

UNIT - IV

Pulse Modulation

Types of Pulse modulation:-

There are two types. They are;

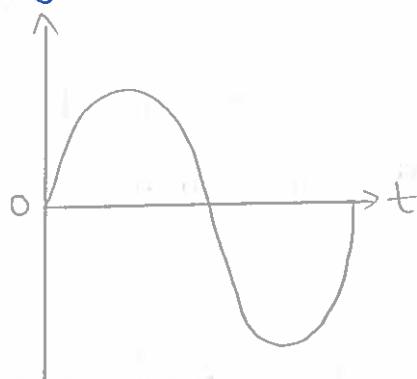
- i) Pulse Analog modulation \rightarrow PAM, PDM (or) PPM, PFM
- ii) Pulse Digital modulation \rightarrow PCM, DPCM, DM, ADPCM.

Pulse modulation Definition:-

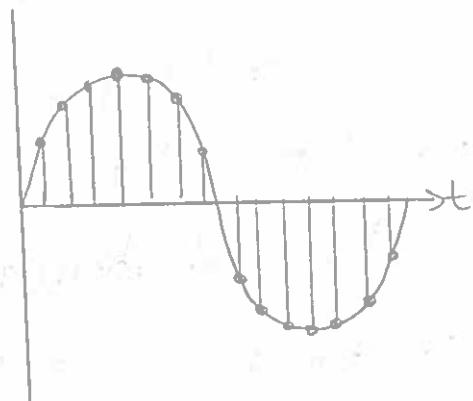
The Sampling process is used to convert continuous time signal into discrete time signal.

Sampling theorem:-

A continuous time signal can be completely represented in its samples and recovered back if the sampling frequency $f_s \geq 2\omega$. Here, f_s is Sampling frequency and ' ω ' is the maximum frequency present in the signal.

