GoldenEar Technology™
A Passion for Sonic Perfection™
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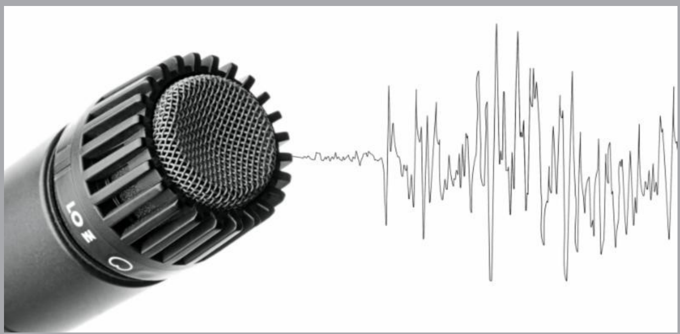
How Do You Do Digital?

(The first in a 2-part article series dealing with digital recording ...)



Audio aficionados may have several musical libraries available to them. CD or LP collections, Internet radio, local streaming and storage sources, the “cloud”, portable devices like MP3 players, cell phones, tablets and more. The list seems to be constantly expanding and our media access is enormous compared to just a decade or two ago. If you look back at that list you’ll find most current sources are digital and many are compressed. A GoldenEar owner who avidly reads these newsletters sent in a question concerning the benefits and drawbacks of compressed digital music storage. Before we go into that, we thought a fairly simplified description of the digital recording process and information about data reduction might be interesting and helpful. We’ll do that in this edition and in the next issue we’ll cover digital formats and compression schemes. So put your “geek” hat on and let’s see.

Analog Versus Digital

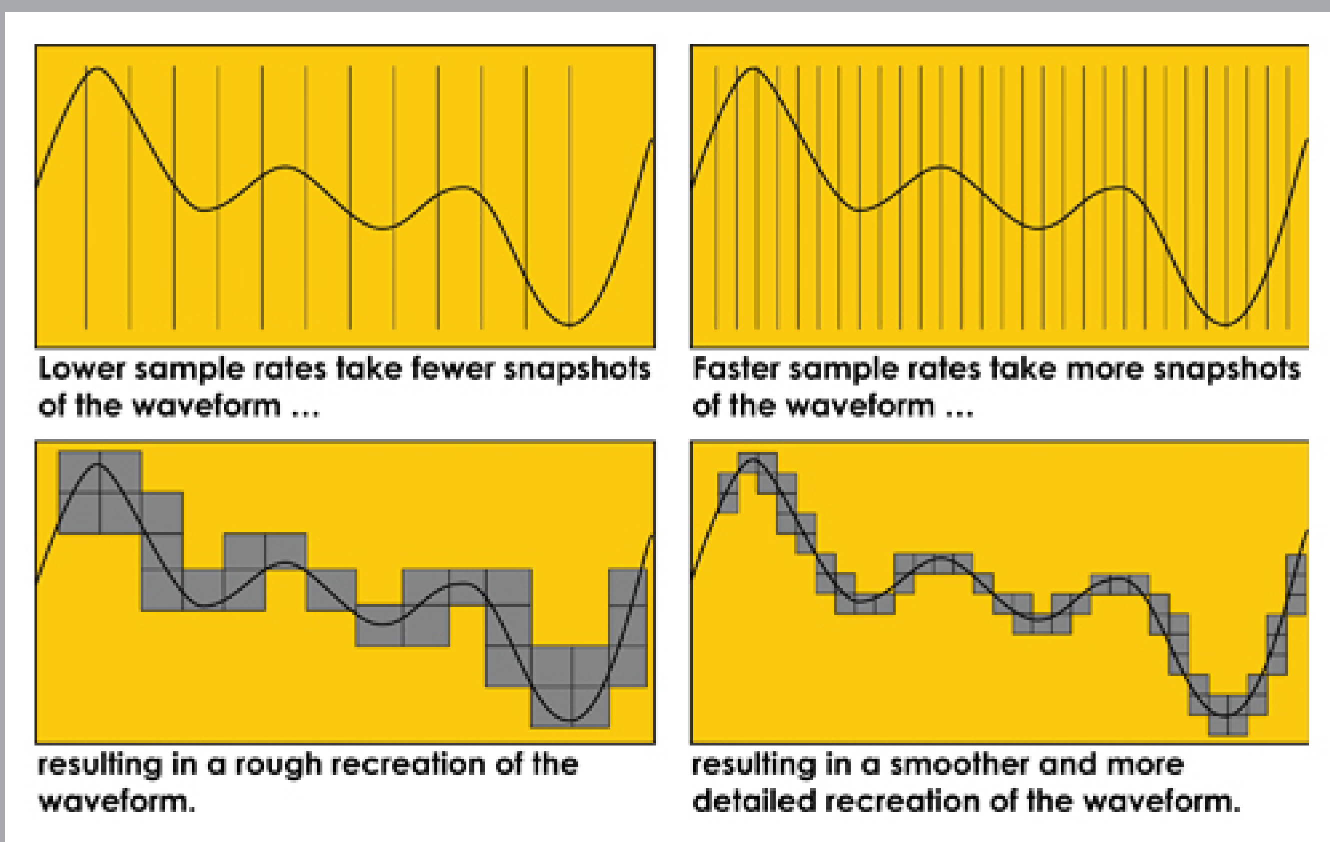
An analog recording takes the electrical signal (waveform) coming from one or more microphones (or other signal source) and records it using an analogue storage medium (tape recorder, vinyl record, etc.) without changing it into a digital format. If you were to look at the voltage output pattern from the source and the stored recording they should be identical, assuming a quality transfer with low distortion. For example, the vinyl record making process uses a mechanical reproduction of that musical voltage carved into a “lacquer disc”. Through successive processes eventually those groves are pressed into hot, soft slabs of vinyl. When the cooled record is played back, the phono needle traces the grooves and (hopefully) re-generates the identical electrical signal to be amplified for your listening pleasure.

