

BRANCH –EE
SEMESTER-6TH

COMMUNICATION ENGINEERING





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01

INTRODUCTION



Elements of an Electrical Communication System

The basic components of electronic communications system are the transmitter, communications channel or medium, receiver, and noise.

Types Of Communication Systems

Depending on Signal specification or technology, the communication system is classified as follows:

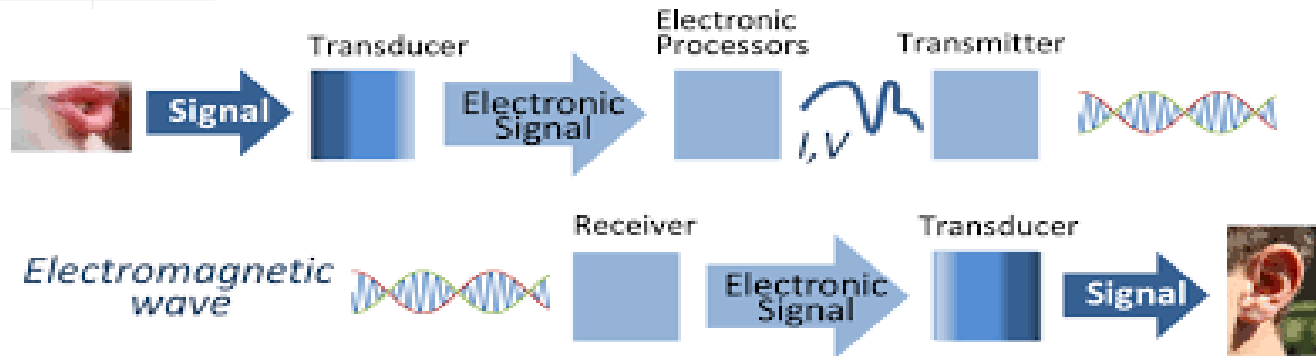
(1) Analog

Analog technology communicates data as electronic signals of varying frequency or amplitude. Broadcast and telephone transmission are common examples of Analog technology.

(2) Digital

In digital technology, the data are generated and processed in two states: High (represented as 1) and Low (represented as 0). Digital technology stores and transmits data in the form of 1s and 0s.

❖ Analog signals (such human voice) or digital signals (binary data) are inputted to the system, processed in the electronic circuits for transmission, and then decoded by the receiver



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

1. Wired (Line communication)

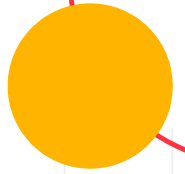
- Parallel wire communication
- Twisted wire communication
- Coaxial cable communication
- Optical fibre communication

2. Wireless (Space communication)

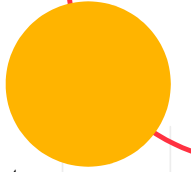
- Ground wave communication
- Skywave communication
- Space wave communication
- Satellite communication

Examples Of Communication Systems

1. Internet
2. Public Switched Telephone network
3. Intranet and Extranet
4. Television



Elements Of Communication Systems



Information

Message or information is the entity that is to be transmitted. It can be in the form of audio, video, temperature, picture, pressure, etc.

Signal

The single-valued function of time that carries the information. The information is converted into an electrical form for transmission.

Transducer

A device or an arrangement that converts one form of energy to the other. An electrical transducer converts physical variables such as pressure, force, temperature into corresponding electrical signal variations. Example: Microphone – converts audio signals into electrical signals.

Photodetector – converts light signals into electrical signals.

Amplifier

The electronic circuit or device that increases the amplitude or the strength of the transmitted signal is called an amplifier. When the signal strength becomes less than the required value, amplification can be done anywhere in between transmitter and receiver. A DC power source will provide for the amplification.

Modulator

As the original message signal cannot be transmitted over a large distance because of their low frequency and amplitude, they are superimposed with high frequency and amplitude wave called carrier wave. This phenomenon of superimposing of message signal with a carrier wave is called modulation. And the resultant wave is a modulated wave which is to be transmitted.



Types of Modulation.

i. Amplitude Modulation (AM)

The process of changing the amplitude of the signal wave by impressing or superimposing it on a high-frequency carrier wave, keeping its frequency constant is called amplitude modulation.

ii. Frequency Modulation (FM)

Frequency modulation is a technique in which the frequency of the message signal is varied by modulating with a carrier wave. It is better than amplitude modulation because it eliminates noise from various sources.

iii. Phase Modulation (PM)

The phase of the carrier wave changes the phase of the signal wave. The phase shift after modulation is dependent on the frequency of the carrier wave as well. Phase modulated waves are immune to noise to a greater extent.

Transmitter

It is the arrangement that processes the message signal into a suitable form for transmission and subsequently reception.

Antenna

An Antenna is a structure or a device that will radiate and receive electromagnetic waves. So, they are used in both transmitters and receivers. An antenna is basically a metallic object, often a collection of wires. The electromagnetic waves are polarised according to the position of the antenna.

Channel

A channel refers to a physical medium such as wire, cables, space through which the signal is passed from the transmitter to the receiver. There are many channel impairments that affect channel performance to a pronounced level. Noise, Attenuation and distortion to mention the major impairments.



CONT...

Channel

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Noise

- Noise is one of the channel imperfection or impairment in the received signal at the destination.
- There are external and internal sources that cause noise. External sources include interference, i.e. interference from nearby transmitted signals (cross talk), interference generated by a natural source such as lightning, solar or cosmic radiation, automobile generated radiation, etc. The external noise can be minimised and eliminated by the appropriate design of the channel, shielding of cables.
- Also by digital transmission external noise can be much minimised.
- Internal sources include noise due to random motion and collision of electrons in the conductors, thermal noise due to diffusion and recombination of charge carriers in other electronic devices.
- Internal noise can be minimised by cooling and using digital technology for transmission.

Attenuation

Attenuation is a problem caused by the medium. When the signal is propagating for a longer distance through a medium, depending on the length of the medium the initial power decreases. The loss in initial power is directly proportional to the length of the medium. Using amplifiers, the signal power is strengthened or amplified so as to reduce attenuation. Also, digital signals are comparatively less prone to attenuation than analogue signals.

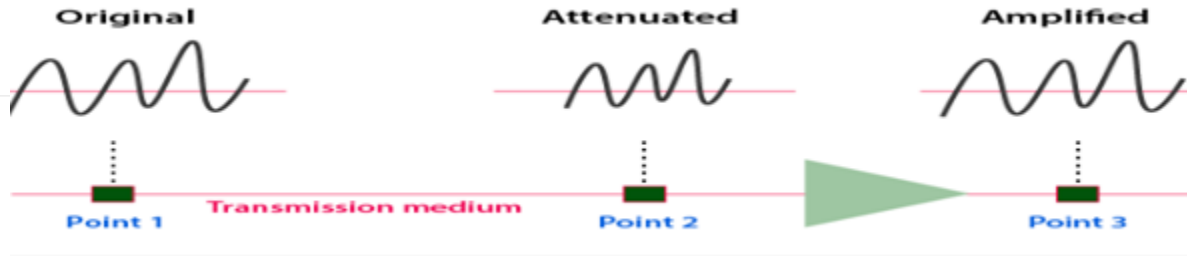


FIGURE OF ATTENUATION



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Distortion

- It is also another type of channel problem. When the signal is distorted, the distorted signal may have frequency and bandwidth different from the transmitted signal.

- The variation in the signal frequency can be linear or non-linear.

•Receiver

- An arrangement that extracts the message or information from the transmitted signal at the output end of the channel and reproduces it in a suitable form as the original message signal is a receiver.

•Demodulator

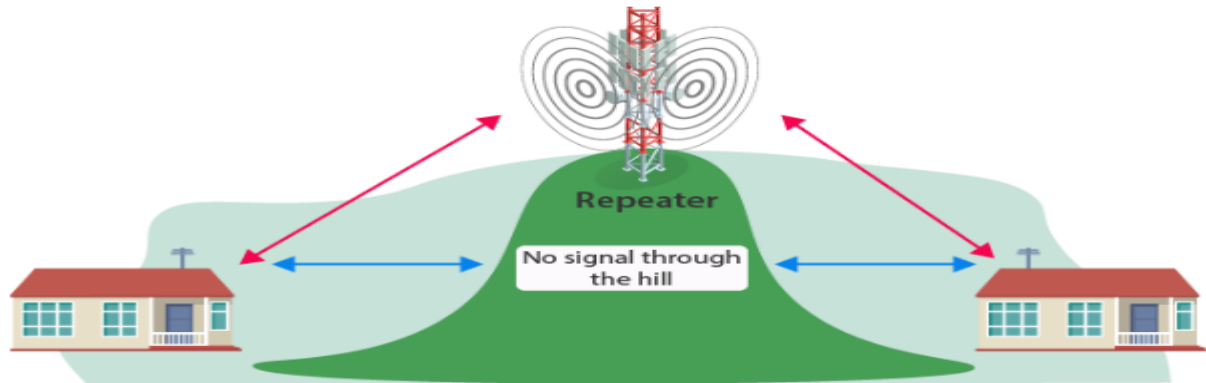
- It is the inverse phenomenon of modulation. The process of separation of message signal from the carrier wave takes place in the demodulator.

- The information is retrieved from the modulated wave.

•Repeaters

- Repeaters are placed at different locations in between the transmitter and receiver.

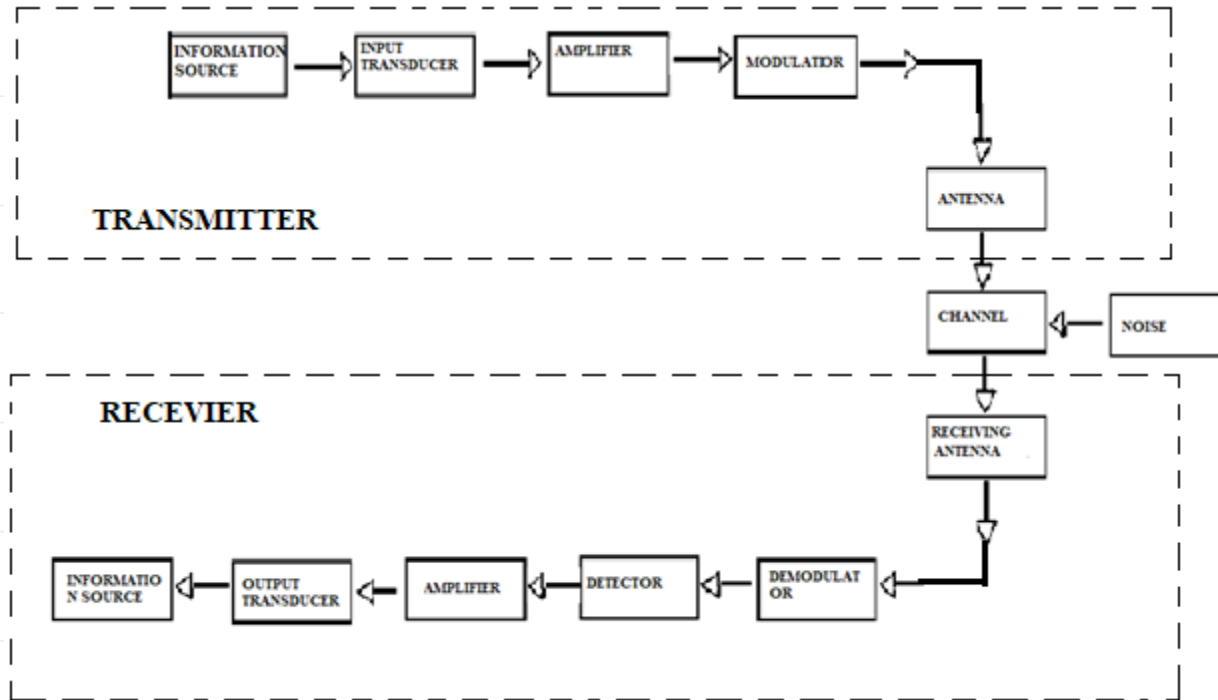
- A repeater receives the transmitted signal, amplifies it and send it to the next repeater without distorting the original signal.



Communication only possible by bouncing the signal through the repeater

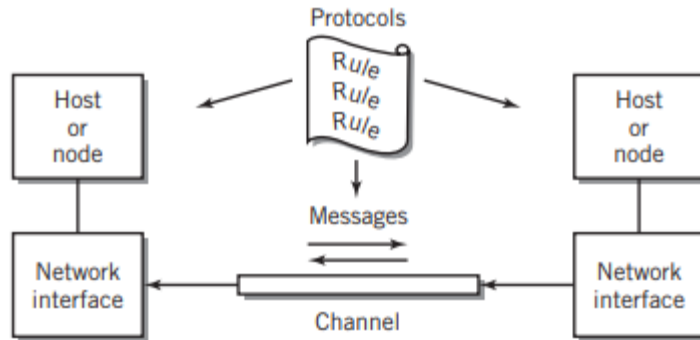


Block Diagram of Communication Systems



Communication Channels

- A **communication channel** refers either to a physical transmission medium such as a wire, or to a logical connection over a multiplexed medium such as a radio channel in telecommunications and computer networking. A channel is used to convey an information signal, for example a digital bit stream, from one or several senders (or transmitters) to one or several receivers.
- A channel has a certain capacity for transmitting information, often measured by its bandwidth in Hz or its data rate in bits per second.

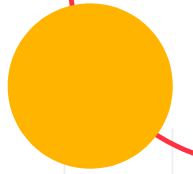


Model of a Communication Channel





Channel performance measures



- These are examples of commonly used channel capacity and performance measures:
- Spectral bandwidth in Hertz
- Symbol rate in baud, pulses/s or symbols/s
- Digital bandwidth bit/s measures: gross bit rate (signalling rate), net bit rate (information rate), channel capacity and maximum throughput
- Link spectral efficiency Signal-to-noise measures: signal-to-interference ratio, carrier-to-interference ratio in decibel
- Bit-error rate (BER), packet-error rate (PER)
- Latency in seconds: propagation time, session time
- Delay jitter



Mathematical Models for Communication Channels Frequency domain analysis of signals and systems

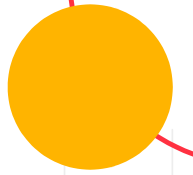
Two types of mathematical tools:

- 1) Time Domain Analysis - Time domain analysis examines the amplitude vs. time characteristics of a measuring signal.
- 2) Frequency Domain Analysis - Frequency domain analysis replaces the measured signal with a group of sinusoids which, when added together, produce a waveform equivalent to the original. - The relative amplitudes, frequencies, and phases of the sinusoids are examined.

Time Domain Analysis

- In time-domain analysis the response of a dynamic system to an input is expressed as a function of time $c(t)$.
- It is possible to compute the time response of a system if the nature of input and the mathematical model of the system are known.
- The time response of a system can be obtained by solving the differential eq. governing the system.
- Alternatively, the response $c(t)$ can be obtained from the transfer function of the system and the input to the system
- Time Domain Analysis
- For a closed loop transfer function, $C(s)/R(s) = G(s)/[1+G(s)H(s)]$ □ Response in s-domain, $C(s) = R(s)*M(s)$
- Response in t-domain, $c(t) = \text{InvLap}[C(s)]$
- Time Domain Analysis Time Domain Specifications
- For specifying the desired performance characteristics of a measurement control system.
- These characteristics of a system of any order may be specified in terms of transient response to a unit step input signal.





Time Domain Analysis

Time Domain Analysis Time Domain Specifications

1. Delay time : It is the time required for the response to reach 50% of the final value in first attempt.
2. Rise time : It is the time required to rise from 0 to 100% of the final value for the under damped system.
3. Peak time : It is the time required for the response to reach the peak of time response or the peak overshoot.
4. Settling time : It is the time required for the response to reach and stay within a specified tolerance band (2% or 5%) of its final value.
5. Peak overshoot : It is the normalized difference between the time response peak and the steady output and is defined as,
6. Steady-state error: It indicates the error between the actual output and desired output as 't' tends to infinity.



Frequency Domain Analysis

Advantages

- Stability of closed loop system can be estimated
- Transfer function of complicated systems can be determined experimentally by frequency tests
- Effects of noise disturbance and parameter variations are relatively easy to visualize.
- Analysis can be extended to certain nonlinear control systems. VIDEO

Frequency Domain Analysis Frequency Domain Specifications

- 1 Resonant Peak- Maximum value of the closed loop transfer function.
- 2 Resonant Frequency- Frequency at which resonant peak occurs.
- 3 Bandwidth- range of frequencies for which the system normalized gain is more than -3db.
- 4 Cut-off rate- It is the slope of the log-magnitude curve near the cut off frequency.
- 5 Gain Margin- The value of gain to be added to system in order to bring the system to the verge of instability.
- 6 Phase Margin- Additional phase lag to be added at the gain cross over freq. in order to bring the system to the verge of instability.

Frequency Domain Analysis Frequency Response Plots

- Frequency domain analysis of a system can be carried either analytically or graphically. The various graphical techniques are 1 Bode Plot 2 Polar Plot 3 Nichols Plot 4 M and N circles
- Frequency response plots are used to determine the frequency domain specifications, to study the stability of the system.
- Example- Measurement from a Solar panel VIDEO



Fourier Transforms

The **Fourier transform** is a mathematical function that can be used to find the base frequencies that a wave is made of. Imagine playing a chord on a piano. When played, the sounds of the notes of the chord mix together and form a sound wave. This works because each of the different note's waves interfere with each other by adding together or canceling out at different points in the wave.

A Fourier transform takes this complex wave and is able to find the frequencies that made it, meaning it can find the notes that a chord is made from.

