Chapter -2

Sampling Theory

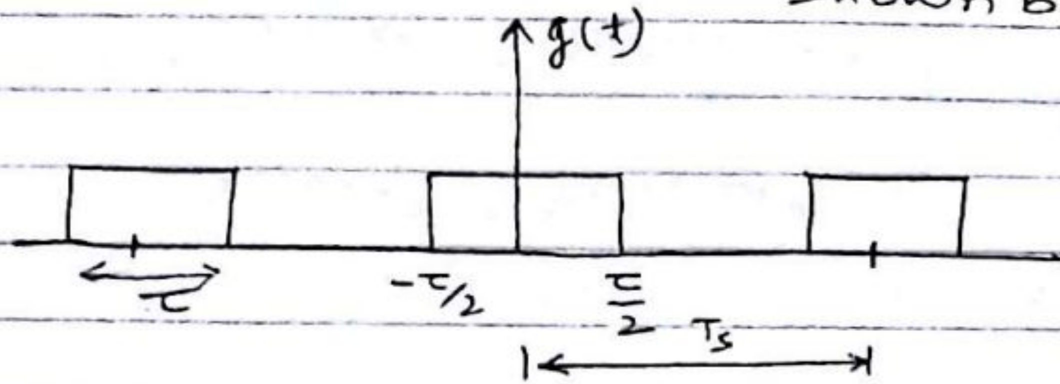
Sampling Theory

- Process of converting analog or continuous signal into discrete signal is called sampling.
- <u>Sampling Theorem/Nquist Criteria</u>
- For band limited signal which do not have maximum frequency greater than f_x
- Sampling Rate or Sampling frequency
- For Transmitting and Receiving end
- $f_s \geq 2 f_x$

Sampling Theory

Let $x(t)$ be the band limited signal, then its sampled signal can be represented as

$\boxed{x_s(t) = x(t)\cdot g(t)}$ — ① where $g(t)$ is the sampling function shown below



$T_s \Rightarrow$ Sampling period

$\tau \Rightarrow$ duration of sampling pulse

Sampling Theory

The sampling function $g(t)$ is given by

$$g(t) = c_0 + 2\sum_{n=1}^{\infty} c_n \cos 2\pi n f_s t \quad - (2)$$

As

$$x_s(t) = x(t)\, g(t)$$

$$x_s(t) = x(t)\left[c_0 + 2\sum_{n=1}^{\infty} c_n \cos 2\pi n f_s t\right]$$

$$x_s(t) = c_0 x(t) + 2c_1 x(t) \cos 2\pi f_s t + 2c_2\, x(t) \cos 4\pi f_s t + \cdots \quad - (3)$$

$$c_0 = \frac{1}{T_s}\int_{T_s} x(t)\, dt = \frac{1}{T_s}\int_{-\tau/2}^{\tau/2} 1 \cdot dt = \frac{\tau}{T_s} \quad - (4)$$

Similarly,

$$c_n = \frac{1}{T_s}\int_{T_s} x(t)\, e^{-jn2\pi f_s t}\, dt = \frac{1}{T_s}\int_{-\tau/2}^{\tau/2} e^{-jn2\pi f_s t}\, dt$$

Sampling Theory

$$= \frac{1}{T_s} \left. \frac{e^{-jn2\pi f_s t}}{-jn2\pi f_s} \right|_{-\tau/2}^{\tau/2}$$

$$= -\frac{1}{jn2\pi f_s T_s} \left[e^{-jn2\pi f_s \tau/2} - e^{-jn2\pi f_s -\tau/2} \right]$$

$$= -\frac{1}{jn2\pi f_s T_s} \left[e^{-jn2\pi f_s \tau/2} - e^{jn2\pi f_s \tau/2} \right]$$

$$= \frac{1}{n\pi f_s T_s} \left[\frac{e^{jn2\pi f_s \tau/2} - e^{-jn2\pi f_s \tau/2}}{2j} \right]$$

$$= \frac{1}{n\pi f_s T_s} \sin(n\cancel{2}\pi f_s \tau/\cancel{2})$$

$$= \frac{\tau}{T_s} \frac{\sin(n\pi f_s \tau)}{n \cdot \pi f_s \tau}$$

$$= \frac{\tau}{T_s} \operatorname{sinc}(n\pi f_s \tau)$$

$$\boxed{C_n = \tau f_s \operatorname{sinc}(n\pi f_s \tau)} \quad - (5)$$

Sampling Theory

Taking Fourier Transform (F·T) of eqn ⑤ we get

$$x(t) \xleftrightarrow{F\cdot T} X(f)$$

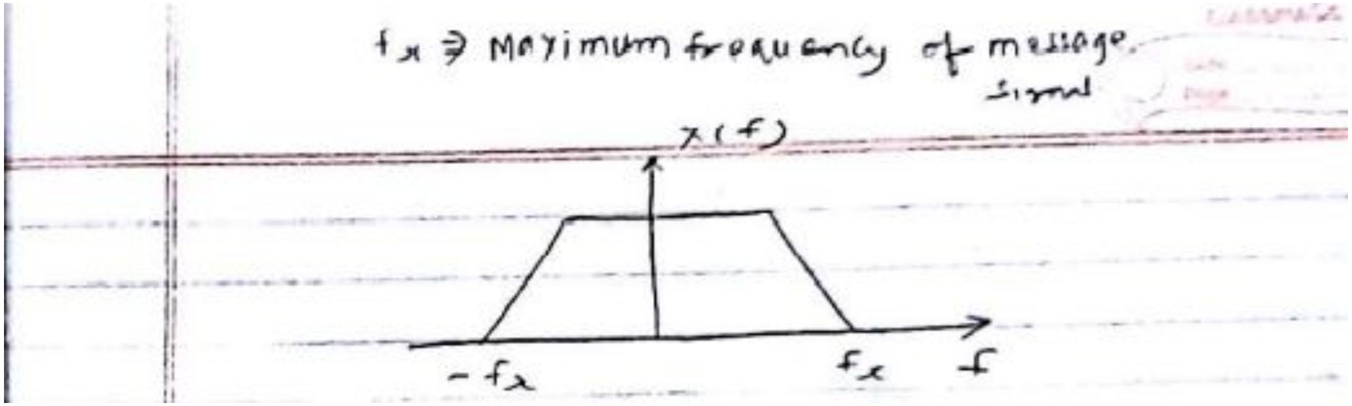
$$\cos w_0 t \xleftrightarrow{F\cdot T} \frac{1}{2}\left[\delta(f-f_0)+\delta(f+f_0)\right]$$

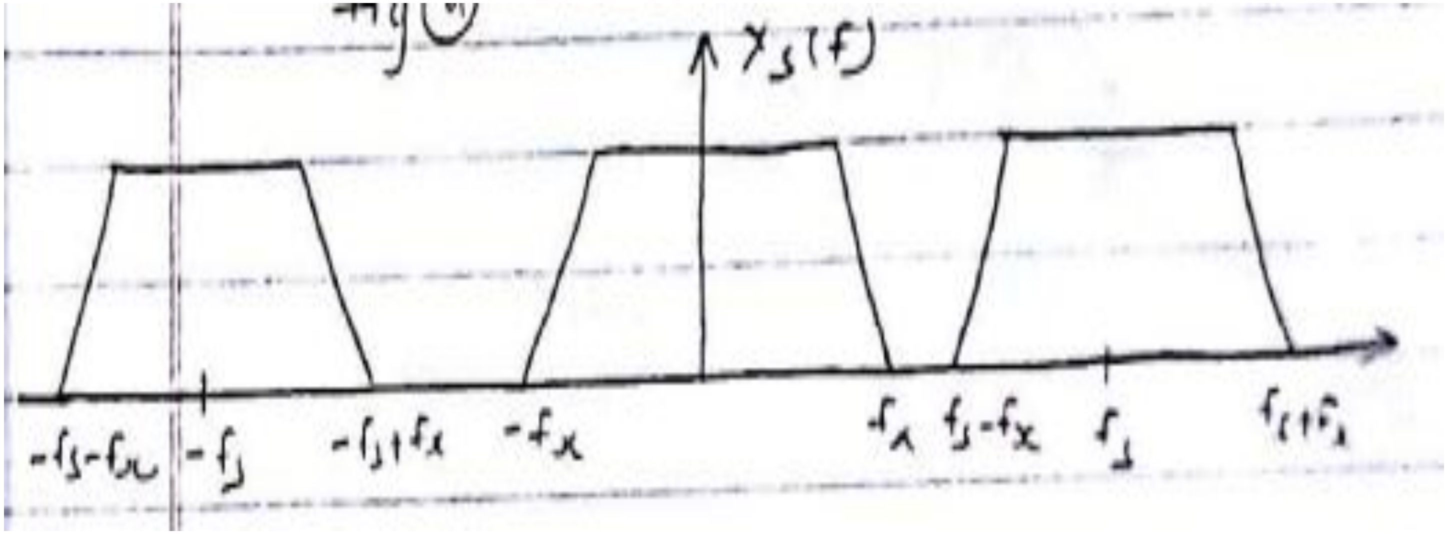
$$x(t)\cos w_0 t \xleftrightarrow{FT} \frac{1}{2}\left[X(f-f_0)+X(f+f_0)\right]$$

$$X_s(f) = C_0 X(f) + C_1 X(f-f_s) + C_1 X(f+f_s) + C_2 X(f-2f_s) + C_2 X(f+2f_s) + \cdots \quad (6)$$

Above eqn can be graphically represented as

Sampling Theory



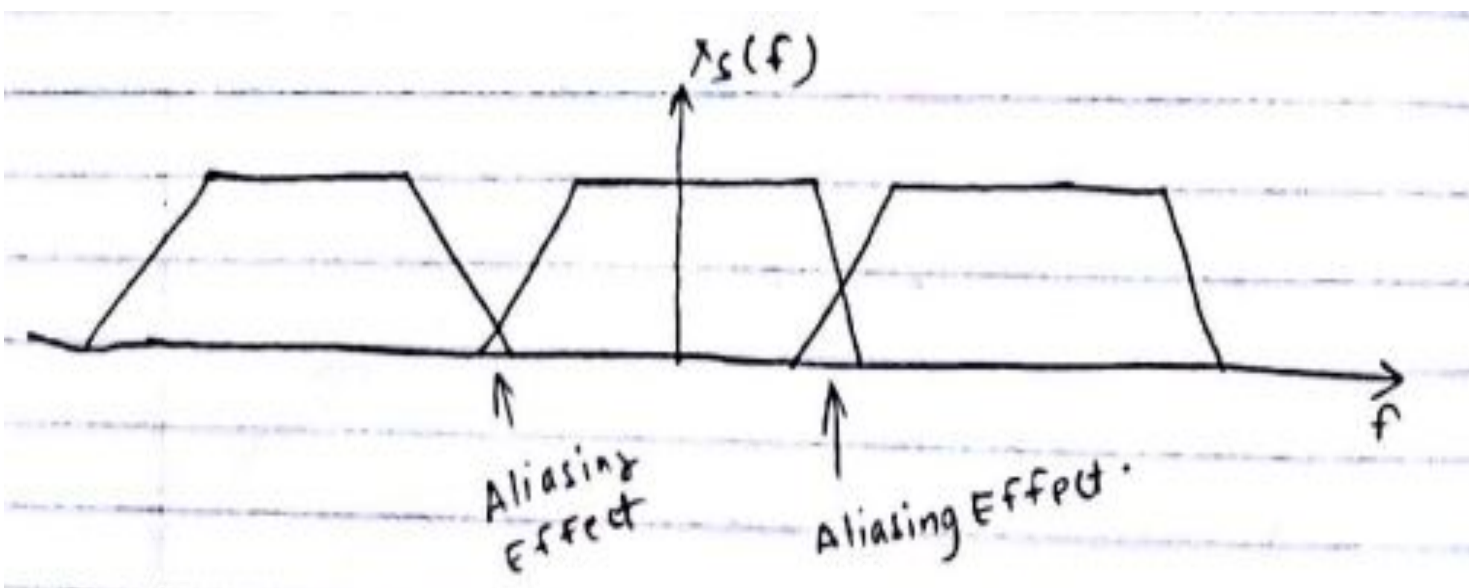


Sampling Theory

- Fig a represents the fourier transform of original message signal x(t).
- Fig b represents the fourier transform of the output of the sampler.
- For error less recovery of the message signal from the spectrum of the sampled signal mus t obey the nyquist sampling theorem as
- $f_s \geq 2 f_x$

Sampling Theory

- Similarly for sampling frequency $f_s < 2 f_x$
- Distortion will occur at the receiving side.
- Samples get overlapped while recovering.
- This phenomena is known as Aliasing.



Instantaneous / Ideal Sampling

- Instantaneous sampling gives train of impulse equal to the instantaneous value of input signal at sampling instant.
- Ideal sampling simply consist of switch Circuit.

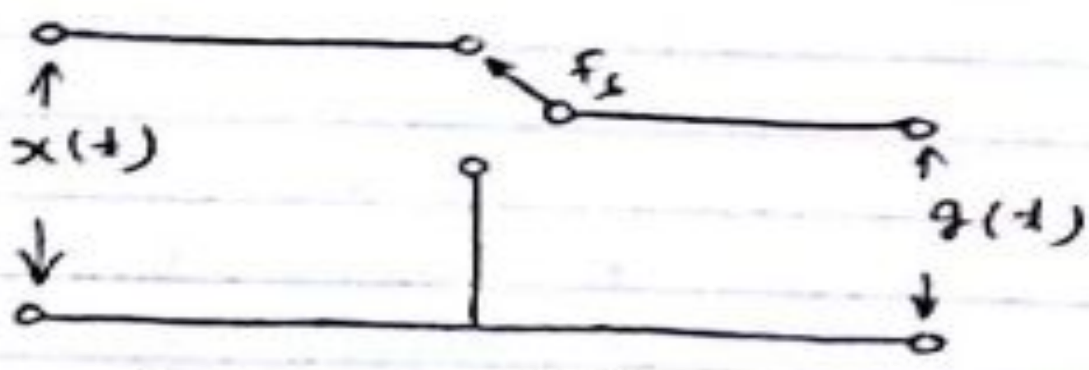


fig :→ switching Sampler

Instantaneous/Ideal Sampling

- For time t, the output g(t) contains the instantaneous value of input signal x(t).
- Sampling function is represented as train of impulse

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad \text{— (1)}$$

output $g(t)$ is expressed as

$$g(t) = x(t) \cdot \delta_{T_s}(t)$$

$$g(t) = x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s) \quad \text{— (2)}$$

Taking Fourier Transform (FT) we get,

$$\boxed{G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)} \quad \text{— (3)}$$

Reconstruction of Sampled Signal

- Process of Reconstructing original Signal x(t) from sampled signal is called Reconstruction.
- Signal x(t) band limited to f_m Hz can be reconstructed by passing the sampled signal through the Ideal Low Pass Filter at cut off frequency f_m
- The expression of sample signal is

  $g(t)= x(t)\ \delta T_s(t)$ -----(1)

Reconstruction of Sampled Signal

- $g(t) = 1/T_s$

$$g(t)=\sum_{n=-\alpha}^{\alpha} x(nts)\,\delta(t - nT_s) \text{-----}(3)$$

To recover the original Signal , the sampled signal is passed through the Ideal Low pass filter of bandwidth f_m Hz. The transform function of LPF is

$H(w) = T_s\ \mathrm{rect}(w/2\pi f_m)$---(4)

The impulse response of the filter is

$h(t) = 2\ f_m\ T_s\ \mathrm{Sinc}(2\pi f_m t)$-----(5)

As the sampling is done at Nyquist rate

$T_s = 1/2\ f_m$, $2 f_m T_s = 1$

$h(t) = 1^*\ T_s\ \mathrm{Sinc}(2\pi f_m t)$

$h(t) = T_s\ \mathrm{Sinc}(2\pi f_m t)$

Reconstruction of Sampled Signal

Output of the filter is

- $x(t)=\Sigma\, x(nT_s)\, h(t-nT_s)$
- $x(t)=\Sigma\, x(nT_s)\, \mathrm{Sinc}\, [2\pi f_m(t-nT_s)]$
- $x(t)=\Sigma\, x(nT_s)\, \mathrm{Sinc}\, [2\pi f_m t-2\pi f_m nT_s)]$
- $x(t)=\Sigma\, x(nT_s)\, \mathrm{Sinc}\, [2\pi f_m t-n\pi)]$ is known as Interpolation formula
- Message or original signal can be recovered from the weighted sum of all sample value.