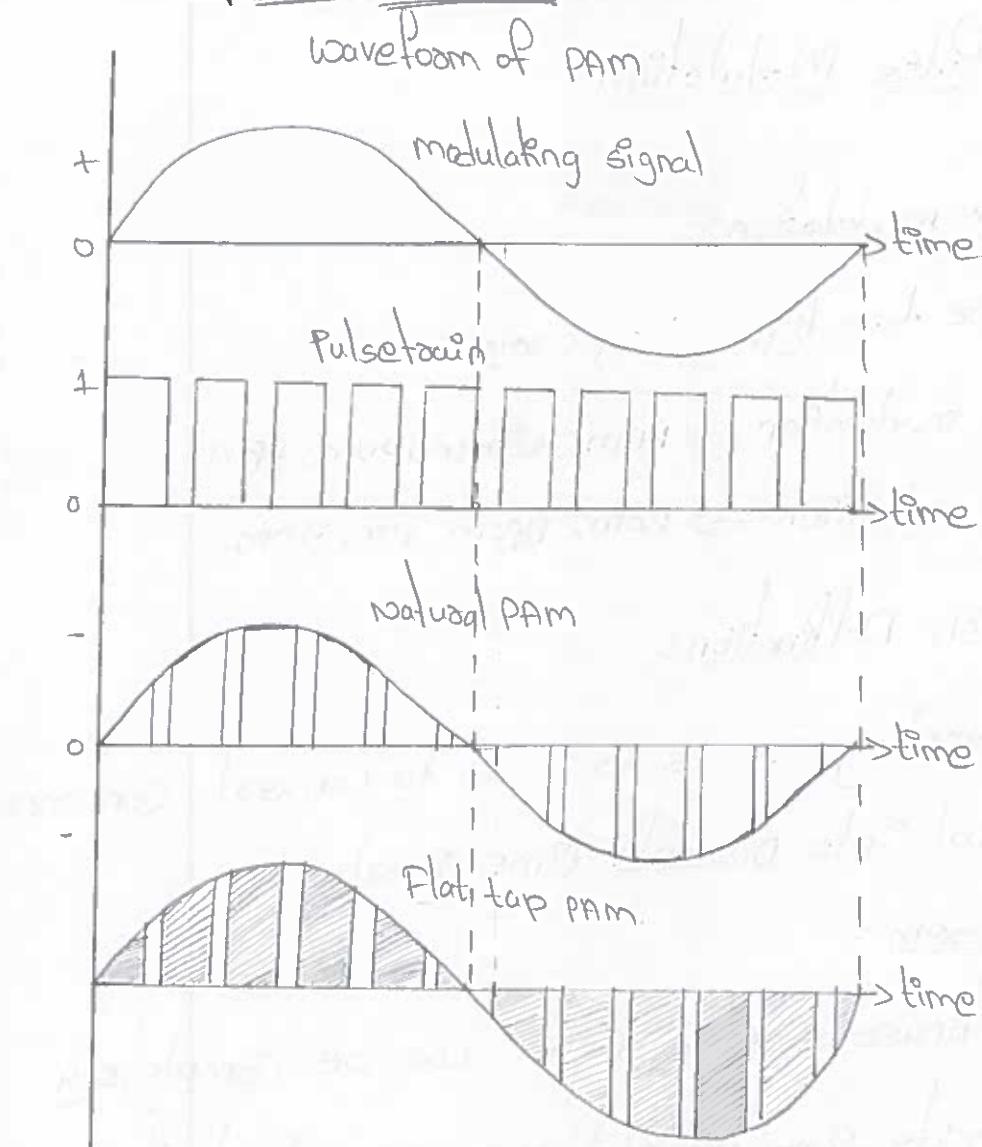


Continuous Signal



Discrete time signal.

Pulse Amplitude modulation :-



Amplitude of the pulse is varied proportional to the Amplitude of modulating signal or message signal.

Pulse width modulation:- (Pwm)

Width of the pulse is varied proportional to the Amplitude of modulating signal or message signal.

Pulse Position modulation (ppm)

Position of the pulse is varied proportional to the Amplitude of modulating signal or message signal.

Composition of FDM and TDM:-

FDM	TDM
* In frequency Division multiplexing total available b.w deviating based on the frequency slots.	* In TDM, the total available bandwidth deviating based on time slots.
* It requires modulators, filters, and demodulators.	* It requires, Commutator at the transmitter end, and a Distributor working in perfect synchronization with commutator at the receiving end.
* Synchronization between transmitter and receiver not required.	* Perfect synchronization b/w transmitter and receiver is required.
* It requires complex circuit at transmitter and Receiver.	* Perfect synchronization b/w * It does not requires very complex circuit.
* Cross talk is present.	* Cross talk is not present.
* It is used in analog signal transmission.	* It is used in Digital signal transmission.

PCM Generation and Reconstruction:-

The analog message signal $s(t)$ is first passed through the low pass filter at cut off frequency f_m Hz. This low pass filter blocks all the frequency components above the f_m Hz. The sample and hold circuit then samples this signal at the rate of f_s i.e., $f_s \geq 2f_m$. The output of sample and hold circuit is given by $n(t)$. This signal is also discrete in time & continuous.

In Amplitude A 'Q' level quantizer compares input nTs with its fixed digital levels. It gives any one of the digital level to nTs with its fixed digital levels. O/p at quantizer is given to binary encoder. This encoder converts o/p signal to binary codes.

