Pulse Modulation

Analog Pulse Modulation

- Pulse Amplitude Modulation (PAM)
- Pulse Width Modulation (PWM)
- Pulse Position Modulation (PPM)

Digital Pulse Modulation

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

Advantage of Pulse modulation: (i) Transmitted power is no longer continuous as in CW Modulation, but pulsed in nature

(ii) Vacant time between pulse occurrence filled by interleaving/multiplexing pulse waveforms of some other Message (TDM)

Sampling Theorem

This provides a mechanism for representing a **continuous time signal** by a **discrete time signal** , taking **sufficient number of samples of signal** so that **original signal is represented in its samples completely.** It can be stated as:

(i) A band-limited signal of finite energy with no frequency component higher **than f_m Hz**, is completely described by **its sample values** which are at uniform intervals **less than or equal to $1/2f_m$** seconds apart. [$\mathbf{T_s = \frac{1}{2fm}}$]where T_s is sampling time.

(ii) **Sampling frequency** must be **equal to or higher than $2f_m$** Hz. **[$fs \geq 2fm$]**

A continuous time signal may be completely represented in samples and recovered back, if $f_s \geq 2f_m$, where f_s is sampling frequency and f_m is maximum frequency component of message signal

Proof of sampling theorem

- **Sampling** of input signal **x(t)** can be obtained by **multiplying x(t)** with an **impulse train δ(t)** of period T_s.
- The output of multiplier is a discrete signal called **sampled signal** which is represented with y(t) in the diagrams,
- **y(t)=x(t).δ(t)……(1)**

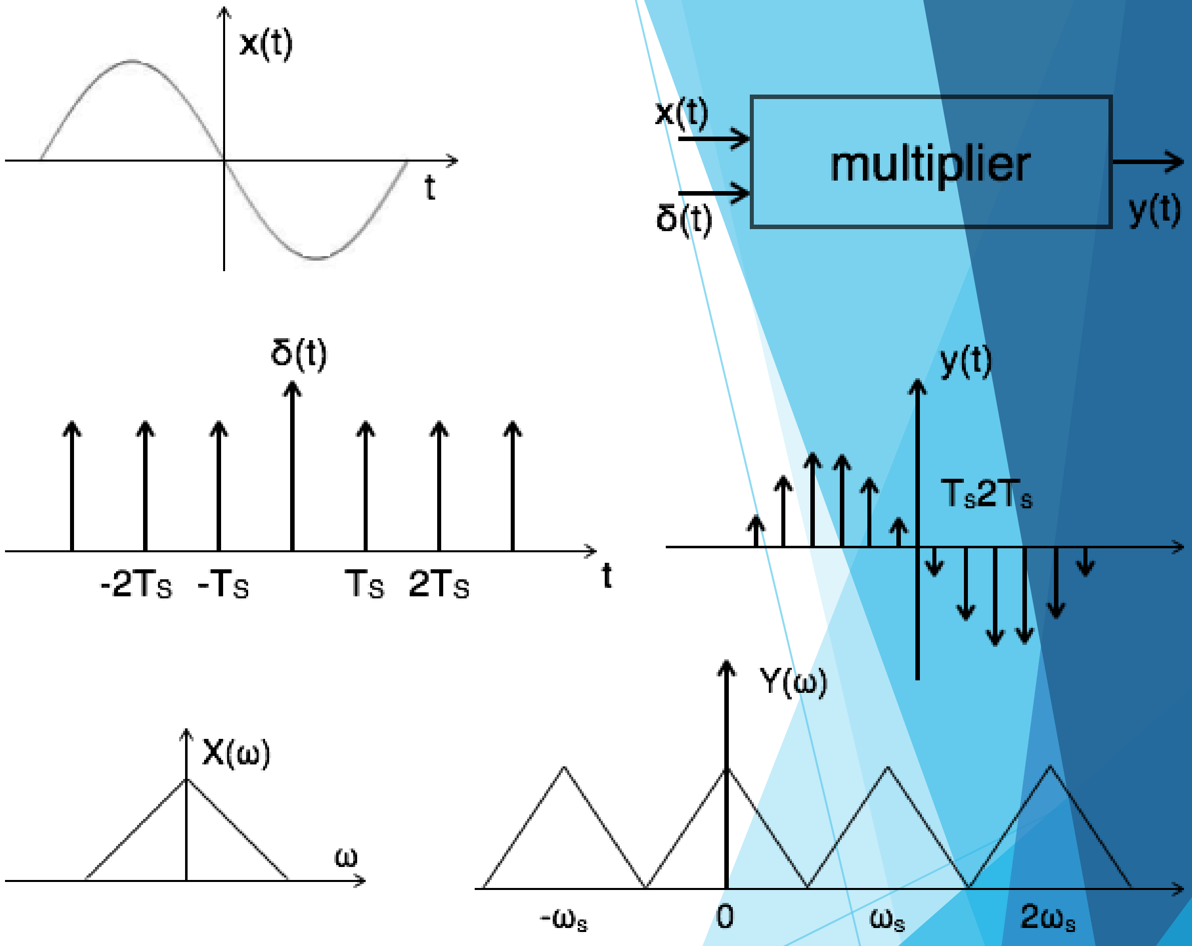
The Fourier series representation of δ(t) :

- $\boldsymbol{\delta(t)=a_0+\sum_{n=1}^{\infty}(a_n\cos n\omega_s t + b_n \sin n\omega_s t)\ldots\ldots(2)}$

where $a_0=\frac{1}{T_s}\int_{-T/2}^{T/2}\delta(t)\,dt = \frac{1}{Ts}\delta(0) = \frac{1}{T_s}$

$a_n=\frac{2}{T_s}\int_{-T/2}^{T/2}\delta(t)\cos n\omega_s dt = \frac{2}{T_s}\delta(0)\cos n\omega_s 0=\frac{2}{T_s}$

$b_n=\frac{2}{T}\int\delta(t)\sin n\omega_s t\,dt = \frac{2}{T}\delta(0)\sin n\omega_s 0=0$



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

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

Substitute δ(t) in equation 1.

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

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

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

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

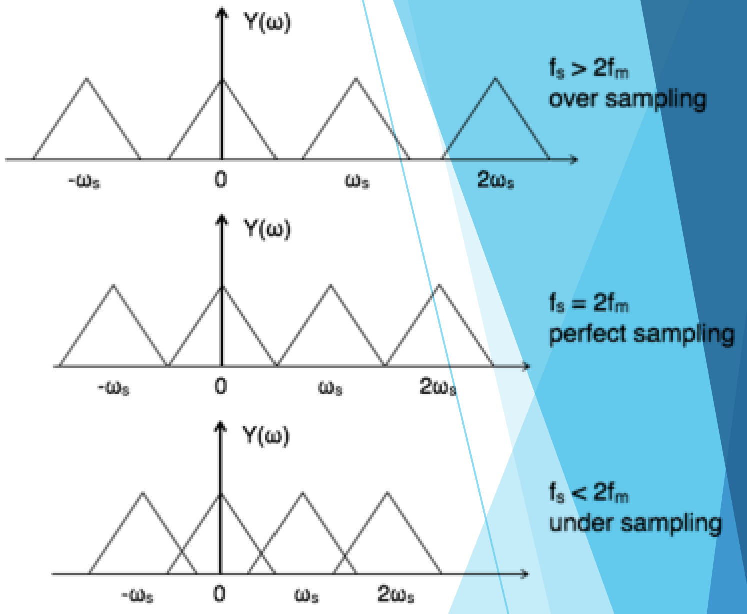
Take Fourier transform on both sides.

$$Y(\omega)=\frac{1}{T_s}[X(\omega)+X(\omega-\omega_s)+X(\omega+\omega_s)+X(\omega-2\omega_s)+X(\omega+2\omega_s)+X(\omega+3\omega_s)+..........]$$

$$Y(\omega)=\frac{1}{T_s}\sum_{-\infty}^{+\infty} X(\omega - n\omega_s)$$

To reconstruct x(t), one has to 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) which is possible if**

$f_s \geq 2f_m$



For **$f_s=2f_m$**, is known as **Nyquist rate.**

$T_s = \frac{1}{2fm}$ **is known as Nyquist interval**

<u>Aliasing Effect</u>

The overlapped region in case of **under sampling**

represents **<u>Aliasing effect</u>**. It can be termed as "**the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a lower-frequency component in the spectrum of its sampled version.**

This effect can be removed by considering

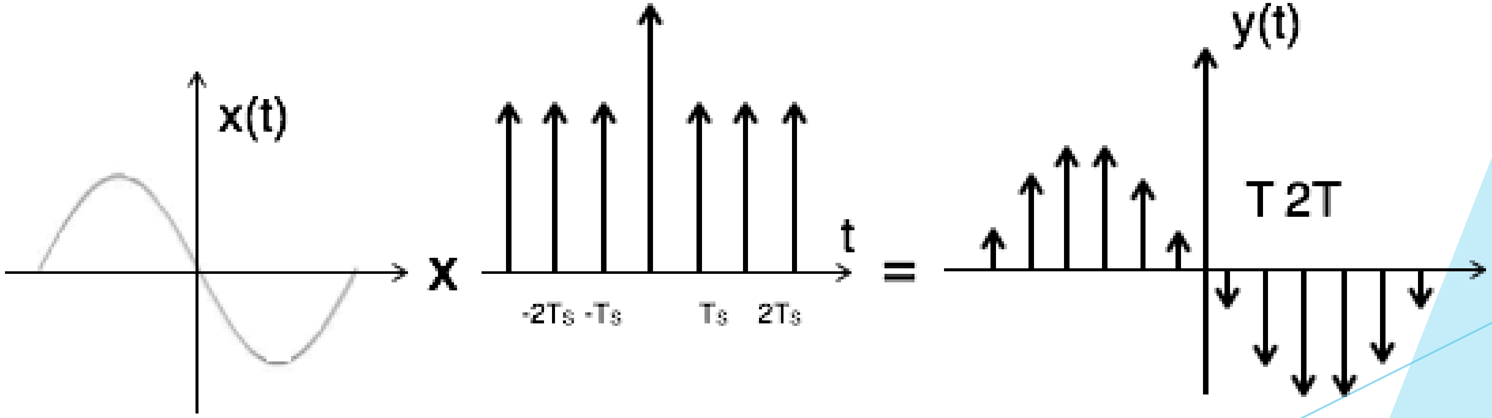
(i) fs $>2f_m$ or

(ii) by using anti aliasing filters which are low pass filters and eliminate high frequency components

Three types of **sampling techniques**:

➢ **<u>Impulse sampling</u>:** Obtained by **multiplying input signal x(t) with impulse train of period 'T_s**.

Also called **ideal sampling**. Practically not used because pulse width cannot be zero and the generation of impulse train not possible.



Natural sampling

- This type of sampling similar to ideal sampling except for the fact that **instead of delta function**, now we use rectangular train of **period T_s**. i.e. multiply input signal x(t) to pulse train
- An **electronic switch** is used to periodically shift between the two contacts at a rate of **f_s = (1/T_s) Hz**, staying on the input contact for C seconds and on the grounded contact for the remainder of each sampling
- The output $x_s(t)$ of the sampler consists of segments of x(t) and hence $X_s(t)$ can be considered as the product of x(t) and sampling function s(t).
- **$X_s(t) = x(t) \times s(t)$**

$$S(t) = \begin{cases} 1 & -\tau/2 < t < \tau/2 \\ 0 & \tau/2 < |t| < Ts/2 \end{cases} \quad \text{-------} \ (1)$$

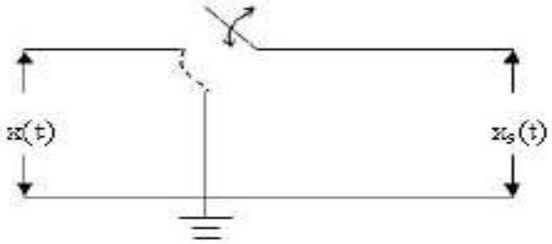


Fig: 2.11 Natural Sampling – Simple Circuit

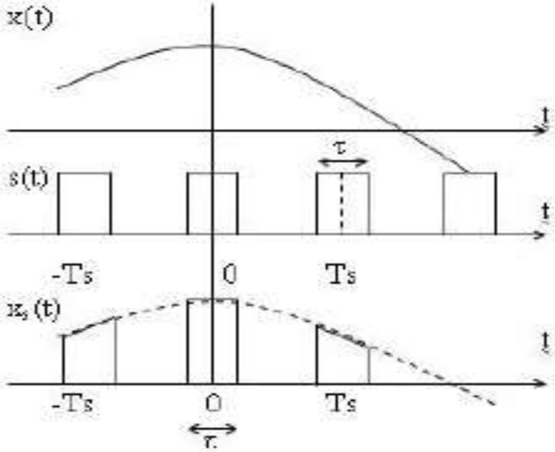
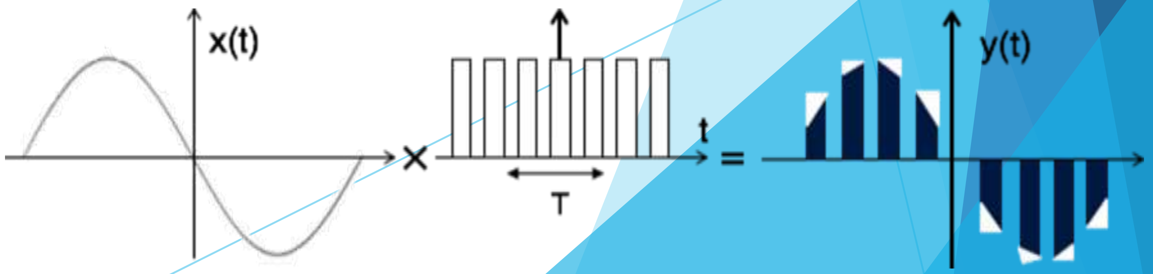


Fig: 2.12 Natural Sampling – Waveforms



Using Fourier series, we can rewrite the signal S(t) as:

$$S(t)= C_0+\sum_{n=1}^{\infty} 2Cn\cos(n\omega_s t)$$

Where the Fourier coefficients $\boldsymbol{C_0=\frac{\tau}{T_s}}$ **and** $\boldsymbol{C_n=f_s\tau \text{sinc}(nf_s\tau)}$

Therefore: $x_s(t)=x(t)[C_0+\sum_{n=1}^{\infty} 2Cn(\cos n\omega_s t)]$

$$\boldsymbol{x_s(t)=C_0x(t)+2C_1x(t)\cos(\omega_s t)+2C_2x(t)\cos(2\omega_s t)+\ldots\ldots}$$

Applying Fourier Transform for the above equation

Using $\boldsymbol{x(t)\leftrightarrow X(f)}$

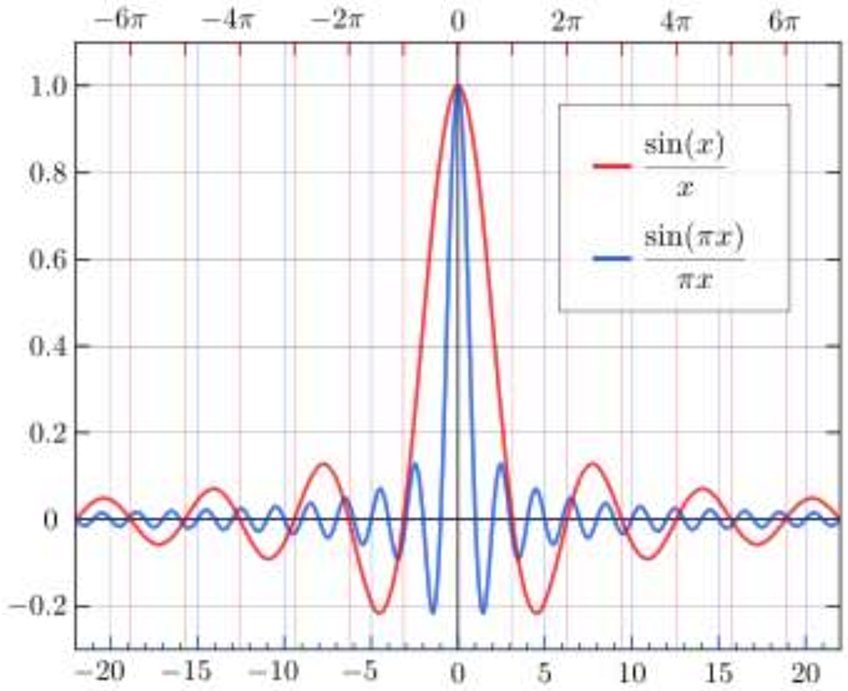
$$\boldsymbol{x(t)\cos(2\pi f_0 t)\leftrightarrow \frac{1}{2}[X(f-f_0)+X(f+f_0)]}$$

$$\boldsymbol{X_s(f)=C_0X(f)+C_1[X(f-f_0)+X(f+f_0)]+C_2[X(f-f_0)+X(f+f_0)]+\ldots\ldots\ldots\ldots}$$

$$\boldsymbol{X_s(f)= C_0X(f)+\sum_{n=-\infty}^{\infty} C_nX(f-nfs)}$$

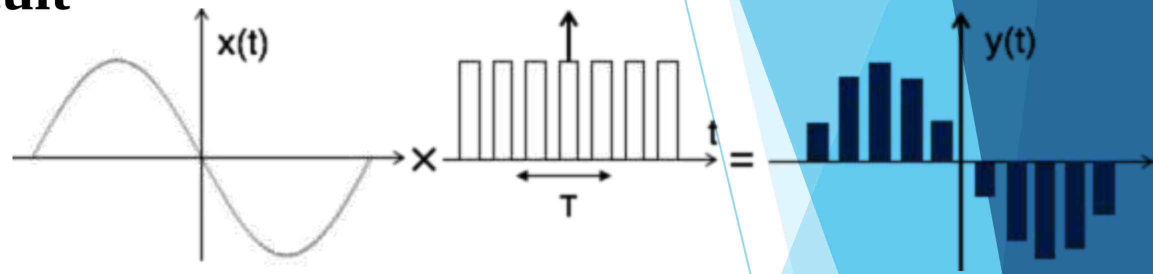
$$\boldsymbol{X_s(f) = A\tau/ Ts .[\Sigma \sin c(n\ fs.\tau)\ X(f-n\ fs)]}$$

The signal $X_s(t)$ has the spectrum which consists of message spectrum and repetition of message spectrum periodically in the frequency domain with a period of fs. But **the message term is scaled by ‘Co”(** **sinc function**) which is **not the case in instantaneous sampling.**

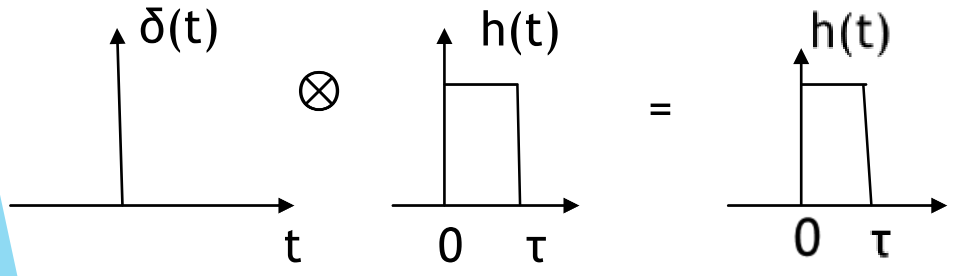


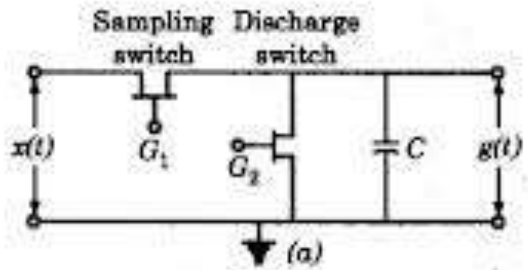
-6π -4π -2π 0 2π 4π 6π
1.0
0.8
0.6
0.4
0.2
0
-0.2
-20 -15 -10 -5 0 5 10 15 20
$\frac{\sin(x)}{x}$
$\frac{\sin(\pi x)}{\pi x}$

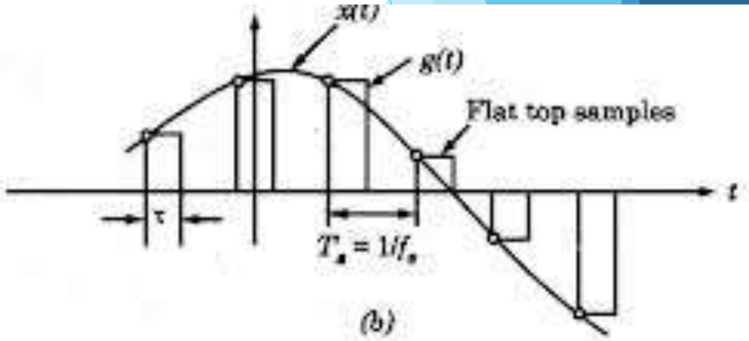
- **<u>Flat Top sampling</u>:** During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the **pulse is in the form of flat top**.
- Here, the top of the samples are flat i.e. they have constant amplitude and is equal to the instantaneous value of the baseband signal x(t) at the start of sampling. Hence, it is called as **<u>flat top sampling or practical sampling</u>**.
- Flat top sampling makes use of **sample and hold circuit**
- Theoretically, **the sampled signal can be obtained by convolution of rectangular pulse h(t) with ideally sampled signal ,$s_\delta(t)$**



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







$f(t) \otimes \delta(t) = f(t)$; property of delta function

Applying a modified form; **s(t) in place of δ(t)**

On convolution of s(t) and h(t), we get a pulse whose duration is equal to h(t) only but amplitude defined by s(t).

Train of impulses given by:

$\boldsymbol{\delta_{Ts}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nTs)}$

Signal s(t) obtained by multiplication of message signal x(t) and δ_{Ts}(t)

Thus, **s(t) = x(t). $\boldsymbol{\delta_{Ts}(t)}$**

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

Now sampled signal g(t) given as:

$$g(t)=s(t) \otimes h(t) \qquad G(f)=S(f)\,H(f)$$

$$=\int_{-\infty}^{\infty} s(\tau)h(t-\tau)d\tau \qquad S(f)=fs\sum X(f-nfs)$$

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

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

Using shifting property of delta function: $\int_{-\infty}^{\infty} f(t)\delta(t-to)=f(t_0)$

$\mathbf{g(t)}=\boldsymbol{\sum_{-\infty}^{\infty} x(nT_s)h(t-nTs)}$

$\mathbf{G(f)}=\boldsymbol{f_s\sum_{-\infty}^{\infty} X(f-nfs)H(f)}$ **Spectrum of flat top samples**

<u>Aperture Effect</u>: Spectrum of flat topped sample is given by;

G(f)=f_s $\sum[\![\boldsymbol{X(f-nf_s)H(f)}]\!]$, where **H(f)= τ.sin c(f_s.t)$\boldsymbol{e}$^($\boldsymbol{-j\pi f\tau}$)**

This equation shows that signal g(t) is obtained by passing the **signal s(t) through a filter having transfer function H(f).**

Figure(a) shows **one pulse of rectangular pulse train** and **each sample of x(t) i.e. s(t)** is **convolved** with **this pulse**

Figure (b) shows the **spectrum of this pulse**. Thus, flat top sampling introduces an **amplitude distortion** in **reconstructed signal x(t) from g(t**). There is a **high frequency roll off** making **H(f) act like a LPF**, thus **attenuating the upper portion of message signal spectrum**. This is known as **<u>aperture effect</u>**

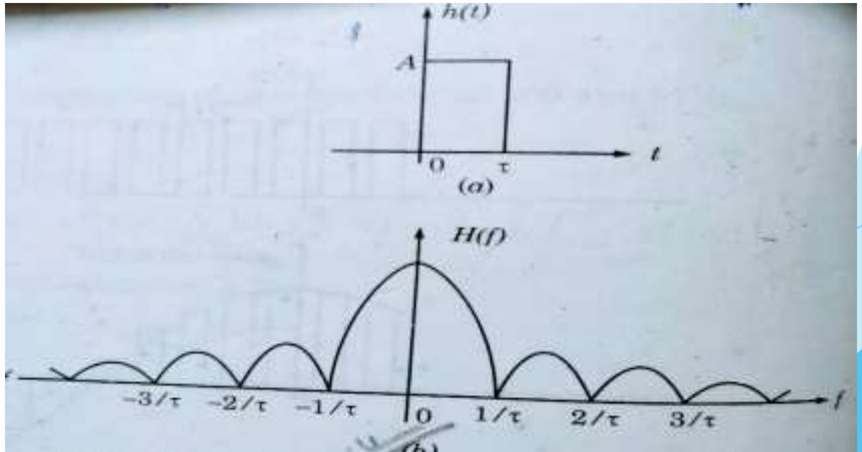
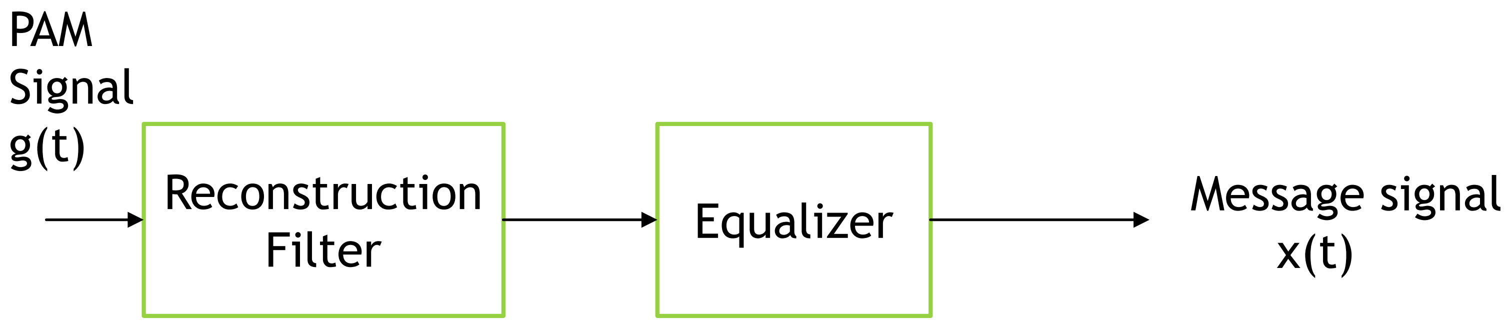


Fig. 3.18 (a) One pulse of rectangular pulse train
(b) spectrum of the pulse shown in figure (a).

<u>How to minimize aperture effect</u>?? An **equalizer** at the receiver end is needed to compensate aperture effect. The receiver contains low pass reconstruction Filter with cut off slightly higher than f_m Hz.



Equalizer in cascade with reconstruction filter has the effect of decreasing the in band loss of reconstruction filter, **frequency increases in such away so as to compensate aperture effect.**

$$\mathbf{H_{eq}(f)} = \frac{K.e^{-j2\pi ftd}}{H(f)},$$

where t_d is time delay introduced by LPF being equal to $\tau/2$

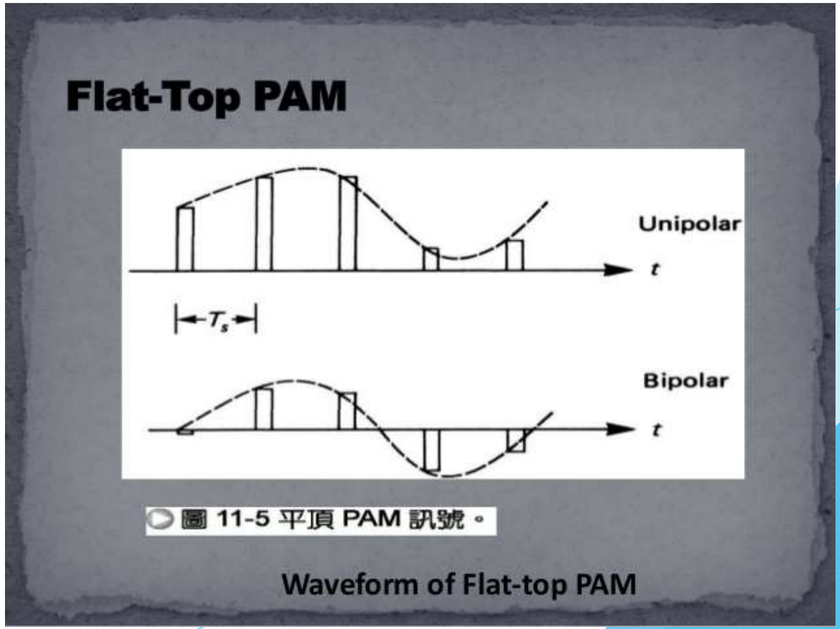
$$\mathbf{H_{eq}(f)} = \frac{K}{\tau \sin c(f\tau)}$$

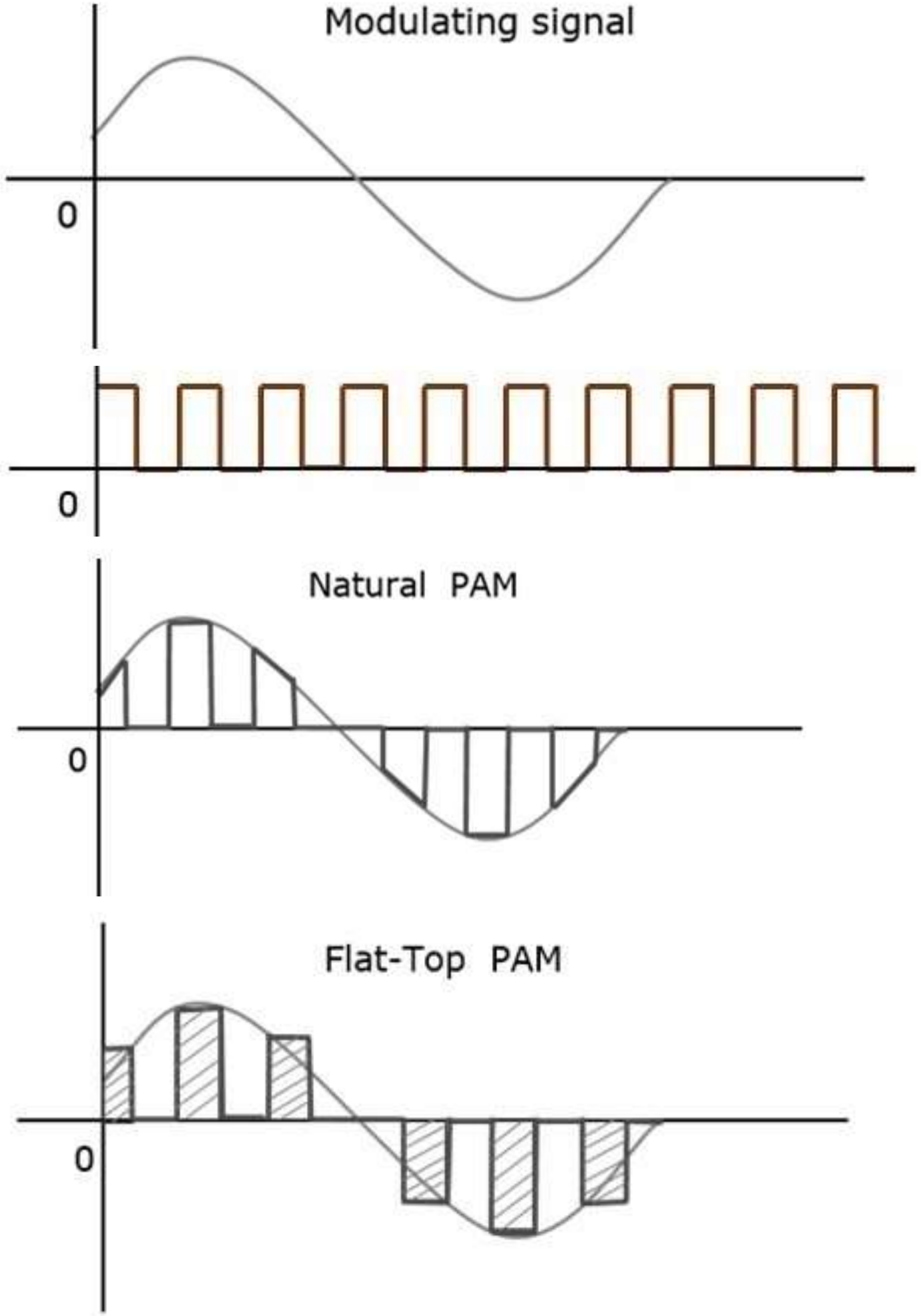
Pulse Amplitude Modulation (PAM)

- Amplitude of **the pulse carrier** varies proportional to the **instantaneous amplitude of the message signal.**
- The **width and positions** of the pulses are **constant** in this modulation.
- **PAM could be:**

(i)Single polarity PAM: A suitable **fixed DC bias is added to the signal** to ensure that all the pulses are positive.

(ii) Double polarity PAM: In this the pulses are both positive and negative.




Modulating signal
0
0
Natural PAM
0
Flat-Top PAM
0

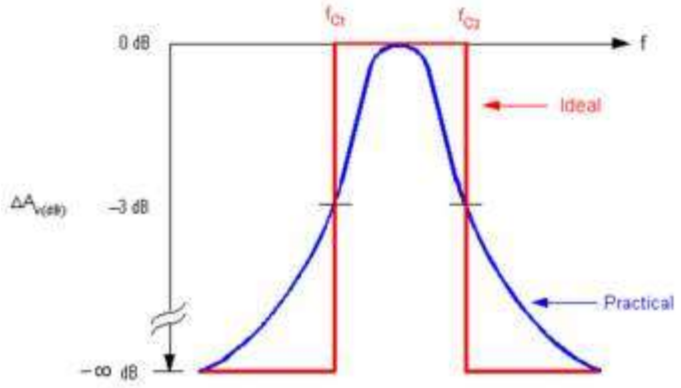
- Depending on type of sampling **PAM can be**:

  (i) Ideal Sampling PAM, (ii) Natural sampling PAM and (iii) Flat top PAM.

- The **advantage** of this modulation is the **generation and detection** is **easy** in this modulation and also **allows multiplexing.**

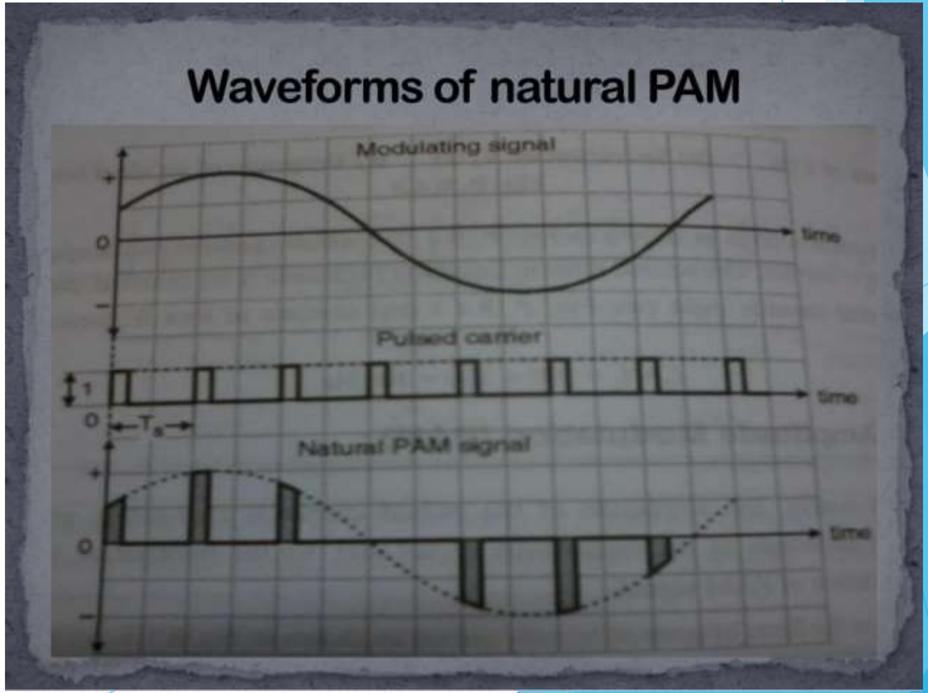
- The **disadvantage** is **large band width** of transmitted signal.

BPF characteristics



- For a **PAM signal** produced with **natural sampling**, the **sampled signal follows the waveform of the input signal** during the time that each sample is taken.
- A PAM signal is generated by using **a pulse train**, called the **sampling signal** (or clock signal) to operate an **electronic switch or "chopper"**. This produces **samples of the analog message signal**.
- The **switch is closed** for the duration of **each pulse**, allowing the **message signal** at that **sampling time to become part of the output.**
- The **switch is open for the remainder of each sampling period making the output zero**. This is known as **Natural PAM**.

In simplest form **PAM** can be visualized as **o/p of an AND gate** whose **two inputs** are **message signal x(t)** and **pulses at sampling rate**



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- For **flat-top sampling**, a sample-and-hold circuit is used in conjunction with the chopper to hold the amplitude of each pulse at a constant level during the sampling time,
- **Flat-top sampling**, produces **pulses whose amplitude remains fixed** during the sampling time. The **amplitude value** of the pulse depends on **the amplitude of the input signal at the time of sampling.**
- **Aperture Effect** seen in this type of **PAM. Equalizers** used at **receiver end**

Transmission Bandwidth in PAM

$\tau \ll T_s$

$f_s \geq 2fm\,; Ts \leq \frac{1}{2fm}$

$\tau \ll Ts \leq \frac{1}{2fm}$

If on and off time of PAM pulse is same **then** $f_{max} = \frac{1}{2\tau}$

$BW \geq fmax;\ \ BW \geq \frac{1}{2\tau}$

$BW \geq \frac{1}{2\tau} \gg fm$

An equalizer is needed to overcome this effect.

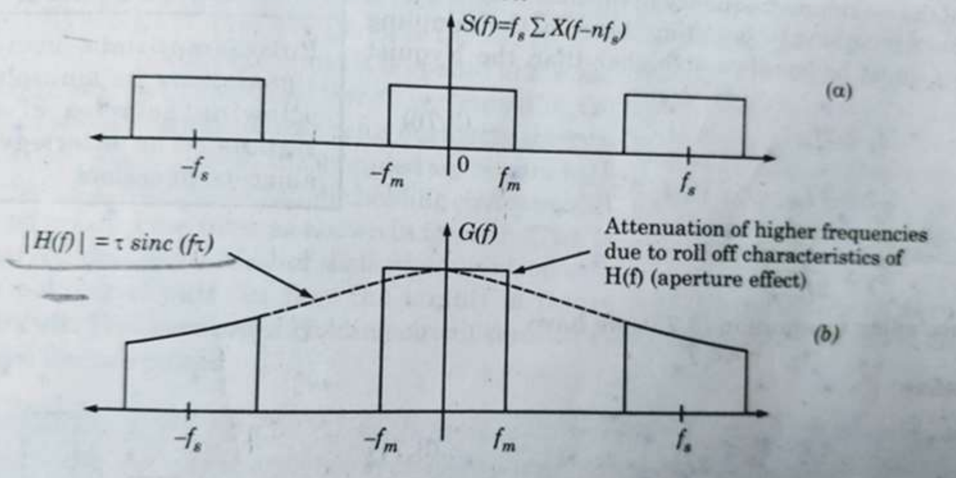


Fig. 3.25. *(a) Spectrum of some arbitrary signal. The signal is sampled at f_s and maximum frequency in the signal is f_m. (b) Spectrum of flat top signal. The dotted curve is $H(f) = t\ sin\ c(f\tau)$*

Transmission of PAM signals

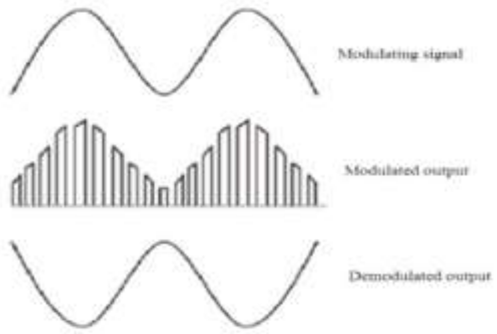
- For PAM signals to be transmitted through space using antennas, they must be **amplitude/ frequency/ phase** modulated by a **high frequency carrier** and only then they can be transmitted. Thus the **overall system is PAM-AM. PAM-FM or PAM-PM** and at receiving end, AM/ FM/PM detection is first employed to get the PAM signal and then message signal is recovered.

Drawbacks of PAM

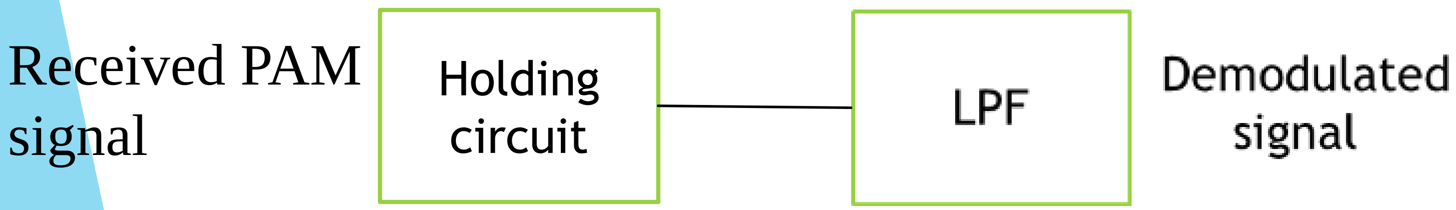
- Bandwidth required for transmission of PAM signal is very large in comparison to maximum frequency present in modulating signal.
- Since amplitude of PAM pulses varies in accordance with modulating signal so interference of noise is maximum in PAM
- Variation of the peak power required by transmitter

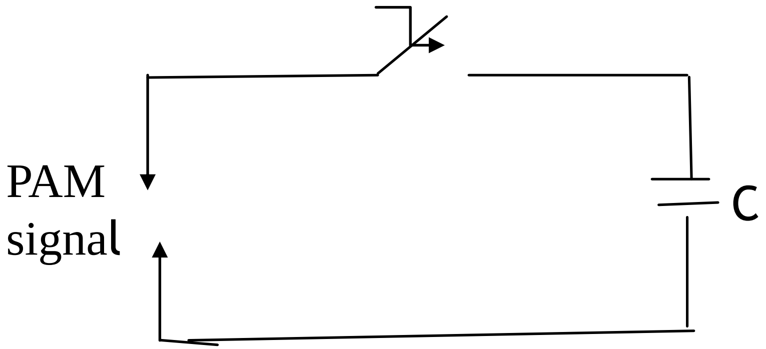
Demodulation of PAM

- PAM signal sampled at Nyquist rate can be **reconstructed at the receiver end** , by passing it through an efficient **Low Pass Filter (LPF)** with exact cut off frequency of **fs/2**. This is known as **Reconstruction or Interpolation Filter**.
- The low pass filter eliminates the high-frequency ripples and generates the demodulated signal. This signal is then applied to the inverting amplifier to amplify its signal level to have the demodulated output with almost equal amplitude with the modulating signal

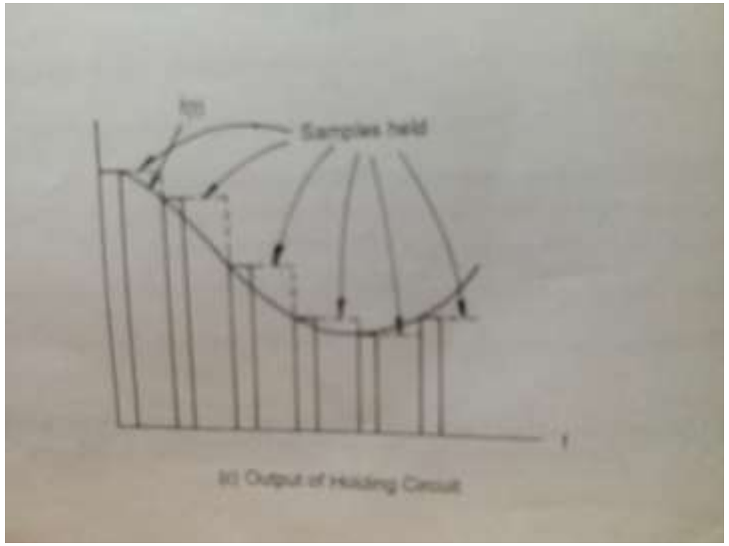


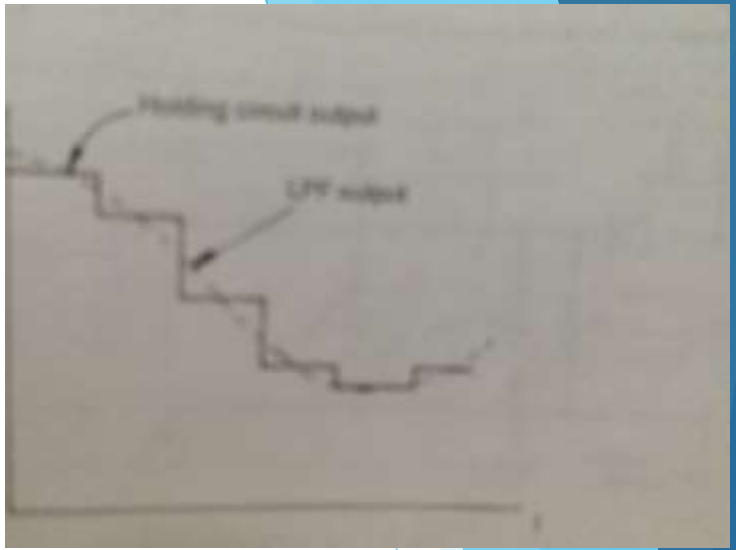
- For a **flat topped PAM**, a **holding circuit** followed by a **LPF** gives demodulated signal





Zero order Holding Circuit





- Switch S closes after the arrival of pulse and opens at the end of pulse.
- Capacitor C charges to pulse amplitude value and holds this value during interval between two pulses.
- The sampled values are shown in fig.
- Holding circuit o/p smoothened in LPF.
- Known as zero order holding circuit, which considers only the previous sample to decide value between two pulses
- First order holding circuit considers previous two samples, second order holding circuit considers previous three samples.

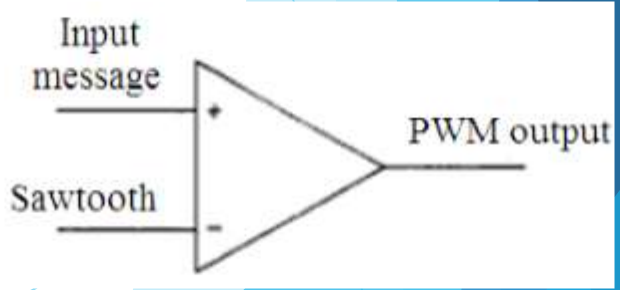
Pulse Time modulation

- In **PTM**, **amplitude of pulse is constant** while **position or width of pulse** is made proportional to the **amplitude of the signal** at the **sampling instant**.
- It can be PWM and PPM
- In both the cases amplitude constant and does not carry information so amplitude limiters can be used (like in FM) providing good noise immunity

Generation of PTM signals can be either by:

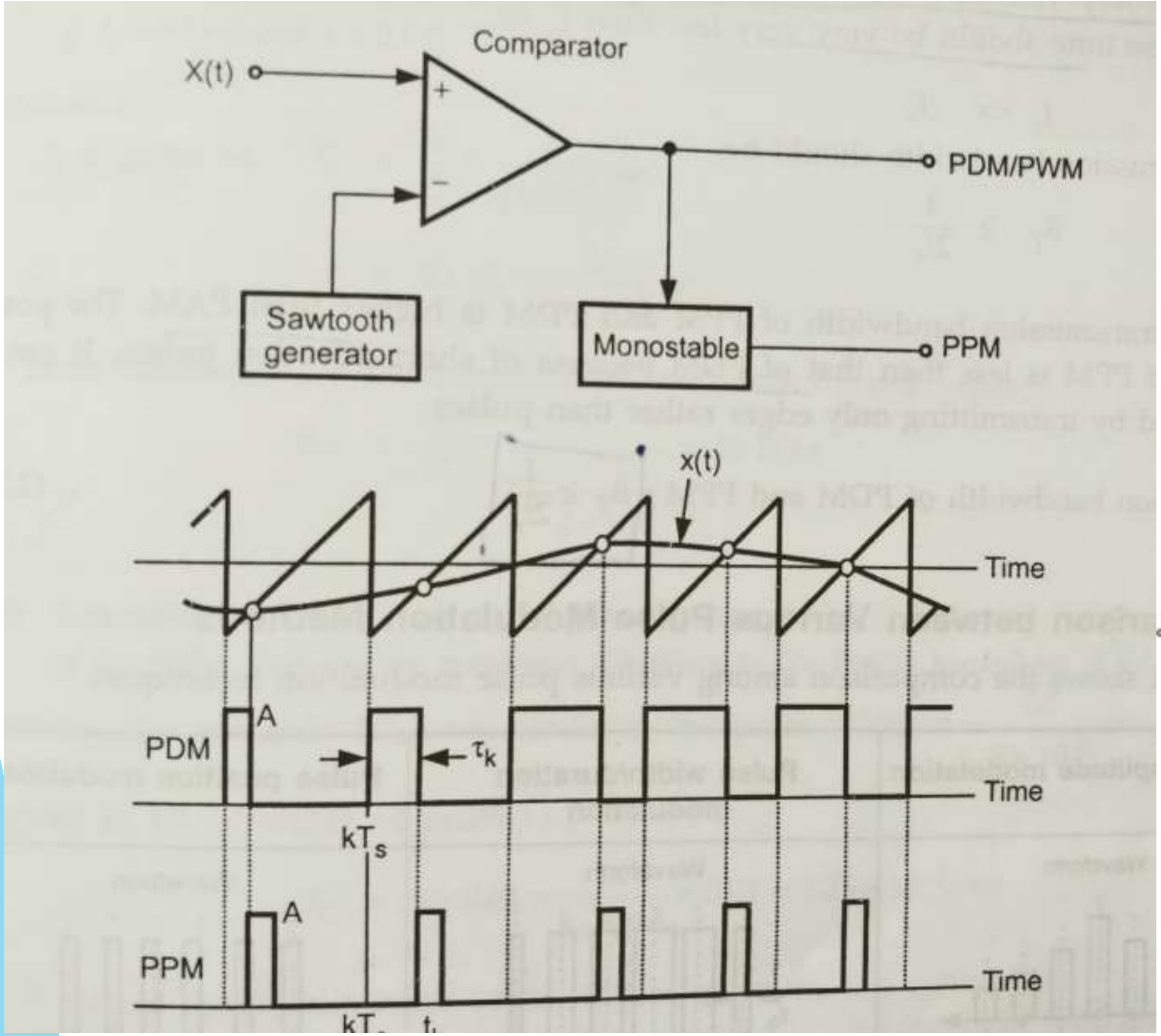
(i) **Indirect Method**: Firstly PAM signals are generated, Synchronized is generated during each pulse interval. These two signals are added and the sum is applied to a comparator whose reference level is suitably chosen. The second crossing of comparator level used for PPM

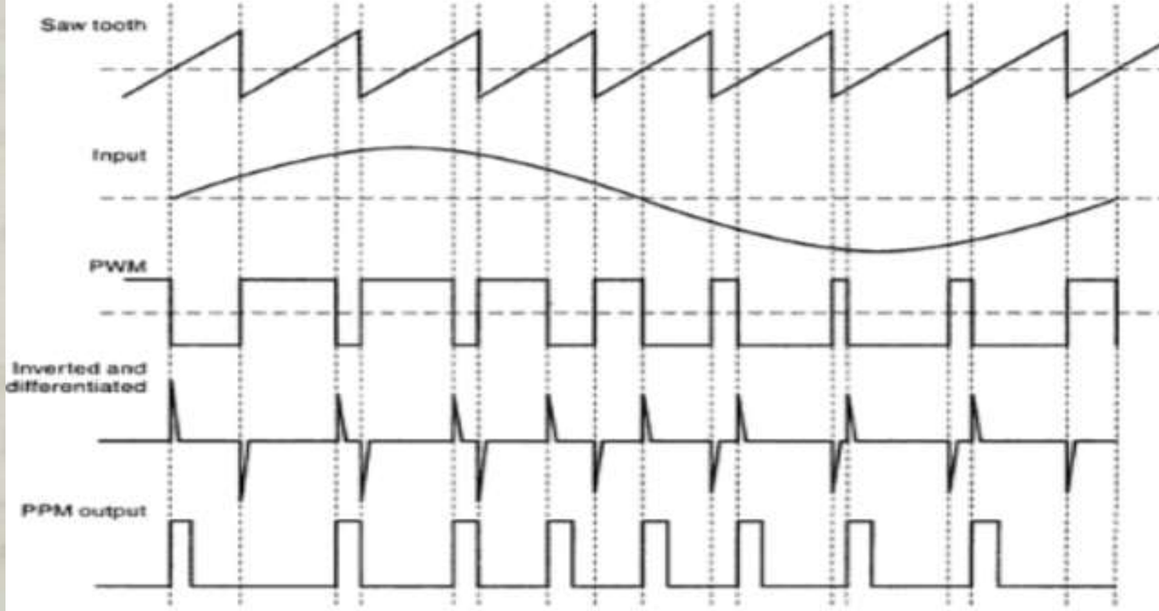
(ii) **Direct method**: PTM waveforms generated without using PAM waveforms



Pulse Width modulation

The pulse width modulation is the modulation of signals by varying the width of pulses. The amplitude and positions of the pulses are constant in this modulation



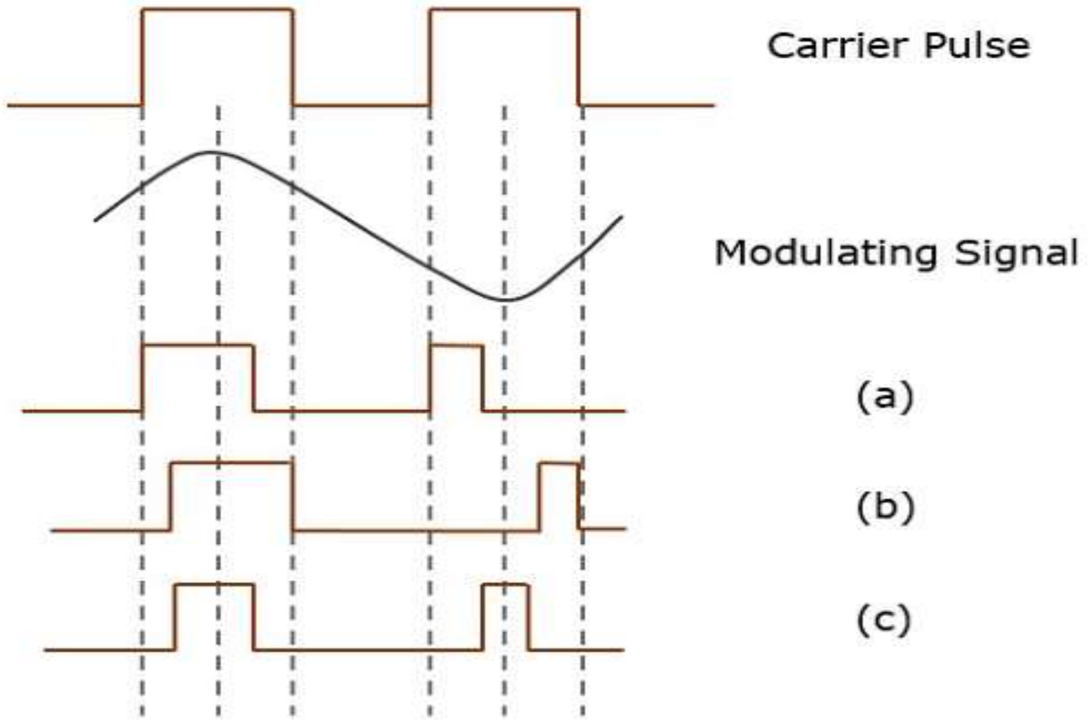


Generation of PWM and PPM by Direct Method

- **The non inverting input** of the **comparato**r is fed by the **input message or modulating signal x(t)** and the other input by a **saw-tooth signal** which operates at carrier frequency.
- The comparator compares the two signals together to generate the PWM signal at its output. Its o/p is high only when the instantaneous value of x(t) is higher than sawtooth waveform.
- The rising edges of the PWM signal occurs at the fixed time period (kT_s) while trailing edge depends on amplitude of message signal x(t).
- When saw-tooth voltage waveform greater than x(t), o/p of comparator is zero, **trailing edge is modulated**
- If **saw-tooth. waveform is reversed**, trailing edge is fixed while **leading edge is modulated**.
- Replacing saw-tooth waveform by **triangular**, both leading and trailing edge modulated. (**symmetrical PWM**)
- The amplitude of PDM/PWM will be positive saturation of the comparator shown as ‘A’, being same for all pulses,

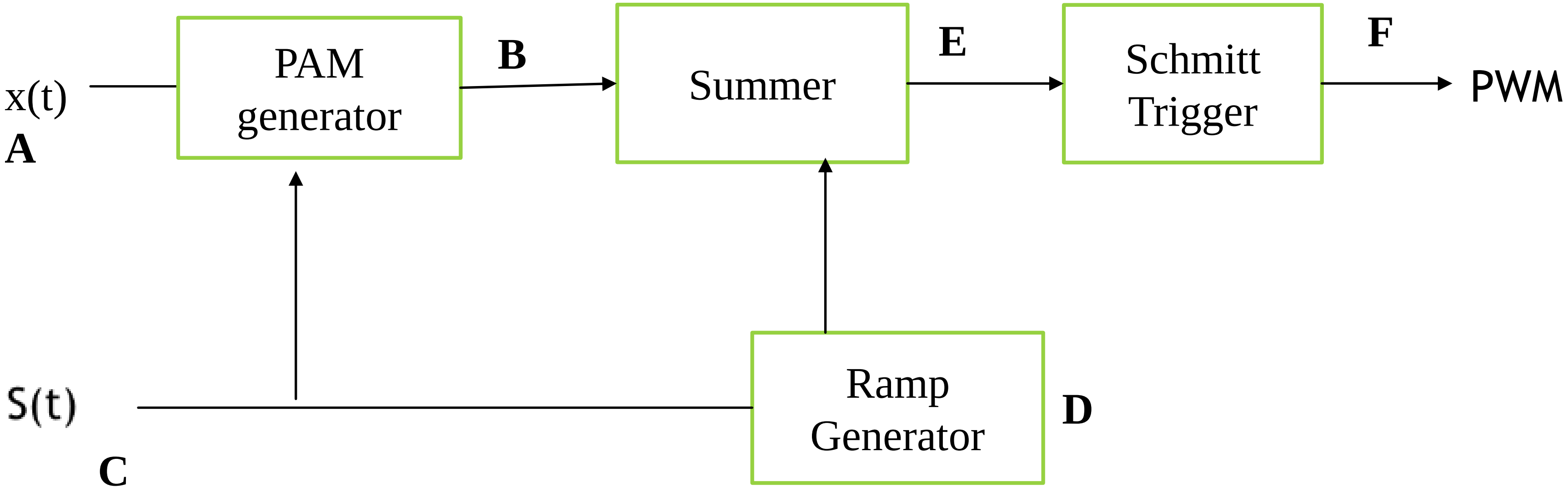
Three types of pulse-width modulation (PWM) are possible:

- The leading edge of the pulse being constant, the trailing edge varies according to the message signal.
- The trailing edge of the pulse being constant, the leading edge varies according to the message signal
- The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal (Symmetrical PWM)

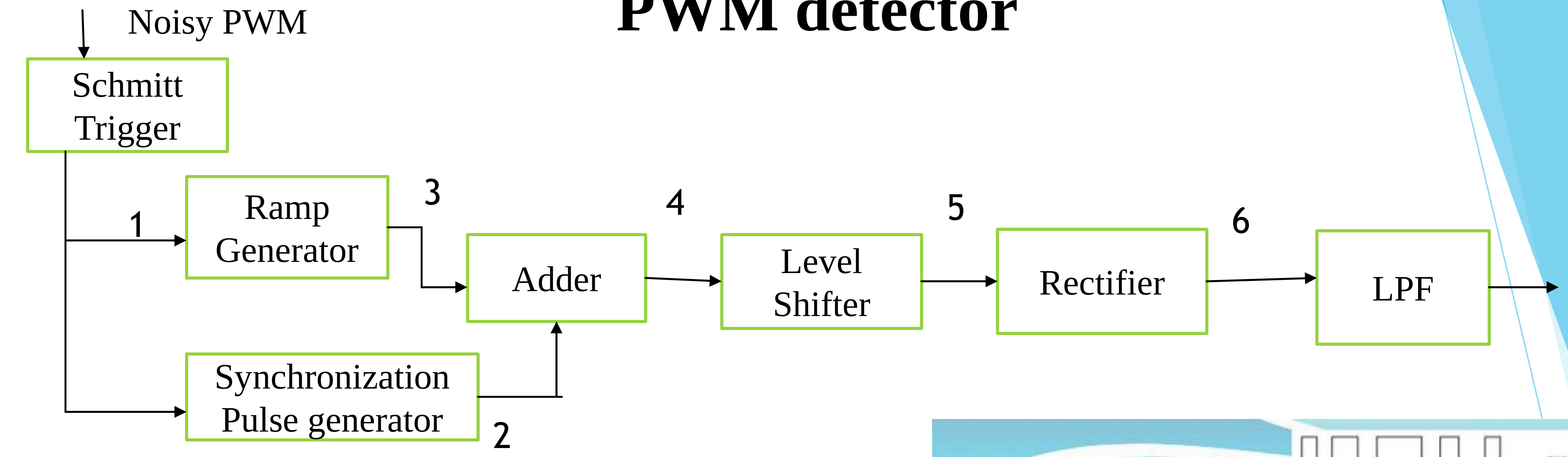


Indirect Method:

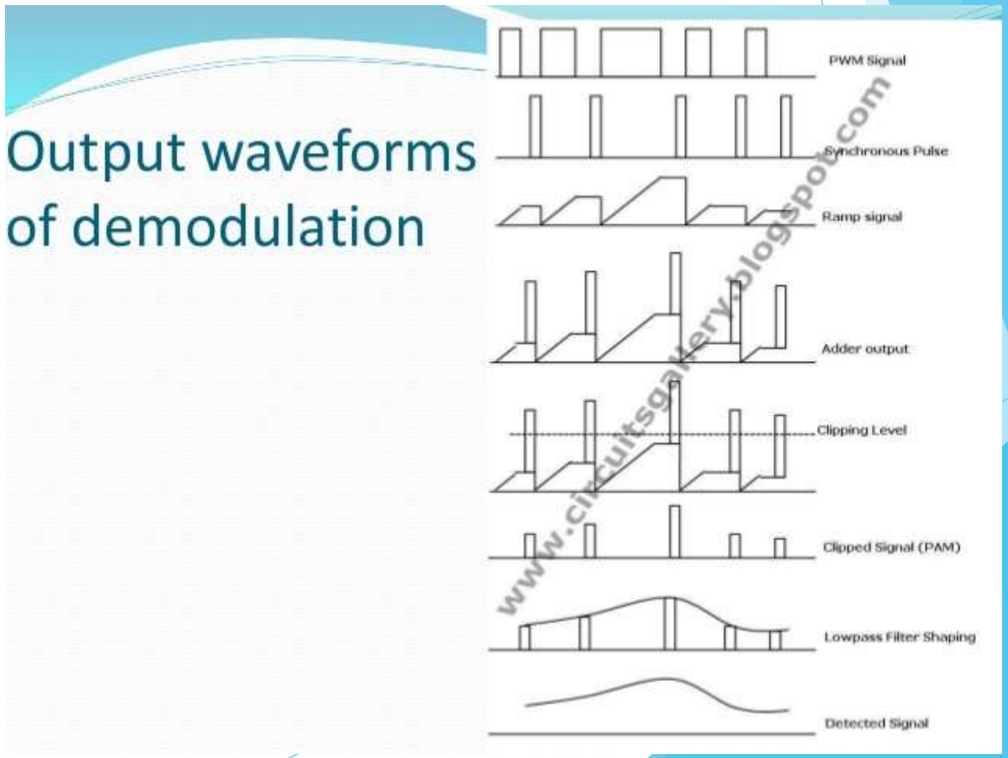
Modulating signal (A) applied to i/p of PAM circuit [s(t) pulse train] and PAM signal generated(B). S(t) also is i/p to Ramp generator(Integrator circuit), all having equal slopes, amplitude and generation(D). These ramp pulses added to PAM pulses to produce varying height samples. These varying height ramp gates a S.T ckt to generate varying width rectangular pulses of PWM.



PWM detector

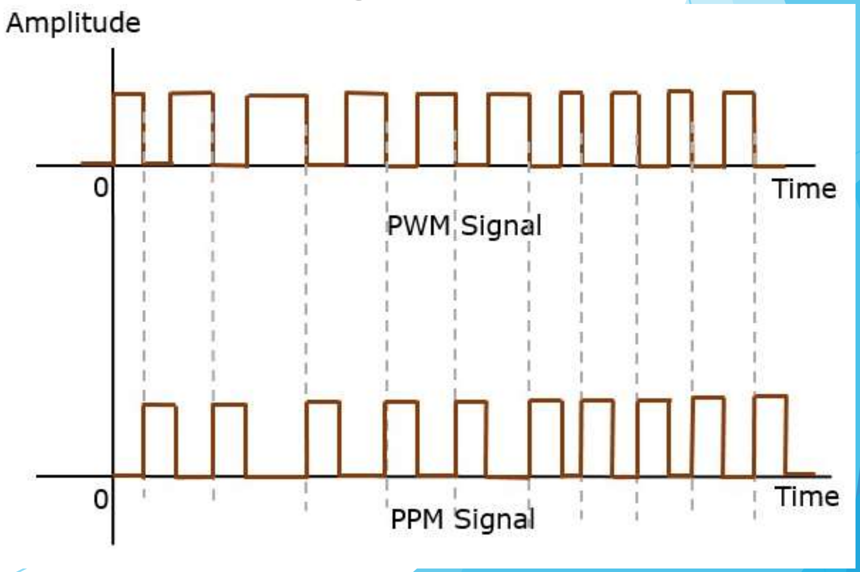


- Received PWM signal applied to ST circuit to remove noise
- Regenerated PWM applied to Ramp generator and synchronization pulse.
- Heights of Ramp proportional to width of pulses.
- Pulse generator produces reference pulses with constant amplitude and width but delayed by specific amount.
- Delayed reference pulses added to o/p of ramp generator
- The o/p given to level shifter, negative offset shifts waveform. Then clipped by rectifier followed by LPF to give message signal.



Pulse position modulation

- **(PPM)** is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the **position of each pulse**, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.
- The transmitter has to send **synchronizing pulses** (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses.
- PPM is done in accordance with the PWM signal.
- PWM signal is used as the trigger input to a monostable multivibrator.
- Its o/p remains zero until it is triggered on the trailing edge of PWM
- O/P of monostable MV switches to positive saturation value A **and remains high for fixed period** then goes low
- Hence, the position of these pulses is proportional to the width of the PWM pulses.



Advantage As the amplitude and width are constant **the power handled is constant**

Disadvantage: Synchronization between Transmitter and receiver is a necessity

- The **PDM is differentiated**, and then **rectified** and **shaped**.
- PPM carries exactly the same information as long as the position of the clock pulses (leading edge) is well defined in the received signal.
- PPM is superior to PDM for message transmission, since the wide pulses of PDM require more energy than PPM when transmitted
- PPM is suited for communication in the presence of noise.
- Very high peak narrow pulses can be transmitted and the pulse position can be determined even when the noise level is high,
- However, transmitting very narrow pulses requires a large band width
- When light is used as the media for transmitting analog signals, **PPM or PCM** are the most suitable types of modulation because **the maximum power output in the modulated light source, such as LED or LASER is achieved when it is pulsed at a very low duty cycle**.
- In PPM, necessary to transmit a series of sync pulses at a much lower repetition rate than the sampling pulses, to avoid interference with original signal and/or minimise the number of pulses transmitted in order to conserve transmission power

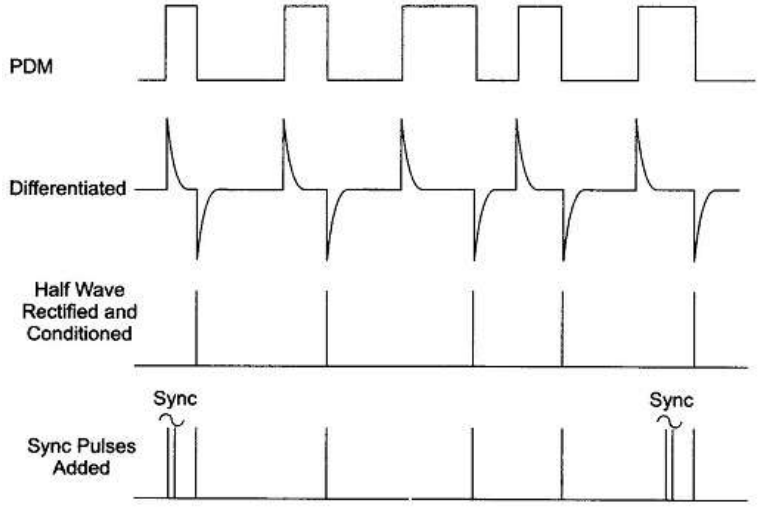


Fig. 18.14 Generation of PPM from PDM (Note: Sync Pulses are Double for Recognition)

Transmission BW of PWM and PPM

- Both PWM and PPM have DC value.
- Both need a shrp rise time and fall time to preserve the message information
- Rise time be very less than Ts i.e. $\mathbf{t_r \ll Ts}$
- **Transmission BW:** $\mathbf{B_T \geq \frac{1}{2tr}}$
- BW higher than PAM

PAM

- The amplitude of the pulse is proportional to the amplitude of modulating the signal.
- **Band width** of transmitting channel depends on the **width of the pulse**
- Instantaneous power of transmitter varies. Noise interference is high
- Complex system. Similar to A.M.

PWM

- Width of pulse is proportional to amplitude of modulating signal.
- The **Bandwidth of transmitting channel** depends on **rise time of the pulse**.
- Instantaneous power of transmitter varies. Noise interference is minimum.
- Simple to implement   Similar to F.M.

PPM

- Relative position of pulse is proportional to amplitude of modulating signal.
- The bandwidth of transmitting channel depends on the rise time of the pulse.
- Instantaneous power remains constant. Noise interference is minimum.
- Simple to implement. Similar to P.M.

Difference Between PAM, PWM, and PPM

Parameter	PAM	PWM	PPM
Type of Carrier:	Train of Pulses	Train of Pulses	Train of Pulses
Variable Characteristic :	Amplitude	Width	Position
Bandwidth Requirement:	Low	High	High
Noise Immunity :	Low	High	High
Information Contained in:	Amplitude Variations	Width Variations	Position Variations
Power efficiency (SNR)	Low	Moderate	High
Transmitted Power	Varies	Varies	Remains Constant
Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
Bandwidth	depends on width of the pulse	rise time of the pulse	rise time of the pulse
Transmitter power	Inst. power varies with amplitude of pulses	Instantaneous power varies with width of the pulses	Constant
Complexity of generation and detection	Complex	Easy	Complex
Similarity with other Modulation Systems	Similar to AM	Similar to FM	Similar to PM

Q. For a PAM transmission of voice signal with f_m=3kHz, calculate the transmission BW. Given that f_s=8kHz and the pulse duration $\tau=0.1T_s$

Soln: $Ts=\frac{1}{fs}$= 125μs

τ=0.1Ts=0.1×125=12.5μs

$\mathbf{BW \geq \frac{1}{2\tau} \geq 40\ kHz}$

Q. For the above signal if rise time is 1% of pulse width, find minimum Tx BW for PWM and PPM?

Soln: $t_r=\tau\times 0{,}01= 1.25\text{x } 10^{-7}$

$B_T \geq \frac{1}{2t_r} \geq \mathbf{4MHz}$

Thus BW of PWM/PPM much higher than PAM

Multiplexing

- Multiplexing refers to the **combination of information streams from multiple sources** for **transmission over a shared medium .**
- Multiplexor is a mechanism that implements the concept. It permits hundreds or even thousands of signals to be combined and transmitted over a single medium. De-multiplexing refers to the separation of a combination, back into separate information streams .

Principle used

- Each sender communicates with a single receiver
- All pairs share a single transmission medium
- Multiplexor combines information from the senders for transmission in such a way that the de multiplexer can separate the information for receivers.
- Cost savings obtained using single channel to send Multiple signals.

The Concept of Multiplexing

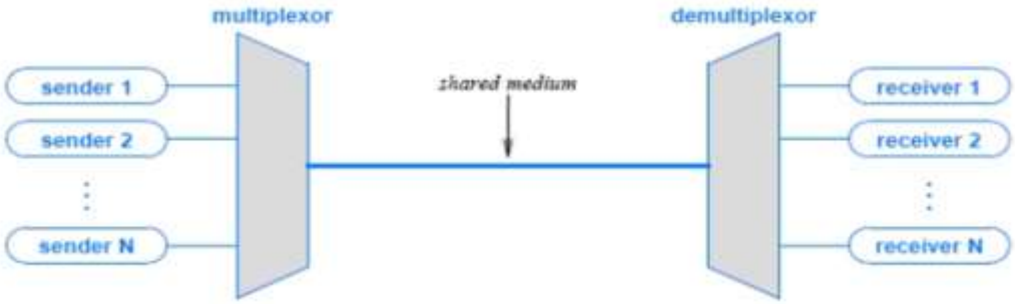


Figure 11.1 The concept of multiplexing in which independent pairs of senders and receivers share a transmission medium.

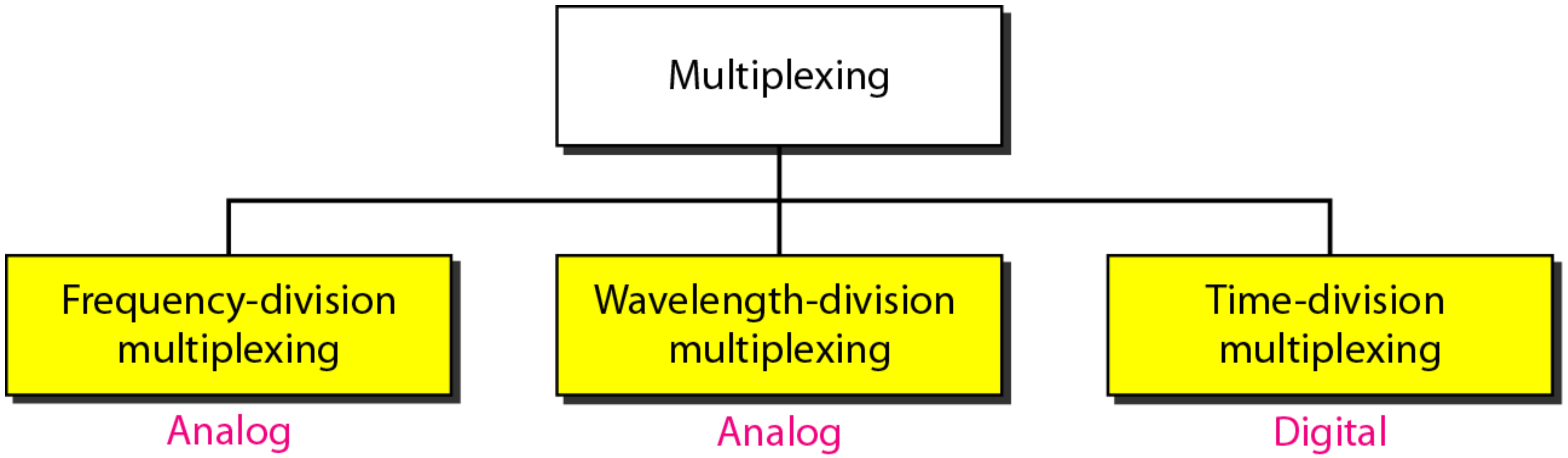
Four basic types of multiplexing

•Frequency Division Multiplexing (FDM)

•Wavelength Division Multiplexing (WDM)

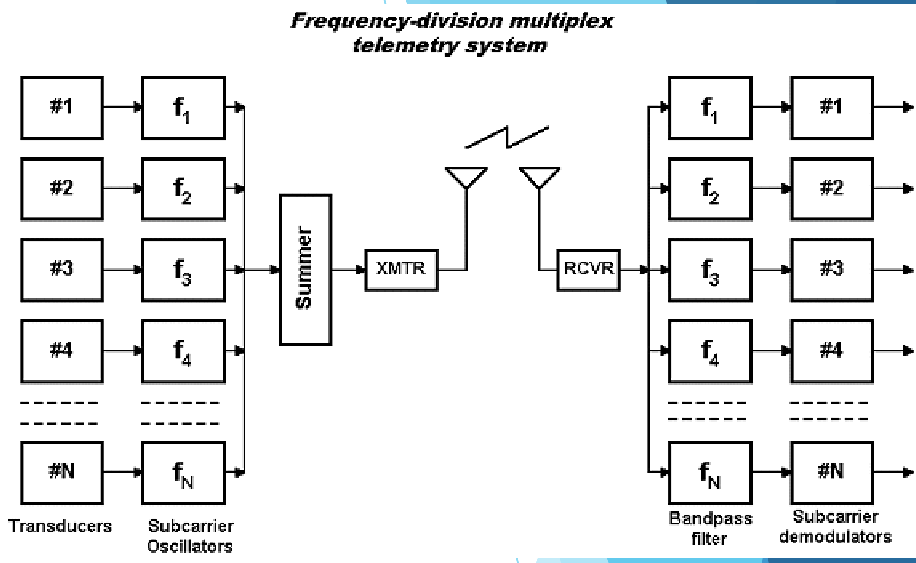
•Time Division Multiplexing (TDM) •

Code Division Multiplexing (CDM)



Frequency Division Multiplex (FDM): **Separation of spectrum into smaller frequency**.

- Channel gets band of the spectrum for the whole time. Each signal allocated different frequency band i.e, Multiple carriers used
- Each message signal is limited to f_m Hz.



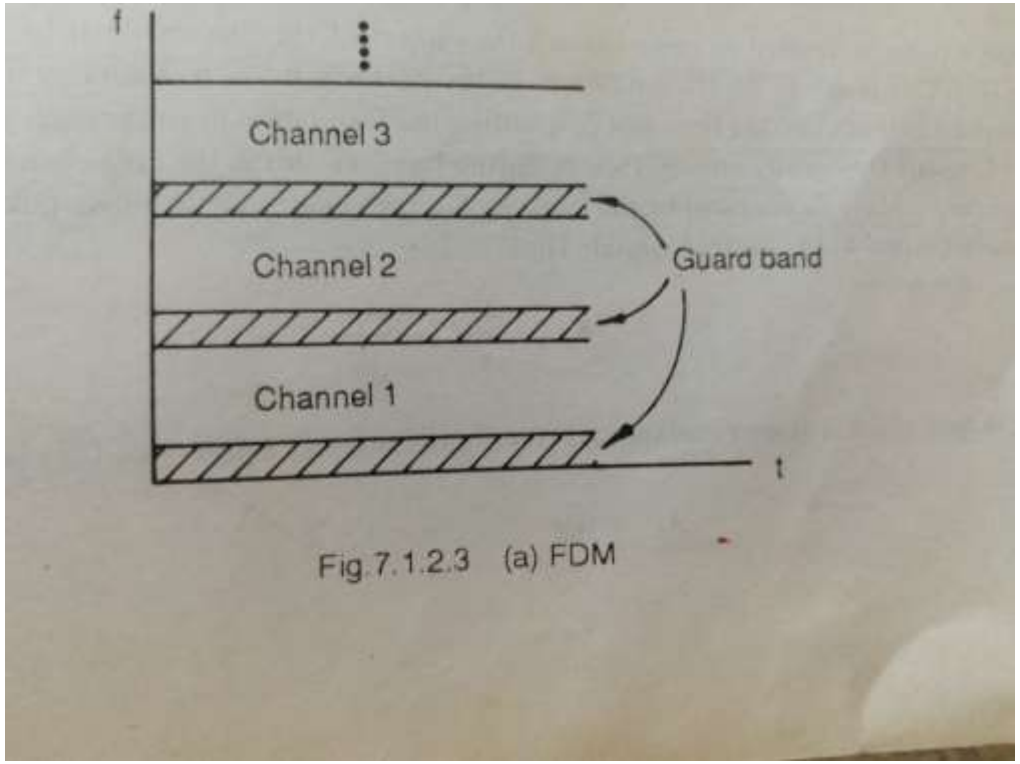
Example: Multiplexing of telephonic signals from n subscribers

- Telephonic message (BW=3kHz) and broadcast signal

limited to 5kHz. Without multiplexing if n channels transmitted,

Interference and no useful information.

- In FDM, each baseband signal translated by Analog Modulation (AM/Angle) to different carrier frequencies.
- Each carrier separated from neighbouring by at least 2fm
- Multiplexed signals can be transmitted over a common channel without interference.
- At receiver, various carrier frequencies selected using BPF tuned to appropriate carrier frequencies and demodulated by separate detector.



FDM System

Advantages: No dynamic coordination needed and works also for analog signals

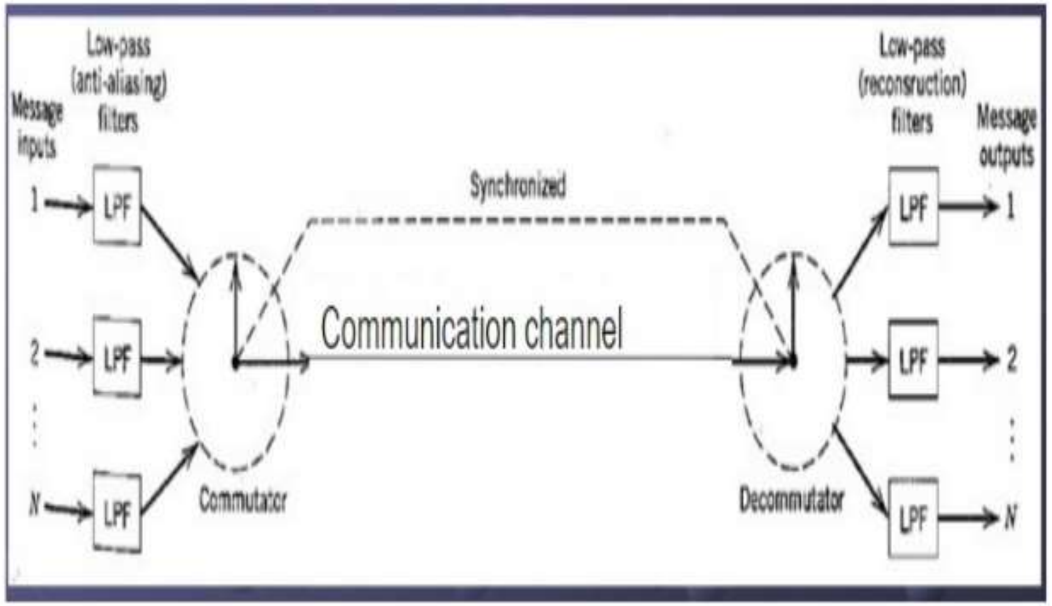
Disadvantages: Waste of bandwidth if traffic distributed uneven; inflexible;

<u>Time Division Multiplexing</u>

- Time Division Multiplexing (TDM) is the time interleaving of samples from several sources so that the information from these sources can be transmitted serially over a single communication channel.
- **At the Transmitter** :Simultaneous transmission of several signals on a time-sharing basis.

  Each signal occupies its own distinct time slot, using all frequencies, for the duration of the transmission. Slots may be permanently assigned on demand
- **At the Receiver** : Decommutator (sampler) has to be synchronized with the incoming waveform
- In Pulse modulation techniques, there is a **free space between any two consecutive pulses** of a signal. This **free space between pulses** can be occupied by **pulses from other channel.** This is **Time Division Multiplexing (TDM)** and makes **maximum utilization of transmission channel.**
- **<u>Applications of TDM</u>:** Digital Telephony, Data communications, Satellite Access, Cellular radio

Block diagram of TDM and PAM-TDM



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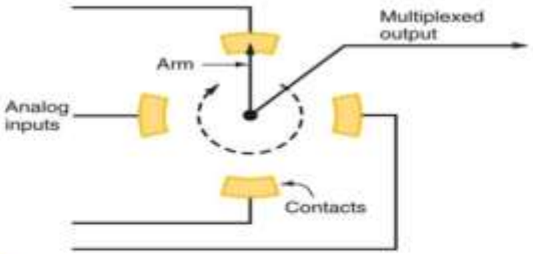
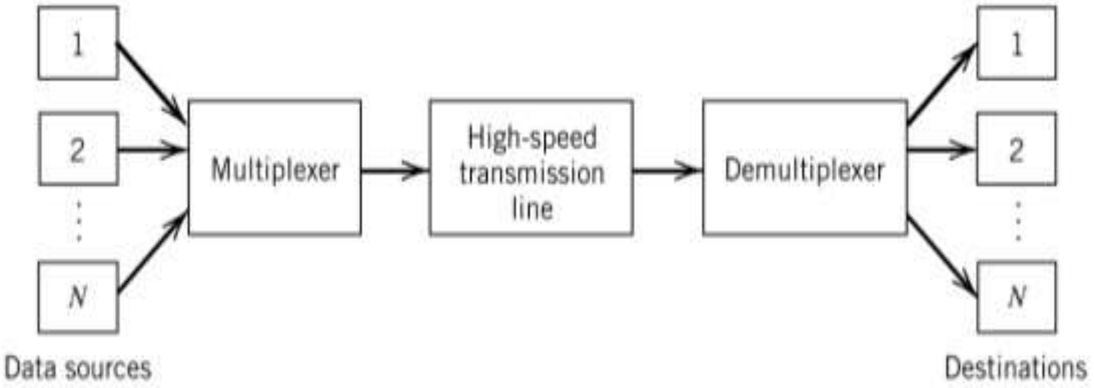
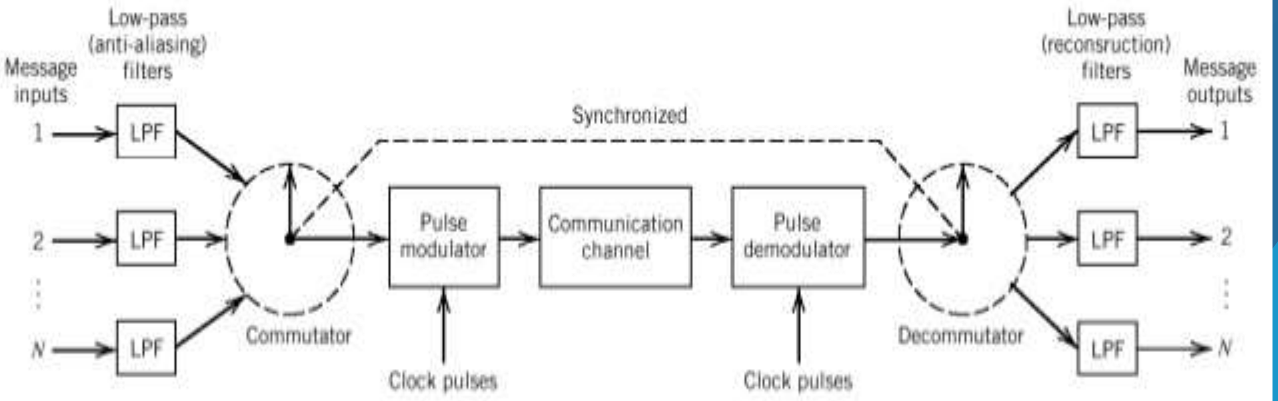
Simple rotary-switch multiplexer



Time Division Multiplexing



Conceptual diagram of multiplexing-demultiplexing.



PAM TDM System

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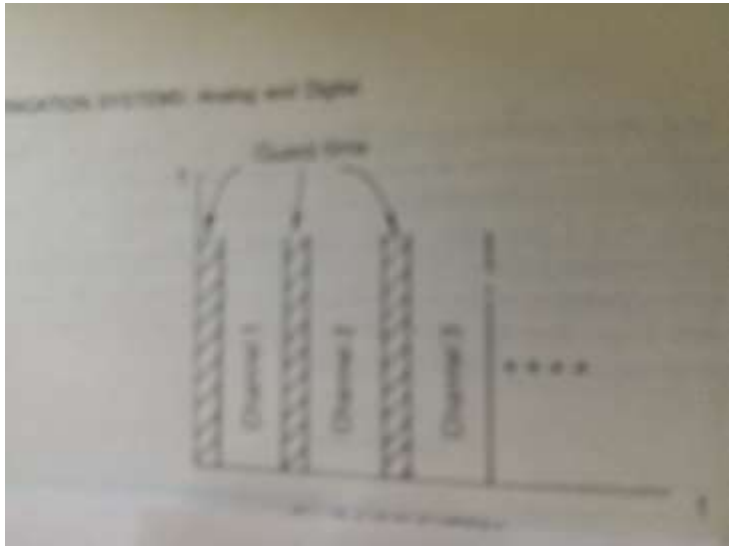
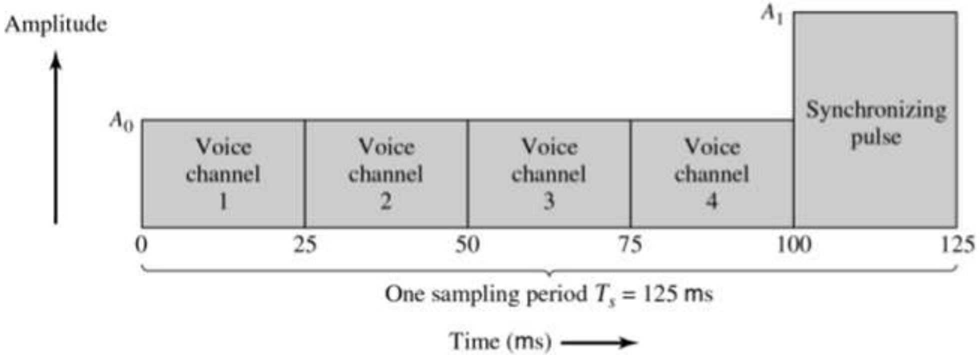


TDM

- Composition of one frame of a multiplexed PAM signal incorporating four voice-signals and a synchronizing pulse.



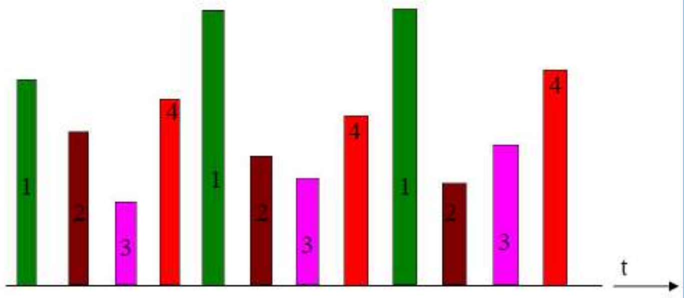
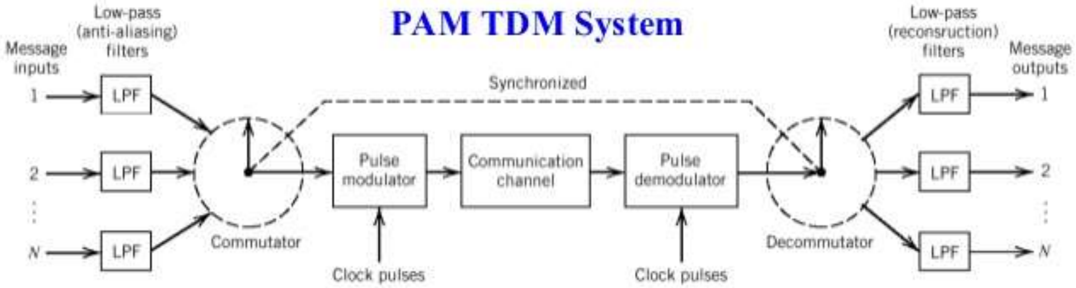
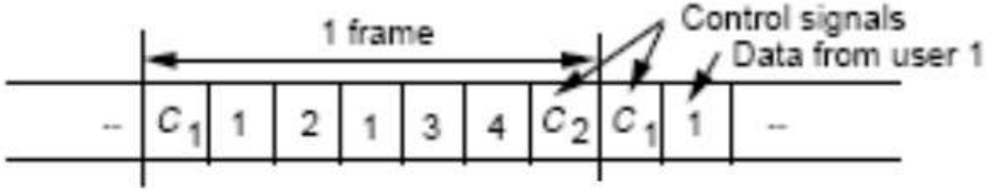


Fig: 2.23 Multiplexing of FOUR signals.

Block diagram of TDM system

PAM TDM System



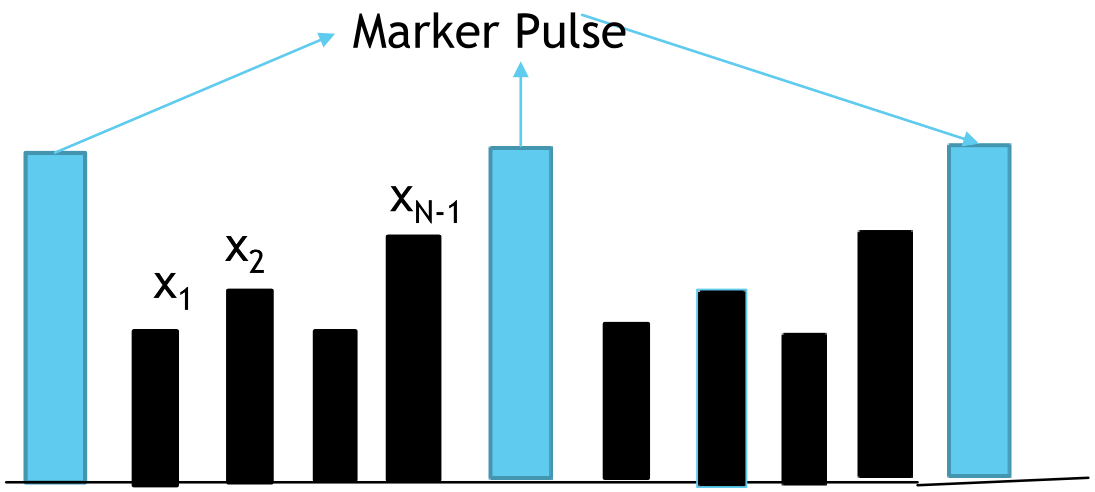


A Typical Framing Structure for TDM

- The system shows **TDM of ‘N’ PAM channels**.
- **Each channel** to be transmitted is passed through **LPF** to band limit its frequency to **f_m Hz (W Hz)**
- Outputs of LPF are connected to **the rotating sampling switch or commutator**.
- It takes **sample from each channel per revolution** and rotates at **the rate of fs.**
- Function of commutator is two fold**: (i) Taking narrow samples of each of N input messages at rate $1/T_s$ (ii) To sequentially interleave N samples Inside a sampling interval T_s**
- The **single signal composed due to multiplexing** of input channels given to **Transmission channel**
- If **W or f_m** is highest signal frequency in the message signal: **$fs \geq 2f_m$ or 2W**; **Ts or $1/fs \leq 1/2fm$**
- Thus time interval Ts contains one sample from each input**. It is called frame. If N input channels multiplexed**, each frame will have one sample from each of N channel's input
- Spacing between two samples: **Ts/N**
- No. of pulses/sec=1/ spacing between two pulses=$1/(Ts/N) = N/Ts$
- Ts=1/fs; No. of pulses per second=Nfs
- The no. of pulses transmitted per second is called **signalling rate of TDM ‘r’**
- **r=Nfs**; fs $\geq$2fm**;** **$r \geq 2Nf_m$ or $r \geq 2NW$**
- Pulsed TDM passed through LPF to convert it to baseband signal whose BW given **by half signalling rate**
- **B.W=$r/2$ = Nf_m=NW; Minimum Transmission Bandwidth**
- **At the receiver, decommutator separates the time multiplexed input channels which then passed through reconstruction filter**

Synchronization in TDM system

- The time division multiplexing (TDM) needs synchronization between multiplexer and demultiplexer. If synchronization is not there between multiplexer and demultiplexer, a bit going to one channel may be received by the wrong channel.
- Because of this reason, one or more synchronization bits are usually added to the beginning of each frame called **Markers** (**highest amplitude**)
- These bits are called framing bits (Marker pulse), allows the demultiplexer to synchronize with the incoming stream so that that it can separate time slot accurately.
- Because of the marker pulse, no of channels to be multiplexed reduced by 1



<u>Crosstalk and Guard Times</u>

- RF transmission of TDM needs further modulation
- TDM signal converted to smooth modulating waveform by passing through a baseband filter
- This filtering gives rise to **inter-channel crosstalk** which means individual signal sample amplitude interfere with each other**.** **<u>Thus interference between adjacent TDM channels is crosstalk</u>**
- This interference can be **reduced by** increasing distance between **individual signal amplitudes**
- The minimum distance between individual signal samples to avoid **crosstalk** is <u>called **guard time**</u>.
- Ideally communication channel over which TDM signal is transmitted should be **infinite** but in practise has a finite BW, known as **band limited channels**
- Whenever signal passed through band limited channel, **shape of signal will change**.
- Whenever a PAM-TDM signal transmitted over band limited channel, signals corresponding to **$x_1(t)$ get mixed with $x_2(t)$ and this overlap causes crosstalk.**
- To keep cross talk below -30dB, $T_g \geq \frac{0.55}{BW}$
- **Thus, guard time required to avoid cross talk decreases with increase in BW**

Transmission Bandwidth for 'N' PAM-TDM channels: Nf_m

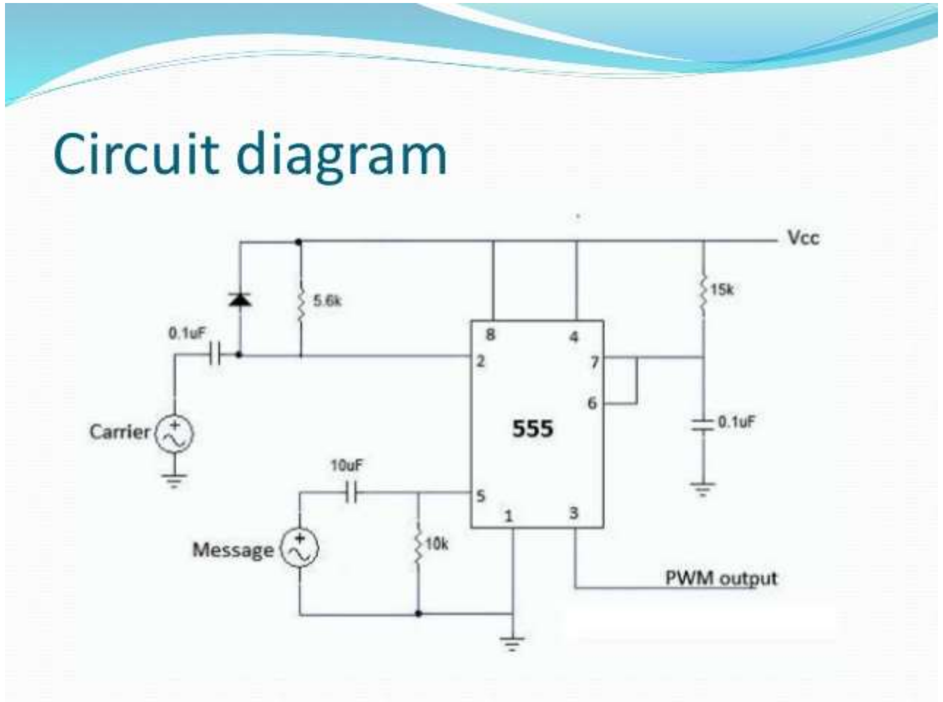
Where f_m is the maximum frequency of baseband signal

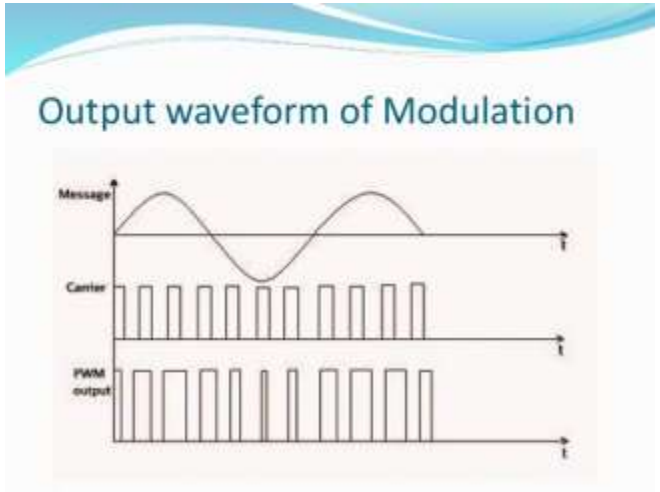
Advantages:

- Full available channel BW can be utilized.
- TDM circuitry not very complex
- Problem of cross talk not very severe

Disadvantages

- Synchronization for proper operation


Circuit diagram
Vcc
5.6k
15k
0.1uF
8
4
2
7
6
555
0.1uF
Carrier
10uF
5
1
3
10k
Message
PWM output


Output waveform of Modulation
Message
Carrier
PWM output
t
t
t

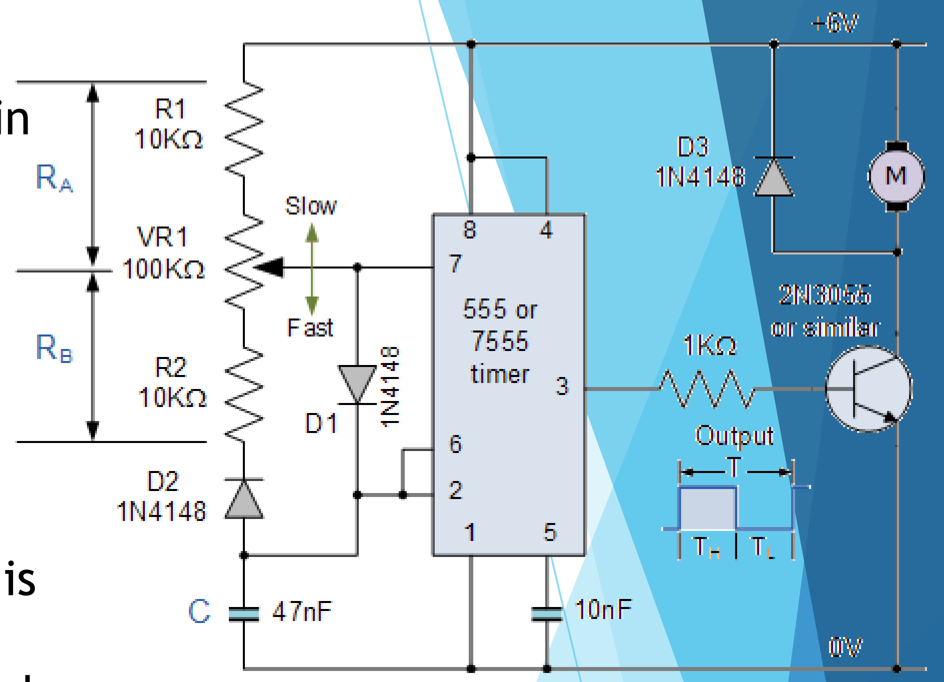
Application of PWM

- Although PWM is also used in communications, its main purpose is **actually to control the power that is supplied to various types of electrical devices,** most especially to inertial loads such as AC/DC motors.
- Pulse-width modulation (PWM) is used for controlling the amplitude of digital signals in order to control devices and **applications requiring power or electricity**. It essentially controls the amount of power, in the perspective of the voltage component, that is given to a device by cycling the **on-and-off phases of a digital signal quickly and varying the width of the "on" phase or duty cycle**. To the device, this would appear as a steady power input with an average voltage value, which is the result of the percentage of the on time. The duty cycle is expressed as the percentage of being fully (100%) on.
- A very powerful benefit of **PWM is that power loss is very minimal**. Compared to regulating power levels using an analog potentiometer to limit the power output by essentially choking the electrical pathway, thereby resulting in power loss as heat, PWM actually turns off the power output rather than limits it. Applications range from **controlling DC motors and light dimming to heating elements.**

This simple circuit based around the familiar NE555 or 7555 timer chip is used to produced the required pulse width modulation signal at a fixed frequency output. The timing capacitor C is charged and discharged by current flowing through the timing networks RA and RB as we looked at in the 555 Timer tutorial.

The output signal at pin 3 of the 555 is equal to the supply voltage switching the transistors fully “ON”. The time taken for C to charge or discharge depends upon the values of RA, RB.

The capacitor charges up through the network RA but is diverted around the resistive network RB and through diode D1. As soon as the capacitor is charged, it is immediately discharged through diode D2 and network RB into pin 7. During the discharging process the output at pin 3 is at 0 V and the transistor is switched “OFF”.



Then the time taken for capacitor, C to go through one complete charge-discharge cycle depends on the values of RA, RB and C with the time T for one complete cycle being given as:

- The time, TH, for which the output is “ON” is: TH = 0.693(RA).C
- The time, TL, for which the output is “OFF” is: TL = 0.693(RB).C
- Total “ON”-“OFF” cycle time given as: T = TH + TL with the output frequency being $f = 1/T$
- With the component values shown, the duty cycle of the waveform can be adjusted from about 8.3% (0.5V) to about 91.7% (5.5V) using a 6.0V power supply. The Astable frequency is constant at about 256 Hz and the motor is switched “ON” and “OFF” at this rate.
- Resistor R1 plus the “top” part of the potentiometer, VR1 represent the resistive network of RA. While the “bottom” part of the potentiometer plus R2 represent the resistive network of RB above.
- These values can be changed to suite different applications and DC motors but providing that the 555 Astable circuit runs fast enough at a few hundred Hertz minimum, there should be no jerkiness in the rotation of the motor.
- Diode D3 is our old favourite the flywheel diode used to protect the electronic circuit from the inductive loading of the motor. Also if the motor load is high put a heatsink on the switching transistor or MOSFET.
- Pulse width modulation is a great method of controlling the amount of power delivered to a load without dissipating any wasted power. The above circuit can also be used to control the speed of a fan or to dim the brightness of DC lamps or LED’s. If you need to control it, then use Pulse Width Modulation to do it.