Sound | Part B

Digital audio encodes analogue signal into digital form by taking multiple snapshots of the sound wave, like frames of a movie film. These snapshots capture split second data on how the membrane of a microphone behaved in response to a sound wave that was filling its sail.

The frequency of these snapshots is known as “sampling rate”.

For example, a common sampling rate used in audio CDs is 44.4 kilohertz which is 44,400 snapshots per second. Each of these snapshot is ultimately just a number that represents by how much the speaker needs to move and how quickly to recreate the original sound. How many zeros and ones are needed to record such a snapshot? Well CDs use the 16 bit format, where zero is a silence and one is the loudest sound possible or in binary, zero is 0000 0000 0000 0000, while 1111 1111 1111 1111 will be the loudest sound possible.

This 44.4 kilohertz 16 bit format is not the only one though. DVDs uses 48 kilohertz and some modern professional recordings go beyond 16 bits rate, to 32 bits. Professional recording equipment is currently switching to 96 kilohertz and even 192 kilohertz.

Now how many samples are enough? Humans hear up to 20 kilohertz and scientists say that the sampling rate needs to be twice as much as that to hear full 20 kilohertz this is known as the Nyquist-Shannon theorem. 8 kilohertz would be a telephone quality it’s often used for speech only applications and we know that in a telephone conversation anything above 4 kilohertz is not really needed to understand the person on the other end.