Generally the low frequencies are most troublesome when mixing. This is very important when dealing with the acoustics of a room and standing waves.

FREQUENCY

We can see that a wavelength will periodically cycle. The amount of time it takes a wavelength to complete a full cycle, in seconds, is known as the frequency (measured in hertz). Frequency is directly related to pitch. If we have two wavelengths and one is twice as fast as the other, we can say that it is double the frequency. For every doubling in frequency, we will get a doubling in pitch.

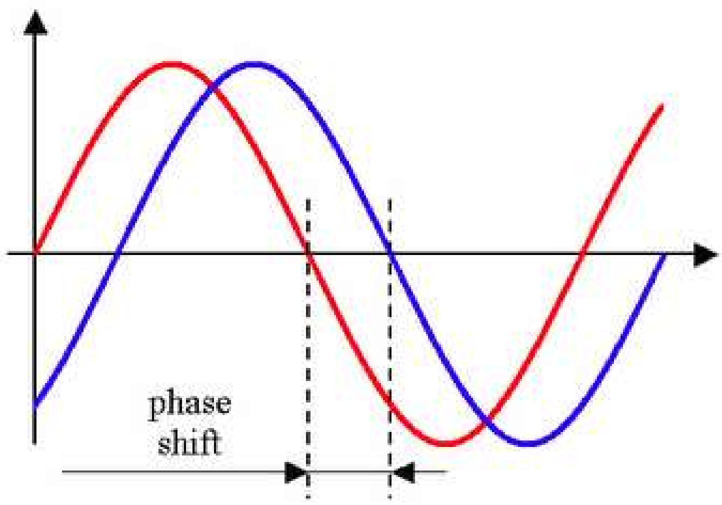
For example, A4 on the keyboard is 440Hz. Therefore A5 will be 880Hz. The A4 and A5 are an octave apart on the keyboard, which is 12 semitones.

PHASE

Phase can be used to describe the relationship between different waveforms, as well as to pinpoint a waveform's amplitude as it cycles through time. This is a crucial concept that is important in many aspects of music production, from microphone placement and sound design, right through to room acoustics and general mixing.

If we analyse a sine wave, we can think of it as positive and negative air pressure. (Compression and expansion of air molecules, as the speaker moves back and forth) We can see how the waveforms air pressure, increases and decreases equally through time (milliseconds).

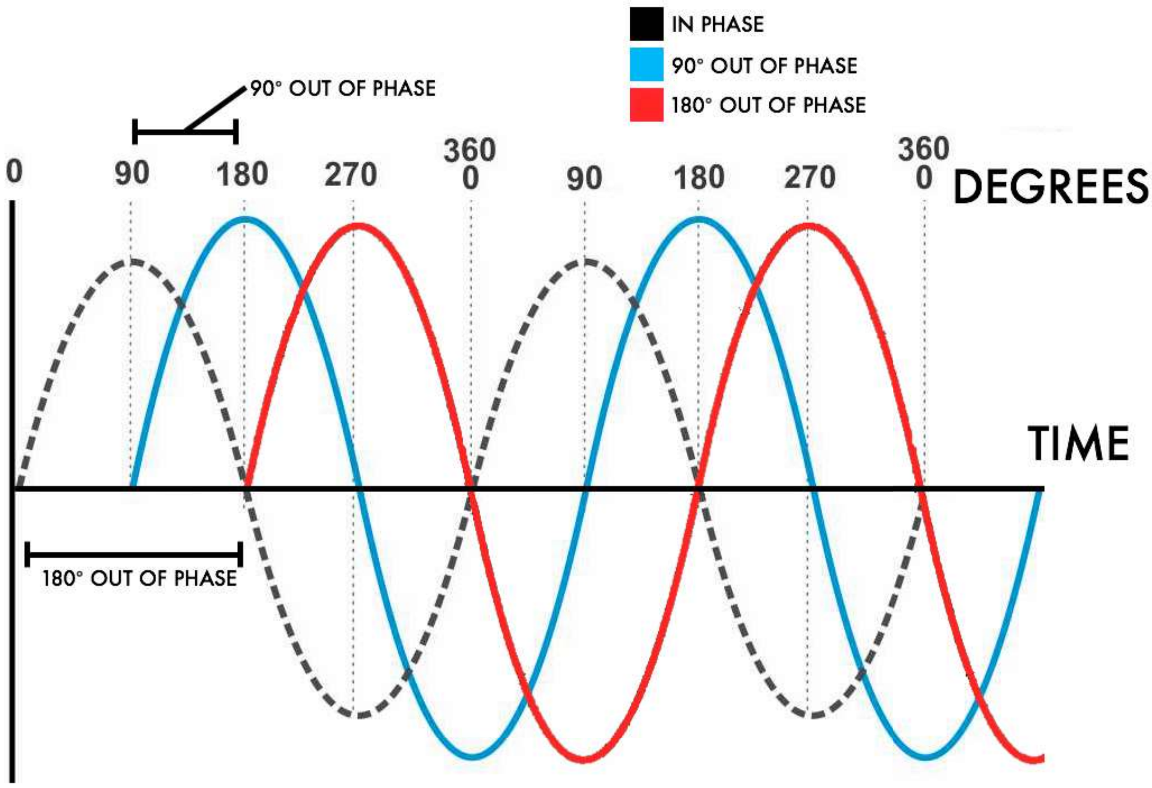
By drawing a zero line through the waveform, we can see that the waveform reaches zero crossing points throughout a single cycle.



