E – Contents

Analog and Digital Communication

Explain the radio frequency spectrum used in communication system

In the radio communication system, the frequencies ranging from a few kilohertz to many gigahertz all are being used for various purposes.

Let us discuss the applications of various frequency bands.

The frequencies most commonly used in early days were from about 300 kHz to 3 MHz and were called as medium frequencies(MF).

The frequencies in the range 30 kHz to 300 kHz are known as the low frequencies (LF).

The frequencies in the range 3 kHz to 30 kHz are called as very low frequencies (VLF).

On the higher frequency side, high frequencies (HF) will cover the frequency range from 3 MHz to 30 MHz.

Then very high frequency (VHF) from 30 MHz TO 300 MHz and so on .

Following table presents the details of entire usable frequency spectrum and its applications.

S.No	Frequency Band	Wavelength	Applications
1.	30 Hz – 300 Hz. Extremely low frequencies(ELF)	10^4 km to 10^3 km	Power transmission
2.	300 Hz – 3 kHz. Voice frequencies (VF)	10^3 km to 100 km	Audio applications
3.	3 kHz – 30 kHz. Very low frequencies (VLF)	100 km to 10 km	Submarine communications. Navy, Military communications
4.	30 kHz – 300 kHz . Low frequencies (LF)	10 km to 1 km. Long waves.	Aeronautical and marine navigation, these frequencies act as sub carriers.
5.	300 kHz – 30 MHz. Medium frequencies (MF)	1 km to 100 m. Medium waves	AM radio broadcast, Marine and aeronautical communications
6.	3 MHz – 30 MHz. High Frequencies (HF)	100 m TO 10 m. Short waves	Shortwave transmission, Amateur and CB communication.
7.	30 MHz – 300 MHz. Very high frequencies (VHF)	10 m to 1 m	T.V. broadcasting. F.M. broadcasting

7.	30 MHz – 300 MHz. Very high frequencies (VHF)	10 m to 1 m	T.V. broadcasting. F.M. broadcasting
8.	300 MHz – 3 GHz. Ultra high frequencies (UHF)	1 m to 10 cm. Microwaves	UHF T.V. Channels, Cellular phones, Military applications
9.	3 GHz – 30 GHz (SHF)	10-1 m to 10-2 m	Satellite communications and Radar
10.	30 -300 GHz (EHF)	10-2 m to 10-3 m	Satellites and specialized Radars.

The radio frequency (RF) spectrum

Explain the Difference Between Analog and Digital Communications

The difference between analog and digital communication system is explained in the table below :

Analog Communication	Digital Communication
Transmitted modulated signal is analog in nature.	Transmitted signal is digital i.e. train of digital pulses.
Amplitude, frequency or phase variations in the transmitted signal represent the information or message.	Amplitude, width or position of the transmitted pulses is constant. The message is transmitted in the form of code words.
Noise immunity is poor for AM, but improved for FM and PM.	Noise immunity is excellent.
It is not possible to separate out noise and signal. Therefore, repeaters cannot be used.	It is possible to separate signal from noise. Therefore, repeaters can be used.
Coding is not possible.	Coding techniques can be used to detect and correct the errors.
Bandwidth required is lower than that for the digital modulation method.	Due to higher bit rates, higher channel bandwidth is required.
FDM is used for multiplexing.	TDM is used for multiplexing.
Not suitable for transmission of secret information in military applications.	Due to coding techniques, it is suitable for military applications.
Analog modulation systems are AM, FM, PM, PAM, AWM, etc.	Digital modulation systems are PCM, DM, ADM, DPCM, etc.

Explain the need for modulation in a communication system

Modulation

In the modulation process, two signals are used namely the modulating signal and the carrier .

The modulating signal is nothing but the baseband signal or information signal while the carrier is a high frequency sinusoidal signal .

In the modulation process, some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal . This modulated signal is then transmitted by the transmitter .

The receiver demodultes the received modulated signal and gets the original information signal back .

Thus, demodulation is exactly opposite to modulation .

In the process of modulation the carrier wave actually acts as carrier which carries the information signal from the transmitter to receiver .

Need of Modulation

You may be ask, when the baseband signal can be transmitted directly why to use the modulation ?

The answer is that the baseband transmission has many limitations which can be overcome using modulation . It is explained below .

In the process of modulation, the baseband signal is translated i.e., shifted from low frequency to high frequency . This frequency shift is proportional to the frequency of carrier.

Advantages of Modulation

1. Reduction in the height of antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Multiplexing is possible
5. Improves quality of reception

We will discuss each of these advantages in detail below .

1. Reduction in the height of antenna

For the transmission of radio signals, the antenna height must be multiple of λ/4 ,where λ is the wavelength . $\lambda = c / f$

where c : is the velocity of light

f: is the frequency of the signal to be transmitted

The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows :

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^3} = 7500\ meters\ i.e. 7.5\ km$$

The antenna of this height is practically impossible to install .

Now, let us consider a modulated signal at f = 1 MHz . The minimum antenna height is given by,

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^6} = 75\ meters$$

This antenna can be easily installed practically . Thus, modulation reduces the height of the antenna .

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver can not separate them from each other .

Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals .

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted . They get heavily attenuated .

The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance .

The modulation process increases the frequency of the signal to be transmitted . Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously .

This is possible only with modulation.

The multiplexing allows the same channel to be used by many signals . Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time .

5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent . This improves quality of reception .

Block Diagram of Communication System with Detailed Explanation

Communication System

Communication is the process of establishing connection or link between two points for information exchange.

OR

Communication is simply the basic process of exchanging information.

The electronics equipements which are used for communication purpose, are called communication equipments. Different communication equipments when assembled together form a **communication system**.
Typical example of communication system are line telephony and line telegraphy, radio telephony and radio telegraphy, radio broadcasting, point-to-point communication and mobile communication, computer communication, radar communication, television broadcasting, radio telemetry, radio aids to navigation, radio aids to aircraft landing etc.

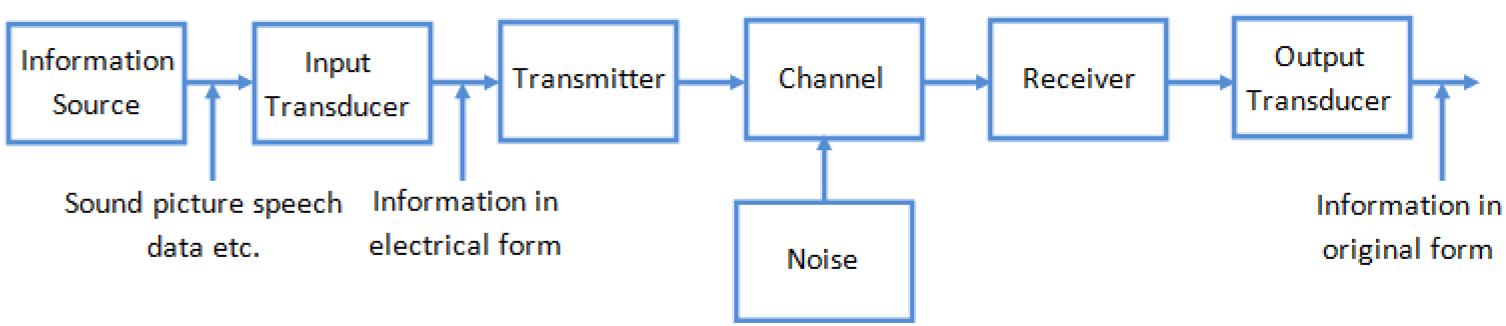
The Communication Process

In the most fundamental sense, communication involves the transmission of information from one point to another through a succession of process as listed below :

1. The generation of a thought pattern or image in the mind of an originator.
2. The description of that image, with a certain measure of precision, by a set of oral visual symbols.
3. The encoding of these symbols in a form that is suitable for transmission over a physical medium of interest.
4. The transmission of the encoded symbols to the desired destination.
5. The decoding and reproduction of the original symbols.
6. The recreation of the original thought pattern or image, with a definable degradation in quality, in the mind of recipient.

Block Diagram of Communication System

Fig. shows the block diagram of a general communication system, in which the different functional elements are represented by blocks.



Fig

The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.

Now, we shall discuss the functioning of these blocks.

(i) Information Source

As we know, a communication system serves to communicate a message or information. This information originates in the information source.

In general, there can be various messages in the form of words, group of words, code, symbols, sound signal etc. However, out of these messages, only the desired message is selected and communicated.

Therefore, we can say that the function of information source is to produce required message which has to be transmitted.

(ii) Input Transducer

A transducer is a device which converts one form of energy into another form.

The message from the information source may or may not be electrical in nature. In a case when the message produced by the information source is not electrical in nature, an input transducer is used to convert it into a time-varying electrical signal.

For example, in case of radio-broadcasting, a microphone converts the information or massage which is in the form of sound waves into corresponding electrical signal.

(iii) Transmitter

The function of the transmitter is to process the electrical signal from different aspects.

For example in radio broadcasting the electrical signal obtained from sound signal, is processed to restrict its range of audio frequencies (upto 5 kHz in amplitude modulation radio broadcast) and is often amplified.

In wire telephony, no real processing is needed. However, in long-distance radio communication, signal amplification is necessary before modulation.

Modulation is the main function of the transmitter. In modulation, the message signal is superimposed upon the high-frequency carrier signal.

In short, we can say that inside the transmitter, signal processings such as restriction of range of audio frequencies, amplification and modulation of are achieved.

All these processings of the message signal are done just to ease the transmission of the signal through the channel.

(iv) The Channel and The Noise

The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver.

There are two types of channels, namely point-to-point channels and broadcast channels.

Example of point-to-point channels are wire lines, microwave links and optical fibres. Wire-lines operate by guided electromagnetic waves and they are used for local telephone transmission.

In case of microwave links, the transmitted signal is radiated as an electromagnetic wave in free space. Microwave links are used in long distance telephone transmission.

An optical fibre is a low-loss, well-controlled, guided optical medium. Optical fibres are used in optical communications.

Although these three channels operate differently, they all provide a physical medium for the transmission of signals from one point to another point. Therefore, for these channels, the term point-to-point is used.

On the other hand, the broadcast channel provides a capability where several receiving stations can be reached simultaneously from a single transmitter.

An example of a broadcast channel is a satellite in geostationary orbit, which covers about one third of the earth’s surface.

During the process of transmission and reception the signal gets distorted due to noise introduced in the system.

Noise is an unwanted signal which tend to interfere with the required signal. Noise signal is always random in character. Noise may interfere with signal at any point in a communication system. However, the noise has its greatest effect on the signal in the channel.

(v) Receiver

The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal. This reproduction of the original signal is accomplished by a process known as the demodulation or detection. Demodulation is the reverse process of modulation carried out in transmitter.

(vi) Destination

Destination is the final stage which is used to convert an electrical message signal into its original form.

For example in radio broadcasting, the destination is a loudspeaker which works as a transducer i.e. converts the electrical signal in the form of original sound signal.

Describe the Classification of Electronic Communication System

Classification of Electronic Communication System

The communication systems may be classified into various cataeories as shown in fig. 1.

