COMPUTER NETWORK AND SECURITY

(According to IOE syllabus)

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SYLLABUS

1. Introduction to Computer Network (5 hours)
1.1 Uses of Computer Network
1.2 Networking model client/server, p2p, active network
1.3 Protocols and Standards
1.4 OSI model and TCP/IP model
1.5 Comparison of OSI and TCP/IP model
1.6 Example network: The Internet, X.25, Frame Relay, Ethernet, VoIP, NGN and MPLS, xDSL.

2. Physical Layer (5 hours)
2.1 Network monitoring: delay, latency, throughput
2.2 Transmission media: Twisted pair, Coaxial, Fiber optic, Line-of-site, Satellite
2.3 Multiplexing, Circuit switching, Packet switching, VC Switching, Telecommunication switching system (Networking of Telephone exchanges)
2.4 ISDN: Architecture, Interface, and Signaling

3. Data Link Layer (5 hours)
3.1 Functions of Data link layer
3.2 Framing
3.3 Error Detection and Corrections,
3.4 Flow Control
3.5 Examples of Data Link Protocol, HDLC, PPP
3.6 The Medium Access Sub-layer
3.7 The channel allocation problem
3.8 Multiple Access Protocols
3.9 Ethernet,
3.10 Networks: FDDI, ALOHA, VLAN, CSMA/CD, IEEE 802.3(Ethernet), 802.4(Token Bus), 802.5(Token Ring), and 802.1(Wireless LAN).

4. Network Layer (9 hours)
4.1 Internetworking &devices: Repeaters, Hubs, Bridges, Switches, Router, Gateway
4.2 Addressing: Internet address, classful address
4.3 Subnetting
4.4 Routing: techniques, static vs. dynamic routing , routing table for classful address
4.5 Routing Protocols: RIP, OSPF, BGP, Unicast and multicast routing protocols
4.6 Routing algorithms: shortest path algorithm, flooding, distance vector routing, link state routing; Protocols: ARP, RARP, IP, ICMP

5. Transport Layer (5 hours)
5.1 The transport service: Services provided to the upper layers
5.2 Transport protocols: UDP, TCP
5.3 Port and Socket
5.4 Connection establishment, Connection release
5.5 Flow control & buffering
5.6 Multiplexing & de-multiplexing
5.7 Congestion control algorithm: Token Bucket and Leaky Bucket Transport Layer

6. Application Layer (5 hours)
6.1 Web: HTTP & HTTPS

6.2 File Transfer: FTP, PuTTY, WinSCP
6.3 Electronic Mail: SMTP, POP3, IMAP
6.4 DNS
6.5 P2PApplications
6.6 Socket Programming
6.7 Application server concept: proxy caching, Web/Mail/DNS server optimization
6.8 Concept of traffic analyzer: MRTG, PRTG, SNMP, Packet tracer, Wireshark.

7. Introduction to IPV6 (4 hours)
7.1 IPv6- Advantages
7.2 Packet formats
7.3 Extension headers
7.4 Transition from IPv4 to IPv6: Dual stack, Tunneling, Header Translation
7.5 Multicasting

8. Network Security (7 hours)
8.1 Properties of secure communication
8.2 Principles of cryptography: Symmetric Key and Public Key
8.3 RSA Algorithm,
8.4 Digital Signatures
8.5 Securing e-mail (PGP)
8.6 Securing TCP connections (SSL)
8.7 Network layer security (IPsec, VPN)
8.8 Securing wireless LANs (WEP)
8.9 Firewalls: Application Gateway and Packet Filtering, and IDS

Practical:
1. Network wiring and LAN setup
2. Router Basic Configuration
3. Static and Dynamic Routing
4. Creating VLAN
5. Router access-list configuration
6. Basic Network setup on Linux
7. Setup of Web Server, DNS Server, DHCP Server
8. Virtualizations

CHAPTER 1 - INTRODUCTION TO COMPUTER NETWORK

1. INTRODUCTION

A computer network is a group of computer systems and other computing hardware devices that are linked together through communication channels to facilitate communication and resource-sharing among a wide range of users. Networks are commonly categorized based on their characteristics.

Application of Networks

- Facilitate communication via email, video conferencing, instant messaging, etc.
- Enable multiple users to share a single hardware device like a printer or scanner
- Enable file sharing across the network
- Allow for the sharing of software or operating programs on remote systems
- Make information easier to access and maintain among network users

There are many types of networks, including:

- Local Area Networks (LAN)
- Personal Area Networks (PAN)
- Home Area Networks (HAN)
- Wide Area Networks (WAN)
- Campus Networks
- Metropolitan Area Networks (MAN)
- Enterprise Private Networks
- Internetworks
- Backbone Networks (BBN)
- Global Area Networks (GAN)
- The Internet

LAN

This is the abbreviation for Local Area Network which is when there are multiple computers and peripheral devices connected to a campus or in an office or other room. They are sharing a common connection that has 10-100 Mbps data transmission speed and are connected by Ethernet cables, usually running on high-speed internet connection. LAN computer terminals may be physically connected using cables or setup wireless, thus called WLAN. LAN is less expensive than WAN or MAN.

WAN

This is the abbreviation for Wide Area Network and is the biggest network which can interconnect networks around the world. Companies such as Microsoft or other worldwide organizations utilize WAN connection between their various branches by communicating via microwave satellites.

WAN has a data transmission speed of 256Kbps to 2Mbps, offering a faster speed than LAN or MAN. WAN is used to connect LANs that are not in the same area and is more expensive than LAN or MAN.

MAN

MAN is the abbreviation for Metropolitan Area Network and bigger than LAN network. It connects computer users that are in a specific geographical area. An example of MAN is your cable television or a large university.

MAN's data transmission speed is 5-10Mbps, which is faster and more expensive than LAN but slower and smaller than WAN.

1.1. USES OF COMPUTER NETWORK

The computer networks are playing an important role in providing services to large organizations as well as to the individual common man.

Service Provided by the Network for Companies:

• Many organizations have a large number of computers in operation. These computers may be within the same building, campus, city or different cities.

• Even though the computers are located in different locations, the organizations want to keep track of inventories, monitor productivity, do the ordering and billing etc.

• The computer networks are useful to the organizations in the following ways:

1. Resource sharing.

2. for providing high reliability.

3. To save money.

4. It can provide a powerful communication medium.

The computer networks offer the following services to an individual person:

1. Access to remote information

2. Person to person communication

3. Interactive entertainment.

Access to remote information:

Access to remote information involves interaction· between a person and a remote database. Access to remote information comes in many forms like:

(i) Home shopping, paying telephone, electricity bills, e-banking, on line share market etc.

(ii) Newspaper is. On-line and is personalized, digital library consisting of books, magazines, scientific journals etc.

(iii)World Wide Web which contains information. About the arts, business, cooking, government, health, history, hobbies, recreation, science, sports etc.

Interactive entertainment:

Interactive entertainment includes:

(i) Multi person real-time simulation games.

(ii) Video on demand.

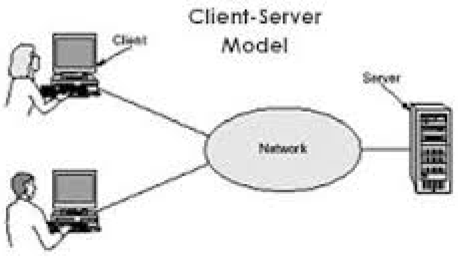
(iii) Participation in live TV programs likes quiz, contest, discussions etc.

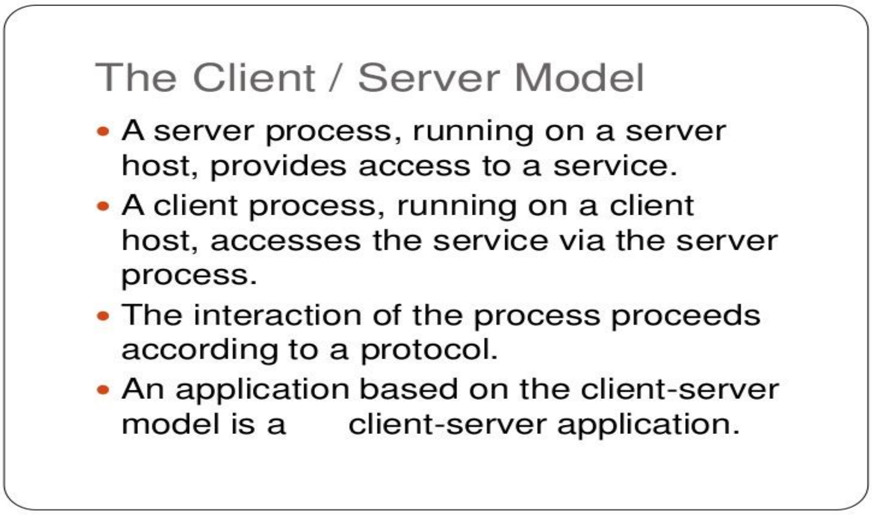
In short, the ability to merge information, communication and entertainment will surely give rise to a massive new industry based on computer networking.

1.2. NETWORKING MODEL

a) Client-Server Model

Client-server architecture (client/server) is a network architecture in which each computer or process on the network is either a *client* or a *server*. Servers are powerful computers or processes dedicated to managing disk drives (*file servers*), printers (*print servers*), or network traffic (network servers). Clients are PCs or workstations on which users run applications. Clients rely on servers for resources, such as files, devices, and even processing power.





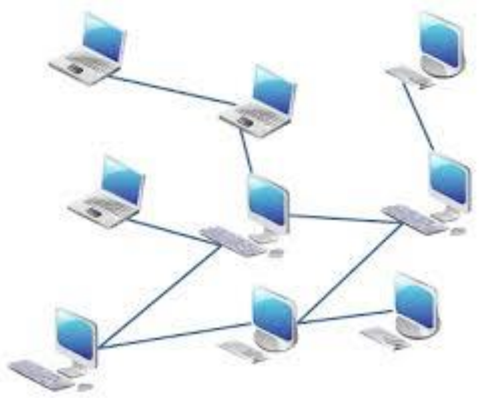
The Client / Server Model

- A server process, running on a server host, provides access to a service.
- A client process, running on a client host, accesses the service via the server process.
- The interaction of the process proceeds according to a protocol.
- An application based on the client-server model is a client-server application.

Fig: Client-Server model

b). P2P model

Peer-to-peer (P2P) is a decentralized communications model in which each party has the same capabilities and either party can initiate a communication session. Unlike the client/server model, in which the client makes a service request and the server fulfills the request, the P2P network model allows each node to function as both a client and server. In its simplest form, a peer-to-peer (P2P) network is created when two or more PCs are connected and share resources without going through a separate server computer. Most P2P programs are focused on media sharing.



c) Active network

An active network is a network in which the nodes are programmed to perform custom operations on the messages that pass through the node. For example, a node could be programmed or customized to handle packets on an individual user basis or to handle multicast packets differently than other packets. Active network approaches are expected to be especially important in networks of mobile users. "Smart packets" use a special self-describing language that allows new kinds of information to be carried within a packet and operated on by a node.

1.3. PROTOCOLS AND STANDARDS

A protocol is the special set of rules that end points in a telecommunication connection use when they communicate. Protocols specify interactions between the communicating entities. Protocols exist at several levels in a telecommunication connection. For example, there are protocols for the data interchange at the hardware device level and protocols for data interchange at the application program level. In the standard model known as Open Systems Interconnection (OSI), there are one or more protocols at each layer in the telecommunication exchange that both ends of the exchange must recognize and observe. Protocols are often described in an industry or international standard. Standard is a common set of rules.

NEED OF LAYERED ARCHITECTURE IN COMPUTER NETWORK

- It simplifies the design process as the functions of each layers and their interactions are well defined.
- The layered architecture provides flexibility to modify and develop network services.
- The number of layers, name of layers and the tasks assigned to them may change from network to network. But for all the networks, always the lower layer offers certain services to its upper layer.
- The concept of layered architecture redefines the way of convincing networks. This leads to a considerable cost savings and managerial benefits.
- Addition of new services and management of network infrastructure become easy.

DESIGN ISSUE OF LAYERED ARCHITECTURE IN COMPUTER NETWORK

There might be a negative impact on the performance as we have the extra overhead of passing through layers instead of calling a component directly.

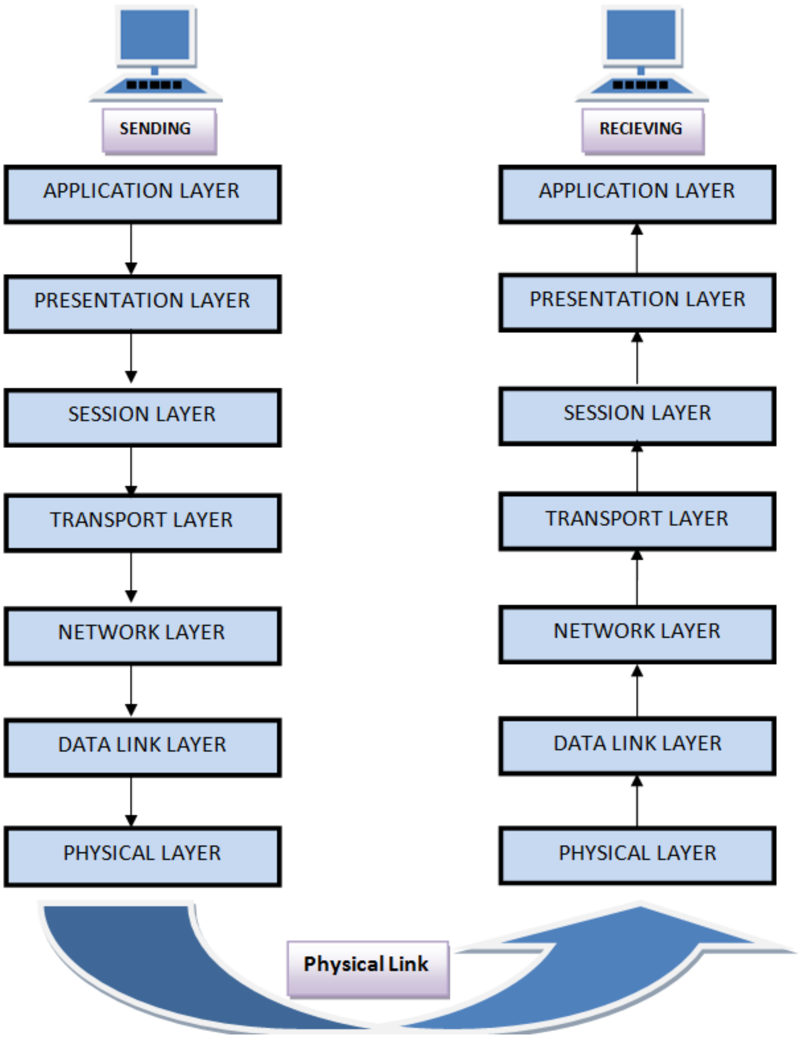
Development of user-intensive applications can sometime take longer if the layering prevents the use of user interface components that directly interact with the database.

The use of layers helps to control and encapsulate the complexity of large applications, but adds complexity to simple applications.

Changes to lower level interfaces tend to percolate to higher levels, especially if the relaxed layered approach is used.

1.4. OSI MODEL AND TCP/IP MODEL

There are many users who use computer network and are located all over the world. To ensure national and worldwide data communication ISO (ISO stands for International Organization of Standardization.) developed this model. This is called a model for open system interconnection (OSI) and is normally called as OSI model.OSI model architecture consists of seven layers.



Layer 1: The Physical Layer:

1. It is the lowest layer of the OSI Model.
2. It activates, maintains and deactivates the physical connection.
3. It is responsible for transmission and reception of the unstructured raw data over network.
4. Voltages and data rates needed for transmission is defined in the physical layer.
5. It converts the digital/analog bits into electrical signal or optical signals.
6. Data encoding is also done in this layer.

Layer 2: Data Link Layer:

1. Data link layer synchronizes the information which is to be transmitted over the physical layer.
2. The main function of this layer is to make sure data transfer is error free from one node to another, over the physical layer.
3. Transmitting and receiving data frames sequentially is managed by this layer.
4. This layer sends and expects acknowledgements for frames received and sent respectively. Resending of non-acknowledgement received frames is also handled by this layer.
5. This layer establishes a logical layer between two nodes and also manages the Frame traffic

control over the network. It signals the transmitting node to stop, when the frame buffers are full.

Layer 3: The Network Layer:

1. It routes the signal through different channels from one node to other.
2. It acts as a network controller. It manages the Subnet traffic.
3. It decides by which route data should take.
4. It divides the outgoing messages into packets and assembles the incoming packets into messages for higher levels.

Layer 4: Transport Layer:

1. It decides if data transmission should be on parallel path or single path.
2. Functions such as Multiplexing, Segmenting or Splitting on the data are done by this layer
3. It receives messages from the Session layer above it, convert the message into smaller units and passes it on to the Network layer.
4. Transport layer can be very complex, depending upon the network requirements.

Transport layer breaks the message (data) into small units so that they are handled more efficiently by the network layer.

Layer 5: The Session Layer:

1. Session layer manages and synchronize the conversation between two different applications.
2. Transfer of data from source to destination session layer streams of data are marked and are resynchronized properly, so that the ends of the messages are not cut prematurely and data loss is avoided.

Layer 6: The Presentation Layer:

1. Presentation layer takes care that the data is sent in such a way that the receiver will understand the information (data) and will be able to use the data.
2. While receiving the data, presentation layer transforms the data to be ready for the application layer.
3. Languages (syntax) can be different of the two communicating systems. Under this condition presentation layer plays a role of translator.
4. It performs Data compression, Data encryption, Data conversion etc.

Layer 7: Application Layer:

1. It is the topmost layer.
2. Transferring of files disturbing the results to the user is also done in this layer. Mail services, directory services, network resource etc are services provided by application layer.
3. This layer mainly holds application programs to act upon the received and to be sent data. Merits of OSI reference model:

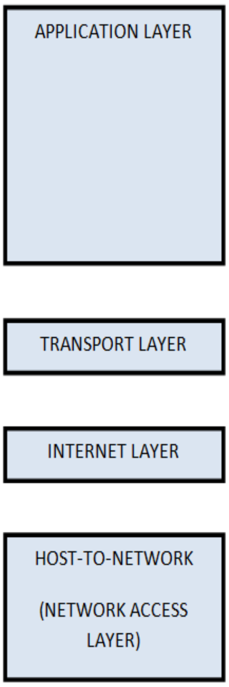
1. OSI model distinguishes well between the services, interfaces and protocols.
2. Protocols of OSI model are very well hidden.
3. Protocols can be replaced by new protocols as technology changes.

4. Supports connection oriented services as well as connectionless service.

Demerits of OSI reference model:

1. Model was devised before the invention of protocols.
2. Fitting of protocols is tedious task.
3. It is just used as a reference model.

TCP/IP is transmission control protocol and internet protocol. Protocols are set of rules which govern every possible communication over the internet. These protocols describe the movement of data between the host computers or internet and offers simple naming and addressing schemes.



Overview of TCP/IP reference model

TCP/IP that is Transmission Control Protocol and Internet Protocol was developed by Department of Defense's Project Research Agency (ARPA, later DARPA) as a part of a research project of network interconnection to connect remote machines.

The features that stood out during the research, which led to making the TCP/IP reference model were:

- Support for a flexible architecture. Adding more machines to a network was easy.
- The network was robust, and connections remained intact until the source and destination machines were functioning.

The overall idea was to allow one application on one computer to talk to(send data packets) another application running on different computer.

Description of different TCP/IP protocols

Layer 1: Host-to-network Layer

1. Lowest layer of the all.
2. Protocol is used to connect to the host, so that the packets can be sent over it.
3. Varies from host to host and network to network.

Layer 2: Internet layer

1. Selection of a packet switching network which is based on a connectionless internetwork layer is called a internet layer.
2. It is the layer which holds the whole architecture together.
3. It helps the packet to travel independently to the destination.
4. Order in which packets are received is different from the way they are sent.
5. IP (Internet Protocol) is used in this layer.

Layer 3: Transport Layer

1. It decides if data transmission should be on parallel path or single path.
2. Functions such as multiplexing, segmenting or splitting on the data is done by transport layer.
3. The applications can read and write to the transport layer.
4. Transport layer adds header information to the data.
5. Transport layer breaks the message (data) into small units so that they are handled more efficiently by the network layer.
6. Transport layer also arrange the packets to be sent, in sequence.

Layer 4: Application Layer

The TCP/IP specifications described a lot of applications that were at the top of the protocol stack. Some of them were TELNET, FTP, SMTP, DNS etc.

1. TELNET is a two-way communication protocol which allows connecting to a remote machine and run applications on it.
2. FTP (File Transfer Protocol) is a protocol that allows File transfer amongst computer users connected over a network. It is reliable, simple and efficient.
3. SMTP (Simple Mail Transport Protocol) is a protocol, which is used to transport electronic mail between a source and destination, directed via a route.
4. DNS (Domain Name Server) resolves an IP address into a textual address for Hosts connected over a network.

Merits of TCP/IP model

1. It operated independently.
2. It is scalable.
3. Client/server architecture.
4. Supports a number of routing protocols.
5. Can be used to establish a connection between two computers.

Demerits of TCP/IP

1. In this, the transport layer does not guarantee delivery of packets.
2. The model cannot be used in any other application.
3. Replacing protocol is not easy.
4. It has not clearly separated its services, interfaces and protocols.

1.5. COMPARISION OF OSI AND TCP/IP

OSI(Open System Interconnection)	**TCP/IP(Transmission Control Protocol / Internet Protocol)**
1. OSI is a generic, protocol independent standard, acting as a communication gateway between the network and end user.	1. TCP/IP model is based on standard protocols around which the Internet has developed. It is a communication protocol, which allows connection of hosts over a network.
2. In OSI model the transport layer guarantees the delivery of packets.	2. In TCP/IP model the transport layer does not guarantees delivery of packets. Still the TCP/IP model is more reliable.
3. Follows vertical approach.	3. Follows horizontal approach.
4. OSI model has a separate Presentation layer and Session layer.	4. TCP/IP does not have a separate Presentation layer or Session layer.
5. OSI is a reference model around which the networks are built. Generally it is used as a guidance tool.	5. TCP/IP model is, in a way implementation of the OSI model.
6. Network layer of OSI model provides both connection oriented and connectionless service.	6. The Network layer in TCP/IP model provides connectionless service.
7. OSI model has a problem of fitting the protocols into the model.	7. TCP/IP model does not fit any protocol
8. Protocols are hidden in OSI model and are easily replaced as the technology changes.	8. In TCP/IP replacing protocol is not easy.
9. OSI model defines services, interfaces and protocols very clearly and makes clear distinction between them. It is protocol independent.	9. In TCP/IP, services, interfaces and protocols are not clearly separated. It is also protocol dependent.
10. It has 7 layers	10. It has 4 layers

1.6. EXAMPLE NETWORK

VoIP

Once upon a time, the public switched telephone system was primarily used for voice traffic with a little bit of data traffic here and there. But the data traffic grew and grew, and by 1999, the number of data bits moved equaled the number of voice bits (since voice is in PCM on the trunks, it can be measured in bits/sec). By 2002, the volume of data traffic was an order of magnitude more than the volume of voice traffic and still growing exponentially, with voice traffic being almost flat (5% growth per year).

As a consequence of these numbers, many packet-switching network operators suddenly became interested in carrying voice over their data networks. The amount of additional bandwidth required for voice is minuscule since the packet networks are dimensioned for the data traffic. However, the average person's phone bill is probably larger than his Internet bill, so the data network operators saw Internet telephony as a way to earn a large amount of additional money without having to put any new fiber in the ground.

Voice over IP (VoIP) commonly refers to the communication protocols, technologies, methodologies, and transmission techniques involved in the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.

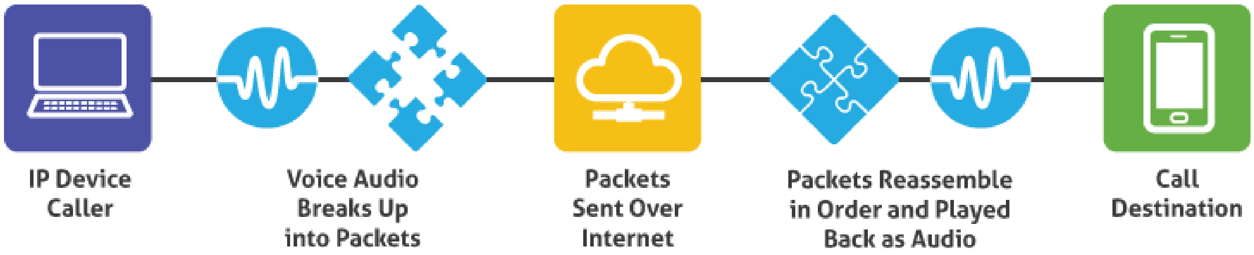
The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream. Even though IP Telephony and VoIP are terms that are used interchangeably, they are actually different; IP telephony has to do with digital telephony systems that use IP protocols for voice communication, while VoIP is actually a subset of IP Telephony. VoIP is a technology used by IP telephony as a means of transporting phone calls.

VoIP systems employ session control protocols to control the set-up and tear-down of calls as well as audio codec which encode speech allowing transmission over an IP network as digital audio via an audio stream. VoIP is available on many smart phones and Internet devices so that users of portable devices that are not phones, may place calls or send SMS text messages over 3G or Wi-Fi.

A VoIP phone is necessary to connect to a VoIP service provider. This can be implemented in several ways:

- Dedicated VoIP phones connect directly to the IP network using technologies such as wired Ethernet or wireless Wi-Fi. They are typically designed in the style of traditional digital business telephones.
- An analog telephone adapter is a device that connects to the network and implements the electronics and firmware to operate a conventional analog telephone attached through a modular phone jack. Some residential Internet gateways and cable modems have this function built in.
- A soft phone is application software installed on a networked computer that is equipped with

a microphone and speaker, or headset. The application typically presents a dial pad and display field to the user to operate the application by mouse clicks or keyboard input.


IP Device Caller
Voice Audio Breaks Up into Packets
Packets Sent Over Internet
Packets Reassemble in Order and Played Back as Audio
Call Destination

Advantages

a. Operational Cost
b. Quality of Service
c. Portability
d. Features like call forwarding, call waiting, three party conversation
e. Flexibility

Disadvantages

a. No service during power outage
b. Reliability
c. Security

NGN

A next-generation network (NGN) is a packet-based network which can provide services including Telecommunication Services and able to make use of multiple broadband, quality of Service-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users. A **N**ext-**G**eneration **N**etwork (**NGN**) is the term given to describe a telecommunications packet-based network that handles multiple types of traffic (such as voice, data, and multimedia). It is the convergence of service provider networks that includes the public switched telephone network (PSTN), the data network (the Internet), and, in some instances, the wireless network as well.

The NGN is characterized by the following fundamental aspects:

- Packet-based transfer
- Separation of control functions among bearer capabilities, call/session, and application/ service
- Decoupling of service provision from network, and provision of open interfaces
- Support for a wide range of services, applications and mechanisms based on service building

blocks (including real time/ streaming/ non-real time services and multi-media)

- Broadband capabilities with end-to-end QoS and transparency
- Interworking with legacy networks via open interfaces
- Generalized mobility
- Unrestricted access by users to different service providers
- A variety of identification schemes which can be resolved to IP addresses for the purposes of routing in IP networks
- Unified service characteristics for the same service as perceived by the user
- Converged services between Fixed/Mobile
- Independence of service-related functions from underlying transport technologies
- Compliant with all Regulatory requirements, for example concerning emergency communications and security/privacy, etc.

MPLS

Multiprotocol Label Switching (**MPLS**) is a mechanism in high-performance telecommunications networks that directs data from one network node to the next based on short path labels rather than long network addresses, avoiding complex lookups in a routing table. The labels identify virtual links (paths) between distant nodes rather than endpoints. MPLS can encapsulate packets of various network protocols. MPLS supports a range of access technologies, including T1/E1, ATM, Frame Relay, and DSL.

MPLS is a highly scalable, protocol agnostic, data-carrying mechanism. In an MPLS network, data packets are assigned labels. Packet-forwarding decisions are made solely on the contents of this label, without the need to examine the packet itself. This allows one to create end-to-end circuits across any type of transport medium, using any protocol. The primary benefit is to eliminate dependence on a particular OSI model data link layer technology, such as Asynchronous Transfer Mode (ATM), Frame Relay, Synchronous Optical Networking (SONET) or Ethernet, and eliminate the need for multiple layer-2 networks to satisfy different types of traffic. MPLS belongs to the family of packet-switched networks.

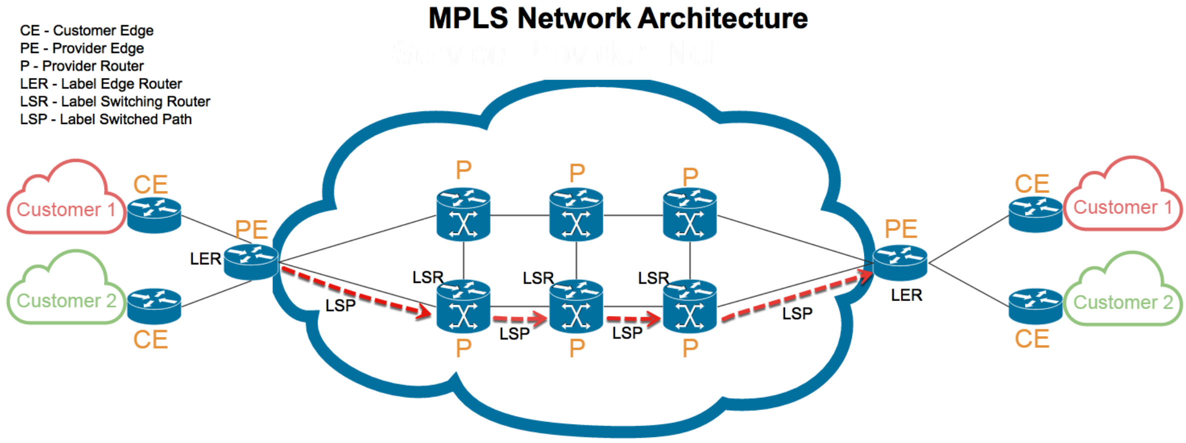
MPLS operates at a layer that is generally considered to lie between traditional definitions of layer 2 (data link layer) and layer 3 (network layer), and thus is often referred to as a “layer 2.5” protocol. It was designed to provide a unified data-carrying service for both circuit-based clients and packet-switching clients which provide a datagram service model. It can be used to carry many different kinds of traffic, including IP packets, as well as native ATM, SONET, and Ethernet frames. A number of different technologies were previously deployed with essentially identical goals, such as Frame Relay and ATM. MPLS technologies have evolved with the strengths and weaknesses of ATM in mind. Many network engineers agree that ATM should be replaced with a protocol that requires less overhead, while providing connection-oriented services for variable-length frames. MPLS is currently replacing some of these technologies in the marketplace. It is highly possible that MPLS will completely replace these technologies in the future, thus aligning these technologies with current and future technology needs.

Features

a. Packet classification
b. Congestion avoidance
c. Congestion management
d. Path Protection
e. Security

Advantage

a. Scalability of network layer routing
b. Flexibility of delivering routing services
c. Increased performance



xDSL

When the telephone industry finally got to 56 kbps, it patted itself on the back for a job well done. Meanwhile, the cable TV industry was offering speeds up to 10 Mbps on shared cables, and satellite companies were planning to offer upward of 50 Mbps. As Internet access became an increasingly important part of their business, the telephone companies began to realize they needed a more competitive product. Their answer was to start offering new digital services over the local loop. Services with more bandwidth than standard telephone service are sometimes called broadband, although the term really is more of a marketing concept than a specific technical concept.

Initially, there were many overlapping offerings, all under the general name of xDSL (Digital Subscriber Line), for various x. Below we will discuss these but primarily focus on what is probably going to become the most popular of these services, ADSL (Asymmetric DSL).

The reason that modems are so slow is that telephones were invented for carrying the human voice and the entire system has been carefully optimized for this purpose. Data have always been stepchildren. At the point where each local loop terminates in the end office, the wire runs through a filter that attenuates all frequencies below 300 Hz and above 3400 Hz. The cutoff is not sharp—300 Hz and 3400 Hz are the 3 dB points—so the bandwidth is usually quoted as 4000 Hz even though the distance between the 3 dB points is 3100 Hz. Data are thus also restricted to this narrow band.

The trick that makes xDSL work is that when a customer subscribes to it, the incoming line is connected to a different kind of switch, one that does not have this filter, thus making the entire capacity of the local loop available. The limiting factor then becomes the physics of the local loop, not the artificial 3100 Hz bandwidth created by the filter. Unfortunately, the capacity of the local loop depends on several factors, including its length, thickness, and general quality.

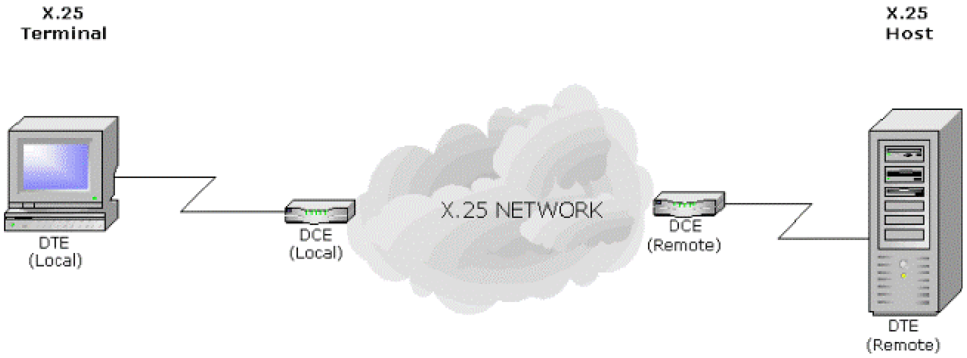
The xDSL services have all been designed with certain goals in mind. First, the services must work over the existing twisted pair local loops. Second, they must not affect customers' existing telephones and fax machines. Third, they must be much faster than 56 kbps. Fourth, they should be always on, with just a monthly charge but no per-minute charge.

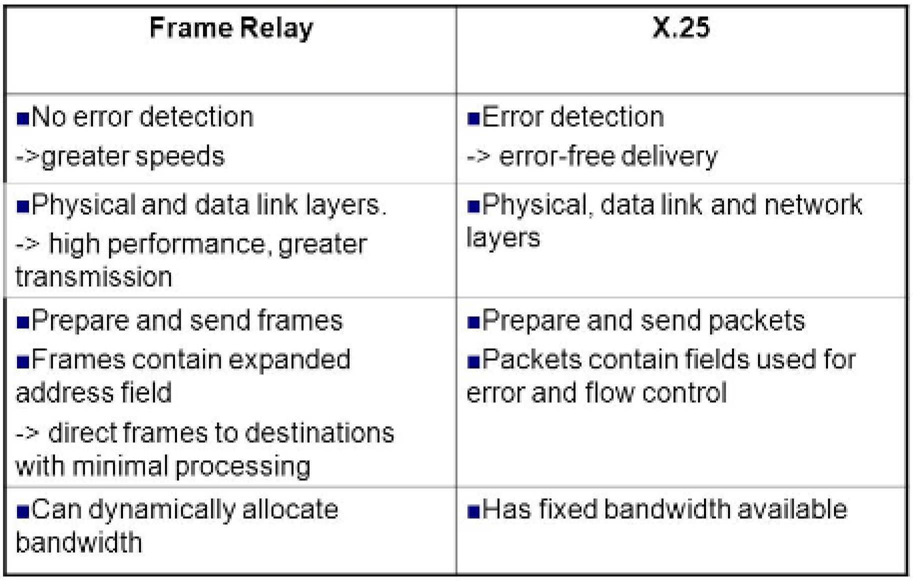
X.25

A connection-oriented network is X.25, which was the first public data network. It was deployed in the 1970s at a time when telephone service was a monopoly everywhere and the telephone company in each country expected there to be one data network per country—theirs. To use X.25, a computer first established a connection to the remote computer, that is, placed a telephone call. This connection was given a connection number to be used in data transfer packets (because multiple connections could be open at the same time). Data packets were very simple, consisting of a 3-byte header and up to 128 bytes of data. The header consisted of a 12-bit connection number, a packet sequence number, an acknowledgement number, and a few miscellaneous bits. X.25 networks operated for about a decade with mixed success.

Frame Relay

In the 1980s, the X.25 networks were largely replaced by a new kind of network called frame relay. The essence of frame relay is that it is a connection-oriented network with no error control and no flow control. Because it was connection-oriented, packets were delivered in order (if they were delivered at all). The properties of in-order delivery, no error control, and no flow control make frame relay akin to a wide area LAN. Its most important application is interconnecting LANs at multiple company offices. Frame relay enjoyed a modest success and is still in use in places today.





Frame Relay	X.25
■No error detection ->greater speeds	■Error detection -> error-free delivery
■Physical and data link layers. -> high performance, greater transmission	■Physical, data link and network layers
■Prepare and send frames ■Frames contain expanded address field -> direct frames to destinations with minimal processing	■Prepare and send packets ■Packets contain fields used for error and flow control
■Can dynamically allocate bandwidth	■Has fixed bandwidth available

Ethernet (IEEE 802.3) Local Area Network (LAN)

Ethernet protocols refer to the family of local-area network (LAN) covered by the IEEE 802.3. In the Ethernet standard, there are two modes of operation: half-duplex and full-duplex modes. In the half duplex mode, data are transmitted using the popular Carrier-Sense Multiple Access/Collision Detection (CSMA/CD) protocol on a shared medium. The main disadvantages of the half-duplex are the efficiency and distance limitation, in which the link distance is limited by the minimum MAC frame size. This restriction reduces the efficiency drastically for high-rate transmission. Therefore, the carrier extension technique is used to ensure the minimum frame size of 512 bytes in Gigabit Ethernet to achieve a reasonable link distance.

Four data rates are currently defined for operation over optical fiber and twisted-pair cables:

- 10 Mbps – 10Base-T Ethernet (IEEE 802.3)
- 100 Mbps – Fast Ethernet (IEEE 802.3u)
- 1000 Mbps – Gigabit Ethernet (IEEE 802.3z)
- 10-Gigabit – 10 Gbps Ethernet (IEEE 802.3ae).

In this document, we discuss the general aspects of the Ethernet. The specific issues regarding Fast Ethernet, Gigabit and 10 Gigabit Ethernet will be discussed in separate documents.

The Ethernet system consists of three basic elements: 1. the physical medium used to carry Ethernet signals between computers, 2. a set of medium access control rules embedded in each Ethernet interface that allow multiple computers to fairly arbitrate access to the shared Ethernet channel, and 3. an Ethernet frame that consists of a standardized set of bits used to carry data over the system.

As with all IEEE 802 protocols, the ISO data link layer is divided into two IEEE 802 sub layers, the Media Access Control (MAC) sub layer and the MAC-client sub layer. The IEEE 802.3 physical layer corresponds to the ISO physical layer.

The MAC sub-layer has two primary responsibilities:

- Data encapsulation, including frame assembly before transmission, and frame parsing/error detection during and after reception

- Media access control, including initiation of frame transmission and recovery from transmission failure

The MAC-client sub-layer may be one of the following:

- Logical Link Control (LLC), which provides the interface between the Ethernet MAC and the upper layers in the protocol stack of the end station. The LLC sub layer is defined by IEEE 802.2 standards.
- Bridge entity, which provides LAN-to-LAN interfaces between LANs that use the same protocol (for example, Ethernet to Ethernet) and also between different protocols (for example, Ethernet to Token Ring). Bridge entities are defined by IEEE 802.1 standards.

Each Ethernet-equipped computer operates independently of all other stations on the network: there is no central controller. All stations attached to an Ethernet are connected to a shared signaling system, also called the medium. To send data a station first listens to the channel, and when the channel is idle the station transmits its data in the form of an Ethernet frame, or packet.

After each frame transmission, all stations on the network must contend equally for the next frame transmission opportunity. Access to the shared channel is determined by the medium access control (MAC) mechanism embedded in the Ethernet interface located in each station. The medium access control mechanism is based on a system called Carrier Sense Multiple Access with Collision Detection (CSMA/CD).

As each Ethernet frame is sent onto the shared signal channel, all Ethernet interfaces look at the destination address. If the destination address of the frame matches with the interface address, the frame will be read entirely and be delivered to the networking software running on that computer. All other network interfaces will stop reading the frame when they discover that the destination address does not match their own address.

When it comes to how signals flow over the set of media segments that make up an Ethernet system, it helps to understand the topology of the system. The signal topology of the Ethernet is also known as the logical topology, to distinguish it from the actual physical layout of the media cables. The logical topology of an Ethernet provides a single channel (or bus) that carries Ethernet signals to all stations.

CHAPTER 2- PHYSICAL LAYER

PHYSICAL LAYER

Physical layer is the lowest layer of all. It is responsible for sending bits from one computer to another. This layer is not concerned with the meaning of the bits and deals with the physical connection to the network and with transmission and reception of signals.

This layer defines electrical and physical details represented as 0 or a 1. How many pins a network will contain, when the data can be transmitted or not and how the data would be synchronized.

FUNCTIONS OF PHYSICAL LAYER:

1. **Representation of Bits:** Data in this layer consists of stream of bits. The bits must be encoded into signals for transmission. It defines the type of encoding i.e. how 0’s and 1’s are changed to signal.
2. **Data Rate:** This layer defines the rate of transmission which is the number of bits per second.
3. **Synchronization:** It deals with the synchronization of the transmitter and receiver. The sender and receiver are synchronized at bit level.
4. **Interface:** The physical layer defines the transmission interface between devices and transmission medium.
5. **Line Configuration:** This layer connects devices with the medium: Point to Point configuration and Multipoint configuration.
6. **Topologies:** Devices must be connected using the following topologies: Mesh, Star, Ring and Bus.
7. **Transmission Modes:** Physical Layer defines the direction of transmission between two devices: Simplex, Half Duplex, Full Duplex.
8. Deals with baseband and broadband transmission.

2.1. NETWORK MONITORING:

a) Delay

Network delay is an important design and performance characteristic of a computer **network** or telecommunications **network**. The **delay** of a **network** specifies how long it takes for a bit of data to travel across the **network** from one node or endpoint to another. It is typically measured in multiples or fractions of seconds.

b) Latency

Network latency is an expression of how much time it takes for a packet of data to get from one designated point to another. In some environments (for example, AT&T), **latency** is measured by sending a packet that is returned to the sender; the round-trip time is considered the **latency**.

c) Throughput

A benchmark can be used to measure **throughput**. In data transmission, **network throughput** is the amount of data moved successfully from one place to another in a given time period, and typically measured in bits per second (bps), as in megabits per second (Mbps) or gigabits per second (Gbps).

2.2. TRANSMISSION MEDIA

These are the means by which a communication signal is carried from one system to another. These media can carry information from a source to a destination. The transmission media can usually be free space such as: satellite, microwave, radio and infrared systems, metallic cables such as: twisted pair, or coaxial cable, or fiber-optic cable.

In telecommunication, transmission media can be divided into two broad categories:

i. Guided transmission media

ii. Unguided transmission media

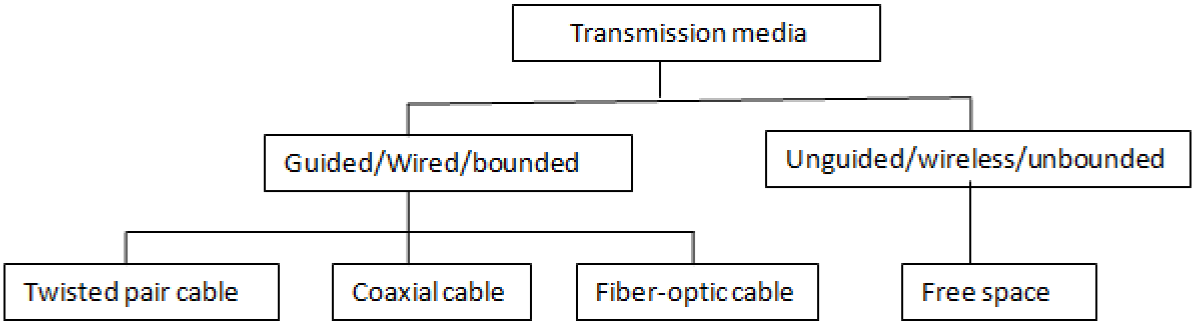


Fig: Classes of transmission media

Guided Transmission Media

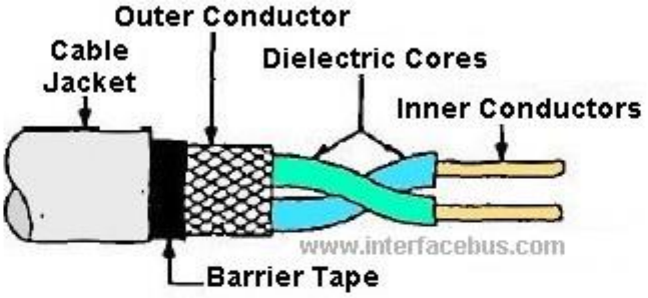
Guided Transmission media uses a cabling system that guides the data signals along a specific path. They provide the physical path way for the transmission of the data from the source to the destination. The data signals travelling along any of these media is directed and contained by the physical limits of the medium. Twisted-pair and coaxial cable use metallic (copper) conductor that accept and transport signals in the form of electric current. Optical fiber is a cable that accepts and transport signals in the form of light. Guided Media are also known as Bound media or wired media.

a) Twisted Pair Cable

Twisted pair cabling is a type of wiring in which two conductors of a single circuit are twisted together for the purposes of canceling out electromagnetic interference (EMI) from external sources; for instance, electromagnetic radiation from unshielded twisted pair (UTP) cables, and crosstalk between neighboring pairs.

This cable is the most commonly used and is cheaper than others. It is lightweight, cheap, can be installed easily, and they support many different types of network. Some important points:

- Its frequency range is 0 to 3.5 kHz.
- Typical attenuation is 0.2 dB/Km @ 1kHz.
- Typical delay is 50 µs/km.
- Repeater spacing is 2km.



Twisted Pair is of two types :

- Unshielded Twisted Pair (UTP)
- Shielded Twisted Pair (STP)

Unshielded Twisted Pair Cable

It is the most common type of telecommunication when compared with Shielded Twisted Pair Cable which consists of two conductors usually copper, each with its own colour plastic insulator. Identification is the reason behind colored plastic insulation.

Shielded Twisted Pair Cable

This cable has a metal foil or braided-mesh covering which encases each pair of insulated conductors. Electromagnetic noise penetration is prevented by metal casing. Shielding also eliminates crosstalk (explained in KEY TERMS Chapter).

It has same attenuation as unshielded twisted pair. It is faster the unshielded and coaxial cable. It is more expensive than coaxial and unshielded twisted pair.

b) Coaxial Cable

Coaxial is called by this name because it contains two conductors that are parallel to each other. Copper is used in this as centre conductor which can be a solid wire or a standard one. It is surrounded by PVC installation, a sheath which is encased in an outer conductor of metal foil, braid or both.

Outer metallic wrapping is used as a shield against noise and as the second conductor which completes the circuit. The outer conductor is also encased in an insulating sheath. The outermost part is the plastic cover which protects the whole cable.

Here the most common coaxial standards.

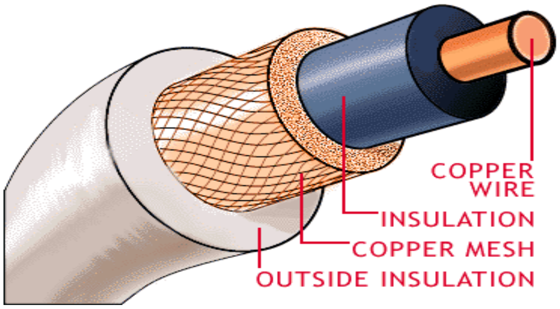
- 50-Ohm RG-7 or RG-11 : used with thick Ethernet.
- 50-Ohm RG-58 : used with thin Ethernet
- 75-Ohm RG-59 : used with cable television
- 93-Ohm RG-62 : used with ARCNET.

Advantages:

- Bandwidth is high
- Used in long distance telephone lines.
- Transmits digital signals at a very high rate of 10Mbps.
- Much higher noise immunity
- Data transmission without distortion.
- The can span to longer distance at higher speeds as they have better shielding when compared to twisted pair cable

Disadvantages:

- Single cable failure can fail the entire network.
- Difficult to install and expensive when compared with twisted pair.
- If the shield is imperfect, it can lead to grounded loop.



C) Fiber Optic Cable

These are similar to coaxial cable. It uses light signals to transmit data. At the centre is the glass core through which light propagates.

In multimode fibers, the core is 50microns, and In single mode fibers, the thickness is 8 to 10 microns.

The core in fiber optic cable is surrounded by glass cladding with lower index of refraction as compared to core to keep all the light in core. This is covered with a thin plastic jacket to protect the cladding. The fibers are grouped together in bundles protected by an outer shield.

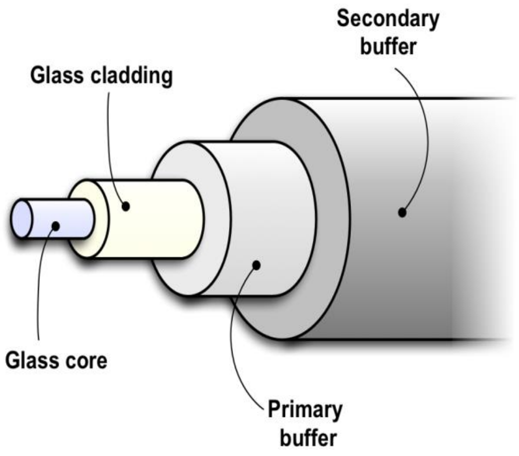
Fiber optic cable has bandwidth more than **2 gbps (Gigabytes per Second)**

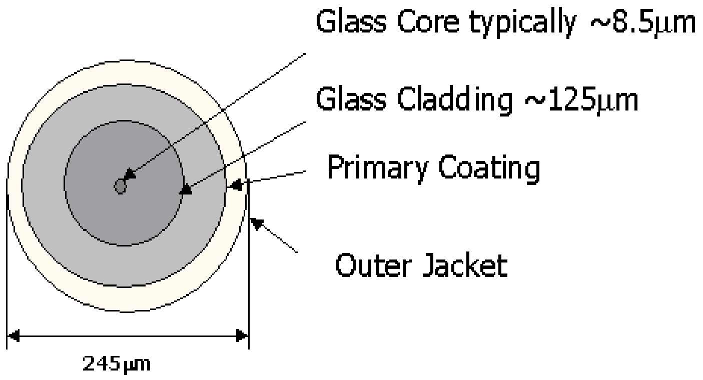
Advantages:

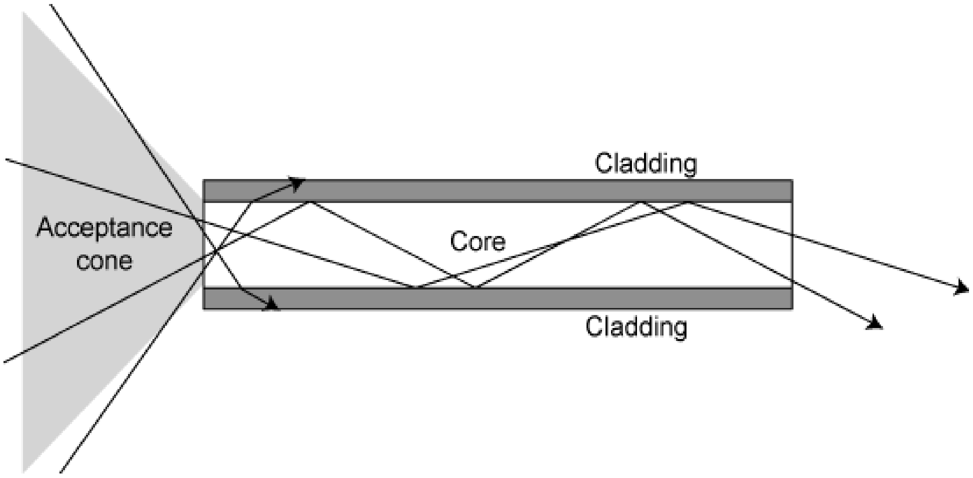
- Provides high quality transmission of signals at very high speed.
- These are not affected by electromagnetic interference, so noise and distortion is very less.
- Used for both analog and digital signals.

Disadvantages:

- It is expensive
- Difficult to install.
- Maintenance is expensive and difficult.
- Do not allow complete routing of light signals.







II) Unguided Media

The unguided media is the wireless media. It simply transports electromagnetic waves without using any physical conductor. Signals are normally broadcast through the air and thus are available to anyone who has the device capable of receiving them.

Wireless transmission

Wireless communication is the transfer of information over a distance without the use of electrical conductors or “wires”. The distances involved may be short (a few meters as in television remote control) or very long (thousands or even millions of kilometers for radio communication). Wireless communication is generally considered to be a branch of telecommunications.

Three ways for wireless data propagation

(i) Radio wave

(ii) Microwave

(iii) Infrared

Ground propagation

Radio waves travel through the lowest portion of the atmosphere, hugging he earth. These low frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet. Distance depends on the amount of power in the signal, that is the greater the power, the greater the distance.

Sky propagation

Higher-frequency radio waves radiate upward into the ionosphere where they are reflected back to earth. This type of transmission allows for greater distances with lower power output.

Line-of-sight propagation

In line-of-sight (LOS) propagation very high frequency signals are transmitted in straight lines directly from antenna-to-antenna. Antenna must be directional, facing each other and either tall enough or close enough together not to be affected by the curvature of the earth. LOS propagation is trekking because radio transmission cannot be completely focused.

Radio waves:

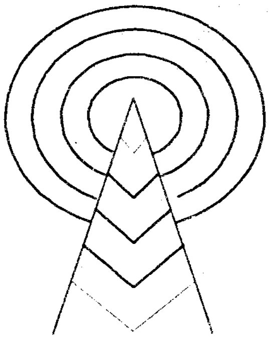
Electromagnetic wave ranging in frequencies between 3 KHz and 1GHz are normally called radio waves.

Radio waves are omnidirectional when an antenna transmits radio waves they are propagated in all directions. This means that sending and receiving antenna do not have to be aligned. A sending antenna can send waves that can be received by any receiving antenna. This omnidirectional property is the disadvantage that the radio waves transmitted by tone antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Radio waves particularly those waves that propagate in sky mode, can travel long distances. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Radio waves particularly those of low and medium frequencies can penetrate walls. It is an advantage because; an AM radio can receive signals inside a building. It is the disadvantage because we cannot isolate a communication to first inside or outside a building. The radio waves band is relatively narrow just under I GHz, compared to the microwave band. When this band is divided into sub-band, the sidebands are also narrow, leading to a low data rate for digital communications.