Reconstruction of Sampled Signal

$\therefore h(t) = 1 \; \text{sinc}(2\pi f_m t)$

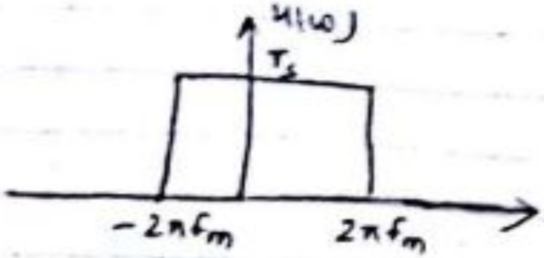


fig:→ Transfer function of LPF

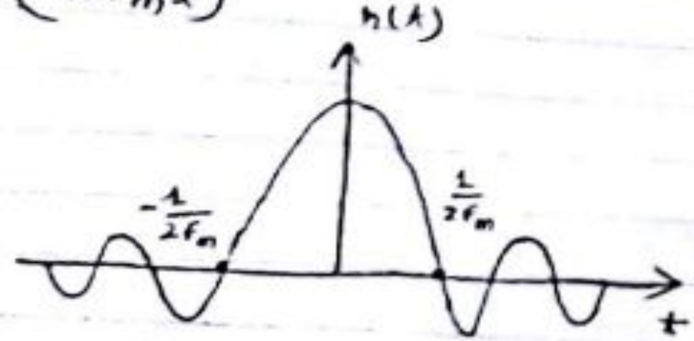


fig:→ Impulse response of LPF

Effect of Under Sampling: Aliasing

- Continuous time band limited signal sampled at rate lower than the Nyquist rate($f_s < 2 f_m$)
- Aliasing effects occurs by overlapping of successive cycle of spectrum.
- Higher frequency overlaps or undertakes over lower frequency component.
- Low Pass Filter called prealias filter to limit band of frequency of the signal to f_m .
- Select sampling frequency $f_s \geq 2 f_m$.

Sampling of Band Pass Signal

- Also Known as Sub Sampling Theorem.
- Sampling Theorem of band pass signal x(t) can be recovered from its sample if it is sampled with minimum rate of twice of bandwidth.

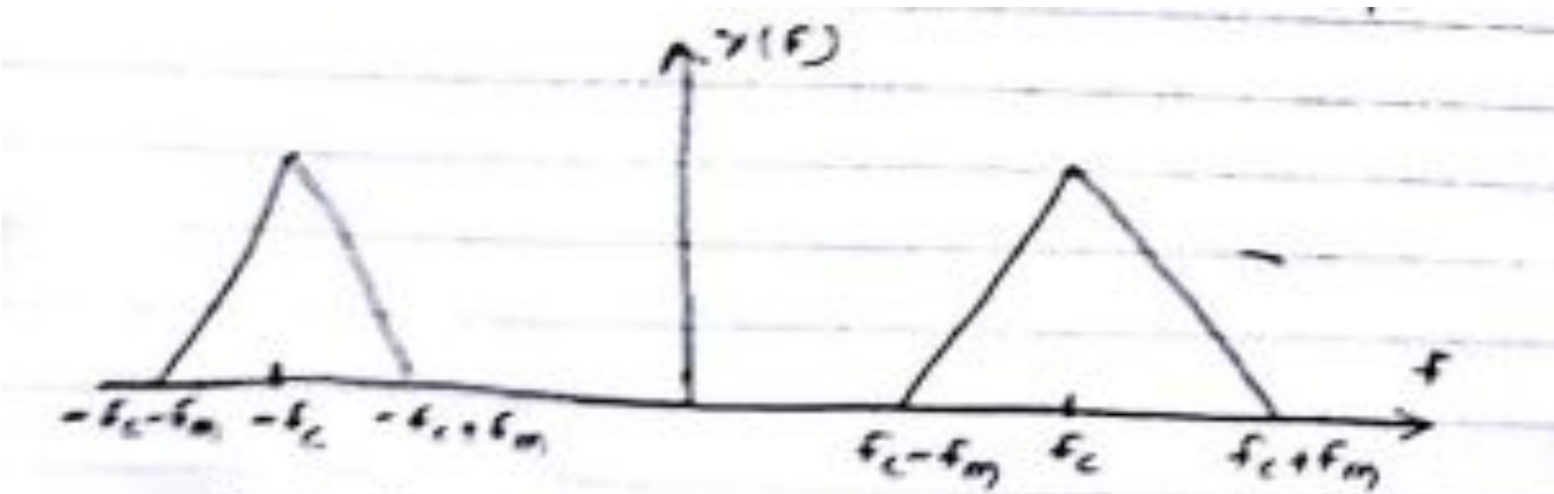


fig: Spectrum of band pass signal

Sampling of Band Pass Signal

- For bandwidth of spectrum of band pass signal of 2 f_m .
- Sampling rate of band pass signal must be 4 f_m
  Signal x(t) can be expressed in terms of Inphase and Quadrature are expressed as

- $X(t)=x_I(t)\cos(2\pi f_c t) - x_Q(t)\text{Sin}(2\pi f_c t)$---(1)

Sampling of Band Pass Signal

By solving

we get $x(t) = \sum_{n=-\infty}^{\infty} x\left(\frac{n}{4f_m}\right) \operatorname{sinc}\left(2f_m t - \frac{n}{2}\right) \cos\left[2\pi f_c\left(t - \frac{n}{4f_m}\right)\right]$

Compairing this reConstruction formula with low pass signals

$$x(t) = \sum x(nT_s) \operatorname{sinc}[2\pi f_m t - n\pi]$$

$x\left(\frac{n}{4F_m}\right) = x(nT_s)$ = sampled version of bandpass signal

$$T_s = \frac{1}{4f_m}$$

$\therefore$ $\boxed{\text{Minimum sampling rate} = \text{Twice of bandwidth} = 4f_m}$

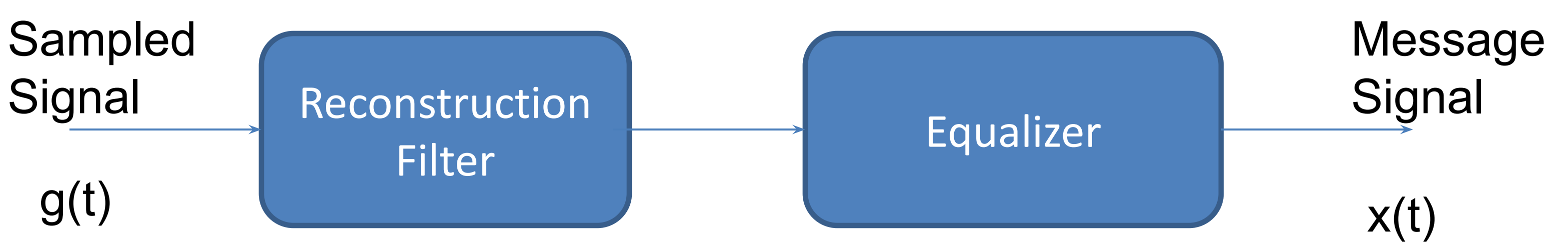
Aperture Effect

- Attenuation of upper portion of message signal spectrum is called Aperture Effect.
- It effects the high frequency component introducing Amplitude Distortion.
- It is depended upon the duration or pulse width of each sample (τ).
- Large or greater value of pulse width ‘τ’ introduces Aperture Effect.

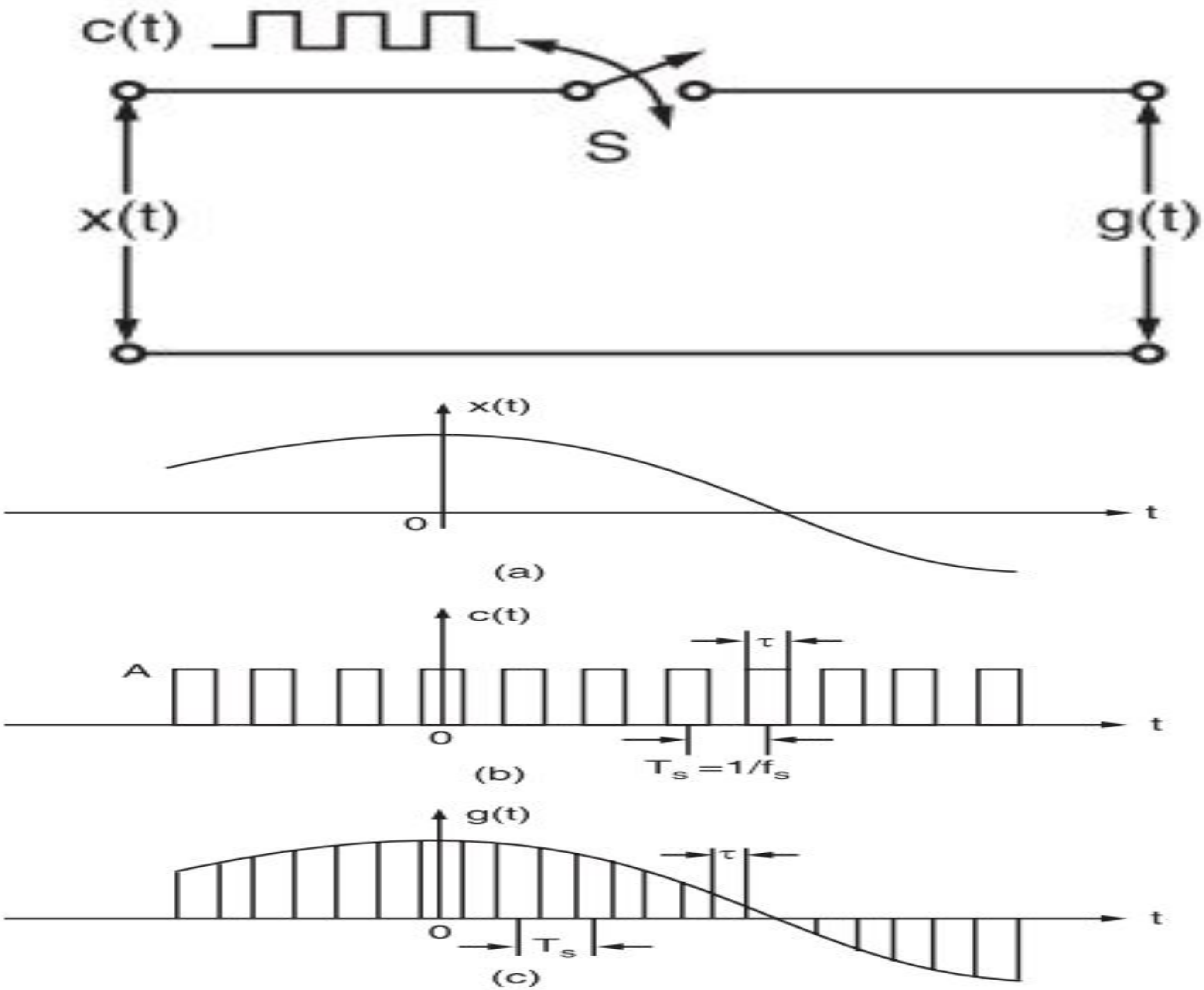
Aperture Effect

Mitigation of Aperture Effect

- We use the pulse width ‘τ’ or duration as low / small/narrow as possible.
- By use of Equalizer while Reconstruction, which will compensate attenuation caused by Low Pass Reconstruction Filter.
- Introducing Guard band between frequency.



Natural Sampling



Natural Sampling

- Natural Sampler is a practical method whose periodic pulse of sampling function c(t) is of width ‘τ’.
- Continuous Input signal x(t) is sampled at the rate of Sampling frequency satisfying Nyquist Criterion.
- Sampled Signal
- g(t)=x(t).C(t)------(1)

Natural Sampling

We know that the exponential Fourier series for any periodic waveform is expressed as

$$c(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi nt/T_0}$$

$$c(t) = \sum_{n=-\infty}^{\infty} C_n . e^{j2\pi f_s nt} \quad \text{with} \quad \frac{1}{T_0} = f_s$$

Since C(t) is rectangular pulse train

$$C_n = \frac{\tau.A}{T_s} \cdot \sin c(f_{n.}\tau)$$

$$c(t) = \sum_{n=-\infty}^{\infty} \frac{\tau.A}{T_s} \cdot \sin c(f_{n.}\tau)\, e^{j2\pi f_s \cdot nt}$$

$$g(t) = \frac{\tau A}{T_s} \cdot \sum_{n=-\infty}^{\infty} \sin c(f_{n.}\tau).\, e^{j2\pi f_s nt} .x(t)$$

