Introduction

The purpose of a Communication System is to transfer an information bearing signal from a source to a user destination through a communication channel.

MODEL OF A COMMUNICATION SYSTEM

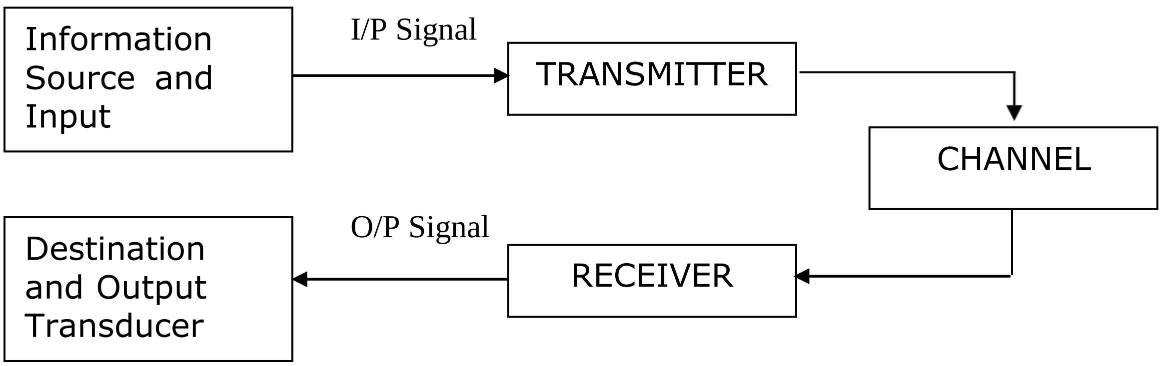


Fig. 1.1: Block diagram of Communication System.

The three basic elements of every communication systems are Transmitter, Receiver and Channel. The Overall purpose of this system is to transfer information from one point to another point, the user destination.

The message produced by a source, normally, is not electrical. Hence an input transducer is used for converting the message to a time – varying electrical quantity called message signal. Similarly, at the destination point, another transducer converts the electrical waveform to the appropriate message.

The transmitter is located at one point in space, the receiver is located at some other point separate from the transmitter, and the channel is the medium that provides the electrical connection between them. The purpose of the transmitter is to transform the message signal produced by the source of information into a form suitable for transmission over the channel.

The received signal is normally corrupted version of the transmitted signal, which is due to channel imperfections, noise and interference from other sources. The receiver has the task of operating on the received signal so as to reconstruct a recognizable form of the original message signal and to deliver it to the user destination.

ELEMENTS OF DIGITAL COMMUNICATION SYSTEMS:

The figure 1.2 shows the functional elements of a digital communication system.

Source of Information: 1. Analog Information Sources.
2. Digital Information Sources.

Analog Information Sources → Microphone actuated by a speech, TV Camera scanning a scene, continuous amplitude signals.

Digital Information Sources → These are teletype or the numerical output of computer which consists of a sequence of discrete symbols or letters.

An Analog information is transformed into a discrete information through the process of sampling and quantizing.

Digital Communication System

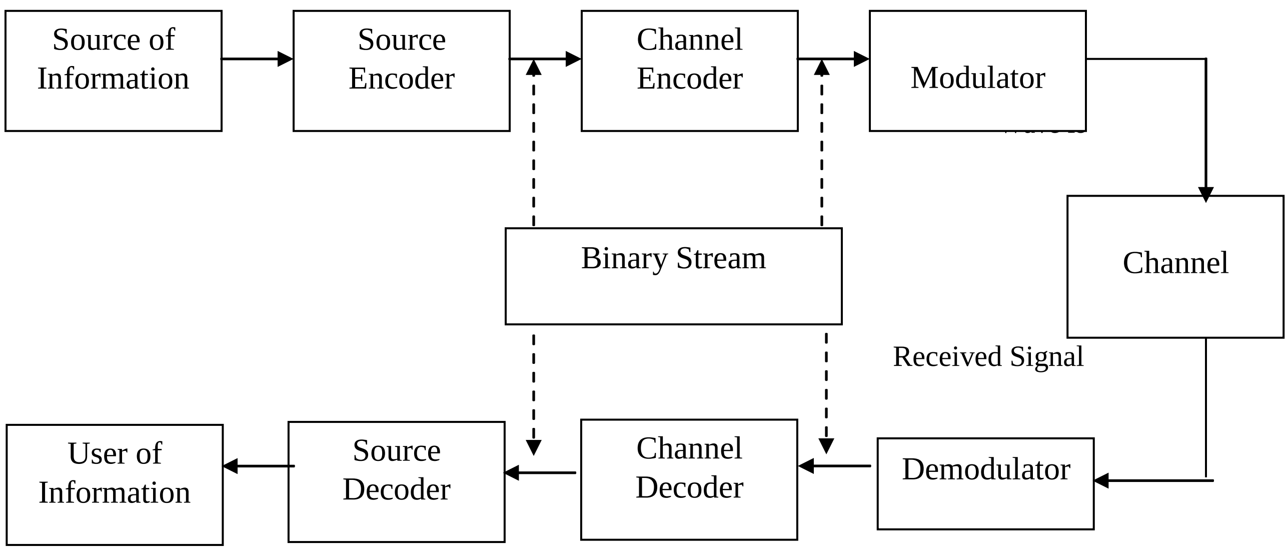


Fig 1.2: Block Diagram of a Digital Communication System

SOURCE ENCODER / DECODER:

The Source encoder (or Source coder) converts the input i.e. symbol sequence into a binary sequence of 0's and 1's by assigning code words to the symbols in the input sequence. For eg. :- If a source set is having hundred symbols, then the number of bits used to represent each symbol will be 7 because 2^7=128 unique combinations are available. The important parameters of a source encoder are **block size, code word lengths, average data rate and the efficiency** of the coder (i.e. actual output data rate compared to the minimum achievable rate)

At the receiver, the source decoder converts the binary output of the channel decoder into a symbol sequence. The decoder for a system using fixed – length code words is quite simple, but the decoder for a system using variable – length code words will be very complex.

Aim of the source coding is to remove the redundancy in the transmitting information, so that bandwidth required for transmission is minimized. Based on the probability of the symbol code word is assigned. Higher the probability, shorter is the codeword.

Ex: Huffman coding.

CHANNEL ENCODER / DECODER:

Error control is accomplished by the channel coding operation that consists of systematically adding extra bits to the output of the source coder. These extra bits do not convey any information but helps the receiver to detect and / or correct some of the errors in the information bearing bits.

There are two methods of channel coding:

1. Block Coding: The encoder takes a block of 'k' information bits from the source encoder and adds 'r' error control bits, where 'r' is dependent on 'k' and error control capabilities desired.
2. Convolution Coding: The information bearing message stream is encoded in a continuous fashion by continuously interleaving information bits and error control bits.

The Channel decoder recovers the information bearing bits from the coded binary stream. Error detection and possible correction is also performed by the channel decoder.
The important parameters of coder / decoder are: Method of coding, efficiency, error control capabilities and complexity of the circuit.

MODULATOR:

The Modulator converts the input bit stream into an electrical waveform suitable for transmission over the communication channel. Modulator can be effectively used to minimize the effects of channel noise, to match the frequency spectrum of transmitted signal with channel characteristics, to provide the capability to multiplex many signals.

DEMODULATOR:

The extraction of the message from the information bearing waveform produced by the modulation is accomplished by the demodulator. The output of the demodulator is bit stream. The important parameter is the method of demodulation.