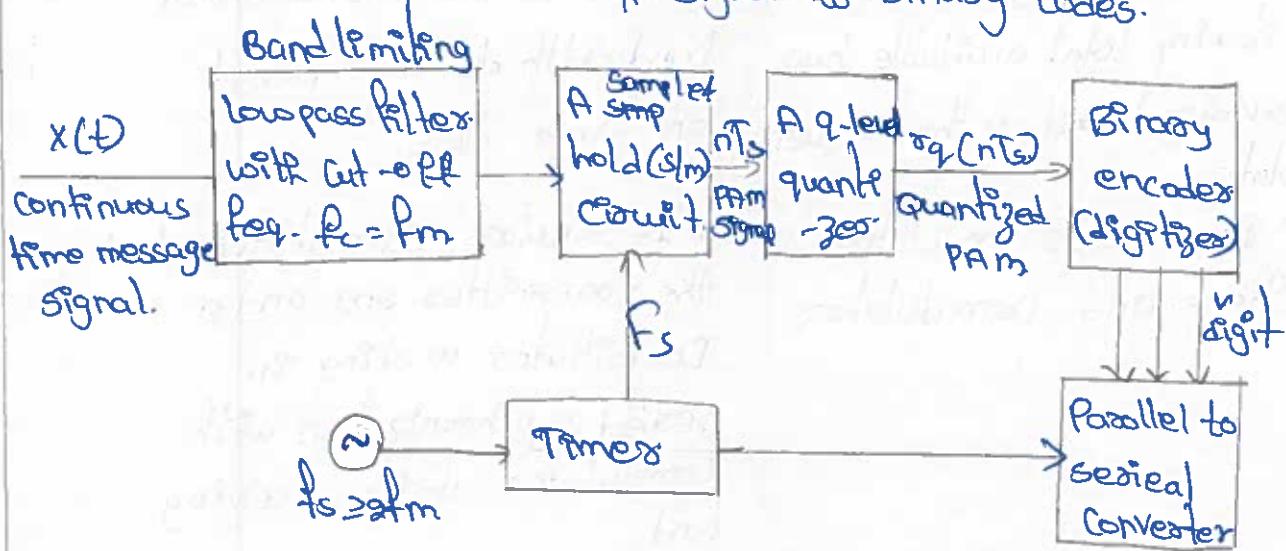


Fig:- A practical PCM Generator.

Quantization noise (or) Quantization Error :-

Quantization:- It is the process of approximation. The samples are approximated or rounded off to the nearest quantization level.

Quantization Error:-

It is the difference between quantized signal and original signal. Its maximum value is $\pm \frac{\Delta}{2}$

$$\text{Quantization error, } E = \hat{x}_q(nT_s) - x(nT_s)$$

Quantization error reducing by reducing stepsize Delta (Δ). This is possible by increasing the no. of quantization levels.

Classification of quantization process:-

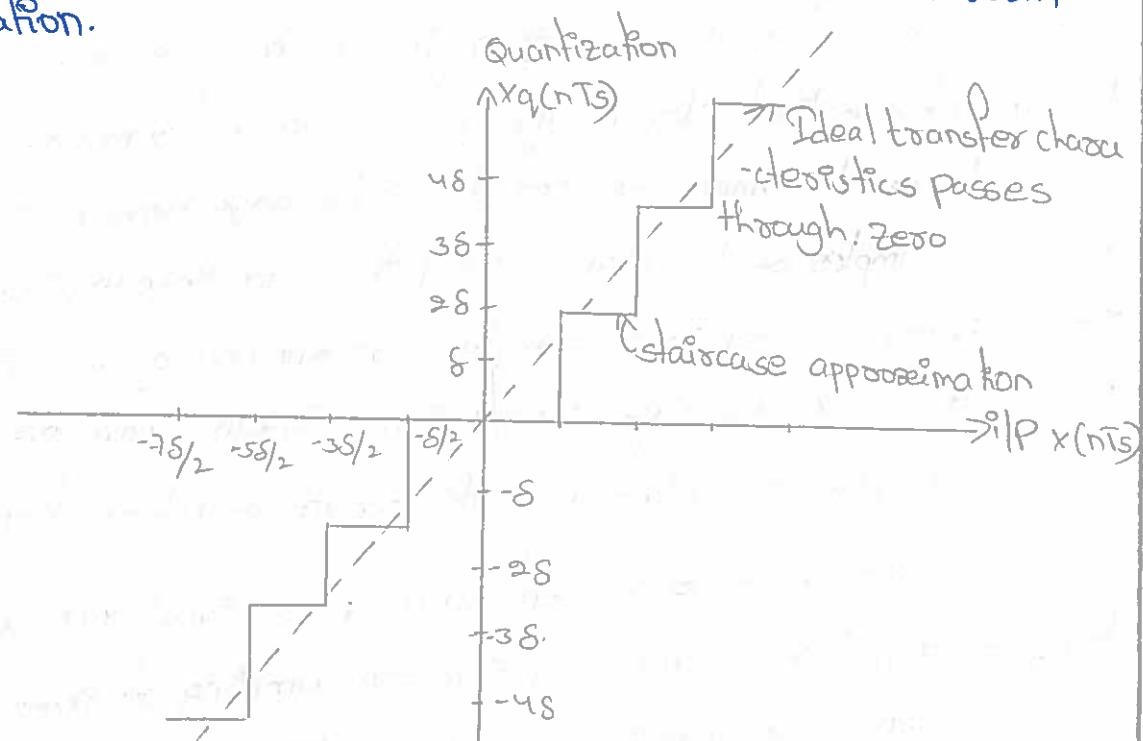
There are two types of quantization process. They are;

i) Uniform Quantization:

