

Department : CSE

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Answer to the question no : 1 (a)

Layering in network : It is the concept of layering to simplify to reduces the complexity of its working. The first computer network were designed with the hardware as the main concern and the software as an afterthought.

The necessity of having layering in a network are described below -

- ① It breaks network communication into smaller, more manageable parts.

- ⑩ It standardizes network components to allow multiple vendor development and support.
 - ⑪ It allows different types of network hardware and software to communicate with each other.
 - ⑫ It prevents changes in one layer from affecting other layers.
 - ⑬ It divides network communication into smaller parts to make learning it easier to understand.
- ②

Adjacent layers: It refers to how the adjacent networking layers in the same computer interact to each other. The higher-layer protocol uses the next lower-layer protocol to perform the service if needed.

We have seen a few examples:

- ① Application layer protocol HTTP uses the transport layer, TCP to perform the error recovery service if needed.
- ② Transport layer protocol UDP uses the internet layer protocol IP to perform

the IP address and routing, service if needed.

③ Internet layer protocol IP uses the link layer protocol Ethernet to perform the host to host physical communication service if needed.

④

Answer to the question no. 1 (B)

The key functions of the Data Link Layer are,

Framing: Divides the stream of bits received from network layer into manageable data units called frames.

Physical Addressing: Add a header to the frame to define the physical address of the source and the destination machine.

Flow Control: Impose a flow-control - control rate at which data is transmitted so as not to flood the receiver.

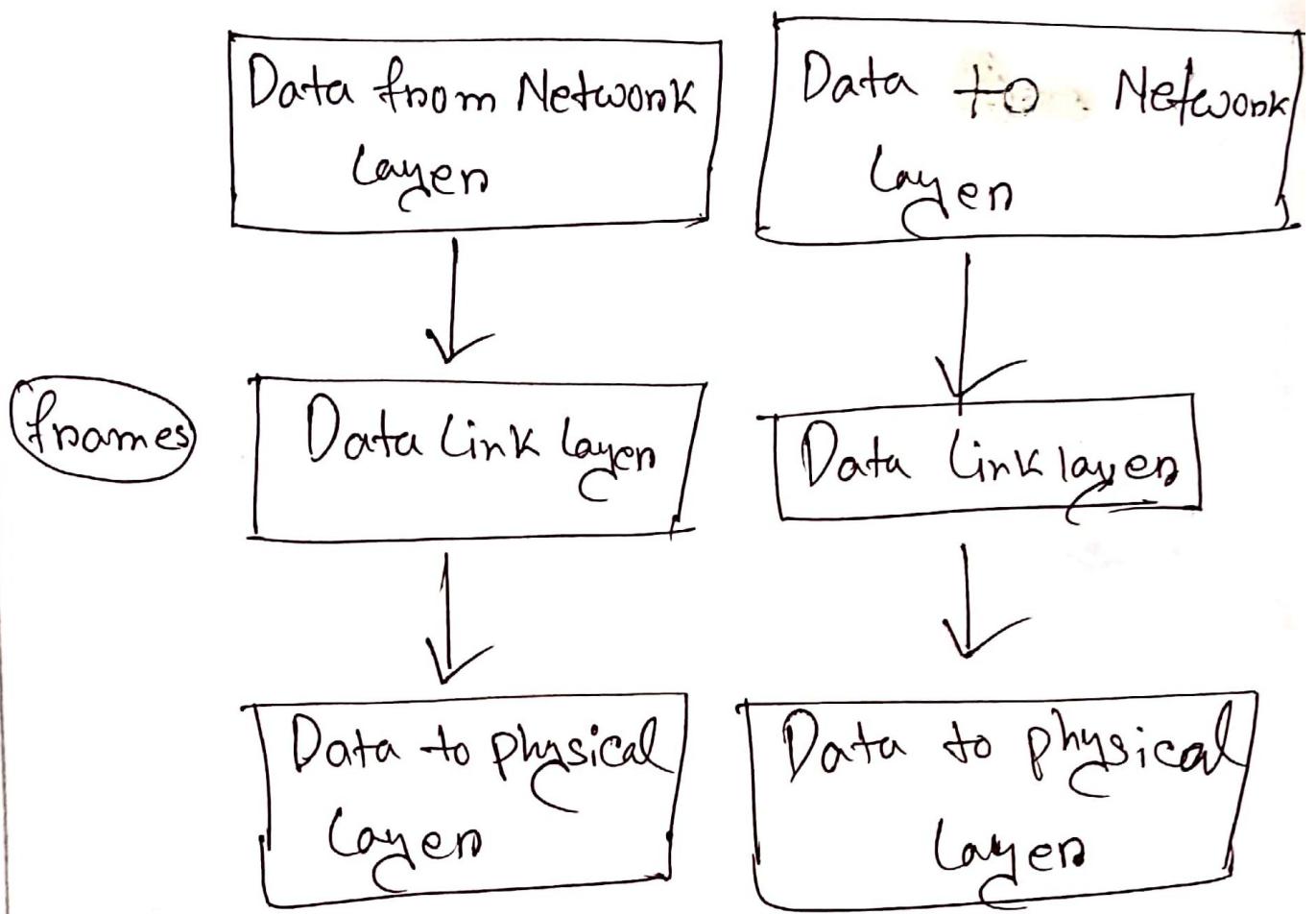
(h)