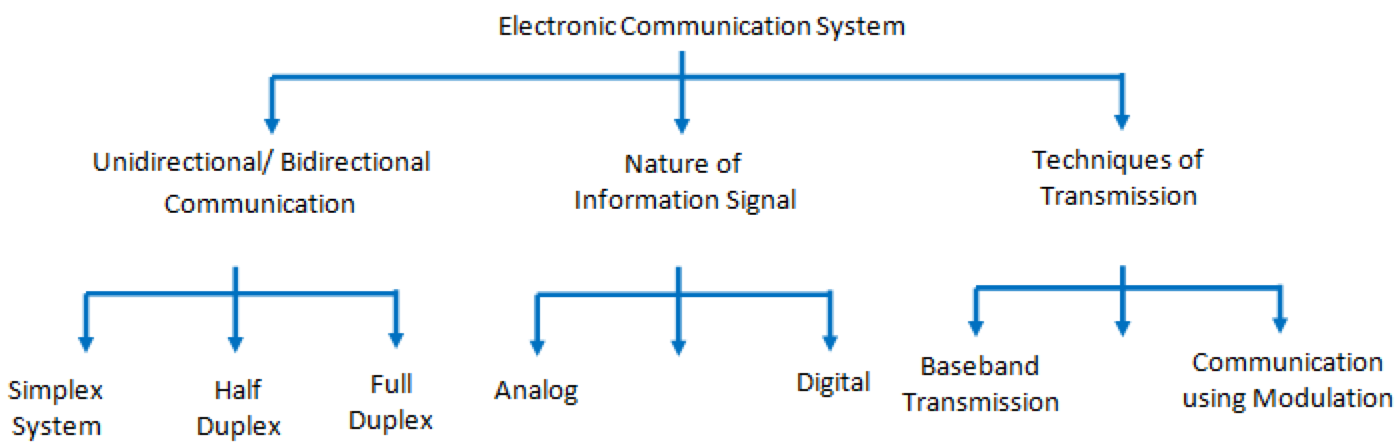


Fig 1: Classification of electronic communication system

It shows that the electronic communication system may be basically categorised into three groups based on :

1. Whether the system is unidirectional or bidirectional

2. Whether it uses as analog or digital signal
3. Whether the system uses baseband transmission or uses some kind of modulation

Classification Based on Direction of Communication

Based on whether the system communicates only in one direction or otherwise, the communication systems are classified as under :

1. Simplex System
2. Half duplex System
3. Full duplex System

Fig. 2 shows this classification .

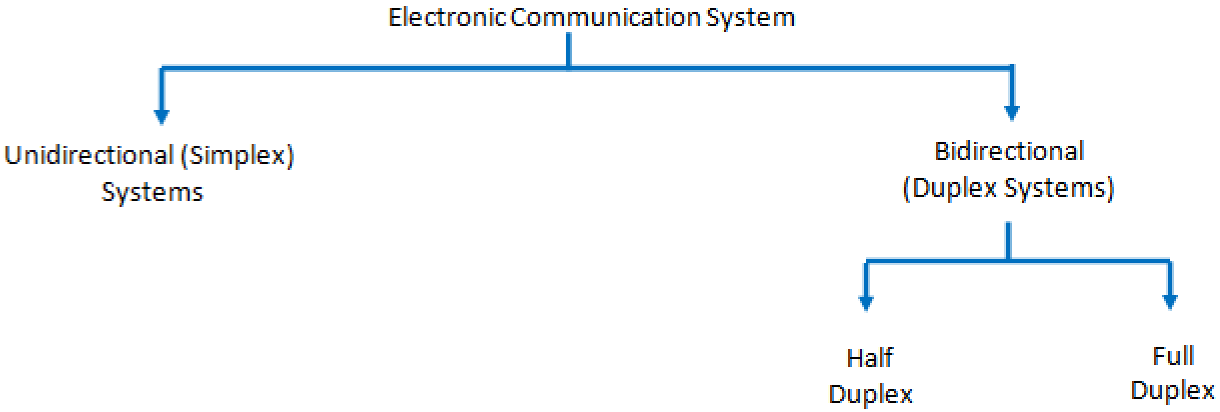


Fig. 2

Simplex System

In these systems, the information is communicated in only one direction .

For example, the radio or TV broadcasting system can only transmit, they can not receive .

Another example of simplex communication is the information transmitted by the telemetry system of a satellite to earth . The telemetry system transmits information about the physical status of the satellite such as its position or temperature .

Half duplex System

These systems are bidirectional, i.e. they can transmit as well as receive but not simultaneously .

At a time, these systems can either transmit or receive, for example, a transreceiver or walky talky set .

The direction of communication alternates . The radio communications such as those in military, fire fighting, citizen band (CB) and amateur radio are half duplex system .

Full duplex System

These are truly bidirectional systems as they allow the communication to take place in both the directions simultaneously .

These systems can transmit as well as receive simultaneously . For example, the telephone systems.

However, the bulk of electronic communications is two -way.

The best example of full duplex communication system is telephone system .

Classification Based On The Nature of Information Signal

Fig.3 shows another way of classifying the electronic communication system .

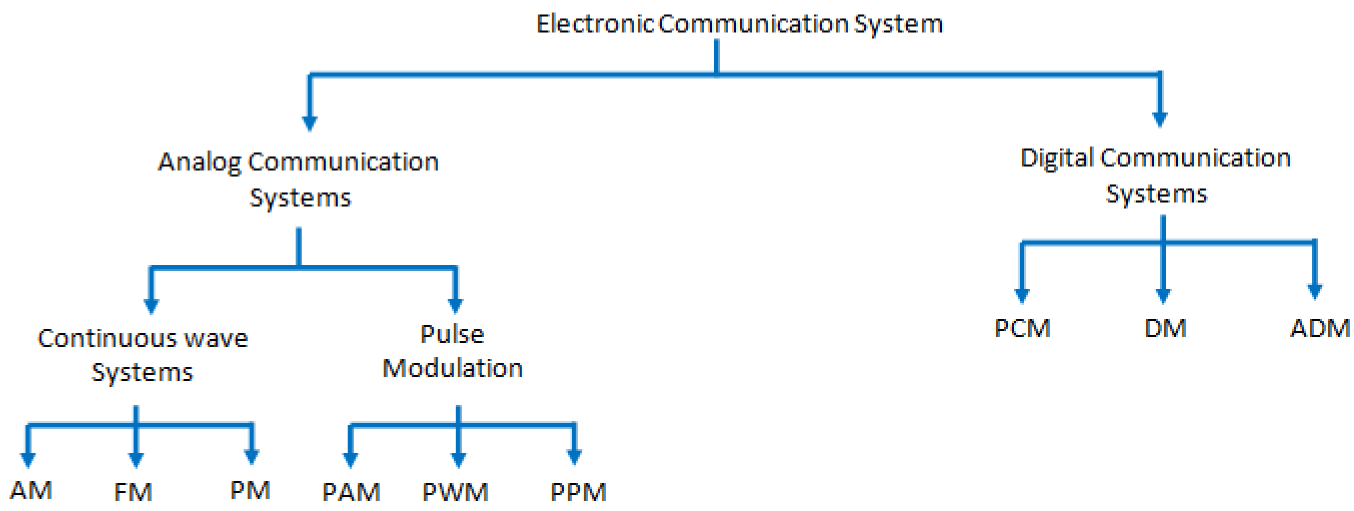


Fig. 3

They are classified into two categories namely :

1. Analog communication system
2. Digital communication system

Analog communication

The modulation systems or techniques in which one of the characteristics of the carrier is varied in proportion with the instantaneous value of modulating signal is called as analog modulation system .

If the carrier is sinusoidal, then its amplitude, frequency or phase is changed in accordance with the modulating signal to obtain AM, FM or PM respectively . These are continuous wave modulation systems.

Analog modulation can be pulsed modulation as well . Here the carrier is in the form of rectangular pulse . The amplitude, width or position of the carrier pulses is varied in accordance with the modulating signal to obtain the PAM, PWM or PPM outputs .

Examples of analog modulation

Following are the examples of analog modulation systems :

1. Amplitude modulation (AM)
2. Frequency modulation (FM)
3. Phase modulation (PM)
4. Pulse Amplitude Modulation (PAM)
5. Pulse Width Modulation (PWM)
6. Pulse Position Modulation (PPM)

Advantages of analog communication

1. Transmitters and receivers are simple
2. Low bandwidth requirement
3. FDM (Frequency division multiplexing) can be used

Drawbacks of analog communication

1. Noise affects the signal quality
2. It is not possible to separate noise and signal
3. Repeaters can not be used between transmitter and receiver
4. Coding is not possible
5. It is not suitable for the transmission of secret information

Applications

1. Radio broadcasting (AM and FM)

2. TV broadcasting
3. Telephones

Digital Communication

The modulation system or technique in which the transmitted signal is in the form of digital pulses of constant amplitude, constant frequency and phase is called as digital modulation system .

Pulse code modulation (PCM) and delta modulation (DM) are the examples of digital modulation .

In PCM and DM , a train of digital pulses is transmitted by the transmitter . All the pulses are of constant amplitude, width and position . The information is contained in the combination of the transmitted pulses .

Advantages of Digital Communication

1. Due to the digital nature of the transmitted signal, the interference of additive noise does not introduce many errors . Hence, digital communication has a better noise immunity .
2. Due to the channel coding techniques used in digital communication, it is possible to detect and correct the errors introduced during the data transmission .
3. Repeaters can be used between transmitter and receiver to regenerate the digital signal . This improves the noise immunity further .
4. Due to the digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc .
5. TDM (Time Division Multiplexing) technique can be used to transmit many voice channels over a single common transmission channel .
6. Digital communication is useful in military applications where only a few permitted receivers can receive the transmitted signal .
7. Digital communication is becoming simpler and cheaper as compared to the analog communication due to the nvention of high speed computers and integrated circuits (ICs) .

Drawbacks of Digital Communication

1. The bit rates of digital systems are high . Therefore, they require a larger channel bandwidth as compared to analog system .
2. Digital modulation needs synchronization in case of synchronous modulation .

Applications of Digital Communications

1. Long distance communication between earth and space ships .
2. Satellite communication
3. Military communication
4. Telephone systems
5. Data and computer communications

Classification Based on the Technique of Transmission

Based on the technique used for the signal transmission, we can categorise the electronic communication system as under :

1. Baseband transmission system
2. Communication systems using modulation

Baseband Transmission

In baseband transmission systems, the baseband signals (original information signals) are directly transmitted .

Example of these type of systems are telephone networks where the sound signal converted into the electrical signal is placed directly on the telephone lines for transmission .

Another example of baseband transmission is computer data transmission over the coaxial cables in the computer networks .

Thus, the baseband transmission is the transmission of the original information signal as it is .

Limitation of Baseband Transmission

The baseband transmission can not be used with certain mediums e.g., it can not be used gor the radio transmission where the medium is free space . This is because the voice signal can not travel long distance in air . It gets suppressed after a short distance . Therefore, for the radio communication of baseband signals, a technique called modulation is used .

Modulation

In the modulation process, two signals are used namely the modulating signal and the carrier .

The modulating signal is nothing but the baseband signal or information signal while the carrier is a high frequency sinusoidal signal .

In the modulation process, some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal . This modulated signal is then transmitted by the transmitter .

The receiver demodultes the received modulated signal and gets the original information signal back .

Thus, demodulation is exactly opposite to modulation .

In the process of modulation the carrier wave actually acts as carrier which carries the information signal from the transmitter to receiver .