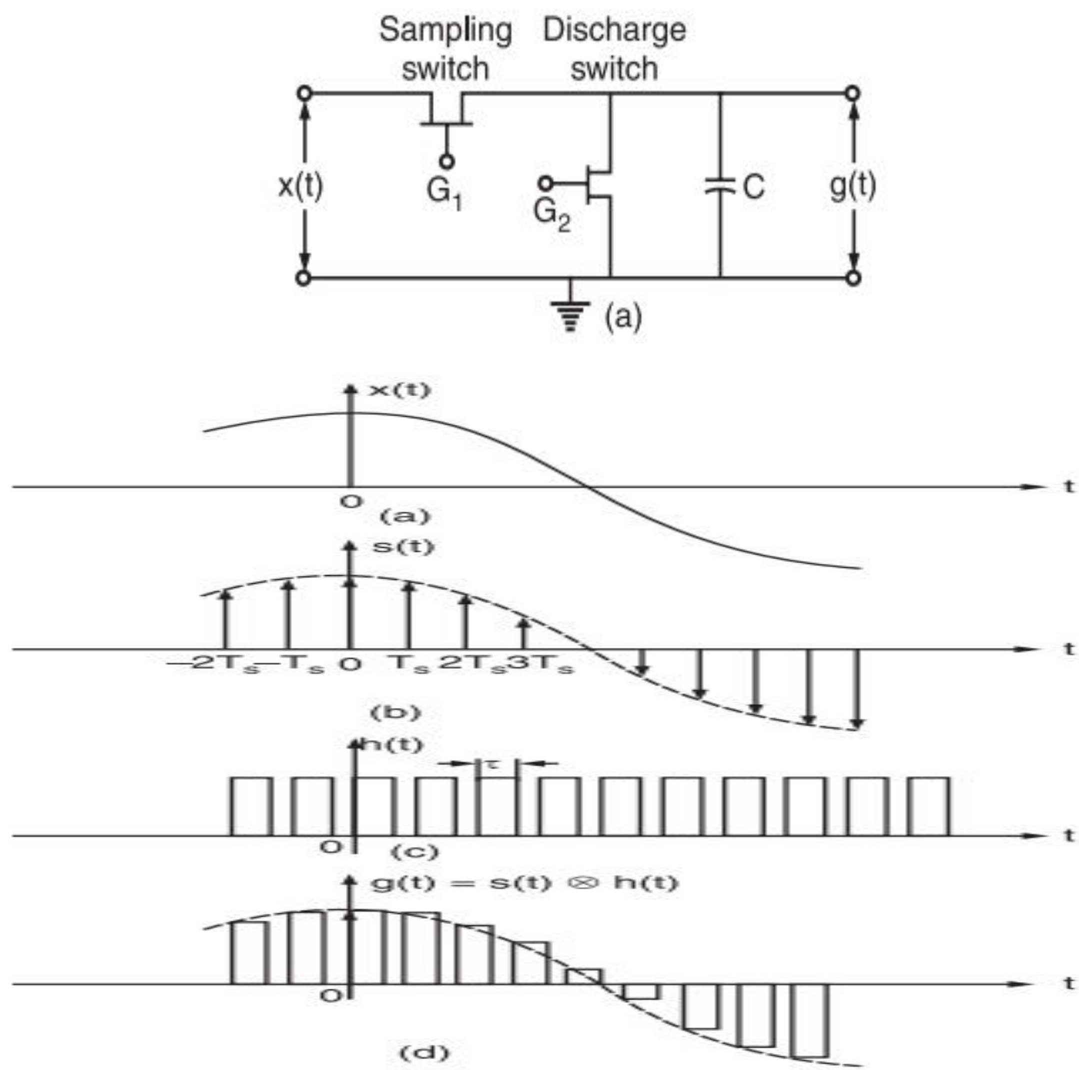
Taking Fourier Transform the spectrum of Natural Sampled signal

$$G(f) = \frac{\tau A}{T_s} \cdot \sum_{n=-\infty}^{\infty} \sin c(nf_{s.}\ \tau)\, X(f - nf_s)$$

Natural Sampling

- The output sample has varying top in accordance with continuous time analog signal.
- Difficult to determine the shape of top of pulse.
- Amplitude detection at Reconstruction may not be exact.
- More susceptible of contamination of Noise.

Flat Top Sampling



Flat Top Sampling

$$g(t) = s(t) \otimes h(t)$$

$$s(t) = x(t) \cdot \delta_{T_s}(t)$$

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

$$g(t) = s(t) \otimes h(t)$$

$$g(t) = \int_{-\infty}^{\infty} s(\tau) h(t - \tau)\, d\tau$$

$$g(t) = \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(\tau - nT_s) h\,(t - \tau)\, d\tau$$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \delta(\tau - nT_s) h\,(t - \tau)\, d\tau$$

Flat Top Sampling

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t - nT_s)$$

$$g(t) = s(t) \otimes h(t)$$

Taking Fourier transform of both sides of above equation, we get

$$G(f) = S(f)\, H(f)$$

We know that S(f) is given as

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Therefore,

$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \cdot H(f)$$

Thus, spectrum of flat top sampled signal:

$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)\, H(f)$$

Flat Top Sampling

- The output sample has constant top in accordance with continuous time analog signal.
- Easy to determine the shape of top of pulse.
- More immune to Noise.
- Sample and Hold Circuit is used to get Flat Top Sampled .
- It consist of Two FET Switches and a Capacitor.

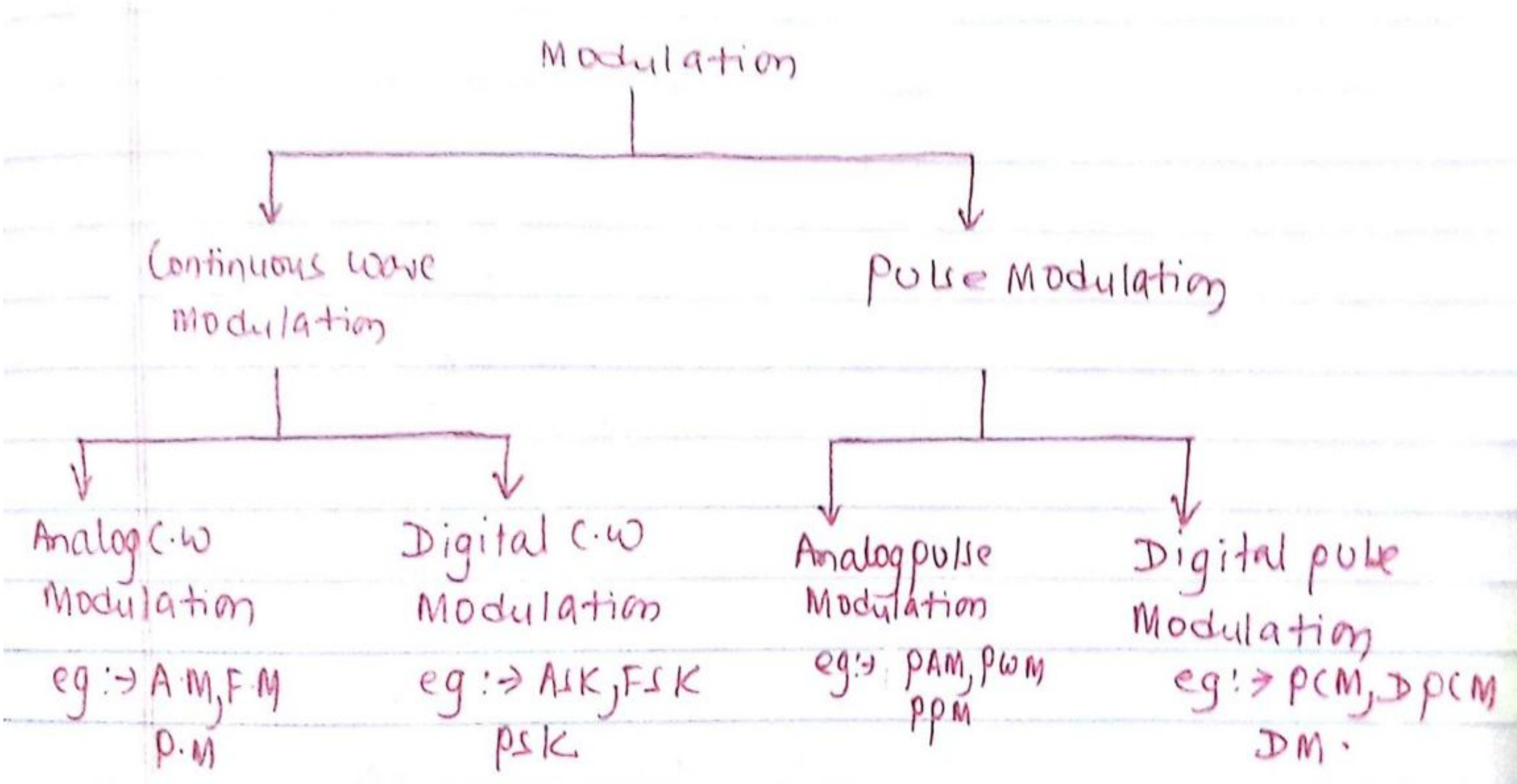
Flat Top Sampling

- Sampling Switch G1 is closed for short duration.
- Capacitor gets quickly charged up to voltage equal to the instantaneous sample value of the incoming signal x(t).
- Sampling switch G1 is opened. With the applied pulse to gate Discharge switch G2.
- Capacitor then gets discharged producing output of sample and Hold Circuit.

Chapter -3

Pulse Modulation System

Modulation


Modulation
Continuous wave Modulation
Pulse Modulation
Analog C.W Modulation
eg:→ A.M, F.M P.M
Digital C.W Modulation
eg:→ ASK, FSK PSK
Analog pulse Modulation
eg:→ PAM, PWM PPM
Digital pulse Modulation
eg:→ PCM, DPCM DM.

Pulse Modulation Systems

- In Pulse Modulation the carrier is pulse train instead of sinusoidal carrier.
- Characteristics or parameter of carrier signal is Amplitude, Width and Position which is changed w.r.t instantaneous value of modulating signal.
- PAM, PLM/PWM/PDM and PPM.
- Benefits of Pulse modulation: It Permits simultaneous transmission of several signals on time sharing basis. Time Division Multiplexing.

Pulse Amplitude Modulation(PAM)

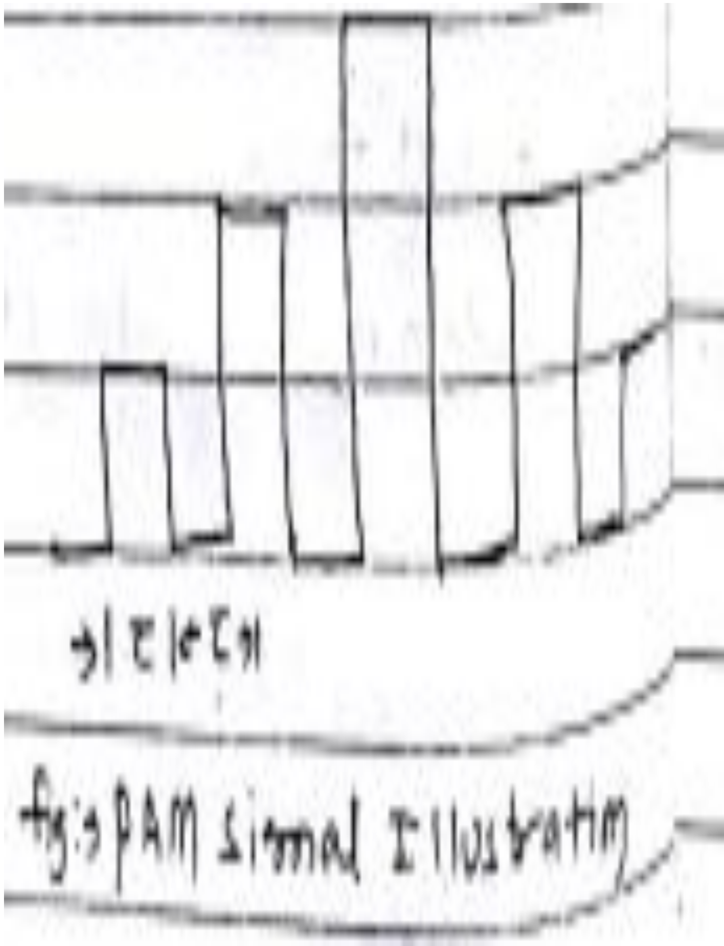
- Amplitude of carrier of periodic train of rectangular pulse is changed in proportion to sample value of message signal.
- Flattop sampling is used over Natural Sampling.
- Flattop Sampling has better noise immunity over Natural Sampling.

Transmission Bandwidth of PAM

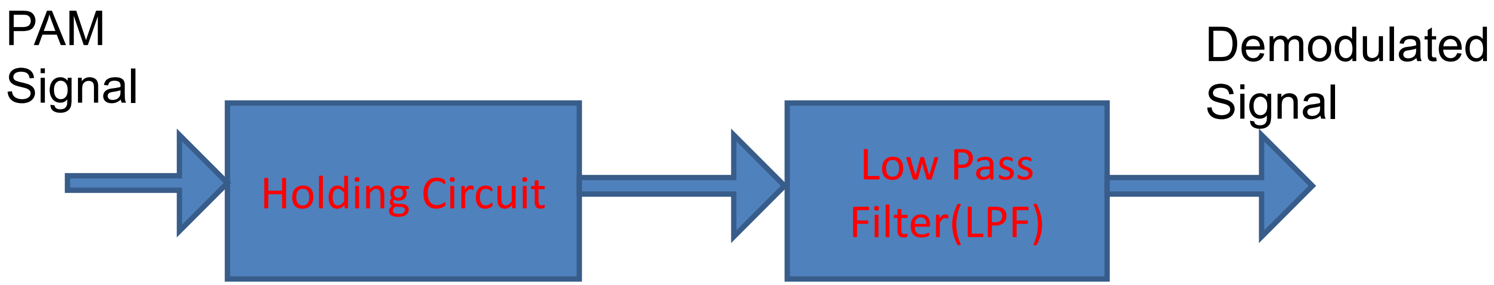
- Pulse duration is "τ" is very small in comparision to sampling period "T_S" between two sample.
- $\tau << T_S$ -----(1)
- According to Sampling Theorem
- $f_s \geqq 2 f_m$ ------(2)
- $1/ T_S \geqq 2 f_m$ ----(3)
- $1/ 2 f_m \geq \geq T_S$ ----(4)

Transmission Bandwidth of PAM

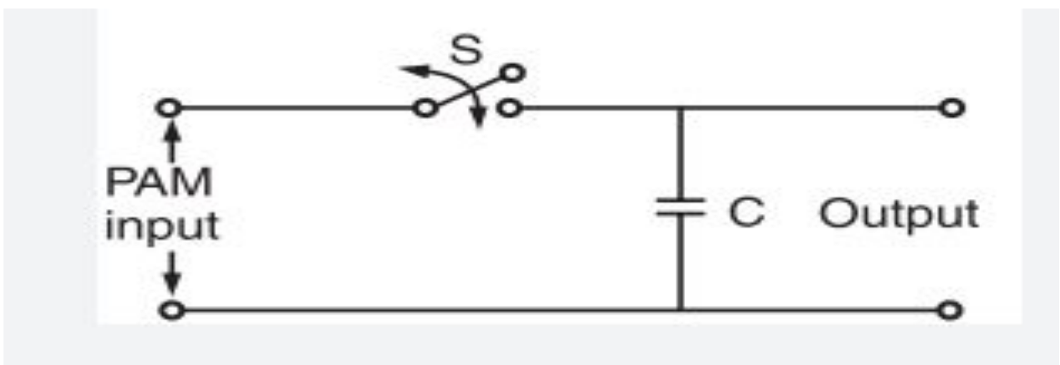
- From equation (1 & 3)
- $\tau << T_S \leq 1/\ 2\ f_m$
- For both ON and OFF Time
- Maximum frequency of PAM
- $f_{max} = 1/\tau + \tau = 1/2\tau$
- Transmission bandwidth
- $BW >> f_{max} \geq 1/2\tau$



Demodulation(Reconstruction)



