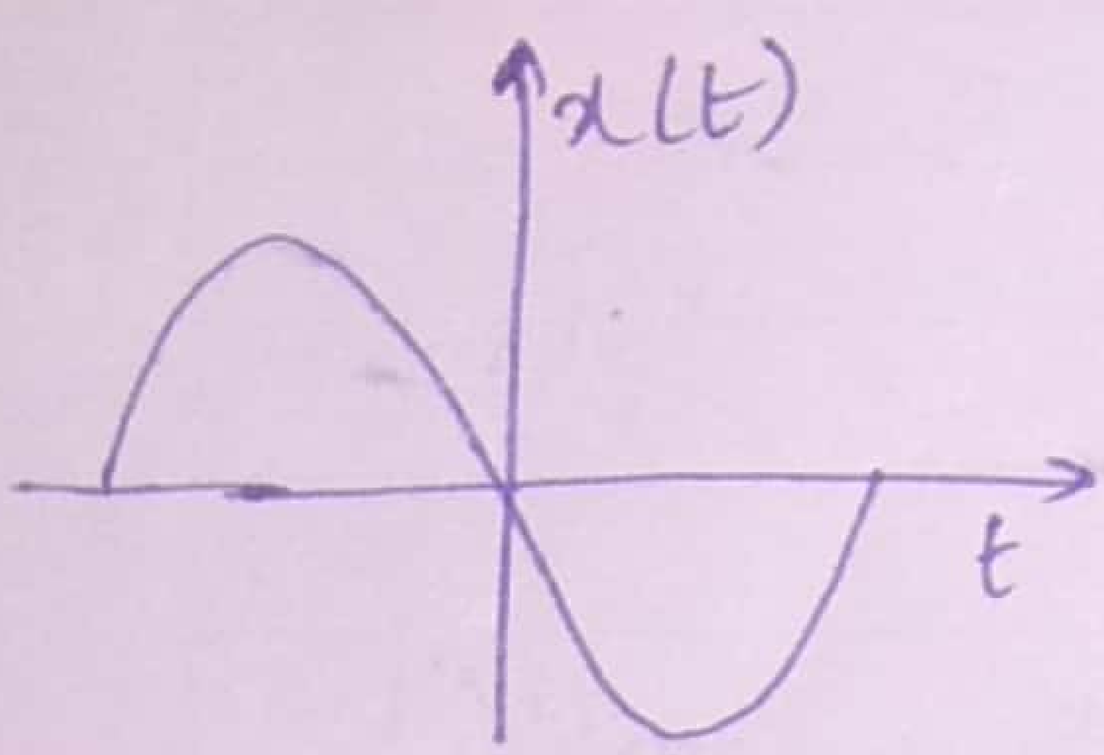
Suruchi
1819316

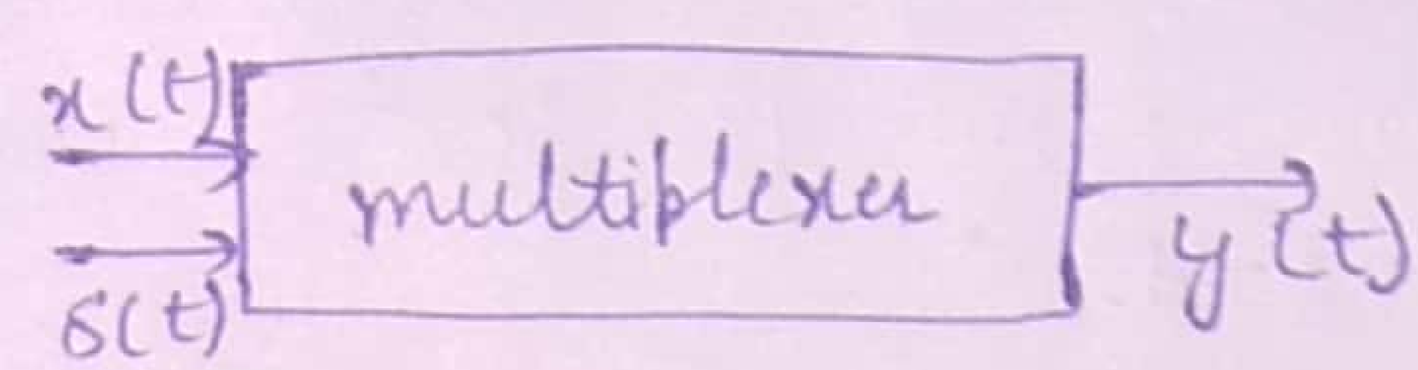
Ques : 1

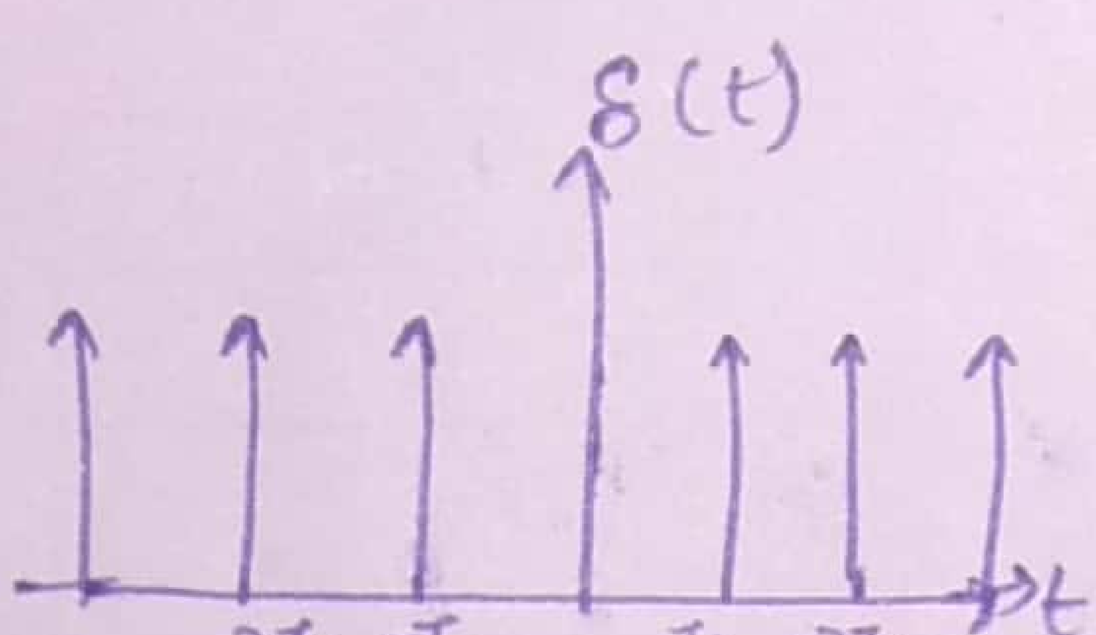
Sampling theorem:- A continuous time signal can be represented in its samples and can recovered back when sampling frequency f_s is greater that or equal to the twice the highest frequency component of message signal i.e.

