Data Communication Fundamentals

Tilak De Silva

Preface

Data Communication Fundamentals is a book, which has most of the required basic theory for data communication. The whole book was written from my experience and I tried to present all concepts with the minimum involvement of mathematics. For SLT Technical Staff, I am sure that this will refresh your knowledge of theory. This is the first book of this series. The second book has more details about networking and TCP/IP.

In order to understand the new services such as CDMA, ADSL, MPLS, WiMAX etc. the knowledge of data communication is essential.

I hope that this book will useful for you as a data communication basic handbook.

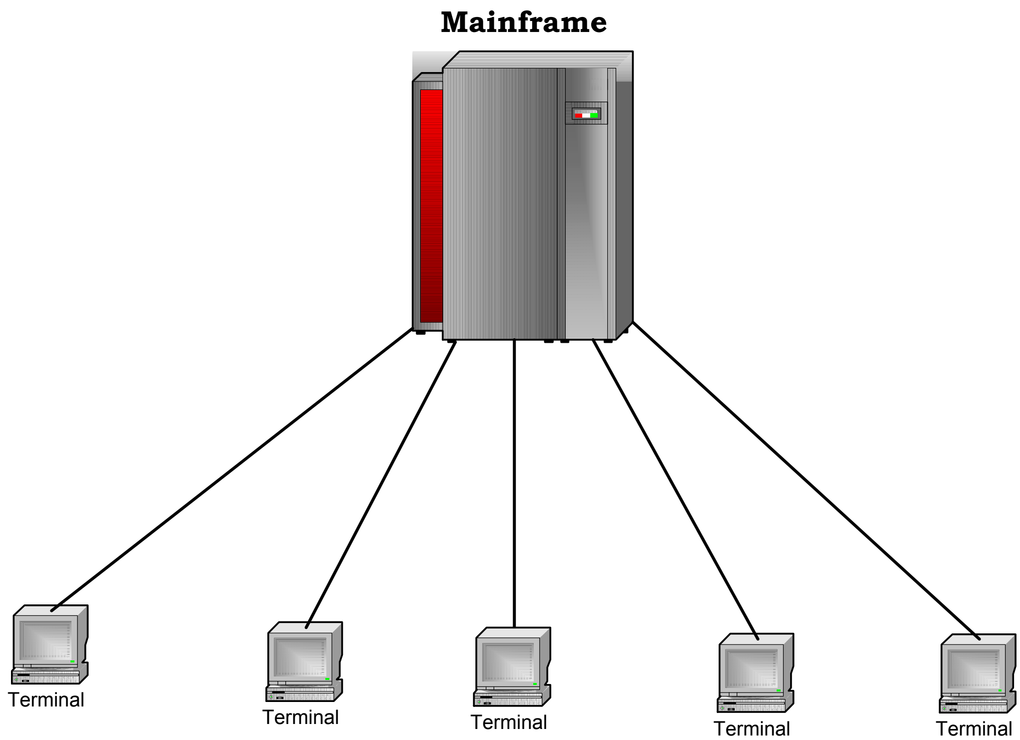
Tilak De Silva

B.Sc. Eng., M.Sc. (Brunel) C. Eng., CITP, FIE(SL), FIEE(UK), MBCS(UK)
24th March 2006

Outline of Data Communication

The first generation of computers started in 1940s to be used for World War II. The computer ENIAC was used in 1946. It consisted of 18,000 vacuum tubes, 1500 relays which weighed 30 tons and consumed 14 kW of power. With the development of electronics the computer became smaller and smaller and today you can see even very small palmtop computers.

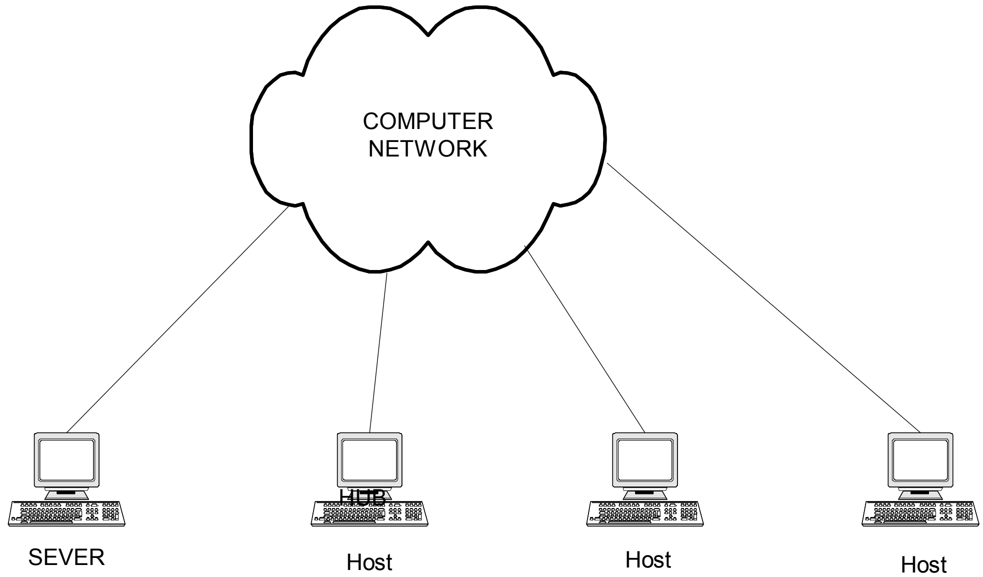
In 1970s mainframe computes were used and people connected to it with unintelligent terminals. This was the first kind of computer network and several persons could use the computer simultaneously.



When the computer became cheaper and smaller people tried to maintain large amounts of data in one computer. Then the database management concept immerged. One high-end computer called a server was used to maintain the database and others could connect to the server from their PCs. They worked in a room or floor or in the same building. Such networks were called Local Area Networks [LAN]. This was further extended to share printers, files etc.

The next step was to do the same functions with computers in remote locations. This type of network was called a Wide Area Network [WAN].

The LANs and WANs are called Computer Networks.



The databases, files, printers etc. can be called resources. Therefore, the main objective of computer networking is resource sharing. Apart from this, networking eliminates the barrier of physical separation. Although we stay thousands of miles apart, by networking we can work just as if we are in the same location.

Communication Fundamentals

In order to understand the basics of data communication following fundamentals are to be studied.

- Analog and Digital Signals
- Pulse Code Modulation [PCM]
- Multiplexing [FDM and TDM]
- Primary Mux [E1 or T1]
- High Order Muxes
- Modulation [Analog and Digital]
- Transmission Media Characteristics
- Copper, Fibre and Radio Transmission
- Satellite and Mobile Communication

Then we have to study,

- Data Transmission Principles
- LAN Concepts
- WAN Concepts
- Internetworking and Network Devices

For networking many network devices are needed. In order to purchase different devices from different vendors some kind of standard was needed. If the devices are not compatible they will not be able to communicate with each other. Therefore, International Standards Organization [ISO] defined the Open Systems Interconnection [OSI] Seven Layers. Today data communication and computer networking is based on the ISO-OSI seven layers. Therefore understanding this is very essential.

ISO-OSI Seven Layers

Application
Presentation
Session
Transport
Network
Data Link
Physical

Layer 1- Physical Layer

- interface with the communications hardware and transmission medium.
- transmission of an unstructured stream of data bits.

Layer 2 - Data-link Layer

- transmission of frames containing data and/or control information.
- provides error control and flow control over the data link.

Layer 3 - Network Layer

- effective where the end-to-end path consists of a series of data links either within one subnetwork [WAN] or over a collection of subnetworks [internet].
- provides routing and relaying over the subnetwork[s].

Layer 4 - Transport Layer

- provides a reliable end-to-end transfer of data between the two communicating systems.
- provides service independent of the underlying subnetwork[s].
- acts as a separator between the end system related protocols [interworking] and the subnetwork related protocols [interconnection].

Layer 5 - Session Layer

- manages the session [establishment, dialogue exchange, recovery, termination].

Layer 6 - Presentation Layer

- resolves differences in data representation in end systems.
- provides common transfer syntax.

Layer 7 - Application Layer

- provides network services for user application processes.
- file transfer, remote terminal access, messaging, remote job entry, management, security and directory services.

Other Standard Bodies

ANSI	-	American National Standards Institute
ITU-T	-	International Telecommunication Union – Technical [Former CCITT]
EIA	-	Electrical Industries Association
TIA	-	Telecommunications Industry Association
ECMA	-	European Computer Manufacturers Association

Services

Depending on different networking requirements different services to be provided. The services are categorized according to the different OSI layers.

Protocols

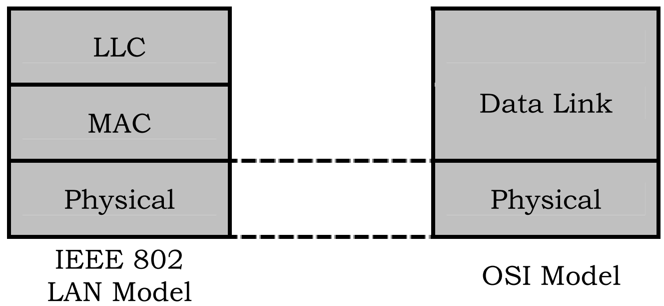
In order to facilitate exchange of information an agreed set of rules are to be followed and it is called a Protocol.

Depending on the requirement the services can be provided with different protocols.

E.g. Data Link Layer
WAN has different protocols.
HDLC, PPP, SLIP etc.
LAN has different protocols
Ethernet, Token Ring, Token Bus etc.

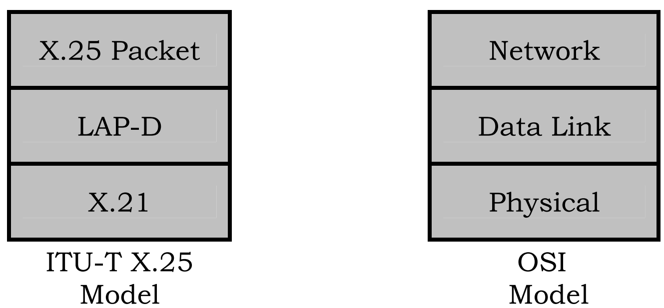
LAN Standards

The LAN standards are defined by IEEE. They have a separate reference model and it has a relationship to ISO-OSI Layers.



WAN Standards

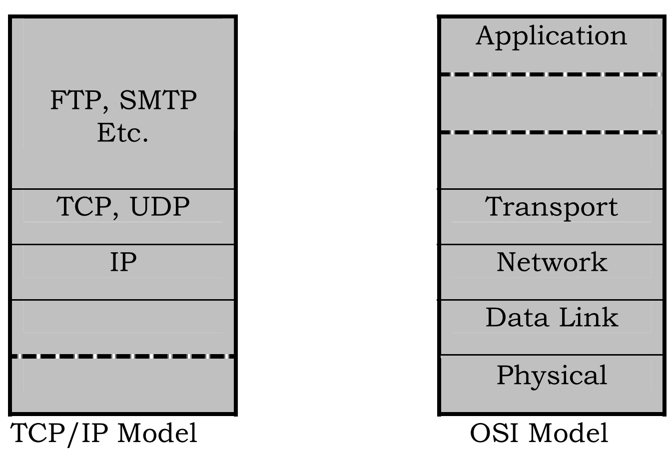
ITU-T defines a WAN Model called X.25.



- X.21 is the Physical Layer standard.
- LAP-D is the Data Link Layer Protocol.
- X.25 is the Network Layer Protocol.

TCP/IP

Internet organization defines the TCP/IP Model.



For TCP/IP, Physical Layer and Data Link Layers are not defined. TCP/IP can work with any Physical and Data Link Layer, which are compatible with TCP/IP. IP is the Network Layer Protocol. UDP and TCP are Transport Layer Protocols. Its Application Layer is equivalent to top three layers of OSI Model.

TCP/IP Application Layer Protocols are FTP, SMTP etc.

Internetworking

In order to connect computers to a network or connect networks, network devices are needed. Some of the network devices are,

- Hubs
- Switches
- Routers
- Gateways etc.

Network Software

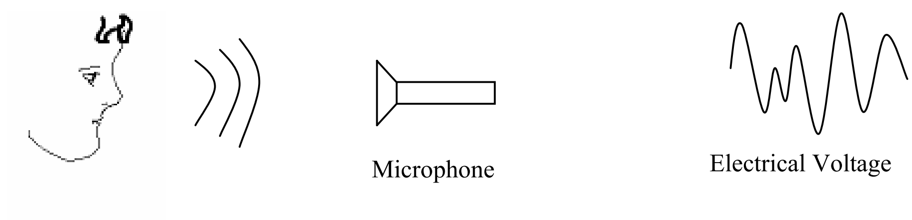
In order to operate the network efficiently and securely different types of software is required. Some of the network software are,

- Network operating system software
- Domain network software
- Address configuration software
- Network management software etc.

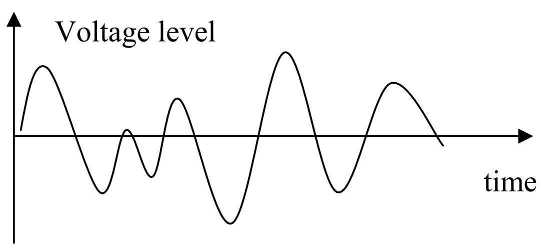
Communication Fundamentals

Analog Signal

A Signal is an electrical voltage or current, which varies with time. It is used to carry some information from one end to another. A typical example is a voice signal.



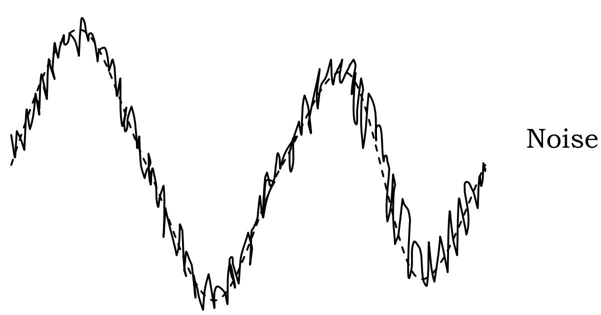
The microphone converts the sound signal to an electrical voltage.



This signal is continuously varying with time. This type of signal is called an analog signal.

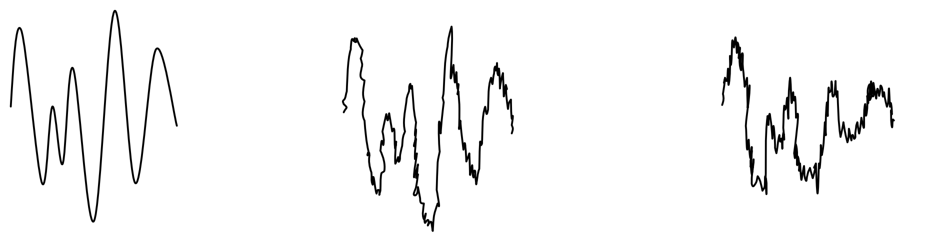
Noise

Noise is an unwanted signal. There are some freely moving electrons in the conductors. An electron movement is a current. Unwanted movement of electrons create unwanted currents. That is an unwanted signal or noise. Please note that there are some other ways of creating noise such as transistor noise, shot noise, galactic noise etc.



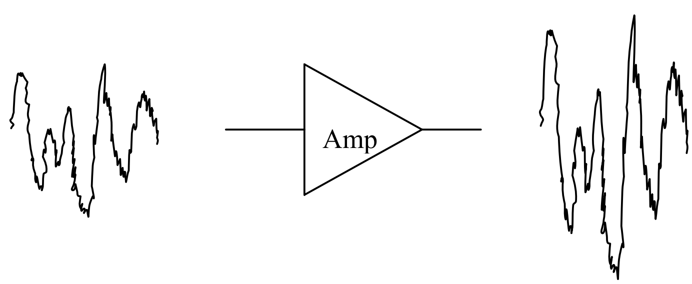
If any signal goes through a conductor it mixes with noise. The ratio of signal level and noise level is called the S/N or signal to noise ratio. The quality of a signal is measured as S/N. Higher the S/N, better the signal quality.

If an analog signal travels a long distance, more and more noise is added to it. Therefore, the S/N reduces and the signal quality is degraded. The biggest disadvantage of an analog signal is, the noise cannot be removed and it accumulates.

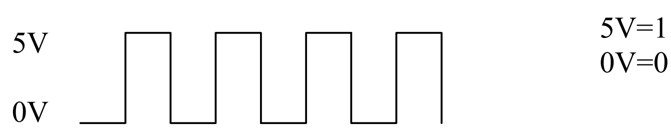


Original Signal | After traveling of Distance *l* | After traveling of Distance 2*l*

The other problem is, if the signal is amplified the noise is also amplified.

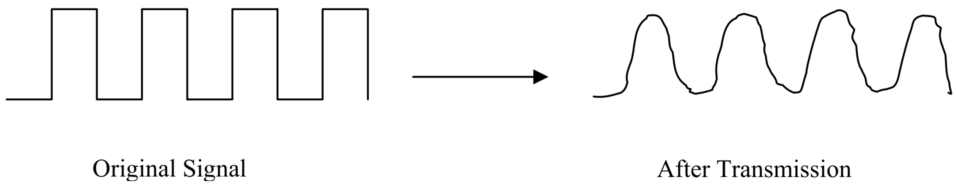


Digital Signal

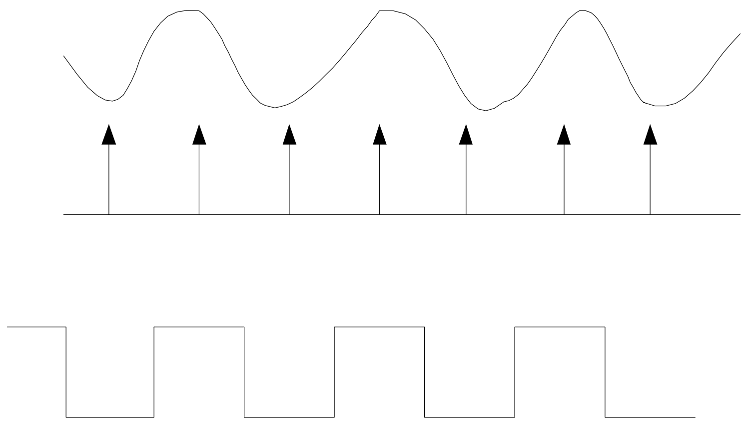


A discrete electrical signal, which has only two levels, is called a digital signal. These two levels are named as “1” and “0”. Normally a digital signal has a fixed number of bits and travels within a particular duration. This is called the pulse rate or bit rate.

The digital signal also gets mixed with noise.



The centre location of the digital signal can be identified by using a special bit pattern called a clock signal.



By checking the level at the centre location of each bit it can be decided whether it is a “0” or a “1”. This process is called the regeneration of the signal. By this method, the original signal can be generated and noise can be completely eliminated. This is the main advantage of a digital signal over an analog signal.

But there is a possibility to change the bit from 1 to 0 or 0 to 1 due to high noise. This is called an error. However there are many methods to correct these errors. Hence at the receive end the original bit pattern can be obtained. Hence digital signal quality will not depend on the distance traveled by the signal.

How to convert an analog signal to a digital signal?

The most commonly used method is the Pulse Code Modulation. Normally all voice telephone channels use this method.

Voice telephone channel frequency band is = 0.3 kHz to 3.4 kHz.

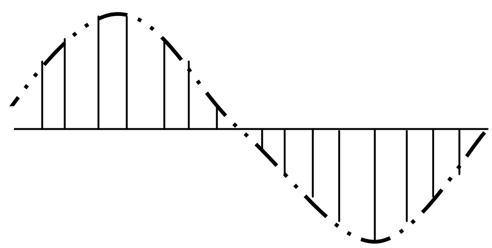
The process can be described as follows.

(i) Sampling
(ii) Quantizing
(iii) Encoding

What is sampling?

The samples of an analog signal are taken.

The sampled signal is called a pulse amplitude modulated signal.



It can be shown that the original signal can be constructed at the receive end using these samples.

Sampling Theorem

In order to completely reconstruct the original signal from the samples, the sample rate should be at least twice its highest frequency.

i.e. sampling rate $\geq$ 2 X highest frequency

The highest frequency of telephone voice channel is 3.4 kHz.

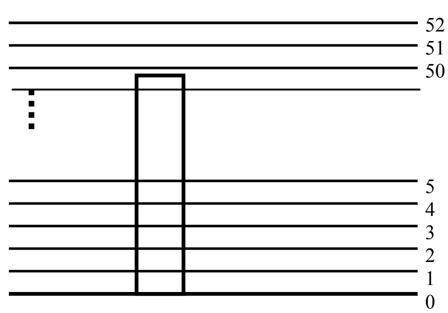
Hence sampling rate $\geq$ 2 X 3.4
$\geq$ 6.8 kHz

Hence a sample rate of 8 kHz is selected.

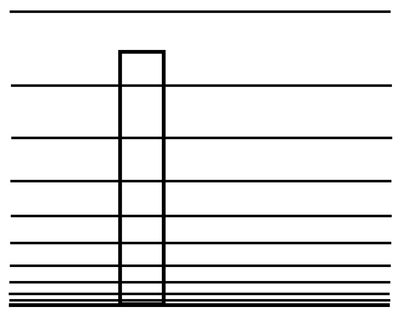
I.e. An analog signal is sampled at a rate of 8000 samples per second.

Quantizing

The samples are divided into many discrete levels. Then each sample is numbered according to their corresponding level.



There is no exact level for the above sample. The approximate level of the above sample is 50. Therefore the level of the sample is considered as 50. Hence an error will be introduced. This is called the quantizing error. This will reflect as noise at the receive end and it affects to the signal to noise ratio at the receive signal. It can be shown that, higher amplitude pulses will have high S/N and small amplitude pulses have low S/N. But we expect equal S/N for all pulses. In order to achieve this, non-linear quantizing is introduced.



It can be shown that using this method equal S/N can be obtained for all pulses.

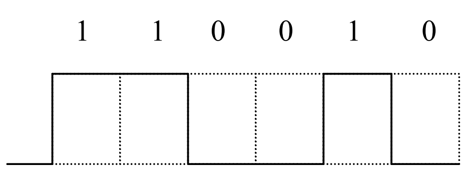
Encoding

After quantizing the corresponding level it is to be represented in some manner.

E.g. If the level is 50, it can be represented as,

Decimal	-	50
Hexa	-	32
Octal	-	62
Binary	-	110010

The 110010-bit pattern should be represented as an electrical signal, i.e. current or voltage. To represent a decimal number 10 voltage levels are required. Likewise 16, 8 and 2 voltage levels are required for hexa, octal and binary respectively. But practically representing more than two voltage levels is difficult. The most convenient and reliable method is using two levels. I.e. binary



This is called a bit stream.

Then we have to decide, how many quantizing levels are required. The more quantizing levels are used, more bits are required. It may cause to increase the bit stream and hence the bandwidth. Therefore, an optimum number of levels are to be selected. The standard number of levels is 256.

$$2^8 = 256$$

In order to represent 256 levels 8 bits are required. Hence each pulse is encoded to 8 bits.

1 sample	=	8 bits
Signal	=	8000 samples/sec
	=	8000 x 8 bits /sec
	=	64000 bits/sec
	=	64 kb/s

Therefore, bit rate of a digital telephone channel is 64 kb/s.

Analog Signal → 8000 samples/sec → 64kb/s

PCM Signal

Modulation

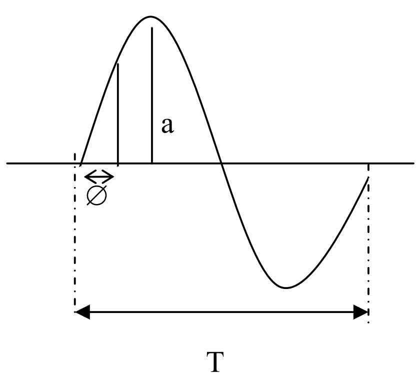
Modulation is a technique used to send information by modifying the characteristics of a basic electromagnetic signal. The basic signal is called the carrier signal.

The characteristics of a signal are amplitude, frequency and phase.

A signal can be represented by

$$a \,\mathrm{Sin}\,(\omega t + \varnothing)$$

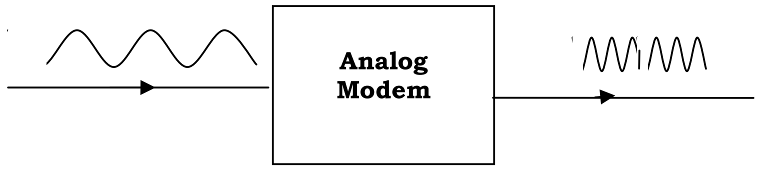
a - amplitude
ω - $2\pi f$ f - frequency
$\varnothing$ - phase



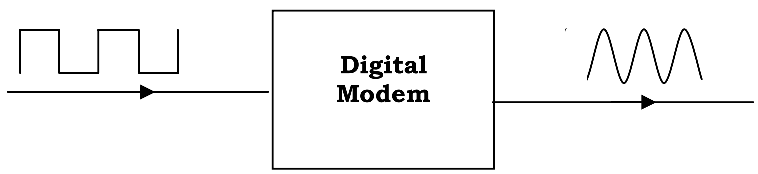
T = Period

$$f = \frac{1}{T}$$

Modulation can be used to convert a low frequency analog signal to a high frequency analog signal,

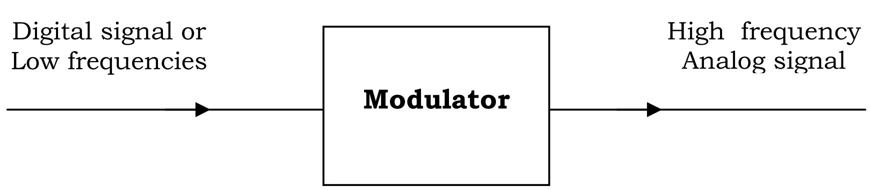


or a digital signal to an analog signal. For example a modem falls into the second category



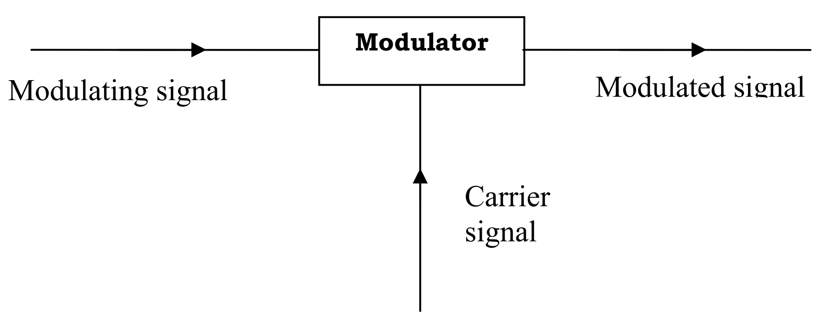
The input bit rate can be 9.6, 14.4, 19.2, 28.8, 56 kb/s. The output is an analog signal of frequency band 0.3 – 3.4 kHz.

Another application of modulation is to convert an analog or a digital signal to a very high frequency radio signal to transmit it through free space. [Broadband Radio Transmission]



[R*adio Transmission is discussed in another section*]

Modulation Process



Modulating Signal

This is the useful signal. This can be an analog signal or a digital signal. If the modulating signal is analog it is called analog modulation. If the modulating signal is digital, it is called digital modulation.

Carrier Signal

This is a high frequency analog signal.

Modulated Signal

The three characteristics of any signal are amplitude, frequency and phase. One of these characteristics are changed according to the shape of the input analog signal or the bit pattern of the input digital signal.