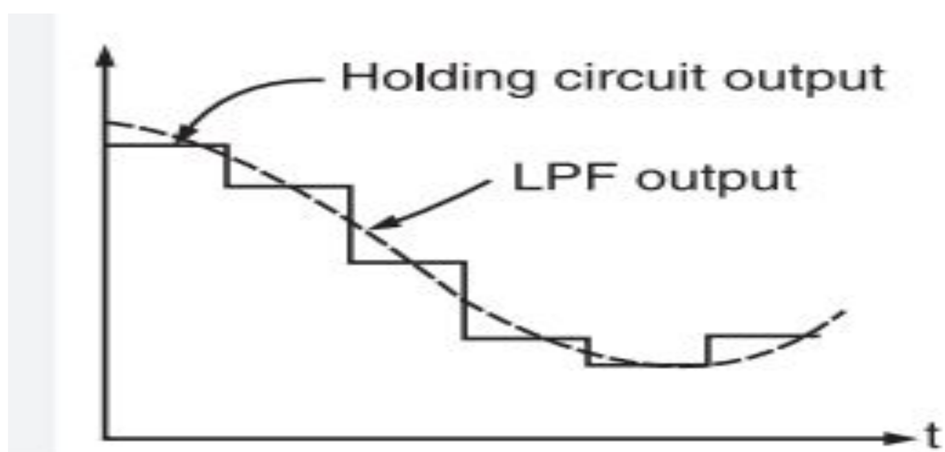
Block Diagram of PAM Demodulator



Holding Circuit (Zero-Order Holding Circuit)

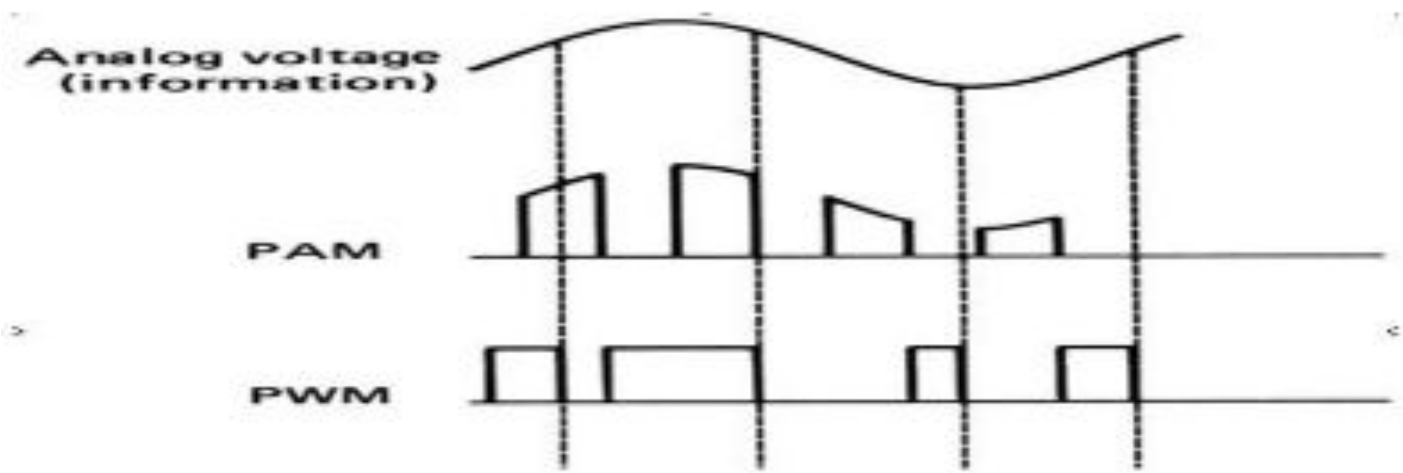
Demodulation(Reconstruction)

- Recovering of the message signal from the modulated signal is called Reconstruction.
- Demodulation is done by Holding Circuit.
- Switch S is closed after arrival of pulse.
- Capacitor is charged to the pulse amplitude value and passed through LPF to recover the modulating signal.



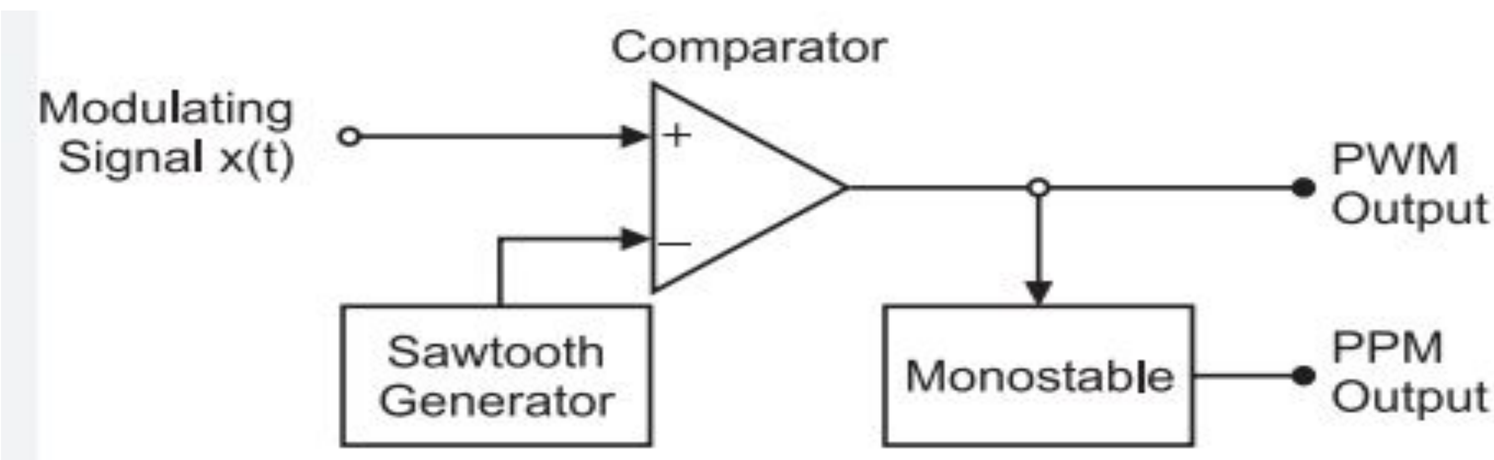
PLM/PWM/PDM

- In Pulse Length/Width/Duration Modulation the pulse width or length or duration is changed in proportion to the amplitude of the modulating signal.



PLM/PWM/PDM

- Saw tooth Generator generates saw tooth signal of frequency f_s sampling frequency
- Message/Modulating signal x(t) is applied to Non Inverting Terminal of comparator.
- Output of Comparator is PLM/PWM/PDM.



PLM/PWM/PDM

Advantages:-

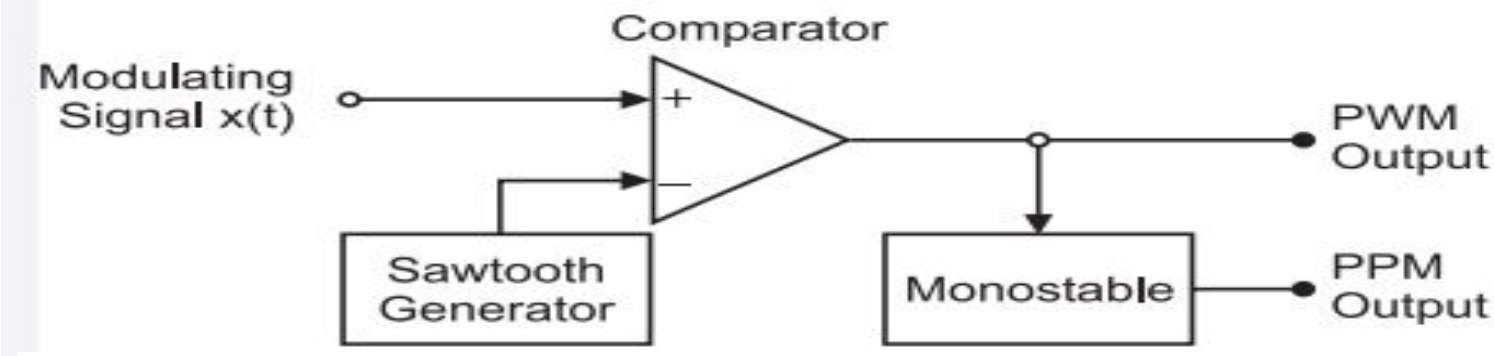
- Better Noise Immunity.
- Synchronization between Tx and Rx is not required.
- Possible to reconstruct the signal from Noise.

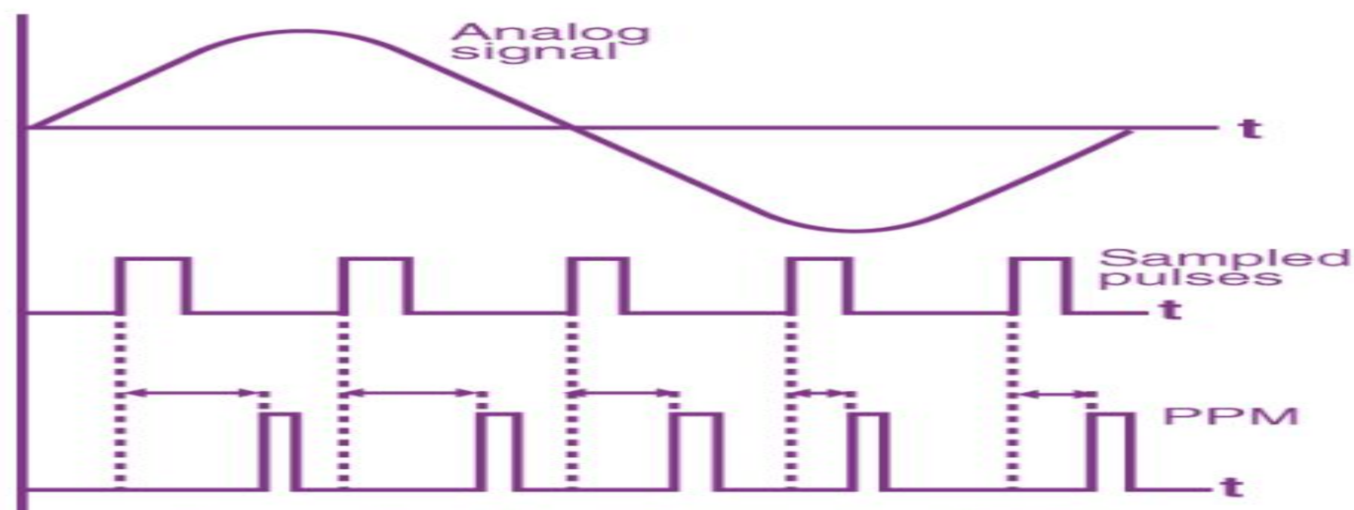
Disadvantages:-

- Requires large Bandwidth compared to PAM Signal.
- Tx must handle the power content of pulse with maximum width.

Pulse Position Modulation(PPM)

- Amplitude and Width is Kept Constant.
- Position of each pulse is changed with respect to the amplitude of sampled Value.





Pulse Position Modulation (PPM)

Pulse Position Modulation(PPM)

- Pulse Width Modulated signal:
- Output of Comparator is fed to monostable multivibrator.
- Multistable Vibrator is negative edge triggered.
- Pulse position modulated signal is obtained at the falling edge of triggering clock Pulse.

Pulse Position Modulation(PPM)

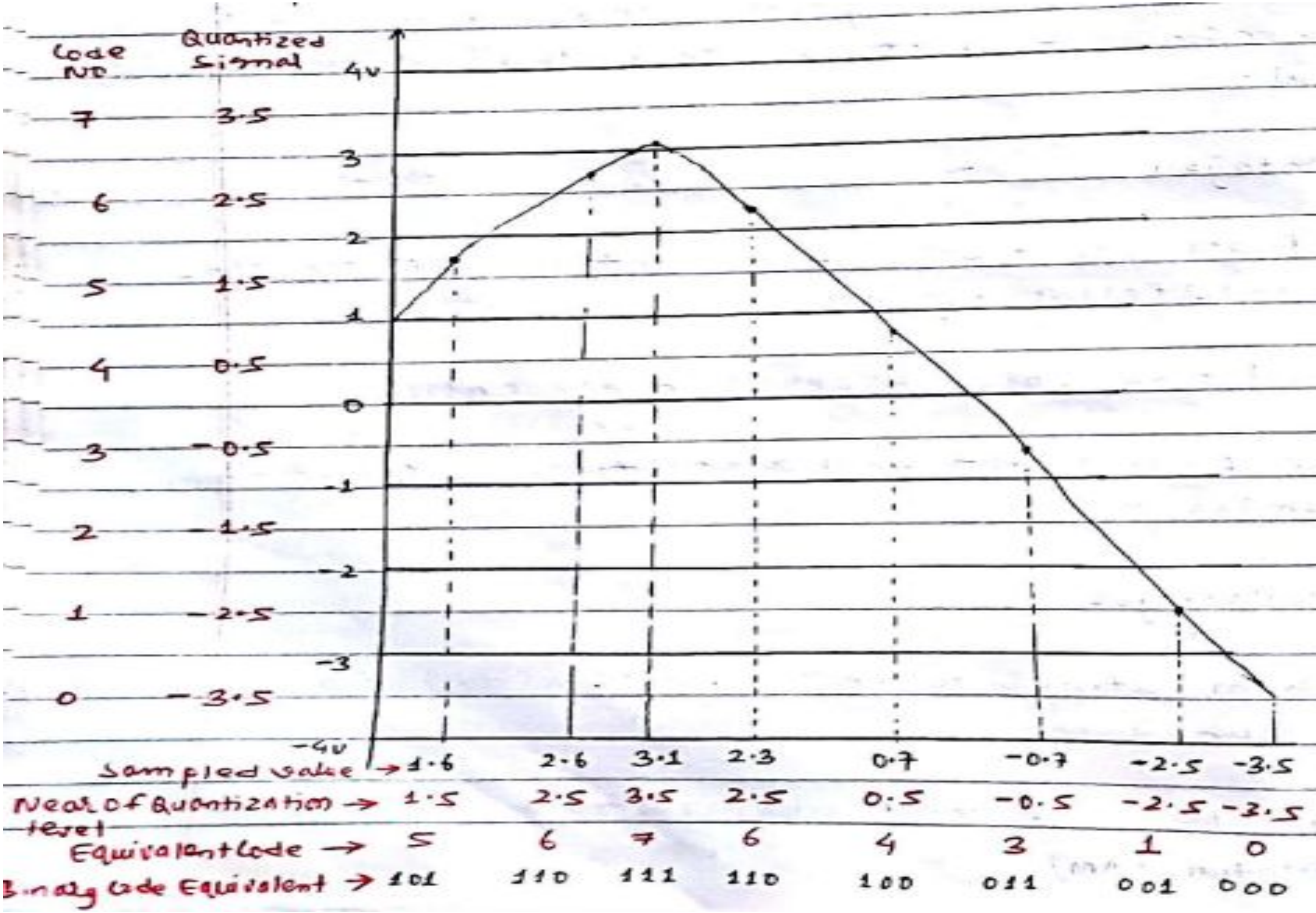
Advantages:-

- PPM has less interference of noise.
- Separation of Signal and Noise is easy.
- Pulse length/Pulse amplitude is constant: Requirement of transmission power is same.

Disadvantages:-

- Synchronization between Tx and Rx is required.
- Large / More bandwidth is required as compared to PAM.

