FACTFILE:
GCSE
DIGITAL TECHNOLOGY
Unit 1 – DIGITAL DATA

Fact File 3: Representing Sound

Learning Outcomes

Students should be able to:

- Describe factors that affect sound quality, including:
  - sample rate;
  - bit depth;
  - bit rate;

  ... when recording sound.
- Explain the need for analogue-to-digital conversion in sound recording.

Digital Representation of Sound

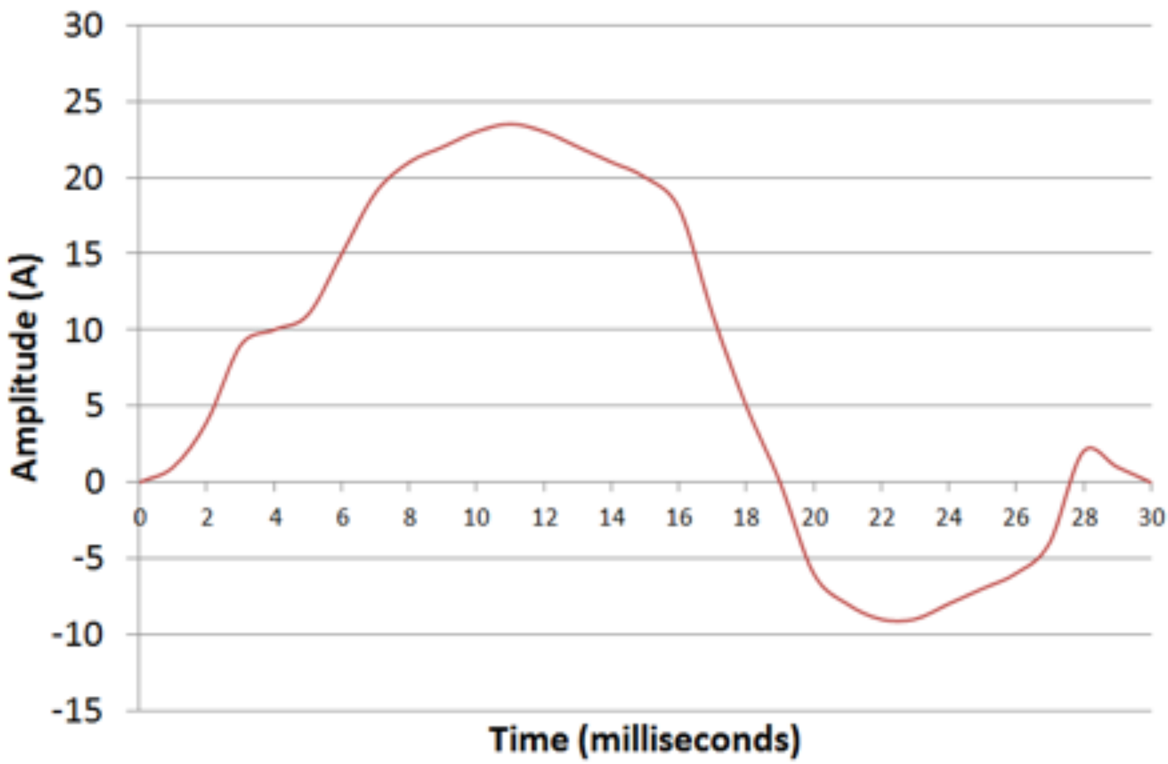
The Need for Analogue-To-Digital Conversion

Sound is heard when vibrations in the air reach the human ear, and a computer SPEAKER (found either on a loudspeaker or within a set of headphones) will vibrate a conical structure in order to push air vibrations towards an audience.

A MICROPHONE detects sound waves and converts them to voltage variations that can subsequently be captured digitally by a computer through the process of ANALOGUE-TO-DIGITAL (ADC) CONVERSION. The original sound wave from the air is ANALOGUE in nature and so is the continuously varying voltage produced when a microphone picks it up. This means that between any two points in time there are infinitely many minor variations in the wave of sound energy. With space limitations in digital storage and with human brains being incapable of detecting all of the minor variations, human need can be adequately met by taking an appropriate quantity of regular measurements of the sound wave, via the microphone, and storing these measurements DIGITALLY so that the original sound wave can be produced later by a speaker. This is called SOUND SAMPLING.