Modulation Methods

If the modulation signal is an analog signal, the three modulation methods are called,

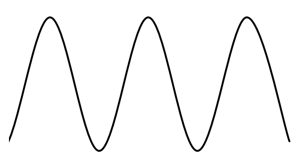
- Amplitude Modulation [AM]
- Frequency Modulation [FM]
- Phase Modulation [PM]

If the modulating signal is a digital signal, the three modulation methods are called,

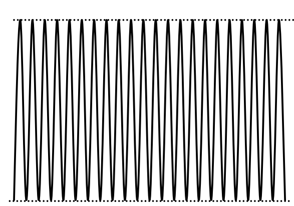
- Amplitude Shift Keying [ASK]
- Frequency Shift Keying [FSK]
- Phase Shift Keying [PSK]

Analog Modulation

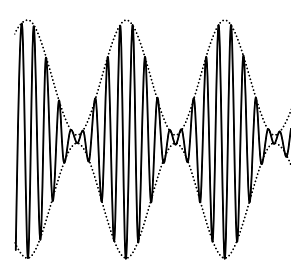
Amplitude Modulation [AM]



Modulating Signal



Carrier Signal

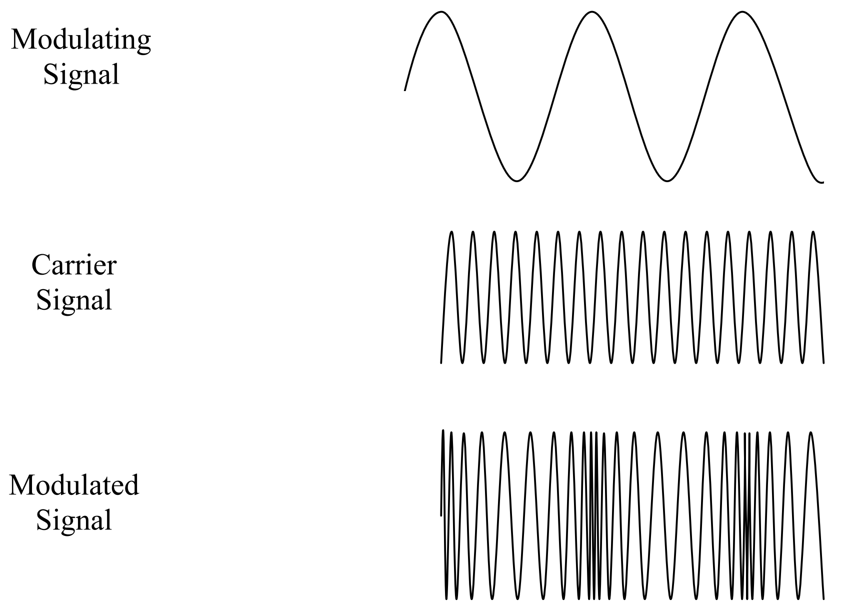


Modulated Signal

Amplitude of carrier signal varies according to the amplitude of modulating signal. The envelop of modulated signal is same as the shape of modulating signal.

Please note that the frequency or phase of the carrier signal is not changed.

Frequency Modulation



The carrier signal frequency changes according to the amplitude of the modulating signal. When amplitude increases, the modulated carrier signal's frequency increases. If the modulating signal amplitude is negative, the frequency of the modulated carrier signal is decreased.

Please note that the amplitude and phase of the carrier signal is not changed.

Phase Modulation

Same as AM or FM. Instead of Carrier Amplitude or Frequency the carrier phase is changed.

It is not possible to show it pictorially.

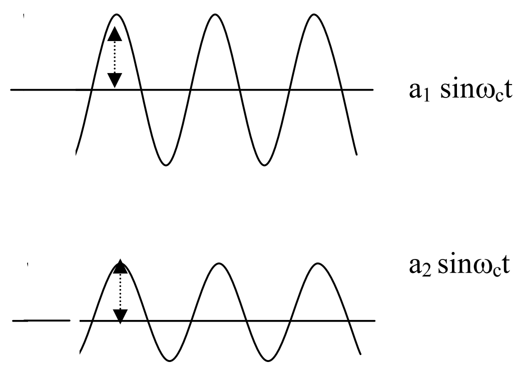
Digital Modulation

The digital signals are transmitted as 1s and 0s. The characteristic of the carrier signal is changed according to 1 or 0. That means there can be two states of amplitude, frequency or phase. The modulator switches [keying] the carrier to relevant state.

Amplitude Shift Keying [ASK]

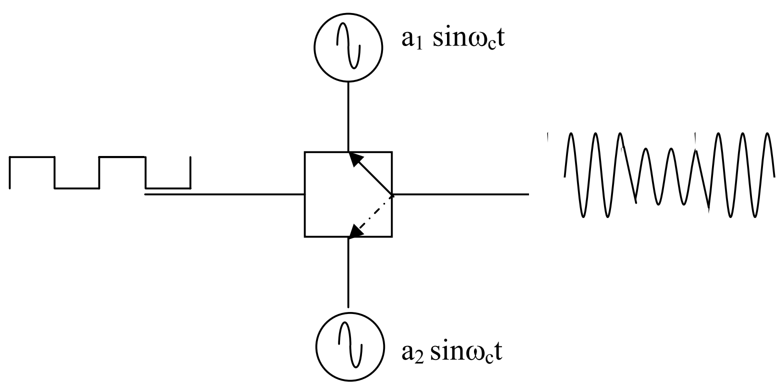
The two states are,

0 – amplitude 1 [a_1]
1 – amplitude 2 [a_2]

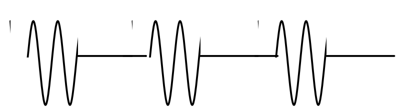


ω_c - $2\Pi f_C$ f_c – Carrier frequency

Please note that the frequency of both carrier signals are same.



If a_1 = 1V an a_2 = 0V input bit stream is 1 0 1 0 1 0, then the modulated signal pattern will be,

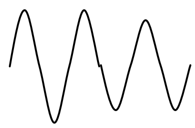


Frequency Shift Keying [FSK]

The two states are,

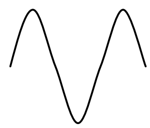
0 - frequency 1 [f1]
1 - frequency 2 [f2]

f_1



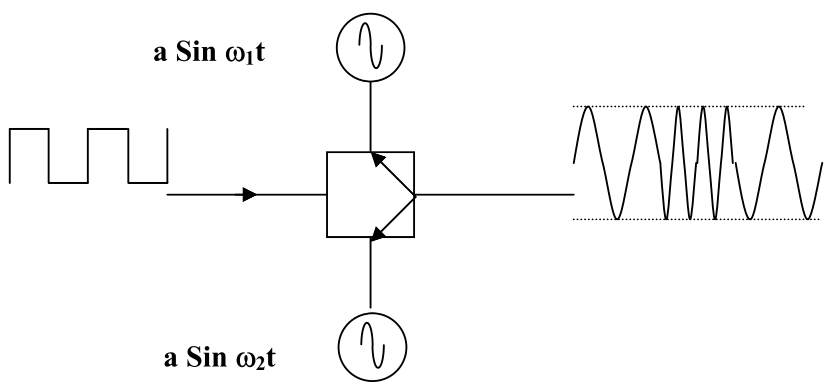
a Sin $\omega_1 t$ $\omega_1 = 2\pi f_1$

f_2



a Sin $\omega_2 t$ $\omega_2 = 2\pi f_2$

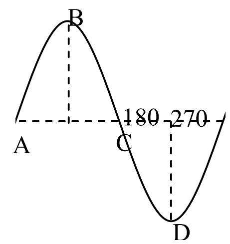
Note that the amplitude of both signals are same.



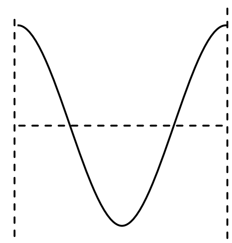
Phase Shift Keying [PSK]

In this method, the carrier signal phase is shifted according to the input digital signal.

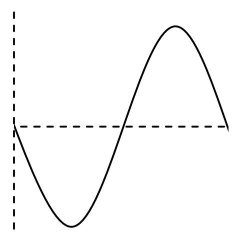
Let us first understand the phase of a signal.



The phase difference between A and B is 90^0.
In other words the point B is 90^0 phase shifted.

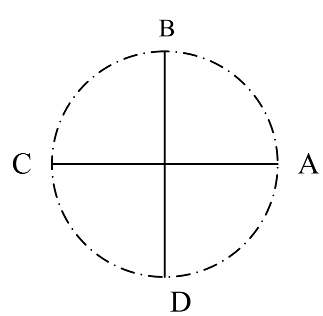


90^0 phase shifted signal.



180^0 phase shifted signal.

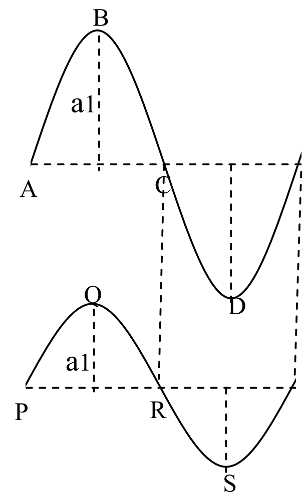
This also can be represented by using a phaser diagram.



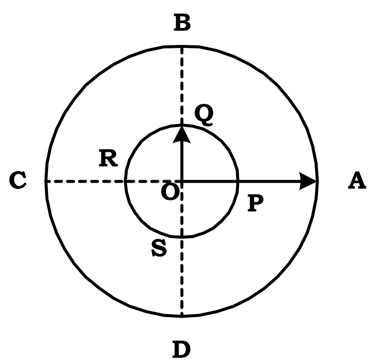
Consider two sinusoidal signals which have the same frequency but different amplitudes.

Signal 1 $a_1 \text{ Sin } \omega t$

Signal 2 $a_2 \text{ Sin } \omega t$



The phaser diagram can be drawn as follows.

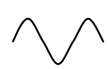


OA=a_1
OQ=a_2

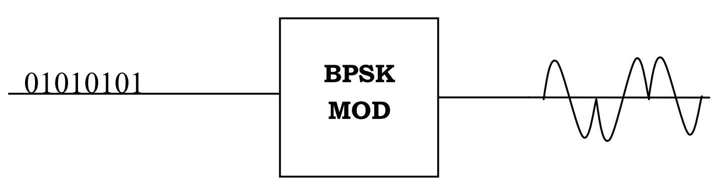
The PSK has different versions.
BPSK, QPSK, 8PSK, 16PSK etc.

Bipolar Phase Shift Keying [BPSK]

There are only two phases.

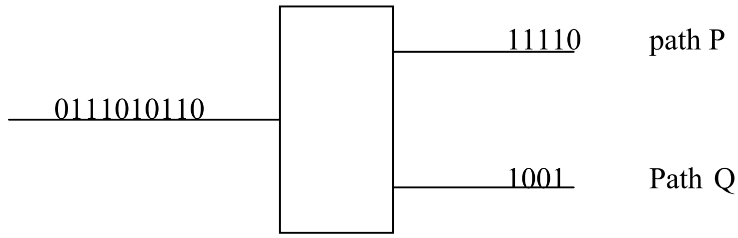
0 - no phase shift. 

1 - 180^0 phase shift 



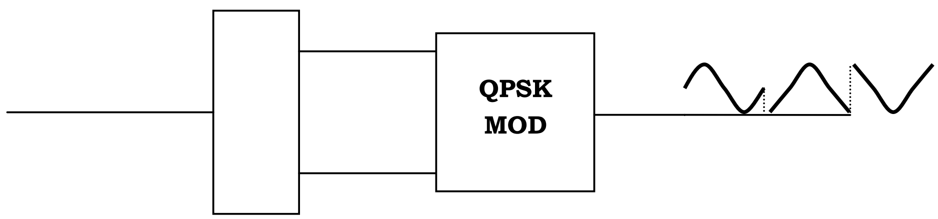
Quadrature Phase Shift Keying [QPSK or 4PSK]

In this method, first the input data stream is divided into two parallel streams.



Divider

First bit goes to P, second bit goes to Q, third bit goes to P, forth bit goes to Q and so on.



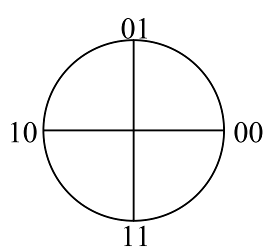
Divider

At the input of the QPSK Modulator, four types of bit combinations can be expected.
That is 00
01
10
11

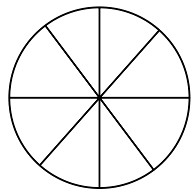
These bit combinations will have four different phases.

00 - 0^0, 01 - 90^0 , 10 - 180^0, 11 - 270^0

Phaser diagram



Similarly the 8PSK phaser diagram can be represented as follows.



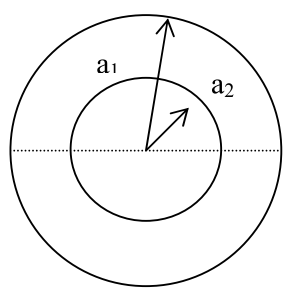
Bit Combination	Phase Shift [Degrees]
000	0
001	45
010	90
011	135
100	180
101	225
110	275
111	305

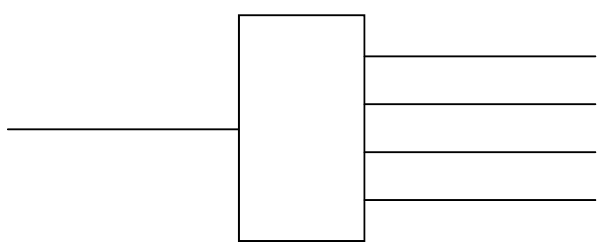
Hybrid Modulation

This is a combination of ASK and PSK.
This method of modulation is called Amplitude Phase Shift Keying [APSK] or Quadrature Amplitude Modulation [QAM].
It can be 16 QAM, 64 QAM etc.
In this method two carrier signals with different amplitudes are involved.

a_1 Sin ωt a_2 Sin ωt

The phaser diagram can be drawn as follows.



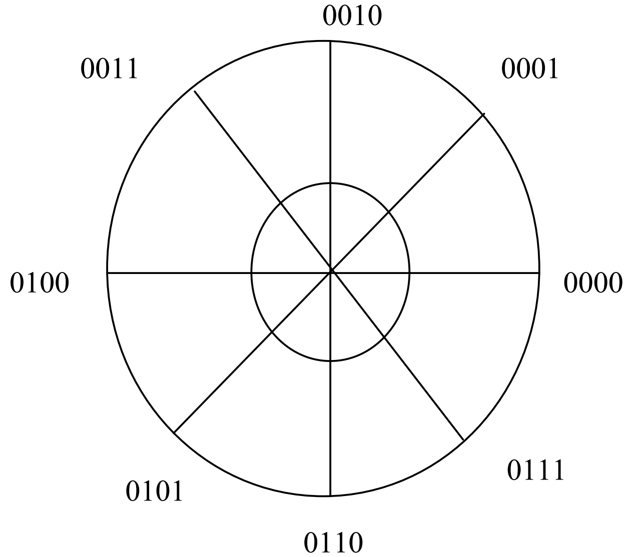


Divider

In this example, there are 16 combinations.

0000, 0001, 0010 …………….. 1111

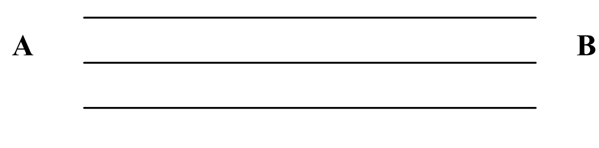
The 16 locations of the phaser diagram is as follows.



The inner circle corresponding locations represent 1000, 1001, 1010……..1111.

If the circle is divided into 16, 32 QAM can be represented.
If the circle is divided into 32, 64 QAM can be represented.

Multiplexing



Suppose we need to transmit four 64 kb/s signals from A to B. For this purpose, it is required to have four channels. Each channel needs at least 2 wires. If the length from A to B is 100m, we need 4 X 2 X 100 = 800m Copper Cable. If the length is 1000m the required length increases to 8000m.

If we can combine all four channels together without any mixing, a single pair of cable is sufficient. This type of combination (packing) of signal is called Multiplexing.

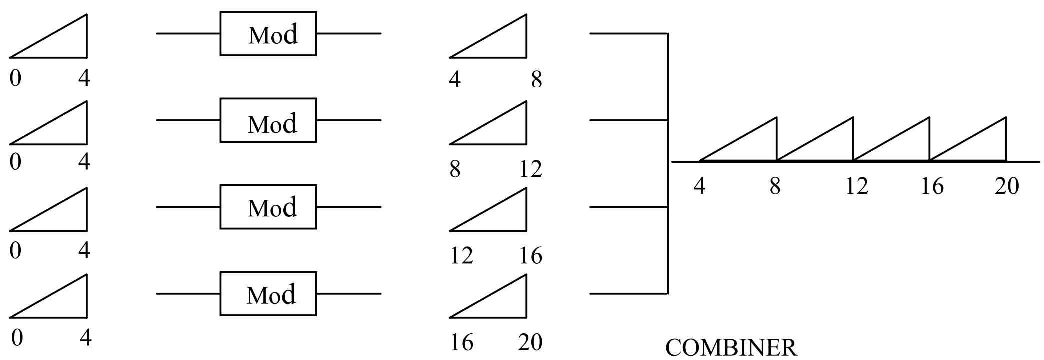
There are mainly two types of Multiplexing.

- Frequency Division Multiplexing [for Analog Signals]
- Time Division Multiplexing [for Digital Signals]

Frequency Division Multiplexing

Let us consider multiplexing of telephone channels.
One Channel - 0 – 4 kHz. [actually it is 0.3 – 3.4 kHz].

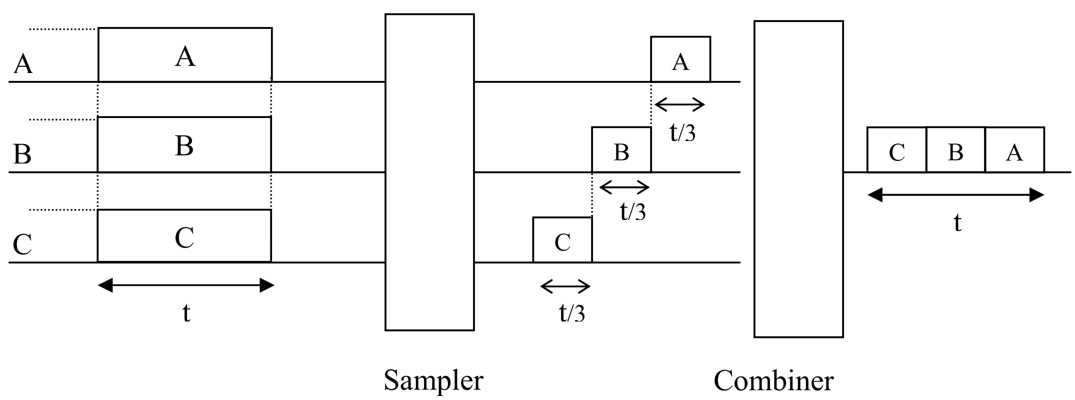
The frequency band can be shifted by modulation.



Here it can be seen that there is no interference of channels. This process is called Frequency Division Multiplexing.

Time Division Multiplexing [TDM]

Suppose we want to multiplex three Digital Signals, which have the same bit rate.



This process is called Time Division Multiplexing.
Suppose the input bit rate is n bits/sec
Time duration is t

t second $\longrightarrow$ 1 bit

1 second $\longrightarrow \frac{1}{t}$ bits $= n$ bits/sec

At the output

$$t \text{ second} \longrightarrow 3 \text{ bits}$$

$$1 \text{ second} \longrightarrow \frac{3}{t} \text{ bits}$$

$$= 3 \text{ X } \frac{1}{t} \text{ bits}$$

$$= 3 \text{ n bits/sec.}$$

It can be seen that in the TDM process, the output bit rate is increased.

Note : If A, B, C are single bits, the TDM method is called “bit interleaving”.
If A, B, C are each 8 bits, the TDM method is called “word interleaving”.
8 bits are also called a Byte or a Time Slot [TS].

TDM Systems

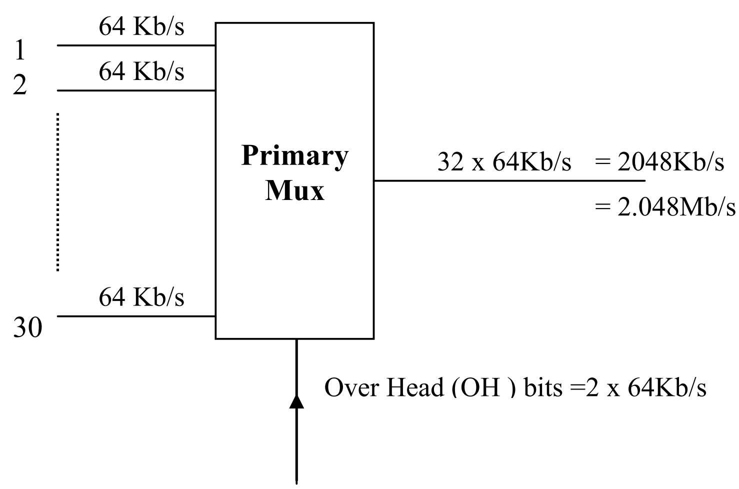
In actual systems, in addition to channel data, additional data is added. They are called the Over Head Bits. [OH bits]

E.g. Synchronization bits

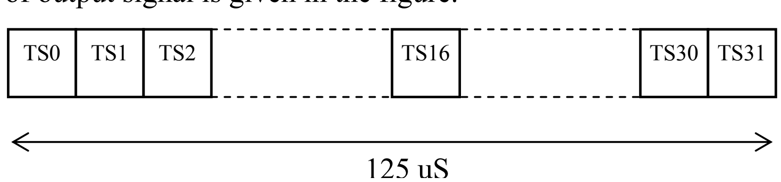
C	B	A	Syn	C	B	A	Syn

Primary Mux [E1 Channel]

By multiplexing 30 channels [each channel is 64 kb/s] the primary mux output is formed.



The frame structure of output signal is given in the figure.



Time Slot 0 [TS0] and Time Slot 16 [TS16] are overhead bits.

One Time Slot	=	8 bits
Therefore, 1 frame	=	8 X 32 bits
	=	256 bits.

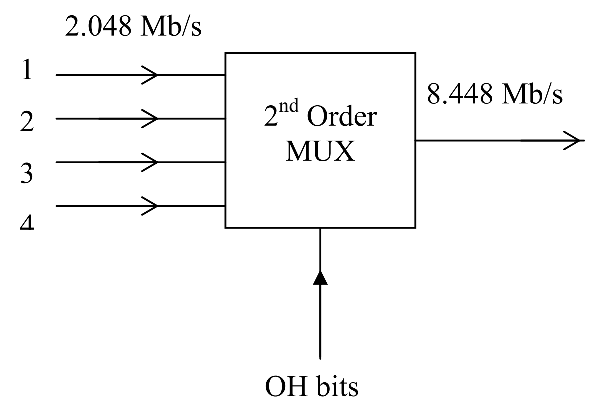
There is another Primary Mux which will multiplex 24 channels, and its output bit rate is 1.544 Mb/s. This is called a T1 channel.

Note : In Sri Lanka E1 multiplexing is used.
One TS carries data of one channel.
One channel is 64 kb/s.
Therefore, one TS is = 64 kb/s.
If you need a 64 kb/s data channel, the data circuit provider allocates you one Time Slot. If you need 128 kb/s data circuit, two Time Slots are allocated. Similarly for 512 kb/s data channel, 8 Time Slots are allocated. If you need a 2.048 Mb/s data channel the whole E1 is allocated.

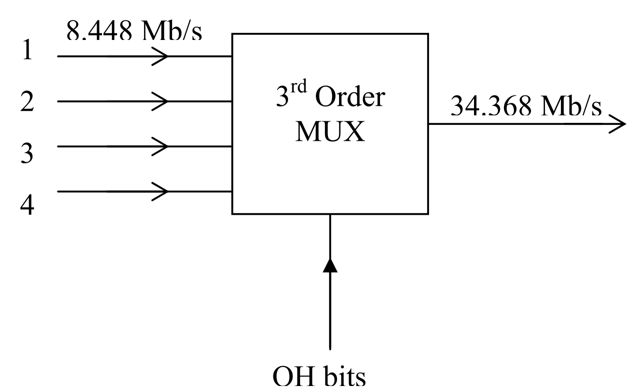
Higher Order Muxes

The primary mux is also called a 1st order mux. Four primary mux output can be again multiplexed and a 2nd order mux output is made.

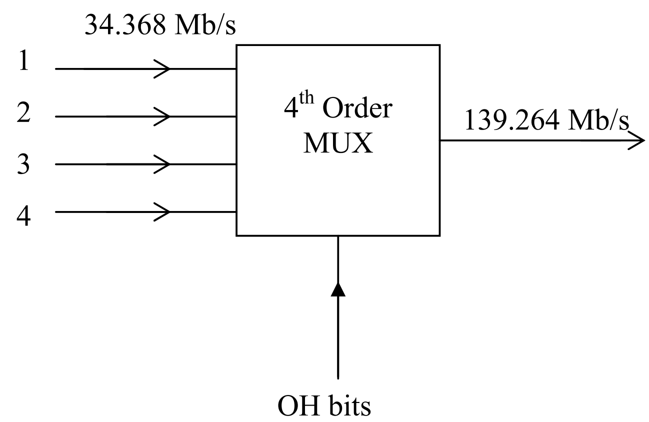
2nd Order Mux [E2 Channel]



3rd Order Mux [E3 Channel]

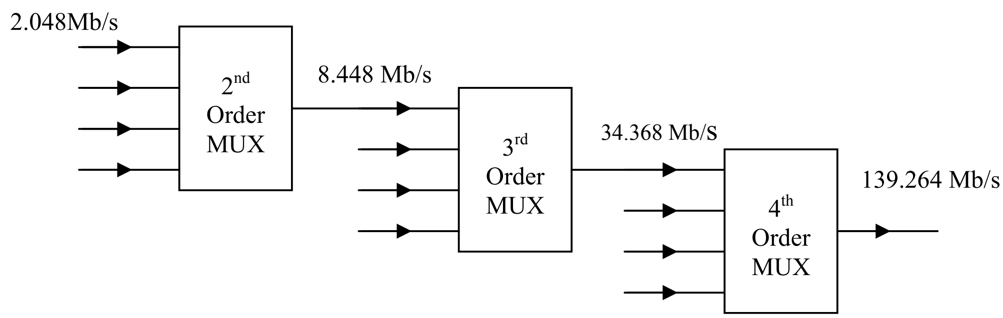


4th Order Mux [E4 Channel]



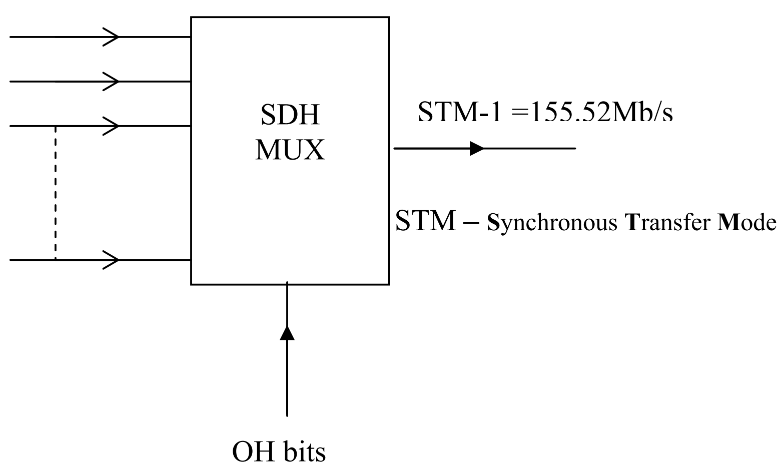
Plesiochronous Digital Hierarchy [PDH]

This is one of the digital multiplexing hierarchies.



Synchronous Digital Hierarchy [SDH]

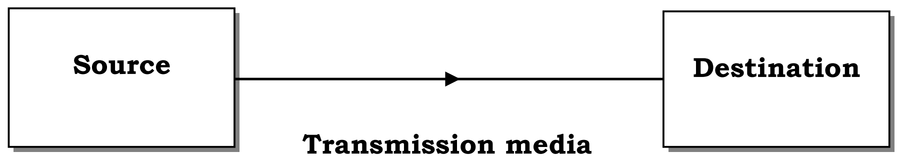
This is the modern digital multiplexing hierarchy.



The input can be E1 or E2 or E3 or E4.
The inputs can be configured.
The output bit rate is 155.52 Mb/s.