Continuous Fourier Equation

The Fourier transform is defined by the equation

$$\mathcal{F}\{x(t)\} = X(\xi) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi\xi t} dt$$

And the inverse is

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

These equations allow us to see what frequencies exist in the signal $x(t)$. A more technical phrasing of this is to say these equations allow us to translate a signal between the time domain to the frequency domain. Note that these equations use a ξ (the Greek letter Xi) to imply frequency instead of ω (Omega) which generally refers to angular frequency ($\omega = 2\pi\xi$). The Fourier transform of a time dependent signal produces a frequency dependent function.

A lot of engineers use omega because it is used in transfer functions, but here we are just looking at frequency. If we use the angular frequency instead of frequency, then we would have to apply a factor of 2π to either the transform or the inverse. The general rule is that the unit of the Fourier transform variable is the inverse of the original function's variable.



CONT...

Example Transformations

Let's kick these equations around a bit. Let's try the super simple function $x(t) = 2$. Plugging this equation into the Fourier transform, we get:

$$X(\xi) = \int_{-\infty}^{\infty} 2e^{-j2\pi\xi t} \delta t = \int_{-\infty}^{\infty} 2[\cos(-2\pi\xi t) + j \sin(-2\pi\xi t)] \delta t$$

Integrals around infinity start to behave oddly, so this example will not be mathematically rigorous, but intuitively we can see that the integral of $\sin(\xi t)$ and $\cos(\xi t)$ as t goes from negative infinity to positive infinity should be 0, unless $\xi = 0$. At that point the equation simplified dramatically to:

$$X(0) = \int_{-\infty}^{\infty} 2[\cos(0) + j \sin(0)] \delta t = \int_{-\infty}^{\infty} 2[1 + j \cdot 0] \delta t = \int_{-\infty}^{\infty} 2 \delta t$$

We can write the equation for $X(\xi)$ using the Dirac delta function, $\delta(x)$, which is defined as:

$$\delta(x) = \begin{cases} \infty, & x = 0 \\ 0, & x \neq 0 \end{cases}$$

So, putting it all together, for $x(t) = 2$, $X(\xi) = 2 \delta(\xi)$. This means that the magnitude of $X(\xi)$ is 0 everywhere except at $\xi=0$, where it is roughly 2∞ . A more mathematically rigorous process, which you can find here, rests on the transform of the unit step function, which rests on the transform of an exponential decay.

The purpose here is just to show that the transform of a DC signal will exist only at 0 Hz.

Now let's look at the Fourier transform of a sine wave of frequency 1kHz.

$$X(\xi) = \int_{-\infty}^{\infty} \sin(2\pi 1000t) e^{-j2\pi\xi t} \delta t = \int_{-\infty}^{\infty} \sin(2000\pi t) [\cos(2\pi\xi t) - j \sin(2\pi\xi t)] \delta t$$



CONT...

We can see that this is going to come to zero except for the case where $\xi = \pm 1000$. At that point, using $\xi = 1000$ for this example, the equation becomes:

$$\begin{aligned} X(1000) &= \int_{-\infty}^{\infty} \sin(2000\pi t) [\cos(2000\pi t) - j \sin(2000\pi t)] \delta t \\ &= \int_{-\infty}^{\infty} \sin(2000\pi t) \cos(2000\pi t) \delta t - j \int_{-\infty}^{\infty} \sin^2(2000\pi t) \delta t \end{aligned}$$

We can apply the trigonometric identity of $\sin(kt)\cos(kt) = \sin(2kt)/2$ and $\sin^2(kt) = (1-\cos(2kt))/2$, and we get:

$$\begin{aligned} \int_{-\infty}^{\infty} \frac{1}{2} \sin(4000\pi t) \delta t - j \int_{-\infty}^{\infty} \frac{1}{2} \delta t + j \int_{-\infty}^{\infty} \frac{1}{2} \cos(4000\pi t) \delta t = \\ -j \int_{-\infty}^{\infty} \frac{1}{2} \delta t + 0 + 0 = -j \int_{-\infty}^{\infty} \frac{1}{2} \delta t \end{aligned}$$

Similarly, at $\xi = -1000$, we will get:

$$X(-1000) = j \int_{-\infty}^{\infty} \frac{1}{2} \delta t + 0 + 0$$

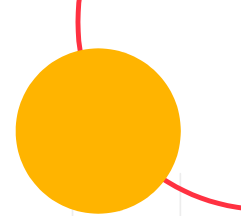
Using the Dirac function, we see that the Fourier transform of a 1kHz sine wave is:

$$\mathcal{F}\{\sin(2000\pi t)\} = X(\xi) = \frac{-j\delta(\xi - 1000)}{2} + \frac{j\delta(\omega + 1000)}{2}$$

We can use the same methods to take the Fourier transform of $\cos(4000\pi t)$, and get:

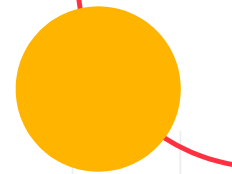
$$\mathcal{F}\{\cos(4000\pi t)\} = X(\xi) = \frac{\delta(\xi - 2000)}{2} + \frac{\delta(\xi + 2000)}{2}$$





A few things jump out here. The first is that the Dirac function has an offset, which means we get the same spike that we saw for $x(t) = 2$, but this time we have spikes at the signal frequency and the negative of the signal frequency. This makes more sense when you remember that $\sin(-\theta) = -\sin(\theta)$, and $\cos(-\theta) = \cos(\theta)$. The second piece that should jump out is that the Fourier transform of the sine function is completely imaginary, while the cosine function only has real parts. This means that the angle of the transform of the sine function, which is the arctan of real over imaginary, is 90° off from the transform of the cosine, just like the sine and cosine functions are 90° off from each other.





The Discrete Time Fourier Transform

A common form of the DFT equation is

$$\mathcal{F}[x[n]] = X[m] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi n m / N}$$

Here, N is the total number of points included in the equation, and m, the input to the transformed function, roughly correlates to the frequency in the same way that n correlates to time with the equation:

$$f_{analysis}[m] = \frac{m f_s}{N}$$

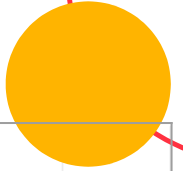
Let's say we're gathering 10 samples sampled at 100uS. For an input signal of $x(t)=2$, we will get $x(0) = 2$, $x(1) = 2$, $x(2) = 2$, etc. Plugging this into the DFT, we get:

$$X[0] = \sum_{n=0}^9 2 e^{-j2\pi n 0 / 10} = \sum_{n=0}^9 2 = 20$$

$$X[1] = \sum_{n=0}^9 2 e^{-j2\pi n 1 / 10} = 2 \sum_{n=0}^9 \left(\cos\left(\frac{2\pi n}{10}\right) - j \sin\left(\frac{2\pi n}{10}\right) \right)$$



Energy and Power



	Energy	Power
Definition	Energy is the capacity to do work. Energy is power integrated over time.	Power is the rate at which work is done, or energy is transmitted.
Unit	joules = watt-seconds or joule = Newton-meter	watt = joules/second
Common symbol(s)	W	P
Example	I left a 60W on for 30 days, which raised my electric bill by 43.2 kWh (kilowatt-hours).	My car's battery can provide 500 amps at 12 volts, which equals 6kW of power.



Different Forms of Power and Energy

There are different forms of energy. These include kinetic energy, thermal, gravitational, electromagnetic, sound, light and elastic. The form of energy is dependent on the frame of reference, and can be transformed into other forms. For instance potential energy is dependent on the position of the object, whereas kinetic energy is the energy required to accelerate an object to a particular speed, and so on. Different forms of power could be electric power, which is the rate at which electrical energy is transferred by a circuit, human power, and optical power.

Transformation of Energy

Various device can be used to convert one form of energy into another. For instance, a battery converts chemical to electric energy, chemical explosion converts chemical energy in to kinetic and thermal energy and so on. Power cannot be converted or transformed.

Measuring Energy vs. Power

- Although it is not possible to directly measure energy, the work done can be defined and measured. The methods involves using a calorimeter, which measures the heat absorbed or released in chemical reactions or physical changes, thermometer, which measures temperature or bolometer that is employed to measure the intensity of radiation. Energy generated can be stored whereas power cannot.
- Since power is energy per unit of time, in theory it can be calculated after measuring the energy used per second. When calculating the real power consumption of an electrical device, it is essential to measure the voltage applied and the current consumed, taking into account the power that is dissipated in the circuit.



Difference in Units

Energy is usually measured in Joules, the other units include ergs and calories. Power is measured in Watts, which is Joules per second, or ergs per second. Machine are usually described by its power rating, the higher the number the more powerful the machine.

Equation

Power is an energy per unit of time. As a rate of change of work done or the energy of a subsystem, power is:

$$P = \frac{W}{t} \quad \text{where } P \text{ is power, } W \text{ is work and } t \text{ is time.}$$



Sampling and Band limited signals

A **signal** is said to be **band-limited** if the amplitude of its spectrum goes to zero for all frequencies beyond some threshold called the **cutoff** frequency.

If a Continuous Time (C.T.) signal is to be uniquely represented and recovered from its samples, then the signal must be band-limited. Further we have to realize that the samples must be sufficiently close and the Sampling Rate must bear certain relation with the highest frequency component of the original signal.



Band-limited signals:

A Band-limited signal is one whose Fourier Transform is non-zero on only a finite interval of the frequency axis.

Specifically, there exists a positive number **B** such that **X(f)** is non-zero only in $f \in [-B, B]$. **B** is also called the Bandwidth of the signal.

To start off, let us first make an observation about the class of Band-limited signals.

Lets consider a Band-limited signal **x(t)** having a Fourier Transform **X(f)**.

Let the interval for which **X(f)** is non-zero be $-B \leq f \leq B$.

Then, $x(t) = \int_{-B}^B X(f) e^{j2\pi ft} df$ converges.

The RHS of the above equation is differentiable with respect to **t** any number of times as the integral is performed on a bounded domain and the integrand is differentiable with respect to t. Further, in evaluating the derivative of the RHS, we can take $\frac{d}{dt}$ inside the integral.

$$\frac{dx(t)}{dt} = \int_{-B}^B (j2\pi f) X(f) e^{j2\pi ft} df$$

In general,

$$\frac{d^n x(t)}{dt^n} = \int_{-B}^B (j2\pi f)^n X(f) e^{j2\pi ft} df$$

This implies that band limited signals are **infinitely differentiable**, therefore, very **smooth** .

We now move on to see how a Band-limited signal can be reconstructed from its samples.

Reconstruction of Time-limited Signals

Consider first a signal **y(t)** that is **time-limited**, i.e. it is non-zero only in $[-T/2, T/2]$.

Its Fourier transform **Y(f)** is given by:



$$= \int_{-\infty}^{\infty} \tilde{y}(t) e^{-j2\pi f t} dt \rightarrow (1)$$



Where $\tilde{y}(t)$ is the periodic extension of $y(t)$ as shown

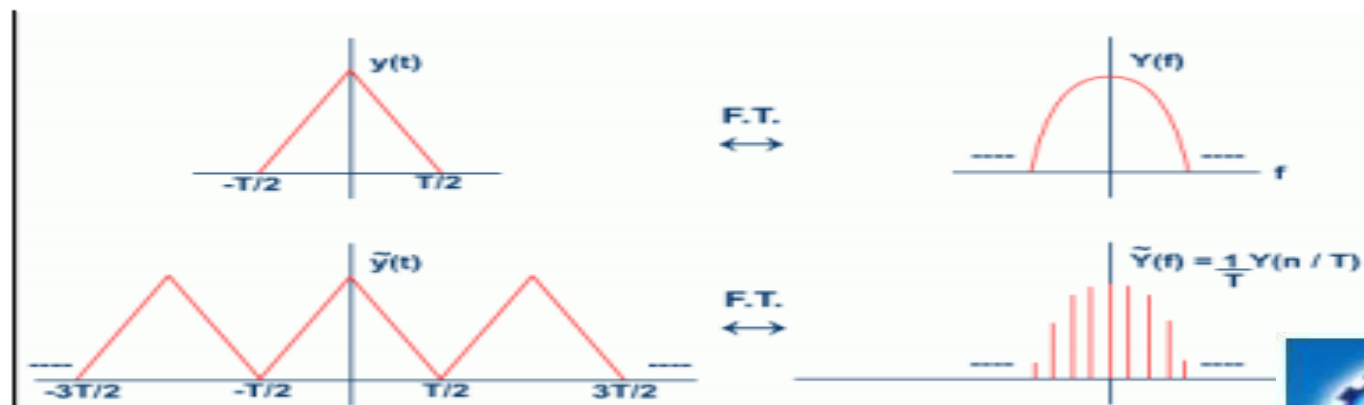
Now, Recall that the coefficients of the Fourier series for a periodic signal $x(t)$ are given by :

$$a_n = \frac{1}{T} \int_{-T/2}^{T/2} x(t) e^{-j2\pi n f_0 t} dt \quad \text{where } f_0 = \frac{1}{T} \quad (2)$$

Comparing (1) and (2), you will find

$$a_n = \frac{1}{T} Y\left(\frac{n}{T}\right)$$

That is, the Fourier Transform of the periodic signal $\tilde{y}(t)$ is nothing but the samples of the original transform.



Therefore, given that; $y(t)$ is time-limited in $[-T/2, T/2]$ and periodic, the entire information about $y(t)$ is contained in just **equispaced samples of its Fourier transform!** It is the dual of this result that is the basis of Sampling and Reconstruction of Band-limited signals :-

Knowing the **Fourier transform is limited** to, say $[-B, B]$, the entire information about the transform (and hence the signal) is contained in just **uniform samples of the (time) signal!**

Reconstruction of Band-limited signals

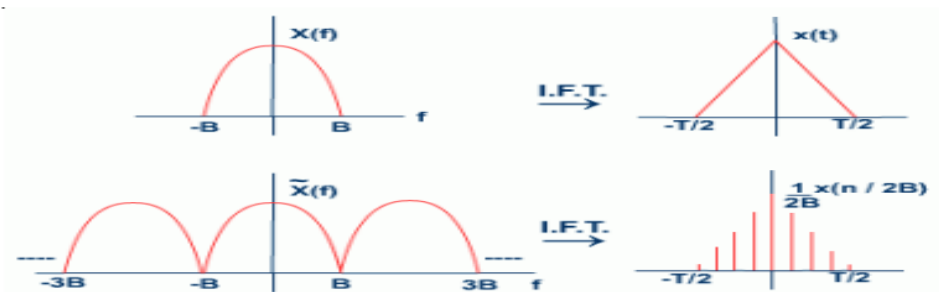
Let us now apply the dual reasoning of the previous discussion to Band-limited signals.

$x(t)$ is Band-limited, with its Fourier transform $X(f)$ being non-zero only in $[-B, B]$. The dual reasoning of the discussion in previous slide will imply that we can reconstruct $X(f)$ perfectly in $[-B, B]$ by using only the samples $x(n / 2B)$. Let's see how.

$$x(t) = \int_{-\infty}^{+\infty} X(f) e^{j2\pi ft} df = \int_{-B}^{+B} X(f) e^{j2\pi ft} df$$

$$\therefore x\left(\frac{n}{2B}\right) = \int_{-B}^{+B} X(f) e^{j2\pi \frac{n}{2B} f} df$$

This time, $\left\{ \frac{1}{2B} x\left(\frac{n}{2B}\right) \right\}$ is the $-n^{\text{th}}$ Fourier series co-efficient of $\tilde{X}(f)$, the periodic extension of $X(f)$.



$$\therefore \tilde{X}(f) = \sum_{n=-\infty}^{+\infty} \frac{1}{2B} x\left(\frac{n}{2B}\right) e^{j2\pi f \frac{n}{2B}}$$

(Fourier series in f -- fundamental period is $2B$ and $\frac{1}{2B} x\left(\frac{n}{2B}\right)$ is the $-n^{\text{th}}$ Fourier series coefficient)

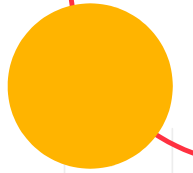


Band pass signals or Pass band

- A passband is the range of frequencies or wavelengths that can pass through a filter. For example, a radio receiver contains a bandpass filter to select the frequency of the desired radio signal out of all the radio waves picked up by its antenna.
- In signal processing, undersampling or **bandpass sampling** is a technique where one samples a **bandpass**-filtered signal at a **sample** rate below its Nyquist rate (twice the upper cutoff frequency), but is still able to reconstruct the signal.
- A **bandpass** random **process** is a random **process** whose power spectrum is contained in a **band** of frequencies centered around a center frequency .
- In a receiver, a **bandpass filter** allows signals within a selected range of frequencies to be heard or decoded, while preventing signals at unwanted frequencies from getting through.
- This band pass value is **calculated** as being $f_r^2 = f_{(UPPER)} \times f_{(LOWER)}$. A **band pass filter** is regarded as a second-order (two-pole) type **filter** because it has “two” reactive components within its circuit structure, then the phase angle will be twice that of the previously seen first-order **filters**, ie, 180° .



