**Chapter 3 Notes**

**Lesson 3 – Digital Audio Basics (RCW)**

**DIGITAL REPRESENTATION OF SOUND**

SAMPLING RATE AND NYQUIST FREQUENCY

* We take many discrete samples of the sound waves instantaneous amplitude, store the info, then reproduce those amps at same rate to create illusion of continuous wave
* The job of a mic is to transduce change in air pressure into analog change in electrical voltage
  + Convert one form of energy to another
* SAMPLE AND HOLD:
  + Continuously changing voltage sampled periodically
  + Voltage at that instant is sampled and held constant until next sample is taken
    - This reduces total amount of info to a certain number of discrete voltages
* ANALOG-TO-DIGITAL CONVERTER (ADC)
  + Receives discrete voltages from sample and hold device and assigns a numerical value to each amp
* QUANTIZATION:
  + The process of converting voltages to numbers
  + Numbers expressed as binary digits (1 or 0)
  + Numbers stored on digital audio tape, hard disk etc
* MEMORY AND STORAGE
  + The more sampling rates and bit the more memory it will take up

**ADVANTAGES OF DIGITAL AUDIO**

SYNTHESIZING DIGITAL AUDIO

* any number could be a consideration as a digital representation for sound

MANIPULATING DIGITAL SIGNALS

* any sound in digital form is just a series of numbers
* multiplication is equivalent to audio amplications
* addition is equivalent to audio mixing
  + y=xn+Ayn-1000

EMAILING MUSIC FILES: WHAT IS LOST WHEN FILES ARE COMPRESSED

* MP3 – particular type of digital audio format that compresses music to fir within reasonable size while keeping near CD quality sound

HOW DIGITAL SAMPLING WORKS

* Software listens to music and takes digital snapshots of the music at particular points in time

LOSSY COMPRESSED FORMATS

* Sampling original file removing ranges of sounds that the average listener can’t hear
* Lossy files don’t sound as good as original
* You lose something in the translation
* Most popular lossy compress format is MP3
* Others include…
  + AAC – Advanced Audio Coding
  + OMA, OMG - ATRAC3
  + LAT, LQT, LSL – Liquid Audio
  + MP3
  + MP3PRO
  + OGG – OGG Vorbis
  + MOV – quick time audio
  + RA, RM, RMA – RealAudio Media
  + WMA – Window’s Media Audio

LOSSLESS COMPRESSED FORMATS

* Creates high fidelity digital archive
* Work more lip ZIP files compression
* Are not as small as lossy files
* Popular lossless formats include…
  + ALAC, M4A – Apple Lossless Audio Codec
  + FLAC – Free Lossless Codec
  + WMA – Windows Media Audi oLossless

NON-COMPRESSED FORMATS

* In order to archive your music in its original, unaltered form – use one of the following formats…
  + AU
  + AIF, AIFF – Audio Interchange Format
  + CDA – Compact Disc Digital Audio
  + SDII – For Pro Tools on MACs only
  + WAV - Waveform Sound Files

**Lesson 3 – Chapter 6; Digital Audio Technology (MRT)**

THE LANGUAGE OF DIGITAL

* Basic theory of digital audio is processed, stored and reproduced over time through the use of a Binary Number System
* Numeric system provides a fast and efficient means for manipulating and storing digital data
* Digital audio system works by measuring instant voltage level of analog signals at specific intervals - then converting samples into words that digitally rep analogous voltage level

DIGITAL BASICS

* Various stages that involved in encoding analog signals to digital and vice versa

SAMPLING

* Analog audio
  + Signals are recorded, stored and reproduced as changes in voltage levels that vary over time in a continuous fashion
* Digital audio
  + Taking periodic samples of an analog waveform over time
    - Then snapshots samples are calculated into binary words that digitally rep voltage levels as they change over time
* Incoming analog signal is sampled at discrete and precise time intervals
  + Analog signal is momentarily held while converter determines voltage level
* This process is continuously repeated throughout the recording process
* Analog signal is continuous in nature
* Digital signal makes use of periodic sampling to encode information
* SAMPLING RATE:
  + The number of measurements that are periodically taken over a course of a second
* Recordings made at a high sample rate will be capable of storing a wider range of frequencies
  + Increasing signals bandwidth at upper limit

QUANTIZATION

* Represent amplitude component of digital sampling process
* Used to translate voltage levels of continuous analog signal into binary digits to coincide digitally

THE NYQUIST THEOREM

* If frequencies more than twice the sample rate are let into the sampling process – these frequencies will have a shorter sample rate than what was actually captured
* To eliminate effect of ALIASING
  + A low pass filter is applied to the circuit before the sampling process takes place
* A slightly higher sample rate should be chosen to account for cutoff slop that’s required