4 X STM - 1	=	STM - 4
4 X STM - 4	=	STM - 16
4 X STM - 16	=	STM - 64

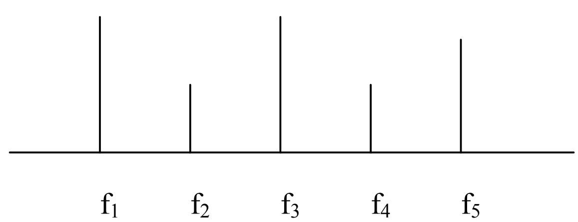
Bandwidth of a Signal



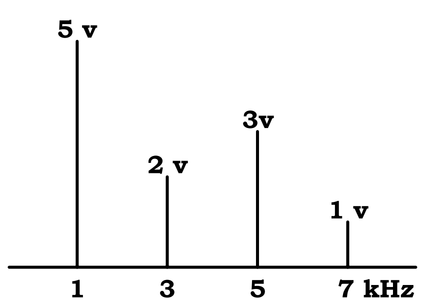
Any signal should travel from one point to another point. The starting point is called the source and the ending point is called the destination. Also it is called Transmitter and Receiver. The source and destination is connected by using a transmission media. It can be a copper cable, fibre optic cable or radio. The media bandwidth is a major cost factor of the system since the media cost depend on the bandwidth of the signal.

Bandwidth

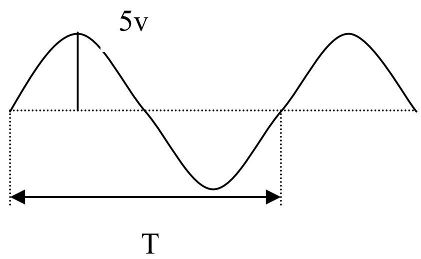
The correct term should be the frequency bandwidth. Any signal can be considered as a combination of sinusoidal signals. This is proved from a theory called Fourier Analysis which will be discussed later. The spread of frequencies can be shown pictorial in the following manner and it is called the frequency spectrum.



The vertical height shows the amplitude of the signal.
According to the above frequency spectrum the bandwidth of the signal is f_5 minus f_1 $[f_5 - f_1]$.
E.g.



The bandwidth is 7-1 = 6kHz or 1kHz to 7 kHz.
The 1kHz signal is



$T = 1/1 \times 10^3$ Seconds
$= 1$ ms

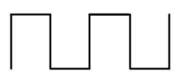
The 3kHz, 5kHz, 7kHz signals also can be represented in a similar manner.

Fourier Analysis

The signal transmitted from source to destination can be,

a digital signal
an analog modulated signal
a digital modulated signal etc.

How can we find the bandwidth of these signals?

Digital Signal ⟶ 

It is very strange if somebody says this digital signal is a mixture of sinusoidal signals. But there is a mathematical formula called Fourier Analysis [or Fourier Transformation] which proves that any signal is a combination of sinusoidal signals.

Bandwidth of a Digital Signal

Consider a digital signal (unipolar) which has a bit pattern of
1 0 1 0 1 0 1 0 1 0 1 0 [Alternative 1s and 0s]
Suppose the bit rate is n bits/second [n b/s].
It can be shown from the Fourier Analysis that the fundamental frequency [lowest frequency of the spectrum] is $n/2$ Hz.

Note : Fourier signals consist of fundamental and harmonics frequencies. Harmonic frequencies are integer multiples of fundamental frequency.
Fundamental frequency - f
Harmonics – 2f, 3f, 4f, 5f etc. or 3f, 5f, 7f etc. or 2f, 4f, 6f etc.

Consider a digital signal sent from source to destination. In the media it travels as individual sinusoidal [analog] signals. In order to regenerate the signal at the destination, at least the fundamental frequency is needed. Therefore, the media should support the travel of, at least up to $n/2$ Hz. Therefore, we can say the media bandwidth should be from 0Hz to $n/2$ Hz. Therefore, the bandwidth is $n/2 - 0 = n/2$ Hz.

Therefore, in general, we say that if the bit rate is *n* b/s, the required media bandwidth should be at least *n*/2 Hz.

In other words we say, if the media has N Hz bandwidth, it supports up to 2N b/s.

E.g. : If the media bandwidth is 3 kHz, we can send a digital signal which has maximum bit rate of 2 X 3 kb/s = 6 kb/s.

In general we can write the following expression. The maximum bit rate a media supports is also called its capacity [C].

$$\text{Capacity} = 2 \times \text{Bandwidth}$$
$$C = 2B$$

The above expression is also correct for ASK, FSK and BPSK modulated signals. But there will be a difference in QPSK, 8PSK, 16QAM, and 64QAM signals. They are called multilevel signals. The level [L] is defined by how many bit combinations is considered at the modulator input.

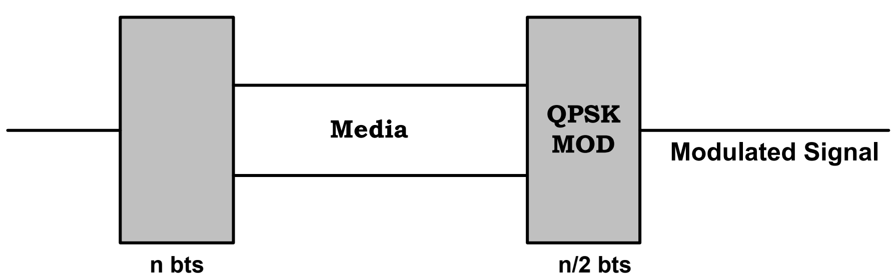
For examples:	QPSK	L = 4	8PSK	L = 8
	16QAM	L = 16	64QAM	L = 64

For multilevel signals, the above equation can be modified as follows.

$$C = 2B \log_2 L$$
$$\log_2 4 = 2 \quad \log_2 8=3 \quad \log_2 16=4 \quad \log_2 64=6$$

For modulated signals the bandwidth depends on the change of rate of carrier frequency. This is called the "baud rate".

Consider a QPSK signal.



Divider

The change of rate of Carrier Frequency is *n*/2 Hz. Therefore, the bandwidth of the modulated signal is *n*/2 Hz. The bandwidth of unmodulated signal is *n* Hz. Therefore, QPSK modulation reduces the required bandwidth to ½ [half].

Similarly for a 8PSK signal the required bandwidth is reduced to $1/3^{rd}$.

In other words, if the media bandwidth is not changed, QPSK can increase the input bit rate by 2 times. 8PSK can increase the input bit rate by 3 times.

Actual Bandwidth of a Media

In the above explanation we considered only the bandwidth requirements. The actual bandwidth available in a media depends on the signal level and the noise level. This relationship was given by Shannon and it is called Shannon's Law.

Shannon's Law

$$C = B\, log_2 (1+S/N)$$

C	-	Capacity (b/s)
B	-	Bandwidth of the media (Hz)
S	-	Signal Level
N	-	Noise Level

Therefore, in practical problems, we should follow the following steps.

Calculate the input bit rate [capacity] which can be increased by multilevel digital modulation. Use the following equation.

$$C_1 = 2\, B\, log_2 L$$

Note : We used the modulation technique to increase the input bit rate.

Next we have to check whether the transmission media supports the above bit rate. Therefore, check the highest bit rate supported by the media under the presence of noise by using the following equation.

$$C_2 = B\, log_2 (1+S/N)$$

If $C_1 < C_2$ the modulated signal can be sent through the media.

Transmission Media Characteristics

A source sends data through a transmission media. We cannot send an unlimited bandwidth through the media due to many limitations. The major problems in any transmission media is,

- Noise
- Attenuation
- Group Delay
- Interference

Noise

Noise is an unwanted electrical signal [voltage or current]. This mainly occurs due to random movement of electrons. This is called “thermal noise” or “white noise”. Copper [metal] conductors are highly affected by thermal noise.

Noise which mainly affects copper conductors are,

- cross talk - adjacent channel’s signal is induced.
- impulse noise - occurs from another electromagnetic source.

Radio signals are affected by atmospheric noise due to atmospheric water vapour, dust particles etc. Another type of noise affected by radio signals are the “Galactic Noise” due to unwanted electromagnetic waves radiated form some stars.

Fibre optics have no much effect from noise.

Attenuation

Assume that there is no noise in the media. Then can we transmit a signal to any distance? No, since the signal strength reduces when it ravel through the media. This effect is called attenuation.

In copper conductors this is due to heat dissipation. The signal goes as an electrical current. A current i dissipate i^2 R [R-Resistance] thermal power. This is a waste of energy of useful signal. Therefore, the signal level is degraded.

In radio transmission, the signal is attenuated due to atmospheric absorption by water vapour, dust etc.

In fibre optic transmission attenuation occurs due to scattering, absorption, bending and this will be discussed later.

Group Delay

The velocity of an electromagnetic signal travelling through a transmission media depends on the frequency of the signal. We noticed that any signal is a combination of many sinusoidal waves, which have different frequencies. Therefore, the signal wave components travel with different velocities and reach the destination at different times. This effect is called the group delay. Due to this effect the destination end should wait until all sinusoidal frequency components are received to reconstruct the original signal. The disadvantage of this effect is some frequency components of previous signal reach the destination after some frequency components of following signal are reached. The signal is reconstructed at the receive end and it is not similar to the original signal due to group delay and it is called the delay distortion or intersymbol distortion. In order to avoid this effect the bit rate should be limited.

The effect for copper cables due to group delay is negligible. For radio transmission it has considerable effect. For fibre optic transmission, this has a very bad effect and it is called "dispersion".

Interference

Radio signals are transmitted through free space. Since there are many frequencies [carriers] transmitted through free space, one carrier can interfere with another carrier.

For copper cables interference can occur due to lightening where it produces many electromagnetic frequencies.

For fibre optic transmission electromagnetic interference does not occur since it operates at very high frequencies.

Transmission Media

Let us see the characteristics of different transmission media. It is important to study and decide the most cost effective transmission media when designing computer networks.

Transmission media can be mainly divided into two categories.

- Guided transmission media
- Unguided transmission media

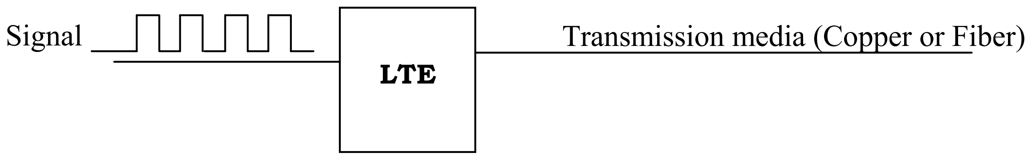
Guided Media
- Copper Cables
- Fiber Optic Cables

Unguided Media
- Domestic Radio
- Satellite Communication
- Mobile Radio

Guided Media

A B

It is a point-to-point communication. The signal can be transmitted without changing the frequencies. These signals normally cannot be interfered with other signals. Only the line coding should be done which we will study later.



Line Terminal Equipment

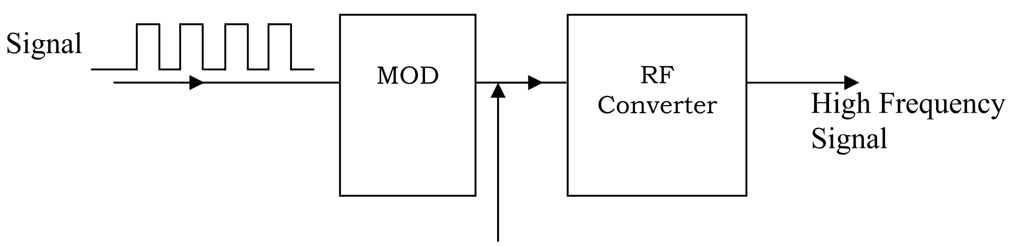
Functions of LTE are,

- Line coding
- Add overhead bits for supervisory purposes
- Power feeding for repeaters [for copper cable]
- Electrical to optical conversion [for fibre optic]

At the receive end LTE will do the reverse functions of the transmit end LTE.

Unguided Media

The signal is transmitted into free space. Therefore, each signal should operate with a unique frequency. If two signals have the same frequency, then those two signals can interfere. [Just like the flying of airplanes. They should fly at different heights, if not they can collide]. Therefore, the original signal should be converted to a unique high frequency. This is done by modulation.



Modulation converts the original signal to an Intermediate Frequency [IF]. This is normally a fixed frequency [e.g. 70 MHz]. Then the RF converter [RF – Radio Frequency, normally this is the term used for high frequencies] converts the IF signal to the required frequency.

Important : Please note that modulation is needed only for radio transmission. For line [copper or fibre], transmission, modulation is not necessary.

Copper Cables

Copper cables are used for different purposes.

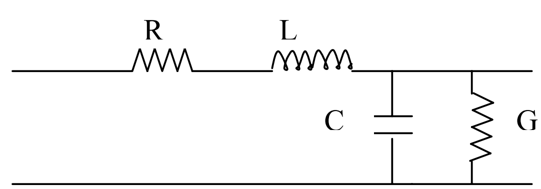
- For voice communication in telecommunication systems.
  [Exchange to DP, DP to home]. The DP to home copper cables are called Aerial Cable.

- For multichannel [high bandwidth] signal transmission. These are called, Coaxial Cables.

- For data transmission, Unshielded Twisted Pair [UTP] or Shielded Twisted Pair [STP] is used.

Common Characteristics of a Copper Cable

A copper cable pair has the resistive, capacitive, inductive and conductive effect and it can be represented as follows.

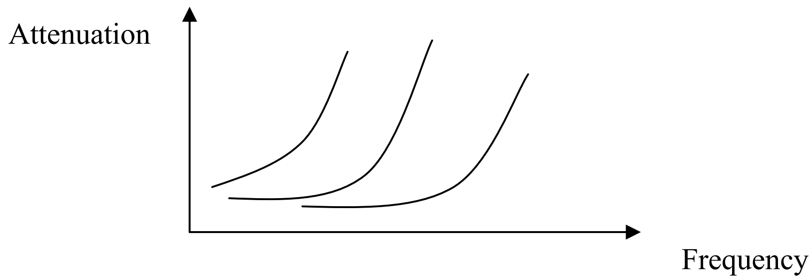


R - Resistance
L - Inductance
C - Capacitance
G - Conductance

The capacitor has high impedance at low frequencies and the inductor has high impedance at high frequencies. Therefore, capacitances and inductors can be used as frequency filters. Since the cable acts as a capacitor and inductor it filters some frequencies. Therefore, the transmit end and the receive end Amplitude- frequency characteristics can have a difference, as shown in the following figure.

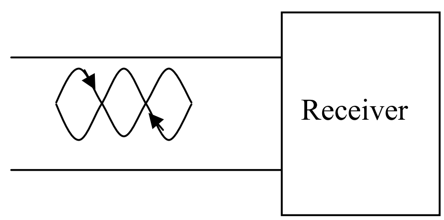
This effect is called amplitude distortion. This can be corrected by using amplitude equalizers. This does not have an effect for short distances.

Attenuation VS Frequency Characteristic



Attenuation increases with the frequency. The above figure shows the characteristic of three different copper cables. Therefore, we should select a copper cable which has low attenuation for the whole required signal bandwidth.

Reflection



We need two cables [a pair] for TX and two cables [a pair] for RX.

Part of the signal goes to the receiver and the remaining part is reflected at the receiver. These two signals are called the incident signal and the reflected signal respectively.

Reflection is an unnecessary occurrence. We need to send the whole signal to the receiver. Since part of the signal energy is reflected, it can be considered as loss of signal energy.

The characteristic Impedance [Zo] of a cable is defined as,

$$Z_0 = \frac{R+j\omega C}{G+j\omega C}$$

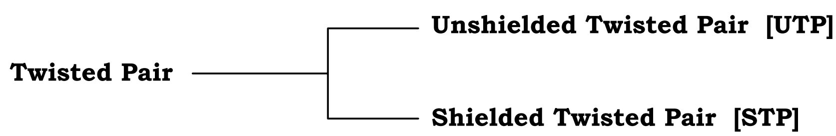
It can be shown that if the receiver input impedance is equal to the characteristic impedance of the cable then there will be no reflection. This condition is called "matched condition" The standard characteristic impedance of cables are ohms 50, 75, 120, 300 etc.

Note : The above characteristics are true for any metallic cable , not only for copper cables.

Types of Copper Cables used in data networks

Twisted Pair Cable

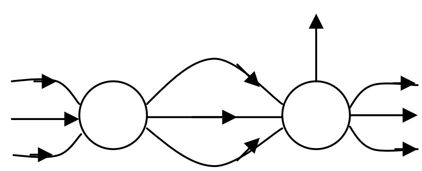
A twisted pair consists of two insulated copper wires. These wires are twisted together in a helical form. This twisted form is used to reduce cross talk. [electrical interference of adjacent channels/cable pairs].



If an individual pair has a metallic shield it is called a STP Cable.

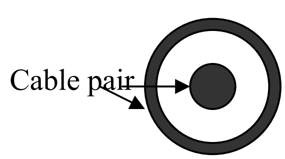
But now high quality UTP Cables are produced which can carry 10 Mb/s, 100 Mb/s and even 1000 Mb/s [1Gb/s]. Therefore, present Computer Networks [LAN] use UTP cables and there are many categories called Cat5, Cat5e and Cat6 where the standards are defined by EIA/TIA standards body.

Coaxial Cable



The above figure shows the magnetic flux pattern of a current [signal] carrying cable pair. You can see that some flux are going to free space and some magnetic energy is lost by the cables.

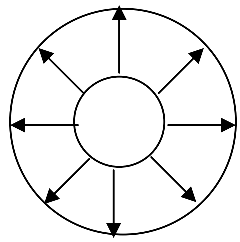
If the two cables are arranged in the following manner, there will be no such loss of energy.



In between the cables, there is an insulator.

Since both cables have the same axis, this is called a Coaxial Cable.

The magnetic flux pattern is as follows.



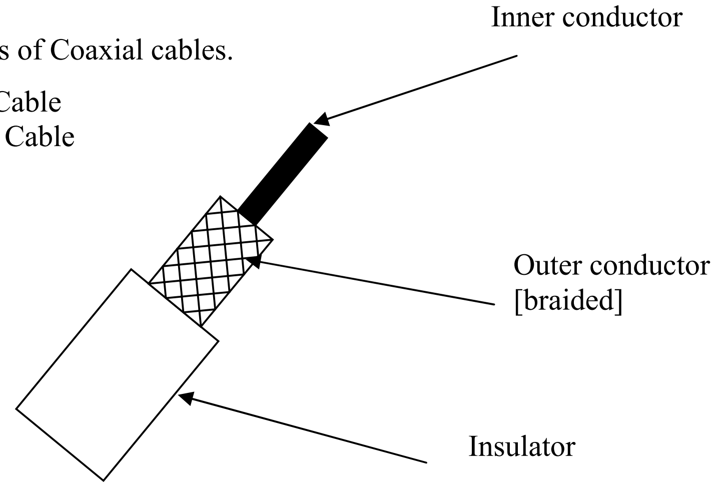
There is no loss of energy.

Therefore, Coaxial Cables can be used for long distance transmission.

A typical Coaxial Cable is given in the figure.
Normally the outer conductor is braided copper.

There are two types of Coaxial cables.

- Thin Coaxial Cable
- Thick Coaxial Cable



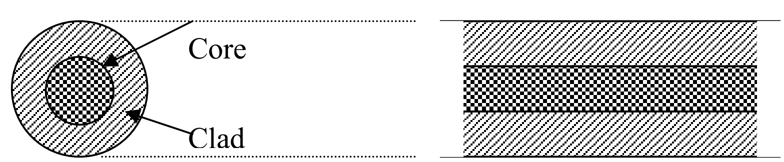
Fibre Optics

Signals can be transmitted as optical signals. For this purpose a fibre optic cable can be used.

Some of the advantages of optical fibre are,

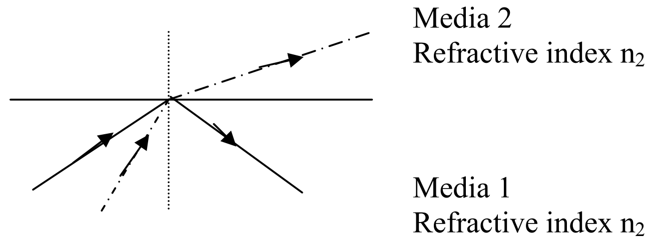
- The information carrying capacity is high. [That means it has a greater bandwidth]
- Not electrically conductive, therefore no interference from electrical signals.
- Less attenuation, therefore signal can travel a long distance without repeaters.

The fibre optic cable consists of two parts. The inner fibre [core] and the outer fibre [clad].



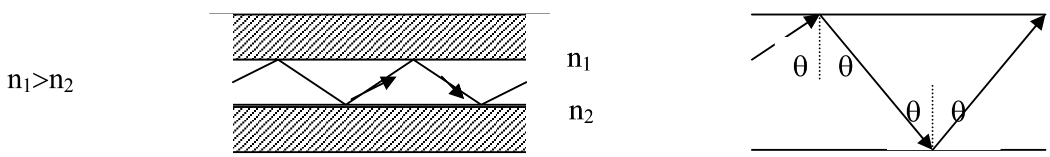
The Core and Clad can be glass or plastic.

Principles of light transmission in a fibre



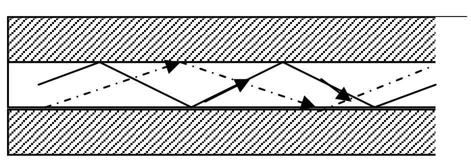
If the incident angle is less than the critical angle, the light ray is refracted to media 2.

If the incident angle is greater than the critical angle, the incident light ray reflects back to the same media [media 1]. This is called the total internal reflection.



The incident angle to the core is θ where θ > critical angle. Therefore, it is reflected back to the same media. Again it is reflected from the opposite surface in a similar manner. Hence the light ray goes through the core in a zigzag path.

Depending on different incident angles of light rays, they can travel in different paths.



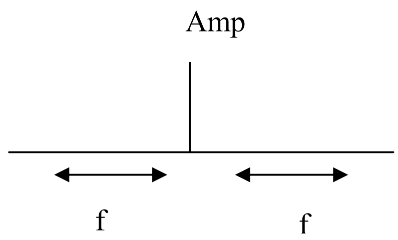
These are called the different modes of optical ray. If the radius increases, many modes of signals [light rays] can travel through the core.

If only one ray which goes through the axis is allowed by the fibre, it is called Single Mode Fiber.

If many rays are allowed, it is called multimode fibre.

	Single mode	Multimode
Core diameter (μm)	8	50/62.5
Clad diameter (μm)	125	125

Why high bandwidth?



If the carrier frequency is f, the theoretical possible bandwidth is, left side f Hz and for symmetry right side is also f. [i.e. 2f Hz]

The light rays operate at 10^{14} Hz.
Therefore, the possible bandwidth is 2 x 10^{14} Hz = 200 THz.
This is a very high bandwidth.

Attenuation Characteristics

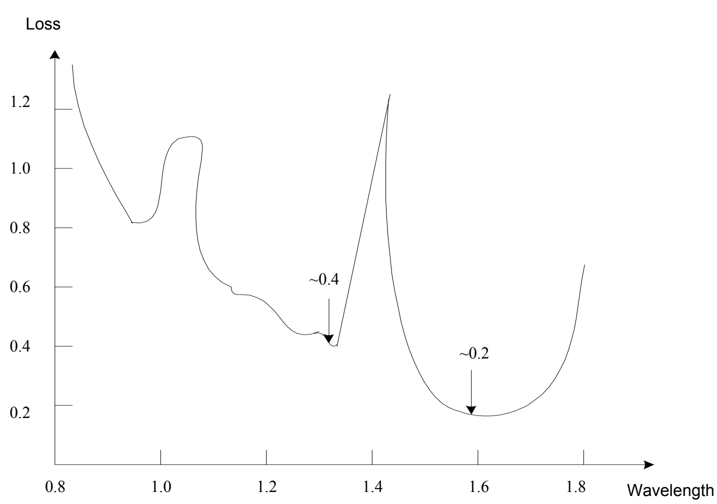
Material absorption Loss

The energy of the signal is absorbed by the water contamination and by the ion impurities.

Scattering Loss

During the glass forming process there can be a density variation of core and clad.
This will result in scattering of portion of the light passing through the core.

Attenuation Characteristic



By considering the attenuation characteristics, three operating wavelengths are selected.

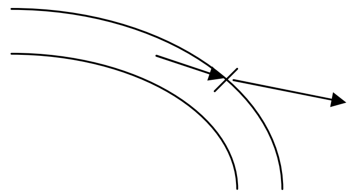
800nm, 1300nm, 1550nm

Normally used wavelength are 1300nm or 1550nm.

Other losses

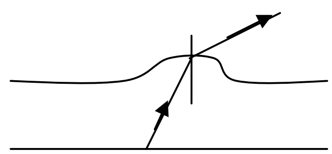
Bending losses - There are two types of bending losses. They are constant radius bending and micro bending.

Constant radius bending



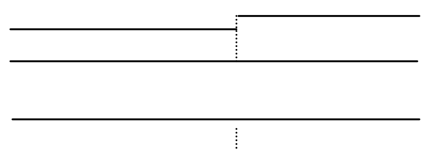
At the time of installation the fibre may have bends. The incident angle at a bend can be less than the critical angle and part of the signal can be refracted and it will cause a signal loss.

Microscopic bending



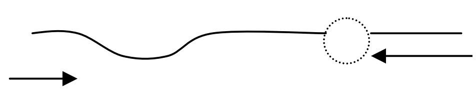
It is a microscopic bending of the core of the fibre that results at the time of manufacturing. This may change the incident angle, part of the signal is refracted and will cause a signal loss.

Coupling losses - Imperfectly formed splices or imperfectly aligned connectors.



This will cause to reflect back part of the signal at a joint or at a connector.

Another type of loss can occur at the core, clad interface due to imperfections such as small variation in the core diameter or air bubbles in the glass.



The incident angle may be changed and part of the signal can be refracted.

Dispersion

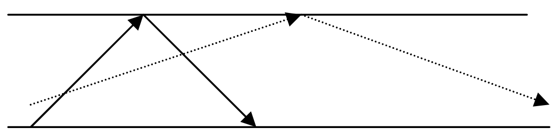
The pulse is broadened due to dispersion.
Δt - Dispersion time.

The possible maximum bit rate R = $\dfrac{1}{t+\Delta t}$

When Δt increases, R decreases.
Higher the dispersion, lower the pulse (bit) rate.
Therefore, the bit rate is limited due to dispersion.

There are different types of dispersions.

Intermodal Dispersion



The different modes of signals have different incident angles and they travel different distances and the pulse gets broadened at the receive end. This is called Intermodal Dispersion.

Since the single mode fibre has only one mode, it has only one incident angle. Therefore, there is no intermodal dispersion for single mode fibres.

Material Dispersion

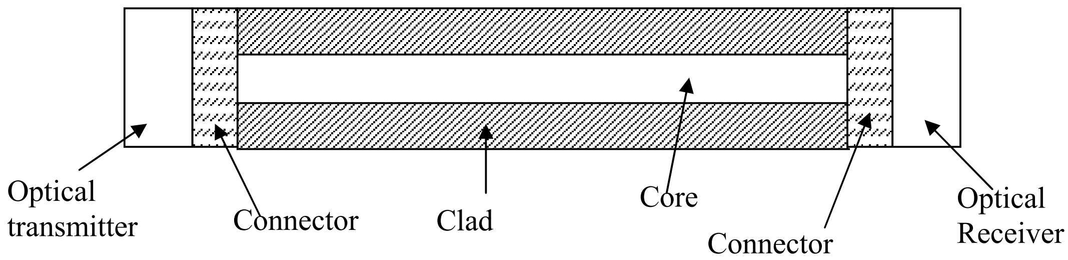
The refractive index of glass depends on the wavelength of the optical signal. Again the speed of the signal depends on the refractive index.

Normally the optical ray is not a single wavelength. It has several different wavelength and they travel at different speeds. Therefore, the signal is subjected to dispersion and it is called material dispersion.

Wave guide Dispersion

The plane of polarization of the signal can vary with the time and it affects the speed. Therefore, the signal is subjected to dispersion. This is called wave-guide dispersion.

Optical System



Optical Transmitter

The optical transmitter converts the signal from electrical energy to optical energy. The typical optical transmitters are,

- Light Emitting Diode [LED]
- Semiconductor Laser Diode [SLD]

Optical Receiver

The optical receiver converts the signal from optical energy to electrical energy. The typical optical receivers are,

- p-*i*-n photo diode
- Avalanche photo diode

Connectors

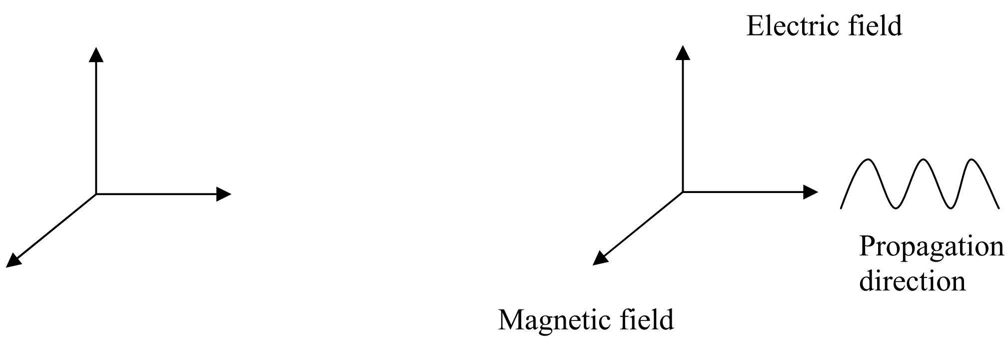
There are two types of connectors.