Almost all the entire band is regulated by authorities using from the authorities. Radio waves use omnidirectional antennas that send our signals in all directions based on the wave lengths, strength and purpose of transmission, we can have several types of antennas.



Omnidirectional Antenna

Applications:

The directional characteristics of radio waves make them useful for multicasting. In which there is one sender but many receiver. Radio waves are used multicast communications such as radio (AM, FM), TV, paging system, and cordless phones.

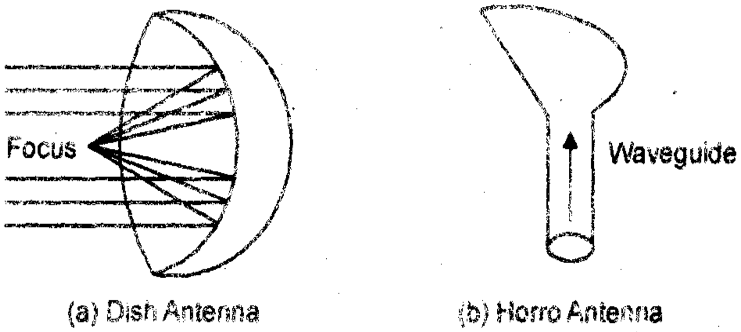
Microwaves:

Electromagnetic waves having frequencies between 1 GHz and 300 GHz are called microwaves.

Microwaves are unidirectional; when an antenna transmits microwaves they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas. The microwave band is relatively wide almost 299GHz. Therefore wider sub-band can be assigned and a high data rate is possible.

On the other hand microwaves:

Propagation is line-of-sight. Since the towers with the mounted antennas needs to be in direct sight of each other, towers that are for apart need to be very tall. The curvature of the earth as well as other blocking obstacles does not allow two short towers to communicate using microwaves. Repeaters are often needed for long distance communication very high frequency microwaves cannot penetrate wall. Use of certain portion of band requires permission from authority. Parabolic dish antenna and horn antenna are used for this means of transmission.



(a) Dish Antenna (b) Horro Antenna

Applications:

Microwaves are used for unicast communication such as cellular telephones, satellite networks and wireless LAN.

Infrared:

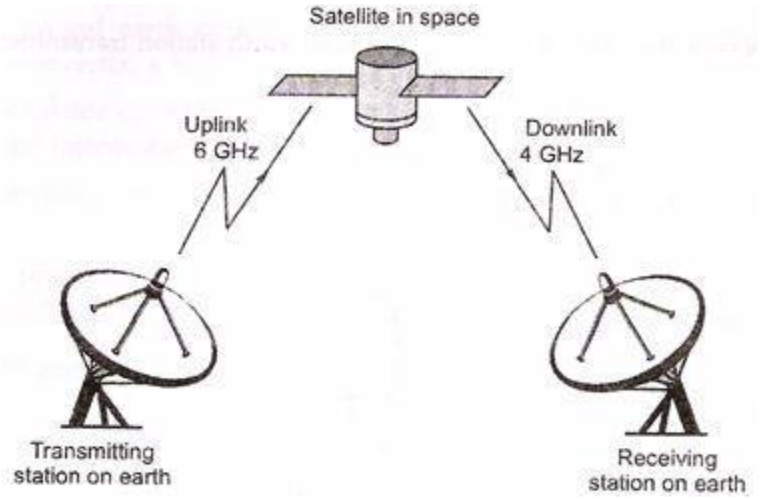
Infrared signals with frequencies ranges from 300 GHz to 400 THz can be used for short range communication. Infrared signals, having high frequencies, cannot penetrate walls. This helps to prevent interference between one system and another. In this, one room cannot be affected by the infrared waves in another room. However, the same characteristics make infrared signals useless for long range communications. In addition, we cannot use infrared waves outside a building because the sun’s rays contain infrared waves that can interfere with the communication.

Applications:

Infrared band, almost 400 THz, has an excellent potential for data transmission. So this will transfer digital data with a very high frequency. There are number of computer devices which are used to send the data through infrared medium e.g. keyboard mice, PCs and printers. There are some manufacturers provide a special part called the IrDA port that allows a wireless keyboard to communicate with a PC.

Communication Satellite:

The concept of satellite based networks is to transmit and receive signals from ground stations. The purpose of satellite communication is to use it for video transmission and sharing. In simple words a satellite is a device which revolves around the earth either for collecting useful information or for helping transfer of information.



LEO is called Low earth orbit, MEO is called Medium Earth Orbit and GEO is called Geostationary orbit. LEO are about 500 Km to 1500 Km above the earth, so the delay is very small and the losses is small too. MEO are installed at 5000 to 12000 km above the earth and generally used for navigation communications like GPS. GEO is about 35800 Km above the equator, the delay and losses are greater, but the advantages is more coverage (it covers 40% of the earth) and there no need to track the satellite, so the earth terminal is cheaper.

Geo-Stationary Earth Orbit

These satellites have almost a distance of 36,000 km to the earth.

E.g. All radio and TV, whether satellite etc, are launched in this orbit.

Advantages of Geo-Stationary Earth Orbit

1. It is possible to cover almost all parts of the earth with just 3 geo satellites.

2. Antennas need not be adjusted every now and then but can be fixed permanently.

3. The life-time of a GEO satellite is quite high usually around 15 years.

Disadvantages of Geo-Stationary Earth Orbit

1. Larger antennas are required for northern/southern regions of the earth.

2. High buildings in a city limit the transmission quality.

3. High transmission power is required.

4. These satellites cannot be used for small mobile phones.

5. Fixing a satellite at Geo stationary orbit is very expensive.

Medium Earth Orbit

Satellite at different orbits operates at different heights. The MEO satellite operates at about 5000 to 12000 km away from the earth's surface.

These orbits have moderate number of satellites.

Advantages of Medium Earth Orbit

1. Compared to LEO system, MEO requires only a dozen satellites.

2. Simple in design.

3. Requires very few handovers.

Disadvantages of Medium Earth Orbit

1. Satellites require higher transmission power.

2. Special antennas are required.

Low Earth Orbit

LEO satellites operate at a distance of about 500-1500 km.

Advantages of Low Earth Orbit

1. The antennas can have low transmission power of about 1 watt.

2. The delay of packets is relatively low.

3. Useful for smaller foot prints.

Disadvantages of Low Earth Orbit

1 If global coverage is required, it requires at least 50-200 satellites in this orbit.

2. Special handover mechanisms are required.

3. These satellites involve complex design.

4. Very short life: Time of 5-8 years. Assuming 48 satellites with a life-time of 8 years each, a new satellite is needed every 2 months.

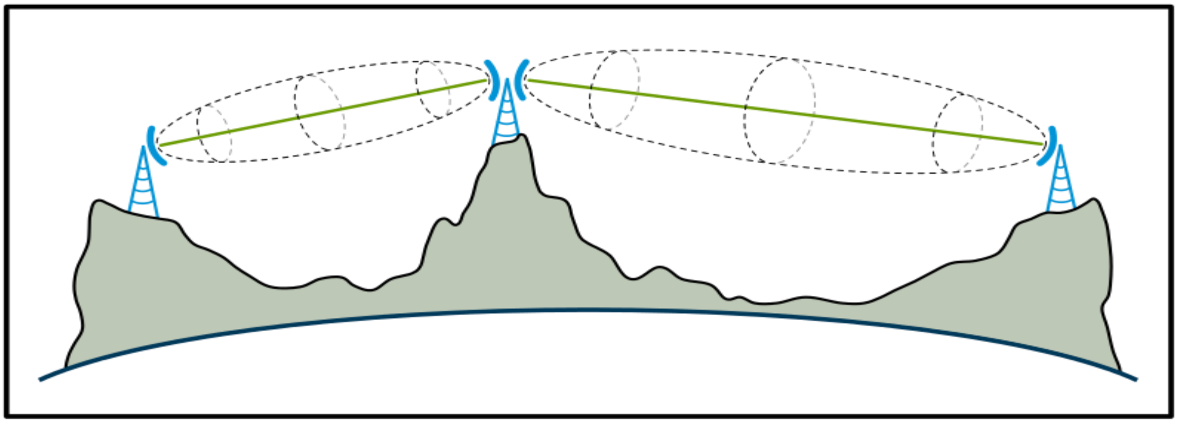
5. Data packets should be routed from satellite to satellite.

LINE OF SIGHT

Line of sight (LoS) is a type of propagation that can transmit and receive data only where transmit and receive stations are in view of each other without any sort of an obstacle between them. FM radio, microwave and satellite transmission are examples of line-of-sight communication.

Long-distance data communication is more effective through wireless networks but geographical obstacles and the curvature of the earth bring limitations to line-of-sight transmission. However, these issues can generally be mitigated through planning, calculations and the use of additional technologies.

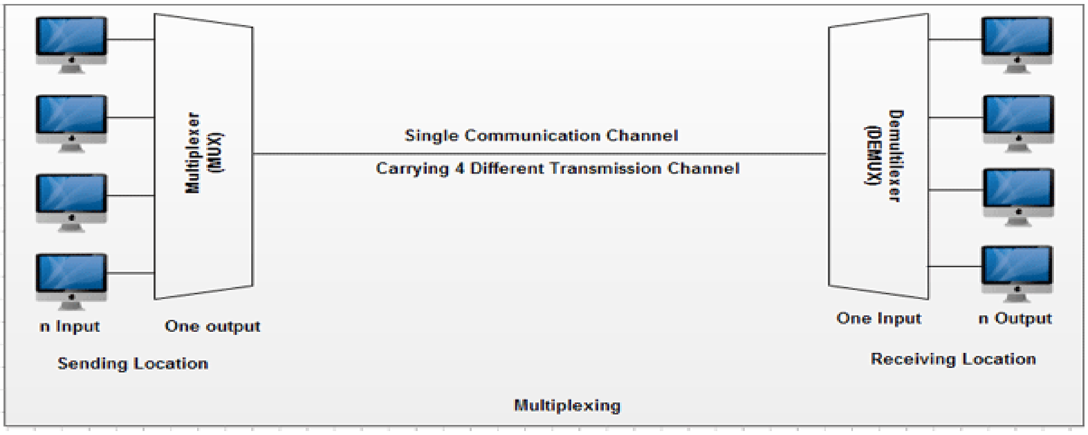
For example, mobile phones use a modified line-of-sight transmission, which is made possible through a combination of effects like diffraction, multipath reflection, local repeaters and rapid handoff.



2.3 MULTIPLEXING

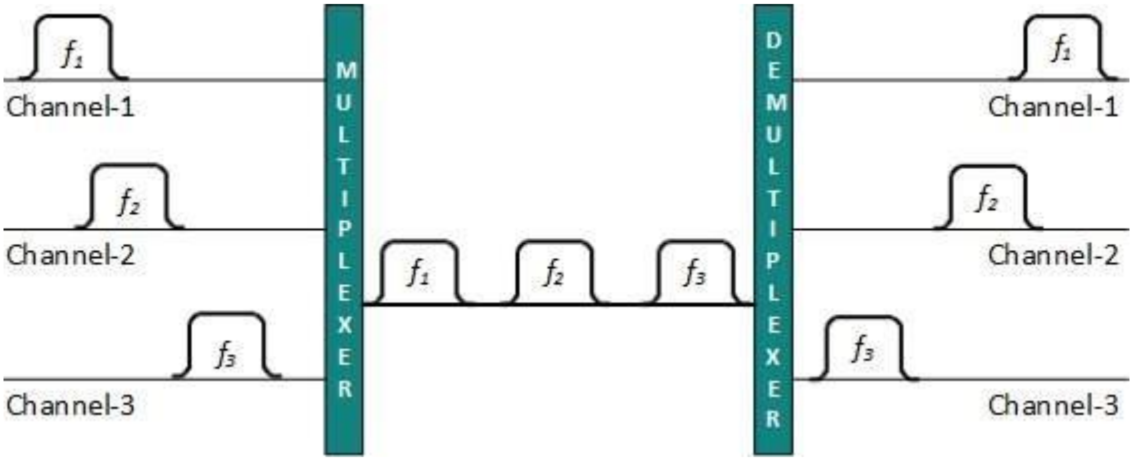
Multiplexing is a technique by which different analog and digital streams of transmission can be simultaneously processed over a shared link. Multiplexing divides the high capacity medium into low capacity logical medium which is then shared by different streams.

When multiple senders try to send over a single medium, a device called Multiplexer divides the physical channel and allocates one to each. On the other end of communication, a De-multiplexer receives data from a single medium, identifies each, and sends to different receivers.



Frequency Division Multiplexing

When the carrier is frequency, FDM is used. FDM is an analog technology. FDM divides the spectrum or carrier bandwidth in logical channels and allocates one user to each channel. Each user can use the channel frequency independently and has exclusive access of it. All channels are divided in such a way that they do not overlap with each other. Channels are separated by guard bands. Guard band is a frequency which is not used by either channel.



In the

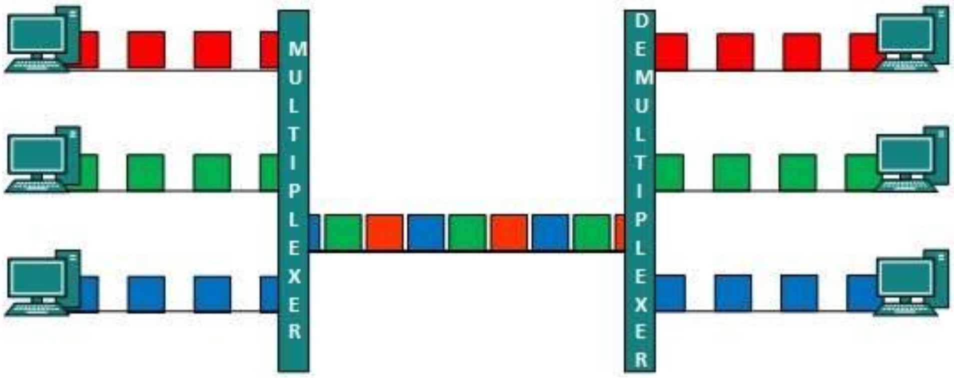
20th century, many telephone companies used frequency-division multiplexing for long distance connections to multiplex thousands of voice signals through a coaxial cable system.

Time Division Multiplexing

TDM is applied primarily on digital signals but can be applied on analog signals as well. In TDM the shared channel is divided among its user by means of time slot. Each user can transmit data within the provided time slot only. Digital signals are divided in frames, equivalent to time slot i.e. frame of an optimal size which can be transmitted in given time slot.

TDM works in synchronized mode. Both ends, i.e. Multiplexer and De-multiplexer are timely synchronized and both switch to next channel simultaneously.

When channel A transmits its frame at one end, the De-multiplexer provides media to channel A on the other end. As soon as the channel A's time slot expires, this side switches to channel B. On the other end, the De-multiplexer works in a synchronized manner and provides media to channel B. Signals from different channels travel the path in interleaved manner.



Synchronous Time Division Multiplexing

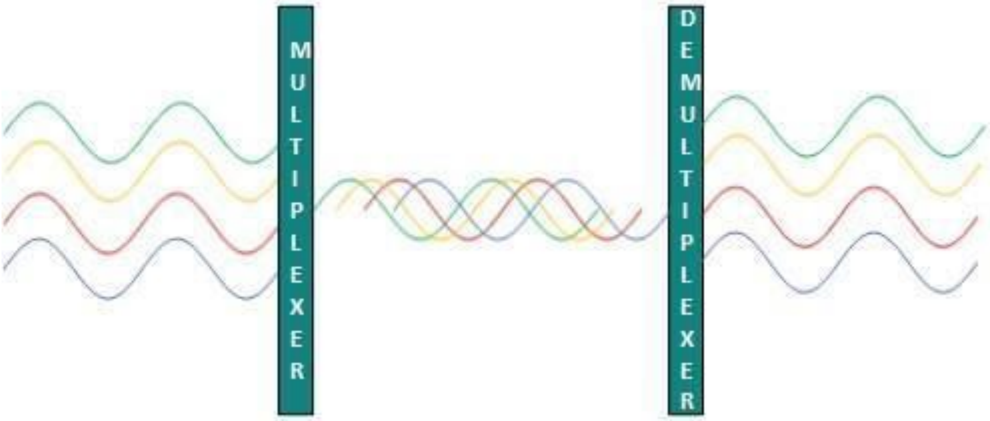
Synchronous time division multiplexing can be used for both analog and digital signals. In synchronous TDM, the connection of input is connected to a frame. If there are 'n' connections, then a frame is divided into 'n' time slots – and, for each unit, one slot is allocated – one for each input line. In this synchronous TDM sampling, the rate is same for all the signals, and this sampling requires a common clock signal at both the sender and receiver end. In synchronous TDM, the multiplexer allocates the same slot to each device at all times.

Asynchronous Time-Division Multiplexing

In asynchronous time-division multiplexing, the sampling rate is different for different signals, and it doesn't require a common clock. If the devices have nothing to transmit, then their time slot is allocated to another device. Designing of a commutator or de-commutator is difficult and the bandwidth is less for time-division multiplexing. This type of time-division multiplexing is used in asynchronous transfer mode networks.

Wavelength Division Multiplexing

Light has different wavelength (colors). In fiber optic mode, multiple optical carrier signals are multiplexed into an optical fiber by using different wavelengths. This is an analog multiplexing technique and is done conceptually in the same manner as FDM but uses light as signals.



Wavelength division multiplexing (WDM) is a technology in fiber optic communications; and, for the high capacity communication systems, wavelength division multiplexing is the most promising concept. This system uses multiplexer at transmitter to join signals and demultiplexer to split the signals apart, at the receiver end. The purpose of WDM is to combine multiple light sources into a single light source at the multiplexer; and, at the demultiplexer the single light is converted into multiple light sources.

Code Division Multiplexing

Multiple data signals can be transmitted over a single frequency by using Code Division Multiplexing. FDM divides the frequency in smaller channels but CDM allows its users to full bandwidth and transmit signals all the time using a unique code. CDM uses orthogonal codes to spread signals.

Each station is assigned with a unique code, called chip. Signals travel with these codes independently, inside the whole bandwidth. The receiver knows in advance the chip code signal it has to receive.

SWITCHING

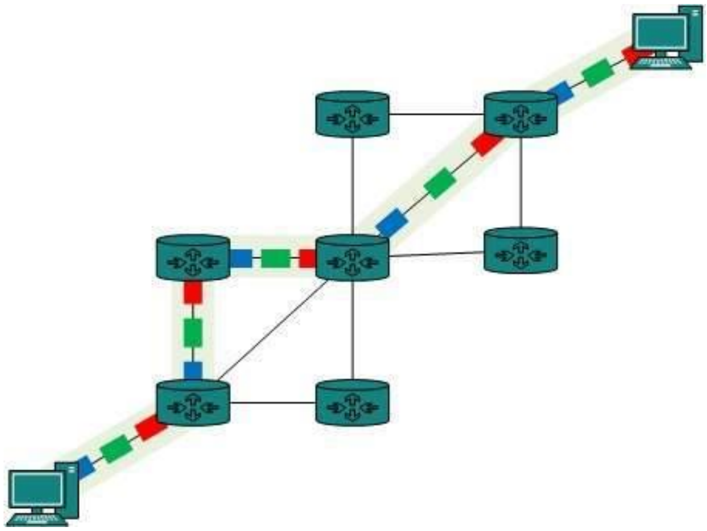
Switching is process to forward packets coming in from one port to a port leading towards the destination. When data comes on a port it is called ingress, and when data leaves a port or goes out it is called egress. A communication system may include number of switches and nodes. At broad level, switching can be divided into two major categories:

Circuit Switching

When two nodes communicate with each other over a dedicated communication path, it is called circuit switching. There 'is a need of pre-specified route from which data will travel and no other data is permitted. In circuit switching, to transfer the data, circuit must be established so that the data transfer can take place.

Circuits can be permanent or temporary. Applications which use circuit switching may have to go through three phases:

- Establish a circuit
- Transfer the data
- Disconnect the circuit

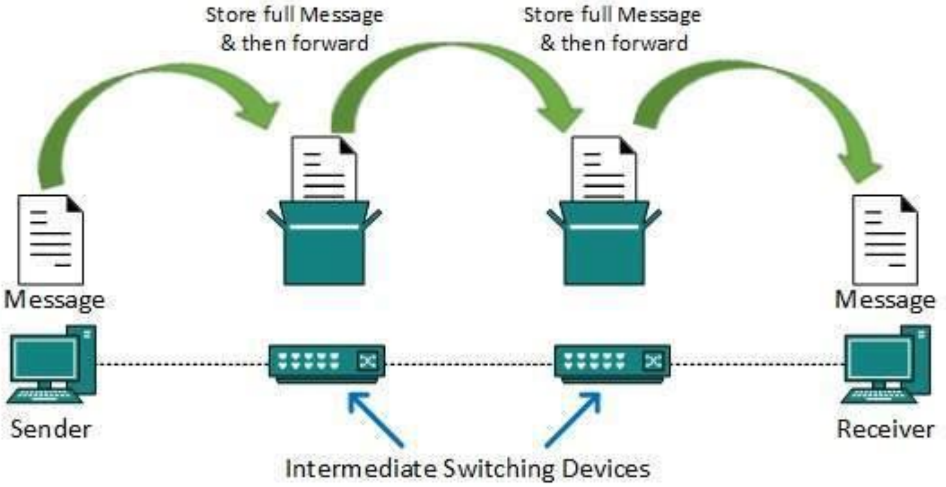


Circuit switching was designed for voice applications. Telephone is the best suitable example of circuit switching. Before a user can make a call, a virtual path between caller and callee is established over the network.

Message Switching

This technique was somewhere in middle of circuit switching and packet switching. In message switching, the whole message is treated as a data unit and is switching / transferred in its entirety.

A switch working on message switching, first receives the whole message and buffers it until there are resources available to transfer it to the next hop. If the next hop is not having enough resource to


Store full Message & then forward
Store full Message & then forward
Message
Message
Sender
Receiver
Intermediate Switching Devices

Accommodate large size message, the message is stored and switch waits.

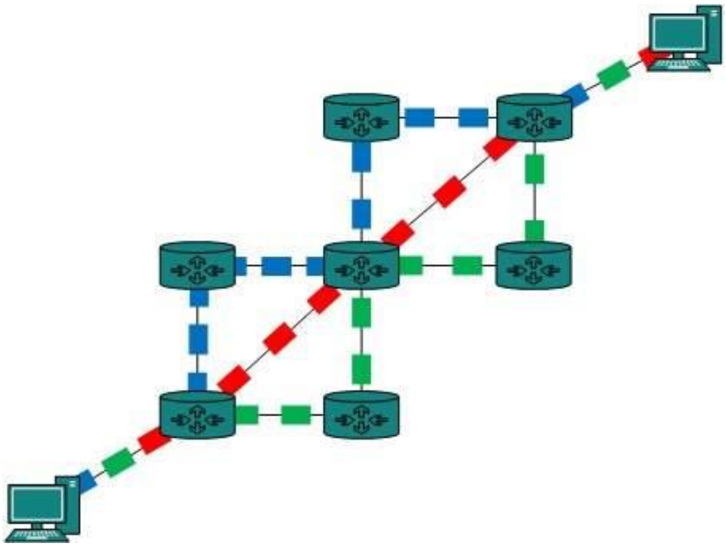
This technique was considered substitute to circuit switching. As in circuit switching the whole path is blocked for two entities only. Message switching is replaced by packet switching. Message switching has the following drawbacks:

- Every switch in transit path needs enough storage to accommodate entire message.
- Because of store-and-forward technique and waits included until resources are available, message switching is very slow.
- Message switching was not a solution for streaming media and real-time applications.

Packet Switching

Shortcomings of message switching gave birth to an idea of packet switching. The entire message is broken down into smaller chunks called packets. The switching information is added in the header of each packet and transmitted independently.

It is easier for intermediate networking devices to store small size packets and they do not take



Many resources either on carrier path or in the internal memory of switches.

Packet switching enhances line efficiency as packets from multiple applications can be multiplexed over the carrier. The internet uses packet switching technique. Packet switching enables the user to differentiate data streams based on priorities. Packets are stored and forwarded according to their priority to provide quality of service.

Virtual Circuit Switching

Virtual circuit (**VC**) is a means of transporting data over a switched computer in such a way that it appears as though there is a dedicated layer link between the source and destination end systems of this data. The term virtual circuit is synonymous with **virtual connection** and virtual channel. Before a connection or virtual circuit may be used, it has to be established, between two or more nodes or software applications, by configuring the relevant parts of the interconnecting network. After that, a bit stream or byte stream may be delivered between the nodes; hence, a virtual circuit protocol allows higher level protocols to avoid dealing with the division of data into segments, packets, or frames.

Virtual circuit communication resembles circuit switching, since both are connection oriented, meaning that in both cases data is delivered in correct order, and signaling overhead is required during a connection establishment phase. However, circuit switching provides a constant bit rate and latency, while these may vary in a virtual circuit service due to factors such as:

- varying packet queue lengths in the network nodes,
- varying bit rate generated by the application,
- varying load from other users sharing the same network resources by means of statistical multiplexing, etc.

Many virtual circuit protocols, but not all, provide reliable communication service through the use of data retransmissions because of error detection and automatic repeat request (ARQ).

Datagram packet switching

Datagram packet-switching is a packet switching technology by which each packet, now called a datagram, and is treated as a separate entity. Each packet is routed independently through the network. Therefore packets contain a header with the full information about the destination. The intermediate nodes examine the header of a packet and select an appropriate link to another node which is nearer to the destination. In this system, the packets do not follow a pre-established route, and the intermediate nodes do not require prior knowledge of the routes that will be used.

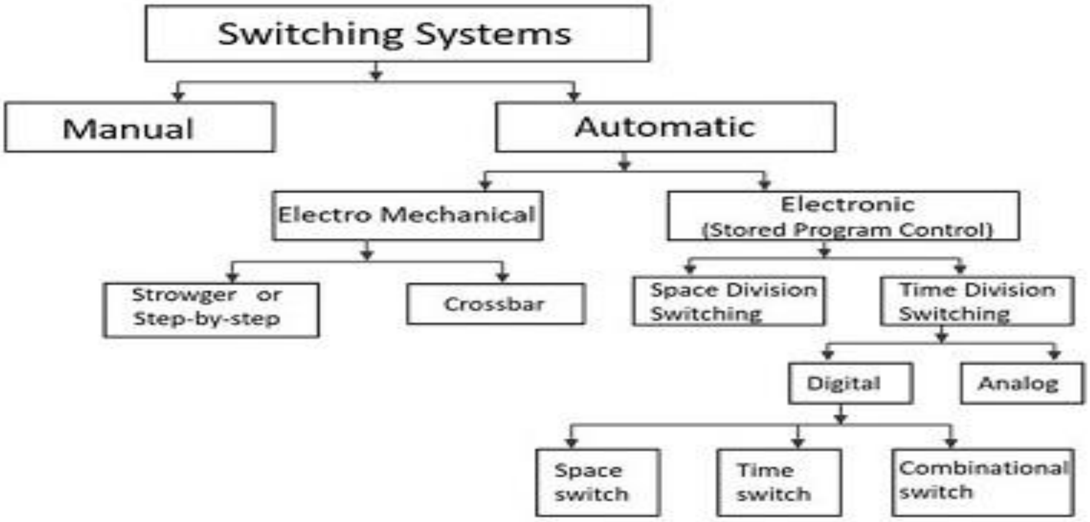
The individual packets which form a data stream may follow different paths between the source and the destination. As a result, the packets may arrive at the destination out of order. When this occurs, the packets will have to be reassembled to form the original message.

Because each packet is switched independently, there is no need for connection setup and no need to dedicate bandwidth in the form of a circuit.

Datagram packet switches use a variety of techniques to forward traffic; they are differentiated by how long it takes the packet to pass through the switch and their ability to filter out corrupted packets. A datagram network is a best effort network. Delivery is not guaranteed. Reliable delivery must be provided by the end systems (i.e. user's computers) using additional protocols.The most common datagram network is the Internet, which uses the IP network protocol.

TELECOMMUNICATION SWITCHING SYSTEM (Networking of Telephone Exchange)

In the early stages of telecommunication systems, the process and stages of switching, played an important to make or break connections. At the initial stages, the switching systems were operated manually. These systems were later automated. The following flowchart shows how the switching systems were classified. From the earliest days of the telephone, it was observed that it was more practical to connect different telephone instruments by running wires from each instrument to a central switching point, or telephone exchange, than it was to run wires between all the instruments. In 1878 the first telephone exchange was installed in New Haven, Connecticut, permitting up to 21 customers to reach one another by means of a manually operated central switchboard. The manual switchboard was quickly extended from 21 lines to hundreds of lines. Each line was terminated on the switchboard in a socket (called a jack), and a number of short, flexible circuits (called cords) with a plug on both ends of each cord were also provided. Two lines could thus be interconnected by inserting the two ends of a cord in the appropriate jacks.



The switching systems in the early stages were operated **manually**. The connections were made by the operators at the telephone exchanges in order to establish a connection. To minimize the disadvantages of manual operation, automatic switching systems were introduced.

The **Automatic** switching systems are classified as the following −

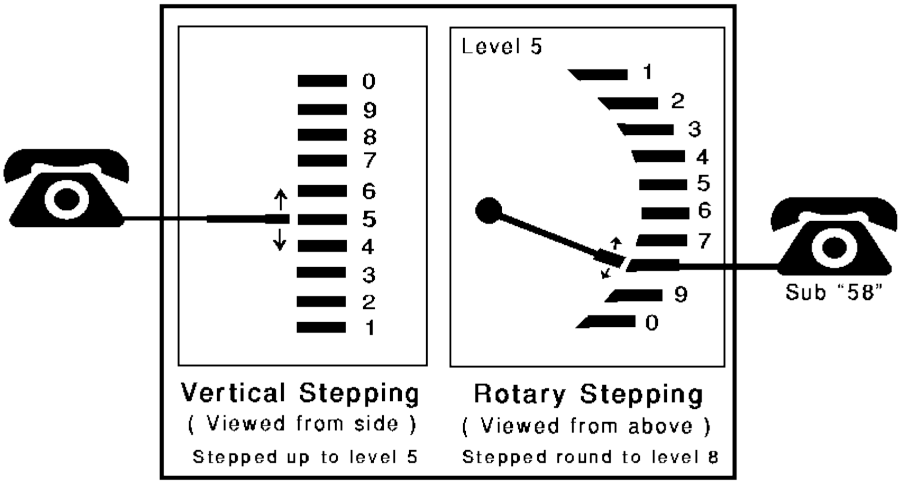
- **Electromechanical Switching Systems** − Here, mechanical switches are electrically operated.
- **Electronic Switching Systems** − Here, the usage of electronic components such as diodes, transistors and ICs are used for the switching purposes.

Electromechanical Switching Systems

The Electromechanical switching systems are a combination of mechanical and electrical switching types. The electrical circuits and the mechanical relays are deployed in them. The Electromechanical switching systems are further classified into the following.

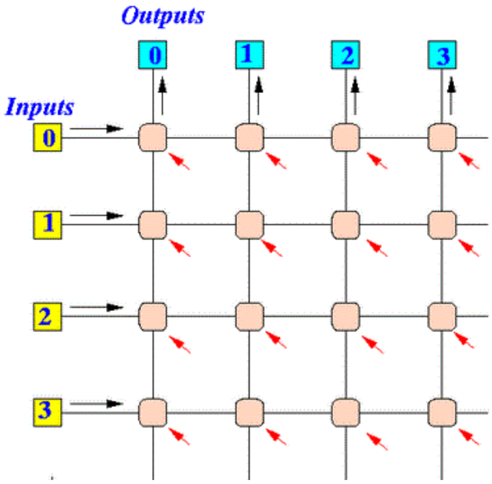
Step-by-step or Strowger

The **Step-by-step** switching system is also called the **Strowger** switching system after its inventor A B Strowger. The control functions in a Strowger system are performed by circuits associated with the switching elements in the system. The Strowger switch consisted of essentially two parts: an array of 100 terminals, called the bank, that were arranged 10 rows high and 10 columns wide in a cylindrical arc; and a movable switch, called the brush, which was moved up and down the cylinder by one ratchet mechanism and rotated around the arc by another, so that it could be brought to the position of any of the 100 terminals. The stepping movement was controlled directly by pulses from the telephone instrument. In the original systems, the caller generated the pulses by rapidly pushing a button switch on the instrument.



Crossbar

The **Crossbar** switching systems have hard-wired control subsystems which use relays and latches. These subsystems have limited capability and it is virtually impossible to modify them to provide additional functionalities. A **crossbar switch** (**cross-point switch**, **matrix switch**) is a collection of switches arranged in a matrix configuration. A crossbar switch has multiple input and output lines that form a crossed pattern of interconnecting lines between which a connection may be established by closing a switch located at each intersection, the elements of the matrix. Originally, a crossbar switch consisted literally of crossing metal bars that provided the input and output paths. Later implementations achieved the same switching topology in solid state semiconductor chips.

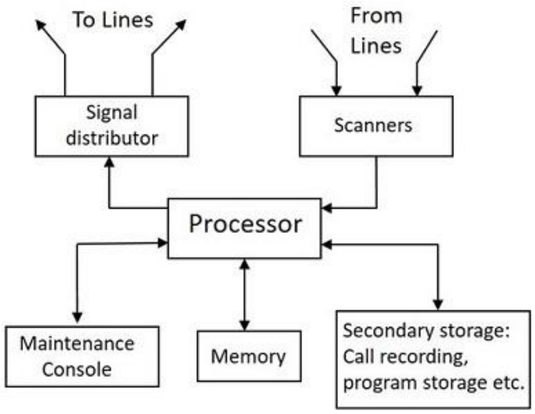


Electronic Switching Systems

The Electronic Switching systems are operated with the help of a processor or a computer which control the switching timings. The instructions are programmed and stored on a processor or computer that controls the operations. This method of storing the programs on a processor or computer is called the **Stored Program Control (SPC)** technology. New facilities can be added to a **SPC** system by changing the control program.

There are two types of SPCs −

- Centralized SPC
- Distributed SPC



It permits the features like abbreviated dialing, call forwarding, call waiting, etc. The Stored Program Control concept is where a program or a set of instructions to the computer is stored in its memory and the instructions are executed automatically one by one by the processor.

Dual processor architecture may be configured to operate in three modes like −

- Standby Mode
- Synchronous Duplex Mode
- Load Sharing Mode

Standby Mode

As the name implies, in the two processors present, one processor is active and the other is in the standby mode. The processor in the standby mode is used as a backup, in case the active one fails.

Synchronous Duplex Mode

In the Synchronous Duplex mode, two processors are connected and operated in synchronism. Two processors P1 and P2 are connected and separate memories like M1 and M2 are used. These processors are coupled to exchange the stored data. A Comparator is used in between these two processors. The Comparator helps in comparing the results.

Load Sharing Mode

Load sharing mode is where a task is shared between two processors. The Exclusion Device (ED) is used instead of the comparator in this mode. The processors call for ED to share the resources, so that both the processors do not seek the same resource at the same time.In this mode, both the processors are simultaneously active. These processors share the resources of the exchange and load. In case one of the processor fails, the other one takes over the entire load of the exchange with the help of ED. Under normal operation, each processor handles one-half of the calls on a statistical basis. The exchange operator can however vary the processor load for maintenance purpose.

Space Division Switching or Time Division Switching

The switching scheme used by the electronic switching systems may be either **Space Division Switching or Time Division Switching.** In space division switching, a dedicated path is established between the calling and the called subscribers for the entire duration of the call. In time division switching, sampled values of speech signals are transferred at fixed intervals.

Analog or Digital

The time division switching may be analog or digital. In analog switching, the sampled voltage levels are transmitted as they are. However, in binary switching, they are binary coded and transmitted.

Space Switching and Time Switching

If the coded values are transferred during the same time interval from input to output, the technique is called **Space Switching**. If the values are stored and transferred to the output at a time interval, the technique is called **Time Switching**. A time division digital switch may also be designed by using a combination of space and time switching techniques.

T1 AND E1 HIERARCHY

T1 and E1 are equivalent digital data transmission formats that carry signals. T1 and E1 lines can be interconnected for international use.

This topic contains the following sections:

T1 Overview

T1 is a digital data transmission medium capable of handling 24 simultaneous connections running at a combined 1.544 Mbps. T1 combines these 24 separate connections, called channels or time slots, onto a single link. T1 is also called DS1.

The T1 data stream is broken into frames. Each frame consists of a single framing bit and 24 8-bit channels, totaling 193 bits per T1 frame. Frames are transmitted 8,000 times per second, at a data transmission rate of 1.544 Mbps (8,000 x 193 = 1.544 Mbps).

As each frame is received and processed, the data in each 8-bit channel is maintained with the channel data from previous frames, enabling T1 traffic to be separated into 24 separate flows across a single medium. For example, in the following set of 4-channel frames (without a framing bit), the data in channel 1 consists of the first octet of each frame, the data in channel 2 consists of the second octet of each frame, and so on:

E1 Overview

E1 is the European format for DS1 digital transmission. E1 links are similar to T1 links except that they carry signals at 2.048 Mbps. Each signal has 32 channels, and each channel transmits at 64 Kbps. E1 links have higher bandwidth than T1 links because it does not reserve one bit for overhead. Whereas, T1 links use 1 bit in each channel for overhead.

T1 and E1 Signals

T1 and E1 interfaces consist of two pairs of wires—a transmit data pair and a receive data pair. Clock signals, which determine when the transmitted data is sampled, are embedded in the T1 and E1 transmissions.

Typical digital signals operate by sending either zeros (0s) or ones (1s), which are usually represented by the absence or presence of a voltage on the line. The receiving device need only detect the presence of the voltage on the line at the particular sampling edge to determine whether the signal is 0 or 1. T1 and E1, however, use bipolar electrical pulses. Signals are represented by no voltage (0), positive voltage (1), or negative voltage (1). The bipolar signal allows T1 and E1 receivers to detect error conditions in the line, depending on the type of encoding that is being used.

Encoding

The following are common T1 and E1 encoding techniques:

- Alternate mark inversion (AMI)—T1 and E1
- Bipolar with 8-zero substitution (B8ZS)—T1 only
- High-density bipolar 3 code (HDB3)—E1 only

Common Characteristics:-

- Both are having Same Sampling Frequency i.e. 8kHz.
- In both (E1 & T1) Number of samples/telephone signal = 8000/sec.

- In both (E1 & T1) Length of PCM Frame = 1/8000s = 125µs.
- In both (E1 & T1) Number of Bits in each code word = 8.
- In both (E1 & T1) Telephone Channel Bit Rate = 8000/s x 8 Bit = 64 kbit/s.

Differing Characteristics:-

- In E1 Encoding/Decoding is followed by A-Law while in T1 Encoding/Decoding is followed by µ-Law.
- In E1 – 13 Number of Segments in Characteristics while in T1 – 15 Number of Segments in Characteristics.
- In E1 – 32 Number of Timeslots / PCM Frame while in T1 – 24 Number of Timeslots / PCM Frame.
- In E1 – 8 x 32 = 256 number of bits / PCM Frame while in T1 – 8 x 24 + 1* = 193 number of bits / PCM Frame. (* Signifies an additional bit).
- In E1 – (125µs x 8)/256 = approx 3.9µs is the length of an 8-bit Timeslot while in T1 – (125µs x 8)/193 = approx 5.2µs is the length of an 8-bit Timeslot.
- In E1 – 8000/s x 256 bits = 2048kbit/s is the Bit Rate of Time-Division Multiplexed Signal while in T1 – 8000/s x 193 bits = 1544kbit/s is the Bit Rate of Time-Division Multiplexed Signal.

INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

These are a set of communication standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network. Before Integrated Services Digital Network (ISDN), the telephone system was seen as a way to transmit voice, with some special services available for data. The main feature of ISDN is that it can integrate speech and data on the same lines, which were not available in the classic telephone system.

ISDN is a circuit-switched telephone network system, but it also provides access to packet switched networks that allows digital transmission of voice and data. This results in potentially better voice or data quality than an analog phone can provide. It provides a packet-switched connection for data in increments of 64 kilobit/s. It provided a maximum of 128 kbit/s bandwidth in both upstream and downstream directions. A greater data rate was achieved through channel bonding. Generally ISDN B-channels of three or four BRIs (six to eight 64 kbit/s channels) are bonded.

In the context of the OSI model, ISDN is employed as the network in data-link and physical layers but commonly ISDN is often limited to usage to Q.931 and related protocols. These protocols introduced in 1986 are a set of signaling protocols establishing and breaking circuit-switched connections, and for advanced calling features for the user. ISDN provides simultaneous voice, video, and text transmission between individual desktop videoconferencing systems and group videoconferencing systems.

ISDN Interfaces:

The following are the interfaces of ISDN:

Basic Rate Interface (BRI) –

There are two data-bearing channels (‘B’ channels) and one signaling channel (‘D’ channel) in BRI to initiate connections. The B channels operate at a maximum of 64 Kbps while the D channel operates at a maximum of 16 Kbps. The two channels are independent of each other. For example, one channel is used as a TCP/IP connection to a location while the other channel is used to send a fax to a remote location. In iSeries ISDN supports basic rate interface (BRl).

The basic rate interface (BRl) specifies a digital pipe consisting two B channels of 64 Kbps each and one D channel of 16 Kbps. This equals a speed of 144 Kbps. In addition, the BRl service itself requires an operating overhead of 48 Kbps. Therefore a digital pipe of 192 Kbps is required.

Primary Rate Interface (PRI) –

Primary Rate Interface service consists of a D channel and either 23 or 30 B channels depending on the country you are in. PRI is not supported on the iSeries. A digital pipe with 23 B channels and one 64 Kbps D channel is present in the usual Primary Rate Interface (PRI). Twenty-three B channels of 64 Kbps each and one D channel of 64 Kbps equals 1.536 Mbps. The PRI service uses 8 Kbps of overhead also. Therefore PRI requires a digital pipe of 1.544 Mbps.

Broadband-ISDN (B-ISDN) –

Narrowband ISDN has been designed to operate over the current communications infrastructure, which is heavily dependent on the copper cable however B-ISDN relies mainly on the evolution of fiber optics. According to CCITT B-ISDN is best described as ‘a service requiring transmission channels capable of supporting rates greater than the primary rate.

ISDN Services:

ISDN provides a fully integrated digital service to users. These services fall into 3 categories- bearer services, teleservices and supplementary services.

Bearer Services –

Transfer of information (voice, data and video) between users without the network manipulating the content of that information is provided by the bearer network. There is no need for the network to process the information and therefore does not change the content. Bearer services belong to the first three layers of the OSI model. They are well defined in the ISDN standard. They can be provided using circuit-switched, packet-switched, frame-switched, or cell-switched networks.

Teleservices –

In this the network may change or process the contents of the data. These services corresponds to layers 4-7 of the OSI model. Teleservices relay on the facilities of the bearer services and are designed to accommodate complex user needs. The user needs not to be aware of the details of the process. Teleservices include telephony, teletex, telefax, videotex, telex and teleconferencing. Though the ISDN defines these services by name yet they have not yet become standards.

Supplementary Service –

Additional functionality to the bearer services and teleservices are provided by supplementary services. Reverse charging, call waiting, and message handling are examples of supplementary services which are all familiar with today's telephone company services.

Principle of ISDN:

The ISDN works based on the standards defined by ITU-T (formerly CCITT). The Telecommunication Standardization Sector (ITU-T) coordinates standards for telecommunications on behalf of the International Telecommunication Union (ITU) and is based in Geneva, Switzerland. The various principles of ISDN as per ITU-T recommendation are:

To support switched and non-switched applications

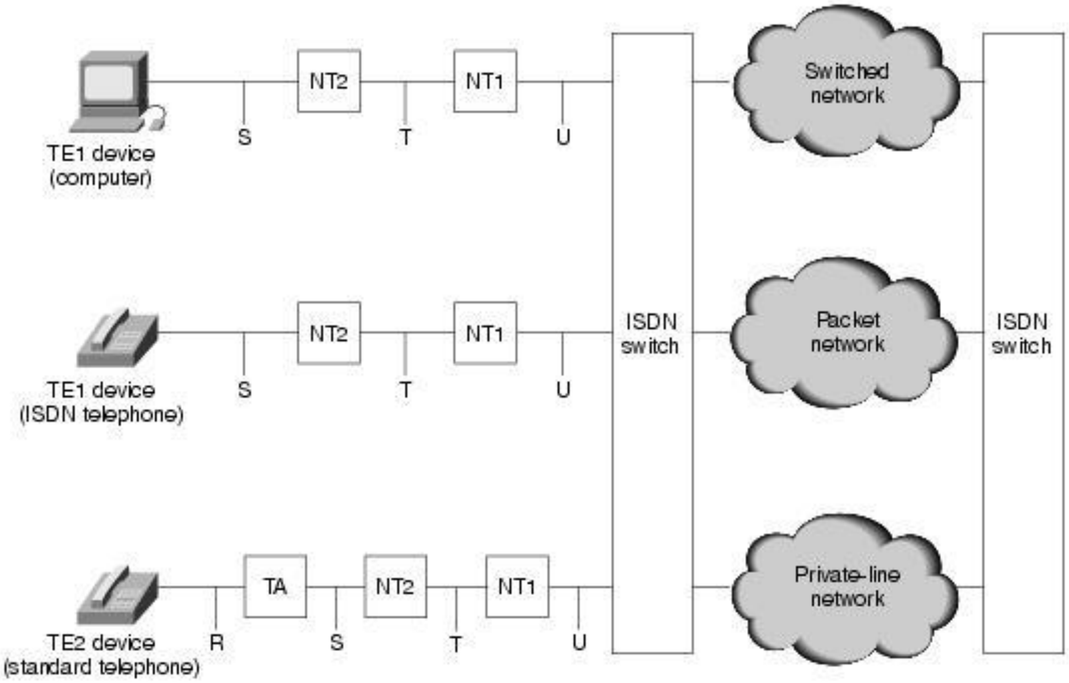
To support voice and non-voice applications

Reliance on 64-kbps connections

Intelligence in the network

Layered protocol architecture

Variety of configurations


NT2
NT1
TE1 device
(computer)
S
T
U
Switched
network
NT2
NT1
ISDN
switch
Packet
network
ISDN
switch
TE1 device
(ISDN telephone)
S
T
U
TA
NT2
NT1
Private-line
network
TE2 device
(standard telephone)
R
S
T
U

CHAPTER 3- DATA LINK LAYER

Data Link Layer is second layer of OSI Layered Model. This layer is one of the most complicated layers and has complex functionalities and liabilities. Data link layer hides the details of underlying hardware and represents itself to upper layer as the medium to communicate.

Data link layer works between two hosts which are directly connected in some sense. This direct connection could be point to point or broadcast. Systems on broadcast network are said to be on same link. The work of data link layer tends to get more complex when it is dealing with multiple hosts on single collision domain.

Data link layer is responsible for converting data stream to signals bit by bit and to send that over the underlying hardware. At the receiving end, Data link layer picks up data from hardware which are in the form of electrical signals assembles them in a recognizable frame format, and hands over to upper layer.

Data link layer has two sub-layers:

- **Logical Link Control:** It deals with protocols, flow-control, and error control
- **Media Access Control:** It deals with actual control of media

FUNCTIONALITY OF DATA-LINK LAYER

Data link layer does many tasks on behalf of upper layer. These are:

- **Framing** Data-link layer takes packets from Network Layer and encapsulates them into Frames. Then, it sends each frame bit-by-bit on the hardware. At receiver' end, data link layer picks up signals from hardware and assembles them into frames.
- **Addressing** Data-link layer provides layer-2 hardware addressing mechanism. Hardware address is assumed to be unique on the link. It is encoded into hardware at the time of manufacturing.
- **Synchronization** When data frames are sent on the link, both machines must be synchronized in order to transfer to take place.
- **Error Control**

Sometimes signals may have encountered problem in transition and the bits are flipped. These errors are detected and attempted to recover actual data bits. It also provides error reporting mechanism to the sender.

- **Flow Control** Stations on same link may have different speed or capacity. Data-link layer ensures flow control that enables both machines to exchange data on same speed.

- **Multi-Access** When host on the shared link tries to transfer the data, it has a high probability of collision. Data-link layer provides mechanism such as CSMA/CD to equip capability of accessing a shared media among multiple Systems.

- Reliable delivery. When a link-layer protocol provides reliable delivery service, it guarantees to move each network-layer datagram across the link without error.

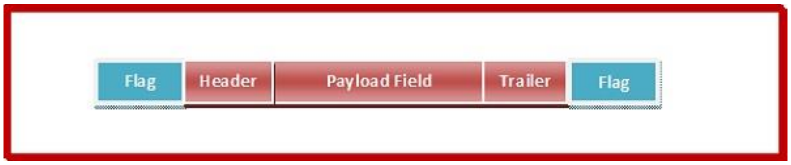
FRAMING

Framing is a point-to-point connection between two computers or devices consists of a wire in which data is transmitted as a stream of bits. However, these bits must be framed into discernible blocks of information. Framing is a function of the data link layer. It provides a way for a sender to transmit a set of bits that are meaningful to the receiver.

Parts of a Frame

A frame has the following parts −

- Frame Header − It contains the source and the destination addresses of the frame.
- Payload field − It contains the message to be delivered.
- Trailer − It contains the error detection and error correction bits.
- Flag − It marks the beginning and end of the frame.



METHODS OF FRAMING

- Encoding Violations
- Bit stuffing
- Flag byte with Byte Stuffing

- Character Count

Bit stuffing:

- Allows frame to contain arbitrary number of bits and arbitrary character size. The frames are separated by separating flag.
- Each frame begins and ends with a special bit pattern, 01111110 called a flag byte. When five consecutive l's are encountered in the data, it automatically stuffs a '0' bit into outgoing bit stream.
- In this method, frames contain an arbitrary number of bits and allow character codes with an arbitrary number of bits per character. In his case, each frame starts and ends with a special bit pattern, 01111110.
- In the data a 0 bit is automatically stuffed into the outgoing bit stream whenever the sender's data link layer finds five consecutive 1s.
- This bit stuffing is similar to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data.
- When the receiver sees five consecutive incoming i bits, followed by a o bit, it automatically de stuffs (i.e., deletes) the 0 bit. Bit Stuffing is completely transparent to network layer as byte stuffing. The figure1 below gives an example of bit stuffing.
- This method of framing finds its application in networks in which the change of data into code on the physical medium contains some repeated or duplicate data. For example, some LANs encodes bit of data by using 2 physical bits.

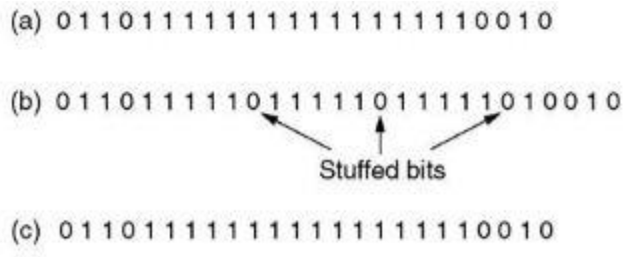

(a) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0
(b) 0 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 0 0 1 0
Stuffed bits
(c) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0

Fig1: Bit stuffing

Byte stuffing:

- In this method, start and end of frame are recognized with the help of flag bytes. Each frames starts with and ends with a flag byte. Two consecutive flag bytes indicate the end of one frame and start of the next one. The flag bytes used in the figure 2 used is named as "ESC" flag byte.
- A frame delimited by flag bytes. This framing method is only applicable in 8-bit character codes which are a major disadvantage of this method as not all character codes use 8-bit characters e.g. Unicode.
- Four example of byte sequences before and after stuffing:

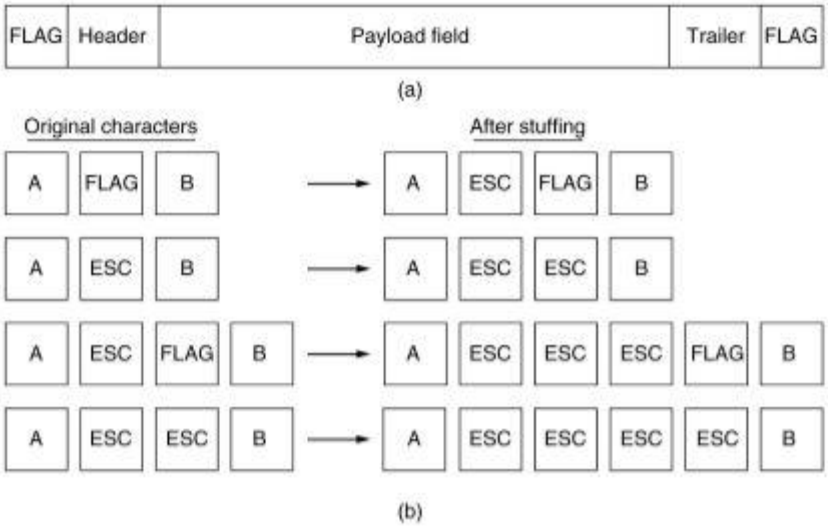
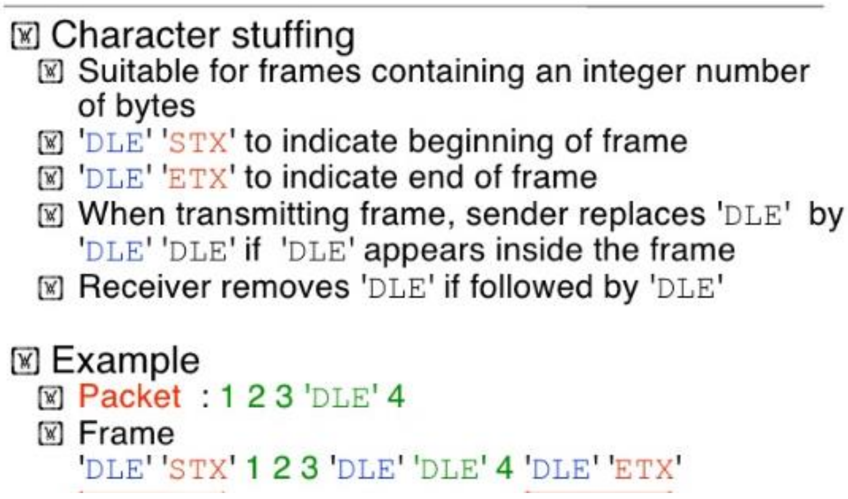


Fig2: Framing with Byte stuffing

Character Count

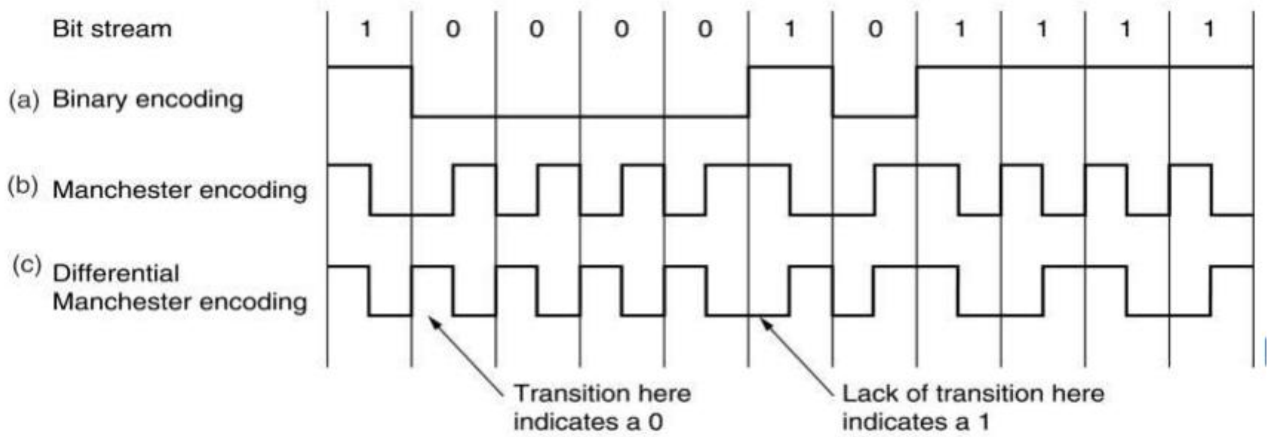
Each frame starts with the ASCII character sequence DLE STX and ends with the sequence DLE ETX.(where DLE is Data Link Escape, STX is Start of Text and ETX is End of Text.) This method overcomes the drawbacks of the character count method. If the destination ever loses synchronization, it only has to look for DLE STX and DLE ETX characters. If however, binary data is being transmitted then there exists a possibility of the characters DLE STX and DLE ETX occurring in the data. Since this can interfere with the framing, a technique called character stuffing is used. The sender's data link layer inserts an ASCII DLE character just before the DLE character in the data. The receiver's data link layer removes this DLE before this data is given to the network layer. However character stuffing is closely associated with 8-bit characters and this is a major hurdle in transmitting arbitrary sized characters.