Objective Questions



1. Out of the following, which is not an essential element of a communication system?

- a) Transmitter b) Transducer
- c) Receiver d) Communication channel

ANSWER:d

2. A device which provides output in electrical form or it has input in electrical form is called a?

- a) Transmitter b) Receiver
- c) Repeater d) Transducer

ANSWER:d

3. Decrease in signal strength due to energy losses is called?

- a) Distortion b) Interference
- c) Attenuation d) Noise

ANSWER:c

4. The disturbance or distortion in the transmission and processing of message signals is called?

- a) Noise b) Attenuation
- c) Interference d) None of these

ANSWER:a

5. A repeater is a combination of?

- a) Receiver b) Amplifier
- c) Transmitter d) All of these

ANSWER:b



6. Which filter attenuates any frequency outside the pass band?

- a) Band-pass filter
- b) Band-reject filter
- c) Band-stop filter
- d) All of the mentioned

Answer: a

7. Narrow band-pass filters are defined as

- a) $Q < 10$
- b) $Q = 10$
- c) $Q > 10$
- d) None of the mentioned

Answer: c

8. Find the center frequency of wide band-pass filter

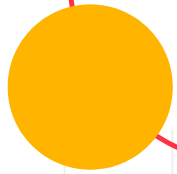
- a) $f_c = \sqrt{f_h \times f_L}$
- b) $f_c = \sqrt{f_h + f_L}$
- c) $f_c = \sqrt{f_h - f_L}$
- d) $f_c = \sqrt{f_h / f_L}$

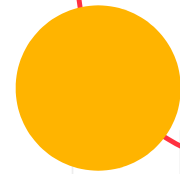
Answer: a

9. Find out the voltage gain magnitude equation for the wide band-pass filter.

- a) $A_{Ft} \times (f/f_L) / \sqrt{[1+(f/f_h)^2] \times [1+(f/f_L)^2]}$
- b) $A_{Ft} / \sqrt{\{[1+(f/f_h)^2] \times [1+(f/f_L)^2]\}}$
- c) $A_{Ft} / \sqrt{\{[1+(f/f_h)^2] / [1+(f/f_L)^2]\}}$
- d) $[A_{Ft} / (f/f_L)] / \sqrt{\{[1+(f/f_h)^2] / [1+(f/f_L)^2]\}}$

Answer: a





10. The advantage of narrow band-pass filter is

- a) f_c can be changed without changing gain
- b) f_c can be changed without changing bandwidth
- c) f_c can be changed without changing resistors
- d) All of the mentioned

Answer: d

11. The type of filter which passes all frequencies within a band between a lower and an upper critical frequency and rejects all others outside this band

- (A) low-pass filter
- (B) high-pass filter
- (C) band-pass filter
- (D) band-stop filter

Answer: band-pass filter

12. For any given signal, average power in its 6 harmonic components as 10 mW each and fundamental component also has 10 mV power. Then, average power in the periodic signal is _____

- a) 70
- b) 60
- c) 10
- d) 5

Answer: b

Explanation: We know that according to Parseval's relation, average power is equal to the sum of the average powers in all of its harmonic components.

$$\therefore P_{\text{avg}} = 10 \times 6 = 60.$$



13. The property of Fourier Transform which states that the compression in time domain is equivalent to the expansion in the frequency domain is

- a) Duality
- b) Scaling
- c) Time scaling
- d) Frequency shifting

Answer: c

14. Notch is a

- a) High pass filter
- b) Low pass filter
- c) Band stop filter
- d) Band pass filter

Answer: c

15. Sin wave is

- a) Aperiodic Signal
- b) Periodic Signal
- c) Random Signal
- d) Deterministic Signal

Answer: b

16. What is the role of channel in communication system?

- a) acts as a medium to send message signals from transmitter to receiver
- b) converts one form of signal to other
- c) allows mixing of signals
- d) helps to extract original signal from incoming signal

Answer: a



17. Distance between direct broadcasting satellite and earth's surface is _____

- a) 4500km
- b) 33000km
- c) 5200km
- d) 36000km

Answer: d

18. UHF stands for _____

- a) Ultra High Electric Field
- b) Ultraviolet High Frequency
- c) Ultra High Frequency
- d) Ultra High Magnetic Field

Answer: c

19. Effective noise at high frequencies is _____

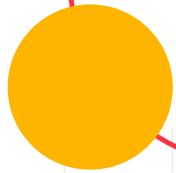
- a) Johnson noise
- b) Flicker noise
- c) transit-time noise
- d) Partition noise

Answer: c

20. What do you understand by isotropic antenna?

- a) it radiates its power uniformly in all directions
- b) it radiates its power non-uniformly in all directions
- c) it radiates its power specifically in one direction only
- d) it does not radiate any power

Answer: a



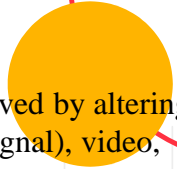


02

Analog signal transmission and reception



Introduction to modulation



- Modulation is the process of encoding information from a message source in a way that is suitable for transmission. This is achieved by altering the characteristics of a wave. By superimposing a message on to a high frequency signal known as a carrier wave (or sinusoidal signal), video, voice and other data can be transmitted.
- In the modulation process, a parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal. This variation acts as a code for data transmission.
- This modulated signal is then transmitted by the transmitter.
- The receiver demodulates the received modulated signal and gets the original information signal back.

Different types of modulation

There are two main types of modulation that are used in telecommunications; analog and digital.

Analog modulation

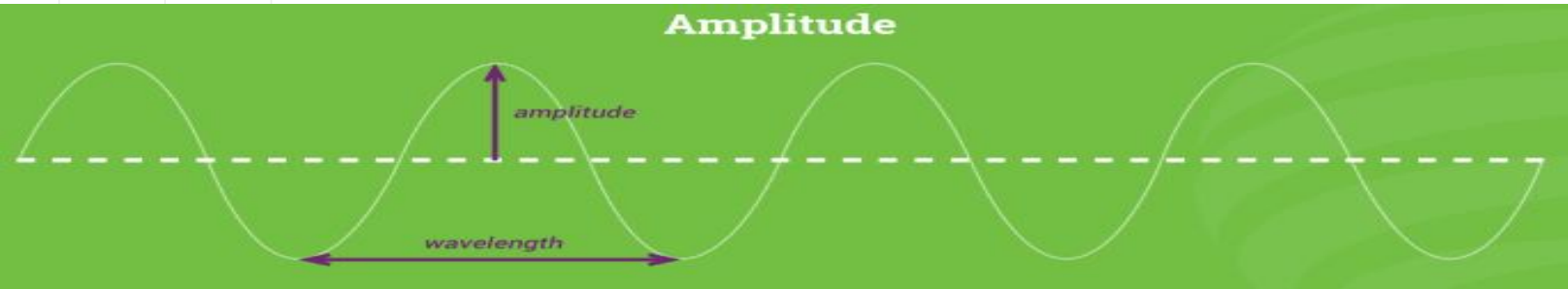
In analog modulation, a continuously varying sine wave is used as a carrier wave that modulates the message signal or data signal.

How analog modulation works

A carrier wave has three defining properties, which are amplitude, frequency and phase. These three defining properties are used to create three types of modulation:

Amplitude:

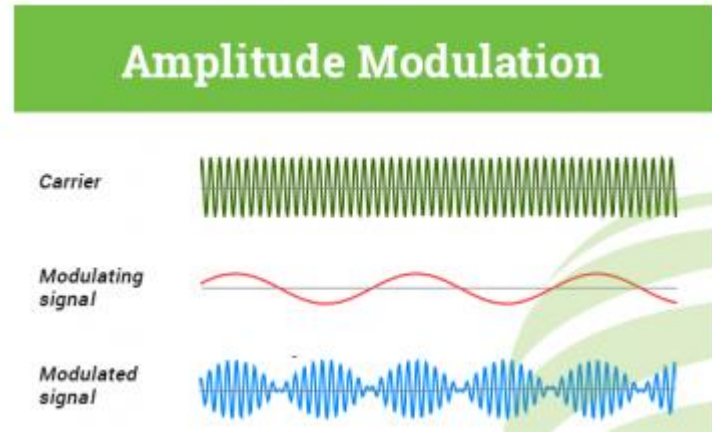
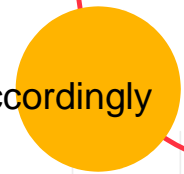
A high energy wave is characterized by a high amplitude; a low energy wave is characterized by a low amplitude.



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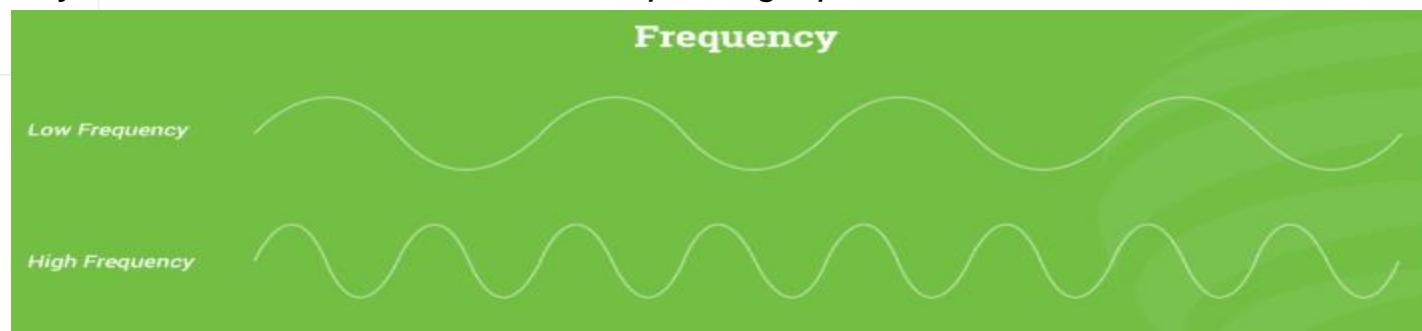
Amplitude modulation

Amplitude modulation or AM is the method of varying the instantaneous amplitude of carrier signal accordingly with instantaneous amplitude of message signal.



Frequency:

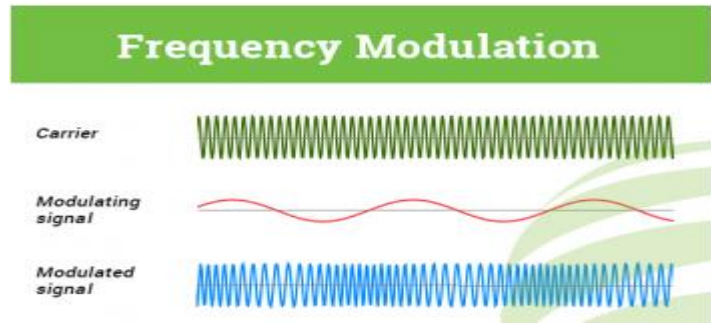
The frequency of a wave is the number of waves passing a point in a certain time.



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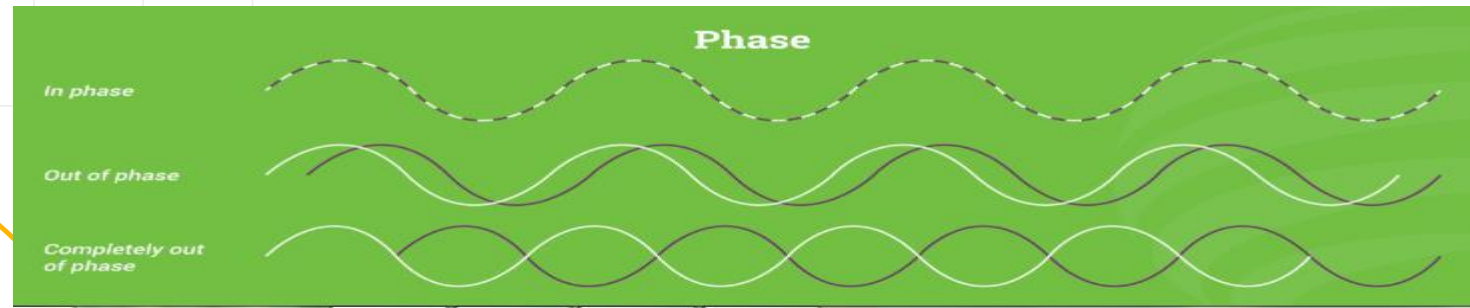
Frequency modulation

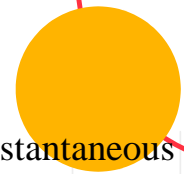
FM or Frequency modulation is the process of varying the instantaneous frequency of Carrier signal accordingly with instantaneous amplitude of message signal.



Phase:

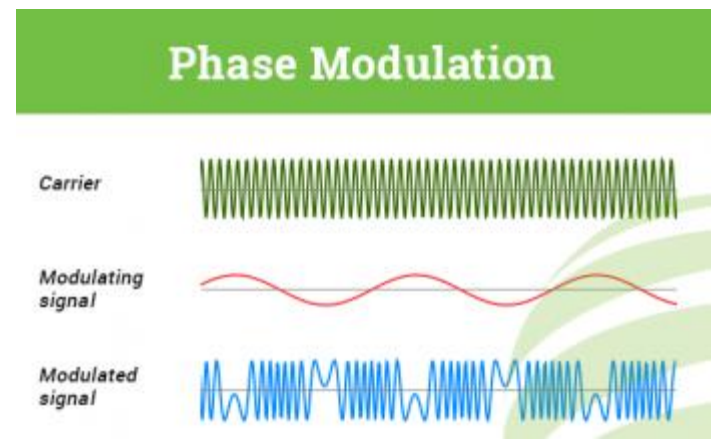
Phase is the position of a point in time (an instant) on a waveform cycle.





Phase modulation

PM or Phase modulation is the process of varying the instantaneous phase of Carrier signal accordingly with instantaneous amplitude of message signal.



Digital Modulation

Digital modulation is similar analog except base band signal is of discrete amplitude level. For binary signal it has only two levels, either high or logic 1 or low or logic 0.

The modulation scheme can be broken down in to three main types.

ASK or Amplitude Shift Key

FSK or Frequency Shift Key

PSK or Phase Shift Key



Why is modulation used in telecommunication?

There are many key problems in telecommunications that modulation helps to solve, including:

Problem: Certain types of data such as voice, video, and bit streams from computers have lower frequency bands and therefore they cannot travel very far wirelessly because they cannot propagate through space.

Solution: When using modulation, the carrier wave (which has a much higher frequency) is used to carry the signal, meaning it can travel further and more quickly through space.

Problem: Antenna length would be impractical for voice, video and bit streams without modulation.

Solution: By using the high frequency carrier waves, the necessary antenna size is greatly reduced.

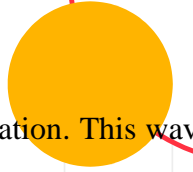
Problem: Without modulation, transmission of video, voice or data bit streams being sent on the same frequency range would cause overlapping of channels (resulting in interference).

Solution: Carrier radio waves have a wide range of frequencies which allow you to select individual non-overlapping channels to carry the information.



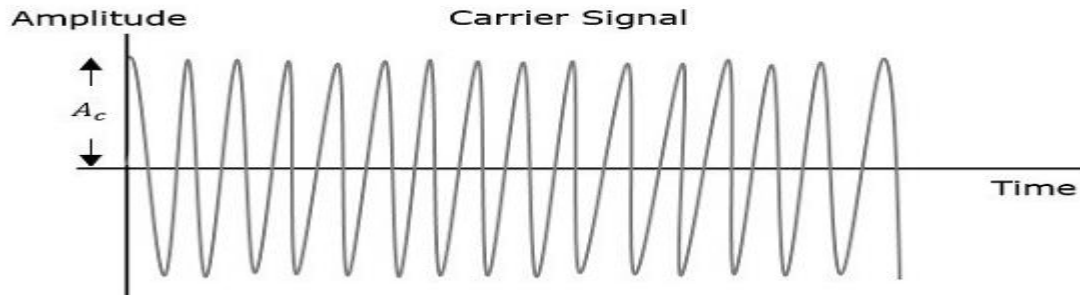
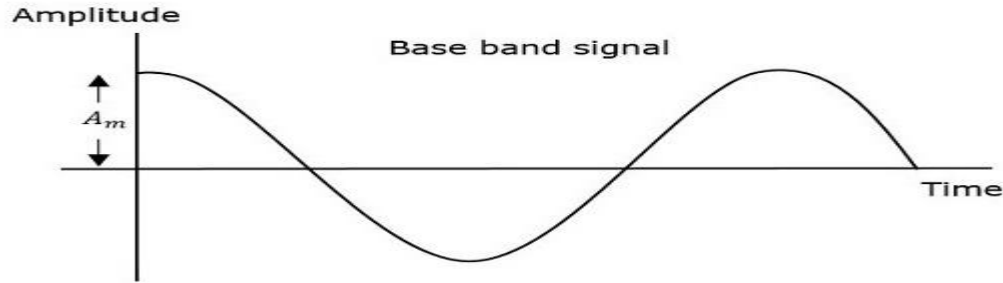


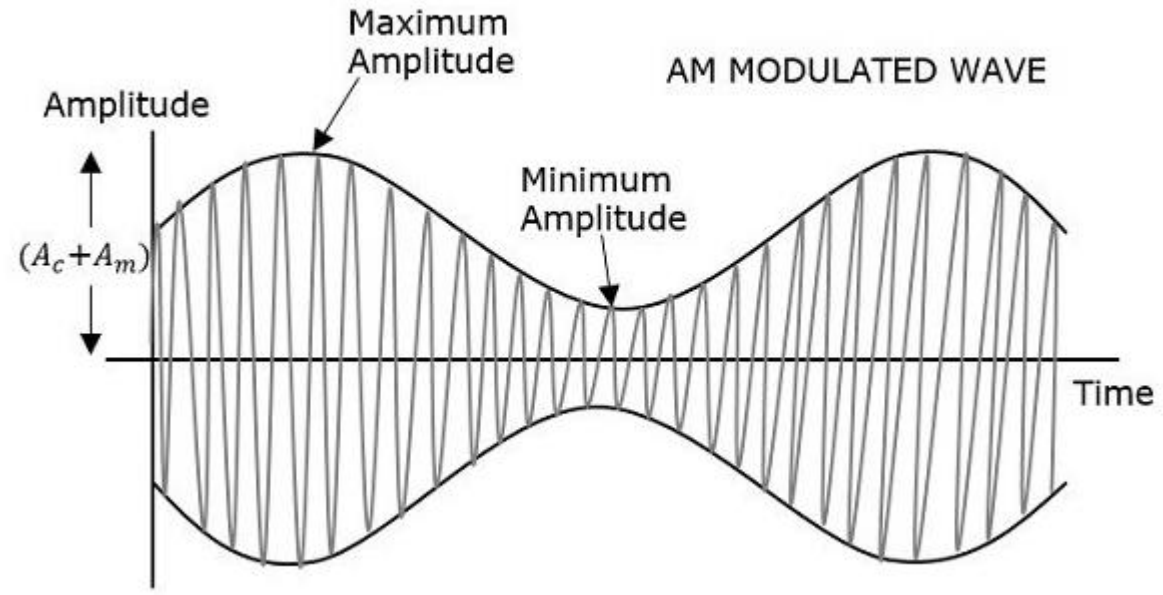
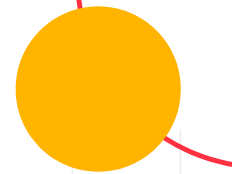
Amplitude Modulation (AM)



A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.

