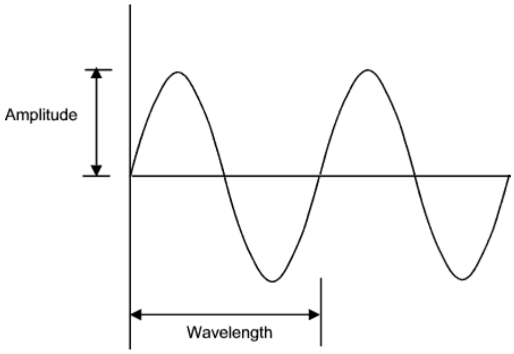
Creative Applications of Sound Synthesis

At a purely physical level sound is simply a *mechanical disturbance of a medium*, which may be air, solids, liquid, gas or composite. This disturbance to the medium causes molecules to move to and fro each other in a springlike manner. As one molecule hits the next the disturbance moves through the medium causing sounds to travel. The most basic type of sound wave is known as a sinusoidal wave form and if one were to plot it on the Cartesian plane it might look like this:



The portion of the sine wave which sits on top of the x-axis is in a state of compression whilst the portion of the sine wave under the x-axis is in a state of rarefaction. Sinusoidal waves are the most important type of waveform as they are the building blocks of all sounds and are extremely important to people who wish to synthesis or create their own sounds.

Frequency

Frequency can be defined as the number of repetitions/cycles that take place in a second. The more repetitions that take place a second the higher the frequency will be. Given any frequency we can work out how long each repetition takes to complete by using this simple formula:

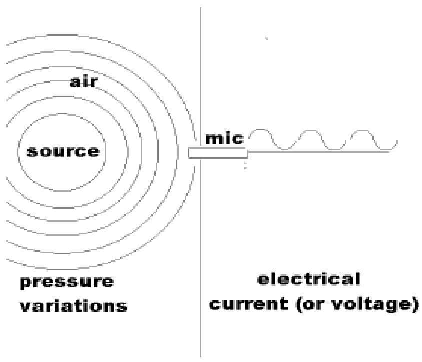
$$f = \frac{1}{T}$$

(*f* is frequency and *T* is time)

Transduction

Acoustical energy in the form of pressure waves can be converted into an electrical signal by using a transducer. A transducer produces electrical pressure, i.e., voltage, that changes constantly in sympathy with the vibrations of the sound wave in the air.

The continuous variation of pressure is therefore 'transduced' into continuous variation of voltage or current. The greater the variation of pressure the greater the voltage that is sent down the cable of the recording device to the computer.



Ideally, the transduction process should be as transparent and clean as possible: i.e., whatever goes in comes out unspoilt. In real-world situations however, this is never the case. Noise and distortion are always inserted into the signal. Every time sound passes through a transducer or is transmitted electrically a loss of signal quality will occur. Noise in this case can be classified as any unwanted signal captured during the transduction process. This normally manifests itself as an unwanted ‘hiss’.

Most guitarist's will be familiar with the term 'distortion' but few are aware of the science behind it. There are three main types of distortion:

1) **Amplitude distortion** is the most common type of distortion and is the one used or modelled in most guitar effects units. Amplitude distortion is caused by non-linearity's in the response of a device to various values of input amplitudes.

2) **Frequency distortion** is caused by a device’s inability to respond the same way to all frequencies

3) **Phase distortion** is the most complex of all distortions and is caused by differences in time response to different frequencies. When a system cannot handle all frequencies with a uniform time response delays are added to different frequencies. This results in a unwanted filtering of the signal being input to a device.

Problems with Analogue Systems

In general, analogue systems can be very unreliable when it comes to noise and distortion. Every time something is copied or transmitted an amount of noise and distortion is introduced in the process. If this is done many times, the cumulative effect can deteriorate the signal quite a lot. It's because of this the music industry has turned to digital technology which so far offers the best solutions to this problem.

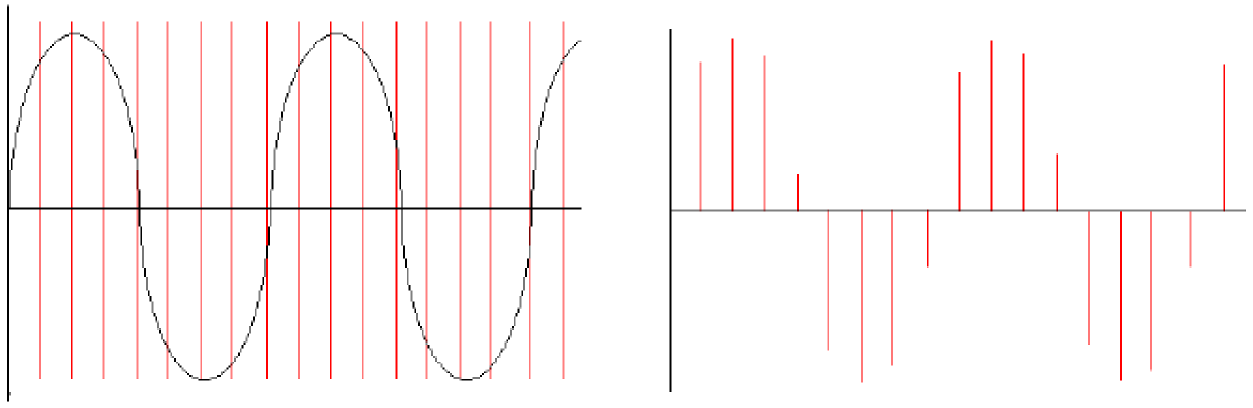
In digital systems, sound is stored as numbers, so a signal can be effectively “cloned”. Mathematical routines can be applied to prevent errors in transmission, which could introduce noise in the signal.