Frequency Translation in Modulation Process

The baseband signal or modulating signal is a low frequency signal . For example, the audio signal is present in the frequency range from 20 Hz TO 20 kHz . But due to modulation, the same signal now gets translated to a higher frequency range .

Explain Delta Modulation in detail with suitable diagram.

Delta Modulation

In PCM the signaling rate and transmission channel bandwidth are quite large since it transmits all the bits which are used to code a sample. To overcome this problem, Delta modulation is used.

Working Principle

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Input signal x(t) is approximated to step signal by the delta modulator.This step size is kept fixed.

The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e., +Δ and -Δ.

Now, if the difference is positive, then approximated signal is increased by one step, i.e., 'Δ'. If the difference is negative, then approximated signal is reduced by 'Δ' .

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted.

Hence, for each sample, only one binary bit is transmitted.

Fig.1 shows the analog signal x(t) and its staircase approximated signal by the delta modulator.

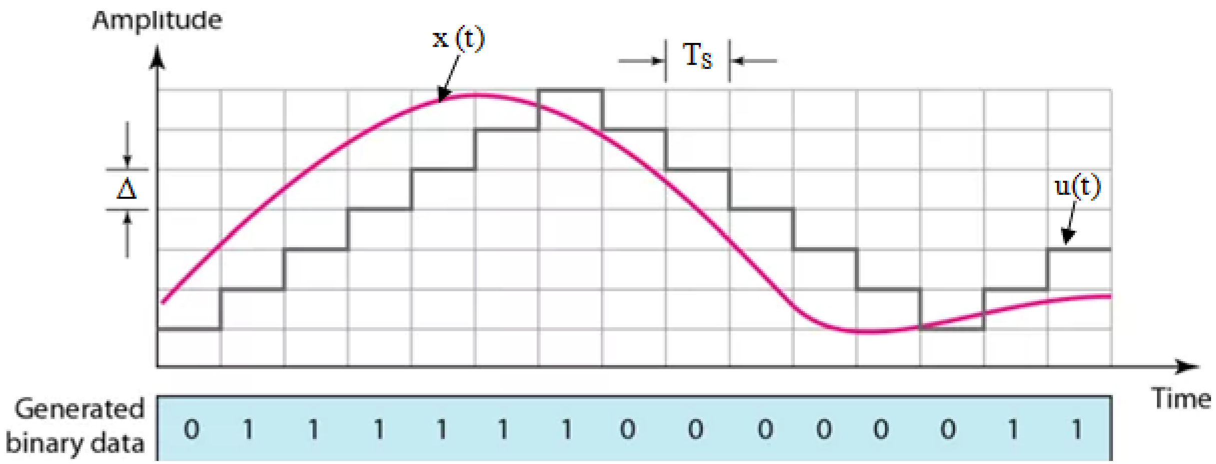


Fig.1. Delta Modulation Waveform

Mathematical Expressions

The error between the sampled value of x(t) and last approximated sample is given as:

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

Where e(nT_s) = error at present sample
$x(nT_s)$ = sampled signal of x(t)
$\hat{x}(nT_s)$ = last sample approximation of the staircase waveform

If we assume $u(nT_S)$ as the present sample approximation of staircase output, then

$$u[(n-1)T_s) = \hat{x}\ (nT_s)$$

$$= \text{last sample approximation of staircase waveform}$$

Let us define a quantity $b(\ nT_S)$ in such a way that,

$$b(nT_s) = \Delta\, sgn[e(nT_s)]$$

This means that depending on the sign of error $e(\ nT_S)$, the sign of step size Δ is decided. In other words we can write

$$b(nT_s) = \begin{cases} +\Delta & if\ x(nT_s) \geq \hat{x}\ (nT_s) \\ -\Delta & if\ x(nT_s)\hat{x}\ < (nT_s) \end{cases}$$

Also if $b\ (nT_S) = +\Delta$ then a binary '1' is transmitted
and if $b\ (nT_S) = -\Delta$ then a binary '0' is transmitted
Here T_S = sampling interval.

Transmitter

Fig. 2 (a) shows the transmitter . It is also known as Delta modulator.

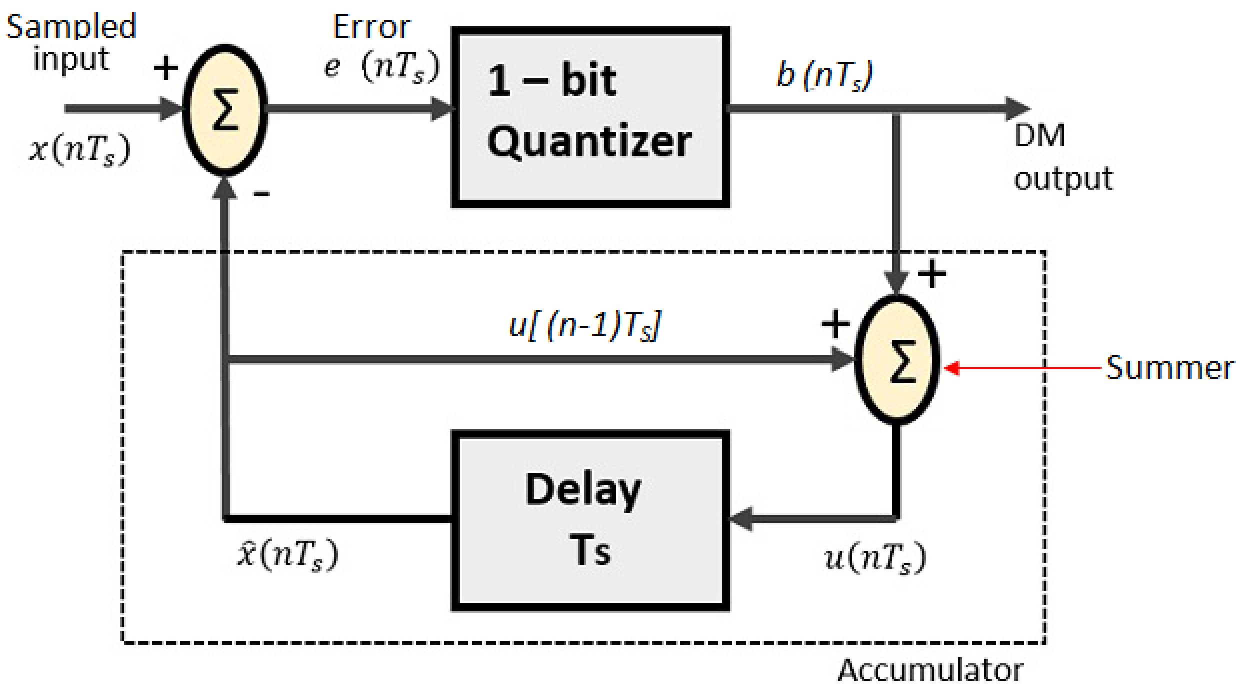


Fig.2 (a) Delta Modulation Transmitter

It consists of a 1-bit quantizer and a delay circuit along with two summer circuits.

The summer in the accumulator adds quantizer output ($\pm\Delta$) with the previous sample approximation . This gives present sample approximation. i.e.,

$$u(nT_s) = u((nT_s - T_s) + [\pm\Delta]$$

$$or \qquad u(nT_s) = u[(n-1)T_s] + b(nT_s)$$

The previous sample approximation $u[(n\text{-}1)T_s]$ is restored by delaying one sample period T_s .

The samples input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.
Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of $+\Delta$ or $-\Delta$.
If the step size is $+\Delta$, then binary ‘1’ is transmitted and if it is $-\Delta$, then binary ‘0’ is transmitted .

Receiver

At the receiver end also known as delta demodulator, as shown in fig.2 (b) , it comprises of a low pass filter(LPF), a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

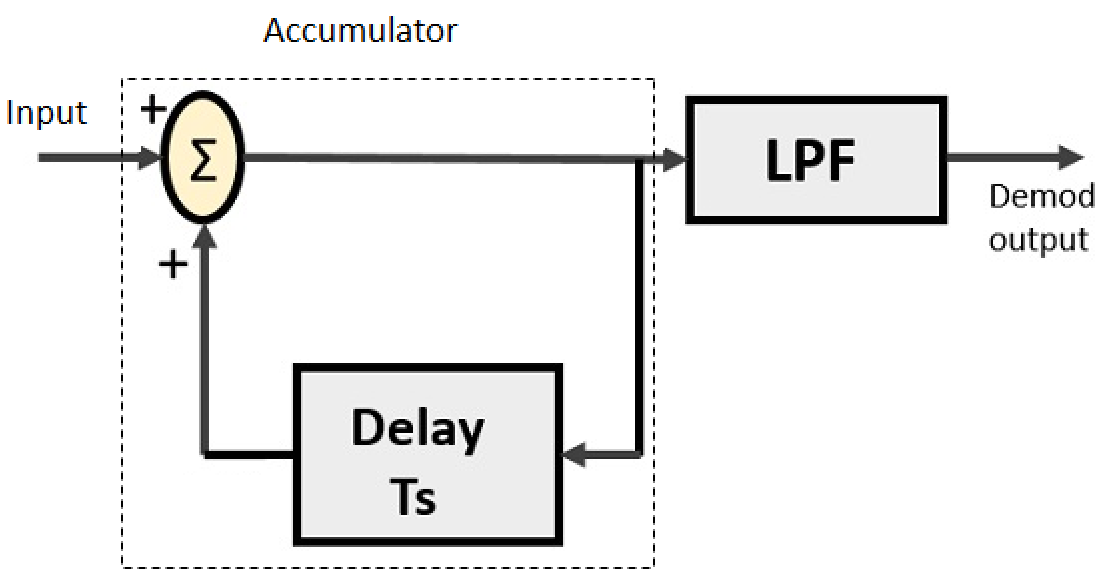


Fig.2 (b) Delta Modulation Receiver

The accumulator generates the staircase approximated signal output and is delayed by one sampling period T_s.
It is then added to the input signal.

If the input is binary ‘1’ then it adds +Δ step to the previous output (which is delayed).

If the input is binary ‘0’ then one step ‘Δ’ is subtracted from the delayed signal.

Also, the low pass filter smoothens the staircase signal to reconstruct the original message signal x(t) .

Advantages and Disadvantages of Delta Modulation

Advantages of Delta Modulation

The delta modulation has certain advantages over PCM as under :

1. Since, the delta modulation transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM .
2. The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter required in delta modulation.

Disadvantages of Delta Modulation

The delta modulation has two major drawbacks as under :

1. Slope overload distortion
2. Granular or idle noise

Now, we will discuss these two drawbacks in detail.

1.Slope Overload Distortion

This distortion arises because of large dynamic range of the input signal.

