Sree Siddaganga Education Society ®Estd: 1966-67

Sree Siddaganga College of Arts Science & Commerce

Affiliated to Tumkur University

Admission for Both Boys & Girls

A College with a difference-NAAC B++

B.H.Road, Tumkur-572102.Ph:0816-2278569, 8277338148

Website: - www.sscasc.in

e-mail:-principal.sscasc@gmail.com

STUDY MATERIAL

Subject:-Computer Networks

DEPARTMENT OF COMPUTER SCIENCE

6th Semester BSc

CHAPTER 1.DATA COMMUNICATIONS

DATA COMMUNICATION:

"Data communication refers to exchange of digital information between two digital devices".

COMPONENTS OF DATA COMMUNICATION:

Message: It is the information to be communicated such as text, numbers, and images. Sound, video, etc.,

Sender: The device that sends the messages.

Ex: a computer, a telephone handset, video camera.

Receiver: The device that receives the message.

Ex: a computer, a telephone handset, video camera.

Medium: The physical path by which a message travels from sender to receiver.

Ex: Twisted pair wires, coaxial cable, and optical fiber.

Protocol:- A set of rules that govern data communication.

Ex: HTTP [**H**yper **T**ext **T**ransfer **P**rotocol]
FTP [**F**ile **T**ransfer **P**rotocol]
TELNET [**Tel**ecommunication **Net**work]
SMTP [**S**imple **M**ail **T**ransfer **P**rotocol].
TCP [**T**ransmission **C**ontrol **P**rotocol]

COMPUTER NETWORKS:

"A Computer network is set of interconnected autonomous computers".

Elements of Network:

Every network includes:

- At least two computers server and client.
- Networking Interface card.
- A transmission medium.
- Network Operating System.(windows NT,Novell NetWare,UNIX,LINUX)

GOALS OF NETWORK:

- Sharing of resources.
- Providing Reliability.
- Reducing Cost.
- Providing Better Performance.
- Providing powerful communication medium to the users.

NETWORK CRITERIA:

A network is considered effective and efficient if the following criteria are met:

- **Performance:** performance of the network is measured in terms of
  i. **Transit time** - The amount of time required for the message to travel from one device to another.
  ii. **Response Time**- the elapsed time between request and response.
- **Reliability:** Network reliability is measured in terms of
  i. A*ccuracy* of deliver

ii. *Frequency* of deliver

iii. *Recovery* time of a network from failure.

- **Security:** Protecting data from unauthorized access and viruses.

LINE CONFIGURATION:

It refers to the manner in which two or more communication devices attach to a link.

Two types:

Point-to-point: It provides a dedicated link between two devices.

Ex: Remote controlled TV

Multi-point: In this more than two devices share a single link.

NETWORK TOPOLOGY:

"A network topology is the shape /physical connectivity of a network".

BUS TOPOLOGY:

"*In a bus topology all the network devices are connected to single long cable called BUS/BACKBONE of the network by means of connectors*"

Advantages:

1. Easy to use and install.
2. Requires less cabling.
3. Failure of one node does not affect the rest of the network.
4. Less expensive.

Disadvantages:

1. Cannot be suitable for larger networks.
2. Fault identification is difficult.
3. Failure of the cable will shut down the entire network.

STAR TOPOLOGY:

"*In a star topology all the network devices are connected to central device HUB*".

Each device requires a single cable to connect to HUB.

If any device wants to send the data to any other device on the network. First it sends to the HUB then the HUB relays the data to the destination device.

This is the most widely used topology.

Advantages:

1. Easy to use and install.
2. Easy to reconfigure.
3. Failure of one node does not affect the rest of the network.
4. Fault identification is easy.

Disadvantages:

1. If the HUB fails the entire network fails.
2. Require large amount of cable.
3. It is expensive.

RING TOPOLOGY:

"*In a ring topology all the devices are connected to one another in the shape of a closed ring so that each device is connected to only two devices on either side of it*"

Data is transmitted around the ring in one direction only from device to device until it reaches its destination.

Advantages:

1. Easy to use and install.
2. Easy to reconfigure.
3. All the nodes on the network have equal access to the network.
4. Fault identification is easy.

Disadvantages:

1. Signal is passed only in one direction.
2. A break in the ring can disable the entire network.
3. Adding or removing the node disrupts the entire network.

MESH TOPOLOGY:

"*In a mesh topology each node as a connection to every other node in the network*". So a fully connected mesh network has **n (n-1)/2** channels to link „n"

Devices.

To accommodate that many links every device on the network must have **„n-1"** input output ports.

Advantages:

1. It is robust and reliable.
2. Easy to reconfigure.
3. Data transfer rate is very fast.
4. Eliminates traffic problem due to dedicated link.
5. Fault isolation is easy.

Disadvantages:

1. Installation and re-configuration is very difficult.
2. It is very expensive.
3. Cable requirement is very high.
4. Large numbers of input or output ports are required for each node.

HYBRID TOPOLOGY:

"It is the combination of two or more topologies"

TRANSMISSION MODES:

The term transmission mode is used to define the "***direction of data signal flow between two linked devices".***

There are 3 types of transmission modes: **Simplex, Half Duplex, and Full Duplex**

Simplex::-In this mode, communication is unidirectional i.e., ***data flows in only one direction*** from transmitter to receiver.
Ex: Radio and TV broad casting, data flow from Keyboard to the computer or from computer to the monitor.

Half Duplex::- In this mode, "*the communication can takes place in both directions, but only in one direction at a time.*"
Ex: Internet Browsing, walkie- talkie.

Full Duplex::-In this mode, the data communication takes place in both directions at the same time.

Ex: Telephone Conversation

TYPES OF NETWORK:

Networks are classified into different type sbased on the size and structure.

LOCAL AREA NETWORK (LAN)

- Interconnection between devices within a single building.
- It covers an area up to 1-2 km.
- Usually owned by a single person or small organization.

METRAPOLITAN AREA NETWORK (MAN)

- Interconnection between devices within a city.
- It covers an area up to 10km.
- Owned by large organization like government.

WIDE AREA NETWORK (WAN)

- It provides long distance transmission of data over large geographical area.

Ex: Internet.

INTERNETWORK (Internet)

"*An internetwork is a network of network*"

CHAPTER 2_PROTOCOLS AND STANDARDS 3:-APPLICATION LAYER

EVOLUTION OF NETWORK ARCHITECTURE AND SERVICES

A communication network is a set of equipment and facilities that provide service like the transmission of information between the users located at various geographical points.

There is a tremendous improvement in transmission bit rate (rate rate) over the last 150 years. The evolution of network services can be schematically represented as follows:

Telegraph Networks

Message switching & digital transmission

→

Telephone networks

Circuit switching

Analog transmission Digital transmission

TELEGRAPH („tele" means „far" and „graphain" means „writing" in GREEK) NETWORKS AND MESSAGE SWITCHING

"*A telegraph network is a* ***telecommunication network*** *for* ***transmitting text messages*** *over a* ***long distance*** *using some form of* ***„on-off"*** *coding system."*

- ✓ The first telegraphy network was demonstrated by "**Samuel B. Morse**" in **1837**.
- ✓ In Morse method the text was encoded into sequence of "**dots & dashes**" called „**Morse**" code.
- ✓ Each dot or dash is communicated by transmitting short and long pulses of electrical current over a copper wire.
- ✓ "In this network a message (telegram) would arrive at a telegraph station, and an operator would store the message until the desired communication line became available and then would forward the message to next appropriate station".
- ✓ The store and forward process would be repeated at each intermediate station until the message arrived at the destination station. This technique is called **message switching**.

TELEPHONE („tele" means „far" and „phone" means „voice" in GREEK) NETWORKS AND CIRCUIT SWITCHING

"A telephone network is a telecommunication network used for telephone calls between two or more parties".

- ✓ The first telephone device was developed by "**Alexander Graham Bell**" in 1876.
- ✓ A telephone network consists of your phone at home that is connected to the central office. The central office is in turn connected

to a hierarchical phone network.

- ✓ In 1878 telephone switches were introduced to allow human operator to interconnect telephone user on demand.

Traditionally a telephone call has 3 phases.

1. **Call set up phase**:

- In this phase the originating user picks the telephone and in the process activates signal in the circuit that connects it to the telephone office.
- The signal alerts the operator in a central office (CO) that a connection is requested.
- Then the operator speaks to the originating user and takes the requested destination station number and number and checks to see whether the desired user is available.
- If so, operator establishes a connection by inserting the two ends of a cord to the socket that terminates the lines of the two users.

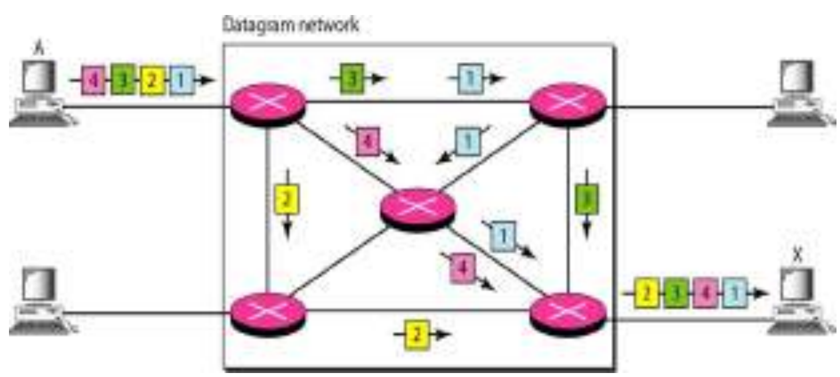
2. **Information Transfer Phase:** Information flows from one user to another.

3. **Connection Released:** When users are done with their conversations, they "hang up" their telephones, which generate a signal that the call is complete.

The connections are then made available for others.

PACKET SWITCHING NETWORKS and INTERNET

Packet switching networks are designed to provide packet transfer service, where a packet is a variable – length block of information.



As shown in fig packets are transferred from switch to switch until they are delivered at the destination. The messages are then recovered by reassembling individual packets at the destination.

.ARPANET

- The „ARPANET" was the first packet switching network developed in late 1960s to interconnect computers using packet switching across a wide area Network [WAN].
- Each packet consists of a header with a destination address attached to user information and is transmitted as a single unit across a network.
- The ARPANET consists of packet switches interconnected by communication lines that provide multiple paths for interconnecting host computers over wide geographical distances.
- ARPANET packet transmission service is ***connectionless*** in the sense no connection setup was required prior to the transmission of a packet.
- Each packet switch in the ARPANET contains a limited amount of ***buffering*** for holding packets before processing.
- To prevent packet switches from being congested, an ***end-to-end congestion control*** was developed.

INTERNET

- An internet is the "***interconnection of multiple networks into a single large network***".
- A protocol suite **TCP/IP** was developed to allow ***communication across multiple diverse networks*** on the internet.
- The **Internet Protocol (IP)** was developed to provide the ***connectionless transfer of packets*** called "data grams" across an internetwork.
- In IP the components of networks are interconnected by special packet switches called "***Gateways***" or "***Routers***". Each router directs the transfer of IP packets across the internet.
- IP provides **best-effort service**.

IP makes every effort to deliver the packets but takes no additional actions when packets are lost, corrupted or delivered out of order.
Hence IP provides unreliable service.

- IP uses hierarchical address space.

IP address consists of two parts: **Net-Id** and **Host-Id** and expressed in dotted-decimal notation
For Ex: 125.125.125.125.

- The Internet provides a space to refer to machines connected to the internet. For Ex: www.google.com
- Automatic translation of names to IP address is performed by Domain Name System (DNS).
- The Transmission Control Protocol (TCP) provides a reliable transfer of information over the Internet.

* The User datagram Protocol (UDP) provides an unreliable transfer of information blocks called datagram.

OSI MODEL

OSI (Open System Interconnect) model is a standard description of how messages should be transmitted between any two communicating parties in a network.

It was introduced by an ISO (International Standard Organization) in 1984. In this model divides the communication functions into 7 layers.

APPLICATION LAYER

It provides user interface and support for services such as e-mail, file transfer, database sharing specific responsibilities of application layer are:

1. NVT [**Network** virtual terminal]

   It consists of a telnet protocol which a local computer to logon to a remote computer.
2. **Accessing transferring & managing the files.**
3. **Mail services.**
4. **Directory services.**

PRESENTATION LAYER

It is the **"translator"** of the network.

Specific responsibilities of presentation layer are:

1. **Translation**

   Changing the format of a message that is used by the sender into mutually acceptable for transmission then at the destination changing that format into one understood by the receiver.
2. **Encryption**

   Encryption & decryption of data for security purpose.
3. **Compression**

   Compressing & decompressing data to make transmission more efficient.
4. **Security.**

   Validating password and login codes.

SESSION LAYER

The session layer is that network dialog controller specific responsibilities of session layer are:-

1. **Session management**

   Dividing a session into sessions by the introduction of checkpoints(marking the significant parts of the message) and separating long messages into shorter units called dialog units.
2. **Synchronization**

   Deciding in what order to pass the dialog units & making sure that the previous request has been fulfilled before the next one is sent.

3. **Dialog control**
   Deciding who sends the data and when.
4. **Graceful close:-**Ensuring that the exchange of data has been completed appropriately before the session closes

TRANSPORT LAYER

The transport layer is responsible for source to destination of delivery of the entire message specific responsibilities of transport layer are:

1. **End to end delivery of messages.**
2. **Segmentation and reassembly**

   Dividing a message into segments and marking each segment with a sequence number are used to reassembly the message correctly at the destination.
3. **Service point addressing**

   Garneting delivery of a message to the appropriate program running on the destination computer.

NETWORK LAYER

Network layer is responsible for end to end delivery of individual packets.

This layer provides two related services

- *Switching*
- *Routing*

Switching

Switching refers to temporary connection between physical links.

Routing

Selecting the best path for sending a packet

Specific responsibilities of network layer are:

1. **Source to destination delivery of packet.**
2. **Logical addressing**

   Inclusion of the source and destination address in the head of each packet.
3. **Address transformation**

   Interpreting logical address to find their physical equivalence.
4. **Multiplexing**

   Using a single physical link to carry data between many devices at the same time.

DATA LINK LAYER

Data link layer is responsible for moving frames from one node to another.

Specific responsibilities of data link layer are:

1. **Framing**

   It divides stream of bits received from the n/w layer into manageable data units called frames.
2. **Physical addressing**

   It adds header & failure that contain addresses and other control information to the beginning and end of the frame.
3. **Flow control**

   If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender. The data link layer imposes a flow control mechanism to avoid overwhelming the receiver.
4. **Error control**

   It adds a mechanism to delete and retransmit the damaged frames.
5. **Synchronization**

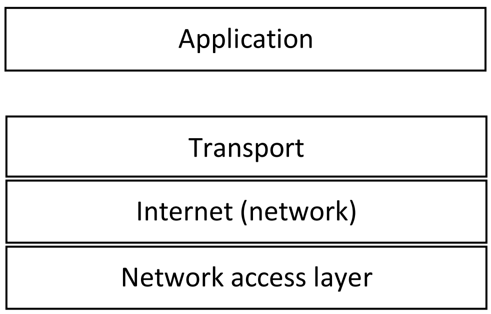
   Header contains bits to inform the receiver that a frame is arriving.

PHYSICAL LAYER

- The physical layer co-ordinates the functions required to transmit a bit stream over a physical medium.
- It is concerned with changing of bit stream into electromechanical signals and their transmission on to and across a medium.

TCP/IP MODEL

TCP/IP is a system allowing heterogeneous computers to communicate with each other.
The TCP/IP model consists of 4 layers: **Application, Transport, Internet, Network access layer.**



It is generally referred as protocol suite as various protocols are implemented at the different layers.

Fig: TCP/IP Protocol suite

APPLICATION LAYER:

The application layer provides the user with the interface to communication. The protocols reside inside the application layer are:

FTP (File Transfer Protocol): It takes care of the transmission files between computers.
HTTP (Hypertext transfer Protocol): It takes care of communication between web browser and a web server.
SMTP (Simple Mail Transfer Protocol): It is responsible for transmission of e-mails.
SNMP (Simple Network Management Protocol): SNMP is used for administration of the network. ***DHCP (Dynamic Host configuration Protocol):*** It is used for Dynamic allocation of IP addresses. **DNS (Domain Name Server):** DNS servers are responsible for translating domain names into IP addresses.

TRANSPORT LAYER:

Transport layer is responsible for reliable transmission of data i.e., all the data must arrive at the destination in the same order in which they are sent by the sender without any errors.
Transport layer uses 2 protocols: **TCP and UDP**

1) **TCP (Transmission Control Protocol):**

TCP provides **connection-oriented** and **reliable** transport data. i.e.,

1. A connection must be established between sender and the receiver before data transmission.
2. It has a mechanism to detect and retransmit the damaged or lost data.

At the sender TCP divides a stream of data into **segments** and adds **sequence number** to each segment. This numbers are used at the receiver to reassemble the message correctly and to identify and replace the segments lost during transmission.

2) **UDP (User Datagram Protocol):**

It provides **unreliable** and **connectionless** transport of data between two end points.

<u>INTERNET LAYER:</u>

Internet layer is concerned with network to network communication.
It is responsible for packetization, logical addressing and routing of data on the network.

<u>The most important protocol used in this layer is:</u>

IP (Internetworking protocol*):*

It is an **unreliable** and **connectionless** protocol.
Since it does not provide any error checking or tracking methods and does not guarantee data transmission.
It transport data as packets and called diagrams, each of which is routed separately.

<u>The IP uses 4 supporting Protocols:</u>

ARP (Address Resolution Protocol): It is used to find the physical address of the node when its IP address is known.

RARP (Reverse Address Resolution Protocol): It allows a host to discover its Internet address when its physical address is given.

ICMP (Internet Control Message Protocol):
It is a mechanism used by hosts and gateways to send query and error reporting messages.

IGMP (Internet Group Message Protocol):
It is used to facilitate the simultaneous transmission of a message to a group of recipients.

<u>NETWORK ACCESS LAYER:</u>

This layer is concerned with specifying the **transmission medium**, the **nature of the signals**, the data rate and related matters.

<u>Two important protocols implemented at this layer are:</u>

SLIP (Serial Line Internet Protocol):
A very simple protocol that provides only basic framing for IP.

PPP (Pont-to-Point Protocol):
A complex protocol that provides framing as well as many additional features that improve security and performance.