Pulse Code Modulation(PCM)



Pulse Code Modulation(PCM)

- Digital Pulse modulation Technique.
- Analog signal is sampled and converted into digital encoded signal.
- Encoded signal is represented with n-bit binary code.
- Three basic operation in PCM

- Sampling
- Quantization
- Encoding

Pulse Code Modulation(PCM)

- Quantization:- Process of representing analog sampled values to a finite set of level.
- Finite set of level is a discrete amplitude value from 0 to maximum level.
- Two types of Quantization:-

- Uniform Quantization.
- Non Uniform Quantization.

Pulse Code Modulation(PCM)

Uniform Quantization:-

- Quantization level are uniformly spaced.
- Same step size.
- Input is divided into interval of equal size.
- Two types of Uniform Quantization

- Mid Rise type Quantizer - Mid Tread Quantizer

Mid Rise Quantizer :-

- Origin lies in the middle of a raising part of the stair-case like graph.
- The quantization levels are even in number.

Pulse Code Modulation(PCM)

Mid Tread Quantizer:-

- Origin lies in the middle of a tread of the stair-case.
- Quantization levels are odd in number.

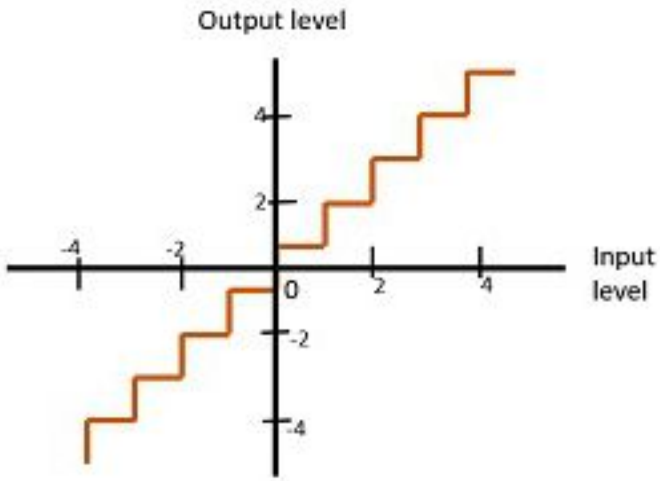


Fig 1 : Mid-Rise type Uniform Quantization

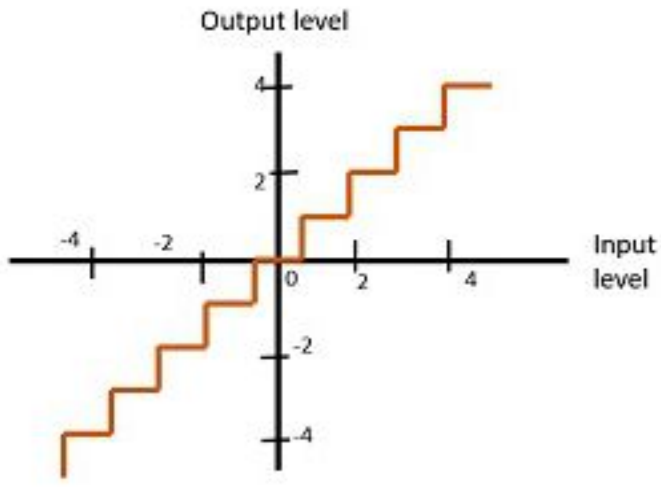


Fig 2 : Mid-Tread type Uniform Quantization

Pulse Code Modulation(PCM)

Non Uniform Quantization:-

- Step size is not same and fixed.
- Step size is small or reduced for small amplitude or weak signal.
- Step size is increased or big for large amplitude or strong signal.
- Non Uniform Quantization is achieved practically by Companding. (Improves SQNR).

Companding

- Process of compressing signal at the Transmitter (Tx) side.
- Expansion on the Receiver(Rx) Side.
- Improves the SQNR.
- Two types of Companding Technique:-

- μ Law Companding.
- A Law Companding.

μ Law Companding

- Compressor Characteristics is Continuous.
- Approximately linear for small Value of input level.
- Approximately logarithmic for high input level.
- Compressed Output is given as

$|v| = \log(1+ \mu |x|/ x_{max}) / \log(1+ \mu)$

V= Normalized Compressed Output voltage.

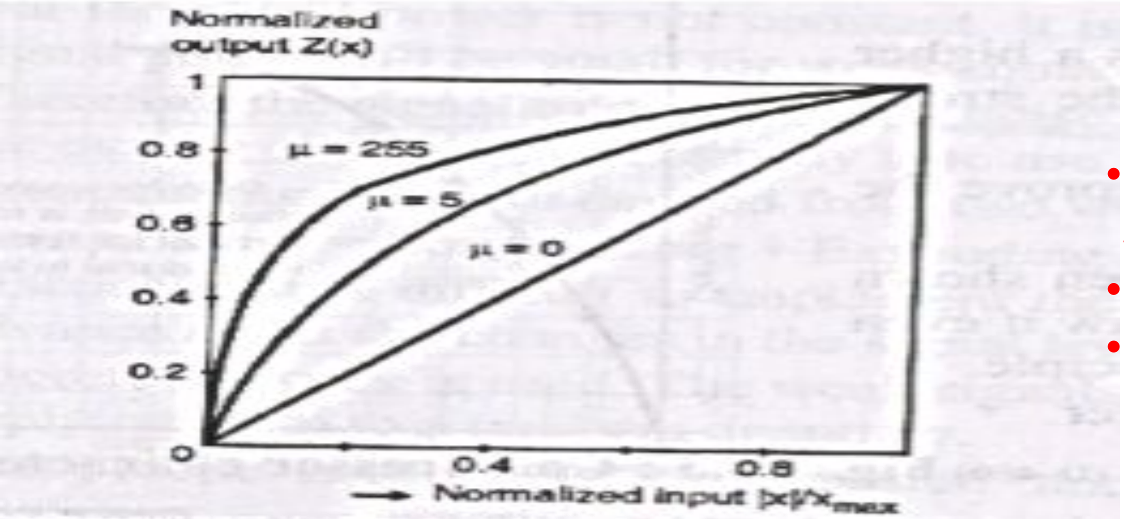
μ = Parameter to define amount of compression

x_{max} = Maximum Value

$|x|/ x_{max}$ = Normalized value of Input W.r.t maximum Value.

μ Law Companding

- American Standard. U.S, Canada and Japan.
- Compressor Characteristics neither strictly Linear nor strictly Logarithmic.
- Practical value of μ=255.



- Used for speech and music signal. PCM Telephones.
- μ= 0 for Uniform Quantization
- Better in Signal Quality.

Compressor Characteristics of a μ Law Compressor

A Law Companding

- Compressor Characteristics is Piecewise.
- Linear segment for low level input.
- Logarithmic curve for high level input.

$|V| = \log(1+ A\, |x|/ x_{max}\ ) / 1+ \log A$ for $1/A \leq |x|/x_{max} \leq 1$

$|V| = A\, |x|/ x_{max}\ ) / 1+ \log A$ for $0 \leq |x| \leq 1/A$

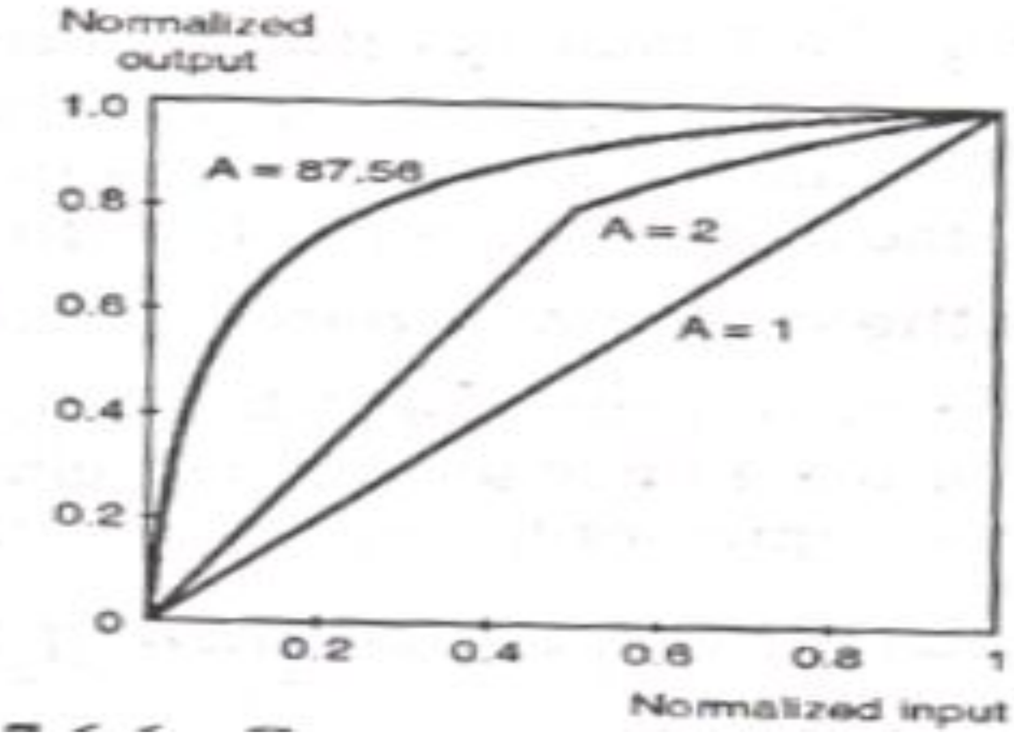
V= Normalized Compressed Output voltage.

A = Parameter to define amount of compression

x_{max} = Maximum Value

$|x|/ x_{max}$ = Normalized value of Input W.r.t maximum Value.

A Law Companding



-European Standard used in Europe and rest of the world.
-Used in PCM Telephone System.

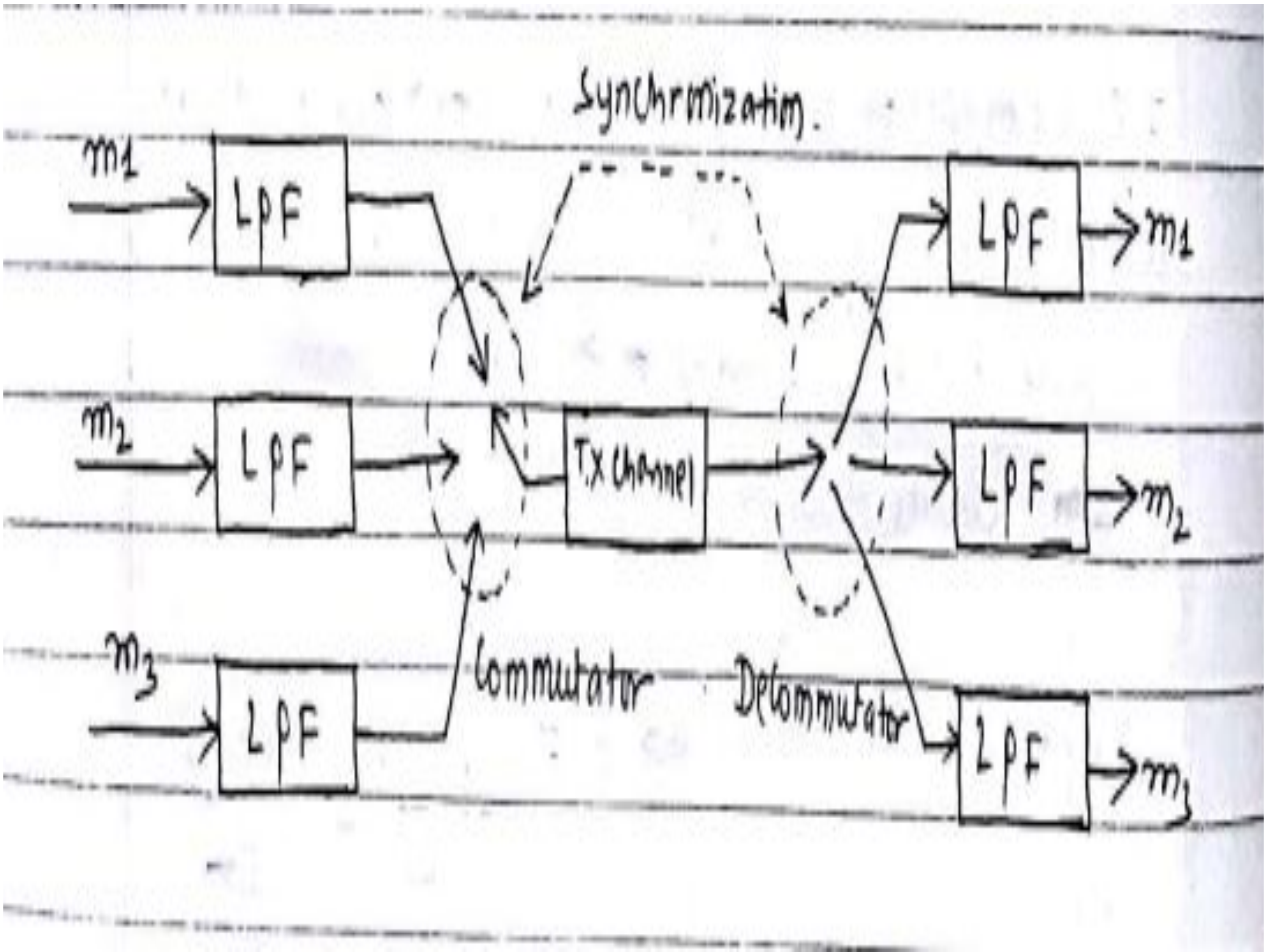
Necessity of Non Uniform Quantization for Speech Signal

- Crest factor = Peak Value of signal/ rms value of signal.
- X(t) is the input to quantizer with its amplitude $-X_{max}$ to $+ X_{max}$
- Peak Value of Signal= X_{max}
- rms value of signal=$\sqrt{X^2(t)}$
- Power is defined as P= $X^2(t)/R$
- For normalized power R=1

Necessity of Non Uniform Quantization for Speech Signal

- $P = X^2(t)$
- Crest factor = $X_{max} \,/\, \sqrt{X^2(t)}$
- For normalized signal x(t) has $X_{max} = 1$
- Crest factor = $1 \,/\, \sqrt{X^2(t)}$
- Crest factor = $1/\sqrt{P}$
- $P<<1$ will decrease the SQNR.

