Department : Computer Science & Engineering

Program : BSc. in CSE

Course No : CSE 3211

Course Title : Data Communication

Examination : Semester Final

Semester (session) : Spring 2020

Student ID : 170204070

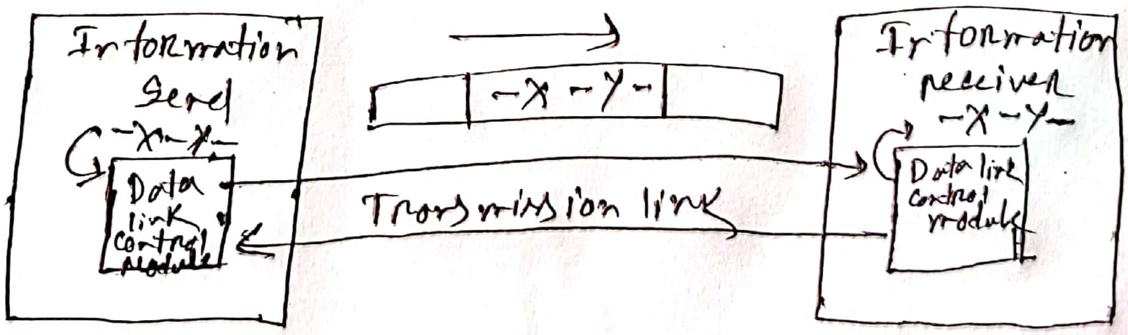
Signature & Date : 08.06.21

Ans to the Ques tion no - 1

1(a)

a) Answer: Layering ~~or~~ OR layered approach is a valid approach to deal with a complex problem. The communication functions are partitioned into hierarchical set of layers. Essentially this layered approach is a divide and conquer technique.

It is very important for a network.



~~The Job of the~~
This communication between layers will takes place through the layers boundaries in a

network. If the ~~laying~~ layering are not ~~the~~ judiciously defined then communication may become a problem or information to be communicated will become troublesome.

for example, we have to choose layering boundaries to minimize information flow across the boundaries. We are decomposing the data communication system into a number of layers somewhere ~~simulation to make~~ similar to make multitasking or parallel processing. A particular task is divided into a number of ~~of~~ parallel task or process.

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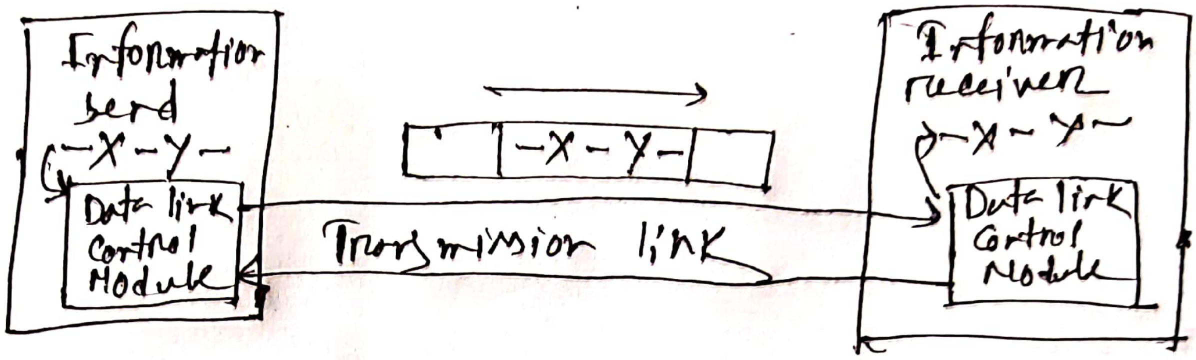
1(b)

b) Answer:



The Job of the data link layer is to make the communication on the physical link reliable and efficient.

The data link layer is the second layer in the OSI Model.

There are three main functions of the data link layer

→ Framing
→ Physical addressing
→ flow control
→ Error Control
→ Access Control.

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1. Framing: The data link layers divides the stream of bits received from the network layer into manageable data units called frames.

2. Physical ~~layer~~ addressing: If frames are to be distributed to different systems on the network, the data link layers adds a header to the frame. If the frame is intended for a system outside the senders network, the receiver address is the address of the device that connect the network to the next one.

