Unit-IV-Pulse modulation & Digital Modulation

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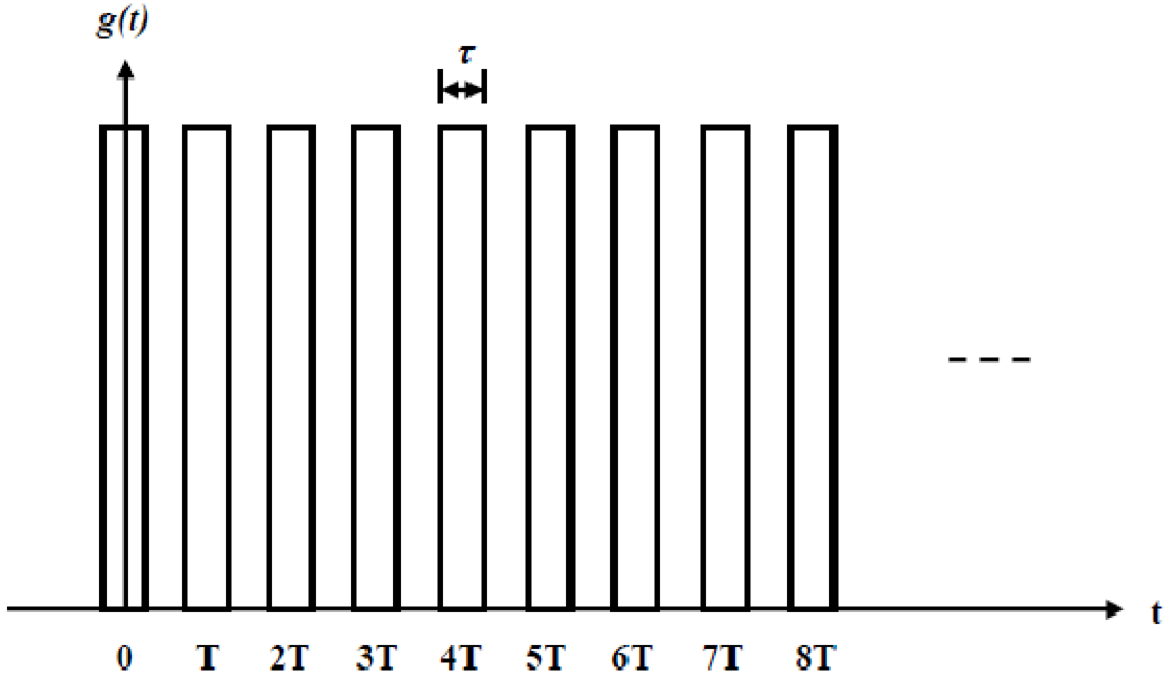
Objective

1. To introduce the concepts of pulse modulation
2. To introduce and understand sampling theorems
3. To discuss about different types of pulse modulation techniques

Pulse Modulation

- The process of changing the characteristics of pulse carrier in accordance with the modulating signal is called pulse modulation.
- The major points of difference between analog modulation and pulse modulation are; in analog modulation technique simple or complex sinusoidal signal is considered as a carrier signal, whereas in pulse modulation, carrier signals are periodic rectangular trains of pulse signals.



- Fig Carrier signal format in pulse modulation

Types of Pulse modulation

- The various types of analog pulse modulation techniques based on these characteristics are:
  1. Pulse amplitude modulation (PAM)
  2. Pulse width modulation (PWM)
  3. Pulse position modulation (PPM)

- Types of Digital Pulse modulation:
  1. Pulse code modulation (PCM)
  2. Delta modulation
  3. Adaptive delta modulation

Pulse amplitude modulation (PAM)

- The process of changing the amplitude of the pulse carrier signal in accordance with the modulating signal is called "Pulse Amplitude Modulation" (PAM) which is also referred to as "*sampling process*".

