text{difference tone} = f_1 - f_2$$

Difference tones can be easily heard when they are below the frequency of both tones' fundamentals. For example, the combination of 2000 and 2500Hz produces a difference tone of 500Hz.

12. MASKING

Masking is the phenomenon by which loud signals prevent the ear from hearing softer sounds. The greatest masking effect occurs when the frequency of the sound and the frequency of the masking noise are close to each other. For example, a 4-kHz tone will mask a softer 3.5-kHz tone but has little effect on the audibility of a quiet 1000-Hz tone. Masking can also be caused by harmonics of the masking tone (e.g., a 1-kHz tone with a strong 2-kHz harmonic might mask a 1900-Hz tone). This phenomenon is one of the main reasons why stereo placement and equalization are so important to the mixdown process. An instrument that sounds fine by itself can be completely hidden or changed in character by louder instruments that have a similar timbre. Equalization, mic choice or mic placement might have to be altered to make the instruments sound different enough to overcome any masking effect.

13 PERCEPTION OF SOUND WITH DIRECTION

Although one ear can't discern the direction of a sound's origin, two ears can. This capability of two ears to localize a sound source within an acoustic space is called spatial or binaural localization. This effect is the result of three acoustic cues that are received by the ears:

- Interaural intensity differences
- Interaural arrival-time differences
- The effects of the pinnae (outer ears).

Middle to higher frequency sounds originating from the right side will reach the right ear at a higher intensity level than the left ear, causing an interaural intensity difference. This volume difference occurs because the head casts an acoustic block or shadow, allowing only reflected

sounds from surrounding surfaces to reach the opposite ear (Figure 2.25). Because the reflected sound travels farther and loses energy at each reflection—in our example the intensity of sound perceived by the left ear will be greatly reduced, resulting in a signal that's perceived as originating from the right.

This effect is relatively insignificant at lower frequencies, where wavelengths are large compared to the head's diameter, allowing the wave to easily bend around its acoustic shadow. For this reason, a different method of localization (known as interaural arrival-time differences) is employed at lower frequencies (Figure 2.26). In both Figures 2.8 and 2.9, small time differences occur because the acoustic path length to the left ear is slightly longer than the path to the right ear. The sound pressure therefore arrives at the left ear at a later time than the right. This method of localization (in combination with interaural intensity differences) helps to give us lateral localization cues over the entire frequency spectrum.



Figure 2.8 The head casts an acoustic shadow that helps with localization at middle to upper frequencies.

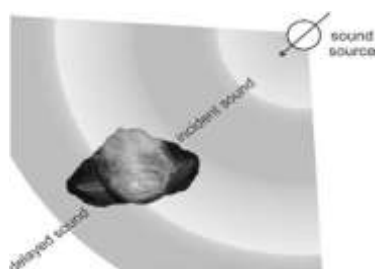


Figure 2.9 Interaural arrival-time differences occurring at lower frequencies.

Intensity and delay cues allow us to perceive the direction of a sound's origin but not whether the sound originates from the front, behind or below. The pinna (Figure 2.9), however, makes use of two ridges that reflect sound into the ear. These ridges introduce minute time delays between the direct sound (which reaches the entrance of the ear canal) and the sound that's reflected from the ridges (which varies according to source location). It's interesting to note that beyond 130° from the front of our face, the pinna is able to reflect and delay sounds by 0 and 80 microseconds

(μsec), making rear localization possible. Ridge 2 (see Figure 2.9) has been reported to produce delays of between 100 and 330 μsec that help us to locate sources in the vertical plane. The delayed reflections from both ridges are then combined with the direct sound to produce frequency–response colorations that are compared within the brain to determine source location. Small movements of the head can also provide additional position information.

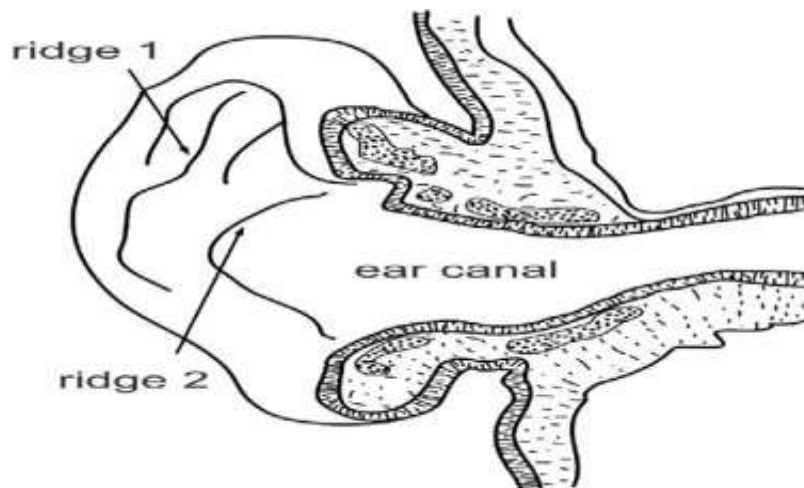


Figure 2.9 The pinna and its reflective ridges for determining vertical location information.

If there are no differences between what the left and right ears hear, the brain assumes that the source is the same distance from each ear. This phenomenon allows us to position sound not only in the left and right loud-speakers but also monophonically between them. If the same signal is fed to both loudspeakers, the brain perceives the sound identically in both ears and deduces that the source must be originating from directly in the center. By changing the proportion that's sent to each speaker, the engineer changes the relative interaural intensity differences and thus creates the illusion of physical positioning between the speakers. This placement technique is known as panning (Figure 2.10).

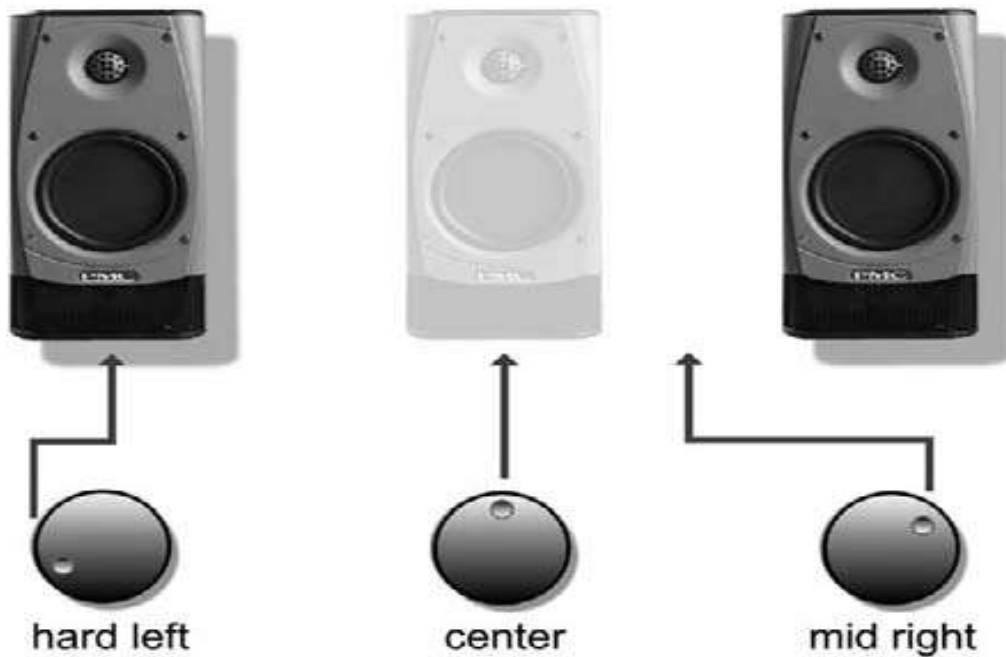


Figure 2.10 Pan pot settings and their relative spatial positions.

14. PERCEPTION OF SOUND WITH SPACE

In addition to perceiving the direction of sound, the ear and brain combine to help us perceive the size and physical characteristics of the acoustic space in which a sound occurs. When a sound is generated, a percentage reaches the listener directly, without encountering any obstacles. A larger portion, however, is propagated to the many surfaces of an acoustic enclosure. If these surfaces are reflective, the sound is bounced back into the room and toward the listener. If the surfaces are absorptive, less energy will be reflected back to the listener. Three types of reflections are commonly generated within an enclosed space (Figure 2.11)

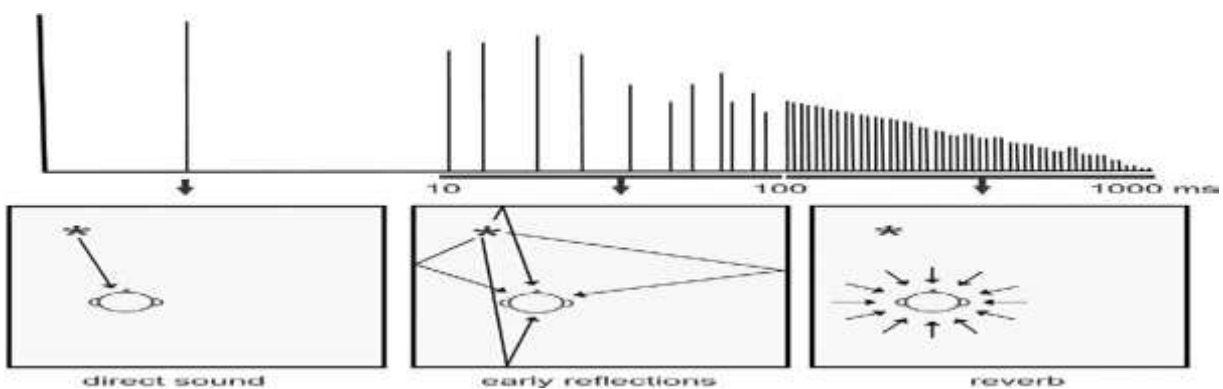


Figure 2.11 The three sound field types that are generated within an enclosed space.

- **Direct sound**
- **Early reflections**
- **Reverberation.**

15 DIRECT SOUND

In air, sound travels at a constant speed of about 1130 feet per second, so a wave that travels from the source to the listener will follow the shortest path and arrive at the listener's ear first. This is called the direct sound. Direct sounds determine our perception of a sound source's location and its size and conveys the true timbre of the source.

16. EARLY REFLECTIONS

Waves that bounce off of surrounding surfaces in a room must travel further than direct sound to reach the listener and therefore arrive after the direct sound and from a multitude of directions. These waves form what are called early reflections. Early reflections give us clues as to the reflectivity, size and general nature of an acoustic space. These sounds generally arrive at the ears less than 50msec after the brain perceives the direct sound and are the result of reflections off of the largest, most prominent boundaries within a room. The time elapsed between hearing the direct sound and the beginning of the early reflections helps to provide information about the size of the performance room. Basically, the farther the boundaries are from the source and listener, the longer the delay before it's reflected back to the listener.

Another aspect that occurs with early reflections is called temporal fusion. Early reflections arriving at the listener within 30msec of the direct sound are not only audibly suppressed, but are also fused with the direct sound. In effect, the ear can't distinguish the closely occurring reflections and considers them to be part of the direct sound. The 30-msec time limit for temporal fusion isn't absolute; rather, it depends on the sound's envelope. Fusion breaks down at 4msec for transient clicks, whereas it can extend beyond 80msec for slowly evolving sounds (such as a sustained organ note or legato violin passage). Despite the fact that the early reflections are suppressed and fused with the direct sound, they still modify our perception of the sound, making it both louder and fuller.

17. REVERBERATION

Whenever room reflections continue to bounce off of room boundaries, a randomly decaying set of sounds can often be heard after the source stops in the form of reverberation. A highly reflective surface absorbs less of the wave energy at each reflection and allows the sound to

persist longer after the initial sound stops (and vice versa). Sounds reaching the listener 50-msec later in time are perceived as a random and continuous stream of reflections that arrive from all directions. These densely spaced reflections gradually decrease in amplitude and add a sense of warmth and body to a sound. Because it has undergone multiple reflections, the timbre of the reverberation is often quite different from the direct sound (with the most notable difference being a roll-off of high frequencies and a slight bass emphasis).

The time it takes for a reverberant sound to decrease to 60 dB below its original level is called its decay time or reverb time and is determined by the room's absorption characteristics. The brain is able to perceive the reverb time and timbre of the reverberation and uses this information to form an opinion on the hardness or softness of the surrounding surfaces. The loudness of the perceived direct sound increases rapidly as the listener moves closer to the source, while the reverberation levels will often remain the same, because the diffusion is roughly constant throughout the room. This ratio of the direct sound's loudness to the reflected sound's level helps listeners judge their distance from the sound source.

Whenever artificial reverb and delay units are used, the engineer can generate the necessary cues to convince the brain that a sound was recorded in a huge, stone-walled cathedral—when, in fact, it was recorded in a small, absorptive room. To do this, the engineer programs the device to mix the original un-reverberated signal with the necessary early delays and random reflections. Adjusting the number and amount of delays on an effects processor gives the engineer control over all of the necessary parameters to determine the perceived room size, while decay time and frequency balance can help to determine the room's perceived surfaces. By changing the proportional mix of direct-to-processed sound, the engineer/producer can place the sound source at either the front or rear of the artificially created space.

Doubling

By repeating a signal using a short delay of 4 to 20msec (or so), the brain can be fooled into thinking that the apparent number of instruments being played is doubled. This process is called doubling. Often, acoustic doubling and tripling can be physically re-created during the overdub phase by recording a track and then going back and laying down one or more passes while the musicians listen to the original track. When this isn't possible, delay devices can be cost effectively and easily used to simulate this effect. If a longer delay is chosen (more than about 35msec), the repeat will be heard as discrete echoes, causing the delay (or series of repeated

delays) to create a slap echo or slap back. This and other effects can be used to double or thicken up a sound ... anybody want vocals that sound like a 1950s pop star?

18. FILTERS

Before diving straight in and talking about how equalizers behave, we'll start with the basics and look at four different types of filters. Just like a coffee filter keeps coffee grounds trapped while allowing coffee to flow through, an audio filter lets some frequencies pass through unaffected while reducing the level of others.

Low-pass Filter

One of the conceptually simplest filters is known as a *low-pass filter* because it allows low frequencies to pass through it. The question, of course, is "how low is low?" The answer lies in a single frequency known as the *cutoff frequency* or f_c . This is the frequency where the output of the filter is 3.01 dB lower than the maximum output for any frequency (although we normally round this off to -3 dB which is why it's usually called the *3 dB down point*). "What's so special about -3 dB?" I hear you cry. This particular number is chosen because -3 dB is the level where the signal is at one half the power of a signal at 0 dB. So, if the filter has no additional gain incorporated into it, then the cutoff frequency is the one where the output is exactly one half the power of the input. (Which explains why some people call it the *half-power point*.) As frequencies get higher and higher, they are attenuated more and more. This results in a *slope* in the frequency response graph which can be calculated by knowing the amount of extra attenuation for a given change in frequency. Typically, this slope is specified in decibels per octave. Since the higher we go, the more we attenuate in a low pass filter, this value will always be negative.

The slope of the filter is determined by its *order*. If we oversimplify just a little, a first-order low-pass filter will have a slope of -6.02 dB per octave above its cutoff frequency (usually rounded to -6 dB/oct). If we want to be technically correct about this, then we have to be a little more specific about where we finally reach this slope. Take a look at the frequency response plot in Figure 2.12. Notice that the graph has a nice gradual transition from a slope of 0 (a horizontal line) in the really low frequency region to a slope of -6 dB/oct in the really high frequencies. In the area around the cutoff frequency, however, the slope is changing. If we want to be really accurate, then we have to say that the slope of the frequency response is really 0 for frequencies less than one tenth of the cutoff frequency. In other words, for frequencies more than one decade below the cutoff frequency. Similarly, the

slope of the frequency response is really -6.02 dB/oct for frequencies more than one decade above the cutoff frequency.

If we have a higher-order filter, the cutoff frequency is still the one where the output drops by 3 dB, however the slope changes to a value of $-6.02n$ dB/oct, where n is the order of the filter. For example, if you have a 3rd-order filter, then the slope is

output drops by 3 dB, however the slope changes to a

$$\text{slope} = \text{order} * -6.02 \text{ dB/oct} \tag{2.4}$$

$$= 3 * -6.02 \text{ dB/oct} \tag{2.5}$$

$$= -18.06 \text{ dB/oct} \tag{2.6}$$

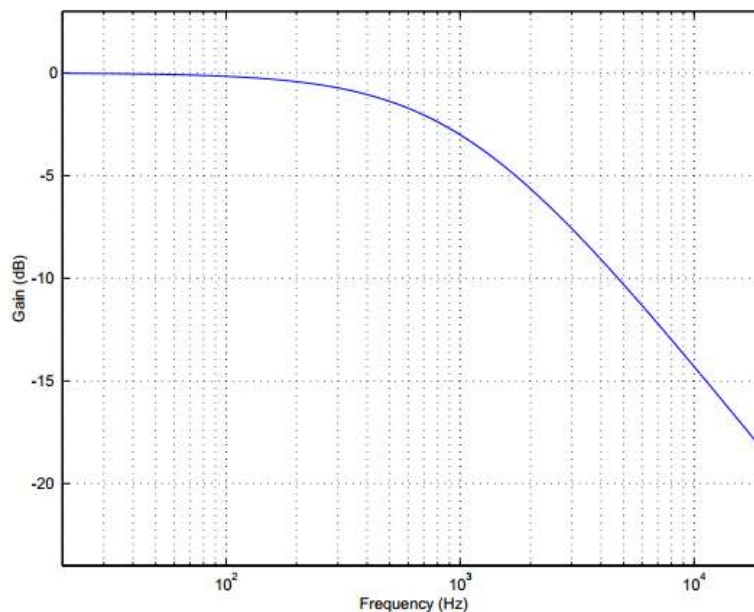


Figure 2.12: The frequency response of a first-order low-pass filter with a cutoff frequency of 1 kHz. Note that the cutoff frequency is where the response has dropped in level by 3 dB. The slope can be calculated by dividing the drop in level by the change in frequency that corresponds to that particular drop.

High-pass Filter

A high-pass filter is essentially exactly the same as a low-pass filter, however, it permits high frequencies to pass through while attenuating low frequencies as can be seen in Figure 2.13. Just like in the previous section, the cutoff frequency is where the output has a level of-

3.01 dB but now the slope below the cutoff frequency is positive because we get louder as we increase frequency. Just like the low-pass filter, the slope of the high-pass filter is dependent on the order of the filter and can be calculated using the equation $6.02n \text{ dB/oct}$, where n is the order of the filter.

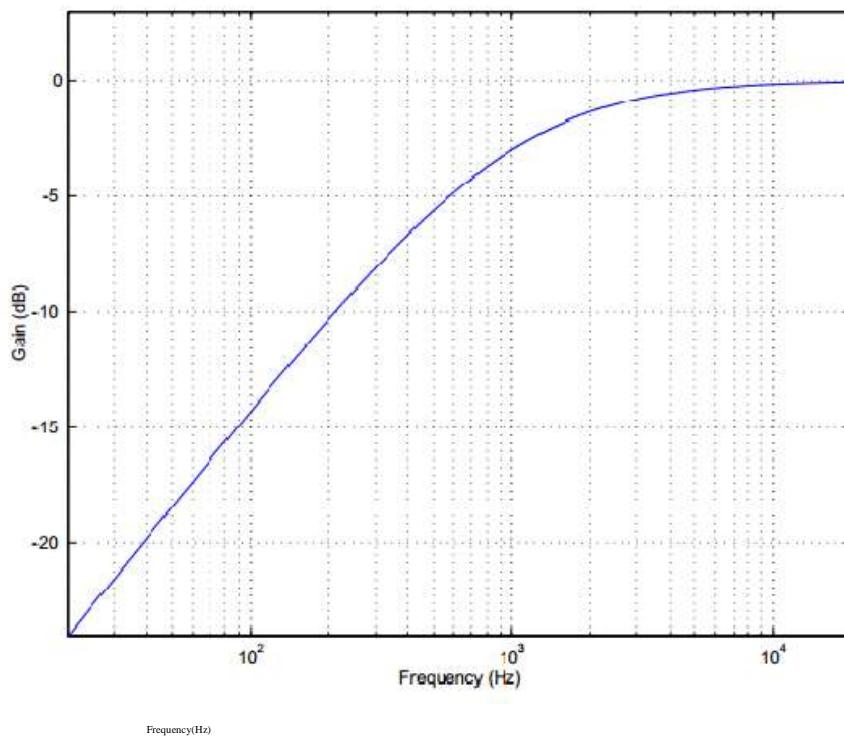


Figure 2.13: The frequency response of a first-order high-pass filter with a cutoff frequency of 1 kHz.

Remember as well that the slope only applies to frequencies that are at least one decade away from the cutoff frequency.

Band-pass Filter

Let's take a signal and send it through a high-pass filter and a low-pass filter in series, so the output of one feeds into the input of the other. Let's also assume for a moment that the two cutoff frequencies are more than a decade apart.

The result of this probably won't hold any surprises. The high-pass filter will attenuate the low frequencies, allowing the higher frequencies to pass through. The low-pass filter will attenuate the high frequencies, allowing the lower frequencies to pass through. The result is that at the high and low frequencies are attenuated, with a middle band (called the *passband*) that's allowed to pass relatively unaffected.

Bandwidth

This resulting system is called a *bandpass filter* and it has a couple of specifications that we should have a look at. The first is the width of the passband. This *bandwidth* is calculated using the difference between two cutoff frequencies which we'll label f_{c1} for the lower one and f_{c2} for the higher one. Consequently, the bandwidth is calculated using the equation:

$$BW = f_{c2} - f_{c1} \quad (2.7)$$

So, using the example of the filter frequency response shown in Figure 4, the bandwidth is $10,000\text{Hz} - 20\text{Hz} = 9980\text{Hz}$.

Centre Frequency

We can also calculate the middle of the passband using these two frequencies. It's not quite so simple as we'd like, however. Unfortunately, it's not just the frequency that's halfway between the low and high frequency cutoff's. This is because frequency specifications don't really correspond to the way we hear things. Humans don't usually talk about frequency — they talk about pitches and notes. They say things like “Middle C” instead of “262Hz.” They also say things like “one octave” or “one semitone” instead of things like “a bandwidth of 262Hz.”

Consider that, if we play the A below Middle C on a well-tuned piano, we'll hear a note with a fundamental of 220Hz. The octave above that is 440Hz and the octave above that is 880Hz. This means that the bandwidth of the first of these two octaves is 220Hz (it's $440\text{Hz} - 220\text{Hz}$), but the bandwidth of the second octave is 440Hz ($880\text{Hz} - 440\text{Hz}$). Despite the fact that they have different bandwidths, we hear them each as one octave, and we hear the 440Hz note as being halfway between the other two notes. So, how do we calculate this? We have to find what's known as the *geometric mean* of the two frequencies. This can be found using the equation

$$f_{\text{centre}} = \sqrt{f_{c1} f_{c2}} \quad (2.8)$$

Q

Let's say that you want to build a bandpass filter with a bandwidth of one octave. This isn't difficult if you know the centre frequency and if it's never going to change. For example, if the centre frequency was 440Hz, and

the bandwidth was one octave wide, then the cutoff frequencies would be 311 Hz and 622 Hz (we won't worry too much about how I arrived at these numbers). What happens if we leave the bandwidth the same at 311 Hz, but change the centre frequency to 880 Hz? The result is that the bandwidth is now no longer an octave wide—it's one half of an octave. So, we have to link the bandwidth with the centre frequency so that we can describe it in terms of a fixed musical interval. This is done using what is known as the

quality or Q of the filter, calculated using the equation:

$$Q = F_{\text{center}} / \text{BW}$$

Now, instead of talking about the bandwidth of the filter, we can use the Q which gives us an idea of the width of the filter in musical terms. This is because, as we increase the centre frequency, we have to increase the bandwidth proportionately to maintain the same Q . Notice however, that if we maintain a centre frequency, the smaller the bandwidth gets, the bigger the Q becomes, so if you're used to talking in terms of musical intervals, you have to think backwards. A big Q is a smaller interval as can be seen in the plot of a number of different Q 's in Figure 2.14.

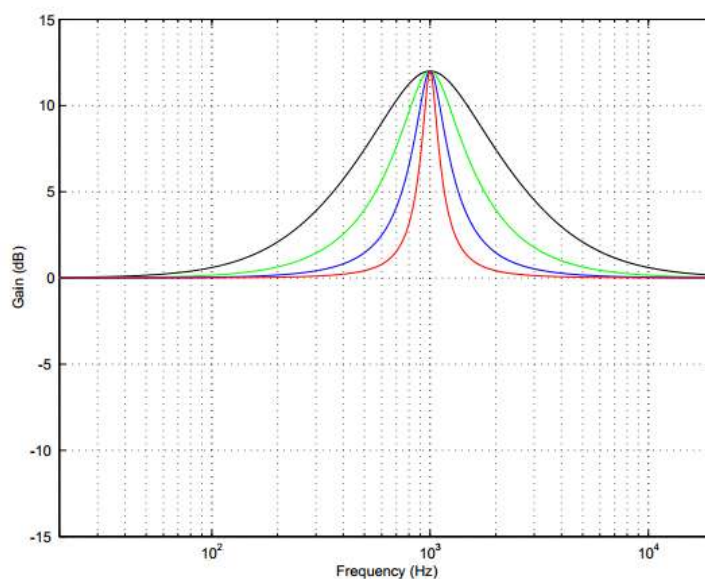


Figure 2.14: The frequency responses of various bandpass filters with different Q 's and a matched centre frequency of 1 kHz.

Notice in Figure

2.14 that you can have a very high Q , and therefore a very narrow bandwidth for a bandpass filter. All of the definitions still hold, however. The cutoff frequencies are still the points where we're 3 dB lower than the maximum value and the bandwidth is still the distance in Hertz between these two points and soon...

Band-reject Filter

Although bandpass filters are very useful at accentuating a small band of frequencies while attenuating others, sometimes we want to do the opposite.

We want to attenuate a small band of frequencies while leaving the rest alone. This can be accomplished using a *band-reject*

filter (also known as a *bandstop filter*) which, as its name implies, rejects (or usually just attenuates) a band of frequencies without affecting the surrounding material. As can be seen in Figure

2.15, this winds up looking very similar to a bandpass filter drawn upside down.

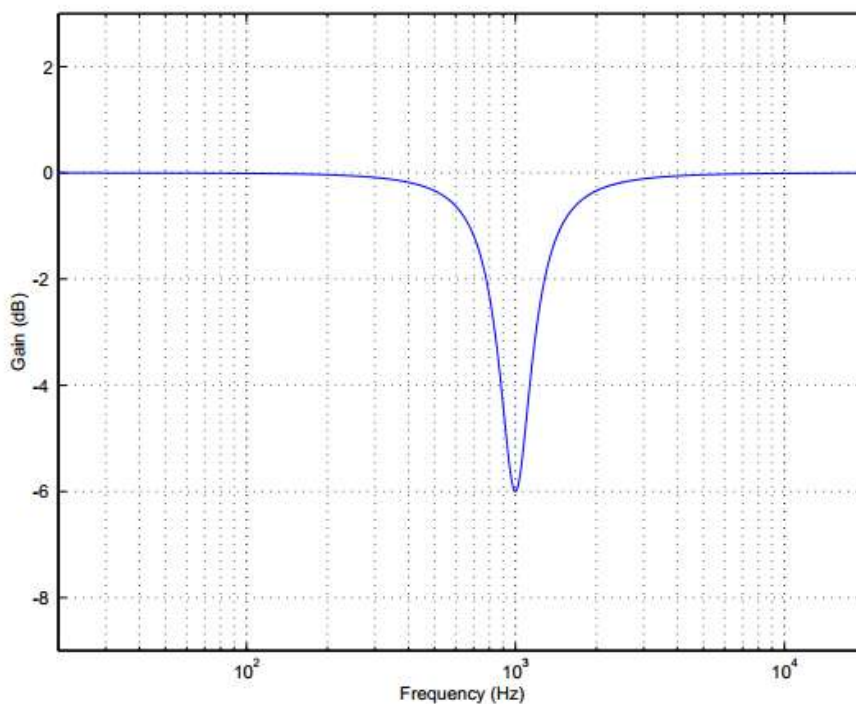


Figure 2.15: The frequency response of a band-reject filter with a center frequency of 1 kHz.

The thing to be careful of when describing band-reject filters is the fact that cutoff frequencies are still defined as the points where we've dropped in level by 3 dB. Therefore, we don't really get an intuitive idea of how much we drop at the center frequency. Looking at Figure 2.15 we can see that, although the band-reject filter looks just like the bandpass filter upside down, the bandwidth is quite different. This is a fairly important point to remember a little later on in this section on symmetry.

Equalizers

Unlike its counterpart from the days of long-

distance phone calls, a modern equalizer is a device that is capable of attenuating and boosting frequencies according

to the desire and expertise of the user. There are four basic types of equalizers, but we'll have to talk about a couple of issues before getting into the nitty-gritty.

An equalizer typically consists of a collection of filters, each of which permits you to control one or more of three things: the *gain*, *centre*

frequency and *Q* of the filter. There are some minor differences in these filters from the ones we discussed above, but we'll sort that out before moving on.

Also, the filters in the equalizer may be connected in *parallel* or in *series*, depending on the type of equalizer and the manufacturer.

To begin with, as we'll see, a

filter in an equalizer comes in three basic models, the *bandpass*, and the *bandreject*, which are typically chosen by the user by manipulating the gain of the filter. On a decibel scale,

positive gain results in a bandpass, whereas negative gain produces a bandreject. In addition, there is the *shelving filter* which is a variation on the highpass and lowpass filters.

The principal difference between filters in an equalizer and the filters defined in Section 1 is that, in a typical equalizer, instead of attenuating all frequencies outside the passband, the filter typically leaves them at a gain of 0 dB. An example of this can be seen in the plot of an equalizer's bandpass filter in Figure

2.16. Notice now that, rather than attenuating all

unwanted frequencies, the filter is applying a known gain to the passband. The further away

you get from the passband, the less the signal is affected. Notice, however, that we still measure the bandwidth using the two points that are 3 dB down from the peak of the curve.

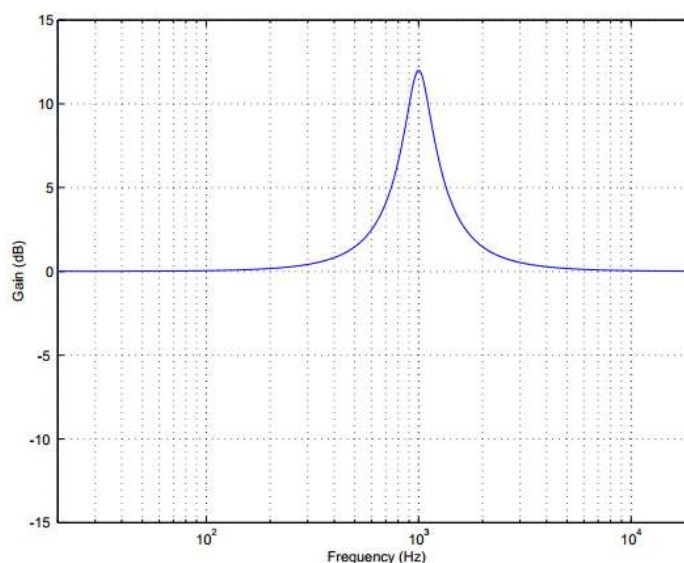


Figure
2.16: The frequency response of a bandpass filter with a centre frequency of 1 kHz, a *Q* of 4, and a gain of 12 dB in a typical equalizer.

Graphic Equalizer

Graphic equalizers are seen just about everywhere these days, primarily because they're intuitive to use. In fact, they are probably the most-used piece of signal processing equipment in recording. The name "graphic equalizer" comes from the fact that the device is made up of a number of filters with center frequencies that are regularly spaced, each with a slider used for gain control. The result is that the arrangement of the sliders gives a graphic representation of the frequency response of the equalizer. The most common frequency resolutions available are one-octave, two-third-octave and one-third-octave, although resolutions as fine as one-twelfth-octave exist. The sliders on most graphic equalizers use

ISO standardized band center frequencies. They virtually always employ reciprocal peak/dip filters wired in parallel. As a result, when two adjacent bands are boosted, there remains a comparatively large dip between the two peaks. This proves to be a great disadvantage when attempting to boost a frequency between two center frequencies. Drastically excessive amounts of boost may be required at the band centers in order to properly adjust the desired frequency. This problem is eliminated in graphic EQ's using the much-less-common combining filters. In this system, the filter banks are wired in series, thus adjacent bands have a cumulative effect. Consequently, in order to boost a frequency between two center frequencies, the given filters need only be boosted a minimal amount to result in a higher-boosted mid-frequency.

Virtually all graphic equalizers have fixed frequencies and a fixed Q. This makes them simple to use and quick to adjust, however they are generally a compromise. Although quite suitable for general purposes, in situations where a specific frequency or bandwidth adjustment is required, they will prove to be inaccurate.

Paraphraphic Equalizer

One attempt to overcome the limitations of the graphic equalizer is the *paraphraphic equalizer*. This is a graphic equalizer with fine frequency adjustment on each slider. This gives the user the ability to sweep the center frequency of each filter somewhat, thus giving greater control over the frequency response of the system.

19. LOUDNESS AND NOISE REDUCTION

Loudness

Although we rarely like to admit it, we humans aren't perfect. This is true in many respects, but for the purposes of this discussion, we'll concentr

atespecificallyonourabilitiestohearthings.Unfortunately,ourearsdon'thavethesamefrequencyresponseatalllisteninglevels.Atveryhighlisteninglevels,wehavea relativelyflat frequencyresponse, butasthe leveldrops,sodoesoursensitivitytohighandlowfrequencies.Asaresult,ifyoumixatune at a very highlistening level and thenreduce the level,it willappear tolacklowendandhighend.Similarly,ifyoumixatalowlevelandturnitup,you'lltendtogetmorelowendandhighend.

Onepossibleuseforanequalizeristocompensatefortheperceivedlackof information in extreme frequency ranges at low listening levels.Essentially, when you turndown themonitorlevels,youcanuseanequalizertoincreasethellevelsofthelowandhighfrequencycontent to compensatefordeficienciesinthehumanhearing mechanism.Thisfilteringisidentical tothatwhichisengagedwhenyoupressthe“loudness”buttononmosthomestereosystems.Ofcourse, the dangerwithsuchequalizationisthatyoudon'tknowwhatfrequencyrangestoalter,andhowmuchtoalter them—soitisnotrecommendabletodosuchcompensationwhenyou're mixing,onlywhenyou'reathomelistingtosomethingthat'salreadybeenmixed.

NoiseReduction

It'spossibleinsomespecificcasestouseequalizationtoreducenoiseinrecordings,butyouhavetobeawareofthedamagethatyou'reinflictingonsomeotherpartsofthesignal.

High-frequencyNoise(Hiss)

Let'ssaythatyou'vegotarecordingofanelectricbassonareallynoisy analogtapedeck.Since mostoftheperceivablenoiseisgoingtobehigh-frequencystuffandsincemostofthesignalthatyou'reinterestedinisgoingtobelow-frequencystuff,allyouneedtodoistorolloffthehighendtoreducethenoise.Ofcourse,thisis thebestofallpossibleworlds.It'smorelikelythatyou'regoingtobecopingwithasignalthat has somehigh-frequencycontent(likeyourleadvocals,forexample...)soifyoustartrollingoffthehighend tooomuch,youstartlosingalotofbrightnessandsparklefromyoursignal,possiblymakingthe endresultworsehatyoustarted.Ifyou'reusingequalizationtoreducenoiselevels,don't forgettooccasionallyhitthe“bypass”switchoftheequalizeronceandawhiletoheartheoriginal.Youmayfindwhenyourefreshyourmemorythatyou'vegonealittletoofarinyourattempt to makethingsbetter.

Low-frequency Noise (Rumble)

Almost every console in the world has a little button every input strip that has a symbol that looks like a little ramp with the slope on the left. This is a high-pass filter that is typically a second-order filter with a cutoff frequency around 100 Hz or so, depending on the manufacturer and the year it was built. The reason that filter is there is to help the recording or sound reinforcement engineer get rid of low-frequency noise like “stagerumble” or microphone handling noise. In actual fact, this filter won't eliminate all of your problems, but it will certainly reduce them. Remember that most signals don't go below 100 Hz (this is about an octave and a half below middle C on a piano) so you probably don't need everything that comes from the microphone in this frequency range—in fact, chances are, unless you're recording pipe organ, electric bass or space shuttle launches, you won't need nearly as much as you think below 100 Hz.

Hummmmmmm...

There are many reasons, forgivable and unforgivable, why you may wind up with an unwanted hum in your recording. Perhaps you work with a poorly-installed system. Perhaps your recording took place under a buzzing street lamp. Whatever the reason, you get a single frequency (and perhaps a number of its harmonics) singing all the way through your recording. The nice thing about this situation is that, most of the time, the hum is at a predictable frequency (depending on where you live, it's likely a multiple of either 50 Hz or 60 Hz) and that frequency never changes. Therefore, in order to reduce, or even eliminate this hum, you need a very narrow band-reject filter with a lot of attenuation. Just the sort of job for a notch filter. The drawback is that you also attenuate any of the music that happen to be at or very near the notch center frequency, so you may have to reach a compromise between eliminating the hum and having too detrimental of an effect on your signal.

Self Assessment Questions

- Q 1 Draw a sinusoidal waveform and highlight graphic representation of the values p_0 (the stasis pressure), p (instantaneous amplitude) , p (instantaneous pressure), and P (maximum peak pressure).
- Q2 Define the following
- A damping with the help of an example
 - B Transverse wave
 - C Longitudinal wave
 - D Torsional wave
- Q3 what are the three major types of interaction effects explain them briefly?
- Q4 Explain the behavior of ear due to the perception of sound with direction?
- Q5 Describe filter and types of filters?
- Q6 briefly describe the following
- A Hiss
 - B Rumble
 - C Hmmmmm

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Web links for short videos on different topics of the unit.

http://www.tonmeister.ca/main/textbook/intro_to_sound_recording20.html#fn1x5

<https://www.youtube.com/watch?v=oqBHBO8cqLI>

https://www.youtube.com/watch?v=qxDvW8_fm7I

<https://www.youtube.com/watch?v=pqJzn8Y1HFw>

<https://www.youtube.com/watch?v=oyTdcQI9oN4>

<https://www.youtube.com/watch?v=5-3rzTwk00I>

<https://www.youtube.com/watch?v=elcDChdqpUQ>

Unit 3

Microphone Basics

Written by: Umer Mehmood

Reviewer: Muhammad Awais Khan

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Learning Outcomes

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- 3. Mic Level & Line Level**
- 4. Phantom Power**
- 5. Microphone Polar Patterns**
- 6. Microphone Impedance**
- 7. Microphone Frequency Response**
- 8. Typical Placement**
- 9. Looking After Your Microphones**

Self Assessment Questions

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INTRODUCTION

Dear students a good engineer will have made hundreds of recordings using dozens of different microphones. Each session is an opportunity to make a new discovery. The engineer will make careful notes of the setup, and will listen to the results many times to build an association between the technique used and the sound achieved.

David Edward Hughes invented a carbon microphone in the 1870s. The first microphone that enabled proper voice telephony was the loose-contact carbon microphone.

Dear students the microphones are of different types. In this unit we will learn about how different microphones convert the sound signal into an electrical signal. This will help the students in choosing microphone according to the recording requirement.

Dear Students the microphone generates an electrical current which is very small sometimes before using the signal its needs to be amplified. Students the microphone's directionality or polar pattern indicates how sensitive it is to sounds arriving at different angles. In this unit the students will also learn the practice of microphone placement and how to achieve better results for different environments.

Dear Students microphones every microphone have a frequency response which refers to the way a microphone responds to different frequencies. For example, a frequency response which favors low frequencies means that the resulting audio output will sound more bass than the original sound.

In this unit students will explore the microphone and will learn the professional use of the microphone.

OBJECTIVES

After completion of this lesson student will:

- Be able to identify the different types of microphones.
- Be able to discuss issues regarding microphones.
- Learn polar pattern indicates how sensitive it is to sounds arriving
- Select a microphone that effectively matches the characteristics of an instrument.
- Position a microphone in a way that accommodates an instrument's complex sound radiation patterns.
- Devise multiple microphone configurations in order to attain more accurate or compelling recordings.
- Learn about microphone placement to achieve better results for different environments
- Work with close as well as distant microphone configurations.
- Utilize stereo microphones to convey the spatial qualities of instruments and ensembles.
- Execute a moderately complex recording session with multiple simultaneous performers.

1. MICROPHONE

A microphone is an electronic device that converts sound into an electrical signal. Electromagnetic transducers facilitate the conversion of acoustic signals into electrical signals. Sound information exists as patterns of air pressure; the microphone changes this information into patterns of electric current. The recording engineer is always interested in the accuracy of this transformation. Microphones are used in many applications such as public address systems, motion picture production, live and recorded audio engineering, two-way radios, megaphones, radio and television broadcasting.

Most microphones today use electromagnetic induction (dynamic microphones), capacitance change (condenser microphones) or piezoelectricity (piezoelectric microphones) to produce an electrical signal from air pressure variations. Microphones typically need to be connected to a preamplifier before the signal can be amplified with an audio power amplifier and a speaker or recorded.



Electronic Symbol

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 aluminum) which vibrates when it is struck by sound waves. In a typical hand-held microphone like the one below, the diaphragm is located in the head of the microphone.



Location of Microphone Diaphragm

When the diaphragm vibrates, it causes other components in the microphone to vibrate. These vibrations are converted into an electrical current which becomes the audio signal.

2. TYPES OF MICROPHONE

There are a number of different types of microphone in common use. The differences can be divided into two areas:

a. The type of conversion technology they use

This refers to the technical method the microphone uses to convert sound into electricity. The most common technologies are dynamic, condenser, ribbon and crystal. Each has advantages and disadvantages, and each is generally more suited to certain types of application.

b. The type of application they are designed for

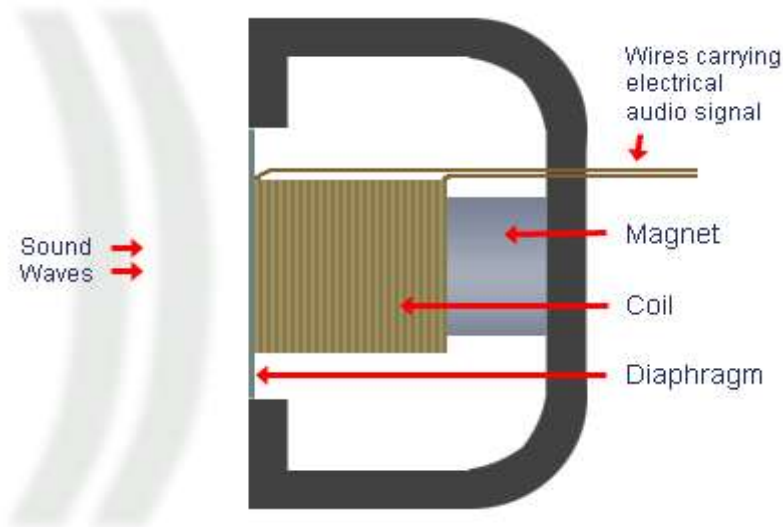
Some microphones are designed for general use and can be used effectively in many different situations. Others are very specialized and are only really useful for their intended purpose.

2.1 Dynamic Microphones

Dynamic microphones are versatile and ideal for general-purpose use. They use a simple design with few moving parts. They are relatively strong and flexible to rough handling. They are also better suited to handling high volume levels, such as from certain musical instruments or amplifiers. They have no internal amplifier and do not require batteries or external power.

When a magnet is moved near a coil of wire an electrical current is generated in the wire. Using this electromagnet principle, the dynamic microphone uses a wire coil and magnet to create the audio signal.

The diaphragm is attached to the coil. When the diaphragm vibrates in response to incoming sound waves, the coil moves backwards and forwards past the magnet. This creates a current in the coil which is channeled from the microphone along wires. A common configuration is shown below.



Cross-Section of Dynamic Microphone

Earlier we mentioned that loudspeakers perform the opposite function of microphones by converting electrical energy into sound waves. This is demonstrated perfectly in the dynamic microphone which is basically a loudspeaker in reverse. When you see a cross-section of a speaker you'll see the similarity with the diagram above. Dynamics do not usually have the same flat frequency response as condensers. Instead they tend to have tailored frequency responses for particular applications.

Neodymium magnets are more powerful than conventional magnets, meaning that neodymium microphones can be made smaller, with more linear frequency response and higher output level.

Advantages

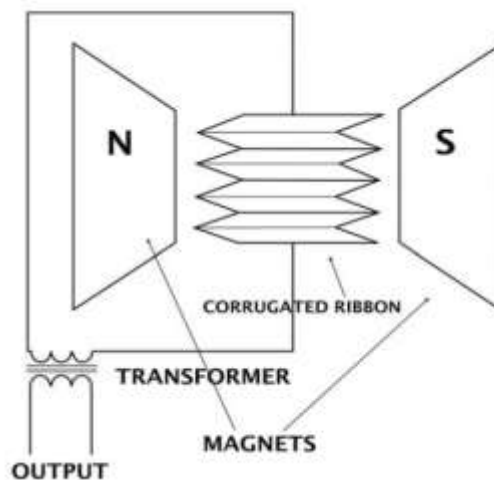
Robust and durable, can be relatively inexpensive, insensitive to changes in humidity, need no external or internal power to operate, can be made fairly small.

Disadvantages

Resonant peak in the frequency response, typically weak high frequency response beyond 10kHz.

The ribbon microphone

The ribbon microphone operates almost the same as the moving coil microphone. The major difference is that the transducer is a strip of extremely thin aluminum foil wide enough and light enough to be vibrated directly by the moving molecules of air of the sound wave, so no separate diaphragm is necessary. However, the electrical signal generated is very small compared to a moving coil microphone, so an output transformer is needed to boost the signal to a usable level.



Block diagram of ribbon microphone

Like the dynamic microphone, the high frequency response is governed by the mass of the moving parts. But because the diaphragm is also the transducer, the mass is usually a lot less than a dynamic type. As a result, the upper frequency response tends to reach slightly higher, to around 14kHz. The frequency response is also generally flatter than for a moving coil microphone.

All good studio ribbon mics provide more opportunity to EQ to taste since they “take” EQ well. Ribbon mics have their resonance peak at the bottom of their frequency range, which means that a ribbon just doesn’t add any extra high frequency hype like condenser mics do.

Advantages

Relatively flat frequency response, extended high frequency response as compared to dynamics, needs no external or internal power to operate.

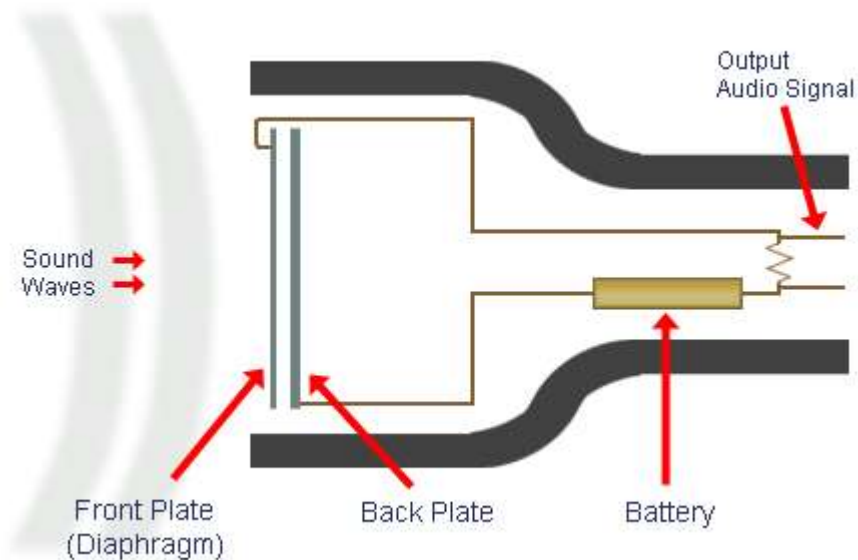
Disadvantages

Fragile—requires care during operation and handling, moderately expensive.

2.3 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.

Condenser microphones require power from a battery or external source. The resulting audio signal is stronger signal than that from a dynamic. Condensers also tend to be more sensitive and responsive than dynamics, making them well-suited to capturing faint nuances in a sound. They are not ideal for high-volume work, as their sensitivity makes them flat to distort.



Cross-Section of a Typical Condenser Microphone

A capacitor has two plates with a voltage between them. In the condenser microphone, 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 microphone or by external phantom power.

Advantages Excellent high frequency and upper harmonic response can have excellent low frequency response.

Disadvantages Moderate to very expensive, requires external powering, can be relatively bulky; low cost (and some expensive) models can suffer from poor or inconsistent frequency response, two mics of the same model may sound quite different, humidity and temperature affect performance.

2.4 The Electret Condenser Microphone

The electret condenser microphone uses a special type of capacitor which has a permanent voltage built in during manufacture. This is somewhat like a permanent magnet, in that it doesn't require any external power for operation. However, good electret condenser microphones usually include a pre-amplifier which does still require power.

Other than this difference, you can think of an electret condenser microphone as being the same as a normal condenser.

- Condenser microphones have a flatter frequency response than dynamics.
- A condenser microphone works in much the same way as an electrostatic tweeter (although obviously in reverse).

3. MIC LEVEL & LINE LEVEL

The electrical current generated by a microphone is very small. Referred to as mic level, this signal is typically measured in millivolts. Before it can be used for anything serious the signal needs to be amplified, usually to line level (typically 0.5 -2V). Being a stronger and more robust signal, line level is the standard signal strength used by audio processing equipment and common domestic equipment such as CD players, tape machines, VCRs, etc.

This amplification is achieved in one or more of the following ways:

- Some microphones have tiny built-in amplifiers which boost the signal to a high mic level or line level.
- The microphone can be fed through a small boosting amplifier, often called a line amp.

- Sound mixers have small amplifiers in each channel. Attenuators can accommodate microphones of varying levels and adjust them all to an even line level.
- The audio signal is fed to a power amplifier, a specialized amp which boosts the signal enough to be fed to loudspeakers.

