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| **BIT DEPTH** |  | The difference between the actual analog value and quantized digital value. This error is either due to [rounding](http://en.wikipedia.org/wiki/Rounding) or [truncation](http://en.wikipedia.org/wiki/Truncation). The error signal is sometimes modeled as an additional random signal |
| **ALIASING** |  | It occurs when a signal above half the sample rate is allowed into, or created within, a digital system. The effect produces unwanted frequencies |
| **DAC** |  | The number of samples of a sound that are taken per second to represent the event digitally. |
| **QUANTISATION NOISE** |  | States that a signal must be sampled at a rate greater than twice the highest frequency component of the signal to accurately reconstruct the waveform; otherwise, the high-frequency content will alias at a frequency inside the spectrum of interest. |
| **ADC** |  | Digital to analogue converter. Takes digital information and translates it back into analogue form |
| **SAMPLE RATE** |  | Describes the potential accuracy of a particular piece of hardware or software that processes audio data. In general, the more bits that are available, the more accurate the resulting output from the data being processed. |
| **Pulse Code Modulation(PCM)** |  | Analogue to digital converter. Translates normal waveforms(sound) and stores it as discrete steps of a numeric value |
| **DIGITAL** |  | A method used to [digitally](http://en.wikipedia.org/wiki/Digital_signal) represent sampled [analog signals](http://en.wikipedia.org/wiki/Analog_signal). It is the standard form of [digital audio](http://en.wikipedia.org/wiki/Digital_audio) in computers, [Compact Discs](http://en.wikipedia.org/wiki/Compact_Disc), [digital telephony](http://en.wikipedia.org/wiki/Digital_telephony) and other digital audio applications. |
| **The Nyquist theorem** |  | A system where information is stored in a series of discrete numeric values that can processed, manipulated and stored by a computer system |
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Digital Audio Key words