MIXING EXPLAINED

Mixing is the process of 'combining', 'blending' or 'balancing' multiple sound sources together. This gives us our resulting mix.

The word 'mix', is an abbreviation of the word mix-down, which means that we will be Mixing 'X' number of tracks down, into a single stereo track.

We can think of mixing as being lots of cumulative processes that achieve a final result. It only takes one tiny detail to be wrong in any one of these processes, and the integrity of the entire mix could be ruined. Mixing is quite a subjective artform, so there are plenty of different ways to get the same job done, as well as many different opinions on how things should be achieved.

First and foremost, we need to realize that everyone has different tastes. What might sound amazing to one person may sound like rubbish to someone else. As our years of production accumulate, we will begin to build ourselves a preferred workflow and style. Further to this, as time passes, our ears will become much more finely tuned to what we are hearing, allowing for more analytical listening. This means that when we are listening to individual sounds within a mix, or the entire mix-down itself, we can understand exactly what it is we want to achieve, and how we plan to achieve it.

For example you may be asked for:

- A punchier kick drum
- A warmer bass-line
- More prominent and upfront vocals
- More of a 3 dimensional mix

All of these ideas are more easily achievable once you have a grasp of the tools at your disposal and the fundamentals of mixing. Not only will these different tastes and styles change from producer to producer, but other factors will need to be considered, such as the genre of music, the track itself, the emotions you wish to convey in the track, and even things such as whether the record will be pressed onto vinyl or not.

Factors such as these can seem irrelevant and are often overlooked, but these small details matter. A common example of this is how tracks intended for vinyl need to be more dynamic than their digital counterparts. Vinyl tracks need careful attention with regards to the extreme high & low frequencies to ensure proper groove geometry, as well as to prevent unwanted distortion that could make the needle skip out the grooves.

Within the mixing process there are a few key functions that will need to be controlled and manipulated to give us a well-polished mix.

These key functions are:

- Volume (Balance)
- Pan & Stereo Imaging
- Equalization
- Dynamics
- Space & Depth
- Effects
- Automation

Once we have got all of these different elements under control, the end result will be a 'bounce' of the sum of all our tracks. Typically this will give us our stereo-interleaved audio file, which will be ready for mastering.

Many producers will mix their tracks as they are composing. This technique is fine, because it can help with a producer's creative flow; however, we strongly suggest revisiting the entire track from a mixing perspective. Different mixing methods are covered extensively in a later eBook from this mixing in Ableton Live series.

MIXING EXPLAINED SUMMARY & KEY POINTS

- Mixing is the process of combining multiple sound sources together

- Different genres and music mediums require different mix-downs

- Mixing is very subjective; each producer has his or her own way of working

- A 'bounce' is when we print a project or piece of audio to a file, such as a .wav or .aiff

- The mix process can be split down into key functions such as volume, panning, stereo imaging, EQ, dynamics, space & depth, effects and automation

BASICS OF SOUND AND WAVES

In this section, we will take a brief look at the basics of sound waves, frequencies and harmonics. It's important that we understand the basics of sound, so that we can fully grasp the reasons why we do certain things in the latter stages of the mix process.

BASIC WAVE THEORY

Firstly we have basic wave theory, the most common waveforms are:

- Sine Wave
- Triangle Wave
- Square Wave
- Pulse Wave
- Sawtooth Wave
- White Noise

