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Comprehensive Theory *with* Solved Examples

Civil Services Examination

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Digital Communication

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CHAPTER 1

Pulse Modulation

Introduction

The modulation technique in which transmitted signal is in the form of digital pulses is called digital modulation system. Normally, the signals produced from various sources is analog in nature. e.g. audio signal captured in microphone, video signal (infinite possibilities of colour at single point and hence is continuous) all are analog signals. These can be converted into digital form using ADC (Analog to Digital converters) because there are certain advantages of digital transmission over analog transmission.

1.1 Analog Communication Versus Digital Communication

1.1.1 Advantages of Digital Communication

- Due to digital nature of transmitted signals, the interference of additive noise (analog) is less. Hence better noise immunity.
- Channel coding techniques makes it possible to detect and correct the errors introduced during transmission.
- Repeaters used between transmitter and receiver helps to regenerate digital signal.
- It is simple and cheap.
- Multiplexing technique can be used to transmit many voice signals over common channel.

1.1.2 Drawback of Digital Communication

- Bandwidth requirements are high.
- Synchronization is needed in case of coherent digital communication systems.

1.1.3 Applications of Digital Communication

- Long distance communication between earth and spaceships.
- Satellite communication.
- Military communication which needs coding.
- Data and computer communications.

In continuous wave modulation studied earlier, some parameter of a sinusoidal carrier wave is varied continuously in accordance with message signal. Similarly, in pulse modulation, which we are going to study in this chapter, some parameter of a pulse train is varied in accordance with the message signal.

Pulse modulation can be classified in two families:

1. Analog Pulse Modulation

In this, a periodic pulse train is used as the carrier wave. Some characteristic of each pulse (e.g. amplitude, duration or position) is varied in a continuous manner in accordance with corresponding sample value of message signal. Here, information is transmitted in analog form but transmission occurs at discrete times.

2. Digital Pulse Modulation

Here, message signal is transmitted in the form that is discrete in both domains-amplitude as well as time. For conversion of any analog signal into digital domain, two operations are necessary, namely, sampling and quantization.

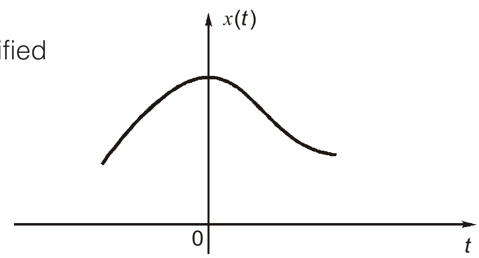
1.2 Sampling Theory

The very basic operation in digital processing and digital communication is sampling. In sampling process, an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time domain.

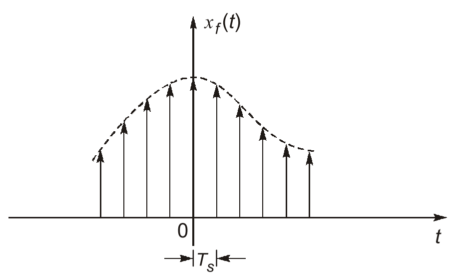
Consider an arbitrary signal $x(t)$ of finite energy which is specified for all time as shown in figure.

A segment of this signal $x(t)$ is shown as in the figure:

Suppose that we sample the signal $x(t)$ instantaneously and at a uniform rate, once every T_s seconds, as shown below :



Message signal



Sampled signal

Hence, an infinite sequence of samples is obtained spaced T_s seconds apart and denoted by $x(nT_s)$ where n is any integer.

Here, T_s = Sampling period

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

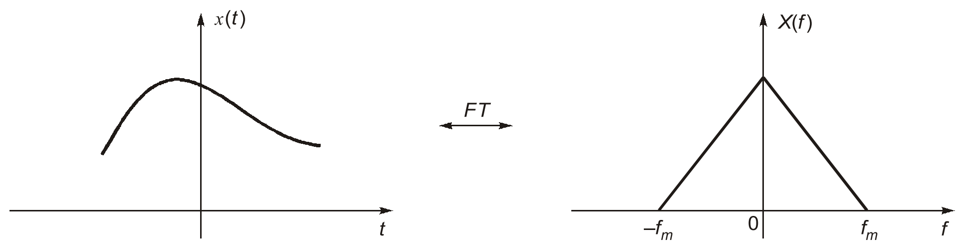
1.2.1 Sampling Theorem

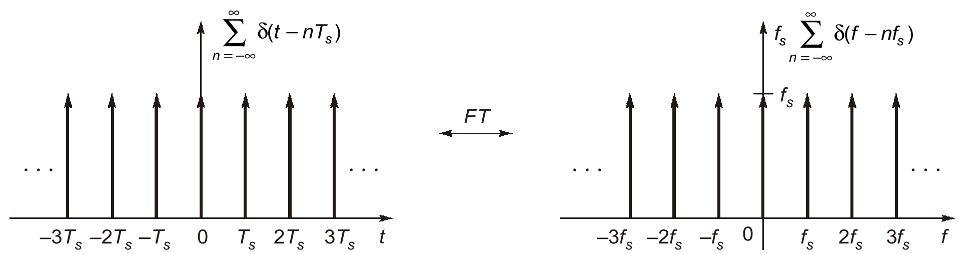
Sampling theorem provides a method of reconstruction of original signal from sampled values and also gives a precise upper bound on the sampling rate required for distortionless reconstruction.

Sampling theorem can be stated in two ways:

1. A continuous message signal band limited to f_m Hz, can be converted to its sampled equivalent without information loss provided $f_s \geq 2f_m$ samples/sec or $T_s \leq \frac{1}{2f_m}$ sec.
2. A continuous message signal band limited to f_m Hz can be perfectly recovered from its sampled equivalent provided $f_s \geq 2f_m$ or $T_s \leq \frac{1}{2f_m}$ sec.

