Lesson 5 Chapter 11

Timecode Refreshment / JAM SYNC

Timecode Refreshment - this process reads the degraded timecode information from a previously recorded track and then amplifies and regenerates the square wave back into its original shape so it can be freshly recorded to a new track or read by another device

Jam Sync - refers to the synchronizer's ability to output the next timecode value, even though one has not appeared at its input. Two forms of Jam Sync: Freewheeling, Continuous

Synchronization using SMPTE timecode

Synchronizer - is to control one or more tape, computer-based or film transports (designed as slave machines) so their speeds and relative positions are made to accurately follow one specific transport (designated as the master)

SMPTE OFFSET TIMES - set time code not 00:00:00:00

Distribution of SMPTE SIGNALS - In a basic audio production system, the only connection that's usually required between the master machine and a synchronizer is the LTC timecode track. Generally, when connecting analog slave devices, two connections will need to be made between each transport and the syncronizer.

Timecode levels - One problem that can plague systems using timecode is crosstalk. This happens when a high-level signal leaks into adjacent signal paths or analog tape tracks.

MIDI-based SYNCHRONIZATION

MIDI REAL-TIME messages - one MIDI device must be designated as the master device in order to provide the timing information to which all other slaved devices are locked.

- Timing Clock - a clock timing that's transmitted to all devices in the MIDI system at a rate 24 pulses per quater note (ppq).
- START - Upon receipt of a timing clock message, the start command instruction all connected devices to begin playing from the begining of their internal sequences.
- Stop - Upon the transmission of a MIDI stop command, all devices in the system stop at their current positions and wait for a message to follow.
- Continue - following the receipt of a MIDI stop command, a MIDI continue message instructs all instruments and devices to ~~resuming~~ playing from ~~the~~ precise point at which the ~~sequence~~ was stopped.

MIDI Time code

MIDI Timecode - was developed to allow electronic musicians, project studios, video facilities and virtually all other production environments to cost effective and easily translate timecode into timestamped messages that can be transmitted via MIDI.

MIDI Time code control structure

- Timecode
- MIDI CUEING

MIDI Time code structure

- Quarter-frame messages
- Full message
- MIDI Cueing messages

SMPTE/MTC CONVERSION - a converter is used to read incoming SMPTE time code and convert into MIDI timecode (and vice versa)

Master / slave relationship - since sync is based on the timing relationship between two or more devices, it follows that the logical way to achieve sync is to have one or more devices (slaves) follow the relative movements of a single transport or device (master).

- Audio recorders
- VCR's
- Software application
- Digital workstation
- DAW support for video + picture sync
- Routing timecode to and from your computer.

Keeping Out of Trouble

Guidlines

- Familiarize yourself with hardware + software BEFORE the session start.
- When in doubt about frame rates, special requirement or anything else, for the matter - ASK! the client
- Fully document timecode setting, offset, start times, etc.
- If the project is not in-house, ASK the producer what the proper frame rate should be. DONT ASSUME ANYTHING!
- When begining a new session, always stripe the master contiguously from beinging to end before the session begins. STRIPE AN extra tape.
- Start generating new code at a point before midnight (Ø1:ØØ:ØØ:ØØ ØØ:Ø1:ØØ:ØØ ALLOW FOR PREROLL) If not in-house ASK the producer what the start times should be. DONT ASSUME!
- Never dub (copy) timecode directly. ALWAYS make a refreshed (JAM SYNC) copy of the original timecode (from analog master) before session.
- Disable noise reduction on analog audio tracks (audio + VIDEO)
- Never use slow videotape speeds. EP mode on VHS is too slow.
- Work with copies of the original production video, and make a new one when sync troubles appear.
- When dropouts occur on the track, switch to freewheel once transport have locked.

Lesson 5 Chapter 12

<u>Amplifier saturation</u> results when the input signal is so large that its DC output supply isn't large enough to produce the required, corresponding output signal. Over driving an amp in such a way will cause a mild to severe waveform -distortion effect known as <u>clipping</u>.

The best way to avoid undesirable distortion is to be aware of the various amps and device gain stages throughout the studio's signal chains.

<u>Operational Amp</u> (op-amp) is a stable, high gain, high bandwith amp that has a high-input impedance and a low-output impedance.

<u>Negative feedback</u> - is a technique that applies a portion of the out-put signal through a limiting resistor back into the negative or phase-inverted input terminal. By feeding a portion of the amps output back into the input out of phase, the device's out-put signal level is reduced. This has the effect of controlling the gain (by varying the negative resistor value) in a way that also serves to stabilize the amp and further reduce distortion.

<u>Preamplifiers</u> - this amp type is often used in a wide range of applications, such as boosting a mic's signal to line level, providing variable gain for various signal types and isolating input signals and equalization, just to name a few. Preamps are an important component in audio engineering because they often set the "Tone" of how a device or system will sound.