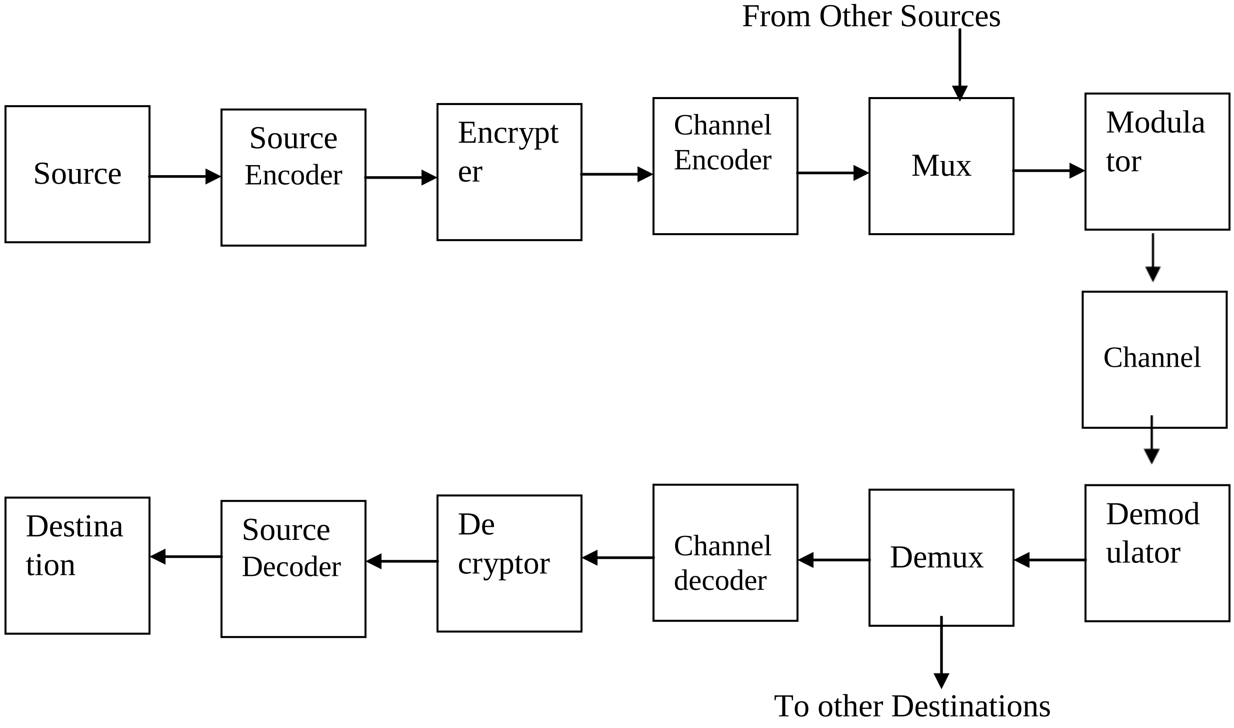
<u>CHANNEL:</u>

The Channel provides the electrical connection between the source and destination. The different channels are: Pair of wires, Coaxial cable, Optical fibre, Radio channel, Satellite channel or combination of any of these.

The communication channels have only finite Bandwidth, non-ideal frequency response, the signal often suffers amplitude and phase distortion as it travels over the channel. Also, the signal power decreases due to the attenuation of the channel. The signal is corrupted by unwanted, unpredictable electrical signals referred to as noise.

The important parameters of the channel are Signal to Noise power Ratio (SNR), usable bandwidth, amplitude and phase response and the statistical properties of noise.

<u>Modified Block Diagram:</u> (With additional blocks)



<u>Fig 1.3 : Block diagram with additional blocks</u>

Some additional blocks as shown in the block diagram are used in most of digital communication system:

- Encryptor: Encryptor prevents unauthorized users from understanding the messages and from injecting false messages into the system.

- MUX : Multiplexer is used for combining signals from different sources so that they share a portion of the communication system.

- DeMUX: DeMultiplexer is used for separating the different signals so that they reach their respective destinations.

- Decryptor: It does the reverse operation of that of the Encryptor.

Synchronization: Synchronization involves the estimation of both time and frequency coherent systems need to synchronize their frequency reference with carrier in both frequency and phase.

Advantages of Digital Communication

1. The effect of distortion, noise and interference is less in a digital communication system. This is because the disturbance must be large enough to change the pulse from one state to the other.

2. Regenerative repeaters can be used at fixed distance along the link, to identify and regenerate a pulse before it is degraded to an ambiguous state.

3. Digital circuits are more reliable and cheaper compared to analog circuits.

4. The Hardware implementation is more flexible than analog hardware because of the use of microprocessors, VLSI chips etc.

5. Signal processing functions like encryption, compression can be employed to maintain the secrecy of the information.

6. Error detecting and Error correcting codes improve the system performance by reducing the probability of error.

7. Combining digital signals using TDM is simpler than combining analog signals using FDM. The different types of signals such as data, telephone, TV can be treated as identical signals in transmission and switching in a digital communication system.

8. We can avoid signal jamming using spread spectrum technique.

Disadvantages of Digital Communication:

1. Large System Bandwidth:- Digital transmission requires a large system bandwidth to communicate the same information in a digital format as compared to analog format.

2. System Synchronization:- Digital detection requires system synchronization whereas the analog signals generally have no such requirement.

Channels for Digital Communications

The modulation and coding used in a digital communication system depend on the characteristics of the channel. The two main characteristics of the channel are BANDWIDTH and POWER. In addition the other characteristics are whether the channel is linear or nonlinear, and how free the channel is free from the external interference.

Five channels are considered in the digital communication, namely: telephone channels, coaxial cables, optical fibers, microwave radio, and satellite channels.

Telephone channel: It is designed to provide voice grade communication. Also good for data communication over long distances. The channel has a band-pass characteristic occupying the frequency range 300Hz to 3400hz, a high SNR of about 30db, and approximately linear response.

For the transmission of voice signals the channel provides flat amplitude response. But for the transmission of data and image transmissions, since the phase delay variations are important an equalizer is used to maintain the flat amplitude response and a linear phase response over the required frequency band. Transmission rates upto16.8 kilobits per second have been achieved over the telephone lines.

Coaxial Cable: The coaxial cable consists of a single wire conductor centered inside an outer conductor, which is insulated from each other by a dielectric. The main advantages of the coaxial cable are wide bandwidth and low external interference. But closely spaced repeaters are required. With repeaters spaced at 1km intervals the data rates of 274 megabits per second have been achieved.

Optical Fibers: An optical fiber consists of a very fine inner core made of silica glass, surrounded by a concentric layer called cladding that is also made of glass. The refractive index of the glass in the core is slightly higher than refractive index of the glass in the cladding. Hence if a ray of light is launched into an optical fiber at the right oblique acceptance angle, it is continually refracted into the core by the cladding. That means the difference between the refractive indices of the core and cladding helps guide the propagation of the ray of light inside the core of the fiber from one end to the other.

Compared to coaxial cables, optical fibers are smaller in size and they offer higher transmission bandwidths and longer repeater separations.

Microwave radio: A microwave radio, operating on the line-of-sight link, consists basically of a transmitter and a receiver that are equipped with antennas. The antennas are placed on towers at sufficient height to have the transmitter and receiver in line-of-sight of each other. The operating frequencies range from 1 to 30 GHz.

Under normal atmospheric conditions, a microwave radio channel is very reliable and provides path for high-speed digital transmission. But during meteorological variations, a severe degradation occurs in the system performance.

Satellite Channel: A Satellite channel consists of a satellite in geostationary orbit, an uplink from ground station, and a down link to another ground station. Both link operate at microwave frequencies, with uplink the uplink frequency higher than the down link frequency. In general, Satellite can be viewed as repeater in the sky. It permits communication over long distances at higher bandwidths and relatively low cost.

SAMPLING PROCESS

SAMPLING: A message signal may originate from a digital or analog source. If the message signal is analog in nature, then it has to be converted into digital form before it can transmitted by digital means. The process by which the continuous-time signal is converted into a discrete– time signal is called Sampling.
Sampling operation is performed in accordance with the sampling theorem.

SAMPLING THEOREM FOR LOW-PASS SIGNALS:-
Statement:- "If a band –limited signal g(t) contains no frequency components for ׀f׀ > W, then it is completely described by instantaneous values g(kTs) uniformly spaced in time with period Ts ≤ 1/2W. If the sampling rate, fs is equal to the Nyquist rate or greater ($f_s \geq 2W$), the signal g(t) can be exactly reconstructed.

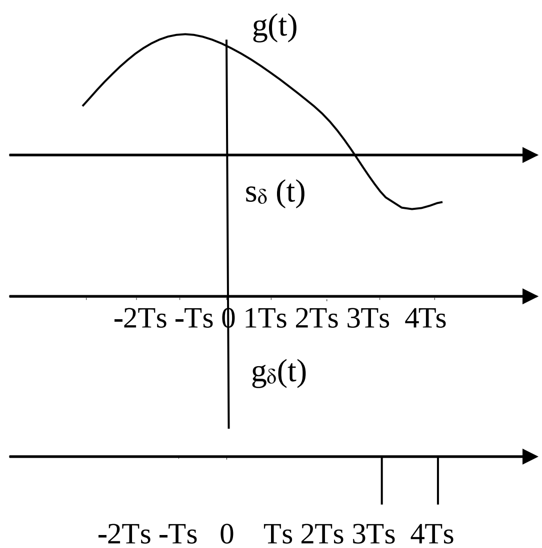


Fig 1.4: Sampling process

Proof:- Consider the signal g(t) is sampled by using a train of impulses $s_\delta(t)$.