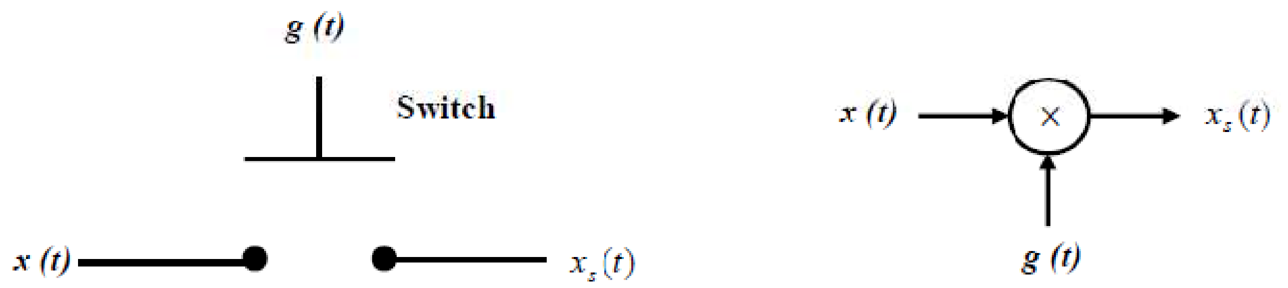


Fig: Model of a sampler

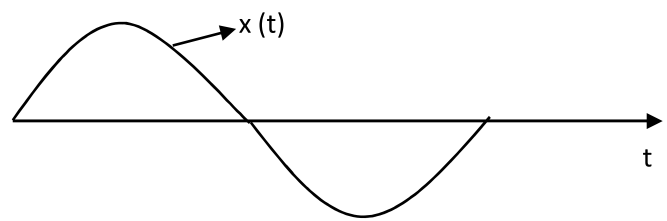
Generation of PAM signals / Sampling operation

- PAM signals can be generated using electronic switches. The inputs for the electronic switches are continuous time modulating signal and train of periodic pulses. When the switch is closed, the corresponding instant message signals are arrived at the output side. The output is zero when the switch is open. Continuous time signals are converted into discrete-time signal due to this process. The output signals are sampled signals.

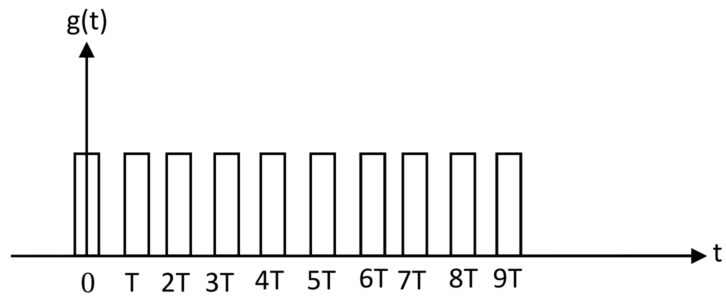
Sampling operation

Input or

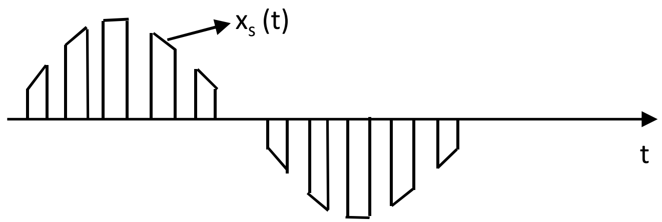
message signal = x (t)



Carrier signal = g(t)



Sampled signal = x_s (t)



Mathematical representation

Let x(t) be the modulating signal (continuous)

g(t) be the carrier (pulse) signal and $x_s(t)$ sampled signal.

Now g(t) are periodic train of pulses. Any periodic signal can represent by Fourier series. Hence Fourier series representation of carrier signal g(t) is given by

$$g(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f_s t} \qquad (1)$$

where C_n is a n^{th} Fourier coefficients and it can be expressed as,

$$C_n = \frac{1}{T} \int_{-\frac{f_s}{2}}^{\frac{f_s}{2}} g(t) e^{-j2\pi n f_s t} \qquad (2)$$

The output of a electronic switch or sampler circuit is given as

$$x_s(t) = x(t) g(t) \qquad (3)$$

This expression describes the sampling operation in time domain.