Equalizers - is nothing more than a freq-discriminating amplifier. In most analog designs, EQ is achieved through the use of resistor/capacitor networks that are located in an op-amps negative feedback loops in order to boost or cut (attenuate) certain freq in the audible spectrum.

Summing Amp - (combining amp) is designed to combine any number of discrete input into a single output signal bus, while providing a high degree of isolation between them. The summing amp is an important component in analog console/mixer design because the large number of iternal signal paths require a high-degree of isolation in order to prevent signals from inadvertenly leaking into other audio path.

Distrabution Amp - isn't used to provide gain but instead will amplify the signals' current (power) that's being delivered to one or more loads.

Power Amp - are used to boost the audio output to a level that can drive one or more loudspeakers at their rated volume levels. Although these are often reliable devices, power amp designs have their own special set of problems. These include the fact that transistor don't like to work at the high temperatures that can be generated during continuous, high-level operation. Such temps can also result in changes in the unit's response and distortion characteristics _ or outright falure. This often requiers that protective measures (such as fuse or thermal protection. under a wide range of circuit conditions (such as load shorts, mismatched loads and even open, "no-load circuit) and are usually designed to work with speaker impedance loads raging between 4 + 16 ohms (most models 8 ohms)

Lesson 5 Chapter 13

Audio Production console or mixer - its basic purpose is to give us full control over volume, tone, blending, and spatical paristoning for any or all signals that are applied to its inputs from mic's, electronic instruments, effects devices, recording systems, + other audio devices

Recording Process
- Recording (BASICS)
- Overdubbing
- Mixdown

Recording - phase involves the physical process of capturing live or sequenced instruments onto a recorded medium.
- All instruments recorded in one live pass
- musicans lay down basic tracks (foundation) usually rythem.
- Vocals + other instruments can be add during overdub phase.

Basics (rythem, bed) - consist of instruments that provide the rythmic foundations of a song and often include drums, bass, rythem guitar + keyboards. Optional Vocal guide (SCRATCH) can also be recorded to help musicans + vocalist to properly capture the proper tempo + the All-important feel of the song.

Grouping - When recording an instrument with multible mics this grouping can be recorded down to one track or a stereo pair. By ~~[illegible]~~ bussing the various channel over to ~~input~~ the same output bus.

General Rule : far more difficult to make changes to a recorded group of tracks, because a change to one instrument will almost directly effect the overall group mix.

The name of the game is to capture the best performance with the highest quality, and at optimum signal levels (without regard to level balances on other tracks.

The rhythm tracks are often the driving backbone of a popular song, and recording them improperly can definitly get the project off to a bad start.

Monitoring - a separate mix is often set up in order for the artist, producer + engineer to hear the instruments in their proper musical perspective.

Overdubbing - recording instuments that are not present during the origial performance that can be added at a later time on to an existing multitrack project.

Mixdown - Once all the musical parts have been performed ~~and~~ recorded to everyone's satisfaction the mixdown can begin. In mixdown the audio is repeated played while adjustments in level, panning, EQ, effects, etc. are made for each track and/or grouping Throughout this artistic process, the individually recorded signals are blended into a composite stereo, surround or mono signal that's fed to the master mixdown recorder (or software). When a number of mixes have been made a single version has been approved, this is the master or final mix can be mastered to its intended medium and/or assembled into a final product.

the mixing surface -
- input section
- through a send section
- into an EQ
- Passing through a monitor mix
- to an output fader
- into a routing section

①. Channel input. - I/O module.
- insert point - mono or sterio

②. Aux send. - ability to send effects, monitor, headphone mix

③. EQ
- to correct for specific problems
- to overcome deficiences in freq response
- to allow contrasting sounds to better blend
- to alter sound purely for musical or creative reasons.

④ Dynamics

⑤. Monitor
- IN-Line
- Separate
- Direct insertion

⑥. Channel fader
- solo
- mute

⑦. Output

⑧. Channel assignment

⑨. Grouping

⑩. Monitor level

⑪. Patch bay
- Open
- Half-normaled
- Normaled
- Paralled

⑫. Metering

Power + Ground related issues

- Keep all studio electronics on the same AC electrical circuit
- Keep audio wiring away from AC wiring
- If you only hear hum comming from a particular input channel check for grounding issues
- Dis connect all the devices - then methodically plug them back in one by one (listen through headphones)
- Check cables for bad connections or improper polarity
- Possible ground loops on the rack mount
- Investigate the use of balanced power source.

Balance power - circuit can reduce line noise of all systems plugged in. It can not eliminate noise the gear is making.

Power Conditioning -

- Voltage regulation
- Eliminating power interruptions
- keeping the lines quiet

Mixing + Balancing Basics

- relative level
- Spatial Positioning
- EQ
- Dynamic processing
- effect's processing