3.3.7 Representing sound

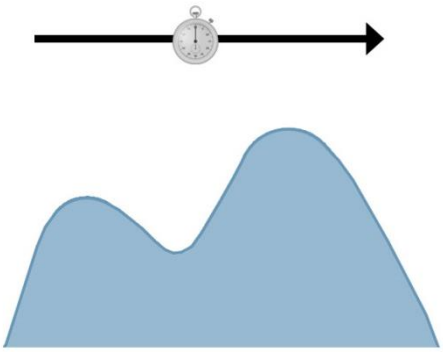
Teacher notes

cse1.net/recaps/13-av.html has some very good background information on this topic.

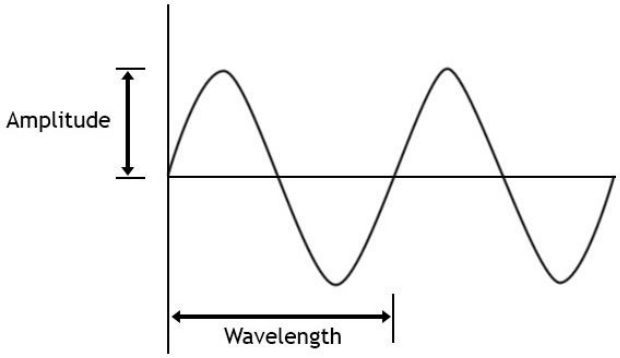
What is analogue sound?

All naturally occurring sounds are analogue in nature – sound is generated as a continuous waveform over time.

Basic analogue wave



Analogue wave showing positive and negative amplitude



By definition, analogue forms must be continuous, ie have no break. We can say therefore that it has infinite resolution. (You can zoom in on the wave indefinitely and it will always be a continuous line with no gaps in it.)

Sound can have a positive and negative amplitude and it is more usual to see it represented using the continuous sine wave format. The fact that positive and negative values are used creates their own challenges for the way the sound is represented when it is captured or 'digitised'.

© 2016 AQA. Created by Teachit for AQA

Capturing analogue sound – the digitisation of sound

All sound files are simply text files written in a specific format eg MP3, WAV which the computer knows how to interpret and play back to the user.

In order for sound to be captured it must be sampled digitally. This involves 'slicing up' the sound vertically (by time duration) and horizontally (by amplitude) and capturing the exact value at the moment as a digital snapshot. The quality of the sound produced is wholly dependent on how many samples are taken and the resolution of the sampling.

There are two key measurable properties of the sound:

- Sampling rate is the number of samples taken in 1 second.
- Amplitude is measured by taking horizontal slices of the sound to give the sampling resolution and is the number of bits (slices) per sample.

Digital sound is represented as discrete (single addressable) points captured by 'mapping out' an analogue sound wave – a binary value representing each point is stored in a recognised file format and reproduced by the digital media player used eg Windows Media Player.

To capture a sound at CD quality, we need to take 44100 samples per second.

Determining the size of sound files

File size (bits) = Sampling rate (Hz) × Resolution (bits) × Length of sample (seconds)

Note that the specification refers to file size in bits which means that the resolution also needs to be in bits. If the resolution is in bytes and the question requires a file size in bits then then multiply by 8.

Example:

We record a song by an artist in stereo, the song lasts four minutes – sound is CD quality.

Sampling rate =	**44,100**	**Resolution** =	16 bits
Length of sample =	4 × 60 = **240**	**File size = 44,100 × 240 × 2 = 169,344,000** (bits)*	

***** This is far larger than a conventional MP3 would be.

Students will complete a quiz with simple file size calculations.

© 2016 AQA. Created by Teachit for AQA