- Character stuffing
  - Suitable for frames containing an integer number of bytes
  - 'DLE' 'STX' to indicate beginning of frame
  - 'DLE' 'ETX' to indicate end of frame
  - When transmitting frame, sender replaces 'DLE' by 'DLE' 'DLE' if 'DLE' appears inside the frame
  - Receiver removes 'DLE' if followed by 'DLE'
- Example
  - Packet : 1 2 3 'DLE' 4
  - Frame
    'DLE' 'STX' 1 2 3 'DLE' 'DLE' 4 'DLE' 'ETX'

Physical Layer Coding Violations

This Framing Method is used only in those networks in which Encoding on the Physical Medium contains some redundancy. Some LANs encode each bit of data by using two Physical Bits′ i.e. Manchester coding is Used. Here, Bit 1 is encoded into high-low(10) pair and Bit 0 is encoded into low-high(01) pair. The scheme means that every data bit has a transition in the middle,′ making it easy for the receiver to locate the bit boundaries. The combinations high-high and low-low are not used for data but are used for delimiting frames in some protocols.


Bit stream
1
0
0
0
0
1
0
1
1
1
1
(a) Binary encoding
(b) Manchester encoding
(c) Differential Manchester encoding
Transition here indicates a 0
Lack of transition here indicates a 1

NUMERICAL

Suppose the following bit string is received by the data link layer from the network layer: 01110111101111101111110. What is the resulting string after bit stuffing? Bold each bit that has been added.

Answer:

011101111011111**0**011111**0**10

3.3. ERROR DETECTION AND CORRECTIONS

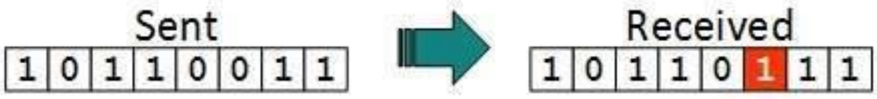
There are many reasons such as noise, cross-talk etc., which may help data to get corrupted during transmission. The upper layers work on some generalized view of network architecture and are not aware of actual hardware data processing. Hence, the upper layers expect error-free transmission between the systems. Most of the applications would not function expectedly if they receive erroneous data. Applications such as voice and video may not be that affected and with some errors they may still function well.

Data-link layer uses some error control mechanism to ensure that frames (data bit streams) are transmitted with certain level of accuracy. But to understand how errors is controlled, it is essential to know what types of errors may occur.

Types of Errors

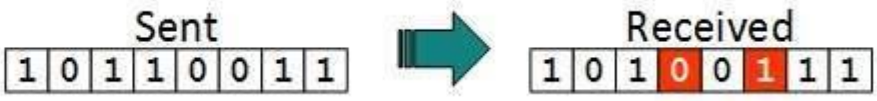
There may be three types of errors:

- **Single bit error**



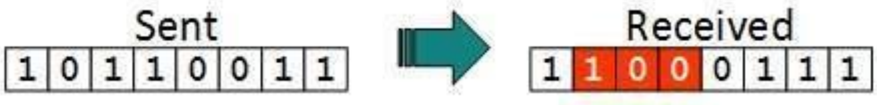
In a frame, there is only one bit, anywhere though, which is corrupt.

- **Multiple bits error**



Frame is received with more than one bits in corrupted state.

- **Burst error**



Frame contains more than1 consecutive bits corrupted.

Error control mechanism may involve two possible ways:

- Error detection
- Error correction

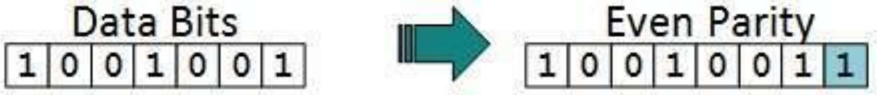
Error Detection

Errors in the received frames are detected by means of Parity Check and Cyclic Redundancy Check (CRC). In both cases, few extra bits are sent along with actual data to confirm that bits received at other end are same as they were sent. If the counter-check at receiver' end fails, the bits are considered corrupted.

Parity Check

One extra bit is sent along with the original bits to make number of 1s either even in case of even parity, or odd in case of odd parity.

The sender while creating a frame counts the number of 1s in it. For example, if even parity is used and number of 1s is even then one bit with value 0 is added. This way number of 1s remains even. If the number of 1s is odd, to make it even a bit with value 1 is added.

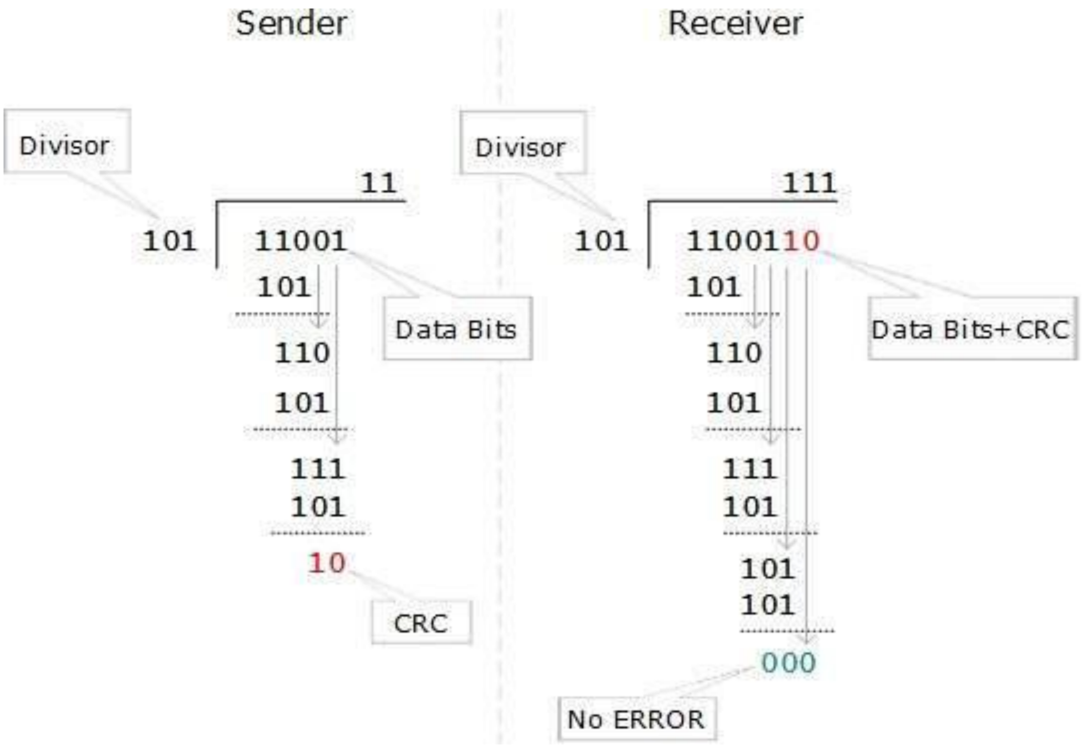


The receiver simply counts the number of 1s in a frame. If the count of 1s is even and even parity is used, the frame is considered to be not-corrupted and is accepted. If the count of 1s is odd and odd parity is used, the frame is still not corrupted.

If a single bit flips in transit, the receiver can detect it by counting the number of 1s. But when more than one bits are erroneous, then it is very hard for the receiver to detect the error.

Cyclic Redundancy Check (CRC)

CRC is a different approach to detect if the received frame contains valid data. This technique involves binary division of the data bits being sent. The divisor is generated using polynomials. The sender performs a division operation on the bits being sent and calculates the remainder. Before sending the actual bits, the sender adds the remainder at the end of the actual bits. Actual data bits plus the remainder is called a codeword. The sender transmits data bits as codeword.



At the other end, the receiver performs division operation on codewords using the same CRC divisor. If the remainder contains all zeros the data bits are accepted, otherwise it is considered as there some data corruption occurred in transit.

Error Correction

In the digital world, error correction can be done in two ways:

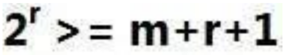
- **Backward Error Correction** When the receiver detects an error in the data received, it requests back the sender to retransmit the data unit.

- **Forward Error Correction** When the receiver detects some error in the data received, it executes error-correcting code, which helps it to auto-recover and to correct some kinds of errors.

The first one, Backward Error Correction, is simple and can only be efficiently used where retransmitting is not expensive. For example, fiber optics. But in case of wireless transmission retransmitting may cost too much. In the latter case, Forward Error Correction is used.

To correct the error in data frame, the receiver must know exactly which bit in the frame is corrupted. To locate the bit in error, redundant bits are used as parity bits for error detection. For example, we take ASCII words (7 bits data), then there could be 8 kind of information we need: first seven bits to tell us which bit is error and one more bit to tell that there is no error.

For m data bits, r redundant bits are used. r bits can provide 2r combinations of information. In m+r bit codeword, there is possibility that the r bits themselves may get corrupted. So the number of r bits used must inform about m+r bit locations plus no-error information, i.e. m+r+1.



$$2^r >= m+r+1$$

Hamming Code

Hamming code is a set of error-correction codes that can be used to **detect and correct the errors** that can occur when the data is moved or stored from the sender to the receiver. It is **technique developed by R.W. Hamming for error correction**.

Redundant bits –

Redundant bits are extra binary bits that are generated and added to the information-carrying bits of data transfer to ensure that no bits were lost during the data transfer. The number of redundant bits can be calculated using the following formula:

2^r ≥ m + r + 1

where, r = redundant bit, m = data bit

Suppose the number of data bits is 7, then the number of redundant bits can be calculated using:
= 2^4 ≥ 7 + 4 + 1
Thus, the number of redundant bits= 4

Parity bits –

A parity bit is a bit appended to a data of binary bits to ensure that the total number of 1's in the data are even or odd. Parity bits are used for error detection. There are two types of parity bits:

1. **Even parity bit:**
   In the case of even parity, for a given set of bits, the number of 1's are counted. If that count is odd, the parity bit value is set to 1, making the total count of occurrences of 1's an even number. If the total number of 1's in a given set of bits is already even, the parity bit's value is 0.
2. **Odd Parity bit –**
   In the case of odd parity, for a given set of bits, the number of 1's are counted. If that count is even, the parity bit value is set to 1, making the total count of occurrences of 1's an odd number. If the total number of 1's in a given set of bits is already odd, the parity bit's value is 0.

General Algorithm of Hamming code –

The Hamming Code is simply the use of extra parity bits to allow the identification of an error.

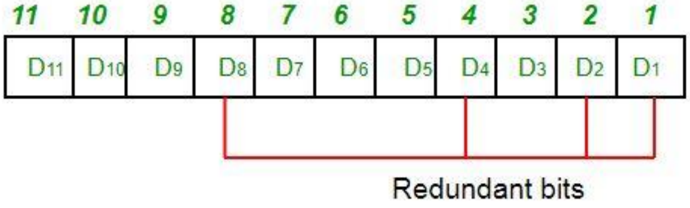
1. Write the bit positions starting from 1 in binary form (1, 10, 11, 100, etc).
2. All the bit positions that are a power of 2 are marked as parity bits (1, 2, 4, 8, etc).

3. All the other bit positions are marked as data bits.
4. Each data bit is included in a unique set of parity bits, as determined its bit position in binary form.
   a. Parity bit 1 covers all the bits positions whose binary representation includes a 1 in the least significant position (1, 3, 5, 7, 9, 11, etc).
   b. Parity bit 2 covers all the bits positions whose binary representation includes a 1 in the second position from the least significant bit (2, 3, 6, 7, 10, 11, etc).
   c. Parity bit 4 covers all the bits positions whose binary representation includes a 1 in the third position from the least significant bit (4–7, 12–15, 20–23, etc).
   d. Parity bit 8 covers all the bits positions whose binary representation includes a 1 in the fourth position from the least significant bit bits (8–15, 24–31, 40–47, etc).
   e. In general each parity bit covers all bits where the bitwise AND of the parity position and the bit position is non-zero.
5. Since we check for even parity set a parity bit to 1 if the total number of ones in the positions it checks is odd.
6. Set a parity bit to 0 if the total number of ones in the positions it checks is even.

Determining the position of redundant bits –

These redundancy bits are placed at the positions which correspond to the power of 2.
As in the above example:

1. The number of data bits = 7
2. The number of redundant bits = 4
3. The total number of bits = 11
4. The redundant bits are placed at positions corresponding to power of 2- 1, 2, 4, and 8

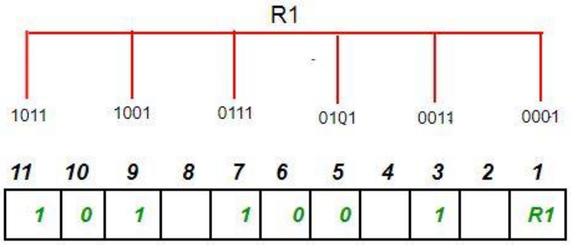


Suppose the data to be transmitted is 1011001, the bits will be placed as follows:

11	10	9	8	7	6	5	4	3	2	1
1	0	1	R8	1	0	0	R4	1	R2	R1

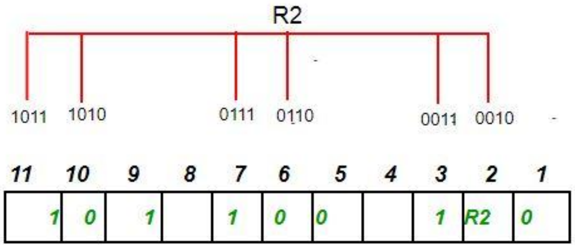
Determining the Parity bits –

1. R1 bit is calculated using parity check at all the bits positions whose binary representation includes a 1 in the least significant position.
   R1: bits 1, 3, 5, 7, 9, 11



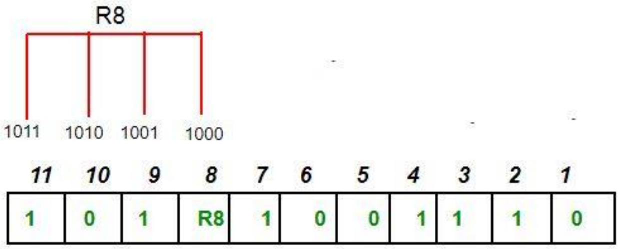
11	10	9	8	7	6	5	4	3	2	1
1	0	1		1	0	0		1		R1

1. To find the redundant bit R1, we check for even parity. Since the total number of 1's in all the bit positions corresponding to R1 is an even number the value of R1 (parity bit's value) = 0
2. R2 bit is calculated using parity check at all the bits positions whose binary representation includes a 1 in the second position from the least significant bit.
   R2: bits 2,3,6,7,10,11



11	10	9	8	7	6	5	4	3	2	1
1	0	1		1	0	0		1	R2	0

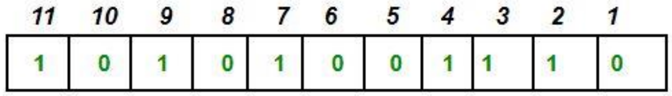
1. To find the redundant bit R4, we check for even parity. Since the total number of 1's in all the bit positions corresponding to R4 is an odd number the value of R4(parity bit's value) = 1
2. R8 bit is calculated using parity check at all the bits positions whose binary representation includes a 1 in the fourth position from the least significant bit.
   R8: bit 8,9,10,11



11	10	9	8	7	6	5	4	3	2	1
1	0	1	R8	1	0	0	1	1	1	0

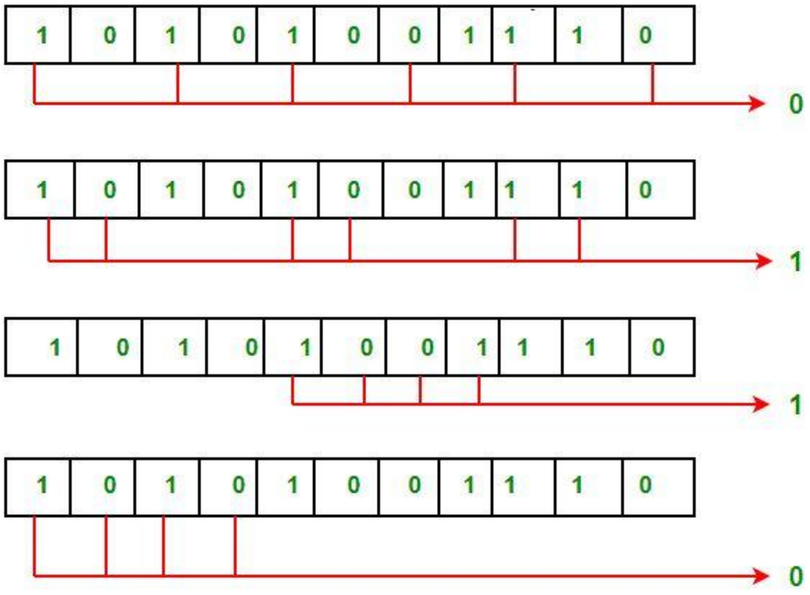
To find the redundant bit R8, we check for even parity. Since the total number of 1's in all the bit positions corresponding to R8 is an even number the value of R8(parity bit's value)=0.

Thus, the data transferred is:



11	10	9	8	7	6	5	4	3	2	1
1	0	1	0	1	0	0	1	1	1	0

Error detection and correction –
Suppose in the above example the 6th bit is changed from 0 to 1 during data transmission, then it gives new parity values in the binary number:



The bits give the binary number as 0110 whose decimal representation is 6. Thus, the bit 6 contains an error. To correct the error the 6th bit is changed from 1 to 0.

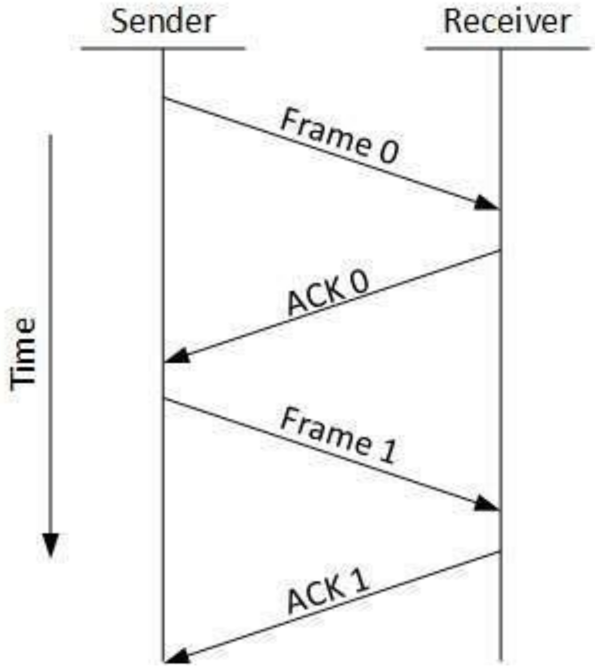
3.4. FLOW CONTROL

When a data frame (Layer-2 data) is sent from one host to another over a single medium, it is required that the sender and receiver should work at the same speed. That is, sender sends at a speed on which the receiver can process and accept the data. What if the speed (hardware/software) of the sender or receiver differs? If sender is sending too fast the receiver may be overloaded, (swamped) and data may be lost.

Two types of mechanisms can be deployed to control the flow:

- **Stop and Wait**

This flow control mechanism forces the sender after transmitting a data frame to stop and wait until the acknowledgement of the data-frame sent is received.



- **Sliding Window**

  In this flow control mechanism, both sender and receiver agree on the number of data-frames after which the acknowledgement should be sent. As we learnt, stop and wait flow control mechanism wastes resources, this protocol tries to make use of underlying resources as much as possible.

Error Control

When data-frame is transmitted, there is a probability that data-frame may be lost in the transit or it is received corrupted. In both cases, the receiver does not receive the correct data-frame and sender does not know anything about any loss.In such case, both sender and receiver are equipped with some protocols which helps them to detect transit errors such as loss of data-frame. Hence, either the sender retransmits the data-frame or the receiver may request to resend the previous data-frame.

Requirements for error control mechanism:

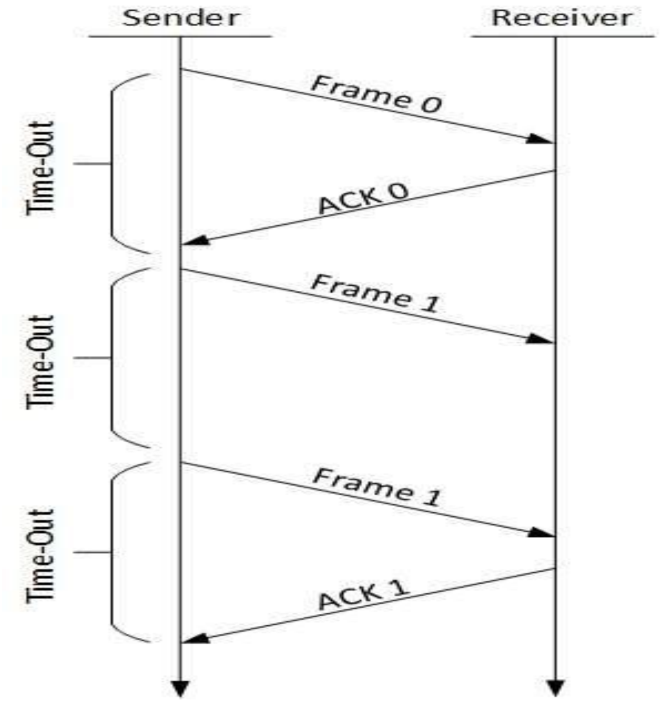
- **Error detection** - The sender and receiver, either both or any, must ascertain that there is some error in the transit.

- **Positive ACK** - When the receiver receives a correct frame, it should acknowledge it.

- **Negative ACK** - When the receiver receives a damaged frame or a duplicate frame, it sends a NACK back to the sender and the sender must retransmit the correct frame.
- **Retransmission:** The sender maintains a clock and sets a timeout period. If an acknowledgement of a data-frame previously transmitted does not arrive before the timeout the sender retransmits the frame, thinking that the frame or it's acknowledgement is lost in transit.

There are three types of techniques available which Data-link layer may deploy to control the errors by Automatic Repeat Requests (ARQ):

- **Stop-and-wait ARQ**



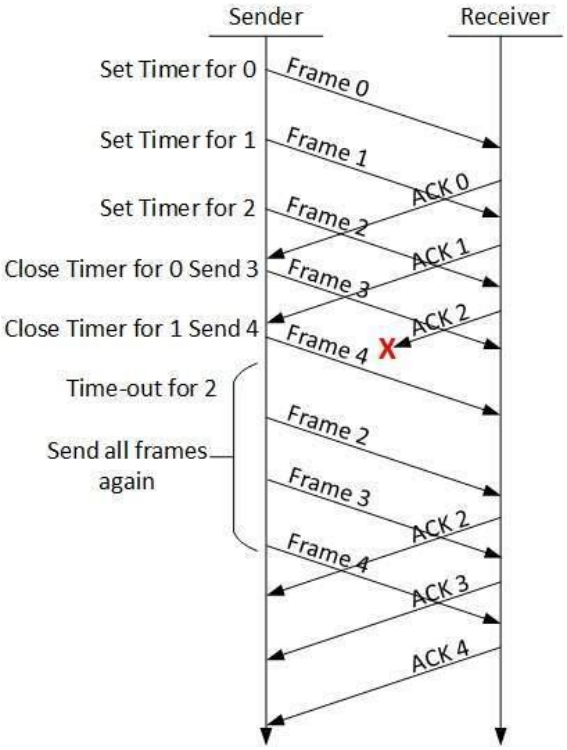
The following transition may occur in Stop-and-Wait ARQ:

- The sender maintains a timeout counter.
- When a frame is sent, the sender starts the timeout counter.
- If acknowledgement of frame comes in time, the sender transmits the next frame in queue.
- If acknowledgement does not come in time, the sender assumes that either the frame or its acknowledgement is lost in transit. Sender retransmits the frame and starts the timeout counter.

- If a negative acknowledgement is received, the sender retransmits the frame.

- **Go-Back-N ARQ**

Stop and wait ARQ mechanism does not utilize the resources at their best.When the acknowledgement is received, the sender sits idle and does nothing. In Go-Back-N ARQ method, both sender and receiver maintain a window.

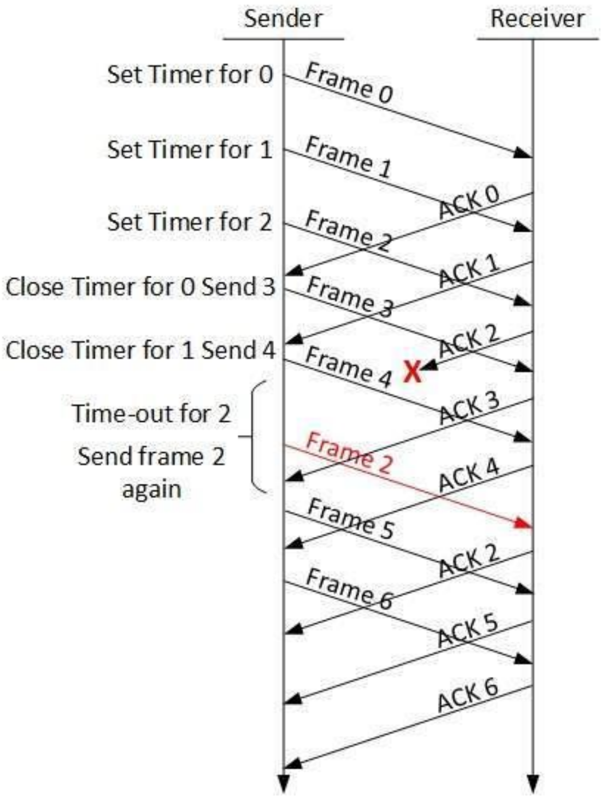


The sending-window size enables the sender to send multiple frames without receiving the acknowledgement of the previous ones. The receiving-window enables the receiver to receive multiple frames and acknowledge them. The receiver keeps track of incoming frame’s sequence number.

When the sender sends all the frames in window, it checks up to what sequence number it has received positive acknowledgement. If all frames are positively acknowledged, the sender sends next set of frames. If sender finds that it has received NACK or has not received any ACK for a particular frame, it retransmits all the frames after which it does not receive any positive ACK.

- **Selective Repeat ARQ**

In Go-back-N ARQ, it is assumed that the receiver does not have any buffer space for its window size and has to process each frame as it comes. This enforces the sender to retransmit all the frames which are not acknowledged.



In Selective-Repeat ARQ, the receiver while keeping track of sequence numbers, buffers the frames in memory and sends NACK for only frame which is missing or damaged.

The sender in this case, sends only packet for which NACK is received.

PIGGYBACKING

Piggybacking data is a bit different from Sliding Window Protocol used in the OSI model. In the data frame itself, we incorporate one additional field for acknowledgment (called ACK). Whenever party A wants to send data to party B, it will send the data along with this ACK field.

THE MEDIUM ACCESS SUBLAYER

- The channel allocation problem
- Multiple access protocol
- Ethernet

- Wireless LAN(802.11)

THE CHANNEL ALLOCATION PROBLEM

- THE CHANNEL ALLOCATION PROBLEM: how to allocate a single broadcast channel among competing users. The channel might be a portion of the wireless spectrum in a geographic region, or a single wire or optical fiber to which multiple nodes are connected.

Static Channel Allocation

- The traditional way of allocating a single channel among multiple competing users is to chop up its capacity by using one of the multiplexing schemes such as FDM (Frequency Division Multiplexing). If there are *N users, the bandwidth is divided into N equal-sized* portions, with each user being assigned one portion. Since each user has a private frequency band, there is no interference among users.

- When there are only a small and constant number of users, each of which has a steady stream or a heavy load of traffic, this division is a simple and efficient allocation mechanism. A wireless example is FM radio stations. Each station gets a portion of the FM band and uses it most of the time to broadcast its signal. However, when the number of senders is large and varying or the traffic is bursty, FDM presents some problems.

  - If the spectrum is cut up into *N regions while* fewer than *N users are currently interested in communicating, a large piece of* valuable spectrum will be wasted.

  - If more than *N users want to communicate,* some of them will be denied permission for lack of bandwidth.

  - Even if the number of users held constant at *N;* dividing the single available channel into some number of static sub channels is inherently inefficient. The basic problem is that when some users are quiescent, their bandwidth is simply lost. They are not using it, and no one else is allowed to use it either.

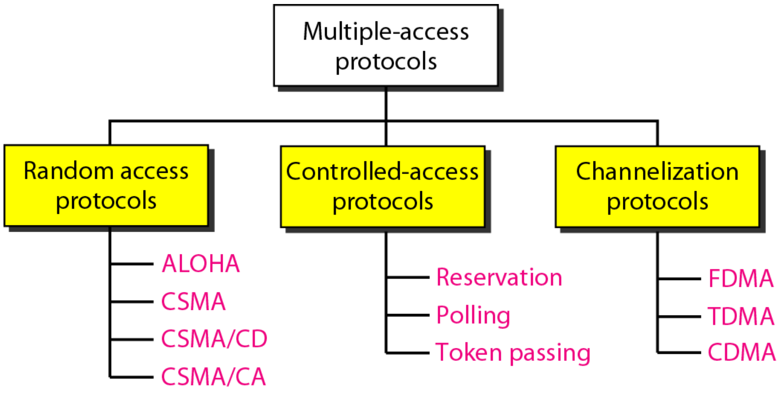
A static allocation is a poor fit to most computer systems, in which data traffic is extremely bursty, often with peak traffic to mean traffic ratios of 1000:1. Consequently, most of the channels will be idle most of the time.

Dynamic channel allocation

Channel not pre-divided into the number of users. Dynamic channel allocation. A mapping can be established when a new station appears, and the mapping can be removed when the station disappears. KEY ASSUMPTIONS:- Broadcast Network Station Model – The model consists of N independent stations (computers, telephones, personal communicators), each with a program or user that generates frames for transmission. Once a frame has been generated, the station is blocked till it has been fully transmitted. 2. Single channel – A single channel is available for all communication. All stations can transmit on it and all can receive from it. Multiple users working on same channel multiple transmitters and multiple

Collision – When 2 frames are transmitted simultaneously, they overlap in time and the resulting signal is garbled. Slotted Time – Fixed time slots. Transmission begins only at start of a slot. Carrier Sense – Station can tell if the channel is already in use or not. Time division multiplexing Here we fix a discrete time interval for dynamic allocation and new allocations will be there only when first have executed their data. If 4 users come channel is divided into 4 parts. Next time when for new 7 users channel is divided into 7 parts but only when next dynamic allocation will be done. FOR EXAMPLE: - The set of cell phones that are operating in the range of a given cell tower varies constantly. Dynamic channel allocation

MULTIPLE ACCESS PROTOCOLS



Channelization Protocols

The term channelization refers to the sharing of a point-to-point communications medium. For example, many telephone conversations (or in our context, computer-to-computer network transactions) can be submitted simultaneously on a single wire, with each conversation being on a separate channel. The notion of a channel is very closely related to the household concept of radio and TV channels. The frequency spectrum for television, for instance, is divided into subranges called channels, and these correspond to our everyday concept of TV channels. Each channel is used to transmit different information, all simultaneously. There are three main ways of doing this:

- In time-division multiplexing (TDMA), different sources transmit on the line at different times, each taking (very short) turns. This is used in long-distance phone lines

- In frequency-division multiplexing (FDMA), the different sources attached to the line send on different frequencies (e.g. different radio frequencies, or different light frequencies, i.e. different colors). This is used for radio and television transmission, and increasingly for computer-to-computer network transactions.

- In code-division multiplexing (CDMA), all nodes on the network send at the same time, on the same frequency, but using different codes. (Think of one node using a 4B/5B code, another using a second kind of code, and so on.) This is used in some cellular telephone systems.

TDMA

Time Division Multiple Access (TDMA) is a digital cellular telephone communication technology. It facilitates many users to share the same frequency without interference. Its technology divides a signal into different timeslots, and increases the data carrying capacity.

Time Division Multiple Access (TDMA) is a complex technology, because it requires an accurate synchronization between the transmitter and the receiver. TDMA is used in digital mobile radio systems. The individual mobile stations cyclically assign a frequency for the exclusive use of a time interval.

In most of the cases, the entire system bandwidth for an interval of time is not assigned to a station. However, the frequency of the system is divided into sub-bands, and TDMA is used for the multiple accesses in each sub-band. Sub-bands are known as **carrier frequencies**. The mobile system that uses this technique is referred as the **multi-carrier systems**.

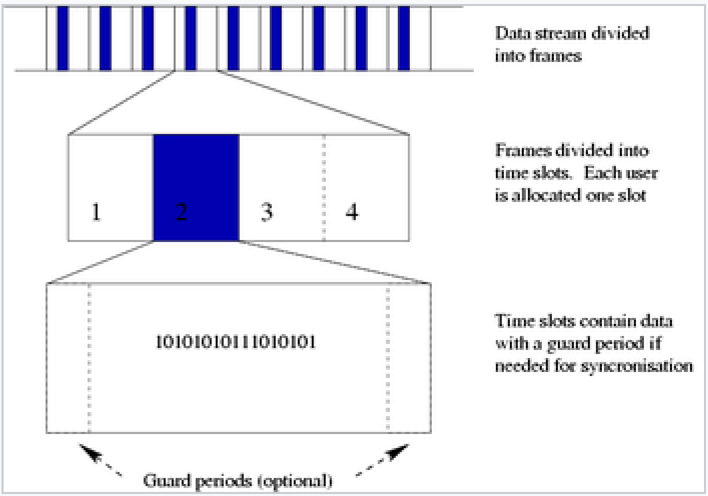
In the following example, the frequency band has been shared by three users. Each user is assigned definite **timeslots** to send and receive data. In this example, user **‘B’** sends after user **‘A,’** and user **‘C’** sends thereafter. In this way, the peak power becomes a problem and larger by the burst communication.

Advantages of TDMA

- Permits flexible rates (i.e. several slots can be assigned to a user, for example, each time interval translates 32Kbps, a user is assigned two 64 Kbps slots per frame).
- Can withstand gusty or variable bit rate traffic. Number of slots allocated to a user can be changed frame by frame (for example, two slots in the frame 1, three slots in the frame 2, one slot in the frame 3, frame 0 of the notches 4, etc.).
- No guard band required for the wideband system.
- No narrowband filter required for the wideband system.

Disadvantages of TDMA

- High data rates of broadband systems require complex equalization.
- Due to the burst mode, a large number of additional bits are required for synchronization and supervision.
- Call time is needed in each slot to accommodate time to inaccuracies (due to clock instability).
- Electronics operating at high bit rates increase energy consumption.
- Complex signal processing is required to synchronize within short slots.


Data stream divided into frames
Frames divided into time slots. Each user is allocated one slot
1
2
3
4
10101010111010101
Time slots contain data with a guard period if needed for syncronisation
Guard periods (optional)

TDMA frame structure showing a data stream divided into frames and those frames divided into time slots.

FDMA

Frequency Division Multiple Access (FDMA) is one of the most common analogue multiple access methods. The frequency band is divided into channels of equal bandwidth so that each conversation is carried on a different frequency (*as shown in the figure below*).

In FDMA method, guard bands are used between the adjacent signal spectra to minimize crosstalk between the channels. A specific frequency band is given to one person, and it will received by identifying each of the frequency on the receiving end. It is often used in the first generation of analog mobile phone.

Advantages of FDMA

As FDMA systems use low bit rates (large symbol time) compared to average delay spread, it offers the following advantages −

- Reduces the bit rate information and the use of efficient numerical codes increases the capacity.
- It reduces the cost and lowers the inter symbol interference (ISI)
- Equalization is not necessary.
- An FDMA system can be easily implemented. A system can be configured so that the improvements in terms of speech encoder and bit rate reduction may be easily incorporated.

- Since the transmission is continuous, less number of bits are required for synchronization and framing.

Disadvantages of FDMA

Although FDMA offers several advantages, it has a few drawbacks as well, which are listed below −

- It does not differ significantly from analog systems; improving the capacity depends on the signal-to-interference reduction, or a signal-to-noise ratio (SNR).
- The maximum flow rate per channel is fixed and small.
- Guard bands lead to a waste of capacity.
- Hardware implies narrowband filters, which cannot be realized in VLSI and therefore increases the cost.

CDMA

Code division multiple access (CDMA) is a channel access method used by various radio communication technologies.

CDMA is an example of multiple access, where several transmitters can send information simultaneously over a single communication channel. This allows several users to share a band of frequencies (see bandwidth). To permit this without undue interference between the users, CDMA employs spread-spectrum technology and a special coding scheme (where each transmitter is assigned a code.

Code Division Multiple Access (CDMA) is a digital cellular technology used for mobile communication. CDMA is the base on which access methods such as cdmaOne, CDMA-2000, and WCDMA are built. CDMA cellular systems are deemed superior to FDMA and TDMA, which is why CDMA plays a critical role in building efficient, robust, and secure radio communication systems.

A Simple Analogy

Let’s take a simple analogy to understand the concept of CDMA. Assume we have a few students gathered in a classroom who would like to talk to each other simultaneously. Nothing would be audible if everyone starts speaking at the same time. Either they must take turns to speak or use different languages to communicate.

The second option is quite similar to CDMA − students speaking the same language can understand each other, while other languages are perceived as noise and rejected. Similarly, in radio CDMA, each group of users is given a shared code. Many codes occupy the same channel, but only those users associated with a particular code can communicate.

Salient Features of CDMA

CDMA, which is based on the spread spectrum technique has following salient features −

- In CDMA, every channel uses the full available spectrum.
- Individual conversations are encoded with a pseudo-random digital sequence and then transmitted using a wide frequency range.
- CDMA consistently provides better capacity for voice and data communications, allowing more subscribers to connect at any given time.
- CDMA is the common platform on which 3G technologies are built. For 3G, CDMA uses 1x EV-DO and EV-DV.

<u>Random Access Protocols</u>

ALOHA: ALOHA is a system for coordinating and arbitrating access to a shared communication Networks channel. It was developed in the 1970s by Norman Abramson and his colleagues at the University of Hawaii. The original system used for ground based radio broadcasting, but the system has been implemented in satellite communication systems.

A shared communication system like ALOHA requires a method of handling collisions that occur when two or more systems attempt to transmit on the channel at the same time. In the ALOHA system, a node transmits whenever data is available to send. If another node transmits at the same time, a collision occurs, and the frames that were transmitted are lost. However, a node can listen to broadcasts on the medium, even its own, and determine whether the frames were transmitted.

There are two different types of ALOHA:

(i)PureALOHA
(ii) Slottecl ALOHA

(i) Pure ALOHA

• **In** pure ALOHA, the stations transmit frames whenever they have data to send.

• When two or more stations transmit simultaneously, there is collision and the frames are destroyed.

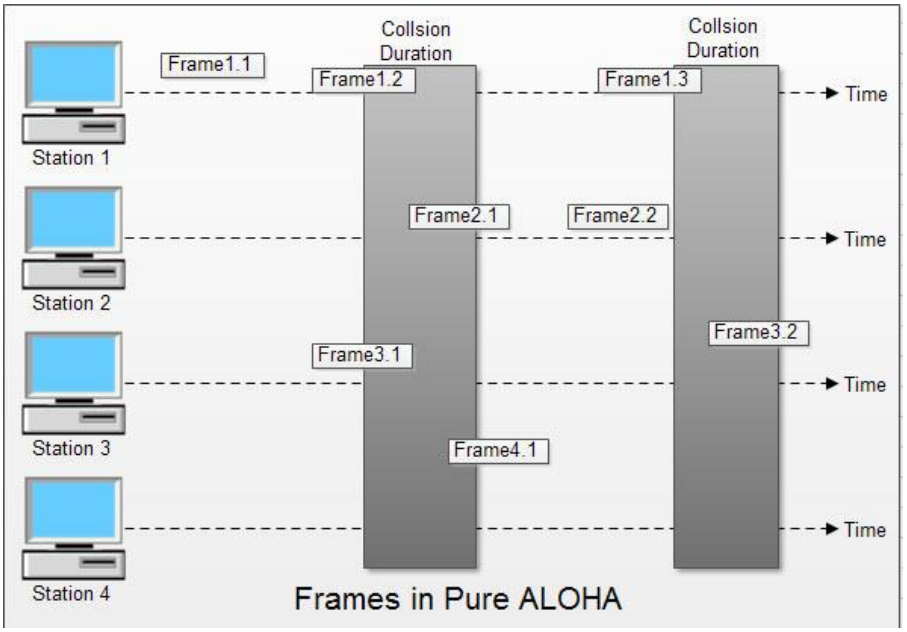
• In pure ALOHA, whenever any station transmits a frame, it expects the acknowledgement from the receiver.

• If acknowledgement is not received within specified time, the station assumes that the frame (or acknowledgement) has been destroyed.

• If the frame is destroyed because of collision the station waits for a random amount of time and sends it again. This waiting time must be random otherwise same frames will collide again and again.

• Therefore pure ALOHA dictates that when time-out period passes, each station must wait for a random amount of time before resending its frame. This randomness will help avoid more collisions.

• Figure shows an example of frame collisions in pure ALOHA.



• In fig there are four stations that .contended with one another for access to shared channel. All these stations are transmitting frames. Some of these frames collide because multiple frames are in contention for the shared channel. Only two frames, frame 1.1 and frame 2.2 survive. All other frames are destroyed.

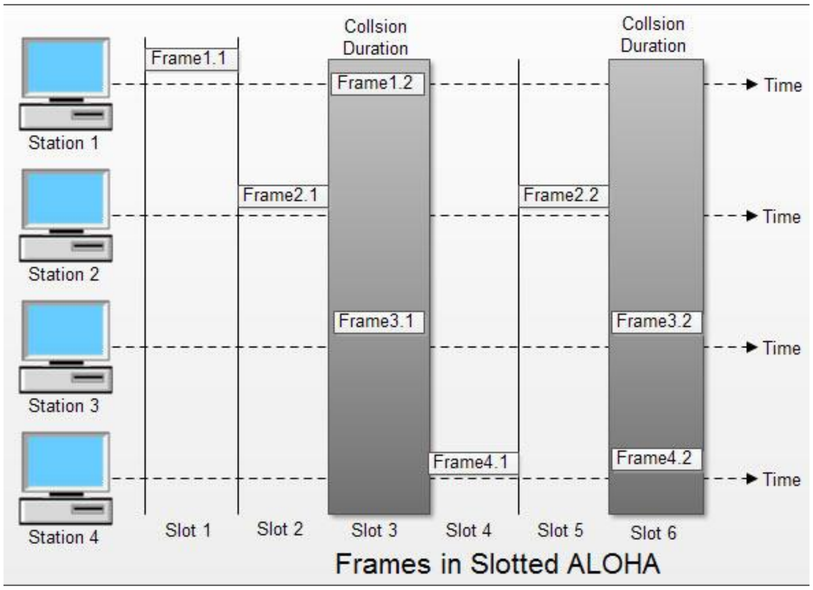
- Whenever two frames try to occupy the channel at the same time, there will be a collision and both will be damaged. If first bit of a new frame overlaps with just the last bit of a frame almost finished, both frames will be totally destroyed and both will have to be retransmitted.

(ii) Slotted ALOHA

- Slotted ALOHA was invented to improve the efficiency of pure ALOHA as chances of collision in pure ALOHA are very high.

- In slotted ALOHA, the time of the shared channel is divided into discrete intervals called slots.

- The stations can send a frame only at the beginning of the slot and only one frame is sent in each slot.



- In slotted ALOHA, if any station is not able to place the frame onto the channel at the beginning of the slot *i.e.* it misses the time slot then the station has to wait until the beginning of the next time slot.

- In slotted ALOHA, there is still a possibility of collision if two stations try to send at the beginning of the same time slot as shown in fig.

- Slotted ALOHA still has an edge over pure ALOHA as chances of collision are reduced to one-half.

Key Differences Between Pure ALOHA and Slotted ALOHA

1. Pure ALOHA was introduced by Norman and his associates at the university of Hawaii in 1970. On the other hand, Slotted ALOHA was introduced by Roberts in 1972.

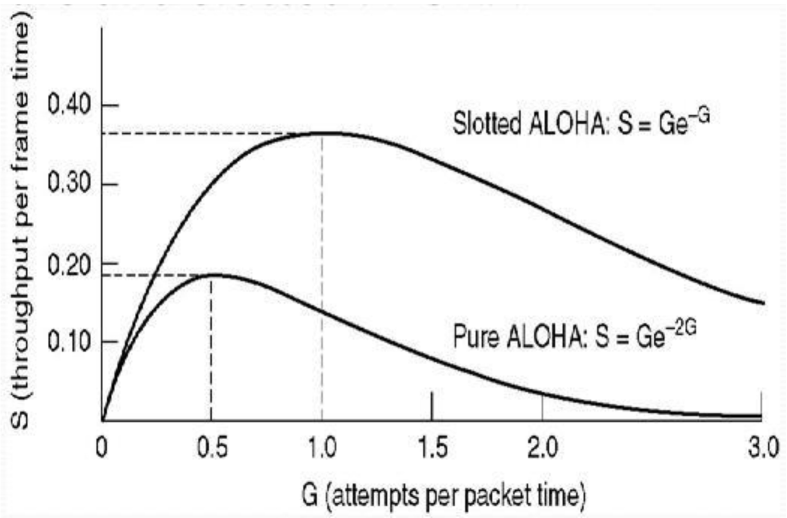
2. In pure ALOHA, whenever a station has data to send it transmits it without waiting whereas, in slotted ALOHA a user wait till the next time slot beings to transmit the data.

3. In pure ALOHA the time is continuous whereas, in Slotted ALOHA the time is discrete and divided into slots.

4. In pure ALOHA the probability of successful transmission is S=G*e^-2G. On the other hand, in slotted ALOHA the probability of successful transmission is S=G*e^-G.

5. The time of sender and receiver in pure ALOHA is not globally synchronized whereas, the time of sender and receiver in slotted ALOHA is globally synchronized.

6. The maximum throughput occurs at G=1/2 which is 18 % whereas, the maximum throughput occurs at G=1 which is 37%.



CSMA, or listen with random access carrier

Technical CSMA (Carrier Sense Multiple Access) is to listen to the channel before transmitting. This significantly reduces the risk of collision, but does not eliminate them completely. If during the propagation time between the couple of the more remote stations (vulnerability period), a coupler does not detect the transmission of a frame, and there may be signal superposition. Therefore, it is necessary to subsequently retransmit lost frames.

Numerous variations of this technique have been proposed, which differ by three Features:

• The strategy followed by the module after detecting the channel status.

• The way collisions are detected.

• The message retransmission after collision policy.

Its main variants are:

• **Non-persistent CSMA**. The coupler the listening channel when a frame is ready to be sent. If the channel is free, the module emits. Otherwise, it starts the same process after a random delay.

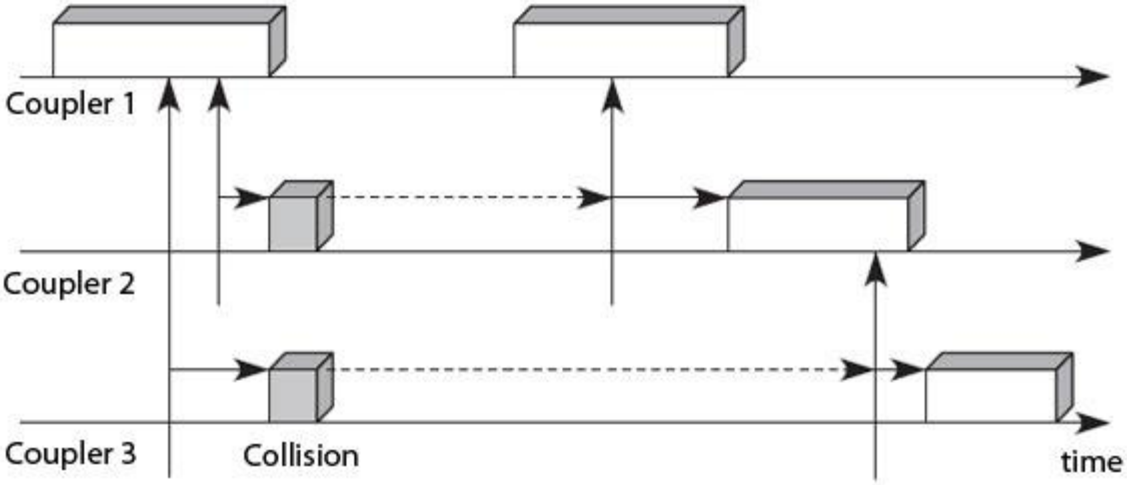
• **Persistent CSMA -** A loan coupler to transmit the channel and previously listening forwards if it is free. If it detects the occupation of carrier, it continues to listen until the channel is clear and transmits at that time. This technique allows lose less time than in the previous case, but it has the disadvantage increase the likelihood of collision, since the frames that accumulate during the busy time are all transmitted simultaneously.

• P-persistent CSMA - The algorithm is the same as before, but when the

Channel becomes free; the module transmits with probability p. In other words, the coupler differs his show with probability 1 - p. This algorithm reduces the likelihood of collision. Assuming both terminals simply making the collision is inevitable in the standard case. With the new algorithm, there is a probability 1 - p that each terminal does not transmit, thereby avoiding the collision. However, it increases the time before transmission, since a terminal may choose not to transmit, with a probability 1 - p, while the channel is free.

• **CSMA / CD (Carrier Sense Multiple Access / Collision Detection) -** This technique normalized random access by the IEEE 802.3 working group is currently the longer used. At a preliminary listening to the network is added listening during transmission. Coupler to issue a loan that detected free channel transmits and continues to listen the channel. The coupler continues to listen, which is sometimes indicated by the CSMA / CD persistent acronym. If there is a collision, it interrupts its transmission as soon as possible and sends special signals, called padding bits so that all couplers are notified of the collision. He tries again his show later using an algorithm that we present later.

Figure shows the CSMA/CD. In this example, the couplers 2 and 3 attempt broadcasting for the coupler 1 transmits its own frame. The couplers 2 and 3 begin to listen and transmit at the same time, the propagation delay around, from the end of the Ethernet frame transmitted by the coupler 1. A collision ensues. Like the couplers 2 and 3 continue to listen to the physical media, they realize the collision, stop their transmission and draw a random time to start retransmission process.



Operating Principle of CSMA / CD

The CSMA/CD creates an efficiency gain compared to other techniques random access because there are immediate collision detection and interruption of current transmission. Issuers couplers recognize a collision by comparing the transmitted signal with the passing on the line. The collisions are no longer recognized by absence of acknowledgment but by detecting interference. This conflict detection method is relatively simple, but it requires sufficient performance coding techniques to easily recognize a superposition signal. It is generally used for this differential coding technique, such as differential Manchester code.

• **CSMA / CA -** Less known than the CSMA / CD access CSMA / CA (Carrier Sense Multiple Access / Collision Avoidance) starts to be heavily used in Wi-Fi networks, that is to say, the wireless Ethernet IEEE 802.11. This is a variation of the CSMA / CD, which allows the CSMA method run when collision detection is not possible, as in the radio. Its operating principle is to resolve contention before the data are transmitted using acknowledgments and timers.

The couplers are testing wishing to transmit the channel several times to ensure that no activity is detected. Every message received shall be immediately paid by the receiver. Sending new messages takes place only after a certain period, so as to ensure a transport without loss of information. The non-return of an acknowledgment, after a predetermined time interval, to detect if there was a collision. This strategy not only makes it possible to implement an acknowledgment mechanism in frame but has the advantage of being simple and economic, since it does not require collision detection circuit, unlike the CSMA/ CD.

There are various techniques of CSMA with collision resolution, including the CSMA / CR (Carrier Sense Multiple Access / Collision Resolution). Some variants use the CSMA also priority mechanisms that may come under this term, that avoid collisions by separate priority levels associated with different stations connected to the network.

Controlled Access Protocols

In controlled access, the stations consult one another to find which station has the right to send. A station cannot send unless it has been authorized by other stations. The three popular controlled-access methods are as follows.

Polling:

Polling works with topologies in which one device is designated as a primary station and the other devices are secondary stations. All data exchanges must be made through the primary device even when the ultimate destination is a secondary device.

The primary device controls the link; the secondary devices follow its instructions. It is up to the primary device to determine which device is allowed to use the channel at a given time. The primary device, therefore, is always the initiator of a session. Consider the following figure.

If the primary wants to receive data, it asks the secondaries if they have anything to send, this is called poll function. If the primary wants to send data, it tells the secondary to get ready to receive; this is called select function.

Select:

The select function is used whenever the primary device has something to send. If it has something to send, the primary device sends it. It has to know whether the target device is prepared to receive or not. So the primary must alert the secondary to the upcoming transmission and wait for an acknowledgment of the secondary's ready status. Before sending data, the primary creates and transmits a select (SEL) frame, one field of which includes the address of the intended secondary.

Poll:

The poll function is used by the primary device to solicit transmissions from the secondary devices. When the primary is ready to receive data, it must ask (poll) each device in turn if it has anything to send. When the first secondary is approached, it responds either with a NAK frame if it has nothing to send or with data (in the form of a data frame) if it does. If the response is negative (a NAK frame), then the primary polls the next secondary in the same manner until it finds one with data to send. When the response is positive (a data frame), the primary reads the frame and returns an acknowledgment (ACK frame), verifying its receipt.

