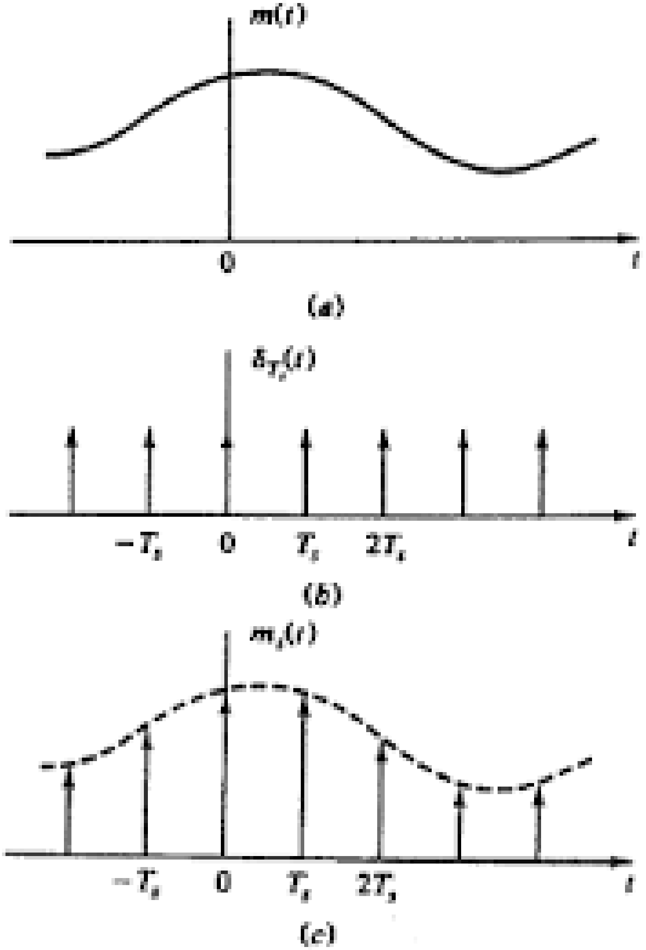
Digital Communication

Satish Chandra

Sampling Process

- It is an operation that is basic to digital signal processing and digital communications.
- Through use of the sampling process an analog signal is converted into a corresponding sequence of samples that are usually spaced in time.



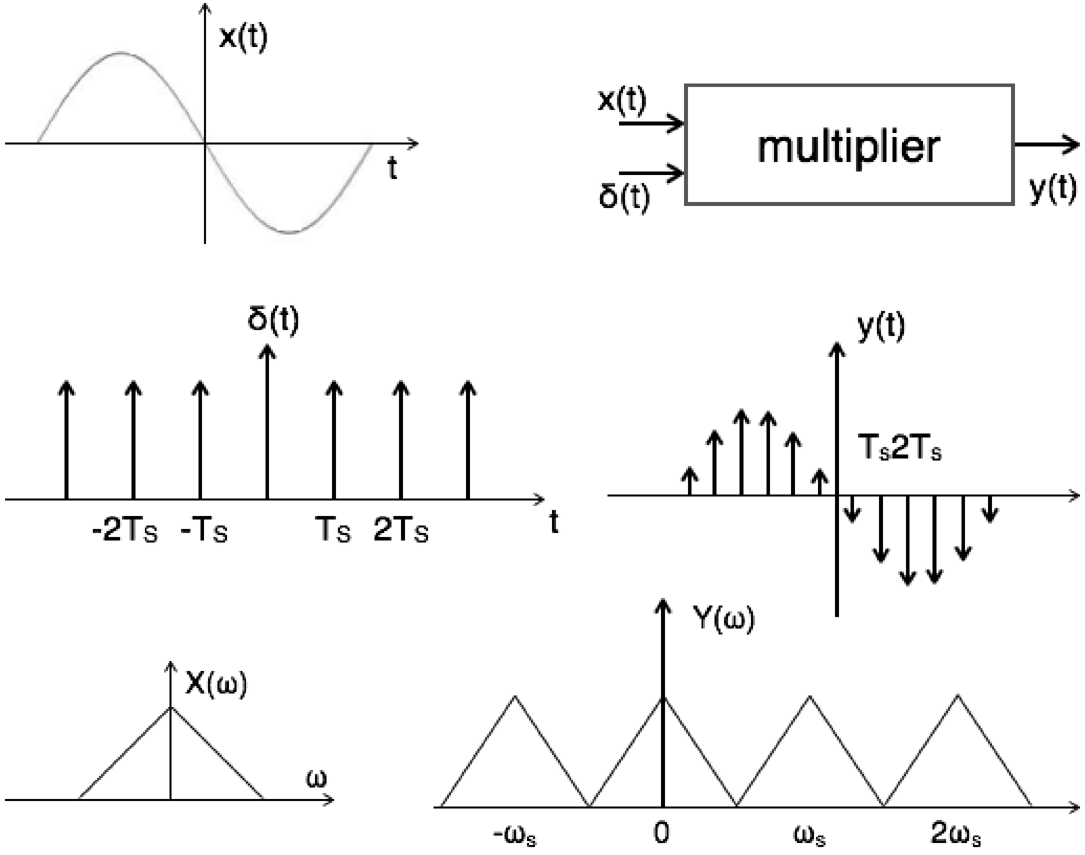
Sampling Theorem

- The sampling theorem shows that a band-limited continuous signal can be perfectly reconstructed from a sequence of samples if the highest frequency of the signal does not exceed half the rate of sampling.
- The Sampling theorem states that, if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed with minimal distortion.

$$f_s \geq 2f_m$$

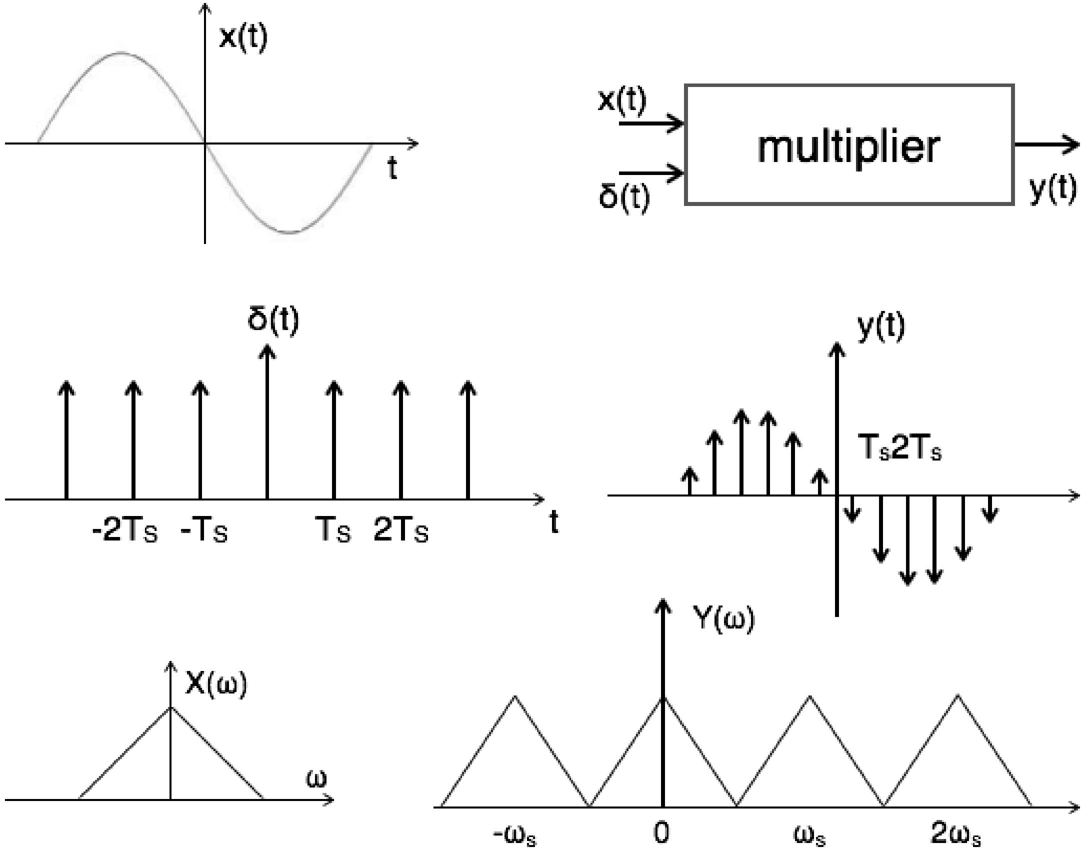
Sampling Theorem

- A signal x(t) which is to be sampled.
- A periodic train of pulses δ(t) of unit amplitude and of period T_s.
- The pulses are narrow, having a width dt.
- The two signals x(t) and δ(t) are applied to a multiplexer, which then yield as an output y(t)=x(t).δ(t)



Sampling Theorem

- The product is the signal sampled at the occurrence of each pulse.
- That is, when a pulse occurs, the multiplexer output has the same values as does x(t) has, and at all other times it is zero.



Sampling Theorem

The process of sampling can be explained by the following mathematical expression:

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

The trigonometric Fourier series representation of $\delta(t)$ is given by

$$\delta(t) = a_0 + \sum_{n=1}^{\infty} (a_n \cos n\omega_S t + b_n \sin n\omega_S t)$$

Sampling Theorem

Where, $a_0 = \frac{1}{T_s}\int_{-T/_2}^{T/_2} \delta(t)dt = \frac{1}{T_s}\delta(0) = \frac{1}{T_s}$

$$a_n = \frac{2}{T_s}\int_{-T/_2}^{T/_2} \delta(t)\cos n\omega_S t\, dt = \frac{2}{T_s}\delta(0)\cos n\omega_S t\,[0] = \frac{2}{T_s}\cos n\omega_S t$$

$$b_n = \frac{2}{T_s}\int_{-T/_2}^{T/_2} \delta(t)\sin n\omega_S t\, dt = \frac{2}{T_s}\delta(0)\sin n\omega_S t\,[0] = 0$$

$$\delta(t) = \frac{1}{T_S} + \sum_{n=1}^{\infty}\left(\frac{2}{T_S}\cos n\omega_S t + 0\right)$$

Sampling Theorem

So,

$$y(t) = x(t).\delta(t) = x(t)\left[\frac{1}{T_s} + \sum_{n=1}^{\infty}\left(\frac{2}{T_S}\cos n\omega_S t + 0\right)\right]$$

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

$$y(t) = \frac{1}{T_s}[x(t) + 2\cos\omega_S t\,.\,x(t) + 2\cos 2\omega_S t\,.\,x(t) + 2\cos 3\omega_S t\,.\,x(t) + ..]$$

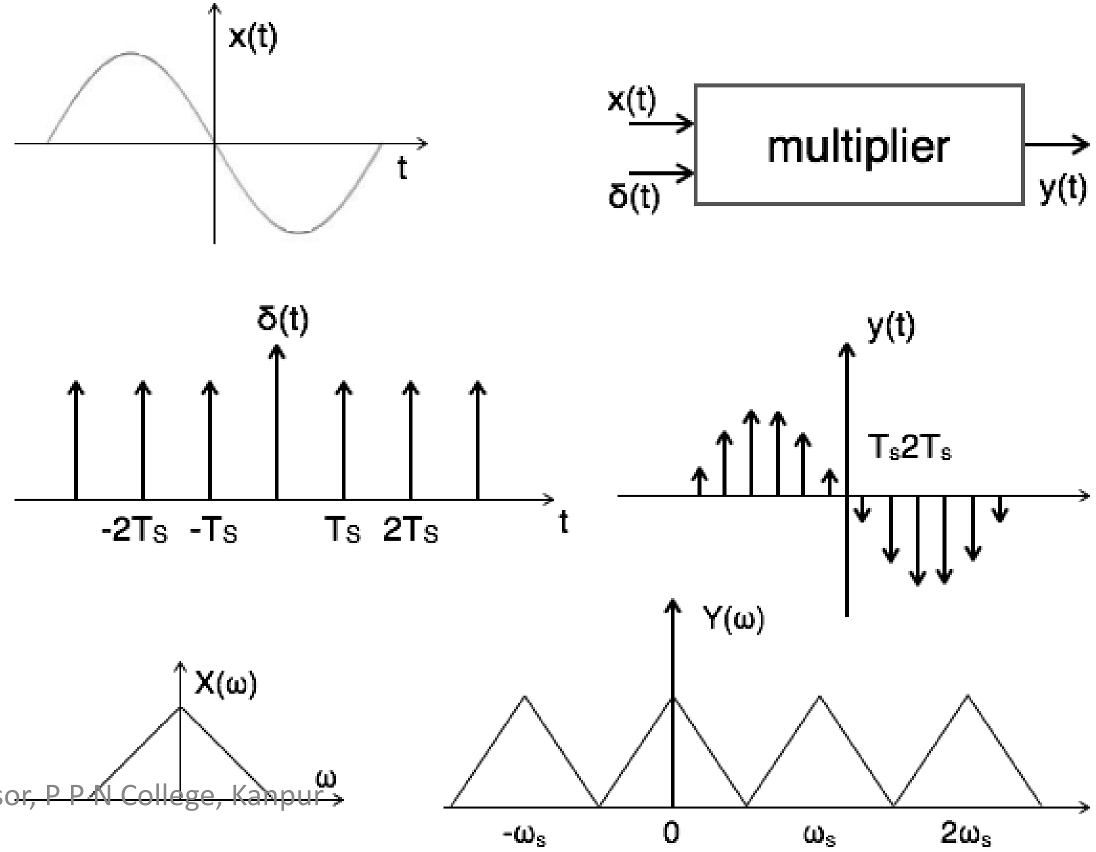
Sampling Theorem

Take Fourier transform on both sides.

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

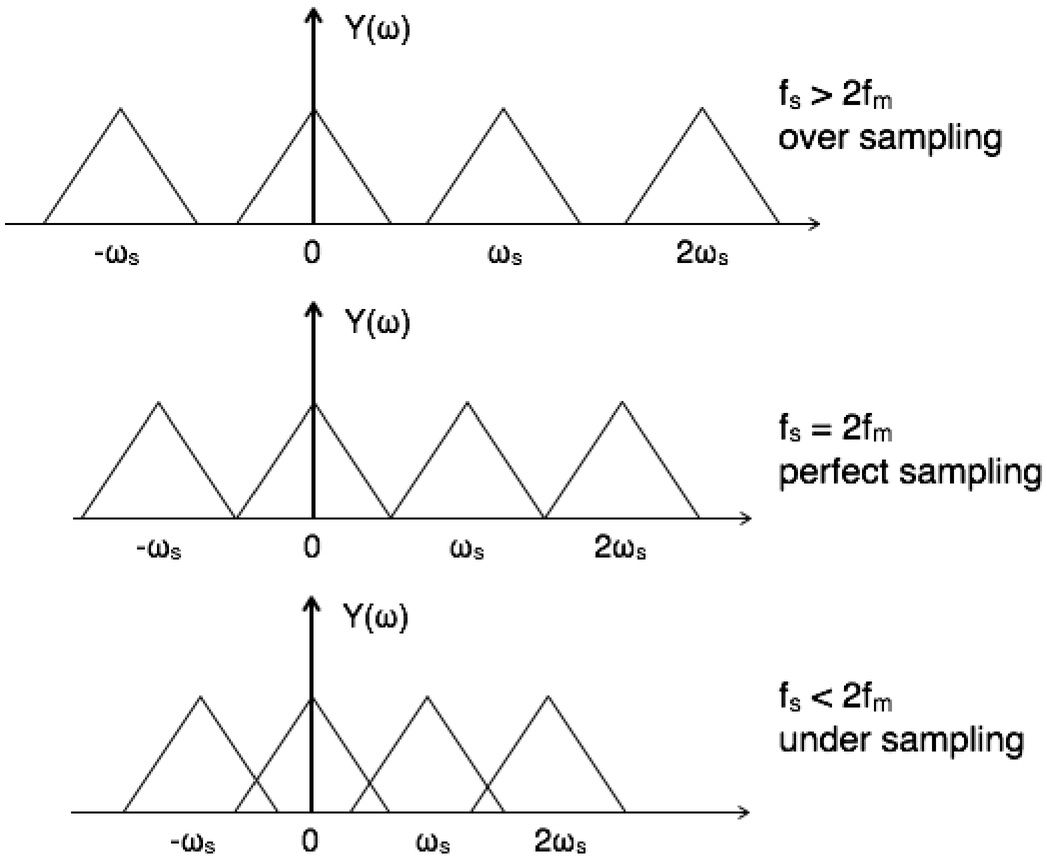
$$Y(\omega) = \frac{1}{T_S}\sum_{n=-\infty}^{\infty} X(\omega - n\omega_S)$$

$$n = 0, \pm 1, \pm 2, \pm 3, \ldots.$$



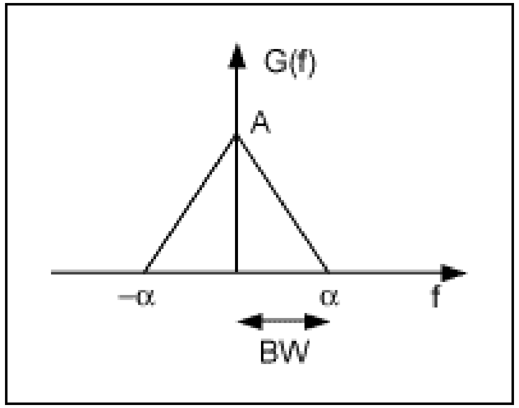
Sampling Theorem

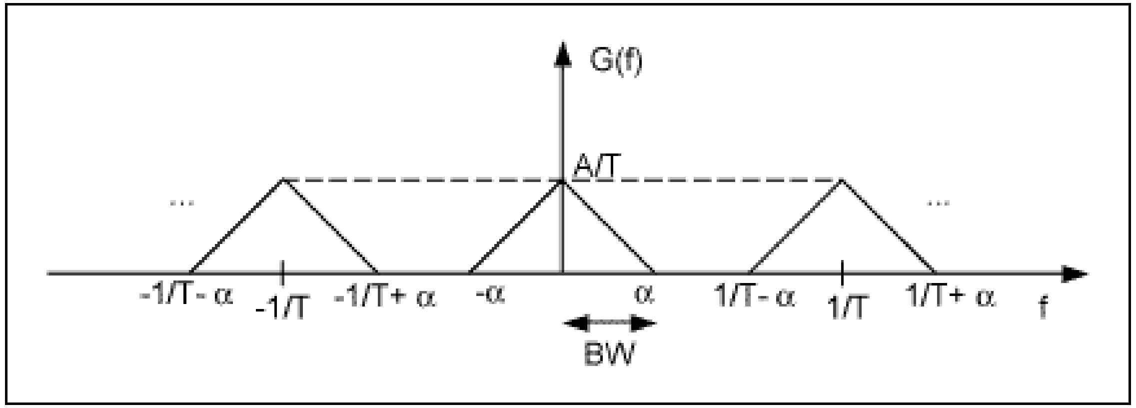
- To reconstruct x(t), recover input signal spectrum X(ω) from sampled signal spectrum Y(ω), which is possible when there is no overlapping between the cycles of Y(ω).
- Possibility of sampled frequency spectrum with different conditions is given in diagram.



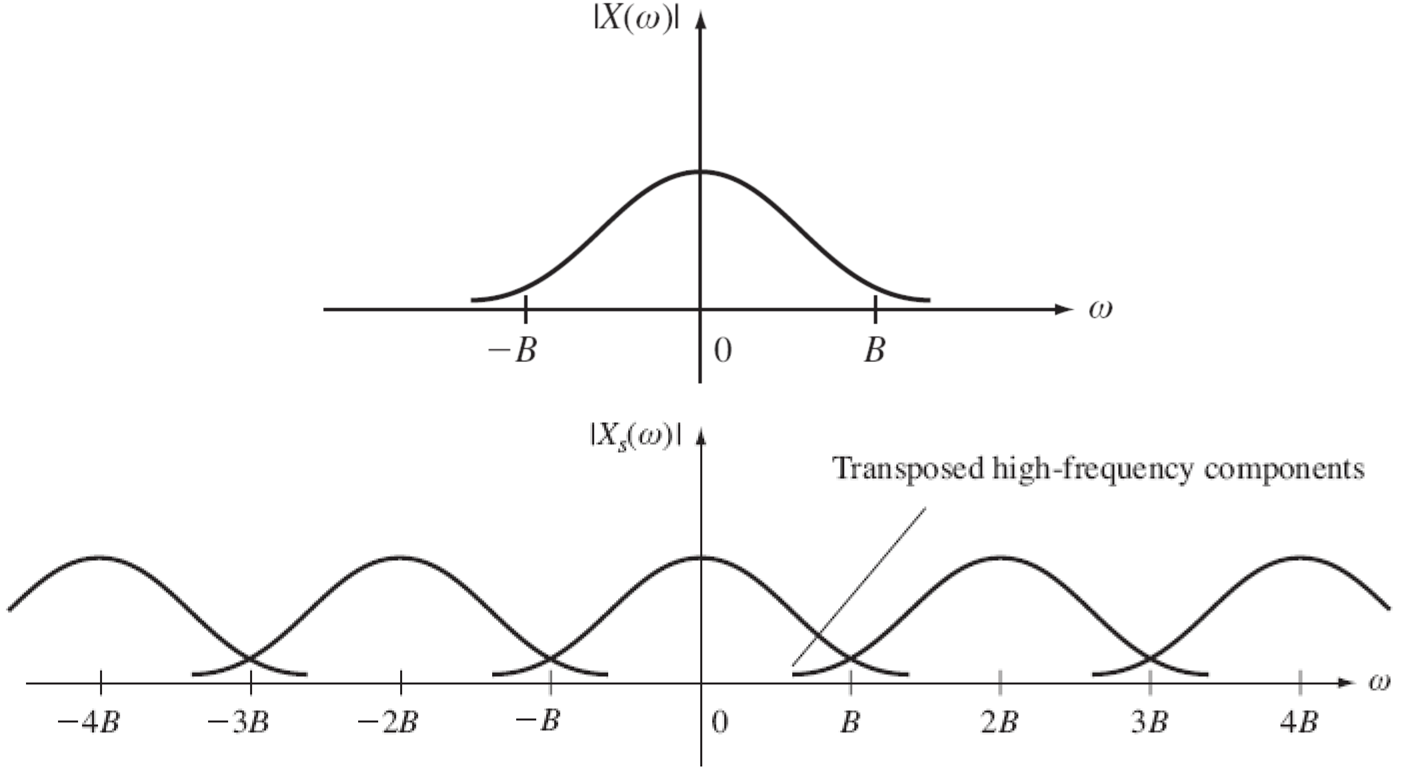
Band Limited Signal

- A signal is said to be band-limited if the amplitude of its spectrum goes to zero for all frequencies beyond some threshold called the cutoff frequency α.
- If the signal is not band-limited distortion will occur when the signal is sampled. We refer to this distortion as ***aliasing***.





Band Limited Signal



Sampling Theorem

Aliasing Effect

- The overlapped region in case of under sampling represents aliasing effect, which can be removed by
  - considering $f_s > 2f_m$
  - by using anti aliasing filters.

Nyquist rate

- The Nyquist sampling rate is two times the highest frequency of the input signal. For instance, if the input signal has a high-frequency component of 1 kHz, then the sampler must sample at least 2 kHz, or the signal might *alias*.
- When the sampling rate becomes exactly equal to $2f_m$ samples per second, then it is called the **Nyquist rate**. It is given by,

$$f_s = 2f_m$$

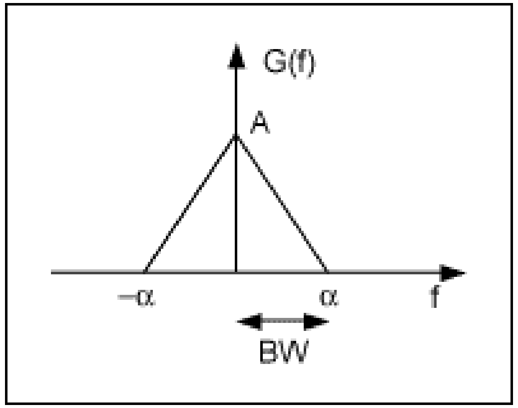
- It is also called the **minimum sampling rate**.
- The f_m is the highest frequency component of a bandlimited signal.

