DELTA MODULATION

(i) Reason to use Delta Modulation

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

(ii) Working Principle

Delta modulation transmits only one bit per sample.
Here, the present sample value is compared with the previous sample value & this result whether the Amplitude is increased or decreased is transmitted.
Input signal $x(t)$ is Approximated to step signal by the delta modulator. This step size is kept fixed.
The difference between the input signal $x(t)$ & staircase Approximated signal is confined to two levels, i.e. $+\Delta$ & $-\Delta$.
Now, if the differences is positive, the Approximated signal is increased by one step, i.e. 'Δ'
If the difference is negative, the Approximated signal is decreased by one step 'Δ'.
When the step is reduced, '0' is transmitted & if the step is increased '1' is transmitted.

Hence for each sample, only one bit is transmitted

(iii) Mathematical Expression

The error between the sampled value of $x(t)$ & last approximated sample is given as

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

where,

$e(nT_s)$ = Error at present sample

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

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

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

$$u[(n-1)T_s] = \hat{x}(nT_s) \quad \text{------(ii)}$$

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

$$b(nT_s) = \Delta \, \text{sign}[e(nT_s)] \quad \text{------(iii)}$$

This means that depending upon the sign of error $e(nT_s)$, the sign of step size Δ is decided.

We can write,

$$b(nT_s) = \begin{cases} +\Delta & \text{if } x(nT_s) \geq \hat{x}(nT_s) \\ -\Delta & \text{if } x(nT_s) < \hat{x}(nT_s) \end{cases} \quad \text{---(iv)}$$

T_s = Sampling Interval.

