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SCHOOL OF SCIENCE AND HUMANITIES

DEPARTMENT OF PHYSICS

SPHA5304 – COMMUNICATION ELECTRONICS

UNIT I SIGNAL ANALYSIS

UNIT 1 SIGNAL ANALYSIS

Fourier transform of gate functions, delta functions at the origin – Two delta function and periodic delta function –properties of Fourier transform – Frequency shifting – Time shifting – Convolution theorem – Frequency convolution theorem – Sampling theorem.

Signal Analysis

Signal Analysis provides the abstract mathematics and functional analysis which is missing from the backgrounds of many readers, especially undergraduate science and engineering students and professional engineers. The students can begin comfortably with the basic ideas. This chapter gradually dispenses the mathematics of Hilbert spaces, complex analysis, distributions, modern integration theory, random signals, and analog Fourier transforms; the less mathematically adept reader is not over- whelmed with hard analysis. There has been no easy route from standard signal processing texts to the latest treatises on wavelets, Gabor transforms, and the like. The gap must be spanned with knowledge of advanced mathematics. And this has been a problem for too many engineering students, classically-educated applied researchers, and practicing engineers. We hope that *Signal Analysis* removes the obstacles. It has the signal processing fundamentals, the signal analysis perspective, the mathematics, and the bridge from all of these to crucial developments that began in the mid-1980s.

Signals: Analog, Discrete, and Digital

Analog, discrete, and digital signals are the raw material of signal processing and analysis. Natural processes, whether dependent upon or independent of human control, generate analog signals; they occur in a continuous fashion over an interval of time or space. The mathematical model of an analog signal is a function defined over a part of the real number line. Analog signal conditioning uses conventional electronic circuitry to acquire, amplify, alter, and transmit these signals. At some point, digital processing may take place; today, this is almost always necessary. Per- haps the application requires superior noise immunity. Intricate processing steps are also easier to implement on digital computers. Furthermore, it is easier to improve and correct computerized algorithms than systems comprised of hard-wired analog components. Whatever the rationale for digital processing, the analog signal is captured, stored momentarily, and then converted to digital form. In contrast to an ana- log signal, a discrete signal has values only at isolated points. Its mathematical representation is a function on the integers; this is a fundamental difference. When the signal values are of finite precision, so that they can be stored in the registers of a computer, then the discrete signal is more precisely known as a digital signal. Digital signals thus come from sampling an analog signal, and—although there is such a thing as an analog computer—nowadays digital machines perform almost all analytical computations on discrete signal data.

What is Signal?

Signal is a time varying physical phenomenon which is intended to convey information (or) Signal is a function of time. (or) Signal is a function of one or more independent variables, which contain some information. Example**:** voice signal, video signal, signals on telephone wires etc. Noise is also a signal, but the information conveyed by noise is unwanted hence it is considered as undesirable.

What is System?

System is a device or combination of devices, which can operate on signals and produces corresponding response. Input to a system is called as excitation and output from it is called as response. For one or more inputs, the system can have one or more outputs. Example: Communication System

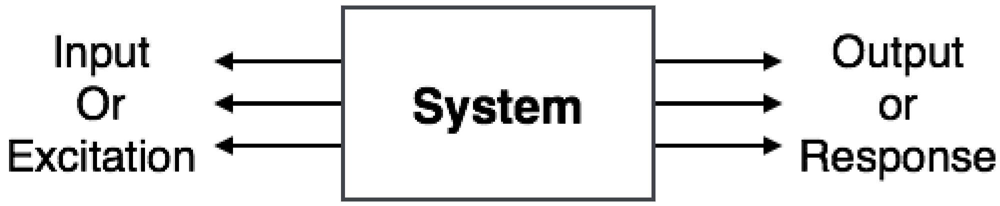


Fig: 1 Simple block diagram of a Communication System

TYPES OF BASIC SIGNALS

1. **Unit Step Function**

Unit step function is denoted by u(t). It is defined as $u(t) = \begin{cases} 1 & t \geqslant 0 \\ 0 & t<0 \end{cases}$

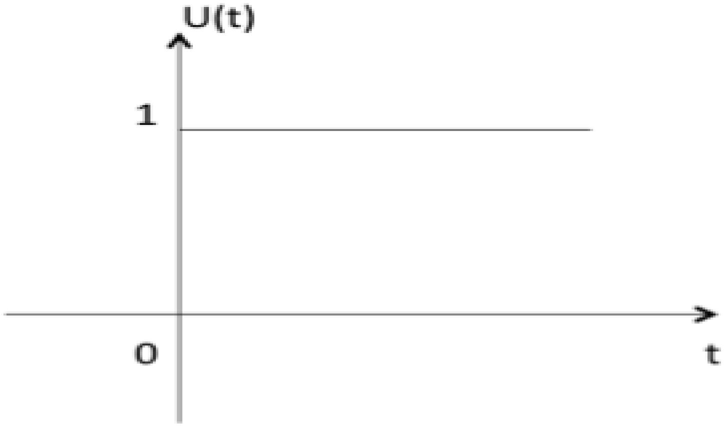


Fig:2 Representation of Unit Step Function

- It is used as best test signal.
- Area under unit step function is unity.

2. Unit Impulse Function

Impulse function is denoted by δ(t). and it is defined as $\delta(t) = \begin{cases} 1 & t=0 \\ 0 & t\neq 0 \end{cases}$

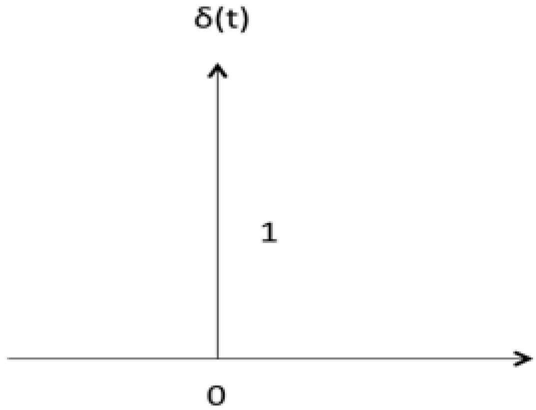


Fig:3 Representation of Unit Impulse Function

$$\int_{-\infty}^{\infty} \delta(t)dt = u(t)$$

$$\delta(t) = \frac{du(t)}{dt}$$

3. Ramp Signal

Ramp signal is denoted by r(t), and it is defined as $r(t) = \begin{cases} t & t\geqslant 0 \\ 0 & t<0 \end{cases}$

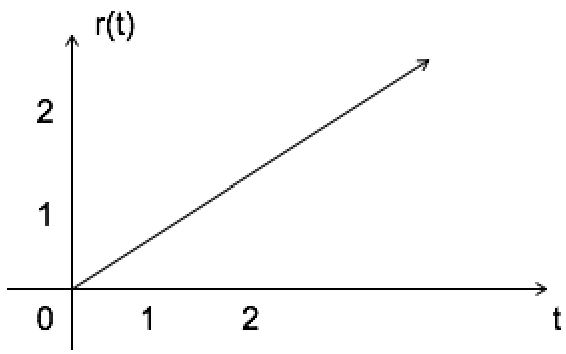


Fig:4 Representation of Ramp signal

$$\int u(t) = \int 1 = t = r(t)$$
$$u(t) = dr(t)/dt$$

Area under unit ramp is unity.

4. Parabolic Signal

Parabolic signal can be defined as x(t) = $\begin{cases} t^2/2 & t \geqslant 0 \\ 0 & t < 0 \end{cases}$

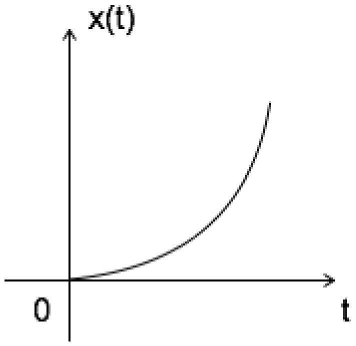


Fig:5 Representation of Parabolic Signal

$$\iint u(t)dt = \int r(t)dt = \int tdt = \frac{t^2}{2} = parabolicsignal$$

$$\Rightarrow u(t) = \frac{d^2x(t)}{dt^2}$$

$$\Rightarrow r(t) = \frac{dx(t)}{dt}$$

5. Signum Function

Signum function is denoted as sgn(t). It is defined as sgn(t) = $\begin{cases} 1 & t > 0 \\ 0 & t = 0 \\ -1 & t < 0 \end{cases}$

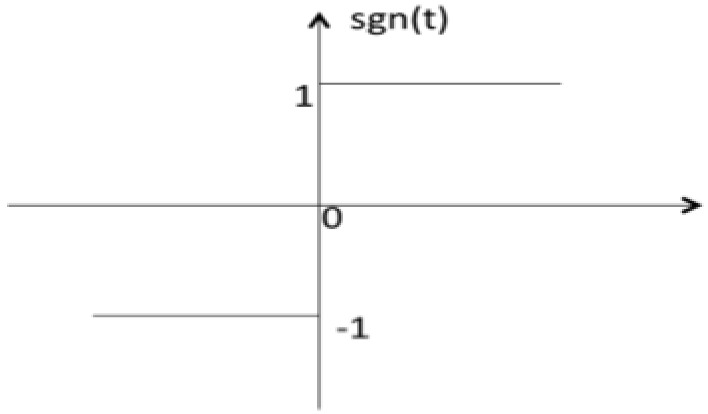


Fig:6 Representation of Signum Function

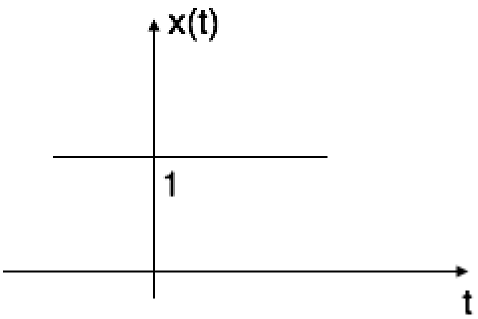
sgn(t) = 2u(t) – 1

6. Exponential Signal

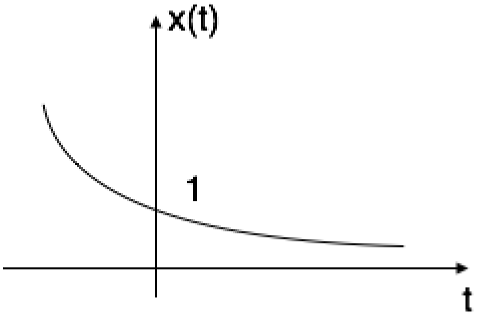
Exponential signal is in the form of x(t) = $e^{\alpha t}$.

The shape of exponential can be defined by α

Case i: if α = 0 $\rightarrow$ x(t) = e^{0} = 1



Case ii: if α < 0 i.e. -ve then x(t) = $e^{-\alpha t}$. The shape is called decaying exponential.



Case iii: if α > 0 i.e. +ve then x(t) = $e^{\alpha t}$. The shape is called raising exponential.

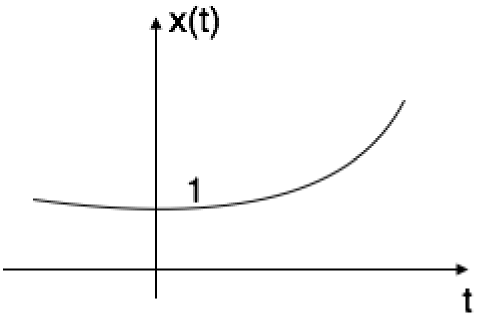


Fig:7 Representation of Exponential Signal when α = 0 ; α < 0 ; α > 0.

7. Rectangular Signal

Let it be denoted as x(t) and it is defined as

$$x(t) = A\, rect \left[\frac{r}{T}\right] \qquad \text{ex:} \; 4\, rect \left[\frac{r}{6}\right]$$

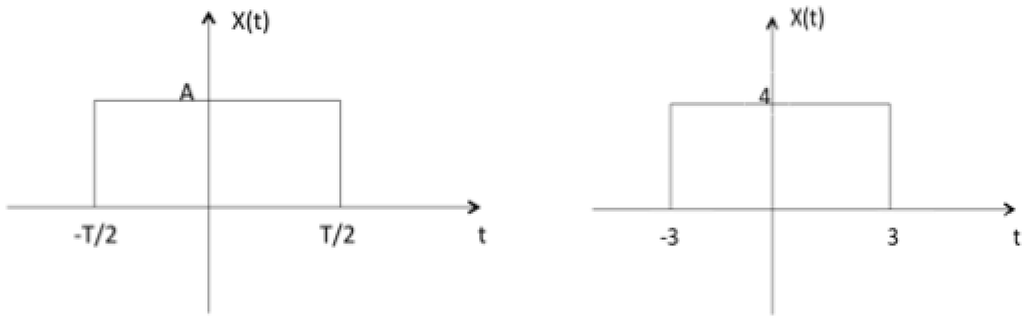


Fig:8 Representation of Rectangular Signal

8. Triangular Signal

Let it be denoted as x(t)

$x(t) = A\left[1 - \frac{|t|}{T}\right]$ $\qquad$ $\text{ex}: x(t) = A\left[1 - \frac{|t|}{5}\right]$

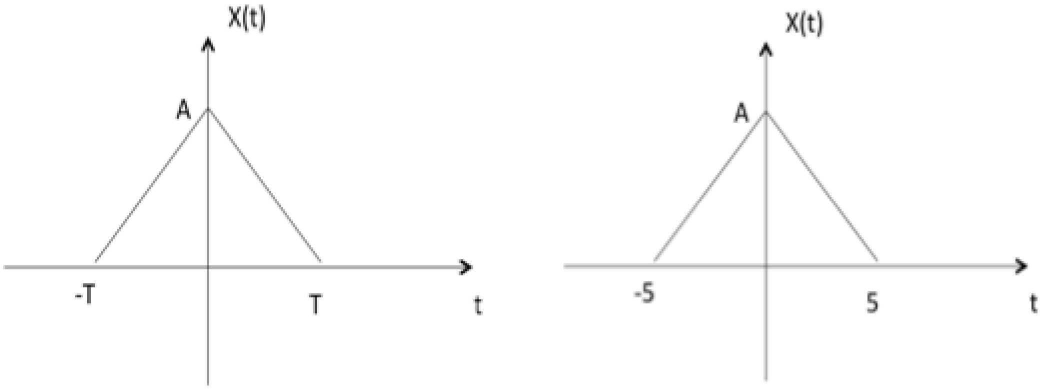
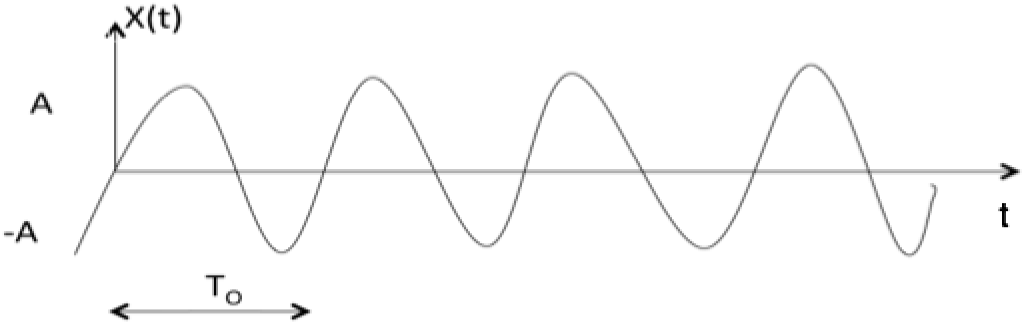


Fig:9 Representation of Triangular Signal

9. Sinusoidal Signal

Sinusoidal signal is in the form of x(t) = A cos($w_0 \pm \phi$) or A sin($w_0 \pm \phi$)



Where $T_0 = 2\pi/w_0$

10. **Sinc Function**

It is denoted as sinc(t) and it is defined as sinc(t)=sinπt/πt

=0 for t =±1, ±2, ±3...

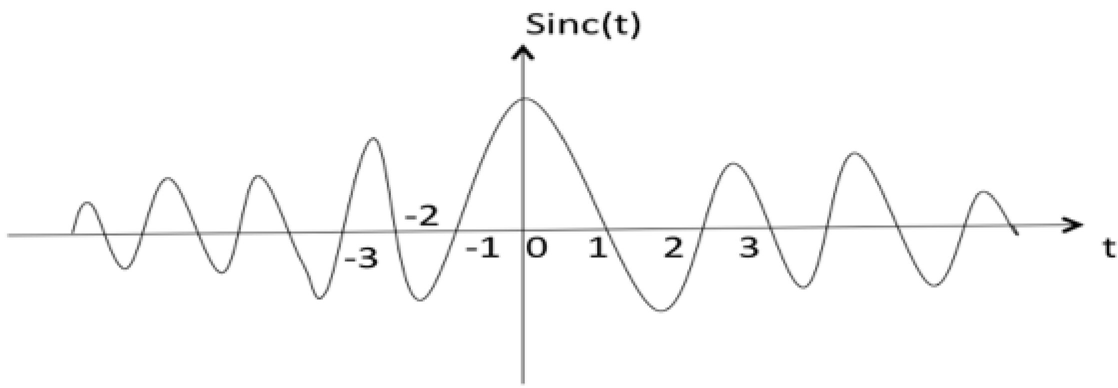


Fig:9 Representation of Sinc Function

11.Sampling Function

It is denoted as sa(t) and it is defined as

$$sa\ (t)=\sin t/t$$
$$=0 \text{ for } t=\pm\pi, \pm 2\pi, \pm 3\pi\ldots$$

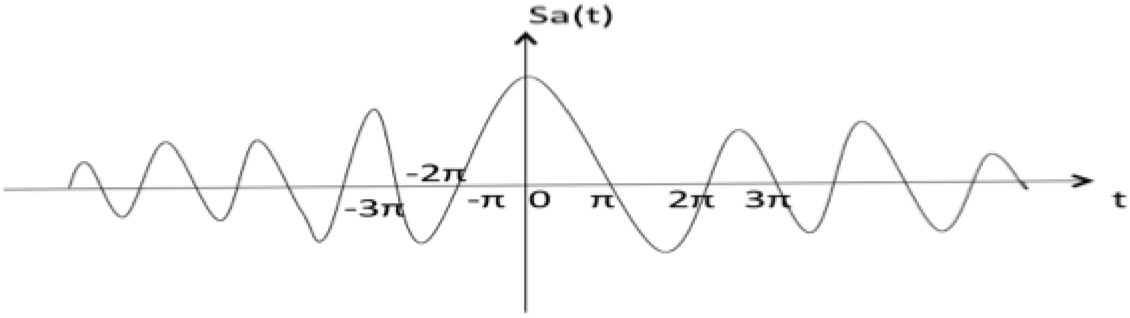


Fig:10 Sampling Fucntion

- ❖ Signals are classified into the following categories:
- Continuous Time and Discrete Time Signals
- Deterministic and Non-deterministic Signals
- Even and Odd Signals
- Periodic and Aperiodic Signals
- Energy and Power Signals
-
- Real and Imaginary Signals

1. Continuous Time and Discrete Time Signals

A signal is said to be continuous when it is defined for all instants of time.

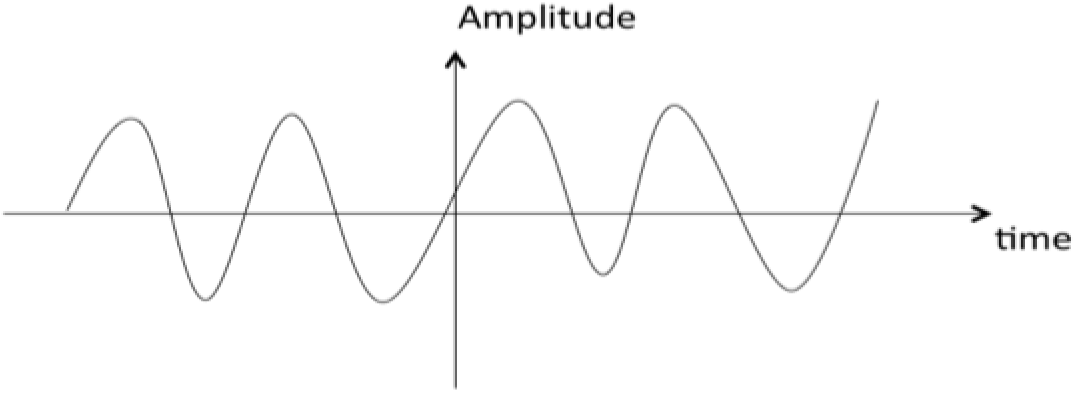


Fig:11 Representation of Continuous Signal

A signal is said to be discrete when it is defined at only discrete instants of time

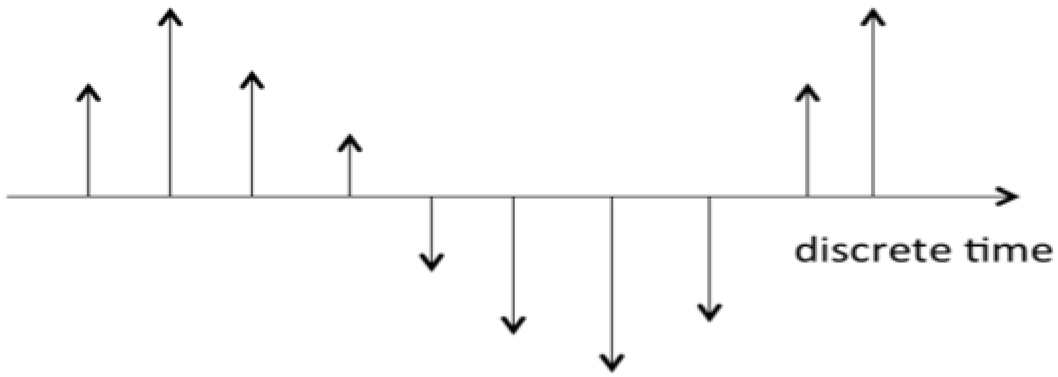


Fig:12 Representation of Discrete Signal

Deterministic and Non-deterministic Signals

A signal is said to be deterministic if there is no uncertainty with respect to its value at any instant of time. (or), signals which can be defined exactly by a mathematical formula are known as deterministic signals.

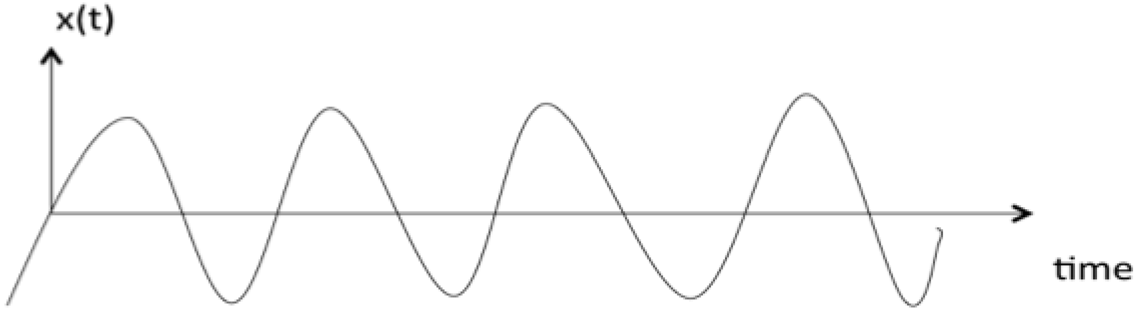


Fig:13 Representation of Deterministic Signal

A signal is said to be non-deterministic if there is uncertainty with respect to its value at some instant of time. Non-deterministic signals are random in nature hence they are called random signals. Random signals cannot be described by a mathematical equation. They are modelled in probabilistic terms.

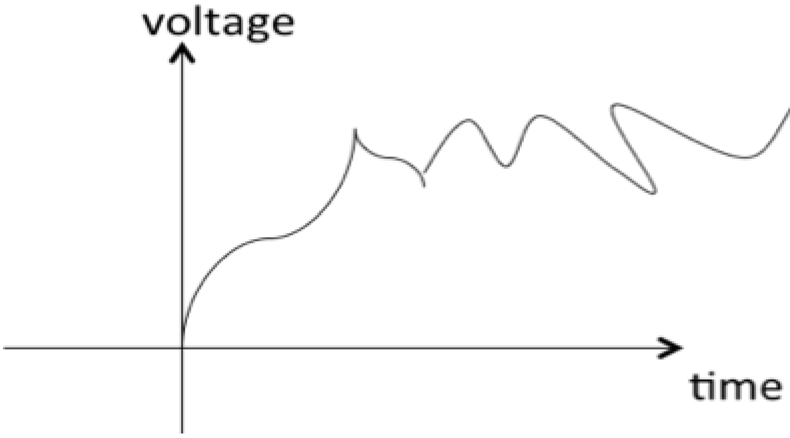


Fig:14 Representation of Non-Deterministic Signal

Even and Odd Signals

A signal is said to be even when it satisfies the condition x(t) = x(-t)

A signal is said to be odd when it satisfies the condition x(t) = -x(-t)

Periodic and Aperiodic Signals

A signal is said to be periodic if it satisfies the condition x(t) = x(t + T) or x(n) = x(n + N).

Where

T = fundamental time period,

1/T = f = fundamental frequency.

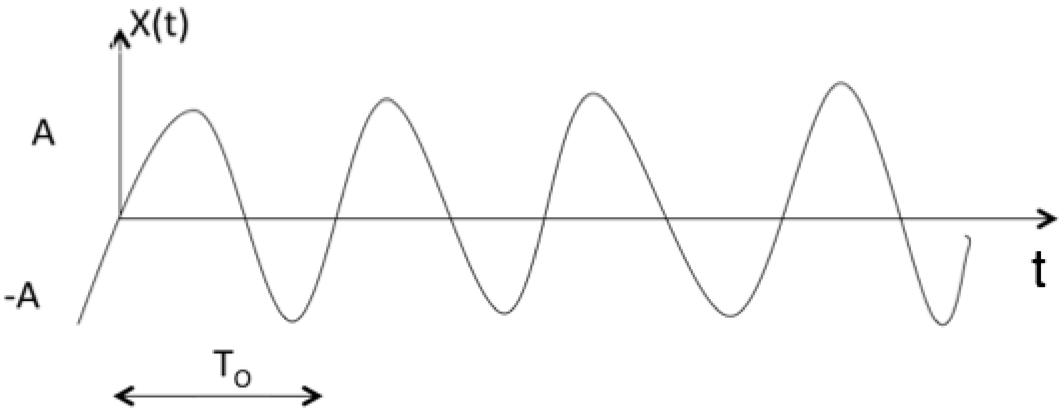


Fig:15 Representation of Periodic Signal

The above signal will repeat for every time interval T_0 hence it is periodic with period T_0.

Energy and Power Signals

A signal is said to be energy signal when it has finite energy.

$$\text{Energy } E = \int_{-\infty}^{\infty} x^2(t)dt$$

A signal is said to be power signal when it has finite power.

$$\text{Power } P = \lim_{T\to\infty} \frac{1}{2T} \int_{-T}^{T} x^2(t)dt$$

Point to Note : A signal cannot be both, energy and power simultaneously. Also, a signal may be neither energy nor power signal.

Power of energy signal = 0

Energy of power signal = ∞

Real and Imaginary Signals

A signal is said to be real when it satisfies the condition x(t) = x*(t)

A signal is said to be odd when it satisfies the condition x(t) = -x*(t)

Example:

If x(t)= 3 then x*(t)=3*=3 here x(t) is a real signal.

If x(t)= 3j then x*(t)=3j* = -3j = -x(t) hence x(t) is a odd signal.

Point to Note: For a real signal, imaginary part should be zero. Similarly, for an imaginary signal, real part should be zero.

There are two variable parameters in general:

1. Amplitude
2. Time

The following operation can be performed with amplitude:

Amplitude Scaling

C x(t) is a amplitude scaled version of x(t) whose amplitude is scaled by a factor C.

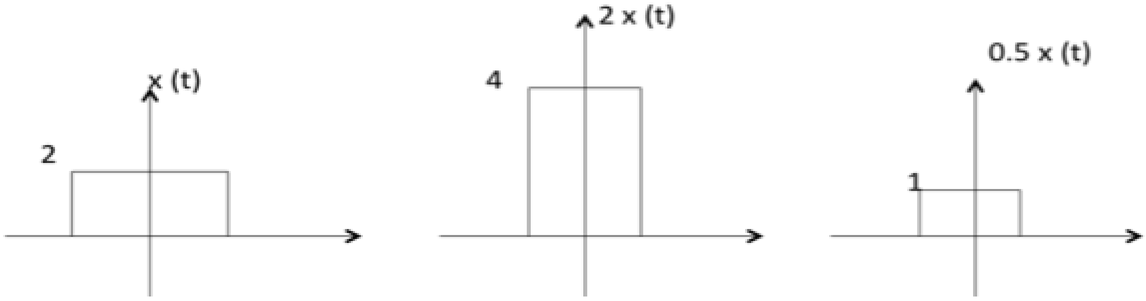


Fig:16 Amplitude Scaling

Addition

Addition of two signals is nothing but addition of their corresponding amplitudes. This can be best explained by using the following example:

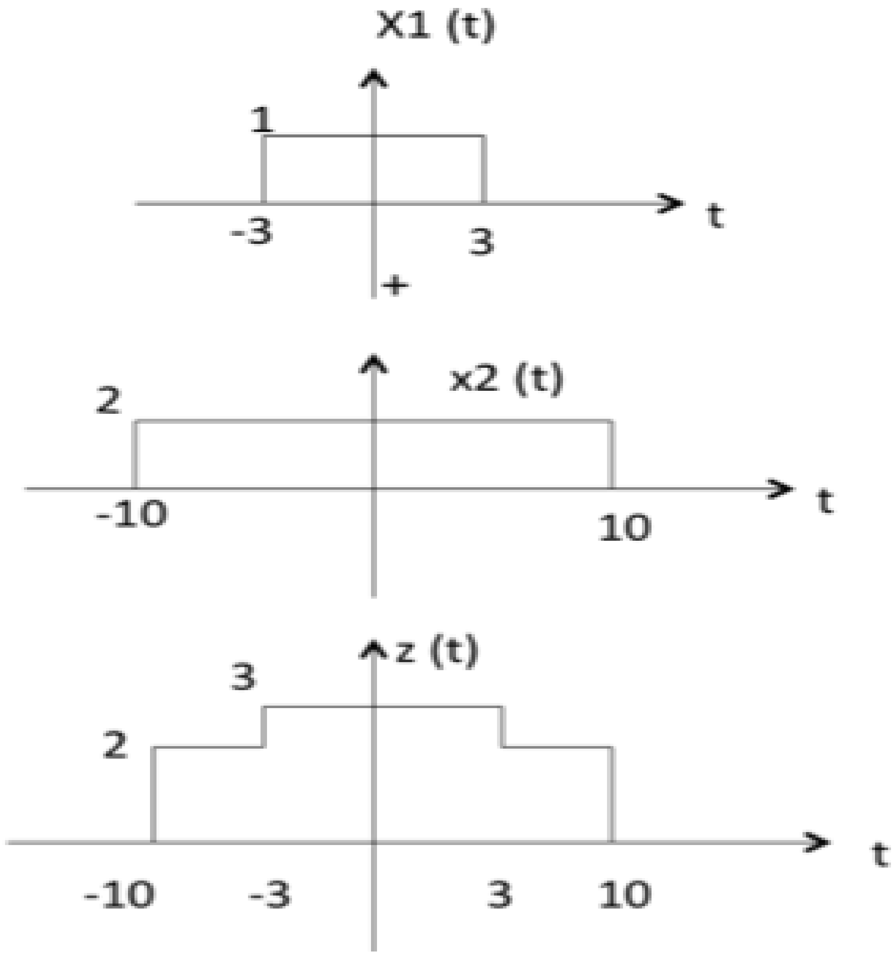


Fig:17 Addition of Two Signals

As seen from the diagram above,

$-10 < t < -3$ amplitude of z(t) = x1(t) + x2(t) = 0 + 2 = 2

$-3 < t < 3$ amplitude of z(t) = x1(t) + x2(t) = 1 + 2 = 3

$3 < t < 10$ amplitude of z(t) = x1(t) + x2(t) = 0 + 2 = 2

Subtraction

subtraction of two signals is nothing but subtraction of their corresponding amplitudes. This can be best explained by the following example:

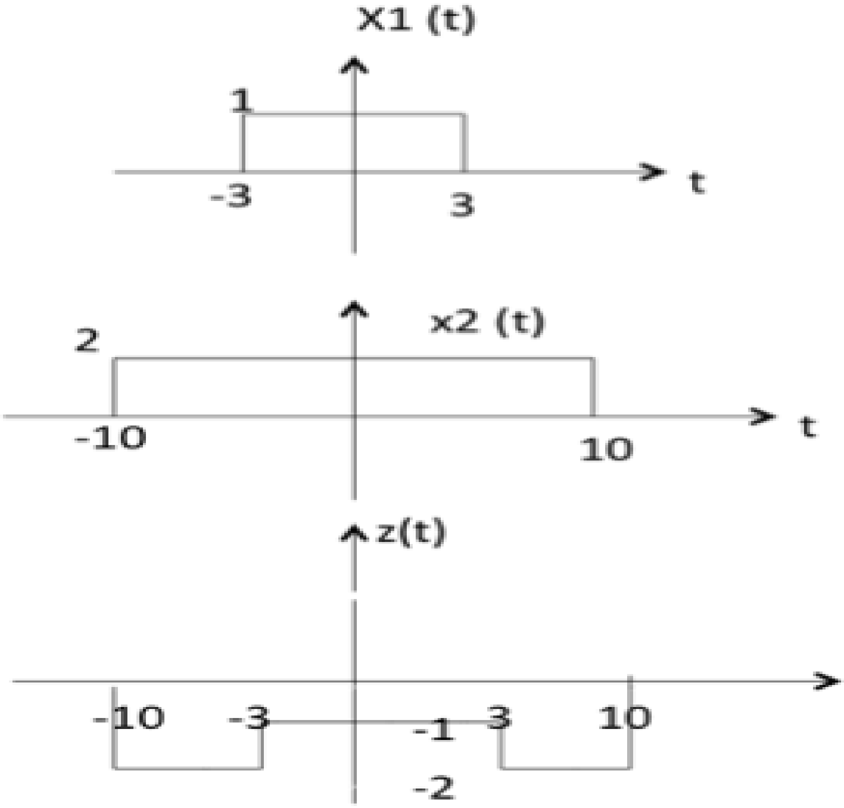


Fig:18 Subtraction of Two Signals

As seen from the above diagram ,

-10 < t < -3 amplitude of z (t) = x1(t) - x2(t) = 0 - 2 = -2

-3 < t < 3 amplitude of z (t) = x1(t) - x2(t) = 1 - 2 = -1

3 < t < 10 amplitude of z (t) = x1(t) + x2(t) = 0 - 2 = -2

Multiplication

Multiplication of two signals is nothing but multiplication of their corresponding amplitudes. This can be best explained by the following example:

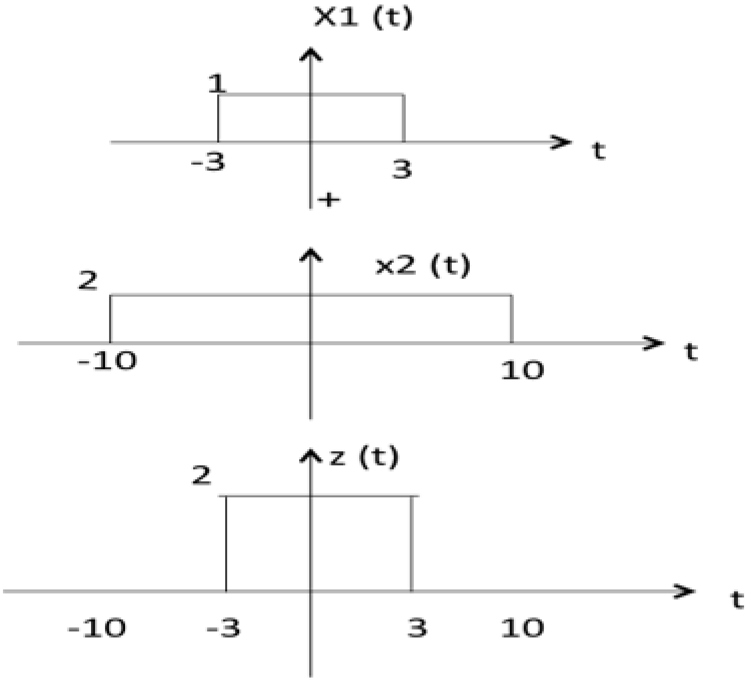


Fig:18 Multiplication of Two Signals

As seen from the above diagram,

$-10 < t < -3$ amplitude of z (t) = x1(t) ×x2(t) = 0 ×2 = 0

$-3 < t < 3$ amplitude of z (t) = x1(t) ×x2(t) = 1 ×2 = 2

$3 < t < 10$ amplitude of z (t) = x1(t) × x2(t) = 0 × 2 = 0

The following operations can be performed with time:

Time Shifting

$x(t \pm t_0)$ is time shifted version of the signal x(t).

$x(t + t_0)$ →→ negative shift

$x(t - t_0)$ →→ positive shift

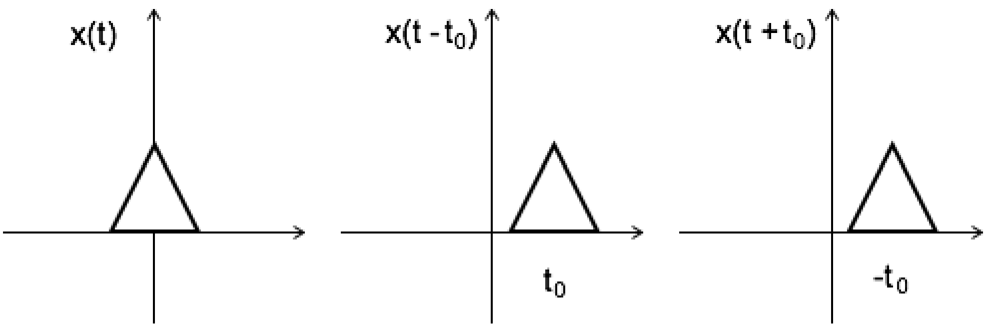


Fig:19 Time Shifting Property

Time Scaling

x(At) is time scaled version of the signal x(t). where A is always positive.

|A| > 1 →→ Compression of the signal
|A| < 1 →→ Expansion of the signal

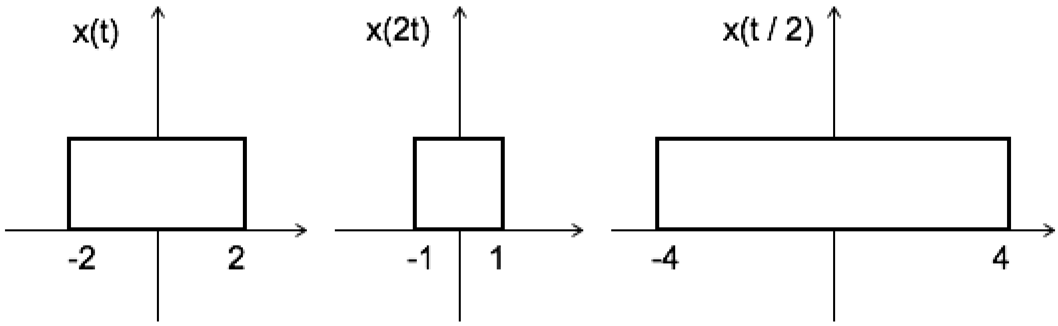


Fig:20 Time Scaling Property

Point to Note: u(at) = u(t) time scaling is not applicable for unit step function.

Time Reversal

x(-t) is the time reversal of the signal x(t).

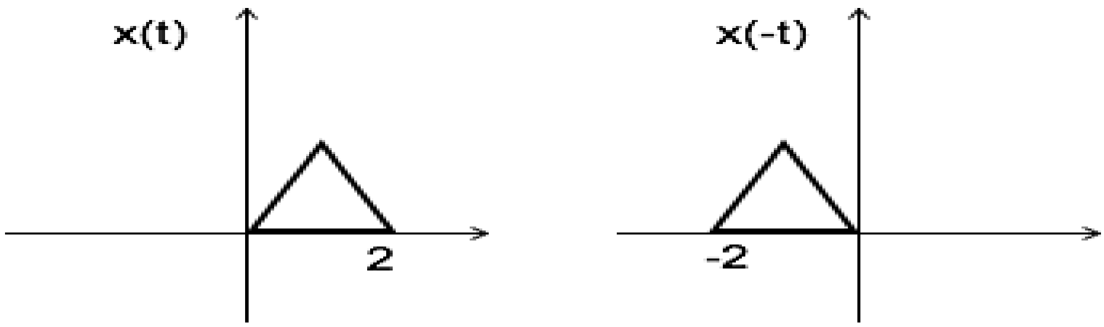


Fig:21 Time Reversal Property

Continuous Time Fourier Transform

Any continuous time periodic signal x(t) can be represented as a linear combination of complex exponentials and the Fourier coefficients (or spectrum) are discrete. The Fourier series can be applied to periodic signals only but the Fourier transform can also be applied to non-periodic functions like rectangular pulse, step functions, ramp function etc. The Fourier transform of Continuous Time signals can be obtained from Fourier series by applying appropriate conditions. The Fourier transform can be developed by finding Fourier series of a periodic function and the tending T to infinity.

Representation of Aperiodic signals:

Starting from the Fourier series representation for the continuous-time periodic square wave:

$$x(t) = \begin{cases} 1 & \text{for } |t| < T1 \\ 0 & \text{for } T1 < |t| < T/2 \end{cases} \qquad 1.1$$

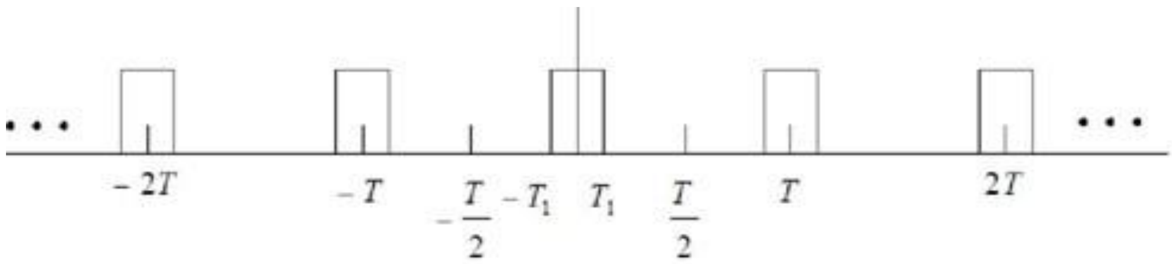


Fig:22 Representation of Aperiodic Signal

The Fourier coefficients a_k for this square wave are

$$a_k = \frac{2\sin(k\omega_0 T_1)}{k\omega_0 T}. \qquad 1.2$$

or alternatively

$$Ta_k = \left.\frac{2\sin(\omega T_1)}{\omega}\right|_{\omega=k\omega_0} \qquad 1.3$$

where 2 sin(wT1) / w represent the envelope of Tak

When T increases or the fundamental frequency w 0 2p / T decreases, the envelope is sampled with a closer and closer spacing. As T becomes arbitrarily large, the original periodic square wave approaches a rectangular pulse. Tak becomes more and more closely spaced samples of the envelope, as T →∞, the Fourier series coefficients approaches the envelope function.

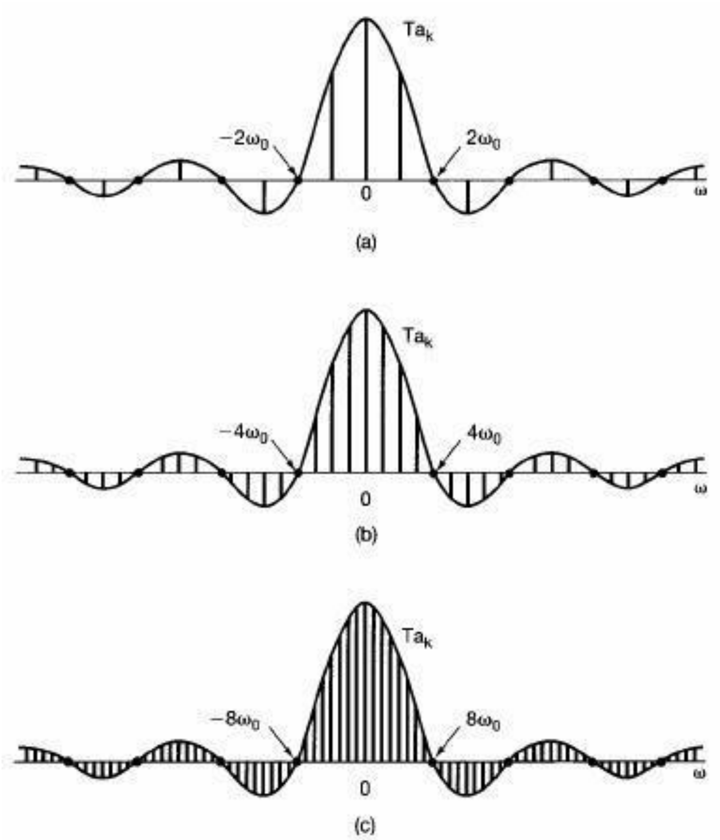
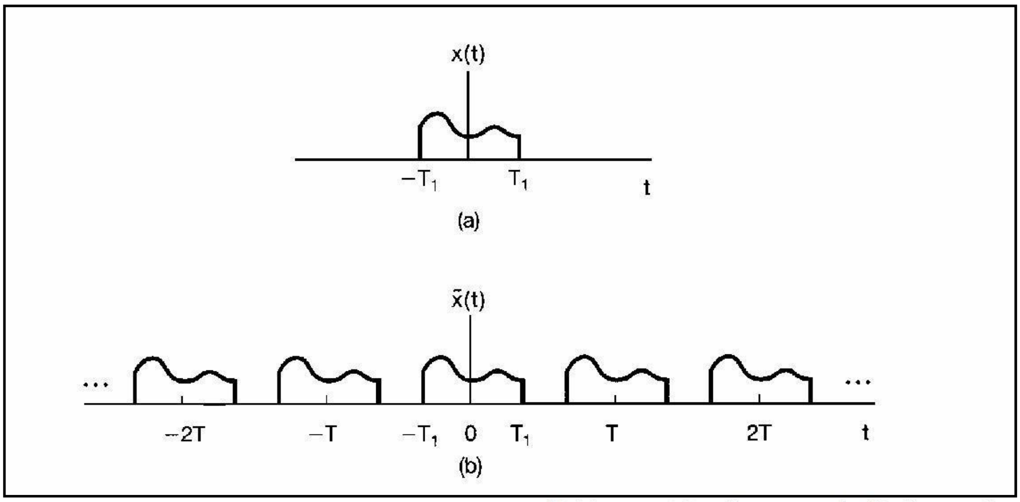


Fig:23 Fourier Series Coefficients of Envelope function

This sample illustrates the basic idea behind Fourier's development of a representation for aperiodic signals. Based on this idea, we can derive the Fourier transform for aperiodic signals.
From this aperiodic signal, we construct a periodic signal (t) , shown in the figure below.