Nyquist Interval

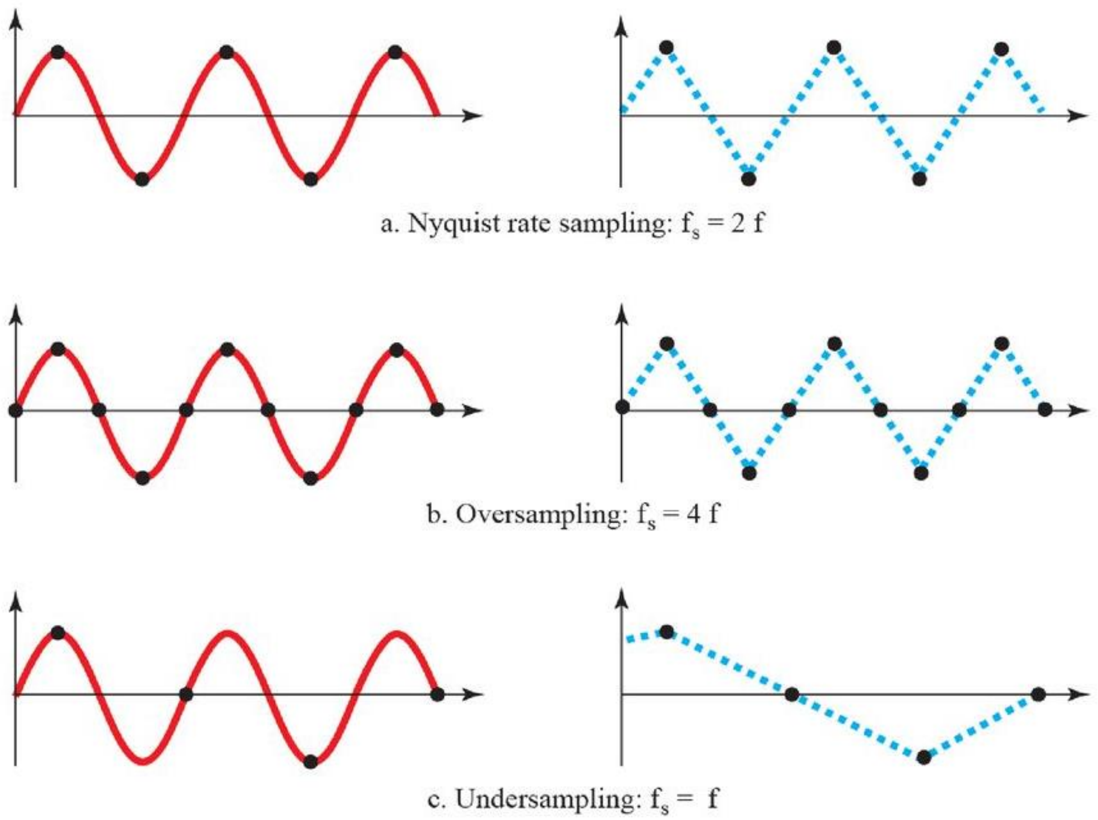
- Similarly, maximum sampling interval is called **Nyquist interval**. It is given by,

$$T_s = \frac{1}{2f_m}$$

- The sampling frequency, $\omega_s = 2BW$, is referred to as the **Nyquist sampling frequency**.



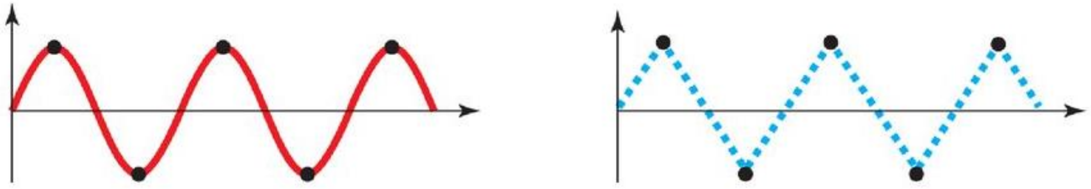
Nyquist rate

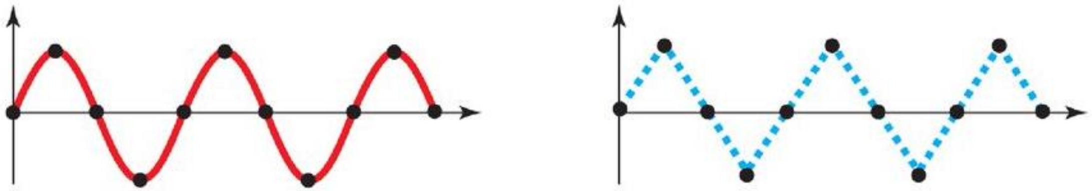


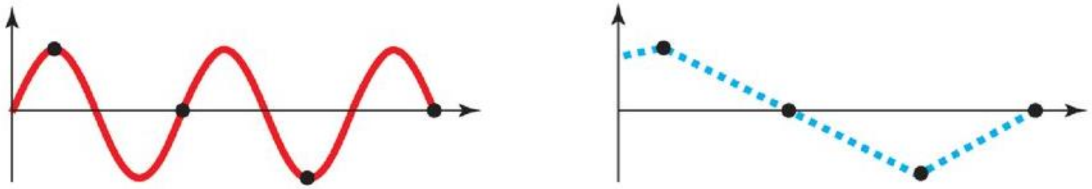
Let us sample a simple sine wave at three sampling rates:

- $f_s = 4f_m$ (2 times the Nyquist rate),
- $f_s = 2f_m$ (Nyquist rate), and
- $f_s = f_m$ (one-half the Nyquist rate).

Nyquist rate

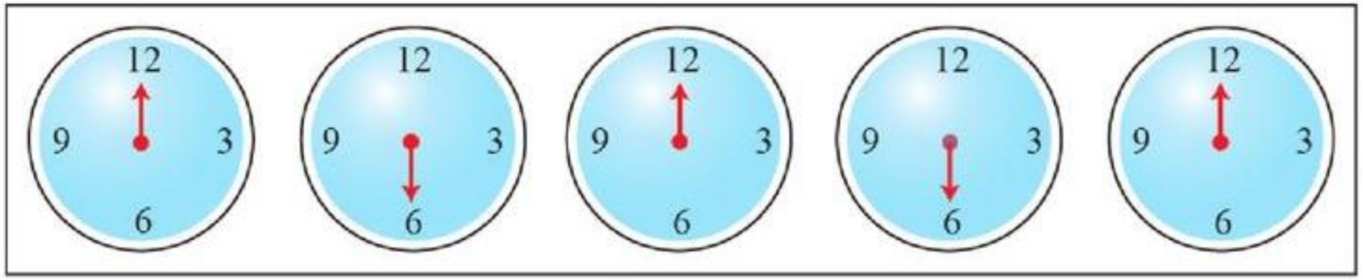

a. Nyquist rate sampling: $f_s = 2\ f$


b. Oversampling: $f_s = 4\ f$


c. Undersampling: $f_s = f$

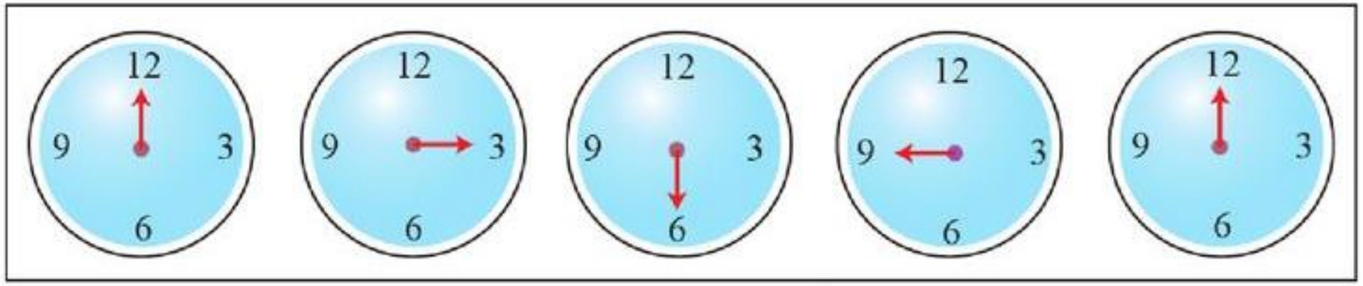
- It can be seen that sampling **at the Nyquist rate** can create a good approximation of the original sine wave.
- **Oversampling** can also create the same approximation, but it is redundant and unnecessary.
- Sampling **below the Nyquist rate** does not produce a signal that looks like the original sine wave.

Nyquist rate



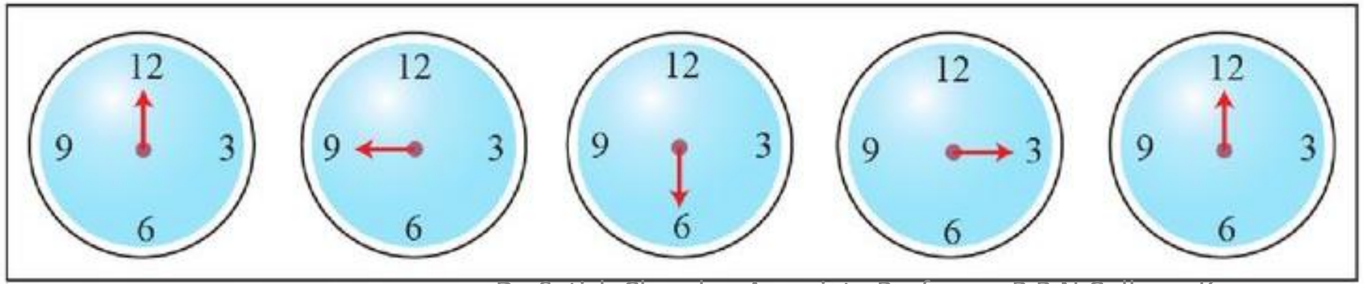
Samples can mean that the clock is moving either forward or backward. (12-6-12-6-12)

a. Sampling at Nyquist rate: $T_s = T\frac{1}{2}$



Samples show clock is moving forward. (12-3-6-9-12)

b. Oversampling (above Nyquist rate): $T_s = T\frac{1}{4}$



Samples show clock is moving backward. (12-9-6-3-12)

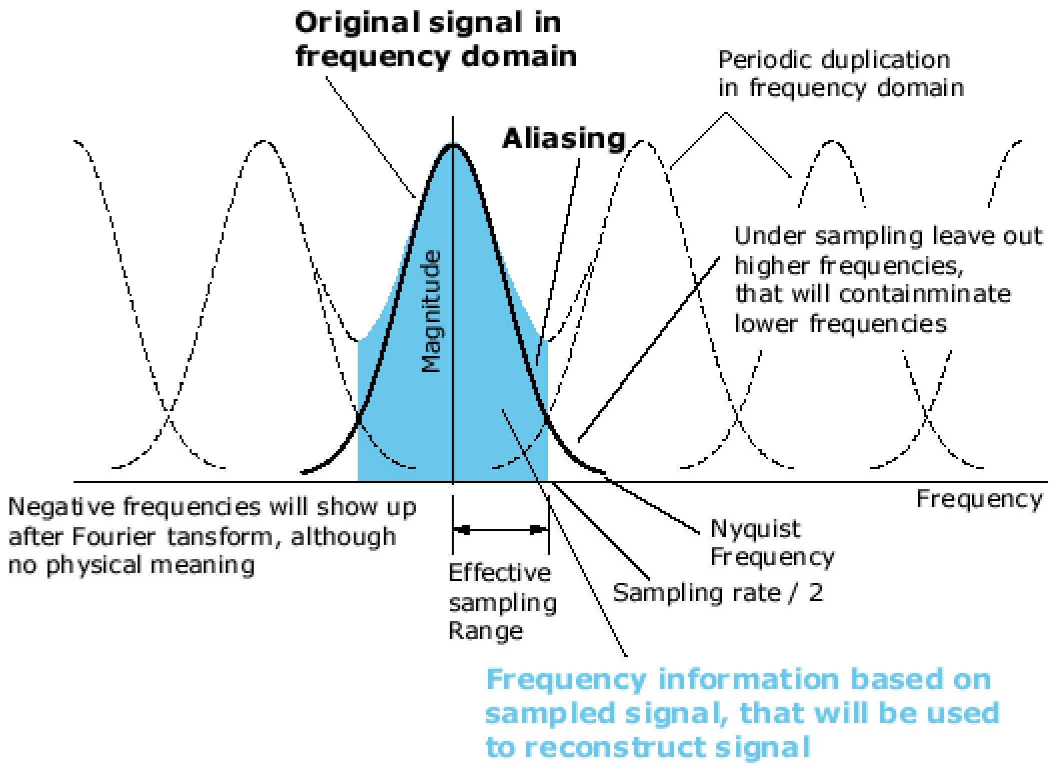
c. Undersampling (below Nyquist rate): $T_s = T\frac{3}{4}$

Nyquist rate

- Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz.
- The sampling rate therefore is 8000 samples per second.

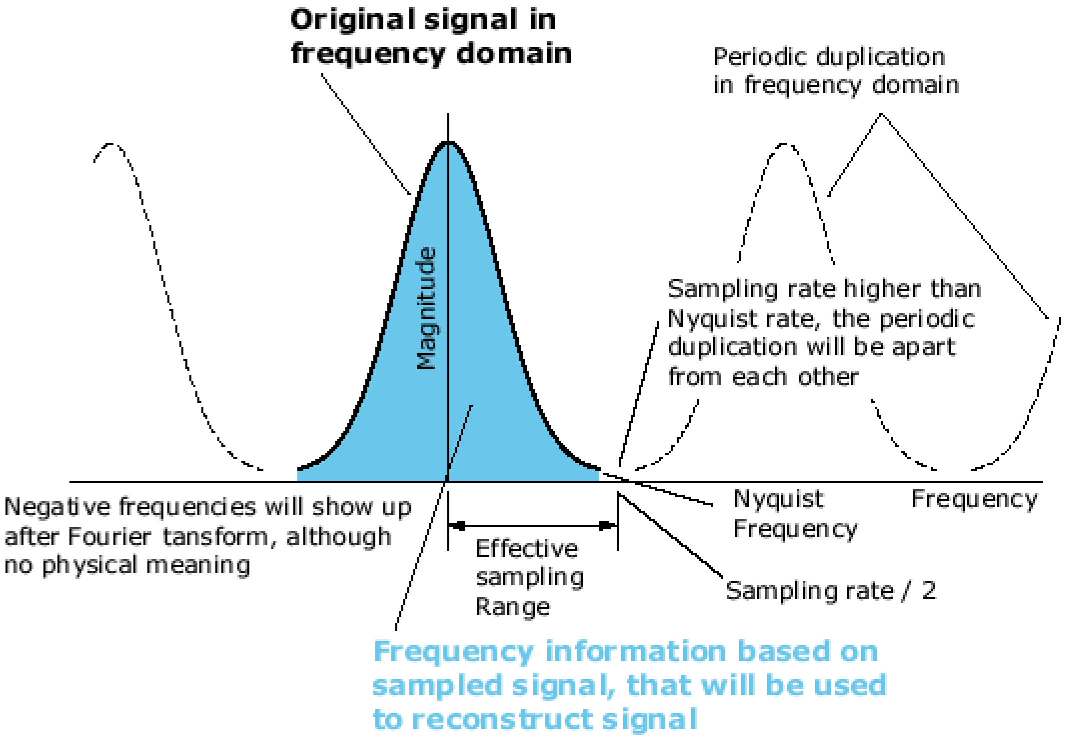
Aliasing

- Under sampling causes frequency components that are higher than half of the sampling frequency to overlap with the lower frequency components.
- As a result, the higher frequency components roll into the reconstructed signal and cause distortion of the signal.
- This type of signal distortion is called **aliasing**.



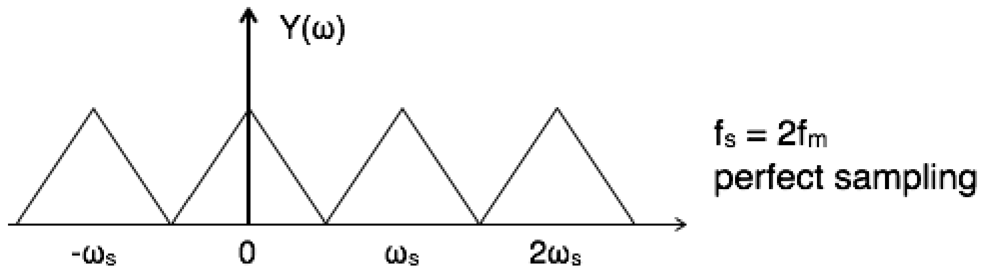
Aliasing

- If the sampling rate is sufficiently high, i.e., greater than the Nyquist rate, there will be no overlapped frequency components in the frequency domain.
- A "cleaner" signal can be obtained to reconstruct the original signal.



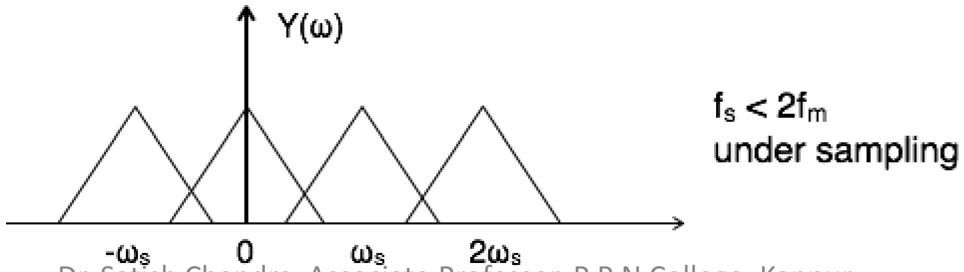
Aliasing

- When the continuous band limited signal is **sampled at Nyquist rate** ($f_s = 2f_m$), the sampled spectrum contains non-overlapping spectrum, repeating periodically. But the successive cycles of spectrum touch each other.
- Therefore, the original spectrum can be recovered from the sampled spectrum by using **low-pass filter** with a cut-off frequency f_m.



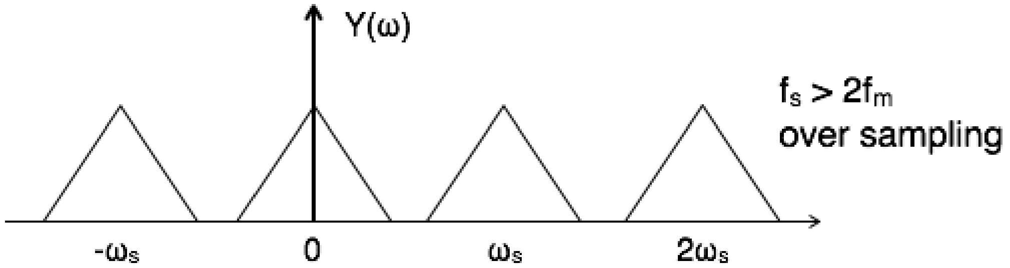
Aliasing

- When the continuous band limited signal is sampled at a rate lower than Nyquist rate ($f_s < 2f_m$), then the successive cycles of the spectrum of the sampled signal overlap with each other.
- Hence, the signal is **under-sampled** and aliasing is produced.
- Thus, aliasing is the phenomenon in which *a high-frequency component in the frequency spectrum of the signal takes identity of a lower frequency component* in the spectrum of the sampled signal.

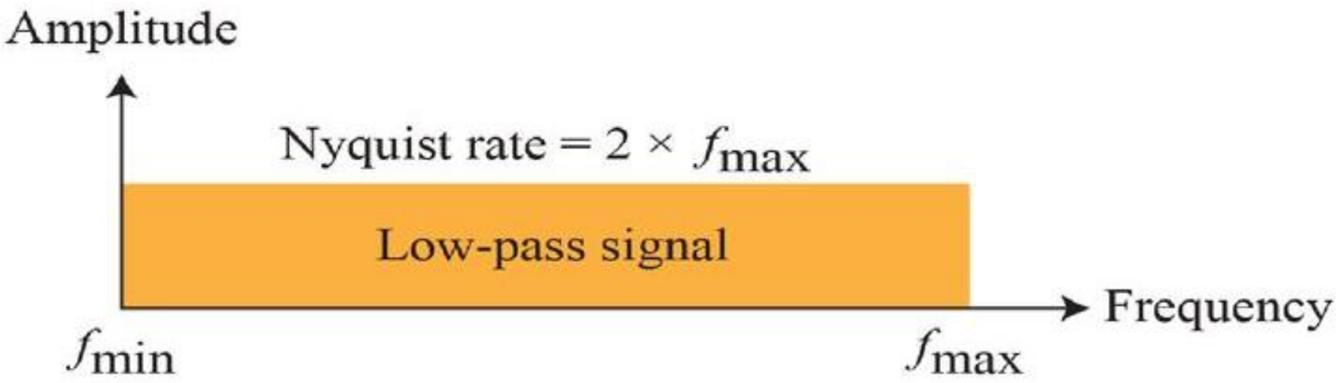


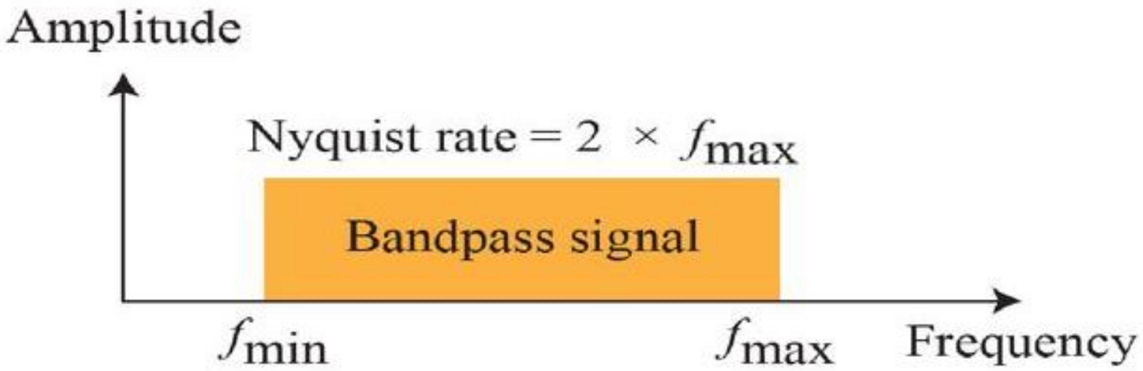
Aliasing

- Due to aliasing, it is not possible to recover original signal from sampled signal by low pass filter, since the spectral components in the overlap regions add and hence the signal is distorted.
- To avoid aliasing,
  - A low pass anti-aliasing filter is used, prior to sampling, to attenuate high frequency components of the signal.
  - Sample at a rate slightly higher than the Nyquist rate, i.e., $f_s < 2f_m$.



Aliasing


Amplitude
Nyquist rate = 2 × f_{max}
Low-pass signal
Frequency
f_{min}
f_{max}


Amplitude
Nyquist rate = 2 × f_{max}
Bandpass signal
f_{min}
f_{max}
Frequency

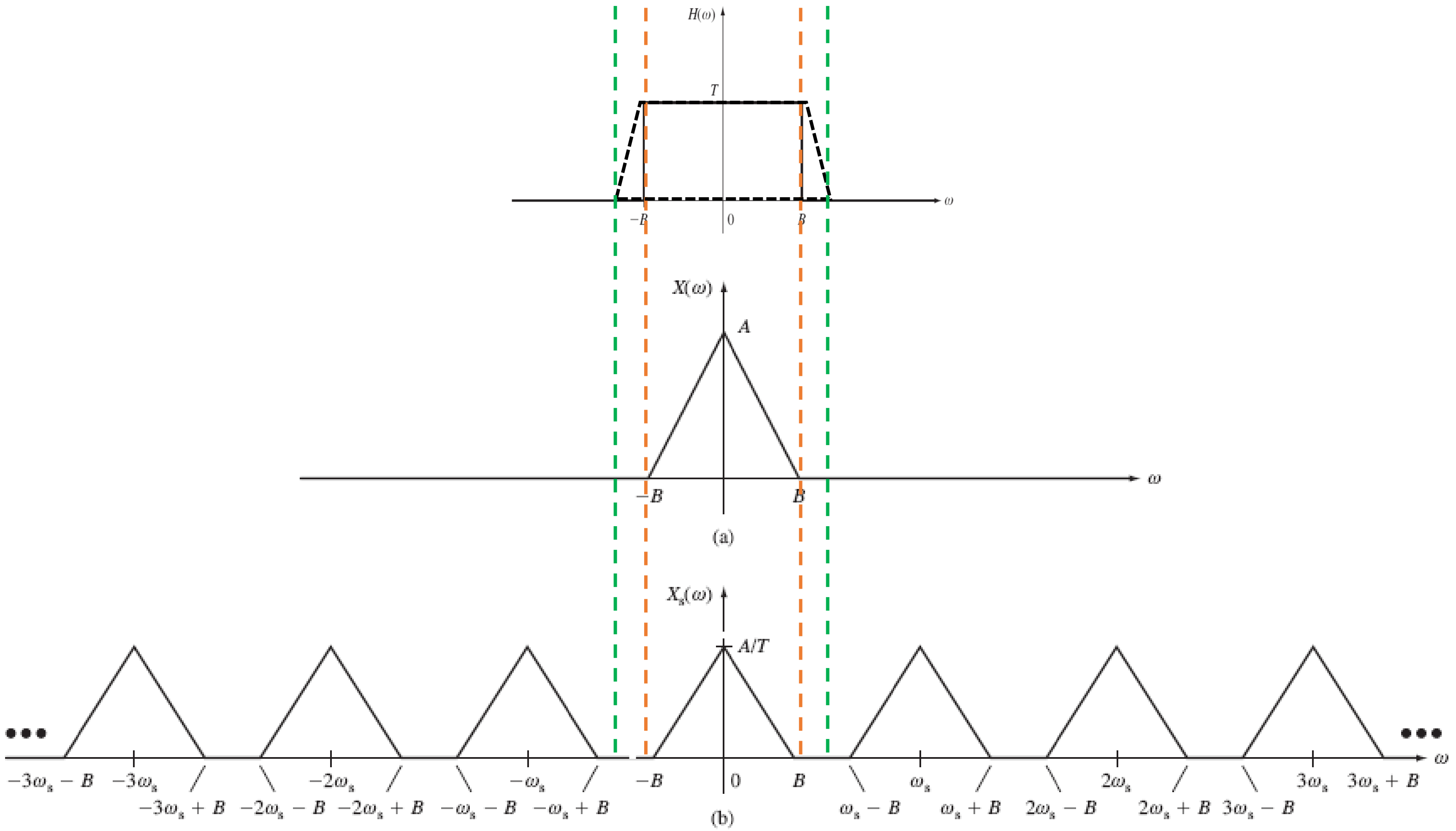
Low-Pass Signal

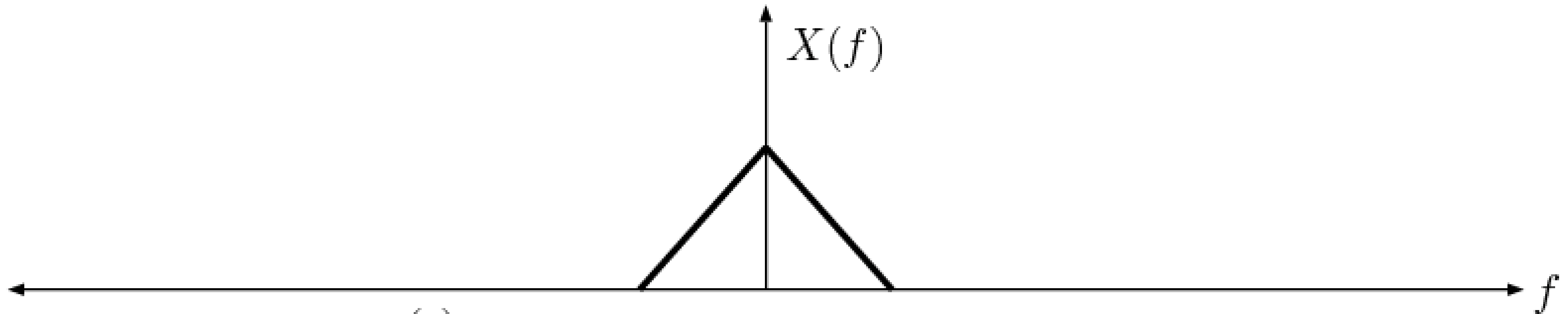
- The low-pass filter is used to recover original signal from its samples.
- It passes only low frequencies up to a specified cut-off frequency and rejects all other frequencies above cut-off frequency.
- A sharp change in response at cut-off frequency in **ideal low-pass filter** is not practically possible.
- In case of **practical low-pass filter**, the amplitude response decreases slowly to become zero.
- The practical low-pass filter can be used in reconstruction of original signal, if sampled at a rate higher than Nyquist rate.

