**Topic: Representation of sound**

Reading Time: 20 mins

**·        Note\* Highlight important/core points while reading**

·        Read the content and write the answers given in the document in your words, to get the solid grip on topic.

**Representation of Sound**

Computers store and process **sound** in **digital format**. However, sound in the real world exists as **analog waves**, which need to be converted into a digital format that computers can understand. This process is called **sampling**.

**1. How is Sound Stored in a Computer?**

Since computers only understand **binary (0s and 1s)**, sound must be converted from **analog** (continuous wave) to **digital** (discrete values). This is done through a process called **analog-to-digital conversion (ADC)**.

**Steps in Sound Digitization:**

1. **Sampling:** The sound wave's **amplitude** is measured at fixed time intervals (sampling rate).
2. **Quantization:** The measured values are rounded to the nearest available digital value.
3. **Encoding:** The quantized values are converted into **binary numbers** and stored in a digital file.

**2. How is Sampling Used to Record a Sound Clip?**

**Step-by-Step Process:**

·         The **amplitude** of the sound wave is measured at fixed time intervals (sampling rate).

·         This produces an **approximate digital representation** of the sound wave.

·         Each **sample** is converted into a **binary number** and stored in the computer.

This method allows computers to store and process audio files such as **MP3, WAV, and FLAC**.

**3. Key Factors Affecting Sound Quality**

**(i) Sampling Rate**

* The number of times the sound wave is measured **per second** (measured in **Hertz (Hz)**).
* Higher sampling rates produce **better sound quality** but require **more storage**.

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| --- | --- |
| **Sampling Rate** | **Quality Example** |
| 8 kHz | Telephone calls |
| 44.1 kHz | CD-quality audio |
| 96 kHz | Professional audio |

**Example:** A CD-quality sound recording has a **sampling rate of 44.1 kHz**, meaning **44,100 samples per second** are taken.

**(ii) Bit Depth**

* The number of **bits** used to store each sample.
* Higher bit depth means **more precise** sound recording.

|  |  |
| --- | --- |
| **Bit Depth** | **Possible Levels per Sample** |
| 8-bit | 256 levels |
| 16-bit | 65,536 levels |
| 24-bit | 16.7 million levels |

**Example:** A **16-bit** system can store **65,536 different amplitude levels**, making the sound **clearer** than an 8-bit system.

**(iii) Bit Rate**

* **Bit rate = Sampling Rate × Bit Depth × Channels**
* Measured in **kbps (kilobits per second)**.
* Higher bit rate = **better quality but larger file size**.

**Example:**

* **128 kbps MP3** = Medium quality
* **320 kbps MP3** = High quality

### ****4. Sound File Types****

Different formats are used to store digital sound:

|  |  |  |
| --- | --- | --- |
| **File Format** | **Compression** | **Quality** |
| **WAV** | No | High |
| **MP3** | Yes (Lossy) | Medium |
| **FLAC** | Yes (Lossless) | High |
| **AAC** | Yes | High |

**5. Comparing Analog and Digital Sound**

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| --- | --- | --- |
| **Feature** | **Analog Sound** | **Digital Sound** |
| **Nature** | Continuous wave | Discrete samples |
| **Storage** | Physical medium (vinyl, tape) | Binary (0s and 1s) |
| **Quality** | Depends on medium | Controlled by sampling rate & bit depth |
| **Editing** | Difficult | Easy (cut, edit, mix) |
| **Examples** | Microphone, vinyl records | MP3, CDs |

**A-Rated Questions/Answers By Examiner**

**Q1: What is meant by ‘sampling’ in digital sound recording?**

**Answer:** Sampling is the process of measuring the **amplitude** of a sound wave at **regular intervals** and storing these values as **binary data** in a computer.

**Q2: How does a higher sampling rate affect sound quality?**

**Answer:** A **higher sampling rate** captures **more details** of the sound wave, resulting in **better sound quality** but **larger file size**.