If we look at this waveform without the concept of time, we can actually change this waveform into a circle, the only reason we don't look at waveforms like this, is because we usually need to look at them relative to time.

We can now see that this cycle's zero crossing points can be measured in degrees of a circle. This is also how we measure phase.

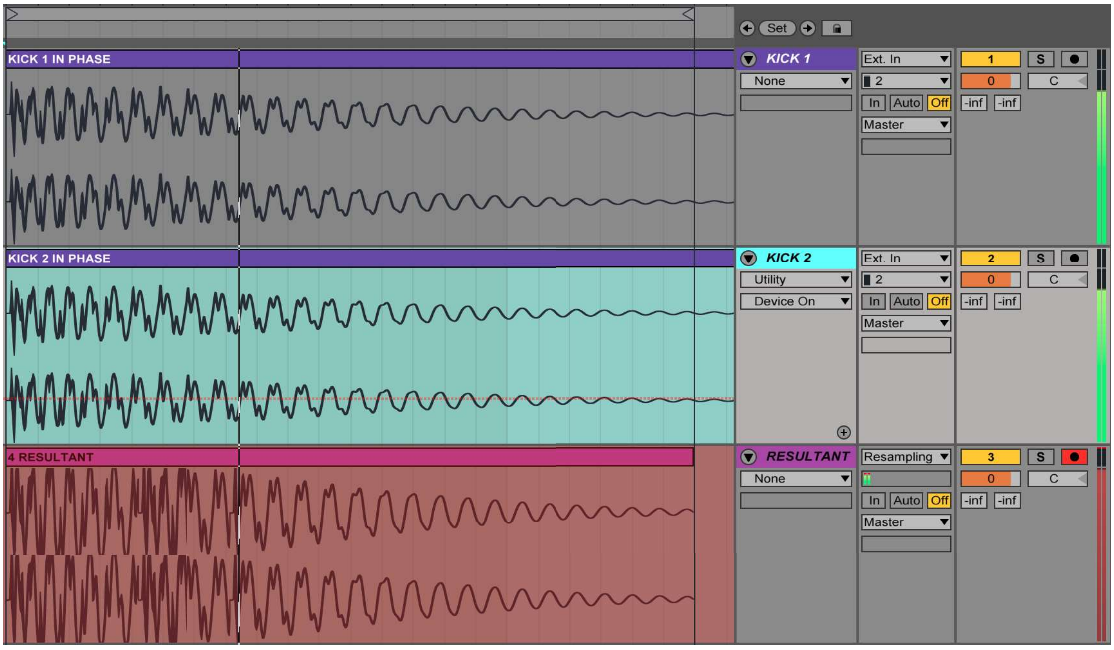
If we start two waveforms a few milliseconds apart from each other, they will be out of phase, due to the cycle starting at different points in time.