Low-Pass Filter

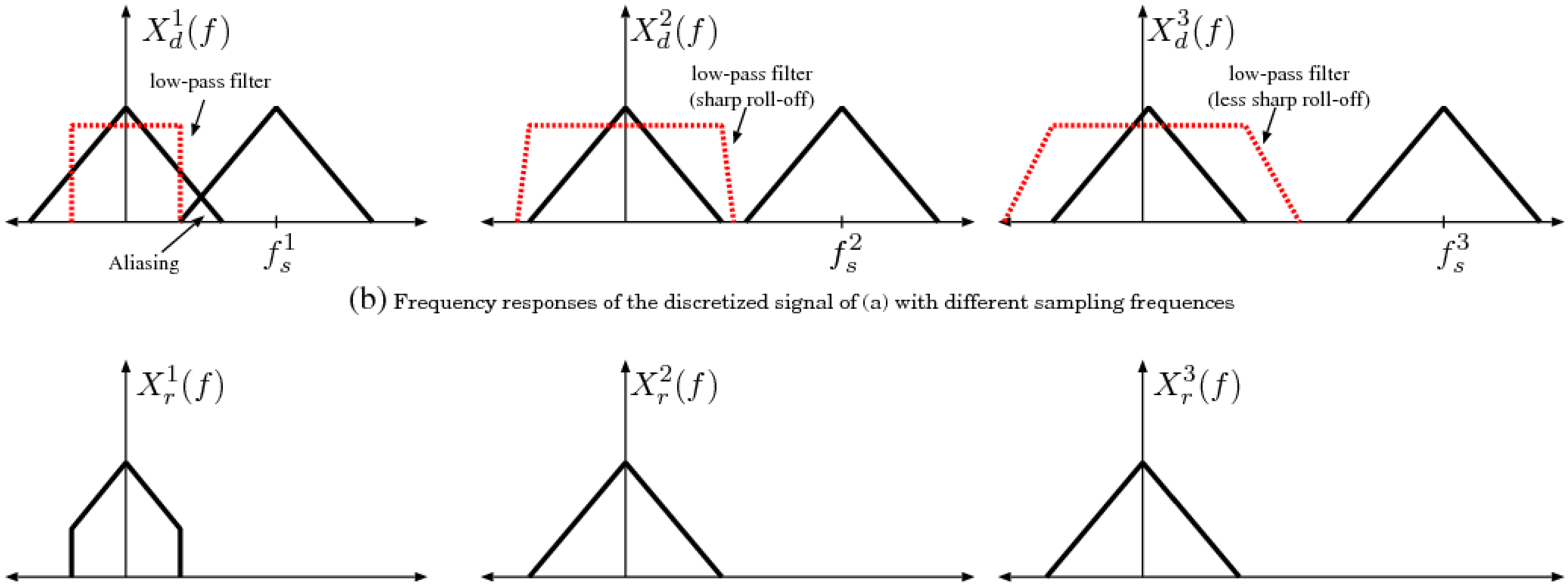
Low pass filter pass only low frequencies upto a cut off frequency

Low-Pass Signal





(a) Frequency response of the original analog signal : baseband signal

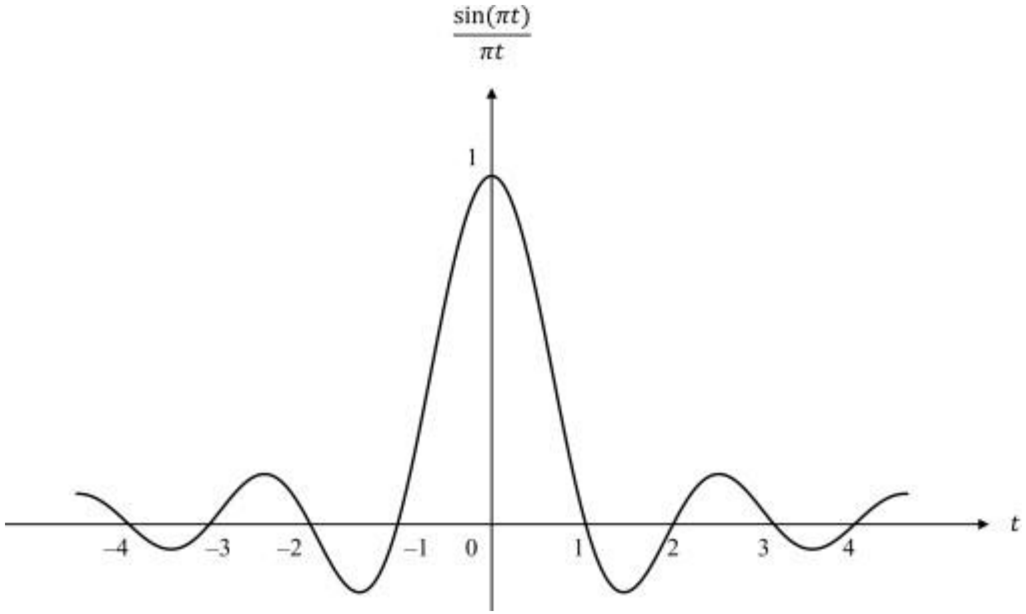


(b) Frequency responses of the discretized signal of (a) with different sampling frequences

(c) Frequency responses of recovered signal by low-pass filtering

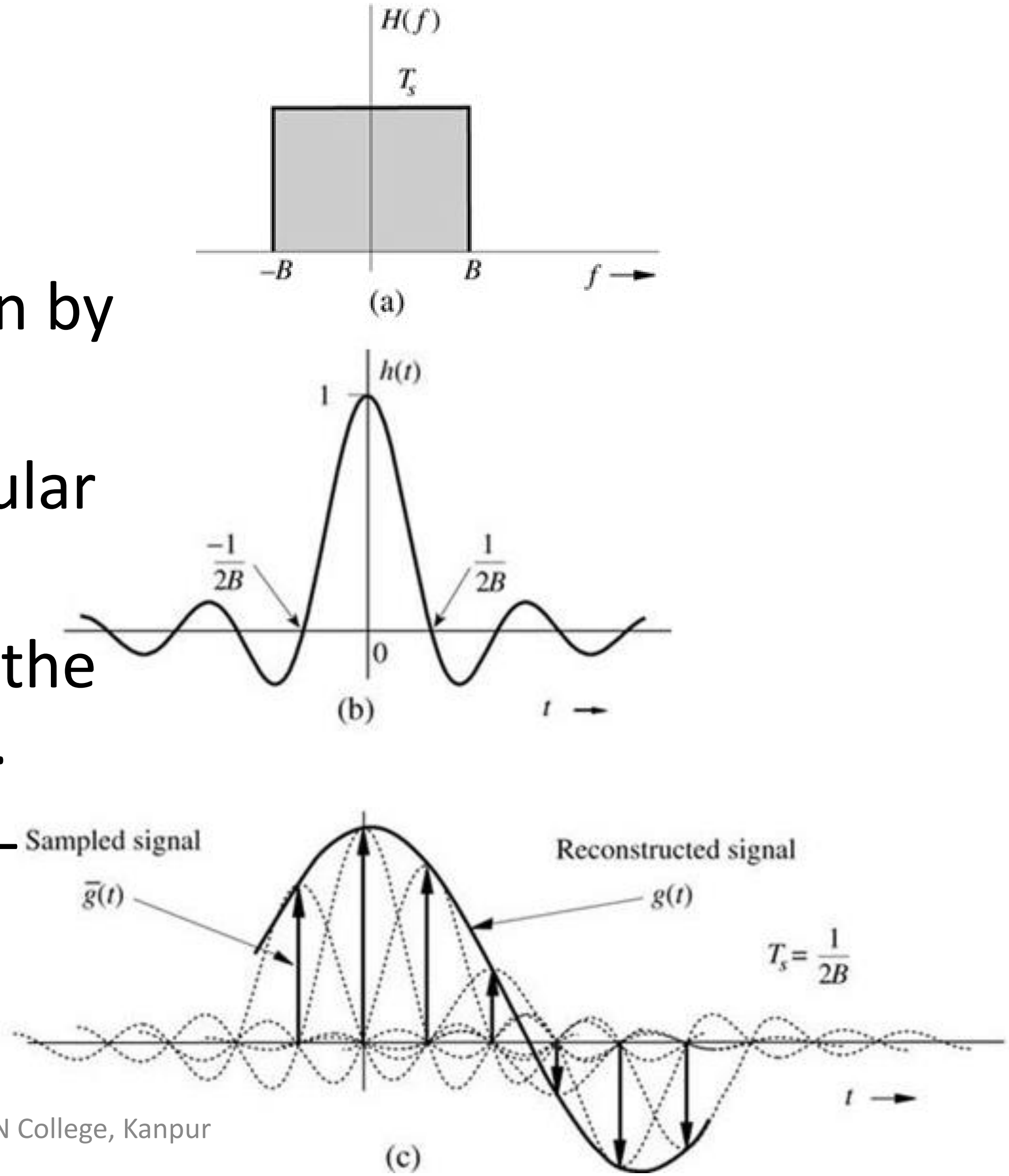
Sinc function

$$\mathrm{sinc}(x) = \frac{\sin x}{x}$$

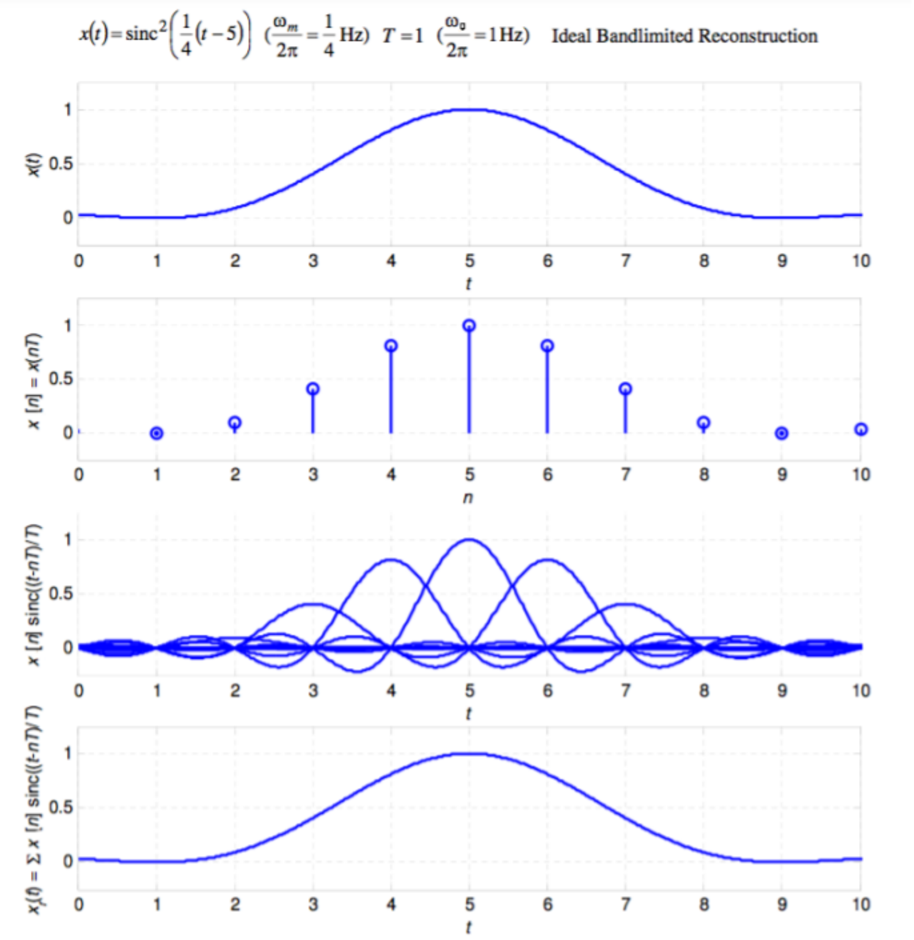


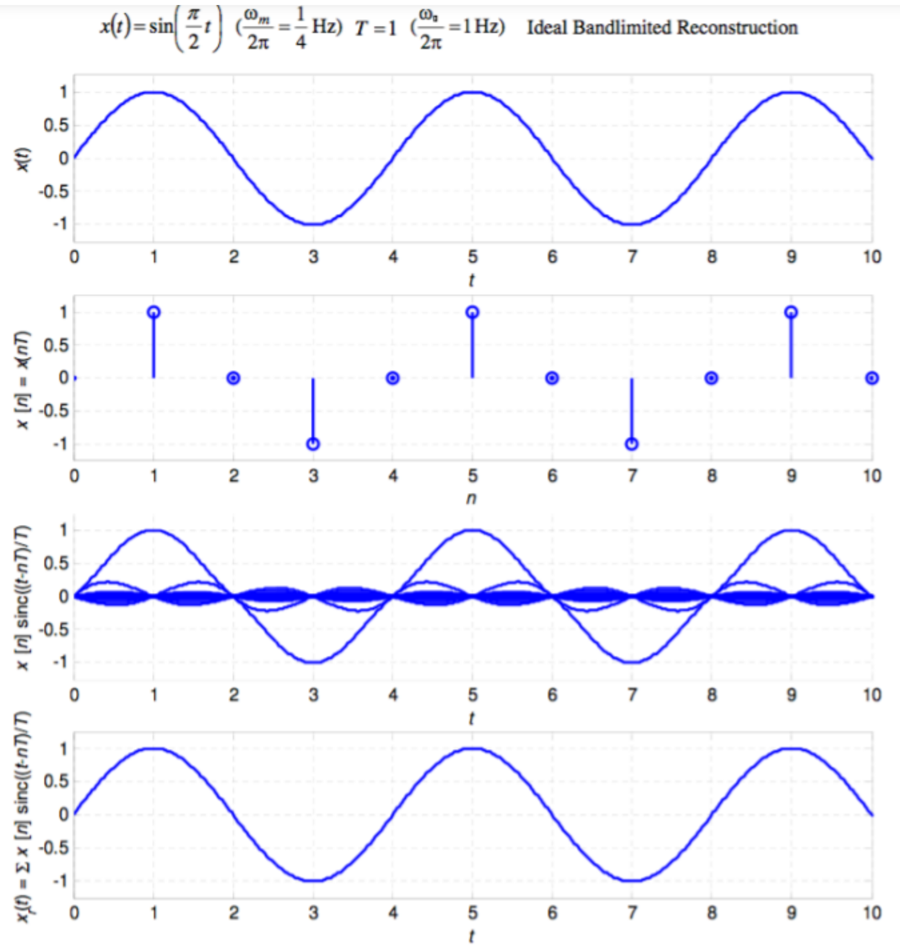
Signal Reconstruction

- We can get back the original continuous time signal x(t) from the sampled version by applying an ideal low-pass filter.
- The time domain version of the rectangular filter is a *sinc* (interpolation) function.
- We can use sinc function to reconstruct the samples and get back the original signal.
- However, sinc function goes on forever – not very practical to use in real system



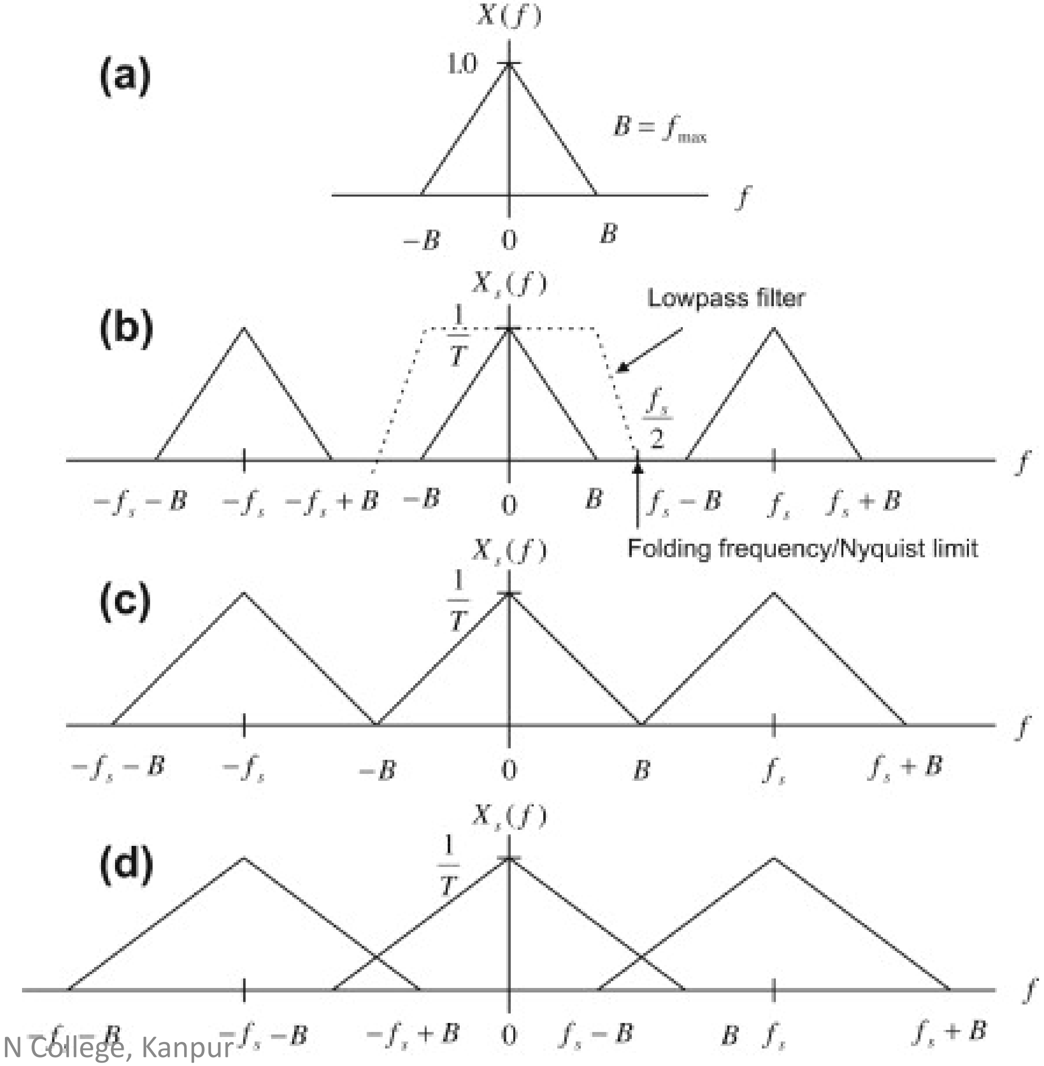
Signal Reconstruction





Signal Reconstruction

- So, in practice, we don’t use the *sinc* function for reconstruction.
- Instead, we sample the signal at highest than Nyquist rate, introducing the gap, and therefore allow us to use much practical filter with less abrupt changes in the spectral domain.



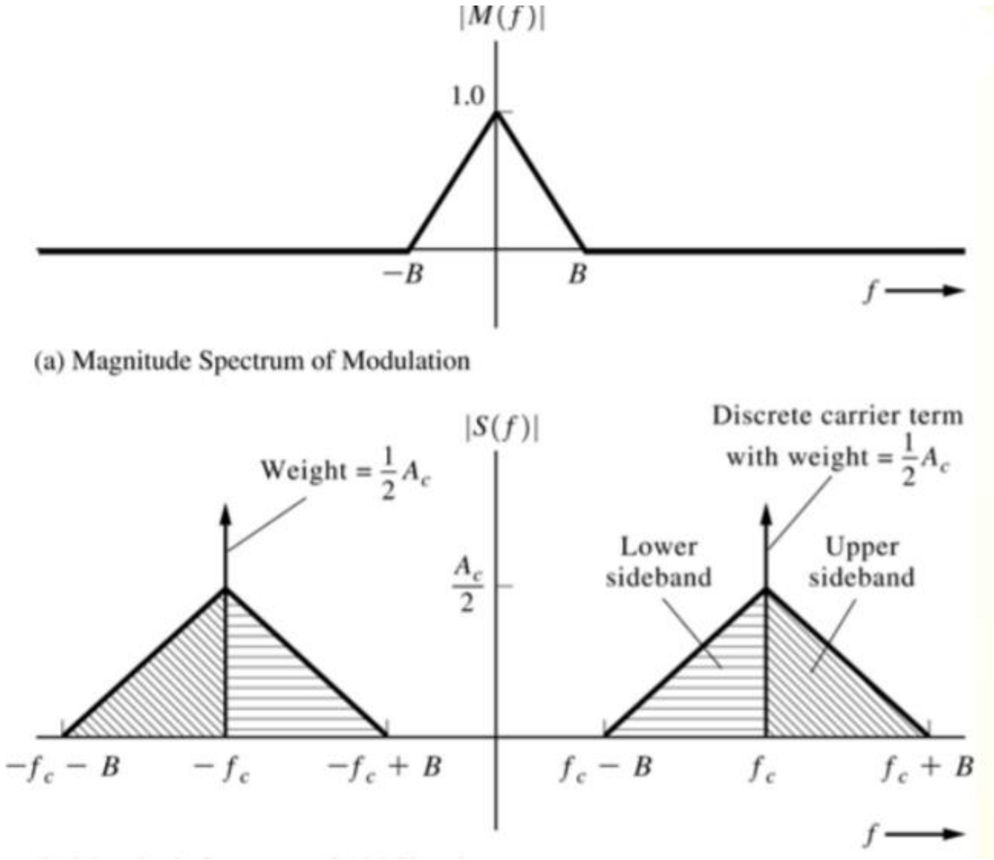
Baseband Vs Bandpass

- A **baseband** waveform has a spectral magnitude that in nonzero for frequencies in the vicinity of the origin (i.e. f=0) and negligible elsewhere.

- A **bandpass** waveform has a spectral magnitude that is nonzero for frequencies in some band concentrated about a frequency $f = \pm f_c$ where f_c is much greater than zero.