Sampling

The analog voltage that corresponds to an acoustic signal changes continuously so that at each instant in time it will have a different value. It is not possible for a computer soundcard to receive the value of the voltage for every instant because of the physical limitations of both the computer and the data converters (remember also that there are an infinite number of instances between every two instances!).

What the soundcard can do however is measure the power of the analogue voltage at intervals of equal duration. This is how all digital recording works and is known as sampling. The result of this sampling process is a discrete or digital signal which is no more than a sequence of numbers corresponding to the voltage at each successive sample time.

Below is a diagram showing a sinusoidal waveform. The vertical lines that run through the diagram on the left represents the points in time when a snapshot is taken of the signal. On the right we can see the information which have been recorded by the computer.



At every sampling period (the vertical lines in the above diagram), a value is obtained from the input analogue signal. This value is called a ***sample***. The sampling period, i.e. how often the soundcard is going to capture a sample is defined by a sampling frequency or *sampling rate* (SR). The usual sampling rate for CD-quality audio is 44100Hz or 44.1KHz.

The Sampling Theorem

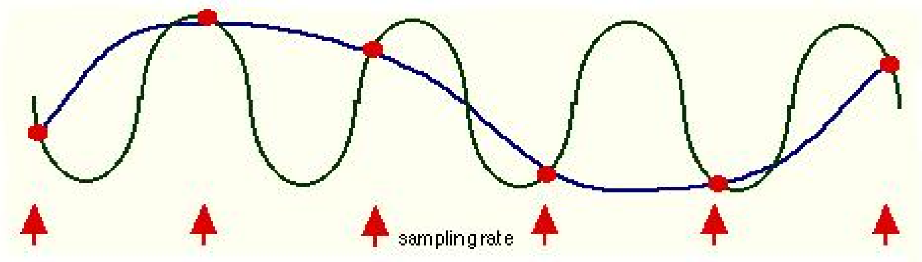
To sample an audio signal correctly it's important to pay attention to the sampling theorem:

> *To represent digitally a signal containing frequencies up to X Hz, it is necessary to use a sampling rate of at least 2X samples per second*

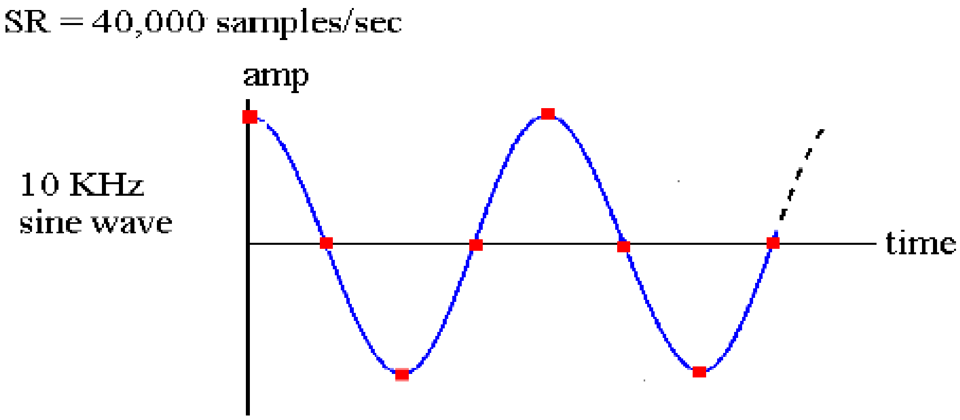
According to this theorem, a soundcard or any other digital recording device will not be able to represent any frequency above 1/2 the sampling rate. Half the sampling rate is also referred to as the ***Nyquist*** frequency, after the Swedish physicist Harry Nyquist who formalised the theory in the 1920's. What it all means is that any signal with frequencies above the Nyquist frequency will be misrepresented. Whats more is that it will result in a frequency lower than the ne being sampled. When this happens it results in what's known as aliasing.