As $T \to \infty$, $\tilde{x}(t) = x(t)$, for any infinite value of t.

The Fourier series representation of $\tilde{x}(t)$ is

$$\tilde{x}(t) = \sum_{}^{+\infty} a_k e^{jk\omega_0 t},$$

Since $\tilde{x}(t) = x(t)$ for $|t| < T/2$, and also, since $x(t) = 0$ outside this interval, so we have

$$a_k = \frac{1}{T}\int_{-T/2}^{T/2} \tilde{x}(t) e^{-jk\omega_0 t}\, dt.$$

$$a_k = \frac{1}{T}\int_{-T/2}^{T/2} x(t) e^{-jk\omega_0 t}\, dt = \frac{1}{T}\int_{-\infty}^{\infty} x(t) e^{-jk\omega_0 t}\, dt.$$

Define the envelope X (jw) of Tak as,

$$X(j\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t}\, dt.$$

2.6

we have for the coefficients a_k,

Then $\tilde{x}(t)$ can be expressed in terms of $X(j\omega)$, that is

$$\tilde{x}(t) = \sum_{k=-\infty}^{+\infty} \frac{1}{T} X(jk\omega_0) e^{jk\omega_0 t} = \frac{1}{2\pi} \sum_{k=-\infty}^{+\infty} X(jk\omega_0) e^{jk\omega_0 t} \omega_0.$$

X(t) can be expressed in terms of x(jw) that is

$$\tilde{x}(t) = \sum_{k=-\infty}^{+\infty} \frac{1}{T} X(jk\omega_0) e^{jk\omega_0 t} = \frac{1}{2\pi} \sum_{k=-\infty}^{+\infty} X(jk\omega_0) e^{jk\omega_0 t} \omega_0$$

As $T \to \infty$, $\tilde{x}(t) = x(t)$ and consequently,

This results in the following Fourier Transform.

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(j\omega) e^{j\omega t} d\omega \quad \textit{Inverse Fourier Transform}$$

and

$$X(j\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt \quad \textit{Fourier Transform}$$

Convolution Theorem:

Fourier transform of the convolution operation remarkably turns out to be a single complex multiplication of the respective Fourier transforms in the frequency domain. The quantities multiplied in the frequency domain are the complex spectra of the convolved signals. A convolution is defined as the integral over all space of one function at x times another function at u-x. The integration is taken over the variable x (which may be a 1D or 3D variable), typically from minus infinity to infinity over all the dimensions. So the convolution is a function of a new variable u, as shown in the following equations. The cross in a circle is used to indicate the convolution operation.

$$\mathrm{f}(x) \otimes \mathrm{g}(x) = \mathrm{g}(x) \otimes \mathrm{f}(x)$$

$$\mathrm{C}(u) = \int_{-\infty}^{\infty} \mathrm{f}(x)\mathrm{g}(u-x)dx$$

$$x' = u - x, \; dx' = -dx$$

$$\mathrm{C}(u) = -\int_{\infty}^{-\infty} \mathrm{f}(u-x')\mathrm{g}(x')dx'$$

`

Note that, because the sign of the variable of integration changed, we have to change the signs of the limits of integration. Because these limits are infinite, the shift of the origin (by the vector u) doesn't change the magnitude of the limits.

Now we reverse the order of the limits, which changes the sign of the equation, and swap the order of the functions g and f.

$$\mathrm{C}(u) = \int_{-\infty}^{\infty} \mathrm{g}(x')\mathrm{f}(u - x')dx'$$

It doesn't matter whether we call the variable of integration x' or x, so we put back x, to get the result we wanted to prove.

$$\mathrm{C}(u) = \int_{-\infty}^{\infty} \mathrm{g}(x)\mathrm{f}(u - x)dx$$

Because there will be so many Fourier transforms in the rest of this presentation, it is useful to introduce a shorthand notation. T will be used to indicate a forward Fourier transform, and its inverse to indicate the inverse Fourier transform.

$$\mathrm{T}(\mathrm{f}(\mathbf{r})) = \int_{space} \mathrm{f}(\mathbf{r})\exp(2\pi i\mathbf{s}\cdot\mathbf{r})d\mathbf{r}$$

There are two ways of expressing the convolution theorem:

The Fourier transform of a convolution is the product of the Fourier transforms. The

Fourier transform of a product is the convolution of the Fourier transforms.

$$\mathrm{T}(\mathrm{f}\otimes\mathrm{g}) = \mathrm{T}(\mathrm{f})\mathrm{T}(\mathrm{g})$$
$$\mathrm{T}(\mathrm{f}\,\mathrm{g}) = \mathrm{T}(\mathrm{f})\otimes\mathrm{T}(\mathrm{g})$$

The convolution theorem is useful, in part, because it gives us a way to simplify many calculations. Convolutions can be very difficult to calculate directly, but are often much easier to calculate using Fourier transforms and multiplication.

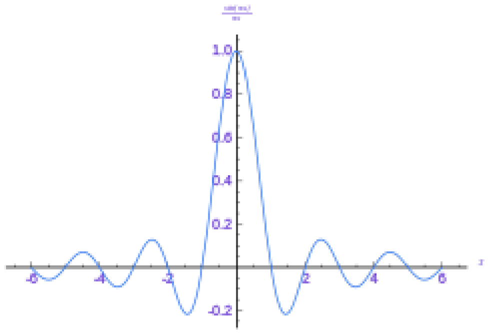
Sampling Theorem

In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).A sample is a value or set of values at a point in time and/or space. A sampler is a subsystem or operation that extracts samples from a continuous signal. A theoretical ideal sampler produces samples equivalent to the instantaneous value of the continuous signal at the desired points.

Sampling Theorem is a process of converting a signal (for example, a function of continuous time and/or space) into a numeric sequence (a function of discrete time and/or space)

If a function x(t) contains no frequencies higher than B hertz, it is completely determined by giving its ordinates at a series of points spaced 1/(2B) seconds apart. A sufficient sample-rate is therefore 2B samples/second, or anything larger. Equivalently, for a given sample rate fs, perfect reconstruction is guaranteed possible for a bandlimit B < fs/2.

When the bandlimit is too high (or there is no bandlimit), the reconstruction exhibits imperfections known as aliasing. Modern statements of the theorem are sometimes careful to explicitly state that x(t) must contain no sinusoidal component at exactly frequency B, or that B must be strictly less than ½ the sample rate. The two thresholds, 2B and fs/2 are respectively called the Nyquist rate and Nyquist frequency. And respectively, they are attributes of x(t) and of the sampling equipment. The condition described by these inequalities is called the Nyquist criterion, or sometimes the Raabe condition. The theorem is also applicable to functions of other domains, such as space, in the case of a digitized image. The only change, in the case of other domains, is the units of measure applied to t, fs, and B.



The normalized sinc function:sin(πx) / (πx) ... showing the central peak atx= 0, and zero-crossings at the other integer values of x. The symbol T = 1/fs is customarily used to represent the interval between samples and is called the sample period or sampling interval. And the samples of function x(t) are commonly denoted by x[n] = x(nT) (alternatively "xn" in older signal processing literature), for all integer values of n. A mathematically ideal way to interpolate the sequence involves the use of sinc functions. Each sample in the sequence is replaced by a sinc function, centered on the time axis at the original location of the sample, nT, with the amplitude of the sinc function scaled to the sample value, x[n]. Subsequently, the sinc functions are summed into a continuous function. A mathematically equivalent method is to convolve one sinc function with a series of Dirac delta pulses, weighted by the sample values. Neither method is numerically practical. Instead, some type of approximation of the sinc functions, finite in length, is used. The imperfections attributable to the approximation are known as interpolation error.

Practical digital-to-analog converters produce neither scaled and delayed sinc functions, nor ideal Dirac pulses. Instead they produce a piecewise- constant sequence of scaled and delayed rectangular pulses (the zero-order hold), usually followed by an "anti-imaging filter" to clean up spurious high-frequency content.

Various types of distortion can occur, including: Aliasing. Some amount of aliasing is inevitable because only theoretical, infinitely long, functions can have no frequency content above the Nyquist frequency. Aliasing can be made arbitrarily small by using a sufficiently large order of the anti-aliasing filter. Aperture error results from the fact that the sample is obtained as a time average within a sampling region, rather than just being equal to the signal value at the sampling instant. In a capacitor-based sample and hold circuit, aperture error is introduced because the capacitor cannot instantly change voltage thus requiring the sample to have non-zero width. Jitter or deviation from the precise sample timing intervals. Noise, including thermal sensor noise, analog circuit noise, etc. Slew rate limit error, caused by the inability of the ADC input value to change sufficiently rapidly.

Quantization as a consequence of the finite precision of words that represent the converted values. Error due to other non-linear effects of the mapping of input voltage to converted output value (in addition to the effects of quantization). Although the use of oversampling can completely eliminate aperture error and aliasing by shifting them out of the pass band, this technique cannot be practically used above a few GHz, and may be prohibitively expensive at much lower frequencies. Furthermore, while oversampling can reduce quantization error and non-linearity, it cannot eliminate these entirely. Consequently, practical ADCs at audio frequencies typically do not exhibit aliasing, aperture error, and are not limited by quantization error. Instead, analog noise dominates. At RF and microwave frequencies where oversampling is impractical and filters are expensive, aperture error, quantization error and aliasing can be significant limitations.

Jitter, noise, and quantization are often analyzed by modeling them as random errors added to the sample values. Integration and zero-order hold effects can be analyzed as a form of low-pass filtering. The non-linearities of either ADC or DAC are analyzed by replacing the ideal linear function mapping with a proposed nonlinear function.

Communication Electronics.

Fourier series (periodic signals or continous signal)

$$F[x(t)] = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t}\, dt$$

Fourier transform (a periodic signals).

Signals
Unit step
unit ramp
exponential fn
Impluse fn or Delta fn [$\delta(t)$]

$X(\omega)$ frequency domain representation of time domain function $x(t)$

$$F^{-1}[X(\omega)] = x(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega)\, e^{j\omega t}\, d\omega$$

Dirichlet's rule.

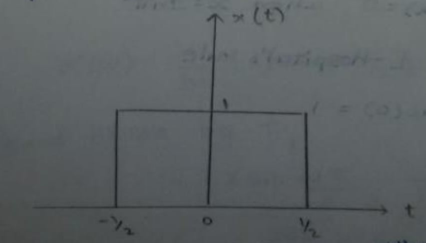
$x(t)$ satisfies Dirichlet's condition.

i, $x(t)$ is a single valued function with a finite number of maxima, minima, finite and a number of dicontinous in any finite time interval.

ii, $x(t)$ is a absolute integralble

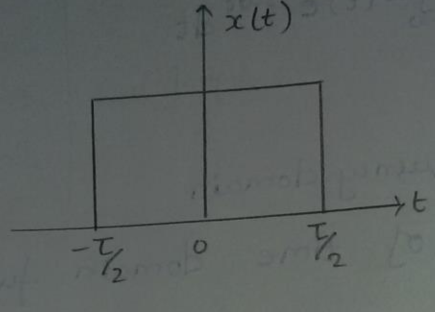
$$\int_{-\infty}^{\infty} |x(t)|\, dt < \infty$$

Gate function. (rectangular pulse)



rect) or rect($\frac{t}{\tau}$) $\left[\frac{\text{unit amplitude}}{\text{width } \tau}\right]$

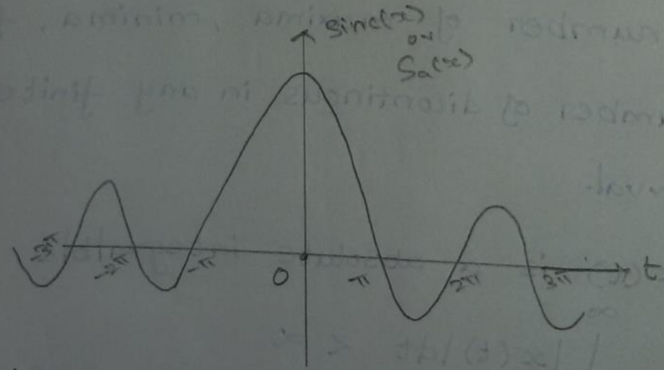
$$x(t) = rect\left(\frac{t}{\tau}\right) = \begin{cases} 1 & -\tau/2 < t < \frac{\tau}{2} \\ 0 & \text{otherwise.} \end{cases}$$



Sampling function (Interpolating fn or Sinc function)

It is also called filtering function

$$sinc(x) = S_a(x) = \frac{\sin x}{x}.$$



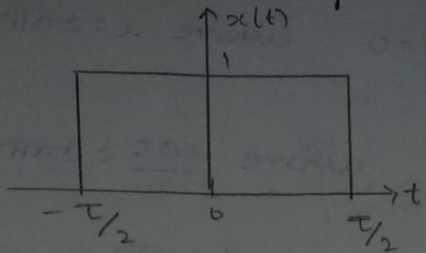
Condition.

i, $sinc(x)$ should be even fn of x.

ii, $sinc(x) = 0$ when $x = \pm n\pi$

iii, Using L-Hospital's rule

Find the Fouier transform of Gate junction



$$x(t) = \text{rect}\left(\frac{t}{\tau}\right) = \begin{cases} 1 & -\tau/2 < t < \tau/2 \\ 0 & \text{otherwise.} \end{cases}$$

Fourier transformen

$$X(\omega) = F[x(t)] = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t}\, dt$$

$$= \int_{-\infty}^{\infty} \text{rect}\left(\frac{t}{\tau}\right) e^{-j\omega t}\, dt$$

$$= \int_{-\tau/2}^{\tau/2} 1 \cdot e^{-j\omega t}\, dt$$

$$= \left[\frac{e^{-j\omega t}}{-j\omega}\right]_{-\tau/2}^{\tau/2}$$

$$= \frac{-1}{j\omega}\left[e^{-j\omega \tau/2} - e^{j\omega\tau/2}\right]$$

$$= \frac{1}{j\omega}\left[e^{j\omega\tau/2} - e^{-j\omega\tau/2}\right]$$

$$X(\omega) = \frac{1}{j\omega}\left(2j \sin \omega \tau/2\right)$$

we have $\mathrm{Sinc}(x) = 0$ where $x = \pm n\pi$

$\mathrm{sinc}\left(\frac{\omega\tau}{2}\right) = 0$ where $\frac{\omega\tau}{2} = \pm n\pi$

$\omega = \pm\frac{2n\pi}{\tau}$ $(n = 1, 2, 3 \ldots)$.

Unit impulse function. (Delta function).

$\delta(t) = 0$, $t \neq 0$.

$$\int_{-\infty}^{\infty} \delta(t)\,dt = 0.$$

Shifting property of impulse function.

$$\int_{-\infty}^{\infty} x(t)\,\delta(t)\,dt = x(0)\cdot 1 \quad [\text{when } t = 0]$$

$$= x(0)$$

when $t = t_0$

$$\int_{-\infty}^{\infty} x(t)\,\delta(t - t_0)\,dt = x(t_0)$$

Find fouirer transform $\int_{-\infty}^{\infty}[t^2+1]\,\delta(t)\,dt$

$$\int_{-\infty}^{\infty}[t^2+1]\,\delta(t)\,dt = \int_{-\infty}^{\infty} t^2\,\delta(t)\,dt + \int_{-\infty}^{\infty}\delta(t)\,dt$$

using shifting property.

when $t = 0$.

$$0 + 1 = 1.$$

Find fouirer transform. $\int_{-1}^{2}(t^4+1)\,\delta(t-1)\,dt$

$$\int_{-1}^{2}(t^4+1)\,\delta(t-1)\,dt$$

$$= \int_{-1}^{2} t^4\,\delta(t-1)\,dt + \int_{-1}^{2}\delta(t-1)\,dt.$$

Using Shifting property.

when, $t = 1$.

$$= [t^4]_{t=1} + \cancel{\int_{-\infty}^{\infty}\delta(t)\,dt}\ \int_{-\infty}^{\infty}\delta(1-1)\,dt \quad t=1$$

$\int\delta(t)\,dt = 1$

$$= 1 + 1 = 2.$$

Find fouier transform of Impulse function

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

Fouier transform,

$$X(\omega) = F[x(t)] = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t}\, dt$$

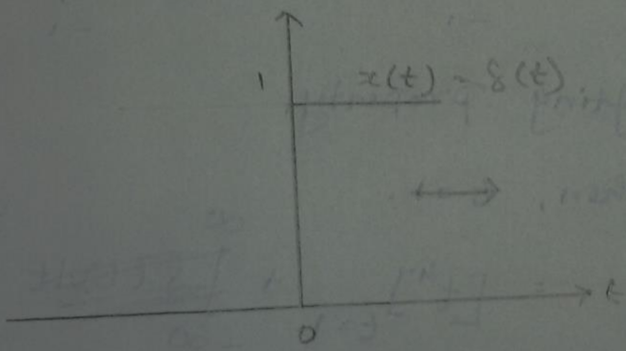
$$= \int_{-\infty}^{\infty} \delta(t)\, e^{-j\omega t}\, dt$$

Using shifiting property.

$$X(\omega) = \left[e^{-j\omega t}\right]_{t=0}.$$

$$X(\omega) \leftrightarrow 1.$$

$$\delta(t) \longleftrightarrow 1$$



i) Unit impulse function contains entire frequency components having identical magnitude.

ii) Band width of Unit impulse function

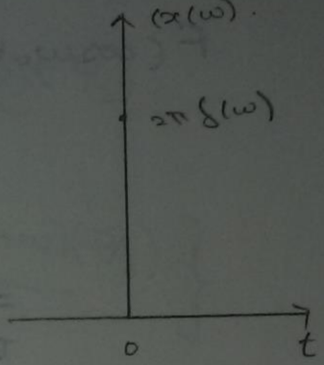
Find inverse fouier transform of $\delta(\omega)$.

$$F^{-1}[\delta(\omega)] = \frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

Using shifting property.

$$= \frac{1}{2\pi}\left[e^{j\omega t}\right]_{\omega=0}.$$

$$= \frac{1}{2\pi}$$



$$F\left[\frac{1}{2\pi}\right] = \delta(\omega)$$

$$\frac{1}{2\pi} \longleftrightarrow \delta(\omega)$$

$$1 \longleftrightarrow 2\pi\,\delta(\omega)$$

Find fouier transform sinusodal $\cos \omega_0 t$.

$$x(t) = \cos \omega_0 t$$

Fouier transform

$$X(\omega) = F[x(t)] = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt.$$

$$= \int_{-\infty}^{\infty} \cos \omega_0 t \; e^{-j\omega t} dt.$$

$$= \int_{-\infty}^{\infty} \frac{e^{j\omega_0 t} + e^{-j\omega_0 t}}{2} \cdot e^{-j\omega t} dt$$

$$= \frac{1}{2}\left[2\pi\delta(\omega-\omega_0) + 2\pi\delta(\omega+\omega_0)\right]$$

Impulse.

Fourier transform of $\cos\omega_0 t$. over the interval $(-\infty,\infty)$

$$F(\cos\omega_0 t) = \underset{\tau\to\infty}{Lt}\int_{-\tau/2}^{\tau/2} \cos\omega_0 t\, e^{-j\omega t}\,dt$$

$$= \underset{\tau\to\infty}{Lt}\int_{-\tau/2}^{\tau/2}\left[\cos\omega_0 t\cos\omega t - j\cos\omega_0 t\sin\omega t\right]dt$$

$$2\cos A\cos B = \cos(A-B) + \cos(A+B)$$

$$2\cos A\sin B = \sin(A+B) + \sin(A-B)$$

$$\Rightarrow \underset{\tau\to\infty}{Lt}\int_{-\tau/2}^{\tau/2}\frac{\cos(\omega_0-\omega)t + \cos(\omega_0+\omega)t}{2} - j\left(\frac{\sin(\omega_0+\omega)t + \sin(\omega_0-\omega)t}{2}\right.$$

$$= \underset{\tau\to\infty}{Lt}\ \frac{1}{2}\left[\frac{\sin(\omega_0-\omega)t}{\omega_0-\omega} + \frac{\sin(\omega_0+\omega)t}{\omega_0+\omega} + j\frac{\cos(\omega_0+\omega)t}{\omega_0+\omega}\right.$$

$$\left. + j\frac{\cos(\omega_0-\omega)t}{(\omega_0-\omega)}\right]_{-\tau/2}^{\tau/2}$$

$$= \underset{\tau\to\infty}{Lt}\ \frac{1}{2}\left[\frac{\sin(\omega_0-\omega)\frac{\tau}{2} + j\cos(\omega_0-\omega)\frac{\tau}{2}}{\omega_0-\omega}\right.$$

$$+\ \frac{\sin(\omega_0+\omega)\frac{\tau}{2} + j\cos(\omega_0+\omega)\frac{\tau}{2}}{\omega_0+\omega}$$

$$-\left\{\frac{\sin(\omega_0-\omega)\left(-\frac{\tau}{2}\right) + j\cos(\omega_0-\omega)\left(-\frac{\tau}{2}\right)}{\omega_0-\omega}\right.$$

$$\left.+\ \frac{\sin(\omega_0+\omega)\left(-\frac{\tau}{2}\right) + j\cos(\omega_0+\omega)\left(-\frac{\tau}{2}\right)}{\omega_0+\omega}\right\}.$$

$$\Rightarrow \underset{\tau\to\infty}{Lt}\ \frac{1}{2}\left[\frac{\sin(\omega_0-\omega)\frac{\tau}{2}}{\omega_0-\omega} + \cancel{j\,\frac{\cos(\omega_0-\omega)\frac{\tau}{2}}{\omega_0-\omega}}\right.$$

$$+\ \frac{\sin(\omega_0+\omega)\frac{\tau}{2}}{\omega_0+\omega} + \cancel{j\,\frac{\cos(\omega_0+\omega)\frac{\tau}{2}}{\omega_0+\omega}}$$

$$+\ \frac{\sin(\omega_0-\omega)\frac{\tau}{2}}{\omega_0-\omega} - \cancel{j\,\frac{\cos(\omega_0-\omega)\frac{\tau}{2}}{\omega_0-\omega}}$$

$$\left.+\ \frac{\sin(\omega_0+\omega)\frac{\tau}{2}}{\omega_0+\omega} - \cancel{j\,\frac{\cos(\omega_0+\omega)\frac{\tau}{2}}{\omega_0+\omega}}\right]$$

Properties of FT.

1. Symmetry.

$$f(t) \longleftrightarrow F(\omega)$$

$$F(t) \longleftrightarrow 2\pi f(\omega).$$

Proof $f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(\omega) e^{j\omega t} d\omega.$

$$2\pi f(t) = \int_{-\infty}^{\infty} F(\omega) e^{j\omega t} d\omega.$$

$f(t) = f(-t)$

$$2\pi f(-t) = \int_{-\infty}^{\infty} F(\omega) e^{-j\omega t} d\omega.$$

change t by ω

ω by ω.

$$2\pi f(-\omega) = \int_{-\infty}^{\infty} F(t) e^{-j\omega t} d\omega.$$

$$2\pi f(-\omega) = F[F(t)]$$

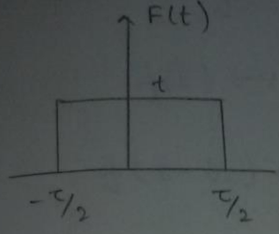
$$2\pi f(-\omega) \longleftrightarrow F(t)$$

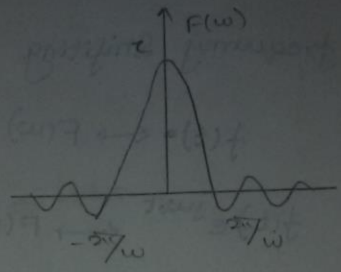
Limitation.

$F(t)$ is an even fn. Symmetric holds

Perfectly $f(-\omega) = f(\omega)$

$$2\pi f(\omega) \longleftrightarrow F(t)$$





2. Linearity.

$$f_1(t) \longleftrightarrow F_1(\omega)$$
$$f_2(t) \longleftrightarrow F_2(\omega)$$

Proof

$$c_1 f_1(t) + c_2 f_2(t) \longleftrightarrow c_1 F_1(\omega) + c_2 F_2(\omega)$$

c_1 and c_2 are constant.

3. Scaling property.

$$f(t) \longleftrightarrow F(\omega)$$
$$f(ct) \longleftrightarrow \frac{1}{c} F\left(\frac{\omega}{c}\right) \qquad c \rightarrow \text{real constant}$$

Proof.

$$F[f(ct)] = \int_{-\infty}^{\infty} f(ct)\, e^{-j\omega t}\, dt$$

Let $ct = z$

$$t = \frac{z}{c}, \quad dt = \frac{dz}{c}$$

$$= \int_{-\infty}^{\infty} f(z)\, e^{-j\omega \frac{z}{c}}\, \frac{dz}{c}$$

$$F[f(ct)] = \frac{1}{c} F\left(\frac{\omega}{c}\right)$$

4. frequency Shifting.

$$f(t) \longleftrightarrow F(\omega)$$

$$f(t)e^{j\omega_0 t} \longleftrightarrow F(\omega-\omega_0).$$

Proof.

$$F[f(t)e^{j\omega_0 t}] = \int_{-\infty}^{\infty} f(t)e^{j\omega_0 t} e^{-j\omega t} dt.$$

$$= \int_{-\infty}^{\infty} f(t)e^{-j(\omega-\omega_0)t} dt$$

$$= F(\omega-\omega_0).$$

A fn is multiplied by a factor $e^{j\omega_0 t}$ results in a shift of ω_0 in a frequency domain. This also results frequency translation Theorem.

eg: $f(t)\cos\omega_0 t = f(t)\left[\dfrac{e^{j\omega_0 t}+e^{-j\omega_0 t}}{2}\right]$

$$= \frac{1}{2}f(t)e^{j\omega_0 t} + \frac{1}{2}f(t)e^{-j\omega_0 t}$$

$$= \frac{1}{2}[F(\omega-\omega_0) + F(\omega+\omega_0)]$$

5, Time Shifting.

$$f(t) \leftrightarrow F(\omega).$$

$$f(t-t_0) \longleftrightarrow F(\omega) e^{-j\omega_0 t}.$$

$$F[f(t-t_0)] = \int_{-\infty}^{\infty} f(t-t_0) e^{-j\omega t} dt.$$

$$t - t_0 = z.$$

$$t = z + t_0$$

$$dt = dz.$$

$$= \int_{-\infty}^{\infty} f(z) e^{-j\omega(z+t_0)} dz.$$

$$= F(\omega) e^{-j\omega t_0}$$

$f(t)$ is shifted in time domain t_0 seconds

its magnitude Spectrum $|F(\omega)|$ is not affected by its phase spactrum and it is shifted by $-\omega t_0$

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DEPARTMENT OF PHYSICS

SPHA5304 – COMMUNICATION ELECTRONICS

UNIT II PULSE MODULATION AND COMMUNICATION

UNIT 2 PULSE MODULATION AND COMMUNICATION

Pulse amplitude modulation – Natural sampling -Instantaneous sampling Transmission of PAM signals – Pulse width modulation – Time division multiplexing and frequency division multiplexing – Band width requirements for PAM signals – Pulse code modulation – Principles of PCU – Quantizing noise – Generation and demodulation of PCM – Effects of noise – Advantages and application of PCM – Differential PCM (DPCM) – Delta modulation.

Communication can be defined as the process of exchange of information through means such as words, actions, signs, etc., between two or more individuals. For any living being, while co-existing, there occurs the necessity of exchange of some information. Whenever a need for exchange of information arises, some means of communication should exist. While the means of communication, can be anything such as gestures, signs, symbols, or a language, the need for communication is inevitable. Language and gestures play an important role in human communication, while sounds and actions are important for animal communication. However, when some message has to be conveyed, a communication has to be established.

Parts of Communication System

Any system which provides communication, consists of the three important and basic parts as shown in the following figure.

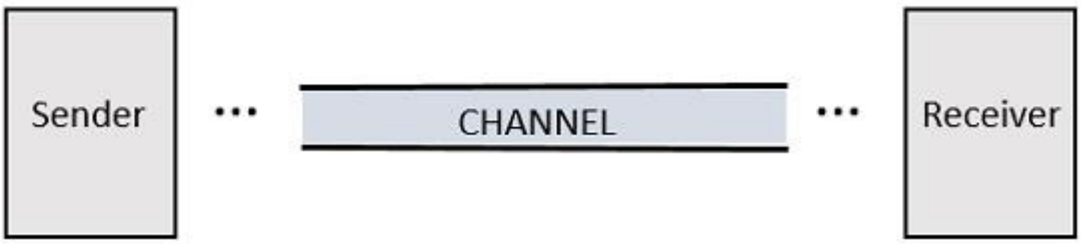

Fig:1 A simple Communication System

- The **Sender** is the person who sends a message. It could be a transmitting station from where the signal is transmitted.
- The **Channel** is the medium through which the message signals travel to reach the destination.
- The **Receiver** is the person who receives the message. It could be a receiving station where the signal transmitted is received.

Signal?

Conveying an information by some means such as gestures, sounds, actions, etc., can be termed as signaling. Hence, a signal can be a source of energy which transmits some information. This signal helps to establish communication between a sender and a receiver. An electrical impulse or an electromagnetic wave which travels a distance to convey a message, can be termed as a signal in communication systems. Depending on their characteristics, signals are mainly classified into two types: Analog and Digital. Analog and Digital signals are further classified, as shown in the below figure.

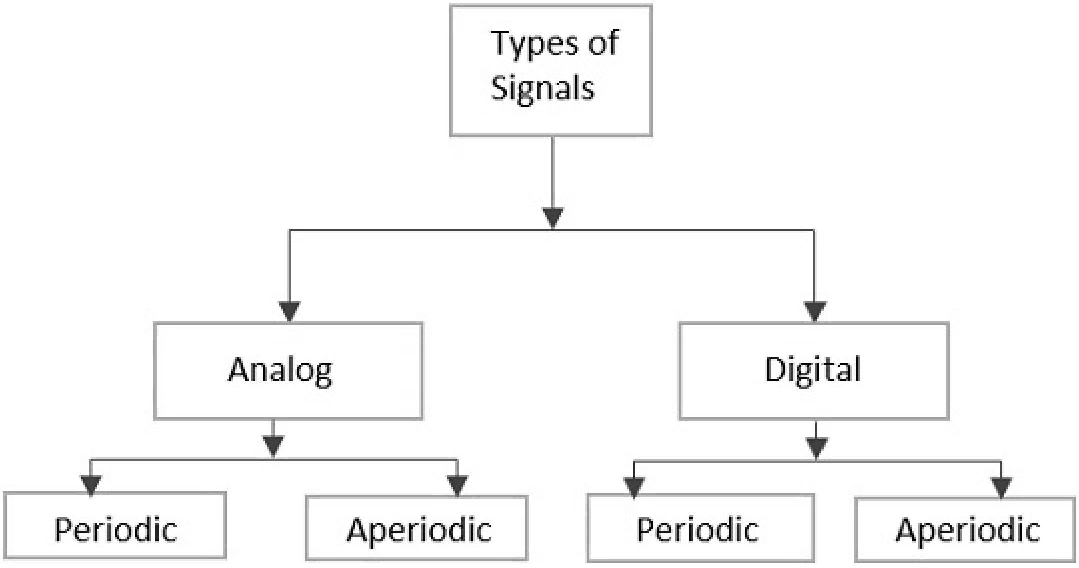

Fig:2 Classifications of Signals

Analog Signal

A continuous time varying signal, which represents a time varying quantity can be termed as an Analog Signal. This signal keeps on varying with respect to time, according to the instantaneous values of the quantity, which represents it.

Digital Signal

A signal which is discrete in nature or which is non-continuous in form can be termed as a Digital signal. This signal has individual values, denoted separately, which are not based on the previous values, as if they are derived at that particular instant of time.

The binary digits which has only 1s and 0s are mostly termed as digital values. Hence, the signals which represent 1s and 0s are also called as digital signals. The communication based on digital signals and digital values is called as Digital Communication.

Periodic Signal

Any analog or digital signal, that repeats its pattern over a period of time, is called as a Periodic Signal. This signal has its pattern continued repeatedly and is easy to be assumed or to be calculated.

Aperiodic Signal

Any analog or digital signal, that doesn't repeat its pattern over a period of time, is called as Aperiodic Signal. This signal has its pattern continued but the pattern is not repeated and is not so easy to be assumed or to be calculated. In general, the signals which are used in communication systems are analog in nature, which are transmitted in analog or converted to digital and then transmitted, depending upon the requirement. But for a signal to get transmitted to a distance, without the effect of any external interferences or noise addition and without getting faded away, it has to undergo a process called as Modulation.

What is Signal Modulation?

A message carrying signal has to get transmitted over a distance and for it to establish a reliable communication, it needs to take the help of a high frequency signal which should not affect the original characteristics of the message signal. The characteristics of the message signal, if changed, the message contained in it also alters. Hence it is a must to take care of the message signal. A high frequency signal can travel up to a longer distance, without getting affected by external disturbances. We take the help of such high frequency signal which is called as a carrier signal to transmit our message signal. Such a process is simply called as Modulation.

Modulation is the process of changing the parameters of the carrier signal, in accordance with the instantaneous values of the modulating signal.

Need for Modulation

The baseband signals are incompatible for direct transmission. For such a signal, to travel longer distances, its strength has to be increased by modulating with a high frequency carrier wave, which doesn't affect the parameters of the modulating signal.

Advantages of Modulation

The antenna used for transmission, had to be very large, if modulation was not introduced. The range of communication gets limited as the wave cannot travel to a distance without getting distorted. Following are some of the advantages for implementing modulation in the communication systems.

- Antenna size gets reduced.
- No signal mixing occurs.
- Communication range increases.
- Multiplexing of signals occur.
- Adjustments in the bandwidth is allowed.
- Reception quality improves.

Signals in the Modulation Process

Following are the **three types of signals** in the modulation process.

1. Message or Modulating Signal

The signal which contains a message to be transmitted, is called as a message signal. It is a baseband signal, which has to undergo the process of modulation, to get transmitted. Hence, it is also called as the modulating signal.

2. Carrier Signal

The high frequency signal which has a certain phase, frequency, and amplitude but contains no information, is called a carrier signal. It is an empty signal. It is just used to carry the signal to the receiver after modulation.

3. Modulated Signal

The resultant signal after the process of modulation, is called as the modulated signal. This signal is a combination of the modulating signal and the carrier signal.

Types of Modulation

There are many types of modulations. Depending upon the modulation techniques used, they are classified as shown in the following figure.

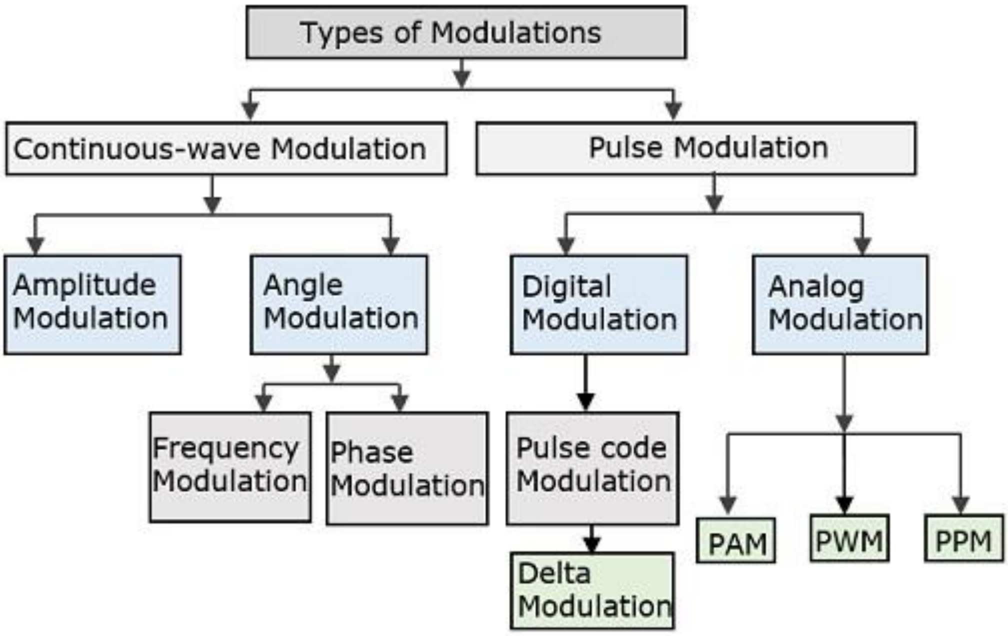

Fig:3 Types of Modulation

The types of modulations are broadly classified into continuous-wave modulation and pulse modulation.

Continuous-wave Modulation

In the continuous-wave modulation, a high frequency sine wave is used as a carrier wave. This is further divided into amplitude and angle modulation.

If the amplitude of the high frequency carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, then such a technique is called as **Amplitude Modulation.** If the angle of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Angle Modulation**.

The angle modulation is further divided into frequency and phase modulation. If the frequency of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Frequency Modulation**. If the phase of the high frequency carrier wave is varied in accordance with the instantaneous value of the modulating signal, then such a technique is called as **Phase Modulation**.

Pulse Modulation

In Pulse modulation, a periodic sequence of rectangular pulses, is used as a carrier wave. This is further divided into analog and digital modulation. In analog modulation technique, if the amplitude, duration or position of a pulse is varied in accordance with the instantaneous values of the baseband modulating signal, then such a technique is called as Pulse Amplitude Modulation (PAM) or Pulse Duration/Width Modulation (PDM/PWM), or Pulse Position Modulation (PPM).

In digital modulation, the modulation technique used is Pulse Code Modulation (PCM) where the analog signal is converted into digital form of 1s and 0s. As the resultant is a coded pulse train, this is called as PCM. This is further developed as Delta Modulation (DM), which will be discussed in subsequent chapters. Hence, PCM is a technique where the analog signals are converted into a digital form. In any communication system, during the transmission of the signal, or while receiving the signal, some unwanted signal gets introduced into the communication, making it unpleasant for the receiver, questioning the quality of the communication. Such a disturbance is called as Noise.

Noise?

Noise is an unwanted signal which interferes with the original message signal and corrupts the parameters of the message signal. This alteration in the communication process, leads to the message getting altered. It is most likely to be entered at the channel or the receiver. The noise signal can be understood by taking a look at the following example.

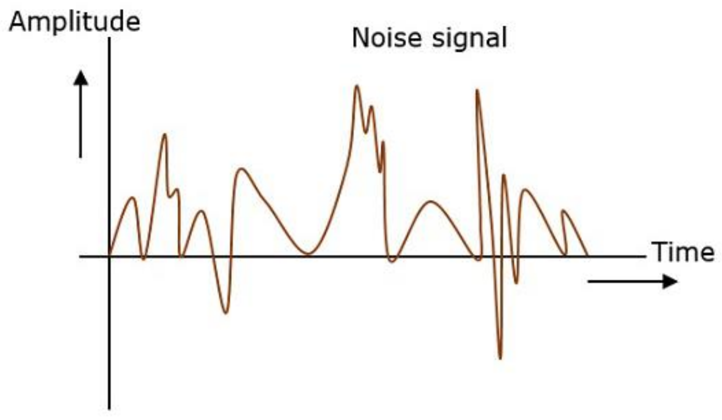

Fig:4 Noise Signal

Hence, it is understood that noise is some signal which has no pattern and no constant frequency or amplitude. It is quite random and unpredictable. Measures are usually taken to reduce it, though it can't be completely eliminated.

Most common examples of noise are −

- **Hiss** sound in radio receivers
- **Buzz** sound amidst of telephone conversations
- **Flicker** in television receivers, etc.

Effects of Noise

Noise is an inconvenient feature which affects the system performance. Following are the effects of noise.

Noise limits the operating range of the systems

Noise indirectly places a limit on the weakest signal that can be amplified by an amplifier. The oscillator in the mixer circuit may limit its frequency because of noise. A system's operation depends on the operation of its circuits. Noise limits the smallest signal that a receiver is capable of processing.

Noise affects the sensitivity of receivers

Sensitivity is the minimum amount of input signal necessary to obtain the specified quality output. Noise affects the sensitivity of a receiver system, which eventually affects the output.

Types of Noise

The classification of noise is done depending on the type of the source, the effect it shows or the relation it has with the receiver, etc. There are two main ways in which noise is produced. One is through some external source while the other is created by an internal source, within the receiver section.

External Source

This noise is produced by the external sources which may occur in the medium or channel of communication, usually. This noise cannot be completely eliminated. The best way is to avoid the noise from affecting the signal.

Examples

- Most common examples of this type of noise are −
- Atmospheric noise (due to irregularities in the atmosphere)
- Extra-terrestrial noise, such as solar noise and cosmic noise.
- Industrial noise.

Internal Source

This noise is produced by the receiver components while functioning. The components in the circuits, due to continuous functioning, may produce few types of noise. This noise is quantifiable. A proper receiver design may lower the effect of this internal noise.

Examples

Most common examples of this type of noise are −

- Thermal agitation noise (Johnson noise or Electrical noise).
- Shot noise (due to the random movement of electrons and holes).
- Transit-time noise (during transition).

Miscellaneous noise is another type of noise which includes flicker, resistance effect and mixer generated noise, etc.

Pulse Amplitude Modulation

Pulse amplitude modulation is the basic form of pulse modulation. In this modulation, the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the modulating signal. Modulation is a process of changing the characteristics of a carrier signal like amplitude, frequency and width, etc. It is the process of adding information to the carrier signal. A carrier signal is a steady waveform with constant amplitude and frequency.

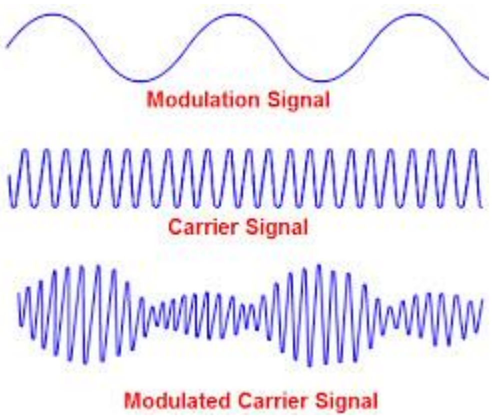

Fig:5 Process of Modulation

Modulation is normally applied to electromagnetic signals like radio laser and optical signals. The Audio, video, images and text data are added to the carrier signal for transmission over telecommunication.

Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling. This technique transmits the data by encoding in the amplitude of a series of signal pulses.

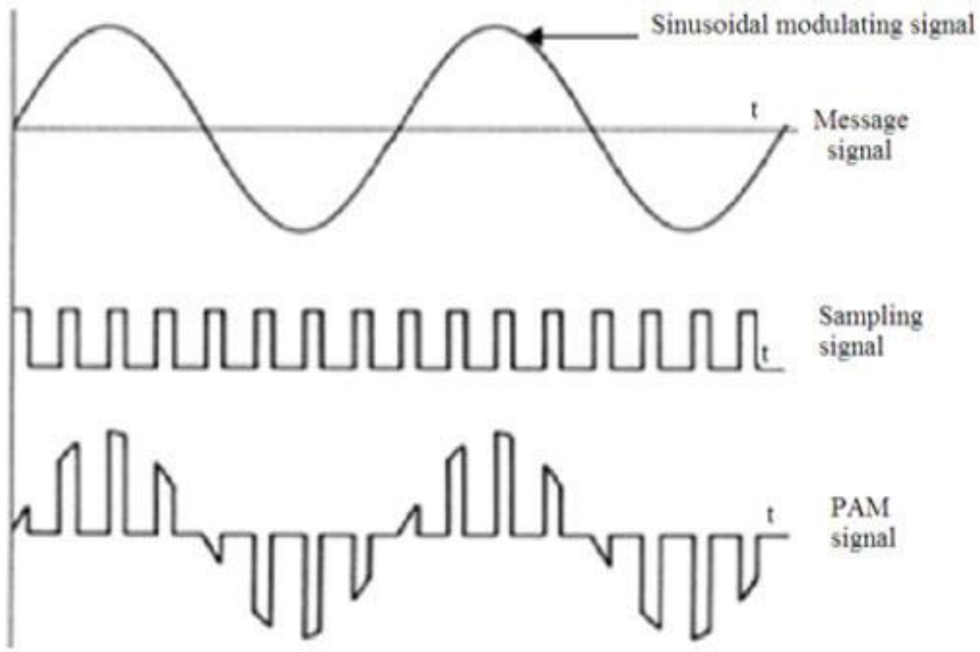

Fig:6 PAM Signal

There are two types of sampling techniques for transmitting a signal using PAM. They are:

- Flat Top PAM
- Natural PAM

Flat Top PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

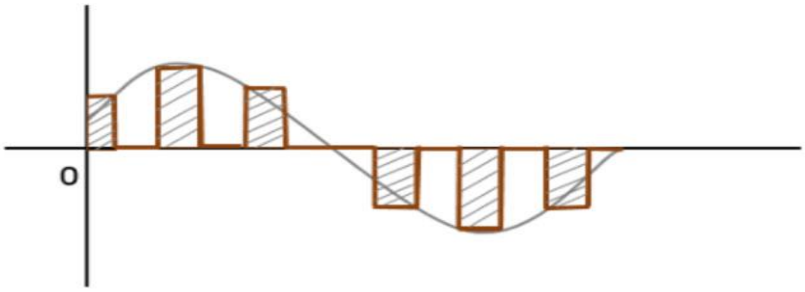

Fig:7 Flat Top PAM

Natural PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. Then follows the amplitude of the pulse for the rest of the half-cycle.

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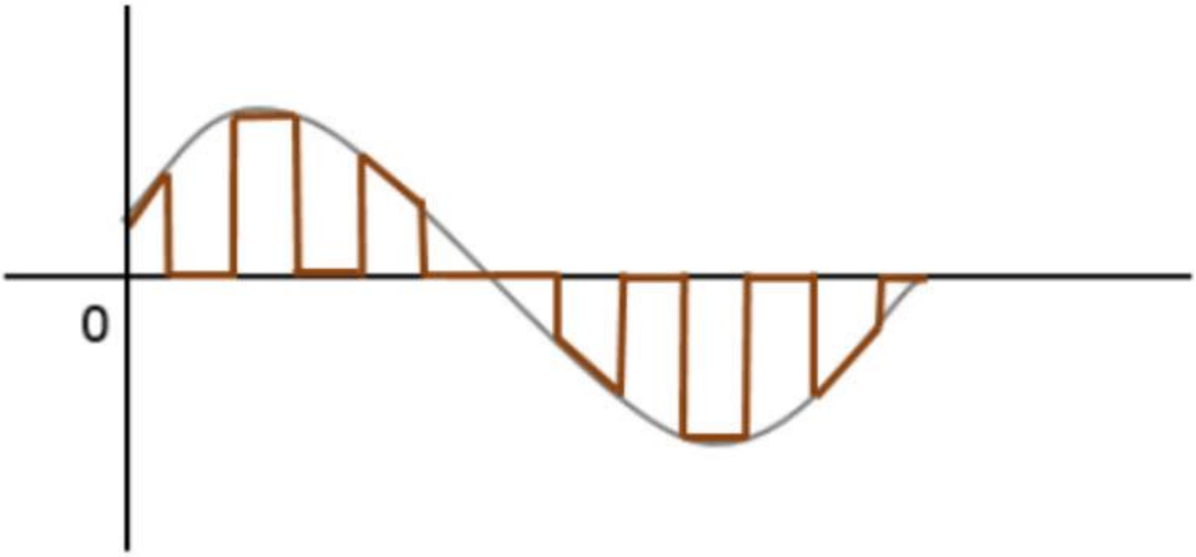


Fig:8 Natural PAM

Demodulation of PAM

For the demodulation of the PAM signal, the PAM signal is fed to the low pass filter. The low pass filter eliminates the high-frequency ripples and generates the demodulated signal. This signal is then applied to the inverting amplifier to amplify its signal level to have the demodulated output with almost equal amplitude with the modulating signal.

