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UNIT 1

INTRODUCTION TO COMPUTER NETWORKS

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STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- know about networking in general.
- use networking, know its purpose, and learn how the networking helps us in working faster and easier.
- explain the networking's various goals and applications.
- describe OSI Reference Mode as a technology for connecting various computers using the networking.
- know about Novell Netware as a networking software from Novell Netware.
- explain about ARPANET as another networking software.
- know about NSFNET software as another networking software.
- describe about the Internet, which is a popular application of Networking.

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1.1 INTRODUCTION

When more than two computers are connected to each other and sharing information, resources and remote systems then this is called Networking.

Technical definitions:

"A network of data processing nodes that are interconnected for the purpose of data communication".

"An interconnection of three or more communicating entities".

Classification of Computer Networks

Network Layer

In this computer networks follow the industry standards of OSI reference model and TCP/IP model. Where as OSI is of seven layers and TCP/IP is defined in five layers.

Scale

It can be classified as:

- Local area network (LAN)
- Campus area network (CAN)
- Metropolitan area network (MAN)
- Wide area network (WAN).
- Personal area network (PAN).

Connection Method

The connection methods available are:

- Ethernet
- Power line communication
- Wireless LAN
- HomePNA

Functional Relationship

This exists between the network elements:

- Peer-to-peer
- Client-server

Network Topology

It is a logical layouts of the network. Topologies are:

- Star network
- Ring network
- Bus network

- Tree network
- Star-bus network
- Mesh network

Services

It provides following services:

- Wireless community network
- Server
- Storage area networks
- Process control networks
- Value-added network

Protocol

On network protocols are used as communication language. Several types of protocols are available:

- TCP/IP
- Network IPX/SPX

Types of Networks

Following is the list of the most common types of computer networks in order of scale.

Personal Area Network (PAN)

PAN is used for communication among the personal devices (intrapersonal communication), or for connecting to a higher level network and Internet. The reach of a PAN is typically a few meters.

A personal area network (PAN) is used for communication among computer devices. For example,

- Telephones
- Personal digital assistants

Personal area networks may be wired with computer buses such as USB and FireWire. A wireless personal area network (WPAN) can also be made possible with network technologies such as IrDA and Bluetooth.

Local Area Network (LAN)

A network covering a small geographic area, like a home, office, or building. Current LANs are most likely to be based on Ethernet technology. The defining characteristics of LANs, in contrast to WANs (wide area networks), include their much higher data transfer rates, smaller geographic range, and lack of a need for leased telecommunication lines.

Campus Area Network (CAN)

A network that connects two or more LANs but that is limited to a specific (possibly private) geographical area such as a college campus, industrial complex, or a military

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base. A CAN, may be considered a type of MAN (metropolitan area network), but is generally limited to an area that is smaller than a typical MAN.

Metropolitan Area Network (MAN)

A network that connects two or more Local Area Networks or CANs together but does not extend beyond the boundaries of the immediate town, city, or metropolitan area. Multiple routers, switches & hubs are connected to create a MAN

Wide Area Network (WAN)

A WAN is a data communications network that covers a relatively broad geographic area and that often uses transmission facilities provided by common carriers, such as telephone companies. WAN technologies generally function at the lower three layers of the OSI reference model: the physical layer, the data link layer, and the network layer.

Internetwork

Two or more networks or network segments connected using devices that operate at layer 3 (the 'network' layer) of the OSI Basic Reference Model, such as a router. Any interconnection among or between public, private, commercial, industrial, or governmental networks may also be defined as an internetwork.

Internet

A specific internetwork, consisting of a worldwide interconnection of governmental, academic, public, and private networks based upon the Advanced Research Projects Agency Network (ARPANET) developed by ARPA of the U.S. Department of Defense – also home to the World Wide Web (WWW) and referred to as the 'Internet' with a capital 'I' to distinguish it from other generic internetworks.

Extranet

A network or internetwork that is limited in scope to a single organization or entity but which also has limited connections to the networks of one or more other usually, but not necessarily, trusted organizations or entities (e.g., a company's customers may be provided access to some part of its intranet thusly creating an extranet while at the same time the customers may not be considered 'trusted' from a security standpoint). Technically, an extranet may also be categorized as a CAN, MAN, WAN, or other type of network, although, by definition, an extranet cannot consist of a single LAN, because an extranet must have at least one connection with an outside network.

