Synchronization and Timecode

Synchronization

- Synchronization causes two or more devices to operate at the same time and speed.
- In audio, the term means an interface between two or more independent, normally free running machines -two multi tracks, or an audio tape recorder and a video player

Classification of synchronization

- For two machine to be synchronized there must be a means of determining both their movements and adjusting them so that two operate at the same time.
- This means that each machine must provide a sync signal that reflects its movement. The sync signals from both machine must be compatible, each must show the machine's movement in the same way.

Feed back control systems (closed-loop)

- One machine is taken to be the master, and slave is adjusted in reference to this master.
- A separate synchronizer compares the slave’s sync signal to the master’s sync signal and generates an error signal, which drives the slave’s motor to follow the master.
- As long as the sync signals are compatible and readable, feedback control will bring the two machines into lock.

Non feedback control (open-loop control)

- This simpler method is used when slave machine can be driven directly by the master’s time code.

TIMECODE:

SMPTE/EBU Timecode

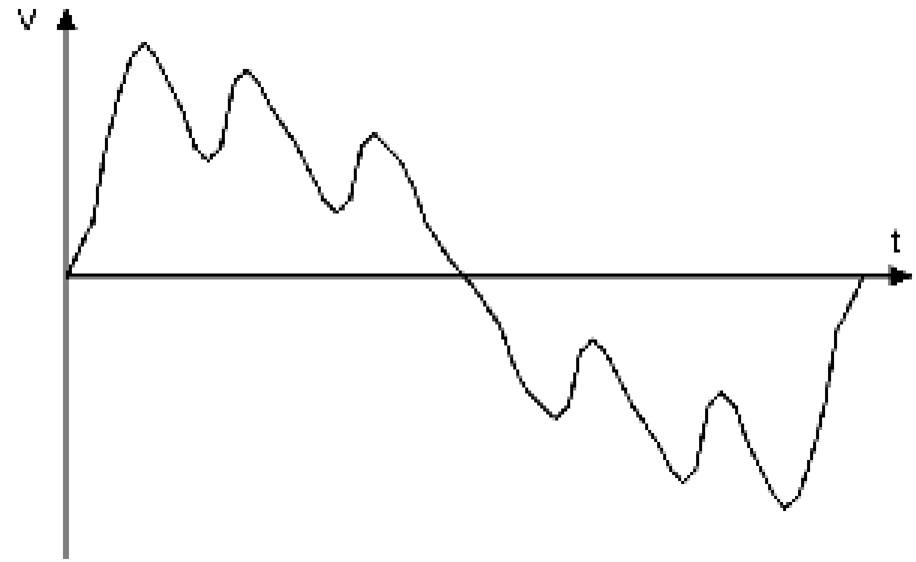
- Society of Motion Picture Television Engineers/European Broadcast Union.
- SMPTE time code is a synchronization standard adopted in the united states in the early 1960’s for video editing.
- 00:00:00:00,hour:minutes:second:frames

Nagra Recorder

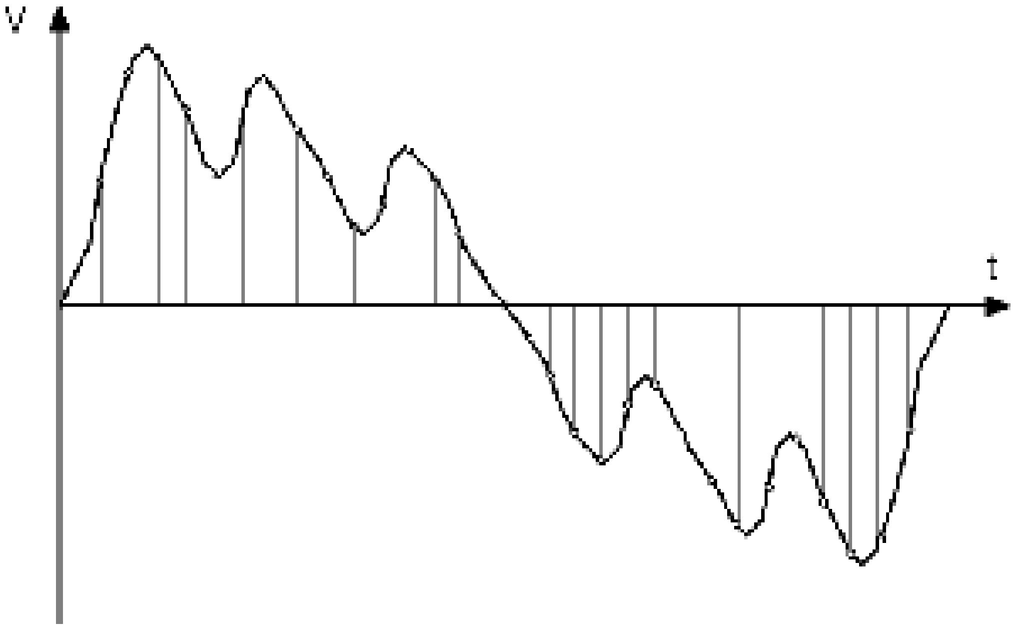
Nagra Digital recorder

How Analog-to-Digital Converter (ADC) Works

- consider the analog signal found on Figure. Let’s assume that it is an audio signal,