According to the standard definition, “The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.” Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures.



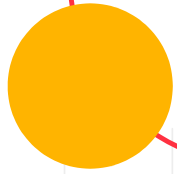


The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as **Envelope**. It is the same as that of the message signal.



Angle Modulation



The modulation in which, the angle of the carrier wave is varied according to the baseband signal.

- An important feature of this modulation is that it can provide better discrimination against noise and interference than amplitude modulation.
- An important feature of Angle mod. is that it can provide better discrimination against noise and distortion
- Two types: • Phase modulation • Frequency Modulation

Frequency Modulation In FM, the message signal $m(t)$ controls the frequency f_c of the carrier.

Consider the carrier $v_c(t) = V_c \cos(\omega_c t)$ then for FM

we may write: FM signal $v_s(t) = V_c \cos(2\pi(f_c + \text{frequency deviation})t)$, where the frequency deviation will depend on $m(t)$.

Given that the carrier frequency will change we may write for an instantaneous carrier signal

$$V_c \cos(\omega_i t) = V_c \cos(2\pi f_i t) = V_c \cos(\phi_i)$$

where ϕ_i is the instantaneous angle $= \omega_i t = 2\pi f_i t$ and f_i is the instantaneous frequency.

Frequency Modulation Since $\phi_i = 2\pi f_i t$ then $d\phi_i = 2\pi f_i dt$ or $\frac{1}{2\pi} \frac{d\phi_i}{dt} = f_i$

i.e. frequency is proportional to the rate of change of angle.

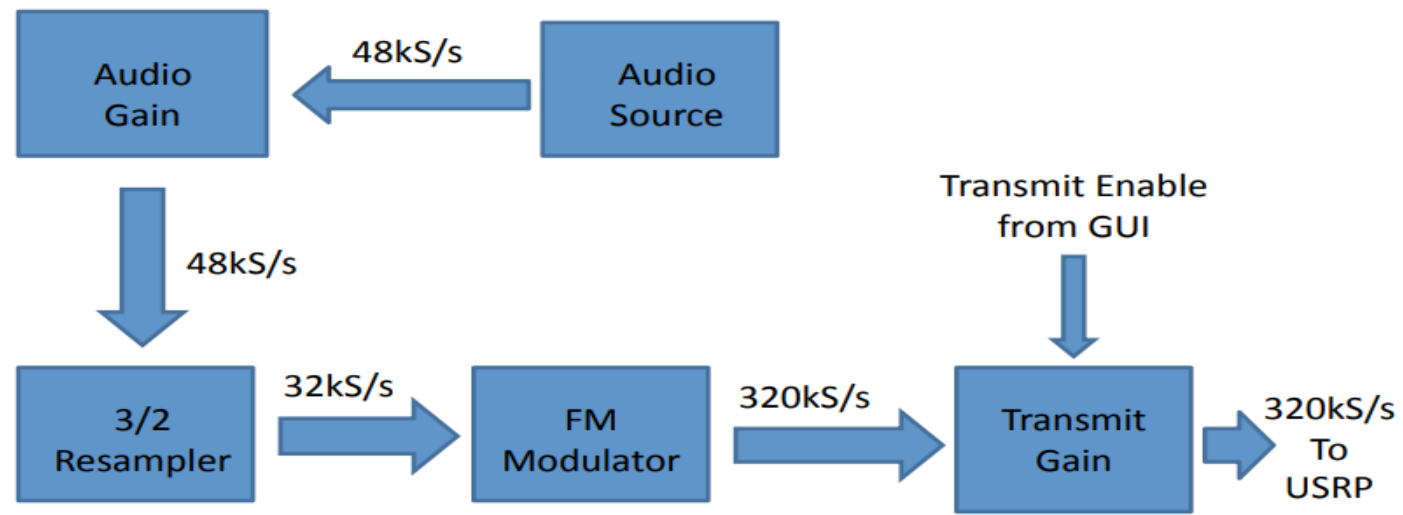
If f_c is the unmodulated carrier and f_m is the modulating frequency, then we may deduce that $f_i = f_c + \Delta f_c \cos(\omega_m t)$,
i.e. $\frac{1}{2\pi} \frac{d\phi_i}{dt} = f_c + \Delta f_c \cos(\omega_m t)$,

i.e. $d\phi_i = 2\pi f_c dt + 2\pi \Delta f_c \cos(\omega_m t) dt$ Hence, we have $2\pi \frac{d\phi_i}{dt} dt$



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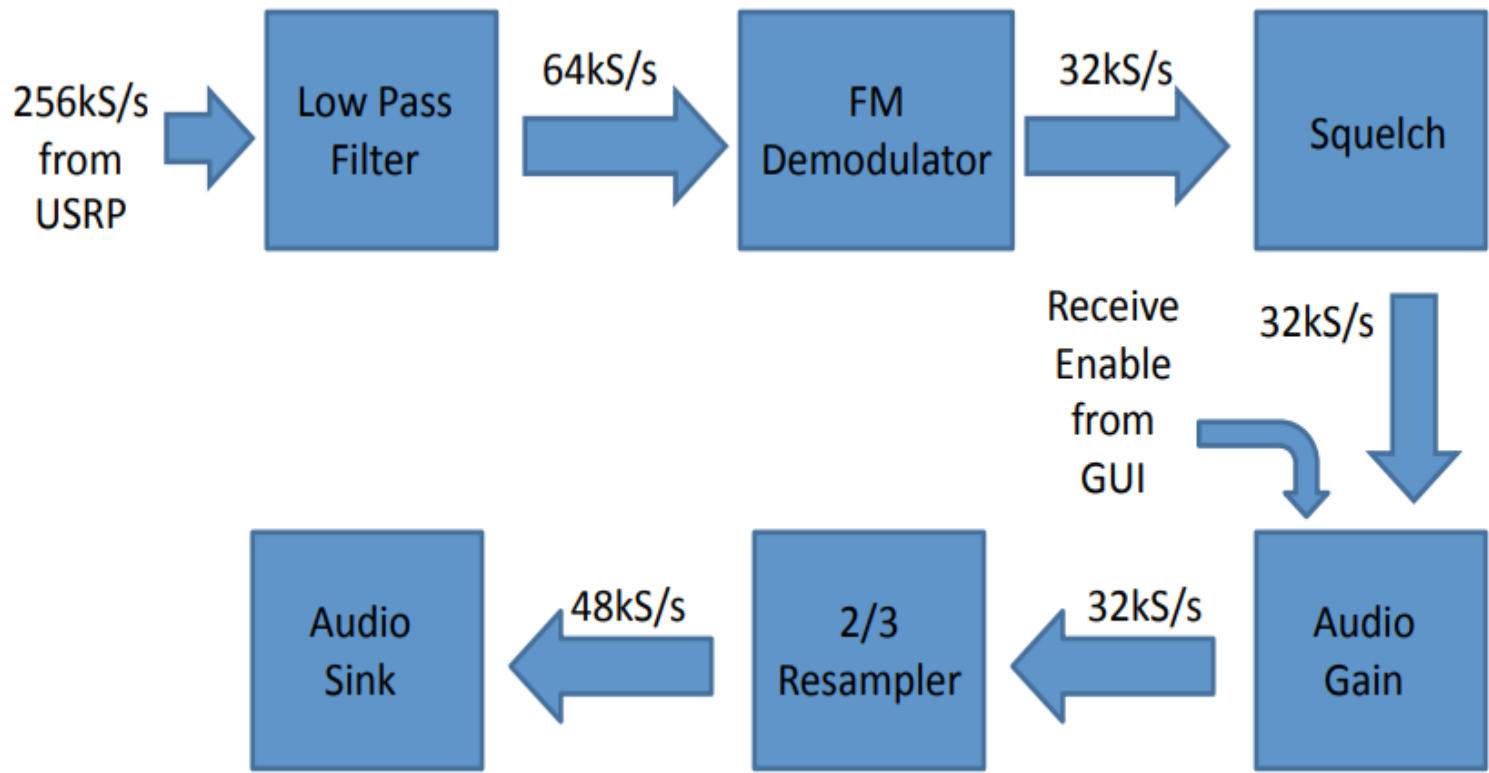
- Frequency Modulation Δf The ratio is called the Modulation Index denoted by β i.e. $\beta = \frac{\text{Peak frequency deviation}}{\text{modulating frequency}}$
- Note – FM, as implicit in the above equation for $v_s(t)$, is a non-linear process – i.e. the principle of superposition does not apply.
- The FM signal for a message $m(t)$ as a band of signals is very complex. Hence, $m(t)$ is usually considered as a 'single tone modulating signal' of the form $m(t) = V_m \cos(\omega_m t)$



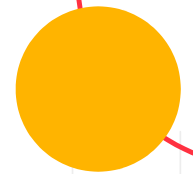
TRANSMIT PATH-ALWAYS RUNNING



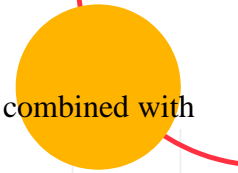
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RECEIVING PATH-ALWAYS RUNNING



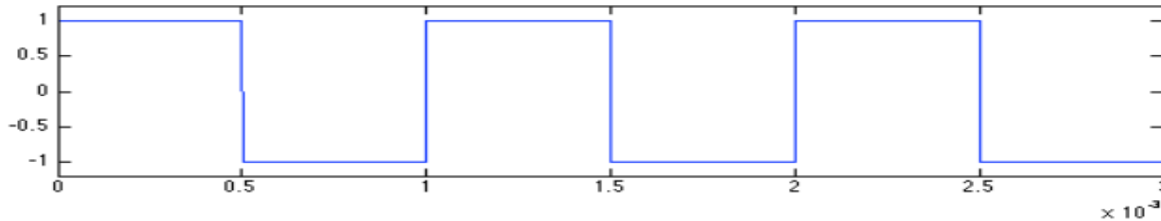
Phase Modulation



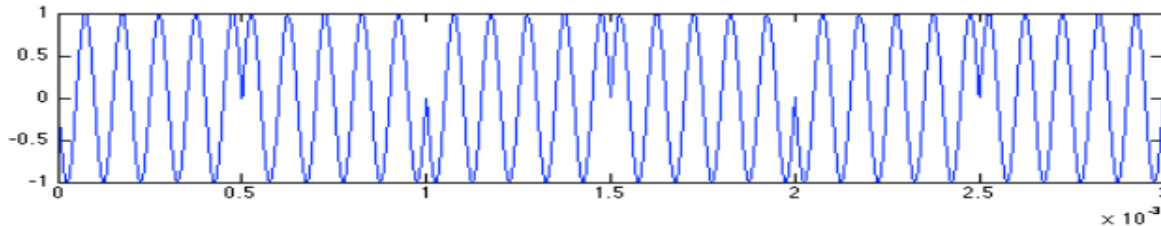
- A method of transmitting signals in which the value of the signal is proportional to the phase angle of a carrier wave. It is often combined with amplitude modulation to transmit digital information via modems.
- With **phase modulation**, the slope of the baseband signal governs how quickly the **phase** changes, and the rate at which the **phase** changes is equivalent to frequency. So in a PM waveform, high baseband slope corresponds to high frequency, and low baseband slope corresponds to low frequency.

For pulse train base band signal

Baseband signal, $x(t)$



PM signal



Advantages, disadvantages and applications of AM

DSB-SC: It has lower power consumption and it is simple technique of modulation. But it is complex in detection at AM receiver. It is used in analog TV transmission systems to transmit color information.

- SSB-SC: It is used for efficient management of spectrum. But generation of SSB modulation is difficult and it is complex in detection at receiver. It is used for 2-way radio FDM.
- VSB-SC: It is compromise between DSB and SSB types. But demodulation system is complex. Bandwidth of VSB-SC is 25% higher than SSB-SC. It is used for analog TV broadcast systems.

Advantages, disadvantages and applications of FM

- Advantages: Increased immunity to noise.
- Disadvantage: Requires larger bandwidth.
- Application: Radio Broadcasting, Direct Satellite Broadcasting

Advantages, disadvantages and applications of PM

- Advantages: Increased immunity to noise.
- Disadvantages: More complex hardware at receiver.
- Applications: Used in data communication systems.





Feature	AM	FM	PM
Function	amplitude of carrier wave varies as per amplitude or voltage of modulating signal input.	Frequency of carrier wave varies as per voltage of modulating signal input.	Phase of carrier wave varies as per voltage of modulating signal input.
Carrier parameter	frequency of carrier wave is kept constant	amplitude of carrier wave is kept constant	amplitude of carrier wave is kept constant
Types	AM types include DSB-SC, SSB, VSB etc. Refer DSB-SC vs SSB-Scion	Digital FM types include FSK, GFSK, Offset FSK etc. Refer MSK and GMSK modulation	Digital PM types include BPSK, QPSK, QAM(combination of amplitude and phase modulation types) Refer BPSK and QPSK,QAM modulation types.





Radio and Television broadcasting

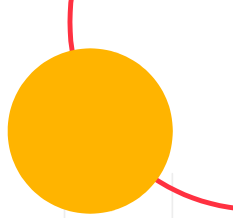


- Television broadcasting: form of broadcasting in which a television signal is transmitted by radio waves from a terrestrial (Earth based) transmitter of a television station to TV receivers having an antenna.
- Transmission of radio and television programs from a radio or television station to home receivers by radio waves is referred to as "over the air" (OTA) or terrestrial broadcasting and in most countries requires a broadcasting license.
- In radio communication, used in radio and television broadcasting, cell phones, two-way radios, wireless networking and satellite communication among numerous other uses, radio waves are used to carry information across space from a transmitter to a receiver, by modulating the radio signal.
- program is transmitted by adding it to a radio wave called a carrier. This process is called modulation.
- Frequency modulation is used for FM radio broadcast; amplitude modulation is the technique used by AM radio stations.





MCQ



1) Amplitude modulation is

- a. Change in amplitude of the carrier according to modulating signal
- b. Change in frequency of the carrier according to modulating signal
- c. Change in amplitude of the modulating signal according to carrier signal
- d. Change in amplitude of the carrier according to modulating signal frequency

ANSWER: (a) Change in amplitude of the carrier according to modulating signal

2) The ability of the receiver to select the wanted signals among the various incoming signals is termed as

- a. Sensitivity
- b. Selectivity
- c. Stability
- d. None of the above

ANSWER: (b) Selectivity

3) Emitter modulator amplifier for Amplitude Modulation

- a. Operates in class A mode
- b. Has a low efficiency
- c. Output power is small
- d. All of the above

ANSWER: (d) All of the above

4) Super heterodyne receivers

- a. Have better sensitivity
- b. Have high selectivity
- c. Need extra circuitry for frequency conversion
- d. All of the above

ANSWER: (d) All of the above



5) **The AM spectrum consists of**

- a. Carrier frequency
- b. Upper side band frequency
- c. Lower side band frequency
- d. All of the above

ANSWER: (d) All of the above

6) **Standard intermediate frequency used for AM receiver is**

- a. 455 MHz
- b. 455 KHz
- c. 455 Hz
- d. None of the above

ANSWER: (b) 455 KHz

7) **In the TV receivers, the device used for tuning the receiver to the incoming signal is**

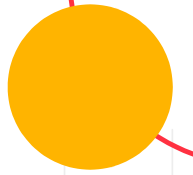
- a. Varactor diode
- b. High pass Filter
- c. Zener diode
- d. Low pass filter

ANSWER: (a) Varactor diode

8) **The modulation technique that uses the minimum channel bandwidth and transmitted power is**

- a. FM
- b. DSB-SC
- c. VSB
- d. SSB

ANSWER: (d) SSB



9) Calculate the bandwidth occupied by a DSB signal when the modulating frequency lies in the range from 100 Hz to 10KHz.

