e-Notes

Digital Communication

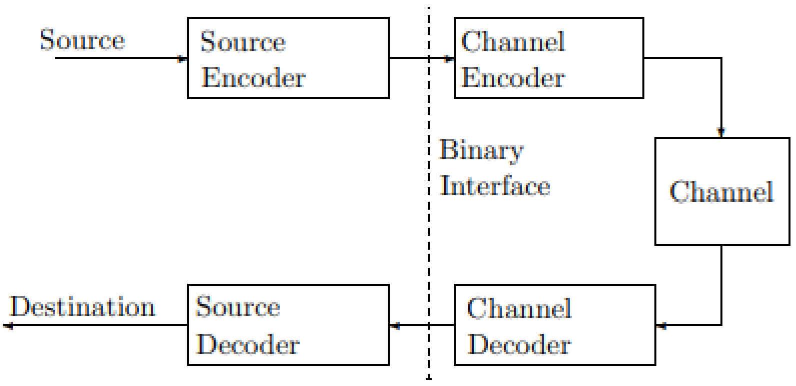
Chapter-1

Introduction

Communication has been one of the greatest needs of the human race. It is essential to form social unions, to educate the young, and to express a myriad of emotions and needs. Good communication is central to a civilized society.

1.1 Digital communication system

Digital communication systems are communication systems where the information propagates through the system in the form of symbols that are discrete or digital. It uses digital sequence as an interface between the source and the channel input (and likewise between the channel output and final destination).



Block diagram of digital communication system

1. Information Source and Input Transducer: The source of information can be analog or digital, e.g. analog: audio or video signal, digital: like teletype signal. In digital communication the signal produced by this source is converted into digital signal which consists of 1′s and 0′s. For this we need a source encoder.

2. Channel Encoder: The information sequence is passed through the channel encoder. The purpose of the channel encoder is to introduce, in controlled manner, some redundancy in the binary information sequence that can be used at the receiver to overcome the effects of noise and interference encountered in the transmission on the signal through the channel. For example take k bits of the information sequence and map that k bits to unique n bit sequence called code word.

3. **Channel:** The communication channel is the physical medium that is used for transmitting signals from transmitter to receiver. In wireless system, this channel consists of atmosphere, for traditional telephony, this channel is wired, there are optical channels, under water acoustic channels etc.We further discriminate this channels on the basis of their property and characteristics, like AWGN channel etc.

4. Channel Decoder: This sequence of numbers then passed through the channel decoder which attempts to reconstruct the original information sequence from the knowledge of the code used by the channel encoder and the redundancy contained in the received data

5. Source Encoder: In digital communication we convert the signal from source into digital signal as mentioned above. The point to remember is we should like to use as few binary digits as possible to represent the signal. In such a way this efficient representation of the source output results in little or no redundancy. This sequence of binary digits is called information sequence

6. Source Decoder: At the end, if an analog signal is desired then source decoder tries to decode the sequence from the knowledge of the encoding algorithm. And which results in the approximate replica of the input at the transmitter end.

Advantage of digital communication

1. Digital communication can be done over large distances though internet and other things.
2. Digital communication gives facilities like video conferencing which save a lot of time, money and effort.
3. It is easy to mix signals and data using digital techniques.
4. The digital communication is fast, easier and cheaper.
5. It can be tolerated the noise interference.
6. It can be detect and correct error easily because of channel coding.
7. Used in military application.
8. It has excellent processing techniques are available for digital signals such as data compression, image processing, channel coding and equalization etc.

Limitation of digital communication

1) Generally, more bandwidth is required than that for analog systems.

2) Synchronization is required.

3) High power consumption (Due to various stages of conversion).

4) Complex circuit, more sophisticated device making is also drawbacks of digital system.

5) Introduce sampling error

6) As square wave is more affected by noise, That's why while communicating through channel we send sine waves but while operating on device we use square pulses.

1.2 Comparison Analog and Digital Communication

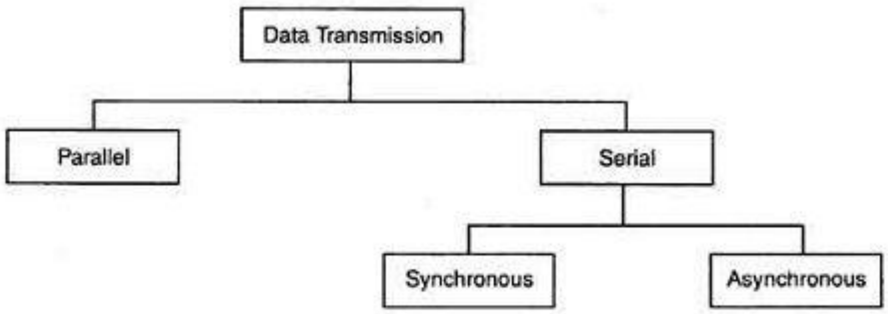
PARAMETERS	ANALOG COMMUNICATION	DIGITAL COMMUNICATION
Definiton	Analog Communication is the technology which uses Analog signal for the transmission of information.	Digital Communication is the technology which uses digital signal for the transmission of information.
Noise and Distortion	Get affected by Noise	Immune from Noise and Distortion
Error Probability	Error Probability is high due to parallax.	Error Probability is low
Hardware	Hardware is complicated and less flexible than digital system.	Hardware is flexible and less complicated than Analog system.
Cost	Low Cost	High Cost
Bandwidth Requirement	Low bandwidth requirement	High bandwidth Requirement
Power Requirement	High power is required	Low Power Requirement
Portability	Less Portable as the components are heavy	More portable due to compact equipments.
Modulation Used	Amplitude and Angle Modulation	Pulse coded Modulation or PCM, DPCM etc.
Representation of Signal	Analog signal can be represented by sine wave.	Digital signal is represented by square wave.
Signal Values	Consists of continuous values	Consists of discrete values
Example of Signal	Analog signal comprises of voice, sound etc.	Digital signals are used in computers

1.3 Principles of Digital Communication

When we enter data into the computer via keyboard, each keyed element is encoded by the electronics within the keyboard into an equivalent binary coded pattern, using one of the standard coding schemes that are used for the interchange of information. To represent all characters of the keyboard, a unique pattern of 7 or 8 bits in size is used. The use of 7 bits means that 128 different elements can be represented, while 8 bits can represent 256 elements. A similar procedure is followed at the receiver that decodes every received binary pattern into the corresponding character.

Data transmission refers to the movement of data in form of bits between two or more digital devices.This transfer of data takes place via some form of transmission media (for example, coaxial cable, fiber optics etc.)

Types of transmission



<u>Parallel transmission</u>

Within a computing or communication device, the distances between different subunits are too short. Thus, it is normal practice to transfer data between subunits using a separate wire to carry each bit of data. There are multiple wires connecting each sub-unit and data is exchanged using a parallel transfer mode. This mode of operation results in minimal delays in transferring each word.

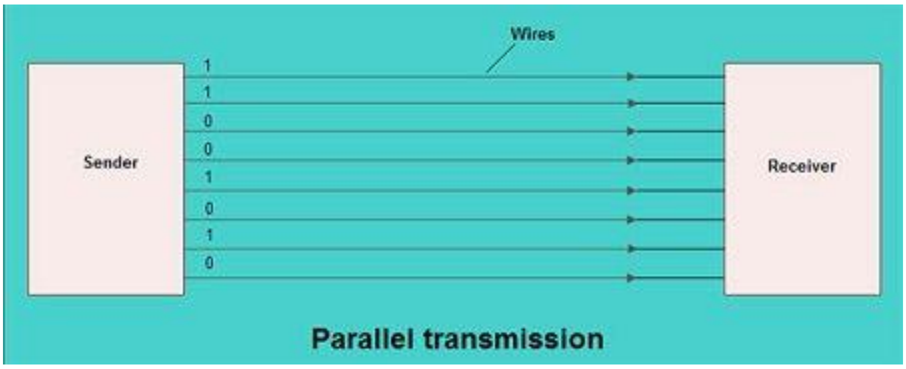
• In parallel transmission, all the bits of data are transmitted simultaneously on separate communication lines.

• In order to transmit n bits, n wires or lines are used. Thus each bit has its own line.

• All n bits of one group are transmitted with each clock pulse from one device to another i.e. multiple bits are sent with each clock pulse.

• Parallel transmission is used for short distance communication.

• As shown in the fig, eight separate wires are used to transmit 8 bit data from sender to receiver.



Advantage of parallel transmission

It is speedy way of transmitting data as multiple bits are transmitted simultaneously with a single clock pulse.

Disadvantage of parallel transmission

It is costly method of data transmission as it requires n lines to transmit n bits at the same time.

<u>Serial Transmission</u>

When transferring data between two physically separate devices, especially if the separation is more than a few kilometers, for reasons of cost, it is more economical to use a single pair of lines. Data is transmitted as a single bit at a time using a fixed time interval for each bit. This mode of transmission is known as bit-serial transmission.

• In serial transmission, the various bits of data are transmitted serially one after the other.

• It requires only one communication line rather than n lines to transmit data from sender to receiver.

• Thus all the bits of data are transmitted on single line in serial fashion.

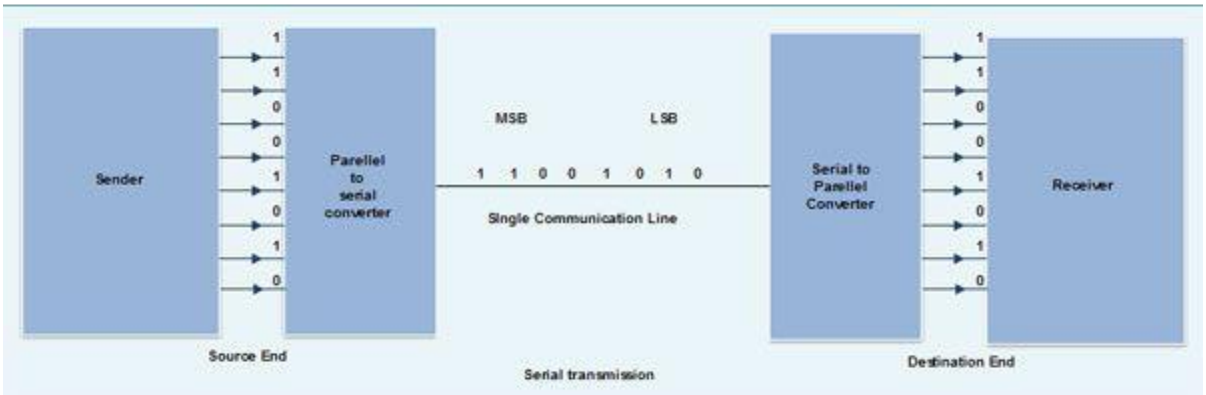
• In serial transmission, only single bit is sent with each clock pulse.

• As shown in fig., suppose an 8-bit data 11001010 is to be sent from source to destination. Then least significant bit (LSB) i,e. 0 will be transmitted first followed by other bits. The most significant bit (MSB) i.e. 1 will be transmitted in the end via single communication line.

• The internal circuitry of computer transmits data in parallel fashion. So in order to change this parallel data into serial data, conversion devices are used.

• These conversion devices convert the parallel data into serial data at the sender side so that it can be transmitted over single line.

• On receiver side, serial data received is again converted to parallel form so that the interval circuitry of computer can accept it



• Serial transmission is used for long distance communication.

Advantage of Serial transmission

Use of single communication line reduces the transmission line cost by the factor of n as compared to parallel transmission.

Disadvantages of Serial transmission

1. Use of conversion devices at source and destination end may lead to increase in overall transmission cost.

2. This method is slower as compared to parallel transmission as bits are transmitted serially one after the other.

Types of Serial Transmission

There are two types of serial transmission-synchronous and asynchronous both these transmissions use 'Bit synchronization'

Bit Synchronization is a function that is required to determine when the beginning and end of the data transmission occurs.

Bit synchronization helps the receiving computer to know when data begin and end during a transmission. Therefore bit synchronization provides timing control.

A) Asynchronous Transmission

• Asynchronous transmission sends only one character at a time where a character is either a letter of the alphabet or number or control character i.e. it sends one byte of data at a time.

• Bit synchronization between two devices is made possible using start bit and stop bit.

• Start bit indicates the beginning of data i.e. alerts the receiver to the arrival of new group of bits. A start bit usually 0 is added to the beginning of each byte.

• Stop bit indicates the end of data i.e. to let the receiver know that byte is finished, one or more additional bits are appended to the end of the byte. These bits, usually 1s are called stop bits.

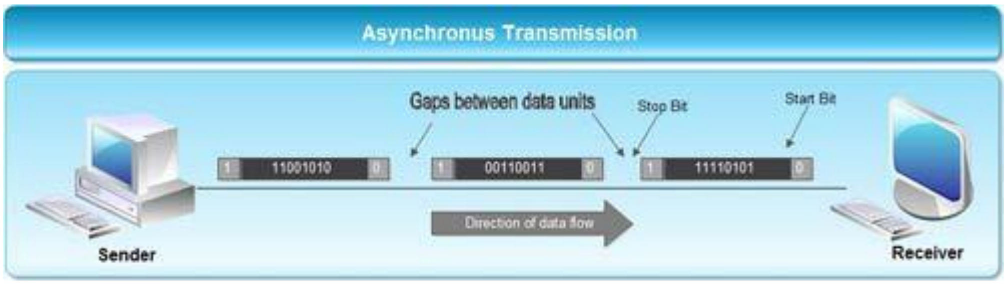
• Addition of start and stop increase the number of data bits. Hence more bandwidth is consumed in asynchronous transmission.