Band-Pass Signal

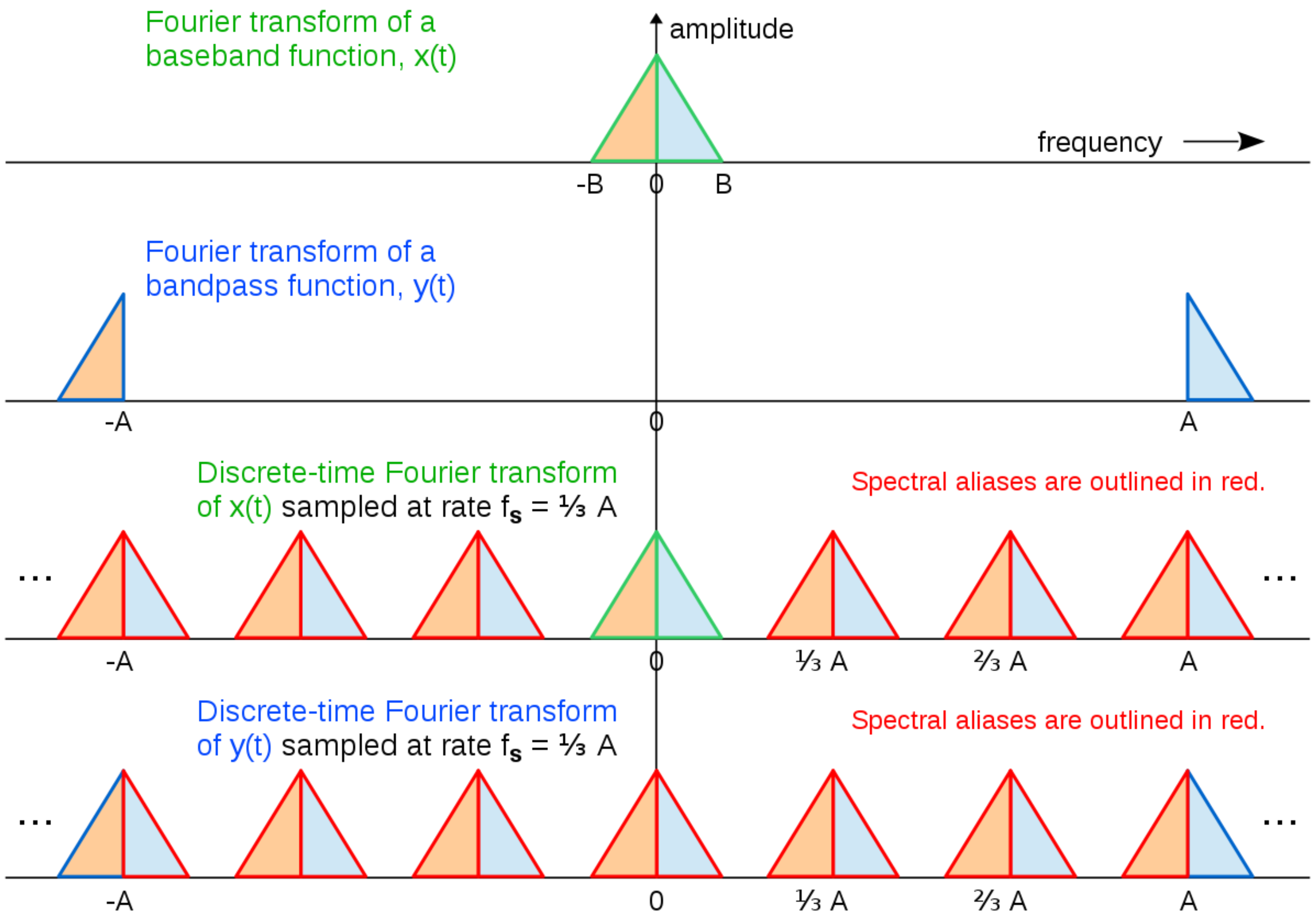
- If the signal is a bandpass signal whose maximum bandwidth is BW=2f_m, then it can be completely recovered from its sample, if it is sampled at the minimum rate of twice the bandwidth.
- Hence, if the bandwidth is 2f_m, then the minimum sampling rate for the bandpass signal must be 4f_m sample per second.



$$T_S = \frac{1}{2B} = \frac{1}{4f_m}$$

Band-Pass Signal

- The bandpass signal of bandwidth B = $2f_m$ can be completely recovered from its samples.
- Figure shows the spectrum of a bandpass signal and the sampled bandpass signal

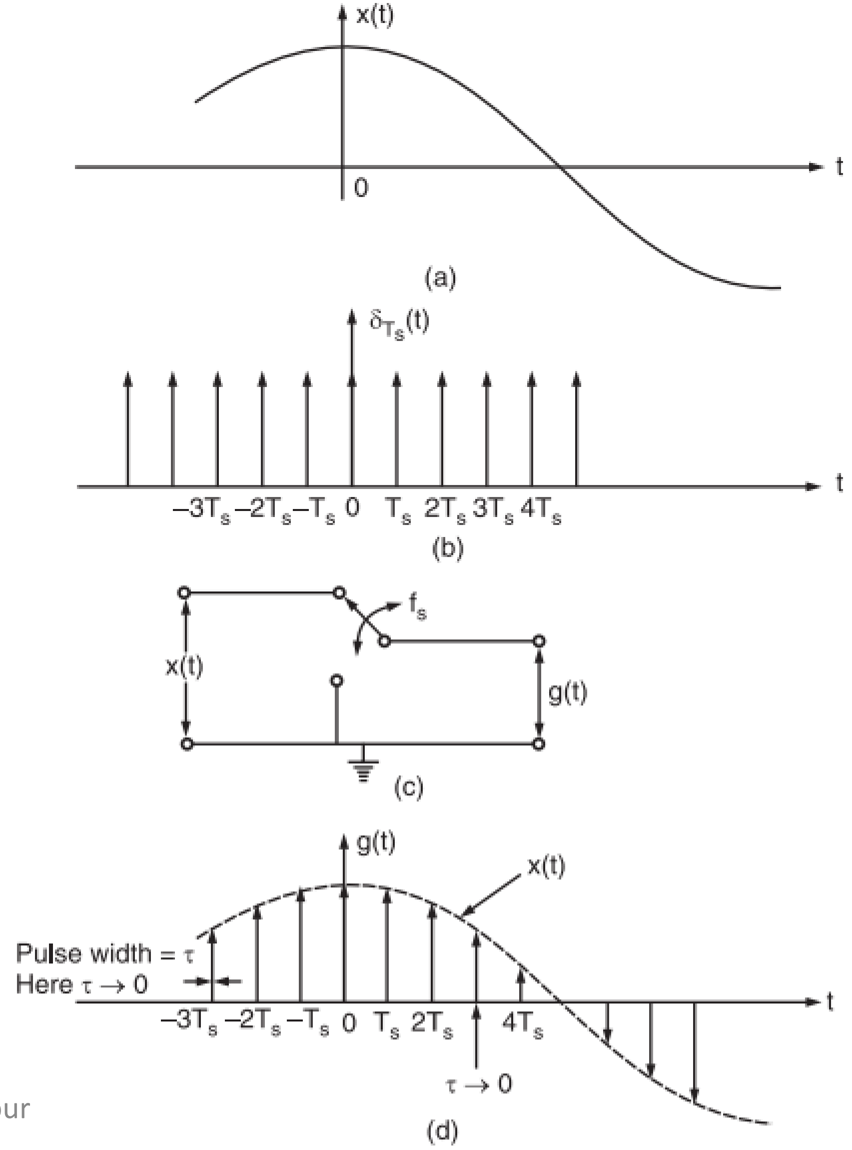


Signals Sampling Techniques

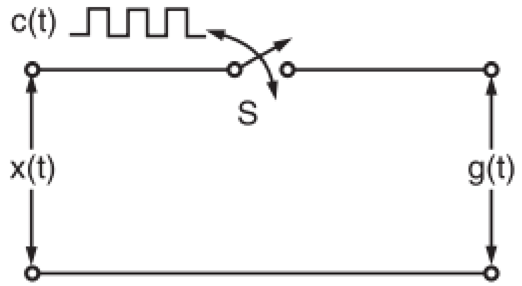
- There are three types of sampling techniques:
  - Instantaneous sampling.
  - Natural sampling.
  - Flat Top sampling.
- Out of these three, instantaneous sampling is called ideal sampling whereas natural sampling and flat-top sampling are called practical sampling methods.

Ideal Sampling

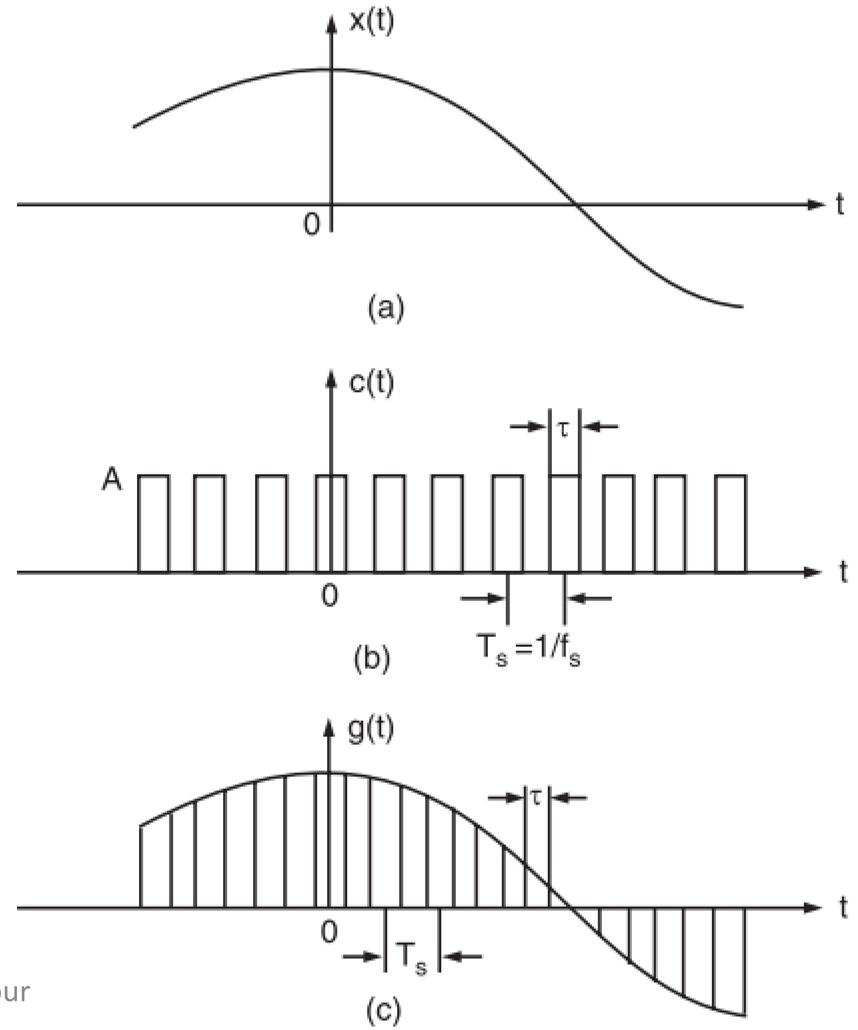
- **Ideal** Sampling is also known as **Instantaneous** sampling or **Impulse** Sampling.
- Train of impulse is used as a carrier signal for ideal sampling.
- In this sampling technique the sampling function is a **train of impulses** and the principle used is known as **multiplication principle**.



Natural Sampling

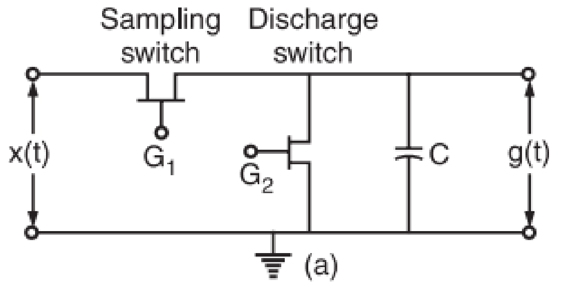


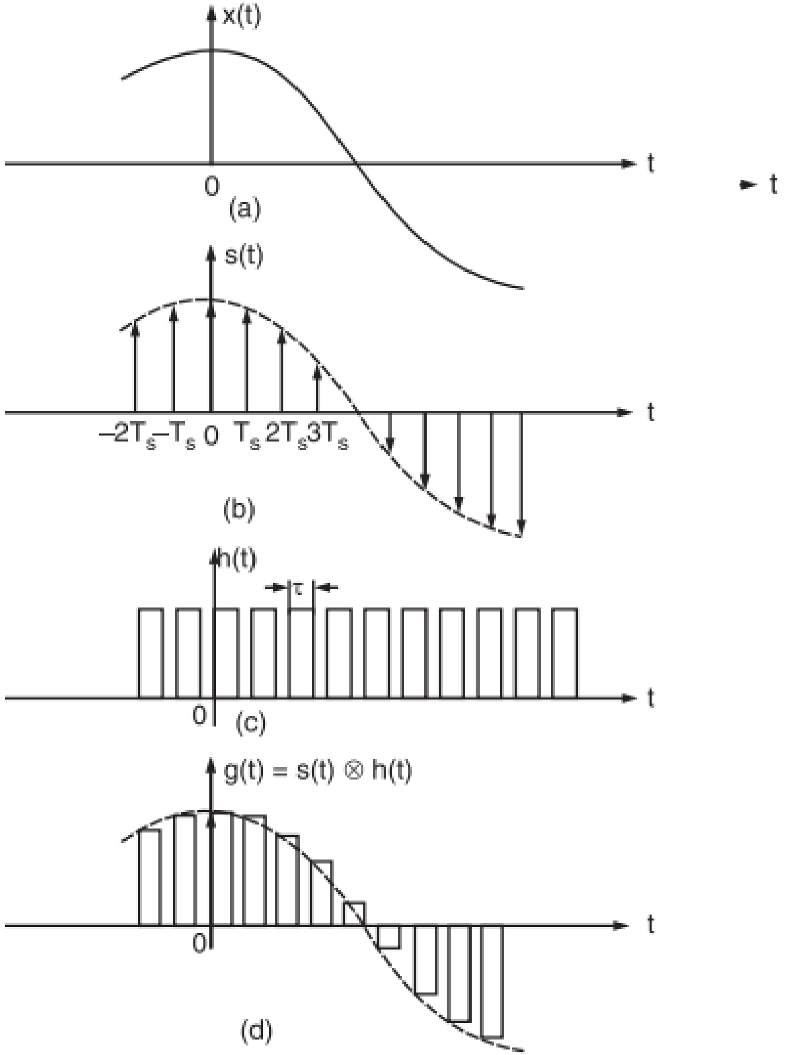
- Natural Sampling is a practical method of sampling in which pulse have **finite width** equal to τ.
- Sampling is done in accordance with the carrier signal which is digital in nature.
- With the help of functional diagram of a Natural sampler, a sampled signal g(t) is obtained by multiplication of sampling function c(t) and the input signal x(t).



Flat Top Sampling

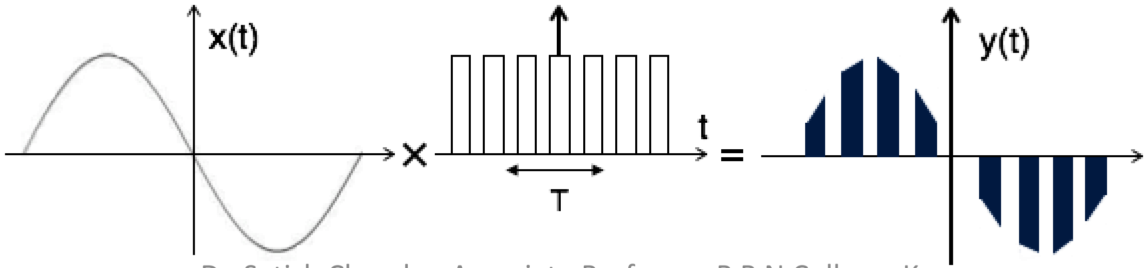
- Flat top sampling is like natural sampling *i.e.*, practical in nature.
- In comparison to natural sampling flat top sampling can be easily obtained.
- In this sampling techniques, the top of the samples remains constant and is equal to the instantaneous value of the message signal x(t) **at the start** of sampling process.
- **Sample and hold circuit** are used in this type of sampling.





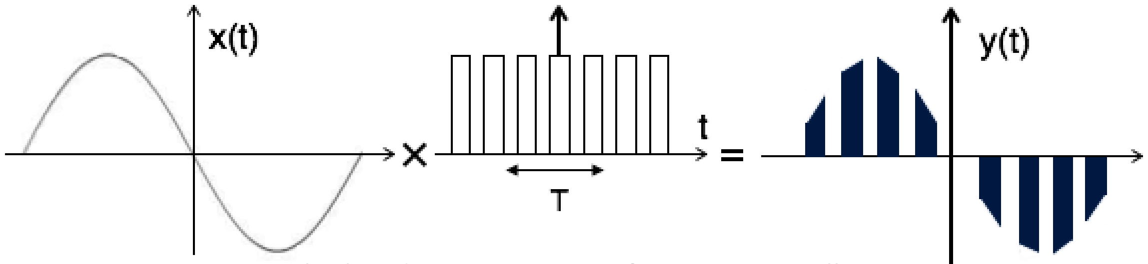
Natural Sampling

- Natural Sampling is much more reasonable manner of sampling than instantaneous sampling.
- The reason is that, instantaneous samples, at the transmitting end of the channel, have **infinitesimal energy**.
- It give rise to signals having peak value which is **infinitesimally small**, when transmitted through a band limited channel.

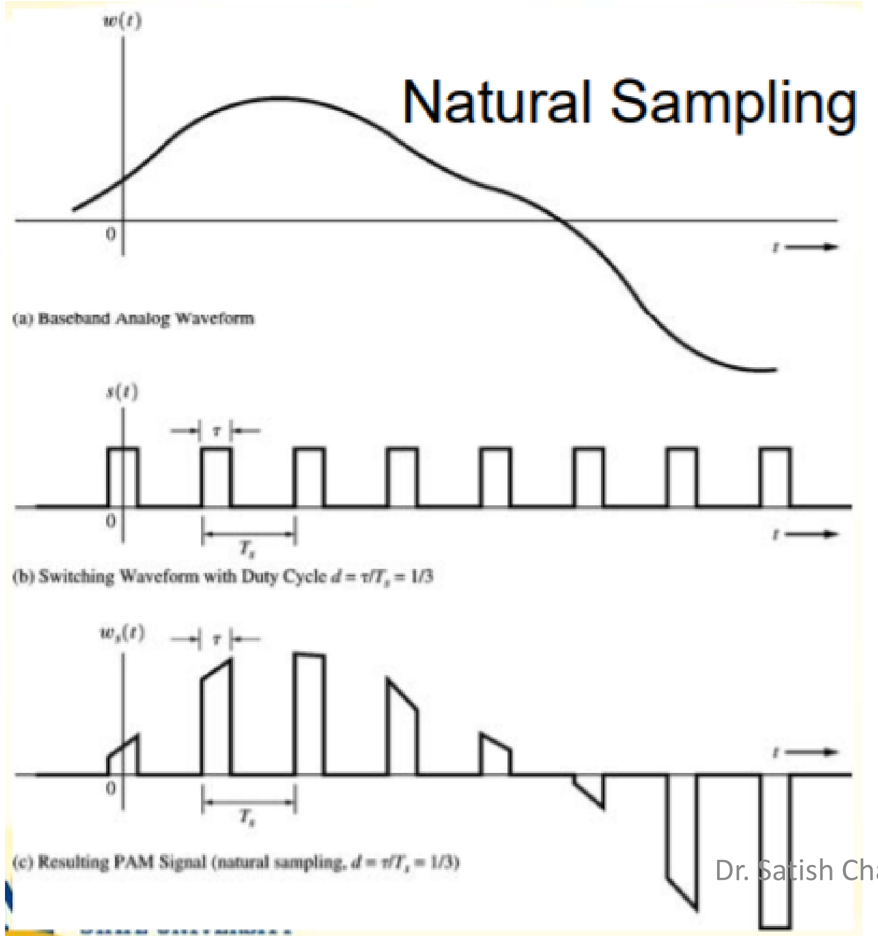


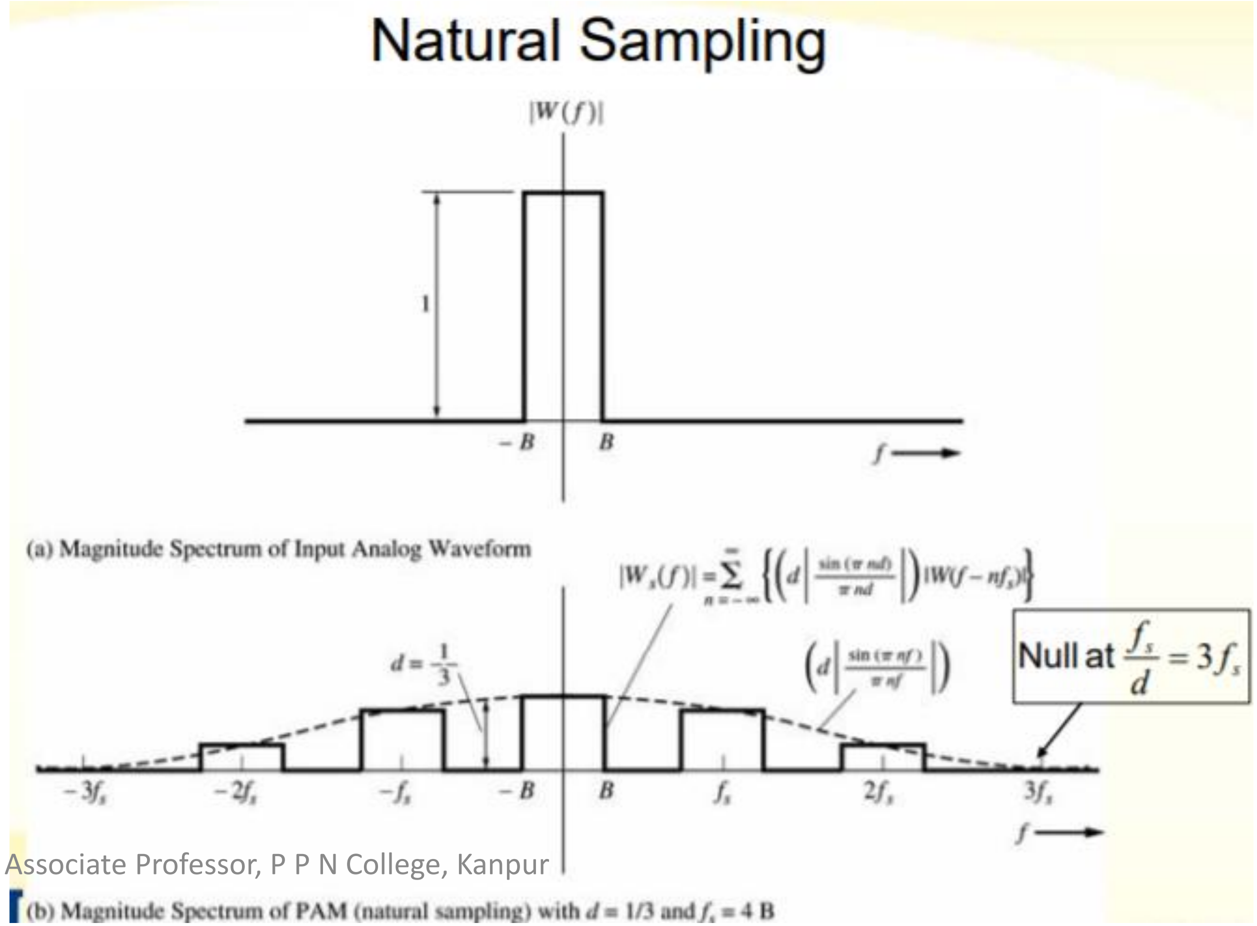
Natural Sampling

- In natural sampling the sampling waveform y(t) consists of a train of pulses having duration τ and separated by the sampling time T_s.
- The baseband signal x(t) and the sampled signal y(t) is shown in the figure.
- The sampled signal consists of a sequence of pulses of varying amplitude whose tops are not flat but follows the waveform of the signal x(t)