IP ADDRESS[logical address]

Ip address is a **numerical address** used to identify nodes on the computer network. It is a unique **32-bit number represented** by **dotted-decimal notation:**
i.e., It is written as **four 8-bit numbers** separated by **periods.**

The address structure is divided into 2 parts: **NETID** and **HOSTID.**
The network ID identifies the network to which the host computer is connected and is assigned by the InerNIC (Internet Network Information Center).
The hostID identifies the actual computer on the network and is assigned by the network administrator at the local site.

The IP address structure is divided into five classes:

Class A

0	1 7	8 31
0	Network ID	Host ID

- Class A reserved for governments.
- Begins from 1 to 126
- They have 8-bits for netID and remaining 24-bits for host ID.
- Ex: **35.**0.0.0

Class B

0	1	2 15	16 31
1	0	Network ID	Host ID

- Class B reserved for medium companies.
- Begins from 128 to 191.
- They have 16-bits for netID and 16-bits for hostID.
- Ex: **128.5.**0.0

Class C

0	1	2	3 23	24 31
1	1	0	Network ID	Host ID

- Class C reserved for small organizations.
- Begins from 192 to 223.
- They have 21-bits for netID and 8-bits for hostID.
- Ex:**192.33.33.**0

Class D

0	1	2	3	4 31
1	1	1	1	Multicast Address

- Class D reserved for multicast services that allow a host to send information to a group of host simultaneously.
- **255.255.255.255** (11111111.11111111.11111111.11111111)

Class E :-Reserved for experiments and future use.

LOOPBACK ADDRESS:

The IP address begins from **127 (127.0.0.1)** is used as loop back address. This means that it is used by the host computer **to send a message back to itself**. It is commonly used for troubleshooting and network testing.

DOMAIN NAME: Textual names given to each node of a computer network. it is written as words separated by periods.

Ex: www.google.com, www.w3schools.com

DOMAIN NAME SERVERS:

The database system which converts the domain name (www.howstuffworks.com) into IP addresses (70.42.25.251.42)

TCP/IP UTILITIES:

TCP/IP provides several tools used for troubleshooting, investigating and analyzing the network.

PING

- PING utility is used to test the connectivity across the network. i.e., to determine whether a host is online and available.
- PING uses ICMP echo messages to check connections between hosts by sending an echo packets and then listening for the reply packets.
- The destination then responds with an ECHO-RESPONSE packet.
- PING is used to measure the round-trip delay between two hosts.
  Round-trip-delay: is the time required for a packet to travel from source to destination and then back again from destination to source.
- The round-trip delay is indicated by the time-to-live(TTL) value.
- The TTL is the maximum no. of hops an IP packet is allowed to remain in network. When TTL reaches 0, router discard the packet.

TRACE ROUTE

- The trace route (tracert) utility is used to determine the route taken by data to reach from its local host to a particular destination.
- The sender first sends a UDP datagram with TTL=1 and invalid port to the specified destination.
- The first router to see the datagram sets the TTL field to zero, discards the datagram , and sends an ICMP Time Exceeded message to the sender.
- The information allows the sender to identify the first machine in the route. These way remaining nodes are identified by sending larger TTL fields.

NETSTAT

- NETSTAT utility is used display all active network connections (both incoming & outgoing) ,routing tables and other network statistics.
- NETSTAT will display statistics for both TCP and UDP, including protocol,

local address, foreign address and the TCP connection state.

- Because UDP is connectionless no connection information will be shown for UDP packets

IPCONFIG

- The IPCONFIG utility is used to display the IP Address, subnet mask, and default gateway for the host.
- It can be used to get information for each IP network interface for the host like DNS hostname, IP addresses of DNS servers, physical address of the NIC.

Cryptography is a method of protecting information and communications through the use of codes so that only those for whom the information is intended can read and process it. The pre-fix "crypt" means "hidden" or "vault" and the suffix "graphy" stands for "writing."

Introduction to Security

When you create systems that store and retrieve data, it is important to protect the data from unauthorized use, disclosure, modification or destruction. Ensuring that users have the proper authority to see the data, load new data, or update existing data is an important aspect of application development. Do all users need the same level of access to the data and to the functions provided by your applications? Are there subsets of users that need access to privileged functions? Are some documents restricted to certain classes of users? The answers to questions like these help provide the basis for the security requirements for your application.

CHAPTER 4.DATA LINK LAYER

ERROR DETECTION AND CORRECTION

What is an Error?

An error is the change or the mismatching take place between the data unit sent by transmitter and the data unit received by the receiver.

Ex: 1010101**0** sent by sender 1010101**1** received by receiver.
Here is an error of 1 bit.

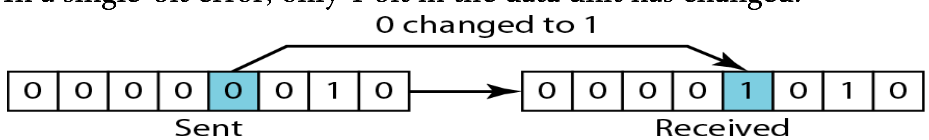
Types of Errors:

1) Single bit error

3) Burst error

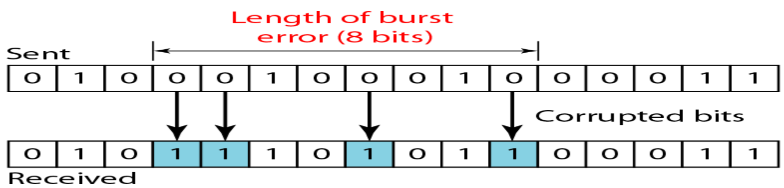
Single bit error

In a single-bit error, only 1 bit in the data unit has changed.



Burst error

A burst error means that 2 or more bits in the data unit have changed.



Error Control

Error control refers to mechanisms to **detect** and **correct** errors that occur in the transmission of frames.

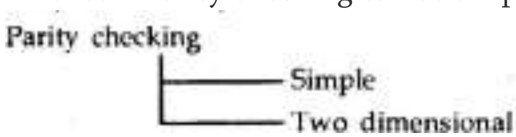
Error detection uses the concept of **redundancy,** which means adding extra bits for detecting errors at the destination.

There are 4 types of redundancy checks are common in data communication:

(a)Parity check
(b) Two dimensional parity check
(c)Checksum.
(d) Cyclic Redundancy check (CRC)

Parity check Method

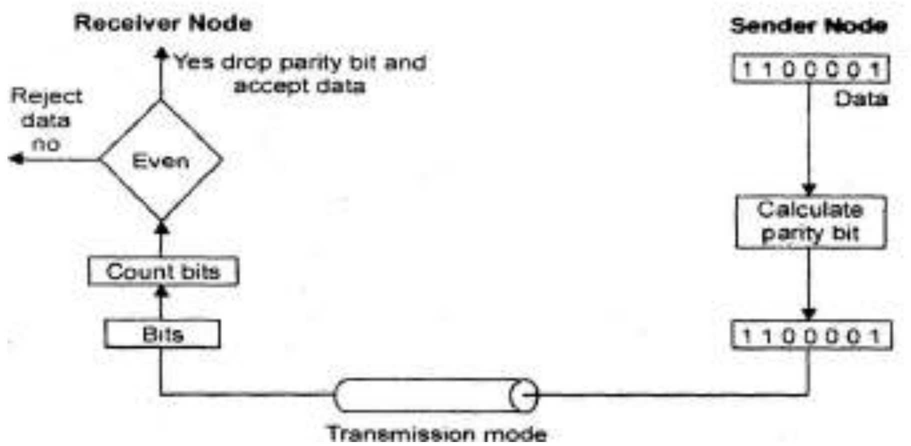
The most common and least expensive mechanism for error detection is the parity check. Parity checking can be simple or two-dimensional.



Simple Parity Check

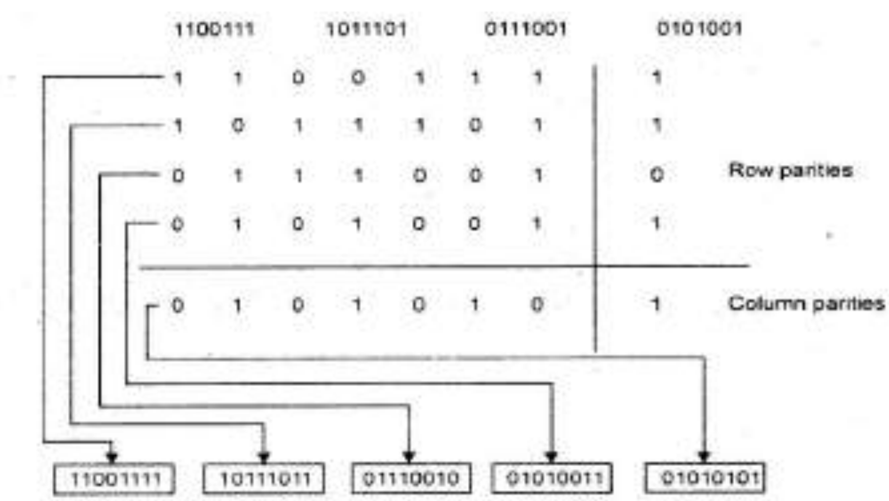
In this technique, a redundant bit, called a parity bit, is added to every data unit so that the total number of 1s in the unit (including the parity bit) becomes even (or odd).

Ex: Suppose we want to transmit the binary data unit 1100001



Two Dimensional Parity Check

- ✓ In this approach a block of bits is organized in a table (rows and columns).
- ✓ First we calculate the parity bit for each data unit.
- ✓ Then we organize them into table. Shows in fig we have four data units shown in four rows and eight columns.
- ✓ We then calculate the parity hit for each column and create a new row of 8 bits.
- ✓ They are the parity bits for the whole block.
- ✓ The first parity bit in the fifth row is calculated based on all first bits; the second parity bit is calculated based on all second bits, and so on.

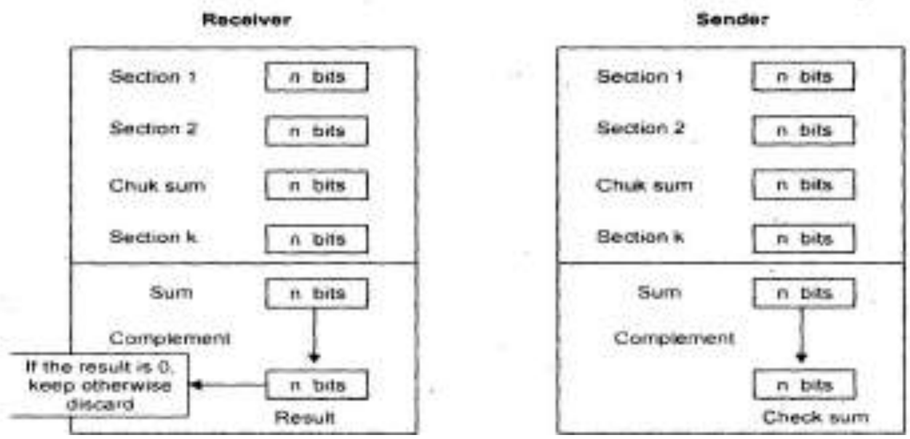


Data and Parity bits

- ✓ We then attach the 8 parity bits to the original data and sent them to the receiver.

Check sum

- ✓ Checksum is the one of the method used for error detection, based on the concept of redundancy.
- ✓ In this mechanism, the unit is divided into K sections, each of n bits.
- ✓ All sections are added using ones complement to get the sum.
- ✓ This is complemented and becomes the check sum.
- ✓ There after this check sum is sent with the data.
- ✓ At the receiver side the unit is divided into K sections each of n bits.
- ✓ All sections are added using ones complement to get the sum.
- ✓ The sum is complemented. If the result is zero data are accepted otherwise rejected.



CRC method of Error Detection

Cyclic Redundancy Check (CRC): Cyclic Redundancy check method is most powerful mechanism of error detecting.

Algorithms for computing the CRC:

Step1: Let „**r**" be the degree of generator polynomial **G(x).**
Multiply the message frame **M(x)** by $\mathbf{X}^r$. This is done by adding „r" zero bits to the low order message bit.
Now the message frame M(x) corresponds to the polynomial $\mathbf{X}^{r}\cdot$M(x).

Step 2: Divide the appended polynomial $\mathbf{X}^{r}\cdot$M(x) by G(x) using modulo-2 division to obtain the reminder **r(x).**

Step 3: Add the reminder r(x) to low order $\mathbf{X}^{r}\cdot$**M(x)** using **modulo-2** addition. The resultant code is the check summed frame or the CRC code to be transmitted.
This CRC is considered as **T(X).**
Step 4: The receiver while divide the **T(x)** with **G(x)** using **modulo-2 division**.

Step 5: if the result of this division gives a remainder as „0" then the data is accepted else it is discarded.

Modulo-2 Arithmetic

In modulo-2 arithmetic, we use only 0 and 1. Operations in this arithmetic are very simple. There is no carry over when we add and no borrow when we subtract one digit from another in a column

Modolo-2 Addition:	0+0=0	0+1=1	1+0=1	1+1=0
Modolo-2 subtraction:	0-0=0	0-1=1	1-0=1	1-1=0

NOTE: Associate bits with coefficients of a polynomial AS FOLLOWS:

1 0 1 1 0 1 1

$1x^6+0x^5+1x^4+1x^3+0x^2+1x+1$

$= x^6+x^4+x^3+x+1$

EXAMPLE :

Sender

- **M(x)** = 110011 x^5+x^4+x+1 (6 bits)
- **G(x)** = 11001 x^4+x^3+1 (5 bits, n = 4)

   4 bits of redundancy

- Form $\mathbf{x^r\ M(x)}$ 110011 0000

   $x^9+x^8+x^5+x^4$

- Divide $\mathbf{x^r M(x)}$ by **G(x)** to find **r(x)**

```
              100001
11001     ) 1100110000
            11001
            -----
                 10000
                 11001
                 -----
                  1001
```

r(x) = 1001

Send the block **110011 1001**

Receiver **100001**

```
          100001
      ___________
1100) 1100111001
      11001
          11001
          11001
          00000
```

No remainder

 Accept

HAMMING CODES

Hamming codes are the error correction codes that are widely used in reliable communication. The code is named after its inventor R.Hamming.

The code uses the number of parity bits located at certain position in the code word.

Hamming distance:

"The Hamming distance between two words is the number of differences between corresponding bits."

Valid Code Word: 10110
Error Code Word: 10100

XOR 00010

Hamming distance d=0+0+0+1+0=1.

The Hamming distance between

1011101 and **1001001** is 2.

Calculating the Hamming Code

1. All the positions in the code word that are powers of 2 (i.e., positions 1, 2, 4, 8….) Are for parity bits P_1, P2, P3, P4
2. The rest of the positions (i.e., positions 3, 5, 7, 9….) are for messages bits M1, M2, M3, and M4.
3. Construct bit location table as shown below.

Bit Designation	M7	M6	M5	P4	M3	P2	P1
Bit Location	7	6	5	4	3	2	1
Binary Location Number	111	110	101	100	011	010	001

Assignment of P1:

- The binary location number of Parity bit P1 has a 1 for its right most digit.
- Therefore parity bit P1 checks bit locations 1,3,5,and 7 and assigns value 0 or 1 according to even or odd parity.

Assignment of P2:

- The binary location number of Parity bit P2 has a 1 for its middle bit digit.
- Therefore parity bit P2 checks bit locations 2, 3, 6, and 7 and assigns value 0 or 1 according to even or odd parity.

Assignment of P4:

- The binary location number of Parity bit P4 has a 1 for its left most digit.
- Therefore parity bit P4 checks bit locations 4, 5, 6,and 7 and assigns value 0 or 1 according to even or odd parity.

Correcting Error:

- ✓ Once hamming code is constructed for the given information bits, it is sent to the receiver.
- ✓ At the receiver side, each parity bit, along with its corresponding group of bits is checked for proper parity.
- ✓ The correct parity of individual parity check is marked as „0" whereas wrong result is marked as „1".
- ✓ This word gives bit locations where error has occurred.
- ✓ If word has all bits „0" then there is no error in the hamming code.

Ex: Receiving hamming code is 1100111. Even parity is used. Locate and correct error.

Step1:

Constructing the bit location table:

Bit Designation	M7	M6	M5	P4	M3	P2	P1
Bit Location	7	6	5	4	3	2	1
Binary Location Number	111	110	101	100	011	010	001
Received code	1	1	0	0	1	1	1

Step 2: Check for parity bits

P1 checks bit locations 1,3,5 and 7
There are three 1"s in the group
Therefore parity check even parity is wrong…............1(LSB)

P2 checks bit locations 2, 3, 6, and 7
There are four 1"s in the group
Therefore parity check even parity is right… 0

P4 checks bit locations 4, 5, 6, and 7

There are two 1"s in the group
Therefore parity check even parity is right… 0

The resultant word is **001.**

This says that bit ion the position 1 is in error. Therefore the data is 1100110.

PEER TO PEER PROTOCOLS

Peer to Peer protocol involves the interaction of two processes or entities.
The purpose of a protocol is to provide service to a higher layer by executing layer-n protocol.

Service Models

The service provided a protocol is described by a service model.
The service model of a given layer specifies the manner in which information is transmitted.
There are 2 broad categories of service models:

1. Connection-oriented
2. Connectionless

Connection-oriented

In this connection is established between two end systems and then the data is transmitted. This involves 3 phases:

1) **Connection establishment:** In this phase connection is established between 2 end systems and during this process dedicated resources will be allocated for connection establishment and data transfer.
2) **Data Transfer:** In this phase the actual data is transfer takes place.

Once the data is transmitted it will reach the destination in the proper order as it is sent from the sender.
The acknowledgement for the received data is sent to the receiver.

3) Connection termination:

In this phase connection between two end systems is terminated.
The resources that are allocated for communication are released.

Ex: TCP is a connection oriented protocol.

Connectionless

- ✓ In this service connection setup is not done, instead individual blocks of information are transmitted and delivered based on the destination address provided.
- ✓ This service doesn‟t provide acknowledgement for transmitted information.
- ✓ If data is lost during transmission, no effort is made to be transmitted.

Ex: IP and UDP are connectionless protocols.

ARQ PROTOCOLs

- ✓ Error detection is one of the responsibilities of data link layer.
- ✓ If any error is detected in the transmission, specified frames are retransmitted .This process is called **Automatic Repeat reQuest.**
- ✓ The set of rules that will determine the operations for the sender and the receiver are named the ARQ protocols.