3. Token Passing:

In the token-passing method, the stations in a network are organized in a logical ring. In other words, for each station, there is a predecessor and a successor. The predecessor is the station which is logically before the station in the ring; the successor is the station which is after the station in the ring. The current station is the one that is accessing the channel now. The right to this access has been passed from the predecessor to the current station. The right will be passed to the successor when the current station has no more data to send.

In this method, a special packet called a token circulates through the ring. The possession of the token gives the station the right to access the channel and send its data. When a station has some data to send, it waits until it receives the token from its predecessor. It then holds the token and sends its data. When the station has no more data to send, it releases the token, passing it to the next logical station in the ring. The station cannot send data until it receives the token again in the next round.

Token management is needed for this access method. Stations must be limited in the time they can have possession of the token. The token must be monitored to ensure it has not been lost or destroyed. For example, if a station that is holding the token fails, the token will disappear from the network. Another function of token management is to assign priorities to the stations and to the types of data being transmitted. And finally, token management is needed to make low- priority stations release the token to high priority stations.

3.5. DATA LINK PROTOCOL

a) HDLC

A high-level data link control (HDLC) is a protocol that is a bit-oriented synchronous data link layer. HDLC ensures the error-free transmission of data to the proper destinations and controls the data transmission speed.

HDLCs can provide both connection-oriented and connectionless services.

A high-level data link control defines rules for transmitting data between network points. Data in an HDLC is organized into units called frames and is sent across networks to specified destinations. HDLC also manages the pace at which data is transmitted. HDLC is commonly used in the open systems interconnection (OSI) model's layer.

HDLC frames are transmitted over synchronous links or asynchronous links, which do not mark the start and end of frames. This is done using a frame delimiter or flag, which contains unique sequence of bits that are not visible inside a frame.

There are three types of HDLC frames:

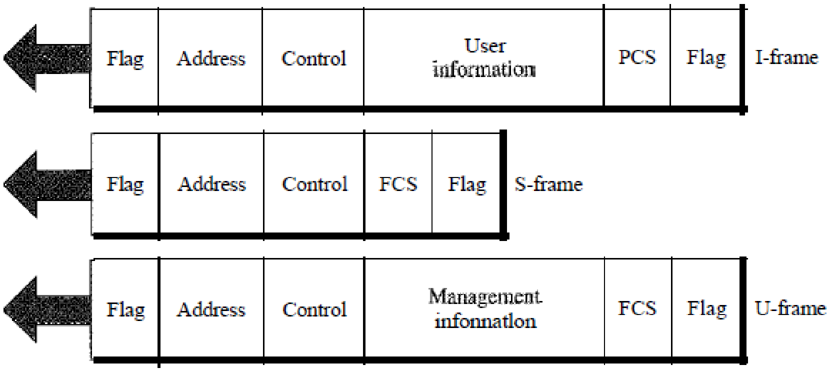
- Information frames/User data (I-frames)
- Supervisory frames/Control data (S-frames)
- Unnumbered frames (U-frames)

The common fields within an HDLC frame are:

- Flag
- Address
- Control information
- Frame check sequence

The HDLC protocol is used by a variety of standards implemented in the protocol stacks of X.25, V.42 and ISDN and many other protocol stacks.

Figure 11.27 *HDLC frames*



o Flag field. The flag field of an HDLC frame is an 8-bit sequence with the bit pattern 01111110 that identifies both the beginning and the end of a frame and serves as a synchronization pattern for the receiver.

o Address field. The second field of an HDLC frame contains the address of the secondary station. If a primary station created the frame, it contains a to address. If a secondary creates the frame, it contains a from address. An address field can be 1 byte or several bytes long, depending on the needs of the network. One byte can identify up to 128 stations (l bit is used for another purpose). Larger networks require multiple-byte address fields. If the address field is only 1 byte, the last bit is always a 1. If the address is more than 1 byte, all bytes but the last one will end with 0; only the last will end with 1. Ending each intermediate byte with 0 indicates
to the receiver that there are more address bytes to come.

o Control field. The control field is a 1- or 2-byte segment of the frame used for flow and error control. The interpretation of bits in this field depends on the frame type. We discuss this field later and describe its format for each frame type.

o Information field. The information field contains the user's data from the network layer or management information. Its length can vary from one network to another.

o FCS field. The frame check sequence (FCS) is the HDLC error detection field. It can contain either a 2- or 4-byte ITU-T CRC.

b) PPP

In computer networking, **Point-to-Point Protocol** (**PPP**) is a data link (layer 2) protocol used to establish a direct connection between two nodes. It connects two routers directly without any host or any other networking device in between. It can provide connection authentication, transmission encryption (using ECP, RFC 1968), and compression.

PPP is used over many types of physical networks including serial cable, phone line, trunk line, cellular telephone, specialized radio links, and fiber optic links such as SONET.

1. It defines link control protocol (LCP) for:-

(a) Establishing the link between two devices.

(b) Maintaining this established link.

(c) Configuring this link.

(d) Terminating this link after the transfer.

2. It defines how network layer data are encapsulated in data link frame.

3. PPP provides error detection.

4. Unlike SLIP that supports only IP, PPP supports multiple protocols.

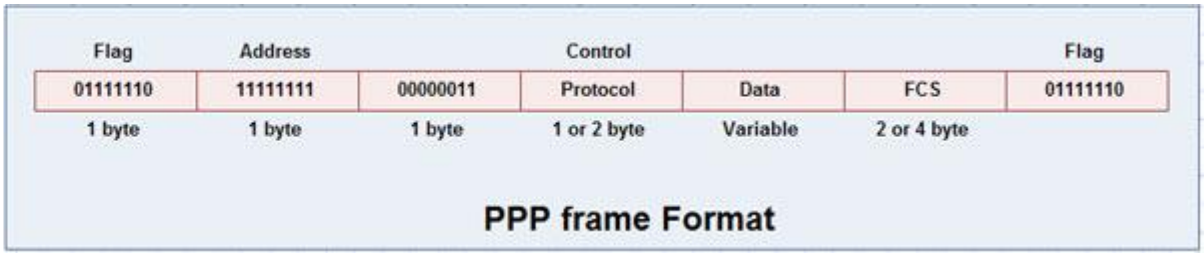
5. PPP allows the IP address to be assigned at the connection time i.e. dynamically. Thus a temporary IP address can be assigned to each host.

6. PPP provides multiple network layer services supporting a variety of network layer protocol. For this PPP uses a protocol called NCP (Network Control Protocol).

8. It also defines how two devices can authenticate each other.

PPP Frame Format

The frame format of PPP resembles HDLC frame. Its various fields are:



Flag	Address		Control			Flag
01111110	11111111	00000011	Protocol	Data	FCS	01111110
1 byte	1 byte	1 byte	1 or 2 byte	Variable	2 or 4 byte	

PPP frame Format

1. **Flag field**: Flag field marks the beginning and end of the PPP frame. Flag byte is 01111110. (1 byte).

2. **Address field**: This field is of 1 byte and is always 11111111. This address is the broadcast address i.e. all the stations accept this frame.

3. **Control field**: This field is also of 1 byte. This field uses the format of the U-frame (unnumbered) in HDLC. The value is always 00000011 to show that the frame does not contain any sequence numbers and there is no flow control or error control.

4. **Protocol field**: This field specifies the kind of packet in the data field i.e. what is being carried in data field.

5. **Data field**: Its length is variable. If the length is not negotiated using LCP during line set up, a default length of 1500 bytes is used. It carries user data or other information.

6. **FCS field**: The frame checks sequence. It is either of 2 bytes or 4 bytes. It contains the checksum.

3.9. ETHERNET (IEEE 802.3) LOCAL AREA NETWORK (LAN)

Ethernet protocols refer to the family of local-area network (LAN) covered by the IEEE 802.3. In the Ethernet standard, there are two modes of operation: half-duplex and full-duplex modes. In the half duplex mode, data are transmitted using the popular Carrier-Sense Multiple Access/Collision Detection (CSMA/CD) protocol on a shared medium. The main disadvantages of the half-duplex are the efficiency and distance limitation, in which the link distance is limited by the minimum MAC frame size. This restriction reduces the efficiency drastically for high-rate transmission. Therefore, the carrier extension technique is used to ensure the minimum frame size of 512 bytes in Gigabit Ethernet to achieve a reasonable link distance.

Four data rates are currently defined for operation over optical fiber and twisted-pair cables:

- 10 Mbps – 10Base-T Ethernet (IEEE 802.3)
- 100 Mbps – Fast Ethernet (IEEE 802.3u)
- 1000 Mbps – Gigabit Ethernet (IEEE 802.3z)
- 10-Gigabit – 10 Gbps Ethernet (IEEE 802.3ae).

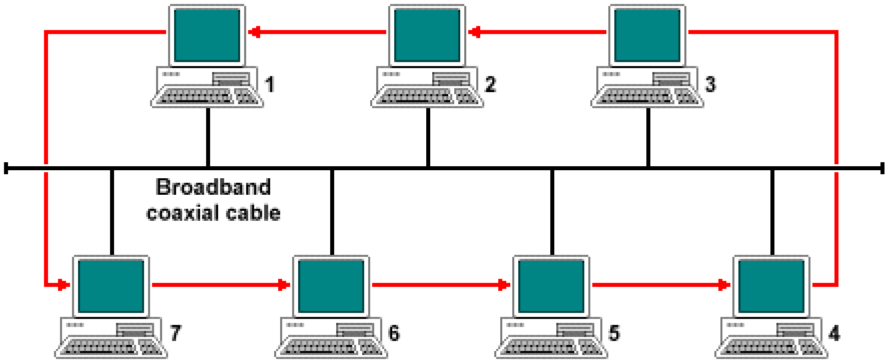
In this document, we discuss the general aspects of the Ethernet. The specific issues regarding Fast Ethernet, Gigabit and 10 Gigabit Ethernet will be discussed in separate documents.

The Ethernet system consists of three basic elements: 1. the physical medium used to carry Ethernet signals between computers, 2. a set of medium access control rules embedded in each Ethernet interface that allow multiple computers to fairly arbitrate access to the shared Ethernet channel, and 3. an Ethernet frame that consists of a standardized set of bits used to carry data over the system.

3.10. NETWORKS

Token Bus (IEEE 802.4)

Token Bus is described in the IEEE 802.4 specification, and is a Local Area Network (LAN) in which the stations on the bus or tree form a *logical ring*. Each station is assigned a place in an ordered sequence, with the last station in the sequence being followed by the first, as shown below. Each station knows the address of the station to its "left" and "right" in the sequence.



A Token Bus network

This type of network, like a Token Ring network, employs a small data frame only a few bytes in size, known as a *token*, to grant individual stations exclusive access to the network transmission medium. Token-passing networks are *deterministic* in the way that they control access to the network, with each node playing an active role in the process. When a station acqires control of the token, it is allowed to transmit one or more data frames, depending on the time limit imposed by the network. When the station has finished using the token to transmit data, or the time limit has expired, it relinquishes control of the token, which is then available to the next station in the logical sequence. When the ring is initialized, the station with the highest number in the sequence has control of the token.

Topology of the network is either a bus or a tree, although the order in which stations are connected to the network is not important. The network topology means that we are essentially dealing with a broadcast network, and every frame transmitted is received by all attached stations. With the exception of broadcast frames, however, frames will only be read by the station to which they are addressed, and ignored by all other stations. As the token frame is transmitted, it carries the destination address of the next station in the logical sequence. As each individual station is powered on, it is allocated a place in the ring sequence (note that in the diagram above, station two is not participating in the ring). The Token Bus medium access control protocol allows stations to join the ring or leave the ring on an ad-hoc basis.

Token Bus networks were conceived to meet the needs of automated industrial manufacturing systems and owe much to a proposal by General Motors for a networking system to be used in their own manufacturing plants - *Manufacturing Automation Protocol* (MAP). Ethernet was not considered suitable for factory automation systems because of the contention-based nature of its medium access control protocol, which meant that the length of time a station might have to wait to send a frame was unpredictable. Ethernet also lacked a priority system, so there was no way to ensure that more important data would not be held up by less urgent traffic.

A token-passing system in which each station takes turns to transmit a frame was considered a better option, because if there are *n* stations, and each station takes *T* seconds to send a frame, no station has to wait longer than *nT* seconds to acquire the token. The ring topology of existing token-passing systems, however, was not such an attractive idea, since a break in the ring would cause a general network failure. A ring topology was also considered to be incompatible with the linear topology of assembly-line or process control systems. Token Bus was a hybrid system that provided the robustness and linearity of a bus or tree topology, whilst retaining the known worst-case performance of a token-passing medium access control method.

The transmission medium most often used for broadband Token Bus networks is 75 Ohm coaxial cable (the same type of cable used for cable TV), although alternative cabling configurations are available. Both single and dual cable systems may be used, with or without head-ends. Transmission speeds vary, with data rates of 1, 5 and 10 Mbps being common. The analogue modulation schemes that can be used include:

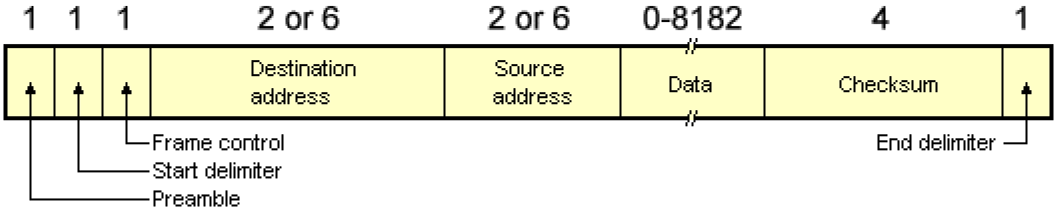
- Phase continuous frequency shift keying
- Phase coherent frequency shift keying
- Multilevel duo binary amplitude modulated phase shift keying

The Token Bus MAC layer protocol

When the ring is initialized, tokens are inserted into it in station address order, starting with the highest. The token itself is passed from higher to lower addresses. Once a station aquires the token, it has a fixed time period during which it may transmit frames, and the number of frames which can be transmitted by each station during this time period will depend on the length of each frame. If a station has no data to send, it simply passes the token to the next station without delay.

The Token Bus standard defines four classes of priority for traffic - 0, 2, 4, and 6 - with 6 representing the highest priority and 0 the lowest. Each station maintains four internal queues that correspond to the four priority levels. As a frame is passed down to the MAC sublayer from a higher-layer protocol, its priority level is determined, and it is assigned to the appropriate queue. When a station acquires the token, frames are transmitted from each of the four queues in strict order of priority. Each queue is allocated a specific time slot, during which frames from that queue may be transmitted. If there are no frames waiting in a particular queue, the token immediately becomes available to the next queue. If the token reaches level 0 and there are no frames waiting, it is immediately passed to the next station in the logical ring. The whole process is controlled by timers that are used to allocate time slots to each priority level. If any queue is empty, its time slot may be allocated for use by the remaining queues.

The priority scheme guarantees level 6 data a known fraction of the network bandwidth, and can therefore be used to implement a real-time control system. As an example, if a network running at 10 Mbps and having fifty stations has been configured so that level 6 traffic is allocated one-third of the bandwidth, each station has a guaranteed bandwidth of 67 kbps for level 6 traffic. The available high priority bandwidth could thus be used to synchronize robots on an assembly line, or to carry one digital voice channel per station, with some bandwidth left over for control information.



The Token Bus frame format

The Token Bus frame format is shown above. The *Preamble* field is used to synchronise the receiver's clock. The *Start Delimeter* and *End Delimeter* fields are used to mark the start and end of the frame, and contain an analogue encoding of symbols other than 0s and 1s that cannot occur accidentally within the frame data. For this reason, a length field is not required.

The *Frame Control* field identifies the frame as either a data frame or a control frame. For data frames, it includes the priority level of the frame, and may also include an indicator requiring the destination station to acknowledge correct or incorrect receipt of the frame. For control frames, the field specifies the frame type.

The *Destination* and *Source* address fields contain either a 2-byte or a 6-byte hardware address

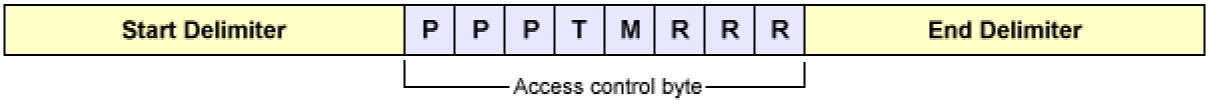
for the destination and source stations respectively (a given network must use either 2-byte or 6-byte addresses consistently, not a mixture of the two). If 2-byte addresses are used, the *Data Field* can be up t0 8,182 bytes. If 6-byte addresses are used, it is limited to 8,174 bytes.
The *Checksum* is used to detect transmission errors. The various control frames used are shown in the table below.

Token Ring (IEEE 802.5)

Token Ring was developed by IBM in the 1970s and is described in the IEEE 802.5 specification. It is no longer widely used in LANs. Token passing is the method of medium access, with only one token allowed to exist on the network at any one time. Network devices must acquire the token to transmit data, and may only transmit a single frame before releasing the token to the next station on the ring. When a station has data to transmit, it acquires the token at the earliest opportunity, marks it as busy, and attaches the data and control information to the token to create a data frame, which is then transmitted to the next station on the ring. The frame will be relayed around the ring until it reaches the destination station, which reads the data, marks the frame as having been read, and sends it on around the ring. When the sender receives the acknowledged data frame, it generates a new token, marks it as being available for use, and sends it to the next station. In this way, each of the other stations on the ring will get an opportunity to transmit data (even if they don't have any data to transmit!).

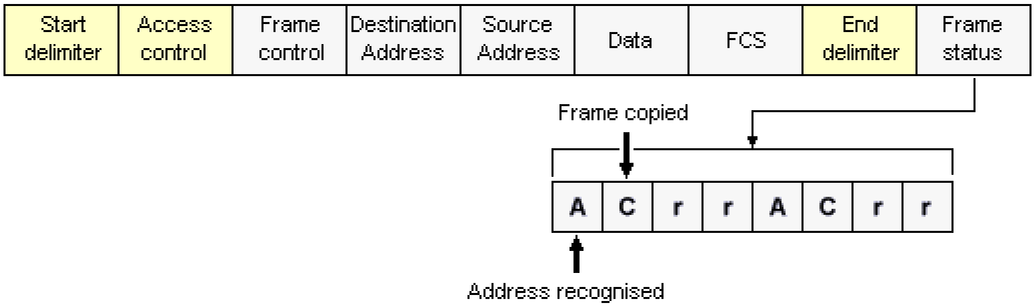
Token Ring networks provide a priority system that allows administrators to designate specific stations as having a higher priority than others, allowing those stations to use the network more frequently by setting the priority level of the token so that only stations with the same priority or higher can use the token (or reserve the token for future use). Stations that raise a token's priority must reinstate the priority level previously in force once they have used the token. In a Token Ring network, one station is arbitrarily selected to be the *active monitor*. The active monitor acts as a source of timing information for other stations, and performs various maintenance functions, such as generating a new token as and when required, or preventing rogue data frames from endlessly circulating around the ring. All of the stations on the ring have a role to play in managing the network, however. Any station that detects a serious problem will generate a *beacon frame* that alerts other stations to the fault condition, prompting them to carry out diagnostic activities and attempt to re-configure the network.

Two basic frame types are used - tokens, and data/command frames. The token is three bytes long and consists of a *start delimiter*, an *access control byte*, and an *end delimiter*. The format of the token is shown below.



The Token Ring token

A data/command frame has the same fields as the token, plus several additional fields. The format of the data/command frame is shown below.



The Token Ring frame format

- *Start delimiter* - alerts each station of the arrival of a token or frame.

- *Access control byte* - contains the *priority field*, the *reservation field*, the *token bit* and a*monitor bit*.

- *Frame control byte* - indicates whether the frame contains data or control information. In a control frame, this byte specifies the type of control information carried.

- *Destination and source addresses* - two six-byte fields that identify the destination and source station MAC addresses.

- *Data* - the maximum length is limited by the ring *token holding time*, which defines the maximum time a station can hold the token

- *Frame check sequence (FCS)* - filled by the source station with a calculated value dependent on the frame contents. The destination station recalculates the value to determine whether the frame was damaged in transit. If so, the frame is discarded.

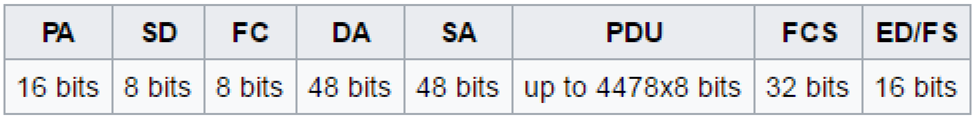
- *End delimiter* - signals the end of the token or frame, and contains bits that may be used to indicate a damaged frame, and to identify the last frame in a logical sequence.

- *Frame status* - a one-byte field that terminates a frame, and includes the one-bit *address-recognized* and *frame-copied* fields. These one-bit fields, if set, provide confirmation that the frame has been delivered to the source address and the data read. Both fields are duplicated within the frame status byte.

FDDI

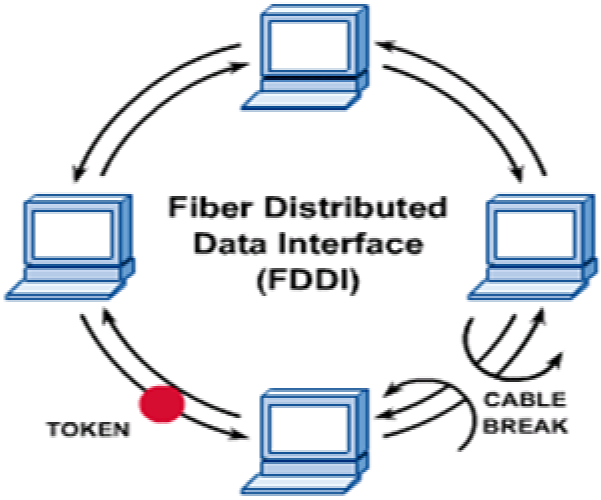
Fiber Distributed Data Interface (**FDDI**) is a standard for data transmission in a local area network. It uses optical fiber as its standard underlying physical medium. FDDI (Fiber Distributed Data Interface) is a set of ANSI and ISO standards for data transmission on fiber optic lines in a local area network (LAN) that can extend in range up to 200 km (124 miles). The FDDI protocol is based on the token ring protocol. In addition to being large geographically, an FDDI local area network can support thousands of users. FDDI is frequently used on the backbone for a wide area network (WAN). An FDDI network contains two token rings, one for possible backup in case the primary ring fails. The primary ring offers up to 100 Mbpscapacity. If the secondary ring is not needed for backup, it can also carry data, extending capacity to 200 Mbps. The single ring can extend the maximum distance; a dual ring can extend 100 km (62 miles).

The FDDI data frame format is:



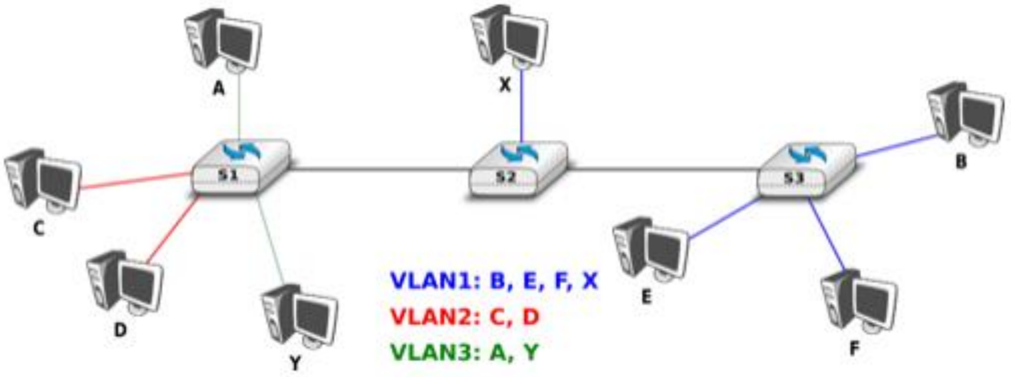
PA	SD	FC	DA	SA	PDU	FCS	ED/FS
16 bits	8 bits	8 bits	48 bits	48 bits	up to 4478x8 bits	32 bits	16 bits

Where **PA** is the preamble, **SD** is a start delimiter, **FC** is frame control, **DA** is the destination address, **SA** is the source address, **PDU** is the protocol data unit (or packet data unit), **FCS** is the frame check Sequence (or checksum), and **ED/FS** are the end delimiter and frame status.



VLANS

Virtual LANs Another important advantage of Ethernet switches is the ability to create Virtual Local Area Networks (VLANs). A virtual LAN can be defined as a set of ports attached to one or more Ethernet switches. A switch can support several VLANs and it runs one MAC learning algorithm for each Virtual LAN. When a switch receives a frame with an unknown or a multicast destination, it forwards it over all the ports that belong to the same Virtual LAN but not over the ports that belong to other Virtual LANs. Similarly, when a switch learns a source address on a port, it associates it to the Virtual LAN of this port and uses this information only when forwarding frames on this Virtual LAN. The figure below illustrates a switched Ethernet network with three Virtual LANs. VLAN2 and VLAN3 only require a local configuration of switch S1. Host C can exchange frames with host D, but not with hosts that are outside of its VLAN. VLAN1 is more complex as there are ports of this VLAN on several switches. To support such VLANs, local configuration is not sufficient anymore. When a switch receives a frame from another switch, it must be able to determine the VLAN in which the frame originated to use the correct MAC table to forward the frame. This is done by assigning an identifier to each Virtual LAN and placing this identifier inside the headers of the frames that are exchanged between switches. Virtual Local Area Networks in a switched Ethernet network



Virtual Local Area Networks in a switched Ethernet network

CHAPTER 4-NETWORK LAYER

4.1. INTERNETWORKING DEVICES

An internetworking device is a widely-used term for any hardware within networks that connect different network resources. Key devices that comprise a network are routers, bridges, repeaters and gateways.

Hub

Hub is one of the basic icons of networking devices which works at physical layer and hence connect networking devices physically together. Hubs are fundamentally used in networks that use twisted pair cabling to connect devices. They are designed to transmit the packets to the other appended devices without altering any of the transmitted packets received. They act as pathways to direct electrical signals to travel along. They transmit the information regardless of the fact if data packet is destined for the device connected or not.

Hub falls in two categories:

Active Hub: They are smarter than the passive hubs. They not only provide the path for the data signals in fact they regenerate, concentrate and strengthen the signals before sending them to their destinations. Active hubs are also termed as 'repeaters'.

Passive Hub: They are more like point contact for the wires to built in the physical network. They have nothing to do with modifying the signals.

Repeater

A repeater connects two segments of your network cable. It retimes and regenerates the signals to proper amplitudes and sends them to the other segments. When talking about, ethernet topology, you are probably talking about using a hub as a repeater. Repeaters require a small amount of time to regenerate the signal. This can cause a propagation delay which can affect network communication when there are several repeaters in a row. Many network architectures limit the number of repeaters that can be used in a row. Repeaters work only at the physical layer of the OSI network model.

Switches

Switches are the linkage points of an Ethernet network. Just as in hub, devices in switches are connected to them through twisted pair cabling. But the difference shows up in the manner both the devices; hub and a switch treat the data they receive. Hub works by sending the data to all the ports on the device whereas a switch transfers it only to that port which is connected to the destination device. A switch does so by having an in-built learning of the MAC address of the devices connected to it. Since the transmission of data signals are well defined in a switch hence the network performance is consequently enhanced. Switches operate in full-duplex mode where devices can send and receive data from the switch at the simultaneously unlike in half-duplex mode. The transmission speed in switches is double than in Ethernet hub transferring a 20Mbps connection into 30Mbps and a 200Mbps connection to become 300Mbps. Performance improvements are observed in networking with the extensive usage of switches in the modern days.

The following method will elucidate further how data transmission takes place via switches:

- Cut-through transmission: It allows the packets to be forwarded as soon as they are received. The method is prompt and quick but the possibility of error checking gets overlooked in such kind of packet data transmission.
- Store and forward: In this switching environment the entire packet are received and 'checked' before being forwarded ahead. The errors are thus eliminated before being propagated further. The downside of this process is that error checking takes relatively longer time consequently making it a bit slower in processing and delivering.
- Fragment Free: In a fragment free switching environment, a greater part of the packet is examined so that the switch can determine whether the packet has been caught up in a collision. After the collision status is determined, the packet is forwarded.

Bridges

A bridge is a computer networking device that builds the connection with the other bridge networks which use the same protocol. It works at the Data Link layer of the OSI Model and connects the different networks together and develops communication between them. It connects two local-area networks; two physical LANs into larger logical LAN or two segments of the same LAN that use the same protocol.

Apart from building up larger networks, bridges are also used to segment larger networks into smaller portions. The bridge does so by placing itself between the two portions of two physical networks and controlling the flow of the data between them. Bridges nominate to forward the data after inspecting into the MAC address of the devices connected to every segment. The forwarding of the data is dependent on the acknowledgement of the fact that the destination address resides on some other interface. It has the capacity to block the incoming flow of data as well. Today Learning bridges have been introduced that build a list of the MAC addresses on the interface by observing the traffic on the network. This is a leap in the development field of manually recording of MAC addresses.

Routers

Routers are network layer devices and are particularly identified as Layer- 3 devices of the OSI Model. They process logical addressing information in the Network header of a packet such as IP Addresses.

Router is used to create larger complex networks by complex traffic routing. It has the ability to connect dissimilar LANs on the same protocol. It also has the ability to limit the flow of broadcasts. A router primarily comprises of a hardware device or a system of the computer which has more than one network interface and routing software.

Functionality:

When a router receives the data, it determines the destination address by reading the header of the packet. Once the address is determined, it searches in its routing table to get know how to reach the destination and then forwards the packet to the higher hop on the route. The hop could be the final destination or another router.

Routing tables play a very pivotal role in letting the router makes a decision. Thus a routing table is ought to be updated and complete. The two ways through which a router can receive information are:

- Static Routing: In static routing, the routing information is fed into the routing tables manually. It does not only become a time-taking task but gets prone to errors as well. The manual updating is also required in case of statically configured routers when change in the topology of the network or in the layout takes place. Thus static routing is feasible for tinniest environments with minimum of one or two routers.
- Dynamic Routing: For larger environment dynamic routing proves to be the practical solution. The process involves use of peculiar routing protocols to hold communication. The purpose of these protocols is to enable the other routers to transfer information about to other routers, so that the other routers can build their own routing tables.

Gateway

Gateway is a device which is used to connect multiple networks and passes packets from one packet to the other network. Acting as the 'gateway' between different networking systems or computer programs, a gateway is a device which forms a link between them. It allows the computer programs, either on the same computer or on different computers to share information across the network through protocols. A router is also a gateway, since it interprets data from one network protocol to another.

Others device such as bridge converts the data into different forms between two networking systems. Then a software application converts the data from one format into another. Gateway is a viable tool to translate the data format, although the data itself remains unchanged. Gateway might be installed in some other device to add its functionality into another.

4.2. ADDRESSING

An IP address is a unique numerical value that is used to identify a computer on network. There are two kinds of IP addresses, public (also called globally unique IP addresses) and private. Public IP addresses are assigned by the Internet Assigned Numbers Authority (IANA).

A network address is an identifier for a node or network interface of a telecommunications network.

Network addresses are often designed to be unique across the network, although some networks allow for local or private addresses that may not be unique.

Four levels of addresses are used in an internet employing the TCP/IP protocols: physical (link) addresses, logical (IP) addresses, port addresses, and specific addresses.

Figure: Addresses in TCP/IP

Each address is related to a specific layer in the TCP/IP architecture, as shown in Figure

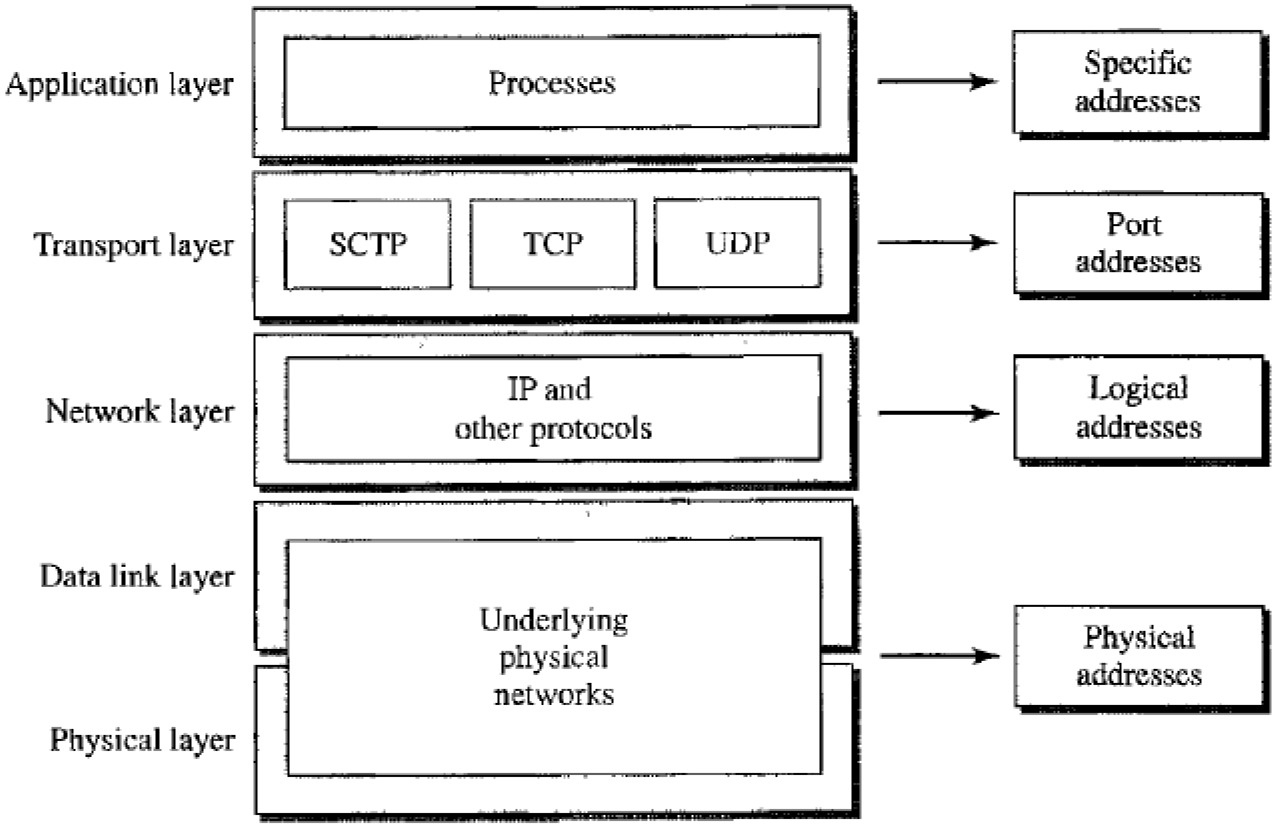


Figure: Relationship of Layers and Addresses in TCP/IP

Physical Addresses

The physical address, also known as the link address, is the address of a node as defined by its LAN or WAN. It is included in the frame used by the data link layer. It is the lowest-level address.

The physical addresses have authority over the network (LAN or WAN). The size and format of these addresses vary depending on the network. For example, Ethernet uses a 6-byte (48-bit) physical address that is imprinted on the network interface card (NIC). LocalTalk (Apple), however, has a 1-byte dynamic address that changes each time the station comes up.

07:01:02:01:2C:4B
A 6-byte (12 hexadecimal digits) physical address

Logical Addresses

Logical addresses are necessary for universal communications that are independent of underlying physical networks. Physical addresses are not adequate in an internetwork environment where different networks can have different address formats. A universal addressing system is needed in which each host can be identified uniquely, regardless of the underlying physical network.

The logical addresses are designed for this purpose. A logical address in the Internet is currently a 32-bit address that can uniquely define a host connected to the Internet. No two publicly addressed and visible hosts on the Internet can have the same IP address.

Port Addresses

The IP address and the physical address are necessary for a quantity of data to travel from a source to the destination host. However, arrival at the destination host is not the final objective of data communications on the Internet. A system that sends nothing but data from one computer to another is not complete. Today, computers are devices that can run multiple processes at the same time. The end objective of Internet communication is a process communicating with another process. For example, computer A can communicate with computer C by using TELNET. At the same time, computer A communicates with computer B by using the File Transfer Protocol (FTP). For these processes to receive data simultaneously, we need a method to label the different processes. In other words, they need addresses. In the TCP/IP architecture, the label assigned to a process is called a port address. A port address in TCP/IP is 16 bits in length.

Internet Address

An Internet Protocol address (IP address) is a numerical label assigned to each device (e.g., computer, printer) participating in a computer network that uses the Internet Protocol for communication. An IP address serves two principal functions: host or network interface identification and location addressing. Its role has been characterized as follows: "A name indicates what we seek. An address indicates where it is. A route indicates how to get there."

The designers of the Internet Protocol defined an IP address as a 32-bit number and this system, known as Internet Protocol Version 4 (IPv4), is still in use today. However, because of the growth of the Internet and the predicted depletion of available addresses, a new version of IP (IPv6), using 128 bits for the address, was developed in 1995. IPv6 was standardized as RFC 2460 in 1998, and its deployment has been ongoing since the mid-2000s.

IP addresses are usually written and displayed in human-readable notations, such as 172.16.254.1 (IPv4), and 2001:db8:0:1234:0:567:8:1 (IPv6).

The organization that doles out IP addresses to the world reserves a range of IP addresses for private networks. Private networks can use IP addresses anywhere in the following ranges:

- 192.168.0.0 - 192.168.255.255 (65,536 IP addresses)
- 172.16.0.0 - 172.31.255.255 (1,048,576 IP addresses)
- 10.0.0.0 - 10.255.255.255 (16,777,216 IP addresses)

The assumption is that these private address ranges are not directly

Classful Address

A classful network is a network addressing architecture used in the Internet from 1981 until the introduction of Classless Inter-Domain Routing in 1993. The method divides the address space for Internet Protocol Version 4 (IPv4) into five address classes by address range.

With an IPv4 IP address, there are five classes of available IP ranges: Class A, Class B, Class C, Class D and Class E, while only A, B, and C are commonly used. Each class allows for a range of valid IP addresses, shown in the following table.

Class	Address Range	Supports
Class A	1.0.0.1 to 126.255.255.254	Supports 16 million hosts on each of 127 networks.
Class B	128.1.0.1 to 191.255.255.254	Supports 65,000 hosts on each of 16,000 networks.
Class C	192.0.1.1 to 223.255.254.254	Supports 254 hosts on each of 2 million networks.
Class D	224.0.0.0 to 239.255.255.255	Reserved for multicast groups.
Class E	240.0.0.0 to 254.255.255.254	Reserved for future use, or Research and Development Purposes.

Ranges 127.x.x.x are reserved for the loopback or localhost, for example, 127.0.0.1 is the loopback address. Range 255.255.255.255 broadcasts to all hosts on the local network.

4.3. SUBNETTING

The practice of dividing a network into multiple subnetworks is called subnetting. Computers that belong to a subnet are addressed with a common, identical, most-significant bit-group in their IP address.

Subnetting is a process of breaking large network in small networks known as subnets. Subnetting happens when we extend default boundary of subnet mask. Basically we borrow host bits to create networks. Let's take a example
Being a network administrator you are asked to create two networks, each will host 30 systems.
Single class C IP range can fulfill this requirement, still you have to purchase 2 class C IP range, one for each network. Single class C range provides 256 total addresses and we need only 30 addresses, this will waste 226 addresses. These unused addresses would make additional route advertisements slowing down the network.

Advantage of Subnetting

- Subnetting breaks large network in smaller networks and smaller networks are easier to manage.
- Subnetting reduces network traffic by removing collision and broadcast traffic, that overall improve performance.
- Subnetting allows you to apply network security polices at the interconnection between subnets.
- Subnetting allows you to save money by reducing requirement for IP range.

Each IP class is equipped with its own default subnet mask which bounds that IP class to have prefixed number of Networks and prefixed number of Hosts per network. Classful IP addressing does not provide any flexibility of having less number of Hosts per Network or more Networks per IP Class.

CIDR or Classless Inter Domain Routing provides the flexibility of borrowing bits of Host part of the IP address and using them as Network in Network, called Subnet. By using subnetting, one single Class A IP address can be used to have smaller sub-networks which provides better network management capabilities.

There are two types of subnetting,

1. **FLSM (Fixed Length Subnet Mask)**

   FLSM stands for Full Length Subnet Mask. It means all the subnets are of the same size. In FLSM, the subnet mask remains the same for all the subnets.

2. **VLSM (Variable Length Subnet Mask)**

VLSM - Internet Service Providers may face a situation where they need to allocate IP subnets of different sizes as per the requirement of customer. VLSM stands for Variable Length Subnet Mask. It means the size of the subnet varies according to the needs. In VLSM, the subnet mask is different normally but it can be same for any two or more subnets depending upon the situation.

IPv4

IP addresses are displayed in dotted decimal notation, and appear as four numbers separated by dots. Each number of an IP address is made from eight individual bits known as octet. Each octet can create number value from 0 to 255. An IP address would be 32 bits long in binary divided into the two components, network component and host component. Network component is used to identify the network that the packet is intended for, and host component is used to identify the individual host on network.

IP addresses are broken into the two components:
Network component: - Defines network segment of device.
Host component: - Defines the specific device on a particular network segment

Subnet mask

Subnet mask is a 32 bits long address used to distinguish between network address and host address in IP address. Subnet mask is always used with IP address. Subnet mask has only one purpose, to identify which part of an IP address is network address and which part is host address.

- In decimal notation subnet mask value 1 to 255 represent network address and value 0 [Zero] represent host address.
- In binary notation subnet mask ON bit [1] represent network address while OFF bit [0] represent host address.

In decimal notation

```
IP address       192.168.1.10
Subnet mask      255.255.255.0
```

Network address is 192.168.1 and host address is 10.

In binary notation

```
IP address 11000000.10101000.00000001.00001010
Subnet mask 11111111.11111111.11111111.00000000
```

Network address is 11000000.10101000.00000001 and host address is 00001010

Network ID

First address of subnet is called network ID. This address is used to identify one segment or broadcast domain from all the other segments in the network.

Block Size

Block size is the size of subnet including network address, hosts addresses and broadcast address.

Broadcast ID

There are two types of broadcast, direct broadcast and full broadcast.
Direct broadcast or local broadcast is the last address of subnet and can be hear by all hosts in subnet.
Full broadcast is the last address of IP classes and can be hear by all IP hosts in network. Full broadcast address is 255.255.255.255
The main difference between direct broadcast and full broadcast is that routers will not propagate local broadcasts between segments, but they will propagate directed broadcasts.

Host Addresses

All address between the network address and the directed broadcast address is called host address for the subnet. You can assign host addresses to any IP devices such as PCs, servers, routers, and switches.

Class A Subnets

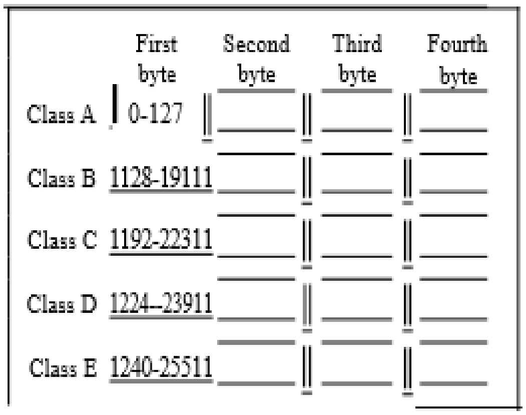
In Class A, only the first octet is used as Network identifier and rest of three octets are used to be assigned to Hosts (i.e. 16777214 Hosts per Network). To make more subnet in Class A, bits from Host part are borrowed and the subnet mask is changed accordingly.

Class B Subnets

By default, using Classful Networking, 14 bits are used as Network bits providing (2^{14}) 16384 Networks and (2^{16}-2) 65534 Hosts. Class B IP Addresses can be subnetted the same way as Class A addresses, by borrowing bits from Host bits

Class C Subnets

Class C IP addresses are normally assigned to a very small size network because it can only have 254 hosts in a network.



	First byte	Second byte	Third byte	Fourth byte
Class A	0-127			
Class B	1128-19111			
Class C	1192-22311			
Class D	1224--23911			
Class E	1240-25511			

CIDR [Classless Inter Domain Routing]

Classless Inter-Domain Routing (CIDR) is a method for allocating IP addresses and IP routing. The Internet Engineering Task Force introduced CIDR in 1993 to replace the previous addressing architecture of classful network design in the Internet. Its goal was to slow the growth of routing tables on routers across the Internet, and to help slow the rapid exhaustion of IPv4 addresses. CIDR reduced the problem of wasted address space by providing a new and more flexible way to specify network addresses in routers.

CIDR is a slash notation of subnet mask. CIDR tells us number of on bits in a network address.

- Class A has default subnet mask 255.0.0.0. that means first octet of the subnet mask has all on bits. In slash notation it would be written as /8, means address has 8 bits on.
- Class B has default subnet mask 255.255.0.0. that means first two octets of the subnet mask have all on bits. In slash notation it would be written as /16, means address has 16 bits on.
- Class C has default subnet mask 255.255.255.0. that means first three octets of the subnet mask have all on bits. In slash notation it would be written as /24, means address has 24 bits on.

4.4. ROUTING

When a device has multiple paths to reach a destination, it always selects one path by preferring it over others. This selection process is termed as Routing. Routing is done by special network devices called routers or it can be done by means of software processes. The software based routers have limited functionality and limited scope.

A router is always configured with some default route. A default route tells the router where to forward a packet if there is no route found for specific destination. In case there are multiple path existing to reach the same destination, router can make decision based on the following information:

- Hop Count
- Bandwidth
- Metric
- Prefix-length
- Delay

Routing Technique

Routing is the process of selecting best paths in a network. In packet switching networks, routing directs packet forwarding (the transit of logically addressed network packets from their source toward their ultimate destination) through intermediate nodes.

Static Vs Dynamic Routing

In static routing the routes are described by fixed paths through a data network. The routes are entered by system administrator. The whole network can be configured by using static routes.

Advantages of Static Routing

Minimal CPU/Memory overhead
No bandwidth overhead (updates are not shared between routers)
Granular control on how traffic is routed Disadvantages of Static Routing • Infrastructure changes must be manually adjusted
No "dynamic" fault tolerance if a link goes down
Impractical on large network

Dynamic routing protocols are the applications which discover network destinations dynamically. Routers will communicate the adjacent routers which informs the network to which each router is connected. These routers adjusts automatically in a network when traffic changes.

Advantages of Dynamic Routing

Simpler to configure on larger networks
Will dynamically choose a different (or better) route if a link goes down
Ability to load balance between multiple links

Disadvantages of Dynamic Routing

Updates are shared between routers, thus consuming bandwidth

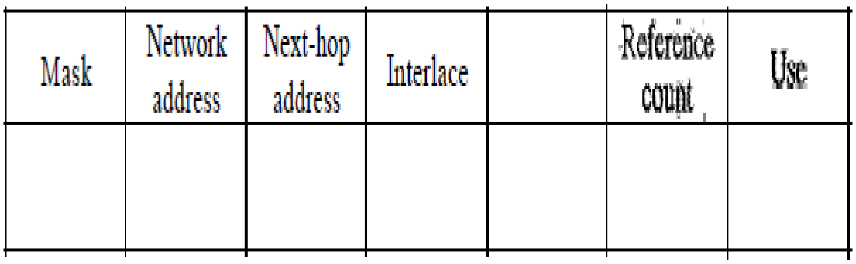
Static routing manually sets up the optimal paths between the source and the destination computers. On the other hand, the dynamic routing uses dynamic protocols to update the routing table and to find the optimal path between the source and the destination computers.

- The static routing is suitable for very small networks and they cannot be used in large networks. As against this, dynamic routing is used for larger networks. The manual routing has no specific routing algorithm. The dynamic routers are based on various routing algorithms like OSPF (Open Shortest Path First), IGRP (Interior Gateway Routing Protocol) and RIP (Routing Information Protocol).
- The static routing is the simplest way of routing the data packets from a source to a destination in a network. The dynamic routing uses complex algorithms for routing the data packets.
- The static routing has the advantage that it requires minimal memory. Dynamic router, however, have quite a few memory overheads, depending on the routing algorithms used.
- The network administrator finds out the optimal path and makes the changes in the routing table in the case of static routing. In the dynamic routing algorithm, the algorithm and the protocol is responsible for routing the packets and making the changes accordingly in the routing table.

Routing Table

A routing table is a set of rules, often viewed in table format that is used to determine where data packets traveling over an Internet Protocol (IP) network will be directed. All IP-enabled devices, including routers and switches, use routing tables.

A routing table contains the information necessary to forward a packet along the best path toward its destination. Each packet contains information about its origin and destination. When a packet is received, a network device examines the packet and matches it to the routing table entry providing the best match for its destination. The table then provides the device with instructions for sending the packet to the next hop on its route across the network.



Mask	Network address	Next-hop address	Interlace		Reference count	Use

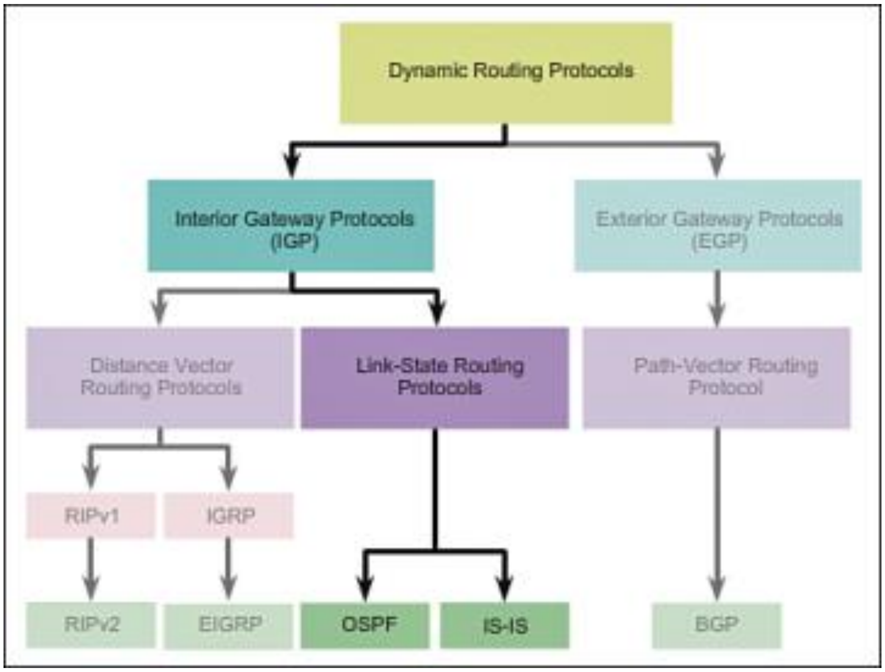
A basic routing table includes the following information:

- Destination: The IP address of the packet's final destination
- Next hop: The IP address to which the packet is forwarded
- Interface: The outgoing network interface the device should use when forwarding the packet to the next hop or final destination
- Metric: Assigns a cost to each available route so that the most cost-effective path can be chosen
- Routes: Includes directly-attached subnets, indirect subnets that are not attached to the device but can be accessed through one or more hops, and default routes to use for certain types of traffic or when information is lacking.

Routing tables can be maintained manually or dynamically. Tables for static network devices do not change unless a network administrator manually changes them. In dynamic routing, devices build and maintain their routing tables automatically by using routing protocols to exchange information about the surrounding network topology. Dynamic routing tables allow devices to "listen" to the network and respond to occurrences like device failures and network congestion.

4.5. ROUTING PROTOCOL

A routing protocol specifies how routers communicate with each other, disseminating information that enables them to select routes between any two nodes on a computer network. Routing algorithms determine the specific choice of route. Each router has a priori knowledge only of networks attached to it directly.


Dynamic Routing Protocols
Interior Gateway Protocols (IGP)
Exterior Gateway Protocols (EGP)
Distance Vector Routing Protocols
Link-State Routing Protocols
Path-Vector Routing Protocol
RIPv1
IGRP
RIPv2
EIGRP
OSPF
IS-IS
BGP

ROUTING INFORMATION PROTOCOLS (RIP)

RIP (Routing Information Protocol) is a forceful protocol type used in local area network and wide area network. RIP (Routing Information Protocol) type is categorized interior gateway protocol within the use of distance vector algorithm. Routing information protocols defined in 1988. It also has version 2 and nowadays both versions are in use. Technically it is outdated by more sophisticated techniques such as (OSPF) and the OSI protocol IS-IS.

Each RIP router maintains a routing table, which is a list of all the destinations (networks) it knows how to reach, along with the distance to that destination. RIP uses a distance vector algorithm to decide which path to put a packet on to get to its destination. It stores in its routing table the distance for each network it knows how to reach, along with the address of the "next hop" router -- another router that is on one of the same networks -- through which a packet has to travel to get to that destination. If it receives an update on a route, and the new path is shorter, it will update its table entry with the length and next-hop address of the shorter path; if the new path is longer, it will wait through a "hold-down" period to see if later updates reflect the higher value as well, and only update the table entry if the new, longer path is stable.

Using RIP, each router sends its entire routing table to its closest neighbors every 30 seconds. (The neighbors are the other routers to which this router is connected directly -- that is, the other routers on the same network segments this router is on.) The neighbors in turn will pass the information on to their nearest neighbors, and so on, until all RIP hosts within the network have the same knowledge of routing paths, a state known as convergence.

OPEN SHORTEST PATH FIRST (OSPF)

Open Shortest Path First (OSPF) is an active routing protocol used in internet protocol. Particularly it is a link state routing protocol and includes into the group of interior gateway protocol. Open Shortest Path First (OSPF) operating inside a distinct autonomous system.

Routers connect networks using the Internet Protocol (IP), and OSPF (Open Shortest Path First) is a router protocol used to find the best path for packets as they pass through a set of connected networks. OSPF is designated by the Internet Engineering Task Force (IETF) as one of several Interior Gateway Protocols (IGPs) -- that is, protocols aimed at traffic moving around within a larger autonomous system network like a single enterprise's network, which may in turn be made up of many separate local area networks linked through routers.

The OSPF routing protocol has largely replaced the older Routing Information Protocol (RIP) in corporate networks. Using OSPF, a router that learns of a change to a routing table (when it is reconfigured by network staff, for example) or detects a change in the network immediately multicasts the information to all other OSPF hosts in the network so they will all have the same routing table information. Unlike RIP, which requires routers to send the entire routing table to neighbors every 30 seconds, OSPF sends only the part that has changed and only when a change has taken place? When routes change -- sometimes due to equipment failure -- the time it takes OSPF routers to find a new path between endpoints with no loops (which is called "open") and that minimizes the length of the path is called the convergence time.