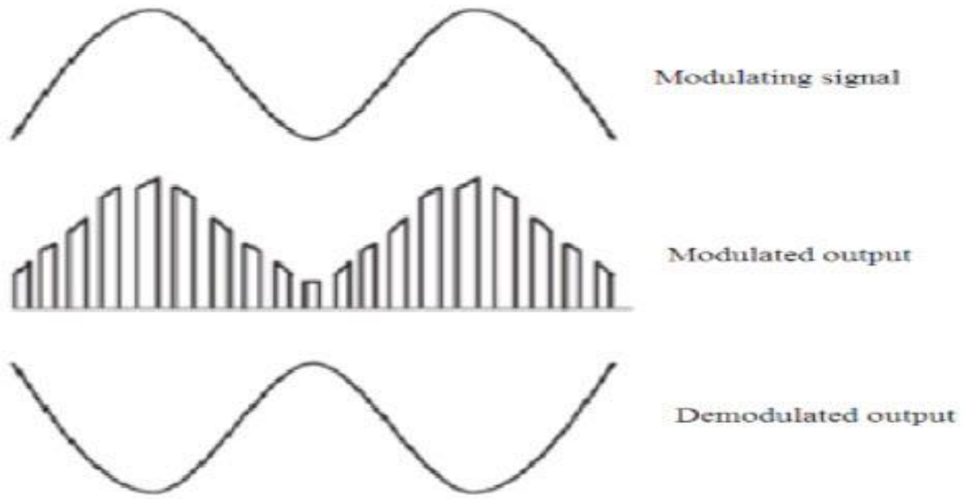


Fig:9 Demodulation of PAM signal

Applications of PAM

It is used in Ethernet communication.

It is used in many micro-controllers for generating the control signals.

It is used in Photo-biology.

It is used as an electronic driver for LED lighting.

Advantages

It is a simple process for both modulation and demodulation.

Transmitter and receiver circuits are simple and easy to construct.

PAM can generate other pulse modulation signals and can carry the message at the same time.

Disadvantages

Bandwidth should be large for transmission PAM modulation.

Noise will be great.

Pulse amplitude signal varies so the power required for transmission will be more.

Pulse Width Modulation

Pulse Width Modulation (PWM) or Pulse Duration Modulation (PDM) or Pulse Time Modulation (PTM) is an analog modulating scheme in which the duration or width or time of the pulse carrier varies proportional to the instantaneous amplitude of the message signal. The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude, to a desired level and hence the noise is limited.

The following figures explain the types of Pulse Width Modulations.

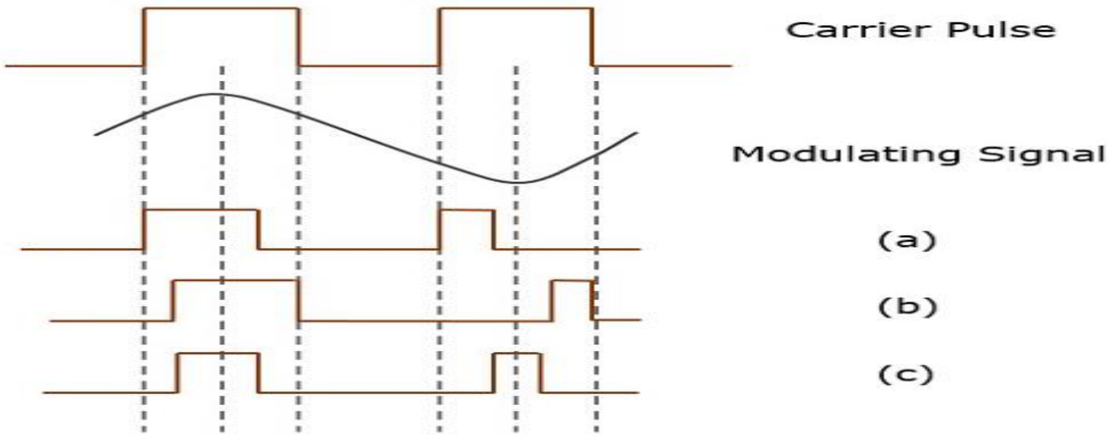


Fig:10 Types of PWM signals

There are three variations of PWM. They are −

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

Pulse Position Modulation

Pulse Position Modulation (PPM) is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal. The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses. The following figures explain the Pulse Position Modulation.

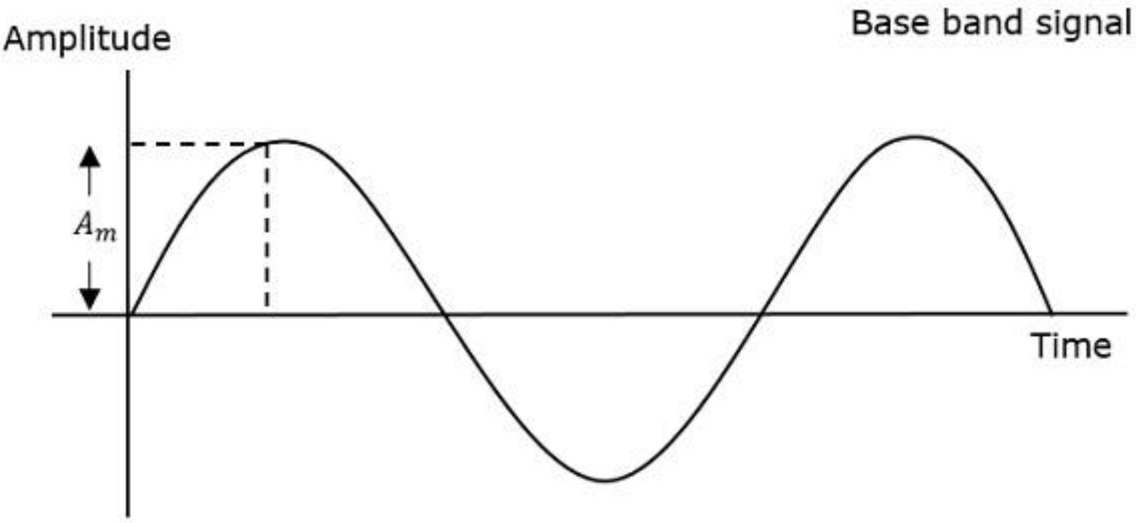
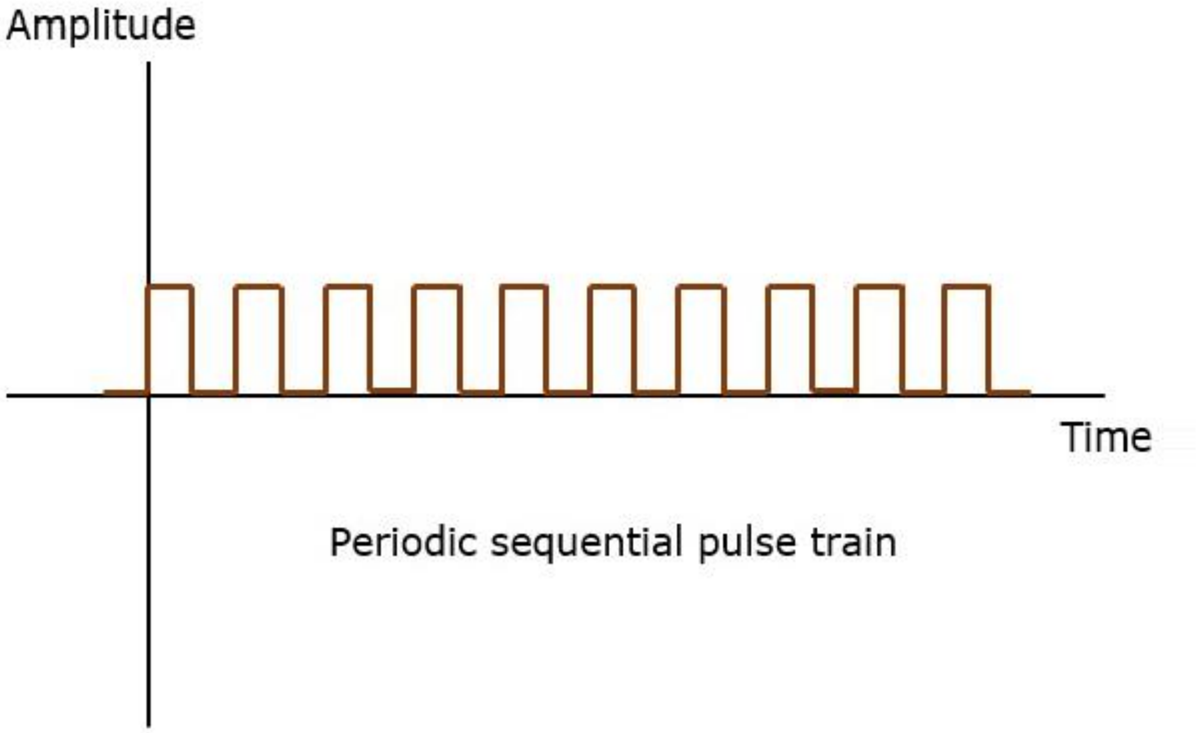


Fig:11 PPM Input signal



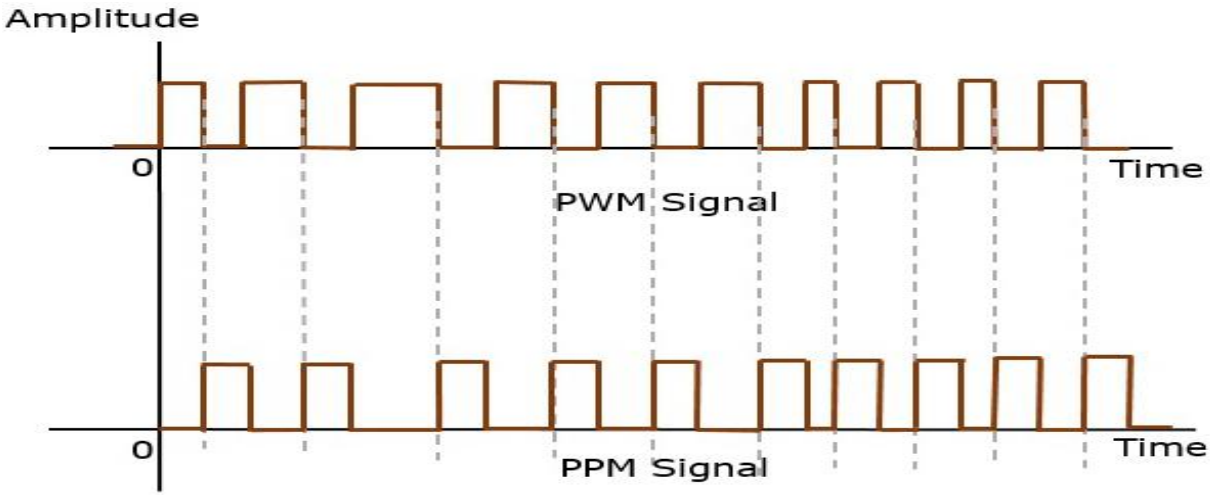


Fig:12 Modulated PWM and PPM signal

Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses.

Advantage

As the amplitude and width are constant, the power handled is also constant.

Disadvantage

The synchronization between transmitter and receiver is a must.

Comparison between PAM, PWM, and PPM

.PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low

Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

Multiplexing is used in the cases where the signals of lower bandwidth and the transmitting media is having higher bandwidth. In this case, the possibility of sending a number of signals is more. In this the signals are combined into one and are sent over a link which has greater bandwidth of media than the communicating nodes.

Time Division Multiplexing (TDM)

This happens when data transmission rate of media is greater than that of the source, and each signal is allotted a definite amount of time. These slots are so small that all transmissions appear to be parallel. In frequency division multiplexing all the signals operate at the same time with different frequencies, but in time division multiplexing all the signals operate with same frequency at different times.

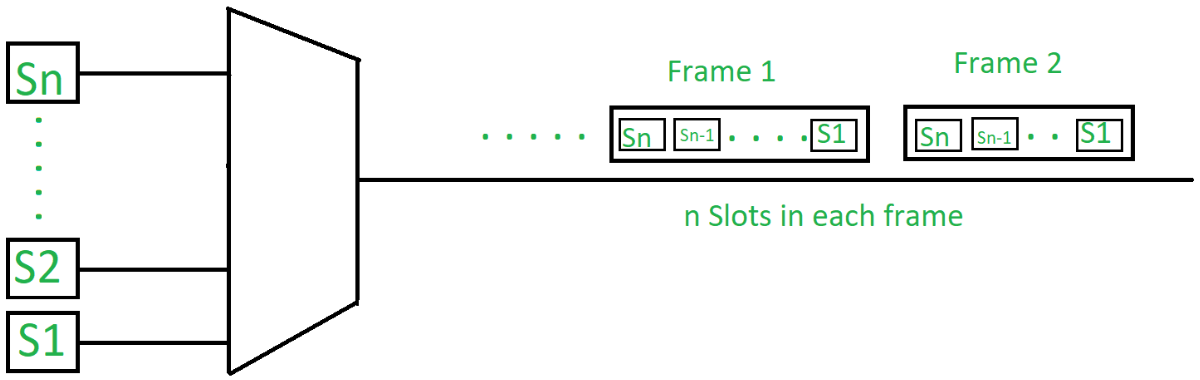


Fig: 13 Time Division Multiplexing

Synchronous TDM

The time slots are pre-assigned and fixed. This slot is even given if the source is not ready with data at this time. In this case the slot is transmitted empty. It is used for multiplexing digitized voice stream.

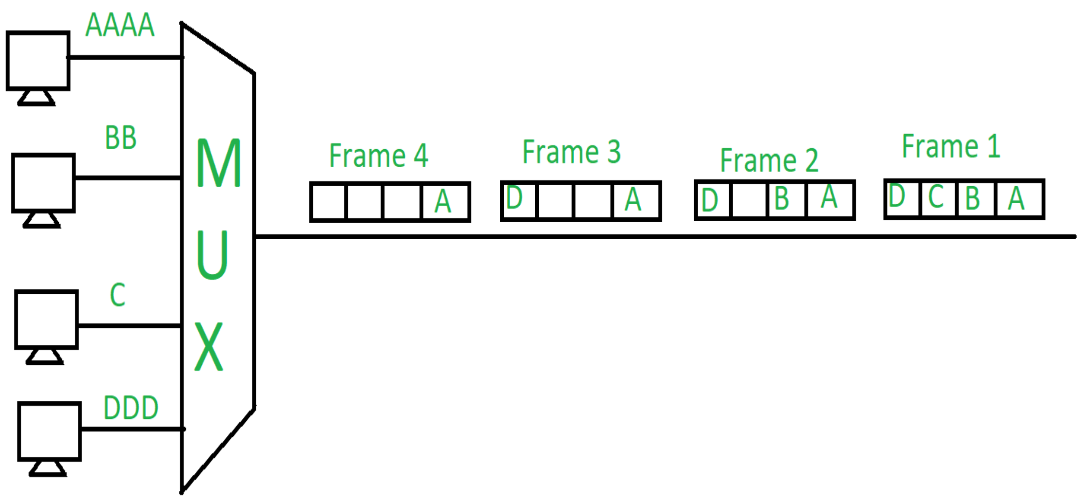


Fig: 14 Synchronous Time Division Multiplexing

Asynchronous (or statistical) TDM

The slots are allocated dynamically depending on the speed of source or their ready state. It dynamically allocates the time slots according to different input channel's needs, thus saving the channel capacity.

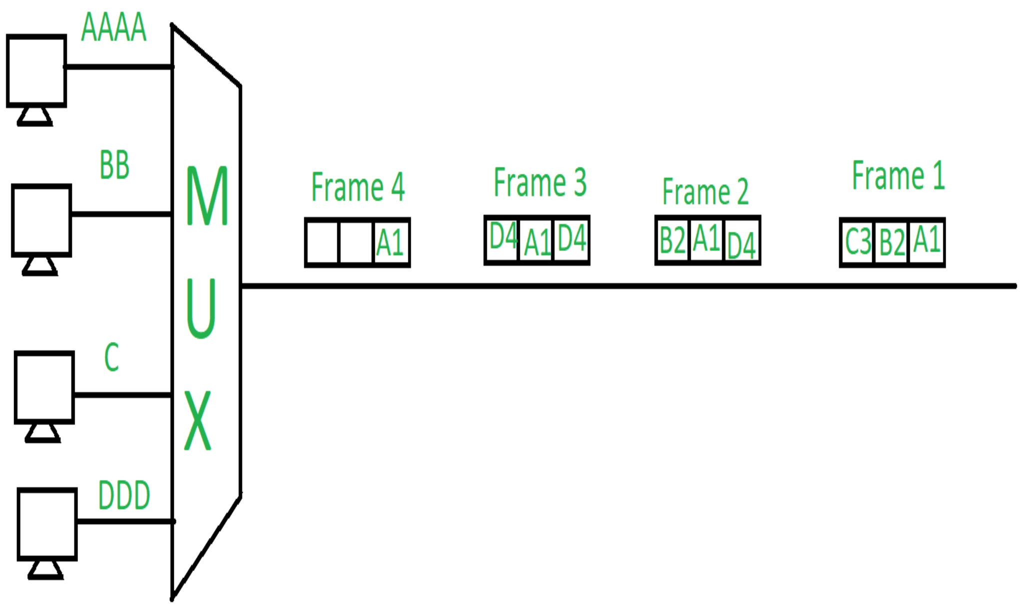


Fig: 15 Asynchronous Time Division Multiplexing

Frequency Division Multiplexing (FDM)

In this a number of signals are transmitted at the same time, and each source transfers its signals in the allotted frequency range. There is a suitable frequency gap between the 2 adjacent signals to avoid over-lapping. Since the signals are transmitted in allotted time so this decreases the probability of collision. The frequency spectrum is divided into several logical channels, in which every user feels that they possess a particular bandwidth. A number of signals are sent simultaneously on the same time allocating separate frequency band or channel to each signal. It is used in radio and TV transmission. Therefore, to avoid interference between two successive channels Guard bands are used.

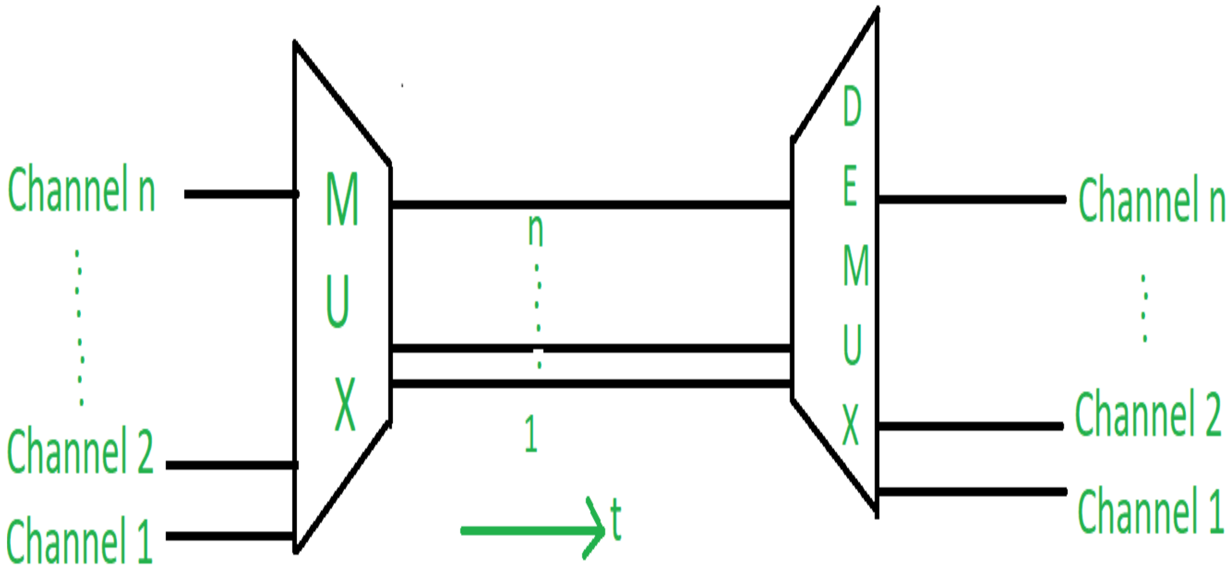


Fig: 16 Frequency division Multiplexing

Pulse Code Modulation

A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.

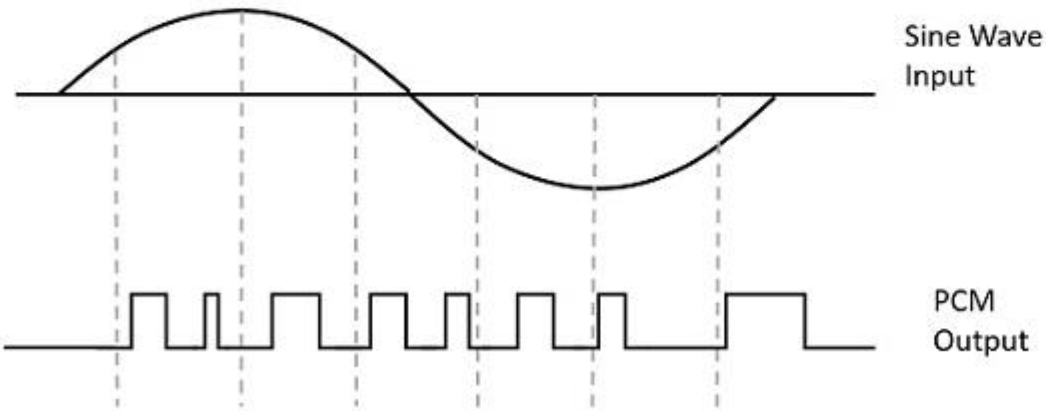


Fig: 17 Output of Pulse Code Modulation

Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant. In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

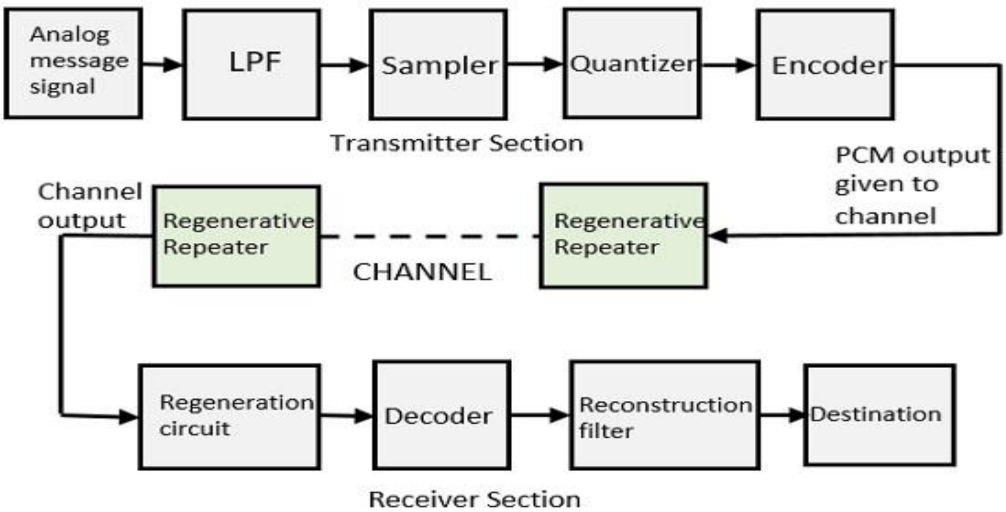


Fig: 18 Block Diagram of PCM

The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.

Elements of PCM

Low Pass Filter (LPF)

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component W of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal. Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Quantization

The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as Quantization.

Quantizing an Analog Signal

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.

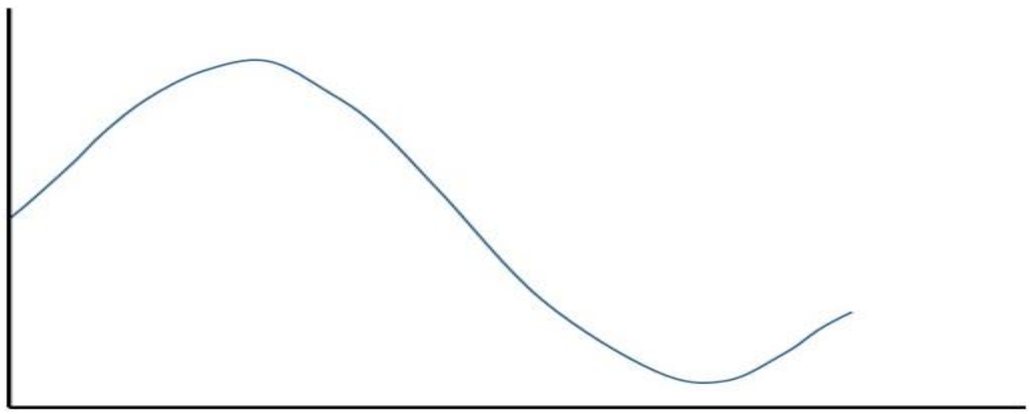


Fig: 19 Analog Signal

The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal. The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.

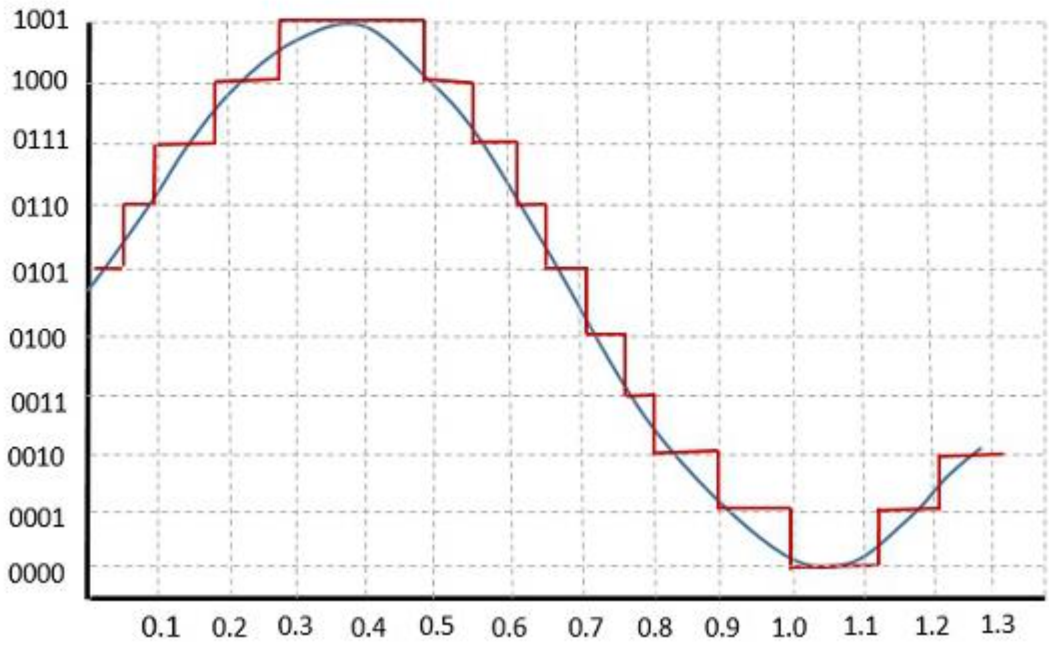


Fig: 20 Quantizing an Analog Signal

Quantization

Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation levels is called a quantum or step-size. The below figure shows the resultant quantized signal which is the digital form for the given analog signal.

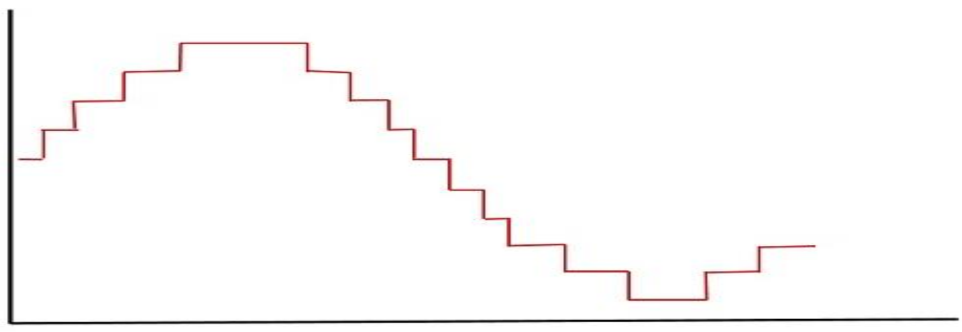


Fig: 21 Stair Case Signal

Resultant Quantized Signal: This is also called as Stair-case waveform, in accordance with its shape.

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

The type of quantization in which the quantization levels are uniformly spaced is termed as a Uniform Quantization. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non-uniform Quantization. There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

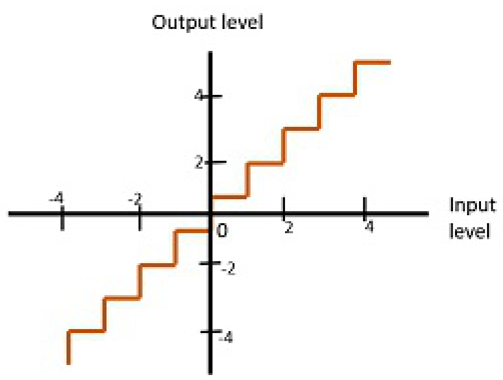


Fig: 22 Mid-rise

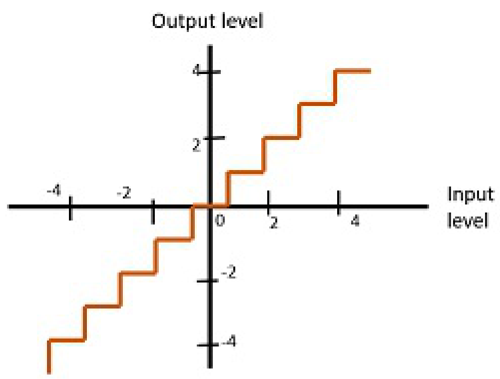


Fig:23 Mid-Tread

Quantization Types

Figure 22 shows the mid-rise type and figure 23 shows the mid-tread type of uniform quantization.

The Mid-Rise type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number. The Mid-tread type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number. Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values. The difference between an input value and its quantized value is called a Quantization

Error. A Quantizer is a logarithmic function that performs Quantization rounding off the value. An analog-to-digital converter (ADC) works as a quantizer.

Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as Quantization Noise.

Companding in PCM

The word Companding is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques. They are

A-law Companding Technique

- Uniform quantization is achieved at A = 1, where the characteristic curve is linear and no compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.

µ-law Companding Technique

- Uniform quantization is achieved at µ = 0, where the characteristic curve is linear and no compression is done.
- µ-law has mid-tread at the origin. Hence, it contains a zero value.
- µ-law companding is used for speech and music signals.
- µ-law is used in North America and Japan.

For the samples that are highly correlated, when encoded by PCM technique, leave redundant information behind. To process this redundant information and to have a better output, it is a wise decision to take a predicted sampled value, assumed from its previous output and summarize them with the quantized values. Such a process is called as **Differential PCM** DPCM technique.

DPCM Transmitter

The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.

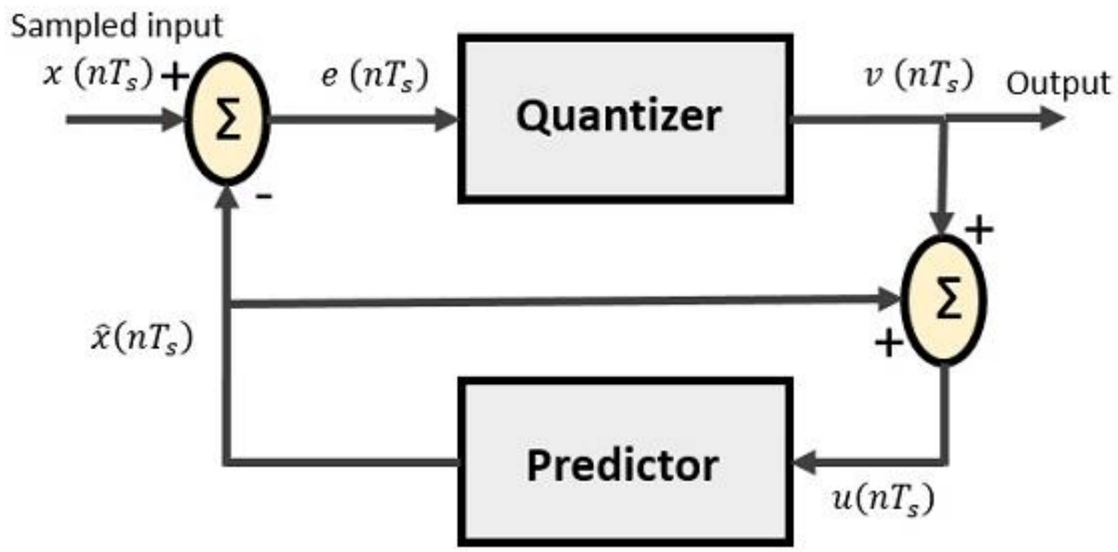


Fig: 24 DPCM Transmitter

The signals at each point are named as −

- x(nTs) is the sampled input
- x^(nTs) is the predicted sample
- e(nTs) is the difference of sampled input and predicted output, often called as prediction error
- v(nTs) is the quantized output
- u(nTs) is the predictor input which is actually the summer output of the predictor output and the quantizer output

The predictor produces the assumed samples from the previous outputs of the transmitter circuit. The input to this predictor is the quantized versions of the input signal x(nTs).

Quantizer Output is represented as

$$v(nTs) = Q[e(nTs)]$$

$$= e(nTs)+q(nTs)$$

Where **q (nT_s)** is the quantization error

Predictor input is the sum of quantizer output and predictor output,

$$u(nT_s) = \hat{x}(nT_s) + v(nT_s)$$

$$u(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$

$$u(nT_s) = x(nT_s) + q(nT_s)$$

The same predictor circuit is used in the decoder to reconstruct the original input.

DPCM Receiver

The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit. Following is the diagram of DPCM Receiver.

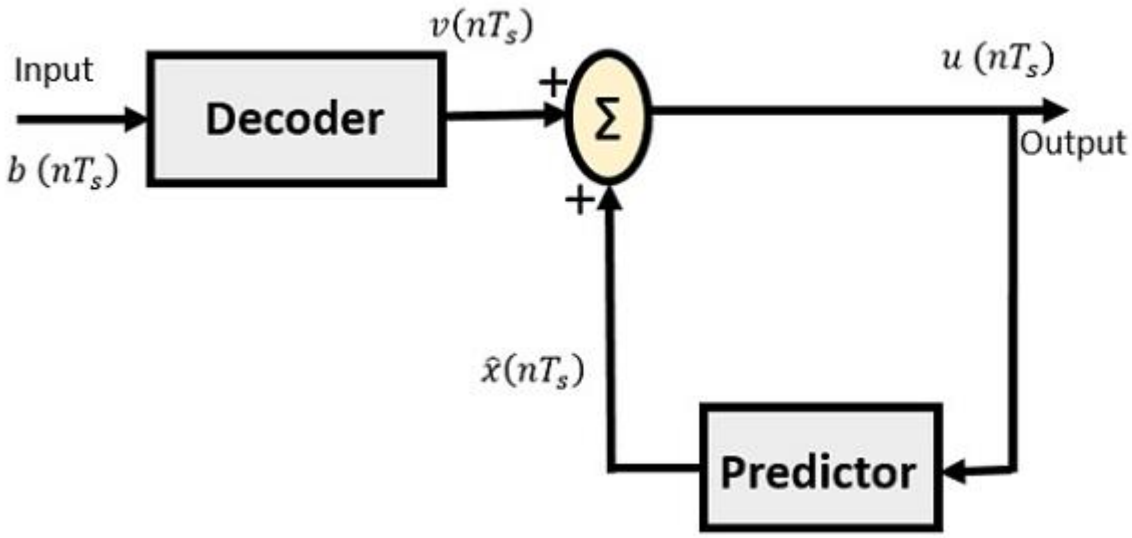


Fig: 25 DPCM Receiver

The notation of the signals is the same as the previous ones. In the absence of noise, the encoded receiver input will be the same as the encoded transmitter output. As mentioned before, the predictor assumes a value, based on the previous outputs. The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtain a better output.

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in Differential PCM is reduced considerably, the sample to-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size will be very small i.e., **Δ** delta.

Delta Modulation

The type of modulation, where the sampling rate is much higher and in which the step size after quantization is of a smaller value **Δ**, such a modulation is termed as **delta modulation**.

Features of Delta Modulation

Following are some of the features of delta modulation.

- An over-sampled input is taken to make full use of the signal correlation.
- The quantization design is simple.
- The input sequence is much higher than the Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., **Δ** delta.
- The bit rate can be decided by the user.
- This involves simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as **1-bit DPCM scheme**. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

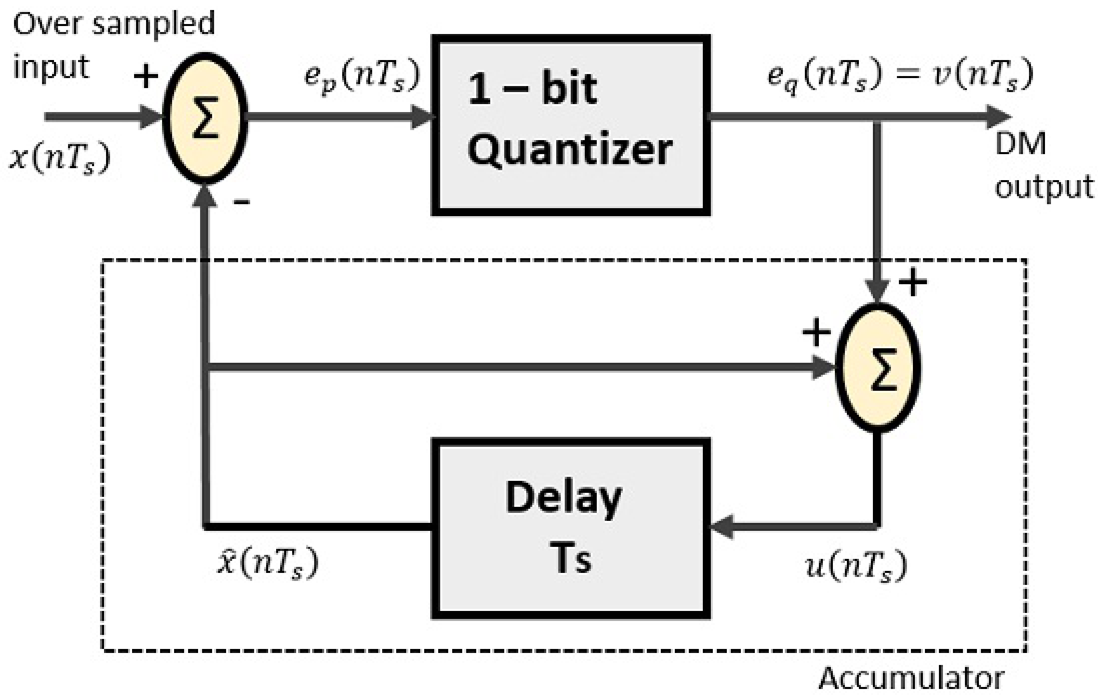


Fig: 26 Delta Modulator

The predictor circuit in DPCM is replaced by a simple delay circuit in DM.

From the above diagram, we have the notations as −

- x(nTs) = over sampled input
- ep(nTs) = summer output and quantizer input
- eq(nTs) = quantizer output = v(nTs)
- x^(nTs) = output of delay circuit
- u(nTs) = input of delay circuit

Using these notations, now we shall try to figure out the process of delta modulation.

ep(nTs)=x(nTs)−x^(nTs) ---------equation 1

=x(nTs)−u([n−1]Ts)

=x(nTs)−[x^[[n−1]Ts]+v[[n−1]Ts]] ---------equation 2

Further,

v(nTs)=eq(nTs)=S.sig.[ep(nTs)]---------equation 3

u(nTs)=x^(nTs)+eq(nTs)

Where,

- $x\hat{}(nTs)$ = the previous value of the delay circuit
- $eq(nTs)$ = quantizer output = $v(nTs)$

Hence,

$u(nTs)=u([n-1]\ Ts)+v(nTs)$ ---------equation 4

Which means,

The present input of the delay unit

=The previous output of the delay unit **+** the present quantizer output

Assuming zero condition of Accumulation,

$$u(nTs) = S\sum_{j=1}^{n} sig[ep(jTs)]$$

$$\textbf{Accumulated version of DM output} = \sum_{j=1}^{n} v(jTs) \text{ ---------equation 5}$$

Now, note that

$$x\hat{}(nTs) = u([n-1]\ Ts)$$

$$= \sum_{j=1}^{n-1} v(jTs) \text{ ---------equation 6}$$

Delay unit output is an Accumulator output lagging by one sample. From equations 5 & 6, we get a possible structure for the demodulator. A Stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

Following is the diagram for delta demodulator.

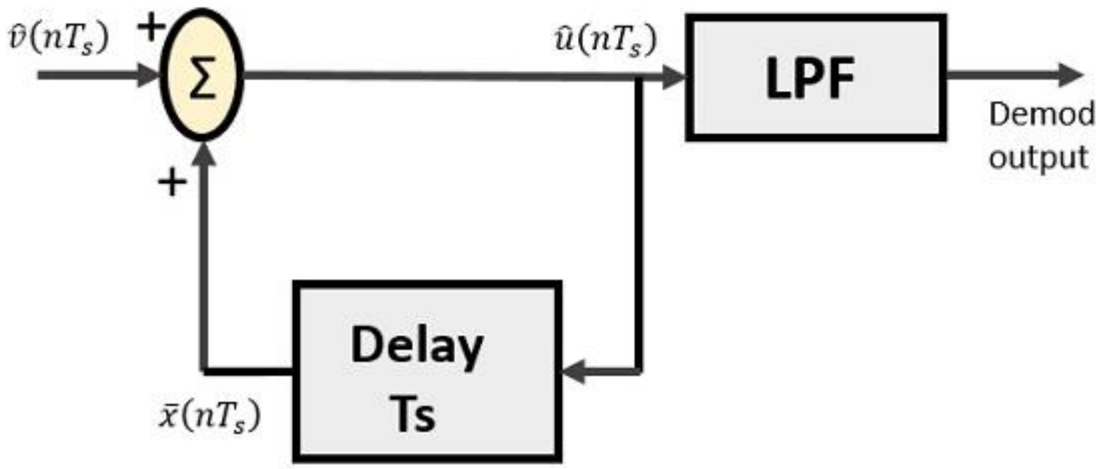


Fig: 26 Delta Demodulator

From the above diagram, we have the notations as −

- v^(nTs) is the input sample
- u^(nTs) is the summer output
- x¯(nTs) is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF. Low pass filter is used for many reasons, but the prominent reason is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages of DM Over DPCM

- 1-bit quantizer
- Very easy design of the modulator and the demodulator

However, there exists some noise in DM.

- Slope Over load distortion (when **Δ** is small)
- Granular noise (when **Δ** is large)

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SCHOOL OF SCIENCE AND HUMANITIES

DEPARTMENT OF PHYSICS

SPHA5304 – COMMUNICATION ELECTRONICS

UNIT III BROAD BAND COMMUNICATION

UNIT 3 BROAD BAND COMMUNICATION

Coaxial cable circuit - Parallel wire line circuit – Computer communication – Digital data communication – Modems – Microwave communication links – LOS links –Tropospheric scatter microwave links – Integrated Service Digital Network (ISDN) – Architecture – Broadband ISDN – Local Area Network (LAN) – LAN topologies – Private Branch Exchange (PBX).

In General Data communications refers to the transmission of this digital data between two or more computers and a computer network or data network is a telecommunications network that allows computers to exchange data. The physical connection between networked computing devices is established using either cable media or wireless media. The best-known computer network is the Internet.

Network Basic Understanding

A system of interconnected computers and computerized peripherals such as printers is called computer network. This interconnection among computers facilitates information sharing among them. Computers may connect to each other by either wired or wireless media.

Network Engineering

Networking engineering is a complicated task, which involves software, firmware, chip level engineering, hardware, and electric pulses. To ease network engineering, the whole networking concept is divided into multiple layers. Each layer is involved in some particular task and is independent of all other layers. But as a whole, almost all networking tasks depend on all of these layers. Layers share data between them and they depend on each other only to take input and send output.

Internet

A network of networks is called an internetwork, or simply the internet. It is the largest network in existence on this planet. The internet hugely connects all WANs and it can have connection to LANs and Home networks. Internet uses TCP/IP protocol suite and uses IP as its addressing protocol. Present day, Internet is widely implemented using IPv4. Because of shortage of address spaces, it is gradually migrating from IPv4 to IPv6.

Internet enables its users to share and access enormous amount of information worldwide. It uses WWW, FTP, email services, audio and video streaming etc. At huge level, internet works on Client-Server model. Internet uses very high-speed backbone of fiber optics. To inter-connect various continents, fibers are laid under sea known to us as submarine communication cable.

Applications of Communication & Computer Network

Computer systems and peripherals are connected to form a network. They provide numerous advantages:

- Resource sharing such as printers and storage devices
- Exchange of information by means of e-Mails and FTP
- Information sharing by using Web or Internet
- Interaction with other users using dynamic web pages
- IP phones
- Video conferences
- Parallel computing
- Instant messaging

DIGITAL DATA COMMUNICATION

The communication that occurs in our day-to-day life is in the form of signals. These signals, such as sound signals, generally, are analog in nature. When the communication needs to be established over a distance, then the analog signals are sent through wire, using different techniques for effective transmission.

The Necessity of Digitization

The conventional methods of communication used analog signals for long distance communications, which suffer from many losses such as distortion, interference, and other losses including security breach. In order to overcome these problems, the signals are digitized using different techniques. The digitized signals allow the communication to be clearer and more accurate without losses. The following figure indicates the difference between analog and digital signals. The digital signals consist of **1s** and **0s** which indicate High and Low values respectively.

