A-level Computer Science

Sound

Lesson Objectives

Students will learn about:

- Storing sound on a computer
- Sampling and Nyquist's theorem
- Factors that affect the audio quality of audio files in detail
- Musical instrument digital interface (MIDI) files
- How to calculate the size of an audio file

1.

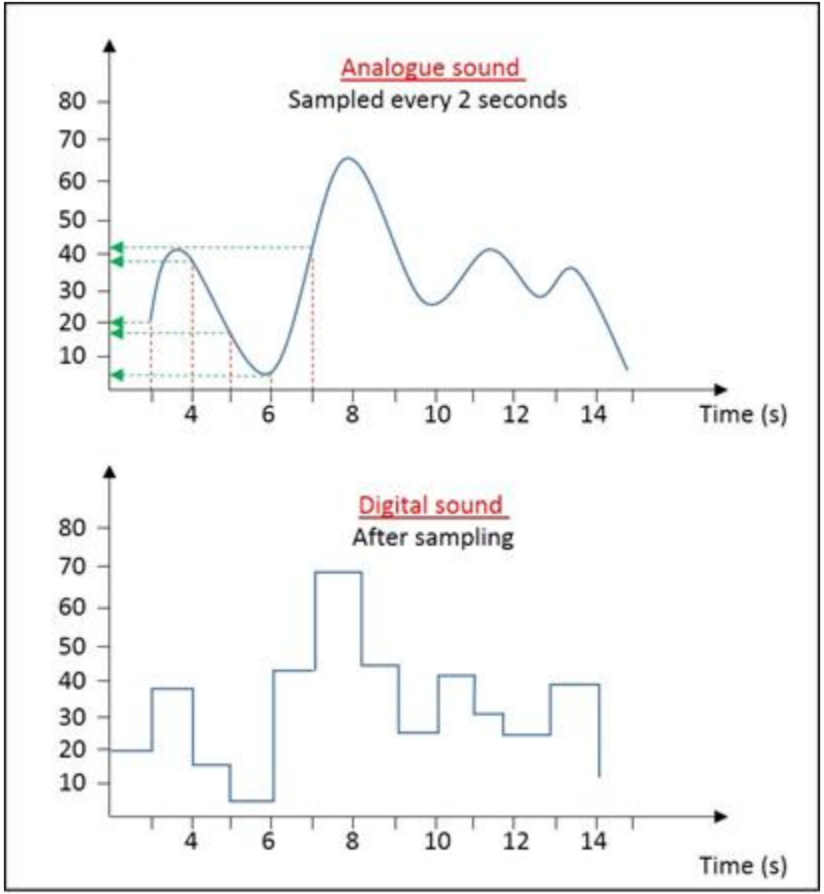
Content

Introduction

- Sound is another form of data that is stored on a computer.
- Similar to other types of information such as numbers, text, and images, the sound is also stored in digital form.
- But, sound, when produced, is in analogue form, that is, continuous varying data.
- To store sound in the computer, it is sampled to obtain thousands of samples per second where each sample corresponds to a binary value.

