Chapter 3: Pulse Code Modulation (PCM)

3.1 PCM transmission format (T1 and E1 format)

3.2 Frame and Multiform

3.3 Frame and multi frame alignment strategy

3.4 Higher order PCM

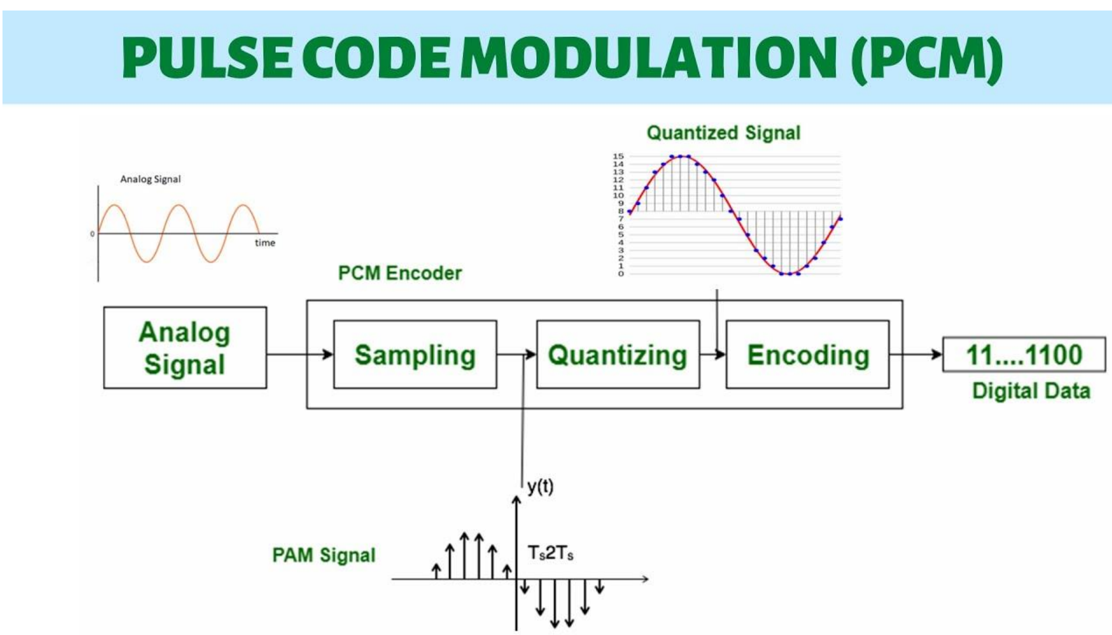
3.5 Plesiochronous Digital Hierarchy (PDH), Synchronous Digital Hierarchy (SDH) and SONET

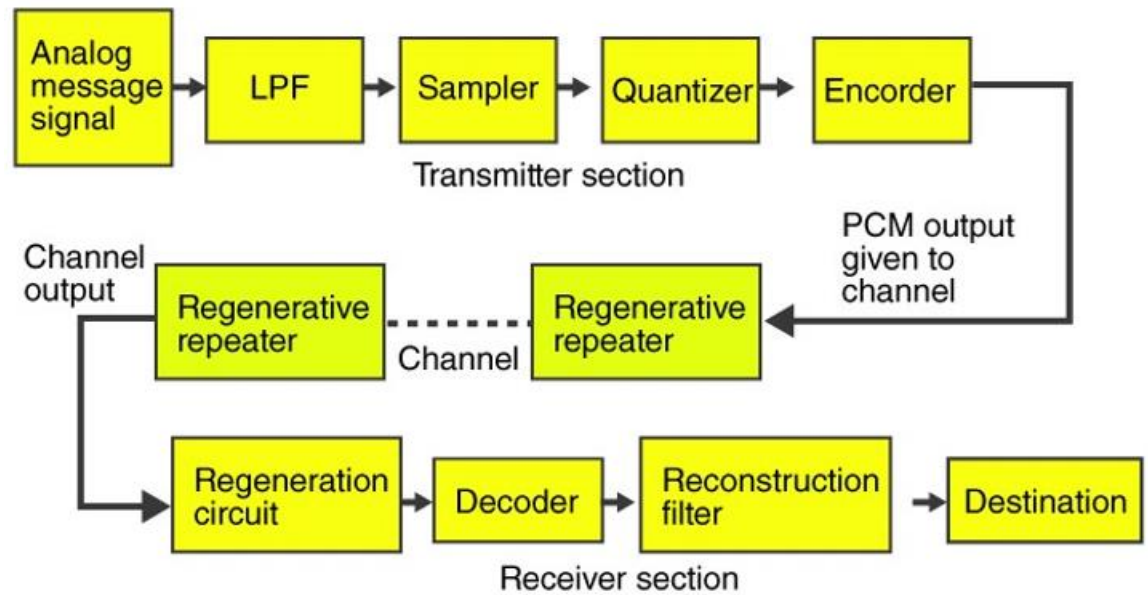
PCM:

PCM is a technique used for remodeling analog signal into digital signal. It ensures high noise immunity and compatibility with digital systems. It is widely used in telecommunication, digital audio, and video systems. PCM formats like T1 and E1 enable efficient time-division multiplexing for voice and data transmission.

The Pulse Code Modulation process is done through the following steps:

- Sampling
- Quantization
- Coding





Low Pass Filter: It helps in removing the high-frequency components included in the input of the analog signal and avoid aliasing (a continuous signal is sampled at too low rate that is) of the message signal.