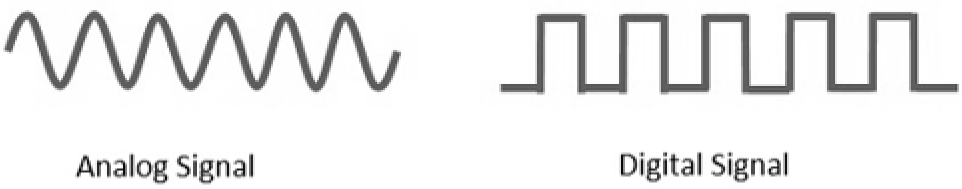

Fig: 1 Representation of Signals

Advantages of Digital Communication

As the signals are digitized, there are many advantages of digital communication over analog communication, such as −

- The effect of distortion, noise, and interference is much less in digital signals as they are less affected.
- Digital circuits are more reliable.
- Digital circuits are easy to design and cheaper than analog circuits.
- The hardware implementation in digital circuits, is more flexible than analog.
- The occurrence of cross-talk is very rare in digital communication.
- The signal is un-altered as the pulse needs a high disturbance to alter its properties, which is very difficult.
- Signal processing functions such as encryption and compression are employed in digital circuits to maintain the secrecy of the information.
- The probability of error occurrence is reduced by employing error detecting and error correcting codes.
- Spread spectrum technique is used to avoid signal jamming.
- Combining digital signals using Time Division Multiplexing TDMTDM is easier than combining analog signals using Frequency Division Multiplexing FDMFDM.
- The configuring process of digital signals is easier than analog signals.
- Digital signals can be saved and retrieved more conveniently than analog signals.
- Many of the digital circuits have almost common encoding techniques and hence similar devices can be used for a number of purposes.

- The capacity of the channel is effectively utilized by digital signals.

Elements of Digital Communication

The elements which form a digital communication system is represented by the following block diagram for the ease of understanding.

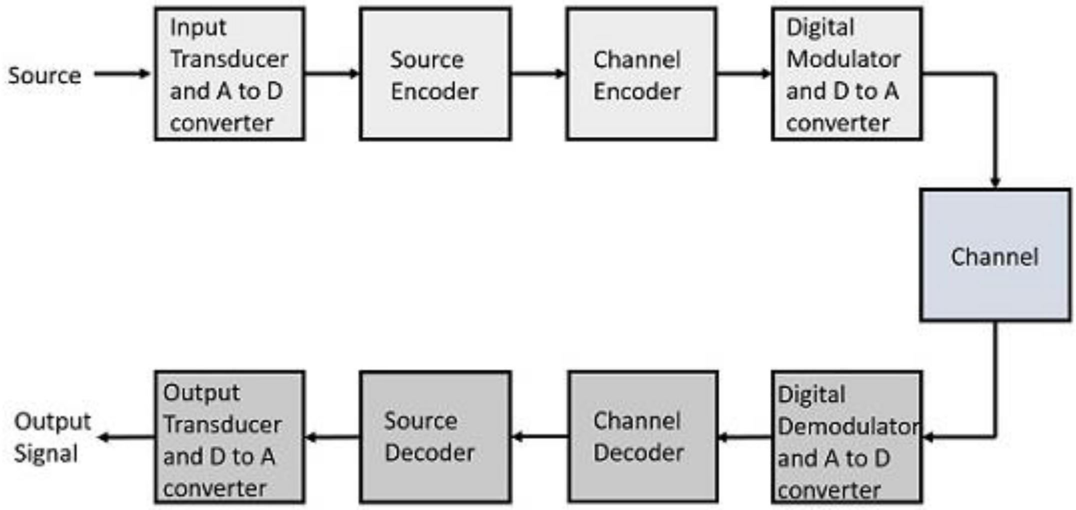

Fig:2 Basic Elements of Digital communication system

Source

The source can be an **analog** signal. **Example**: A Sound signal

Input Transducer

This is a transducer which takes a physical input and converts it to an electrical signal (**Example**: microphone). This block also consists of an **analog to digital** converter where a digital signal is needed for further processes. A digital signal is generally represented by a binary sequence.

Source Encoder

The source encoder compresses the data into minimum number of bits. This process helps in effective utilization of the bandwidth. It removes the redundant bits unnecessary excess bits, i.e., zeroes unnecessary excess bits, i.e., zeroes.

Channel Encoder

The channel encoder, does the coding for error correction. During the transmission of the signal, due to the noise in the channel, the signal may get altered and hence to avoid this, the channel encoder adds some redundant bits to the transmitted data. These are the error correcting bits.

Digital Modulator

The signal to be transmitted is modulated here by a carrier. The signal is also converted to analog from the digital sequence, in order to make it travel through the channel or medium.

Channel

The channel or a medium, allows the analog signal to transmit from the transmitter end to the receiver end.

Digital Demodulator

This is the first step at the receiver end. The received signal is demodulated as well as converted again from analog to digital. The signal gets reconstructed here.

Channel Decoder

The channel decoder, after detecting the sequence, does some error corrections. The distortions which might occur during the transmission, are corrected by adding some redundant bits. This addition of bits helps in the complete recovery of the original signal.

Source Decoder

The resultant signal is once again digitized by sampling and quantizing so that the pure digital output is obtained without the loss of information. The source decoder recreates the source output.

Output Transducer

This is the last block which converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output (**Example**: loud speaker).

Output Signal

This is the output which is produced after the whole process. **Example** − The sound signal received. This unit has dealt with the introduction, the digitization of signals, the advantages and the elements of digital communications. In the coming chapters, we will learn about the concepts of Digital communications, in detail.

What is Modem

A **modem** –"modulator-demodulator" – is a hardware device that converts data from a digital format, intended for communication directly between devices with specialized wiring, into one suitable for a transmission medium such as telephone lines or radio.

A modem modulates one or more carrier wave signals to encode digital information for transmission, and demodulates signals to decode the transmitted information. The goal is to produce a signal that can be transmitted easily and decoded reliably to reproduce the original digital data. Modems can be used with almost any means of transmitting analog signals, from light-emitting diodes to radio. A common type of modem is one that turns the digital data of a computer into a modulated electrical signal for transmission over telephone lines, to be demodulated by another modem at the receiver side to recover the digital data.

History of the modem

The first modem, known as the Dataphone, was released by AT&T in 1960. It later became more common for home users when Dennis Hayes and Dale Heatherington released the 80-103A modem in 1977. Dial-up modems were commonly used by computers to connect to the Internet through the early 2000s until broadband Internet started to be more widely available. As broadband Internet became available and popular, dial-up modems were used by fewer computer users. Today, computers no longer come with a dial-up modem, requiring users who need one to purchase and install it.

Types of Modems

There are three types of modems: cable, digital subscriber line (DSL) and dial-up. A cable modem uses coaxial cables that connect to the back of the modem and the bolt-like outlet in your wall or on your cable box. This type of modem delivers high speed internet to your device. DSL and dial-up modems use a cable that connects to your phone line. DSL, however, still allows you to use your landline telephone while connected to the internet. Fiber-optic technology doesn’t require a modem for its Internet service.

speeds of modems

Modem speed is measured in bps and Kbps, which is the speed the modem can send and receive data. Today, a 56 K (56,000 bps) modem is the fastest solution and speed used with today's dial-up modem. Earlier speeds of modems included 110 baud, 300 baud, 1200 baud, 2400 baud, 4800 baud, 9600 baud, 14.4k, 28.8k, and 33.6k.

Modem features and standards

A modem may also include some or all of the features and specifications listed below.

- Auto-answer - A modem's ability to automatically answer the phone after the phone rings a set amount of time.
- Data/Voice - Modems with voice capability that switch between a voice and data communication.
- Fax - Fax modems can send and receive a fax with the proper software.
- V.90 - The standard that the modem uses also allows it to communicate at an optimal speed. When first introduced, there were multiple standards, but nearly all 56 K modems use the V.90 standard.

Parts Inside a MODEM

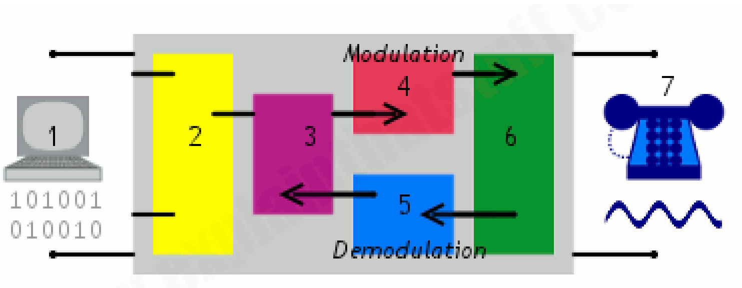

Fig:3 Structure of Modem

A modem is the "interpreter" between a digital computer and the traditionally analog phone network, which is sometimes (especially in technical books) referred to as the PSTN (public switched telephone network). The main components shown here are:

1. Computer or computer network—entirely digital.
2. Computer interface.
3. Modem controller—essentially the modem's independent, central processor.
4. Digital to analog converter (for outgoing, transmitted data)—turns computer data into phone-like analog signals.
5. Analog to digital converter (for incoming, received data)—turns phone signals into digital computer data.
6. Data access arrangement (DAA) makes modem's circuits compatible with the electrical requirements of the phone network.
7. PSTN—the "analog" phone network.

Internal Structure of Modem

Fig:4 Internal View of a MODEM

1. **On/off button**: Spring-loaded switch turns the power on and off.
2. **Capacitors**: Have a variety of jobs to do in a modem, including smooth out current peaks. (See our article on capacitors for more on how they work.)
3. **Volume control**: Controls the loudspeaker volume.
4. **Loudspeaker**: Relays what's happening on the phone line as your modem dials. Read more about loudspeakers.
5. **Modem chip**: Modulates (add digital information to the outgoing telephone signal) and demodulates (separate the digital information from the incoming signal).
6. **Other chips**: Control modem chip and other components.
7. **Microphone**: Allows you to send your own voice down the phone line. Discover how microphones work.
8. **Serial connection**: Connects the modem to your computer's serial (RS-232) port. Newer modems connect to the USB port instead.
9. **Microphone socket**: Connects an external microphone so you can record messages in higher quality than if you use the built-in microphone.
10. **Telephone sockets**: Connect your modem to a phone socket with a standard (RJ11) telephone cable. There's a second socket where you can plug a telephone handset into your modem. This lets you to use your phone through the modem when your computer's not already using the line.

11. **Power input**: Connects the modem to an external power supply unit (electricity transformer) to your modem.

Basic Microwave Communication System/Link

Microwave signal are used for communication over long distance continental or intercontinental. Microwave is the communication link which make the communication possible. The basic block diagram of microwave communication system is shown in below figure.

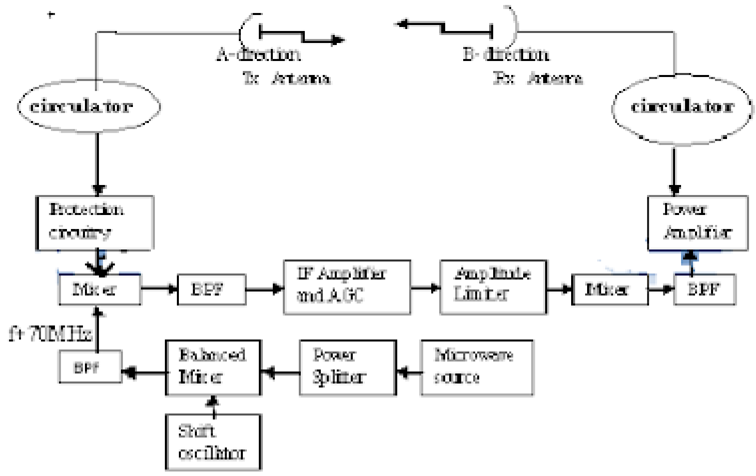

Fig: 5 Block diagram of a Microwave communication Link

Antenna:- Mostly a parabolic refractor types of antenna are used which is used to transmit and receive the signal.

Circulator: A circulator is used to isolate transmitter with the receiver input and to couple transmitter to antenna and antenna to receiver input.

Protection Circuitry: It provides safety to the mixer from overloads.

Mixer (Receiver): It has two outputs. One is the incoming signal and other is the signal from lower band pass filter (BPF). The mixer gives an IF signal of 70Mhz.

Band pass filter (BPF): It provides the necessary selectivity to the receiver and it prevents the interference.

IF amplifier and AGC: - It amplifies the signal up to a intermediate frequency of 70Mhz. and its gain is controlled through AGC (automatic gain control)

Amplitude limiter: As the signal is frequency modulated one so as amplitude limiter is used to avoid unwanted amplitude variations.

Mixer (Transmitter): It is used to convert IF frequency to transmitting microwave frequency band to pass through it and hence prevent interference.

POWER AMPLIFIER: -This amplifier amplifies the transmitted power from a repeater section in the range of 0.2W to 10W.

MICROWAVE SOURCE: - Klystron & Gunn Oscillators were used as microwave source. Now, V H F transistor crystal oscillators are used for microwave source.

POWER SPLITTER: - It divides the output power from a microwave source and feeds a large portion to the transmitter mixer, which converts it into transmitting microwave frequency.

SHIFT OSCILATOR: - It provides one of the inputs to the balanced mixer so that it produces 70MHz IF at the output of receiver mixer.

This microwave link communicates with 600 to 2700 channels per carrier. Thus, the number of carriers in each direction can be four to twelve.

Microwave Link in Electronic Communication:

A Microwave Link in Electronic Communication performs the same functions as a copper or optic fiber cable, but in a different manner, by using point-to-point microwave transmission between repeaters. Many links operate in the 4- and 6-GHz region, but some links operate at frequencies as low as 2 GHz and others at frequencies as high as 13 GHz. Propagation is of course by means of the space wave and therefore limited to line of sight. Typical repeater spacings are close to 50 km, unless a city repeater is located on top of a special tower, Or a country one on a hill. Even then, much larger repeater spacings cannot be used because of the very high attenuation with distance to which radio waves are subject.

A Microwave Link in Electronic Communication terminal has a number of similarities to a coaxial cable terminal. The multiplex equipment will be very similar, if not identical, as will be the channel capacity. Where a cable system uses a number of coaxial cable pairs, a Microwave Link in Electronic Communication will use a number of carriers at various frequencies within the bandwidth allocated to the system. The effect is much the same, and once again a spare carrier is used as a "protection" bearer in case one of the working bearers fails. Finally, there are interconnections at the terminal to other microwave or cable systems, local or-trunk.

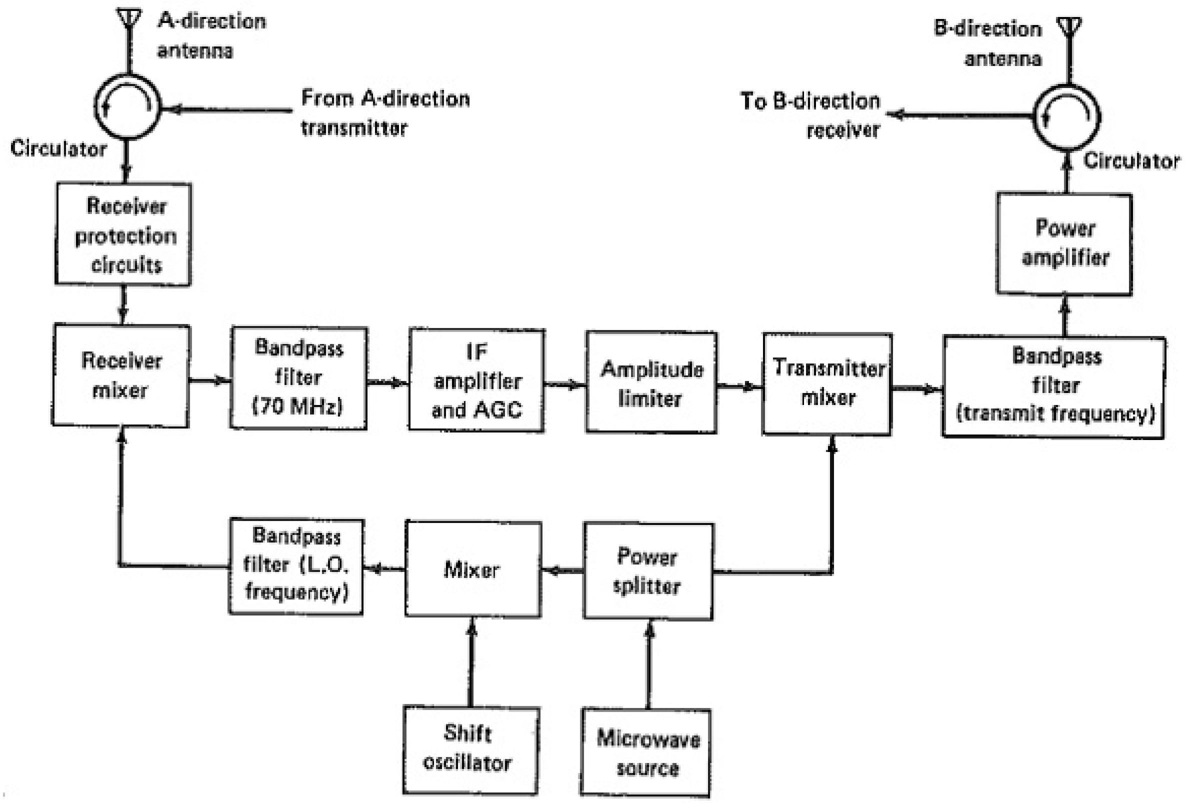

Fig:6 Block diagram of a MW link carrier chain transmitting from A to B

The similarities are in what is done, and the differences lie in the specific detail of how it is done. To illustrate the latter point, the simplified block diagram of a typical microwave repeater is shown in Figure 6 . Essentially, the repeater receives a modulated microwave signal from one repeater and transmits to the next one, and an identical chain is provided for working in the other direction. The only difference here is that the transmissions in the two directions are somewhat different in frequency to avoid interference; the frequency difference is typically a few hundred megahertz at the 4- or 6-GHz operating frequencies.

The block diagram in Figure 6 shows no amplification of the received signal at the radio frequency. Rather, there is conversion down to an IF which is almost invariably 70 MHz, and this is the frequency at which the bulk of amplification takes place in the link shown. Indeed, low-power links have a modulated output oscillator rather than a power output amplifier, and in those links all of the amplification will take place at 70 MHz. The reason for this frequency conversion in existing links is noise reduction: until recently, it has been a lot easier to produce a very low-noise amplifier at 70 MHz than 4 GHz or above. A typical Microwave Link in Electronic Communication consists of several repeaters between the end points, and of course noise is additive for analog systems. The latest developments in microwave transistors have dramatically reduced their noise figures, and so microwave links (especially digital ones) are beginning to appear with RF preamplifiers.

The fact that having a low-noise, sensitive receiver allows the designer to reduce transmit power in proportion; if receiver noise figure can be halved, so can the required link output power. In turn, this allows cost and size reductions in every repeater of what might be a very long chain. The antennas most frequently used are those with parabolic reflectors. Hoghorn antennas are preferred for high-density links, since they are broadband and low-noise. They also lend themselves to so-called frequency reuse, by means of separation of signals through vertical and horizontal polarization. Hoghons are widely used in the very common United States Microwave Link in Electronic Communication in the TD-2C and TD-3C series.

The circulator ensures a connection between the adjoining ports in the direction of the arrow but, not between any other ports. In Figure 6, this means that the transmitter is connected to the antenna and the antenna to the receiver, but the transmitter has no direct connection to the receiver input. If this were not ensured, the receiver mixer would be burned out with remarkable rapidity, The mixer is further safeguarded by protection circuits from overloads caused by any transmission, often but not always generated by transmitters connected to the same antenna.

The receiver mixer is nowadays almost exclusively a Schottky-barrier diode, since this is a very low-noise device. Indeed, other mixer diodes in older systems have generally been replaced through retrofitting with Schottky diodes. The mixer is followed by a bandpass filter, usually operating at 70 MHz and having a bandwidth in the vicinity of 12 MHz. The filter provides the selectivity of the system, ensuring that signals belonging to the other carriers in the system are rejected adequately. The IF amplifier comes next and, as mentioned, provides most of the gain of the repeater. It is almost invariably a low-noise, ultra-linear, very broadband transistor amplifier, which consists of several stages and has AGC applied to it.

The amplitude limiter follows the IF amplifier, to prevent spurious amplitude modulation. In modem links a carrier is injected at this point if the preceding link has failed and no signal is being received. If this were not done, a lot of noise would be transmitted by the link, since AGC would disappear and IF amplifier gain would rise to a maximum.

Varactor diodes are most often used in the transmitter mixer, whose function it is to bring the IF output up to the transmitting microwave frequency. This mixer is followed by a bandpass filter to prevent any straying into unauthorized portions of the frequency spectrum or interference to other carriers in the link.

The output power of a link varies, depending on the bandwidth and therefore the number of circuits per carrier, and on the distance to the next repeater. In most cases powers between 0.25 and 10 W are. transmitted, with 2 to 5 W most common. For powers of 0.5 W or less, a power amplifier is not required, and a power oscillator is used instead. This is most likely to be a reflex klystron in older equipment, a Gunn diode or an IMPATT diode in more modern equipment. The semiconductor devices are preferred for their greater reliability, lower power consumption and simpler power supply requirements. For powers of 1 to 5 W, at frequencies not exceeding 6 GHz, output amplifiers are used, being most commonly push-pull metal-ceramic disk-seal triodes or single-ended TWT amplifiers. Equipment installed during the 1980s is most likely to use FET power amplifiers. For powers in excess of about 5 W, and certainly at frequencies above 6 GHz, traveling-wave tubes are almost universal as power amplifiers. They are then preferred to semiconductor devices because of their much higher available output powers.

The microwave source was a klystron up to the 1960s, and a Gunn oscillator with AFC in the 1970s, but it is nowadays most likely to be a VHF transistor crystal oscillator, with a varactor multiplier. Multiplication factors are of the order of 20 to 40, and the power output is in the vicinity of 200 mW. The power splitter sends approximately 75 percent of the power to the transmitter mixer, and the rest to the mixer which is also fed by the shift oscillator. The function of this circuit is to ensure that the receiver mixer is fed with a frequency 70 MHz higher than the incoming signal, so as to provide the 70-MHz frequency difference for the IF amplifier. This assumes that the receive and transmit frequencies are the same and implies that the receive and transmit frequencies in the A direction in Figure 6 are a few hundred megahertz higher or lower than in the B direction for which the figure is drawn. Some links operate slightly differently, and their receive and transmit frequencies in a given direction are somewhat different. The shift oscillator provides the appropriately different frequency, to ensure still that an IF or 70 MHz is available. The function of the bandpass filter is to remove the unwanted frequencies from the output of the balanced mixer which precedes it.

The typical number of carriers (in each direction) in a Microwave Link in Electronic Communication is at least four, and sometimes as many as 12. There are normally 600 to 2700 channels per carrier. In difficult locations, diversity may be used, in which case it is most likely to be space diversity incorporating pairs of antennas for the same direction. Also, it must be reiterated that the repeaters are not directly involved in the modulation process. This is because they are simply repeaters; their function is to receive, amplify and retransmit. The fact that frequency changing takes place is extraneous to their function and should certainly not be confused with IF amplification in ordinary receivers (where IF amplifiers are followed by demodulators). Modulation does of course take place, as does demodulation, but only at the terminals, not at repeaters.

The towers used for Microwave Link in Electronic Communication range in height up to about 25 m, depending on the terrain, length of that particular link and location of the tower itself. Such link repeaters are unattended, and, unlike coaxial cables where direct current is fed down the cable, repeaters must have their own power supplies. The 200 to 300 W of dc power required by a link is generally provided by a battery. In turn, the power is replenished by a generator, which may be diesel, wind-driven or, in some (especially desert) locations, solar. The antennas themselves are mounted near the top of the tower, a few meters apart in the case of space diversity. They must be accurately aligned to the next repeater in the link, because beamwidths are less than 2°, and any misalignment causes a power loss. Alignment is one of the many items checked at each periodical maintenance visit to a repeater.

It was stated at the beginning of this section that microwave links and coaxial cables perform essentially the same functions. Given that, it may be thought that the two media are in competition. So they are, up to a point, but not to the extent that any one system is likely to oust the other. Basically, microwave links are cheaper and have better properties for TV transmission, although coaxial cable is much less prone to interference. (Coaxial cables are more prone to the kind of industrial interference caused by people using bulldozers and other digging appliances without first checking a map!) The preference for microwave links in transmitting TV programs to distant stations for rebroadcasting is due to the lesser number of repeaters for a given distance, as compared with a coaxial cable. In turn, this reduces the cumulative phase and amplitude distortion over the large bandwidth occupied by TV. On the other hand, a microwave link is far more subject to impulse noise, or “hits,” than the cable, which is protected’ and a closed-circuit system. The overall result of these considerations is that the two media are complementary over the “backbone” routes in most developed countries, although Microwave Link in Electronic Communication predominate over the lesser routes.

Modern Commercial Microwave Links

A microwave link is a communications system that uses a beam of radio waves in the microwave frequency range to transmit video, audio, or data between two locations, which can be from just a few feet or meters to several miles or kilometers apart. Examples of Commercial Microwave links from CableFree may be see here. Modern Microwave Links can carry up to 400Mbps in a 56MHz channel using 256QAM modulation and IP header compression techniques. Operating Distances for microwave links are determined by antenna size (gain), frequency band, and link capacity. The availability of clear Line of Sight is crucial for Microwave links for which the Earth’s curvature has to be allowed. Microwave links are commonly used by television broadcasters to transmit programmes across a country, for instance, or from an outside broadcast back to a studio. Mobile units can be camera mounted, allowing cameras the freedom to move around without trailing cables. These are often seen on the touchlines of sports fields on Steadicam systems.

Planning of microwave links

Cable Free Microwave links have to be planned considering the following parameters:

- Required distance (km/miles) and capacity (Mbps)
- Desired Availability target (%) for the link
- Availability of Clear Line of Sight (LOS) between end nodes
- Towers or masts if required to achieve clear LOS
- Allowed frequency bands specific to region/country
- Environmental constraints, including rain fade
- Cost of licenses for required frequency bands

Microwave Frequency Bands

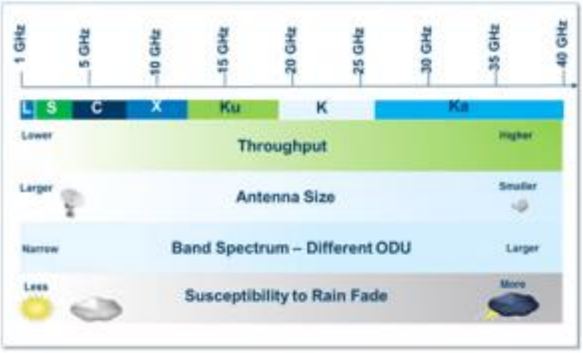

Fig: 7 Microwave Frequency Bands

Microwave signals are often divided into three categories:

ultra high frequency (UHF) (0.3-3 GHz);

super high frequency (SHF) (3-30 GHz); and

extremely high frequency (EHF) (30-300 GHz).

In addition, microwave frequency bands are designated by specific letters.

Designation Frequency range	Microwave frequency bands
L band	1 to 2 GHz
S band	2 to 4 GHz
C band	4 to 8 GHz
X band	8 to 12 GHz
Ku band	12 to 18 GHz
K band	18 to 26.5 GHz
Ka band	26.5 to 40 GHz
Q band	30 to 50 GHz
U band	40 to 60 GHz
V band	50 to 75 GHz
E band	60 to 90 GHz
W band	75 to 110 GHz
F band	90 to 140 GHz
D band	110 to 170 GHz

Lower Microwave frequencies are used for longer links, and regions with higher rain fade. Conversely, Higher frequencies are used for shorter links and regions with lower rain fade.

Uses of microwave links

- Backbone links and “Last Mile” Communication for cellular network operators
- Backbone links for Internet Service Providers (ISPs) and Wireless ISPs (WISPs)
- Corporate Networks for Building to Building and campus sites
- Telecommunications, in linking remote and regional telephone exchanges to larger (main) exchanges without the need for copper/optical fibre lines.
- Broadcast Television with HD-SDI and SMPTE standards.

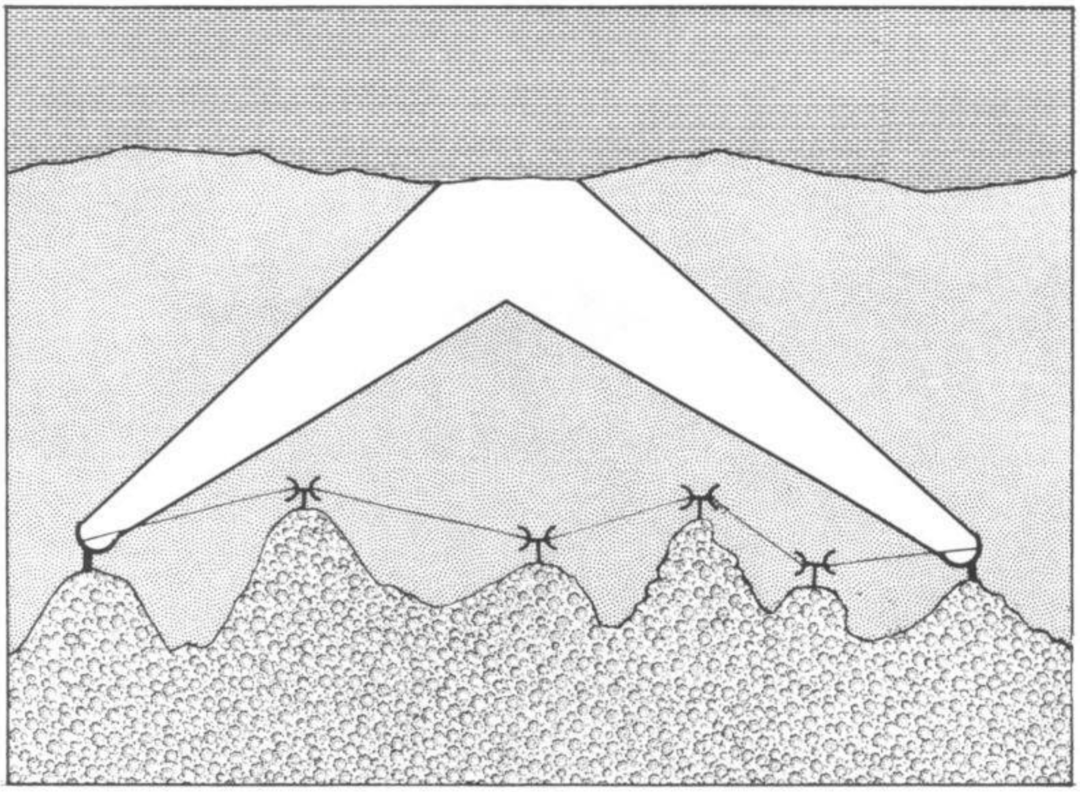

Fig: 8 Tropospheric propagation

Tropospheric scatter (also known as tropo-scatter) is a method of communicating with microwave radio signals over considerable distances – often up to 300 kilo-metres (190 mi), and further depending on terrain and climate factors. This method of propagation uses the tropospheric scatter phenomenon, where radio waves at UHF and SHF frequencies are randomly scattered as they pass through the upper layers of the troposphere. Radio signals are transmitted in a narrow beam aimed just above the horizon in the direction of the receiver station. As the signals pass through the troposphere, some of the energy is scattered back toward the Earth, allowing the receiver station to pick up the signal.

Normally, signals in the microwave frequency range travel in straight lines, and so are limited to line of sight applications, in which the receiver can be 'seen' by the transmitter. Communication distances are limited by the visual horizon to around 30–40 miles (48–64 km).

Tropo-scatter allows microwave communication beyond the horizon. It was developed in the 1950s and used for military communications until communications satellites largely replaced it in the 1970s.

Because the troposphere is turbulent and has a high proportion of moisture the tropospheric scatter radio signals are refracted and consequently only a tiny proportion of the radio energy is collected by the receiving antennas. Frequencies of transmission around 2 GHz are best suited for tropospheric scatter systems as at this frequency the wavelength of the signal interacts well with the moist, turbulent areas of the troposphere, improving signal to noise ratios. Radio waves can propagate over the horizon when the lower atmosphere of the earth bends, scatters, and/or reflects the electromagnetic fields. These effects are collectively known as tropospheric propagation, or tropo for short. Tropospheric propagation can affect wireless communications, sometimes enhancing the usable range, but also compounding interference problems.

The most well-known form of tropo is called bending. Air reduces radio-wave propagation speed compared with the speed in a vacuum. The greater the air density, the more the air slows the waves, and thus the greater is the index of refraction. The density and index of refraction are highest near the surface, and steadily decrease with altitude. This produces a tendency for radio waves at very-high frequencies (VHF, 30 to 300 MHz) and ultra-high frequencies (UHF, 300 MHz to 3 GHz) to be refracted toward the surface. A wave beamed horizontally can follow the curvature of the earth for hundreds of miles.

The lower atmosphere scatters electromagnetic radiation over a vast range, including radio wavelengths. This effect is known as tropospheric scatter, or tropo-scatter. In general, tropo-scatter is most pronounced at UHF and microwave radio frequencies (300MHz and above). A radio wave beamed slightly above the horizon can be scattered at altitudes up to several miles, making over-the-horizon communication possible. The greatest communications range can be realized over flat land or over water. Scattered waves are weak, so high-power transmitters and sensitive receivers are necessary.

A less common, but often dramatic, form of tropo is called ducting or duct effect. This occurs when there is a defined, horizontal boundary between air masses having different densities. When a cool air mass is overlain by a warm air mass, as is the case along and near warm fronts and cold fronts, radio waves at VHF and UHF are reflected at the boundary if they strike it at a near-grazing angle from beneath (within the cooler air mass). Because radio waves are also reflected from the earth's surface, the result can be efficient propagation for hundreds or, in some cases, upwards of 1,000 miles, as the waves alternately bounce off the frontal boundary and the surface. Ducting can allow long-distance radio reception in the frequency-modulation (FM) broadcast band between 88 and 108 MHz. It can also affect the lower VHF television channels if receiving antennas (rather than cable networks) are used.

characteristics of typical tropo-scatter systems:

- Path distances: upwards of 250 km, point to point
- Frequencies: 1800 to 2400 MHz, 4400 to 5000 MHz (primary), and 7100 to 7400 MHz
- Antenna sizes: 2 meter to 9 meter diameter parabolic reflectors, on towers and on trailers
- Frequency diversity: two frequencies (at 1% or more spacing)
- Space diversity: reception on two antennas 100 wavelengths or more apart
- Angle diversity: vertical beam angles separated by approximately one 3 dB antenna beam
- width Dual and quad diversity: most systems are quad diversity. Short links, 100 km or less,
- may be able to use dual diversity. Path availability: typically designed for 99.999% of all hours of the year or better
- Equipment reliability: 50,000 hours to 100,000 hours (6 to 12 years) MTBF or better
- Modulation: QPSK
- Capacity: up to 22 Mb/s, current production modems
- Transmit power: 100 watts to 2,000 watts, employing solid state power amplifiers
- Adaptive link power control: Automatically adjusts transmit power to supply minimum
- needed to maintain link integrity within set parameters. Monitor and control: fully automated with built-in equipment performance monitoring and radio link performance monitoring and logging.
- Service channels: digital service channels for voice order wire and monitoring
- Installations: equipment shelters, trailers, buildings.

Integrated Services Digital Network

ISDN was first defined in the CCITT red book in 1988.The Integrated Services of Digital Networking, in short ISDN is a telephone network based infrastructure that allows the transmission of voice and data simultaneously at a high speed with greater efficiency. This is a circuit switched telephone network system, which also provides access to Packet switched networks.

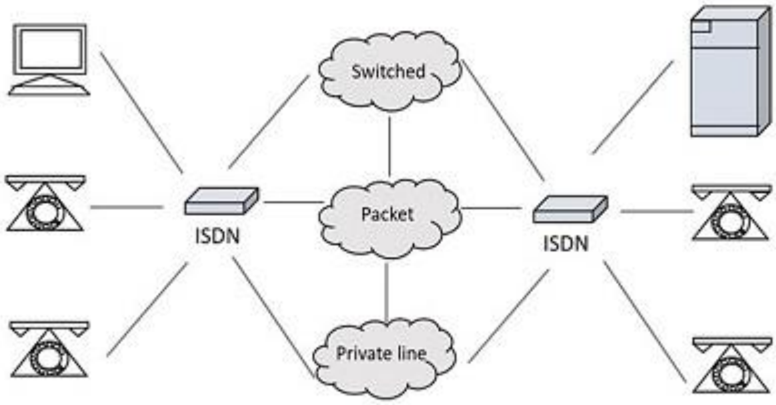

Fig:9 Model of a Practical ISDN

ISDN supports a variety of services. A few of them are listed below −

- Voice calls
- Facsimile
- Videotext
- Teletext
- Electronic Mail
- Database access
- Data transmission and voice
- Connection to internet
- Electronic Fund transfer
- Image and graphics exchange
- Document storage and transfer
- Audio and Video Conferencing
- Automatic alarm services to fire stations, police, medical etc.
-

Types of ISDN

Among the types of several interfaces present, some of them contains channels such as the **B-Channels** or Bearer Channels that are used to transmit voice and data simultaneously; the **D-Channels** or Delta Channels that are used for signaling purpose to set up communication.

The ISDN has several kinds of access interfaces such as −

- Basic Rate Interface (BRI)
- Primary Rate Interface (PRI)
- Narrowband ISDN

- Broadband ISDN

Basic Rate Interface (BRI)

The Basic Rate Interface or Basic Rate Access, simply called the ISDN BRI Connection uses the existing telephone infrastructure. The BRI configuration provides two data or bearer channels at 64 Kbits/sec speed and one control or delta channel at 16 Kbits/sec. This is a standard rate. The ISDN BRI interface is commonly used by smaller organizations or home users or within a local group, limiting a smaller area.

Primary Rate Interface (PRI)

The Primary Rate Interface or Primary Rate Access, simply called the ISDN PRI connection is used by enterprises and offices. The PRI configuration is based on T-carrier or T1 in the US, Canada and Japan countries consisting of 23 data or bearer channels and one control or delta channel, with 64kbps speed for a bandwidth of 1.544 M bits/sec. The PRI configuration is based on E-carrier or E1 in Europe, Australia and few Asian countries consisting of 30 data or bearer channels and two-control or delta channel with 64kbps speed for a bandwidth of 2.048 M bits/sec. The ISDN BRI interface is used by larger organizations or enterprises and for Internet Service Providers.

Narrowband ISDN

The Narrowband Integrated Services Digital Network is called the **N-ISDN**. This can be understood as a telecommunication that carries voice information in a narrow band of frequencies. This is actually an attempt to digitize the analog voice information. This uses 64kbps circuit switching. The narrowband ISDN is implemented to carry voice data, which uses lesser bandwidth, on a limited number of frequencies.

Broadband ISDN

The Broadband Integrated Services Digital Network is called the **B-ISDN**. This integrates the digital networking services and provides digital transmission over ordinary telephone wires, as well as over other media. The CCITT defined it as, "Qualifying a service or system requiring transmission channels capable of supporting rates greater than primary rates." The broadband ISDN speed is around 2 MBPS to 1 GBPS and the transmission is related to ATM, i.e., Asynchronous Transfer Mode. The broadband ISDN communication is usually made using the fiber optic cables.

As the speed is greater than 1.544 Mbps, the communications based on this are called **Broadband Communications**. The broadband services provide a continuous flow of information, which is distributed from a central source to an unlimited number of authorized receivers connected to the network. Though a user can access this flow of information, he cannot control it.

Advantages of ISDN

ISDN is a telephone network based infrastructure, which enables the transmission of both voice and data simultaneously. There are many advantages of ISDN such as −

- As the services are digital, there is less chance for errors.
- The connection is faster.
- The bandwidth is higher.
- Voice, data and video − all of these can be sent over a single ISDN line.

Disadvantages of ISDN

The disadvantage of ISDN is that it requires specialized digital services and is costlier. However, the advent of ISDN has brought great advancement in communications. Multiple transmissions with greater speed are being achieved with higher levels of accuracy.

ISDN specifies a number of reference points that define logical interfaces between functional groups, such as TAs and NT1s. ISDN reference points include the following:

- R—The reference point between non-ISDN equipment and a TA.
- S—The reference point between user terminals and the NT2.
- T—The reference point between NT1 and NT2 devices.
- U—The reference point between NT1 devices and line-termination equipment in the carrier network.

The U reference point is relevant only in North America, where the NT1 function is not provided by the carrier network. Sample ISDN configuration shows three devices attached to an ISDN switch at the central office. Two of these devices are ISDN compatible, so they can be attached through an S reference point to NT2 devices. The third device (a standard, non-ISDN telephone) attaches through the reference point to aTA. Any of these devices also could attach to an NT1/2 device, which would replace both the NT1 and the NT2. In addition, although they are not shown, similar user stations are attached to the far-right ISDN switch.

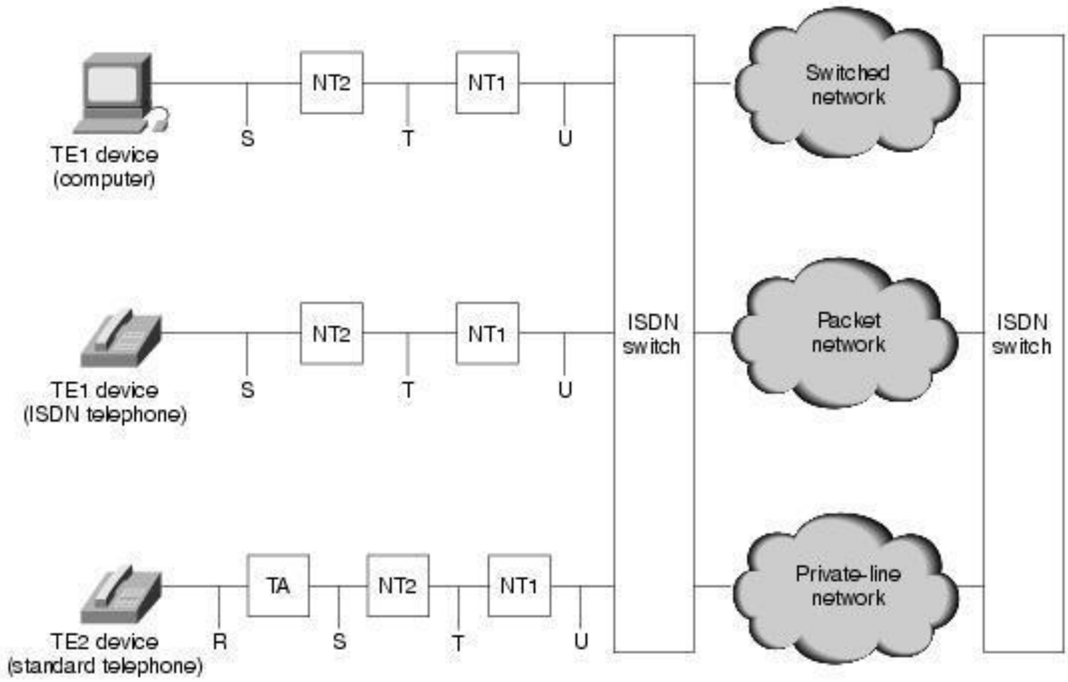

Fig:10 ISDN configuration

ISDN Specifications This section describes the various ISDN specifications for Layer 1, Layer 2, and Layer 3. Layer 1 ISDN physical layer (Layer 1) frame formats differ depending on whether the frame is outbound (from terminal to network) or inbound (from network to terminal). Both physical layer interfaces are shown in Figure.

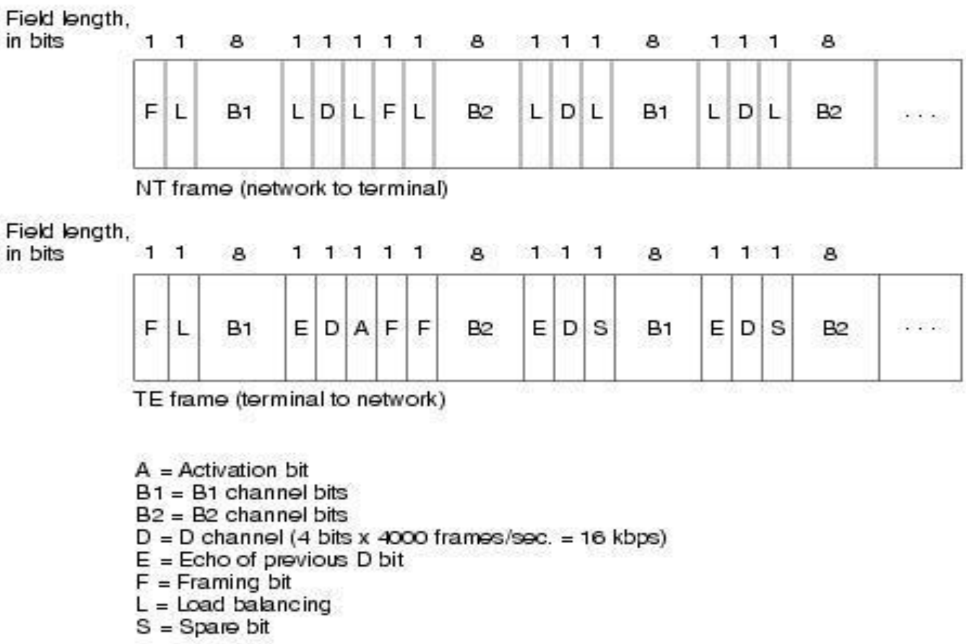

Fig:11 Physical Layer Interface

The frames are 48 bits long, of which 36 bits represent data. The bits of an ISDN physical layer frames are used as follows:

• F—Provides synchronization

• L—Adjusts the average bit value

• E—Ensures contention resolution when several terminals on a passive bus contend for a channel

• A—Activates devices

• S—Is unassigned

• B1, B2, and D—Handle user data

LOCAL AREA NETWORK

A Local Area Network (LAN) is a private network that connects computers and devices within a limited area like a residence, an office, a building or a campus. On a small scale, LANs are used to connect personal computers to printers. However, LANs can also extend to a few kilometers when used by companies, where a large number of computers share a variety of resources like hardware (e.g. printers, scanners, audiovisual devices etc), software (e.g. application programs) and data.

Features of LAN

- Network size is limited to a small geographical area, presently to a few kilometers.
- Data transfer rate is generally high. They range from 100 Mbps to 1000 Mbps.
- In general, a LAN uses only one type of transmission medium, commonly category 5 coaxial cables.
- A LAN is distinguished from other networks by their topologies. The common topologies are bus, ring, mesh, and star.
- The number of computers connected to a LAN is usually restricted. In other words, LANs are limitedly scalable.
- IEEE 802.3 or Ethernet is the most common LAN. They use a wired medium in conjuncture with a switch or a hub. Originally, coaxial cables were used for communications. But now twisted pair cables and fiber optic cables are also used. Ethernet’s speed has increased from 2.9 Mbps to 400 Gbps.