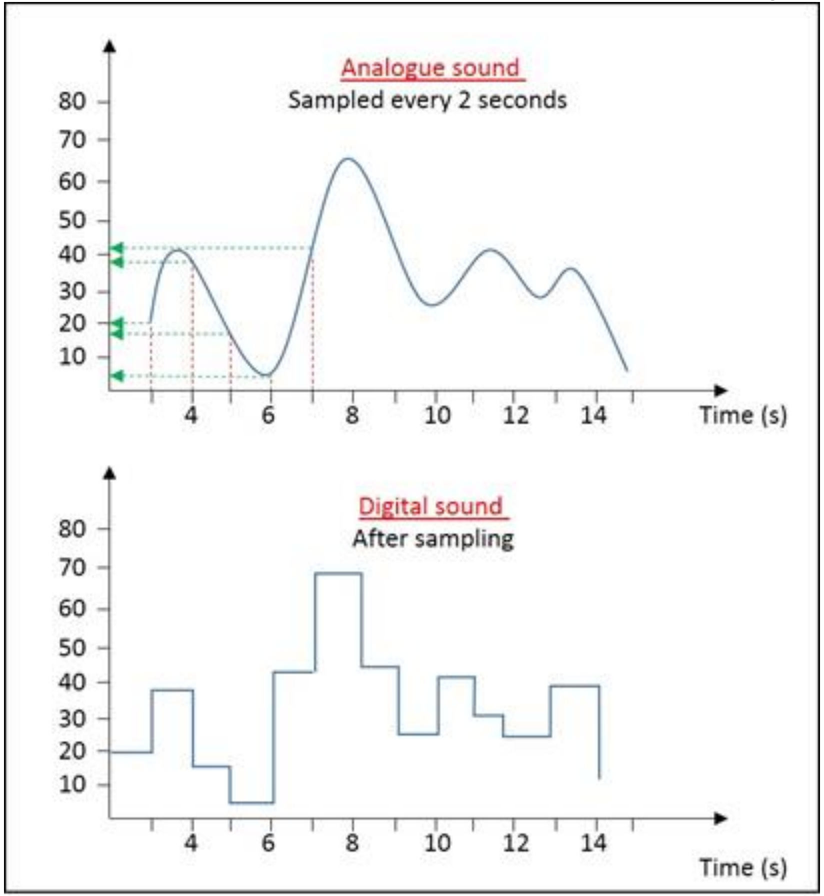
Sampling

- Sampling is the process that converts analogue sound into discrete digital data that can be stored on a computer.
- An analogue sound signal is shown.
- This sound is sampled every two seconds to obtain discrete digital sound.



Sampling

- We can see that the shape of digital sound is quite similar to analogue sound, but the curves are not smooth.
- A listener will feel that the digital sound is the same as the analogue sound but with a reduced quality.