Sampler: It helps to collect the sample data at any time of the message signal, in order to reform the original signal. As per the sampling theorem, the sampling rate is greater than the highest frequency component of the message signal. ($f_s \geq 2f_m$)

Quantizer: It helps to minimize the error through the process known as quantizing by reducing the unnecessary bits and also helps in compressing the obtained values.

Encoder: It helps to allot each quantized level through a binary code. The sample-and-hold process is adopted in this. Low pass filter, sampler, and quantizer aids to convert analog to digital forms. Encoding also aids in minimizing the usage of bandwidth.

Regenerative Repeater: It is used to compensate for the signal loss and also reform the signal. It also helps to increase signal strength

Decoder: It forms the original signal by decoding the pulse coded waveform. Decoder acts as the demodulator.

Reconstruction Filter: The reconstruction filter helps to obtain the original signal

Advantages of Pulse Code Modulation (PCM)

High Noise Immunity & Compatibility: PCM signals are less affected by noise and interference, making them ideal for long-distance transmission and compatible with modern digital systems.

Reliability & Quality: Error detection/correction techniques ensure reliable transmission, and PCM delivers superior audio and video quality for applications like CDs and communication systems.

Disadvantages of Pulse Code Modulation (PCM)

High Bandwidth & Complexity: PCM requires a large bandwidth and involves complex encoding/decoding processes compared to analog systems.

Quantization Error & Power Consumption: Quantization introduces errors, and the high processing demands lead to increased power consumption, limiting suitability for low-power applications.

Pulse code modulations are of two types:

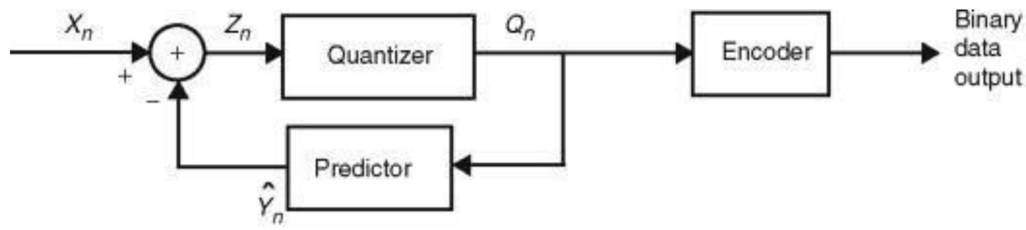
- Differential pulse code modulation (DPCM)
- Adaptive differential pulse code modulation (ADPCM)

1) Differential Pulse Code Modulation DPCM

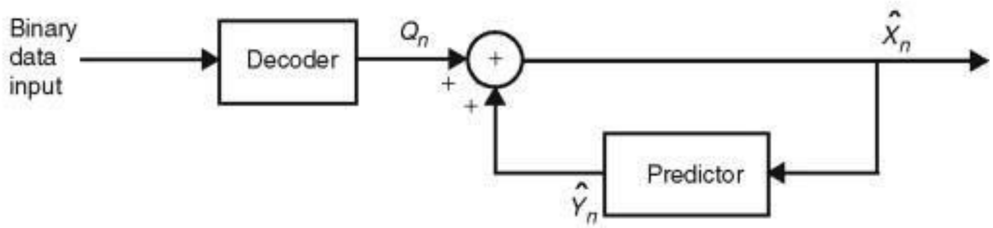
It is a technique that exploits the redundancy in a signal by transmitting the difference between consecutive samples, rather than the absolute value of each sample. This reduces the required bitrate for transmission. It adds functionalities based on the prediction of the samples of the signal.

How it works:

- **Prediction:** The current sample value is predicted based on previous samples
- **Differencing:** The difference between the actual sample value and the predicted value is calculated
- **Quantization:** The difference is quantized to a finite number of levels
- **Encoding:** The quantized difference is encoded into a binary code



DPCM encoder



DPCM decoder

- If you take samples of an analog signal faster than the Nyquist rate (which is the minimum rate to avoid loss of information), the samples will look very similar to each other
- The values (amplitudes) of one sample and the next are almost the same, with only small differences
- When you process (quantize and encode) these similar samples, a lot of the information ends up being repetitive or redundant, making the transmission less efficient
- In DPCM, instead of sending every sample, only the difference between one sample and the next is sent. This reduces redundant data and compresses the signal for more efficient transmission

How DPCM Works (Simplified):

Sampling: The analog signal is sampled at regular intervals to create a series of digital values.

Example: A signal might have sampled values like 12, 14, 15, and 16.

Prediction: A predicted value is generated for the next sample based on past samples. Usually, the simplest prediction assumes the next sample will be the same as the current one.

For instance, if the current sample is 12, we predict the next sample will also be 12.

Difference Calculation: The difference (or error) between the actual sample value and the predicted value is calculated.

Example:

Actual sample: 14

Predicted sample: 12

Difference: 14 - 12 = +2

Quantization: The difference is quantized. This means we round the difference to the nearest value from a fixed set of levels (to simplify storage and transmission).

For example, if the difference is +2 and our quantization levels are {+3, +2, +1, 0, -1, -2, -3}, it stays as +2. If it were +2.3, it would be rounded to +2.

Encoding and Transmission: Only the quantized difference is transmitted, not the original sample value.