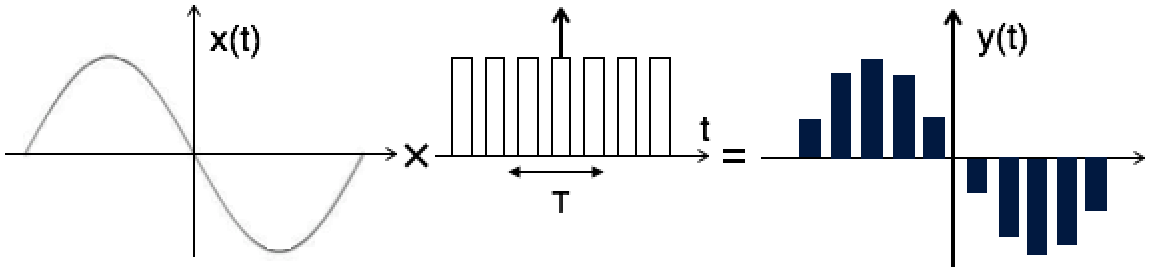
Natural Sampling



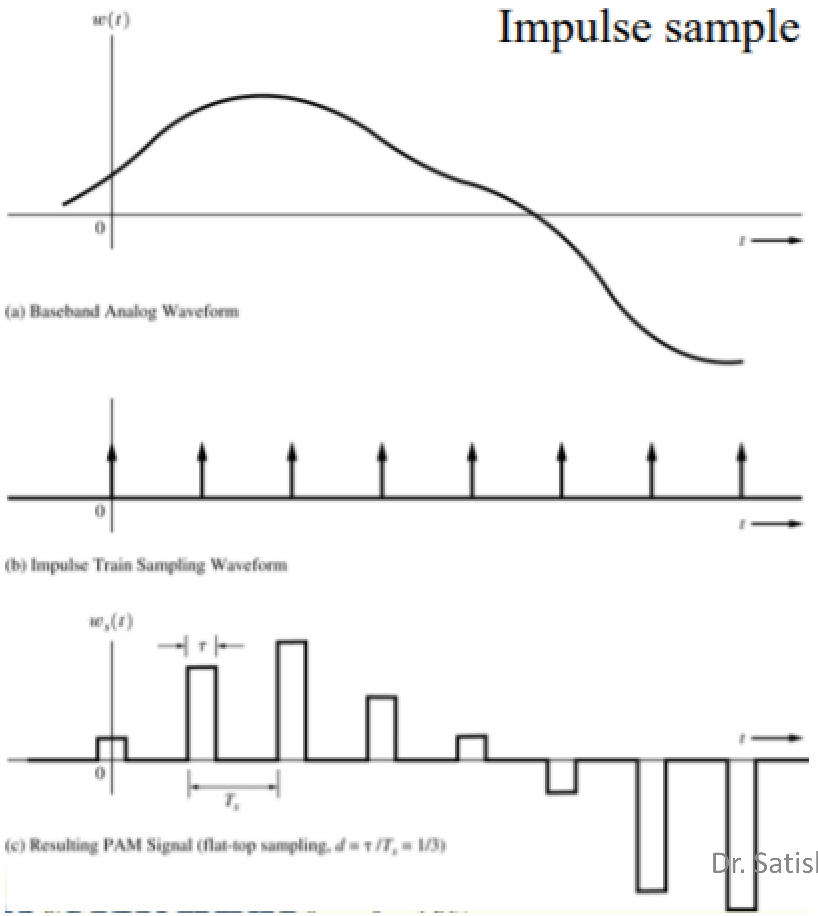


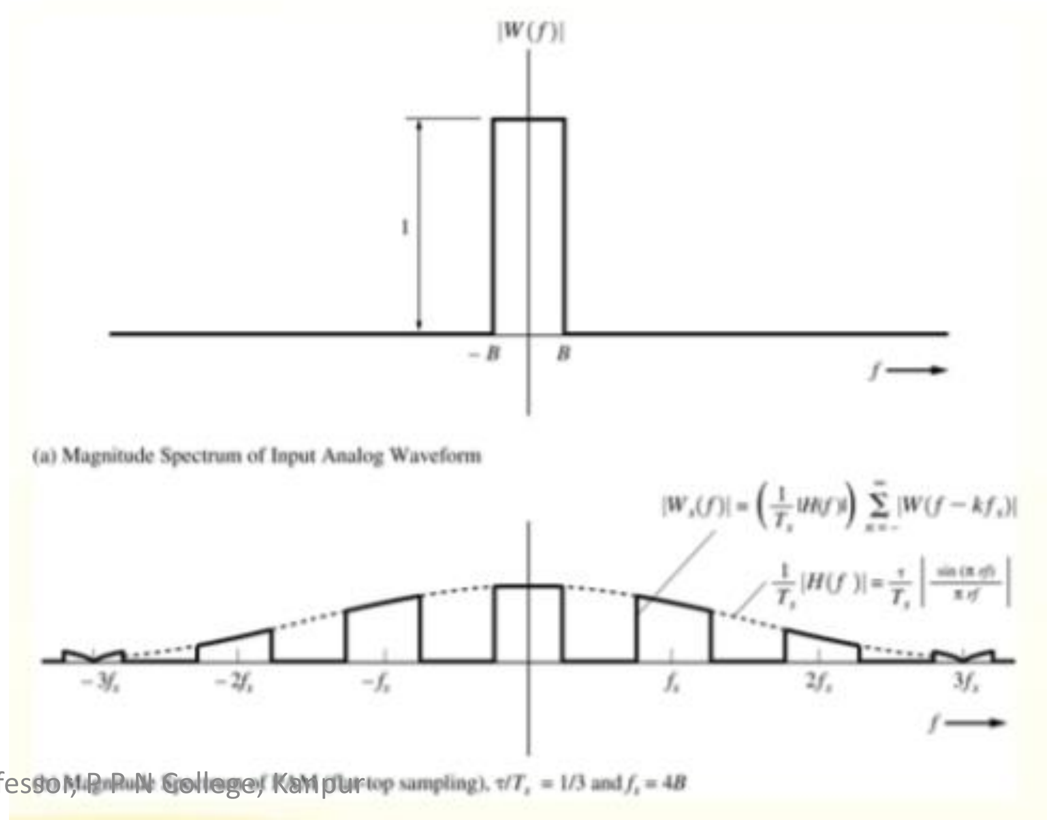
Flat Top Sampling

- Pulses shown in last figure with tops follow the waveform of the signal are actually not used. Instead flat-topped pulse are used.
- A flat topped pulse has a constant amplitude fixed by the sample value of the signal at same point within the pulse interval.
- In figure, the signal has been sampled at the beginning of the pulse.
- For flat-top sampling, design of the circuitry used to sample is simple.



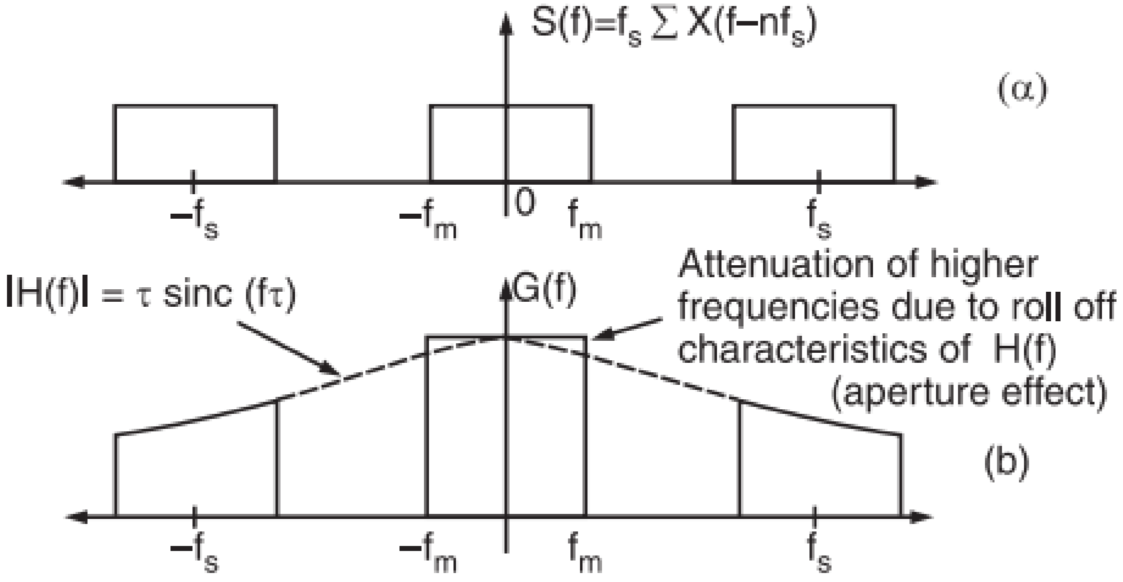
Flat Top Sampling





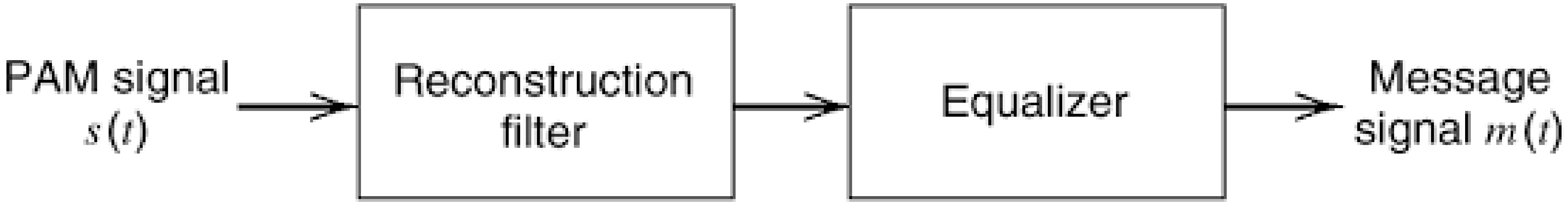
Aperture Effect in Flat top Sampling

- In flat-top sampling, due to the lengthening of the sample, amplitude distortion and delay are introduced.
- This distortion is referred to as the **aperture effect** and occurs during the reconstruction of g(t) form s(t).
- Aperture effect can be improved by selecting value of pulse width τ to be very small and by using equalizer circuit.



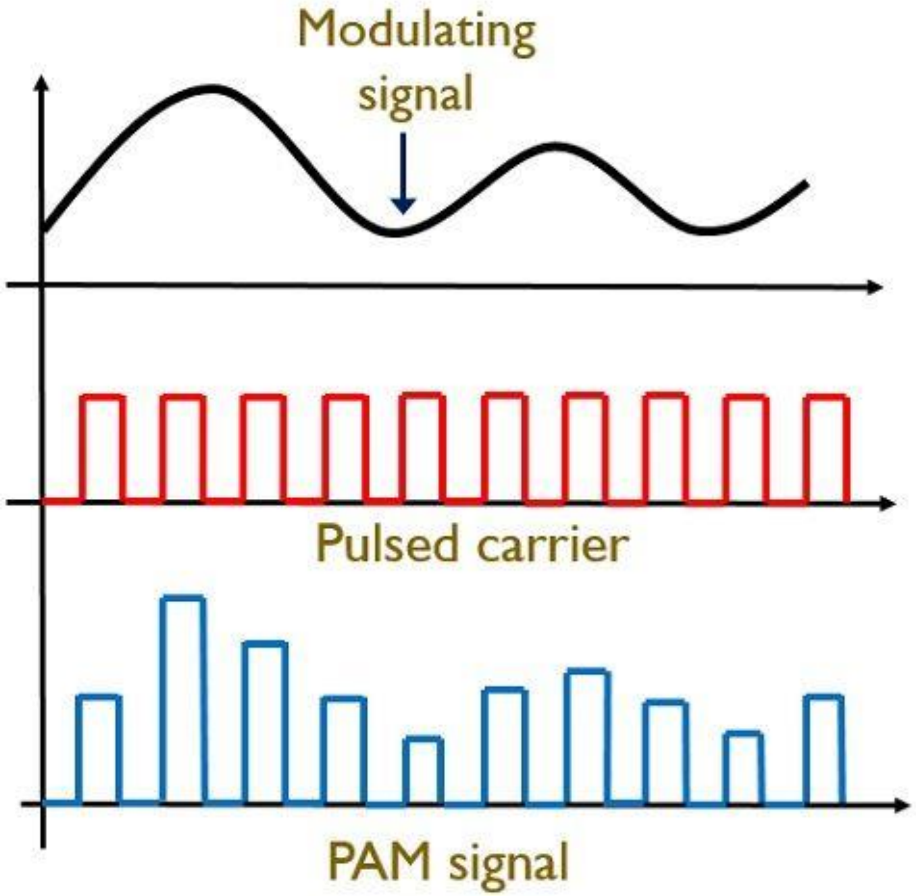
Aperture Effect in Flat top Sampling

- During flat top sampling, to convert varying amplitudes of pulse to flat top pulses we use a **sinc function**. Because of this, there would be decrease in the amplitude.
- This distortion is named as **Aperture effect**.
- This may be eliminated by using an **equalizer** in cascade with the output **low pass filter**.



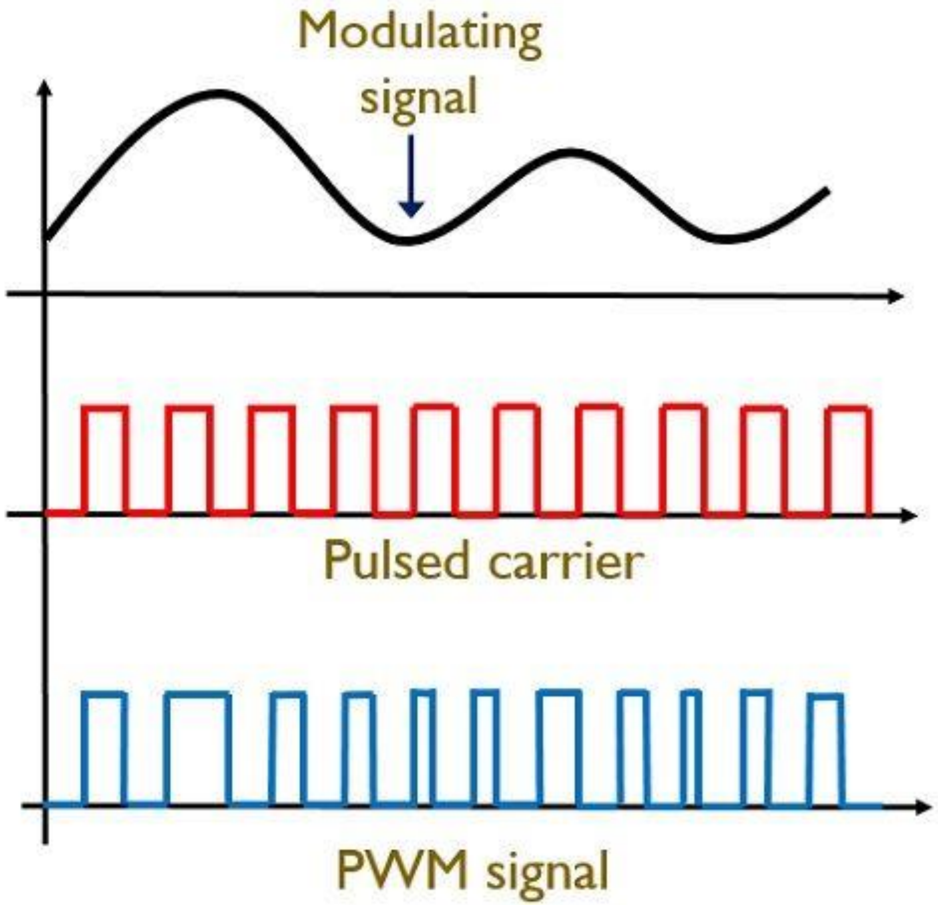
PAM

- PAM stands for **pulse amplitude modulation**.
- Technique in which the amplitude of the pulsed carrier signal is changed according to the amplitude of the message signal.
- The amplitude of the pulses is varying with respect to the amplitude of analog modulating signal, like in case of amplitude modulation.
- But the major difference is that unlike AM, here the carrier wave is a pulse train rather than continuous wave signal.



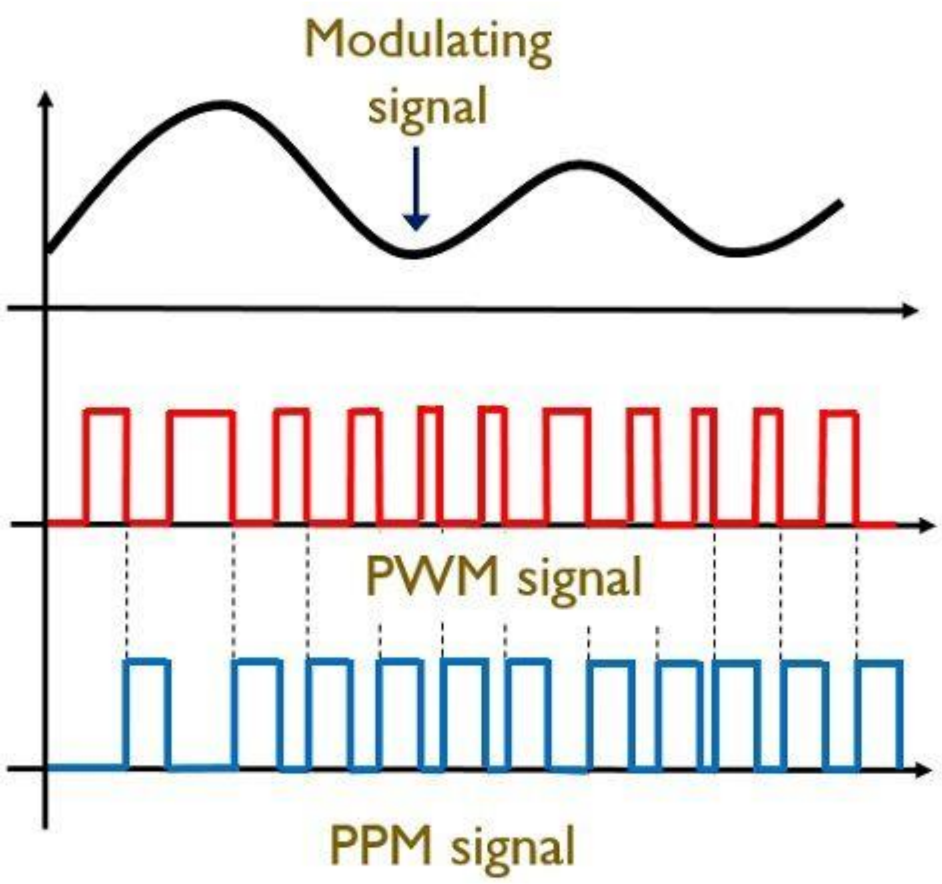
PWM

- PWM is an acronym used for **pulse width modulation**.
- The width of the pulses is varied according to the amplitude of the message signal.
- Unlike PAM, the amplitude of the signal is constant and only the width is varying.
- Technique is similar to frequency modulation because, by the variation in the width of the pulses, the frequency of the pulses in the PWM signal shows variation.



PPM

- PPM is used for **pulse position modulation**.
- The position of the pulses is changed in accordance with the amplitude of the modulating signal.
- The pulse amplitude and the pulse width are the two constant that does not show variation with the amplitude of the modulating signal but only the position shows variation.
- The position of the pulse changes according to the reference pulses, and these reference pulses are nothing but PWM pulses.
- Basically, the falling edge of PWM pulses acts as the starting of the PPM pulses.



Comparison

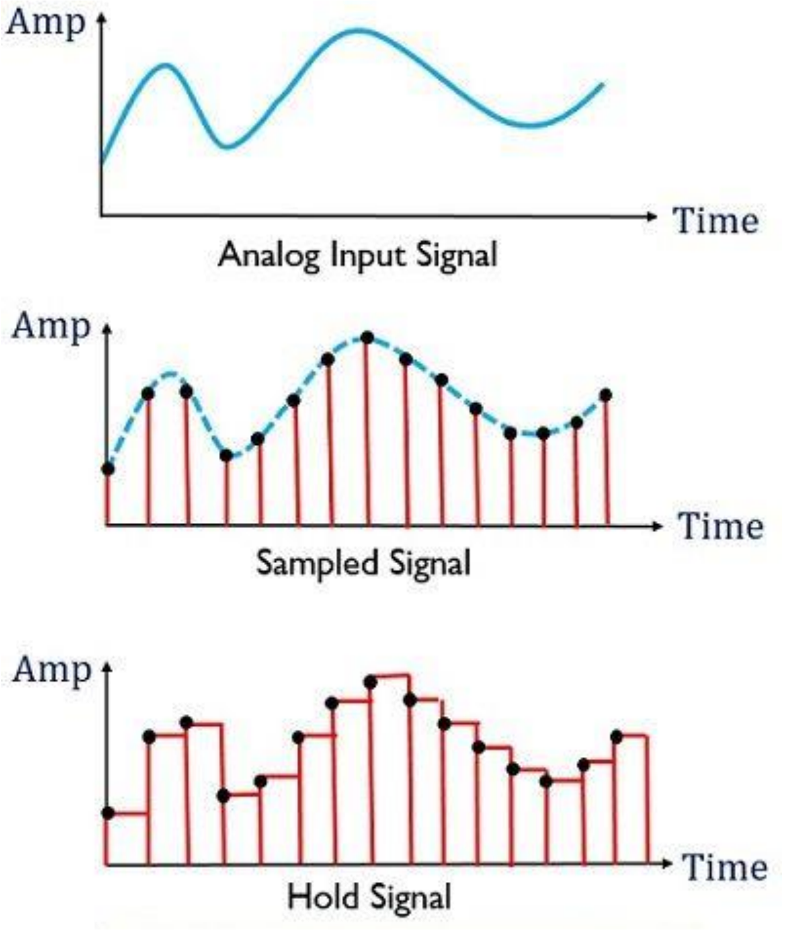
Basis for Comparison	PAM	PWM	PPM
Varying parameter	Amplitude	Width	Position
Immunity towards noise	Low	High	High
Signal to noise ratio	Low	Moderate	Comparitively high
Need of synchronization pulse	Not exist	Not exist	Exist
Bandwidth dependency	On pulse width	On rise time of pulse	On rise time of pulse
Transmission power	Variable	Variable	Constant
Bandwidth requirement	Low	High	High
Similarity of implementation	Similar to AM	Similar to FM	Similar to PM
Synchronization between Transmitter and Receiver	Not needed	Not needed	Needed

Pulse Modulation

- PAM, PWM and PPM all three are analog **pulse modulation** techniques.
- The major difference between PAM, PWM & PPM lies in the parameter of a pulsed carrier that varies according to the modulating signal.
- In **PAM**, the amplitude of the pulsed carrier signal is varied according to the amplitude of analog modulating signal.
- In **PWM**, the width of the pulses of the carrier wave is varied according to the modulating signal.
- As against, in **PPM**, the position of the pulses shows variation according to the modulating signal.

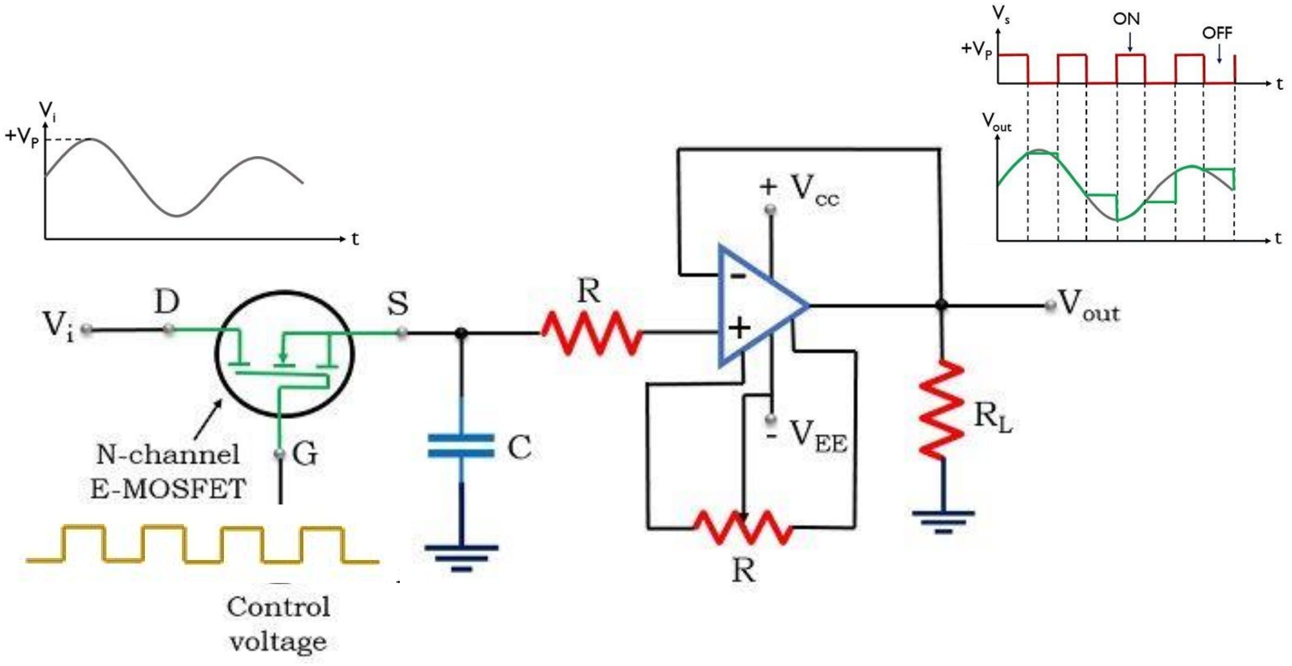
Signal Recovery Through Holding

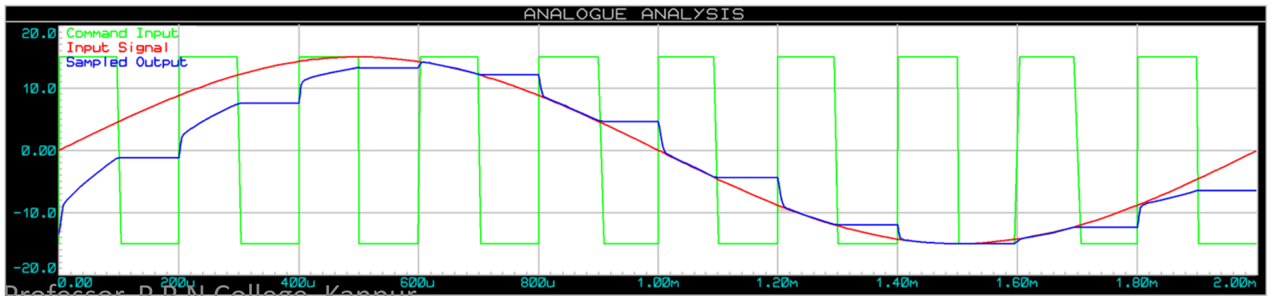
- There are two operations involved in the generation of the PAM signal.
  - Instantaneous sampling of the message signal $V_i(t)$ every T_s sec, where the sampling rate f_s is chosen in accordance with the sampling theorem.
  - Lengthening the duration of each sample, so obtained so some constant value T_h.
- These two operations are referred as **sample and hold**.