- ST [bayonet]
- SMA [threaded]

Radio Transmission

A radio signal is an electromagnetic wave, which travels through free space [unguided media].

Electromagnetic Wave



If a signal has an electric field and a magnetic field together and if their strengths are varying with time, it can be shown mathematically that the energy propagates [travels] as shown in the above figure.

Note : If the electric field and magnetic filed are orthogonal [90^0 apart] then the energy [signal] travels through the other orthogonal axis as shown in the figure. Such an electro magnetic wave is called a Transverse Electromagnetic wave. But this is not a mandatory requirement. That means it is not necessary for electric field and magnetic field to be orthogonal.

The signals travelling through a copper cable or a fiber optic cable are also electro magnetic waves.

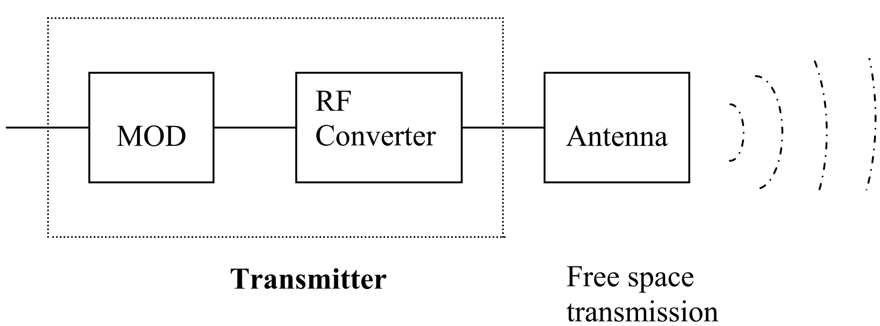
When a radio signal travels through free space its electric and magnetic fluxes can be subjected to changes due to other influences. Also they can be subjected to reflection refraction etc. In some waves these changes cause considerable influence but in some it does not. This mainly depends on the frequency of the wave. Therefore, the radio frequency spectrum [range] is divided in to different ranges and categorized with different names.

Category	***Abbreviation***	***Frequency Band***
Very Low Frequency	VLF	3-30 KHz
Low Frequency	LF	30-300 KHz
Medium Frequency	MF	300-3000 KHz
High Frequency	HF	3-30 KHz
Very High Frequency	VHF	30-300 KHz
Ultra High Frequency	UHF	300-3000 KHz
Super High Frequency	SHF	3-30 GHz
Extra High Frequency	EHF	30-300 GHz
Optical	Optical	100 THz range

Velocity of Electromagnetic Wave

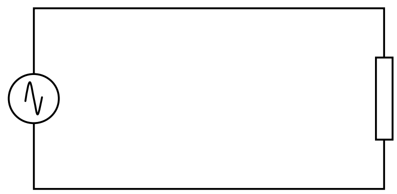
The velocity in the free space and fibre are about $3x10^8$ m/s. The velocity in the copper is about $2x10^8$ m/s.

Radio System

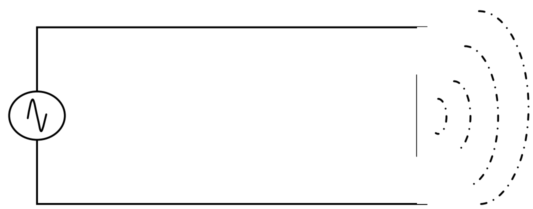


The main components of any radio system is given in the figure.

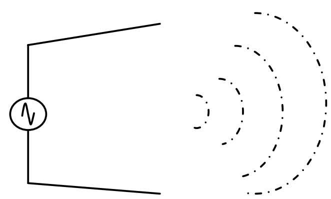
Antenna



Suppose a signal source is connected to a load by using two wires. Suppose the load resistor is removed. Now how can the signal travel?

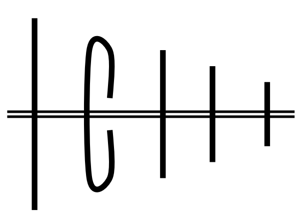


Now the signal travels to free space.

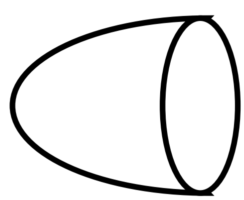


By changing the shape of the two-wires the amount of wave [energy] and the directions etc. can be changed. This is the simplest antenna. There are different antennas used for different applications/different frequencies.

Yagi Antenna

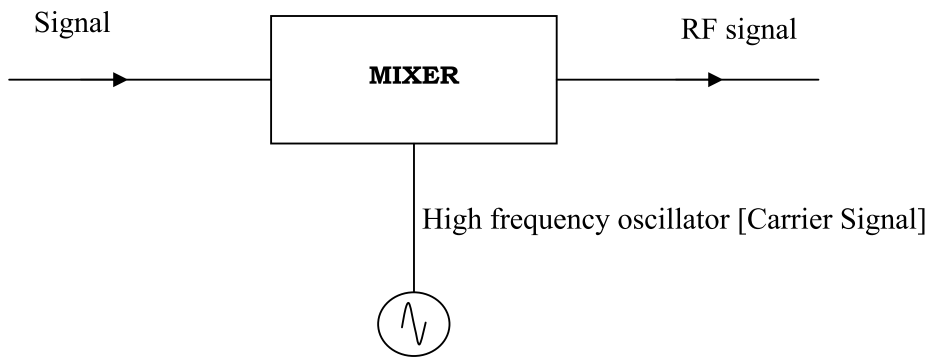


Parabolic



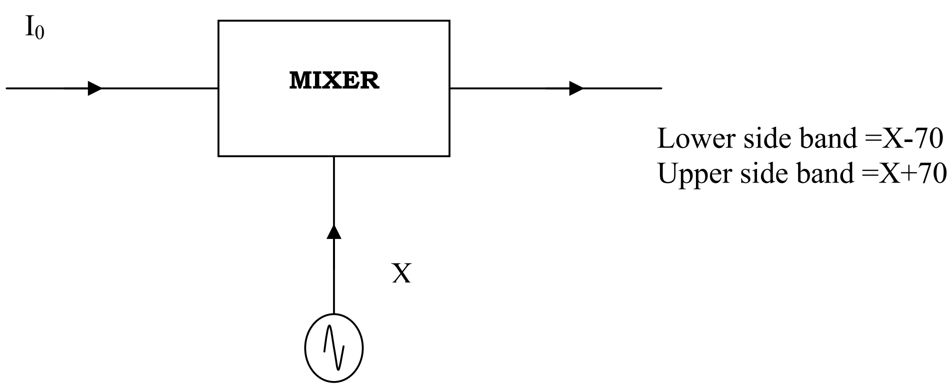
Omni-directional Antenna

RF Converter [Up Converter]



The mixer is an analog modulator [e.g. AM modulator]. The modulator output is a fixed frequency [IF]. E.g. 70 MHz.

Suppose we want to convert the IF signal to 2.4 GHz [2400 MHz].



Both side bands exist at the output. By using a filter we can select either. Suppose we selected the LSB.

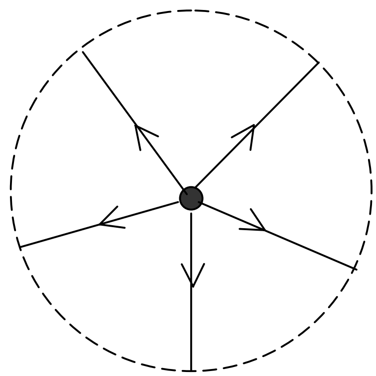
Then $X - 70 = 2400$

$X = 2470$ MHz.

Then we have to adjust the carrier signal to 2470 MHz. In a Similar way by adjusting the carrier signal the required output frequency can be obtained.

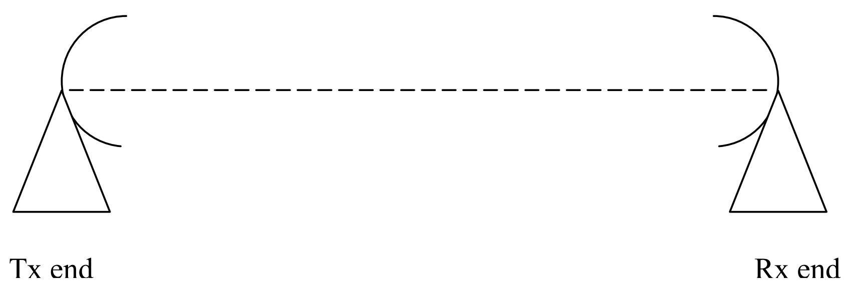
Signal Radiation

Pattern

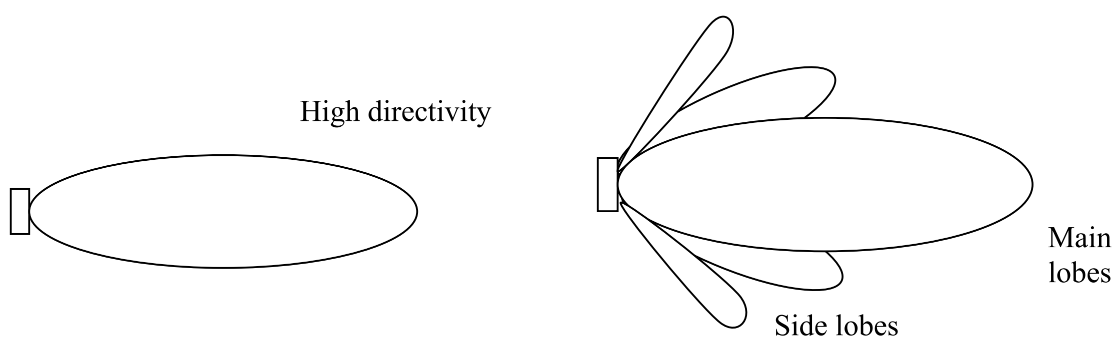


If the signal is transmitted to the free space without an antenna the signal will travel in all directions.

But our requirement is to transmit the signal to a particular destination or area. For this purpose an antenna is required.

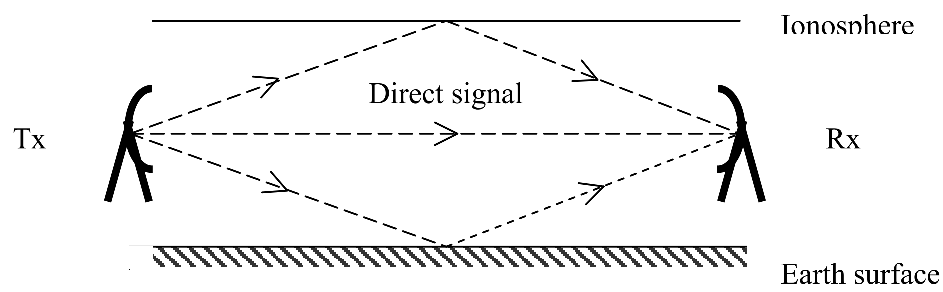


The pattern of the signal propagation is called the radiation pattern.



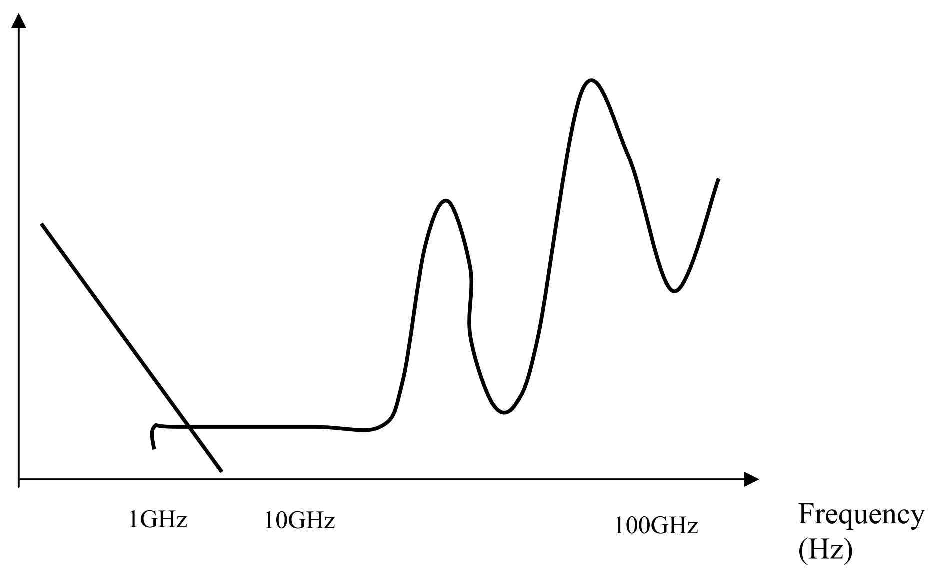
The radiation pattern depends on the type of antenna and the frequency of the signal. Higher the frequency, higher the directivity.

Reflection



The upper surface of the atmosphere is called the ionosphere. The signal can be reflected from both the earth surface and the ionosphere. The receiver output will be a mixture of all these signals. The resultant signal may have less strength than the direct signal or vise versa. This badly affects microwaves.

Effect of Noise



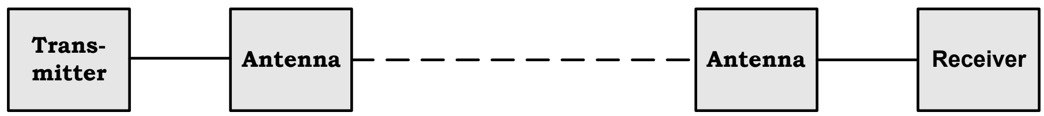
This has the biggest effect for radio signals. It can be seen that the best operating frequency range is 1 GHz – 10 GHz. This is also called the Radio Window. Therefore, most of the applications use this frequency range.

E.g.	Mobile	1-2 GHz
	Satellite	4-6 GHz
	Wireless LAN	2.4 GHz and 5 GHz.

Note: For different bands different letters are used.

S band
C band

Typical Radio System



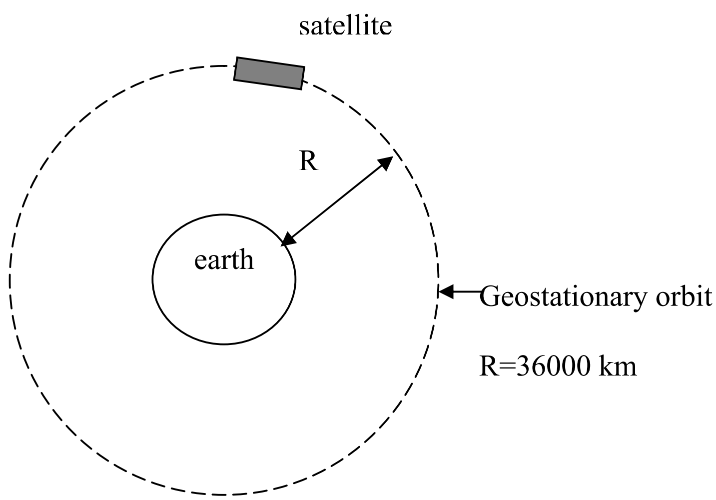
Receiver performs the reverse function of the Transmitter. If the transmitter and receiver are in one unit it is called a Transceiver.

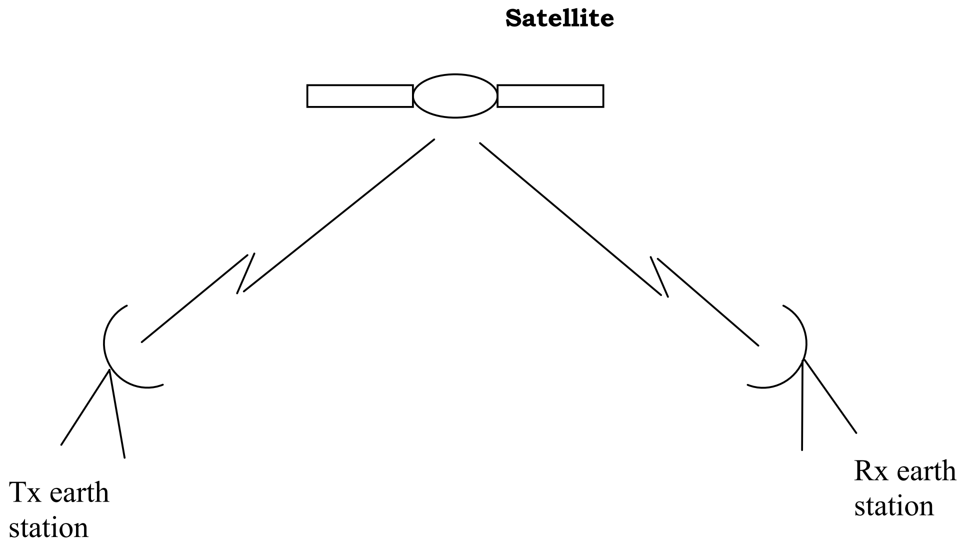
Applications of frequency bands

The following are some of the applications of different frequency bands.

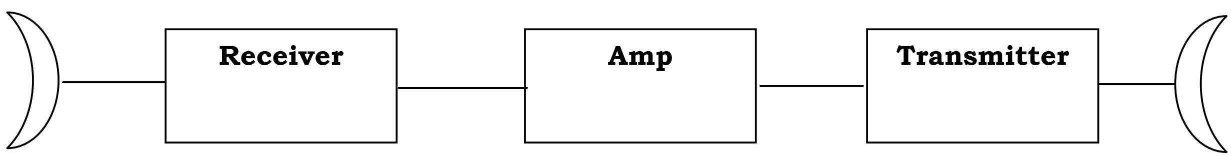
VLF	-	Telegraph transmission for navigation.
LF	-	Sound broadcast through earth surface. It can travel about 1500 km.
MF, HF	-	Commercial radio broadcasting.
VHF,UHF	-	TV broadcasting
Microwaves	-	Domestic Carrier [wide band] transmission by Telcos. Satellite Communication

Satellite Communication





For long distance communication satellite communication can be used. The main function of the satellite is to receive the signal, amplify and transmit back to earth.



There are two frequency bands used for satellite communication.

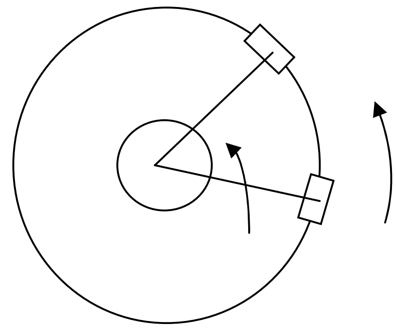
C-band	Tx	-	5.925	-	6.425	GHz
	Rx	-	3.7	-	4.2	GHz

Ku-band	Tx	-	14 GHz range
	Rx	-	11/12 GHz range

C-band has less effect on attenuation and noise compared to ku-band.

Satellite Orbit

A satellite can travel in any orbit around the earth. But the most useful orbit is the Geostationary Orbit where it takes 24 hrs. to travel one orbit in the same direction as the earth's travel. Therefore, both travel the same angle for a given time.



Hence the satellite is stationary with respect to the earth. This orbit is called the Geostationary orbit and it is located at about 3600 km away from the earth surface.

Signal travels from earth to the satellite and comes back from the satellite to earth. This is called one hop.

One hop travels 2 x 3600 = 7200 km.

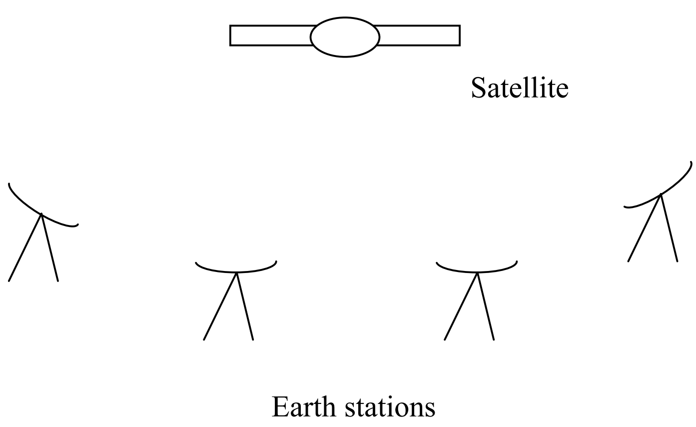
The velocity of the signal is = 3×10^8 m/s

Therefore, the time taken to travel one hop is =

$$\frac{7200 \times 10^5}{3 \times 10^8} \, Seconds$$

The propagation delay of one satellite hop is approximately 250 ms. This causes a considerable effect on data communication.

Satellite Access Methods



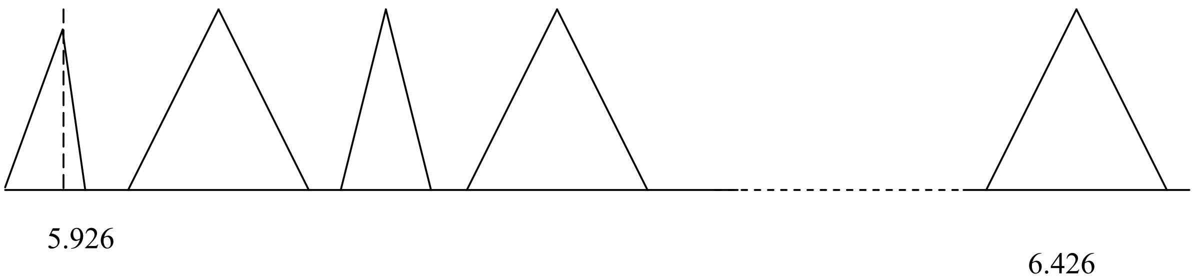
There are three satellite access methods.

- Frequency Division Multiple Access [FDMA]
- Time Division Multiple Access [TDMA]
- Code Division Multiple Access [CDMA]

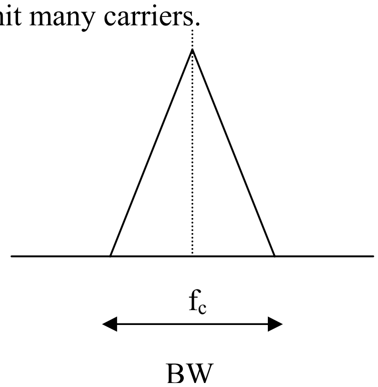
FDMA

Consider the c-band where the Tx frequency range is 5.925 – 6.425 GHz.

There is a 500 MHz band.



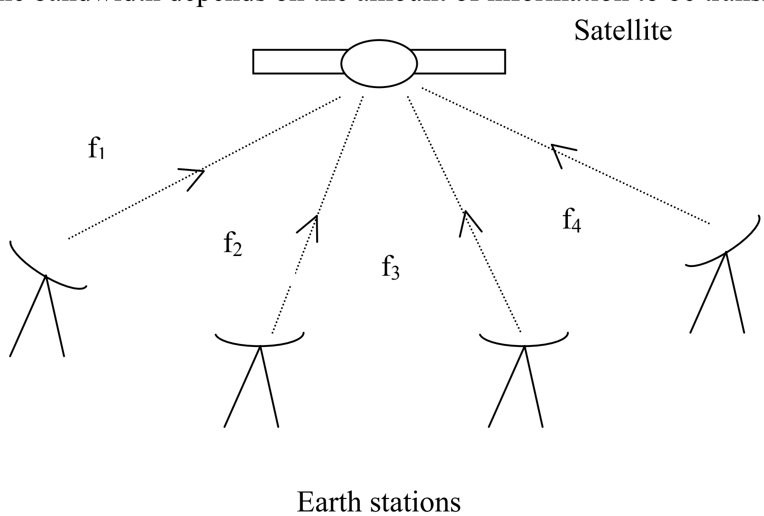
Different frequencies are allocated for different carriers [earth station]. One earth station can transmit many carriers.



Fc	-	Centre Frequency
BW	-	bandwidth

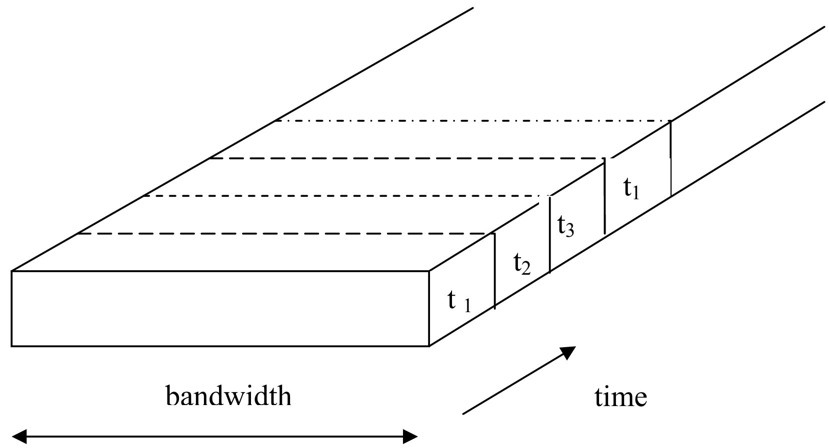
e.g. fc = 6000 MHz.
BW = 5 MHz.

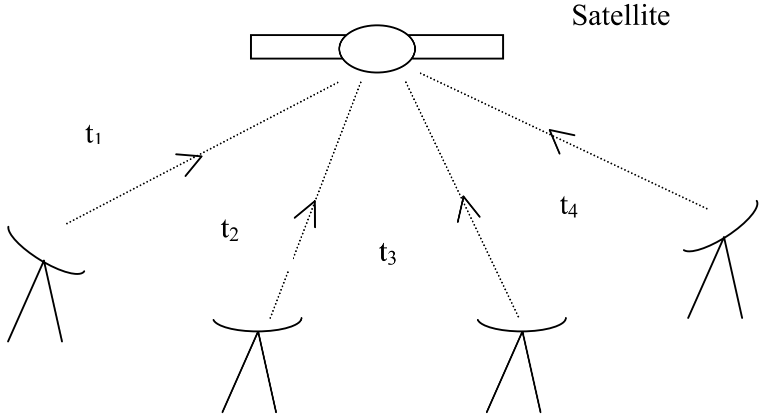
The bandwidth depends on the amount of information to be transmitted.



There is no information of signals at the satellite or free space. Since they operate at different frequencies.

TDMA





Earth station

In this method the whole bandwidth is used by all carriers. But they do not transmit continuously as FDMA.

Carrier 1	transmits	-	t_1 ms
Carrier 2	transmits	-	t_2 ms
Carrier 3	transmits	-	t_3 ms
Carrier 4	transmits	-	t_4 ms

Again

Carrier 1	transmits	-	t_1 ms
Carrier 2	transmits	-	t_2 ms

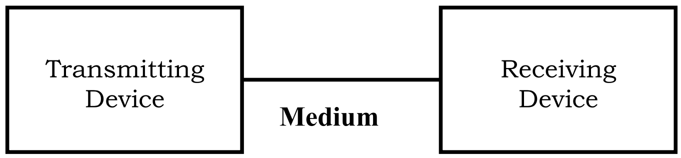
And this pattern is repeated.

Very Small Aperture Terminal [VSAT]

Normally very large antennas such as 18 m diameter antennas are used for satellite communication. These antennas have a very large aperture. But if we transmit a carrier of about 512 kb/s, such a big antenna is not economical. Therefore, especially for data communication or low information transmission, specially designed earth stations are used. They use very small aperture antennas such as 5 m diameter. Such earth stations are called VSATS. The operation of a VSAT is similar to a normal earth station.

Data Communication - Fundamentals

Communication Model



To transmit information between two locations, it is necessary to have a transmitter, receiver and a transmission medium which provides the connection.

Line Connections

Dedicated Line

If the transmitter and the receiver are in the same premises they can be connected by using a piece of cable. E.g. Local Area Network [LAN]. This is called a dedicated line.