Let $g_\delta(t)$ denote the ideally sampled signal, can be represented as

$$g_\delta(t) = g(t).s_\delta(t) \text{---------------------------- } 1$$

where $s_\delta(t)$ – impulse train defined by

$$s_\delta(t) = \sum_{k=-\infty}^{+\infty} \delta(t - kT_s) \text{------------------------ } 2$$

Therefore $g_\delta(t) = g(t) . \sum_{k=-\infty}^{+\infty} \delta(t - kT_s)$

$$= \sum_{k=-\infty}^{+\infty} g(kT_s).\delta(t - kT_s) \text{ ----------- } 3$$

The Fourier transform of an impulse train is given by

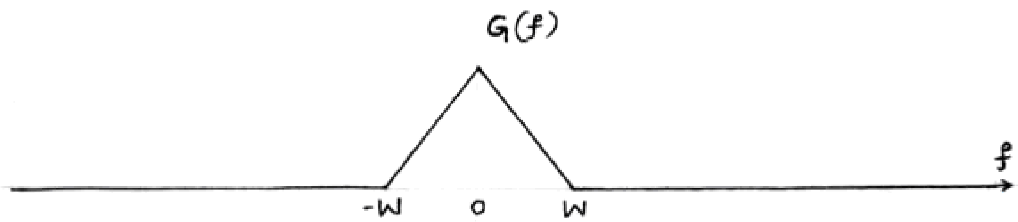
$$S_\delta(f) = F[s_\delta(t)] = f_s \sum_{n=-\infty}^{+\infty} \delta(f - nf_s) \quad \text{------------------- } 4$$

Applying F.T to equation 2.1 and using convolution in frequency domain property,

$$G_\delta(f) = G(f) * S_\delta(f)$$

Using equation 2.4, $G_\delta(f) = G(f) * f_s \sum_{n=-\infty}^{+\infty} \delta(f - nf_s)$

$$G_\delta(f) = f_s \sum_{n=-\infty}^{+\infty} G(f - nf_s) \quad \text{----------------- } 5$$



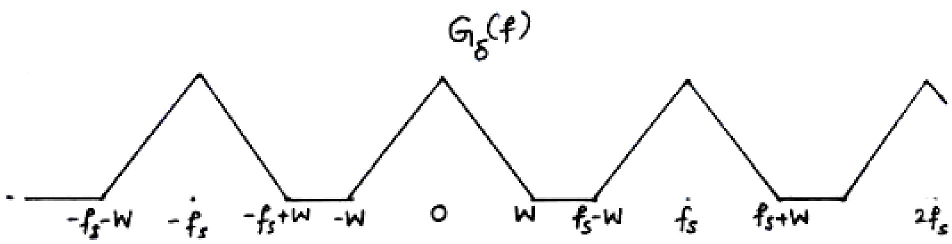


Fig. 1.5 Over Sampling ($f_s > 2W$)

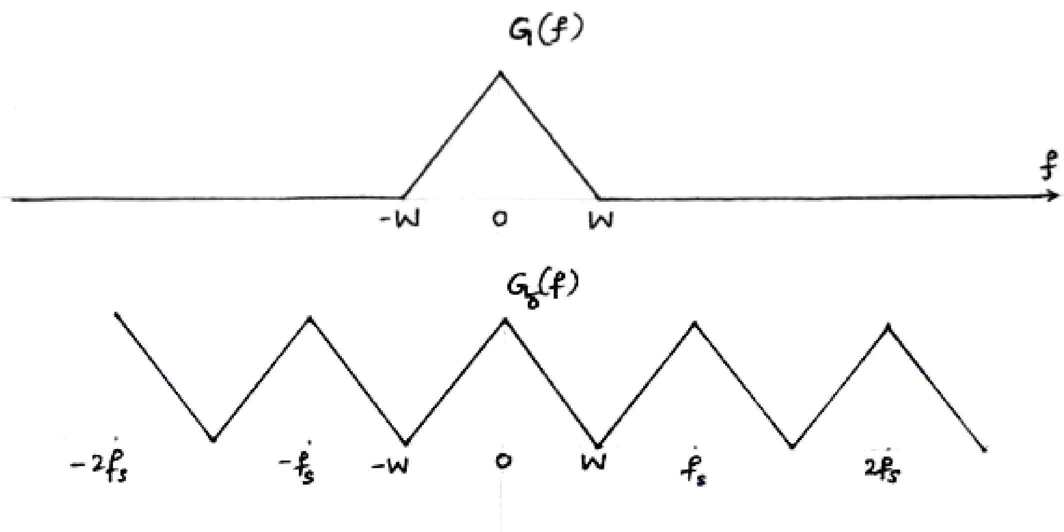


Fig. 1.6 Nyquist Rate Sampling ($f_s = 2W$)

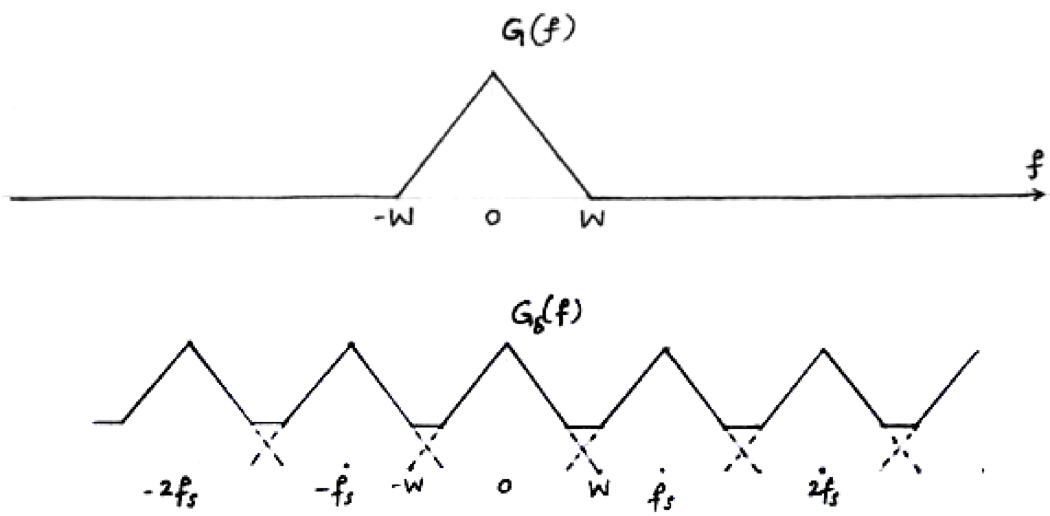
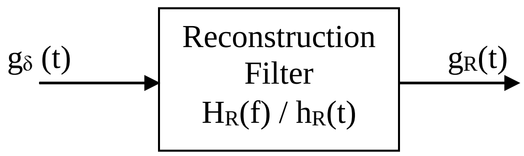


Fig. 1.7 Under Sampling ($f_s < 2W$)

Reconstruction of g(t) from $g_\delta(t)$:

By passing the ideally sampled signal $g_\delta(t)$ through an low pass filter (called Reconstruction filter) having the transfer function $H_R(f)$ with bandwidth, B satisfying the condition $W \le B \le (f_s - W)$, we can reconstruct the signal g(t). For an ideal reconstruction filter the bandwidth B is equal to W.



The output of LPF is, $g_R(t) = g_\delta(t) * h_R(t)$

where $h_R(t)$ is the impulse response of the filter.

In frequency domain, $G_R(f) = G_\delta(f) . H_R(f)$.

