GOVERNMENT POLYTECHNIC BHUBANESWAR

LECTURENOTE

ON

ANALOG & DIGITAL COMMUNICATION –TH-3

FACULTY NAME- SOMA DASH, LECTURER, E&TC Deptd.

Branch-Electronics and Telecommunication Engg.
SEMESTER-5^{TH}

DEPARTMENT OF TELECOMMUNICATION ENGINEERING

ANALOG & DIGITAL COMMUNICATION

This subjects deals with different types of Analog & Digital Electronics Communication Systems includes basic processes, principles & methods of different Systems including Transmitters & Receivers for conveying messages/exchange information at a distance. 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 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 more clear and accurate without loss. The challenges in digital transmission was to deal with the increased bandwidth requirement of digital signals. Analog Communication is a data transmitting technique in a format that utilizes continuous signals to transmit data including voice, image, video, electrons etc. An analog signal is a variable signal continuous in both time and amplitude which is generally carried by use of modulation. Digital communications is any exchange of data that transmits the data in a digital form. Communications done over the Internet is a form of digital communication. A digital communication system is designed to transport a message from an information source through a transmission medium (i.e., channel) to an information sink. The goal is to accomplish this task such that the information is efficiently transmitted with a certain degree of reliability.

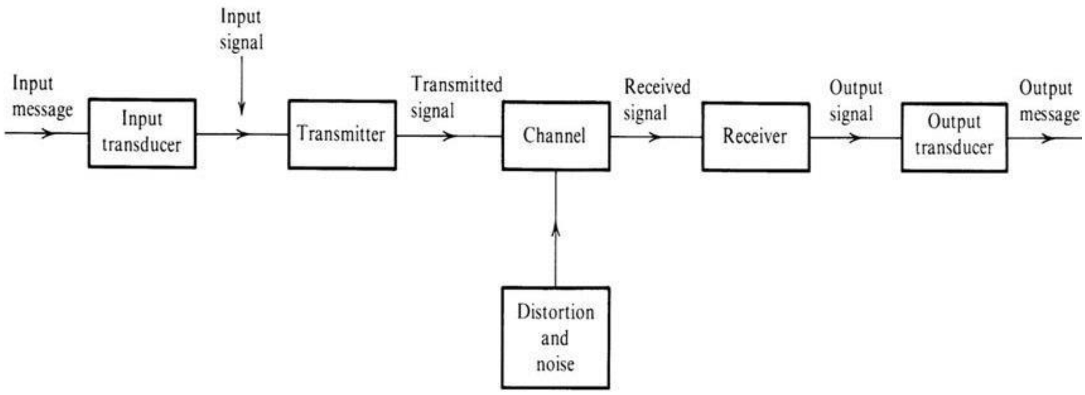
Unit-1: Elements of Communication Systems.

Source of information & Communication Channels.

The key components of a communication system are as follows.

Source :

- The **source** originates a message, such as a human voice, a television picture, an e-mail message, or data.
- If the data is nonelectric (e.g., human voice, e-mail text, television video), it must be converted by an input transducer into an electric waveform referred to as the baseband signal or message signal through physical devices such as a microphone, a computer keyboard, or a CCD camera.



Transmitter :

- The **transmitter** modifies the baseband signal for efficient transmission.
- The transmitter may consist of one or more subsystems: an A/D converter, an encoder, and a modulator.
- Similarly, the receiver may consist of a demodulator, a decoder, and a D/A converter.

Channel :

- The channel is a medium of choice that can convey the electric signals at the transmitter output over a distance.
- A typical channel can be a pair of twisted copper wires (telephone and DSL), coaxial cable (television and internet), an optical fiber, or a radio link. Additionally, a channel can also be a point-to-point connection in a mesh of interconnected channels that form a communication network.

Receiver:

- The receiver reprocesses the signal received from the channel by reversing the signal modifications made at the transmitter and removing the distortions made by the channel.
- The receiver output is fed to the output transducer, which converts the electric signal to its original form-the message.
- The destination is the unit to which the message is communicated.

Noise:

- In a practical environment, signals passing through communication channels not only experience channel distortions but also are corrupted along the path by undesirable interferences and disturbances lumped under the broad term **noise**.
- These interfering signals are random and are unpredictable from sources both **external** and **internal**.
- External noise includes interference signals transmitted on nearby channels, human-made noise generated by faulty contact switches of electrical equipment, automobile ignition radiation, fluorescent lights or natural noise from lightning, microwave ovens, and cellphone emissions, as well as electric storms and solar and intergalactic radiation.
- With proper care in system design, external noise can be minimized or even eliminated in some cases.
- Internal noise results from thermal motion of charged particles in conductors, random emission, and diffusion or recombination of charged carriers in electronic devices.
- Proper care can reduce the effect of internal noise but can never eliminate it.
- Noise is one of the underlying factors that limit the rate of telecommunications.

Classification of Communication systems (Line & Wireless or Radio)

Depending on the communication channel, the communication system is categorized as follows:

1. Wired (Line communication)

- Parallel wire communication
- Twisted wire communication
- Coaxial cable communication
- Optical fibre communication

2. Wireless (Space communication)

- Ground wave communication
- Skywave communication
- Space wave communication
- Satellite communication

In designing communication systems, it is important to understand and analyze important factors such as the channel and signal characteristics, the relative noise strength, the maximum number of bits that can be sent over a channel per second, and, ultimately, the signal quality.

In a given (digital) communication system, the fundamental parameters and physical limitations that control the rate and quality are the channel bandwidth B and the signal power Ps.

The **bandwidth** of a channel is the range of frequencies that it can transmit with reasonable fidelity. . For example, if a channel can transmit with reasonable fidelity a signal whose frequency components vary from 0 Hz (de) up to a maximum of 5000 Hz (5 kHz), the channel bandwidth B is 5 kHz. Likewise, each signal also has a bandwidth that measures the maximum range of its frequency components.

The signal power P_s plays a dual role in information transmission. First, Ps is related to the quality of transmission. Increasing Ps strengthens the signal pulse and diminishes the effect of channel noise and interference. In fact, the quality of either analog or digital communication systems varies with the signal-to-noise ratio (SNR). In any event, a certain minimum SNR at the receiver is necessary for successful communication. Thus, a larger signal power Ps allows the system to maintain a minimum SNR over a longer distance, thereby enabling successful communication over a longer span.

Channel bandwidth limits the bandwidth of signals that can successfully pass through, whereas signal SNR at the receiver determines the recoverability of the transmitted signals. Higher SNR means that the transmitted signal pulse can use more signal levels, thereby carrying more bits with each pulse transmission. Higher bandwidth B also means that one can transmit more pulses (faster variation) over the channel. Hence, SNR and bandwidth B can both affect the underlying channel "throughput." The peak throughput that can be reliably carried by a channel is defined as the channel capacity.

One of the most commonly encountered channels is known as the additive white Gaussian noise (AWGN) channel. The AWGN channel model assumes no channel distortions except for the additive white Gaussian noise and its finite bandwidth B. This ideal model captures application cases with distortionless channels and provides a performance upper bound for more general distortive channels.

The band-limited A WGN channel capacity was dramatically highlighted by Shannon's equation,

$$C = B \log2 (1 + SNR) \text{ bit/s}$$

Here the channel capacity C is the upper bound on the rate of information transmission per second. In other words, C is the maximum number of bits that can be transmitted per second with a probability of error arbitrarily close to zero; that is, the transmission is as accurate as one desires.

Modulation Process, Need of modulation and classify modulation process

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

Increase the Signal Strength

The baseband signals transmitted by the sender are not capable of direct transmission. The strength of the message signal should be increased so that it can travel longer distances. This is where modulation is essential. The most vital need of modulation is to enhance the strength of the signal without affecting the parameters of the carrier signal.

Wireless Communication System

Modulation has removed the necessity for using wires in the communication systems. It is because modulation is widely used in transmitting signals from one location to another with faster speed. Thus, the modulation technique has helped in enhancing wireless communication systems.

Prevention of Message Signal From Mixing

Modulation and its types prevent the interference of the message signal from other signals. It is because a person sending a message signal through the phone cannot tell such signals apart. As a result, they will interfere with each other. However, by using carrier signals having a high frequency, the mixing of the signals can be prevented. Thus, modulation ensures that the signals received by the receiver are entirely perfect.

Size of the Antenna

The signals within 20 Hz to 20 kHz frequency range can travel only a few distances. To send the message signal, the length of the antenna should be a quarter wavelength of the used frequency. Thus, modulation is required to increase the frequency of the message signal and to enhance its strength to reach the receiver.

Length of the antenna can be easily calculated using this formula:

$L = \lambda = u/v$

Here,
L = length of antenna
λ = wavelength of the transmitted signal
u=3 x 108 m/sec
v= carrier wave frequency

Digital Signal: A digital signal is a signal that represents data as a sequence of discrete values; at any given time it can only take on one of a finite number of values.

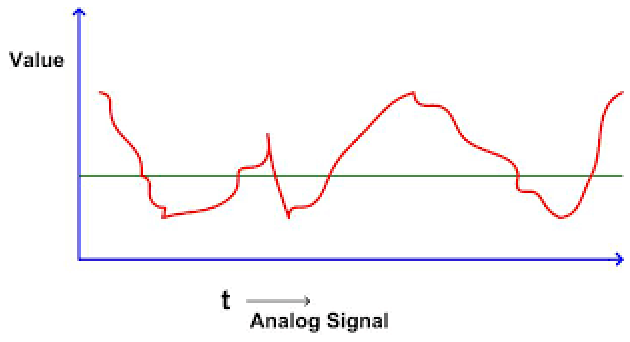
Analog Signal: An analog signal is any continuous signal for which the time varying feature of the signal is a representation of some other time varying quantity i.e., analogous to another time varying signal.

Data

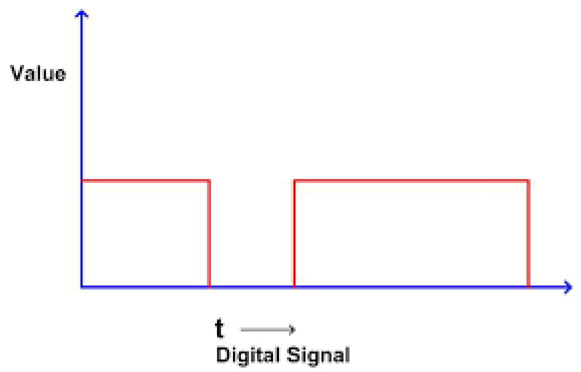
- Data refers to information that conveys some meaning based on some mutually agreed up rules or conventions between a sender and a receiver and today it comes in a variety of forms such as text, graphics, audio, video and animation.
- Data can be of two types; analog and digital.
- Analog data take on continuous values on some interval. Typical examples of analog data are voice and video. The data that are collected from the real world with the help of transducers are continuous-valued or analog in nature.
- On the contrary, digital data take on discrete values. Text or character strings can be considered as examples of digital data. Characters are represented by suitable codes, e.g. ASCII code, where each character is represented by a 7-bit code.

Signal

- It is electrical, electronic or optical representation of data, which can be sent over a communication medium.
- Stated in mathematical terms, a signal is merely a function of the data. For example, a microphone converts voice data into voice signal, which can be sent over a pair of wire. Analog signals are continuous-valued; digital signals are discrete-valued. The independent variable of the signal could be time (speech, for example), space (images), or the integers (denoting the sequencing of letters and numbers in the football score).
- Figure below shows an analog signal.



- Digital signal can have only a limited number of defined values, usually two values 0 and 1, as shown in below figure.



<u>**Signal Characteristics**</u>

A signal can be represented as a function of time, i.e. it varies with time. However, it can be also expressed as a function of frequency, i.e. a signal can be considered as a composition of different frequency components. Thus, a signal has both time-domain and frequency domain representation.

Time-domain concepts

- A signal is continuous over a period, if $\lim_{t\to a} s(t) = s(a)$, for all a, i.e., there is no break in the signal.
- A signal is discrete if it takes on only a finite number of values.
- A signal is periodic if and only if $s(t+T) = s(t)$ for $-\alpha < t < \alpha$, where T is a constant, known as period.
- The period is measured in seconds. In other words, a signal is a periodic signal if it completes a pattern within a measurable time frame.
- A periodic signal is characterized by the following three parameters.

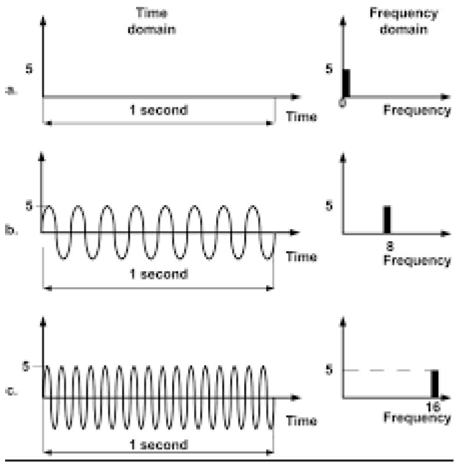
Amplitude: It is the value of the signal at different instants of time. It is measured in volts.

Frequency: It is inverse of the time period, i.e. $f = 1/T$. The unit of frequency is Hertz (Hz) or cycles per second.

Phase: It gives a measure of the relative position in time of two signals within a single period. It is represented by ϕ in degrees or radian.

Frequency domain concepts

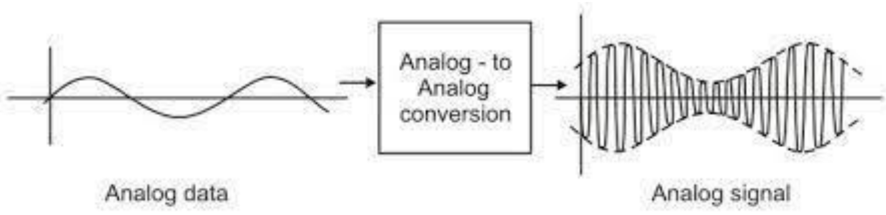
The time domain representation displays a signal using time-domain plot, which shows changes in signal amplitude with time. The time-domain plot can be visualized with the help of an oscilloscope. The relationship between amplitude and frequency is provided by frequency domain representation, which can be displayed with the help of spectrum analyser. Time domain and frequency domain representations of three sine waves of three different frequencies are shown in Fig. 2.1.6.


Time
domain
Frequency
domain
a.
5
1 second
Time
5
0
Frequency
b.
5
1 second
Time
5
8
Frequency
c.
5
1 second
Time
5
16
Frequency

Unit-2: Amplitude (linear) Modulation System

Introduction

Although transmission of digital signal is preferred, it is not always feasible to transmit in digital form because it requires channel of high bandwidth having low pass characteristics. On the other hand, an analog transmission requires lower bandwidth having band pass characteristics. The process involved in analog transmission is known as modulation, which requires manipulation of one or more of the parameters of the carrier that characterizes the analog signal. Figure below depicts the modulation process to get analog signal.



Some of the important advantages of modulation are summarized below:

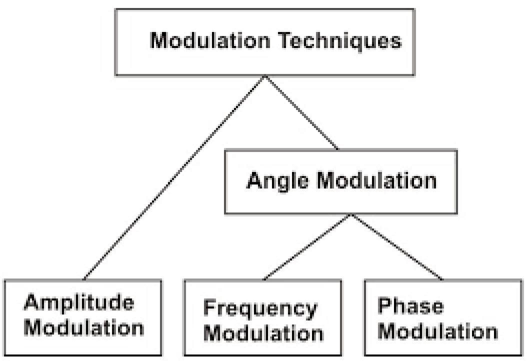
Frequency translation: Modulation translates the signal from one region of frequency domain to another region. This helps to transmit the modulated signal with minimum attenuation through a particular medium.

Practical size of antenna: Modulation translates baseband signal to higher frequency, which can be transmitted through a bandpass channel using an antenna of smaller size. This has made communication practical.

Narrowbanding: As modulation translates a signal from lower frequency domain to higher frequency domain, the ratio between highest to lowest frequency of the modulated signal becomes close to 1.

Multiplexing: Different base band signals originating from different sources can be translated to different frequency ranges. This allows transmission of different signals through the same medium using frequency division multiplexing (FDM) to be discussed in the following lesson.

The modulation technique can be broadly divided into two basic categories; **Amplitude modulation and Angle modulation**. The Angle modulation can be further divided into two more categories; Frequency and Phase modulations as shown in below figure.



Amplitude Modulation (AM):

This is the simplest form of modulation where the amplitude of the carrier wave is modulated by the analog signal known as the modulating signal. A signal to be modulated, a carrier and the modulated signal are shown in below figure.

Let the modulating waveform is given by $e_m(t) = Em \cos(2\pi f_m t)$ and the carrier signal is given by $e_c(t) = E_c \cos(2\pi f_c t + \Phi_c)$. Then the equation of the modulated signal is given by

$$s(t) = (E_c + E_m \cos 2\pi f_m t) \cos 2\pi f_c t$$

Modulation Index: The modulation index, represented by m, is given by

$m = (Emax - Emin)/(Emax + Emin)$

$= Em / Ec$,

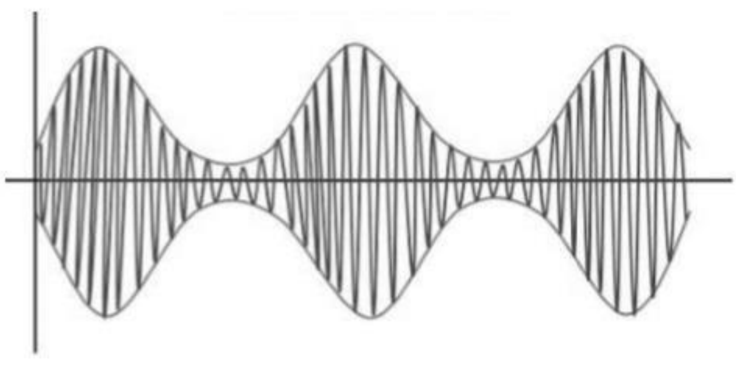
where $E_{max} = E_c + E_m$, $Emin = E_c - E_m$, and $s(t) = E_c (1 + m \cos 2\pi f_m t) \cos 2\pi f_c t$,

The envelope of the modulated signal is represented by $1+m\ e_m(t)$ for $m < 1$. Loss of information occurs when $m > 1$.

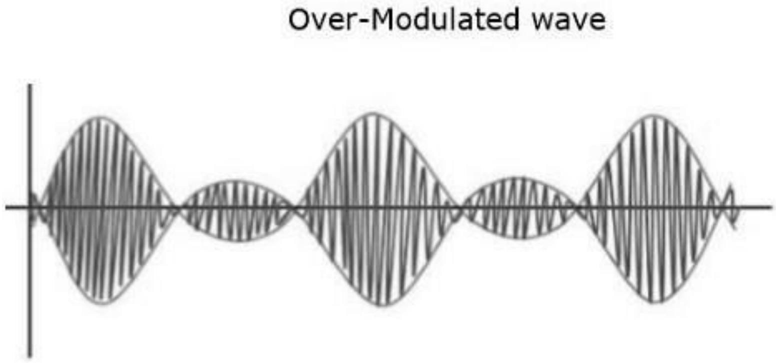
For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%.

For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as Under-modulation. Such a wave is called as an under-modulated wave.