Proof of Sampling Theorem





Fourier transform of $s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$ is given by

$$s(t) = \sum_{k=-\infty}^{\infty} a_k\, e^{jk\omega_o t}$$

$$a_k = \frac{1}{T}\int_{-T/2}^{T/2} s(t)\, e^{-jk\omega_o t} dt = \frac{1}{T_s}\int_{-T_s/2}^{T_s/2} \delta(t) e^{-jk\omega_o t} dt$$

Now, $\delta(t)\, F(t) = \delta(t)\, F(0)$

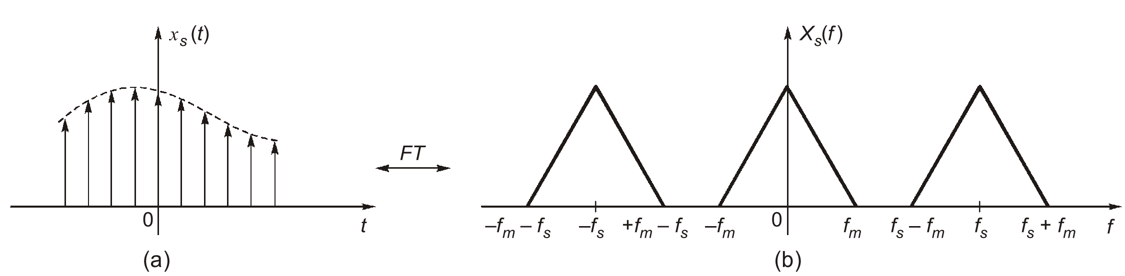
$\therefore$ $$a_k = \frac{1}{T_s}\int_{-T_s/2}^{T_s/2} \delta(t)\, dt = \frac{1}{T_s}$$

By using property of representation of Fourier transform of periodic signals, Fourier transform of $\sum_{n=-\infty}^{\infty} \delta(t - nT_s)$ is given by

$$= a_k \sum_{n=-\infty}^{\infty} \delta(f - nf_s) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

Now, $$x_s(t) = x(t) \times \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \;\Rightarrow\; X_s(f) = X(f) * f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

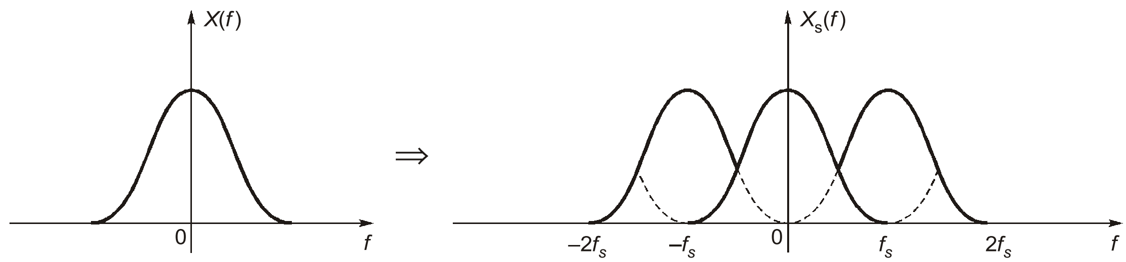
$$X_s(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$



Sampled signals (b) Spectrum of a sampled signal with $f_s > 2f_m$

The derivation of sampling theorem, as described, is based on the assumption that the signal $g(t)$ is band limited to f_m Hz. In general, however, a signal is not band limited. Consequently, some aliasing is produced by the sampling process.

"Aliasing refers to the phenomenon of high frequency component in the spectrum of the signal seemingly taking on the identity of a lower frequency in the spectrum of its sampled version."

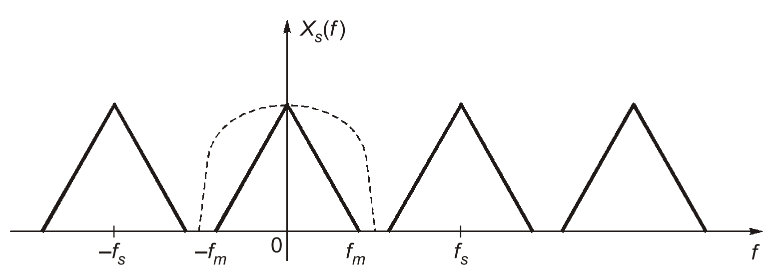


Spectrum of signal under aliasing ($f_s < 2f_m$)

To avoid aliasing, following measures are used:

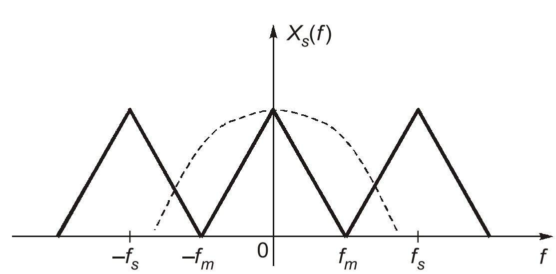
1. Prior to sampling, a low pass pre-alias filter is used to attenuate those high frequency components of the signal that are not essential to the information being conveyed by the signal.
2. Filtered signal is sampled at a rate higher than Nyquist rate.

(i) **Over sampling:** $(f_s > 2f_m)$ or $\left(T_s < \frac{1}{2f_m} \text{ sec.}\right)$



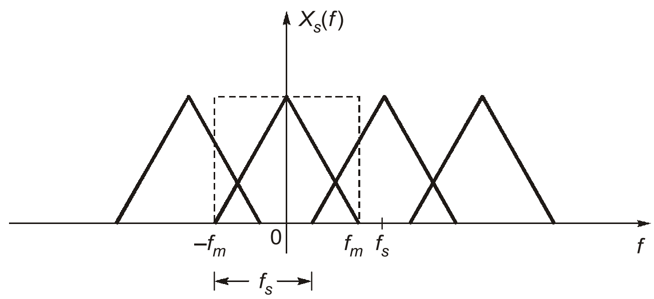
This is most preferred way of sampling because even practical LPF can be used for reconstruction.