Signal Recovery Through Holding

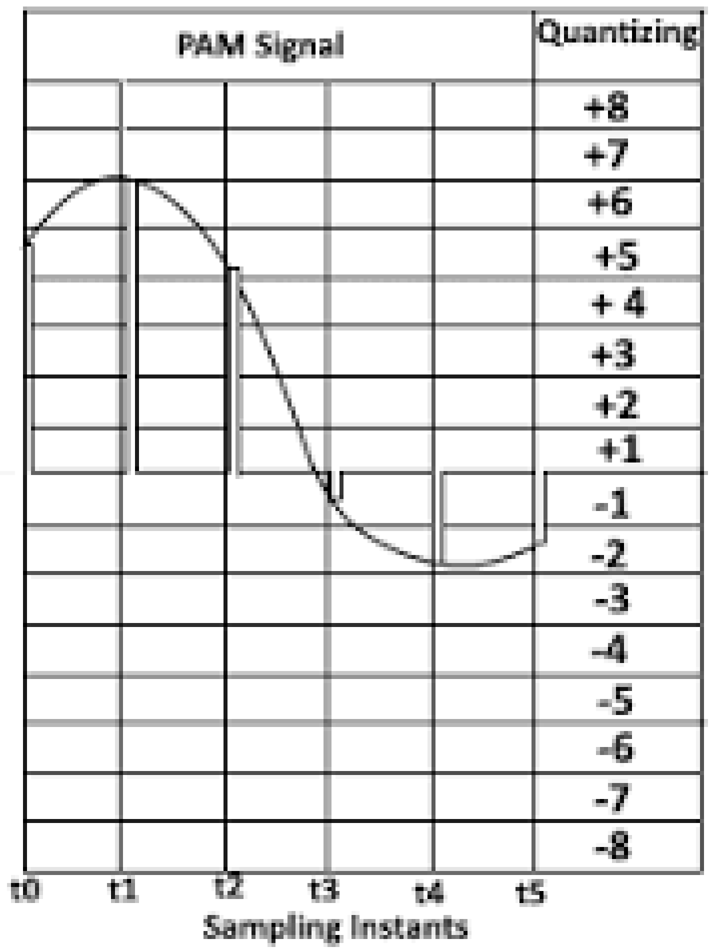
- In the circuit, the baseband signal $V_i(t)$ and its flat topped samples are shown.
- At the receiving end the sample pulses are extended, i.e., the sample value of each individual baseband signal is held until the occurrence of the next sample of that same baseband signal.
- The output waveform consists of an **up-down staircase waveform** with no bank intervals.





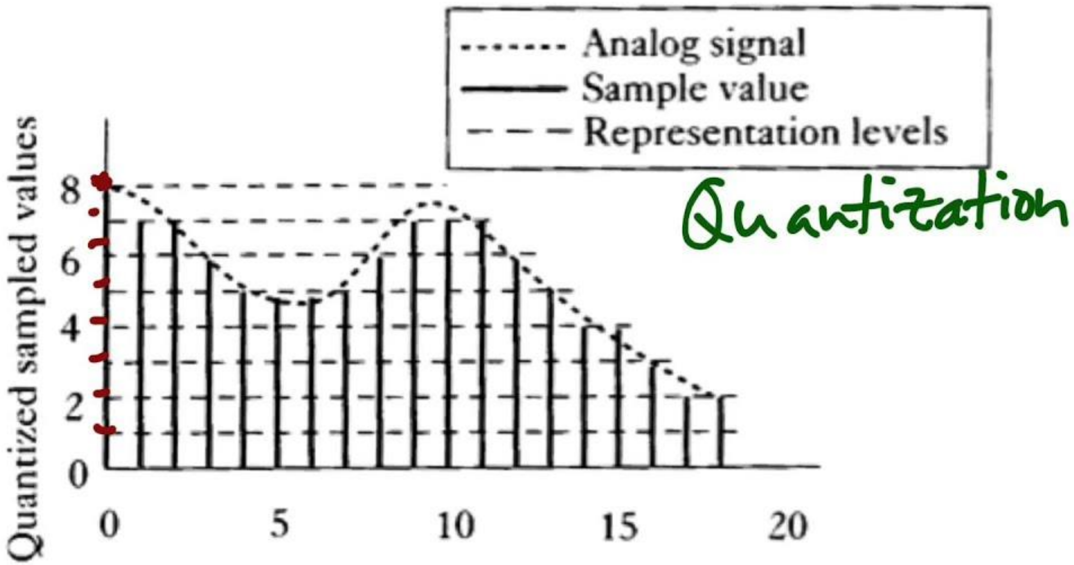
Quantization of Signals

- The modulating signal can assume an infinite number of different levels between the two limit values (V_L and V_H)
- Which define the range of the modulating signal is divided up into a number of small subrange.
- The number of subrange will depend on the nature of the modulating signal and will be from 8 to 256 levels.
- A number that is an integer power of two is chosen because of generating binary codes.

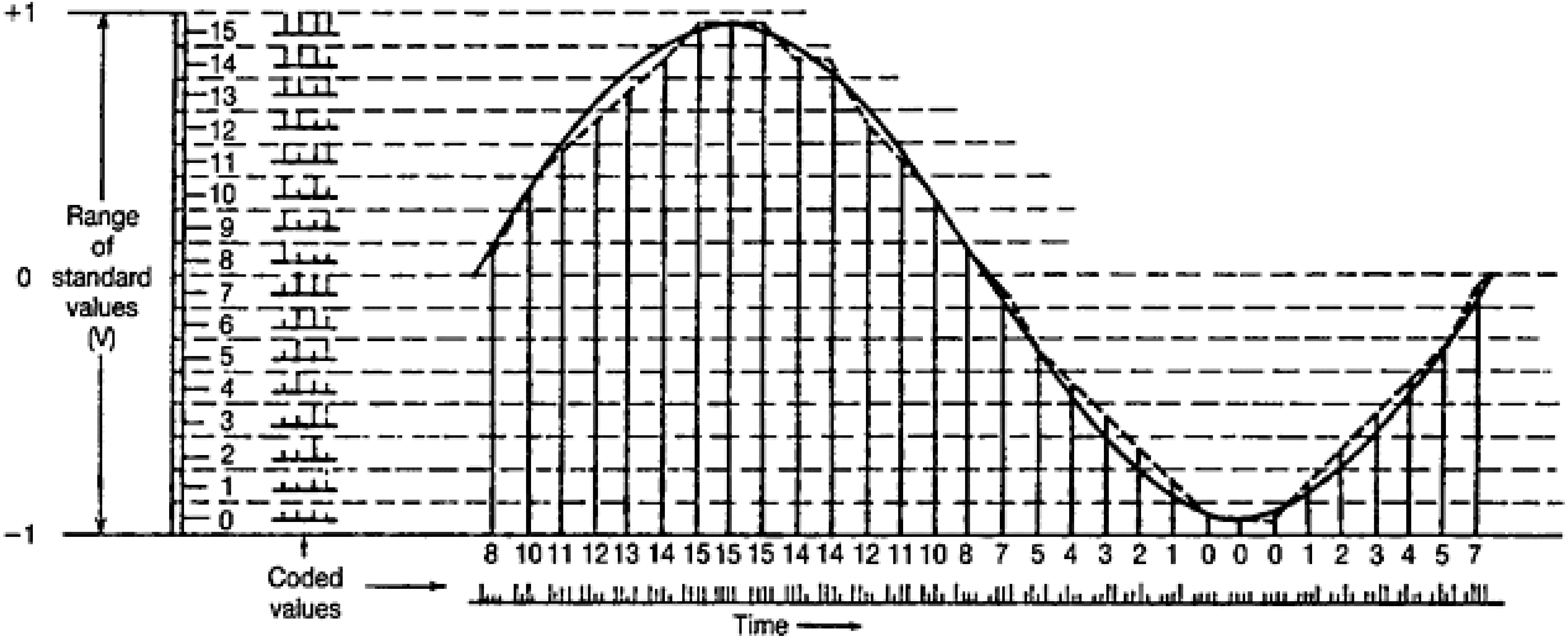


Quantization of Signals

- A new signal is generated by producing for each sample a voltage level corresponding to the midpoint level of the subrange in which the sample falls.
- The result is stepped waveform which follows the contour of the original modulating signal, with each step synchronized to the sampling period.



(a) Quantized sampling with 8 representation levels (3 bits per sample).



Quantization and resulting coding using 16 quantizing steps.

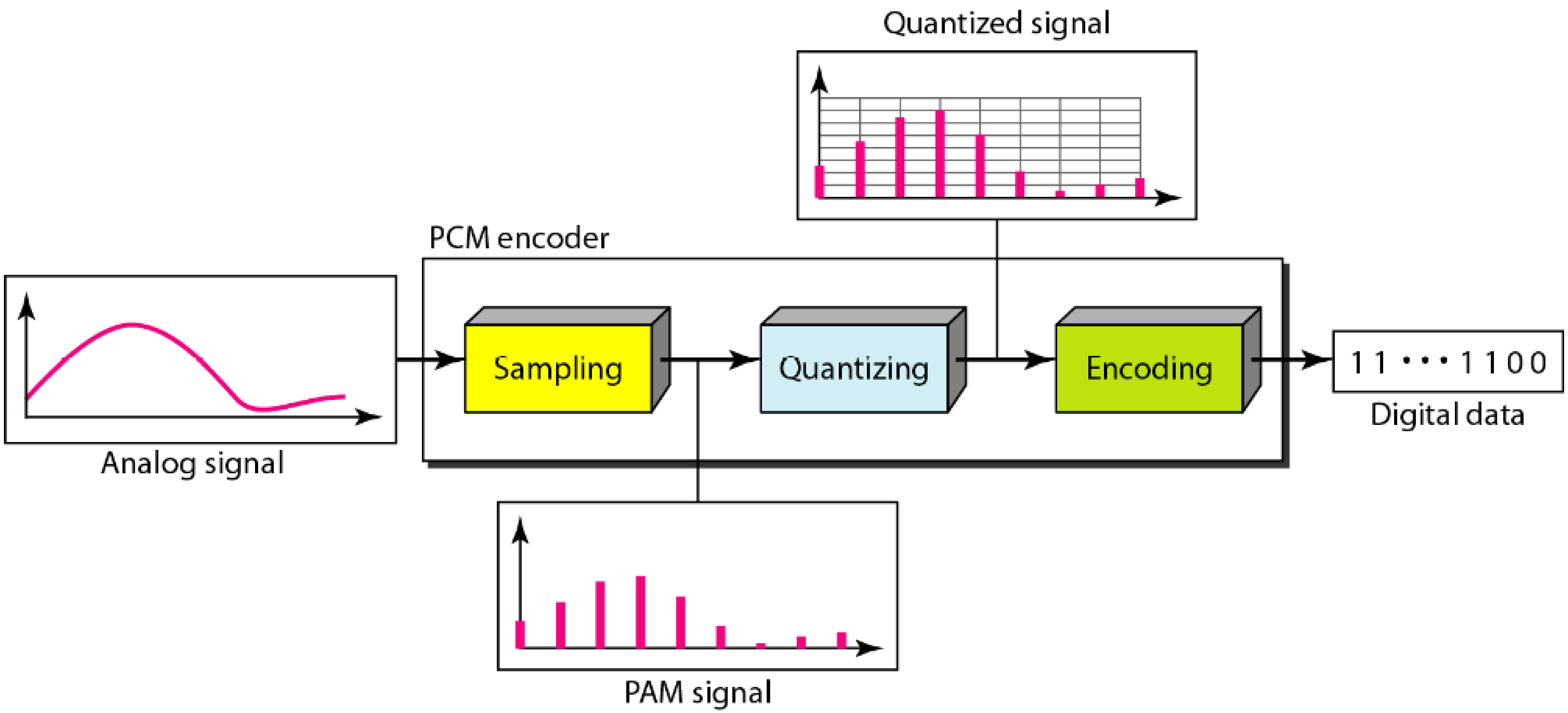
Quantization Error

- The quantized staircase waveform is an approximation to the original waveform.
- The difference between the two waveforms amount to noise added to the signal by the quantizing circuit, and is called **quantization error**.
- The **mean square quantization error** voltage has a value of

$$\overline{e^2} = \frac{S^2}{12}$$

- Where *e* is the difference between the original and quantized signal voltage and *S* is the voltage of each steps or the subrange voltage.

Encoding

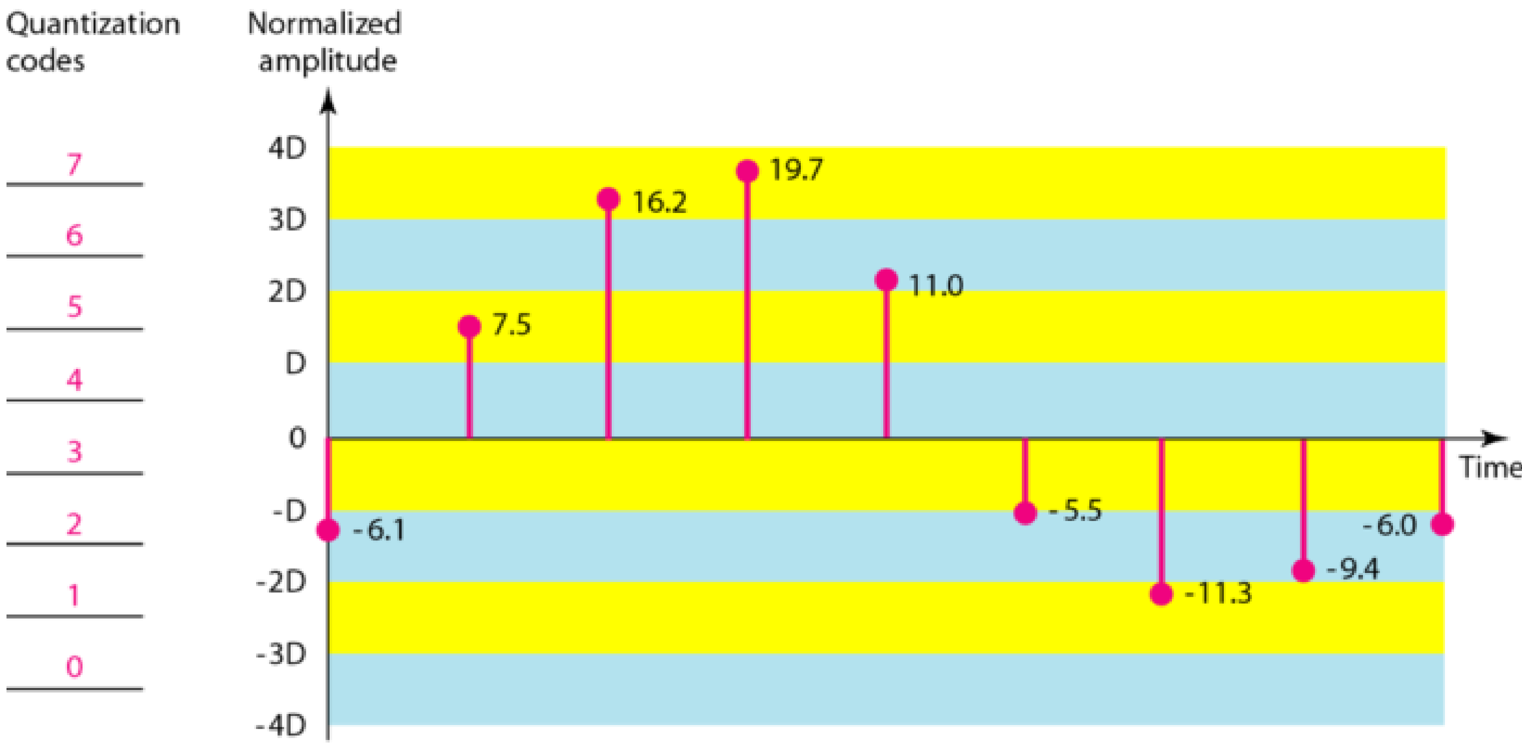

Quantized signal
PCM encoder
Sampling
Quantizing
Encoding
11 • • • 1100
Digital data
Analog signal
PAM signal

Coding of Signals

Sampling through holding

Quantization

Encoding



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

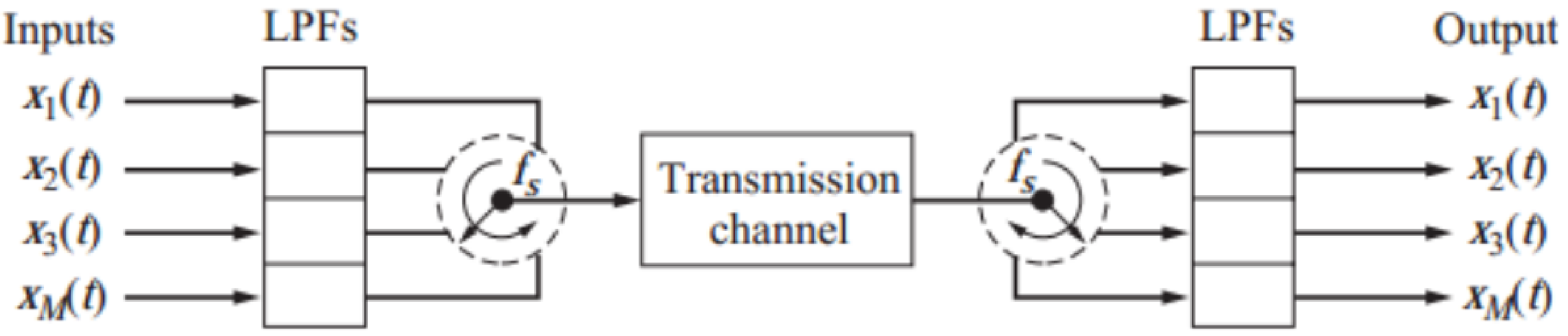
Encoding

- Encoding is the process of representing the sampled values as a **binary number** in the range 0 to n.
- The value of n is chosen as a **power of 2**, depending on the accuracy required.
- **Increasing n** reduces the step size between adjacent quantization levels and hence **reduces the quantization noise**.
- The down side of this is that the amount of digital data required to represent the analogue signal increases.

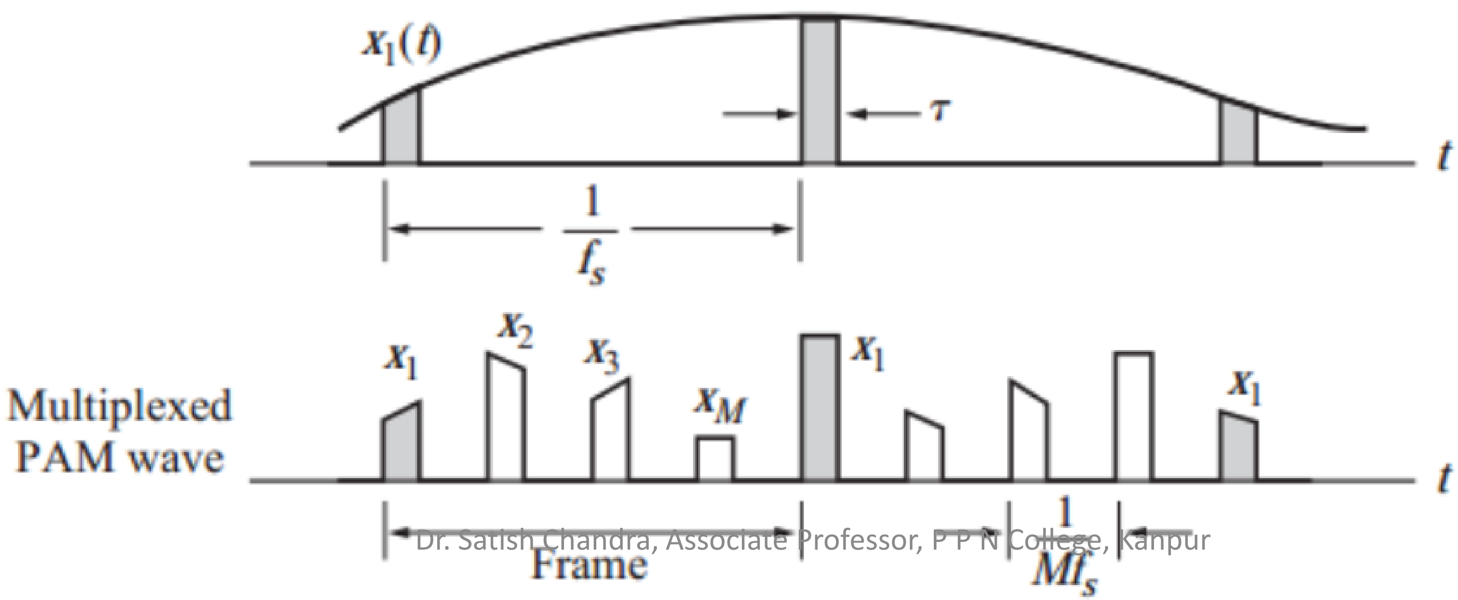
Multiplexing

- Multiplexing may be defined as a technique which allows many users to share a common communication channel, simultaneously.
- There are two major types of multiplexing techniques
  - Frequency division multiplexing (FDM)
  - Time division multiplexing (TDM)
- In FDM the frequency spectrum is divided among the logical channels (stations), with each user having exclusive possession of his frequency band.
- In TDM the frequency spectrum is completely allotted to one user for a fixed time slot. Each user, in this way, gets the entire bandwidth periodically for small time interval.

Time Division Multiplexing

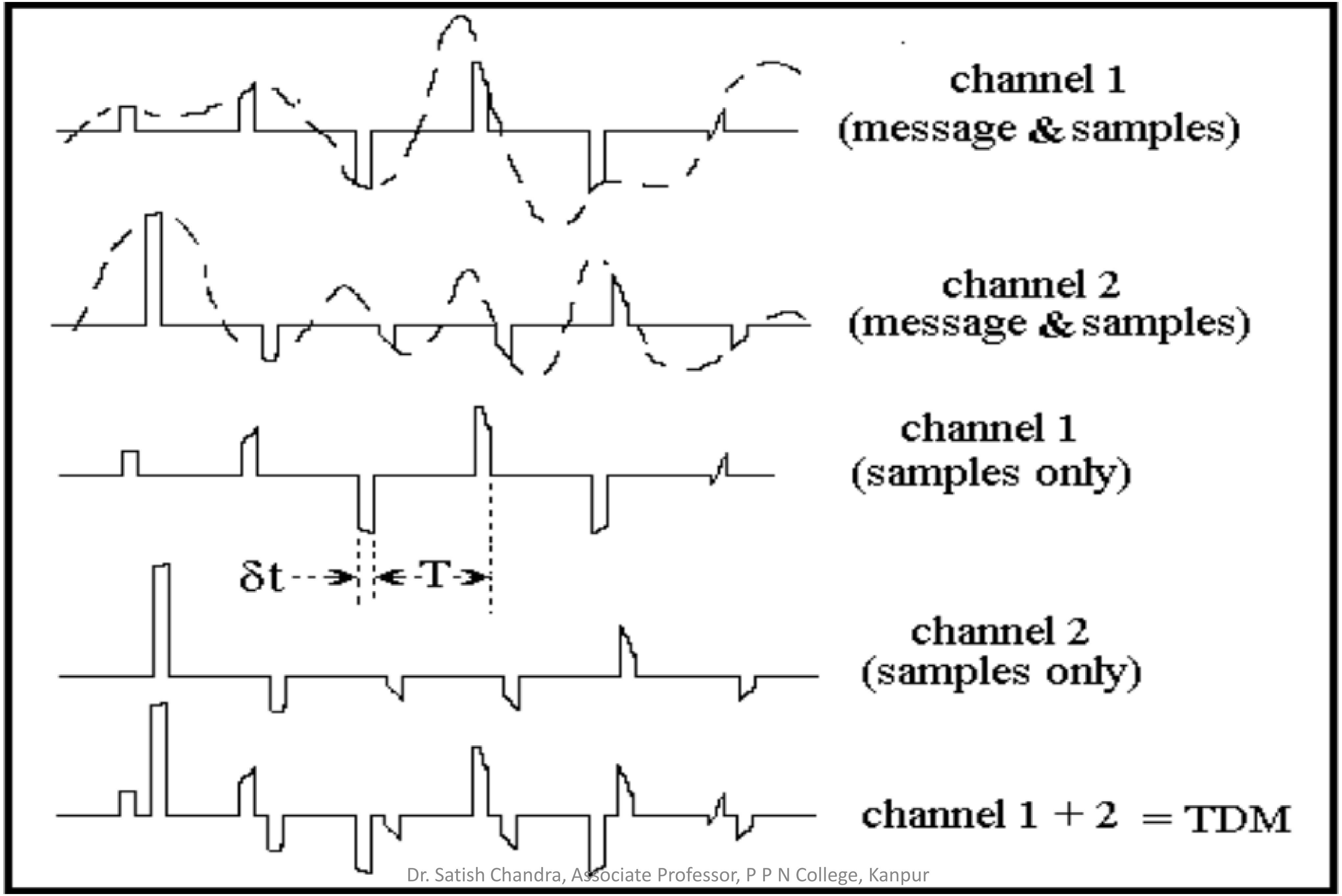


(a)



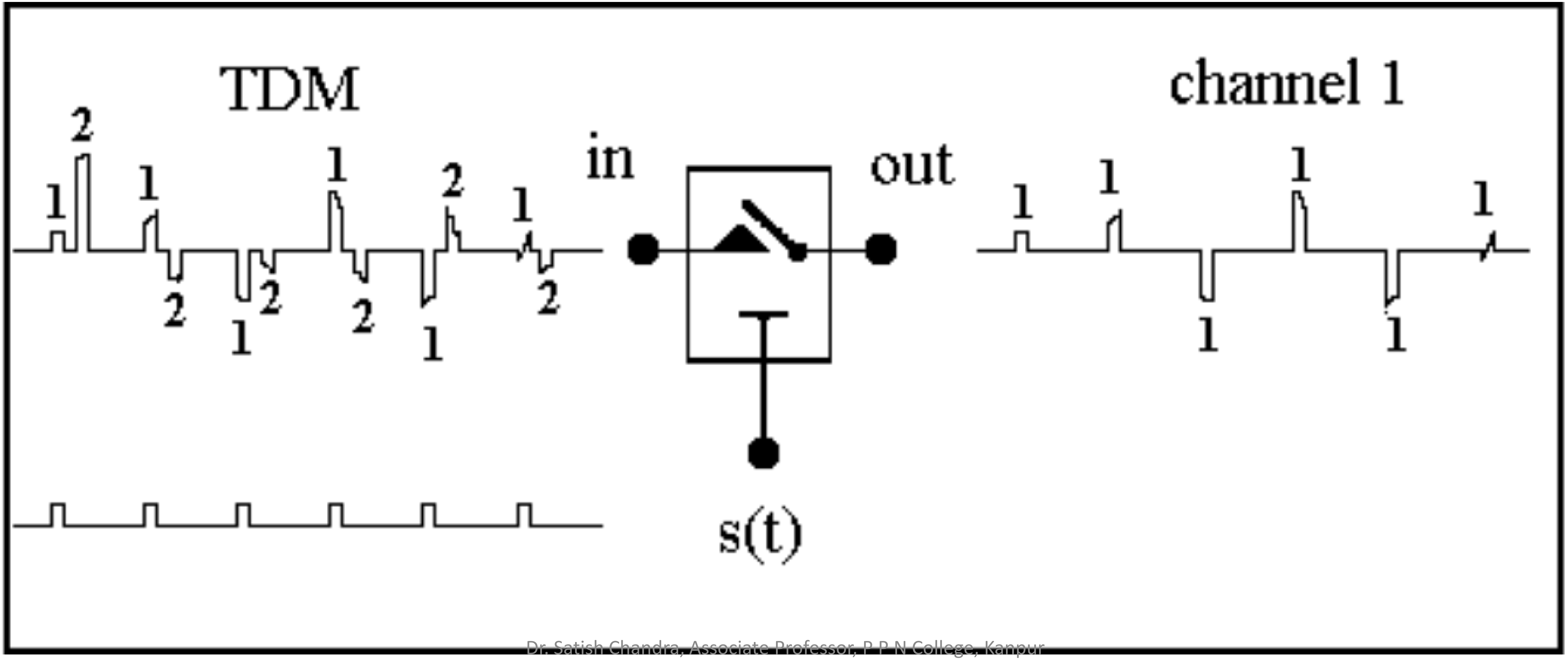
PAM-TDM

- The signal that is to be converted to PAM is fed to one input and pulses (at the sampling frequency) are applied to other input of the AND gate, to open it during the desired time intervals.
- The output of the AND gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant.
- The pulses are then passed through a pulse-shaping network which gives them flat tops.
- Frequency modulation (FM) is then employed so that the system becomes PAM-FM.


channel 1
(message & samples)
channel 2
(message & samples)
channel 1
(samples only)
δt
T
channel 2
(samples only)
channel 1 + 2 = TDM

PAM-TDM

- In the receiver, the pulses are first recovered with a standard FM demodulator.
- They are then fed to an ordinary diode detector which is followed by a low-pass filter.
- If the cut-off frequency of this LPF is high enough to pass the highest signal frequency, but low enough to remove the sampling frequency ripple, an undistorted replica of the original signal is reproduced.