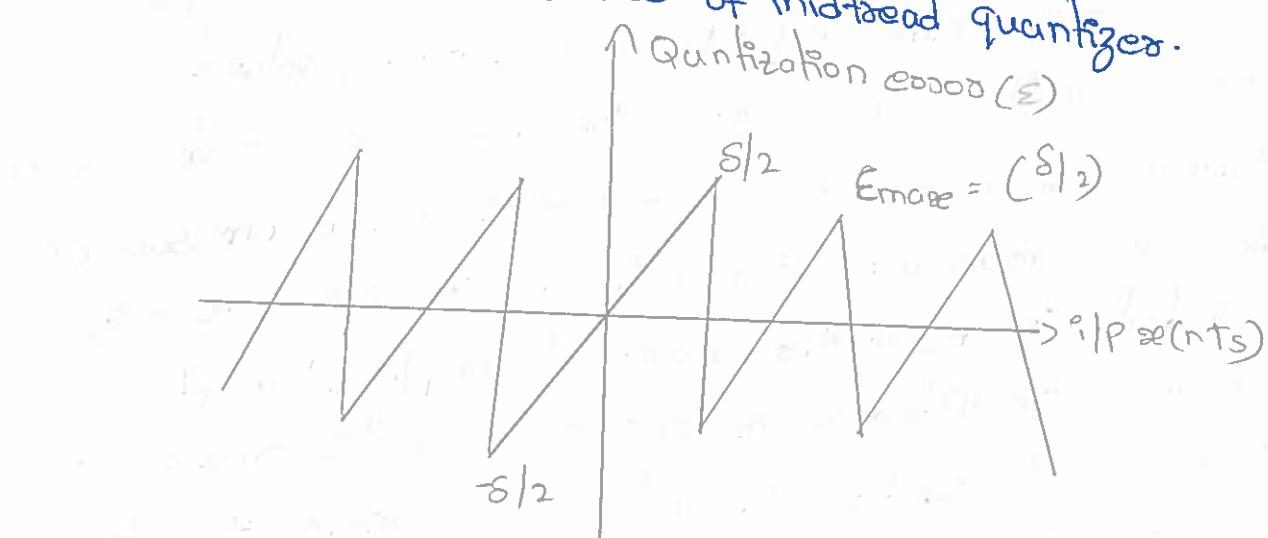
If the stepsize remains constant throughout the input range. Then the quantization is known as Uniform quantization.

ii) Non Uniform Quantization:

If the stepsize varies depending on the size of input. Then the quantization is known as non-uniform quantization.



a) Quantization characteristics of mid-tread quantizer.



b) Quantization Error

Companding :-

The compression of signal at the transmitter and expansion at the receiver is called combinedly as Companding.

Companding = Compressing + Expanding.

Differential Pulse Code Modulation (DPCM) :-

It may be observed that the samples of a signal are highly correlated with each other. This is because any signal does not change fast. This means that its value from present sample to next sample does not differ by large amount. The adjacent samples of the signal carry the same information with little difference. When these samples are encoded by a standard PCM system, the resulting encoded signal contains some redundant information. Fig: 10.19 illustrates this redundant information.

Fig: 10.19 shows a continuous time signal $s(t)$ by dotted line. This signal is sampled by flat top sampling at intervals T_s , $2T_s$, $3T_s$..., nT_s . The sampling by using 3 bit PCM. The sample is quantized to the nearest digital each level as shown by small circles in the figure 10.19. The encoded binary value of each sample is written on the top of the samples. We can observe from Fig. 10.19. that information can be carried only by one sample. But three samples are carrying the same information means it is redundant. Consider another example of sample taken at $9T_s$ and $10T_s$. The difference between these samples only due to last but bit and first two bits are redundant, since they do not change.