(ii) **Critical sampling:** $\left(f_s = 2f_m \text{ or } T_s = \frac{1}{2f_m}\right)$



Only ideal LPF can be used for reconstruction because when practical LPF is used, some extra frequency components are present which causes distortion.

(iii) Under sampling: $\left(f_s < 2f_m \text{ or } T_s > \dfrac{1}{2f_m}\right)$



In this case, loss of message signal is irrecoverable and even ideal LPF doesn't help. This method is never preferred.

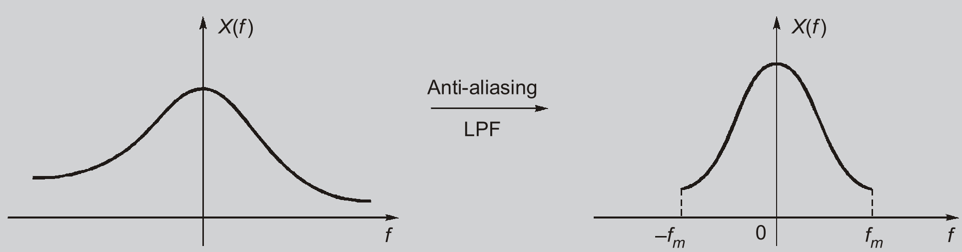
1.2.2 Nyquist Rate and Nyquist Interval

The minimum possible sampling rate and maximum possible sampling interval allowed to avoid aliasing are called as nyquist rate and nyquist interval respectively.

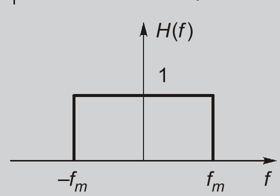
Nyquist rate = $2f_m$ samples/sec ; Nyquist interval = $\dfrac{1}{2f_m}$ seconds.

NOTE

The LPF used prior to sampling to band limit the signal is called as anti-aliasing filter.

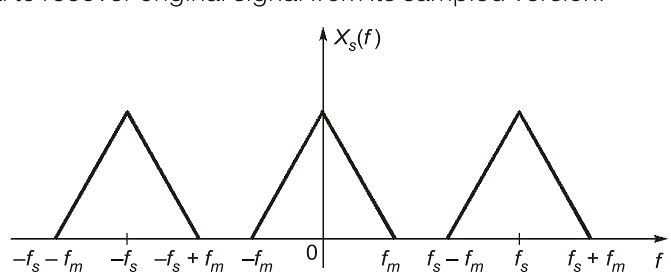


Anti-aliasing LPF =

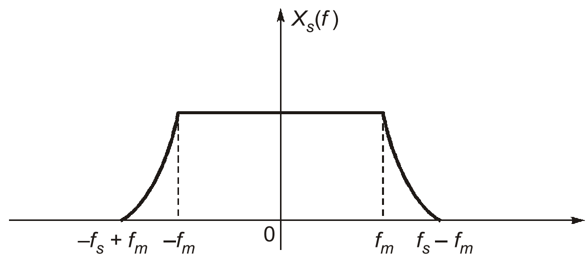


1.2.3 Reconstruction Filter

The use of sampling rate higher than the nyquist rate also has the beneficial effect of easing the design of the reconstruction filter used to recover original signal from its sampled version.



Reconstruction Filter Response



Reconstruction filter is low pass with a pass band extending from $-f_m$ to $+f_m$ which is itself determined by antialiasing filter.

Reconstruction filter has a transition band extending (for positive frequencies) from f_m to $f_s - f_m$ where f_s is sampling rate.

1.2.4 Types of Sampling

1. Natural Sampling

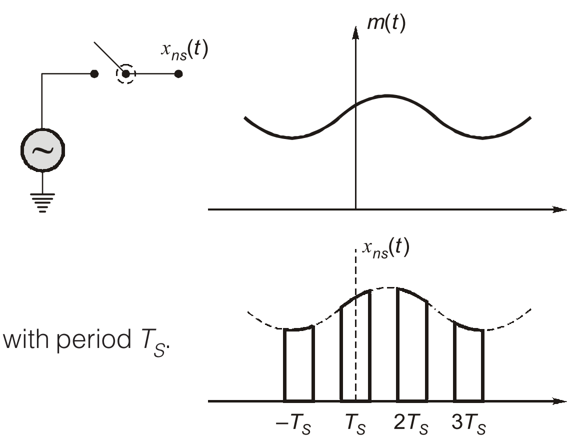
A practical way of sampling a band-limited analog signal $m(t)$ is performed by high-speed switching circuits. An equivalent circuit employing a mechanical switch and the resulting sampled signal are shown

The sampled signal $x_{ns}(t)$ can be written as

$$x_{ns}(t) = m(t) \cdot x_p(t)$$

where $x_p(t)$ in the periodic train of rectangular pulses with period T_S.