Reconstruction at Receiver: At the receiver, the original signal is reconstructed by adding the received difference to the predicted value.

Example: If the predicted value is 12 and the received difference is +2, the reconstructed sample is 12 + 2 = 14.

For Example:

Our actual differences (before quantization) were:

+2.3, +1.7, +1.4

➔ Step Size = 1:

Quantization levels: {-3, -2, -1, 0, +1, +2, +3}

Quantized differences:

+2.3 → +2 (rounded to the nearest level)

+1.7 → +2 (rounded to the nearest level)

+1.4 → +1 (rounded to the nearest level)

Here, smaller step size results in less error because the levels are close together.

➔ Step Size = 2:

Quantization levels: {-6, -4, -2, 0, +2, +4, +6}

Quantized differences:

+2.3 → +2 (still fairly accurate)

+1.7 → +2 (less accurate)

+1.4 → +2 (rounding causes a larger error)

A larger step size reduces the precision, especially for small differences.

Advantages of DPCM

- Better Compression
- Improved Signal Quality

Disadvantages of DPCM

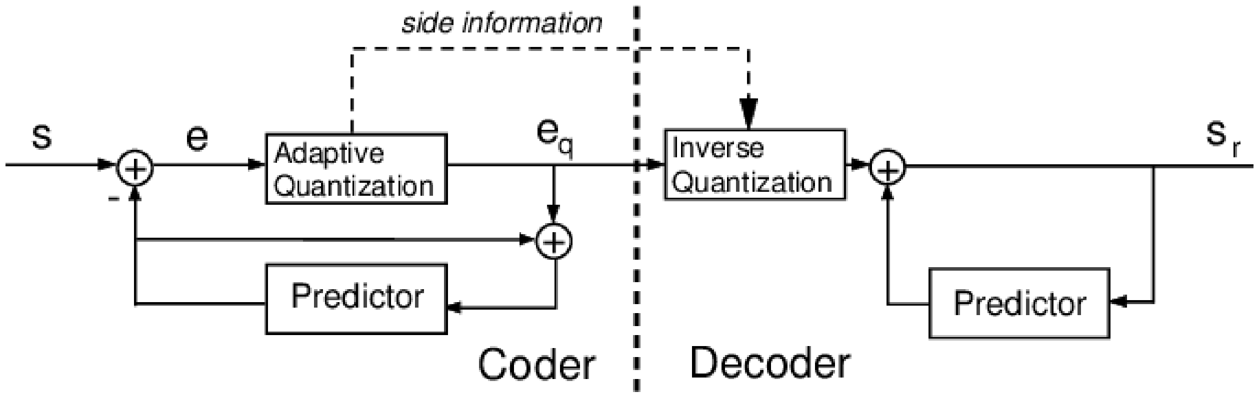
- Complexity: Requires sophisticated hardware or software due to the more complex implementation
- Prediction Errors: Inaccurate prediction models can introduce errors into the encoded signal

2) Adaptive Differential Pulse Code Modulation (ADPCM)

It is an extension of DPCM that adapts to the characteristics of the input signal. It dynamically adjusts the quantization step size to match the signal's dynamic range, further improving compression efficiency. It is a technique in which the size of the quantization step is varied, to allow the further reduction of the required data bandwidth to a given signal-to-noise ratio.

How it works:

- **Adaptive Quantization:** The quantizer adjusts its step size based on the statistical properties of the input signal.
- **Adaptive Prediction:** The predictor adapts its coefficients to better predict the current sample value



Higher-Order Pulse Code Modulation

Higher-order PCM refers to PCM systems that utilize higher levels of sophistication in terms of bit resolution, sampling rate, and techniques to improve fidelity, reduce error, or optimize the efficiency of digital communication systems.

Delta Modulation (DM)

- It is an analog to digital and digital to analog signal conversion technique.
- It uses one bit PCM code to realize digital transmission of analog signal.
- With delta modulation, instead of transmit a coded illustration of a sample solely one bit is transmitted, that merely indicates whether or not the sample is larger or smaller than the previous sample.
- If signal is large, the next bit in digital data is 1 otherwise 0.
- This helps to reduce bandwidth (BW = N f_s, as N=1, BW = f_s).
- We are *not transmitting the actual sampled value*, we are sending the information whether present sample is higher than or lower than the approximated value

https://www.youtube.com/watch?v=sHRgdFAt-nQ

PDH- Plesiochronous Digital Hierarchy

Plesio = Near   Chronous = Time

- Different Stages are almost synchronized and was very famous multiplexing technology in early age
- ***Although they have the same nominal bit rate, they commonly originate from different crystal oscillators and can vary within the clock tolerance***
- The basic data rate is 2.048Mbps for European standard E1and for North American standard T1 is 1.544Mbps

PCM

Pulse code modulation is the basic block for PDH SDH and SONET. Sampling frequency must be twice the higher frequency component of the signal.

Voice frequency sampling rate 8 KHz (0.3 to 3.4 KHz = 4 KHz).

Each sample encoded into 8 bits (256 level quantization).

Bit rate = 8000 * 8 = 64 Kbps.