4. PHANTOM POWER

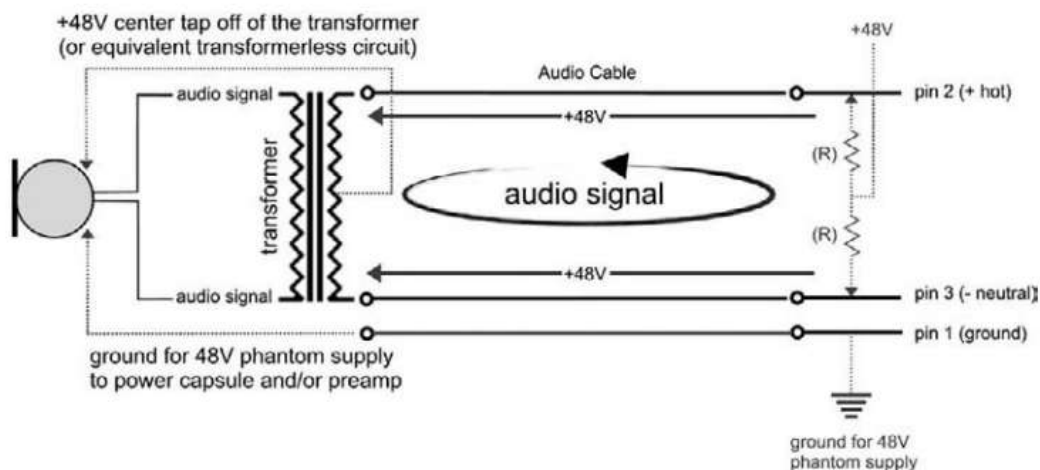
Phantom power is a means of distributing a DC current through audio cables to provide power for microphones and other equipment.

The supplied voltage is usually between 12 and 48 Volts, with 48V being the most common. Individual microphones draw as much current from this voltage as they need.

A balanced audio signal connected to a 3 pin XLR has the audio signal traveling on the two wires – usually connected to pin 2 (+ve) and pin 3 (-ve). Pin 1 is connected to the shield, which is earthed. The audio signal is an AC (alternating current), whereas phantom power is DC (direct current).

The DC phantom power is transmitted simultaneously on both pin 2 and 3, with the shield (pin 1) being the ground. Since the DC voltage on the ‘hot’ and ‘cold’ pins (2 & 3) is identical, it is seen by equipment as “common mode” noise and rejected, or ignored, by the equipment.

If you put a volt meter on pins 1 & 2, or pins 1 & 3, you will see the 48v DC phantom power, but if you meter pins 2 & 3 (the audio carrying wires) you will see no voltage.



Schematic diagram of phantom power system

The DC voltage can be harnessed however, and used to power microphones, mic-line amps, or indeed a video camera (in this case the DC voltage would travel up the video cable – and would need special equipment to filter this voltage).

In summary, audio signals transmit as AC current, whereas powered equipment requires DC current to operate. Phantom power is a clever way of using one cable to transmit both currents.

Generate Phantom Power

Phantom power can be generated from sound equipment such as mixing consoles and preamplifiers. Special phantom power supplies are also available.

Phantom Power effect on Audio

Despite occasional reports of damage or unwanted audio disturbance, it is generally accepted that phantom power does not affect the quality of audio and is quite safe to use. However it is recommended that you do not supply phantom power to microphones which do not require it, especially ribbon microphones.

Note:

- Most earth-lift switches will disable phantom power.
- High-impedance microphones, or microphones with unbalanced outputs, are not compatible with phantom power.
- Some non-standard consumer components use a feature they call phantom power, but is not true phantom power. These devices may cause damage when connected to a true phantom-powered device.

5. MICROPHONE POLAR PATTERNS

A microphone's directionality or polar pattern indicates how sensitive it is to sounds arriving at different angles about its central axis. Some microphone designs combine several principles in creating the desired polar pattern.

Directional Properties

Every microphone has a property known as directionality. This describes the microphone's sensitivity to sound from various directions. Some microphones pick up sound equally from all directions; others pick up sound only from one direction or a particular combination of directions. The types of directionality are divided into three main categories:

1. **Omnidirectional**

Picks up sound evenly from all directions (omni means "all" or "every").

2. **Unidirectional**

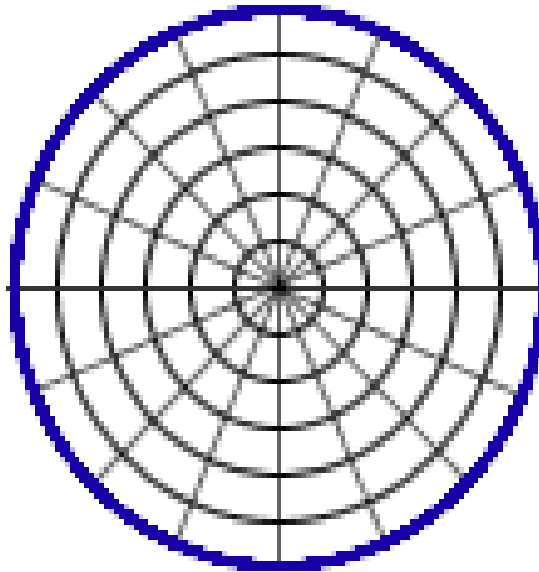
Picks up sound predominantly from one direction. This includes cardioid and hyper cardioid microphones.

3. **Bidirectional**

Picks up sound from two opposite directions.

To help understand the directional properties of a particular microphone, user manuals often include a graphical representation of the microphone's directionality. This graph is called a polar pattern.

Omnidirectional



Omnidirectional pattern

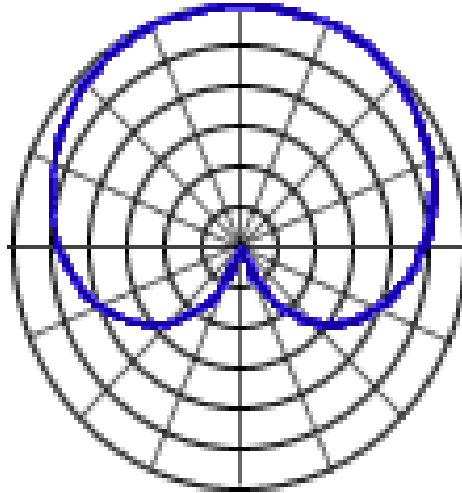
An omnidirectional microphone response is generally considered to be a perfect sphere in three dimensions. As with directional microphones, the polar pattern for an "omnidirectional" microphone is a function of frequency. The body of the microphone is not infinitely small and, as a consequence, it tends to get in its own way with respect to sounds arriving from the rear, causing a slight flattening of the polar response. This flattening increases as the diameter of the microphone reaches the wavelength of the frequency. Therefore, the smallest diameter microphone gives the best omnidirectional characteristics at high frequencies.

Captures sound equally from all directions. It is used for capturing ambient noise; Situations where sound is coming from many directions; Situations where the microphone position must remain fixed while the sound source is moving.

Notes:

- Although omnidirectional microphones are very useful in the right situation, picking up sound from every direction is not usually what you need. Omni sound is very general and unfocused - if you are trying to capture sound from a particular subject or area it is likely to be overwhelmed by other noise.

Cardioid



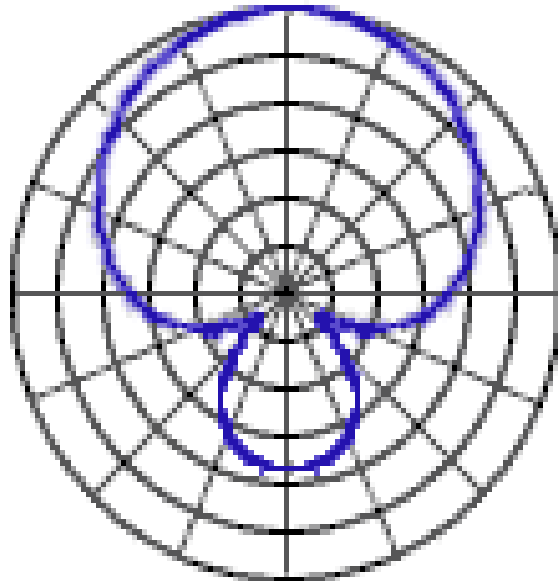
Cardioid pattern

Cardioid means "heart-shaped", which is the type of pick-up pattern these microphone use. A cardioid pickup pattern is **a highly flexible pickup pattern that is great for all-purpose use**. Cardioid microphones come in all shapes and sizes. A cardioid microphone is slightly directional. Cardioid microphones will still pickup background noise if they are not in a controlled environment. Sound is picked up mostly from the front, but to a lesser extent the sides as well. It is used for emphasizing sound from the direction the microphone is pointed even as leaving some latitude for microphone movement and ambient noise.

Notes:

- The cardioid is a very versatile microphone, ideal for general use.
- Handheld microphones are usually cardioid.
- There are many variations of the cardioid pattern (such as the hypercardioid).

Hypercardioid



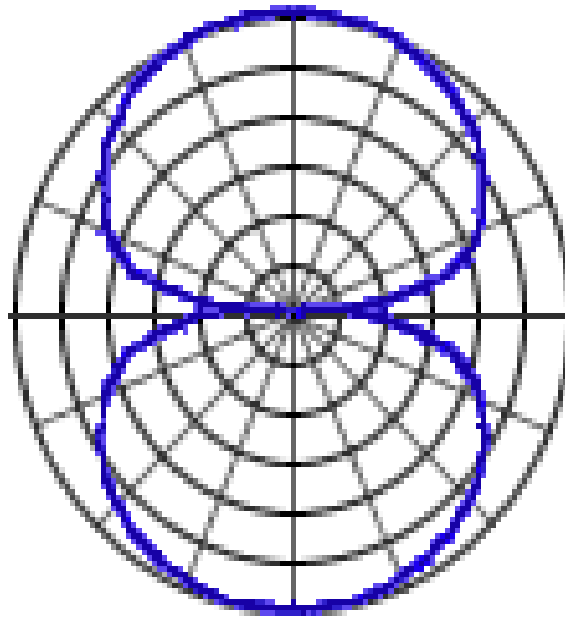
Hypercardioid pattern

It is very directional and eliminates most sound from the sides and rear. Due to the long thin design of hypercardioids, they are often referred to as shotgun microphones. It is used for isolating the sound from a subject or direction when there is a lot of ambient noise.

Notes:

- By removing all the ambient noise, unidirectional sound can sometimes be a little unnatural..
- You need to be careful to keep the sound consistent. If the microphone doesnot stay pointed at the subject you will lose the audio.
- Shotguns can have an area of increased sensitivity directly to the rear.

Bidirectional



Bidirectional pattern

A bidirectional microphone is designed to **pick-up audio equally from the front and back of the microphone**. Typically, bidirectional microphones are used for radio interview recording. Uses a figure-of-eight pattern and picks up sound equally from two opposite directions. It is used as you can imagine, there aren't a lot of situations which require this polar pattern. One possibility would be an interview with two people facing each other (with the microphone between them).

6. MICROPHONE IMPEDANCE

When dealing with microphones, one consideration which is often misunderstood or overlooked is the microphone's impedance rating factor. In order to ensure the best quality and most reliable audio, attention should be given in getting this factor right.

Impedance

Impedance is an electronics term which measures the amount of opposition a device has to an AC current. Technically speaking, it is the combined effect of capacitance, inductance, and resistance on a signal. The letter "Z" is often used as shorthand for the word impedance, for example, Hi-Z or Low-Z.

Impedance is measured in ohms, shown with the Greek Omega symbol " Ω ". A microphone with the specification 600Ω has an impedance of 600 ohms.

Microphone Impedance

All microphones have a specification referring to their impedance. This specification may be written on the microphone itself, or you may need to consult the manual.

You will often find that microphones with a hard-wired cable and 1/4" plug are high impedance, and microphones with separate balanced audio cable and XLR connector are low impedance.

There are three general classifications for microphone impedance. Different manufacturers use slightly different guidelines but the classifications are roughly:

1. Low Impedance (less than 600Ω)
2. Medium Impedance (600Ω - $10,000\Omega$)
3. High Impedance (greater than $10,000\Omega$)

Impedance to Choose

High impedance microphones are usually quite cheap. Their main disadvantage is that they do not perform well over long distance cables - after about 5 or 10 meters they begin producing poor quality audio. In any case these microphones are not a good choice for serious work. In fact, although not completely reliable, one of the clues to a microphone's overall quality is the impedance rating.

Low impedance microphones are usually the preferred choice.

Matching Impedance with Other Equipment

Microphones are not the only things with impedance. Other equipment, such as the input of a sound mixer, also has an ohms rating. Again, you may need to consult the appropriate manual or website to find these values. Be aware that what one system calls "low impedance" may not be the same as your low impedance microphone - you really need to see the ohms value to know exactly what you're dealing with.

A low impedance microphone should generally be connected to an input with the same or higher impedance. If a microphone is connected to an input with lower impedance, there will

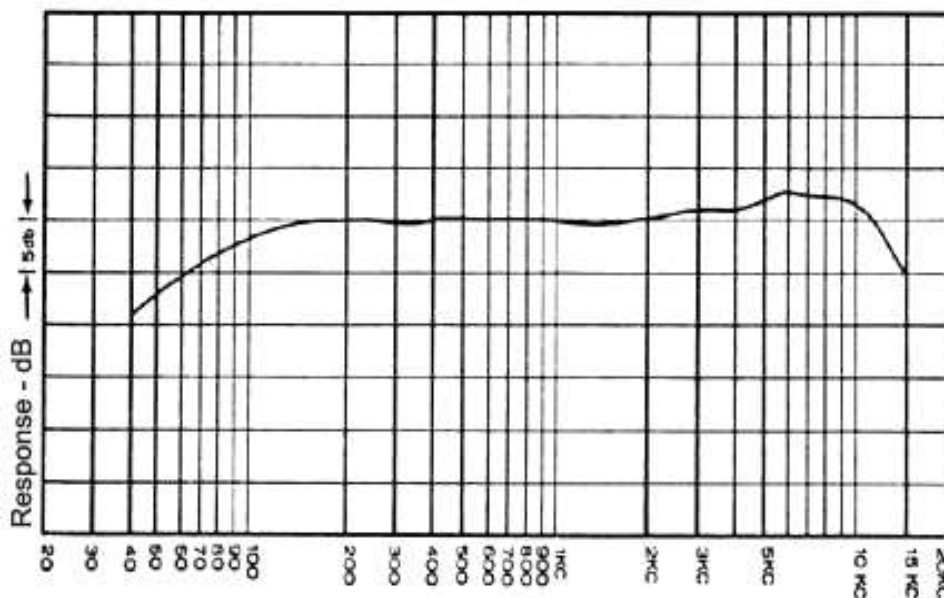
be a loss of signal strength. In some cases you can use a line matching transformer, which will convert a signal to different impedance for matching to other components.

7. MICROPHONE FREQUENCY RESPONSE

Frequency response refers to the way a microphone responds to different frequencies. It is a characteristic of all microphones that some frequencies are exaggerated and others are attenuated (reduced). For example, a frequency response which favors high frequencies means that the resulting audio output will sound more trebly than the original sound.

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 - Cycles per Second

Response Curve

An ideal "flat" frequency response means that the microphone is equally sensitive to all frequencies. In this case, no frequencies would be exaggerated or reduced (the chart above would show a flat line), resulting in a more accurate representation of the original sound. We therefore say that a flat frequency response produces the purest audio.

In the real world a perfectly flat response is not possible and even the best "flat response" microphones have some deviation.

More importantly, it should be noted that a flat frequency response is not always the most desirable option. In many cases a tailored frequency response is more useful. For example, a response pattern designed to emphasize the frequencies in a human voice would be well suited to picking up speech in an environment with lots of low-frequency background noise.

The main thing is to avoid response patterns which emphasize the wrong frequencies. For example, a vocal microphone is a poor choice for picking up the low frequencies of a bass drum.

Frequency Response Ranges

You will often see frequency response quoted as a range between two figures. This is a simple way to see which frequencies a microphone is capable of capturing effectively. For example, a microphone which is said to have a frequency response of 20 Hz to 20 kHz can reproduce all frequencies within this range. Frequencies outside this range will be reproduced to a much lesser extent or not at all.

This specification makes no mention of the response curve, or how successfully the various frequencies will be reproduced. Like many specifications, it should be taken as a guide only.

Condenser vs. Dynamic

Condenser microphones generally have flatter frequency responses than dynamic. All other things being equal, this would usually mean that a condenser is more desirable if accurate sound is a prime consideration.

8. TYPICAL PLACEMENT

8.1 Single microphone

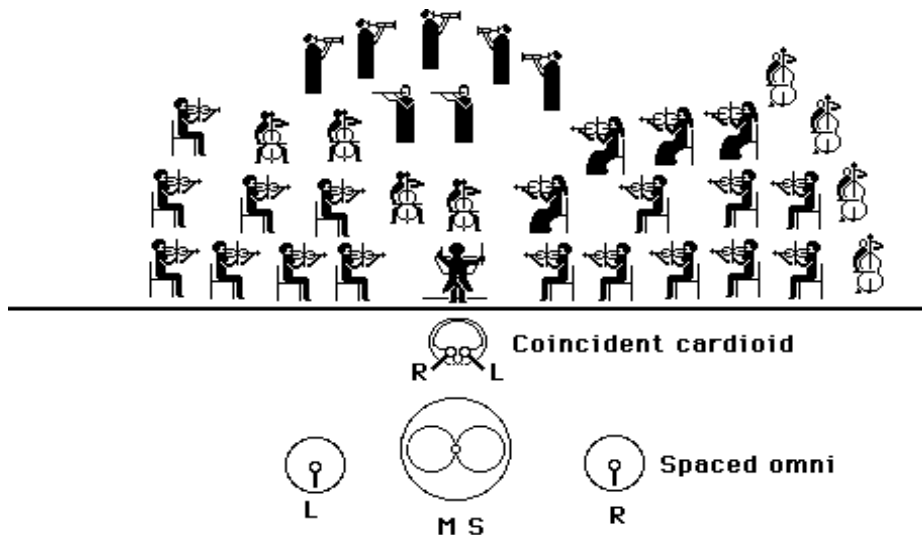
Use of a single microphone is pretty straightforward. Having chosen one with appropriate sensitivity and pattern, (and the best distortion, frequency response, and noise characteristics), you simply mount it where the sounds are. The practical range of distance between the instrument and the microphone is determined by the point where the sound overloads the microphone or console at the near end, and the point where ambient noise becomes objectionable at the far end. Between those extremes it is largely a matter of taste and experimentation.

If you place the microphone close to the instrument, and listen to the results, you will find the location of the microphone affects the way the instrument sounds on the recording. The timbre may be odd, or some notes may be louder than others. That is because the various components of an instrument's sound often come from different parts of the instrument body, and we are used to hearing an evenly blended tone. A close in microphone will respond to some locations on the instrument more than others because the difference in distance from each to the microphone is proportionally large. A good rule of thumb is that the blend zone starts at a distance of about twice the length of the instrument. If you are recording several instruments, the distance between the players must be treated the same way.

If you place the microphone far away from the instrument, it will sound as if it is far away from the instrument. We judge sonic distance by the ratio of the strength of the direct sound from the instrument to the strength of the reverberation from the walls of the room. When we are physically present at a concert, we use many cues beside the sounds to keep our attention focused on the performance, and we are able to ignore any distractions there may be. When we listen to a recording, we don't have those visual clues to what is happening, and find anything extraneous that is very audible annoying. Some engineers prefer to use close miking techniques to keep noise down and add artificial reverberation to the recording, others solve the problem by mounting the microphone very high, away from audience noise but where adequate reverberation can be found.

8.2 Stereo

Stereo sound is an illusion of spaciousness produced by playing a recording back through two speakers. The success of this illusion is referred to as the image. A good image is one in which each instrument is a natural size, has a distinct location within the sound space, and does not move around. The main factors that establish the image are the relative strength of an instrument's sound in each speaker, and the timing of arrival of the sounds at the listener's ear. In a studio recording, the stereo image is produced artificially. Each instrument has its own microphone, and the various signals are balanced in the console as the producer desires. In a concert recording, where the point is to document reality, and where individual microphones would be awkward at best, it is most common to use two microphones, one for each speaker.



Microphone placement for stereo recording

8.3 Spaced microphones

The simplest approach is to assume that the speakers will be eight to ten feet apart, and place two microphones eight to ten feet apart to match. Either omnis or cardioids will work. When played back, the results will be satisfactory with most speaker arrangements. The big disadvantage of this technique is that the microphones must be rather far back from the ensemble—at least as far as the distance from the leftmost performer to the rightmost. Otherwise, those instruments closest to the microphones will be too prominent. There is

usually not enough room between stage and audience to achieve this with a large ensemble, unless you can suspend the microphones or have two very tall stands.

8.4. Coincident cardioids

There is another disadvantage to the spaced technique that appears if the two channels are ever mixed together into a monophonic signal or broadcast over the radio. Because there is a large distance between the microphones, it is quite possible that sound from a particular instrument would reach each microphone at slightly different times (sound takes 1 millisecond to travel a foot). This effect creates phase differences between the two channels, which results in severe frequency response problems when the signals are combined. You seldom actually lose notes from this interference, but the result is an uneven, almost shimmering sound. The various coincident techniques avoid this problem by mounting both microphones in almost the same spot.

This is most often done with two cardioid microphones, one pointing slightly left, one slightly right. The microphones are often pointing toward each other, as this places the diaphragms within a couple of inches of each other, totally eliminating phase problems. No matter how they are mounted, the microphone that points to the left provides the left channel. The angle between the microphones is critical, depending on the actual pickup pattern of the microphone. If the microphones are too parallel, there will be little stereo effect. If the angle is too wide, instruments in the middle of the stage will sound weak, producing a hole in the middle of the image.

You may place the microphones fairly close to the instruments when you use this technique. The problem of balance between near and far instruments is solved by aiming the microphones toward the back row of the ensemble; the front instruments are therefore off axis and record at a lower level. You will notice that the height of the microphones becomes a critical adjustment.

8.5 M.S. or Middle side technique

The most elegant approach to coincident miking is the M.S. or middle-side technique. This is usually done with a stereo microphone in which one element is omnidirectional, and the other bidirectional. The bidirectional element is oriented with the axis running parallel to the

stage, rejecting sound from the center. The omni element, of course, picks up everything. To understand the next part, consider what happens as instrument is moved on the stage. If the instrument is on the left half of the stage, a sound would first move the diaphragm of the bidirectional microphone to the right, causing a positive voltage at the output. If the instrument is moved to center stage, the microphone will not produce any signal at all. If the instrument is moved to the right side, the sound would first move the diaphragm to the left, producing a negative voltage. The instruments on one side of the stage are 180 degrees out of phase with those on the other side, and the closer they are to the center, the weaker the signal produced.

Now the signals from the two microphones are not merely kept in two channels and played back over individual speakers. The signals are combined in a circuit that has two outputs; for the left channel output, the bidirectional output is added to the omni signal. For the right channel output, the bidirectional output is subtracted from the omni signal. This gives stereo, because an instrument on the right produces a negative signal in the bidirectional microphone, which when added to the omni signal, tends to remove that instrument, but when subtracted, increases the strength of the instrument. An instrument on the left suffers the opposite fate, but instruments in the center are not affected, because their sound does not turn up in the bidirectional signal at all.

M.S. produces a very smooth and accurate image, and is entirely mono compatible. The only reason it is not used more extensively is the cost of the special microphone and decoding circuit.

8.6 Large ensembles

The above techniques work well for concert recordings in good halls with small ensembles. When recording large groups in difficult places, you will often see a combination of spaced and coincident pairs. This does produce a kind of chorusing when the signals are mixed, but it is an attractive effect and not very different from the sound of string or choral ensembles any way. When balance between large sections and soloists cannot be achieved with the basic setup, extra microphones are added to highlight the weaker instruments. A very common problem with large halls is that the reverberation from the back seems late when

compared to the direct sound taken at the edge of the stage. This can be helped by placing a microphone at the rear of the audience area to get the ambient sound into the recording.

8.7 Studio techniques

These are just a few things you might see if you dropped in on the middle of a session.

Individual microphones on each instrument

This provides the engineer with the ability to adjust the balance of the instruments at the console, or, with a multi-track recorder. There may be eight or nine microphones on the drum set alone.

Close microphone placement

The microphones will usually be placed rather close to the instruments. This is partially to avoid problems that occur when an instrument is picked up in two non-coincident microphones, and partially to modify the sound of the instruments.

Acoustic fences around instruments, or instruments in separate rooms

The interference that occurs when an instrument is picked up by two microphones that are mixed is a very serious problem. You will often see extreme measures, such as a bass drum stuffed with blankets to muffle the sound, and then electronically processed to make it sound like a drum again.

Everyone wearing headphones

Studio musicians often play to "click tracks", which are not recorded metronomes, but someone tapping the beat with sticks and occasionally counting through tempo changes. This is done when the music must be synchronized to a film or video, but is often required when the performer cannot hear the other musicians because of the isolation measures described above.

20 or 30 takes on one song.

Recordings require a level of perfection in intonation and rhythm that is much higher than that acceptable in concert. The finished product is usually a composite of several takes.

Pop filters in front of microphones

Some microphones are very sensitive to minor gusts of wind-so sensitive in fact that they will produce a loud pop if you breathe on them. To protect these microphones engineers will often mount a nylon screen between the microphone and the artist. This is not the most common reason for using pop filters but vocalists like to move around when they sing; in particular, they will lean into microphones. If the singer is very close to the microphone, any motion will produce drastic changes in level and sound quality. Many engineers use pop filters to keep the artist at the proper distance. The performer may move slightly in relation to the screen, but that is a small proportion of the distance to the microphone.

9. LOOKING AFTER YOUR MICROPHONES

- Obey the normal common-sense rules of electronic equipment care, e.g. avoid very high temperatures, dust, dampness, high humidity, physical shocks, etc.
- Many performers think it's cool to swing the microphone by its lead and generally throw it around the place. Unless you own the microphone and you can afford to replace it regularly, don't do this.
- Don't blow into the microphone. The diaphragm is designed to respond to sound waves, not wind.
- Don't tap the head of the microphone. This can damage the microphone and/or speakers.
- If applicable, turn microphone off when not in use. Remove and replace batteries regularly. The action of removing and inserting batteries can help keep the contacts clean.
- Don't subject microphones to volume levels greater than their design capabilities.
- Always be careful with phantom power. Although it will not generally harm your microphone, it's prudent to play it safe.
- Keep all leads safely secured. If someone trips over a lead there may be all sorts of problems from damaged microphones to lawsuits.

- If the performance of a microphone deteriorates over time, it may be possible to have the diaphragm cleaned. You will need to talk to the supplier or manufacturer for details.

Self Assessment Questions

1. Define different microphone types?
2. Explain different studio techniques for using a microphone?
3. How to use a microphone by middle-side technique?
4. Describe the maintenance of a microphone?
5. Explain the placement of single microphone?
6. Define cardioid and hypercardioid pattern?

Suggested Readings/Reference

- http://www.shure.eu/support_download/videos/how-to/microphone-basics
- <http://www.mediacollege.com/audio/microphones/how-microphones-work.html>
- http://artsites.ucsc.edu/ems/music/tech_background/te-20/teces_20.html
- <https://soundtech.wordpress.com/education/the-fundamentals-of-audio/microphone-basics/>
- <http://en.wikipedia.org/wiki/Microphone>
- <https://www.youtube.com/watch?v=27dLLnisH2M>
- <https://www.youtube.com/watch?v=jGj27JkDcpc>
- <https://www.youtube.com/watch?v=PQWeQHy6HV0&list=PLnLOmVwRMCqQmQaiT58c94oKm8A7w5ZT0&index=6>
- <https://www.youtube.com/watch?v=1Uf79xgcKdI>

UNIT 4

Recording Environment

Written By: Syyed Muhammad Saadullah Shah
Reviewer : Muhammad Awais Khan

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- 2. Control Room**
- 3. Project Studio**
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- 7. Recording Process**
- 8. Preparation**
- 9. Recording**
- 10. Overdubbing**
- 11. Mix down**
- 12. Mastering**
- 13. Sequence Editing**

Self Assessment Questions

References

INTRODUCTION

There is a range of ways to record audio – from the smart phone in your pocket (perhaps) to digital recorders, microphones and laptops, to fully kit out recording studios. Basically, the equipment you use should be appropriate to what you want the audio for – and appropriate to how you're going to be doing the recording (e.g. using a smart phone may be the easiest way of recording whilst 'on the move').

However, there are also a range of challenges in recording audio, which may mean you don't get the result you want. Remember: If you have a bad audio recording to begin with, you won't be able to 'make it good' in the edit. You should consider from the beginning what kind of equipment you'll need, and anticipate some of the audio recording challenges you'll be likely to face.

To start with the quality audio recording/editing you required a professional studio with the sound lock environment to achieve the quality productions.

Control room has a vital role for any kind of audio broadcasting. All broadcasting and recording studios are attached with the control room to ensure the studios activity and control and monitoring the broadcasted signals.

Most of the portable studios designed professionally for the different locations only for the recording/editing the audio productions.

Outdoor audio recording is a salient feature of audio productions. Live on location recording purposely cover the live events or to record the impressions/interviews at the location.

Professional peoples behind any audio project are the important and like the backbone of any audio productions.

Recording process is an essential segment of any audio production. Engineers/Producers make necessary arrangements before recordings to ensure the broadcasted quality.

Another audio production segment is the audio dubbing/editing, sequence editing, mix down the project and finally mastering. It is much important to finish any audio recording project before it goes on air.

OBJECTIVES

After reading this unit the student will be able to:

1. Learning after the recording environment in a digital environment in detail will provide the students a practical professional knowledge for their professional carrier.
2. This course leads the maximum practical skills for those professional students who made their future in the field of audio broadcasting.
3. This is digital era and the audio productions completely deal with the digital technology as the audio productions are the computer aided productions.
4. After the completion of this task the students will be able to handle any type of digital audio recordings for their best performance in the field of audio productions.

1 RECORDING STUDIO

How to Make a Cheap Recording Studio

As computer technology has developed, more and more performance is possible on a lower and lower budget. As a result, building a simple home recording studio around your existing computer can be quite inexpensive. Learning how to make a cheap recording studio at home requires an assessment of exactly what you'll be using the studio for and what quality of sound you need. The guide below outlines what to look for in each piece of equipment.

Steps



Step 1

Purchase a computer. If you don't already have a computer to use in your recording setup, you will need to purchase one. Important considerations are processing speed and amount of memory, as recording software tends to use your computer's resources heavily. Both Windows and Mac

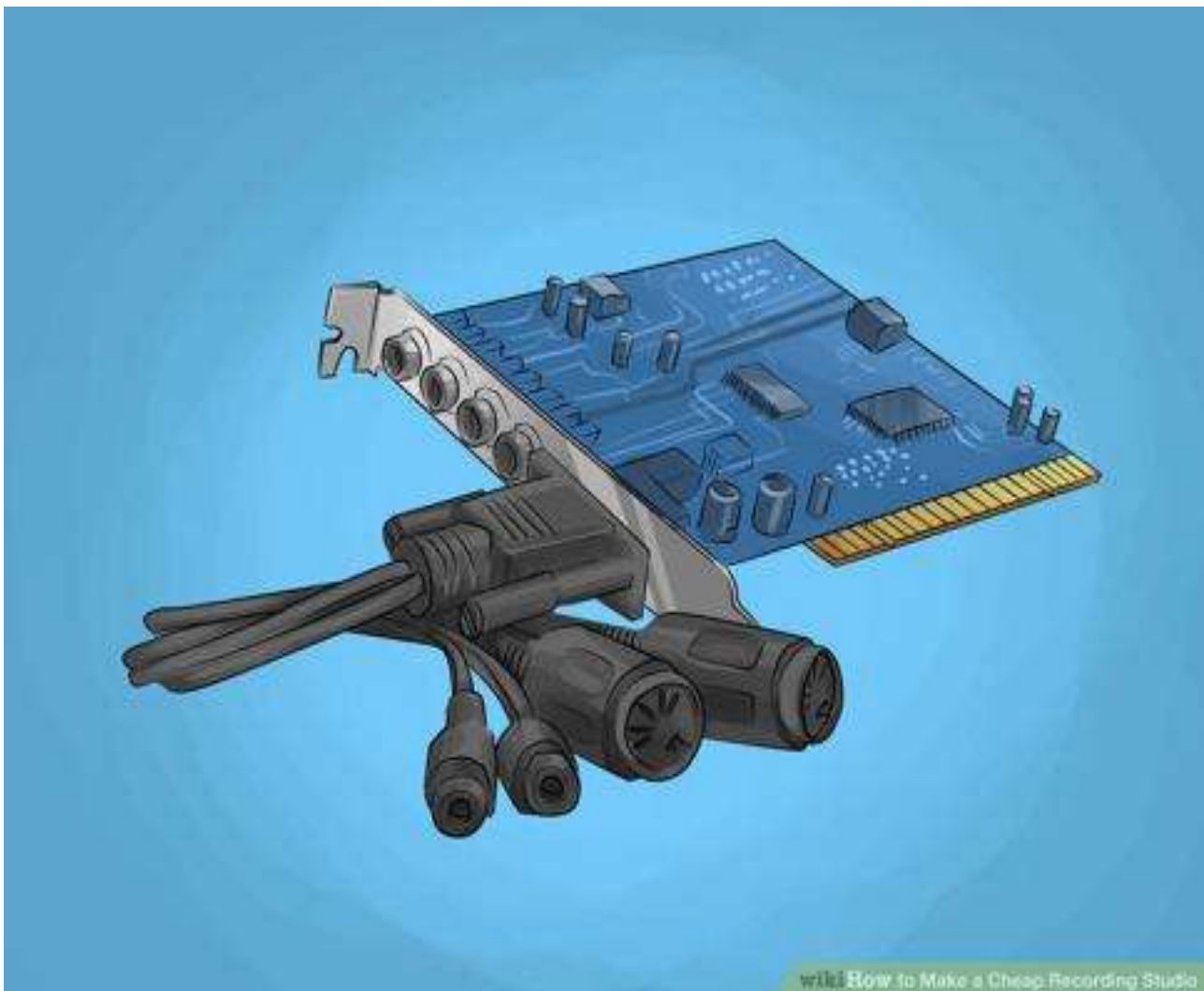
platforms will work well; however, Windows machines typically allow for easier upgrading of the sound card. Factory installed sound cards are not usually robust enough to produce high-quality recordings, so upgrading is a good idea

Step 2

Choose a piece of recording software. The recording software provides the interface through which you will manage your recordings on your computer. There are several options for small budgets. Generally, the more expensive applications offer greater functionality and flexibility.

For recording on a very small budget, you can use recording software licensed as freeware or shareware. Audacity and Garage Band are 2 popular choices for low budget recording.

With a slightly higher budget, you can purchase near professional quality recording software such as Ableton Live or Cakewalk Sonar. Both of these applications are also available in entry level versions that are less expensive but less powerful.



Step 3

Purchase and install an audio interface. An audio interface is a piece of hardware that replaces your computer's sound card and allows you to connect your instruments and microphones to your computer through a mixer. On a PC, you will usually install your audio interface in an empty PCI slot. On a Mac, you may need to purchase an interface that can be connected through a USB or FireWire cable.

At the least, make sure your audio interface has 2 input and 2 output jacks. This will allow you to record in stereo. For more flexibility, choose an interface with 4 input jacks.

One of the top manufacturers of audio interfaces for home use is M-Audio. They produce both entry level and high end models.

Step 4

Buy an audio mixer. A mixer is an essential piece of equipment for any home recording studio. The mixer handles all your inputs (such as microphones, guitars, and keyboards), allows you to adjust each input's settings, and routes the output to your audio interface and into your computer.

The basic functions on an inexpensive mixer will usually be adequate for home recording needs.

At the least, make sure each channel on your mixer includes adjustments for panning, volume, and 3-band equalization. Four channels will be more than adequate for home recording.

Popular brands for entry-level mixers are Behringer, Alesis, and Yamaha.



Step 5

Chose studio monitors and headphones for your studio. The speakers you use to listen to your mix during editing are called studio monitors (sometimes referred to as reference speakers). Studio monitors differ from other speakers in that they are meant to deliver a perfectly flat frequency response. This means that you are hearing your recording exactly as it exists digitally, without any frequency adjustment.

When choosing studio monitors, make sure to look for "near-field" models. These are de-signed to be listened to from about a yard (1 m) away, and so eliminate any effects due to the acoustics of your room.

Studio monitors can be purchased used from online classifieds sites or audio retailers. The robust, simple construction of loudspeakers makes them an ideal component to buy used and save money. In addition to or in place of monitors, you can buy a set of headphones. Headphones provide the advantage of being cheaper, smaller, and less likely to disturb a neighbor or housemate. Headphones can be used in conjunction with studio monitors to assess very low-volume components of your recordings.

Step 6

Decide on a microphone(s) to use in your studio. An inexpensive home recording studio can be managed with only a single microphone if necessary.

If you only buy 1 mic, make sure to choose a dynamic mic. This type of construction is more robust and versatile, and is self-powered. An industry standard dynamic mic is the Shure SM-57, which can be used for vocals and instruments.



If you need to record very quiet or expressive instruments, such as an acoustic guitar or piano, a condenser mic will provide better results. Condenser mics aren't as rugged or versatile as dynamic mics, but provide more sensitive response. A cheap recording studio can readily make do with 1 dynamic and 1 condenser microphone.



Answered Questions

Is a mixer necessary?

If you are working with more than one instrument, it is necessary since you want to be able to change the tone of each instrument so it does not out-sound another. If you are planning to use one instrument, such as a guitar, you might need a guitar mixer, which take cares of all the sounds and tones of a guitar. There are different mixers for different instruments, including voice. So, if you are using more than one, then definitely consider buying one. If you are using only one instrument, you could still get one but it's not necessary.

Will a headset with a mic work?

Yes, but headsets with mics generally don't have very good quality. You'll get better sound if you use a real microphone and separate headphones.

How can I make a soundproof device?

You can use a sound cancellation material, like star foam or the package your eggs come in.

I am new in this field. What instruments do I need?

The most essential instrument has got to be the Midi keyboard. If you have any money to spare, you can also get a drum kit.

Tips

Building your recording studio inexpensively often means building from what you already own. Using existing components such as microphones and computers, even when they are not ideally suited to the task at hand, will keep your budget low.

- Additional equipment may be needed depending on your recording needs. If you are interested in using the "soft-synth" instruments including with your recording software, for instance, you will need a MIDI interface and keyboard.
- If you do not have any recording equipment, you can get the following to set up a reasonably cheap yet efficient set up:
 - Apple Mac Mini
 - 2.3GHz Quad-Core Intel Core i7 (Turbo Boost up to 3.3GHz) with 6MB L3 cache
 - 1TB (5400-rpm) hard drive2
 - Intel HD Graphics 4000
 - 4GB (two 2GB) of 1600MHz DDR3 memory
 - M Audio Studiophile AV 30
 - Focusrite Scarlett 2i2 USB 2.0 Audio Interface
 - Samson C01 Large Diaphragm Condenser
 - Samson RH300/Samson SR850/ Audio Technica ATH M30 or JVC Harx 700 Reference Headphones

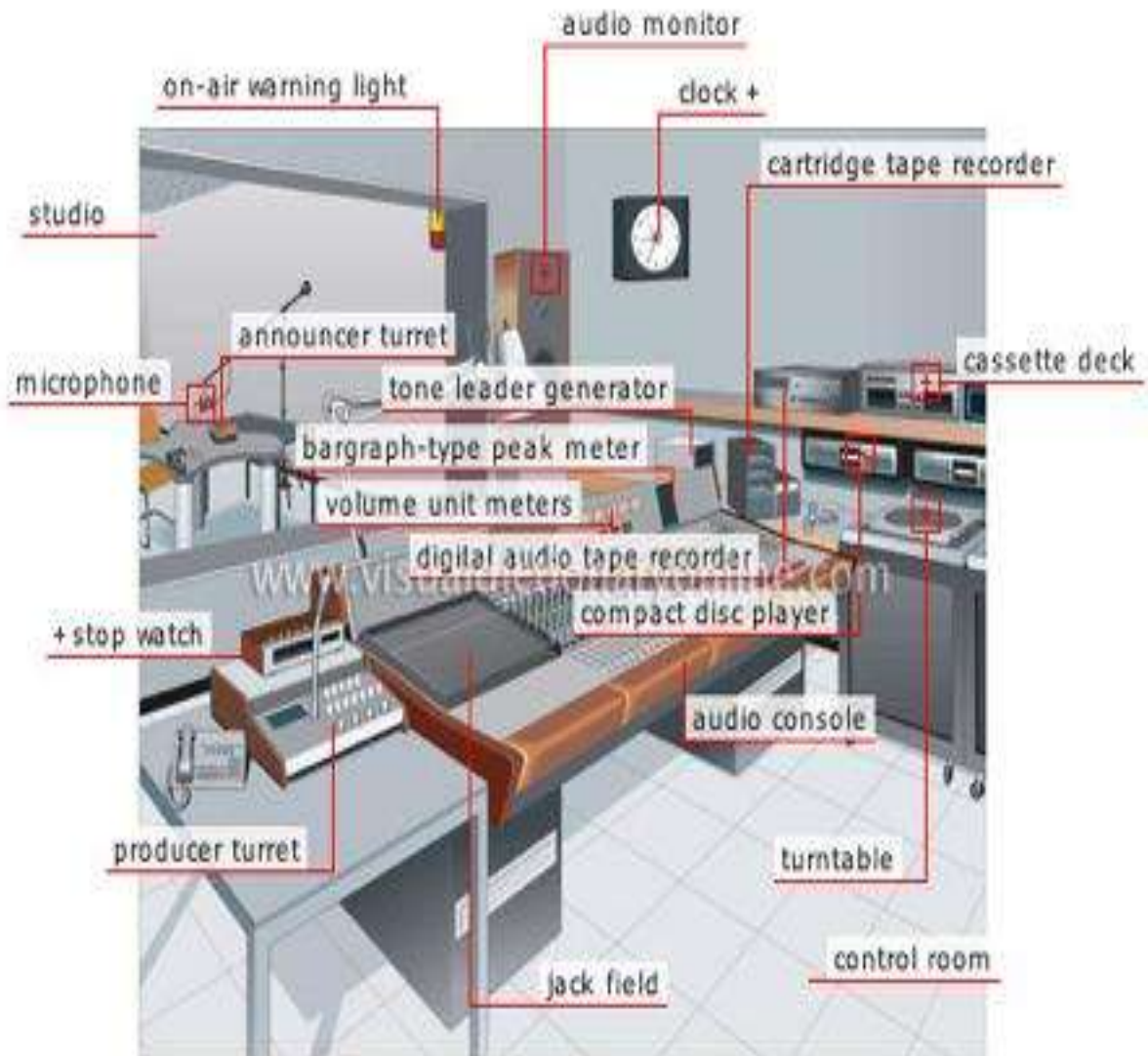
Things You'll Need

- Computer
- Recording software
- Audio interface
- Audio mixer
- Studio monitors
- Headphones
- Microphones
- MIDI keyboards

2 CONTROL ROOM

Radio studio and control room

Area with two rooms separated by a glass window where audio programs are produced, recorded or broadcast.



Control room

Room adjacent to the studio that is equipped with sound control and recording equipment; the director monitors the on-air program from here.

Bargraph type peak meter

Instrument measuring peak sound intensity in a predetermined time period.

Audio console

Console made up of all the devices used to control, adjust and mix sound.

Jack field

Series of connector sockets (jacks) allowing various pieces of equipment to be linked to the audio console.

Producer turret

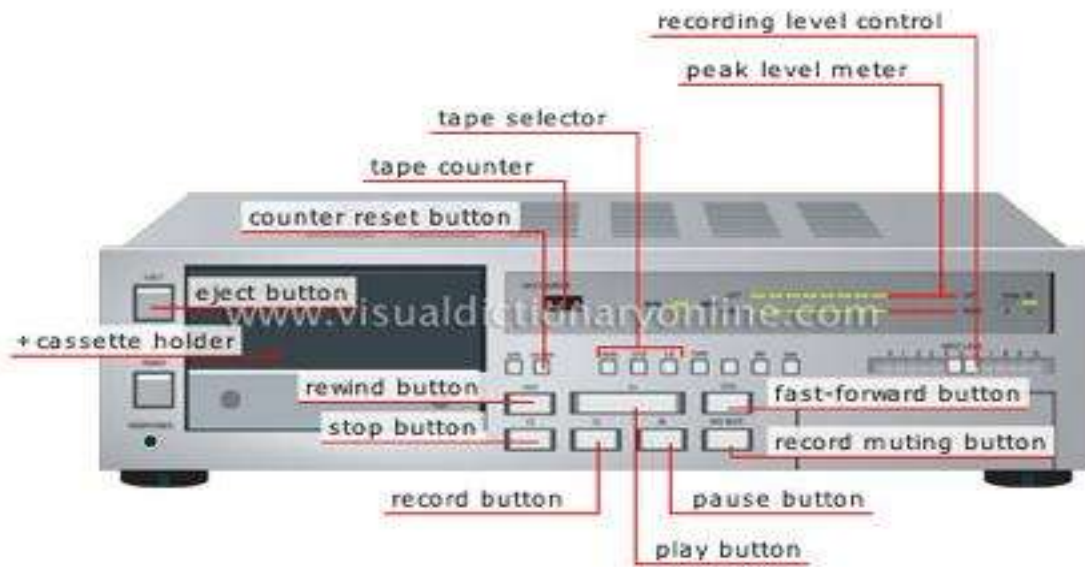
Control unit with a microphone that is used by the program's producer to communicate with the announcer.

Stop watch

Instrument that precisely measures time in minutes, seconds and fractions of seconds.

Cassette deck

Device used to play back and record sounds on a recording tape cassette.



Playing window

Opening allowing the recording tape to advance in front of the playback head of the cassette tape deck.

Housing tape-guide

Part that holds and guides the recording tape in front of the playing window.

Guide roller

Spool that guides the recording tape.

Recording tape

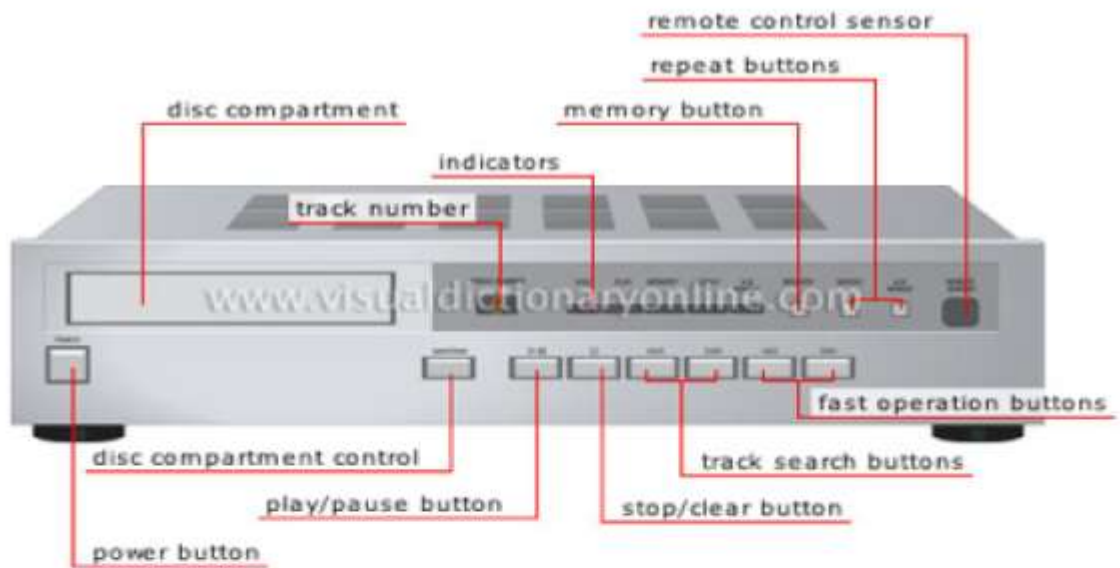
Flexible tape whose surface is covered with a magnetic substance; it is used as a recording medium.

Take-up reel

Cylindrical part on which the recording tape winds.

Compact disc player

Device using a laser beam to play back sounds recorded on a compact disc (CD).



Remote control sensor

Device that receives infrared signals emitted by a remote control so that certain functions can be operated from a distance.

Fast operation buttons

Buttons that accelerate forward or backward sound reproduction.

Track search buttons

Buttons used to move to the next or previous track.

Stop/clear button

Button that interrupts playback of a disc or erases programs held in memory.

Play/pause button

Button that temporarily starts or stops playback of a disc.

Disc compartment control

Button that opens and closes the disc tray.

Power button

Mechanical connection that turns the player on or off.

Disc compartment

Compartment that contains the tray into which discs are inserted for playback.

Track number

Liquid crystal display that shows the number of the track being played.

Indicators

Indicator lights showing the operations carried out or the functions selected.

Memory button

Button for programming playback of a number of tracks in a given order.

Repeat buttons

Buttons allowing repeated playback of one or several tracks.

Digital audio tape recorder

Device using a small magnetic tape cartridge to digitally record a program for later broadcast.

Cartridge tape recorder

Device for analog recording of a program for later broadcast using a magnetic tape cartridge.

Tone leader generator

Device producing the tracking or technical tuning signals that is inserted at the beginning of a recording.

Clock

Clock used to time a program.

Volume unit meters

Instruments measuring the relative intensity of the various sounds being broadcast or recorded.

Audio monitor

Device that reproduces the audio portion of an on-air program to monitor its sound quality.

Loudspeakers

Case enclosing one or several speakers, which convert electric pulses into sound waves by means of an amplifier.



Left channel

Speaker cover

Thin grille made of fabric or metal that covers and protects the speakers.

Right channel

Diaphragm

Cone-shaped flexible part that vibrates to create sound waves in the air.

Woofer

Loudspeaker designed to reproduce the low frequencies of the sound signal.

Midrange

Loudspeaker designed to reproduce the middle frequencies of the sound signal.

Tweeter

Loudspeaker designed to reproduce the high frequencies of the sound signal.

On-air warning light

Light indicating that a program is being broadcast.

Announcer turret

Control unit used by a program host or announcer mainly to turn a microphone on or off.

Microphone

Device that converts electric pulses into broadcast or recorded sounds.

Dynamic microphone.



Cable Housing Connector

Device used to connect the cable to the microphone.

On-off switch

Power-connecting device used to turn the microphone on or off.

Windscreen

Screen covering and protecting a microphone; it muffles the speaker's breathing and the sound of the wind.

Studio

Soundproof area designed for sound recording; radio programs are produced here.

3 PROJECT STUDIO

Dear students for the details of Project Studio we found a very effective video clip which elaborated well the concept of this studio. Please watch this video at the link given below.

