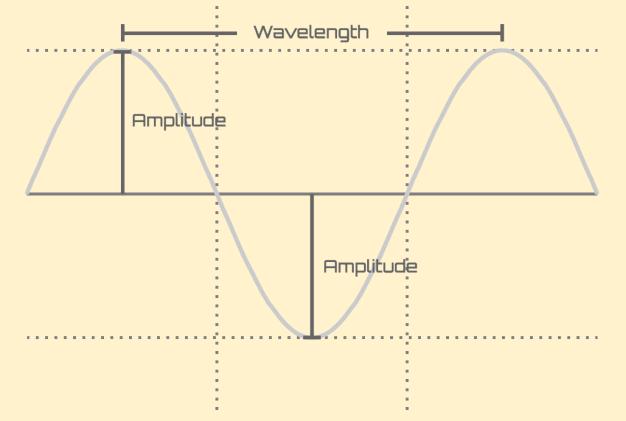
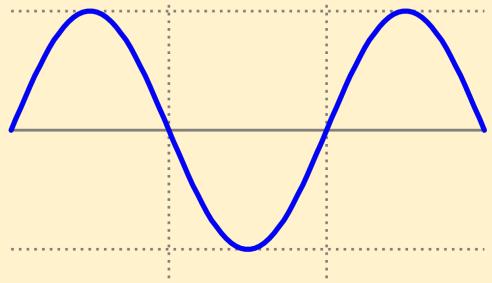
Sound travels through the air as a wave. These change the air pressure that affects your ear

drums.



Sound waves are **analogue signals**. This means that they are continuous, without any breaks.

To store them in a computer we need to break them up so that we can represent them as numbers.



The process of breaking up sound waves is called sampling.

Sampling lets us turn analogue information into digital data.

 The pixels in a digital image are samples individual points that make up the image. We need to do the same with sound waves.

To change analogue information into **digital** form we need to **sample** it.

This means we take readings at **regular** intervals along the sound wave to measure the **amplitude** (volume) and convert these into **numbers**.

The numbers can then be stored using binary.

A **sample** is a measure of **amplitude** at a point in time.