- The sampled signal in frequency domain can also be expressed by finding the Fourier transform of a signal $x_s(t)$

$$x(t) \xrightarrow{FT} x(f)$$

$$x(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt \qquad (4)$$

The frequency domain signal of sampled signal can be obtained in a similar manner.

$$x_s(t) \xrightarrow{FT} x_s(f)$$

$$x_s(f) = \int_{-\infty}^{\infty} x_s(t) e^{-j2\pi f t} dt \qquad (5)$$

Sub (3) in (5), we get

$$x_s(f) = \int_{-\infty}^{\infty} x(t) g(t) e^{-j2\pi f t} dt \qquad (6)$$

Sub value of g(t) from eq. (1) to eq. (6)

$$= \int_{-\infty}^{\infty} x(t) \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f_s t} e^{-j2\pi f t} dt$$

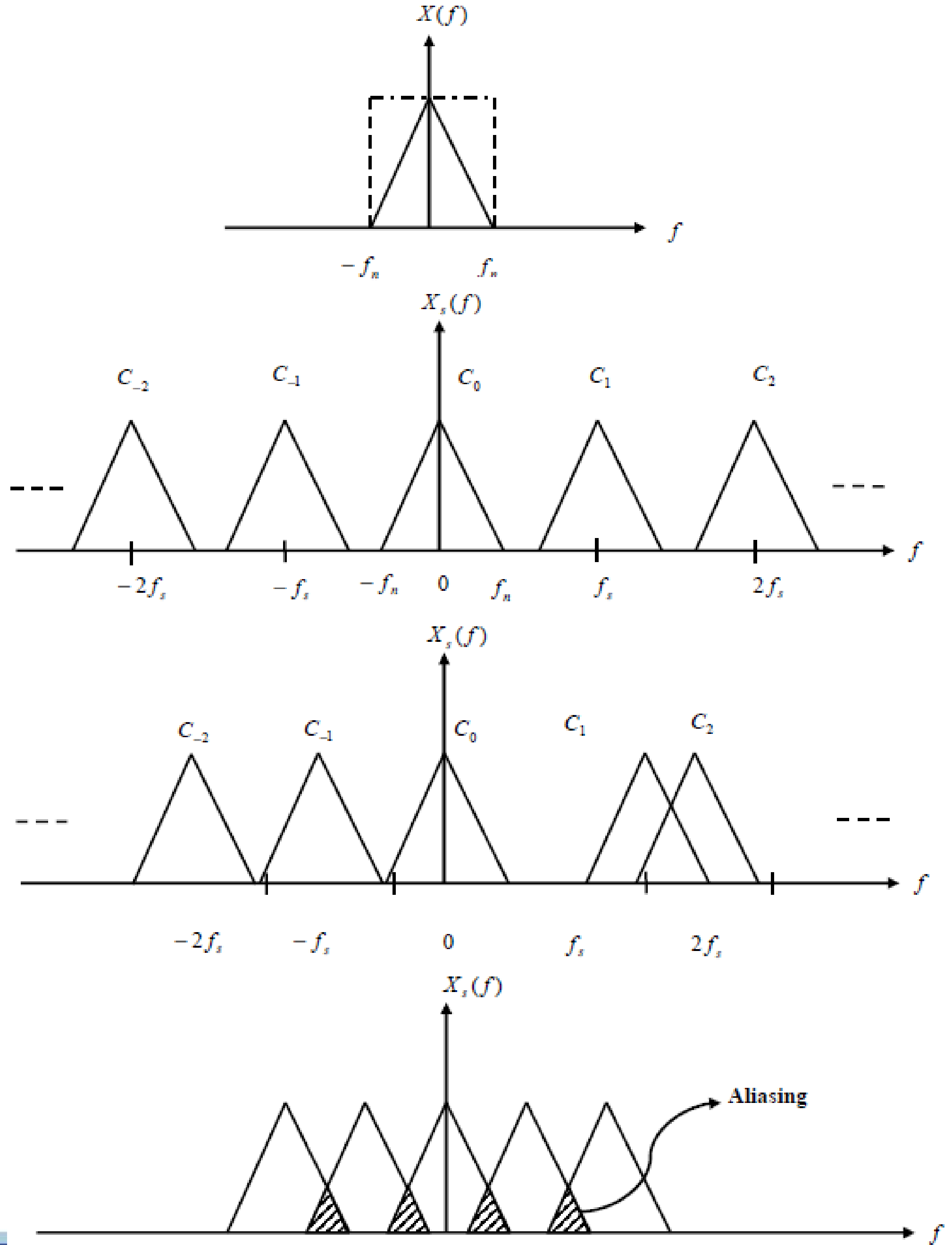
$$= \sum_{n=-\infty}^{\infty} C_n \int_{-\infty}^{\infty} x(t) e^{-j2\pi(f-nf_s)} dt$$