<https://youtu.be/2q9H2pfFcuo>

4 PORTABLE STUDIO



A simple portable recording studio.

Peter Valentino/Stock Xchnng

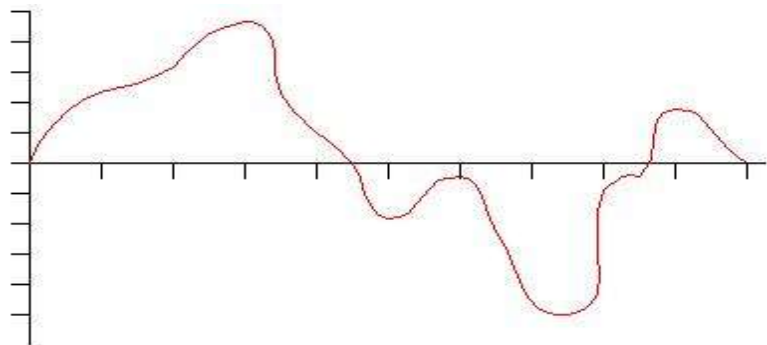
The most important piece of equipment in any portable studio is the digital audio work-station (DAW), also known as a computer. Depending on the software on the computer, a DAW could act as a recording device, mixer and sequencer. By handling so many tasks, a good DAW reduces the need for additional equipment.

Handling audio files requires a lot of computer horsepower, particularly if you're mixing lots of channels. For that reason, it's important to choose a computer with a fast micro-processor. For a while, it seemed like Mac computers would always reign supreme in the world of media computing. But some audio engineers say that the differences between Mac and PC performance are negligible. As long as the computer you pick has a powerful CPU and a large, fast hard drive, you're in good shape.

Another piece of the portable studio setup is the audio interface. While many computers have input and output ports and sound cards, they aren't always capable of recording or playing back professional-quality sound. For that reason, many engineers who set up portable studios rely on additional audio interface devices. These devices range in size from a handheld gadget to a machine the size of a hefty VCR.

Audio interface devices usually have multiple input and output ports. Many have both analog and digital ports, which covers all musical instruments and microphones. Some also act as analog-to-digital converters (ADCs). That means the device can accept an analog signal and then digitize it. It converts sound into information that a computer can manipulate.

Analog signals are continuous waves that vary in frequency and amplitude. An analog audio signal's frequency corresponds to the sound's pitch. The wave's amplitude represents the sound's volume. Digital signals aren't continuous. Instead, a digital signal is a series of snapshots called samples. The number of times a computer takes a snapshot of an analog signal per second is the sampling rate. Higher sampling rates translate into smoother, more natural sound.



An analog sound wave is continuous.

Not all audio interfaces are also ADCs. Some audio engineers might prefer to use a dedicated ADC, then run the signal coming from the ADC through the audio interface and into the DAW. Either way, the audio interface carries the signal to the DAW. Audio engineers use the DAW to manipulate individual channels and mix the sound into a final track.

The DAW might be the most important hardware component in a portable studio, but it's useless without the right software. Keep reading to learn about the applications audio engineers use to produce music.

Channels and Mixing

You've probably heard terms like 4-channel or 16-channel devices. What does that mean? In general, it means the equipment can accept a certain number of individual input devices when recording. A 4-channel device can accept four inputs, for example. In re-cording, each channel remains separate from the other channels. That's where a mixer comes in. Audio engineers use mixers to tweak each channel before combining all inputs into a single recording.

The Rest of the Gear

Audio experts like eSessions.com CEO Gina Fant-Saez suggest audio engineers purchase an external hard drive to augment their DAWs. That's because computer hard drives tend to record more slowly than external hard drives. To record audio reliably, you need a hard drive that can spin at faster speeds [source: O'Reilly Digital Media]. Other hardware an audio engineer needs to complete a portable studio includes:

Headphones

Microphones

Speakers

Cables

5 LIVE ON LOCATION RECORDING

The 5 Best Methods for Recording on Location



While musicians have access to state-of-the-art recording studios, many travel to record their tunes on location. Challenging perhaps, for producers to get the sound quality they need, the practice of recording on location is a rewarding way to make music.

Moreover, there are many possibilities for recording music with top-notch production and great sounds that rival the studio setting. The following five methods are among the best ways to record in a non-studio setting.

1. Choose Location with Care

Many artists choose to record at a particular location due to its optimal acoustics.

Some rooms are known for their great drum sound while some locations may enhance vocals. If you're recording indoors, it's essential to prevent leakage from heating or cooling elements from getting through to your microphones and muddying the clarity of your recordings.

While a lot of tweaking can be accomplished by editing with modern recording programs, it's somehow always better when you get that organically pure sound. Depending on whether you're recording a single musician or an orchestra, consider your location with care; discover its pros and cons before you fly the band in so you're ready to tackle any issues before they can detract from your recording process.

2. Field Recording

If you are in a situation where the natural sounds of your setting won't interfere and may even complement your live recording, you can opt for field recording.

Ocean waves might have more ambiance than the local airport, but there are many natural settings that can enhance the acoustics of a performance. Of course, if you are recording a band, it is still best to lay down vocal tracks separately. The various nuances of a vocal performance can easily be lost in a live recording situation even with the best of microphones.

Of course, there's no harm in trying, but whenever possible, it's usually recommended to record vocals in isolation from other instruments to avoid muddled lyrics and so forth. Zoom Handheld Recorders are now popularly used for many types of field recordings.

3. Microphones—Best You Can Buy

There's always a lot of hype about various recording programs—the latest and greatest upgrades Pro Tools has to offer!

However, no matter what program you choose, you can avoid a lot of editing headaches simply by recording with the best microphones for your recording situation. Stereo microphones and a minimum of leakage can allow you to capture a great recording outside of the studio.

It'll take some time to set up the microphones to capture each instrument as perfectly as possible, but tweaking your microphone setup is one key method of getting the most out of a recorded performance.

4. Mobile Studio

At one time, you had to be as successful as the Rolling Stones to afford a portable studio.

Today, there are many recorders you can easily move from place to place and remain well in budget. Today's technology will allow you to record with a laptop given the right soft-ware. Whether you are going with a minimum of gear or a bus-full of recording equipment, you can now bring the studio to just about any setting you want to record at.

5. Self-Recording Keyboards

Keyboards like the Korg MS-1 that specialize in sampling can also record music. You won't capture a lengthy performance the way you would with a laptop, but using an instrument typically keyboards—that can record themselves along with other sounds is another interesting method to use when on location. Its built-in microphone is also handy and its editing capabilities and software make this particular keyboard ideal for recording situations.

Recording on Location

Recording outside of the studio setting is more popular than ever. Given the right equipment for your plans, you can capture a performance anywhere.

You can also take your tracks back to a studio to tweak them or dump them into your editing program to make the changes you need. Research your gear and try to hire the best recording engineer for the job and no one is ever likely to know you recorded anywhere but in a professional recording studio.

6 PEOPLE BEHIND THE PRODUCT

- **Engineers**
- **Producers**
- **Technicians**
- **Composers**
- **Editors**

7 RECORDING PROCESS

PHASES OF THE RECORDING PROCESS

Following are the typical phases of the recording process:

PRE-PRODUCTION

During this stage, all the initial decisions are made regarding the recording. What is the purpose of the recording (i.e., demo for shopping or submission, indie release, song for download sales, just for fun, etc.)? What style of music will you be recording? Who will be playing what instruments, or what sounds will you use for the recording? Where will you record, mix and master the recording? Where is the budget coming from, and/or how will you raise the money for the project? How many songs will you be recording, and who will be involved in the writing? Will you need to hire musicians to play on the recording or will you and/or your band perform everything? When do you need to complete the recording by?

SOUND SOURCE SELECTION

This does not have to be a ‘formal’ process, but is nonetheless an important part of the project recording process. Everything that happens down the line in terms of the quality of the recording will be influenced by the sound source selection. Sound sources include the brand and model of instruments, the quality of the samples, and the caliber of the soft synths and virtual instruments. Of course, the vocals are an important sound source, so the quality of the vocalist counts as well. Use sound sources of low quality, and you will be paying the price the rest of the way. Great sounding sound sources (performed and recorded well, of course) will make it much easier to mix the songs later on.

SELECTION OF PERFORMERS

Naturally, if you are either in a band or performing as a solo artist, you will most likely pick yourself or your band members to play all the instruments and handle all the performances. The selection of the performers is the next most important thing after the sound source selection. Great sounding (high-quality and well maintained) instruments played with passion by great performers can overcome even a bad recording. Therefore, put your ego aside and get the best performers to perform on the recording. If you consider yourself the best performer by virtue of the fact that it is your music and you a unique passion for the performance and understand the songs the best, then that can also be a valid argument for performing the songs yourself.

WRITING

Once the pre-production is complete, the writing process can proceed efficiently. Time isn't wasted trying to figure out all the things that have already been covered in the pre-production stage, and instead everyone can focus on writing the best songs they can possibly come up with for the recording.

REHEARSAL

Not all bands have the time or patience to rehearse prior to their recording. Nevertheless, it is a crucial part of the process because this is where you can discover different ways to perform the song; including what the right key is, how the tempo feels, etc. Often, this part is combined with the writing process. Some performers use the recording process to rehearse, which is a waste of time and money.

RECORDING

This is the stage where the actual recording happens. Important decisions will be made regarding the best ways to capture the sounds. This is where an experienced and/or knowledgeable engineer plays a crucial role. Unless you already have the sounds in your computer, or on tape (e.g., from samples, soft synths, and/or virtual instruments), there are only three ways to record a sound; either by microphone, or via direct injection (DI), or a combination of the two. The quality of your microphones and DI boxes are extremely important to the final results, along with the mic-pres, additional processors (e.g., compressor, deesser, EQ, etc.), and A/D converter. Of additional importance is 'sound' of the room in which you are recording, the maintenance of the equipment being recorded, along with mic placement, the quality and length of mic cables, the techniques used by the recording engineer, among other things. Along with great sound sources and great performers (and performances), getting the recording right will carry you into a great sounding mix and master.

EDITING

Once everything is recorded, additional editing can be done in order to get the recording to sound perfect (if perfection is what you are after) and get the tracks ready for mixing. Some musicians prefer to leave the recording exactly as it was recorded for a more authentic and natural sound. For others, additional work is needed in order to piece together the perfect lead vocal take, create a

powerful guitar solo, tune a voice or instrument, fix drum timing, apply time stretching to a track, adjust an early or late take, cut and paste background vocals from one chorus to another or replace a word sung in the wrong place, and so on. Editing the tracks into their final form allows the mixing process to be just about mixing, instead of spending valuable time editing.

MIXING

The mixing stage is the stage where you take all the individual tracks that have been re-recorded (e.g., vocals, guitars, bass, kick, snare, keyboard, flute, violin, samples, etc.) and apply processing (e.g., volume levels, panning, compression, EQ, reverb, chorus, delay, flange, phaser, gating, etc.) in order to make everything sound as good as possible. This stage allows the most flexibility in manipulating the sounds, since processing can be applied to each individual track as necessary. Bad decisions made here will negatively affect the next stage (mastering). Any problems you have with the mix should be taken care of at this stage, and not left to be addressed later during the mastering stage where it becomes much more difficult, if not impossible to fix things. It is infinitely easier to focus in on a problem area, whether it is volume, tone, or character, and manipulate it using volume and panning as well as processing like compression, EQ, expansion, chorus, gating, etc., during the mixing stage. The final result of the mix is that all the individual tracks are mixed from multiple tracks 'down' to two (2) tracks as a stereo mix.

MASTERING

The mastering stage is the stage where you take the mixed 2-track source and apply any additional processing that might be necessary and create a master suitable for replication. It can be said that good mastering creates a more 'finished' product that appears to have more sheen, heft, depth, punch, and clarity than the mix alone. Generally speaking, the processing applied during mastering is mainly high-quality equalization, compression, the occasional multi-band processing (compression, expansion) if necessary, stereo enhancement/correction, noise removal, and volume maximizing (limiting). All processing applied during mastering will affect the entire mix, unlike during mixing, where each track is processed individually. Some processes typically applied during the mixing stage (e.g., chorus, delay, flange, phaser, etc.) are not normally applied during the mastering stage, unless in moderate amounts for special effect.

In addition to signal processing, some other important things take place during mastering; like sequencing the songs into the correct order, selecting the correct length of space (silence or room noise) in between each song, inserting metadata (ISRC, UPC, title, artist, copyright info, etc.), assuring no errors are on the final master, supplying a master (or more accurately, a pre-master) to

the replicating plant suitable for replication, etc. Mastering will almost always make the mix sound better. However, the quality of the mix will greatly affect the quality of the master. Some things cannot be fixed in the mastering stage, and should more suitably be dealt with at the mixing stage (or even further back, depending on the issue).

IN CONCLUSION

This is a general description of the different phases of the recording process, from conception to completion. The important thing to remember is that the more attention you pay to each phase in the process, the better (more professional) the final master will sound. Do not wait until the very final stage to try and correct all the mistakes made or shortcuts taken along the way.

8 PREPARATION

8 Tips for Preparing Audio Scripts for Recording



Tips-for-preparing-script

What does it take to turn a storyboard into a script that a narrator can easily read? Whether you are recording at a studio or in-house, whether you are using a professional voice over artist or a coerced colleague, there are certain conventions that make the task easier.

Here are tips for formatting and organizing your script that apply to all types of recording at work, home, and in professional studios. Some are known conventions and some are simply what I have discovered through trial and error.

8.1. Double-check for Errors

Every script has errors. It could be incorrect content or a misuse of grammar. Each error you discover during the recording session slows things down and stops the narrator's momentum. If the error requires contacting the SME, that can be a real headache.

Therefore, go over that script with a thorough eye for detail and read it aloud. Ask someone else to review it for errors too. Making real-time corrections during a recording are not uncommon, but the less this happens, the smoother your recording will go.

8.2. Indicate Emphasized Words

I remember the first time I was at a recording for a script that I wrote, I was surprised that the voice talent didn't always intonate sentences the way I intended. In hindsight, this seems obvious. How could someone else, who is not even familiar with the content, read a script with the same voice modulation I had in my mind.

That's when I started to add emphasis in every script. Typically this is done through text formatting, such as using bold or italicized text. To avoid confusion, use one method for the entire script and communicate this convention to your narrator in a prerecording briefing.

8.3. Provide Pronunciation for Little Known Terms

Using terms that are unique to a field can slow down a recording. If you use medical, technical or other specialized vocabularies, find a way to communicate the pronunciation of these words and acronyms in the script. Point these out to the narrator before recording begins.

For example, you can write out the phonetic spelling of a term in brackets, so the narrator can quickly see the pronunciation. If the script uses acronyms, indicate whether the term should be pronounced by its letters or as a word. For example, when the letters alone are used, I write it with dashes, as in U-S-A.

8.4. Indicate Where You Need Pauses

If you allow for pauses in the script, it is easier to accommodate graphical changes on the screen, such as animations and progressive reveals. You can add an ellipsis (three dots) to the script or write the word “pause” in brackets when you need that extra half-second of silence.

Let the narrator know that at these points, you would like a pause of “one beat.” That nearly imperceptible moment of silence will help you synchronize the audio and visuals seamlessly during course production.

8.5. Insert Page Numbers

During a recording at a professional studio, one of my team members created a script without page numbers. The audio engineer teased him about this the whole time. That was enough to ensure I always remember page numbers. They are essential because during a live recording because everyone present will need to reference them. Also, be sure the page numbers are located in a very obvious place, such as bottom center.

8.6. Avoid Page Turns

When using a paper version of a script (which I find many professional voiceover artists prefer), be sure that he or she will not need to turn the page in the middle of a sentence or paragraph. The sound of paper turning usually gets picked up by the mic. Actually, this is a good tip even if the narrator is reading the script online. The time it takes to find and press the Page Down key can ruin the sound byte.

8.7. Name Your Audio Files

When recording for an eLearning course, I like to prepare the script so that the audio segment for each screen is associated with a unique file name. Devise a naming strategy that makes sense in your production environment.

Therefore, my scripts have two columns a narrow one on the left and a wider one on the right. The column on the left indicates the name of the audio file; the column on the right holds the script. This ensures that everything is well-defined for the person (even if it's you) doing the postproduction work of breaking up the audio into smaller files.

8.8. Make it Easy for the Voice Talent

Regardless of whether the script is read online or from a print-out, double-space the text and use an easy to read typeface so the script is highly legible. The physical attributes of the script should be transparent to the narrating process.

Always provide the script to your narrator a few days before the recording session. Professionals always ask for a script ahead of time, so it makes sense to give it to your col-leagues too. Not only will the recording have fewer retakes, your narrator will feel more comfortable and prepared.


9 RECORDING

Record audio with Sound Recorder

You can use Sound Recorder to record a sound and save it as an audio file on your computer. You can record sound from different audio devices, such as a microphone that's plugged into the sound card on your computer. The types of audio input sources you can record from depend on the audio devices you have and the input sources on your sound card. (Applies to Windows 7)

To record audio with Sound Recorder

Make sure you have an audio input device, such as a microphone, connected to your computer.

Open Sound Recorder by clicking the Start button . In the search box, type Sound Recorder, and then, in the list of results, click Sound Recorder.

Click Start Recording.

To stop recording audio, click Stop Recording.

Optional if you want to continue recording audio, click Cancel in the Save As dialog box, and then click Resume Recording. Continue to record sound, and then click Stop Recording.

Click the File name box, type a file name for the recorded sound, and then click Save to save the recorded sound as an audio file.

Note

To use Sound Recorder, you must have a sound card and speakers installed on your computer. If you want to record sound, you also need a microphone (or other audio input device).

You can play the saved audio file on your computer by using a media player program.

<http://windows.microsoft.com/en-US/windows7/Record-audio-with-Sound-Recorder>

10 OVERDUBBING

The process of making an overdub, or overdubs is a technique used in audio recording, whereby a performer listens to an existing recorded performance (usually through headphones in a recording studio) and simultaneously plays a new performance along with it, which is also recorded. The intention is that the final mix will contain a combination of these "dubs".

Tracking (or "laying the basic tracks") of the rhythm section (usually including drums) to a song, then following with overdubs (solo instruments, such as keyboards or guitar, then finally vocals), has been the standard technique for recording popular music since the early 1960s. Today, overdubbing can be accomplished even on basic recording equipment, or a typical PC equipped with a sound card, using digital audio workstation software.

Examples

Overdubs can be made for a variety of reasons. One of the most obvious is for convenience; for example, if a bass guitarist is temporarily unavailable, the recording can be made and the bass track added later. Similarly, if only one or two guitarists are available, but a song calls for multiple guitar parts, a guitarist can play both lead and rhythm guitar. Overdubbing is also used to solidify a weak singer; double tracking allows a singer with poor intonation to sound more in tune. (The opposite of this is often used with sampled instruments; detuning the sample slightly can make the sound more lifelike.)

Overdubbing has sometimes been viewed negatively, when it is seen as being used to artificially enhance the musical skills of an artist or group, such as with studio-recorded inserts to live recordings, or backing tracks created by session musicians instead of the credited performers. The early records of The Monkees were made by groups of studio musicians pre-recording songs (often in a different studio, and some before the band was even formed), which were later overdubbed with the Monkees' vocals. While the songs became hits, this practice drew criticism. Michael Nesmith in particular disliked what overdubbing did to the integrity of the band's music. Additionally in working with producer Butch Vig, Kurt Cobain had expressed a disdain for double-

track recording. Vig had to reportedly convince Cobain to use the recording technique by saying, "The Beatles did it on everything. John Lennon loved the sound of his voice double-tracked."

11 MIX DOWN

In sound recording and reproduction, audio mixing (or “mix down”) is the process which commences after all tracks are recorded (often, tracked) and edited as individual parts. The mixing-process can consist of various processes but are not limited to setting levels, setting equalization, using stereo panning, and the addition of effects. The way the song is mixed has as much impact on the way it sounds as each of the individual parts that have been recorded. Dramatic impacts on how the song affects the listeners can be created by minor adjustments in the relationship among the various instruments within the song.

Audio mixing is utilized as part of creating an album or single. Mixing is largely dependent on both the arrangement and the recordings. The mixing stage often follows a multi track recording. The process is generally carried out by a mixing engineer, though sometimes it is the musical producer, or even the artist, who mixes the recorded material. After mixing, a mastering engineer prepares the final product for reproduction on a CD, for radio, or otherwise.

Prior to the emergence of digital audio workstations (DAWs), the process of mixing used to be carried out on a mixing console. Currently, more and more engineers and independent artists are using a personal computer for the process. Mixing consoles still play a large part in the recording process. They are often used in conjunction with a DAW, although the DAW may only be used as a multi track recorder and for editing or sequencing, with the actual mixing being performed on the console.



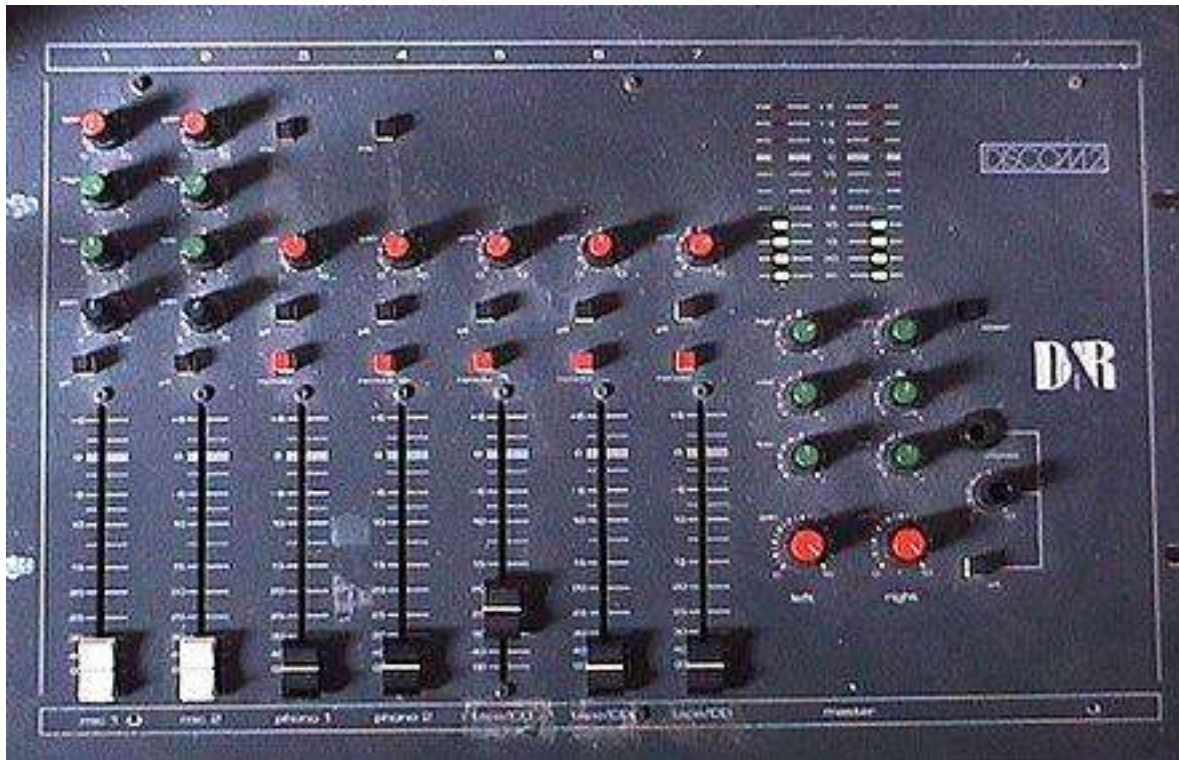
The role of audio mixing

In its simplest form an audio mixer combines several incoming signals into a single output signal, however this is not as simple as connecting the input signals in parallel and sending them to a single output signal because they could influence each other. In order to combine different signals, they must be mixed first so that each signal has a relationship of hierarchy (each signal's volume one step below the next). The role of a music producer is not necessarily a technical one, with the physical aspects of recording being assumed by the audio engineer, and so producers often leave the similarly technical mixing process to a specialist audio mixer. Even producers with a technical background may prefer that a mixer comes in to take care of the final stage of the production process. Noted producer and mixer Joe Chiccarelli has said that it is often better for a project that an outside person comes in because:

"when you're spending months on a project you get so mired in the detail that you can't bring all the enthusiasm to the final [mixing] stage that you'd like. [You] need somebody else to take over those responsibilities so that you can sit back and regain your objectivity."

Equipment

Mixing Consoles



A mixer, or mixing console, or mixing desk, or mixing board, or software mixer is the operational heart of the mixing process. Mixers offer a multitude of inputs, each is fed by a track from a multitrack recorder. Mixers typically have 2 main outputs (in the case of two-channel stereo mixing) or 8 (in the case of surround).

Mixers offer three main functionalities:

Mixing – summing signals together, which is normally done by a dedicated summing amplifier or in the case of digital by a simple algorithm.

Routing – allows the routing of source signals to internal buses or external processing units and effects.

Processing – many mixers also offer on-board processors, like equalizers and compressors.

Simple mixing console

Mixing consoles used for dubbing are often large and intimidating, with exceptional amount of controls. These controls luckily consist of a great deal of duplication among them, so by studying just one area of a console, one learns nearly all of the areas. The mixing console can at the end of the day be broken down into two ingredients, processing and configuration. Sound processes are the devices that is used to manipulate the sound, all the way from simple internal level controls to sophisticated outboard reverberation units whereas the configuration issues consist out the signal routing from the input to the output of the console through the various processes.

Digital Audio Workstations (DAW's) today have many mixing features which potentially have more processes available to that of a major console. The distinction between DAW's equipped with a control surface and large consoles is usually that, if the console is digital, it will consist of dedicated digital signal processors for each channel and is thus designed not to “overload” under the burden of signal processing and which may possibly crash or lose signals. DAW's can dynamically assign resources like digital audio signal processing power and so could run out if many signal processes were in simultaneous use. The upside of this is that this can be solved fairly easily by just plugging in more hardware into the DAW, but the downside would be that the cost of this endeavor may approach that of a major console.

12 MASTERING

A form of audio post production, is the process of preparing and transferring recorded audio from a source containing the final mix to a data storage device (the master); the source from which all copies will be produced (via methods such as pressing, duplication or replication). In recent years digital masters have become usual although analog masters, such as audio tapes, are still being used by the manufacturing industry, notably by a few engineers who have chosen to specialize in analog mastering.

Mastering requires critical listening; however, software tools exist to facilitate the process. Mastering is a crucial gateway between production and consumption and, as such, it involves technical knowledge as well as specific aesthetics. Results still depend upon the accuracy of speaker monitors and the listening environment. Mastering engineers may also need to apply corrective equalization and dynamic compression in order to optimize sound translation on all playback systems.[2] It is standard practice to make a copy of a master recording, known as a safety copy, in case the master is lost, damaged or stolen.

Digital technology

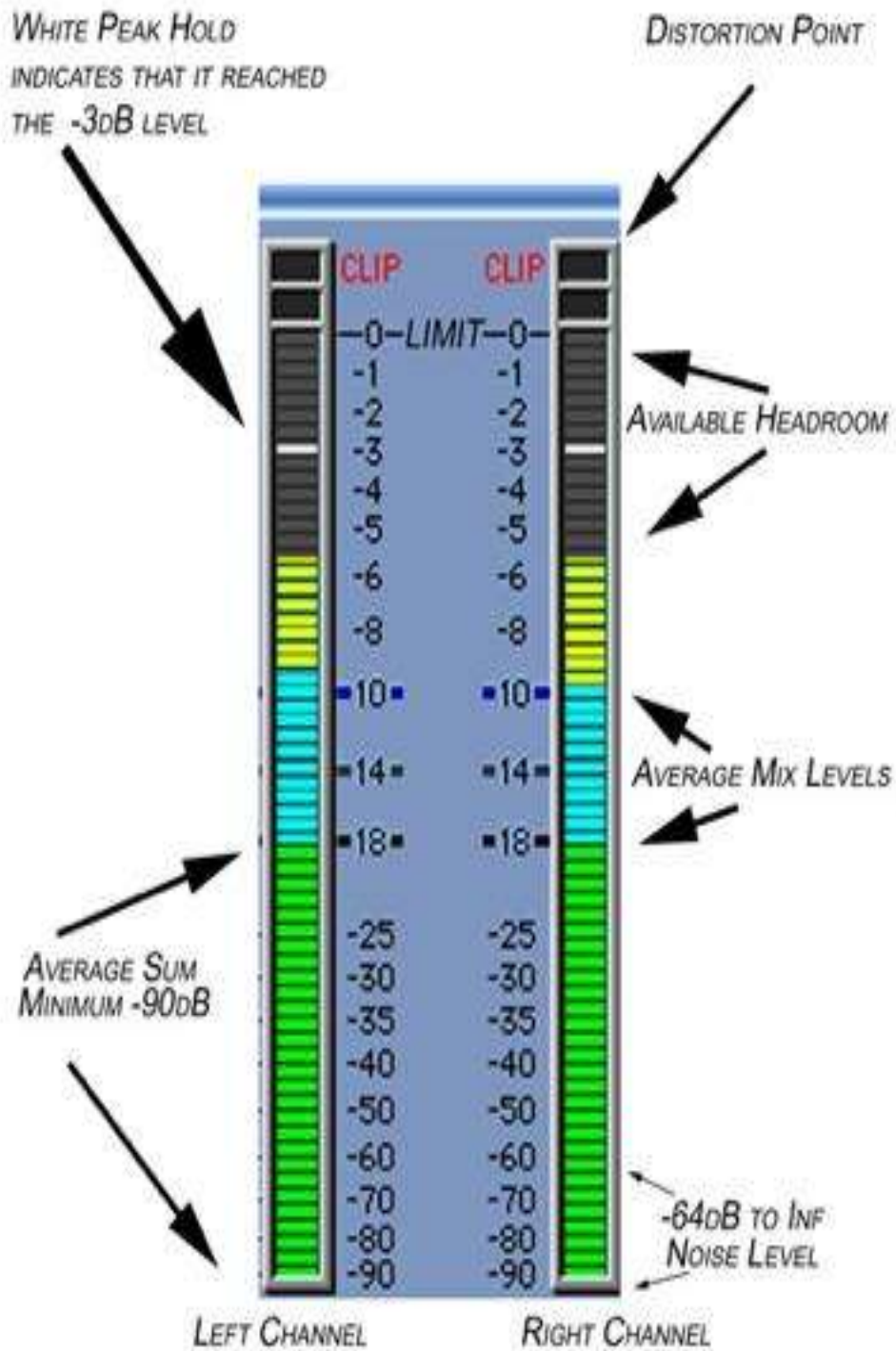
Optimum Digital Levels with respect to the Full Digital Scale (dBFS)

In the 1990s, electro-mechanical processes were largely superseded by digital technology, with digital recordings stored on hard disk drives or digital tape and transferred to CD.

The digital audio workstation (DAW) became common in many mastering facilities, allowing the off-line manipulation of recorded audio via a graphical user interface (GUI). Although many digital processing tools are common during mastering, it is also very common to use analog media and processing equipment for the mastering stage. Just as in other areas of audio, the benefits and drawbacks of analog digital technology compared to analog technology are still a matter for debate. However, in the field of audio mastering, the debate is usually over the use of digital versus analog signal processing rather than the use of digital technology for storage of audio.

Although there is no "optimum mix level for mastering", the example in the picture to the right only suggests what mix levels are ideal for the studio engineer to render and for the mastering engineer to process. It is important to allow enough headroom for the mastering engineer's work. Reduction of headroom by the mix or mastering engineer has resulted in a loudness war in

commercial recordings.



OPTIMUM MIX LEVELS FOR MASTERING

Steps of the process typically include the following:

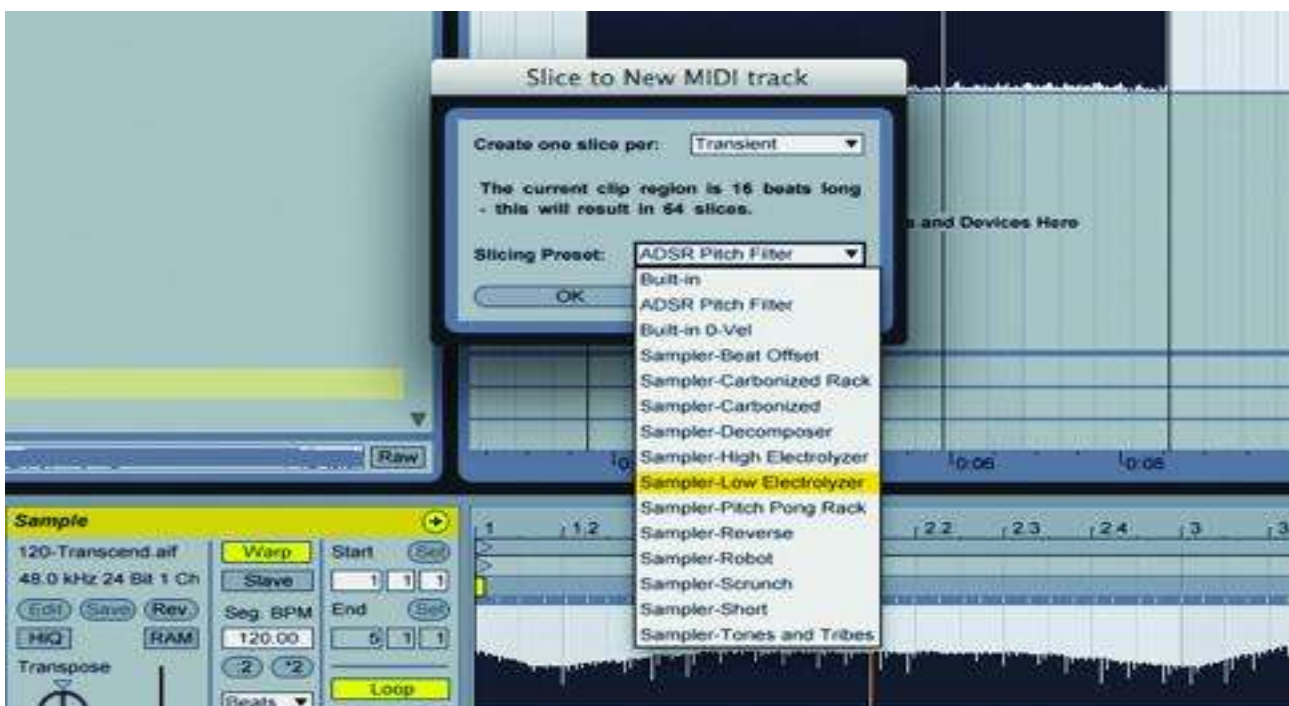
- Transferring the recorded audio tracks into the Digital Audio Workstation (DAW).
- Sequence the separate songs or tracks as they will appear on the final release.
- Adjust the length of the silence between songs.
- Process or "sweeten" audio to maximize the sound quality for the intended medium (e.g. applying specific EQ for vinyl).
- Transfer the audio to the final master format (CD-ROM, half-inch reel tape, PCM 1630 U-matic tape, etc.).

Examples of possible actions taken during mastering:

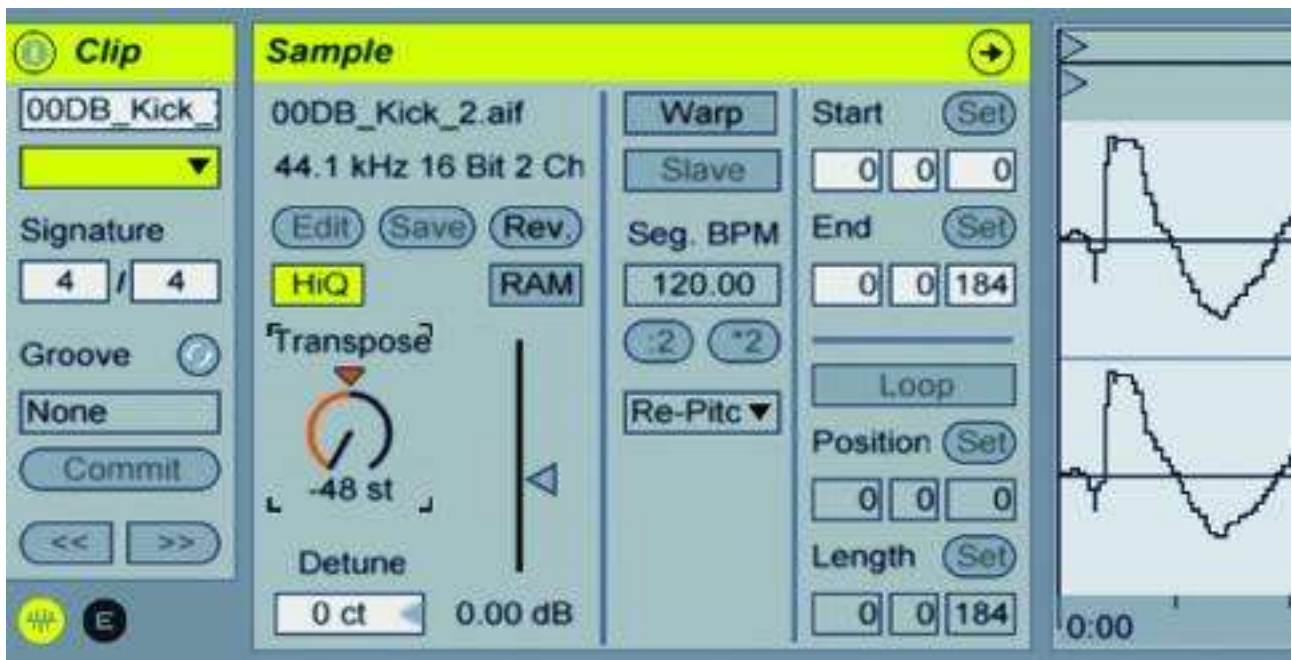
- Editing minor flaws.
- Applying noise reduction to eliminate clicks, dropouts, hum and hiss.
- Adjusting stereo width.
- Adding ambience.
- Equalize audio across tracks for the purpose of optimized frequency distribution.
- Adjust volume.
- Dynamic range compression or expansion.
- Peak limit.
- Dither.

13 SEQUENCE EDITING

Sequencing and editing options...



1: There are three ways to manipulate audio: using smaller, sequenced individual events, manipulating from within a loop or recording of a performance, or by using Live's Slice To New MIDI Track function (from right-click menu) to get audio into a MIDI-controllable form. The latter can utilise techniques from the previous set of steps here, but our re-commendation is to fully explore the presets available



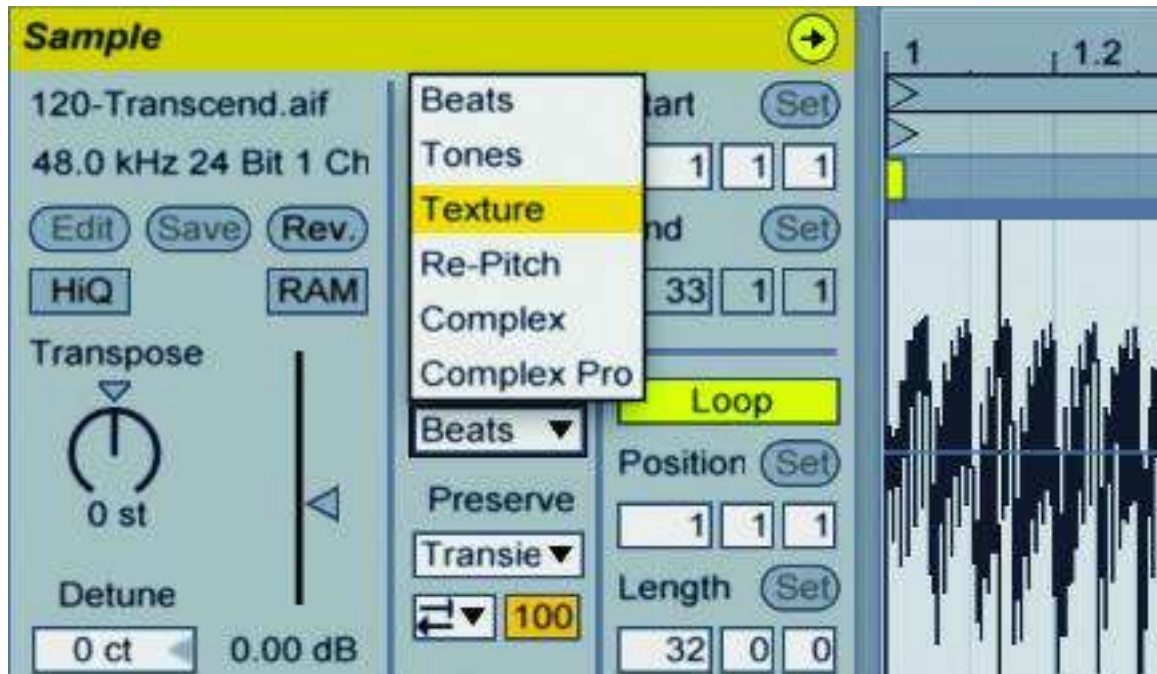
2: The first few variables for an audio file can be discovered quite easily with single sounds. Starting with Warp mode disabled, Transpose will let you pitch the audio up and down with variable speed. Extreme settings of an octave or more in either direction will cause great sonic changes which can generate very interesting textures and sounds to start an idea with.



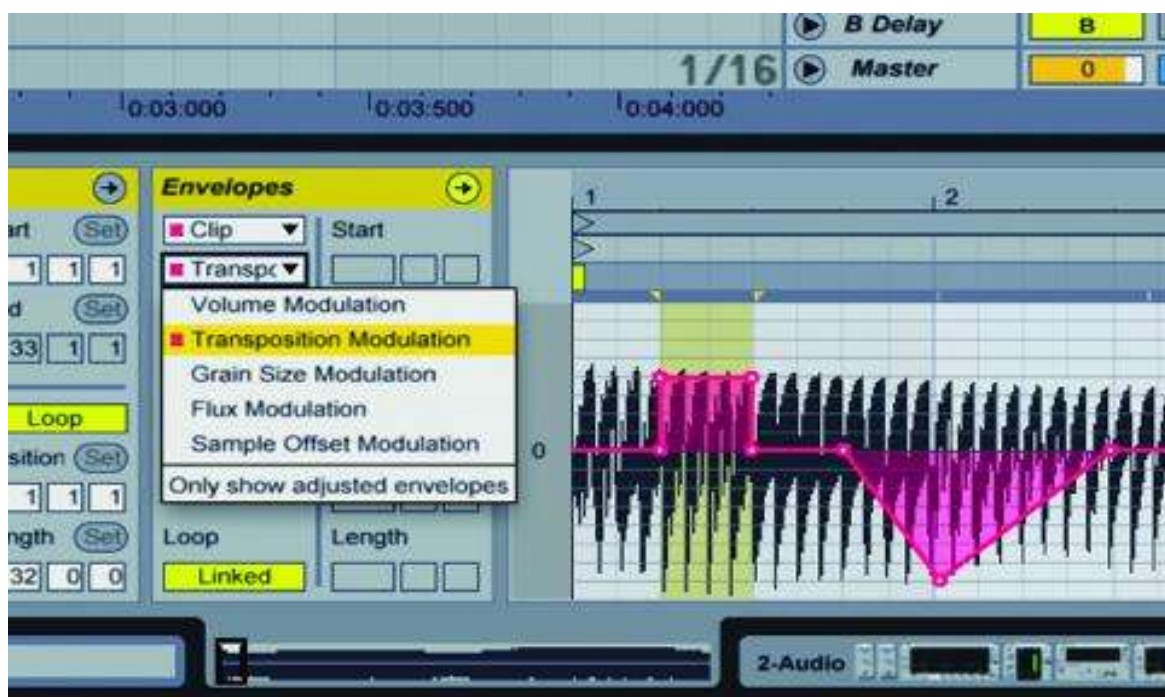
3: As soon as Warp mode is enabled you open up a whole new world of possibilities, the first being an overall time-stretch effect which, like vari-speed transpose, can create dramatic tonal changes at more extreme settings. Hit the half-time button a few times for an immediate granular-type effect.



4: Warp-based time-stretching and pitch change are easily applied to loops or recordings, but in order for single sounds to benefit from this you first need to turn them from a sequence into a new single audio file. To do this, highlight all parts on a track and select Consolidate from the Edit menu



5: To quickly explore the effects of warp markers, use the half- and double-time buttons. Warp modes can be chosen from the dropdown menu below the: 2& *2 buttons and all but re-pitch modes will generate a unique, stretched timbre. Warp markers themselves can be moved around for an in-loop variation of time-stretch and compress effects.



6: Transpose is another useful tool for broad variation. First open the Envelope box by pressing the small E button underneath the Clip box, then select Transposition Modulation and alter the envelope over time. All stretch modes can have their other parameters changed as well, so explore the new tones these can offer you.

Self Assessment Question

Q # 1. How many steps are there to make a cheap recording studio?

Q # 2. What is the importance of control room?

Q # 3. Why a control room is necessary for a broadcasting house?

Q # 4. What is the most important piece of equipment in any portable studio?

Q # 5. How many best methods are there for recording on location?

Q # 6. Describe the typical phases of the recording process?

Q # 7 What is dubbing, Please explain with example?

Q # 8. Where the sound card used?

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RECORDING SESSION

Written By: Syed Salman Ali Zaidi
Reviewer: Umer Mehmood Qureshi

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Objectives

- 1. Difference between analog and digital audio**
- 2. Analog recording technology**
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- 6. Recording process**
- 7. Signal distribution**
- 8. Digital audio recorders**
- 9. Digital audio workstation**
- 10. Launching the Software**
- 11. How to record with Adobe Audition**

Self Assessment Questions

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INTRODUCTION

Dear Students this unit provides a brief overview of audio and its recording on the computer. After reading this unit, you should have a general idea of sound and audio recording and some ideas about where to learn more. Topics include difference between digital and analogue audio, recording techniques, recording devices and creating first recording track through adobe audition. Students the modern audio recording involves an ever-expanding set of specialized tools. Each year, there are number of applications using audio increases. In this unit you will find many possible tools and technologies. In addition to what you may already have in your computer, you will find an abundance of shareware and freeware music and audio software available.

Dear Students in the past 20 years, audio has moved from analog recording (tape cassettes) as the playback medium to totally digital recording using computers with digital surround sound playback.

Dear Students today, the Musical Instrument Digital Interface (MIDI), the sequencer, sound cards, and synthesizers allow anyone to create music right on their desktop. Even a modest setup can create surprisingly realistic audio tracks. You probably have many of these tools in your computer already.

Dear students, even if you're not a musician, the technology surrounds you. Songs can be downloaded as MIDI files that will play on your computer's soundcard. Computer-based audio files can now deliver your favorite songs over the Internet. Download a group of your favorite songs as MP3 files and then create a custom CD to listen to at your next party. In this unit the students will learn to create greatest hits collections of your favorite tunes to share with your friends, add a soundtrack and sound effects to your latest video recording.

Dear students in this unit you will learn about creating your own music and audio files. You have much software for the purpose. Basically skills are important, how to produce a music file or an interview etc. In this unit you will learn about recording of music, program or audio file and steps involved in recording to a finished product. Students after this unit you will learn to create your own audio track.

OBJECTIVES:

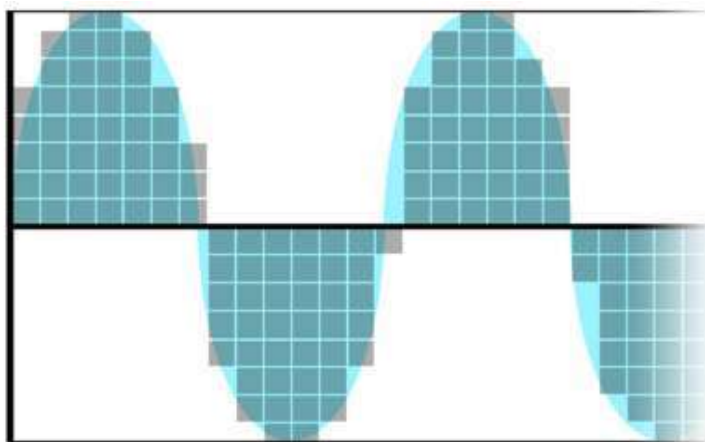
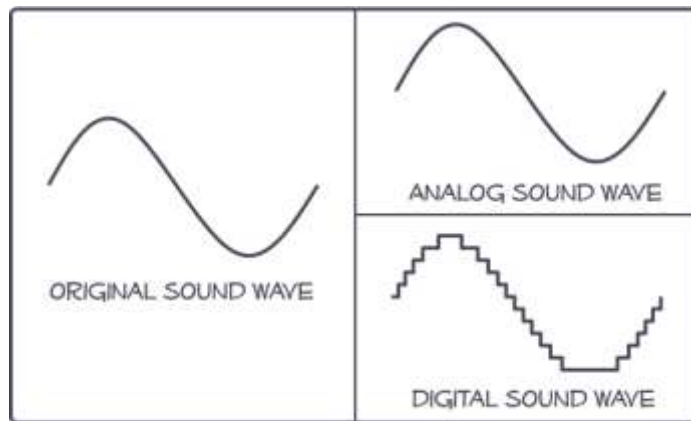
After studying this unit you will be able to:

1. The students will understand the difference between analog and digital audio.
2. Well knowledge of analog and digital audio recording techniques.
3. Comprehensive information about digital audio recorders.
4. How to record music or an audio file from scratch to a finished product.

So after learning difference between digital and analogue audio what do you think which audio is better. You can better analyse after listening both the audios recorded by an analogue or digital technology.

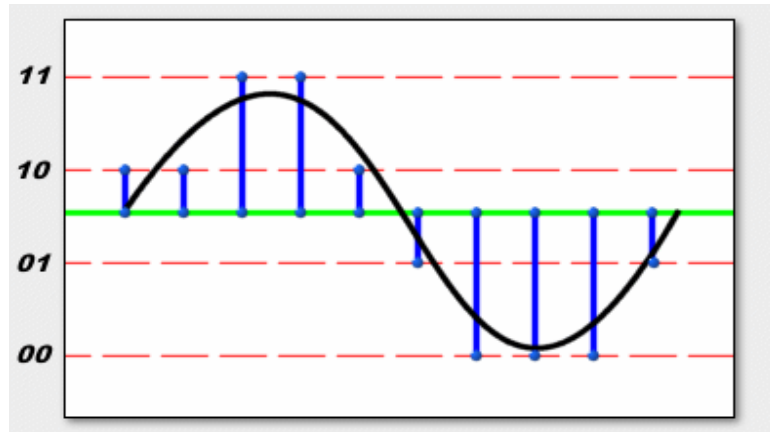
1. DIFFERENCE BETWEEN ANALOG AND DIGITAL AUDIO

Audio come in two basic types, analog and digital. Analog refers to audio recorded using methods that replicate the original sound waves. Vinyl records and cassette tapes are examples of analog mediums. Digital audio is recorded by taking samples of the original sound wave at a specified rate. CDs and Mp3 files are examples of digital mediums.