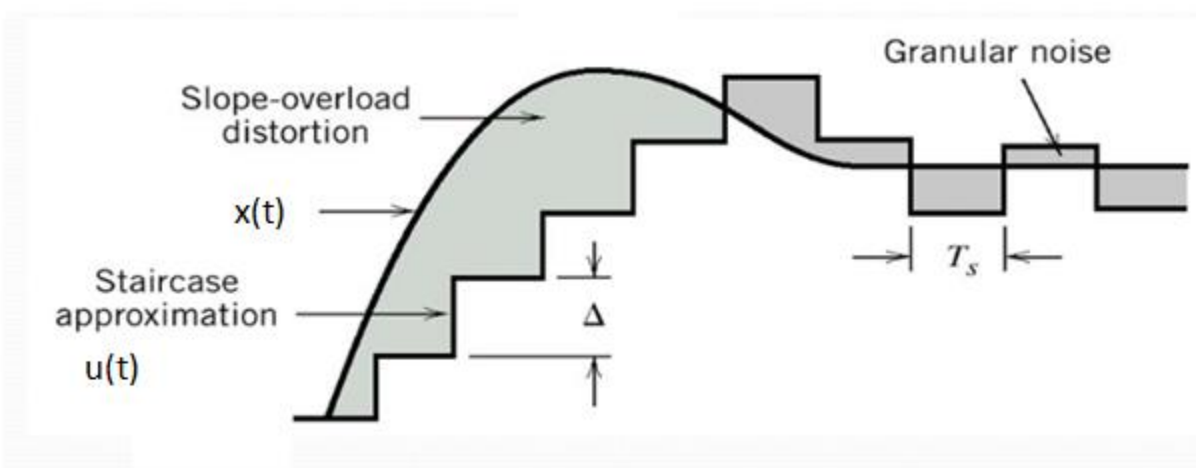


Fig.1 : Quantization errors in Delta modulation

We can observe from fig.1 , the rate of rise of input signal x(t) is so high that the staircase signal can not approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t).

Hence, there is a large error between the staircase approximated signal and the original input signal x(t).

This error or noise is known as **slope overload distortion** .
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 remain fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is known as **Linear Delta Modulator (LDM)** .

2. Granular or Idle Noise

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.

Fig.1 shows that when the input signal is almost flat , the staircase signal u(t) keeps on oscillating by ±Δ around the signal.

The error between the input and approximated signal is called **granular noise**.
The solution to this problem is to make the step size small .

Explain Differential Pulse Code Modulation

Differential Pulse Code Modulation

It may be observed that the samples of a signal are highly correlated with each other. This is due to the fact that any signal does not change fast. Which means , its value from present sample to next sample does not vary by a large amount.

The adjacent samples of the signal carry the same information with a little difference.

When these samples are encoded by a standard PCM system, the resulting encoded signal contains some redundant information.

Redundant Information in PCM

Fig.1 shows a continuous time signal x(t) by dotted line. This signal is sampled by flat top sampling at intervals T_s , $2T_s$, $3T_s$ ….. nT_s .

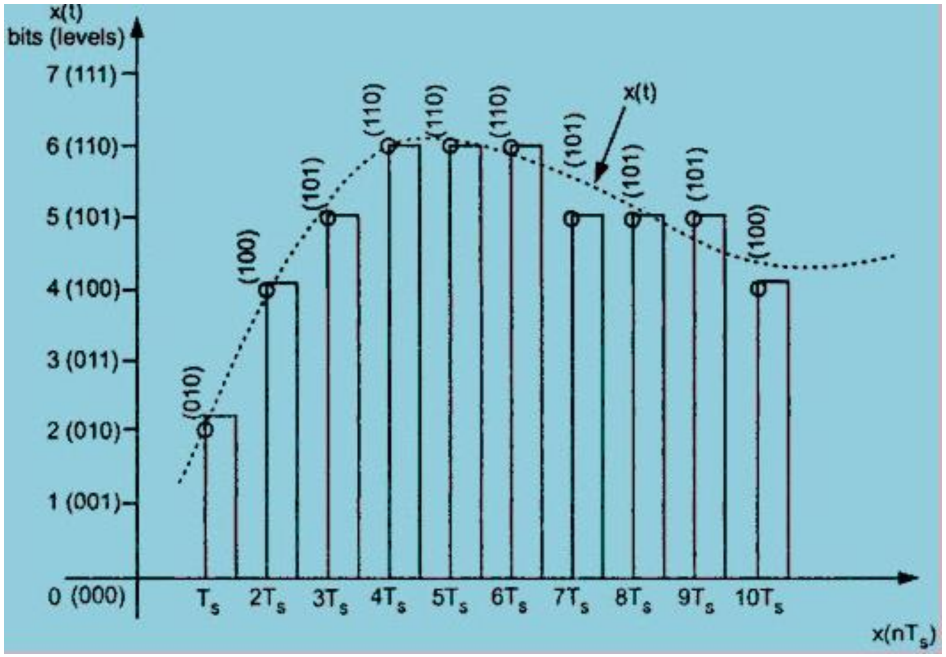


Fig.1 : Illustration of redundant information in PCM

The sampling frequency is selected to be higher than nyquist rate.

The samples are encoded by using 3 bit (7 levels) PCM.

The sample is quantized to the nearest digital level as shown by small circles in fig.1 .

The encoded binary value of each sample is written on the top of the samples.

We can observe from fig.1 that the samples taken at $4T_s$, $5T_s$ and $6T_s$ are encoded to same value of (110). This information can be carried only by one sample.

But three smaples are carrying the same information means that it is redundant .

We consider another example of samples taken at $9T_s$ and $10T_s$. The difference between these samples only due to last bit and first two bits are redundant, as they do not change.
If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit one sample will also be reduced.

This type of digital pulse modulation technique is called as Differential Code Modulation (DPCM).

Working Principle

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples.

The prediction may not be exact but it is very close to the actual sample value.

Fig.2 shows the transmitter of DPCM system.

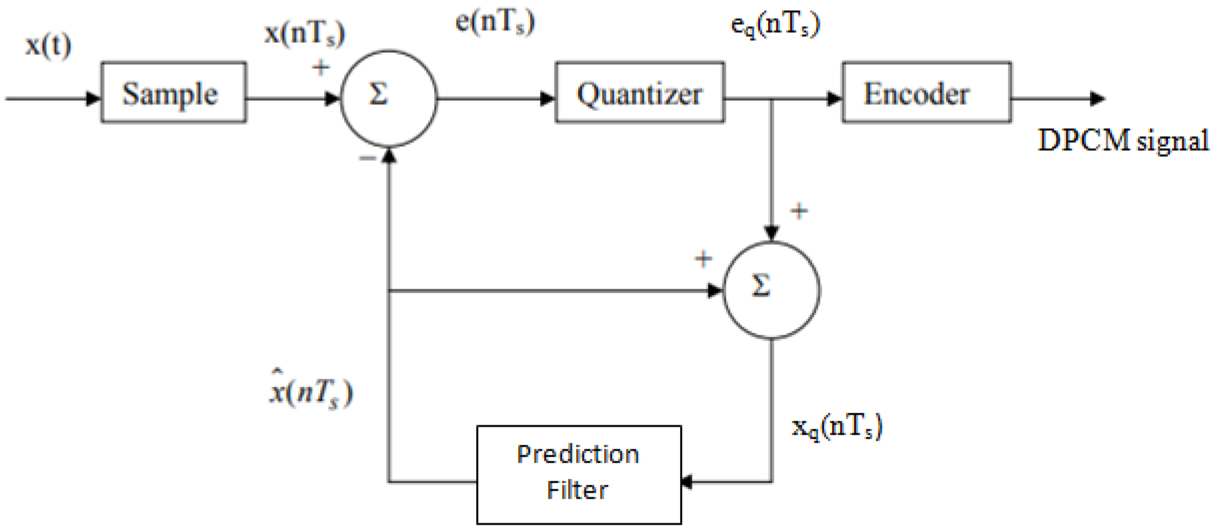


Fig.2 : A Differential pulse code modulation

The sampled signal is denoted by $x(nT_s)$ and predicted signal is denoted by $x\hat{}(nT_s)$.
The comparator finds out the difference between the actual sample value $x(nT_s)$ and predicted sample value $x\hat{}(nT_s)$.
This is known as prediction error and it is denoted by $e(nT_s)$.
It can be defined as ,

$e(nT_s) = x(nT_s) – x\hat{}(nT_s)$……………………….(1)
The predicted value is produced by using a prediction filter.

The quantizer output signal gap $e_q(nT_s)$ and previous prediction is added and given as input to the prediction filter.This signal is called $x_q(nT_s)$.
This makes the prediction more and more close to the actual sampled signal.

We can observe that the quantized error signal $e_q(nT_s)$ is very small and can be encoded by using small number of bits.
Thus number of bits per sample are reduced in DPCM.

The quantizer output can be written as ,

$e_q(nT_s) = e(nT_s) + q(nT_s)$………………………..(2)
Here, $q(nT_s)$ is the quantization error.
As shown in fig.2, the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantizer output. i.e.,
$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s)$……………………..(3)
Substituting the value of $e_q(nT_s)$ from eq.(2) in the above eq. (3) , we get,
$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$ ………………….(4)
eq.(1) is written as,

$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$
$\therefore \quad e(nT_s) + \hat{x}(nT_s) = x(nT_s)$
Therefore, substituing the value of $e(nT_s) + \hat{x}(nT_s)$ from the above equation into eq. (4), we get,
$x_q(nT_s) = x(nT_s) + q(nT_s)$ …………………..(5)

Reception of DPCM Signal

Fig.3 shows the block diagram of DPCM receiver.

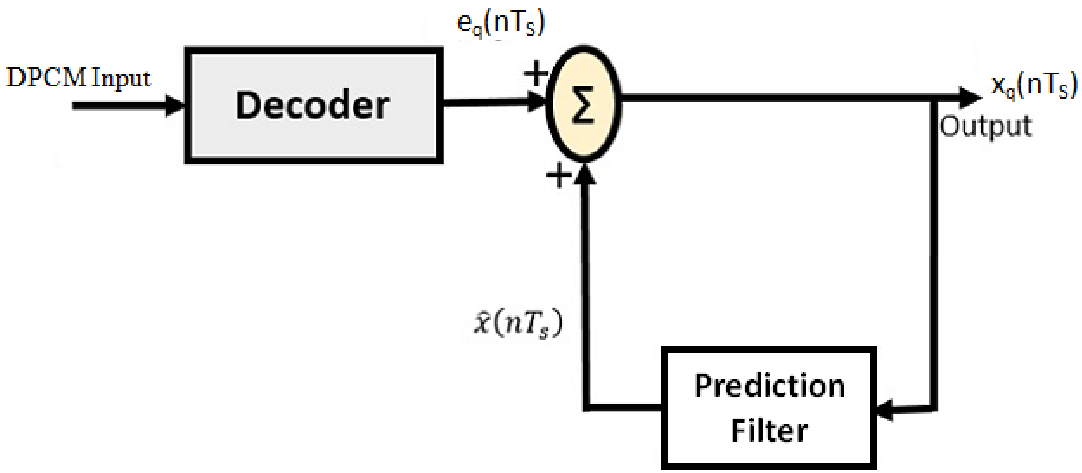


Fig.3 : DPCM Receiver

The decoder first reconstructs the quantized error signal from incoming binary signal.

The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal.

Thus the signal at the receiver differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

Advantages of DPCM

1. As the difference between $x(nT_s)$ and $x\hat{}(nT_s)$ is being encoded and transmitted by the DPCM technique, a small difference voltage is to be quantized and encoded.
2. This will require less number of quantization levels and hence less number of bits to represent them.
3. Thus signaling rate and bandwidth of a DPCM system will be less than that of PCM.

Comparison Between PCM, DM, ADM and DPCM

In this article we will compare Pulse Code Modulation (PCM), Delta Modulation (DM), Adaptive Delta Modulation (ADM) and Differential Pulse Code Modulation.
We have already discussed all these modulation techniques in our previous articles.