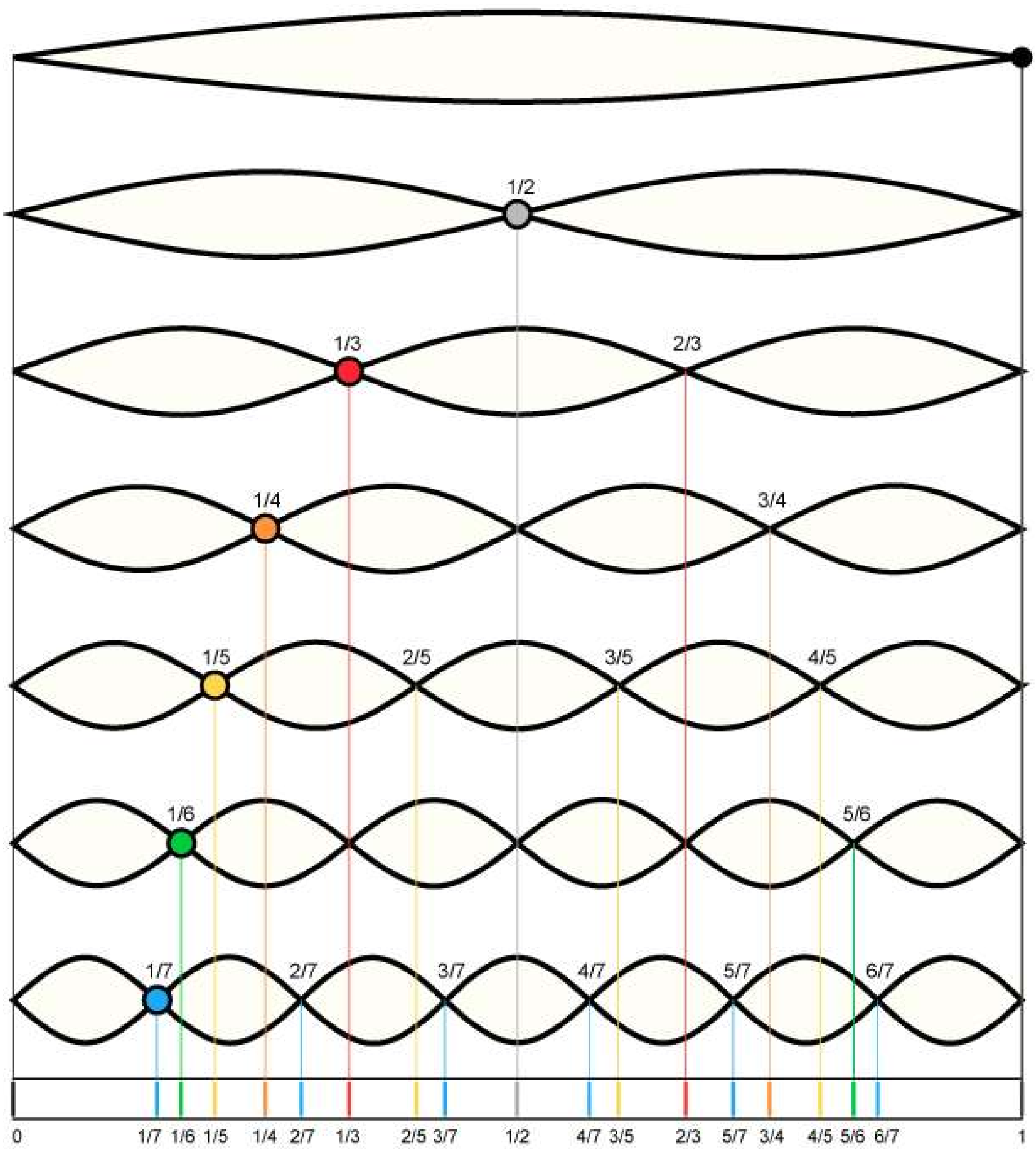
It is unusual in the mixing stage, that waveforms will be this simple. Usually the audio we work with will be complex waveforms, which can be thought of as being made up of lots of simple waveforms stacked together, as well as a lot of effects and processing.

As we are aware, a sound is simply a vibration or oscillation. If a vibration cycles periodically, then they will have a musical tone.

Complex tones can be broken down into a fundamental frequency and overtones, which together make the composite sound. (Fourier analysis) This is the technique used in spectral analysers, which is why they are also commonly referred to as FFT's. (Fast Fourier Transform.)

The reason we get these harmonics (overtones) in complex sounds is because of how an object can cause micro vibrations within itself. For example a guitar string can be split in half, third, quarter and so on, which will subsequently produce the 2nd, 3rd and 4th harmonics.

The amplitude and phase of these harmonics will also play a considerable role in the overall timbre of the composite sound.



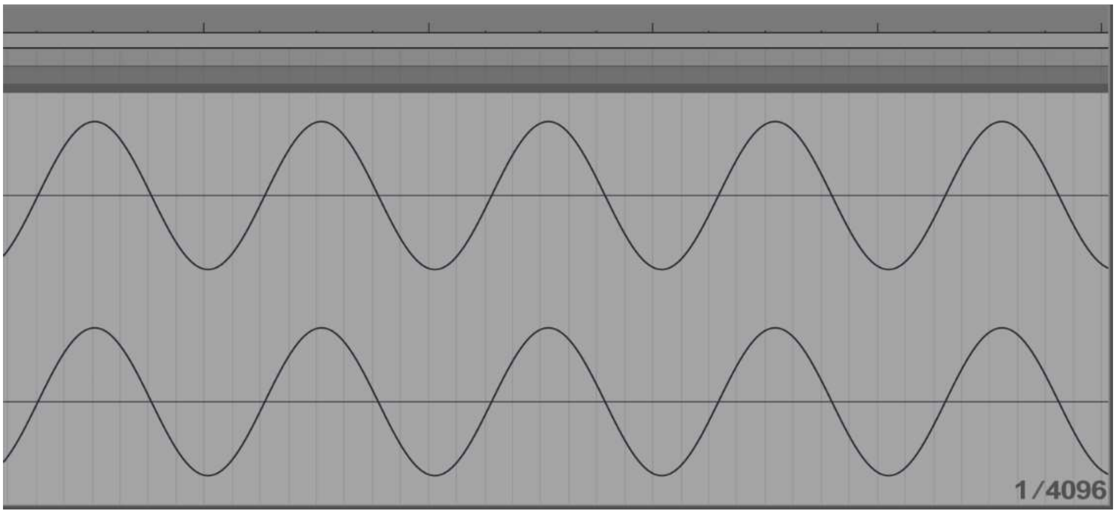
Splitting a vibrating object into whole number multiples shows a breakdown of the harmonics. This theory will also become useful later to understand the concept of standing waves.

When we listen to sounds, we will be hearing them in the human hearing range. This is between 20Hz–20KHz. Generally, as we get older, most humans can't hear much audio past around 15-18kHz.