From the definition of Fourier transform,

$$\int_{-\infty}^{\infty} x(t) e^{-j2\pi(f-nf_s)} dt = X(f-nf_s)$$

$$x_s(f) = \sum_{n=-\infty}^{\infty} C_n X(f-nf_s) \qquad (7)$$

Eq. (7) can be represented pictorially, as shown in figure. It shows sampling in time domain introducing periodicity in the frequency domain. i.e. the same band limited spectrum repeated for every sample interval.


$X(f)$
f
$-f_n$
f_n
$X_s(f)$
C_{-2}
C_{-1}
C_0
C_1
C_2
$-2f_s$
$-f_s$
$-f_n$
0
f_n
f_s
$2f_s$
Aliasing

- The conclusion drawn from the figure are:

1. Band limited signal $x(t)$ can be perfectly recovered at the receiver side only when the sampling frequency is $f_s \geq 2 f_h$.
2. When sampling frequency does not satisfy the above condition, recovery of the original signal at the receiver side is not possible. Also the signals are also affected by adjacent samples. This effect is referred to as "Aliasing effect". Hence the condition for Aliasing : $f_s \leq 2 f_h$.

In other words, for perfect reconstruction of sampling interval (T_s), is always higher than $\frac{1}{2T}$. i.e. rate of closer of electronic switch at the transmitter side must satisfy the condition of Nyquist interval.

Sampling Theorem or Nyquist Criterion for sampling operation

A band limited signal $x(t)$ *which has no frequency components above* f_h*, can be completely specified by samples at a rate greater than or equal to* $2f_h$.

i.e. $f_s \geq 2f_h$ *(8)*

Perfect recovery of the signal is not possible at the receiver side, when the sampling interval at the transmitter side is not followed.

Detection of PAM signal

- Just passing the spectrum of $x_s(f)$ to low pass filter may be done for reconstructing the signal. Low pass filter allows the spectrum from the band $-f_h$ to $+f_h$ and suppresses all other side bands. Fig. illustrates the detection circuit of PAM signal.

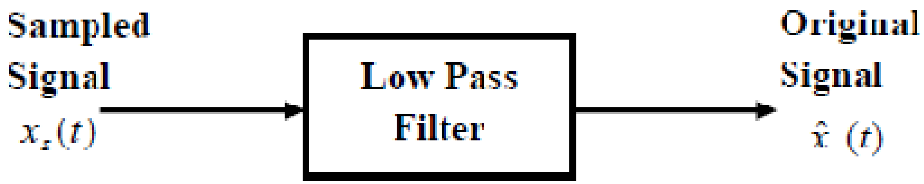


Fig: Detection of PAM signal