PDH Hierarchy for E1 System:

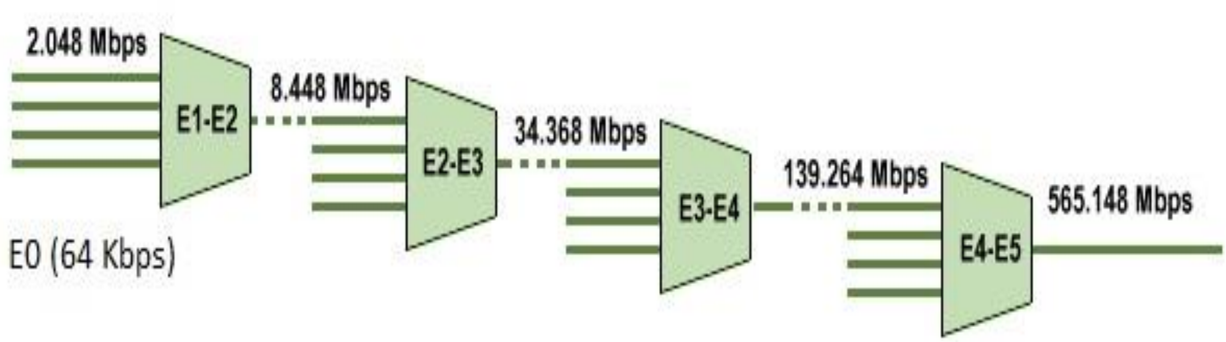
•30 Channel PCM = 2 Mbps 30 Voice channel

•2 Mbps x 4 = 8 Mbps 120 Voice channel

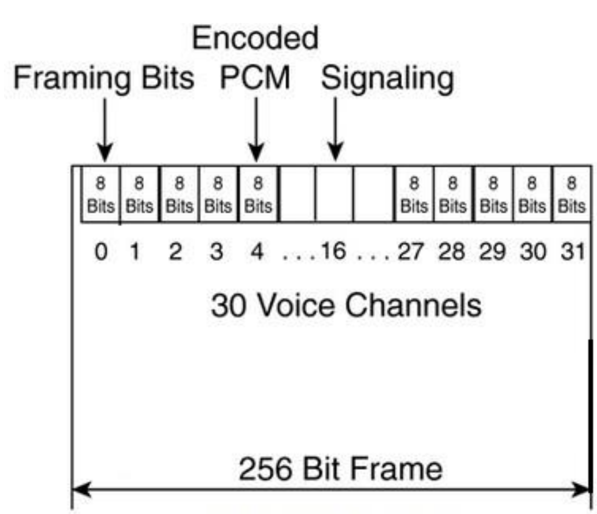
•8 Mbps x 4 = 34 Mbps 480 Voice channel

•34 Mbps x 4 = 140 Mbps 1920 Voice channel

•140 Mbps x 4 = 565 Mbps 7680 Voice channel



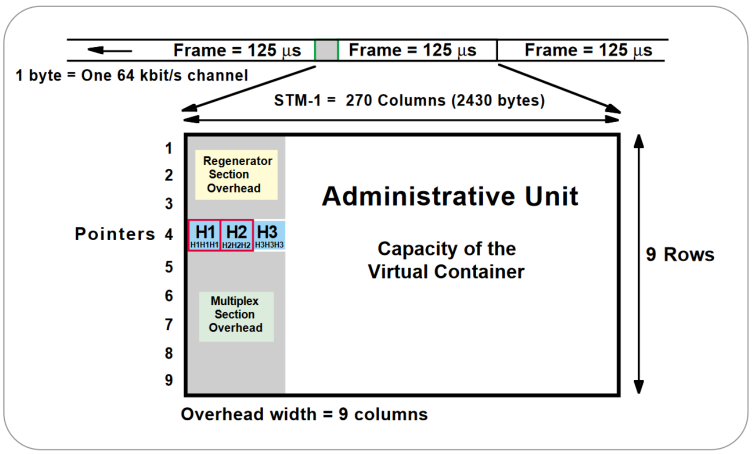
PDH Level Hierarchy (Europe)



E1- Frame Format

PDH disadvantages

- Every apparatus has its own clock
- Input tributary clock rate is slow than multiplexing equipment
- No network level synchronization and bit stuffing is needed
- No monitoring overhead and No management overhead
- No common standard among vendors



Frame = 125 µs
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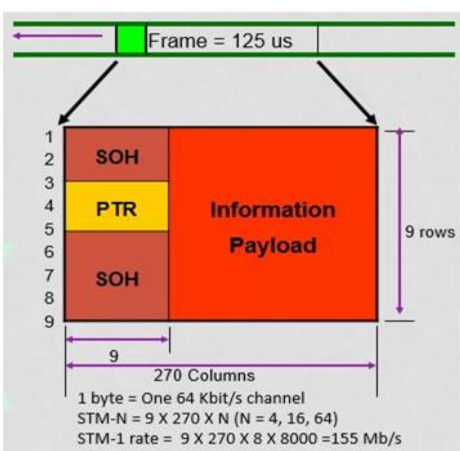
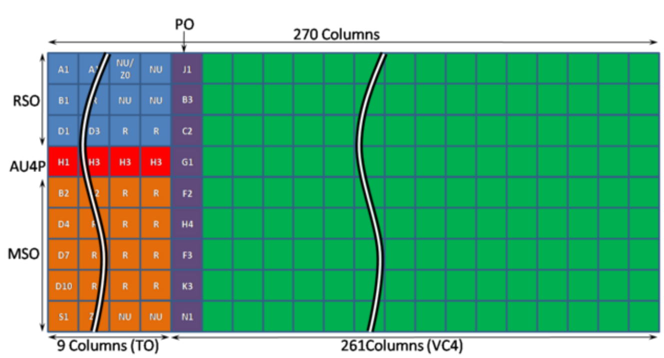
1 byte = One 64 kbit/s channel
STM-1 = 270 Columns (2430 bytes)
1
2
3
Pointers 4
5
6
7
8
9
Regenerator Section Overhead
H1
H2
H3
Multiplex Section Overhead
Administrative Unit
Capacity of the Virtual Container
9 Rows
Overhead width = 9 columns

