Digitising Sound - Reading

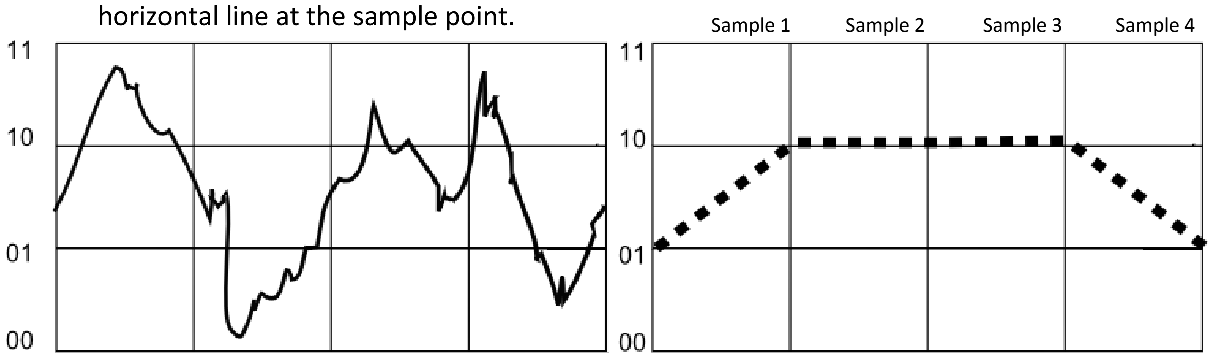
Sound is input into a computer with a microphone and output from it with a speaker. Sound, unlike say a key stroke, is continuous movement of air and so an **analogue signal**. Computers need to store things digitally as 1s and 0s.

A sound card will contain an **analogue-to-digital converter (ADC)** to convert the signal from a microphone to digital for the computer. It will have a **digital-to-analogue converter (DAC)** to convert the computer's digital signal to an analogue one for the speaker.

This is how it works:

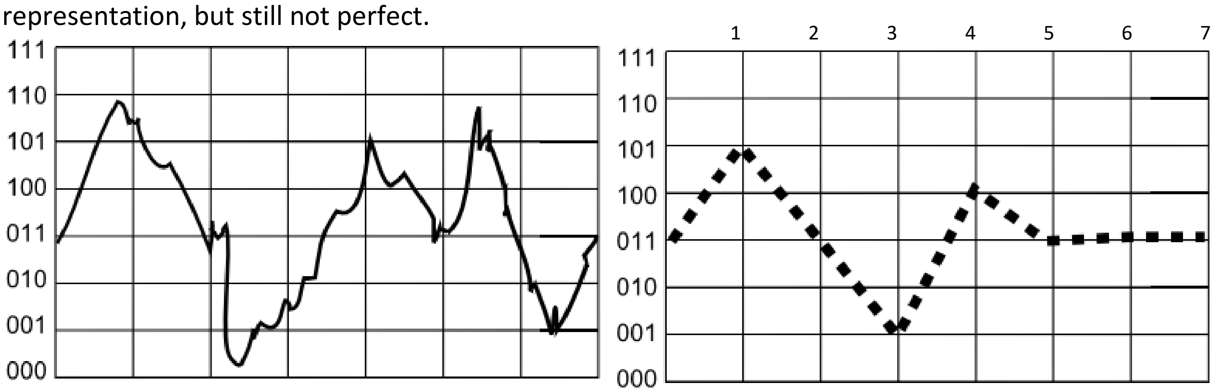
1. Start with a sound wave (the line on the right shows 1 second of audio)

2. Now sample the data (here we are taking 4 samples per second – so 4Hz) – this is the vertical lines. This is known as the **sampling rate**. We also choose a **sample resolution** (in this case 2 bits) which is how accurate each sample will be. Notice that because of the low sample resolution we need to round up or down if it doesn't cross a horizontal line at the sample point.



3. The binary to store the right hand graph will be: 10 10 10 01.

Using a higher sample rate this time (7Hz) and higher bit rate (3 bits) we get a more accurate representation, but still not perfect.



The binary to store this would be: 101 011 001 100 011 011 011

File size (bits) = **sampling rate * sample resolution * seconds**

A CD uses 16 bits per sample * 44.1 KHz sample rate * 2 channels (left and right)
= 16*44100*2 = 1 411 200 bit/s data rate = (1411200 / 8)/1000 = 176.4 kilobytes per second