- a. 28 KHz
- b. 24.5 KHz
- c. 38.6 KHz
- d. 19.8 KHz

ANSWER: (d) 19.8 KHz

10) In Amplitude Demodulation, the condition which the load resistor R must satisfy to discharge capacitor C slowly between the positive peaks of the carrier wave so that the capacitor voltage will not discharge at the maximum rate of change of the modulating wave (W is message bandwidth and ω is carrier frequency, in rad/sec) is

- a. $RC < 1/W$
- b. $RC > 1/W$
- c. $RC < 1/\omega$
- d. $RC > 1/\omega$

ANSWER: (a) $RC < 1/W$

11) A modulation index of 0.5 would be same as

- a. 0.5 of Modulation Depth
- b. 1/2% of Modulation Depth
- c. 5% of Modulation Depth
- d. 50% of Modulation Depth

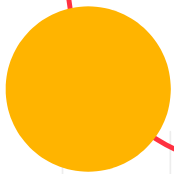
ANSWER: (d) 50% of Modulation Depth

12) A 3 GHz carrier is DSB SC modulated by a signal with maximum frequency of 2 MHz. The minimum sampling frequency required for the signal so that the signal is ideally sampled is

- a. 4 MHz
- b. 6 MHz
- c. 6.004 GHz
- d. 6 GHz

ANSWER: (c) 6.004 GHz





13) The function of multiplexing is

- a. To reduce the bandwidth of the signal to be transmitted
- b. To combine multiple data streams over a single data channel
- c. To allow multiple data streams over multiple channels in a prescribed format
- d. To match the frequencies of the signal at the transmitter as well as the receiver

ANSWER: (b) To combine multiple data streams over a single data channel

14) Aliasing refers to

- a. Sampling of signals less than at Nyquist rate
- b. Sampling of signals greater than at Nyquist rate
- c. Sampling of signals at Nyquist rate
- d. None of the above

ANSWER: (a) Sampling of signals less than at Nyquist rate

15) The amount of data transmitted for a given amount of time is called

- a. Bandwidth
- b. Frequency
- c. Noise
- d. Signal power

ANSWER: (a) Bandwidth

16) An AM broadcast station transmits modulating frequencies up to 6 kHz. If the AM station is transmitting on a frequency of 894 kHz, the values for maximum and minimum upper and lower sidebands and the total bandwidth occupied by the AM station are:

- a. 900 KHz, 888 KHz, 12 KHz
- b. 894 KHz, 884 KHz, 12 KHz
- c. 894 KHz, 888 KHz, 6 KHz
- d. 900 KHz, 888 KHz, 6 KHz

ANSWER: (a) 900 KHz, 888 KHz, 12 KHz

Explanation:

Maximum Frequency $f_{USB} = 894 + 6 = 900$ kHz

Minimum Frequency $f_{LSB} = 894 - 6 = 888$ kHz .Bandwidth $BW = f_{USB} - f_{LSB} = 900 - 888 = 12$ kHz OR $= 2(6 \text{ kHz}) = 12$ kHz



17) The total power in an Amplitude Modulated signal if the carrier of an AM transmitter is 800 W and it is modulated 50 percent.

- a. 850 W
- b. 1000.8 KW
- c. 750 W
- d. 900 W

ANSWER: (d) 900 W

Explanation:

The total power in an Amplitude Modulated wave is given by

$$P_T = P_C (1 + m^2/2)$$

Here, $P_C = 800W$,

$$m = 0.5$$

therefore,

$$P_T = 800 (1 + (0.5)^2/2) = 900 W$$



18) • An unmodulated AM signal produces a current of 5.4 A. If the modulation is 100 percent, calculate: • • •

(a) the carrier power,

(b) the total power,

(c) the sideband power when it is transmitted through an antenna having an impedance of 50Ω .

a. 1458 W, 2187.5 W, 729.25 W

b. 278 W, 2187.5 W, 1917.25 W

c. 1438 W, 2187.5 W, 759.25 W

d. 280 W, 2187.5 W, 750.25 W

ANSWER: (a) 1458 W, 2187.5 W, 729.25 W

Explanation:

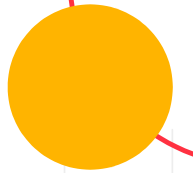
$$\text{a) } P_C = I^2 R = (5.4)^2 * 50 = 1458 \text{ W}$$

$$\text{b) } I_T = I_c \sqrt{1+m^2/2} = 5.4 \sqrt{1+1^2/2} \\ = 6.614 \text{ A}$$

$$P_T = I_T^2 R \\ = (6.614)^2 * 50 \\ = 2187.25 \text{ W}$$

$$\text{c) } P_{SB} = P_T - P_C \\ = 2187.25 - 1458 \text{ W} \\ = 729.25 \text{ W (for two bands)}$$

$$\text{For single band, } P_{SB} = 729.25/2 \\ = 364.625 \text{ W}$$



19) Calculate the depth of modulation when a transmitter radiates a signal of 9.8KW after modulation and 8KW without modulation of the signal.

- a. 80%
- b. 67%
- c. 50%
- d. 100%

ANSWER: (b) 67%

Explanation:

$$P_{\text{total}} = 9.8\text{KW}$$

$$P_c = 8\text{KW}$$

Power of the signal (P_{total}) transmitted by a transmitter after modulation is given by

$$P_{\text{total}} = P_c (1 + m^2/2)$$

Where P_c is the power of carrier i.e., without modulation

M is the modulation index

Therefore,

$$9.8 = 8 (1 + m^2/2), \quad 9.8/8 = 1 + m^2/2, \quad m = 0.67 = 67\%$$



20) When AM signal is of 25KHz, calculate the number of channels required in Medium Frequency (MF) band of 300KHz-3000KHz.

- a. 94
- b. 69
- c. 85
- d. 54

ANSWER: (d) 54

Explanation:

Medium Frequency (MF) is the band of frequencies from 300 KHz to 3MHz. The lower portion of the MF band (300to 500 kilohertz) is used for ground-wave transmission for reasonably long distances. The upper and lower ends of the mf band are used for naval purpose.

Frequency available in MF band= $3000 - 300 = 2700$ KHz

The bandwidth required by 25 KHz signal = $2 * 25 = 50$ KHz

Therefore the number of channels available = $2700 / 50 = 54$





03

Pulse modulation systems





Pulse amplitude modulation



An analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time.

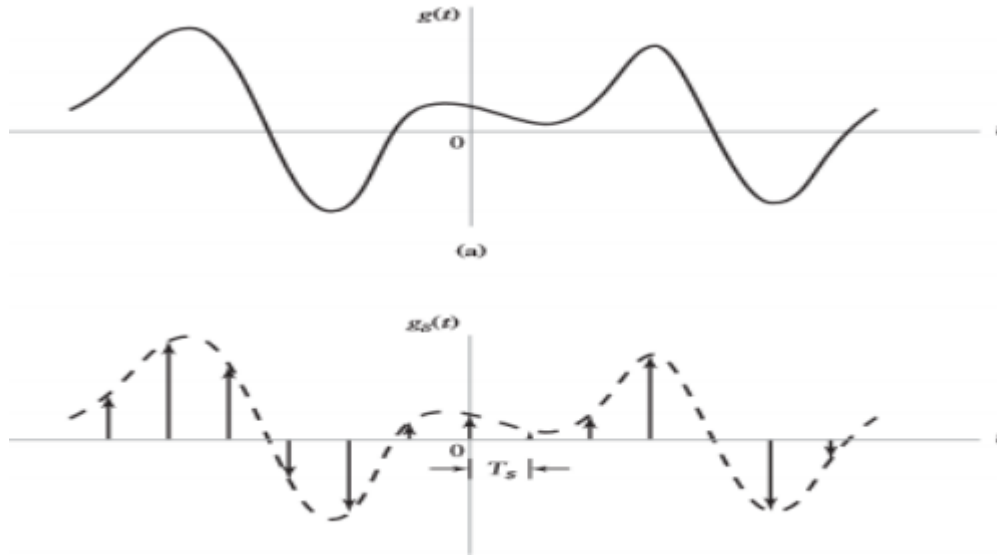


Figure: Illustration of sampling process. (a) Analog waveform (b) Instantaneously sampled representation of the analog Signal



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$$g_{\delta}(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t - nT_s) \quad (1)$$

$$G_{\delta}(f) = f_s \sum_{m=-\infty}^{\infty} G(f - mf_s) \quad (2)$$

Where

$G(f)$ is the Fourier Transform of the original signal $g(t)$, and f_s is the sampling rate.

• The above equation states that the process of uniformly sampling a continuous time signal of finite energy results in a periodic spectrum with a period equal to sampling rate.

$$G_{\delta}(f) = \sum_{n=-\infty}^{\infty} g(nT_s)\exp(-j2\pi n f T_s) \quad (3)$$

Suppose, that the signal $g(t)$ is strictly band-limited, with no frequency components higher than W Hertz. That is, the Fourier transform of $g(t)$ has the property that $G(f)$ is zero for $f > W$, as illustrated in below figure. Also we choose, $T_s = 1/2W$, then the corresponding spectrum shown in figure b.



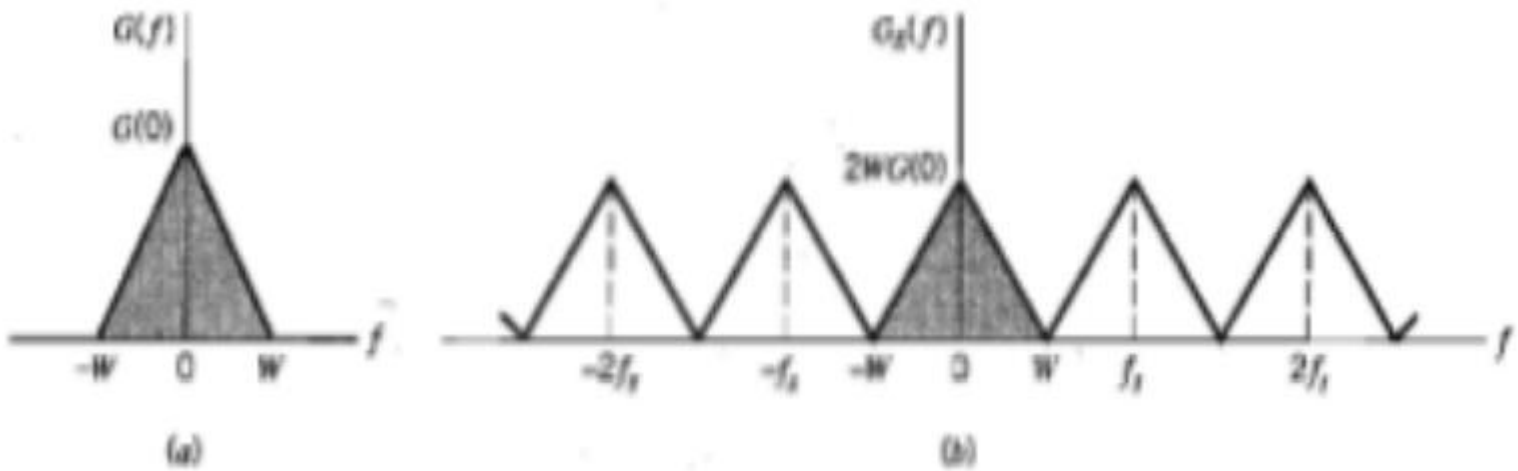
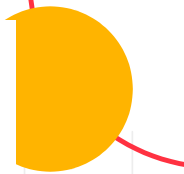


Figure: (a) Spectrum of a strictly band-limited signal $g(t)$
 (b) Spectrum of the sampled version of $g(t)$ for a sampling period $T_s=1/2W$

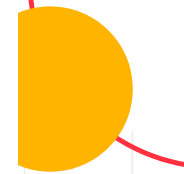
$$G_{\delta}(f) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \exp\left(-\frac{j\pi n f}{W}\right) \quad (4)$$

The Fourier Transform of $g_{\delta}(t)$ may also be expressed as



CONT...
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$$G_{\delta}(f) = f_s G(f) + f_s \sum_{\substack{m=-\infty \\ m \neq 0}}^{\infty} G(f - mf_s) \quad (5)$$



Hence, under the following two conditions,

1. $G(f) = 0$ for $|f| \geq W$
2. $f_s = 2W$

We find from equation (5) that

$$G(f) = \frac{1}{2W} G_{\delta}(f), \quad -W < f < W \quad (6)$$

substitute equation (4) into the above equation, we may also write

$$G(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \exp\left(-\frac{j\pi n f}{W}\right), \quad -W < f < W \quad (7)$$

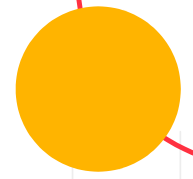
• Therefore, if the sampled values of $g(n/2W)$ of a signal $g(t)$ are specified for all n , then the Fourier transform $G(f)$ of the signal is uniquely determined using eq (7). Because the sequence $g(n/2W)$ has all the information contained in $g(t)$. • The expression for reconstructing the original signal $g(t)$ from the sequence of sample values $\{g(n/2W)\}$ is



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W_t

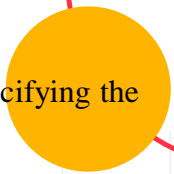
$$g(t) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W_t}\right) \text{sinc}(2W_t t - n), \quad -\infty < t < \infty \quad (8)$$



Sinc (2wt) playing the role of an interpolation function. Each sample is multiplied by a delayed version of interpolation function, and all the resulting waveforms are added to obtain g(t)



Sampling Theorem



• A band-limited signal of finite energy, which has no frequency components higher than W Hertz, is completely described by specifying the values of the signal at instants of time separated by $1/2W$ seconds.

OR

• A band-limited signal of finite energy, which has no frequency components higher than W Hertz, may be completely recovered from a knowledge of its samples taken at the rate of $2W$ samples per second. • The sampling rate of $2W$ samples per second, for a signal bandwidth of W Hertz, is called Nyquist rate; its reciprocal $1/2W$ is called Nyquist interval.

Aliasing Effect

• If , an information bearing signal is not strictly band-limited, some aliasing is produced by the sampling process. • Aliasing refers to the phenomenon of a high frequency component in the spectrum of the signal seemingly taking on the identity of a lower frequency in the spectrum

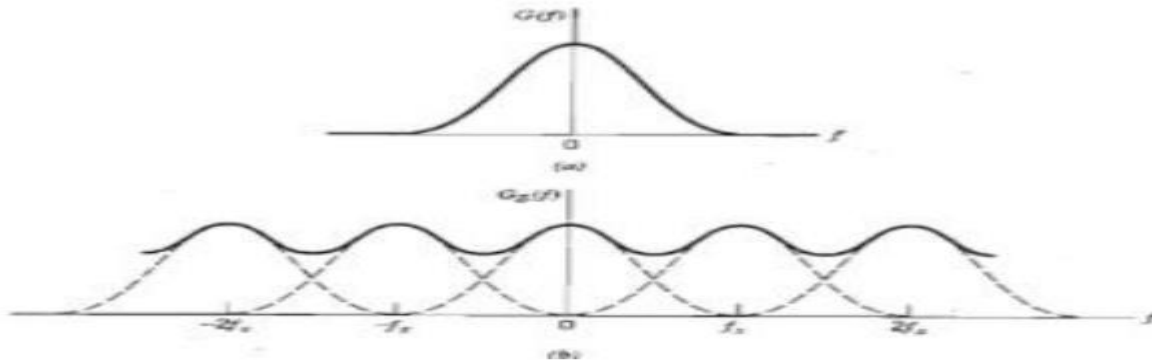
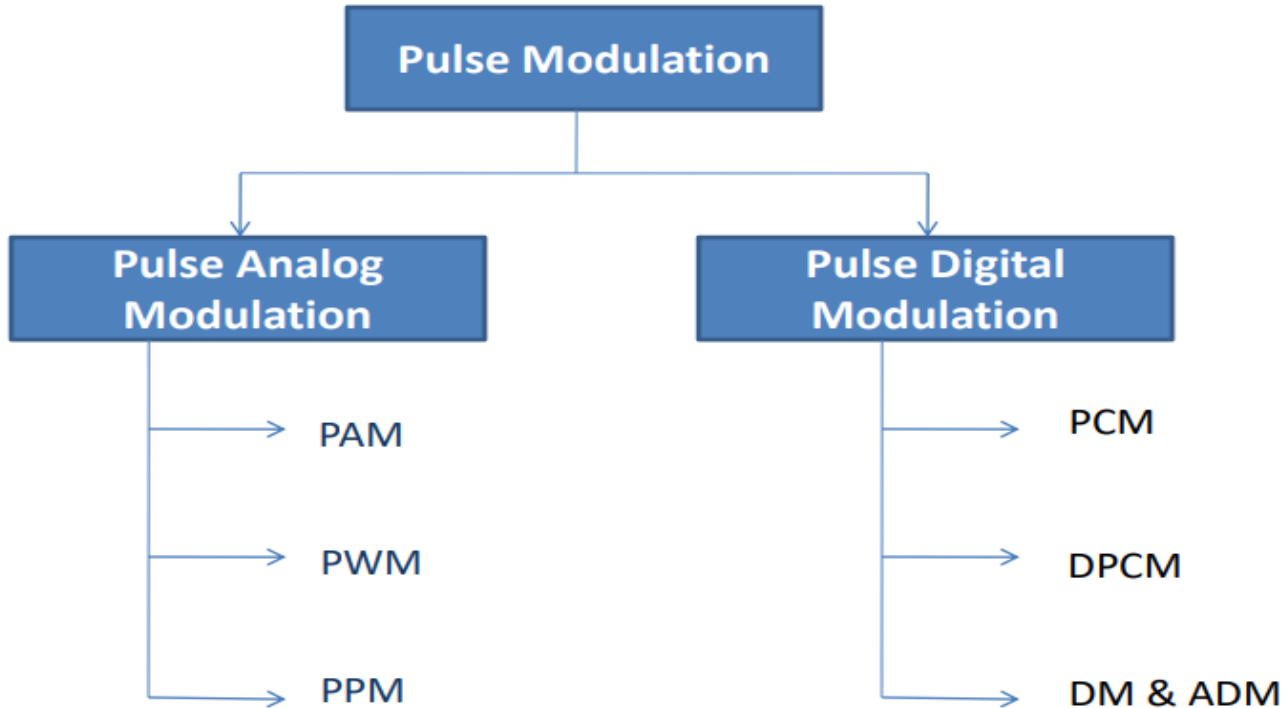


Figure:
(a) Spectrum of a signal
(b) Spectrum of an under-sampled version of the signal exhibiting the aliasing phenomenon

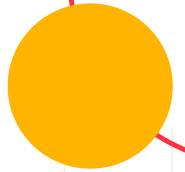


Types of Pulse Modulation





Pulse Amplitude Modulation



In fact the pulses in a PAM signal may be of Flat-top type or natural type or ideal type.

- The Flat-top PAM is most popular and is widely used. The reason for using Flat-top PAM is that during the transmission, the noise interferes with the top of the transmitted pulses and this noise can be easily removed if the PAM pulse is Flat-top.
- In natural samples PAM signal, the pulse has varying top in accordance with the signal variation. Such type of pulse is received at the receiver, it is always contaminated by noise. Then it becomes quite difficult to determine the shape of the top of the pulse and thus amplitude detection of the pulse is not exact.