Error Control: Error Control is achieved by adding a trailer at the end of the frame. Duplication of frames are also prevented by using the mechanism.

Access Control: Protocols of this layer determine which of the devices has control over the link at any given time, when two or more devices are connected to the same link.

(6)

Synchronization: When data frames are sent on the link, both machines must be synchronized in order to transfer to take place.



2

Answers to the question no: 1(c)

Given that,

$$2 \text{ million bytes} = 2000 \times 8 \text{ kb}$$
$$= 16000 \text{ kb}$$

Time taken to download 56 kbps channel

$$= \frac{16000}{56}$$
$$= 285.7 \text{ s}$$

Time taken to download 1 mbps channel

$$= \frac{16000}{1024}$$
$$= 15.6 \text{ s}$$

⑧

Answers to the Question no: 2 (a)

The Nyquist theorem, also known as the sampling theorem, is a principle that engineers follow in the digitization of analog signals. For analog-to-digital conversion (ADC) to result in a faithful reproduction of the signal, slices, called samples of the analog wave form must be taken frequently.

⑨ Any Analog signal consists of components at various frequencies. The simplest case is the sine wave.

in which all signal energy is concentrated at one frequency.

In practice, analog signals usually have complex wave forms with components at many frequencies.

The highest frequency the bandwidth of that signal.

Suppose the highest frequency component, in hertz, for a given signal is f_{max} . According to the Nyquist theorem, the sampling rate must be at least $2f_{max}$, or twice

⑩

the highest analog frequency component. If the sampling rate is less than $2f_{\text{max}}$, some of the highest frequency component in the analog input signal will not be correctly represented in the digitized output.

When a such analog signal is converted back to analog form by a digital to analog converter, false frequency. This undesirable condition is a form of distortion called ~~also~~ aliasing.

11

2-(b) ①

$$\begin{array}{r}
 10100110110 \\
 10111 \overline{-} 10100 \quad 11110000 \\
 10111
 \end{array}$$

$$\begin{array}{r}
 00111 \\
 00000 \\
 \hline
 \end{array}$$

$$\begin{array}{r}
 01111 \\
 00000 \\
 \hline
 \end{array}$$

11111

10111

$$\begin{array}{r}
 10001 \\
 10111 \\
 \hline
 \end{array}$$

01100

00000

11000

10111

11110

10110

10010

10111

01010

0000

1010

$q = 101001110110$

②

(ii)

$$\begin{array}{r} 10011 \\ \hline 10111 \} 101001100 \\ 10111 \\ \hline 00111 \\ 00000 \\ \hline 01111 \\ 00000 \\ \hline 11110 \\ 10111 \\ \hline 10010 \\ 10111 \\ \hline 0101 \end{array}$$