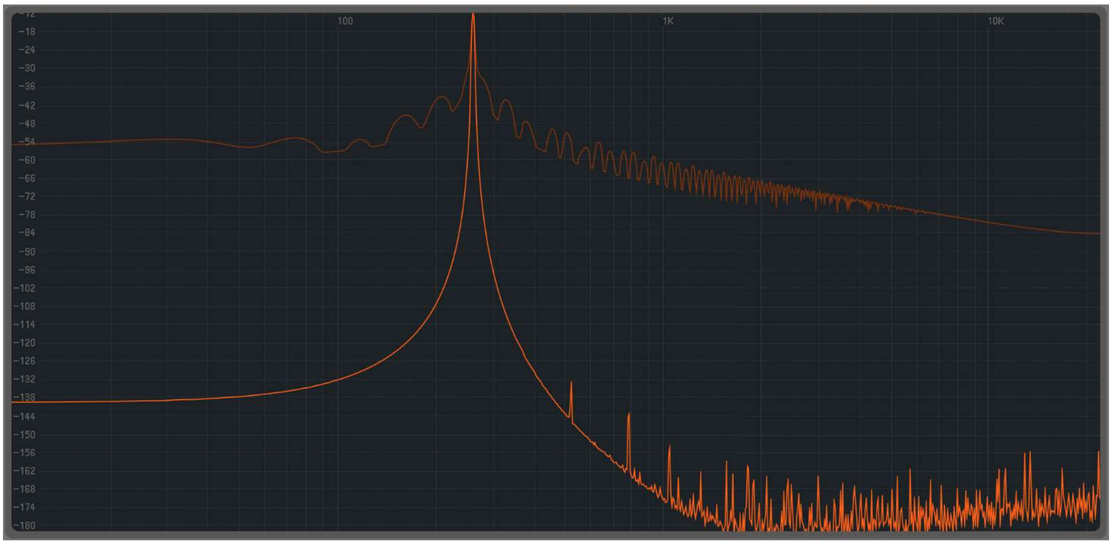
We will take a look at each of these waveforms and their harmonic content using a frequency analyser. Spectral analysers can be set to work logarithmically or in a linear manor. We recommend keeping your spectral analyser in logarithmic mode, which is it's default setting. This is because there are is a huge range of frequencies that we need to be able to see. Using a logarithmic graph allows us to see all of the frequencies in a more condensed and useful format.

Sine Waves

Are the purest waveforms and only consist of the fundamental, with no harmonics. This means filters have little effect on their overall sound. They are often used for adding sub-bass to sounds.