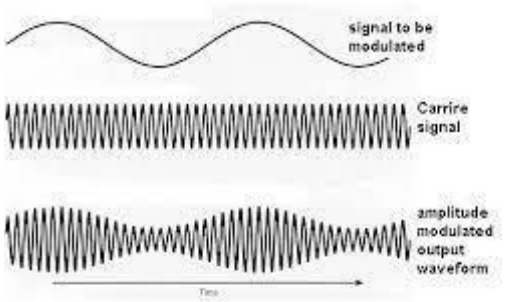
Under-Modulated wave



If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an over-modulated wave. It would look like the following figure.



As the value of the modulation index increases, the carrier experiences a 180^0 phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such an over-modulated wave causes interference, which cannot be eliminated.



Frequency Spectrum: Frequency spectrum of the sinusoidal AM signal can be represented by

$s(t) = E_c *1 + m \cos 2\pi f_m t + \cos 2\pi f_c t = E_c \cos 2\pi f_c t + m E_c \cos 2\pi f_m t \cos 2\pi f_c t$

$= E_c \cos 2\pi f_c t + m/2\ E_c \cos 2\pi(fc - fm)t + m/2\ Ec \cos 2\pi(fc + fm)t$

It may be noted that there are three frequency components; Carrier wave of amplitude Ec, Lower sideband of amplitude m/2 Ec and Higher sideband of amplitude m/2 Ec.

A carrier of 1 MHz with peak value of 10V is modulated by a 5 KHz sine wave amplitude 6V. Determine the modulation index and frequency spectrum.

Answer: The modulation index m = 6/10 = 0.6. The side frequencies are (1000 – 5) = 995 KHz and (1000 + 5) = 1005 KHz having amplitude of 0.6 × 10/2 = 3V

Bandwidth of AM Wave

Bandwidth (BW) is the difference between the highest and lowest frequencies of the signal. Mathematically, we can write it as

$BW = f_{max} - f_{min}$

Consider the following equation of amplitude modulated wave.

$s(t) = A_c * 1 + \mu \cos(2\pi f_m t) + \cos(2\pi f_c t)$

$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + A_c \mu \cos(2\pi f_c t) \cos(2\pi f_m t)$

$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + A_c \mu/2 \cos * 2\pi(f_c + f_m)t + + A_c \mu/2 \cos * 2\pi(f_c - f_m)t +$

Hence, the amplitude modulated wave has three frequencies. Those are carrier frequency f_c, uppersideband frequency $f_c + f_m$ and lower sideband frequency $f_c - f_m$

Here,

$f_{max} = f_c + f_m$ and $f_{min} = f_c - f_m$

Substitute, f_{max} and f_{min} values in bandwidth formula.

$BW = f_c + f_m - (f_c - f_m)$

$\Rightarrow BW = 2f_m$

Thus, it can be said that the bandwidth required for amplitude modulated wave is twice the frequency of the modulating signal.

Power:

Power Calculations of AM Wave

Consider the following equation of amplitude modulated wave.

$s(t) = A_c \cos(2\pi f_c t) + A_c \mu/2 \cos * 2\pi(f_c + f_m)t + + A_c \mu/2 \cos * 2\pi(f_c - f_m)t +$

Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.

$P_t = P_c + P_{USB} + P_{LSB}$

We know that the standard formula for power of cos signal is

$P = V_{rms}2 / R = (V_m/\sqrt{2})2/R$

Where,

V_{rms} is the rms value of cos signal.

V_m is the peak value of cos signal.

First, let us find the powers of the carrier, the upper and lower sideband one by one.

Carrier power

$P_c=(A_c/\sqrt{2})2 / R= (A_c)2/2R$

Upper sideband power

$P_{USB}=(A_c\ \mu/2\sqrt{2})2/ R=A_c2\ \mu2/ 8R$

Similarly, we will get the lower sideband power same as that of the upper side band power.

$P_{USB}= A_c2\ \mu2/ 8R$

Now, let us add these three powers in order to get the power of AM wave.

$P_t=A_c2/2R + A_c2\ \mu2/ 8R+ A_c2\ \mu2/ 8R$

$P_t=A_c2/2R\ (1+ \mu2/4+ \mu2/4)$

$P_t= P_c\ \ (1+ \mu2/2)$

We can use the above formula to calculate the power of AM wave, when the carrier power and the modulation index are known.

If the modulation index μ=1then the power of AM wave is equal to 1.5 times the carrier power. So, the power required for transmitting an AM wave is 1.5 times the carrier power for a perfect modulation.

A modulating signal m(t)=10cos(2π×103t) is amplitude modulated with a carrier signal c(t)=50cos(2π×105 t). Find the modulation index, the carrier power, and the power required for transmitting AM wave.

$P_c=(50)2 / 2(1)=1250W$

Therefore, the Carrier power, Pc is 1250 watts.

We know the formula for power required for transmitting AM wave is

$P_t= P_c\ \ (1+ \mu2/2)$

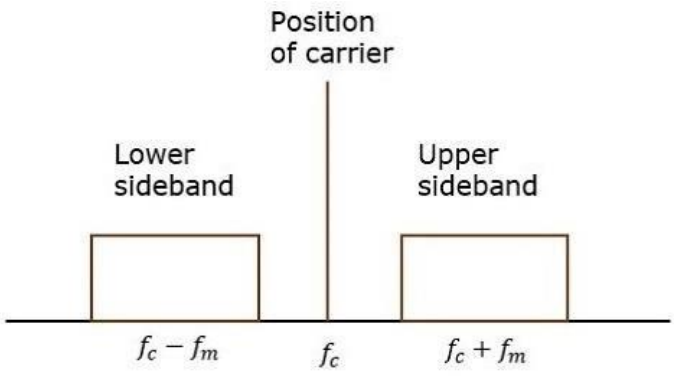
- Average power developed across a resistor R for the carrier signal is $P_c = E_c{}^2/2R$.
- For each of the sideband frequencies the power is $P_{SF} = (mE_c / 2)^2/ 2R = P_c\ m2/4$.
- So, the total power required for transmission is $= P_c (1 + 2(m^2/4)) = P_c (1 + m^2/2)$.

To minimize power for transmission, there are two other alternatives:

- **Double-Sideband with Suppressed Carrier (DSBSC) Modulation**
- **Single Side Band (SSB) Modulation**

Double-Sideband with Suppressed Carrier (DSBSC) Modulation utilizes the transmitted power more efficiently than DSB AM. On the other hand, Single Side Band (SSB) Modulation not only conserves energy, it also reduces bandwidth. It may be noted that one of the two side bands needs to be transmitted.

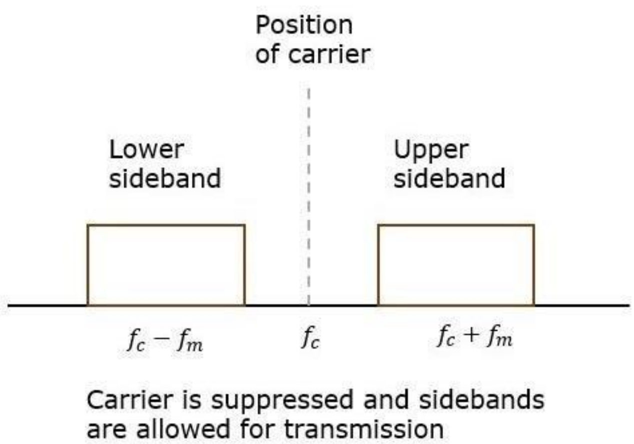
The transmission of a signal, which contains a carrier along with two sidebands can be termed as Double Sideband Full Carrier system or simply DSBFC. It is plotted as shown in the following figure.



However, such a transmission is inefficient. Because, two-thirds of the power is being wasted in the carrier, which carries no information.

DSBSC

If this carrier is suppressed and the saved power is distributed to the two sidebands, then such a process is called as Double Sideband Suppressed Carrier system or simply DSBSC. It is plotted as shown in the following figure.



Mathematical Expressions

Let us consider the same mathematical expressions for modulating and carrier signals as we have considered earlier.

m(t)=Em cos(2πfmt)

$c(t)=E_c \cos(2\pi f_c t)$

Mathematically, we can represent the equation of DSBSC wave as the product of modulating and carrier signals.

$s(t)=m(t)c(t)$

$\Rightarrow s(t)= E_m E_c \cos(2\pi f_m t) \cos(2\pi f_c t)$

Bandwidth of DSBSC Wave

$BW=f_{max}-f_{min}$

Consider the equation of DSBSC modulated wave.

$s(t)= E_m E_c \cos(2\pi f_m t) \cos(2\pi f_c t)$

$\Rightarrow s(t)= E_m E_c /2 \cos*2\pi(f_c+f_m)t++ E_m E_c /2 \cos*2\pi(f_c-f_m)t+$

The DSBSC modulated wave has only two frequencies. So, the maximum and minimum frequencies are fc+fmfc+fm and fc−fmfc−fm respectively.

$f_{max}=f_c+f_m$ and $f_{min}=f_c-f_m$

Substitute, f_{max} and f_{min} values in the bandwidth formula.

$BW=f_c+f_m-(f_c-f_m)$

$BW=2f_m$

Thus, the bandwidth of DSBSC wave is same as that of AM wave and it is equal to twice the frequency of the modulating signal.

Power Calculations of DSBSC Wave

Consider the following equation of DSBSC modulated wave.

$s(t)= E_m E_c /2 \cos*2\pi(f_c+f_m)t++ E_m E_c /2 \cos*2\pi(f_c-f_m)t+$

Power of DSBSC wave is equal to the sum of powers of upper sideband and lower sideband frequency components.

$P_t=P_{USB}+P_{LSB}$

We know the standard formula for power of cos signal is

$P=V_{rms}2 /R=(V_m/\sqrt{2})2/R$

Upper sideband power

$P_{USB}=(E_m E_c /2\sqrt{2})2/ R= E_m 2 E_c 2/ 8R$

Similarly, we will get the lower sideband power same as that of upper sideband power.

$P_{LSB} = (E_m E_c / 2\sqrt{2})^2 / R = E_m^2 E_c^2 / 8R$

Now, let us add these two sideband powers in order to get the power of DSBSC wave.

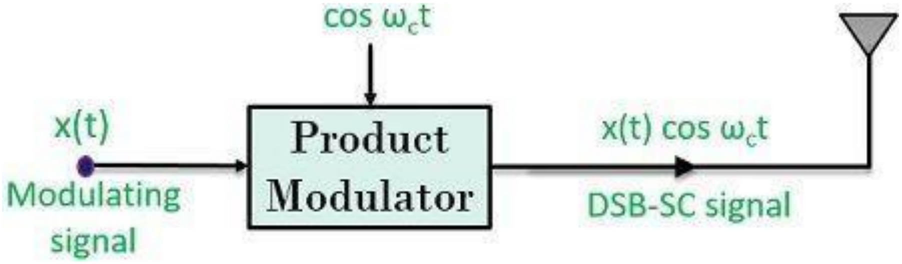
$P_t = P_{USB} + P_{LSB}$

$= E_m^2 E_c^2 / 8R + E_m^2 E_c^2 / 8R$

$= E_m^2 E_c^2 / 4R$

Generation of DSB-SC signal

Let's have a look at the block diagram of the DSB-SC system shown below:



Here, by observing the above figure, we can say that a product modulator generates a DSB-SC signal.

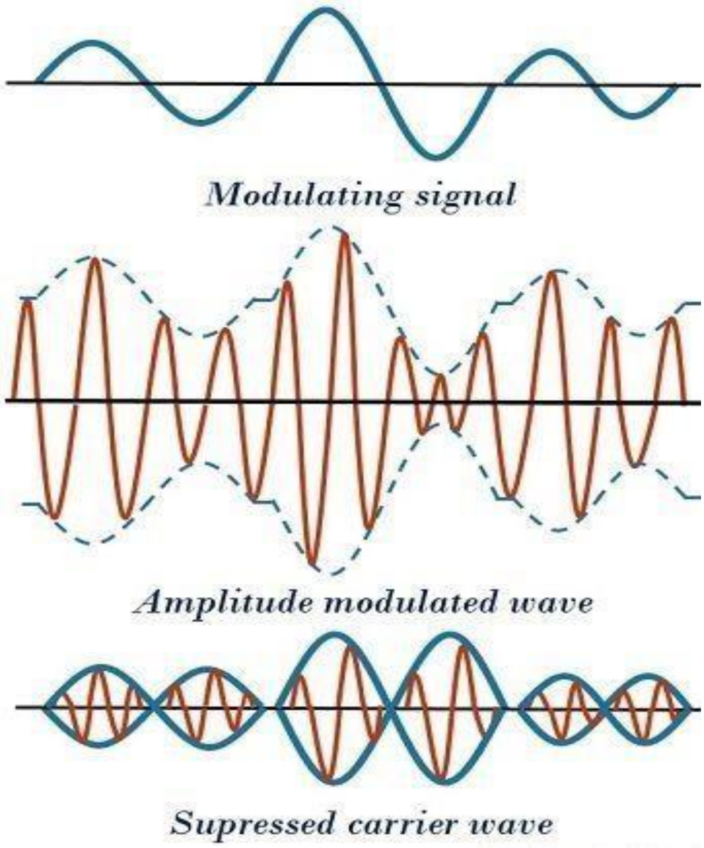
The signal is obtained by the multiplication of baseband signal x(t) with carrier signal cos $\omega_c t$

By frequency shifting property of Fourier transform-

$$x(t) \cos \omega_c t \longleftrightarrow \frac{1}{2}[\, X(\omega + \omega_c) + (\omega - \omega_c)]$$

From the above equation, it is clear that only 2 components are present in the spectrum. These two are the two sidebands that are placed at $+\omega_c$ and $-\omega_c$.

Let's have a look at the pictorial representation of a waveform for DSB-SC system-



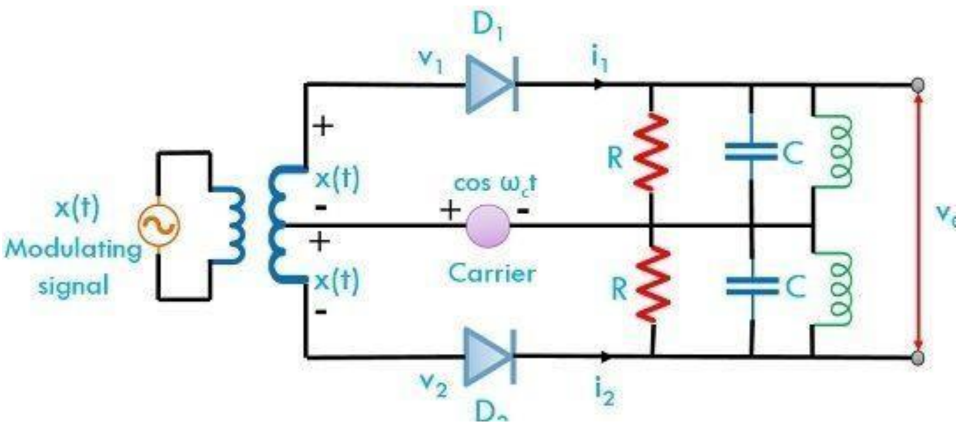
Carrier suppression in DSB-SC (Balanced Modulator)

The carrier without any information content is suppressed by a balanced modulator. Its principle of operation is such that, when two signals of the different frequency are passed through a non-linear resistance then an amplitude modulated signal with the suppressed carrier is achieved at the output.

It can be a diode, JFET or BJT that possess non-linear resistance characteristic.

A non-linear device has the capability to produce 2 sidebands with a carrier. But, a balanced mode connection of 2 non-linear devices produces a DSB-SC signal.

Let's have a look at the circuit of the balanced modulator using diodes:



As we can see that the baseband input signal is applied at the input of 2 diodes that are 180: phase reversed with each other through a centre tapped transformer.

Hence the input at D1,

v1 = cos ωct + x(t)

and input at D2,

v2 = cos ωc t – x(t)

At the output side tuned bandpass filter is obtained by parallel connection of RLC circuit.

So, the current through D1 is given as

$$i_1 = a v_1 + b v_1^2$$

$$i_1 = a\,[x\,(t) + \cos \omega_c t] + b\,[x\,(t) + \cos \omega_c t]^2$$

$$i_1 = a\,x(t) + a \cos \omega_c t + b x^2(t) + 2\,b\,x(t) \cos \omega_c t + b \cos^2 \omega_c t$$

Similarly,

$$i_2 = a v_2 + b v_2^2$$

$$i_2 = a\,[\cos \omega_c t - x\,(t)] + b\,[\cos \omega_c t - x\,(t)]^2$$

$$i_2 = a \cos \omega_c t - a\,x\,(t) + b x^2(t) - 2\,b\,x\,(t) \cos \omega_c t + b \cos^2 \omega_c t$$

The output voltage is given by

vo = i1 R – i2 R

On substituting the above-given value of i1 and i2 in the output equation, we will have,

vo = R *2 a x(t)+ (+ 4b x (t) cos ωct)+

Therefore, the output is,

vo = 2aR x(t) + 4bRx (t) cos ωct

: 2aR x(t) = modulating signal

bR x(t) cos ωct = DSB-SC signal

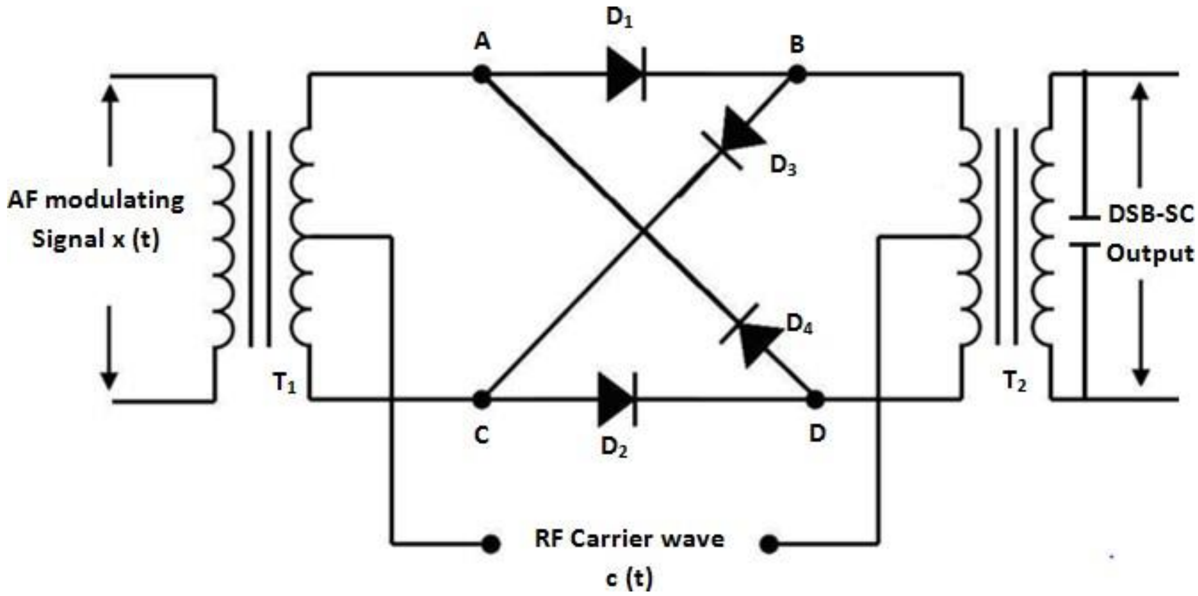
Thus, from the above expression, it is clear that output voltage is a combination of modulating signal along with the DSB-SC signal. After the elimination of the modulating signal, the DSB-SC signal is then passed to the LC bandpass and is received at the output.