For the ideal LPF $H_R(f) = \begin{cases} K & -W \le f \le +W \\ 0 & \text{otherwise} \end{cases}$

then impulse response is $h_R(t) = 2WT_s. \text{Sinc}(2Wt)$

Correspondingly the reconstructed signal is

$$g_R(t) = [\, 2WT_s \,\text{Sinc}\,(2Wt)] * [g_\delta(t)]$$

$$g_R(t) = 2WT_s \sum_{K=-\infty}^{+\infty} g(kTs).Sinc(2Wt) * \delta(t - kTs)$$

$$g_R(t) = 2WT_s \sum_{K=-\infty}^{+\infty} g(kTs).Sinc[2W(t - kTs)]$$

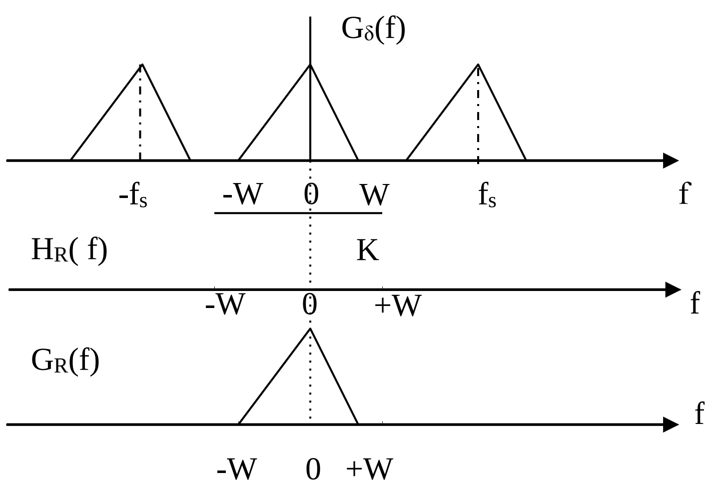


Fig: 1.8 Spectrum of sampled signal and reconstructed signal

Sampling of Band Pass Signals:

Consider a band-pass signal g(t) with the spectrum shown in figure.

Band width = B

Upper Limit = f_u

Lower Limit = f_l

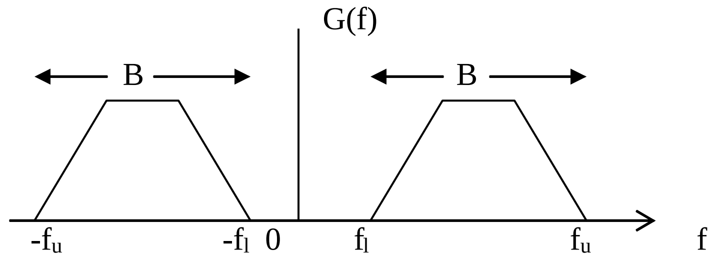


Fig 1.9: Spectrum of a Band-pass Signal

The signal g(t) can be represented by instantaneous values, g(kTs) if the sampling rate fs is ($2f_u/m$) where m is an integer defined as

$$((f_u / B) - 1) < m \leq (f_u / B)$$

If the sample values are represented by impulses, then g(t) can be exactly reproduced from it's samples by an ideal Band-Pass filter with the response, H(f) defined as

$$H(f) = \begin{cases} 1 & f_l < |f| < f_u \\ 0 & \text{elsewhere} \end{cases}$$

If the sampling rate, $fs \geq 2fu$, exact reconstruction is possible in which case the signal g(t) may be considered as a low pass signal itself.

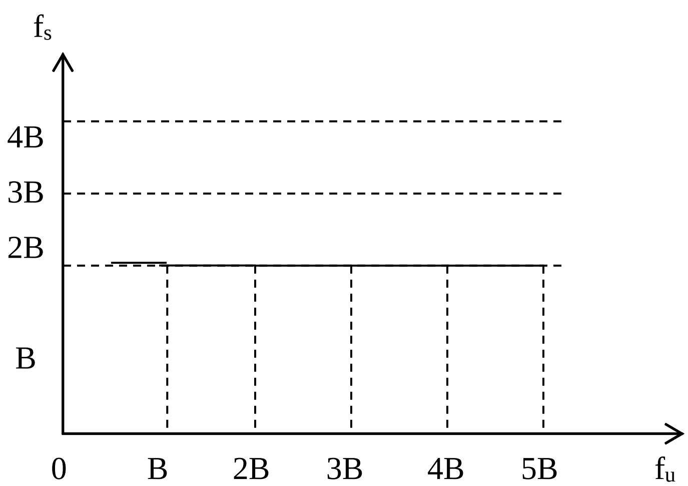


Fig 1.10: Relation between Sampling rate, Upper cutoff frequency and Bandwidth.

Example-1 :

Consider a signal g(t) having the Upper Cutoff frequency, f_u = 100KHz and the Lower Cutoff frequency f_l = 80KHz.

The ratio of upper cutoff frequency to bandwidth of the signal g(t) is

$$f_u / B = 100K / 20K = 5.$$

Therefore we can choose m = 5.

Then the sampling rate is $f_s = 2f_u / m = 200K / 5 = 40KHz$

Example-2 :

Consider a signal g(t) having the Upper Cutoff frequency, f_u = 120KHz and the Lower Cutoff frequency f_l = 70KHz.

The ratio of upper cutoff frequency to bandwidth of the signal g(t) is

$$f_u / B = 120K / 50K = 2.4$$

Therefore we can choose m = 2. ie.. m is an integer less than (f_u /B).

Then the sampling rate is $f_s = 2f_u / m = 240K / 2 = 120KHz$

Quadrature Sampling of Band – Pass Signals:

This scheme represents a natural extension of the sampling of low – pass signals.

In this scheme, the band pass signal is split into two components, one is in-phase component and other is quadrature component. These two components will be low–pass signals and are sampled separately. This form of sampling is called quadrature sampling.

Let g(t) be a band pass signal, of bandwidth '2W' centered around the frequency, fc, (fc>W). The in-phase component, $g_I(t)$ is obtained by multiplying g(t) with cos(2πfct) and then filtering out the high frequency components. Parallelly a quadrature phase component is obtained by multiplying g(t) with sin(2πfct) and then filtering out the high frequency components..

The band pass signal g(t) can be expressed as,

$$g(t) = g_I(t).\cos(2\pi f_c t) - g_Q(t)\sin(2\pi f_c t)$$

The in-phase, $g_I(t)$ and quadrature phase $g_Q(t)$ signals are low–pass signals, having band limited to ($-W < f < W$). Accordingly each component may be sampled at the rate of 2W samples per second.

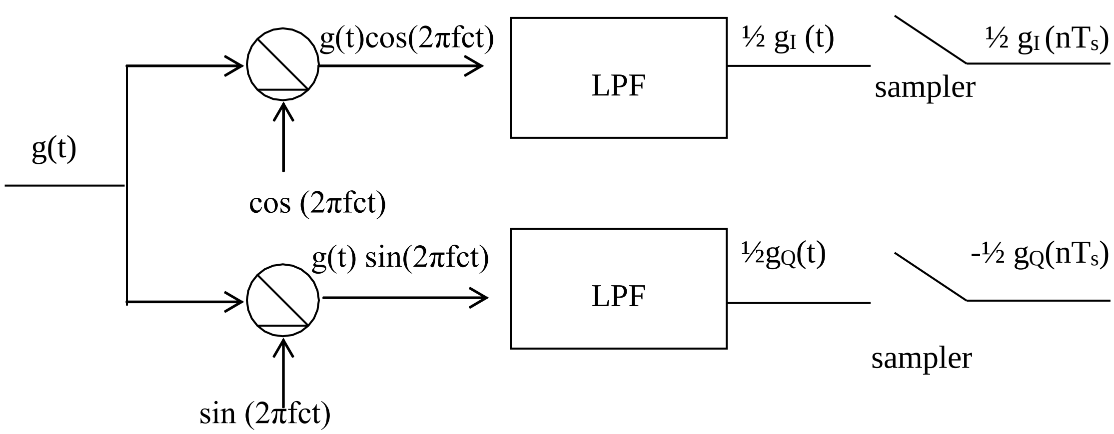
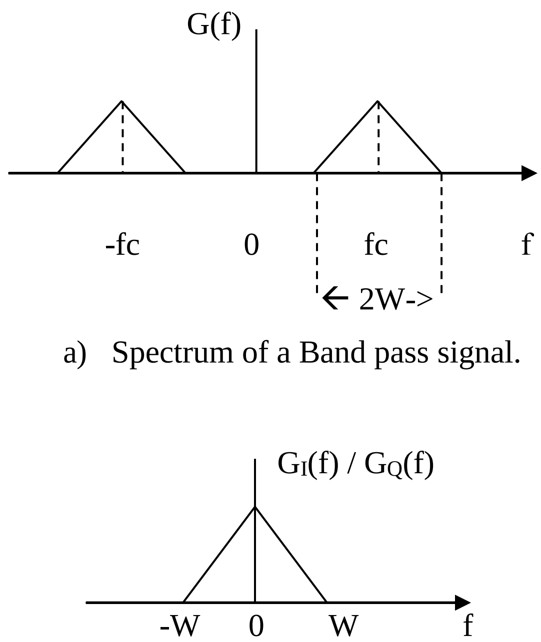