• There is idle time between the transmissions of different data bytes. This idle time is also known as Gap

• The gap or idle time can be of varying intervals. This mechanism is called Asynchronous, because at byte level sender and receiver need not to be synchronized. But within each byte, receiver must be synchronized with the incoming bit stream.

Application of Asynchronous Transmission

1. Asynchronous transmission is well suited for keyboard type-terminals and paper tape devices. The advantage of this method is that it does not require any local storage at the terminal or the computer as transmission takes place character by character.



2. Asynchronous transmission is best suited to Internet traffic in which information is transmitted in short bursts. This type of transmission is used by modems.

Advantages of Asynchronous transmission

1. This method of data transmission is cheaper in cost as compared to synchronous e.g. If lines are short, asynchronous transmission is better, because line cost would be low and idle time will not be expensive.

2. In this approach each individual character is complete in itself, therefore if character is corrupted during transmission, its successor and predecessor character will not be affected.

3. It is possible to transmit signals from sources having different bit rates.

4. The transmission can start as soon as data byte to be transmitted becomes available.

5. Moreover, this mode of data transmission in easy to implement.

Disadvantages of asynchronous transmission

1. This method is less efficient and slower than synchronous transmission due to the overhead of extra bits and insertion of gaps into bit stream.

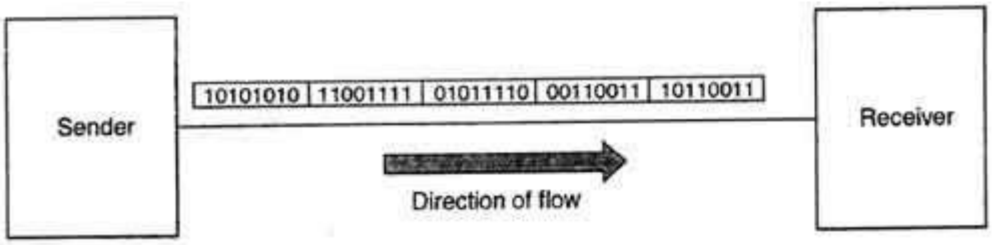
2. Successful transmission inevitably depends on the recognition of the start bits. These bits can be missed or corrupted.

B) Synchronous Transmission

• Synchronous transmission does not use start and stop bits.

• In this method bit stream is combined into longer frames that may contain multiple bytes.

• There is no gap between the various bytes in the data stream.



Synchronous Transmission

• In the absence of start & stop bits, bit synchronization is established between sender & receiver by 'timing' the transmission of each bit.

• Since the various bytes are placed on the link without any gap, it is the responsibility of receiver to separate the bit stream into bytes so as to reconstruct the original information.

• In order to receive the data error free, the receiver and sender operates at the same clock frequency.

Application of Synchronous transmission

• Synchronous transmission is used for high speed communication between computers.

Advantage of Synchronous transmission

1. This method is faster as compared to asynchronous as there are no extra bits (start bit & stop bit) and also there is no gap between the individual data bytes.

Disadvantages of Synchronous transmission

1. It is costly as compared to asynchronous method. It requires local buffer storage at the two ends of line to assemble blocks and it also requires accurately synchronized clocks at both ends. This leads to increase in the cost.

2. The sender and receiver have to operate at the same clock frequency. This requires proper synchronization which makes the system complicated.

Comparison between Serial and Parallel transmission

Sr. No.	Factor	Serial	Parallel
1.	Number of bits transmitted at one clock pulse	One bit	*n* bits
2.	No. of lines required to transmit *n* bits	One line	*n* lines
3.	Speed of data transfer	Slow	Fast
4.	Cost of transmission	Low as one line is required	Higher as *n* lines are required.
5.	Application	Long distance communication between two computers	Short distance communication. like computer to printer.

Comparison between Asynchronous and Synchronous.

Sr. No.	Factor	Asynchronous	Synchronus
1.	Data send at one time	Usually 1 byte	Multiple bytes
2.	Start and Stop bit	Used	Not used
3.	Gap between Data units	Present	Not present
4.	Data transmission speed	Slow	Fast
5.	Cost	Low	High

Chapter-2

Sampling Thereom and its Basic Concept

2.1 WHAT IS SAMPLING?

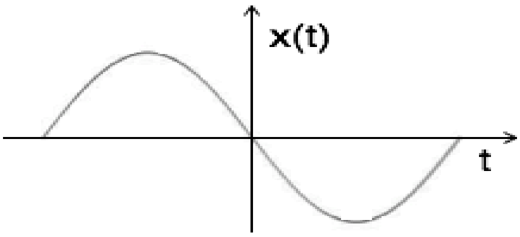
Sampling may be defined as the procedure in which a sample is selected from an individual or a group of people of certain kind for research purpose. In sampling, the population is divided into a number of parts called sampling units.

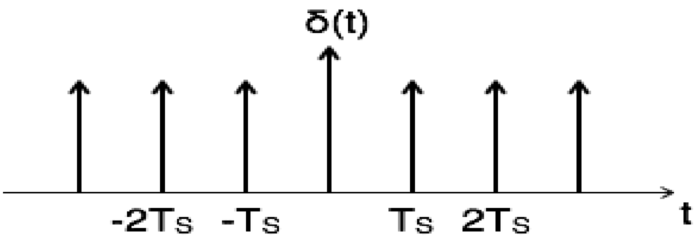
2.2 SAMPLING THEOREM

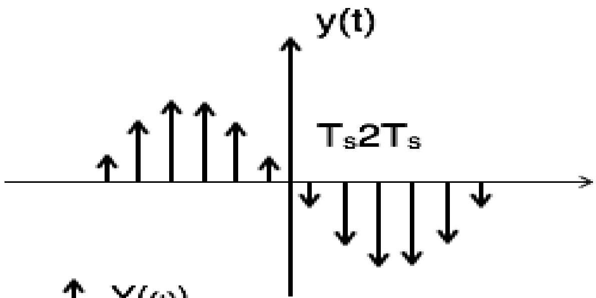
Statement: A continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal. i. e. $fs \geq 2fm$.$fs \geq 2fm$.

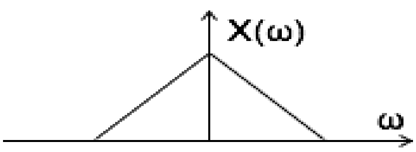
Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to f_m Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

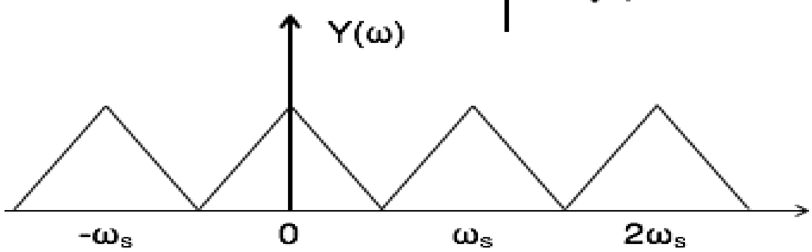
Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period T_s. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) in the following diagrams:











ADVANTAGES OF SAMPLING

1. Low cost of sampling
2. Less time consuming in sampling
3. Scope of sampling is high
4. Accuracy of data is high

DISADVANTAGE OF SAMPLING

1. Chances of bias
2. Difficulties in selecting truly a representative sample
3. Need for subject specific knowledge
4. changeability of sampling units
5. impossibility of sampling

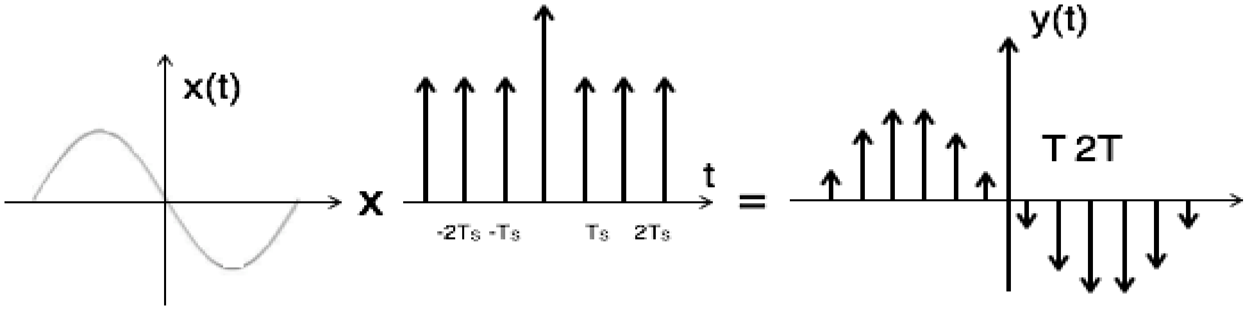
TYPES OF SAMPLING

There are three types of sampling techniques:

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

2.3 Impulse Sampling

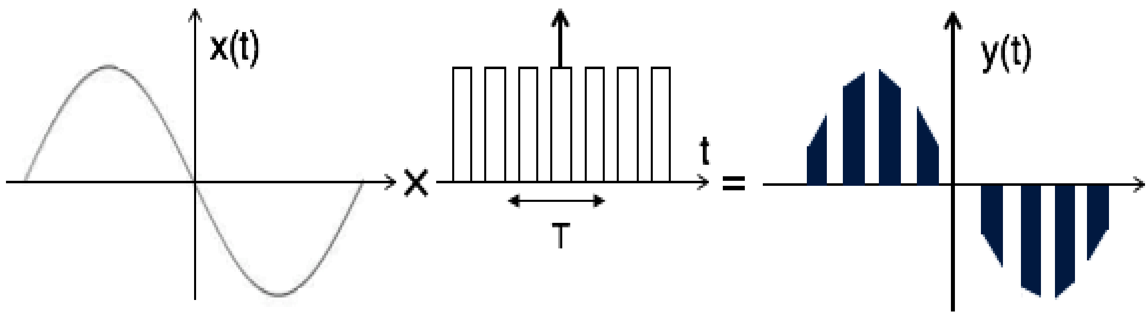
Impulse sampling can be performed by multiplying input signal x(t) with impulse train of period 'T'. Here, the amplitude of impulse changes with respect to amplitude of input signal x(t). The output of sampler is given by



This is called ideal sampling or impulse sampling. You cannot use this practically because pulse width cannot be zero and the generation of impulse train is not possible practically.

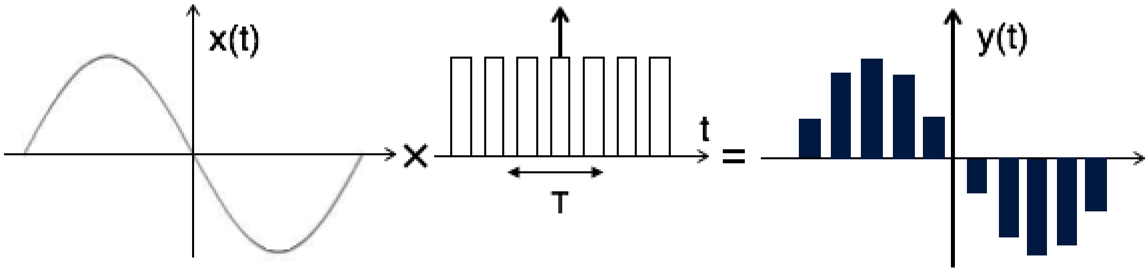
2.4 Natural Sampling

Natural sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period T. i.e. you multiply input signal x(t) to pulse train as shown below

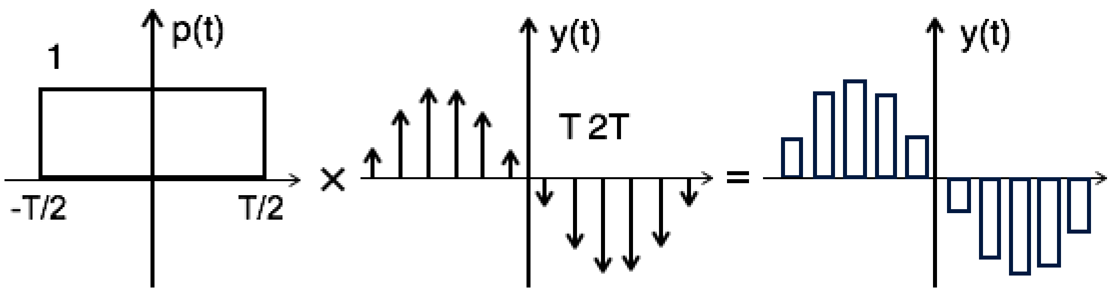


2.5 Flat Top Sampling

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



Theoretically, the sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal say $y_\delta(t)$ as shown in the diagram:i.e. $y(t)=p(t)\times y\delta(t)$......(1)



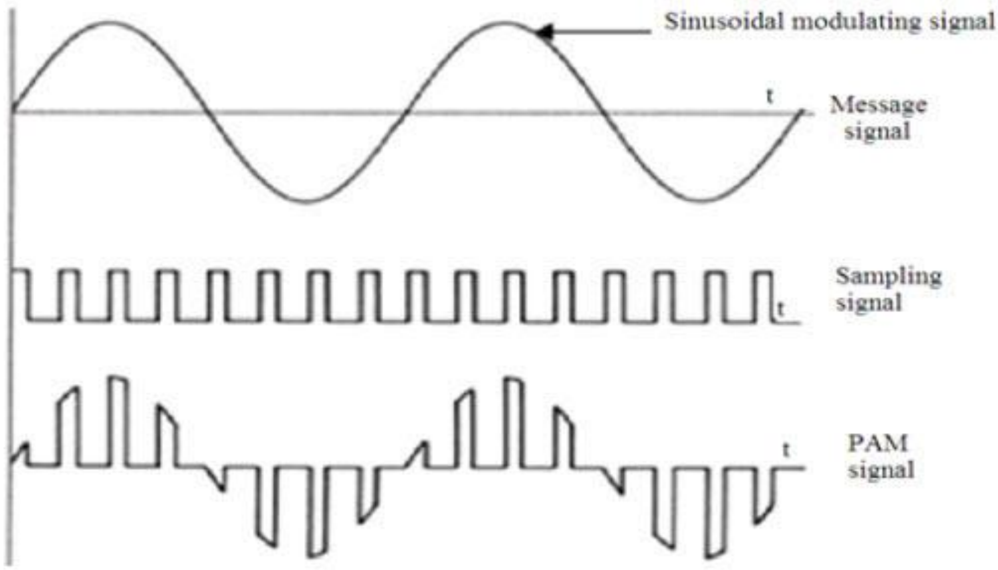
Nyquist Rate

It is the minimum sampling rate at which signal can be converted into samples and can be recovered back without distortion.