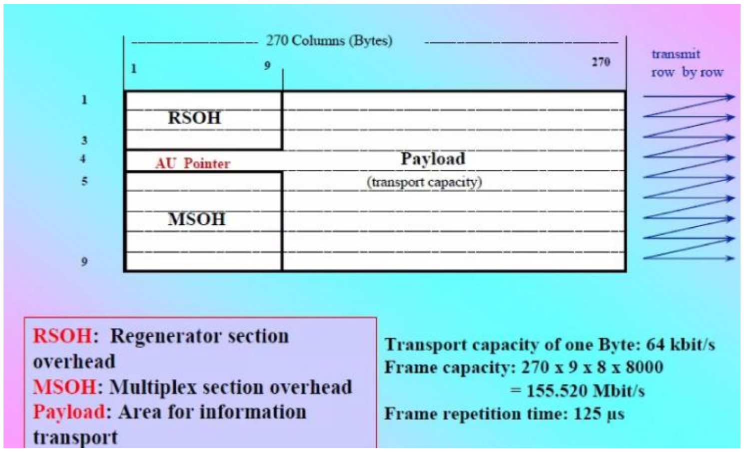
SDH Synchronous Digital Hierarchy

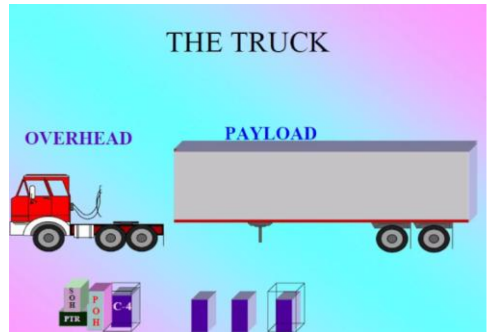
This method is used in optical fiber transmission system called SDH system.

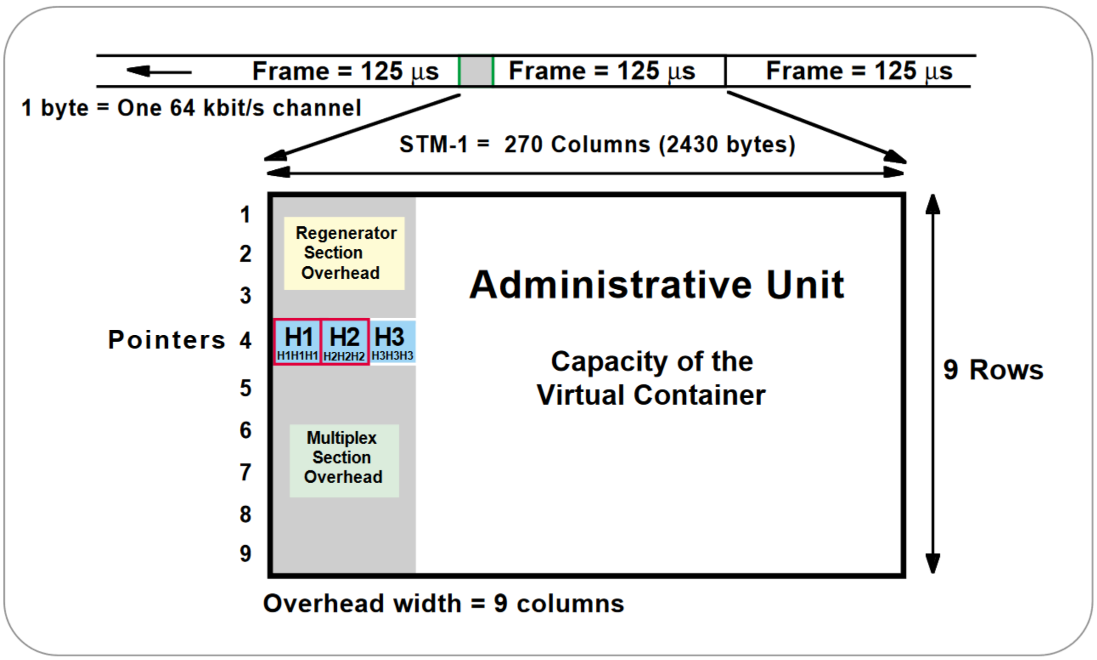
- Synchronous: All systems clocks are synchronised with a master clock. All clocks may be out of phase but run at the same frequency.
- Plesiochronous: All system clocks run at the same frequency with a defined precision.
- Asynchronous: System clocks are not synchronised. Tx and Rx have an independent clock.