Intranets and extranets may or may not have connections to the Internet.

If connected to the Internet, the intranet or extranet is normally protected from being accessed from the Internet without proper authorization. The Internet itself is not considered to be a part of the intranet or extranet, although the Internet may serve as a portal for access to portions of an extranet.

Basic Hardware Components

All networks are made up of basic hardware building blocks to interconnect network nodes, such as Network Interface Cards (NICs), Bridges, Hubs, Switches, and Routers. In addition, some method of connecting these building blocks is required, usually in

the form of galvanic cable (most commonly Category 5 cable). Less common are microwave links (as in IEEE 802.11) or optical cable ("optical fiber").

Network Interface Cards

A **network card, network adapter** or **NIC** *(network interface card) is a piece of* computer hardware designed to allow computers to communicate over a **computer network**. It provides physical access to a networking medium and provides a low-level addressing system through the use of MAC addresses. It allows users to connect to each other either by using cables or wirelessly.

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Bridges

A **network bridge** connects multiple network segments at the data link layer (layer 2) of the OSI model. Bridges are similar to repeaters or network hubs, devices that connect network segments at the physical layer, however a bridge works by using bridging where traffic from one network is managed rather than simply rebroadcast to adjacent network segments.

Hubs

A hub is a piece of hardware which provides the connectivity of a segment of a network by directing traffic through the network. It does this in a rudimentary way, it simply copies the data to all of the Nodes connected to the hub. Hubs are commonly used to connect segments of a LAN. A hub contains multiple ports. When a packet arrives at one port, it is copied to the other ports so that all segments of the LAN can see all packets.

Switches

Switches are the device of networking that directs traffic to the correct node by filtering and forwarding packets between Nodes. Switches operate at the data link layer (layer 2) and sometimes the network layer (layer 3) of the OSI Reference Model and therefore support any packet protocol. LANs that use switches to join segments are called switched LANs or, in the case of Ethernet networks, switched Ethernet LANs. In a circuit-switched data network, a switch is used to create a virtual circut between the pairs of endpoints. This means that it creates a path to the destination node from the source node.

Routers

Routers are the networking device that forwards data packets along networks by using headers and forwarding tables to determine the best path to forward the packets. Routers also provide interconnectivity between like and unlike devices on the network. This is accomplished by examining the Header of a data packet. They use protocols such as ICMP to communicate with each other and configure the best route between any two hosts. A router is connected to at least two networks, commonly two LANs or WANs or a LAN and its ISP's network.

Routers are usually located at gateways, the places where two or more networks connect. Many household DSL and Cable Modems are also routers.

Building a Simple Computer Network

A simple computer network may be constructed from two computers by adding a network adapter (Network Interface Controller (NIC)) to each computer and then

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connecting them together with a special cable called a crossover cable. This type of network is useful for transferring information between two computers that are not normally connected to each other by a permanent network connection or for basic home networking applications. Alternatively, a network between two computers can be established without dedicated extra hardware by using a standard connection such as the RS-232 serial port on both computers, connecting them to each other via a special crosslinked *null modem* cable.

Practical networks generally consist of more than two interconnected computers and generally require special devices in addition to the Network Interface Controller that each computer needs to be equipped with. Examples of some of these special devices are hubs, switches and routers.

1.2 USES OF NETWORKS: GOALS AND APPLICATIONS

The most important usuage of networking is Resource Sharing, and the goal is to make all programs, equipment, and especially data available to anyone on the network without regard to the physical location of the resource and the user. In other words, the mere fact that a user happends to be 1000 km away from his data should not prevent him from using the data as though they were local: A second goal is to provide high reliability by having alternative sources of supply. For example, all files could be replicated on two or more machines, so if one of them is unavailable (due to hardware failure), the other copies could be used.

Small computers have a much better price/performance ratio rather than large ones. Mainframes are roughly a factor of ten faster than personal computers, buy they cost a thousand times more. The imbalance has caused many systems designers to build system consisting of personal computers, one per user, with data kept on one or more shared file server machines.