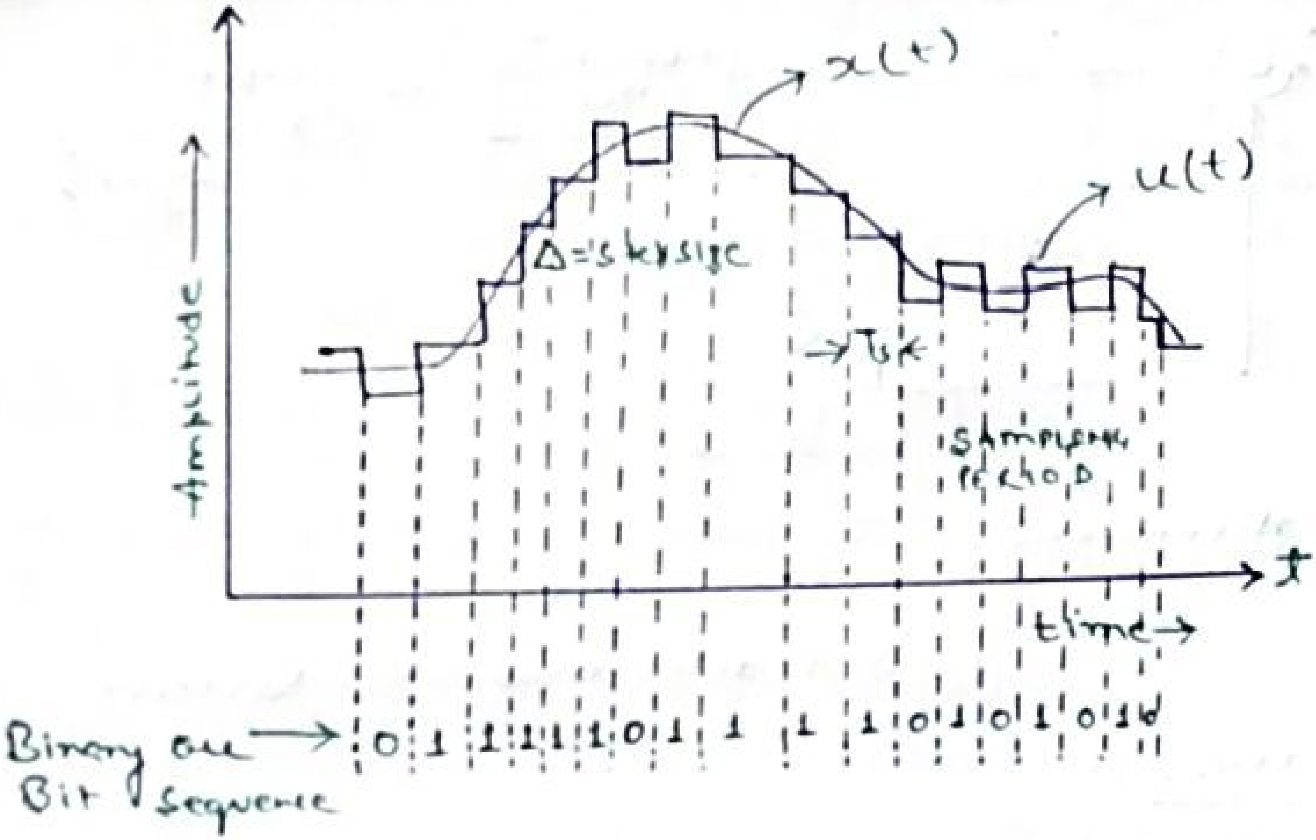

fig. Delta Modulation waveform

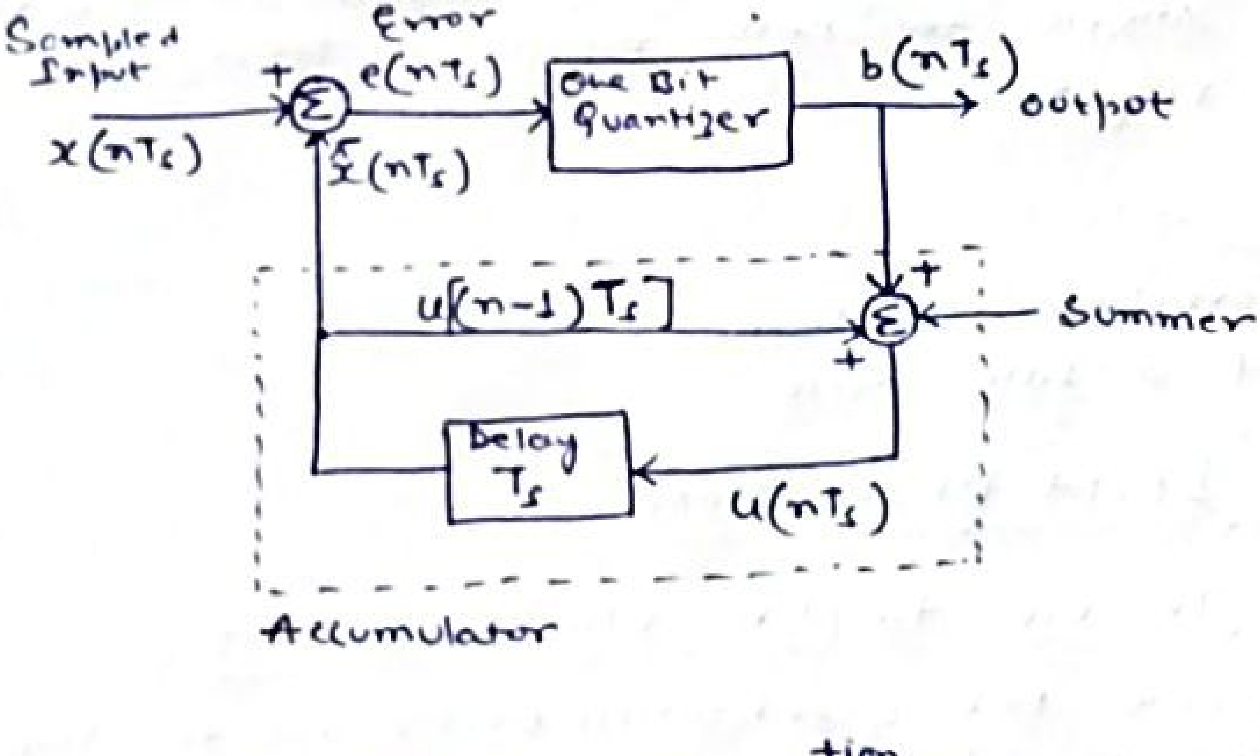

fig (a) A Delta Modulation Transmitter

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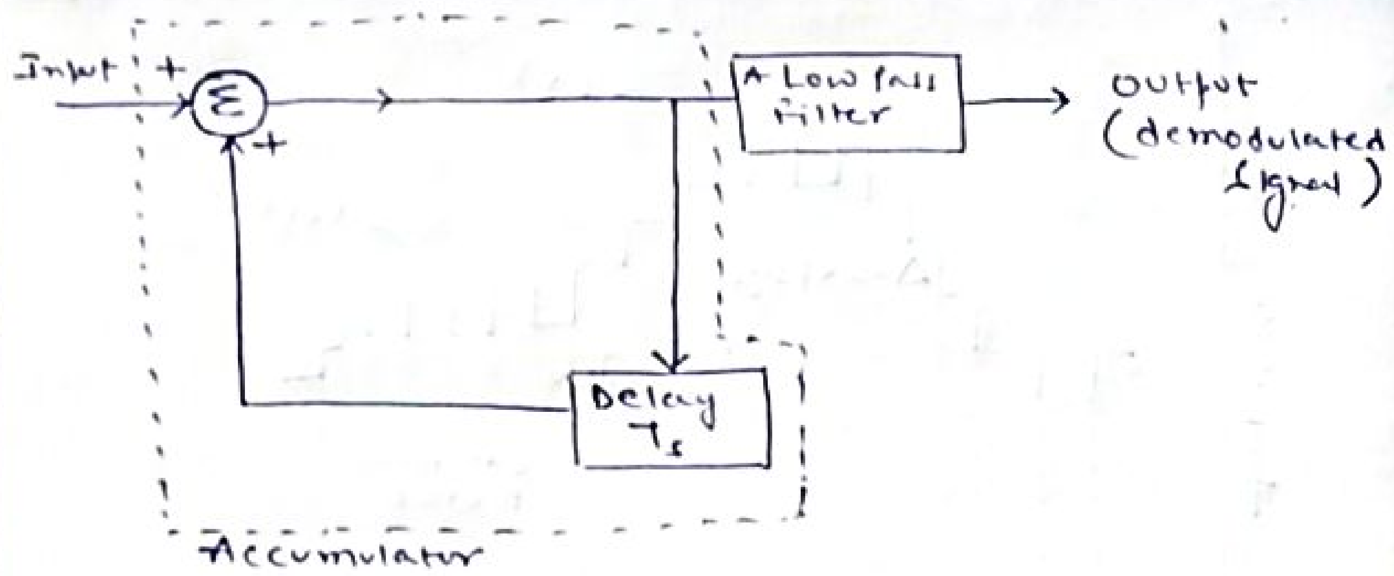


