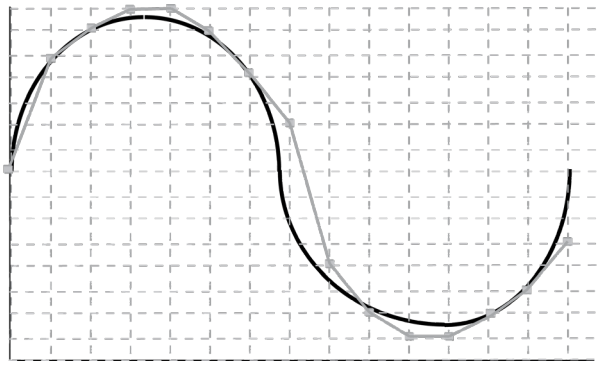
Bits, Bit Depth, Sample Rate and Conversion **Karen Collins www.gamessound.com**

A **bit**, derived from binary digit, is the smallest unit of information in computer language, a one (1) or zero (0) (also sometimes referred to as "on or off", or "white or black"). In referring to processors, the number of bits indicates how much data a computer's main processor can manipulate simultaneously. For instance, an 8-bit computer can process 8 bits of data at the same time.

Bits can also be used to describe sound fidelity or resolution. **Bit depth** is used to describe the number of bits available in a byte. Higher bit depths result in better the quality, but larger file sizes. 8 bits can represent 28 (binary being base 2), or 256 variations in a byte. Adding one bit doubles the accuracy, or number of levels. At 16 bits, there are 65,536 possible states (216 = 65,536), or variations of sixteen ones and zeroes. When recording sound, 256 divisions are not very accurate, since the amplitude of a wave is rounded up or down to fit the nearest available point of resolution. This process, known as quantisation, distorts the sound or adds noise. CD quality sound is considered 16-bit, although often the CDs are recorded in 24-bit and converted to 16-bit before release.



This diagram simplifies the process, by showing a 4-bit sample (16 sample points along the positive and negative amplitudes), with amplitudes sampled at 16 times per second. The black wave line shows the original sound wave, and the grey dots show the sample points that would occur. As you can see from the grey line, the original sound is considerably changed by the sampling of the sound at a low rate.

Sample Rate

A **sample** is a measurement of amplitude. A sample contains the information of the amplitude value of a waveform measured over a period of time. The sample rate is the number of times the original sound is sampled per second, or the number of measurements per second. Sample rate is also known as sample frequency: A CD quality sample rate of 44.1 KHz means that 44100 samples per second were recorded. If the sample rate is too low, a distortion known as aliasing will occur, and will be audible when the sample is converted back to analogue by a digital to analogue converter (DAC). Analogue to digital converters will typically have an anti-aliasing filter which removes harmonics above the highest frequency that the sample rate can accommodate.

DACs

The recreation of a sound wave from sample data (binary code) to an analogue current (an electrical pressure soundwave) is performed by a **digital to analogue converter**, or DAC. DACs have bit depths and sample rates. The higher the bit rate and sample rate, the better the resulting sound. DACs most often work through pulse code modulation. **Pulse Code Modulation** (PCM, otherwise known as raw, or AI2 synthesis), refers to an analogue sound converted into digital sound, by sampling an analogue waveform. The data is stored in binary, which is then decoded and played back as it was originally recorded. The down side of this method is the amount of space required to store the samples: as a result, most PCM samples are limited to those sounds with a short envelope, such as percussion. 8-bit PCM samples commonly have an audible hiss due to the resolution problems. **ADPCM, adaptive differential PCM** (also known as adaptive *delta* PCM), is essentially a method of compressing a PCM sample. The difference between two adjacent sample values is quantified, reducing the pitch or raising the pitch slightly, to reduce the amount of data required. ADPCM uses only 4 bits per sample, therefore requiring only one quarter of the space of a 16-bit PCM sample. This works well for lower frequencies, but at higher frequencies can lead to distortion. PCM and ADPCM are now subclasses of the Microsoft waveform, WAV, although Windows system hardware does not understand ADPCM, and so it must be decompressed before playing as a PCM sample.

Pulse Width Modulation works by generating variable-width pulses to represent the amplitude of an analog input signal (sample). The PWM method can attain higher volume, and achieve a range of interesting timbres (such as a pseudo-chorus or phasing sound), but the samples are low quality (4-bit). On the Commodore 64, the noise channel could double as a simple PWM sampler. PWM was used for sampling short sounds like percussion, and to simulate an LFO to the volume (creating a tremolo effect, as heard on *Parallax*)