A sine waveform



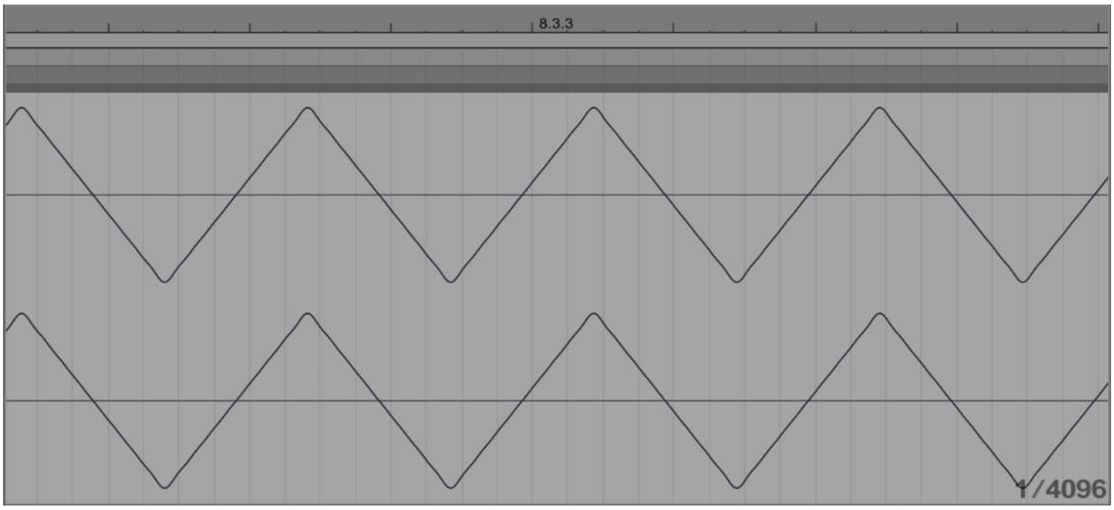
A sine waves spectral analysis

Triangle Waves

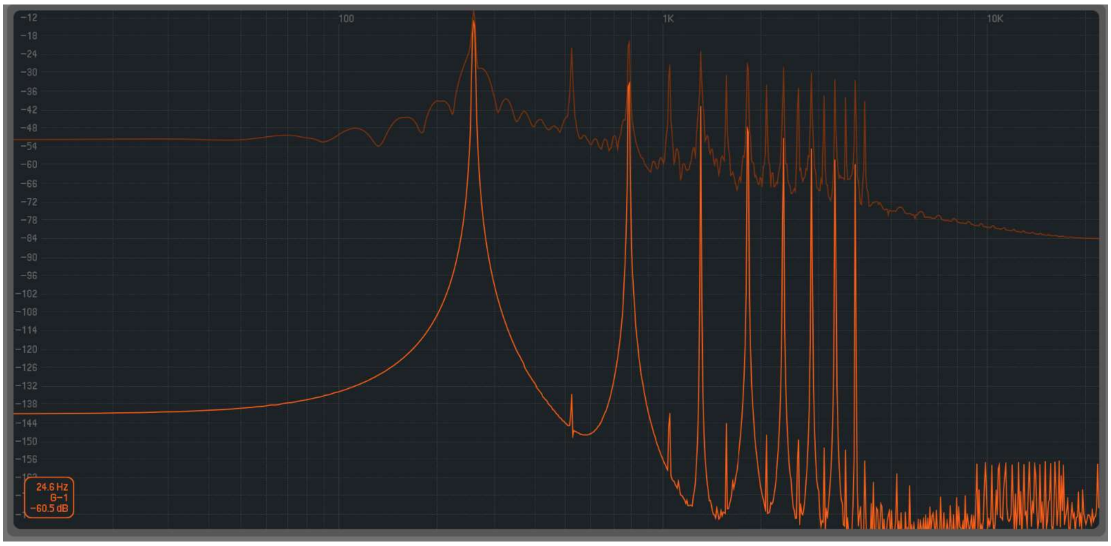
Contains only odd harmonics. The harmonics amplitude can be calculated using the formula:

Amplitude = $1/N^2$
(N = number of the harmonic, 1 being the fundamental)

Triangle waves have a sound that is somewhere in between a sine wave and a square wave.



A triangle wave



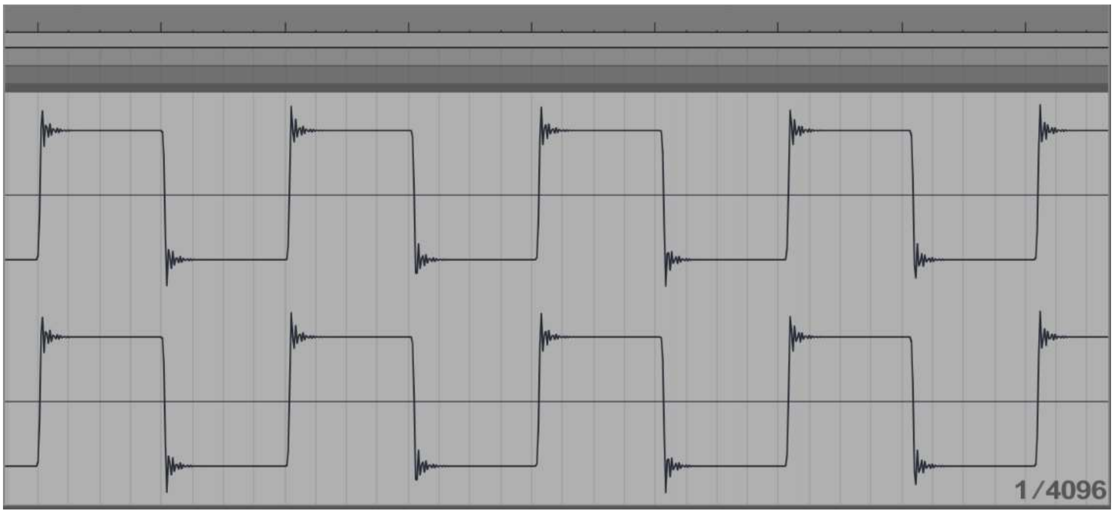
Spectral analysis of a triangle wave

Square Waves

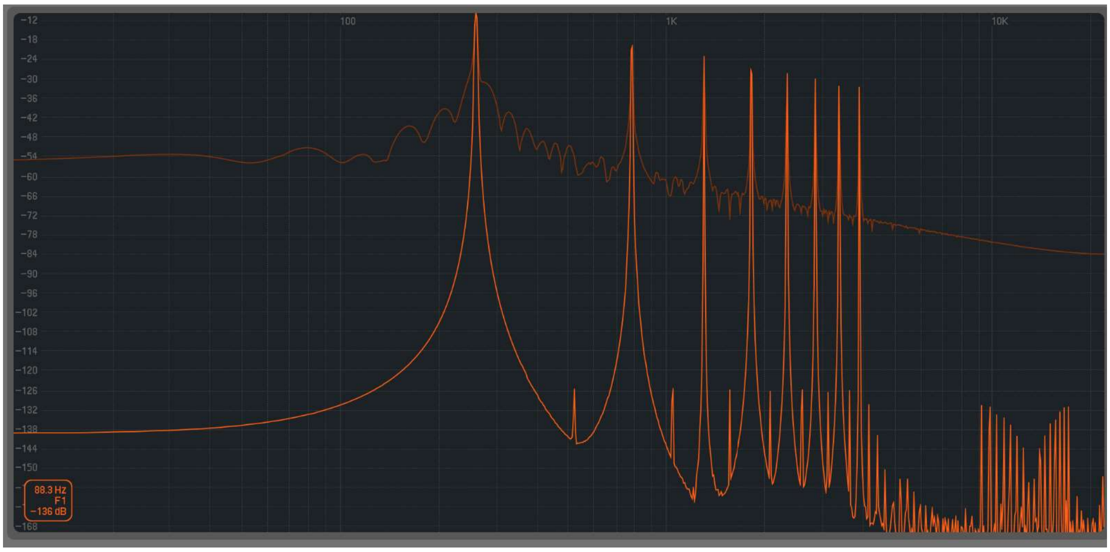
Contain only the odd harmonics; this gives them a hollow-type sound, which is good for recreating any form of wood instrument. Square waves can sometimes be referred to as pulse waves. Altering the pulse width can change a pulse wave's tonality.