OVERSAMPLING

* Increases effective sampling rate by factors ranging between 12 and 128 times the original
  + Nyquist Filters
    - Expensive and difficult to design
    - By increasing sample bandwidth – a simple inexpensive filter can be used
  + Oversampling
    - Results in higher quality analog-to-digital (A/D) and digital-to-analog (D/A) converters that sound better

SIGNAL-TO-ERROR RATIO

* Used to measure the quantization process
* Signal-To-Noise Ratio – ANALOG
* Signal-To-Error – DIGITAL

DITHER

* Relating to quantization
* Commonly used during recording process to increase overall bit resolution of a recorded signal when converting from higher to lower bit rate
* By adding a small amount of random noise into the A/D path you can…
  + Improve resolution of conversion process below and least significant bit level
  + Reduce harmonic distortion in a way that greatly improves signal’s performance
* Used to smooth and round data values

FIXED- VS. FLOATING-POINT PROCESSING

* Many new digital audio and DAW systems use floating-point arithmic in order to process, mix, and output digital audio

THE DIGITAL RECORDING/REPRODUCTION PROCESS

* The recording process
* The playback process
* Sound file sample rate
  + Sample rate of recorded digital audio bitstream relates to res at which will be digitally captured
* Sound file bit rate
  + Relates to number of quantization steps that are encoded into bitstream
* Regarding digital audio levels
  + The loudness war
* Digital audio transmission
  + AES/EBU
  + S/PDIF
  + SCMS
  + MADI
  + ADAT lightpipe
  + TDIF
* Signal distribution
  + Daisy chain the data from one device to the next in a straightforward way
    - Woks well if there are only a few devices chained together
  + Use a distribution device to connect all the devices together
* Jitter
  + Controversial and misunderstood phenomenon
  + Time base error
  + Caused by varying time delays in circuit paths from component to component
  + 2 common causes of jitter…
    - Poorly designed phased locked loops (PLLs)
    - Waveform distortion
      * Due to mismatched impedances/reflection in the signal path
* Wordclock
  + The need for locking (Synchronization) the sample clocks within a connected digital audio system to a common timing reference
  + The internal clocks of all the digital devices within a connected chain must be reference to a singer “master” wordclock timing element

DIGITAL AUDIO RECORDING SYSTEMS

SAMPLERS

* The drum machine was one of the first production applications in digital audio
  + The ability to trigger pre-recorded drum/percussion sounds from a single instrument
* From the drum machine came a major class of sample and synthesizer technology
* The sampler was born…
  + Devices were developed that were capable of recording, transposing, processing and reproducing segments of digitized audio directly from RAM

HARD-DISK RECORDING

* After the comp tech came into play – so did the hard disk where you could record, edit and playback audio

HARD-DISK MULTITRACK RECORDERS

* Mimics basic transport operation as traditional multitrack recorders

PORTABLE STUDIOS

* Include all of the required hardware and control system interface to record, edit mix down and playback a project anywhere you are

FLASH MEMORY HANDHELDS

* New class of portable recording device
* Include a set of built in high quality condensers mics

OLDER TECHNOLOGIES

* Minidisc
  + Viable medium for recording and storing CD, MP3 tracks and original recordings
* The rotating-head digital audio recorder
  + Backbone of wide range of media production systems
* The rotary head
* Digital audio tape (DAT) system
* DAT Tape/Transport Format
* The modular digital multi-track
* ADAT MDM format
  + Made use of standards S-VHS video taoe to record PCM audio t 44.1k/48k at 16/20- bit word depths
* DTRS MDM format
  + Capable of recording up to 108 minutes of digital audio onto standard 120 minute hi-8mm videotape

**Chapter 3 – Mandatory Supplement Reading**

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

In order for you to get the closest snapshot of the original sound wave, your sample rate should be higher. This is because the higher the sample rate, the closer it will be to its original analog waveform. Low sample rates limit the range of frequencies that can be recorded and results in an inaccurate representation of the original sound wave.

1. Do your best to explain Sample Rate and Bit Depth. Pull from your memory and what you can research.

*Sample Rate* – Sample rate, in my understanding, is crucial to the sound quality of a project. It is the number of digital snapshots taken of an audio signal each second and is measured in hertz or kilohertz. This rate helps the process of determining the frequency range of an audio file. It is taken from a continuous signal to make a discrete signal.

*Bit Depth* – Bit Depth determines the dynamic range of a sound wave. The higher the bit depth the better sound recording you will receive with more detail as opposed to having a lower bit depth where accuracy will be off and there will be a lot of sounds lost. Finally, the measuring unit used to measure bit depth is binary digits, bits.