Comparison between all these modulation techniques is shown in the table below.

S.NO	Parameter of Comparison	Pulse Code Modulation (PCM)	Delta Modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
1.	Number of bits	It can use 4,8, or 16 bits per sample.	It uses only one bit for one sample	It uses only one bit for one sample	Bits can be more than one but are less than PCM.
2.	Levels and step size	The number of levels depends on number of bits. Level size is fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies.	Number of levels is fixed.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise are present.	Quantization noise is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4.	Transmission bandwidth	Highest bandwidth is required since numbers of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is less than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6.	Complexity of Implementation	System is complex.	Simple	Simple	Simple

Conversion of Analog Signals to Digital Signals

In communication systems, sometimes it happens that we are available with an analog signal, and we have to transmit a digital signal for that particular application.

In such cases, we have to convert the analog signal to digital signal. That means that we have to convert a continuous time signal in the form of digits.

To see how a signal can be converted from analog signal to digital form, let us consider an analog signal x(t) as shown in fig.1(a).

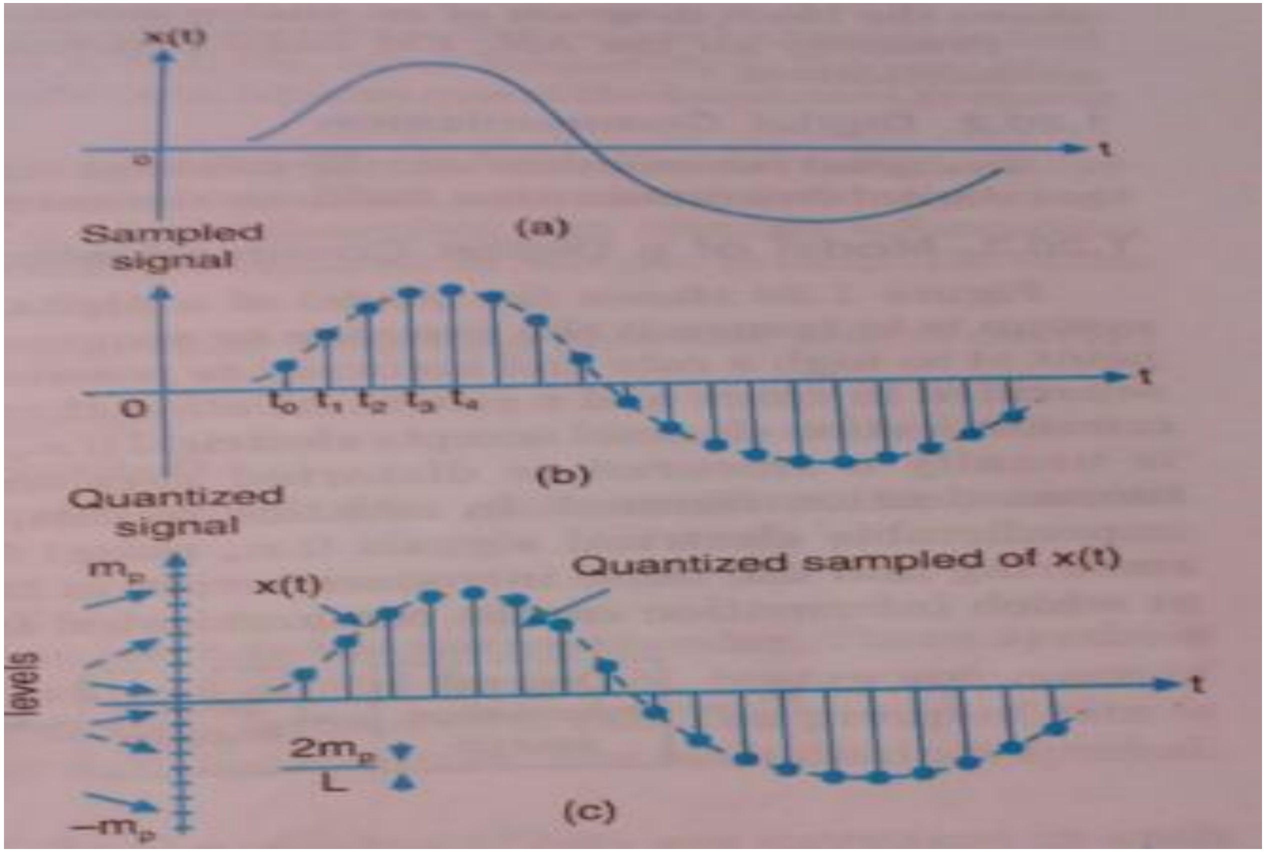


Fig.1 : (a) An Analog Signal, (b) Samples of Analog signal, (c) Quantization

First of all , we get sample of this signal according to the sampling theorem.

For this purpose, we mark the time-instants t_0, t_1 ,t_2 and so on , at equal time-intervals along the time axis. At each of these time-instants , the magnitude of the signal is measured and thus samples of the signal are taken. Fig.1(b) shows a representation of the signal of fig.1(a) in terms of its samples.

Now, we can say that the signal in fig.1(b) is defined only at the sampling instants.

This means that, it no longer is a continuous function of time, but rather, it is a discrete-time signal.

However, since the magnitude of each sample can take any value in a continuous range, the signal in fig.1(b) is still an analog signal.

This difficulty is neatly resolved by a process known as quantization. In quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels.

As shown in fig.1(c), amplitudes of the signal x(t) lie in the range ($-m_p$, m_p) which is partitioned into L intervals, each of magnitude $\Delta v = 2m_p / L$.
Now, each sample is approximated or rounded off to the nearest quantized level as shown in fig.1(c) .

Since each sample is now approximated to one of the L numbers, therefore, the information is digitized.

The quantized signal is an approximation of the original one. We can improve the accuracy of the quantized signal to any desired degree simply by increasing the number of levels L .

Comparison of PCM and Analog Modulation

Compairing PCM and Analog Modulation

In this tutorial we shall compare PCM and Analog Modulation in detail.

The threshold effect in PCM is similar to a property of analog modulation methods such as FM or PPM.

The property is that , these systems tend to reduce the wideband noise above the threshold levels.

The PCM also provides the wideband noise reduction if it is operated above its threshold which is given by:

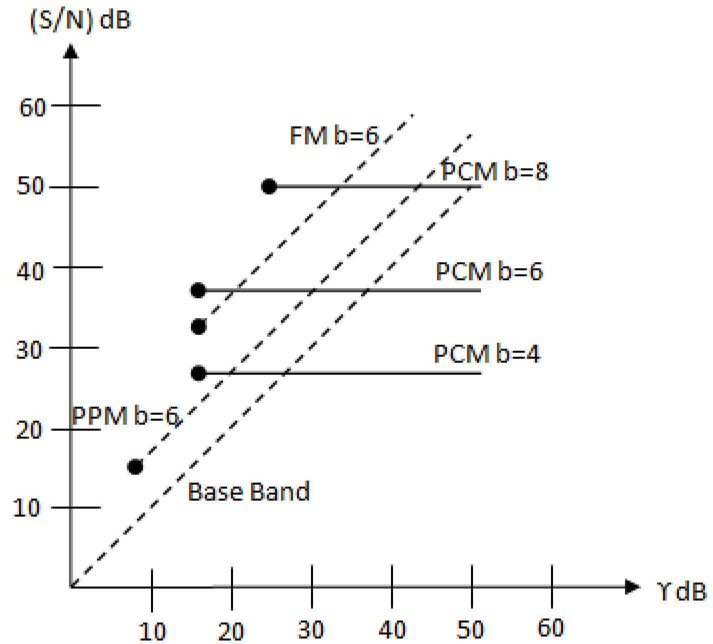
$$(\frac{S}{N})_D = 3q^2 S_x$$

Where $q=2^v$ for binary PCM and $q= M^v$ for M-ary PCM.
We assume that the sampling frequency is close to the Nyquist rate and bandwidth BW = Nf_m Hz.
Then, $q=M^v= M^b$ where b = BW/ f_m is known as the bandwidth ratio.
Therefore, we have,

$$(\frac{S}{N})_D = 3 \times (M^V)^2 S_x = 3 \times M^{2V} S_x$$

Here, $(S/N)_D$ = Signal to noise ratio at the destination
S_x = Signal power at the destination
Here, it may be noted that the signal to noise ratio $(S/N)_D$ is proportional to M^{2b} which is much higher than the $(S/N)_D$ of the wideband FM which is proportional to only b or b^2.
Hence, PCM performs better than FM .

Fig.1 shows the performance of various modulation types as a function of γ .



Comparison of PCM and analog communication

Fig.1

All the curves in fig.1 have been plotted for $S_x = 1/2$. The dots indicate the threshold points. The PCM curves have been drawn for M = 2 and v = b.

Conclusions

Some of the important observations from fig.1 may be listed as under:

1. For PCM if b is constant , then increase in γ beyond the threshold value γth (corresponding to the threshold point) does not increase $(S/N)_D$ at all . Let us observe the flat PCM curves in fig.1. hence, PCM must be operated just above the threshold.

2. Near threshold, the PCM does offer some advantages over FM and PPM, with thesame values of b and $(S/N)_D$.
3. However, this advantage is gained at the expense of more complicated and expensive circuitry.
4. The $(S/N)_D$ for FM and PPM increases linearly with increase in the value of γ and becomes better than that of PCM for higher values of γ .

Benefits of PCM

Fig.1 reveals the following benefits of using the PCM :

1. PCM allows the use of regenerative repeaters. This improves its noise performance.
2. PCM allows the transmission of analog signals in the form of digital signals.

PCM is not used for Radio Broadcasting

In Radio Broadcasting, a relatively large signal to noise ratio (typically of the order of 60 dB) is required.

To get this level of $(S/N)_D$, the PCM with $b>8$ is needed.
However, we can obtain the same performance with an FM system with $b = 6$ and with much simpler transmitter and receiver circuits.

Therefore, higher bandwidth requirement and complicated circuitry are the drawbacks of PCM which does not allow its use fr the radio, TV broadcasting applications .