Sampling An Analogue Sound

Consider a very short sound clip of 30 milliseconds in length. Here is a visual representation of its analogue form:



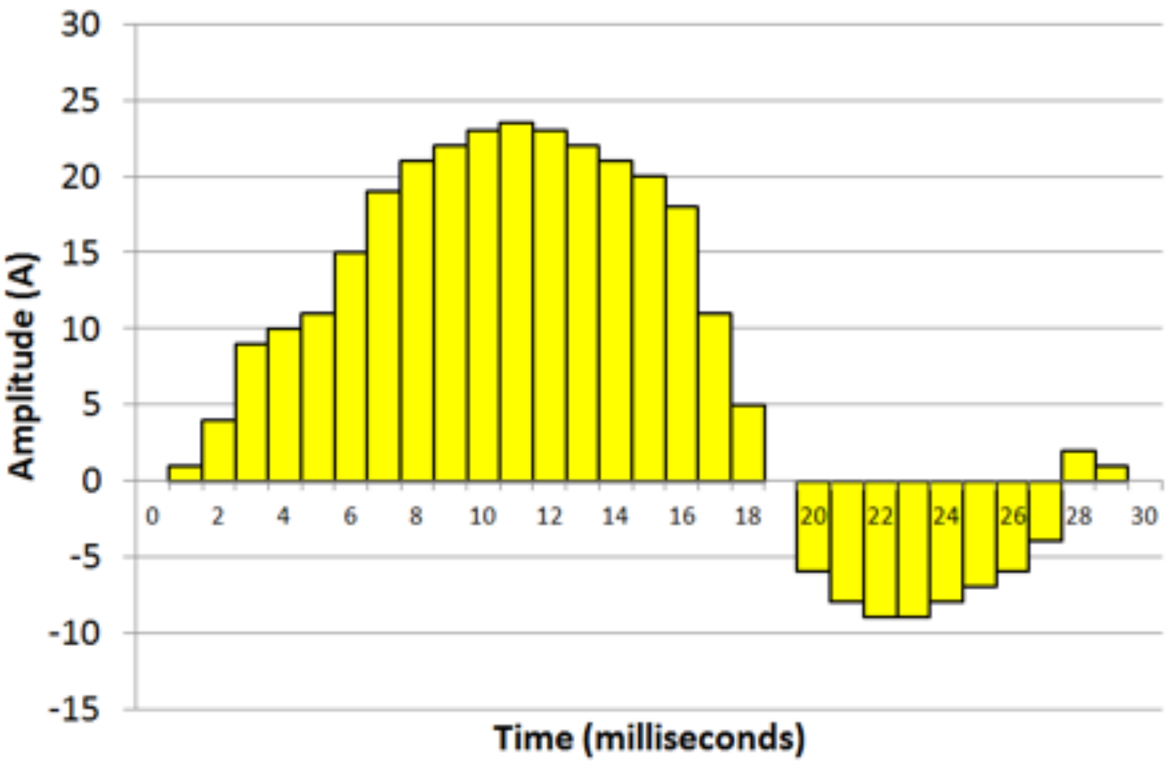
Over the space of 30 milliseconds, the manner in which the amplitude of the sound wave rises and falls varies enormously. Between 0 and 3 milliseconds it rises exponentially, between 17 and 20 milliseconds it descends rapidly and unevenly, and between 23 and 26 seconds it rises steadily. To save this sound clip digitally, we must measure the amplitude of the wave, ideally at regular intervals to minimise the complexity of reconstructing the wave when it is played back later.

It is decided that on this occasion a sample will be taken every millisecond. Here are the readings taken from the analogue wave:

Time (milliseconds)	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Amplitude (A)	0	1	4	9	10	11	15	19	21	22	23	23.5	23	22	21	20

Time (milliseconds)	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30
Amplitude (A)	18	11	5	0	–6	–8	–9	–9	–8	–7	–6	–4	2	1	0

This data can thus be stored as a sequence of binary numbers in the computer's external storage. This is a DIGITAL representation of the wave, and can be visualised as follows:



Reconstructing the original sound wave from its digital samples involves turning the saved numbers back into voltages that the speaker can handle. This will not bring back a perfect copy of the original analogue sound wave, but the resemblance will be close enough to be acceptable to the human ear.

Factors Affecting Sound Quality

In the sampling process described so far, there were variables at work that shall now be further studied in order to determine how they may affect the quality of the sound as perceived by the human listener.

We know already that SOUND SAMPLING is the process of regularly measuring the voltage of signals (provided by a microphone) and storing the sequence of measurements in digital storage media.