The 3 commonly used ARQ protocols are:

1. Stop and wait ARQ
2. Go Back N ARQ
3. Selective Repeat ARQ

Stop and wait ARQ

Working:

- Sender sends one frame and waits for acknowledgement.

- Receiver acknowledges the receiving of the frame.
- After receiving the acknowledgement, sender sends the next frame.
- In the case

1.The transmitted frame or returned acknowledgement was lost and
2.The sender‟s timer will timeout.

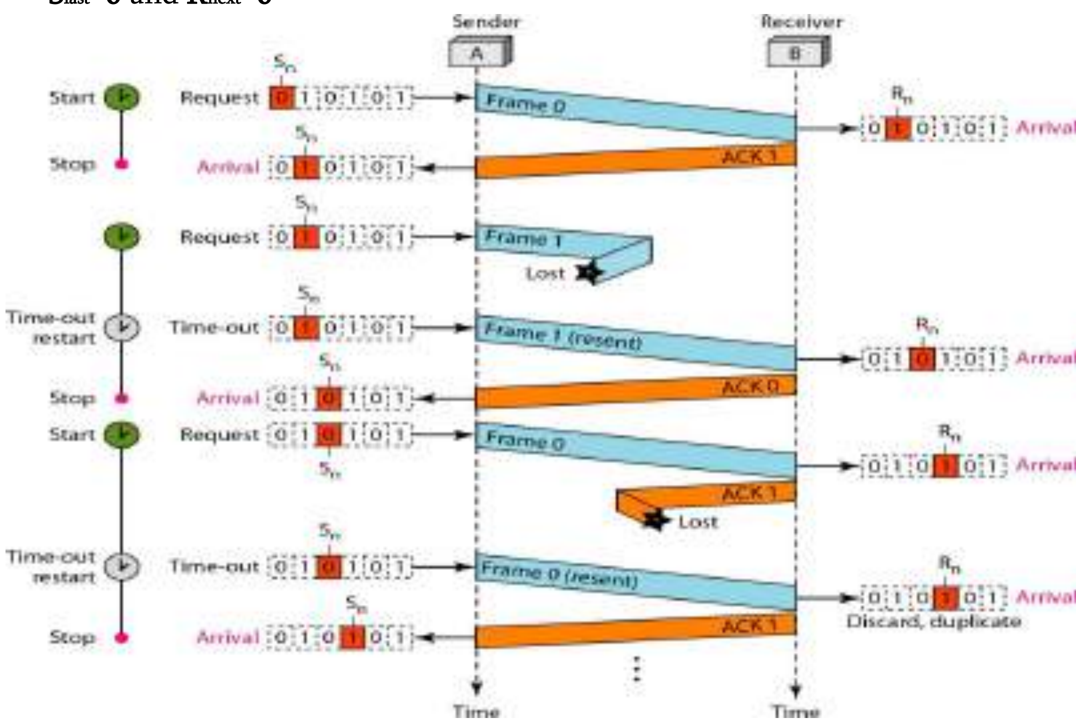
The sender retransmits the frame.

Use of ACK numbers.

❖ The ACK numbers always announces the sequence number of the next frame expected by the receiver.

❖ Ex: If frame 0 has arrived, the receiver sends an ACK frame with ACK 1(meaning frame 1 is expected next).

For this purpose, here sender and receiver will have a variable called as S_{last} and R_{next} initially **S_{last}=0** and **R_{next}=0**



1. Sender A sends a frame 0 and starts timer and store copy of the frame Here **S_{last}=0** and **R_{next}=0**

2. Receiver site **R_{next}=0** indicates Receiver is expecting frame with Sequence **number 0**. It will accept the frame and sends ACK frame with number=1 indicating receiver is expecting next frame with sequence **number=1.**

3. Sender receives **ACK 1** and transmission is success.

4.Now **S_{last} =1** i.e., sender now sends frame with **Seq no =1** but **frame 1 lost**.

5.Timeout occurs at sender site and assuming frame is lost sender retransmits frame 1.

6. Now **R_{next}=1** and receiver accepts frame 1 and sends ACK 0 i.e., it is expecting next frame = frame 0.

7.Now **S_{last}=0,** sender sends frame 0.

8. **R_{next}=0** Receiver will accept frame 0 and sends ACK 1, but ACK 1 lost.

9. Sender side timeout occurs and assuming frame is lost, it retransmits frame 0.

10. **R_{next}=1** so receiver will discard the frame 0, which is a duplicate frame and assuming previous ACK lost. It will resend ACK 1 to sender.

Disadvantages:

1) **Stop and wait ARQ works well only on channels that have low propagation delay.**
2) **In Stop and wait after each frame sent the host must wait for an ACK.** During that time it is wasting the bandwidth and becomes Inefficient.

To improve efficiency ACK should be sent after multiple frames

Alternatives: Sliding Window protocol

Go-back-N ARQ

Selective Repeat ARQ

Pipelining: "The process of beginning the next task before that previous task has ended".

Go-Back-N ARQ and Selective Repeat ARQ implements pipelining.

GO-BACK-N ARQ:

- To overcome the inefficient transmission that occurs in stop and wait ARQ, multiple frames must be in transition to keep the channel busy while waiting for acknowledgement.
- In this protocol we can send several frames before receiving acknowledgements; we keep a copy of these frames until the acknowledgement arrives.
- ACK to one frame validates all frames ahead of this frame called accumulated ACK
- If ACK for a frame is not received before timeout, all outstanding frames are retransmitted – That is why the protocol is called GO-BACK-N ARQ.

SEQUENCE NUMBERS:

- ✓ Frames from a sending station are numbered sequentially.
- ✓ Usually sequence number of each frame is included in the frame header.
- ✓ If the header of the frame allows m-bits for the sequence number, the sequence numbers range from **0 to 2^m-1.**

Ex:

If **m=2**, then the only sequence numbers are **0, 1, 2, 3**.

SLIDING WINDOW:

Go-Back-N uses the sliding window concept, which defines the range of sequence numbers.

It maintains 2 windows:

1) **Send sliding window** 2) **Receiver sliding window**.

Send Sliding Window:

- ✓ Send window is an imaginary box covering the sequence numbers of the frames which can be send.
- ✓ The maximum size of the window is **2^m-1**.

✓ The send window divides possible sequence numbers into **4** regions:

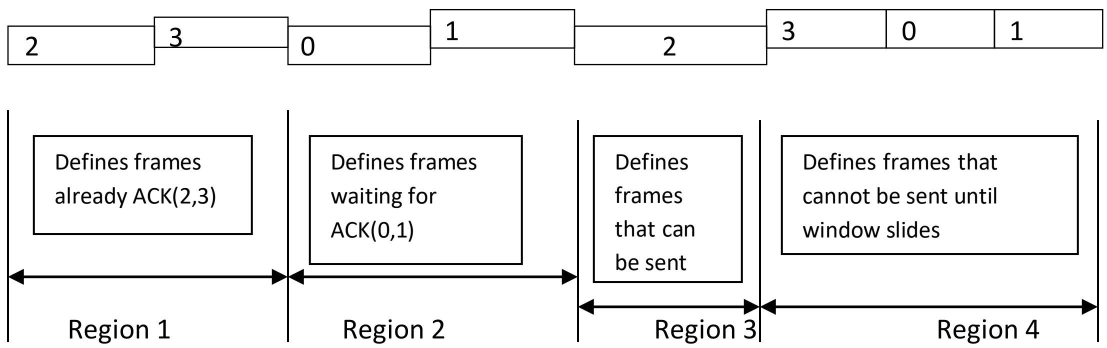


Fig: Send sliding window before sliding.

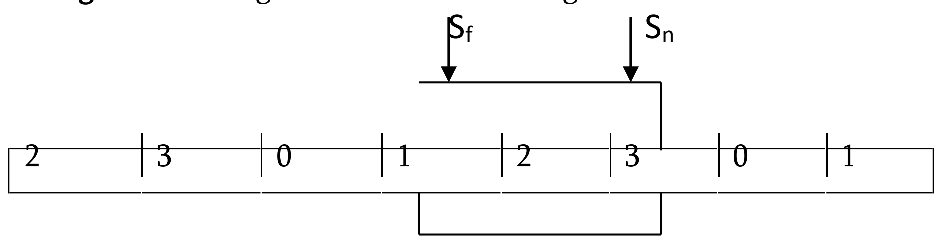


Fig: Send sliding window after sliding.

__**Receive window**__:

✓ The receive window makes that the correct data frames are received and that correct ACK are sent.

✓ The size of the receive window is always 1.

✓ The receiver expecting arrival of a specific frame.

✓ Any frame arrives out of order will be discarded and needs to be sent.

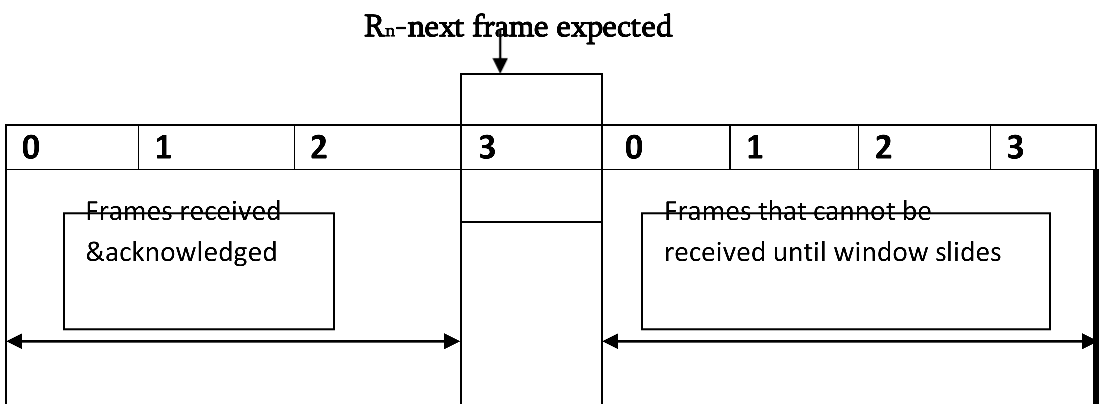


Fig: Receive window before sliding

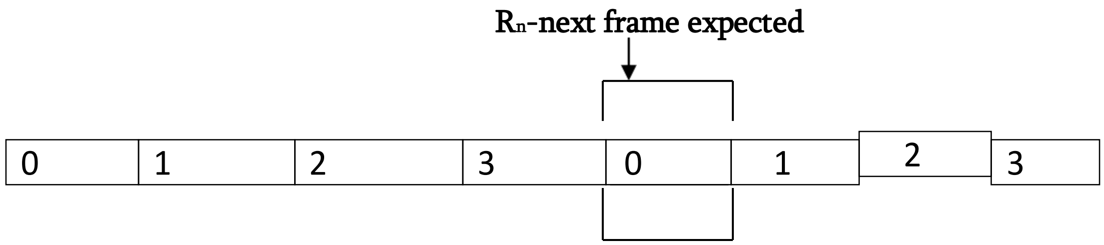


Fig: Receive window after sliding

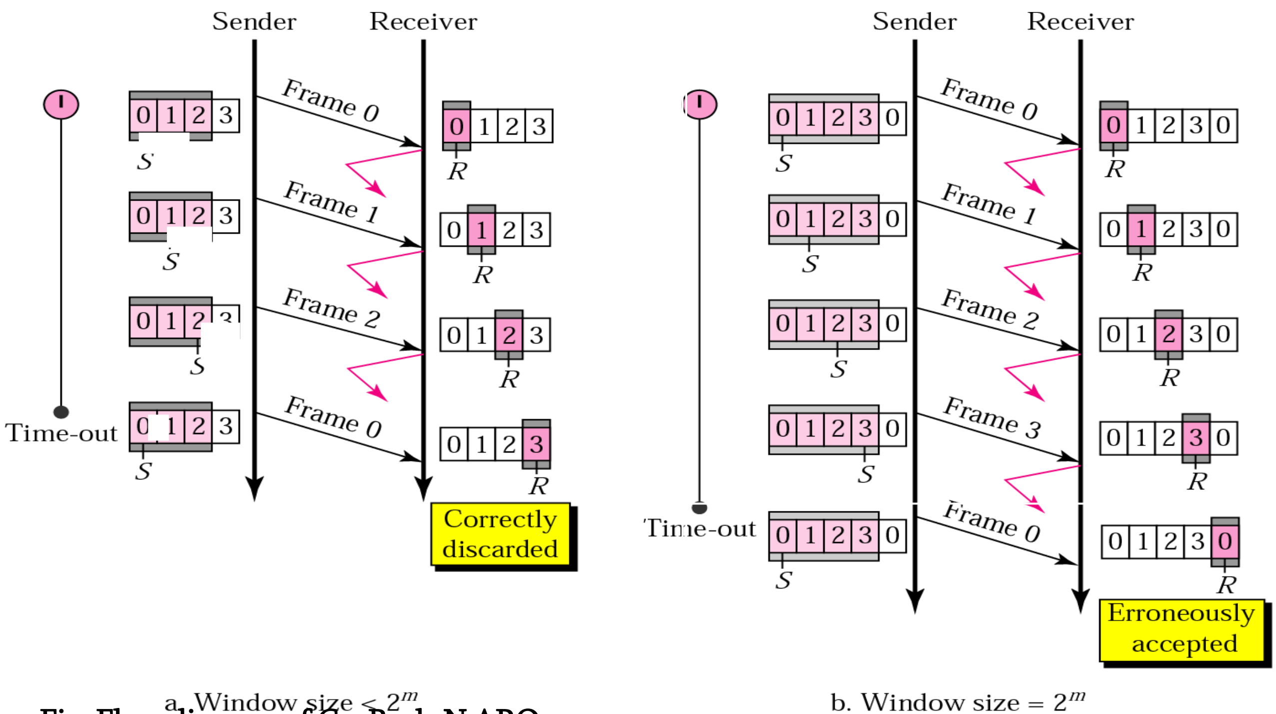


Fig: Flow diagram of Go-Back-N ARQ

1. Sender window size is 2^2-1=4-1=3

S_f =0 i.e., sent frame=0.

S_n i.e., next frame can be sent =1,2 Frame 0 sent

2. Receiver side Rn **=0** i.e., Receiver expecting frame=0, So it will accept frame 0 & sends ACK1.
3. ACK1 is lost.
4. Sender side **S_f=0,1**

   S_n =2

   Frames 1 sent
5. **R_n =1** , So it will accept frame 1 and send ACK2. ACK2 lost.
6. Frame 2 is sent , **R_n=2** accepted & sent ACKs and ACK is lost.
7. Timeout occurs at sender side, and sender did not receive ACK for none of the sent frames.
8. Assuming frames are lost, frame 0 is resent but now, **R_n=3.**

   i.e.**, receiver is expecting frame 3 but it receives frame 0 which is a duplicate frame so it discards that frame.**

SELECTIVE REPEAT ARQ:

- ✓ In **Go-Back-N ARQ** we have seen that if one frame is corrupted then all other N frames were needed to be resent. This drawback is overcome in this protocol.
- ✓ Instead of transmitting N frames, here only corrupted frame is retransmitted. This mechanism is called Selective Repeat ARQ.

Windows:

- ✓ This protocol uses two windows: **Send window** and **Receive Window**.
- ✓ The send window maximum size is $\mathbf{2^{m-1}}$. **Ex:** if m=2, $2^{2-1}=2$.\
- ✓ Sequence numbers ranges from 0 to 2^m-1. **Ex:** if m=2, $2^2-1=3$ 0,1,2,3,0,1,2,3,0,1,2,3………
- ✓ Receiver window size is also same as the sender window size $\mathbf{2^{m-1}}$.

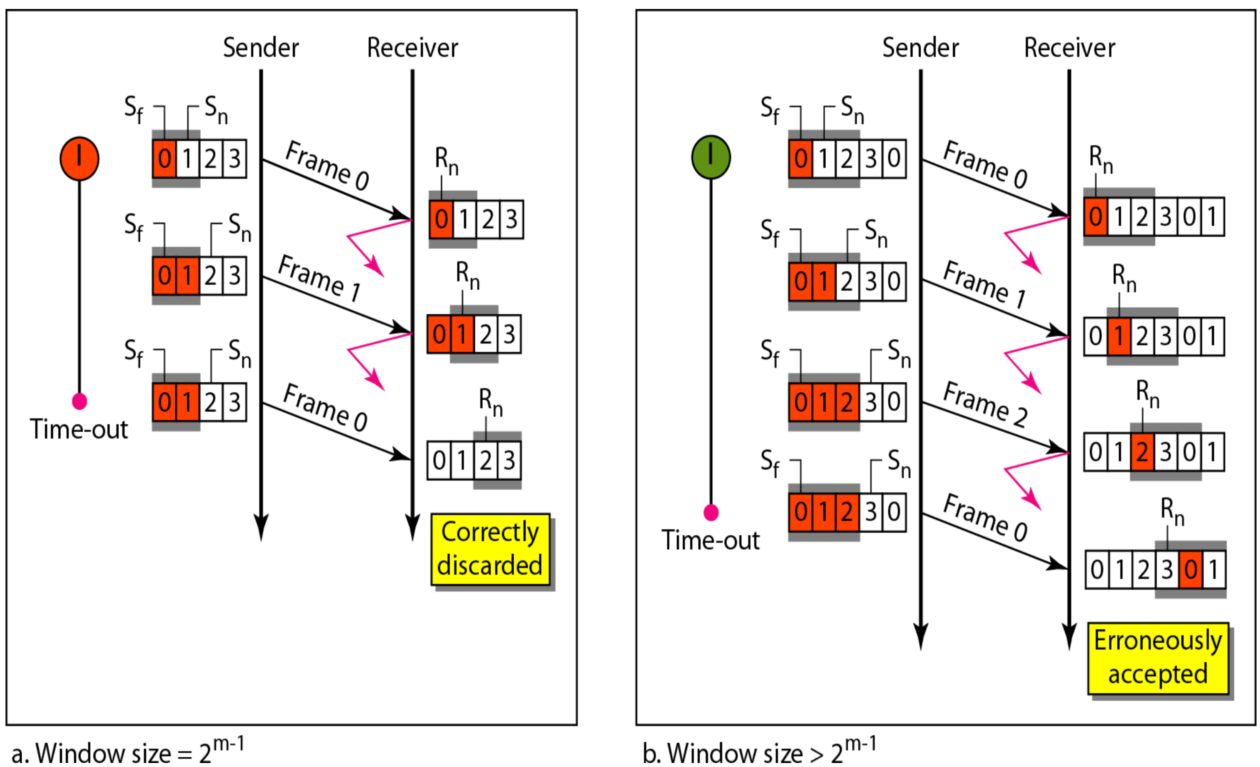


Fig: Flow diagram of Selective repeat ARQ

1. Sender window size is $\mathbf{2^{m-1}}= 2^{2-1}=2$. Receiver window size is $\mathbf{2^{m-1}}= 2^{2-1}=2$.

2. Sender A transmits frame 0. $S_f=0$ and Sn=1.

3. Receiver side Rn=0,1.
   Receiver will accept and sends ACK=1 but ACK 1 is lost.

4. Again sender transmits frame 1 . Receiver window shows 1,2. So it will accept frame and sends ACK2 but ACK2 is lost.

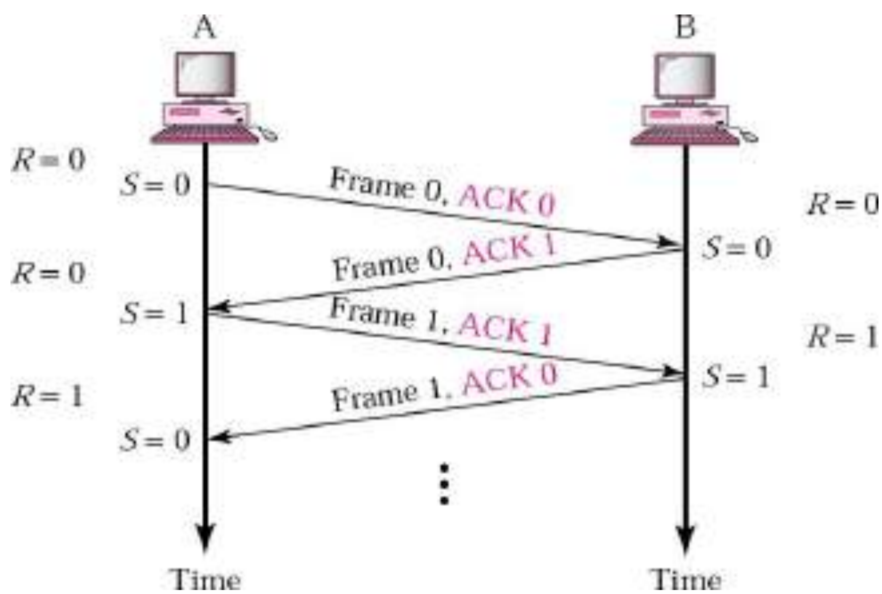
5.Timeout occurs at the sender site and sender didn‟t receive ACK for frame 0 & frame 1. So sender will retransmit frame 0.