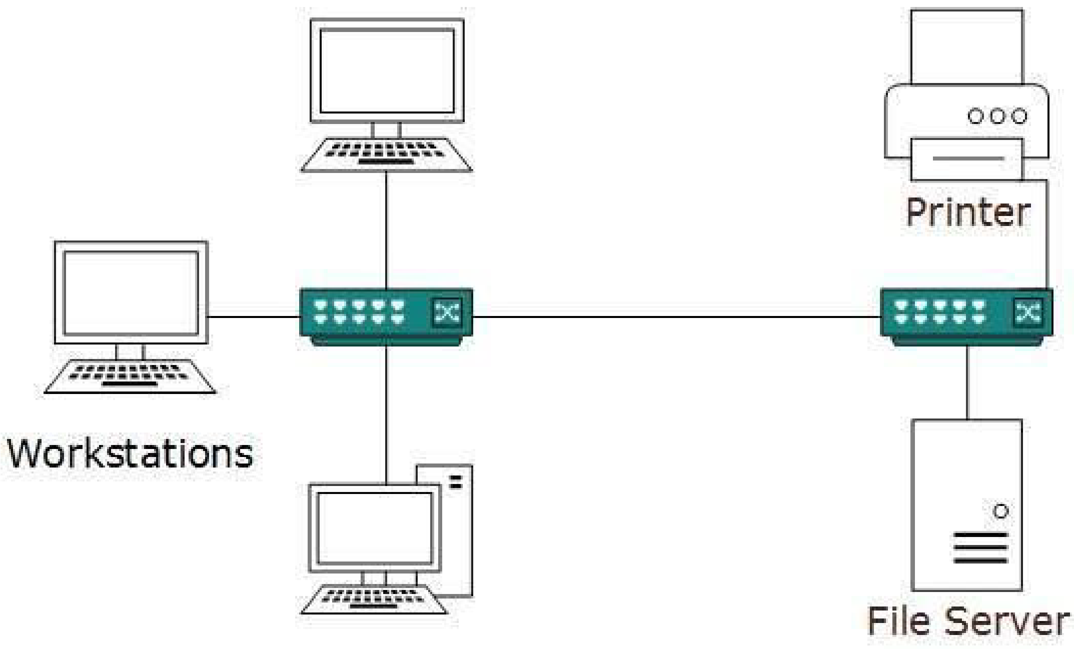

Fig: 12 A Simple LAN Network

Wireless LANs (WLAN)

Wireless LANs use high-frequency radio waves instead of cables for communications. They provide clutter free homes, offices and other networked places. They have an Access Point or a wireless router or a base station for transferring packets to and from the wireless computers and the internet. Most WLANs are based on the standard IEEE 802.11 or Wi-Fi.

Virtual LANs (VLAN)

Virtual LANs are a logical group of computers that appear to be on the same LAN irrespective of the configuration of the underlying physical network. Network administrators partition the networks to match the functional requirements of the VLANs so that each VLAN comprise a subset of ports on a single or multiple switch. This allows computers and devices on a VLAN to communicate in the simulated environment as if it is a separate LAN.

The way in which devices are interconnected to form a network is called network topology. Some of the factors that affect choice of topology for a network are −

- **Cost** − Installation cost is a very important factor in overall cost of setting up an infrastructure. So cable lengths, distance between nodes, location of servers, etc. have to be considered when designing a network.
- **Flexibility** − Topology of a network should be flexible enough to allow reconfiguration of office set up, addition of new nodes and relocation of existing nodes.

- **Reliability** − Network should be designed in such a way that it has minimum down time. Failure of one node or a segment of cabling should not render the whole network useless.
- **Scalability** − Network topology should be scalable, i.e. it can accommodate load of new devices and nodes without perceptible drop in performance.
- **Ease of installation** − Network should be easy to install in terms of hardware, software and technical personnel requirements.
- **Ease of maintenance** − Troubleshooting and maintenance of network should be easy.

Bus Topology

Data network with bus topology has a linear transmission cable, usually coaxial, to which many network devices and workstations are attached along the length. Server is at one end of the bus. When a workstation has to send data, it transmits packets with destination address in its header along the bus.

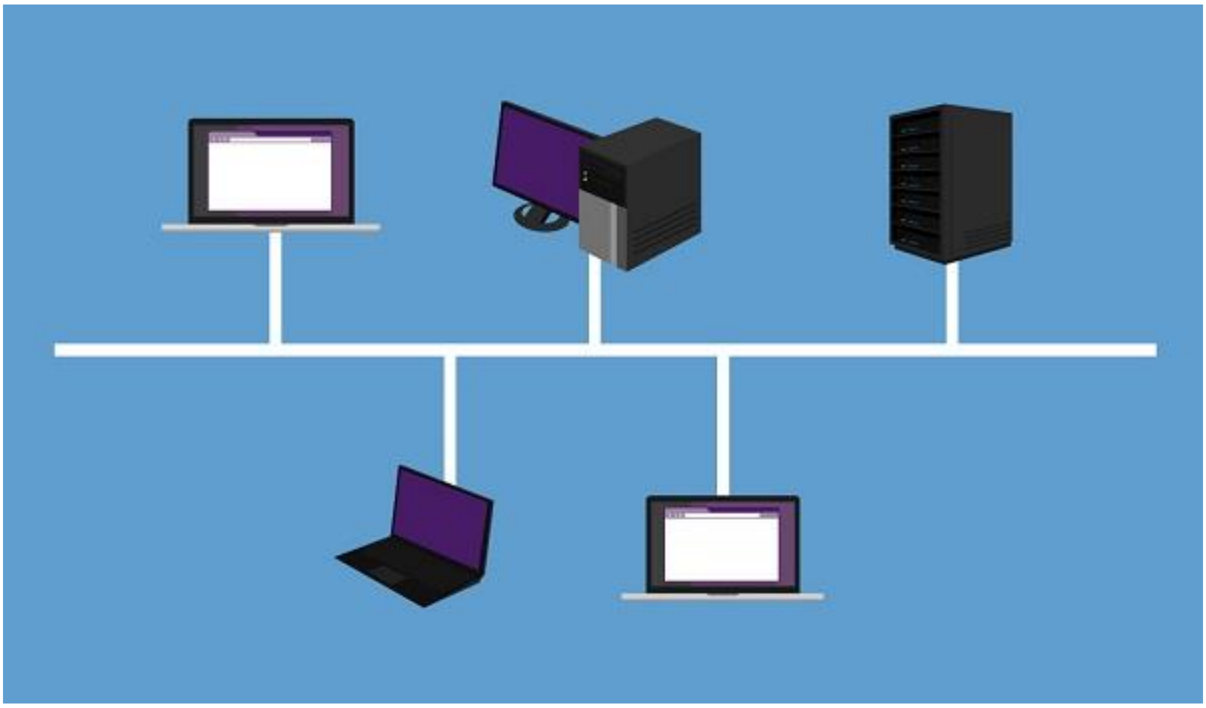

Fig : 13 Bus Topology

The data travels in both the directions along the bus. When the destination terminal sees the data, it copies it to the local disk.

Advantages of Bus Topology

- Easy to install and maintain
- Can be extended easily

- Very reliable because of single transmission line

Disadvantages of Bus Topology

- Troubleshooting is difficult as there is no single point of control
- One faulty node can bring the whole network down
- Dumb terminals cannot be connected to the bus

Ring Topology

In **ring topology** each terminal is connected to exactly **two nodes**, giving the network a circular shape. Data travels in only one pre-determined direction.

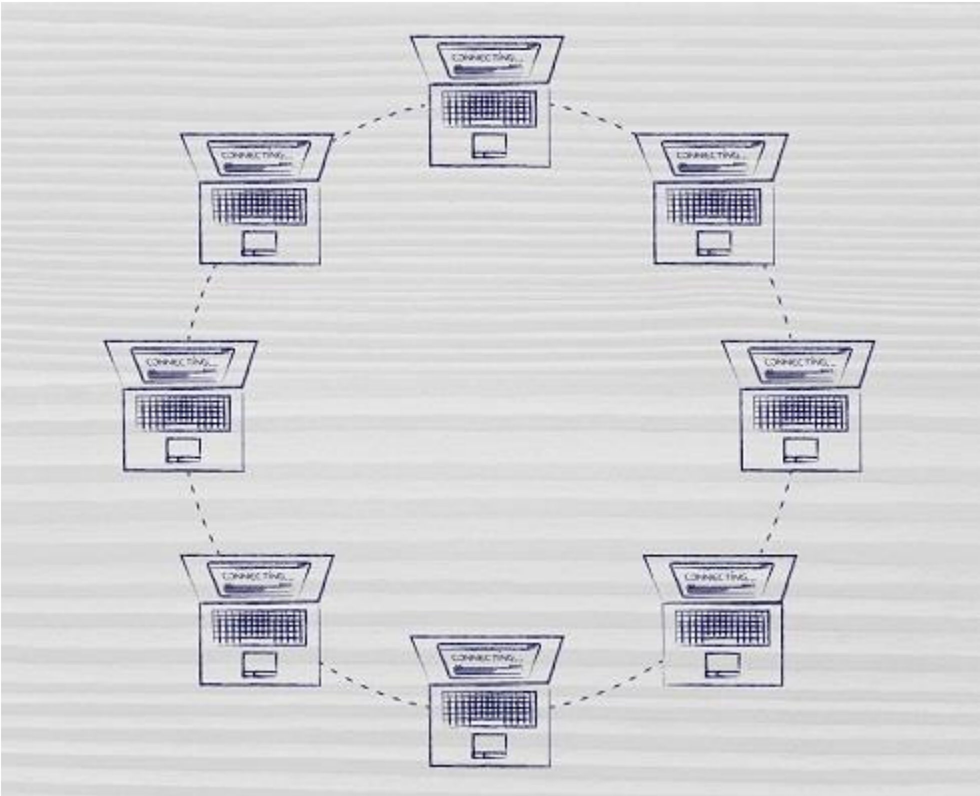

Fig: 14 Ring Topology

When a terminal has to send data, it transmits it to the neighboring node which transmits it to the next one. Before further transmission data may be amplified. In this way, data reverses the network and reaches the destination node, which removes it from the network. If the data reaches the sender, it removes the data and resends it later.

Advantages of Ring Topology

- Small cable segments are needed to connect two nodes
- Ideal for optical fibres as data travels in only one direction
- Very high transmission speeds possible

Disadvantages of Ring Topology

- Failure of single node brings down the whole network
- Troubleshooting is difficult as many nodes may have to be inspected before faulty one is identified
- Difficult to remove one or more nodes while keeping the rest of the network intact

Star Topology

In star topology, server is connected to each node individually. Server is also called the central node. Any exchange of data between two nodes must take place through the server. It is the most popular topology for information and voice networks as central node can process data received from source node before sending it to the destination node.

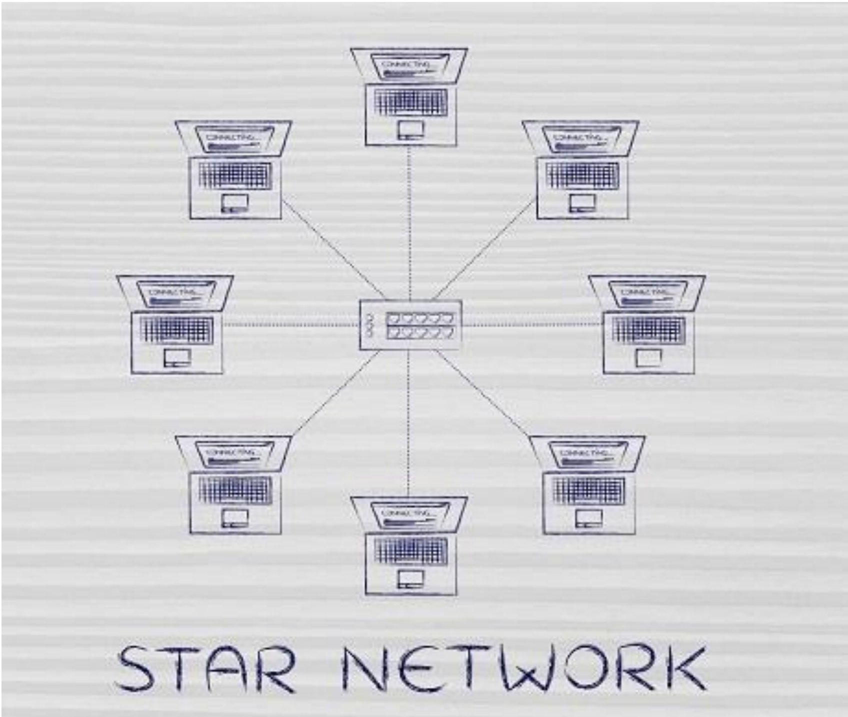

Fig: 15 Star Topology

Advantages of Star Topology

- Failure of one node does not affect the network
- Troubleshooting is easy as faulty node can be detected from central node immediately
- Simple access protocols required as one of the communicating nodes is always the central node

Disadvantages of Star Topology

- Long cables may be required to connect each node to the server
- Failure of central node brings down the whole network

Tree Topology

Tree topology has a group of star networks connected to a linear bus backbone cable. It incorporates features of both star and bus topologies. Tree topology is also called hierarchical topology.

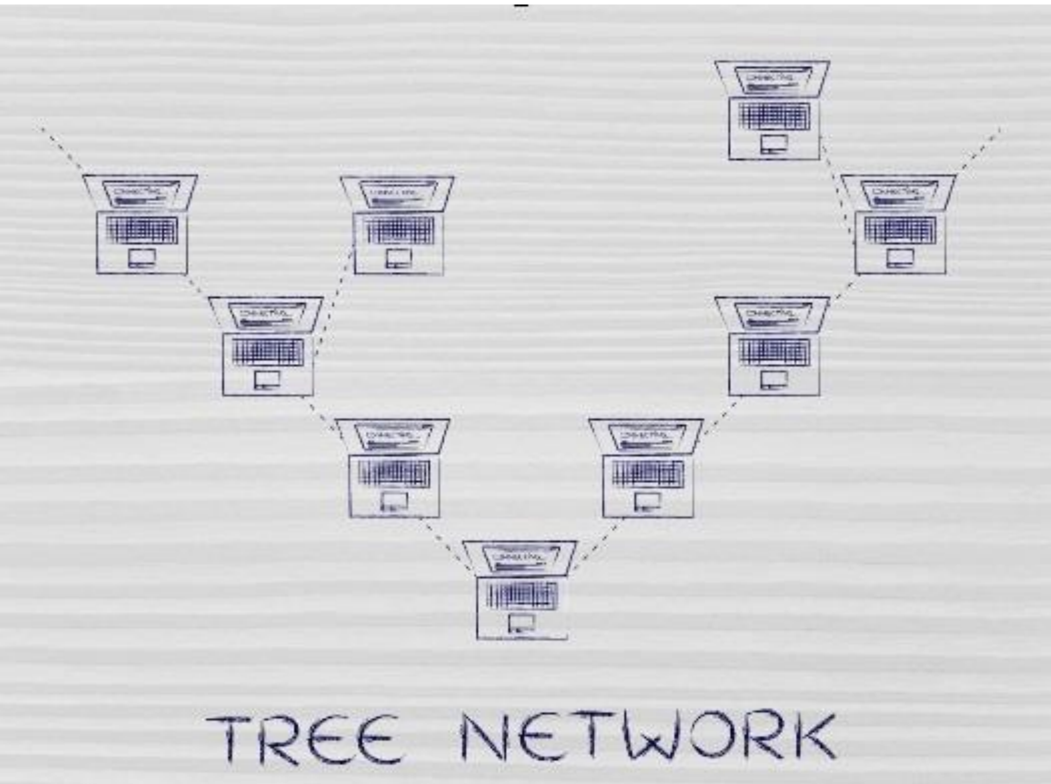

Fig:16 Tree Topology

Advantages of Tree Topology

- Existing network can be easily expanded
- Point-to-point wiring for individual segments means easier installation and maintenance
- Well suited for temporary networks

Disadvantages of Tree Topology

- Technical expertise required to configure and wire tree topology
- Failure of backbone cable brings down entire network
- Insecure network
- Maintenance difficult for large networks

Mesh Topology

In this type of topology, a host is connected to one or multiple hosts.This topology has hosts in point-to-point connection with every other host or may also have hosts which are in point-to-point connection to few hosts only.

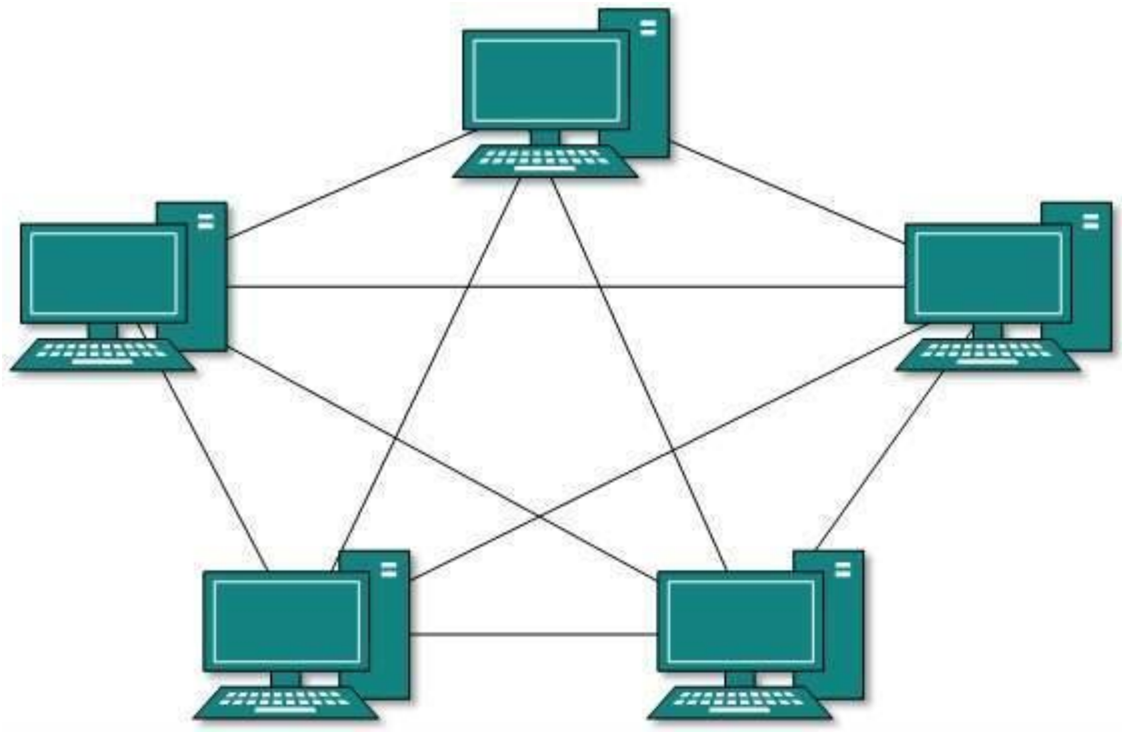

Fig: 17 Mesh Topology

Hosts in Mesh topology also work as relay for other hosts which do not have direct point-to-point links. Mesh technology comes into two types:

- **Full Mesh**: All hosts have a point-to-point connection to every other host in the network. Thus for every new host n(n-1)/2 connections are required. It provides the most reliable network structure among all network topologies.
- **Partially Mesh**: Not all hosts have point-to-point connection to every other host. Hosts connect to each other in some arbitrarily fashion. This topology exists where we need to provide reliability to some hosts out of all.

Hybrid Topology

A network structure whose design contains more than one topology is said to be hybrid topology. Hybrid topology inherits merits and demerits of all the incorporating topologies.

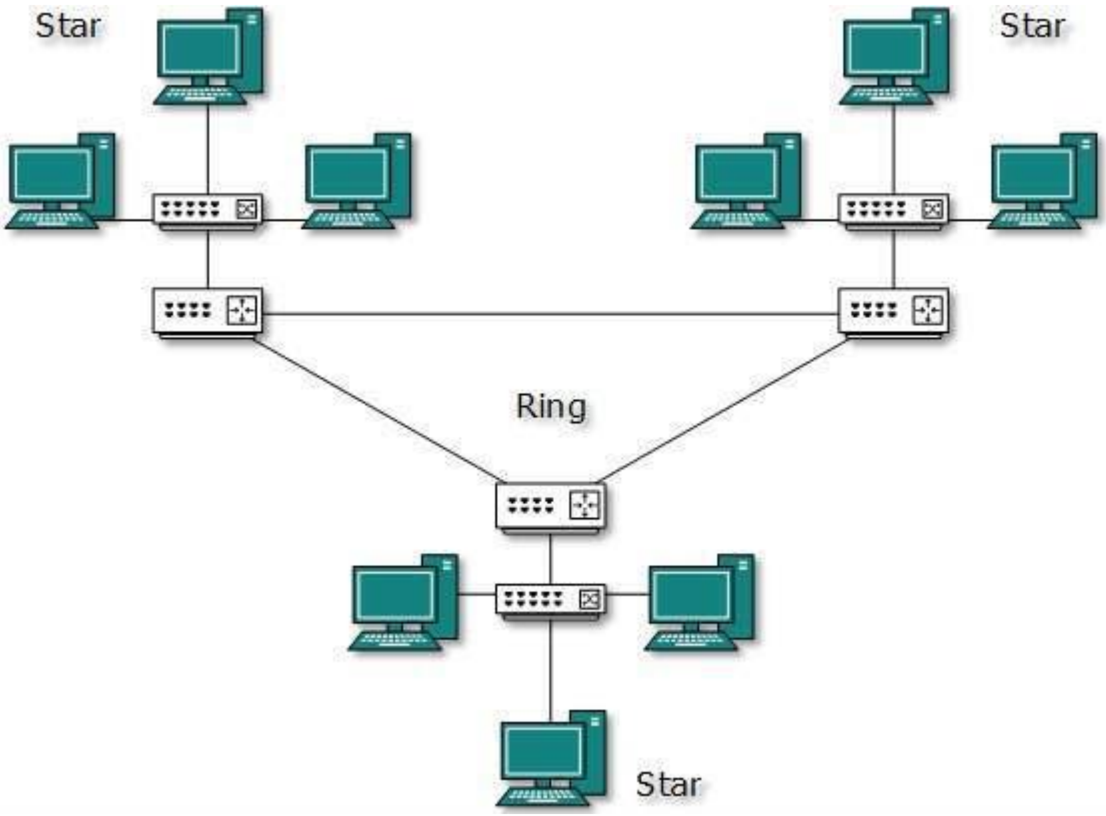

Fig: 18 Hybrid Topology

The above figure 18 represents an arbitrarily hybrid topology. The combining topologies may contain attributes of Star, Ring, Bus, and Daisy-chain topologies. Most WANs are connected by means of Dual-Ring topology and networks connected to them are mostly Star topology networks. Internet is the best example of largest Hybrid topology.

Private Branch Exchange (PBX)

Private Branch Exchange (PBX) is private phone arrange that permits clients to converse with one another. Diverse equipment parts cooperate to give availability to phone arrange. A PBX works as association's inner phone arrangement. PBX system keeps up directing and propelled calling highlights for inbound and outbound calls. Introducing PBX was and is quite difficult. An association helps at least one system administrators with many years of involvement with broadcast communications. You will likewise require physical space to put PBX system in workplace, for example, pantry or worker room. To all more likely value highlights and advantages of business-grade PBX, we should initially discuss telephone system.

A conventional phone system is just old phone system or POTS for short. It depends on curved pair wires from local telephone organization to structure. POTS is essential, dependable and has not changed much in 140 years. Phone organizations interface calls with others utilizing freely exchanged phone arrange (PTSN). PSTN makes it conceivable to fix calls locally to Verizon client AT&T client. Giving business telephone administration is no less. A basic business telephone bill can undoubtedly be in thousands to many lines every month. Hence, there must be superior way.

A PBX permits business to work interior telephone system and utilize fewer telephone lines from telephone organization. THE top PBX system offers alternative of overseeing voice messages, auto orderlies, and recorded messages. It incorporates telephone augmentations for everybody in organization. PBX has upgraded business calling process, which gives noteworthy update to past limitations. Before that, PBX was claimed and worked by troublesome individuals.

Today, PBX systems are essentially improved. Calls are currently made utilizing Voice over Internet Protocol (VoIP) innovation, not at all like for local phone organization. Rather than simple lines, SIP gives availability to small amount of trunking cost. PBX system enables IT, pioneers, to deal with their current gadgets with all-advanced spine. By giving diverse business telephone numbers for various expansions. Then again, cloud consolidates PBX with expansion of telephone system that completely deals with best of two universes.

Types of PBX Phone Systems:

- Hosted PBX
- On-Campus PBX
- PBX Sip Trunking

As indicated by FCC, some phone tricks communicate costly international calls to guiltless representatives utilizing heritage PBX frameworks. PBX running in cloud is expensive in light of fact that its highlights are refreshed consistently and you do not have to introduce arrange foundation. For organizations that cannot be completely overhauled, PBX Sip Trunking is significant option. Inspire your PBX with your equipment with new adaptability and lower communication costs. Organizations that select for hosted PBX or SIP trunking solution can lessen their telecom costs by up to 60%.

Advantages:

- **Scalability** –
Since quantity of cell phones on your system is not constrained to specific pieces of more seasoned call

exchanging gadgets, search for extra associations just to plug another telephone into your broadband association and boot. It interfaces with Internet by means of your PBX in cloud and can be completely initiated right away.

- **Flexibility** –
  Not just is it simple to associate another telephone, however, you can likewise include another telephone anywhere and fundamentally interface with current phone organization. Since all administrations are worked through Internet, there are no limitations on topographical design of your framework. For instance, you may have client care working from base camp in Boston, far off branch workplaces in Florida, and San Francisco – and these areas work proficiently through solitary cloud PBX framework.
- **Mobility** –
  Since worldwide business exercises have changed, individuals have voyaged great deal. This is to get call without objective of setting off to your office work area telephone when you are several miles away. Since call directing through administration is for you on Internet, they will naturally check rundown of telephone numbers you give online to expansions and proceed with chase until you make ground approach to your mobile phone.

Disadvantages:

- **Broadband required**

  Access to all cloud-put together administrations depends with respect to your capacity to get to Internet. Facilitated PBX is same. Notwithstanding, quality and dependability of your broadband circuit are significantly more significant when you intend to layer your organizations on voice traffic. The nature of administration equipment can be thought about, at end of day, dependability of association will decide nature of your involvement in voice administrations. Despite fact that it is conceivable to lay voice-over DSL and link modem circuits, we suggest broadband network with higher help level arrangements, for example, T1, Fiber, and Ethernet where accessible.
- **Current system constraints**

  This is anything but major inconvenience, however, can be costly relying upon your present system status. Since PBX is IP based, you need difficult information to organize. This implies base Category 5 change to deal with cabling and different telephones. In event that you need to isolate your voice and information traffic for quality, you additionally need switch equipped for parting these administrations. Likewise, on off chance that you do not as of now have this capacity, you can consider Power driving over Ethernet (PoE). In the event that this extra equipment is absent, it will be remembered for PBX arrangement.

- **More clients, higher MRC**

For predetermined number of clients, PBX might be more affordable than base-based PBX. This level shifts from business to business. We found that organizations somewhere in range of two and twenty clients are commonly acceptable contenders to have PBX administrations, their correspondence needs, and what arrangement they right now have. As this sort of administration is paid for or added to every client seat, month to month repeating expense (MRC) increments with number of clients.

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SCHOOL OF SCIENCE AND HUMANITIES

DEPARTMENT OF PHYSICS

SPHA5304 – COMMUNICATION ELECTRONICS

UNIT IV SATELLITE COMMUNICATION

UNIT 4 SATELLITE COMMUNICATION

Introduction – Communication satellite systems – Transmitting and receiving earth station – Satellite orbits – Satellite frequency bands – Satellite multiple access formats – FDMA – CDMA – Satellite channel, Power flow – Polarization antenna gain – Parabolic dish antenna – Power loss – Rainfall effect – Receiver noise –satellite system power budget: EIRP, received power Carrier to noise ratio, G/T ratio. – Satellite link analysis – Up link – Down link –Cross link – Direct Home TV broadcasting – Satellite transponders.

Introduction

If communication takes place between any two earth stations through a satellite, then it is called as satellite communication. In this communication, electromagnetic waves are used as carrier signals. These signals carry the information such as voice, audio, video or any other data between ground and space and vice-versa. In general terms, a satellite is a smaller object that revolves around a larger object in space. For example, moon is a natural satellite of earth.

Communication refers to the exchange (sharing) of information between two or more entities, through any medium or channel. In other words, it is nothing but sending, receiving and processing of information. If the communication takes place between any two earth stations through a satellite, then it is called as satellite communication. In this communication, electromagnetic waves are used as carrier signals. These signals carry the information such as voice, audio, video or any other data between ground and space and vice-versa.

A satellite is basically a self-contained communications system with the ability to receive signals from Earth and to retransmit those signals back with the use of a transponder—an integrated receiver and transmitter of radio signals. Soviet Union had launched the world's first artificial satellite named, Sputnik 1 in 1957. Nearly after 18 years, India also launched the artificial satellite named, Aryabhata in 1975.

Types of Communication Satellites

There are nine different types of satellites i.e. Communications Satellite, Remote Sensing Satellite, Navigation Satellite, LEO, MEO, HEO, GPS, **GEOs**, Drone Satellite, Ground Satellite, Polar Satellite.

Need of Satellite Communication

The following two kinds of propagation are used earlier for communication up to some distance.

- **Ground wave propagation** − Ground wave propagation is suitable for frequencies up to 30MHz. This method of communication makes use of the troposphere conditions of the earth.
- **Sky wave propagation** − The suitable bandwidth for this type of communication is broadly between 30–40 MHz and it makes use of the ionosphere properties of the earth.

The maximum hop or the station distance is limited to 1500KM only in both ground wave propagation and sky wave propagation. Satellite communication overcomes this limitation. In this method, satellites provide communication for long distances, which is well beyond the line of sight.

Since the satellites locate at certain height above earth, the communication takes place between any two earth stations easily via satellite. So, it overcomes the limitation of communication between two earth stations due to earth's curvature.

Working of a Satellite

A satellite is a body that moves around another body in a particular path. A communication satellite is nothing but a microwave repeater station in space. It is helpful in telecommunications, radio and television along with internet applications.

A repeater is a circuit, which increases the strength of the received signal and then transmits it. But this repeater works as a transponder. That means, it changes the frequency band of the transmitted signal from the received one. The frequency with which, the signal is sent into the space is called as Uplink frequency. Similarly, the frequency with which, the signal is sent by the transponder is called as Downlink frequency. The following figure illustrates this concept clearly.

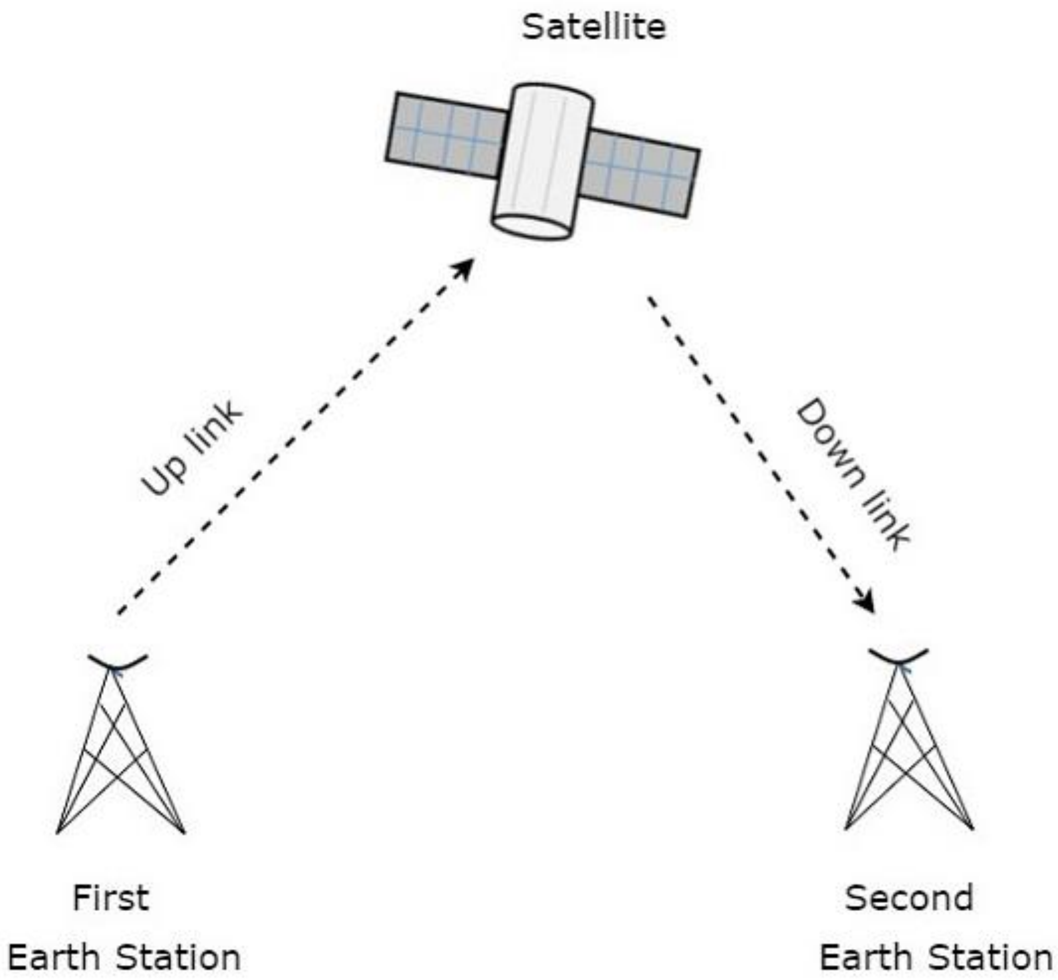

Fig:1 Representation of Satellite Communication

The transmission of signal from first earth station to satellite through a channel is called as uplink. Similarly, the transmission of signal from satellite to second earth station through a channel is called as downlink.

Uplink frequency is the frequency at which, the first earth station is communicating with satellite. The satellite transponder converts this signal into another frequency and sends it down to the second earth station. This frequency is called as Downlink frequency. In similar way, second earth station can also communicate with the first one.

The process of satellite communication begins at an earth station. Here, an installation is designed to transmit and receive signals from a satellite in an orbit around the earth. Earth stations send the information to satellites in the form of high powered, high frequency (GHz range) signals.

The satellites receive and retransmit the signals back to earth where they are received by other earth stations in the coverage area of the satellite. Satellite's footprint is the area which receives a signal of useful strength from the satellite.

Advantages of Satellite Communication

- Area of coverage is more than that of terrestrial systems
- Each and every corner of the earth can be covered
- Transmission cost is independent of coverage area
- More bandwidth and broadcasting possibilities

Disadvantages of using satellite communication

- Launching of satellites into orbits is a costly process.
- Propagation delay of satellite systems is more than that of conventional terrestrial systems.
- Difficult to provide repairing activities if any problem occurs in a satellite system.
- Free space loss is more
- There can be congestion of frequencies.

Applications of Satellite Communication

Satellite communication plays a vital role in our daily life. Following are the applications of satellite communication −

- Radio broadcasting and voice communications
- TV broadcasting such as Direct to Home (DTH)
- Internet applications such as providing Internet connection for data transfer, GPS applications, Internet surfing, etc.
- Military applications and navigations
- Remote sensing applications
- Weather condition monitoring & Forecasting

ORBIT

The path of satellite revolving around the earth is known as **orbit**. This path can be represented with mathematical notations. Orbital mechanics is the study of the motion of the satellites that are present in orbits. So, we can easily understand the space operations with the knowledge of orbital motion.

Satellite should be properly placed in the corresponding orbit after leaving it in the space. It revolves in a particular way and serves its purpose for scientific, military or commercial. The orbits, which are assigned to satellites with respect to earth are called as Earth Orbits. The satellites present in those orbits are called as Earth Orbit Satellites.

We should choose an orbit properly for a satellite based on the requirement. For example, if the satellite is placed in lower orbit, then it takes less time to travel around the earth and there will be better resolution in an onboard camera. Similarly, if the satellite is placed in higher orbit, then it takes more time to travel around the earth and it covers more earth's surface at one time.

Following are the three important **types of Earth Orbit satellites**

- Geosynchronous Earth Orbit Satellites
- Medium Earth Orbit Satellites
- Low Earth Orbit Satellites

Geosynchronous Earth Orbit Satellites

A Geo-synchronous Earth Orbit (GEO) Satellite is one, which is placed at an altitude of 22,300 miles above the Earth. This orbit is synchronized with a side real day (i.e., 23 hours 56 minutes). This orbit can have inclination and eccentricity.

It may not be circular. This orbit can be tilted at the poles of the earth. But, it appears stationary when observed from the Earth. These satellites are used for satellite Television. The same geo-synchronous orbit, if it is circular and in the plane of equator, then it is called as Geostationary orbit. These Satellites are placed at 35,900kms (same as Geosynchronous) above the Earth's Equator and they keep on rotating with respect to earth's direction (west to east).

The satellites present in these orbits have the angular velocity same as that of earth. Hence, these satellites are considered as stationary with respect to earth since, these are in synchronous with the Earth's rotation. The advantage of Geostationary orbit is that no need to track the antennas in order to find the position of satellites. Geostationary Earth Orbit Satellites are used for weather forecasting, satellite TV, satellite radio and other types of global communications. The following figure shows the difference between Geo-synchronous and Geo-stationary orbits. The axis of rotation indicates the movement of Earth.

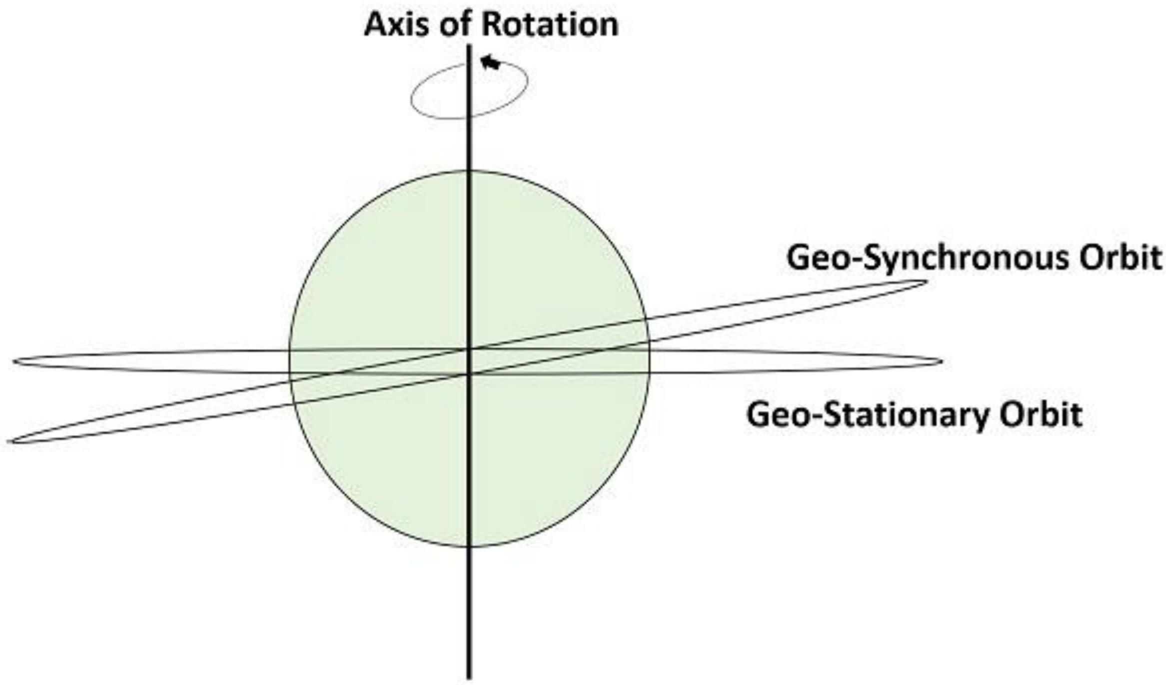

Fig: 2 Difference between Geo-synchronous and Geo-stationary orbits.

Point to Note: Every Geostationary orbit is a Geo-synchronous orbit. But, the converse need not be true.

Medium Earth Orbit Satellites

Medium Earth Orbit (MEO) satellites will orbit at distances of about 8000 miles from earth's surface. Signals transmitted from a MEO satellite travel a shorter distance. Due to this, the signal strength at the receiving end gets improved. This shows that smaller and light weight receiving terminals can be used at the receiving end.

Transmission delay can be defined as the time it takes for a signal to travel up to a satellite and back down to a receiving station. In this case, there is less transmission delay. Because, the signal travels for a shorter distance to and from the MEO satellite.

For real-time communications, the shorter the transmission delay, the better will be the communication system. As an example, if a GEO satellite requires 0.25 seconds for a round trip, then MEO satellite requires less than 0.1 seconds to complete the same trip. MEOs operate in the frequency range of 2 GHz and above. These satellites are used for High speed telephone signals. Ten or more MEO satellites are required in order to cover entire earth.

Low Earth Orbit Satellites

Low Earth Orbit LEO) satellites are mainly classified into three categories. Those are little LEOs, big LEOs, and Mega-LEOs. LEOs will orbit at a distance of 500 to 1000 miles above the earth's surface. These satellites are used for satellite phones and GPS.

This relatively short distance reduces transmission delay to only 0.05 seconds. This further reduces the need for sensitive and bulky receiving equipment. Twenty or more LEO satellites are required to cover entire earth. Little LEOs will operate in the 800 MHz (0.8 GHz) range. Big LEOs will operate in the 2 GHz or above range, and Mega-LEOs operates in the 20-30 GHz range. The higher frequencies associated with Mega-LEOs translates into more information carrying capacity and yields to the capability of real-time, low delay video transmission scheme.

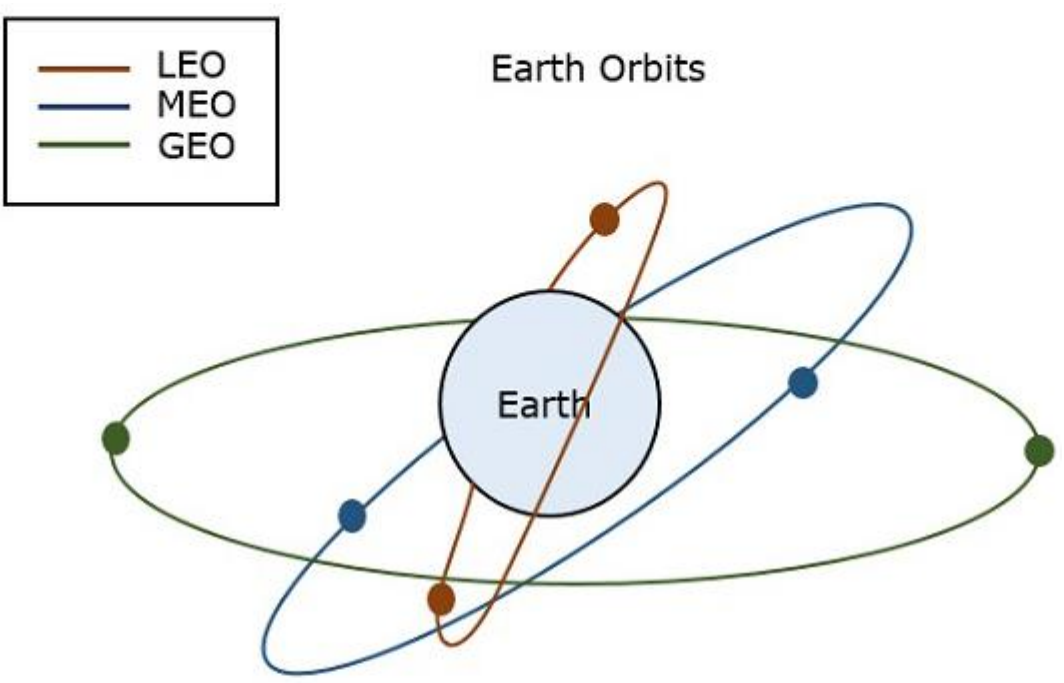

Fig:3 Depicts the paths of LEO, MEO and GEO

Orbital Elements

Orbital elements are the parameters, which are helpful for describing the orbital motion of satellites. Following are the **orbital elements**.

- Semi major axis
- Eccentricity
- Mean anomaly

- Argument of perigee
- Inclination
- Right ascension of ascending node

The above six orbital elements define the orbit of earth satellites. Therefore, it is easy to discriminate one satellite from other satellites based on the values of orbital elements.

Semi major axis

The length of **Semi-major axis (a)** defines the size of satellite's orbit. It is half of the major axis. This runs from the center through a focus to the edge of the ellipse. So, it is the radius of an orbit at the orbit's two most distant points.

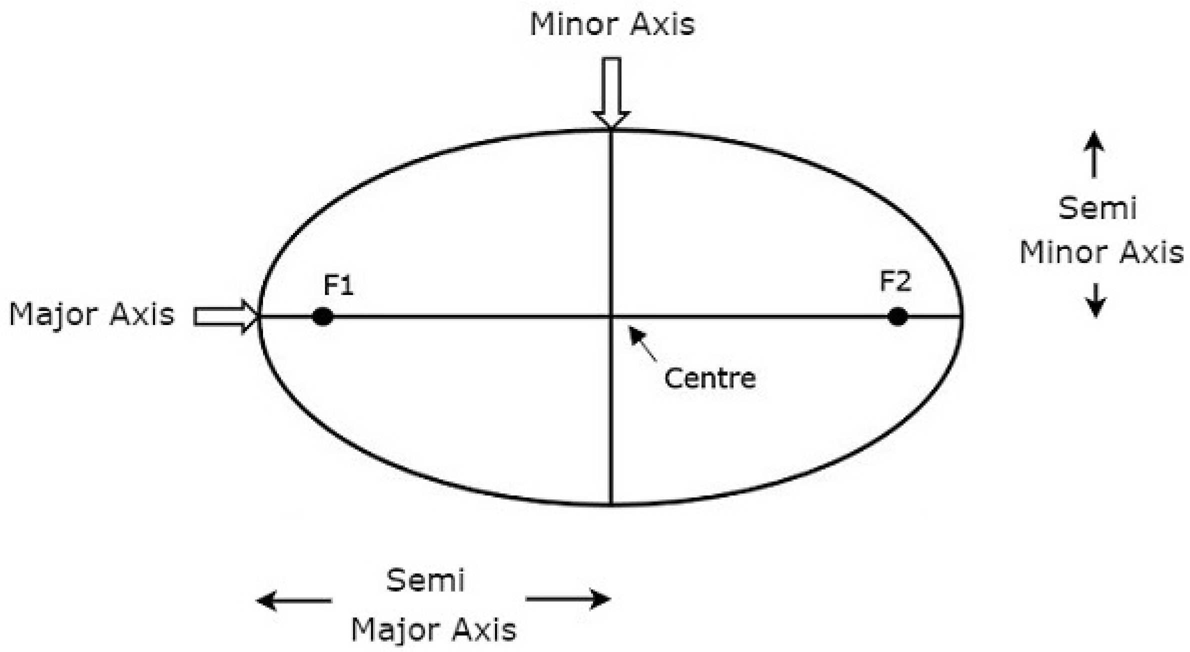

Fig:4 Satellite Semi Major and minor Axis

Both semi major axis and semi minor axis are represented in above figure. Length of semi **major axis (a)** not only determines the size of satellite's orbit, but also the time period of revolution. If circular orbit is considered as a special case, then the length of semi-major axis will be equal to **radius** of that circular orbit.