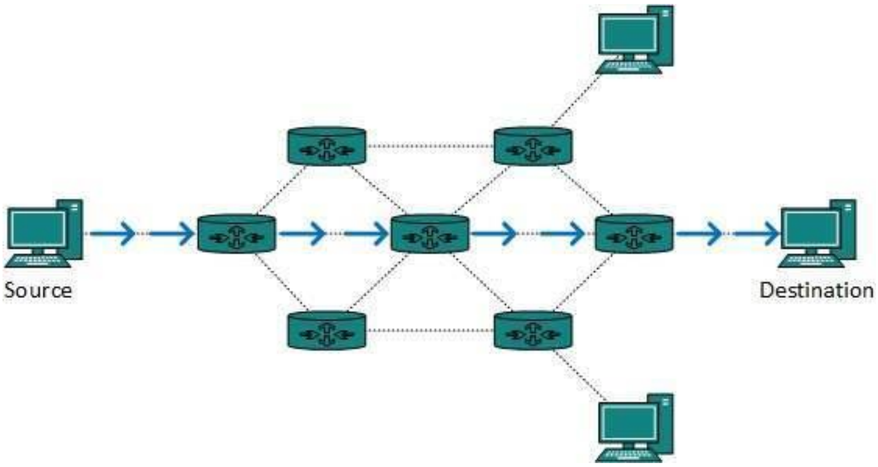
Rather than simply counting the number of router hops between hosts on a network, as RIP does, OSPF bases its path choices on "link states" that take into account additional network information, including IT-assigned cost metrics that give some paths higher assigned costs. For example, a satellite link may be assigned higher cost than a wireless WAN link, which in turn may be assigned higher cost than a metro Ethernet link.

BORDER GATEWAY PROTOCOL (BGP)

Border Gateway Protocol (BGP) are the core routing protocol of the internet and responsible to maintain a table of Internet protocol networks which authorize network reaching capability between AS. The Border Gateway Protocol (BGP) expressed as path vector protocol. BGP router maintains a standard routing table used to direct packets in transit. This table is used in conjunction with a separate routing table, known as the routing information base (RIB), which is a data table stored on a server on the BGP router. The RIB contains route information both from directly connected external peers, as well as internal peers, and continually updates the routing table as changes occurs. BGP is based on TCP/IP and uses client-server topology to communicate routing information, with the client-server initiating a BGP session by sending a request to the server.

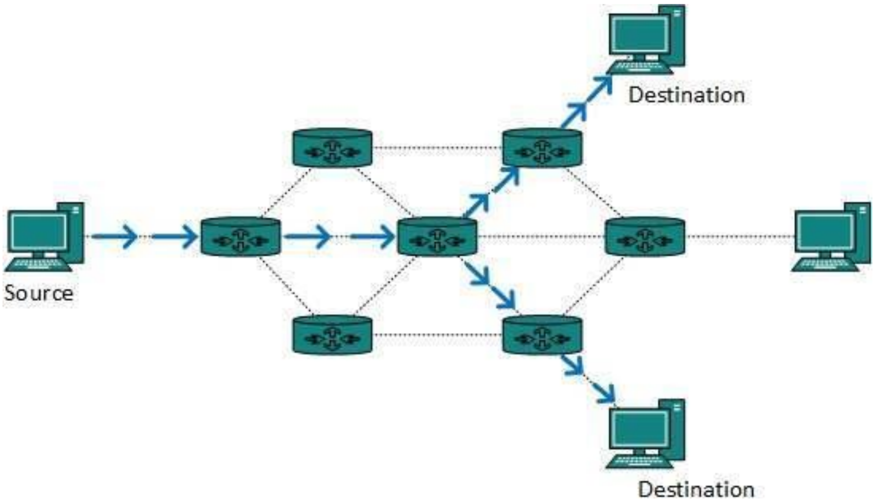
UNCAST ROUTING

Most of the traffic on the internet and intranets known as unicast data or unicast traffic is sent with specified destination. Routing unicast data over the internet is called unicast routing. It is the simplest form of routing because the destination is already known. Hence the router just has to look up the routing table and forward the packet to next hop.



MULTICAST ROUTING

Multicast routing is special case of broadcast routing with significance difference and challenges. In broadcast routing, packets are sent to all nodes even if they do not want it. But in Multicast routing, the data is sent to only nodes which wants to receive the packets.



The router must know that there are nodes, which wish to receive multicast packets (or stream) then only it should forward. Multicast routing works spanning tree protocol to avoid looping.

Multicast routing also uses reverse path Forwarding technique, to detect and discard duplicates and loops.

4.6. ROUTING ALGORITHM

ADDRESS RESOLUTION PROTOCOL (ARP)

The address resolution protocol (ARP) is a protocol used by the Internet Protocol (IP) , specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet.

The term address resolution refers to the process of finding an address of a computer in a network. The address is "resolved" using a protocol in which a piece of information is sent by a client process executing on the local computer to a server process executing on a remote computer. The information received by the server allows the server to uniquely identify the network system for which the address was required and therefore to provide the required address. The address resolution procedure is completed when the client receives a response from the server containing the required address.

There are four types of ARP messages that may be sent by the ARP protocol. These are identified by four values in the "operation" field of an ARP message. The types of message are:

1. ARP request
2. ARP reply
3. RARP request
4. RARP reply

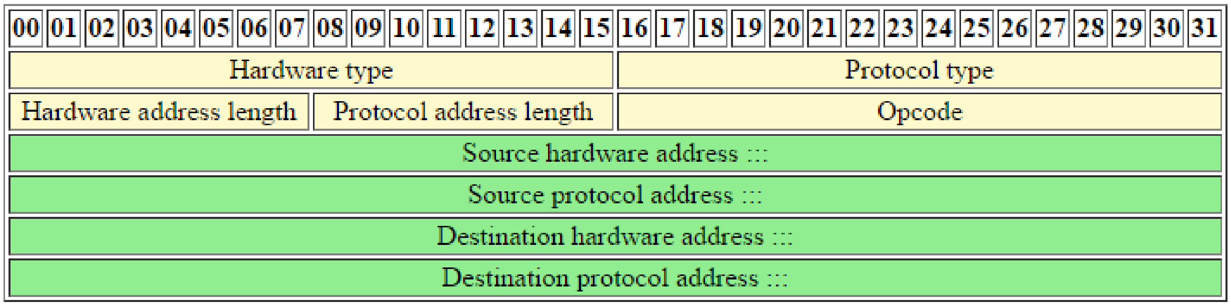
To reduce the number of address resolution requests, a client normally caches resolved addresses for a (short) period of time. The ARP cache is of a finite size, and would become full of incomplete and obsolete entries for computers that are not in use if it was allowed to grow without check. The ARP cache is therefore periodically flushed of all entries. This deletes unused entries and frees space in the cache. It also removes any unsuccessful attempts to contact computers which are not currently running.

If a host changes the MAC address it is using, this can be detected by other hosts when the cache entry is deleted and a fresh ARP message is sent to establish the new association. The use of gratuitous ARP (e.g. triggered when the new NIC interface is enabled with an IP address) provides a more rapid update of this information.

RARP

RARP (Reverse Address Resolution Protocol) is a protocol by which a physical machine in a local area network can request to learn its IP address from a gateway server's Address Resolution Protocol (ARP) table or cache. A network administrator creates a table in a local area network's gateway router that maps the physical machine (or Media Access Control - MAC address) addresses to corresponding Internet Protocol addresses. When a new machine is set up, its RARP client program requests from the RARP server on the router to be sent its IP address. Assuming that an entry has been set up in the router table, the RARP server will return the IP address to the machine which can store it for future use.

RARP packet:



00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Hardware type																Protocol type															
Hardware address length								Protocol address length								Opcode															
Source hardware address :::																															
Source protocol address :::																															
Destination hardware address :::																															
Destination protocol address :::																															

IP

The Internet Protocol (IP) is the method or protocol by which data is sent from one computer to another on the Internet. Each computer (known as a host) on the Internet has at least one IP address that uniquely identifies it from all other computers on the Internet.

When you send or receive data (for example, an e-mail note or a Web page), the message gets divided into little chunks called packets. Each of these packets contains both the sender's Internet address and the receiver's address. Any packet is sent first to a gateway computer that understands a small part of the Internet. The gateway computer reads the destination address and forwards the packet to an adjacent gateway that in turn reads the destination address and so forth across the Internet until one gateway recognizes the packet as belonging to a computer within its immediate neighborhood or domain. That gateway then forwards the packet directly to the computer whose address is specified.

Because a message is divided into a number of packets, each packet can, if necessary, be sent by a different route across the Internet. Packets can arrive in a different order than the order they were sent in. The Internet Protocol just delivers them. It's up to another protocol, the Transmission Control Protocol (TCP) to put them back in the right order.

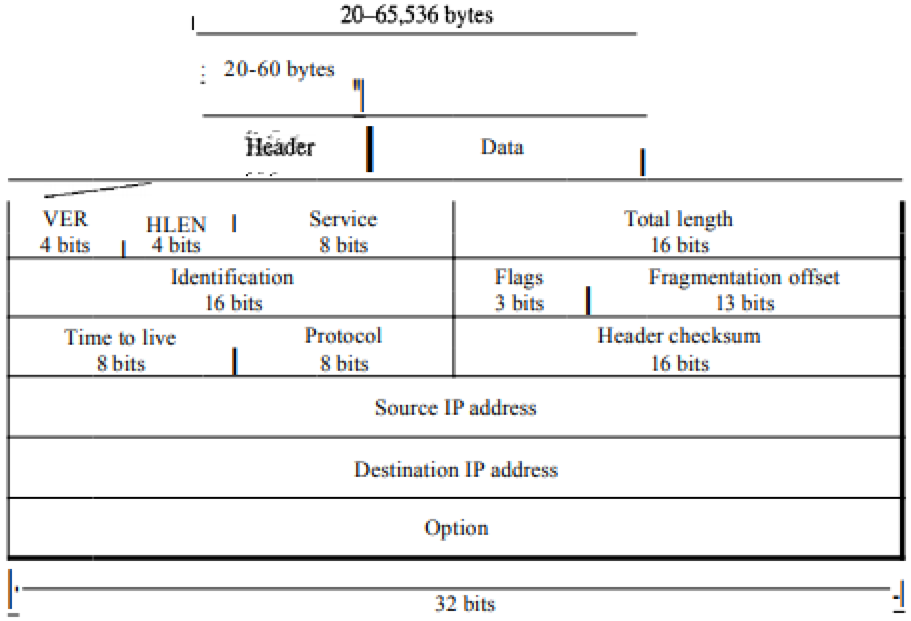
IP is a connectionless protocol, which means that there is no continuing connection between the end points that are communicating. Each packet that travels through the Internet is treated as an independent unit of data without any relation to any other unit of data. (The reason the packets do get put in the right order is because of TCP, the connection-oriented protocol that keeps track of the packet sequence in a message.) In the Open Systems Interconnection (OSI) communication model, IP is in layer 3, the Networking Layer.

The most widely used version of IP today is Internet Protocol Version 4 (IPv4). However, IP Version 6 (IPv6) is also beginning to be supported. IPv6 provides for much longer addresses and therefore for the possibility of many more Internet users. IPv6 includes the capabilities of IPv4 and any server that can support IPv6 packets can also support IPv4 packets.

IPV4

Packets in the IPv4 layer are called Datagrams.

IPV4 Datagram format.



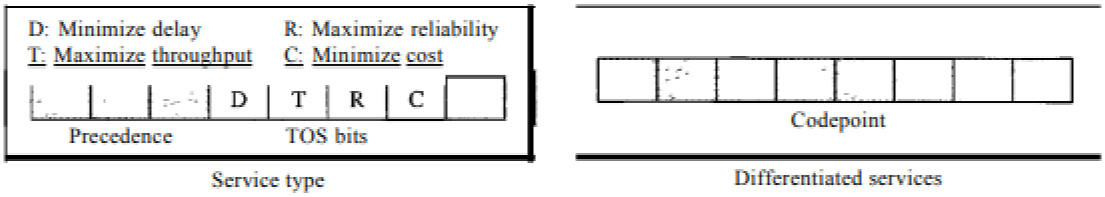
A datagram is a variable-length packet consisting of two parts: header and data. The header is 20 to 60 bytes in length and contains information essential to routing and delivery. It is customary in TCP/IP to show the header in 4-byte sections. A brief description of each field is in order.

o Version (VER). This 4-bit field defines the version of the IPv4 protocol. Currently the version is 4. However, version 6 (or IPng) may totally replace version 4 in the future. This field tells the IPv4 software running in the processing machine that the datagram has the format of version 4. All fields must be interpreted as specified in the fourth version of the protocol. If the machine is using some other version of IPv4, the datagram is discarded rather than interpreted incorrectly.

o Header length (HLEN). This 4-bit field defines the total length of the datagram header in 4-byte words. This field is needed because the length of the header is variable (between 20 and 60 bytes). When there are no options, the header length is 20 bytes, and the value of this field is 5 (5 x 4 = 20). When the option field is at its maximum size, the value of this field is 15 (15 x 4 = 60).

o Services. IETF has changed the interpretation and name of this 8-bit field. This field, previously called service type, is now called differentiated services. We show both interpretations in Figure

Figure 20.6 *Service type or differentiated services*

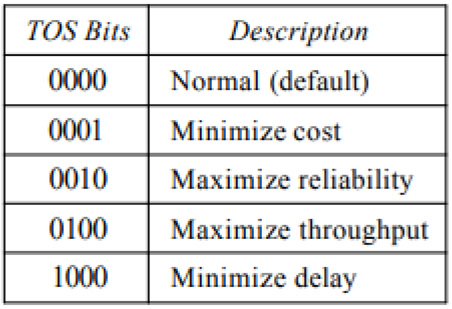


1. Service Type In this interpretation, the first 3 bits are called precedence bits. The next 4 bits are called type of service (TOS) bits, and the last bit is not used.

   a. Precedence is a 3-bit subfield ranging from 0 (000 in binary) to 7 (111 in binary). The precedence defines the priority of the datagram in issues such as congestion. If a router is congested and needs to discard some datagrams, those datagrams with lowest precedence are discarded first. Some datagrams in the Internet are more important than others. For example, a datagram used for network management is much more urgent and important than a datagram containing optional information for a group.

   b. TOS bits is a 4-bit subfield with each bit having a special meaning. Although a bit can be either 0 or 1, one and only one of the bits can have the value of 1 in each datagram. The bit patterns and their interpretations are given in Table 20.1. With only 1 bit set at a time, we can have five different types of services.

Table 20.1 *Types ofservice*



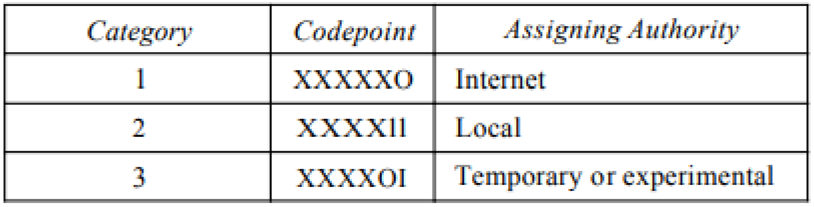
TOS Bits	*Description*
0000	Normal (default)
0001	Minimize cost
0010	Maximize reliability
0100	Maximize throughput
1000	Minimize delay

2. Differentiated Services In this interpretation, the first 6 bits make up the codepoint subfield, and the last 2 bits are not used. The codepoint subfield can be used in two different ways.

a. When the 3 rightmost bits are Os, the 3 leftmost bits are interpreted the same as the precedence bits in the service type interpretation. In other words, it is compatible with the old interpretation.

b. When the 3 rightmost bits are not all Os, the 6 bits define 64 services based on the priority assignment by the Internet or local authorities according to Table 20.3. The first category contains 32 service types; the second and the third each contain 16. The first category (numbers 0, 2,4, ... ,62) is assigned by the Internet authorities (IETF). The second category (3, 7, 11, 15, , 63) can be used by local authorities (organizations). The third category (1, 5, 9, ,61) is temporary and can be used for experimental purposes. Note that the numbers are not contiguous. If they were, the first category would range from 0 to 31, the second from 32 to 47, and the third from 48 to 63. This would be incompatible with the TOS interpretation because XXXOOO (which includes 0, 8, 16, 24, 32, 40, 48, and 56) would fall into all three categories. Instead, in this assignment method all these services belong to category 1. Note that these assignments have not yet been finalized.

Table 20.3 *Values for codepoints*



Category	*Codepoint*	*Assigning Authority*
1	XXXXXO	Internet
2	XXXXll	Local
3	XXXXOI	Temporary or experimental

o Total length. This is a In-bit field that defines the total length (header plus data) of the IPv4 datagram in bytes. To find the length of the data coming from the upper layer, subtract the header length from the total length. The header length can be found by multiplying the value in the HLEN field by 4. Length of data =total length - header length since the field length is 16 bits, the total length of the IPv4 datagram is limited to 65,535 (216 - 1) bytes, of which 20 to 60 bytes are the header and the rest is data from the upper layer.

o Identification. This field is used in fragmentation (discussed in the next section).

o Flags. This field is used in fragmentation (discussed in the next section).

o Fragmentation offset. This field is used in fragmentation (discussed in the next section).

o Time to live. A datagram has a limited lifetime in its travel through an internet. This field was originally designed to hold a timestamp, which was decremented by each visited router. The datagram was discarded when the value became zero. However, for this scheme, all the machines must have synchronized clocks and must know how long it takes for a datagram to go from one machine to another. Today, this field is used mostly to control the maximum number of hops (routers) visited by the datagram. When a source host sends the datagram, it stores a number in this field. This value is approximately 2 times the maximum number of routes between any two hosts. Each router that processes the datagram decrements this number by 1. If this value, after being decremented, is zero, the router discards the datagram. This field is needed because routing tables in the Internet can become corrupted. A datagram may travel between two or more routers for a long time without ever getting delivered to the destination host. This field limits the lifetime of a datagram. Another use of this field is to intentionally limit the journey of the packet. For example, if the source wants to confine the packet to the local network, it can store 1 in this field. When the packet arrives at the first router, this value is decremented to 0, and the datagram is discarded.

o Protocol. This 8-bit field defines the higher-level protocol that uses the services of the IPv4 layer. An IPv4 datagram can encapsulate data from several higher-level protocols such as TCP, UDP, ICMP, and IGMP. This field specifies the final destination protocol to which the IPv4 datagram is delivered. In other words, since the IPv4 protocol carries data from different other protocols, the value of this field helps the receiving network layer know to which protocol the data belong o Checksum. The checksum concept and its calculation are discussed later in this chapter.

o Source address. This 32-bit field defines the IPv4 address of the source. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.

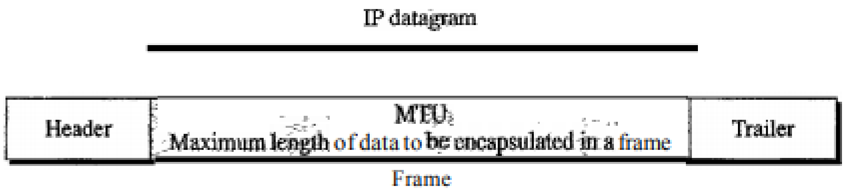
o Destination address. This 32-bit field defines the IPv4 address of the destination. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.

Fragmentation

A datagram can travel through different networks. Each router decapsulates the IPv4 datagram from the frame it receives, processes it, and then encapsulates it in another frame. The format and size of the received frame depend on the protocol used by the physical network through which the frame has just traveled. The format and size of the sent frame depend on the protocol used by the physical network through which the frame is going to travel. For example, if a router connects a LAN to a WAN, it receives a frame in the LAN format and sends a frame in the WAN format.

Maximum Transfer Unit (MTU)

Each data link layer protocol has its own frame format in most protocols. One of the fields defined in the format is the maximum size of the data field. In other words, when a datagram is encapsulated in a frame, the total size of the datagram must be less than this maximum size, which is defined by the restrictions imposed by the hardware and software used in the network (see Figure). The value of the MTU depends on the physical network protocol. Table 20.5 shows the values for some protocols.



To make the IPv4 protocol independent of the physical network, the designers decided to make the maximum length of the IPv4 datagram equal to 65,535 bytes. This makes transmission more efficient if we use a protocol with an MTU of this size. However, for other physical networks, we must divide the datagram to make it possible to pass through these networks. This is called **fragmentation**.

The source usually does not fragment the IPv4 packet. The transport layer will instead segment the data into a size that can be accommodated by IPv4 and the data link layer in use. When a datagram is fragmented, each fragment has its own header with most of the fields repeated, but with some changed. A fragmented datagram may itself be fragmented if it encounters a network with an even smaller MTU. In other words, a datagram can be fragmented several times before it reaches the final destination.

In IPv4, a datagram can be fragmented by the source host or any router in the path although there is a tendency to limit fragmentation only at the source. The reassembly of the datagram, however, is done only by the destination host because each fragment becomes an independent datagram. Whereas the fragmented datagram can travel through different routes, and we can never control or guarantee which route a fragmented datagram may take, all the fragments belonging to the same datagram should finally arrive at the destination host. So it is logical to do the reassembly at the final destination. An even stronger objection to reassembling packets during the transmission is the loss of efficiency it incurs. When a datagram is fragmented, required parts of the header must be copied by all fragments. The option field mayor may not be copied, as we will see in the next section.

The host or router that fragments a datagram must change the values of three fields:

Fields Related to Fragmentation

The fields that are related to fragmentation and reassembly of an IPv4 datagram are the identification, flags, and fragmentation offset fields.

o Identification. This 16-bit field identifies a datagram originating from the source host. The combination of the identification and source IPv4 address must uniquely define a datagram as it leaves the source host. To guarantee uniqueness, the IPv4 protocol uses a counter to label the datagrams. The counter is initialized to a positive number. When the IPv4 protocol sends a datagram, it copies the current value of the counter to the identification field and increments the counter by 1. As long as the counter is kept in the main memory, uniqueness is guaranteed. When a datagram is fragmented, the value in the identification field is copied to all fragments. In other words, all fragments have the same identification number, the same as the original datagram. The identification number helps the destination in reassembling the datagram. It knows that all fragments having the same identification value must be assembled into one datagram.

o Flags. This is a 3-bit field. The first bit is reserved. The second bit is called the do not fragment bit. If its value is 1, the machine must not fragment the datagram. If it cannot pass the datagram through any available physical network, it discards the datagram and sends an ICMP error message to the source host. If its value is 0, the datagram can be fragmented if necessary. The third bit is called the more fragment bit. If its value is 1, it means the datagram is not the last fragment; there are more fragments after this one. If its value is 0, it means this is the last or only fragment (see Figure).

o Fragmentation offset. This 13-bit field shows the relative position of this fragment with respect to the whole datagram. It is the offset of the data in the original datagram measured in units of 8 bytes. Figure 20.11 shows a datagram with a data size of 4000 bytes fragmented into three fragments. The bytes in the original datagram are numbered 0 to 3999. The first fragment carries bytes 0 to 1399. The offset for this datagram is 0/8 = O. The second fragment carries bytes 1400 to 2799; the offset value for this fragment is 1400/8 = 175. Finally, the third fragment carries bytes 2800 to 3999. The offset value for this fragment is 2800/8 =350. Remember that the value of the offset is measured in units of 8 bytes. This is done because the length of the offset field is only 13 bits and cannot represent a sequence of bytes greater than 8191. This forces hosts or routers that fragment datagrams to choose a fragment size so that the first byte number is divisible by 8. Figure 20.12 shows an expanded view of the fragments in Figure 20.11. Notice the value of the identification field is the same in all fragments. Notice the value of the flags field with the more bit set for all fragments except the last. Also, the value of the offset field for each fragment is shown.

Figure 20.11 *Fragmentation example*

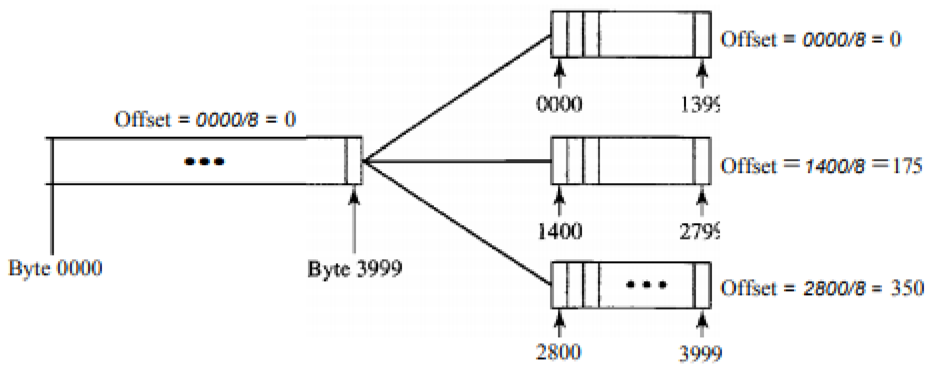
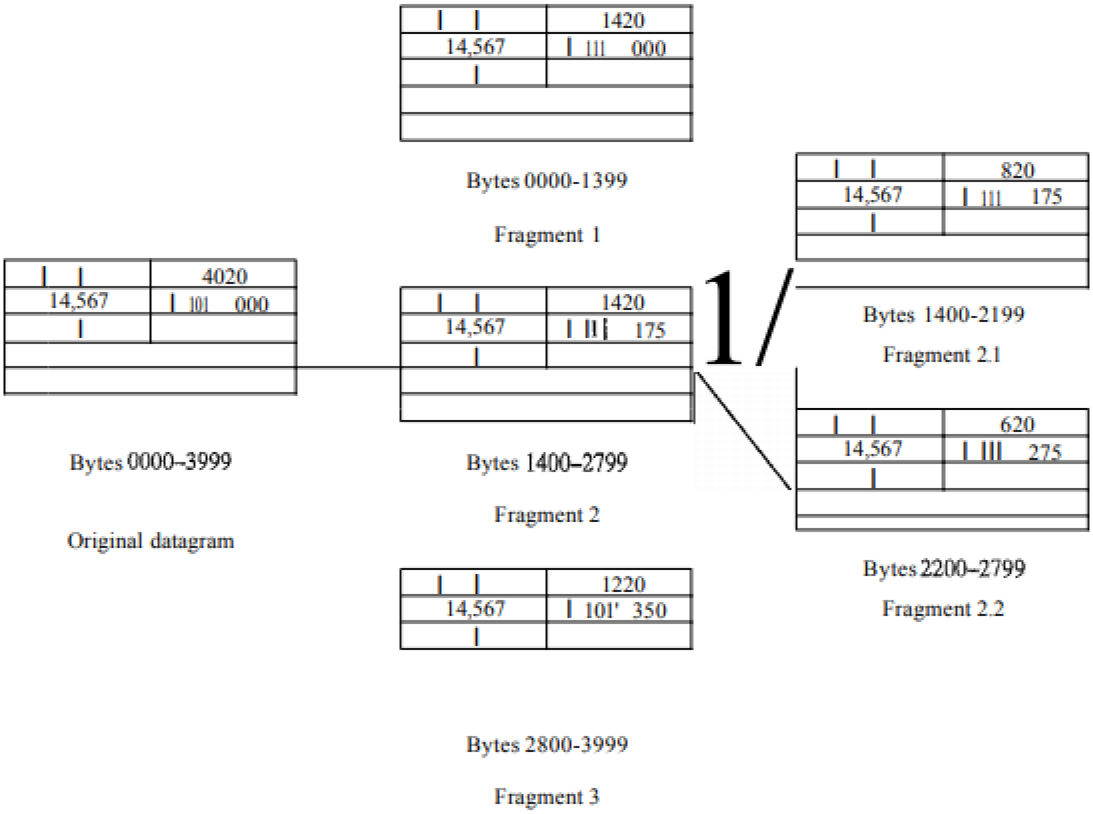


Figure 20.12 *Detailedfragmentation example*



The figure also shows what happens if a fragment itself is fragmented. In this case the value of the offset field is always relative to the original datagram. For example, in the figure, the second fragment is itself fragmented later to two fragments of 800 bytes and 600 bytes, but the offset shows the relative position of the fragments to the original data. It is obvious that even if each fragment follows a different path and arrives out of order, the final destination host can reassemble the original datagram from the fragments received (if none of them is lost) by using the following strategy: 1. The first fragment has an offset field value of zero. 2. Divide the length of the first fragment by 8. The second fragment has an offset value equal to that result. 3. Divide the total length of the first and second fragments by 8. The third fragment has an offset value equal to that result. 4. Continue the process. The last fragment has a more bit value of O.

SHORTEST PATH ROUTING:

Shortest path can be calculated only for the weighted graphs. The edges connecting two vertices can be assigned a nonnegative real number, called the weight of the edge.

The general algorithm is:

1. Initialize the array smallestWeight so that smallestWeight[u] = weights [vertex, u].

2. Set smallestWeight [vertex] = 0.

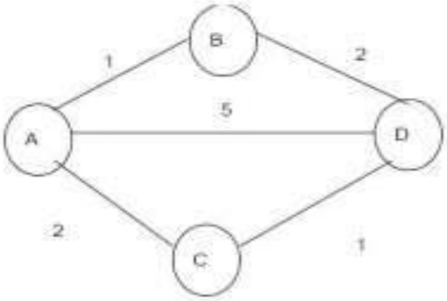
3. Find the vertex, v that is closest to vertex for which the shortest path has not been determined.

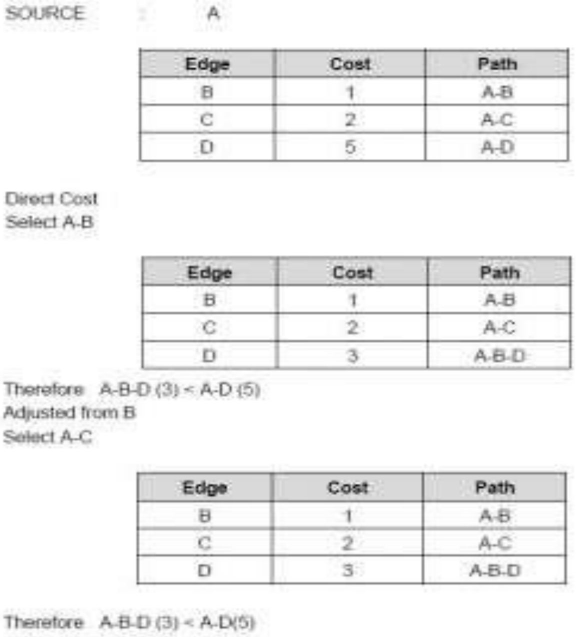
4. Mark v as the (next) vertex for which the smallest weight is found.

5. For each vertex w in G, such that the shortest path from vertex to w has not been determined and an edge (v, w) exists, if the weight of the path to w via v is smaller than its current weight, update the weight of w to the weight of v + the weight of the edge (v, w).

Because there are n vertices, repeat Steps 3 through 5, n – 1 times.

Example: Shortest Path





SOURCE : A

Edge	Cost	Path
B	1	A-B
C	2	A-C
D	5	A-D

Direct Cost
Select A-B

Edge	Cost	Path
B	1	A-B
C	2	A-C
D	3	A-B-D

Therefore A-B-D (3) < A-D (5)
Adjusted from B
Select A-C

Edge	Cost	Path
B	1	A-B
C	2	A-C
D	3	A-B-D

Therefore A-B-D (3) < A-D(5)

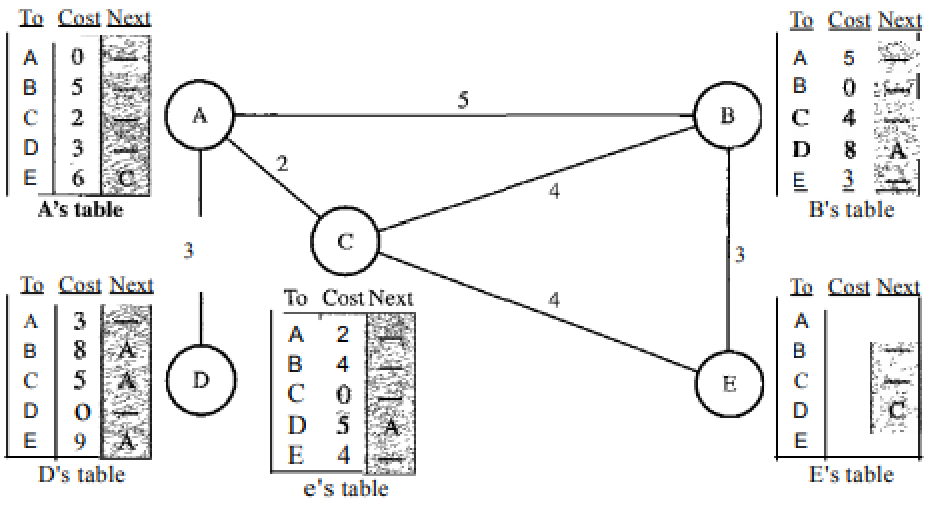
FLOODING:

- Static algorithm
- Every incoming packet is sent to outgoing line except from one it arrived
- Generates a number of duplicate packet
- Solution to have hop counter in each header and is decremented and packet is discarded when counter become 0.

DISTANCE VECTOR ROUTING

In distance vector routing, the least-cost route between any two nodes is the route with minimum distance. In this protocol, as the name implies, each node maintains a vector (table) of minimum distances to every node. The table at each node also guides the packets to the desired node by showing the next stop in the route (next-hop routing). We can think of nodes as the cities in an area and the lines as the roads connecting them. A table can show a tourist the minimum distance between cities. In Figure 22.14, we show a system of five nodes with their corresponding tables.

Figure 22.14 *Distance vector routing tables*



To	Cost	Next
A	0	—
B	5	—
C	2	—
D	3	—
E	6	C

A's table

To	Cost	Next
A	5	—
B	0	—
C	4	—
D	8	A
E	3	—

B's table

To	Cost	Next
A	3	—
B	8	A
C	5	A
D	O	—
E	9	A

D's table

To	Cost	Next
A	2	—
B	4	—
C	0	—
D	5	A
E	4	—

e's table

To	Cost	Next
A		
B		—
C		—
D		C
E		

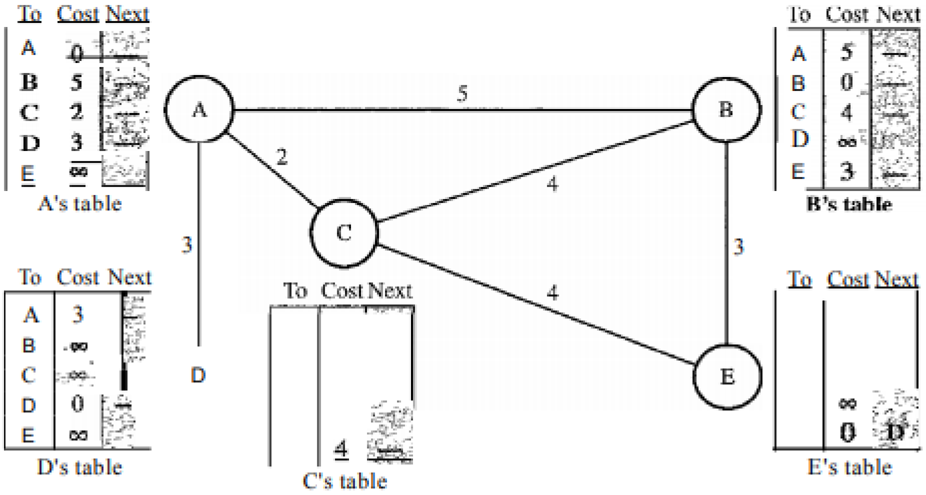
E's table

The table for node A shows how we can reach any node from this node. For example, our least cost to reach node E is 6. The route passes through C.

Initialization

The tables in Figure 22.14 are stable; each node knows how to reach any other node and the cost. At the beginning, however, this is not the case. Each node can know only the distance between itself and its immediate neighbors, those directly connected to it. So for the moment, we assume that each node can send a message to the immediate neighbors and find the distance between itself and these neighbors. Figure 22.15 shows the initial tables for each node. The distance for any entry that is not a neighbor is marked as infinite (unreachable).

Figure 22.15 *Initialization of tables in distance vector routing*



A's table

To	Cost	Next
A	0	—
B	5	—
C	2	—
D	3	—
E	∞	

B's table

To	Cost	Next
A	5	—
B	0	—
C	4	—
D	∞	
E	3	—

D's table

To	Cost	Next
A	3	
B	∞	
C	∞	
D	0	—
E	∞	

Sharing

The whole idea of distance vector routing is the sharing of information between neighbors. Although node A does not know about node E, node C does. So if node C shares its routing table with A, node A can also know how to reach node E. On the other hand, node C does not know how to reach node D, but node A does. If node A shares its routing table with node C, node C also knows how to reach node D. In other words, nodes A and C, as immediate neighbors, can improve their routing tables if they help each other. There is only one problem. How much of the table must be shared with each neighbor? A node is not aware of a neighbor's table. The best solution for each node is to send its entire table to the neighbor and let the neighbor decide what part to use and what part to discard. However, the third column of a table (next stop) is not useful for the neighbor. When the neighbor receives a table, this column needs to be replaced with the sender's name. If any of the rows can be used, the next node is the sender of the table. A node therefore can send only the first two columns of its table to any neighbor. In other words, sharing here means sharing only the first two columns

Updating

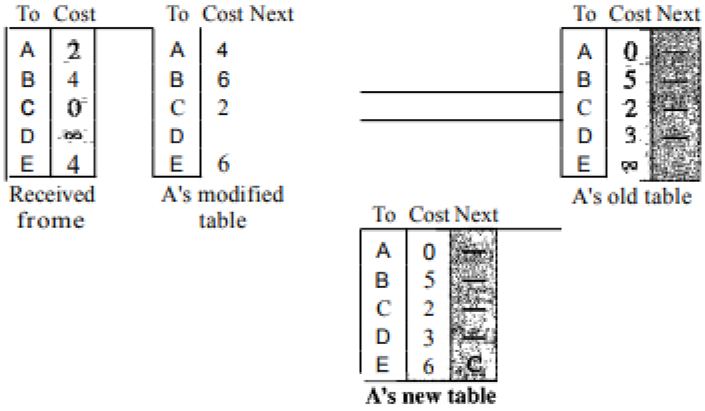
When a node receives a two-column table from a neighbor, it needs to update its routing table. Updating takes three steps:

1. The receiving node needs to add the cost between itself and the sending node to each value in the second column. The logic is clear. If node C claims that its distance to a destination is x mi, and the distance between A and C is y mi, then the distance between A and that destination, via C, is x + y mi.

2. The receiving node needs to add the name of the sending node to each row as the third column if the receiving node uses information from any row. The sending node is the next node in the route. 3. The receiving node needs to compare each row of its old table with the corresponding row of the modified version of the received table. a. If the next-node entry is different, the receiving node chooses the row with the smaller cost. If there is a tie, the old one is kept. b. If the next-node entry is the same, the receiving node chooses the new row. For example, suppose node C has previously advertised a route to node X with distance

3. Suppose that now there is no path between C and X; node C now advertises this route with a distance of infinity. Node A must not ignore this value even though its old entry is smaller. The old route does not exist anymore. The new route has a distance of infinity. Figure 22.16 shows how node A updates its routing table after receiving the partial table from node C.

Figure 22.16 *Updating in distance vector routing*



Received frome

To	Cost
A	2
B	4
C	0
D	∞
E	4

A's modified table

To	Cost	Next
A	4	
B	6	
C	2	
D		
E	6	

A's old table

To	Cost	Next
A	0	—
B	5	—
C	2	—
D	3	—
E	∞	

A's new table

To	Cost	Next
A	0	—
B	5	—
C	2	—
D	3	—
E	6	C

There are several points we need to emphasize here. First, as we know from mathematics, when we add any number to infinity, the result is still infinity. Second, the modified table shows how to reach A from A via C. If A needs to reach itself via C, it needs to go to C and come back, a distance of 4. Third, the only benefit from this updating of node A is the last entry, how to reach E. Previously, node A did not know how to reach E (distance of infinity); now it knows that the cost is 6 via C.

When to Share

The question now is, when does a node send its partial routing table (only two columns) to all its immediate neighbors? The table is sent both periodically and when there is a change in the table. Periodic Update A node sends its routing table, normally every 30 s, in a periodic update. The period depends on the protocol that is using distance vector routing. Triggered Update A node sends its two-column routing table to its neighbors anytime there is a change in its routing table. This is called a triggered update. The change can result from the following.

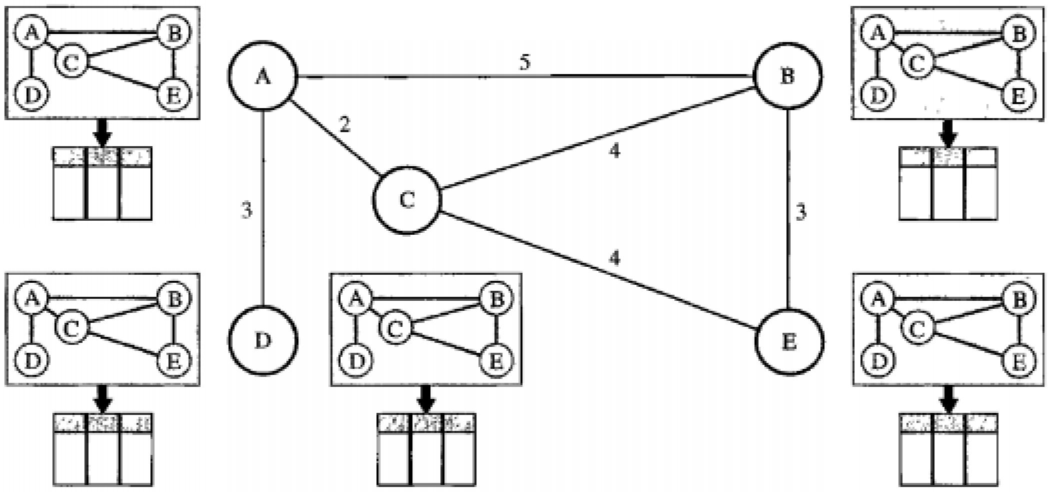
1. A node receives a table from a neighbor, resulting in changes in its own table after updating.

2. A node detects some failure in the neighboring links which results in a distance change to infinity.

LINK STATE ROUTING

Link state routing has a different philosophy from that of distance vector routing. In link state routing, if each node in the domain has the entire topology of the domain the list of nodes and links, how they are connected including the type, cost (metric), and condition of the links (up or down)-the node can use Dijkstra's algorithm to build a routing table. Figure 22.20 shows the concept.

Figure 22.20 *Concept oflink state routing*

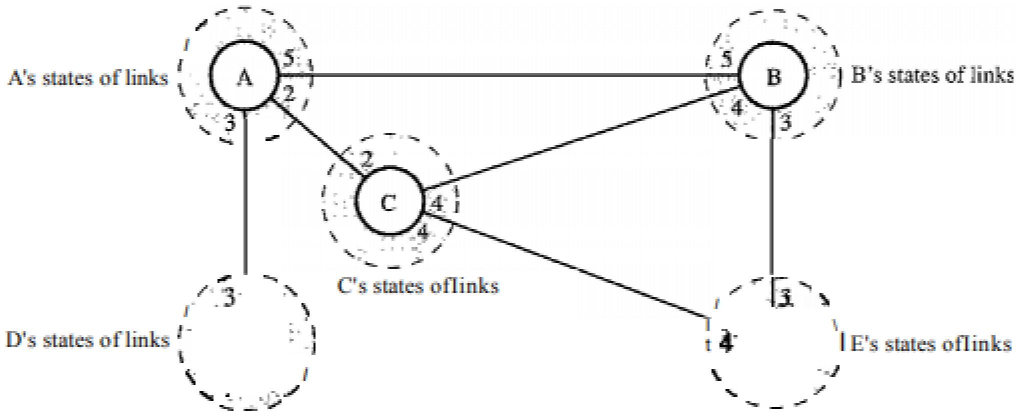


The figure shows a simple domain with five nodes. Each node uses the same topology to create a routing table, but the routing table for each node is unique because the calculations are based on different interpretations of the topology. This is analogous to a city map. While each person may have the same map, each needs to take a different route to reach her specific destination.

The topology must be dynamic, representing the latest state of each node and each link. If there are changes in any point in the network (a link is down, for example), the topology must be updated for each node. How can a common topology be dynamic and stored in each node? No node can know the topology at the beginning or after a change somewhere in the network.

Link state routing is based on the assumption that, although the global knowledge about the topology is not clear, each node has partial knowledge: it knows the state (type, condition, and cost) of its links. In other words, the whole topology can be compiled from the partial knowledge of each node. Figure 22.21 shows the same domain as in Figure 22.20, indicating the part of the knowledge belonging to each node.

Figure 22.21 *Link state knowledge*



Node A knows that it is connected to node B with metric 5, to node C with metric 2, and to node D with metric 3. Node C knows that it is connected to node A with metric 2, to node B with metric 4, and to node E with metric 4. Node D knows that it is connected only to node A with metric 3. And so on. Although there is an overlap in the knowledge, the overlap guarantees the creation of a common topology-a picture of the whole domain for each node.

Building Routing Tables

In link state routing, four sets of actions are required to ensure that each node has the routing table showing the least-cost node to every other node.

1. Creation of the states of the links by each node, called the link state packet (LSP).

2. Dissemination of LSPs to every other router, called flooding, in an efficient and reliable way.
3. Formation of a shortest path tree for each node.

4. Calculation of a routing table based on the shortest path tree. Creation of Link State Packet (LSP) A link state packet can carry a large amount of information. For the moment, however, we assume that it carries a minimum amount of data: the node identity, the list of links, a sequence number, and age. The first two, node identity and the list of links, are needed to make the topology. The third, sequence number, facilitates flooding and distinguishes new LSPs from old ones. The fourth, age, prevents old LSPs from remaining in the domain for a long time.

LSPs are generated on two occasions:

1. When there is a change in the topology of the domain. Triggering of LSP dissemination is the main way of quickly informing any node in the domain to update its topology.

2. On a periodic basis. The period in this case is much longer compared to distance vector routing. As a matter of fact, there is no actual need for this type of LSP dissemination. It is done to ensure that old information is removed from the domain. The timer set for periodic dissemination is

normally in the range of 60 min or 2 h based on the implementation. A longer period ensures that flooding does not create too much traffic on the network.

Flooding of LSPs

After a node has prepared an LSP, it must be disseminated to all other nodes, not only to its neighbors. The process is called flooding and based on the following:

1. The creating node sends a copy of the LSP out of each interface.

2. A node that receives an LSP compares it with the copy it may already have. If the newly arrived LSP is older than the one it has (found by checking the sequence number), it discards the LSP. If it is newer, the node does the following:

a. It discards the old LSP and keeps the new one.

b. It sends a copy of it out of each interface except the one from which the packet arrived. This guarantees that flooding stops somewhere in the domain (where a node has only one interface).

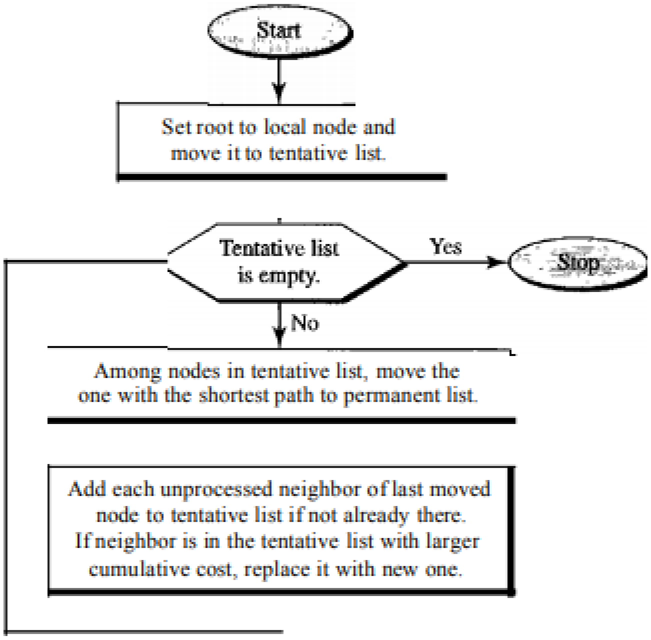
Formation of Shortest Path Tree: Dijkstra Algorithm

After receiving all LSPs, each node will have a copy of the whole topology. However, the topology is not sufficient to find the shortest path to every other node;

a shortest path tree is needed. A tree is a graph of nodes and links; one node is called the root. All other nodes can be reached from the root through only one single route. A shortest path tree is a tree in which the path between the root and every other node is the shortest. What we need for each node is a shortest path tree with that node as the root.

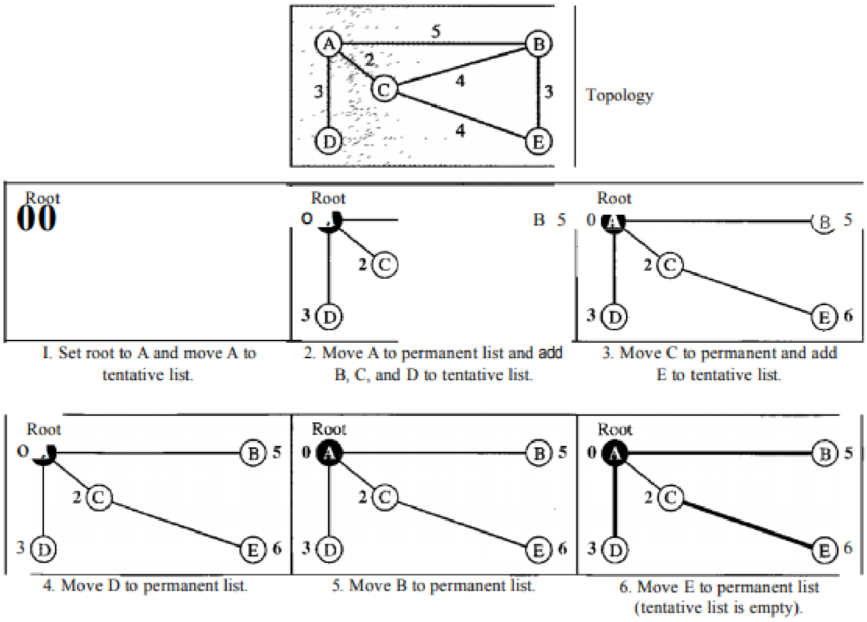
The Dijkstra algorithm creates a shortest path tree from a graph. The algorithm divides the nodes into two sets: tentative and permanent. It finds the neighbors of a current node, makes them tentative, examines them, and if they pass the criteria, makes them permanent. We can informally define the algorithm by using the flowchart in Figure 22.22. Let us apply the algorithm to node A of our sample graph in Figure 22.23. To find the shortest path in each step, we need the cumulative cost from the root to each node, which is shown next to the node. The following shows the steps. At the end of each step, we show the permanent (filled circles) and the tentative (open circles) nodes and lists with the cumulative costs.

Figure 22.22 *Dijkstra algorithm*



Start
Set root to local node and move it to tentative list.
Tentative list is empty.
Yes
Stop
No
Among nodes in tentative list, move the one with the shortest path to permanent list.
Add each unprocessed neighbor of last moved node to tentative list if not already there. If neighbor is in the tentative list with larger cumulative cost, replace it with new one.

Figure 22.23 *Example offormation ofshortest path tree*



A
B
C
D
E
5
2
3
4
3
4
Topology
Root
00
1. Set root to A and move A to tentative list.
Root
O
B 5
2 C
3 D
2. Move A to permanent list and add B, C, and D to tentative list.
Root
0 A
B 5
2 C
3 D
E 6
3. Move C to permanent and add E to tentative list.
Root
O
B 5
2 C
3 D
E 6
4. Move D to permanent list.
Root
0 A
B 5
2 C
3 D
E 6
5. Move B to permanent list.
Root
0 A
B 5
2 C
3 D
E 6
6. Move E to permanent list (tentative list is empty).

1. We make node A the root of the tree and move it to the tentative list. Our two lists are Permanent list: empty Tentative list: A(O)

2. Node A has the shortest cumulative cost from all nodes in the tentative list. We move A to the permanent list and add all neighbors of A to the tentative list. Our new lists are Permanent list: A(O) Tentative list: B(5), C(2), D(3)

3. Node C has the shortest cumulative cost from all nodes in the tentative list. We move C to the permanent list. Node C has three neighbors, but node A is already processed, which makes the unprocessed neighbors just B and E. However, B is already in the tentative list with a cumulative cost of 5. Node A could also reach node B through C with a cumulative cost of 6. Since 5 is less than 6, we keep node B with a cumulative cost of 5 in the tentative list and do not replace it. Our new lists are Permanent list: A(O), e(2) Tentative list: B(5), 0(3), E(6)

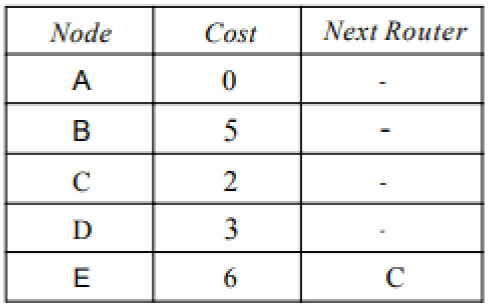
4. Node D has the shortest cumulative cost of all the nodes in the tentative list. We move D to the permanent list. Node D has no unprocessed neighbor to be added to the tentative list. Our new lists are Permanent list: A(O), C(2), 0(3) Tentative list: B(5), E(6)

5. Node B has the shortest cumulative cost of all the nodes in the tentative list. We move B to the permanent list. We need to add all unprocessed neighbors of B to the tentative list (this is just node E). However, E(6) is already in the list with a smaller cumulative cost. The cumulative cost to node E, as the neighbor of B, is 8. We keep node E(6) in the tentative list. Our new lists are Permanent list: A(O), B(5), C(2), 0(3) Tentative list: E(6)

6. Node E has the shortest cumulative cost from all nodes in the tentative list. We move E to the permanent list. Node E has no neighbor. Now the tentative list is empty. We stop; our shortest path tree is ready. The final lists are Permanent list: A(O), B(5), C(2), D(3), E(6) Tentative list: empty

Calculation of Routing Table from Shortest Path Tree Each node uses the shortest path tree protocol to construct its routing table. The routing table shows the cost of reaching each node from the root. Table 22.2 shows the routing table for node A.

Table 22.2 *Routing table for node A*



Node	*Cost*	*Next Router*
A	0	-
B	5	-
C	2	-
D	3	-
E	6	C

Compare Table 22.2 with the one in Figure 22.14. Both distance vector routing and link state routing end up with the same routing table for node A.