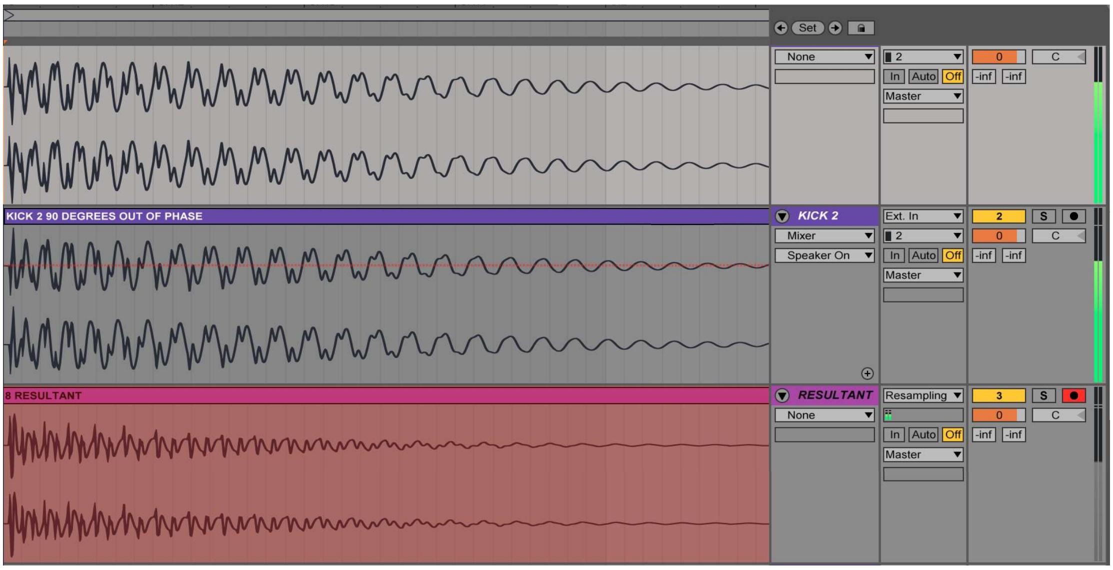
When we mix sounds, we will often be adding multiple waveforms together. If we stack some sine waves on top of each other, we can see how their phase relationships will cause them to add together (sum), or cancel each other out.

Two identical signals directly out of phase will cause complete phase cancellation.

Two identical waveforms in phase with each other will sum together and increase in volume.

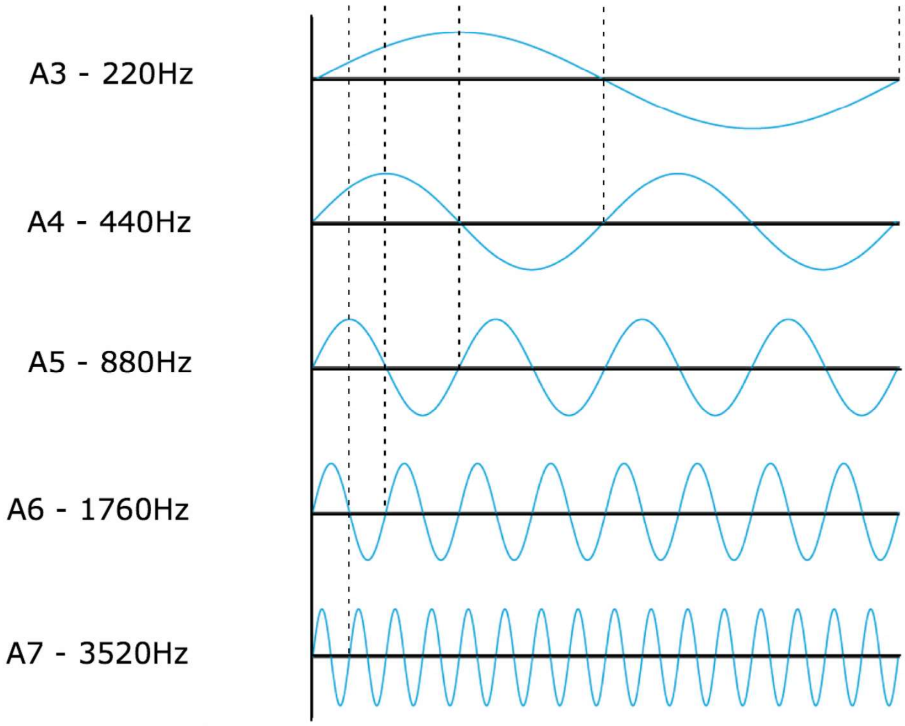


Two waveforms out of phase may have a mixture of summing and cancellation at different frequencies.



