Transmission

Q1.

1.1 Transmission essentially involves transporting of information from one place to another. Explain why digital transmission cannot be achieved by merely following the Sampling Theorem.

Sampling theorem involves in three processes i.e. Sampling such that $f_s \langle 2f_m$ (f_m is the maximum frequency of the band limited signal), Media, by low-pass filtering the original signal can be obtained. Here reproducing the samples through a transmission media cannot be achieved. Hence a technique is used to resolve this. i.e. PCM.

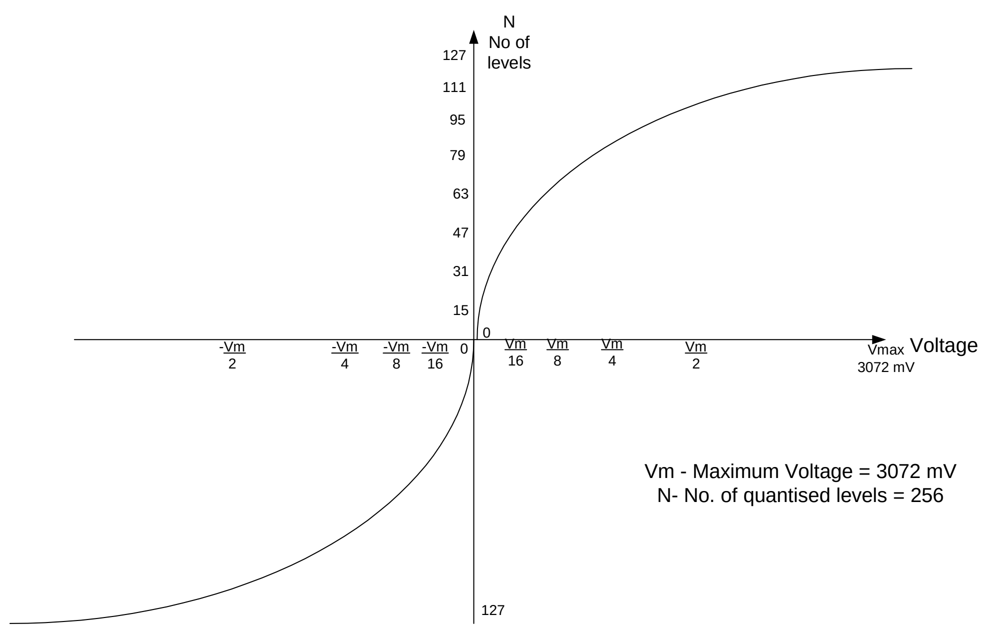
1.2 Explain the basic process of Pulse Code modulation.

Sampling ---- Quantizing – Encoding – Transmission

1.3 In practice, non-linear quantizing is deployed as against linear quantizing, explain the reasons behind. 256 levels are used for non-linear quantizing and there are 8 segments each positive and negative sample. Explain the basic characteristics of non-linear quantizing used for European system (the A-law signal compression of 13 linear segments)

NLQ is used to provide a good S/N ratio for a wider span of signals. This is essential as about 90% of the signals lie between $+V_m/2$ and $-V_m/2$.

The following shows the NLQ characteristics.



1.4 Each quantized level will be converted to binary before transporting the quantized level (related to the sample) into the transmission media. Assuming a telephone will generate a maximum of ±3072 mV calculate the encoded format that will be sent in the transmission media for the following sample values.

i. + 2210mV
ii. + 128mV
iii. – 14mV
iv. –416mV

i. +2210mV = 1 111 0110
ii. +128mV = 1 011 0100
iii. –14mV = 0 000 1000
iv. –416mV = 0 101 0000

Q2.

(a) Time Division Multiplexing (TDM) is deployed in the PCM system in order to maximize the utilization of the transmission media. Explain briefly the TDM systems deployed in the Europe and America / Japan.

Digital Transmission is governed by sampling theorem. According to the sampling theorem voice band (20KHz) is limited to a telephone channel of 4kHz and this limited 4 kHz signal is sampled at 8 kHz, the resultant samples has to be reproduced over the other end of the transmission media by low-pass filtering. This process involves 8,000 samples per second for a given voice. This has resulted in sending one sample in every 125us. In other words 125us period is being used by the transmission media if we use to transmit only for one telephone channel. This idling period can be further utilized to send some other similar telephone channels sampling information which will follow the same destination. The concept of Time Division Multiplexing derive how to utilize the transmission media more efficiently. In the Europe system 32 channel samples are being sent during this 125us while in the American and Japanese system 24 channel samples are used. If we assume 8 bits per sample for 125us the Europe system will send 256 bits and the American system will send 192 bits resulting 2.048 Mbps and 1.542 Mbps respectively.