6.Receiver window has 2, 3.

So receiver will correctly discard that frame 0 which is duplicate frame and sends an ACK for frame 0 and frame 1.

Piggybacking:

- ✓ Piggybacking is a technique used to improve the efficiency of bidirectional protocols.
- ✓ When a frame is carrying data from A to B, it can also carry control



information about frames from B and vice versa.

- ✓ Both the sender and receiver maintain control variable S and R.
- ✓ The sender sends Frame0 and ACK0 appended along with it.
- ✓ Similarly the receiver sends Frame0 with ACK1 appended to it.
- ✓ This way both Frame and Acknowledgement will concurrently increase optimal efficiency of bandwidth utilization.

FRAMING

- ✓ Framing involves identifying the beginning and end of a block of information.
- ✓ Frames can be **fixed** or **variable size**.
- ✓ In fixed size framing, there is no need for defining the boundaries of the frames. (Size itself is a delimiter).
- ✓ In variable size framing, we have defined the end of the current frame and beginning of the next frame.

There are two methods available for framing:

- **Byte stuffing(Character stuffing):**
  Use special characters to identify beginning and end of the frames.
- **Bit stuffing:**
  Use special bit patterns called flags to identify the beginning and end of the frames.

Byte stuffing:

- Also referred to as character stuffing.
- ASCII characters are used as framing delimiters (e.g. DLE STX and DLE ETX)
- The problem occurs when these character patterns occur within the “transparent” data.

Solution: sender stuffs an **extra DLE** into the data stream just before each occurrence of an "accidental" DLE in the data stream.

The data link layer on the receiving end unstuffs the DLE before giving the data to the network layer.

A	DLE	B	ETX	DLE	STX	E

Original data

DLE	**STX**	A	DLE	B	ETX	DLE	STX	E	**DLE**	**ETX**

After framing

DLE	**STX**	A	**DLE**	DLE	B	ETX	**DLE**	DLE	STX	E	**DLE**	**ETX**

After Byte stuffing

- **DLE(Data Link Escape)**
- **STX(Start of Text)**
- **ETX(End of Text)**

Bit stuffing:

- Each frame begins and ends with a special bit pattern called a flag byte [01111110]. {Note this is 7E in hex}
- Whenever sender data link layer encounters *five consecutive ones* in the data stream, it automatically stuffs a 0 bit into the outgoing stream.
- When the receiver sees *five consecutive incoming ones followed by a 0 bit*, it automatically dyestuffs the 0 bit before sending the data to the network layer.

Original data:

0110111111100111110111111111100000

After framing:

01111110 0110111111100111110111111111100000 **01111110**

After stuffing:

01111110 01101111101100111110011111011111000000 **01111110**

POINT –TO-POINT PROTOCOL

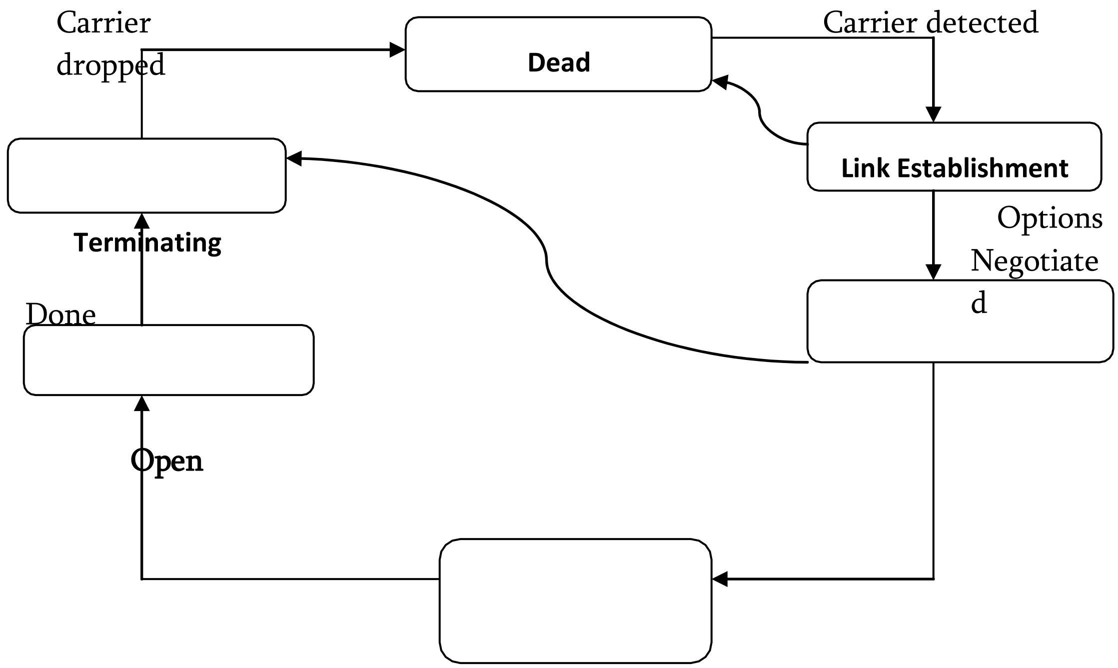
- Point-To-Point protocol is a collection of protocols(LCP,NCP,PAP, CHAP) Used for point-to-point communication.
- It is a byte-oriented protocol.

Services Provides PPP are:

1. PPP defines the format of the frame that has to be exchanged between two devices.
2. It uses LCP (Link Control Protocol) to establish communication over a PPP link.
3. Provides a method for encapsulating multiple data grams.

i.e., It transfer packets that are produced by different network layer protocols using NCP (Network Control Protocol).

<u>**The Service provided by PPP can be understood by a simple phase diagram:**</u>



Link Dead:
In this phase the link is not used.
There is no active no carrier in this phase

Lank establishment:
If the carrier is detected ppp enter this phase.
In this phase with the help of LCP options are negotiated.
If the negotiation is success, it enters into authentication phase.

Authenticate:
It is an optional phase
After the LCP setup the link, the password authentication protocol (PAP) will authenticate the sender. If authentication fails, PAP informs the LCP to terminate the link, if successful control goes to next phase.

Network layer protocol:
In this phase Negotiation over network layer protocol occurs with the help of network control protocol (NCP).

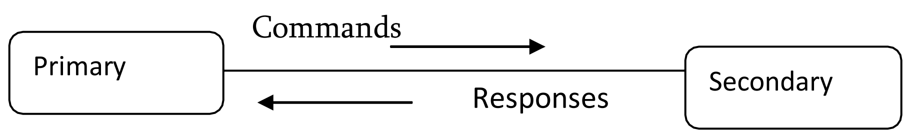
Open:
In this phase actual transfer of data packets takes place.

Terminate:
Once data transfer is completed connection should be terminated NCP configuration

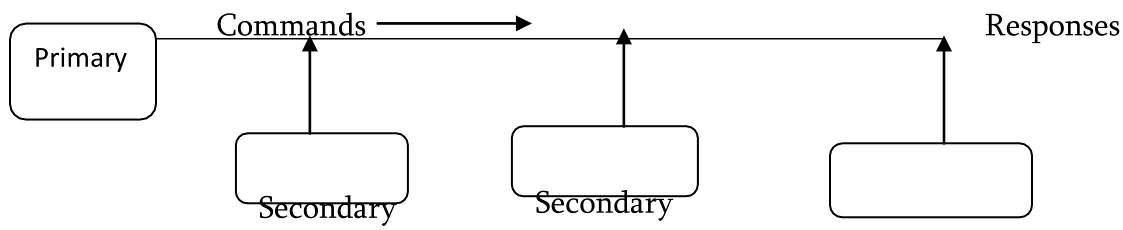
HDLC DEFINES Two TYPES OF DATA TRANSFER MODE:

Normal Response Mode (NRM):

- Here we have one primary station and multiple secondary stations.
- The station configuration is unbalanced.
- The primary station can send commands. And secondary station can only respond.
- This type can be used for point-to-point and multipoint links.



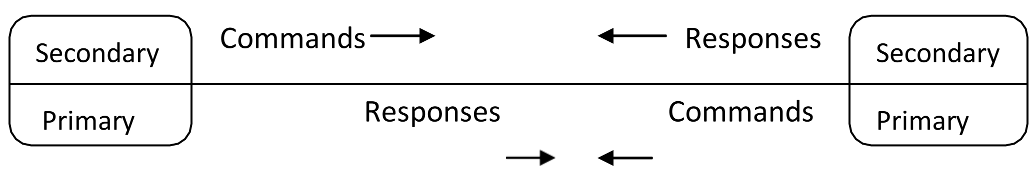
(a) point-to-point



(b) Multi-point

Asynchronous Balanced Mode (ABM) :

- Mainly used in point-to-point links, for communication between combined stations.



HDLC FRAME FORMAT:

Flag	Address	Control	Information	FCS	Flag

Flag field: The Flag field of an HDLC frame is and 8-bit sequence with the pattern 01111110, which indicates the beginning and end of the frame.

Address field: It contains the address of the secondary station.

If a primary station is created the frame, it contains a to address.

If a secondary station created the frame, it contains the From address.

Control field: It is used for flow and error control.

Information field: Contains the actual users data.

FCS: the frame check sequence number is the HDLC error detection field

HDLC DEFINES 3 TYPES OF FRAMES:

1) **Information frames**:
They are used to carry user data and control information related to user data.
2) Supervisory frames.
They are used for error and flow control purposes and hence contains send and receive sequence numbers.
3) Un numbered frames:
They are reserved for system management.

Control field determines the type of the frame:

I-FRAME:

0	N(S)	P/F	N(R)

S-FRAME:

1	0	S	S	P/F	N(R)

RAME:

1	1	M	M	P/F	M	M	M

N(S) =Sender sequence number
N(R) =receiver sequence number

S=supervisory bit
If
SS=00 Receiver Ready (RR) frame.
SS=01 Receiver Reject (REJ) frame.
SS=10 Receiver Not Ready (RNR)
frame. **SS=11**
Selective Reject (SREJ) frame.

P/F=poll or Final bit
If
P=1, then frame sent from primary to secondary
F=1, then frame is sent from secondary to primary.

M=Un numbered frame

Local Area Networks (LANs)

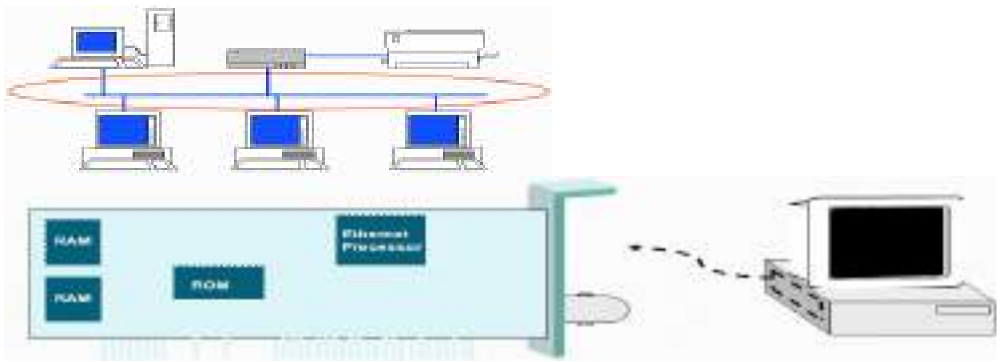
Local Area Networks are privately-owned networks within a small area, usually a single building or campus of up to a few kilometers.

LAN characteristics are determined by

Topologies
MAC (Medium Access Control)
Transmission media
Size of coverage

LAN structure

Explain the components of LAN structure?

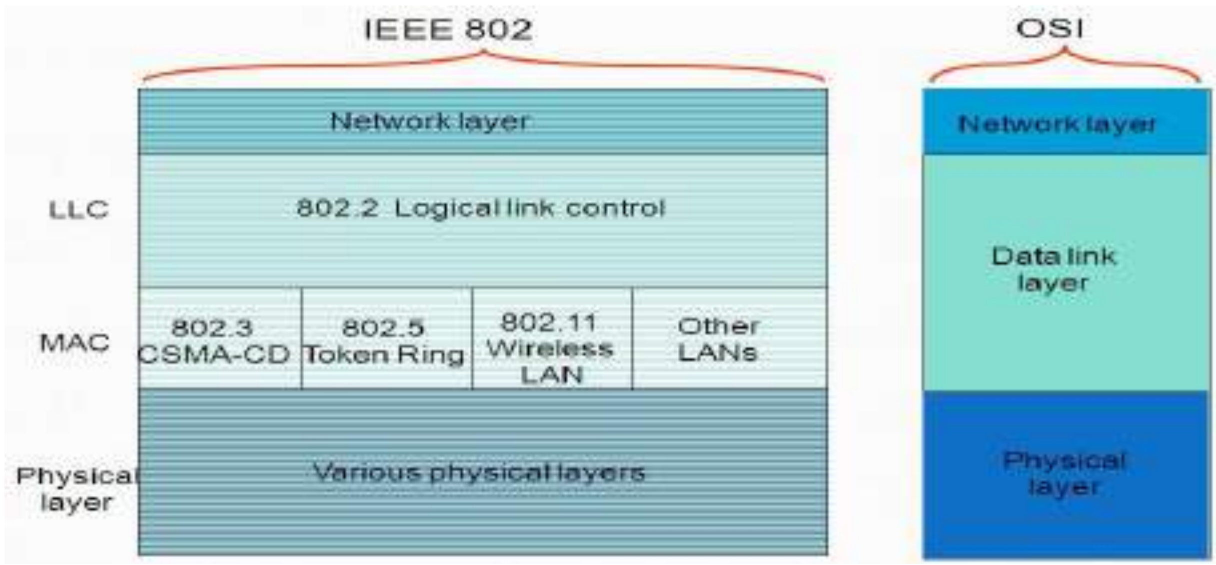


- A typical LAN consists of **number of computers** and **network devices** such **as printers** which are interconnected by **cabling system** as shown below:

- A cabling system can be arranged in **bus, ring, or star topology**.
- The cabling system may use **twisted pair cable**, **coaxial cable**, or **optical fiber** transmissions media.
- The computers and network devices are connected to the cabling system through a **Network Interface Card (NIC)** or **LAN adapter card.**

The MAC Sub-layer

- In 1985, the computer society of the IEEE started a project called project 802, to set standards to enable inter communication among equipment from a variety of manufacturers.
- Project 802 does not replace any part of the OSI model.
- Instead it specifies the functions of physical and data link layer of the major LAN protocols.
- The relationship of the 802 standard to the OSI model is as shown in the fig below:



- The IEEE has subdivided the data link layer into two sub layers:
  LLC (Logical Link Control) and **MAC (**Medium Access Control)
- The MAC sub layer deals with the problem of co-coordinating access to the shared physical medium.
- IEEE has defined several MAC standards namely:
  IEEE 802.3 (Ethernet) IEEE
  802.4(Token Bus) IEEE
  802.5(Token Ring) IEEE
  802.11(Wireless LAN)
- The LLC sub layer is to provide flow and error control for the upper layer protocols.

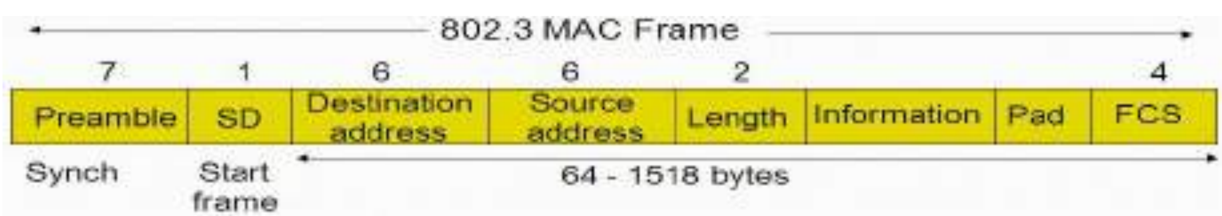
<u>Expand MAC and LLC .Name any one protocol used in MAC?</u>

MAC: Medium Access Control
LLC: Logical Link Control
The protocols used in MAC are: **CSMA-CD, Token Ring.**

<u>IEEE 802.3 (ETHERNET)</u>

It is a LAN protocol that is used in Bus and Star topologies and implements CSMA/CD as the medium access method.