ICMP

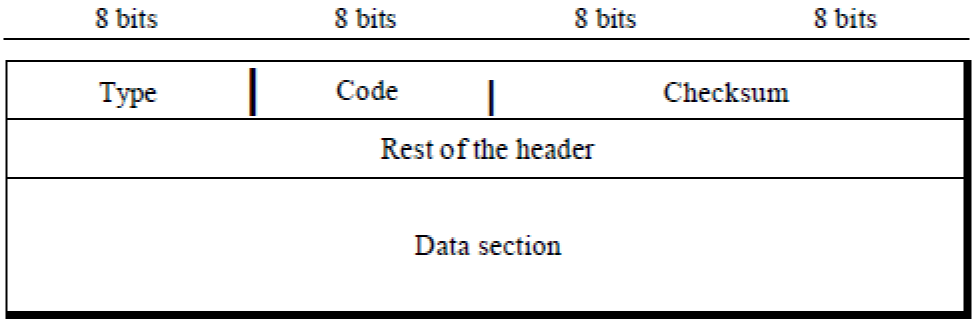
The IP protocol has no error-reporting or error-correcting mechanism. The IP protocol also lacks a mechanism for host and management queries. A host sometimes needs to determine if a router or another host is alive. And sometimes a network administrator needs information from another host or router. The Internet Control Message Protocol (ICMP) has been designed to compensate for the above two deficiencies. It is a companion to the IP protoco1.

Types of Messages

ICMP messages are divided into two broad categories: error-reporting messages and query messages. The error-reporting messages report problems that a router or a host (destination) may encounter when it processes an IP packet. The query messages, which occur in pairs, help a host or a network manager get specific information from a router or another host. For example, nodes can discover their neighbors. Also, hosts can discover and learn about routers on their network, and routers can help a node redirect its messages.

Message Format

An ICMP message has an 8-byte header and a variable-size data section. Although the general format of the header is different for each message type, the first 4 bytes are common to all. The first field, ICMP type, defines the type of the message. The code field specifies the reason for the particular message type. The last common field is the checksum field. The rest of the header is specific for each message type. The data section in error messages carries information for finding the original packet that had the error. In query messages, the data section carries extra information based on the type of the query.

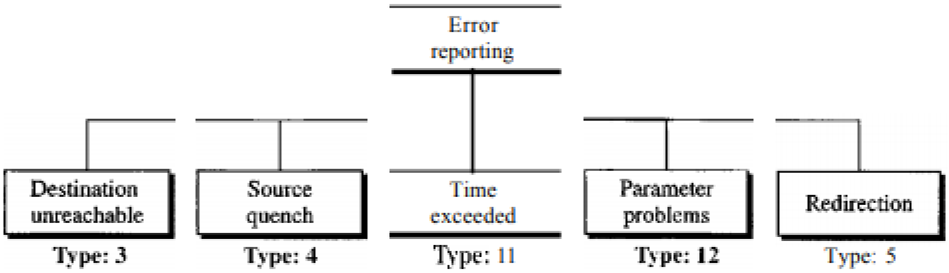


Error Reporting

One of the main responsibilities of ICMP is to report errors. Although technology has produced increasingly reliable transmission media, errors still exist and must be handled. IP, as discussed in Chapter 20, is an unreliable protocol. This means that error checking and error control are not a concern of IP. ICMP was designed, in part, to compensate for this shortcoming. However, ICMP does not correct errors-it simply reports them. Error correction is left to the higher-level protocols. Error messages are always sent to the original source because the only information available in the datagram about the route is the source and destination IP addresses. ICMP uses the source IP address to send the error message to the source (originator) of the datagram.

Five types of errors are handled: destination unreachable, source quench, time exceeded, parameter problems, and redirection

Figure 21.9 *Error-reporting messages*



Destination Unreachable

When a router cannot route a datagram or a host cannot deliver a datagram, the datagram is discarded and the router or the host sends a destination-unreachable message back to the source host that initiated the datagram. Note that destination-unreachable messages can be created by either a router or the destination host.

Source Quench

The IP protocol is a connectionless protocol. There is no communication between the source host, which produces the datagram, the routers, which forward it, and the destination host, which processes it. One of the ramifications of this absence of communication is the lack of flow control. IP does not have a flow control mechanism embedded in the protocol. The lack of flow control can create a major problem in the operation of IP: congestion. The source host never knows if the routers or the destination host has been overwhelmed with datagrams. The source host never knows if it is producing datagrams faster than can be forwarded by routers or processed by the destination host. The lack of flow control can create congestion in routers or the destination host. A router or a host has a limited-size queue (buffer) for incoming datagrams waiting to be forwarded (in the case of a router) or to be processed (in the case of a host). If the datagrams are received much faster than they can be forwarded or processed, the queue may overflow. In this case, the router or the host has no choice but to discard some of the datagrams. The source-quench message in ICMP was designed to add a kind of flow control to the IP. When a router or host discards a datagram due to congestion, it sends a source-quench message to the sender of the datagram. This message has two purposes. First, it informs the source that the datagram has been discarded. Second, it warns the source that there is congestion somewhere in the path and that the source should slow down (quench) the sending process.

Time Exceeded

The time-exceeded message is generated in two cases: As we see in Chapter 22, routers use routing tables to find the next hop (next router) that must receive the packet. Ifthere are errors in one or more routing tables, a packet can travel in a loop or a cycle, going from one router to the next or visiting a series of routers endlessly. As we saw in Chapter 20, each datagram contains a field called time to live that controls this situation. When a datagram visits a router, the value of this field is decremented by 1. When the time-to-live value reaches 0, after decrementing, the router discards the datagram. However, when the datagram is discarded, a time-exceeded message must be sent by the router to the original source. Second, a time-exceeded message is also generated when not all fragments that make up a message arrive at the destination host within a certain time limit.

Parameter Problem

Any ambiguity in the header part of a datagram can Create serious problems as the datagram travels through the Internet. If a router or the destination host discovers an ambiguous or missing value in any field of the datagram, it discards the datagram and sends a parameter-problem message back to the source.

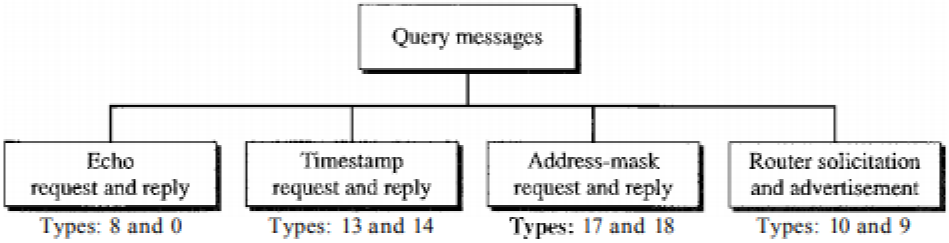
Redirection

When a router needs to send a packet destined for another network, it must know the IP address of the next appropriate router. The same is true if the sender is a host. Both routers and hosts, then, must have a routing table to find the address of the router or the next router. Routers take part in the routing update process, as we will see in Chapter 22, and are supposed to be updated constantly. Routing is dynamic. However, for efficiency, hosts do not take part in the routing update process because there are many more hosts in an internet than routers. Updating the routing tables of hosts dynamically produces unacceptable traffic. The hosts usually use static routing. When a host comes up, its routing table has a limited number of entries. It usually knows the IP address of only one router, the default router. For this reason, the host may send a datagram, which is destined for another network, to the wrong router. In this case, the router that receives the datagram will forward the datagram to the correct router. However, to update the routing table of the host, it sends a redirection message to the host.

Query

In addition to error reporting, ICMP can diagnose some network problems. This is accomplished through the query messages, a group of four different pairs of messages, as shown in Figure 21.12. In this type of ICMP message, a node sends a message that is answered in a specific format by the destination node. A query message is encapsulated in an IP packet, which in turn is encapsulated in a data link layer frame. However, in this case, no bytes of the original IP are included in the message, as shown in Figure 21.13.

Figure 21.12 *Query messages*



Echo Request and Reply

The echo-request and echo-reply messages are designed for diagnostic purposes. Network managers and users utilize this pair of messages to identify network problems. The combination of echo-request and echo-reply messages determines whether two systems (hosts or routers) can communicate with each other. The echo-request and echo-reply messages can be used to determine if there is communication at the IP level. Because ICMP messages are encapsulated in IP datagrams, the receipt of an echo-reply message by the machine that sent the echo request is proof that the IP protocols in the sender and receiver are communicating with each other using the IP datagram.

Timestamp Request and Reply

Two machines (hosts or routers) can use the timestamp request and timestamp reply messages to determine the round-trip time needed for an IP datagram to travel between them. It can also be used to synchronize the clocks in two machines.

Address-Mask Request and Reply

A host may know its IP address, but it may not know the corresponding mask. For example, a host may know its IP address as 159.31.17.24, but it may not know that the corresponding mask is /24. To obtain its mask, a host sends an address-mask-request message to a router on the LAN. If the host knows the address of the router, it sends the request directly to the router. If it does not know, it broadcasts the message. The router receiving the address-mask-request message responds with an address-mask-reply message, providing the necessary mask for the host. This can be applied to its full IP address to get its subnet address.

Router Solicitation and Advertisement

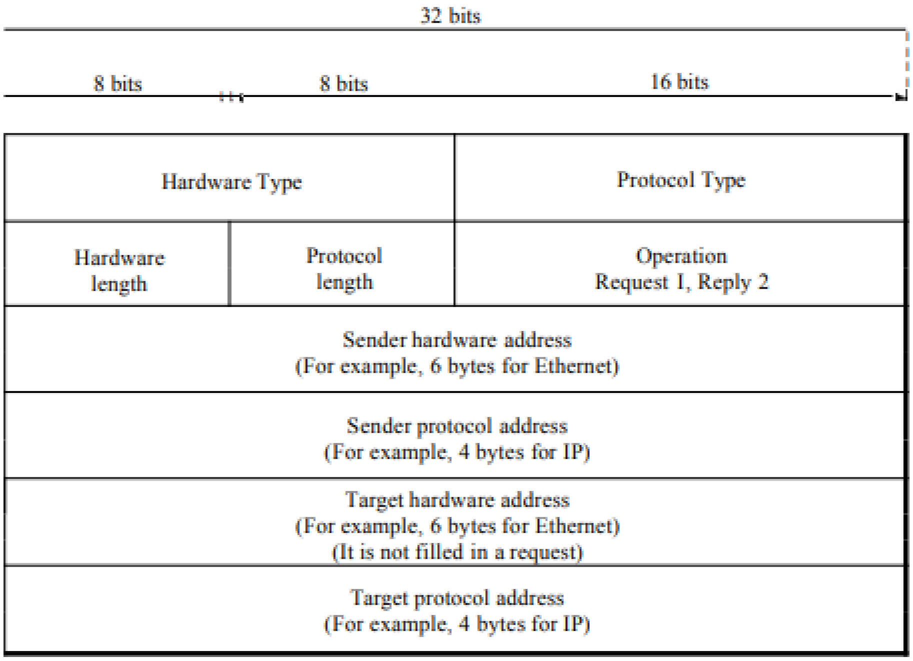
As we discussed in the redirection message section, a host that wants to send data to a host on another network needs to know the address of routers connected to its own network. Also, the host must know if the routers are alive and functioning. The router-solicitation and router-advertisement messages can help in this situation. A host can broadcast (or multicast) a router-solicitation message. The router or routers that receive the solicitation message broadcast their routing information using the router-advertisement message. A router can also periodically send router-advertisement messages even if no host has solicited. Note that when a router sends out an advertisement, it announces not only its own presence but also the presence of all routers on the network of which it is aware.

Checksum

In ICMP the checksum is calculated over the entire message (header and data).

ARP

The address resolution protocol (**ARP**) is a protocol used by the Internet Protocol (IP) [RFC826], specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer.



32 bits

8 bits	8 bits	16 bits
Hardware Type		Protocol Type
Hardware length	Protocol length	Operation Request 1, Reply 2
Sender hardware address (For example, 6 bytes for Ethernet)		
Sender protocol address (For example, 4 bytes for IP)		
Target hardware address (For example, 6 bytes for Ethernet) (It is not filled in a request)		
Target protocol address (For example, 4 bytes for IP)		

The fields are as follows:

o Hardware type. This is a 16-bit field defining the type of the network on which ARP is running. Each LAN has been assigned an integer based on its type. For example, Ethernet is given type 1. ARP can be used on any physical network.

o Protocol type. This is a 16-bit field defining the protocol. For example, the value of this field for the IPv4 protocol is 080016, ARP can be used with any higher-level protocol.

o Hardware length. This is an 8-bit field defining the length of the physical address in bytes. For example, for Ethernet the value is 6.

o Protocol length. This is an 8-bit field defining the length of the logical address in bytes. For example, for the IPv4 protocol the value is 4.

o Operation. This is a 16-bit field defining the type of packet. Two packet types are defined: ARP request (1) and ARP reply (2).

o Sender hardware address. This is a variable-length field defining the physical address of the sender. For example, for Ethernet this field is 6 bytes long. o Sender protocol address. This is a variable-length field defining the logical (for example, IP) address of the sender. For the IP protocol, this field is 4 bytes long.

o Target hardware address. This is a variable-length field defining the physical address of the target. For example, for Ethernet this field is 6 bytes long. For an ARP request message, this field is all Os because the sender does not know the physical address of the target.

o Target protocol address. This is a variable-length field defining the logical (for example, IP) address of the target. For the IPv4 protocol, this field is 4 bytes long.

RARP

Reverse Address Resolution Protocol (RARP) finds the logical address for a machine that knows only its physical address. Each host or router is assigned one or more logical (IP) addresses, which are unique and independent of the physical (hardware) address of the machine. To create an IP datagram, a host or a router needs to know its own IP address or addresses. The IP address of a machine is usually read from its configuration file stored on a disk file. However, a diskless machine is usually booted from ROM, which has minimum booting information. The ROM is installed by the manufacturer. It cannot include the IP address because the IP addresses on a network are assigned by the network administrator. The machine can get its physical address (by reading its NIC, for example), which is unique locally. It can then use the physical address to get the logical address by using the RARP protocol. A RARP request is created and broadcast on the local network. Another machine on the local network that knows all the IP addresses will respond with a RARP reply. The requesting machine must be running a RARP client program; the responding machine must be running a RARP server program. There is a serious problem with RARP: Broadcasting is done at the data link layer. The physical broadcast address, all is in the case of Ethernet, does not pass the boundaries of a network. This means that if an administrator has several networks or several subnets, it needs to assign a RARP server for each network or subnet. This is the reason that RARP is almost obsolete.

SUBNETTING NUMERICALS

- **Write the IP address 222.1.1.20 mask 255.255.255.192 in CIDR notation.**

Decimal 192 =11000000 binary which means that 2 bits of this octet are used for the subnet? Now add the 24 bits 255.255.255 and we have 26 bits. So we write:

222.1.1.20/26

- **Write the IP address 135.1.1.25 mask 255.255. 248.0 in CIDR notation.**

Decimal 248 =11111000 binary which means that 5 bits of this octet are used for the subnet? Now add the 16 bits 255.255. And we have 21 bits. So we write:

135..1.1.25/21

- **You have been allocated a class C network address of 211.1.1.0 and are using the default subnet mask of 255.255.255.0 how may hosts can you have?**

A class C address has 8 bits of the host which will give 28 -2 =254 hosts

Subnet the Class C IP Address 205.11.2.0 so that you have 30 subnets.

What is the subnet mask for the maximum number of hosts?

How many hosts can each subnet have?

What is the IP address of host 3 on subnet 2 ?

Current mask= 255.255.255.0

Bits needs for 30 subnets =5 =25 =32 possible subnets

Bits left for hosts = 3 = 23 = 8-2=6 possible hosts.

So our mask in binary =11111000= 248 decimal

Final Mask =255.255.255.248

Address of host 3 on subnet 2 is

Subnet 2 =00010000 host 3 =000000011

Add the two together =00010011=19

Therefore IP address of host 3 on subnet 2 =205.11.2.19

- **Subnet the Class C IP Address 195.1.1.0 So that you have at least 2 subnets each subnet must have room for 48 hosts. What are the two possible subnet masks?**

Current mask= 255.255.255.0

Bits needs for 48 hosts = 6 = 26 = 64-2=62 possible hosts.

Bits needs for 2 subnets =1 =21 =2 possible subnets

Total of 7 bits needed so therefore we can use either 1 bit or 2 bits for the subnet. So we could have

1 bit subnet 7 bits hosts or 2 bits subnet 6 bit host

Masks are 10000000 and 11000000 =128 decimal and 192 decimal.

Final possible masks are:

255.255.255.128 and 255.255.255.192

- **Change the following IPv4 addresses from binary notation to dotted-decimal notation.**

a. 10000001 00001011 00001011 11101111

b. 11000001 10000011 00011011 11111111

Solution

We replace each group of 8 bits with its equivalent decimal number (see Appendix B) and add

dots for separation.

a. 129.11.11.239

b. 193.131.27.255

- **Change the following IPv4 addresses from dotted-decimal notation to binary notation.**

a. 111.56.45.78

b. 221.34.7.82

Solution

We replace each decimal number with its binary equivalent (see Appendix B).

a.• 01101111 00111000 00101101 01001110

b. 11011101 00100010 00000111 01010010

- **Find the error, if any, in the following IPv4 addresses.**

a. 111.56.045.78

b. 221.34.7.8.20

c. 75.45.301.14

d. 11100010.23.14.67

Solution

a. There must be no leading zero (045).

b. There can be no more than four numbers in an IPv4 address.

c. Each number needs to be less than or equal to 255 (301 is outside this range).

d. A mixture of binary notation and dotted-decimal notation is not allowed.

- **Find the class of each address.**

a. 00000001 00001011 00001011 11101111

b. 11000001 10000011 00011011 11111111

c. 14.23.120.8

d. 252.5.15.111

Solution

a. The first bit is O. This is a class A address.

b. The first 2 bits are 1; the third bit is O. This is a class C address.

c. The first byte is 14 (between 0 and 127); the class is A.

d. The first byte is 252 (between 240 and 255); the class is E

- **A block of addresses is granted to a small organization. We know that one of the addresses is 205.16.37.39/28. What is the first address in the block ?**

Solution

First Address The first address in the block can be found by setting the 32 - n rightmost bits in the binary notation of the address to Os.

The binary representation of the given address is 11001101 00010000 00100101 00100 I 11. If we set 32 - 28 rightmost bits to 0, we get 11001101 000100000100101 0010000 or 205.16.37.32.

The first address in the block can be found by setting the rightmost 32 - n bits to Os.

Find the last address for the block?

Last Address The last address in the block can be found by setting the 32 - n rightmost bits in the binary notation of the address to Is.

The last address in the block can be found by setting the rightmost 32 - n bits to Is.

The binary representation of the given address is 11001101 000100000010010100100111. If we set 32 - 28 rightmost bits to 1, we get 11001101 00010000 001001010010 1111 or 205.16.37.47.

Find the number of addresses?

The value of n is 28, which means that number of addresses is 232- 28 or 16.

- **Suppose an organization is given the block 17.12.40.0/26, which contains 64 addresses. The organization has three offices and needs to divide the addresses into three subblocks of 32, 16, and 16 addresses.**

We can find the new masks by using the following arguments:

1. Suppose the mask for the first subnet is n1, then 232- n1 must be 32, which means that n1 =27.

2. Suppose the mask for the second subnet is n2, then 232- n2 must be 16, which means that n2 = 28.

3. Suppose the mask for the third subnet is n3, then 232- n3 must be 16, which means that n3 =28.

This means that we have the masks 27, 28, 28 with the organization mask being 26.

a. In subnet 1, the address 17.12.14.29/27 can give us the subnet address if we use the mask /27 because

Host: 00010001 00001100 00001110 00011101

Mask: /27

Subnet: 00010001 00001100 00001110 00000000.... (17.12.14.0)

b. In subnet 2, the address 17.12.14.45/28 can give us the subnet address if we use the mask /28 because

Host: 00010001 00001100 00001110 00101101

Mask: /28

Subnet: 00010001 00001100 00001110 00100000.... (17.12.14.32)

c. In subnet 3, the address 17.12.14.50/28 can give us the subnet address if we use the mask /28 because

Host: 00010001 00001100 00001110 00110010

Mask: /28

Subnet: 00010001 00001100 00001110 00110000.... (17.12.14.48)

- **An ISP is granted a block of addresses starting with 190.100.0.0/16 (65,536 addresses). The ISP needs to distribute these addresses to three groups of customers as follows:**

a. The first group has 64 customers; each needs 256 addresses.

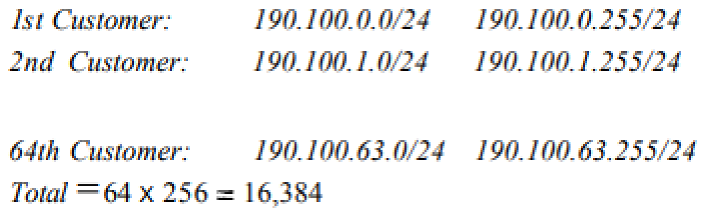
b. The second group has 128 customers; each needs 128 addresses.

c. The third group has 128 customers; each needs 64 addresses.

Design the subblocks and find out how many addresses are still available after these allocations.

1. Group 1

For this group, each customer needs 256 addresses. This means that 8 (log2256) bits are needed to define each host. The prefix length is then 32 - 8 =24. The addresses are

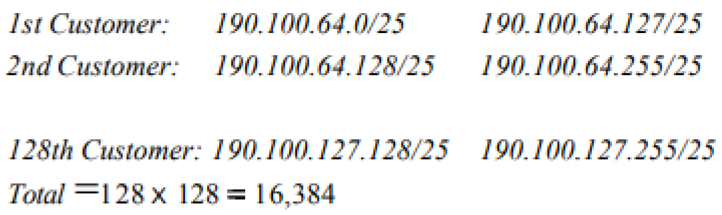


1st Customer:	*190.100.0.0/24*	*190.100.0.255/24*
2nd Customer:	*190.100.1.0/24*	*190.100.1.255/24*
64th Customer:	*190.100.63.0/24*	*190.100.63.255/24*

Total =64 x 256 = 16,384

Group2

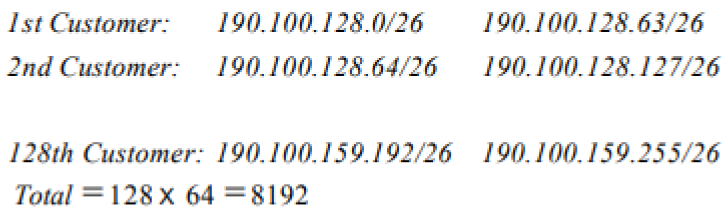
For this group, each customer needs 128 addresses. This means that 7 (10g2 128) bits are needed to define each host. The prefix length is then 32 - 7 =25. The addresses are



1st Customer:	190.100.64.0/25	190.100.64.127/25
2nd Customer:	190.100.64.128/25	190.100.64.255/25
128th Customer:	190.100.127.128/25	190.100.127.255/25

Total =128 x 128 = 16,384

Group3 For this group, each customer needs 64 addresses. This means that 6 (logz 64) bits are needed to each host. The prefix length is then 32 - 6 =26. The addresses are



1st Customer:	190.100.128.0/26	190.100.128.63/26
2nd Customer:	190.100.128.64/26	190.100.128.127/26
128th Customer:	190.100.159.192/26	190.100.159.255/26

Total =128 x 64 =8192

Number of granted addresses to the ISP: 65,536

Number of allocated addresses by the ISP: 40,960

Number of available addresses: 24,576

Numerical reference: B. Forouzan

CHAPTER 5-TRANSPORT LAYER

THE TRANSPORT SERVICES

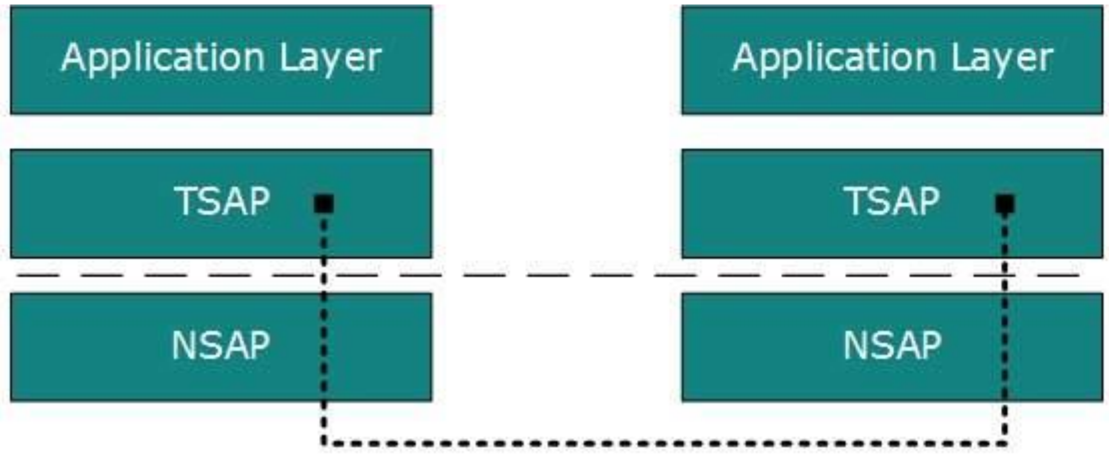
Transport layer offers peer-to-peer and end-to-end connection between two processes on remote hosts. Transport layer takes data from upper layer (i.e. Application layer) and then breaks it into smaller size segments, numbers each byte, and hands over to lower layer (Network Layer) for delivery.

Functions

- This Layer is the first one which breaks the information data, supplied by Application layer in to smaller units called segments. It numbers every byte in the segment and maintains their accounting.
- This layer ensures that data must be received in the same sequence in which it was sent.
- This layer provides end-to-end delivery of data between hosts which may or may not belong to the same subnet.
- All server processes intend to communicate over the network are equipped with well-known Transport Service Access Points (TSAPs) also known as port numbers.

End-to-End Communication

A process on one host identifies its peer host on remote network by means of TSAPs, also known as Port numbers. TSAPs are very well defined and a process which is trying to communicate with its peer knows this in advance.



For example, when a DHCP client wants to communicate with remote DHCP server, it always requests on port number 67. When a DNS client wants to communicate with remote DNS server, it always requests on port number 53 (UDP).

The two main Transport layer protocols are:

- **Transmission Control Protocol**

  It provides reliable communication between two hosts.

- **User Datagram Protocol**

  It provides unreliable communication between two hosts

TRANSPORT SERVICE PREMITIVE

A service is specified by a set of primitives. A primitive means operation. To access the service a user process can access these primitives. These primitives are different for connection oriented service and connectionless service.

There are five types of service primitives:

1. LISTEN : When a server is ready to accept an incoming connection it executes the LISTEN primitive. It blocks waiting for an incoming connection.
2. CONNECT : It connects the server by establishing a connection. Response is awaited.
3. RECIEVE: Then the RECIEVE call blocks the server.
4. SEND : Then the client executes SEND primitive to transmit its request followed by the execution of RECIEVE to get the reply. Send the message.
5. DISCONNECT : This primitive is used for terminating the connection. After this primitive one can't send any message. When the client sends DISCONNECT packet then the server also sends the DISCONNECT packet to acknowledge the client. When the server package is received by client then the process is terminated.

TRANSPORT PROTOCOLS: TCP AND UDP

TRANSMISSION CONTROL PROTOCOL (TCP)

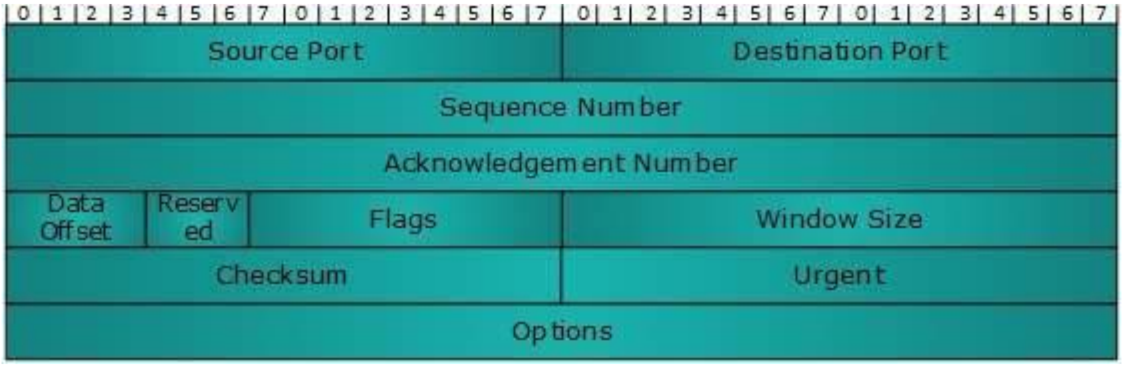
The transmission Control Protocol (TCP) is one of the most important protocols of Internet Protocols suite. It is most widely used protocol for data transmission in communication network such as internet.

Features

- TCP is reliable protocol. That is, the receiver always sends either positive or negative acknowledgement about the data packet to the sender, so that the sender always has bright clue about whether the data packet is reached the destination or it needs to resend it.
- TCP ensures that the data reaches intended destination in the same order it was sent.
- TCP is connection oriented. TCP requires that connection between two remote points be established before sending actual data.
- TCP provides error-checking and recovery mechanism.
- TCP provides end-to-end communication.
- TCP provides flow control and quality of service.
- TCP operates in Client/Server point-to-point mode.
- TCP provides full duplex server, i.e. it can perform roles of both receiver and sender.

Header

The length of TCP header is minimum 20 bytes long and maximum 60 bytes.



- **Source Port (16-bits)** - It identifies source port of the application process on the sending device.
- **Destination Port (16-bits)** - It identifies destination port of the application process on the receiving device.
- **Sequence Number (32-bits)** - Sequence number of data bytes of a segment in a session.
- **Acknowledgement Number (32-bits)** - When ACK flag is set, this number contains the next sequence number of the data byte expected and works as acknowledgement of the previous data received.

- **Data Offset (4-bits)** - This field implies both, the size of TCP header (32-bit words) and the offset of data in current packet in the whole TCP segment.
- **Reserved (3-bits)** - Reserved for future use and all are set zero by default.
- **Flags (1-bit each)**
    - **NS** - Nonce Sum bit is used by Explicit Congestion Notification signaling process.
    - **CWR** - When a host receives packet with ECE bit set, it sets Congestion Windows Reduced to acknowledge that ECE received.
    - **ECE** -It has two meanings:
        - If SYN bit is clear to 0, then ECE means that the IP packet has its CE (congestion experience) bit set.
        - If SYN bit is set to 1, ECE means that the device is ECT capable.
    - **URG** - It indicates that Urgent Pointer field has significant data and should be processed.
    - **ACK** - It indicates that Acknowledgement field has significance. If ACK is cleared to 0, it indicates that packet does not contain any acknowledgement.
    - **PSH** - When set, it is a request to the receiving station to PUSH data (as soon as it comes) to the receiving application without buffering it.
    - **RST** - Reset flag has the following features:
        - It is used to refuse an incoming connection.
        - It is used to reject a segment.
        - It is used to restart a connection.
    - **SYN** - This flag is used to set up a connection between hosts.
    - **FIN** - This flag is used to release a connection and no more data is exchanged thereafter. Because packets with SYN and FIN flags have sequence numbers, they are processed in correct order.
- **Windows Size** - This field is used for flow control between two stations and indicates the amount of buffer (in bytes) the receiver has allocated for a segment, i.e. how much data is the receiver expecting.

- **Checksum** - This field contains the checksum of Header, Data and Pseudo Headers.
- **Urgent Pointer** - It points to the urgent data byte if URG flag is set to 1.
- **Options** - It facilitates additional options which are not covered by the regular header. Option field is always described in 32-bit words. If this field contains data less than 32-bit, padding is used to cover the remaining bits to reach 32-bit boundary.

USER DATAGRAM PROTOCOL (UDP)

The User Datagram Protocol (UDP) is simplest Transport Layer communication protocol available of the TCP/IP protocol suite. It involves minimum amount of communication mechanism. UDP is said to be an unreliable transport protocol but it uses IP services which provides best effort delivery mechanism.

In UDP, the receiver does not generate an acknowledgement of packet received and in turn, the sender does not wait for any acknowledgement of packet sent. This shortcoming makes this protocol unreliable as well as easier on processing.

Requirement of UDP

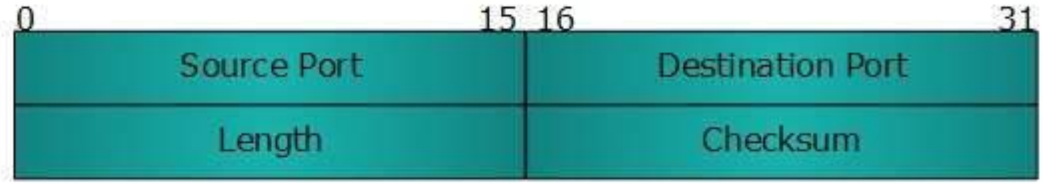
A question may arise, why do we need an unreliable protocol to transport the data? We deploy UDP where the acknowledgement packets share significant amount of bandwidth along with the actual data. For example, in case of video streaming, thousands of packets are forwarded towards its users. Acknowledging all the packets is troublesome and may contain huge amount of bandwidth wastage. The best delivery mechanism of underlying IP protocol ensures best efforts to deliver its packets, but even if some packets in video streaming get lost, the impact is not calamitous and can be ignored easily. Loss of few packets in video and voice traffic sometimes goes unnoticed.

Features

- UDP is used when acknowledgement of data does not hold any significance.
- UDP is good protocol for data flowing in one direction.
- UDP is simple and suitable for query based communications.
- UDP is not connection oriented.
- UDP does not provide congestion control mechanism.
- UDP does not guarantee ordered delivery of data.
- UDP is stateless.
- UDP is suitable protocol for streaming applications such as VoIP, multimedia streaming.

UDP Header

UDP header is as simple as its function.



UDP header contains four main parameters:

- **Source Port** - This 16 bits information is used to identify the source port of the packet.
- **Destination Port** - This 16 bits information, is used identify application level service on destination machine.
- **Length** - Length field specifies the entire length of UDP packet (including header). It is 16-bits field and minimum value is 8-byte, i.e. the size of UDP header itself.
- **Checksum** - This field stores the checksum value generated by the sender before sending. IPv4 has this field as optional so when checksum field does not contain any value it is made 0 and all its bits are set to zero.

UDP application

Here are few applications where UDP is used to transmit data:

- Domain Name Services
- Simple Network Management Protocol
- Trivial File Transfer Protocol
- Routing Information Protocol
- Kerberos

PORT AND SOCKET

- At the transport layer, we need a transport layer address, called a port number, to choose among multiple processes running on the destination host.
- The destination port number is needed for delivery; the source port number is needed for the reply.
- In the Internet model, the port numbers are 16-bit integers between 0 and 65,535.
- The client program defines itself with a port number, chosen randomly by the transport layer

software running on the client host. This is the ephemeral port number.

- The server process must also define itself with a port number.
- This port number, however, cannot be chosen randomly
- The Internet has decided to use universal port numbers for servers; these are called well-known port numbers.
- For example, while the Daytime client process, discussed above, can use an ephemeral (temporary) port number 52,000 to identify itself, the Daytime server process must use the well-known (permanent) port number 13.

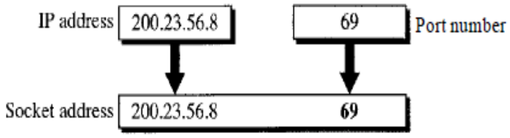
lANA Ranges

The lANA (Internet Assigned Number Authority) has divided the port numbers into three ranges: well known, registered, and dynamic (or private

- Well-known ports. The ports ranging from 0 to 1023 are assigned and controlled by lANA. These are the well-known ports.
- Registered ports. The ports ranging from 1024 to 49,151 are not assigned or controlled by lANA. They can only be registered with lANA to prevent duplication.
- Dynamic ports. The ports ranging from 49,152 to 65,535 are neither controlled nor registered. They can be used by any process. These are the ephemeral ports.

Socket Address

- Process-to-process delivery needs two identifiers, IP address and the port number, at each end to make a connection.
- The combination of an IP address and a port number is called a socket address.
- The client socket address defines the client process uniquely just as the server socket address defines the server process uniquely
- A transport layer protocol needs a pair of socket addresses: the client socket address and the server socket address.



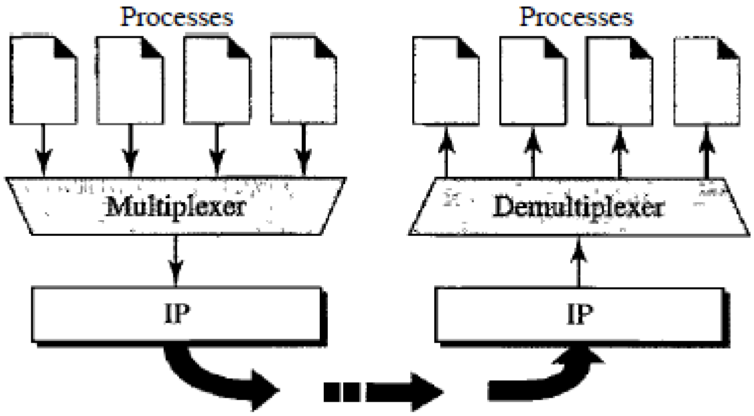
MULTIPLEXING AND DEMULTIPLEXING

Multiplexing

- At the sender site, there may be several processes that need to send packets.
- However, there is only one transport layer protocol at any time. This is a many-to-one relationship and requires multiplexing.
- The protocol accepts messages from different processes, differentiated by their assigned port numbers.
- After adding the header, the transport layer passes the packet to the network layer.

Demultiplexing

- At the receiver site, the relationship is one-to-many and requires demultiplexing.
- The transport layer receives datagrams from the network layer.
- After error checking and dropping of the header, the transport layer delivers each message to the appropriate process based on the port number.



- Internet Protocol (IP) provides a packet delivery service across an internet
- however, IP cannot distinguish between multiple processes (applications) running on the same computer
- fields in the IP datagram header identify only computers
- a protocol that allows an application to serve as an end-point of communication is known as a transport protocol or an end-to-end protocol
- the TCP/IP protocol suite provides two transport protocols:
  - the User Datagram Protocol (UDP)
  - the Transmission Control Protocol (TCP)

CONNECTIONLESS VERSUS CONNECTION-ORIENTED SERVICE

Connectionless Service

- In a connectionless service, the packets are sent from one party to another with no need for connection establishment or connection release.
- The packets are not numbered; they may be delayed or lost or may arrive out of sequence.
- There is no acknowledgment either.
- UDP, is connectionless.

Connection Oriented Service

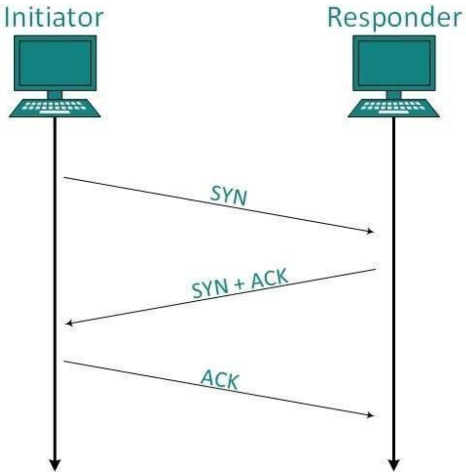
- In a connection-oriented service, a connection is first established between the sender and the receiver.
- Data are transferred. At the end, the connection is released.

- A prior connection setup is needed in connection-oriented service but not in connectionless service.
- Connection-oriented service guarantees reliability but not connectionless service.
- Congestion is very unlikely in connection-oriented service but not in connectionless.
- Lost data retransmission is possible in connection-oriented service but not in connectionless service.
- Connection-oriented is suitable for long connection while connectionless is suitable for a bursty connection
- Packets reach the destination following the same route in connection-oriented service, but for connectionless, the packets can take different paths.
- Resource allocation is needed in the connection-oriented but not in the case of connectionless service.
- The transfer is slower in the connection-oriented due to connection setup time and ACK but is faster in connectionless service due to missing initial setup and ACK.

CONNECTION ESTABLISHMENT, CONNECTION RELEASE

Connection Management

TCP communication works in Server/Client model. The client initiates the connection and the server either accepts or rejects it. Three-way handshaking is used for connection management.



Establishment

Client initiates the connection and sends the segment with a Sequence number. Server acknowledges it back with its own Sequence number and ACK of client's segment which is one more than client's Sequence number. Client after receiving ACK of its segment sends an acknowledgement of Server's response.

Release

Either of server and client can send TCP segment with FIN flag set to 1. When the receiving end responds it back by Acknowledging FIN, that direction of TCP communication is closed and connection is released.

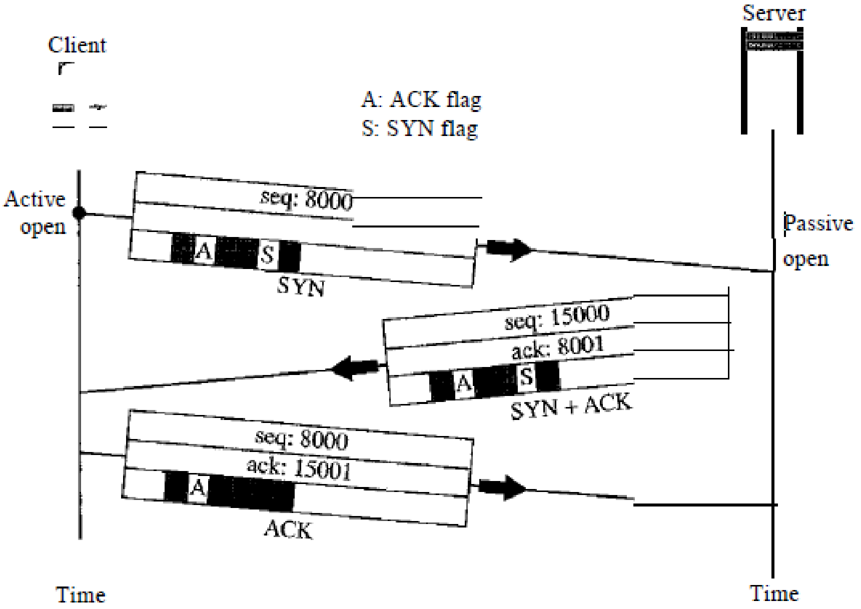
TCP Connection: Connection establishment

- TCP transmits data in full-duplex mode.
- When two TCPs in two machines are connected, they are able to send segments to each other simultaneously.
- This implies that each party must initialize communication and get approval from the other party before any data are transferred

THREE-WAY HANDSHAKING

The connection establishment in TCP is called three way handshaking. In our example, an application program, called the client, wants to make a connection with another application program, called the server, using TCP as the transport layer protocol.

Figure 23.18 *Connection establishment using three-way handshaking*



The three steps in this phase are as follows.

1. The client sends the first segment, a SYN segment, in which only the SYN flag is set. This segment is for synchronization of sequence numbers. It consumes one sequence number. When the data transfer starts, the sequence number is incremented by 1. We can say that the SYN segment carries no real data, but we can think of it as containing 1 imaginary byte.

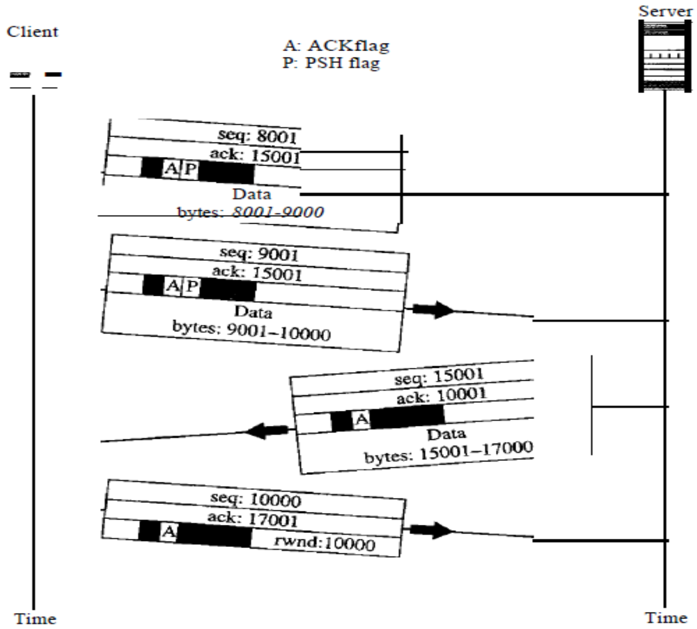
2. The server sends the second segment, a SYN +ACK segment, with 2 flag bits set: SYN and ACK. This segment has a dual purpose. It is a SYN segment for communication in the other direction and serves as the acknowledgment for the SYN segment. It consumes one sequence number.

3. The client sends the third segment. This is just an ACK segment. It acknowledges the receipt of the second segment with the ACK flag and acknowledgment number field. Note that the sequence number in this segment is the same as the one in the SYN segment; the ACK segment does not consume any sequence numbers.

Data Transfer

After connection is established, bidirectional data transfer can take place. The client and server can both send data and acknowledgments. We will study the rules of acknowledgment later in the chapter; for the moment, it is enough to know that data traveling in the same direction as an acknowledgment are carried on the same segment. The acknowledgment is piggybacked with the data. Figure 23.19 shows an example. In this example, after connection is established, the client sends 2000 bytes of data in two segments. The server then sends 2000 bytes in one segment. The client sends one more segment. The first three segments carry both data and acknowledgment, but the last segment carries only an acknowledgment because there are no more data to be sent. Note the values of the sequence and acknowledgment numbers. The data segments sent by the client have the PSH (push) flag set so that the server TCP knows to deliver data to the server process as soon as they are received. The segment from the server, on the other hand, does not set the push flag. Most TCP implementations have the option to set or not set this flag.

Figure 23.19 *Data transfer*



Connection Termination

Three-Way Handshaking Most implementations today allow three-way handshaking

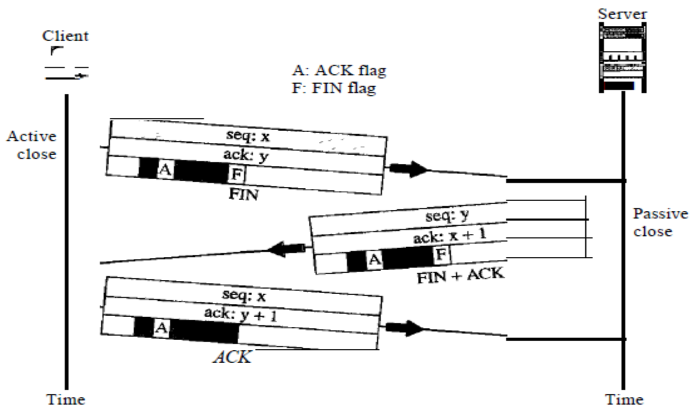
for connection termination.

1. In a normal situation, the client TCP, after receiving a close command from the client process, sends the first segment, a FIN segment in which the FIN flag is set. Note that a FIN segment can include the last chunk of data sent by the client, or it can be just a control segment. If it is only a control segment, it consumes only one sequence number.

2.The server TCP, after receiving the FIN segment, informs its process of the situation and sends the second segment, a FIN +ACK segment, to confirm the receipt of the FIN segment from the client and at the same time to announce the closing of the connection in the other direction. This segment can also contain the last chunk of data from the server. If it does not carry data, it consumes only one sequence number.

3.The client TCP sends the last segment, an ACK segment, to confirm the receipt of the FIN segment from the TCP server. This segment contains the acknowledgment number, which is 1 plus the sequence number received in the FIN segment from the server. This segment cannot carry data and consumes no sequence numbers.

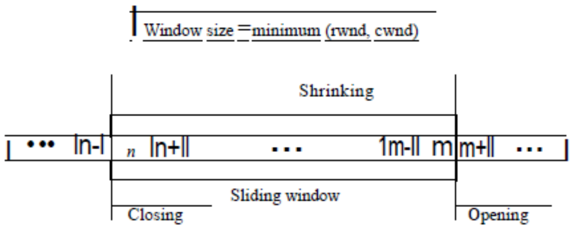
Figure 23.20 *Connection termination using three-way handshaking*



FLOW CONTROL AND BUFFERING

- TCP uses a sliding window, to handle flow control.
- The sliding window protocol used by TCP, however, is something between the Go-Back-N and Selective Repeat sliding window.
- The sliding window protocol in TCP looks like

Figure 23.22 *Sliding window*



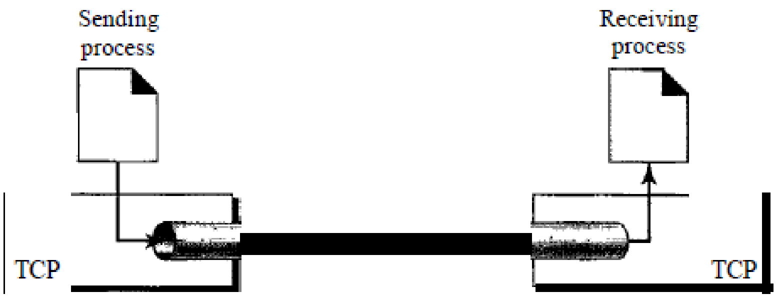
- A sliding window is used to make transmission more efficient as weD as to control the flow of data so that the destination does not become overwhelmed with data.

- TCP sliding windows are byte-oriented.
- Some points about TCP sliding windows:
- o The size of the window is the lesser of rwnd and cwnd. o The source does not have to send a full window's worth of data.
- o The window can be opened or closed by the receiver, but should not be shrunk.
- o The destination can send an acknowledgment at any time as long as it does not result in a shrinking window.
- o The receiver can temporarily shut down the window; the sender, however, can always send a segment of 1 byte after the window is shut down.

Stream Delivery Service: Buffer

- TCP, on the other hand, allows the sending process to deliver data as a stream of bytes and allows the receiving process to obtain data as a stream of bytes.
- TCP creates an environment in which the two processes seem to be connected by an imaginary "tube“ that carries their data across the Internet.

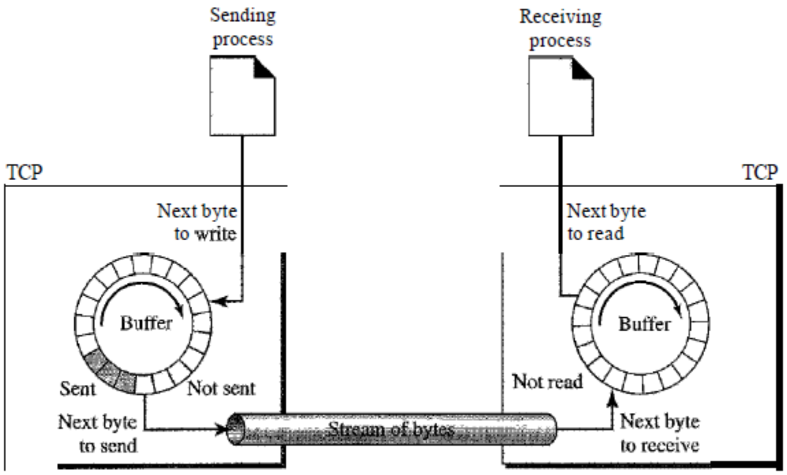
Figure 23.13 *Stream delivery*



Sending and Receiving Buffers

- Because the sending and the receiving processes may not write or read data at the same speed, TCP needs buffers for storage.
- There are two buffers, the sending buffer and the receiving buffer, one for each direction

Figure 23.14 *Sending and receiving buffers*



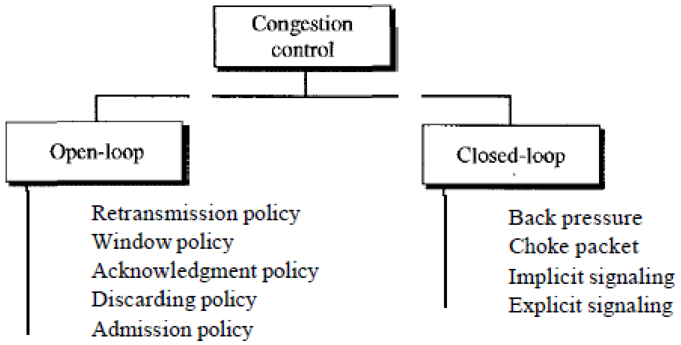
CONGESTION

An important issue in a packet-switched network is **congestion.** Congestion in a network may occur if the **load** on the network-the number of packets sent to the network-is greater than the capacity of the network-the number of packets a network can handle. **Congestion control** refers to the mechanisms and techniques to control the congestion and keep the load below the capacity.

CONGESTION CONTROL TECHNIQUES

Congestion control refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. In general, we can divide congestion control mechanisms into two broad categories: open-loop congestion control (prevention) and closed-loop congestion control (removal)

Figure 24.5 *Congestion control categories*



Open-Loop Congestion Control

In open-loop congestion control, policies are applied to prevent congestion before it happens. In these mechanisms, congestion control is handled by either the source or the destination.

Retransmission Policy

Retransmission is sometimes unavoidable. If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. Retransmission in general may increase congestion in the network. However, a good retransmission policy can prevent congestion. The retransmission policy and the retransmission timers must be designed to optimize efficiency and at the same time prevent congestion. For example, the retransmission policy used by TCP (explained later) is designed to prevent or alleviate congestion.

Window Policy

The type of window at the sender may also affect congestion. The Selective Repeat window is better than the Go-Back-N window for congestion control. In the Go-Back-N window, when the timer for a packet times out, several packets may be resent, although some may have arrived safe and sound at the receiver. This duplication may make the congestion worse. The Selective Repeat window, on the other hand, tries to send the specific packets that have been lost or corrupted.

Acknowledgment Policy

The acknowledgment policy imposed by the receiver may also affect congestion. If the receiver does not acknowledge every packet it receives, it may slow down the sender and help prevent congestion. Several approaches are used in this case. A receiver may send an acknowledgment only if it has a packet to be sent or a special timer expires. A receiver may decide to acknowledge only N packets at a time. We need to know that the acknowledgments are also part of the load in a network. Sending fewer acknowledgments means imposing less load on the network.

Discarding Policy

A good discarding policy by the routers may prevent congestion and at the same time may not harm the integrity of the transmission. For example, in audio transmission, if the policy is to discard less sensitive packets when congestion is likely to happen, the quality of sound is still preserved and congestion is prevented or alleviated.

Admission Policy

An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual-circuit networks. Switches in a flow first check the resource requirement of a flow before admitting it to the network. A router can deny establishing a virtual circuit connection if there is congestion in the network or if there is a possibility of future congestion.

Closed-Loop Congestion Control

Closed-loop congestion control mechanisms try to alleviate congestion after it happens.

Several mechanisms have been used by different protocols.

Backpressure

The technique of backpressure refers to a congestion control mechanism in which a congested node stops receiving data from the immediate upstream node or nodes. This may cause the upstream node or nodes to become congested, and they, in turn, reject data from their upstream nodes or nodes. And so on. Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow, to the source. The backpressure technique can be applied only to virtual circuit networks, in which each node knows the upstream node from which a flow of data is corning.