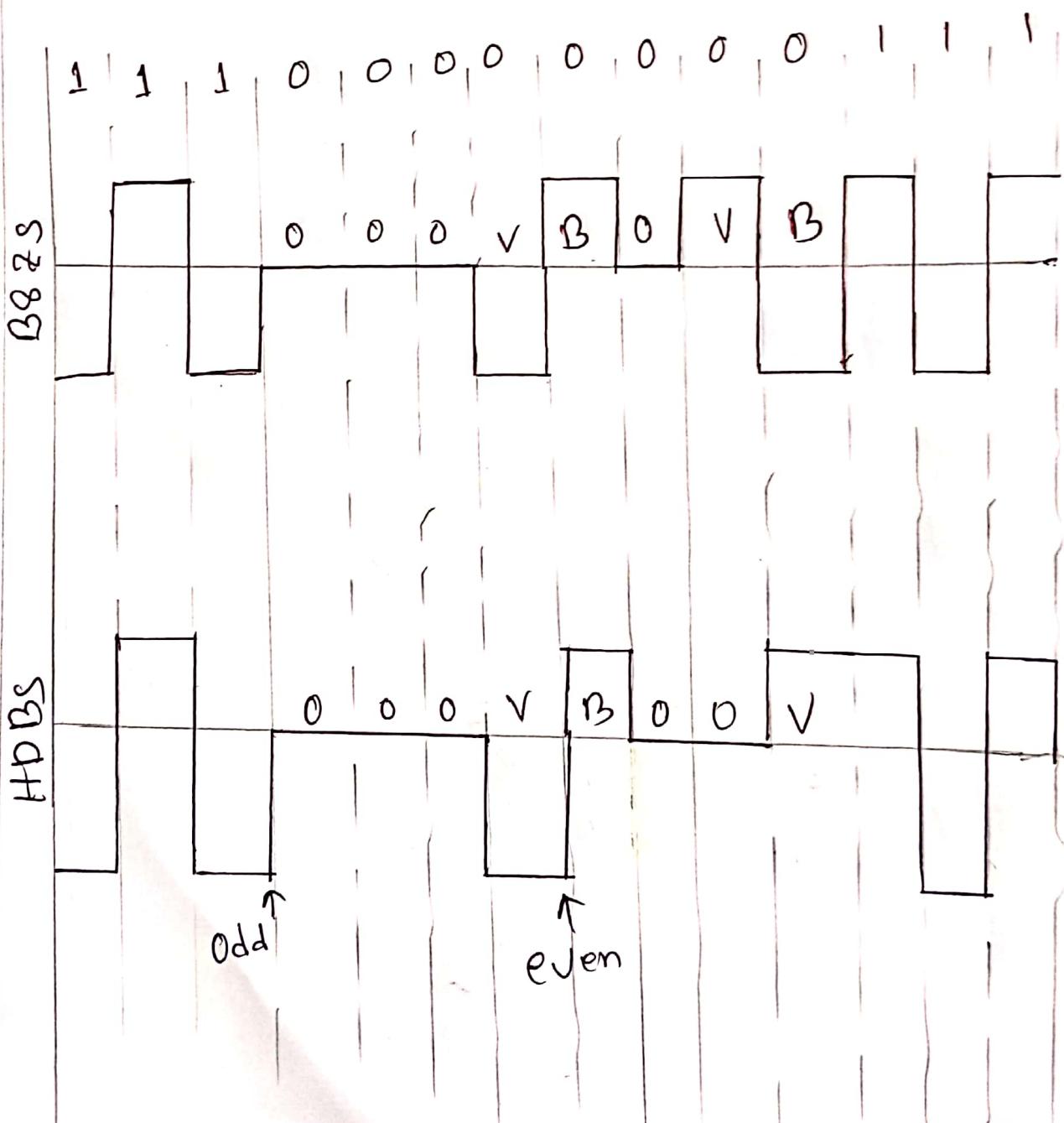
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Answer to the question no: 2(c)

Mg id is 170204105

105 → odd

111 0000 0000 111



Answer to the question no: 3(a)