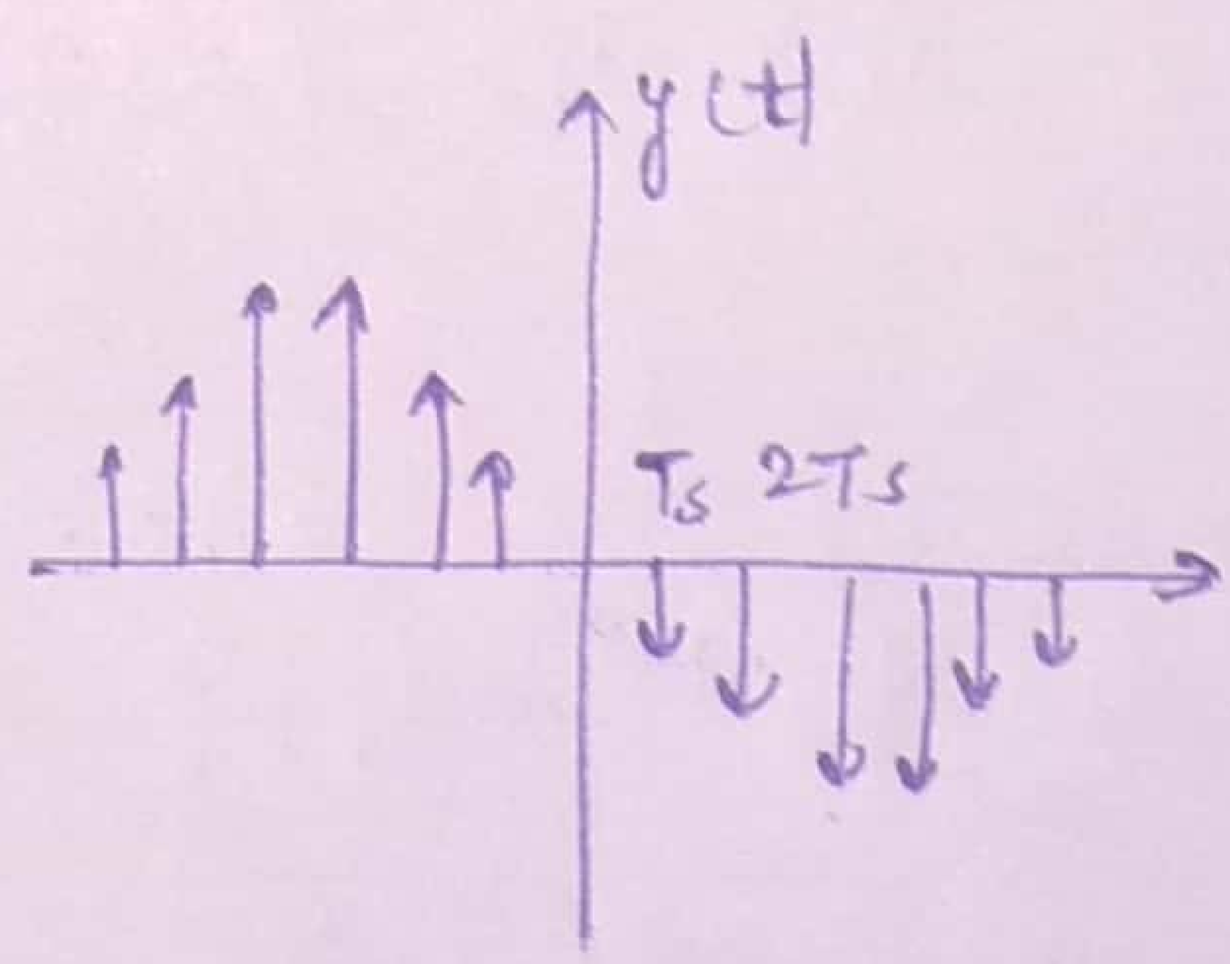
$$f_s \geq 2 f_m .$$

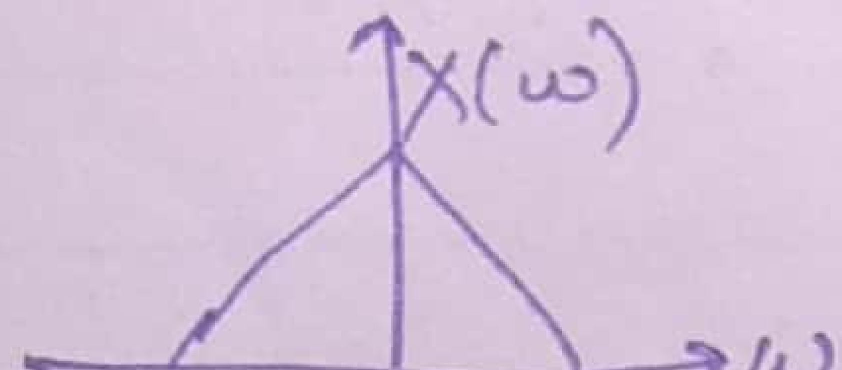
Proof:- Consider a continuous time signal $x(t)$. The spectrum of $x(t)$ is a band limited to f_m Hz i.e. the spectrum of $x(t)$ is zero for $|\omega| > \omega_m$.

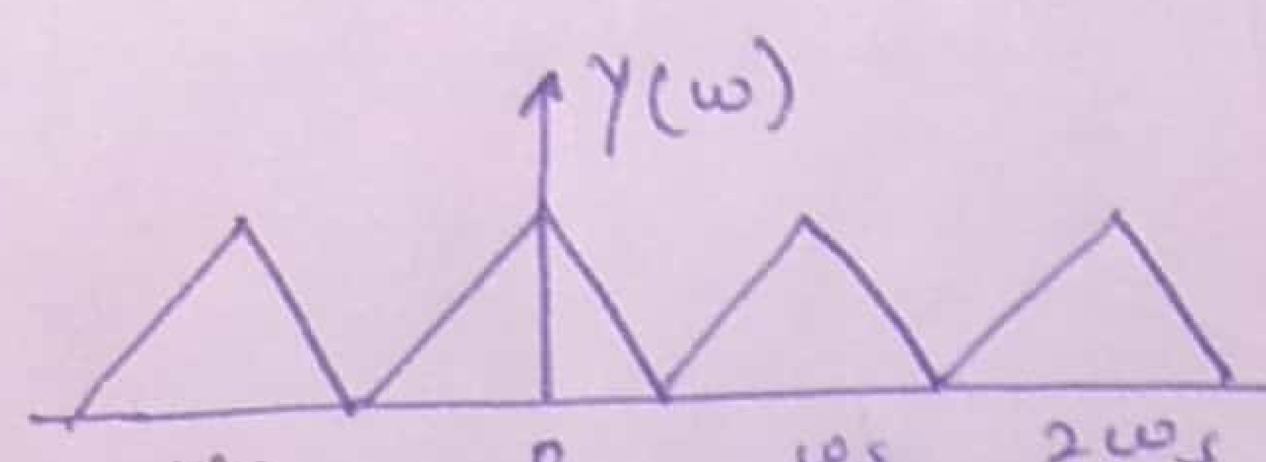












Here, you can observe that the sampled signal
$y(t) = x(t) \cdot \delta(t)$ ---- (1)

The trignometric fourier series representation of $\delta(t)$ is

$$\delta(t) = a_0 + \sum_{n=1}^{\infty} (a_n \cos n\omega_s t + b_n \sin n\omega_s t) \quad \text{---} (2)$$

where $a_0 = \frac{1}{T_s} \int_{-T/2}^{T/2} \delta(t)\, dt = \frac{1}{T_s} \delta(0) = \frac{1}{T_s}$

$$a_n = \frac{2}{T_s} \int_{-T/2}^{T/2} \delta(t) \cos n\omega_s dt = \frac{2}{T_2} \delta(0) \cos n\omega_s 0 = \frac{2}{T}$$

$$b_n = \frac{2}{T_s} \int_{-T/2}^{T/2} \delta(t) \sin \omega_s t\, dt = \frac{2}{T_s} \delta(0) \sin n\omega_s 0 = 0$$

substitute above values in equation(2).

$$\therefore \delta(t) = \frac{1}{T_s} + \sum_{n=1}^{\infty} \left(\frac{2}{T_s} \cos n\omega_s t + 0\right).$$

substitute $\delta(t)$ in equation(1).

$$y(t) = x(t) \cdot \delta(t).$$

$$= x(t) \left[\frac{1}{T_s} + \sum_{n=1}^{\infty} \left(\frac{2}{T_s} \cos n\omega_s t\right)\right.$$

$$= \frac{1}{T_s} \left[x(t) + 2 \sum_{n=1}^{\infty} (\cos n\omega_s t)\, x(t)\right]$$