Nyquist rate $f_N = 2f_m$ hz

2.6 Pulse Amplitude Modulation

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



Applications of PAM

- It is used in Ethernet communication.
- It is used in many micro-controllers for generating the control signals.
- It is used in Photo-biology.
- It is used as an electronic driver for LED lighting.

Advantages

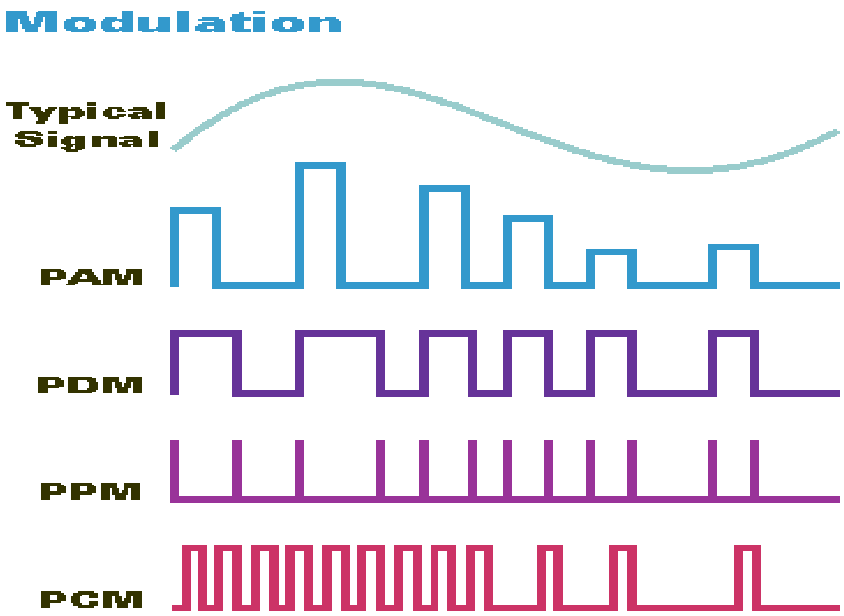
- It is the simple process for both modulation and demodulation.
- Transmitter and receiver circuits are simple and easy to construct.
- PAM can generate other pulse modulation signals and can carry the message at the same time.

Disadvantages

- Bandwidth should be large for transmission PAM modulation.
- Noise will be great.
- Pulse amplitude signal varies so power required for transmission will be more.

2.7 PULSE POSITION MODULATION

Pulse position modulation works by sending electrical, electromagnetic, or optical pulses to a computer or other device in order to communicate simple data. It requires both devices to be synchronized to the same clock so that when a series of pulses is sent, the device decodes the information based on when the pulses were broadcasted. Alternately, another form of pulse position modulation known as differential pulse position modulation, allows all signals to be encoded based on the difference between broadcast times. This means that a receiving device only has to observe the difference in arrival times in order to decode a transmission.



Applications

Pulse position modulation has many purposes, especially in RF (Radio Frequency) communications. For example, pulse position modulation is used in remote controlled aircraft, cars, boats, and other vehicles and is responsible for conveying a transmitter's controls to a receiver. Each pulse's position may describe an analogue controller's physical direction, while the number of pulses may describe the number of possible commands that the device may receive.

Advantages

Pulse position modulation conveys simple commands that other forms of signal modulation are either simply not made for or are too complex to use in certain situations. Because pulse position modulation only communicates simple commands from a transmitter to a receiver, it is often used in lightweight applications due to its low system requirements.

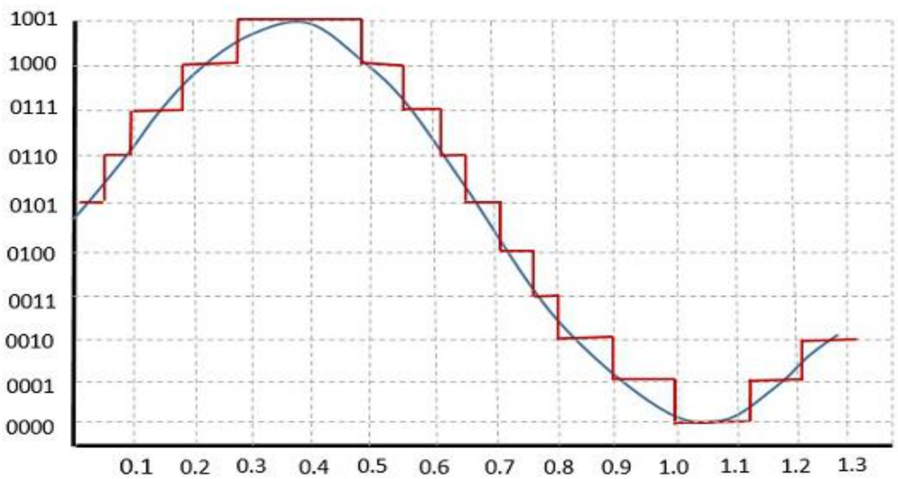
Disadvantages

Pulse position modulation requires that both devices are synchronized or differential pulse position modulation is used. Also, pulse position modulation is highly sensitive to multi-pathway interference, such as echoing, that can disrupt a transmission by altering the difference in arrival times of each signal.

2.8 QUANTIZATION

Quantization, in mathematics and digital signal processing, is the process of mapping input values from a large set (often a continuous set) to output values in a (countable) smaller set, often with a finite number of elements. Rounding and truncation are typical examples of quantization processes. Quantization is involved to some degree in nearly all digital signal processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms.

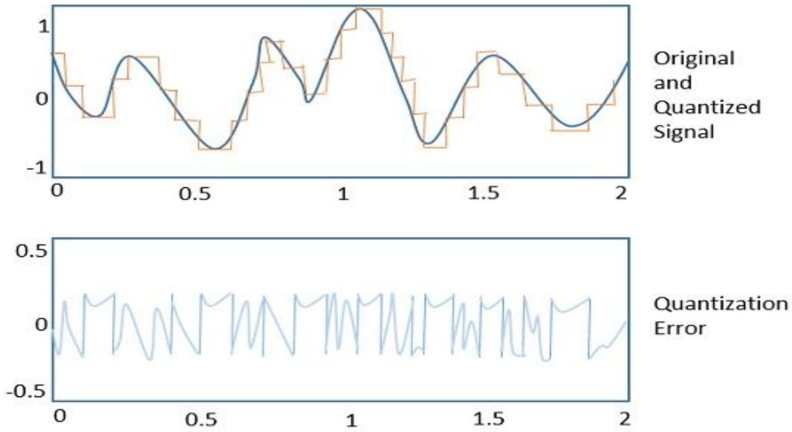
The difference between an input value and its quantized value (such as round-off error) is referred to as quantization error. A device or algorithmic function that performs quantization is called a quantizer. An analog-to-digital converter is an example of a quantizer.



2.9 Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values. The difference between an input value and its quantized value is called a **Quantization Error**. A **Quantizer** is a logarithmic function that performs Quantization (rounding off the value). An analog-to-digital converter (**ADC**) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.



Quantization Noise

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

2.10 Companding in PCM

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

There are two types of Companding techniques. They are −

A-law Companding Technique

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

µ-law Companding Technique

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

µ-law is used in North America and Japan.

A-Law Companding

A-law is the CCITT recommended companding standard used across Europe. Limiting the linear sample values to 12 magnitude bits, the A-law compression is defined by Equation 1, where A is the compression parameter (A=87.7 in Europe), and x is the normalized integer to be compressed.

$$F(x) = \begin{cases} \dfrac{A*|x|}{1+\ln(A)} & 0 \le |x| < \dfrac{1}{A} \\[2ex] \dfrac{\operatorname{sgn}(x)*(1+\ln(A|x|))}{1+\ln(A)} & \dfrac{1}{A} \le |x| \le 1 \end{cases}$$

A-Law Equation

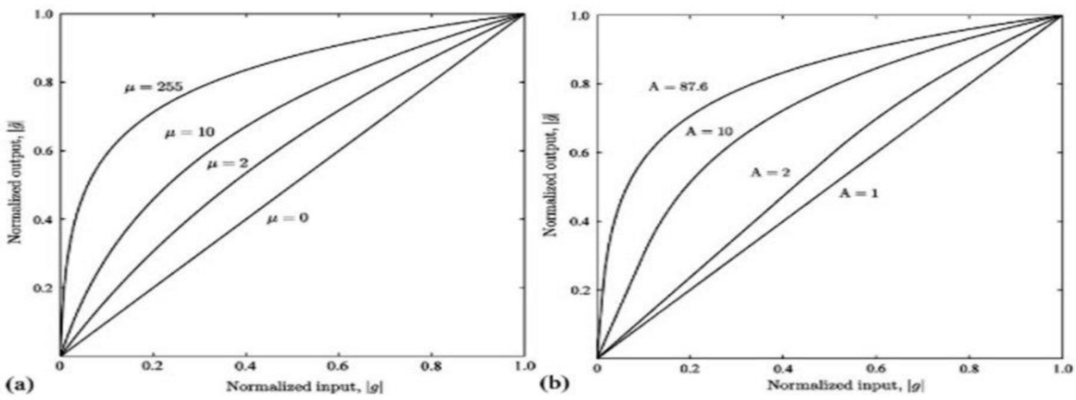
µ-law companding

The United States and Japan use µ-law companding. Limiting the linear sample values to 13 magnitude bits, the µ-law compression is defined by Equation 2, where m is the compression parameter (m =255 in the U.S. and Japan) and x is the normalized integer to be compressed.

The encoding and decoding process for µ-law is similar to that of A-law. There are, however, a few notable differences: 1) µ-law encoders typically operate on linear 13-bit magnitude data, as opposed to 12-bit magnitude data with A-law, 2) before chord determination a bias value of 33 is added to the absolute value of the linear input data to simplify the chord and step calculations, 3) the definition of the sign bit is reversed, and 4) the inversion pattern is applied to all bits in the 8bit code. Table 3 illustrates a µ-law encoding table. The sign bit of the linear input data is omitted from the table. The sign bit (S) for the 8-bit code is set to 1 if the input sample is positive, and is set to 0 if the input sample is negative.

$$F(x) = \frac{\operatorname{sgn}(x)*\ln(1+\mu|x|)}{\ln(1+\mu)} \quad 0 \le |x| \le 1$$

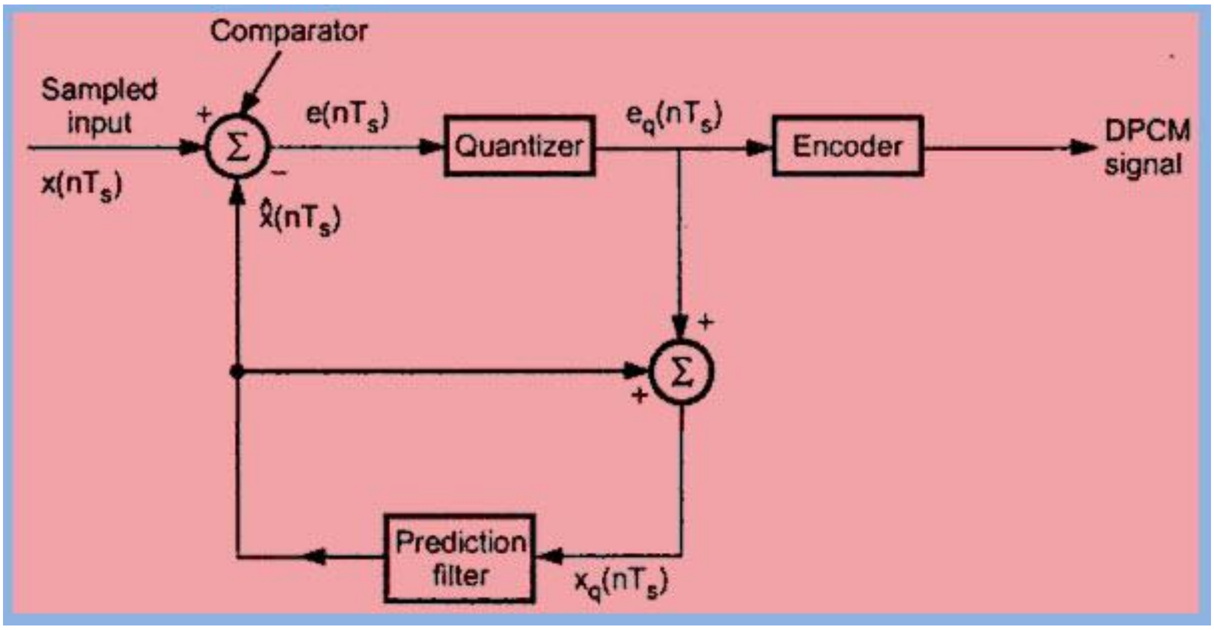
u Law Equation



2.11 Differential Pulse Code Modulation

If the redundancy is reduced, then the overall bit rate will decrease and the number of bits required to transmit one sample will also reduce. This type of digital pulse modulation technique is called differential pulse code modulation. The DPCM works on the principle of prediction. The value of the present sample is predicted from the previous samples. The prediction may not be exact, but it is very close to the actual sample value.

