SAMPLE RATE

In developing an audio sound for computers or telecommunication, the sample rate is the number of samples of a sound that are taken per second to represent the event digitally.

The more samples taken per second, the more accurate the digital representation of the sound can be. For example, the current sample rate for CD-quality audio is 44,100 samples per second. This sample rate can accurately reproduce the audio frequencies up to 20,500 hertz, covering the full range of human hearing.

[Source: http://whatis.techtarget.com/definition/sample-rate]

Sampling rate or sampling frequency defines the number of samples per second (or per other unit) taken from a continuous signal to make a discrete or digital signal. For time-domain signals like the waveforms for sound (and other audio-visual content types), frequencies are measured in hertz (Hz) or cycles per second. The Nyquist –Shannon sampling theorem (Nyquist principle) states that perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled. For example, if an audio signal has an upper limit of 20,000 Hz (the approximate upper limit of human hearing), a sampling frequency greater than 40,000 Hz (40 kHz) will avoid aliasing and allow theoretically perfect reconstruction.

[Source: http://www.digitizationguidelines.gov/term.php?term=samplingrateaudio]

Nyquist Sampling Theorem

The Nyquist Sampling Theorem explains the relationship between the sample rate and the frequency of the measured signal. It states that the sample rate f_s must be greater than twice the highest frequency component of interest in the measured signal. This frequency is often referred to as the Nyquist frequency, f_N.

$$f_s > 2 * f_N$$