fig. (b) A delta modulation Receiver

ADVANTAGES :-

(i) Signaling rate & Transmission Bandwidth is quite small for delta modulation compared to PCM.

(ii) The transmitter & Receiver implementation is very much simple for Delta modulation. there is no ADC required.

Drawbacks :-

(i) Slope overload distortion.

(ii) Granual or Idle noise.

ADAPTIVE DELTA MODULATION

(i) Reason To use Adaptive delta Modulation

To overcome the quantization error due to slope overload & granual noise, the step size (Δ) is made Adaptive to variations in the input signal $x(t)$. Particularly in the step segment of the signal $x(t)$, the step size is increased. Also if the input is varrying slowly, the step size is reduced. Then, this method is known as Adaptive delta modulation (ADM).

The adaptive delta modulators can take continious charges in step size or discrete changes in step size.

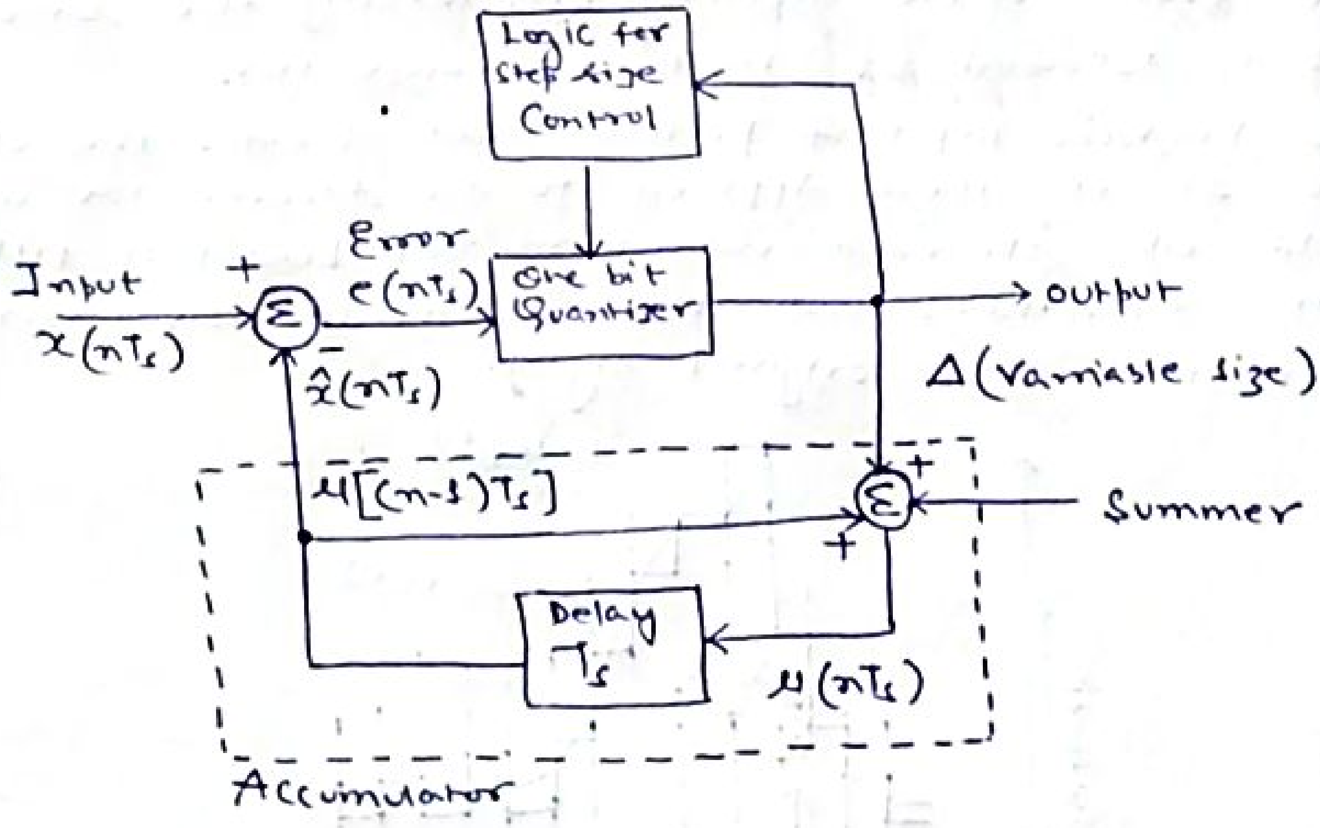


fig.(a) Adaptive delta Modulator Tx

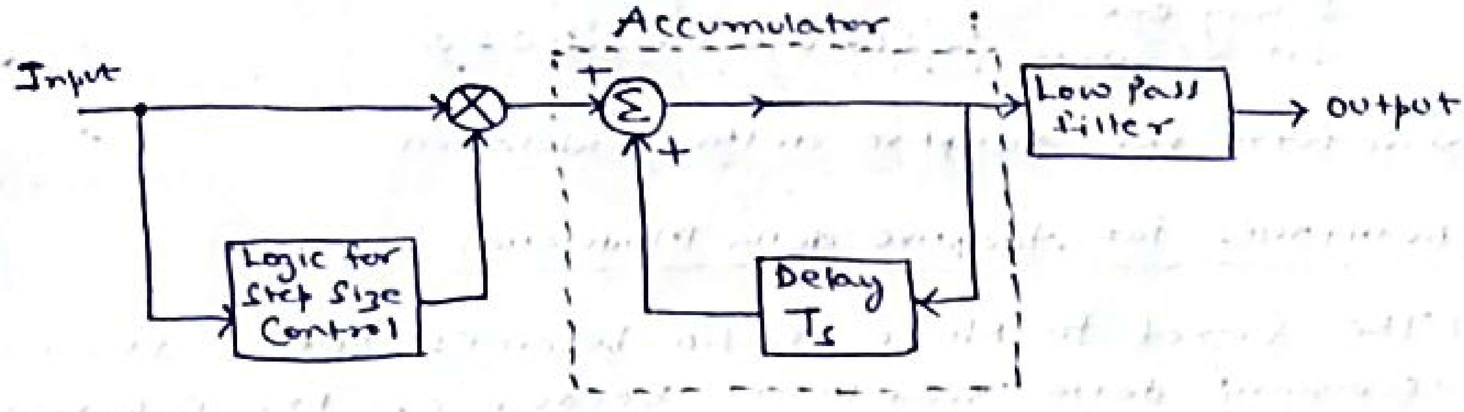


fig. (b) Rx

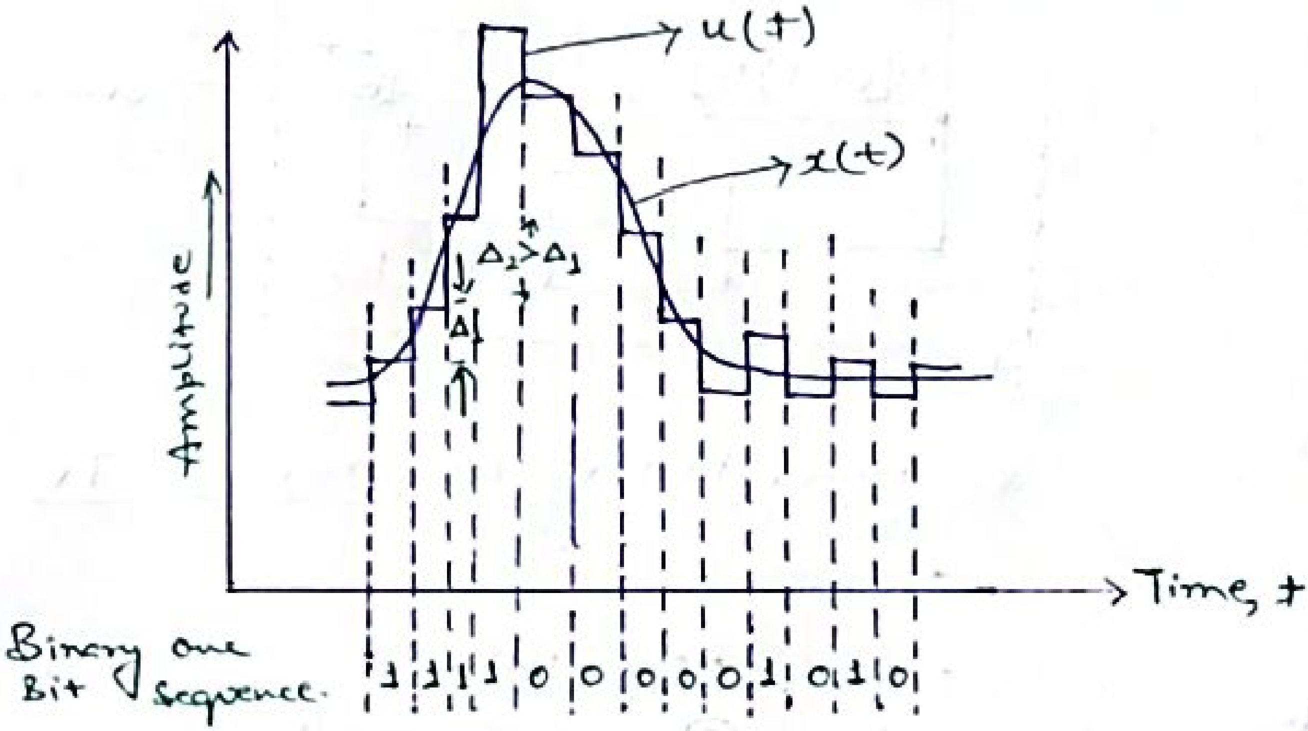
TRANSMITTER PART

The Logic for step size control is Added in the diagram. The step size increases or decreases According to a specified rule depending on one bit Quantizer O/P.