Fig 1.11: Generation of in-phase and quadrature phase samples



a) Spectrum of a Band pass signal.

b) Spectrum of $g_I(t)$ and $g_Q(t)$

Fig 1.12 a) Spectrum of Band-pass signal g(t)

b) Spectrum of in-phase and quadrature phase signals

RECONSTRUCTION:

From the sampled signals $g_I(nTs)$ and $g_Q(nTs)$, the signals $g_I(t)$ and $g_Q(t)$ are obtained. To reconstruct the original band pass signal, multiply the signals $g_I(t)$ by $\cos(2\pi fct)$ and $\sin(2\pi fct)$ respectively and then add the results.

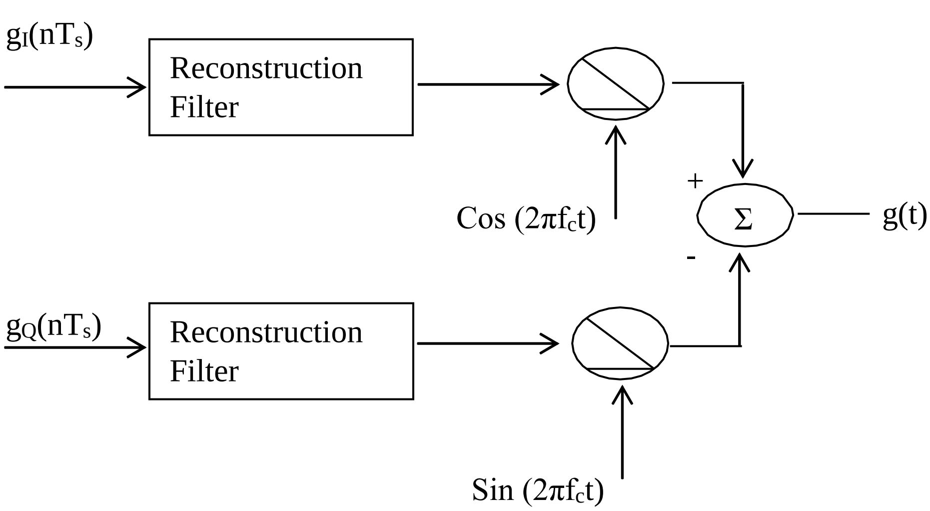
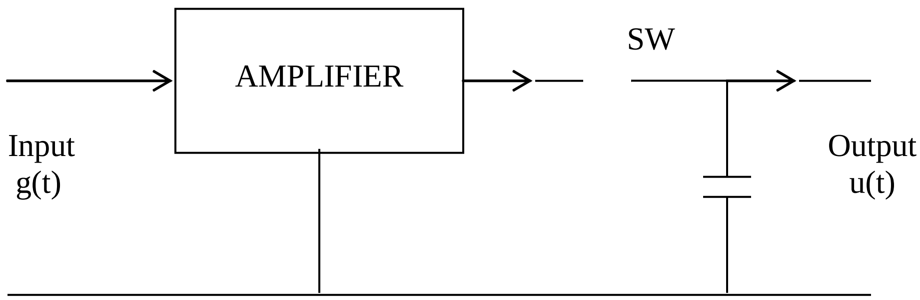


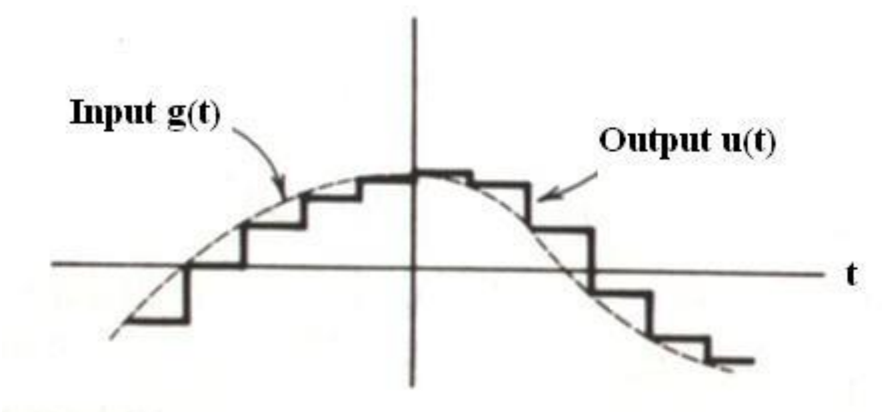
Fig 1.13: Reconstruction of Band-pass signal g(t)

Sample and Hold Circuit for Signal Recovery.

In both the natural sampling and flat-top sampling methods, the spectrum of the signals are scaled by the ratio τ/Ts, where τ is the pulse duration and Ts is the sampling period. Since this ratio is very small, the signal power at the output of the reconstruction filter is correspondingly small. To overcome this problem a sample-and-hold circuit is used .



a) Sample and Hold Circuit



b) Idealized output waveform of the circuit

Fig: 1.14 Sample Hold Circuit with Waveforms.

The Sample-and-Hold circuit consists of an amplifier of unity gain and low output impedance, a switch and a capacitor; it is assumed that the load impedance is large. The switch is timed to close only for the small duration of each sampling pulse, during which time the capacitor charges up to a voltage level equal to that of the input sample. When the switch is open , the capacitor retains the voltage level until the next closure of the switch. Thus the sample-and-hold

circuit produces an output waveform that represents a staircase interpolation of the original analog signal.
The output of a Sample-and-Hold circuit is defined as

$$u(t) = \sum_{n=-\infty}^{+\infty} g(nTs)\, h(t - nTs)$$

where h(t) is the impulse response representing the action of the Sample-and-Hold circuit; that is

$$h(t) = \begin{cases} 1 & \text{for } 0 < t < Ts \\ 0 & \text{for } t < 0 \text{ and } t > Ts \end{cases}$$

Correspondingly, the spectrum for the output of the Sample-and-Hold circuit is given by,

$$U(f) = f_s \sum_{n=-\infty}^{+\infty} H(f)G(f - nf_s))$$

where G(f) is the FT of g(t) and

$H(f) = Ts\ \mathrm{Sinc}(fTs)\exp(-j\pi fTs)$

To recover the original signal g(t) without distortion, the output of the Sample-and-Hold circuit is passed through a low-pass filter and an equalizer.

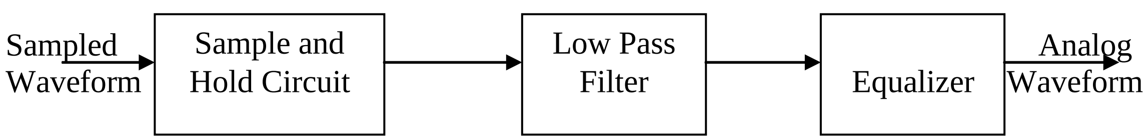


Fig. 1.15: Components of a scheme for signal reconstruction

Signal Distortion in Sampling.

In deriving the sampling theorem for a signal g(t) it is assumed that the signal g(t) is strictly band-limited with no frequency components above 'W' Hz. However, a signal cannot be finite in both time and frequency. Therefore the signal g(t) must have infinite duration for its spectrum to be strictly band-limited.

In practice, we have to work with a finite segment of the signal in which case the spectrum cannot be strictly band-limited. Consequently when a signal of finite duration is sampled an error in the reconstruction occurs as a result of the sampling process.

Consider a signal g(t) whose spectrum G(f) decreases with the increasing frequency without limit as shown in the figure 2.19. The spectrum, $G_\delta(f)$ of the ideally sampled signal , $g_\delta(t)$ is the sum of G(f) and infinite number of frequency shifted replicas of G(f). The replicas of G(f) are shifted in frequency by multiples of sampling frequency, fs. Two replicas of G(f) are shown in the figure 2.19.