Eccentricity

The value of **Eccentricity (e)** fixes the shape of satellite's orbit. This parameter indicates the deviation of the orbit's shape from a perfect circle. If the lengths of semi major axis and semi minor axis of an elliptical orbit are a & b, then the mathematical expression for **eccentricity (e)** will be

$$e=\sqrt{a^2-b^2}/a$$

The value of eccentricity of a circular orbit is **zero**, since both a & b are equal. Whereas, the value of eccentricity of an elliptical orbit lies between zero and one. The following **figure** shows the various satellite orbits for different eccentricity (e) values

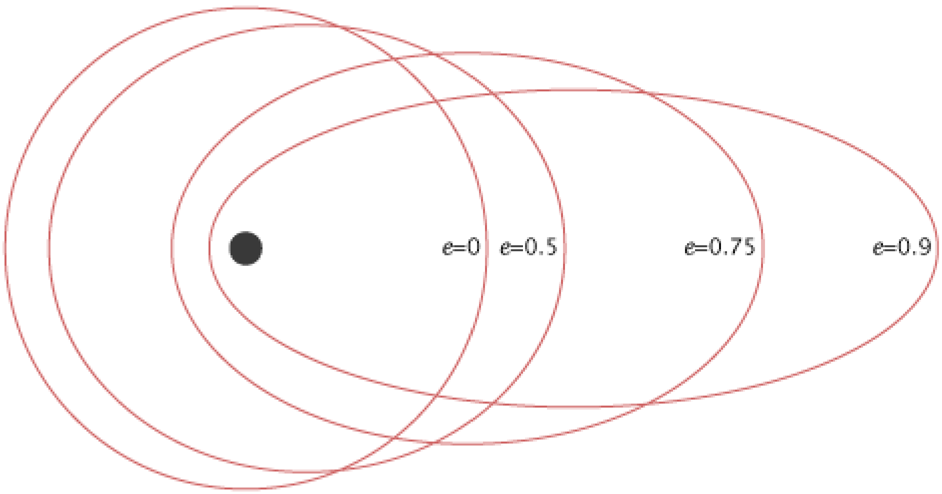

Fig:5 satellite orbits for different eccentricity (e) values

In above figure, the satellite orbit corresponding to eccentricity (e) value of zero is a circular orbit. And, the remaining three satellite orbits are of elliptical corresponding to the eccentricity (e) values 0.5, 0.75 and 0.9.

Mean Anomaly

For a satellite, the point which is closest from the Earth is known as Perigee. **Mean anomaly** (M) gives the average value of the angular position of the satellite with reference to perigee.

If the orbit is circular, then Mean anomaly gives the angular position of the satellite in the orbit. But, if the orbit is elliptical, then calculation of exact position is very difficult. At that time, Mean anomaly is used as an intermediate step.

Argument of Perigee

Satellite orbit cuts the equatorial plane at two points. First point is called as **descending node**, where the satellite passes from the northern hemisphere to the southern hemisphere. Second point is called as **ascending node**, where the satellite passes from the southern hemisphere to the northern hemisphere.

Argument of perigee (ω) is the angle between ascending node and perigee. If both perigee and ascending node are existing at same point, then the argument of perigee will be zero degrees. Argument of perigee is measured in the orbital plane at earth's center in the direction of satellite motion.

Inclination

The angle between orbital plane and earth's equatorial plane is known as **inclination (i)**. It is measured at the ascending node with direction being east to north. So, inclination defines the orientation of the orbit by considering the equator of earth as reference.

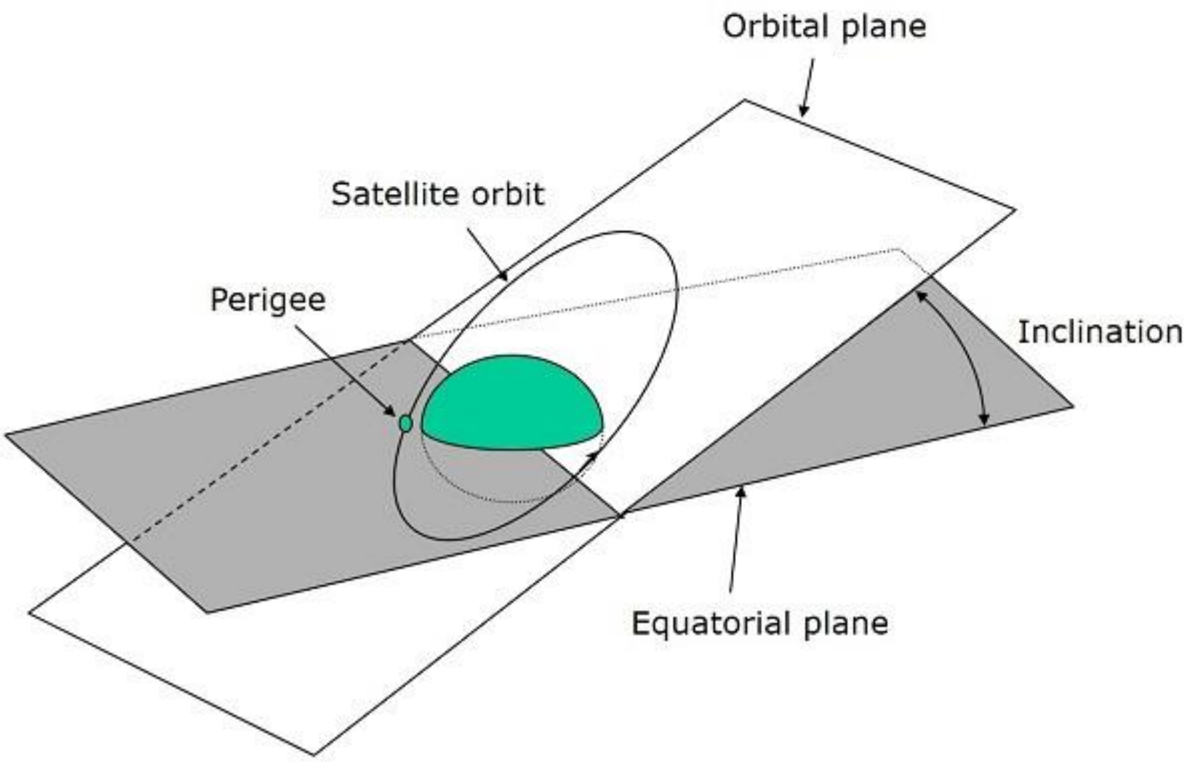

Fig:6 Inclination Angle

There are four types of orbits based on the angle of inclination.

- **Equatorial orbit** − Angle of inclination is either zero degrees or 180 degrees.
- **Polar orbit** − Angle of inclination is 90 degrees.
- **Prograde orbit** − Angle of inclination lies between zero and 90 degrees.
- **Retrograde orbit** − Angle of inclination lies between 90 and 180 degrees.

Right Ascension of Ascending node

We know that **ascending node** is the point, where the satellite crosses the equatorial plane while going from the southern hemisphere to the northern hemisphere. Right Ascension of ascending node **(Ω)** is the angle between line of Aries and ascending node towards east direction in equatorial plane. Aries is also called as vernal and equinox.

Satellite's **ground track** is the path on the surface of the Earth, which lies exactly below its orbit. The ground track of a satellite can take a number of different forms depending on the values of the orbital elements.

Orbital Equations

Forces acting on Satellite

A satellite, when it revolves around the earth, it undergoes a pulling force from the earth due to earth's gravitational force. This force is known as **Centripetal force** (F_1) because this force tends the satellite towards it.

Mathematically, the **Centripetal force** (F_1) acting on satellite due to earth can be written as

$$F_1=GMm/R^2$$

Where,

- ***G*** is universal gravitational constant and it is equal to 6.673×10^{-11} $N·m^2/kg^2$.
- ***M*** is mass of the earth and it is equal to 5.98×10^{24} Kg.
- ***m*** is mass of the satellite.
- ***R*** is the distance from satellite to center of the Earth.

A satellite, when it revolves around the earth, it undergoes a pulling force from the sun and the moon due to their gravitational forces. This force is known as **Centrifugal force** (F_2) because this force tends the satellite away from earth.

Mathematically, the **Centrifugal force** (F_2) acting on satellite can be written as

$$F2=mv^2/R$$

Where, **v** is the orbital velocity of satellite.

Orbital Velocity

Orbital velocity of satellite is the velocity at which, the satellite revolves around earth. Satellite doesn't deviate from its orbit and moves with certain velocity in that orbit, when both Centripetal and Centrifugal forces are **balance** each other.

So, **equate** Centripetal force (F_1) and Centrifugal force (F_2).

$$GMm/R^2=mv^2/R$$

$$=>GM/R=v^2$$

$$=>v=\sqrt{GM/R}$$

Therefore, the **orbital velocity** of satellite is

$$V=\sqrt{GM/R}$$

Where,

- ***G*** is gravitational constant and it is equal to 6.673×10^{-11} $N \cdot m^2/kg^2$.
- ***M*** is mass of the earth and it is equal to 5.98×10^{24} Kg.
- ***R*** is the distance from satellite to center of the Earth.

So, the orbital velocity mainly **depends** on the distance from satellite to center of the Earth (R), since G & M are constants.

Transponder

The subsystem, which provides the connecting link between transmitting and receiving antennas of a satellite is known as Transponder. It is one of the most important subsystem of space segment subsystems. Transponder performs the functions of both transmitter and receiver (Responder) in a satellite. Hence, the word 'Transponder' is obtained by the combining few letters of two words, Transmitter (Trans) and Responder (ponder).

Block diagram of Transponder

Transponder performs mainly **two functions**. Those are amplifying the received input signal and translates the frequency of it. In general, different frequency values are chosen for both uplink and down link in order to avoid the interference between the transmitted and received signals.

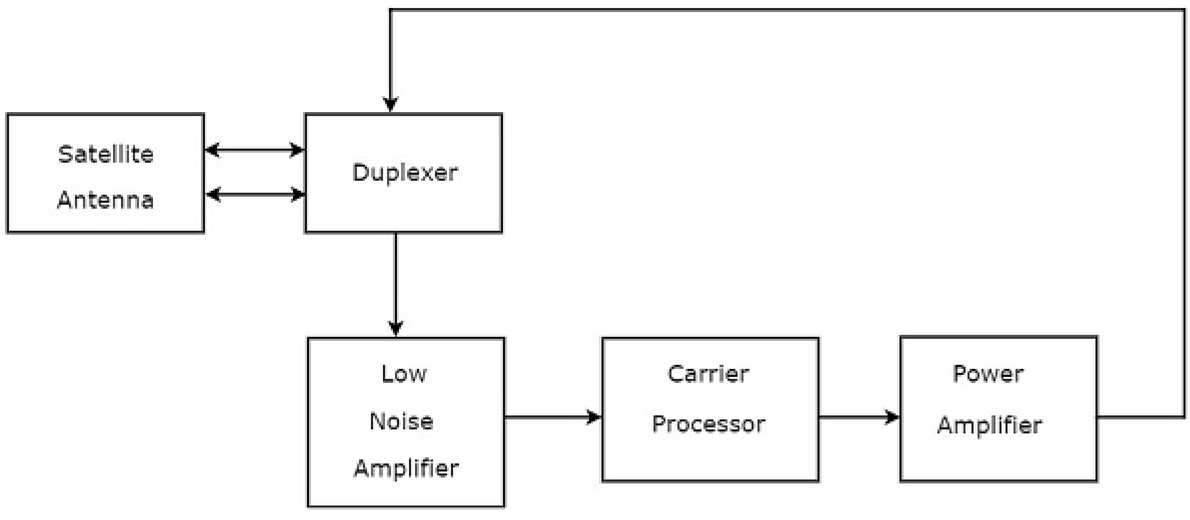

Fig:7 Block Diagram of a Transponder

The function of each block is mentioned below.

- **Duplexer** is a two-way microwave gate. It receives uplink signal from the satellite antenna and transmits downlink signal to the satellite antenna.
- **Low Noise Amplifier** (LNA) amplifies the weak received signal.
- **Carrier Processor** performs the frequency down conversion of received signal (uplink). This block determines the type of transponder.
- **Power Amplifier** amplifies the power of frequency down converted signal (down link) to the required level.

Types of Transponders

Basically, there are **two types** of transponders. Those are Bent pipe transponders and Regenerative transponders.

Bent Pipe Transponders

Bent pipe transponder receives microwave frequency signal. It converts the frequency of input signal to RF frequency and then amplifies it. Bent pipe transponder is also called as repeater and **conventional transponder**. It is suitable for both analog and digital signals.

Regenerative Transponders

Regenerative transponder performs the functions of Bent pipe transponder. i.e., frequency translation and amplification. In addition to these two functions, Regenerative transponder also performs the demodulation of RF carrier to baseband, regeneration of signals and modulation. Regenerative transponder is also called as Processing transponder. It is suitable only for digital signals. The main **advantages** of Regenerative transponders are improvement in Signal to Noise Ratio (SNR) and have more flexibility in implementation.

The earth segment of satellite communication system mainly consists of two earth stations. Those are transmitting earth station and receiving earth station. The transmitting earth station transmits the information signals to satellite. Whereas, the receiving earth station receives the information signals from satellite. Sometimes, the same earth station can be used for both transmitting and receiving purposes.

In general, earth stations receive the baseband signals in one of the following forms. Voice signals and video signals either in analog form or digital form. Initially, the analog modulation technique, named FM modulation is used for transmitting both voice and video signals, which are in analog form. Later, digital modulation techniques, namely Frequency Shift

Keying (FSK) and Phase Shift Keying (PSK) are used for transmitting those signals. Because, both voice and video signals are used to represent in digital by converting them from analog.

Block Diagram of Earth Station

Designing of an **Earth station** depends not only on the location of earth station but also on some other factors. The location of earth stations could be on land, on ships in sea and on aircraft. The depending factors are type of service providing, frequency bands utilization, transmitter, receiver and antenna characteristics.

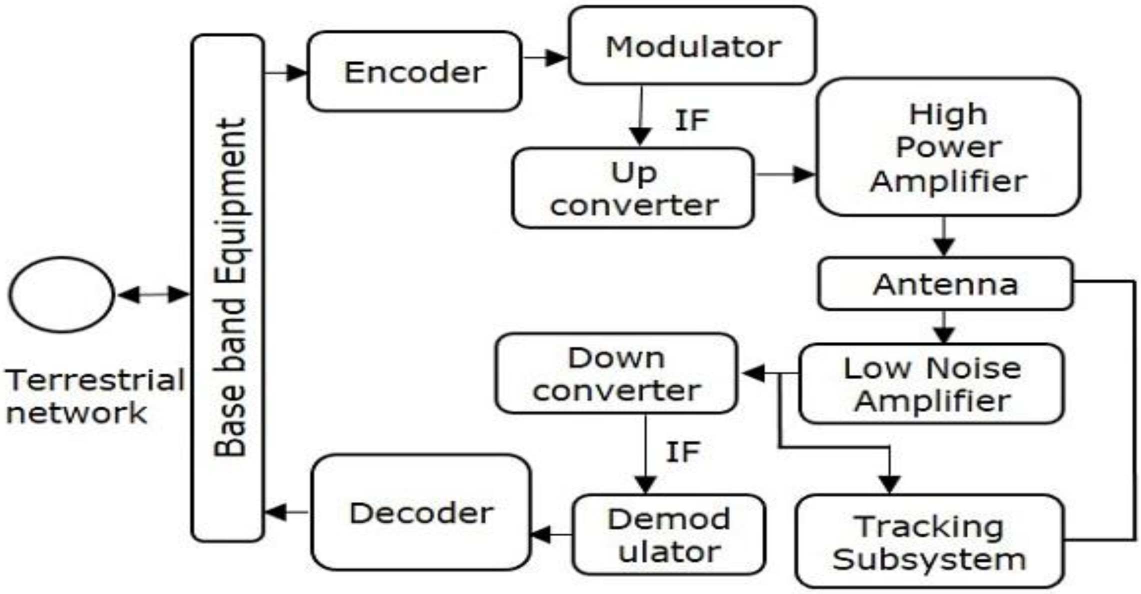

Fig:8 Block Diagram of Digital Earth Station

There are four major **subsystems** that are present in any earth station. Those are transmitter, receiver, antenna and tracking subsystem.

Transmitter

The binary (digital) information enters at base band equipment of earth station from terrestrial network. **Encoder** includes error correction bits in order to minimize the bit error rate. In satellite communication, the Intermediate Frequency **(IF)** can be chosen as 70 MHz by using a transponder having bandwidth of 36 MHz. Similarly, the IF can also be chosen as 140 MHz by using a transponder having bandwidth of either 54 MHz or 72 MHz. Up converter performs the frequency conversion of modulated signal to higher frequency. This signal will be amplified by using High power amplifier. The earth station antenna transmits this signal.

Receiver

During reception, the earth station antenna receives downlink signal. This is a low-level modulated RF signal. In general, the received signal will be having less signal strength. So, in order to amplify this signal, Low Noise Amplifier (LNA) is used. Due to this, there is an improvement in Signal to Noise Ratio (SNR) value. RF signal can be down converted to the Intermediate Frequency (IF) value, which is either 70 or 140 MHz. Because, it is easy to demodulate at these intermediate frequencies.

The function of the decoder is just opposite to that of encoder. So, the decoder produces an error free binary information by removing error correction bits and correcting the bit positions if any. This binary information is given to base band equipment for further processing and then delivers to terrestrial network.

Earth Station Antenna

The major parts of Earth station Antenna are feed system and Antenna reflector. These two parts combined together radiates or receives electromagnetic waves. Since the feed system obeys reciprocity theorem, the earth station antennas are suitable for both transmitting and receiving electromagnetic waves.

Parabolic reflectors are used as the main antenna in earth stations. The gain of these reflectors is high. They have the ability of focusing a parallel beam into a point at the focus, where the feed system is located.

Tracking Subsystem

The **Tracking subsystem** keeps track with the satellite and make sure that the beam comes towards it in order to establish the communication. The Tracking system present in the earth station performs mainly **two functions**. Those are satellite acquisition and tracking of satellite. This tracking can be done in one of the following ways. Those are automatic tracking, manual tracking & program tracking.

Satellite frequency bands

Satellite technology is developing fast, and the applications for satellite technology are increasing all the time. Not only can satellites be used for radio communications, but they are also used for astronomy, weather forecasting, broadcasting, mapping and many more applications. With the variety of satellite frequency bands that can be used, designations have been developed so that they can be referred to easily.

The higher frequency bands typically give access to wider bandwidths, but are also more susceptible to signal degradation due to 'rain fade' (the absorption of radio signals by atmospheric rain, snow or ice). Because of satellites' increased use, number and size, congestion has become a serious issue in the lower frequency bands. New technologies are being investigated so that higher bands can be used.

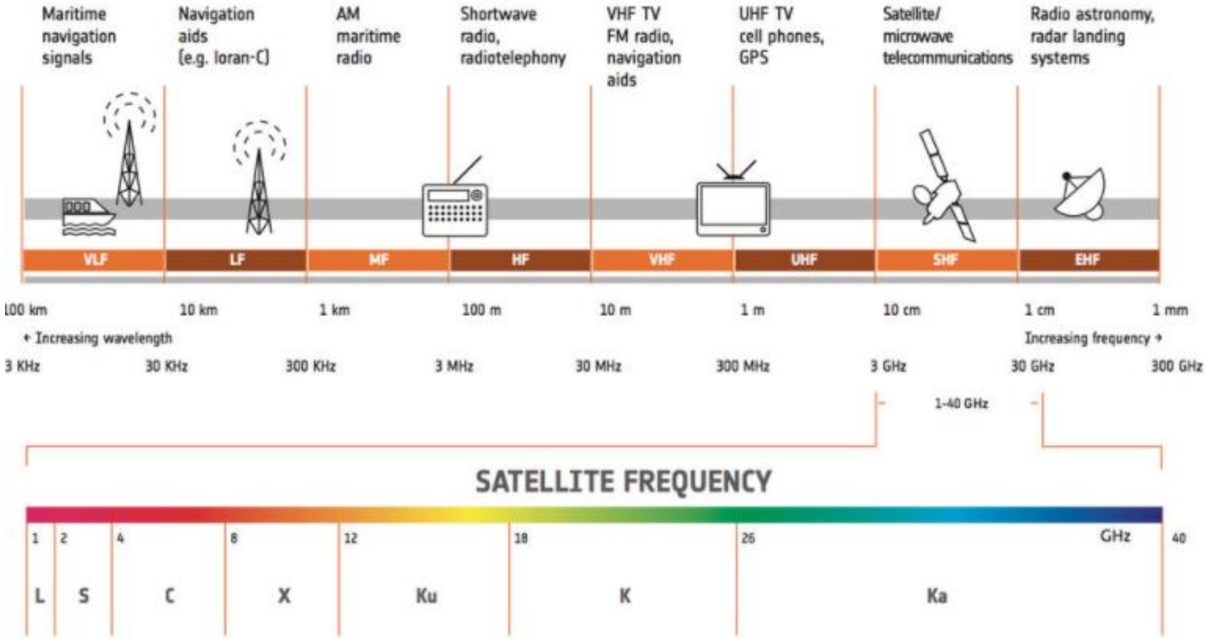

Fig: 9 Frequencies used for Satellite Communication

L-band (1–2 GHz)

Global Positioning System (GPS) carriers and also satellite mobile phones, such as Iridium; Inmarsat providing communications at sea, land and air; World Space satellite radio.

S-band (2–4 GHz)

Weather radar, surface ship radar, and some communications satellites, especially those of NASA for communication with ISS and Space Shuttle. In May 2009, Inmarsat and Solaris mobile (a joint venture between Eutelsat and Astra) were awarded each a 2×15 MHz portion of the S-band by the European Commission.

C-band (4–8 GHz)

Primarily used for satellite communications, for full-time satellite TV networks or raw satellite feeds. Commonly used in areas that are subject to tropical rainfall, since it is less susceptible to rain fade than Ku band (the original Telstar satellite had a transponder operating in this band, used to relay the first live transatlantic TV signal in 1962).

X-band (8–12 GHz)

Primarily used by the military. Used in radar applications including continuous-wave, pulsed, single-polarization, dual- polarization, synthetic aperture radar and phased arrays. X-band radar frequency sub-bands are used in civil, military and government institutions for weather monitoring, air traffic control, maritime vessel traffic control, defence tracking and vehicle speed detection for law enforcement.

Ku-band (12–18 GHz)

Used for satellite communications. In Europe, Ku-band downlink is used from 10.7 GHz to 12.75 GHz for direct broadcast satellite services, such as Astra.

Ka-band (26–40 GHz)

Communications satellites, uplink in either the 27.5 GHz and 31 GHz bands, and high-resolution, close-range targeting radars on military aircraft.

Satellite Multiple Access Formats

A satellite's service is present at a particular location on the earth station and sometimes it is not present. That means, a satellite may have different service stations of its own located at different places on the earth. They send carrier signal for the satellite. In this situation, we do multiple access to enable satellite to take or give signals from different stations at time without any interference between them.

Following are the **three types of multiple access techniques**.

- FDMA (Frequency Division Multiple Access)
- TDMA (Time Division Multiple Access)
- CDMA (Code Division Multiple Access)

FDMA

In this type of multiple access, we assign each signal a different type of frequency band (range). So, any two signals should not have same type of frequency range. Hence, there won't be any interference between them, even if we send those signals in one channel. One perfect example of this type of access is our radio channels. We can see that each station has been given a different frequency band in order to operate.

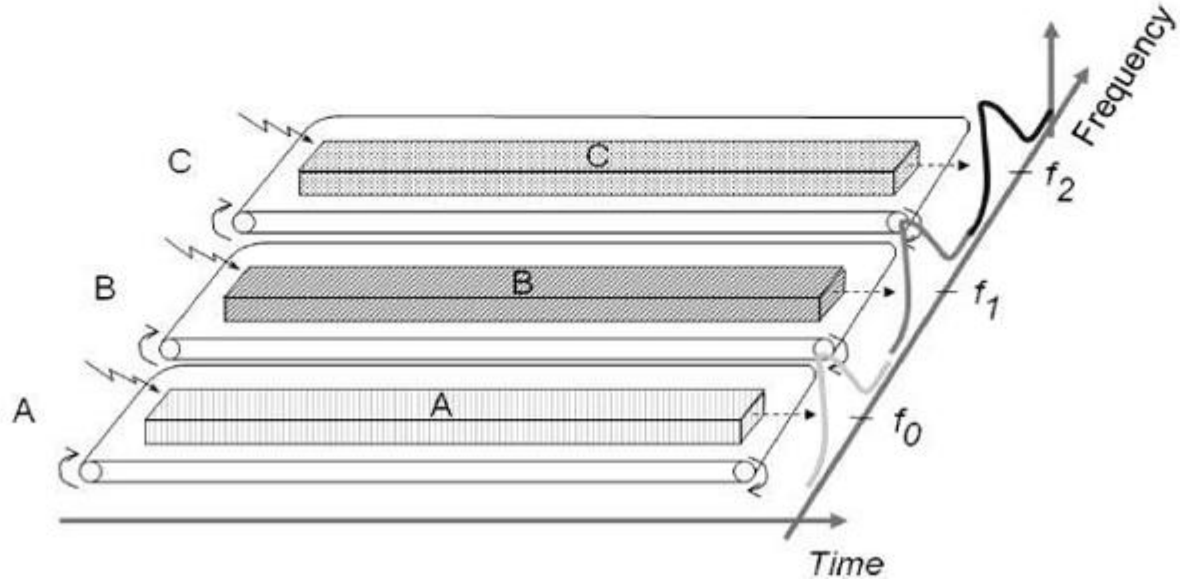

Fig:10 Frequency Division Multiple Access

Let's take three stations A, B and C. We want to access them through FDMA technique. So we assigned them different frequency bands. As shown in the figure, satellite station A has been kept under the frequency range of 0 to 20 HZ. Similarly, stations B and C have been assigned the frequency range of 30-60 Hz and 70-90 Hz respectively. There is no interference between them. The main disadvantage of this type of system is that it is very burst. This type of multiple access is not recommended for the channels, which are of dynamic and uneven. Because, it will make their data as inflexible and inefficient.

TDMA

As the name suggests, TDMA is a time based access. Here, we give certain time frame to each channel. Within that time frame, the channel can access the entire spectrum bandwidth. Each station got a fixed length or slot. The slots, which are unused will remain in idle stage.

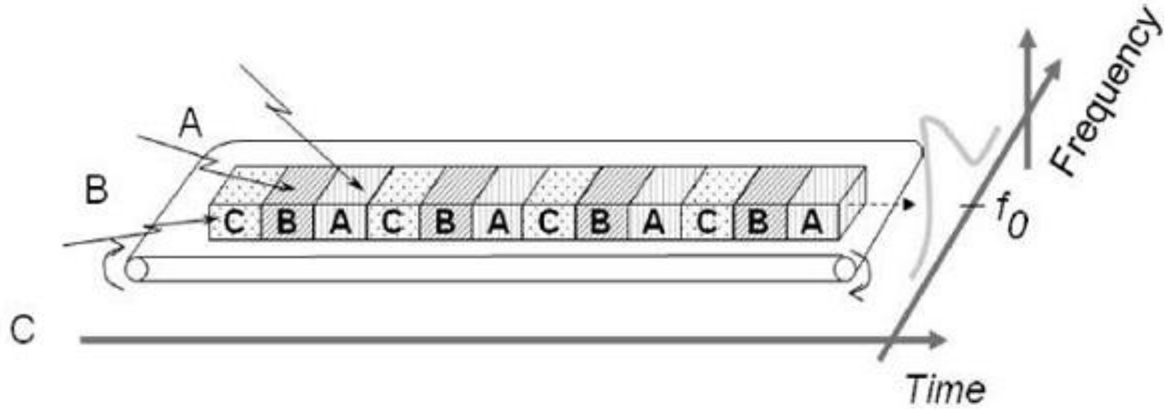

Fig:11 Time Division Multiple Access

Suppose, we want to send five packets of data to a particular channel in TDMA technique. So, we should assign them certain time slots or time frame within which it can access the entire bandwidth. In above figure, packets 1, 3 and 4 are active, which transmits data. Whereas, packets 2 and 5 are idle because of their non-participation. This format gets repeated every time we assign bandwidth to that particular channel.

Although, we have assigned certain time slots to a particular channel but it can also be changed depending upon the load bearing capacity. That means, if a channel is transmitting heavier loads, then it can be assigned a bigger time slot than the channel which is transmitting lighter loads. This is the biggest advantage of TDMA over FDMA. Another advantage of TDMA is that the power consumption will be very low.

Point to Note : In some applications, we use the combination of both TDMA and FDMA techniques. In this case, each channel will be operated in a particular frequency band for a particular time frame. In this case, the frequency selection is more robust and it has greater capacity over time compression.

CDMA

In CDMA technique, a unique code has been assigned to each channel to distinguish from each other. A perfect example of this type of multiple access is our cellular system. We can see that no two persons' mobile number match with each other although they are same X or Y mobile service providing company's customers using the same bandwidth. In CDMA process, we do the decoding of inner product of the encoded signal and chipping sequence. Therefore, mathematically it can be written as

$$\text{Encoded signal} = \text{Original data} \times \text{chipping sequence}$$

The basic advantage of this type of multiple access is that it allows all users to coexist and use the entire bandwidth at the same time. Since each user has different code, there won't be any interference. In this technique, a number of stations can have number of channels unlike FDMA and TDMA. The best part of this technique is that each station can use the entire spectrum at all time.

Basic Terminology

An isotropic radiator (antenna) radiates equally in all directions. But, it doesn't exist practically. It is just a theoretical antenna. We can compare the performance of all real (practical) antennas with respect to this antenna.

Power flux density

Assume an isotropic radiator is situated at the center of the sphere having radius, r. We know that power flux density is the ratio of power flow and unit area.

Power flux density, Ψi of an isotropic radiator is

$$\Psi i = ps4\pi r^2$$

Where, Ps is the power flow. In general, the power flux density of a practical antenna varies with direction. But, it's maximum value will be in one particular direction only.

Antenna Gain

The gain of practical antenna is defined as the ratio of maximum power flux density of practical antenna and power flux density of isotropic antenna.

Therefore, the Gain of Antenna or Antenna gain, G is

$$G = \Psi_m / \Psi_i$$

Where, Ψm is the maximum power flux density of practical antenna. And, Ψi is the power flux density of isotropic radiator (antenna).

Equivalent Isotropic Radiated Power

Equivalent isotropic radiated power (EIRP) is the main parameter that is used in measurement of link budget. Mathematically, it can be written as

$$EIRP=GP_s$$

We can represent EIRP in decibels as

$$[EIRP]=[G]+[P_s]dBW$$

Where, G is the Gain of Transmitting antenna and Ps is the power of transmitter.

Transmission Losses

The difference between the power sent at one end and received at the receiving station is known as Transmission losses. The losses can be categorized into 2 types.

- Constant losses
- Variable losses

The losses which are constant such as feeder losses are known as constant losses. No matter what precautions we might have taken, still these losses are bound to occur. Another type of loses are variable loss. The sky and weather condition is an example of this type of loss. Means if the sky is not clear signal will not reach effectively to the satellite or vice versa. Therefore, our procedure includes the calculation of losses due to clear weather or clear sky condition as 1st because these losses are constant. They will not change with time. Then in 2nd step, we can calculate the losses due to foul weather condition.

Link budget calculations

There are two types of link budget calculations since there are two links namely, uplink and downlink.

Earth Station Uplink

It is the process in which earth is transmitting the signal to the satellite and satellite is receiving it. Its mathematical equation can be written as

$$(C/N0)U=[EIRP]U+(G/T)U-[LOSSES]U-K$$

Where,

- [CN0] is the carrier to noise density ratio
- [GT] is the satellite receiver G/T ratio and units are dB/K

Here, Losses represent the satellite receiver feeder losses. The losses which depend upon the frequency are all taken into the consideration. The EIRP value should be as low as possible for effective UPLINK. And this is possible when we get a clear sky condition. Here we have used the (subscript) notation "U", which represents the uplink phenomena.

Satellite Downlink

In this process, satellite sends the signal and the earth station receives it. The equation is same as the satellite uplink with a difference that we use the abbreviation "D" everywhere instead of "U" to denote the downlink phenomena.

Its mathematical equation can be written as;

$$[C/N_0]_D = [EIRP]_D + [GT]_D - [LOSSES]_D - K$$

Where,

- [C/N0] is the carrier to noise density ratio
- [G/T] is the earth station receiver G/T ratio and units are dB/K

Here, all the losses that are present around earth stations. In the above equation we have not included the signal bandwidth B. However, if we include that the equation will be modified as follows.

$$[C/N_0]_D = [EIRP]_D + [GT]_D - [LOSSES]_D - K - B$$

Link Budget

If we are taking ground satellite in to consideration, then the free space spreading loss (FSP) should also be taken into consideration. If antenna is not aligned properly then losses can occur. so we take AML (Antenna misalignment losses) into account. Similarly, when signal comes from the satellite towards earth it collides with earth surface and some of them get absorbed. These are taken care by atmospheric absorption loss given by "AA" and measured in db.

Now, we can write the loss equation for free sky as

$$Losses = FSL + RFL + AML + AA + PL$$

Where,

- RFL stands for received feeder loss and units are db.
- PL stands for polarization mismatch loss.

Now the decibel equation for received power can be written as

$$PR = EIRP + GR + Losses$$

Where,

- PRPR stands for the received power, which is measured in dBW.
- Gr is the receiver antenna gain.

The designing of down link is more critical than the designing of uplink. Because of limitations in power required for transmitting and gain of the antenna. In satellite communication systems, there are two types of power calculations. Those are transmitting power and receiving power calculations. In general, these calculations are called as Link budget calculations. The unit of power is decibel.

Parabolic dish antenna

The parabolic antenna was invented by German physicist Heinrich Hertz during his discovery of radio waves in 1887. He used cylindrical parabolic reflectors with spark-excited dipole antennas at their focus for both transmitting and receiving during his historic experiments.

A parabolic antenna is an antenna that uses a parabolic reflector, a curved surface with the cross-sectional shape of a parabola, to direct the radio waves. The most common form is shaped like a dish and is popularly called a dish antenna or parabolic dish. The main advantage of a parabolic antenna is that it has high directivity. It functions similarly to a searchlight or flashlight reflector to direct the radio waves in a narrow beam, or receive radio waves from one particular direction only.

Parabolic antennas have some of the highest gains, meaning that they can produce the narrowest beamwidths, of any antenna type. In order to achieve narrow beamwidths, the parabolic reflector must be much larger than the wavelength of the radio waves used, so parabolic antennas are used in the high frequency part of the radio spectrum, at UHF and microwave (SHF) frequencies, at which the wavelengths are small enough that conveniently-sized reflectors can be used.

Parabolic antennas are used as high-gain antennas for point-to-point communications, in applications such as microwave relay links that carry telephone and television signals between nearby cities, wireless WAN/LAN links for data communications, satellite communications and spacecraft communication antennas. They are also used in radio telescopes. The other large use of parabolic antennas is for radar antennas, in which there is a need to transmit a narrow beam of radio waves to locate objects like ships, airplanes, and guided missiles, and often for weather detection. With the advent of home satellite television receivers, parabolic antennas have become a common feature of the landscapes of many modern countries.

Frequency Range

The frequency range used for the application of Parabolic reflector antennas is above 1MHz. These antennas are widely used for radio and wireless applications.

Principle of Operation

The standard definition of a parabola is - Locus of a point, which moves in such a way that its distance from the fixed point (called focus) plus its distance from a straight line (called directrix) is constant.

The point F is the focus (feed is given) and V is the vertex. The line joining F and V is the axis of symmetry. PQ are the reflected rays where L represents the line directrix on which the reflected points lie (to say that they are being collinear). Hence, as per the above definition, the distance between F and L lie constant with respect to the waves being focused.

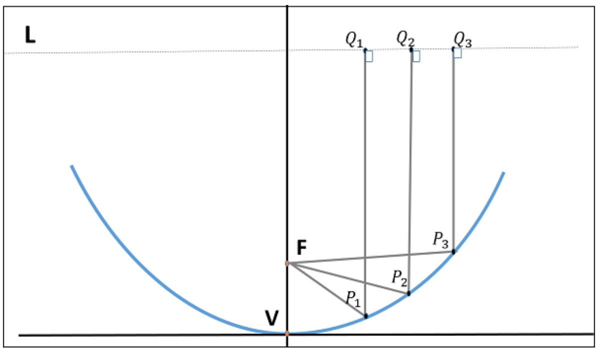

Fig:12 The Geometry of Parabolic Reflector.

The reflected wave forms a colllimated wave front, out of the parabolic shape. The ratio of focal length to aperture size (ie., f/D) known as “f over D ratio” is an important parameter of parabolic reflector. Its value varies from 0.25 to 0.50.

The law of reflection states that the angle of incidence and the angle of reflection are equal. This law when used along with a parabola, helps the beam focus. The shape of the parabola when used for the purpose of reflection of waves, exhibits some properties of the parabola, which are helpful for building an antenna, using the waves reflected.

Properties of Parabola

- All the waves originating from focus, reflects back to the parabolic axis. Hence, all the waves reaching the aperture are in phase.
- As the waves are in phase, the beam of radiation along the parabolic axis will be strong and concentrated.

Construction & Working of a Parabolic Reflector

If a Parabolic Reflector antenna is used for transmitting a signal, the signal from the feed, comes out of a dipole or a horn antenna, to focus the wave on to the parabola. It means that, the waves come out of the focal point and strike the Paraboloidal reflector. This wave now gets reflected as collimated wave front, as discussed previously, to get transmitted.

The same antenna is used as a receiver. When the electromagnetic wave hits the shape of the parabola, the wave gets reflected onto the feed point. The dipole or the horn antenna, which acts as the receiver antenna at its feed, receives this signal, to convert it into electric signal and forwards it to the receiver circuitry.

Fig:13 Parabolic Reflector Antenna

The gain of the paraboloid is a function of aperture ratio (D/λ). The Effective Radiated Power (ERP) of an antenna is the multiplication of the input power fed to the antenna and its power gain.

Usually a wave guide horn antenna is used as a feed radiator for the paraboloid reflector antenna. Along with this technique, we have another type of feed given to the paraboloid reflector antenna, called as Cassegrain feed.

Cassegrain Feed

Cassegrain is another type of feed given to the reflector antenna. In this type, the feed is located at the vertex of the paraboloid, unlike in the parabolic reflector. A convex shaped reflector, which acts as a hyperboloid is placed opposite to the feed of the antenna. It is also known as secondary hyperboloid reflector or sub-reflector. It is placed such that its one of the foci coincides with the focus of the paraboloid. Thus, the wave gets reflected twice.

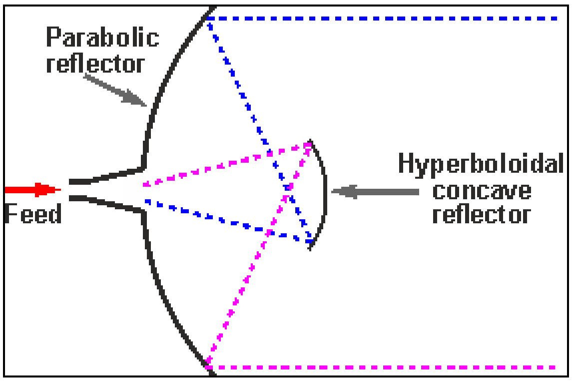

Fig:14 The working model of cassegrain feed.

Working of a Cassegrain Antenna

When the antenna acts as a transmitting antenna, the energy from the feed radiates through a horn antenna onto the hyperboloid concave reflector, which again reflects back on to the parabolic reflector. The signal gets reflected into the space from there. Hence, wastage of power is controlled and the directivity gets improved.

When the same antenna is used for reception, the electromagnetic waves strike the reflector, gets reflected on to the concave hyperboloid and from there, it reaches to the feed. A wave guide horn antenna presents there to receive this signal and sends to the receiver circuitry for amplification.

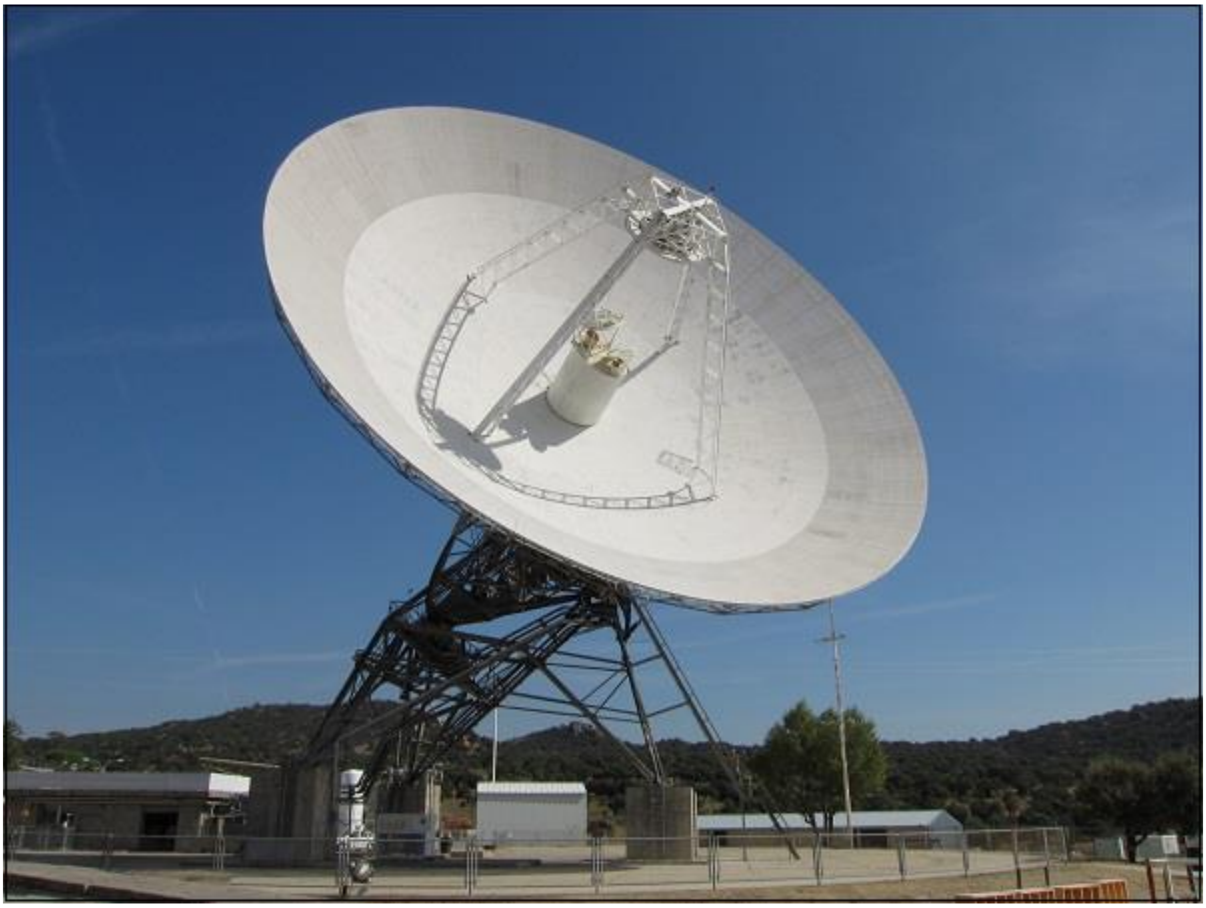

Fig:15 paraboloid reflector with cassegrain feed

Advantages

- Reduction of minor lobes
- Wastage of power is reduced
- Equivalent focal length is achieved
- Feed can be placed in any location, according to our convenience
- Adjustment of beam (narrowing or widening) is done by adjusting the reflecting surfaces

Disadvantage

- Some of the power that gets reflected from the parabolic reflector is obstructed. This becomes a problem with small dimension paraboloid.

Applications

- The cassegrain feed parabolic reflector is mainly used in satellite communications.
- Also used in wireless telecommunication systems.

Gregorian Feed

This is another type of feed used. A pair of certain configurations are there, where the feed beamwidth is progressively increased while antenna dimensions are held fixed. Such a type of feed is known as Gregorian feed. Here, the convex shaped hyperboloid of casssegrain is replaced with a concave shaped paraboloid reflector, which is of course, smaller in size

These **Gregorian feed** type reflectors can be used in four ways −

- Gregorian systems using reflector ellipsoidal sub-reflector at foci F1.
- Gregorian systems using reflector ellipsoidal sub-reflector at foci F2.
- Cassegrain systems using hyperboloid sub-reflector (convex).
- Cassegrain systems using hyperboloid sub-reflector (concave but the feed being very near to it.)

These are all just to mention because they are not popular and are not widely used. They have got their limitations.

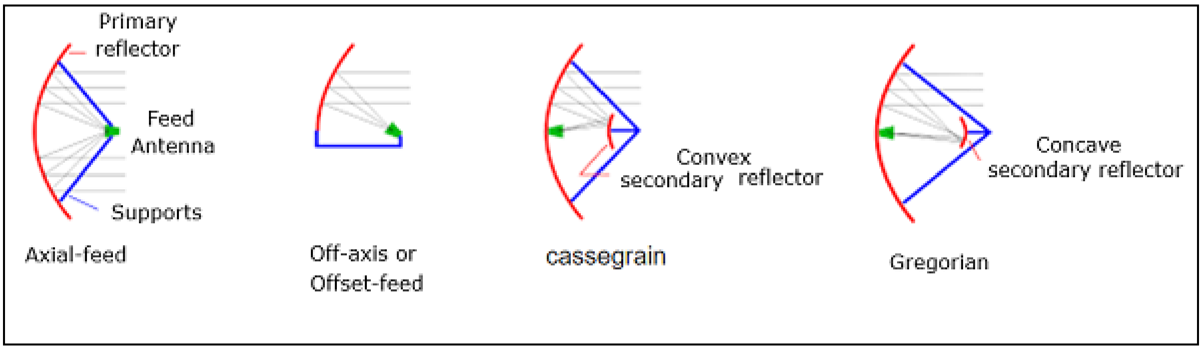

Fig:16 Working Pattern of all the Types of Reflectors

There are other types of paraboloid Reflectors such as −

- Cut- paraboloid
- Parabolic cylinder
- Pill-box paraboloid

However, all of them are seldom used because of the limitations and disadvantages they have in their working conditions. Hence, of all the types of reflector antennas, the simple parabolic reflectors and the cassegrain feed parabolic reflectors are the most commonly used ones.

Direct to home Broadcast (DTH):

DTH stands for Direct-To-Home television. DTH is defined as the reception of satellite programmes with a personal dish in an individual home. DTH Broadcasting to home TV receivers take place in the ku band(12 GHz). This service is known as Direct To Home service.

DTH services were first proposed in India in 1996. Finally, in 2000, DTH was allowed. The new policy requires all operators to set up earth stations in India within 12 months of getting a license. DTH licenses in India will cost $2.14 million and will be valid for 10 years. Working principal of DTH is the satellite communication. Broadcaster modulates the received signal and transmit it to the satellite in KU Band and from satellite one can receive signal by dish and set top box.