3. flow Control: If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid ~~over heating~~ overwhelming the receiver

4. Error Control: The data link layer adds reliability to the physical layer by adding mechanism to detect and retransmit damaged or lost frame. Error control is normally achieved through a trailer added to the end of the frame.

5. Access Control: when two or more device are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time.

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1(C)

c) Answer: 2 million bytes $= 2000 \times 8$ Kb

$= 16000$ Kb

Time taken to download 56 ~~Kbps~~ Kbps channel $= \frac{16000}{56}$ second

$= 285.7$ second

Time taken to download at 1 Mbps Channel

$= \frac{16000 \text{ Kbps}}{1 \text{ Mbps}}$

$= \frac{16000 \text{ Kbps}}{1024 \text{ Kbps}}$

$= 15.6$ second

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As to the Question no - 2
2(a)

2)(a) Answer: The Nyquist theorem also known as the sampling theorem, it is a principle that engineers follow in the digitalization 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. The number of samples per second called the sampling rate or sampling frequency.

Any analog signal consists of components of various frequencies, the

Simpliest case is, the sine wave in which all signal energy is concentrated at one frequency

Suppose, the highest frequency components in Hertz, for a given analog signal is f_{max}, According to Nyquist theorem, the sampling rate must be at least $2f_{max}$, or twice the highest analog frequency, ~~the greater the bandwidth, it is all other factors are held constant~~. component. So, the sampling in an analog to digital converter is activated by a clock pulse generator. If the sampling rate less than $2f_{max}$, 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 ~~from~~ form by a DAC, false frequency components appear that were not in the original signal. This ~~Understan~~ undesireable condition is a form of distortion called aliasing.

2(b)

b) Answer:

```
                 1001101110          extra
        ---------------------------
10111 ) 10100 11110 0000   ← augmented dataword
        10111
        -----
         00111
         00000
         -----
          01111
          00000
          -----
           11111
           10111
           -----
            10001
            10111
            -----
             01100
             00000
             -----
              11000
              10111
              -----
               11110
               10111
               -----
                10010
                10111
                -----
                 01010
                 00000
                 -----
                  1010   remainder
```

dataword | Remainder.

Code word : | 1010011110 | 1010 |

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~~Ans to the Question no-02~~

~~2(a)~~

Sum of my student id = (1+7+2+4+7)
= 21 (odd)

So, second last digit of code word contains error.

Code word : | 1010011110 | 1010 |

error

```
         100110110
10111 ) 10100 11110 1000
        10111
        00111 1
        00000
         01111
         00000
          11111
          10111
           10000
           10111
            01100
            00000
             11001
             10111
              11100
              10111
               10110
               10111
                00010
                00000
                00010
```