Time Division Multiplexing with PCM



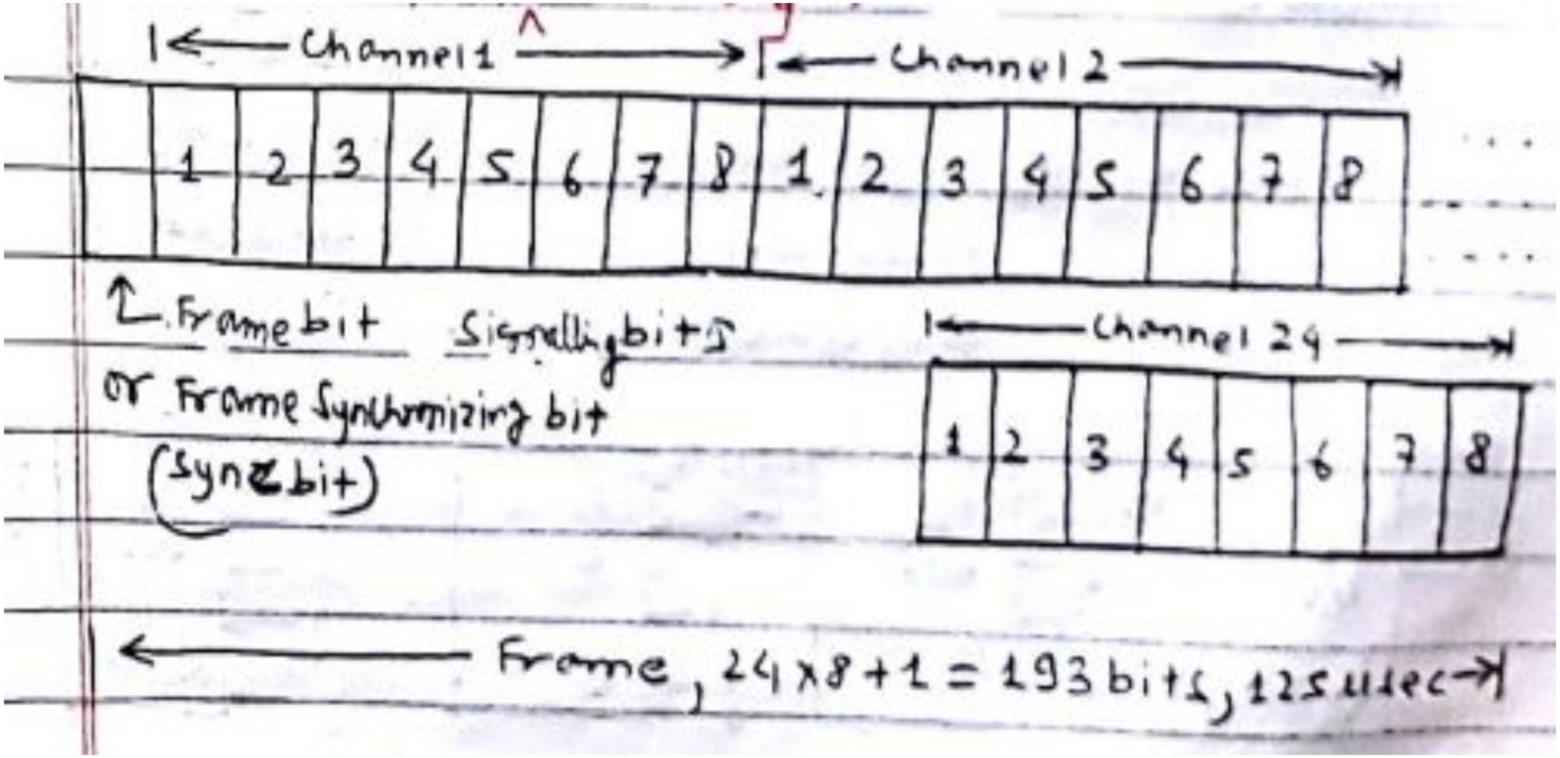
Time Division Multiplexing with PCM

- In Time Division Multiplexing, samples of messages are transmitted at some fixed interval of time.
- Samples of signal are transmitted serially and recovered separately on destination.
- Commutator takes low pass filtered signal sequentially at fixed interval of time over same channel.

Time Division Multiplexing with PCM

- Decommutator at receiving end separates the signal.
- Commutator and Decommutator are synchronized
- Low Pass Filter(LPF) at the receiving end converts the sampled signal into original signal.

T1 TDM-PCM Telephone Hierarchy



T1 = 8000 frames/sec x ((24 x 8) + 1) bits/frame

= 1544000 bits/sec

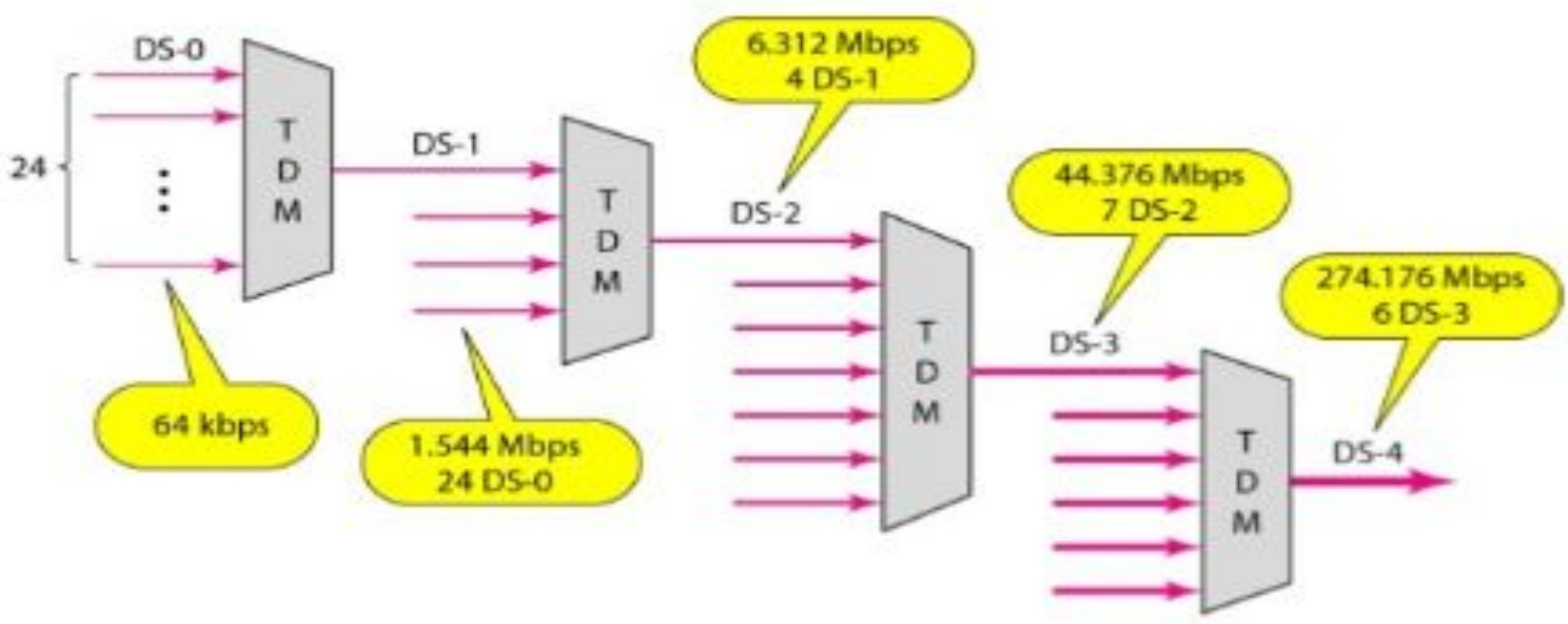
= 1.544 Mbps

T1 TDM-PCM Telephone Hierarchy

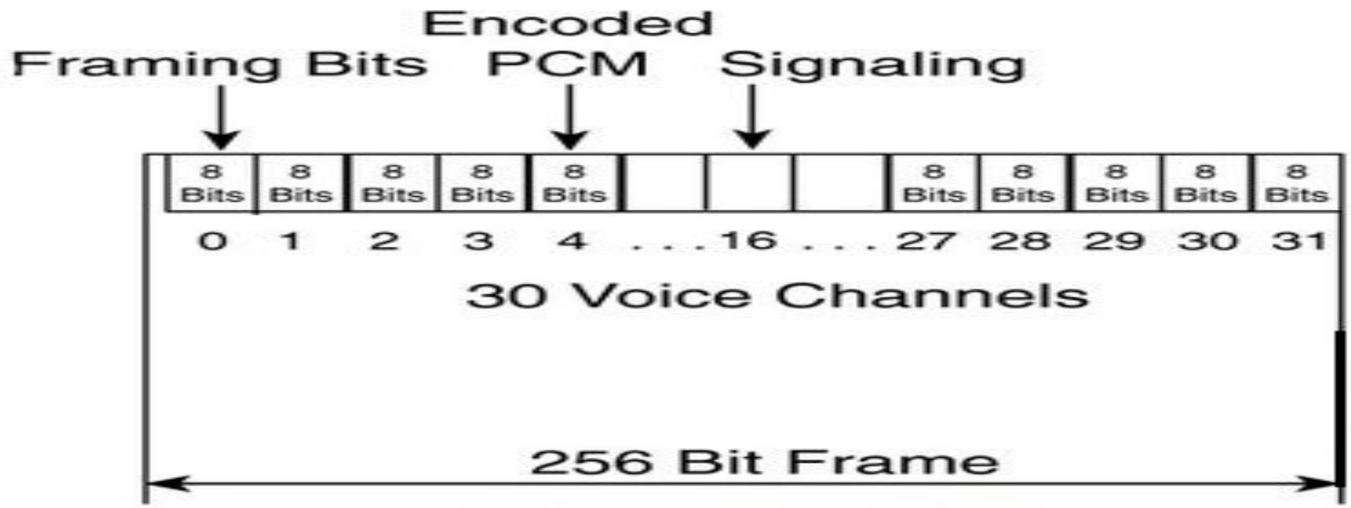
- T1 System is North American digital multiplexing standard recognized by ITU-T.
- 24 voice channel band limited to 300-3400 MHz.
- Sampling frequency of 8 KHz.
- Each Sample converted to 7 bit word and 1 bit is reserved for synchronization.
- Similarly there are other standard of T1 as:-

T1 TDM-PCM Telephone Hierarchy

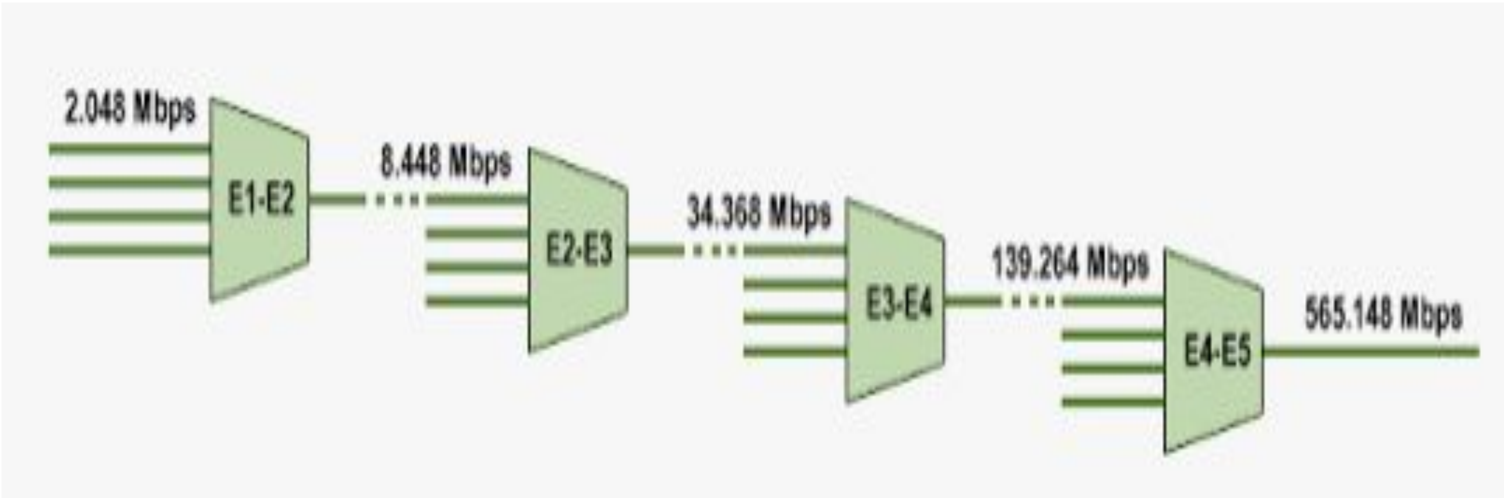
Standard	Channel	Signaling Rate
T2 = 4*T1	96 Channel	6.312 Mbps
T3 = 7 * T2	672 Channel	44.736 Mbps
T4 = 6 * T3	4032 Channel	274.176 Mbps
T5 = 2 * T4	8064 Channel	560.16 Mbps



E1 TDM-PCM Telephone Hierarchy



E1- Frame Format



E1 TDM-PCM Telephone Hierarchy

- E1 System is European digital multiplexing standard recognized by ITU-T.
- 32 voice channel band limited to 300-3400 MHz.
- Sampling frequency of 8 KHz.
- Each frame is divided into 32 equal time slot.
- Two time slot is reserved for Signaling & Controlling.

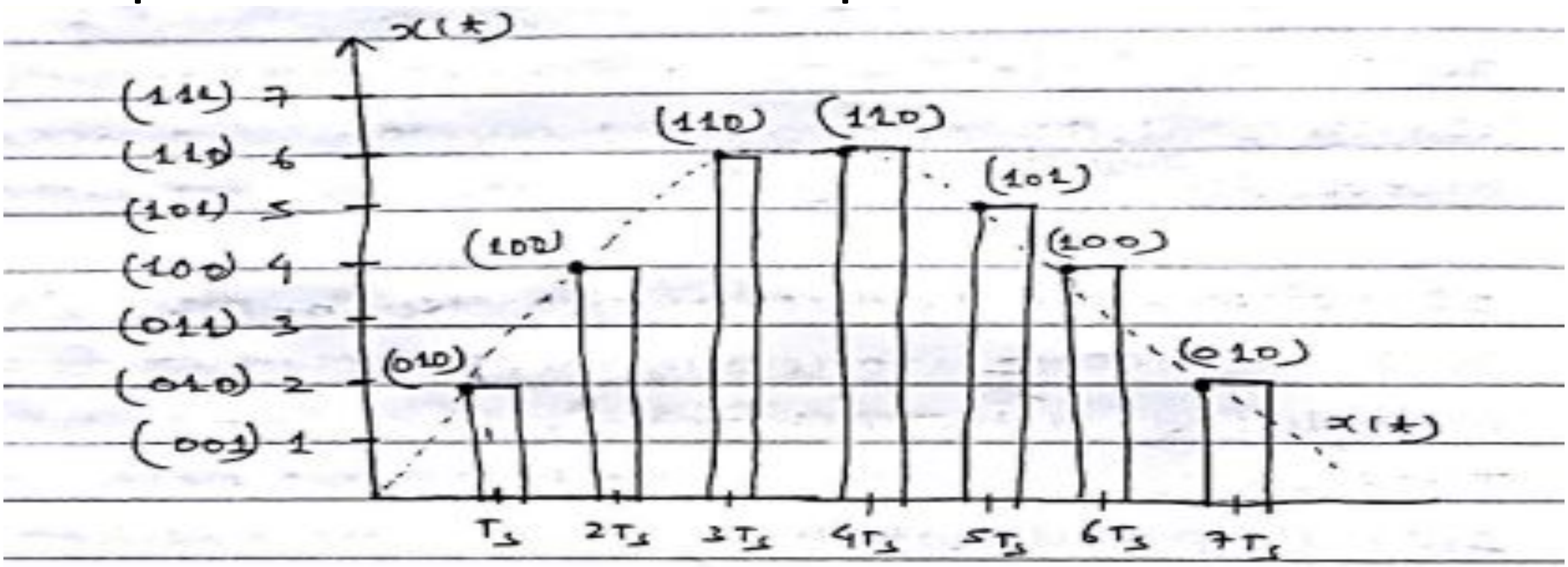
E1 TDM-PCM Telephone Hierarchy

- Bit rate = 32*8 bits/frame * 8000 sample/sec
- Bit Rate = 2.048 Mbps.
- Other standards