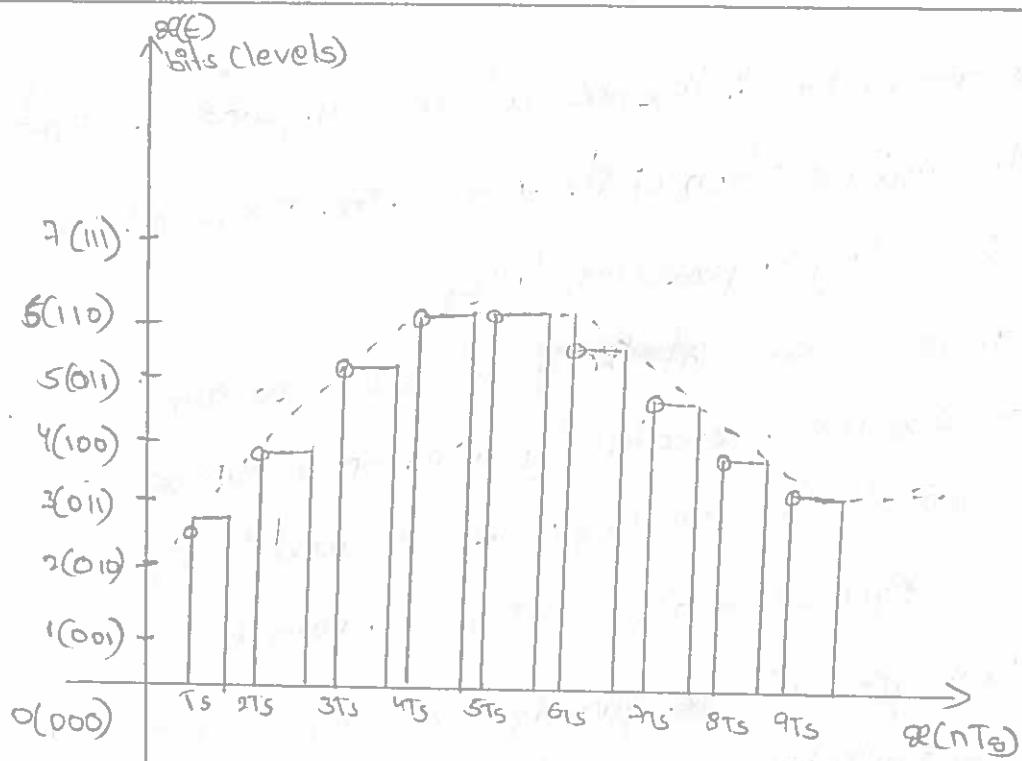


Fig 10.19: Illustration of redundant information in PCM.

If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit one sample will also be reduced. This type of digital pulse modulation scheme is known as Differential Pulse Code Modulation.

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples.

This is known as Prediction error and it is denoted by $e(nTs)$. It can be defined as:

$$e(nTs) = s(nTs) - \hat{s}(nTs)$$

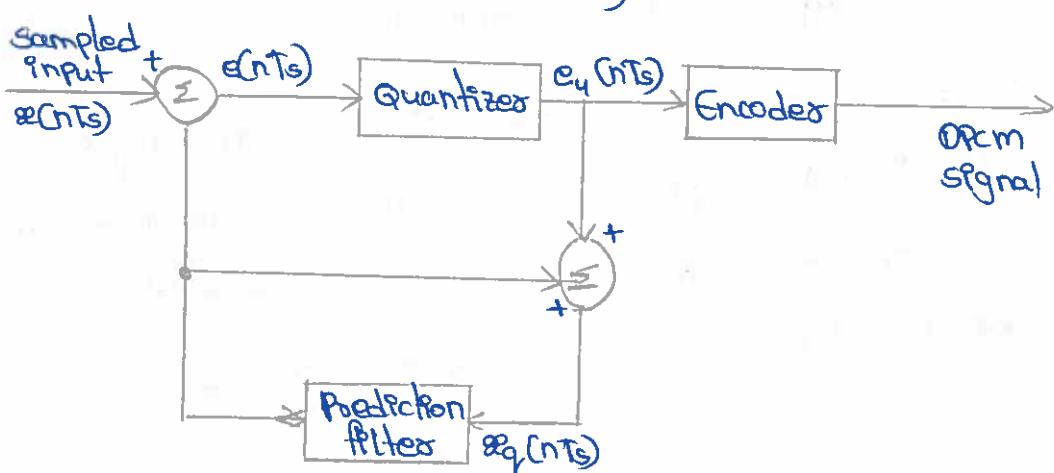


Fig. 10.20. A Differential pulse code modulation transmitter.