As you can see in the diagrams, the analog sound wave replicates the original sound wave, whereas the digital sound wave only replicates the sampled sections of the original sound wave. The potential fidelity (reliability) of an analog recording depends on the sensitivity of the equipment and medium used to record and playback the recording.

Among other factors, digital audio fidelity heavily depends on the rate at which the recording equipment sampled the original sound wave over a specified increment of time. Even with the newest technologies and techniques, digital audio still cannot create exact replications of an original sound wave.



Figure

Many times, we hear fancy words like “Uncompressed” and “Lossless”. These words are very misleading as all digital audio features some compression and loss of the original signal. However, even the best trained human ear may not be able to tell the difference between a high quality digital signal and an analog audio signal.

An easy way to visualize digital audio is to consider the difference between a regular light bulb and a strobe light.

Recording basics

Both digital and analog recordings have their merits. No matter which recording process is used, analog or digital, both are created by a microphone turning air pressure (sound) into an electrical analog signal.

An analog recording is made by then imprinting that signal directly onto the master tape (via magnetization) or master record (via grooves) – from which copies can be made into cassette tapes and vinyl records.



Figure: Tape Recorders

Below are some vinyl recorders:





Digital recordings take that analog signal and convert it into a digital representation of the sound, which is essentially a series of numbers for digital software to interpret.

After the analog signal is digitalized, the recording can be copied and placed onto a compact disc, hard drive or streamed online.





Compact Disc(CD)

Hard Disk

Audio Bandwidth

Bandwidth is the ability of a recorded signal to be reproduced at varying degrees of resolution. Think of it like enlarging a low-resolution image versus a high-resolution image. After a certain point, enlarging a lower-resolution image will become pixelated and difficult to see, where the high-resolution image will resize clearly.

Like images, audio signals can have a limited bandwidth if recorded digitally. Once a digital recording is made, the bandwidth is set in place.

An analog recording is considered unlimited. Therefore, it can move to a higher and higher resolution without losing its original quality.



Pixelated Image

Signal to noise ratio

The signal-to-noise ratio (SNR) is the amount of noise generated by the recording signal to your speakers. Digital recordings can have a greater signal-to-noise ratio depending on the bit depth of the recording.

The smooth analog signal matches the recorded sound wave better than the steps of a digital recording. However, the analog medium (vinyl or magnetized tape) the recording is imprinted can have tiny imperfections that cause cracking and popping noise.

Mobility of Media

Digital music can be stored, played and streamed on multiple transportable digital products (CD's, phones, mp3 players, etc.). Outside of tape players, analog-recorded music is fairly immobile.

Loss of audio quality

Digital recordings can be played and copied endlessly without ever losing their original quality. Over time, vinyl records and tapes can lose their audible value when being played or copied.

2. ANALOG RECORDING TECHNOLOGY

Analog recording methods store signals as a continuous signal in or on the media. The signal may be stored as a physical texture on a phonograph record, or a fluctuation in the field strength of a magnetic recording.

Following are some machines used for analog recording:

1. The Phonograph
2. Gramophone
3. Telegraphon
4. Magnetophone

THE PHONOGRAPH

The Phonograph was the first machine used to capture analog sound, and was invented by the well-known inventor Thomas Edison in 1877.



Edison incorporated various elements into his Phonograph that would become staples that can be found in recording devices to this day.

Recording

For a sound to be recorded by the Phonograph, it has to go through three distinct steps. First, the sound enters a cone-shaped component of the device, called the microphone diaphragm. That sound causes the microphone diaphragm, which is connected to a small metal needle, to vibrate. The needle then vibrates in the same way, causing its sharp tip to engrave a distinctive groove into a cylinder, which was made out of tinfoil.

Playback

In order to playback the sound recorded on one of the tinfoil cylinders, the recording process is essentially reversed. As the cylinder spins, the needle follows the groove created by the previous recording session. This causes the needle to vibrate, and then the diaphragm. This vibration comes out of the diaphragm, which is now functioning as a sort of sound amplification device, much like the bell on any wind instrument. The result is an audible reproduction of the originally recorded sound.

GRAMOPHONE

Fans of modern record players are already familiar with one very early improvement on the phonograph, known as the gramophone. Inventor Emile Berliner created the device in 1887, only ten years after Edison's original device.

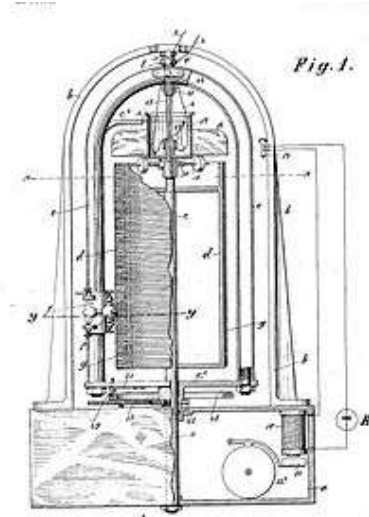


Advantages

Berliner's main improvement to the phonograph was related to the component of the device that actually held the recorded information. The previously used tinfoil cylinders were awkwardly shaped, making them hard to store. They could also not be reproduced economically, which was another reason why they were not seen as a viable option for recorded music. Berliner realized these disadvantages, and set out to create a better version of the tinfoil cylinder. What he came up with was not a cylinder at all, but was rather a flat circular disc much like modern vinyl records. These discs could not only be easily stacked and stored for safe-keeping, but were also comparatively easy to reproduce. This quality allowed for the mass production of recorded discs, which was the first step towards commercially recorded music.

TELEGRAPHONE

The next great advancement in analog sound recording came in the form of the telegraphone, which was created by Danish inventor Valdemar Poulsen between 1898 and 1900.



Early mechanical drawings of the telegraphone

This machine was vastly different from the gramophone or the phonograph, in that instead of record sound mechanically, it records using a process called electromagnetism.

MAGNETOPHON

In 1935, inventor Fritz Pfleumer took the electromagnetic recording idea and took it to the next level. Rather than using heavy, expensive, and dangerous steel wire like Poulsen, Pfleumer realized that he could coat normal strips of paper with tiny particles of iron. The iron would allow the paper to be magnetized in the same way as the steel wire, but would eliminate most of its shortcomings.



The Magnetophon

The magnetophon operated with a process nearly identical to that of the telegraphone. An inscriber, called the recording head, passes over the electromagnetic paper strip, creating patterns of varying magnetic polarity within it, which can later be played back. The playback is achieved using a reversal of the recording process. The pre-magnetized paper, which had come to be known as tape, passed over a coil, creating changes in magnetic flux. These changes were translated into an electric current, which when amplified produced a replica of the previously recorded sounds.

Advantages

There were many advantages of tape recording, but the most important was that it led to the development of multi tracking. Multi tracking occurs when multiple takes of a performance, which were recorded at separate times, are brought together to play simultaneously. This is the method all recording studios use to this day, in order to record all of the separate instruments of a song, and get the best possible takes from all of the musicians.

A reel of tape could also hold far more recorded information than previous mediums. For instance, Berliner's discs held only a few minutes of recording, meaning that each disc usually contained a single song, or multiple short clips. Pflumer's tape reels, on the other hand, could hold up to thirty minutes of sound. This ability is what eventually led to the concept of a music "album", or collection of multiple songs.

3. DIGITAL RECORDING TECHNOLOGY

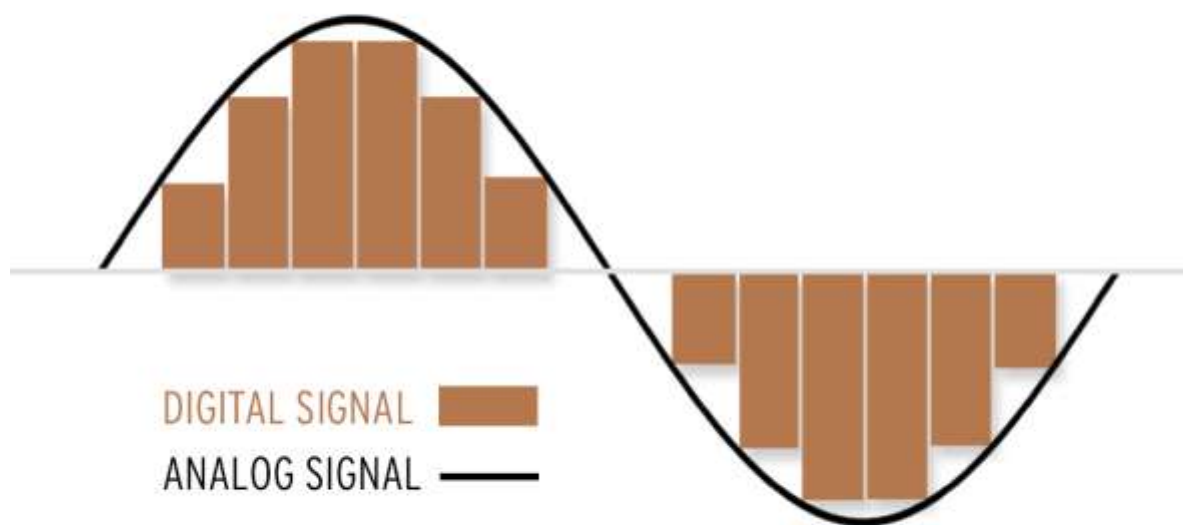
In a Compact Disc or any other digital recording technology, the goal is to create a recording with very high similarity between the original signal and the reproduced signal and perfect recording sounds the same every single time you play it no matter how many times you play it. To accomplish these two goals, digital recording converts the analog wave into a stream of numbers and records the numbers instead of the wave. The conversion is done by a device called an analog-to-digital converter (ADC).

To play back the music, the stream of numbers is converted back to an analog wave by a digital-to-analog converter (DAC). The analog wave produced by the DAC is amplified and fed to the speakers to produce the sound. The analog wave produced by the DAC will be the same every time, as long as the numbers are not corrupted. The analog wave produced by the DAC will also

be very similar to the original analog wave if the analog-to-digital converter sampled at a high rate and produced accurate numbers.

You can understand why CDs have such high fidelity if you understand the analog-to-digital conversion process better. Let's say you have a sound wave, and you wish to sample it with an ADC.

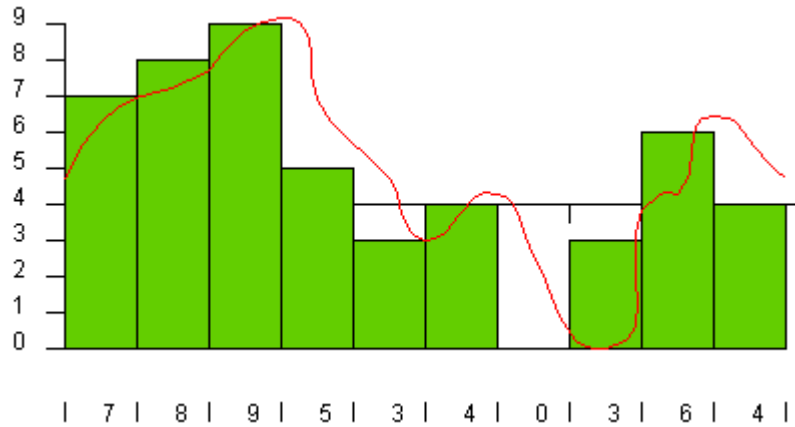
The figure shows a typical wave:



When you sample the wave with an analog-to-digital converter, you have control over two variables:

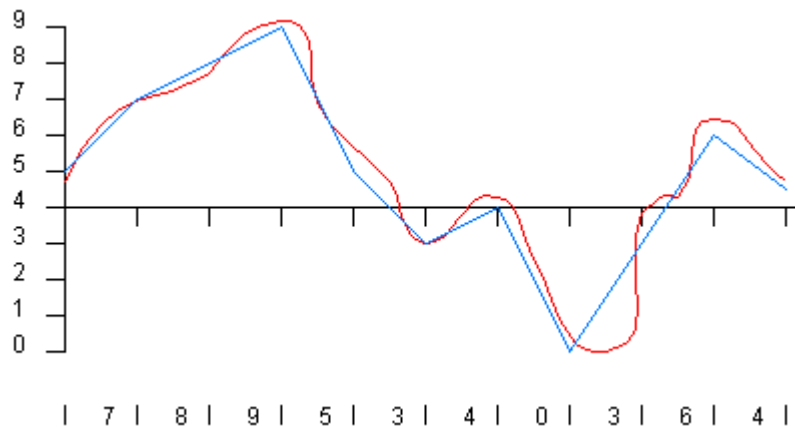
- The **sampling rate** - Controls how many samples are taken per second
- The **sampling precision** - Controls how many different gradations (quantization levels) are possible when taking the sample

In the following figure, let's assume that the sampling rate is 1,000 per second and the precision is 10:



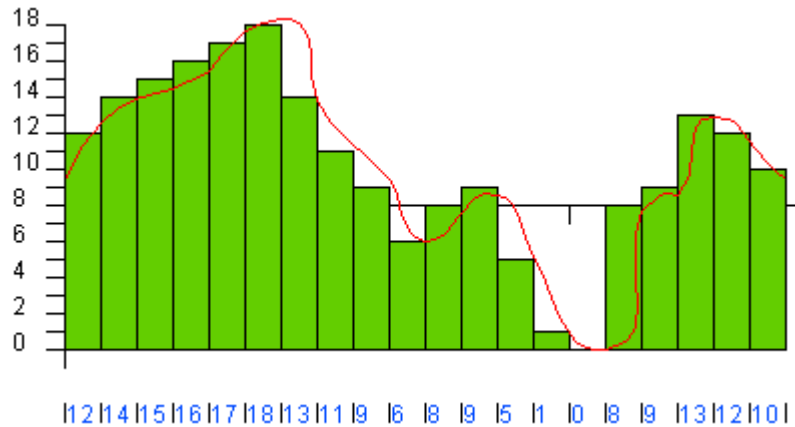
The green rectangles represent samples. Every one-thousandth of a second, the ADC looks at the wave and picks the closest number between 0 and 9. The number chosen is shown along the bottom of the figure. These numbers are a digital representation of the original wave.

When the DAC recreates the wave from these numbers, you get the blue line shown in the following figure:

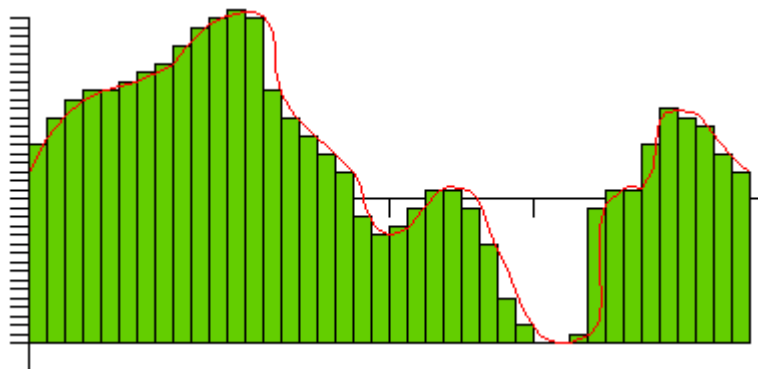


You can see that the blue line lost quite a bit of the detail originally found in the red line, and that means the fidelity of the reproduced wave is not very good. This is the sampling error. You reduce sampling error by increasing both the sampling rate and the precision.

In the following figure, both the rate and the precision have been improved by a factor of 2 (20 gradations at a rate of 2,000 samples per second):



In the following figure, the rate and the precision have been doubled again (40 gradations at 4,000 samples per second):



You can see that as the rate and precision increase, the fidelity (the similarity between the original wave and the DAC's output) improves. In the case of CD sound, fidelity is an important goal, so the sampling rate is 44,100 samples per second and the number of gradations is 65,536. At this level, the output of the DAC so closely matches the original waveform that the sound is essentially "perfect" to most human ears.

4. DIGITAL RECORDING BASICS

In the context of audio, "Analog" refers to the method of representing a sound wave with voltage fluctuations that are analogous to the pressure fluctuations of the sound wave. Analog fluctuations are infinitely varying rather than the discrete changes at sample time associated with digital recording. Simply put, "digital audio" refers to the representation of sound in digital form.

Portable Recorders

At the present time, there are four mainstream types of portable digital recorders:

1. Solid State

Which uses flash memory.

2. Hard-Disk Based

Which records to an internal or external hard drive.

3. CD Recorders

Which records on CD (Compact Disks)

4. Direct-to-computer recording

Which uses an analog-to-digital converter and sends the signal into the computer via firewire or a USB connection.

Recorder inputs

Portable recorders possess a variety of input types:

XLR inputs:



XLR

XLR *inputs* are the highest quality analog inputs. The connection is a “balanced” signal. A balanced connection enables the linking of analog audio devices, including microphones, to a recorder through impedance-balanced cables. Usually associated with professional-level audio equipment, these allow for longer cable lengths and reduce the addition of external noise to the signal. Balanced cables have either XLR or TRS plugs. Professional level digitization will usually

involve balanced outputs on the analog playback device and balanced inputs on the analog-to-digital converter. XLR cables transport a mono signal. Stereo recorders with XLR inputs will contain two XLR inputs. Single Point Stereo microphones will contain a modified version of XLR. In certain cases, this can be a 5-pin connector that splits into two XLR male cables.

TRS (tip-ring-sleeve):

A $\frac{1}{4}$ inch. **TRS (tip-ring-sleeve)** contains the same technology as the XLR connector, but in the form of a $\frac{1}{4}$ inch connector. The balanced TRS connector carries a mono signal and requires balanced inputs.



1/4 in. TRS

This allows for greater compatibility with higher-end recorders. A 1/4" balanced input is fairly rare in portable digital audio recorders.

1/8 in. mini-plug inputs:

The **1/8 in. mini-plug inputs** are typically associated with lower-end recorders. On the connector, one ring signifies a mono jack and two rings signify a stereo jack.



1/8 in. mini-plug

The mini-plug's advantage is portability, but the preamps associated with this input type are usually sub-par. The mini-plug can be temperamental because it does not lock into the microphone input jack and because touching the jack while plugged in can cause static.

Recorder Settings

Sample Rate is the number of samples or “snapshots” taken of the signal and is measured in hertz/second. The higher the sample rate the better the digital representation will be. CD quality equals 44,100 samples per second or 44.1 KHz and is the minimum recommended sample rate for field recordings.

Bit Depth refers to the number of bits used to represent a single sample. For example, 16-bit is a common sample size. While 8-bit samples take up less memory (and hard disk space), they are inherently noisier than 16-bit or 24-bit samples. The higher the bit depth, the better the recording; however, higher bit depths also lead to larger file sizes.

Bit Rate refers to a measurement of digital audio based on the following equation and is usually expressed in kilobits/second

$$\text{Bit rate} = (\text{bit depth}) \times (\text{sampling rate}) \times (\text{number of channels})$$

Again, CD quality equals 16-bit and should be the minimum bit depth used for field recordings. As flash-based storage media dramatically decreases in cost, 24-bit field recordings are beginning to catch on in the field. These 24-bit recordings are becoming the standard bit depth for archival quality. Although many repositories and individuals continue to digitize using 16-bit.

Channels

Single-channel recording is known as monaural or mono recording. Stereo recording involves the recording of two channels (left and right). In interviewing situations, the two channels associated with stereo recording allow the separation and isolation of a channel for the interviewer and the interviewee.

Recording quality and compression

Digital recorders provide a variety of parameters regarding recording quality. Field recordings should be recorded without compression. Compressed recording formats are usually measured by bit rate, which is calculated by an equation involving bit depth, sample rate, and the number of channels being recorded.

For archival purposes, uncompressed is the best way to record. Compressed audio is best utilized for creating web-deliverable files, not for recording the original interview.

File Formats

Uncompressed:

The standard file formats associated with uncompressed recordings are Wave (.wav), Broadcast Wave, and .AIFF. Portable field recorders normally utilize Wave files. AIFF files usually are associated with Mac computer applications. Since both are uncompressed, the quality is the same.

Compressed

MPEG recording will dramatically decrease your data footprint and, thus, increase your recording time. The compression, however, degrades your recording quality. MPEG is ideal for placing audio files on the Web. Indeed Mp3 files have become a standard uncompressed codec almost universally accepted by most computer players. Mp3 files are usually measured by bit rate rather than by sample rate and bit depth.

We will learn in about file formats and audio standards in detail in unit 8.

File Size

File sizes for recordings are calculated by combing bit depth, sample rate, channels, and recording time.

- 16-bit/44.1KHz/Stereo/.wav =635.04 mb perhour
- 24-bit/96 KHz/Stereo/.wav = 2 gb per hour

Recording Levels

- You want to record as strong a signal as possible.
- You do not want your recording levels to clip.
- Higher bit depth recording is more forgiving when boosting the levels of a low-level recording.
- A recording that has average peaks will normally have a greater amount of noise when the levels are boosted to optimal levels. I prefer to stay away from averaging because of the

unpredictability of an interview. If clipping occurs, don't panic and gently back the levels down.

MANUAL LEVEL CONTROL

Manual level control involves the operator adjusting the levels by use of the input level or recording level controls. When recording with manual level control it is best to use a limiter to protect against clipping.

LIMITER

A limiter sets a threshold above which the signal will be gently pushed down in order to prevent clipping.

5. SAMPLING AND QUANTIZATION

Sampling

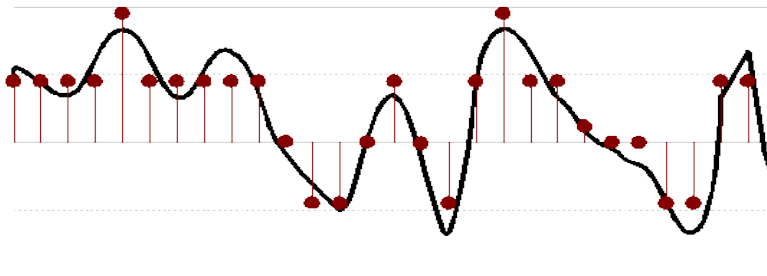
Before digital recording took over the audio and video industries, everything was recorded in analog. Audio was recorded to devices like cassette tapes and records. Video was recorded to Beta and VHS tapes. The media was even edited in analog format, using multichannel audio tapes (such as 8-tracks) for music, and film reels for video recordings. This method involved a lot of rewinding and fast-forwarding, which resulted in a time-consuming process.

Fortunately, digital recording has now almost completely replaced analog recording. Digital editing can be done much more efficiently than analog editing and the media does not lose any quality in the process. However, since what humans see and hear is in analog format (linear waves of light and sound), saving audio and video in a digital format requires converting the signal from analog to digital. This process is called sampling.

Sampling involves taking snapshots of an audio or video signal at very fast intervals, usually tens of thousands of times per second. The quality of the digital signal is determined largely by the sampling rate, or the bit rate the signal is sampled at. The higher the bit rate, the more samples are created per second, and the more realistic the resulting audio or video file will be. For example, CD-quality audio is sampled at 44.1 kHz, or 44,100 samples per second. The difference between a

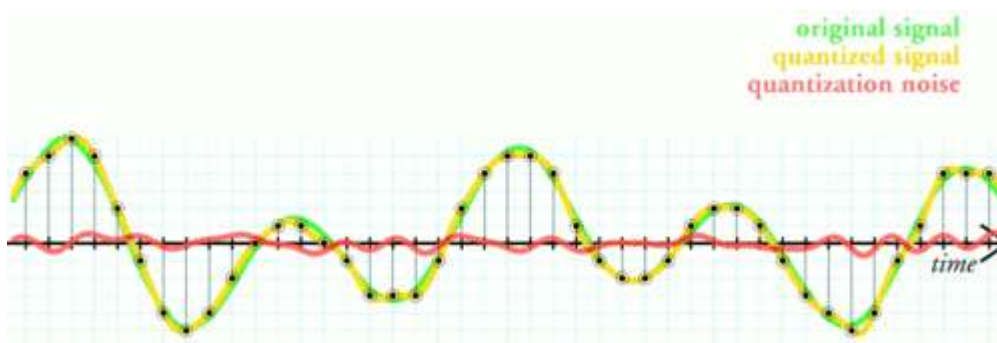
44.1 kHz digital recording and the original audio signal is imperceptible to most people. However, if the audio was recorded at 22 kHz (half the CD-quality rate), most people would notice the drop in quality right away.

Samples can be created by sampling live audio and video or by sampling previously recorded analog media. Since samples estimate the analog signal, the digital representation is never as accurate as the analog data. However, if a high enough sampling rate is used, the difference is not noticeable to the human senses. Because digital information can be edited and saved using a computer and will not deteriorate like analog media, the quality/convenience tradeoff involved in sampling is well worthwhile.



Quantization

The simplest way to quantize a signal is to choose the digital amplitude¹ value closest to the original analog amplitude. This example shows the original analog signal (green), the quantized signal (black dots), the signal reconstructed from the quantized signal (yellow) and the difference between the original signal and the reconstructed signal (red). The difference between the original signal and the reconstructed signal is the quantization error and, in this simple quantization scheme, is a deterministic function of the input signal.



¹Amplitude

Quantization, in mathematics and digital signal processing, is the process of mapping a large set of input values to a (countable) smaller set. Rounding and truncation are typical examples of quantization processes. Quantization is involved to some degree in nearly all digital signals processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms. The difference between an input value and its quantized value (such as round-off error) is referred to as quantization error. A device or algorithmic function that performs quantization is called a quantizer. An analog-to-digital converter is an example of a quantizer.

6. RECORDING PROCESS

The recording process is an exciting journey filled with art, fulfillment, and satisfaction. More specifically, the process is the steps required to make finished recording. The steps in the recording process may vary from project to project. It depends on what your end goal is. For the most part though, the following steps will get you on the path to successful recording.

For the most popular recording-a CD-here is the basic recording process:

- Choose your equipment
- Mic placement & technique
- Record the audio
- Editing
- Do any mixing needed
- Do any mastering needed
- Replicate your discs

Let's look at each one of these steps:

Choose you equipment

While this is not an action step in the recording process, it is utterly important to get the right equipment.

There is a lot of recording equipment available, for a lot of different uses. By defining your needs you can soon decide what you need and what you do not need.

Some of the things you might need are:

- Computer interface
- Computer software
- Digital recording console
- Mixer
- Microphones
- Mic stands
- Microphone preamp
- Headphones

Its looks like quite a list but you will not need all of the things listed.

Mic Placement and Technique

What makes the difference between different recordings? Why do some CD's sound better than others?

The answer lies in various places, but one of the important answers is mic placement.

Mic placement is how you position your microphones in relation to the instrument or voice. Technique is the general strategy used in mic placement. Where you put your mics while recording, will make a big difference in the sound you get. You can just plunk a mic down and record, but a little experimentation can go a long way in getting a better sound.

Record the audio

This is one of the most exciting parts of recording, when you actually record the tracks to disk. Before you do this, make sure you accurately follow the previous steps to get the best sound

possible. You have quality mics, placed in excellent strategic positions. You will be recording with excellent equipment (digital recording console or computer). Now, just push the record button.

Sit back, and enjoy the wonder of it all.

When you are done, go back and listen to it again. Enjoy this feeling. At this stage, you are in the heart of the recording process.

Editing

Once everything is recorded, additional editing can be done in order to get the recording to sound perfect and get the tracks ready for mixing. In editing you can delete or cut the portions where something wrong is recorded or during recording there is some pause or gap. You can apply various sound effects on your recording in editing process.

Mixing

When you are baking a cake, you take the ingredients (like flour and sugar) and mix them together to make the chocolate cake. Everything gets mixed together, and out comes a beautiful cake (or sound). That is what audio mixing is about too. Only now you're dealing with audio tracks instead of eggs and baking powder.

There are a lot of things you can do when mixing besides just combining 4 outputs into 1.

You can:

- adjust the levels of each track
- adjust the panning of each track
- add effects to each track

This is where audio engineers spend most of their time during the whole recording process.

Mastering

Basically, mastering means the process of putting the final touches on your mix to make it ready. It is one of the final steps in the recording process.

This can entail subtle eq, compression, and a touch of reverb, as well as adjusting the levels of each individual track to you can't hear a difference when you're listening to the CD.

The goal is to make the end result work well together.

Replication

The last step in the recording process is replication. It means having copies made of your CDs. Duplication is another word that means almost the same, but with slight differences.

In duplication, the music is burned onto blank CDs. This is usually used for small runs, like 25 - 500 CDs. From 500 on up in quantity, it is usually more cost effective to go the replication route.

In replication, the process is different. Make a glass copy of master copy. From this, stamp out your new CDs. The bottoms are silver and it looks a bit more professional.

7. SIGNAL DISTRIBUTION

Signal distribution is done by an audio distribution amplifier (DA) designed to reproduce the same audio signal to multiple outputs. Unlike an adapter cable that simply splits the audio and produces a weaker signal output, a distribution amplifier ensures the same signal strength that goes into the unit also leaves the unit for each output. Recording studios, media production companies, and complicated home theatre installations typically use an audio distribution amplifier to route a common audio source to many locations.

Though the design of audio distribution amplifiers changes with new technologies, their role of providing the same audio signal to a number of destinations remains essentially the same. They can be used for feeding the same audio to a number of tape duplicators, distributing audio throughout a home or business, or with an audio kiosk that allows a number of headphones to connect to the same audio source. Professionally, they can be found providing the same audio to a variety of components used in a recording studio.



Generally, an audio distribution amplifier is designed according to specific purposes. Some use inputs and outputs that require raw wire to be attached with small retention screws or the same kind of connectors found on speaker components. Others are manufactured using specific kinds of audio connectors, such as balanced three-pin XLR or unbalanced Radio Corporation of America (RCA) jacks.



RCA Jacks



XLR Cables

The internal circuitry typically will reflect the kind of application for which the distribution amplifier is designed. Some distribution amplifiers provide only a few additional outputs, while others provide dozens of outputs. Generally, the more outputs provided by a distribution amplifier, the more the device will cost.

Additional features may include the ability to apply gain, or increase the volume, for the entire unit or for each individual output. An audio distribution amplifier used for professional audio installations typically uses high quality electronic components that maintain the original quality of

the audio signal. Poorly manufactured units may add noise or unwanted distortion that could degrade the audio quality.

Most distribution amplifiers are also designed to include video. Unlike audio, which can be split without noticeable degradation in quality, video cannot be split into two paths without the signal becoming unwatchable. Home theatre distribution amplifiers frequently include both audio and video signal paths in the same unit.

Older units designed for analog audio have been replaced by digital models. Digital distribution amplifiers use digital audio protocols. As audio technology advances, so does the need to bundle outputs from a single or monophonic audio channel, to multiple audio channels.



8. DIGITAL AUDIO RECORDER

It is a device that converts sound, such as speech and other sounds, into a digital file that can be moved from one electronic device to another, played back by a computer, tablet or smart phone and stored like any other digital file. Using the proper software tools, digital audio can also be enhanced and edited like any other digital file.

All compact audio recording devices use flash memory. Some of them use various types of removable memory but some are limited to the built-in memory. Generally, digital audio recorders have both built-in memory and the ability to store files on flash memory cards. Most of the audio recorders have a capacity to record a whole days worth of recording time. One important feature that is absolutely necessary is the ability to connect the device to a computer to transfer audio files. This feature is present in nearly all the current digital audio recorders, but some may lack this feature.



Portable Digital Audio recorders

Audio Quality

There are several common file formats for storing audio recordings. The initial distinction is between devices designed to record music and those aimed primarily at an audio market. It may seem obvious that a device designed to record music would work well with voice recording. That may be true, but not necessarily. Some voice recording devices are designed to block out ambient noise which may make for poor music recording quality especially in a live concert. The Digital Audio Recorder supports MP3 file format.

Some of the other formats are:

- WAV — waveform audio file a standard for IBM and Microsoft
- AIFF – Audio Interchange File Format developed by Apple
- AU — AU audio file format developed by Sun Microsystems.

For most audio only applications, MP3 files are sufficient.

9. DIGITAL AUDIO WORKSTATION

A digital audio workstation (D.A.W.) is an electronic device or computer software application for recording, editing and producing audio files such as songs, musical pieces, human speech or sound effects. DAWs come in a wide variety of configurations from a single software program on a laptop, to an integrated stand-alone unit, all the way to a highly complex configuration of numerous components controlled by a central computer. Regardless of configuration, modern DAWs have a central interface that allows the user to alter and mix multiple recordings and tracks into a final produced piece.



Music production using a digital audio workstation (DAW) with multi-monitor set-up DAWs are used for the production and recording of music, radio, television, podcasts, multimedia and nearly any other situation where complex recorded audio is needed.

Integrated DAW

An integrated DAW consists of a mixing console, control surface, audio converter, and data storage in one device. Integrated DAWs were more popular before commonly available personal computers became powerful enough to run DAW software. As computer power and speed increased and price decreased, the popularity of costly integrated systems with console automation dropped.



An **integrated DAW** consisted of: a control screen, 48-track digital mixer integrated on hard disk recorder including data storage and audio interface.

DAW Software

"DAW" can simply refer to the software itself, but traditionally, a computer-based DAW has four basic components: a computer, a sound card or audio interface, digital audio editor software, and at least one input device for adding or modifying data. This could be as simple as a mouse (if no external instruments are used) or as sophisticated as a piano-style MIDI² controller keyboard or automated fader board for mixing track volumes.

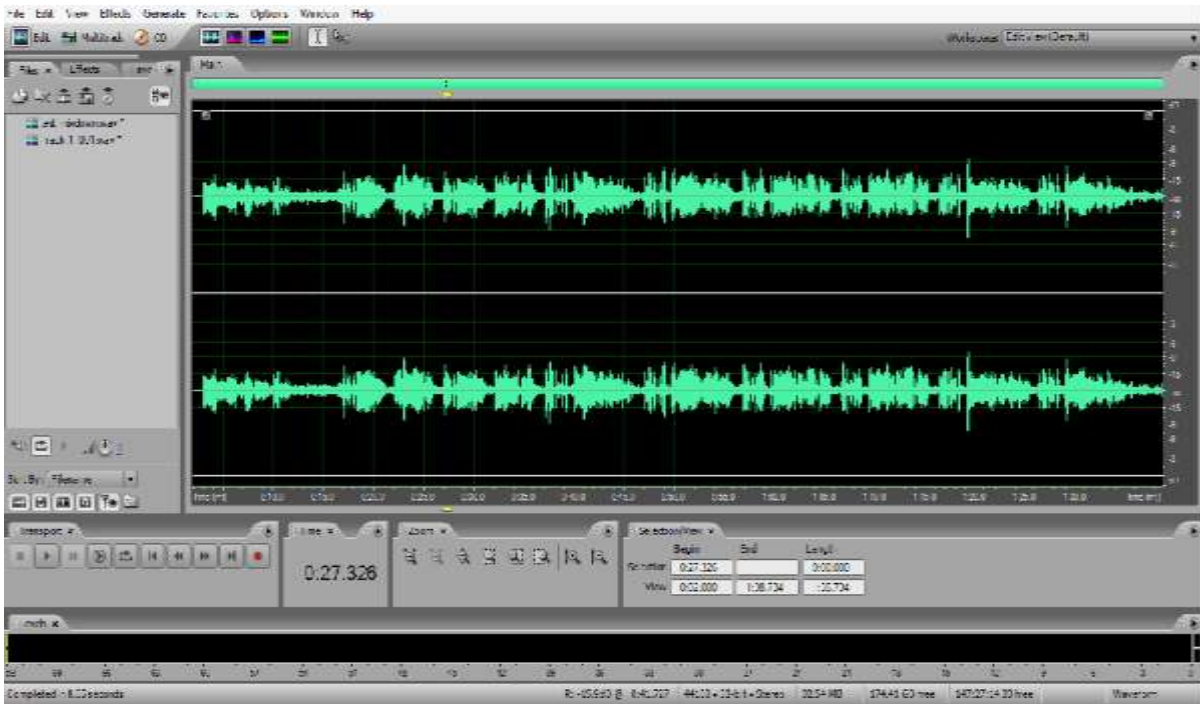
The computer acts as a host for the sound card/audio interface, while the software provides the interface and functionality for audio editing. The sound card/external audio interface typically converts analog audio signals into digital form, and digital back to analog audio when playing it back; it may also assist in further processing of the audio. The software controls all related hardware components and provides a user interface to allow for recording, editing, and playback.

Computer-based DAWs have extensive recording, editing, and playback capabilities (some even have video-related features). For example, musically, they can provide a near-infinite increase in additional tracks to record on, polyphony, and virtual synthesizer or sample-based instruments to use for recording music. A DAW with a sampled string section emulator can be used to add string

MIDI (/ˈmɪdi/; short for Musical Instrument Digital Interface) is a technical standard that describes a protocol, digital interface and connectors and allows a wide variety of electronic musical instruments, computers and other related devices to connect and communicate with one another.

accompaniment "pads" to a pop song. DAWs can also provide a wide variety of effects, such as reverb, to enhance or change the sounds themselves.

Simple smart phone-based DAWs, called Mobile Audio Workstation (MAWs), are used (for example) by journalists for recording and editing on location.



A

screenshot of a typical software DAW (Adobe Audition).

Common Functionality

As software systems, DAWs are designed with many user interfaces, but generally they are based on a multitrack tape recorder metaphor, making it easier for recording engineers and musicians already familiar with using tape recorders to become familiar with the new systems. Therefore, computer-based DAWs tend to have a standard layout that includes transport controls (play, rewind, record, etc.), track controls and a mixer, and a waveform display. Single-track DAWs display only one (mono or stereo form) track at a time. The term "track" is still used with DAWs, even though there is no physical track as there was in the era of tape-based recording.

Multitrack DAWs support operations on multiple tracks at once. Like a mixing console, each track typically has controls that allow the user to adjust the overall volume, equalization and stereo balance (pan) of the sound on each track. In a traditional recording studio additional rackmount processing gear is physically plugged into the audio signal path to add

reverb, compression, etc. However, a DAW can also route in software or use software plug-in (or VSTs) to process the sound on a track.

DAWs are capable of many of the same functions as a traditional tape-based studio setup, and in recent years have almost completely replaced them. Modern advanced recording studios may have multiple types of DAWs in them and it is not uncommon for a sound engineer and/or musician to travel with a portable laptop-based DAW, although interoperability between different DAWs is poor.

Perhaps the most significant feature available from a DAW that is not available in analog recording is the ability to 'undo' a previous action, using a command similar to that of the "undo" button in word processing software. Undo makes it much easier to avoid accidentally permanently erasing or recording over a previous recording. If a mistake or unwanted change is made, the undo command is used to conveniently revert the changed data to a previous state. Cut, Copy, Paste, and Undo are familiar and common computer commands and they are usually available in DAWs in some form. More common functions include the modifications of several factors concerning a sound. These include wave shape, pitch, tempo, and filtering.

Commonly DAWs feature some form of automation, often performed through "envelopes". Envelopes are procedural line segment-based or curve-based interactive graphs. The lines and curves of the automation graph are joined by or comprise adjustable points. By creating and adjusting multiple points along a waveform or control events, the user can specify parameters of the output over time (e.g., volume or pan). Automation data may also be directly derived from human gestures recorded by a control surface or controller. MIDI is a common data protocol used for transferring such gestures to the DAW.

MIDI recording, editing, and playback is increasingly incorporated into modern DAWs of all types, as is synchronization with other audio and/or video tools.

PLUG-INS

There are countless software plugins for DAW software, each one coming with its own unique functionality, thus expanding the overall variety of sounds and manipulations that are possible. Some of the functions of these plugins include distortion, resonators, equalizers, synthesizers, compressors, chorus, virtual amp, limiter, phaser and flangers. Each have their own form of

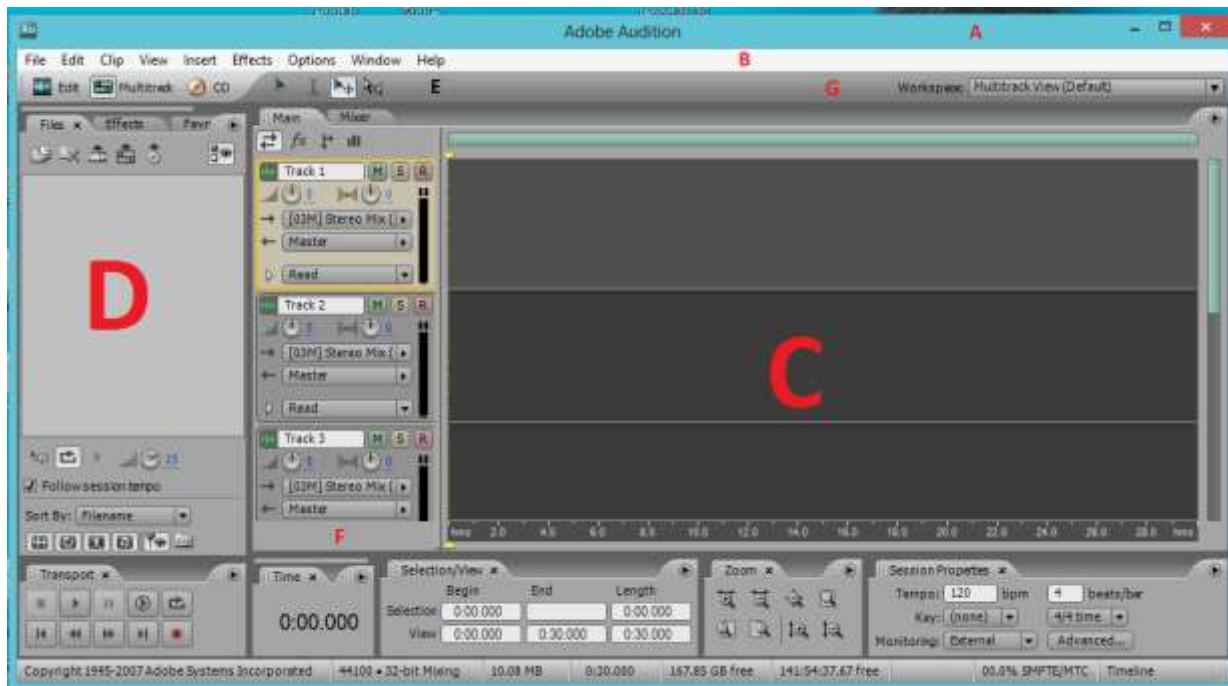
manipulating the sound waves, tone, pitch, and speed of a simple sound and transform it into something different. To achieve an even more distinctive sound, multiple plugins can be used in layers, and further automated to manipulate the original sounds and mold it into a completely new sample.

10. LAUNCHING THE SOFTWARE

For recording purpose we will use software called Adobe Audition. For launching you can simple double click the icon of Adobe Audition on your computer desktop.



It will open adobe audition on your computer screen.



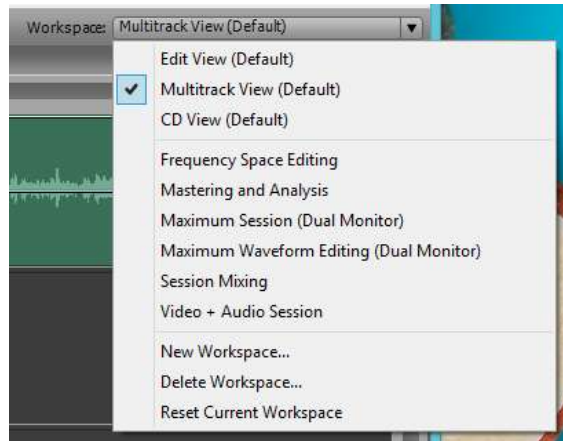
Screen shot of Adobe Audition 3.

A: Title Bar **B:**Menu Bar **C:** Main Window **D:**Files Window **E:** Tool Bar **F:**Multitracks

G: Workspace Menu,**H:** Status Bar,**I:**Other different menu

Defining Different Panel

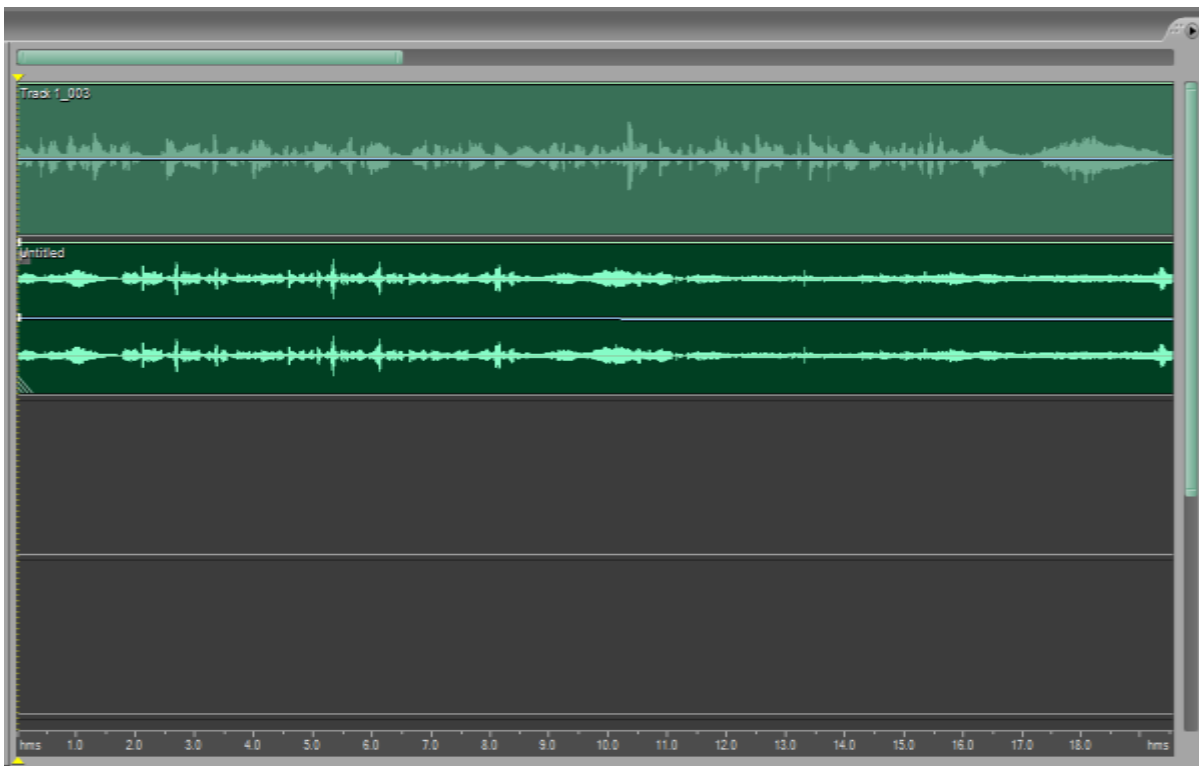
Workspace It is present on the right upper corner of the main window. When you click workspace, you can see Edit View (Default), Multitrack View (Default), CD View (Default). You can chose here view for your file, in which you want to work. Different views have their related options and panels. You can define your own workspace and save it to your hard disk. Next time you want to work with your own workspace you can load it from this menu.



In below picture you can see Title, Menu and Tool Bars



Main Panel in adobe audition.



Armed Multitrack Panel ready for recording the two tracks with different inputs.



Side Panel with loaded files views, effects and favourite menus.



Window menu is very important in menu bar. When you click a drop down menu display, from where you can show or hide different panels like Tools, Mixer, Files, Effects etc. on the workspace.

11. HOW TO RECORD WITH ADOBE AUDITION

Steps involved in recording an audio are as below.

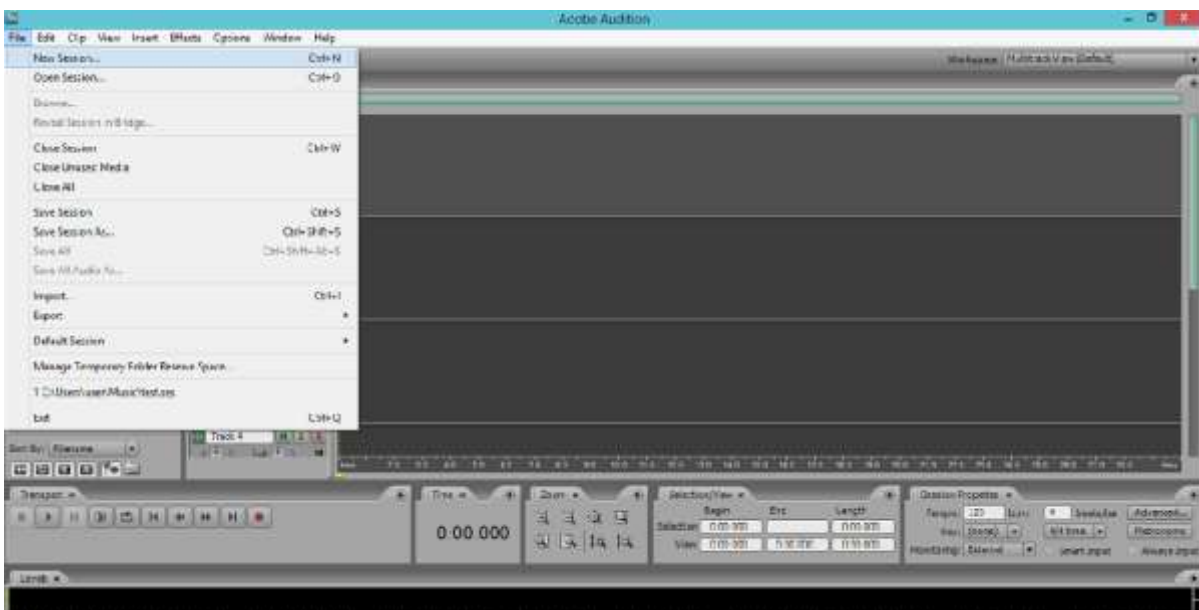
Step 1: Materials

What you'll need for this is:

- A microphone
- A Computer
- Adobe Audition

Step 2: Recording

First things first, you need to open a new multitrack. Click File>New Session.