Standard	Channel	Signaling Rate
E2 = 4*E1	120	8.448 Mbps
E3 = 4*E2	480	34.368 Mbps
E4 = 4*E3	1920	139.264 Mbps

Differential Pulse Code Modulation(DPCM)

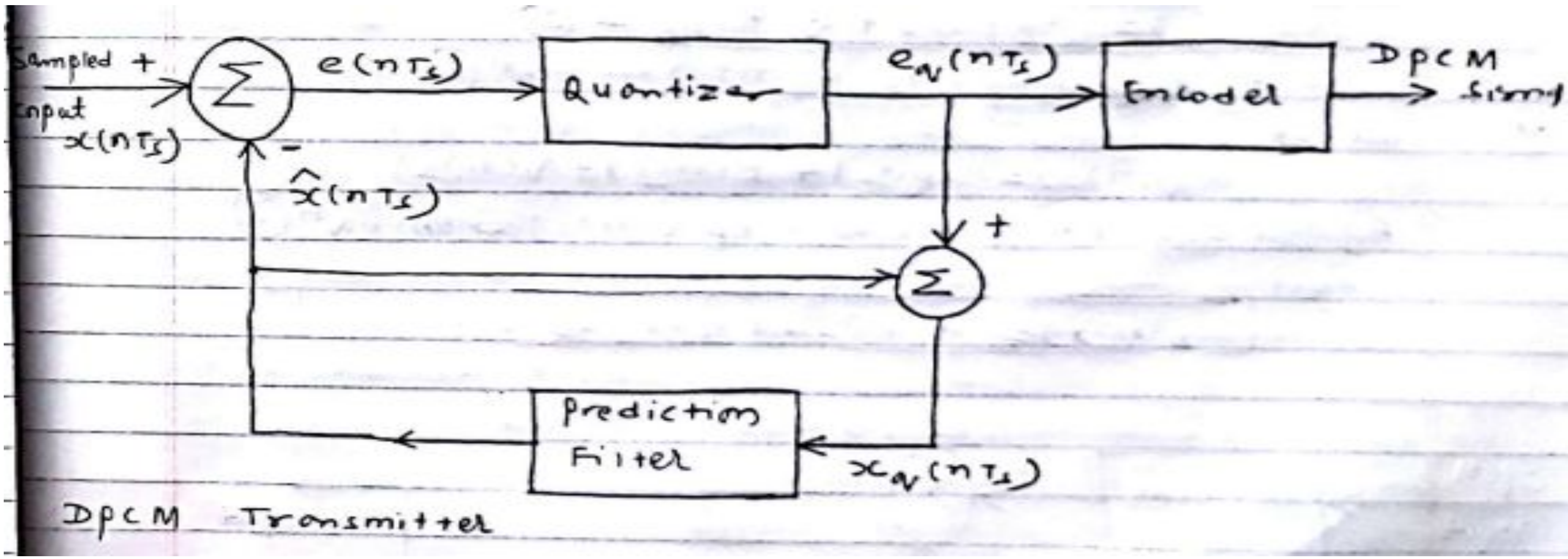
- Each sample is quantized in independent manner in PCM.
- Previous sample value has no effect on quantization of new sample.



Differential Pulse Code Modulation(DPCM)

- Samples taken at 3Ts and 4Ts are encode with same value (110).
- Single sample can be send in DPCM.
- Samples at 5Ts and 6Ts @ Difference between sample is last bit only.
- Two repeated bit(Redundant bit) can be removed and only the difference third bit can be send to represent the whole sample value.

Differential Pulse Code Modulation(DPCM)



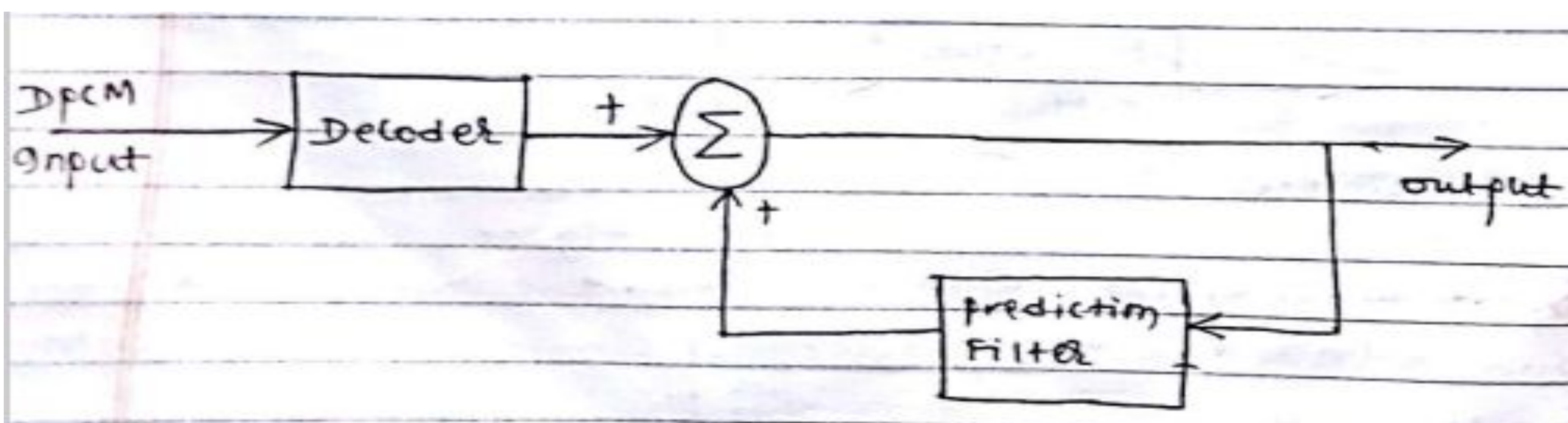
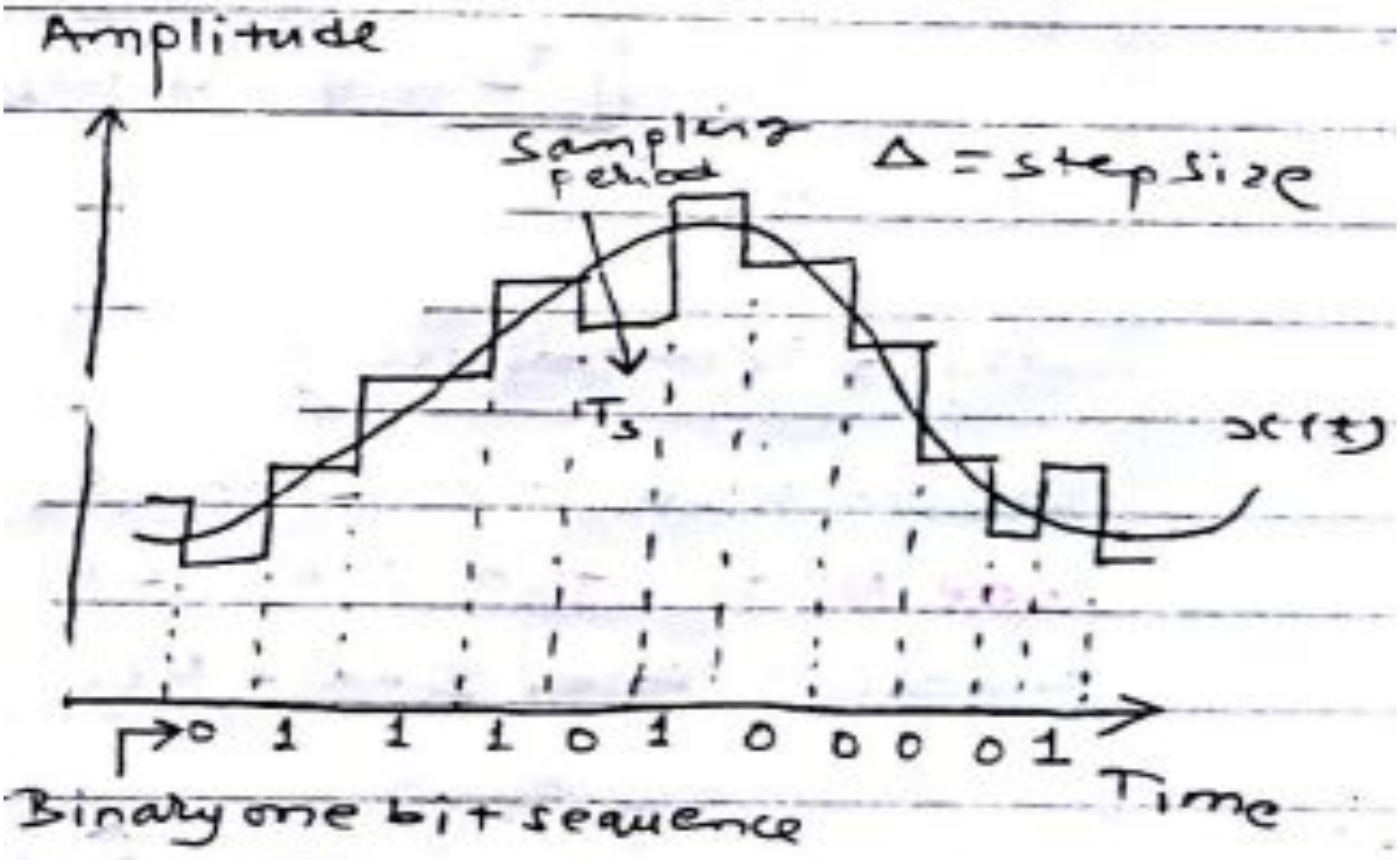


fig :→ DPCM Receiver

Differential Pulse Code Modulation(DPCM)

- DPCM works on the principle of prediction.
- Present Sample is predicted on the basis of past sample.
- Prediction may not be exact.
- Very Close to actual sample value.

Delta Modulation



Delta Modulation

- Delta modulation transmits one bit per sample.
- DPCM works on the principle of Comparison.
- Present Sample is compared with the previous sample value.
- Present sample value is smaller than previous sample value : defined by $-\Delta$ level and “0” is transmitted.

Delta Modulation

- Present sample value is smaller than previous sample value : defined by +Δ level and “1” is transmitted.
- It is staircase approximation of the input waveform.
- Each Step is represented by 1 for the rise of step.
- Each Step is represented by 0 for the fall of step.