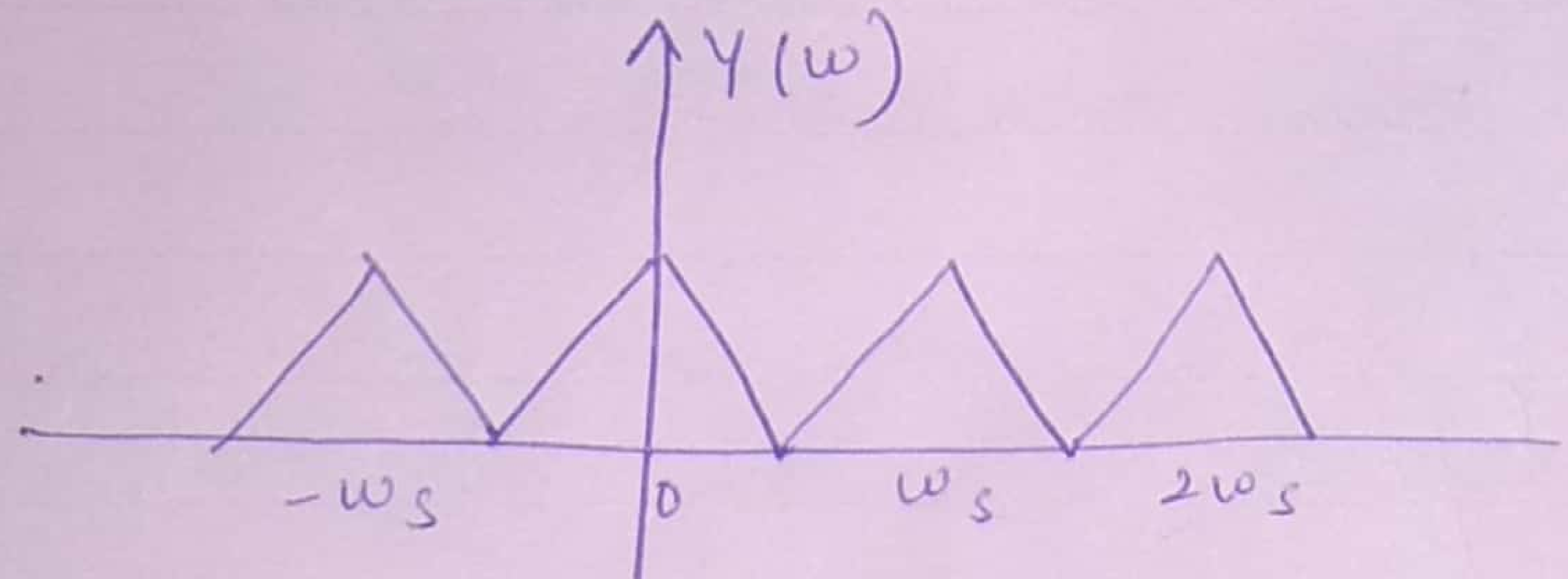
$$y(t) = \frac{1}{T_s}\left[x(t) + 2\cos \omega_s t \cdot x(t) + 2\cos 2\omega_s t \cdot x(t) + 2\cos 3\omega_s t \cdot x(t) \text{----}\right]$$

Take fourier series on both sides

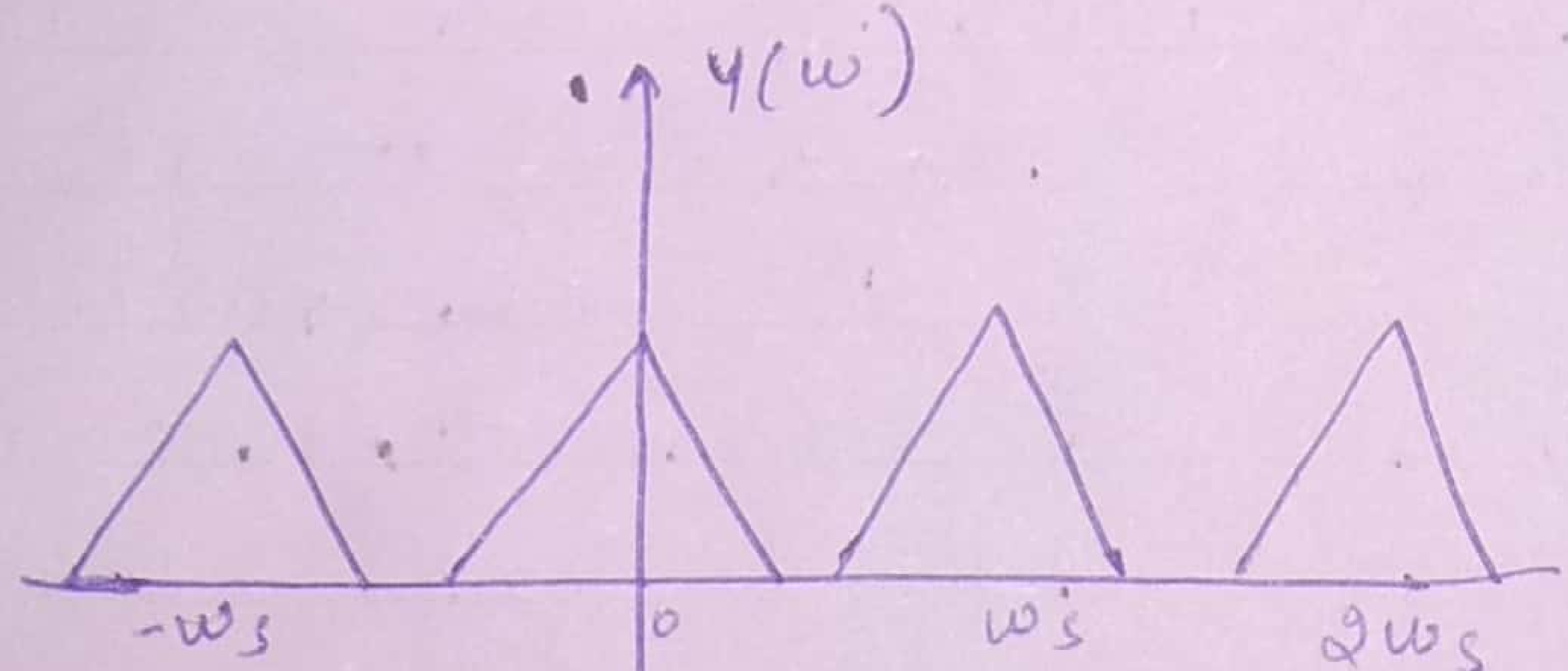
$$Y(\omega) = \frac{1}{T_s}\left[X(\omega) + X(\omega-\omega_s) + X(\omega+\omega_s) + X(\omega-2\omega_s) + X(\omega+2\omega_s) + \cdots\right]$$

$$\therefore \; Y(\omega) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} X(\omega - n\omega_s) \quad \text{where } n = 0, \pm 1, \pm 2, \ldots$$