Differential Pulse Code Modulation Transmitter

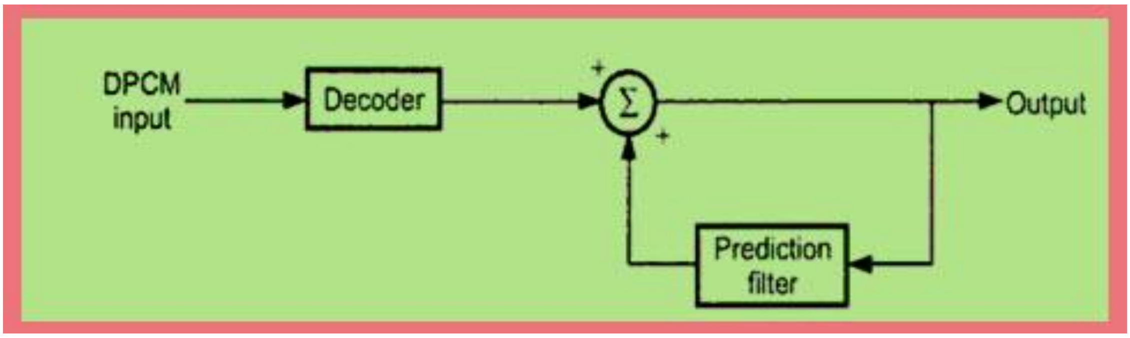


The sampled signal is denoted by x(nTs) and the predicted signal is indicated by x^(nTs). The comparator finds out the difference between the actual sample value x(nTs) and the predicted value x^(nTs). This is called signal error and it is denoted as e(nTs) e(nTs)= x(nTs)- x^(nTs) …….(1)

Here the predicted value x^(nTs) is produced by using a prediction filter(signal processing filter). The quantizer output signal eq(nTs) and the previous prediction is added and given as input to the prediction filter, this signal is denoted by xq(nTs). This makes the prediction closer to the actually sampled signal. The quantized error signal eq(nTs) is very small and can be encoded by using a small number of bits. Thus the number of bits per sample is reduced in DPCM.

Differential Pulse Code Modulation Receiver

In order to reconstruct the received digital signal, the DPCM receiver (shown in below figure) consists of a decoder and prediction filter. In the absenteeism of noise, the encoded receiver input will be the same as the encoded transmitter output.



As we discussed above, the predictor undertakes a value, based on the previous outputs. The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtain a better output. That means here first of all the decoder will reconstruct the quantized form of original signal. Therefore the signal at the receiver differs from the actual signal by quantization error q(nTs), which is introduced permanently in the reconstructed signal.

Applications of DPCM

The DPCM technique mainly used Speech, image and audio signal compression. The DPCM conducted on signals with the correlation between successive samples leads to good compression ratios. In images, there is a correlation between the neighbouring pixels, in video signals, the correlation is between the same pixels in consecutive frames and inside frames (which is same as correlation inside the image).

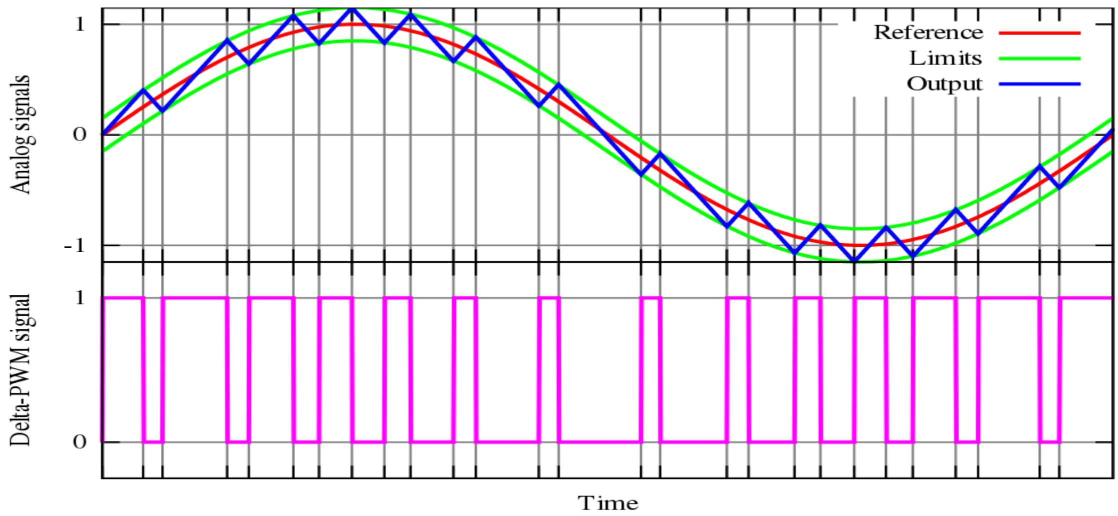
This method is suitable for real Time applications. To understand the efficiency of this method of medical compression and real-time application of medical imaging such as telemedicine and online diagnosis. Therefore, it can be efficient for lossless compression and implementation for lossless or near-lossless medical image compression.

2.12 Delta modulation

A delta modulation (DM or Δ-modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance. DM is the simplest form of differential pulse-code modulation (DPCM) where the difference between successive samples are encoded into n-bit data streams. In delta modulation, the transmitted data are reduced to a 1-bit data stream. Its main features are:

- The analog signal is approximated with a series of segments.
- Each segment of the approximated signal is compared to the preceding bits and the successive bits are determined by this comparison.
- Only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.

To achieve high signal-to-noise ratio, delta modulation must use oversampling techniques, that is, the analog signal is sampled at a rate several times higher than the Nyquist rate



2.13 Frequency-hopping spread spectrum

Frequency-hopping spread spectrum (FHSS) is a method of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudorandom sequence known to both transmitter and receiver. It is used as a multiple access method in the code division multiple access (CDMA) scheme frequency-hopping code division multiple access (FH-CDMA).

Each available frequency band is divided into sub-frequencies. Signals rapidly change ("hop") among these in a predetermined order. Interference at a specific frequency will only affect the signal during that short interval. FHSS can, however, cause interference with adjacent direct-sequence spread spectrum (DSSS) systems.

Adaptive frequency-hopping spread spectrum (**AFH**), a specific type of FHSS, is used in Bluetooth wireless data transfer.

Chapter-3

Digital Modulation Techniques

Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog ones. There are many types of digital modulation techniques and we can even use a combination of these techniques as well. Digital-to-Analog signals is the next conversion we will discuss in this chapter. These techniques are also called as **Digital Modulation techniques**.

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

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

- ASK – Amplitude Shift Keying
- FSK – Frequency Shift Keying
- PSK – Phase Shift Keying

ASK – Amplitude Shift Keying

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

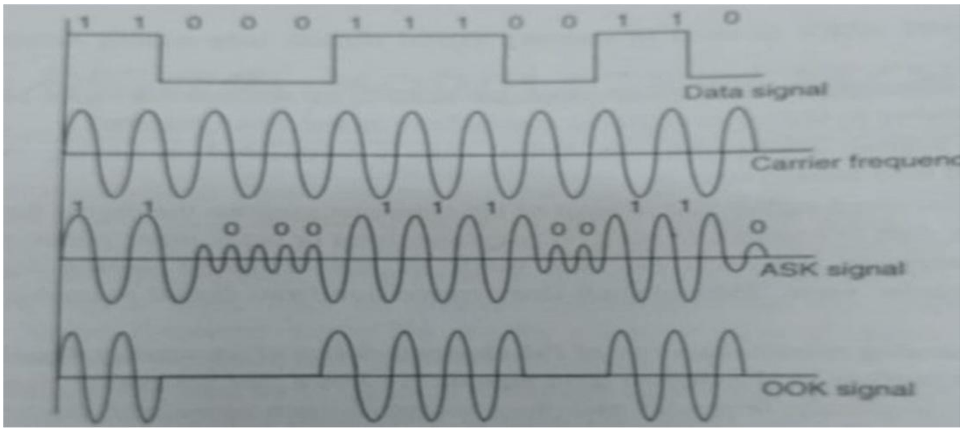
FSK – Frequency Shift Keying

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

PSK – Phase Shift Keying

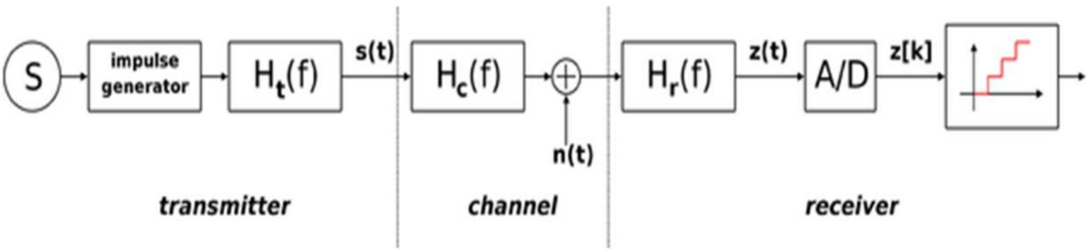
The phase of the output signal gets shifted depending upon the input. These are mainly of two types, namely Binary Phase Shift Keying (BPSK) and Quadrature Phase Shift Keying (QPSK), according to the number of phase shifts. The other one is Differential Phase Shift Keying (DPSK) which changes the phase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

3.1 Amplitude-shift keying (ASK) is a form of amplitude modulation that represents digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by transmitting a fixed-amplitude carrier wave and fixed frequency for a bit duration of T seconds. If the signal value is 1 then the carrier signal will be transmitted; otherwise, a signal value of 0 will be transmitted. The simplest and most common form of ASK operates as a switch, using the presence of a carrier wave to indicate a binary one and its absence to indicate a binary zero. This type of modulation is called on-off keying (OOK), and is used at radio frequencies to transmit Morse code (referred to as continuous wave operation).



Waveform of ASK

Block Diagram of ASK



ASK diagram

ASK system can be divided into three blocks. The first one represents the transmitter, the second one is a linear model of the effects of the channel, the third one shows the structure of the receiver. The following notation is used:

- h_t(f) is the carrier signal for the transmission
- h_C(f) is the impulse response of the channel
- n(t) is the noise introduced by the channel
- h_r(f) is the filter at the receiver
- L is the number of levels that are used for transmission
- T_S is the time between the generation of two symbols

Different symbols are represented with different voltages. If the maximum allowed value for the voltage is A, then all the possible values are in the range [−A, A] and they are given by:

$$v_i = \frac{2A}{L-1}i - A; \quad i = 0, 1, \ldots, L-1$$

the difference between one voltage and the other is:

$$\Delta = \frac{2A}{L-1}$$

Considering the picture, the symbols v[n] are generated randomly by the source S, then the impulse generator creates impulses with an area of v[n]. These impulses are sent to the filter ht to be sent through the channel. In other words, for each symbol a different carrier wave is sent with the relative amplitude.

Out of the transmitter, the signal s(t) can be expressed in the form:

$$s(t) = \sum_{n=-\infty}^{\infty} v[n] \cdot h_t(t - nT_s)$$

In the receiver, after the filtering through hr (t) the signal is:

$$z(t) = n_r(t) + \sum_{n=-\infty}^{\infty} v[n] \cdot g(t - nT_s)$$

$$n_r(t) = n(t) * h_r(t)$$
$$g(t) = h_t(t) * h_c(t) * h_r(t)$$

where * indicates the convolution between two signals. After the A/D conversion the signal z[k] can be expressed in the form:

$$z[k] = n_r[k] + v[k]g[0] + \sum_{n \neq k} v[n]g[k-n]$$

In this relationship, the second term represents the symbol to be extracted. The others are unwanted: the first one is the effect of noise, the third one is due to the intersymbol interference.

If the filters are chosen so that g(t) will satisfy the Nyquist ISI criterion, then there will be no intersymbol interference and the value of the sum will be zero, so:

$$z[k] = n_r[k] + v[k]g[0]$$

the transmission will be affected only by noise.

Advantages and Disadvantages-

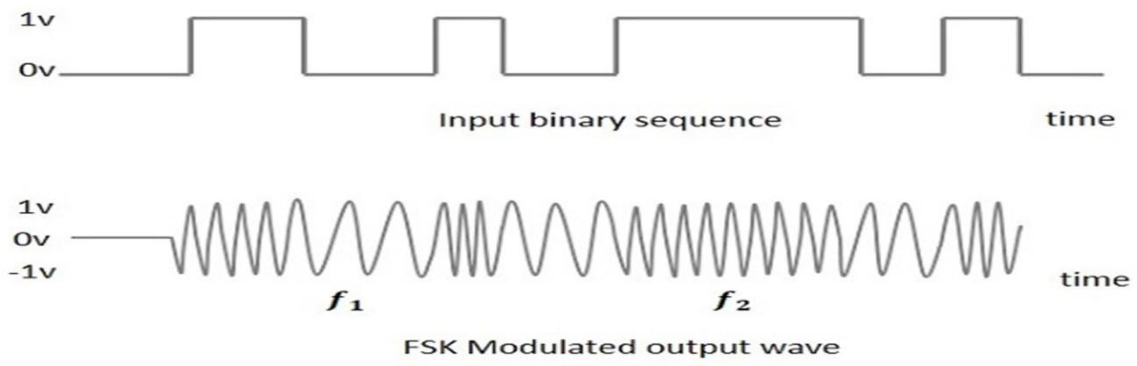
ASK is linear and sensitive to atmospheric noise, distortion, propagation conditions on different routes in PSTN etc. But still ASK modulation and demodulation, both are relatively inexpensive.