Thus we will have,

4bR x(t) cos ωct = K x(t) cos ωct at the output.

Ring Modulator

Following is the block diagram of the Ring modulator.



It consists of four diodes, an audio frequency transformer T1 and an RF transformer T2 .

The carrier signal is assumed to be a square wave with frequency fc and it is connected between the centre taps of the two transformers .

The DSB-SC output is obtained at the secondary of the RF transformer T2 .

Working Operation

The operation of the ring modulator is explained with the assumptions that the diodes act as perfect switches and that they are switched ON and OFF by the RF carrier signal .

This is because the amplitude and frequency of the carrier is higher than that of the modulating signal .

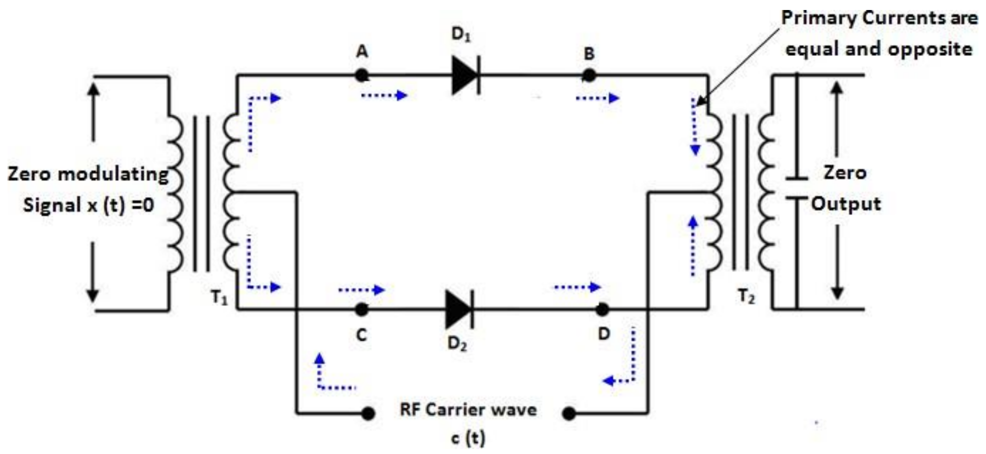
The operation can be divided into different modes without the modulating signal and with the modulating signal as follows :

Mode 1 : Carrier Suppression

To understand how carrier suppression takes place, let us assume that the modulating signal is absent and only the carrier signal is applied.

Hence x(t) = 0

The equivalent circuit for this mode of operation is shown in figure.



As shown in the above figure , the diodes D1 and D2 are forward biased and the diodes D3 and D4 are reverse biased .

We can observe that the direction of currents flowing through the primary windings of output transformer T2 are equal and opposite to each other .

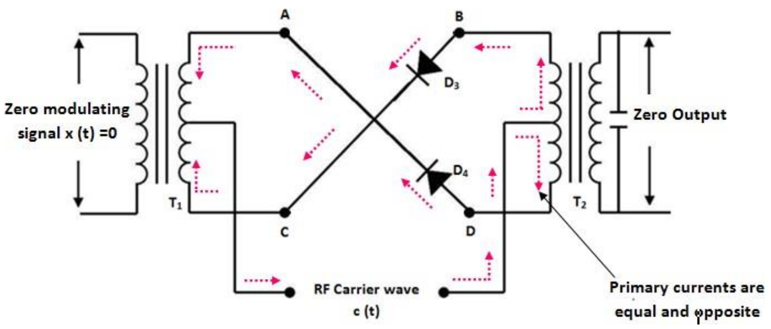
Therefore, the magnetic fields produced by these currents are equal and opposite and cancel each other .

Hence, the induced voltage in secondary winding is zero . Thus, the carrier is supressed in the positive half-cycle .

Operation in the Negative half-cycle of Carrier

In this mode also let us assume that the modulating signal is zero .

In the negative half-cycle of the carrier, the diodes D3 and D4 are forward biased and the diodes D1 and D2 are reverse biased .



The currents flowing in the upper and lower halves of the primary winding of T2 are again equal and in opposite directions . This cancels the magnetic fields as explained in mode 1 .

Thus, the output voltage in this mode also is zero .

Thus, the carrier is suppressed in the negative half-cycle as well .

Advantages of DSB-SC modulation

It provides 100% modulation efficiency.

Due to suppression of carrier, it consumes less power.

It provides a larger bandwidth.

Disadvantages of DSB-SC modulation

It involves a complex detection process.

Using this technique it is sometimes difficult to recover the signal at the receiver.

It is an expensive technique when it comes to demodulation of the signal.

Applications of DSB-SC modulation

During the transmission of binary data, DSB-SC system is used in phase shift keying methods.

In order to transmit 2 channel stereo signals, DSB signals are used in Television and FM broadcasting.

DSB-SC technique allows us to have a transmission that reduces overall power consumption rate, thereby ensuring a stronger signal at the output.

PROBLEMS

The equation of amplitude wave is given by

$\mathbf{s(t)=20[1+0.8\cos(2\pi\times103\ t)]\ \cos(4\pi\times105\ t)}$

Find the carrier power, the total sideband power, and the band width of AM wave.

Given, the equation of Amplitude modulated wave is

$s(t)=20*1+0.8\cos(2\pi\times103\ t)+\cos(4\pi\times105\ t)$

$s(t)=20*1+0.8\cos(2\pi\times103\ t)+\cos(2\times2\pi\times105\ t)$

We know the equation of Amplitude modulated wave is

$s(t)=Ac*1+m\cos(2\pi fmt)+\cos(2\pi fct)$

By comparing the above two equations, we will get

Amplitude of carrier signal as Ac=20volts

Modulation index as m=0.8

Frequency of modulating signal as fm=103 Hz=1KHz

Frequency of carrier signal as fc=2×105 Hz=200KHz

The formula for Carrier power, Pc is

Pc=(Ac)2 /2R

Assume R=1Ωand substitute Ac value in the above formula.

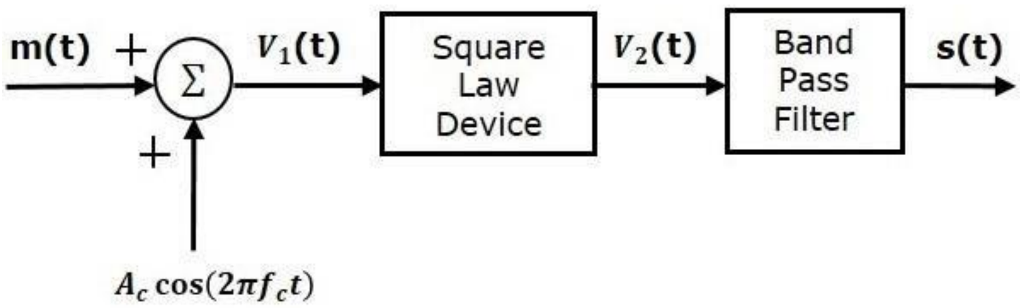
Pc=200W

The following two modulators generate AM wave.

- **Square law modulator**
- **Switching modulator**

Square Law Modulator

Following is the block diagram of the square law modulator



Let the modulating and carrier signals be denoted as m(t) and Acos(2πfct) respectively.

These two signals are applied as inputs to the summer (adder) block.

This summer block produces an output, which is the addition of the modulating and the carrier signal.

Mathematically, we can write it as

V1(t)=m(t)+Ac cos(2πfct)

This signal V1(t) is applied as an input to a nonlinear device like diode. The characteristics of the diode are closely related to square law.

V2(t)=k1V1(t)+k2 V1(t)2..(1)
Where, k1 and k2 are constants.

Substituting V1(t) in Equation (1) we get

V2(t)=k1 *m(t)+Ac cos(2πfct)+ + k2*m(t)+Ac cos(2πfct)+2

V2(t)= k1 m(t)+ k1Ac cos(2πfct)++ k2 m2(t) + k2 Ac2 cos2 (2πfct) +2 k2 m(t) Ac cos(2πfct)

V2(t)= k1 m(t)+ k2 m2(t)+ k2 Ac2 cos2 (2πfct)+ k1Ac *1+ 2 (k2/ k1) m(t)+ cos(2πfct)

The last term of the above equation represents the desired AM wave and the first three terms of the above equation are unwanted. So, with the help of band pass filter, we can pass only AM wave and eliminate the first three terms.

Therefore, the output of square law modulator is

S(t)=k1Ac *1+ 2 (k2/ k1) m(t)+ cos(2πfct)

The standard equation of AM wave is

S(t)=k1Ac *1+ 2 ka m(t)+ cos(2πfct)

Where, Ka is the amplitude sensitivity.

By comparing the output of the square law modulator with the standard equation of AM wave, we will get the scaling factor as k1 and the amplitude sensitivity ka as

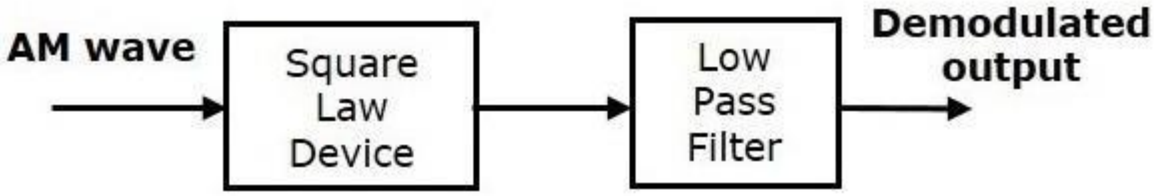
2 (k2/ k1)

The process of extracting an original message signal from the modulated wave is known as detection or demodulation. The circuit, which demodulates the modulated wave is known as the demodulator. The following demodulators (detectors) are used for demodulating AM wave.

Square Law Demodulator

Envelope Detector
Square Law Demodulator

Square law demodulator is used to demodulate low level AM wave. Following is the block diagram of the square law demodulator.



This demodulator contains a square law device and low pass filter. The AM wave V1(t) is applied as an input to this demodulator.

The standard form of AM wave is

$V1(t)=Ac[1+kam(t)]\cos(2\pi fct)$

We know that the mathematical relationship between the input and the output of square law device is

$V2(t)=k1V1(t)+k2\ V1(t)2$ Equation 1

Where,

V1(t) is the input of the square law device, which is nothing but the AM wave

V2(t) is the output of the square law device

k1 and k2 are constants

Substituting V1(t) in Equation 1 we get,

$V2(t)=k1(Ac*1+kam(t)+\cos(2\pi fct))+k2(Ac*1+kam(t)+\cos(2\pi fct))2$

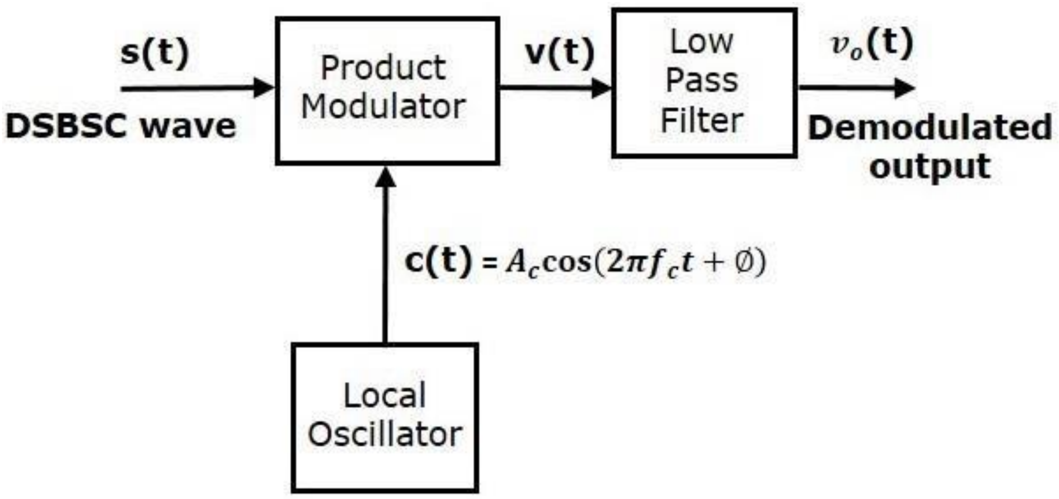
DSB-SC demodulators

The process of extracting an original message signal from DSBSC wave is known as detection or demodulation of DSBSC. The following demodulators (detectors) are used for demodulating DSBSC wave.

- **Coherent Detector**
- **Costas Loop**

Coherent Detector

Here, the same carrier signal (which is used for generating DSBSC signal) is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. Following is the block diagram of the coherent detector.



In this process, the message signal can be extracted from DSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in DSBSC modulation. The resulting signal is then passed through a Low Pass Filter. Output of this filter is the desired message signal.

Let the DSBSC wave be

s(t)=Ac cos(2πfct) m(t)

The output of the local oscillator is

c(t)=Accos(2πfct+φ)

Where, φ is the phase difference between the local oscillator signal and the carrier signal, which is used for DSBSC modulation.

From the figure, we can write the output of product modulator as

v(t)=s(t)c(t)

Substitute, s(t) and c(t) values in the above equation

v(t)=Accos(2πfct)m(t)Accos(2πfct+φ) m(t)

=Ac2 cos(2πfct) cos(2πfct+φ) m(t)

=Ac2/2*cos(4πfct+φ)+cosφ] m(t)

v(t)= Ac2/2 cosφ m(t)+ Ac2/2 cos(4πfct+φ) m(t)

In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.

Therefore, the output of low pass filter is

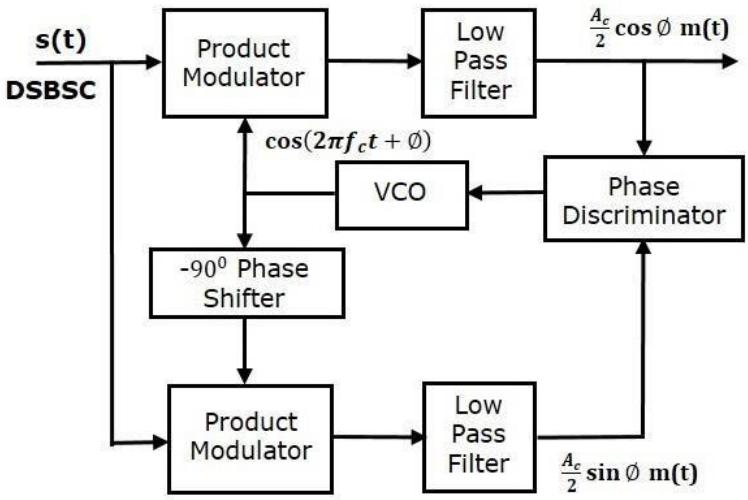
Vo(t)= Ac2/2 cosφ m(t)

The demodulated signal amplitude will be maximum, when φ=00. That's why the local oscillator signal and the carrier signal should be in phase, i.e., there should not be any phase difference between these two signals.

The demodulated signal amplitude will be zero, when φ=±900 . This effect is called as quadrature null effect.

Costas Loop

Costas loop is used to make both the carrier signal (used for DSBSC modulation) and the locally generated signal in phase. Following is the block diagram of Costas loop.



Costas loop consists of two product modulators with common input s(t), which is DSBSC wave. The other input for both product modulators is taken from Voltage Controlled Oscillator (VCO) with −900 phase shift to one of the product modulator as shown in figure.

We know that the equation of DSBSC wave is

s(t)=Accos(2πfct)m(t)

Let the output of VCO be

c1(t)=cos(2πfct+φ)

This output of VCO is applied as the carrier input of the upper product modulator.

Hence, the output of the upper product modulator is

v1(t)=s(t)c1(t)

Substitute, s(t) and c1(t) values in the above equation.

⇒v1(t)=Accos(2πfct)m(t) cos(2πfct+φ)

After simplifying, we will get v1(t) as

v1(t)=Ac/ 2cosφ m(t)+Ac/2cos(4πfct+φ)m(t)

This signal is applied as an input of the upper low pass filter. The output of this low pass filter is

v01(t)=Ac/2 cosφ m(t)

Therefore, the output of this low pass filter is the scaled version of the modulating signal.

The output of −900 phase shifter is

c2(t)=cos(2πfct+φ−900)=sin(2πfct+φ)

This signal is applied as the carrier input of the lower product modulator.

The output of the lower product modulator is

$v2(t)=s(t)c2(t)$

Substitute, s(t) and c2(t) values in the above equation.

$\Rightarrow v2(t)=Accos(2\pi fct)m(t)sin(2\pi fct+\varphi)$

After simplifying, we will get v2(t) as

$v2(t)=Ac/2sin\varphi m(t)+Ac/2sin(4\pi fct+\varphi)m(t)$

This signal is applied as an input of the lower low pass filter. The output of this low pass filter is

$v02(t)=Ac/2\ sin\varphi\ m(t)$

The output of this Low pass filter has −900−900 phase difference with the output of the upper low pass filter.

The outputs of these two low pass filters are applied as inputs of the phase discriminator. Based on the phase difference between these two signals, the phase discriminator produces a DC control signal.

This signal is applied as an input of VCO to correct the phase error in VCO output. Therefore, the carrier signal (used for DSBSC modulation) and the locally generated signal (VCO output) are in phase.

Unit-3: Angle Modulation Systems.

Angle Modulation

In angle modulation, amplitude of the modulated signal is constant. Frequency Modulation (FM) and Phase Modulation (PM) are the special cases of Angle modulation. For Phase Modulation, the phase is proportional to the modulating signal, whereas for frequency modulation, the derivative of the phase is proportional to the modulating signal.

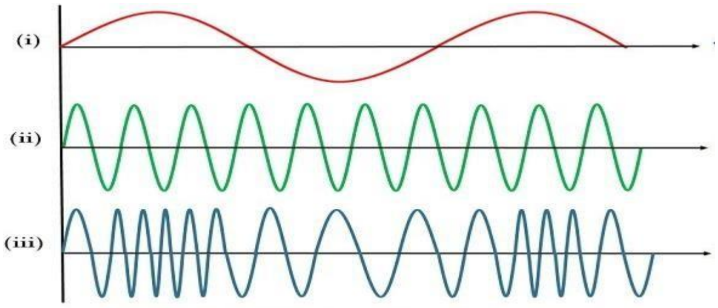
Frequency Modulation is the process of varying the frequency of the carrier signal linearly with the message signal.

Phase Modulation is the process of varying the phase of the carrier signal linearly with the message signal.

Frequency modulation

In amplitude modulation, the amplitude of the carrier varies. But in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.

The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes. This can be better understood by observing the following figures.



(i) Modulating signal
(ii) Carrier waveform
(iii) Frequency modulated signal

The frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero. The frequency increases when the message signal reaches its maximum amplitude.

Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency increases. Likewise, with the decrease in the amplitude of the modulating signal, the frequency also decreases.