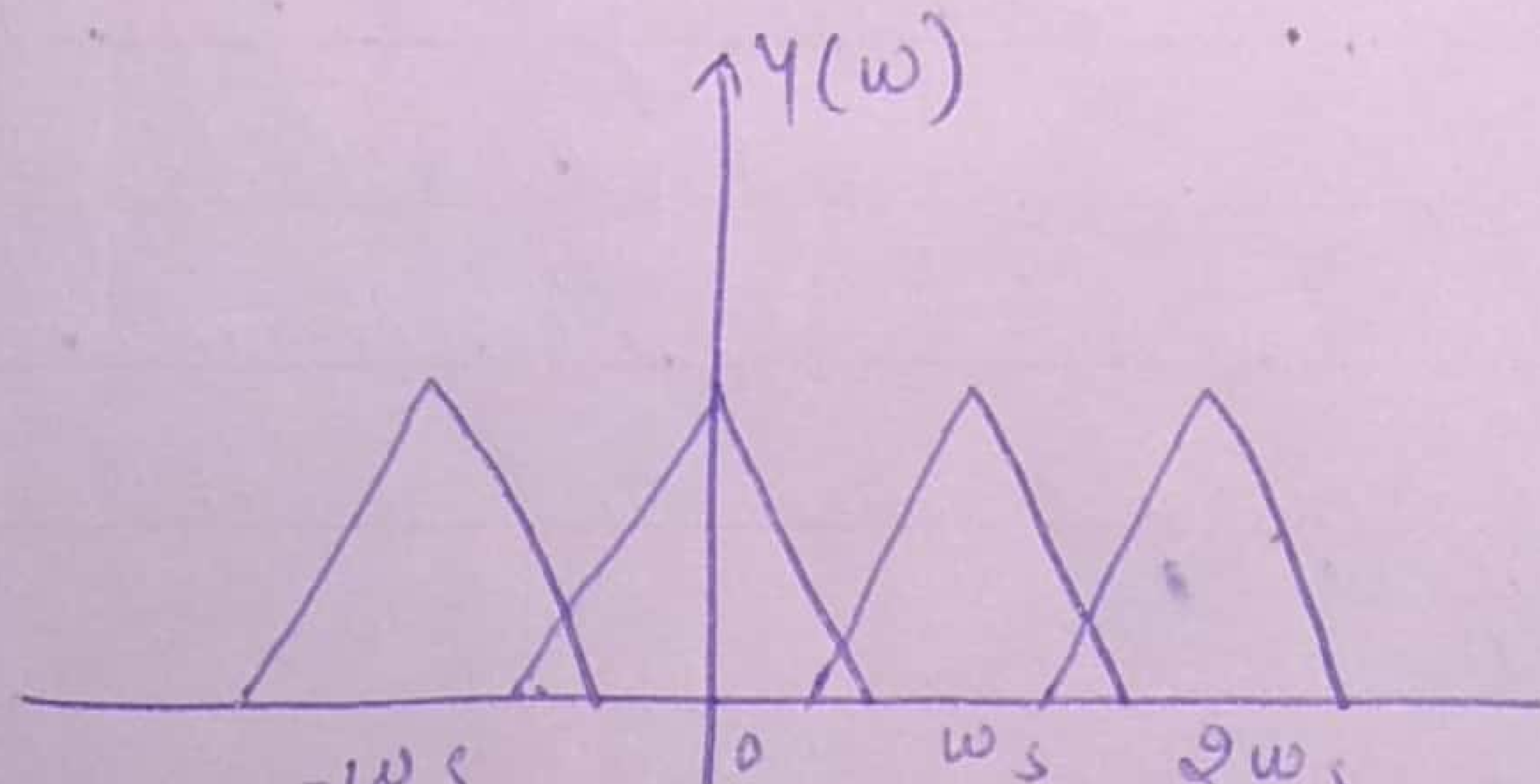
Possibility of sampled frequency spectrum with different conditions is given by the following diagrams :-



$f_s = 2f_m$
perfect sampling



$f_s > 2f_m$
over sampling,



$f_s < 2f_m$
under sampling.

Suruchi
1219316

Quantization is done because, we have an analog signal and we want to process it, store it and we want to analyze it. This all can be done when we convert our signal into digital format. First the signal is sampled and the time axis becomes discreet and the signal becomes discreet. Then the amplitude is converted into discrete levels which is called quantization and this converts the discreet signal into an digital signal. Now if we dont quantize the signal then the amplitude can take values from 0 to infinity. Suppose we want to store such data and we use binary digits to represent these signals, then we would require a large set of numbers, and memory space to store them. that is why we quantize the amplitude by making finite levels and then approximating the amplitude in these levels. if we increase the levels the approximation becomes closer to the real signal. but this has an effect on the memory also.