Sir,

Syndrome, 0010

So, we can say

second bit corrupted

in receiver side

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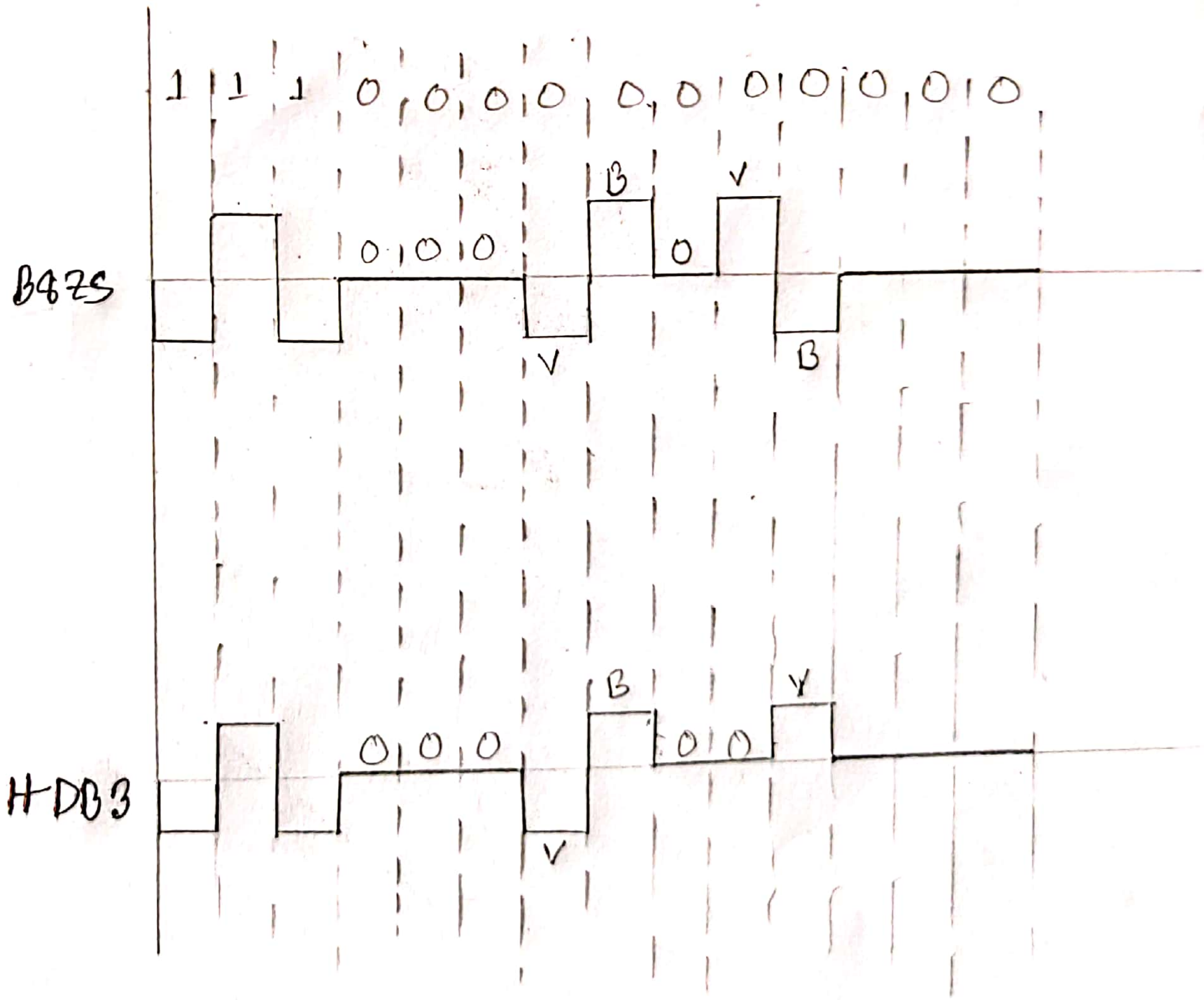
2(C)

2(C) Answer: last digit of my student id is '0', so, this is even. So, $\phi = 0$

So, sequence will be - 1 1 1 0 0 0 0 0 0 0 0 0 0 0 0



Ans to the Question no - 3

3(a)

a) Answer: We use QPSK instead of BPSK because QPSK has the advantage of having twice data rate compared 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:

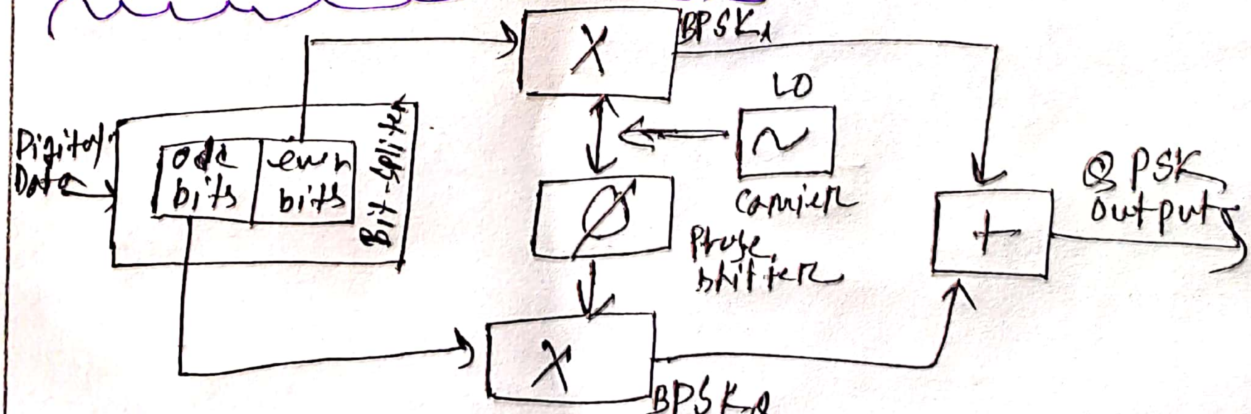


fig: QPSK

The ~~Simplify~~ simplicity of BPSK designers to use 2 bits at a time in each signal element thereby decreasing the baud rate and eventually required bandwidth. The scheme is called quadrature PSK or QPSK.

Because it was two seperate BPSK modulation. One is in the phase 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 T, the duration of each bit sent to the corresponding BPSK signal is 2T.

This means the bit to each BPSK signal has one half the frequency of original signal.

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3(b)

b) Answer:

According to Nyquist theorem, 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 engineers follow in the digitization of analog signals.

We know that,

$$S = \frac{1}{R} \times N$$

(i) $r = \log_2 2 = 1$

$\therefore S = \left(\frac{1}{1} \times 2000\right) = 2000$ baud

(ii) for ASK,

$r = \log_2 2 = 1 \therefore S = \left(\frac{1}{1} \times 4000\right)$ baud

$= 4000$ baud.

(iii) for QPSK,

$r = \log_2 4 = 2$

$\therefore S = \left(\frac{1}{2} \times 6000\right) = 3000$ baud

(iv) for 64-QAM;

$r = \log_2 64 = 6$

$\therefore S = \left(\frac{1}{6} \times 36000\right)$ baud

$= 6000$ baud.

Here we can see that 64-QAM has the highest baud rate while FSK has the lowest baud rate.

3(C)

~~3(C)~~

3(C) Answer:

(I)

(I) Looking at ~~it~~ a signal's frequency domain plot we can say this ~~sig~~ signal is periodic or not.

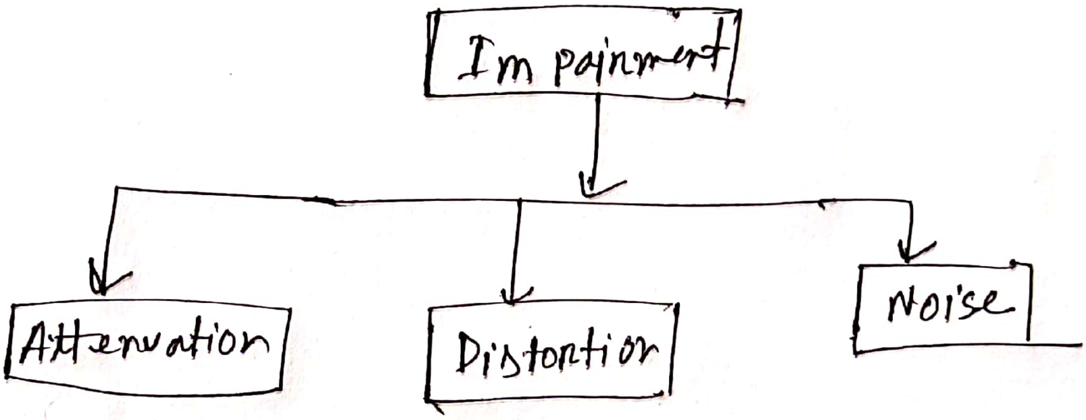
for periodic frequency domain plot the graph will be a straight line corresponding to the frequency. On the other hand, non-periodic signal will ~~have~~ be no straight line.

again ~~...~~.

~~So~~ If the wave form composed of specific narrow band frequencies then it is periodic otherwise non-periodic.

(ii)

(ii) Three causes of impairment are attenuation, distortion and noise



Attenuation: It means loss of energy. When a signal simple or composite travels through medium it loses some of its energy.
Distortion: Distortion means the signal changes its form or shape. Distortion occur in a composite signal made of frequencies.