Mathematical Representation

Let the carrier frequency be fc

The frequency at maximum amplitude of the message signal = fc + Δf

The frequency at minimum amplitude of the message signal = fc − Δf

The difference between FM modulated frequency and normal frequency is termed as Frequency Deviation and is denoted by Δf.

The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the Carrier Swing.

Carrier Swing = 2 × frequency deviation

FM can be divided into Narrowband FM and Wideband FM.

Narrowband FM

The features of Narrowband FM are as follows −

- This frequency modulation has a small bandwidth.
- The modulation index is small.
- Its spectrum consists of carrier, USB, and LSB.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Wideband FM

The features of Wideband FM are as follows −

- This frequency modulation has infinite bandwidth.
- The modulation index is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- This is used in entertainment broadcasting applications such as FM radio, TV, etc.

In case of frequency modulation, the modulating signal $e_m(t)$ is used to vary the carrier frequency. The change is frequency is proportional to the modulating voltage $ke_m(t)$, where k is a constant known as frequency deviation constant, expressed in Hz/V. The instantaneous frequency of the modulated signal can be represented by $f_i(t) = f_c+ke_m(t)$, where f_c is the carrier frequency.

For sinusoidal modulation

$e_m(t) = E_m \cos 2\pi f_m t$ and

$f_i(t) = f_c + k\, e_m(t)$

$= f_c + k E_m \cos 2\pi f_m t = f_c + \Delta f \cos 2\pi f_m t$

Therefore,

$s(t) = E_c \cos \theta(t)$

$= E_c \cos (2\pi f_c t + 2\pi \Delta f \int_0^t \cos 2\pi f_m t\, dt)$

$= E_c \cos (2\pi f_c t + (\Delta f/f_m) \sin 2\pi f_m t)$

The modulation index, denoted by β, is given by

$\beta = (\Delta f / f_m)$

or $s(t) = E_c \cos (2\pi f_c t + \beta \sin 2\pi f_m t)$

Bandwidth:

The modulated signal s(t) will contain frequency components $f_c + f_m$, $f_c + 2f_m$, and so on. It can be best approximated based on Carson’s Rule, when β is small.

$$BW=2(\Delta f+f_m)$$

It may be noted that FM requires greater bandwidth than AM.

The bandwidth is 10 times that of the base band signal.

Power:

As the amplitude remains constant, total average power is equal to that of the unmodulated carrier power. So, the power = Ac2/2. Although Am increases the bandwidth, it does not affect power. Therefore, the transmission power for FM is less compared to AM at the expense of higher bandwidth.

Phase modulation

In case of phase modulation the modulated signal can be represented by

s(t) = Ac cos[wct + Φ(t)+

The angle (wct + Φ(t)) undergoes a modulation around the angle θ = wct. The signal is therefore an angular-velocity modulated signal. When the phase is directly proportional to the modulating signal, i.e,

Φ(t) = np m(t), we call it phase modulation, where np is the phase modulation index. The instantaneous frequency of a phase modulated signal is given by

s(t) = Ec cos (Wct + k'm(t)), where k' is a constant

Unit-4: AM & FM TRANSMITTER & RECEIVER

There is a number of different types of radio:

Tuned radio frequency, TRF :

Regenerative receiver:

Super regenerative receiver:

Superheterodyne receiver:

Direct conversion receiver:

Terms associated with Radio Receiver

Sensitivity

It is the ability to amplify weak signals.

Minimum RF signal level that can be detected at the input to the receiver and still produce a usable demodulated information signal.

Broadcast receivers/ radio receivers should have reasonably high sensitivity so that it may have good response to the desired signal

But should not have excessively high sensitivity otherwise it will pick up all undesired noise signals. It is function of receiver gain and measures in decibels.

Typical sensitivity for commercial broadcast-band AM receiver is 50 µV.

Sensitivity of the receiver depends on :

Noise power present at the input to the receiver

Receiver noise figure

Bandwidth improvement factor of the receiver

The best way to improve the sensitivity is to reduce the noise level.

Selectivity

It is the ability to differentiate desired signal from unwanted signals.

Selectivity is obtained by using tuned circuits, which are tuned to desired frequency.

The quality factor of these LC circuits determines the selectivity.

It is given by, Q=XL/R, For better selectivity 'Q' should be high.

Fidelity

It is defined as – a measure of the ability of a communication system to produce an exact replica of the original source information at the output of the receiver.

Any variations in the demodulated signal that are not in the original information signal is considered as distortion.

Radio receiver should have high fidelity or accuracy.

Example- In an A. M. broadcast the maximum audio frequency is 5 KHz hence receiver with good fidelity must produce entire frequency up to 5 KHz.

Noise Figure

In wireless communication systems, the "noise figure (NF)" or the related "noise factor (F)" is a number used to specify the performance of a radio receiver.

The lower the value of noise figure, the better the performance.

This tutorial discusses this important parameter in more detail and describes three different noise figure measurement procedures.

Noise Figure (NF) is sometimes referred to as Noise Factor (F). The relationship is simply:

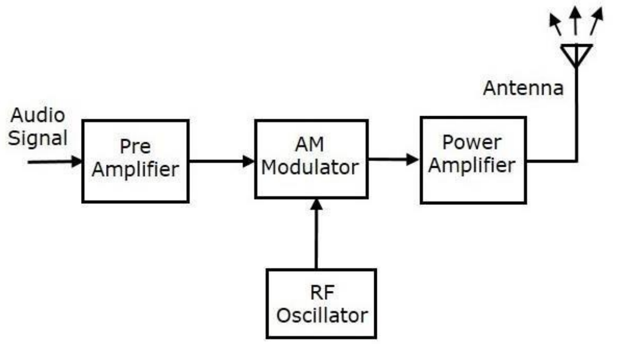
NF = 10 * log10 (F)

Noise Figure (Noise Factor) contains the important information about the noise performance of a RF system. The basic definition is:

$$\text{Noise Factor (F)} = \frac{\text{Total Output Noise Power}}{\text{Output Noise due to Input Source Only}}$$

AM Transmitter

AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted. The block diagram of AM transmitter is shown in the following figure.



The working of AM transmitter can be explained as follows.

The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.

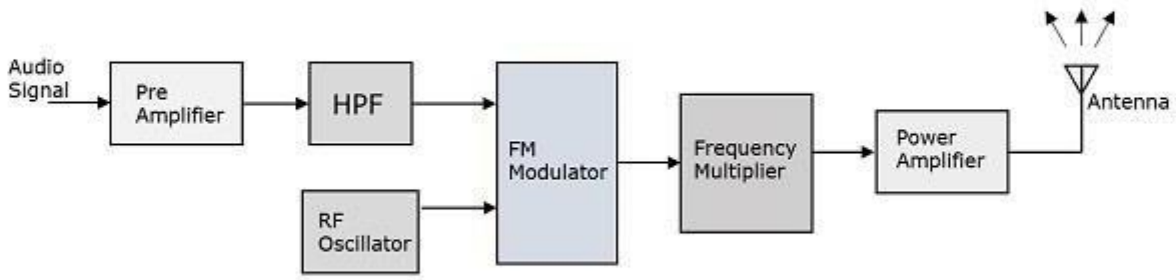
The RF oscillator generates the carrier signal.

Both the modulating and the carrier signal is sent to AM modulator.

Power amplifier is used to increase the power levels of AM wave. This wave is finally passed to the antenna to be transmitted.

FM Transmitter

FM transmitter is the whole unit, which takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted. The block diagram of FM transmitter is shown in the following figure.



The working of FM transmitter can be explained as follows.

The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.

This signal is then passed to high pass filter, which acts as a pre-emphasis network to filter out the noise and improve the signal to noise ratio.

This signal is further passed to the FM modulator circuit.

The oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal.

Several stages of frequency multiplier are used to increase the operating frequency. Even then, the power of the signal is not enough to transmit. Hence, a RF power amplifier is used at the end to increase the power of the modulated signal. This FM modulated output is finally passed to the antenna to be transmitted.

The antenna present at the beginning of the receiver section, receives the modulated wave. First let us discuss the requirements of a receiver.

Requirements of a Receiver

AM receiver receives AM wave and demodulates it by using the envelope detector. Similarly, FM receiver receives FM wave and demodulates it by using the Frequency Discrimination method. Following are the requirements of both AM and FM receiver.

It should be cost-effective.

It should receive the corresponding modulated waves.

The receiver should be able to tune and amplify the desired station.

It should have an ability to reject the unwanted stations.

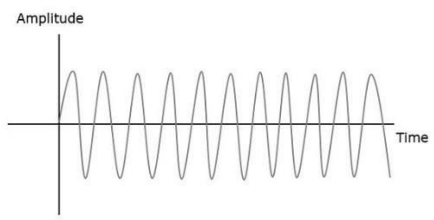
Demodulation has to be done to all the station signals, irrespective of the carrier signal frequency.

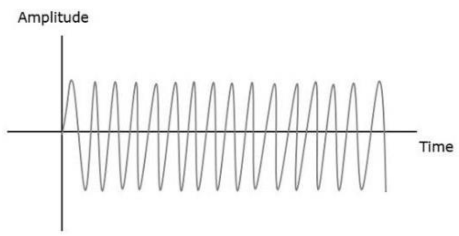
For these requirements to be fulfilled, the tuner circuit and the mixer circuit should be very effective. The procedure of RF mixing is an interesting phenomenon.

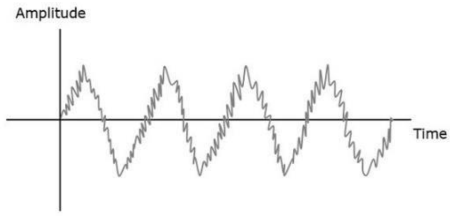
RF Mixing

The RF mixing unit develops an Intermediate Frequency (IF) to which any received signal is converted, so as to process the signal effectively.

RF Mixer is an important stage in the receiver. Two signals of different frequencies are taken where one signal level affects the level of the other signal, to produce the resultant mixed output. The input signals and the resultant mixer output is illustrated in the following figures.

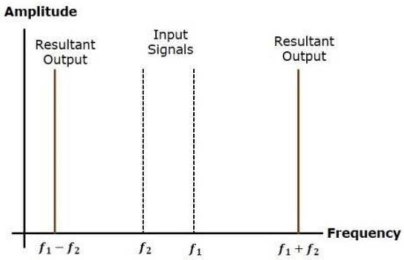






Let the first and second signal frequencies be f1 and f2. If these two signals are applied as inputs of RF mixer, then it produces an output signal, having frequencies of f1+f2 and f1−f2 .

If this is observed in the frequency domain, the pattern looks like the following figure.

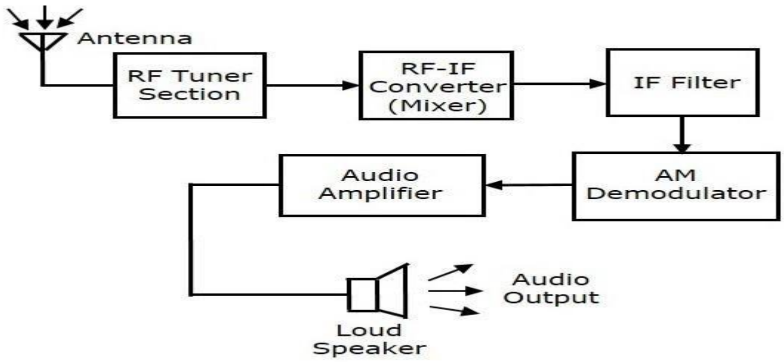


In this case, f1 is greater than f2. So, the resultant output has frequencies f1+f2 and f1−f2.

AM Receiver

The AM super heterodyne receiver takes the amplitude modulated wave as an input and produces the original audio signal as an output. Selectivity is the ability of selecting a particular signal, while rejecting the others. Sensitivity is the capacity of detecting RF signal and demodulating it, while at the lowest power level.

Radio amateurs are the initial radio receivers. However, they have drawbacks such as poor sensitivity and selectivity. To overcome these drawbacks, super heterodyne receiver was invented. The block diagram of AM receiver is shown in the following figure.



RF Tuner Section

The amplitude modulated wave received by the antenna is first passed to the tuner circuit through a transformer. The tuner circuit is nothing but a LC circuit, which is also called as resonant or tank circuit. It selects the frequency, desired by the AM receiver. It also tunes the local oscillator and the RF filter at the same time.

RF Mixer

The signal from the tuner output is sent to the RF-IF converter, which acts as a mixer. It has a local oscillator, which produces a constant frequency. The mixing process is done here, having the received signal as one input and the local oscillator frequency as the other input. The resultant output is a mixture of two frequencies *(f1+f2),(f1−f2)+*(f1+f2),(f1−f2)+ produced by the mixer, which is called as the Intermediate Frequency (IF).

The production of IF helps in the demodulation of any station signal having any carrier frequency. Hence, all signals are translated to a fixed carrier frequency for adequate selectivity.

IF Filter

Intermediate frequency filter is a band pass filter, which passes the desired frequency. It eliminates all other unwanted frequency components present in it. This is the advantage of IF filter, which allows only IF frequency.

AM Demodulator

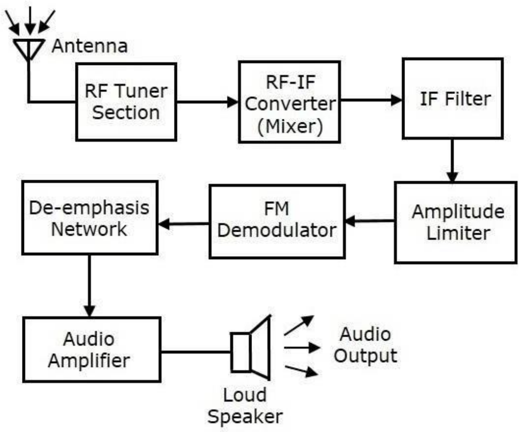
The received AM wave is now demodulated using AM demodulator. This demodulator uses the envelope detection process to receive the modulating signal.

Audio Amplifier

This is the power amplifier stage, which is used to amplify the detected audio signal. The processed signal is strengthened to be effective. This signal is passed on to the loudspeaker to get the original sound signal.

FM Receiver

The block diagram of FM receiver is shown in the following figure.



This block diagram of FM receiver is similar to the block diagram of AM receiver. The two blocks Amplitude limiter and De-emphasis network are included before and after FM demodulator. The operation of the remaining blocks is the same as that of AM receiver.

We know that in FM modulation, the amplitude of FM wave remains constant. However, if some noise is added with FM wave in the channel, due to that the amplitude of FM wave may vary. Thus, with the help of amplitude limiter we can maintain the amplitude of FM wave as constant by removing the unwanted peaks of the noise signal.

In FM transmitter, we have seen the pre-emphasis network (High pass filter), which is present before FM modulator. This is used to improve the SNR of high frequency audio signal. The reverse process of pre-emphasis is known as de-emphasis. Thus, in this FM receiver, the de-emphasis network (Low pass filter) is included after FM demodulator. This signal is passed to the audio amplifier to increase the power level. Finally, we get the original sound signal from the loudspeaker.

Unit-5: ANALOG TO DIGITAL CONVERSION & PULSE MODULATION SYSTEM.

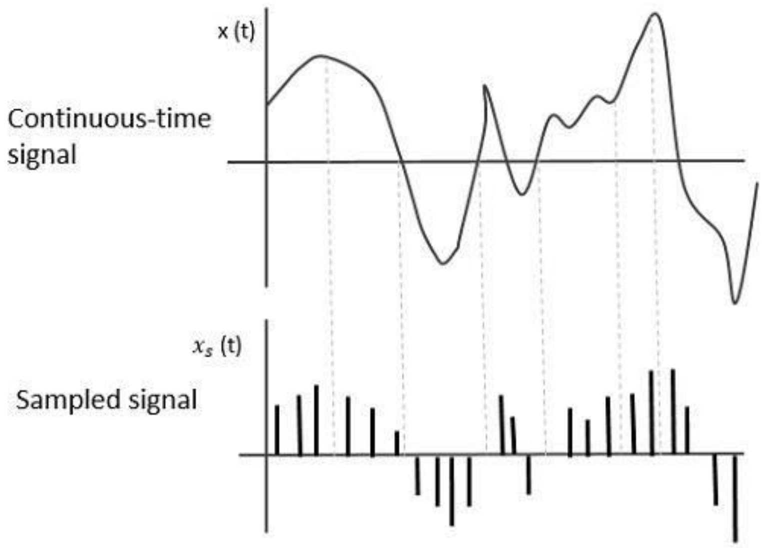
Concept of Sampling Theorem , Nyquist rate & Aliasing

Sampling is defined as, "The process of measuring the instantaneous values of continuous-time signal in a discrete form."

Sample is a piece of data taken from the whole data which is continuous in the time domain.

When a source generates an analog signal and if that has to be digitized, having 1s and 0s i.e., High or Low, the signal has to be discretized in time. This discretization of analog signal is called as Sampling.

The following figure indicates a continuous-time signal x tt and a sampled signal xs tt. When x tt is multiplied by a periodic impulse train, the sampled signal xs tt is obtained.



Sampling Rate

To discretize the signals, the gap between the samples should be fixed. That gap can be termed as a sampling period Ts.

SamplingFrequency=1/Ts=fs

Where,

Ts is the sampling time

fs is the sampling frequency or the sampling rate

Sampling frequency is the reciprocal of the sampling period. This sampling frequency, can be simply called as Sampling rate. The sampling rate denotes the number of samples taken per second, or for a finite set of values.

For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as Nyquist rate.

Nyquist Rate

Suppose that a signal is band-limited with no frequency components higher than W Hertz. That means, W is the highest frequency. For such a signal, for effective reproduction of the original signal, the sampling rate should be twice the highest frequency.

Which means,

$F_S=2W$

Where,

F_S is the sampling rate

W is the highest frequency

This rate of sampling is called as Nyquist rate.

A theorem called, Sampling Theorem, was stated on the theory of this Nyquist rate.

Sampling Theorem

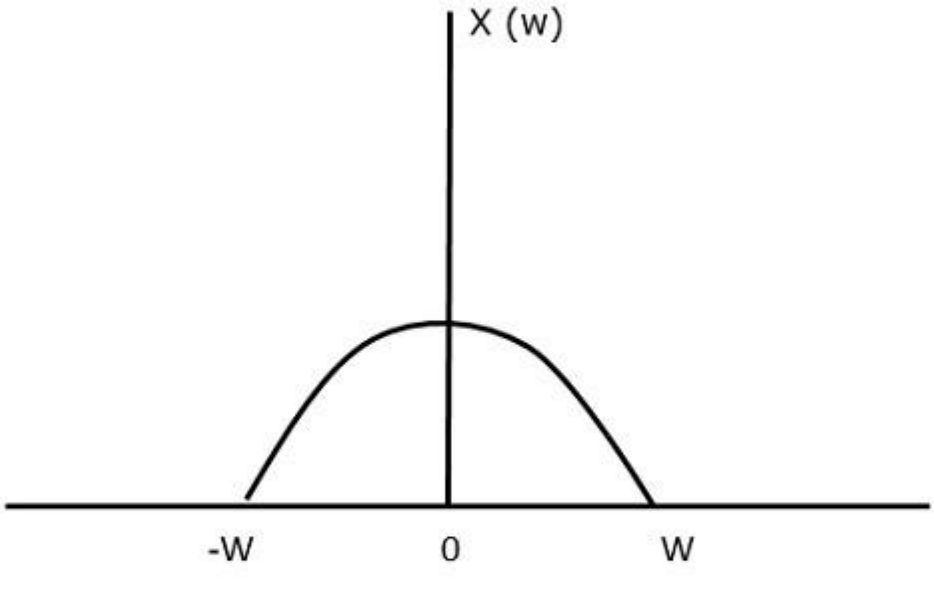
The sampling theorem, which is also called as Nyquist theorem, delivers the theory of sufficient sample rate in terms of bandwidth for the class of functions that are bandlimited.