**Q3: Convert a 16-bit sound sample of ‘1010110010101100’ to decimal.**

**Answer:**

1. **1010110010101100₂**
2. Convert to decimal: **(1×2¹⁵) + (0×2¹⁴) + (1×2¹³) + (0×2¹²) + (1×2¹¹) + (1×2¹⁰) + (0×2⁹) + (0×2⁸) + (1×2⁷) + (0×2⁶) + (1×2⁵) + (0×2⁴) + (1×2³) + (1×2²) + (0×2¹) + (0×2⁰) = 44,300₁₀**

So, **1010110010101100₂ = 44,300₁₀**.

**Q4: Why is MP3 smaller in file size than WAV?**

**Answer:** MP3 uses **lossy compression**, removing **inaudible sound frequencies**, reducing file size while keeping acceptable quality. WAV files are **uncompressed**, keeping all data.

**Q5: What are the three factors that determine the quality of digital sound?**

**Answer:**

1. **Sampling Rate** (how often samples are taken per second).
2. **Bit Depth** (number of bits used per sample).
3. **Bit Rate** (amount of data processed per second).

### Write your Answers on your Notebook and Verify it on Next Screen

**Q6: How does bit depth affect the quality of digital sound?**

**Q7: What is the formula for calculating bit rate in digital sound?**

**Q8: Explain the difference between lossy and lossless compression in audio files.**

**Q9: Why do CDs use a sampling rate of 44.1 kHz?**

**Q10: Convert the 8-bit binary sound sample ‘11001101’ to decimal.**

**6. Answer:**

* Bit depth determines the **number of possible amplitude levels** in a digital sound recording.
* **Higher bit depth** means a more **precise** representation of the sound wave, reducing background noise.
* Example:
  + **8-bit audio** = **256 levels** (low quality, noticeable noise).
  + **16-bit audio** = **65,536 levels** (CD-quality).
  + **24-bit audio** = **16.7 million levels** (professional studio quality).

**Answer:** Higher **bit depth** = **better sound clarity** and **lower noise**.

**7. Answer:**

* **Formula:**  
  **Bit Rate = Sampling Rate × Bit Depth × Number of Channels**
* Measured in **kbps (kilobits per second)**.
* Example:
  + A **44.1 kHz, 16-bit, stereo (2-channel) audio file** has a bit rate of:  
    **44,100 × 16 × 2 = 1,411,200 bits per second (1,411 kbps).**

**Answer:** **Bit Rate = Sampling Rate × Bit Depth × Channels.**

**8. Answer:**

* **Lossy Compression:**
  + Removes **inaudible frequencies** to **reduce file size**.
  + Some sound quality is **lost**.
  + Examples: **MP3, AAC**.
* **Lossless Compression:**
  + **No loss of quality**, but **smaller file size** than uncompressed audio.
  + Perfect for **archiving** and **high-fidelity audio**.
  + Examples: **FLAC, ALAC**.

**Answer:** Lossy removes data to reduce size (**MP3**), while lossless compresses without losing quality (**FLAC**).

**9. Answer:**

* According to the **Nyquist Theorem**, the sampling rate must be at **least twice** the highest frequency to accurately reproduce sound.
* Human hearing range is **20 Hz to 20 kHz**.
* A sampling rate of **44.1 kHz** ensures **all audible frequencies** are captured while preventing distortion.

**Answer:** **44.1 kHz** is chosen to meet **Nyquist’s Theorem** for **accurate sound reproduction**.

**10. Answer:**

1. **11001101₂ → Decimal Calculation:**
   * (1 × 2⁷) + (1 × 2⁶) + (0 × 2⁵) + (0 × 2⁴) + (1 × 2³) + (1 × 2²) + (0 × 2¹) + (1 × 2⁰)
   * 128 + 64 + 0 + 0 + 8 + 4 + 0 + 1 = **205₁₀**.

**Answer:** **11001101₂ = 205₁₀ (decimal).**