<u>Ethernet Frame format</u>



- **Preamble** is followed by start delimiter that consists of the pattern 10101011 (11 indicates start of the frame)

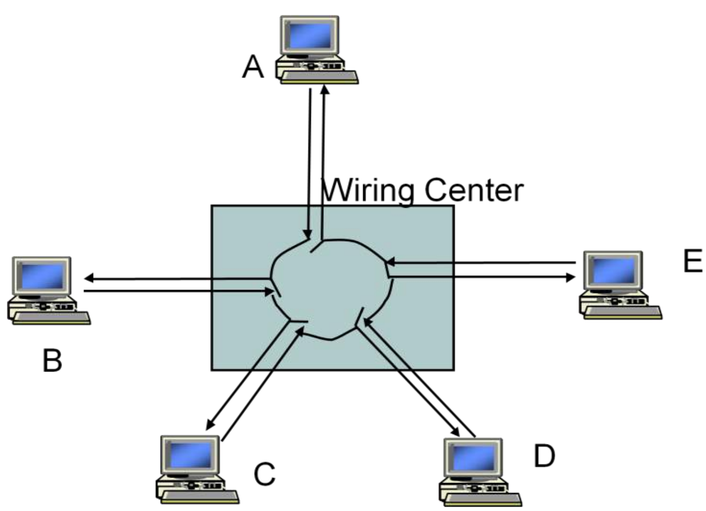
It helps receivers synchronize their clocks to transmitter clock.
Receivers look for change in 10 patterns.

- **Destination and source addresses** are 6 byte long which is used to multicast the frames to a group of users.
- **Length filed** indicates the number of bytes in information field i.e., the longest allowable frame is 1518 bytes.
- **Pad field** ensures minimum frame size is 64 bytes.
- **FCS** field is is 32-bit CRC value used for error checking.

IEEE 802.5 Token Ring

Explain the working and frame format of token ring? Or with a neat frame format explain IEEE 802.5 LAN standard?

- ✓ IEEE 802.5 LAN standard uses ring topology for their connection and the information flows from one direction along the ring from source to the destination and back to the source.
- ✓ The key notion is that MAC is provided via a small frame called a token that circulates around the ring.
- ✓ Possession of the token grants the right to transmit.



IEEE 802.5/Token Ring Frame Formats

Token Frame Format

SD	AC	ED

Data Frame Format

1	1	1	2 or 6	2 or 6		4	1	1
SD	AC	FC	Destination Address	Source Address	Information	FCS	ED	FS

Token frame format is of 3 bytes
Starting delimiter
(SD) Ending
delimiter(ED) Access
control (AC)

Data frame format:

Starting delimiter (SD):
Starts with J & K symbols represents 0 & 1 respectively for their no transmission which consumes one byte of memory.

Access Control:
The second byte in the frame format where
T bit is the token bit
T=0 indicates token frame
T=1 indicates data frame
& AC consumes one byte of memory in nature.

FC (Frame control):
This field indicates whether the frame contains data or MAC information in which it is identified by FF=01 or 00 respectively.

Source & destination address:
These are 6 bytes and will have the address of source & destination in which the frames has to be transferred.

Information: It is the actual data to be sent.
FCS (Frame Check Sequence): CCITT-32 CRC checksum.
ED field: signals the end of token / data frame.
FS (Frame Status): tells the sending device whether the destination device is on the ring and, if it is, whether it copied the frame.

IEEE 802.11 Wireless LAN standard

What is IEEE 802.11 standard? Or Write a short note on wireless LAN?

802.11 refer to a family of specifications developed by the IEEE for ***wireless LAN*** (WLAN) technology. Wireless technology eliminates wires in the process, simplifies installation and motion of equipment as well as provides connectivity between computers.

IEEE 802 .11 Terminology

Basic Service Set (BSS): a group of stations that coordinates the access to the medium under a given instance of the MAC.
Basic service area (BSA): The geographical area covered by the BSS.
Adhoc network: a group of stations within range of each other.
Access point (AP): station integrated into the wireless LAN and the distribution system.

ESS (Extended Service Set):
An ESS is a set of two or more wireless APs connected to the same wired network that defines a single logical network segment bounded by a router.

DS (Distribution System):

The APs of multiple BSSs are interconnected by the DS. This allows for mobility, because STAs can move from one BSS to another BSS.

The DS is the logical component used to interconnect BSSs.

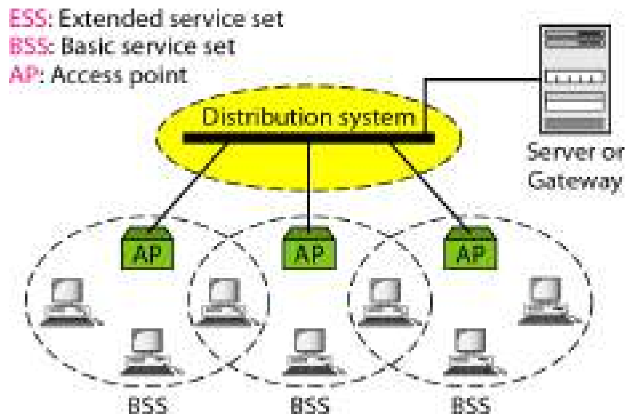
The DS provides distribution services to allow for the roaming of STAs between BSSs.

Portal: bridge to other (wired) networks.

Infrastructure network: the combination of BSSs, a DS and portals.

All STAs in a BSS communicate through the AP.

The AP provides connectivity to the wired LAN.



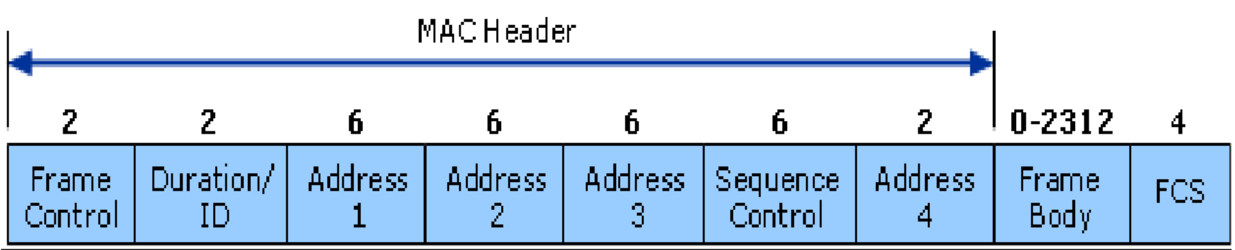
802.11 MAC Frame Format

IEEE 802.11 supports 3 types of frames:

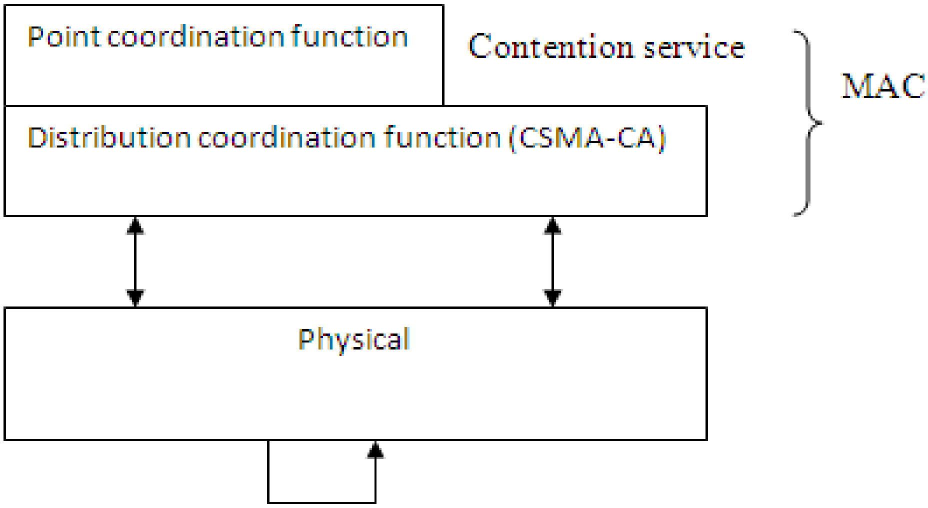
1) **Management frames:** used for station association and disassociation with the AP, timing and synchronization and authentication and de authentication.

2) **Control frames:** Used for handshaking and fro positive acknowledgements during data exchange.

3) **Data frames:** Used for transmission of data.



MAC layers in IEEE 802.11 standard



IEEE 802.11 MAC protocol is specified in terms of
Co-ordination functions that determine when should be station in a BSS is allowed to transmit and when it may be able to receive PDUs over the wireless medium.

Distributed co-ordination function (DCF):

- ✓ It is the basic access method used to support asynchronous data transfer on a best-effort basis.
- ✓ The DCF is based on CSMA/CA protocol.
- ✓ DCF sits on the top of the physical layer and supports contention free services.
- ✓ Carrier sensing involves monitoring the channel to know whether the medium is busy or idle.
- ✓ If the medium is busy it has to wait for some amount of time and start sensing the medium until medium becomes free.
- ✓ To avoid the collision, CSMA/CA uses the method that even when the channel is free it waits for some random amount of time called the inerframe space (IFS) so that it can avoid the collision.

"A handshake procedure was developed to operate with CSMA/CA when there is a hidden –station problem."

IEEE 802.4 Token Bus

IEEE 802.3 has the following Disadvantages

1. **Priority for frames is not present:**
   All traffic (e.g. voice, video, interactive key strokes, background email etc.) is treated the same.

2. **Non-deterministic behavior:**
   No absolute **bound on delay** (waiting time for a station), thus difficult to support real time applications such as voice and video.

Therefore 802.3 was standardized by IEEE in 1985, and is called Token Bus.

- Token Bus is a technique in which the stations on the bus or tree form a logical ring as shown below:

Token Bus: physically a bus, logically a ring.

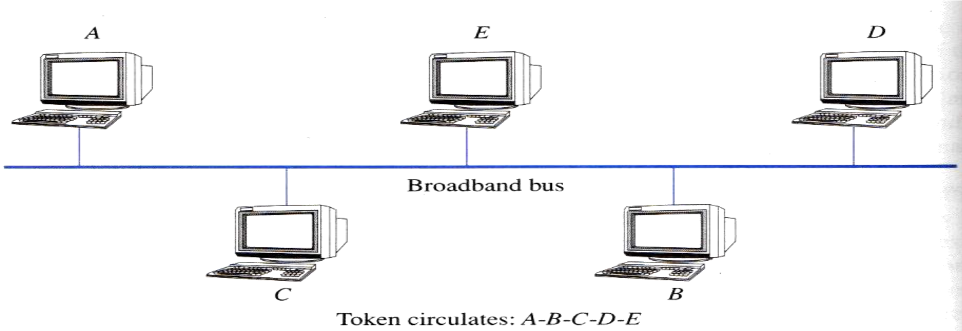


Figure 6.20 Token Bus

The followings are the basic ideas of Token Bus:

1. A special frame called ***token*** is passing around to all the stations.
2. Station that holds the token has the permission to transmit frames.
3. Token holding time limits the number of frames each station can transmit at one time before releasing to next station.
4. It supports 4 classes of priority.

A station transmits its highest priority frames first, and then it‟s next highest, etc. until it has transmitted all of its frames or until its time has expired.

5. The physical location on the bus is not important. The protocol follows a logical ring. Since it is a bus, all stations see all frames.
6. When a new station joins the network, it has to negotiate its order on the passing order.
7. When a station fails, a timing mechanism is used to remove the station from the network.
8. A protocol has to follow to regenerate the token if lost; remove duplicated tokens if generated erroneously.
9.

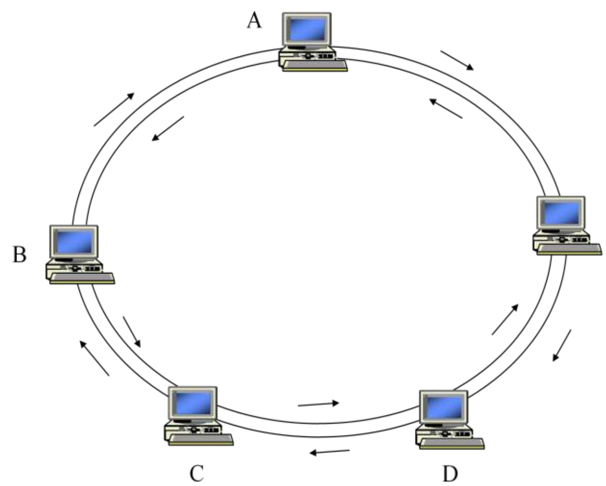
Figure 6.21 Token Bus Frame Format

number of octets

1 or more	1	1	2 or 6	2 or 6	0–8191	4	1
Preamble	SD	FC	DA	SA	....data....	FCS	ED

SD: Start delimiter
FC: Frame control
DA: Destination address
SA: Source address
FCS: Frame check sequence
ED: End delimiter

It has the following components.



- **Preamble:** is used to synchronize the receiver‟s clock.
- **Start delimiter** and **end delimiter:** used to mark the frame boundary.
- The **frame control** field is used to distinguish data frames from control frames.
  - o For data frames, it carries the

a) **frame's priority and**

b) **Indicator requiring the destination to acknowledge the receipt of the frame.**

  - o For control frames, the field is used to specify

The frame type, including token passing frame, **ring maintenance frame,**

- **Destination and source address**:

Either 2-byte address or 6-byte address has to be used.

- The **data field**: can be up to 8182 bytes long when 2-byte address is used; it can be upto 8174 bytes long when 6-byte address is used.
- **Checksum is:** 4 byte long. The same algorithm as that in 802.3 is used to compute and exam the checksum.

<u>FDDI –FIBER DISTRIBUTED DATA INTERFACE</u>

<u>Write a note on FDDI?</u>

- o FDDI is token –ring based LAN standard where it uses ring topology for their connection between the stations of devices.
- o The frames are transmitted through the ring so that , at each and every station the destination address is checked if it matches then stops because it reached the destination else will be transmitted through the ring again until exact address is matched.
- o Here the disadvantage of ring topology is overcome by having dual ring

arrangement as shown in figure.

- If there is any break in the ring then it takes opposite direction of flow from the last station before the break.

FDDI frame structure:

Data Frame Format

8	1	1	2 or 6	2 or 6		4	1	1
PRE	SD	FC	Destination Address	Source Address	Information	FCS	ED	FS

Preamble

Preamble: signals the start of the frame.

SD: start of the frame delimiter signals the start of the frame‟s contents.

FC: signals the type of the frame.

Source & destination address:

These are 6 bytes and will have the address of source & destination in which the frames has to be transferred.

Information: It is the actual data to be sent.

FCS (Frame Check Sequence): CCITT-32 CRC checksum.

ED field: signals the end of token / data frame.

FS (Frame Status): tells the sending device whether the destination device is on the ring and, if it is, whether it copied the frame.

Chapter 5. Network layer-Internetworks

Packet Switching Networks

Packet Switching refers to protocols in which **messages are divided into packets before they are sent.**

Each packet is then transmitted individually and can even follow different routes to its destination.

Once all the packets forming a message arrive at the destination, they are recompiled into the original message.

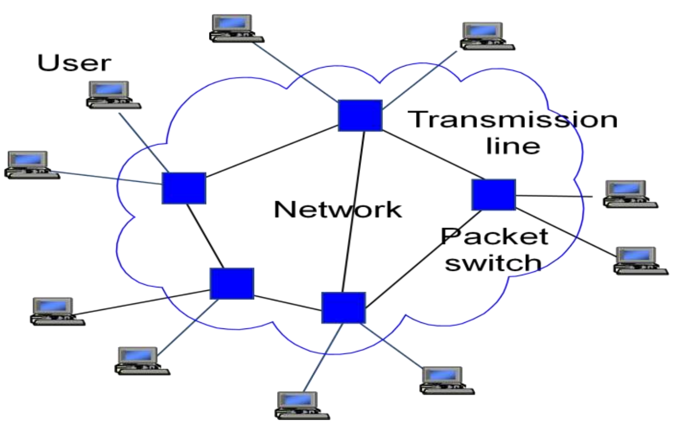
Each packet contains part of the **user's data** and some **control information**.

The control information should at least contain:

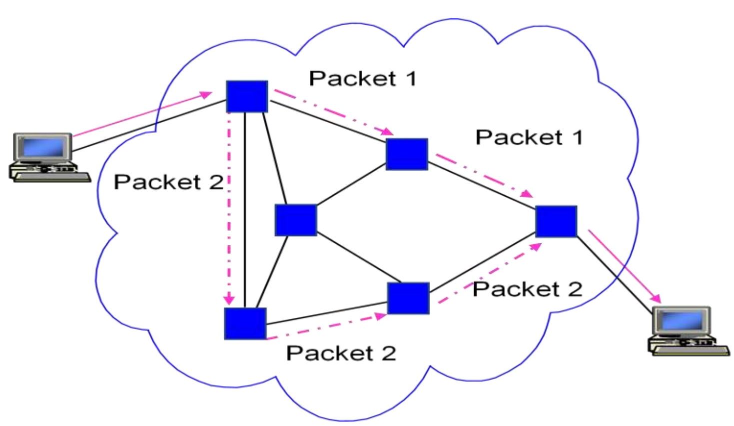
Destination Address

Source Address

Store and forward - Packets are received, stored briefly (buffered) and passed on to the next node.

