FR. CONCEICAO RODRIGUES COLLEGE OF ENGINEERING
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Module 5.0
Pulse Modulation & Multiplexing

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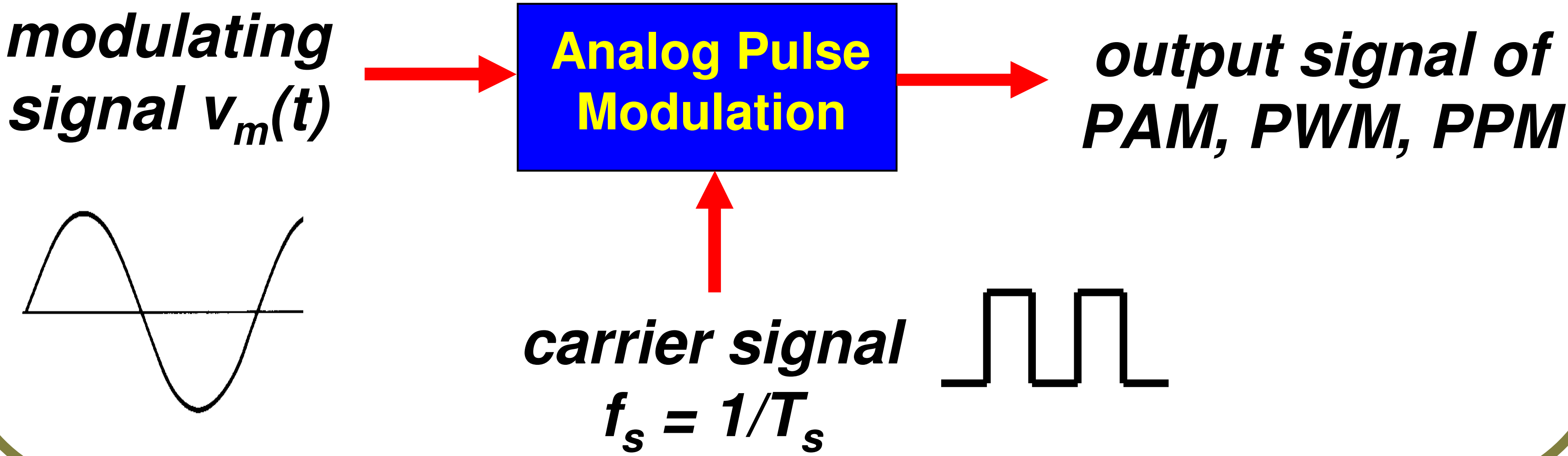
Electronic Circuits & Communication Fundamentals
ECCF (CSC 304) for S.E. (Computer Engg.) – Semester III

Analog Pulse Modulation

- **AF modulating (baseband) signal is given by $v_m(t) = V_m \sin\omega_m t$ OR $v_m(t) = V_m \cos\omega_m t$**
- **HF carrier signal is a train of pulses having a frequency of $f_s = 1/T_s$**



Analog Pulse Modulation

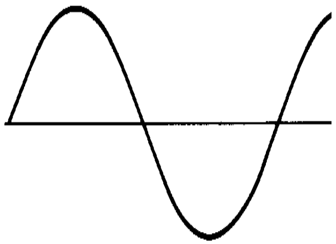
Here the carrier signal is a train of pulses & is ($f_s = 1/T_s$) modulated by $v_m(t)$ as follows :-

- **<u>Pulse Amplitude Modulation (PAM)</u> if carrier pulses amplitude varies with $v_m(t)$**
- **<u>Pulse Width Modulation (PWM)</u> if the carrier pulses time period (T_s) varies with $v_m(t)$**
- **<u>Pulse Position Modulation (PPM)</u> if carrier pulses position varies with $v_m(t)$**

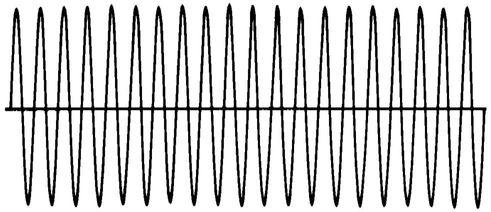
In each case either amplitude, width or position (any two) will remain constant or unchanged

CW & Pulse Modulation (Analog)

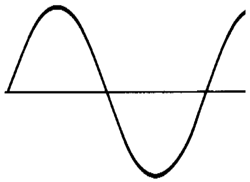
modulating signal $v_m(t)$ → **Continuous Wave Modulation** → ***output signal AM, FM or PM***



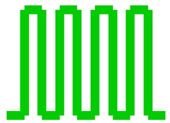
carrier signal $v_c(t)$



Modulating Signal $v_m(t)$ → **Analog Pulse Modulation** → ***Output Signal PAM, PWM, PPM***



Carrier Signal $f_s = 1/T_s$



Pulse Modulation Advantages

Pulse Modulation has the following advantages over continuous wave (CW) modulation :-

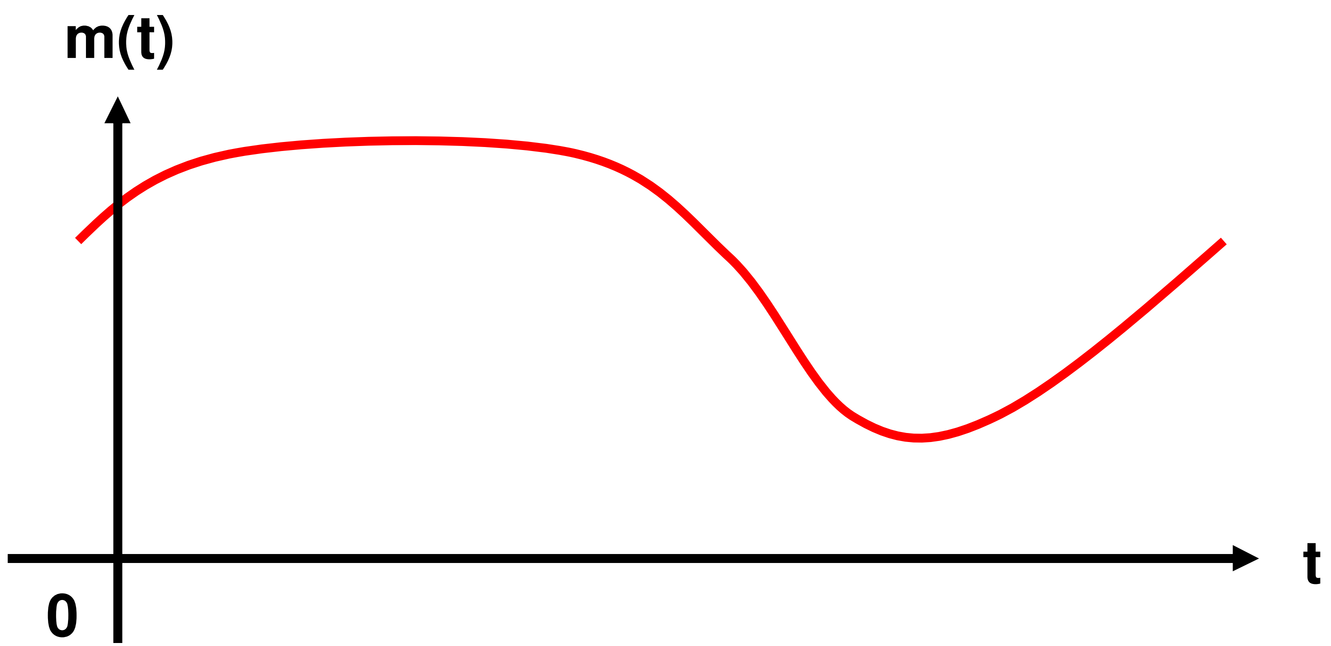
- **Power Saving –** since the pulses operate in ON/OFF fashion, power saving is possible as compared to CW modulation technique
- **Multiplexing –** during OFF time interval of a pulse train, series of other pulses can easily be accommodated, enabling multiplexing

these two important advantages make pulse modulation beneficial over the continuous wave (CW) modulation

How many pulses (samples) ?

Consider analog bandlimited input modulating signal m(t) as shown below :-

SAMPLE m(t) into m(nT) USING SAMPLING



How many pulses (samples) ?

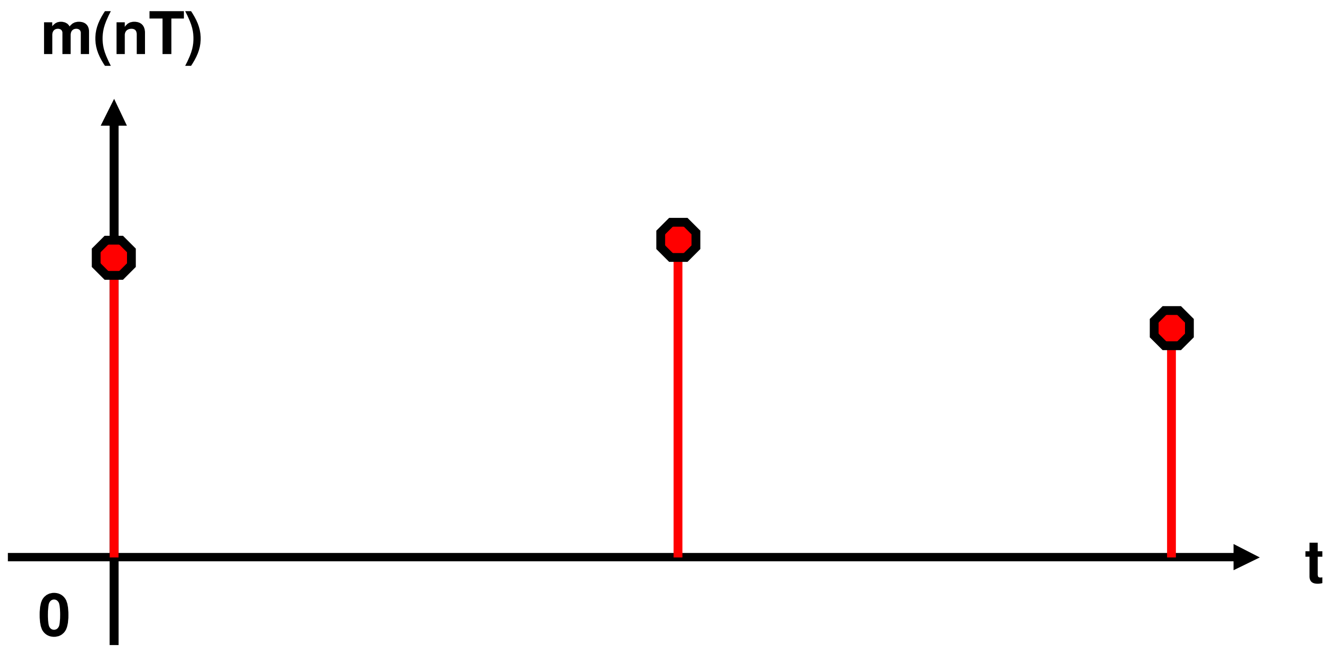
If two (2) samples are made out of m(t), are they sufficient to completely represent m(t) ?

TWO SAMPLES SIMPLY NOT SUFFICIENT !!!

How many pulses (samples) ?

If three (3) samples are made out of m(t), are they sufficient to completely represent m(t) ?

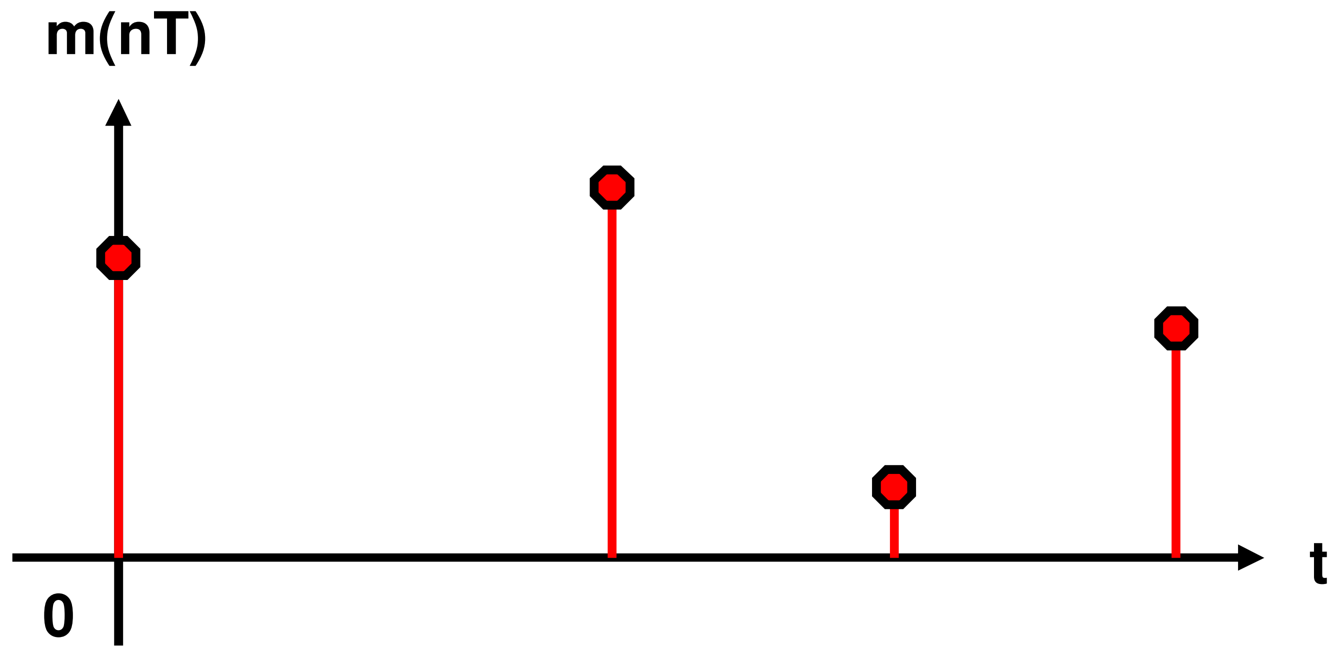
THREE SAMPLES SIMPLY NOT SUFFICIENT !!!



How many pulses (samples) ?

If four (4) samples are made out of m(t), are they sufficient to completely represent m(t) ?

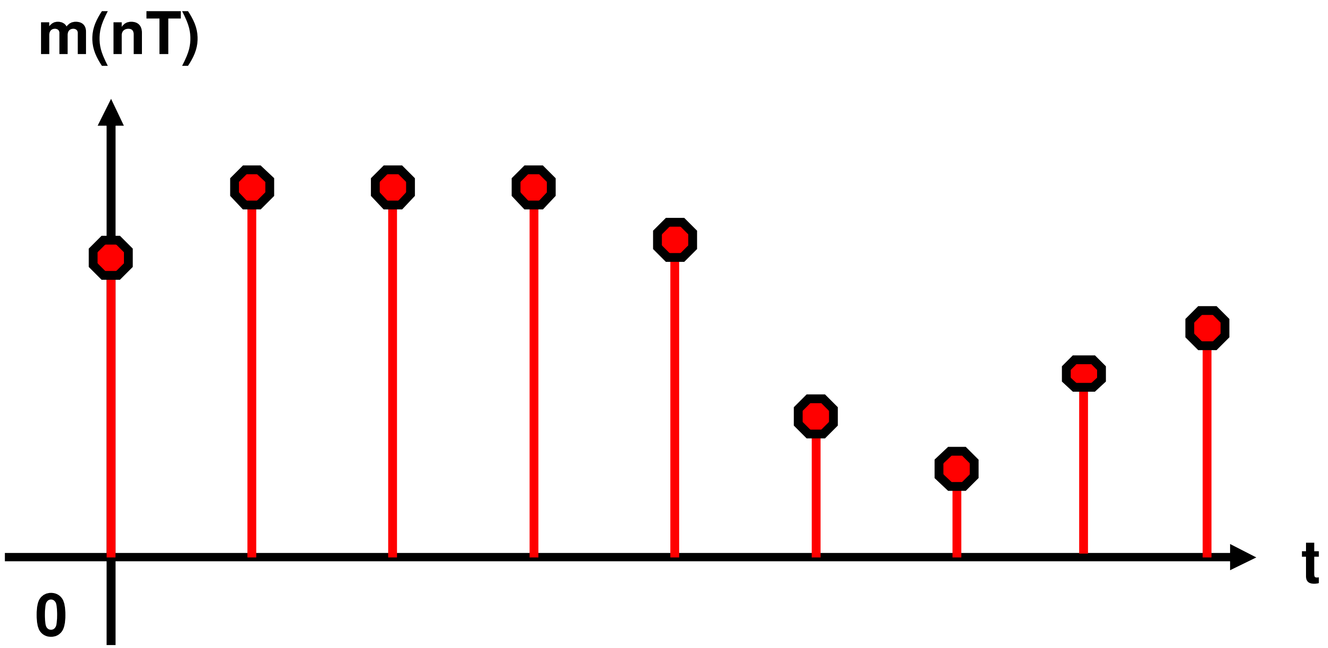
FOUR SAMPLES SIMPLY NOT SUFFICIENT !!!



How many pulses (samples) ?

If eight (8) samples are made out of m(t), are they sufficient to completely represent m(t) ?

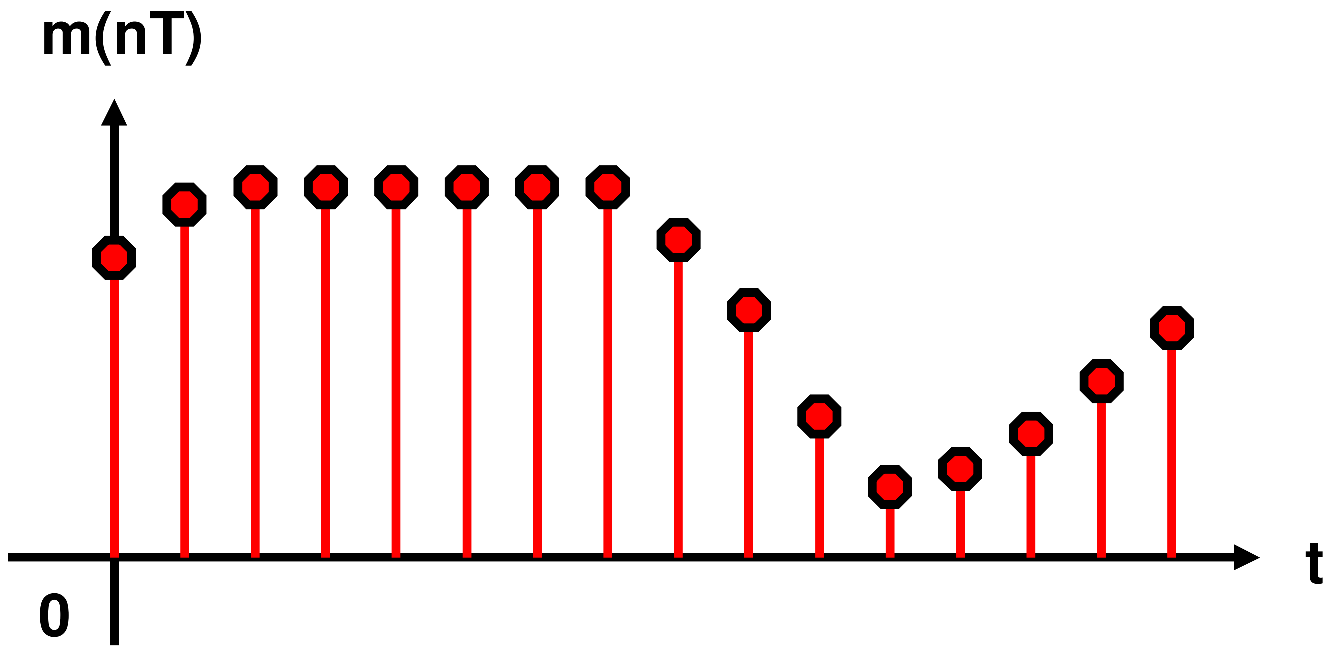
EIGHT SAMPLES ARE FINE – BUT RECOVERY ?



How many pulses (samples) ?

If sixteen (16) samples are made of m(t), are they sufficient to completely represent m(t) ?

THIS IS THE BEST SAMPLING PROCESS !!!



THE SAMPLING THEOREM

For analog input bandlimited modulating signal m(t) the sampling should be done in such a way that :-

- **The sampled version of m(t) should be represented completely in terms of its sampled output**
- **It should be possible to recover the original signal m(t) from its sampled version at the receiver**

Hence the frequency at which sampling should be done is given by the following equation :-

$$\boxed{f_s \geq 2f_m}$$

THE SAMPLING THEOREM

Nyquist – Shannon – Hartley sampling theorem states that the maximum sampling frequency (f_s) should be :-

$$\boxed{f_s \geq 2f_m}$$

Nyquist Rate is simply the sampling frequency itself & is described by the sampling theorem equation :-

$$\boxed{f_s \geq \frac{2}{T_m}}$$

where T_m is maximum time period of input modulating signal

THE SAMPLING THEOREM

Nyquist Interval refers to the total time interval between each successive pulse of the carrier & is given by :-

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

Above equations are necessary & sufficient for all the following conditions :-

- Sampled representation of input signal $v_m(t)$
- Recovery of $v_m(t)$ from it's sampled version

TYPES OF SAMPLING PROCESS

- **Impulse Sampling Process**
- **Natural Sampling Process**
- **Flat – Top Sampling Process**

Each of the above sampling processes or techniques depends on the shape or characteristics of sampled output signal amplitude

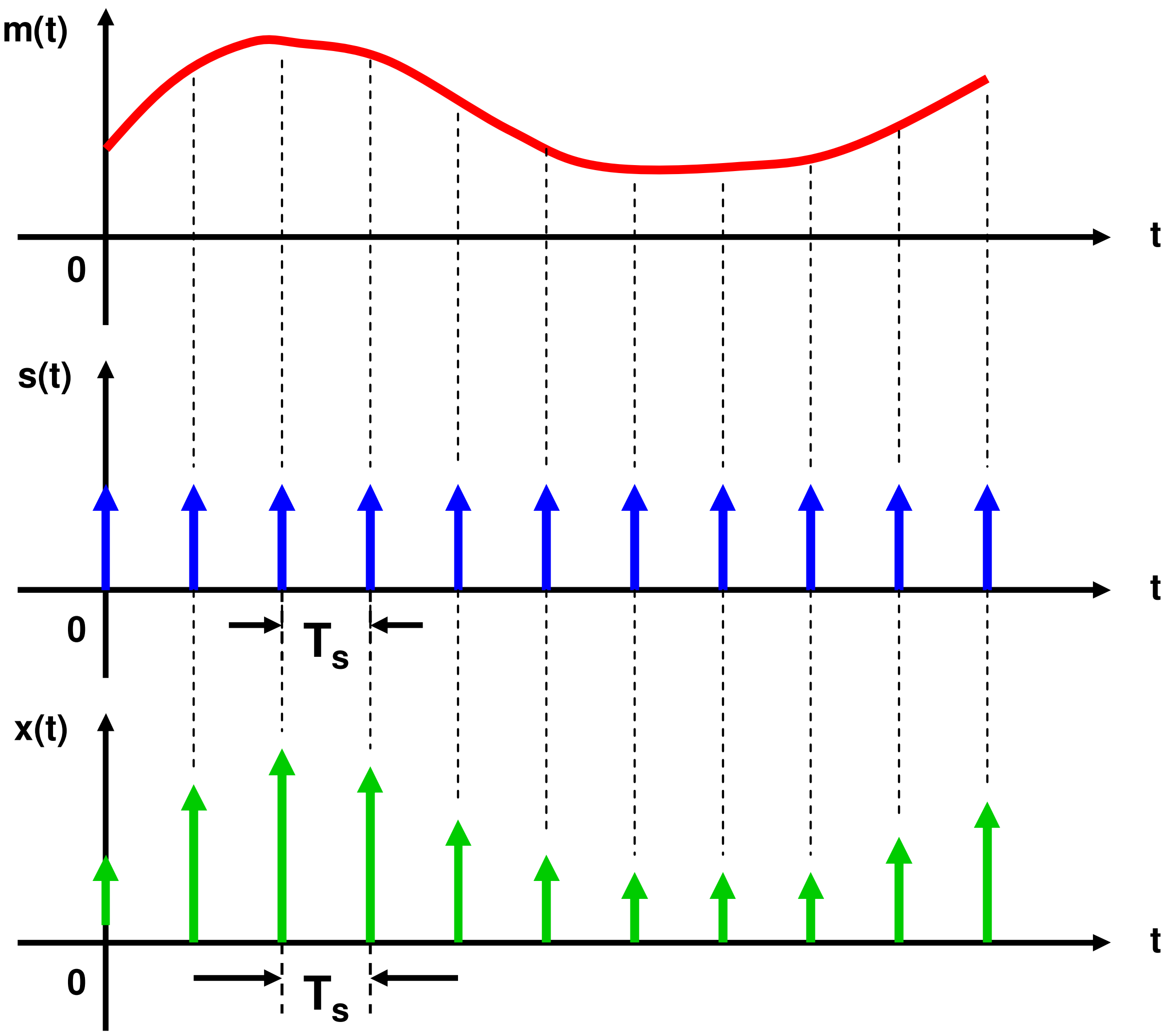
Natural Sampling & Flat – Top Sampling process are physically implementable

1. Impulse Sampling Process

Input Signal

Sampling Pulses

Sampled Signal



1. Impulse Sampling Process :- Advantages & Disadvantages

ADVANTAGES :-

1. Mathematically used to prove the sampling theorem
2. Easiest to generate using a multiplier (ideally)
3. Each impulse of extremely short duration (zero)

DISADVANTAGES :-

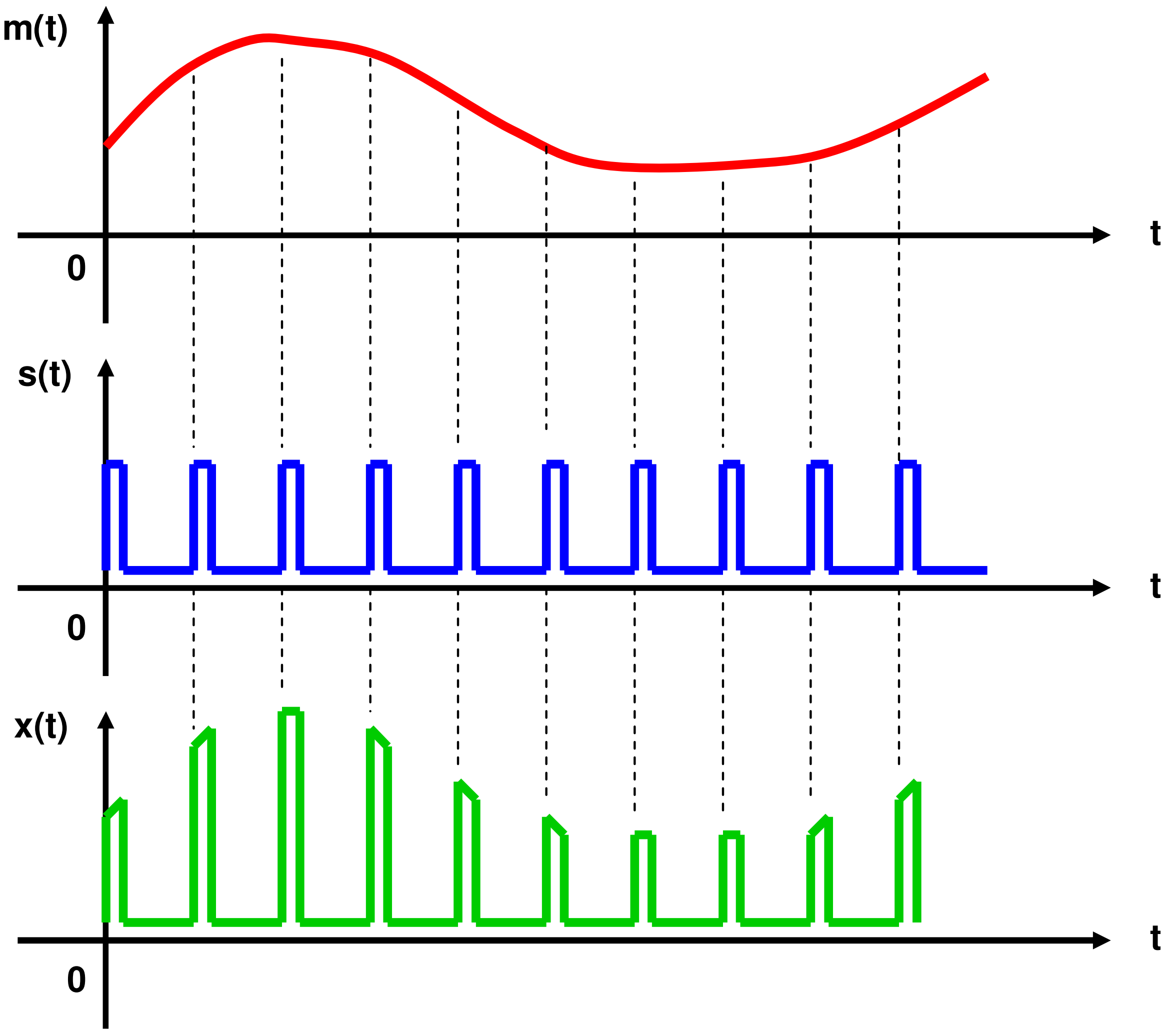
1. Extremely small transmitted power present
2. Very poor signal to noise ratio (SNR)
3. Impossible to generate such narrow width pulses
4. May get lost in noise due to such narrow width
5. Recovery becomes difficult for such conditions

2. Natural Sampling Process

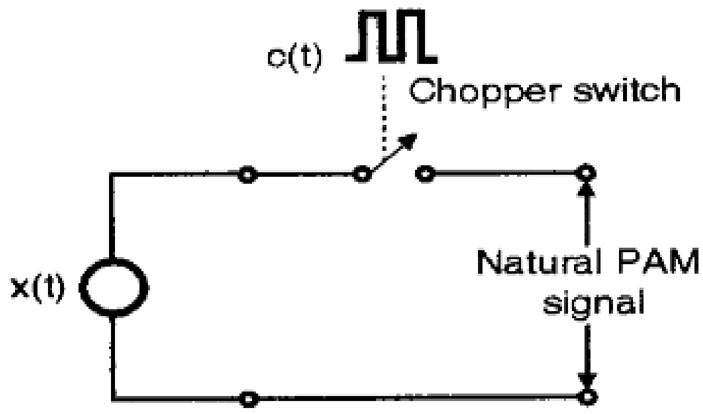
Input Signal

Sampling Pulses

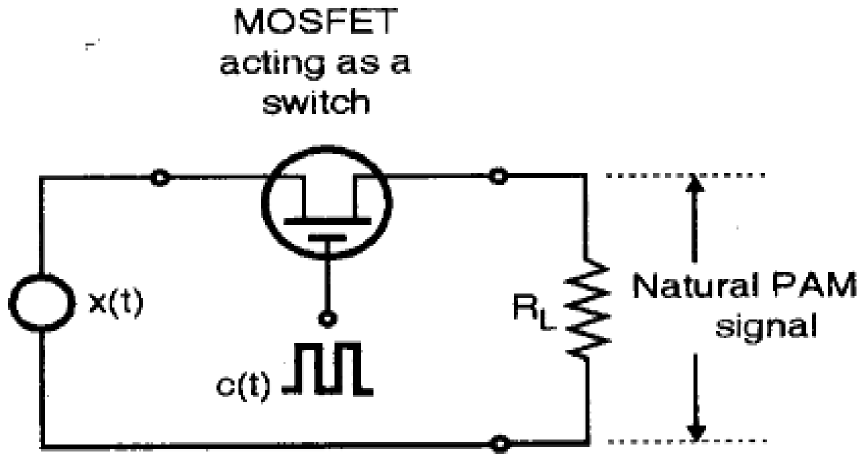
Sampled Signal



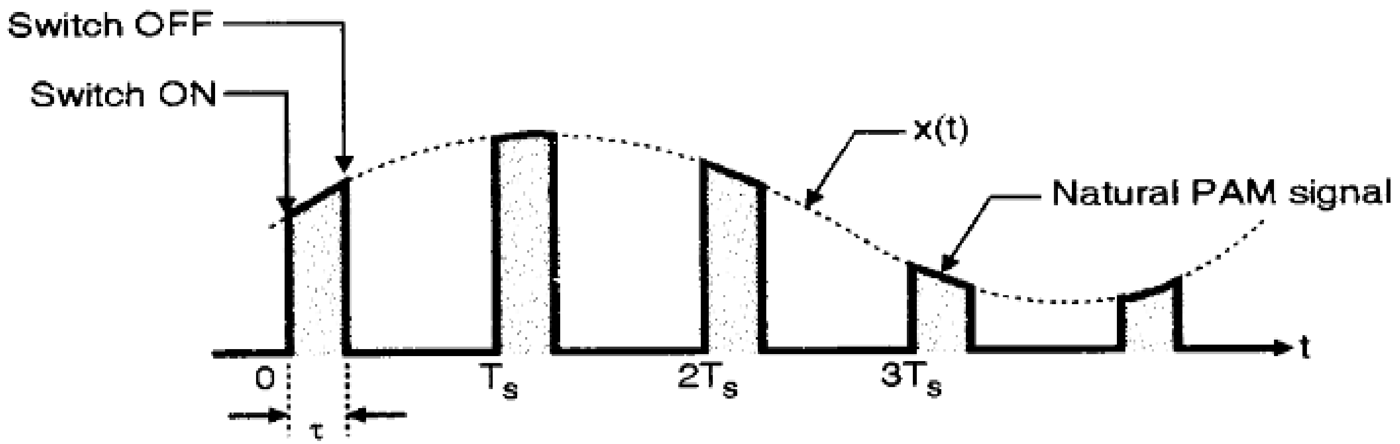
2. Natural Sampling Process :- Circuit Diagram for Generation



(a) Natural PAM



(b) Natural PAM with a MOSFET acting as a switch



2. Natural Sampling Process :- Circuit Diagram for Generation

- Main principle behind natural sampling is to perform the ‘chopping’ of input signal
- This operation is technically called sampling using a high frequency ON – OFF switch
- MOSFET works as a high frequency switch, sampling $v_m(t)$ at certain instants of time
- The time between successive or consecutive samples in the sampling time ‘Ts’
- When ON, part of the signal directly appears at the output, when OFF there’s no signal

2. Natural Sampling Process :- Advantages & Disadvantages

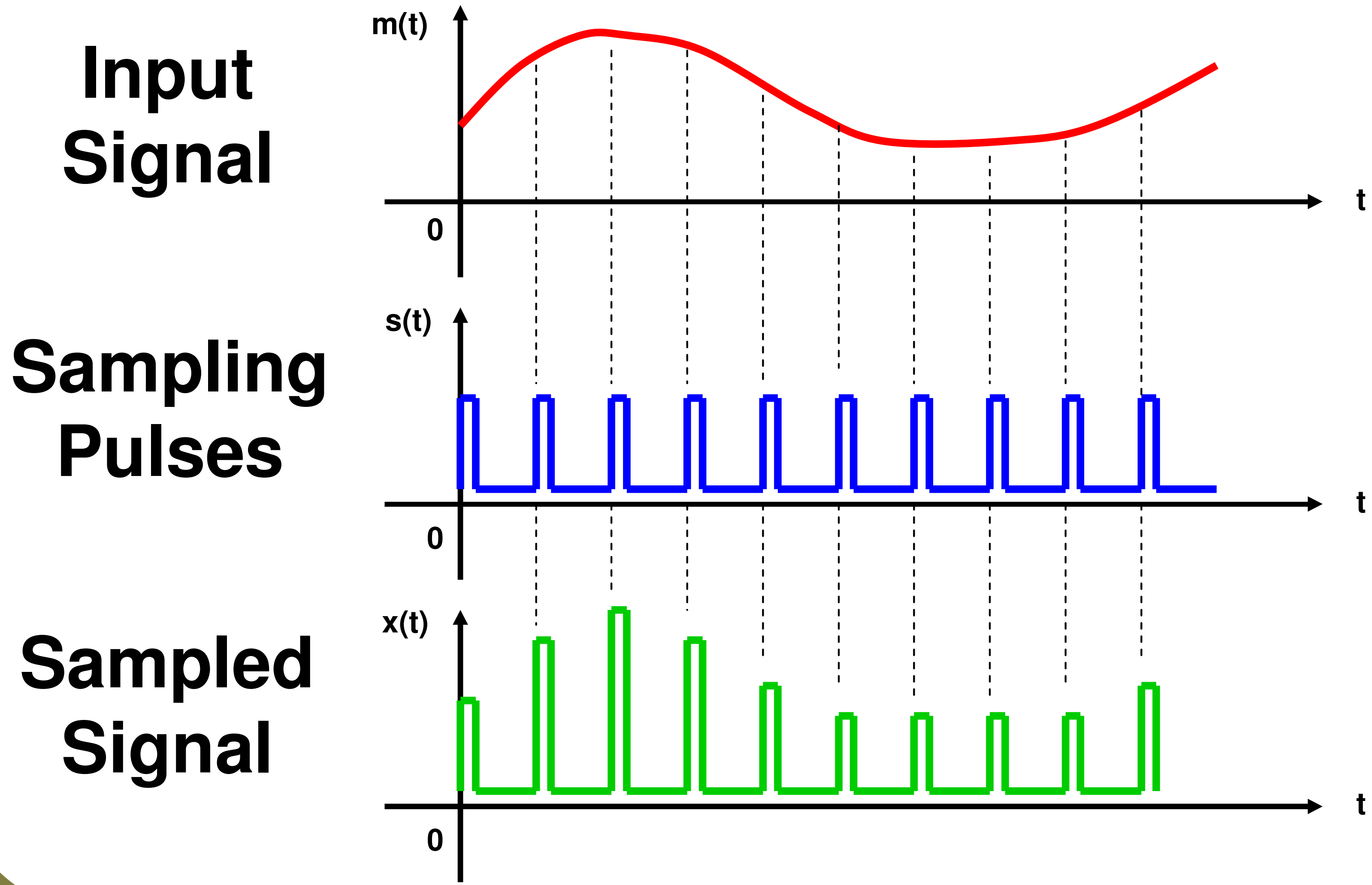
ADVANTAGES :-

1. Easier to generate than ideal sampling impulses
2. Each of the pulses have a finite pulse width
3. Good signal to noise ratio (SNR) than impulses
4. Low effect of noise compared to the impulses
5. Recovery is easy using a low pass filter (LPF)
6. Simple & easy practical circuit construction

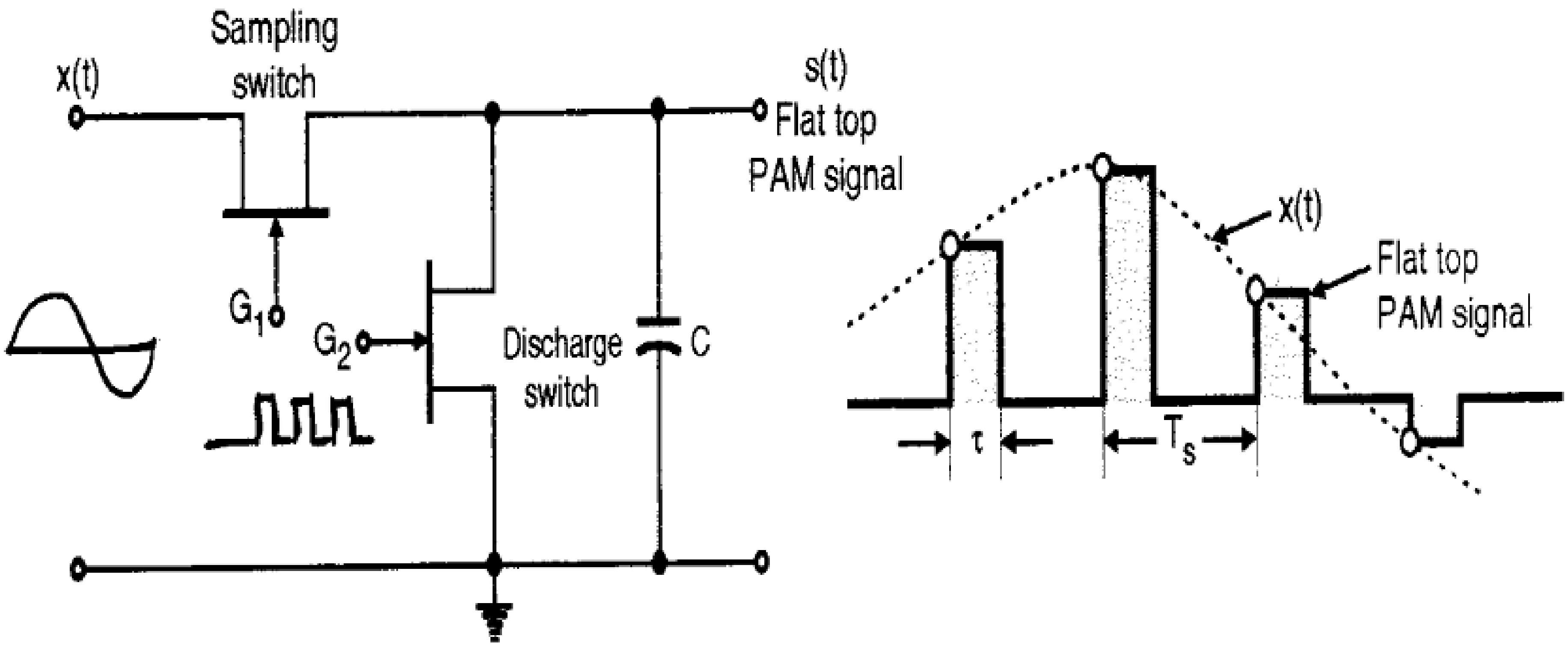
DISADVANTAGES :-

1. Aperture effect due to finite pulse width (distortion)
2. Possibility of crosstalk due to large pulse width
3. Some distortion is introduced as the high frequency components are decreased (attenuated)

3. Flat – Top Sampling Process



3. Flat – Top Sampling Process :- Circuit Diagram for Generation



please refer class notes for the complete theoritical description

3. Flat – Top Sampling Process :- Circuit Diagram for Generation

- Here the output pulses have a flat – shaped tops against natural sampling techniques
- MOSFET works as a high frequency switch, sampling $v_m(t)$ at certain instants of time
- These sampled values of $v_m(t)$ are stored by the capacitor 'C' (later on discharged)
- Each sample is held to a constant value by capacitor 'C' leading to flat – top pulses
- Before the next sample is stored, previously stored samples have to be discharged

3. The Flat – Top Sampling Process :- Advantages & Disadvantages

ADVANTAGES :-

1. Easier to generate than ideal sampling impulses
2. Each of the pulses have a finite pulse width
3. Good signal to noise ratio (SNR) than impulses
4. Low effect of noise compared to the impulses
5. Recovery is easy using a low pass filter (LPF)
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DISADVANTAGES :-

1. Aperture effect due to finite pulse width (distortion)
2. Possibility of crosstalk due to large pulse width
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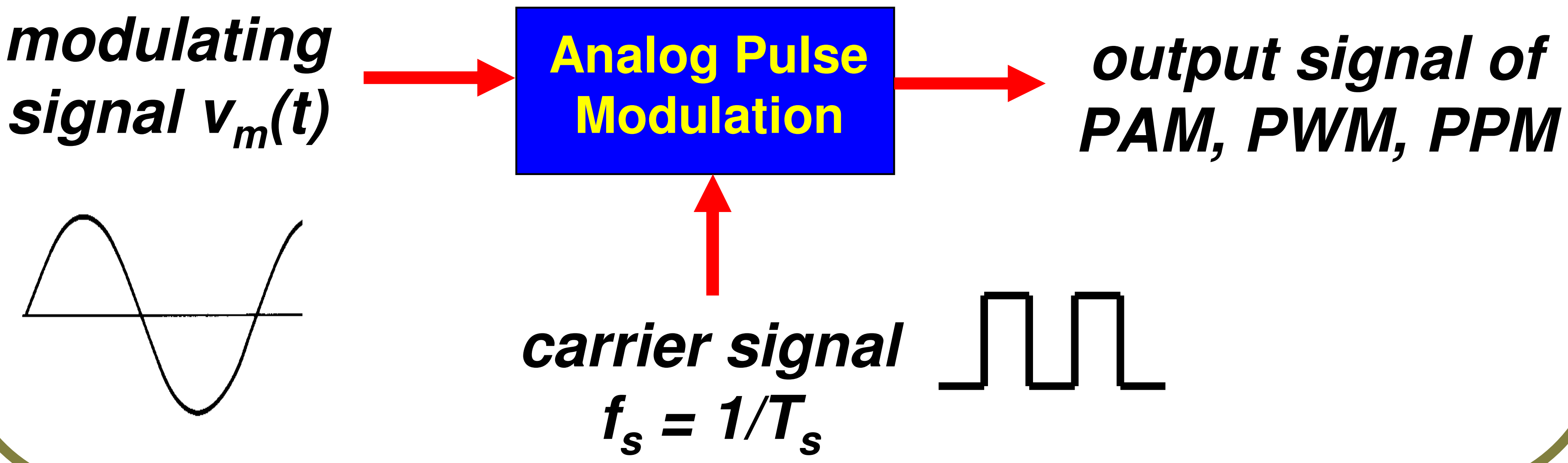
Comparison of Ideal, Natural & Flat – Top Sampling Techniques

- **Nature of samples**
- **Circuit Description**
- **Practical Realization**
- **Circuit Diagram**
- **Nature of Waveforms**
- **Sampling Rate**
- **Signal Power**
- **Bandwidth Needed**
- **Effect of Noise**

refer class notes for comparison

Techniques of PAM, PWM, PPM

- **AF modulating (baseband) signal is given by $v_m(t) = V_m \sin\omega_m t$ OR $v_m(t) = V_m \cos\omega_m t$**
- **HF carrier signal is a train of pulses having a frequency of $f_s = 1/T_s$**



Techniques of PAM, PWM, PPM

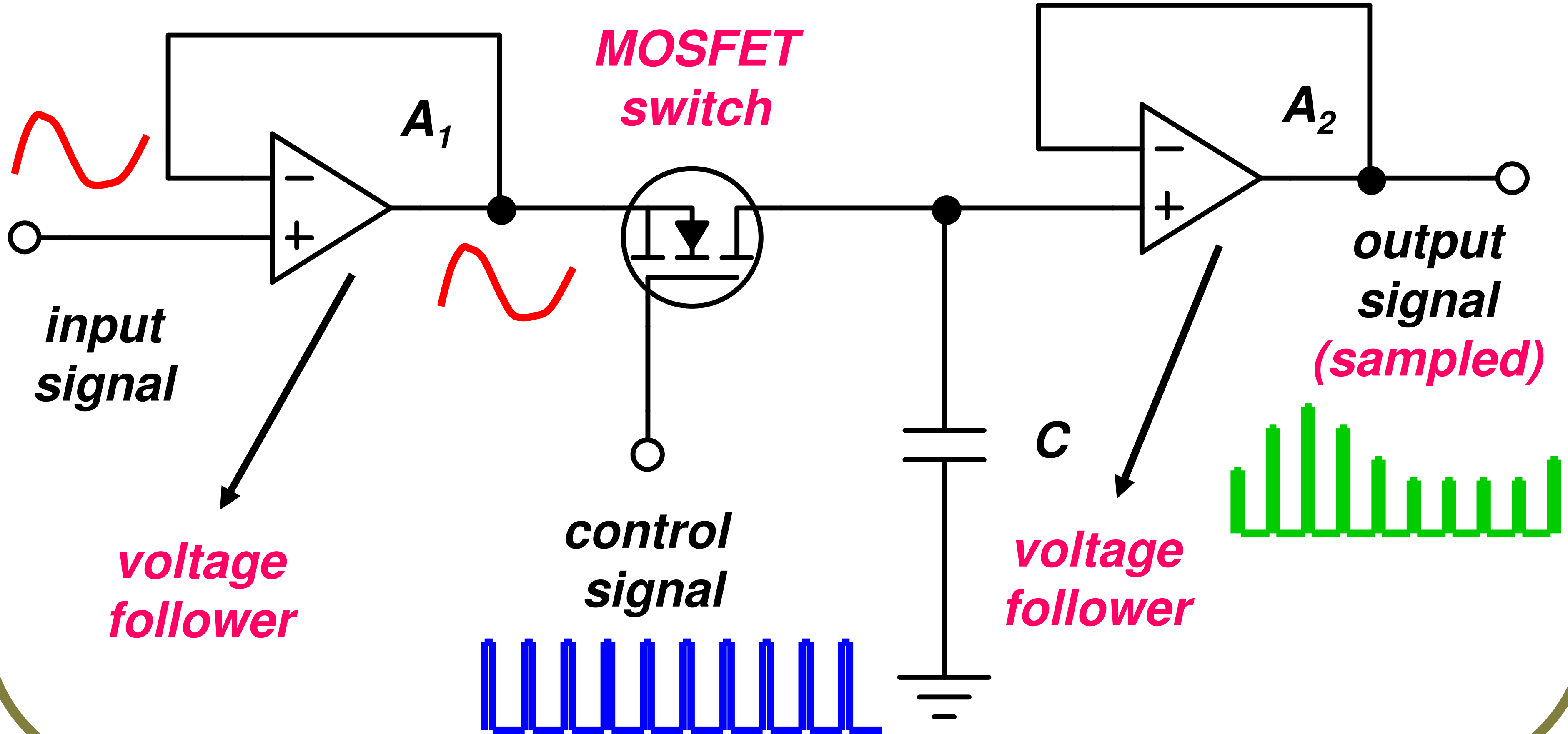
Here the carrier signal is a train of pulses & is ($f_s = 1/T_s$) modulated by $v_m(t)$ as follows :-

- **Pulse Amplitude Modulation (PAM)** if carrier pulses amplitude varies with $v_m(t)$
- **Pulse Width Modulation (PWM)** if the carrier pulses time period (T_s) varies with $v_m(t)$
- **Pulse Position Modulation (PPM)** if carrier pulses position varies with $v_m(t)$

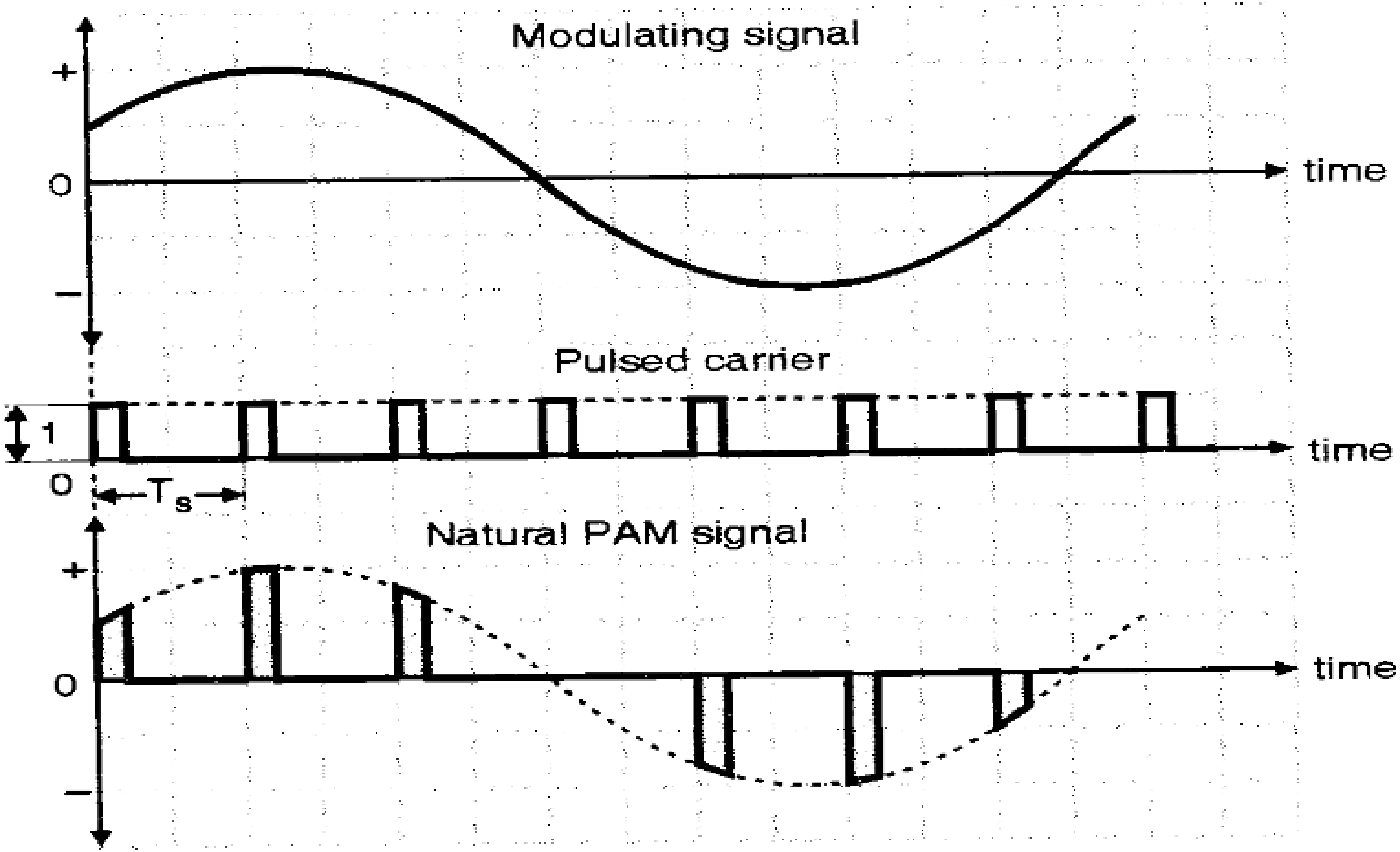
In each case either amplitude, width or position (any two) will remain constant or unchanged

1. Pulse Amplitude Modulation

(a) Generation/Modulation :-



1. Pulse Amplitude Modulation



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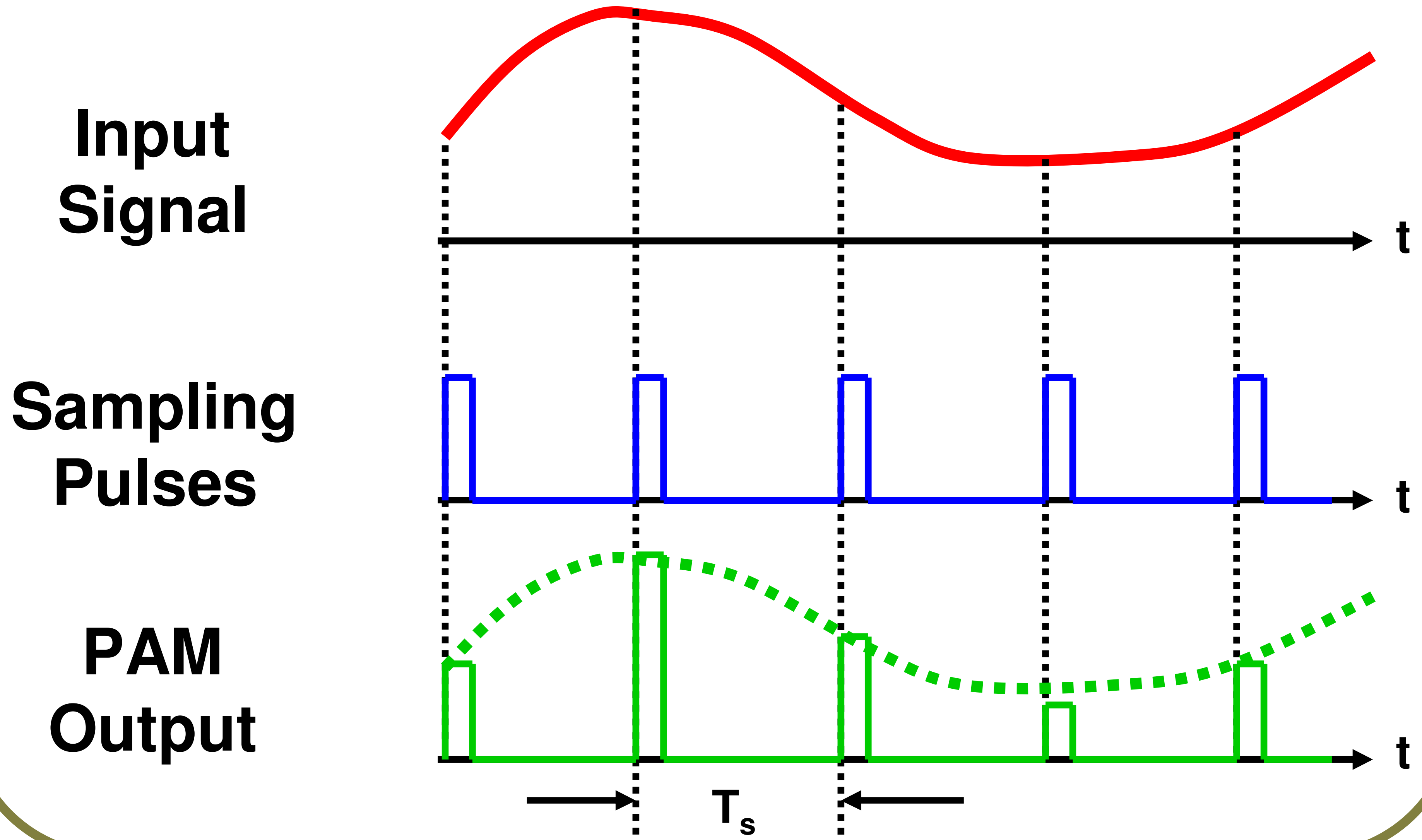
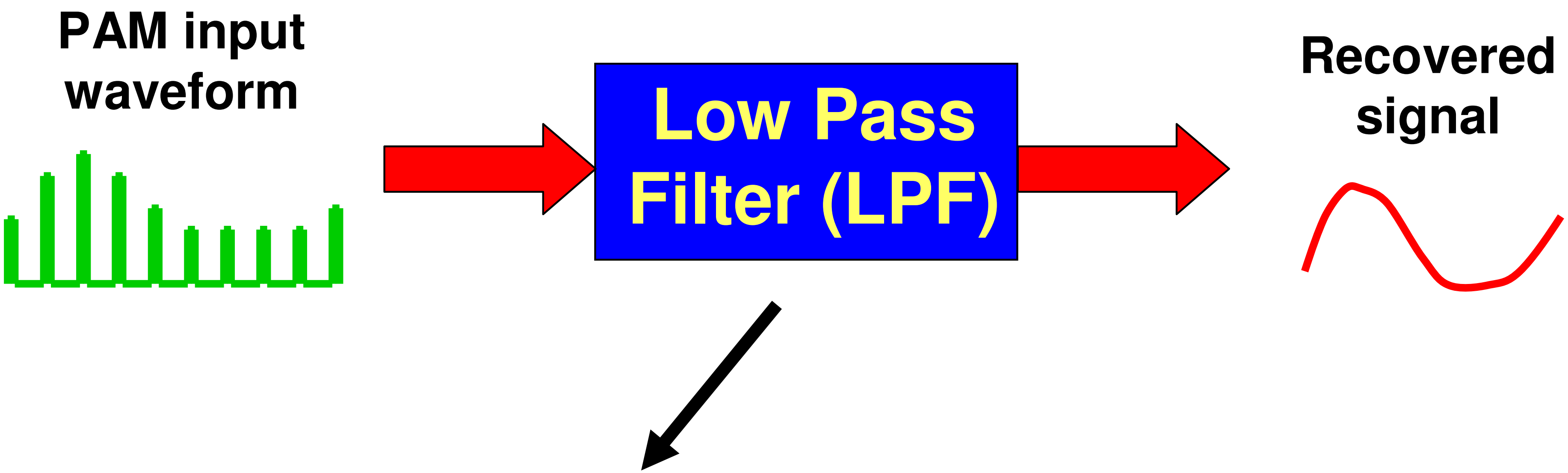


1. Pulse Amplitude Modulation (PAM) – (a) Generation/Modulation

- Op – Amps A_1 & A_2 work as the unity gain voltage followers at the input & output
- MOSFET works as a high frequency switch, sampling $v_m(t)$ at certain instants of time
- These sampled values of $v_m(t)$ are stored by the capacitor ‘C’ (later on discharged)
- Each sample is held to a constant value by capacitor ‘C’ leading to flat – top pulses
- Before the next sample is stored, previously stored samples have to be discharged

1. Pulse Amplitude Modulation

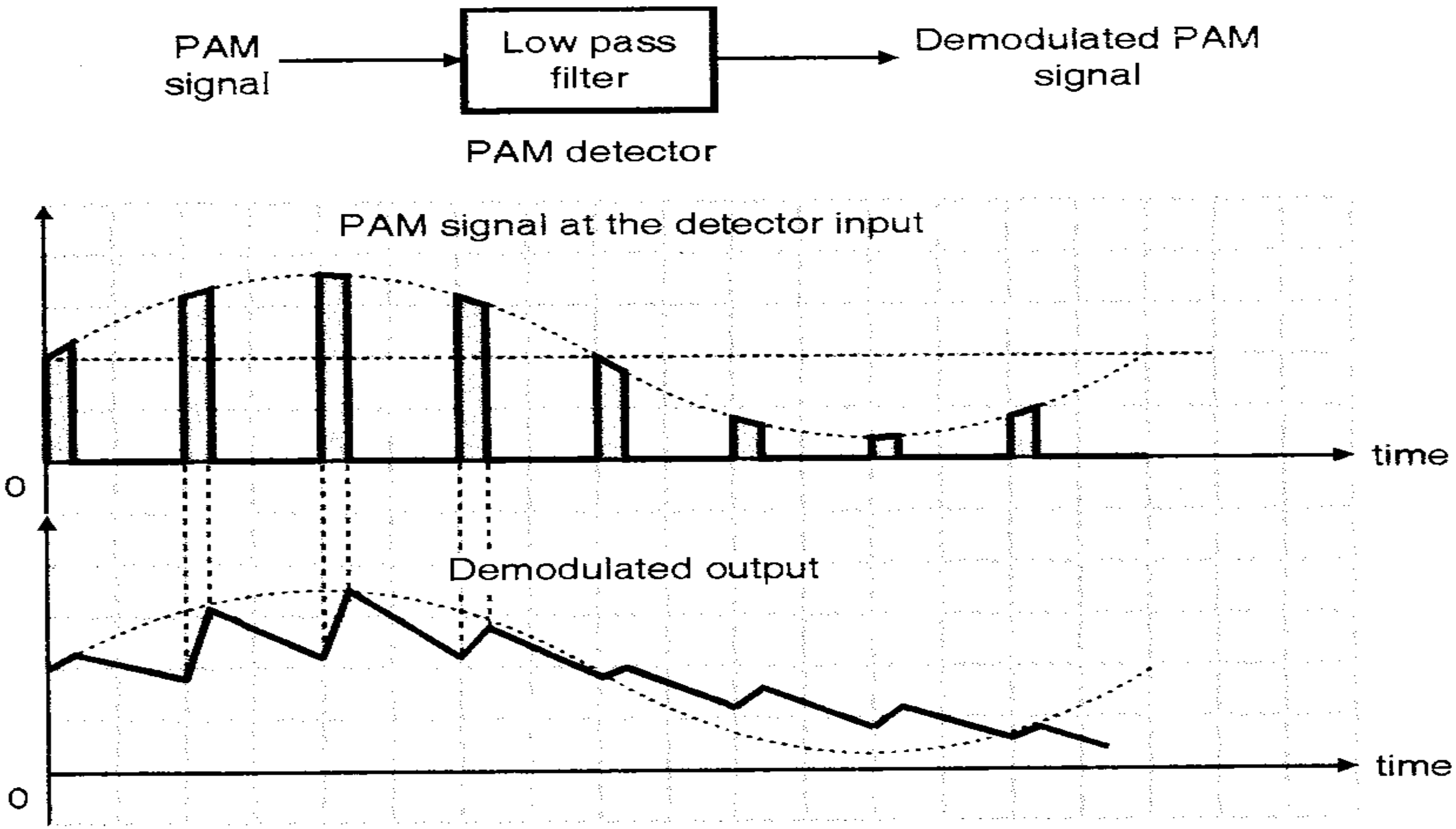
(b) Demodulation/Detection :-



LPF tuned to single frequency of f_m Hz, which is that of the input modulating signal $v_m(t)$

1. Pulse Amplitude Modulation

(b) Demodulation/Detection :-



1. Pulse Amplitude Modulation (PAM) – (b) Detector/Demodulator

- PAM uses non – coherent demodulation for the recovery of the original signal $v_m(t)$
- Receiver consists of a basic low pass filter (LPF) to recover back original signal $v_m(t)$
- Each input pulse charges the capacitor (C) as it appears at the input of the LPF
- Thus the low pass filter (LPF) will remove all the transients in input PAM waveform
- LPF cut – off frequency chosen as frequency of original modulating signal $v_m(t)$

1. Pulse Amplitude Modulation (PAM) – (c) Advantages & Disadvantages

ADVANTAGES :-

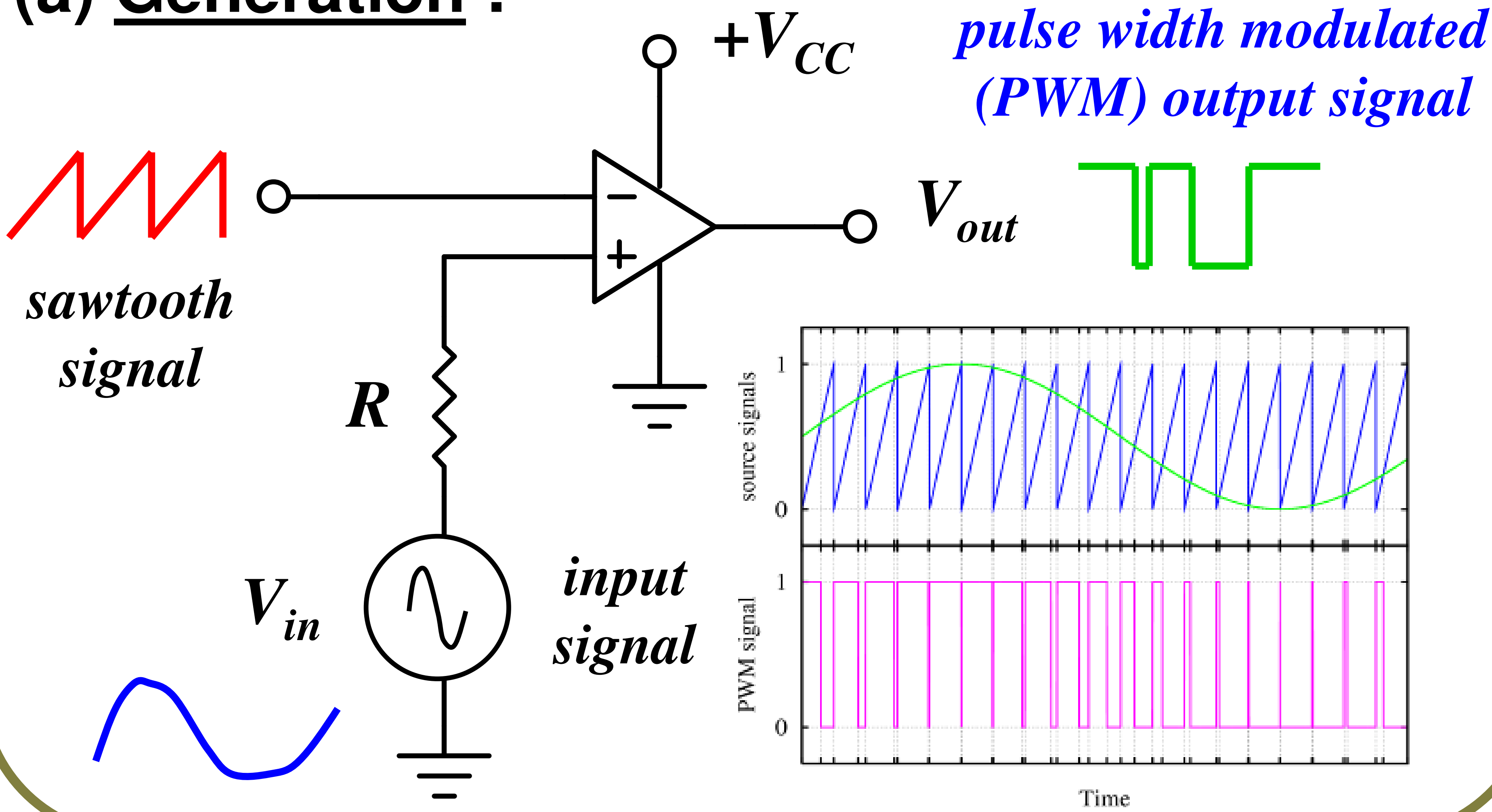
1. Extremely simple & easy hardware configuration for modulation & demodulation against PWM & PPM
2. No synchronization required in PAM as compared to both, PWM & PPM (non – coherent demodulation)
3. Bandwidth requirements smaller than PWM & PPM

DISADVANTAGES :-

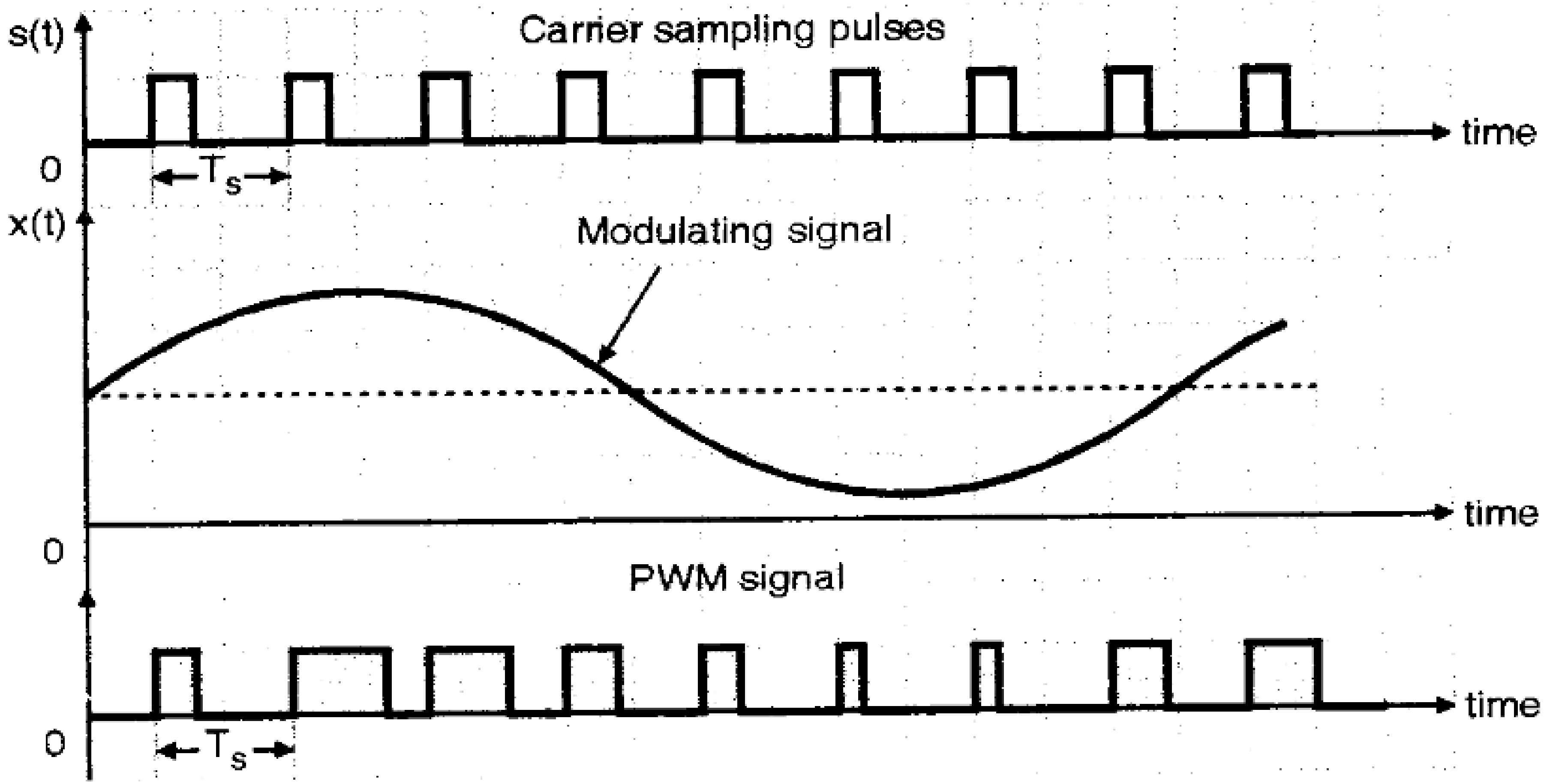
1. Extremely poor noise immunity v/s PWM & PPM
2. High power requirement than PPM since amplitude of each pulse changes constantly at all times

2. Pulse Width Modulation

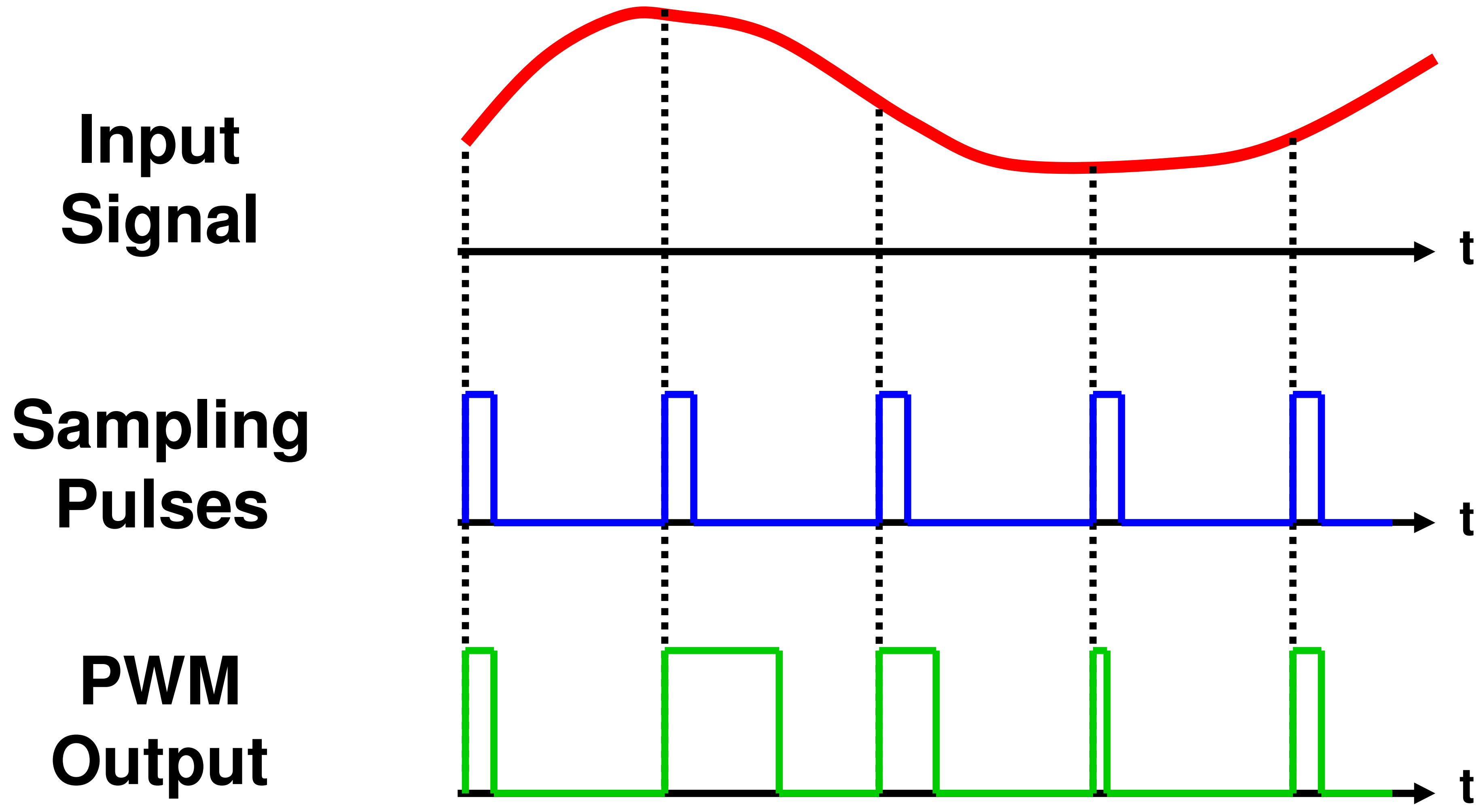
(a) Generation :-



2. Pulse Width Modulation

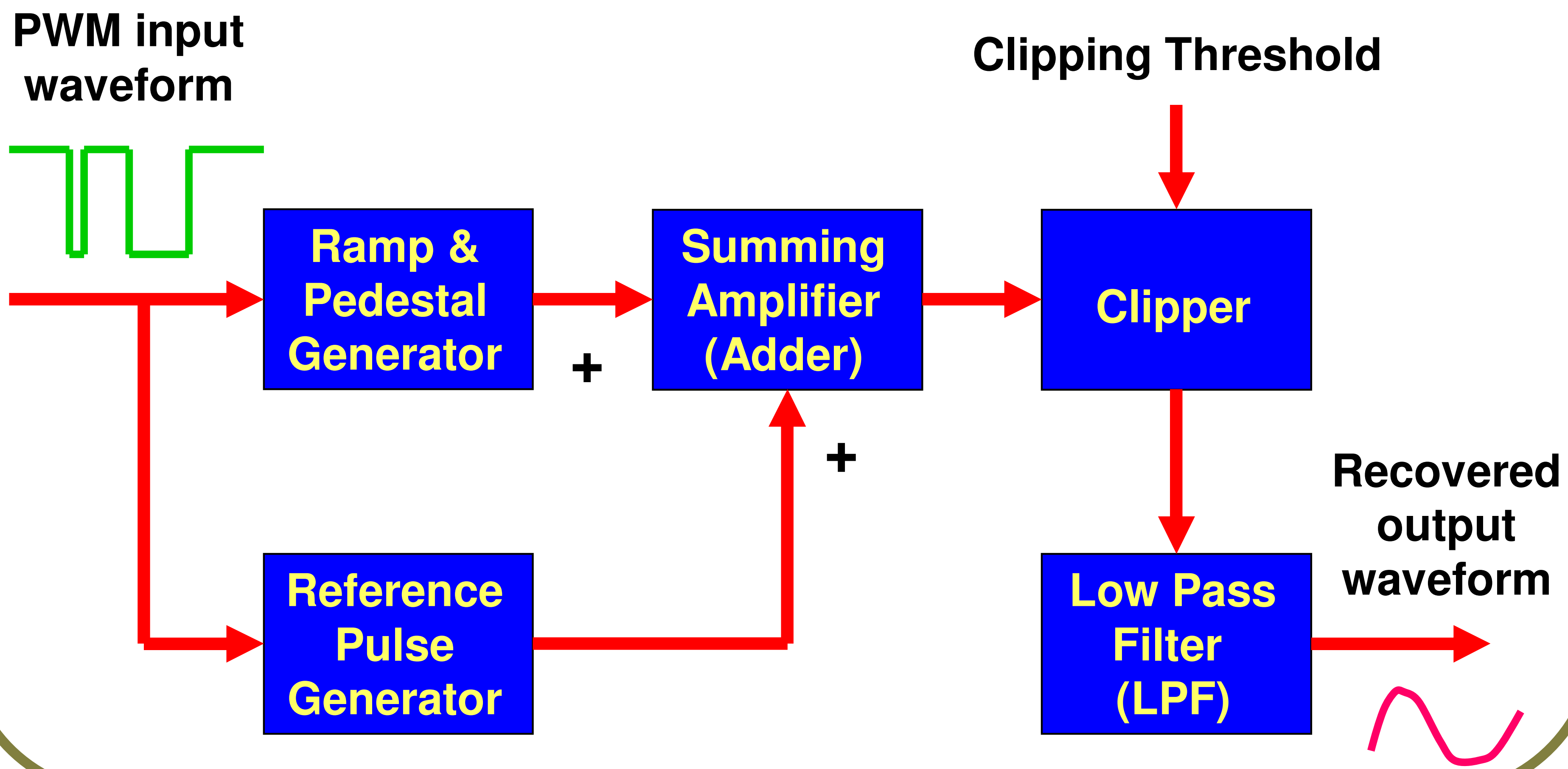


2. Pulse Width Modulation



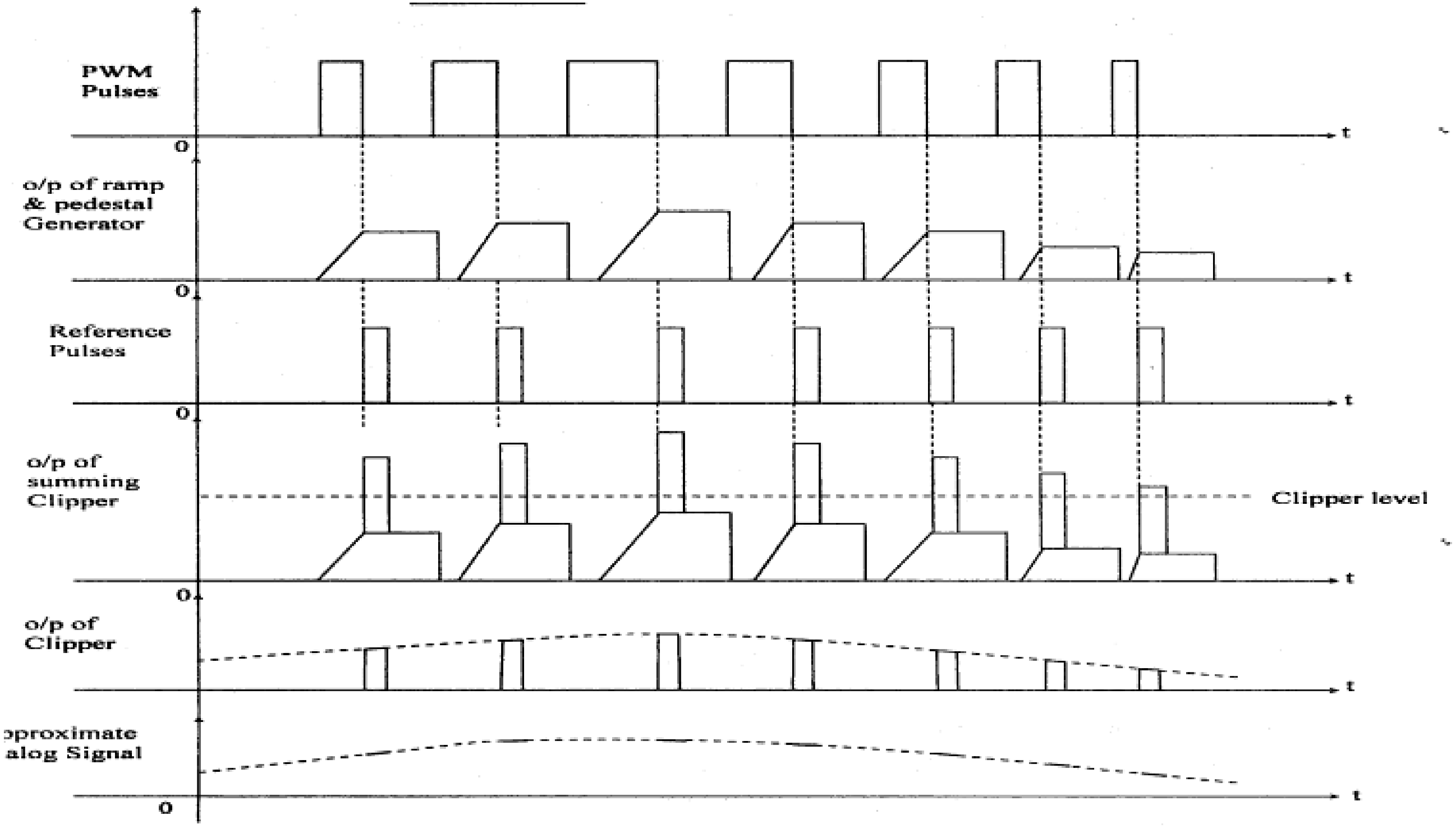
2. Pulse Width Modulation

(b) Demodulation/Detection :-



2. Pulse Width Modulation

(b) Demodulation/Detection :-



2. Pulse Width Modulation (PWM) – (b) Detection/Demodulation

- Received PWM pulses given to the ramp & pedestal generator, giving a slope (ramp)
- Ramp slope depends on a pulse width, while pedestal is at fixed time duration (width)
- This is added with reference pulses giving o/p pulses of varying amplitude, as shown
- By selecting a threshold level, clipper can remove half portion of the waveform
- PWM is thus converted into PAM, which will be demodulated using low pass filter (LPF)

2. Pulse Width Modulation (PWM) – (c) Advantages & Disadvantages

ADVANTAGES :-

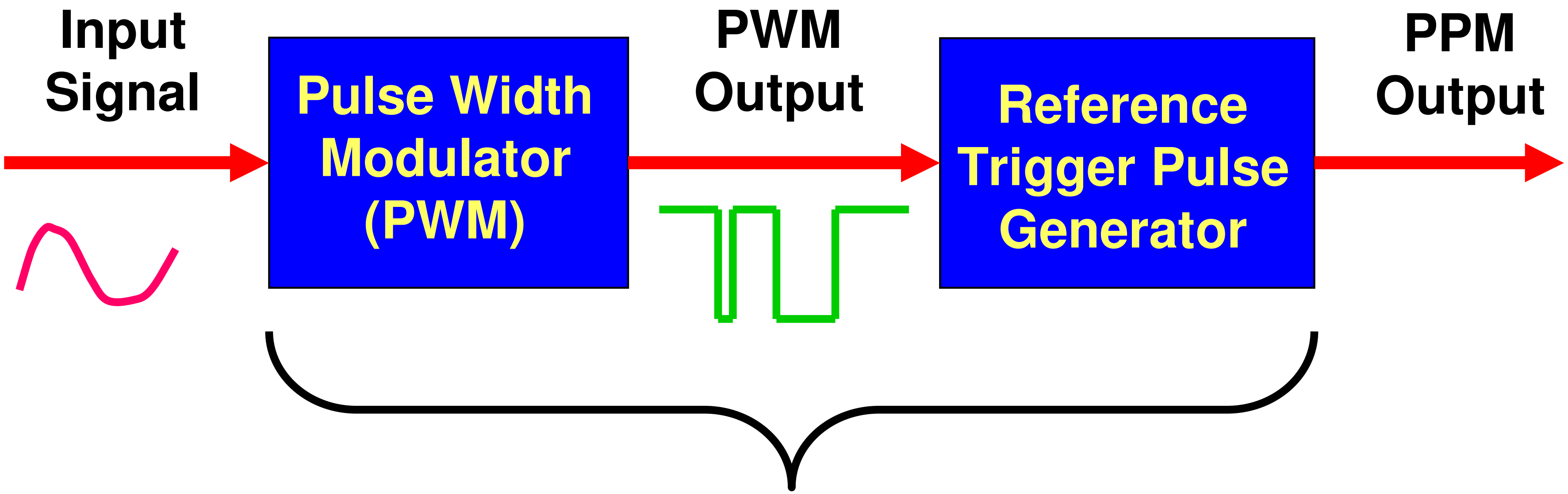
1. Good noise immunity compared to PAM
2. No synchronization required as compared to PPM
3. Complexity in modulation & demodulation is far less as compared to PPM

DISADVANTAGES :-

1. Coherent demodulation technique is required
2. High bandwidth requirement as compared to PAM
3. High power requirement than PAM (variable width)
4. Complexity in generation & detection (indirect) as compared to PAM

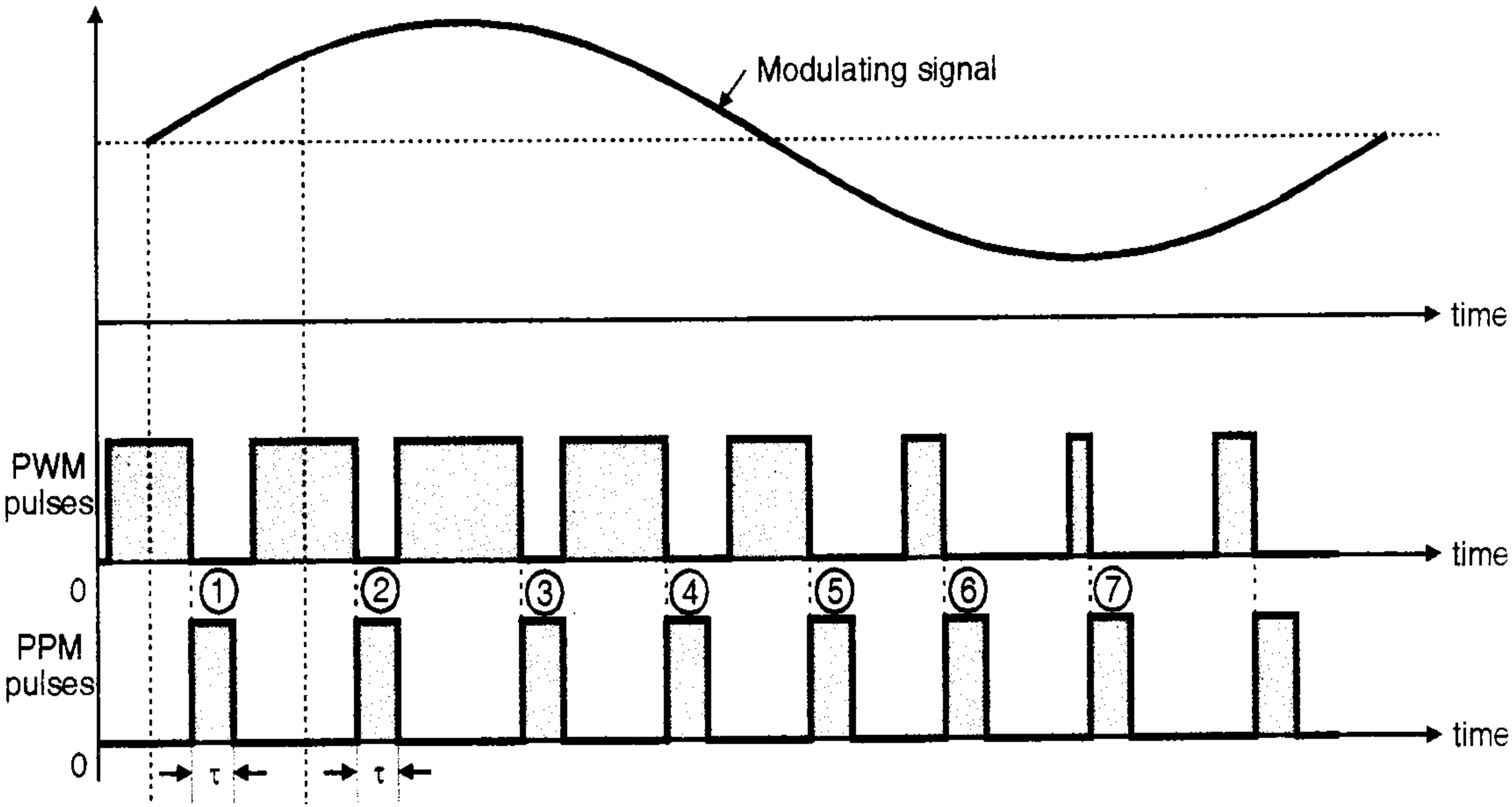
3. Pulse Position Modulation

(a) Generation/Modulation :-

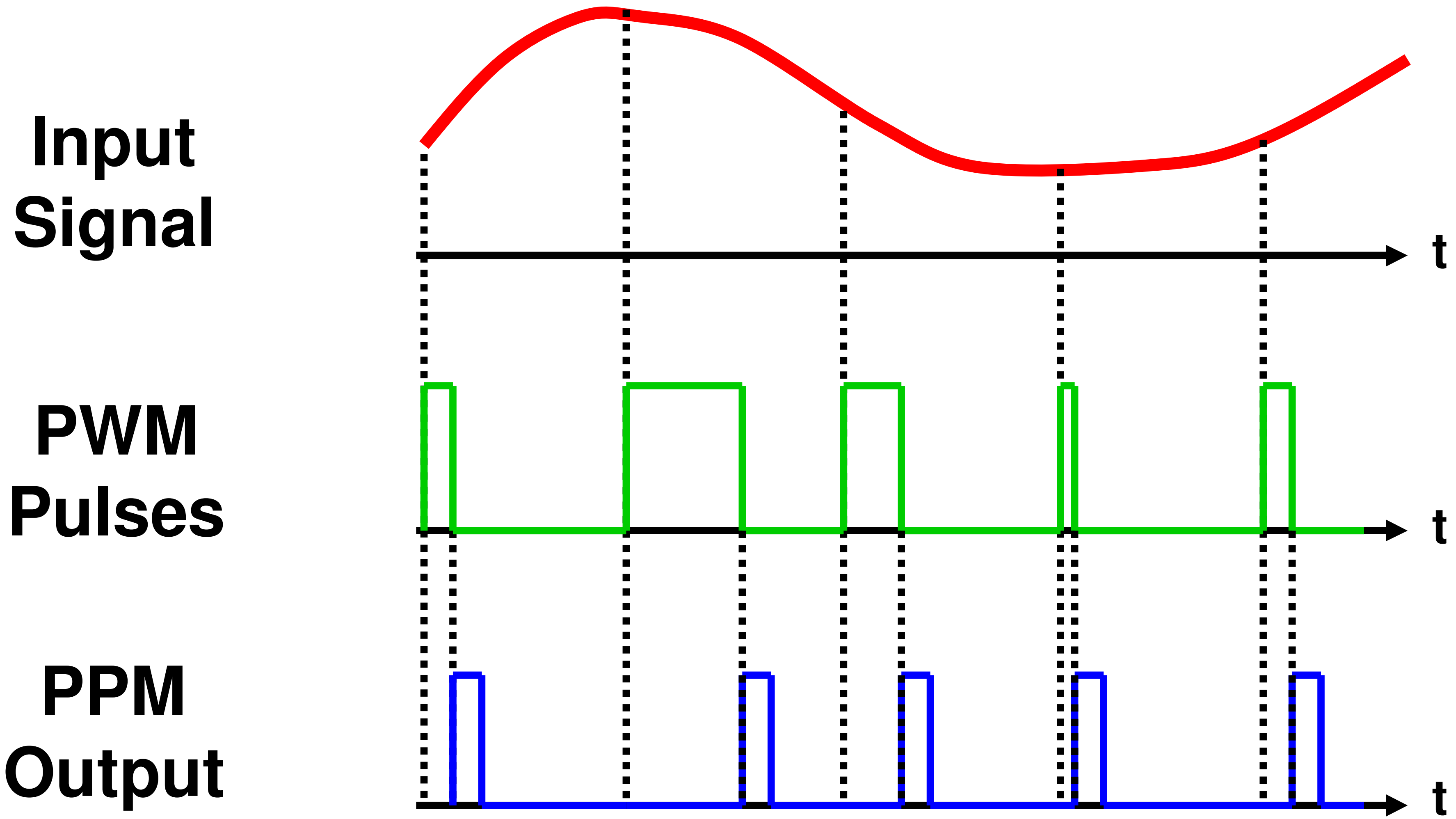


An indirect generation process which first requires PWM & then conversion to PPM

3. Pulse Position Modulation

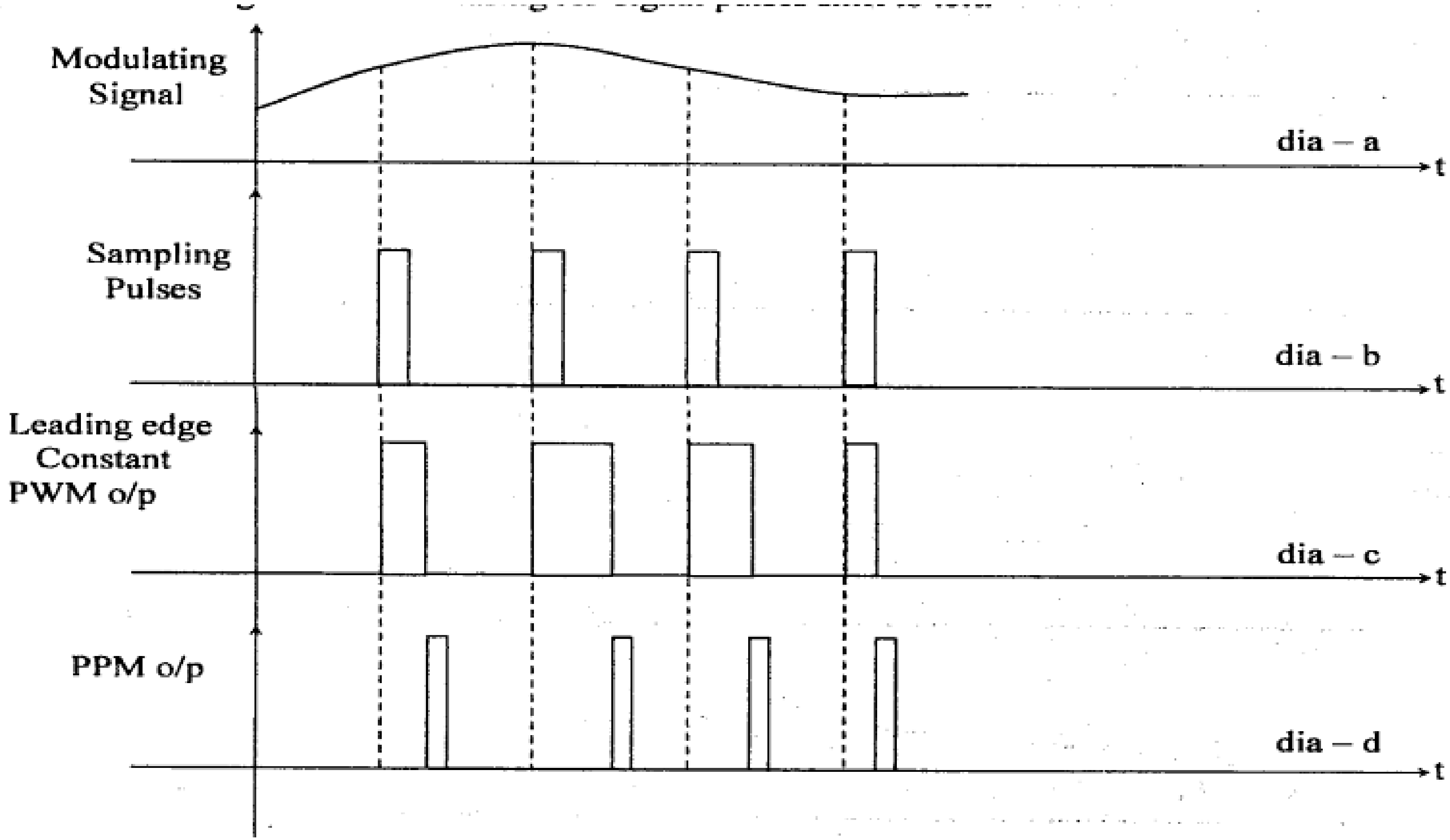


3. Pulse Position Modulation



3. Pulse Position Modulation

(a) Generation/Modulation :-

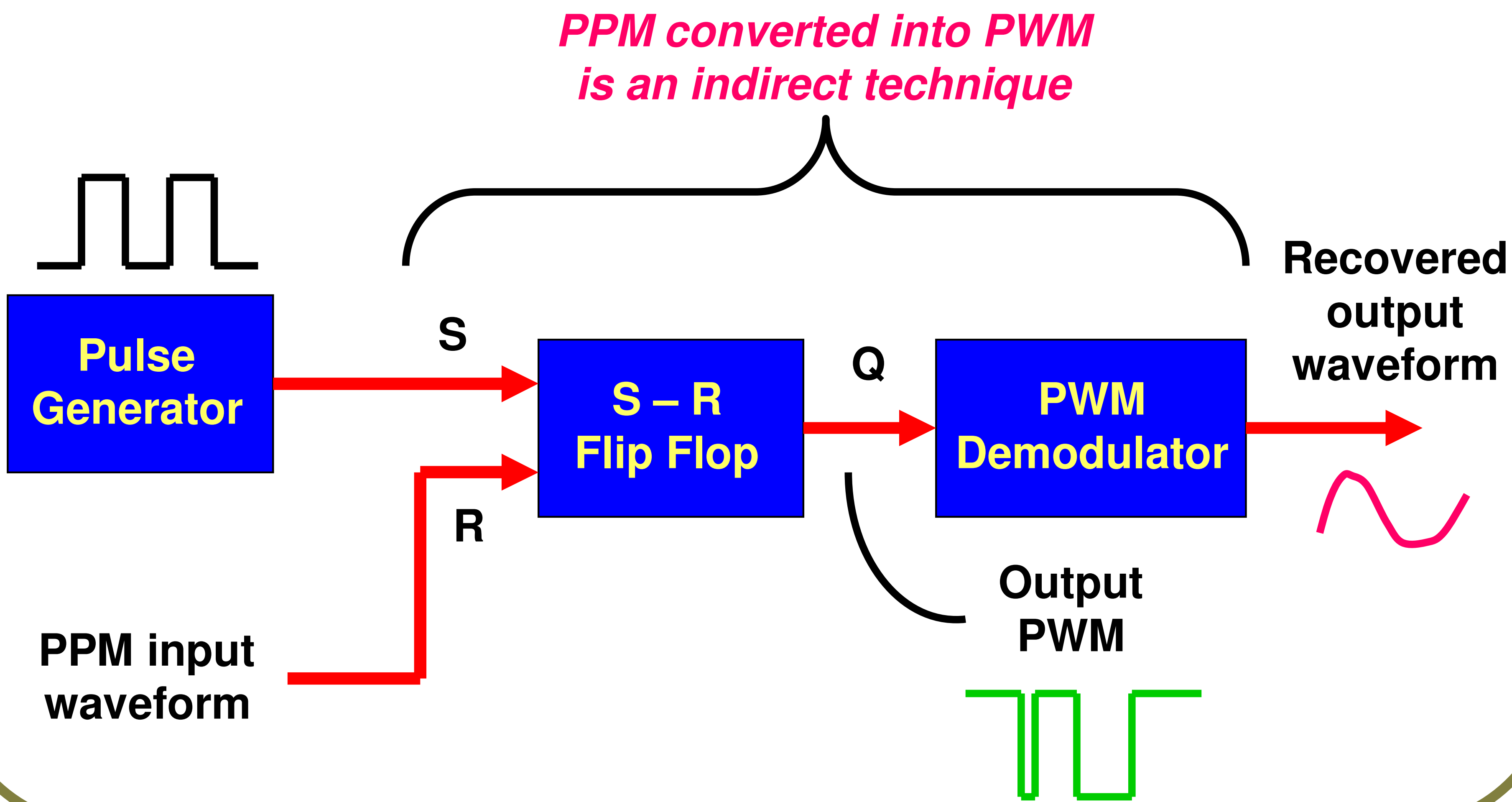


3. Pulse Position Modulation (PPM) – (a) Generation/Modulation

- Input modulating signal, is initially given to the pulse width modulator (PWM) circuit
- These output pulse width modulated pulses trigger the monostable multivibrator
- It goes high for a pre-determined amount of time & then comes back to the low state
- With respect to PWM pulses, output of PPM pulses are shifted back & forth, as shown
- Hence pulse position modulated (PPM) pulse is generated from the input PWM waveform

3. Pulse Position Modulation

(b) Demodulation/Detection :-



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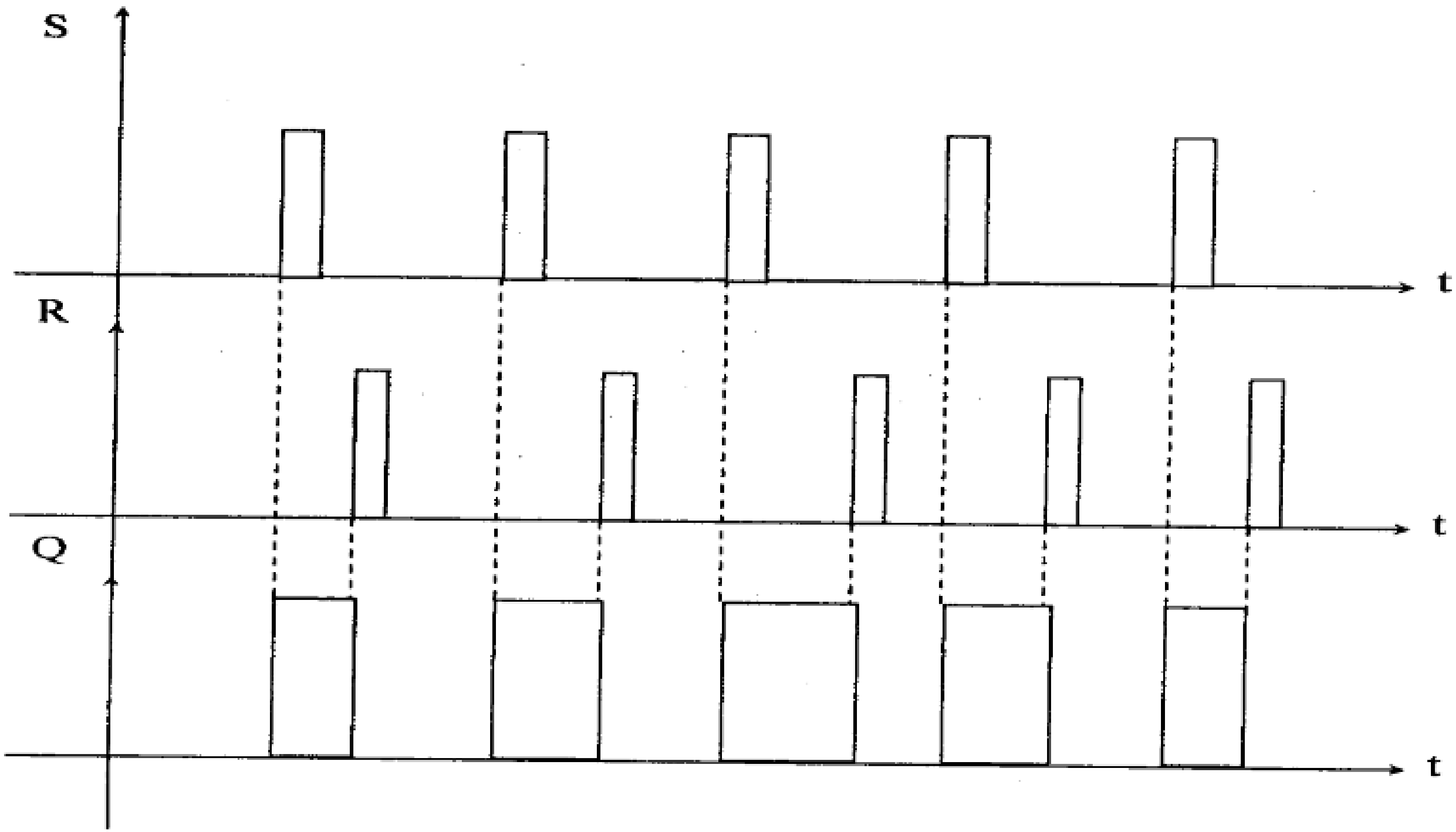
3. Pulse Position Modulation

(b) Demodulation/Detection :-



3. Pulse Position Modulation (PPM) – (b) Detection/Demodulation

- **PPM pulses can be demodulated only when demodulator is synchronized with receiver**
- **Pulse generator generates constant train of pulses for ‘S’ input (set) of SR flip-flop**
- **Incoming PPM pulses are then applied to the ‘R’ input (reset) of the SR flip-flop**
- **Based on time between pulses received at S & R inputs, output ‘Q’ will be high**
- **Input PPM waveform has been changed into PWM waveform, which is demodulated**

3. Pulse Position Modulation (PPM) – (c) Advantages & Disadvantages

ADVANTAGES :-

1. Excellent noise immunity compared to PAM & PWM
2. Due to constant amplitude & pulse width, transmitted power remains constant, compared to PAM & PWM

DISADVANTAGES :-

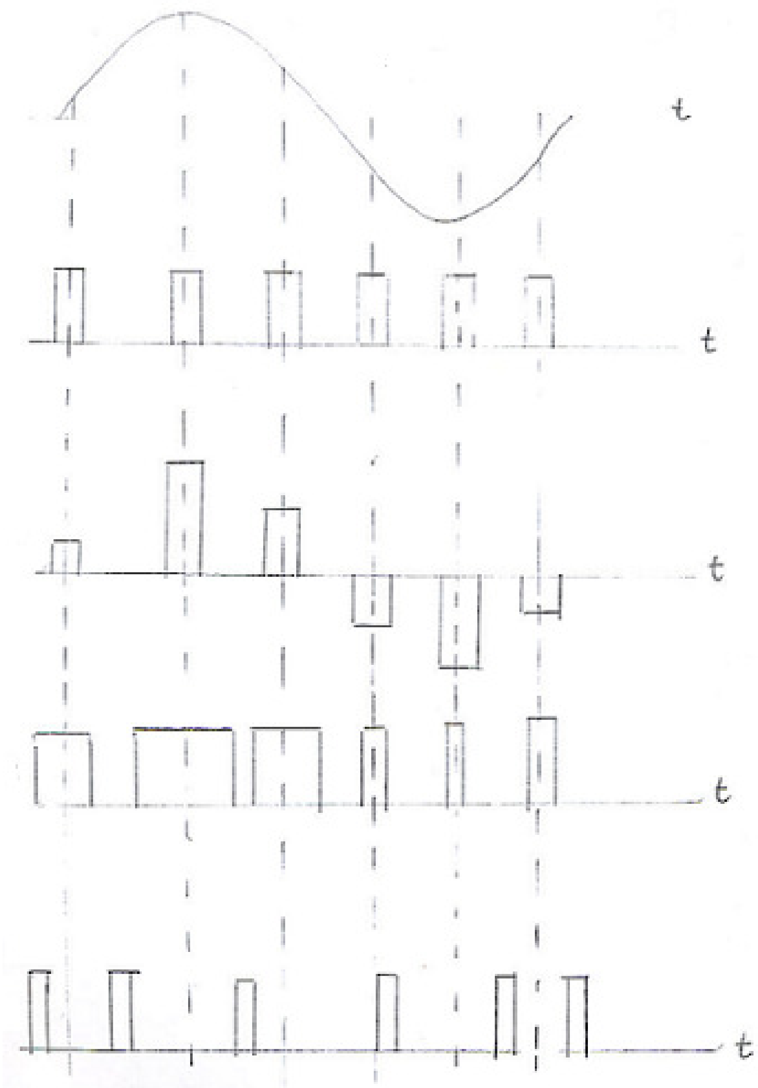
1. Coherent demodulation technique is required
2. Synchronization between transmitter & receiver
3. High bandwidth requirement as compared to PAM
4. Complexity in generation & detection (indirect) as compared to both, PAM & PWM

Comparing PAM, PWM & PPM

- Variable Characteristics
- Bandwidth Requirement
- Noise Immunity
- Constant Characteristics
- Transmitted Power
- Synchronizing Pulses
- Tx & Rx Complexity
- Output Waveforms
- Merits & Demerits

refer class notes for comparison

Review of Analog Pulse Modulation

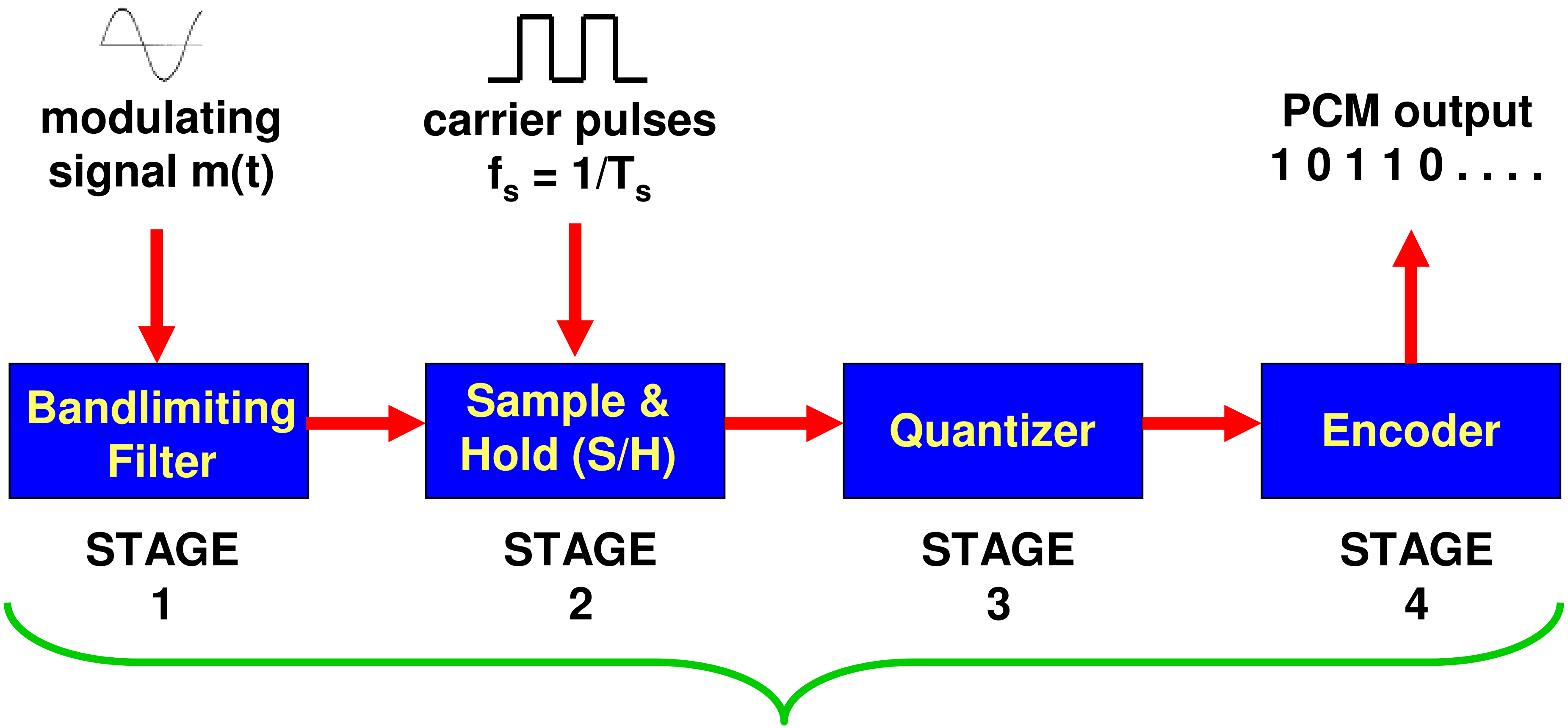


- modulating signal $v_m(t)$
- carrier signal (train of pulses)
- pulse amplitude modulation (PAM)
- pulse width/time modulation (PWM)
- pulse position modulation (PPM)

Pulse Code Modulation (PCM)

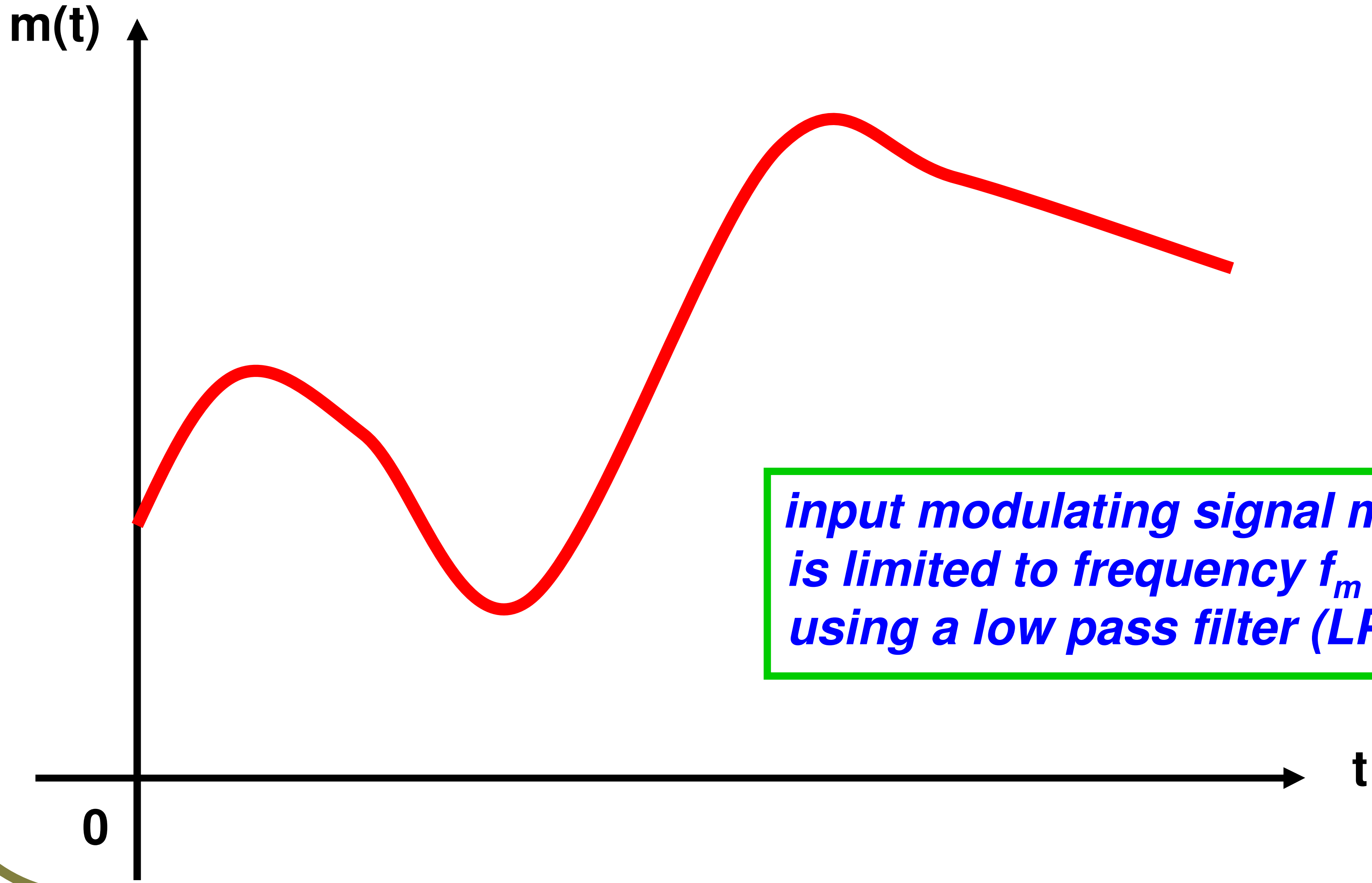
- Superior as compared to PAM, PWM & PPM with respect to the noise immunity
- Output is pulse with each having a constant amplitude, width & position
- Each sample is represented by some unique digital code word or a digital data sequence
- Uses process of quantization (rounding off) to generate codes from the sampled values
- Output is transmitted directly in digital form against analog form of PAM, PWM & PPM

Pulse Code Modulation (PCM) – (a) Transmitter Block Diagram



pulse code modulation (PCM) is basically four (4) stage process

Operation of PCM System – STAGE 1
(1) Bandlimiting Filter

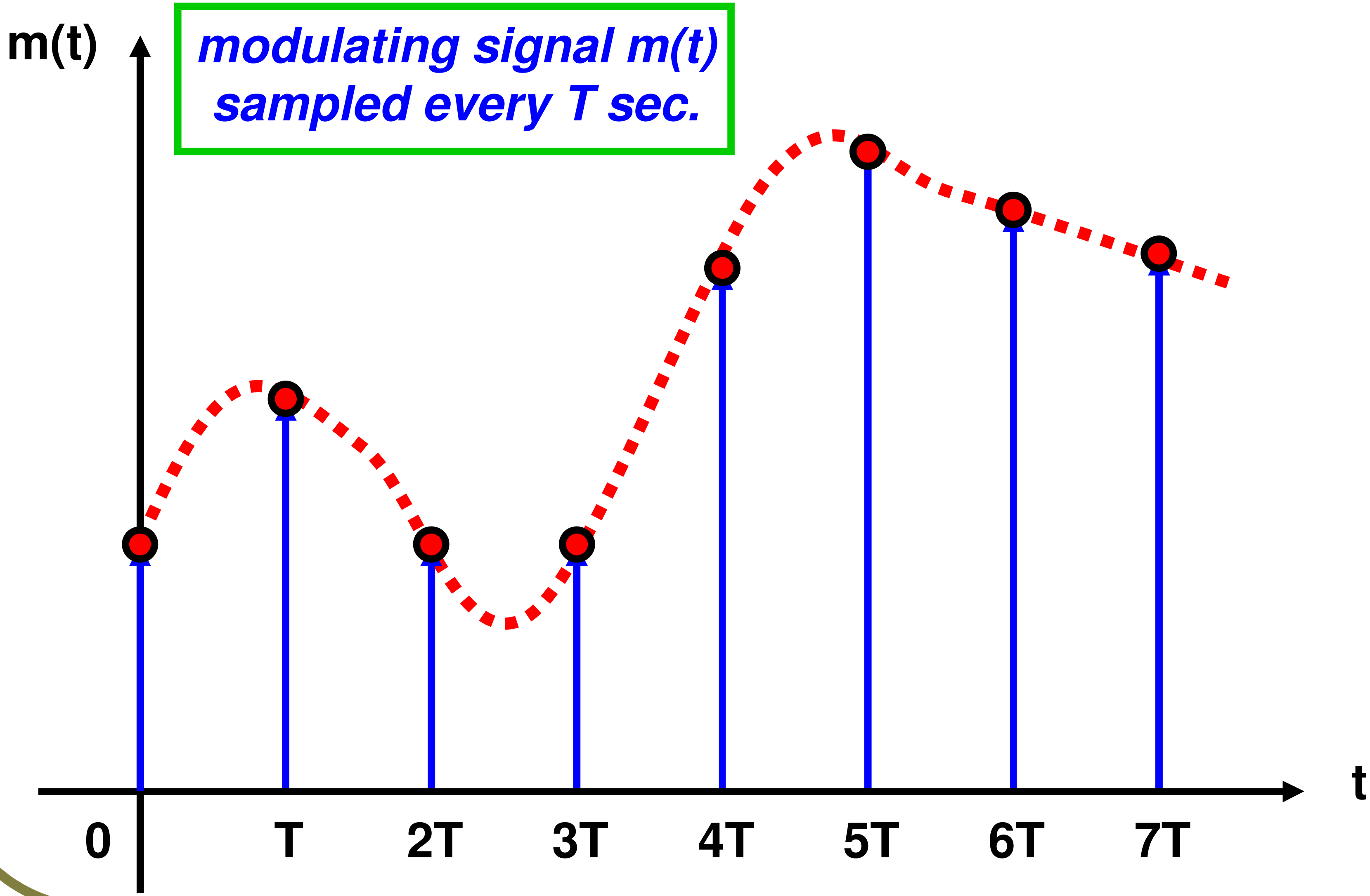


input modulating signal m(t) is limited to frequency f_m Hz using a low pass filter (LPF)

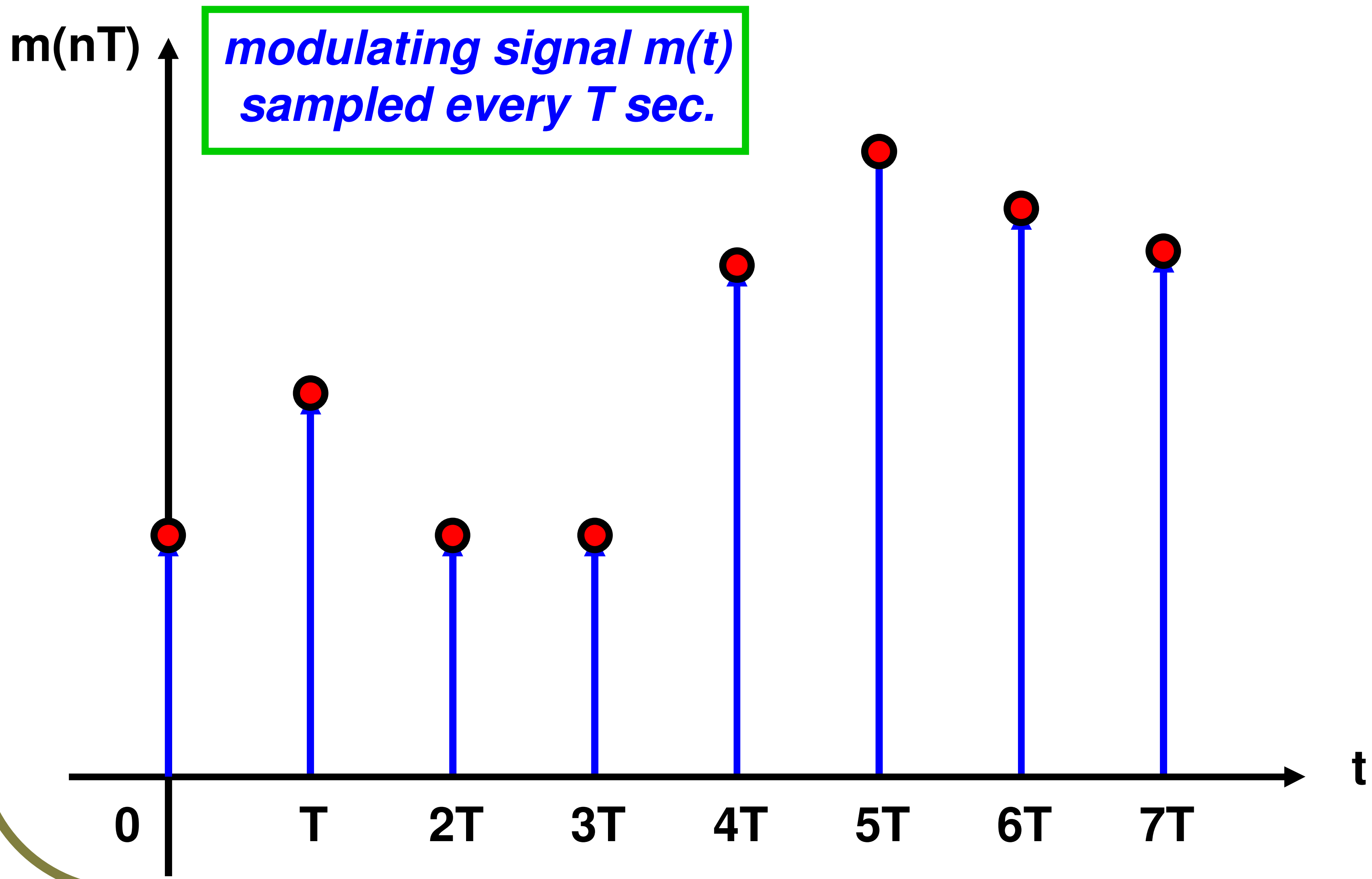
Operation of PCM System – STAGE 1
(1) Bandlimiting Filter

- **Bandlimiting filter is basically low pass filter (LPF) having a cut – off frequency of f_m Hz**
- **It means any frequencies in the modulating input signal m(t) beyond f_m Hz are cut – off**
- **As per sampling theorem criterion baseband m(t) limited to maximum frequency of f_m Hz**
- **It helps to establish an appropriate value of the sampling frequency of $f_s \geq 2f_m$**
- **This helps to prevent any chance of aliasing or fold – over distortion effect**

Operation of PCM System – STAGE 2
(2) Sample & Hold (S/H)



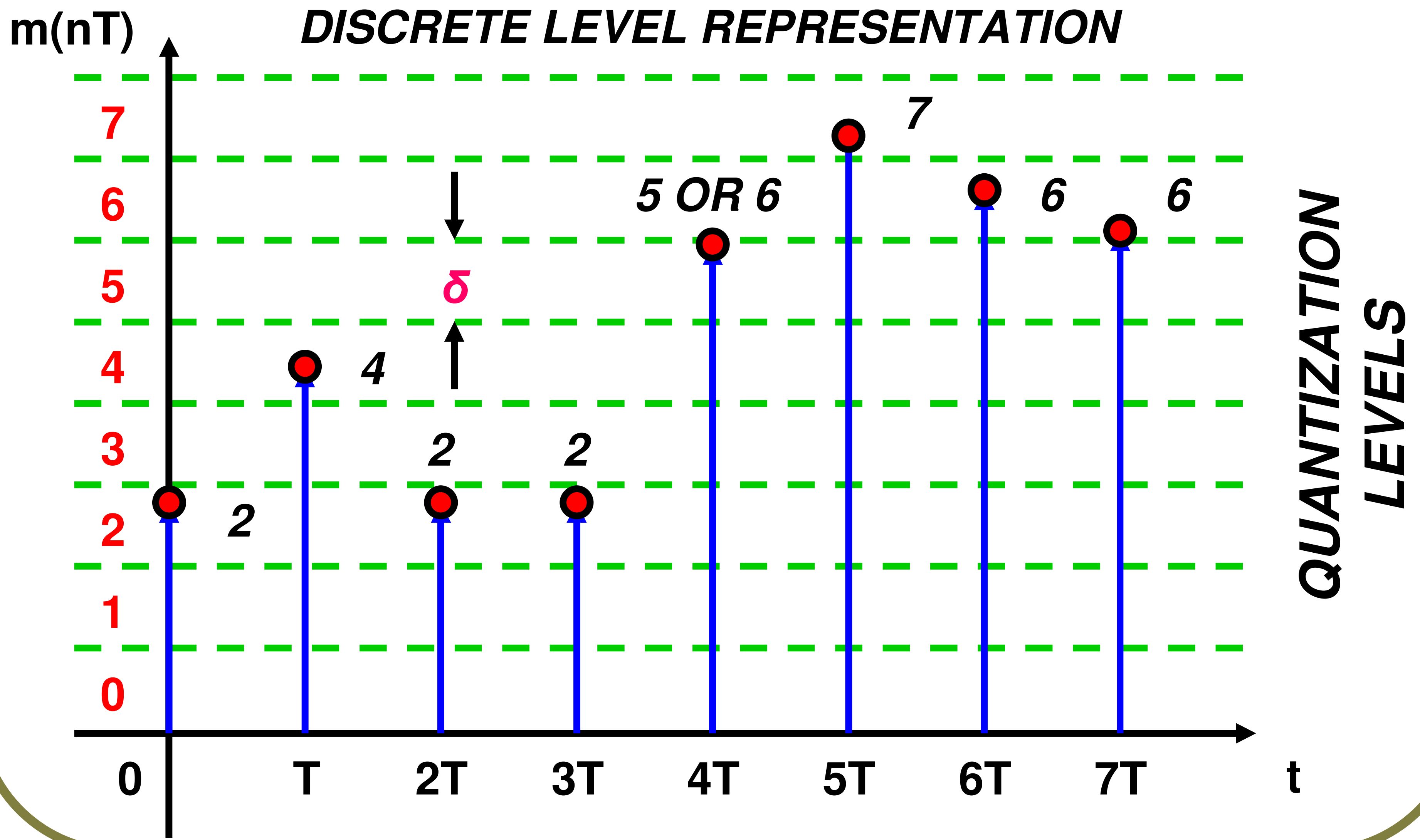
Operation of PCM System – STAGE 2
(2) Sample & Hold (S/H)



Operation of PCM System – STAGE 2
(2) Sample & Hold (S/H)

- Bandlimited signal m(t) is sampled at a rate higher than the Nyquist rate of $f_s \geq 2f_m$
- Each samples are equally spaced over time period of T_s seconds where $T_s = 1/f_s$
- Reconstructing original signal m(t) possible only if it is sampled at a rate of $f_s \geq 2f_m$
- This helps to prevent any chance of aliasing or fold – over distortion effect at $f_s \geq 2f_m$
- Flat – top sampling or natural sampling can be used, as both are practically realizable

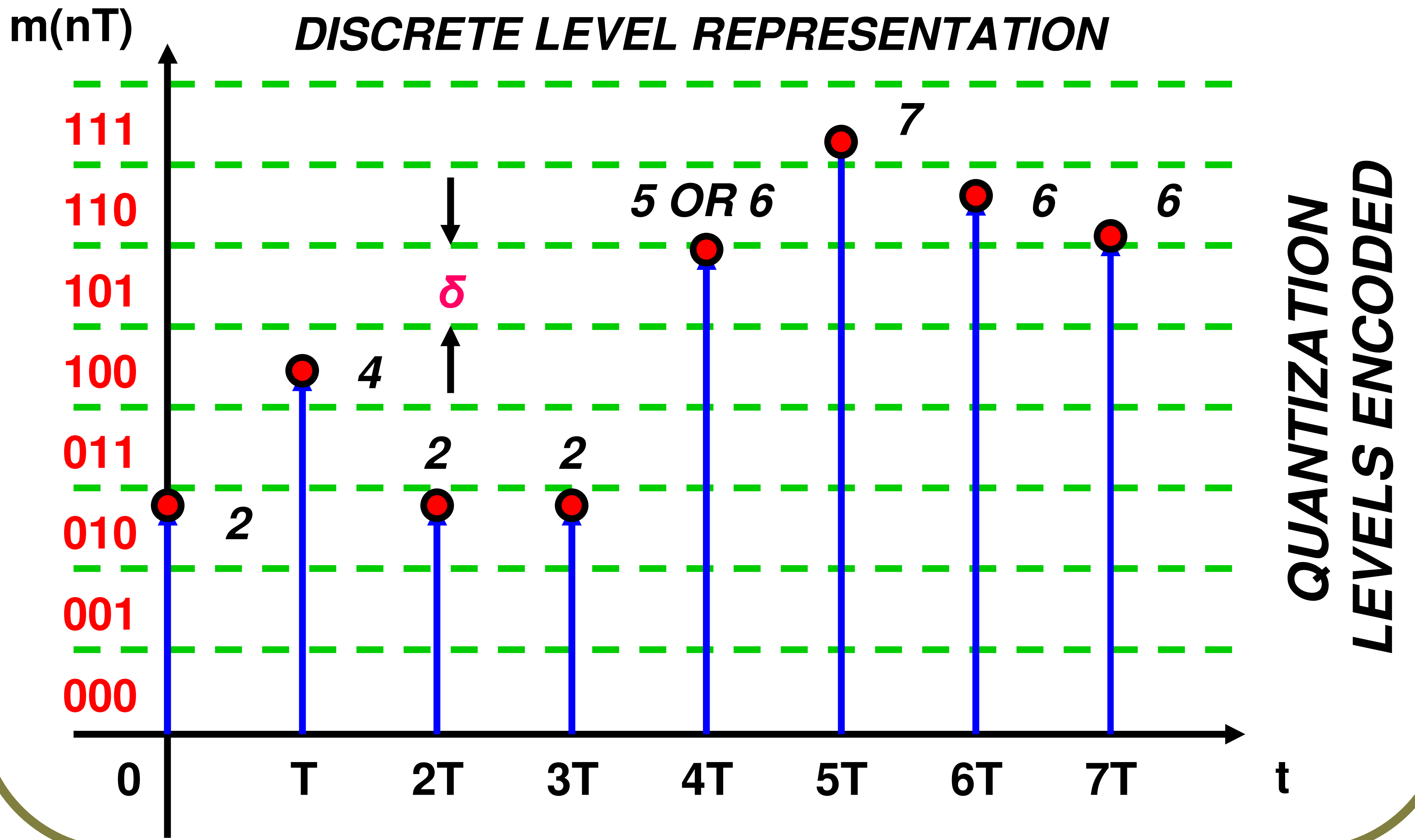
Operation of PCM System – STAGE 3
(3) Quantization Process



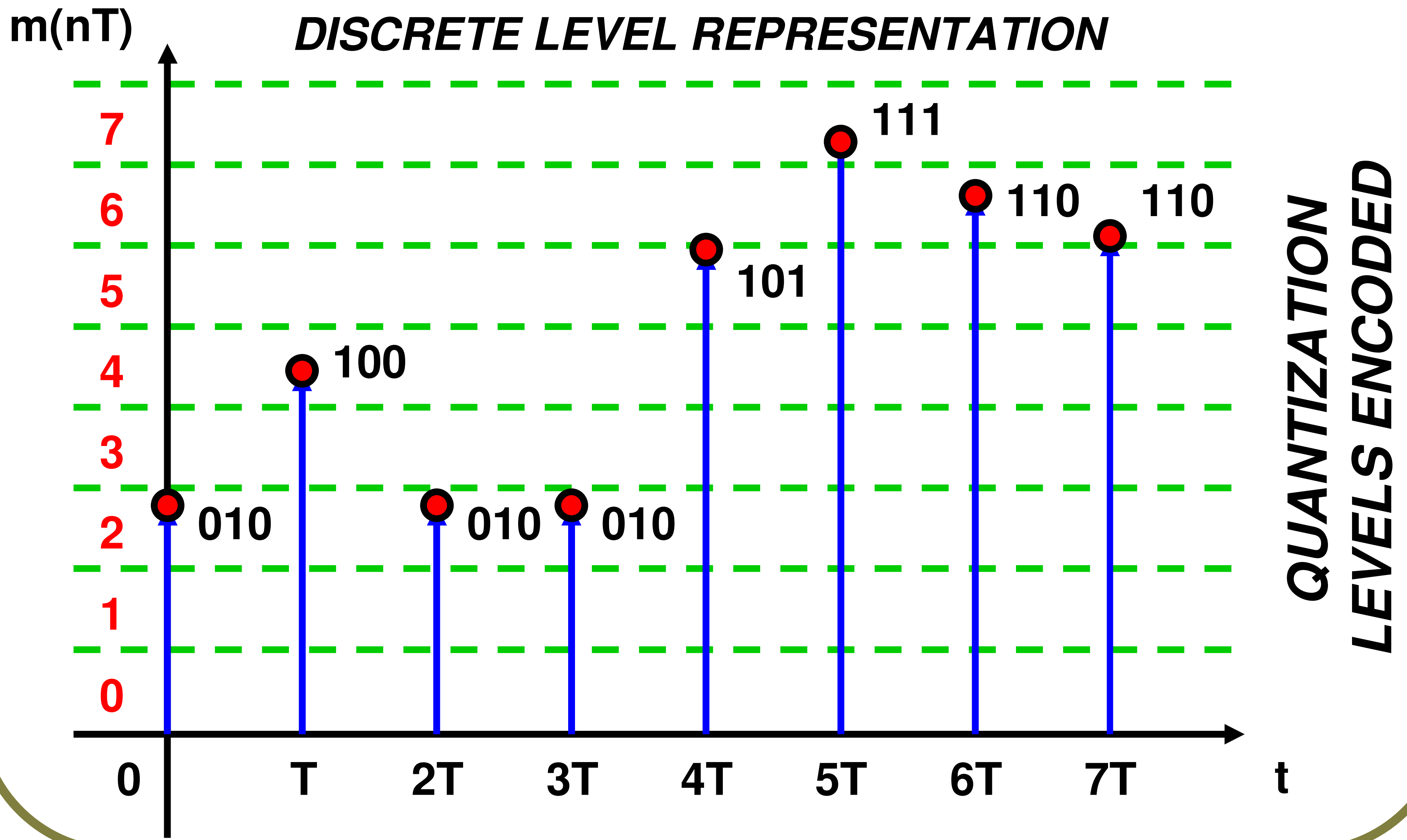
Operation of PCM System – STAGE 3
(3) Quantization Process

- It is a process in which actual sampled value is approximated to nearest standard level
- Total (amplitude) range of input modulating signal is divided into small levels or steps
- This depends upon the nature of modulating signal & no. of bits (N) used for its encoding
- Here the sampled signal's amplitude actually is set to the nearest standard level or step
- Uniform or non – uniform quantization based on the uniformity of the given step size (δ)

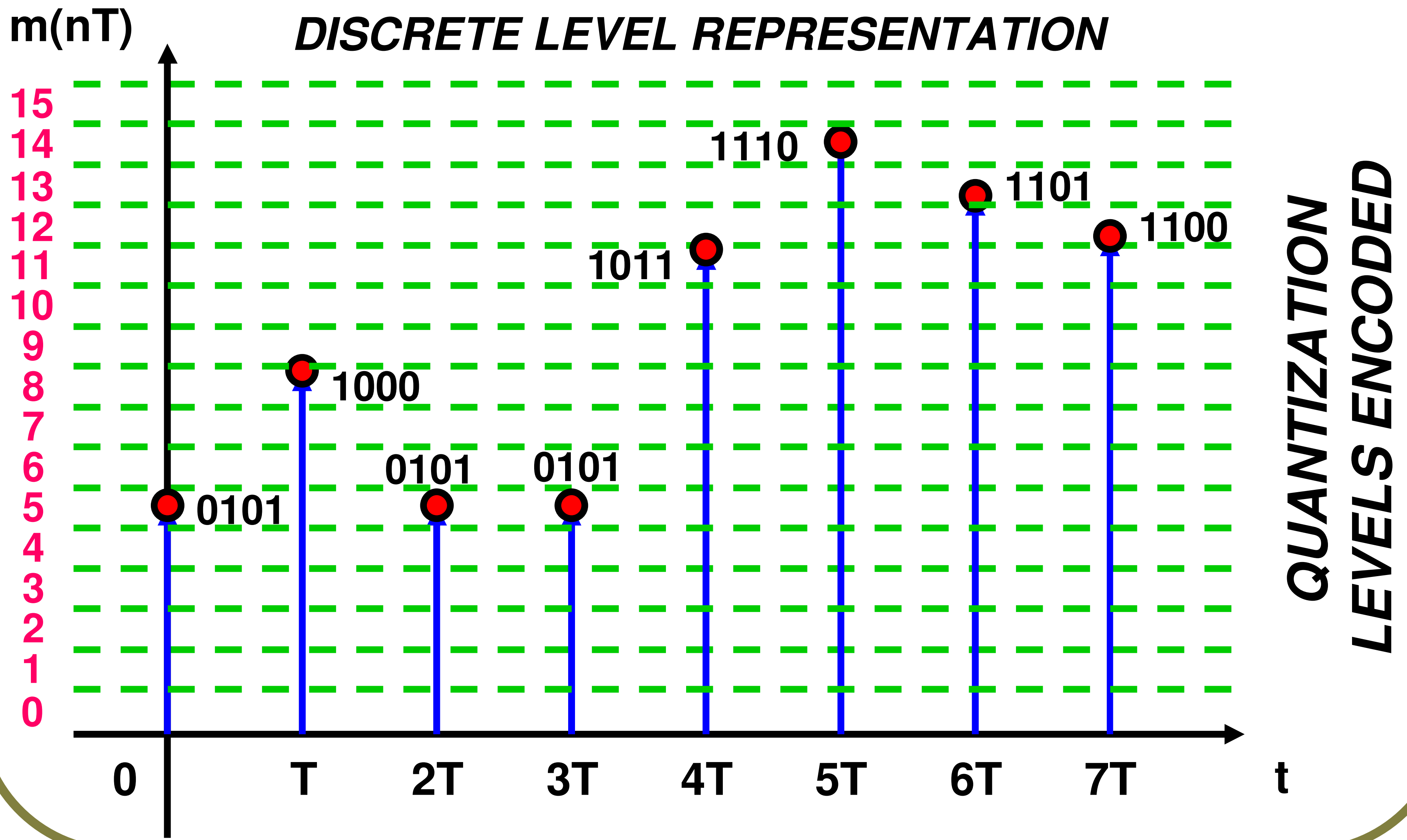
Operation of PCM System – STAGE 4
(4) Encoding Process



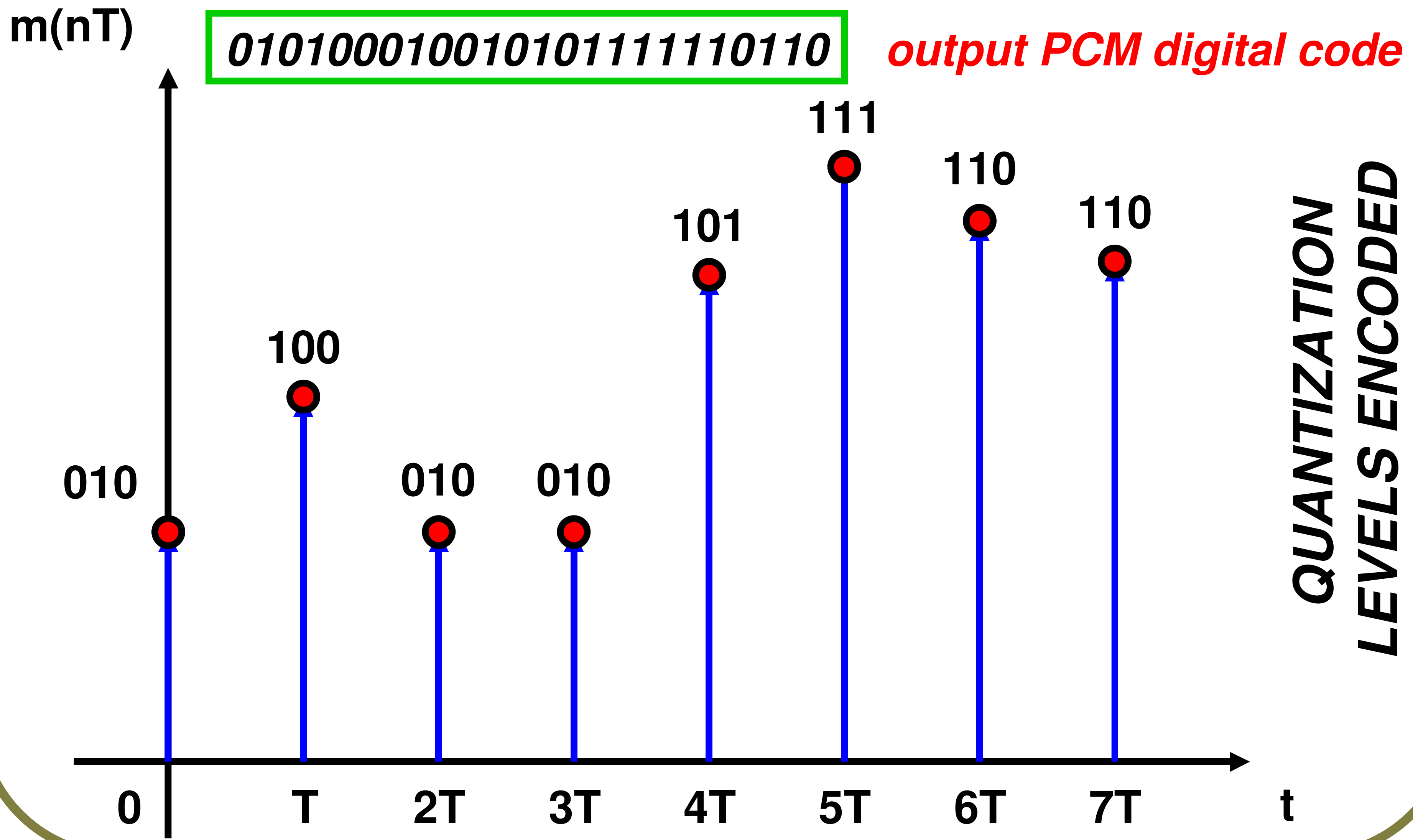
Operation of PCM System – STAGE 4
(4) Encoding Process



Operation of PCM System – STAGE 4
(4) Encoding Process



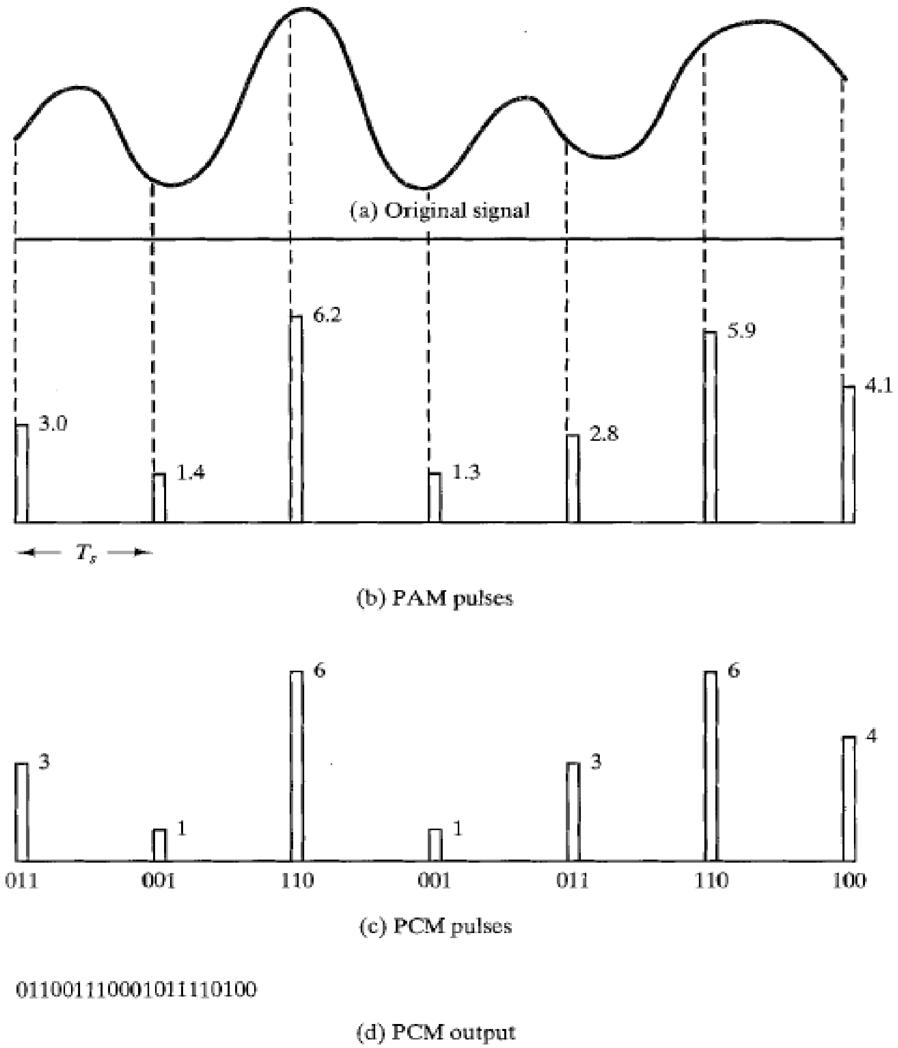
Operation of PCM System – STAGE 4
(4) Encoding Process



Operation of PCM System – STAGE 4
(4) Encoding Process

- **Quantized sampled values are converted in digital (data) codes using binary bits (digits)**
- **This is performed by using an N – bit analog to digital converter (ADC) mechanism**
- **Bits used for encoding (N) decide number of the quantization levels 'Q' since $Q = 2^N$**
- **Step size (δ) can be decreased by increasing 'N' which decreases quantization levels 'Q'**
- **Digital output (1's & 0's) is transmitted over communication channel by using line codes**

Operation of PCM System – Waveforms



original bandlimited input signal m(t)

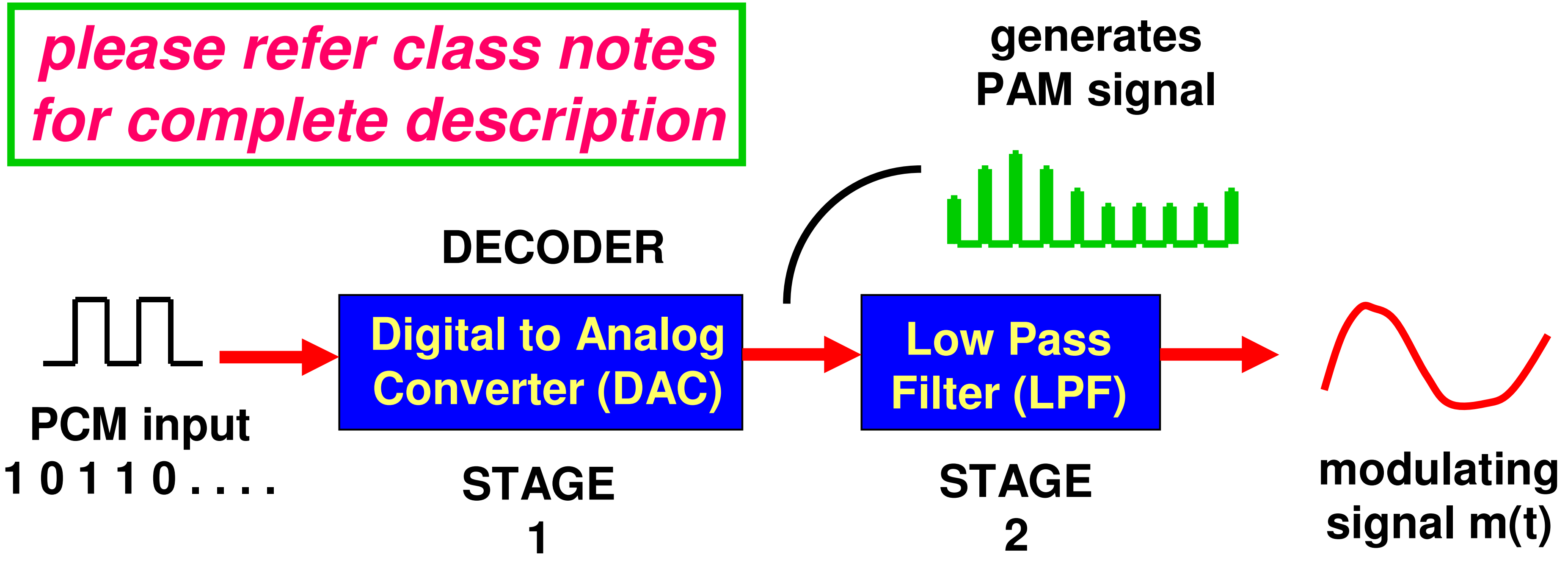
sampled value m(t) at every T_s seconds

Quantized values of sampled signal

PCM output data

Pulse Code Modulation (PCM) – (b) Receiver Block Diagram

please refer class notes for complete description



- **Decoder** – digital to analog converter (DAC) used to generate output pulse amplitude modulation (PAM)
- **Low Pass Filter (LPF)** – demodulates output of PAM to recover the original input modulating signal m(t)

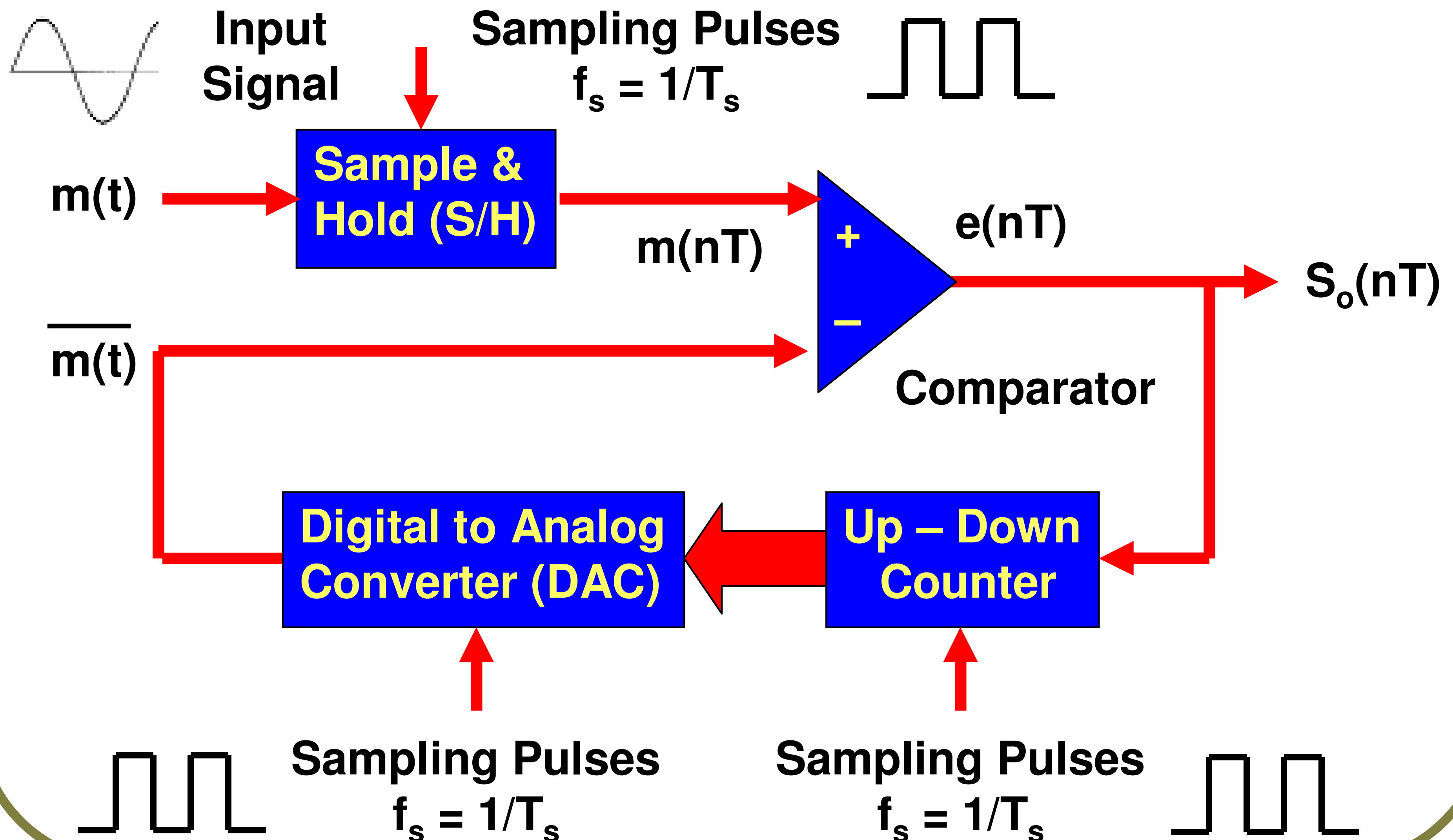
Advantages of PCM

- Excellent noise immunity compared to PAM, PWM & PPM, more noise resistant
- Repeaters can be used which regenerate the PCM signal, reducing effect of noise further
- Possible to store PCM signal (digital nature) of waveforms due to digitization
- Coding techniques can be used for message encryption & data security for transmission
- Excellent signal to noise ratio (SNR) which balances the increase in the bandwidth

Disadvantages of PCM

- Quantization Error (ε_Q) which can easily be corrected by the step size variations (s)
- Extremely large bandwidth required since a sample needs 'N' bits for representation
- Larger signaling rate & a data transmission rate compared to PAM, PWM & PPM
- Encoding, decoding & quantizing circuitry of PCM is extremely complicated
- PCM transmission requires repeaters, which makes hardware costly & complicated

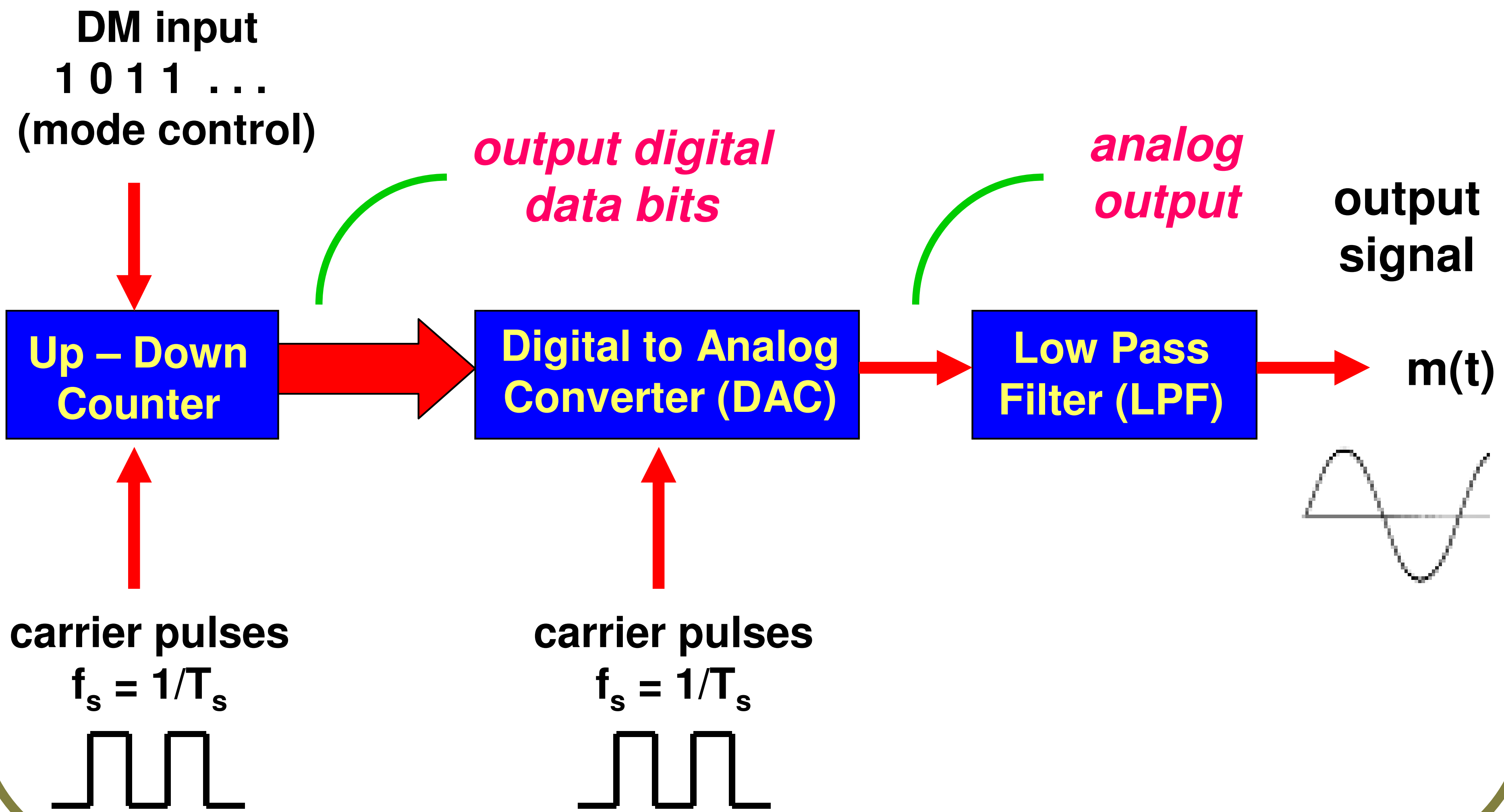
Linear Delta Modulation (LDM)
(a) Transmitter Block Diagram



Linear Delta Modulation (LDM)
(a) Transmitter Block Diagram

- Input m(t) is compared with an approximate version of itself using a comparator device
- At each sampling instant an error signal that results from comparator output given to S/H
- This e(t) = 1 or 0 forms the mode control for up-down counter operating at frequency f_s
- Counter output converted into approximate analog form by digital to analog converter
- This is given through feedback loop back to comparator, giving output single bit (1 or 0)

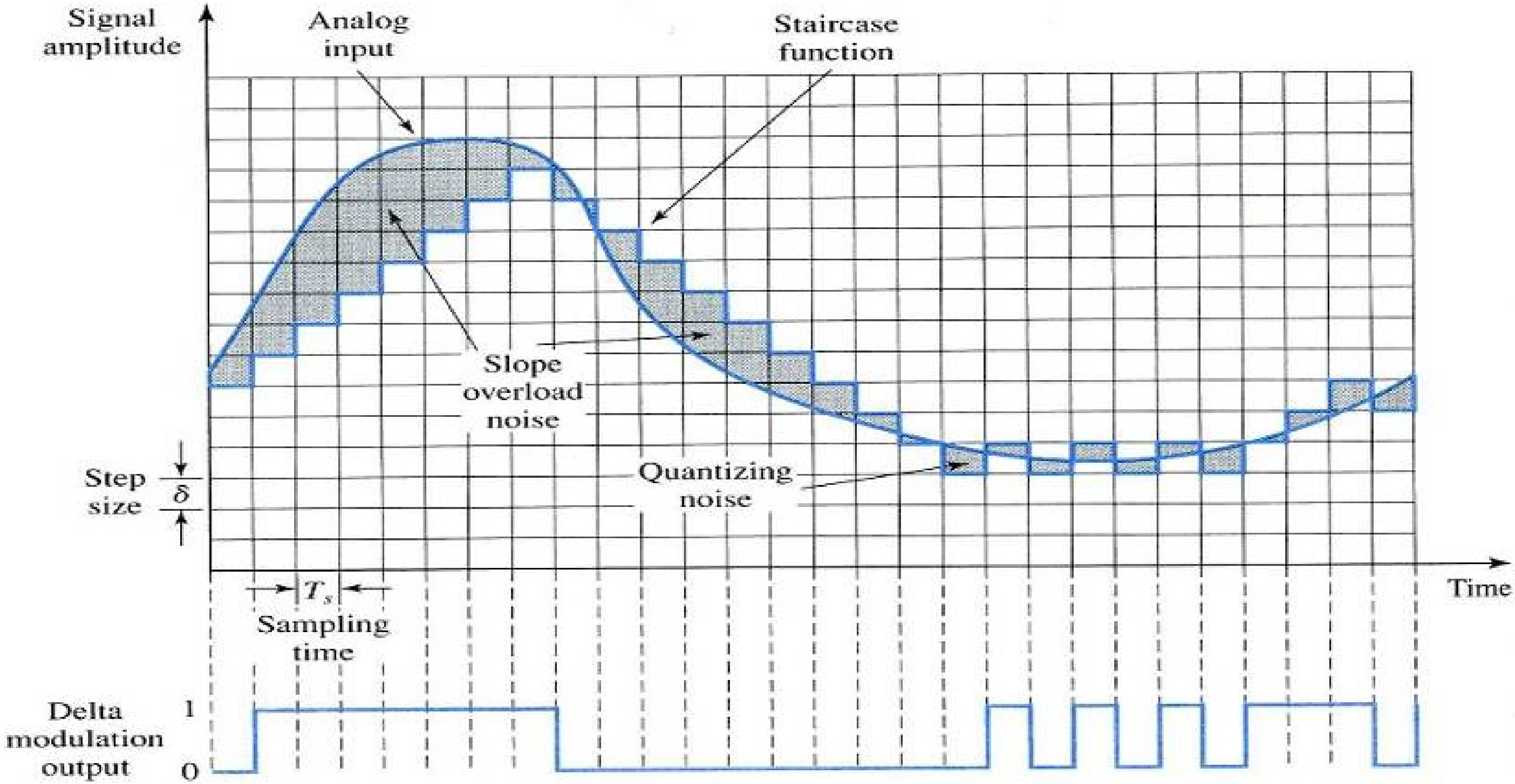
Linear Delta Modulation (LDM)
(b) Receiver Block Diagram



Linear Delta Modulation (LDM) (b) Receiver Block Diagram

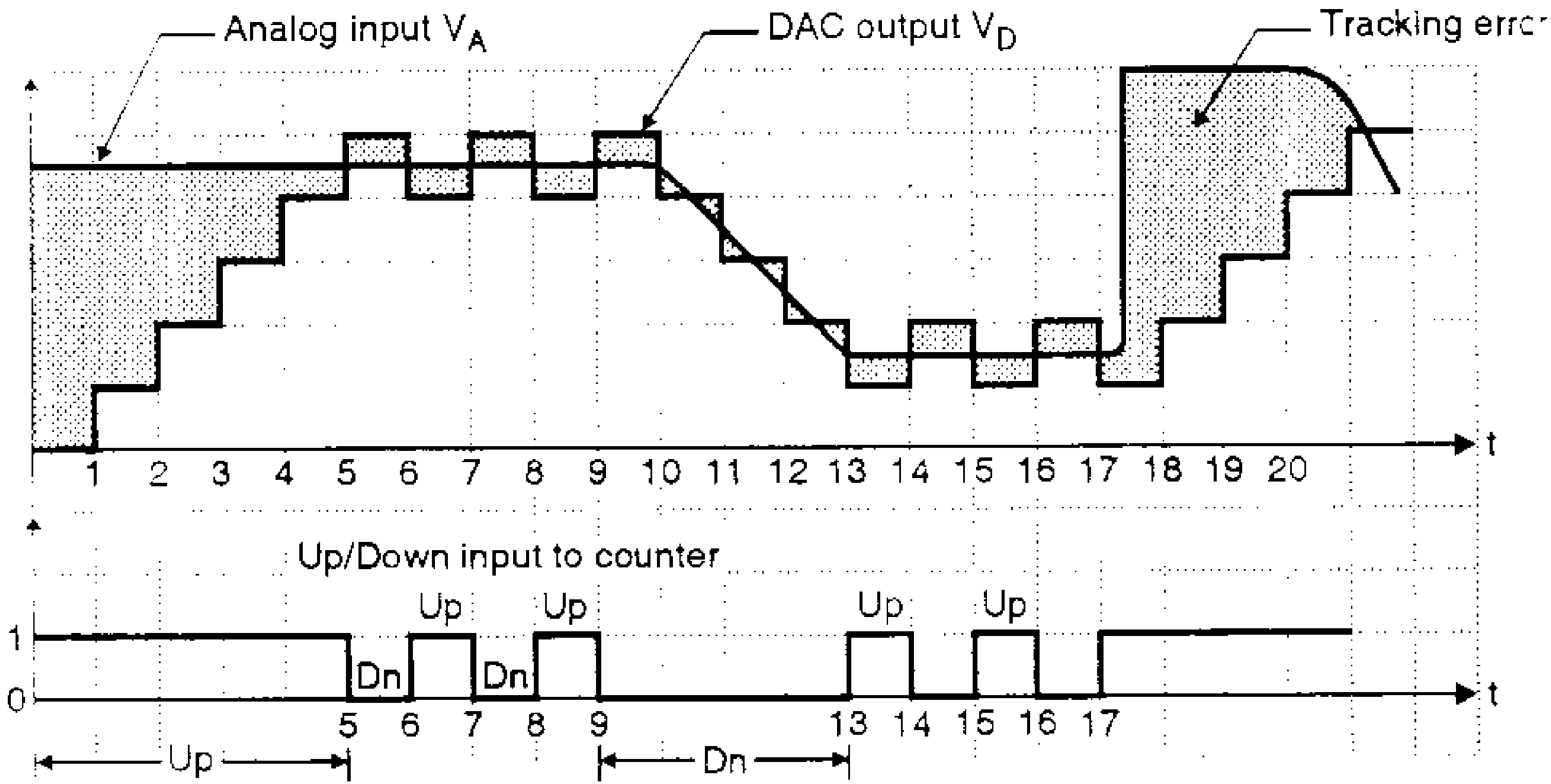
- Transmitted DM output is applied as a mode control to up-down counter at frequency f_s
- Based on 1 or 0 received, counter o/p count increments or decrements (step-wise form)
- This is given to a digital to analog converter (DAC) generating analog output voltage
- Low pass filter (LPF) is tuned to a frequency of the input modulating signal of f_m
- It smoothens output of DAC, generating an approximate version of input signal m(t)

Linear Delta Modulation (LDM) – (c) Input & Output Waveforms



Delta Modulation

Linear Delta Modulation (LDM) – (c) Input & Output Waveforms



Linear Delta Modulation (LDM) – (d) Advantages & Disadvantages

ADVANTAGES :-

1. Only one bit for transmission than PCM & DPCM
2. Lower transmission bandwidth compared to PCM
3. Lower data transmission rate than PCM, DPCM
4. Doesn't require the use of predictor or quantizer

DISADVANTAGES :-

1. Slope Overload Distortion
2. Granular Noise OR Hunting Error
3. Start – up Interval Error

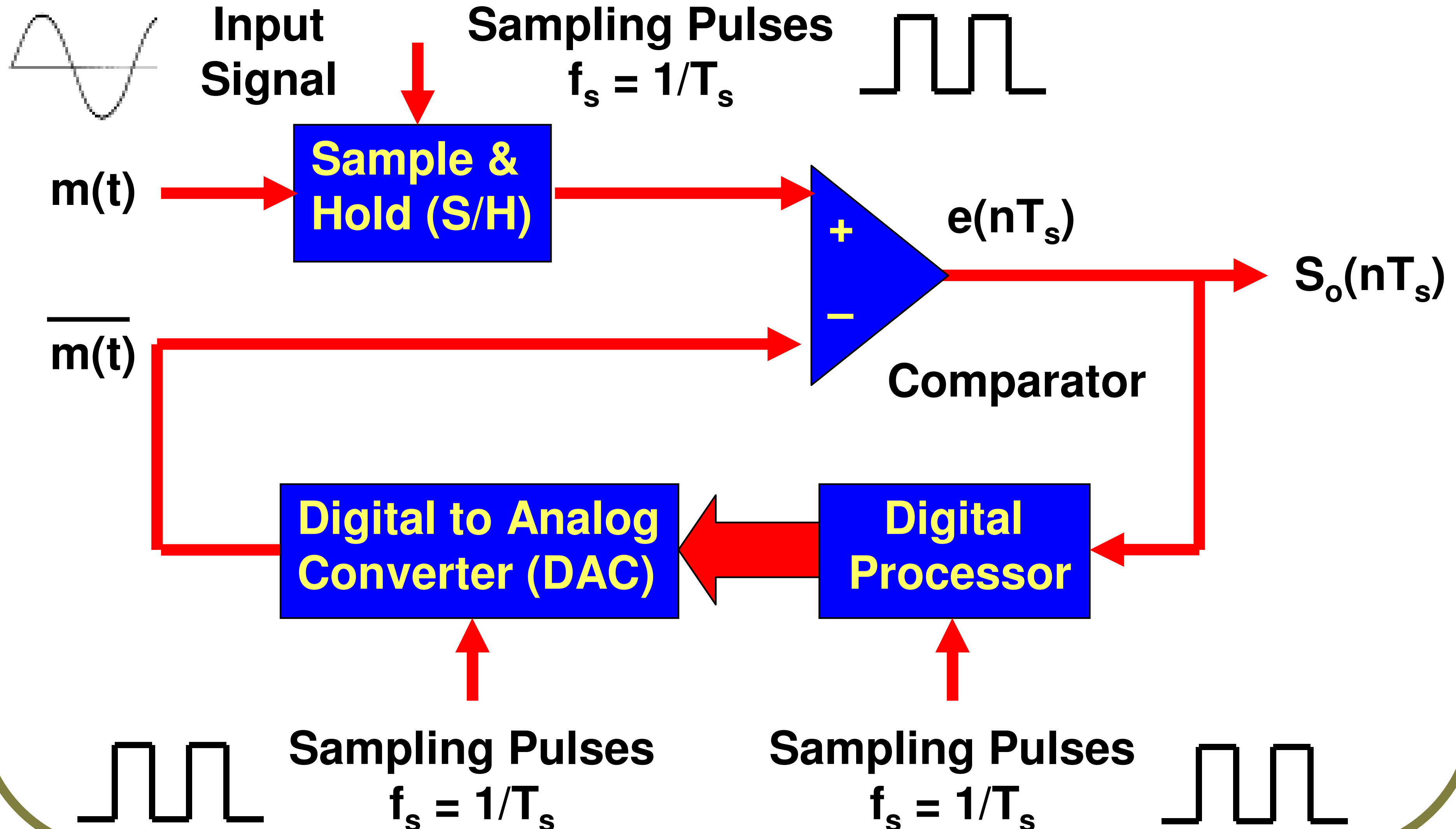
above disadvantages are called errors in delta modulation (DM)

Linear Delta Modulation (LDM) – (e) Summary of Errors in LDM

- **START UP INTERVAL ERROR** occurs when output of DAC takes considerable time to reach upto input modulating signal m(t), due to small step size (δ)
- **SLOPE OVERLOAD DISTORTION** is when the output of DAC fails to follow the slope (rise/fall) of the input modulating signal m(t), due to small step size (δ)
- **GRANULAR NOISE OR HUNTING ERROR** is when an output of DAC, hunts or travels above or below m(t), alternately generating 1 & 0 at the output of DM

these errors can be minimized only by increasing or decreasing step size (δ)

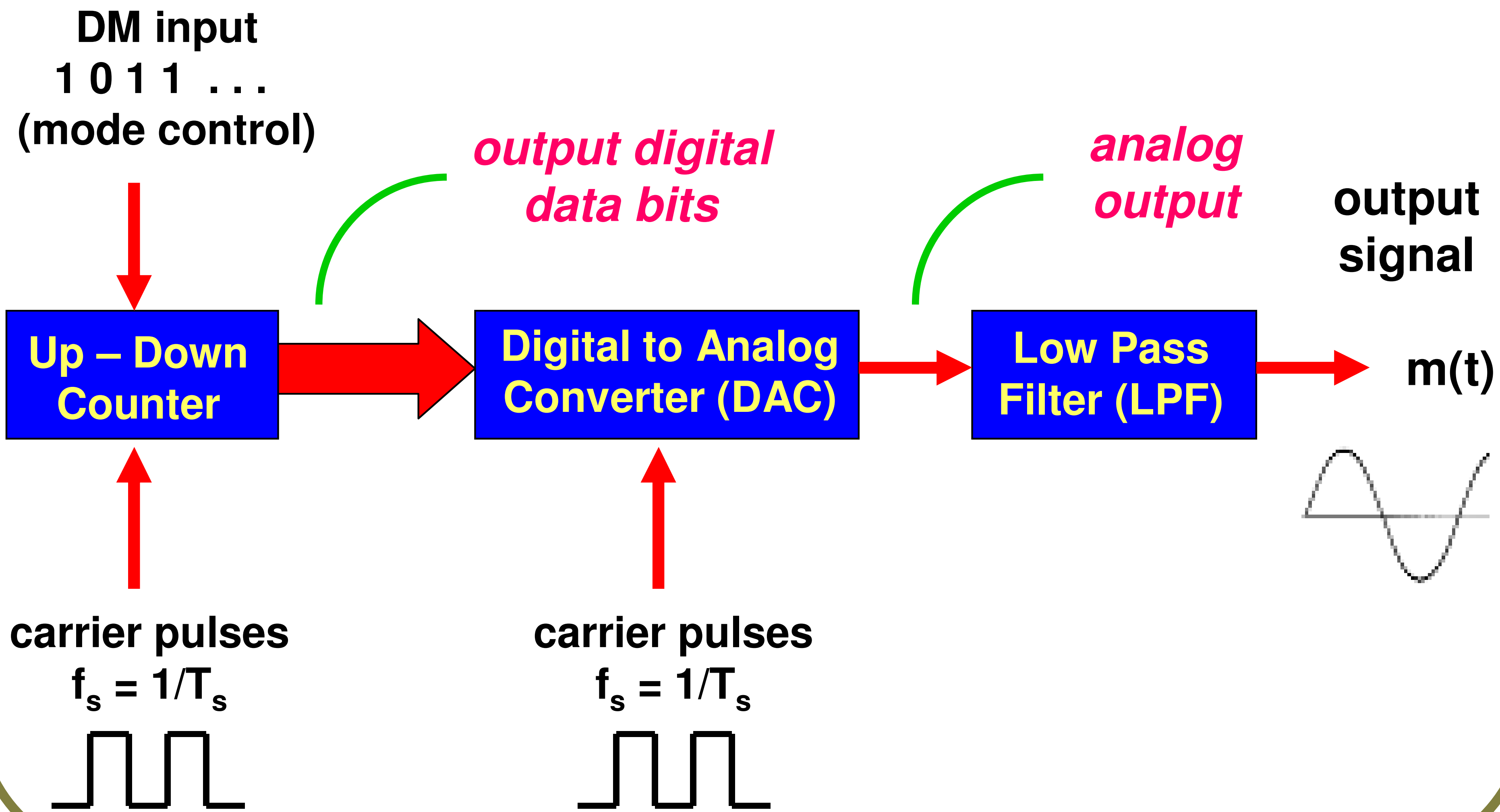
Adaptive Delta Modulation (ADM)
(a) Transmitter Block Diagram



Adaptive Delta Modulation (ADM) (a) Transmitter Block Diagram

- Input m(t) is compared with an approximate version of itself using a comparator device
- At each sampling instant an error signal that results from comparator output given to S/H
- This e(t) = 1 or 0 forms the mode control for the digital processor to vary the step size (δ)
- Digital processor output is converted into an analog form by digital to analog converter
- This is given through feedback loop back to comparator, giving output single bit (1 or 0)

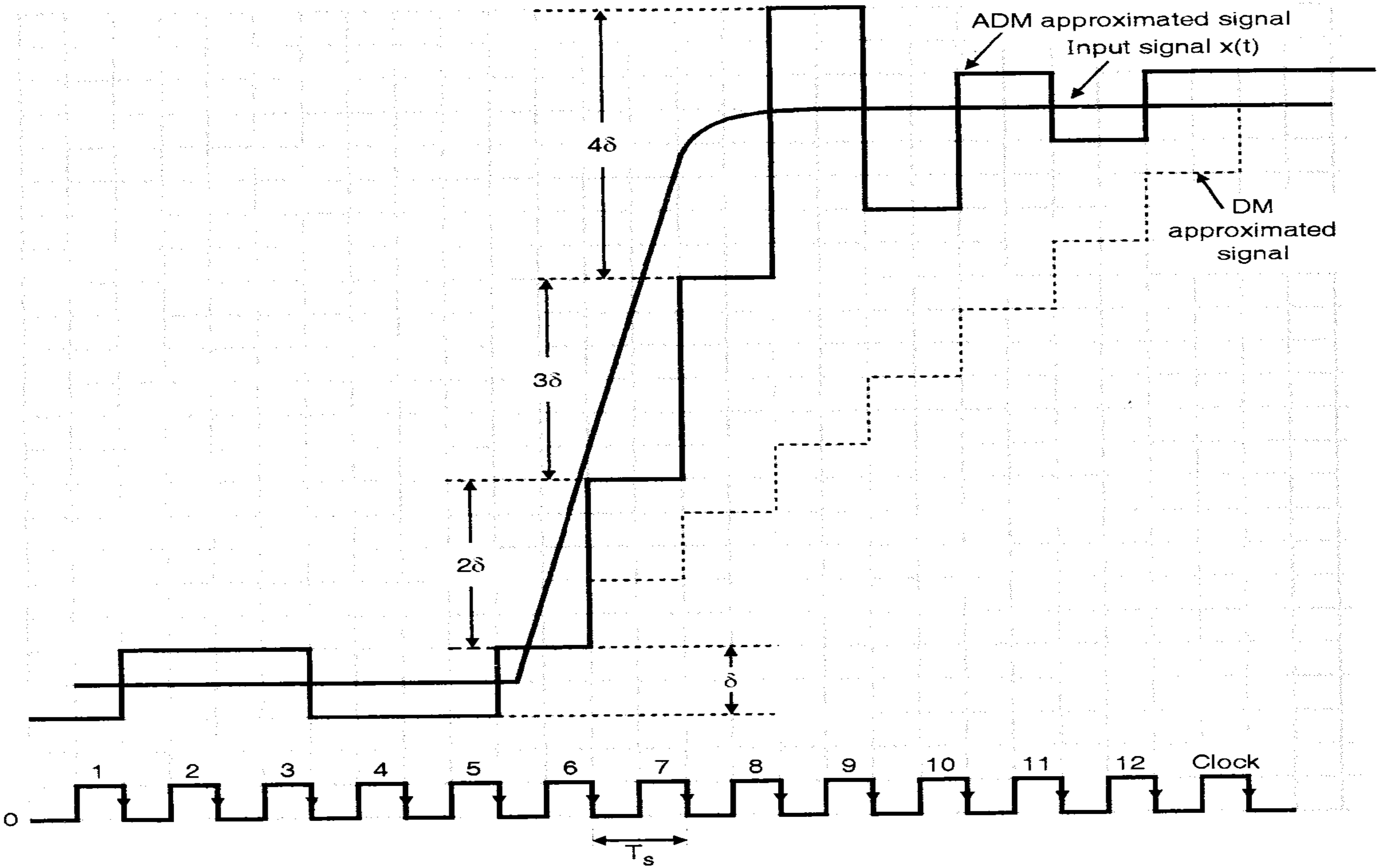
Adaptive Delta Modulation (ADM)
(b) Receiver Block Diagram



Adaptive Delta Modulation (ADM)
(b) Receiver Block Diagram

- Transmitted ADM output is applied as mode control to up-down counter at frequency f_s
- Based on 1 or 0 received, counter o/p count increments or decrements (step-wise form)
- This is given to a digital to analog converter (DAC) generating analog output voltage
- Low pass filter (LPF) is tuned to a frequency of the input modulating signal of f_m
- It smoothens output of DAC, generating an approximate version of input signal m(t)

Adaptive Delta Modulation (ADM) – (c) Input & Output Waveforms



Adaptive Delta Modulation (ADM) – (d) Advantages & Disadvantages

ADVANTAGES :-

1. Only one bit for transmission than PCM & DPCM
2. Lower transmission bandwidth compared to PCM
3. Lower data transmission rate than PCM, DPCM
4. Doesn't require the use of predictor or quantizer
5. Variable step size (δ) to minimize errors in DM

DISADVANTAGES :-

1. Digital processor design is difficult (complicated)
2. Trade – offs is to be maintained between minimizing start – up interval & slope overload distortion or the hunting error as 'δ' can be increased or decreased

Comparison of PCM, DM & ADM

- Principle of operation
- No. of bits per sample
- Step size (δ)
- Configuration
- Data transmission (R)
- Bandwidth (BW)
- Errors (if any)
- Advantages
- Disadvantages

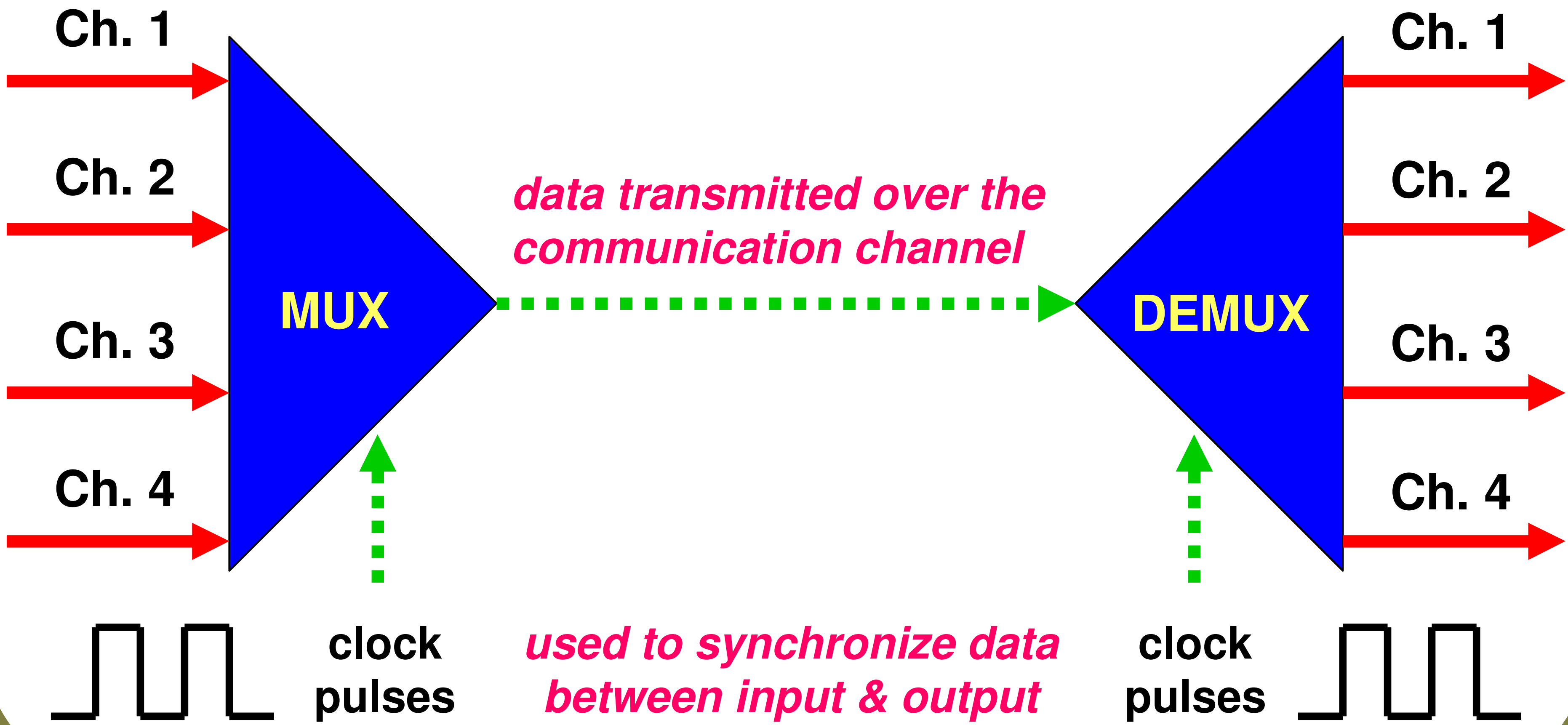
refer to class notes for complete description

Introduction to Multiplexing

- Process of simultaneously transmitting two or more signals over a single channel
- It is possible to increase the number of total channels to transmit more information
- It has multiplexer at the input which accepts multiple different input signals (channels)
- The output consists of demultiplexer used to sort out original input signals (channels)
- Useful in telephony, satellite & optical fiber communication, telemetry etc.

Introduction to Multiplexing

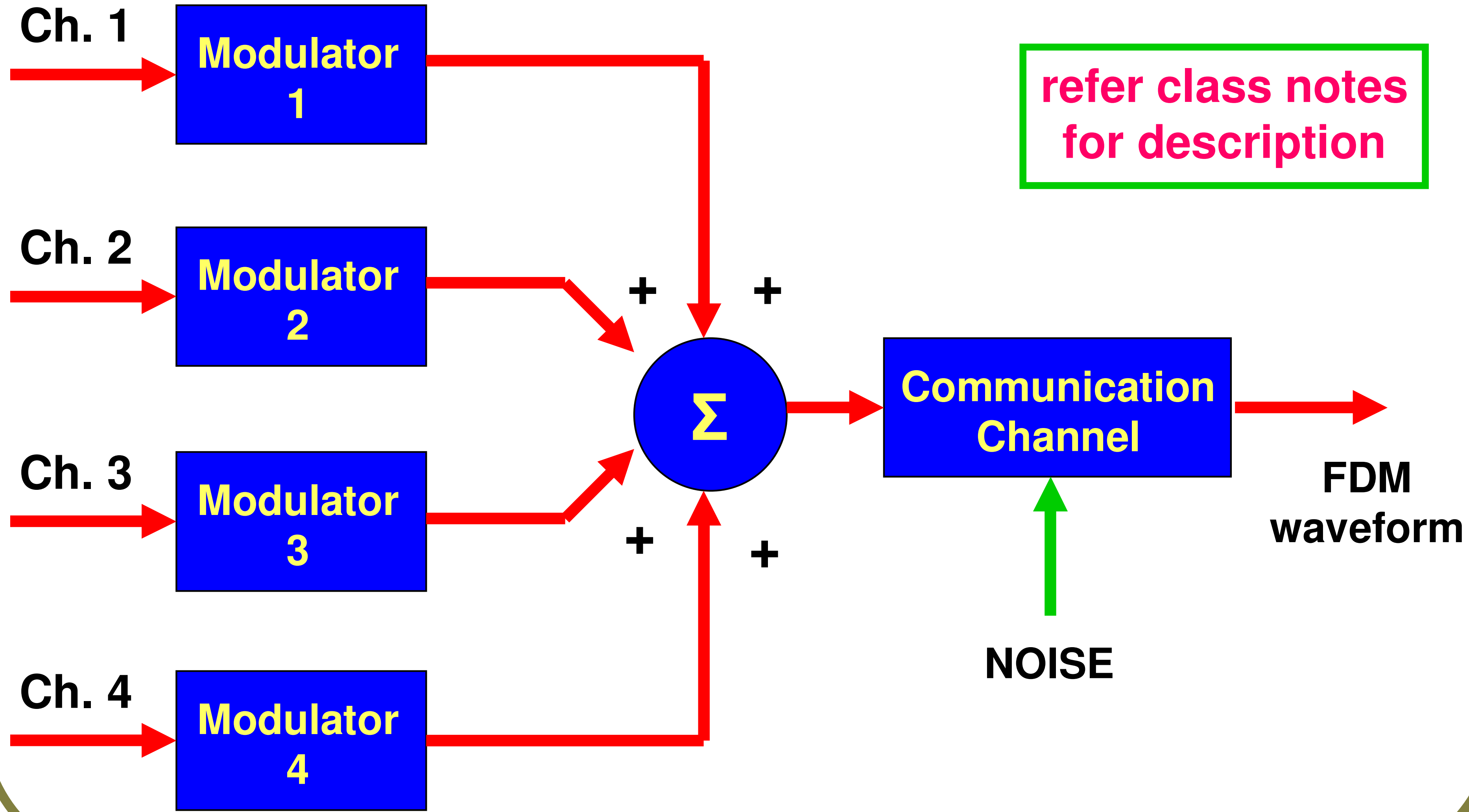
SYSTEM BLOCK DIAGRAM :-



Classification of Multiplexing

- Multiplexing
  - Analog Multiplexing
    - Frequency Division Multiplexing (FDM)
    - Wavelength Division Multiplexing (WDM)
  - Digital Multiplexing
    - Time Division Multiplexing (TDM)

Frequency Division Multiplexing (FDM) – (a) Transmitter Block Diagram



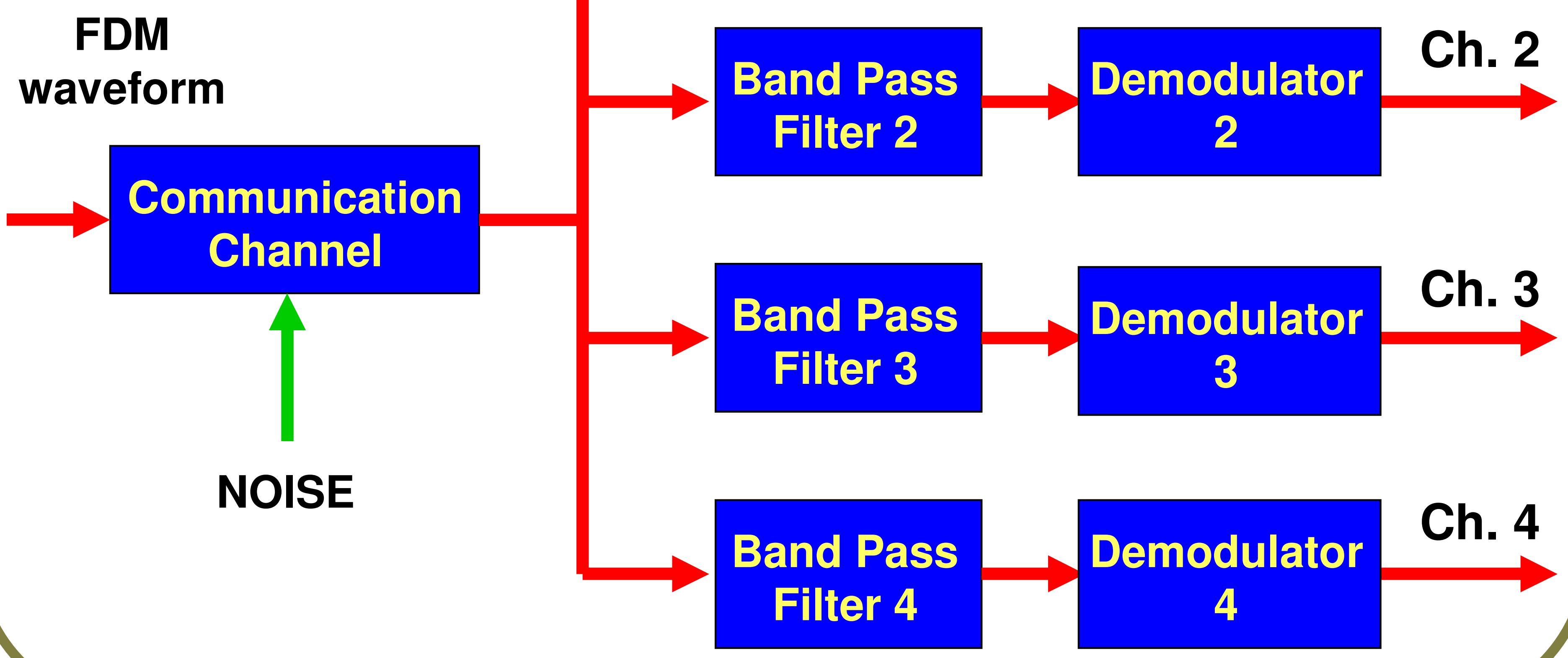
refer class notes for description

Frequency Division Multiplexing (FDM) – (a) Transmitter Block Diagram

- Concept of FDM is based on the sharing of a common bandwidth between different users
- Each of the input signal (channel) modulates different, independent carrier signals
- Due to this, there is shift in all the frequency spectrum of input signals (channels)
- Modulator outputs are added by a summing point block for further transmission
- This single output signal contains all of the individual frequency spectrums as shown

Frequency Division Multiplexing (FDM) – (b) Receiver Block Diagram

refer class notes for description



Frequency Division Multiplexing (FDM) – (b) Receiver Block Diagram

- The composite FDM signal is now applied to group of individual bandpass filters (BPF)
- Each of the bandpass filters (BPF) has cutoff frequency of one of the carriers
- They have a sufficient bandwidth to pass all desired information without any distortion
- Each BPF will only pass its corresponding frequency, thus rejecting all the others
- The channel demodulator then removes the carrier, recovering back the original signal

Frequency Division Multiplexing (FDM) – (d) Advantages of FDM

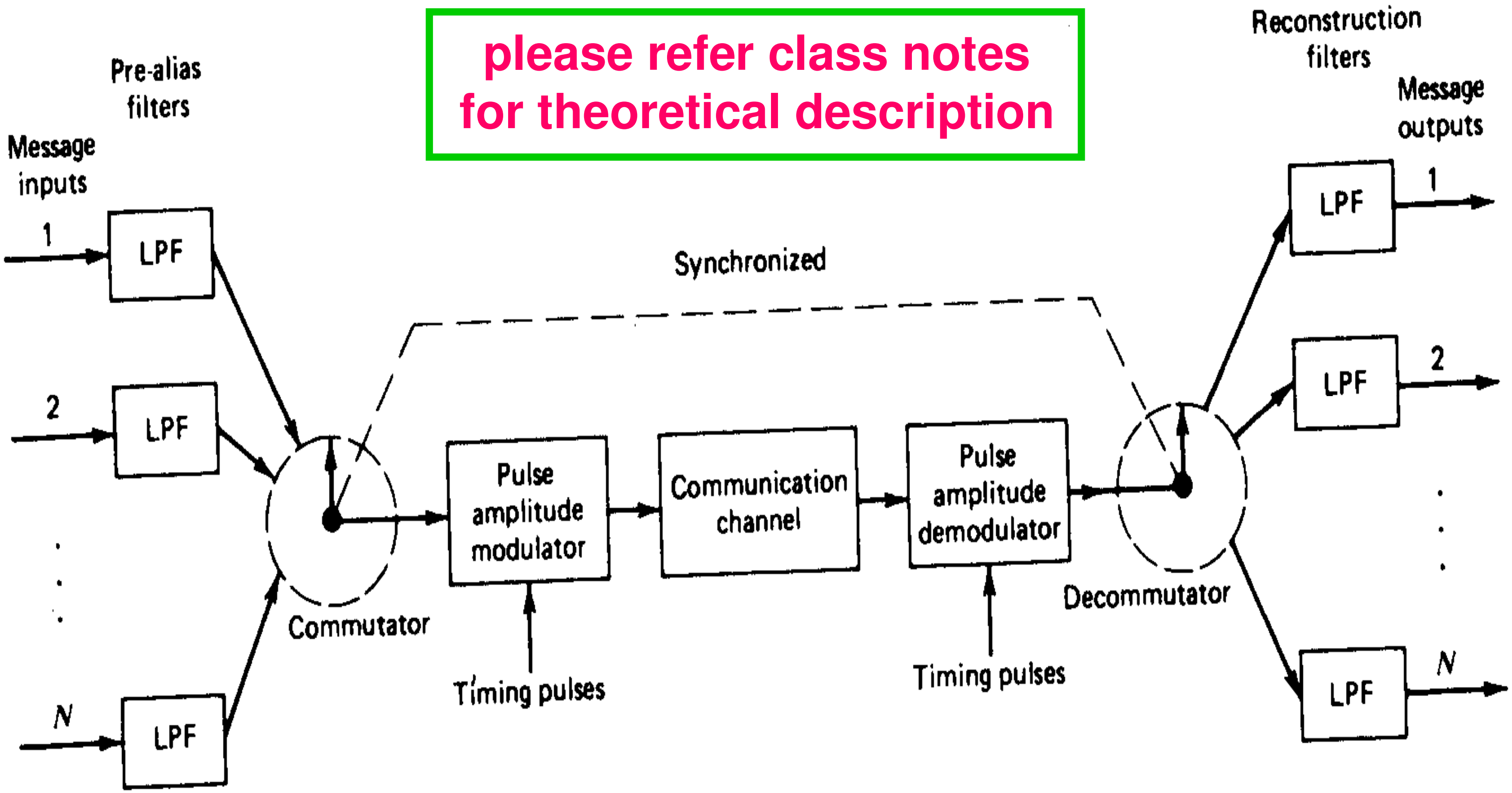
- A large number of channels (signals) can be simultaneously transmitted at all times
- FDM doesn’t need synchronization between the transmitter & the receiver unlike TDM
- Unlike TDM, demodulation of FDM is simple by using demodulators, BPF & LPF
- Due to narrow band fading phenomenon, a single channel may get affected in FDM
- Best suited for analog transmission such as telephone, AM & FM radio broadcasting etc.

Frequency Division Multiplexing (FDM) – (e) Disadvantages of FDM

- Communication channel needs to have high amount of bandwidth for FDM
- If number of users increases, the bandwidth available per user decreases drastically
- Expensive & complicated hardware set – up if the overall number of channels increase
- FDM suffers from the problem of cross – talk unlike in TDM (intermodulation distortion)
- All the FDM channels are affected due to the wideband fading phenomenon

Time Division Multiplexing (TDM) – (a) System Block Diagram (Tx & Rx)

please refer class notes for theoretical description



Time Division Multiplexing (TDM) – (a) System Block Diagram (Tx & Rx)

- Multiplexer here is a commutator, which is a rotating switch at 'f_s' rotations per second
- This rotating switch makes contact with one of the input signal (channels) as shown
- These samples extracted from the switching action are then pulse modulated (PAM)
- These pulses are then transmitted through a channel & received by the de-commutator
- Received pulses are then sent back to their original channels due to synchronization

Time Division Multiplexing (TDM) – (b) Advantages of TDM

- Full channel bandwidth available at all point of time, unlike FDM where it is divided
- No intermodulation distortion in TDM, unlike FDM since no separate carriers are used
- Hardware set – up for TDM is simpler & less complicated unlike in FDM
- Excellent applications in the transmission of digital data or digital pulses
- Problem of crosstalk is almost eliminated in TDM as against FDM, where it is severe

Time Division Multiplexing (TDM) – (c) Disadvantages of TDM

- Only suited for digital data applications, for analog signals – FDM has to be used
- Simultaneous transmission unlike FDM isn't possible since entire bandwidth is used up
- Due to slow narrowband fading effect, all the TDM channels may get wiped out
- Synchronization is essential for an optimum performance between transmitter & receiver
- Marker pulses required between successive time frames to distinguish between them

Comparison of FDM & TDM

- Definition & Principle
- Bandwidth Utilization
- Type of Signals
- Synchronization
- Complexity of Tx & Rx
- Crosstalk Problem
- Wideband Fading
- Narrowband Fading
- Applications (Usage)

refer class notes for comparison