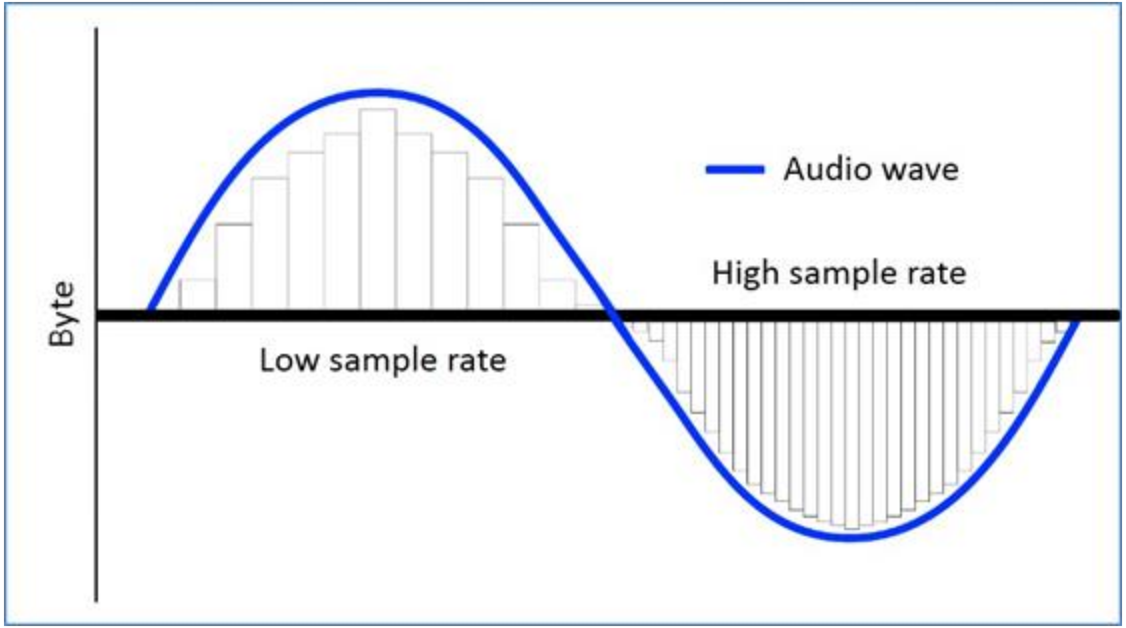
Factors affecting audio quality

Factor	Definition
Sampling rate	Number of samples per second
Bit depth	Number of bits used to represent each clip
Bit rate	Number of bits used per second of audio

Sampling rate/frequency

- The sampling rate is the number of samples taken per second.
- The higher the sampling rate, the higher the sound detail. The ups and downs of the sound wave can be recorded more clearly.
- The unit for sampling rate is also represented in Hertz (Hz).
- Each sample represents the amplitude of the wave at a certain point in time.

Sampling rate/frequency

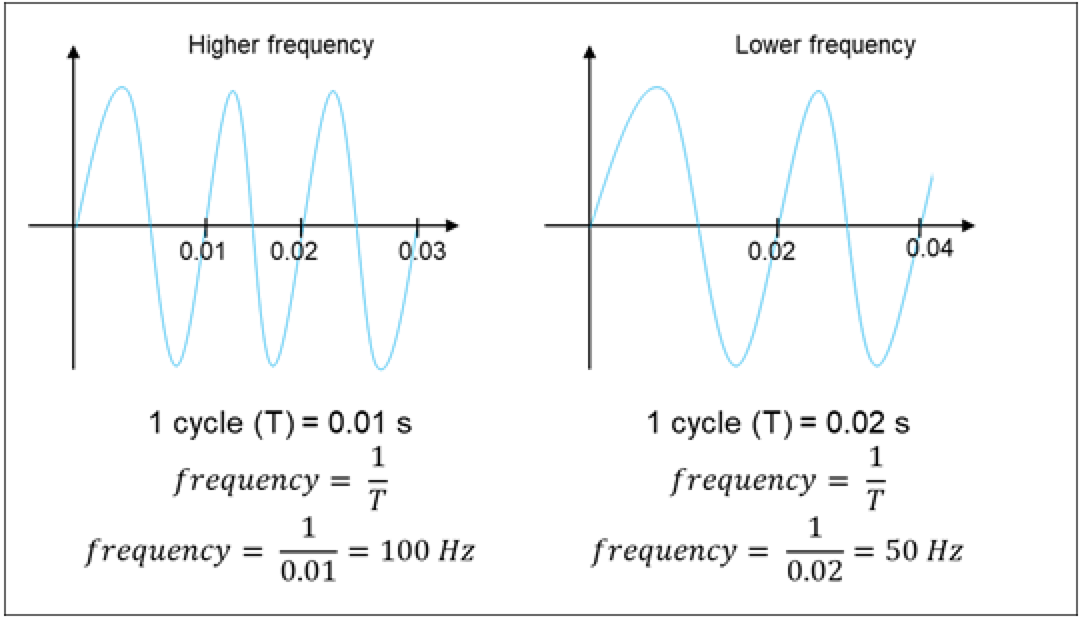


Sampling rate/frequency

- The most common sampling rate for music is 44,100 samples per second, which is 44,100 Hz (=44.1 kHz).
- A voice-over-Internet protocol (VOIP) has a sampling rate of 8 kHz, which is enough for the human voice to be heard clearly, but the quality is reduced to a certain extent.

Frequency

- The number of oscillations per second is called frequency.
- It is measured in Hertz (Hz) and controls the pitch of the sound.
- Sound waves with different frequencies are given.



Nyquist’s theorem

- Nyquist’s theorem states that for accurate sampling, the sampling rate must be at least twice the frequency of the highest frequency in the original sound signal.
- When the sampling rate is less than the frequency suggested by Nyquist’s theorem, the recording will not be accurate to the original sound.