Generation of PAM

There are two operations involved in the generation of PAM signal 1. Instantaneous sampling of the message signal $m(t)$ every T_s seconds, where the sampling rate $f_s = 1/T_s$ is chosen in accordance with the sampling theorem. 2. Lengthening the duration of each sample so obtained to some constant value T .

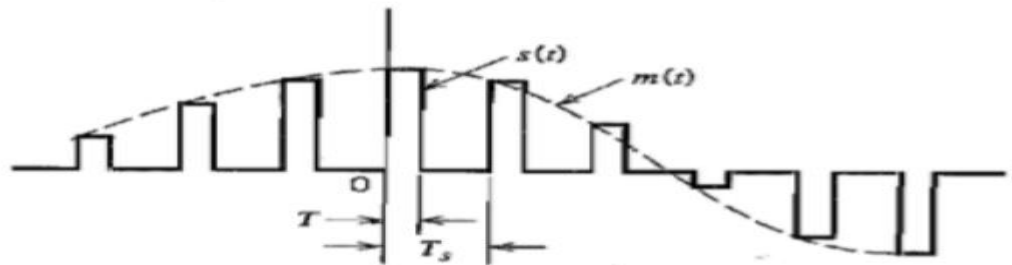


Figure: PAM signal



Sample and Hold Circuit for Generating Flat-top sampled PAM

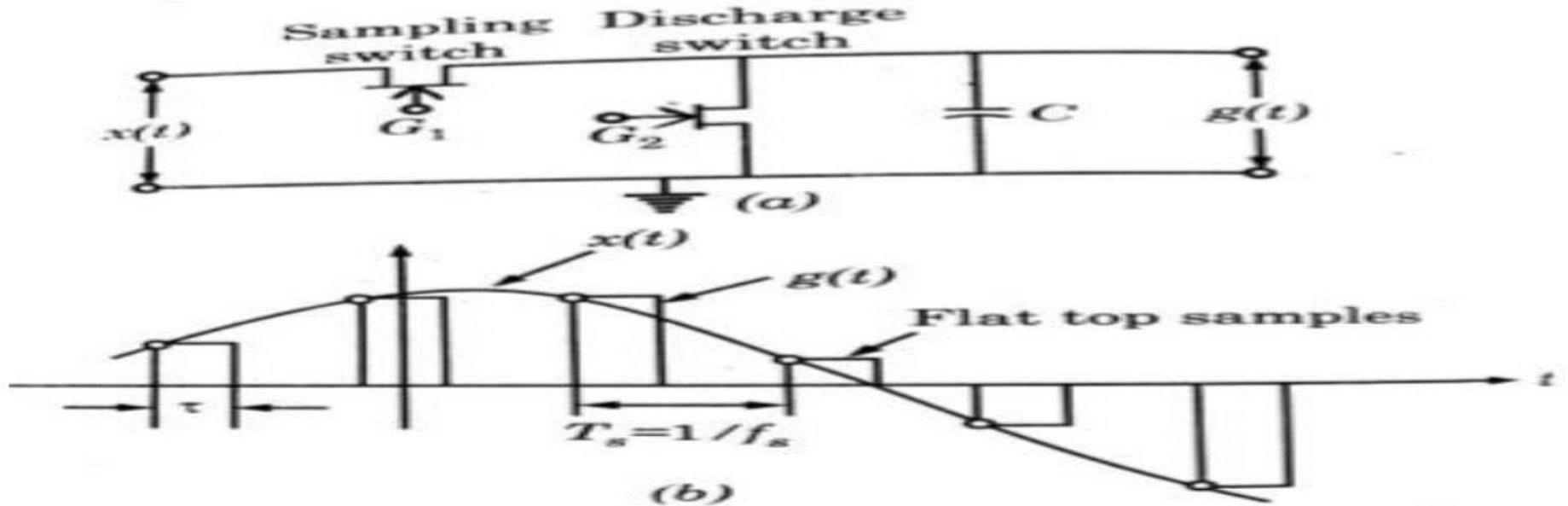
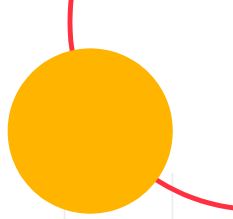


Figure:

- (a) Sample and hold circuit generating flat top sampled PAM
- (b) Waveforms of flat top sampled PAM





- The sample and hold circuit consists of two Field Effect Transistor switches and a capacitor.
 - The sampling switch is closed for a short duration by a short pulse applied to the gate G1 of the transistor. During this period, the capacitor C is quickly charged up to a voltage equal to the instantaneous sample value of the incoming signal.
 - Now, the sampling switch is opened and the capacitor holds the charge. The discharge switch is then closed by a pulse applied to gate G2 of the other transistor. Due to this, the capacitor is discharged to zero volts. The discharge switch is then opened and thus capacitor has no voltage. Hence the output of the sample and hold circuit consists of a sequence of flat-top samples as shown in figure.



Demodulation of PAM

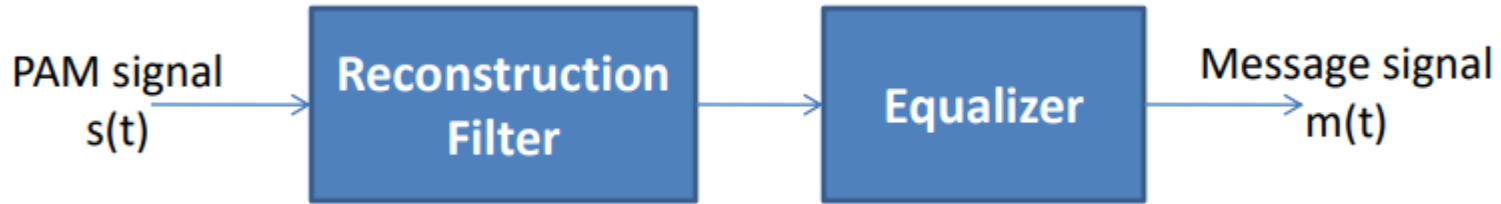
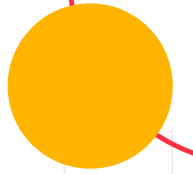


Figure: System for recovering message signal $m(t)$ from PAM signal $s(t)$

- The distortion caused by the use of PAM to transmit an analog information bearing signal is referred to as the aperture effect. This distortion may be corrected by connecting an equalizer in cascade with the low-pass reconstruction filter as shown in fig.
- • The equalizer has the effect of decreasing the in-band loss of the reconstruction filter as the frequency increases in such a manner as to compensate for the aperture effect



Advantages of PAM



- It is the simple and simple process for modulation and demodulation
- Transmitter and receiver circuits are simple and easy to construct.

Drawbacks of PAM signal

- The bandwidth required for the transmission of a PAM signal is very large in comparison to the maximum frequency present in the modulating signal.
- Since the amplitude of the PAM pulses varies in accordance with the modulating signal therefore the interference of noise is maximum in a PAM signal. This noise cannot be removed easily.
- Since the amplitude of the PAM pulses varies, therefore, this also varies the peak power required by the transmitter with modulating signal.





Pulse Time Modulation (PTM)



- In pulse time modulation, amplitude of pulse is held constant, whereas position of pulse or width of pulse is made proportional to the amplitude of signal at the sampling instant.
- There are two types of pulse time modulation. i. Pulse Width Modulation ii. Pulse Position Modulation

Pulse Width Modulation

- PWM is also called Pulse Duration Modulation (PDM), Pulse Length Modulation (PLM)
- In PWM, Width of the pulses of the carrier pulse train is varied in accordance with the modulating signal.

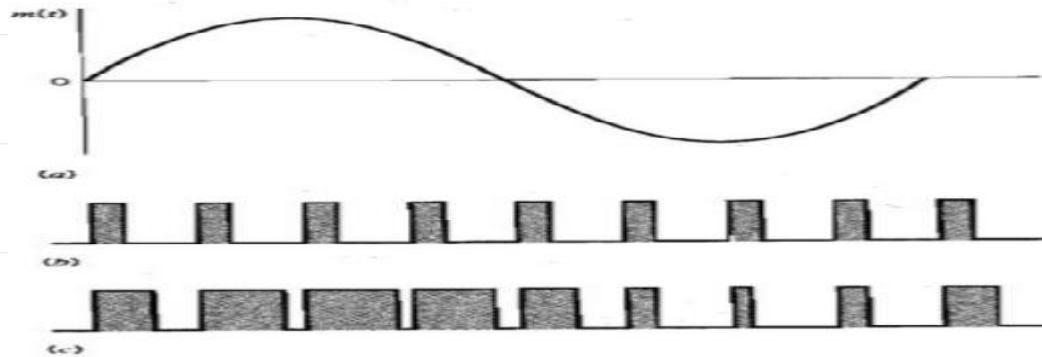
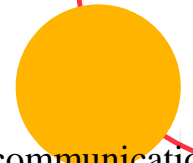


Figure: Illustration of PWM (a) Modulating signal (b) Pulse Carrier (c) PWM signal



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Advantages of PWM

: • Noise is less, since in PWM, amplitude is held constant. • Signal and noise separation is very easy • PWM communication does not required synchronization between transmitter and receiver.

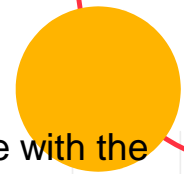
Disadvantages of PWM:

- In PWM, pulses are varying in width and therefore their power contents are variable this requires that the transmitter must be able to handle the power content of the pulse having maximum pulse width.
- Large bandwidth is required for the PWM as compared to PAM PWM Demodulation .

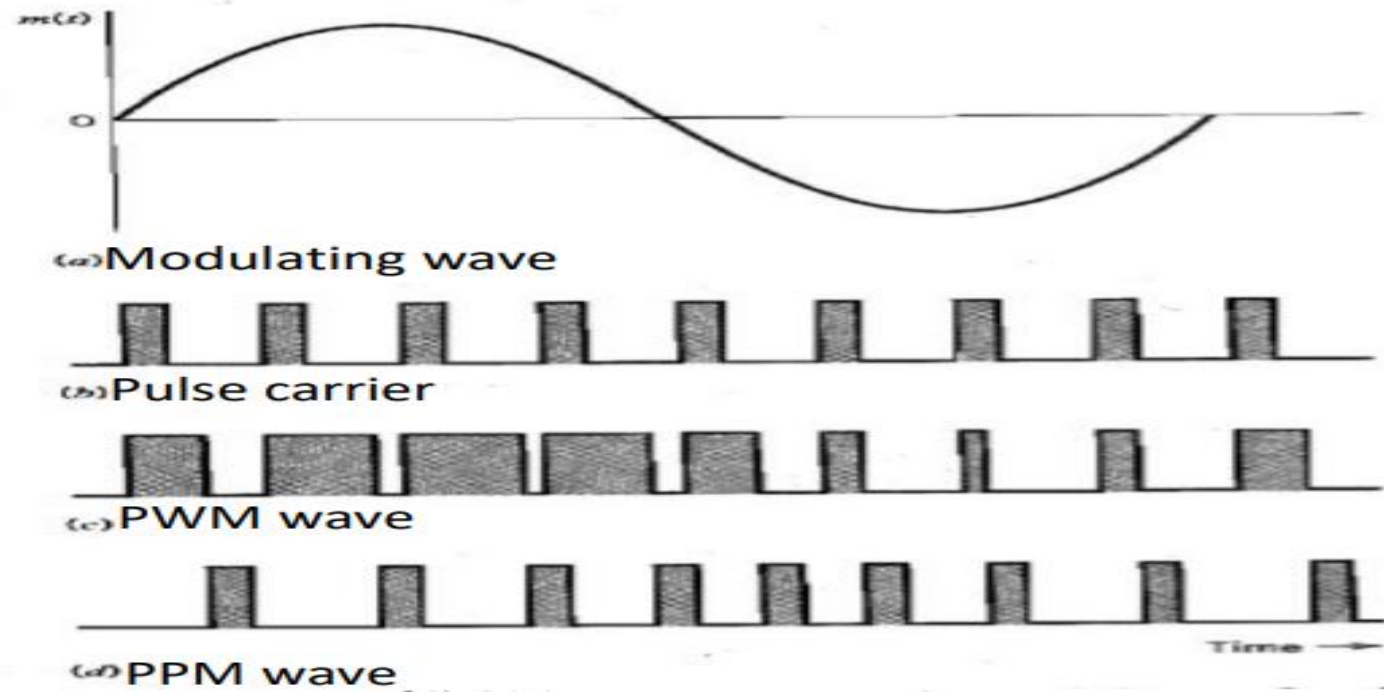


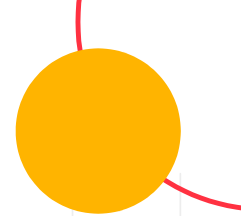
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Pulse Position Modulation (PPM)



In PPM, the position of the pulse relative to its un-modulated time occurrence is varied in accordance with the message signal.





Advantages of PPM:

- Like PWM, in PPM, amplitude is held constant thus less noise interference.
- Signal and noise separation is very easy
- Because of constant pulse widths and amplitudes, transmission power for each pulse is same

Disadvantages of PWM:

- Synchronization between transmitter and receiver is required.
- Large bandwidth is required for the PPM as compared to PAM



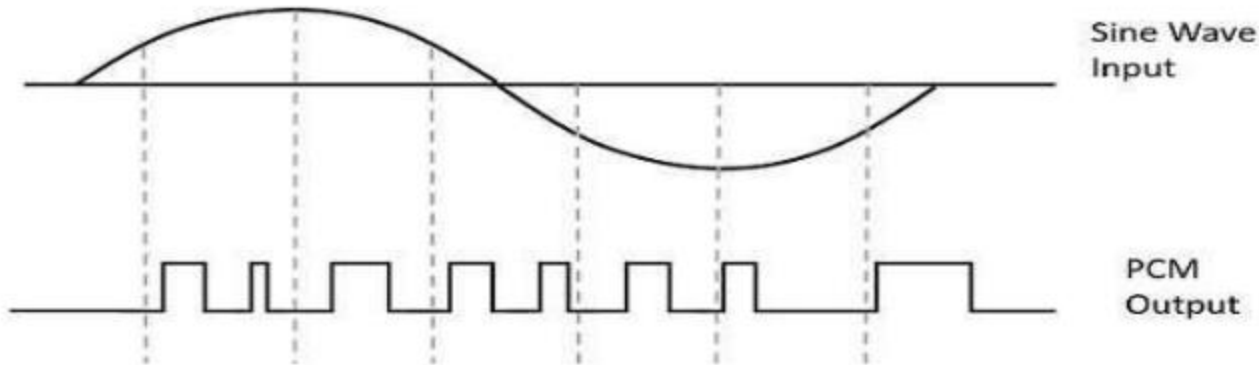


PCM system

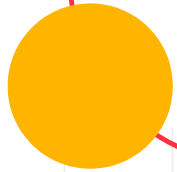


The message signal is the signal which is being transmitted for communication and the carrier signal is a high frequency signal which has no data, but is used for long distance transmission. There are many modulation techniques, which are classified according to the type of modulation employed. Of them all, the digital modulation technique used is Pulse Code Modulation (PCM). A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a PCM will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.

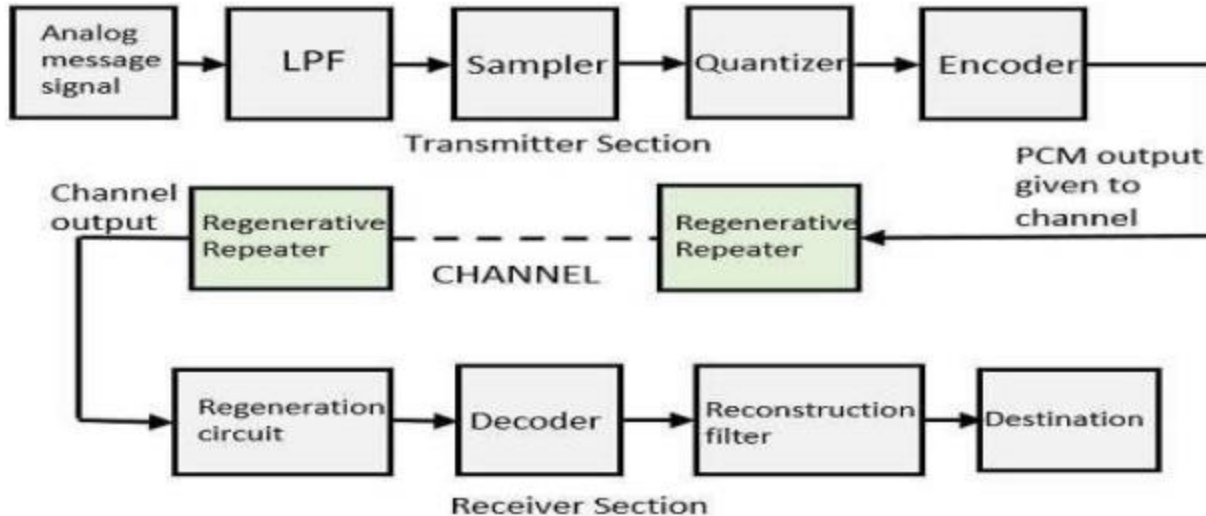
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant. In Pulse Code Modulation, the message signal is represented by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



Basic Elements of PCM



- The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section.
- The low pass filter prior to sampling prevents aliasing of the message signal.
- The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train.
- Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.





Intersymbol interference

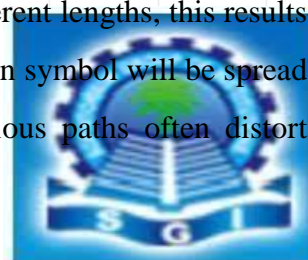


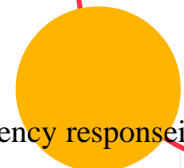
In telecommunication, intersymbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. The spreading of the pulse beyond its allotted time interval causes it to interfere with neighboring pulses. ISI is usually caused by multipath propagation or the inherent linear or non-linear frequency response of a communication channel causing successive symbols to blur together. The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible.