The sampling theorem states that, "a signal can be exactly reproduced if it is sampled at the rate fs which is greater than twice the maximum frequency W."

To understand this sampling theorem, let us consider a band-limited signal, i.e., a signal whose value is non-zero between some –W and W Hertz.

Such a signal is represented as $x(f)=0$ for $|f|>W$

For the continuous-time signal x t, the band-limited signal in frequency domain, can be represented as shown in the following figure.

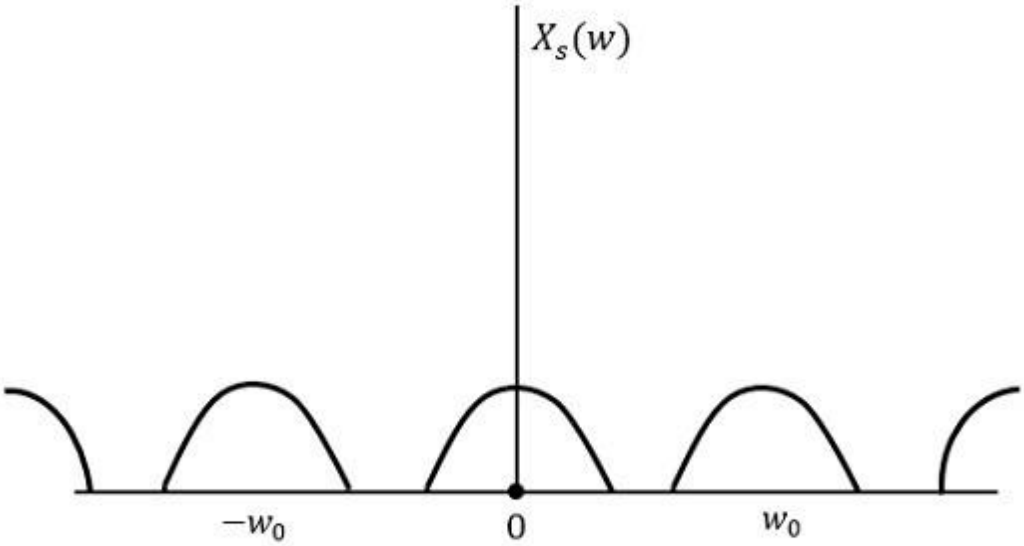


Band limited signal

We need a sampling frequency, a frequency at which there should be no loss of information, even after sampling. For this, we have the Nyquist rate that the sampling frequency should be two times the maximum frequency. It is the critical rate of sampling.

If the signal xtt is sampled above the Nyquist rate, the original signal can be recovered, and if it is sampled below the Nyquist rate, the signal cannot be recovered.

The following figure explains a signal, if sampled at a higher rate than 2w in the frequency domain.



The above figure shows the Fourier transform of a signal xs(t)xs(t). Here, the information is reproduced without any loss. There is no mixing up and hence recovery is possible.

Let us see what happens if the sampling rate is equal to twice the highest frequency (2W)

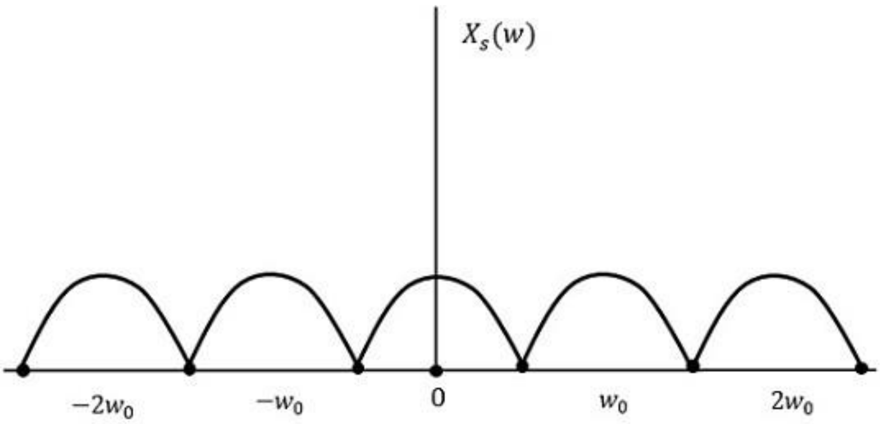
That means,

fs=2W

Where,

fs is the sampling frequency

W is the highest frequency

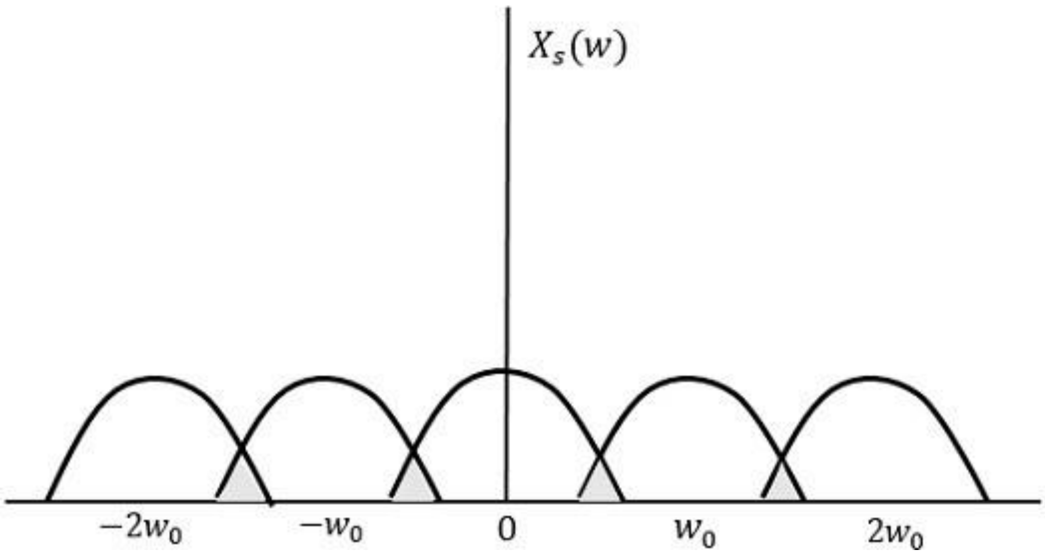


The result will be as shown in the above figure. The information is replaced without any loss. Hence, this is also a good sampling rate.

Now, let us look at the condition,

fs<2Wfs<2W

The resultant pattern will look like the following figure.



We can observe from the above pattern that the over-lapping of information is done, which leads to mixing up and loss of information. This unwanted phenomenon of over-lapping is called as Aliasing.

Aliasing

Aliasing can be referred to as “the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a low-frequency component in the spectrum of its sampled version.”

The corrective measures taken to reduce the effect of Aliasing are −

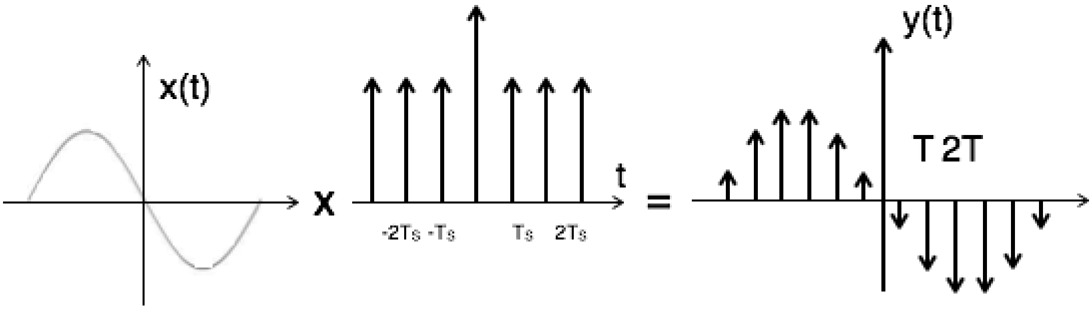
- In the transmitter section of PCM, a low pass anti-aliasing filter is employed, before the sampler, to eliminate the high frequency components, which are unwanted.
- The signal which is sampled after filtering, is sampled at a rate slightly higher than the Nyquist rate.
- This choice of having the sampling rate higher than Nyquist rate, also helps in the easier design of the reconstruction filter at the receiver.

There are three types of sampling techniques:

- **Impulse sampling.**
- **Natural sampling.**
- **Flat Top sampling.**

Impulse Sampling

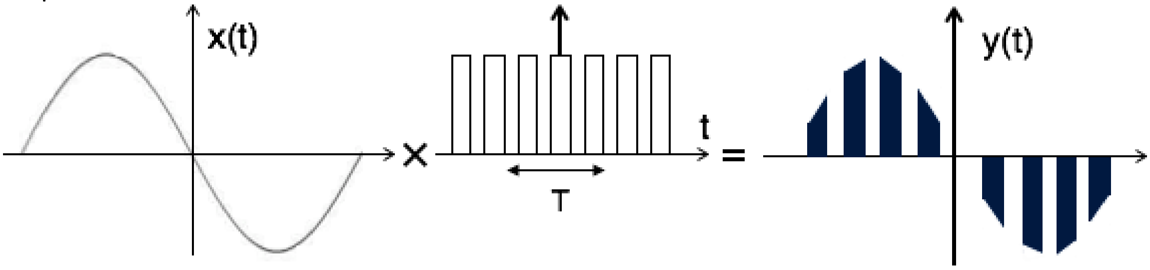
Impulse sampling can be performed by multiplying input signal x(t) with impulse train of period 'T'. Here, the amplitude of impulse changes with respect to amplitude of input signal x(t). The output of sampler is given by



y(t)= x(t)× impulse train

Natural Sampling

Natural sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period T. as shown below

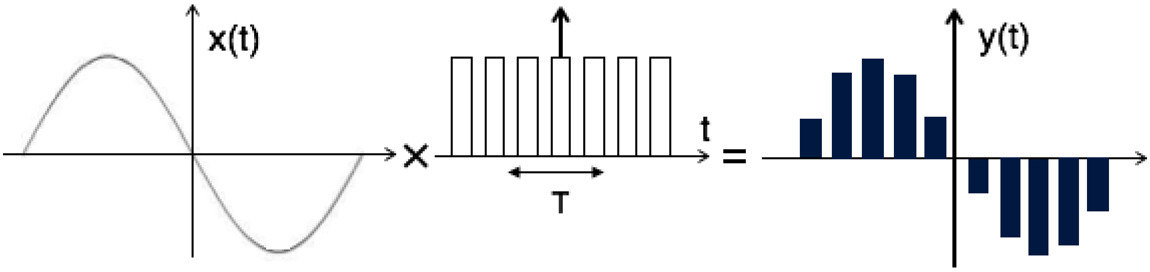


The output of sampler is

y(t)=x(t)×pulse train

Flat Top Sampling

During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the pulse is in the form of flat top. Here, the top of the samples are flat i.e. they have constant amplitude. Hence, it is called as flat top sampling or practical sampling. Flat top sampling makes use of sample and hold circuit.



Theoretically, the sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal say yδ(t) as shown in the diagram:

Analog Pulse Modulation

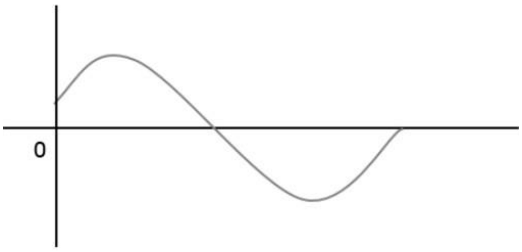
The analog modulation techniques are mainly classified into Pulse Amplitude Modulation, Pulse Duration Modulation/Pulse Width Modulation, and Pulse Position Modulation.

Pulse Amplitude Modulation

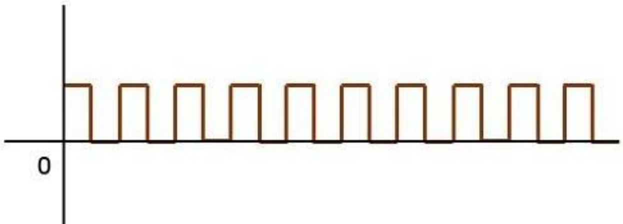
Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. In natural PAM, a signal sampled at the Nyquist rate is reconstructed, by passing it through an efficient Low Pass Frequency (LPF) with exact cutoff frequency

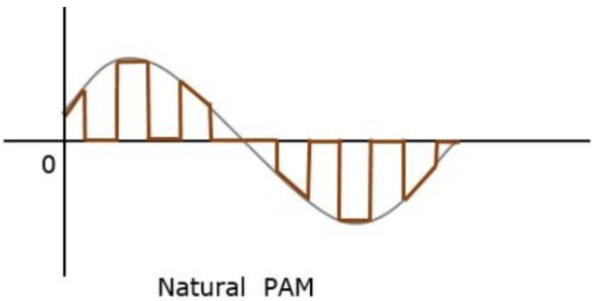
The following figures explain the Pulse Amplitude Modulation.



Modulating signal

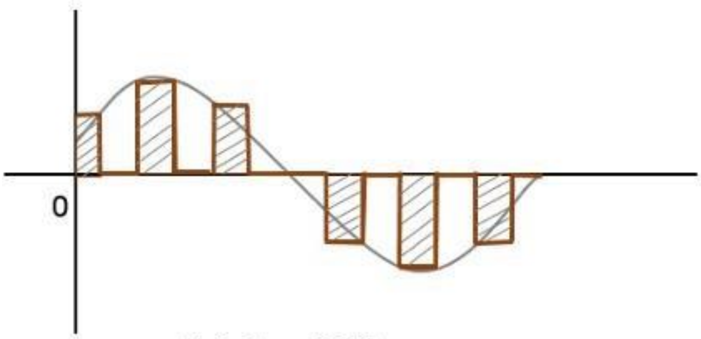


Carrier Pulse train



Natural PAM

Though the PAM signal is passed through an LPF, it cannot recover the signal without distortion. Hence to avoid this noise, flat-top sampling is done as shown in the following figure.



Flat-Top PAM

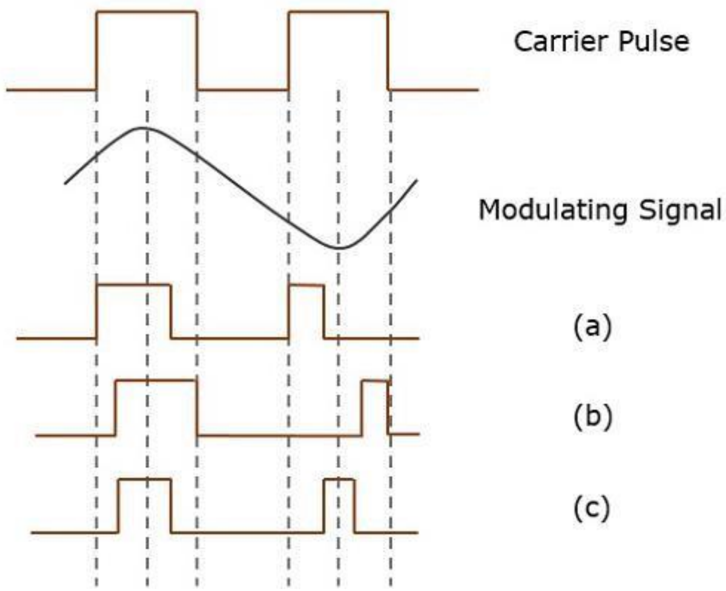
Flat-top sampling is the process in which sampled signal can be represented in pulses for which the amplitude of the signal cannot be changed with respect to the analog signal, to be sampled. The tops of amplitude remain flat. This process simplifies the circuit design.

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.



There are three variations of PWM. They are −

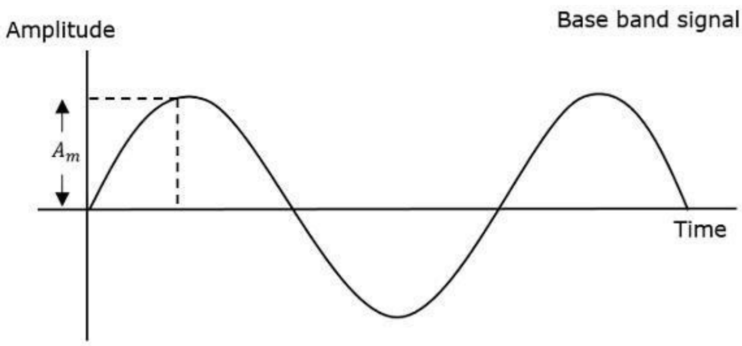
- The leading edge of the pulse being constant, the trailing edge varies according to the message signal.
- The trailing edge of the pulse being constant, the leading edge varies according to the message signal.
- The center of the pulse being constant, the leading edge and the trailing edge varies according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

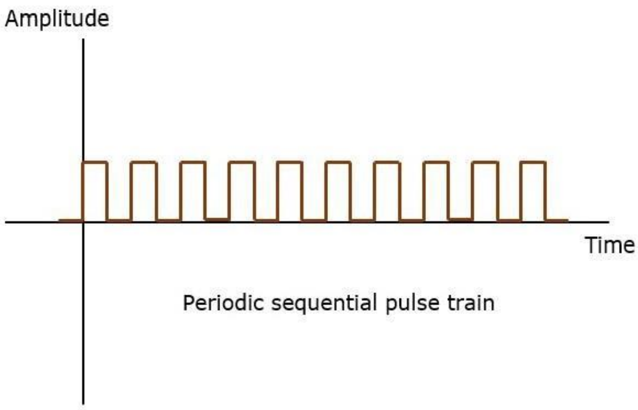
These three types are shown in the above given figure, with timing slots.

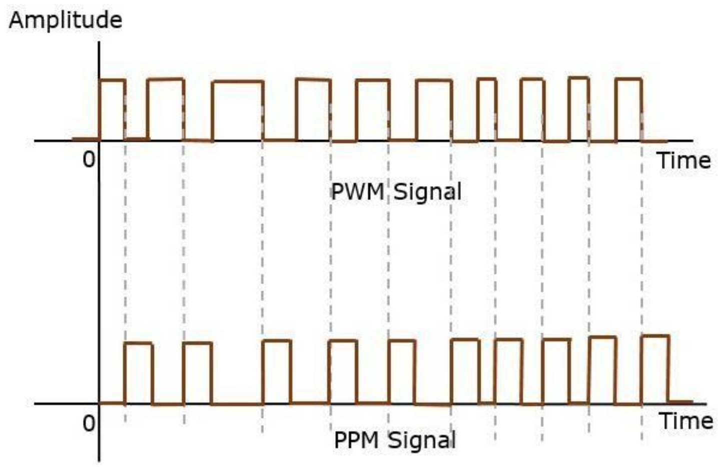
Pulse Position Modulation

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.