Lets assume we have 4 Binary digits we can have 16 levels for the amplitude to take. This will have quantization error. But if we increase the digits to 64, we will get more levels at the same time the space required to hold these binary digits will also increase.

Suruchi
1219316

Ques: 2

Encoding is a process of using various patters of voltage or current levels to represent 1s and 0s of the digital signals on the transmission link

The common types of line encoding are:- unipolar, polar, bipolar, and Manchester.

Encoding Scheme:-
The data encoding scheme is divided into the following types, depending upon the type of data conversion.

→ Analog data to Analog signals:-
The modulation techniques such as amplitude modulation, frequency modulation and phase modulation of analog signals, fall under this category.

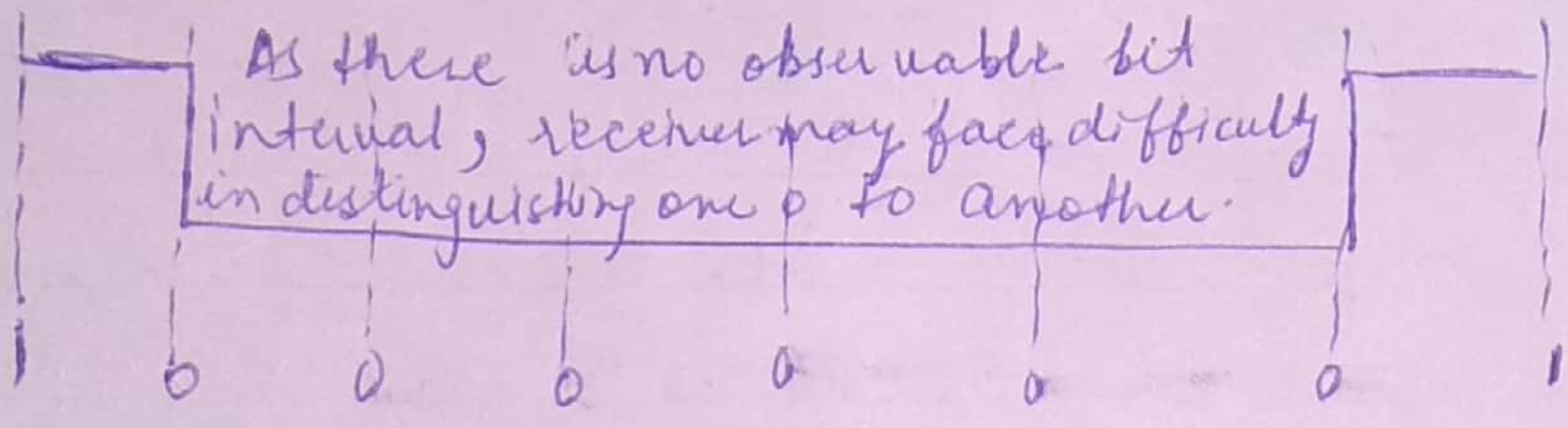
→ Analog data to digital signals-
This process can be termed as digitalization, which is done by pulse code Modulation PCM. Hence, it is nothing but digital modulation. As we have already discussed, samping and quantization are important factors in this. Delta modulation gives a better output than PCM.

Sruchi
12193 16

→ Digital data to digital signals:-
These are in this section. There are several ways to map digital data to digital signals.

• Non Return to zero (NRZ)

NRZ code has 1 for high voltage level and 0 for low voltage level. The main behaviour of NRZ codes is that the voltage level remains constant during bit interval. The end or start of a bit will not be indicated and it will maintain the same voltage state, if the value of the previous bit and the value of the present bit are same.



If the above example is considered as there is a long sequence of constant voltage level and the clock synchroniz-
-ation may be lost due to the absence of bit interval, it becomes difficult for the receiver to differentiate between 0 and 1.

There are two variations in NRZ namely-

→ NRZ-L NRZ-level

→ NRZ-I NRZ- inverted.