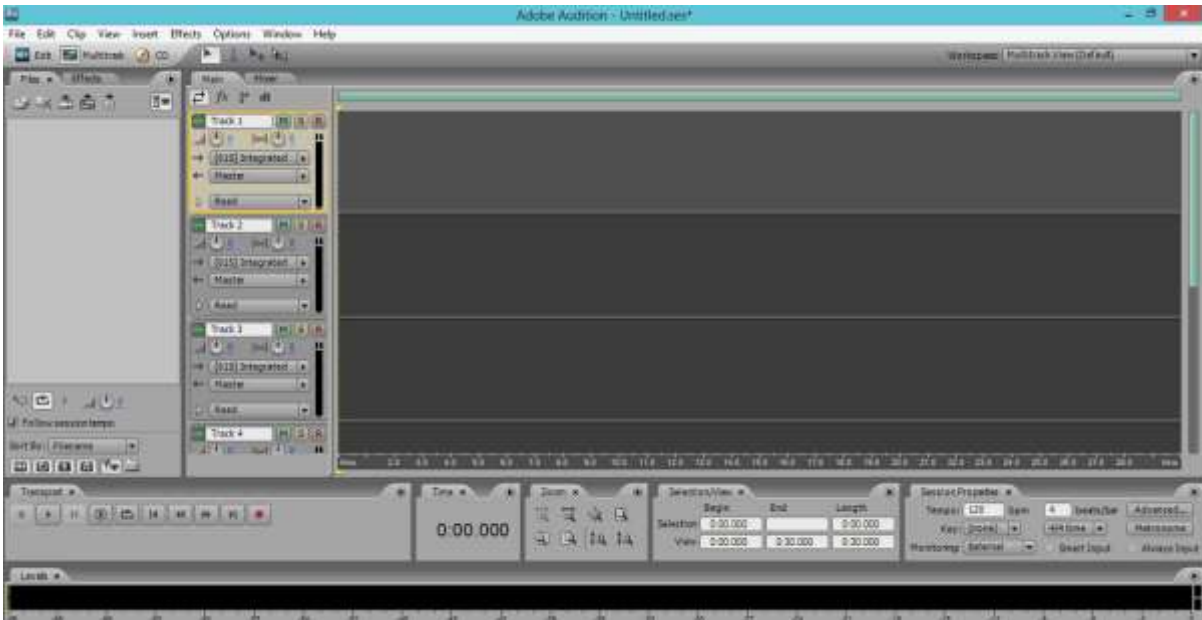


A

new window will open asking you sample rate for the new file you want to create.

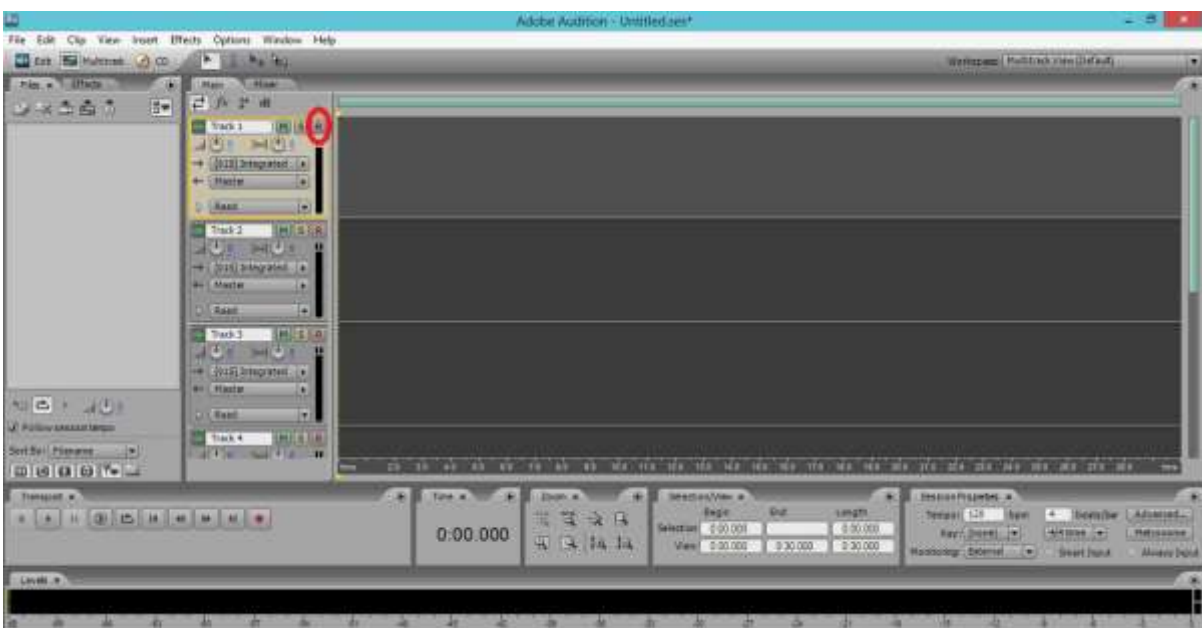


Just give an input for example for audio cd quality select 44100. As you select sample rate and click ok it will open a new multitrack project with at least 6 tracks open simultaneously to work with.

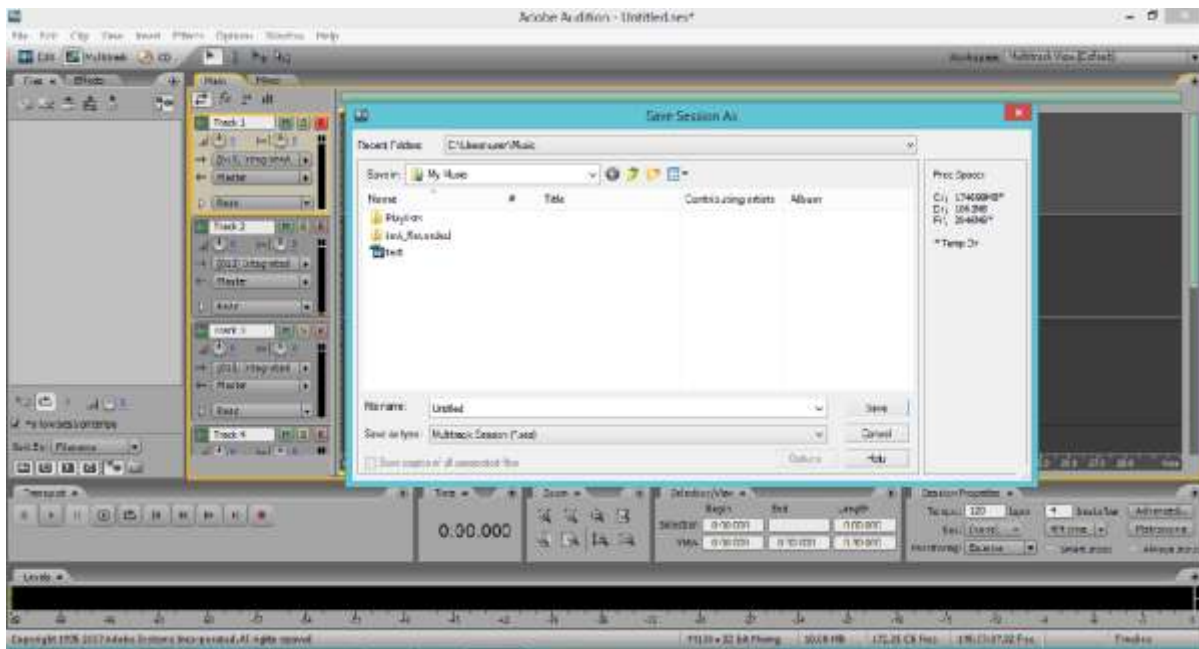


Now that you have your project open you'll need to arm your track to record. To arm it click the R as shown in the pictures above.

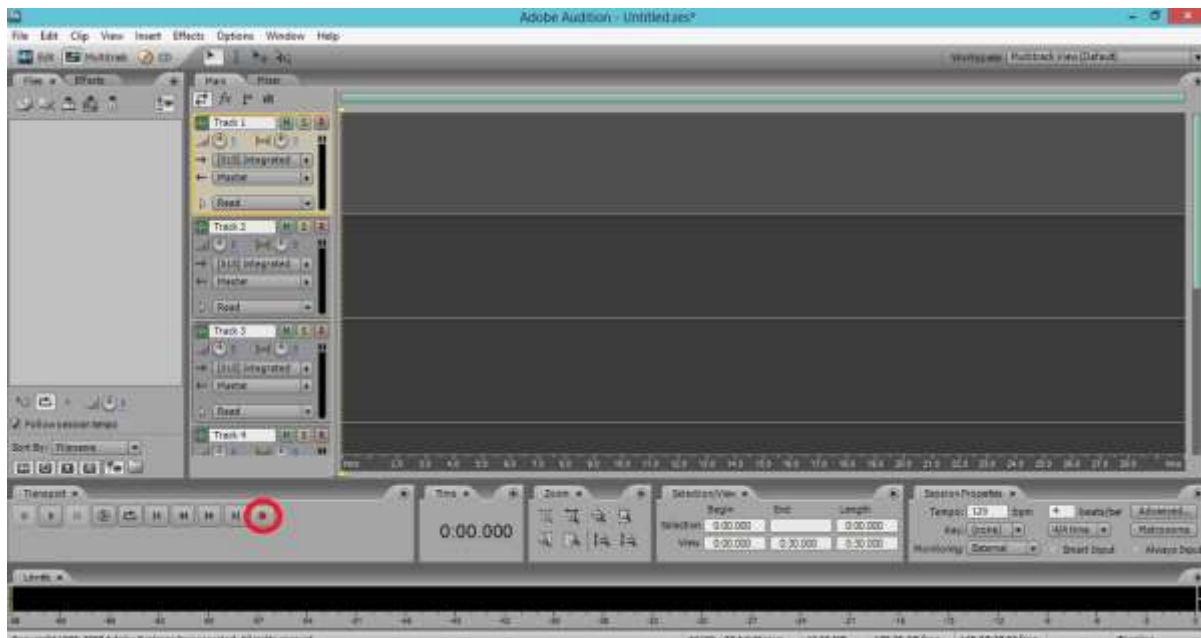
You can have all of the tracks armed at once, which will record on all the tracks.



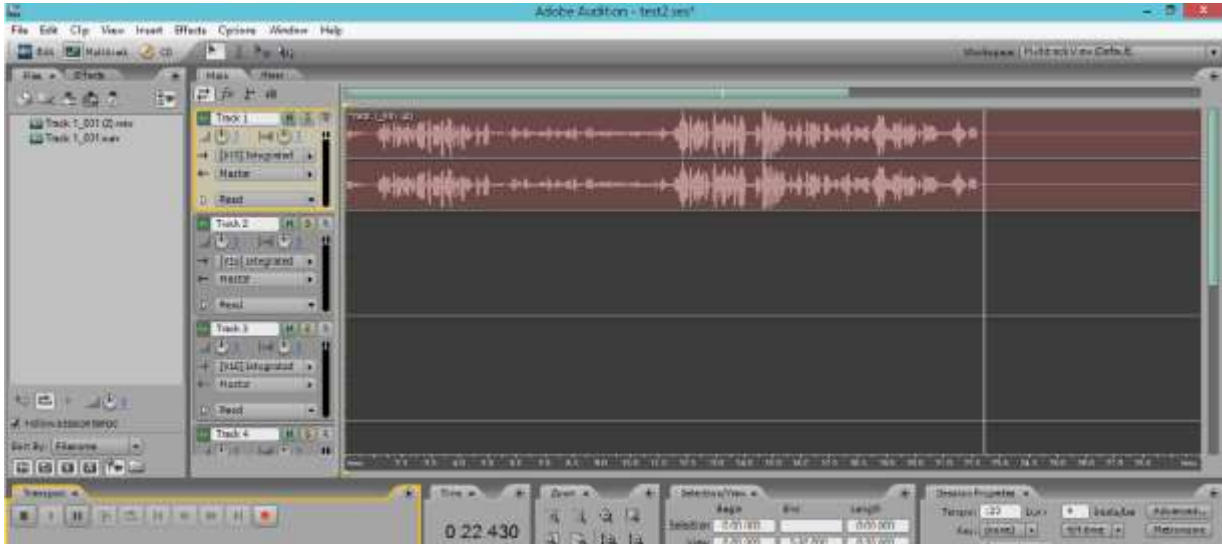
When you click R on the track it will open a new window to save the project. Assign a name to your multitrack project and select the location where you want to save the file. Make sure you chose a file name which can be easily remind you or understand you the project in future.



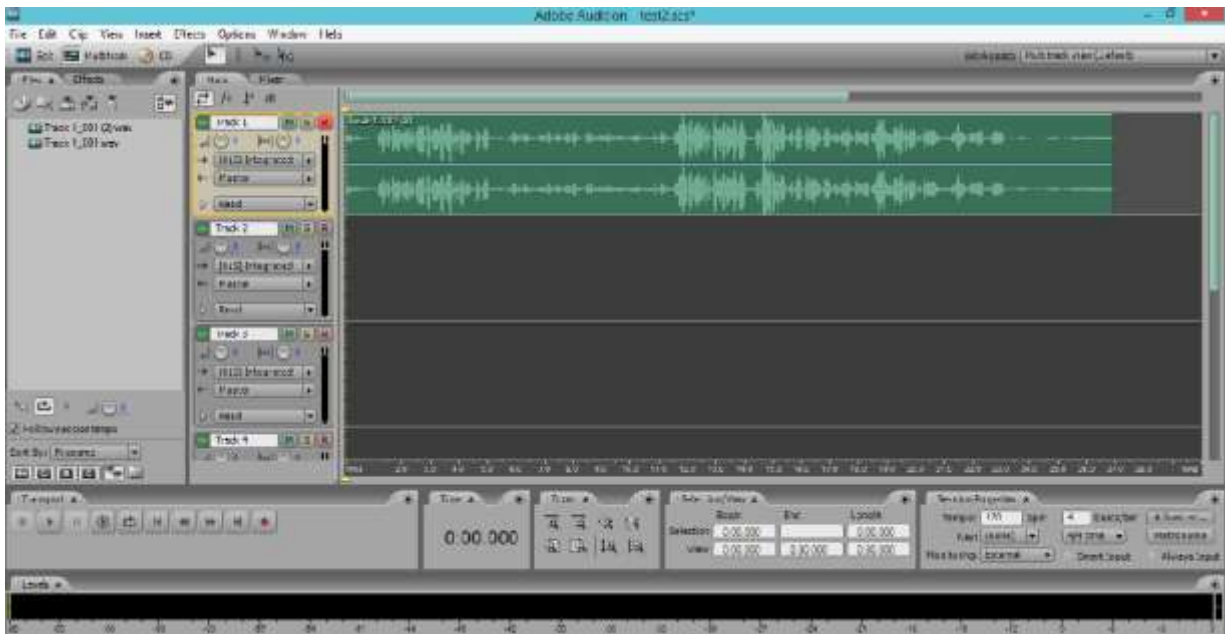
Once you armed the track, it is ready to be recorded on. Click the red recording button on the bottom of your screen as shown below.



Now that the track is recording, the track will turn red, and whenever you speak or it detects audio it will show the sound. The larger the sound bars the louder the sound.



To stop recording simply press the stop button found on the same panel as the record button. Once you've pressed the stop button the track will go from red to green. From here you can play and review your clip, and make any changes you think are necessary.



After making your changes, or if you're happy with what you recorded originally, you should save it. Click File>Save Session. That's it. You have recorded your first file.

Now that you have saved your work, the next step is editing and exporting the project in your desired format. Which you will learn in next units.

SELF ASSESSMENT QUESTIONS

WRITE TRUE / FALSE

1. Audio recordings come in two basic types; analog and digital.
2. CDs and Mp3 files are examples of digital mediums.
3. Vinyl record is an example of digital record.
4. Bandwidth is the ability of a recorded signal to be reproduced at varying degrees of resolution.
5. The signal may be stored as zero and one on a phonograph record.
6. Emile Berliner invented Phonograph in 1877.
7. Thomas Edison invented Gramophone in 1997.
8. The Magnetophone led to the development of multitracking.
9. The sampling rate controls how many samples are taken per second.
10. CD quality equals 44,100 samples per second.
11. Solid State portable recorder uses hard disk.
12. Single-channel recording is known as monaural or mono recording.
13. Analogue recording has now almost completely replaced digital recording.
14. A device or algorithmic function that performs quantization is called a quantizer.
15. The steps in the recording process may vary from project to project.
16. Mic placement is how you position your microphones in relation to the voice.
17. Mastering means the process of editing.
18. Replication is the first step in recording process.
19. Signal distribution is done by an "audio distribution amplifier".
20. All compact audio recording devices use flash memory.
21. The digital audio recorder does not support mp3 format.

22. .wav, .aiff and .au are video formats.
23. An integrated DAW consists of a mixing console, control surface, audio converter, and data storage in one device.
24. The term "track" is still used with DAWs.
25. DAW stands for Dual Audio Workstation.
26. MIDI stands for Musical Instrument Digital Interface.
27. Adobe Audition is the only software for digital audio recording.
28. Microphone is not a necessary item in studio recording.
29. Sample rate for CD quality is 44100.
30. In adobe audition you first arm the track to record.

SHORT ANSWERS.

1. Define Analog and Digital audio?
2. What is Fidelity?
3. Explain the difference between an analog and digital recording?
4. What is an audio bandwidth, give an example?
5. Explain Signal to Noise Ratio?
6. Give three examples of analog recording technology.
7. Who and When invented the Phonograph?
8. How the conversion from analog to digital and digital to analog is done in digital recording?
9. Define sample rate and sample precision?
10. What are the four basic types of Digital Recorders and explain?
11. How many types of audio inputs used in recordings?
12. Explain the two types of audio channels.

13. How many basic types of file formats are?
14. Explain recording levels.
15. Define manual level control and limiter.
16. Give short brief about sampling.
17. What basic steps involved in recording process?
18. If Mic placement is important in recording process then explain why?
19. Explain in your words that editing is important after recording.
20. Explain the process of Mixing an audio.
21. What is Replication?
22. Define Signal Distribution Process.
23. Give a short brief about digital audio recorder.
24. Define Digital Audio Workstation and its types.
25. Which three default views in Workspace of Adobe Audition?

ANSWER KEY TRUE / FALSE

1. True	16. True
2. True	17. False
3. False	18. False
4. True	19. True
5. False	20. True
6. False	21. False
7. False	22. False
8. True	23. True
9. True	24. True
10. True	25. False
11. False	26. True
12. True	27. False
13. False	28. False
14. True	29. True
15. True	30. True

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UNIT 6

EDITING A PROJECT

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Reviewer: Syed Salman Ali Zaidi

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INTRODUCTION

Editing audio digitally offers a multitude of possibilities for manipulating sound. From getting rid of unwanted noises to sonically reshaping a recording beyond any recognisable form to the original, editing can be a very powerful tool. If you tune into a radio show, or play a music CD, what you hear will have been extensively edited. Beginning to edit audio files can be a daunting task and it is worth taking a little time to get familiar with the basics. Editing is a skill and improves with practice. Tasks that seem to take a long time will quickly speed up once you regularly work with digital audio.

When you import audio files into your editing software they can be cut and separated into regions and moved around within an arrangement. When you are happy with the arrangement of all the regions, have finished editing and done all the necessary mixing, you will then be ready to render the arrangement into a final audio file.

Editing tool (Software) playing an important role for editing the audio project. Computer aided recording/editing requires software for the computer which is dedicated for the studio. There are few single/multi channel editing software tools available in the market. Adobe Audition is mostly used for professional audio recording/editing.

When we called interface it means the essential components which helps to communicate the audio signal. Sound card is an integral part of studio hardware. Sound card could be internal or external.

There are different processes in a software (Editing tool). Processes are used according to the project or according to the requirement. Utilisation of software tool is an important factor for any audio project.

It is important to edit your audio project in different scenarios to make your audio productions in a professional way by using the built-in tools like, the trim tool, the scissor tool, sound bite handling etc.

OBJECTIVES

After reading this unit the students will be able to:

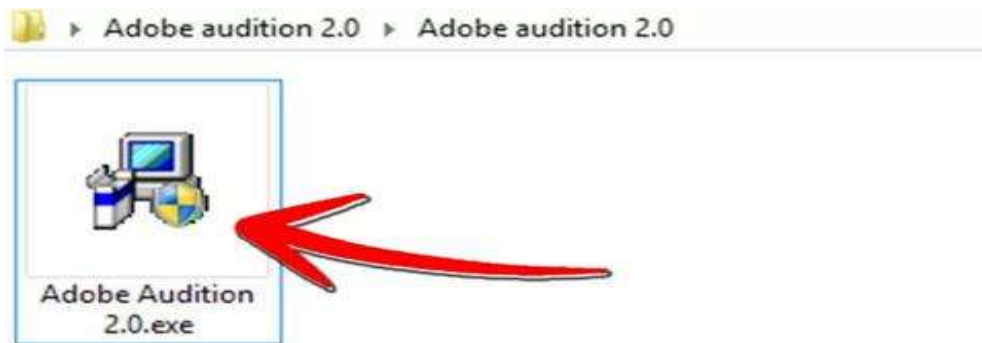
1. It will provide the students a practical professional knowledge for their professional carrier.
2. This course leads the maximum practical skills for those professional students who made their future in the field of audio broadcasting having the knowledge of any audio editing as a project.
3. This is digital era and the audio productions completely deal with the digital technology as the audio productions are the computer aided productions.
4. After the completion of this task the students will be able to handle any type of audio production as a project for their best performance in the field of audio productions.

1. LAUNCHING THE SOFTWARE

How to Install and Activate Adobe Audition

Step 1

Double click the installer



wiki How to Install and Activate Adobe Audition 2.0



Step 2

Run the installation.



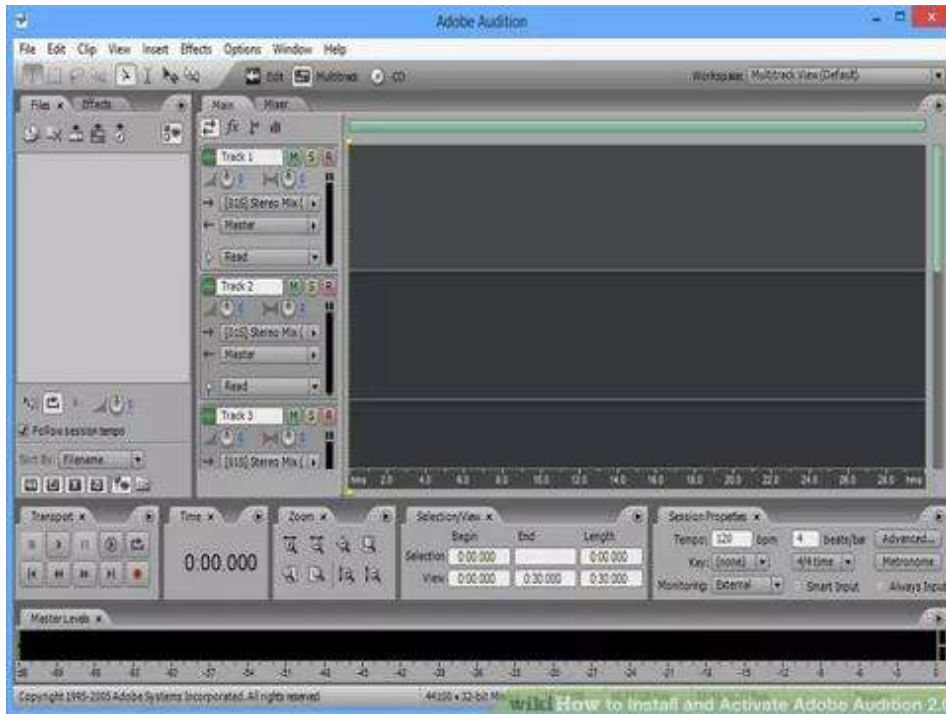
Step 3

Give the serial number. After completing the installation, you will be told to activate it within 30 days.



Step 4

Choose "more activation options" and then choose "activate by phone". You will be asked to give an authorization code.



Step 5

Load adobe audition 2.0 after the preceding steps are completed. Now you are ready to make your professional recordings, mixings and mastering. Enjoy.

2. TOUR OF THE INTERFACE

Audio Interfaces

What is an audio interface?

An audio interface is a piece of hardware that expands and improves the sonic capabilities of a computer. Some audio interfaces give you the ability to connect professional microphones, instruments and other kinds of signals to a computer, and output a variety of signals as well. In addition to expanding your inputs and outputs, audio interfaces can also greatly improve the sound quality of your computer. Every time you record new audio or listen through speakers and

headphones, the audio interface will reproduce a more accurate representation of the sounds. They're an absolutely essential component in computer-based audio production. They're used for recording music and podcasts, and in video post production for recording voice-overs and sound design.



Why would I use an audio interface?

Audio interfaces are used when more a professional level of audio performance is required from a computer, and when one or more professional microphones, instruments and other kinds of signals need to be routed into or out of a computer.



How is an audio interface different from a sound card?

When an audio interface is used with a computer, it acts as the computer's sound card. In this sense, an audio interface is very similar to a consumer sound card. However, the similarities end there. A good audio interface not only enables you to connect an assortment of different analog and digital signals, it also provides a more accurate digital clock and superior analog circuitry that improves the overall sound quality. You can achieve an entirely different level of audio than you would by just using the stock sound card that comes with a computer.

How does an audio interface connect to my computer?

Some audio interfaces connect to computers through common USB ports, while others use more esoteric connections like PCMCIA slots. When you're choosing an audio interface, it's very important to determine the specific kind of port that's available on your computer. This will help you find an audio interface that will be compatible with your computer, and narrow down the number of possible models from which you can choose.



There are lots of audio interfaces available that connect through the USB 1.0 and USB 2.0 ports. There are also many audio interfaces that connect through FireWire ports. If you're using a notebook computer, there are interfaces available that connect through various kinds of Express Card slots, and if you're using a desktop computer, there are models that connect through a variety of PCI card slots. If you know what kind of port you're going to use on your computer, you can start shopping for the ideal audio interface to suit your needs.



Which one is the best port to use to connect an audio interface to a computer?

This depends on your specific needs. If you plan on tracking and overdubbing with multiple microphones or instruments simultaneously, you're better off using a high-speed port such as FireWire. If you don't plan on recording with more than two microphones at a time, you'll likely be fine just using a USB 1.0 interface. The more demanding your needs, the higher the bandwidth of an interface you're going to need. The hierarchy of interface bandwidth speeds from lowest to highest goes from: USB 1.0, USB 2.0, FireWire, PCMCIA/Express Card, PCI.



How many inputs and outputs am I going to need on my audio interface?

That depends entirely on the kind of work you want to do with your audio interface. If you plan on recording with multiple professional microphones, you need to look for an audio interface with multiple XLR microphone inputs. If you're going to be recording voice-overs for video production, you may need an audio interface with only a single XLR input. If you're going to DJ with a computer, it's a good idea to choose an audio interface with four line-level outputs (two outputs are used to send your stereo mix to the house sound system, the other two outputs are used to privately cue songs).



What features does an audio interface need in order to connect professional mics?

If your primary need is the ability to connect microphones to a computer, you should look for an audio interface with XLR microphone inputs. Professional microphones connect with three-pin XLR jacks. XLR connectors are desirable because they lock into place and provide a more secure audio connection. An audio interface outfitted with microphone inputs will typically come with anywhere from one to eight XLR inputs.

Many audio interfaces come with jacks called “combo” inputs. This kind of jack combines a three pin XLR input with a 1/4” TRS input in one socket. Combo inputs tend to confuse people, because they look different than XLR and 1/4” TRS inputs, yet they accept both kinds of plugs. It’s

important to familiarize yourself with combo inputs, so you know what they are when you're deciding which interface to purchase.



What is “Phantom Power” and why would I need it?

Some microphones require a little flow of electricity in order to operate, while other kinds of microphones are capable of picking up sound without any power at all. Certain kinds of microphones run on batteries, while other kinds of microphones are fed power from the device that they're plugged into. It's called “phantom power” when the device that the microphone is plugged into supplies it with electricity. Most audio interfaces that feature mic inputs will also supply phantom power. Because only certain kinds of microphones require phantom power, audio interfaces have a switch to turn it on and off. Phantom power tends to intimidate beginners because it just sounds spooky. Fear not. Using phantom power is about as complicated as flipping a light switch to turn on a table lamp. Besides being called phantom power, it is also referred to as “+48V.”



What are line-level TRS inputs and outputs, and why would I need them?

Line-level inputs and outputs can be very useful; however, to use them properly you must first understand the distinction between mic-level and line-level. Microphones output a very weak signal. The signal is so weak that it needs to be boosted up by a preamp when connected to a mic input. Line-level audio signals are much stronger than mic-level signals, and require no additional amplification. Therefore, line-level signals need a different kind of input than microphones do.

Line-level inputs and outputs on audio interfaces usually show up as 1/4" TRS jacks or 1/4" TS jacks. 1/4" is the diameter of the plug and TRS stands for Tip, Ring and Sleeve; TS for Tip, Sleeve. TRS connections are desirable because they provide a balanced (grounded) connection, which is better at rejecting noise that long cable runs can pick up, or reducing "ground" hum. An example of when you would use line-level inputs is when you're recording the audio from a keyboard. Most professional keyboards have stereo line-level outputs. You can connect these directly to the line-level inputs on an audio interface. When you're connecting studio monitors (powered speakers) to an audio interface, you plug them into the line-level outputs. You can also use line-level inputs and outputs to connect external effects, compressors, limiters and all kinds of stuff. Other connectors include 3/8" mini and RCA (phono) connectors.



What are MIDI ports, and are they important to me?

MIDI ins and outs are found on many audio interfaces. They allow you to send MIDI information into and out of a computer. If you're not familiar with MIDI, just think of it as a simple language that enables pieces of music-oriented hardware to communicate with each other. For example, if you connect the MIDI Out of your audio interface to the MIDI In on a digital piano, you can send a command from your audio software on your computer that tells the digital piano to play a C flat.

People use MIDI ports for all kinds of things. Like in the example mentioned previously, they're often used to connect external MIDI instruments. You could create a MIDI sequence on a synthesizer, and then bring this sequence into your audio software with the MIDI interface on your audio interface. The beauty is that the MIDI sequence is just a series of commands, so when you record it into your DAW you can completely change it and turn it into something new. MIDI ports are also used to connect hardware control surfaces, keyboard controllers and a wide range of other equipment and devices.



What are S/PDIF connectors, and why would I need them?

S/PDIF is simply a digital audio format. Just think of it as a digital version of an analog audio connection. S/PDIF stands for Sony/Phillips Digital Interconnect Format. One of the reasons S/PDIF tends to confuse people is that it's used on different kinds of jacks. The most common kind of S/PDIF connector is a coaxial jack. Unfortunately, this just adds another layer of confusion, because a digital coaxial jack looks exactly like a common analog RCA phono jack. It gets more confusing because a single analog RCA jack can only pass a mono audio signal, while a single coaxial S/PDIF jack can pass a stereo signal. If you weren't confused enough, the S/PDIF format

can also be sent through optical TOSLINK connectors, which look nothing at all like coaxial RCA jacks.

The good news is that you don't have to worry about any of this stuff. S/PDIF connectors are found on many audio interfaces, and they can be really useful. S/PDIF jacks usually come in pairs, with one for input and the other for output. In order to put them to use, you just need other equipment with S/PDIF input and outputs to connect to them. For example, using S/PDIF inputs and outputs is a common way to connect external effects modules.



What is “Direct Monitoring,” and is it something I should have?

Any time you record sound into a computer with an audio interface, you are going to experience some degree of “latency.” If you're not familiar with latency, think of it as the delay in time from the moment you make a command until the moment your command is carried out. If you strike a bell with a mallet, you will hear the sound of the bell ringing instantly. However, when you need to pipe commands through a computer, things don't happen as immediately.

When you plug a microphone into an audio interface and say “Check 1-2-3,” that sound has to travel on a long journey before you can hear it in your headphones:

The sound is picked up by the capsule in the mic;

Then it is sent through the mic cable into the audio interface; It's converted into digital audio and sent to the computer;

The computer routes the digital audio to the DAW audio software; The audio software receives, processes and sends it back out; The digital audio travels back to the audio interface;

The audio interface converts the digital audio back into analog and sends it out to the headphones.

That's a pretty long trip just so you can hear "Check 1-2-3" in the headphones, right? The resulting latency can sometimes distract musicians and make it difficult for them to perform. This is where the direct monitor knob comes in. When you use direct monitoring, you hear the analog audio that is being plugged directly into the interface, as opposed to hearing it after it's been sent out to the computer and back. This nearly eliminates the latency, and makes the musician happier. Direct monitoring is usually only found on USB 1.0 audio interfaces, because their slower speed makes them more latency prone. Unfortunately, this functionality isn't referred to as "direct monitoring" by every manufacturer. Some interfaces have direct monitoring controls, but call it by another name. If you see a USB 1.0 interface with a knob that has "mix" on one side and "computer" on the other, then it has a direct monitoring feature.



What accessories should I get for my audio interface?

Audio interfaces often serve as the heart of a recording studio. Most of the essential tools used in a studio will be connected to the interface directly and indirectly. Of them all, powered studio monitors tend to be the most common tools used with audio interfaces. The cables will vary in

length, depending on your setup, with terminations that are appropriate for each item. These might be ¼" TS to ¼" TS, ¼" TRS to ¼" TRS, ¼" TRS to XLR, XLR to XLR, etc.



With powered monitors in place, you'll be able to properly hear what you're working on. When you need to monitor your work privately, a good pair of studio headphones is an essential tool.



The need to plug professional-quality microphones into a computer is the most common reason people purchase audio interfaces. Naturally, having a few good studio microphones to use with your audio interface is a good idea. Mix it up and buy a variety of mics.

Having a solid dynamic microphone is a great place to start. Adding a large diaphragm condenser microphone will really expand your sonic palette and let you make good use of your phantom-power switches. Small diaphragm condenser microphones are really great for capturing cymbals and various instruments. And a ribbon mic will round out your mic collection with its ability to capture smooth mid frequencies.



The cable that you use to connect the microphone to the interface can make a difference. Spending a little more on a nicely made XLR cable usually proves to be a wise long-term investment (providing that you don't abuse it too much).



The Takeaway

- Audio interfaces expand and improve the sonic capabilities of a computer.
- They add inputs and outputs and can improve the sound quality of your computer. Audio interfaces are an absolutely essential component in computer-based audio production.
- Audio interfaces let you plug pro mics, instruments and other signals into a computer.
- When an audio interface is used with a computer, it can act as the computer's sound card.
- When choosing an audio interface, it's important to determine the specific port that's available on your computer for its use.
- Audio interfaces connect through USB 1.0, USB 2.0, FireWire, PCMCIA/ExpressCard and PCI.
- Professional microphones connect with three-pin XLR jacks.

- “Combo” inputs combine a three-pin XLR input with a 1/4” TRS input in one socket.
- Phantom power is a little flow of electricity that powers condenser microphones.
- 1/4" TRS connections provide a balanced connection, which can provide cleaner-sounding audio.
- MIDI enables music-oriented hardware components to communicate with one another.
- Just think of S/PDIF jacks as a digital version of an analog audio connection.
- ADAT ports are capable of passing eight independent channels of digital audio.
- Word Clock sync is not the same thing as SMPTE time code sync.
- Latency can distract musicians and make it difficult for them to perform.

3. EXPLORING THE PRINCIPLES OF EQUALIZER

Principles of Equalization

Equalization

An equalizer, in its broad description, allows you to boost or cut the volume of specified frequencies. During the mix, equalization can be effectively used in different ways to correct problems that were created during the recording session or from incompatibility among instruments. Equalization can also be used in a creative way in order to produce original effects.

No matter which way you are going to use an equalizer, there are few notions and concepts that you should know before beginning an equalization session. First, equalizers are generally used as inserts on the channel and not as auxiliary sends. Next, you have to be familiar with the most used types of equalizers in a digital audio workstation setting. Among the several types of equalizers available nowadays there are five main categories that have proven to be the most useful in a mixing situation: peak, high shelf, low shelf, high pass and low pass. Table 1 is a list of each equalizer’s description, parameters and common uses. In Figure 1 you can see the symbols with which they are usually indicated.

TYPE OF EQUALIZER

Peak

DESCRIPTION AND PARAMETERS

It allows you to cut or boost frequencies around the centre frequency. Centre frequency: it determines the frequency to cut or boost

Gain: positive gain boosts, negative gain cuts

Q point: it determines the “shape of the bell” or how wide the area around the cut-off point is going to be: the lower the value the larger the bell and vice versa the higher the value the smaller the bell. The Q parameter can usually (but not always) vary from a value of 0.7 (equal to a 2 octaves frequency range) to a 2.8 (1/2 octave)

TYPICAL USES

A Peak Eq. is extremely versatile. It can be used to pinpoint and cut/boost a very precise frequency or it can be used in a broader way to correct wider acoustic problems. It is usually utilized in the middle of the frequency range.

TYPE OF EQUALIZER

High Shelf

DESCRIPTION AND PARAMETERS

It cuts or boosts the frequency at the cut-off and all the frequencies higher than the set cut-off point.

It has only two parameters: the cut-off frequency and the gain.

TYPICAL USES

It is usually used in the mid-high and high end of the spectrum. It can be effectively used to brighten up a track by using a positive gain of 3 or 4 dB and a cut-off frequency of 10 kHz and higher (be careful because this setting can increase the overall noisiness of the track). It can also

be used to reduce the noise of a track by reducing by 3 or 4 dB frequencies around 15 kHz and higher

TYPE OF EQUALIZER

Low Shelf

DESCRIPTION AND PARAMETERS

It cuts or boosts the frequency at the cut-off and all the frequencies lower than the set cut-off point.

It has only two parameters: the cut-off frequency and the gain

TYPICAL USES

It is usually used in the low-mid and low range of the audible spectrum to reduce some of the rumble noise caused by microphone stands and other low end sources

TYPE OF EQUALIZER

High Pass

DESCRIPTION AND PARAMETERS

It cuts all the frequencies below the cut-off point. It has only one parameter which is the cut-off frequency.

TYPICAL USES

It is a very drastic filter. It is often used to cut very low rumble noises below 60 Hz

TYPE OF EQUALIZER

Low Pass

DESCRIPTION AND PARAMETERS

It cuts all the frequencies above the cut-off point. It has only one parameter which is the cut-off frequency.

TYPICAL USES

It is a very drastic filter. It is often used to cut very high hiss noises above 18 kHz. Use with caution in order to avoid cutting too much high end of the track.

Table 1: Characteristics and parameters of the most common types of equalizer

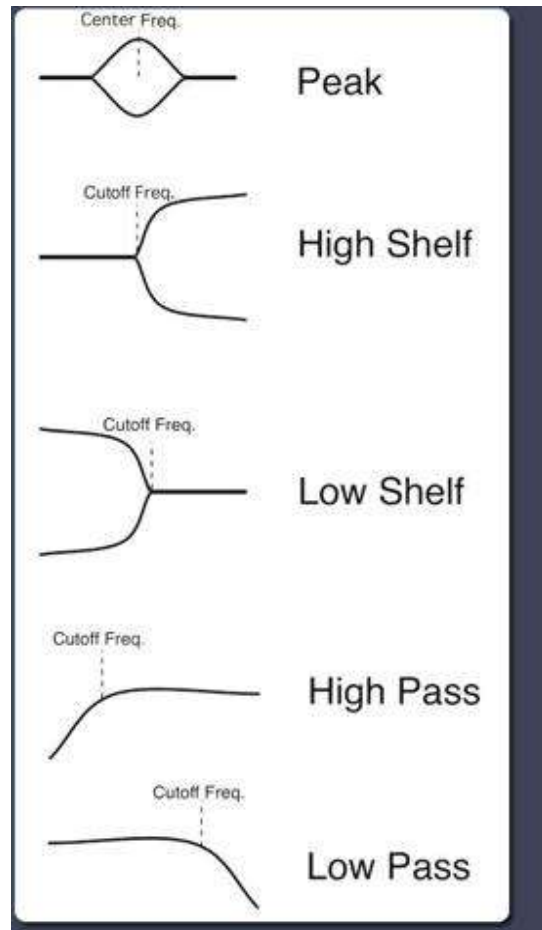


Fig. 1: Conventional symbols for most common types of equalizers

Think First

Remember that equalization is a problem-solving procedure. This means that there's no point in playing around with the settings if you don't know what you want to achieve and how the final result should sound. A good approach to equalization is to listen carefully to the soloed track and come up with a list of things you might want to improve or correct. If you are using a parametric EQ, the next step is to bring up the gain and sweep across the frequency range until you find the frequency range you want to cut or boost. After that, boost or cut as desired.

Keep in mind that when equalizing you will have to make small adjustments every time you add tracks to the mix since the frequencies and respective ranges of the other instruments affect the way an instrument sounds. The most important concept here is to be able to emphasize the characteristic frequencies of the track you are working on and eliminate frequencies that do not enhance its sonic features in any particular way. In fact, you should be able to “carve” a small niche inside the audible range for each instrument and section so that it is clearly intelligible and not masked by other instruments. If the mix sounds muddy and cluttered you should start trying to focus on which instruments contribute to the clutter? Try to use the equalizer to add clarity by gently shifting the centre of each instrument involved so that they do not overlap with each other. As a general rule it is always better to cut than to boost, mainly because the human ear is more used to a reduction than to an augmentation in intensity of frequencies.

While it is hard to generalize, there are a few common settings that are useful to use as a starting point during an equalization session. That summed up in Table 2.

FREQUENCIES

20-60 Hz

APPLICATION

- Cut to reduce rumble and noises related to electric interferences

COMMENTS

It is a good idea to always reduce by 4 to 6 dB this area in order to lower the low frequencies noise

FREQUENCIES

60-80 Hz

APPLICATION

- Boost to add fullness to low frequency instruments such as bass and bass drums

FREQUENCIES

100-200 Hz

APPLICATION

- Boost to add fullness to guitars, French horns, trombones, piano, snares

- Cut to reduce “boomy” effect on mid-range instruments

COMMENTS

This frequency range effectively controls the powerful low-end of a mix

FREQUENCIES

200-300 Hz

APPLICATION

- Cut to reduce low and unwanted resonances on cymbals
- Boost to add fullness to vocal tracks

COMMENTS

Be careful not to boost too much of this frequency range in order to avoid adding muddiness to the mix

FREQUENCIES

400-600 Hz

APPLICATION

- Cut to reduce unnatural “boxy” sound on drums
- Boost to add presence and clarity to bass

COMMENTS

This frequency range can also be effective to boost the low range of the guitar

FREQUENCIES

1.4-1.5 kHz

APPLICATION

- Boost for intelligibility of bass and piano

FREQUENCIES

2.8-3 kHz

APPLICATION

- Boost to add clarity to bass
- Boost to add attack and punch to guitars

COMMENTS

This range can also be used effectively to add clarity on vocal parts

FREQUENCIES

5-6 kHz

APPLICATION

- Boost for vocal presence
- Boost for attack on piano, guitars and drums

COMMENTS

A general mid-range frequency area to add presence and attack.

FREQU

ENCIE

S 7.5-9

kHz

APPLICATION

- Cut to avoid sibilance on vocal and voice
- Boost to add attack on percussions
- Boost to add clarity, breath and sharpness to synthesizers, piano and guitars

COMMENTS

A mid-high range area that controls the clarity and the attack of the mid-high range instruments

FREQUENCIES

10-11 kHz

APPLICATION

- Boost to increase sharpness on cymbals
- Boost to add sharpness on piano and guitars
- Cut to darken piano, guitars, drums and percussions

COMMENTS

High range section that affects clarity and sharpness

FREQUENCIES

14-15 kHz

APPLICATION

- Cut to reduce sharpness on cymbals, piano and guitars
- Boost to add brightness on vocals
- Boost to add real ambience to synthesized and sampled patches

FREQUENCIES

18 kHz

APPLICATION

- Cut to reduce hiss noise
- Boost to add clarity to overall mix

COMMENTS

A delicate high range section that should require drastic positive or negative gain settings only in extreme situations

EQ Rules of Thumb

When equalizing you must pay attention to some of the most common mistakes that sometimes even the seasoned engineer makes. First of all try always to keep your equalization gain parameter at a reasonable level. As a general rule, avoid cutting or boosting by more than 6 dB unless absolutely necessary. If for some reason you see that some of your EQ settings go over this limit try to question why and see if there is a better solution to the problem. The same can be said for situations where you end up boosting (or cutting) several frequencies at the same time that have the only effect of raising (or lowering) the overall volume of the track without really affecting its sonic content. In this case try to bypass the equalizer and experiment with volume changes instead.

You will be surprised how much a small amount of equalization can change, and hopefully improve your mix. Try to hear the sound in your head that you want to achieve through equalization and avoid playing around with the parameters trying to “find” the perfect sound. Leave luck for Las Vegas

4. USING THE COMPRESSOR

Compression is the process of lessening the dynamic range between the loudest and quietest parts of an audio signal. This is done by boosting the quieter signals and attenuating the louder signals.

Effective Ways to Use Compression

Amplitude compression can be one of the most confusing effects for audio engineers.

For some engineers, compression can seem complicated and intimidating. The processing by a compressor depends on many factors including the threshold, ratio, attack time, release time, and knee.

On some compressors these factors can be set by an engineer, while on other compressors these factors are fixed. For some engineers, it may not even be intuitive as to why these controls are helpful to use when performing amplitude compression.

On top of all that, there is more than one way to use a compressor. From subtle dynamic range reduction to extreme tone shaping, a compressor can have several, dramatically different purposes in a mix. This is why it is so important to have all the different controls of a compressor, or at least be able to select the compressor with the correct fixed settings.

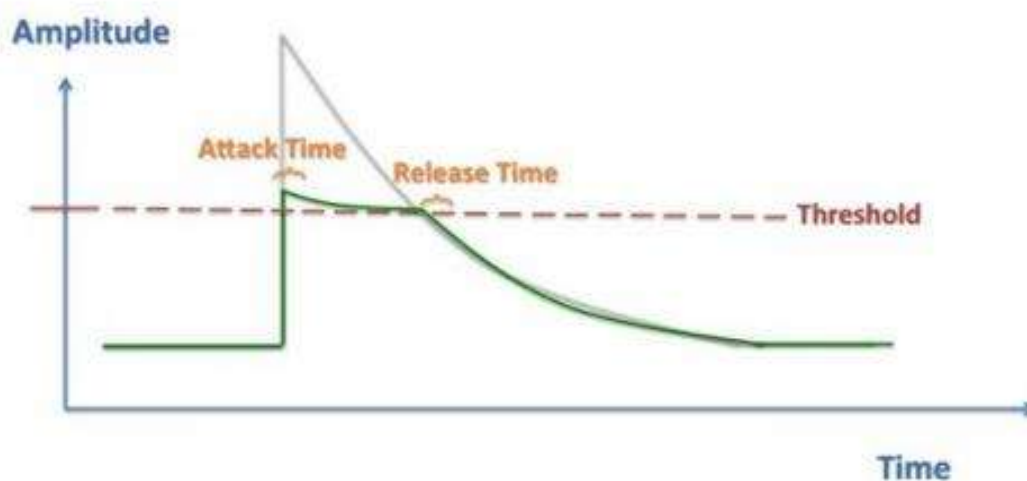
I have heard from many engineers that have had similar experiences with compressors. They think, “Let’s see if a compressor can add anything to this track.” After routing the track through the compressor, they arbitrarily adjust parameters until they stumble upon a desired sound or they give-up and stop using the compressor. This can make the use of a compressor very time consuming.

As much as I wish I could say, “There is no wrong way to use a compressor,” nothing could be further from the truth. Unfortunately, there are many ways to use a compressor incorrectly.

In fact, the incorrect use of a compressor is one of the fastest ways to ruin a mix. For this reason, it is my recommendation to compress with a purpose. In other words, only use a compressor when you have a specific reason. Whether you are trying to solve a problem in your mix or add a textural effect to your mix, you should be able to quickly choose which compressor to use and what settings to start with.

Here are five basic “starting points” for a compressor and their corresponding purpose.

1) Taming Transients: Fast Attack + Fast Release + High Threshold"



Conventional downward compressors detect when a signal’s amplitude is higher than a threshold, and then respond by reducing the amplitude of the signal.

By using a fast attack time and a fast release time, the compressor will respond almost instantaneously when a signal's amplitude crosses the threshold.

Therefore, gain reduction will primarily be applied only when the signal's amplitude is above the threshold. When the amplitude drops below the threshold, gain reduction will not be applied.

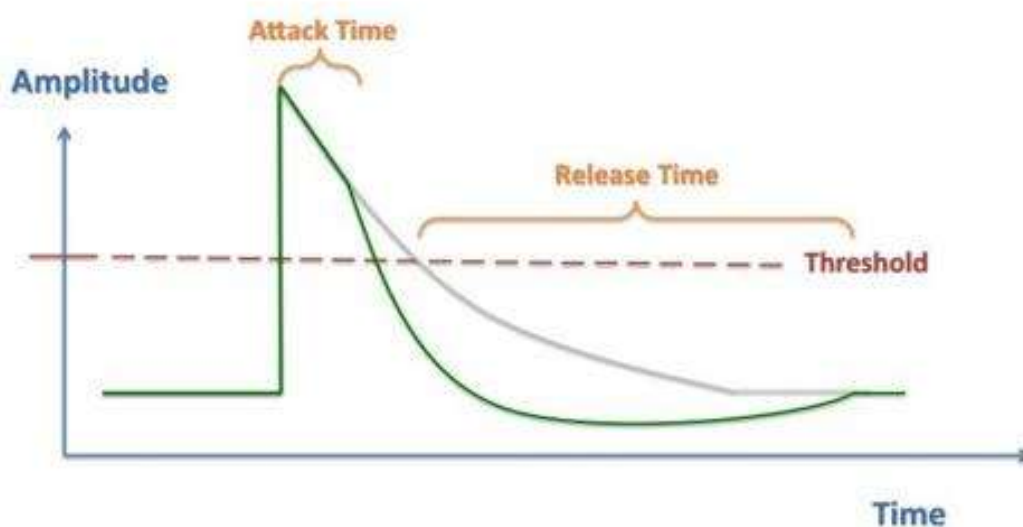
If the threshold is set so that only the attack of a signal triggers the compressor, then these settings on a compressor can be used to tame the transients of a signal and also help pre-vent a signal from clipping.

The threshold can be adjusted to change how much of the transient is changed. Lowering the threshold and increasing the ratio will squash more of the transient, resulting in a higher relative amplitude for the note's sustain. Extreme settings with this technique work well in parallel compression.

This technique works well with an 1176 compressor. The field effect transistor circuit of the compressor means that it can respond very quickly. This technique is great for controlling the attack of drum hits and plucked/strummed notes on a guitar.