Causes

Multipath propagation

One of the causes of intersymbol interference is multipath propagation in which a wireless signal from a transmitter reaches the receiver via multiple paths. The causes of this include reflection (for instance, the signal may bounce off buildings), refraction (such as through the foliage of a tree) and atmospheric effects such as atmospheric ducting and ionospheric reflection. Since the various paths can be of different lengths, this results in the different versions of the signal arriving at the receiver at different times. These delays mean that part or all of a given symbol will be spread into the subsequent symbols, thereby interfering with the correct detection of those symbols. Additionally, the various paths often distort the amplitude and/or phase of the signal, thereby causing further interference with the received signal.





Bandlimited channels

Another cause of intersymbol interference is the transmission of a signal through a bandlimited channel, i.e., one where the frequency response is zero above a certain frequency (the cutoff frequency). Passing a signal through such a channel results in the removal of frequency components above this cutoff frequency. In addition, components of the frequency below the cutoff frequency may also be attenuated by the channel.

This filtering of the transmitted signal affects the shape of the pulse that arrives at the receiver. The effects of filtering a rectangular pulse not only change the shape of the pulse within the first symbol period, but it is also spread out over the subsequent symbol periods. When a message is transmitted through such a channel, the spread pulse of each individual symbol will interfere with following symbols.

Bandlimited channels are present in both wired and wireless communications. The limitation is often imposed by the desire to operate multiple independent signals through the same area/cable; due to this, each system is typically allocated a piece of the total bandwidth available. For wireless systems, they may be allocated a slice of the electromagnetic spectrum to transmit in (for example, FM radio is often broadcast in the 87.5–108 MHz range). This allocation is usually administered by a government agency; in the case of the United States this is the Federal Communications Commission(FCC). In a wired system, such as an optical fiber cable, the allocation will be decided by the owner of the cable. The bandlimiting can also be due to the physical properties of the medium - for instance, the cable being used in a wired system may have a cutoff frequency above which practically none of the transmitted signal will propagate.

Communication systems that transmit data over bandlimited channels usually implement pulse shaping to avoid interference caused by the bandwidth limitation. If the channel frequency response is flat and the shaping filter has a finite bandwidth, it is possible to communicate with no ISI at all. Often the channel response is not known beforehand, and an adaptive equalizer is used to compensate the frequency response.

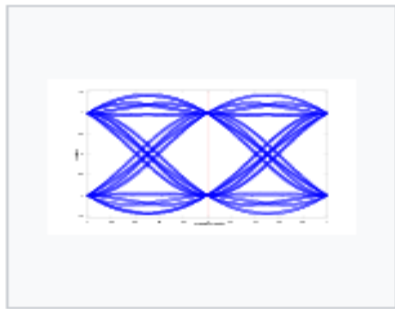
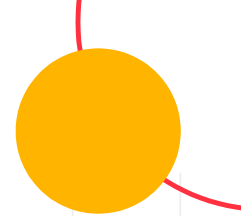


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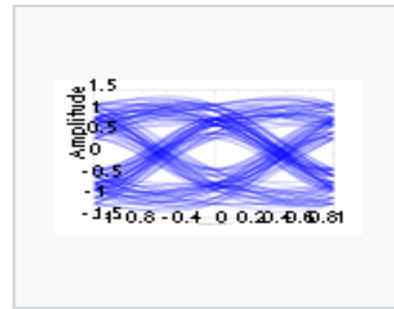
Effects on eye patterns

- One way to study ISI in a PCM or data transmission system experimentally is to apply the received wave to the vertical deflection plates of an oscilloscope and to apply a sawtooth wave at the transmitted symbol rate R ($R = 1/T$) to the horizontal deflection plates.
- The resulting display is called an eye pattern because of its resemblance to the human eye for binary waves.
- The interior region of the eye pattern is called the eye opening. An eye pattern provides a great deal of information about the performance of the pertinent system.
- The width of the eye opening defines the time interval over which the received wave can be sampled without error from ISI.
- It is apparent that the preferred time for sampling is the instant of time at which the eye is open widest.
- The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.
- The height of the eye opening, at a specified sampling time, defines the margin over noise.
- An eye pattern, which overlays many samples of a signal, can give a graphical representation of the signal characteristics.
- The first image below is the eye pattern for a binary phase-shift keying (PSK) system in which a one is represented by an amplitude of -1 and a zero by an amplitude of $+1$.
- The current sampling time is at the center of the image and the previous and next sampling times are at the edges of the image.
- The various transitions from one sampling time to another (such as one-to-zero, one-to-one and so forth) can clearly be seen on the diagram.
- The noise margin the amount of noise required to cause the receiver to get an error - is given by the distance between the signal and the zero amplitude point at the sampling time; in other words, the further from zero at the sampling time the signal is the better.
- For the signal to be correctly interpreted, it must be sampled somewhere between the two points where the zero-to-one and one-to-zero transitions cross. Again, the further apart these points are the better, as this means the signal will be less sensitive to errors in the timing of the samples at the receiver.
- The effects of ISI are shown in the second image which is an eye pattern of the same system when operating over a multipath channel.
- The effects of receiving delayed and distorted versions of the signal can be seen in the loss of definition of the signal transitions.
- It also reduces both the noise margin and the window in which the signal can be sampled, which shows that the performance of the system will be worse (i.e. it will have a greater bit error ratio).





The eye diagram of a binary PSK system



The eye diagram of the same system with multipath effects added





Equalization



In telecommunication, equalization is the reversal of distortion incurred by a signal transmitted through a channel. When a channel has been equalized the frequency domain attributes of the signal at the input are faithfully reproduced at the output.

Telephones, DSL lines and television cables use equalizers to prepare data signals for transmission.

Equalizers are critical to the successful operation of electronic systems such as analog broadcast television. In this application the actual waveform of the transmitted signal must be preserved, not just its frequency content. Equalizing filters must cancel out any group delay and phase delay between different frequency components.

Equalization :is the process of remove ISI and noise effects from the channel.

- It's located at receiver end of the channel.
- The goal of equalization is to mitigate the effects of ISI. However, this goal must be balanced so that in the process of removing ISI, the noise power in the received signal is not enhanced.





Comanding


•Comanding refers to a technique for compressing and then expanding (or decompressing) an analog or digital signal. It is a combination of the words "compressing" and "expanding."

This twin-sequential process is non-linear overall but linear over short periods of time. Data is compressed before being transmitted. Then, it is expanded at the receiving end using the same non-linear scale to restore it to its original form, but with reduced noise and crosstalk levels (meaning reduced disruption of, or interference with, signals in an adjacent circuit).

This disruption or interference is commonly from alternating current (AC), direct current (DC) or other transmission lines.

The electronic circuit responsible for comanding is called the compandor.

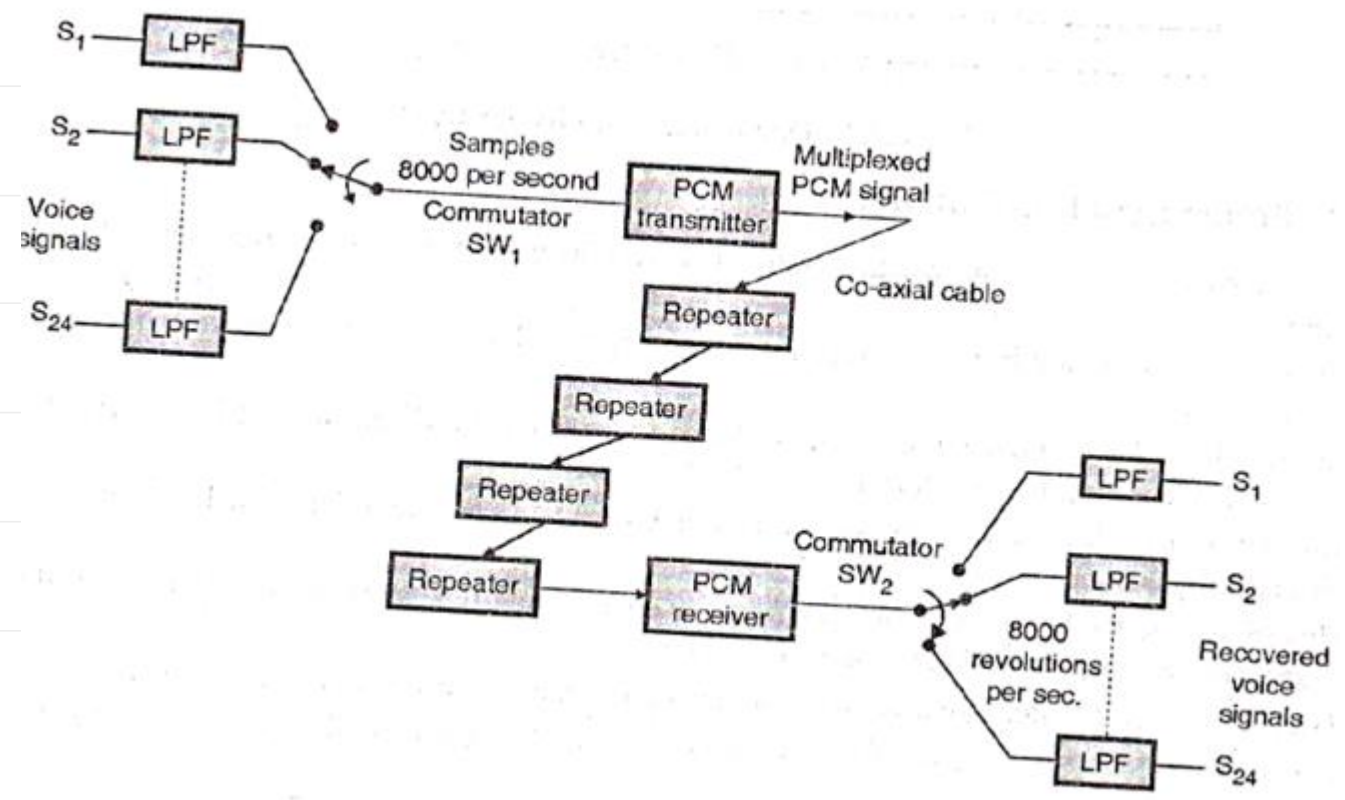
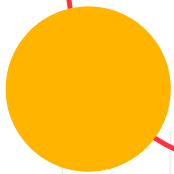
This term is also known as compansion.



Time Division Multiplexing of PCM signals

- Time division multiplexing (TDM) is a communications process that transmits two or more streaming digital signals over a common channel. In TDM, incoming signals are divided into equal fixed-length time slots.
- After multiplexing, these signals are transmitted over a shared medium and reassembled into their original format after demultiplexing. Time slot selection is directly proportional to overall system efficiency.
- Time division multiplexing (TDM) is also known as a digital circuit switched.
- TDM is comprised of two major categories: TDM and synchronous time division multiplexing (sync TDM). TDM is used for long-distance communication links and bears heavy data traffic loads from end users. Sync TDM is used for high-speed transmission.
- TDM can also be used within time division multiple access (TDMA) where stations sharing the same frequency channel can communicate with one another. GSM utilizes both TDM and TDMA.





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- **Time-division multiplexing (TDM)** is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the data rate of the transmission medium exceeds that of signal to be transmitted.
- When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required.
- Figure shows the basic time division multiplexing scheme, called as the T1T1 **digital system**.
- This system is used to convey multiple signals over telephone lines using wideband coaxial cable.
- Operation of the T1T1 system.
- The operation of the PCM-TDM system is as follows
- This system has been designed to accommodate 24 voice channels marked S1S1 to S24S24 .
- Each signal is bandlimited to 3.3kHz, and the sampling is done at a standard rate of 8 kHz.
- This is higher than the Nyquist rate.
- The sampling is done by the commutator switch SW1SW1 .
- These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW1SW1.
- These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW1SW1.



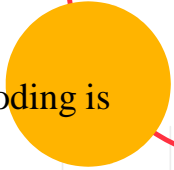


- Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and companding, as explained earlier.
- The resulting digital waveform is transmitted over a co-axial cable.
- Periodically, after every 6000 ft, the PCM-TDM signal is regenerated by amplifiers called “repeaters”.
- They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output.
- This ensures that the received signal is free from the distortions and noise.
- At the destination the signal is companded, decoded and demultiplexed, using a PCM receiver.
- The PCM receiver output is connected to different low pass filters via commutator switch SW1SW1 and SW2SW2 .
- Synchronization between the transmitter and receiver commutators SW1SW1 and SW2SW2 is essential in order to ensure proper communication.





Line codes



•A line code is the code used for data transmission of a digital signal over a transmission line. This process of coding is chosen so as to avoid overlap and distortion of signal such as inter-symbol interference.

Properties of Line Coding

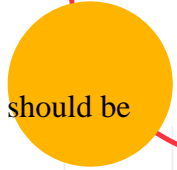
- As the coding is done to make more bits transmit on a single signal, the bandwidth used is much reduced.
- For a given bandwidth, the power is efficiently used.
- The probability of error is much reduced.
- Error detection is done and the bipolar too has a correction capability.
- Power density is much favorable.
- The timing content is adequate.
- Long strings of **1s** and **0s** is avoided to maintain transparency.

Types of Line Coding

- There are 3 types of Line Coding
- Unipolar
- Polar
- Bi-polar



Bandwidth of PCM system



- In digital we always refer sampling **rate** $R_s = \text{No. of samples (sent) per sec}$ as **bandwidth**. As per Nyquist theorem, sampling freq should be atleast twice the maximum content of the signal freq. Therefore, **bandwidth is half the signalling rate in PCM transmission**.
- Bandwidth of PCM signal depends on the bit rate and the pulse shape.

Let the bit rate be R (of the PCM signal generated), then

$$R = n \cdot f_s$$

n = number of bits on the PCM word ($M = 2^n$... M is no. of levels of quantization)

f_s = sampling rate to which analog signal is sampled

For no aliasing i.e. the Nyquist rate : $f_s = 2B$,

where B is the bandwidth of the analog signal that is to be converted.

Bandwidth of the PCM (BPCM) waveform is bounded by

$$B_{PCM} \geq R/2$$

$$B_{PCM} \geq n \cdot B$$

(so $R/2$ is the minimum bandwidth of the PCM signal)

For one using a rectangular pulse with polar NRZ (Non return to zero) line code:

$$B_{PCM} = R = n \cdot f_s = n \cdot 2 \cdot B$$

This means the bandwidth of PCM is greater than the bandwidth of the analog signal.





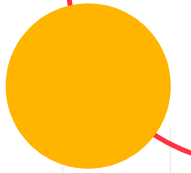
Noise in PCM systems



- There are mainly two types of noises that affect the performance of a PCM system, Channel noise, Quantization noise, quantization error.
- Noise is an unwanted disturbance in an electrical signal. Noise generated by electronic devices varies greatly as it is produced by several different effects.
- Some salient features of a PCM system are: Immunity to transmission noise and interference. It is possible to regenerate the coded signal along the transmission path. ... The quantizing of an analog signal is done by approximating the signal with a number of quantization levels.
- Quantization noise is typically caused by small differences (mainly rounding errors) between the actual analog input voltage of the audio being sampled and the specific bit resolution of the analog-to-digital converter being used. This noise is nonlinear and signal dependent.
- We can minimize the channel noise by sampling at the Nyquist rate of $2B(z)$. Finally, the PCM signal is obtained from the quantized PAM signal; a specific code must be developed to represent a particular quantized level. For example, the code used in the PCM signal is called Gray code.
- The performance of a PCM system is influenced by two major sources of noise: Channel noise, which is introduced anywhere between the transmitter output and the receiver input. Channel noise is always present, once the equipment is switched on.



MCQ



1. Companding is used

to overcome quantizing noise in PCM
in PCM transmitters, to allow amplitude limiting in the receivers
to protect small signals in PCM from quantizing distortion.
in PCM receivers, to overcome impulse noise.

Answer: to overcome quantizing noise in PCM

2. The biggest disadvantages of PCM is

its inability to handle analog signals
the high error rate which its quantizing noise introduces
its incompatibility with TDM
the large bandwidths that are required for it.

Answer: the large bandwidths that are required for it.