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

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Ans to the Question no - 5

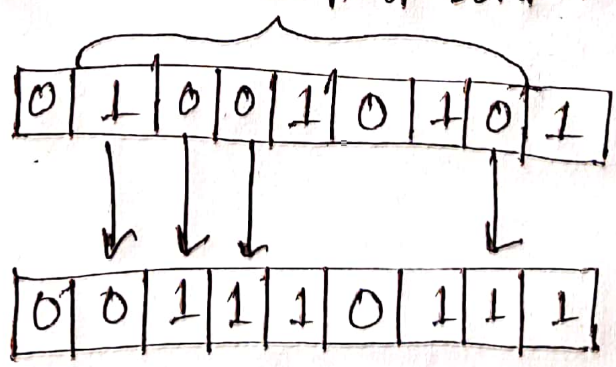
5(a)

5) **Answer:**

A burst error means that two or more bits in the data unit have changed.

A burst error is more ~~likely to be~~ affected by noise because the duration of noise is normally longer than the duration of 1 bit, which means that when noise affects data, it affects a set of bits.

Length of burst error (7 bits)



The correction errors is more difficult than the detection. In error detection, we are looking only to see if any error has occurred. The answer is simple yes or no. We are not even interested in the number of errors. a single bit error is the same of us as a burst error.

But in error correction, we need to know exact number of bits that are corrupted and more importantly, their location in the message. The number of errors and the size of the message is important factor here.
Suppose, we need to correct single error in 8 bit data we have to check 8 possible error location

Again if we check two errors in 8 bit data we need to consider 28 possibilities. So, error detection is easier than the correction.

5 (b)

b) given that,

burst rate = 2 ms

so, $= 2 \times 10^{-3}$ second.

(I) for 3000 bps,

bit corrupted = ~~3000 bit~~ $(3000 \times (2 \times 10^{-3}))$ bit

$= 6$ bits

(II) for 24 kbps,

bit corrupted $= (\cancel{24} \times (2 \times 10^{-3}))$ bit, with 24×10^3 written above the struck 24

$= 48$ bits

(III) for 100 kbps,

$$\text{bit corrupted} = \left(100 \times 10^3 \times \left(2 \times 10^{-3}\right)\right) \text{ bits}$$
$$= 200 \text{ bits}$$

(IV) for 100 Mbps,

$$\text{bits corrupted} = \left(100 \times 10^6 \times \left(2 \times 10^{-3}\right)\right) \text{ bits}$$
$$= 200000 \text{ bits.}$$

So, we can say, for same duration of noise, 3000 bps corrupt less but 100Mbps can effect most.

5(c)

5(c) Answer: In simple parity check code that can detect an odd number of errors.

~~A~~ In simple parity check,

$$r_0 = a_3 + a_2 + a_1 + a_0$$

Syndrome, $S_0 = b_3 + b_2 + b_1 + b_0 + r_0$

~~if~~, ~~s~~

In single parity check, sender sends

~~cod~~ Code word $= a_3\, a_2\, a_1\, a_0\, r_0$

and receiver receive,

$$b_3\, b_2\, b_1\, b_0\, r_0$$

So, if Syndrome $S_0 = 0$, there is no error.

but, single parity checker cannot detect even no of error

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for, two dimention parity check, it is the upgrade version of single parity check.
In single parity check K-bit data word changed to n-bit code word where $n = K+1$:
The extra bit is parity bit using Modulo-2
~~I~~ Single parity only detects error but cannot ~~Bot~~ correct errors.

Again in two-dimentional parity data organized in row columns, for each row & each column parity check is calculated, where as in single parity check we only calculated the row parity.

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Ans to the Question no - 7(a)

7(a)

~~A~~ answer

(i)

The size of data word say K & the Size of code word say n, r = redundancy

So, n is always greater than K

and, n = K + r.

(ii)

The remainder is always one bit smaller than the divisor.

(iii)

The degree of ~~generator~~ Polynomial generator is one less than the size of the divisor.

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(iv)

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

7(b)

My Id: 17 0204 070
second last digit of my ID is 7 which is odd.
So, the generated data word = 1101000

P.T.O

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Simulation of division in CRC encoder: