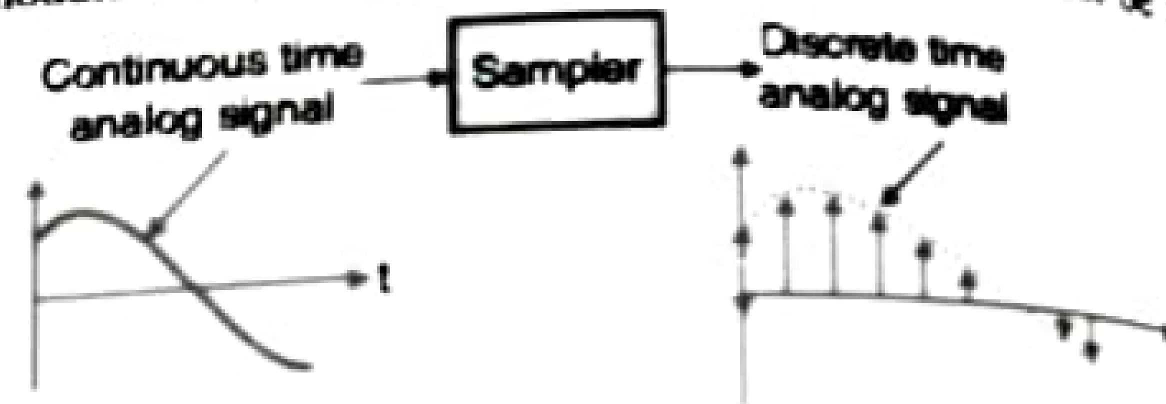
2 Sampling Process :

In the pulse modulation and digital modulation systems, the signal to be transmitted must be in the discrete time form.

If the message signal is coming from a digital source (e.g. a digital computer) then it is in the proper for a digital communication system to be processed.



(L-156) **Fig. 8.2.1 : Sampling process**

But this is not always the case. The message signal can be analog in nature (e.g. speech or video signal).

In such a case it has to be first converted into a discrete time signal. We use the "sampling process" to do this.

Thus using the sampling process we convert a continuous time signal into a discrete time signal.

For the sampling process to be of practical utility it is necessary to choose the sampling rate properly. The sampling process should satisfy the following requirements :

1. Sampled signal should represent the original signal faithfully.
2. We should be able to reconstruct the original signal from its sampled version.

In order to represent the original message signal "faithfully" (without loss of information), it is necessary to take as many samples of the original signal as possible.

Higher the number of samples, closer is the representation.

The number of samples depends on the "sampling rate" and the maximum frequency of the signal to be sampled.

Sampling theorem was introduced to the communication theory in 1949 by Shannon. Therefore this theorem is also called as "**Shannon's sampling theorem**".

The statement of sampling theorem in time domain, for the bandlimited signals of finite energy is as follows :

Statement :

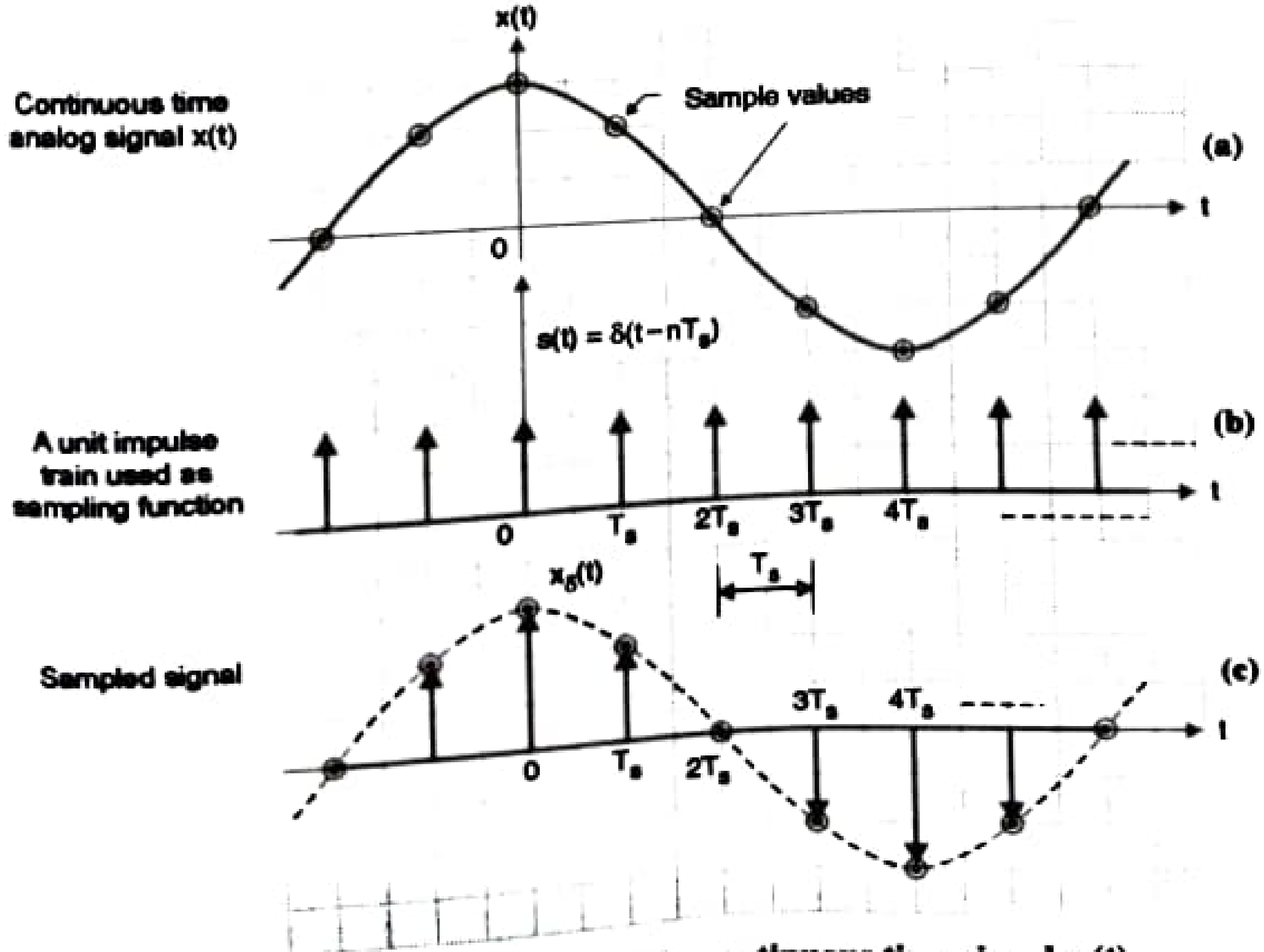
If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.

If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (1/2W) seconds apart.

Let us now prove the sampling theorem in time domain. The assumptions made for this proof are as follows :

Assumptions :

- Let x (t) be a continuous time analog signal as shown in Fig. 8.3.1.



(D-480) **Fig. 8.3.1 : Sampling of a continuous time signal x (t)**

- Let x (t) be a signal with finite energy and infinite duration.
- Let x (t) be a strictly bandlimited signal. That means it does not contain any frequency components above "W" Hz.
- Let s (t) be the sampling function as shown in Fig. 8.3.1. It is a train of unit impulses, spaced by a period of T_s seconds. This sampling function samples the original signal at a rate of "f_s" samples per second. Therefore "T_s" represents the sampling period such that,

$$T_s = \frac{1}{f_s} = \text{Sampling period} \qquad ...(8.3.1)$$

$$\text{and} \quad f_s = \frac{1}{T_s} = \text{Sampling rate.}$$

Spectrum of the sampled signal :

Step 1 : Represent the sampling function s (t) mathematically :

- Fig. 8.3.1 shows the sampling function s (t) which is a train of unit impulses.
- The spacing between the adjacent unit impulses is T_s seconds, therefore the frequency of the sampling function is equal to the sampling frequency f_s.
- The sampled signal is denoted by $x_\delta(t)$ and it is as shown in Fig. 8.3.1.
- The sample function s (t) can be represented mathematically as follows :

$$s(t) = \dots\dots\delta(t+2T_s) + \delta(t+T_s) + \delta(t) + \delta(t-T_s) + \delta(t-2T_s) + \dots$$

$$\therefore \; s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \qquad \dots(8.3.2)$$

Step 2 : Represent the sampled signal $x_\delta(t)$ mathematically :

- Fig. 8.3.1 shows the sampled signal $x_\delta(t)$ graphically. It is present only at the sampling instants i.e. T_s, $2T_s$ etc. and its instantaneous amplitude is equal to the amplitude of original signal x (t) at the sampling instants.
- This is shown by the encircled points in Fig. 8.3.1. Let us represent the instantaneous amplitude of x (t) at the various sampling points $t = nT_s$ as $x(nT_s)$. This is the amplitude of the encircled points of Fig. 8.3.1.
- Looking at the sampled signal $x_\delta(t)$ we can say that the sampled signal is obtained by multiplying x (t) and s (t).

$$\therefore \; x_\delta(t) = x(t) \times s(t) = x(nT_s) \times s(t) \qquad \dots(8.3.3)$$

Substituting the expression for s (t) from Equation (8.3.2) we get the mathematical expression for the sampled signal $x_\delta(t)$ as,

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s) \qquad \dots(8.3.4)$$

Step 3 : Obtain the Fourier transform of the sampled signal :

- The fourier transform of a train of impulses (dirac delta function) is given by.

$$X(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$$

- Here we have the similar pulse train as sampling function s (t). Therefore the Fourier transform of the sampling function is given by,

$$S(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \qquad ...(8.3.5)$$

- Note that f_0 has been replaced by f_s in the above equation.
- The sampled signal in the time domain is represented as product of x (t) and s (t).

$$\text{i.e.} \quad x_\delta(t) = x(t) \times s(t) \qquad ...(8.3.6)$$

- Taking the Fourier transform of both the sides we get,

$$\text{i.e.} \quad X_\delta(f) = X(f) * S(f) \qquad ...(8.3.7)$$

- This is because the Fourier transform of the product of two signals in the time domain is the convolution of their Fourier transforms. Substituting the value of S(f) from Equation (8.3.5) we get,

$$\therefore \quad X_\delta(f) = X(f) * \left[f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \right] \qquad ...(8.3.8)$$

where * denotes convolution. Interchanging the orders of convolution and summation results in :

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f) * \delta(f - nf_s) \qquad ...(8.3.9)$$

- From the properties of delta function, we find that the convolution of X (f) and $\delta(f - nf_s)$ is equal to $X(f - nf_s)$. Hence the above equation can be simplified as follows :

$$\text{F.T. of the sampled signal, } X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \qquad ...(8.3.10)$$

where X (f) = Fourier transform of the original signal x (t).

Conclusion from Equation (8.3.10) :

1. The term $X(f - nf_s)$ in Equation (8.3.10) represents the shifted version of the spectrum $X(f)$ of the original signal $x(t)$. Thus depending on the value of "n" (which extends from $-\infty$ to $+\infty$) we will get infinite number of original spectrums $X(f)$ centered at frequencies $0, \pm f_s, \pm 2f_s, \pm 3f_s, \pm 4f_s$ ...etc. In other words,

$$X(f - nf_s) = X(f) \text{ at } f = 0, \pm f_s, \pm 2f_s, \pm 3f_s \qquad \text{...(8.3.11)}$$

2. This concept will be clear if we open Equation (8.3.10) and write the terms separately as shown below.

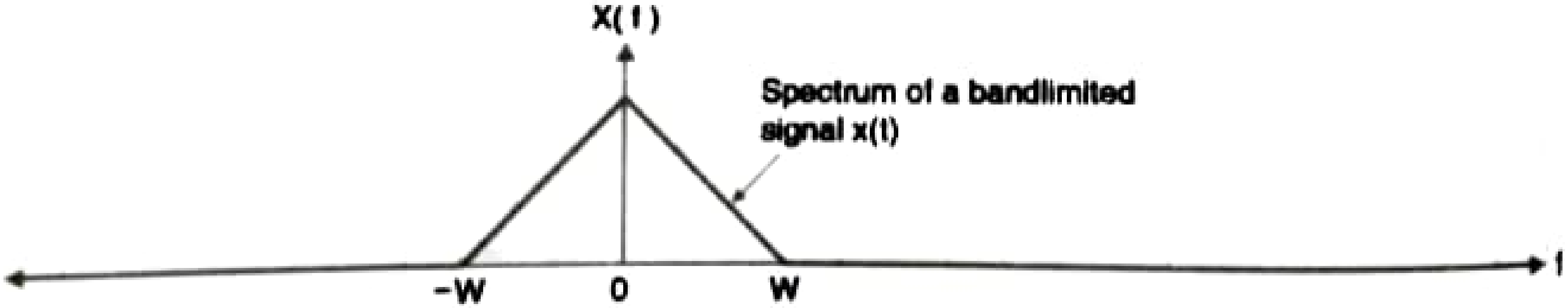
Now open the summation sign in Equation (8.3.10) to get, (D-[illegible])

$$X_\delta(f) = \ldots\ldots + \underbrace{f_s X(f + 2f_s)}_{X(f) \text{ shifted left by } 2f_s} + \underbrace{f_s X(f + f_s)}_{X(f) \text{ shifted left by } f_s} + \underbrace{f_s X(f)}_{\text{Spectrum } X(f)} + \underbrace{f_s X(f - f_s)}_{X(f) \text{ shifted right by } f_s} + \ldots\ldots$$

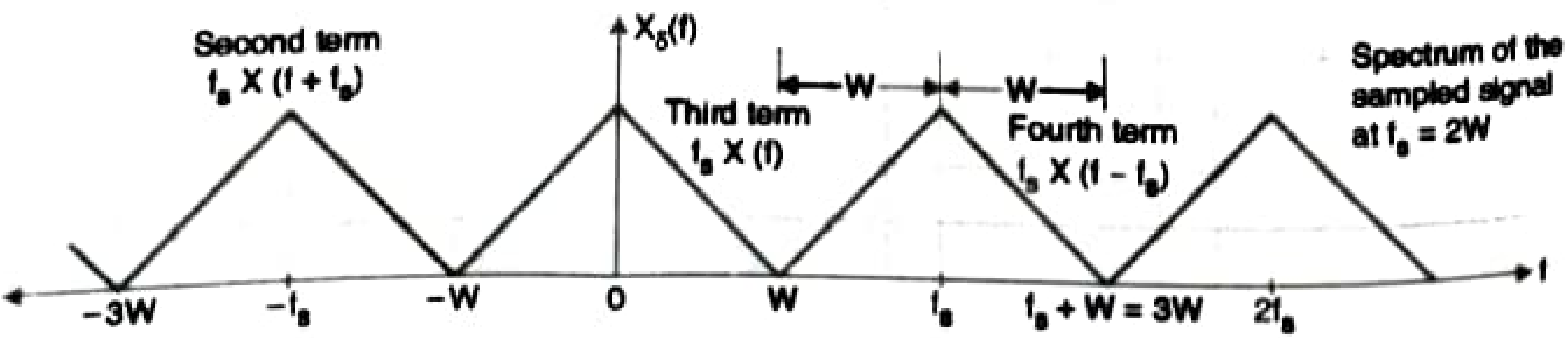
The spectrum $X_\delta(f)$ of the sampled signal is plotted as shown in Fig. 8.3.2.

Equation (8.3.10) can also be written as :

$$X_\delta(f) = f_s X(f) + \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} f_s X(f - nf_s) \qquad \text{... (8.3.12)}$$



(a) Spectrum of the original signal x(t)



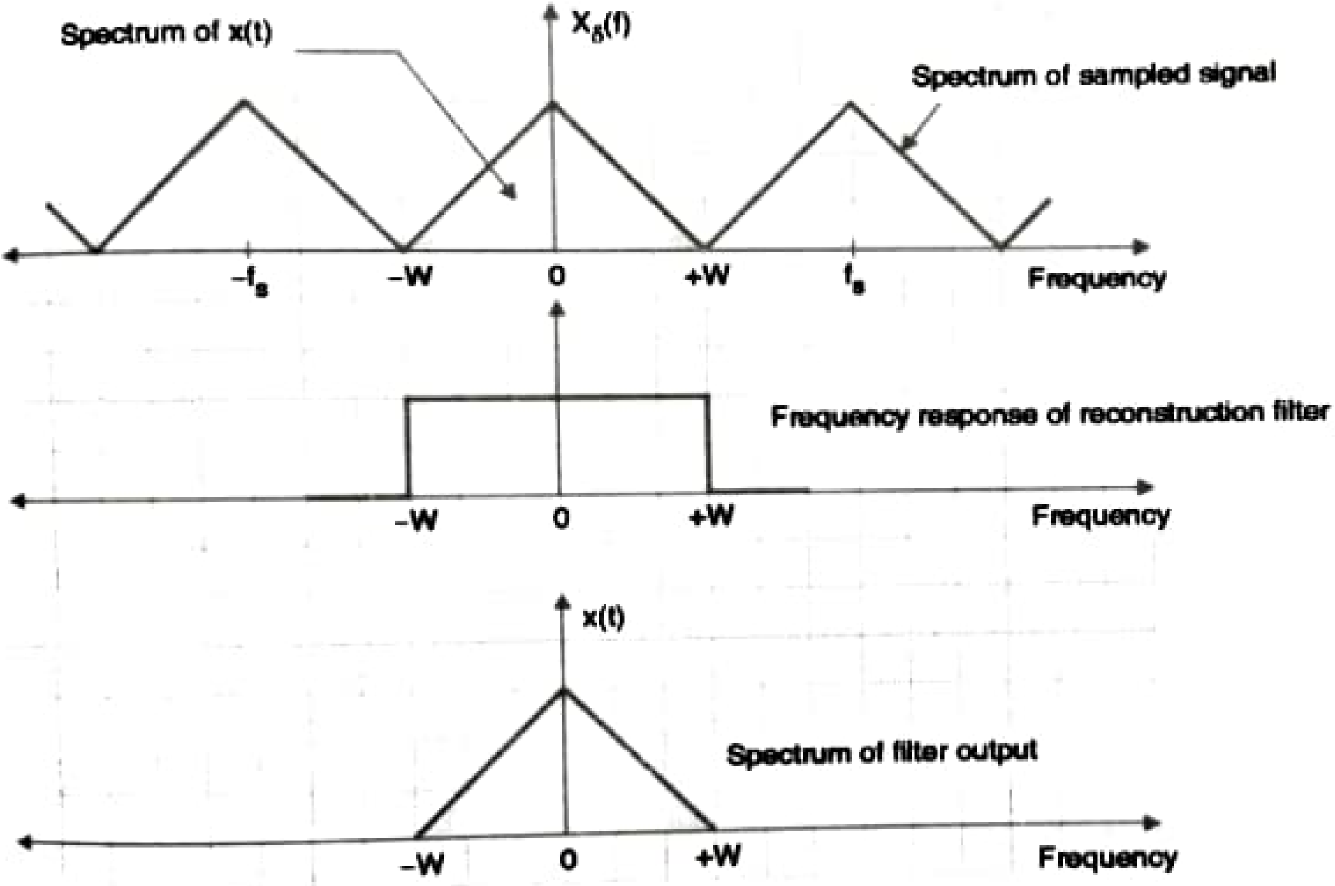
(b) Spectrum of the sampled signal $x_\delta(t)$ with $f_s = 2W$, i.e. Sampling is done exactly at Nyquist rate

Actual reconstruction of the original signal by using a low pass filter :

- Thus the peaks of the sinc pulses represent the amplitudes of the samples.
- The signal x (t) expressed in Equation (8.3.19) is then passed through an ideal low pass filter to recover the original signal x (t). This low pass filter is therefore called as the reconstruction filter. This is shown in the graphical representation of Fig. 8.3.5(a).

(D-413) **Fig. 8.3.5(a) : Reconstruction filter**

- Assume that the cut-off frequency of the ideal low pass filter is adjusted precisely to W Hz. The frequency response of the reconstruction filter is shown in Fig. 8.3.5(b).



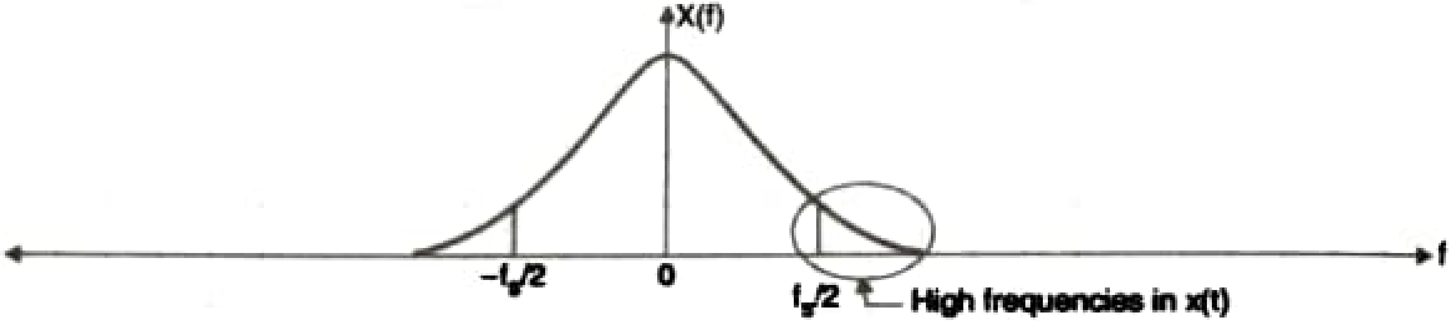
(D-414) **Fig. 8.3.5(b) : Operation of reconstruction**

- When the sampled signal x_δ (t) is applied at the input, this filter will allow only the shaded portion in the spectrum of x_δ (t) to pass through to the output and will block all other frequency components.
- Thus the frequency components only corresponding to x (t) will be passed through to the output and the original signal x (t) is recovered.

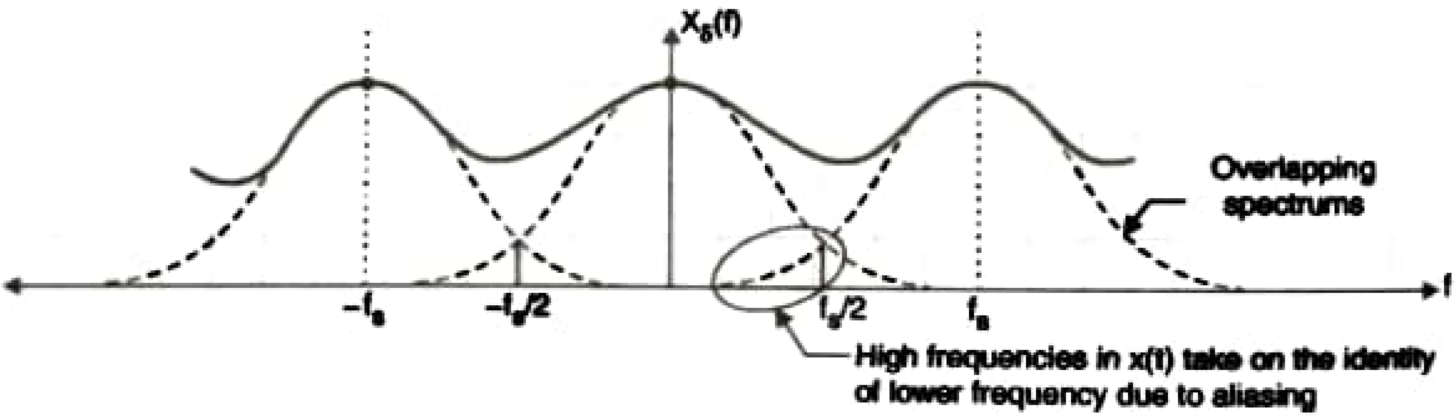
If the signal x (t) is not strictly bandlimited and / or if the sampling frequency f_s is less than 2W, then an error called **aliasing** or **foldover error** is observed. The adjacent spectrums overlap if $f_s < 2W$. This is shown in Fig. 8.3.6(b).

The signal x (t) is not strictly bandlimited. The spectrum of signal x (t) is shown in Fig. 8.3.6(b).

The spectrum $X_\delta(f)$ of the discrete time signal $x_\delta(t)$ is shown in Fig. 8.3.6(b) which is nothing but the sum of X (f) and infinite number of frequency shifted replicas of it as explained earlier.



(a) Spectrum of a continuous time signal x(t)



(b) Spectrum of the sampled version of x(t) with $f_s < 2W$

(D-415) **Fig. 8.3.6**

Consider the two replicas of X (f) which are centered about the frequencies f_s and $-f_s$.

If we use a reconstruction filter with its pass-band extending from $-f_s/2$ to $+f_s/2$ then its output will not be an undistorted version of the original signal x (t). Some distortion will be present in the filter output.

The distortion occurs due to the overlapping of the adjacent spectrums as shown in Fig. 8.3.6(b). Due to this overlapping, it is seen that the portions of the frequency shifted replicas are "folded over" inside the desired spectrum.

Due to this "fold over", high frequencies in X (f) are reflected into low frequencies in $X_\delta(f)$. This can be understood by comparing the shaded portions of the spectra shown in Fig. 8.3.6(a) and (b).

- **Aliasing :** This phenomenon of a high frequency in the spectrum of the original signal x (t), taking on the identity of lower frequency in the spectrum of the sampled signal $x_\delta(t)$ is called as aliasing or fold over error.

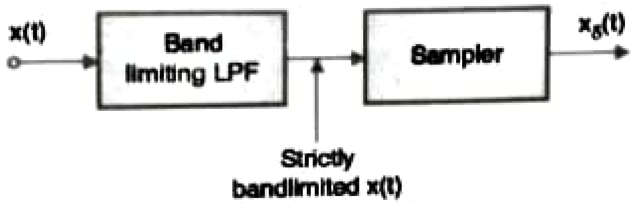
Effect of aliasing :

Due to aliasing some of the information contained in the original signal x (t) is lost in the process of sampling.

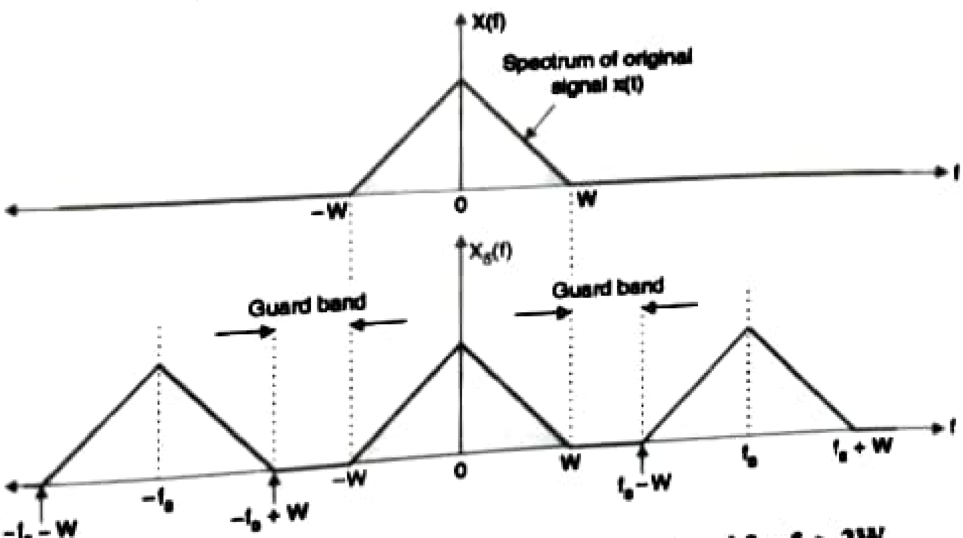
How to eliminate aliasing ?

Aliasing can be completely eliminated if we take the following action :

- Use a bandlimiting low pass filter and pass the signal x (t) through it before sampling as shown in Fig. 8.3.7(a).
- This filter has a cutoff frequency at f_c = W, therefore it will strictly bandlimit the signal x (t) before sampling takes place. This filter is also called as antialiasing filter or prealias filter.



(D-416) **Fig. 8.3.7(a) : Use of a bandlimiting filter to eliminate aliasing**



(D-417) **Fig 8.3.7(b) : Spectrum of a sampled signal for $f_s > 2W$**

- Increase the sampling frequency f_s to a great extent i.e. $f_s >>> 2W$. Due to this, even though x (t) is not strictly bandlimited, the spectrums will not overlap. A guard band is created between the adjacent spectrums as shown in Fig. 8.3.7(b).

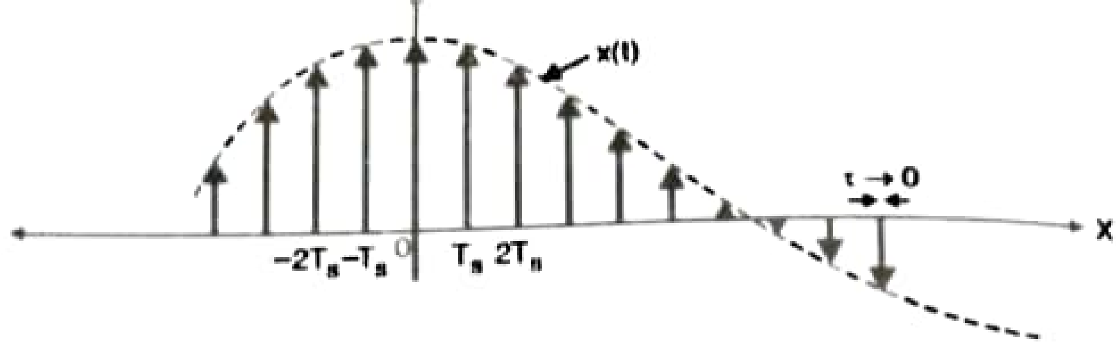
Thus aliasing can be prevented by :

1. **Using an antialiasing or prealiasing filter and**
2. **Using the sampling frequency $f_s > 2W$.**

8.3.3 Nyquist Criteria :

- The minimum sampling rate of "2W" samples per second for a signal x (t) having maximum frequency of "W" Hz is called as "Nyquist rate". The reciprocal of Nyquist rate i.e. 1/2W is called as the Nyquist interval.

$$\text{Nyquist rate} = 2W \text{ Hz}$$
$$\text{Nyquist interval} = 1/2W \text{ seconds}$$



(D-428) Fig. 8.4.1 : An ideally sampled signal $x_\delta(t)$

Disadvantages of ideal sampling :

1. The disadvantage of ideal sampling is that due to very narrow samples, the transmitted power is very small and the S/N ratio is low. Thus the ideally sampled pulses may get lost in the background noise.
2. Ideal sampling is not possible to achieve practically, because it is practically impossible to have pulses of widths approaching zero. Therefore practically natural or flat top sampling is used. Ideal sampling was used only to prove the sampling theorem.

Practical sampling techniques :

The practical sampling techniques are different from the ideal sampling in the following ways :

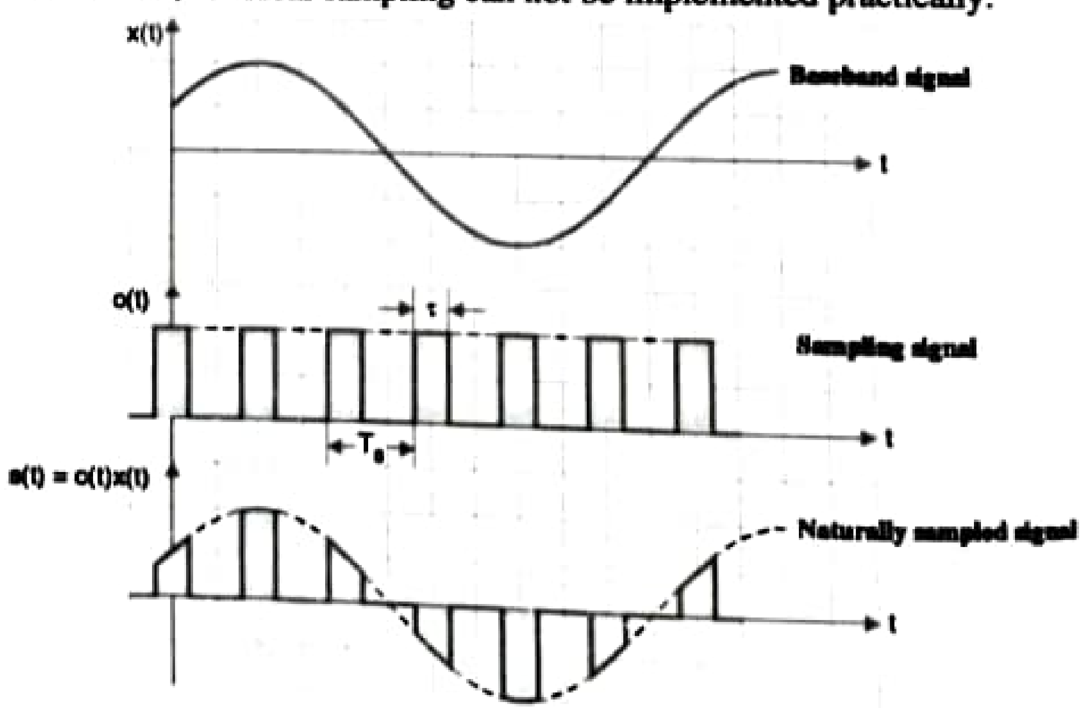
1. In practical sampling methods, the duration of the sampling pulses is finite and the amplitude of the pulses is also finite.
2. Practical sampling methods use the practical low pass filters for reconstruction. Guard band between the adjacent spectrums in the sampled signal is necessary to avoid distortions. Ideal filters are not used.
3. The signal x (t) which is to be sampled is not strictly bandlimited. Due to this there are problems faced while deciding the sampling rate f_s.

There are two popularly used practical sampling techniques. They are :

1. Natural sampling or chopper sampling and
2. Flat top sampling.

8.4.2 Natural Sampling or Chopper Sampling :

As explained earlier, the ideal sampling can not be implemented practically.



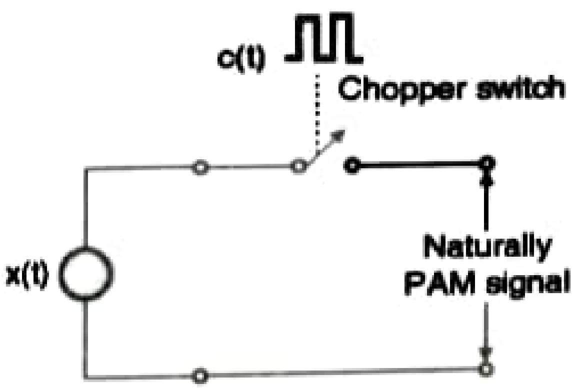
(D-429) Fig. 8.4.2 : Process of natural sampling

A more reasonable and practically feasible manner of sampling is called as "Natural Sampling", as shown in Fig. 8.4.2. Looking at the waveforms in Fig. 8.4.2 we note the following important points :

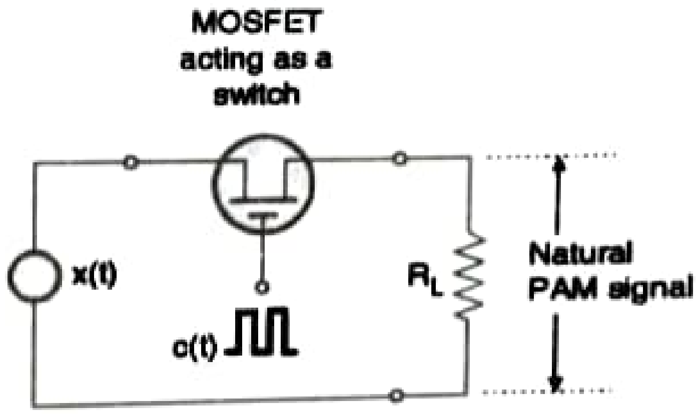
- Here the sampling waveform c (t) consists of a train of pulses each having a duration "τ" and separated by the sampling time T_s.
- The baseband or modulating signal x (t) and the sampled signal s (t) = c (t) x (t) are as shown in Fig. 8.4.2. The sampled signal is obtained by multiplication of x (t) and c (t).
- The sampled signal is a train of pulses of width τ, whose amplitudes are varying. These pulses do not have flat tops but their tops follow the waveform of the signal x (t).
- The sampling rate is greater than or equal to the Nyquist rate.

Circuit arrangement for natural sampling :

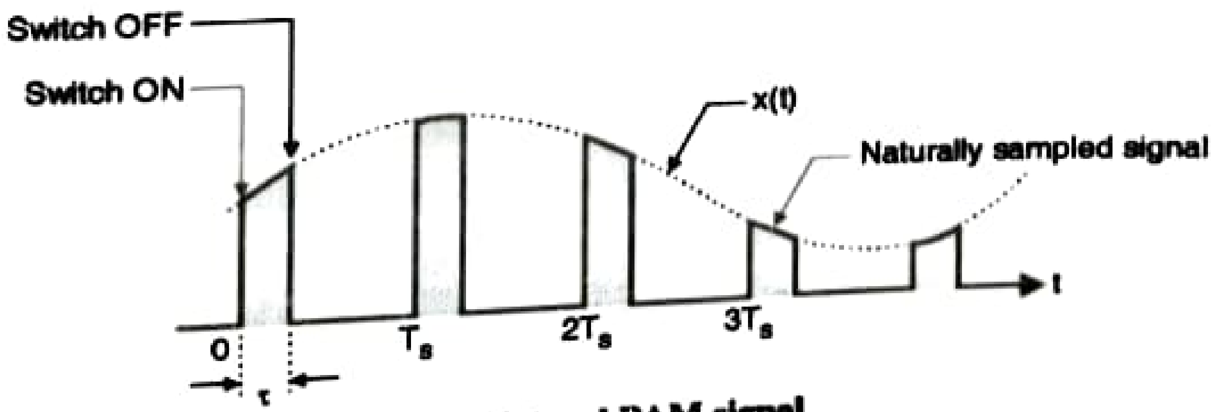
- Natural sampling is sometimes called as chopper sampling because the waveform of the sampled signal appears to be chopped off from the continuous time signal x (t).
- The chopper arrangement is as shown in Fig. 8.4.3 where the chopper switch is being operated by the sampling function "c (t)".



(a) Natural PAM



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



(c) Natural PAM signal

(D-430) **Fig. 8.4.3**

Reconstruction :

The reconstruction technique for natural sampling is similar to that for the instantaneous sampling.

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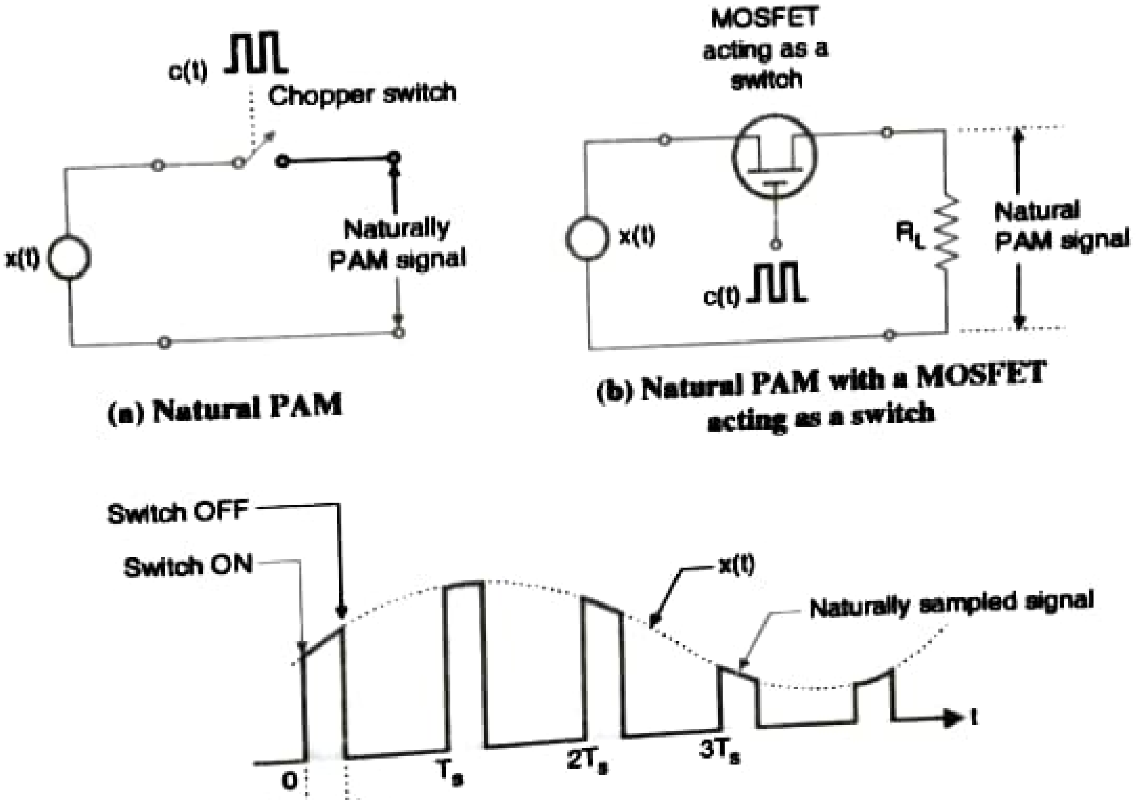


(a) Natural PAM

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

(c) Natural PAM signal

(D-430) Fig. 8.4.3

Reconstruction :

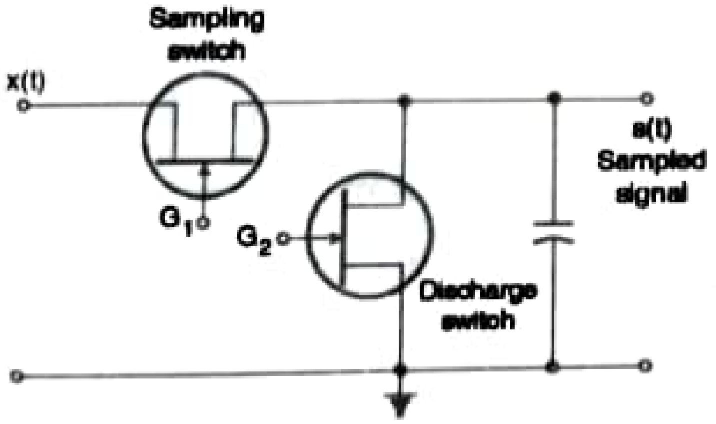
- The reconstruction technique for natural sampling is similar to that for the instantaneous sampling.

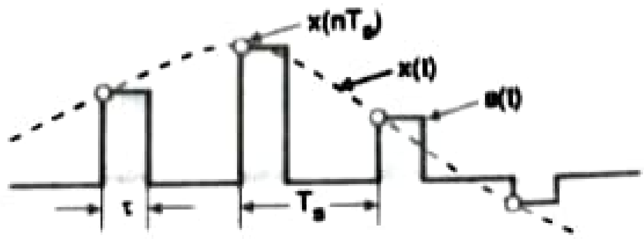
(D-432) Fig. 8.4.5

Merits and demerits of natural sampling :

1. Generation is easy.
2. We can use practical low pass filter for reconstruction.
3. The amplitudes of high frequency components decrease therefore some distortion is introduced.
4. Increased SNR due to finite pulse width of the sampling function and that of the sampled signal.
5. For large values of "τ" there is a possibility of crosstalk.

- The natural sampling is rarely employed in practice. Instead the other practical sampling technique called flat top sampling is employed in practice.
- In the flat top sampling technique, the analog signal x (t) is sampled instantaneously at the rate $f_s = \frac{1}{T_s}$ and the duration of each sample is lengthened to a duration "τ" as shown in Fig. 8.4.6(b).
- Thus the amplitudes of these pulses are constant and equal to the corresponding sampled values.





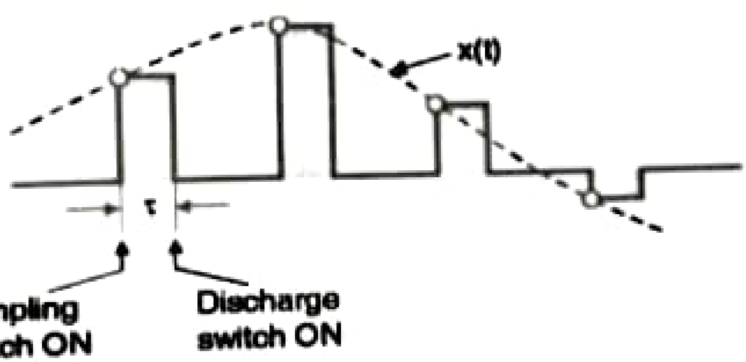
(a) Sample and hold circuit to obtain the flat topped samples **(b) Flat top sampled signal**

(D-433) **Fig. 8.4.6**

- The flat top pulses can be obtained by using the sample and hold circuit shown in Fig. 8.4.6(a).

Operation of the sample and hold circuit :

- The sample and hold circuit consists of two FET switches and a capacitor as shown in Fig. 8.4.6(a). The analog signal x (t) is applied at the input of this circuit and the sampled signal s (t) is obtained across the capacitor.
- A gate pulse will be applied to gate G_1 at the instant of sampling for a very short time. The sampling switch will turn on and the capacitor charges through it to the sample value x (nT_s).
- The sampling switch is then turned off. Both the FETs will remain OFF for a duration of "τ" seconds and the capacitor will hold the voltage across it constant for this period. Thus the pulse is stretched to "τ" seconds.
- At the end of the pulse interval (τ), a pulse is applied to G_2 i.e. gate terminal of discharge FET. This will turn on the discharge FET and short circuit the capacitor. The output voltage then reduces to zero. This is as shown in Fig. 8.4.7.



In pulse modulation, the carrier is in the form of train of periodic rectangular pulses.

Pulse modulation can be either analog or digital.

In the analog pulse modulation, the amplitude, width or position of the rectangular carrier pulses is changed in accordance with the modulating signal.

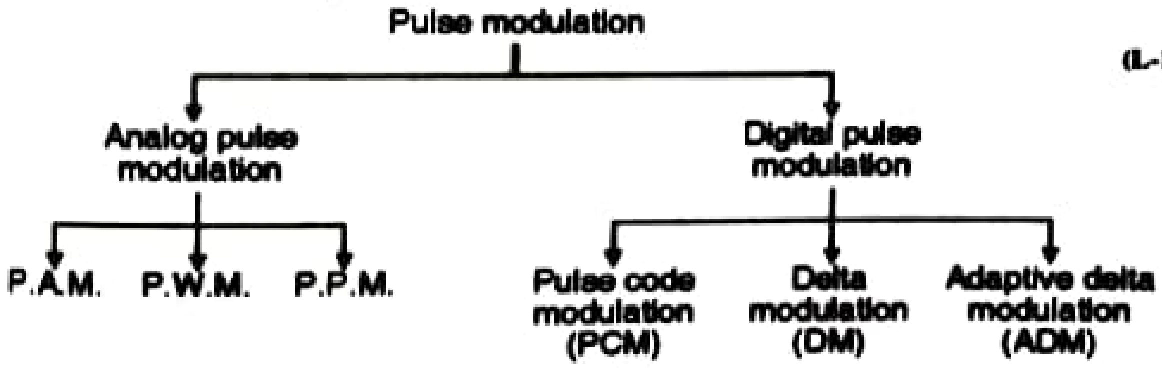
This will result in PAM (pulse amplitude modulation), PWM (pulse width modulation) or PPM (pulse position modulation) respectively.

PAM, PWM and PPM the examples of analog pulse modulation.

The pulse modulation can be digital as well. The well known examples of digital pulse modulation are pulse code modulation (PCM), delta modulation (DM), adaptive delta modulation (ADM), etc.

The classification of the pulse modulation system is as follows :

(L-153)



In this chapter we are going to discuss only the analog pulse modulation schemes.

Pulse Amplitude Modulation (PAM) :

The amplitude of a constant width, constant position rectangular carrier is varied in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 9.1.1(c).

Pulse Width Modulation (PWM) :

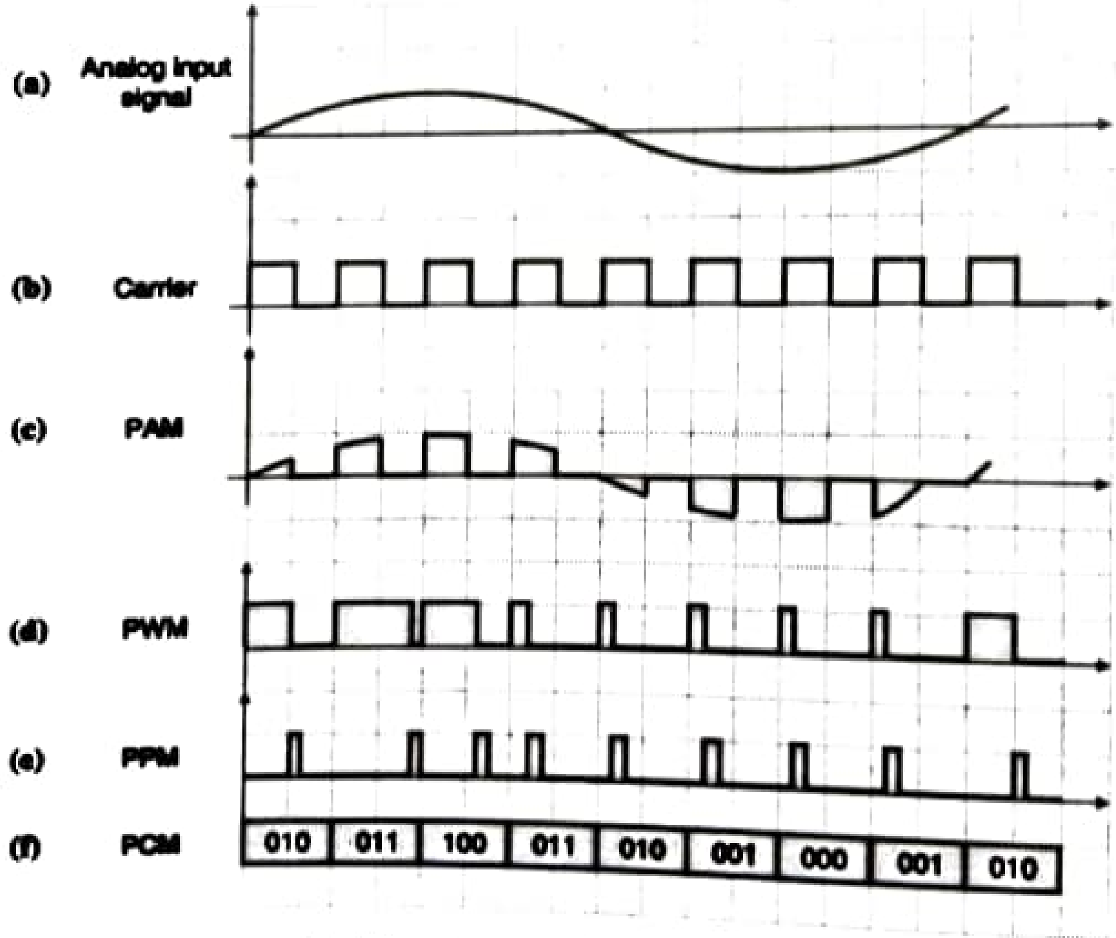
- The width of carrier pulses is made to vary in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 9.1.1(d).
- PWM is also called as pulse duration modulation (PDM) or pulse length modulation (PLM).

Pulse Position Modulation (PPM) :

- In PPM the amplitude and width of the pulses is kept constant but the position of each pulse is varied in accordance with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.
- The PPM pulses can be derived from the PWM pulses as shown in Fig. 9.1.1(e). Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

4. **Pulse Code Modulation (PCM) :**

- The analog message signal is sampled and converted to a fixed length, serial binary number as shown in Fig. 9.1.1(f).



(L-154) **Fig. 9.1.1 : Pulse modulation**

- In other words a binary code is transmitted. Hence the name pulse code modulation.
- The PAM, PWM and PPM are called as the analog pulse communication systems whereas PCM, delta modulation (DM) are the examples of digital pulse communication systems.

What is the difference between analog pulse communication and digital pulse communication ?

- For analog as well as digital pulse communication systems, the transmitted signal is a discrete time signal.
- In analog pulse communication, the information is transmitted in the form of change in amplitude, width or position of the rectangular carrier pulses. So the transmitted pulsed signal is still an analog signal.
- In digital pulse communication, the information is transmitted in the form of codes. Codeword are formed by grouping the digital pulses.
- Note that for digital pulse communication we do not change amplitude, frequency or phase of the transmitted signal. Thus the transmitted signal in digital pulse communication is a digital signal.

Practical use :

- PAM does not have a good noise immunity. So its practical use is restricted.
- PWM and PPM are used for some military applications but are not used for commercial communication applications.
- PCM is the most useful method of all.

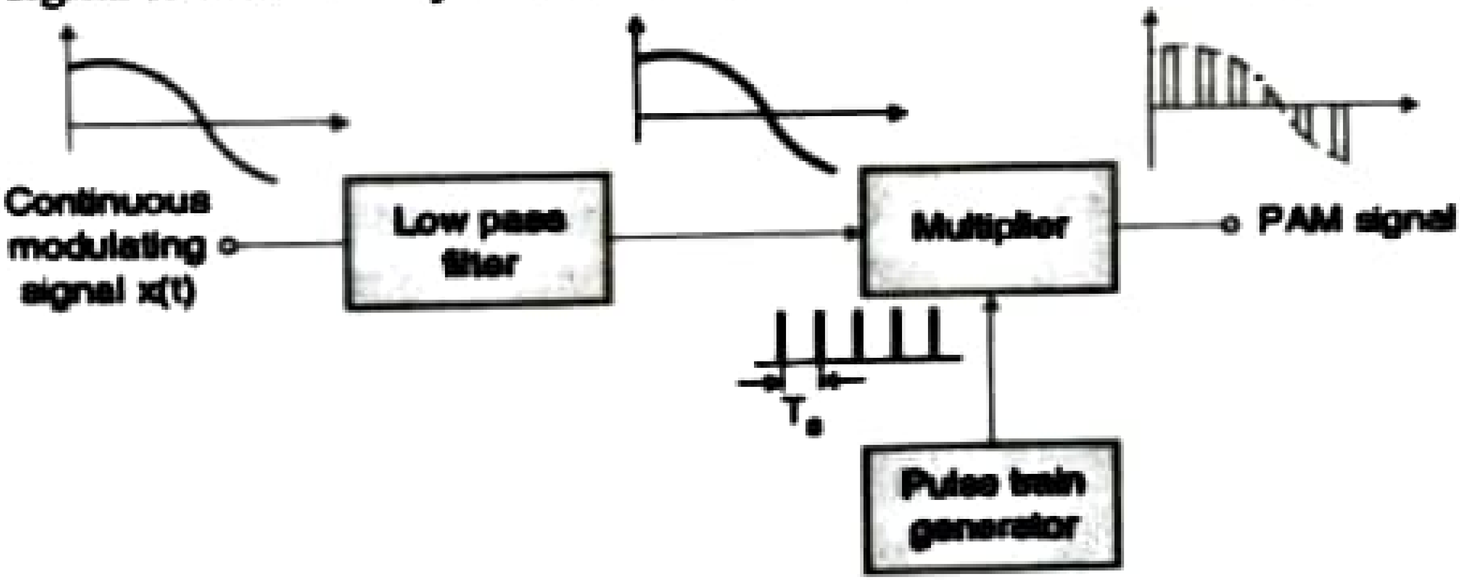
Pulse Amplitude Modulation (PAM) :

In the PAM system, the amplitude of the pulsed carrier is changed in proportion with the instantaneous amplitude of the modulating signal x(t). So the information is contained in the amplitude variation of PAM signal.

The carrier is in the form of train of narrow pulses as shown in Fig. 9.2.2.

If you compare the PAM system with the sampling process, you will find that these two processes are identical.

The PAM signal is then sent by either wire or cable or it is used to modulate a carrier.



(L-181) Fig. 9.2.1 : Generation of PAM

Types of PAM :

There are two types of PAM :

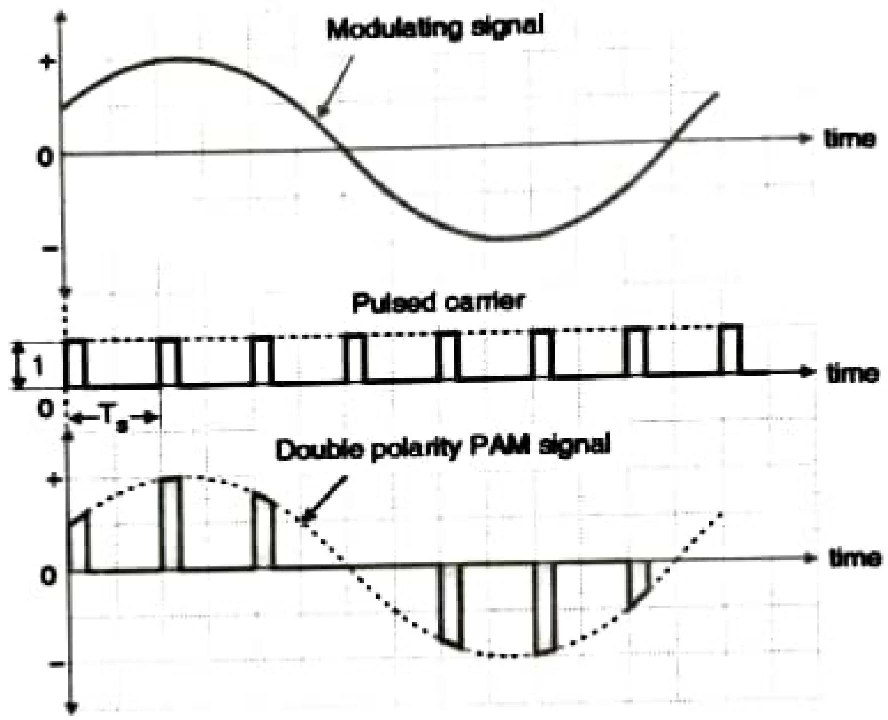
1. Natural PAM
2. Flat top PAM

Refer Fig. 9.2.1 to understand the generation of natural PAM.

The continuous modulating signal x(t), is passed through a low pass filter. The LPF will bandlimit this signal to f_m. That means all the frequency components higher than the frequency f_m are removed. Bandlimiting is necessary to avoid the "aliasing" effect in the sampling process.

The pulse train generator generates a pulse train at a frequency f_s, such that $f_s \geq 2f_m$. Thus the Nyquist criteria is satisfied.

The rectangular narrow carrier pulses generated by the pulse train generator would carry out the uniform "sampling" in the multiplier block, to generate the PAM signal as shown in Fig. 9.2.2. The samples are placed T_s seconds away from each other.



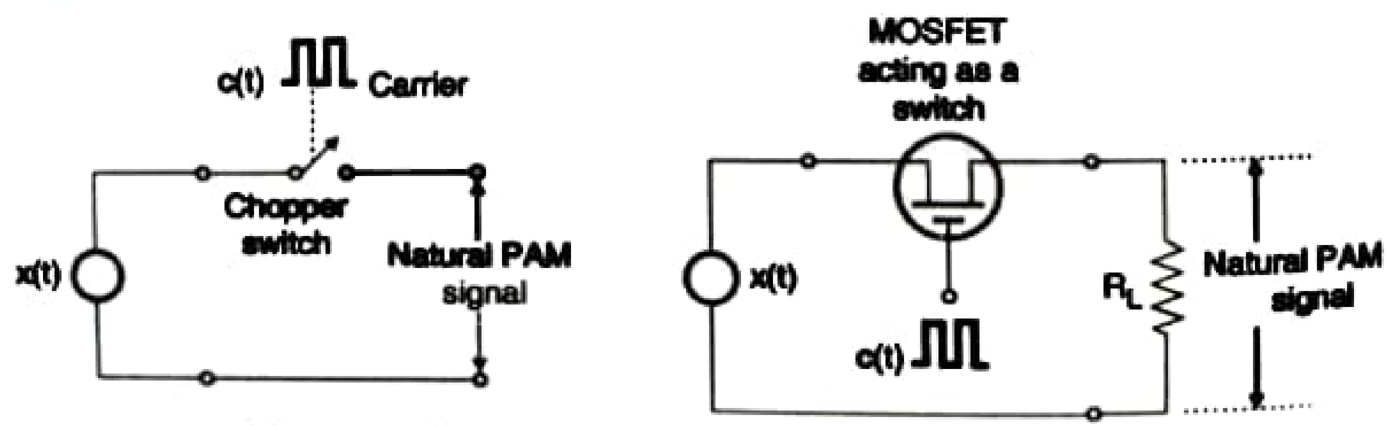
(L-182) **Fig. 9.2.2 : Waveform of natural PAM**

The "information" in the modulating signal is contained in the "Amplitude variations" of the pulsed carrier. Therefore this system is similar to the AM system discussed earlier.

Natural PAM is sometimes called as chopper sampled PAM because the waveform of the sampled signal appears to be chopped off from the continuous time signal x(t).

The chopper arrangement is as shown in Fig. 9.2.3 where the chopper switch is being operated by the pulsed carrier "c (t)".

Circuit arrangement for natural PAM :



(a) Natural PAM

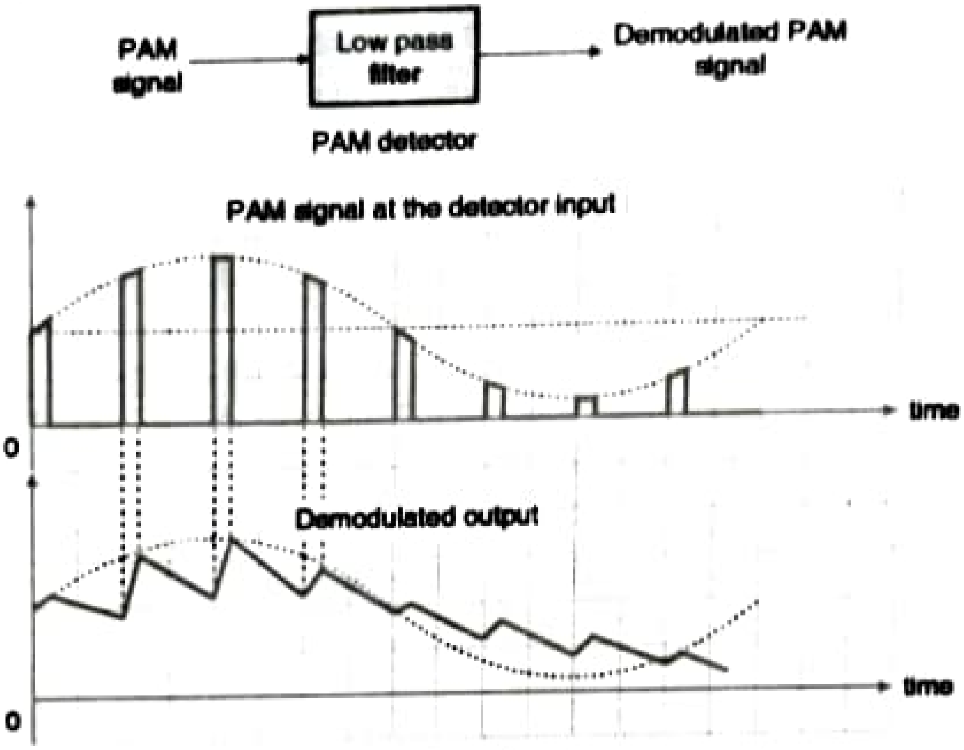
(b) Natural PAM with a MOSFET acting as a switch

The PAM signal can be detected (demodulated) by passing it through a low pass filter.

The low pass filter cutoff frequency is adjusted to f_m so that all the high frequency ripple is removed and the original modulating signal is recovered back.

The PAM detection and the corresponding waveforms are as shown in Fig. 9.2.5.

From the waveforms, it is seen that the demodulated output signal is close to the original modulating signal x (t).

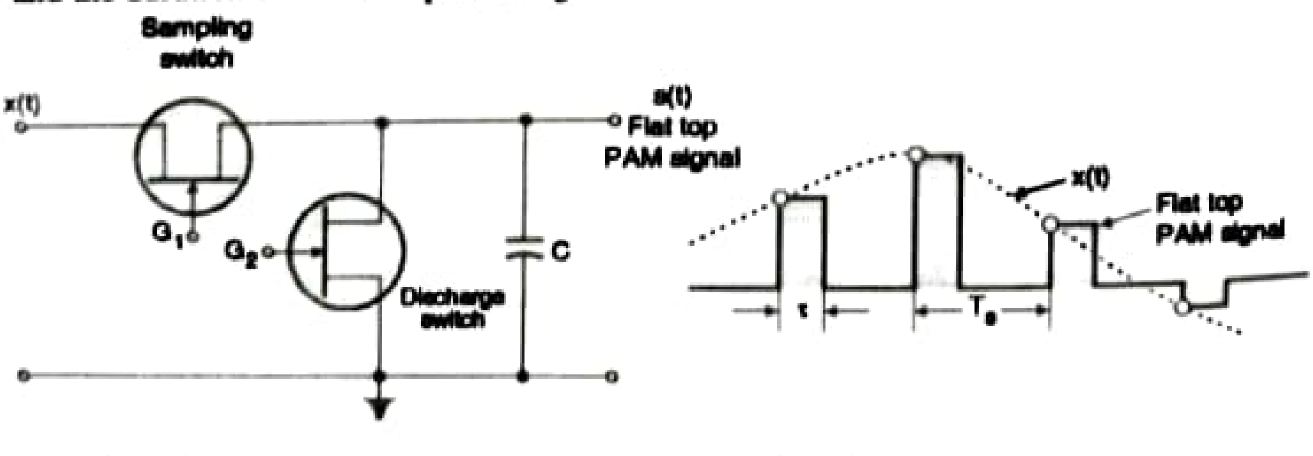


(L-185) **Fig. 9.2.5 : Detection of PAM and waveforms**

.3 Flat Top PAM :

The natural PAM is rarely employed in practice. Instead the flat top PAM is employed in practice.

In the flat top PAM technique, the analog signal x(t) is sampled instantaneously at the rate $f_s = \frac{1}{T_s}$ and the duration of each sample is lengthened to a duration "τ" as shown in Fig. 9.2.6.



(a) Sample and hold circuit to obtain the flat top PAM

(b) Flat top PAM signal

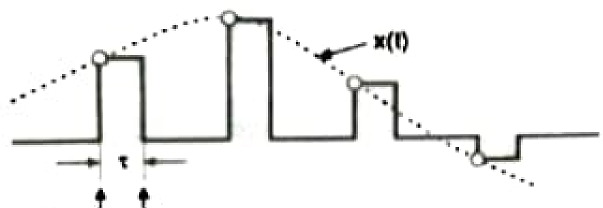
(D-449) **Fig. 9.2.6**

Thus the amplitudes of these pulses are constant and equal to the corresponding sampled values.

The flat top PAM can be obtained by using the sample and hold circuit shown in Fig. 9.2.6(a).

Operation of the sample and hold circuit :

- The sample and hold circuit consists of two FET switches and a capacitor as shown in Fig. 9.2.6(a). The analog signal x (t) is applied at the input of this circuit and the flat topped PAM signal s (t) is obtained across the capacitor.
- A gate pulse will be applied to gate G_1 at the instant of sampling, for a very short time. The sampling switch will turn on and the capacitor charges through it to the sample value $x(nT_s)$. This is instantaneous value of x (t) at instant $t = nT_s$ where n = 0, 1, 2.... The sampling switch is then turned off. Both the FETs will remain OFF for a duration of "τ" seconds and the capacitor will hold the voltage across it constant for this period. Thus the pulse is stretched to "τ" seconds and we get a pulse with a flat top.
- At the end of the pulse interval (τ), a pulse is applied to G_2 i.e. gate terminal of discharge FET. This will turn on the discharge FET and short circuit the capacitor. The output voltage then reduces to zero. This is as shown in Fig. 9.2.7.



(D-450) **Fig. 9.2.7 : Operation of sample and hold circuit**

Principle of generating the flat top PAM pulses :

- Flat top PAM is same as flat top sampled signal discussed in section 9.2.3. So the methods of generation, detection, spectrum, aperture effect etc will be exactly the same.

9.2.4 Spectrum of Flat Top PAM Signal :

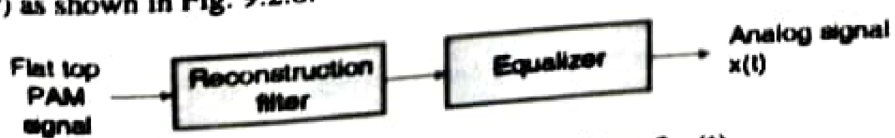
The spectrum of flat top PAM signal is same as that of the flat top sampled signal. It is given by :

Spectrum of flat top PAM signal

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - n f_s) \cdot H(f) \qquad \ldots(9.2.2)$$

9.2.5 Reconstruction of Original Signal x (t) :

- Due to the aperture effect discussed earlier, an amplitude distortion as well as a delay is introduced in the flat top sampled signal.
- This distortion can be corrected by connecting an equalizer after the reconstruction filter (low pass filter) as shown in Fig. 9.2.8.



(D-451) **Fig. 9.2.8 : Reconstruction of x (t)**

9.2.6 Merits and Demerits of Flat Top PAM :

1. Better SNR due to increased signal power. This is due to the finite width "τ ... s.
2. Generation is easy.
3. Practical filters can be used for reconstruction.
4. Aperture effect introduces distortion.

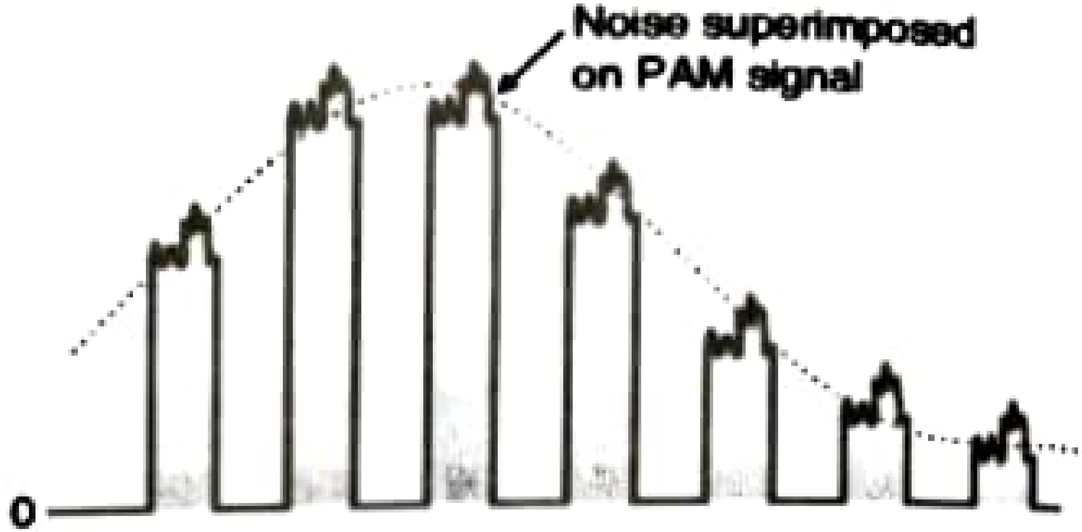
.10 Effect of Noise on PAM :

The amplitude of the pulsed carrier is being changed in proportion with the amplitude of modulating signal in PAM.

Hence all the "information" is contained in the amplitude variation of the PAM signal.

When PAM signal travels over a communication channel, noise gets added to it as shown in Fig. 9.2.9.

Note that the noise distorts the amplitude of PAM signal. Since the information is contained in the amplitude, the noise will contaminate the information.



(L-190) Fig. 9.2.9 : Effect of noise on PAM signal

Therefore the noise performance of PAM system is very poor.

The PWM and PPM systems have a better noise performance.

11 Applications of PAM :

PAM is used in the PAM-TDM system.

PAM is used in the PAM-FM system.

Pulse Width Modulation (PWM) or Pulse Duration

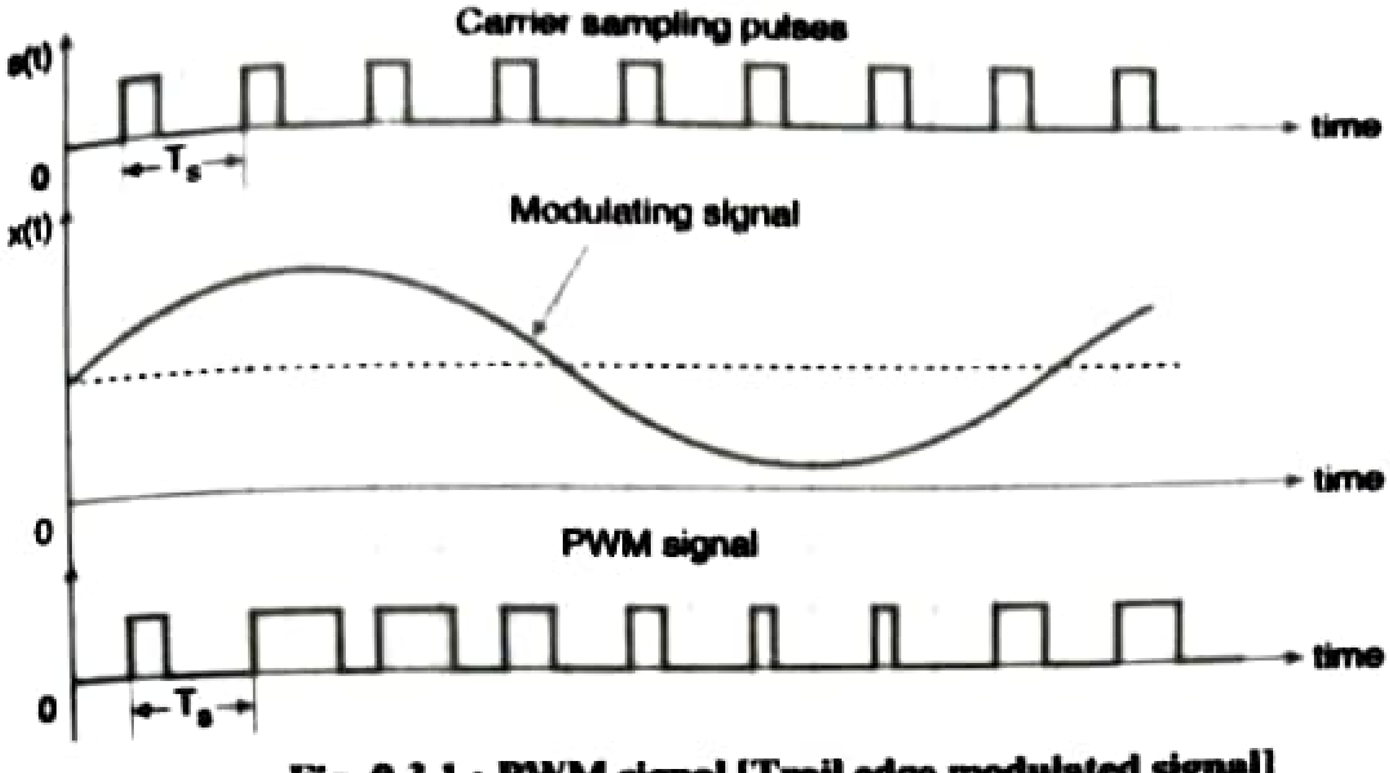
The other type of a pulse analog modulation is the pulse width modulation (PWM). In PWM, the width of the carrier pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM are as shown in Fig. 9.3.1.

As seen from the waveforms, the amplitude and the frequency of the PWM wave remains constant. Only the width changes.

That is why the "information" is contained in the width variation. This is similar to FM. As the noise is normally "additive" noise, it changes the amplitude of the PWM signal.

At the receiver, it is possible to remove these unwanted amplitude variations very easily by means of a limiter circuit.

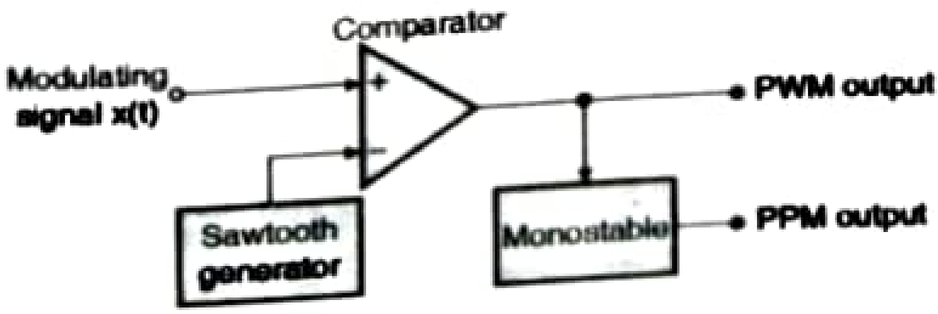
As the information is contained in the width variation, it is unaffected by the amplitude variations introduced by the noise. Thus the PWM system is more immune to noise than the PAM signal.



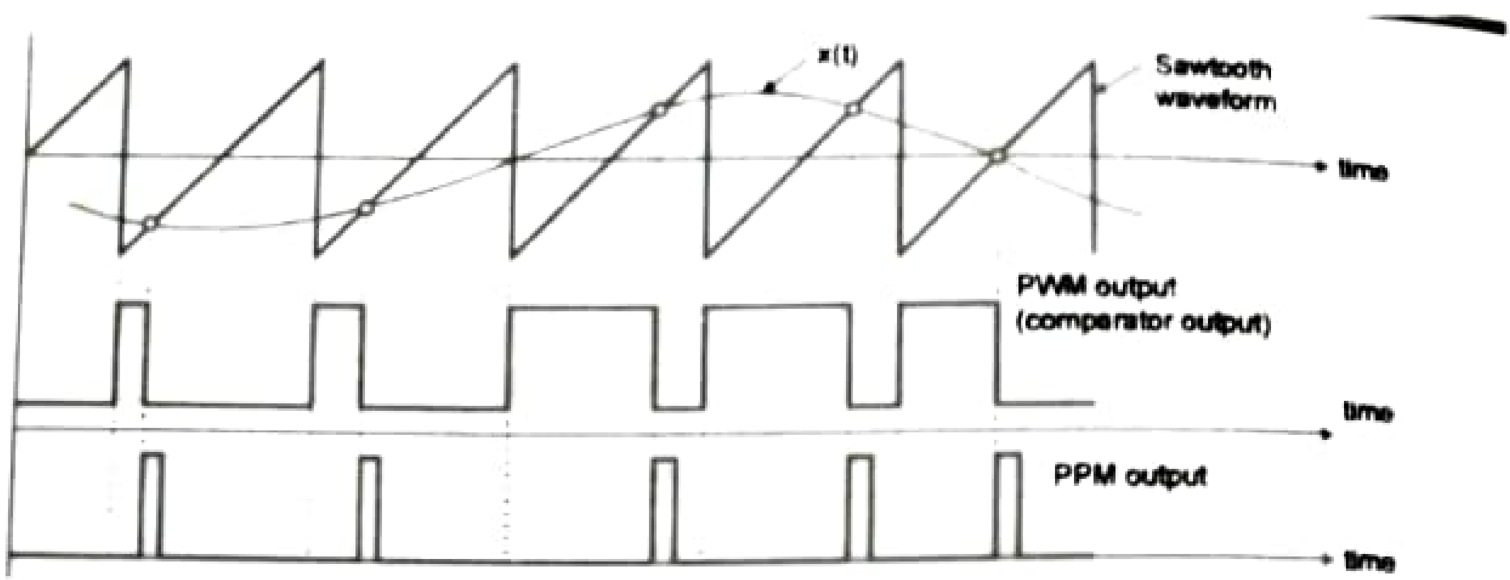
(D-454) Fig. 9.3.1 : PWM signal [Trail edge modulated signal]

Generation of PWM Signal :

The block diagram of Fig. 9.3.2(a) can be used for the generation of PWM as well as PPM.



(D-455) **Fig. 9.3.2(a) : PWM and PPM generator**



(D-456) **Fig. 9.3.2(b) : Waveforms**

A sawtooth generates a sawtooth signal of frequency f_s, therefore the sawtooth signal in this case is a sampling signal. It is applied to the inverting terminal of a comparator.

The modulating signal x (t) is applied to the non-inverting terminal of the same comparator.

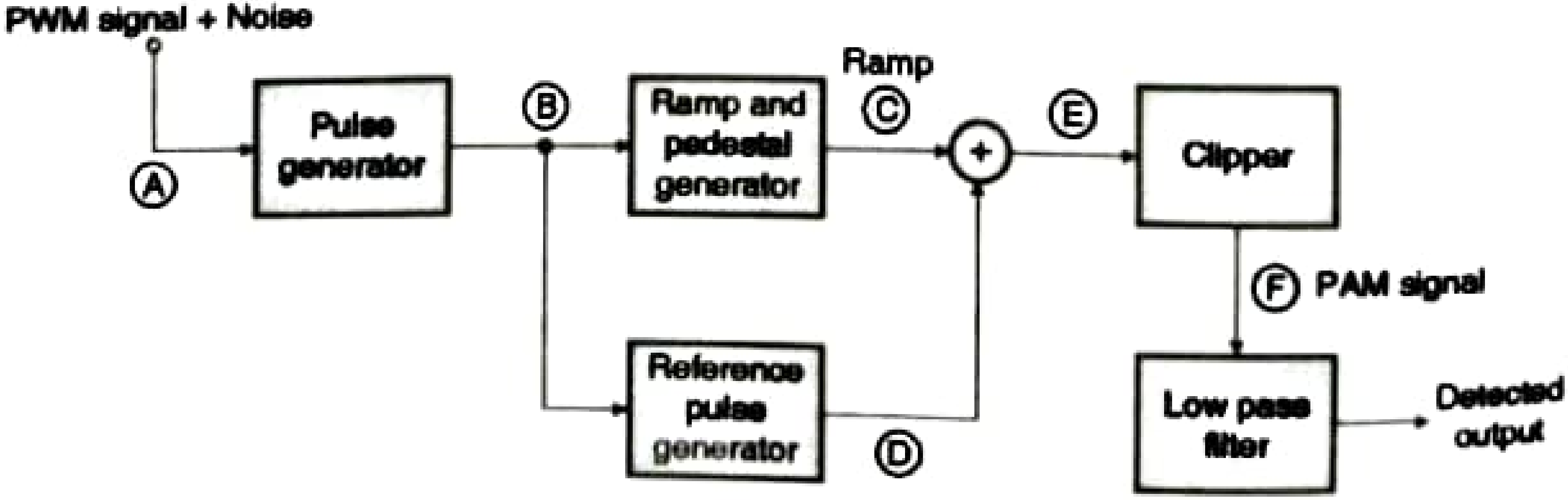
The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal.

This gives rise to a PWM signal at the comparator output as shown in Fig. 9.3.2(b).

Note that the leading edges of the PWM waveform coincide with the falling edges of the ramp signal.

Thus the leading edges of PWM signal are always generated at fixed time instants. However the occurrence of its trailing edges will be dependent on the instantaneous amplitude of x (t).

Therefore this PWM signal is said to be trail edge modulated PWM.



(D-457) **Fig. 9.3.3 : PWM detection circuit**

eration :

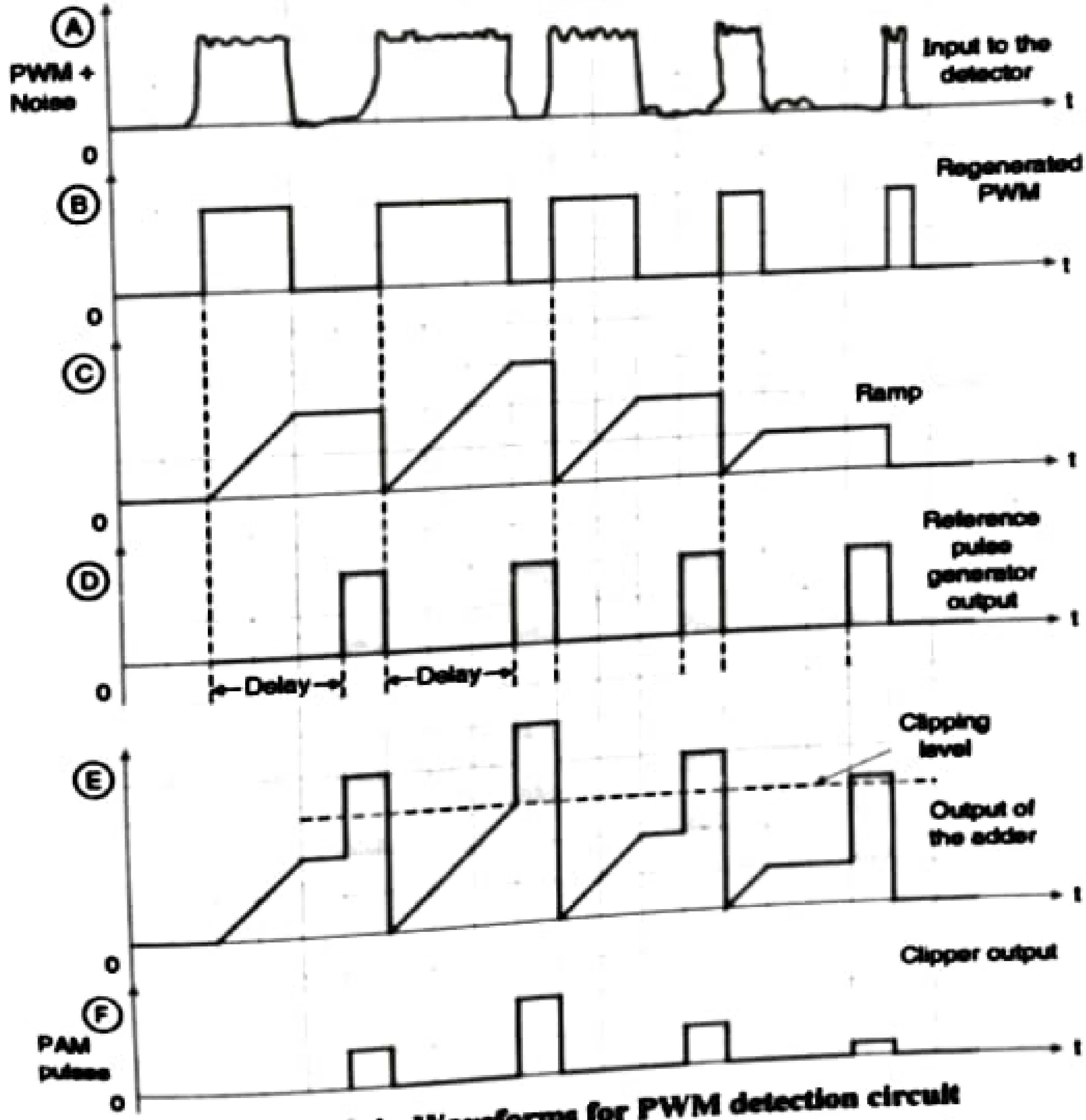
The PWM signal received at the input of the detection circuit is contaminated with noise. This signal is applied to pulse generator circuit which regenerates the PWM signal. Thus some of the noise is removed and the pulses are squared up.

The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude, constant width pulses. These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval.

The regenerated PWM pulses are also applied to a ramp generator. At the output of it we get a constant slope ramp for the duration of the pulse. The height of the ramp is thus proportional to the widths of the PWM pulses. At the end of the pulse a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.

The constant amplitude pulses at the output of reference pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper.

A low pass filter is used to recover the original modulating signal back from the PAM signal.



(D-458) Fig. 9.3.4 : Waveforms for PWM detection circuit

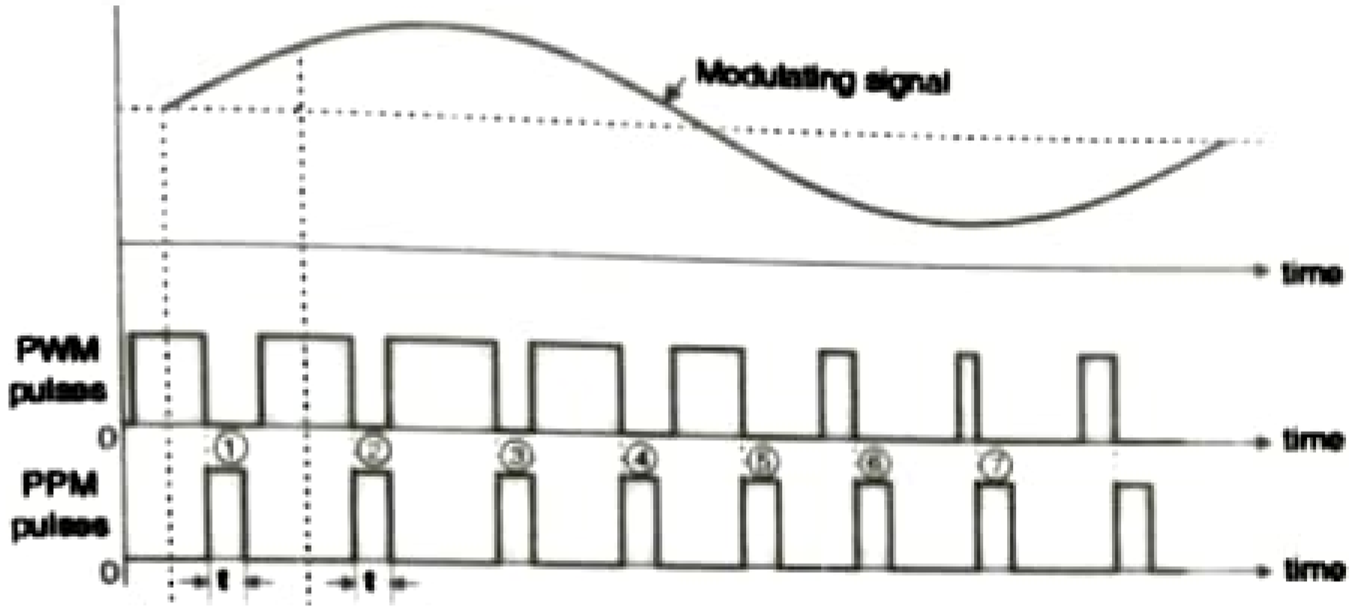
1. Less effect of noise i.e. very good noise immunity.
2. Synchronization between the transmitter and receiver is not essential. (Which is essential in PPM).
3. It is possible to reconstruct the PWM signal from a noise contaminated PWM, as discussed in the detection circuit. Thus it is possible to separate out signal from noise (which is not possible in PAM).

Due to the variable pulse width, the pulses have variable power contents. So the transmitter must be powerful enough to handle power corresponding to the maximum width pulse. The average power transmitted can be as low as 50% of this maximum power.
In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to BW of PAM.

Applications of PWM :

In the optical fiber communication. 2. In some military applications.

- In PPM the amplitude and width of the pulsed carrier remains constant but the position of each pulse is varied in proportion with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.
- The PPM pulses can be derived from the PWM pulses as shown in Fig. 9.4.1. Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.
- The vertical dotted lines drawn in Fig. 9.4.1 are treated as reference lines to measure the shift in position of PPM pulses. The leading edge of each PPM pulse coincides with the trailing pulse of a PWM pulse.



(D-460) **Fig. 9.4.1 : PPM pulses generated from PWM signal**

The PPM pulses marked 1, 2 and 3 etc. in Fig. 9.4.1 go away from their respective reference lines.

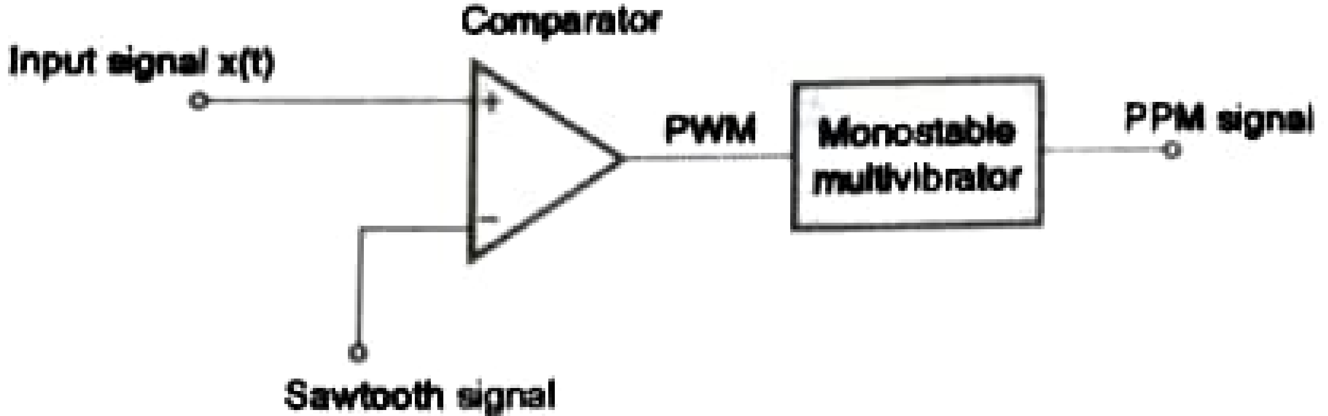
This is corresponding to increase in the modulating signal amplitude. Then as the modulating voltage decreases the PPM pulses 4, 5, 6, 7 come progressively closer to their respective reference lines.

4.1 Generation of PPM Signal :

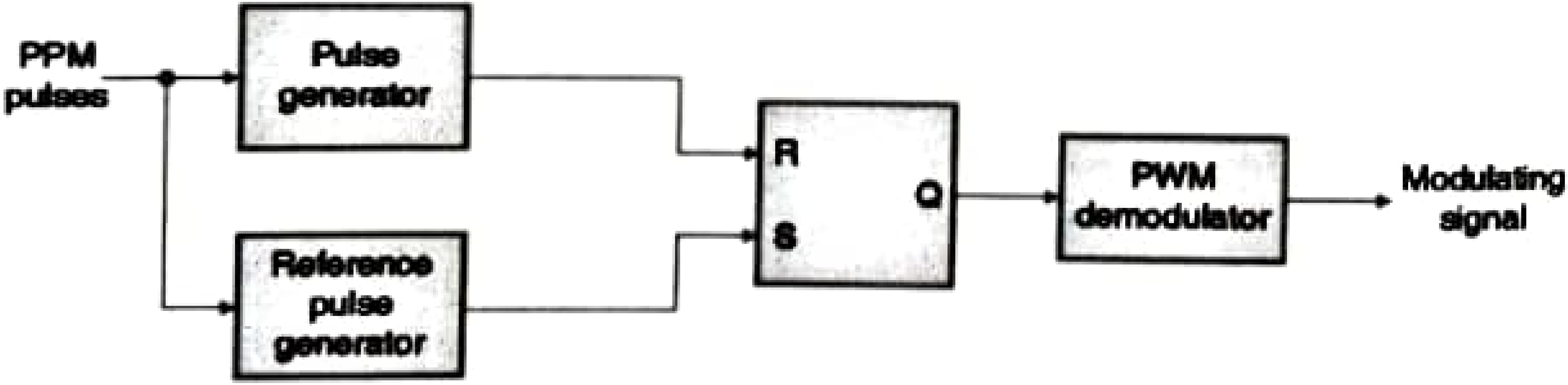
The PPM signal can be generated from PWM signal as shown in Fig. 9.3.2(a). The same block diagram has been repeated in Fig. 9.4.2 as shown.

The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered.

- Hence corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components.
- Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting as shown in Fig. 9.4.1.



The PPM demodulator block diagram is as shown in Fig. 9.4.3.



(D-462) **Fig. 9.4.3 : PPM demodulator circuit**

operation of the demodulator circuit is explained as follows :

The noise corrupted PPM waveform is received by the PPM demodulator circuit.

The pulse generator develops a pulsed waveform at its output of fixed duration and apply these pulses to the reset pin (R) of a SR flip-flop.

A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses.

Due to the set and reset signals applied to the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator. This is same as the one discussed in section 9.3.2.

Due to constant amplitude of PPM pulses, the information is not contained in the amplitude. Hence the noise added to PPM signal does not distort the information. Therefore it has good noise immunity. This is same as that explained for PWM in section 9.3.2.

It is possible to reconstruct PPM signal from the noise contaminated PPM signal. This is also possible in PWM but not possible in PAM.

Due to constant amplitude of pulses, the transmitted power always remains constant. It does not change as it used to, in PWM.

As the position of the PPM pulses is varied with respect to a reference pulse, a transmitter has to send synchronizing pulses to operate the timing circuits in the receiver. Without them the demodulation won't be possible to achieve.

Large bandwidth is required to ensure transmission of undistorted pulses.

Applications :

PPM is used in some military applications.