This is fundamentally how we use up headroom when we sum all our tracks together to one stereo track, which will cumulatively get louder as we add more instruments. Hence, we can usually get more power, focus and volume with fewer instruments.

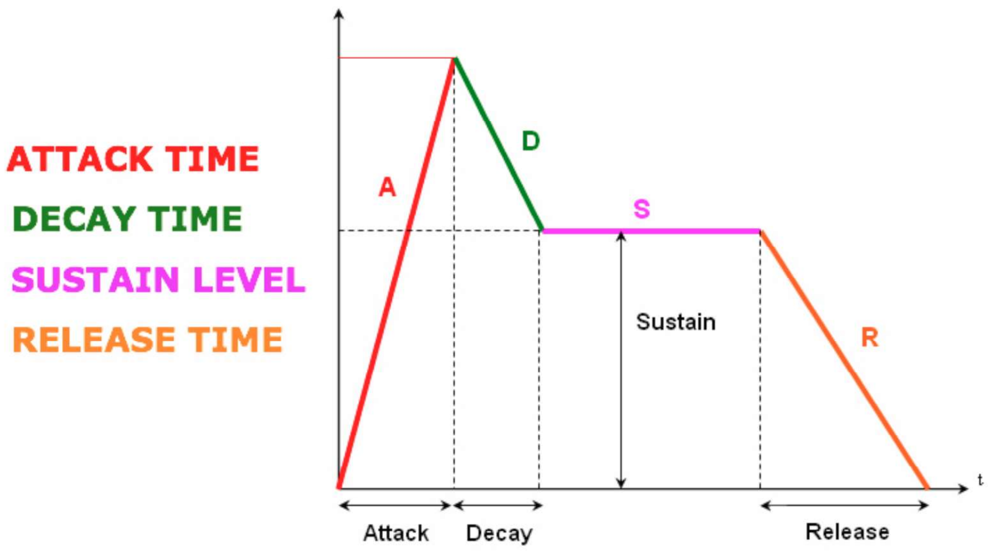
Finally, if we have two sine waves one octave apart, and we sum them together, we can see how these sine waves slot together, which means they are in harmony with each other.
The resultant wave will be a combination of the two waves.



ENVELOPES

We also perceive sound by their amplitude envelopes. This is a more logical, three-dimensional way of perceiving the amplitude of a sound over a period of time, rather than any instantaneous moment.

We can break a sound down into different sections or phases:



There are more complicated versions of this 'envelope', but this is the type that we are going to be using most frequently.

The **attack** phase is the initial period of time, when the sound builds up to its peak or transient.

Decay is the time for a sound to go from its peak volume, down to its sustain level.

Sustain is the level at which a note will sustain if a key is pressed. For example, a VST synthesizer will commonly loop at a certain sustain level when a key is pressed on a midi controller.

Release the time it takes for the note to reduce to silence once we take our finger off the key.

These factors make up how audio sounds the way it does.
We could say that most drum samples have a very sharp attack, giving them a percussive sound. They also tend to have a medium/sharp decay and no sustain or release.

A string on the other hand, will have a very slow attack, and a more gradual release. A piano will have a fast attack and a fairly fast decay, then the sustain will continue if the sustain pedal is used, followed by a fairly sharp release.

DECIBELS

Now we will briefly go into the unit of measure for amplitude, the decibel. This is quite an unusual unit of measure, because the decibel is actually a relative unit of measure, which can be used in many different contexts.

Decibels are used to express a ratio between a reference level and the level we are measuring. These can be different for power, sound pressure, digital, and analog equipment.

The meaning of a decibel changes completely depending on what suffix is used after the DB. This is because the part that comes after the decibel will indicate what the decibel is being referenced to.

For example:

We have dBu which is used in analog equipment to measure voltage, 0dBu is equivalent to 0.775volts. This is not to be compared to a VU meter. A VU meter is a slow analog meter that measures the average amplitude of the signal, (RMS) this closely replicates the human ear's perception of sound. A VU meter can only be compared to another unit of measure when using a sine wave test tone. 0 VU corresponds to +4 dBu.