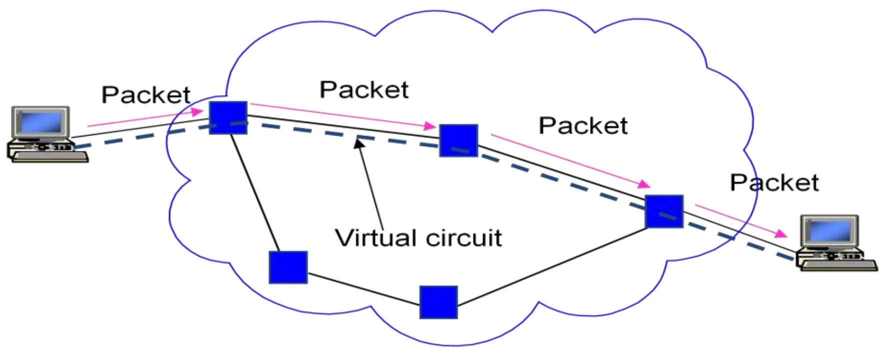


Packet Switching – Datagram



1. This method is also referred as connection less packet switching.
2. Packets in this approach are called datagram.
3. Each packet is treated independently of all others.
4. To do this each packet must be individually addressed to determine where its source and destination are.
5. Each node chooses the path for each packet, taking into account the information from its neighboring node regarding traffic, line failures and so on.
6. The packets with same destination address may or may not follow the same path, and they may arrive out of sequence at the destination.
7. Packets may also lost or dropped because of lack of resources.

Packet Switching – Virtual Circuit



1. This method is also known as connection-oriented packet switching.
2. In the VC approach a pre planned route/logical connection is established before any packets are sent.
3. Once the route is established, all the packets follow the same route and therefore arrive in sequence.
4. This logical communication path between the two devices remains active as long as the two devices are available and can be used to send packets once.
5. After the sending process is completed it is terminated.
6. Each packet contains a virtual circuit identifier instead of destination address along with data.
7. No routing decision is required for each packet (as they follow the same path) and hence this method has less routing (processing) time.

Routing Algorithms

- ✓ The routing algorithm is network layer software which decides routes and the data structures they use for an incoming packet to be transmitted further.
- ✓ The routing may be within a network or between networks depending upon the source and destination.

Routing Algorithm Requirements

Desirable features of routing algorithms

i) Correctness
ii) Responsiveness to changes
iii) Optimality
iv) Robustness
v) Simplicity

Routing algorithms are of two types:

1) Non adaptive
2) Adaptive

Non adaptive or static routing algorithms

When the information about the network topology and traffic is known, non adaptive algorithms are used.

When the topology is static and traffic is not changing, the following algorithms are used.

a) Shortest path routing
b) Flooding
c) Flow based routing

Two commonly implemented shortest path routing algorithms are:

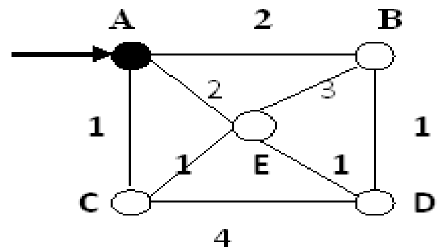
1) Dijikstra‟s algorithm
2) Bellman Ford algorithm

DIJIKSTRA‟S ALGORITHM

The principle of dijikstra‟s algorithm is **"To progressively identify the shortest path from the source node to all other nodes by developing the path in order of increasing path length"**

Working:

Step 1: Each node is labeled with its distance from the source node along the best known path.
Step 2: Initially assume that no paths are found and all nodes can be labeled with infinity. **Step 3:**As the algorithm proceeds and the paths are found , the labels may change, reflecting better paths.
Step 4: The labels are initially tentative. When it is known that label represents the shortest possible path from the source to that node, it is made permanent and no changes are made thereafter.

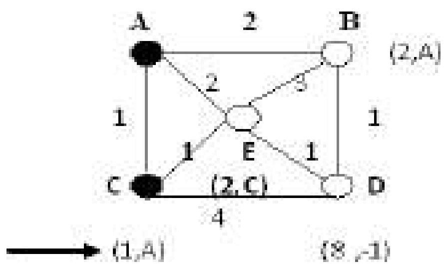


Example:

Consider the following undirected graph

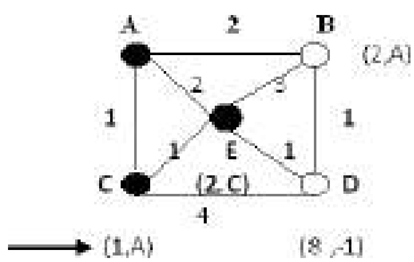
Step 1:

- ✓ Since A is the source, start making node A as permanent, indicated by a filled circle.
- ✓ Then examine the adjacent nodes to A.
- ✓ The adjacent nodes are B, C and E.
- ✓ **Node C** is with the smallest label and makes it permanent as in the figure.



Step 2:

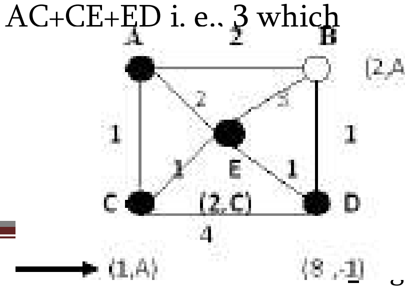
- ✓ Now start at C and examine all the nodes adjacent to it. The adjacent nodes are E and D.
- ✓ We have E(2,A) the total distance from A to E through C is AC+CE i.e., 2
- ✓ For node D, from A through is AC+CD i.e., 5.
- ✓ So the new label on D is D (5, C) as in the figure.



Step 3:

- ✓ Mark E as permanent because E has smallest label as D.
- ✓ Starting from E, the adjacent nodes to E are B and D.
- ✓ The distance from A to D through B and E is AE+EB+BD i.e., 6 which is not the shortest.
- ✓ But the route from A to D through C and E are AC+CE+ED i. e.. 3 which is shorter than other route.

Therefore finally the shortest path from A to D using dijikstra‟s algorithm is



AC ⟶ CE ⟶ ED which is 3.

Hence **ACED** is the shortest path.

BELLMAN FORD ALGORITHM

The principle of d Bellman Ford algorithm is **"Each neighbor of source node knows the shortest path to the destination node.**

The source node can then determine the shortest path to destination node by calculating distance to destination node through each of its neighbors and then picking up the minimum.

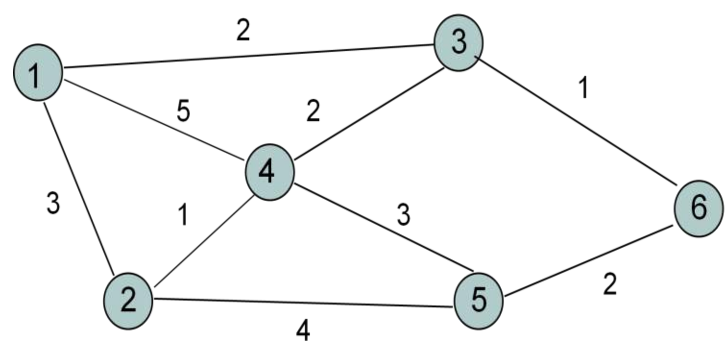
Working:

1) Initially mark all the nodes except source as ∞.
2) And the distance to destination D_d=0.

3) Find the minimum distance to the destination through neighbors: For each i≠d.

D_i-$\min_j(C_{ij}+D_j)$, for all j≠i.

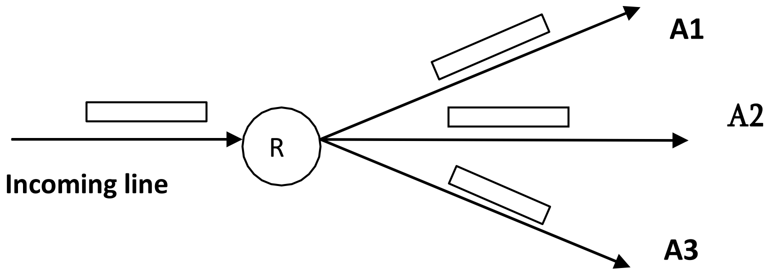
4)Repeat step 2 until destination is reached..



Iteration	Node 1	Node 2	Node 3	Node 4	Node 5
Initial	(-1, □)	(-1, □)	(-1, □)	(-1, □)	(-1, □)
1	(-1, □)	(-1, □)	(6, 1)	(-1, □)	(6,2)
2	(3,3)	(5,6)	(6, 1)	(3,3)	(6,2)
3	(3,3)	(4,4)	(6, 1)	(3,3)	(6,2)

FLOODI NG ALGORITHM

"It is an approach in which, every incoming packet at each node is sent out on every outgoing line except the one which it arrived".



Advantages:

1. Increased reliability, since the message will be sent at least once to every host.
2. Widely used in military applications, wireless networks and distributed database applications.

Dis Advantages:

1. It is very useful in terms of networks total bandwidth.

While a message may Only have one destination it has to be sent to every host.

2. Messages can also become duplicated in the network further increasing the load.
3. Two approaches are implemented to prevent duplication of packets.

FLOW BASED ROUTING:

1. This algorithm considers both topology and load(traffic).
2. This technology requires certain information in advance.

a) Subnet topology

b) Traffic matrix (F_{ij})

c) Capacity matrix (C_{ij}) [capacity of each line in bps] finally a routing algorithm must be chosen.

Working principle:

Step1: The idea behind the analysis is that for a given line, if the capacity and average flow are known, it is possible to compute the mean packet delay on that line from queuing theory.

Step 2: From the mean delays on all the lines, it is straightforward to calculate a flow-weighted average to get the mean packet delay for the whole subnet.

Step 3: The routing problem then reduces to finding the routing algorithm that produces the minimum average delay for the subnet.

Mean delay can be calculated from **$T(sec)=1/\mu\ (C-\lambda)$**

Where C(in bps)=Line capacity, λ=maen flow (in pac/sec),$1/\mu$=mean packet size

ADAPTIVE ALGORITHM

Here the routing strategies react to the changing conditions within the network.

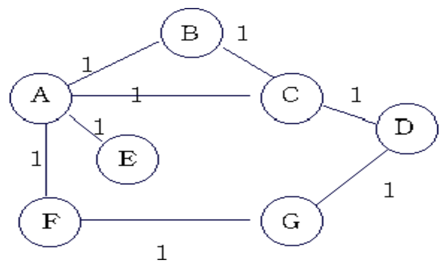
The following algorithms are used:

a) Distance vector routing

b) Link state routing

c) Hierarchical routing

Distance vector routing



Each node constructs a one-dimensional array containing the "distances"(costs) to all other nodes and distributes that vector to its immediate neighbors.

1. The starting assumption for distance-vector routing is that each node knows the cost of the link to each of its directly connected neighbors.
2. A link that is down is assigned an infinite cost. 3.

Information Stored at Node	Distance to Reach Node						
	A	B	C	D	E	F	G
A	0	1	1	?	1	1	?
B	1	0	1	?	?	?	?
C	1	1	0	1	?	?	?
D	?	?	1	0	?	?	1
E	1	?	?	?	0	?	?
F	1	?	?	?	?	0	1
G	?	?	?	1	?	1	0

Initial distances stored at each node (global view).

Final distances stored at each node (global view).

Information Stored at Node	Distance to Reach Node A	B	C	D	E	F	G
A	0	1	1	2	1	1	2
B	1	0	1	2	2	2	3
C	1	1	0	1	2	2	2
D	2	2	1	0	3	2	1
E	1	2	2	3	0	2	3
F	1	2	2	2	2	0	1
G	2	3	2	1	3	1	0

We can represent each node's knowledge about the distances to all other nodes as a table like the one given in Table 1.

Note that each node only knows the information in one row of the table.

1. Every node sends a message to its directly connected neighbors containing its personal list of distance. (For example, **A** sends its information to its neighbors **B, C, E**, and **F**.)
2. If any of the recipients of the information from **A** find that **A** is advertising a path shorter than the one they currently know about, they update their list to give the new path length and note that they should send packets for that destination through **A**. (node **B** learns from **A** that node **E** can be reached at a cost of 1; **B** also knows it can reach **A** at a cost of 1, so it adds these to get the cost of reaching **E** by means of **A**. **B** records that it can reach **E** at a cost of 2 by going through **A**.)
3. After every node has exchanged a few updates with its directly connected neighbors, all nodes will know the least-cost path to all the other nodes.
4. In addition to updating their list of distances when they receive updates, the nodes need to keep track of which node told them about the path that they used to calculate the cost, so that they can create their forwarding table. (for example, **B** knows that it was **A** who said " I can reach **E** in one hop" and so **B** puts an entry in its table that says " To reach **E**, use the link to **A**.)

LINK STATE ROUTING

Link state routing technique consists of 5 steps:

1.) Discover your neighbors and learn their addresses.

Send “Hello” packet on each point-to-point line. Destination node replies with its address.

2.) Measure the cost (delay) to each neighbor.

Send an “ECHO” packet over the line.

Destination is required to respond to “ECHO” packet immediately.

Measure the time required for this operation.

3.) Construct a packet containing all this information

Each router creates a *link state packet* (LSP) which contains names (e.g. network addresses) and cost to each of its neighbours

The LSP is transmitted to *all* other routers, who each update their own records

When a router receives LSPs from all routers, it can use (collectively) that information to make topology-level decisions.

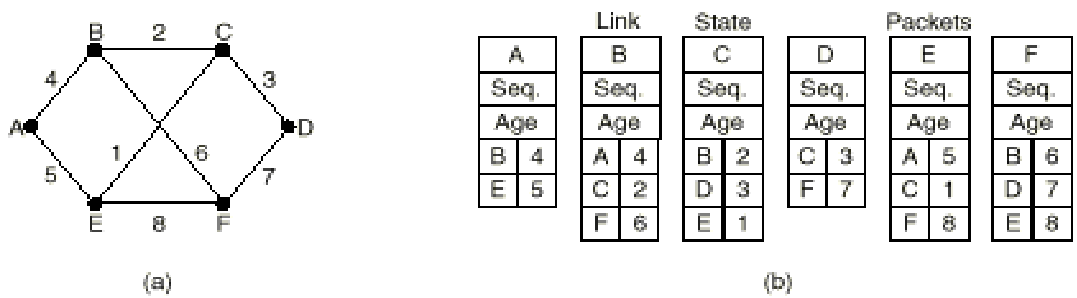


Fig. 5-15. (a) A subnet. (b) The link state packets for this subnet.

4.) Send this packet to all other routers.

Use selective flooding

Sequence numbers prevent duplicate packets from being propagated

Lower sequence numbers are rejected as obsolete

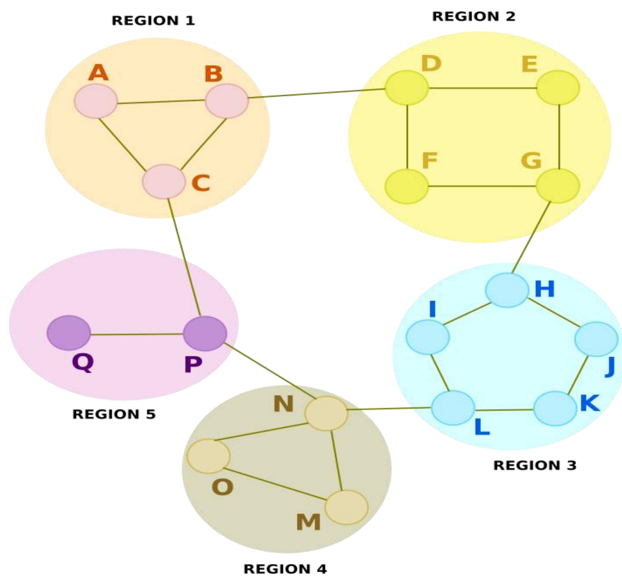
5.) Compute the shortest path to every other router.

Dijkstra‟s Shortest Path algorithm is used to determine the shortest path to each destination.

HIERARCHICAL ROUTING

1. As the network grows the number of routers in the network increases.
2. The size of routing tables becomes enormous and makes routing impossible.
3. We use hierarchical routing to overcome this problem.
4. In the hierarchical routing, routers are classified in groups as regions. Each router has only the information about the routers in its own region and has no information about routers in other regions.
5. If A wants to send packets to any router in region 2 (D, E, F, or G), it sends them to B and so on.
6. As you can see, in this type of routing, the tables can be summarized, so the network efficiency improves.
7. The example shows two-level hierarchical routing.

Destination	Line	Weight
A	N/A	N/A
B	B	1
C	C	1
Region 2	B	2
Region 3	C	4
Region 4	C	3
Region 5	C	2



IPv4 (Internet Protocol Version 4)

IPv4 is a connectionless protocol for a packet switching network that uses the Datagram approach. IPv4 is defined and specified in IETF publication RFC 791.

Datagram is handled independently and each datagram can follow a different route to the destination.

Datagram Frame format

Packets in the IPv4 layer called **Datagram.** A Datagram is a variable length packet consisting of two parts header and Data; header is 20 to 60 bytes in length and contains information for routing.

VERSION: Version of the IP protocol (4 bits), which is 4 for IPv4

HLEN: IP header length (4 bits), which is the number of 32 bit words in the header. The minimum value for this field is 5 and the maximum is 15.

Type of service: Low Delay, High Throughput, Reliability (8 bits)

Total Length: Length of header + Data (16 bits), which has a minimum value 20 bytes and the maximum is 65,535 bytes.

Identification: Unique Packet Id for identifying the group of fragments of a single IP datagram (16 bits)

Flags: 3 flags of 1 bit each: reserved bit (must be zero), do not fragment flag, more fragments flag (same order)

Fragment Offset: Represents the number of Data Bytes ahead of the particular fragment in the particular Datagram. Specified in terms of number of 8 bytes, which has the maximum value of 65,528 bytes.

Time to live: Datagram's lifetime (8 bits), It prevents the datagram to loop through the network by restricting the number of Hops taken by a Packet before delivering to the Destination.