The use of a low-pass reconstruction filter with it's pass band extending from (-fs/2 to +fs/2) no longer yields an undistorted version of the original signal g(t). The portions of the frequency shifted replicas are folded over inside the desired spectrum. Specifically, high frequencies in G(f) are reflected into low frequencies in $G_\delta(f)$. The phenomenon of overlapping in the spectrum is called as Aliasing or Foldover Effect. Due to this phenomenon the information is invariably lost.

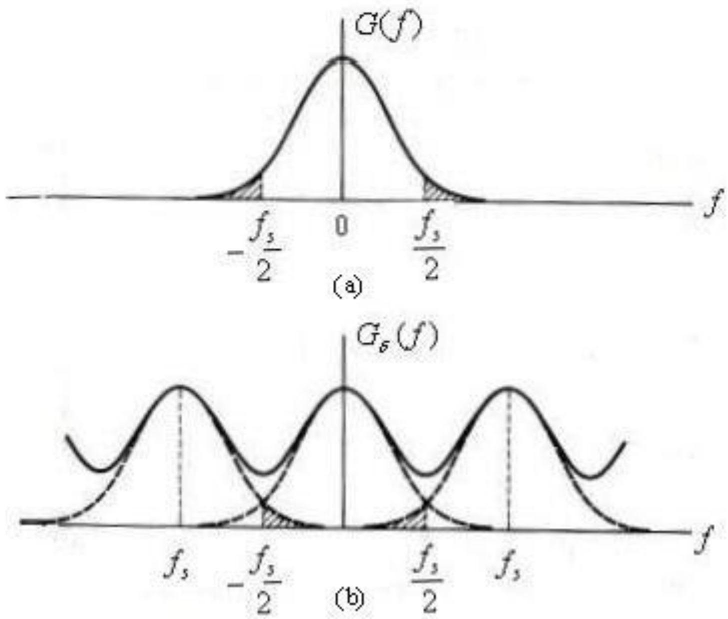


Fig. 1.16 : a) Spectrum of finite energy signal g(t)
b) Spectrum of the ideally sampled signal.

<u>Bound On Aliasing Error:</u>

Let g(t) be the message signal, g(n/fs) denote the sequence obtained by sampling the signal g(t) and $g_i(t)$ denote the signal reconstructed from this sequence by interpolation; that is

$$g_i(t) = \sum_n g\left(\frac{n}{f_s}\right) Sinc(f_s t - n)$$

Aliasing Error is given by, ε = | g(t) - gi(t) |

Signal g(t) is given by

$$g(t) = \int_{-\infty}^{\infty} G(f)\exp(j2\pi ft)df$$

Or equivalently

$$g(t) = \sum_{m=-\infty}^{+\infty} \int_{(m-1/2)fs}^{(m+1/2)fs} G(f)\exp(j2\pi ft)df$$

Using Poisson's formula and Fourier Series expansions we can obtain the aliasing error as

$$\varepsilon = \left| \sum_{m=-\infty}^{+\infty} [1 - \exp(-j2\pi m f_s t)] \int_{(m-1/2)fs}^{(m+1/2)fs} G(f)\exp(j2\pi ft)df \right|$$

Correspondingly the following observations can be done :

1. The term corresponding to m=0 vanishes.
2. The absolute value of the sum of a set of terms is less than or equal to the sum of the absolute values of the individual terms.
3. The absolute value of the term 1- exp(-j2πmfst) is less than or equal to 2.
4. The absolute value of the integral in the above equation is bounded as

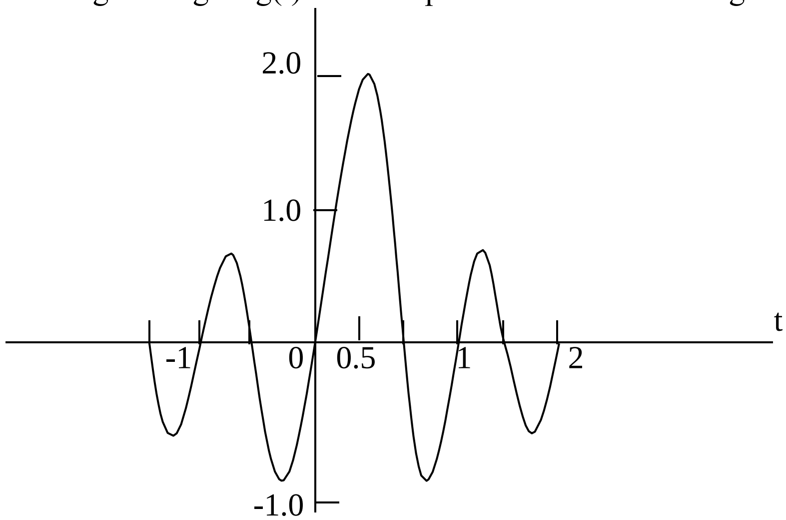
$$\left| \int_{(m-1/2)fs}^{(m+1/2)fs} G(f)\exp(j2\pi ft)df \right| < \int_{(m-1/2)fs}^{(m+1/2)fs} |G(f)|\,df$$

Hence the aliasing error is bounded as

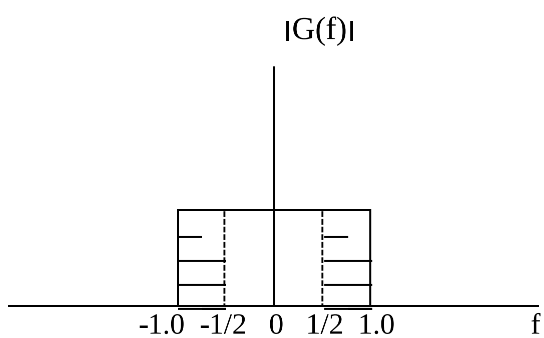
$$\varepsilon \le \ 2\int_{|f|>fs/2} |G(f)|\,df$$

Example: Consider a time shifted sinc pulse, $g(t) = 2\,\mathrm{sinc}(2t - 1)$. If $g(t)$ is sampled at rate of 1sample per second that is at t = 0, ± 1, ±2, ±3 and so on , evaluate the aliasing error.

Solution: The given signal g(t) and it's spectrum are shown in fig. 2.20.



a) Sinc Pulse



(b) Amplitude Spectrum, |G(f)|

Fig. 1.17

The sampled signal g(nTs) = 0 for n = 0, ± 1, ±2, ±3 . . . . .and reconstructed signal

$g_i(t) = 0$ for all t.

From the figure, the sinc pulse attains it's maximum value of 2 at time t equal to ½. The aliasing error cannot exceed max|g(t)| = 2.

From the spectrum, the aliasing error is equal to unity.

Natural Sampling:

In this method of sampling, an electronic switch is used to periodically shift between the two contacts at a rate of fs = (1/Ts) Hz, staying on the input contact for C seconds and on the grounded contact for the remainder of each sampling period.

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).

$$x_s(t) = x(t) \cdot s(t)$$

The sampling function s(t) is periodic with period Ts, can be defined as,

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

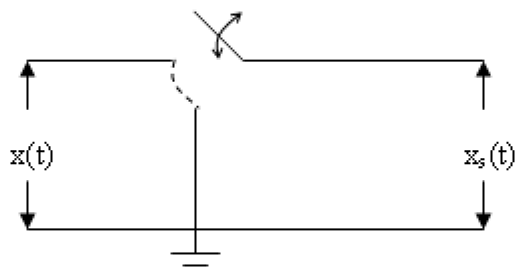


Fig: 1.18 Natural Sampling – Simple Circuit.

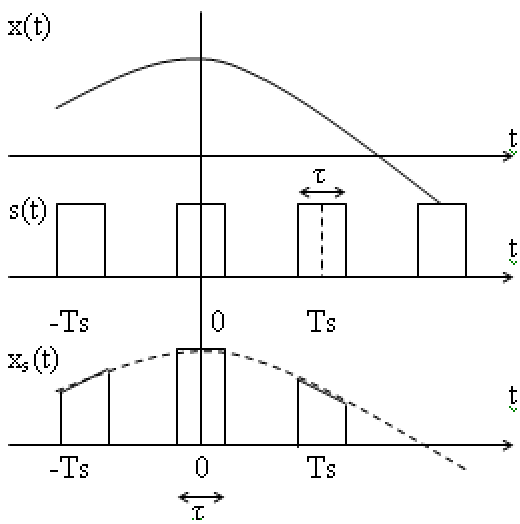


Fig: 1.19 Natural Sampling – Waveforms.