Bit depth

- Bit depth is the number of bits available for each sample.
- The higher the bit depth, the higher the quality the audio will be.
- A CD has a bit depth of 16 bits, and a DVD has a bit depth of 24 bits.
- An n bit system can have 2^n different values. Hence, a CD can represent values from 0 to 65535(2^{16} -1).

Bit depth

- High-quality audio files are created as pulse-code modulation (PCM).
- PCM is the process for digitising a sound file and creating an uncompressed file.
- WAV and AIFF are a few examples of uncompressed audio file formats.

Bit rate

- Bit rate is the number of bits of data used to store data sampled every second. The unit for bit rate is kilobits per second (kbps).

$$\text{Bit rate} = \text{Sampling rate} \times \text{bit depth} \times \text{channels}$$

- An audio file has 44,100 samples per second, a bit depth of 16-bits and 2 channels (stereo audio). A bit rate of this file can be calculated as:

$$\text{Bit rate} = 44100 \times 16 \times 2 = 1{,}411{,}200 \text{ bits per second} = 1411.2 \text{ kbps}$$

Bit rate & audio quality

- A reasonable music audio must have a minimum bit rate of 128 kbps.
- The more the bit rate, the better the quality.
- This is the reason why the audio quality of a music CD is better than downloading from the internet.

Bit rate & file size

- A three-minute audio file with a sampling rate of 44,100 samples per second, bit depth 16 bits and 2 channels, has a bit rate of 1411.2 kbps per second.
- For 3 minutes, the number of bits required is, 1,411,200×180=254,016,000 bits.
- This value is equal to 254016000 ÷ 8 = 31752000 bytes = 31.75 megabytes (MB). This is the file size of a three-minute audio file.

Analogue to digital conversion

- An analogue signal is a continuous signal which represents physical measurements.
- A signal from a microphone is an example of an analogue signal.
- Sound signals in analogue form can be recorded in tape.
- The signals are amplified and played via a speaker.