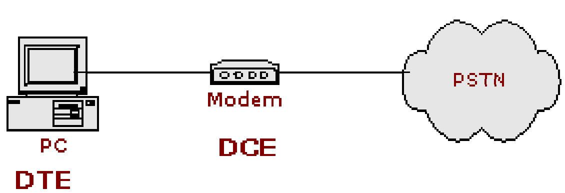
Permanent Line

If the transmitter and receiver are located very far [E.g. transmitter in Colombo and receiver in Kandy], a physical connection by a cable is not possible. If a permanent connection is needed, in such cases the connection can be obtained from a Public Data Network . This type of line is called a permanent Line.

Switched Line

If there is no necessity to have a permanent connection between the transmitter and the receiver, a cost effective solution is to establish the connection whenever necessary. This is called a Switched Line. E.g. A connection via Public Switched Telephone Network [PSTN].

DTE/DCE



Data Terminal Equipment [DTE]

The terminal equipment [PC or Dumb terminal] connected to a network is called Data Terminal Equipment [DTE].

Data Communication Equipment [DCE]

The interface connected to the DTE and network is called DCE. This is also called Data Circuit Terminating Equipment.

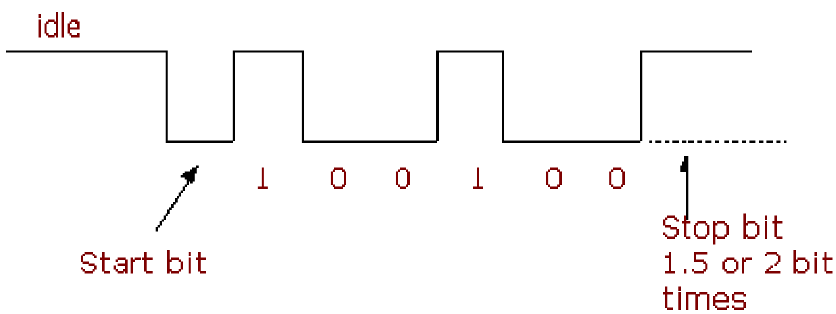
Transmission Modes

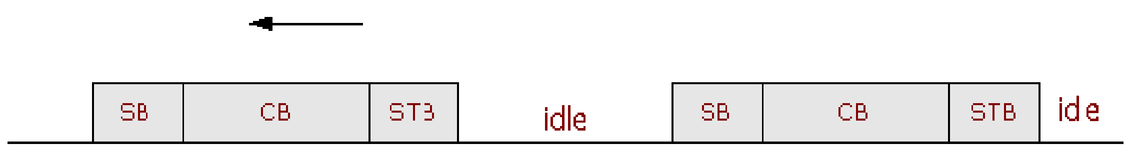
There are two transmission modes.

- Asynchronous Transmission
- Synchronous Transmission

Asynchronous Transmission

Data is not transmitted continuously. A character can be represented by a group of bits. [E.g. 8 bits] Each set is sent with a start bit and a stop bit.





SB - Start Bit
STB - Stop Bit
CB - Character Bit

Characters are terminated intermittently.

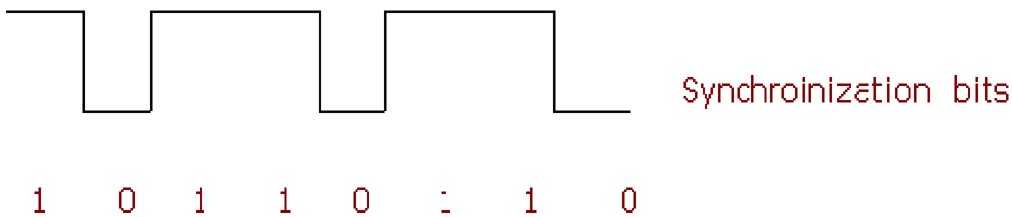
At the idle condition the line voltage can be positive. When the start bit is received by the Receiver it detects the start bit by changing the voltage to zero level. The end of bits is detected by a stop bit which has longer bit duration.

Synchronous Transmission

Data is transmitted continuously as frames.

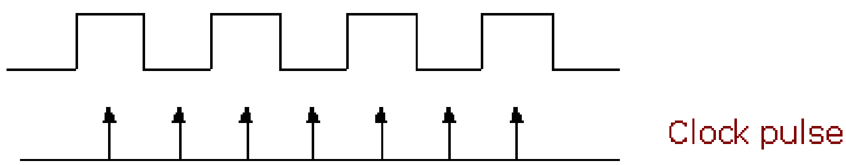
Frame

A frame consists of synchronization bits and data bits.



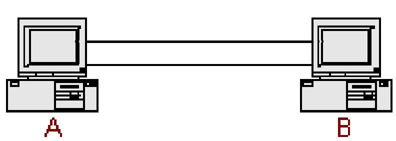
Synchronization bits are a predefined bit pattern. The receiver scans all received bits until it receives the synchronization bits. After it detects the SYN bits, the receiver knows that the next fixed number of bits are data bits.

In order to detect a bit a clock pulse is used.



Note : *Clock pulses are a continuous pulse stream which has a bit rate equivalent to data bit rate and a very small pulse width. This can be generated separately or from the incoming data bits. If it is generated from the incoming bit stream it is called clock recovery.*

Transmission Techniques



Simplex - Transmit in only one direction.

→ Data Flow

2 wire circuit

Half Duplex - Transmits in both directions. But not simultaneously. [2 wire circuit]

→ Data Flow
← (Not at the same time)

A to B transmits and stop, then B to A transmits.

2 wire circuit

Full Duplex -

↔ Data Flow

A to B and B to A transmit simultaneously.
This is normally a 4 wire circuit.

(A → B 2 wire, A ← B 2wire)

Note* : *By using Frequency Division Multiplexing, this can be done using two wires.

Base band

The whole bandwidth of the cable is occupied by the signal.

Broad Band

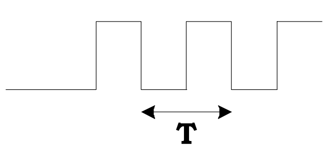
The bandwidth of the cable is used by more than one multiplexed signal.

Data Encoding

Data encoding means the bit pattern is modified for better clock recovery and less attenuation. There are different types of line codes such as AMI, Manchester 2 BIQ etc.

In order to understand line codes, we should be familiar with the following terms.

Cell or Unit Interval [UI]



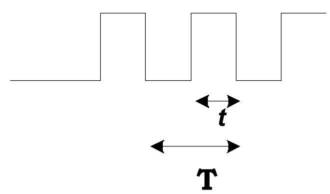
The time allocated for one bit is called the Unit Interval. That is the period of the pulse train.

Unit Interval = T

Bit Rate

Number of bits per second. Bit Rate = $\dfrac{1}{\mathrm{T}}$

Duty Cycle



Duty Cycle = $\dfrac{t}{\mathrm{T}} \times 100\%$

<u>Unipolar Signals</u>

Only one potential relative to ground level is available.

There are two types of unipolar signals.

(i) Non Return to Zero [NRZ]

(ii) Return to Zero [RZ]

NRZ signals have a 100% duty cycle.
RZ signals have less than 100% duty cycle.

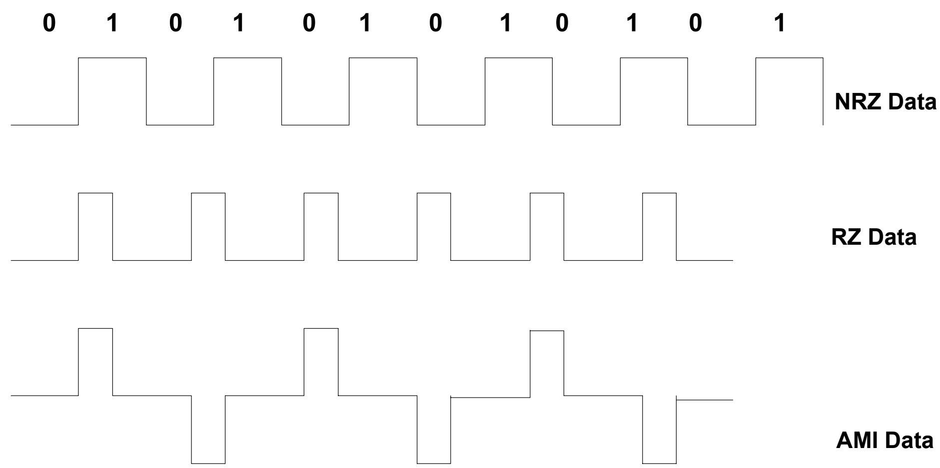
<u>Types of Line Codes</u>

Alternate Mark Inversion [AMI]

Properties of the AMI Code

(i) Zeros are transmitted as zeros without any change.

(ii) Alternative marks are inverted to opposite polarity.



Manchester Coding

1 - Low to High signal [Transition occurs at the center]

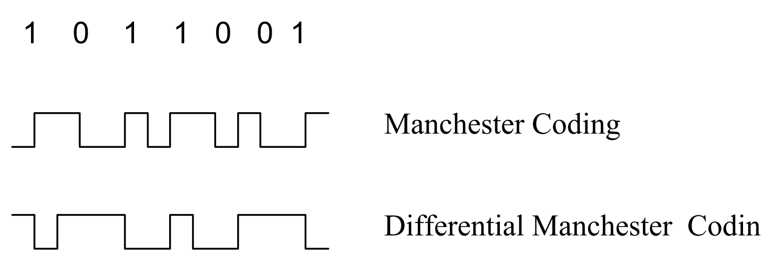
2 - High to Low signal [Transition occurs at the center]

Differential Manchester Coding

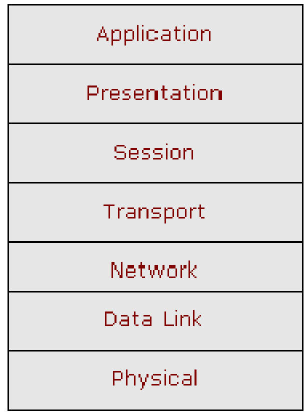
0 - Transition occurs at the start of the bit

1 - Transition occurs at the middle of the bit

Transition means, if it changes the current state irrespective of 1 or 0.



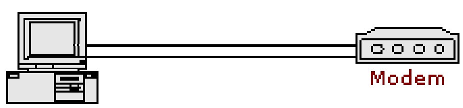
ISO-OSI Seven Layers



The International Standards Organization [ISO] defined Open System Interconnection [OSI] Seven Layers.

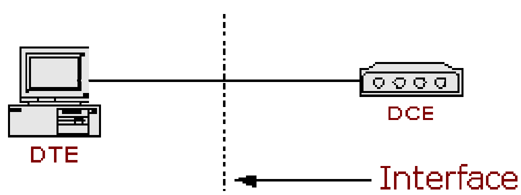
What is the use of this ?

Suppose we purchased a PC from one vender and a Modem from another vender.



When we try to connect them, the computer connector may not be compatible with the modem connector. If they were manufactured according to the ISO-OSI standards this problem will not arise.

Physical Layer



The physical layer defines the DTE/DCE interface standards. It defines the following characteristics.

Mechanical Characteristics

- The shape of the connector.
- Number of pins.
- The diameter of a pin.
- The distance between two pins etc.

Electrical Characteristics

The voltage levels used.
e.g. '1' - 5V
0 - 0 V

Both the DTE and the DCE should use the same electrical signal levels.

Functional Characteristics

Function of each pin should be defined.

Example :

- Pin 1 - Ground
- Pin 2 - DTE TX
- Pin 3 - DTE RX

Procedural Characteristics

The procedure of communication between DTE and DCE.
E.g. DTE requests from DCE to send data.
Then DCE says OK etc.

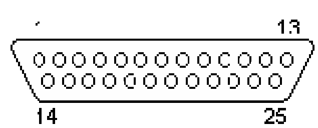
The ISO-OSI defines the basic features of the standards. The other industrial or service oriented associations/institutions introduce the interfaces according to the ISO-OSI recommendations. The two main associations/institutions that introduce such interfaces are,

- Electrical Industrial Association [EIA]
- International Telecommunication Union [ITU-T]
  E.g. EIA introduced RS-232D interface standard
  ITU-T introduced the V.35, X.21, I.440 etc.

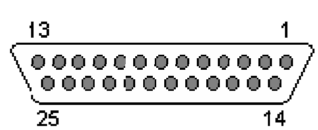
RS232D

What are the mechanical characteristics ?

Connector is named as DB25.



25 pin d-sub Male [At the DTE]



25 pin d-sub Female [At the DCE]

25 PIN D-SUB MALE at the DTE [Computer].
25 PIN D-SUB FEMALE at the DCE [Modem].

Functional Characteristics

Pin	Name	I/O	Description
1	GND	-	Shield Ground
2	TXD	O →	Transmit Data
3	RXD	I ←	Receive Data
4	RTS	O →	Request to Send
5	CTS	I ←	Clear to Send
6	DSR	I ←	Data Set Ready
7	GND	-	System Ground
8	CD	I ←	Carrier Detect
9	-	-	RESERVED
10	-	-	RESERVED
11	STF	O →	Select Transmit Channel
12	S.CD	I ←	Secondary Carrier Detect
13	S.CTS	I ←	Secondary Clear to Send
14	S.TXD	O →	Secondary Transmit Data
15	TCK	I ←	Transmission Signal Element Timing
16	S.RXD	I ←	Secondary Receive Data
17	RCK	I ←	Receiver Signal Element Timing
18	LL	O →	Local Loop Control
19	S.RTS	O →	Secondary Request to Send
20	DTR	O →	Data Terminal Ready
21	RL	O →	Remote Loop Control
22	RI	I ←	Ring Indicator
23	DSR	O →	Data Signal Rate Selector
24	XCK	O →	Transmit Signal Element Timing
25	TI	I ←	Test Indicator

Note: *Do not connect SHIELD(1) to GND(7).*

Electrical Characteristics

+3V to +15V - ON or '1'

-3V to -15V - OFF or '0'

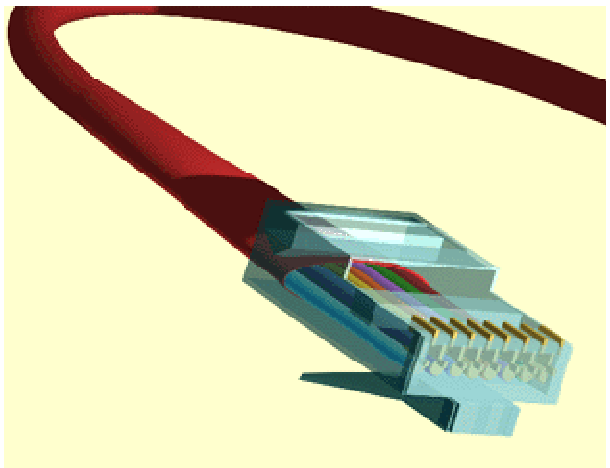
The voltage range from -3V to +3V is considered as the transition region, that has no effect upon the condition of the circuit.

Procedural Characteristics

The DTE needs to send data to the DCE.
The DTE activates the DTE Ready [DTR], Pin 20
The DCE responds by activating DCE ready Pin 6.
After some other steps the DTE Transmits Data [TXD], Pin 2 etc.
Similarly for any DTE-DCE interface above all are to be defined.
E.g. V.34 28.8 kbps modem interface.

RJ-45 Jack

Most DCEs are designed to connect to the PSTN by using RJ45 Jacks.



The RJ-45 Connector has 8 pins.
Pins 3,4 Tx from DTE to DCE
Pins 5,6 Rx to DTE from DCE

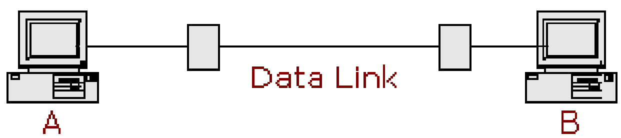
Wiring Diagrams for Straight Through, Cross Over Cables

Note: *The hook is underneath in all cases and Pin one is always on the Left.*

Straight Through Cable	Color Code
Pin 1	white orange
Pin 2	orange
Pin 3	white green
Pin 4	blue
Pin 5	white blue
Pin 6	green
Pin 7	white brown
Pin 8	brown

Cross Over Cable	Color Code
Pin 1	white green
Pin 2	green
Pin 3	white orange
Pin 4	blue
Pin 5	white blue
Pin 6	orange
Pin 7	white brown
Pin 8	brown

Data Link Layer



Suppose there is a physical connection between two computers. Computer **A** needs to transfer data to computer **B**.

Before sending data, computer **A** should establish a live connection with computer **B**. Then transfer the data. After completion of the sending data, the live link should be terminated. This is the responsibility of the Data Link Layer.

That is
- Establishment of the link
- Data transfer
- Termination of the link

When transferring data, it is to be made sure that all data will be transferred without any error.

How can we transfer the data without any error?

Detect the error, then correct the error.
i.e.
- Error detection
- Error correction

Error Detection

The simplest method of error detection is parity check. Suppose the frame size is 8 bits. 7 data bits and 1 parity bit.

0	1	1	0	0	1	1	P

Odd Parity - the parity bit (P) is set to 1 or 0 to make the total number of '1's to an odd number.
For above example P = 1

Even Parity - the parity bit (P) is set to 1 or 0 to make the total number of 1s to an even number.
For above example P = 0

At the receive end the total number of '1's in the frame is checked. If we use odd parity, the number of '1's should be an odd number. If not, there is an error. Then the same frame is requested from the transmit end.

Note **:** *This method is correct only for single bit errors.*

Cyclic Redundancy Check [CRC]

The data bit [n number] is divided by another number [m bits where m<n]

e.g. data - 1010001101
division - 110101
Modulo 2 arithmetic is used.

After dividing, the remainder can be obtained. This is called the ***"Frame Check Sequence (FCS)"***

For above example it can be shown that FCS = 01110
The data and FCS is transmitted together.
101000110101110

At the receive end it is divided by the same division (110101). If the remainder is zero, there was no error. If not there are errors and the same frame is requested from the transmit end.

Error Correction

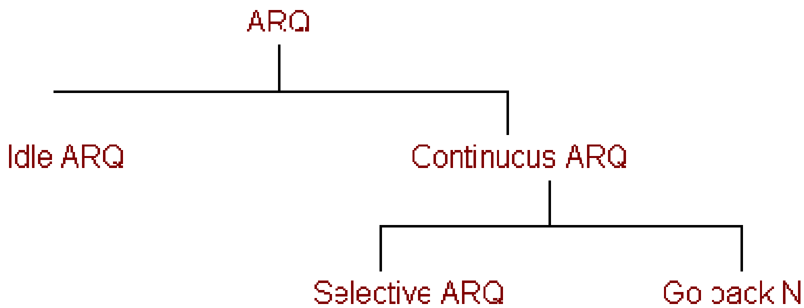
Forward Error Correction (FEC)

Detect the errors and correct the errors at the receive end.

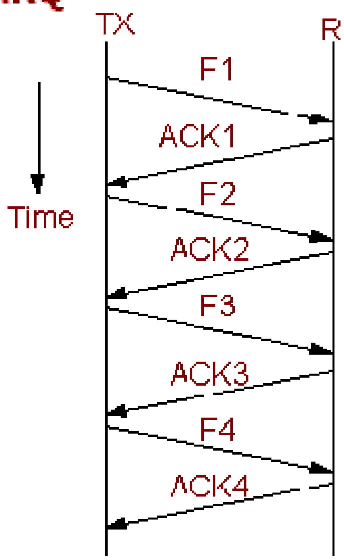
Backward Error Correction

Data is transferred by frames. If any error is detected, it is requested from the transmit end to retransmit the frame. Normally this process is done automatically. Therefore it is called the **A**utomatic **R**epeat re**Q**uest [ARQ].

ARQ methods



Idle ARQ



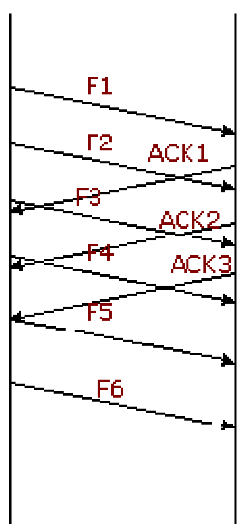
ACK - Acknowledge

NAK Negative Acknowledge (where an error is detected)

If NAK is received the same frame is retransmitted.

This is also called the stop and **wait method.**

Continuous ARQ



The frame 3 has errors and NAK3 was received.

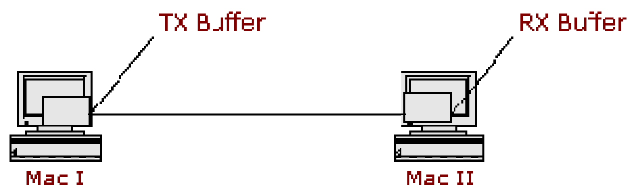
Selective Retransmission

- retransmit F3
- Next transmit F7, F8 etc.

Go back N

- retransmit F3
- Next transmit F4, F5 etc.

Flow Control



Because of error control, another problem arises. That is the frames are temporarily stored at the receive end, checked and if errors are detected request the frame from the transmit end. The Transmit end also keeps the frames in a temporary store, until the confirmation from the receive end is received. The temporary store is called the ***buffer*** and it has a limited memory space.

Suppose the receive buffer has only 5 frames space. If there are 5 frames already in the buffer and another frame is also received, it has no memory space in the buffer. Therefore the frame will be overflowed and the data is lost. In such a situation the receive end should inform the transmit end to suspend sending of frames. When the memory has free space, it is informed to the transmit end to resume sending of the frames.

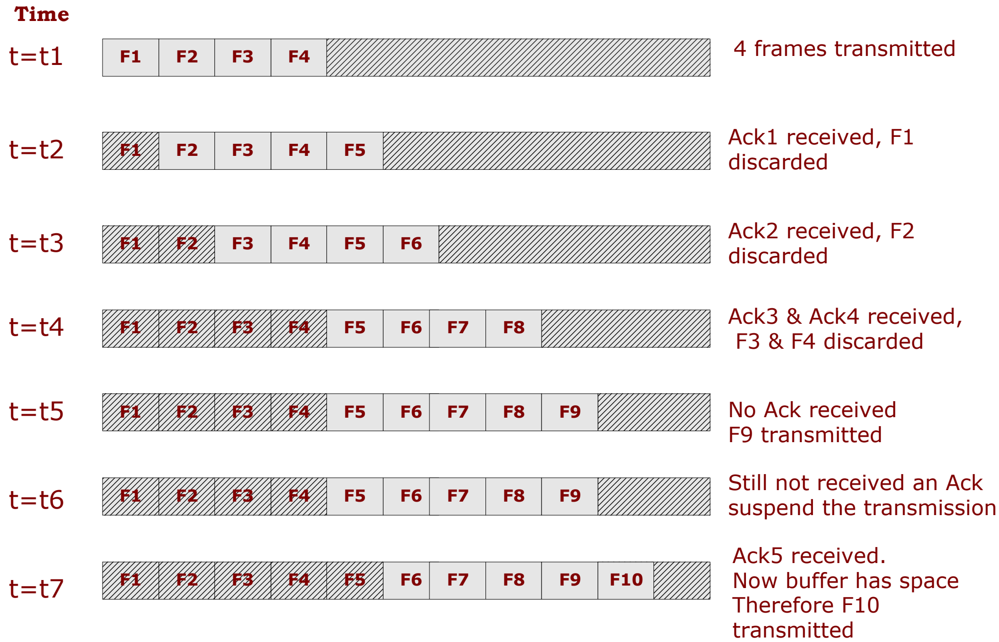
Similarly if the transmit buffer is full the transmission of frames is to be suspended until sufficient space is available in the transmit buffer.

This process is called ***flow control***. The most popular method for flow control is the sliding window mechanism.

Sliding Window Mechanism

Suppose the TX buffer size is 5 frames.

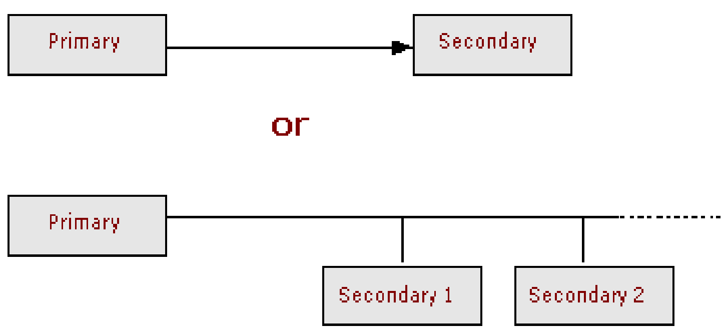
Consider the buffer content at different times.

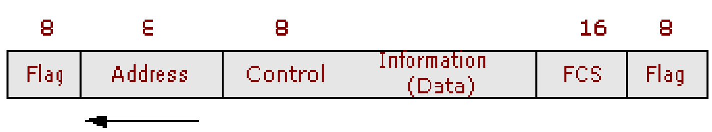


This mechanism is just like sliding a window. Therefore, it is called Sliding Window Mechanism.

High Level Data Link Control [HDLC]

This is a general frame format used in the Data Link Layer. This is a point to point link protocol.



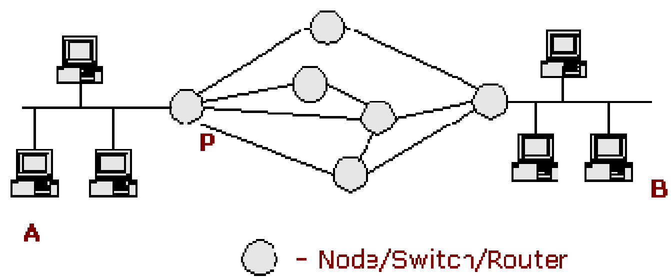


Flag - For synchronization
Address - Address of secondary. [if more than one secondary is available]
Control - For link establishment, termination, ACK, NAK and other control information.

Other Data Link Layer Protocols

LAP - B - X.25 DLL Protocols.
LAP - D - ISDN DLL Protocols.
Point-to-Point Protocols. [PPP].
Logical Link Control [LLC] and Media Access Control [MAC] Protocols for LANs.

Network Layer



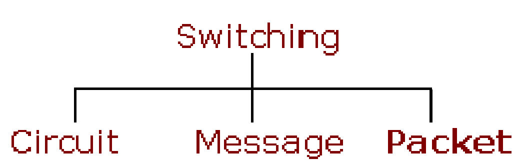
The above figure shows a switched network. Computer A needs to connect to computer B. When data goes to P, it has many routers and it should switch the data to the correct path. Therefore at each node, the data has to be switched to the correct route. This process is called routing.

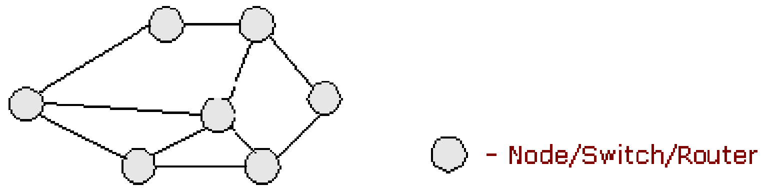
The group of bits sent by the network layer is called a packet. The packet has data and a Network Header [NH].

The NH has the Source and Destination logical addresses and other information related to the Network Layer. At each node the destination logical address is analyzed and switching is done accordingly.

It is the responsibility of the Network layer to carry a data packet from the source computer to the destination computer.

Switching





Switched Network

Circuit Switching

For PSTN the nodes are telephone exchanges. The telephone exchange analyzes the destination telephone number and switches the signal accordingly. The method of switching is called circuit switching. There are two types of circuit switching.

- Space Switching
- Time [memory] Switching

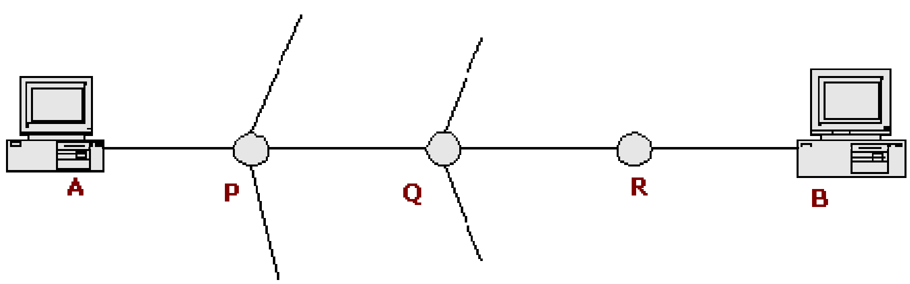
There will be an end to end dedicated connection and also this is a real time connection. This means that one end sends data [voice] to the other and it is received at that instant. However normally the real time connection is required for voice communication [interactive communication]. But this is not essential for computer to computer communication.

