Lesson 3
Chapter 6

Digital Audio Technology

<u>digital audio</u> - a process by which numeric representation of analog signals (in the form of V levels) are encoded, processed, stored and reproduced over time through the use of a binary number system
Ex: C A T (0100 0011)(0100 0001)(0101 0111) (1 = ON 0 = off)

<u>Sampling</u> - <u>analog audio</u> - signals are recorded, stored and reproduced as changes in voltage levels that continuously vary over time in continuous fashion.
<u>digital audio</u> - doesn't operate the same, it operates by taking periodic samples of an analog audio waveform over time, then calculating each of these snapshot samples into a grouped binary words that digitally represent these voltage levels as they change over time, as accurately as possible.

<u>Sample rate</u> - the number of measurements (samples) that are periodicly taken over the course of a second. It's reciprical (<u>sampling time</u>) is the elapsed time that occurs between each sampling period.
Ex. sample rate 44.1 kHz corresponds to a sample time of 1/44,100 of a second.

<u>Quantization</u> - represents the amplitude component of the digital sampling process. It's ~~sample~~ translate the V levels of a continuous analog signal (at discrete sample points over time) into binary digits (bits) for the purpose of manipulating or ~~and~~ storing audio data in the digital domain.

- Sampling (in the ~~truest~~ sense of the word) an analog V signal at precise intervals in time.
- Converting these samples into a digital word value that most accurately represents these V levels.
- Storing them with in a digital memory device.

The Nyquist Therom - in order for the desired freq bandwith to be faithfully recorded in the digital domain, the selected sample rate must be at ~~that~~ least twice as high as the highest freq to be recorded (sample rate ≥ 2 x highest freq).

~~Should freq that an~~

Alias or Aliasing - freq that are greater than twice the sample rate be allowed into the sampling process, these high freq would have a periodic nature that's shorter than the sample rate can actually capture.
To eleminate the effects of aliasing, a low-pass filter is placed into the circuit before the sampling process takes place.

Oversampling - increases the effective sample rate by factors raging between 12 + 128 times the original rate.

Signal-to-error ratio - is used to measure the quantization process. A digital system's STOR is closely akin (although not identical) to the analog ~~signal~~ concept of signal-to-noise (S/N) ratio. Wheras a S/N ratio is used to indicat the overall dynamic range of an analog system, the (S/E) ratio of a digital audio device indicates the degree of accuracy that's used to capture a sampled ~~level~~ and its step-related effects.

Dither - is commonly used during the recording or conversion process to increased the overall bit resolution (+ therefore low-level noise + signal clarity) of a recorded signal, when converting from a higher to lower bit rate. Dither is the addition of very small amounts of randomly generated noise to an existing bitstream that allows the S/N and distortion figures to fall to levels that approach that their approach their theorectial limits

Recording Process

Digital recording base included a low pass filter, a sample-and-hold circuit, an analog-to-digital converter, circuitry for signal coding and error correction. Following the low-pass filter, a sample+hold (S/H) circuit holds + measures the analog Vo level that's present during the sample period. This period is determined by the sample rate ((1/44,100th of a second for a 44.1 rate).

The most common form of digital audio data coding is pulse-code manipulation, or PCM.

Sound file sample rates of a recorded digital audio bitstream directly relates to the resolution at which a recorded sound will be digitaly captured

32k - broadcasting - w/ high-quality converter
44.1k - standard in consumer + pro audio production, CD
48k - early standard sample rate - VIDEO + DVD Production
88.2k - multiple of 44.1 - High resolution recording
192K - high resolution + uncommon in pro audio productions + high storage + media requirements.

Sound file bite rates of digitally recorded sound file directly relates to the number of quantization steps that are encoded into the bit stream

16 bits - consumer + pro audio (CD) standard - minimum depth for high-quality pro audio
20 bits - standard for high-bit-depth resolution - uncommon
24 bits - standard for high definition audio application w/ 96k sample rate.

Regaurding digital audio levels - average peak levels above full scale can easily ruin a recording. As such digital has a wider dynamic range than analog, it's always good idea to ~~[illegible]~~ reduced your levels so that they peak from -12 to -20 dB. This will accurately capture the peaks without clipping, without introducing an appreciable amount of noise into the mix.

✱ Using a Y-Cord to split a digital signal between 2 devices MAJOR NO-NO ✱

Digital Audio Transmition

- AES/EBU (AUDIO ENGINEERING SOCIETY AND THE EUROPEAN BROADCAST GROUP)
- S/PDIF (Sony/Phillips Digital Interface)
- SCMS (Serial Copy Managment System)
- MADI (Multichannel Audio Digital Interface)
- ADAT LIGHTPIPE (Alesis lightpipe system)
- TDIF (TASCAM DIGITAL INTERFACE)

Jitter - is time-base error. Caused by varying time delays in the circuit paths from component to component in the signal path - the most common causes of jitter are poorly designed Phase Locked Loops (PLLs) and waveform distortion due to mismatched impedances and/or reflections in the signal path.

Wordlock - through the use of a single, master timing reference known as word lock, the overall sample-and-hold conversions states during both the record and playback process for all digital audio channels and devices within the system will occurr at exactly the same point in time.

Superclock (256Fs) is transmitted over the same type and cable as wordlock and was eventually abandoned in favor of the industry standard wordlock protocol.

Digital Audio Recording Systems

Samplers - capable of recording, musically transposing, processing and reproducing segments of digitized audio directly from RAM

HARD-DISK RECORDING

- ability to handle multible samplefiles
- random-acess editing
- Nondestructive editing
- DSP - Digital signal processing

HARD DISK MULTITRACK RECORDERS - mimics the basic transport, operational and remote controls of a traditional multitrack recorder

Portable Studio - all-in-one system include all of the required hardware and the control system's interface to record, edit, mixdown and playback a project virtually anywhere when using an AC adapter or batteries.

Flash memory handhelds - recording to a solid-state flash memory, handhelds often included a set of built-in high-quality electret-condenser mics and offer many of the recording and overdub features that you might expect from a larger portable recording system, such as a USB link (allowing for easy data trasfer or for directly turning the device into a USB interface with built-in mics), built-in effects, microphone emulation (allowing the mic's character to be changed to conform to a number of mic styles), a guitar tuner, etc. Some handhelds allow for pro mic's to be pluged in.

Older Tech

- Sony Mini Disk
- Rotating-Head Digital Audio recorder
- Rotary Head
- Digital Audio Tape (DAT) SYSTEM
- DAT TAPE/TRANSPORT FORMAT
- MODULAR DIGITAL MULTITRACK
- ADAT MDM FORMAT
- DTRS MDM FORMAT