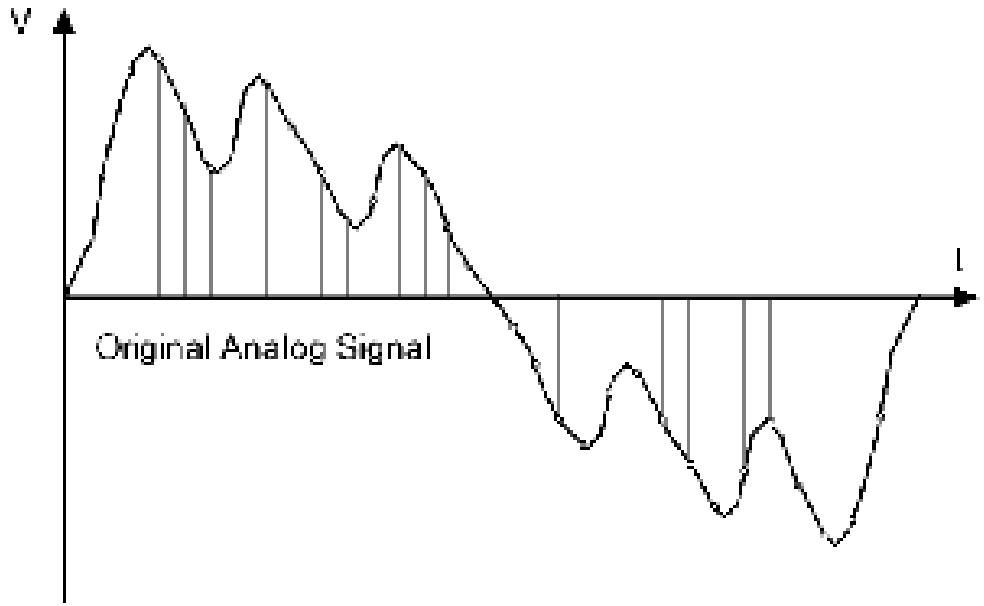


- What the ADC circuit does is to take samples from the analog signal from time to time. Each sample will be converted into a number, based on its voltage level.



- The frequency on which the sampling will occur is called sampling rate. If a sampling rate of 22,050 Hz is used, for example, this means that in one second 22,050 points will be sampled. Thus, the distance of each sampling point will be of 1 / 22,050 second (45.35 μs, in this case). If a sampling rate of 44,100 Hz is used, it means that 44,100 points will be captured per second. In this case the distance of each point will be of 1 / 44,100 second or 22.675 μs. And so on.

- During the digital-to-analog conversion, the numbers will be converted again into voltages. The waveform resulted from the digital-to-analog conversion won’t be perfect, as it won’t have all the points from the original analog signal, just some of them. In other words, the digital-to-analog converter will connect all the points captured by the analog-to-digital converter, any values that existed originally between these points will be suppressed.


V
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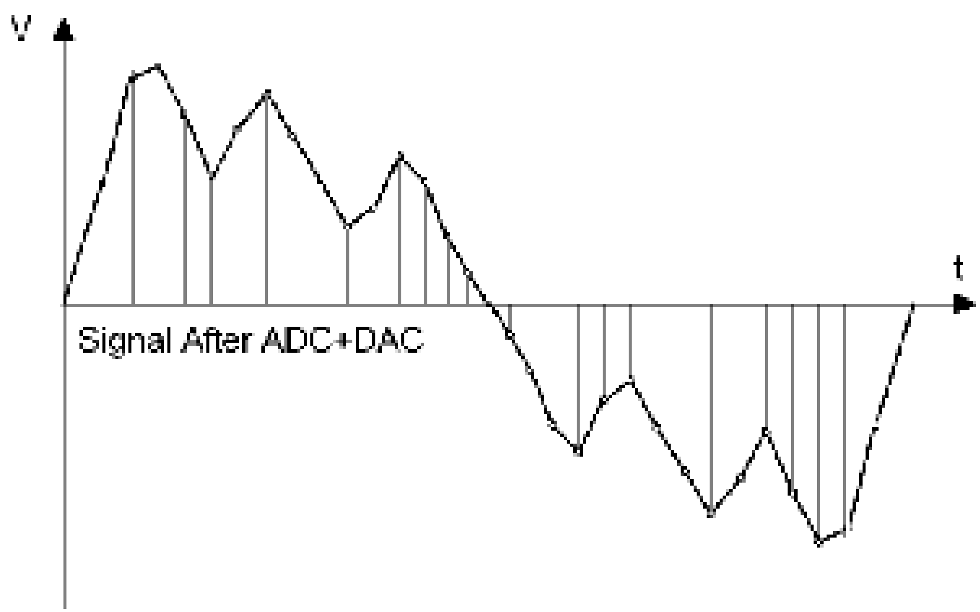
Original Analog Signal


V
t
Signal After ADC+DAC

- The more sampling points we use – i.e. the higher the sampling rate –, the more perfect will be the analog signal produced by the digital-to-analog converter (DAC). However, the more samples we capture more storage space is necessary to store the resulting digital data. For example, an analog-to-digital conversion using a 44,100 Hz sampling rate will generate twice the number of data as a conversion using a 22,050 Hz sampling rate, as it will capture twice the samples from the original waveform.

Nyquist Theorem.

- This theorem states that the sampling rate on analog-to-digital conversions must be at least two times the value of the highest frequency you want to capture.