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

The comparison between the above modulation processes is presented in a single table.

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

Analog to Digital Conversion

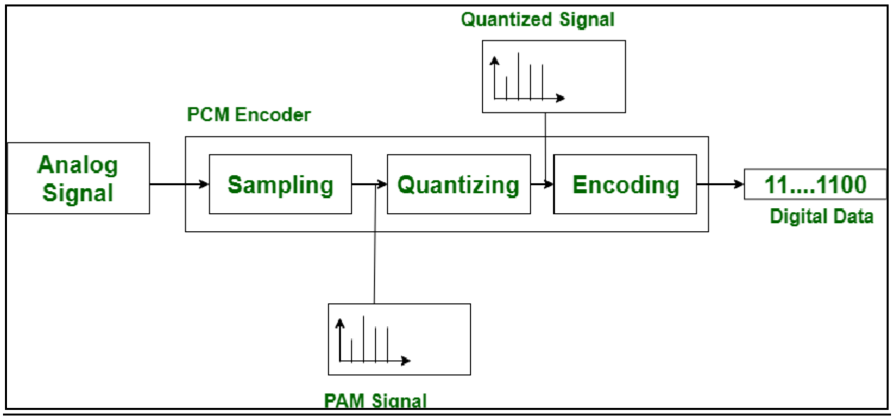
Analog data such as voice, video and music can be converted into digital signal communication through transmission media. This allows the use of modern digital transmission and switching equipment's. The device used for conversion of analog data to digital signal and vice versa is called a coder (coder-decoder).

There are two basic approaches: - Pulse Code Modulation (PCM) - Delta Modulation (DM)

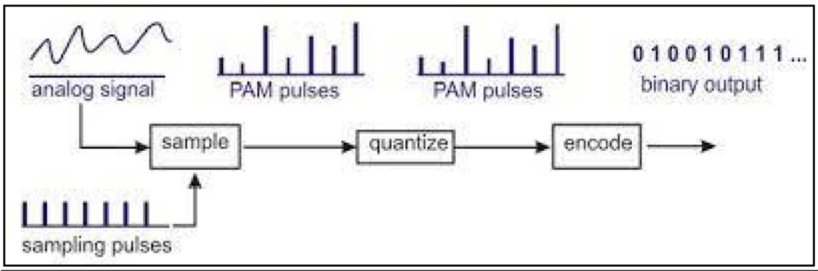
Pulse Code modulation

Pulse Code Modulation involves the following three basic steps .

- Sampling – PAM
- Quantization
- Line coding



Sampling: This process is based on Shannon's sampling theorem. Numbers of samples of the signal are taken at regular intervals, at a rate higher than twice the highest significant signal frequency. This basic step is known as Pulse Amplitude Modulation (PAM) as shown in below figure. For example, during the sampling of voice data, in the frequency range 300 to 4000 Hz, 8000 samples per second are sufficient for the coding.



Quantization: The PAM samples are quantized and approximated to n-bit integer by using analog-to-digital converter. For example, if n = 4, then there are 16 (=24) levels available for approximating the PAM signals. This process introduces an error are known as quantization error. Quantization error depends on step size. Use of uniform step size leads to poorer S/N ratio for small amplitude signals. With the constraint of a fixed number of levels, the situation can be improved using variable step size. The effect of quantization error can be minimized by using a technique known as companding. In this case, instead of using uniform stage sizes, the steps are close together at low signal amplitude and further apart at high signal amplitude as shown in Fig. 2.4.18. It uses a compressor before encoding and expander after decoding. This helps to improve the S/N ratio of the signal.

Encoding

The digitization of the analog signal is done by the encoder. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an n bit code. Encoding also minimizes the bandwidth used.

DELTA MODULATION :

Since PCM is a very complex technique, other techniques have been developed to reduce the complexity of PCM. The simplest is delta Modulation. Delta Modulation finds the change from the previous value.

Unit-6: DIGITALMODULATION TECHNIQUES.

Multiplexing is a technique used to combine and send the multiple data streams over a single medium. The process of combining the data streams is known as multiplexing and hardware used for multiplexing is known as a multiplexer.

Multiplexing is achieved by using a device called Multiplexer (**MUX**) that combines n input lines to generate a single output line. Multiplexing follows many-to-one, i.e., n input lines and one output line.

Demultiplexing is achieved by using a device called Demultiplexer (**DEMUX**) available at the receiving end. DEMUX separates a signal into its component signals (one input and n outputs). Therefore, we can say that demultiplexing follows the one-to-many approach.

he transmission medium is used to send the signal from sender to receiver. The medium can only have one signal at a time.

If there are multiple signals to share one medium, then the medium must be divided in such a way that each signal is given some portion of the available bandwidth. For example: If there are 10 signals and bandwidth of medium is100 units, then the 10 unit is shared by each signal.

When multiple signals share the common medium, there is a possibility of collision. Multiplexing concept is used to avoid such collision.

Transmission services are very expensive.

Multiplexing is the process of combining multiple signals into one signal, over a shared medium. If the analog signals are multiplexed, then it is called as analog multiplexing. Similarly, if the digital signals are multiplexed, then it is called as digital multiplexing.

Multiplexing was first developed in telephony. A number of signals were combined to send through a single cable. The process of multiplexing divides a communication channel into several number of logical channels, allotting each one for a different message signal or a data stream to be transferred. The device that does multiplexing can be called as Multiplexer or MUX.

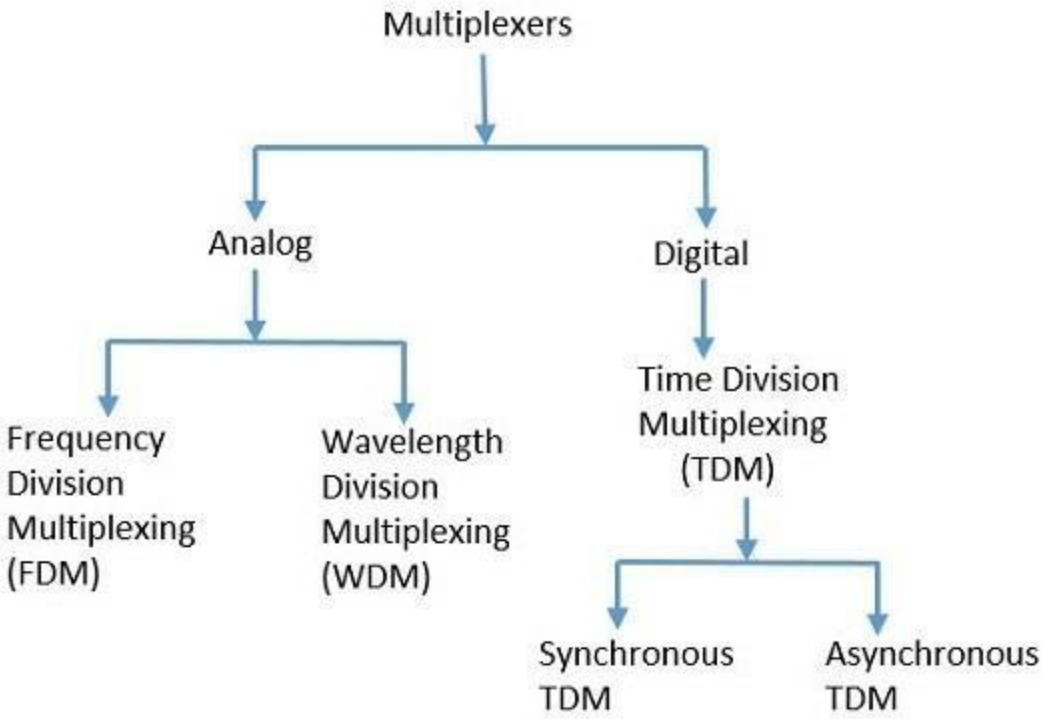
The reverse process, i.e., extracting the number of channels from one, which is done at the receiver is called as de-multiplexing. The device that does de-multiplexing can be called as de-multiplexer or DEMUX.

The following figures illustrates the concept of MUX and DEMUX. Their primary use is in the field of communications.

Multiplexing and Demultiplexing

Types of Multiplexers

There are mainly two types of multiplexers, namely analog and digital. They are further divided into Frequency Division Multiplexing (FDM), Wavelength Division Multiplexing (WDM), and Time Division Multiplexing (TDM). The following figure gives a detailed idea about this classification.

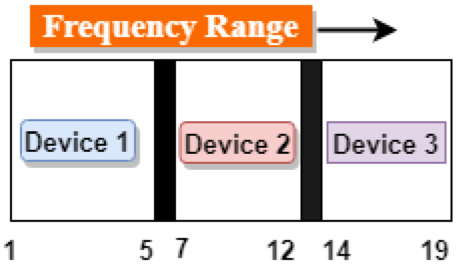


There are many types of multiplexing techniques. Out of which, we have the main types with general classification, mentioned in the above figure. Let us take a look at them individually.

Frequency-division Multiplexing (FDM)

It is an analog technique.

Frequency Division Multiplexing is a technique in which the available bandwidth of a single transmission medium is subdivided into several channels.



In the above diagram, a single transmission medium is subdivided into several frequency channels, and each frequency channel is given to different devices. Device 1 has a frequency channel of range from 1 to 5.

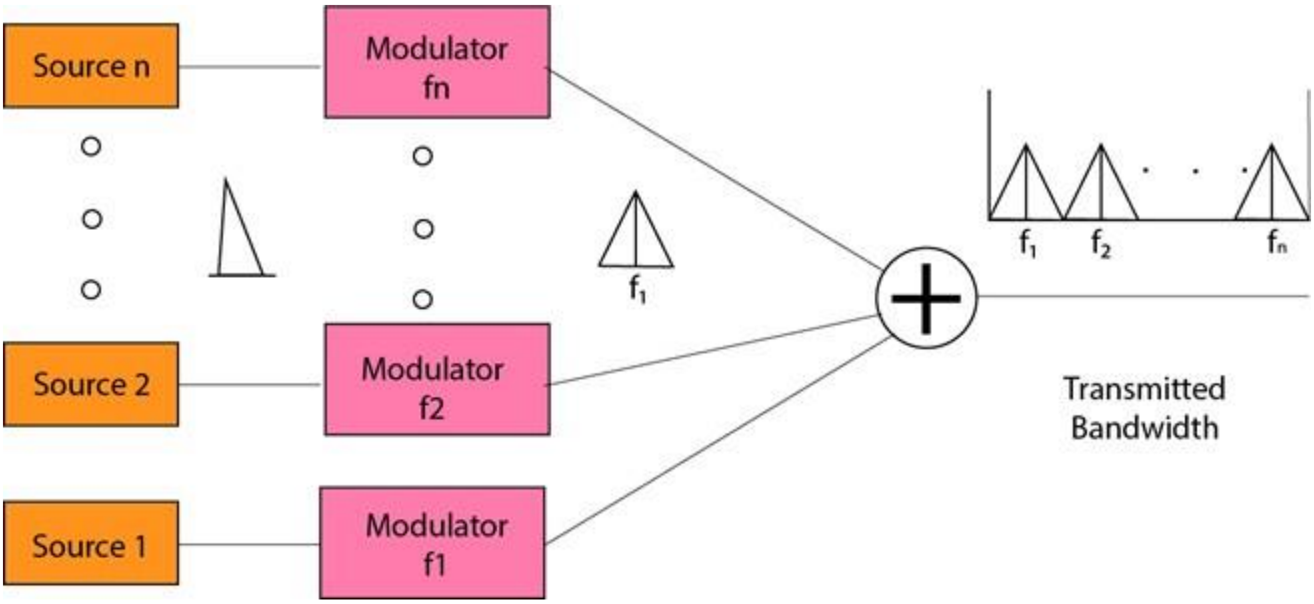
The input signals are translated into frequency bands by using modulation techniques, and they are combined by a multiplexer to form a composite signal.

The main aim of the FDM is to subdivide the available bandwidth into different frequency channels and allocate them to different devices.

Using the modulation technique, the input signals are transmitted into frequency bands and then combined to form a composite signal.

The carriers which are used for modulating the signals are known as sub-carriers. They are represented as f1,f2..fn.

FDM is mainly used in radio broadcasts and TV networks.



Advantages of FDM:

FDM is used for analog signals.

FDM process is very simple and easy modulation.

A Large number of signals can be sent through an FDM simultaneously.

It does not require any synchronization between sender and receiver.

Disadvantages of FDM:

FDM technique is used only when low-speed channels are required.

It suffers the problem of crosstalk.

A Large number of modulators are required.

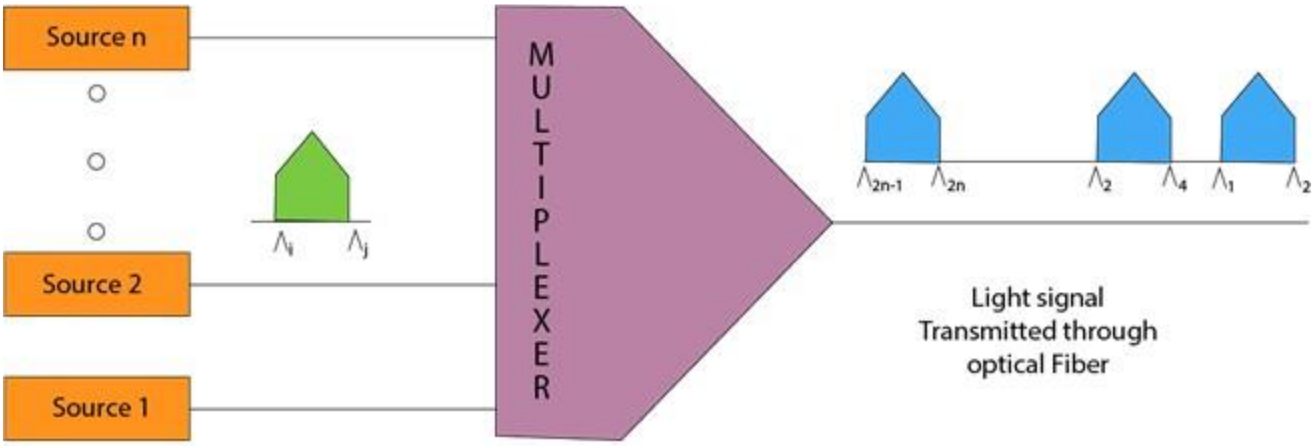
It requires a high bandwidth channel.

Applications of FDM:

FDM is commonly used in TV networks.

It is used in FM and AM broadcasting. Each FM radio station has different frequencies, and they are multiplexed to form a composite signal. The multiplexed signal is transmitted in the air.

Wavelength Division Multiplexing (WDM)



WDM Transmitter

Wavelength Division Multiplexing is same as FDM except that the optical signals are transmitted through the fibre optic cable.

WDM is used on fibre optics to increase the capacity of a single fibre.

It is used to utilize the high data rate capability of fibre optic cable.

It is an analog multiplexing technique.

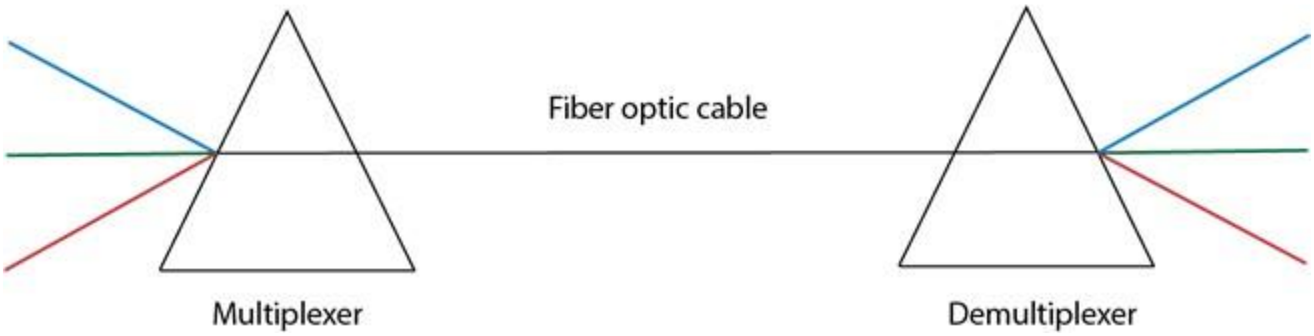
Optical signals from different source are combined to form a wider band of light with the help of multiplexer.

At the receiving end, demultiplexer separates the signals to transmit them to their respective destinations.

Multiplexing and Demultiplexing can be achieved by using a prism.

Prism can perform a role of multiplexer by combining the various optical signals to form a composite signal, and the composite signal is transmitted through a fibre optical cable.

Prism also performs a reverse operation, i.e., demultiplexing the signal.



Time Division Multiplexing

It is a digital technique.

In Frequency Division Multiplexing Technique, all signals operate at the same time with different frequency, but in case of Time Division Multiplexing technique, all signals operate at the same frequency with different time.

In Time Division Multiplexing technique, the total time available in the channel is distributed among different users. Therefore, each user is allocated with different time interval known as a Time slot at which data is to be transmitted by the sender.

A user takes control of the channel for a fixed amount of time.

In Time Division Multiplexing technique, data is not transmitted simultaneously rather the data is transmitted one-by-one.

In TDM, the signal is transmitted in the form of frames. Frames contain a cycle of time slots in which each frame contains one or more slots dedicated to each user.

It can be used to multiplex both digital and analog signals but mainly used to multiplex digital signals.

There are two types of TDM:

- **Synchronous TDM**
- **Asynchronous TDM**

Synchronous TDM

A Synchronous TDM is a technique in which time slot is preassigned to every device.

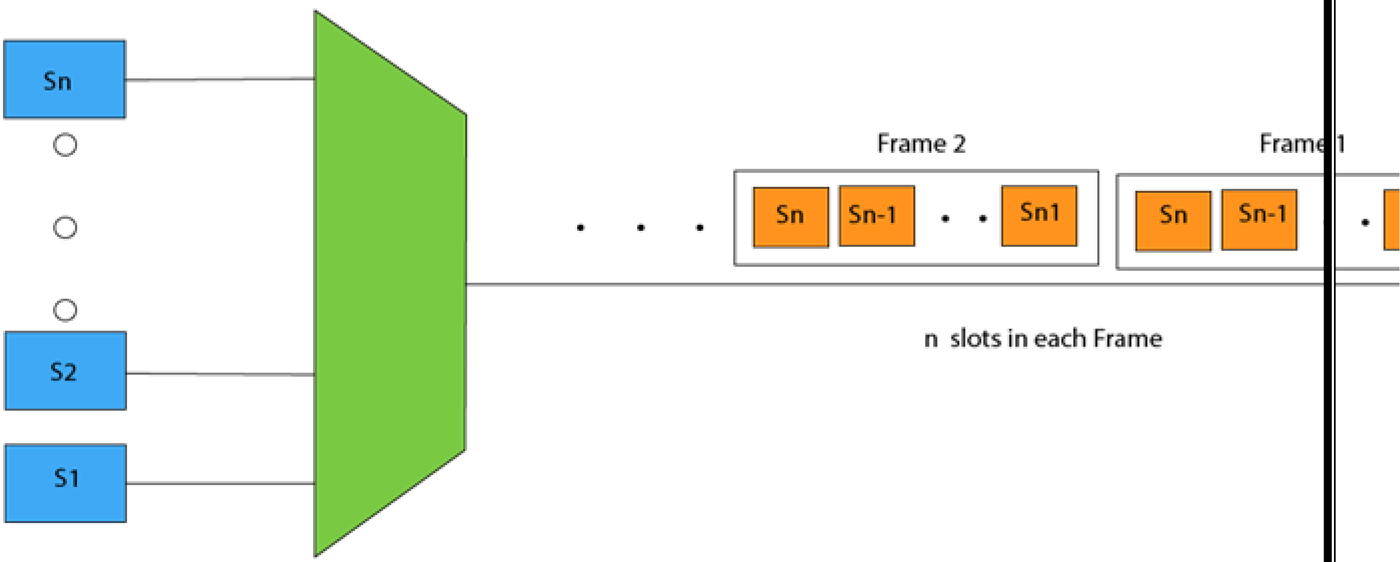
In Synchronous TDM, each device is given some time slot irrespective of the fact that the device contains the data or not.