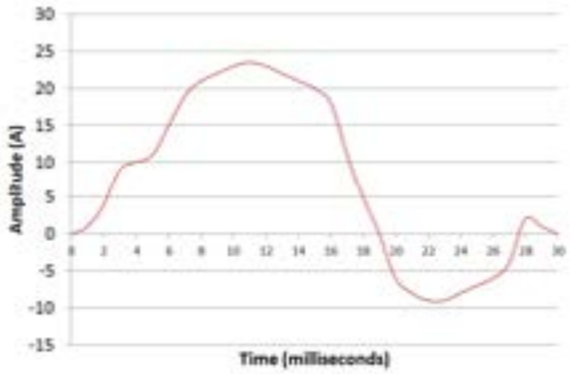
Sample Rate (Sampling Frequency)

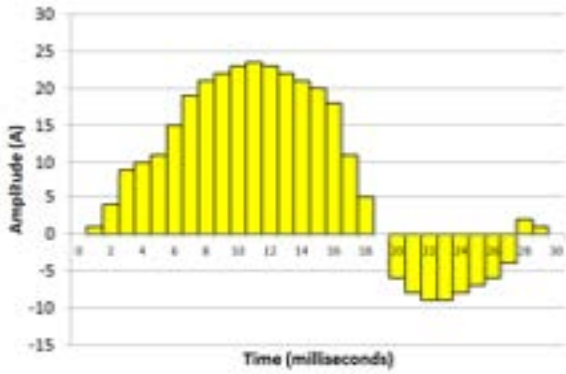
The SAMPLE RATE (or SAMPLING FREQUENCY) is the quantity of samples taken per second. It is measured in Hertz (Hz). We tend to use kilohertz as a suitable measure for sound:

1 kilohertz = 1 kHz = 1,000 Hz

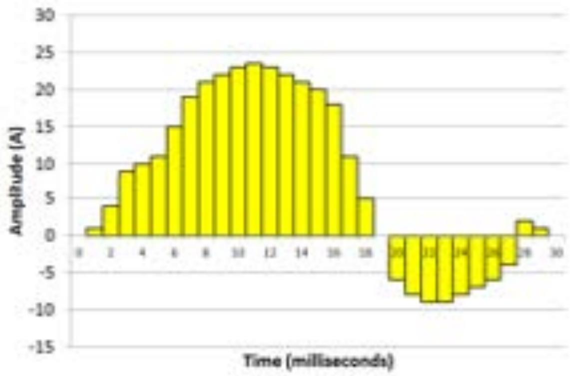
Consider the original example which had a SAMPLE RATE of 1 kHz:

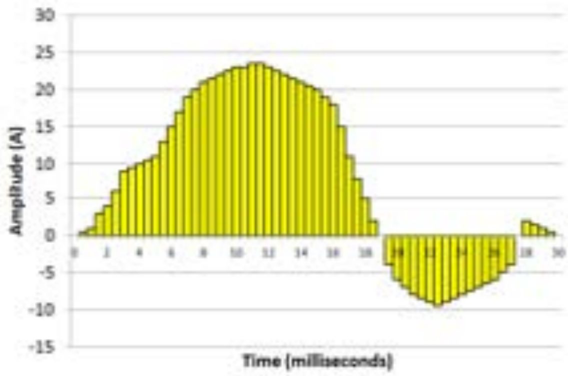
- 30 samples in 30 milliseconds = 1 sample per millisecond = 1000 samples per second





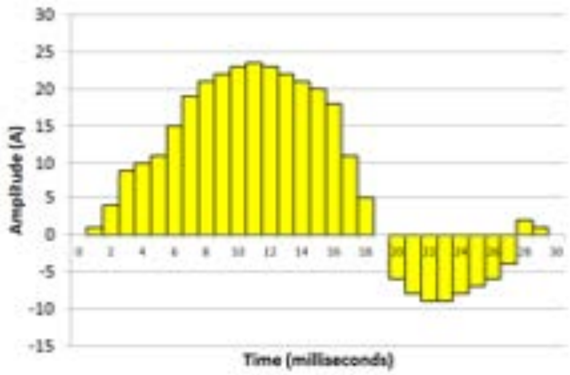
Compare this with a HIGHER SAMPLE RATE of 2000 samples per second, or 2kHz:

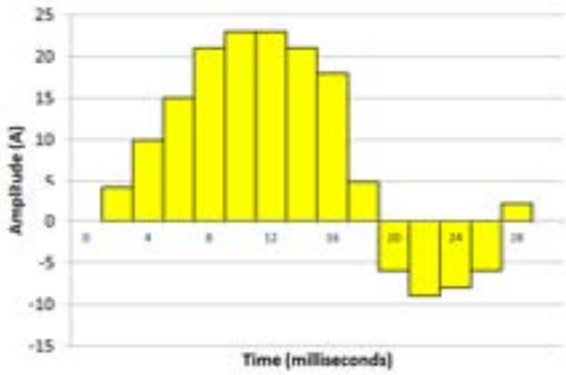




By making the sample rate **higher**, then the digital representation of the wave becomes a **stronger resemblance** to the original analogue sound wave. However, this will demand more storage space and more CPU time and processing power in order to keep up with the ongoing measurement of the incoming voltage signals during sampling.