A computer network can provide a powerful communication medium among widely separated employees. Using a network, it is easy for two or more people who live far apart to write a report together. When one worker makes a change to an on-line document, the others can see the change immediately, instead of waiting several days for the letter.

1.3 OSI REFERENCE MODE

In 1977, the International Organization for Standardization (ISO), began to develop its OSI networking suite. OSI has two major components: an abstract model of networking (the Basic Reference Model, or seven-layer model), and a set of concrete protocols. The standard documents that describe OSI are for sale and not currently available online.

Parts of OSI have influenced Internet protocol development, but none more than the abstract model itself, documented in ISO 7498 and its various addenda. In this model, a networking system is divided into layers. Within each layer, one or more entities implement its functionality. Each entity interacts directly only with the layer immediately beneath it, and provides facilities for use by the layer above it.

In particular, Internet protocols are deliberately not as rigorously architected as the

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OSI model, but a common version of the TCP/IP model splits it into four layers. The Internet Application Layer includes the OSI Application Layer, Presentation Layer, and most of the Session Layer. Its End-to-End Layer includes the graceful close function of the OSI Session Layer as well as the Transport Layer. Its Internetwork Layer is equivalent to the OSI Network Layer, while its Interface layer includes the OSI Data Link and Physical Layers. These comparisons are based on the original seven-layer protocol model as defined in ISO 7498, rather than refinements in such things as the Internal Organization of the Network Layer document.

Protocols enable an entity in one host to interact with a corresponding entity at the same layer in a remote host. Service definitions abstractly describe the functionality provided to a (N)-layer by an (N-1) layer, where N is one of the seven layers inside the local host.

Layer 7: Application Layer

The application layer is the seventh level of the seven-layer OSI model. It interfaces directly to and performs common application services for the application processes; it also issues requests to the presentation layer. Note carefully that this layer provides services to user-defined application processes, and not to the end user. For example, it defines a file transfer protocol, but the end user must go through an application process to invoke file transfer. The OSI model does not include human interfaces.

The common application services sublayer provides functional elements including the Remote Operations Service Element (comparable to Internet Remote Procedure Call), Association Control, and Transaction Processing (according to the ACID requirements).

Above the common application service sublayer are functions meaningful to user application programs, such as messaging (X.400), directory (X.500), file transfer (FTAM), virtual terminal (VTAM), and batch job manipulation (JTAM).

Layer 6: Presentation Layer

The Presentation layer transforms the data to provide a standard interface for the Application layer. MIME encoding, data encryption and similar manipulation of the presentation are done at this layer to present the data as a service or protocol developer sees fit. Examples of this layer are converting an EBCDIC-coded text file to an ASCII-coded file, or serializing objects and other data structures into and out of XML.

Layer 5: Session Layer

The Session layer controls the dialogues/connections (sessions) between computers. It establishes, manages and terminates the connections between the local and remote application. It provides for either full-duplex or half-duplex operation, and establishes checkpointing, adjournment, termination, and restart procedures. The OSI model made this layer responsible for "graceful close" of sessions, which is a property of TCP, and also for session checkpointing and recovery, which is not usually used in the Internet protocols suite. Session layers are commonly used in application environments that make use of remote procedure calls (RPCs).

iSCSI, which implements the Small Computer Systems Interface (SCSI)) encapsulated into TCP/IP packets, is a session layer protocol increasingly used in Storage Area

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Networks and internally between processors and high-performance storage devices. iSCSI leverages TCP for guaranteed delivery, and carries SCSI command descriptor blocks (CDB) as payload to create a virtual SCSI bus between iSCSI initiators and iSCSI targets.

Layer 4: Transport Layer

The Transport layer provides transparent transfer of data between end users, providing reliable data transfer services to the upper layers. The transport layer controls the reliability of a given link through flow control, segmentation/desegmentation, and error control. Some protocols are state and connection oriented. This means that the transport layer can keep track of the segments and retransmit those that fail.

The best known example of a layer 4 protocol is the Transmission Control Protocol (TCP).

The transport layer is the layer that converts messages into TCP segments or User Datagram Protocol (UDP), Stream Control Transmission Protocol (SCTP), etc. packets.

