TERMS

SAMPLING RATE

In digital audio, **44,100 Hz** (alternately represented as **44.1 kHz**) is a common sampling frequency. Analog audio is recorded by sampling it 44,100 times per second, and then these samples are used to reconstruct the audio signal when playing it back.

44.1 kHz audio is widely used, due to this being the sampling rate used in Compact Discs, dating back to its use by Sony from 1979.

- Nyquist-Shannon sampling theorem says the sampling frequency must be greater than twice the maximum frequency one wishes to reproduce. Since human hearing range is roughly 20 Hz to 20,000 Hz, the sampling rate had to be greater than 40 kHz.

SAMPLE

A single, scalar value representing the amplitude of the sound wave in one channel of audio data.

CHANNEL

An independent waveform in the audio data. The number of channels is important: one channel is “Mono,” two channels is “Stereo” – there are different waves for the left and right speakers. 5.1 surround sound has 5 channels, one of which is for the lowest sounds and is usually sent to a subwoofer. Again, each channel holds audio data that is independent of all the other channels, although all channels will be the same overall length.

FRAME

A frame is like a sample, but in multichannel format – it is a snapshot of all the channels at a specific data point.

SAMPLING RATE

The number of **samples** (or **frames**) that exist for each second of data. This field is represented in Hz, or "per second." For example, CD-quality audio has 44,100 samples per second. A higher sampling rate means higher fidelity audio.

BIT DEPTH / BITS PER SAMPLE

The number of bits available for one sample. Common bit depths are 8-bit, 16-bit and 32-bit. A sample is almost always represented by a native data type, such as byte, short, or int. A higher bit depth means each sample can be more precise, resulting in higher fidelity audio.

BLOCK ALIGN

This is the number of bytes in a frame. This is calculated by multiplying the number of channels by the number of bytes (not bits) in a sample. To get the number of bytes per sample, we divide the **bit depth** by 8 (assuming a byte is 8 bits). The resulting formula to calculate block align looks like **blockAlign = nChannels * (bitsPerSample / 8)**. For 16-bit stereo format, this gives you **2 channels * 2 bytes = 4 bytes**.

AVERAGE BYTES PER SECOND

Used mainly to allocate memory, this measurement is equal to **sampling rate * block align**.

PCM

Pulse-code modulation (**PCM**) is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, Compact Discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.