Making the sample rate LOWER will reduce the likeness to the original sound wave and thus reduce the quality of the sound as perceived by the human listener. Here is what happens when the sample rate is halved to 0.5 kHz (500 Hz):





A common standard sample rate for sound is in the range 40kHz to 50kHz. The MP3 file type and CD-quality music both use 44.1kHz as their sample rate.

Bit Depth (Sample Resolution)

The BIT DEPTH is the number of bits that is allocated per sample taken.

As long as the sample rate is known, there is no need to store the time values, only the sample (A) values, thanks to the fact that the samples are taken at **regular intervals**.

Consider this 5-millisecond fragment of the original sound wave's data:

Time (milliseconds)	0	1	2	3	4
Amplitude (A)	0	1	4	9	10

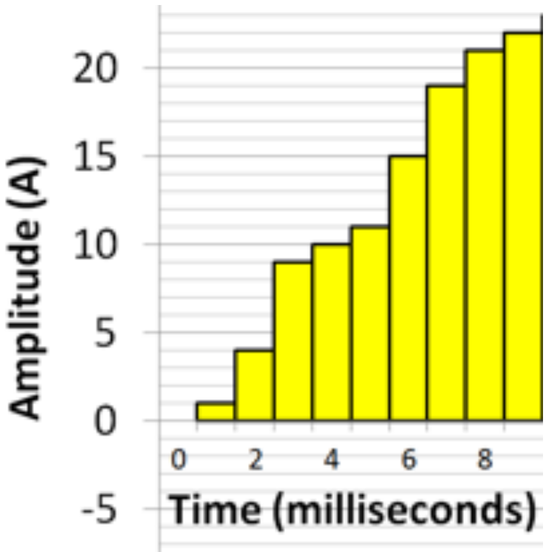
This can be stored in binary by representing each sample (i.e. each value of A) as 1 byte:

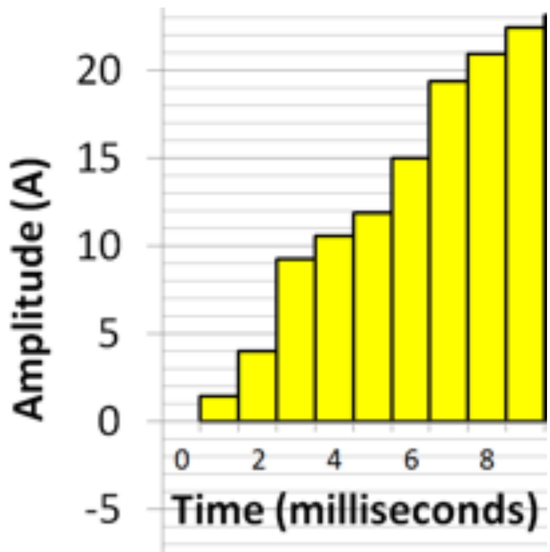
Time (milliseconds)	0	1	2	3	4
Amplitude (A)	00000000	0000001	00000100	00001001	00001010

With N bits, 2^N different binary values are possible. In this case, the numbers could be whole numbers (integers) in the range 0 to 255, or perhaps -128 to +127. However, restricting the values to being only whole numbers means that we cannot store the ½ measurements, and if we allow ½ measurements then the range of numbers shrinks to e.g. 0 to 127.5, or -64 to +63.5.

The amplitude A of the wave represents the volume of the sound at a given point in time.

By INCREASING THE BIT DEPTH, this can INCREASE THE QUALITY of the digital sound. This is because the samples taken can more precisely measure the exact value of A for every sample. The difference is not obvious at close range. Study the two diagrams below to see the subtle differences in how closely aligned to the gridlines the samples are:





For CD-quality sound the bit depth will be 2B (also known as 16-bit) per sample, but this can be reduced to 1B for generic online use where lower quality is more acceptable.

Bit Rate

The BIT RATE is the number of bits required to store 1 second of sound. The unit of measurement is bits per second (bps) or kilobits per second (kbps).

