Understanding Digital Audio

Analogue vs Digital Audio

You've most likely encountered conversations regarding analogue vs digital sound (vinyl vs MP3, for instance) - it's a widely debated topic online and amongst music and audio enthusiasts.

But just what is the difference between these signal paths and what makes the sound different?
You probably understand that your CD player, iPod and computer are digital audio devices and record players are analogue audio devices, but it's important to understand how digital audio works on the inside of these devices so that you can make a proper choice on your digital audio equipment.

An analogue signal, by definition, is:
"*A nominally continuous electrical signal that varies in amplitude or frequency in response to changes in sound, light, heat, position, or pressure".*

Analogue can be electrical or mechanical but the key word here is "continuous."

An analogue signal path implies a continuous signal in contrast to a digital signal path, which breaks everything into numbers. This is the primary difference between analogue and digital sound.

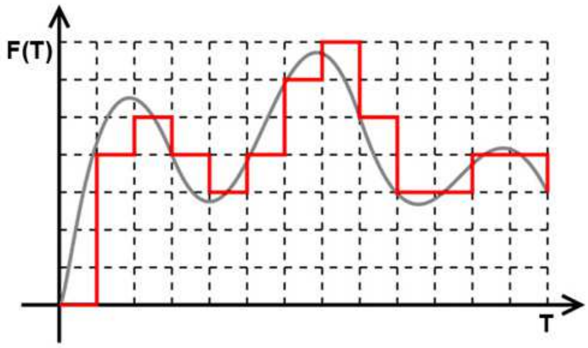
Until the mid 1980′s almost all audio recording devices were analogue. That is to say they all used a mechanical or electrical recording methods to capture a continuous waveform. Around this time digital recording started to become affordable and eventually it became the most cost effective way to create music – which is why so many of us use digital devices to create sound today.

DAC – Digital to Analogue Converter.

A Digital to Analogue Converter (DAC) is something that most of us take for granted. There's one in your satellite TV box, one in your CD Player and one in your computer. This device **is the heart of your hearing experience** with all forms of digital audio. When it comes to professional audio we want to use a high quality DAC to create a cleaner and sometimes more enjoyable experience.

When recording audio into your computer the Analogue to Digital Converter (ADC) is the soul of your recording experience. This is what turns your guitar or voice etc into binary data to be used by the computer.

Digital to Analogue converters are manufactured almost exclusively on integrated circuits (microchips) and the best ones are created by a few companies who specialize in this type of chip architecture. Therefore many audio interfaces share the same DAC circuits. There are many kinds of DAC circuits, however, and the industry is constantly trying to create better chips.



Bit Depth / Resolution

The waveform above represents an analogue signal/sample (grey) and a digital signal/sample (red.) Notice that the analogue signal is a smooth curve, whereas the digital signal is broken into a grid-like shape. While this grid is not entirely accurate (it's more for the sake of example), it helps to illustrate the idea of "bit depth."

Bit depth describes the number of bits of information recorded for each sample.

When you read "16 bit" or "24 bit," the bits represent the resolution – how many dots will help create that nice curve of the waveform. The fewer dots, or bits, the more grid-like your wave will be and the more "grainy" the reproduction of sound will be; the more bits, the more accurate the curve and therefore more accurate sound.

16 bit is the standard resolution for CDs and is generally acceptable for analogue to digital recording. 24 bit will give you a cleaner sound and more accurate representation of the curve. Some systems go even higher than this but understand that with more bit depth you will be pushing your processor to work harder.

Sampling Rate

Bit depth and sampling rate determine the quality or accuracy of a digital recording. While bit depth is sort of easy to explain, Sampling Rate is a bit trickier.

Sampling rate *defines the number of samples per unit of time (usually seconds) taken from a continuous signal to make a discrete signal*.
For time related signals the unit for sampling rate is Hertz. Perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled, or equivalently, when the Nyquist frequency (processing speed - half the sample rate) exceeds the highest frequency of the signal being sampled.

In practice this means that a minimum sampling rate of 40kHz should allow for accurate reproduction of 20Hz – 20,000kHz – the accepted range of human hearing. This was initially the standard for digital audio recording until it was realized that human beings could actually perceive sounds above and below that range. For this reason you now find interfaces that record at 48kHz and higher, with CDs using the standard: 44.1kHz sample rate.