Mapping of course / module to the Programme Learning Outcomes

Learning Outcome of the course	Programme Outcomes											
	01	02	03	04	05	06	07	08	09	10	11	12

Analog Transmission

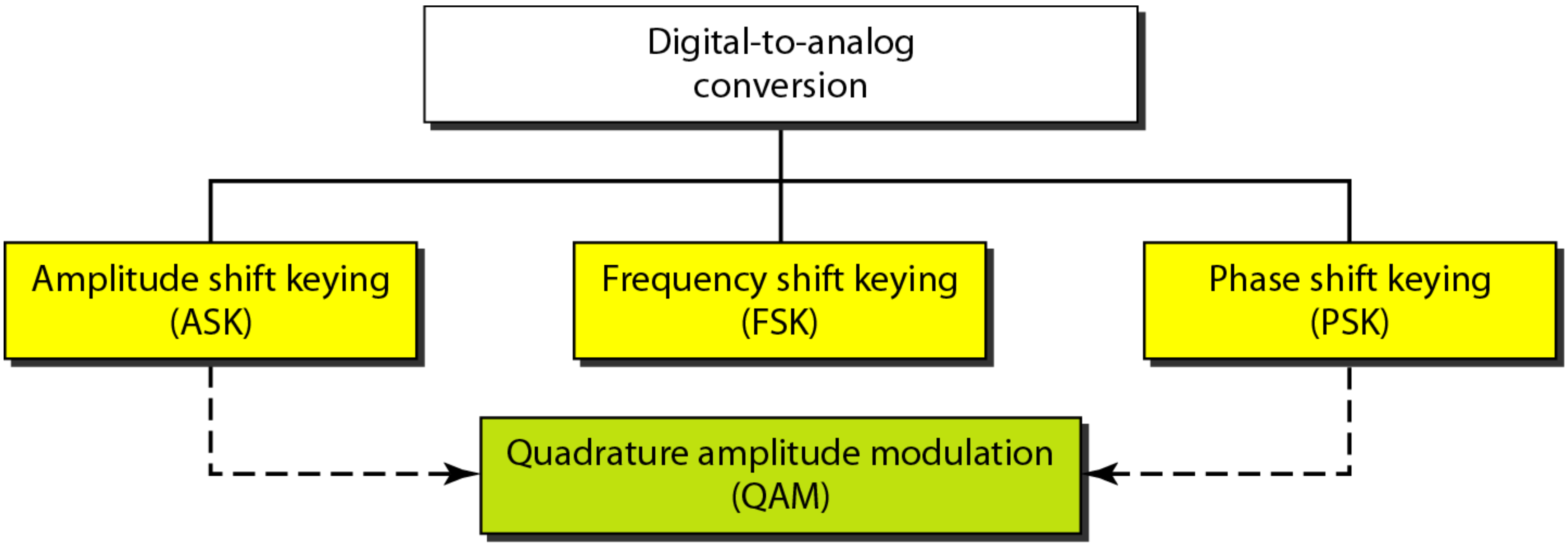
•Three mechanisms of *modulating* digital data into an analog signal by *altering* any of the three *characteristics* of analog signal:

→ ASK : Amplitude shift keying

→ FSK : Frequency shift keying

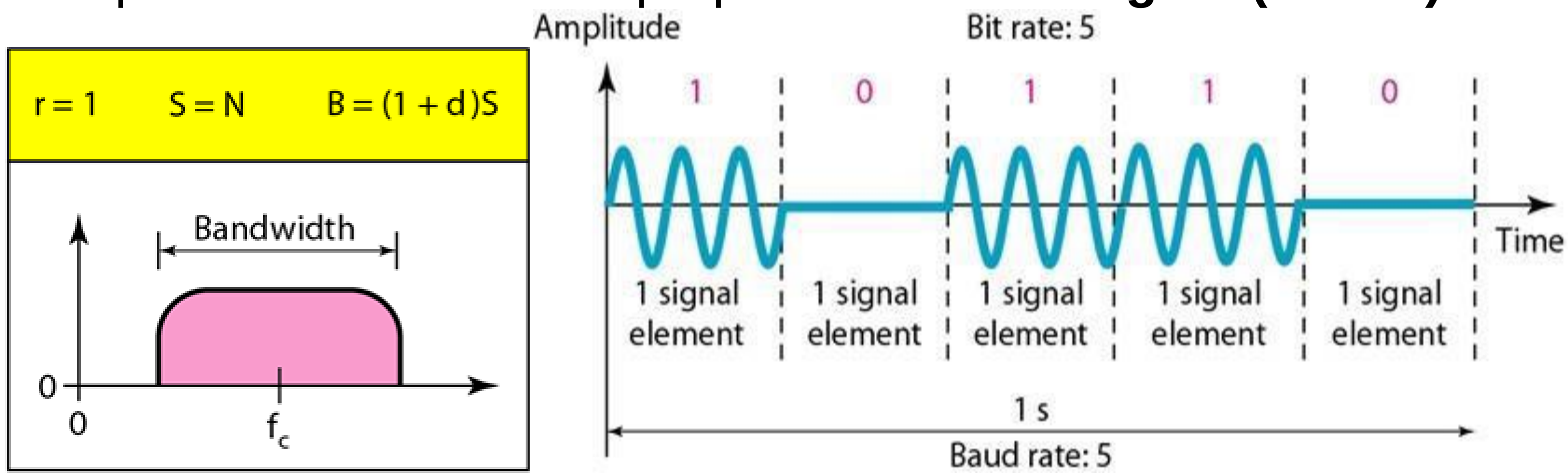
→ PSK : Phase shift keying

Types of Analog Transmission



Amplitude Shift Keying

modulation produces aperiodic composite signal, with continuous set of frequencies bandwidth is proportional to the **signal (baud) rate**



bandwidth is divided into two with two carrier frequencies, as The figure shows the positions of two carrier frequencies and the bandwidths.
The available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.
• •

Merits and Demerits

- Values represented by different amplitudes of carrier
- Usually, one amplitude is zero – i.e. presence and absence of carrier is used
- Susceptible to sudden gain changes
- Inefficient
- Typically used up to 1200bps on voice grade lines
- Used over optical fiber

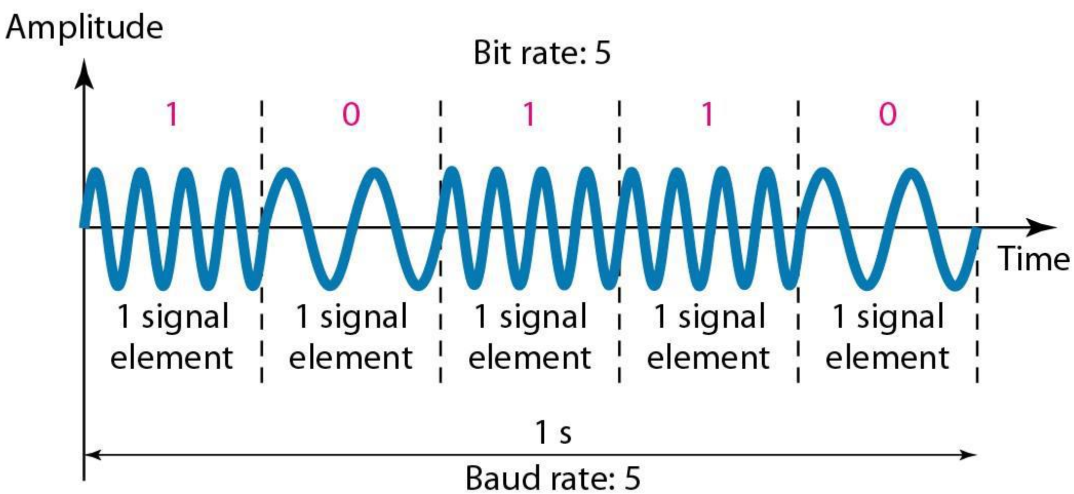
Frequency Shift Keying

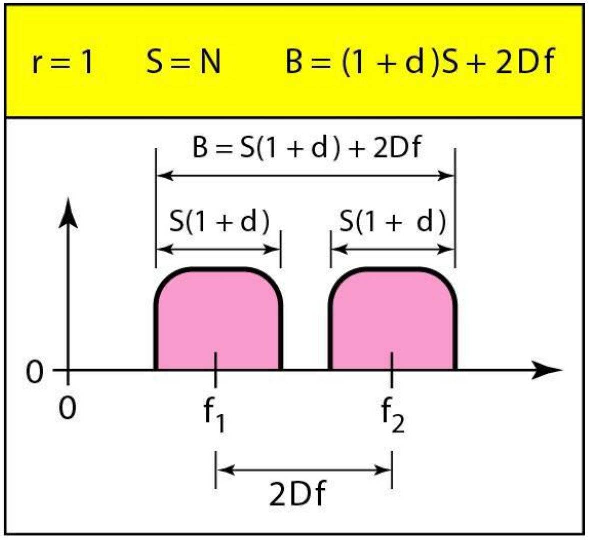
•

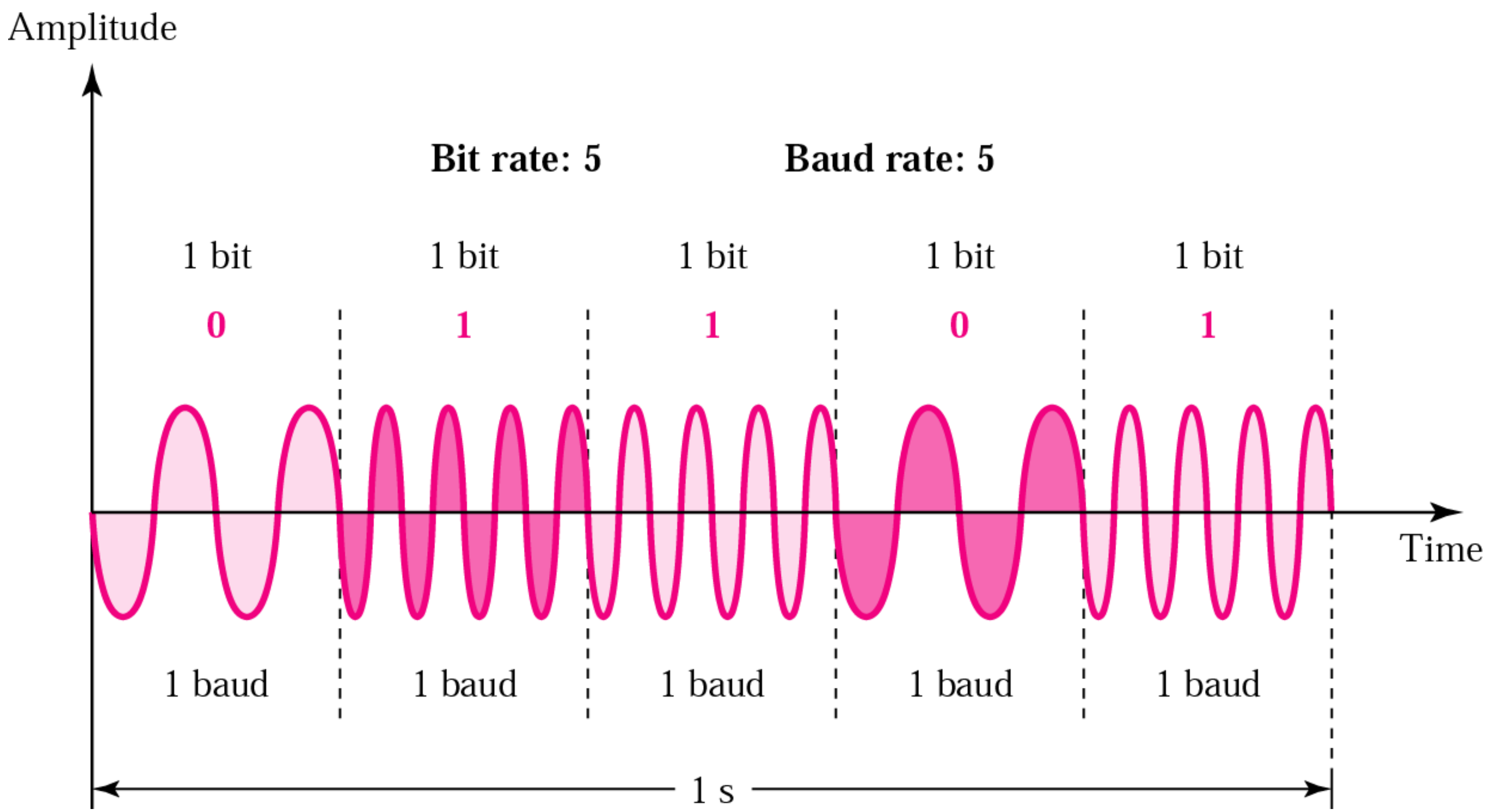
•*frequency* of the carrier signal is *varied* to represent data frequency of the modulated signal is constant for the duration of one signal element and changes for the next signal element if the data element changes amplitude Amplitude and Phase remain constant for all signal elements