Application-

1. ASK is commonly used to transmit digital data over optical fiber.
2. ASK is also used for transmitting Morse code at radio frequency.
3. It is used extensively for commercial terrestrial application

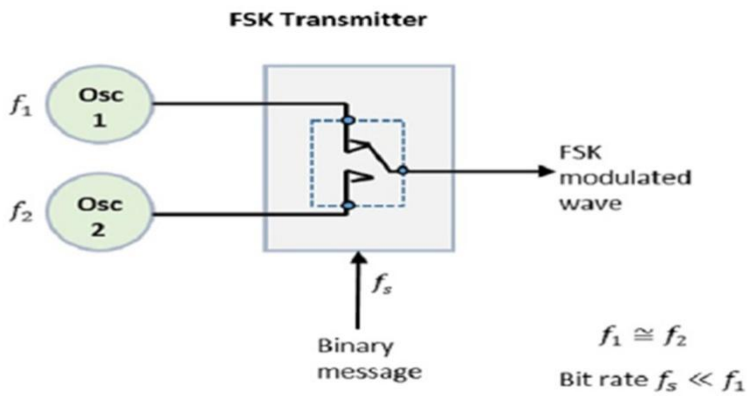
3.2 Frequency Shift Keying (FSK) is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. FSK is a scheme of frequency modulation. The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input. The binary **1s** and **0s** are called Mark and Space frequencies.

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



FSK Modulator

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



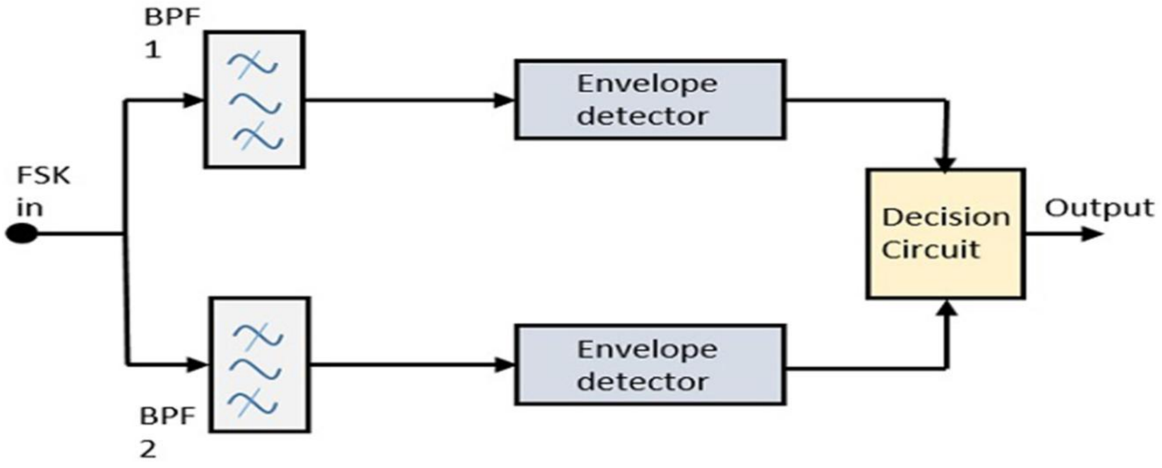
The two oscillators, producing a higher and a lower frequency signals, are connected to a switch along with an internal clock. To avoid the abrupt phase discontinuities of the output waveform during the transmission of the message, a clock is applied to both the oscillators, internally. The binary input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

FSK Demodulator

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

Asynchronous FSK Detector

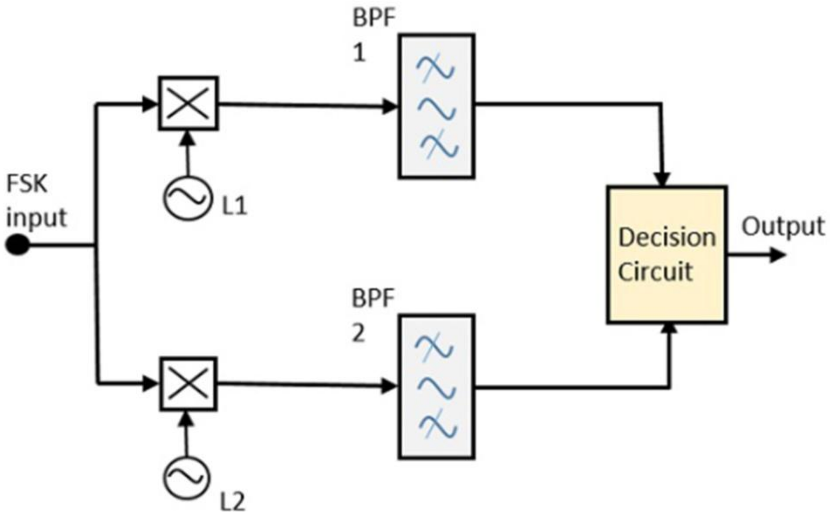
The block diagram of Asynchronous FSK detector consists of two band pass filters, two envelope detectors, and a decision circuit. Following is the diagrammatic representation.



The FSK signal is passed through the two Band Pass Filters (BPFs), tuned to **Space** and **Mark** frequencies. The output from these two BPFs look like ASK signal, which is given to the envelope detector. The signal in each envelope detector is modulated asynchronously. The decision circuit chooses which output is more likely and selects it from any one of the envelope detectors. It also re-shapes the waveform to a rectangular one.

Synchronous FSK Detector

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



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

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

3.3 Phase Shift Keying (PSK) is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.PSK is of two types, depending upon the phases the signal gets shifted. They are −

Binary Phase Shift Keying (BPSK)

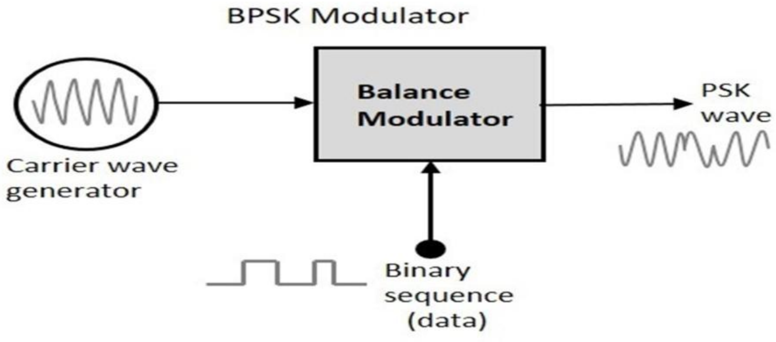
This is also called as 2-phase PSK or Phase Reversal Keying. In this technique, the sine wave carrier takes two phase reversals such as 0° and 180°. BPSK is basically a Double Side Band Suppressed Carrier (DSBSC) modulation scheme, for message being the digital information.

Quadrature Phase Shift Keying (QPSK)

This is the phase shift keying technique, in which the sine wave carrier takes four phase reversals such as 0°, 90°, 180°, and 270°. If this kind of techniques are further extended, PSK can be done by eight or sixteen values also, depending upon the requirement.

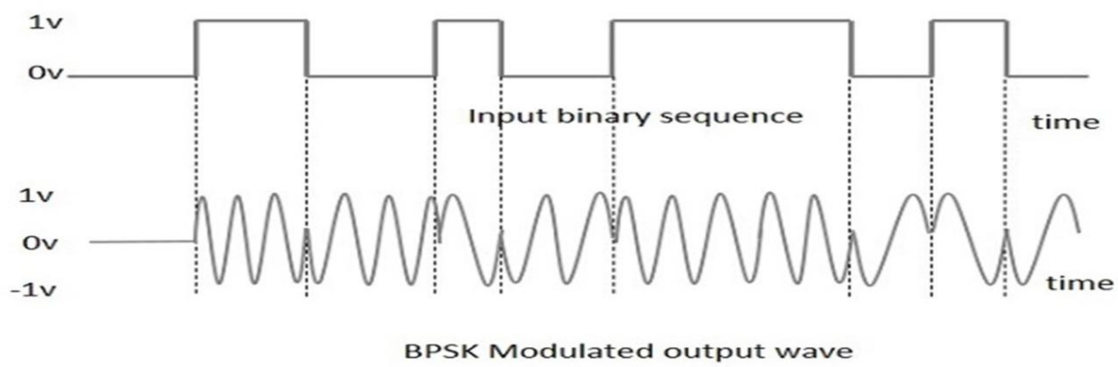
BPSK Modulator

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



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

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

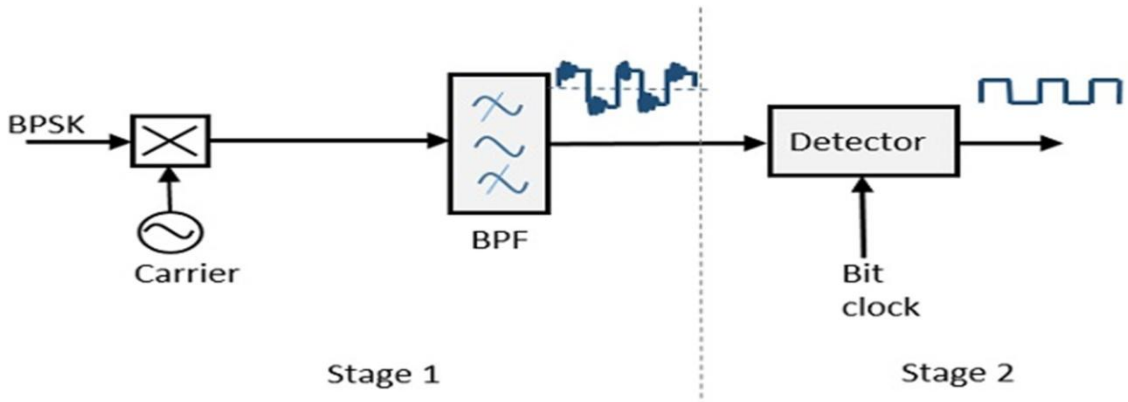


The output sine wave of the modulator will be the direct input carrier or the inverted (180° phase shifted) input carrier, which is a function of the data signal.

BPSK Demodulator

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

BPSK Demodulator



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

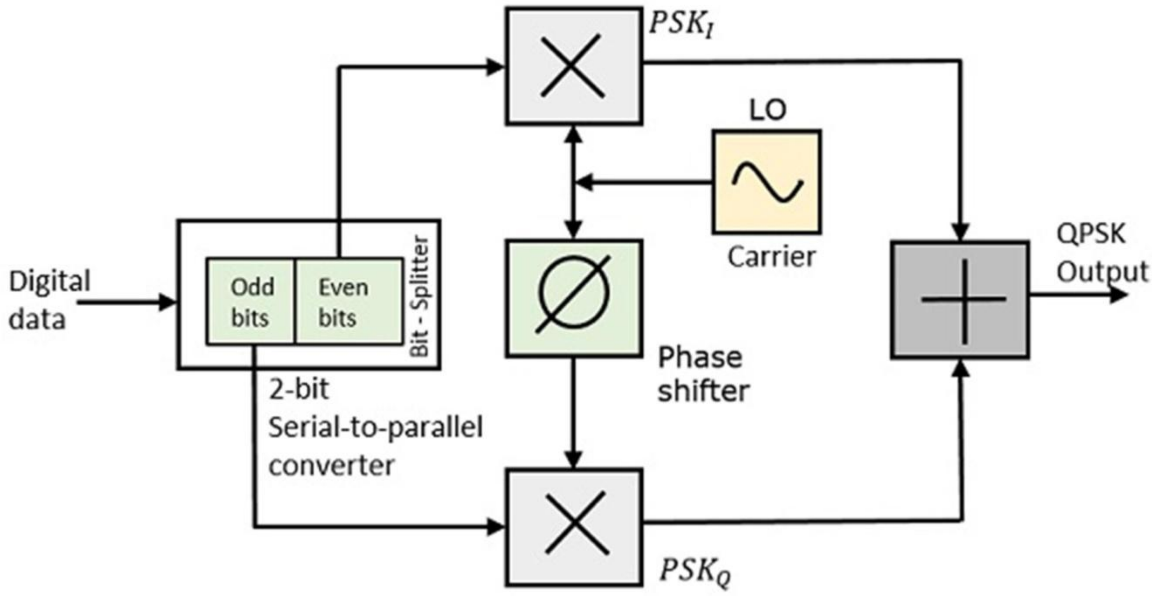
In the next stage of demodulation, the bit clock rate is needed at the detector circuit to produce the original binary message signal. If the bit rate is a sub-multiple of the carrier frequency, then the bit clock regeneration is simplified. To make the circuit easily understandable, a decision- making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

3.4 Quadrature Phase Shift Keying (QPSK)

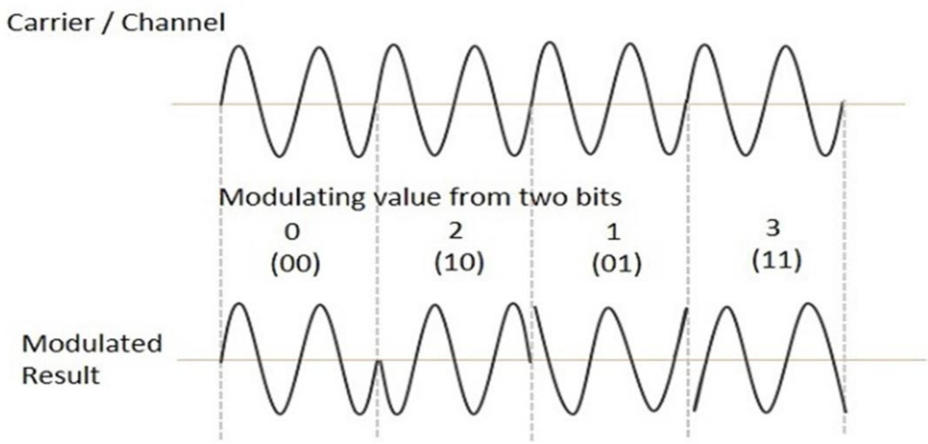
The Quadrature Phase Shift Keying (QPSK) is a variation of BPSK, and it is also a Double Side Band Suppressed Carrier (DSBSC) modulation scheme, which sends two bits of digital information at a time, called as **bigits**. Instead of the conversion of digital bits into a series of digital stream, it converts them into bit pairs. This decreases the data bit rate to half, which allows space for the other users.