The disadvantages of circuit switching in terms of Data Communication are,

- Dedicated end to end connection during idle time [although no data is transferred, connection is there]. This is a waste.

- If the destination computer [telephone] is busy or any in between link is busy the connection cannot be made. The whole connection process should be started from the beginning.

The solution for this problem is store and forward method.



Computer A sends data to node P. Node P will switch the data to path PQ. If the link or node is busy the data is stored in P and when the link and node Q is free the data is transferred. The same is done from Q to R and from R to B. This is not a real time connection. This is called Message Switching.

Message Switching

The whole message [data] is send at once. e.g. computer file, electronic mail. The destination address is included in the message. The message is then passed through the network from node to node. At each node the entire message is received, stored, and when the link and the next node is free, transmitted to the next node.

The problem of this method is, if a long file is transmitted [e.g. say 1 hour duration], if one error occurs at the last moment [e.g.. 58th minute] the whole file has to be retransmitted. [Then it takes 1 hour and 58 minutes]. The solution for this problem is packet switching.

Packet Switching

The data is divided into small groups of bits and these are called Packets. Each packet is given a number and transmitted one by one. Each host in the network is given a unique identification called address. The packet also includes the source host address and the destination host address. The packet is analyzed by the Packet Switching Exchange [PSE]. It checks the destination host address and directs to the correct route at each PSE/node/router. This process is called Packet Switching. There are two types of packet switching.

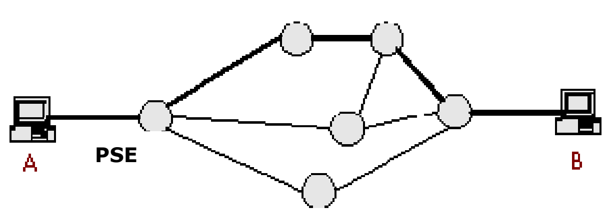
- Datagram
- Virtual Circuit

Datagram

Each packet is analyzed at each node, the route is found and the packet is directed to that route. If 1000 packets are sent from one end to the other end, each node should analyze all 1000 packets. Therefore nodes need considerable processing power. This method is suitable to transmits small number of packets. Also packets can go through different routes and the packets may not reach the receiver end in the correct order. This means that packet 12 can reach the destination before packet 10. The receiver end removes the headers of each packet and assembles the data in the correct order. In this process there is no initial connection setup between transmit end and receive end. Therefore, this is called a connectionless process.

Virtual Circuit

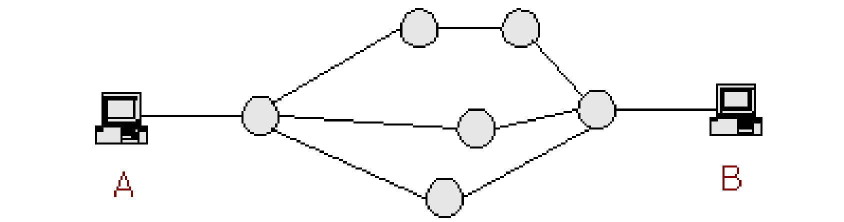
Initially a special packet is sent to find the route. This packet is analyzed at each node and one route is selected at each node to transmit the packets. Then the path is setup between transmit end and receive end. This is called a virtual circuit.



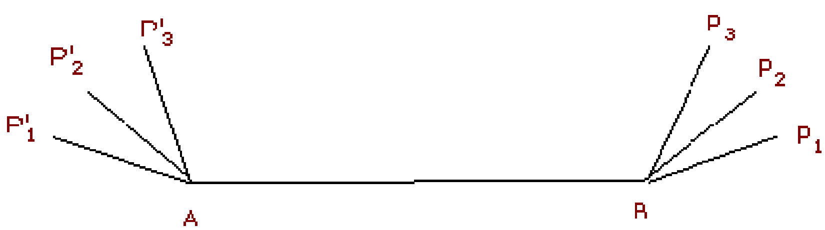
The dark line shows the virtual circuit path between **A** and **B**.

Now A can transmit all data packets to B through this path. The packets will reach A in the correct order. The advantage of this method is that each packet is not analyzed at each node to find out which virtual circuit each belongs to. Therefore a node needs less processing power compared to the datagram method. This is also called a connection oriented process.

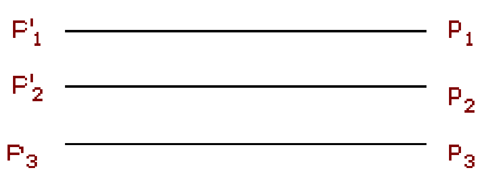
Transport Layer



Suppose computer **B** is the server and computer **A** is the client. **B** can run several processes [programs] at the same time. Normally these are called server processes. **A** can work with all these processes, but **A** should run the corresponding client processes.



P_1, P_2, P_3 - Server processes
P'_1, P'_2, P'_3 - Client processes

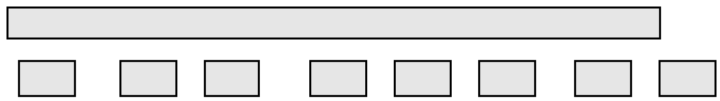


Carrying the data from A to B is the responsibility of the Network Layer.

Establishment of connection between processes is the responsibility of the Transport Layer.

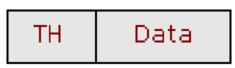
Each process is gives an identification number. This is called a "Port number".

Another responsibility of the Transport Layer is segmentation. The Session Layer [The layer above Transport Layer] sends data to the Transport Layer. The Transport Layer divides the data into small units.



This process is called segmentation.

Then, each data unit is combined with the Transport Header [H] which consists of some information related to the Transport Layer, including the Source and Destination Port numbers.



TH - Transport Header

Therefore the responsibilities of the Transport Layer are,

- Segmentation of data (Tx end) and reassembly of data [receive end].
- Establish connection between process.
- Flow control of data [process to process]
- Error control of data [process to process]
- After completion of transferring data terminate the connection.

Session Layer

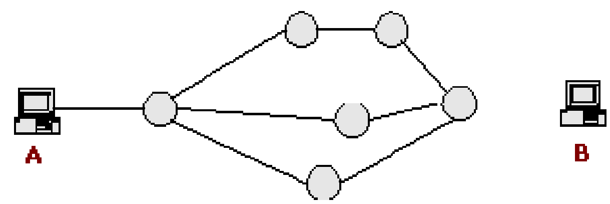
Keep the data separate from different applications.

Presentation Layer

Special processing such as encryption.
E.g. ASCII , EBCIDIC , JPEG

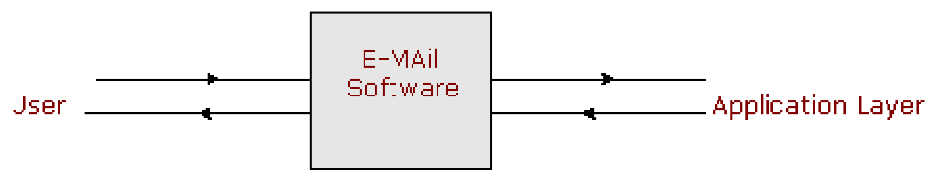
Application Layer

Application Layer interacts with the user and the Presentation Layer. The other six layers support to carry the application data from transmit end to receive end. Example of an application is E-Mail.

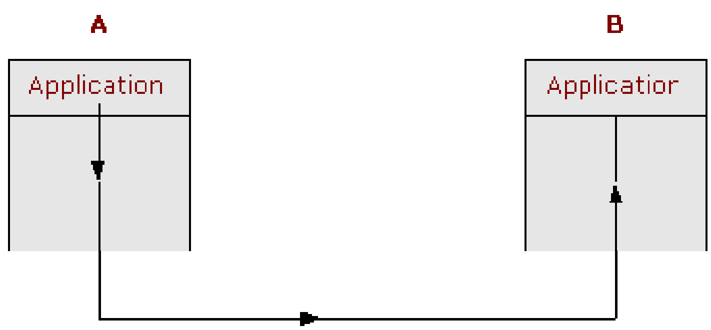


How can we send an E-Mail from A to B?

Suppose **A** is the client and **B** is the Mail Server.
After **A** makes a connection with **B**, **A** types a letter [data]
This data is sent to Presentation, Session and Transport Layers.



The Transport Layer segments the data, adds the transport Header and sends to the Network Layer. Then it is sent to the Data Link Layer, the Physical Layer and finally put to the transmission media.



B will do the reverse process of **A**.

Important points of ISO-OSI Seven Layers

Segment - group of bits at Transport Layer level.
Packet - group of bits at Network Layer level.
Frame - group of bits at Data Link Layer level

Process to process [(users' program to program] connection is made by Transport Layer [TL].

Computer [Host] to computer [Host] connection is made by Network Layer [NL].

A network has many links. Data packets are switched to the correct link [route] at a node.

Carrying Data from one end of the link to the other end is done by Data Link Layer [DLL)].

Port Address	-	identifies the process. Related to the Transport Layer.
Physical Address	-	identifies the computer. Related to the Network Layer.
Physical Address	-	identifies the computers at two ends of a link. Related to the Data Link Layer.

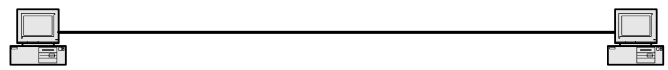
An example of a logical address is IP Address.

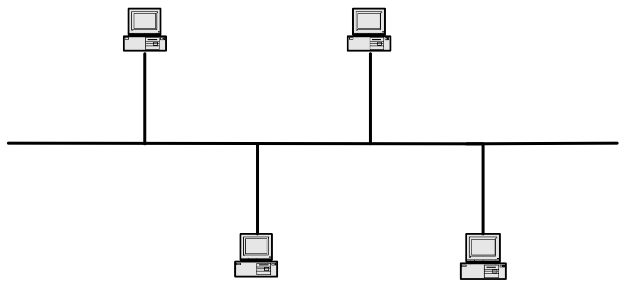
An example of a physical address is MAC Address.

Computer Networks & Data Communication

How can we connect two or more computers?

If the computers are located in close proximity, cables can be used.

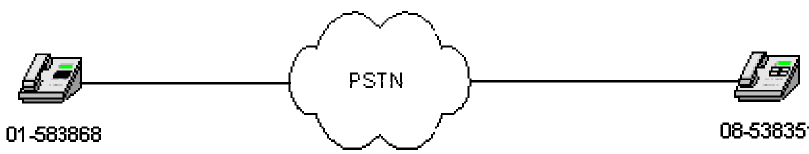




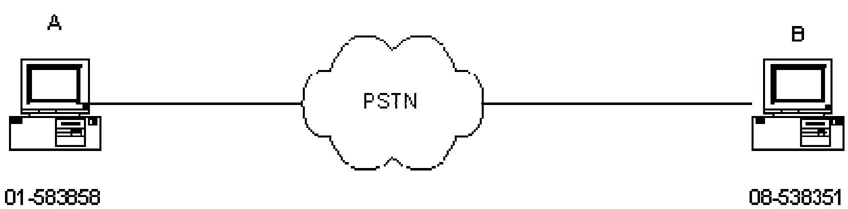
This type of network is called a Local Area Network. [LAN]

If the computers are located very far away a Public Network can be used.

A well-known Public Network is the Telephone Network. Normally this is called a Public Switched Telephone Network. [PSTN]



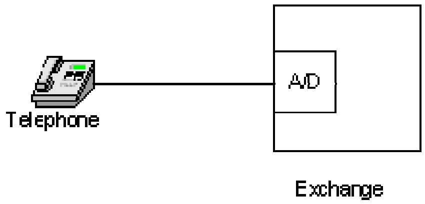
We can remove two telephones and connect two computers instead.



If computer **A** dials the number 08-538351, it can connect to the computer **B.**

This type of connection is called a dial up connection. Also this type of a network is called a Wide Area Network. [WAN]

However, there is a problem in connecting a computer directly to PSTN.

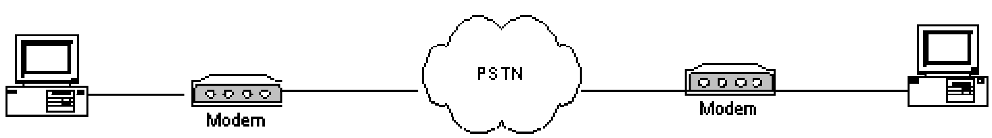


The telephone output signal is an analog signal with a bandwidth of 0.3 – 3.4 KHz.

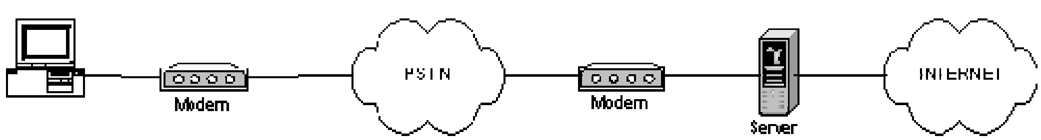
At the telephone exchange this analog signal is converted to a digital signal. [Analog to Digital conversion or A/D Conversion]. For this purpose Pulse Code Modulation [PCM] is used.

Since the exchange expects an analog signal, the computer output digital signal should be converted to an analog signal. For this purpose a Modulator is required. Similarly for the received side of computer, the analog signal sent by the exchange has to be converted to a digital signal. For this purpose a Demodulator is required.

Normally the Modulator and Demodulator comes as one unit and it is called a MODEM.



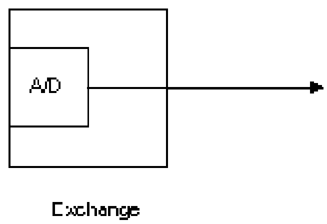
How to connect Internet through PSTN



To connect to the SLT Internet Server, you have to dial the number 150.

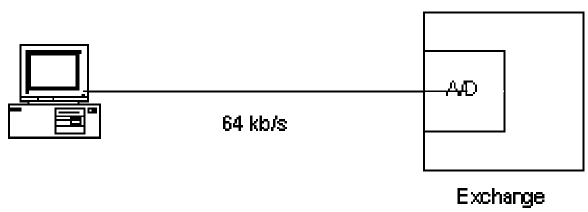
After connecting a Modem to a computer, it should be configured using the appropriate software which comes in a diskette along with the Modem.

The bit rates supported by a normal modem is 4.8 kb/s, 9.6 kb/s, 14.4 kb/s, 19.2 kb/s, 28.8 kb/s, 56 kb/s.



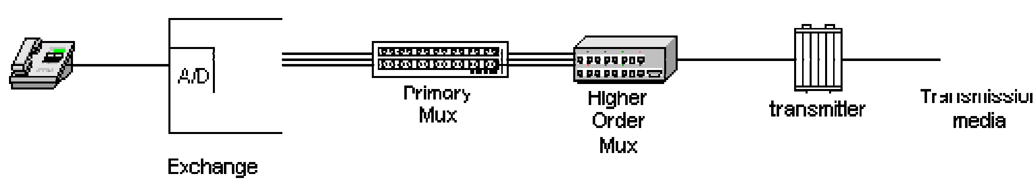
The A/D conversion process at the exchange is PCM.

The PCM output is 64 kb/s.

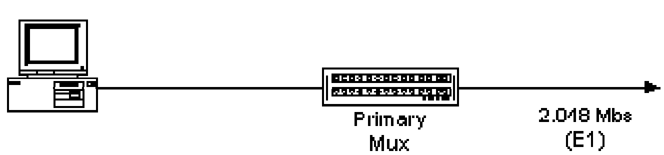


If we bypass the A/D conversion part of exchange, we can send a 64 kb/s digital signal from the computer to the exchange.

Normal telecommunication system is as follows.



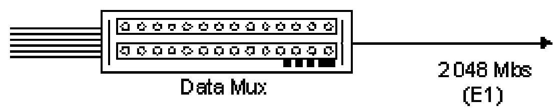
If
we by-pass the exchange the computer can be directly connect to the Primary MUX.



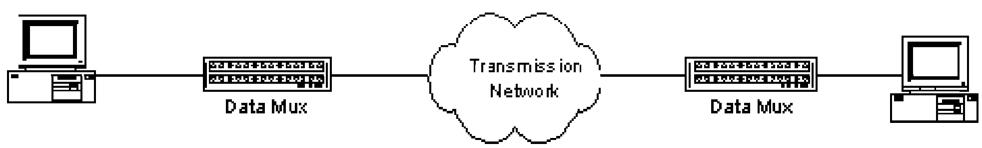
Each input channel of the Primary MUX is 64 kb/s. Therefore the computer can send 64 kb/s digital signal.

However, the Primary MUX does not have the ability to provide service for bit rates of multiples of 64 kb/s. That is 128 kb/s, 192 kb/s, 256 kb/s, 512 kb/s etc.

Therefore for data transmission purposes the Data MUX is used.

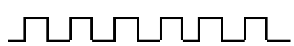


The input ports can be configured for different bit rates. That is 128 kb/s, 192 kb/s, 256 kb/s, 512 kb/s etc.



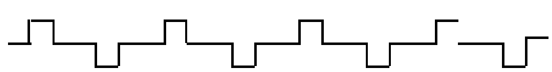
This type of circuit is called a leased circuit.

The computer output signal is a unipolar binary signal.



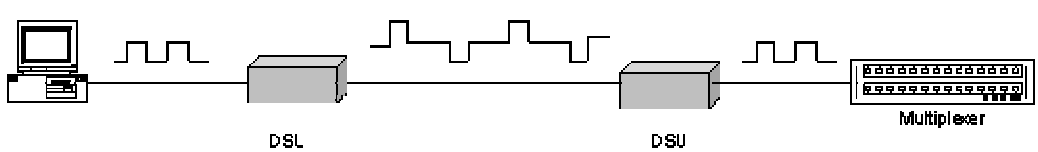
When such a signal travels more than about 10 m it can get attenuated.

In order to send a Computer signal to a long distance, it should be converted to some pattern of a bipolar signal.



This is called a line code.

By using a Digital Service Unit [DSU], line coding can be achieved.



For leased digital circuits we have to use a DSU at the computer and DSU at the Data Mux.

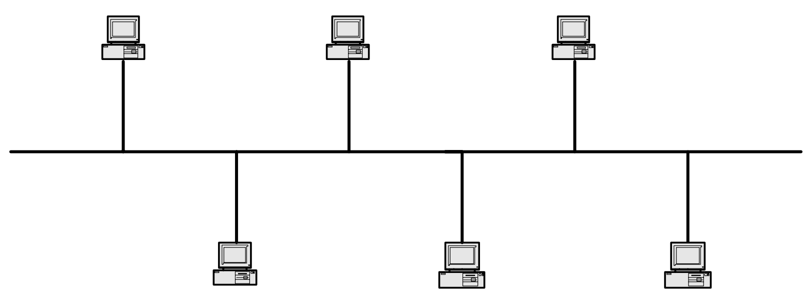
For dial up connections we have to use a Modem at the computer.

Local Area Networks [LAN]

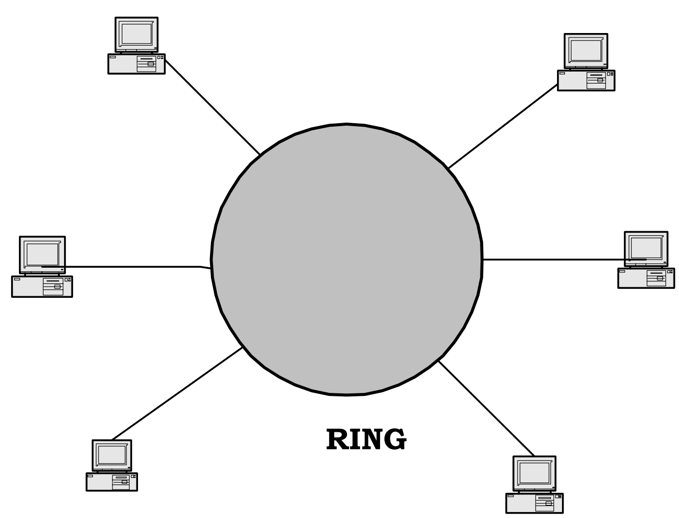
If the computers are connected by cables, [which are in close proximity], it is called a Local Area Network [LAN].

Network topologies

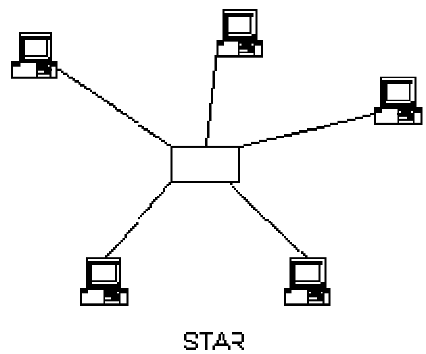
The way of connection of computers is called "Topology".



BUS



RING



The computers can be connected as in the configurations above. These are called network topologies.

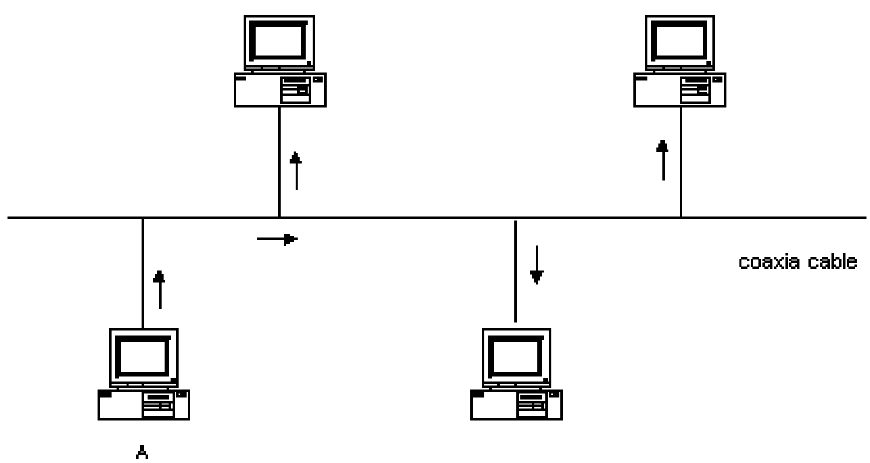
If a signal is put to a bus or a ring, all hosts connected to that network can receive the signal. Therefore, bus and ring networks are called “Broadcast Network”.

The widely used topology is Bus.

The computers can be connected using a coaxial cable.

There are two types of coaxial cables.

- Thin Coaxial
- Thick Coaxial



If computer **A** transmits a signal, it will be received by all other computers. This feature is called broadcasting. Therefore, the bus network is a broadcasting network.

Media Access Control (MAC)

In a bus network or ring network, if two computers release signals to the cable [media], there will be a collision of signals. In order to avoid collisions, the access to the media has to be controlled. The method of media access control is called MAC Protocol.

The widely used two MAC protocols are CSMA/CD and Token.

CSMA/CD

Before transmitting a signal to the media it should be checked whether a signal is already in the media. This is called Carrier Sense [CS].

By this method multiple computers can access the media at different times. This is called Multiple Access [MA]. However, there is a possibility to transmit signals by two computers at exactly the same time. Then there can be a collision. Therefore, after releasing the signal to the media it should be monitored whether there is any collision. This is called collision Detection [CD]. If a collision is detected the signal has to be retransmitted. This is one of the MAC protocols. It is called CSMA/CD.

Control Token

A small frame called token is continuously going through the ring or bus. If any host in the network wants to transmit, first it checks the token in the media. If the token is available, it gets the token and releases the data frame. The data frame goes round the network [ring or bus] and come back to the transmitted host. Then it releases the token to the media.

Ethernet

If the topology is Bus and the MAC protocol is CSMA/CD, it is called Ethernet.

Thin Coaxial → **Thin Ethernet**

Thick Coaxial → **Thick Ethernet**

The bit rate of a coax cable is 10 Mb/s.

If there is a LAN, of 100 Mb/s, it is called Fast Ethernet. If the bit rate is 1000 Mb/s, it is called Gigabit Ethernet.

Token Ring

If the topology is Ring and the MAC protocol is token it is called a "Token Ring"

Token Bus

If the topology is Bus and the MAC protocol is token it is called a "Token Bus"

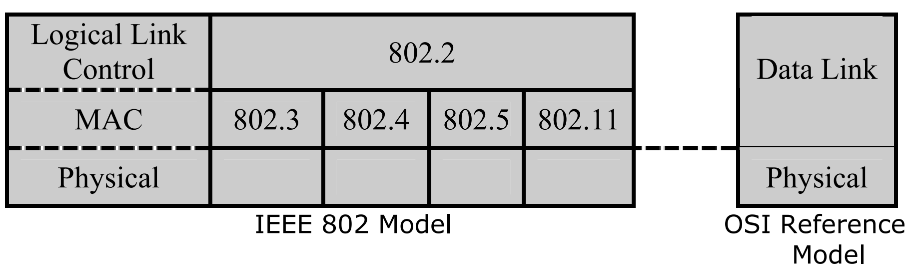
MAC Address

In order to identify computers, each computer is given a 48-bit identification. This is called MAC Address or Physical Address.

IEEE 802 LAN Standards

The LAN standards are defined by the IEEE 802 recommendations.

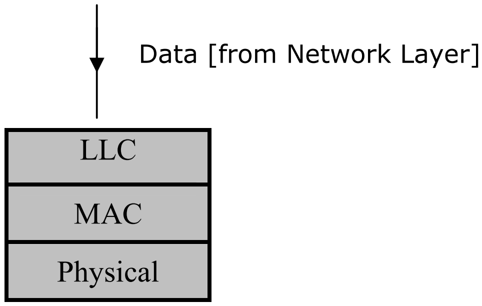
IEEE 802 Reference Model



802.2	-	Logical Link Control [LLC] Protocol.
802.3	-	CSMA/CD Bus
802.4	-	Token Bus
802.5	-	Token Ring
802.11	-	Wireless

The LLC Layer is common for all MAC Protocols.

LLC Frame Structure



The LLC Layer receives data from the Network Layer.

1	1	1	
DSAP	**SSAP**	**Control**	**Information**

DSAP - Destination Service Access Point - 1 byte
SSAP - Source Service Access Point - 1 byte

These two fields are used to keep the protocol information on destination and source Network Layer.

The control field also has one byte length. This is used for supervisory functions and for sending and receiving frame sequence numbers etc.

IEEE 802.3 Ethernet Frame Structure

<u>Ethernet</u>

If the topology is Bus and the MAC protocol is CSMA/CD it is called an Ethernet .

When a packet of data is handed over to the MAC layer to be transmitted over the physical channel, MAC layer appends some other information to it and creates a frame known as the ***Ethernet frame.***

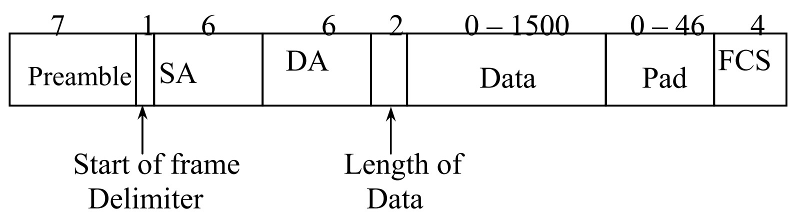
Ethernet Frame

Data	**Overhead Bits**

The data is transmitted as a frame. The frame will have some additional bits in addition to data. They are called Overhead [OH] bits. It carries Source Address, Destination Address, and Synchronization etc.

When a frame is released to the media it is received by all computers. But the computer, which has the MAC Address equal to the Destination Address of the frame, will accept the signal. Others ignore it.

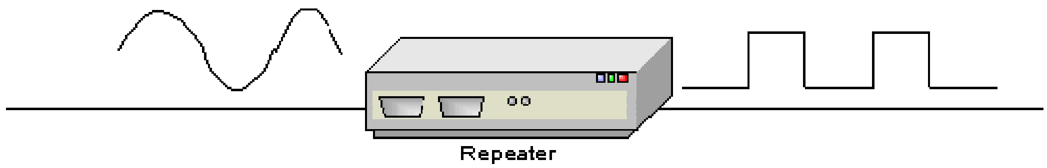
The IEEE 802.3 Ethernet frame structure is as follows,



Each frame starts with a preamble of 7 bytes, each containing the bit pattern 10101010. The Manchester encoding technique is used to encode this pattern to a set of electrical pulses.