If the device does not have any data, then the slot will remain empty.

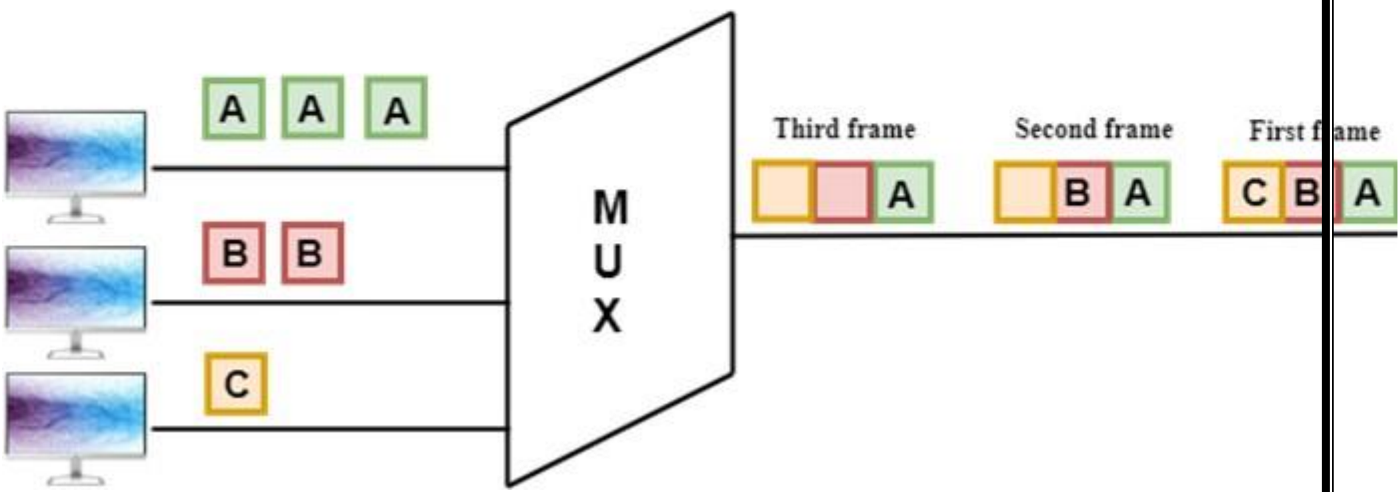
In Synchronous TDM, signals are sent in the form of frames. Time slots are organized in the form of frames. If a device does not have data for a particular time slot, then the empty slot will be transmitted.

The most popular Synchronous TDM are T-1 multiplexing, ISDN multiplexing, and SONET multiplexing.

If there are n devices, then there are n slots.



Concept Of Synchronous TDM



In the above figure, the Synchronous TDM technique is implemented. Each device is allocated with some time slot. The time slots are transmitted irrespective of whether the sender has data to send or not.

Disadvantages Of Synchronous TDM:

The capacity of the channel is not fully utilized as the empty slots are also transmitted which is having no data. In the above figure, the first frame is completely filled, but in the last two frames, some slots are empty. Therefore, we can say that the capacity of the channel is not utilized efficiently.

The speed of the transmission medium should be greater than the total speed of the input lines. An alternative approach to the Synchronous TDM is Asynchronous Time Division Multiplexing.

Asynchronous TDM

An asynchronous TDM is also known as Statistical TDM.

An asynchronous TDM is a technique in which time slots are not fixed as in the case of Synchronous TDM. Time slots are allocated to only those devices which have the data to send. Therefore, we can say that Asynchronous Time Division multiplexor transmits only the data from active workstations.

An asynchronous TDM technique dynamically allocates the time slots to the devices.

In Asynchronous TDM, total speed of the input lines can be greater than the capacity of the channel.

Asynchronous Time Division multiplexor accepts the incoming data streams and creates a frame that contains only data with no empty slots.

In Asynchronous TDM, each slot contains an address part that identifies the source of the data.

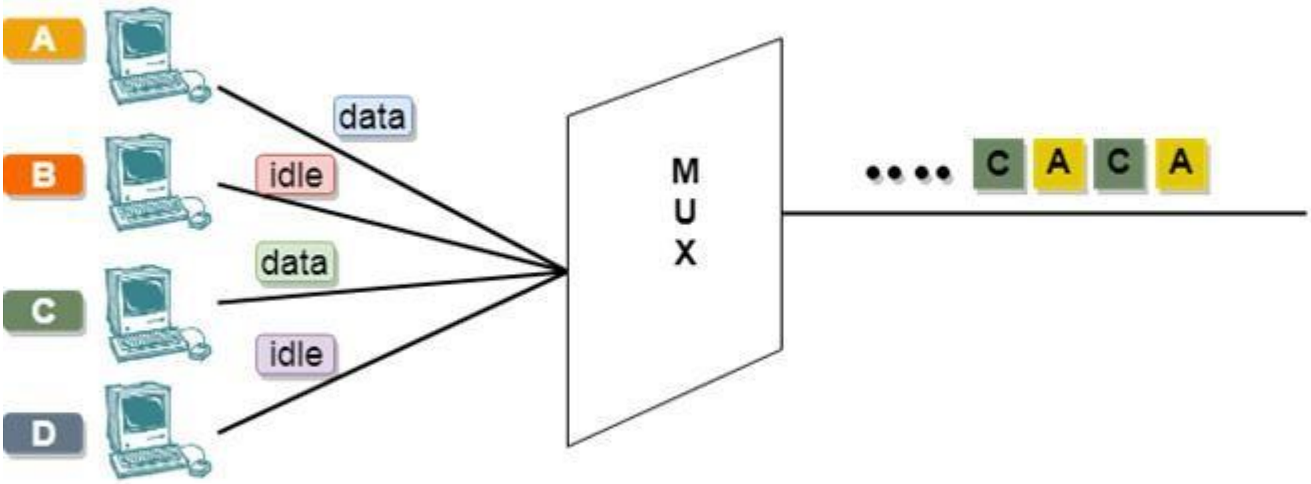
ADDRESS	DATA

The difference between Asynchronous TDM and Synchronous TDM is that many slots in Synchronous TDM are unutilized, but in Asynchronous TDM, slots are fully utilized. This leads to the smaller transmission time and efficient utilization of the capacity of the channel.

In Synchronous TDM, if there are n sending devices, then there are n time slots. In Asynchronous TDM, if there are n sending devices, then there are m time slots where m is less than n (m<n).

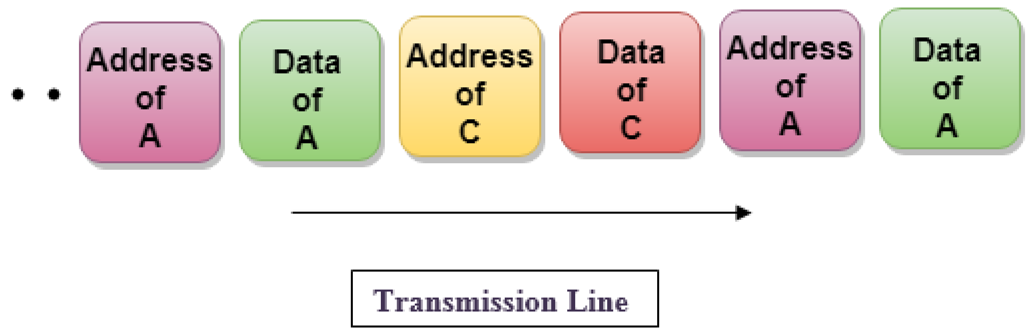
The number of slots in a frame depends on the statistical analysis of the number of input lines.

Concept Of Asynchronous TDM



In the above diagram, there are 4 devices, but only two devices are sending the data, i.e., A and C. Therefore, the data of A and C are only transmitted through the transmission line.

Frame of above diagram can be represented as:



Advantages of digital communication system over Analog system

- 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.

Digital modulation techniques & types.

Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog modulation techniques.

There are many types of digital modulation techniques and also their combinations, depending upon the need.

ASK – Amplitude Shift Keying

The amplitude of the resultant output depends upon the input data whether it should be a zero level or a variation of positive and negative, depending upon the carrier frequency.

FSK – Frequency Shift Keying

The frequency of the output signal will be either high or low, depending upon the input data applied.

PSK – Phase Shift Keying

The phase of the output signal gets shifted depending upon the input. These are mainly of two types, namely Binary Phase Shift Keying BPSKBPSK and Quadrature Phase Shift Keying QPSKQPSK, according to the number of phase shifts. The other one is Differential Phase Shift Keying DPSKDPSK which changes the phase according to the previous value.

M-ary Encoding

M-ary Encoding techniques are the methods where more than two bits are made to transmit simultaneously on a single signal. This helps in the reduction of bandwidth.

The types of M-ary techniques are −

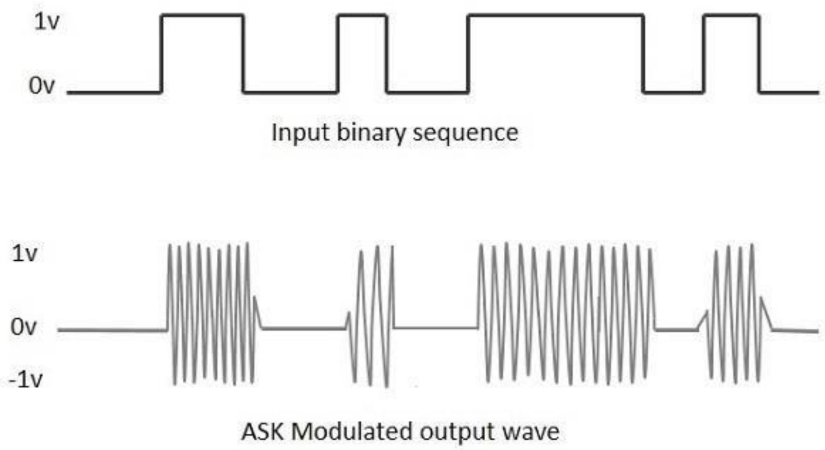
- **M-ary ASK**
- **M-ary FSK**
- **M-ary PSK**

Amplitude Shift Keying (ASK)

Amplitude Shift Keying (ASK) is a type of Amplitude Modulation which represents the binary data in the form of variations in the amplitude of a signal.

Any modulated signal has a high frequency carrier. The binary signal when ASK modulated, gives a zero value for Low input while it gives the carrier output for High input.

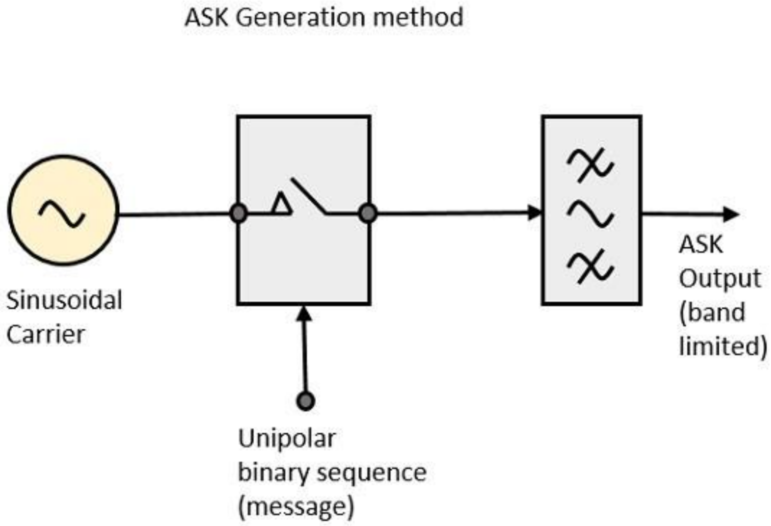
The following figure represents ASK modulated waveform along with its input.



To find the process of obtaining this ASK modulated wave, let us learn about the working of the ASK modulator.

ASK Modulator

The ASK modulator block diagram comprises of the carrier signal generator, the binary sequence from the message signal and the band-limited filter. Following is the block diagram of the ASK Modulator.



The carrier generator, sends a continuous high-frequency carrier. The binary sequence from the message signal makes the unipolar input to be either High or Low. The high signal closes the switch, allowing a carrier wave. Hence, the output will be the carrier signal at high input. When there is low input, the switch opens, allowing no voltage to appear. Hence, the output will be low.

The band-limiting filter, shapes the pulse depending upon the amplitude and phase characteristics of the band-limiting filter or the pulse-shaping filter.

ASK Demodulator

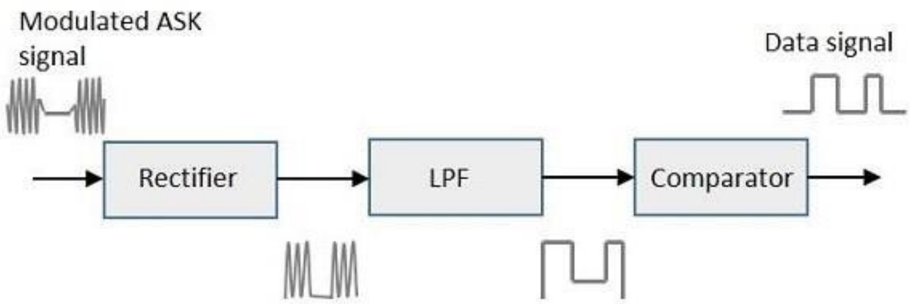
There are two types of ASK Demodulation techniques. They are −

- **Asynchronous ASK Demodulation/detection**
- **Synchronous ASK Demodulation/detection**

The clock frequency at the transmitter when matches with the clock frequency at the receiver, it is known as a Synchronous method, as the frequency gets synchronized. Otherwise, it is known as Asynchronous.

Asynchronous ASK Demodulator

The Asynchronous ASK detector consists of a half-wave rectifier, a low pass filter, and a comparator. Following is the block diagram for the same.

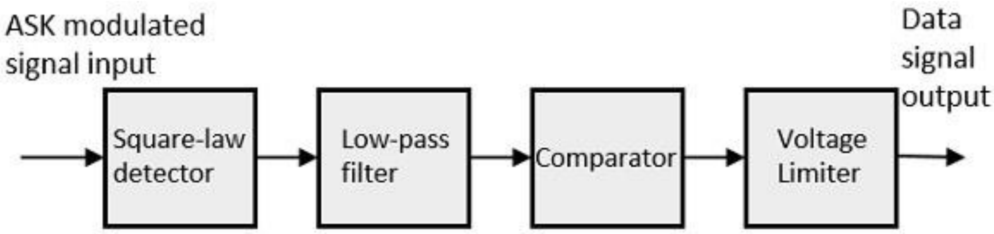


Asynchronous ASK detector

The modulated ASK signal is given to the half-wave rectifier, which delivers a positive half output. The low pass filter suppresses the higher frequencies and gives an envelope detected output from which the comparator delivers a digital output.

Synchronous ASK Demodulator

Synchronous ASK detector consists of a Square law detector, low pass filter, a comparator, and a voltage limiter. Following is the block diagram for the same.



Synchronous ASK detector

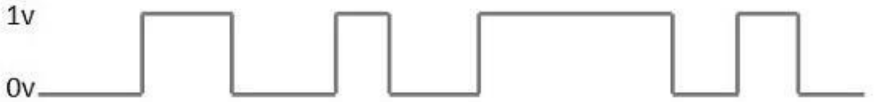
The ASK modulated input signal is given to the Square law detector. A square law detector is one whose output voltage is proportional to the square of the amplitude modulated input voltage. The low pass filter minimizes the higher frequencies. The comparator and the voltage limiter help to get a clean digital output.

Frequency Shift Keying (FSK)

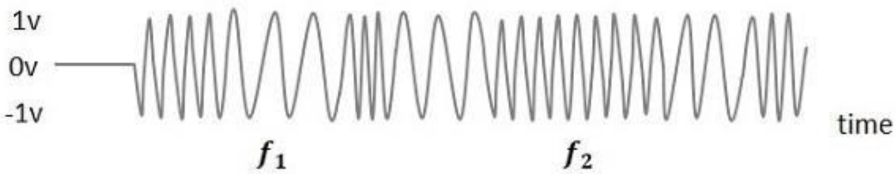
Frequency Shift Keying (FSK) is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. FSK is a scheme of frequency modulation.

The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input. The binary 1s and 0s are called Mark and Space frequencies.

The following image is the diagrammatic representation of FSK modulated waveform along with its input.



Input binary sequence

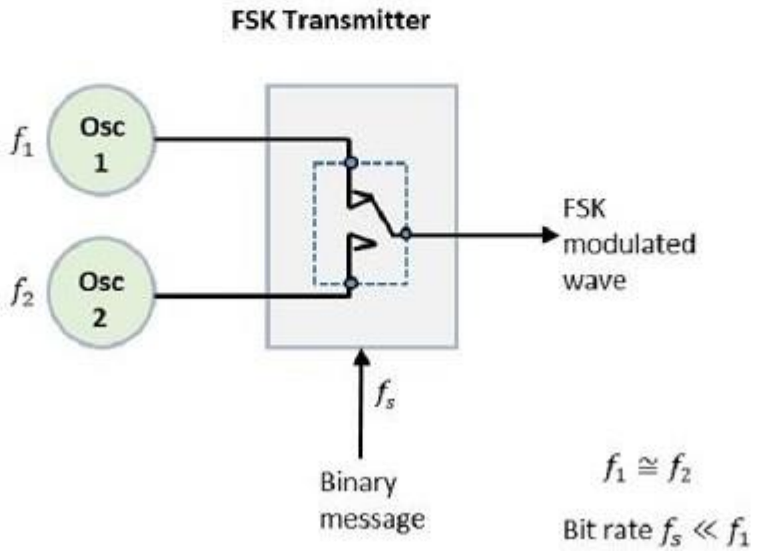


FSK Modulated output wave

To find the process of obtaining this FSK modulated wave, let us know about the working of a FSK modulator.

FSK Modulator

The FSK modulator block diagram comprises of two oscillators with a clock and the input binary sequence. Following is its block diagram.