Reciever Part :-

In the reciever of Adaptive delta modulator, there are two portions. The first portion produces the step size from each incoming bit. Exactly the same process is followed as that in transmitter.

The Previous input + present input decides the step size. It is then Applied to an Accumulator which build up staircase waveform. The Low pass filter then smoothens out the staircase waveform to reconstruct the original signal.



⇒ Wave form for Adaptive delta Modulation

Advantages for Adaptive delta Modulation :-

(i) The Signal to Noise Ratio becomes better then ordinary delta modulation because of the reduction in slope overload distortion & Idle noise.

(ii) Because of the Varriable step size, the dynamic range of ADM is wider than simple DM.

(iii) Utilization of Bandwidth is better than delta modulation.

DIFFERENTIAL PULSE CODE MODULATION (DPCM)

i) Reason To Use DPCM

It may be observed that the samples of a signal are highly correlated with each other. This is due to the fact that 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 a little difference.
When these samples are encoded by a standard PCM system, the resultant encoded signal contains redundant information.

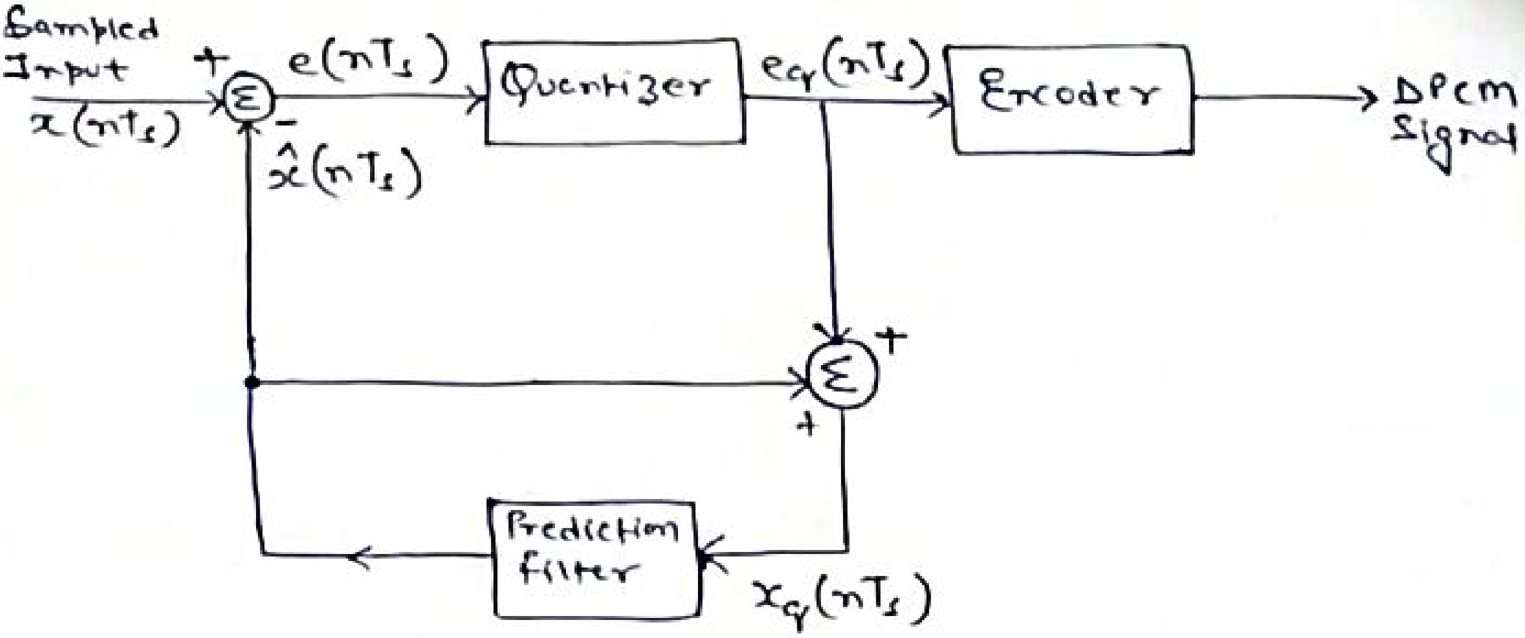
If this redundancy is reduced, then overall bit rate will decrease & no. of bits required to transmit one sample will also be reduced.
This type of digital pulse modulation scheme is known as Differential pulse code Modulation (DPCM).