Data rates used under Europe and America/Japan

Europe:	2.048Mbit/s
	8.448Mbit/s
	34.368Mbit/s
	139.264Mbit/s
North American:	1.544Mbit/s
	6.312Mbit/s
	44.376Mbit/s
Japan:	1.544Mbit/s
	6.312Mbit/s
	32.064Mbit/s
	97.728Mbit/s

(b) In practice, non-linear quantizing is deployed as against linear quantizing, explain the reasons behind.

The quality factor of any transmission media is measured by Signal to Noise Ratio. In linear quantizing SNR is good only high valued samples where as in the practical application 90 percent of the samples are lying around + or – $V_{max}/2$ where V_{max} is the highest voltage generated by a telephone. Hence linear quantizing will not be suitable for the practical applications for the transmission of human voice. Hence non-linear quantizing is used. In the non-linear quantizing out of 256 quantized levels only 16 quantizing levels are used for a range of $V_{max}/2$ to V_{max} to suit practically.

(c) 256 levels are used for non-linear quantizing and there are 8 segments for each positive and negative sample. Explain the basic characteristics of non-linear quantizing used for European system (the A-law signal compression of 13 linear segments)

Y axis: Number of levels - 8 segments, each with 16 sub-levels (i.e. +127 to -127)

X axis: Voltage in mV - From +Vm to $-V_{m,}$ where V_m= 3,072mV

Gradients of upper 0 and 1 segments and lower 0 and 1 segments are same. Therefore, 15 total number of segments are reduced to 13 segments.

This makes it a 13 segment graph.

Segment No.	Voltage Range (mV)	Level Range	Increment/Level
7	1,536 – 3,072	111 – 127	96
6	768 – 1,536	95 – 111	48
5	384 – 768	79 – 95	24
4	192 – 384	63 – 79	12
3	96 – 192	47 – 63	6
2	48 – 96	31 – 47	3
1	24 – 48	15 – 31	1.5
0	0 – 24	0 – 15	1.5

According to the voltage of the sampled signal, a segment number and a level number is determined.

This creates a 8-bit representation of the sampled signal as shown below;

Sign: + or – (1-bit)

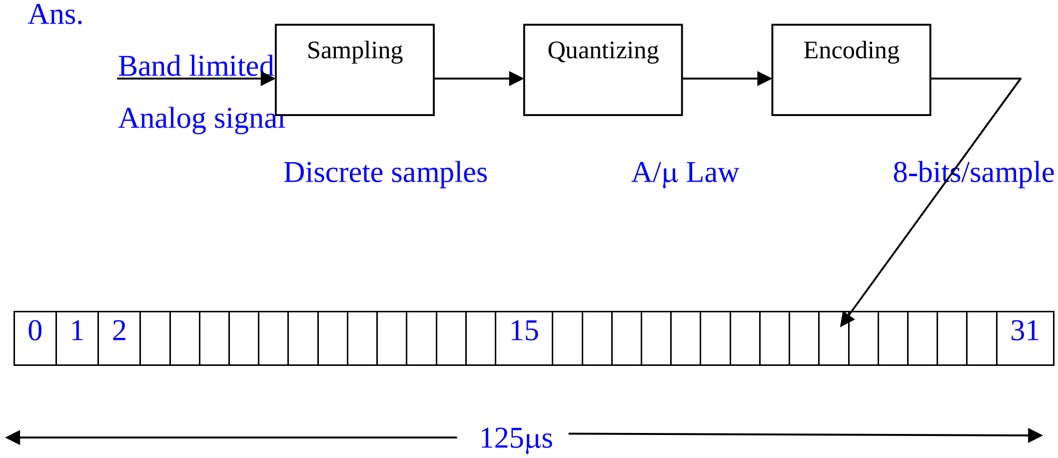
Segment No.: 3-bits

Level No.: 4-bits

Q3.

3.1 Draw a block diagram of PCM clearly explaining how a basic frame structure is achieved in each block.

Ans.