Perhaps an easy way to visualize the Transport Layer is to compare it with a Post Office, which deals with the dispatch and classification of mail and parcels sent. Do remember, however, that a post office manages the outer envelope of mail. Higher layers may have the equivalent of double envelopes, such as cryptographic Presentation services that can be read by the addressee only.

Roughly speaking, tunneling protocols operate at the transport layer, such as carrying non-IP protocols such as IBM's SNA or Novell's IPX over an IP network, or end-to-end encryption with IPsec. While Generic Routing Encapsulation (GRE) might seem to be a network layer protocol, if the encapsulation of the payload takes place only at endpoint, GRE becomes closer to a transport protocol that uses IP headers but contains complete frames or packets to deliver to an endpoint. L2TP carries PPP frames inside transport packets.

Layer 3: Network Layer

The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer. The Network layer performs network routing functions, and might also perform fragmentation and reassembly, and report delivery errors. Routers operate at this layer—sending data throughout the extended network and making the Internet possible. This is a logical addressing scheme – values are chosen by the network engineer.

The addressing scheme is hierarchical. The best known example of a layer 3 protocol is the Internet Protocol (IP).

Perhaps it's easier to visualize this layer as managing the sequence of human carriers taking a letter from the sender to the local post office, trucks that carry sacks of mail to other post offices or airports, airplanes that carry airmail between major cities, trucks that distribute mail sacks in a city, and carriers that take a letter to its destinations. Think of fragmentation as splitting a large document into smaller envelopes for shipping, or, in the case of the network layer, splitting an application or transport record into packets.

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Layer 2: Data Link Layer

The Data Link layer provides the functional and procedural means to transfer data between network entities and to detect and possibly correct errors that may occur in the Physical layer. The best known example of this is Ethernet. This layer manages the interaction of devices with a shared medium. Other examples of data link protocols are *HDLC* and *ADCCP* for point-to-point or packet-switched networks and *Aloha* for local area networks. On IEEE 802 local area networks, and some non-IEEE 802 networks such as FDDI, this layer may be split into a Media Access Control (MAC) layer and the IEEE 802.2 Logical Link Control (LLC) layer. It arranges bits from the physical layer into logical chunks of data, known as frames.

This is the layer at which the bridges and switches operate. Connectivity is provided only among locally attached network nodes forming layer 2 domains for unicast or broadcast forwarding. Other protocols may be imposed on the data frames to create tunnels and logically separated layer 2 forwarding domain.

The data link layer might implement a sliding window flow control and acknowledgment mechanism to provide reliable delivery of frames; that is the case for SDLC and HDLC, and derivatives of HDLC such as LAPB and LAPD. In modern practice, only error detection, not flow control using sliding window, is present in modern data link protocols such as Point-to-Point Protocol (PPP), and, on local area networks, the IEEE 802.2 LLC layer is not used for most protocols on Ethernet, and, on other local area networks, its flow control and acknowledgment mechanisms are rarely used. Sliding window flow control and acknowledgment is used at the transport layers by protocols such as TCP.

Layer 1: Physical Layer

The Physical layer defines all the electrical and physical specifications for devices. In particular, it defines the relationship between a device and a physical medium. This includes the layout of pins, voltages, and cable specifications. Hubs, repeaters, network adapters and Host Bus Adapters (HBAs used in Storage Area Networks) are physical-layer devices.

To understand the function of the physical layer in contrast to the functions of the data link layer, think of the physical layer as concerned primarily with the interaction of a single device with a medium, where the data link layer is concerned more with the interactions of multiple devices (i.e., at least two) with a shared medium. The physical layer will tell one device how to transmit to the medium, and another device how to receive from it, but not, with modern protocols, how to gain access to the medium. Obsolescent physical layer standards such as RS-232 do use physical wires to control access to the medium.

The major functions and services performed by the physical layer are:

- Establishment and termination of a connection to a communications medium.
- Participation in the process whereby the communication resources are effectively shared among multiple users. For example, contention resolution and flow control.
- Modulation, or conversion between the representation of digital data in user equipment and the corresponding signals transmitted over a communications channel. These are signals operating over the physical cabling (such as copper and fiber optic) or over a radio link.