(ii) Working Principle :-

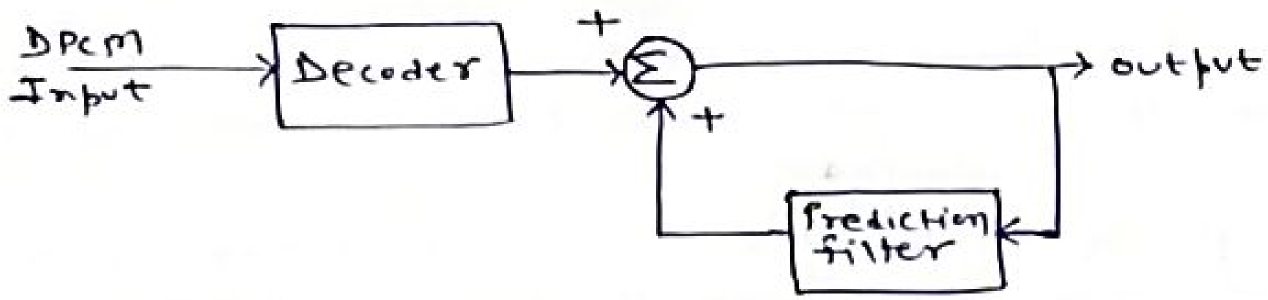
In fact the differential pulse code modulation works on the principle of prediction.
The value of the present sample is predicted from the past sample.
The prediction may not be exact but it is very close to the actual sample value.

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

Hence, the quantized version of the signal $x_q(nT_s)$ is the sum of the original sample value & quantization error $q(nT_s)$. The quantization error can be positive or negative.



(a) DPCM Transmitter



(b) DPCM Receiver

(iii) Reception of DPCM Signal

The decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output & quantized error signals are summed up to give quantized version of the original signal. Thus the signal at the reciever differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

Types of Predictors :-

(i) One Tap Predictors.
(ii) N- Tap Predictors.

Advantage of DPCM

(i) This will require less number of Quantization levels & hence less number of bits to represent them.

(ii) Signaling rate & bandwidth of DPCM system will be less than that of PCM.

A PCM – TDM System : T1 Carrier System

When a Large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required.
This system is used to convey multiple signals over telephony lines using wideband Co-axial cable.

Working Operations

(i) The system has been designed to accomodate 24 voice channels marked S_1 to S_{24}. Each signal is bandlimited to 3.3 KHz & the sampling is done at a standard rate of 8KHz. The sampling is done by the commutator switch SW_1.

(ii) These voice signals are selected one by one & connected to a PCM transmitter by the commutator switch SW_1.

(iii) Each sampled signal is then Applied to the PCM transmitter which converts it into a digital signal by the process of A to D Conversion & Companding.