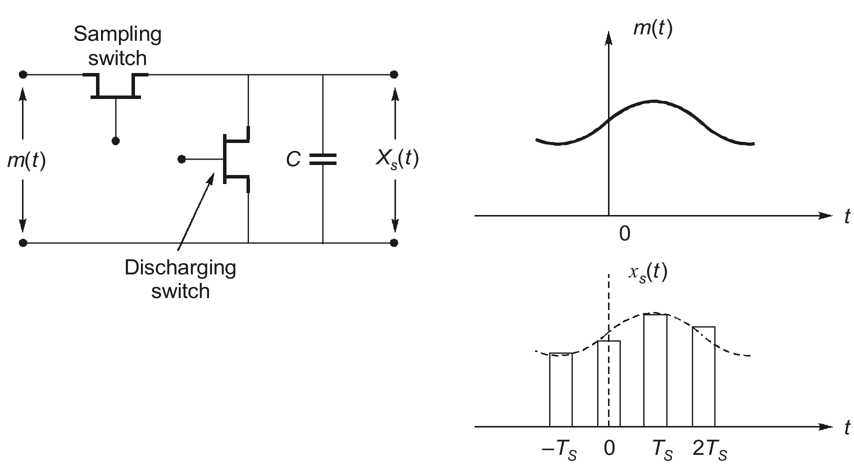
The sampling here is termed natural sampling since the top of each pulse in $x_{ns}(t)$ retains the shape of its corresponding analog segment during the pulse interval.



Natural Sampling

2. Flat-top Sampling

The simplest and thus most popular practical sampling method is actually performed by a functional block termed the sample and hold (S/H) circuit. This circuit produces a flat-top sampled signal $x_s(t)$.



Flat top sampling

Example - 1.1 Find the Nyquist rate of the following signals:

(i) sin(4000πt) (ii) sin(3000πt)sin(5000πt) (iii) sinc(400t) (iv) sinc(100t)sinc(200t)

Solution:

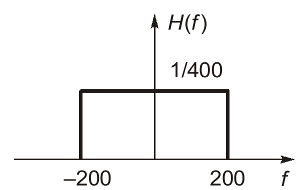
(i) sin(4000πt)

Here,
$$f_m = 2000 \text{ Hz}$$
$$f_s = 2\, f_m = 4 \text{ kHz}$$

(ii) $$\sin(3000\pi t)\sin(5000\pi t) = \frac{1}{2}(2\sin 3000\pi t \sin 5000\pi t) = \frac{1}{2}(\cos 2000\pi t - \cos 8000\pi t)$$

$$f_{\max} = 4000 \text{ Hz}$$
$$f_s = 8 \text{ kHz}$$

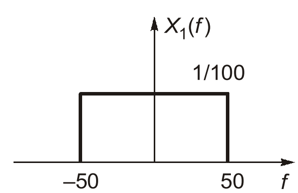
(iii) $$\text{sinc}(400t) = \frac{\sin 400\,\pi t}{400\,\pi t} \longleftrightarrow$$



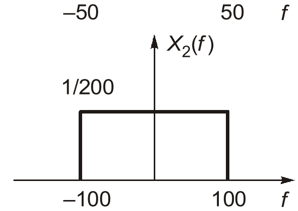
$$f_s = 400 \text{ Hz}$$

(iv) sinc(100t)sinc(200t)

$$x_1(t) = \text{sinc}100t \longleftrightarrow$$

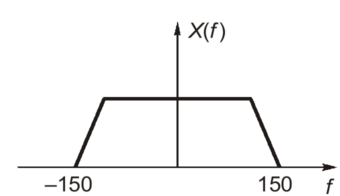


$$x_2(t) = \text{sinc}200t \longleftrightarrow$$



Multiplication in time domain is convolution in frequency domain.

$$f_{\max} = 150$$
$$f_s = 2f_{\max} = 300 \text{ Hz}$$
$$T_s = \frac{1}{f_s} = \frac{1}{300} = 0.3 \times 10^{-2} \text{sec.}$$



Example - 1.2 A signal $x_1(t)$ is band-limited to 2 kHz while $x_2(t)$ is bandlimited to 3 kHz. Find the nyquist sampling rate of the following signals:

(a) $x_1(2t)$
(b) $x_2(t-3)$
(c) $x_1(t) + x_2(t)$
(d) $x_1(t)\, x_2(t)$
(e) $x_1(t) * x_2(t)$

Solution:

(a) The spectrum of $x_1(2t)$ (time compression) stretches to 4 kHz. Thus the Nyquist rate is 8 kHz.

(b) The spectrum of $x_2(t-3)$ (time shift changes only phase) extends to 3 kHz, the Nyquist rate is 6 kHz.

(c) The spectrum of $x_1(t) + x_2(t)$ [sum of spectra] extends to 3 kHz. Thus the Nyquist rate is 6 kHz.

(d) The spectrum of $x_1(t)\,x_2(t)$ (convolution in frequency domain) extends to 5 kHz. Thus the Nyquist rate is 10 kHz.

(e) The spectrum of $x_1(t) * x_2(t)$ (product in frequency domain) extends to 2 kHz. Thus the Nyquist rate is 4 kHz.

Example - 1.3 Find the Nyquist sampling rate and the Nyquist sampling interval for each of the following signals:

(a) $5\cos(1000\pi t)\cos(4000\,\pi t)$ (b) $\text{sinc}(100\,\pi t)$

(c) $\text{sinc}^2(100\,\pi t)$ (d) $\text{sinc}(100\,\pi t) + 3\,\text{sinc}^2(60\pi t)$

(e) $\text{sinc}(50\pi t).\text{sinc}(100\pi t)$

Solution:

(a) $$5\cos 1000\pi t\cos 4000\,\pi t = 2.5[\cos(3000\,\pi t) + \cos(5000\,\pi t)]$$

The signal is band-limited to 2500Hz. Hence the Nyquist rate is 5000 Hz and the Nyquist interval is $\frac{1}{5000}s = 0.2$ ms

(b) $$\text{sinc}(100\,\pi t) \leftrightarrow \frac{1}{100\pi}\Pi\left(\frac{f}{100\pi}\right)$$

Thus the bandwidth is 50 πHz

Nyquist rate is 100 πHz

Nyquist interval is $\frac{1}{100\pi} = 3.18$ ms

(c) $$\text{sinc}^2(100\,\pi t) \leftrightarrow \frac{1}{100\pi}\Pi\left(\frac{f}{100\pi}\right) * \frac{1}{100\pi}\Pi\left(\frac{f}{100\pi}\right)$$