Suruchi
1219316

• Bi-phase Encoding

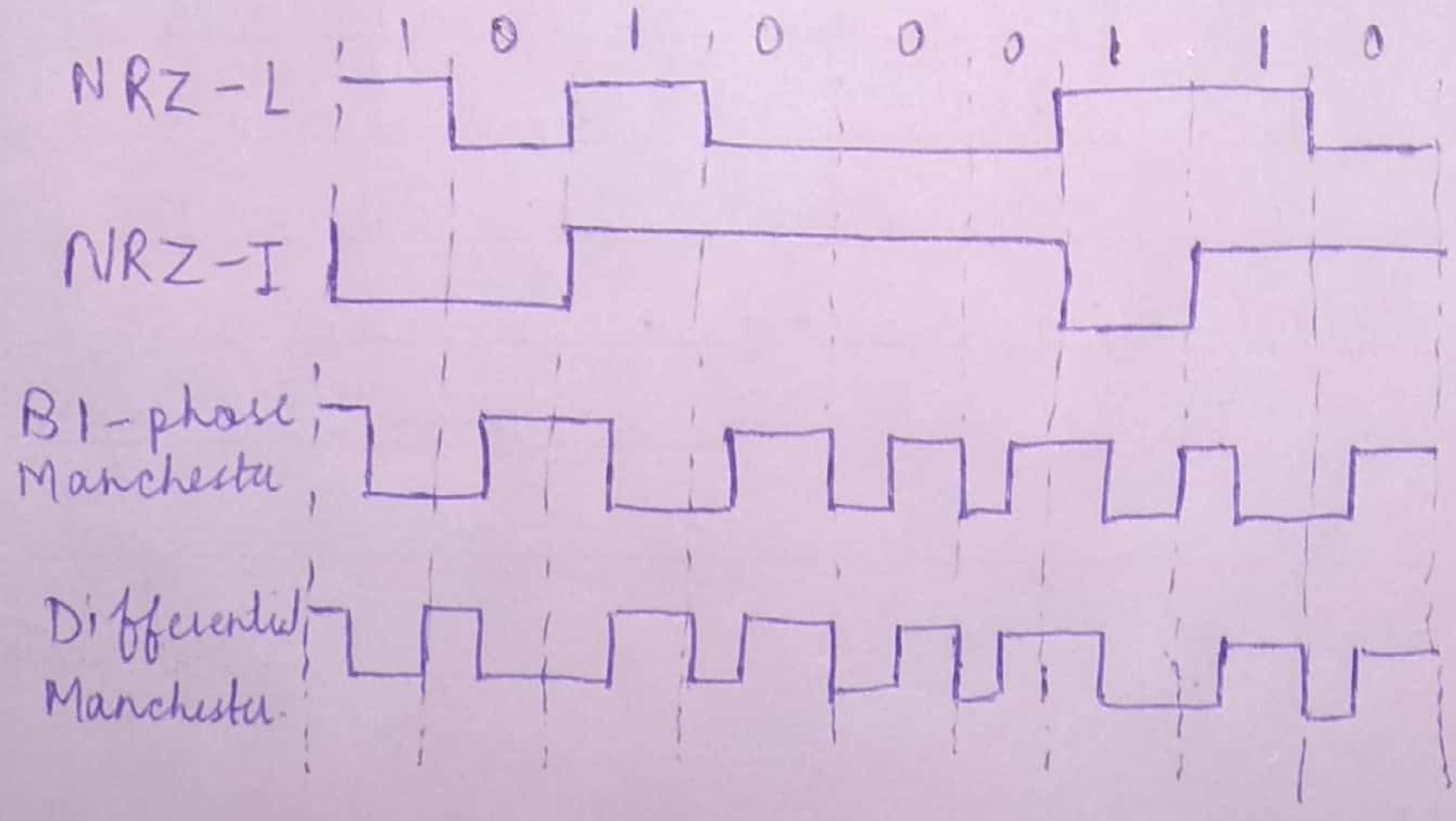
The signal level is checked twice for every bit time, both initially and in the middle. Hence, the clock rate is double the data transfer rate and thus the modulation rate is also doubled. The clock is taken from the signal itself. The bandwidth required for this encoding is greater.

There are two types of Bi-phase encoding:-

- Bi-phase Manchester.
- Differential Manchester.

Biphase Manchester:

In this type of coding, there always occurs a transition in the middle of the bit interval. If there occurs a transition at the beginning of the bit interval, then the input bit is 0, if no transition occurs at the beginning of the bit interval, then the input bit is 1.



Suruchi
1219316

Block Coding:-

Among the types of block coding, the famous ones are 4B/5B encoding and 8B/6T encoding. The number of bits are processed in different manners, in both of these processes.

4B/5B encoding:- In manchester encoding, to send the data, the clocks with double speed is required rather than NRZ coding. Here, as the name implies, 4-bits of code is mapped with 5-bits, with a minimum number of 1 bits in the group.

The basic idea of selecting a 5-bit code is that, it should have one leading 0 and it should have no more than two trailing 0s. Hence, these words are chosen such that two ~~transitio~~ transactions take place per block of bits.

8B/6T encoding:- We have used two voltage levels to send a single bit over a single signal. But if we use more than 3 voltage levels, we can send more bits per signal.

Suruchi
1219316

Ques 3

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of a smaller value Δ, such a modulation is termed as delta modulation.

Features of delta modulation :-

→ An over sampled input is taken to make full use of the signal correlation.
→ The quantization design is simple.
→ The input sequence is much higher than the Nyquist rate.
→ The quality is moderate
→ The design of the modulator & the demodulator is simple
→ The stair-case approximation of output waveform.
→ The step-size is very small, i.e. Δ delta.
→ The bit rate can be decided by the user
→ This involves simpler implementation