Thus, $e(nT_s)$ is the difference between unquantized input sample $\hat{x}(nT_s)$ and prediction of its $\hat{x}(nT_s)$. The predicted value is produced by using a prediction filter.

From we can see that the quantized error signal $q(nT_s)$ is very small and can be encoded by using small number of bits. Thus number of bits per sample are deduced in DPCM.

$$e_q(nT_s) = e(nT_s) + q(nT_s) \quad (10.5_1)$$

Hence $q(nT_s)$ is the quantization error. As shown in figure 10.20, the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantizer output.

$$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s) \quad (10.5_2)$$

Putting the value of $e_q(nT_s)$ from equation 10.5₁ in the above equation we

$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s) \quad 10.5_3$$

Equation 10.5₀ is written as

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

$$e(nT_s) + \hat{x}(nT_s) = x(nT_s)$$

\therefore Putting the value of $e(nT_s) + \hat{x}(nT_s)$ from above equation (10.5₃) we get,

$$x_q(nT_s) = x(nT_s) + q(nT_s).$$

Thus the quantized version of the signal $x_q(nT_s)$ is the sum of original sample value and quantization error $q(nT_s)$. The quantization error can be positive or negative. Thus equation (10.5₃) does not depend on the prediction filter characteristics.

Delta modulation (cm):-

We have observed in PCM that it transmits all the bits which are used to code the sample. Hence signaling rate and transmission channel bandwidth are large in PCM. To overcome this problem, Delta modulation is used.

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value.

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Thus for each sample, only one binary bit is transmitted. Fig 10.14 shows the analog signal $\hat{s}(t)$ and staircase approximated signal by the delta modulator.

Thus, the principle of delta modulation can be explained by following equations

The error between the samples value of $s(t)$ and last approximated sample is given as,

$$e(nT_s) = s(nT_s) - \hat{s}(nT_s)$$

Here, $e(nT_s)$ = error at present sample.

$s(nT_s)$ = Sampled signal of $s(t)$

$\hat{s}(nT_s)$ = last sample approximation of the staircase waveform.

If we assume $u(nT_s)$ as the present sample approximation of staircase output,

$$\text{then, } u(nT_s) = \hat{s}(nT_s)$$

= last sample approximation of staircase waveform.

Let us define a quantity $b(nT_s)$ in such a way that,

$$b(nT_s) = \text{sgn} [e(nT_s)]$$

This means that depending on the sign of error $e(nT_s)$, the sign of step size s is decided. In other words, we can write.

$$\begin{aligned} b(nT_s) &= +s && \text{if } e(nT_s) \geq \hat{e}(nT_s) \\ &= -s && \text{if } e(nT_s) < \hat{e}(nT_s) \end{aligned}$$

Also if $b(nT_s) = +s$ binary '1' is transmitted

and if $b(nT_s) = -s$ a binary '0' is transmitted.

T_s = Sampling interval.

The summer in the accumulator adds quantized output ($\pm s$) with the previous sample approximation. This gives Present sample approximation. i.e.,

$$u(nT_s) = u(nT_s - T_s) + [\pm s]$$

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

The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s . The sampled $e(nT_s)$ and staircase approximated signal $\hat{e}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of $+s$ or $-s$. If the step size is $+s$, then binary '1' is transmitted and if it is $-s$, then binary of '0' is transmitted.

The accumulator and low-pass filter are used. The accumulator staircase approximated signal output and is delayed by one sampling period T_s . It is then added to the I/P signal.

If input is binary '1' then it adds +s step to the previous output. If input is binary '0' then one step 's' is subtracted from the delayed signal. the low-pass filter has the cutoff frequency equal to highest frequency in $\alpha(t)$.

Advantages of Delta Modulation:-

The delta modulation has certain advantages over PCM as under:

1. Since, the delta modulation transmits only one bit for one sample, therefore the signaling rate and transmitted only channel bandwidth is quite small for delta modulation compared to PCM.
2. The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter required in delta modulation.

Disadvantages of Delta modulation:-

The delta modulation has two major drawbacks as under:

- i) Slope overload distortion
- ii) Granular or Idle noise.

Adaptive Dm:-

To overcome the quantization errors due to slope overload and granular noise, the step size is made adaptive to variations in the i/p signal $\alpha(t)$. Particularly in the steps segment of the signal $\alpha(t)$, the step size is increased.