The amplitude of a square waves harmonics can be calculated using the formula:

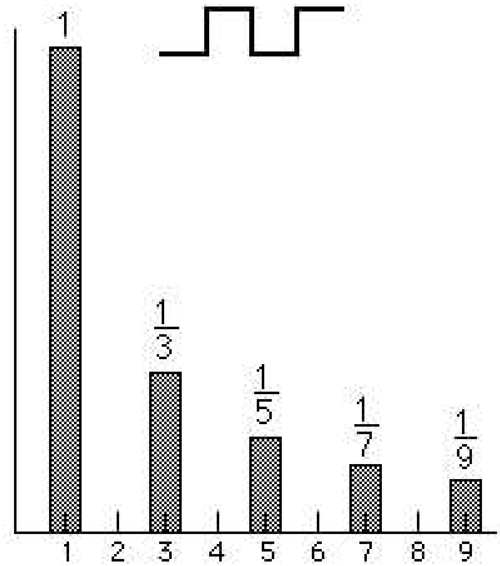
AMPLITUDE = 1/N



A square wave



Spectral analysis of a square wave (Sq8)



A diagram to show a square wave. It's characteristic 'hollow' or 'wooden' sound is achieved from its lack of even harmonics. This diagram also shows the change in amplitude per harmonic, in relation to the fundamental frequencies amplitude.

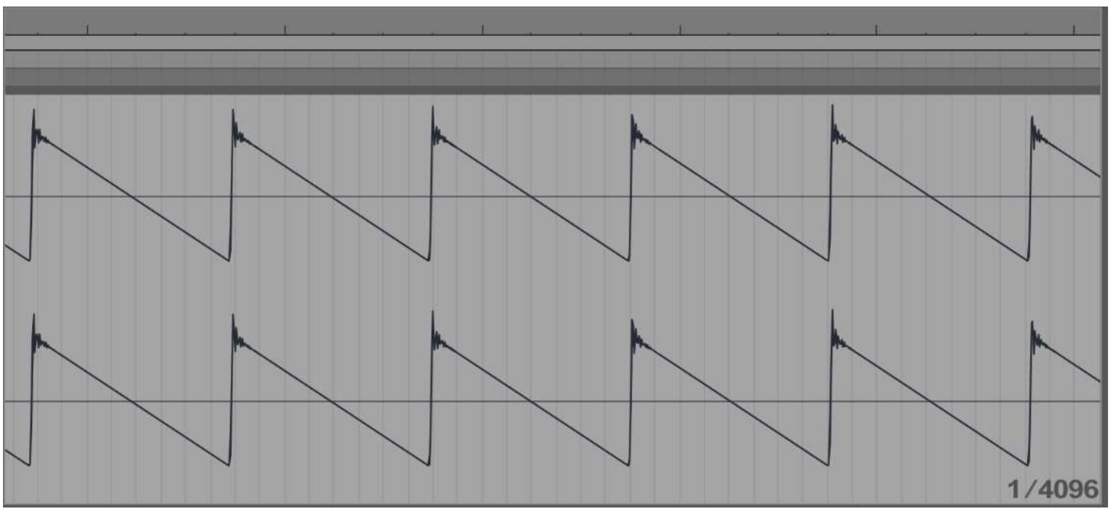
Sawtooth Waves

Contain all the frequencies in the harmonic series. Each harmonic gets quieter as it goes up the scale, this causes Saw waves to produce a very rich sound, which makes them perfect for shaping with the use of filters.