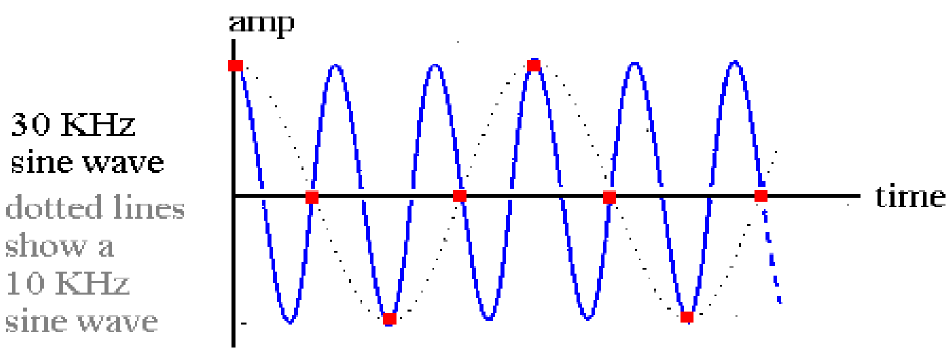
Aliasing

Here is an example of how aliasing occurs:



The sinusoidal wave form is being sampled at each circle. The line that joins the circles together is the captured waveform. As you can see the captured wave form and the original waveform are different frequencies. Here is another example:





Bits, Bytes and Words. Understanding binary.

All digital computers represent data as a collection of ***bits****(short for binary digit)*. A bit is the smallest possible unit of information. One bit can only be one of two states - off or on, 0 or 1. The meaning of the bit, which can represent almost anything, is unimportant at this point. The thing to remember is that *all* computer data - a text file on disk, a program in memory, a packet on a network - is ultimately a collection of bits.

Bits in strings of eight are called **bytes**, and one byte usually represents a single character of data in the computer. It's a little used term, but you might be interested in knowing that a nibble is half a byte (usually 4 bits).

The Binary System

All digital computers work in a environment that has only two variables, 0 and 1. All numbers in our decimal system therefore must be translated into 0's and 1's of the binary system. If you think of binary numbers in terms of switches. With 1 switch you can represent up to 2 different numbers.

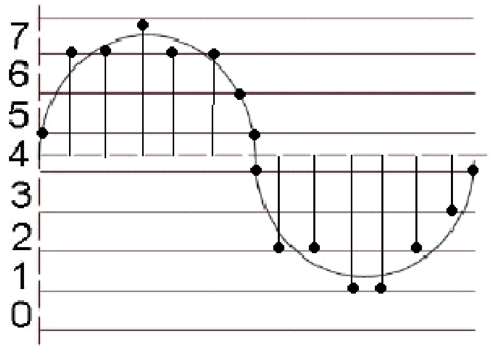
0 (OFF) = Decimal 0
1 (ON) = Decimal 1

Thus, a single bit represents 2 numbers, two bits give 4 numbers, three bits show 8 numbers, four bits represent 16 numbers, and so forth up to a byte, or eight bits, which represents 256 numbers. Therefore each added bit doubles the amount of possible numbers that can be represented. Put simply, the more bits you have at your disposal the more information you can store.

Bit-depth resolution

Another parameter which can affect the fidelity of a digital signal is the accuracy with which each sample is known, in other words knowing how strong is each voltage is. Every sample obtained is set to a specific amplitude (the measure of strength for each voltage) level. The number of levels depends on the precision of the measurement, in bits, i.e., how many binary digits are used to store the samples. The number of bits that a system can use is normally referred to as the *bit-depth resolution*.

If the bit-depth resolution is 3 then there are 8 possible levels of amplitude that we can use for each sample. The norm for CD's is 16 bit which allows for 65536 different possible amplitude levels. Using bit rates lower than 16 is not such a great idea as it results in noise being added to the signal.



In the above diagram there are 8 different amplitude levels(3-bit). At each sampling period the soundcard plots an amplitude. As we are only using a 3-bit system the resolution is not good enough to mark the correct amplitude. We can see in the digram that some vertical lines go past the point of the real signal. This is because our bit-depth resolution is not good enough to sample the correct amplitude level at each sampling period.

ADC / DAC

The entire process, as described above, of taking an analogue signal and converting it to a digital one is referred to as **analogue to digital conversion** or ADC. Of course **digital to analogue conversion,** DAC, is also possible. This is how we get to hear our music through our PC’s headphones or speakers. For example, if one plays a sound from Media Player or iTunes the software will send a series of number to the computer soundcard. In fact it will most likely send 44100 numbers a second. If the audio that's playing is 16 bit then these numbers will range from -32768 to +32768.

When the sound card receives these numbers from the audio stream it will output corresponding voltages to a loudspeaker. When the voltages reach the loudspeaker they cause the loudspeakers magnet to move inwards and outwards thus causing a disturbance in the air around it causing sound.