**Sound Representation**What is sound

sound is a form of energy that needs a medium (like air, water, or a solid) to travel from one place to another through vibrations in that medium. These vibrations create alternating patterns of high- and low-pressure regions, which our ears can detect and interpret as sound.

Key terms used in encoding sound

**Analogue and Digital Data**

* **Analogue Data**: Sound waves in the physical world are continuous and represented by analogue signals, which are smooth and continuously varying over time.
* **Digital Data**: To store and process sound on computers and digital devices, analogue sound waves must be converted into digital data, which is a discrete representation of the continuous analogue signal.

**Sampling**

* **Sampling**: The process of converting a continuous analogue signal into a sequence of discrete digital values at specific intervals of time.
* **Sample**: A single digital value that represents the amplitude (or loudness) of the analogue sound wave at a specific point in time.

**Sampling Rate**

* **Sampling Rate**: The number of samples captured per second during the sampling process.
* **Nyquist Theorem**: According to the Nyquist theorem, the sampling rate must be at least twice the highest frequency present in the analogue signal to accurately reconstruct the original signal from the digital samples.
* **Common Sampling Rates**: Common sampling rates used for audio include 44.1 kHz (CD quality), 48 kHz (professional audio), and 96 kHz (high-definition audio).

**Sampling Resolution**

* **Sampling Resolution**: The number of bits used to represent the amplitude (or loudness) of each sample.
* **Bit Depth**: The number of bits used to represent the amplitude value of a single sample. Common bit depths include 8 bits (256 levels), 16 bits (65,536 levels), and 24 bits (16.7 million levels).
* **Quantization**: The process of approximating the continuous analogue amplitude values to discrete digital values based on the sampling resolution.
* **Quantization Error**: The error introduced by quantization, which can lead to noise or distortion in the digital representation of the sound.

Stages of sound processing

To create digital sound files that can closely recreate the original recorded sounds, the computer needs to represent the analogue sound waves in a digital form.

The first step is to pick up the sound waves using a microphone or other sound recording device which converts the vibrations in the air into electrical signals.

The signals are then processed by an **analogue to digital converter (ADC)**, which converts the electrical signals into digital values that can be stored on a computer.

The continuous electrical signal from the microphone is **sampled** at regular intervals by the ADC circuit. This process is called **sampling** and involves taking measurements of the **level of the analogue signal**, called **samples**, at regular time intervals.

During sampling, the amplitude (loudness) of the signal is measured and quantized to the nearest digital value based on the sampling resolution or bit depth.

* + The sampling rate determines how many samples are taken per second, affecting the maximum frequency that can be captured.
  + The bit depth determines the number of possible digital values that can represent the amplitude of each sample, influencing the dynamic range and audio quality.
  + The digitized samples are encoded into a specific digital audio format, typically using Pulse-Code Modulation (PCM) for uncompressed audio or a compressed format like MP3, AAC, or FLAC.

Once in a digital format, you can edit sounds with sound processing programs.

Playback

To listen to the recorded audio, the process is reversed:

* + - The digital audio data is read from the storage medium.
    - The encoded data is decoded back into a sequence of digital samples.
    - A Digital-to-Analogue Converter (DAC) reconstructs the analogue electrical signal from the digital samples.

The analogue electrical signal is amplified and sent to speakers or headphones, which convert the electrical signal back into sound waves that our ears can perceive.

Calculating the file size of Audio

File size (in bytes) = (Sample rate \* Bits per sample \* Channels) \* (Duration in seconds \* 8)

Where "Channels" is either 1 for mono or 2 for stereo

Example, if you have a stereo audio file with a sample rate of 44.1 kHz, a bit depth of 16 bits per sample, and a duration of 5 minutes (300 seconds), the file size can be calculated as follows:

File size = (44100 \* 16 \* 2) \* (300 \* 8)  
File size = 2067200 \* 300  
File size = 6201600000 bytes = 620.16 MB

**Summary and Revision**

The sampling rate, also known as the sample rate, is the number of times per second that a sound waveform is measured and recorded as a series of samples. The higher the sampling rate, the more accurately the sound can be captured and reproduced. The standard sampling rate for CD-quality audio is 44.1 kHz, but higher sampling rates (such as 96 kHz or 192 kHz) can be used for higher-quality recordings.

On the other hand, the sound resolution, also known as the bit depth, is the number of bits used to represent each sample. The higher the bit depth, the more accurately the sound can be represented, resulting in a wider dynamic range and lower noise floor. The standard bit depth for CD-quality audio is 16 bits, but higher bit depths (such as 24 bits or 32 bits) can be used for higher-quality recordings.

In summary, changing the sound sampling rate and resolution can have a significant impact on the sound quality of recorded audio. A higher sampling rate will capture the sound more accurately and a higher resolution will provide a wider dynamic range and lower noise floor. However, increasing the sampling rate and resolution also increases the file size and storage requirements.

**Sample rate**

Another aspect to consider is how often the ADC should sample the sound wave. If the ADC takes samples with a low frequency (i.e. not very often), the resulting sound will sound very different to the original one. On the other hand, if the ADC takes samples too frequently, the digitised sound will resemble the original one, but the sound file will be very large, because it has to store a large number of samples.

The frequency used to sample an analogue signal is called the **sampling rate**. It is defined as the **number of samples taken per second** and it is measured in hertz (one hertz is equal to one sample per second). The higher the sampling rate, the better the quality of the audio recording, and the bigger the file size of the recording.

**Sample Resolution**

Every sample corresponds to a level of the electrical signal that was sampled, and that in turn represents the value of the height of the sound wave at a point in time. The **sample resolution** is the number of bits used to represent each sample.

The sample resolution determines the number of available digital values that can be used during sampling. If you have a low sample resolution (very few bits per sample), then the available binary patterns for the samples will be very limited, and the variations in the analogue signal will not be represented accurately in a digital form. This means that the values of the analogue signal will have to be stored as the closest digital value that is available. For example, in the below diagram, using a sample resolution of only 2 bits means that measurements of different levels of the signal have to be stored using the same binary pattern. A sample resolution of 3 bits allows for a more accurate representation of each sample.

On the other hand, if you have a high sample resolution, then the file size of the recording will be larger, so the file will take up more storage space. This might not be practical, depending on the intended use of the recording. For example, the sample resolution for a CD is 16 bits per sample, which allows for 216 = 65536 binary patterns to be used to represent the samples