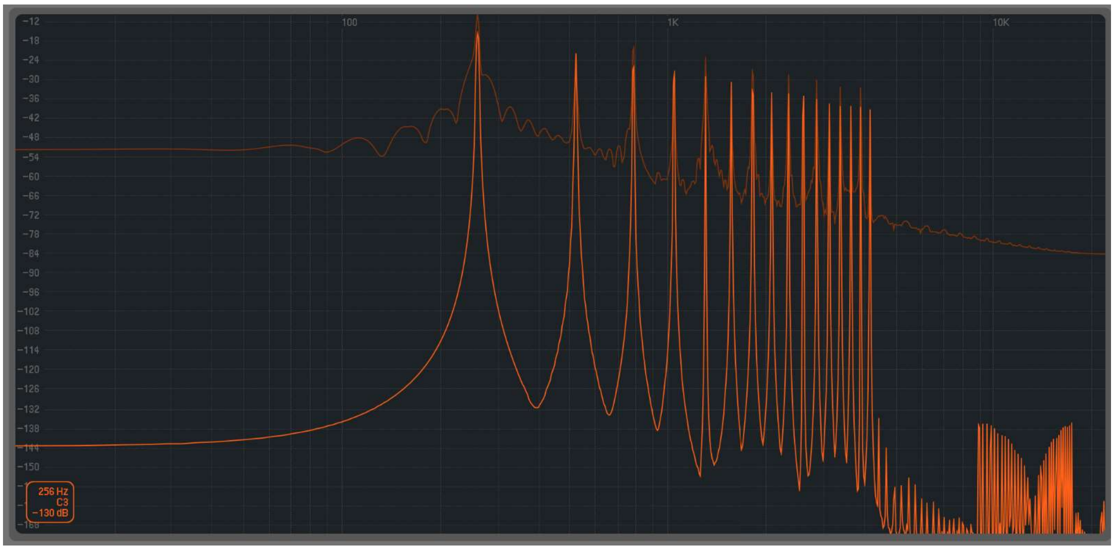
The amplitude of a sawtooth waves harmonics can be calculated using the formula:

AMPLITUDE = 1/N

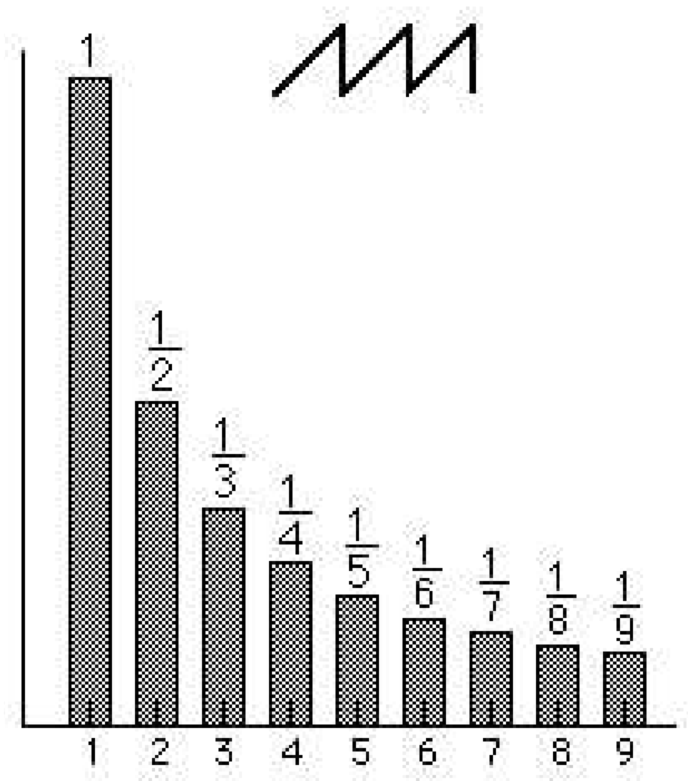
For example, Harmonic 5 will be a 5th of the amplitude in comparison to the fundamental.



