Analogue Data Representation

Sounds

Sound is **analogue data**. This needs to be converted to numbers to be able to be stored digitally

Sounds cause air particles to vibrate. Our ears interpret these vibrations as sound.

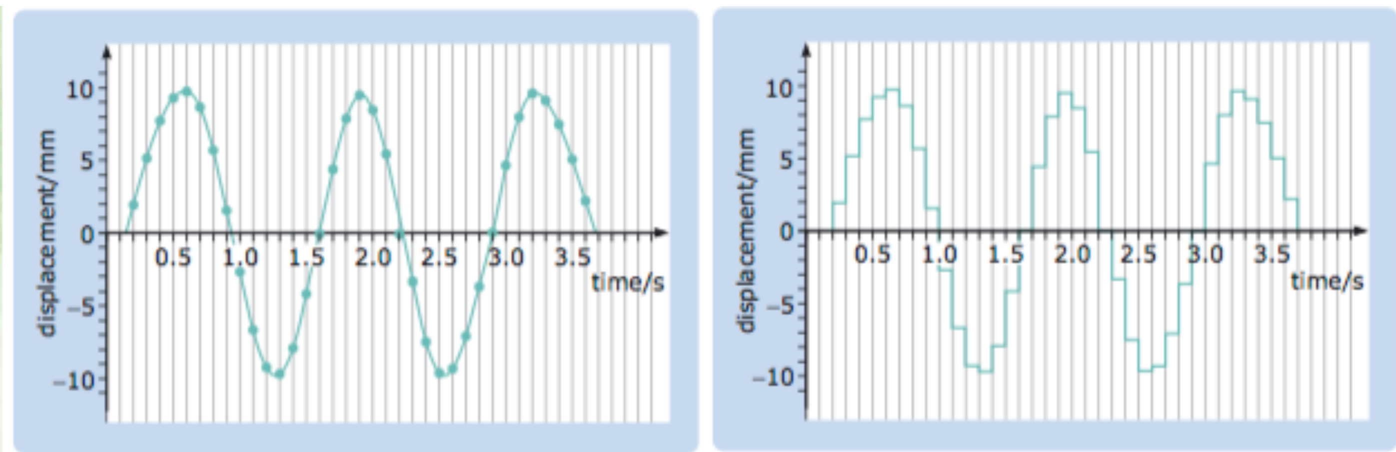
The vibrations move through the air as **analogue signals** (sound waves). Playing an instrument or singing creates these analogue signals. It has been possible to record these signals for decades as **analogue** recordings on tape recorders or records.

Sampling Audio:

CDs were the first major form of digital sound format. When CDs were introduced in 1982 they often had more capacity (700 MB) than the hard drives of most personal computers

To store sound information on a computer we need to create a **digital** version of the sound. To do this we need to find a way of representing the sound as a set of numbers.

This requires **sampling**. Sampling measures the **amplitude** (the volume) of an analogue signal (sound wave) at regular intervals. The measurement of the signal at each point can be turned into a number, although the process turns the smooth analogue signal (wave) into a stepped digital profile.



Sampling every 0.1 second

Sampling Rate

The number of samples per second is called the **sampling rate**. The more samples each second, the closer to the original sound the digital recording will be. At the same time, the more samples per second the bigger the file size will be because more data is being stored per second of sound.

1 Hertz is one complete **cycle** of a sound wave per second
kHz means **kiloHertz** – 1,000 Hertz. So 1kHz is 1,000 samples per second.

Sound **sample rates** are measured in **Hertz (Hz)**. 1 Hz is 1 sample per second. The standard sampling rate for CD quality audio is 44.1 kHz. This means that 44,100 samples are taken each second. For a VOIP app such as Skype, the sampling rate might be only 8kHz.

Activity 1:

a) Describe how **analogue** sound is converted into a digital format using **sampling**
b) Define the term **amplitude**
c) Explain how **sampling rate** effects the quality of a digital audio file

Sampling Resolution

The **sampling resolution** of an audio file is how many bits are used to store each sample. The higher the sample resolution, the more data is stored about each sample. This will effect the quality of the digital audio file – the higher the sampling resolution the higher the sound quality should be.

In science the term **resolution** means the smallest interval on a measuring instrument.

CD quality sound uses 16 bit sample resolution. This means that 16 bits are used to store each sample (a range of numbers from 0 to 65,536).

DVD quality sound uses 24 bit sample resolution (0 to 16.7 million).

Sampling Rate, Resolution and File Size

The more data being stored the greater the file size.

file size in bits = sampling rate x sample resolution x seconds of audio

Remember, sample rate is in Hz or kHz – the number of samples per second

So, an audio file 10 seconds long sampled at a rate of 20Hz using 8 bit resolution would be:

file size = 20 x 8 x10 = 1600 bits

If you need to give the file size in Bytes simply divide by 8. You could then convert to kB, MB, GB…

Activity 2:

a) Explain the meaning of the term **sampling resolution**
b) Explain how **sampling rate** and **sample resolution** effect file size
c) Calculate the file size in bits in each of the following cases
 (i) 3 seconds of audio sampled at 10Hz at 8 bit resolution
 (ii) 10 seconds of audio sampled at 4Hz at 16 bit resolution
 (iii) 5 seconds of audio sampled at 40kHz at 24 bit resolution

Analogue v Digital

Natural sounds we hear (like voice, engine noise, the sea, birds etc…) are **analogue**. This data changes smoothly along a curve. Technically you can continue to split analogue data up forever – you can always make the gap between the next two points smaller.

This means that, technically, the quality of analogue audio is "better" than that of **digital** audio. Digital data has been sampled and eventually you get to a stage when you can no longer split a sample.

The key is to use a good enough sample rate and resolution to "fool" the ear of most people, so that the audio sounds just as good most of the time – or "good enough" anyway.

Is there a difference between analogue and digital sound quality?

I have a record player and a CD player attached to the same speaker and amplifier. If you play the same track on a vinyl record and a CD player there is a noticeable difference in quality – the vinyl sounding much "brighter" – to my ear anyway