This transient tamer technique can be used to perceptually push an instrument back in a mix because the attack does not cut through as much.

2) Enhancing Transients: Medium attack + Synchronized release + High Threshold



The complement of the transient tamer is the transient enhancer.

Rather than using the compressor to reduce the amplitude of the signal's attack, the compressor can be used to reduce the amplitude of the signal's sustain.

The threshold should still be set so that gain reduction is triggered only by the transient. However, the attack time should be increased so that gain reduction is delayed until after the attack passes through the compressor. By delaying the gain reduction, the sustain of the signal will be reduced. With this technique, the compressor's release can be set so that gain reduction continues until the start of the next note.

Therefore, the release time should be set to be synchronized with the notes of the instrument and groove. If a snare drum is playing on the 2-4 backbeat, gain reduction of the previous note should return to 0 dB just as the next note is played.

This technique is great for drums, bass, guitar, piano, and anywhere else you want to sharpen the instrument's attack. Changing the compressors ratio from low settings (2:1) to high settings (8:1) will change how much the sustain is squashed.

To dial in these settings on a compressor, it can be helpful to use a compressor with fine control over the attack time and release time.

The following is an order of operation that can be used to quickly dial in the appropriate settings based on your song.

Start with fast attack/fast release and set the threshold to detect the signal's transients.

Slow down the release time so that the gain reduction lasts until just before the start of the next transient.

Finally, slow down the attack time until you hear the transient pass through.

There are also many compressors that have fixed settings that perform this technique automatically.

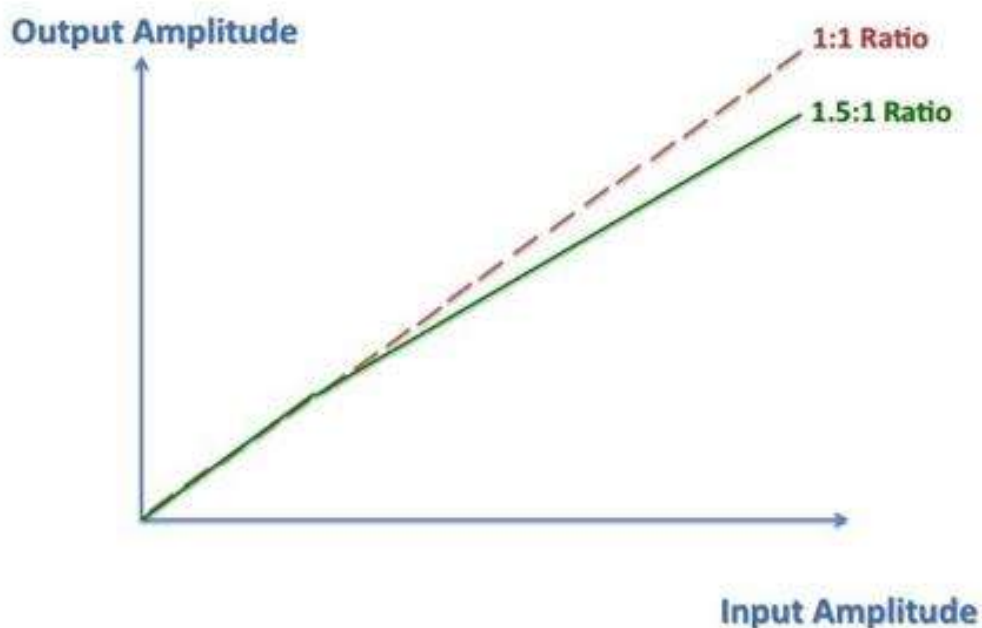
The dbx 160 and LA-2A are examples of compressors that have a fixed attack time, which allows a signal's transient to pass through.

The dbx 160 is great for drums, while the LA-2A is great for bass. The release time of the LA-2A is program dependent, meaning that it will synchronize, in some ways, with the instrument's notes automatically.

On an SSL console, the compressor has a switch for the attack time. In one position the attack time is fast and in the other position the attack time is slower. Essentially the switch assumes that an engineer only needs two settings: either an attack time that will compress a transient or an attack time that will allow the transient to pass through. In general, this can be a helpful way to think about using a compressor's attack control. Even though a variable knob can be used to set a wide range of attack times, it can be much simpler to ask the question, "Transient? Yes/no?" when choosing your attack setting.

This transient enhancer technique can be used to perceptually push an instrument forward in a mix because the transient will cut through more.

3) Transparent Dynamic Range Reduction: Low threshold + Low Ratio



In theory, a compressor is an amplitude processor that can be used to reduce (compress) the dynamic range of a signal. This can be accomplished by reducing the amplitude of the parts of a signal that have high amplitude.

However, a compressor can also be used to increase (expand) the dynamic range of a signal, accomplished with the transient enhancer technique.

Nonetheless, many times the purpose of using a compressor is to actually reduce the dynamic range of a signal. With the transient tamer technique, the threshold is conventionally set so that the compressor only responds to the transient portion of a signal. Another technique to reduce a

signal's dynamic range is to set the threshold so that the compressor responds to both the transient and sustain portions of a signal.

If a low ratio is used, this can be a very transparent method to reduce dynamic range. One place to start would be to use a ratio of 1.2:1 or 1.5:1.

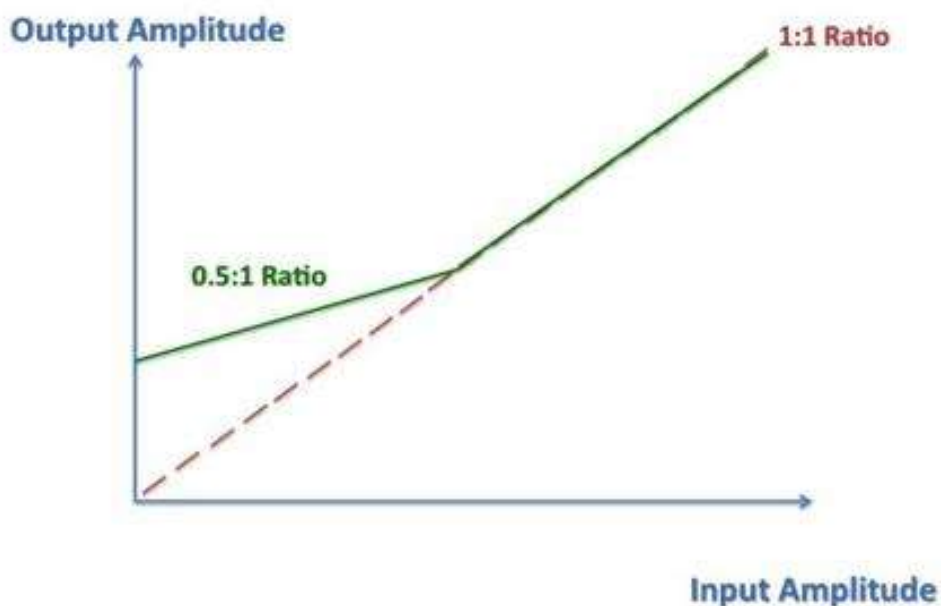
This technique requires a compressor with a variable ratio that goes down to 1:1. After the low ratio is set, the threshold can be reduced so that nearly the entire signal triggers the compressor.

With this technique, the effective gain reduction for portions of the signal that are slightly above the threshold will be smaller than the effective gain reduction for portions of the signal that are significantly above the threshold.

As an example: with a ratio of 1.5:1, if an input signal is 6 dB above the threshold, the output will be 4 dB above the threshold. The effective gain reduction is 2 dB. If an input signal is 15 dB above the threshold, the output will be 10 dB above the threshold for an effective gain reduction of 5 dB.

Therefore, this technique is an effective way to slightly reduce the gain of low amplitude signals and gently reduce the gain of high amplitude signals. With such a low ratio, compressors can be very transparent.

4) Lengthening Sustain: Conventional Upwards Compression



The dynamic range of a signal can be reduced by decreasing the amplitude when it is above the threshold (downwards compression).

Additionally, the dynamic range of a signal can be reduced by increasing the amplitude when it's below a threshold. This is called upwards compression.

This mode of operation was not typically found on analog hardware compressors. However, it is becoming more common on digital software compressors.

This type of compression is helpful if your purpose is to change the amplitude of the quieter parts of your signal. One example would be if you would like to increase the amplitude of an instrument's sustain.

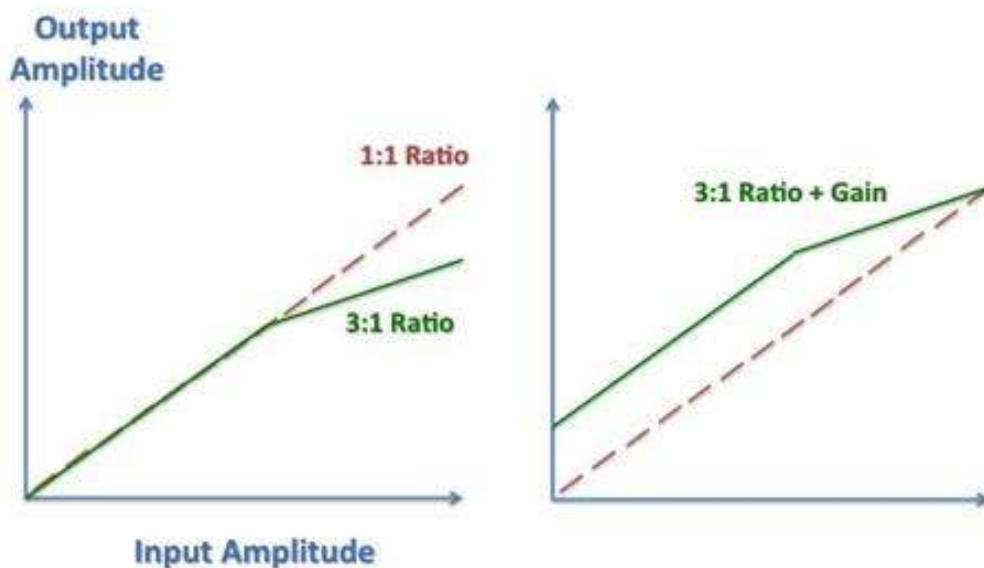
Common places to use this technique are for vocals, lead instruments, and bass. The result of upwards compression is a signal that has a much more consistent level. Therefore, it can help a vocal part stay out front of the rest of mix. It can also help make sure a bass sits in the context of a mix without ever getting too loud or too quiet.

When it comes the compressor's ratio, upwards compression uses a ratio less than 1:1.

As an example: if the ratio is 0.5:1, this means an input signal 4 dB below the threshold will be increased so that the output is only 2 dB below the threshold. When actually setting the ratio for upwards compression, a little goes a long way. Even settings of 0.9:1 can have a significant impact.

This type of compression is found on many software compressors including Waves Maxx-Volume/MV2 and several iZotopeplug-ins.

5) Upwards Compression: Downwards Compression + Gain



Lastly, another version of upwards compression can be accomplished using a combination of downwards compression and make-up gain.

Therefore, if your compressor does not have a specific setting for upwards compression, you can still achieve a similar result.

Downwards compression can be used to reduce the amplitude of the signal when it is above a threshold.

After downwards compression is performed, there is additional headroom to increase the gain of a signal without clipping. By applying a constant make-up gain, both the amplitude above the threshold and below the threshold will be increased.

If performed carefully, this type of compression can actually increase the amplitude of the portions of a signal with low amplitude, while having almost no net change to the amplitude of the portions with high amplitude.

This type of compression can be used very subtly or very dramatically. Try it both in series and in parallel. Experiment with this technique on everything: drums, vocals, piano, etc.

Almost any compressor can be used with this technique. You could try “All Buttons Mode” on an 1176 for your drum room mics, gentler settings with a CL-1B on vocals, or a Fair-child on the mix buss.

Conclusion

Compressors are one of the most versatile and powerful processors for an audio engineer to use.

Even though there are many parameters and possible combinations of parameters, a compressor can be mastered for use in common mix situations.

After learning several basic “starting points,” it is only a matter of making minor adjustments to make the compression more/less audible.

When you compress, make sure you compress with a purpose.

5. USING THE EFFECT GENERATOR

An Effect changes the audio in some way.

A Generator creates new audio, either in an existing track or in a new track. For further details please check this link:

http://manual.audacityteam.org/man/index_of_effects_generators_and_analyzers.html

6. CUSTOMIZING THE INTERFACE

How to Customize VLC Media Player Interface

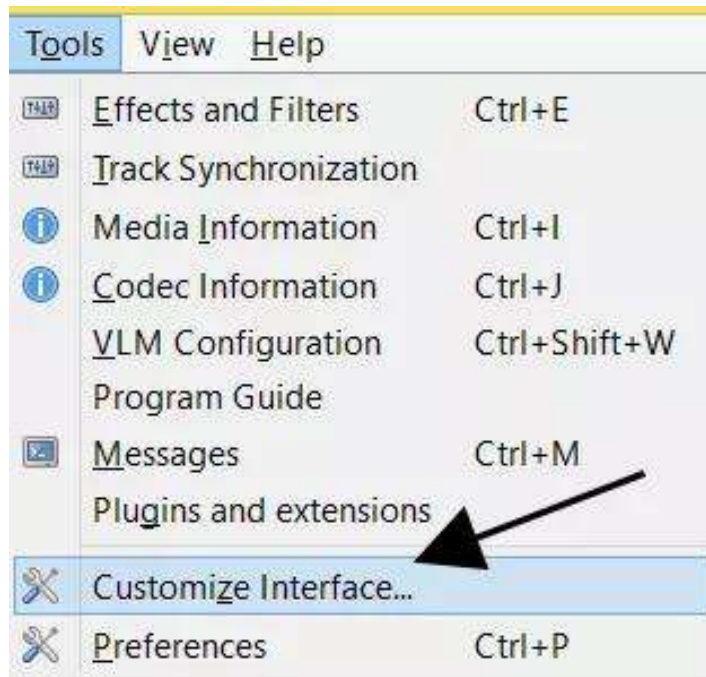
When it comes to VLC, a lot is customizable in terms of the minimally present user interface. You can easily change where the play, pause, stop, next, previous and other video/audio control buttons are placed. You can also change the position of the individual buttons. Additionally, you can even change the position of all the control button set in-regards to where it shows up in the player. All you have to do is go to Tools > Customize Interface.

From the customize interface or Toolbars Editor option you can drag the individual buttons around, add new buttons and remove the ones that you don't need. You can also configure the time toolbar and customize the full screen buttons separately. Additional customization options in terms of the button sizes and designs are also available.

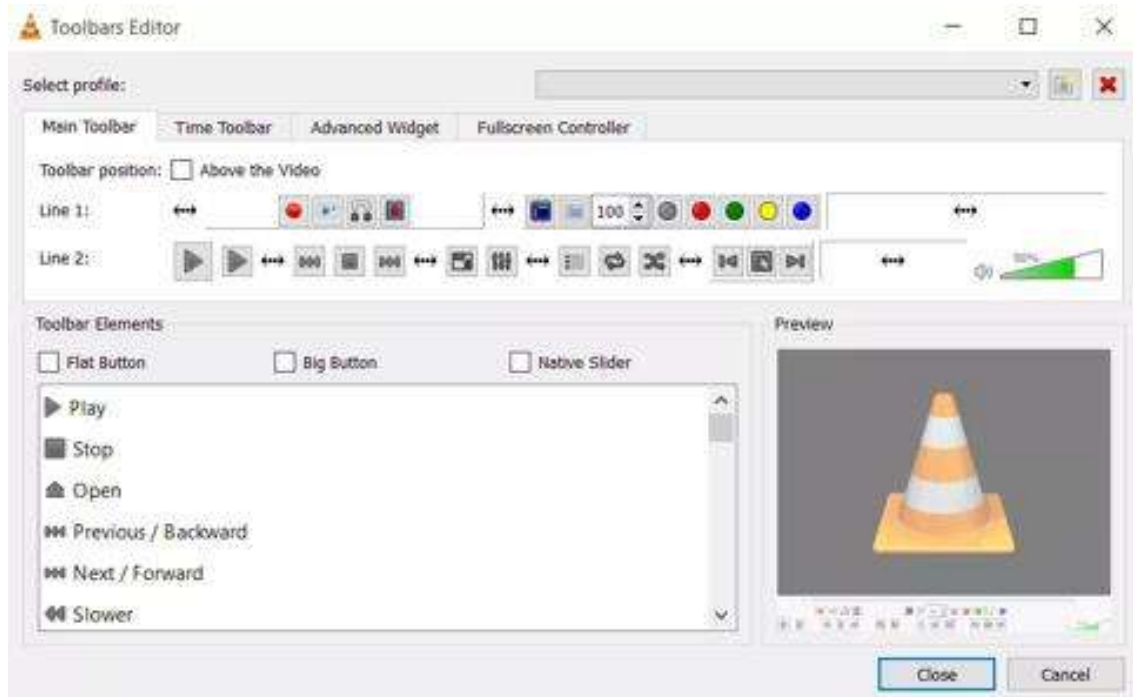
Here are the detailed steps/explanations to customize your VLC Media Player Interface and buttons:

To Access Toolbars Editor

In the menu bar or via the right click menu, select Tools > Customize Interface.



The Toolbars Editor will open.



In the editor, you will see different tabs:

1. Main Toolbar: This is the toolbar that is displayed when VLC is running in Window mode. You can change the position of the player controls to place it above the video by checking the appropriate box. There are two lines of controls and line 2 has the most commonly used buttons.

2. Time Toolbar: This one allows you to customize the time toolbar that shows the position of the video or audio that you are currently playing.

3. Advanced Widget: This is the advanced widget that is displayed when View > Advanced Controls is activated. You can place buttons that can record, cut, loop or navigate frame by frame. These are the buttons that aren't frequently used.

4. Full screen Controller: These are the controls that show up when your video is playing in full-screen mode. You can have a different set of controls in full screen.

Not that there is a Select profile option above everything else. This one allows you to save your interface configurations as profiles. You can switch between profiles to switch to different interfaces easily.

To Edit the Interface

Use your mouse and simply drag the buttons in the toolbar editor.

Choose the tab for which you want to change the controls.

To add new buttons drag them from the Toolbar Elements to the main toolbar, full screen controller, advanced widget or time toolbar.

To edit/move the existing buttons click and hold using your mouse, and drag them to where you want to.

To remove buttons just drag them outside the toolbars editor.

Click Close to save your changes or if you messed up, click Cancel.

See the Preview area to get an idea of what your player looks like after the changes are saved. Make use of the options to display the toolbar under or above the video. We are used to the toolbar being placed below the video but you never know whether you'd like it placed above; unless you try it.

Note:

If you are trying to add new buttons to the main toolbar, you'll want to add it to line 2 because that's your usual control toolbar.

Make sure to create a new profile or make use of saving and retrieving using the profile feature. This prevents you from messing it all up.



VLC Options Related to Displaying in Windows System Tray and Task Bar

7. UTILISATION OF TOOLS

5 Tools for Cleaning Up Audio in Adobe Audition

Integrate Adobe Audition into your post production workflow Utilize Audition's powerful tools for fixing common audio problems like back-ground noise, hum, clipping, clicks and pops.

Noise Reduction in Audition

Adobe Audition has powerful noise reduction tools that be accessed in the Waveform Editor. If you are in a Multitrack Session, double click on a track to go into the Waveform Editor.

Click and drag to select several seconds of background/ambient only sound. The more time you have to sample the better your results will be. Make sure you do not select any audio with voices or other noises

Go to Effects > Noise Reduction (process). Click “Capture Noise Print” and then “Select Entire File”.

Click “Noise Only” to hear what you are removing (deselect it before you click apply). Click the green button on & off to toggle the effect as you adjust the “Noise Reduction” & “Reduce by” sliders.

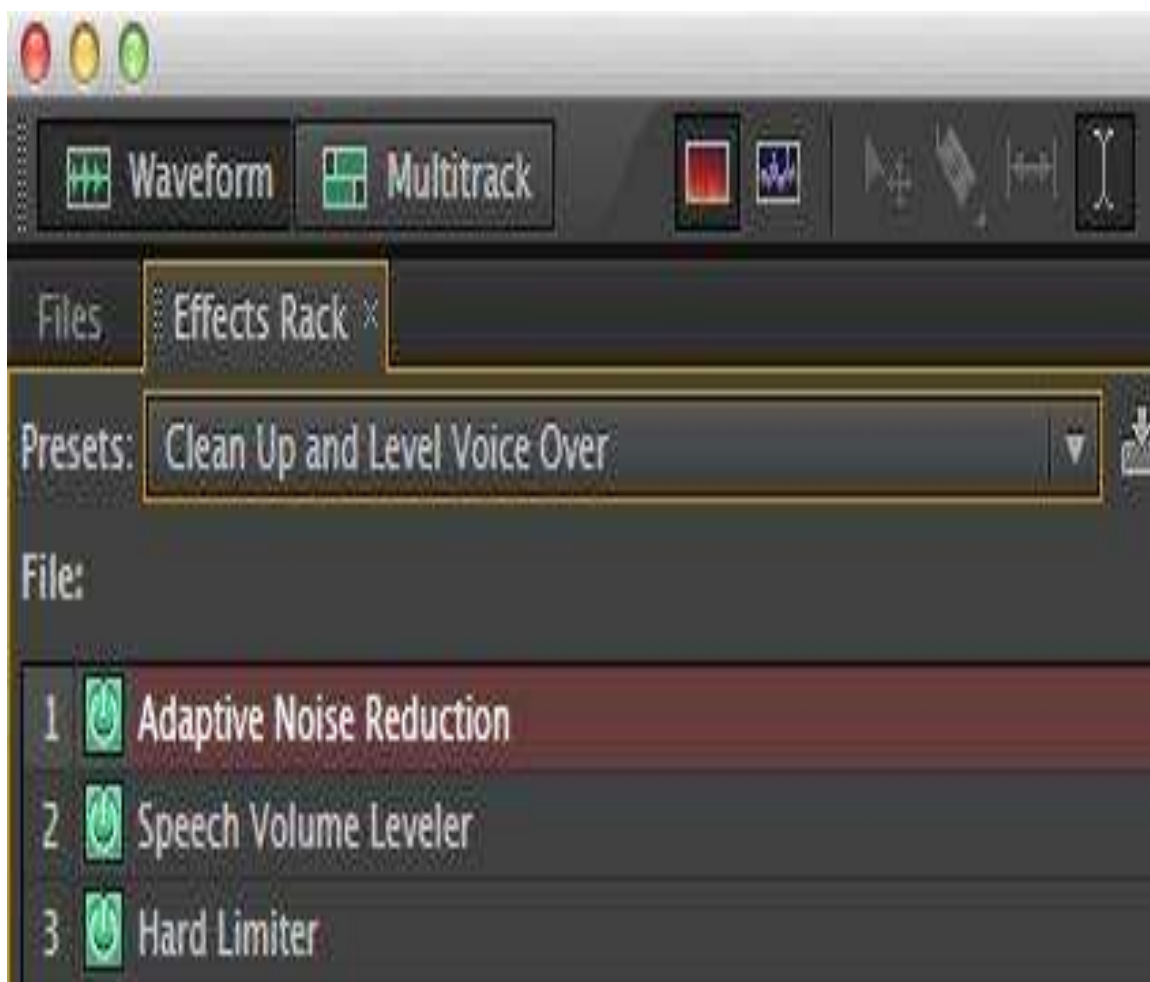
If you prefer shortcuts, use Shift +P to save a noise print and CMD/CNTRL/Shift + P to open the “Noise Reduction” Effect. I suggest making shortcuts for effects you commonly use (do this by accessing the shortcut editor in the menu bar, Edit >Keyboard Shortcuts).



Adaptive Noise Reduction

Adaptive Noise Reduction automatically learns what noise is, as long as you have back-ground noise before people start speaking. To take advantage of this tool, it is a good habit to always record 4-5 seconds of audio before your talent starts speaking.

In Adobe Audition, you can also combine Adaptive Noise Reduction with other effects in the Effects Rack (which you can't do with standard Noise Reduction). It is part of several presets like "Clean up and Level Voice-Over" that can help you get started if you are new to audio effects.

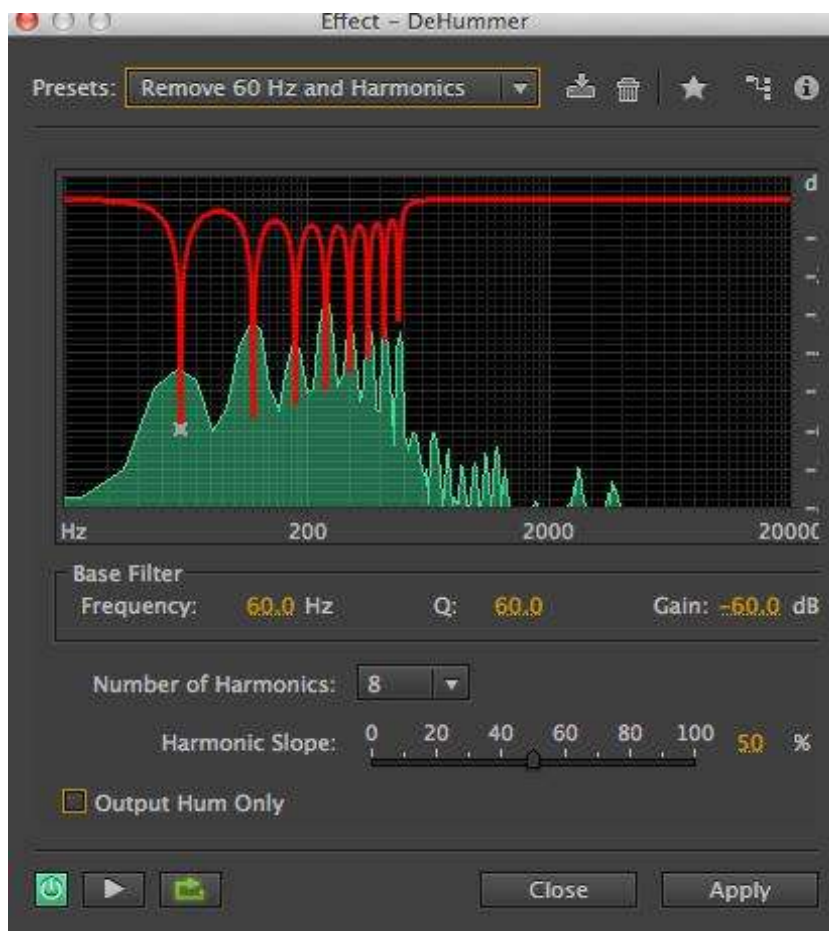


Remove Hum in Audition

This Adobe Audition effect comes in handy if you are doing a lot of location filming where you can't control the production environment.

Use this to remove AC hum (lights, power lines, and electronics). In my example I was picking up hum from an Xbox 360 in the room.

Go to Effects > Noise Reduction/Restoration > Dehummer. Select your preset based on country. I'm in the states so I picked 60Hz.

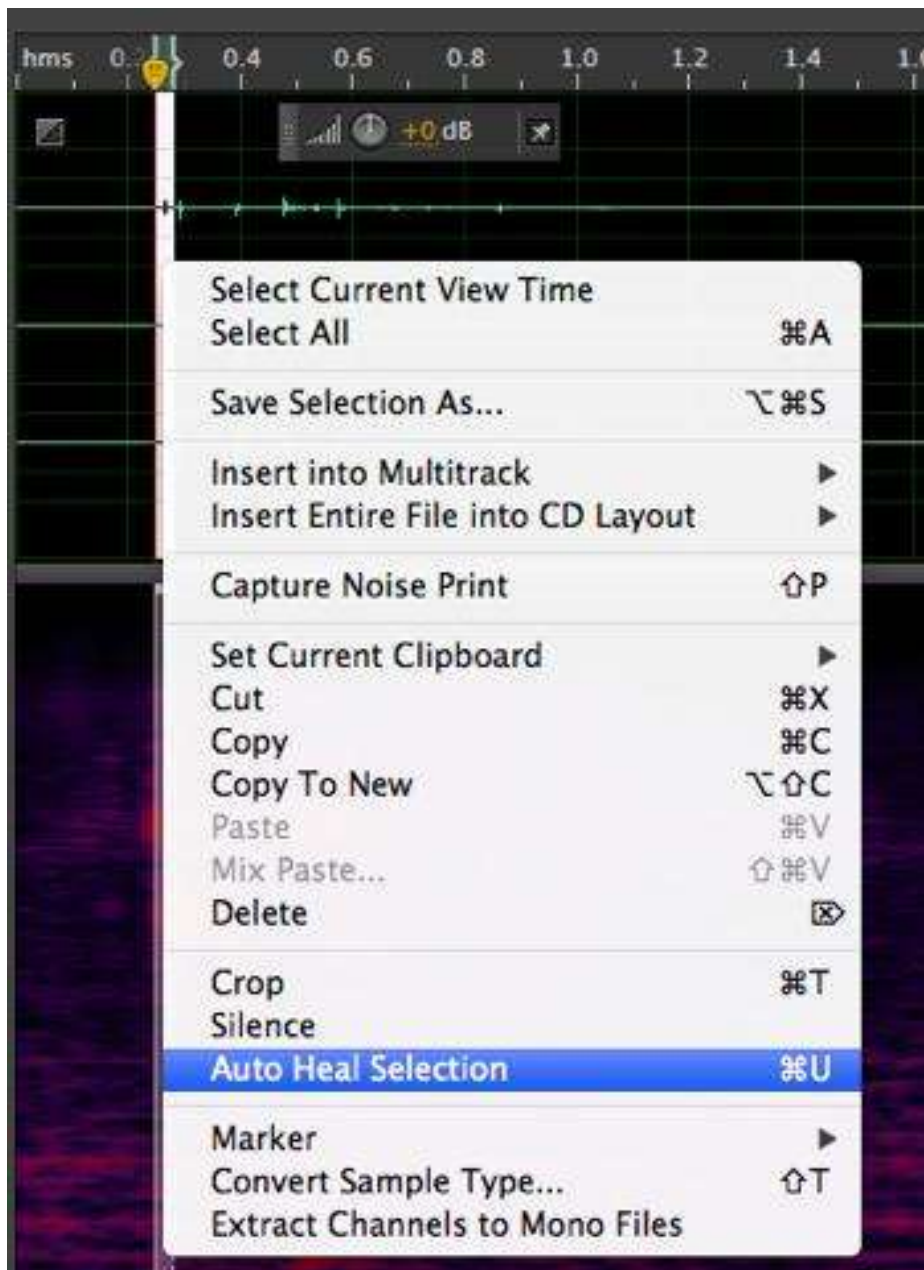


Auto Heal & Spot Healing

You can use Auto Heal & the Spot Healing Brush to remove clicks, pops, and other short noises you want to remove from your audio.

Zoom in by pressing the plus key and select the pop. Right click and select “Auto Heal” (Comm/Control + U).

You can also paint a selection with the spot healing brush(B) by dragging over the area to fix in the “Spectral Frequency” display.



Using the Diagnostic Panel

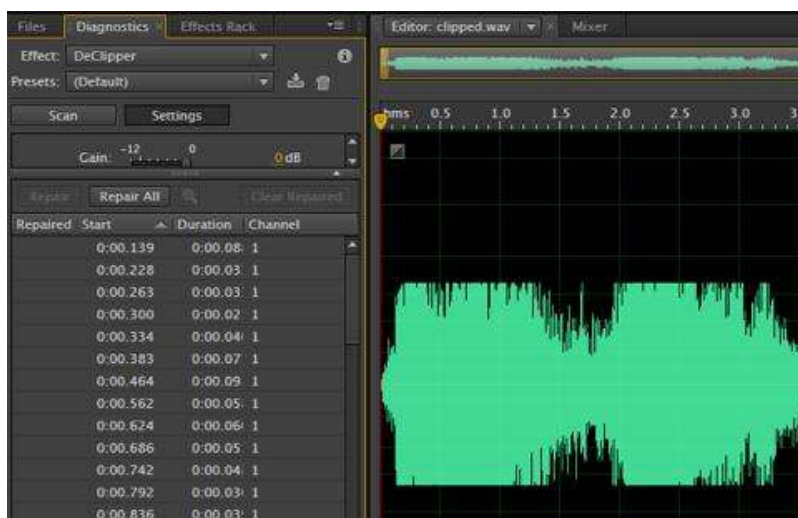
Access the Diagnostics panel in Audition by choosing Window > Diagnostics from the menu bar. The powerful diagnostic panel provides tools to fix clipping, clicks and pops in your audio.

The Declipper is handy for repairing clipped audio. Select the DeClipper Effect in the Diagnostic panel. Click Scan and your clipping areas will be listed. Select a listed problem to move to it in the waveform.

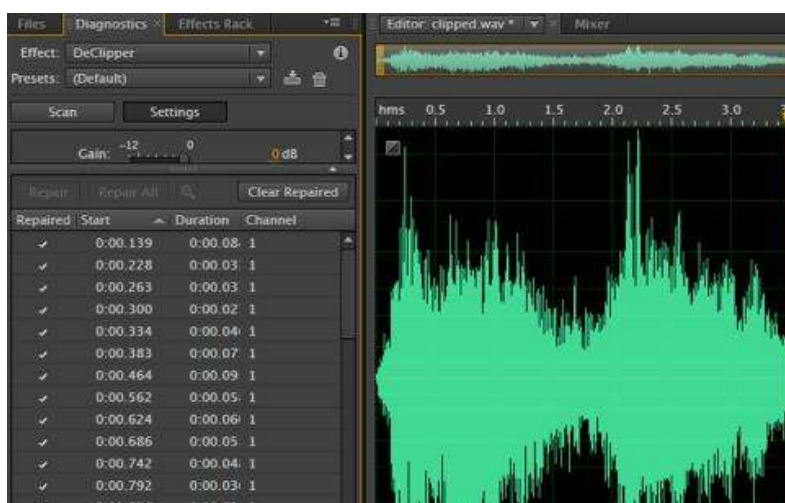
You can fix each one at a time or click Repair All.

Note: Depending on your audio it may still appear clipped, as Audition works in 32 bit floating point. Decrease the amplitude or use Normalize to see that the audio isn't actually clipped. I have had varied results with the DeClipper, so if it doesn't fix your issues, you the manual method mentioned above using Auto Heal & Spot Healing Brush.

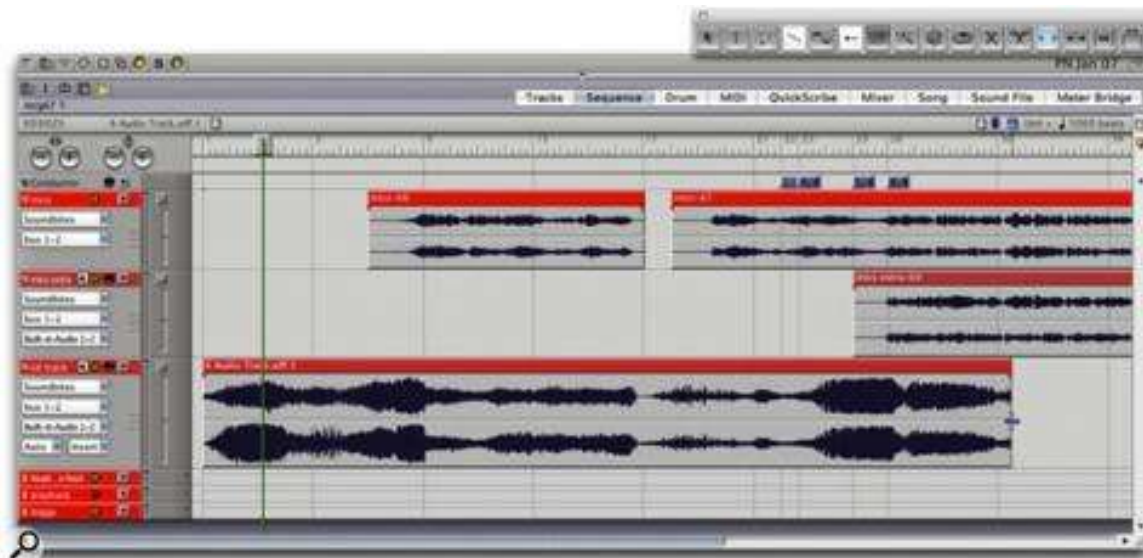
Before:



After:



8. EDITING DIFFERENT AUDIO SCENARIOS



The Trim Tool



One of the most common audio editing scenarios is stripping away unwanted sound bite audio either side of a good take, or isolating a small vocal fragment (say) in a longer speech sound bite. The easiest way to do this is to edge-edit the sound bite, as I described last month, by pointing the mouse at the right or left boundary of the sound bite and dragging using the blue edge-edit cursor.

Sometimes, though, if you're working with very long sound bites or are really zoomed in to the Sequence Editor, sound bite edges aren't visible. Edge-editing is then a hassle: you have to scroll forward (or zoom out) to locate the sound bite edge, drag the edge-edit, then try to get back to what you were doing. Surely there must be an easier way?

DP 5's extended Tool Palette, with the Scissors tool already selected and the Trim tool about to be clicked.

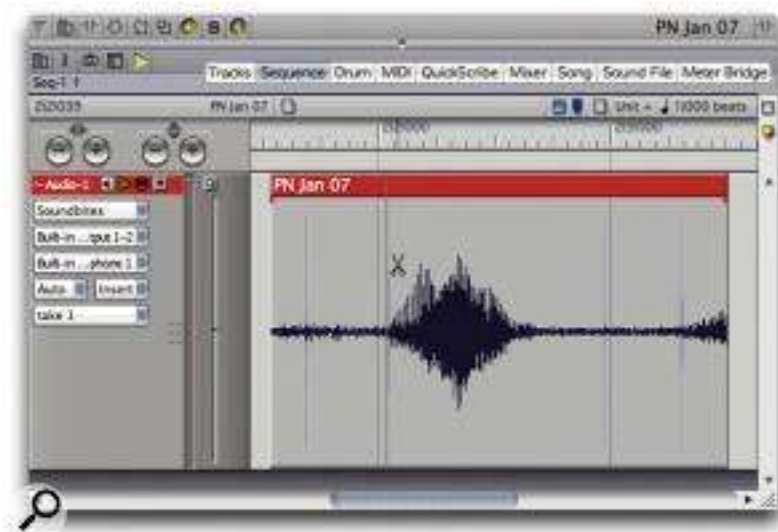
DP 5's extended Tool Palette, with the Scissors tool already selected and the Trim tool about to be clicked.

In Digital Performer 5, there is. The Tool Palette, which you can easily open by hitting Shift-O, now sports four new tools, one of which is the Trim tool. As with all the other tools, it's miles better for workflow to select this with a keyboard shortcut rather than the mouse, and the Trim tool's default shortcut is the forward slash (commonly found next to the right-hand shift key). Try out the following:

1. With a sound bite visible in your Sequence Editor and the Tool Palette open, hold down the forward-slash key. Your mouse pointer should take on the appearance of a square bracket with blue arrows either side — just the same as when you do a normal edge-edit.
2. Point anywhere in your sound bite and click. The right-hand boundary of the sound bite jumps back to that point. You've effectively edge-edited the sound bite, but in a single click, and with no need to go near the right-hand sound bite edge.
3. Hit Apple-Z to Undo your last action and get your full-length sound bite back.
4. Now hold down the forward-slash key together with the Alt/Option key and again point at your sound bite. You'll notice the mouse pointer's square bracket now points in the opposite direction to before.
5. Click somewhere in your sound bite and now the left-hand boundary of the sound bite jumps to that point. So Alt/Option-slash is like edge-editing the left-hand side of the sound-bite.

It's worth noticing that you can actually click and drag with the Trim tool, just as if you were edge-editing, and DP will 'scrub' the audio as you do so (with, typically, comical results). Also, if you have the Edit grid enabled and you hold down the Apple key while trimming, the trim will be constrained to that grid, which is handy when you need edits to snap to a strict rhythmical pattern.

The Scissor Tool



DP's Scissors tool has been around for a while and performs all manner of useful editing tasks, chopping up longer sound bites into shorter ones, and isolating beats or phrases ready to be duplicated and used elsewhere in your sequence. Let's look at it in context.

1. Load up or record a sound bite, have it showing in the Sequence Editor, and be prepared to hack it to bits. For now, make sure the Edit and Beat grids are turned off, by deselecting their blue toggle switches at the top right of the editor window.

2. Regardless of whether the Tool Palette is open or closed, hold down the 'C' key (think 'Cut') to engage the Scissors tool. When you point at your sound bite, you'll see an additional cursor superimposed on it to confirm exactly where the cut will occur.

When you're working with beat analysed audio, and with DP's Beat grid enabled -- note the blue tick-box towards the upper right of the window & mdash; Scissor tool cuts are constrained to audio beats. Here, a short vocal syllable is being cut on a detected beat, shown by very faint vertical lines superimposed on the waveform.

When you're working with beat analysed audio, and with DP's Beat grid enabled -- note the blue tick-box towards the upper right of the window — Scissor tool cuts are constrained to audio beats. Here, a short vocal syllable is being cut on a detected beat, shown by very faint vertical lines superimposed on the waveform.

3. Simply click to cut the sound bite into two. You can then continue to click if necessary to make additional cuts in other places. Simple as that.

The Scissors tool has got some tricks up its sleeve, though. One is the ability to make cuts that are automatically aligned with Edit or Beat grid divisions. This makes it really easy to exactly chop out a four-beat section (for example) from a longer sound bite. And, assuming your audio has been analysed for beats, cutting out a single drum hit or even an entire groove becomes trivially easy, even if it doesn't exactly align with DP 's time ruler.



When you're using the Scissors in conjunction with the Edit or Beat grids, there's another technique you can try. Zoom in to your sequence sufficiently that you can see all the divisions of your currently selected Edit grid (it won't work otherwise). Then click a sound bite to make a cut, but keep your mouse key pressed down and drag left or right over the soundbite. As you do this DP makes multiple cuts, either on Edit or Beat grid divisions, to leave what looks like a very fragmented collection of sound bites. Of course, the audio will still play back without a break, as the sound bites perfectly abut one another, but you can now easily move or copy any of them elsewhere in your sequence (see screen above). Some people also delete or mute some of the resulting sound bites, to achieve 'stutter' or glitch effects.

Sound bite Handling

Layering

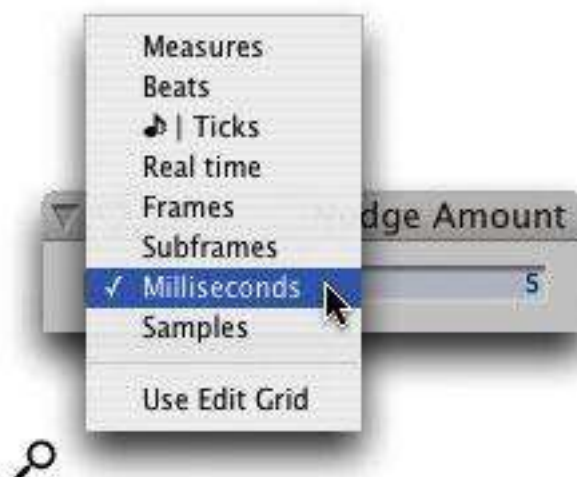


The Audio menu gives access to DP's sound bite layering commands. With these, you can control how overlapping sound bites appear on the screen, and consequently how the resulting edit sounds during playback.

The Audio menu gives access to DP's sound bite layering commands. With these, you can control how overlapping sound bites appear on the screen, and consequently how the resulting edit sounds during playback.

When you're working with sound bites, it doesn't take long to realise that they can be over-lapped in the same track. On playback, DP plays whatever portion of a sound bite is visible at any one time, so the layering of overlapping sound bites becomes important. Just as in a graphics program, DP has functions for moving selected sound bites forward or back in the layering. To access these, go to Audio menu / Layering and choose one of the four sub-menu items that appears (see screen below). Alternatively, use the keyboard shortcuts Shift-Apple-[minus sign] and Shift Apple-[equals] to push a sound bite back or forward in the track layering, respectively.

Nudge, Nudge



The Audio menu gives access to DP's sound bite layering commands. With these, you can control how overlapping sound bites appear on the screen, and consequently how the resulting edit sounds during playback.

Nudge is a wonderfully helpful feature that is also easy to use. It simply moves any selected data (including sound bites) by a user-specified amount, in response to a press of the left or right arrow key on your Mac keyboard. For audio editing, you can set it up to move a sound bite one millisecond (or less) at a time, to hugely assist in finding good edit points, for example. Equally, the Nudge amount can be much larger, making it easy to move data and audio around your sequence regardless of the zoom level.

To set the current Nudge Amount, hit Alt-Apple-N. Then, in the little window that appears, configure your desired time/amount by choosing from the pop-up menu and configuring any necessary text boxes. One handy option is 'Use Edit Grid', but all the others are useful too. For example, if you needed to move your entire sequence four bars further along, to make room for some new material at the start, you'd only need to hit Apple-A (to Select All), Alt-Apple-N (to bring up the Nudge Amount window), choose 'Measures' and type in '4', hit Return, then, back in the Sequence Editor, press the right arrow key. If you do a lot of Nudging, you can even keep the Nudge Amount window open and reconfigure it whenever necessary as you encounter different editing tasks.

SELF ASSESSMENT QUESTION

Q # 1. How you can launch the software?

Q # 2. Name one of the famous audio editing tool and why it is mostly used?

Q # 3. What is interface in the means of communication?

Q # 4. What is sound card and how it works?

Q # 5. How many types of sound cards are there?

Q # 6. What is customizing the interface?

Q # 7. Brief any scenario?

Q # 8. What is the utilisation of tool?

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UNIT 7

Editing Efficiency and Multi-track Mixing

Written By: Syyed Muhammad Saadullah Shah

Reviewer: Syed Salman Ali Zaidi

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- 2. Effect Rack**
- 3. Amplitude Statistics**
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- 6. Multi-track Session**
- 7. Adding Multiple Track**
- 8. Mixing and placing Audio**

Self Assessment Questions

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INTRODUCTION

Early recordings were all completed live, using a single complete take; due to the technology not being advanced enough to patch in mistakes it all had to be done through one take only. Audiences of music were more concerned with how it was performed than the perfection of it, so if a wrong note was played, that didn't bother people as much as it would now. Now, with the exception of a few 'live' albums, most takes are edited to make them as 'perfect' as possible. Audio editing can involve taking part of a track and replacing it with another section that is the same, or adding/removing content from the track. The modern listening audience has become very selective regarding mistakes and what doesn't sound good to the ear. We can edit through our DAW to enhance the track using Beat Detective, Flex Time, Time Stretch etc.

Multi-track mixing essentially the blending of a number of audio files or regions to create an overall mix that sounds well balanced. This introductory guide is written for those who wish to conduct very basic mixing of small multi-track projects and who have little or no knowledge or experience of the subject.

A mix is the putting together of audio (on separate tracks or channels in your software) to create a final mix file. With the creation of a new audio resource, this could be in the form of separate audio files of spoken word, adding background music to voice recordings, or blending in special effects sound files with other source files.

It discusses the fundamental aspects of mixing in digital audio software, such as volume, pan and automation, and what mixing down to mono, stereo and surround actually means.

This guide also advises on how to finalise your mix, the starting blocks of the mastering processes.

OBJECTIVES

After the reading this unit the student will be able to:

1. Learning after the Editing efficiency and multi track mixing in detail will provide the students a practical professional knowledge for their professional carrier.
1. This course leads the maximum practical skills for those professional students who made their future in the field of all type of audio productions.
2. This is digital era and the audio productions completely deals with the digital technology as the audio productions are the computer aided productions.
4. After the completion of this task the students will be able to handle any type of digital audio professional productions for their best performance in the field of audio productions.

1. MARKERS

Working with markers

Markers (sometimes called cues) are locations that you define in a waveform. Markers make it easy to navigate within a waveform to make a selection, perform edits, or play back audio.

In Adobe Audition, a marker can be either a point or a range. A point refers to a specific time position within a waveform (for instance, 1:08.566 from the start of the file). A range has both a start time and an end time (for example, all of the waveform from 1:08.566 to 3:07.379). You can drag start and end markers for a range to different times.

In the timeline at the top of the Editor panel, markers have white handles you can select, drag, or right click to access additional commands.



Examples of markers

A. Marker point B. Marker range

Note:

To preserve markers when you save a file, select Include Markers and Other Metadata.

Add, select, and rename markers

Though you can add markers directly in the Editor panel, you use the Markers panel (Windows > Markers) to define and select markers.

To hide or show information such as Duration and Type, choose Markers Display from the panel menu.

Add a marker

Do one of the following:

- Play audio.
- Place the current-time indicator where you want a marker point to be.
- Select the audio data you want to define as a marker range.
- Either press the M key, or click the Add Marker button in the Markers panel.

To automatically create markers where silence occurs, see Delete Silence and Mark Audio options.

Select markers

- Click a marker in the Editor or Markers panel. Or double-click to move the current-time indicator to that location and select the area for range markers.
- To select adjacent markers, click the first marker you want to select in the Markers panel, and then Shift click the last.
- To select nonadjacent markers, Ctrl+click (Windows) or Command+click (Mac OS) them in the Markers panel.
- To move the current-time indicator to the nearest marker, choose Edit > Marker > Move CTI to Next or Previous.

Rename a marker

- In the Markers panel, select the marker.
- Click the marker name, and enter a new name.

Adjust, merge, convert, or delete markers

After creating markers, you can fine-tune them to best address the needs of an audio project.

Reposition markers

- In the Editor panel, drag marker handles to a new location.
- In the Markers panel, select the marker, and enter new Start values for point markers, or Start, End, and Duration values for range markers.

Merge individual markers

- In the Markers panel, select the markers you want to merge, and click the Merge button.

The new merged marker inherits its name from the first marker. Merged point markers become range markers.

Convert a point marker to a range marker

- Right-click the marker handle, and choose Convert to Range.

The marker handle splits into two handles.

Convert a range marker to a point marker

- Right-click a marker handle, and choose Convert to Point.

The two parts of the range marker handle merge into a single handle, with the start time of the range becoming the time for the point marker.

Delete markers

- Select one or more markers, and click the Delete button in the Markers panel.
- Right-click the marker handle in the Editor panel, and choose Delete Marker.

Save audio between markers to new files

- In the Waveform Editor, choose Window > Markers.
- Select one or more marker ranges. (See Working with markers .)
- Click the Export Audio button in the Markers panel.
- Set the following options:

Use Marker Names In Filename

Uses the marker name as the prefix for the filename.

Prefix

Specifies a filename prefix for the new files.