TDM
channel 1
in
out
s(t)
1
2
1
2
1
2
1
2
2
1
1
2
1
1
1
1
1
1
1

Channel Bandwidth for PAM-TDM Signal

- Suppose we have M independent signal $x_1(t), x_2(t), x_3(t), \ldots$ etc., each of which is bound limited to f_m, i.e., the highest signal frequency present in all the channels is f_m.
- Then, by sampling theorem, the sampling frequency f_s must be

$$f_s \geq 2f_m$$

- and

$$T_s \leq \frac{1}{2f_m}$$

- because

$$T_s = \frac{1}{f_s}$$

Channel Bandwidth for PAM-TDM Signal

- Hence, the sampling interval T_s consists of one sample from each signal, and thus total M samples. Therefore, spacing between two samples will be equal to T_s/M.
- Hence, the number of pulses per second or pulse frequency will be the reciprocal of spacing between two pulses.
- It is also known as signalling rate of PAM-TDM signal. Hence,

$$signalling\ rate = \frac{M}{T_S} = Mf_s$$

- Since, $f_s \geq 2f_m$ $\qquad signalling\ rate \geq 2Mf_m$

Channel Bandwidth for PAM-TDM Signal

- As the signal $x_1(t)$, $x_2(t)$, $x_3(t)$, ... etc., are baseband signal, they must be obtained by passing pulsed TDM signal through low pass filter.
- The bandwidth B_{LPF} of this low pass filter is given by half of signalling rate. Therefore,

$$B_{LPF} = \frac{1}{2} M f_s$$

- The transmission bandwidth (B) of PAM-TDM channel will be equal to bandwidth of the low pas filter (B_{LPF}). Thus,

$$B = \frac{1}{2} M f_s$$

Channel Bandwidth for PAM-TDM Signal

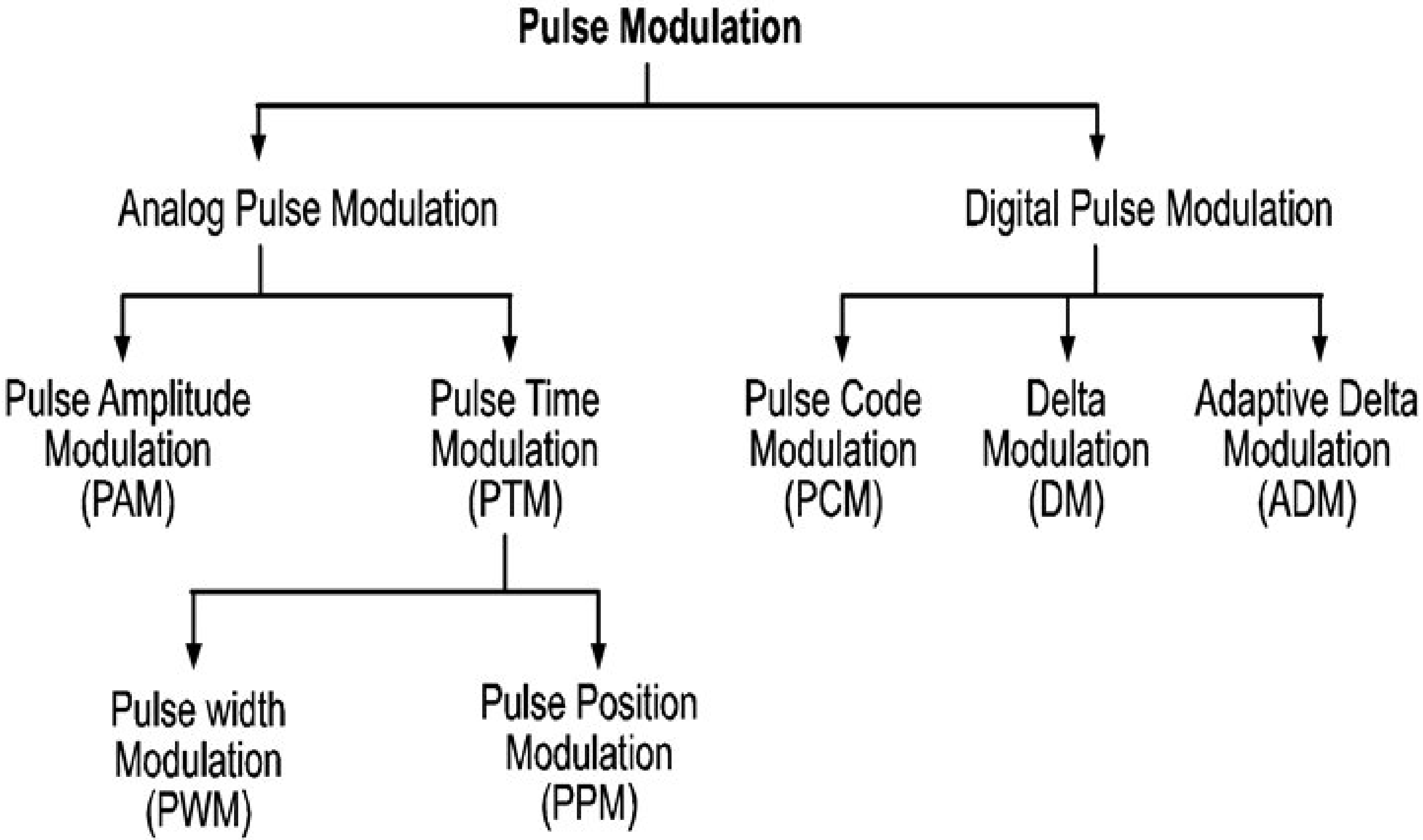
- If sampling rate becomes equal to Nyquist rate, then

$$f_s = 2f_m$$

$$B = Mf_m$$

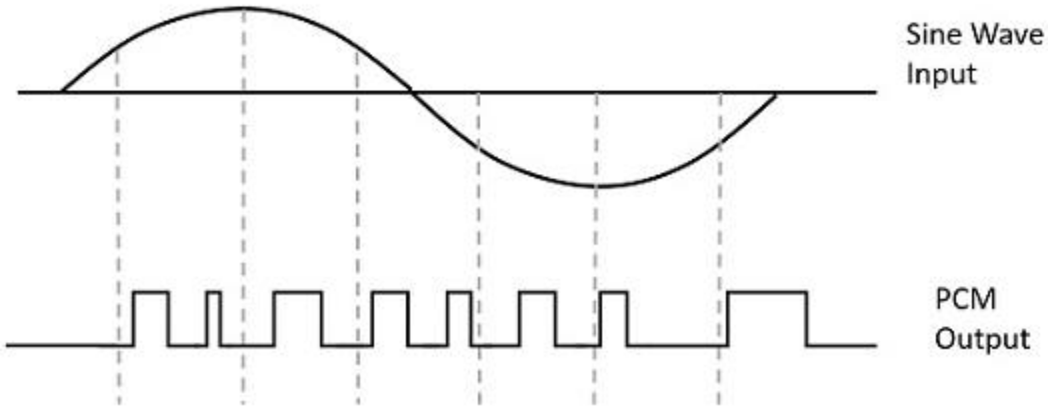
- The B is the minimum transmission bandwidth of PAM-TDM channel.
- If there are M channels in TDM, band limited to f_m, then minimum bandwidth of the transmission channel will be equal to Mf_m.

Pulse Modulation Technique



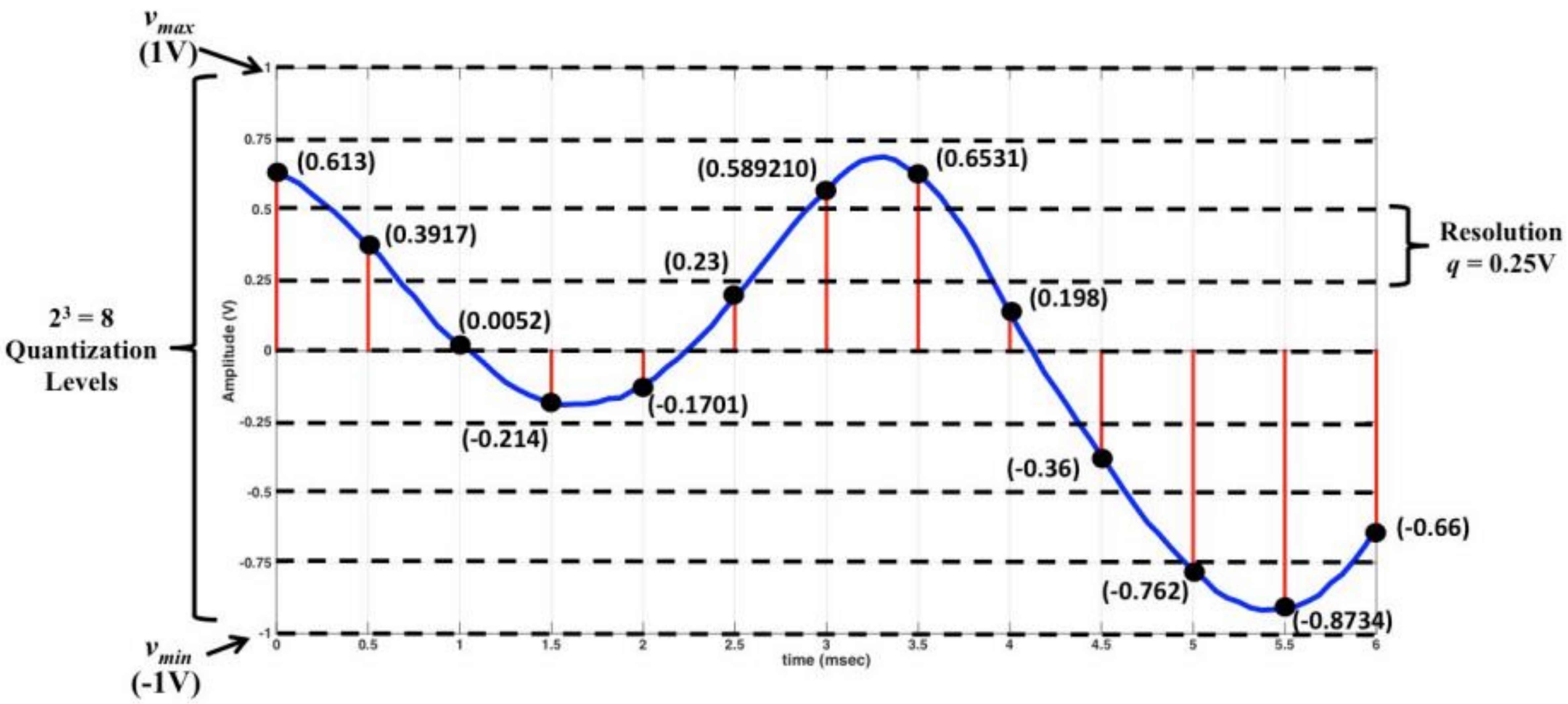
Pulse Code Modulation (PCM)

- Pulse code modulation is used to convert analog signals into binary digital form.
- In the PCM system, group of pulses or codes are transmitted which represent binary numbers corresponding to modulating voltage levels.
- Recovery of the transmitted information does not depend on the height, width or energy content of the individual pulses, but only on their presence or absence.

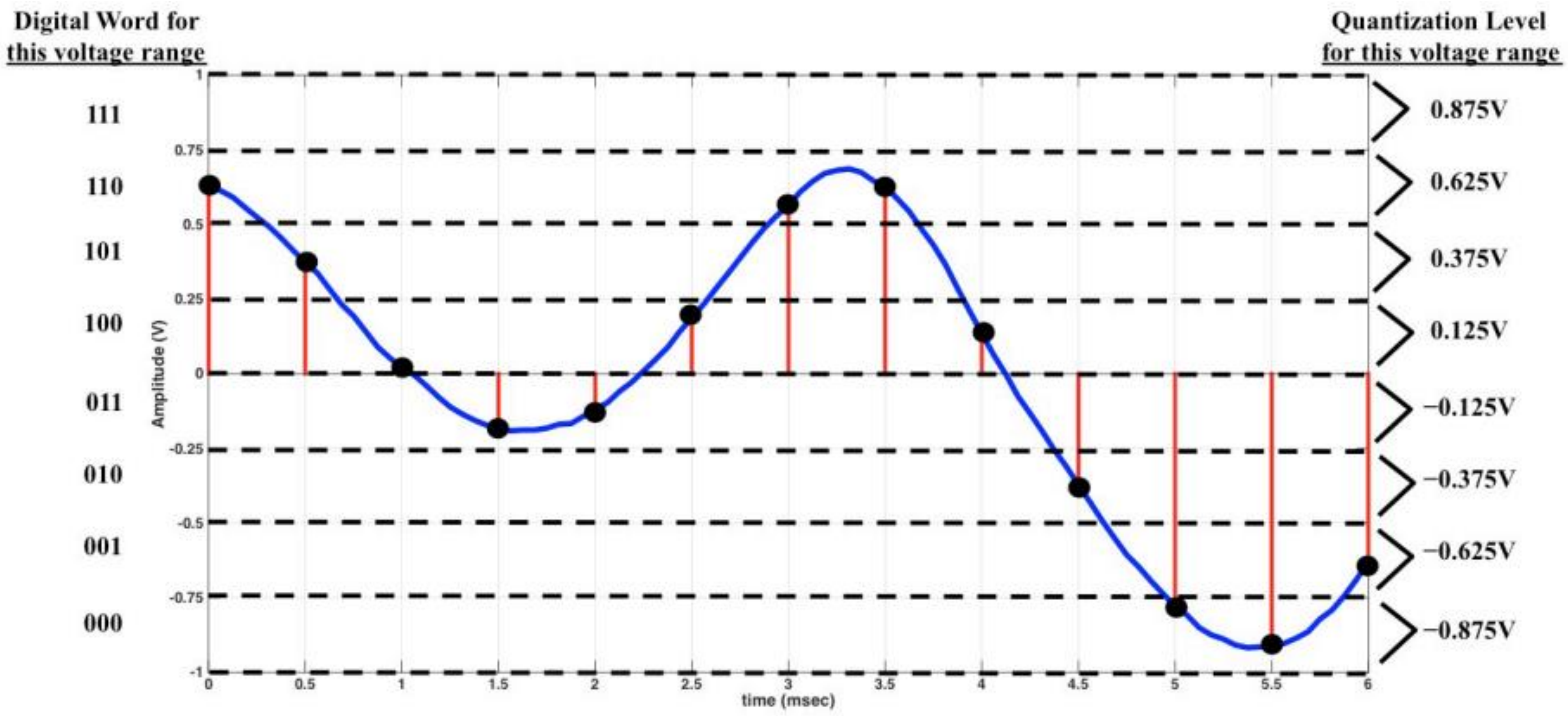


Pulse Code Modulation (PCM)

- PCM system tend to be very immune to interference and noise. Since it is relatively easy to recover pulses under these condition, even in the presence of large amount of noise and distortion.
- Regeneration of the pulse on the way is also relatively easy resulting in a system that produces excellent results for long distance communication.
- This system is very economical to realize, because it corresponds exactly to the process of A/D conversion.
- At the receiver end, each binary set of pulses are reconstructed into a pulse whose amplitude is proportional to the original digit sent.



v_{max} (1V)
v_{min} (-1V)
$2^3 = 8$ Quantization Levels
Resolution $q = 0.25V$
(0.613)
(0.3917)
(0.0052)
(-0.214)
(-0.1701)
(0.23)
(0.589210)
(0.6531)
(0.198)
(-0.36)
(-0.762)
(-0.8734)
(-0.66)
Amplitude (V)
time (msec)

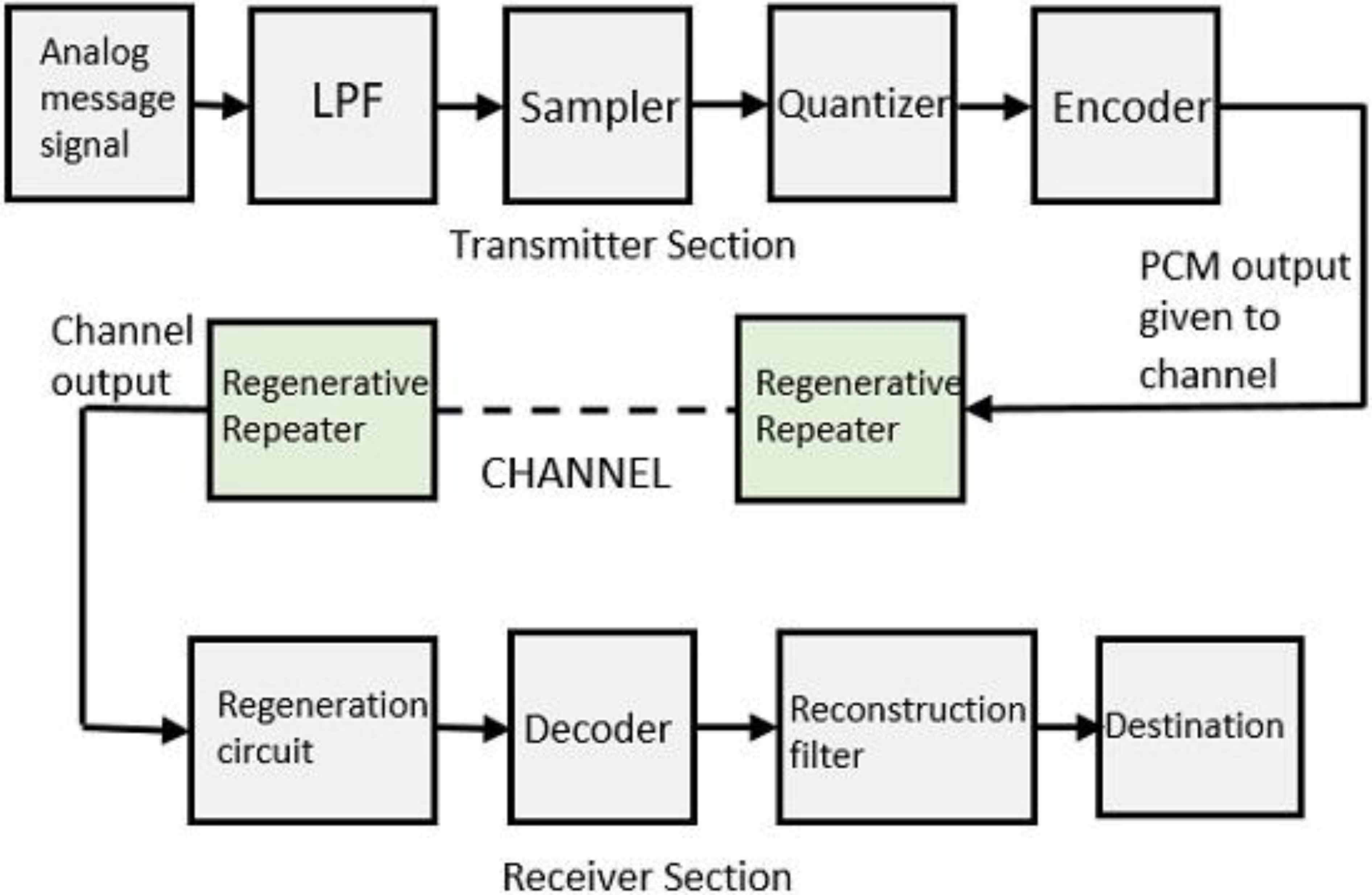


The binary representation of the above signal is:

110 101 100 011 011 100 110 110 100 010 000 000 001.

Basic Elements of PCM

- The **transmitter section** of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section.
- The low pass filter prior to sampling prevents aliasing of the message signal.
- The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train.
- Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.


Analog message signal
LPF
Sampler
Quantizer
Encoder
Transmitter Section
PCM output given to channel
Channel output
Regenerative Repeater
CHANNEL
Regenerative Repeater
Regeneration circuit
Decoder
Reconstruction filter
Destination
Receiver Section

Basic Elements of PCM

Low Pass Filter

- This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

- This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal.
- The sampling rate must be greater than twice the highest frequency component f_m of the message signal, in accordance with the sampling theorem.

Basic Elements of PCM

Quantizer

- Quantizing is a process of reducing the excessive bits and confining the data.
- The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

- The digitization of analog signal is done by the encoder.
- It designates each quantized level by a binary code.
- The sampling done here is the sample-and-hold process.
- These three sections LPF, Sampler, and Quantizer will act as an **analog to digital converter**.
- Encoding minimizes the bandwidth used.

Basic Elements of PCM

Regenerative Repeater

- This section increases the signal strength.
- The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

Decoder

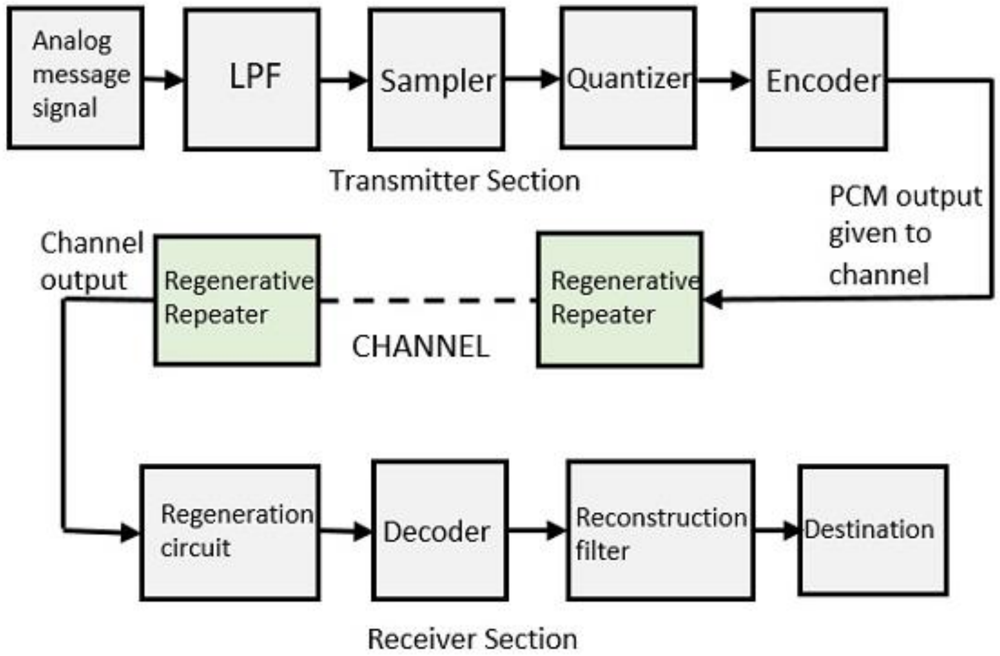
- The decoder circuit decodes the pulse coded waveform to reproduce the original signal.
- This circuit acts as the **demodulator**.