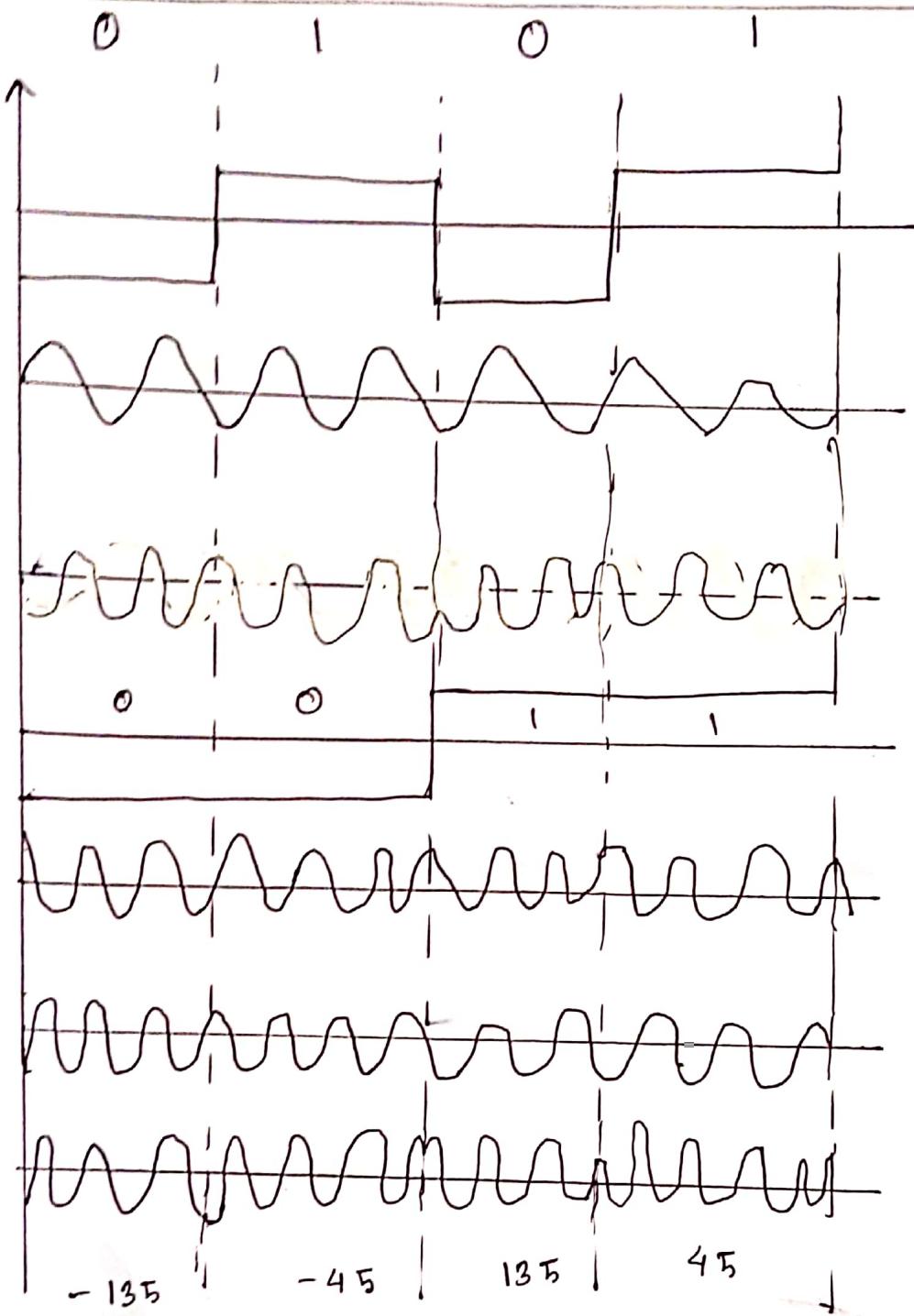
We can QPSK instead of BPSK because QPSK has advantages of having double data rate compare to BPSK. This is due to support of two bit per carrier in QPSK compare to one bit per carrier in the case of BPSK.