(iv) The resulting digital waveform is transmitted over a co-axial cable

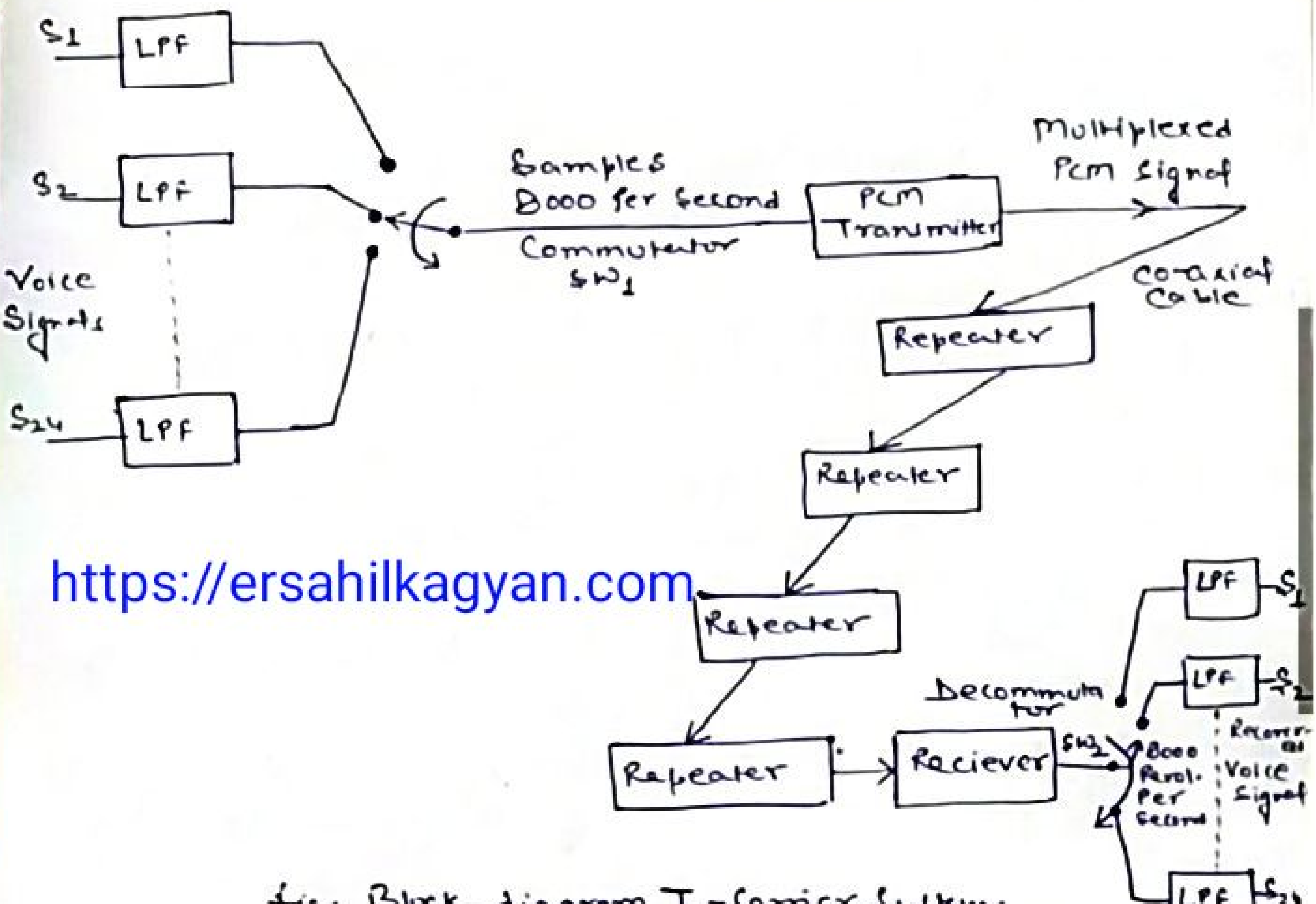


fig. Block diagram T_1-Carrier System.

(v) Periodically, after every 6000 ft, the PCM-TDM signal is generated by Amplifiers called Repeaters. They eliminate the distortion introduced by the channel & remove the superimposed Noise & regenerate a clean PCM-TDM signal at their output.

(vi) At the destination, the signal is companded, decoded & demultiplexed, using a PCM reciever. The PCM reciever output is connected to different low pass filters via the decommutator switch SW_2

(vii) Synchronization between the transmitter & Reciever Commutators SW_1 & SW_2 is essential in order to ensure proper communication.

MATCHED FILTER

The device for the optimum detection of a pulse involves the use of a linear-time-invariant (LTI) filter, which is known as a matched filter.

The matched filter is so called because its impulse response is matched to the pulse signal.

The ~~optimum~~ Integrate filter (~~optimum~~ i.e... Reciever) / DUMP FILTER

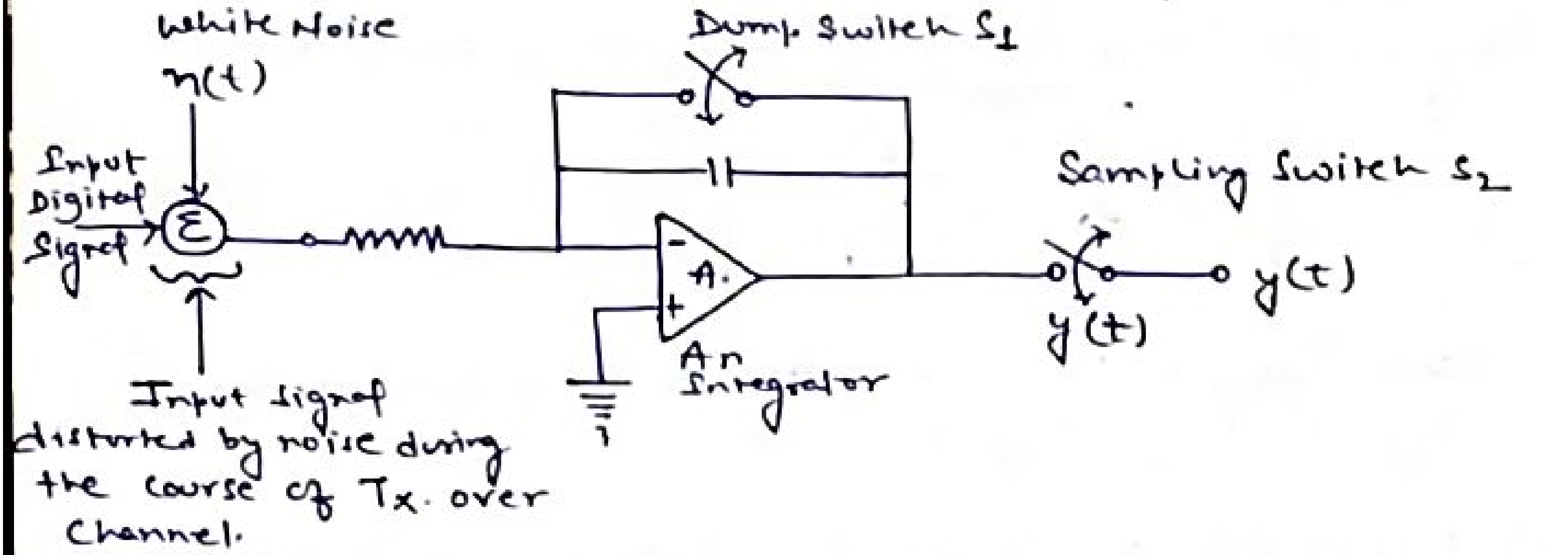


fig. Integrate & Dump filter

Let us consider a very simple & basic detector ckt. for the detection of digital signals.