Choke Packet

A choke packet is a packet sent by a node to the source to inform it of congestion. Note the difference between the backpressure and choke packet methods. In backpressure, the warning is from one node to its upstream node, although the warning may eventually reach the source station. In the choke packet method, the warning is from the router, which has encountered congestion, to the source station directly. The intermediate nodes through which the packet has traveled are not warned. We have seen an example of this type of control in ICMP.

Implicit Signaling

In implicit signaling, there is no communication between the congested node or nodes and the source. The source guesses that there is a congestion somewhere in the network from other symptoms. For example, when a source sends several packets and there is no acknowledgment for a while, one assumption is that the network is congested. The delay in receiving an acknowledgment is interpreted as congestion in the network; the source should slow down. We will see this type of signaling when we discuss TCP congestion control later in the chapter.

Explicit Signaling

The node that experiences congestion can explicitly send a signal to the source or destination. The explicit signaling method, however, is different from the choke packet method. In the choke packet method, a separate packet is used for this purpose; in the explicit signaling method, the signal is included in the packets that carry data. Explicit signaling, as we will see in Frame Relay congestion control, can occur in either the forward or the backward direction.

Backward Signaling A bit can be set in a packet moving in the direction opposite to the congestion. This bit can warn the source that there is congestion and that it needs to slow down to avoid the discarding of packets.

Forward Signaling A bit can be set in a packet moving in the direction of the congestion. This bit can warn the destination that there is congestion. The receiver in this case can use policies, such as slowing down the acknowledgments, to alleviate the congestion.

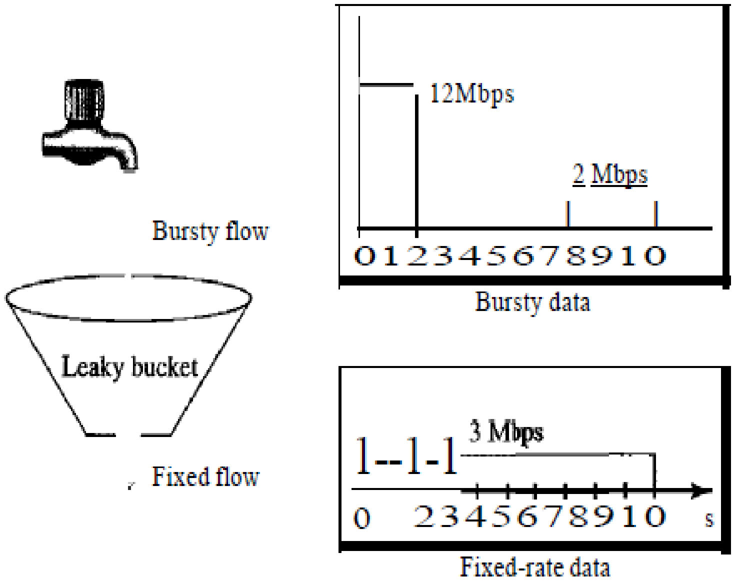
TRAFFIC SHAPING (Congestion Control Algorithm)

Traffic shaping is a mechanism to control the amount and the rate of the traffic sent to the network. Two algorithms can shape traffic: leaky bucket and token bucket.

Leaky Bucket

If a bucket has a small hole at the bottom, the water leaks from the bucket at a constant rate as long as there is water in the bucket. The rate at which the water leaks does not depend on the rate at which the water is input to the bucket unless the bucket is empty. The input rate can vary, but the output rate remains constant. Similarly, in networking, a technique called leaky bucket can smooth out bursty traffic. Bursty chunks are stored in the bucket and sent out at an average rate.

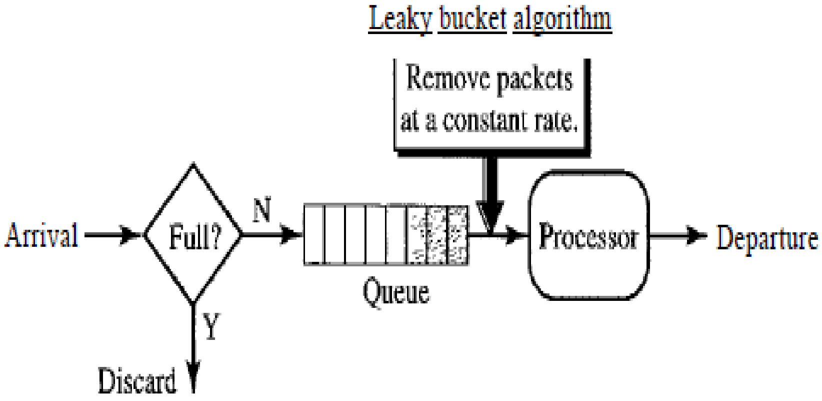
Figure 24.19 *Leaky bucket*



In the figure, we assume that the network has committed a bandwidth of 3 Mbps for a host. The use of the leaky bucket shapes the input traffic to make it conform to this commitment. In Figure 24.19 the host sends a burst of data at a rate of 12 Mbps for 2 s, for a total of 24 Mbits of data. The host is silent for 5 s and then sends data at a rate of 2 Mbps for 3 s, for a total of 6 Mbits of data. The leaky bucket smooths the traffic by sending out data at a rate of 3 Mbps during the same 10 s. Without the leaky bucket, the beginning burst may have hurt the network by consuming more bandwidth than is set aside for this host. We can also see that the leaky bucket may prevent congestion. As an analogy, consider the freeway during rush hour (bursty traffic). If, instead, commuters could stagger their working hours, congestion on our freeways could be avoided.

A simple leaky bucket implementation is shown in Figure 24.20. A FIFO queue holds the packets. If the traffic consists of fixed-size packets (e.g., cells in ATM networks), the process removes a fixed number of packets from the queue at each tick of the clock. If the traffic consists of variable-length packets, the fixed output rate must be based on the number of bytes or bits.

Figure 24.20 *Leaky bucket implementation*



The following is an algorithm for variable-length packets:

1. Initialize a counter to *n* at the tick of the clock.

2. If *n* is greater than the size of the packet, send the packet and decrement the counter by the packet size.

3.Repeat this step until *n* is smaller than the packet size.

4.Reset the counter and go to step 1.

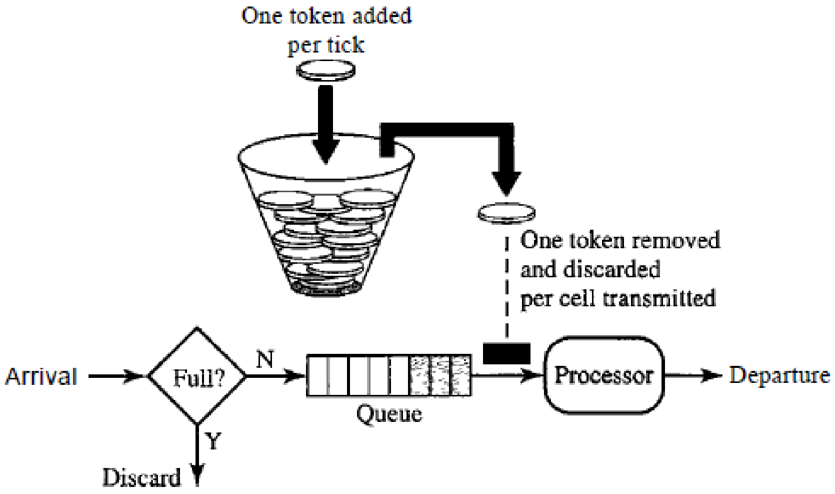
Token Bucket

The leaky bucket is very restrictive. It does not credit an idle host. For example, if a host is not sending for a while, its bucket becomes empty. Now if the host has bursty data, the leaky bucket allows only an average rate. The time when the host was idle is not taken into account. On the other hand, the token bucket algorithm allows idle hosts to accumulate credit for the future in the form of tokens. For each tick of the clock, the system sends *n* tokens to the bucket. The system removes one token for

every cell (or byte) of data sent. For example, if *n* is 100 and the host is idle for 100 ticks, the bucket collects 10,000 tokens. Now the host can consume all these tokens in one tick with 10,000 cells, or the host takes 1000 ticks with 10 cells per tick. In other words, the host can send bursty data as long as the bucket is not empty. Figure 24.21 shows the idea. The token bucket can easily be implemented with a counter. The token is initialized to zero. Each time a token is added, the counter is incremented by 1. Each time a unit of data is sent, the counter is decremented by 1. When the counter is zero, the host

cannot send data.

Figure 24.21 *Token bucket*



CHAPTER 6-APPLICATION LAYER

The application layer is a layer in the Open Systems Interconnection (OSI) seven-layer model and in the TCP/IP protocol suite. It consists of protocols that focus on process-to-process communication across an IP network and provides a firm communication interface and end-user services.

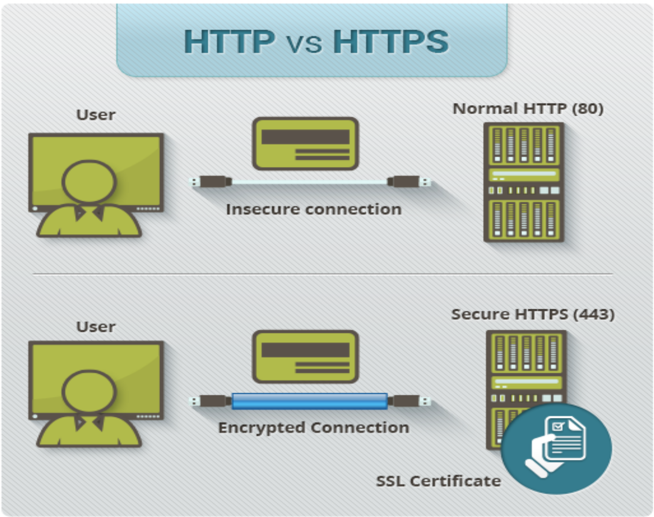
WEB: HTTP AND HTTPS

"World Wide Web " or "the *Web*", a hypertext system that operates over the Internet. *Web* (*web* browser) (previously Epiphany), the *web* browser included with GNOME desktop environment. *Web*.com, a public company that offers websites and other services for small businesses and consumers.

The Hypertext Transfer Protocol (HTTP) is an application protocol for distributed, collaborative, hypermedia information systems. HTTP is the foundation of data communication for the World Wide Web. Hypertext is structured text that uses logical links (hyperlinks) between nodes containing text.

Hyper Text Transfer Protocol Secure (HTTPS) is the secure version of HTTP, the protocol over which data is sent between your browser and the website that you are connected to. The 'S' at the end of HTTPS stands for 'Secure'. It means all communications between your browser and the website are encrypted. HTTPS is often used to protect highly confidential online transactions like online banking and online shopping order forms.

Web browsers such as Internet Explorer, Firefox and Chrome also display a padlock icon in the address bar to visually indicate that a HTTPS connection is in effect.



HTTPS pages typically use one of two secure protocols to encrypt communications - SSL (Secure Sockets Layer) or TLS (Transport Layer Security). Both the TLS and SSL protocols use what is known as an 'asymmetric' Public Key Infrastructure (PKI) system. An asymmetric system uses two 'keys' to encrypt communications, a 'public' key and a 'private' key. Anything encrypted with the public key can only be decrypted by the private key and vice-versa.

Instead of Hyper Text Transfer Protocol (HTTP), this website uses **Hyper Text Transfer Protocol Secure (**HTTPS**)**.
Using HTTPS, the computers agree on a "code" between them, and then they scramble the messages using that "code" so that no one in between can read them. This keeps your information safe from hackers.

HTTP

Http stands for Hyper Text Transfer Protocol. It allows World Wide Web users to transferring information like image, text, video, music, graphic and other files on web pages. Http is basically used to access html pages and also other resources can be accessible using HTTP.

HTTP is a request-response protocol in the client-server computing model. When you enter http:// in front of the address tells the browser to connect over HTTP. For example, when you enter a URL (http://www.abc.com) in your web browser, this sends an HTTP command to the Web server to fetch and transfer the requested web page. Here, your web browser is your client and your website host as a server.

HTTPS

HTTPS stands for Hypertext Transfer Protocol Secure. HTTPS is a protocol which uses an encrypted HTTP connection by transport-layer security.

Sometimes, the clients may be exchanging private information with a server, which needs to be secured for preventing some hacking issue. For this reason, HTTPS was developed by Netscape Corporation to allow authorization and secured transactions.

Here is the fact of HTTP:

- HTTPS uses a port 443 by default to transfer the information.
- HTTPS URLs begin with “https://”.
- The HTTPS is first used in HTTPS V1.1 and defined in RFC 2616.

When you interact with a website, such as trying to retrieve a webpage, data is sent back and forth between you and the web server. The S in HTTPS signals the inclusion of the "Secure Sockets Layer" - more colloquially known as "SSL" - whose function is to encrypt the data that is exchanged between you and the website.

When you connect to the website your computer will receive their SSL Certificate and checks it against the server's credentials. Your computer and the web server then figure out the best way to encrypt information, exchange special "keys" with one another, and then give it a small test drive to ensure they can properly share encrypted information. Once the two are ready to go, they each give the green light and exchange encrypted information.

Because both your computer and the website's server have to verify their identities, set up their special way of encoding/decoding that is unique to them, and always transfer information in a secure fashion many of the typical attack vectors available to hackers are rendered insignificant. They will have spent more resources in terms of computing power and time than they could hope to get back through your information, which will likely have changed over the years it would take to properly decrypt. And the method of decryption is unique to each connection.

FILE TRANSFER

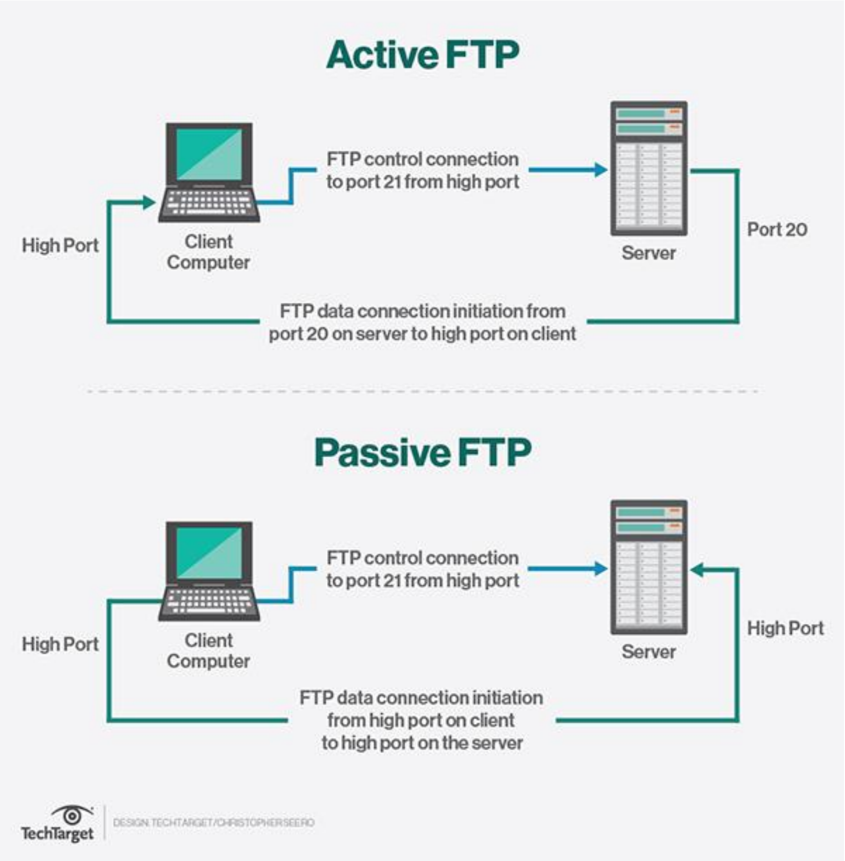
FTP

The File Transfer Protocol (FTP) is a standard network protocol used to transfer computer files between a client and server on a computer network. FTP is built on client-server model architecture and uses separate control and data connections between the client and the server.

File Transfer Protocol (FTP) is a standard Internet protocol for transmitting files between computers on the Internet over TCP/IP connections.

FTP is a client-server protocol that relies on two communications channels between client and server: a command channel for controlling the conversation and a data channel for transmitting file content. Clients initiate conversations with servers by requesting to download a file. Using FTP, a client can upload, download, delete, and rename, move and copy files on a server. A user typically needs to log on to the FTP server, although some servers make some or all of their content available without login, also known as anonymous FTP.

FTP sessions work in passive or active modes. In active mode, after a client initiates a session via a command channel request, the server initiates a data connection back to the client and begins transferring data. In passive mode, the server instead uses the command channel to send the client the information it needs to open a data channel. Because passive mode has the client initiating all connections, it works well across firewalls and Network Address Translation (NAT) gateways.


Active FTP
FTP control connection
to port 21 from high port
High Port
Client
Computer
Server
Port 20
FTP data connection initiation from
port 20 on server to high port on client
Passive FTP
FTP control connection
to port 21 from high port
High Port
Client
Computer
Server
High Port
FTP data connection initiation
from high port on client
to high port on the server
TechTarget
DESIGN: TECHTARGET/CHRISTOPHERSEERO

FTP was originally defined in 1971, prior to the definition of TCP and IP, and has been redefined many times -- e.g., to use TCP/IP (RFC 765 and RFC 959), and then Internet Protocol Version 6 (IPv6), (RFC 2428). Also, because it was defined without much concern for security, it has been extended many times to improve security: for example, versions that encrypt via a TLS connection (FTPS) or that work with Secure File Transfer Protocol (SFTP), also known as SSH File Transfer Protocol.

PUTTY

PuTTY is a free and open-source terminal emulator, serial console and network file transfer application. It supports several network protocols, including SCP, SSH, Telnet, rlogin, and raw socket connection. It can also connect to a serial port. The name "PuTTY" has no definitive meaning.

PuTTY was originally written for Microsoft Windows, but it has been ported to various other operating systems. Official ports are available for some Unix-like platforms, with work-in-progress ports to Classic Mac OS and macOS, and unofficial ports have been contributed to platforms such as Symbian,Windows Mobile and Windows Phone.

PuTTY was written and is maintained primarily by Simon Tatham and is currently beta software.

Features

PuTTY supports many variations on the secure remote terminal, and provides user control over the SSH encryption key and protocol version, alternate ciphers such as 3DES, Arcfour, Blowfish, DES, and Public-key authentication. It also can emulate control sequences from xterm, VT102 or ECMA-48 terminal emulation, and allows local, remote, or dynamic port forwarding with SSH (including X11 forwarding). The network communication layer supports IPv6, and the SSH protocol supports the zlib@openssh.com delayed compression scheme. It can also be used with local serial port connections.

PuTTY comes bundled with command-line SCP and SFTP clients, called "pscp" and "psftp" respectively, and plinks, a command-line connection tool, used for non-interactive sessions. PuTTY is a program that connects one device to another over the network. It supports SSH and Telnet, among others. PuTTY is a “client” application that talks to a “host”. The host must be running an SSH server (which is often the case for iMX Linux enabled systems).A Windows version exists and this is ideal for transferring files between your Windows PC and a Linux Platform.

WinSCP (Windows Secure Copy) is a free and open source SFTP, FTP, WebDAV and SCP client for Microsoft Windows. Its main function is secure file transfer between a local and a remote computer. Beyond this, WinSCP offers basic file manager and file synchronization functionality. For secure transfers, it uses Secure Shell (SSH) and supports the SCP protocol in addition to SFTP.[3]

Development of WinSCP started around March 2000 and continues. Originally it was hosted by the University of Economics in Prague, where its author worked at the time. Since July 16, 2003, it is licensed under the GNU GPL and hosted on SourceForge.net.

WinSCP is based on the implementation of the SSH protocol from PuTTY and FTP protocol from FileZilla. It is also available as a plug-in for Altap Salamander file manager, and there exists a third-party plug-in for the FAR file manager.

- Graphical user interface
- Translated into several languages
- Integration with Windows (Drag-and-drop, URL, shortcut icons)
- All common operations with files
- Support for SFTP and SCP protocols over SSH-1 and SSH-2, FTP protocol and WebDAV protocol.[8]
- Batch file scripting, command-line interface and .NET wrapper
- Directory synchronization in several semi or fully automatic ways
- Integrated text editor
- Support for SSH password, keyboard-interactive, public key and Kerberos (GSS) authentication

- Integrates with Pageant (PuTTY authentication agent) for full support of public key authentication with SSH
- Choice of Windows Explorer–like or Norton Commander–like interfaces
- Optionally stores session information
- Optionally import session information from PuTTY sessions in the registry
- Able to upload files and retain associated original date/timestamps, unlike FTP clients

ELECTRONIC MAIL:

E-mail (electronic mail) is the exchange of computer-stored messages by telecommunication. (Some publications spell it email; we prefer the currently more established spelling of e-mail.) E-mail messages are usually encoded in ASCII text.

Overview

SMTP, POP3 and IMAP are TCP/IP protocols used for mail delivery. If you plan to set up an email server such as hMailServer, you must know what they are used for. Each protocol is just a specific set of communication rules between computers.

SMTP

SMTP stands for Simple Mail Transfer Protocol. SMTP is used when email is delivered from an email client, such as Outlook Express, to an email server or when email is delivered from one email server to another. SMTP uses port 25.

SMTP provides a set of codes that simplify the communication of email messages between email servers (the network computer that handles email coming to you and going out). It's a kind of shorthand that allows a server to break up different parts of a message into categories the other server can understand. When you send a message out, it's turned into strings of text that are separated by the code words (or numbers) that identify the purpose of each section.

SMTP provides those codes, and email server software is designed to understand what they mean. As each message travels towards its destination, it sometimes passes through a number of computers as well as their individual MTAs. As it does, it's briefly stored before it moves on to the next computer in the path. Think of it as a letter going through different hands as it winds its way to the right mailbox.

POP3

POP3 stands for Post Office Protocol. POP3 allows an email client to download an email from an email server. The POP3 protocol is simple and does not offer many features except for download. Its design assumes that the email client downloads all available email from the server, deletes them from the server and then disconnects. POP3 normally uses port 110.

(1) POP is short for **P**ost **O**ffice **P**rotocol, a protocol used to retrieve e-mail from a mail server. Most e-mail applications (sometimes called an e-mail client) use the POP protocol, although some can use the newer IMAP (Internet Message Access Protocol).
There are two versions of POP. The first, called POP2, became a standard in the mid-80's and requires SMTP to send messages. The newer version, POP3, can be used with or without SMTP.
(2) Pop is short for **p**oint **o**f **p**resence, an access point to the Internet. ISPs have typically multiple POPs. A point of presence is a physical location, either part of the facilities of a telecommunications provider that the ISP rents or a separate location from the telecommunications provider, that houses servers, routers, ATM switches and digital/analog call aggregators.
(3) Pop is short for **P**rogrammed **Op**erator (POP), a pseudo-opcode in a virtual machine language executed by an interpretive program. The Programmed Operator instructions provide the ability to define an instruction set for efficient encoding by calling subprograms into primary memory.
(4) POP is short for **p***icture*-**o***utside*-**p***icture* POP is a feature found on some televisions that allows the user to divide the screen into two same-size pictures, enabling you to view a second program. *Compare with picture-in-picture (*PIP*).*

IMAP

IMAP stands for Internet Message Access Protocol. IMAP shares many similar features with POP3. It, too, is a protocol that an email client can use to download email from an email server. However, IMAP includes many more features than POP3. The IMAP protocol is designed to let users keep their email on the server. IMAP requires more disk space on the server and more CPU resources than POP3, as all emails are stored on the server. IMAP normally uses port 143. Here is more information about IMAP.

Short for **I***nternet* **M***essage* **A***ccess* **P***rotocol,* a protocol for retrieving e-mail messages. The latest version, *IMAP4,* is similar to *POP3* but supports some additional features. For example, with IMAP4, you can search through your e-mail messages for keywords while the messages are still on mail server. You can then choose which messages to download to your machine.
IMAP was developed at Stanford University in 1986.

Examples

Suppose you use hMailServer as your email server to send an email to bill@microsoft.com.

1. You click Send in your email client, say, Outlook Express.
2. Outlook Express delivers the email to hMailServer using the SMTP protocol.
3. hMailServer delivers the email to Microsoft's mail server, mail.microsoft.com, using SMTP.

4. Bill's Mozilla Mail client downloads the email from mail.microsoft.com to his laptop using the POP3 protocol (or IMAP).

DNS

Short for **D***omain* **N***ame* **S***ystem* (or **S***ervice* or **S***erver*), an Internet service that translates *domain names* into IP addresses. Because domain names are alphabetic, they're easier to remember. The Internet however, is really based on IP addresses. Every time you use a domain name, therefore, a DNS service must translate the name into the corresponding IP address. For example, the domain name *www.example.com* might translate to *198.105.232.4*.
The DNS system is, in fact, its own network. If one DNS server doesn't know how to translate a particular domain name, it asks another one, and so on, until the correct IP address is returned.
Short for **d***igital* **n***ervous* **s***ystem*, a term coined by Bill Gates to describe a network of personal computers that make it easier to obtain and understand information.

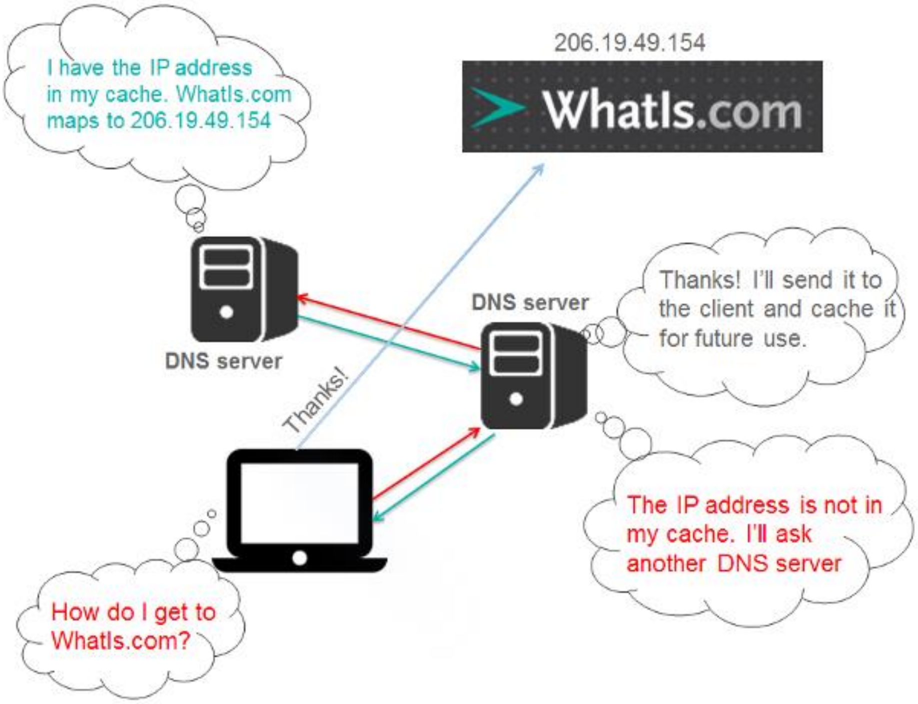
Domain names give people a more intuitive way to access content or services than IP addresses: www.techtarget.com instead of 206.19.49.149, for example. Most URLs are built around the domain name of the web server fielding the request: e.g., http://searchnetworking.techtarget.com/definition/DNS-attack. Web browsing and most other internet activity rely on DNS behind the scenes to quickly provide the information necessary to connect users to remote hosts.

Why is DNS important?

Having a single DNS server somewhere that maintained a complete central list of domain name or IP address mappings would be impractical. There are too many mappings, they change too often and the number of requests for address or name lookups would overwhelm any system. As a result, DNS is distributed throughout the internet in a hierarchy of authority. Access providers and enterprises, as well as governments, universities and other organizations, typically have their own assigned ranges of IP addresses and an assigned domain name; they also typically run DNS servers to manage the mapping of those names to those addresses.

How does DNS work?

DNS servers answer questions from both inside and outside their own domains. When a server receives a request from outside the domain for information about a name or address inside the domain, it provides the authoritative answer. When a server receives a request from inside its own domain for information about a name or address outside that domain, it passes the request out to another server -- usually one managed by its internet service provider. If that server does not know the answer or the authoritative source for the answer, it will reach out to the DNS servers for the top-level domain -- e.g., for all of .com or .edu. Then, it will pass the request down to the authoritative server for the specific domain -- e.g., techtarget.com or stkate.edu; the answer flows back along the same path.


I have the IP address in my cache. WhatIs.com maps to 206.19.49.154
206.19.49.154
WhatIs.com
DNS server
DNS server
Thanks! I'll send it to the client and cache it for future use.
Thanks!
The IP address is not in my cache. I'll ask another DNS server
How do I get to WhatIs.com?

P2P APPLICATION

What is Peer to Peer (P2P) Application?

P2P is nothing but just Peer to Peer networking. As we have Server - Client Model and Peer to Peer network in the same way these P2P applications work. You need a P2P program that will be installed on your computer it creates a community of P2P application users and it creates a virtual network between these users. For the user it will look as it is in a Peer to Peer network and he can share files from his local computer and download files shared by other users. It is very similar to our Instant Messaging like Yahoo, AOL or GTalk where even though to who we are taking to are on a different network but a virtual network is created where it looks like we are on a same network and we can share files and chat. The P2P application has been very much in demand from last couple of years. A P2P application is mainly used for sharing Music, Movies, Games and other files.

What are the disadvantages of Peer to Peer (P2P) Application?

Is it estimated that for any given ISP 60 to 80% of their traffic is consumed by P2P traffic. So even in your office if people are using P2P application they will consume a huge amount of bandwidth without production.P2P application is very famous for distributing Pirated software. Your users might be using pirated software on their computers and Auditors will never appreciate that. Symantec Underground Economy says that "The annual global cost to businesses of software piracy in one 2007 study puts the cost at nearly $40 billion"

You can never trust the file you are downloading from a remote user in P2P environment.90% of the files contain malwares. Thus if your users are using P2P application there is very high rate of Virus Outbreak in your network that too very frequently. In 2008 10% of malware were propagated via P2P applications. Even the very infamous W32.Downadup also propagated and updated itself via P2P applications.

P2P is a very famous mechanism for distributing Bots, Spywares, Adware, Trojans, Rootkits, Worms and other types of malwares.

Since it is very easy to change the port for these P2P applications it is very difficult to block this traffic. It is strictly not advised to have P2P application allowed in your network. Enterprises should take measures to prevent P2P clients from being installed on any computers on the network. End users who download files from P2P networks should scan all such files with a regularly updated antivirus product.

How to block Peer to Peer Applications (P2P) using Symantec Endpoint Protection?

There are 3 ways of blocking Peer to Peer Applications on your network using Symantec Endpoint Protection.

1. Blocking Peer to Peer Applications using Intrusion Prevention System

Open Symantec Endpoint Protection Manager

Click on Policies -> Intrusion Prevention -> Edit Intrusion Prevention Policies .go to Exceptions -> Click on Add.

Then under Show Category scroll it down and Select Peer to Peer.

On the bottom right hand side of the policy click on Select all -> click next

Action -Block

Log - Log the Traffic

SOCKET PROGRAMMING

Sockets provide the communication mechanism between two computers using TCP. A client program creates a socket on its end of the communication and attempts to connect that socket to a server.

When the connection is made, the server creates a socket object on its end of the communication. The client and the server can now communicate by writing to and reading from the socket.

The java.net.Socket class represents a socket, and the java.net.ServerSocket class provides a mechanism for the server program to listen for clients and establish connections with them.

The following steps occur when establishing a TCP connection between two computers using sockets −

- The server instantiates a ServerSocket object, denoting which port number communication is to occur on.
- The server invokes the accept () method of the ServerSocket class. This method waits until a client connects to the server on the given port.
- After the server is waiting, a client instantiates a Socket object, specifying the server name and the port number to connect to.
- The constructor of the Socket class attempts to connect the client to the specified server and the port number. If communication is established, the client now has a Socket object capable of communicating with the server.
- On the server side, the accept() method returns a reference to a new socket on the server that is connected to the client's socket.

After the connections are established, communication can occur using I/O streams. Each socket has both an OutputStream and an InputStream. The client's OutputStream is connected to the server's InputStream, and the client's InputStream is connected to the server's OutputStream.

TCP is a two-way communication protocol, hence data can be sent across both streams at the same time. Following are the useful classes providing complete set of methods to implement sockets.

APPLICATION SERVER CONCEPT

PROXY SERVER

In computer networks, a **proxy server** is a server (a computer system or an application) that acts as an intermediary for requests from clients seeking resources from other servers.[1] A client connects to the proxy server, requesting some service, such as a file, connection, web page, or other resource available from a different server and the proxy server evaluates the request as a way to simplify and control its complexity.

PROXY CACHING

Proxy caching is a feature of **proxy** servers that stores content on the **proxy** server itself, allowing web services to share those resources to more users. The **proxy**server coordinates with the source server to **cache** documents such as files, images and web pages.

CONCEPT OF TRAFFIC ANALYZER

A) MRTG

The Multi Router Traffic Grapher (MRTG) is a tool to monitor the traffic load on network links. MRTG generates HTML pages containing PNG images which provide a LIVE visual representation of this traffic. Check http://www.stat.ee.ethz.ch/mrtg/ to see what it does.

MRTG consists of a Perl script which uses SNMP to read the traffic counters of your routers and a fast C program which logs the traffic data and creates beautiful graphs representing the traffic on the monitored network connection. These graphs are embedded into WebPages which can be viewed from any modern Web-browser.

In addition to a detailed daily view, MRTG also creates visual representations of the traffic seen during the last seven days, the last five weeks and the last twelve months. This is possible because MRTG keeps a log of all the data it has pulled from the router. This log is automatically consolidated so that it does not grow over time, but still contains all the relevant data for all the traffic seen over the last two years. This is all performed in an efficient manner. Therefore you can monitor 200 or more network links from any halfway decent UNIX box.

MRTG is not limited to monitoring traffic, though. It is possible to monitor any SNMP variable you choose. You can even use an external program to gather the data which should be monitored via MRTG. People are using MRTG, to monitor things such as System Load, Login Sessions, Modem availability and more. MRTG even allows you to accumulate two or more data sources into a single graph.

B) PRTG

Paessler Router Traffic Grapher, renamed **PRTG Network Monitor** from version 7 in 2008, is a server up-time and utilization, network monitoring and bandwidth usage software package for server infrastructure from Paessler AG. It can monitor and classify bandwidth usage in a network using SNMP, packet sniffing and Net flow. It services Microsoft Windows and Linux. It was derived from the open-source Multi Router Traffic Grapher (MRTG) project. A version with a limited number of sensors is available free of charge.

C) SNMP

Simple Network Management Protocol (SNMP) is a popular protocol for network management. It is used for collecting information from, and configuring, network devices, such as servers, printers, hubs, switches, and routers on an Internet Protocol (IP) network. Microsoft Windows Server 2003 provides SNMP agent software that works with third-party SNMP management software to monitor the status of managed devices and applications.

D) Packet Tracer

Cisco Packet Tracer is a powerful network simulation program that allows students to experiment with network behavior and ask "what if" questions. As an integral part of the Networking Academy comprehensive learning experience, Packet Tracer provides simulation, visualization, authoring, assessment, and collaboration capabilities and facilitates the teaching and learning of complex technology concepts.

Packet Tracer supplements physical equipment in the classroom by allowing students to create a network with an almost unlimited number of devices, encouraging practice, discovery, and troubleshooting. The simulation-based learning environment helps students develop 21st century skills such as decision making, creative and critical thinking, and problem solving. Packet Tracer complements the Networking Academy curricula, allowing instructors to easily teach and demonstrate complex technical concepts and networking systems design.

E) Wire shark

Wireshark is a free and open source packet analyzer. It is used for network troubleshooting, analysis, software and communications protocol development, and education. Originally named **Ethereal**, the project was renamed Wireshark in May 2006 due to trademark issues.[4]

Wireshark is cross-platform, using the Qt widget toolkit in current releases to implement its user interface, and using pcap to capture packets; it runs on Linux, macOS, BSD, Solaris, some other Unix-like operating systems, and Microsoft Windows. There is also a terminal-based (non-GUI) version called TShark. Wireshark, and the other programs distributed with it such as TShark, are free software, released under the terms of the GNU General Public License.

CHAPTER 7-IPV6

IPV6

Internet Protocol version 6 (IPv6) is the most recent version of the Internet Protocol (IP), the communications protocol that provides an identification and location system for computers on networks and routes traffic across the Internet. IPv6 was developed by the Internet Engineering Task Force (IETF) to deal with the long-anticipated problem of IPv4 address exhaustion. IPv6 is intended to replace IPv4.

With the rapid growth of the Internet after commercialization in the 1990s, it became evident that far more addresses would be needed to connect devices than the IPv4 address space had available. IPv6 uses a 128-bit address, theoretically allowing 2^{128}, or approximately 3.4×10^{38} addresses. Several IPv6 transition mechanisms have been devised to permit communication between IPv4 and IPv6 hosts.

IPv6 addresses are represented as eight groups of four hexadecimal digits with the groups being separated by colons, for example 2001:0db8:0000:0042:0000:8a2e:0370:7334, but methods to abbreviate this full notation exist.

LIMITATIONS OF IPV4

The network layer protocol in the TCPIIP protocol suite is currently IPv4 (Internetworking Protocol, version 4). IPv4 provides the host-to-host communication between systems in the Internet. Although IPv4 is well designed, data communication has evolved since the inception of IPv4 in the 1970s. IPv4 has some deficiencies (listed below) that make it unsuitable for the fast-growing Internet.

o Despite all short-term solutions, such as subnetting, classless addressing, and NAT, address depletion is still a long-term problem in the Internet.

o The Internet must accommodate real-time audio and video transmission. This type of transmission requires minimum delay strategies and reservation of resources not provided in the IPv4 design.

o The Internet must accommodate encryption and authentication of data for some applications. No encryption or authentication is provided by IPv4.

ADVANTAGES OF IPV6

The next-generation IP, or IPv6, has some advantages over IPv4 that can be summarized

as follows:

- Larger address space. An IPv6 address is 128 bits long, Compared with the 32-bit address of IPv4, this is a huge increase in the address space.
- Better header format. IPv6 uses a new header format in which options are separated from the base header and inserted, when needed, between the base header and the upper-layer data. This simplifies and speeds up the routing process because most of the options do not need to be checked by routers.
- New options. IPv6 has new options to allow for additional functionalities.
- Allowance for extension. IPv6 is designed to allow the extension of the protocol if required by new technologies or applications.
- Support for resource allocation. In IPv6, the type-of-service field has been removed, but a mechanism has been added to enable the source to request special handling of the packet. This mechanism can be used to support traffic such as real-time audio and video.
- Support for more security. The encryption and authentication options in IPv6 provide

PACKET FORMAT

The IPv6 packet is shown in Figure. Each packet is composed of a mandatory base header followed by the payload. The payload consists of two parts: optional extension headers and data from an upper layer. The base header occupies 40 bytes, whereas the extension headers and data from the upper layer contain up to 65,535 bytes of information.

Base Header

Figure shows the base header with its eight fields. These fields are as follows:

o Version. This 4-bit field defines the version number of the IP. For IPv6, the value is 6.

o Priority. The 4-bit priority field defines the priority of the packet with respect to traffic congestion.

Figure 20.15 *IPv6 datagram header and payload*

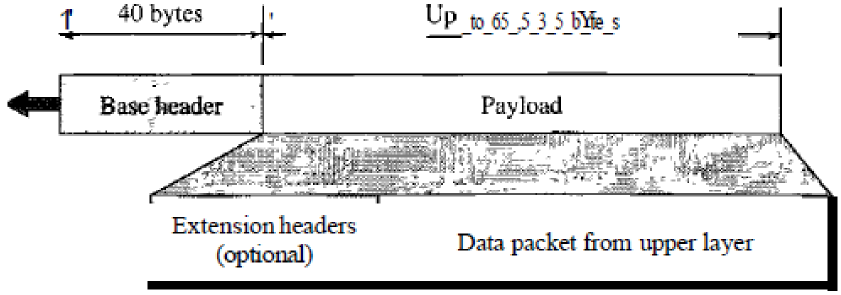
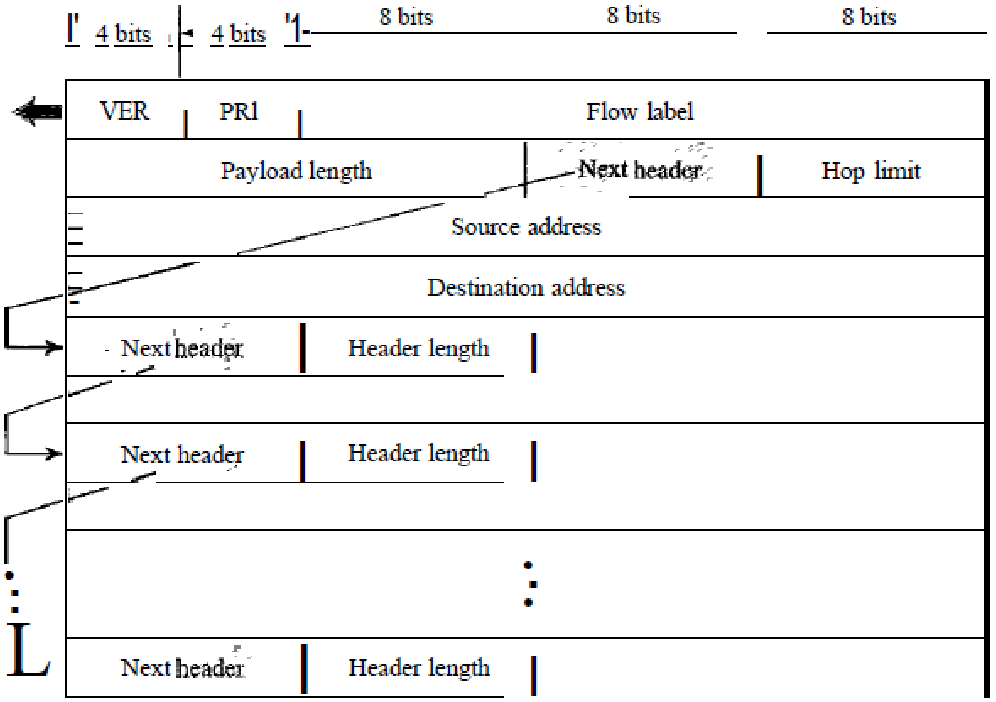


Figure 20.16 *Format of an IPv6 datagram*



- Flow label. The flow label is a 3-byte (24-bit) field that is designed to provide special handling for a particular flow of data.
- Payload length. The 2-byte payload length field defines the length of the IP datagram

excluding the base header.

- Next header. The next header is an 8-bit field defining the header that follows the base header in the datagram. The next header is either one of the optional extension headers used by IP or the header of an encapsulated packet such as UDP or TCP. Each extension header also contains this field. Table 20.6 shows the values of next headers. Note that this field in version 4 is called the protocol.
- Hop limit. This 8-bit hop limit field serves the same purpose as the TIL field in IPv4.
- Source address. The source address field is a 16-byte (128-bit) Internet address that identifies the original source of the datagram.
- Destination address. The destination address field is a 16-byte (128-bit) Internet address that usually identifies the final destination of the datagram. However, if source routing is used, this field contains the address of the next router.

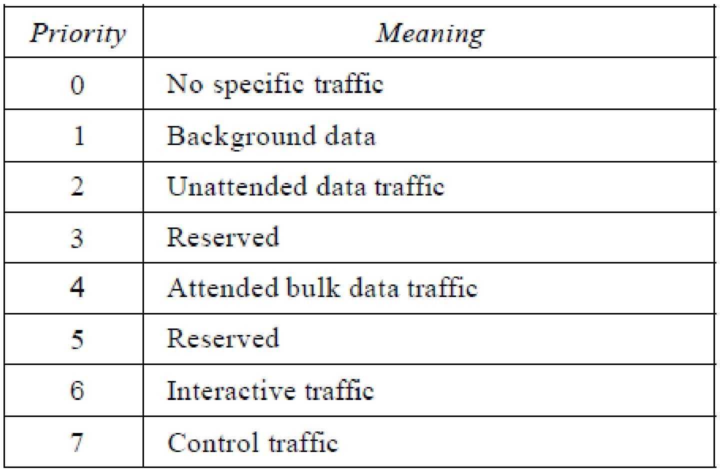
Priority

The priority field of the IPv6 packet defines the priority of each packet with respect to other packets from the same source. For example, if one of two consecutive Datagrams must be discarded due to congestion, the datagram with the lower packet priority will be discarded. IPv6 divides traffic into two broad categories: congestion-controlled and noncongestion-controlled.

Congestion-Controlled Traffic If a source adapts itself to traffic slowdown when there is congestion, the traffic is referred to as congestion-controlled traffic. For example, TCP, which uses the sliding window protocol, can easily respond to traffic. In congestion-controlled traffic, it is understood that packets may arrive delayed, lost, or out of order. Congestion-controlled data are assigned priorities from 0 to 7, as listed in

Table 20.7. A priority of 0 is the lowest; a priority of 7 is the highest.

Table 20.7 *Priorities for congestion-controlled traffic*



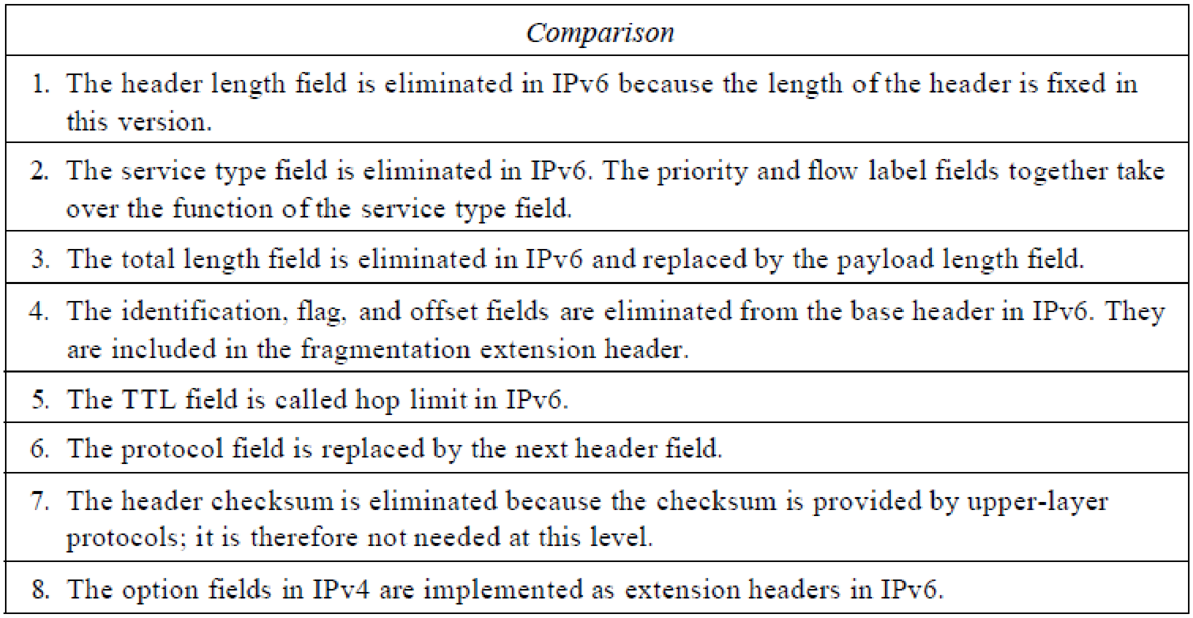
Priority	*Meaning*
0	No specific traffic
1	Background data
2	Unattended data traffic
3	Reserved
4	Attended bulk data traffic
5	Reserved
6	Interactive traffic
7	Control traffic

Noncongestion-Controlled Traffic: This refers to a type of traffic that expects minimum delay. Discarding of packets is not desirable. Retransmission in most cases is impossible. In other words, the source does not adapt itself to congestion. Real-time audio and video are examples of this type of traffic.

Flow Label

A sequence of packets, sent from a particular source to a particular destination that needs special handling by routers is called a flow of packets. The combination of the source address and the value of the flow label uniquely define a flow of packets. To a router, a flow is a sequence of packets that share the same characteristics, such as traveling the same path, using the same resources, having the same kind of security, and so on. A router that supports the handling of flow labels has a flow label table. In its simplest form, a flow label can be used to speed up the processing of a packet by a router. When a router receives a packet, instead of consulting the routing table and going through a routing algorithm to define the address of the next hop, it can easily look in a flow label table for the next hop.

Table 20.9 *Comparison between IPv4 and IPv6 packet headers*

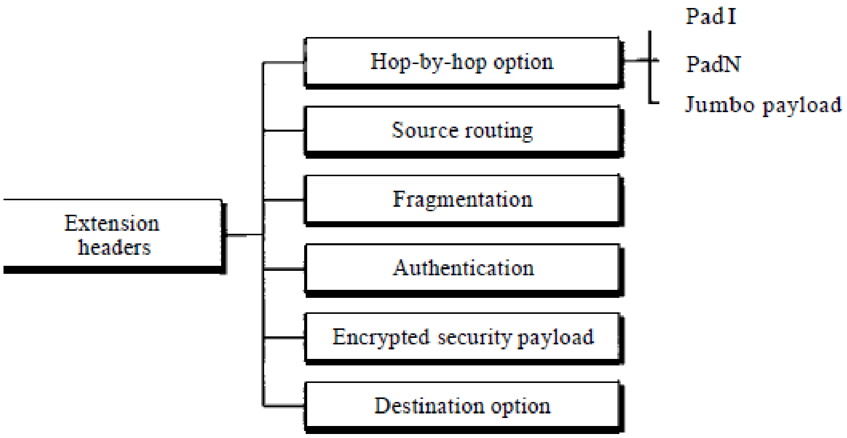


Comparison
1. The header length field is eliminated in IPv6 because the length of the header is fixed in this version.
2. The service type field is eliminated in IPv6. The priority and flow label fields together take over the function of the service type field.
3. The total length field is eliminated in IPv6 and replaced by the payload length field.
4. The identification, flag, and offset fields are eliminated from the base header in IPv6. They are included in the fragmentation extension header.
5. The TTL field is called hop limit in IPv6.
6. The protocol field is replaced by the next header field.
7. The header checksum is eliminated because the checksum is provided by upper-layer protocols; it is therefore not needed at this level.
8. The option fields in IPv4 are implemented as extension headers in IPv6.

EXTENSION HEADERS

The length of the base header is fixed at 40 bytes. However, to give greater functionality to the IP datagram, the base header can be followed by up to six extension headers. Many of these headers are options in IPv4. Six types of extension headers have been defined,

Figure 20.17 *Extension header types*



Hop-by-Hop Option

The hop-by-hop option is used when the source needs to pass information to all routers visited by the datagram. So far, only three options have been defined: Padl, PadN, and jumbo payload. The Padl option is 1 byte long and is designed for alignment purposes. PadN is similar in concept to Padi. The difference is that PadN is used when 2 or more bytes are needed for alignment. The jumbo payload option is used to define a payload longer than 65,535 bytes.

Source Routing

The source routing extension header combines the concepts of the strict source route and the loose source route options of IPv4.

Fragmentation

The concept of fragmentation is the same as that in IPv4. However, the place where fragmentation occurs differs. In IPv4, the source or a router is required to fragment if the size of the datagram is larger than the MTU of the network over which the datagram travels. In IPv6, only the original source can fragment. A source must use a path MTU discovery technique to find the smallest MTU supported by any network on the path. The source then fragments using this knowledge.

Authentication

The authentication extension header has a dual purpose: it validates the message sender and ensures the integrity of data.

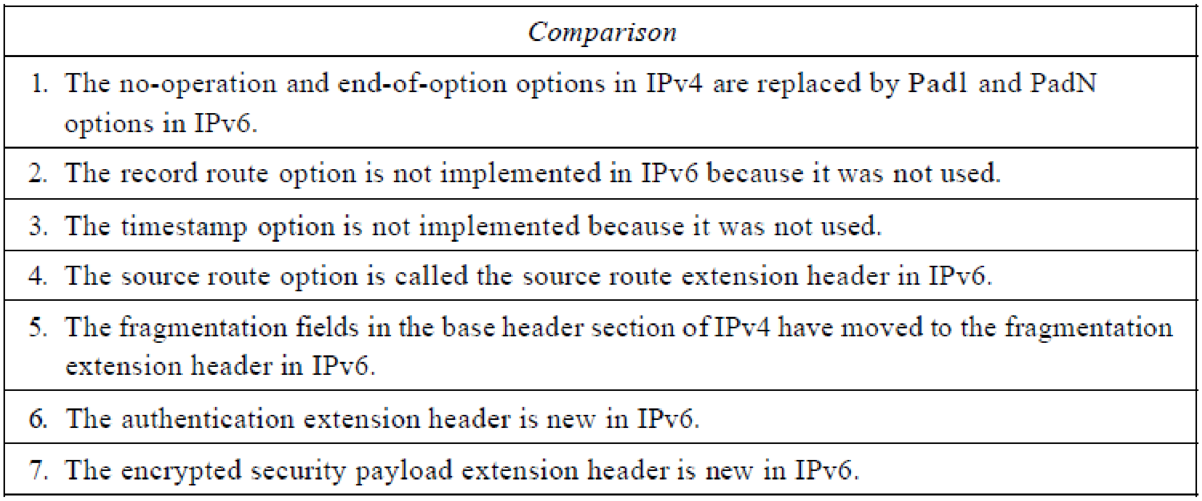
Encrypted Security Payload

The encrypted security payload (ESP) is an extension that provides confidentiality and guards against eavesdropping.

Destination Option

The destination option is used when the source needs to pass information to the destination only. Intermediate routers are not permitted access to this information.

Table 20.10 *Comparison between IPv4 options and IPv6 extension headers*

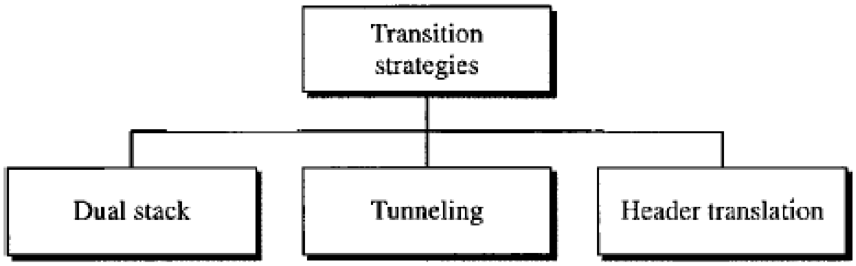


Comparison
1. The no-operation and end-of-option options in IPv4 are replaced by Pad1 and PadN options in IPv6.
2. The record route option is not implemented in IPv6 because it was not used.
3. The timestamp option is not implemented because it was not used.
4. The source route option is called the source route extension header in IPv6.
5. The fragmentation fields in the base header section of IPv4 have moved to the fragmentation extension header in IPv6.
6. The authentication extension header is new in IPv6.
7. The encrypted security payload extension header is new in IPv6.