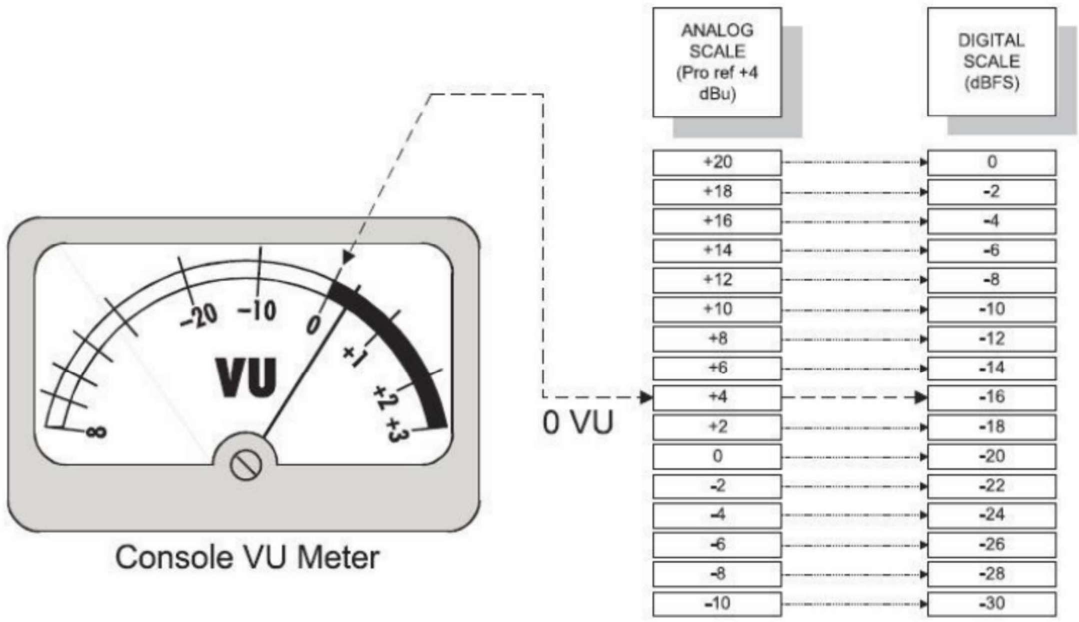
We also have dBv, which is very similar to dBu, but is referenced to 1 Volt rather than 0.775 volts.
When we mix within DAWs we can use a combination of different metering, knowing how each of them compare to each other is important.

We should mix VU meters so that the needle dances around the 0 VU point. This means we have the optimum signal to noise ratio (equivalent to +4dbu)

When we are mixing on the channel faders in our DAW, 0 VU (+4dBu) should be equivalent to between -12 and -16 dB Full Scale

The reason for this change between analog hardware and DAW's is because from -10dBfs to 0dBfs within a DAW is actually supposed to be used as headroom.

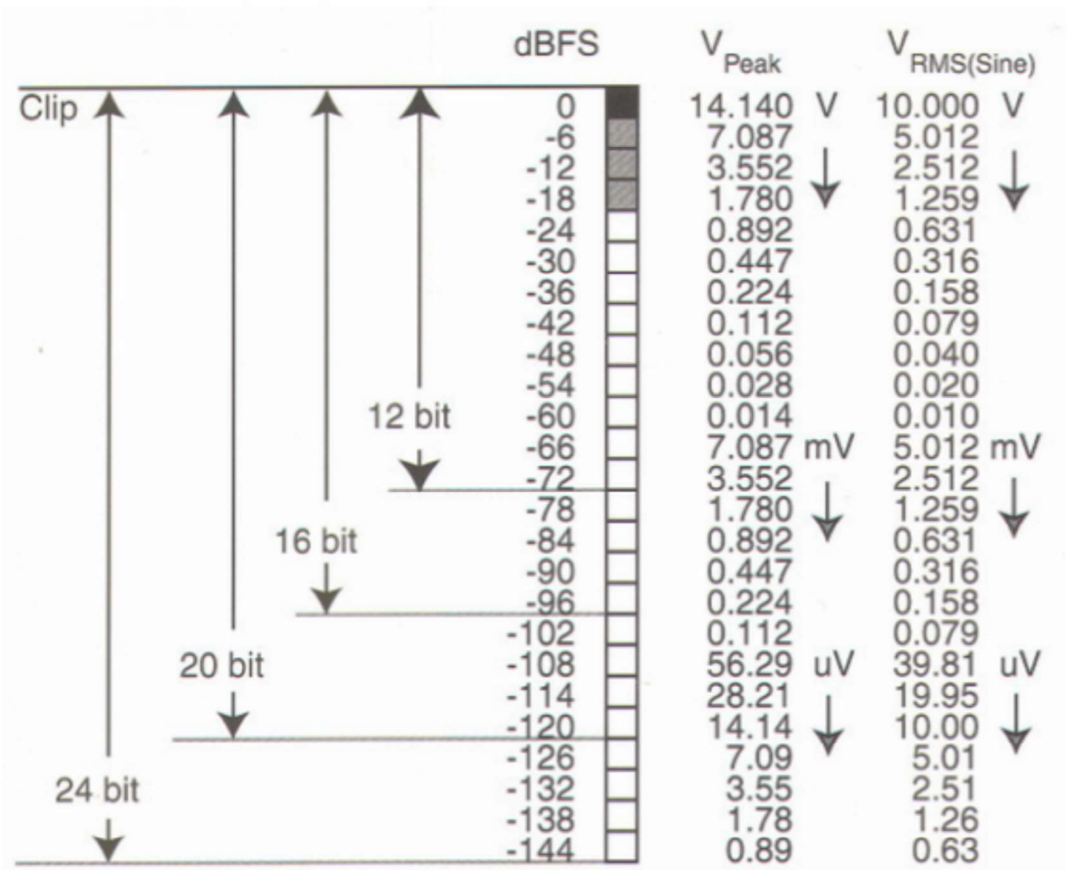
We have dB SPL, which is 'Decibels Sound Pressure Level.'
This is referenced to the quietest sound the human ear can hear, which is 20 micro-Pascal's. This is the unit of measure used for things such as, measuring the loudness of a set of headphones. 1 Pascal is equivalent to 94dB SPL. (The sound of a motorbike going past)