QPSK Modulator

The QPSK Modulator uses a bit-splitter, two multipliers with local oscillator, a 2-bit serial to parallel converter, and a summer circuit. Following is the block diagram for the same.

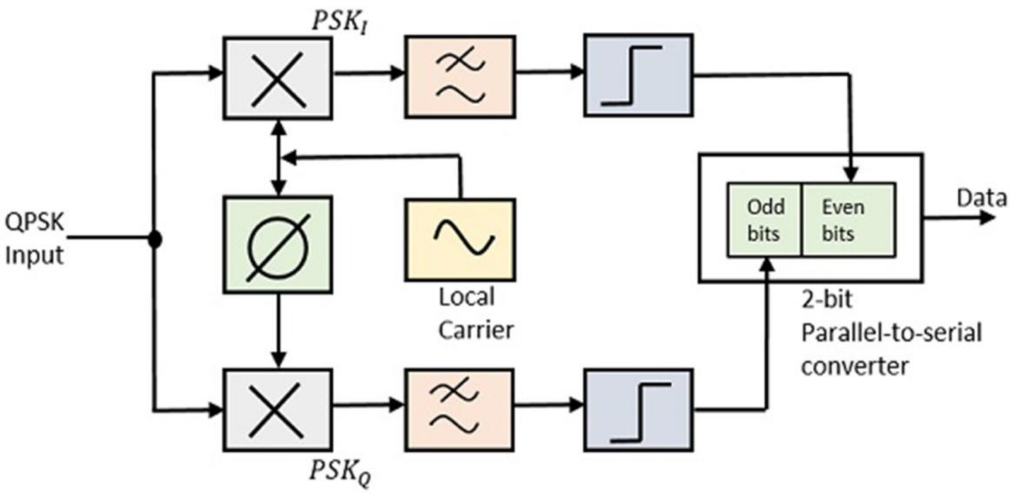


At the modulator's input, the message signal's even bits (i.e., 2nd bit, 4th bit, 6th bit, etc.) and odd bits (i.e., 1st bit, 3rd bit, 5th bit, etc.) are separated by the bits splitter and are multiplied with the same carrier to generate odd BPSK (called as **PSK_I**) and even BPSK (called as **PSK_Q**). The **PSK_Q** signal is anyhow phase shifted by 90° before being modulated. The QPSK waveform for two-bits input is as follows, which shows the modulated result for different instances of binary inputs.



QPSK Demodulator

The QPSK Demodulator uses two product demodulator circuits with local oscillator, two band pass filters, two integrator circuits, and a 2-bit parallel to serial converter. Following is the diagram for the same.



The two product detectors at the input of demodulator simultaneously demodulate the two BPSK signals. The pair of bits are recovered here from the original data. These signals after processing, are passed to the parallel to serial converter.

Chapter-4

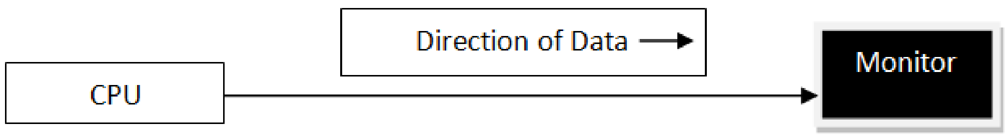
Data Transmission Circuits

In telecommunication, a data transmission circuit is the transmission media and the intervening equipment used for the data transfer between data terminal equipment (DTEs). A data transmission circuit includes any required signal conversion equipment. A data transmission circuit may transfer information in (a) one direction only, (b) either direction but one way at a time, or (c) both directions simultaneously.

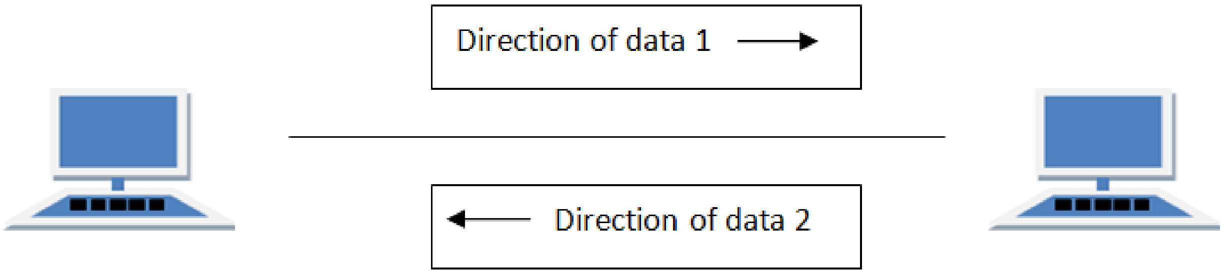
4.1 Transmission Modes

Transmission mode refers to the mechanism of transferring of data between two devices connected over a network. It is also called **Communication Mode**. These modes direct the direction of flow of information. There are three types of transmission modes. They are:

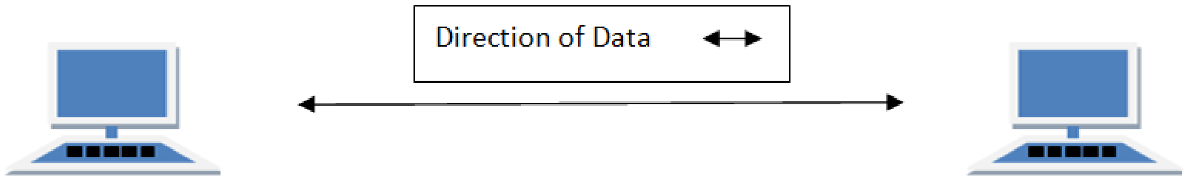
1. Simplex Mode
2. Half duplex Mode
3. Full duplex mode

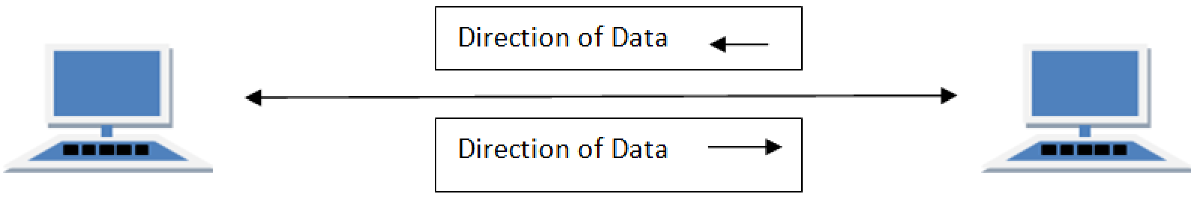


Simplex A to B only



Half duplex A to B or B to A

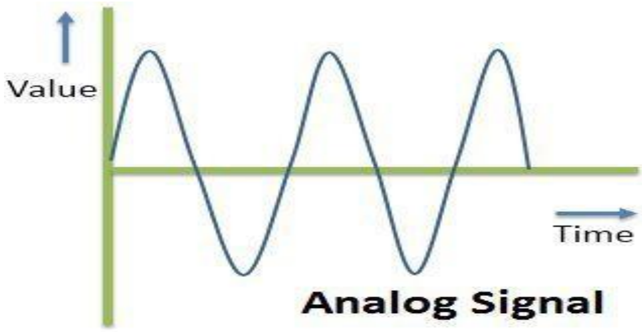




Full duplex A to B and B to A

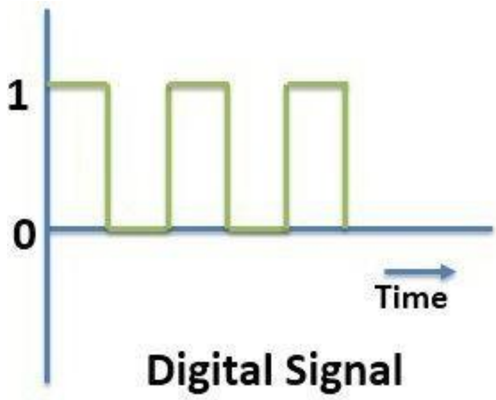
4.2 Digital and Analog Transmission

Analog signal is a kind of continuous wave form that changes over time. An analog signal is further classified into simple and composite signals. A simple analog signal is a sine wave that cannot be decomposed further. On the other hand, a composite analog signal can be further decomposed into multiple sine waves. An analog signal is described using amplitude, period or frequency and phase. Amplitude marks the maximum height of the signal. Frequency marks the rate at which signal is changing. Phase marks the position of the wave with respect to time zero.



An analog signal is not immune to noise hence; it faces distortion and decrease the quality of transmission. The range of value in an analog signal is not fixed.

Digital signals also carry information like analog signals but is somewhat is different from analog signals. Digital signal is non-continuous, discrete time signal. Digital signal carries information or data in the binary form i.e. a digital signal represent information in the form of bits. Digital signal can be further decomposed into simple sine waves that are called harmonics. Each simple wave has different amplitude, frequency and phase. Digital signal is described with bit rate and bit interval. Bit interval describes the time require for sending a single bit. On the other hand, bit rate describes the frequency of bit interval.



4.3 Multiplexing

The process of combining various digital signals into one signal over a shared medium is called multiplexing. This process is done at the sender's end where the signals are combined to form a composite signal using the multiplexer. Multiplexer is a digital electronic device which is used to combine signals at the sender's end. De-multiplexing is the reverse process done at receiver's end and thus extracts the original signal at the receiver's end.

4.4 Bandwidth

The range of frequencies contained in a composite signal is called **bandwidth**. The bandwidth of a composite signal is the difference between the highest and lowest frequency contained in that signal. For example, if a composite signal contains frequencies between 1000Hz to 4000Hz, then the bandwidth is (4000 -1000) 3000Hz. It is the characteristic measure of the network performance. In easier terms, the bandwidth refers to the size of the medium through which data travels or the capacity of the medium.

4.5 Digital to Analog Conversion

DACs are commonly used in music players to convert digital data streams into analog audio signals. They are also used in televisions and mobile phones to convert digital video data into analog video signals which connect to the screen drivers to display monochrome or color images. These two applications use DACs at opposite ends of the frequency/resolution trade-off. The audio DAC is a low-frequency, high-resolution type while the video DAC is a high-frequency low- to medium-resolution type.

4.6 Modulation

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a modulating signal that typically contains information to be transmitted. Most radio systems in the 20th century used frequency modulation (FM) or amplitude modulation (AM) for radio broadcast.

Bit Rates of Digital Transmission Systems

System	Bit Rate	Observations
Telephone twisted pair	33.6-56 kbps	4 kHz telephone channel
Ethernet twisted pair	10 Mbps, 100 Mbps	100 meters of unshielded twisted copper wire pair
Cable modem	500 kbps-4 Mbps	Shared CATV return channel
ADSL twisted pair	64-640 kbps in, 1.536-6.144 Mbps out	Coexists with analog telephone signal
2.4 GHz radio	2-11 Mbps	IEEE 802.11 wireless LAN
28 GHz radio	1.5-45 Mbps	5 km multipoint radio
Optical fiber	2.5-10 Gbps	1 wavelength
Optical fiber	>1600 Gbps	Many wavelengths

4.7 Synchronous and Asynchronous Data Transmission

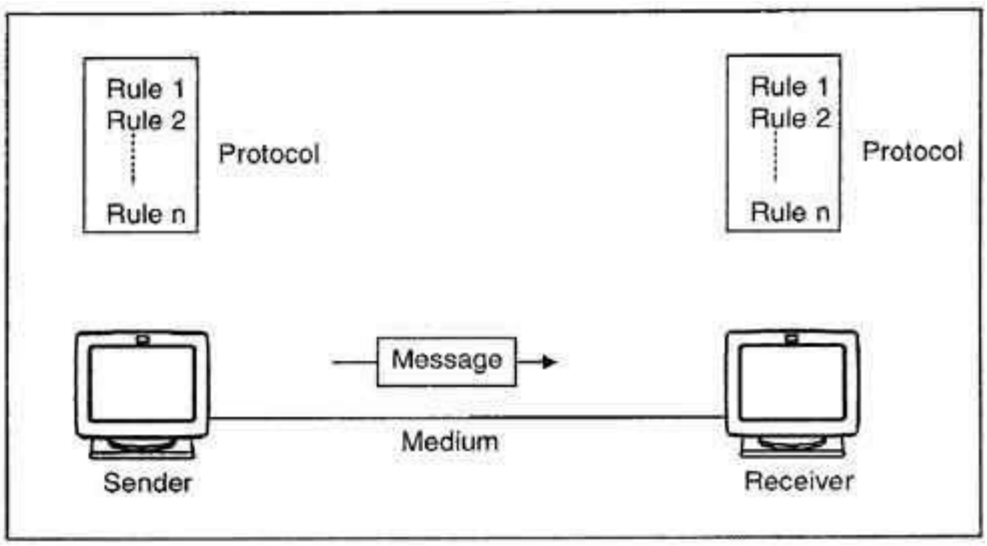
In **synchronous transmission**, data moves in a completely paired approach, in the form of chunks or frames. Synchronization between the source and target is required so that the source knows where the new byte begins, since there are no spaces included between the data.