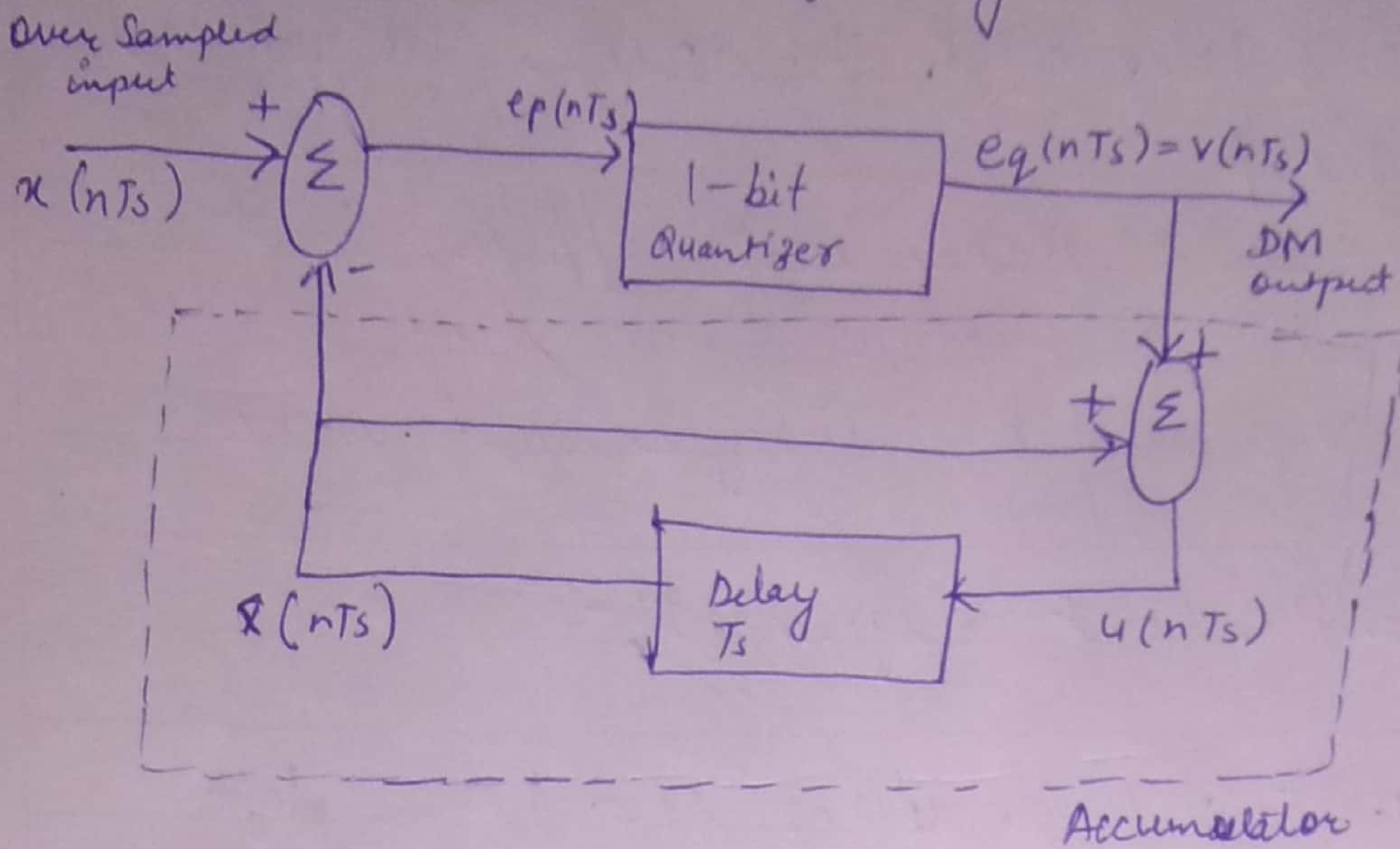
Delta modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

The Delta Modulator comprises of a 1-bit quantizer & a delay circuit along with two summer circuit.

Sruchi
1219316

Following is the block diagram of a delta modulator:



The predictor circuit in DPCM is replaced by a simple delay circuit in DM.

→ $x(nT_s)$ = over sampled input.

→ $ep(nT_s)$ = summer output & quantizer input.

→ $eq(nT_s)$ = quantizer output = $v(nT_s)$

→ $\hat{x}(nT_s)$ = output of delay circuit.

→ $u(nT_s)$ = input of delay circuit.

using these notations, now we shall try to figure out the process of delta modulation.

$$ep(nT_s) = x(nT_s) - \hat{x}(nT_s) \quad - \text{eq}^n (1)$$

$$= x(nT_s) - u([n-1]T_s)$$

$$= x(nT_s) - [\hat{x}[[n-1]T_s] + v[[n-1]T_s]] \quad \text{eq}^n (2).$$

Further $v(nT_s) = eq(nT_s) = S \cdot sig \cdot [ep(nT_s)]$ — $\text{eq}^n (3)$

$$u(nT_s) = \hat{x}(nT_s) + eq(nT_s).$$

Sruchi
1819316

where,

→ $\hat{x}(nT_s)$ = the previous value of the delay circuit.

→ $e_q(nT_s)$ = quantizer output = $v(nT_s)$

Hence,

$$u(nT_s) = u([n-1]T_s) + v(nT_s) \quad - \text{eq } (4).$$

Which means,

The present input of the delay unit =

The previous output of the delay unit + the present quantizer output

Assuming zero condition of Accumulation,

$$u(nT_s) = S\sum_{j=1}^{n} \operatorname{sig}[e_p(jT_s)]$$

Accumulated version of DM output = $\sum_{j=1}^{n} v(jT_s)$ — eqn (5)

Now, note that

$$\hat{x}(nT_s) = u([n-1]T_s)$$
$$= \sum_{j=1}^{n-1} v(jT_s) \quad — \text{eq}^n (6)$$

Delay unit output is an accumulator output lagging by one sample.

From equation (5) & (6), we get a possible structure for the demodulator.

A stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (Δ). The output quality of the waveform is moderate.

Advantages of Delta modulation :-

- Design is easy & simple.
- It is a 1 bit quantizer
- Modulator & demodulator can be designed easily.
- In delta modulation, the quantization design is very simple.
- The bit rate can be designed by the user.

Disadvantages of Delta modulation :-

- When the value of delta is small, slope overload distortion is seen, which is a type of noise.
- when the value of delta is large, granular noise is seen, which is a type of noise.

Uses.

- Delta modulation is more useful in system where timely data delivery at the receiver is more important than the data quality.
- This modulation is applied to ECG waveform for database reduction & real time signal processing.
- For analog to PCM encoding this modulation method is used.
- Delta modulation is applied in television system.