The adaptive delta modulation can take continuous changes in step size or discrete changes in step size.

The previous $\delta/I/P$ and present $\delta/I/P$ decides the step size. It is then applied to an accumulator which builds up staircase waveform. The low-pass filter then smoothens out the staircase waveform to reconstruct the original signal.

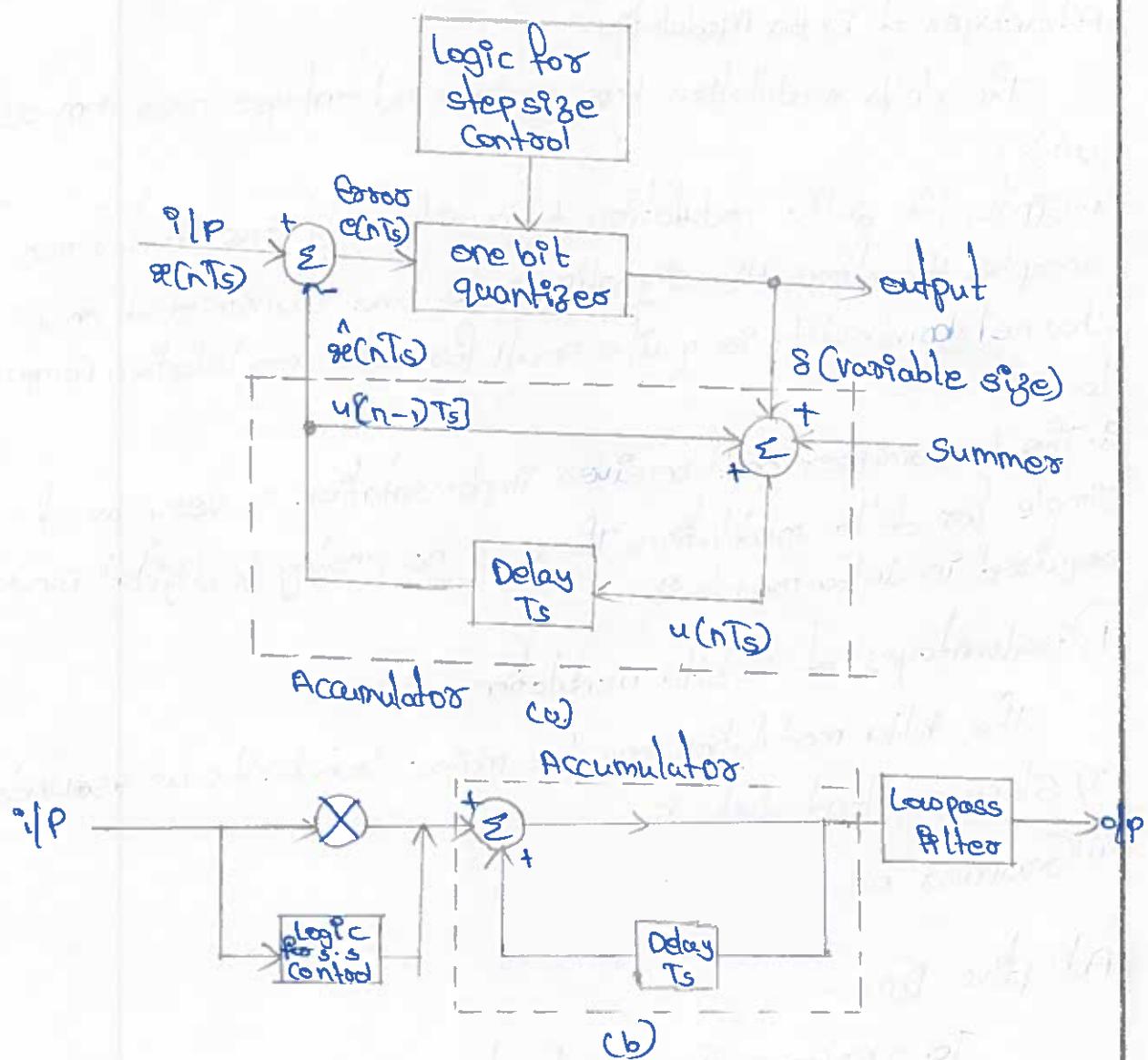


Fig: 10.17 Adaptive Delta modulator

a) Transmitter b) Receiver.