Here the digital signal $x(t)$ is distorted by white noise $n(t)$ during the transmission over channel.

The noisy signal $[x(t) + n(t)]$ is applied to the input of integrate & Dump filter.

The capacitor is discharged fully at the begining of the bit interval.

This is achieved by temporarily closing switch S_2 at the begining of the bit interval.

The integrator then integrates noisy input signal over one bit period.

for the square pulse input, the o/p of the integrator would be the triangular pulse.

At the end of the bit period $t = T$, the magnitude of $y(t)$ attains its maximum amplitude.

Detection in integrate & dump filter is unaffected by the values of previous bits.

The o/p of integrator would decrease after $t > T$.

⇒ Signal to Noise ratio of Integrate & Dump filter

$$\boxed{\left[\frac{S}{N}\right]_0 = \frac{A^2 T}{N_0/2}}$$

This signal to Noise ratio is also known as <u>figure of Merit</u>

⇒ Probability of Error for Integrate & Dump filter.

$$P_e = \frac{1}{2}\,\mathrm{erfc}\sqrt{\frac{A^2 T}{N_0}}$$

Since, $A^2 T = E$ = Energy of the bit.

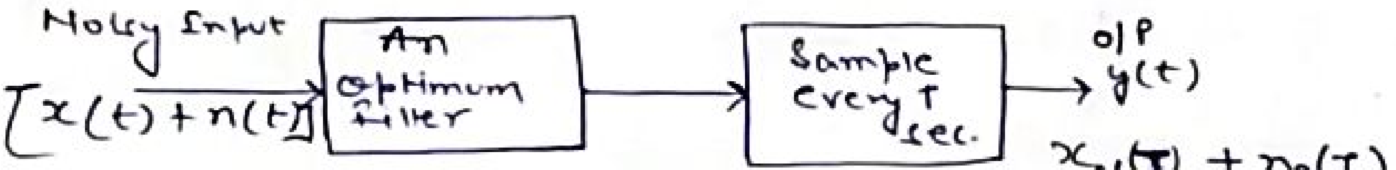
$$\boxed{P_e = \frac{1}{2}\,\mathrm{erfc}\sqrt{\frac{E}{N_0}}}$$

The OPTIMUM FILTER (OPTIMUM RECEIVER)

Previously we discussed about the Integrate & Dump filter. But at this point, let us think whether Integrate & Dump filter is an optimum* filter for the purpose of minimizing the probability of Error P_e.

For this purpose we used a generalized filter to recieve binary coded signals. It is known as <u>Optimum filter</u>

Here we considered the generalized Gaussian Noise which is having zero mean.



$x_{01}(T) + n_0(T)$
or
$x_{02}(T) + n_0(T)$

$x(t)$ will be $x_1(t)$ or $x_2(t)$

Noise $n(t)$ is added in the channel & has Power spectral density PSD $S_{ni}(f)$

Input to the optimum filter is $[x(t) + n(t)]$

Output from the filter is $x_{01}(T) + n_0(T)$ or $x_{02}(T) + n_0(T)$

In the Absence of Noise $n(t)$ the o/p of the Reciever will be

$$y(t) = x_{01}(T) \quad \text{if } x(t) = x_1(t)$$
$$y(t) = x_{02}(T) \quad \text{if } x(t) = x_2(t)$$

Hence in the Absence of noise, decisions are taken clearly.

* Here the word optimum mean to have max. Advantage.

The decision boundary will be in the middle of $x_{01}(T)$ & $x_{02}(T)$

$\therefore$ Decision Boundary $= \dfrac{x_{01}(T) + x_{02}(T)}{2}$

Probability of Error (P_e)

$$\boxed{P_e = \frac{1}{2}\,\text{erfc}\left[\frac{x_{01}(T) - x_{02}(T)}{2\sqrt{2}\,\sigma}\right]}$$

Transfer function for the optimum filter

The optimum filter which minimizes the probability of error (P_e) has to maximize the ratio

$$\left[\frac{x_{01}(T) - x_{02}(T)}{S}\right]^2.$$

To maximize this ratio, the filter has the transfer function given by the following Expression

$$H(f) = k.\frac{X(f)}{S_{ni}(f)}e^{-j2\pi fT}$$

& the Maximized ratio is Expressed as

$$\left(\frac{S}{N}\right)_{o\,max} = \left[\frac{x_0^2(T)}{\sigma^2}\right]_{max} = \left[\frac{x_{01}(T) - x_{02}(T)}{\sigma}\right]^2_{max.}$$

or

$$\left(\frac{S}{N}\right)_{o\,max} = \int_{-\infty}^{\infty}\frac{|X(f)|^2}{S_{ni}(f)}\,df$$