Delta Modulation

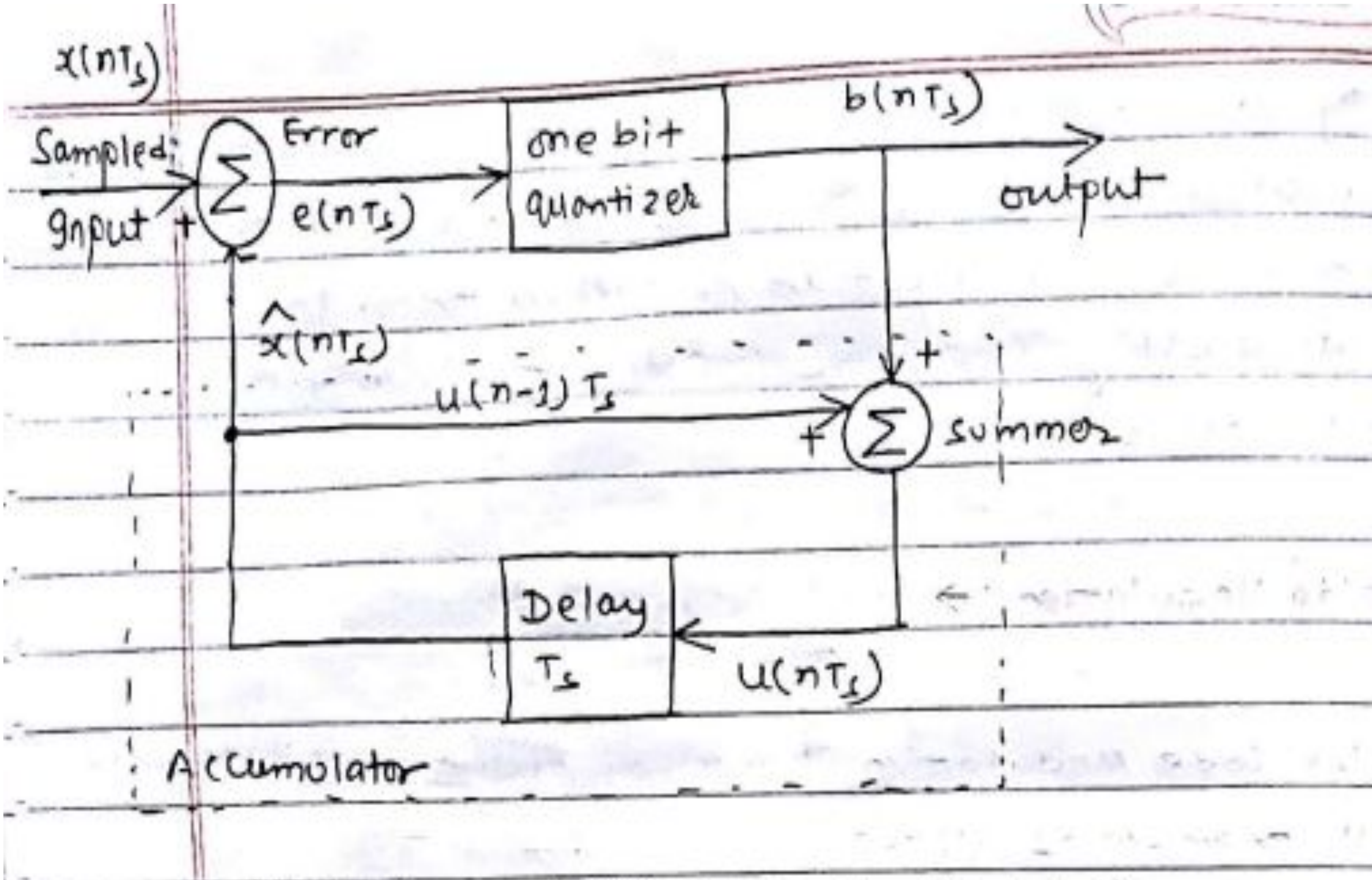


fig:→ Transmitter of Delta Modulation

Delta Modulation

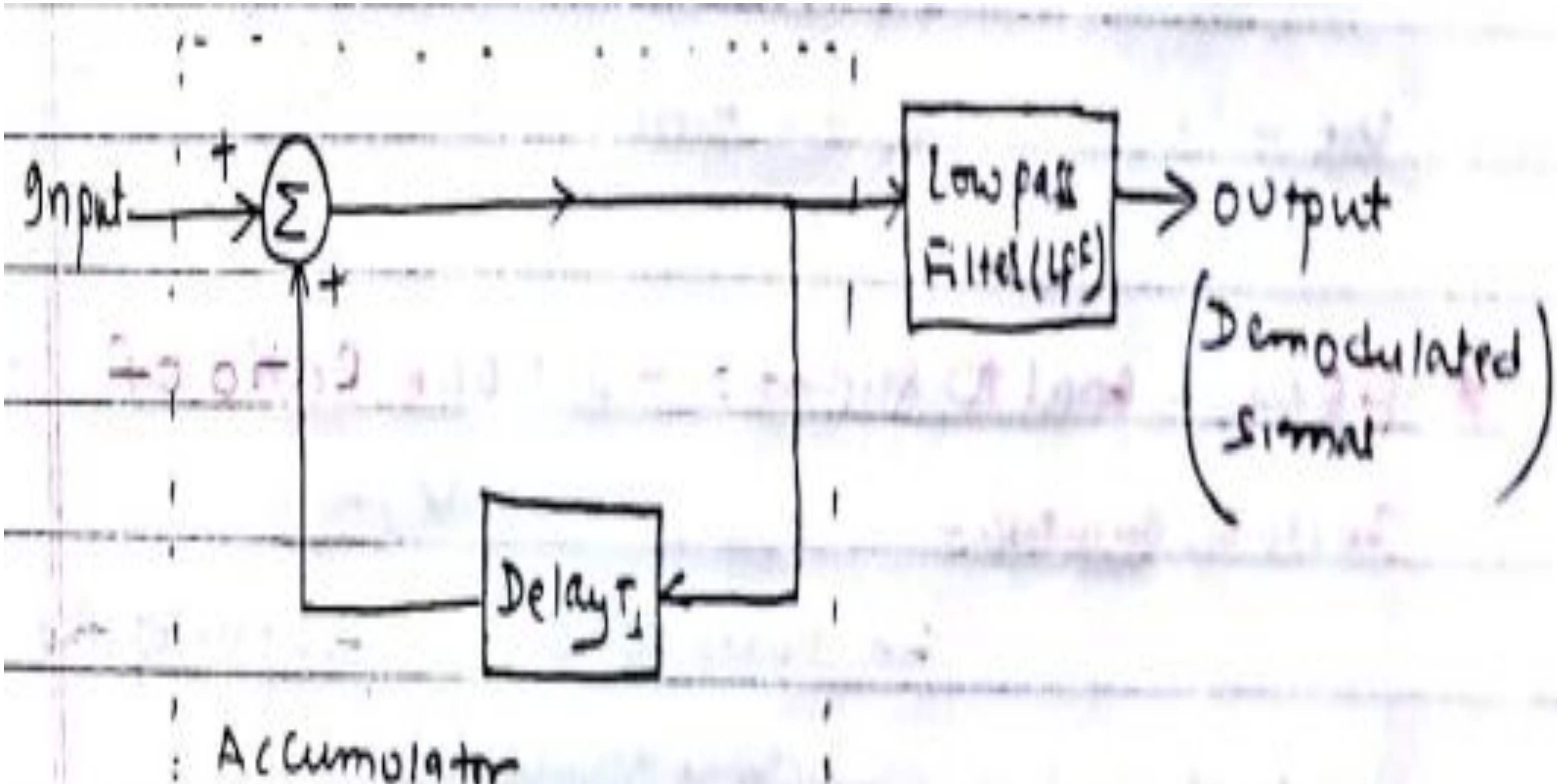
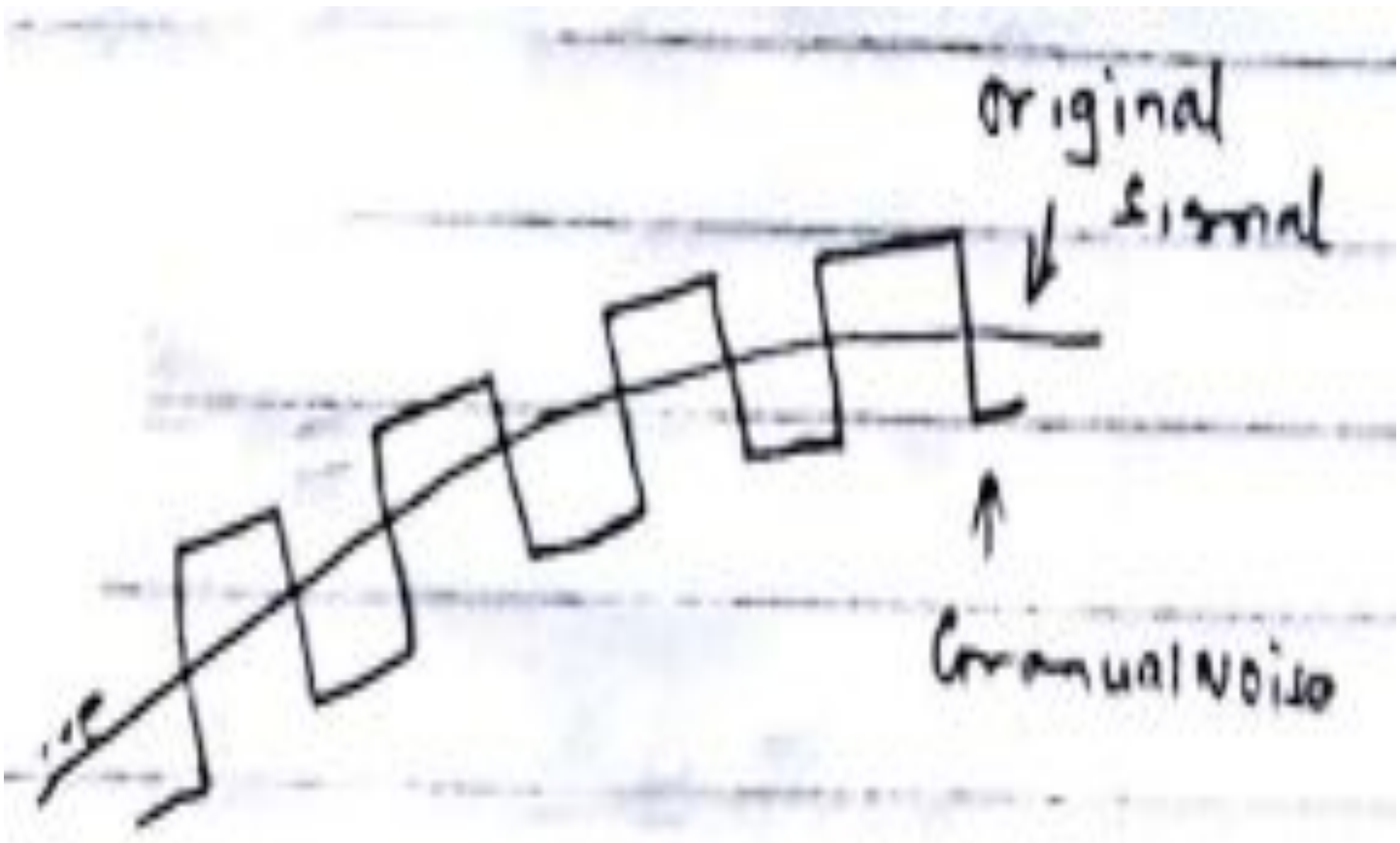


fig:→ Delta Modulation Receiver

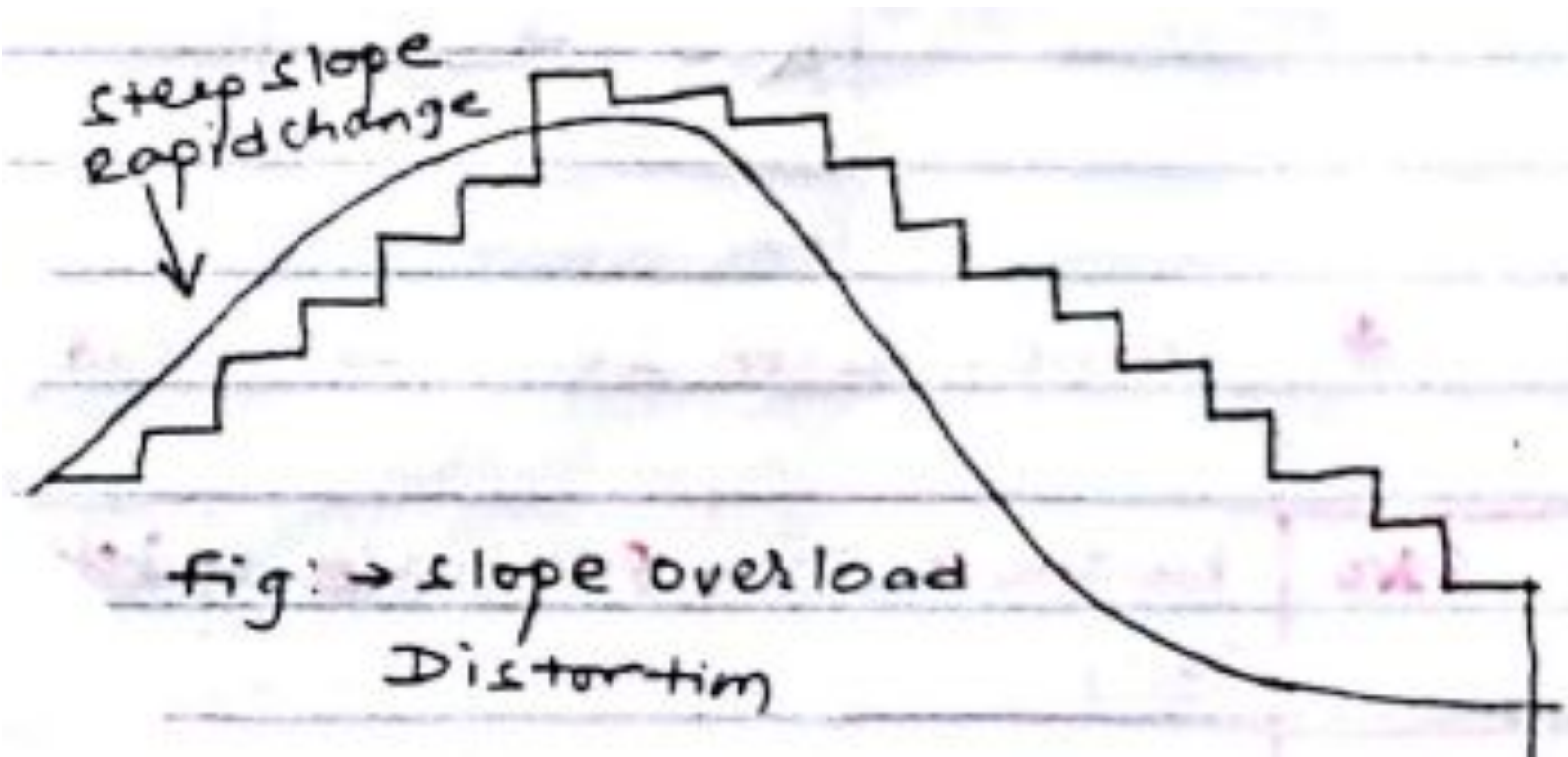
Quantization Noise in Delta Modulation

- Granual noise is introduced if the step size is very large compared to the input signal.



Quantization Noise in Delta Modulation

- Slope Overload Distortion is introduced if the input signal slope is very large compared to the input signal.



Parametric Speech Coding Vocoders, Linear Prediction Coding

- Digital Speech coders classified into two categories:-
- Waveform Coder:- Uses Algorithm to encode and decode.

- Output is the approximation of the input.
- Provides high quality of signal but requires relatively high bandwidth / bit rate.

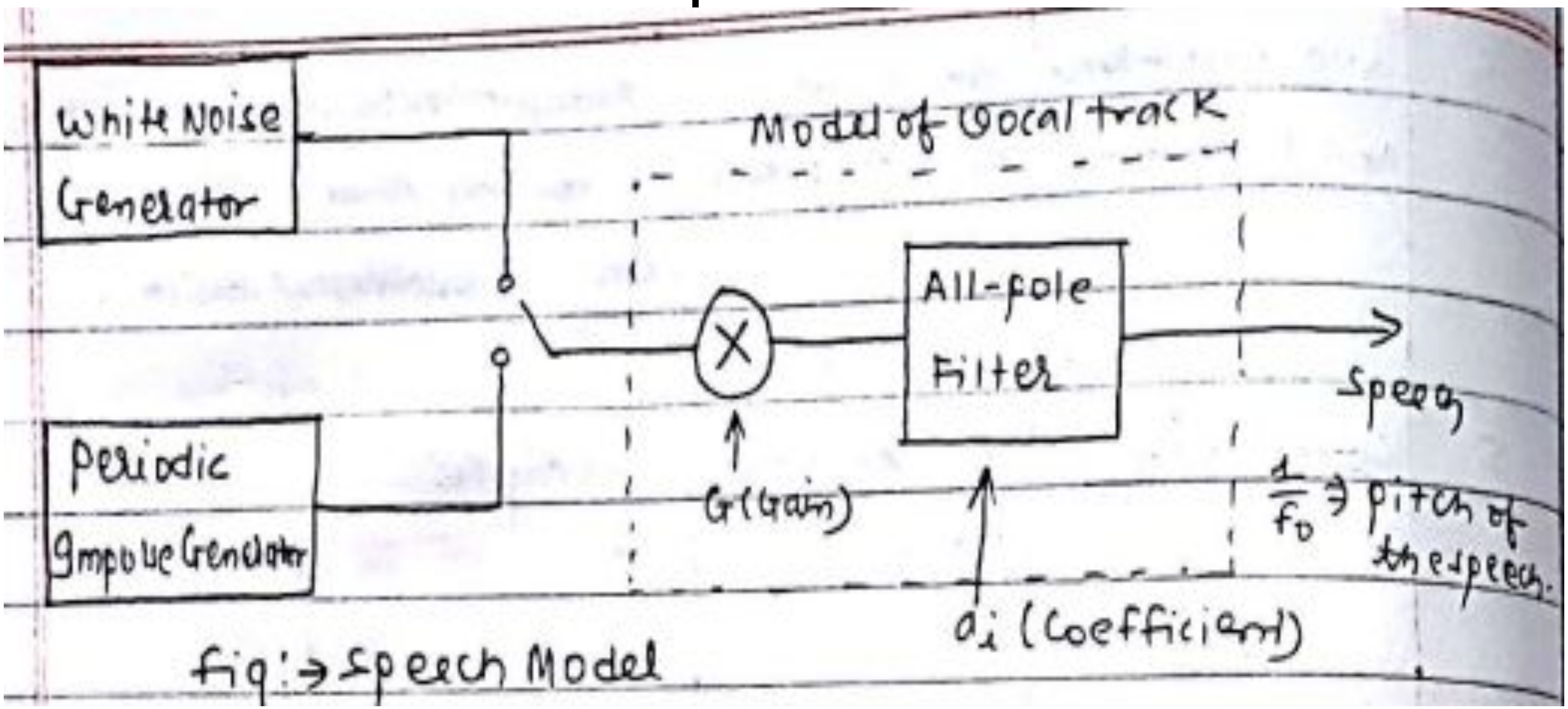
Parametric Speech Coding Vocoders, Linear Prediction Coding

- VoCoder:-

- Encode the speech signal by modeling the signal extracting set of parameters.
- Original voice is predicted using these parameters extracted at the transmitter.
- This technique of coding of speech is Linear Prediction Coding (LPC).
- It requires relatively less bandwidth / bit rate.

Parametric Speech Coding Vocoders, Linear Prediction Coding

- VoCoder:- Speech signal is modeled with parameter like repetition frequency “Fo”, all pole filter “ Ao” Gain parameter “G”.



Unit-3

Thank you