# Homework 5 Representing sound Answers

1. An airline company is making noise-cancelling headphones used to provide a quiet environment for travellers.

The headphones consist of:

* Two small speakers
* One microphone
* A programmable microprocessor with permanent memory
* One analogue-to-digital convertor (ADC)
* One digital-to-analogue convertor (DAC)
	1. Firstly, external sound around the user is recorded by the microphone. What type of signal is output from the microphone? [1]

1 mark for:

* Analogue (voltage) signal
	1. This signal is passed into the analogue-to-digital converter where it is sampled to produce a digital version of the sound recorded. Explain how this sampling process is achieved. [3]

1 mark for each point:

* Analogue signal is sampled at regular intervals
* The rate is measured in Hertz
* Each sample is given a digital value based on a fixed number of bits
* This is known as sampling resolution
* These values are transmitted as a digital signal for later processing
	1. The choice of sampling frequency should be made based on the Nyquist theory. Explain, with a suitable formula, how this sampling rate can be determined in order to replicate the original signal. [2]

1 mark for each point:

* Sampling rate should be more than twice the highest frequency
* fs > 2fmax
	1. The maximum frequency humans hear is 22,000 Hertz.

	Components inside the headphone used to sample the ambient sound work at a sampling rate of 30,000 Hertz.

	The signal received by the processor is inverted and transmitted to the speakers. The DAC changes the digital signal into an analogue voltage signal used to drive the speakers.

	Evaluate how effective the speakers will be at cancelling the sound recorded by the microphone. [2]

1 mark for each point:

* Maximum sample rate is not more than twice the highest frequency
* As such, the signal will miss out components of high frequency sounds
* Resolution also not enough to accurately record the level of each sample
* Samples will be rounded to the nearest binary value
* Whilst this missing information will be small, in a quiet environment even the slightest sound will be noticeable
	1. The processor requires a buffer to hold 2 seconds of stereo audio with left and right channels. The headphones sample sound at a resolution of 16 bits. Calculate how big the buffer will need to be in kB. [3]

1 mark for each point:

* 1 second of audio will produce: 30,000 x 16 = 480,000 bits
* 2 seconds produces: 960,000 bits
* Left and right channels x 2 = 1,920,000 bits
* /8 = 240,000 bytes
* /1000 = **240kB**
1. A manager uses her mobile phone to record voice memos for later transcription when she returns to the office.

The diagram below shows a portion of the phone’s memory containing three samples of the original sound wave.

|  |
| --- |
| 1001 1011 1110 0101 |
| 1111 0001 1111 1111 |
| 0010 1101 1111 0010 |

1. What sampling resolution has been used? [1]

1 mark for:

* 16 bits
1. The phone recorded the audio at a sampling rate of 96,000 Hertz. Explain what this means. [2]

1 mark for each point:

* Hertz means number of samples taken per second
* A sample is a measurement of the sound wave
1. The manager is finding she is running out of storage space when making just a few recordings. She would like to be able to store more recordings and has the option to adjust the sampling rate.

Explain how lowering this value will create smaller files. [2]

1 mark for each point:

* Fewer samples taken per second
* Therefore entire sound recording will contain less measurements
* Resulting in a fewer total number of bits
1. There is 1 GB of storage left on the phone. Estimate how many complete 20-second recordings could be made if the sampling rate is 44 KHz with an 8-bit resolution. [2]

1 mark for each point:

* Each second of recording produces: 44,000 x 8 = 352,000 bits
* Each 20-second recording produces: 352,000 x 20 = 7,040,000 bits
* 7,040,000 bits / 8 = 880,000 bytes
* Number of recordings calculated as: 1,000,000,000 bytes / 880,000 bytes
* Correct answer (rounded down to ensure whole track recorded): 1136
1. A MIDI system is used to generate a short jingle before a public announcement.

The MIDI system represents sound as a sequence of event messages, each of two or
three bytes.

Give **two** advantages of using MIDI to represent music instead of using sampled sound. [2]

Less storage space required // sounds can be represented using less data;

Easy to modify / edit notes;

Easy to change instruments;

Musical score can be generated directly from a MIDI file;

No data lost about musical notes // through sampling;

Better quality needs to be substantiated e.g. background noise cannot be picked up.

Simple method to compose algorithmically;

Total 20 Marks