- The next byte containing '10101011' denotes the start of the frame.
- Each Ethernet Card Address has 48 bits or 6 bytes. This address is used in LANs to identity the parties involved in a communication session. The next two slots, i.e. Destination address & the Source address represent the Ethernet address of the destination computer and the source computer respectively.
- Next two bytes represent the total number of bytes in the data field.
- The next field consists of the actual data bytes to be sent from the source computer to the destination computer.
- In order to distinguish valid frames from garbage frames IEEE defined a minimum frame length for the valid frames. That is 64 bytes from Destination address field to checksum field. Therefore if the data field is less than 46 bytes, the Pad field is used to fill out the frame to the minimum length. Also the maximum size of data is 1500 bytes.
- The final field of the Ethernet frame is the checksum field. These 32 bits are used to prepare a code for the data to detect erroneous situations. The cyclic redundancy algorithm is used to generate these check bits.

Repeater



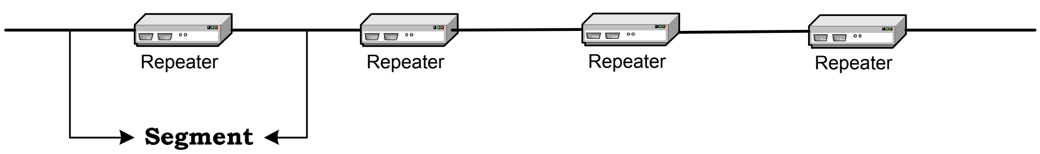
The repeater can reconstruct the original signal.

If thin Coaxial cable is used, a repeater is used after 200 m.

One side of a repeater is called a segment. The maximum number of repeaters recommended for one bus is four. In other words, the maximum number of segments is five.

Segment of Ethernet

The maximum number of repeaters allowed is four.

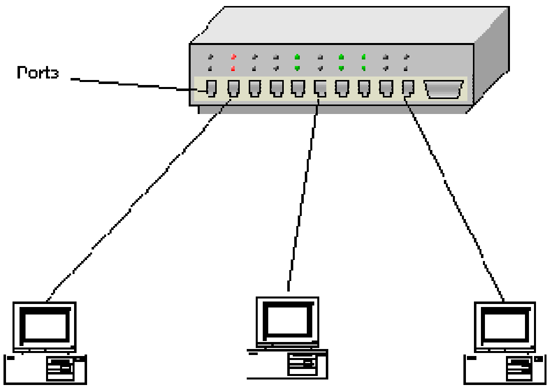


If thin coax is used, the maximum segment length is 200 m.
If thick coax is used, the maximum segment length is 500 m.

Therefore, the maximum length of a coax LAN can be 1000 m or 2500 m. [end to end]. This has 5 segments but act as a single bus. If any two computers transmit frames to the cable at the same time, those two frames will collide. Therefore, we say that the whole LAN is in one-collision domain.

Hub

A hub has many connecting points. They are called ports.



Each port is connected to a computer by using an unshielded Twisted Pair [UTP] Cable with a RJ45 connector. The hub is a "Multi Port Repeater".

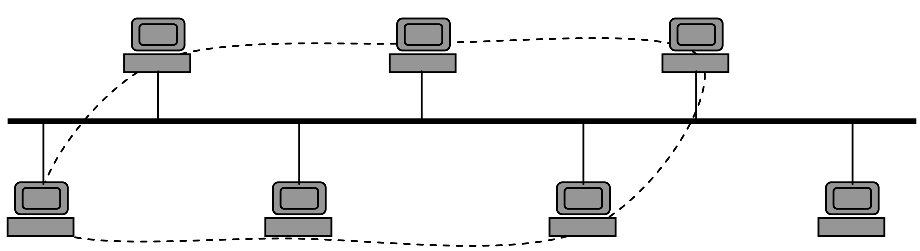
If computer **A** transmits a signal it is repeated and retransmitted to all other computers. That is, the hub broadcasts the message. This is same as the bus network.

Therefore, the hub, logically works as a bus network [physically it is a star network]. The MAC protocol used is CSMA/CD. Therefore, this is an Ethernet.

Nowadays all Ethernet LANs are connected using hubs. Unlike coaxial cable, the hub can easily increase the number of computers connected to the network.

Token Bus [IEEE 802.4]

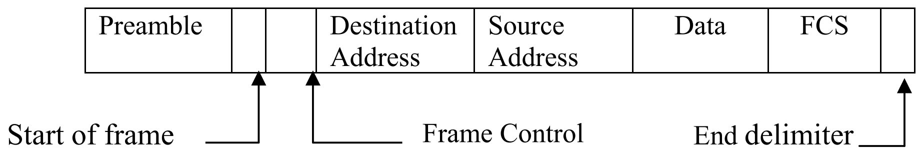
Physically the token bus is a linear cable in to which the stations are attached. Logically these stations are organized into a ring as illustrated in the diagram below, with each station knowing the address of the station to its 'left' and right.



This station is not currently in the logical ring

When the ring is initialized, stations are inserted into the logical ring in order of station address, from highest to lowest. A special control packet known as a token propagates around this logical ring, with only the token holder being permitted to transmit frames. The token holder can transmit frames for a certain amount of time and then it passes the token to its neighbour. If the station has no data to be sent, it passes the token to the next station immediately. Since only one station at a time holds the token, collisions do not occur in this algorithm.

Frame format of IEEE 802.4 is as follows.



Frame control field is used to specify the frame type. I.e. is data frames and control frames.

Token Ring [IEEE 802.5]

This standard was defined for the networks with ring topology. In this standard a special bit pattern called the token circulates around the ring whenever all stations are idle. When a station wants to transmit a frame, it is required to seize the token. Only the token holder is permitted to transmit a frame over the channel. This standard also solves the collision problem as the token bus.

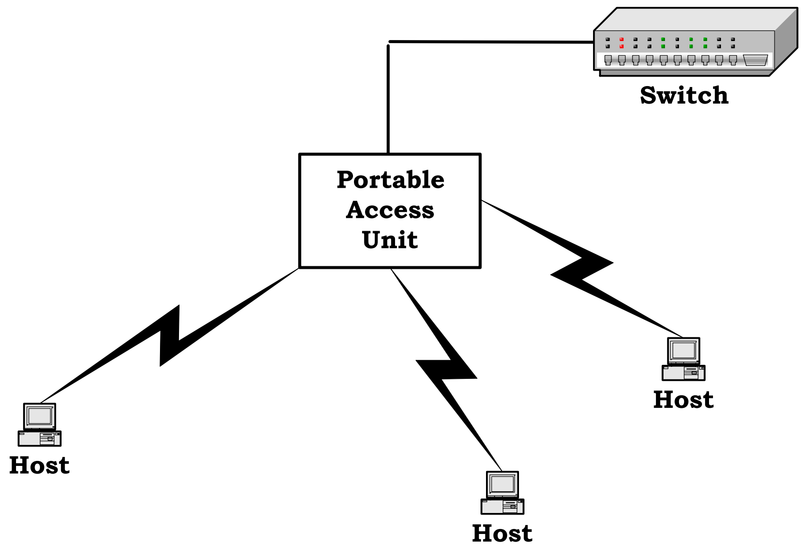
This also has the facility to prioritize the token. That is, different stations can have their own priority level which is defined by a control station called a Monitor.

A station wishing to transmit must wait for a free token with the existing priority is less than the required priority. If this is not satisfied it reserves the token to its required priority.

If the token priority is the same as the set priority it seizes the token and transmits the normal Data Frame. After completion of sending data it releases the token with priority same as reserved priority.

Wireless LANs

The cabling of LANs is a problem in some occasions such as open office spaces. Sometimes the cables or connectors can be damaged and also it is not nice looking since the cables are hanging here and there. The solution for such occasion is the wireless LANs.



The computers are linked to the portable Access Unit [PAU]. The PAU is wired to a switch or other network devices. The link between Host and PAU is wireless. This link can be ratio or infrared.

The MAC methods that can be used are,

- CDMA
- CSMA/CD
- CSMA/CA
- TDMA

There are four transmission schemes used with radio wireless LANs.

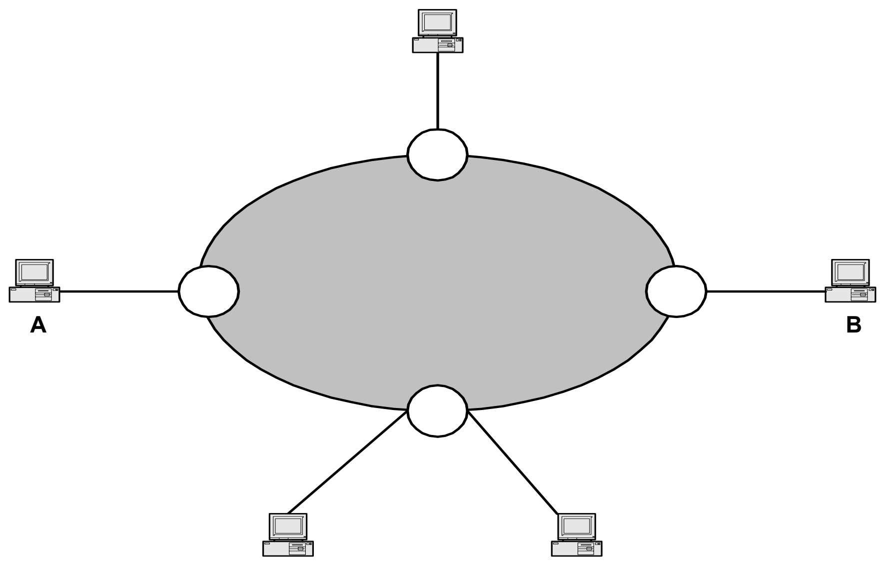
- Direct Sequence Spread Spectrum
- Frequency Hopping Spread Spectrum
- Single Carrier Modulation
- Multi Subcarrier Modulation.

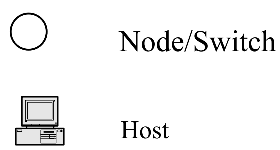
The IEEE standard for wireless LANs is 802.13.

Wide Area Network [WAN]

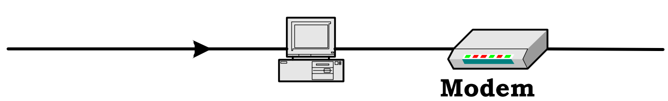
If the computers are far away it is not practicable to connect them by using a physical wire like in LANs. In this case an existing network can be used to establish a connection.

E.g. Public Switched Telephone Network [PSTN]





If PSTN is used, the telephone is removed at one point and a computer and modem is connected instead.

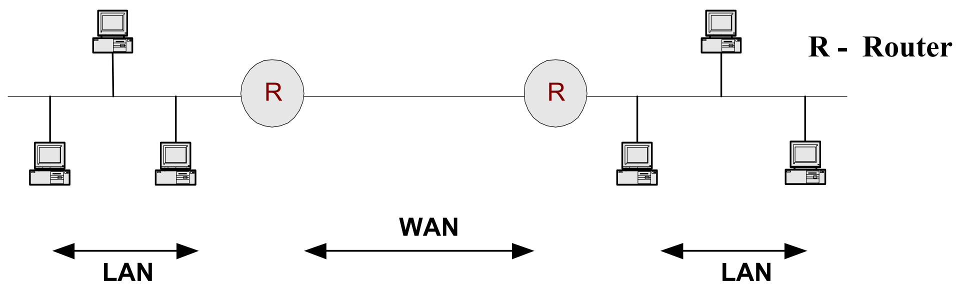


The computer dials the destination telephone number and establishes the link.

Apart from PSTN, the other methods that can be used are,

- Leased Line
- Packet Switched Connection
- Frame Relay
- ISDN
- ADSL
- IP-VPN

The above connection links are used to connect two remotely located LANs.



The Router has two types of ports.

- LAN Ports
- WAN Ports

The WAN Port is connected to the Leased Line or Frame Relay or any of the above types of connection links.

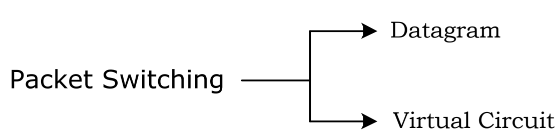
PSTN

The PSTN connection is not a dedicated connection. Therefore, it is suitable for occasional connection such as access to Internet from home.

Leased Line

This link is secure, dedicated, but the cost is high. We can get bit rates of 64 kb/s, 128 kb/s, 256kb/s, 512kb/s, 1024 kb/s and E1 [2048 kb/s] etc.

Packet Switched Connection



Datagram

This is a connectionless concept. That means before transferring the data, a connection is not established. The packet is analyzed at each node/switch/router and directed it to the correct destination. Also there is no flow control, error control etc. This is a simple technique but unreliable.

One of the protocols used with datagrams is Internet protocol [IP].

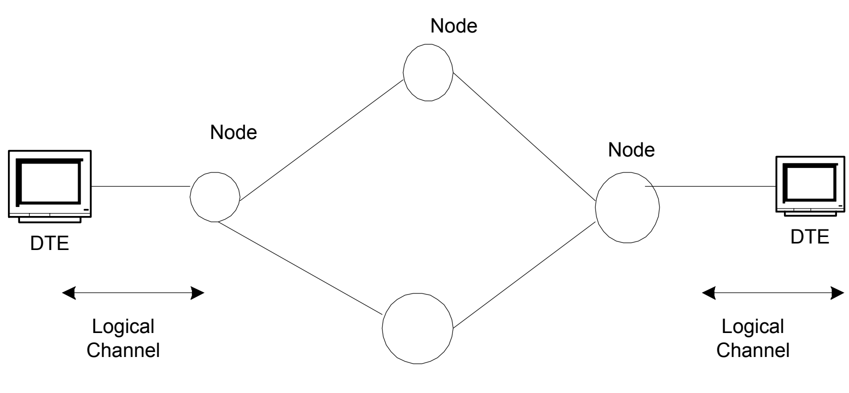
Virtual Circuit [VC]

The analogy is similar to exchange of messages by using telephone calls. The DTE [terminal] sends a special packet to its Packet Switching Exchange [PSE] with an identifier called Logical Channel Identifier [LCI] and the destination host address. The PSE analyzes the packet and sends it to the destination PSE through the intermediate PSES. The destination PSE allocates another LCI to the destination, DTE and sends the response to the source DSE. The intermediate PSES keep information about the LCIs. Now a temporary path is established between the two PSES and it is called a Virtual Circuit. Then the information packets are transferred with the stamp of LCI. Therefore, the PSEs do not need to analyze the packets for routing. The work load at any PSE is less than the work load of Datagram method. The information packets are transferred serially.

Although the virtual circuit may appear to be similar to circuit switching, the virtual circuit has following features.

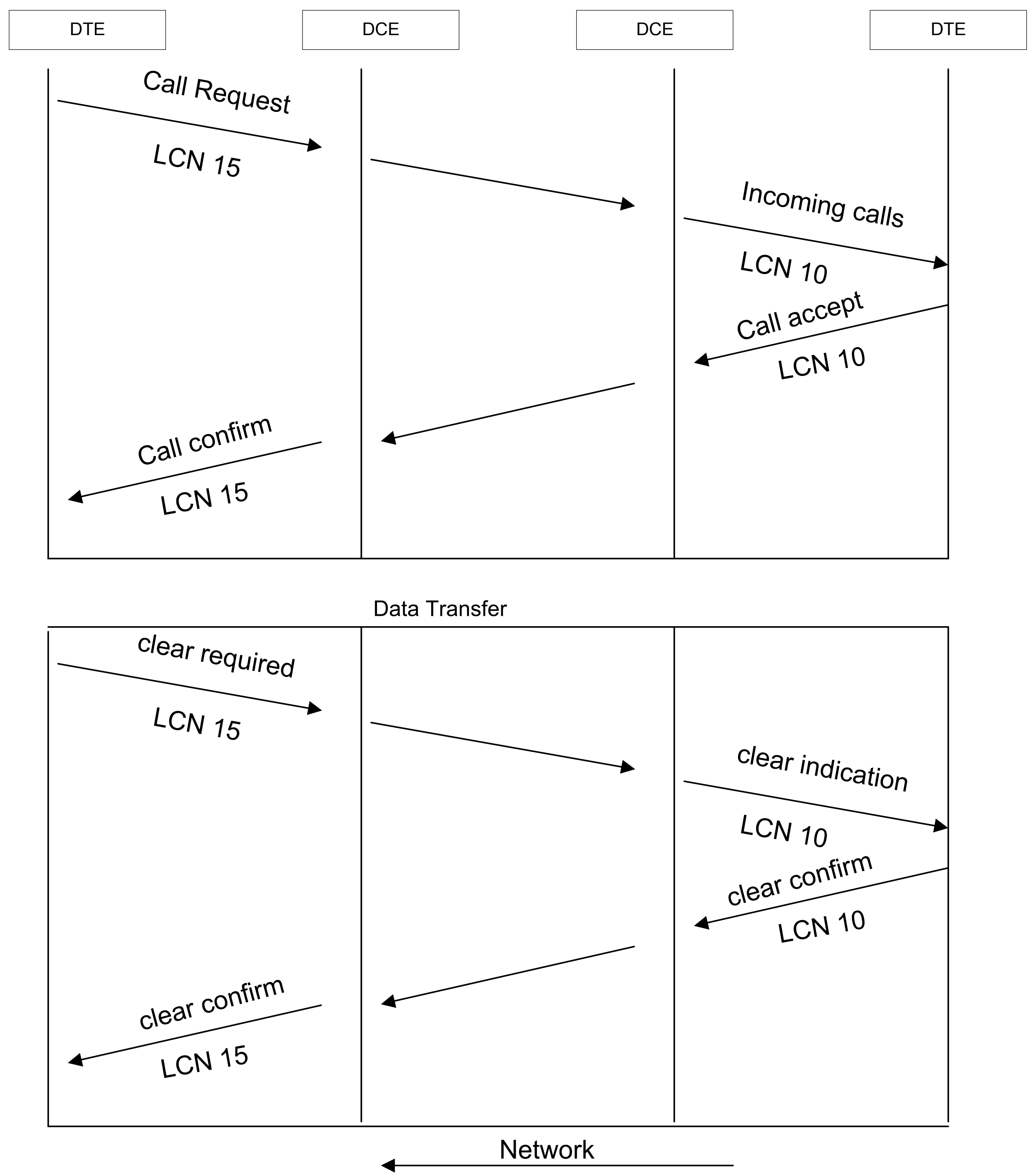
- Virtual circuit is purely conceptual.
- Error control and flow control is available in packet level and link level.
- Many virtual circuits can be established for one DTE simultaneously.

Normally, a virtual circuit is cleared and the appropriate logical channel identifiers are released after all data relating to a call have been exchanged. If the user has bulk traffic to be transferred everyday, it is possible to established the virtual circuit permanently. This is known as a permanent virtual circuit [PVC].



Virtual Circuit

Switched Virtual Circuit Operation



Datagram

E.g. SVC

Suppose DTE-A wants to connect to DTE-B. DTE-A sends a packet to Node A with the DTE-B address and the Logical Channel Number [or LCI] = 15. By analyzing the DTE-B address, Node A routes the packet to Node B. Node B allocates LCN = 10 and send the packet to DTE-B. The DTE-B accepts the call and replies to Node B with LCN = 10 [This is the channel allocated to communicate between Node and DTE-B). Node B informs this to Node A. Then Node A confirms the call to DTE-A through the LCN = 15.

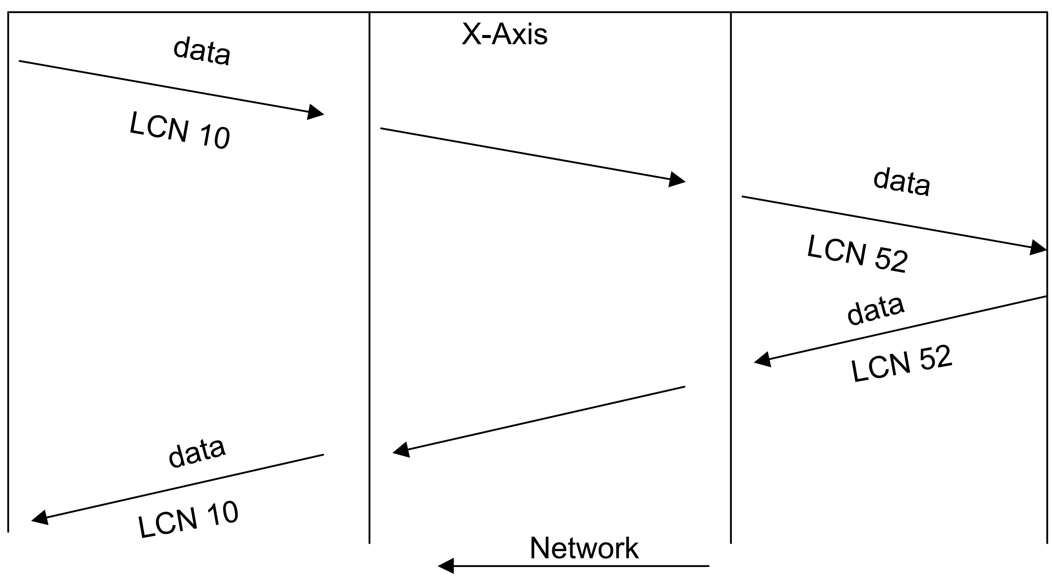
Now the connection has been established. The next step is to transfer the data.

After completion of sending the data, DTE-A wants to clear the channel. That also is done through a similar process used for call establishment.

Please note that once the LCN is allocated, the same LCN is used to communicate between the DTE and the Node [PSE]

Permanent Virtual Circuit Operation

PVC Operation



E.g. PVC

The LCN number is permanently allocated and the end to end virtual circuit is also available permanently. The connection is always available. Therefore, data can be transmitted at anytime.

X.25 Protocol Suite

ITU-T X.25 Protocol Suite

During 1970's many proprietary packet switched networking emerged in developed countries. Later it was decided to prepare global standards for packet switching networking and then CCITT [ITU-T] defined the set of rules and it was named as X.25.
X.25 has three layers in OSI modes.

Network	X.25 PL P
Data Link	LAP – B
Physical	X.21, X.21 bis

X.25 Physical Layer

The physical layer provides the physical signalling and connections between the DTE and the DCE. At this layer, standards such as X.21, X.21 bis, EIA – 232 – E and V.35 are used. Also it can be implemented with Ethernet interfaces and E1 carriers.

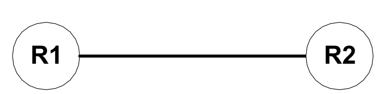
X.25 Data Link Layer (DLL)

The primary objective of the X.25 DLL is to transport error free packets between the DTE and the Packet Switching Exchange. This layer uses the LAP - B [Link Access Procedure – Balanced] Protocol. LAP – B is a version of HDLC. It is also responsible for normal functions of Data Link Layer and informing the X.25 network layer about unusual link problems such as excessive errors on a link failure.

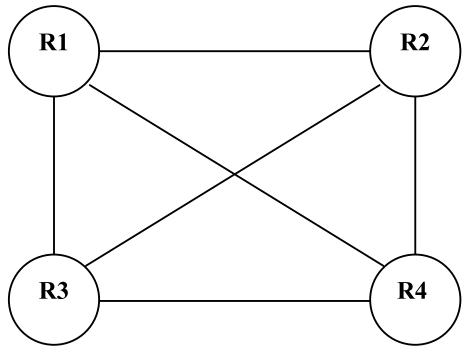
X.25 Network Layer

X.25 network nodes and DTEs employ Statistical Time Division Multiplexing [STDM] technique to transfer the users' traffic into and out of the network. Multiplexing of more than one user is a feature of virtual circuit.

Why we need frame relay?

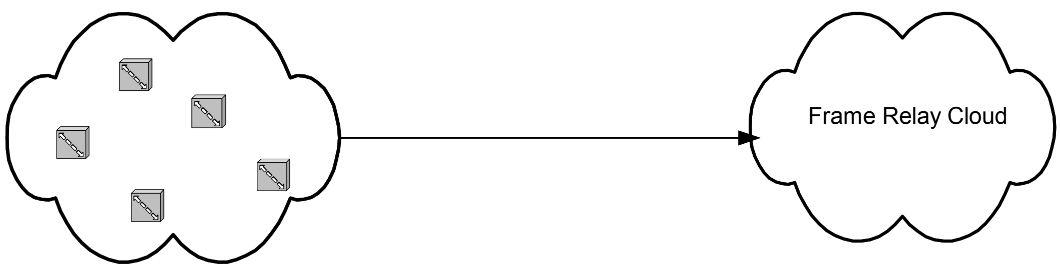


Consider the Router R1 and Router R2 connected by using a Leased Line. If the Leased Line has any problem we cannot communicate between the two routers. In order to have redundancy we should have another Leased Line [standby]. This will increase the operational cost of the network.

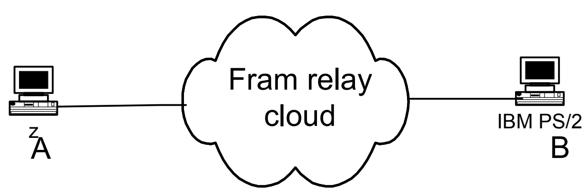


If we have four Routers [R1, R2, R3 and R4], to have connectivity among all Routers, 6 Leased Lines are required. Again this is a costly network.

The solution for this kind of problem is "Frame Relay"

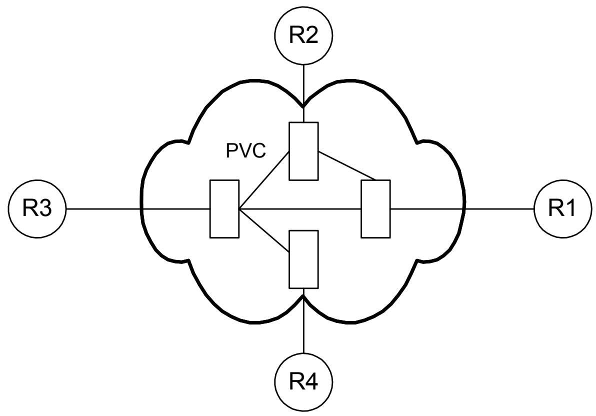


The Frame Relay cloud is same as PSTN. It has many Frame Relay Switches.



If a Frame Relay link is needed between A and B, a special number is given to A to reach B and another special number is given to B to reach A. These numbers are called Data Link Connection Identifier [DLCI]

Normally we give a unique DLCI for each Router. Frame Relays switches are configured to make the connection. The connection path is same as X.25 and it is called a virtual circuit. Normally Frame Relay uses PVCs only.



Consider four routers connected to a Frame Relay cloud as shown in the figure.
DLCIs of the four Routers are 50,51,52 & 53.

The frame format is as follows.

F R Header	**Data**	**FR Trailer**

The main fields of Header are,

- DLCI
- FECN [Forward Explicit Congestion Notification]
- BECN [Backward Explicit Congestion Notification]

The Trailer is the Frame Check Sequence [FCS] for error detection.

When R1 sends a frame to R2, the DLCI is put as 52. When this frame comes to FR switch it knows to which route it is to be switched.

Frame Relay Switches are configured in a way that if the normal link fail it is automatically switched to an alternative route. Therefore, we do not need to maintain separate redundant links.

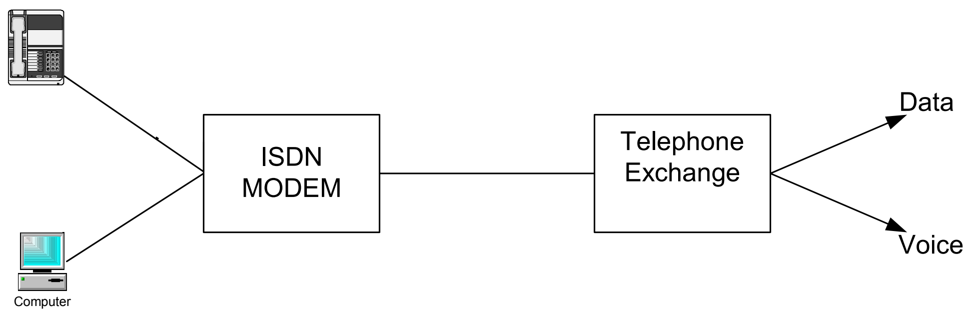
Also we can get PVCs with alternate Routers.

E.g. R1 to R2
One PVC – direct R1 – R2
Second PVC – R1 – R3 – R2

The PVC is cheaper than the Leased Line.