Bits/Time slot	=	8
Total number of bits in the PCM frame	=	8 * 32
	=	256
Data rate of the frame	=	256/125µs
	=	2.048Mbit/s
Time taken for one slot	=	125/32
	=	3.9µs
Data rate of a slot	=	8/3.9 µs
	=	64kBit/s

This gives a PCM frame of 32 time slots (each with 64Kbit/s rate) at a data rate of 2.048Mbit/s where time slot 0 is used for synchronizing while time slot 16 is used for synchronizing.

3.2 A PCM system will comprise of sampling, quantizing and encoding. Explain each block. clearly.

Sampling: Sampling a band limited analog signal at a frequency higher than twice the maximum frequency of the analog signal

Quantizing: Non-linear quantizing of sampled values

Encoding: Assigning ‘0’s and ‘1’s for the quantized voltage levels

Sign: + or – (1-bit)

Segment No.: 3-bits

Level No.: 4-bits

3.3 Three consecutive voice samples in a 30 channel PCM system are identified as follows.

+50V, 0V, -68V

Draw the wave forms that you can observe in the transmission media, assuming HDB3 transcoding. (Assuming A-Law compression for non-linear quantizing)

(Refer the Attachment-I)

Model answers-Website.xls

Q4.

i. Explain the following with reference to PCM.

Sampling

Quantizing

Encoding

Sampling: Sampling a band limited analog signal at a frequency higher than twice the maximum frequency of the analog signal

Quantizing: Non-linear quantizing of sampled values

Encoding: Assigning ‘0’s and ‘1’s for the quantized voltage levels

Sign: + or – (1-bit)

Segment No.: 3-bits

Level No.: 4-bits

ii. What are the applications of TS0 and TS16 in the PCM frame.

TS0 - for Synchronization

TS16 - for sending Supervisory signals of Ch.1 to Ch.15 and Ch.17 to Ch.31

iii. Briefly explain why higher order multiplexing is necessary in telecommunication.

In SMW4 submarine cable system, STM-64 is used which is equivalent to 10Gbit/s. Briefly Explain how a bit in a sample is multiplexed and de-multiplexed if used SMW4.

Telecommunication involves transporting user information from one place to another. When the number of simultaneous calls increase, capacity of the transporting media too has to be expanded to carry the additional information content This increases the cost hence the tariff..

As capacity/bandwidth is a crucial factor in telecommunication link design, it is necessary to combine many user information in a transmitting station in such a way that individual information contents are separated and retrieved at the receiving station.

Higher order multiplexing is used for this in telecommunication.

Different PCM streams are first multiplexed using either PDH or SDH technology and transported through a same medium. At the receiving station, these PCM streams are de-multiplexed.

This makes it possible to share the same transporting medium (microwave, copper cable or optical fibers) by different PCM streams.

Q5.

C

"Digital transmission is essential in today's telecommunication as it is the most efficient way of reproducing the transmitted signal at the receiving end."

Explain the above statement with reference to different stages in digital transmission.

In digital transmission, first a band-limited analog signal is sampled using the Nyquist theorem. This makes it possible to reproduce the original signal with a minimum noise at the receiver. These discrete samples are then quantized using non-liner quantization. As most of the signals lie Within $+V_m/2$ and $-V_m/2$, non-linear quantizing is used to have a S/N ratio all the sampled values.

Quantized values are then encoded so that every sampled and quantized values are represented as 8-bits. Finally, a PCM frame is built with such encoded patterns and transmitted through the medium. In such a way, in all stages, attention is paid to reduce the noise introduced to the original signal while undergoing this process.

Therefore, digital transmission is essential in today's networks as it ensures reproduction of original signal at the receiving end.

5.1 What are the different types of repeaters?

i. Active repeaters – feeds power, repeats the signal

ii. Passive repeaters – functions as a reflector

5.2 Repeaters are extensively used in Pulse Code Modulated (PCM) transmission links in Cable PCM transmission systems.

Explain different sections of a repeater with reference to the functions of each unit.

Repeaters carryout 3 different functions in cable PCM networks.

i. Reshaping
ii. Retiming
iii. Power feeding

Different functional units in a repeater:

i. Line equalize – Amplifies low frequency signals with a high gain and high frequency signals with a low gain, to equalize different frequency signals received

ii. Rectifier – O/p values is rectified

iii. Tuned amplifier – Driven by the rectifier o/p and regenerates the clock pulse

iv. + Threshold detector – Gives an o/p when the signal value is higher than the threshold

v. – Threshold detector - Gives an o/p when the signal value is lower than the negative threshold

vi. Differentiator – Provides a short spark when there is an output from the Tuned amplifier

vii. + Reshaper – O/p a signal if there is a value when +Threshold detector gives a spark

viii. – Reshaper - O/p a signal if there is a value when -Threshold detector gives a spark

ix. Power feeding equipment

Q6.