Synchronous transmission is effective, dependable, and often utilized for transmitting a large amount of data. It offers real-time communication between linked devices. An example of synchronous transmission would be the transfer of a large text file. Before the file is transmitted, it is first dissected into blocks of sentences. The blocks are then transferred over the communication link to the target location.

In **asynchronous transmission**, data moves in a half-paired approach, 1 byte or 1 character at a time. It sends the data in a constant current of bytes. The size of a character transmitted is 8 bits, with a parity bit added both at the beginning and at the end, making it a total of 10 bits. It doesn't need a clock for integration—rather, it utilizes the parity bits to tell the receiver how to translate the data.

4.8 Data Transmission Circuit

Data communication refers to the exchange of data between a source and a receiver via form of transmission media such as a wire cable. Data communication is said to be local if communicating devices are in the same building or a similarly restricted geographical area. The meanings of source and receiver are very simple. The device that transmits the data is known as source and the device that receives the transmitted data is known as receiver. Data communication aims at the transfer of data and maintenance of the data during the process but not the actual generation of the information at the source and receiver.



Transmission over Short Distances

When the source and destination registers are part of an integrated circuit (within a microprocessor chip, for example), they are extremely close (thousandths of an inch). Consequently, the bus signals are at very low power levels, may traverse a distance in very little time, and are not very susceptible to external noise and distortion.

Transmission over Medium Distances

Computer peripherals such as a printer or scanner generally include mechanisms that cannot be situated within the computer itself. Our first thought might be just to extend the computer's internal buses with a cable of sufficient length to reach the peripheral. Doing so, however, would expose all bus transactions to external noise and distortion even though only a very small percentage of these transactions concern the distant peripheral to which the bus is connected.

Transmission over Long Distances

Data communications through the telephone network can reach any point in the world. The volume of overseas fax transmissions is increasing constantly, and computer networks that link thousands of businesses, governments, and universities are pervasive. Transmissions over such distances are not generally accomplished with a direct-wire digital link, but rather with digitally- modulated analog carrier signals. This technique makes it possible to use existing analog telephone voice channels for digital data, although at considerably reduced data rates compared to a direct digital link.

4.9 Characteristics of Data Transmission Circuits:

Bandwidth Requirements – Characteristics of Data Transmission Circuits is the most instances consists of pulse-type energy. The data stream is similar to a square-wave signal with rapid transitions from one voltage level to another, with the repetition rate depending on the binary representation of the data word. For instance, if an 8-bit word has the value 01010101, the resulting voltage graph would appear as a series of four square waves with each negative half- cycle equal to each positive half-cycle

Chapter-5

Modem

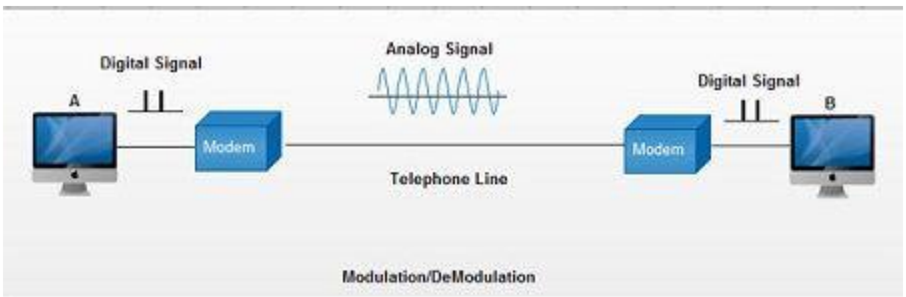
A modem converts the digital signal into analog data signals. Modem stand for modulation demodulation. They can be installed inside the computer in an expansion slot available for it. External modem are also available and they can be connected to a computer through a serial or USB port. Two general type of modem are **Standard modem and Window modem.**

The standard modem use generic device drivers and they can be integral as well as external ones. On the other hand a window modem is an integral plug and play device. It needs a special device driver provided by the window operating system to function properly. the internal modem do not require much physical configuration. They can be installed into a compatible through a cable called a null modem cable. Most of the home computer use a DSL modem which bridges the data from the phone line to a format usable by the interwork interface card and computer. Here DSL stand for digital subscriber line. A symmetric al DSL (SDSL) modem can send and receive at the same speed.

5.1 WHAT IS A MODEM

A modem is a device which is used to connect a computer to a standard telephone line so that one transmit and receive electronically transmitted data between two points in a network.

Modem is the key that opens up the world of the internet and World Wide Web (WWW), and e- mail on –line commercial services and bulletin board system (BBS). Modem is used to establish a communication link using phone lines between computer.



A modem is a modulator and demodulator. The main function of a modem is to covert analog data transmitter over phone lines into digital data that computer can understand and also to convert digital data (from computer)to analog data so that it can be transmitter o analog phone lines. Telephone lines coming to household are usually analog. It means that these line are designed to carry a voltage signal which is an exact replica of sound wave coming out of the mouth of the user. Such a voltage signal is known as analog and has a varying frequency and amplitude. However the computer are designed to understand and work on binary signal which have either high or low level (1s or 0s)different of these one and zero s reverent text commands and graphics in a computer.

5.2 NEED OF A MODEM

The modems are used in every communication system. The basic reason behind that is the analog communication channel that can transmit only the analog signal on them. .so if the data or signal is a digital data to need a conversion from digital data to analog signal that can be easily transmitted on the transmitter channel. The modem performs this tasking that is why it is required in most of the commutation circuits it accepts the digital information in from of sequence of bits from the source CPU and after analyzing convert these received signal back into digital data and hands it over to the destination CPU.

The role of the modem is as explained below:

Data Compression:

For reducing the amount of time, it takes for sending data and for cutting down on the amount of error in the signal, modems need to employ data compression. The data compression technique decreases the size of the signal that is needed for sending the required data.

Error Correction:

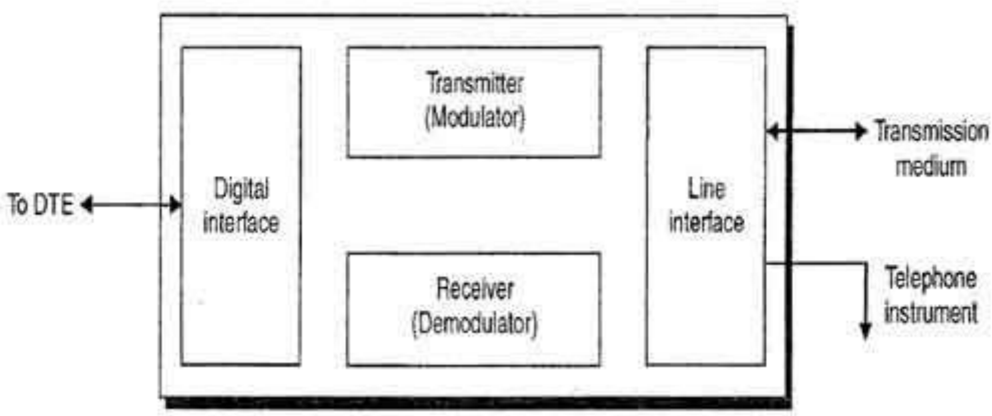
This is the process in which the Modem checks the information they have received is undamaged. Sometimes damage of data is being noticed in the form of altered or lost data. To get rid of this issue, the modem uses error correction. Batches of the information are being made and those frames are tagged with a checksum.

Flow Control:

The speed of sending information differs from modem to modem. There is a huge necessity of slowing down the speed of the fast modems so that the slow ones can work properly. Modem plays an important role in the networking of your computer. With the changing time and improving technology the working of these devices has changed and now they are providing much better service than ever before.

5.3 Working of a Modem

When an analog facility is used for data communication between two digital devices called Data Terminal Equipment (DTE), modems are used at each end. DTE can be a terminal or a computer. The modem at the transmitting end converts the digital signal generated by DTE into an analog signal by modulating a carrier. This modem at the receiving end demodulates the carrier and hand over the demodulated digital signal to the DTE.



Building blocks of a modem

The transmission medium between the two modems can be dedicated circuit or a switched telephone circuit. If a switched telephone circuit is used, then the modems are connected to the local telephone exchanges. Whenever data transmission is required connection between the modems is established through telephone exchanges.

5.4 Types of Modems

Internal Modems:

As the name signifies the internal modems are being connected with the motherboard of our computer. The internal modems are generally, dial-up or wireless (Wi-Fi). The telephone network is being used for sending and receiving signals in case of dial-ups. Authentication is required for the connection. In comparison to other modems, dial-up are markedly slower.

Coming to the Wi-Fi modems, there is no need to connect them with the telephone network and authentication is not required for such devices.

External Modems:

An external modem is an unconnected unit packed in a case. Basically, we connect an external modem with the telephone line and the computer through cables.

Cable Modems:

The name says it all! We have seen these sorts of modems at our homes or at cable operators place. The radio frequency that range that cable television uses is also being used by the cable modems. The using of the existing cable television infrastructure that allows the cable TV companies to provide Internet services.

ADSL Modems:

Asymmetric Digital Subscriber Line or what we call ADSL modems to use telephone lines for sending and receiving data. ASDL modems are really faster than any conventional voice band modem. The ASDL,as well as Cable modems, are used for providing the broadband internet connection. These types of modems allow more data transfer and that make the using of internet faster.

There are two types of data transmission used by Modems and those are synchronous and asynchronous. To make it more clear for you timing signals are used for Synchronous transmission and error-correcting formulas are used for asynchronous transmission. These device scan be used for one method of transmission or the other or can be used for both the ways.

Half duplex and full duplex Modems

Half duplex

A half duplex modem permits transmission in one direction at a time. If a carrier is detected on the line by the modem, It gives an indication of the incoming carrier to the DTE through a control signal of its digital interface.

(a) Half Duplex Modem (b) Full Duplex Modem

Full duplex

A full duplex modem allows simultaneous transmission in both directions. Therefore, there are two carriers on the line, one outgoing and the other incoming.

The basic modulation techniques used by a modem to convert digital data to analog signals are :

• Amplitude shift keying (ASK).

• Frequency shift keying (FSK).

• Phase shift keying (PSK).

• Differential PSK (DPSK).

These techniques are known as the binary continuous wave (CW) modulation.

• Modems are always used in pairs. Any system whether simplex, half duplex or full duplex requires a modem at the transmitting as well as the receiving end.

• Thus a modem acts as the electronic bridge between two worlds - the world of purely digital signals and the established analog world.

5.5 Modem Interconnection

Modems differ according to the method of interfacing with the communications circuits. If the circuit is a short and dedicated line, a limited distance modem can be used. This type of modem can be relatively simple in its circuitry since it does not have to drive a line which utilizes switching systems and line control devices such as echo suppressors.

The majority of data circuits utilize telephone channels provided by public carriers. These channels generally pass through switching facilities and are provided with equipment designed to enhance the use of the channel for voice applications. This type of equipment is not designed specifically for data transmission, so that the modems must be designed to compensate for any inadequacies of the voice-grade channel. Two broad types of modems are available for this type of service, the hard-wired modem and the acoustically coupled data set.

A hard-wired modem connects directly to the communication circuit in a semipermanent way. Such modems may be self-contained devices which connect to terminals and business machines, or they may be incorporated in the business machine. Connected to the communications circuit at all times, the hard-wired units can be polled (automatically contacted by the computer) and interrogated at any time. If associated with proper business machines and computers, these modems can send and receive data without human intervention. The one limitation of the hard-wired modem is that it precludes mobility since, being hard-wired, the equipment must remain connected to the circuit terminals.

The acoustically coupled modem solves the mobility problem. A standard telephone handset can be placed in the foam cups of an acoustic coupler, and the transmitter and receiver sounds will be conveyed to and from the telephone channel by transmit and receive elements of the acoustic coupler. The Data Communication Circuit using Modem components of the acoustic coupler form an interface with the business machine. Using this device, a person is able to interconnect with any computer system which has dial-up interconnect capability. Acoustic couplers are often built into briefcase-sized units which include a typewriter like terminal and a printer, providing the ability to access and manipulate data from any telephone. The portability and ease of connection afforded by the acoustic coupler are obtained at the expense of other capabilities. Since standard telephone circuits are typically used, speed of transmission is limited. The ability to have the system "on line" continuously is obviously not possible.

5.6 Modem Data Transmission Speed

Data Communication Circuit using Modem are generally classified according to the important characteristic of transmission speed as follows:

MODEM CLASSIFICATION	DATA RATE HANDLED (BPS)
Low-speed	Up to 600
Medium-speed	600 to 2400
High-speed	2400 to about 10,800

All of the above modems can operate within a single 300- to 3400-Hz (4-kHz) telephone channel. As speed increases beyond approximately 19,000 bps, a wideband modem is needed, as is a wideband channel. Wideband circuits are available generally in multiples of 4-kHz circuits, but the cost is significantly greater than for voice-grade circuits.