A comparison of VU, analog and digital metering.

Finally we also have dBFS or 'decibels full-scale' which is used in modern day DAW's and digital audio. the dBFS scale works backwards from 0 dBFS. Zero dBFS is equivalent to the maximum volume achievable by a piece of digital equipment without clipping or distorting. Therefore a reading should never exceed 0dBFS, else it will cause digital clipping.

Because this level is variable, we measure backwards from 0dBFS, down to the noise floor. This scale is logarithmic, so we will struggle hearing sounds after around -48dBFS. As we can see decibels and loudness can be quite complicated, but to keep things simple, loudness in every day life, is most commonly referred to in dB SPL whilst levels on analog equipment are commonly measured in dBu. However, throughout this eBook, we will be working within the Ableton Live DAW, so we will be referring to dBFS (Decibels Full Scale) unless otherwise stated. The dynamic range of a DAW can change dependant on the bit-depth. The more bits we have, the larger the dynamic range, because the bits can be used to represent a larger range of values.



A diagram to show how bit-depth affects the dynamic range

LOGARITHMIC SCALES

As mentioned in the last section, decibels use a logarithmic scale. The reason for this is because it is similar to how the human ear interprets sound. The dynamic range of human hearing goes from 0dB SPL to 120dB SPL. This range could not be plotted on a scale in a conventional manor.

This is where logarithms come in. a Logarithmic scale is basically a condensed scale that allows us to plot more easily onto graphs.
It does this using a mathematical function, so that logarithmic numbers increase exponentially. We can also see this on the meters of our volume faders.

We can also see a logarithmic scale by opening up a spectrum analyser; the frequency lines are spaced differently to show certain frequency ranges in more detail than others.

Here is a chart of the logarithmic value in decibels. This should help you to understand that a doubling in dB doesn't necessarily correlate to a doubling in loudness.

dB	Relative Power Increase
0	1
1	1.26
3	2
10	10
20	100
30	1000
50	100'000
100	10'000'000'000

The part where this gets confusing is that as we can see, 3 dB is a doubling in power (amplitude). However, a doubling of voltage would be equivalent to 6dB's, (sound pressure level) whilst in terms of psychoacoustics, our perception of double the loudness would be more like 10dBs.

This can all get very confusing, but as long as we can grasp the concept that we measure our dynamic range in decibels, and that this is an exponential scale, then we will have more than enough knowledge to mix tracks down effectively.