A sawtooth wave



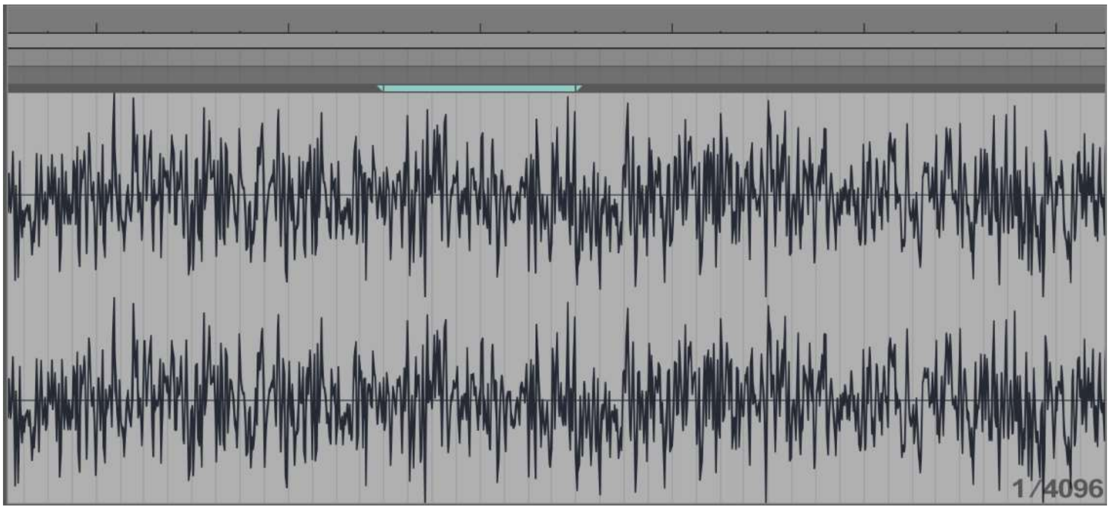
Spectral analysis of a sawtooth wave



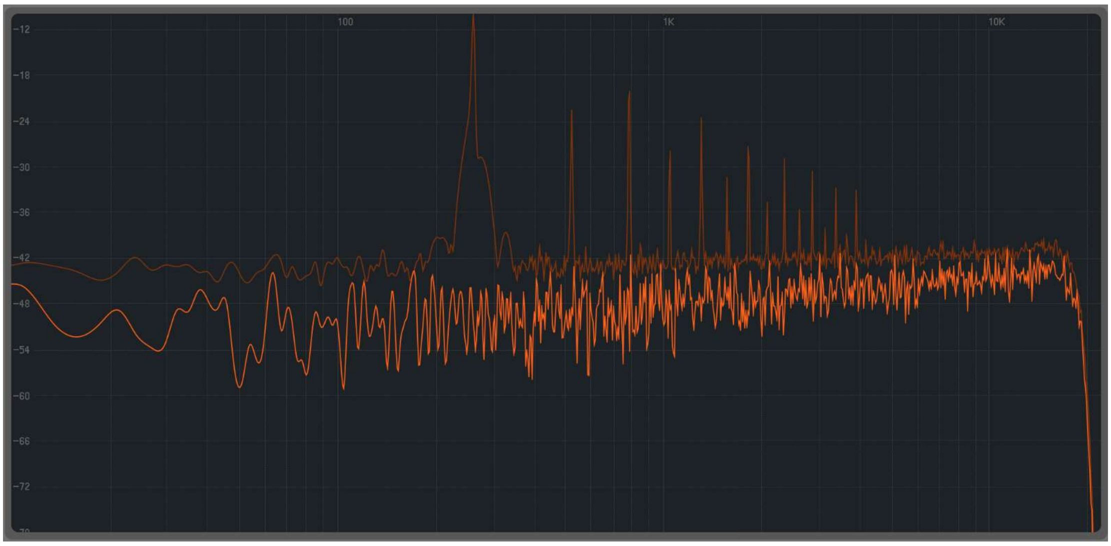
A diagram to show a sawtooth waves harmonics and their relative amplitudes in comparison with the fundamental frequency.

White Noise

Contains all the frequencies within the frequency spectrum at full amplitude. White noise has many different uses, from synthesizing drums, to creating textures and swells.



White Noise



Spectral analysis of white noise

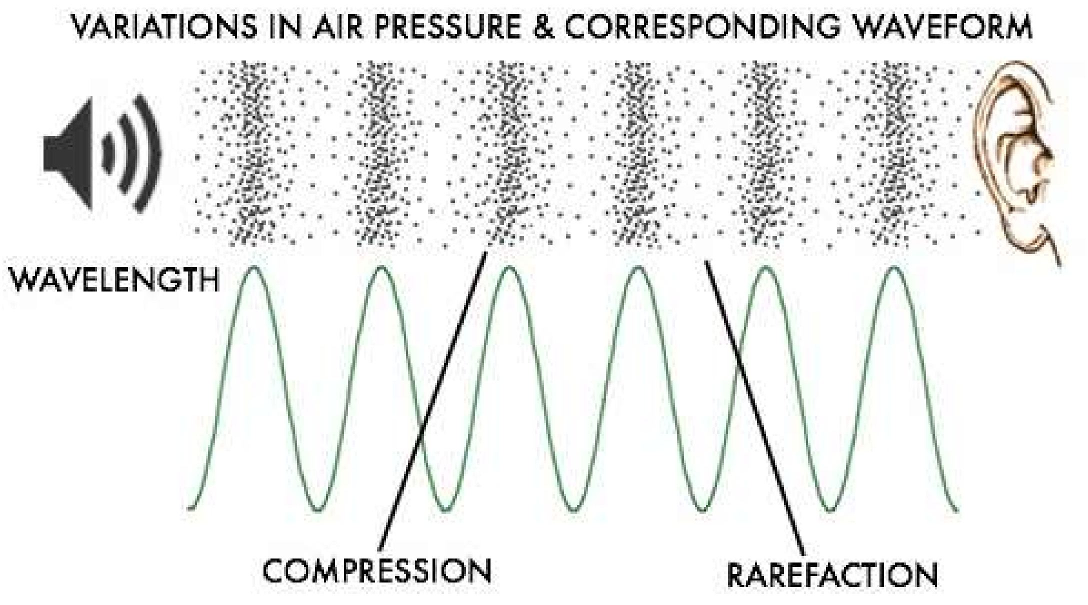
PROPAGATION OF SOUND

To be able to understand how sound is affected by different environments, we must first take a look at what it is, and how it propagates through a medium such as air.

WHAT IS SOUND

Sound is simply waves of compressed and refracted air particles. These can be thought of as waves, or vibrations that travel through the air. Sound can be created by a voltage, which moves a magnet within a speaker. This speaker, which is transducing electrical energy into kinetic energy, is making the speaker driver move back and forth. This in turn excites the air particles in front of the speaker cone.

These air particles propagate through a medium (in this case air) to our ears, where our eardrum's (another form of transducer) detect the vibrations and turn them into neural-electrical signals for our brains to process.