Direct-to-Home satellite broadcasting or DTH is the distribution of television signals from high-powered geostationary satellites to small dish antennas and satellite receivers in homes across the country. Early satellite television was broadcast in C band - radio in the 3.4-gigahertz (GHz) to 7-GHz frequency range. Digital broadcast satellite transmits programming in the Ku frequency range (10 GHz to 14 GHz). The DTH signals can be received directly at homes with the help of a small sized dish receive unit containing a Dish Antenna of diameter 60 to 90 cm installed at the building's roof-top or on the wall facing clear south and one indoor. Set-Top-Box unit facilitating viewing of demultiplexed signals from DTH channel bouquet on TV set.

The DTH signals can be received anywhere across the country irrespective of the terrain conditions provided the area comes under the footprint of the Satellite. DTH transmission eliminates the intervening role of a local cable operator since a user is directly connected to the DTH service. DTH Transmission is most preferred in Ku- Band so as to avoid the need of larger Dish sizes for suitably receiving the DTH signals. As the DTH telecast is in the Digital mode, user is able to reap all the benefits of Digital transmission. Programs in the DTH bouquet are having higher resolution picture and better audio quality than traditional analog signals.

DTH Block Diagram

A DTH network consists of a broadcasting centre, satellites, encoders, multiplexers, modulators and DTH receivers. The encoder converts the audio, video and data signals into the digital format and the multiplexer mixes these signals. It is used to provide the DTH service in high populated area A Multi Switch is basically a box that contains signal splitters and A/B switches. A outputs of group of DTH LNBs are connected to the A and B inputs of the Multi Switch.

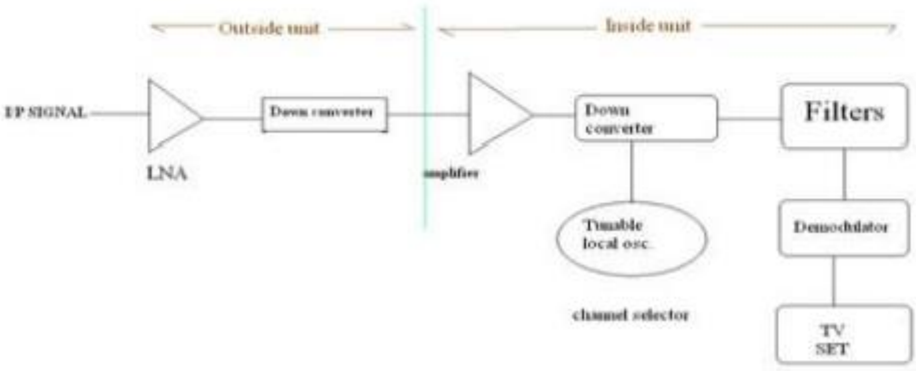

Fig: 17 DTH Service

Advantage:

The main advantage is that this technology is equally beneficial to everyone. As the process is wireless, this system can be used in all remote or urban areas. High quality audio and video which are cost effective due to absence of mediators. Almost 4000 channels can be viewed along with 2000 radio channels. Thus the world's entire information including news and entertainment is available to subscriber at home. As there are no mediators, a complaint can be directly expressed to the provider. With a single DTH service consumer will be able to use digital quality audio, video and also high-speed broadband. It also offers interactive channels and program guides with customers having the choice to block out programming which they consider undesirable.

DTH also offers digital quality signals which do not degrade the picture or sound quality. It also offers interactive channels and program guides with customers having the choice to block out programming which they consider undesirable. One of the great advantages of the cable industry has been the ability to provide local channels, but this handicap has been overcome by many DTH providers using other local channels or local feeds. The other advantage of DTH is the availability of satellite broadcast in rural and semi-urban areas where cable is difficult to install.

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UNIT V RADAR SYSTEMS AND OPTICAL FIBER

UNIT 5 RADAR SYSTEMS AND OPTICAL FIBER

Introduction, Basic Radar systems, Radar systems – Radar range – Pulsed radar system – A Scope – Plan Position Indicator (PPI) – Search Radar – Tracking Radar – Moving Target Indicator (MTI) – Doppler Effect – MTI principle – Digital MTI – Radar Beacons. Optical Fiber: Introduction to light, optical fiber and fiber cables, optical fiber characteristics and classification, losses, Fiber optic components and systems, Installation, testing and repair.

Introduction

RADAR is an electromagnetic based detection system that works by radiating electromagnetic waves and then studying the echo or the reflected back waves. The full form of RADAR is **RA**dio **D**etection **A**nd **R**anging. Detection refers to whether the target is present or not. The target can be stationary or movable, i.e., non-stationary. Ranging refers to the distance between the Radar and the target. Radars can be used for various applications on ground, on sea and in space.

The applications of Radars are:

- Controlling the Air Traffic
- Ship safety
- Sensing the remote places
- Military applications

In any application of Radar, the basic principle remains the same.

Basic principle of radar.

Radar is used for detecting the objects and finding their location.

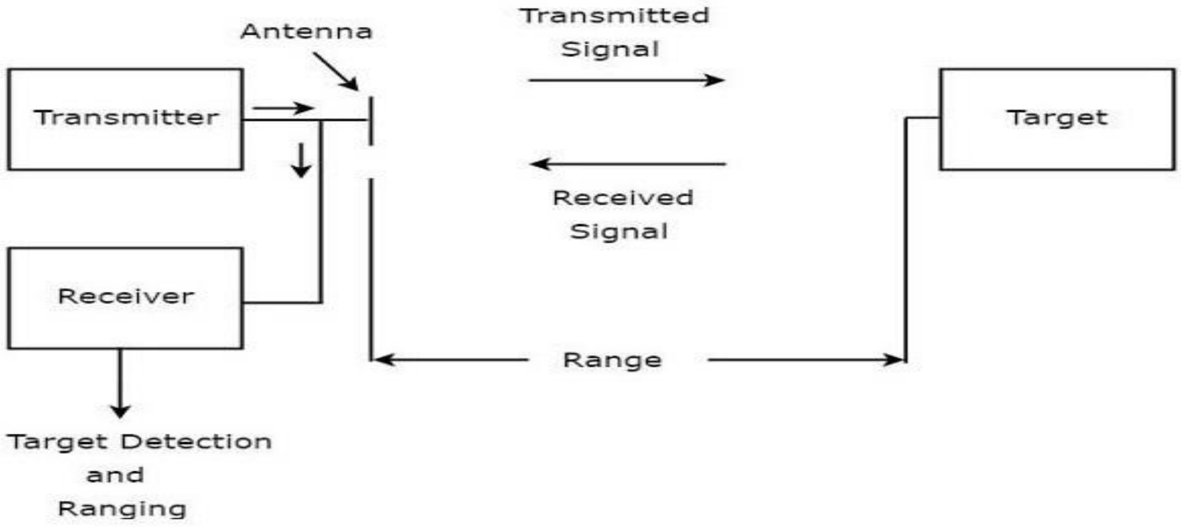

Fig:1 Block Diagram of Radar Communication

Radar mainly consists of a transmitter and a receiver. It uses the same Antenna for both transmitting and receiving the signals. The function of the transmitter is to transmit the Radar signal in the direction of the target present. Target reflects this received signal in various directions. The signal, which is reflected back towards the Antenna gets received by the receiver.

Terminology of Radar Systems

- Range
- Pulse Repetition Frequency
- Maximum Unambiguous Range
- Minimum Range

Range

The distance between Radar and target is called Range of the target or simply range, R. We know that Radar transmits a signal to the target and accordingly the target sends an echo signal to the Radar with the speed of light, C.

Let the time taken for the signal to travel from Radar to target and back to Radar be 'T'. The two way distance between the Radar and target will be 2R, since the distance between the Radar and the target is R.

Now, the following is the formula for **Speed**.

$$\text{Speed} = \text{Distance} / \text{Time}$$

$$\Rightarrow \text{Distance} = \text{Speed} \times \text{Time}$$

$$\Rightarrow 2R = C \times T$$

$$R = CT/2 \qquad \text{Equation1}$$

We can find the **range of the target** by substituting the values of C & T in Equation 1.

Pulse Repetition Frequency

Radar signals should be transmitted at every clock pulse. The duration between the two clock pulses should be properly chosen in such a way that the echo signal corresponding to present clock pulse should be received before the next clock pulse.

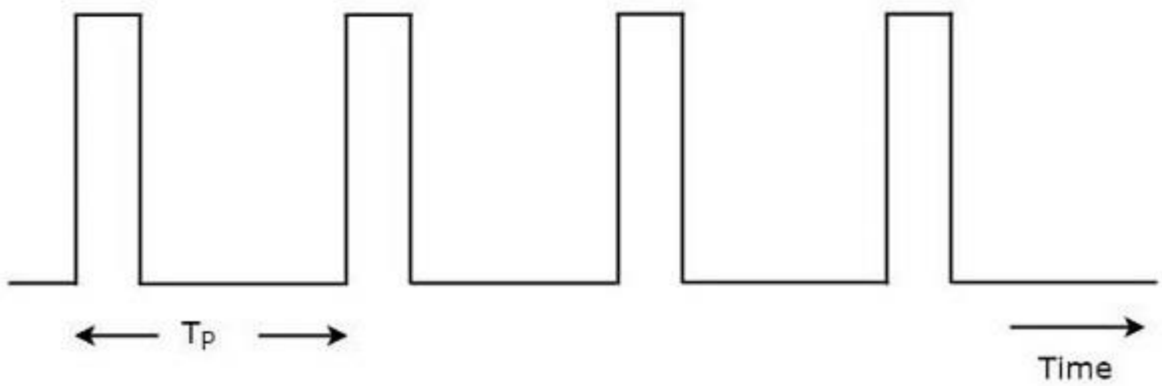

Fig:2 A typical Radar wave form

As shown in the above figure, Radar transmits a periodic signal. It is having a series of narrow rectangular shaped pulses. The time interval between the successive clock pulses is called pulse repetition time, T_P.

The reciprocal of pulse repetition time is called pulse repetition frequency, f_P. Mathematically, it can be represented as

$$f_P = 1/T_P \qquad \text{Equation2}$$

Therefore, pulse repetition frequency is nothing but the frequency at which Radar transmits the signal.

Maximum Unambiguous Range

We know that Radar signals should be transmitted at every clock pulse. If we select a shorter duration between the two clock pulses, then the echo signal corresponding to present clock pulse will be received after the next clock pulse. Due to this, the range of the target seems to be smaller than the actual range.

So, we have to select the duration between the two clock pulses in such a way that the echo signal corresponding to present clock pulse will be received before the next clock pulse starts. Then, we will get the true range of the target and it is also called maximum unambiguous range of the target or simply, maximum unambiguous range.

Substitute, $R=R_{un}$ and $T=T_P$ in Equation 1.

$$R_{un}=CT_P/2 \qquad \text{Equation3}$$

From Equation 2, we will get the pulse repetition time, T_P as the reciprocal of pulse repetition frequency, f_P.

Mathematically, it can be represented as

$$T_P=1/f_P \qquad \text{Equation4}$$

Substitute, Equation 4 in Equation 3.

$$R_{un} = C(1/f_P)/2$$

$$R_{un} = C/2f_P \qquad \text{Equation5}$$

We can use either Equation 3 or Equation 5 for calculating maximum unambiguous range of the target. We will get the value of maximum unambiguous range of the target, R_{un} by substituting the values of C and T_P in Equation 3. Similarly, we will get the value of maximum unambiguous range of the target, R_{un} by substituting the values of C and f_P in Equation 5.

Minimum Range

We will get the minimum range of the target, when we consider the time required for the echo signal to receive at Radar after the signal being transmitted from the Radar as pulse width. It is also called the shortest range of the target.

Substitute, $R = R_{min}$ and $T=\tau$ in Equation 1.

$$R_{min} = C\tau/2 \qquad \text{Equation 6}$$

We will get the value of minimum range of the target, R_{min} by substituting the values of C and τ in Equation 6.

Types of Radar

- Pulse Radar
- Continuous Wave Radar

Pulse Radar

The Radar, which operates with pulse signal is called the Pulse Radar. Pulse Radars can be classified into the following two types based on the type of the target it detects.

- Basic Pulse Radar
- Moving Target Indication Radar

Basic Pulse Radar

The Radar, which operates with pulse signal for detecting stationary targets, is called the Basic Pulse Radar or simply, Pulse Radar. It uses single Antenna for both transmitting and receiving signals with the help of Duplexer.

Antenna will transmit a pulse signal at every clock pulse. The duration between the two clock pulses should be chosen in such a way that the echo signal corresponding to the present clock pulse should be received before the next clock pulse.

Moving Target Indication Radar

The Radar, which operates with pulse signal for detecting non-stationary targets, is called Moving Target Indication Radar or simply, MTI Radar. It uses single Antenna for both transmission and reception of signals with the help of Duplexer.

MTI Radar uses the principle of Doppler effect for distinguishing the non-stationary targets from stationary objects.

Continuous Wave Radar

The Radar, which operates with continuous signal or wave is called Continuous Wave Radar. They use Doppler Effect for detecting non-stationary targets. Continuous Wave Radars can be classified into the following two types.

- Unmodulated Continuous Wave Radar
- Frequency Modulated Continuous Wave Radar

Unmodulated Continuous Wave Radar

The Radar, which operates with continuous signal (wave) for detecting non-stationary targets is called Unmodulated Continuous Wave Radar or simply, CW Radar. It is also called CW Doppler Radar. This Radar requires two Antennas. Of these two antennas, one Antenna is used for transmitting the signal and the other Antenna is used for receiving the signal. It measures only the speed of the target but not the distance of the target from the Radar.

Frequency Modulated Continuous Wave Radar

If CW Doppler Radar uses the Frequency Modulation, then that Radar is called the Frequency Modulated Continuous Wave (FMCW) Radar or FMCW Doppler Radar. It is also called Continuous Wave Frequency Modulated Radar or CWFM Radar. This Radar requires two Antennas. Among which, one Antenna is used for transmitting the signal and the other

Antenna is used for receiving the signal. It measures not only the speed of the target but also the distance of the target from the Radar. The Radar, which operates with pulse signal for detecting stationary targets is called Basic Pulse Radar or simply, Pulse Radar.

Block Diagram of Pulse Radar

Pulse Radar uses single Antenna for both transmitting and receiving of signals with the help of Duplexer.

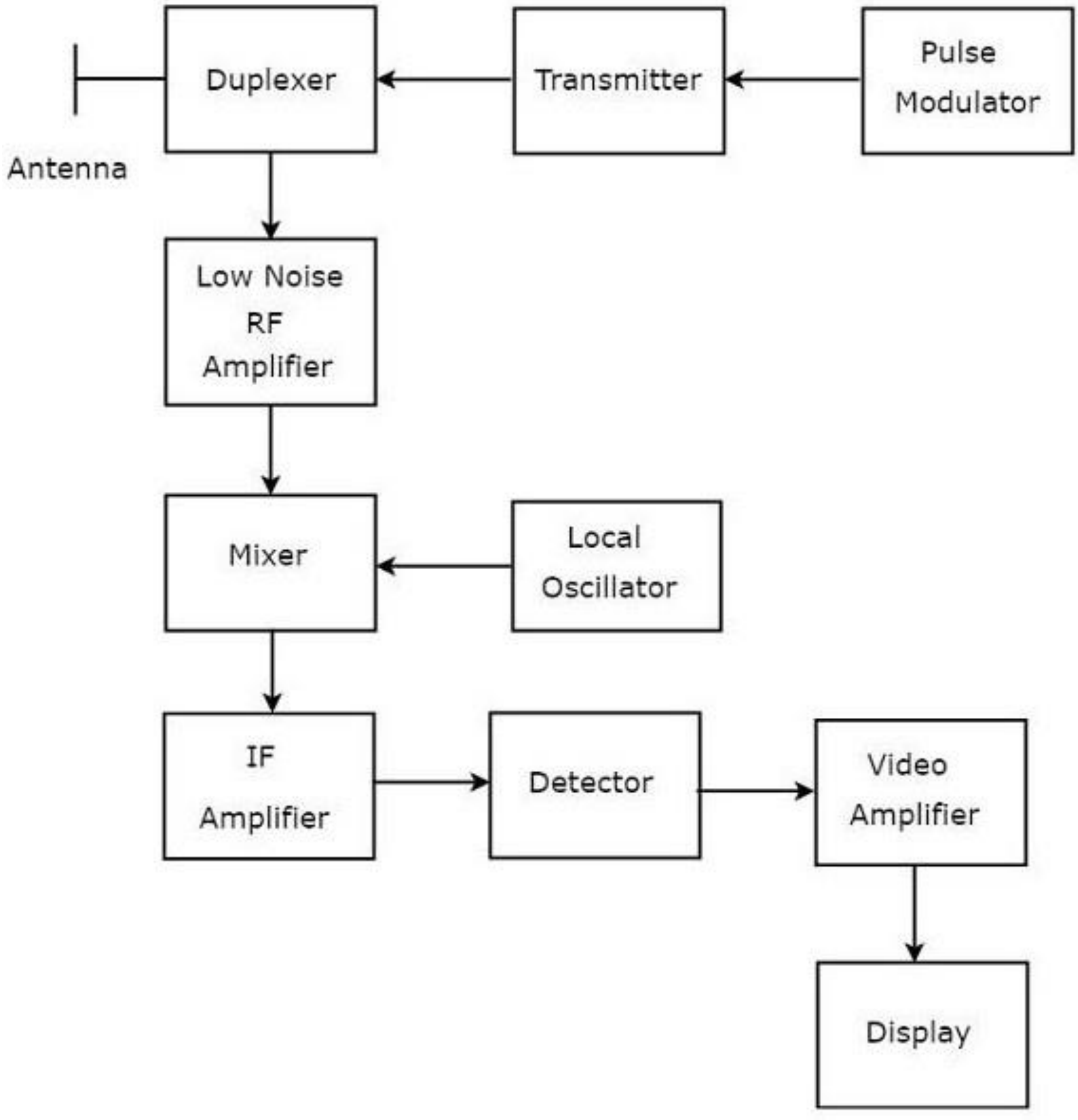

Fig:3 Block Diagram of Pulse Radar

The **function** of each block of Pulse Radar are as follows:

- **Pulse Modulator** − It produces a pulse-modulated signal and it is applied to the Transmitter.

- **Transmitter** − It transmits the pulse-modulated signal, which is a train of repetitive pulses.
- **Duplexer** − It is a microwave switch, which connects the Antenna to both transmitter section and receiver section alternately. Antenna transmits the pulse-modulated signal, when the duplexer connects the Antenna to the transmitter. Similarly, the signal, which is received by Antenna will be given to Low Noise RF Amplifier, when the duplexer connects the Antenna to Low Noise RF Amplifier.
- **Low Noise RF Amplifier** − It amplifies the weak RF signal, which is received by Antenna. The output of this amplifier is connected to Mixer.
- **Local Oscillator** − It produces a signal having stable frequency. The output of Local Oscillator is connected to Mixer.
- **Mixer** − We know that Mixer can produce both sum and difference of the frequencies that are applied to it. Among which, the difference of the frequencies will be of Intermediate Frequency (IF) type.
- **IF Amplifier** − IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure allows only the Intermediate Frequency, which is obtained from Mixer and amplifies it. It improves the Signal to Noise Ratio at output.
- **Detector** − It demodulates the signal, which is obtained at the output of the IF Amplifier.
- **Video Amplifier** − As the name suggests, it amplifies the video signal, which is obtained at the output of detector.
- **Display** − In general, it displays the amplified video signal on CRT screen.

Doppler Effect

If the target is not stationary, then there will be a change in the frequency of the signal that is transmitted from the Radar and that is received by the Radar. This effect is known as the Doppler effect.

According to the Doppler effect, we will get the following two possible cases −

- The **frequency** of the received signal will **increase**, when the target moves towards the direction of the Radar.

- The **frequency** of the received signal will **decrease**, when the target moves away from the Radar.

Basic Radar uses the same Antenna for both transmission and reception of signals. We can use this type of Radar, when the target is stationary, i.e., not moving and / or when that Radar can be operated with pulse signal.

The Radar, which operates with continuous signal (wave) for detecting non-stationary targets, is called Continuous Wave Radar or simply CW Radar. This Radar requires two Antennas. Among which, one Antenna is used for transmitting the signal and the other Antenna is used for receiving the signal.

Block Diagram of CW Radar

The CW Doppler Radar contains two Antennas − transmitting Antenna and receiving Antenna.

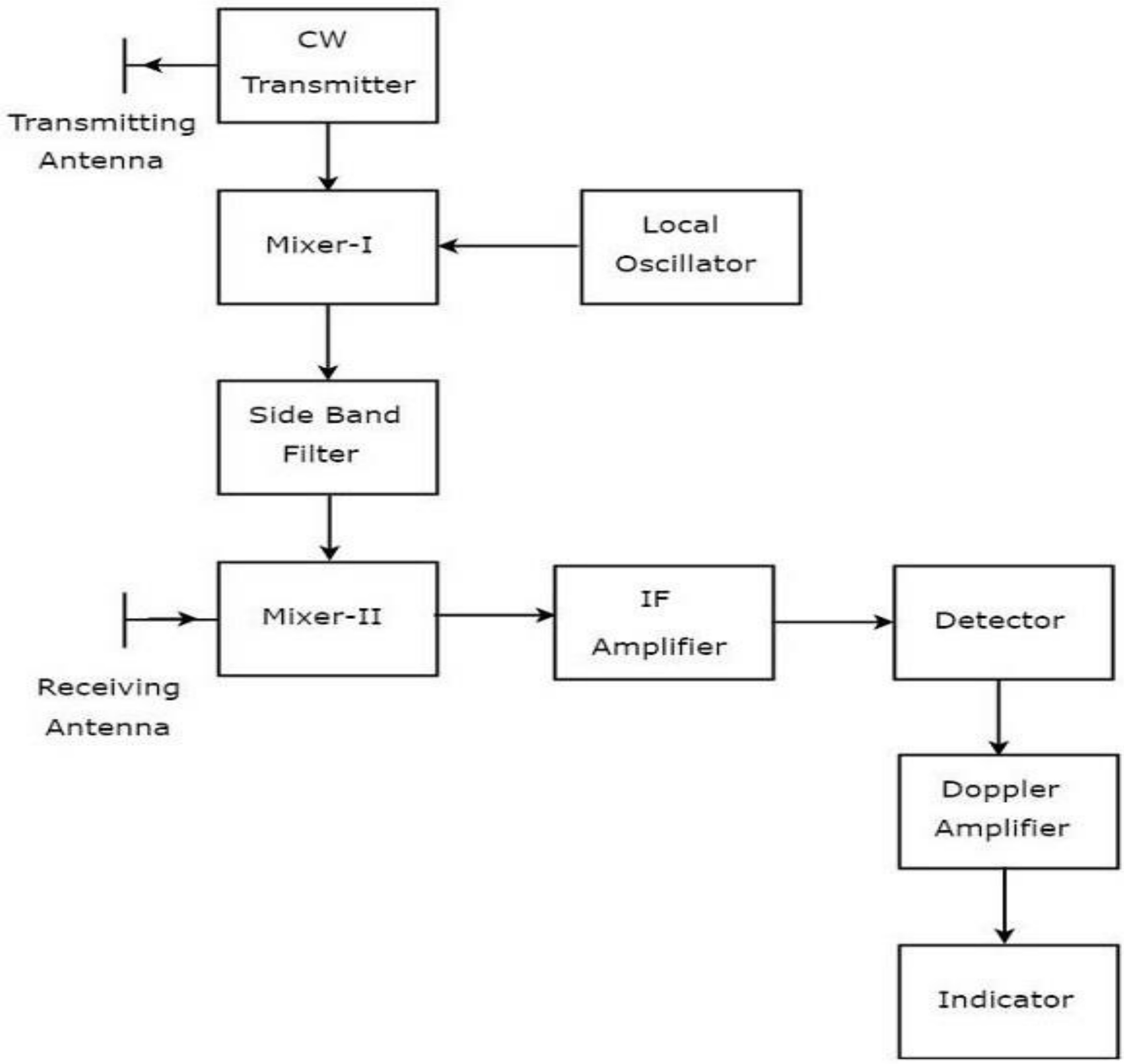

Fig:4 Block Diagram of CW Radar

The block diagram of CW Doppler Radar contains a set of blocks and the **function** of each block is mentioned below.

- **CW Transmitter** − It produces an analog signal having a frequency of fo. The output of CW Transmitter is connected to both transmitting Antenna and Mixer-I.
- **Local Oscillator** − It produces a signal having a frequency of fl. The output of Local Oscillator is connected to Mixer-I.
- **Mixer-I** − Mixer can produce both sum and difference of the frequencies that are applied to it. The signals having frequencies of fo and f_l are applied to Mixer-I. So, the Mixer-I will produce the output having frequencies fo+ f_l or fo− f_l.
- **Side Band Filter** − As the name suggests, side band filter allows a particular side band frequency − either upper side band frequencies or lower side band frequencies. The side band filter shown in the above figure produces only upper side band frequency, i.e., fo+ f_l.
- **Mixer-II** − Mixer can produce both sum and difference of the frequencies that are applied to it. The signals having frequencies of fo+ f_l and fo±fd are applied to Mixer-II. So, the Mixer-II will produce the output having frequencies of 2 fo+ f_l ±fd or f_l ±fd.
- **IF Amplifier** − IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure allows only the Intermediate Frequency, f_l±fd and amplifies it.
- **Detector** − It detects the signal, which is having Doppler frequency, fd.
- **Doppler Amplifier** − As the name suggests, Doppler amplifier amplifies the signal, which is having Doppler frequency, fd.
- **Indicator** − It indicates the information related relative velocity and whether the target is inbound or outbound.

CW Doppler Radars give accurate measurement of relative velocities. Hence, these are used mostly, where the information of velocity is more important than the actual range.

Plan position indicator

A plan position indicator (PPI) is a type of radar display that represents the radar antenna in the center of the display, with the distance from it and height above ground drawn as concentric circles. As the radar antenna rotates, a radial trace on the PPI sweeps in unison with it about the center point. It is the most common type of radar display. The radar antenna sends pulses while rotating 360 degrees around the radar site at a fixed elevation angle. It can then change angle or repeat at the same angle according to the need. Return echoes from targets are received by the antenna and processed by the receiver and the most direct display of those data is the PPI.

The height of the echoes increases with the distance to the radar, as represented in the adjacent image. This change is not a straight line but a curve as the surface of the Earth is curved and sinks below the radar horizon. For fixed-site installations, north is usually represented at the top of the image. For moving installations, such as small ship and aircraft radars, the top may represent the bow or nose of the ship or aircraft, i.e., its heading (direction of travel) and this is usually represented by a lubber line. Some systems may incorporate the input from a gyrocompass to rotate the display and once again display north as "up". Also, the signal represented is the reflectivity at only one elevation of the antenna, so it is possible to have many PPIs at one time, one for each antenna elevation.

The PPI is used in many domains involving display of range and positioning, especially in radars, including air traffic control, ship navigation, meteorology, on board ships and aircraft etc. PPI displays are also used to display sonar data, especially in underwater warfare. However, because the speed of sound in water is very slow compared to microwaves in air, a sonar PPI has an expanding circle that starts with each transmitted "ping" of sound. In meteorology, a competing display system is the CAPPI (Constant Altitude Plan Position Indicator) when a multi-angle scan is available. Using computers to process data, modern sonar and lidar installations can mimic radar PPI displays too.

Search Radars

Search radars scan great volumes of space with pulses of short radio waves. They typically scan the volume two to four times a minute. The waves are usually less than a meter long. Ships and planes are metal, and reflect radio waves. The radar measures the distance to the reflector by measuring the time of the roundtrip from emission of a pulse to reception, dividing this by two, and then multiplying by the speed of light. To be accepted, the received pulse has to lie within a period of time called the range gate. The radar determines the direction because the short radio waves behave like a search light when emitted from the reflector of the radar set's antenna.

- Early Warning (EW) Radar Radar Systems
    - Ground Control Intercept (GCI) Radar

- Airborne Early Warning (AEW)
    - Airborne ground surveillance (AGS)
    - Over-the-Horizon (OTH) Radar
- Target Acquisition (TA, TAR) Radar Systems
    - Surface-to-Air Missile (SAM) Systems
    - Anti-Aircraft Artillery (AAA) Systems
- Surface Search (SS) Radar Systems
    - Surface Search Radar
    - Coastal Surveillance Radar
    - Harbour Surveillance Radar
    - Antisubmarine Warfare (ASW) Radar
- Height Finder (HF) Radar Systems
- Gap Filler Radar System

Tracking Radar

The Radar, which is used to track the path of one or more targets is known as Tracking Radar. In general, it performs the following functions before it starts the tracking activity.

- Target detection
- Range of the target
- Finding elevation and azimuth angles
- Finding Doppler frequency shift

So, Tracking Radar tracks the target by tracking one of the three parameters — range, angle, Doppler frequency shift. Most of the Tracking Radars use the principle of tracking in angle.

Angular Tracking

The pencil beams of Radar Antenna perform tracking in angle. The axis of Radar Antenna is considered as the reference direction. If the direction of the target and reference direction is not same, then there will be angular error, which is nothing but the difference between the two directions.

If the angular error signal is applied to a servo control system, then it will move the axis of the Radar Antenna towards the direction of target. Both the axis of Radar Antenna and the direction of target will coincide when the angular error is zero. There exists a feedback mechanism in the Tracking Radar, which works until the angular error becomes zero.

Following are the two techniques, which are used in angular tracking.

- Sequential Lobing
- Conical Scanning

Sequential Lobing

If the Antenna beams are switched between two patterns alternately for tracking the target, then it is called sequential lobing. It is also called sequential switching and lobe switching. This technique is used to find the angular error in one coordinate. It gives the details of both magnitude and direction of angular error.

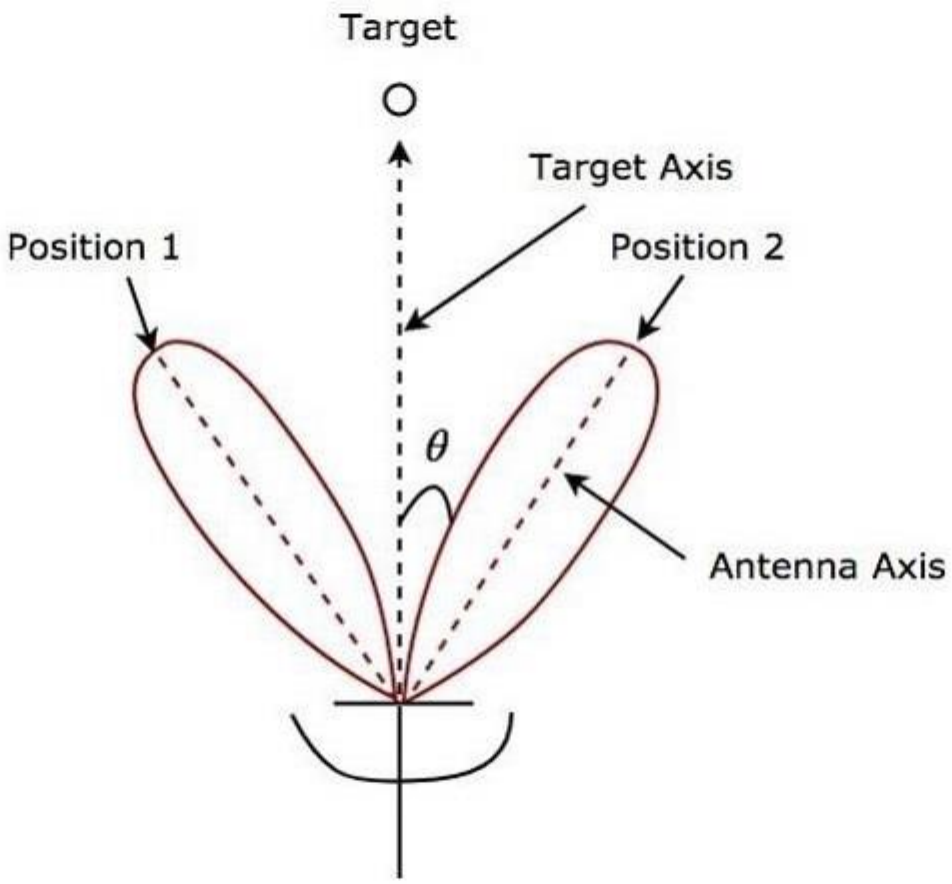

Fig:5 Example of Sequential Lobing in Polar Coordinates

As shown in the above figure, Antenna beams switch between Position 1 and Position 2 alternately. Angular error θ is indicated in the above figure. Sequential lobing gives the position of the target with high accuracy. This is the main advantage of sequential lobing.

Conical Scanning

If the Antenna beam continuously rotates for tracking a target, then it is called conical scanning. Conical scan modulation is used to find the position of the target.

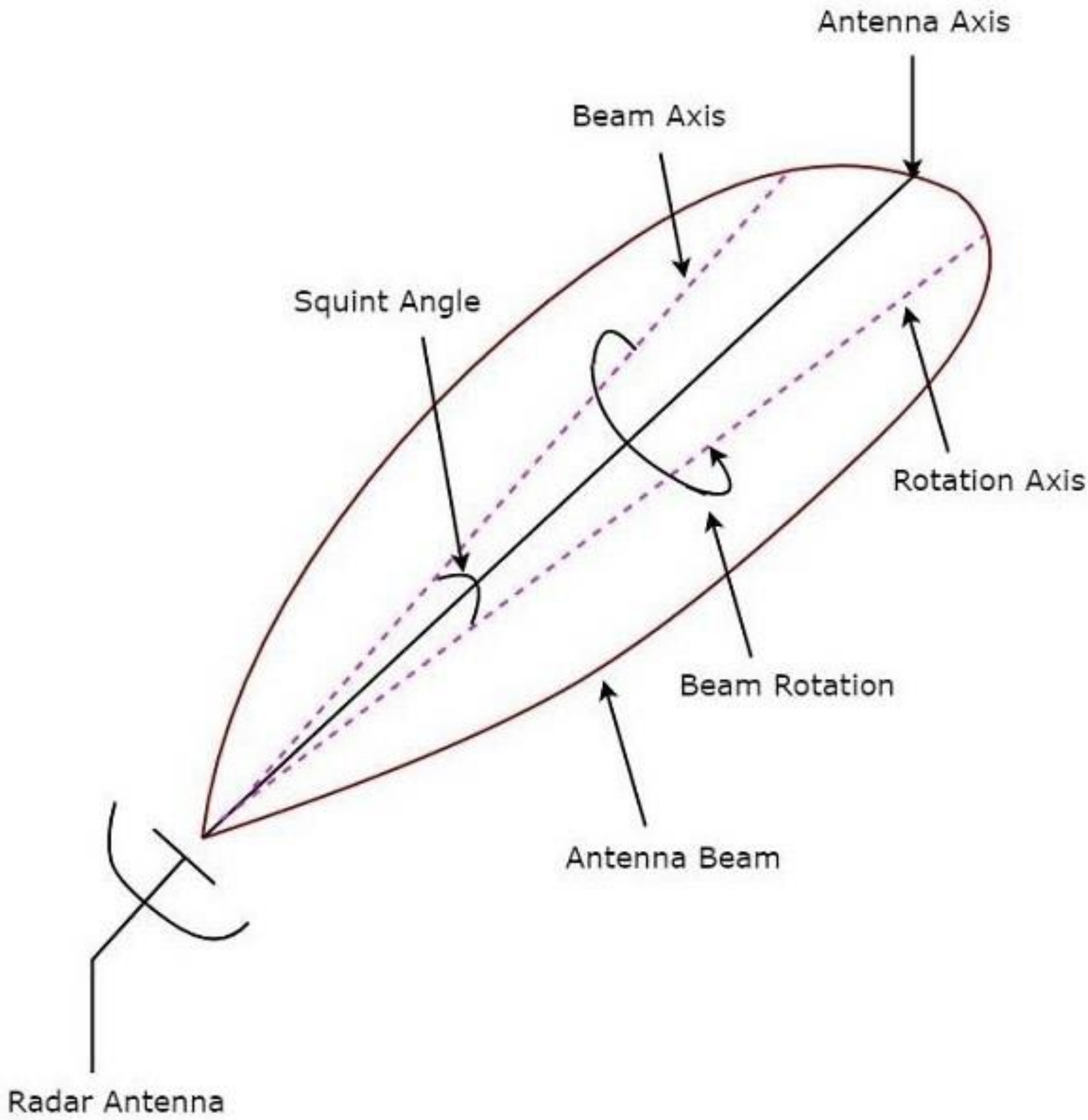

Fig:6 Example of Conical Scanning.

Squint angle is the angle between beam axis and rotation axis and it is shown in the above figure. The echo signal obtained from the target gets modulated at a frequency equal to the frequency at which the Antenna beam rotates.

The angle between the direction of the target and the rotation axis determines the amplitude of the modulated signal. So, the conical scan modulation has to be extracted from the echo signal and then it is to be applied to servo control system, which moves the Antenna beam axis towards the direction of the target.

If the Radar is used for detecting the movable target, then the Radar should receive only the echo signal due to that movable target. This echo signal is the desired one. However, in practical applications, Radar receives the echo signals due to stationary objects in addition to the echo signal due to that movable target.

The echo signals due to stationary objects (places) such as land and sea are called clutters because these are unwanted signals. Therefore, we have to choose the Radar in such a way that it considers only the echo signal due to movable target but not the clutters.

For this purpose, Radar uses the principle of Doppler Effect for distinguishing the non-stationary targets from stationary objects. This type of Radar is called Moving Target Indicator Radar or simply, MTI Radar. According to Doppler effect, the frequency of the received signal will increase if the target is moving towards the direction of Radar. Similarly, the frequency of the received signal will decrease if the target is moving away from the Radar.

Types of MTI Radars

The MTI Radars can be classified into the following two types based on the type of transmitter that has been used.

- MTI Radar with Power Amplifier Transmitter
- MTI Radar with Power Oscillator Transmitter

MTI Radar with Power Amplifier Transmitter

MTI Radar uses single Antenna for both transmission and reception of signals with the help of Duplexer.

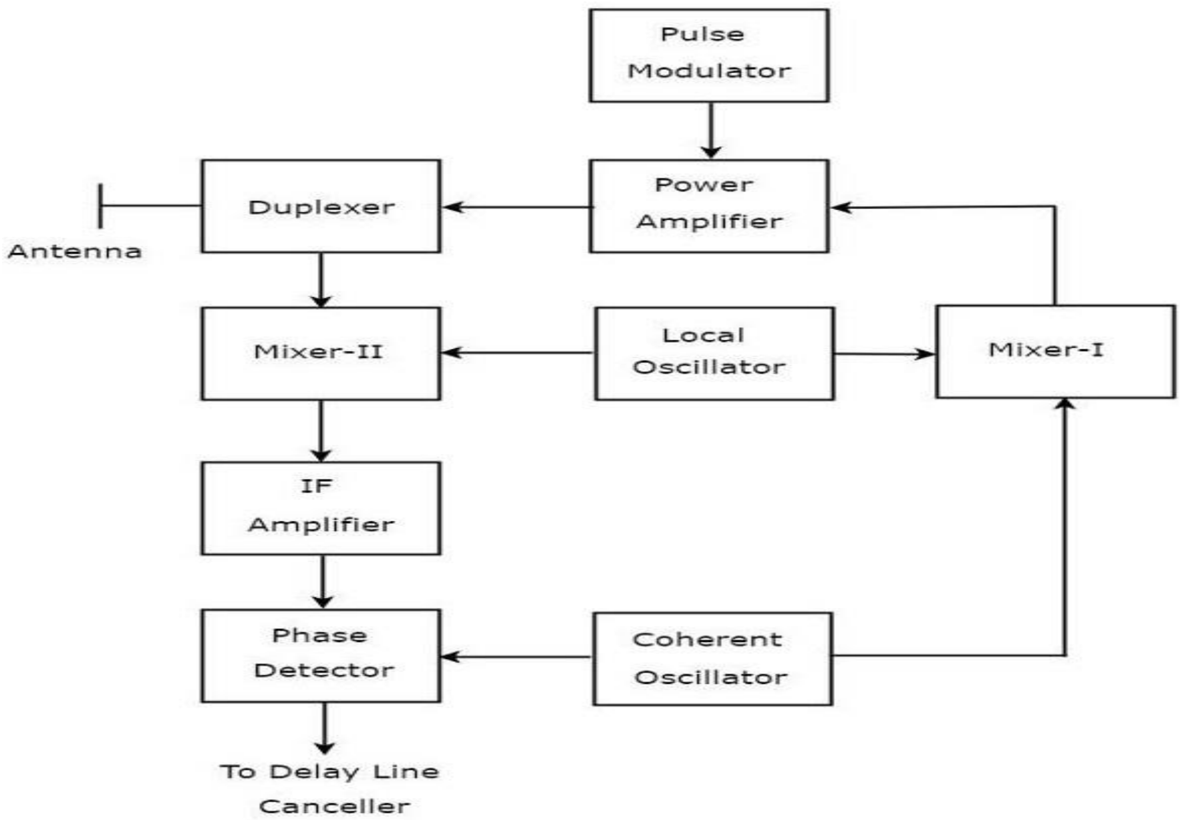

Fig:7 Block Diagram of MTI Radar with Power Amplifier Transmitter

The **function** of each block of MTI Radar with power amplifier transmitter is mentioned below.

- **Pulse Modulator** − It produces a pulse modulated signal and it is applied to Power Amplifier.
- **Power Amplifier** − It amplifies the power levels of the pulse modulated signal.
- **Local Oscillator** − It produces a signal having stable frequency f_l. Hence, it is also called stable Local Oscillator. The output of Local Oscillator is applied to both Mixer-I and Mixer-II.
- **Coherent Oscillator** − It produces a signal having an Intermediate Frequency, fc. This signal is used as the reference signal. The output of Coherent Oscillator is applied to both Mixer-I and Phase Detector.
- **Mixer-I** − Mixer can produce either sum or difference of the frequencies that are applied to it. The signals having frequencies of f_l and fc are applied to Mixer-I. Here, the Mixer-I is used for producing the output, which is having the frequency f_l +fc.
- **Duplexer** − It is a microwave switch, which connects the Antenna to either the transmitter section or the receiver section based on the requirement. Antenna transmits the signal having frequency f_l +fc when the duplexer connects the Antenna to power amplifier. Similarly, Antenna receives the signal having frequency of f_l+fc±fd when the duplexer connects the Antenna to Mixer-II.
- **Mixer-II** − Mixer can produce either sum or difference of the frequencies that are applied to it. The signals having frequencies f_l+fc±fd and f_l are applied to Mixer-II. Here, the Mixer-II is used for producing the output, which is having the frequency fc±fd.
- **IF Amplifier** − IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure amplifies the signal having frequency fc+fd. This amplified signal is applied as an input to Phase detector.
- **Phase Detector** − It is used to produce the output signal having frequency fdfd from the applied two input signals, which are having the frequencies of fc+fd and fc. The output of phase detector can be connected to Delay line canceller.

MTI Radar with Power Oscillator Transmitter

The block diagram of MTI Radar with power oscillator transmitter looks similar to the block diagram of MTI Radar with power amplifier transmitter. The blocks corresponding to the receiver section will be same in both the block diagrams. Whereas, the blocks corresponding to the transmitter section may differ in both the block diagrams.

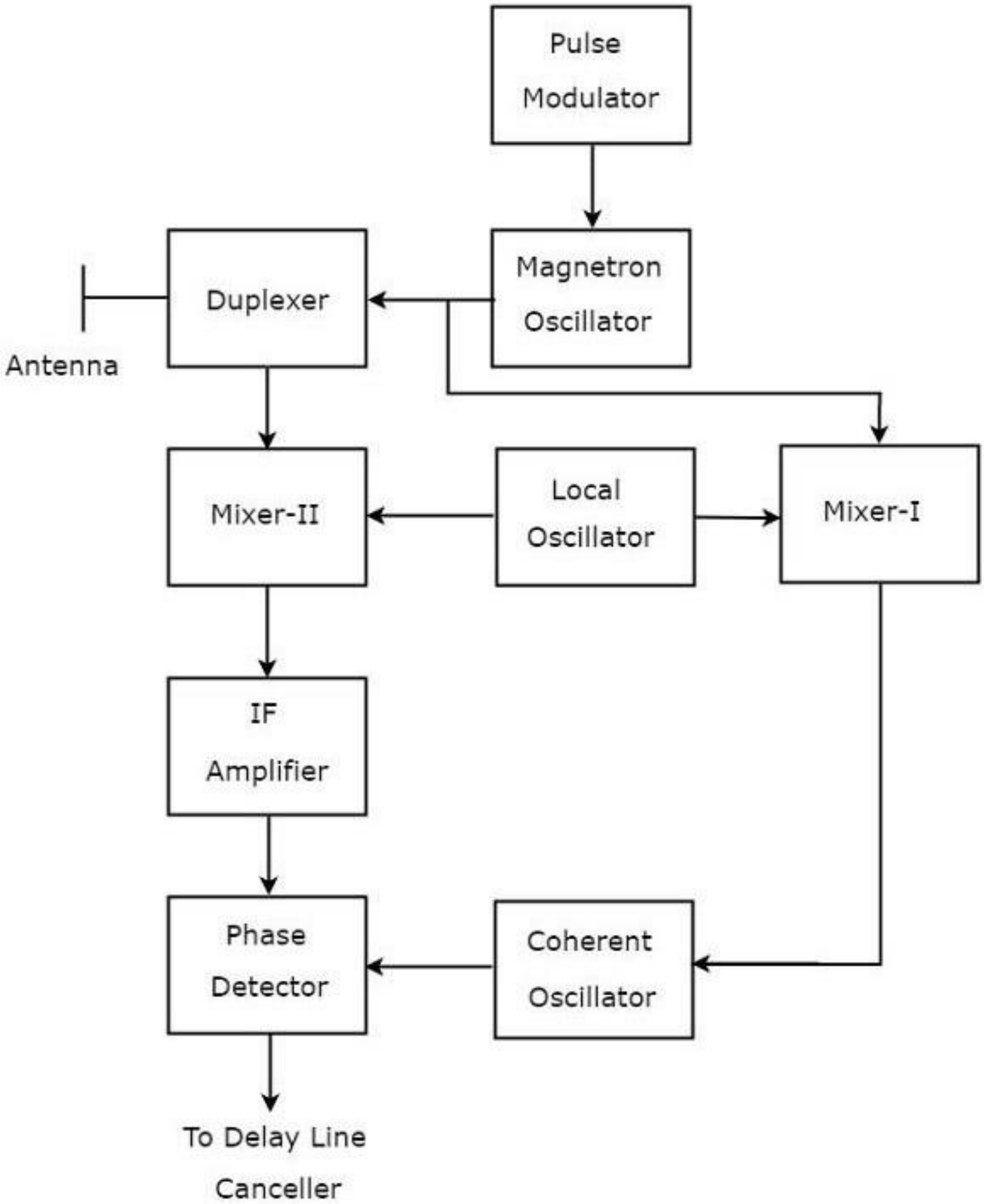

Fig:8 Block Diagram of MTI Radar with Power Oscillator Transmitter

As shown in the above figure, MTI Radar uses the single Antenna for both transmission and reception of signals with the help of Duplexer. The **operation** of MTI Radar with power oscillator transmitter is mentioned below.