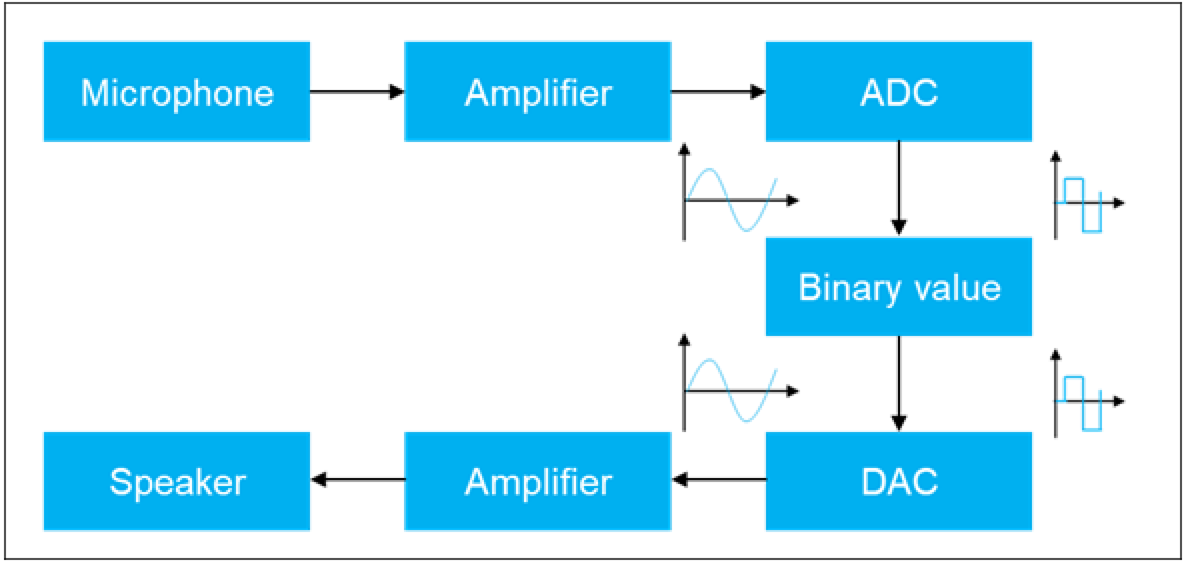
Analogue to digital conversion

- Whereas digital signal is a discrete signal that uses discontinuous values to represent information.
- For example, information stored in compact disc, computer, digital watch.
- An analogue signal is sampled at intervals to convert it to digital values which are then stored in devices such as compact disc and DVDs.

Analogue to digital conversion

- A microphone converts sound energy to electrical energy.
- In this process, sound in analogue form is converted to digital form by sampling the analogue signal at intervals.
- The amplitude at the intervals is measured and converted to binary value based on the bit depth.
- Digital values are converted to analogue signals by a digital to analogue converter (DAC). The signal is amplified and sent to a speaker.

Analogue to digital conversion



Let’s review some concepts

Sampling

Sampling is the process that converts analogue sound into discrete digital data that can be stored in a computer.

Bit depth

The number of bits used to represent each clip. An n-bit system can represent 2^n different values.

Bit rate

Number of bits used per second of audio. Bit rate=Sampling rate × bit depth × channels

Factors affecting audio quality

Sampling rate

Bit depth

Bit rate

Musical instrument digital interface (MIDI)

A communication protocol that enables electronic musical instruments to interact with each other using information and control signals.

Sampling rate

A number of samples per second. Usually represented in Hertz.

Nyquist’s theorem states that for accurate sampling, the sampling rate must be at least twice the frequency of the highest frequency in the original sound signal.

Calculating file size

File size (bits)= bit rate × number of seconds

2.

Activity

Activity-1

Duration: 10 minutes

1. A system has a bit depth of 8 bits. What range of values can it represent? Show your working.
2. Calculate the bit rate of a 32-bit depth system with sampling rate 44,100 samples per second and 2 channels.
3. What is the file size of a 4-minute song with bit rate found in question 2?
4. What happens to the file size and sound quality if the sampling rate is increased?

3.

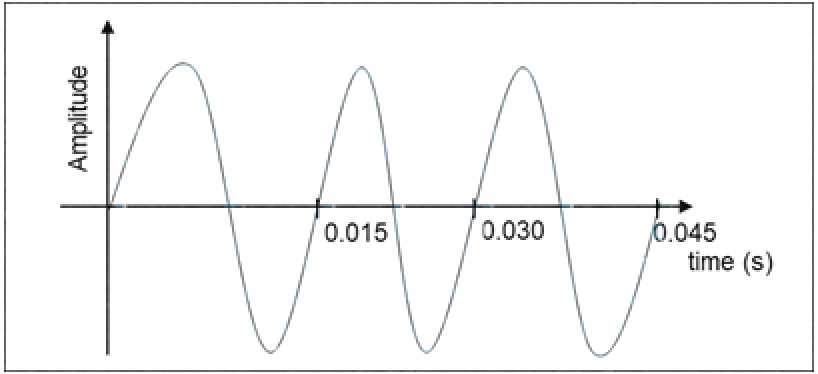
End of topic questions

End of topic questions

1. How is an analogue sound converted to digital form?
2. How is a sound file downloaded from the internet different from a sound file in CD?
3. What happens to a sound file if its bit depth is increased?
4. Why is the sampling rate of sound in VoIP just 8 kHz?
5. What are the factors by which the audio quality of a digital sound is affected?

End of topic questions

6. This question is about the waveform given.



i. What is the frequency of this signal?

ii. What is the minimum sampling frequency required to sample this signal so that an accurate recording can be made?