Bit rate is a straightforward calculation:

Bit depth × Sample rate = BIT RATE

In other words:

- Take the number of bits that are required per sample (**bit depth**);
- Take the number of samples per second (**sample rate**);
- Multiply these together to reveal how many bits are required to store 1 second of sound.

Since a high bit rate indicates that either the bit depth is high or the sample rate is high, a HIGH BIT DEPTH indicates a HIGH QUALITY SOUND as a lot of computer memory ends up being devoted to storing the original analogue wave as accurately as possible, either through very regular measurements of its amplitude (A) or through very accurate measurements of the precise value of A, or indeed both simultaneously.

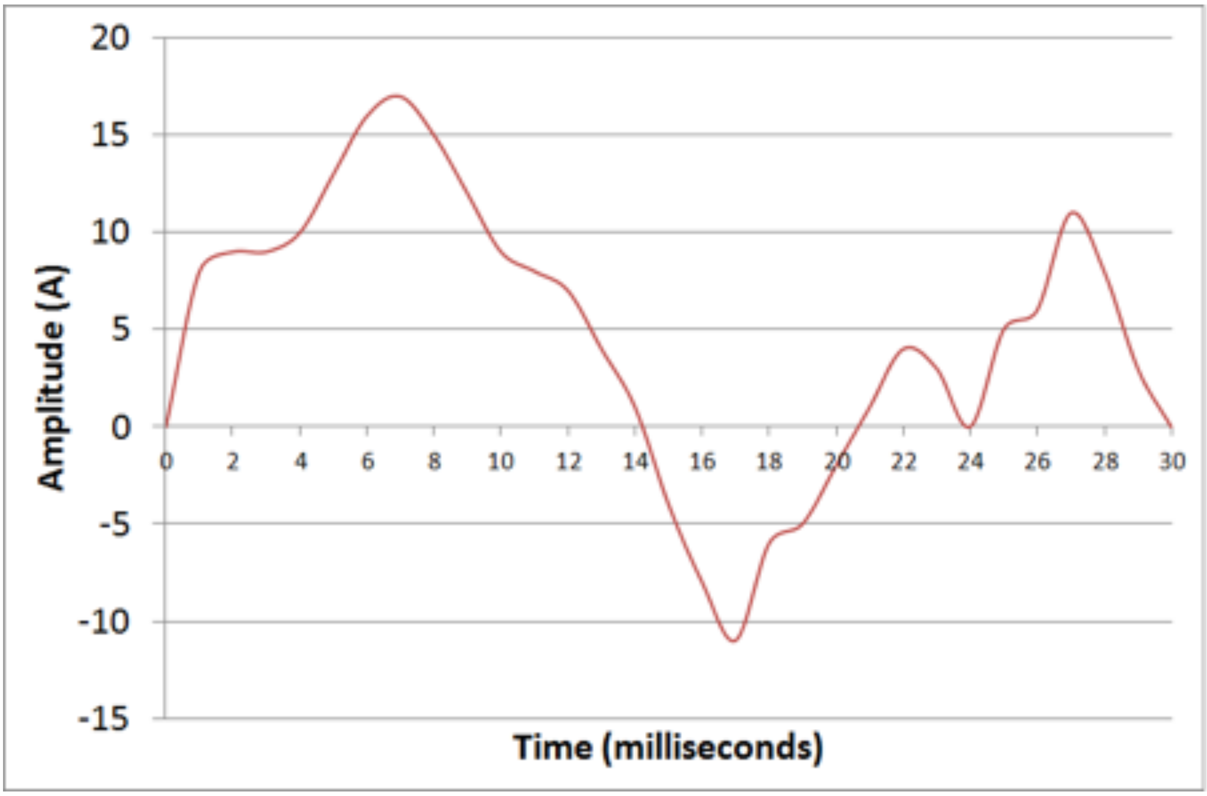
Questions

Q1 With reference to the diagram below, explain what is meant by the following terms:

(a) sampling

(b) sample rate

You may annotate the diagram or a rough sketch of the diagram.



Q2 Explain why analogue sounds cannot be stored in their original form on a digital storage device.

Q3 (a) Explain why increasing the **sample rate** will increase the **quality** of a sound.

(b) Explain one **disadvantage** of increasing the sample rate.

(c) Apart from adjusting the **sample rate**, name and explain two other factors that affect the **quality** of digital sound.

Q4 A sound is sampled at 1 kHz for 10 seconds at a bit rate of 2B. Estimate the file size.

Bibliography

BCS Glossary of Computing and ICT, 13th ed., BCS Academy Glossary Working Party pp49&55

http://searchnetworking.techtarget.com/definition/bit-rate