Bandwidth of the signal is $= 50\pi + 50\pi = 100\pi$

Nyquist rate is 200π Hz

Nyquist rate is $\frac{1}{200\pi}s = 1.59$ ms

(d) $\text{sinc}(100\pi t) + 3\text{sinc}^2(60\pi t) = x(t)$

$$x(t) \xleftrightarrow{FT} \frac{1}{100\pi}\Pi\left(\frac{f}{100\pi}\right) + \frac{3}{60\pi}\Pi\left(\frac{f}{60\pi}\right) * \frac{1}{60\pi}\Pi\left(\frac{f}{60\pi}\right)$$

Bandwidth is $\max\{50\pi, 30\pi + 30\pi\} = 60\pi$

Nyquist rate is 120π Hz

Nyquist interval in $\frac{1}{120\pi} = 2.65$ ms

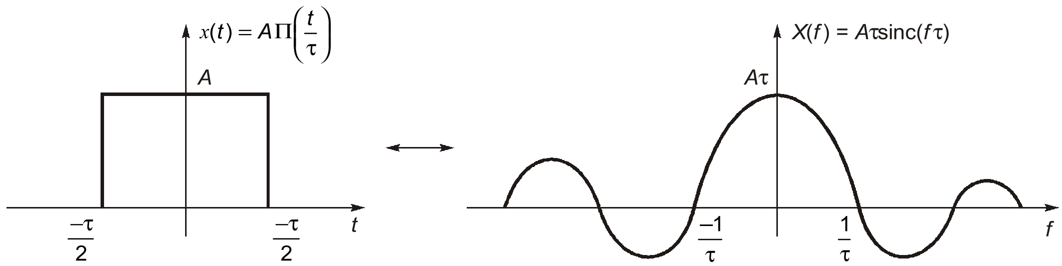
(e) $\text{sinc}(50\,\pi t)\,\text{sinc}(100\pi t) = x(t)$

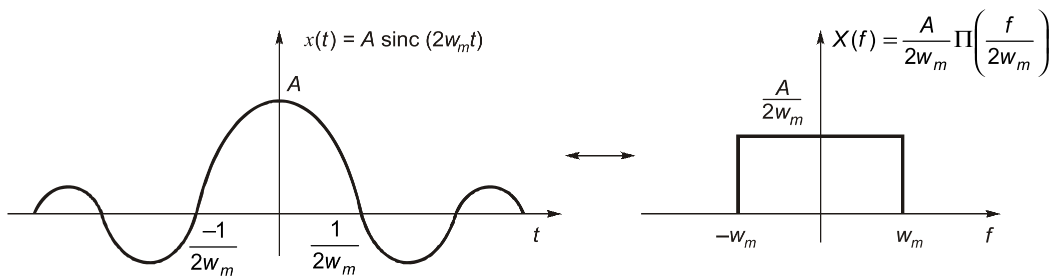
$$x(t) \xleftrightarrow{FT} \left[\frac{1}{50\pi}\Pi\left(\frac{f}{50\pi}\right)\right] * \left[\frac{1}{100\pi}\Pi\left(\frac{f}{100\pi}\right)\right]$$

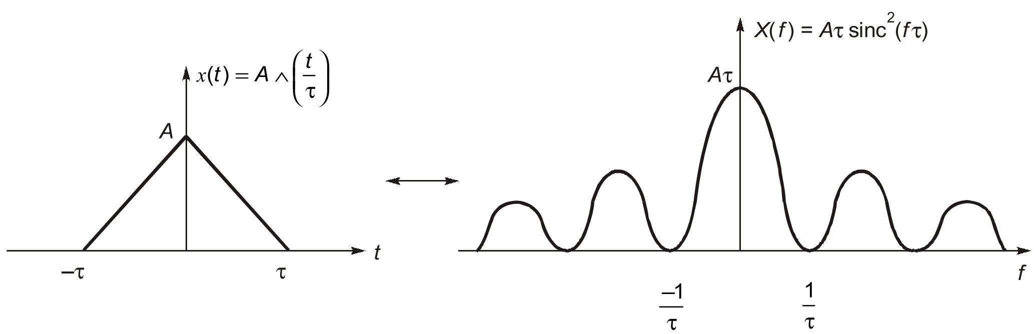
The bandwidth of the signal is $25\pi + 50\pi = 75\pi$ Hz

Nyquist rate is 150 πHz.

Nyquist interval is $\frac{1}{150\pi} = 2.12$ ms







NOTE

The bandpass sampling theorem states that if a bandpass signal $m(t)$ has a spectrum of bandwidth $\omega_B = 2\pi f_B$ and an upper frequency limit $\omega_U = 2\pi f_u$, then $m(t)$ can be recovered from $m_s(t)$ by bandpass filtering if $f_s = \frac{2f_u}{k}$ where k is the largest integer not exceeding $\frac{f_u}{f_B}$. All higher sampling rates are not necessarily usable unless they exceed $2f_u$.

Example - 1.4 Given the signal

$$m(t) = 10 \cos 2000\pi t \cos 8000\pi t$$

(a) What is the minimum sampling rate based on the low-pass uniform sampling theorem?
(b) Repeat (a) based on the bandpass sampling theorem.