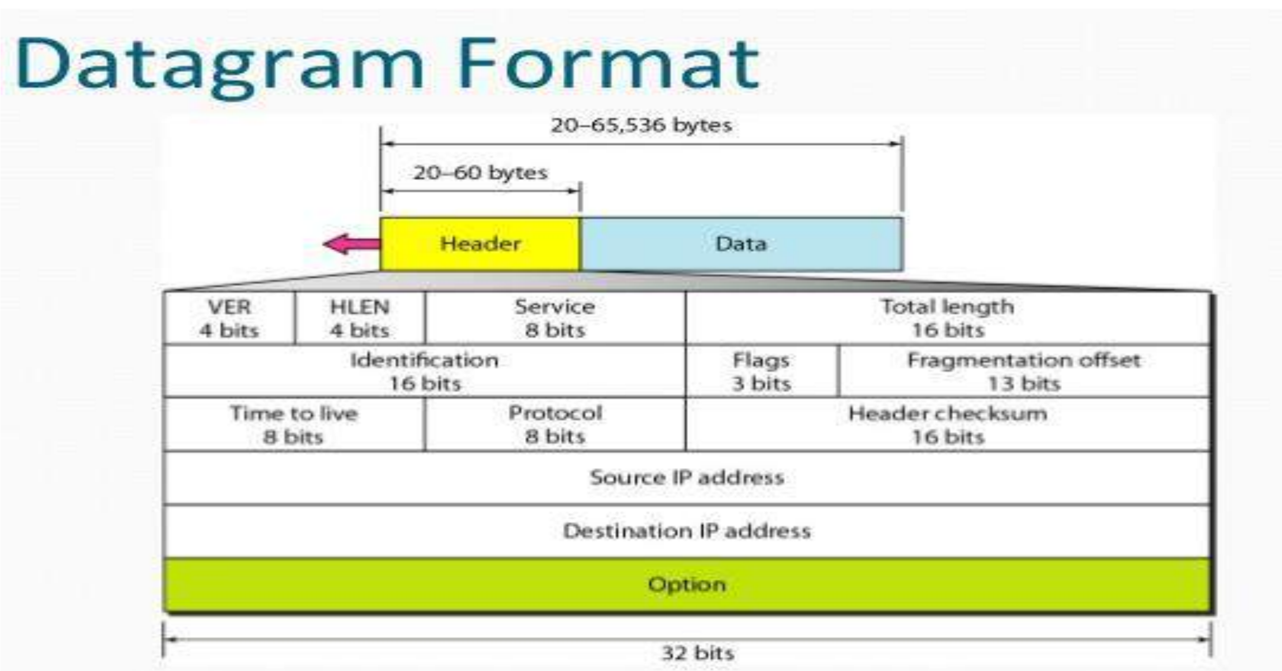
Protocol: Name of the protocol to which the data is to be passed (8 bits)

Header Checksum: 16 bits header checksum for checking errors in the datagram header

Source IP address: 32 bits IP address of the sender

Destination IP address: 32 bits IP address of the receiver

Option: Optional information such as source route, record route. Used by the Network administrator to check whether a path is working or not.



Internet Protocol version 6 (IPv6) Header

Prerequisite: Introduction to Internet Protocol version 6
IP version 6 is the new version of Internet Protocol, which is way better than IP version 4 in terms of complexity and efficiency. Let's look at the header of IP version 6 and understand how it is different from IPv4 header.

IP version 6 Header Format:
Version (4-bits) : Indicates version of Internet Protocol which contains bit sequence 0110.
Traffic Class (8-bits) : The Traffic Class field indicates class or priority of IPv6 packet which is similar to *Service Field* in IPv4 packet. It helps routers to handle the traffic based on priority of the packet. If congestion occurs on router then packets with least priority will be discarded.

As of now only 4-bits are being used (and remaining bits are under research), in which 0 to 7 are assigned to Congestion controlled traffic and 8 to 15 are assigned to Uncontrolled traffic.
Priority assignment of Congestion controlled traffic:

Priority	Meaning
0	No Specific traffic
1	Background data
2	Unattended data traffic
3	Reserved
4	Attended bulk data traffic
5	Reserved
6	Interactive traffic
7	Control traffic

Uncontrolled data traffic is mainly used for Audio/Video data. So we give higher priority to Uncontrolled data traffic.
Source node is allowed to set the priorities but on the way routers can change it. Therefore, destination should not expect same priority which was set by source node.

Flow Label (20-bits) : Flow Label field is used by source to label the packets belonging to the same flow in order to request special handling by intermediate IPv6 routers, such as non-default quality of service or real time service. In order to distinguish the flow, intermediate router can use source address, destination address and flow label of the packets..

Payload Length (16-bits) : It is a 16-bit (unsigned integer) field, indicates total size of the payload which tells routers about amount of information a particular packet contains in its payload. In case length of payload is greater than 65,535 bytes (payload up to 65,535 bytes can be indicated with 16-bits),

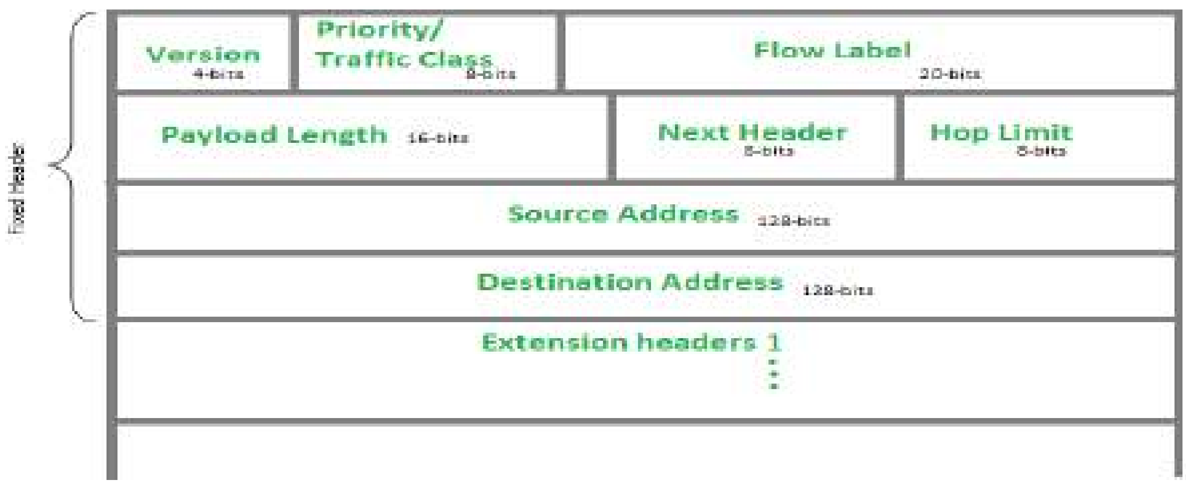
Next Header (8-bits) : Next Header indicates type of extension header(if present) immediately following the IPv6 header. Whereas In some cases it indicates the protocols contained within upper-layer packet, such as TCP, UDP.

Hop Limit (8-bits) : Hop Limit field is same as TTL in IPv4 packets. It indicates the maximum number of intermediate nodes IPv6 packet is allowed to travel. Its value gets decremented by one, by each node that forwards the packet and packet is discarded if value decrements to 0.

Source Address (128-bits) : Source Address is 128-bit IPv6 address of the original source of the packet.

Destination Address (128-bits) : Destination Address field indicates the IPv6 address of the final destination(in most cases). All the intermediate nodes can use this information in order to correctly route the packet.

Extension Headers: In order to rectify the limitations of *IPv4 Option Field*, Extension Headers are introduced in IPversion 6. The extension header mechanism is very important part of the IPv6 architecture. Next Header field of IPv6 fixed header points to the first Extension Header and this first extension header points to the second extension header and so on.



Quality of service

Quality of service (**Q o S**) is the description or measurement of the overall performance of a service, such as a telephony or computer network or a cloud computing service, particularly the performance seen by the users of the network. To quantitatively measure quality of service, several related aspects of the network service are often considered, such as packet loss, bit rate, throughput, transmission delay, availability, jitter, etc.

In the field of computer networking and other packet-switched telecommunication networks, quality of service refers to traffic prioritization and resource reservation control mechanisms rather than the achieved service quality. Quality of service is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

Quality of service is particularly important for the transport of traffic with special requirements. In particular, developers have introduced Voice over IP technology to allow computer networks to become as useful as telephone networks for audio conversations, as well as supporting new applications with even stricter network performance requirements.

Chapter 6.Transport Layer

MULTIPLEXING

Multiplexing is the set of techniques that allows simultaneous transmission of multiple signals across a single data link

Need for multiplexing

- As data and telecommunication usage increases, the traffic increases.
- To accommodate this increase, individual lines are added each time a new channel is needed or higher capacity links are installed and are used to carry multiple signals.
- If the transmission capacity of a link is greater than the transmission needs of the devices connected to it, the excess capacity is wasted. Hence, the transmission capacity of a link can be shared similar to a large water pipe carrying water to several separate houses at a time.
- A technique called multiplexing is designed to reduce the number of links by combing multiple communication paths over a single carrier.
- The aim is to share an expensive resource. For example several phone calls may be transferred using one wire.
- A device that perform the multiplexing is called a multiplexing (MUX), and a device that perform the reverse process is called a demultiplex (DEMUX).

The four device on the left direct their transmission stream to a multiplexer (mux) which combines them into a single stream (many to one) thus sharing the capacity of link

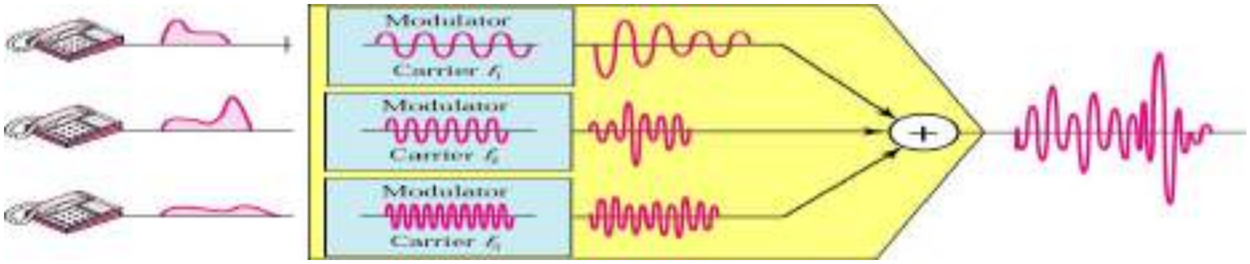
At the receiving end, demultiplexer (DEMUX) separates the signals stream back into its component transmission (one to many) and directs them to their intended device

Frequency Division Multiplexing

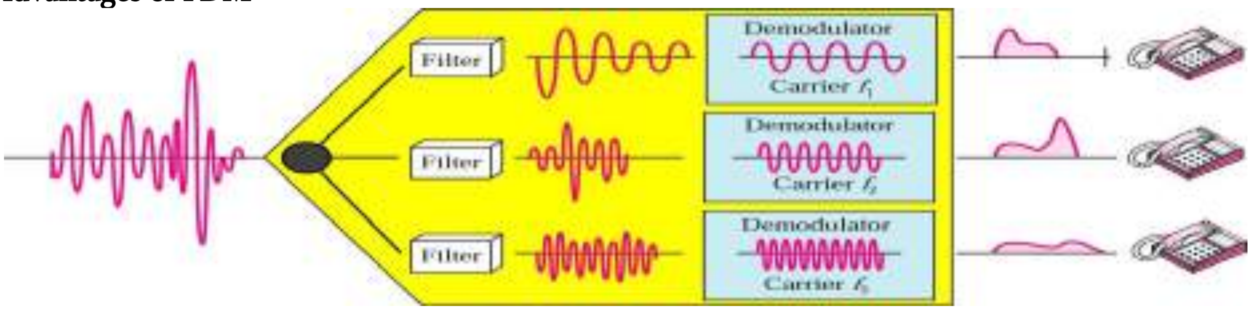
- In analog transmission, signals are commonly multiplexed using frequency division multiplexing
- In FDM the available bandwidth of a communication channel is shared among multiple user
- The carrier bandwidth is divided into sub channels of different frequency width and each sub channel is allocated to the user
- Signals generated by each user are modulated into different carrier frequencies and are then combined into a single composite signals can be transported by the link
- Channel are separated strips of unused bandwidth called guard bands to

prevent signals from overlapping

- FDM is extensively used in broadcast radio and cable television.



Advantages of FDM



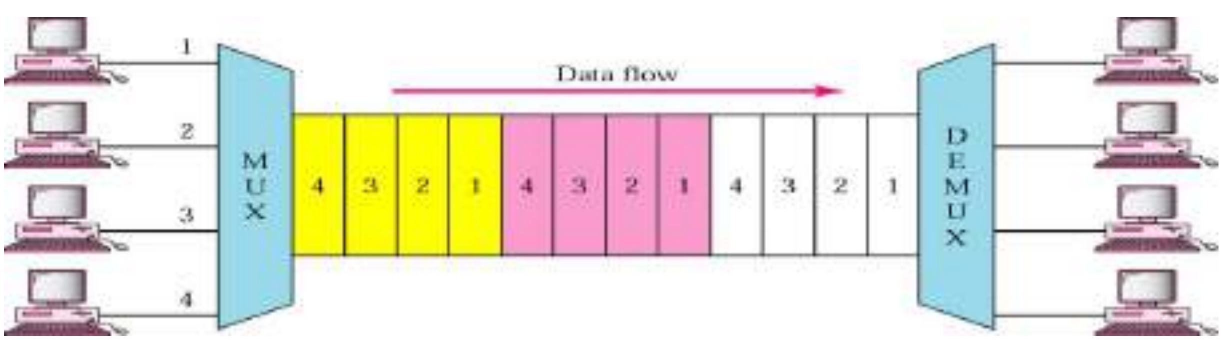
- ➢ Here user can be added to the system by simply adding another pair of transmitter modulator and receiver demodulators.
- ➢ FDM system support full duplex information flow.
- ➢ Noise problem for analog communication has lesser effect.

Disadvantages of FDM

- ➢ In FDM system the initial cost is high. This may include the cable between the two ends and the associated connectors for the cable.
- ➢ In FDM system, a problem from one user can sometime affect others.
- ➢ In FDM system, each user requires a precise carrier frequency

Time Division Multiplexing

- In digital transmission, signals are commonly multiplexed using time-division multiplexing (TDM).
- In TDM the multiple signals are carried over the same channel in alternating time slots.
- Each user of channel is allotted a small time interval by dividing the total time available in the channel. During this time interval the user can transmit the message.
- In TDM, user send message one after the and use full channel bandwidth of channel during his time slice.
- The message sent through the physical channel must be separated at the receiving end. Individual chunks of message sent by each user should be reassembled into a full message.



Advantages of TDM

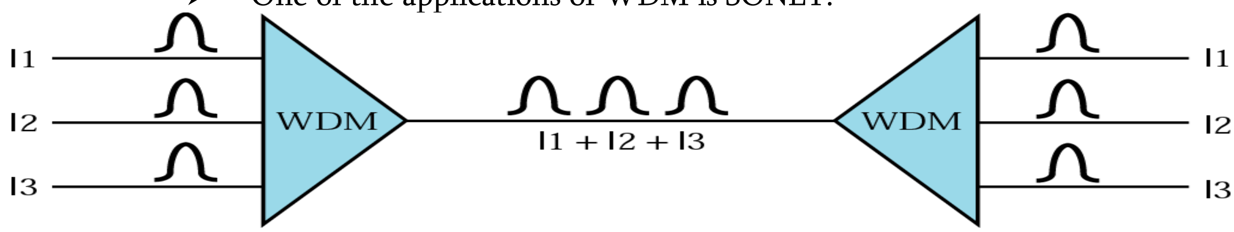
- It uses a single link.
- It does not require carrier matching at both end of the links.
- Use of capacity is high.
- There is no need to include identification of the traffic stream on each packet.

Disadvantages of TDM

- Initial cost is high.
- Technical complexity is more
- TDM can only be used for digital data multiplexing.

Wavelength Division Multiplexing (WDM)

- WDM is a technology that closely resembles FDM, but here several light signals of different frequencies are combined into one single light signal at the multiplexer and sent at the same time through the optical fiber.
- One of the applications of WDM is SONET.



Multiplexing and Demultiplexing in Transport Layer

Multiplexing and Demultiplexing services are provided in almost every protocol architecture ever designed. UDP and TCP perform the demultiplexing and multiplexing jobs by including two special fields in the segment headers: the source port number field and the destination port number field.

Multiplexing –

Gathering data from multiple application processes of sender, enveloping that data with header and sending them as a whole to the intended receiver is called as multiplexing.

Demultiplexing –

Delivering received segments at receiver side to the correct app layer processes is called as demultiplexing.

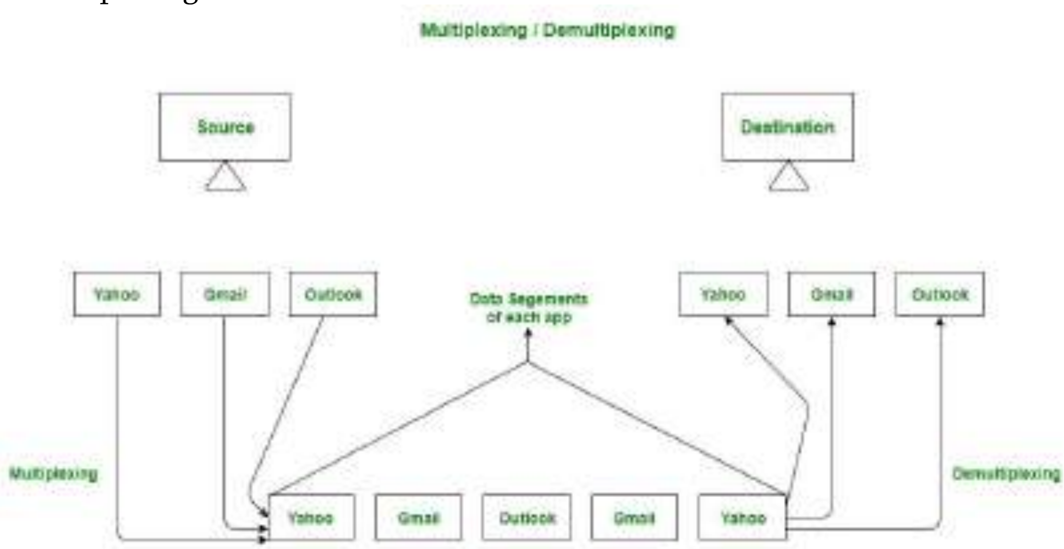


Figure – Abstract view of multiplexing and demultiplexing

Multiplexing and demultiplexing are the services facilitated by the transport layer of OSI model.