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

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

where the Fourier coefficients, $Co = \tau / Ts$ & $Cn = fs\,\tau\,\mathrm{Sinc}(n\,fs\,\tau)$

$$\text{Therefore: } x_s(t) = x(t)\left[Co + \sum_{n=1}^{\infty} 2Cn\cos(nw_s t) \right]$$

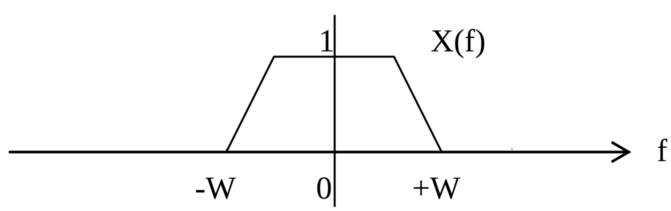
$$x_s(t) = Co.x(t) + 2C_1.x(t)\cos(w_s t) + 2C_2.x(t)\cos(2w_s t) + \ldots\ldots$$

Applying Fourier transform for the above equation

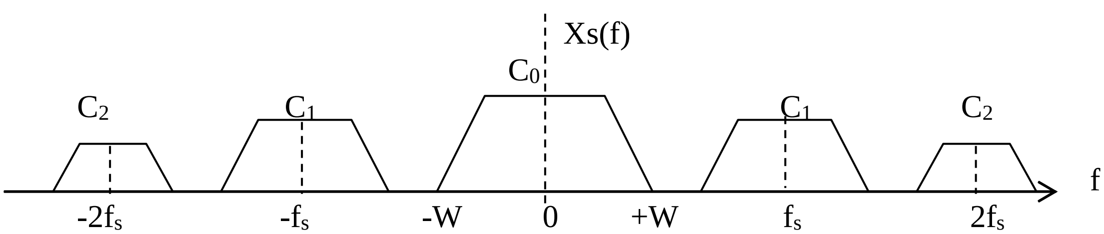
$$\begin{cases} \text{Using } x(t) \overset{FT}{\longleftrightarrow} X(f) \\ x(t)\cos(2\pi f_0 t) \longleftrightarrow \tfrac{1}{2}[X(f-f_0) + X(f+f_0)] \end{cases}$$

$$Xs(f) = Co.X(f) + C_1[X(f-f_0) + X(f+f_0)] + C_2[X(f-f_0) + X(f+f_0)] + \ldots\ldots$$

$$Xs(f) = Co.X(f) + \sum_{\substack{n=-\infty \\ n\neq 0}}^{\infty} Cn.X(f - nfs)$$



Message Signal Spectrum



Sampled Signal Spectrum ($f_s > 2W$)

Fig:1.20 Natural Sampling Spectrum

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 f_s. But the message term is scaled by 'Co". Since the spectrum is not distorted it is possible to reconstruct x(t) from the sampled waveform $x_s(t)$.

Flat Top Sampling:

In this method, the sampled waveform produced by practical sampling devices, the pulse p(t) is a flat – topped pulse of duration, τ.

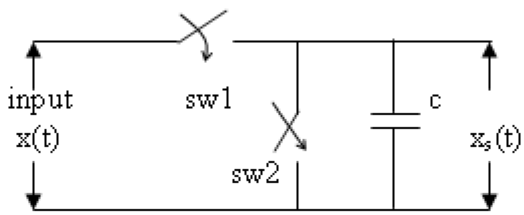


Fig. 1.21: Flat Top Sampling Circuit

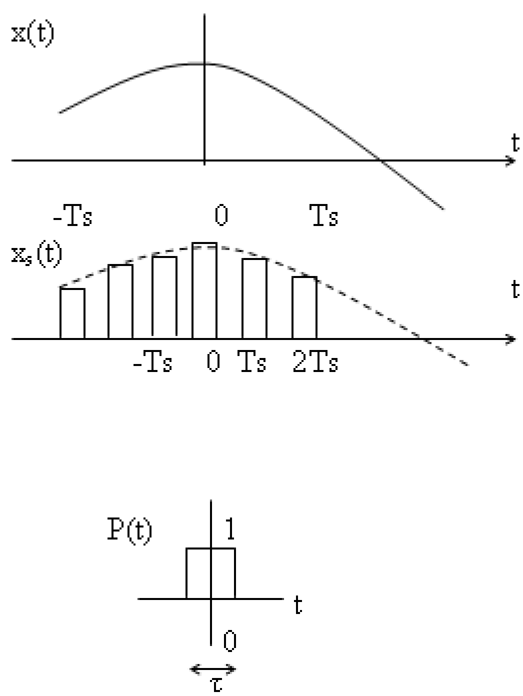


Fig.1.22: Waveforms

Mathematically we can consider the flat – top sampled signal as equivalent to the convolved sequence of the pulse signal p(t) and the ideally sampled signal, $x_\delta(t)$.

$$x_s(t) = p(t) * x_\delta(t)$$

$$x_s(t) = p(t) * \left[\sum_{k=-\infty}^{+\infty} x(kTs).\delta(t - kTs) \right]$$

Applying F.T,

$$X_s(f) = P(f).X_\delta(f)$$

$$= P(f).\ \ fs \sum_{n=-\infty}^{+\infty} X(f - nfs)$$

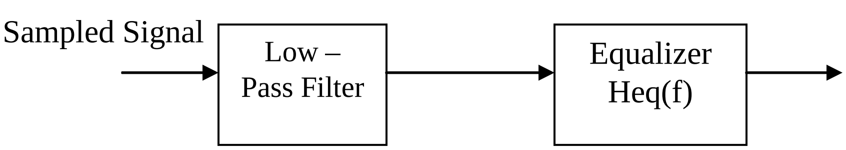
where $P(f) = FT[p(t)]$ and $X_\delta(f) = FT[x_\delta(t)]$

<u>Aperature Effect:</u>

The sampled signal in the flat top sampling has the attenuated high frequency components. This effect is called the <u>Aperture Effect.</u>

The aperture effect can be compensated by:

1. Selecting the pulse width τ as very small.
2. by using an <u>equalizer circuit.</u>



Equalizer decreases the effect of the in-band loss of the interpolation filter (lpf). As the frequency increases, the gain of the equalizer increases. Ideally the amplitude response of the equalizer is

$$|H_{eq}(f)| = 1 / |P(f)| = \frac{1}{\tau.SinC(f\tau)} = \frac{\Box\pi f}{Sin(\pi f \tau)}$$

RECOMMENDED QUESTIONS

1. Explain various communication channels for digital communication.
2. Bring out merits and demerits of digital communication over analog communication.
3. Explain quadrature sampling of band pass signals.
4. With a neat block diagram, explain the basic signal processing operation in digital communication.
5. Define the sampling theorem for band pass signals with necessary diagrams. Explain the generation and reconstruction of band pass signals.
6. Explain the term quadrature sampling of band pass signal with the help of spectrum and block diagrams.
7. State and prove sampling theorem for low pass signals.
8. With necessary equations, explain flat top sampling.
9. What is aperture effect? How to eliminate it?
10. With necessary expression explain, natural sampling.
11. Explain sample and hold circuit.

Unit - 2

PAM, TDM. Waveform Coding Techniques, PCM, Quantization noise and SNR, robust quantization.

7 Hours

Text Book:

1. **Digital communications**, Simon Haykin, John Wiley, 2003.