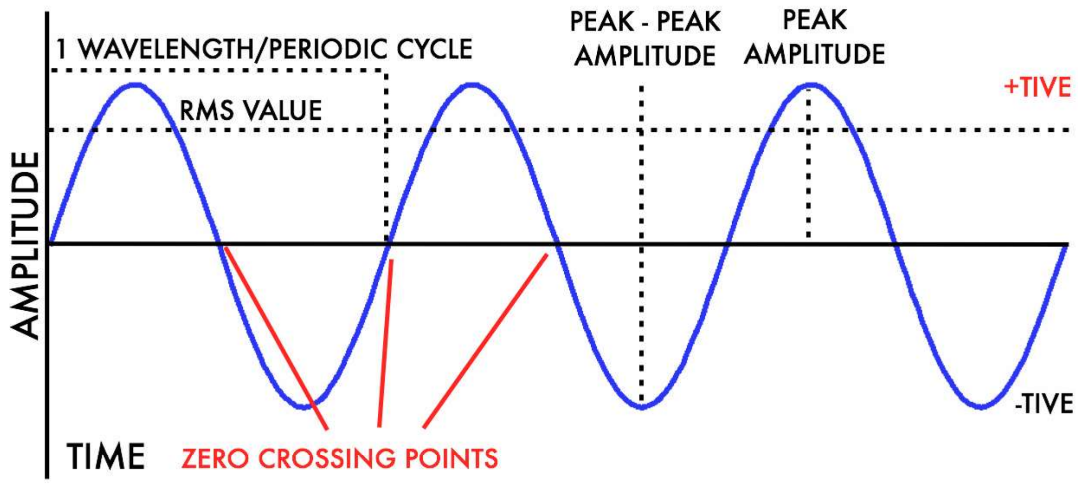
THE SPEED OF SOUND

The speed at which sound travels through a medium is known as a velocity. Sound travels at different velocities dependent on the material it is passing through. Mediums such as wood, water and air will all have different velocities.

For the purposes of mixing we only need to worry about air. The speed of sound through air at room temperature is 343 meters per second. (1125ft per second) This can be approximated to 1 millisecond per foot, or 3ms per meter.

WAVES

We can describe a wave in terms of amplitude, wavelength, velocity, frequency, harmonic content (timbre) phase, and envelope.



AMPLITUDE

The amplitude is the distance from the central line to the furthest distance from the central line. This is commonly measured in Decibels, (dB) or Volts. (V) As we get further away from the central line, the amplitude, or loudness will increase.

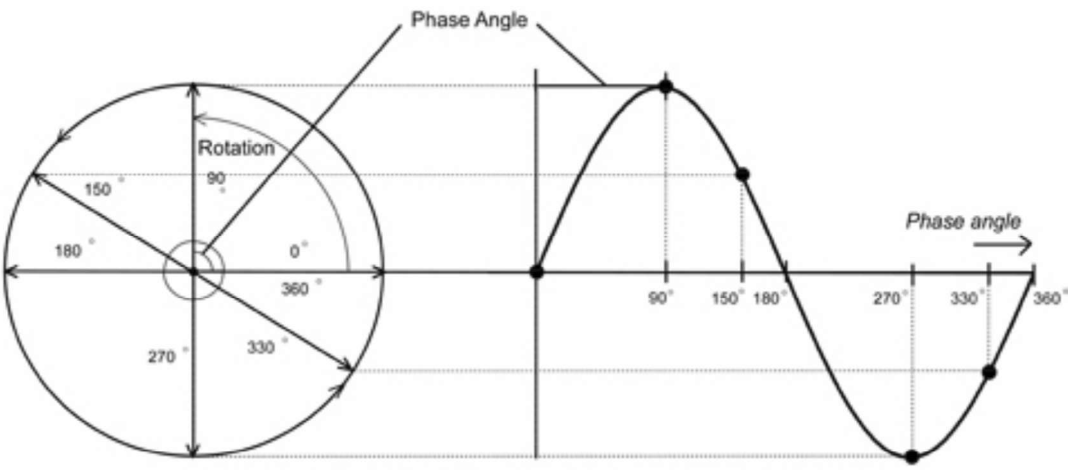
Measuring the highest peak to the central line is known as the peak amplitude, whereas measuring the entire waveform from its +tive, to its –tive value is know as peak to peak.

We can also measure 'RMS' which stands for 'root mean squared'. This will approximate the average amplitude of the wave.
This is important because it gives us an indicator of the overall loudness of our track, whereas the peak volume only really gives us an indicator of whether our fast transient peaks are clipping or not.

CYCLES AND WAVELENGTHS

If we look at the diagram below, we can see one complete cycle of a sine wave. A cycle needs to pass through 360 degrees to become a full cycle. This can actually be shown as a circle; however, we split the circle in half and flip it over. This allows us to represent the wave over time.

We can also see that the waves 'degree' defines its phase. A wave can start, or be 'offset' at any point along the 360 degrees.



The wavelength of a waveform is the 'length' of the 'wave' or the distance from the beginning of the cycle, to the end of the cycle.

We can use this formula to calculate wavelengths:

Λ (Lambda) (Wavelength) = Velocity of sound in air (343) / Frequency (Hz)

This formula proves that low frequencies have large waveforms, between 5-20 meters, whereas higher frequencies can be as small as a matter of centimetres in length.

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.