Implementation of QPSK: The simplicity of BPSK enticed designers to use 2 bits at a time in each signal element, then by decreasing the baud rate and eventually the required bandwidth. The scheme is called quadrature PSK or QPSK because it uses two separate BPSK modulations; one is in-phase

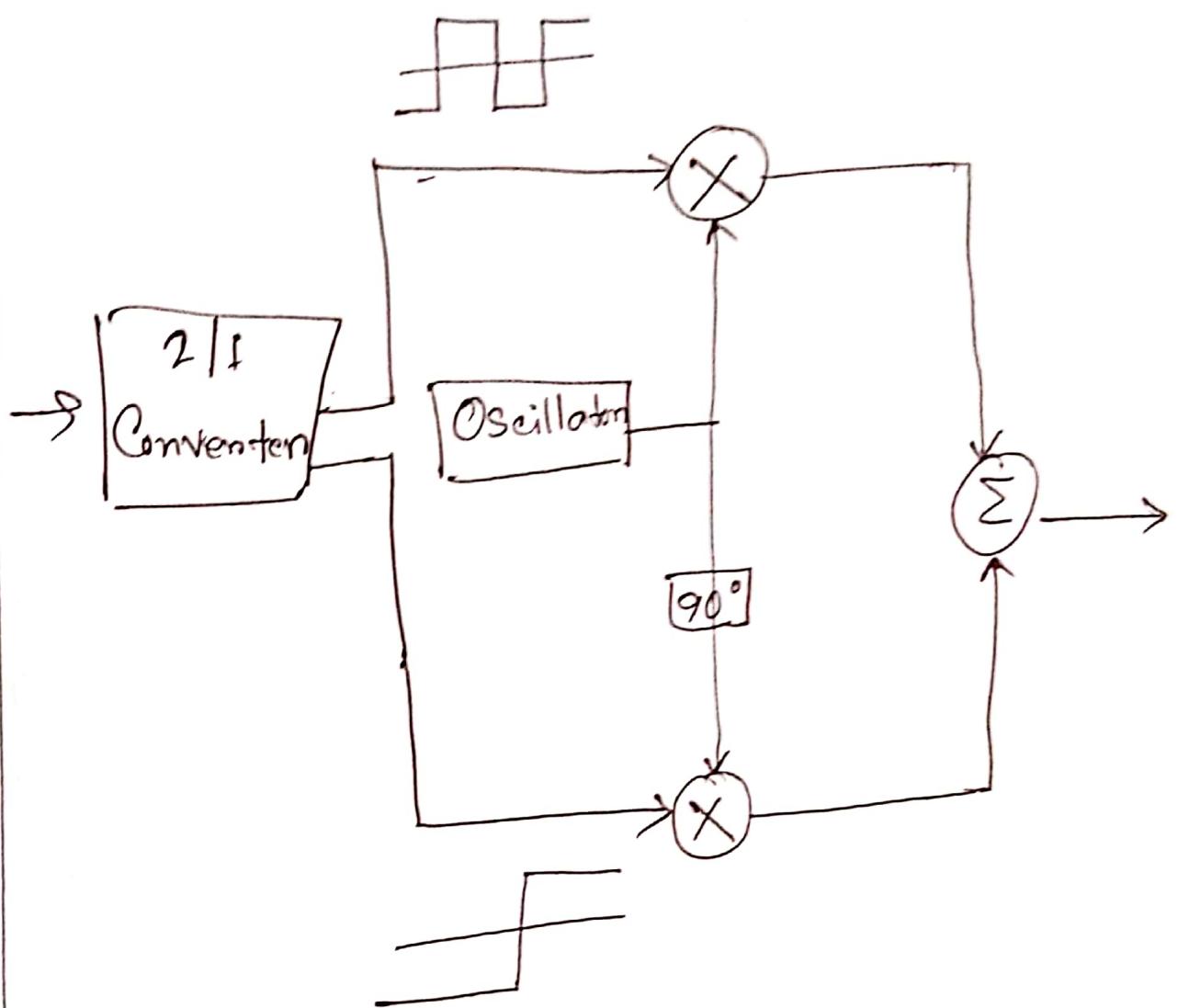
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the other quadrature. The incoming bits are first passed through a serial-to-parallel conversion that sends one bit to one modulator and the next bit to the other modulator. If the duration of each bit in the incoming signal is T , the duration of each bit B sent to the corresponding BPSK signal is $2T$. This means that the bit to each BPSK signal has one-half the frequency of the original signal.

⑩



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Answers to the question no: 3(b)

Nyquist Theorem: Nyquist theorem state that a periodic signal must be sampled at more than twice the highest frequency component of the signal. In practice because of the finite time available a sample rate somewhat higher than this is necessary.