Postfix Starting #

Specifies the number to begin with when adding numbers to the filename prefix. Adobe Audition automatically adds numbers after the prefix (for example, prefix02, prefix03) to distinguish saved files.

Location

Specifies the destination folder for saved files. Click Browse to specify a different folder.

Format

Specifies the file format. The Format Settings area below indicates and data compression and storage modes; to adjust these, click Change. (See Audio format settings.)

Sample Type

Indicates the sample rate and bit depth. To adjust these options, click Change. (See Con-vert the sample rate of a file.)

Include Markers and Other Metadata

Includes audio markers and information from the Metadata panel in processed files.(See Viewing and editing XMP metadata.)

Creating playlists

A playlist is an arrangement of marker ranges that you can play back in any order and loop a specified number of times. A playlist lets you try different versions of an arrangement be-fore you commit to edits. You create playlists in the Playlist panel (Window > Playlist).

Note:

To store a playlist with a file, you must save in WAV format. (See Save audio files.)

Create a playlist

- In the Playlist panel, click the Open Markers Panel button .
- In the Markers panel, select marker ranges you want to add to the playlist. Then click the Insert Selected Range Markers Into Playlist button , or drag the range markers to the Playlist panel.

Change the order of items in a playlist

- Drag the item up or down.

Play items in a playlist

- To play all or part of the list, select the first item you want to play. Then click the Play button at the top of the panel.
- To play a specific item, click the Play button to the left of the item name.

Loop an item in a playlist

Select an item, and enter a number in the Loops column. Each item can loop a different number of times.

Delete items from a playlist

- Select the items, and click the Remove button .
- Delete Silence and Mark Audio options
- Batch process files

2. EFFECT RACK

Sounds

Effect Racks

Effect Racks is a collection of over 200 sophisticated audio effects engineered for instant sound sculpting. It comes packed with an endless variety of Channel Strips, DJ and Live PA effects, Glitch Racks, Amp Racks, Modulators, Filters, Beat Processors and Noise Boxes.



Effect Racks compiles the vast sound manipulation possibilities of Puremagnetik's RackPak series and includes the complete RackPak 1, 2, 3 and AmpPak collections. Effects Racks' presets are grouped into the following categories:

Amp Racks

From overdriven metal to crunchy glitch percussion processors Amp Racks contains a specially programmed selection of all things amplified and distorted. Plug in a live guitar feed, process audio loops or create lo fi beats with this versatile collection of tools.

Channel Strips

Includes custom multiband compressors, voice, guitar and drum channels, tone coloring, imaging and mix controls.

Crush & Destroy

For annihilating all of your audio quickly and safely. This category includes guitar cabinets, distortion menus, overdrives and bit reduction racks.

DJ & Live PA

Add instant dynamicism to a live set with this collection of beat parsers, DJ channel strips, compressors, EQ and filtering effects.

Glitch Racks

A selection of 11 rhythmic dissection racks, stuttering effects and real-time beat tweaking tools.

LoFi&NoFi

8 expertly programmed distortions, bit crunchers and sample massacring racks. Derange and overdrive whatever you feed these effects.

Noise Boxes

The noise boxes category is unique in that it contains 9 white noise instruments that can easily be replaced with your own instruments or soft synths. Use them as is, or as a template for texture generation, feedback processing and more.

Space Modulators

Includes 9 advanced audio racks to explore the sonic stereo spectrum, widen your mixes or freak out your drums.

Time Machines

Throw 4/4 out the window with this collection of 15 slapbacks, meter crunchers and echo filters.

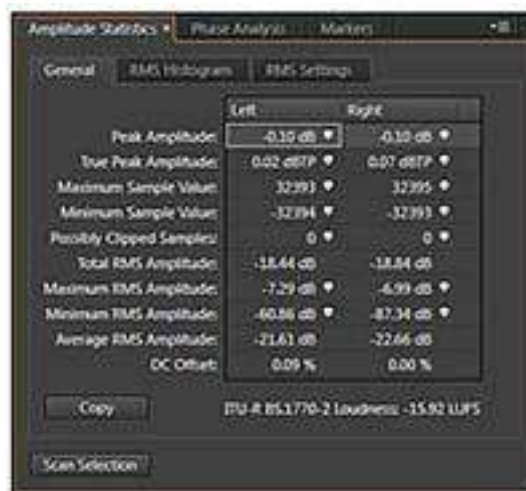
Weird Filters

Banks of 8 band split filters with independent time based effects and dynamic processing per band.



3. AMPLITUDE STATISTICS

1. Audition can analyze your file for amplitude, clipping, DC offset, and other characteristics
 2. If the entire waveform isn't already selected, press Ctrl+A (Command+A).
- Click the Amplitude Statistics tab, and then click Scan Selection.



- The Amplitude Statistics panel shows peak levels, whether some samples may be clipped, average (RMS) amplitude, DC offset, and other statistics. Here's one example of how these statistics can be useful: Note that there's some DC offset in the left channel, which reduces the available amount of headroom. Choose Favorites > Repair DC Offset, and then click Scan Selection again. The left channel no longer has any DC offset.

- Keep this project open for the next lesson.

NOTE

Total RMS Amplitude is another useful statistic, because it represents a file's overall loudness. If two files have very different Total RMS Amplitude statistics, one will probably sound much louder than the other.

4. SHORTCUT KEYS

These partial lists include the shortcuts that Adobe Audition experts find most useful. For a complete list of shortcuts, choose Edit > Keyboard Shortcuts.

Keys for playing and zooming audio

Result

Windows Shortcut

Toggle between Waveform and Multitrack Editor 8

Start and stop playback Spacebar

Move current-time indicator to beginning of timeline Home

Move current-time indicator to end of timeline End

Move current-time indicator to previous marker, clip, or Ctrl+left arrow

selection edge

Move current-time indicator to next marker, clip, or selection edge Ctrl+right arrow

Toggle preference for Return CTI To Start Position On Shift+X

Stop

Zoom in horizontally =

Zoom in vertically Alt+=

Zoom out horizontally -

Results

Windows Shortcut

Zoom out vertically Alt+minus sign

Add marker M or * (asterisk)

Move to previous marker Ctrl+Alt+left arrow

Move to next marker Ctrl+Alt+right arrow

Keys for editing audio files

The following keyboard shortcuts apply only in the Waveform Editor.

Results

Windows Shortcut

Repeat previous command (opening its dialog box and clicking OK)

Shift+R

Repeat previous command (opening its dialog box but not clicking OK)

Ctrl+R

Open Convert Sample Type dialog box

Shift+T

Capture a noise reduction profile for the Noise Reduction effect

Shift+P

Activate left channel of a stereo file for editing

Up arrow

Activate right channel of a stereo file for editing

Down arrow

Results

Windows Shortcut

Make spectral display more logarithmic or
linear

Ctrl+Alt+up or down arrow

Make spectral display fully logarithmic or linear

Ctrl+Alt+Page Up or Down

Increase or decrease spectral resolution

Shift+Ctrl+up or down arrow

Keys for mixing multitrack sessions

The following keyboard shortcuts apply only in the Multitrack Editor.

Results

Windows Shortcut

Select the same input or output for all audio tracks

Ctrl+Shift-select

Activate or deactivate Mute, Solo, Arm For Record, or Ctrl+Shift-click

Adjust knobs in large increments

Shift-drag

Adjust knobs in small increments

Ctrl-drag

Nudge selected clip to the left

Alt+comma

Nudge selected clip to the right

Alt+period

Maintain key frame time position or parameter value Shift-drag

Reposition envelope segment without creating keyframe Ctrl-drag

5. FAVORITES

Adding a Favorite

1. As you begin to work more with Audition, you may find yourself applying the same effects repeatedly. Saving effects as favorites allows you to name and access your effects in a central location. You can even assign keyboard shortcuts to access commonly used effects.

2. In the Edit View, click on the Favorites tab to bring the Favorites panel forward. If the Favorites tab is not visible, choose Window > Favorites to make it visible. Click the Edit Favorites button at the bottom of the Favorites panel. The Favorites window appears. Audition comes installed with certain favorites, such as Fade In and Fade Out.

Click the New button, and in the Name field, enter Pan Left to Right Smooth. In the Press New Shortcut Key field type the letter D.

3. In the Function tab, click the Audition Effect drop-down menu. A list of the effects available in Edit View appears. Choose Amplitude\Stereo Field Rotate (process).



4. To ensure that the effect you are saving is the one you want, click the Edit Settings button. The Stereo Field Rotate window appears. Click to select the Pan Left to Right Smooth preset that you saved in the previous section, and click OK. Click the Save button and the favorite appears at the bottom of the Favorites list.

5. In the Favorites window, click the Up button to move your favorite to the top of the list of favorites. Click the Close button.

6. Click the Files tab to access your list of files. Double-click on the SmackFunkDrm18. Cell file to display the waveform. Click to select the waveform, then press the letter D to apply the Stereo Field Rotate effect. You may need to wait a few moments as Audition calculates and applies the changes. When the waveform changes, the effect has been applied. Press the spacebar to hear the effect.

Note

You can also click the Favorites tab, then double-click the Pan Left to Right Smooth favorite to apply the effect.

7. Choose File > Save As. If necessary, navigate to the AA_03 folder on your hard disk. Rename this file SmackFunkDrm_pan.cel and click Save.

Note

If an alert message appears confirming that you want to overwrite an existing file, click Yes.

8. Press F12 to enter Multitrack View. When working in Edit View, the Multitrack View is also open and all edits made in Edit View are automatically updated. Press the Home key and then press the Play button to hear the session. Pay particular attention to the stereo effect of the first drum clip, as well as when the tambourine begins. The stereo field changes you made have been incorporated into the main mix.

9. Choose File > Save Session and then choose File > Close All.

6. MUTITRACK SESSION

About multitrack sessions

In the Multitrack Editor, you can mix together multiple audio tracks to create layered soundtracks and elaborate musical compositions. You can record and mix unlimited tracks, and each track can contain as many clips as you need—the only limits are hard disk space and processing power. When you're happy with a mix, you can export a mixdown file for use on CD, the web, and more.

The Multitrack Editor is an extremely flexible, realtime editing environment, so you can change settings during playback and immediately hear the results. While listening to a session, for example, you can adjust track volume to properly blend tracks together. Any changes you make are impermanent, or non-destructive. If a mix doesn't sound good next week, or even next year, you can simply remix the original source files, freely applying and removing effects to create different sonic textures.

Adobe Audition saves information about source files and mix settings in session (.sesx) files. Session files are relatively small because they contain only pathnames to source files and references to mix parameters (such as volume, pan, and effect settings). To more easily manage session files, save them in a unique folder with the source files they reference. If you later need to move the session to another computer, you can simply move the unique session folder.

Editing multitrack sessions in the Editor panel and Mixer

In the Multitrack Editor, the Editor panel provides several elements that help you mix and edit sessions. In the track controls on the left, you adjust track specific settings, such as volume and pan. In the timeline on the right, you edit the clips and automation envelopes in each track



Select ranges in the Multitrack Editor



In the toolbar, select the Time Selection tool .

1. In the Editor panel, do one of the following:

- To select only a range, click an empty area of the track display, and drag left or right.
- To select a range and clips, click the center of a clip, and drag a marquee.

Start Time

Sets a start-time offset, helping you match audio in Adobe Audition to the time displayed in video applications.

Advanced settings

To customize Time Display settings for the active session, set the Time Format and Custom Frame Rate settings. For details, see [Change the time display format](#).

Adobe also recommends

- Comparing the Waveform and Multitrack editors
- Create a new multitrack session
- Save multitrack sessions
- Add or delete tracks
- Arranging and editing multitrack clips
- Automating mixes with envelopes

7. ADDING MULTIPLE TRACK

COMPILE WHOLE TRACKS.

Track1, track2, track3, track4, track5.

Import track one and two.

Left click on blank area of left panel of track two to highlight the track.

Go to Edit and click Cut

Click on track one at the very end of the track then go to edit again and click Paste.

Track two will now be added to the end of track one.

If there's a silent gap between the tracks click on the gap to place the selection marker then zoom in, click hold and drag to select the blank section then Edit/Cut.

Repeat for tracks 3, 4 and 5 then export via the file menu to your chosen format.

For a smoother transition between tracks you could crossfade them.

As one track is fading out the next is fading in. There is no stop / start as in the above method.

If a track starts or stops abruptly a fade in or fade out effect can be applied by selecting the beginning / end of a track then selecting fade in or fade out from the effect menu.

Do this first.

If you make an error ctrl+z is the quickest way to undo it.

Import all five tracks.

Apply fade ins / outs if needed.

Select the time shift tool. (icon with the left/right arrow <->)

Click hold and drag on track two to move the whole track to the right until it slightly over-laps the end of track one.

Re-select the selection tool, click on the waveform just before the crossfade point and play the track to check it.

Re-adjust with the time shift tool if needed then do the same for tracks 3 and 4.

Tip: there's an icon with a blank magnifying glass symbol and inward pointing arrows (>--<) which will fit the whole project in the same screen.

The finished project should look something like this with the overlaps: (ignore the lines)

Track one

_____Track two

_____Track three

_____Track four

_____Track five

When exported it's automatically mixed into a single track.

8. MIXING AND PLACING AUDIO

Audio mixing (recorded music)

Digital Mixing Console Sony DMX R-100 used in project studios

In sound recording and reproduction, audio mixing (or mix down) is the process which commences after all tracks are recorded (often, tracked) and edited as individual parts. The mixing-process can consist of various processes but are not limited to setting levels, setting equalization, using stereo panning, and the addition of effects. The way the song is mixed has as much impact on the way it sounds as each of the individual parts that have been recorded. Dramatic impacts on how the song affects the listeners can be created by minor adjustments in the relationship among the various instruments within the song.

Audio mixing is utilized as part of creating an album or single. Mixing is largely dependent on both the arrangement and the recordings. The mixing stage often follows a multitrack recording. The process is generally carried out by a mixing engineer, though sometimes it is the musical producer, or even the artist, who mixes the recorded material. After mixing, a mastering engineer prepares the final product for reproduction on a CD, for radio, or otherwise.

Prior to the emergence of digital audio workstations (DAWs), the process of mixing used to be carried out on a mixing console. Currently, more and more engineers and independent artists are using a personal computer for the process. Mixing consoles still play a large part in the recording process. They are often used in conjunction with a DAW, although the DAW may only be used as a multitrack recorder and for editing or sequencing, with the actual mixing being performed on the console.



Digital Mixing Console Sony DMX R-100 used in project studios

The role of audio mixing

In its simplest form an audio mixer combines several incoming signals into a single output signal, however this is not as simple as connecting the input signals in parallel and sending them to a single output signal because they could influence each other. In order to combine different signals, they must be mixed first so that each signal has a relationship of hierarchy (each signal's volume one step below the next). The role of a music producer is not necessarily a technical one, with the physical aspects of recording being assumed by the audio engineer, and so producers often leave the similarly technical mixing process to a specialist audio mixer. Even producers with a technical background may prefer that a mixer comes in to take care of the final stage of the production process. Noted producer and mixer Joe Chiccarelli has said that it is often better for a project that an outside person comes in because:

when you're spending months on a project you get so mired in the detail that you can't bring all the enthusiasm to the final [mixing] stage that you'd like. [You] need somebody else to take over those responsibilities so that you can sit back and regain your objectivity.

[https://en.wikipedia.org/wiki/Audio_mixing_\(recorded_music\)](https://en.wikipedia.org/wiki/Audio_mixing_(recorded_music))

Mixing refers to the process of combining multiple Audacity tracks which play simultaneously into a single track.

Audacity mixes automatically when playing or exporting, but it can also physically mix selected multiple tracks together into one within the project.

The channel of a track being mixed affects whether it will be mixed into the left channel of the resulting track(s), the right channel, or both (mono). For example, if you have four tracks:

Track 1: left channel

Track 2: left channel

Track 3: right channel

Track 4: mono channel

and you select them all and perform a Mix and Render, you will end up with one stereo track: the left channel will contain a mix of tracks 1, 2, and 4 and the right channel will contain a mix of tracks 3 and 4.

Mixing can be done for a number of reasons, for example mixing speech with background music to make a podcast, or adding different instruments into the same song. Concatenating songs (for example, playing three songs one after the other) does not necessarily involve mixing, but if you wanted the songs to fade into each other it would involve mixing.

Within an Audacity project, you can physically mix selected multiple selected tracks into a single mono or stereo track using either of two explicit mix commands:

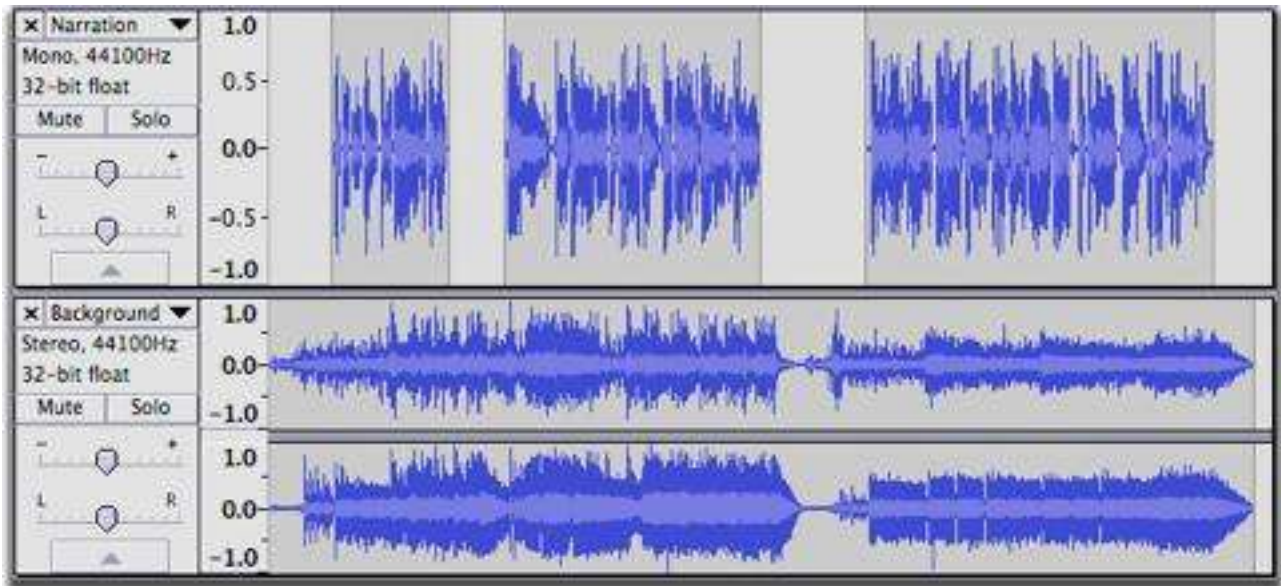
Tracks > Mix and Render (which replaces the original track(s) with the mixed track) or

Tracks > Mix and Render to New Track (which adds the mixed track to the project, preserving the original tracks).

However in Audacity, mixing is automatic. You could just put audio into two different tracks, play to listen to the result then export it as an audio file like MP3 or WAV or burn the WAV to Audio CD.

However once audio has been finally mixed (as in an audio file you might import into Audacity) it is essentially impossible to separate out all the original parts again; it's like trying to take the banana out of a banana milkshake after you've already put it through the blender. There are a few occasions when it actually is possible to separate sounds a bit - you can sometimes isolate the bass, or remove the lead vocals. But these processes don't always work well and usually cause some quality loss. So remember, as long as the multiple tracks are inside an Audacity project, you can manipulate them independently, but once you export as a mixed down file you can't expect to separate the different parts again. So keep your Audacity project around if you plan to continue editing

The controls used for mixing are the Mute and Solo buttons and the Gain (-.....+) and Pan (L.....R) sliders.



In the above example, mixing the mono (upper) and stereo (lower) track means that the audio of the mono track will be heard equally in both left and right channels of the resulting stereo mix.

Muting and Soloing

When working with multiple tracks, it's often important to be able to hear just one at a time. Each track has a Mute and a Solo button, allowing you to temporarily hear just some of your tracks - see the figure above

Mute causes a track to be silenced. More than one track can be muted.

Solo can behave in two different ways depending on the setting made in Tracks Preferences.

Default behaviour is that Solo silences all of the tracks except the ones being soloed. More than one track can be soloed, and soloing overrides muting.

Alternative behaviour is that only one track can be soloed at a time. Soloing still overrides muting.

A third option in Tracks Preferences is to hide the Solo button from tracks, leaving just a Mute button which silences whichever tracks it is applied to.

You can press the Mute and Solo buttons while tracks are playing.

If you're using the keyboard, SHIFT+U toggles muting on the currently focused track which has the yellow border, and SHIFT + S toggles soloing. The solo shortcut works even if you hide the Solo button.

Gain and panning

Above the Mute / Solo buttons, each track has a - / + gain slider which adjusts the track's volume, and an L / R pan slider which adjusts the track's stereo position in the overall mix - whether it comes from the left speaker, right speaker or in-between. To change the value, just click on the slider and drag. For finer control when dragging, hold SHIFT while dragging or double-click on the slider or slider scale to enter a precise value as text.

The normal range of gain is from -36 dB to 36 dB. If you need more, choose Effect > Amplify.

If you're using the keyboard, use:

ALT + SHIFT + UP to increase the gain on the focused track or ALT + SHIFT + DOWN to reduce it

Use ALT + SHIFT + LEFT to pan left on the focused track or ALT + SHIFT + RIGHT to pan right.

Or press SHIFT + G to adjust the gain in a dialog box or SHIFT + P to adjust the pan.

Explicit Mixing and Rendering

While mixing is automatic, there are times when you may want to explicitly tell Audacity to mix several tracks. This is useful in several ways:

You can consolidate tracks which you have finished working on, making it easier to see the other tracks without scrolling up and down

Playback may respond more quickly with fewer tracks

You can see what the final mix will look like as a waveform so as to check the overall level of the final mix before exporting it.

To mix explicitly, select all the tracks you want to mix together then choose either Tracks > Mix and Render or Tracks > Mix and Render to New Track (shortcut CTRL + SHIFT + M).

Several things happen when you choose either Mix and Render command.

All selected tracks are mixed down to a single track called Mix.

If you choose Mix and Render, the resulting Mix track replaces the selected original tracks.

If you choose Mix and Render to New Track, the original tracks are preserved so that the resulting Mix track becomes an additional track in the project.

The Mix track is always placed underneath any non-selected or remaining tracks.

The new mixed track will be stereo unless the tracks you mixed were mono tracks panned to center.

If any of the original tracks did not match the sample rate of the project (set at bottom left of the project window in Selection Toolbar), they will be re-sampled to match the project rate.

Any envelope points defining amplitude modifications will be applied and the previous envelope points removed.

Gain and panning changes will be applied and the sliders reset to normal in the mixed track.

Mute and Solo button states will be released.

You can always Edit > Undo if you're not happy with the results of Mix and Render, then make changes and try it again.

Mixing Levels

The act of mixing multiple tracks adds the waveforms together. In most cases this will cause the mixed track to have a higher peak and RMS (average) level than the individual pre-mixed tracks, though this is not always true by definition. How much (or whether) the peak level increases and how much louder it actually sounds depends on how related the wave-forms of the mixed tracks are.

When peaks or troughs in the waveform coincide, the waveforms will reinforce each other, leading to an increased signal level. In fact if you combined two identical tracks, the signal level would exactly double, leading to an increase in peak level of 6 dB. But when a peak in one track coincides with a trough in another track the waveforms will tend to cancel each other out, leading to a lower level in the mix at that point.

Also the more tracks that have audio at the same point on the Timeline, the higher the mix level is likely to be.

The overall mix level is indicated on the Playback Meter when the project is playing. You can see individual meters for each track (showing the levels as modified by the track's gain/pan sliders and mute/solo buttons) if you enable View > Mixer Board.

SELF ASSESSMENT QUESTIONS

Q # 1.What are the markers and why markers use during audio recording/editing ?

Q # 2.What are the benefits of shortcut keys?

Q # 3.What is multi-track session and what are its benefits?

Q # 4.How you can add multiple tracks?

Q # 5.Brief the mixing and placing audio?

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Unit 8

Exporting and Formats

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REVIEWER: Umer Mehmood Qureshi

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Reference

INTRODUCTION

Dear Students the fact is that the real world is continuous and the Digital world is not. More precisely time, distance, and pressure are all continuous quantities. They can take an infinite number of values and are infinitely divisible.

In this unit we will discuss about the Sampling rate for an audio file. The students will learn why it is very important for our audio files, bit depth and bit rate.

Dear students the sampling rate is only one of several variables in audio files. Others include the sample size, the number of channels, encoding algorithm, the type of compression used, and possibly commands and/or information useful to the operating system for which the file format was developed.

Dear students many audio file formats allow for a variable number of channels. Thus a file could be mono, stereo, or any number of discreet channels. Audio files can be encoded in different ways. Dear students we all know and appreciate the convenience of digital audio, and most people are comfortable with the most common format of MP3. Move beyond this, however, and things can start to get a little confusing. In this unit students will learn the basics of file formats, and hopefully, it will help students to improve their listening experience in the process.

Dear students Data formats designed to represent musical scores, recordings, and other miscellaneous aspects of musical composition have proliferated over the last several decades. While some have been recognized as industry standards, archival records have been generated in a bewildering array of formats specific to particular operating systems and software, in addition to a variety of file interchange formats. This unit is intended as an introduction to the file formats used in audio recordings. The digital representation of music can be broken down into three broad categories. The first category includes file formats that represent actual sound (digital recordings) while the second includes formats that represent musical scores (notation files). A third category includes formats that neither represent a score or recording, but serve to control computer operations that could then generate a score or recording.

The last variables in an audio file are the system-specific commands or information. File formats that make use of these variables are only useful on certain computers. To some extent, the file format used can help to determine the chronology of files in an archive. Some formats have become obsolete as technology improved, and some operating systems have decreased in popularity.

Dear students a description of the file formats and their technical specifications is explained in this unit which will help students learn about file formats.

Dear students this unit also covers the different ways of saving an edited audio file. How to export your saved work or project in different file formats. In the end of this unit students will be able to write your own audio CD.

OBJECTIVES:

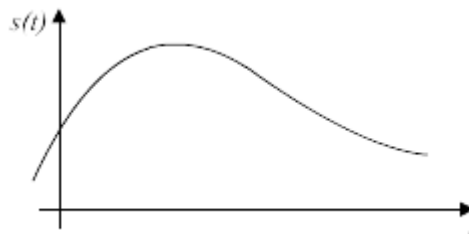
After studying this unit you will be able to:

1. Understand the science of sampling and bit depth in an audio file.
2. Importance of audio channels in recording.
3. Knowledge about the difference of stereo and mono audio.
4. Learn about recording methods and audio quality.
5. Knowledge about the history of mono and stereo.
6. Know about different file formats.
7. Knowledge about compressed, lossless and lossy formats.
8. Learn how to save the audio file.
9. Learn to export multitrack audio project.
10. Able to save an edited audio file and its export to different format and creating audio CD.

1. SAMPLE RATE AND BITRATE

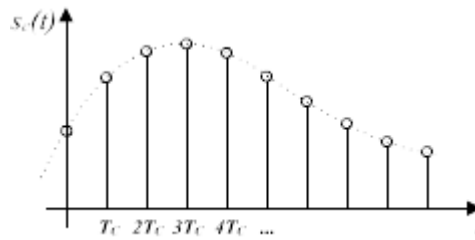
We use digital audio all the time, but people are unclear about how digital audio works. Digital audio has two primary qualities that compose the way the audio is described. Real sounds have frequencies and volumes. In order to measure real world sounds and represent them digitally, we have created sample rate and bitrate as digital's audio qualities. Sample rate determines how analog frequencies are described digitally whereas bitrate determines how analog volume is described digitally. The two qualities need each other in order to describe a sound. You can't have volume without frequency or frequency without volume.

In order to understand about sample rate and bitrate, you need to understand a little bit about how all things digital work. Digital works like a ticking second hand. Whereas time and the world as we know seem continuous and seamless, digital breaks things like time up into little measurements. When we are talking about measurements of time, we are talking about ticks just like the ticking of a second hand. If anything is to happen digitally, it has to happen on a tick. The rate of these ticks is measured in hertz. A 2Ghz computer has 2,000,000,000,000 ticks a second. That's a lot of ticks.



a. Sample Rate

Sample rate is a rate, just like the ticks we just talked about. Analog signal is smooth, just like the image you see to the right here. Like the real world, it just keeps going. In order to get this signal represented in the digital world, we need to measure it into little chunks by defining a rate. On the bottom line of the graph to the right you'll see time is represented by t .



If we start to split up the smooth analog signal into digital chunks, you will start to see something like the second image. With each tick of the clock we measure what is happening at each tick 't'. That measurement is documented, represented by the balls on the signal graph. How often we do these measurements is called the sample rate. The higher the rate, the closer you'll get to the smoothness of the first image. Measuring things in bits like this is called quantizing and the measurements are called samples (hence sample rate).

The sample rate can be thought of as how often or how much the sound is described.

CD quality audio has 44,100 of these measurements a second. That's called 44.1 kilohertz (khz).

b. Bit Depth

With what we just learned, consider that in order for the ticks to make any sense at all they need to actually be measuring something. We are actually measuring volume. Volume is represented by the height of the balls in the image. With each tick a new measurement of the volume is made. This is how we describe the volume. It is a range from 0 to 100? 0 to 2000? 0 to 1? The range of volumes that can be described is the bit rate. Now, in each of these examples, 0 means totally silent and 100, 2000, or 1, respectively, means as-loud-as-it-can-get. So the only difference between each of these ranges is not how loud the sound can be but how many different volumes can be described. We only have two choices for '0 to 1', i.e. Is there a sound or not? But from 0 to 2000 we can have half volume (1000), quarter volume (500), or even somewhere in between (829). The higher the bitrate, the more accurately we can communicate exactly how loud the volume of the 'real' sound we want to describe is.

The bit rate can be thought of as how well the sound is described.

CD quality audio has 65,536 volumes to choose from for every sample that's measured. That's called 16-bit audio (because 2 to the 16th power is 65,536).

c. Putting Them Together

With each sample rate and bitrate there's a limit to how accurately the analog sound can be described. The Nyquist-Shannon sampling theorem states that a sample rate of twice the maximum frequency of the signal being sampled is needed to describe the frequency. Most humans can hear from 20Hz to 20Khz, so the sampling rate of 44.1khz was chosen to be able to capture frequencies up to 22.05khz.

d. Speaking in General Terms

In general, the higher bitrate the 'smoother' the sound will be. 8-bit sounds rather grainy and harsh whereas 16-bit sound sounds quite a bit better. 24-bit sound is used by most audio professionals these days not because it sounds so much better than 16-bit sound but because the higher accuracy is useful because so much is done to the audio in the recording, mixing, and mastering process. Higher bitrate means that each change that is done to the sound produces a more accurate result. Imagine only being able to describe the sounds you are recording with two volumes: on or off. It would be impossible to produce any music at all with such a low bitrate.

Higher sample rates are theoretically able to capture higher frequencies that humans may or may not be able to perceive. There are many things to consider when making these claims, including the quality of the microphones used, the sounds being recorded, the delivery medium, and the quality of the speakers to be used to listen to the material. Since then, most people have come to the agreement that high resolution samples rates are not as important as higher bit rates with respect to pro audio. Again, you cannot replace one with the other, so a balance is required, but 44.1khz/24-bit audio is still the standard when producing 44.1khz/16-bit audio CD quality audio.

When digital audio is played back, the audio processor looks at the information and recreates the waveform from the sample/bit rates. It's actually creating, as best it can, real continuous sound from the quantized digital data.

e. Final Thoughts

Why carry around a dozen cups from shot glasses to five gallon jugs when one 1.5litre travel bottle suits most of your drinking needs throughout the day? It's the same way with sample rates and bitrates. If you don't need super high-res audio, it may not be worthwhile to record it. The higher the bitrate and samplerate, the more data will be recorded, the larger your sessions will be, and the harder your DAW will have to work.

See this table for a few examples.

<i>Bit Depth</i>	<i>Sample Rate</i>	<i>Bit Rate</i>	<i>File Size of one stereo minute</i>	<i>File size of a three minute song</i>
16	44,100	1.35 Mbit/sec	10.1 megabytes	30.3 megabytes
16	48,000	1.46 Mbit/sec	11.0 megabytes	33 megabytes
24	96,000	4.39 Mbit/sec	33.0 megabytes	99 megabytes
mp3 file	128 k/bit rate	0.13 Mbit/Sec	0.94 megabytes	2.82 megabytes

Hard disk requirements for a multi-track 3 minute song

<i>Bit depth/sample rate</i>	<i>number of mono tracks</i>	<i>size per mono track</i>	<i>size per song</i>	<i>songs per 20 gigabyte hard disk</i>	<i>songs per 200 gigabyte hard disk</i>
16/44.1	8	15.1 megs	121 megs	164	1640
16/48	8	16.5megs	132 megs	150	1500
24/96	8	49.5 megs	396 megs	50	500
16/44.1	16	15.1 megs	242megs	82	820

16/48	16	16.5 megs	264 megs	74	740
24/96	16	49.5 megs	792 megs	24	240

2. AUDIO CHANNELS

There are basic two channels in audio file recording:

1. Stereo
2. Mono

Stereo (Stereophonic) is the reproduction of sound using two or more independent audio channels in a way that creates the impression of sound heard from various directions, as in natural hearing.

Mono (Monaural or monophonic) has audio in a single channel, often centered in the “sound field”.

Comparison chart

Stereo sound has almost completely replaced mono because of the improved audio quality that stereo provides.

Mono versus Stereo comparison chart		
	Mono	Stereo
Cost	Less expensive for recording and reproduction	More expensive for recording and reproduction
Recording	Easy to record, requires only basic equipment	Requires technical knowledge and skill to record, apart from equipment. It's important to know the relative position of the objects and events.
Key feature	Audio signals are routed	Audio signals are routed through 2 or more

	through a single channel	channels to simulate depth/direction perception, like in the real world.
Stands for	Monaural or monophonic sound	Stereophonic sound
Usage	Public address system, radio talk shows, hearing aid, telephone and mobile communication, some AM radio stations	Movies, Television, Music players, FM radio stations

a. APPLICATIONS

Mono sound is preferred in radiotelephone communications, telephone networks, and radio stations dedicated to talk shows and conversations, public address system, hearing aids. Stereo sound is preferred for listening to music, in theatres, radio stations dedicated to music, FM broadcasting and Digital Audio Broadcasting (DAB).

b. HISTORY

Until the 1940s mono sound recording was popular and most of the recording was done in mono even though the two-channel audio system was demonstrated by Clément Ader in as early as 1881. In November 1940 Walt Disney's Fantasia became the first commercial motion picture with stereophonic sound. With the advent of magnetic tapes the usage of stereo sound became easier. In the 1960s albums were released as both monaural LPs and stereo LPs because people still had their old mono players and the radio station were mostly AM. Similarly movies were released in both versions because some theatres were not equipped with stereo speakers systems. Today no monaural standards exist for 8-track tape and compact disc and all films are released in stereophonic sound.

c. RECORDING METHODS AND AUDIO QUALITY

Mono sound recording is done mostly with one microphone and only one loudspeaker is required to listen to the sound. For headphones and multiple loudspeakers the paths are mixed into a single signal path and transmitted. The signal contains no level, arrival time

or phase information that would replicate or simulate directional cues. Everyone hears the very same signal and at the same sound level. The sound played for instance by each instrument in a band will not be heard distinctly though it will have full fidelity. Hand held recorders record sound in mono. It is cheaper and easier to record in mono sound.

Stereo recording is done with two or more special microphones. The stereo effect is achieved by careful placement of microphone receiving different sound pressure levels accordingly even the loudspeakers need to have the capability to produce the stereo and they also need to be positioned carefully. These sound systems have two or more independent audio signal channels. The signals 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. It is expensive and it requires skill to record stereo sound.

There are following methods of recording in stereo:

X-Y technique:(intensity stereophony)- In this technique two directional microphones are at the same place, typically pointing at an angle between 90° and 135° to each other.

A-B technique:(time-of-arrival stereophony) - Here two parallel microphones which are not direction specific are kept some distance apart. This results in capturing time-of-arrival stereo information as well as some level (amplitude) difference information.

M/S technique:(Mid/Side stereophony) - A bidirectional microphone facing sideways and another microphone at an angle of 90° are kept facing the sound source. This method is used for films.

Near-coincident technique:(mixed stereophony) - This technique combines the principles of both A-B and X-Y (coincident pair) techniques. The playback is suitable over stereo speakers.

d. COMPATIBILITY

Mono is compatible with and usually found on Phonograph cylinders, Disc records, like 78 rpm and earlier 16 $\frac{2}{3}$, 33 $\frac{1}{3}$, and 45 rpm microgroove, AM radio and some (very few) FM radio stations. Mono and stereo are both found in MiniDisc, compact audio cassette, most FM radio, VCR formats and TV. Mono is not used in 8-track tape and audio CDs.

3. AUDIO FILE FORMAT

An audio file format is for storing digital audio data on a computer system. The bit layout of the audio data (excluding metadata) is called the audio coding format and can be uncompressed, or compressed to reduce the file size, often using lossy compression. The data can be a raw bit stream in an audio coding format, but it is usually embedded in a container format or an audio data format with defined storage layer.

a. FORMAT TYPES

It is important to distinguish between the audio coding format, the container containing the raw audio data, and an audio codec. A codec performs the encoding and decoding of the raw audio data while this encoded data is stored in a container file. Although most audio file formats support only one type of audio coding data, a multimedia container format (as AVI) may support multiple types of audio and video data.

There are three major groups of audio file formats:

1. Uncompressed audio formats, such as WAV, AIFF, AU or raw header-less PCM.
2. Formats with lossless compression, such as FLAC, Monkey's Audio (filename extension `.ape`), Wav Pack (filename extension `.wav`), TTA, ATRAC Advanced Lossless, ALAC(filename extension `.m4a`), MPEG-4 SLS, MPEG-4 ALS, MPEG-4 DST, Windows Media Audio Lossless (WMA Lossless), and Shorten (SHN).
3. Formats with lossy compression such as Opus, MP3, Vorbis, Musepack, AAC, ATRAC and Windows Media Audio Lossy (WMA lossy)

b. UNCOMPRESSED AUDIO FORMAT

One major uncompressed audio format, LPCM, is the same variety of PCM as used in Compact Disc Digital Audio and is the format most commonly accepted by low level audio APIs and D/A converter hardware. Although LPCM can be stored on a computer as a raw audio format, it is usually stored in a `.wav` file on Windows or in a `.aiff` file on Mac OS. The AIFF format is based on the Interchange File Format (IFF), and the WAV format is based on the similar Resource Interchange File Format (RIFF). WAV and AIFF are not inherently lossless. They are designed to store a wide variety of audio formats, lossless and lossy. They just add a small, metadata-containing header before the audio data to declare the format of the audio data, such as LPCM with a particular sample rate, bit depth and number of channels. Since WAV and AIFF are widely supported and can store LPCM, they are suitable file formats for storing and archiving an original recording.

BWF (Broadcast Wave Format) is a standard audio format created by the European Broadcasting Union as a successor to WAV. Among other enhancements, BWF allows more robust metadata to be stored in the file. This is the primary recording format used in many professional audio workstations in the television and film industry. BWF files include a standardized timestamp reference which allows for easy synchronization with a separate picture element. Stand-alone, file based, multi-track recorders from AETA, Sound Devices, Zaxcom, HHB Communications Ltd, Fostex, Nagra, Aaton, and TASCAM all use BWF as their preferred format.

C. LOSSLESS COMPRESSED AUDIO FORMAT

A lossless compressed format stores data in less space without losing any information. The original, uncompressed data can be recreated from the compressed version.

Uncompressed audio formats encode both sound and silence with the same number of bits per unit of time. Encoding an uncompressed minute of absolute silence produces a file of the same size as encoding an uncompressed minute of music. In a lossless compressed format, however, the music would occupy a smaller file than an uncompressed format and the silence would take up almost no space at all.

Lossless compression formats include the common FLAC, WavPack, Monkey's, Audio, ALAC (Apple Lossless). They provide a compression ratio of about 2:1 (i.e. their files take up half the space of PCM). Development in lossless compression formats aims to reduce processing time while maintaining a good compression ratio.

d. LOSSY COMPRESSED AUDIO FORMAT

Lossy compression enables even greater reductions in file size by removing some of the audio information and simplifying the data. This of course results in a reduction in audio quality, but a variety of techniques are used, mainly by exploiting psychoacoustics, to remove the parts of the sound that have the least effect on perceived quality, and to minimize the amount of audible noise added during the process. The popular MP3 format is probably the best-known example, but the AAC format found on the iTunes Music Store is also common. Most formats offer a range of degrees of compression, generally measured in bit rate. The lower the rate, the smaller the file and the more significant the quality loss.

e. LIST OF FORMATS

<i>File Extension</i>	<i>Description</i>
.aac	The Advanced Audio Coding format is based on the MPEG-2 and MPEG-4 standards. AAC files are usually ADTS or ADIF containers.
.aax	Audiobook format, which is a variable-bitrate (allowing high quality) M4B file encrypted with DRM. MPB contains AAC or ALAC encoded audio in an MPEG-4 container.
.act	ACT is a lossy ADPCM 8 kbit/s compressed audio format recorded by most Chinese MP3 and MP4 players with a recording function, and voice recorders
.aiff	Standard audio file format used by Apple. It could be considered the Apple equivalent of wav.

<i>File Extension</i>	<i>Description</i>
.amr	AMR-NB audio, used primarily for speech.
.ape	Monkey's Audio lossless audio compression format.
.au	The standard audio file format used by Sun, Unix and Java. The audio in au files can be PCM or compressed.
.dss	DSS files are an Olympus proprietary format. It is a fairly old and poor codec. GSM or MP3 are generally preferred where the recorder allows. It allows additional data to be held in the file header.
.flac	File format for the Free Lossless Audio Codec, a lossless compression codec.
.gsm	Designed for telephony use in Europe, gsm is a very practical format for telephone quality voice. It makes a good compromise between file size and quality. Note that wav files can also be encoded with the gsm codec.
.m4a	An audio-only MPEG-4 file, used by Apple for unprotected music downloaded from their iTunes Music Store. Audio within the m4a file is typically encoded with AAC, although lossless ALAC may also be used.
.m4b	Audiobook / podcast extension with AAC or ALAC encoded audio in an MPEG-4 container. Both M4A and M4B formats can contain metadata including chapter markers, images, and hyperlinks, but M4B allows "bookmarks" (remembering the last listening spot), whereas M4A does not.
.m4p	A version of AAC with proprietary Digital Rights Management developed by Apple for use in music downloaded from their iTunes Music Store.
.mp3	MPEG Layer III Audio is the most common sound file format used today.

<i>File Extension</i>	<i>Description</i>
.mpc	Musepack or MPC (formerly known as MPEGplus, MPEG+ or MP+) is an open source lossy audio codec, specifically optimized for transparent compression of stereo audio at bitrates of 160–180 kbit/s.
.ogg, .oga	A free, open source container format supporting a variety of formats, the most popular of which is the audio format Vorbis. Vorbis offers compression similar to MP3 but is less popular.
.opus	A lossy audio compression format developed by the Internet Engineering Task Force (IETF) and made especially suitable for interactive real-time applications over the Internet. As an open format standardised through RFC 6716, a reference implementation is provided under the 3-clause BSD license.
.ra, .rm	A RealAudio format designed for streaming audio over the Internet. The .ra format allows files to be stored in a self-contained fashion on a computer, with all of the audio data contained inside the file itself.
.raw	A raw file can contain audio in any format but is usually used with PCM audio data. It is rarely used except for technical tests.
.wav	Standard audio file container format used mainly in Windows PCs. Commonly used for storing uncompressed (PCM), CD-quality sound files, which means that they can be large in size—around 10 MB per minute. Wave files can also contain data encoded with a variety of (lossy) codecs to reduce the file size (for example the GSM or MP3 formats).
.wma	Windows Media Audio format, created by Microsoft. Designed with Digital Rights Management (DRM) abilities for copy protection.
.webm	Royalty-free format created for HTML5 video.

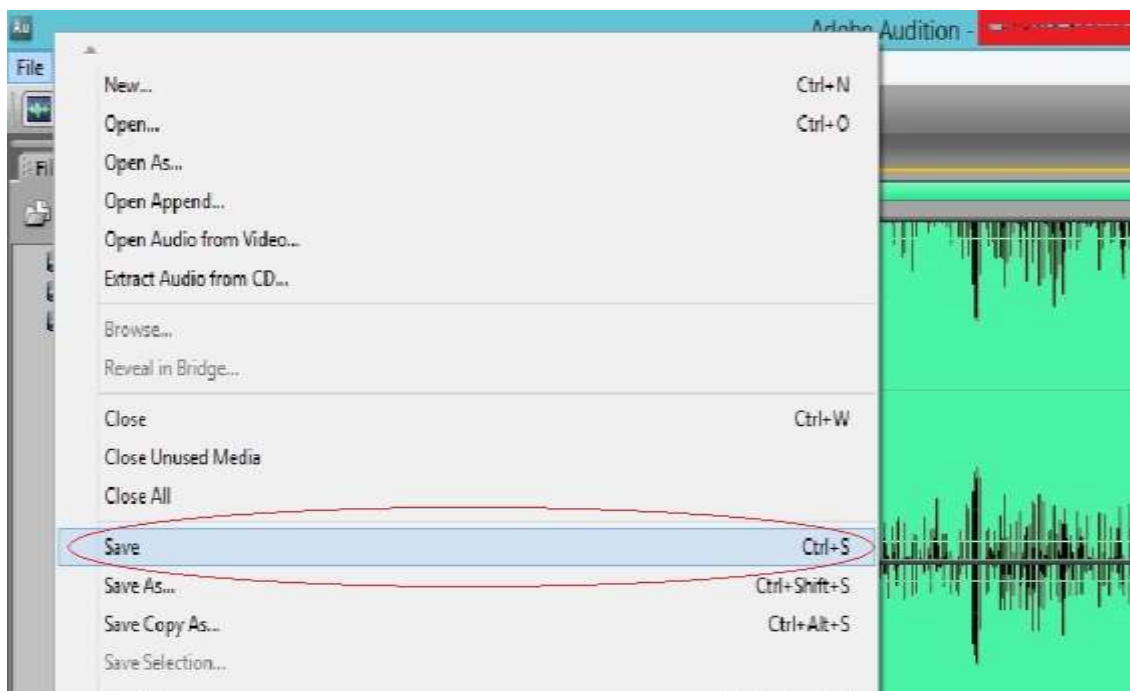
4. SAVING EDITED AUDIO TRACK

There are several different formats for saving a file or group of files are available in edit view of Adobe Audition. We discuss all the options one by one. Choosing a format depends upon your plan to use the file in future. Keep in mind that each format stores unique information that might be discarded if you save a file in a different format.

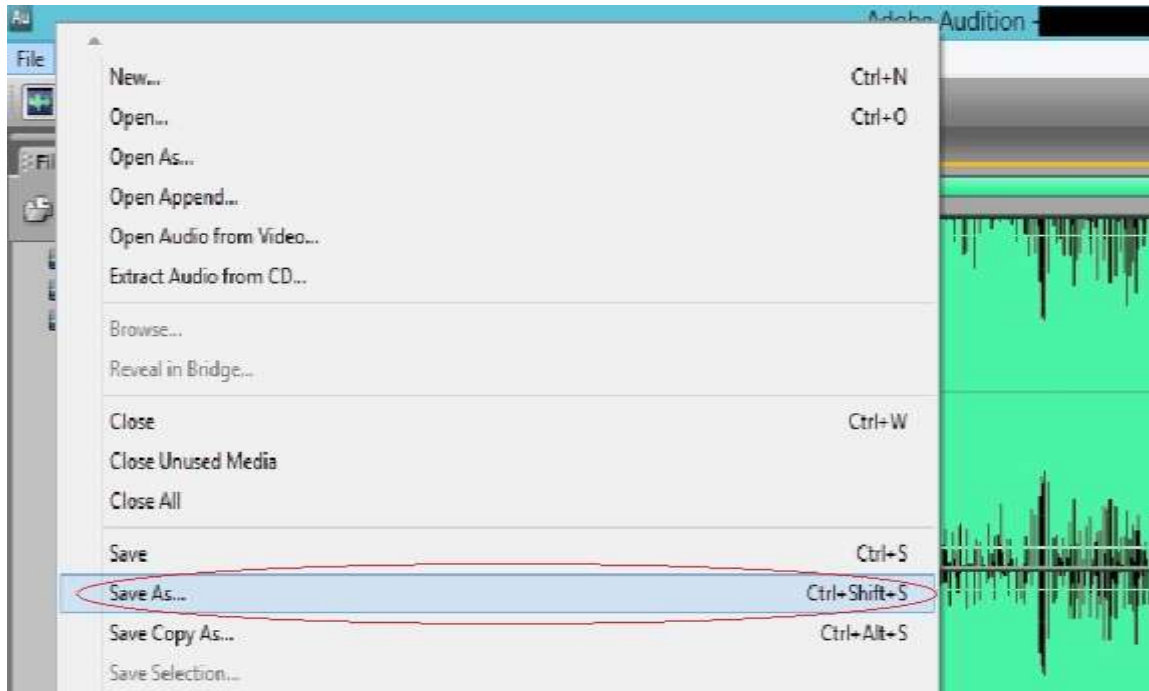
We will discuss the options available in saving a file after defining all the save commands.

a. DIFFERENT WAYS OF SAVING A FILE

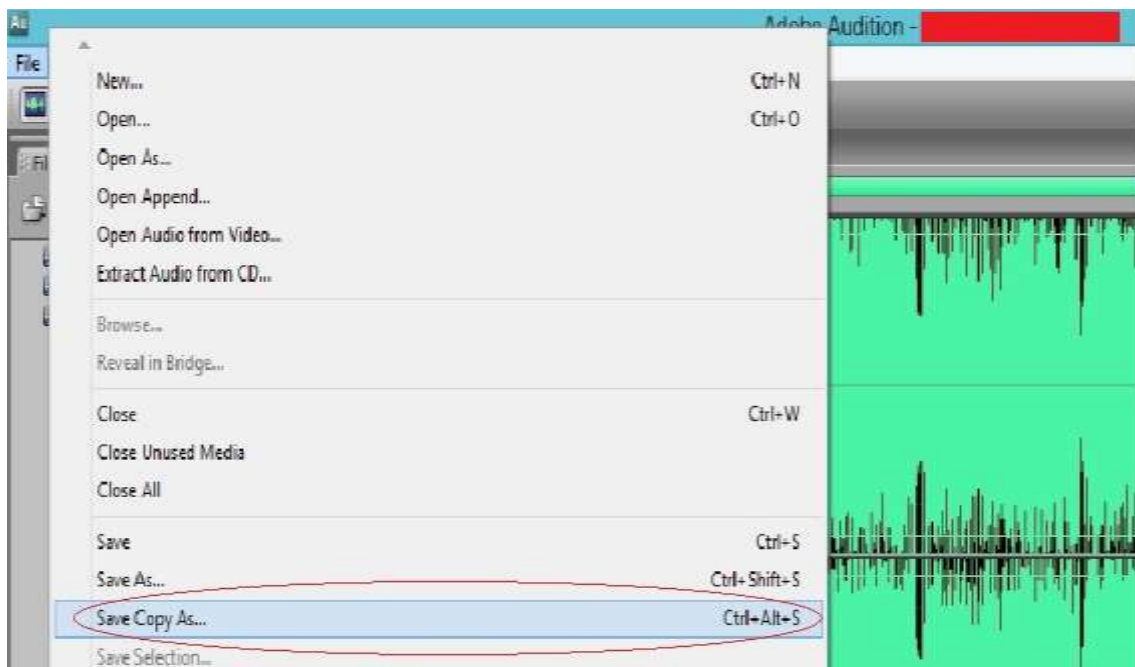
1. To save changes in a current file select File > Save.