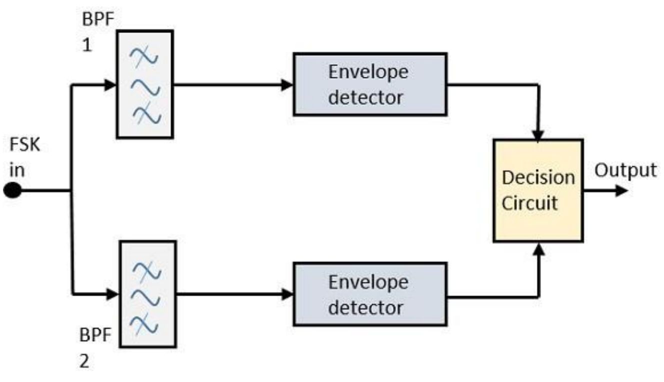
The two oscillators, producing a higher and a lower frequency signals, are connected to a switch along with an internal clock. To avoid the abrupt phase discontinuities of the output waveform during the transmission of the message, a clock is applied to both the oscillators, internally. The binary input sequence is applied to the transmitter so as to choose the frequencies according to the binary input.

FSK Demodulator

There are different methods for demodulating a FSK wave. The main methods of FSK detection are asynchronous detector and synchronous detector. The synchronous detector is a coherent one, while asynchronous detector is a non-coherent one.

Asynchronous FSK Detector

The block diagram of Asynchronous FSK detector consists of two band pass filters, two envelope detectors, and a decision circuit. Following is the diagrammatic representation.

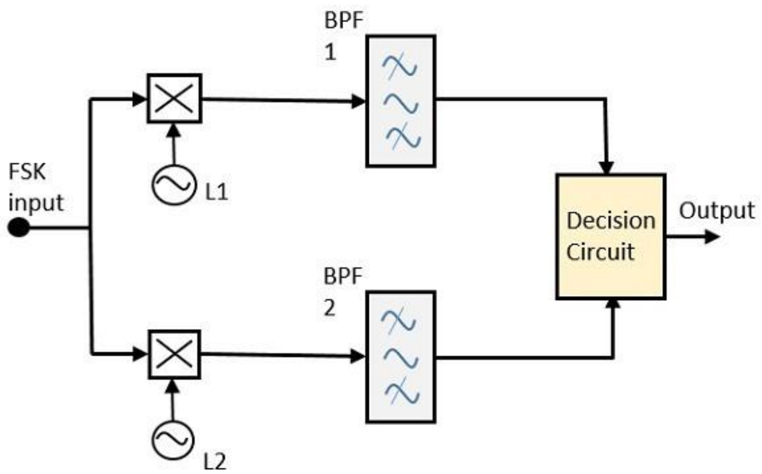


The FSK signal is passed through the two Band Pass Filters BPFsBPFs, tuned to Space and Mark frequencies. The output from these two BPFs look like ASK signal, which is given to the envelope detector. The signal in each envelope detector is modulated asynchronously.

The decision circuit chooses which output is more likely and selects it from any one of the envelope detectors. It also re-shapes the waveform to a rectangular one.

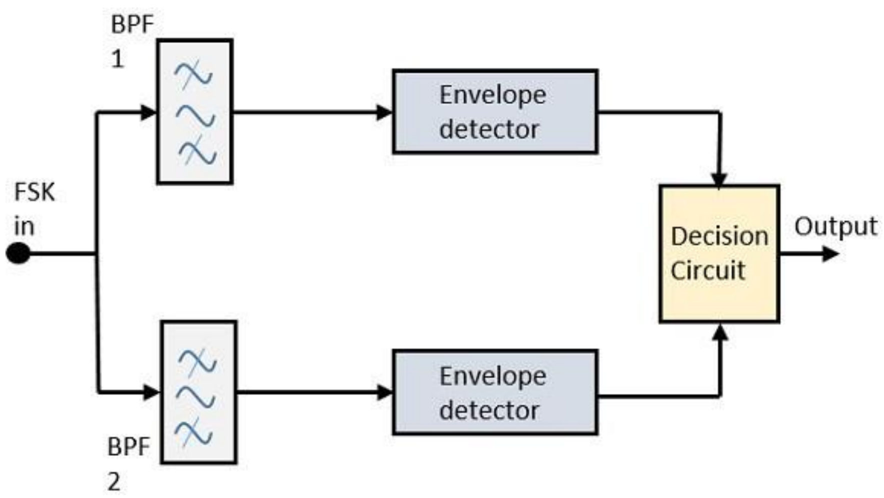
Synchronous FSK Detector

The block diagram of Synchronous FSK detector consists of two mixers with local oscillator circuits, two band pass filters and a decision circuit. Following is the diagrammatic representation.



The FSK signal input is given to the two mixers with local oscillator circuits. These two are connected to two band pass filters. These combinations act as demodulators and the decision circuit chooses which output is more likely and selects it from any one of the detectors. The two signals have a minimum frequency separation.

For both of the demodulators, the bandwidth of each of them depends on their bit rate. This synchronous demodulator is a bit complex than asynchronous type demodulators.



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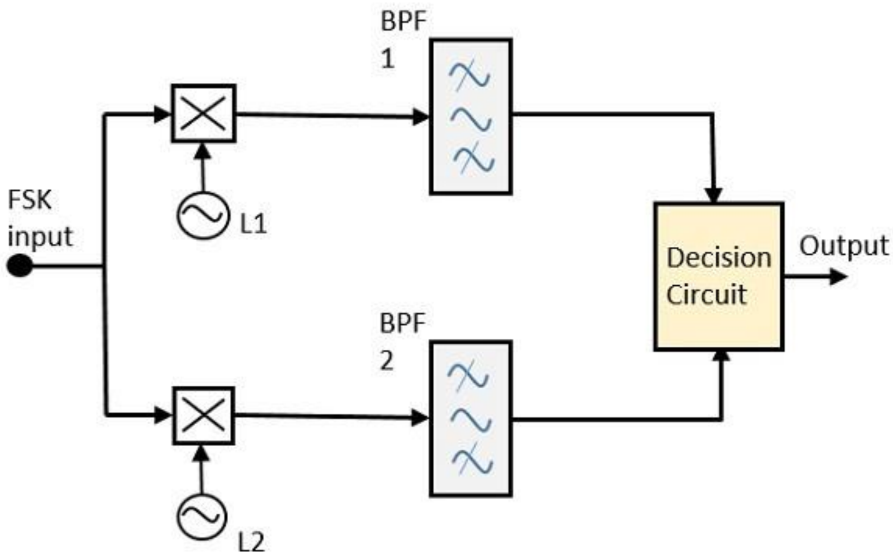

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Phase Shift Keying PSK

PSK is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

PSK is of two types, depending upon the phases the signal gets shifted. They are −

Binary Phase Shift Keying (BPSK)

This is also called as 2-phase PSK or Phase Reversal Keying. In this technique, the sine wave carrier takes two phase reversals such as 0° and 180°.

BPSK is basically a Double Side Band Suppressed Carrier DSBSCDSBSC modulation scheme, for message being the digital information.

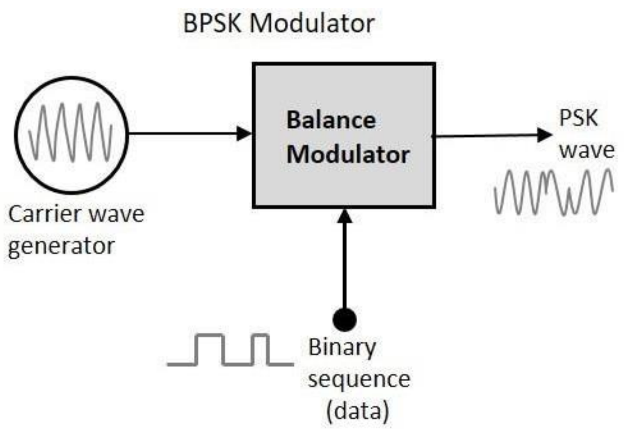
Quadrature Phase Shift Keying (QPSK)

This is the phase shift keying technique, in which the sine wave carrier takes four phase reversals such as 0°, 90°, 180°, and 270°.

If this kind of techniques are further extended, PSK can be done by eight or sixteen values also, depending upon the requirement.

BPSK Modulator

The block diagram of Binary Phase Shift Keying consists of the balance modulator which has the carrier sine wave as one input and the binary sequence as the other input. Following is the diagrammatic representation.



The modulation of BPSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be 0° and for a high input, the phase reversal is of 180°.

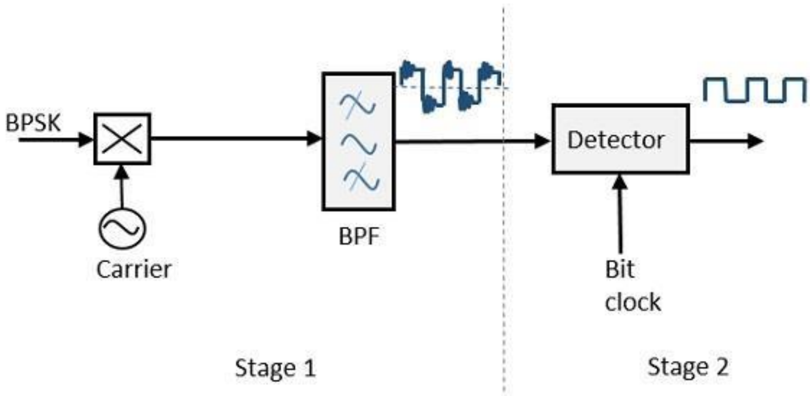
Following is the diagrammatic representation of BPSK Modulated output wave along with its given input.

The output sine wave of the modulator will be the direct input carrier or the inverted 180°phaseshifted180°phaseshifted input carrier, which is a function of the data signal.

BPSK Demodulator

The block diagram of BPSK demodulator consists of a mixer with local oscillator circuit, a bandpass filter, a two-input detector circuit. The diagram is as follows.

BPSK Demodulator



By recovering the band-limited message signal, with the help of the mixer circuit and the band pass filter, the first stage of demodulation gets completed. The base band signal which is band limited is obtained and this signal is used to regenerate the binary message bit stream.

In the next stage of demodulation, the bit clock rate is needed at the detector circuit to produce the original binary message signal. If the bit rate is a sub-multiple of the carrier frequency, then the bit clock regeneration is simplified. To make the circuit easily understandable, a decision-making circuit may also be inserted at the 2nd stage of detection.

Define bit, Baud, symbol & channel capacity formula.(Shannon Theorems)

Data rate refers to the speed of data transfer through a channel. It is generally computed in bits per second (bps). Higher data rates are expressed as Kbps ("Kilo" bits per second, i.e.1000 bps), Mbps ("Mega" bits per second, i.e.1000 Kbps), Gbps ("Giga" bits per second, i.e. 1000 Mbps) and Tbps ("Tera" bits per second, i.e. 1000 Gbps).

One of the main objectives of data communications is to increase the data rate. There are three factors that determine the data rate of a channel:

Bandwidth of the channel

Number of levels of signals that are used

Noise present in the channel

Data rate can be calculated using two theoretical formulae:

- Nyquist Bit Rate – for noiseless channel
- Shannon's Capacity – for noisy channel

Nyquist Bit Rate

Nyquist bit rate was developed by Henry Nyquist who proved that the transmission capacity of even a perfect channel with no noise has a maximum limit.

The theoretical formula for the maximum bit rate is:

maximum bit rate = 2 × Bandwidth × $\log_2 V$

Here, maximum bit rate is calculated in bps

Bandwidth is the bandwidth of the channel

V is the number of discrete levels in the signal

For example, if there is a noiseless channel with a bandwidth of 4 KHz that is transmitting a signal with 4 discrete levels, then the maximum bit rate will be computed as, maximum bit rate = 2 × 4000 × $\log_2 4$ = 16,000 bps = 16 kbps

Shannon's Capacity

Claude Shannon extended Nyquist's work for actual channels that are subject to noise. Noise can be of various types like thermal noise, impulse noise, cross-talks etc. Among all the noise types, thermal noise is unavoidable. The random movement of electrons in the channel creates an extraneous signal not present in the original signal, called the thermal noise. The amount of thermal noise is calculated as the ratio of the signal power to noise power, SNR.

Signal-to-Noise Ratio,

SNR = Average Signal Power/Average Noise Power

Since SNR is the ratio of two powers that varies over a very large range, it is often expressed in decibels, called SNR_{db} and calculated as:

$SNR_{db} = 10 \log_{10} SNR$.

Shannon's Capacity gives the theoretical maximum data rate or capacity of a noisy channel.

It is expressed as:

Capacity = Bandwidth × $\log_2(1+SNR)$

Here, Capacity is the maximum data rate of the channel in bps

Bandwidth is the bandwidth of the channel

SNR is the signal – to – noise ratio

For example, if the bandwidth of a noisy channel is 4 KHz, and the signal to noise ratio is 100, then the maximum bit rate can be computed as:

Capacity = 4000 × log2(1+100) = 26,633 bps = 26.63 kbps

Bit Rate

In telecommunications and computing, "bit rate" is the number of bits communicated or processed per unit of time.

Bit rate is measured in bits per second (symbol: bit/s) and is commonly prefixed with a SI prefix like kilo, mega, Giga, or Tera.

The non-standard term "bps" is sometimes used instead of the conventional sign "bit/s". Therefore, 1 Mbps stands for one million bits per second.

One byte per second (1 B/s) equates to 8 bits per second in most computer and digital communication contexts.

Baud Rate

In telecommunications and electronics, a "baud" is a standard symbol rate measuring unit. It is one of the components that determine the speed of transmission across a data channel.

In pulses per second, symbols per second, it is the unit for symbol rate or modulation rate.

In a digitally modulated signal or a "bd rate line code", it is the number of unique symbol changes (signaling events) made to the transmission medium per second.

The term "baud" refers to the gross bit rate, which is measured in bits per second.

If the system only has two symbols (usually 0 and 1), baud and bit per second (bit/s) are interchangeable.

Both Bit rate and Baud rate are used in data communication. Bit rate and Baud rate are related using below formula.

Bit Rate = Baud rate x the number of bit per baud

T1 Digital Carrier system

T1 digital carrier system is a North American digital multiplexing standard since 1963.

T1 stands for transmission one and specifies a digital carrier system using PCM encoded analog signal.

A T1 carrier system is time division multiplexes PCM encoded samples from 24 voice band channels for transmission over a single metallic wire pair or optical fiber transmission line.

Each voice band channel has BW around 300Hz to 3000KHz.

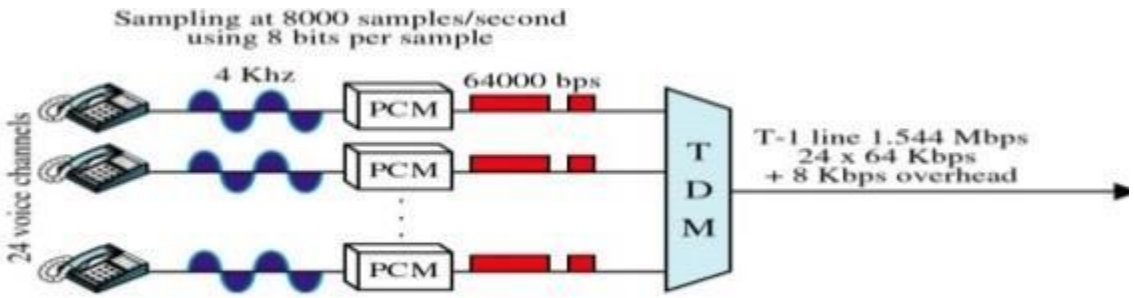


Fig 1: T1 digital system

A multiplexer is simply a digital switch with 24 independent inputs and one time division multiplexed output.

The PCM output signals from 24 voice band channels are sequentially selected and connected through the multiplexer to the transmission line.

With T1 carrier system, there is sampling, encoding and multiplexing of 24 voice band channels.

Each channel contains an 8-bit PCM code and sampled 8000 times a second.

Each channel is sampled at same rate but not at same time.

The figure shows that, each channel is sampled once in each frame, but not at same time.

Each channel's sample is offset from previous channel's sample by 1/24 of total frame time.

Therefore one 64Kbps PCM encoded sample is transmitted for each voice band channel during each frame. The line Speed is calculated as:

$$\frac{24\ Channels}{frame} \times \frac{8Bits}{Channel} = 192\text{bits per frame}$$

$$Thus\ \frac{192\ bits}{frame} \times \frac{8000\ frames}{second} = 1.536Mbps$$

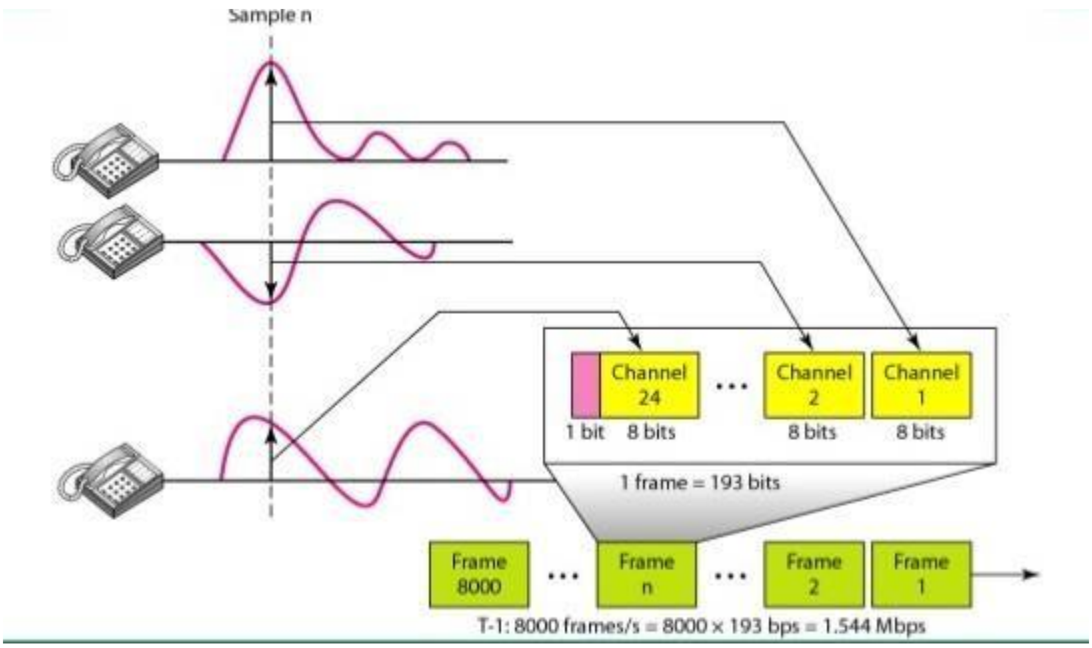


Fig 2: T1 frame structure

An additional bit (called framing bit) is added to each frame.

The framing bit occurs once per frame (8000bps rate) and recovered in receiver, where it is used to maintain frame and sample synchronization between TDM transmitter and receiver.

So each frame contains 193 bits and line speed for T1 digital carrier system is

$$\frac{193\ bits}{frame} \times \frac{8000\ frames}{second} = 1.544Mbps$$

AMI line coding is used for T1 digital Systems