SDH properties

- ☞ Powerful Management (Overhead Structure to handle with multiple signal levels and performing management tasks)
- ☞ Standard Optical Interfaces and Standard Digital Format
- ☞ Easy Cross connection

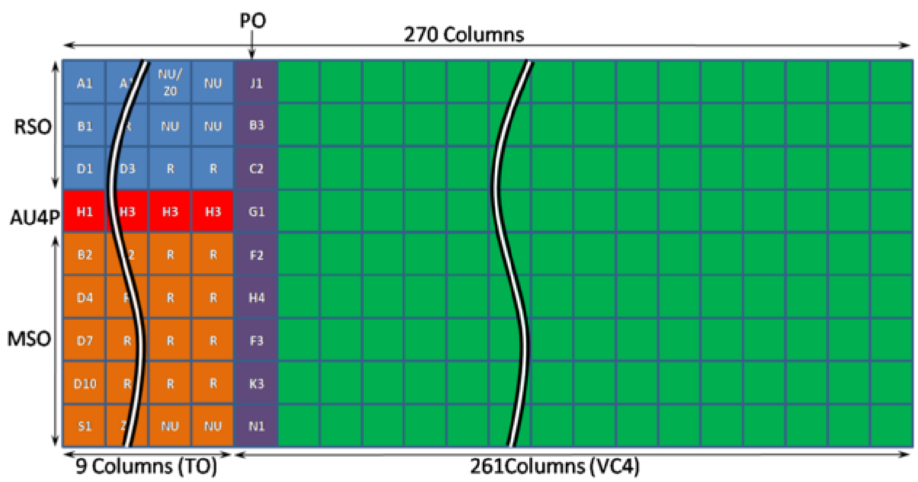
☞ Easy add and drop (SDH performs single stage multiplexing)

☞ Supports various topologies ring, bus, star and etc

☞ It provides both electrical and optical input interfaces

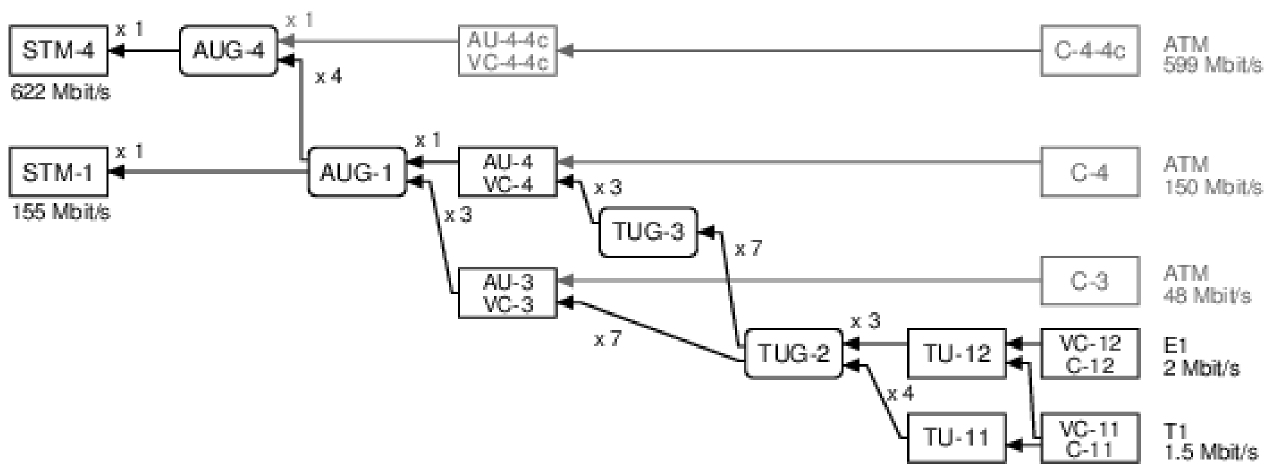
☞ NMS (Network Management System) facility

STM-1 synchronous transport module level 1



- 270 column.
- 9 rows.
- Row by Row Transmission
- Time period is 125us
- Each byte represents 64Kbps
- ITU-T recommendation G.707
- STM-1Frame Calculation

- Total byte= 270*9=2430
- Each byte=8 bits
- Total bits = 2430*8=19440
- Frame Frequency=8000
- Time period=1/8000=125us
- Data rate of one byte = 8*8000=64Kbps
- Data rate of frame =19440*8000=155Mb



POH Path Overhead

Path overhead is a monitoring and alarm information field. Path overhead size varies with container size. POH travel with the container over the path.

Virtual container = POH + Container

VC 11, VC 12, VC 2, VC 3, VC 4.

SOH Section Overhead

Overhead has basically the same job everywhere (Management, Control, Alarm and etc). In SDH network, Overheads are coupled with network architecture to simplify the task.

Regenerator Section Overhead (RSOH)

Pointer (AU-PTR): It indicates the first byte of a virtual container. Its function becomes important in the case of asynchronous case.

Multiplex Section Overhead (MSOH)

SDH Signal Mapping

Container C: It accepts data from pdh level signals. It performs bit stuffing to adjust data rate.

Virtual Container VC: It has container and POH. POH contains alarm information, Performance and Some management information.