Explain the basic difference between SDH and PDH transmission.

SDH support high data rates compared to PDH. Hence higher capacity equipment, transmission media etc., are necessary. Because of these higher bandwidth requirements, it is not possible to use certain transmission media and also have to limit the data rates in certain mediums. (e.g. Microwave radio)

i. Clearly explain the difference between PDH and SDH multiplexing methods.

PDH and SDH are two different multiplexing techniques used in telecommunication. Their main features are shown below in brief;

PDH:

Less synchronized

Easy to operate

Absent of high network management features

Low data rates

High efficiency in allocating bits in the frame

More on manual control

SDH:

Fully synchronized

More software oriented

Network management features available

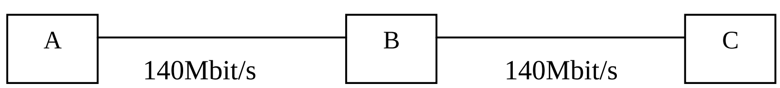
High data rates

Low efficiency in the allocation of bits in the frame

Low user interaction necessary

Interoperability is possible

ii. Assume three transmitting stations A,B and C.



Consider the following E1 distribution between above 3 stations.

Station 1	Station 2	No. of 2Mbit/s terminated
A	B	33
B	C	44
A	C	16

(a) State the total utilization (Used 2Mbit/s/Total 2Mbit/s available) of each link. (AB and BC)

Link AB:

Used E1/Total E1 = 49/64 = 0.77

Link BC:

Used E1/Total E1 = 60/64 = 0.94

(b) State the capacities of transmission equipment to be used, so that the link utilization is optimum, in these 3 stations if use,

PDH technology

Link AB:

Same, i.e. 140Mbit/s

Link BC:

Same, i.e. 140Mbit/s

SDH technology

Link AB:

STM-I (155.52Mbit/s)

Link BC:

STM-1(155.52Mbit/s)

iii. Explain clearly the advantages and disadvantages of SDH.

Advantages:

Provision of high data rates

Availability of advanced network management facilities

Less user interaction

Interoperability is possible

Disadvantages:

High cost

Complex systems

High skill of staff required

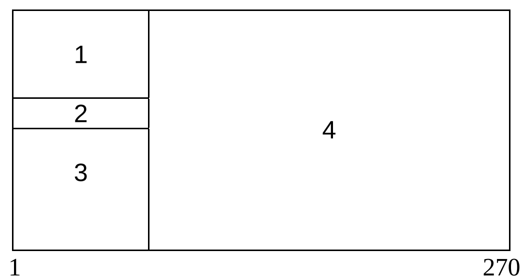
Q7.

7.1 STM-1 Frame structure is defined by 9x270 Matrix, each point of the Matrix is a byte (8 bits). Show that STM-1 will carry 155.52 Mbps.

