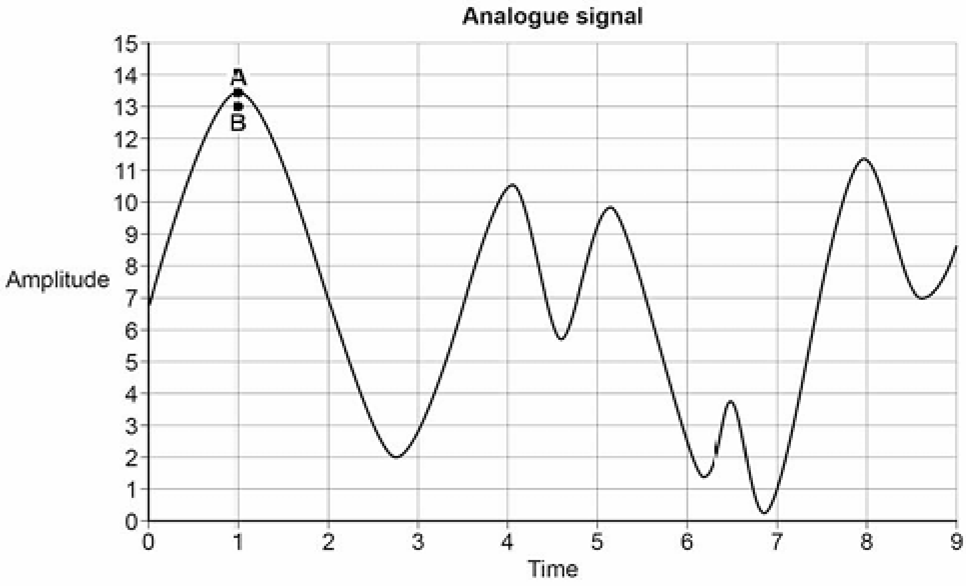
4.5.6.1 Representing Sounds

1) **Figure 1** shows an analogue signal represented as a waveform. The analogue signal is being converted to a digital signal by an analogue to digital convertor (ADC).

Figure 1



Points **A** and **B** in **Figure 1** indicate the amplitude of the waveform (**A**), at a point in time, and the value that was recorded for this measurement when the waveform was sample (**B**).

a) The waveform's amplitude is measure and recorded using a scale with 16 divisions, which are shown on the Y axis.

The recorded digital data indicates which division on the Y axis each measurement is closet to. For example, the closest division to measurement **A** is 13.

What sample resolution has been used?

[1 mark]

- **4 bits**

b) The graph covers a period of 0.01 seconds. During this period, 10 samples have been recorded at the times indicated by the divisions on the X axis in **Figure 1**.

What sample rate has been used?

[1 mark]

- **10 / 0.01 = 1000 Hz**

c) Explain the impact of the difference between **A** and **B** and how this difference could be reduced by redesigning the sampling system.

[2 marks]

- **It will not be possible to reproduce the original signal accurately**
- **The difference could be reduced by increasing the sample resolution**

d) A different analogue signal is being sampled. The highest frequency present in the signal's waveform is 1200 Hz.

What is the minimum sample rate that must be used during sampling to preserve all the frequencies in the waveform?

[1 mark]

- **2400 Hz**

---

2) A sound has been sampled and recorded. The second was sampled for 1 minute and 40 seconds at the sample rate 8000Hz with a 16-bit resolution.

A sample rate of 1Hz means that one sample has been taken every second.

Calculate the minimum amount of storage space, **in bytes**, needed to store the sampled sound.

[2 marks]

- **100 x 16 x 8000 / 8**
- **= 1,600,000 bytes**

---

3) An analogue to digital converter (ADC) was used during the sampling process.

Explain the principles of operation of an ADC.

[2 marks]

- **An analogue signal is sampled at regular time intervals**
- **The amplitude of the signal is measured**
- **This measurement is coded into a fixed number of bits**

---

4) MIDI is a system that can be used to enable musical devices to communicate and to represent music on a computer.

a) Describe how MIDI is used to represent digital music.

[2 marks]

- **Music is represented as a sequence of MIDI event messages**
- **Data that might be contained in a message:**
  - **Channel**
  - **Note on/off**
  - **Pitch /frequency/note number**
  - **Volume / loudness**
  - **Velocity**
  - **Key pressure /aftertouch**
  - **Duration / length**
  - **Timbre**
  - **Instrument**
  - **Pedal effects**
  - **Pitch bend**
- **MIDI messages are usually two or three bytes long**

b) Describe the advantages of using MIDI to represent music instead of using sampled sound.

[3 marks]

- **More compact representation**
- **Easy to modify e.g. change note values, change instruments**
- **Simple method to compose music algorithmically**
- **Musical score can be generated directly from a MIDI file**
- **No data lost about musical notes through sampling**
- **The MIDI file can be directly output to control an instrument/device**

---

5) A digital recording was made using a sampling rate of 44,100Hz with a 16-bit sample resolution.

The file, which stores only the recording is 17.199 megabytes in size.

Calculate the duration of the recording to the nearest second.
You should show your working.

[3 marks]

- **44,100 x 16/8 = 88100 bytes per second**
- **17199000 / 88100**
- **= 195 seconds**

6) A sound is sampled and recorded digitally.

a) The sound is sampled at a rate 48000 samples per second (Hz) for 3 minutes using a 16-bit sample resolution.

Calculate the size of the digital recording, giving your answer in mebibytes.

Give your answer rounded to 2 decimal places. You should show your working.

[2 marks]

- **48000 x 3 x 60 = 8640000 samples in total**
- **8640000 x 2 / 2^{20} = 16.48 mebibytes**

b) The highest frequency component in a different sound is 15000 Hz.

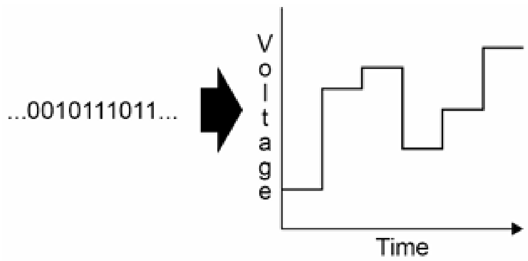
What is the minimum sampling rate that should be used when recording this sound to ensure that all the frequencies in the original waveform are preserved, so that when the recording is played back the original sound is recreated accurately?

[1 mark]

- **30000 Hz**

c) **Figure 2** shows part of the process of playing back a sound that has been sampled. The binary sound data is used to generate an electrical waveform.

Figure 2



A hardware component on a sound card carries out the process shown in **Figure 2.**

State the name of this component.

[1 mark]

- **Digital to analogue converter (DAC)**