CHAPTER 5 DIGITAL AUDIO BASICS

DIGITAL AUDIO

How analog voltage gets converted into digital 1's and 0's.

SAMPLE RATE

Digital audio is simply samples or pictures of sound. The rate at which these audio samples or pictures are taken is the *Sample Rate* and is measured in *Frequency*.

Sample rate determines the frequency range of the sound recorded or played back. The human hearing range is 20 Hz – 20 kHz. The higher the frequency of a sound, the faster the sampling rate must be to capture the sound.

Think of a strobe light. If the strobe light is flashing slower than 24 times a second, motion seems to be broken. It makes a cool effect on the dance floor, but the brain perceives that there are gaps in the visual information. If the strobe light frequency, how often it flashes, increases, more visual information is provided. This is quite analogous to the frequency rate of digital audio. The more pictures that are taken, the higher the resolution of the captured audio and the closer it will relate to the original audio signal.

Q: So, How fast should a digital audio system sample in order to capture the entire hearing spectrum?

A: 40,000 Hz or 40 kHz

This is based on the ***Nyquist Theorem***. The Nyquist Theorem basically states that to accurately sample an audio source, the frequency must be **twice** that of the highest frequency of the source audio.

Q: Why double the frequency?
A: All sound is made up of sine waves. A sine wave has a positive crest and a negative trough that corresponds to a speaker pushing out and retracting in. To represent a complete sine wave, a digital system must represent both the positive crest and negative trough of the waveform. A single cycle of a sine wave has two points of reference that must be represented so the sampling frequency must be double.

Of course, the Nyquist Theorem is only a theory. When only two points of a sine wave are sampled the resulting output is a triangle wave. While the sine wave and the triangle wave sound similar, the triangle wave has harmonic content and is not a true representation of the original sound source even though it fools the ear.

See the Chapter 4 on subtractive synthesis for more information on sine and triangle waveforms and their harmonic content.

Sample Rate:	**Format:**
8 kHz	Telephone, iPod memos
32 kHz	DAT, Semi-pro consumer products
44.1 kHz	CD, DAT, Studio
48 kHz	Film, audio for video
88.2 kHz	Studio
96 kHz	DVD audio, Studio
192 kHz	High-end Pro Studio

BIT RATE

Bit rate determines the **Dynamic Range** of the sampled audio or how a digital system can represent the difference between the softest and loudest sounds.

Each added bit doubles the dynamic range.

Bit Rate:	**Dynamic range possible values:**
1 bit	2
2 bit	4
3 bit	8
4 bit	16
5 bit	32
6 bit	64
7 bit	128
8 bit	256
9 bit	512
10 bit	1,024
11 bit	2,048
12 bit	4,096
16 bit	65,536
20 bit	1,048,576
24 bit	16,777,216
32 bit	4,294,967,296

Early samplers were 8 bit and 12 bit.

4 bit: telephone

8 bit sampler: Ensoniq Mirage – used extensively on Janet Jackson's *Rhythm Nation*

12 bit samplers: EMU SP1200 drum sampler