The Nyquist theorem also known as the sampling theorem. It is a principle that sampling theorem. It is a principle that engineers follow in the digitization of analog signals.

(19)

i) For FSK: $S = 1/n \times N$

$$n = \log_2 2 = 1$$

$$\therefore S = 1/1 \times 2000 = 1/1 \times 2000 = 2000 \text{ baud}$$

ii) For ASK:

$$n = \log_2 2 = 1$$

$$\therefore S = 1/1 \times 4000 = 4000 \text{ baud}$$

iii) For QPSK

$$n = \log_2 4 = 2$$

$$\therefore S = 1/2 \times 6000 = 3000 \text{ baud}$$

iv) For 64-QAM

$$n = \log_2 64 = 6$$

$$\therefore S = 1/6 \times 36000 = 6000 \text{ baud}$$

20) We, can say that 64-QAM has the highest baud rate while FSK has the lowest baud rate.

Answer to the question no: 3(c)(i)

Yes we can say if a signal is periodic or non periodic by just looking its frequency domain plot. For periodic frequency domain plot the graph will be a straight line corresponding to the frequency. On the other hand non-periodic signal will not be a straight line.

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Answers to the Question no: 3(c)(ii)

The cause of impairment →

- ① Attenuation
- ② Distortion
- ③ Noise

~~Attenuation~~: Attenuation

Attenuation: Attenuation means of energy. When a signal simple or composite travels through medium it loses some of its energy in overcoming the resistance of the medium.

② Distortion: Distortion means that the signal changes its form or shape. Distortion can occur

in a composite signal made of different frequencies.

Noise: Noise is another cause of impairment. Several types of noise such as thermal noise, induced noise, crosswalk, and impulse noise may corrupt the signal.

Answer to the question no: 4(a)

The traditional checksum uses a small number of bits (16) to detect errors in a message of any size.

At least three types of errors cannot be detected by the current checksum

Calculation, first, if two data items are swapped during transmission, the sum and the checksum values will not change. Second, if the value of one data item is increased and the value of another one is decreased the same amount, the sum and the checksum

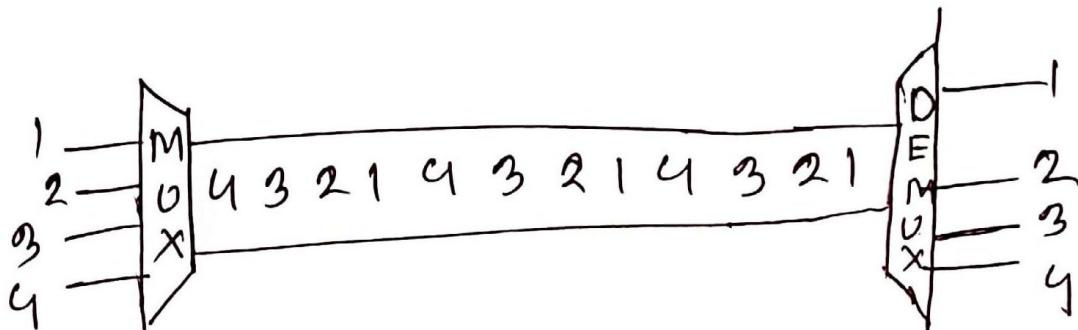
Can not be detect these changes. Third, if one or more data items is changed in such a way that the change is a multiple of $2^{16}-1$, the sum on the checksum can not detect the changes".

However, the tendency in the Internet, particularly in designing new protocols, so, it is to replace the checksum with a CRC.