This is an analog to analog recording/listening process and is still highly regarded by many audiophiles. But it’s quality is variable based on many factors. In any recording, the sound quality of the final product depends upon every step in the process including the desires of the artist, the producer and the recording engineer, not to mention the room/speakers used in the monitoring process and the target market for the music. Yes, they may actually modify the mix and EQ the sound based upon the type of music that’s being recorded. So you can end up with very bass heavy rap and hip hop, loud and extremely compressed (no soft passages, everything is just LOUD) pop music, relatively accurate jazz, country and classical, etc. For analog records the quality of the pressing process, the vinyl used, etc., all impact the final sound too.

Transitioning to the World of Ones and Zeros

Sony™ and Philips™ presented the first "Red Book" digital recording standards detailing the CD recording specifications in 1980. The first CD players came to market shortly thereafter and opened the door to the digital media revolution that's so dramatically changing our world.

The Red Book CD recording process uses something called Pulse Code Modulation (PCM). It works this way; we still have that musical equivalent electrical signal coming from one or more microphones or other sources. But rather than an analogue recording device, we feed it into a computer which "samples" (measures) the incoming voltage 44,100 times every second (that's the sampling frequency). Each time it samples the voltage it gives that value a name. Since we'd run out of names pretty quickly if we called these voltages names like "Warren" or "Marge", we have the computer use a combination of up to 16 ones and zeros for the names. That's where the "16 bit" part of the Red Book CD sampling standard comes from. 16 bit samples give you the potential to identify 65,536 different named voltage values. If you could see one of these names it would look something like 0010111010110011. The 44,100 Hz sampling rate and 16 bit "word depth" are typically stated when describing this recording process as "44.1 kHz, 16 bit" recording, or more often, 44/16.



Lower sample rates take fewer snapshots of the waveform ...

Faster sample rates take more snapshots of the waveform ...

resulting in a rough recreation of the waveform.

resulting in a smoother and more detailed recreation of the waveform.

The higher the sampling rate, the better the resolution is in the reproduction of the original waveform (see simplified example, above), think more pixels in the picture on your TV set, and the higher the frequency the recording can reproduce. The greater the bit rate (bit depth), the greater the dynamic range potential of the recording will be. This is directly related to the signal to noise ratio of the recording; so for example, a 16 bit sample is capable of delivering about a 96dB S/N.

Today you'll find other sampling rates mentioned including 48, 96 and even 192kHz. Just like with the Red Book standard, the computer assigns a "name" to each voltage sample it takes, except these samples are taken at 48, 96 or 192 thousand times per second. Having so many samples means there has to be lots more potential combinations of ones and zeros to create lots more names. These higher sampling rates work with 24 bit word depth instead of the Red Book 16 bit. 24 Bit samples can identify over 16 million possible voltage levels (names). The Blu-Ray and DVD-Audio standards use 16, 20 or 24 bit depth recording and storage systems with high level sampling rates.

SACD's (Super Audio CD) incorporate an alternative digital recording technique that Sony and Philips incorporated and for their purposes named "Direct Stream Digital" (DSD). It was originally developed years earlier and uses "1-bit" Pulse Density Modulation (PDM as opposed to PCM), recording a stream of single bit values at a very high sampling rate of 2.8224 mHz, or 64 times the sampling rate of standard Red Book CDs. However, because the samples are single bit, their bit depth is 1/16th that of Red Book PCM recording. There is some disagreement among experts as to whether DSD actually has the capability to deliver better sound quality than high resolution PCM recordings. The DSD system is more complex to explain and we're gonna leave it here. If you're an engineer, or just want to turn your brain into oatmeal, you can find more information concerning the ones and zeros of DSD and PCM on the Web.

Even with digital recording and playback there are variables that impact the end result sound quality. You still have the artist, producer, etc. and the sound quality of the electronics in the recording and playback systems. But the digital medium has the potential to deliver excellent and consistently repeatable performance with long term stability of the storage mediums used.

See? Simple, huh? Don't you feel like the Magical Mystery Curtain has been pulled back to reveal some of the inner workings of digital recording? Well, just note that our overly simplified explanations have left out lots of important details and processes that are necessary to actually make the system work. And on your end, your audio systems' digital to analog converters (DAC) will ultimately determine how those digital recording actually sounds on your GoldenEar system. And how good those digital recordings can sound has a lot to do with our next article ...

Next Issue: Digital File Formats and Compression