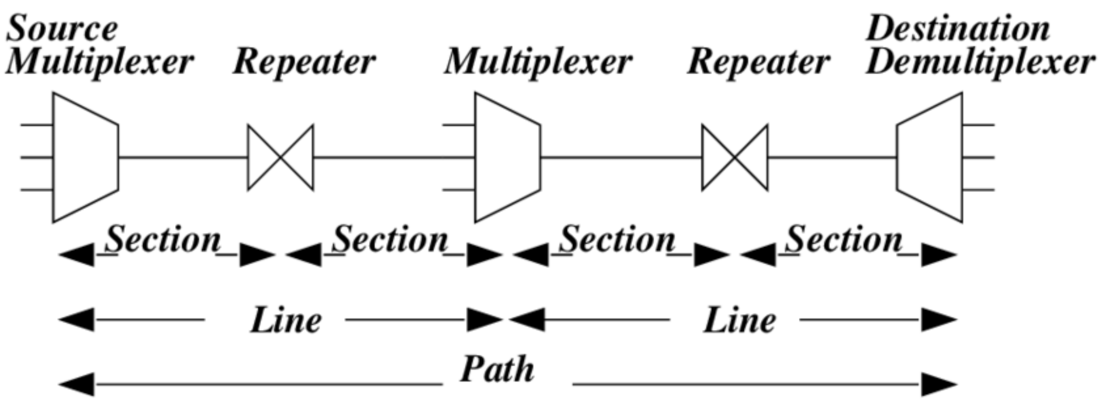
Tributary Unit TU **=** VC + Pointer

Tributary unit Group TUG

Administrative Unit AU

Administrative Unit Group AUG

SDH Network Element



Regenerator

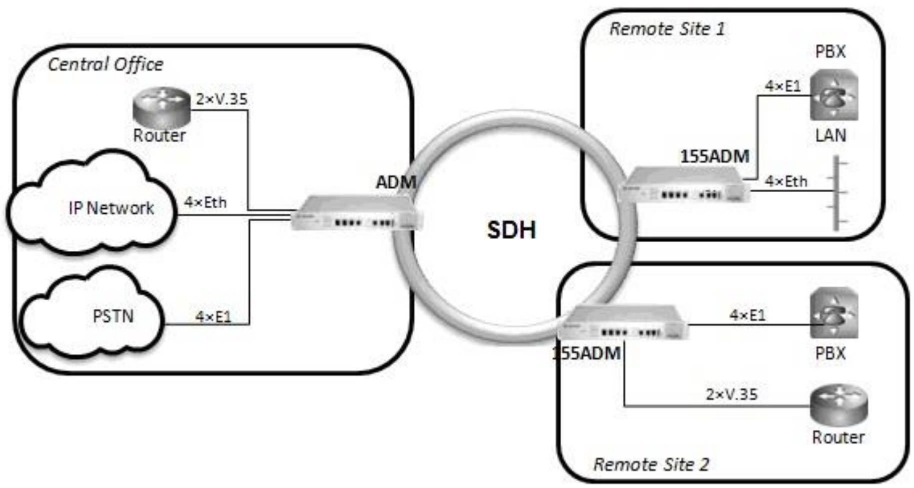
- It regenerates the attenuated signal
- Terminal Multiplexer or Path Terminal Element (PTE)
- It is end point device. It combines the lower level of signals into a STM-N level

ADD/DROP Multiplexer

- In-ring setup, it performs the add and drop of multiple levels Of signals

Digital Cross Connect (DXX)

- It performs the switching function for various sizes of containers



STM-N Levels

The basic rate of transmission 155.54 Mbps. SDH hierarchy based on multiple of 155.54Mbps