Binary FSK

- implemented using ***two*** carrier frequencies:
- F_1,(space frequency) data elements 0
- f_2, (mark frequency) data elements 1
- both f_1 and f_2 are $2\Delta f$ apart






Amplitude
Bit rate: 5
Baud rate: 5
1 bit
1 bit
1 bit
1 bit
1 bit
0
1
1
0
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 s

Merits and Demerits

- Values represented by different frequencies (near carrier)
- Less susceptible to error than ASK
- Typically used up to 1200bps on voice grade lines
- High frequency radio
- Even higher frequency on LANs using co-ax
- Used in cordless and paging system

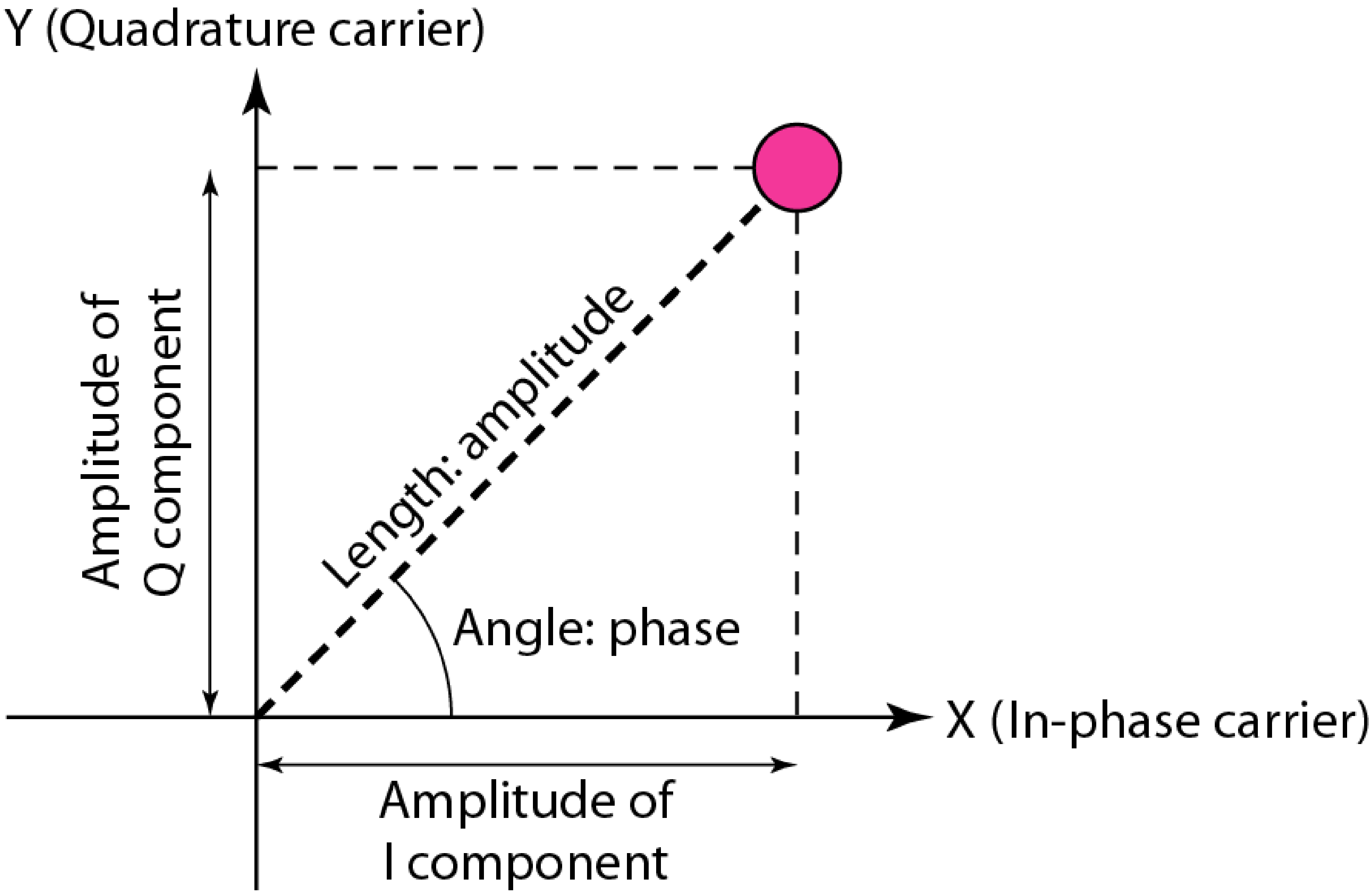
Phase Shift Keying

•*Phase* of the carrier signal is *varied* to represent two or more different signal elements amplitude and frequency remain constant

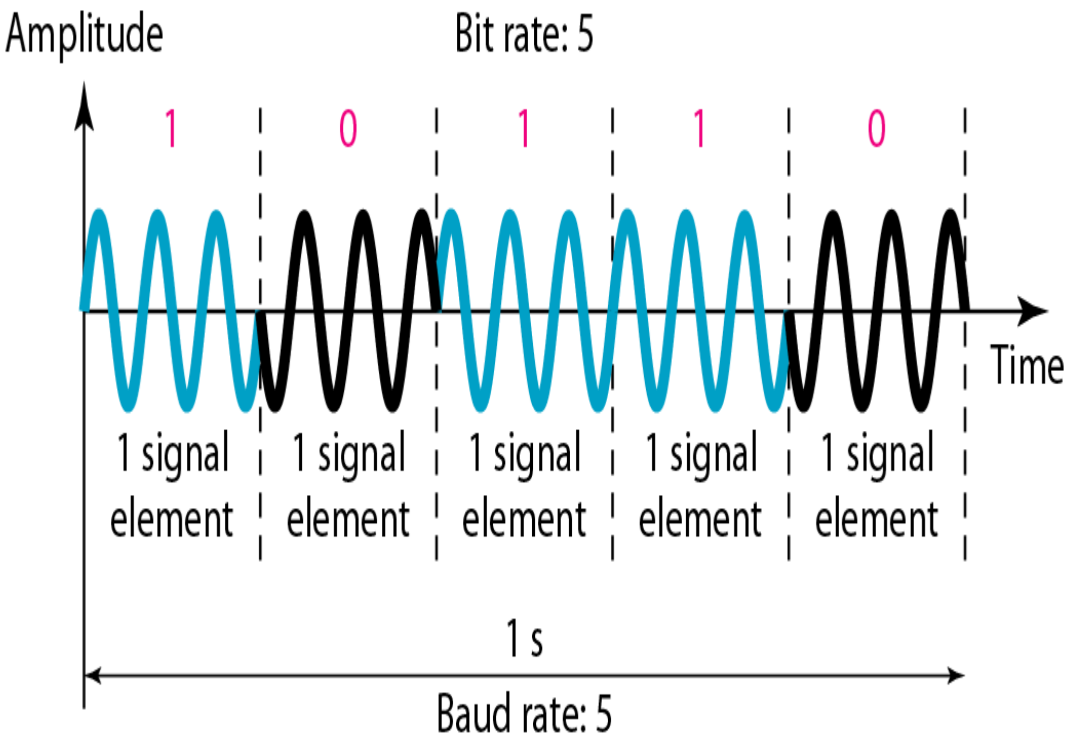
Constellation diagram

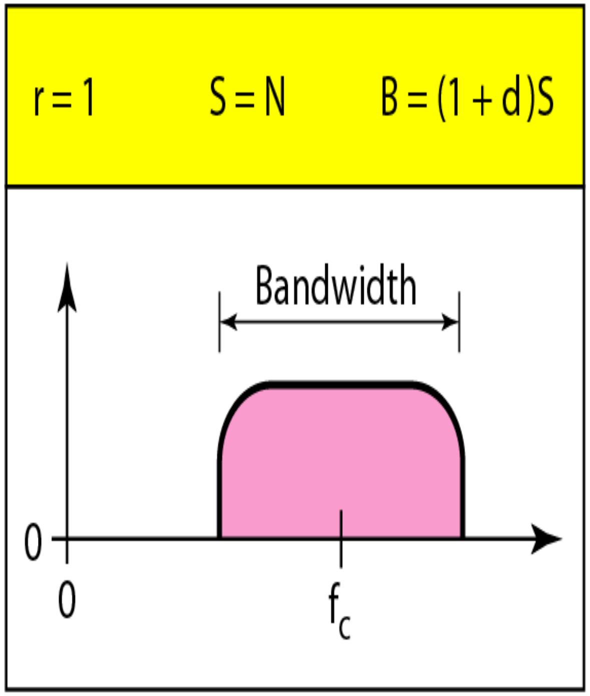
•

•helps defining the amplitude and phase of a signal element signal element type is represented as a dot the bit or combination of bits it carries is written next to the dot diagram has two axes X-axis → related to the in-phase carrier Y-axis → related to the quadrature carrier

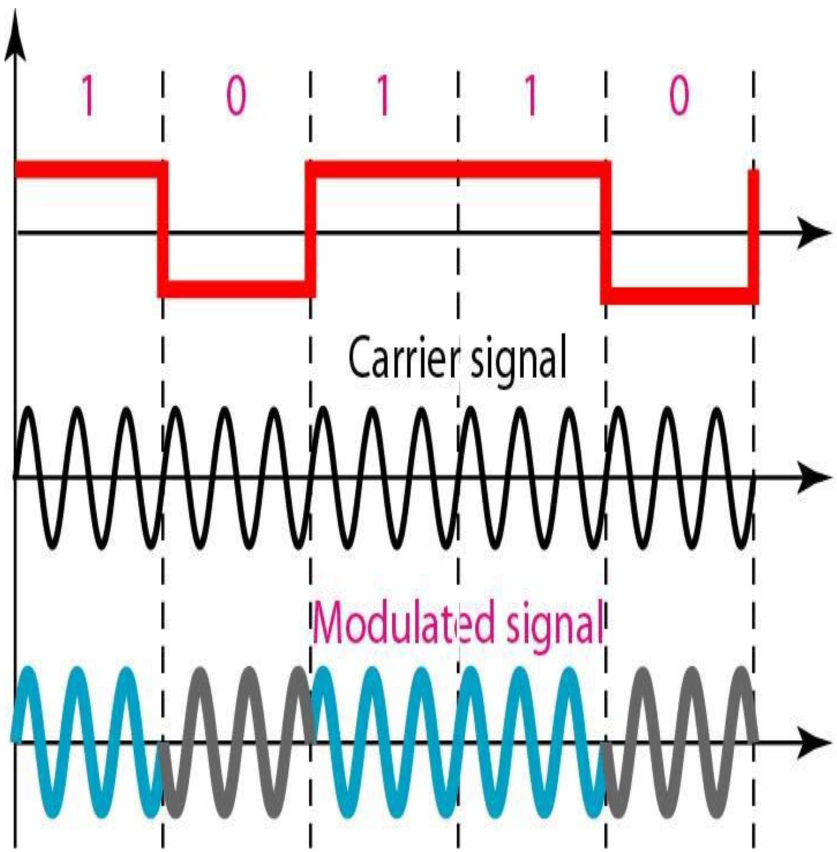

Y (Quadrature carrier)
Amplitude of
Q component
Length: amplitude
Angle: phase
X (In-phase carrier)
Amplitude of
I component