- The output of Magnetron Oscillator and the output of Local Oscillator are applied to Mixer-I. This will further produce an **IF signal**, the phase of which is directly related to the phase of the transmitted signal.
- The output of Mixer-I is applied to the Coherent Oscillator. Therefore, the phase of Coherent Oscillator output will be **locked** to the phase of IF signal. This means, the phase of Coherent Oscillator output will also directly relate to the phase of the transmitted signal.
- So, the output of Coherent Oscillator can be used as reference signal for comparing the received echo signal with the corresponding transmitted signal using **phase detector**.

The above tasks will be repeated for every newly transmitted signal.

Radar Beacons

A Radar Beacons is a small radar set consisting of a receiver, a separate transmitter and an antenna which is often omnidirectional. When another radar transmits a coded set of pulses at the beacon, i.e., interrogates it, the beacon responds by sending back its specific pulse code. The pulses from the beacon, or transponder as it is often called, may be at the same frequency as those from the interrogating radar, in which case they are received by the main station together with its echo pulses. They may alternatively be at a special beacon frequency, in which case a separate receiver is required by the interrogating radar. Note that the beacon does not transmit pulses continuously in the same way as a search or tracking radar but only responds to the correct interrogation.

Applications

One of the functions of a beacon may be to identify itself. The beacon may be installed on a target, such as an aircraft, and will transmit a specific pulse code when interrogated. These pulses then appear on the PPI of the interrogating radar and inform it of the identity of the target. The system is in use in airport traffic control and also for military purposes, where it is called identification, friend or foe (IFF).

Another use of radar beacons is rather similar to that of lighthouses, except that radar beacons can operate over much larger distances. An aircraft or ship, having interrogated a number of beacons of whose exact locations it may be unaware (on account of being slightly lost), can calculate its position from the coded replies accurately and automatically. The presence of a beacon on a target increases enormously the distance over which a target may be tracked. Such active tracking gives much greater range than the passive tracking so far described, because the power transmitted by the beacon (modest though it normally is) is far in excess of the power that this target would have reflected had it not carried a beacon.

Introduction to Light

Light is defined as the electromagnetic radiation with wavelengths between 380 and 750 nm which is visible to the human eye. Electromagnetic radiation, such as light, is generated by changes in movement (vibration) of electrically charged particles, such as parts of 'heated' molecules, or electrons in atoms (both processes play a role in the glowing filament of incandescent lamps, whereas the latter occurs in fluorescent lamps). Electromagnetic radiation extends from γ rays and X-rays through to radio waves and to the long radio waves

Fiber Optics

An optical fiber can be understood as a dielectric waveguide, which operates at optical frequencies. The device **or** a tube, if bent or if terminated to radiate energy, is called a waveguide, in general.

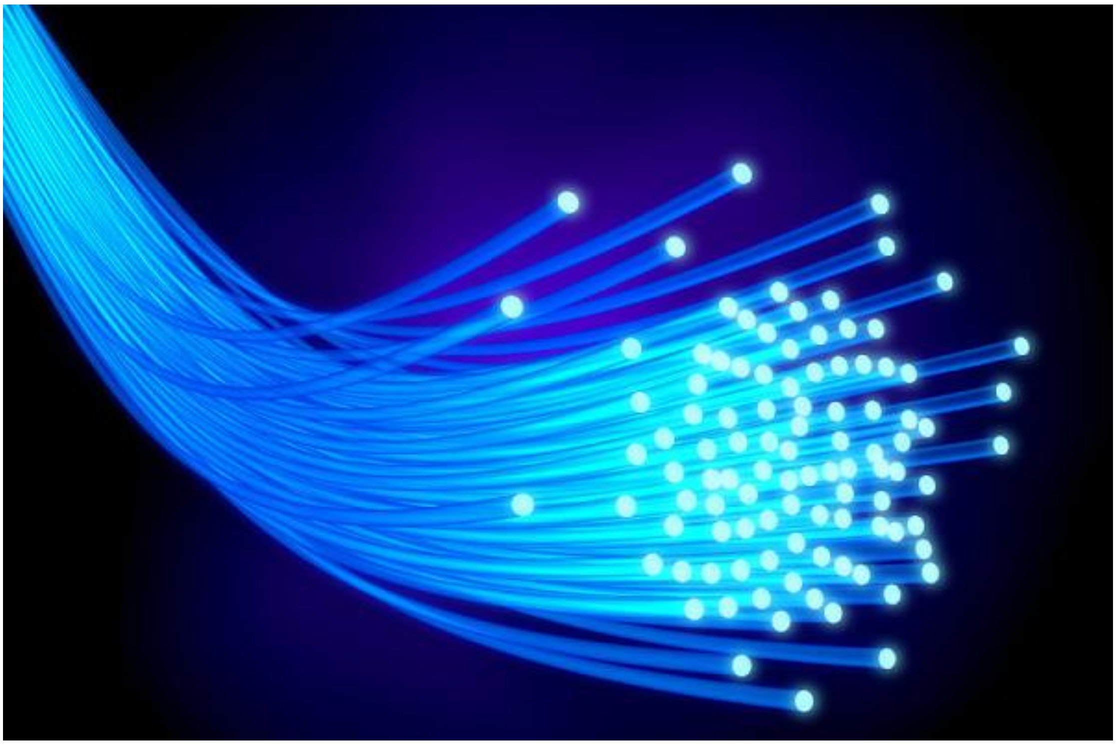

Fig:9 A Bunch of fiber optic cables

The electromagnetic energy travels through it in the form of light. The light propagation, along a waveguide can be described in terms of a set of guided electromagnetic waves, called as modes of the waveguide.

Optical Fiber?

Optical fiber is the technology associated with data transmission using light pulses travelling along with a long fiber which is usually made of plastic or glass. Metal wires are preferred for transmission in optical fiber communication as signals travel with fewer damages. Optical fibers are also unaffected by electromagnetic interference. The fiber optical cable uses the application of total internal reflection of light. The fibers are designed such that they facilitate the propagation of light along with the optical fiber depending on the requirement of power and distance of transmission. Single-mode fiber is used for long-distance transmission, while multimode fiber is used for shorter distances. The outer cladding of these fibers needs better protection than metal wires.

OPTICAL COMMUNICATION SYSTEM

An optical fiber communication system is similar in basic concept to any type of communication system. A block schematic of a general communication system is shown in Figure 10 (a), the function of which is to convey the signal from the information source over the transmission medium to the destination. The communication system therefore consists of a transmitter or modulator linked to the information source, the transmission medium, and a receiver or demodulator at the destination point. In electrical communications the information source provides an electrical signal, usually derived from a message signal which is not electrical (e.g. sound), to a transmitter comprising electrical and electronic components which converts the signal into a suitable form for propagation over the transmission medium. This is often achieved by modulating a carrier, which, as mentioned previously, may be an electromagnetic wave.

The transmission medium can consist of a pair of wires, a coaxial cable or a radio link through free space down which the signal is transmitted to the receiver, where it is transformed into the original electrical information signal (demodulated) before being passed to the destination. However, it must be noted that in any transmission medium the signal is attenuated, or suffers loss, and is subject to degradations due to contamination by random signals and noise, as well as possible distortions imposed by mechanisms within the medium itself. Therefore, in any communication system there is a maximum permitted distance between the transmitter and the receiver beyond which the system effectively ceases to give intelligible communication. For long-haul applications, these factors necessitate the installation of repeaters or line amplifiers.

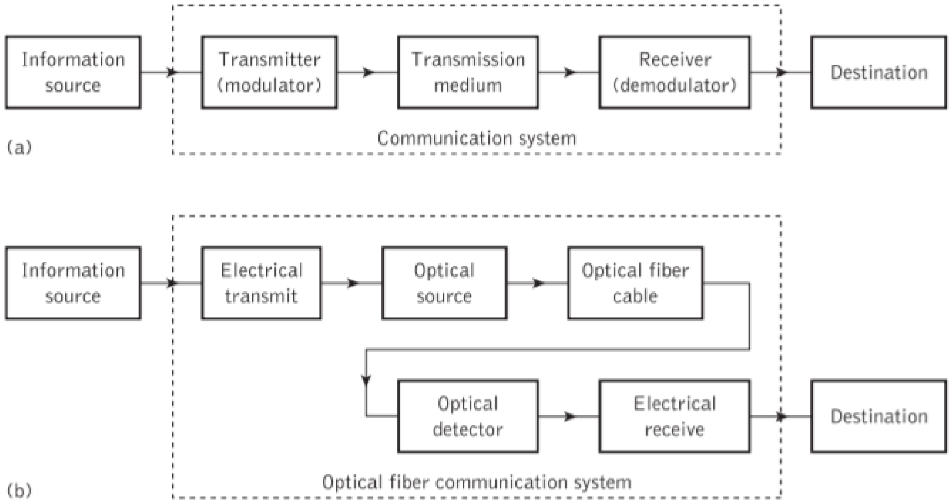

Fig:10 (a) The general communication system. (b) The optical fiber communication system

For optical fiber communications, the system components are given in Figure 10 (b). In this case, the information source provides an electrical signal to a transmitter comprising an electrical stage, which drives an optical source to give modulation of the light wave carrier. The optical source, which provides the electrical–optical conversion, may be either a semiconductor laser or light-emitting diode (LED). The transmission medium consists of an optical fiber cable and the receiver consists of an optical detector, which drives a further electrical stage and hence provides demodulation of the optical carrier. Photodiodes (*p–n, p–i–n* or avalanche) and, in some instances, phototransistors and photoconductors are utilized for the detection of the optical signal and the optical–electrical conversion.

The optical carrier may be modulated using either an analog or digital information signal. In the system shown in Figure 10 (b), analog modulation involves the variation of the light emitted from the optical source in a continuous manner. With digital modulation, however, discrete changes in the light intensity are obtained (i.e. on–off pulses). Although often simpler to implement, analog modulation with an optical fiber communication system is less efficient, requiring a far higher signal-to-noise ratio at the receiver than digital modulation. Also, the linearity needed for analog modulation is not always provided by semiconductor optical sources, especially at high modulation frequencies. For these reasons, analog optical fiber communication links are generally limited to shorter distances and lower bandwidth operation than digital optical links.

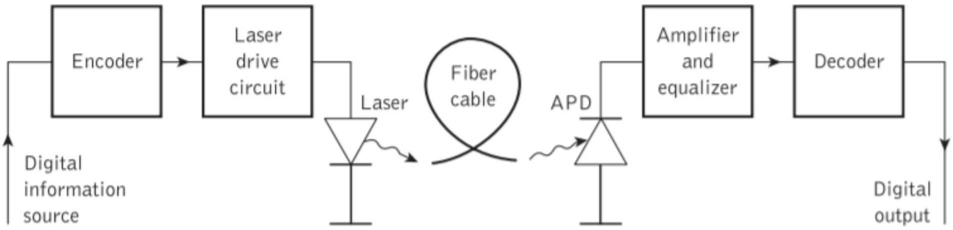

Fig:11 A digital optical fiber link using a semiconductor laser source and an avalanche photodiode (APD) detector.

Figure 11, shows a block schematic of a typical digital optical fiber link. Initially, the input digital signal from the information source is suitably encoded for optical transmission. The laser drive circuit directly modulates the intensity of the semiconductor laser with the encoded digital signal. Hence a digital optical signal is launched into the optical fiber cable.

The avalanche photodiode (APD) detector is followed by a front-end amplifier and equalizer or filter to provide gain as well as linear signal processing and noise bandwidth reduction. Finally, the signal obtained is decoded to give the original digital information.

Advantages of Optical Fiber Communication

Communication using an optical carrier wave guided along a glass fiber has a number of extremely attractive features, several of which were apparent when the technique was originally conceived. Furthermore, the advances in the technology to date have surpassed even the most optimistic predictions, creating additional advantages. Hence it is useful to consider the merits and special features offered by optical fiber communications over more conventional electrical communications. In this context we commence with the originally foreseen advantages and then consider additional features which have become apparent as the technology has been developed.

Enormous potential bandwidth: The optical carrier frequency in the range 10^{13} to 10^{16} Hz (generally in the near infrared around 10^{14} Hz or 10^{5} GHz yields a far greater potential transmission bandwidth than metallic cable systems (i.e. coaxial cable bandwidth typically around 20 MHz over distances up to a maximum of 10 km) or even millimeter wave radio systems (i.e. systems currently operating with modulation bandwidths of 700 MHz over a few hundreds of meters). Indeed, by the year 2000 the typical bandwidth multiplied by length product for an optical fiber link incorporating fiber amplifiers was 5000 GHz km in comparison with the typical bandwidth–length product for coaxial cable of around 100 MHz km. Hence at this time optical fiber was already demonstrating a factor of 50 000 bandwidth improvement over coaxial cable while also providing this superior information-carrying capacity over much longer transmission distances [16].

Small size and weight: Optical fibers have very small diameters which are often no greater than the diameter of a human hair. Hence, even when such fibers are covered with protective coatings they are far smaller and much lighter than corresponding copper cables. This is a tremendous boon towards the alleviation of duct congestion in cities, as well as allowing for an expansion of signal transmission within mobiles such as aircraft, satellites and even ships.

Electrical isolation: Optical fibers which are fabricated from glass, or sometimes a plastic polymer, are electrical insulators and therefore, unlike their metallic counterparts, they do not exhibit earth loop and interface problems. Furthermore, this property makes optical fiber transmission ideally suited for communication in electrically hazardous environments as the fibers create no arcing or spark hazard at abrasions or short circuits.

Immunity to interference and crosstalk: Optical fibers form a dielectric waveguide and are therefore free from electromagnetic interference (EMI), radio-frequency interference (RFI), or switching transients giving electromagnetic pulses (EMPs). Hence the operation of an optical fiber communication system is unaffected by transmission through an electrically noisy environment and the fiber cable requires no shielding from EMI. The fiber cable is also not susceptible to lightning strikes if used overhead rather than underground. Moreover, it is fairly easy to ensure that there is no optical interference between fibers and hence, unlike communication using electrical conductors, crosstalk is negligible, even when many fibers are cabled together.

Signal security: The light from optical fibers does not radiate significantly and there- fore they provide a high degree of signal security. Unlike the situation with copper cables, a transmitted optical signal cannot be obtained from a fiber in a noninvasive manner (i.e. without drawing optical power from the fiber). Therefore, in theory, any attempt to acquire a message signal transmitted optically may be detected. This feature is obviously attractive for military, banking and general data transmission (i.e. computer network) applications.

Low transmission loss: The development of optical fibers over the last 20 years has resulted in the production of optical fiber cables which exhibit very low attenuation or transmission loss in comparison with the best copper conductors. Fibers have been fabricated with losses as low as 0.15 dB km^{-1} and this feature has become a major advantage of optical fiber communications. It facilitates the implementation of communication links with extremely wide optical repeater or amplifier spacings, thus reducing both system cost and complexity. Together with the already proven modulation bandwidth capability of fiber cables, this property has provided a totally compelling case for the adoption of optical fiber communications in the majority of long-haul telecommunication applications, replacing not only copper cables, but also satellite communications, as a consequence of the very noticeable delay incurred for voice transmission when using this latter approach.

Ruggedness and flexibility: Although protective coatings are essential, optical fibers may be manufactured with very high tensile strengths. Perhaps surprisingly for a glassy substance, the fibers may also be bent to quite small radii or twisted without damage. Furthermore, cable structures have been developed which have proved flexible, compact and extremely rugged. Taking the size and weight advantage into account, these optical fiber cables are generally superior in terms of storage, transportation, handling and installation to corresponding copper cables, while exhibiting at least comparable strength and durability.

System reliability and ease of maintenance: These features primarily stem from the low-loss property of optical fiber cables which reduces the requirement for intermediate repeaters or line amplifiers to boost the transmitted signal strength. Hence with fewer optical repeaters or amplifiers, system reliability is generally enhanced in comparison with conventional electrical conductor systems. Furthermore, the reliability of the optical components is no longer a problem with predicted lifetimes of 20 to 30 years being quite common. Both these factors also tend to reduce maintenance time and costs.

Potential low cost: The glass, which generally provides the optical fiber transmission medium, is made from sand, which is not a scarce resource. So, in comparison with copper conductors, optical fibers offer the potential for low-cost line communication. Overall system costs when utilizing optical fiber communication on long-haul links, however, are substantially less than those for equivalent electrical line systems because of the low-loss and wideband properties of the optical transmission medium. The requirement for intermediate repeaters and the associated electronics is reduced, giving a substantial cost advantage. Although this cost benefit gives a net gain for long- haul links, it is not always the case in short-haul applications where the additional cost incurred, due to the electrical–optical conversion (and vice versa), may be a deciding factor.

The reducing costs of optical fiber communications has provided strong competition not only with electrical line transmission systems, but also for microwave and millimeter wave radio transmission systems. Although these systems are reasonably wideband, the relatively short-span ‘line of sight’ transmission necessitates expensive aerial towers at intervals no greater than a few tens of kilometers. Hence, with the exception of the telecommunication access network due primarily to current first installed cost constraints, optical fiber has become the dominant transmission medium within the major industrialized societies. Many advantages are therefore provided by the use of a light wave carrier within a transmission medium consisting of an optical fiber.

Parts of a Fiber

The most commonly used optical fiber is single solid di-electric cylinder of radius a and index of refraction n_1. The following figure explains the parts of an optical fiber.

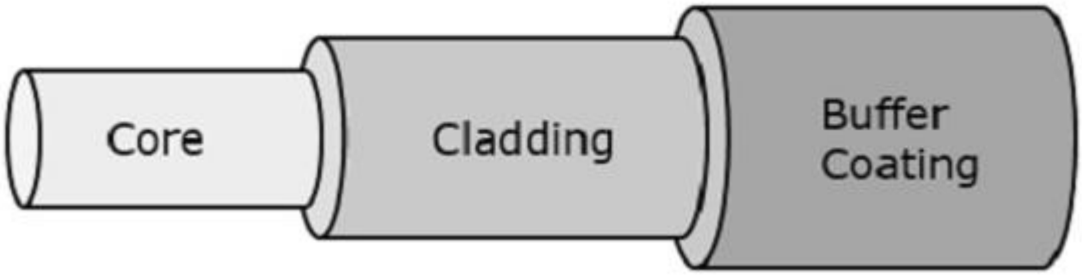

Fig:12 Parts of an Optical Fiber

This cylinder is known as the **Core** of the fiber. A solid di-electric material surrounds the core, which is called as **Cladding**. Cladding has a refractive index n_2 which is less than n_1.

Cladding helps in −

- Reducing scattering losses.
- Adds mechanical strength to the fiber.
- Protects the core from absorbing unwanted surface contaminants.

Types of Optical Fibers

Depending upon the material composition of the core, there are two types of fibers used commonly. They are −

- **Step-index fiber** − The refractive index of the core is uniform throughout and undergoes an abrupt change (or step) at the cladding boundary.
- **Graded-index fiber** − The core refractive index is made to vary as a function of the radial distance from the center of the fiber.

Both of these are further divided into −

- **Single-mode fiber** − These are excited with laser.
- **Multi-mode fiber** − These are excited with LED.

Step index fibers

The optical fiber considered in the preceding sections with a core of constant refractive index *n*1 and a cladding of a slightly lower refractive index *n*2is known as step index fiber. This is because the refractive index profile for this type of fiber makes a step change at the core–cladding interface, as indicated in Figure 13, which illustrates the two major types of step index fiber.

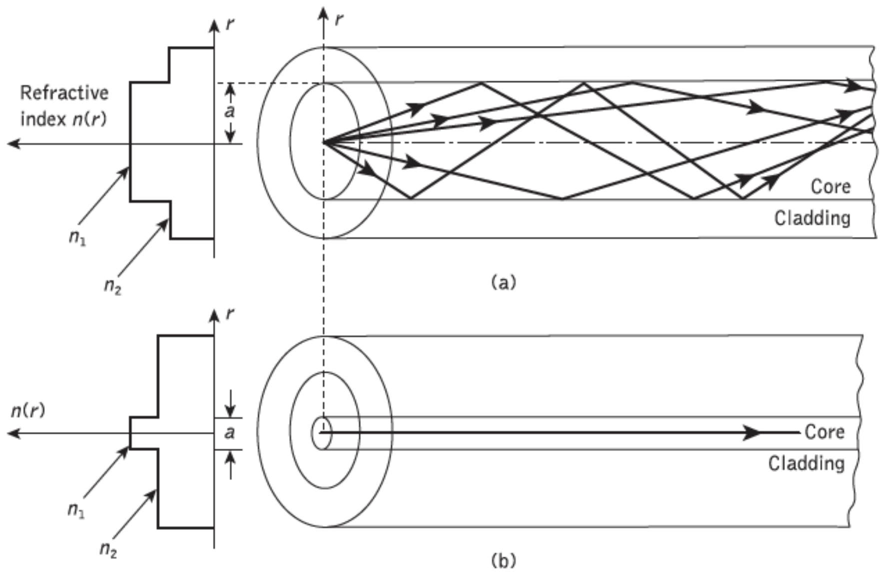

Fig:13 The refractive index profile and ray transmission in step index fibers:
(a) multimode step index fiber; (b) single-mode step index fiber

The refractive index profile for both single mode and multimode step-index fibers may be defined as:

$$n(r) = \begin{cases} n_1 & r < a \quad \text{(core)} \\ n_2 & r \geq a \quad \text{(cladding)} \end{cases}$$

Figure 13 (a) shows a multimode step index fiber with a core diameter of around 50µm or greater, which is large enough to allow the propagation of many modes within the fiber core. This is illustrated in Figure 13(a) by the many different possible ray paths through the fiber. Figure 13 (b) shows a single-mode or mono mode step index fiber which allows the propagation of only one transverse electromagnetic mode (typically HE11), and hence the core diameter must be of the order of 2 to 10µm. The propagation of a single mode is illustrated in Figure 13

(b) as corresponding to a single ray path only (usually shown as the axial ray) through the fiber.

The single-mode step index fiber has the distinct advantage of low intermodal dispersion (broadening of transmitted light pulses), as only one mode is transmitted, whereas with multimode step index fiber considerable dispersion may occur due to the differing group velocities of the propagating modes. This in turn restricts the maximum bandwidth attainable with multimode step index fibers, especially when com- pared with single-mode fibers. However, for lower bandwidth applications multimode fibers have several advantages over single-mode fibers. These are:

- The use of spatially incoherent optical sources (e.g. most light-emitting diodes) which cannot be efficiently coupled to single-mode fibers.
- Larger numerical apertures, as well as core diameters, facilitating easier coupling to optical sources
- Lower tolerance requirements on fiber connectors

Multimode step index fibers allow the propagation of a finite number of guided modes along the channel. The number of guided modes is dependent upon the physical parameters (i.e. relative refractive index difference, core radius) of the fiber and the wavelengths of the transmitted light which are included in the normalized frequency V for the fiber. The total number of guided modes or mode volume Ms for a step index fiber is related to the V value for the fiber by the approximate expression

$$M_s \simeq \frac{V^2}{2}$$

which allows an estimate of the number of guided modes propagating in a particular multimode step index fiber.

Graded index fibers

Graded index fibers do not have a constant refractive index in the core but a decreasing core index n(r) with radial distance from a maximum value of n1 at the axis to a constant value n2 beyond the core radius a in the cladding.

This index variation may be represented as:

$$n(r) = \begin{cases} n_1(1 - 2\Delta(r/a)^{\alpha})^{\frac{1}{2}} & r < a \quad \text{(core)} \\ n_1(1 - 2\Delta)^{\frac{1}{2}} = n_2 & r \geq a \quad \text{(cladding)} \end{cases}$$

where D is the relative refractive index difference and α is the profile parameter which gives the characteristic refractive index profile of the fiber core. Equation above which is a convenient method of expressing the refractive index profile of the fiber core as a variation of α, allows representation of the step index profile when α = ∞, a parabolic profile when α = 2 and a triangular profile when α = 1. This range of refractive index profiles is illustrated in Figure 14.

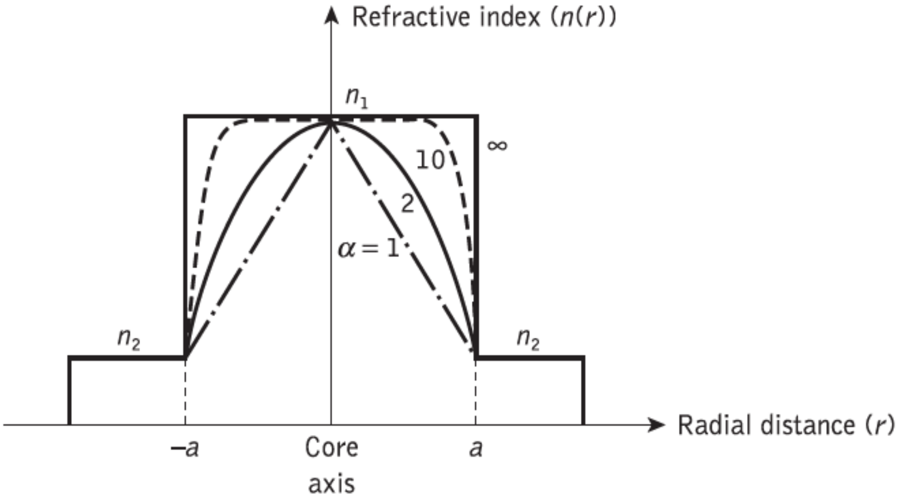

Fig:14 Possible fiber refractive index profiles for different values of α given

The graded index profiles which at present produce the best results for multimode optical propagation have a near parabolic refractive index profile core (α value around 2). Fibers with such core index profiles are well established and consequently when the term 'graded index' is used without qualification it usually refers to a fiber with this profile. For this reason in this section we consider the waveguiding properties of graded index fiber with a parabolic refractive index profile core.

A multimode graded index fiber with a parabolic index profile core is illustrated in Figure 15. It may be observed that the meridional rays shown appear to follow curved paths through the fiber core. Using the concepts of geometric optics, the gradual decrease in refractive index from the center of the core creates many refractions of the rays as they are effectively incident on a large number or high to low index interfaces. This mechanism is illustrated in Figure 15, where a ray is shown to be gradually curved, with an ever-increasing angle of incidence, until the conditions for total internal reflection are met, and the ray travels back towards the core axis, again being continuously refracted.

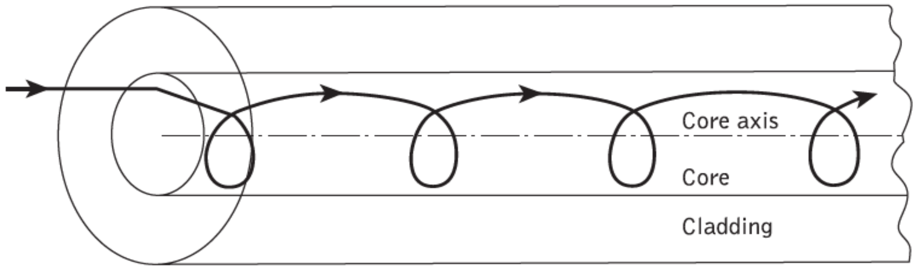

Fig:15 A helical skew ray path within a graded index fiber

Multimode graded index fibers exhibit far less intermodal dispersion than multimode step index fibers due to their refractive index profile. Although many different modes are excited in the graded index fiber, the different group velocities of the modes tend to be normalized by the index grading. Again considering ray theory, the rays traveling close to the fiber axis have shorter paths when compared with rays which travel. However, the near axial rays are transmitted through a region of higher refractive index and therefore travel with a lower velocity than the more extreme rays. This compensates for the shorter path lengths and reduces dispersion in the fiber. Hence, multi-mode graded index fibers with parabolic or near-parabolic index profile cores have transmission bandwidth which may be orders of magnitude greater than multimode step index fiber bandwidths. Consequently, although they are not capable of the bandwidths attainable with single-mode fibers, such multimode graded index fibers have the advantage of large core diameters (greater than 30 μm) coupled with bandwidths suitable for long- distance communication.

Single-mode fiber

The advantage of the propagation of a single mode within an optical fiber is that the signal dispersion caused by the delay differences between different modes in a multimode fiber is eliminated. Hence, for the transmission of a single mode, the fiber must be designed to allow propagation of only one mode, while all other modes are attenuated by leakage or absorption. Following the preceding discussion of multimode fibers, this may be achieved through choice of a suitable normalized frequency for the fiber. For single-mode operation, only the fundamental LP01 mode can exist. Thus single-mode propagation of the LP01 mode in step index fibers is possible over the range:

$$0 \leq V < 2.405$$

as there is no cutoff for the fundamental mode. It must be noted that there are in fact two modes with orthogonal polarization over this range, and the term single-mode applies to propagation of light of a particular polarization.

Optical Fiber Communications

The communication system of fiber optics is well understood by studying the parts and sections of it. The major elements of an optical fiber communication system are shown in the figure 16. The basic components are light signal transmitter, the optical fiber, and the photo detecting receiver. The additional elements such as fiber and cable splicers and connectors, regenerators, beam splitters, and optical amplifiers are employed to improve the performance of the communication system.

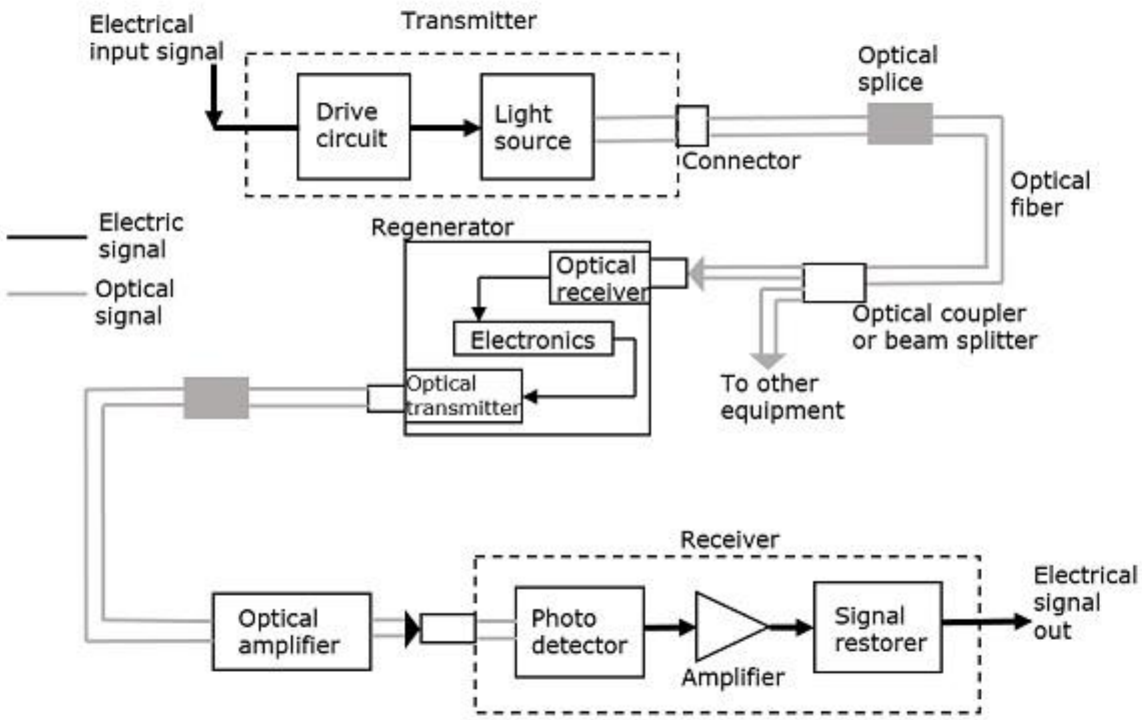

Fig:16 Elements of an Optical Fiber Communication

Functional Advantages

- The transmission bandwidth of the fiber optic cables is higher than the metal cables.
- The amount of data transmission is higher in fiber optic cables.
- The power loss is very low and hence helpful in long-distance transmissions.
- Fiber optic cables provide high security and cannot be tapped.
- Fiber optic cables are the most secure way for data transmission.
- Fiber optic cables are immune to electromagnetic interference.

- These are not affected by electrical noise.

Physical Advantages

- The capacity of these cables is much higher than copper wire cables.
- Though the capacity is higher, the size of the cable doesn't increase like it does in copper wire cabling system.
- The space occupied by these cables is much less.
- The weight of these FOC cables is much lighter than the copper ones.
- Since these cables are di-electric, no spark hazards are present.
- These cables are more corrosion resistant than copper cables, as they are bent easily and are flexible.
- The raw material for the manufacture of fiber optic cables is glass, which is cheaper than copper.
- Fiber optic cables last longer than copper cables.

Disadvantages

- Though fiber optic cables last longer, the installation cost is high.
- The number of repeaters are to be increased with distance.
- They are fragile if not enclosed in a plastic sheath. Hence, more protection is needed than copper ones.

Applications of Fiber Optics

- Used in telephone systems
- Used in sub-marine cable networks
- Used in data link for computer networks, CATV Systems
- Used in CCTV surveillance cameras
- Used for connecting fire, police, and other emergency services.
- Used in hospitals, schools, and traffic management systems.
- They have many industrial uses and also used for in heavy duty constructions.

Losses in optical fiber:

- Absorption loss
- Scattering loss
- Dispersion loss
- Radiation loss
- Coupling loss.

Absorption loss:

Absorption loss is related to the material composition and fabrication process of fiber. Absorption loss results in dissipation of some optical power as hear in the fiber cable. Although glass fibers are extremely pure, some impurities still remain as residue after purification. The amount of absorption by these impurities depends on their concentration and light wavelength. The absorption loss can be further divided into two groups which is discussed below.

Intrinsic absorption

Intrinsic absorption in the ultraviolet region is caused by electronic absorption bands. Basically, absorption occurs when a light particle (photon) interacts with an electron and excites it to a higher energy level. The main cause of intrinsic absorption in the infrared region is the characteristic vibration frequency of atomic bonds. In silica glass, absorption is caused by the vibration of silicon-oxygen (Si-O) bonds. The interaction between the vibrating bond and the electromagnetic field of the optical signal causes intrinsic absorption. Light energy is transferred from the electromagnetic field to the bond.

Extrinsic absorption

Extrinsic absorption is much more significant than intrinsic Caused by impurities introduced into the fiber material during manufacture – Iron, nickel, and chromium Caused by transition of metal ions to higher energy level Modern fabrication techniques can reduce impurity levels below 1 part in 1010. For some of the more common metallic impurities in silica fibre the table shows the peak attenuation wavelength and the attenuation caused by an impurity concentration of 1 in 109.

Radiative losses:

Radiative losses also called bending losses, occur when the fibre is curved. There are two types of radiative losses: Micro bending losses. Macro bending losses.

Scattering loss:

Basically, scattering losses are caused by the interaction of light with density fluctuations within a fiber. Density changes are produced when optical fibers are manufactured.

- **Linear Scattering Losses:** Linear scattering occurs when optical energy is transferred from the dominant mode of operation to adjacent modes. It is proportional to the input optical power injected into the dominant mode. o Linear scattering is divided into two categories: Mie scattering and Rayleigh scattering.

- **Non- Linear Scattering Losses:** Scattering loss in a fiber also occurs due to fiber non-linearity's i.e. if the optical power at the output of the fiber does not change proportionately with the power change at the input of the fiber, the optical fiber is said to be operating in the non-linear mode. Non- Linear scattering is divided into two categories: Stimulated Raman Scattering and Stimulated Brillouin Scattering.

Dispersion loss:

Dispersion is a measure of the temporal spreading that occurs when a light pulse propagates through an optical fiber. Dispersion is sometimes referred to as delay distortion in the sense that the propagation time delay causes the pulse to broaden.

Radiation loss:

When optical fibers are exposed to ionizing radiation such as energetic electrons, protons, neutrons, X-rays, Ɣ-radiation, etc., they undergo 'damage'. The term 'damage' primarily refers to the additional loss of the propagating optical signal leading to decreased power at the output end which could lead to premature failure of the component and or system.

Coupling Losses/Connector losses:

In fiber cables, coupling losses can occur at any of the following three types of optical junctions- light source to fiber connection, fiber to fiber connections and fiber to photo detector connections.

Installing and testing fiber-optic:

Although fiber-optic networks differ significantly from copper-based networks, the methods for testing and troubleshooting them are very similar. The techniques can easily be mastered by technicians with some basic training in fiber optics and network testing. To thoroughly test fiber-optic cabling plant requires that one should test it three times. One should test the cable on the reel for continuity before installing it to ensure that no damage was done in shipment from the manufacturer to the job site. It is generally sufficient just to test for continuity, since most fiber is installed without connectors and then terminated in place, and poor connectors are the most common problem to be uncovered by testing for loss.

After termination, test each segment of the cabling plant individually as it is installed, to ensure that each connector and cable is good. As a final check, test each end-to-end run, from the wiring closet to end-user equipment. Typical fiber-optic cabling plants are composed of a backbone cable connecting patch panels and several short jumper cables that connect the equipment to the cabling plant. Building and campus systems usually have the backbone fiber terminated in wiring closets; short jumpers connect to wall outlets or directly to the equipment. These installations often have no splices at all, since distances are short.

Testing the complete cabling plant is done per a standard test procedure, Optical Fiber Systems Test Procedures-14. This procedure covers the peculiarities of multimode fiber in detail. In fact, it was written to address the problems of controlling mode power distribution in multimode cables but, with the exception of mode power distribution errors, the same procedures apply to single mode fiber.

For multimode fibers, testing is usually done at both 850 and 1300 nanometers, using light-emitting diodes. This proves the performance of the cable for every data communications system, including fiber distributed data interface and enterprise systems connection, and meets the requirements of all network vendors. For single mode fiber cables, testing is usually done at 1300 nm. Sometimes 1550-nm testing is also required to show whether the cable can support wavelength-division multiplexing at 1300 and 1550 nm, for future service expansion. In addition, 1550-nm testing can show micro bending losses that are not obvious at 1300 nm, since the fibers are much more sensitive to bending losses at 1550 nm.

If cabling plant end-to-end loss exceeds total allowable loss, the best solution is to retest each segment of the cabling plant separately, checking suspect cables each way, since the problem is most likely a single bad connector or splice. If the cabling plant is long enough, an optical time-domain reflectometer may be used to find the problem. Bad connectors must then be repolished or replaced to get the loss within acceptable ranges.

The OTDRs (Optical Time Domain Reflectometer) were used for all testing of installed cabling plants. In fact, printouts or pictures of OTDR traces were kept on record for every fiber in every cable. Today, the power meter and source (or optical-loss test set) have replaced the reflectometer for most final qualification testing, since the direct loss test gives a more reliable test of end-to-end loss than does an OTDR.

Do not use an OTDR for measuring end-to-end loss. It does not accurately measure actual link loss as seen by the transmitters and receivers of the fiber-optic link. As normally used, the OTDR does not count the end connectors` loss because it uses a laser with very restricted mode power distribution, which minimizes the loss of the fiber and intermediate connectors. Finally, the difference in backscattering coefficients of various fibers leads to imprecise connector loss measurements.

However, you may have to use an OTDR to find bad splices or optical return loss problems in connectors and splices in single mode cabling plant. Only with a reflectometer can optical return loss problems be located for correction. Typical back reflection test sets give only the total amount of backscatter or return loss; they do not give information about the effects of individual components, which is necessary to locate and fix the problem.

If an OTDR has high enough resolution to record short, individual cable assemblies, it can also be used to find bad connectors or splices in a high-loss cabling plant. However, if the cables are too short or the splices too near the end of the fiber (as is often the case with pigtails spliced onto single mode fiber cables), the only way to localize the problem is to use a visual fault locator, preferably a high-powered, helium-neon laser type, which can shine through the jacket of typical yellow or orange PVC-jacketed single-fiber cables. This method of fault location is easiest with single-fiber cables that have yellow or orange jackets, because these are more translucent to laser light.

Testing networks

To test network cables, use a fiber-optic test kit, which includes a fiber-optic power meter and an LED or laser source. The source should be of the same type and wavelength as the transmitters used in the network being tested. Adapters provide the interface to the connectors used with the network. You will need test cables as launch and receive jumpers for testing the network cables, and a connector coupling kit to interconnect the test jumpers with the cables to be tested.

Connectors and cables should be handled with care. Do not bend cables too tightly, especially near the connectors, because sharp bends can break the fibers. Do not drop the connectors; they can be damaged by a blow to the optical face. Do not pull hard on the connectors because this might break the fiber in the back of the connector or cause pistoning if the bond between the fiber and the connector ferrule is broken.

If there is any question about the condition of the connectors, clean them before testing. Use a fiber-optic inspection microscope with appropriate stages to hold the connectors if there is any doubt about their cleanliness or physical condition. In the installation and testing of fiber-optic networks, a problem routinely encountered is incorrect fiber-optic connections. A fiber-optic link consists of two fibers transmitting in opposite directions to provide duplex communication. It is common to find fibers switched, with the transmit fiber connected to a transmitter and the receive fiber to a receiver.

To avoid this problem, use a visual tracer to verify the proper connections. The tracer can be a flashlight (although it is hard to hold the fiber in place to couple enough light to see it), a modified flashlight, or even a microscope that will hold the fiber steady and couple an adequate amount of power into it. A special test source using a bright red LED, like those used in plastic fiber links, can also serve as a tracer. Shine the visible light from the tracer down the fiber and track the fiber through the cables and patch panels to the far end.

With a tracer, you can trace fibers for up to 2.5 miles (4 kilometers). In addition to tracing fibers, you can use the device to check continuity and find broken fibers in cables. Another recommended use is to check the continuity of every fiber in multifiber cables before installation to ensure that all fibers are functional. Installing a cable with bad fibers can be an expensive proposition. Visual tracers are an inexpensive, valuable tool for every member of the installation crew.

For single mode cables, a more powerful tool is available --a high-power visible laser coupled to fiber. This visual fault locator uses a red laser --either helium-neon or diode --- with enough power to show breaks in the fiber through the jacket of the fiber. It is much more expensive than a simple fiber tracer, however.

Testing installed cabling plant

Fiber-optic networks are always specified to operate over a range of loss, typically called the system margin. Either too much or too little loss can be a problem. If loss is too high, the signal will be low at the receiver, causing a poor signal-to-noise condition in the receiver. If the loss is too low, the power level at the receiver will be too high, causing receiver saturation. Both these conditions will cause high bit-error rates in digital systems and poor analog signal performance.

Using a power meter and source, test the complete cabling plant for loss; include all individual jumper or trunk cables.Use the double-ended method, since system margin specifications include the loss of connectors on both ends of the fiber. If the end-to-end (transmitter-to-receiver) loss measurement for a given fiber is within the network margin specification, the data should be recorded for future reference. If the loss is too low, notation should be made that the fiber will probably need an inline attenuator to reduce receiver power to acceptable levels. If the loss is too high, it will be necessary to retest each link of the complete cable run to find the bad link.

Possible causes of high end-to-end link loss are bad connectors, bad splice bushings in patch panels, cables bent too tightly around corners, broken fibers in cables or even bad launch or receive cables or instruments. There are only two ways to find the problem: Test each segment of the cable individually, or use an OTDR if the lengths are long enough to be viewed with the limited resolution of this instrument.

Testing and troubleshooting networks

The installed network can be tested quickly and easily with a fiber-optic power meter. Set the network transmitter to transmit a clock output or other bit stream of known duty cycle. Set the power meter calibration on the proper wavelength and the reading units on watts. To test the received power --the most critical element in the network --disconnect the cable connector at the receiver, attach the power meter and measure the power.

If the receiver power is low, check the transmitter power by disconnecting the source jumper cable at the first available connector and measuring the power at that point. Alternatively, you can disconnect the cable at the transmitter and use a jumper that has been pre-tested and is known to be good to measure the coupled power. If the output is measured through a short network jumper cable (less than 30 feet), no compensation for jumper loss is necessary. For longer jumpers, some compensation for cable loss may be necessary.

If receiver power is low but transmitter power is high, something is wrong with the cables. They must be tested at every connection to isolate the bad cables or connectors. Starting from either the transmitter or receiver end, follow the network cables to every patch panel. Disconnect the connector and measure the power at each point. By making measurements in decibels, you can easily calculate the loss of the cable network to each point by subtracting successive readings.

If you note a larger-than-expected loss in the cable link, test the suspect cable using one or more of the following methods. If a cable has attenuation that is higher than specifications but still transmits light, check the connectors using a microscope to determine if they have been damaged and should be replaced. If the connectors look good, the best solution may be to replace the cable or switch to a spare. If a visual fault locator is available, use it to locate breaks in the fiber and find broken connectors. Under some circumstances, such as high loss in long jumper or trunk cables, an OTDR can be used to diagnose cable faults.

Transceiver loopback testing

The data communications capabilities of the network can be tested with a loopback test. This test uses a calibrated fiber-optic attenuator placed between the transmitter and receiver on a piece of equipment to see if it can transmit data to itself. Many types of network equipment have diagnostics to do loopback testing. This loopback method tests the transmitter and receiver of the unit under standard data transmission conditions over the specified link loss budget.

Some data communications equipment can also institute an electrical network loopback test, where the loopback path is inside the equipment, looping back over the entire data link to the equipment on the far end of the link. If both ends of the link pass a unit loopback test but fail a network loopback test, the problem is in the cables, which then need testing.

Once installation is complete, the cabling plant is tested and network equipment is running smoothly, what is likely to go wrong in a fiber-optic network? Fortunately, not much. One of the biggest selling points for fiber optics has been its reliability. But there are some potential problems that can be addressed by the end user.

The biggest problem with the cabling plant is "backhoe fade." This happens when someone mistakenly cuts or breaks the cable, either when an electrician is working on cables inside a building or when an underground cable is dug up. Inside buildings, use orange- or yellow-jacketed cable instead of black or gray to make the fiber-optic cable more visible and distinctive. Short distances make OTDRs unusable inside, so a visual fault locator is necessary.

Outdoors, the best defense is to mark where cables are buried and bury a marker tape above the cable, to alert workers to the presence of cable below. Outside cable faults are best found by using a reflectometer to localize the fault, then having personnel search the area for obvious damage. Another problem is broken cable just behind the connectors in patch panels. This is a difficult fault to find, but a visual fault locator is often the best way. Unless the jumper cables are quite long, a reflectometer won`t help at all.

Within the fiber-optic link, the most likely component to fail is the light-emitting diode or laser transmitter, because it is the most highly stressed component in the link. Lasers are feedback-stabilized to maintain a constant output power, so they tend to fail all at once. The power output of LEDs drops as they age, but the time frame is quite long --100,000 to 1 million hours. If there is no power at the receiver, the next place to check should be the transmitter LED or lasers, to isolate the problem to either the transmitter or the cabling plant. Receivers are low-stressed devices and highly reliable, but the electronics behind them can fail. If there is receiver power but no communications, a loopback test to see if the receiver is working is the best test of its status.

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