Answers to the question no: 4(b)

TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

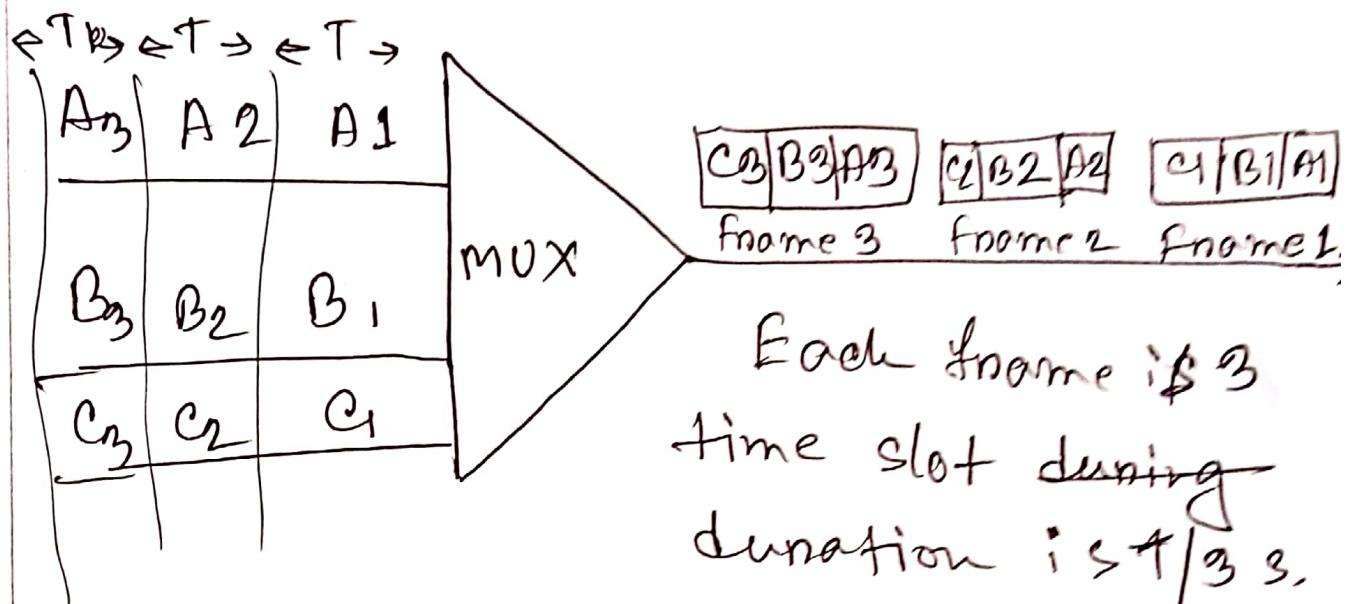
TDM is a digital process that allows several connections to share the high bandwidth of a line instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link.



In synchronous TDM, each input connection has an allotment in the output even if it is not sending data. In synchronous TDM, the data flow of each input connection is divided into units, where each input occupies one input time slot.

A unit can be 1 bit, one character, or one block of data. Each input unit

and occupies one output time slot is n times shorter than the duration of an input time slot. If an input time slot is T_s , the output time slot is T/n_s . Where n is the number of connections. In other words, a unit in the output connection has a shorter duration; it travels faster. The following figure shows an example of synchronous TDM where $n=3$.



Each frame is 3 time slots during duration is $4/3$ s.

In synchronous TDM, a round of data units from each input connection is collected into a frame. If we have n times slots and one slot is allocated for each unit, one for each input line. If the duration

2a

of each slot is T/n and the duration of each frame is T .

Time slots are grouped into frames. A frame consists of one complete cycle of time slots, with one slot dedicated to each sending device. In a system with n input lines, each frame has n slots with each slot allocated to carry data from a specific input line.

Answers to the question no: 4-(e)

$$\text{Each bandwidth} = (849 - 824) \\ = 25$$

$$= (894 - 869) \\ = 25$$

If we divide 25 MHz by 30 kHz, we get 833.33. In reality the band is divided into 832 channels. Of these, 42 channels are used for control, which means only 790 channels are used for control, which means only 790 channels are available for cellular phone users.

③

Ans: 790 channels

Answers to the question no-7 (a)

The relationship (I) between the size of the data word and the size of the codeword is : $n = k + r$

Where,

n = Size of the codeword

k = Size of the data word

r = Size of the remainder

(II)

The remainder is always one bit smaller than the divisor.



(I.F.F)

The degree of generator polynomial
is one less than the size of the divisor

(I.V)

The degree of the generator polynomial
is the same as the size of the
remainder.

Answer to the Question no: 7 (b)

My ID is : 170204105

The second last digit of my Id is 0
which is even.

So,

The augmented dataword = 1001000

Simulation