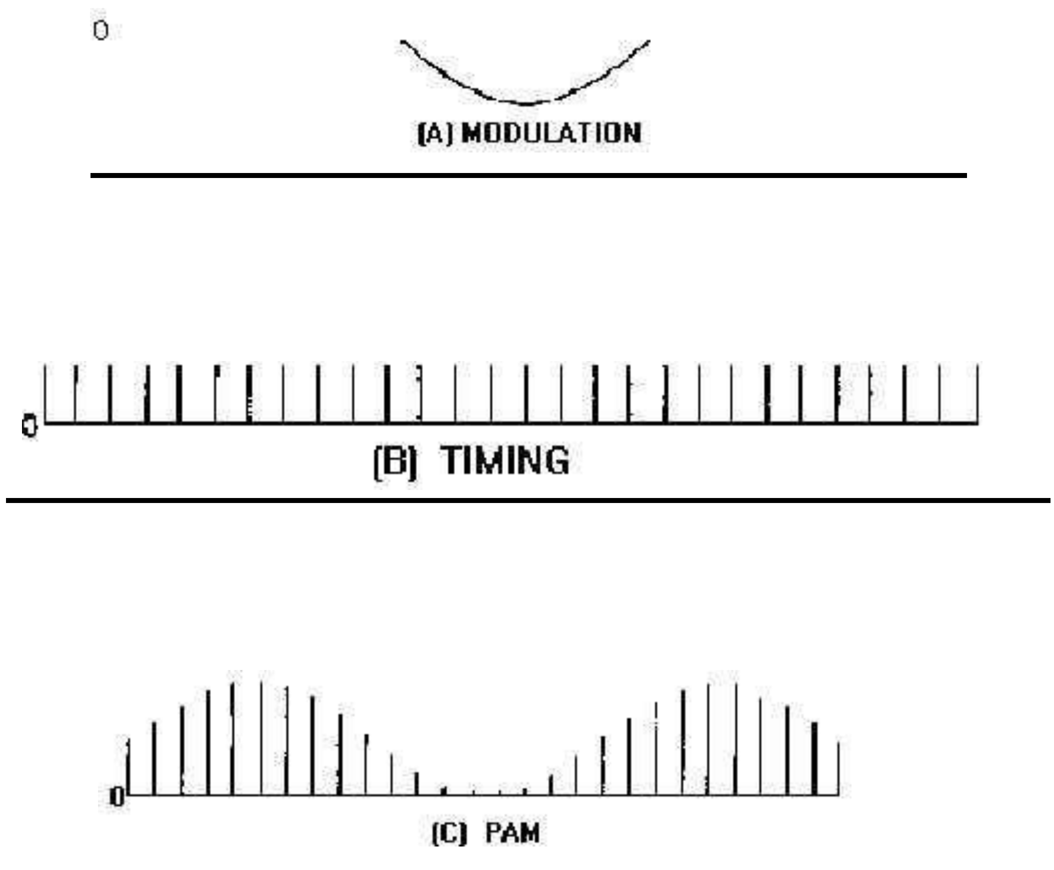
Reference Books:

1. **Digital and analog communication systems & An introduction to Analog and Digital Communication**, K. Sam Shanmugam, John Wiley, 1996. 2.Simon Haykin, John Wiley, 2003
2. **Digital communications** - Bernard Sklar: Pearson education 2007

Unit --2

Pulse amplitude modulation

Pulse-Amplitude Modulation **"** The amplitude of a carrier consisting of a periodic train of rectangular pulses is varied in proportion to sampled values of a message signal".
Fig (A) represents a sinewave of intelligence to be modulated on a transmitted carrier wave. Fig (B) shows the timing pulseswhich determine the sampling interval. Fig (C) shows PULSE-AMPLITUDE MODULATION (PAM).

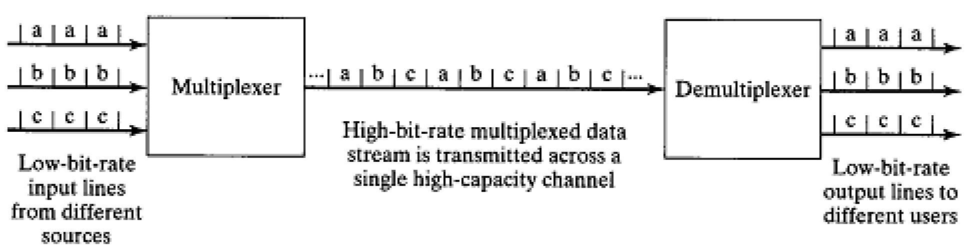


Pulse-amplitude modulation is the simplest form of pulse modulation. It is generated in much the same manner as analog-amplitude modulation. The timing pulses are applied to a pulse amplifier in which the gain is controlled by the modulating waveform. Since these variations in amplitude actually represent the signal, this type of modulation is basically a form of AM. The only difference is that the signal is now in the form of pulses. This means that Pam has the same built- in weaknesses as any other AM signal -high susceptibility to noise and interference. The reason for susceptibility to noise is that any interference in the transmission path will either add to or subtract from any voltage already in the circuit (signal voltage). Thus, the amplitude of the signal will be changed. Since the amplitude of the voltage represents the signal, any unwanted change to the signal is considered a SIGNAL DISTORTION. For this reason, pam is not often used. When pam is used, the pulse train is used to frequency modulate a carrier for transmission. Techniques of pulse modulation other than pam have been developed to overcome problems of noise interference.

Time division multiplexing :

It's often practical to combine a set of low-bit-rate streams, each with a fixed and pre-defined bit rate, into a single high-speed bit stream that can be transmitted over a single channel. This technique is called time division multiplexing (TDM) and has many applications, including wireline telephone systems and some cellular telephone systems. The main reason to use TDM is to take advantage of existing transmission lines. It would be very expensive if each low-bit-rate stream were assigned a costly physical channel (say, an entire fiber optic line) that extended over a long distance.

Consider, for instance, a channel capable of transmitting 192 kbit/sec from Chicago to New York. Suppose that three sources, all located in Chicago, each have 64 kbit/sec of data that they want to transmit to individual users in New York. As shown in Figure 7-2, the high-bit-rate channel can be divided into a series of time slots, and the time slots can be alternately used by the three sources. The three sources are thus capable of transmitting all of their data across the single, shared channel. Clearly, at the other end of the channel (in this case, in New York), the process must be reversed (i.e., the system must divide the 192 kbit/sec multiplexed data stream back into the original three 64 kbit/sec data streams, which are then provided to three different users). This reverse process is called demultiplexing.



Time division multiplexing.

Choosing the proper size for the time slots involves a trade-off between efficiency and delay. If the time slots are too small (say, one bit long) then the multiplexer must be fast enough and powerful enough to be constantly switching between sources (and the demultiplexer must be fast enough and powerful enough to be constantly switching between users). If the time slots are larger than one bit, data from each source must be stored (buffered) while other sources are using the channel. This storage will produce delay. If the time slots are too large, then a significant delay will be introduced between each source and its user. Some applications, such as teleconferencing and videoconferencing, cannot tolerate long delays.

Waveform Coding Techniques

PCM [Pulse Code Modulation]

PCM is an important method of analog –to-digital conversion. In this modulation the analog signal is converted into an electrical waveform of two or more levels. A simple two level waveform is shown in fig .

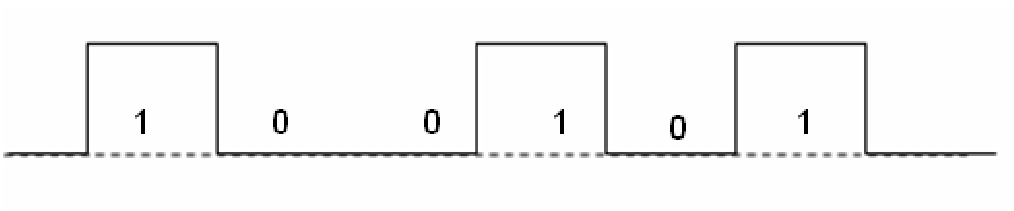


Fig:2.1 A simple binary PCM waveform

The PCM system block diagram is shown in fig 3.2. The essential operations in the transmitter of a PCM system are Sampling, Quantizing and Coding. The Quantizing and encoding operations are usually performed by the same circuit, normally referred to as analog to digital converter.

The essential operations in the receiver are regeneration, decoding and demodulation of the quantized samples. Regenerative repeaters are used to reconstruct the transmitted sequence of coded pulses in order to combat the accumulated effects of signal distortion and noise.

PCM Transmitter:

Basic Blocks:

1. Anti aliasing Filter
2. Sampler
3. Quantizer
4. Encoder

An anti-aliasing filter is basically a filter used to ensure that the input signal to sampler is free from the unwanted frequency components.
For most of the applications these are low-pass filters. It removes the frequency components of the signal which are above the cutoff frequency of the filter. The cutoff frequency of the filter is chosen such it is very close to the highest frequency component of the signal.

Sampler unit samples the input signal and these samples are then fed to the Quantizer which outputs the quantized values for each of the samples. The quantizer output is fed to an encoder