Lesson 3

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Introduction - Digital Audio Basics

1. The sampling rate is the number of measurements taken from an analog signal in one second. (in this lesson in this manual)

2. When we allow frequencies that are twice our sample rate to enter the sampling process, we get oversampling, or sounds that are actually harmonic distortion. (page 206)

3. Quantization is the way Digital Audio Workstation (D.A.W.) records the volume component of the digital recording or sampling process. (page 203)

4. Dither is applied to the process to reduce quantization errors and increase in noise and/or fuzziness that could creep into a bitstream to make it sound more natural. (Page 208-209)

5. List and explain in detail the Nyqist theorem. (page 205) Take at least twice as many samples as the highest frequency you wish to record. The computer can only accurately represent frequencies up to half the sampling rate (referred to as Nyquist frequency).

6. The basic theory of digital audio is processed, stored, and reproduced over time through the use of a binary number system. (page 199)

7. MP4 is the most common type of compression format for e-mailing audio. (page 377-378)

8. When we rip a CD to MP3, we reduce the size 10 percent. (page ~~26~~ 31, Recording Connection Workbook)

MANDATORY SUPPLEMENTAL READING

Lesson 3 – Sample Rate and Bit Depth.

Your own curiosity about audio engineering / producing should naturally lead you to researching additional topics on your own. To help you with this, here is a lesson to expand on a topic you partially covered.

To get the closest snapshot of the original sound wave, do you want your sample rate to be higher or lower, and why? 50 words.

You want your sample rate to be higher because according to the Nyquist Theory there has to be twice as many samples (but not to the point where it's oversampling). You don't want lower because that leads to it being "lossy" or distorted.

Do your best to explain Sample Rate and Bit Depth. Pull from your memory and what you can research. – 50 words for each definition.

Sample rate is the number of audible sounds captured per second. The more samples you have through the movement of time (not to exceed twice the original amount) the better the representation.

Bit depth is the word length for audio and the most common is 16-bit. Bit depth represents the binary number system which is composed of the numbers "0" and "1".

NOTES:

- Still photos for movies are usually 24 frames per second
- The job of a microphone is to transduce (convert one form of energy into another) the change of air pressure into an analogous change in electrical voltage
- At regularly spaced moments in time, the voltage at that instant is sampled and held constant until the next sample is taken; This reduces the total amount of info to a certain number of discrete voltages
- A device known as an Analog-to-Digital-Converter (ADC) receives the discrete voltages from the sample-and-hold device and assigns a numerical value to each amplitude;
- This process of converting voltages to numbers is known as quantization
- The digital-to-analog converter (DAC) converts each number to a voltage and communicates those voltages to an amplifier to increase the amplitude of the voltage
- The standard for compact disc recordings (and for "CD quality" computer audio) is to take 44,100 samples per second for each channel of audio
- Harold Nyquist - Nyquist theorem (twice as many samples as the highest frequency)
- Its essential to use the low-pass filter before the sample and hold process, to remove any frequencies above the Nyquist frequency
- An echo can be created by recalling samples that occured earlier and adding them to the current samples
- Digital signal processing (DSP) is concerned with the effects ~~of~~ digital filters -- formulae formulae for modifying digital signals by combinations of delay, multiplication, addition, and other numerical operations
- MP3 is a particular type of digital audio format that compresses music to fit within reasonably-sized computer files, while maintaining near-CD quality sound
- Lossy compression works by sampling the original file and removing those ranges of sounds that the average listener can't hear
- A lossless encoder uses complex algorithms to determine what sounds a human is able to hear, based on accepted psychoacoustic models, and chops off those sounds outside this range
- Compressed vs non-compressed formats