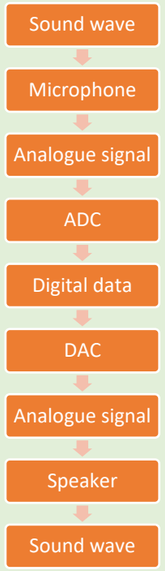
Representing Sound

Analogue and Digital

- Analogue data is the sound wave.
- The microphone (transducer) is a sensor that converts the sound wave into a continuous electrical analogue signal.
- To store the signal on the computer it needs to be converted to digital data using an ADC (Analogue to Digital Converter)
- The digital data is stored as a sequence of discrete values that are used to encode the signal.
- To listen to the digital sound that is stored in the computer a DAC (Digital to Analogue Converter) is used to create an electrical analogue signal.
- A vibrating speaker then converts the electrical signal into a sound wave.



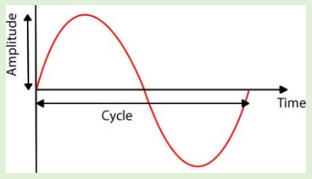
Analogue to digital conversion

For sound to be stored digitally on a computer it needs to be converted from its continuous analogue form into a discrete binary values.

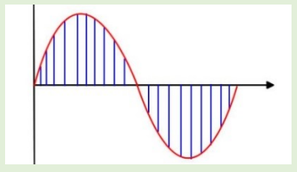
1. The analogue signal is sampled at regular intervals.
2. The samples are approximated to the nearest integer (quantised).
3. Each integer is encoded as a binary number with a fixed number of bits.

```
0001 0010 0011 0100 0100
```

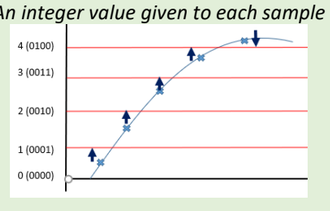
Original analogue signal



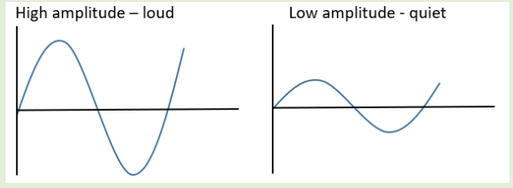
Signal sampled at regular intervals



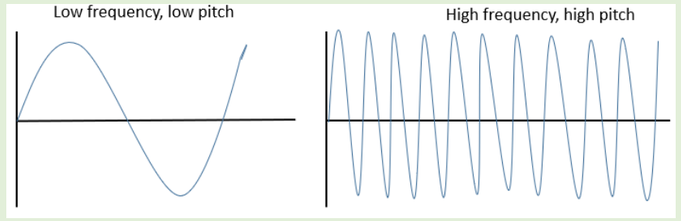
An integer value given to each sample



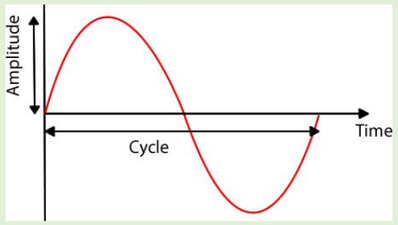
Amplitude - Represents the size of the wave and impacts the volume. The bigger the amplitude the bigger the volume.



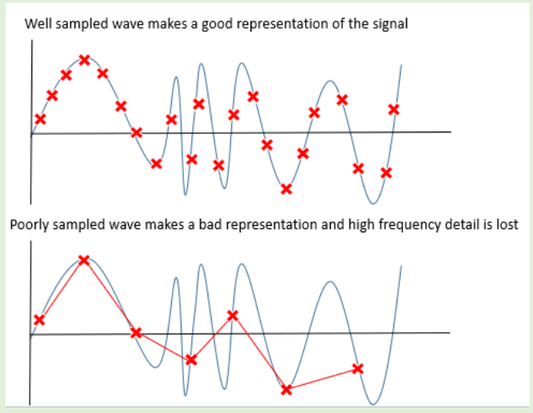
Frequency - Number of cycles per second is measured in Hertz (Hz). Frequency impacts the pitch, so the higher the frequency the higher the pitch.



Sample - Measure of amplitude at a given point in time

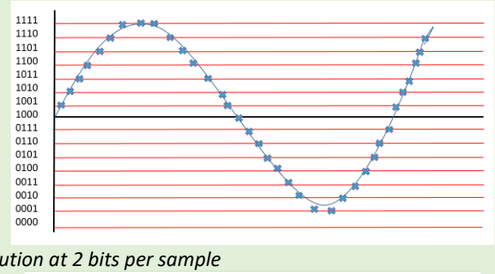


Sample rate is number of samples per second

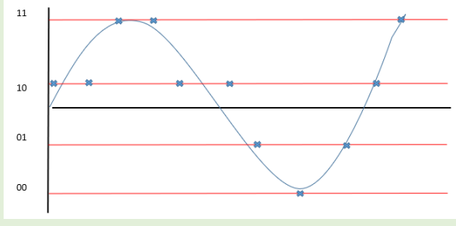


Sampling resolution is the number bits per sample

Sample resolution at 4 bits per sample



Sample resolution at 2 bits per sample



The **size of sound files** can be calculated using the following formula:

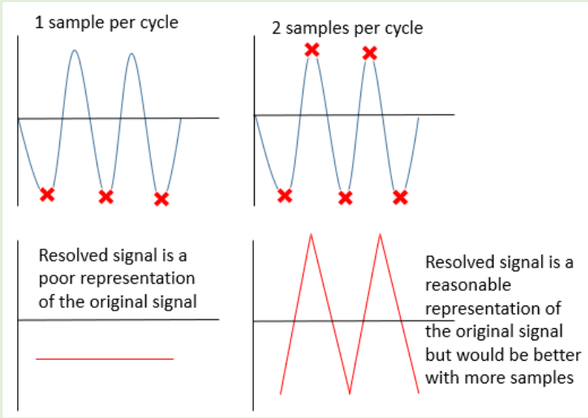
```
File size = length x sample rate x sampling resolution
```

where length is in seconds

Worked example: What is the size of a 30 second sound clip in kilobytes that is sampled at 8000 samples per second with a sample resolution of 16 bits?

30 x 8000 x 16 = 3840000 bits / 8x1000 = **480 Kbytes**

Nyquist theorem states that we need to sample at least at twice the rate of the highest frequency. That is we need to sample twice per cycle in order for us to resolve that frequency.



Musical Instrument Digital Interface (MIDI)

MIDI is synthesised music.

A MIDI file is a set of instructions that contains messages (or events) on how to produce a sound for a digital device.

MIDI is not stored music.

Information contained in a MIDI event

- Note-on
- Note-off (Both note on and note off give the event duration)
- Key pressure (Aftertouch – how hard a key is pressed
- Pitch
- Velocity
- Vibrato
- Volume
- Pitch Bend

This is not an exhaustive list, other information is also contained in a MIDI event.

A collection of events taken together allow us to produce a piece of music

Advantages of MIDI

- MIDI files are generally small.
- Can play using different instruments
- Can easily edit MIDI files so there is no need for re-recording
- Can change the key of a piece of music