Solution:

(a)

$$m(t) = 10 \cos 2000\pi t \cos 8000\pi t = 5 \cos 6000\pi t + 5 \cos 10000\pi t$$

$$f_m = 5000 \text{ Hz} = 5 \text{ kHz}$$

Thus,

$$f_s = 2f_m = 10 \text{ kHz}$$

(b)

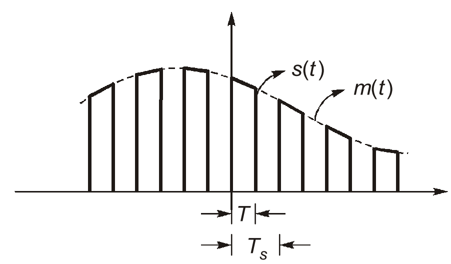
$$f_u = f_m = 5 \text{ kHz}$$
$$f_B = (5-3) = 2 \text{ kHz}$$
$$\frac{f_u}{f_B} = \frac{5}{2} = 2.5 \quad ; \quad k = 2$$

Based on bandpass sampling theorem

$$f_s = \frac{2f_u}{k} = 5 \text{ kHz}$$

1.3 Pulse Amplitude Modulation

After understanding the sampling theorem, pulse amplitude modulation can be defined. In PAM, the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal. PAM waveform is as shown in figure:



Here dashed curve depicts $m(t)$ and sequence of amplitude modulated rectangular pulses shown as solid lines represent corresponding PAM signal, $s(t)$.

There are two operations involved in generation of PAM signal.

Instantaneous Sampling

Message signal is sampled every T_s seconds. Where $T_s = \frac{1}{f_s}$, chosen in accordance with nyquist rate.

Lengthening

The duration of each sample is lengthened to obtain a constant value T. One important reason for intentionally lengthening the duration of each sample is to avoid the use of an excessive channel bandwidth since bandwidth is inversely proportional to pulse duration.

These two operations are jointly referred to as "sample and hold". We may express the PAM signal as discrete convolution sum:

$$s(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\, h(t - nT_s)$$

T_s is sampling period and $m(nT_s)$ is sample value of $m(t)$ obtained at time $t = nT_s$.

$$h(t - nT_s) = \int_{-\infty}^{\infty} h(t-\tau)\, \delta(\tau - nT_s) d\tau$$

Hence $\delta(t)$ becomes

$$s(t) = \int_{-\infty}^{\infty} \left[\sum_{n=-\infty}^{\infty} m(nT_s)\, \delta(\tau - nT_s) \right] h(t-\tau) d\tau$$

Let
$$m_\delta(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\, \delta(\tau - nT_s)$$

$\therefore$
$$s(t) = \int_{-\infty}^{\infty} m_\delta(t)\, h(t-\tau) d\tau$$

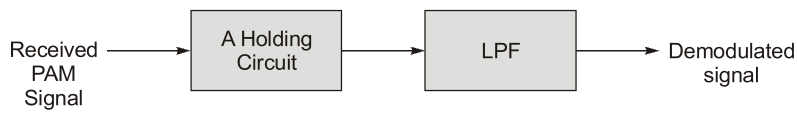
In frequency domain, above equation becomes

$$\Rightarrow \qquad S(f) = M_\delta(f)\,H(f)$$

$$M_\delta(f) = f_s \sum_{K=-\infty}^{\infty} M(f - Kf_s)$$

$$S(f) = f_s \sum_{K=-\infty}^{\infty} M(f - Kf_s)\,H(f)$$

1.3.1 Demodulation of PAM Signals



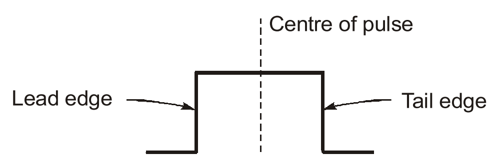
- Demodulation is process of recovering back the original message signal from modulated signal. In holding circuit, the capacitor is charged to the pulse amplitude value and it holds this value during the interval between two pulses.
- LPF is used to smoothen the output of the holding circuit.

1.3.2 Disadvantages of PAM Signal

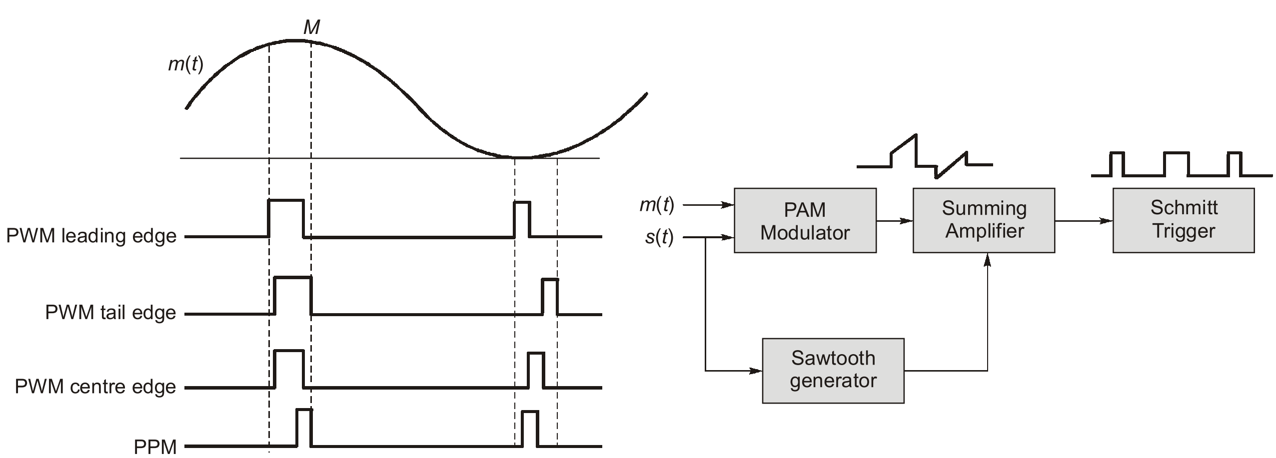
- Bandwidth requirement of PAM transmission is very large.
- Since, the message signal is stored in amplitude variations, hence interference of noise is maximum.
- As the amplitude varies, so peak power variations of transmitter also varies.