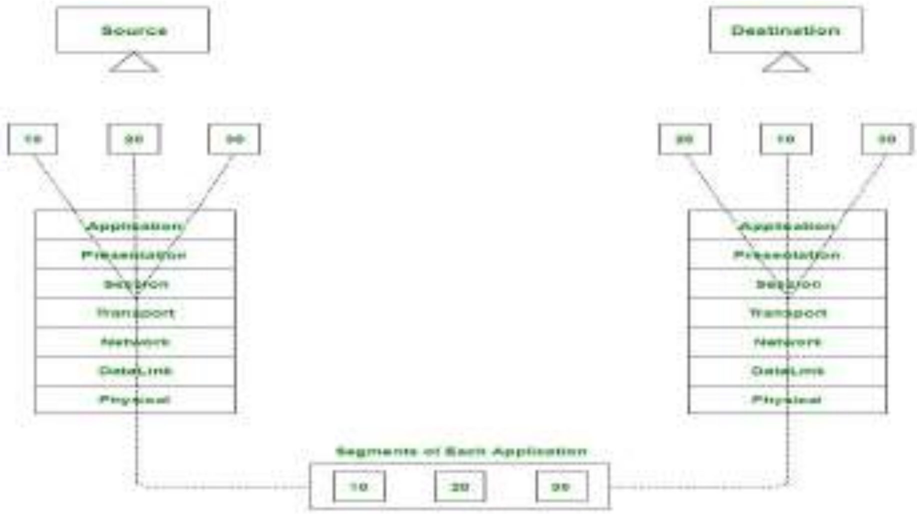


Figure – Transport layer- junction for multiplexing and demultiplexing

There are two types of multiplexing and Demultiplexing :

1. Connectionless Multiplexing and Demultiplexing
2. Connection-Oriented Multiplexing and Demultiplexing

How Multiplexing and Demultiplexing is done –

For sending data from an application at sender side to an application at the destination side, sender must know the IP address of destination and port number of the application (at the destination side) to which he want to transfer the data. Block diagram is shown below :

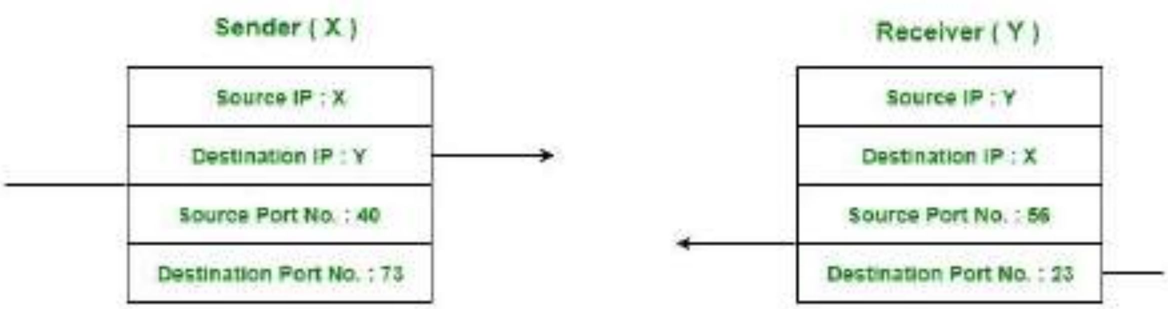


Figure – Transfer of packet between applications of sender and receiver

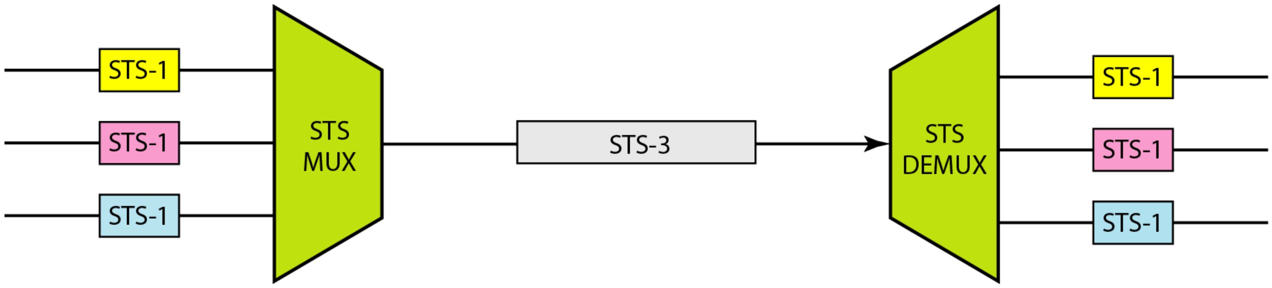
Let us consider two messaging apps that are widely used now a days viz. Hike and whatsapp. Suppose A is the sender and B is the receiver. Both sender and receiver have these applications installed in their system (say smartphone). Suppose A want to send messages to B in whatsapp and hike both. In order to do so, A must mention the IP address of B and destination port number of the whatsapp while sending the message through whatsapp application. Similarly, for the later case, A must mention the IP address of B and destination port number of the hike while sending the message

SONET/SDH (Synchronous Optical Networking / Synchronous Digital Hierarchy)

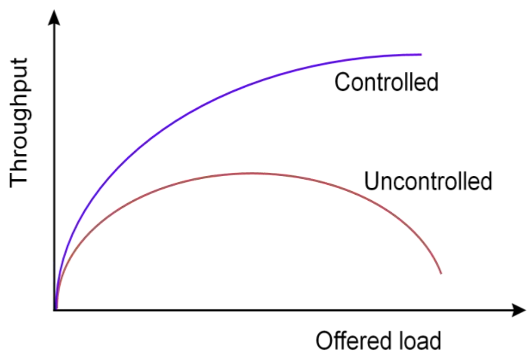
- SONET is a synchronous TDM system controlled by a master clock.

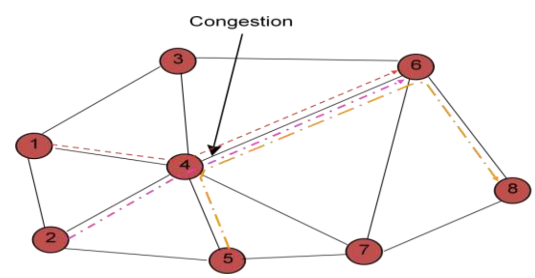
Synchronous: A single clock is used to handle the timing of transmission and equipment across the entire network.

- It was intended to provide a specification for taking advantage of high speed digital transmission capability of optical fiber.



- o SONET uses 51.84 Mbps electrical signal known as the synchronous transport signal level- 1(STS-1).
- o Each STS level (STS-1 to STS-192) supports a certain data rate, specified in megabits per second.
- o The physical links defined to carry each level of STS are called optical carriers (OCs).
- o Lower rate STS s can be multiplexed to make them compatible with higher rate systems.
  - o Ex: Three STS-1 s can be combined into one STS-3 as shown in fig.





CONGESTION

"Congestion is a situation in a network that may occur if load on the network (the no. of packets sent to the network) is greater than the capacity of the network (no. of packets a network can handle).

Factors causing congestion:

1. The input traffic rate exceeds the capacity of the output lines.
2. The routers are too slow to perform book keeping tasks(queuing buffers , updating tables
3. The router‟s buffer is too limited.

CONGESTION CONTROL

In general congestion control mechanisms are divided into 2 broad categories:

1. Open-loop congestion control(prevention)
2. Closed-loop congestion control(removal)

Open loop policies

Open loop policies are applied to prevent congestion before it happens.

Open loop control relies on 3 mechanisms: Admission control, policing, and traffic shaping.

1) Admission control:

Admission control is a network function that computes the resources(bandwidth and buffers)

requirements of a new flow and determines whether the resources are available for the flow.
If the QoS(Quality of Service) of the new flow can be satisfied without violating Qos of existing flows, the flow is accepted; otherwise the flow is rejected.

2)Policing

Once the traffic flow is accepted by an admission control policy, the QoS will be satisfied as long as the source obeys its negotiated traffic.
The process of monitoring and enforcing the traffic flow is called policing.

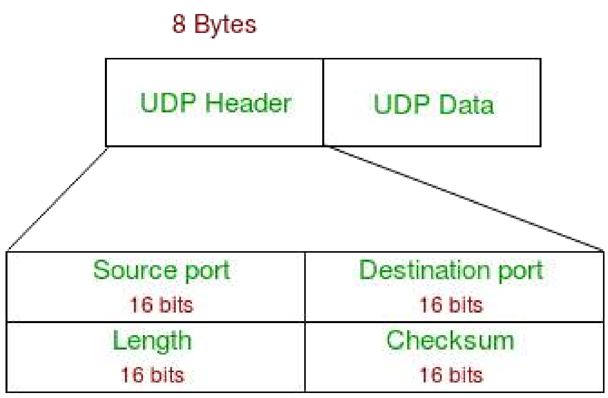
3)Traffic shaping

One of the main causes of congestion is that traffic is often burst.
The method of forcing the packets to be transmitted at a more predictable rate is called traffic shaping.

User Datagram Protocol (UDP)

User Datagram Protocol (UDP) is a Transport Layer protocol. UDP is a part of Internet Protocol suite, referred as UDP/IP suite. Unlike TCP, it is **unreliable and connectionless protocol.** So, there is no need to establish connection prior to data transfer.
Though Transmission Control Protocol (TCP) is the dominant transport layer protocol used with most of Internet services; provides assured delivery, reliability and much more but all these services cost us with additional overhead and latency. Here, UDP comes into picture. For the realtime services like computer gaming, voice or video communication, live conferences; we need UDP. Since high performance is needed, UDP permits packets to be dropped instead of processing delayed packets. There is no error checking in UDP, so it also save bandwidth.
User Datagram Protocol (UDP) is more efficient in terms of both latency and bandwidth.

UDP Header –
UDP header is **8-bytes** fixed and simple header, while for TCP it may vary from 20 bytes to 60 bytes. First 8 Bytes contains all necessary header information and remaining part consist of data. UDP port number fields are each 16 bits long, therefore range for port numbers defined from 0 to 65535; port number 0 is reserved. Port numbers help to distinguish different user requests or process.



1. **Source Port :** Source Port is 2 Byte long field used to identify port number of source.
2. **Destination Port :** It is 2 Byte long field, used to identify the port of destined packet.
3. **Length :** Length is the length of UDP including header and the data. It is 16-bits field.

4. **Checksum :** Checksum is 2 Bytes long field. It is the 16-bit one's complement of the one's complement sum of the UDP header, pseudo header of information from the IP header and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.

Notes – Unlike TCP, Checksum calculation is not mandatory in UDP. No Error control or flow control is provided by UDP. Hence UDP depends on IP and ICMP for error reporting.

Applications of UDP:

- Used for simple request response communication when size of data is less and hence there is lesser concern about flow and error control.
- It is suitable protocol for multicasting as UDP supports packet switching.
- UDP is used for some routing update protocols like RIP(Routing Information Protocol).
- Normally used for real time applications which can not tolerate uneven delays between sections of a received message.
- Following implementations uses UDP as a transport layer protocol:
  - NTP (Network Time Protocol)
  - DNS (Domain Name Service)
  - BOOTP, DHCP.
  - NNP (Network News Protocol)
  - Quote of the day protocol
  - TFTP, RTSP, RIP, OSPF.
- Application layer can do some of the tasks through UDP-
  - Trace Route
  - Record Route
  - Time stamp
- UDP takes datagram from Network Layer, attach its header and send it to the user. So, it works fast.
- Actually UDP is null protocol if you remove checksum field.

Chapter 7.Modern Wireless Communications Systems

What is wireless generation?

The mobile **wireless Generation** (G) generally refers to a change in the nature of the system, speed, technology, frequency, data capacity, latency etc. Each **generation** have some standards, different capacities, new techniques and new features

Generations of Wireless Network: 1G, 2G, 3G & 4G -

1G, which stands for **"first generation,"** refers to the first generation of wireless telecommunication Technology, more popularly known as cellphones. A set of wireless standards developed in the 1980's, 1G technology replaced 0G technology .1G wireless networks used analog radio signals. Through 1G, a voice call gets modulated to a higher frequency of about 150MHz and up as it is Transmitted between radio towers. This is done using a technique called Frequency-Division Multiple Access (FDMA).

Different 1G standard were used in various countries. Advanced Mobile Phone System (AMPS) was a 1G standard used in the United States. Nordic Mobile Telephone (NMT) was a 1G standard used in Nordic countries (Denmark, Finland, Iceland, Norway and Sweden), as well as in its neighboring countries Switzerland and Netherlands, Eastern Europe, and Russia. Italy used a telecommunications system called RTMI.In the United Kingdom, Total Access Communication System (TACS) was used. France used Radiocom 2000.

Second Generation (2G) 2g Architecture-Second generation (2g) telephone technology is based on GSM or in other words global system for mobile communication. Second generation was launched in Finland in the year 1991.How 2G works, Uses of 2G technology/2G network allows for much greater penetration intensity. 2G technologies enabled the various mobile phone networks to provide the services such as text messages, picture messages and MMS (multi media messages)

GPRS is important because it helps operators, vendors, content providers, and users prepare for 3G, as many concepts of GPRS live on in 3G, and we will need these enhancements to 2G networks for ten years or more. At the moment, wireless network technologies are somewhere between 2G and 2.5G. The second generation of mobile communications technology was all about digital PCS.

What is 3G Technology/3G Technology If you want augmented bandwidth, multiple mobile applications and clarity of digital signals, then3G (Thrid Generation Technology) is your gateway. GSM technology was able to transfer circuit switched data over the network. The use of 3G technology is also able to transmit packet switch data efficiently at better and increased bandwidth. 3G mobile technologies proffers more advanced services to mobile users. It can help many multimedia services to function.

The basic feature of 3G Technology (Thrid Generation Technology) is fast data transfer rates. However this feature is not currently working properly because, ITU 200 is still making decision to fix the data rates.

3G technology is much flexible, because it is able to support the 5 major radio technologies.
These radio technologies operate under CDMA, TDMA and FDMA.CDMA holds for IMT-DS
The aim of the 3G is to allow for more coverage and growth with minimum investment.

4G Technology/What is 4G technology

4G Technology is basically the extension in the 3G technology with more bandwidth and services offers in the 3G. But at this time nobody exactly knows the true 4G definition.
Some people say that 4G technology is the future technologies
that are mostly in their maturity period. The expectation for the 4G technology is basically the high quality audio/video streaming over end to end Internet Protocol. If the Internet Protocol (IP) multimedia sub-system movement achieves what it going to do, nothing of this possibly will matter.

4G technology is likely to enable ubiquitous computing, that will simultaneously connects to numerous high date speed networks offers faultless handoffs all over the geographical regions. Many network operators possibly utilize technologies for example; wireless mesh networks and cognitive radio network to guarantee secure connection & competently allocates equally network traffic and bandwidth. Some of the companies trying 4G mobile communication at 100 Mbps for mobile users and up to 1 Gbps over fixed stations.
They planned on publicly launching their first commercial wireless **network around 2010.**

Wireless Communication Protocol

Wireless communication protocols to connect computers, laptops and smartphones. The more widespread and standardized **protocols** are **wireless** LAN (IEEE 802.11) or Bluetooth (IEEE 802.15. 1). The implementation of these **protocols** in a device requires high processing capacity and big energy consumption.

Parameters	Wired	Wireless
Communication Medium	Copper, Fiber etc.	Air
Standard	IEEE 802.3	802.11 family
Mobility and Roaming	Limited	Higher
Security	High	Lower than Wired. Also easy to hack
Speed / Bandwidth	High Speed upto 1 Gbps	Lower speed than Wired Network.
Access to Network	Physical Access Required	Proximity Required
Delay	Low	High
Reliability	High	Lower than Wired
Flexibility to change	Less flexible to changes	More flexible configuration
Working principle	CSMA/CD, operates by detecting the occurrence of a collision.	CSMA/CA , hence reduces possibility of collision be avoiding collision from happening
Interference and Fluctuations vulnerability	Very Less	High
Installation activity	Cumbersome and manpower intensive	Less labor intensive and easy
Installation Time	Takes longer time to perform installation	Very less deployment time
Dedicated / Shared Connection	Dedicated	Shared
Installation Cost	High	Low
Maintenance (Upgrade) cost	High	Low
Related equipment	Router, Switch , Hub	Wireless Router, Access Point
Benefits	• Greater Speed • Higher noise immunity • Highly reliable • Greater Security	• No Hassles of Cable • Best for mobile devices • Greater mobility • Easy installation and management

Mobile technology is a form of technology that is mostly used in cellular communication and other related aspects. It uses a form of platform where by many transmitters have the ability to send data at the same time on a single channel. This platform is called Code-division multiple access (CDMA). This platform allows many users to make use of single frequencies because it restricts the likelihood of interference of frequencies from two or more sources. The mobile technology has improved from a simple device used for phone call and messaging into a multi-tasking device used for GPS navigation, internet browsing, gaming, instant messaging tool etc

Mobile technology was a mystery two decades ago but now, it has become something of necessity to both the rural and the urban areas. The mobile technology started as a remarkable achievement in the world of technology but now, it is transforming into user comfort technology due to its present diverse functionality. The mobile has made it possible for users to transfer files and other files through Bluetooth and wifi.

The mobile is also equipped with internet connectivity, making it easy for the user to gain information and also to download files from the internet. Video call conferencing is another achievement that has come to reality through mobile technology. Business men and clients now have the channel to communicate even without seeing in person. With the use of mobile technology, it is now easy to catch up with every form of entertainment from the comfort of your home.

The diversity of mobile technology is due to the fact that that many mobile operating systems are available for smart phones all of which have their own unique characteristics. Some of these operating systems are: the Android, blackberry, webos, ios, symbian windows and bada mobile some of which will be briefly discussed: The android is an operating system that is developed by google.

Since 2008 the operating system customized their platforms, allowing the user to download any app he wishes to download like the gaming app, the utility apps, the GPS and other tools. Any user that have the knowledge to create an app and also wishes to create one can legally do so. The WebOS basically has the ability to support internet programming languages some of which are HTML, Javascript&CSS.

The internet is now equipped with a 4G network. This network enables a high speed data transmission in the channels, making surfing the internet to be easier. In the future, smart phones will be aware of their environment,

Since the arrival of the mobile, it has helped humans in many ways; some of which are: mobile phones are very important in case of an emergency they safe lives in cases of accidents and other related issues. One of the most important use of smart phones is that they ensure safety. Families can easily communicate with each other while away. To cap it up, mobile technology is here to stay and holds a lot more features in the future to meet even the most of our basic needs and to make life a lot easier.

Transmitter and Receiver

A radio receiver is the opposite of a **radio transmitter**. It uses an **antenna** to capture radio waves, processes those waves to extract only those waves that are vibrating at the desired frequency, extracts the audio signals that were added to those waves, amplifies the audio signals, and finally plays them on a speaker.

In radio communications, a radio **receiver**, also known as a **receiver**, wireless or simply radio is ... Besides broadcast **receivers**, described above, radio **receivers** are **used** in a huge variety of electronic systems in modern technology device with a local computer **network** (WLAN) to exchange data with other **devices**