Data rate = $[(9*270)*8]/125*10^{-6}$

= 155.52Mbits/s

7.2 The following shows the STM-1 frame. Explain the importance of each area shown.



1 – RSOH: carries overhead information for the management of regenerator section

2 – AU Pointer: Used as the VC-4 pointer

3 - MSOH: carries overhead information for the management of multiplexer section

4 - Payload: Contains actual user information

7.3 SDH will use the following 3 processes.

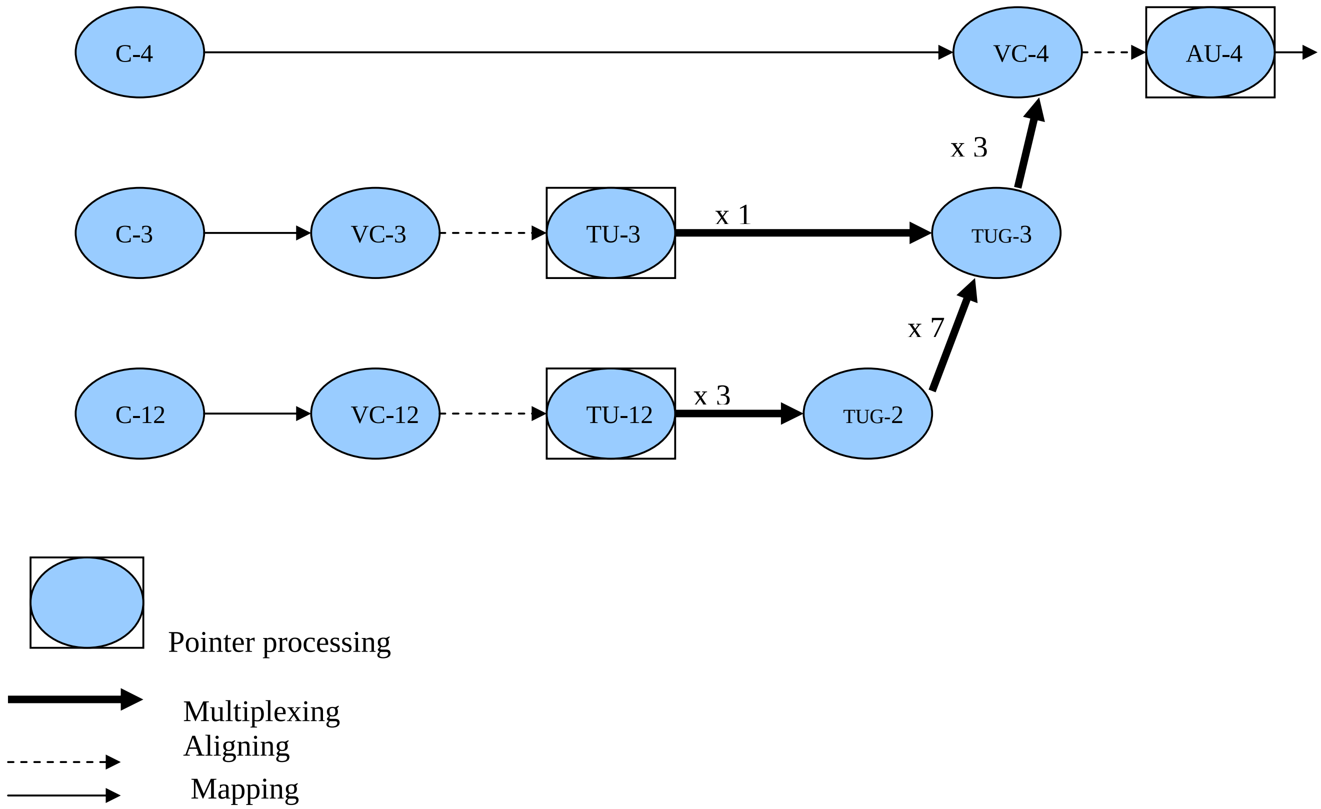
v. Mapping
vi. Aligning
vii. Multiplexing

Briefly explain the applications of above when a PDH 2.048 Mbps, 34.368 Mbps, 139.264 Mbps is converted to STM-1. Draw the relevant diagram.

Mapping – process where a container is converted to a virtual container by adding a POH byte

Aligning – Process of assembling a VC into a tributary unit where a pointer is added to point to the position of the first byte of the VC

Multiplexing – Process where a TUG/AU is formed by TUs by byte interleaving



7.4 Draw the SDH multiplexing hierarchy from STM-1 to STM-256 and show the basic information of 1 bit is shrink from 6.4 ns to 25 ps when the multiplexing is achieved up to STM-256. Clearly show the bit speed of each SDH hierarchy.

STM-1 ------ STM-4 ------- STM-16 -------- STM-64 ------- STM-256

Time duration for 1 byte in STM-1 = $1/155.52*10^6$ = 6.4ns

,, STM-4 = 6.4/4= 1.61ns

,, STM-16 = 6.4/16 = 0.4ns

,, STM-64 = 6.4/64 = 0.1ns

,, STM-256 = 6.4/256 = 0.025ns = 25ps

Bit speeds:

STM-1	155.52Mbit/s	$\simeq$ 155 Mbits/s
STM-4	622.08Mbit/s	$\simeq$ 622 Mbits/s
STM-16	2.488Gbit/s	$\simeq$ 2.5 Gbits/s
STM-64	9.953Gbit/s	$\simeq$ 10 Gbits/s
STM-256	39.8Gbit/s	$\simeq$ 40 Gbits/s

Attachment-I