1.4 Pulse Width Modulation

PAM has the same disadvantage as amplitude modulation of continuous wave signals, namely modulation by noise, crosstalk etc. So properties of pulse other than amplitude are used for modulation. Instead of amplitude, the pulse width is made the variable. There are three types of PWM namely, lead edge, tail edge or centre of pulse.



The lead edge, tail edge or centre of pulse is kept fixed in phase as the duration of the pulse is varied with modulation. Sampling rate is $\frac{1}{2f_m}$, and guard time is same as PAM.



Block diagram of a PWM system

Let Δ be the width of the pulse in unmodulated pulse train in PWM,

$$\Delta \propto V_m$$

Width of pulse in PWM signal is given by

$$\Delta_m = \Delta(1 + V_m)$$

When there is no message i.e. V_m is equal to 0, then width of pulse will be equal to original width Δ. For positive values of message, the width will be proportionately increase by $(1 + V_m)$ factor. For negative values of message, the width decreases by $(1 - V_m)$.

NOTE

- In PWM, amplitude of pulse remains constant.
- PWM is more robust to noise.
- It is generally not used for conversion of analog to digital because here randomness is involved.

1.4.1 Speed Control of DC Motors Using PWM

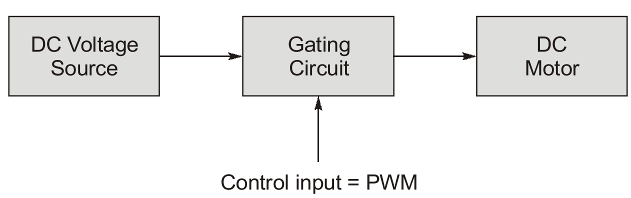
Speed of DC motor depends on average DC voltage applied across its terminals.

Suppose, if V volts is the voltage for running the DC motor at its full speed, then 0 volt implies rest condition. So, speed of motor can be varied from its rest to full speed value by varying DC voltage.

PWM performs this action. Constant DC voltage source is applied across terminals of DC motor through a gating circuit controlled by PWM signal. The gating circuit will essentially convert the constant DC source into a variable DC source.

No modulation $\Rightarrow$ constant width pulses and let this run the DC motor at some speed.

Now when width increases, voltage value increases from its unmodulated case and hence the speed. It happens in the opposite way for decrease in width. Thus, PWM provides a convenient and efficient approach for speed control of DC motors.



1.4.2 Advantage of PWM

- Good noise immunity.
- It is possible to reconstruct the message signal from noisy PWM.
- No synchronization is necessary.

1.4.3 Disadvantage of PWM

- Bandwidth requirement is large.
- Due to variable pulse width, pulses have variable power contents.

1.5 Pulse Position Modulation

Pulse position modulation is defined as the process of varying the position of the pulse with respect to the instantaneous variations of the message signal. If t_p indicates the timing instant of the leading or trailing edge of the pulse in each period of the train in PPM,

$$t_p \propto V_m$$

Increase in modulating voltage shift PPM pulses w.r.t. reference.

Mathematically, the position of leading or trailing edge of pulse (in each period) in PPM signal is given by

$$t_p = f(V_m)$$

when there is no message, then position of leading or trailing edge of pulse will be equal to original and hence $t_p = 0$.

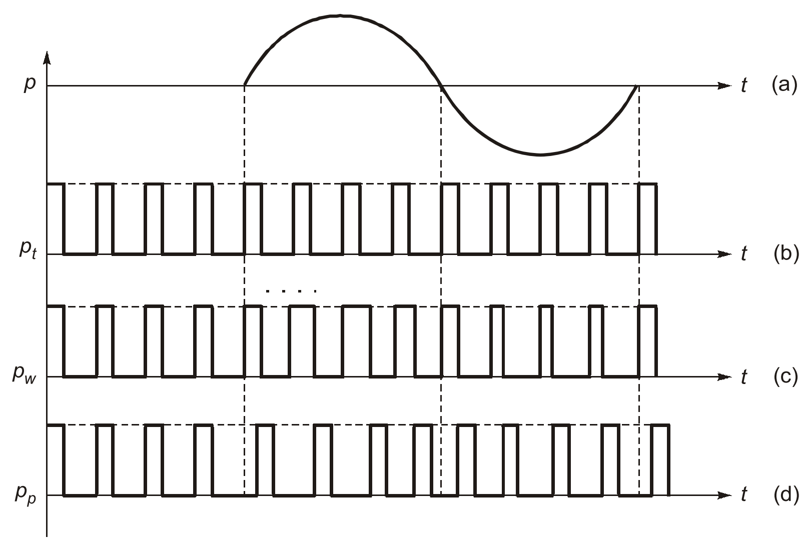
For positive values, $t_p = f(V_m)$

For negative values, $t_p = -f(V_m)$

1.5.1 Generation of PPM

First of all, PWM is generated by varying the width of trailing edge, then this edge will be extracted to get the position of the pulse in each period.

Once the position is extracted, the pulse is placed at this instant. Here, amplitude and width remains constant and same as that of original. Hence PPM is equally robust to noise.



Generation of PPM: (a) Message, (b) Pulse train, (c) PWM and (d) PPM

1.5.2 Advantages of PPM

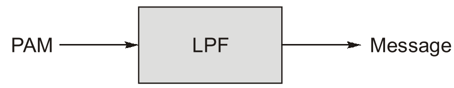
- Due to constant amplitude and storage of information in position of pulses, effect of noise is minimum.
- Hence, message signal can be reconstructed from a noisy PPM.
- Due to constant amplitude of pulses, transmitted power always remains constant.