2. To save changes under another file name select File > Save as option from the menu.



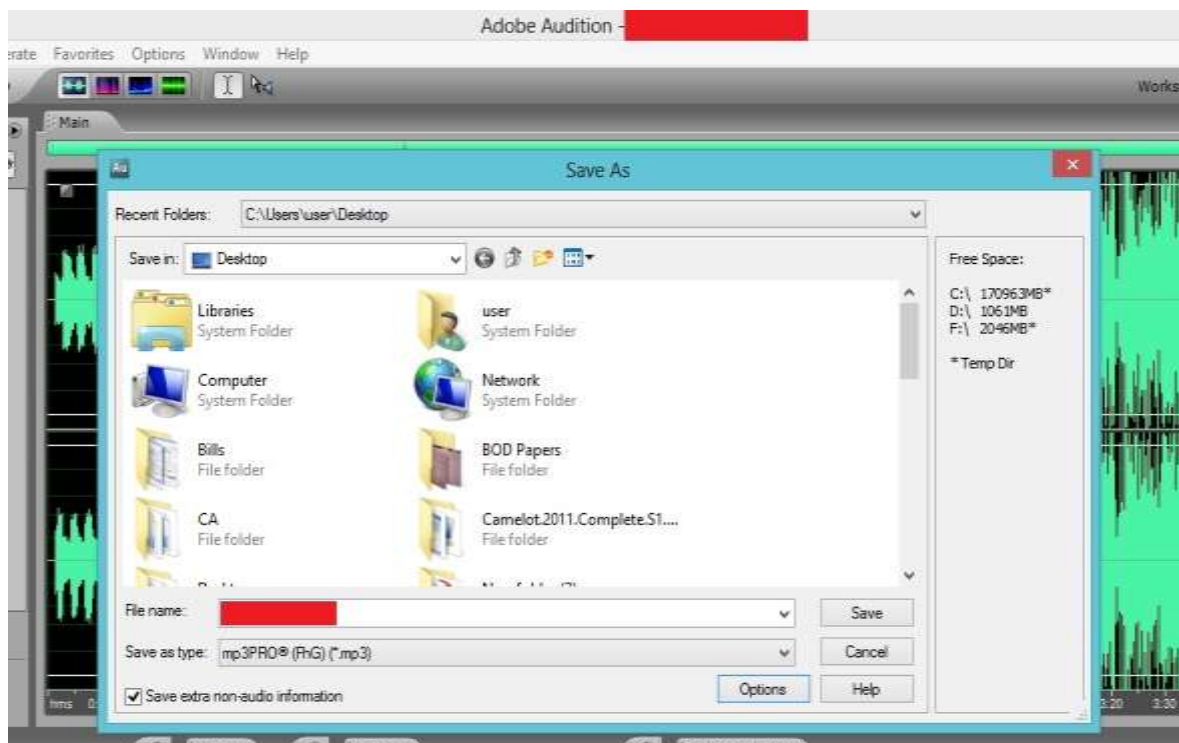
3. To save a copy of the current file while leaving the original version open and active, choose File > Save Copy As.



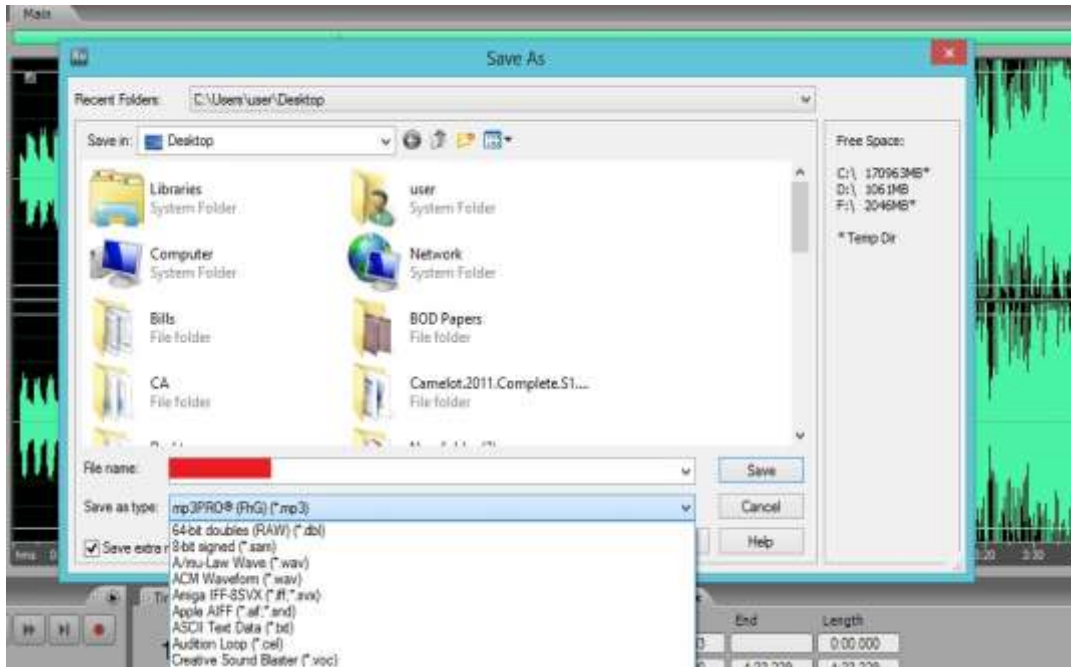
4. To save a selected portion of the edited as a new file choose File > Save Selection.



When you choose any one option from the above it will open a dialog box having different options.



Here you select the folder and its location where you want to put in your saved file. After then input a file name which is understandable to you in future. Select the file type. There

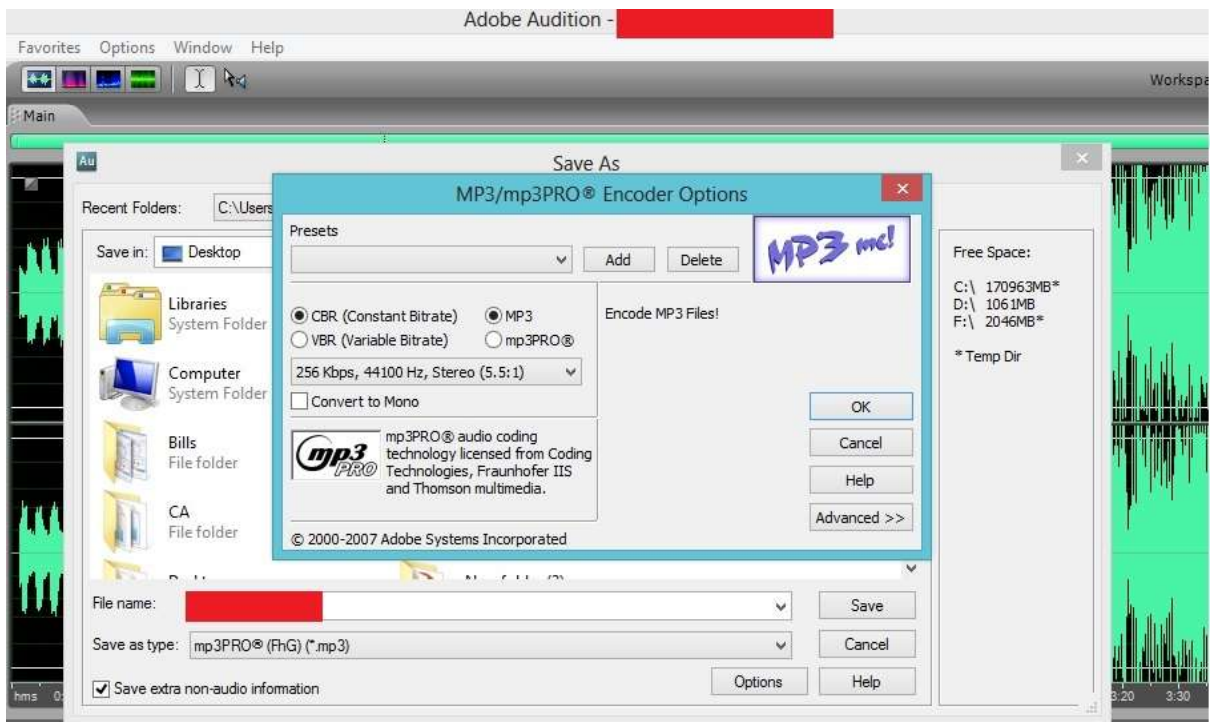


are

many file types available according to your requirement. You are very well aware about the file types.

Below file type select Save Extra Non-Audio Information to preserve markers and file information, such as loop and Broadcast Wave metadata. Deselect this option only, when you plan to burn your CD from another program.

Depending on the format you choose, additional options might be available. To view format-specific options, click Options.

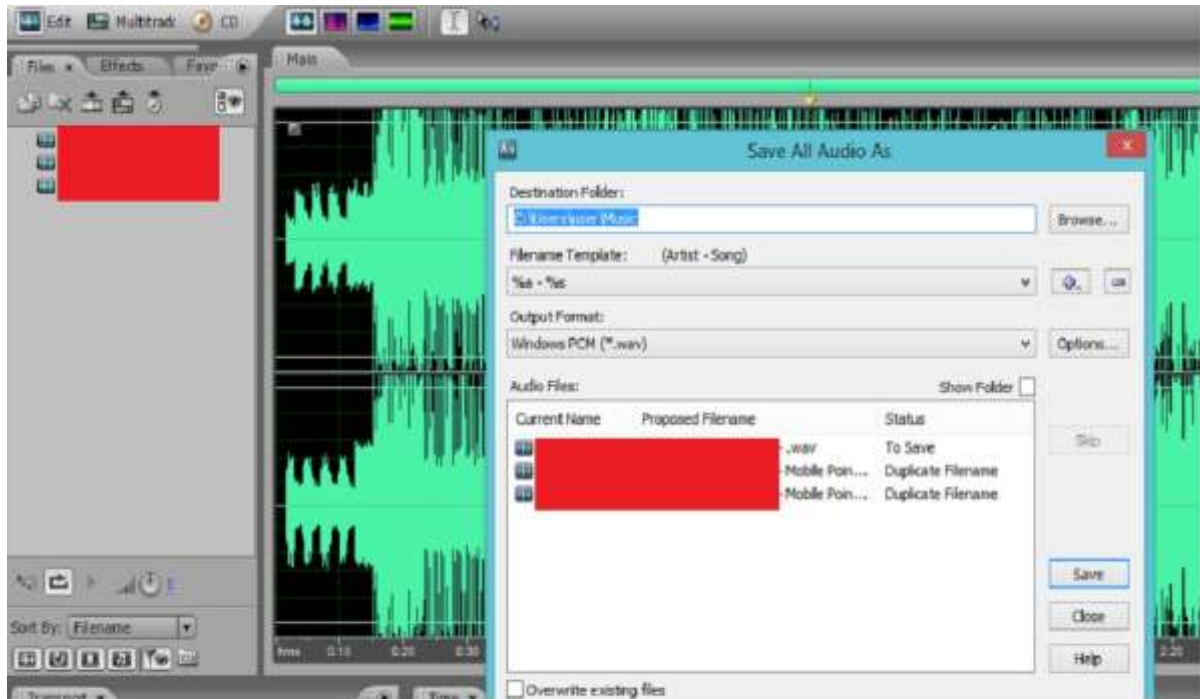


Here you can select extra options for your file if you want to. After selecting all the options click save. It will save your file at a location specified by you.

5. To save all open files in their current format select File > Save All options from the menu.





In either Edit View or Multitrack View, you can save a group of all open audio files to one format of your choice with File > Save All Audio As option. This option automatically choose filenames from file information such as Artist, Album and song if the files extracted from a CD.



Set the following options.

Destination folder: Select the destination folder where do you want to put your files..

Filename Template: You can input your information if files are not extracted from a CD.

You can add another file name template by pressing  sign and delete current filename template by pressing  sign.

Output Format: Select the output format for your files.

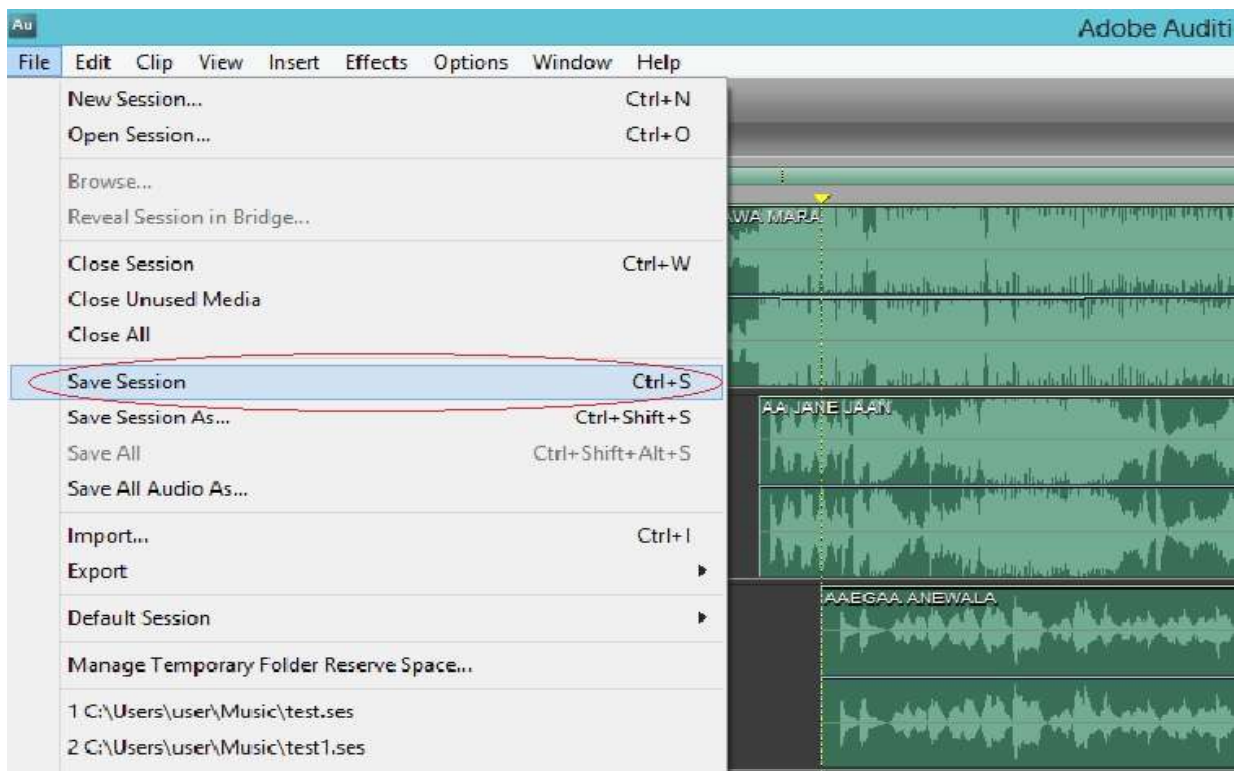
Audio Files: Audio files with specific names are shown here.

Show Folder: Displays the complete path for each file in the Audio Files list.

b. SAVING MULTITRACK SESSION

A multitrack session file is a small, non-audio file. It stores information about locations of related audio files on your hard drive, the duration of each audio file within the session, the envelopes and effects applied to various tracks, and so forth. You can reopen a saved session file later to make further changes to the mix.

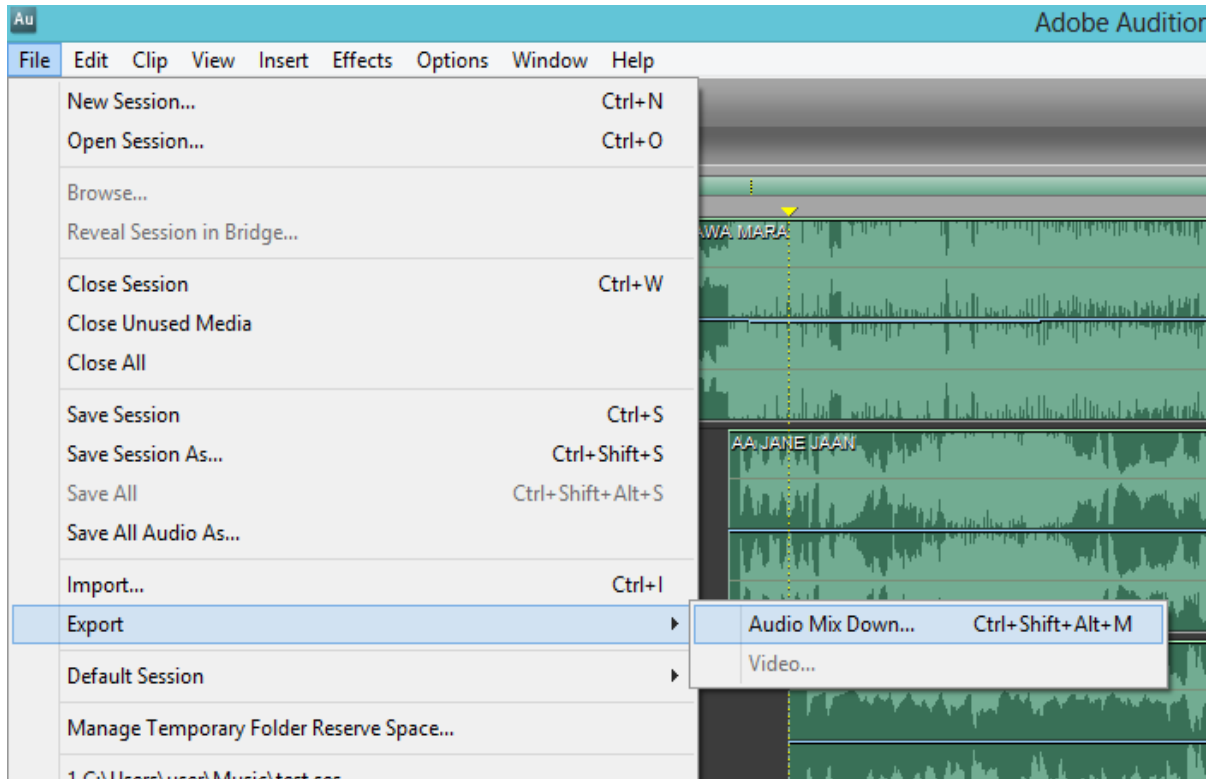
If you create multitrack compositions only for Adobe Audition, save session files in the native SES format and if you have planned to share multitrack compositions with other applications, save sessions in XML format.




It will open a dialogue box having different options. Give the filename, file type and other options mostly discussed in save options of single track. In option you can change the sample rate of the files. Choose any sample rate available in the drop down menu and then save the file.

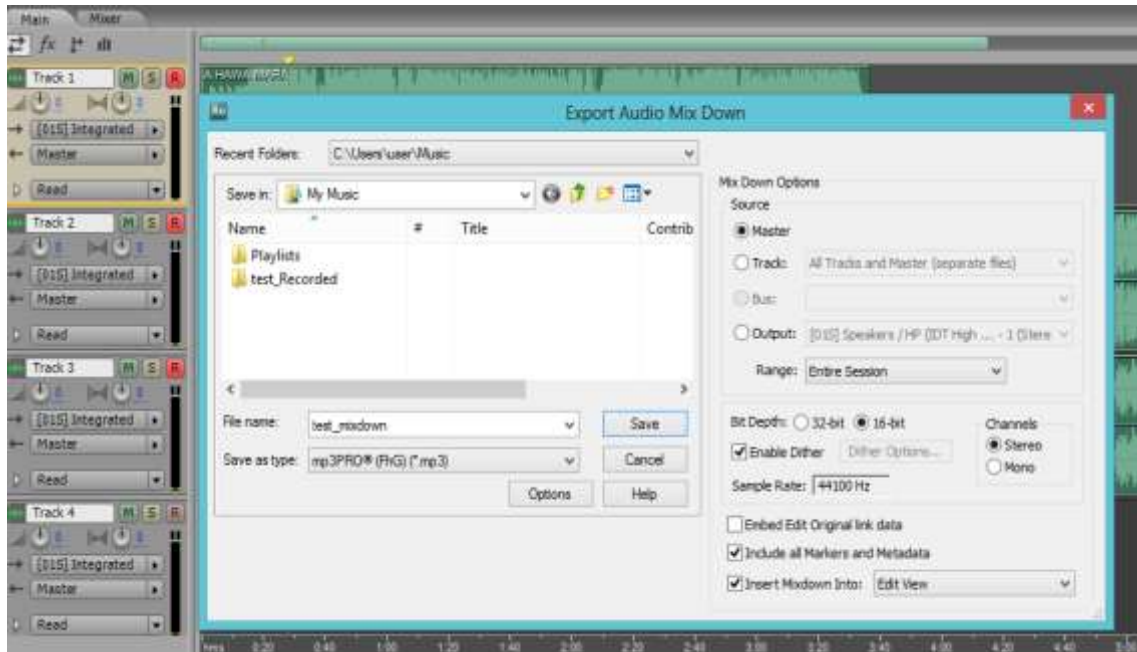
c. EXPORT A MULTITRACK SESSION TO AN AUDIO FILE

To export a multitrack session choose File > Export > Audio Mix Down.



Ctrl+Shift+Alt+M

1. If you want to export part of a session, use the Time Selection tool  to select the desired range.
2. In the Export Audio Mix Down dialog box, specify a location, name, and format for the saved file. If the file format you select can be customized, the Options button is available, click it to review or change settings, and then click OK.
3. In the Mix Down Options section, set Source, Bit Depth, and Metadata options.
4. Click Save.



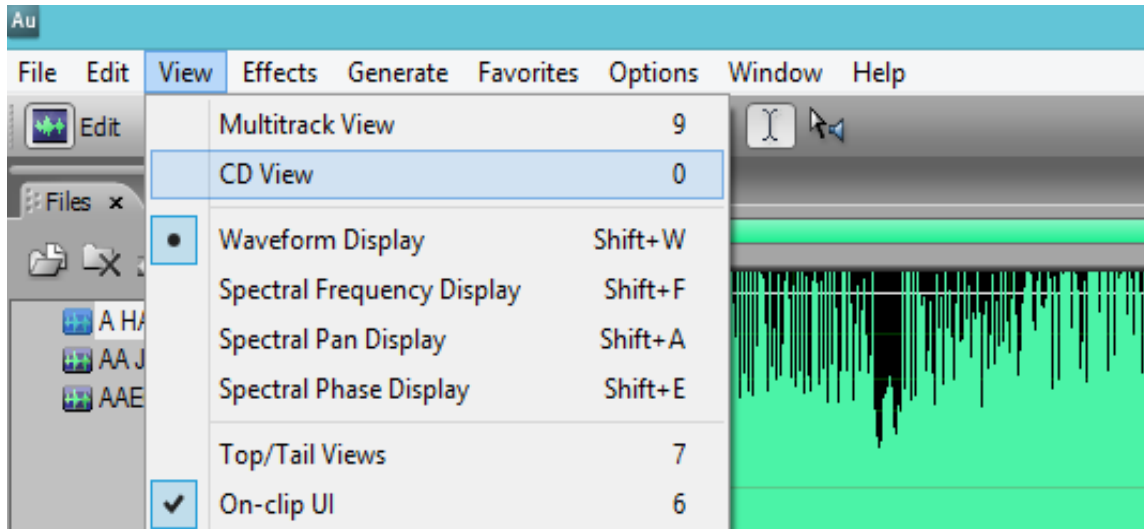
d. CLOSE FILES

Do any of the following:

- To close the current audio file in Edit View, choose File > Close.
- To close the current session file in Multitrack View but leave related media files open, choose File > Close Session.
- To close a CD list in CD View, choose File > Close CD List.
- To close all audio and video files not in use, choose File > Close Unused Media.
- To close all open audio, video, session, and CD list files, choose File > Close All.

e. BURNING AN AUDIO CD

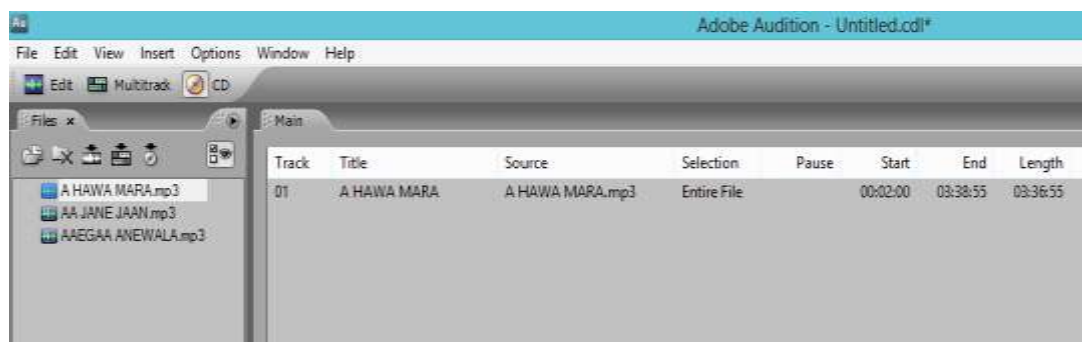
To open CD view click View > CD View or press 0.



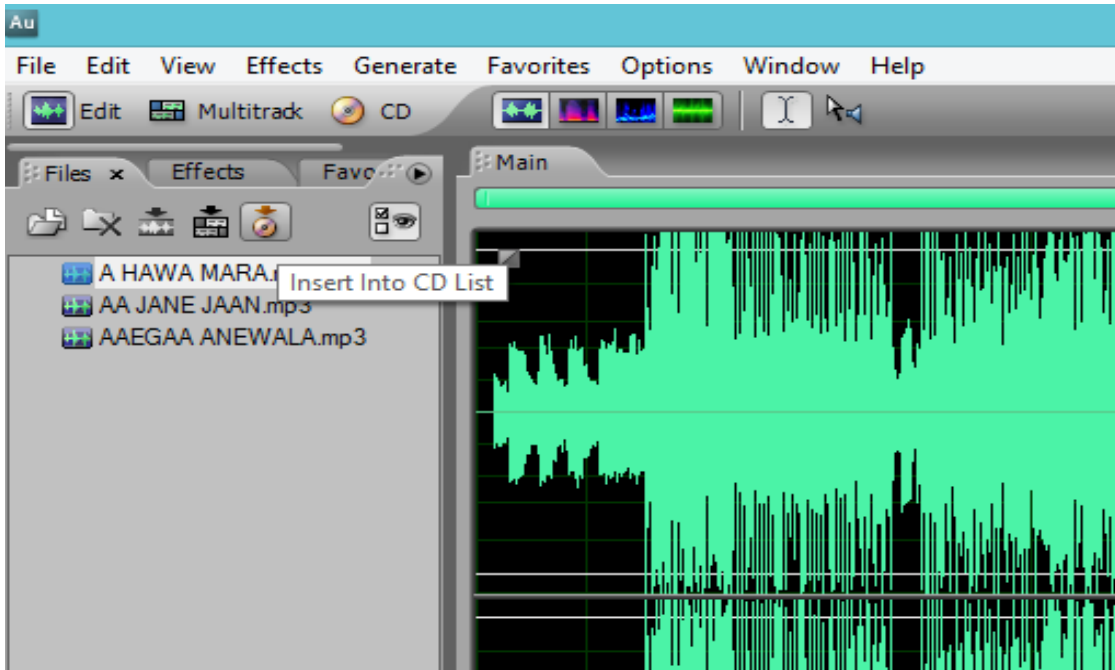
Below is the CD View.



You can insert files in Main list or drag the files into Main Pan or press Alt + Insert.



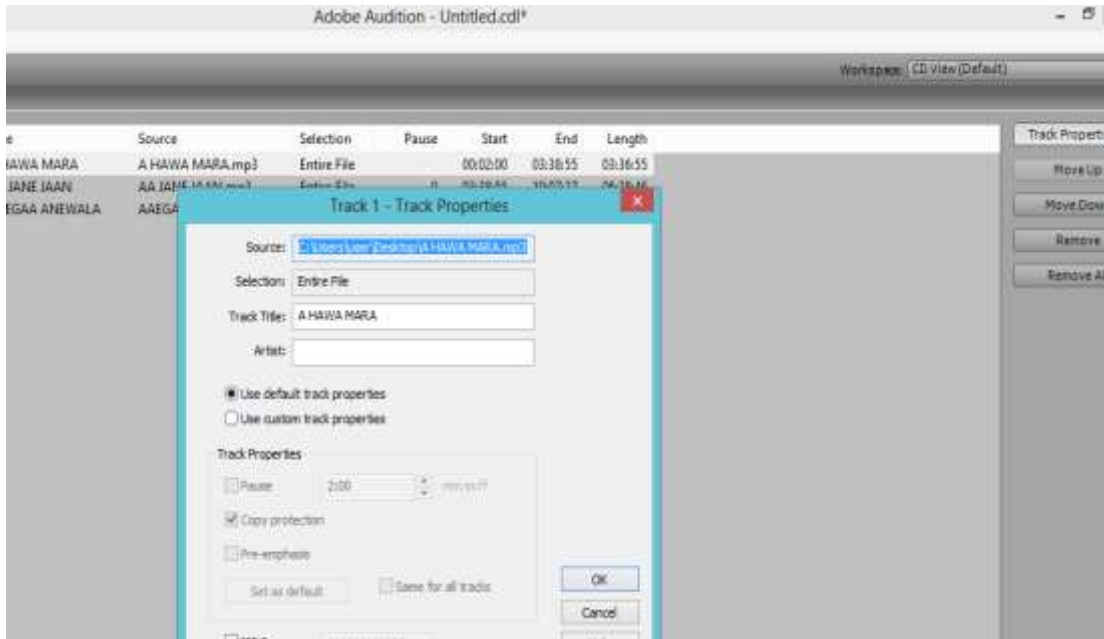
Or click the icon present on up side of the files list.



After inserting the files into CD list. You can select different options and save the .cdl file for future use.

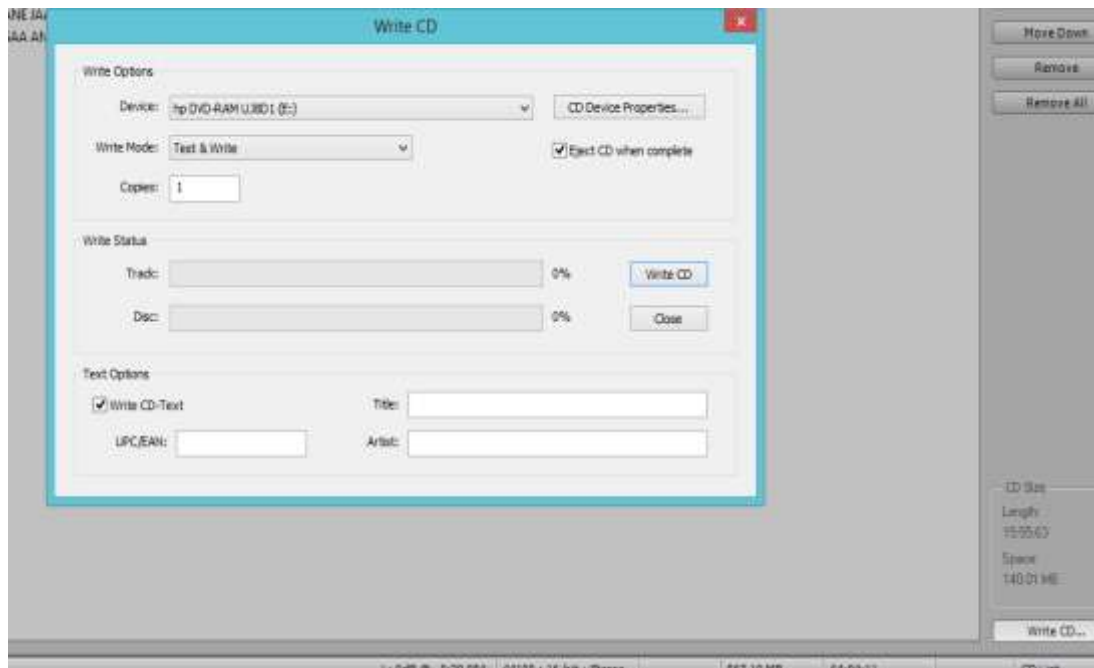


You can select track properties by clicking track properties on the right side of the pan. It will open a dialogue box. Select the source, selection and track title. Click ok. Reassign, re-arrange or remove one or all track in your CD view.



Click Write CD to set the write cd options. Set CD Device Properties.

Enter Text Options. Write CD-Text. Title and Artist. Insert a blank CD into CD drive and click Write CD. It will start the CD writing and shows the status. After burning disk will automatically eject.



That's it your burned CD is ready. Enjoy your first Audio CD.

SELF ASSESSMENT QUESTIONS

1. Explain in detail Sample rate and bit depth.
2. Define basic audio channels and types.
3. What is audio file format? Just name the file formats.
4. Explain lossy compressed audio format with two examples.
5. Give the name of five audio file formats and give description of any two.
6. How many ways we can save an edited audio file? Explain any one.
7. Define the export of your saved project.
8. How can we burn an audio CD?

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UNIT 9

INTERNSHIP

Written By: Muhammad Awais Khan

Reviewer: Zahid Majeed

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 - 3.3 Responsibilities**
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- 4. Internship Completion**
- 5. Supervising an intern**
- 6. Training**
- 7. Orientation**
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- 11. Frequently asked questions**

Student Intern Evaluation form...

Supervisor Evaluation of Student Internship

References

INTRODUCTION

This program is open to all nationalities and all ages over 18. Suitable for gap years or those taking a year out, grown-up gapers, career breakers, anyone interested in gaining overseas qualification experience or an internship for university credit or requirement. Also suitable for anyone just wanting to study and learn about audio and audio Production practice.

1. Internship

Internships are work-related learning experiences that provide students with the chance to gain important knowledge and skills in a career related that may or may not be directly related to their academic study. Therefore internships guide should be with very clearly defined learning objectives related to the professional goals of the learners. This internship will provide an exposure to the career fields of interest without making a permanent commitment.

Dear student the internships are unpaid positions providing you with practical experience. The Institute of Educational Technology, Allama Iqbal Open University will offer the internship to all of you after being enrolled and completion of course for academic credit. You can earn work experience by participating in an internship.

The interning in a field of choice will stand out on your resume and help them with their job search after the completion of course. Every one of you meets during the course of their internship will be a contact. The internship supervisor or guider will help or guide you in career path as you make their way into the business world. They know other people in the industry as well and will provide introductions. Your fellow interns also will become great contacts in the future.

All students should explore the possibility of earning academic credit through their place of internship/university.

1.1 Internship Criteria

Following is the criteria to meet for unpaid internships. According to certain standards listed below:

- Interns are not guaranteed a job at the end of the internship
- Interns must receive training from the company.
- Interns must get hands-on experience with equipment and processes used in the industry.
- Interns' training must primarily benefit them, not the company.

1.2 Why Provide Internships?

Following are the few reasons for providing internship

- Internships allow students the opportunity to apply their knowledge and skills in a
- Professional setting while still doing course.
- Internships offer carefully planned and monitored work experience with the goal being to
- gain additional knowledge from on the job exposure.
- Internships may also be part of an educational program in which students can earn
- academic credits from their college. Internships may be arranged independently from
- the curriculum in which students would gain work experience only.

1.3 Benefits to Department

- Immediate assistance to support projects
- Students will provide new ideas and viewpoints
- Salary Savings = No cost to department
- Effective public relations ambassadors for department; Recruitment and Workforce Planning
- Department/University ties are strengthened and communication is improved
- Permanent State employees can be relieved from performing minor or routine
- tasks allowing them to perform higher priority work
- Students energize a workplace with their enthusiasm and desire to learn.

1.4 Benefits to Students

- Career related experience
- Gains practical knowledge
- Opportunity to explore career avenues
- Valuable work experience for their resumes
- Potential to earn academic credit
- Increased self-confidence
- Enhances conventional classroom learning methods
- Letter of recommendation from departments supervisor
- Obtain references from co-workers

2. Internship Coordinator

The Internship coordinator will:

- Conduct on-campus recruiting to ensure students are aware that the department is offering internship opportunities
- Advertise the department's recruitment opportunities
- Coordinate the recruiting and screening of intern applicants
- Assist in the selection of interns
- Promote internship opportunities within the department
- Serve as the contact regarding the department's internships
- Review and revise the department's internship procedures as needed
- Serve as a liaison between intern supervisors and university.

3. Interns

- Internships allow students the opportunity to apply their knowledge and skills in a professional setting while still completing course.
- Internships offer carefully planned and monitored work experience with the goal being to gain additional knowledge from on the job exposure.
- Internships may also be part of an educational program in which students can earn academic credits from their college. Internships may be arranged independently from the curriculum in which students would gain work experience only.

3.1 How to Begin

Interns should:

- Analyze their skills, values and interests to determine the location and working environment desired.
- Prepare a resume and cover letter and have them critiqued at their career center.
- Network with alumni, college professors, friends, and family.

3.2 How to Apply

Internships are defined at the end of the course through departments, at Student Internship Positions. Please follow the directions on each individual internship flyer.

3.3 Responsibilities

Interns should:

- Adhere to agency policies, procedures, and rules governing professional behavior.
- Be punctual, and work the required number of hours at times agreed to by the intern and their supervisor.
- Notify their supervisor if they are unable to attend as planned.
- Behave and dress appropriately to the particular workplace.
- Respect the confidentiality of the workplace, its clients and its employees.
- If things are slow, take the initiative and volunteer for different tasks or other work.
- Discuss any problems with their supervisor and, if necessary, with the Internship coordinator at the department.

3.4 Intern Rights

Unpaid interns have the same legal rights as employees in regards to protection against discrimination and harassment. However, interns do not have the same rights as employees in the realms of unemployment compensation or termination procedures.

4. Internship Completion

At the end of the internship:

- The intern supervisor will provide the student with a letter of recommendation.

- The student Intern will evaluate the overall internship experience. The evaluation form must be returned to the internship coordinator.

5. Supervising an Intern

An intern will have a designated site supervisor who is responsible for providing orientation and supervision. This will be someone, available to the student on a regular basis, and possesses expertise in the area in which the intern will work. Even if the intern will rotate through various departments in order to gain broad-based experience, there still will be a single overall supervisor who over sees the internship as a whole. Dear student on choosing a supervisor, it is important to choose someone who is interested in working with university students; has the time to invest in the internship, especially during the first few weeks; and possesses qualities such as leadership, strong communication skills, and patience.

Because an internship is defined as a learning experience, proper supervision of the intern is essential. The supervisor is very important element in internship and he/she will be serving as a teacher, mentor, critic, and boss. Ongoing supervision of the student intern is the key to the success of the internship. This is especially true for students who do not have extensive work experience.

Acknowledging and identifying the different expectations between the workplace and school will help interns make a successful transition to the world of work.

An effective method of intern supervision is to have a set time (bi-weekly is recommended) to meet with the intern to review progress on projects, touch base, and provide feedback. Some supervisors do this over lunch; others choose a more formal setting.

The supervisor will oversee and assign the student intern's work. Supervisors will need to monitor the intern's time and submit an intern evaluation form provided by the intern's college for those receiving academic credit. The intern supervisor will also provide the student with a letter of recommendation.

6. Training

Training is as important as supervision. Establishing a training program will give the interns a clear understanding of what is expected, and information about the duties that will be supervised and evaluated.

The Institute of Educational technology, AIOU, Islamabad will designate a supervisor to oversee and assign the student intern's work.

Discuss the following with interns:

- What will the specific duties/responsibilities of the intern be?
- How will you provide the intern with regular feedback, guidance, and support?
- What training will the intern receive (if applicable)?
- What will the intern need to do if they will be absent from work?

7. Orientation

Establish goals and objectives, and clarify these goals and objectives before the intern begins working. Some interns need more guidance than others, and many factors must be taken into consideration. Consider the intern's cultural background, disabilities, learning style and experience. Evaluate his or her level of maturity and confidence. Is the intern a critical thinker or a creative problem-solver?

Plan to include the following in your orientation:

- **Information about the organization.** Office interns will review documents that are important for them to understand the big picture. If available, include an organizational chart that explains various roles and responsibilities of employees.
- **Structure.** Interns might not be familiar with formal workplace procedures (e.g., attendance policies, break times, days off). Make sure to clarify relevant policies and procedures to interns on their first day.
- **Introductions.** Take time in the beginning of the internship to introduce the intern to the people in your program. Allow more time for conversation with those employees who are likely to interact with the intern on a regular basis. Some interns, based on personality or culture, may be reluctant to seek out co-worker on their own. By making a special effort to encourage those contacts early on, interns will feel more comfortable asking for advice or support later.

8. Ongoing Training

Interns, as students, appreciate any opportunity to learn new skills or increase their knowledge. Developing a plan for training throughout the internship will keep students interested in the position and ready to tackle new challenges. Ongoing training may include the following:

- **Skill development.** There may be a need for training in specific skills such as computer programs, office equipment, or other tasks directly related to the job.
Even bright students with great potential will struggle if they are not instructed in the specifics related to successful completion of duties.
- **Shadowing.** Allow interns to participate in activities and meetings. Interns may have leadership potential but not understand the culture of your organization.
They will rely on their supervisor to educate them.
- **Questions.** Interns might not know when to speak or how or what to ask. Assist them in actively learning by explaining and clarifying everything. Suggest and encourage questions at appropriate times.
- **Professional conferences or association meetings.** If possible, offer interns the opportunity to attend training or networking events. It helps interns to get a feel for the overall mission of your organization, and at the same time make them feel that they are valued.

9. Evaluations

Evaluation is important to an intern's development and is an opportunity to identify strengths and weaknesses. It is helpful if supervisors evaluate throughout the entire internship, not just at the end. The evaluation should be planned as a learning experience and an opportunity for two-sided feedback.

Regularly scheduled evaluations help avoid common problems with internships, including miscommunication, misunderstanding of job roles, and lack of specific goals and objectives. You may find it helpful to schedule a preliminary evaluation early in the internship (in the second or third week). This will help you understand whether the intern's orientation and training was sufficient or if there are specific areas in which the intern has questions or needs further training.

Criteria to consider when evaluating an intern:

- Progress towards or accomplishment of learning objectives as stated in the learning agreement.
- Skill development or job knowledge gained over the course of the internship.
- Overall contribution to the mission of the organization.
- Dependability, punctuality, attendance.
- Relations with others, overall attitude.
- Potential in the field.

The student will also evaluate the internship experience, which is important in determining the value of the work experience for future interns. Categories might include:

- Was there educational value or merit in the assignment?
- Did the position live up to its initial description?
- Was the supervisor receptive to your ideas?
- Does the experience relate to your major or career goals?
- Did you receive a proper job orientation?
- Was the supervisor willing and/or capable of answering questions?
- Did you develop good work habits?

10. Internship Completion

An internship should have a clearly stated end date that is identified before the internship begins. Completing a formal evaluation process such as the one described above can help both the site supervisor and the intern bring closure to the experience. A letter of recommendation from the Intern supervisor shall be given to the intern on the last day of work.

You also may want to have some form of acknowledgment such as a lunch with coworkers in the final week of the internship. Because co-workers often have extensive contact with interns, this type of event can be a positive way to recognize the contribution of other employees as well as the intern.

At the end of the internship, the intern supervisor will:

- Provide the student with a letter of recommendation.

- Complete college/university evaluation to assess the intern's progress and skill development (if applicable).
- Evaluate the overall internship experience. This feedback is not only essential for making necessary program improvements, but also for recognizing those departments that provide outstanding learning opportunities. The evaluation form must be returned to the internship coordinator.

11.Frequently Asked Questions

Why should I look at an internship?

Internships allow students the opportunity to apply their knowledge and skills in a professional setting. Students will gain valuable work experience and the opportunity to explore career avenues.

How do I find an internship?

The department advertises their intern positions at local radio stations and production houses.

Is my internship paid?

There are no paid positions available. The department will provide internships to students as volunteers or for academic credit.

What happens at the end of my internship?

You will receive a letter of recommendation from your internship supervisor. You will also have an opportunity to evaluate the department's internship program.

Institute of Educational Technology

Student Intern Performance

Print all information clearly!

Intern's Name _____

Semester of Internship: Spring Autumn

Year: _____

Intern's supervisor: _____

What resources did you use to find your internship? (Check all that apply)

Career Services Office/Internship Coordinator Faculty Internet Site

Family Friend Other: _____

Internship field: _____

Project Title: _____

Details of the final project: _____

Department _____ Date _____

This evaluation is completed by the student.

Please rate the following aspects of your Internship placement on the basis of this scale:

Excellent (Consistently exceeds expectations) Good (Sometimes exceeds expectations)

Average (Meets expectation) Poor (Rarely meets expectations)

N/A Not Applicable (Not applicable to this internship experience)

Select one evaluation level for each area by marking an “X” under the level that represents the internship.

	Excellent	Good	Average	Poor	N/A
Work experience relates to my area of study					
Adequacy of employer supervision					
Helpfulness of supervisor					
Acceptance by fellow workers					
Opportunity to use my training					
Opportunity to develop my human relations skills					
Provided levels of responsibility consistent with my ability and growth					
Opportunity to develop communication skills					
Opportunity to develop my creativity					
Cooperativeness of fellow workers					
Opportunity to problem solve					
Opportunity to develop critical thinking skills					
Provided orientation to the organization					
Attempt to offer feedback on my progress and abilities					
Effort to make it a learning experience for me					
Gave me a realistic preview of my field of interest					
Adequate training					
I feel I am better prepared to enter the world of work after this experience					
I felt I was productive for the department					
Through this internship I had the opportunity to use and develop my:					
Interpersonal/human relation skills					
Oral communication/presentation skills					
Creativity					
Problem solving abilities					
Critical thinking skills					
Writing skills					
My Internship experience:					
Confirmed my interest in a career in this line of work					
Has made me decide to pursue a different career path					
Overall Internship experience (circle one)	Excellent	Good	Fair	Poor	

Comments if any

Would you work for this supervisor again?

Yes

No

Uncertain

Would you work for this agency where doing internship again?

Yes

No

Uncertain

Would you recommend this agency where doing internship to other students?

Yes

No

Uncertain

Why or why not?: _____

Intern's Signature: _____ **Date:** _____

Thank you very much for completing this evaluation of your internship. We take your comments very seriously. Please return this evaluation to the university.

Supervisor Evaluation of Student Internship

Print all information clearly!

Intern's Name: _____

Intern's Supervisor: _____

This internship started on (date) _____ **and**
will be as completed on (date) _____

Do you permit the student to receive a copy of this evaluation? Yes ___ **No** ___

Please rate the following aspects of your Internship placement on the basis of this scale:

Excellent (Always demonstrates this ability/consistently exceeds expectations)

Good (Usually demonstrates this ability/sometimes exceeds expectations)

Average (Sometimes demonstrates this ability/meets expectation)

Poor (Seldom demonstrates this ability/rarely meets expectations)

N/A Not Applicable (Not applicable to this internship experience)

Evaluation of personal qualities of the intern observed during the internship. Select one evaluation level for each area by marking an "X" under the level that represents the intern's performance.

	Excellent	Good	Average	Poor	N/A
Ability to learn					
Observes and/or pays attention to others					
Asks pertinent and purposeful questions					
Seeks out and utilizes appropriate resources					
Accepts responsibility for mistakes and learns from experiences					
Open to new experiences; takes appropriate risks					
Reading/Writing/Computation Skills					
Reads/comprehends/follows written materials					
Communicates ideas and concepts clearly in writing					
Works with mathematical procedures appropriate to the job					
Attention to accuracy and detail					
Listening & Oral Communication Skills					
Listens to others in an active and attentive manner					
Comprehends and follows verbal instructions					
Effectively participates in meetings or group settings					
Demonstrates effective verbal communication skills					
Creative Thinking & Problem Solving Skills					
Seeks to comprehend and understand the "big picture"					
Breaks down complex tasks/problems into manageable pieces					
Brainstorms/develops options and ideas					
Respects input and ideas from other sources and people					
Demonstrates an analytical capacity					
Interpersonal & Teamwork Skills					
Relates to co-workers effectively					
Manages and resolves conflict to a team atmosphere					
Supports and contributes to a team atmosphere					
Controls emotions in a manner appropriate for work					
Demonstrates assertive but appropriate behavior					
Basic Work Habits					
Reports to work as scheduled					
Is prompt in showing up to work and meetings					
Exhibits a positive and constructive attitude					
Dress and appearance are appropriate for this organization					
Character Attributes					
Brings a sense of value and integrity to the job					
Seeks to serve others					
Refrains from gossip/respects the privacy of others					
Behaves in an ethical manner					
Respects the diversity (religious/cultural/ethnic) of co-workers					
Overall performance of the intern (circle one)	Excellent	Good	Average	Poor	

Comments about the project

Would you supervise this intern again?

Yes

No

Uncertain

Would your agency host this intern again?

Yes

No

Uncertain

Would you recommend this student to other organizations?

Yes

No

Uncertain

Why or why not? _____

Evaluator's Signature: _____ **Date:** _____

Title/Position: _____ **Telephone:** _____

Thank you very much for completing this evaluation of your intern. We take your comments very seriously. Please return this evaluation to the FAX number or address listed above.

Reference

https://jobs.ca.gov/PDF/Student_Internship_Program_Guide.pdf