3. Indicate which of the following pulse modulation systems is analog PCM

Differential PCM
PWM
Delta modulation

Answer: PWM

4. Quantizing noise occurs in

time-division multiplexing
FDM
PCM
PWM

Answer: PCM



5. **Quantizing noise** can be reduced by increasing the number of samples per second. It is true,

yes, it is

no, it is not

not necessarily

none of these

Answer: none of these

6. **In PCM a system, the quantization noise depends upon**

the number of quantization levels only

the sampling rate only

both the sampling rate and the number of quantization levels

none of the above is correct

Answer: both the sampling rate and the number of quantization levels

7. **The signal-to quantization noise ratio in PCM system depends upon sampling rate**

number of quantization levels

message signal bandwidth

none of the above

Answer: number of quantization levels

8. **Indicate which of the following systems is digital?**

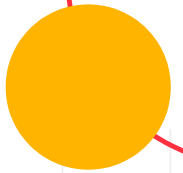
Pulse-position modulation

Pulse-code modulation

Pulse-width modulation

Pulse-frequency modulation

Answer: Pulse-code modulation



9. Quantizing noise occurs in

- time-division multiplex
- frequency-division multiplex
- pulse-code modulation
- pulse-width modulation

Answer: Pulse-code modulation

10. In order to reduce quantizing noise, one must

- increase the number of standard amplitudes
- send pulses whose sides are more nearly vertical
- use an R.F. amplifier in the receiver
- increase the number of samples per second

Answer: increase the number of standard amplitudes

11. The biggest disadvantages of PCM is

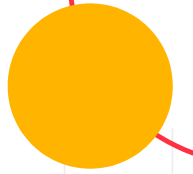
- its inability to handle analog signals
- the high error rate which its quantizing noise reduces
- its incompatibility with TDM
- the large bandwidth that are required for it

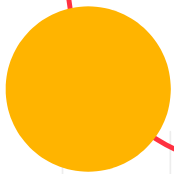
Answer: the large bandwidth that are required for it

12. The main advantage of PCM system is

- a lower bandwidth
- a lower power
- lower noise
- None of the above

Answer: lower noise





13. Quantization noise is produced in

all pulse modulation system

PCM

all modulation system

none of the above

Answer: PCM

14. One of the following systems is analog –

PCM

delta

differential PCM

PAM

Answer: PAM

15. For an efficient communication in PCM system number of samples per second must at least be equal to twice the highest modulating

frequency. Comment

Not necessary

A very important consideration

Who cares

80- 50, true

Answer: A very important consideration

16. In PCM system, output S/N increases

linearly with bandwidth

exponentially with bandwidth

inversely with bandwidth

none of these

Answer: linearly with bandwidth



17. The transmitted pulse in PCM system usually occur at a uniform rate for minimum bandwidth

true

false

Answer: true

18. In a DM system, the granular (idling) noise occurs when the modulating signal

increase rapidly

remains constant

decreases rapidly

the nature of modulating signal has nothing to do with this noise

Answer: remains constant

19. Quantization noise is produced in

all pulse modulation system

PCM

all modulation system

none of these

Answer: PCM

20. The main advantage of PCM signal is

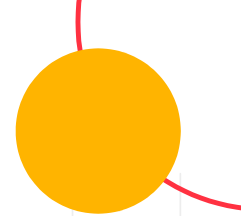
lower bandwidth

higher bandwidth

lower noise

none of these

Answer: lower noise



04

Delta Modulation



Delta Modulation (DM)

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

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

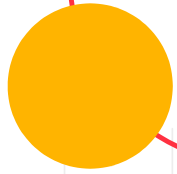
Features of Delta Modulation

Following are some of the features of delta modulation.

- An over-sampled input is taken to make full use of the signal correlation.
- The quantization design is simple.
- The input sequence is much higher than the Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., Δ .
- The bit rate can be decided by the user.
- This involves simpler implementation.
- Delta Modulation is a simplified form of DPCM technique, also viewed as **1-bit DPCM scheme**. As the sampling interval is reduced, the signal correlation will be higher.



Limitations of DM



1. Slope overload distortion

• This distortion arises because of large dynamic range of input signal. To reduce this error, the step size must be increased when slope of signal $x(t)$ is high. Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is known as Linear Delta Modulator (LDM).

2. Granular noise

• Granular noise occurs when step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount because of large step size. The error between the input and approximated signal is called granular noise. The solution to this problem is to make step size small. Adaptive Delta Modulation

• To overcome the quantization error due to slope overload distortion and granular noise, the step size (Δ) is made adaptive to variations in input signal $x(t)$. Particularly in the step segment of the $x(t)$, the step size is increased. Also, if the input is varying slowly, the step size is reduced.

Then this method is known as Adaptive Delta Modulation (ADM). The adaptive delta modulators can take continuous changes in the step size or discrete changes in the step size

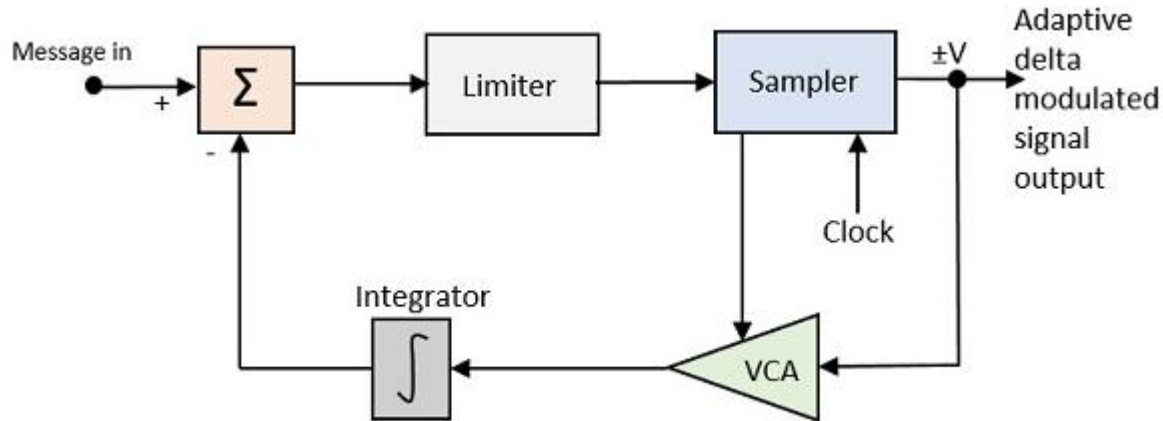


Adaptive Delta Modulation

In digital modulation, we have come across certain problem of determining the step-size, which influences the quality of the output wave.

A larger step-size is needed in the steep slope of modulating signal and a smaller stepsize is needed where the message has a small slope. The minute details get missed in the process. So, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of **Adaptive Delta Modulation**.

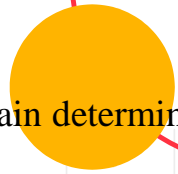
Following is the block diagram of Adaptive delta modulator.



Adaptive Delta Modulation



CONT... 



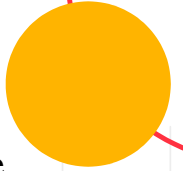
The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step-size and both are proportional.

ADM quantizes the difference between the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next values, for the faithful reproduction of the fast varying values.





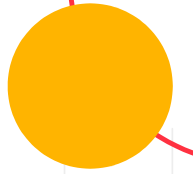
Noise in Delta Modulation



- Quantizing noise in delta modulation systems falls into two categories, granular noise and slope overload noise.
 - Slope overload noise occurs when the slope of the input signal is greater than the delta modulator is capable of reproducing
 - Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size.
 - Quantizing noise in delta modulation systems falls into two categories, granular noise and slope overload noise.
- Granular noise exists because the decoded output signal can assume only a specified number of levels in the range of interest.



Comparison between PCM and DM



- PCM stands for pulse code modulations while in DM is stands for delta modulations.
- In PCM feedback does not exist in transmitter or receiver while in delta modulation feedback existed in the transmitter.
- PCM requires the highest transmitter bandwidth while DM requires the lowest transmitter bandwidth.
- PCM is complex in terms of complexity of implementation, whereas DM is simple in terms of complexity of implementation.
- In PCM per sample 4,8, or 16 bits are used while in DM only one's bit is used per sample.
- PCM may be a technique wont to digitally represent sampled analog signals while in DM convert digital to analog and analog to digital converter.
- PCM is costly, DM is cheap.
- PCM has a good signal to noise ratio while in DM has a poor signal to noise ratio.
- PCM is mostly used in video telephony and audio telephony, DM is mostly used in speeches as well as images.
- PCM signals are required encoder and decoder both sides while in DM signals can modulate and demodulates.
- In PCM Quantization error depends on the number of levels, while in DM slope overload distortion is present.

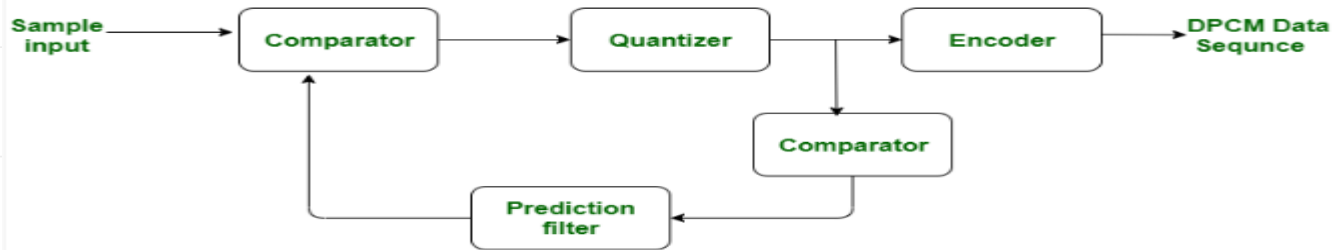


Delta or Differential PCM (DPCM)

Differential Pulse Code Modulation (DPCM):

DPCM stands for Differential Pulse Code Modulation, is same as Pulse Code Modulation technique used for reworking analog signal into digital signal. Differential Pulse Code Modulation has moderate signal to noise magnitude relation. Differential Pulse Code Modulation differs from Pulse Code Modulation as a results of it quantizes the excellence of the actual sample and expected value. that's the reason it's cited as as differential Pulse Code Modulation(DPCM).

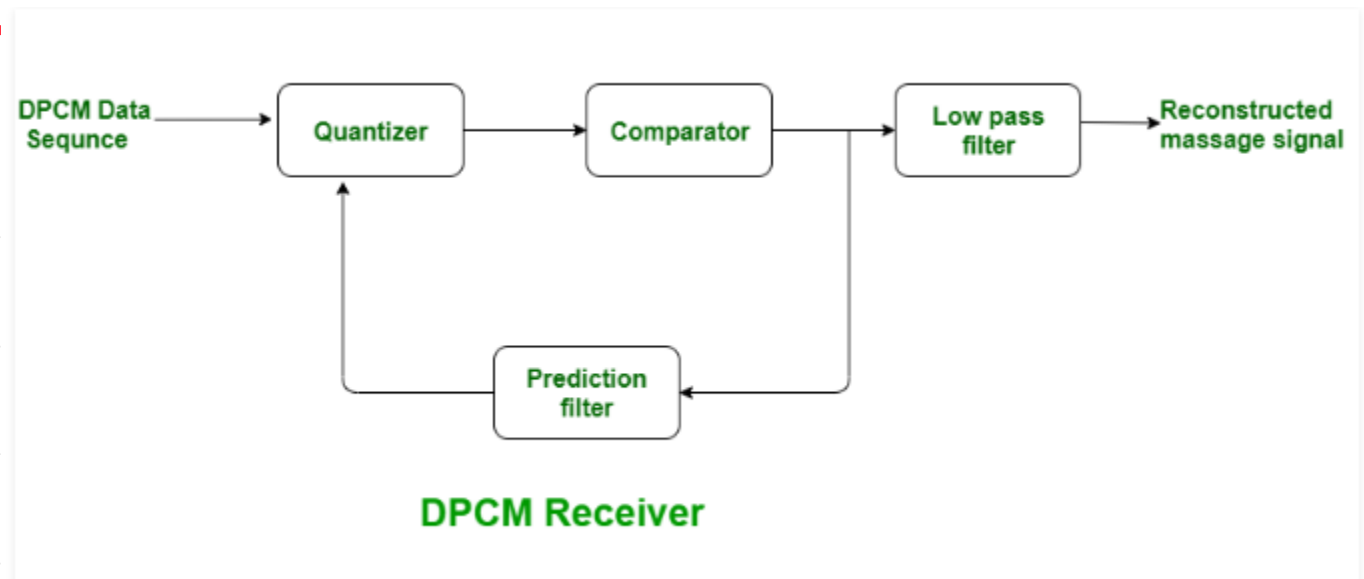
DPCM transmitter and DPCM receiver operations are given below through figure



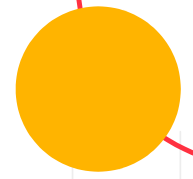
DPCM Transmitter

In the above diagram, if the signal is large then the next bit in digital data is 1 otherwise next bit is 0.



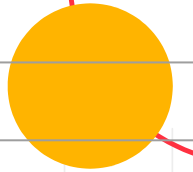


DPCM Receiver

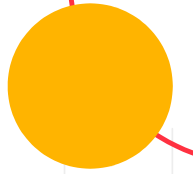


The difference between DM and DPCM

S.NO	Comparison based on	DM	DPCM
1.	Feedback	In DM, feedback exists in transmitter.	Here, feedback exists in both transmitter and receiver.
2.	signal to noise ratio	DM has poor signal to noise ratio.	DPCM has fair signal to noise ratio.
3.	Transmission bandwidth	It requires lowest bandwidth.	Here, DPCM requires less bandwidth than PCM.
4.	Levels, step size	In DM, step size is fixed.	While here, number of levels are fixed.
5.	Efficiency	DM is less efficient than DPCM.	DPCM is more efficient.
6.	Number of bits	In DM, only one bit is used per sample.	Here more than one but less than PCM(Pulse Code Modulation) bits are used.
7.	Quantization error and distortion	Slop overload distortion and granular noise are present.	Slop overload distortion and quantization noise are present.
8.	Applications	It is generally used in speeches	It is mostly used in videos and speeches.



MCQ



• • • • •
• • • • •

1) In uniform quantization process

- a. The step size remains same
- b. Step size varies according to the values of the input signal
- c. The quantizer has linear characteristics
- d. Both a and c are correct

ANSWER: (d) Both a and c are correct

2) The process of converting the analog sample into discrete form is called

- a. Modulation
- b. Multiplexing
- c. Quantization
- d. Sampling

ANSWER: (c) Quantization

3) The characteristics of compressor in μ -law companding are

- a. Continuous in nature
- b. Logarithmic in nature
- c. Linear in nature
- d. Discrete in nature

ANSWER: (a) Continuous in nature

4) The modulation techniques used to convert analog signal into digital signal are

- a. Pulse code modulation
- b. Delta modulation
- c. Adaptive delta modulation
- d. All of the above

ANSWER: (d) All of the above



5) **The sequence of operations in which PCM is done is**

- a. Sampling, quantizing, encoding
- b. Quantizing, encoding, sampling
- c. Quantizing, sampling, encoding
- d. None of the above

ANSWER: (a) Sampling, quantizing, encoding

6) **In PCM, the parameter varied in accordance with the amplitude of the modulating signal is**

- a. Amplitude
- b. Frequency
- c. Phase
- d. None of the above

ANSWER: (d) None of the above

7) **One of the disadvantages of PCM is**

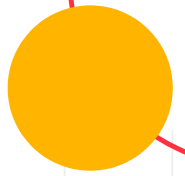
- a. It requires large bandwidth
- b. Very high noise
- c. Cannot be decoded easily
- d. All of the above

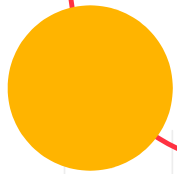
ANSWER: (a) It requires large bandwidth

8) **The expression for bandwidth BW of a PCM system, where v is the number of bits per sample and f_m is the modulating frequency, is given by**

- a. $BW \geq v f_m$
- b. $BW \leq v f_m$
- c. $BW \geq 2 v f_m$
- d. $BW \geq 1/2 v f_m$

ANSWER: (a) $BW \geq v f_m$





) **The error probability of a PCM is**

- a. Calculated using noise and inter symbol interference
- b. Gaussian noise + error component due to inter symbol interference
- c. Calculated using power spectral density
- d. All of the above

ANSWER: (d) All of the above

10) In Delta modulation,

- a. One bit per sample is transmitted
- b. All the coded bits used for sampling are transmitted
- c. The step size is fixed
- d. Both a and c are correct

ANSWER: (d) Both a and c are correct

11) In digital transmission, the modulation technique that requires minimum bandwidth is

- a. Delta modulation
- b. PCM
- c. DPCM
- d. PAM

ANSWER: (a) Delta modulation

12) In Delta Modulation, the bit rate is

- a. N times the sampling frequency
- b. N times the modulating frequency
- c. N times the nyquist criteria
- d. None of the above

ANSWER: (a) N times the sampling frequency

13) In Differential Pulse Code Modulation techniques, the decoding is performed by

- a. Accumulator
- b. Sampler
- c. PLL
- d. Quantizer

ANSWER: (a) Accumulator



14) **DPCM is a technique**

- a. To convert analog signal into digital signal
- b. Where difference between successive samples of the analog signals are encoded into n-bit data streams
- c. Where digital codes are the quantized values of the predicted value
- d. All of the above

ANSWER: (d) All of the above

15) **DPCM suffers from**

- a. Slope over load distortion
- b. Quantization noise
- c. Both a & b
- d. None of the above

ANSWER: (c) Both a & b

16) **The noise that affects PCM**

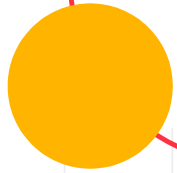
- a. Transmission noise
- b. Quantizing noise
- c. Transit noise
- d. Both a and b are correct

ANSWER: (d) Both a and b are correct

17) **The factors that cause quantizing error in delta modulation are**

- a. Slope overload distortion
- b. Granular noise
- c. White noise
- d. Both a and b are correct

ANSWER: (d) Both a and b are correct



18) Granular noise occurs when

- a. Step size is too small
- b. Step size is too large
- c. There is interference from the adjacent channel
- d. Bandwidth is too large

ANSWER: (b) Step size is too large

19) The crest factor of a waveform is given as –

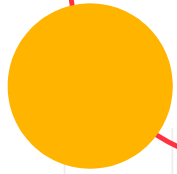
- a. $2 \text{Peak value} / \text{rms value}$
- b. $\text{rms value} / \text{Peak value}$
- c. $\text{Peak value} / \text{rms value}$
- d. $\text{Peak value} / 2\text{rms value}$

ANSWER: (c) Peak value/ rms value

20) The digital modulation technique in which the step size is varied according to the variation in the slope of the input is called

- a. Delta modulation
- b. PCM
- c. Adaptive delta modulation
- d. PAM

ANSWER: (c) Adaptive delta modulation






REFERANCE

1]John G.Proakis,M. Salehi, Communication Systems Engineering, 2nd ed. New Delhi, India. PHI Learning Private Limited, 2009.

[2] R.P Singh and S.D Sapre, Communication Systems Analog & Digital, 2nd





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