TRANSITION FROM IPv4 TO IPv6

Because of the huge number of systems on the Internet, the transition from IPv4 to IPv6 cannot happen suddenly. It takes a considerable amount of time before every system in the Internet can move from IPv4 to IPv6. The transition must be smooth to prevent any problems between IPv4 and IPv6 systems. Three strategies have been devised to help the transition

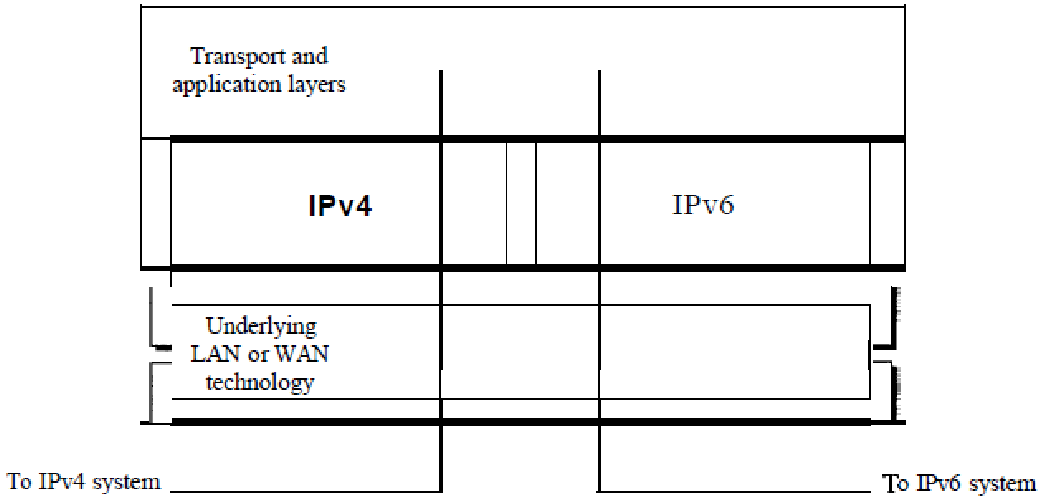
Figure 20.18 *Three transition strategies*



Dual Stack

It is recommended that all hosts, before migrating completely to version 6, have a dual stack of protocols. In other words, a station must run IPv4 and IPv6 simultaneously untilall the Internet uses IPv6.

Figure 20.19 *Dual stack*

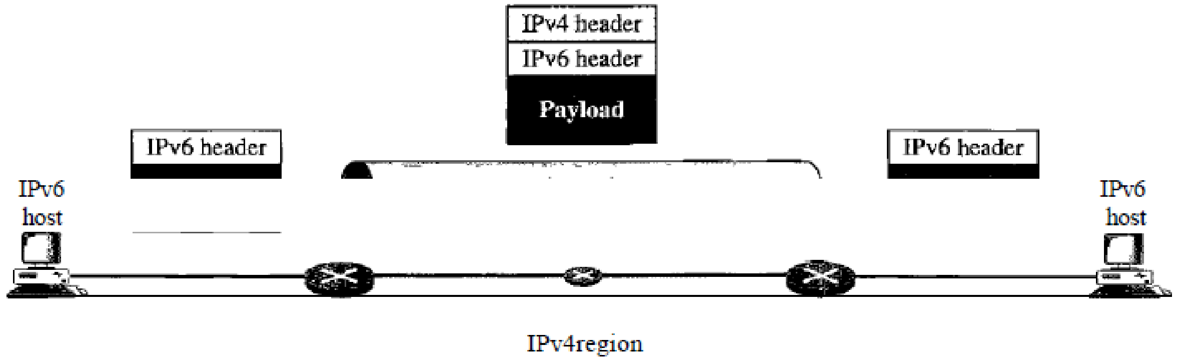

Transport and application layers
IPv4
IPv6
Underlying LAN or WAN technology
To IPv4 system
To IPv6 system

To determine which version to use when sending a packet to a destination, the source host queries the DNS. If the DNS returns an IPv4 address, the source host sends an IPv4 packet. If the DNS returns an IPv6 address, the source host sends an IPv6 packet.

Tunneling

Tunneling is a strategy used when two computers using IPv6 want to communicate with each other and the packet must pass through a region that uses IPv4. To pass through this region, the packet must have an IPv4 address. So the IPv6 packet is encapsulated in an IPv4 packet when it enters the region, and it leaves its capsule when it exits the region. It seems as if the IPv6 packet goes through a tunnel at one end and emerges at the other end. To make it clear that the IPv4 packet is carrying an IPv6 packet as data, the protocol value is set to 41.

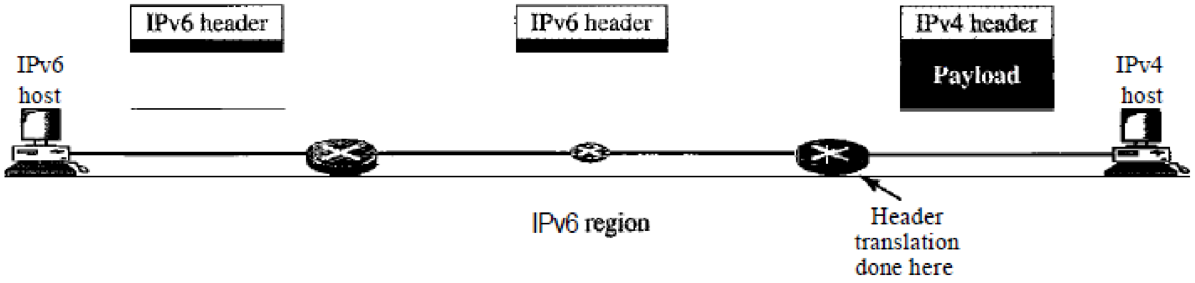
Figure 20.20 *Tunneling strategy*



Header Translation

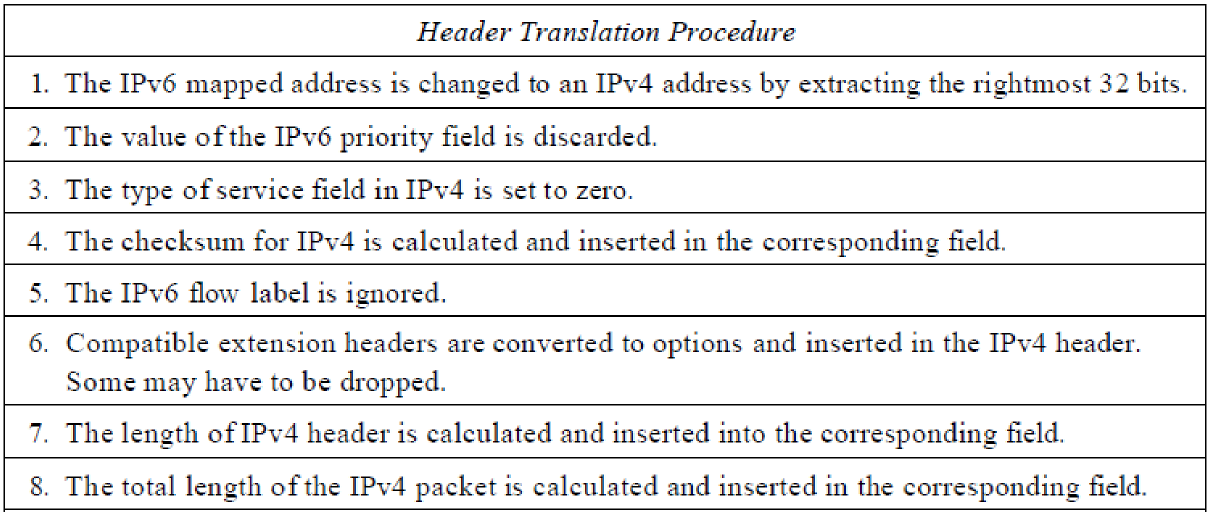
Header translation is necessary when the majority of the Internet has moved to IPv6 but some systems still use IPv4. The sender wants to use IPv6, but the receiver does not understand IPv6. Tunneling does not work in this situation because the packet must be in the IPv4 format to be understood by the receiver. In this case, the header format must be totally changed through header translation. The header of the IPv6 packet is converted to an IPv4 header

Figure 20.21 *Header translation strategy*



Header translation uses the mapped address to translate an IPv6 address to an IPv4 address.

Table 20.11 *Header translation*



Header Translation Procedure
1. The IPv6 mapped address is changed to an IPv4 address by extracting the rightmost 32 bits.
2. The value of the IPv6 priority field is discarded.
3. The type of service field in IPv4 is set to zero.
4. The checksum for IPv4 is calculated and inserted in the corresponding field.
5. The IPv6 flow label is ignored.
6. Compatible extension headers are converted to options and inserted in the IPv4 header. Some may have to be dropped.
7. The length of IPv4 header is calculated and inserted into the corresponding field.
8. The total length of the IPv4 packet is calculated and inserted in the corresponding field.

CHAPTER 8-NETWORK SECURITY

SECURITY

The science and art of transforming messages to make them secure and immune to attack.

PROPERTIES OF SECURE COMMUNICATION

Message Confidentiality

Message confidentiality or privacy means that the sender and the receiver expect confidentiality.

The transmitted message must make sense to only the intended receiver. To all others, the message must be garbage. When a customer communicates with her bank, she expects that the communication is totally confidential.

Message Integrity

Message integrity means that the data must arrive at the receiver exactly as they were sent. There must be no changes during the transmission, neither accidentally nor maliciously. As more and more monetary exchanges occur over the Internet, integrity is crucial. For example, it would be disastrous if a request for transferring $100 changed to a request for $10,000 or $100,000. The integrity of the message must be preserved in a secure communication.

Message Authentication

Message authentication is a service beyond message integrity. In message authentication the receiver needs to be sure of the sender's identity and that an imposter has not sent the message.

Message Nonrepudiation

Message nonrepudiation means that a sender must not be able to deny sending a message that he or she, in fact, did send. The burden of proof falls on the receiver. For example, when a customer sends a message to transfer money from one account to another, the bank must have proof that the customer actually requested this transaction.

Entity Authentication

In entity authentication (or user identification) the entity or user is verified prior to Access to the system resources (files, for example). For example, a student who needs to access her university resources needs to be authenticated during the logging process. This is to protect the interests of the university and the student.

CRYPTOGRAPHY

Cryptography, a word with Greek origins, means "secret writing." However, we use the term to refer to the science and art of transforming messages to make them secure and immune to attacks. Figure 30.1 shows the components involved in cryptography.

Figure 30.1 *Cryptography components*

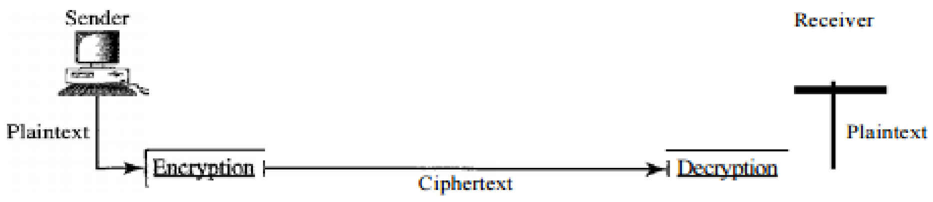
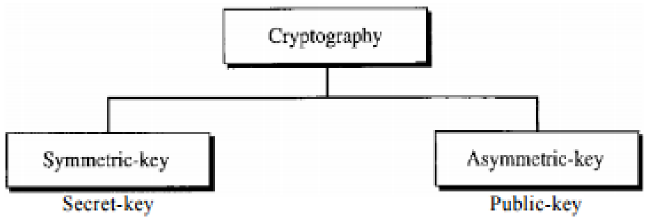


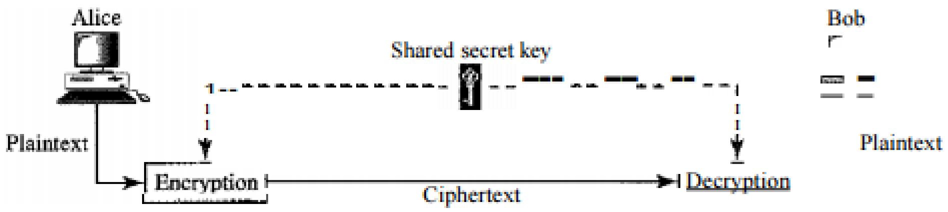
Figure 30.2 *Categories ofcryptography*



SYMMETRIC KEY CRYPTOGRAPHY

- In symmetric-key cryptography, the same key is used by both parties. The sender uses this key and an encryption algorithm to encrypt data; the receiver uses the same key and the corresponding decryption algorithm to decrypt the data

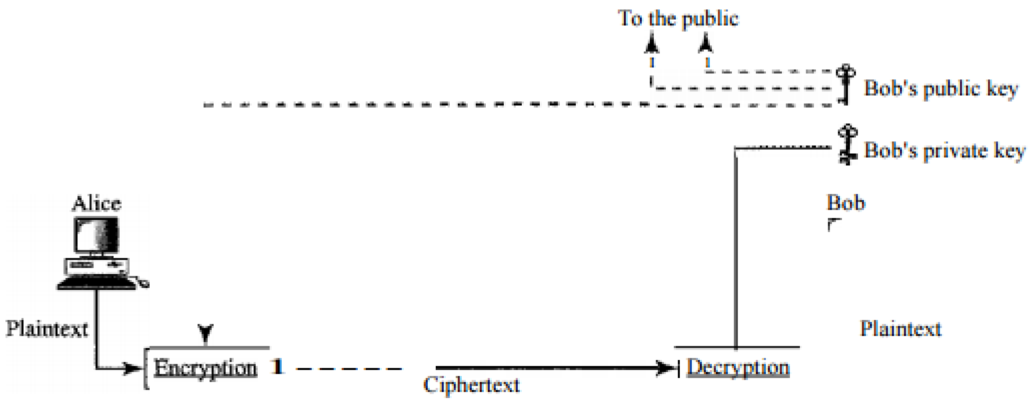
Figure 30.3 *Symmetric-key cryptography*



ASYMMETRIC-KEY CRYPTOGRAPHY

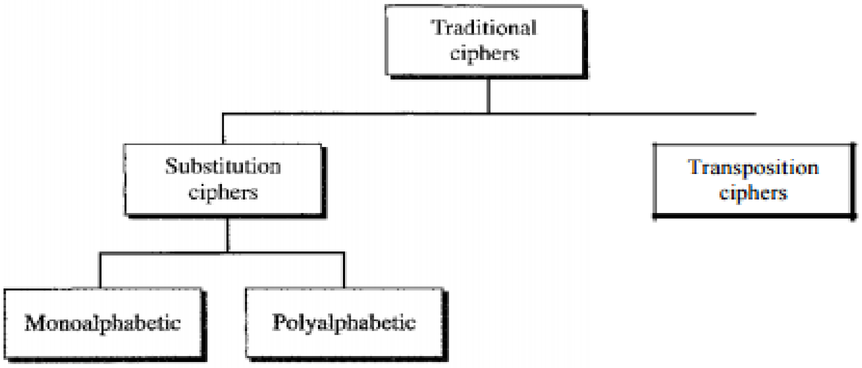
- In asymmetric or public-key cryptography, there are two keys: a private key and a public key. The private key is kept by the receiver. The public key is announced to the public

Figure 30.4 *Asymmetric-key cryptography*



SYMMETRIC-KEY CRYPTOGRAPHY

Figure 30.7 *Traditional ciphers*

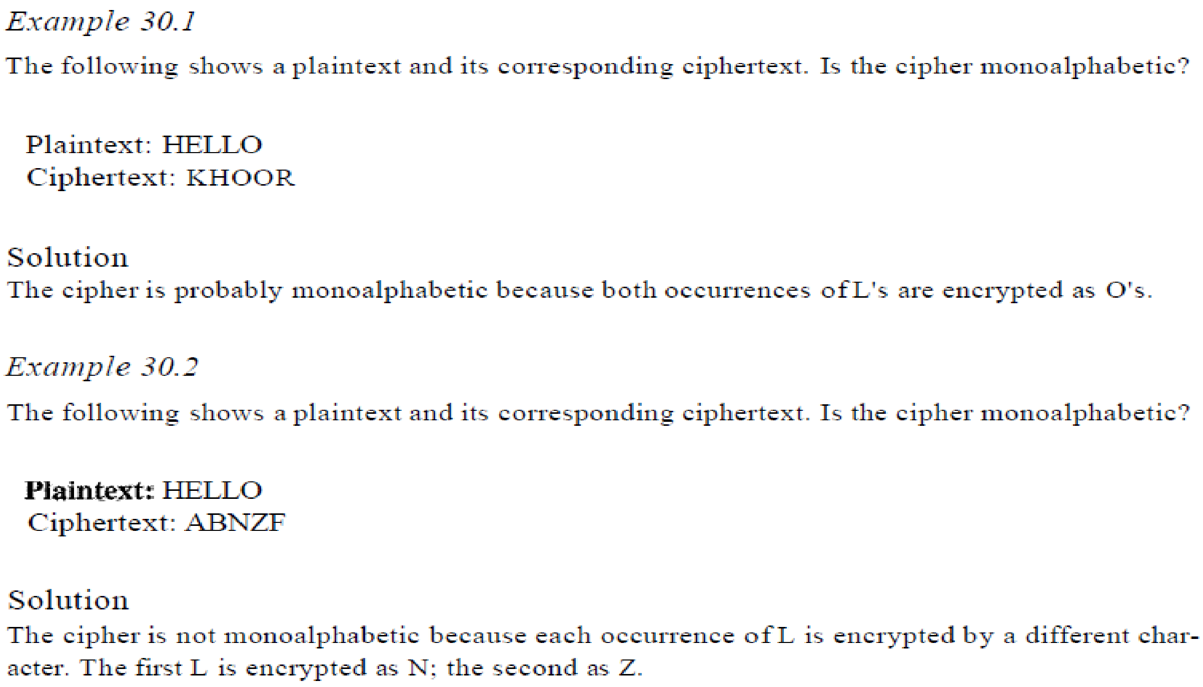


- **Substitution Cipher**
  A substitution cipher substitutes one symbol with another.

- **Monoalphabetic cipher**
  In a monoalphabetic cipher, a character (or a symbol) in the plaintext is always changed to the same character (or symbol) in the cipher text regardless of its position in the text. For example, if the algorithm says that character A in the plaintext is changed to character D, every character A is changed to character D.

- **Polyalphabetic cipher**

- In a polyalphabetic cipher, each occurrence of a character can have a different substitute.



Example 30.1

The following shows a plaintext and its corresponding ciphertext. Is the cipher monoalphabetic?

Plaintext: HELLO
Ciphertext: KHOOR

Solution

The cipher is probably monoalphabetic because both occurrences of L's are encrypted as O's.

Example 30.2

The following shows a plaintext and its corresponding ciphertext. Is the cipher monoalphabetic?

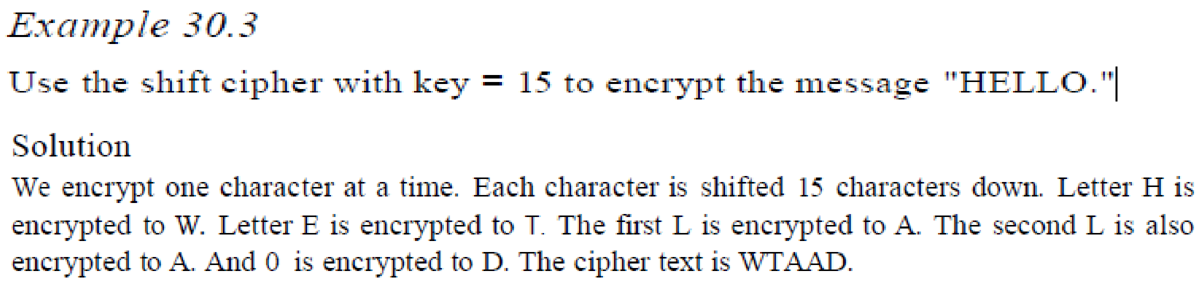
Plaintext: HELLO
Ciphertext: ABNZF

Solution

The cipher is not monoalphabetic because each occurrence of L is encrypted by a different character. The first L is encrypted as N; the second as Z.

- **Shift Cipher**

  The simplest monoalphabetic cipher is probably the shift cipher. We assume that the plaintext and cipher text consist of uppercase letters (A to Z) only. In this cipher, the encryption algorithm is "shift key characters down," with key equal to some number. The decryption algorithm is "shift key characters up."

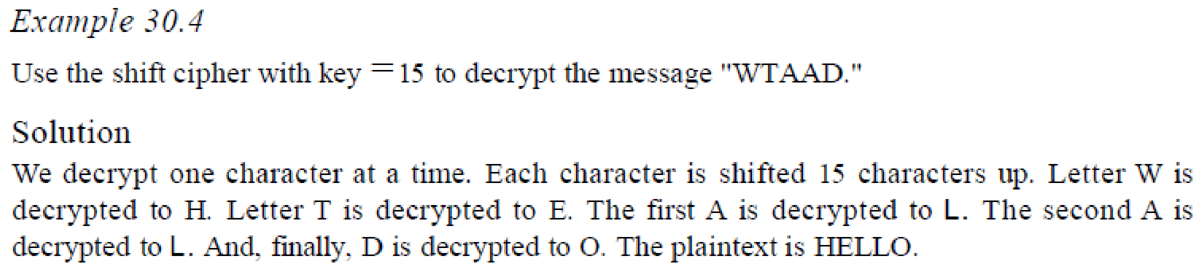


Example 30.3

Use the shift cipher with key = 15 to encrypt the message "HELLO."

Solution

We encrypt one character at a time. Each character is shifted 15 characters down. Letter H is encrypted to W. Letter E is encrypted to T. The first L is encrypted to A. The second L is also encrypted to A. And 0 is encrypted to D. The cipher text is WTAAD.



Example 30.4

Use the shift cipher with key =15 to decrypt the message "WTAAD."

Solution

We decrypt one character at a time. Each character is shifted 15 characters up. Letter W is decrypted to H. Letter T is decrypted to E. The first A is decrypted to L. The second A is decrypted to L. And, finally, D is decrypted to O. The plaintext is HELLO.

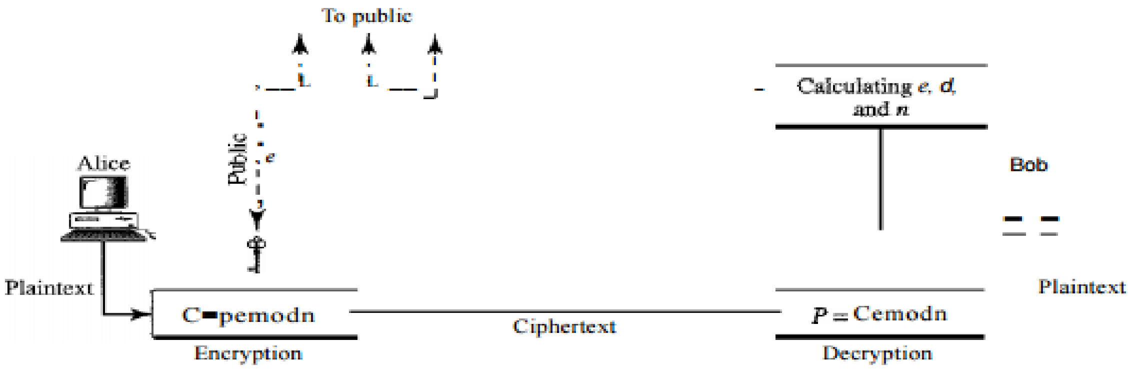
- **Transposition Ciphers**

  In a transposition cipher, there is no substitution of characters; instead, their locations change. A character in the first position of the plaintext may appear in the tenth position of the cipher text.

ASYMMETRIC-KEY CRYPTOGRAPHY

The most common public key algorithm is RSA, named for its inventors Rivest, Shamir, and Adleman (RSA). It uses two numbers, e and d, as the public and private keys

Figure 30.24 *RSA*



Selecting Keys

Bob use the following steps to select the private and public keys:

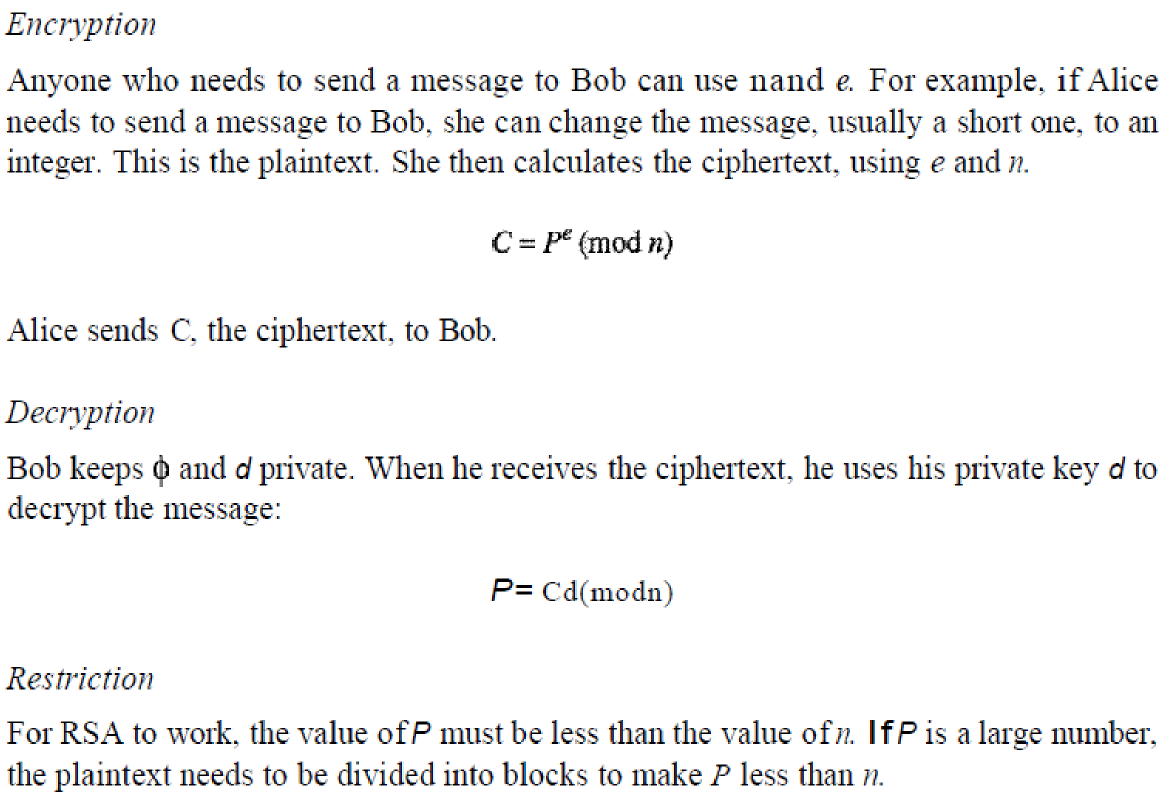
1. Bob chooses two very large prime numbers p and q. Remember that a prime number is one that can be divided evenly only by 1 and itself.

2. Bob multiplies the above two primes to find n, the modulus for encryption and decryption. In other words, n ::: p X q.

3. Bob calculates another number <1> ::: (p -1) X (q - 1).

4. Bob chooses a random integer e. He then calculates d so that d x e::: 1 mod <1>.

5. Bob announces e and n to the public; he keeps <1> and d secret.



Encryption

Anyone who needs to send a message to Bob can use nand *e*. For example, if Alice needs to send a message to Bob, she can change the message, usually a short one, to an integer. This is the plaintext. She then calculates the ciphertext, using *e* and *n*.

$$C = P^e \pmod n$$

Alice sends C, the ciphertext, to Bob.

Decryption

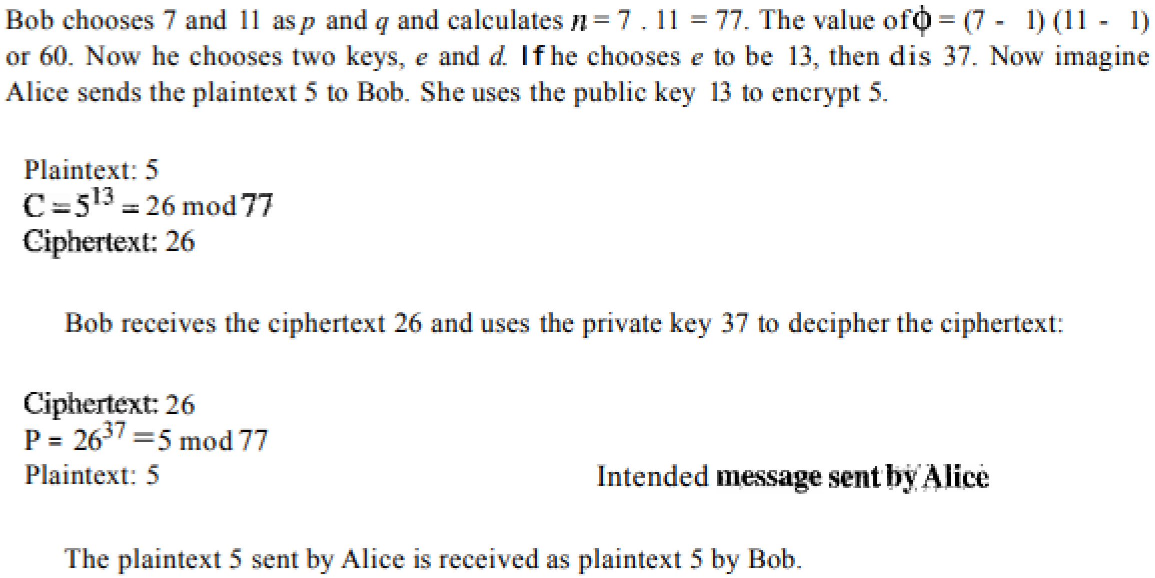
Bob keeps ϕ and *d* private. When he receives the ciphertext, he uses his private key *d* to decrypt the message:

$$P = \mathrm{Cd(modn)}$$

Restriction

For RSA to work, the value of *P* must be less than the value of *n*. If *P* is a large number, the plaintext needs to be divided into blocks to make *P* less than *n*.

Restriction for RSA to work, the value of P must be less than the value of n. If P is a large number, the plaintext needs to be divided into blocks to make P less than n.



Bob chooses 7 and 11 as *p* and *q* and calculates $n = 7 . 11 = 77$. The value of $\phi = (7 - 1)(11 - 1)$ or 60. Now he chooses two keys, *e* and *d*. If he chooses *e* to be 13, then *d* is 37. Now imagine Alice sends the plaintext 5 to Bob. She uses the public key 13 to encrypt 5.

Plaintext: 5
$C = 5^{13} = 26 \bmod 77$
Ciphertext: 26

Bob receives the ciphertext 26 and uses the private key 37 to decipher the ciphertext:

Ciphertext: 26
$P = 26^{37} = 5 \bmod 77$
Plaintext: 5

Intended message sent by Alice

The plaintext 5 sent by Alice is received as plaintext 5 by Bob.

SECURING MAIL: PGP

- Pretty Good Privacy
- Provide security to e-mail
- Developed by Phil Zimmerman in 1995
- Documentation and source code freely available
- Independent of OS and processor
- All user uses public key cryptography
- Uses RSA
- Service provided: Authentication, Confidentiality, Compression, Email compatibility

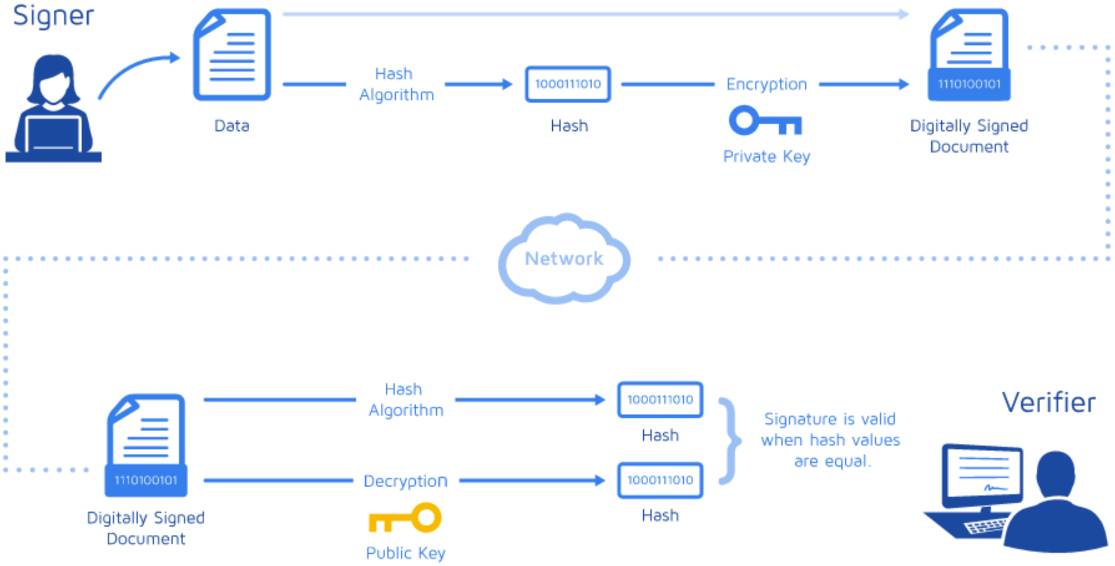
DIGITAL SIGNATURE

- A digital signature is basically a way to ensure that an electronic document (e-mail, spreadsheet, text file, etc.) is authentic.
- Authentic means that you know who created the document and you know that it has not been altered in any way since that person created it.
- Digital signatures rely on certain types of encryption to ensure authentication.
- Encryption is the process of taking all the data that one computer is sending to another and encoding it into a form that only the other computer will be able to decode.
    - Authentication is the process of verifying that information is coming from a trusted source. These two processes work hand in hand for digital signatures.
- Digital signatures are based on public key cryptography, also known as asymmetric cryptography.
- Using a public key algorithm such as RSA, one can generate two keys that are mathematically linked: one private and one public.
- To create a digital signature, signing software (such as an email program) creates a one-way hash of the electronic data to be signed.
- The private key is then used to encrypt the hash. The encrypted hash -- along with other information, such as the hashing algorithm -- is the digital signature.

- Digital signatures, like handwritten signatures, are unique to each signer.
- Digital signature solution providers, such as DocuSign, follow a specific protocol, called PKI.
- PKI requires the provider to use a mathematical algorithm to generate two long numbers, called keys. One key is public, and one key is private.
- When a signer electronically signs a document, the signature is created using the signer's private key, which is always securely kept by the signer.
- The mathematical algorithm acts like a cipher, creating data matching the signed document, called a hash, and encrypting that data.
- The resulting encrypted data is the digital signature. The signature is also marked with the

time that the document was signed. If the document changes after signing, the digital signature is invalidated.

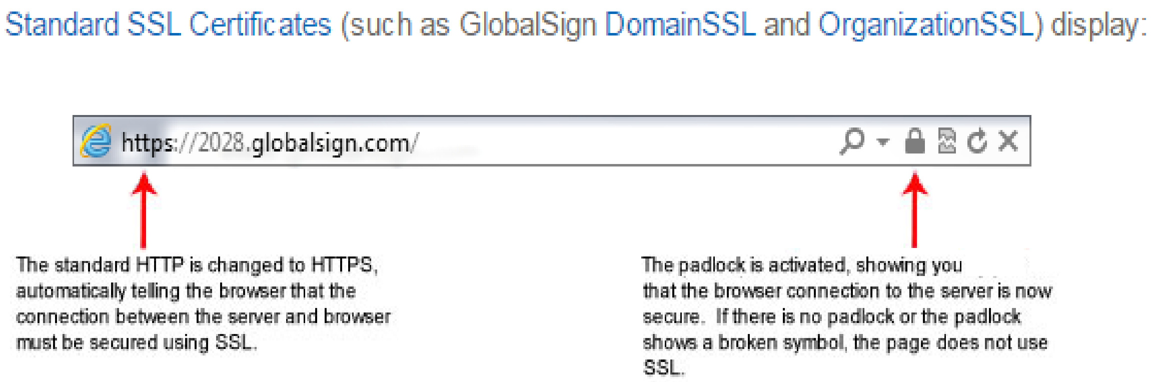
- As an example, Jane signs an agreement to sell a timeshare using her private key. The buyer receives the document. The buyer who receives the document also receives a copy of Jane's public key. If the public key can't decrypt the signature (via the cipher from which the keys were created), it means the signature isn't Jane's, or has been changed since it was signed. The signature is then considered invalid.



SSL (Secure Socket Layer)

- Most widely deployed security protocol used today.
- Essentially a protocol that provides a secure channel between two machines operating over the Internet or an internal network.
- Typically used when a web browser needs to securely connect to a web server over the inherently insecure Internet.

Users are alerted to the presence of SSL when the browser displays a padlock


Standard SSL Certificates (such as GlobalSign DomainSSL and OrganizationSSL) display:

https://2028.globalsign.com/

The standard HTTP is changed to HTTPS, automatically telling the browser that the connection between the server and browser must be secured using SSL.

The padlock is activated, showing you that the browser connection to the server is now secure. If there is no padlock or the padlock shows a broken symbol, the page does not use SSL.

- The authentication process uses public key encryption to validate the digital certificate and to confirm that a server is, in fact, the server it claims to be.
- Once the server has been authenticated, the client and server establish cipher settings and a shared key to encrypt the information they exchange during the remainder of the session.
- This provides data confidentiality and integrity.
- This whole process is invisible to the user. For example, if a webpage requires an SSL connection, the URL will change from HTTP to HTTPS, and a padlock icon will appear in the browser once the server has been authenticated.

- Advantages

- To secure online credit card transactions.
- To secure system logins and any sensitive information exchanged online.
- To secure webmail and applications like Outlook Web Access, Exchange and Office Communications Server.
- To secure workflow and virtualization applications like Citrix Delivery Platforms or cloud-based computing platforms.
- To secure the connection between an email client such as Microsoft Outlook and an email server such as Microsoft Exchange.
- To secure the transfer of files over https and FTP(s) services such as website owners updating new pages to their websites or transferring large files.
- To secure hosting control panel logins and activity like Parallels, cPanel, and others.
- To secure intranet based traffic such as internal networks, file sharing, extranets, and database connections.
- To secure network logins and other network traffic with SSL VPNs such as VPN Access Servers or applications like the Citrix Access Gateway.

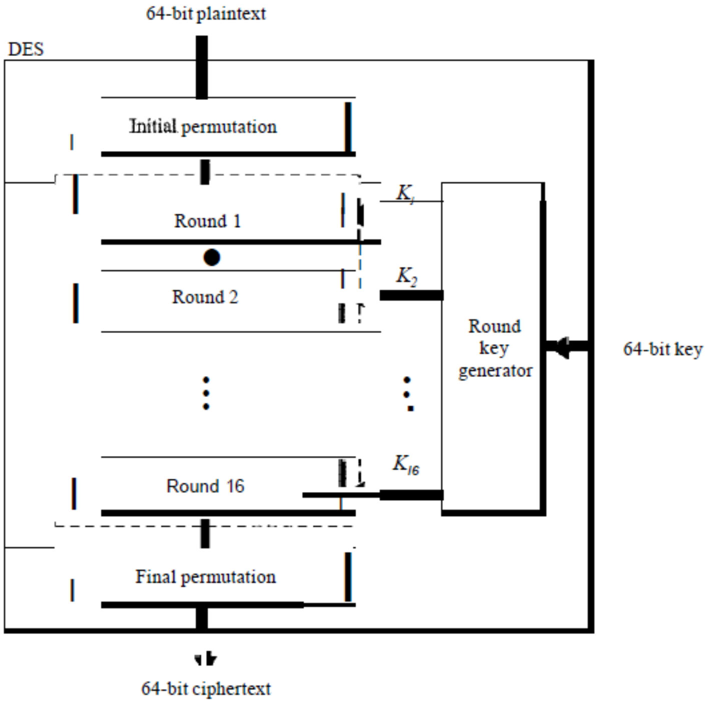
IPSECURITY (IPSEC)

- Internet Protocol security (IPSec) is a framework of open standards for helping to ensure private, secure communications over Internet Protocol (IP) networks through the use of cryptographic security services.
- IPSec supports network-level data integrity, data confidentiality, data origin authentication, and replay protection.
- Because IPSec is integrated at the Internet layer (layer 3), it provides security for almost all protocols in the TCP/IP suite, and because IPSec is applied transparently to applications, there is no need to configure separate security for each application that uses TCP/IP.
- IPSec helps provide defense-in-depth against:
- Network-based attacks from untrusted computers, attacks that can result in the denial-of-service of applications, services, or the network

- Data corruption
- Data theft
- User-credential theft
- Administrative control of servers, other computers, and the network.
- IPSec is a general-purpose security technology that can be used to help secure network traffic in many scenarios.
- Packet filtering
- End-to-end security between specific hosts
- End-to-end traffic through a Microsoft Internet Security and Acceleration (ISA) Server-secured network address translator
- Secure server
- Layer Two Tunneling Protocol (L2TP) over IPSec (L2TP/IPSec) for remote access and site-to-site virtual private network (VPN) connections
- Site-to-site IPSec tunneling with non-Microsoft IPSec gateways
- IPsec provides two choices of security service: Authentication Header (AH), which essentially allows authentication of the sender of data, and Encapsulating Security Payload (ESP), which supports both authentication of the sender and encryption of data as well.

Data Encryption Standard (DES)

One example of a complex block cipher is the Data Encryption Standard (DES). DES was designed by IBM and adopted by the U.S. government as the standard encryption method for nonmilitary and nonclassified use. The algorithm encrypts a 64-bit plaintext block using a 64-bit key,



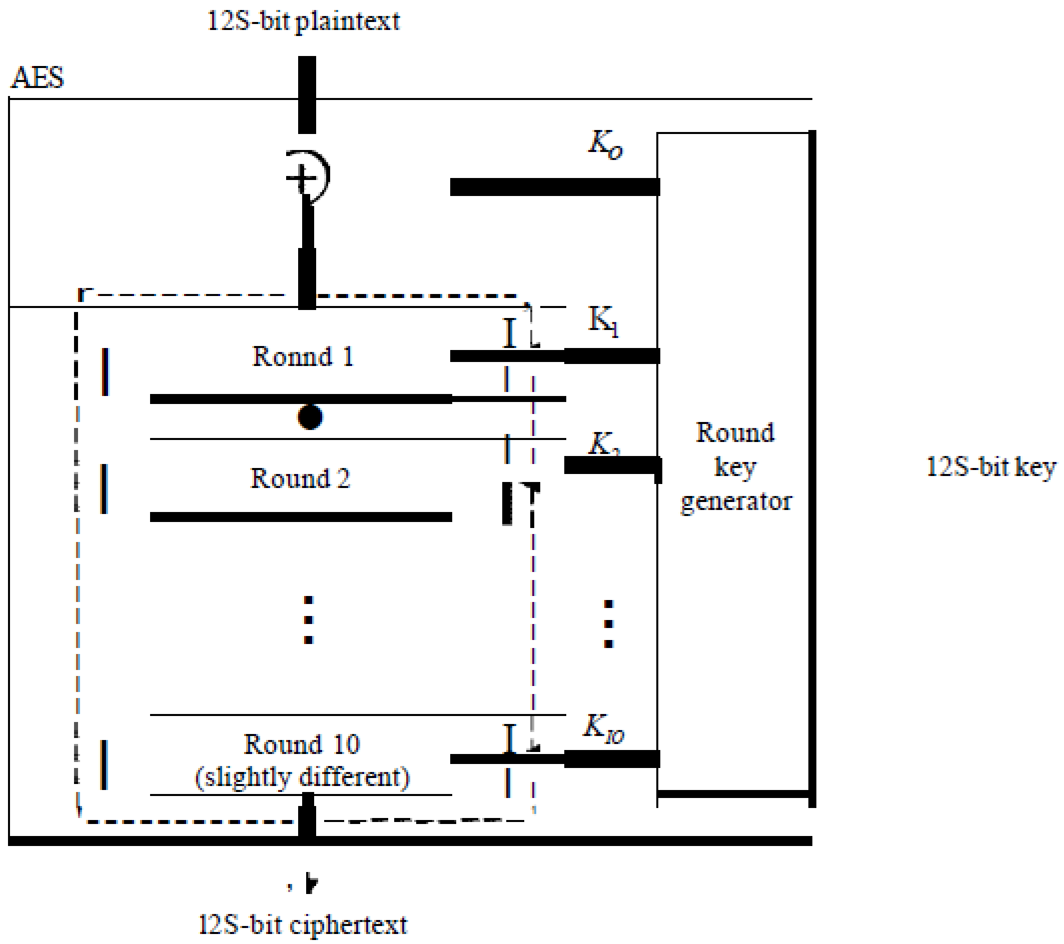
DES has two transposition blocks (P-boxes) and 16 complex round ciphers (they are repeated). Although the 16 iteration round ciphers are conceptually the same, each uses a different key derived from the original key. The initial and final permutations are keyless straight permutations that are the inverse of each other. The permutation takes a 64-bit input and permutes them according

to predefined values.

Advanced Encryption Standard (AES)

The Advanced Encryption Standard (AES) was designed because DES's key was too small. Although Triple DES ODES) increased the key size, the process was too slow. AES is a very complex round cipher. AES is designed with three key sizes: 128, 192, or 256 bits. Table 30.1 shows the relationship between the data block, number of rounds, and key size.

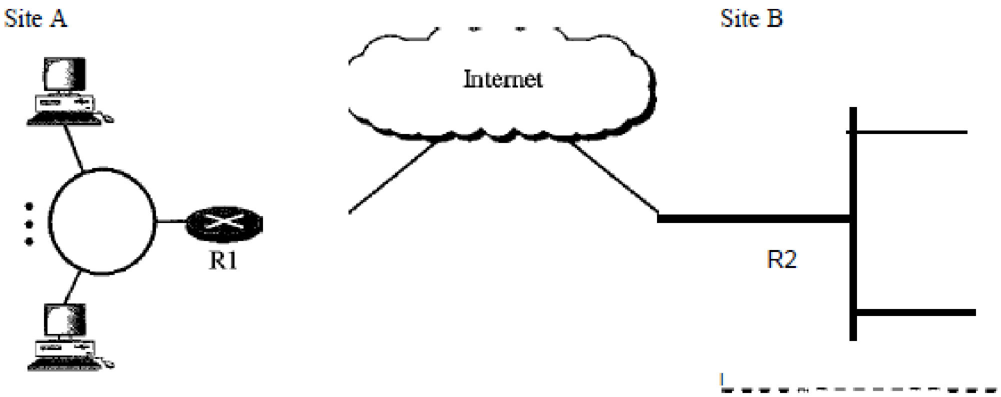
There is an initial XOR operation followed by 10 round ciphers. The last round is slightly different from the preceding rounds; it is missing one operation. Although the 10 iteration blocks are almost identical, each uses a different key derived from the original key.



Virtual Private Network

Virtual private network (VPN) is a technology that is gaining popularity among large organizations that use the global Internet for both intra- and inter organization communication, but require privacy in their internal communications. We discuss VPN here because it uses the IPSec Protocol to apply security to the IP datagram. A technology called virtual private network allows organizations to use the global Internet for both purposes. VPN creates a network that is private but virtual. It is private because it guarantees privacy inside the organization. It is virtual because it does not use real private WANs; the network is physically public but virtually private. Routers Rl and R2 use VPN technology to guarantee privacy for the organization.

Figure 32.12 *Virtual private network*



VPN Technology

VPN technology uses IPSec in the tunnel mode to provide authentication, integrity, and

privacy.

Tunneling To guarantee privacy and other security measures for an organization,

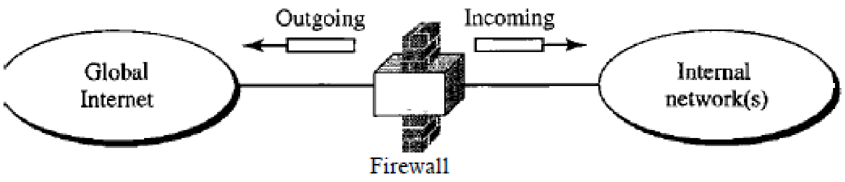
VPN can use the IPSec in the tunnel mode. In this mode, each IP datagram destined for

private use in the organization is encapsulated in another datagram.

FIREWALLS

All previous security measures cannot prevent Eve from sending a harmful message to a system. To control access to a system, we need firewalls. A firewall is a device (usually a router or a computer) installed between the internal network of an organization and the rest of the Internet. It is designed to forward some packets and filter (not forward) others.

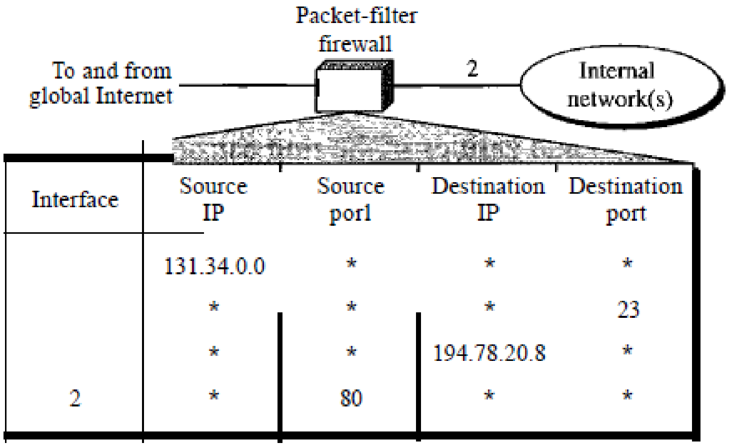
Figure 32.22 *Firewall*



For example, a firewall may filter all incoming packets destined for a specific host or a specific server such as HTTP. A firewall can be used to deny access to a specific host or a specific service in the organization. A firewall is usually classified as a packet-filter firewall or a proxy-based firewall.

Packet-Filter Firewall

A firewall can be used as a packet filter. It can forward or block packets based on the information in the network layer and transport layer headers: source and destination IP addresses, source and destination port addresses, and type of protocol (TCP or UDP). A packet-filter firewall is a router that uses a filtering table to decide which packets must be discarded (not forwarded). Figure shows an example of a filtering table for this kind of a firewall.



Interface	Source IP	Source porl	Destination IP	Destination port
	131.34.0.0	*	*	*
	*	*	*	23
	*	*	194.78.20.8	*
2	*	80	*	*

According to Figure , the following packets are filtered:

1. Incoming packets from network 131.34.0.0 are blocked (security precaution). Note

that the * (asterisk) means "any."

2. Incoming packets destined for any internal TELNET server (port 23) are blocked.

3. Incoming packets destined for internal host 194.78.20.8 are blocked. The organization

wants this host for internal use only.

4. Outgoing packets destined for an Http server (port 80) are blocked. The organization

does not want employees to browse the Internet.

Application Gateway

An application gateway or application level gateway (ALG) is a firewall proxy which provides network security. It filters incoming node traffic to certain specifications which mean that only transmitted network application data is filtered. Such network applications include File Transfer Protocol (FTP), Telnet, Real Time Streaming Protocol (RTSP) and Bit Torrent.

Application gateways provide high-level secure network system communication. For example, when a client requests access to server resources such as files, Web pages and databases, the client first connects with the proxy server, which then establishes a connection with the main server.

The application gateway resides on the client and server firewall. The proxy server hides Internet Protocol (IP) addresses and other secure information on the client's behalf. A computer's internal system may communicate with an external computer using firewall protection. The application gateway and external computer function without client information or knowledge of the proxy server IP address.

Intrusion Detection System

An intrusion detection system (IDS) is a device or software application that monitors a network or systems for malicious activity or policy violations. Any detected activity or violation is typically reported either to an administrator or collected centrally using a security information and event management (SIEM) system. A SIEM system combines outputs from multiple sources, and uses alarm filtering techniques to distinguish malicious activity from false alarms.

There is a wide spectrum of IDS, varying from antivirus software to hierarchical systems that monitor the traffic of an entire backbone network. The most common classifications are network intrusion detection systems (NIDS) and host-based intrusion detection systems (HIDS). A system that monitors important operating system files is an example of a HIDS, while a system that analyzes incoming network traffic is an example of a NIDS. It is also possible to classify IDS by detection approach: the most well-known variants are signature-based detection (recognizing bad patterns, such as malware) and anomaly-based detection (detecting deviations from a model of "good" traffic, which often relies on machine learning). Some IDS have the ability to respond to detected intrusions. Systems with response capabilities are typically referred to as an intrusion prevention system.

Network intrusion detection systems

Network intrusion detection systems (NIDS) are placed at a strategic point or points within the network to monitor traffic to and from all devices on the network. It performs an analysis of passing traffic on the entire subnet, and matches the traffic that is passed on the subnets to the library of known attacks. Once an attack is identified, or abnormal behavior is sensed, the alert can be sent to the administrator. An example of an NIDS would be installing it on the subnet where firewalls are located in order to see if someone is trying to break into the firewall. Ideally one would scan all inbound and outbound traffic, however doing so might create a bottleneck that would impair the overall speed of the network. OPNET and NetSim are commonly used tools for simulation network intrusion detection systems. NID Systems are also capable of comparing signatures for similar packets to link and drop harmful detected packets which have a signature matching the records in the NIDS. When we classify the design of the NIDS according to the system interactivity property, there are two types: on-line and off-line NIDS. On-line NIDS deals with the network in real time. It analyses the Ethernet packets and applies some rules, to decide if it is an attack or not. Off-line NIDS deals with stored data and passes it through some processes to decide if it is an attack or not.

Host intrusion detection systems

Host intrusion detection systems (HIDS) run on individual hosts or devices on the network. A HIDS monitors the inbound and outbound packets from the device only and will alert the user or administrator if suspicious activity is detected. It takes a snapshot of existing system files and matches it to the previous snapshot. If the critical system files were modified or deleted, an alert is sent to the administrator to investigate. An example of HIDS usage can be seen on mission critical machines, which are not expected to change their configurations.

WEP

Wired Equivalent Privacy (WEP) is a security protocol, specified in the IEEE Wireless Fidelity (Wi-Fi) standard, 802.11b, that is designed to provide a wireless local area network (WLAN) with a level of security and privacy comparable to what is usually expected of a wired LAN. A wired local area network (LAN) is generally protected by physical security mechanisms (controlled access to a building, for example) that are effective for a controlled physical environment, but may be ineffective for WLANs because radio waves are not necessarily bound by the walls containing the network. WEP seeks to establish similar protection to that offered by the wired network's physical security measures by encrypting data transmitted over the WLAN. Data encryption protects the vulnerable wireless link between clients and access points; once this measure has been taken, other typical LAN security mechanisms such as password protection, end-to-end encryption, virtual private networks (VPNs), and authentication can be put in place to ensure privacy.

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