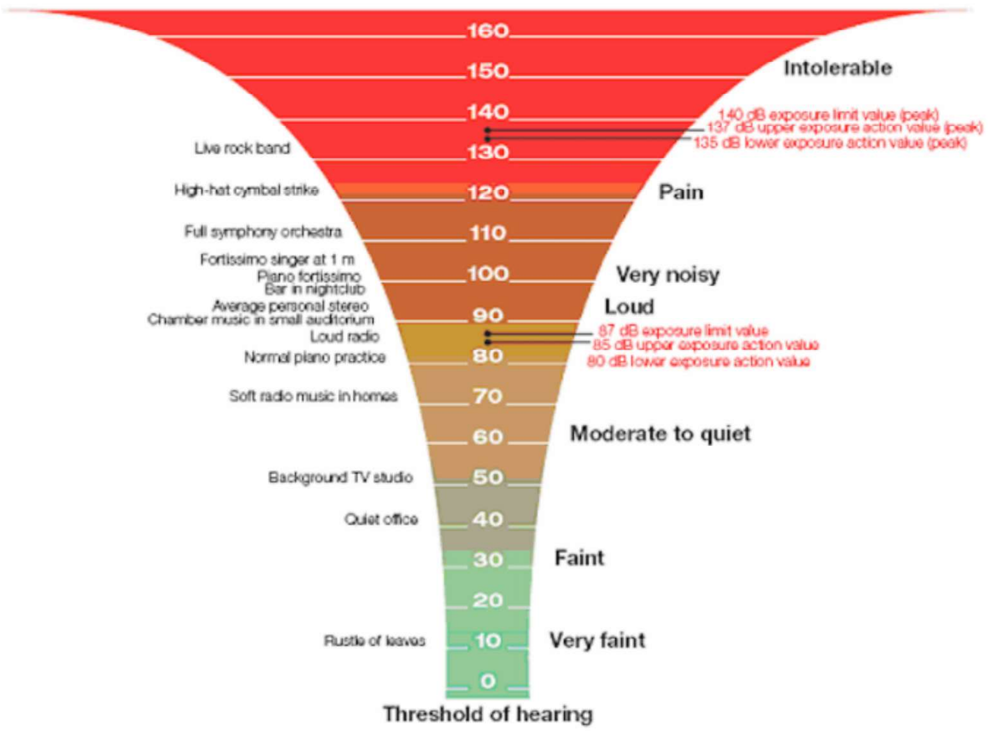
HOW WE HEAR

As we are aware, sound is comprised of fluctuations in pressure. (Compression and rarefaction of atmospheric pressure.) Or in simple terms, a speaker cone moving back and forward, causing vibrations in the air.

When these vibrations reach the ear, they are projected into the ear canal by the outer ear. Once inside the ear canal, the vibrations will then reach the eardrum and cause it to vibrate. These vibrations are transduced into tiny movements, which then go to the inner ear, where thousands of tiny hair receptors transduce these vibrations into a neural signal, which then passes down the acoustic nerve into the brain, where it is interpreted as sound.

These tiny hairs within the inner ear can easily be damaged by loud noises. They will also naturally deteriorate over time, with age. Once they are damaged they do not recover, so be wary of listening at loud levels. Damage to these tiny hairs leads to tinnitus (ringing ears) and permanent hearing damage.