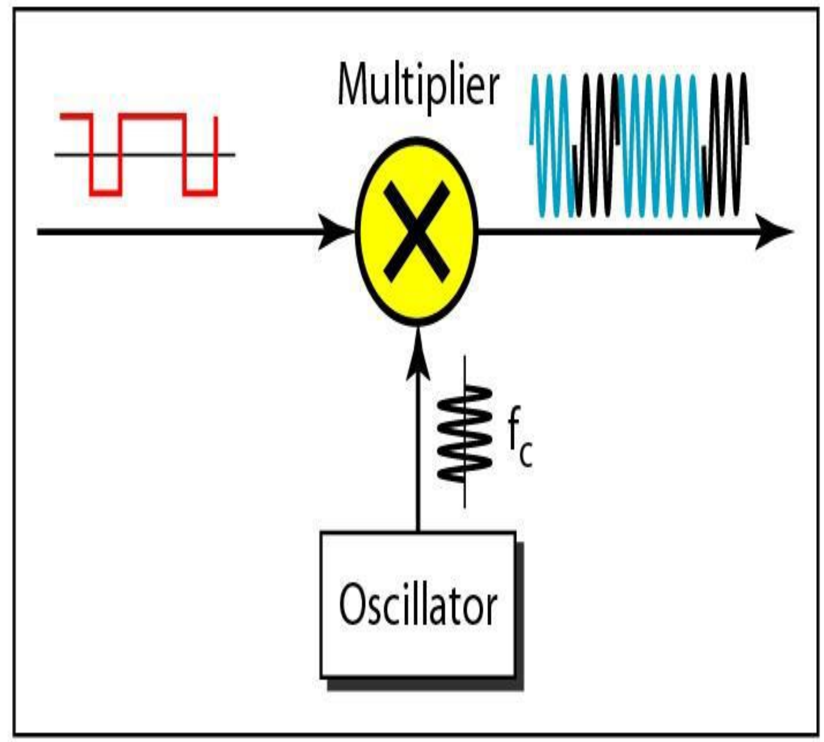
phase 0° → 1 bit ; phase 180° → 0 bit bandwidth requirement is the same as that of ASK





Binary PSK : implementation



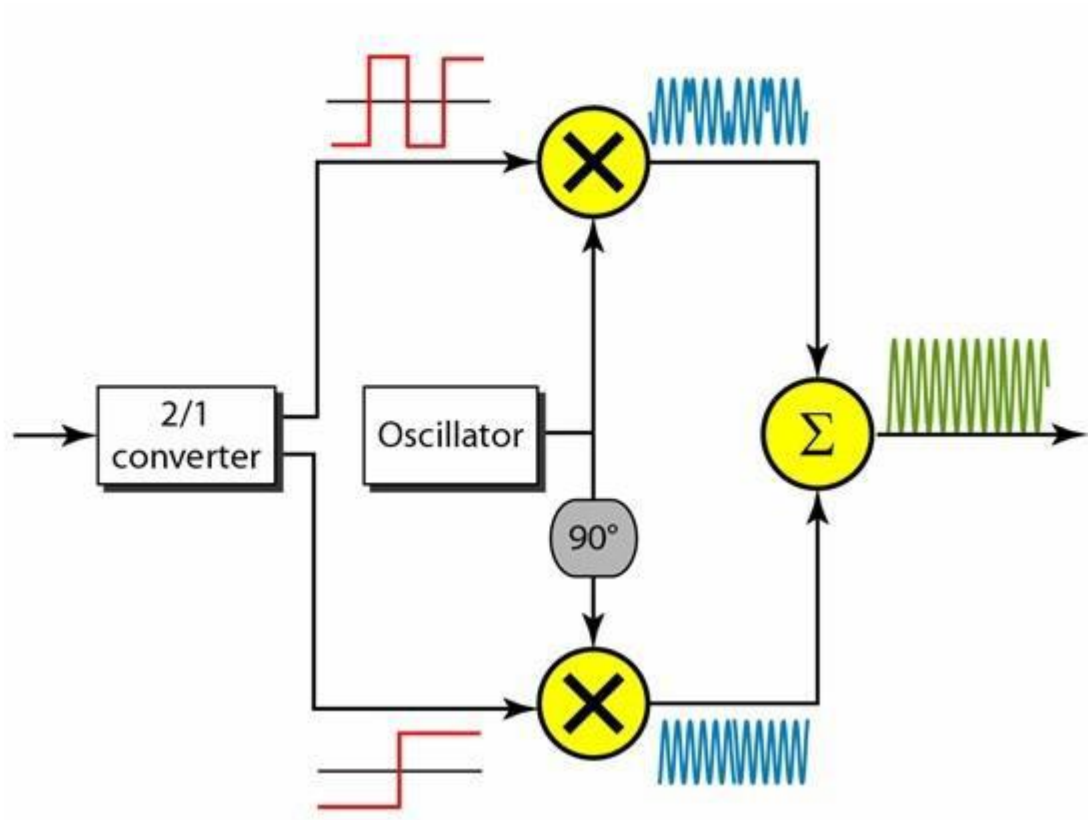


Quadrature PSK

•use of *two bits* at a time in each signal element → decrease of baud rate → reduction of required bandwidth

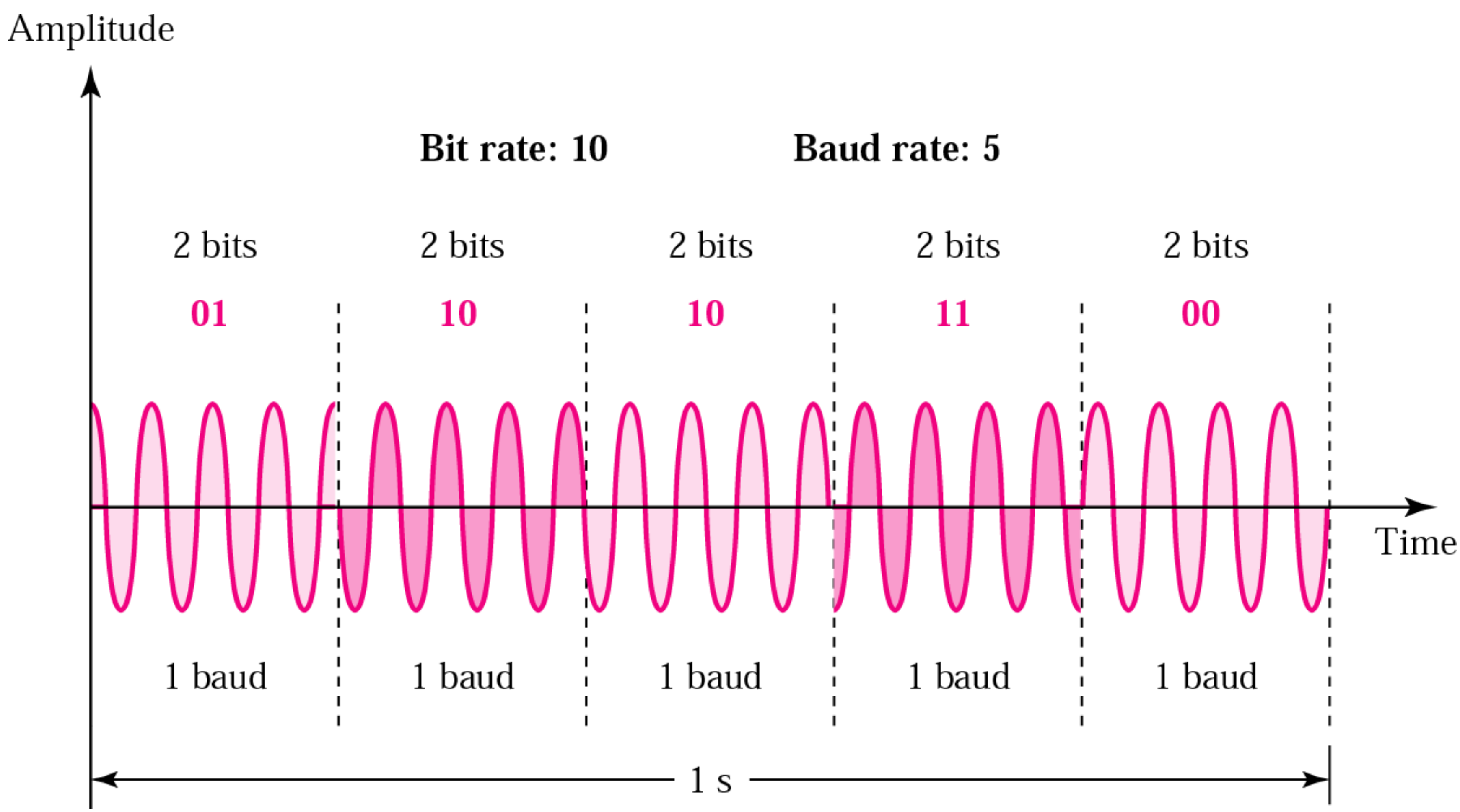
•uses two separate BPSK modulations : one in-phase and the other out-of-phase (quadrature)

Quadrature PSK: implementation



serial to parallel converter sends one bit to one modulator and the next bit to the other modulator

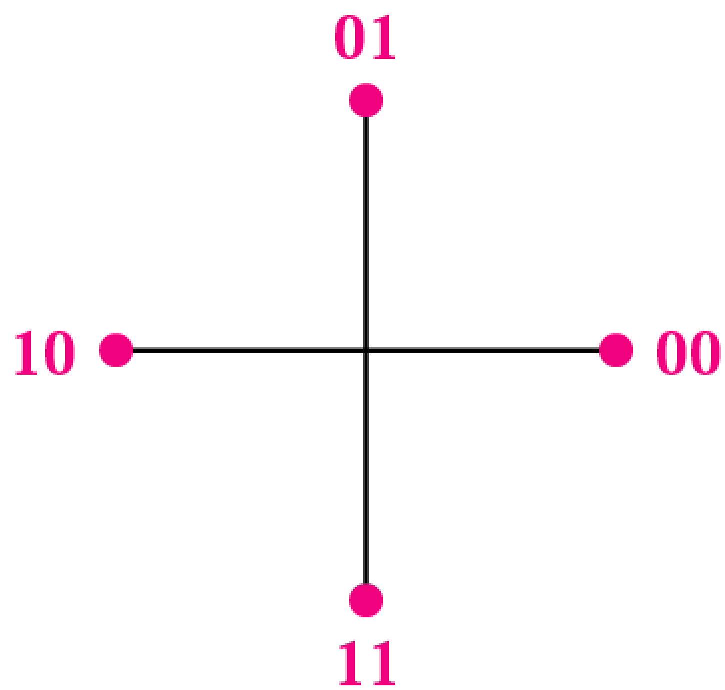
Quadrature PSK



•4-PSK characteristics

Dibit	Phase
00	0
01	90
10	180
11	270

Dibit
(2 bits)



Constellation diagram