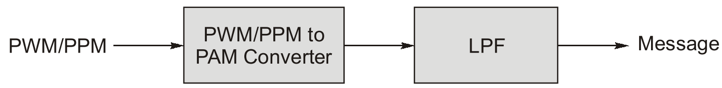
1.5.3 Disadvantage of PPM

- It is indirect way of storing the message information as in PWM and here also randomness is involved.
- Hence, it is of theoretical interest only and has limited use in signal processing and communication.

1.5.4 Demodulation of PPM/PWM Signals

- PAM signals can be demodulated by passing through a low pass filter which retains the low frequency message signal and smoothens out the pulse train information.
- PWM and PPM are demodulated by first converting them into PAM signal and then perform low pass filtering.





S. No.	PAM	PDM or PWM	PPM
1.	Amplitude of the pulse is proportional to amplitude of modulating signal.	Width of the pulse is proportional to amplitude of modulating signal.	The relative position of the pulse is proportional to the amplitude of modulating signal.
2.	The bandwidth of the transmission channel depends on width of the pulse.	Bandwidth of transmission channel depends on rise time of the pulse.	Bandwidth of transmission channel depends on rising time of the pulse.
3.	The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter remains constant.
4.	Noise interference is high and system is complex.	Noise interference is minimum.	Noise interference is minimum.
5.	Similar to amplitude modulation.	Simple to implement similar to frequency modulation.	Simple to implement similar to phase modulation.

Comparison between PAM, PWM and PPM

NOTE

- PWM is generated by using monostable multivibrator.
- PWM is converted to PAM using bistable multivibrator.
- PWM works satisfactorily if synchronization between transmitter and receiver fails, whereas PPM does not.
- Pulse modulation is used in microwave band.

After having discussion of pulse analog modulation, we will discuss about digital pulse modulation. It is of four types:

1. Pulse Code Modulation (PCM)
2. Differential Pulse Code Modulation (DPCM)
3. Delta modulation (DM)
4. Adaptive delta modulation (ADM)

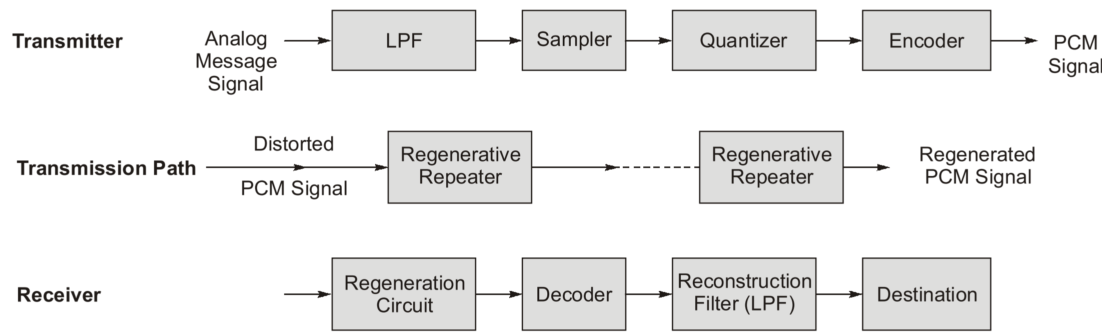
These are discussed in details in following sections:

1.6 PCM (Pulse Code Modulation)

In PCM, a message signal $m(t)$ is represented by a sequence of coded pulses which is accomplished by representing the signal in discrete form in both time and amplitude. This technique is breakthrough for moving to digital communication from analog communication.

In PAM, the time parameter is discrete but the amplitude still remains continuous. In this way, PAM is different from PCM.

1.6.1 Block Diagram of PCM System



1.6.2 Working of PCM System

An analog signal can be converted to a digital signal by means of sampling and quantization.

In PCM, message signal is first sampled and then amplitude of each sample is rounded off to the nearest one of a finite set of allowable values known as quantization levels, so that both time and amplitude are in discrete form. The conversion of analog signal to digital signal involves two steps: sampling and quantization.

- First, we get samples of this signal as par by the sampling theorem.
- For taking samples, time instants t_0, t_1, $t_2 \cdots$ are marked at equal time intervals along time axis.
- At each of these time instants, the magnitude of the signal is taken and these are called samples.
- Signal is now defined only at samples and hence, the signal is discrete. But since the magnitude still can take any value, hence the signal is still analog. This problem is solved by quantization. In quantization, the total amplitude range which the signal may occupy is divided into number of standard levels.

Fo example $\qquad x(t) = V_m \sin(\omega t)$

The amplitude of $x(t)$ varies from $-V_m$ to V_m.

If it is partitioned into L intervals, each having magnitude $\Delta = \dfrac{V_{max} - V_{min}}{L} = \dfrac{2V_m}{L}$, then it is said to be quantized. The amplitude of sample is rounded off to nearest quantization level.

Here Δ is called as step size.

The transmitter consists of mainly three operations–sampling, quantizing and encoding.

Sampling is done at Nyquist rate or higher which results in discrete time signal.

The quantizing and encoding operations are usually performed in the same circuit which is known as an analog to digital convertor (ADC).

In receiver, the essential operations are regeneration of impaired signals, decoding and demodulation of train of quantized samples. These are performed by digital to analog converter (DAC).

NOTE

- During transmission from the transmitter to the receiver, regenerative repeaters are used to reconstruct the transmitted sequence of coded pulses in order to remove the effects of signal distortion and noise.
- PCM, PAM, PPM and PWM, all are types of pulse modulation but **PAM, PPM, and PWM are pulse analog modulation and PCM is a pulse digital modulation.**
- PCM output is in the coded digital form and digital pulses are of form of constant amplitude, pulse and width. Hence the information is transmitted in the form of coded words.
- A PCM consists of PCM encoder at the transmitter side and PCM decoder at the receiver side.
- PCM is not modulation in conventional sense because unlike modulation, no parameter is varied in proportion to message signal.