Integrated Services Digital Network [ISDN]

The PSTN is designed to establish connection among telephones. If we want to use PSTN for data communication the telephone should be disconnected from the line and instead of that a modem and a computer is connected to the line. The main disadvantage of this method is the telephone and computer cannot be used simultaneously. ISDN is the solution for this problem.



If we use and ISDN modem at home both the telephone and the computer can be used simultaneously. Please note that the "ISDN Modem" is not the correct term but normally subscribers use this term.

This has the following facilities.

For normal subscribers,

2 x 64 kb/s channels + 1 x 16 kb/s channel.
One 64 kb/s channel is called a Bearer Channel [B Channel].
The 16 kb/s channel is called a D Channel.
One B channel can be used for voice. [Telephone].
The other B channel can be used for data [Computer] or 2 B [128 kb/s] can be used for data.
D channel is normally used for signaling between the ISDN modem and the exchange. If it is not used for signaling, that also can be used for data.

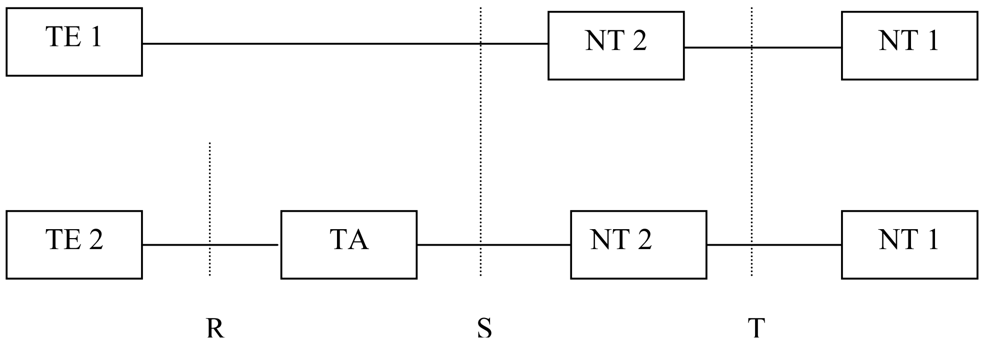
All the channels are represented as,

2B + D

If all channels are used for data, a 144 kb/s bandwidth can be used for data.
In addition to 2B +D, the overhead bits of 48 kb/s are also added. Therefore, the total bit rate will become 192 kb/s.

ISDN Customer Premises Equipment

The TDM of the signals are done at ISDN modem. But normal telephones are not digital. Therefore, in such signals the analog to digital conversion has to be done. Digital telephones are also available. Sometimes a LAN may be connected instead of a computer. Therefore, customer interface equipment is decided by considering all these facts. Therefore, ITU-T defined some interface points R, S and T.



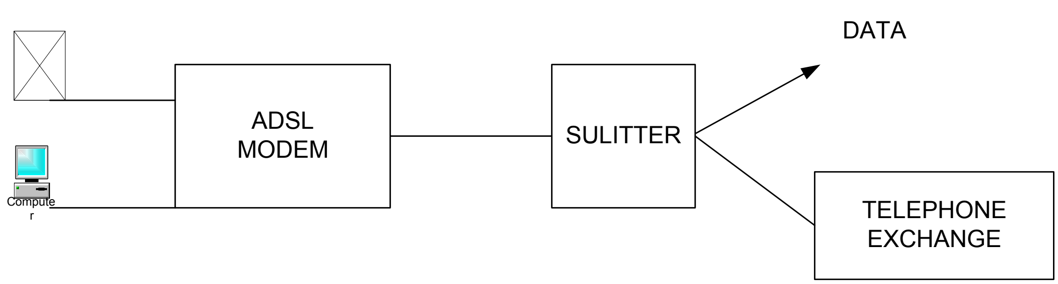
R, S, T - Reference Interface Points
TE - Terminal Equipment
NT - Network Termination

TE1 refers to the devices that support standard ISDN interfaces.
E.g. Digital Telephone, Digital fax.
TE2 refers to non-ISDN equipment. Such equipment should connect through a Terminal Adapter [TA].

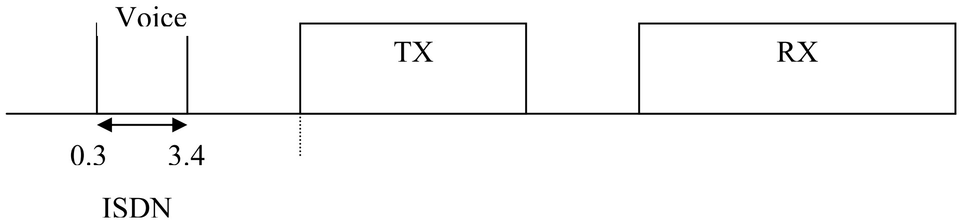
NT1 is connected to the telephone subscriber loop. There are equipment which integrates both NT1 and NT2 and it is named as NT12.

Asynchronous Digital Subscriber Line [ADSL]

This is a further development of better utilization of the normal telephone line. This was mainly developed for Internet users.



The ADSL modem use a special line code and converts the multiplexed [voice and data] to an analog signal. The bandwidth utilization is as follows.



The Tx and Rx bandwidths are not the same. Therefore, it is called Asymmetric. The reason is, for Internet download information [data] is more than the upload information. Therefore, higher bandwidth is allocated for downloading.

Theoretically the upload bandwidth and download bandwidth can go up to 1 Mb/s and 8 Mb/s. But these bandwidths depend on the distance between the exchange and subscriber premises. However, at least 512 Kb/s download can be obtained easily.

Note : ADSL can be used for both normal telephone and data simultaneously.
ADSL can be used for both ISDN and ADSL data simultaneously.

IP-VPN

The Internet is used for the connectivity. Since the Internet is a public network and not secure, special encryption methods are used to send data. The data is decrypted at the receive end. Although Internet is a public network, after encryption it will become virtually a private network. Therefore, it is called a Virtual Private Network [VPN].

TCP/IP Protocol Suit

The TCP/IP is a protocol that can be used for LANs and WANs. It has five layers and its relationship to OSI model is given in the following figure.

SMTP,FTP SNMP TFTP
TCP, UDP
IP
Data Link
Physical

TCP/IP Model

Application
Presentation
Session
Transport
Network
Data Link
Physical

OSI Model

The TCP/IP model has only five layers.
Physical, Data Link, Network, Transport and Application.

TCP/IP Application Layers is equivalent to Session, Presentation and Application Layers of OSI Model.

TCP/IP Model does not define Physical Layer and Data Link Layer. It can work with any Physical Layer and Data Link Layer.

Network Layer

This has only one protocol called Internetworking protocol [IP]

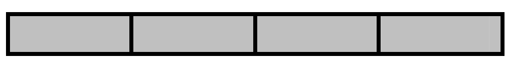
Transport Layer

The Transport Layer has two protocols called Transport Control Protocol [TCP] and User Datagram Protocol [UDP].

Application Layer

This layer has many protocols. SMTP, FTP, SNMP, TFTP, TELNET etc.

The Application Layer sends data to the Transport Layer.



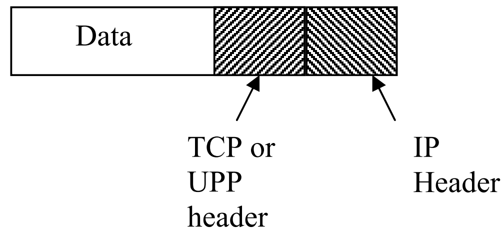
Application Data

The Transport Layer divides it into small parts called “Segments”. To each segment a TCP header or a UDP header is added. [Depending on the Application the relevant protocol should be selected]

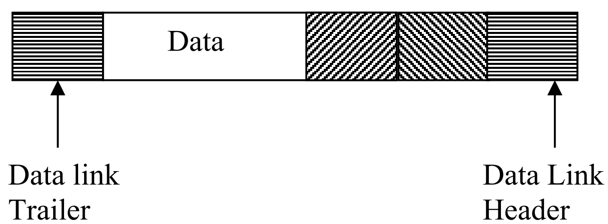
TCP is a connection-oriented protocol. That means it establishes a connection with the other end transport layer.

UDP is a connectionless protocol.

The TCP segment or UDP segment is sent to the IP layer. Then it adds the IP header. IP has the datagram operation. Therefore, it is a connectionless protocol.



This is called a IP packet. The IP packet is sent to Data Link Layer. It adds the Header and Trailer. This is called a frame.



The frame is sent to the Physical Layer and it will convert the frame to 1s and 0s [raw bits] and send it to the physical medium.

It is received at the Receive end and the reverse function is performed at each layer.

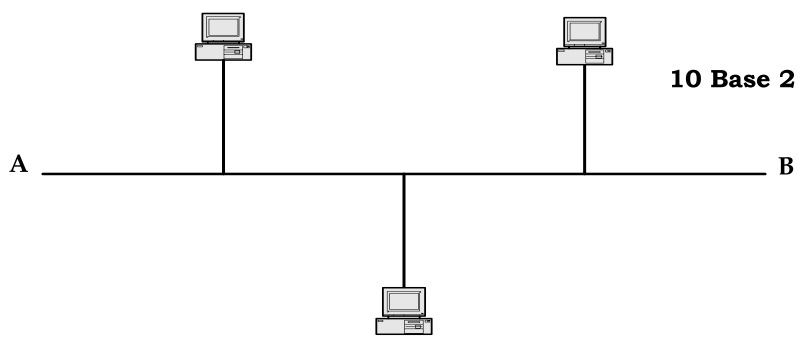
Network Devices and Internetworking

There are many networking devices used in LANs and WANs. Some of the networking devices are,

- Repeater
- Hub
- Bridge
- L2 Switch
- L3 Switch
- Routers

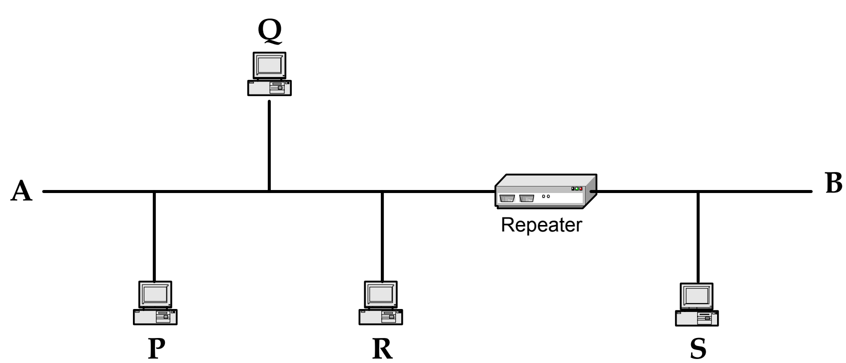
Repeater

Consider a 10 Base 2 LAN. Thin coaxial cables are used and the signal can travel a maximum of 200 m length without repeaters.

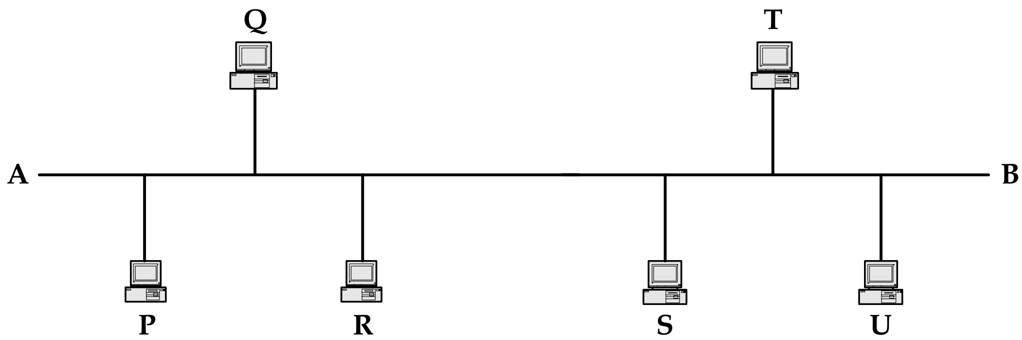


A to ***B*** maximum length can be 200 m.
This is called a segment.

If it is more than 200 m length, a repeater can be used.



The above network has two segments. But it acts as one Ethernet network.



If *P* and *Q* transmit a frame at the same time there will be a collision. Any two computers in any segment transmitting simultaneously will make a collision. Therefore, the above network has one "Collision Domain".

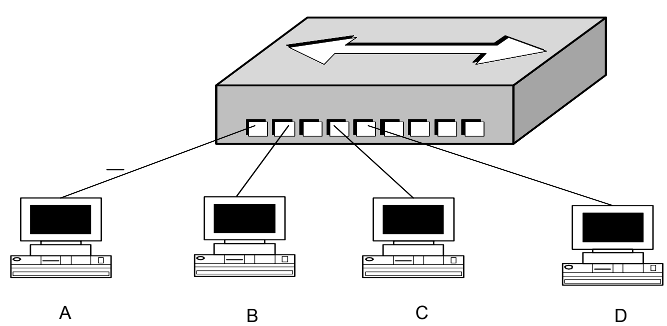
The maximum number of repeaters allowed is four. That is five segments.

5 Segment
1 collision domain

Also if one computer broadcasts a message it will be received by all computers in the above network. Therefore, the network has only one "Broadcast Domain".

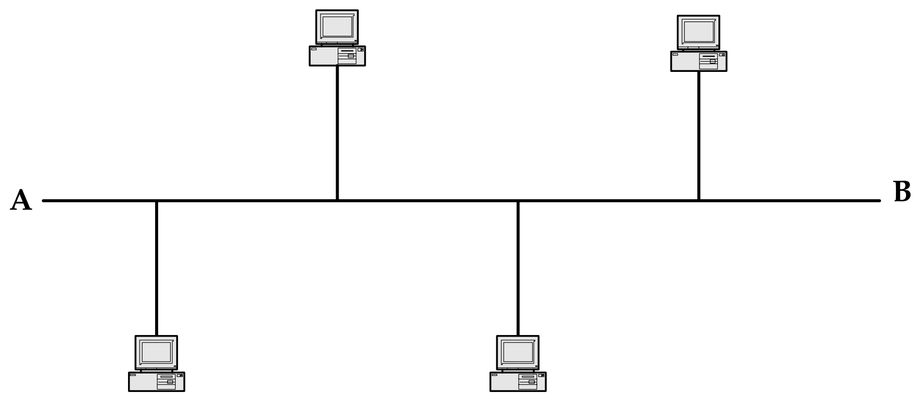
Hub

Installation and expansion [increase the number of hosts] of a network is rather difficult and coaxial cable is costly compared to UTP cables. Now there are UTP cables available which can operate even with 1000 Mb/s [1 Gb/s]. Therefore, now the trend is to use UTP cables instead of coaxial cables. For this purpose a hub is used.



The hub is a repeater which has many ports. It is a passive device. That means it cannot analyze the packets or frames. It can see only raw bits 1s and 0s. Therefore, hub operates at the physical layer.

Consider the following bus network.



If ***A*** sends a signal it will be received by ***B***,***C*** and ***D***. The same thing happens in the hub. Therefore, hub is physically a star but logically a bus.

Therefore, now everybody uses hubs to make bus networks. The maximum length the signal can travel in one port is 100 m.

How to install a LAN using a Hub?

The required items are,

- Hub
- UTP Cables
- RJ45 Connectors
- Network Interface Card [NIC]

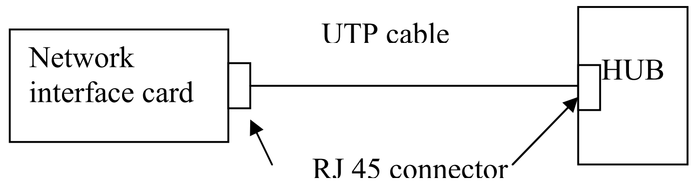
In each computer a NIC should be installed and configured.

Network Interface Card

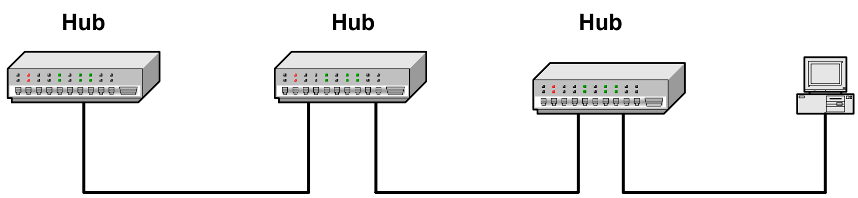
This is the interface between the computer and the LAN media. The NIC performs the following functions.

- Carrier sense
- Conversion of binary signal to Differential Manchester Coded Signal and vice versa.
- Media Access Control

The NIC can operate either in half duplex or full duplex mode. Some NICs can operate at 10 Mb/s whereas some can operate at both 10 Mb/s and 100 Mb/s.

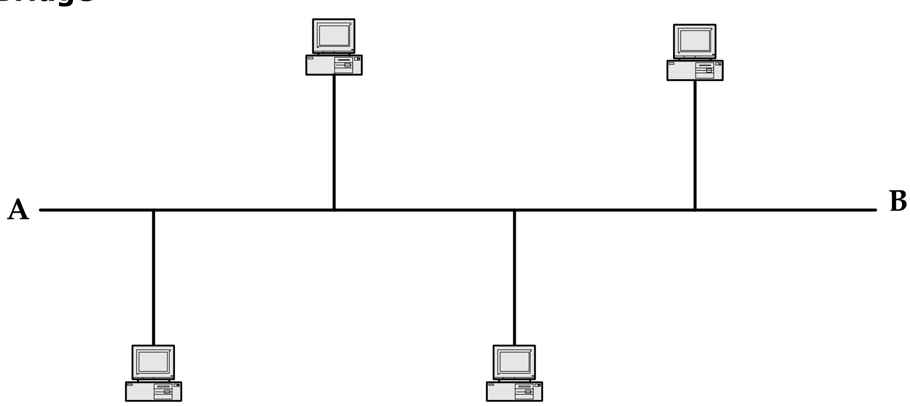


The maximum allowable length is 100 m. If you want to operate at more than 100m cascaded hubs can be used.



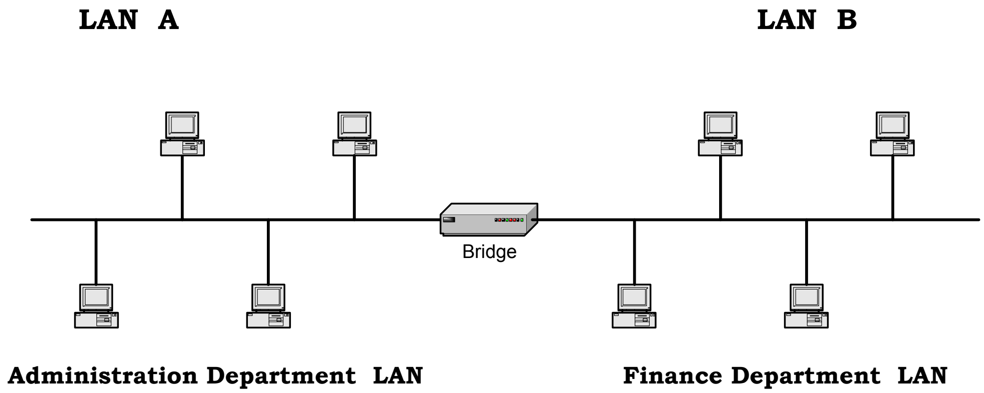
If is not recommended to cascade more than two hubs.

Bridge



Suppose a bus network has 6 Computers which are used by Finance Department and the Administration Department. At this stage collisions may not be affected for the performance of the network.

If the number of Computers increase to 20, it can affect the performance due to collisions. Therefore, it is advisable to separate the two departments into two LANs. But there should be provisions to access from one department to other department, whenever necessary. In order to satisfy this requirement a Bridge can be used.



Bridge is an intelligent device which can connect two or more similar types of LANs.

The operation of the Bridge is as follows.

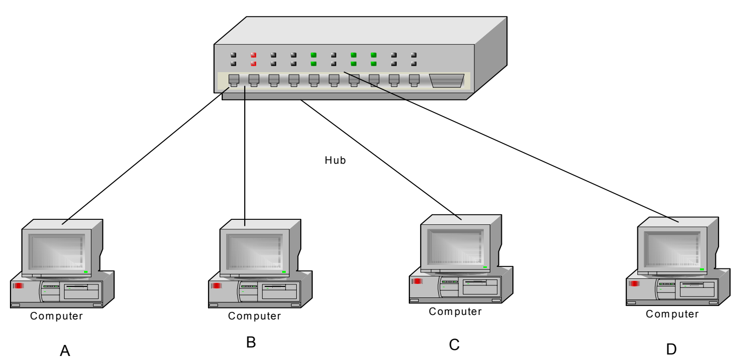
When it receives a frame, the destination MAC address is checked. If it is in the same network it is ignored. If it is in the other network it is transmitted to the other network. This has two advantages. It improves the performance and security.

The Bridge operates at the second layer of OSI model.
If any computer broadcasts a message it will be received by all computers in all the networks connected to the bridge. [Bridge can have more than two ports].

Consider the above example. It has,

- two segments.
- two collision domains.
- one broadcast domain.

Switches



Switch is an intelligent device. When it receives a frame, it analyzes the frame and obtains the destination address. Then it refers the MAC address table and finds out the port to which the destination address host is connected. Then the two ports are switches to make a connection between the two hosts.

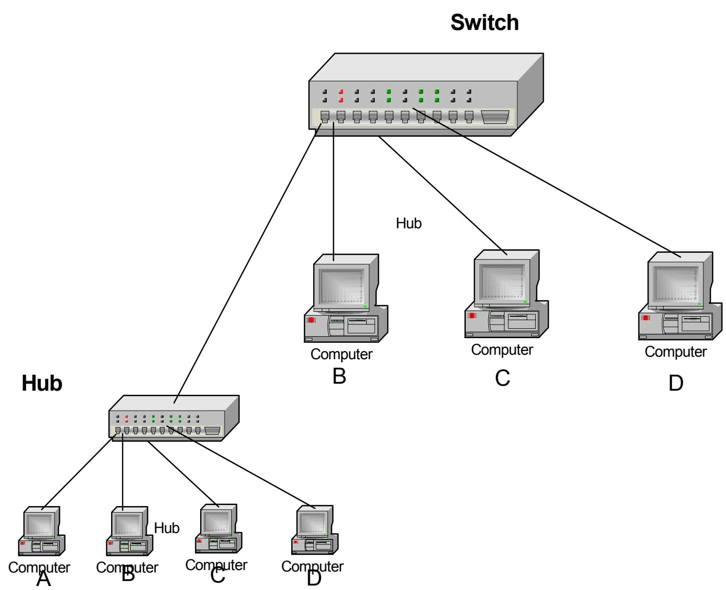
In the above figure, suppose ***A*** wants to send a frame to ***B*** and, ***C*** wants to send a frame to ***D***. The switch makes the connections ***A*** to ***B*** and ***C*** to ***D*** simultaneously. Therefore, switch has higher performance compared to a hub when there are many computers connected to the network.

Switch can operate with half duplex or full duplex modes. Some switches support both 10 Mb/s and 100 Mb/s bandwidths.

If one computer broadcasts a signal it is broadcasted to all computers by the switch.

Therefore, in a switch one port is one segment and one collision domain. All ports are in one broadcast domain.

Switch can operates with hubs also.



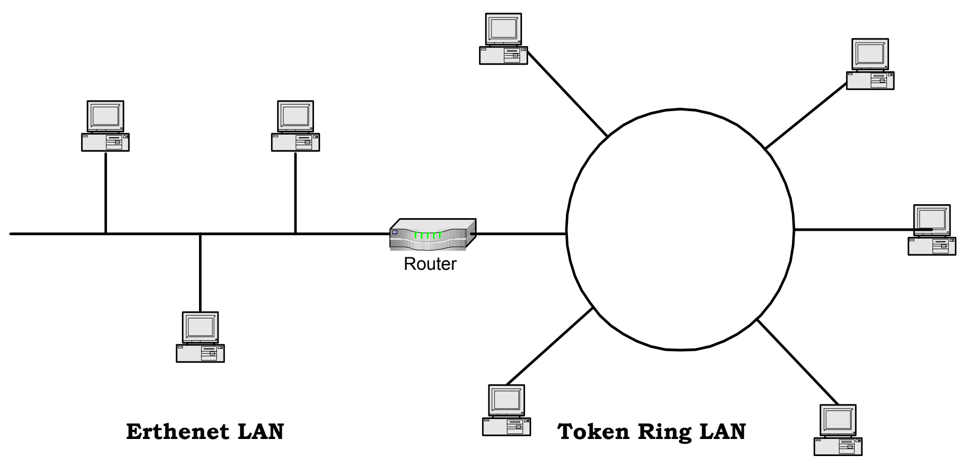
If the distance from the switch to the host is more than 100m, a hub or a repeater can be used in between of the switch and the host.

Normally switches operate at layer 2 and they are called L2 switches or Ethernet switches.

Some switches can operate at layer 3 where they can analyze the network layer packet [e.g. IP Packet] and take decisions. They are called L3 switches.

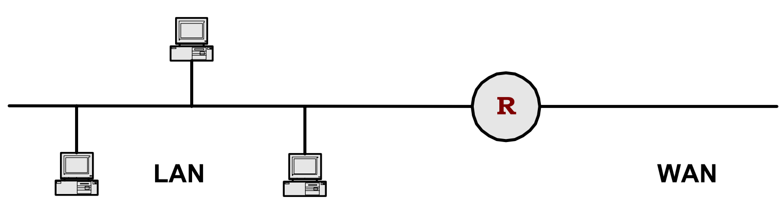
Routers

Routers can be used to connect dissimilar types of LANs or LANs and WANs.



Router has two types of ports.

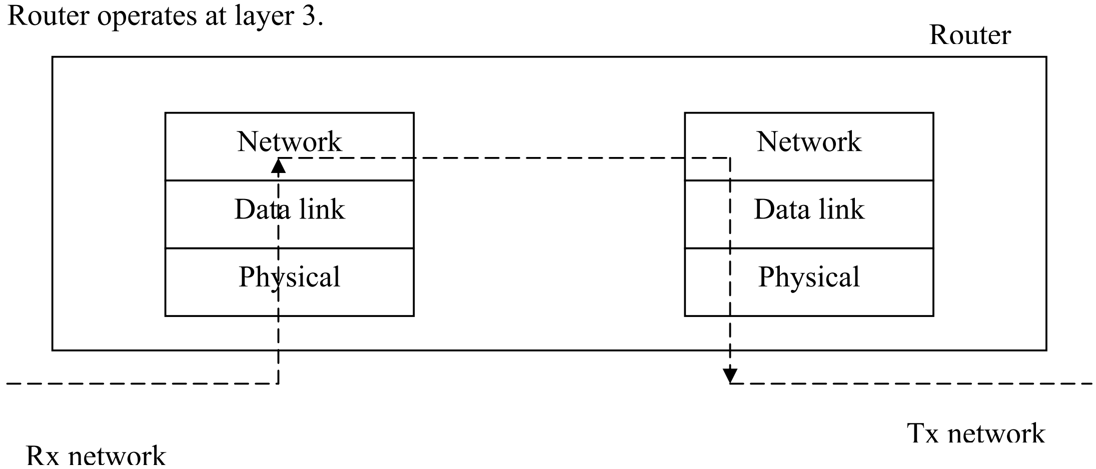
- LAN Ports
- WAN Ports



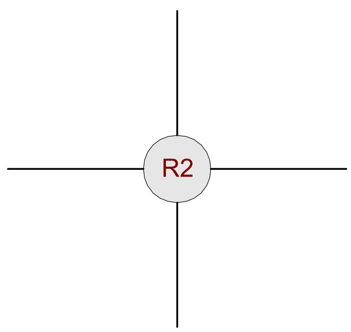
The WAN port can be connected to a,

- Leased Line
- Frame Relay connection
- ISDN connection
- ADSL connection
- Dial Up connection
- IP-VPN connection

Router operates at layer 3.



The raw bits are received from media to Physical Layer. It sends to the data to the Data Link Layer. It removes the DLC header and trailer and sends it to the Network Layer. It analyses the packet and finds out the destination address. The Router maintains a routing table. By referring to the routing table it can decide to which port the packet is to be sent. Then it changes some parameters in the network header [IP header] and sends it to the DLC. It adds the DLC header and trailer and sends to Physical Layer. The Physical Layer sends the raw bits [1s and 0s] to the correct port [interface].



Suppose a Router, has four ports. The router does not perform broadcasting to all ports.

Gateway

The Router operates at the third layer. The gateway can operate at high layers such as Transport or Application Layers.