5.7 Modem modulation Methods

Data Communication Circuit using Modem utilize various types of modulation methods, the most common being frequency-shift keying (FSK), which shifts a carrier frequency to indicate a mark or a space. Encoded data can be transferred through communication systems designed for voice transmission because the frequency shifting is limited to the 4-kHz bandwidth of the voice-grade channel. The FSK signal is also analog in nature, enhancing its compatibility with communications circuits. Other types of modulation schemes are used, such as phase-shift-keying (PSK), four-phase PSK and eight-phase PSK, quadrature AM (QAM) and vestigial sideband AM.

Chapter-6

Space and Time Switching

In large networks, there may be more than one paths for transmitting data from **sender** to receiver. Selecting a path that data must take out of the available options is called **switching**. There are two popular switching techniques – circuit switching and packet switching.

6.1 Circuit Switching

When a dedicated path is established for data transmission between sender and receiver, it is called circuit switching. When any network node wants to send data, be it audio, video, text or any other type of information, a **call request signal** is sent to the receiver and acknowledged back to ensure availability of dedicated path. This dedicated path is then used to send data. ARPANET used circuit switching for communication over the network.

Advantages of Circuit Switching

Circuit switching provides these advantages over other switching techniques −

- Once path is set up, the only delay is in data transmission speed
- No problem of congestion or garbled message

Disadvantages of Circuit Switching

Circuit switching has its disadvantages too −

- Long set up time is required
- A request token must travel to the receiver and then acknowledged before any transmission can happen
- Line may be held up for a long time
-

6.2 Packet Switching

As we discussed, the major problem with circuit switching is that it needs a dedicated line for transmission. In packet switching, data is broken down into small packets with each packet having source and destination addresses, travelling from one router to the next router.

6.3 Message switching techniques

Switched communication networks are those in which data transferred from source to destination is routed between various intermediate nodes. Switching is the technique by which nodes control or switch data to transmit it between specific points on a network. There are 3 common switching techniques:

1. Circuit Switching

2. Packet Switching
3. Message Switching

6.4 Message Switching –

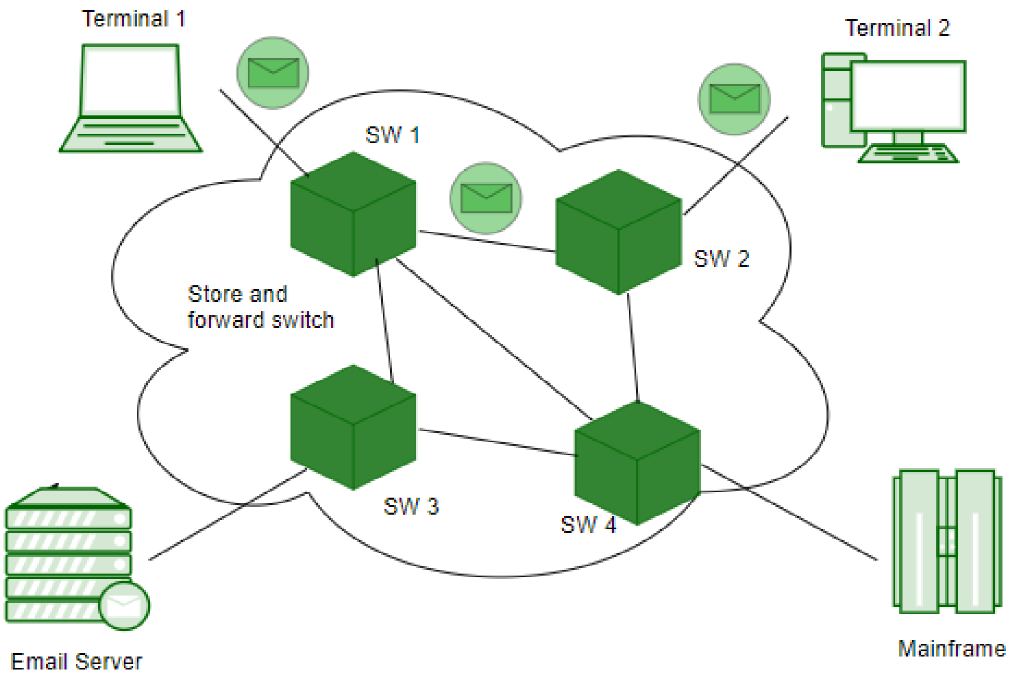
Message switching was a technique developed as an alternate to circuit switching, before packet switching was introduced. In message switching, end users communicate by sending and receiving *messages* that included the entire data to be shared. Messages are the smallest individual unit.

Also, the sender and receiver are not directly connected. There are a number of intermediate nodes transfer data and ensure that the message reaches its destination. Message switched data networks are hence called hop-by-hop systems.

They provide 2 distinct and important characteristics:

1. **Store and forward –** The intermediate nodes have the responsibility of transferring the entire message to the next node. Hence, each node must have storage capacity. A message will only be delivered if the next hop and the link connecting it are both available, otherwise it'll be stored indefinitely. A store-and-forward switch forwards a message only if sufficient resources are available and the next hop is accepting data. This is called the store-and-forward property.
2. **Message delivery –** This implies wrapping the entire information in a single message and transferring it from the source to the destination node. Each message must have a header that contains the message routing information, including the source and destination.

Message switching network consists of transmission links (channels), store-and-forward switch nodes and end stations as shown in the following picture:



6.5 Characteristics of message switching

Message switching is advantageous as it enables efficient usage of network resources. Also, because of the store-and-forward capability of intermediary nodes, traffic can be efficiently regulated and controlled. Message delivery as one unit, rather than in pieces, is another benefit.

However, message switching has certain disadvantages as well. Since messages are stored indefinitely at each intermediate node, switches require large storage capacity. Also, these are pretty slow. This is because at each node, first there us wait till the entire message is received, then it must be stored and transmitted after processing the next node and links to it depending on availability and channel traffic. Hence, message switching cannot be used for real time or interactive applications like video conference.

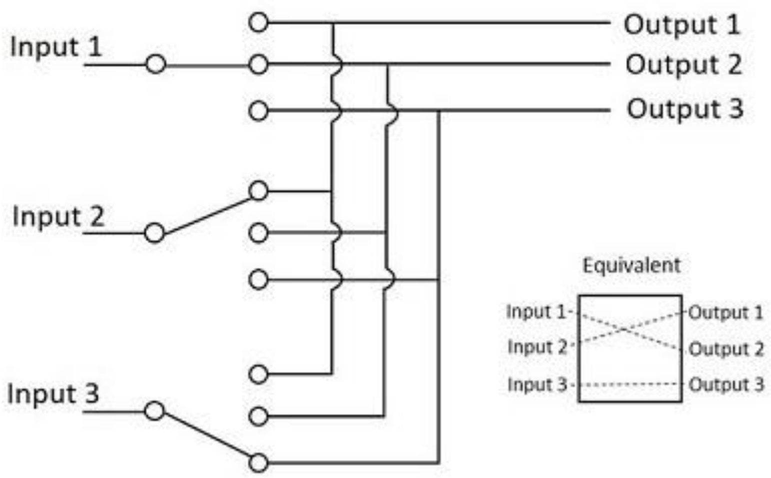
Applications

The store-and-forward method was implemented in telegraph message switching centres. Today, although many major networks and systems are packet-switched or circuit switched networks, their delivery processes can be based on message switching. For example, in most electronic mail systems the delivery process is based on message switching, while the network is in fact either circuit-switched or packet-switched.

6.6 Space Switching

The paths in a circuit are separated from each other, spatially in space division switching. Though initially designed for analog networks, it is being used for both analog and digital switching. A Crosspoint switch is mostly referred to as a space division switch because it moves a bit stream from one circuit or bus to another.

The switching system where any channel of one of its incoming PCM highway is connected to any channel of an outgoing PCM highway, where both of them are spatially separated is called the **Space Division Switching**. The Crosspoint matrix connects the incoming and outgoing PCM highways, where different channels of an incoming PCM frame may need to be switched by different Crosspoints in order to reach different destinations.



Though the space division switching was developed for the analog environment, it has been carried over to digital communication as well. This requires separate physical path for each signal connection, and uses metallic or semiconductor gates.

Advantages of Space Switching

Following is the advantage of Space Division Switching −

- It is instantaneous.

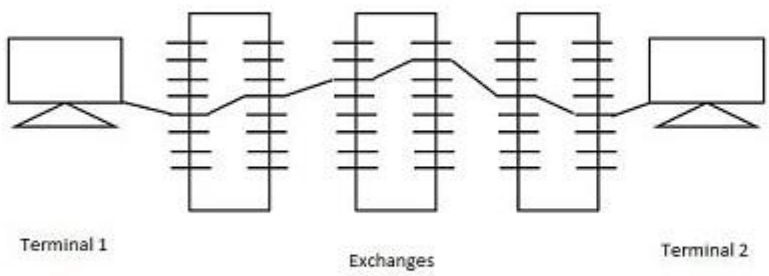
Disadvantages of Space Switching

- Number of Crosspoints required to make space-division switching are acceptable in terms of blocking.

6.7 Time Switching

Time division switching comes under digital switching techniques, where the Pulse Code Modulated signals are mostly present at the input and the output ports. A digital Switching system is one, where the inputs of any PCM highway can be connected to the outputs of any PCM highway, to establish a call. The incoming and outgoing signals when received and re-transmitted in a different time slot, is called **Time Division Switching.** The digitized speech information is sliced into a sequence of time intervals or slots. Additional voice circuit slots, corresponding to other users are inserted into this bit stream of data. Hence, the data is sent in time frames.

The main difference between space division multiplexing and time division multiplexing is sharing of Crosspoints. Crosspoints are not shared in space division switching, whereas they can be shared in time division multiplexing, for shorter periods. This helps in reassigning the Crosspoints and its associated circuitry for other connections as well.



Time division switches use time division multiplexing, in switching. The two popular methods of TDM are TSI (Time and Slot Interchange) and TDM bus. The data sent at the transmitter reaches the receiver in the same order, in an ordinary time division multiplexing whereas, in TSI mechanism, the data sent is changed according to the ordering of slots based on the desired connections. It consists of RAM with several memory locations such as input, output locations and control unit.

Both of the techniques are used in digital transmission. The TDM bus utilizes multiplexing to place all the signals on a common transmission path. The bus must have higher data rate than individual I/O lines. The main advantage of time division multiplexing is that, there is no need of Crosspoints. However, processing each connection creates delay as each time slot must be stored by RAM, then retrieved and then passed on.

6.8 TST & STS Switching

Description: This invention relates in general to time space time (TST) telecommunication system switches, and, in particular, to TST switches for interconnecting digital Time Division Multiplex (TDM) communication lines, using two basic modules, a plurality of time switching modules and a plurality of space switching modules. A time folded TST switch concept is taught and claimed in co-pending application, Ser. No. 497,214, filed Aug. 14, 1974 with two of the co-inventors here being common co-inventors with another inventor thereof. Time Space Time (TST) switches are a particularly useful configuration of switching elements providing both time and space translation between channels of Time Division Multiplexed (TDM) telecommunications. A further object is to achieve improved reliability and lessened maintenance requirements through use of such TST switch systems using two basic modules. Features of this invention useful in accomplishing the above objects include, in a TST switch with combined and distributed state and control stores, the use of two basic modules, a time switching module and a space switching module with three circuit types (excluding clock and control circuits). The three circuit types include the time switch module control portion, the time switch module memory portion, and the space switch module. The two basic modules are interconnect able to realize virtually any size and configuration of a time division switch having distributed operation particularly with the control stores associated with the switching elements incorporated into the time switch and space switch modules. As an example, the space switch is integrated with the space control circuitry into a single LSI circuit. Multiple LSI circuits are configured as basic functional units providing desired flexibility in switch size and maintainability.

BASIS FOR COMPARISON	SYNCHRONOUS TRANSMISSION	ASYNCHRONOUS TRANSMISSION
Meaning	Transmission starts with the block header which holds a sequence of bits.	It uses start bit and stop bit preceding and following a character respectively.
Transmission manner	Sends data in the form of blocks or frames	Sends 1 byte or character at a time
Synchronization	Present with the same clock pulse.	Absent
Transmission Speed	Fast	Slow
Gap between the data	Does not exist	Exist
Cost	Expensive	Economical
Time Interval	Constant	Random
Implemented by	Hardware and software	Hardware only
Examples	Chat Rooms, Video Conferencing, Telephonic Conversations, etcetera.	Letters, emails, forums, etcetera.

BASIS FOR COMPARISON	BASEBAND TRANSMISSION	BROADBAND TRANSMISSION
Type of signalling used	Digital	Analog
Application	Work well with bus topology.	Used with a bus as well as tree topology.
Encoding Used	Manchester and Differential Manchester encoding.	PSK encoding.
Transmission	Bidirectional	Unidirectional
Signal range	Signals can be travelled over short distances	Signals can be travelled over long distances without being attenuated.

Crosstalk is when there are two different signals that are somehow partly connected and getting mixed with each other. This happens on electric circuit boards when a signal on one conductive path is partly radiated and received on a second conductive path. Circuit designers have to build in shielding techniques to prevent this crosstalk. Crosstalk causes noise.

Intersymbol interference deals with only one signal. Something in the channel is distorting and spreading the symbols in the data stream such that the symbols are overlapping. This makes it difficult for a receiver to distinguish the symbols and it leads to bit errors.

Courtesy:

1. Google.com
2. Wikipedia.org