STM-1 155.54Mbps

STM-4 622Mbps

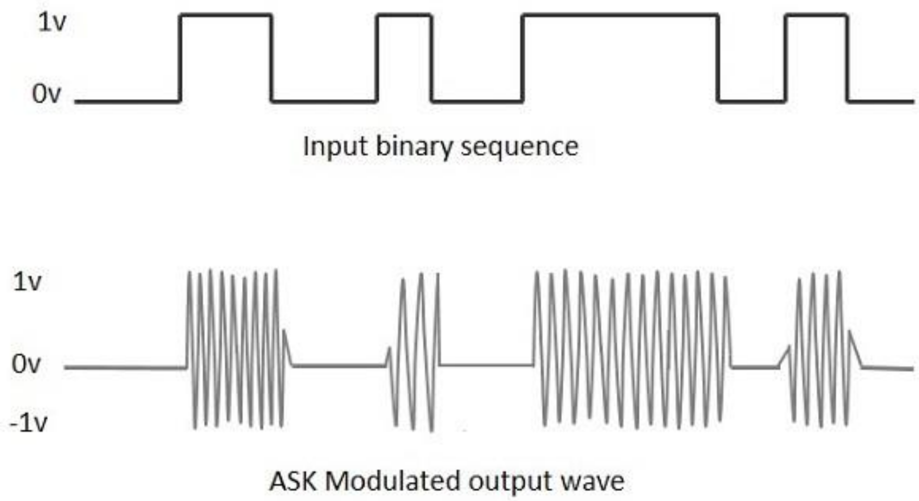
STM-16 2.5Gbps

STM-64 10Gbps

// ASSIGNMENT: SONET

Amplitude Shift Keying

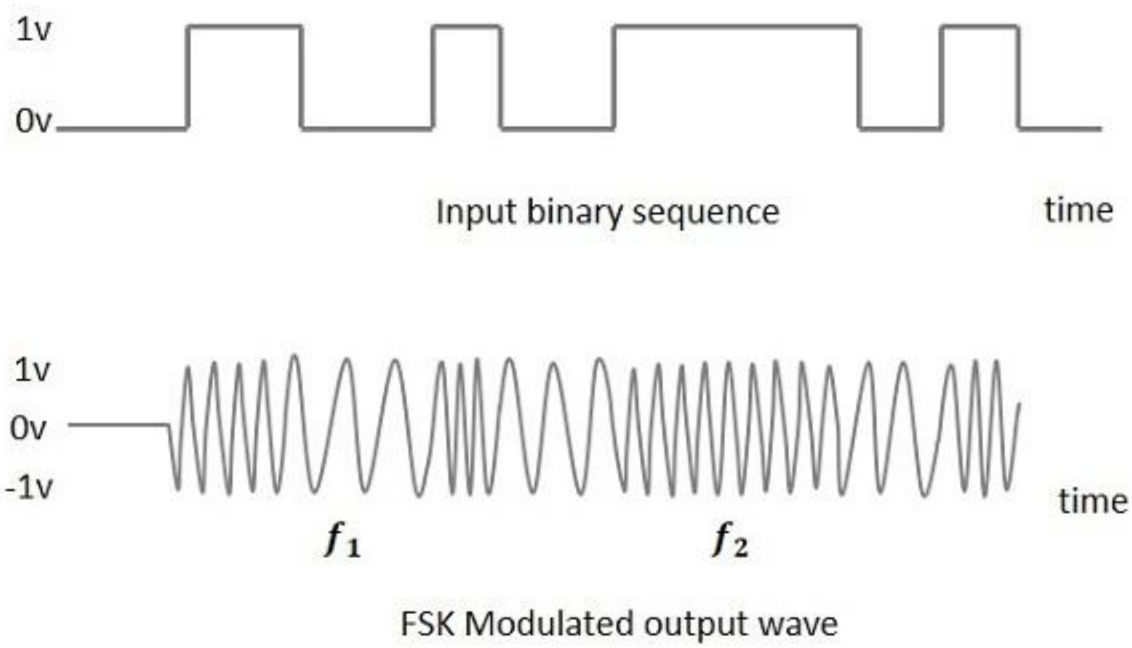
- In Amplitude Shift Keying (ASK), each symbol in the message signal gives a unique amplitude to the carrier wave. There are two types of ASK, Binary and M-ary.
- In Binary ASK, logic 1 is associated with certain amplitude of carrier wave e.g. 12V and logic 0 is associated with different amplitude other than 12V e.g. 0V.
- In M-ary ASK, a group of log2M bits are considered together rather than 1 bit at a time and the amplitude level is associated with this group of bits.
- For example, in 16-ary ASK, a group of 4 bits are considered and are given a respective amplitude. Since there are 16 possible 4 bit binary numbers (24), 16 different amplitude levels are required for modulation. If all such amplitudes are created using a single carrier wave, then it is called as coherent ASK. If multiple carrier wave each with different amplitudes are used for modulation then it is called as non-coherent ASK.



Frequency Shift Keying (FSK)

- In Frequency Shift Keying (FSK), each symbol in the message signal gives a unique frequency to the carrier wave. There are two types of FSK, Binary and M-ary. In Binary FSK, logic 1 is associated with certain frequency of carrier wave e.g. 50MHz and logic 0 is associated with different frequency other than 50MHz e.g. 25MHz. In M-ary FSK, a group of log2M bits are considered together rather than 1 bit at a time and the frequency is associated with this group of bits.

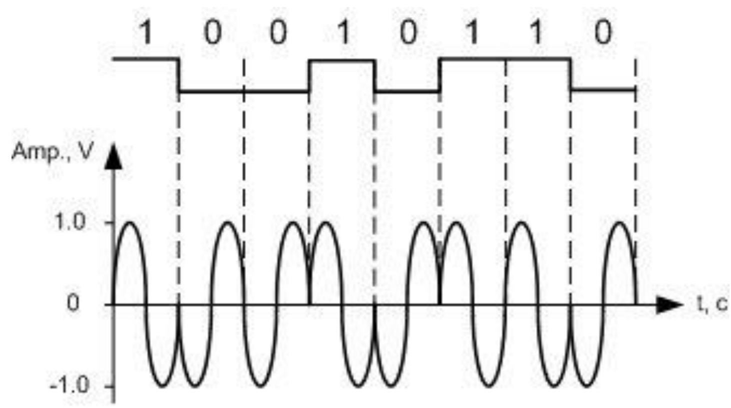
- For example, in 16-ary FSK, a group of 4 bits are considered and are given a respective frequency. Since there are 16 possible 4 bit binary numbers (24), 16 different frequencies are required for modulation. If all such frequencies are created using a single carrier wave, then it is called as coherent FSK. If multiple carrier wave each with different frequencies are used for modulation then it is called as non-coherent FSK.



Phase Shift Keying (PSK)

In Phase Shift Keying (PSK), each symbol in the message signal gives a unique phase shift to the carrier wave. There are two types of PSK, Binary and M-ary. In Binary PSK, logic 1 is associated with certain phase shift of carrier wave e.g. 90° and logic 0 is associated with different phase shift other than 90° e.g. 0°. In M-ary PSK, a group of log2M bits are considered together rather than 1 bit at a time and the phase shift is associated with this group of bits.

For example, in 16-ary PSK, a group of 4 bits are considered and are given a respective phase shift. Since there are 16 possible 4 bit binary numbers (24), 16 different phase shifts are required for modulation. If all such phase shifts are created using a single carrier wave, then it is called as coherent PSK. If multiple carrier wave each with different phase shifts are used for modulation then it is called as non-coherent PSK.



Phase shift keying (PSK)