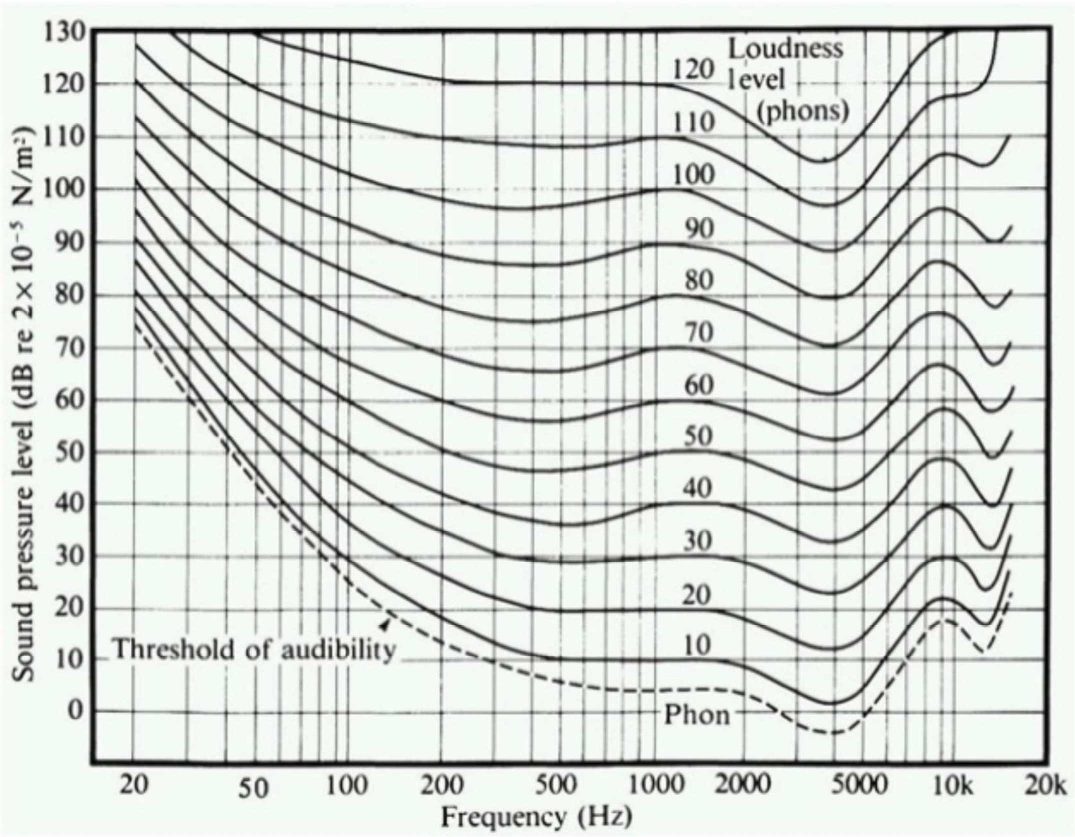
This hearing loss occurs from long exposures of over approximately 90dB SPL. Typically anything over 115dB SPL will cause significant temporary or permanent hearing loss even at relatively short exposures. It's also worth noting that it's generally the higher frequencies that will do the most damage, rather than the sub frequencies, as many people would think.



HEARING IS NOT LINEAR

Human hearing isn't linear. Many experiments have concluded that our hearing is more sensitive around the 3-5KHz range. Frequencies that are in the extreme high or low ranges will often need to be significantly louder for us to perceive them as the same level as a frequency in the 3-5khz range.

This can be shown with this graph of equal loudness curves below.



BASICS OF SOUNDS AND WAVES SUMMARY & KEY POINTS

- The most common waveforms are sine, triangle, square, pulse, saw & white noise
- During mixing we will come across complex combinations of these waves
- Any sound can be created using multiple sine waves. This is the fundamental theory behind additive synthesis
- Sine waves are the purest waveforms, consisting of a fundamental frequency only
- Triangle waves contain odd harmonics and are slightly brighter than sine waves
- Square waves contain odd harmonics and sound hollow; these are good for replicating wooden sounds
- Saw waves contain all the frequencies in the harmonic scale and are very bright
- White noise contains all the frequencies in the spectrum at full amplitude
- Sound propagates through air at 343 meters per second (1125ft per second)
- This can be approximated at 3ms per meter (1ms per foot)
- Waves can be described in terms of amplitude, wavelength, velocity, frequency, timbre, phase & envelope
- A peak measurement is the loudest part of the signal
- RMS means root-mean-square which is a more average way of interpreting a signal
- RMS is closer to how the human ear hears sound
- There's 360 degrees to one periodic cycle
- Hertz (Hz) is the measurement of frequency

- 1Hz is 1 cycle per second
- Wavelength = velocity of sound in air (343 meters) / Frequency (Hz)
- A doubling in frequency is also a doubling in pitch (1 octave)
- The range of human hearing is 20Hz - 20KHz
- Phase is measured in degrees
- Similar sounds that are in phase with each other will sum together
- Similar sounds that are out of phase will cancel each other out
- A common way of representing and manipulating a sounds amplitude over time, is b with ADSR envelopes
- ADSR envelopes and LFO's can also be used as modulation sources
- The decibel is a relative unit of measure to a given reference level
- There are many different types of decibel, within music DB SPL and DBFS are the most common
- 0 DBFS is the loudest a signal can go within software without clipping
- Decibels are logarithmic, a doubling in DB is not the same as a doubling in loudness
- Sound is the compression and rarefaction of air pressure
- Human hearing is not linear; our hearing is most acute in the vocal range of 3-5Khz