Basic Elements of PCM

Reconstruction Filter

- After the **digital-to-analog conversion** is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal.

PCM

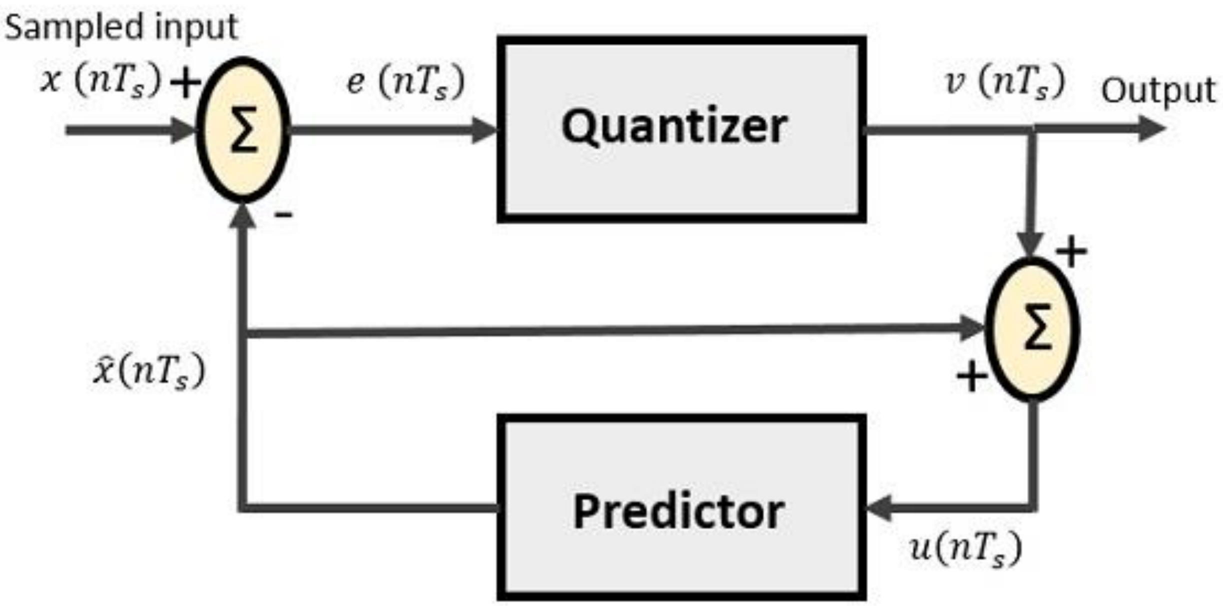


Differential PCM

- Differential pulse code modulation (DPCM) is a technique in which the difference between samples, rather than the sample values themselves, is encoded in binary.
- It is quite similar to PCM, however, each word in DPCM system indicates the difference in amplitude positive or negative between this sample and the previous sample.
- The reason to chose DPCM is that, speech samples do not change drastically from sample to sample, and therefore the difference values can be encoded using fewer bits.
- Smaller bandwidth would be required for the transmission.

DPCM Transmitter

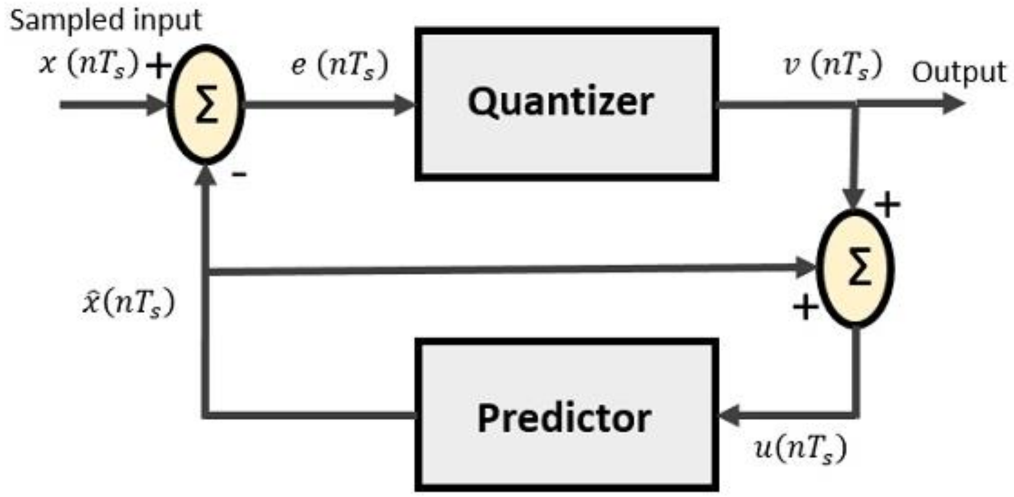
- The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.



DPCM Transmitter

The signals at each point are named as –

- $x(nT_s)$ is the sampled input.
- $\hat{x}(nT_s)$ is the predicted sample.
- $e(nT_s)$ is the difference of sampled input and predicted output, often called as *prediction error.*
- $v(nT_s)$ is the quantized output.
- $u(nT_s)$ is the predictor input which is actually the summer output of the predictor output and the quantizer output

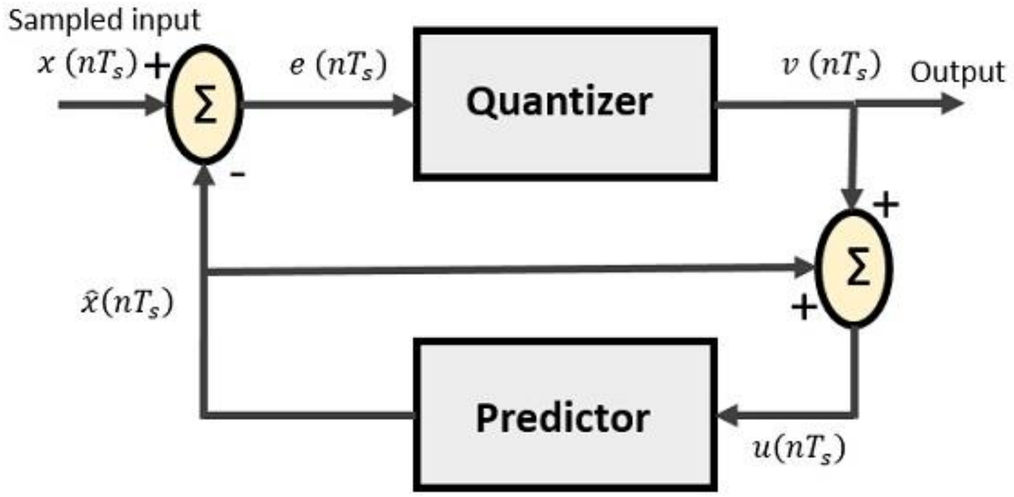


DPCM Transmitter

- The predictor produces the assumed samples from the previous outputs of the transmitter circuit.
- The input to this predictor is the quantized versions of the input signal $x(nT_s)$.
- Quantizer Output is represented as –

$$v(nT_s) = Q[e(nT_s)] = \mathrm{e}(nT_s) + \mathrm{q}(nT_s)$$

- Where $\mathrm{q}(nT_s)$ is the quantization error.



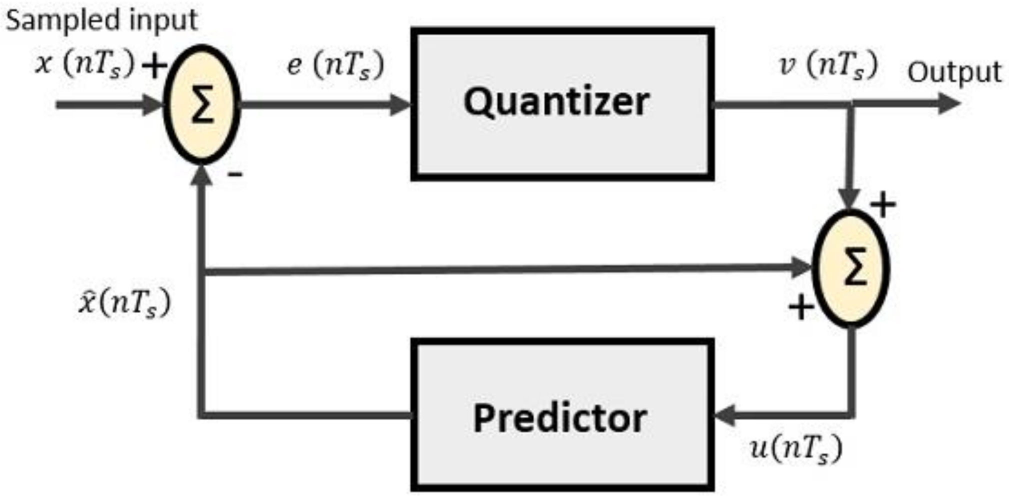
DPCM Transmitter

- Predictor input is the sum of quantizer output and predictor output,

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

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

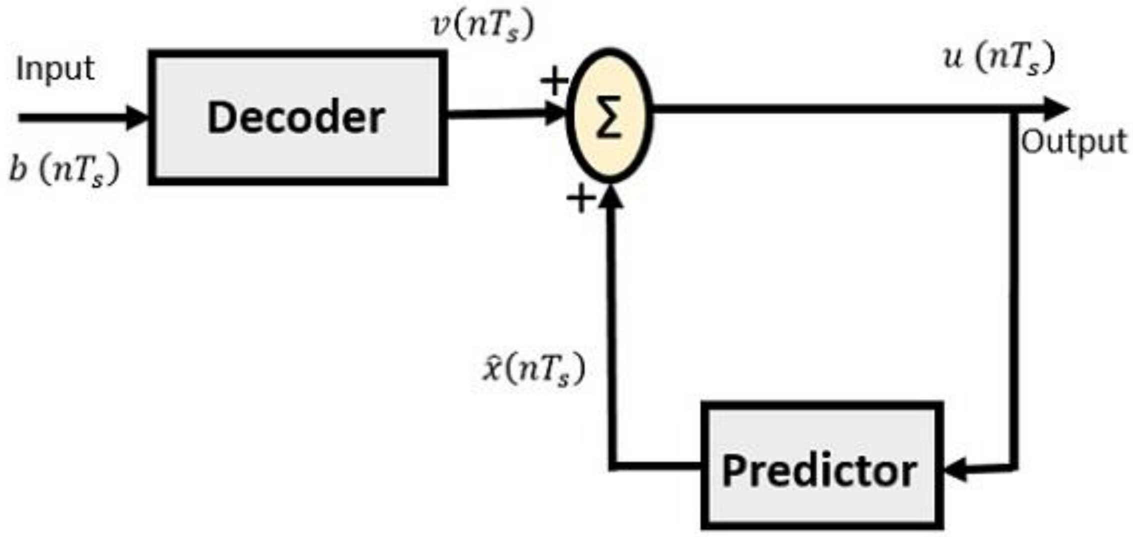
$$u(nT_s) = x(nT_s) + q(nT_s)$$



- The same predictor circuit is used in the decoder to reconstruct the original input.

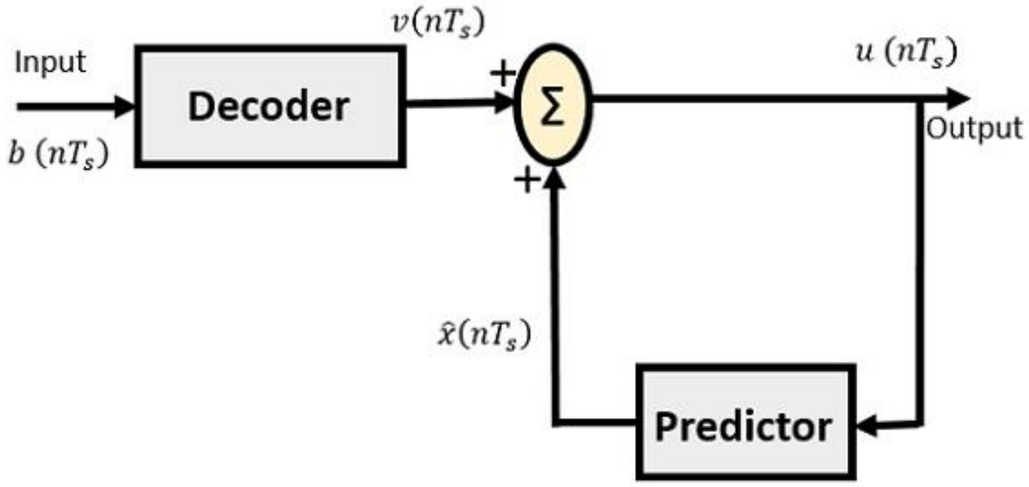
DPCM Receiver

- The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit. Following is the diagram of DPCM Receiver.



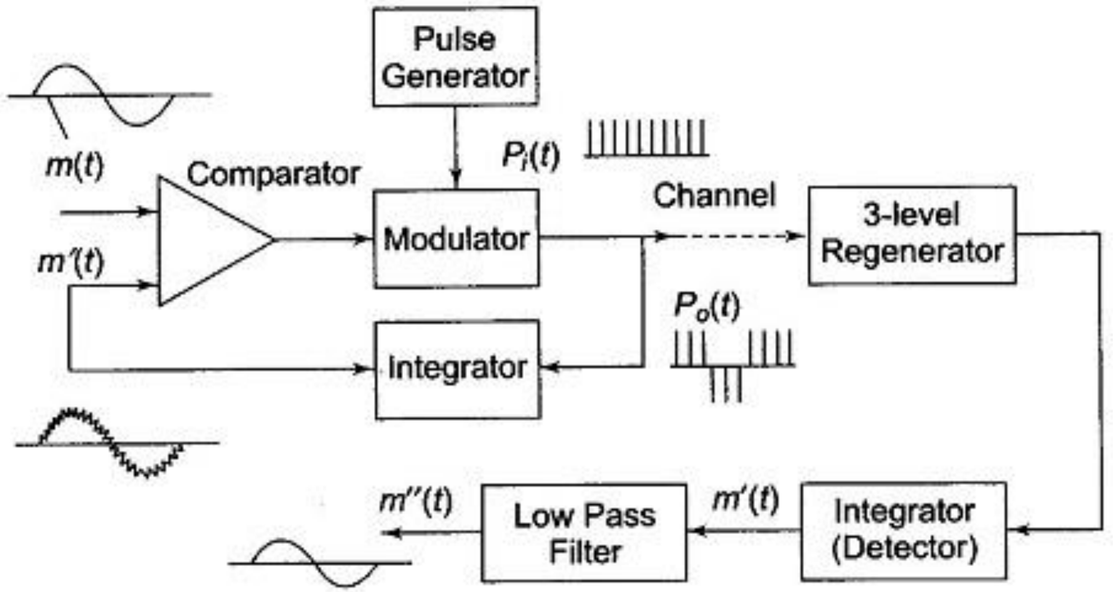
DPCM Receiver

- The receiver for reconstructing the quantized version of the input consists of a decoder to reconstruct quantized error signal, $v(nT_s)$.
- The quantized version of the original input, $u(nT_s)$ is reconstructed using the quantized error signal, $v(nT_s)$ and output of the same prediction filter used in transmission, $\hat{x}(nT_s)$.
- The receiver output is equal to $u(nT_s)$, which differ from the original input $x(nT_s)$ only by the quantization error.



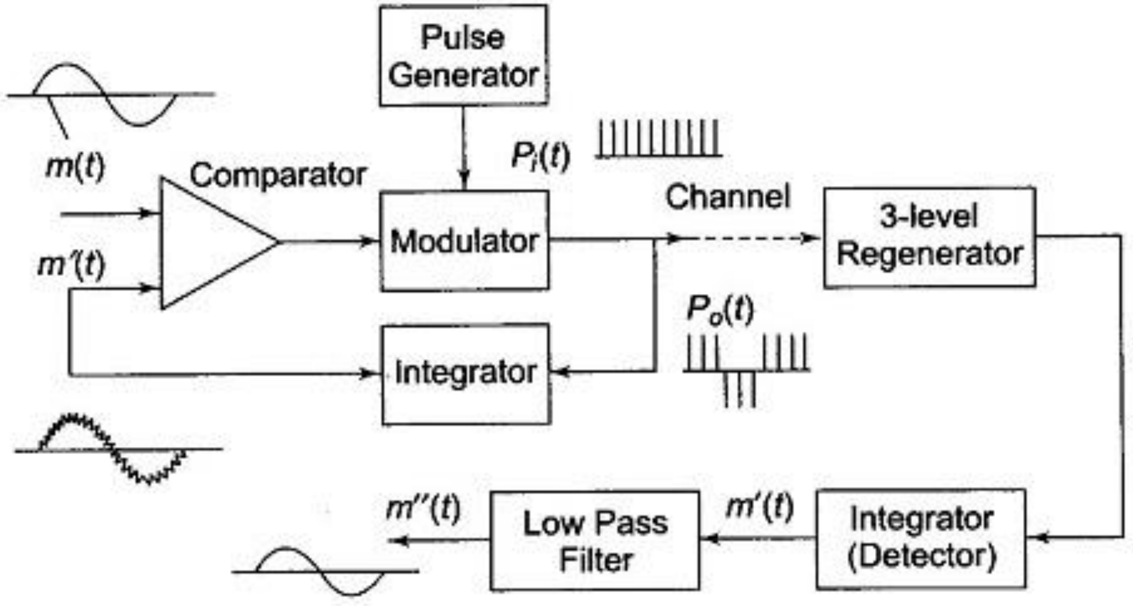
Delta Modulation

- In delta modulation (DM) a single bit is sent per sample to indicate, whether the signal is larger or smaller than the previous sample.
- In DM only the polarity of the difference signal is encoded as output.
- If the difference between the input and feedback signal is positive, it is encoded as a binary 1, and if negative, as a binary 0.



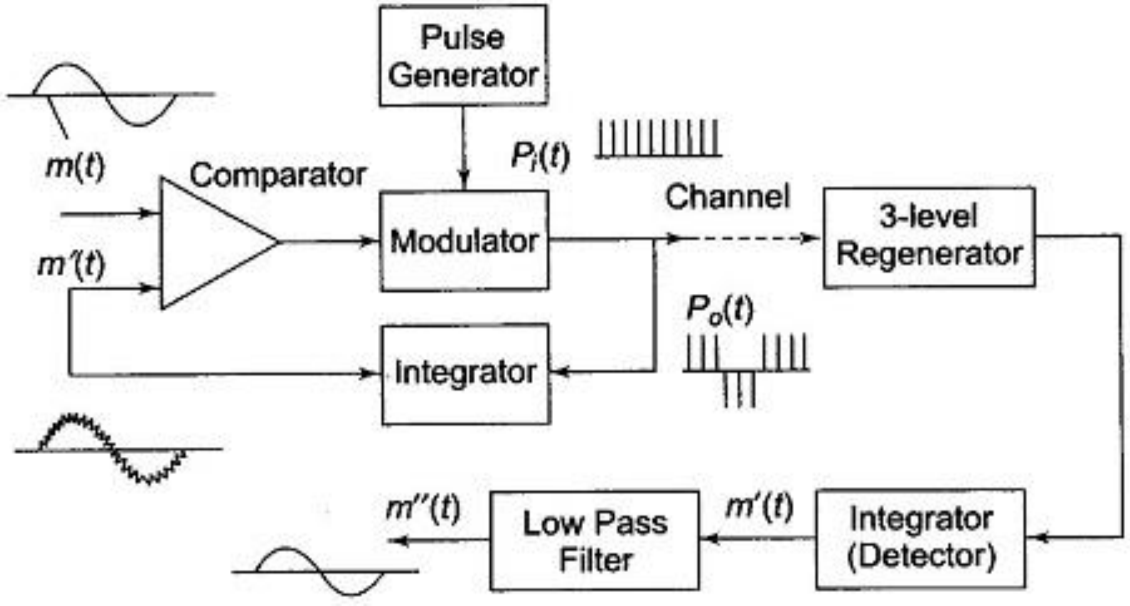
DM Transmitter

- One input to the modulator is a positive train of unipolar impulses (very short pulses) $P_i(t)$ at the sampling frequency f_s.
- The other input is the output from the comparator, which consists of fixed amplitude pulses, whose amplitude depends on the difference signal at the comparator input.
- The polarity is positive, if the analog signal m(t) is greater than and negative, if less than the feedback signal m'(t).



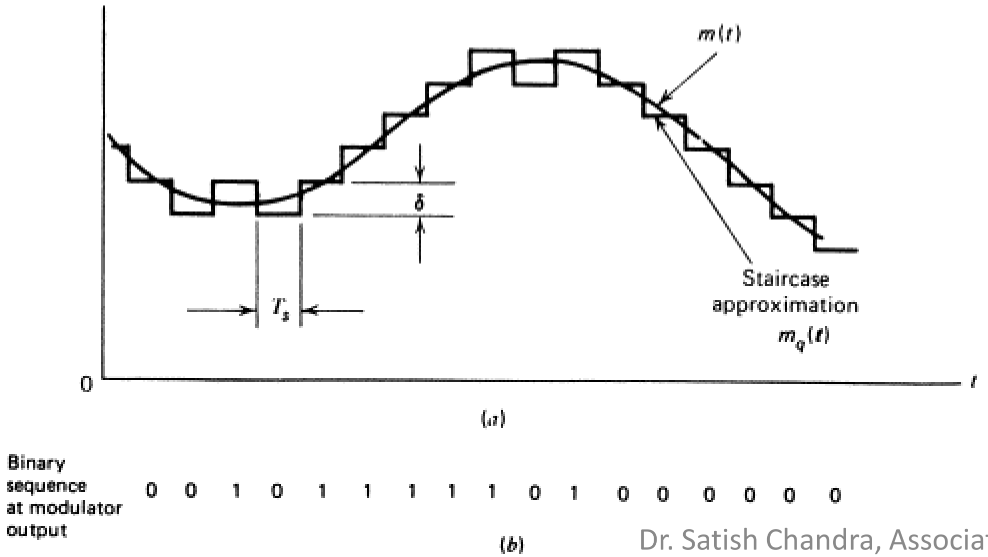
DM Transmitter

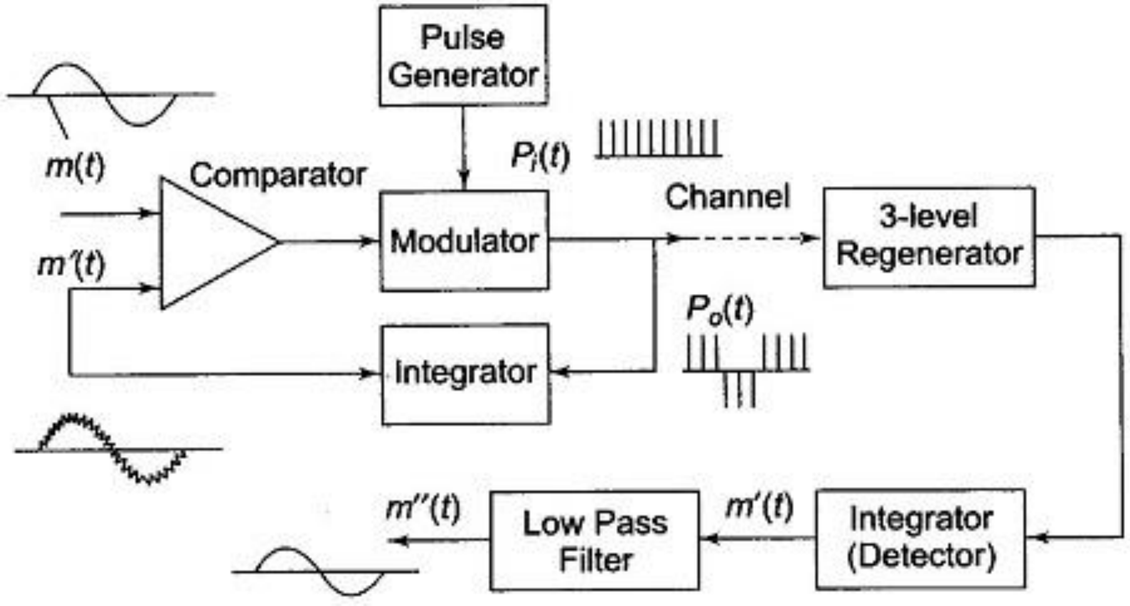
- The modulator output is therefore a sequence of impulses $P_o(t)$ whose polarity depends on the difference signal.
- The feedback signal m'(t) is the integral of the modulator output $P_o(t)$, when it is passed through the integrating circuit.
- The integrator causes m'(t) to rise or fall by a fixed step height for each positive or negative impulse $P_o(t)$ applied to its input.



DM Transmitter

- The m'(t) waveform is a staircase waveform approximation to the original signal m(t).
- The impulse waveform $P_i(t)$ is converted to a line waveform and transmitted.





DM Receiver

- A regenerator is used to recover Pi(t) impulse waveform, which is integrated to produce the staircase waveform approximation similar to that at the transmitter.
- The replica m'(t) of the original signal m(t) is recovered from the integrator output through the use of LPF.
- The recovered waveform will exhibits the effect of granular noise and slope overload and will not be the exact replica of the original input.