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Parallel SCSI buses operate in this layer, although it must be remembered that the logical SCSI protocol is a transport-layer protocol that runs over this bus.. Various physical-layer Ethernet standards are also in this layer; Ethernet incorporates both this layer and the data-link layer. The same applies to other local-area networks, such as Token ring, FDDI, and IEEE 802.11, as well as personal area networks such as Bluetooth and IEEE 802.15.4.

1.4 NOVELL NETWARE

The most popular, at one stage, network system in the PC world was Novell NetWare. It was designed to be used by companies downsixing from a mainframe to a network of PCs. In such systems, each user has a desktop PC functioning as a client. In addition, some number of powerful PCs operate as servers, providing file services, database services, and other services to a collection of clients. In other words, Novell Netware is based on client-server mode.

1.5 ARPANET

ARPANET (Advanced Research Projects Agency Network), created by a small research team at the head of the Massachusetts Institute of Technology and the Defense Advanced Research Projects Agency (DARPA) of the United States Department of Defense, was the world's first operational packet switching network, and the predecessor of the contemporary global Internet. The packet switching of the ARPANET was based on designs by Lawrence Roberts, of the Lincoln Laboratory.

Packet switching, now the dominant basis for data communications worldwide, then was a new and important concept. Data communications had been based upon the idea of circuit switching, as in the old, typical telephone circuit, wherein a dedicated circuit is occupied for the duration of the telephone call, and communication is possible only with the single party at the far end of the circuit.

With packet switching, a data system could use one communications link to communicate with more than one machine by disassembling data into datagrams, then gather these as packets. Thus, not only could the link be shared (much as a single post box can be used to post letters to different destinations), but each packet could be routed independently of other packets.

Software & Protocols

The starting point for host-to-host communication on the ARPANET was the 1822 protocol, which defined how a host computer transmitted messages to an ARPANET IMP. The message format was designed to work unambiguously with a broad range of computer architectures. An 1822 message essentially consisted of (i) a message type, (ii) a numeric host address, and (iii) a data field. To send a data message to another host, the transmitting host would format a data message containing the destination host's address and the data message being sent, and then transmit the message through the 1822 hardware interface. The IMP then delivered the message to its destination address, either by delivering it to a locally connected host, or by delivering it to another IMP. When the message was ultimately delivered to the destination host, the receiving IMP would transmit a Ready for Next Message (RFNM) acknowledgement to the sending, host IMP.

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Unlike modern Internet datagrams, the ARPANET was designed to reliably transmit 1822 messages, and to inform the host computer when it loses a message; the contemporary IP is unreliable, whereas the TCP is reliable. Nonetheless, the 1822 protocol proved inadequate for handling multiple connections among different applications residing in a host computer. This problem was addressed with the Network Control Program (NCP), which provided a standard method to establish reliable, flow-controlled, bidirectional communications links among different processes in different host computers. The NCP interface allowed application software to connect across the ARPANET by implementing higher-level communication protocols, an early example of the protocol layering concept incorporated to the OSI model. In 1983, TCP/IP protocols replaced NCP as the ARPANET's principal protocol, and the ARPANET then became one component of the early Internet.

Network Applications

NCP provided a standard set of network services that could be shared by several applications running on a single host computer. This led to the evolution of application protocols that operated, more or less, independently of the underlying network service. When the ARPANET migrated to the Internet protocols in 1983, the major application protocols migrated with it.

E-mail: In 1971, Ray Tomlinson, of the BBN company sent the first network e-mail. By 1973, e-mail constituted 75 per cent of ARPANET traffic

File transfer: By 1973, the File Transfer Protocol (FTP) specification had been defined and implemented, enabling file transfers over the ARPANET

Voice traffic: The Network Voice Protocol (NVP) specifications were defined in (RFC 741), then implemented, but, because of technical shortcomings, conference calls over the ARPANET never worked well; the contemporary Voice over Internet Protocol (packet voice) was decades away

Growth

In March, 1970, the ARPANET reached the east coast of the United States, when a BBN company IMP was connected to the network. Thereafter, the ARPANET grew: 9 IMPs by June 1970 and 13 IMPs by December 1970, then 18 by September 1971 (when the network included 23 university and government hosts); 29 IMPs by August 1972, and 40 by September, 1973. By June 1974, there were 46 IMPs, and in July 1975, the network numbered 57 IMPs. By 1981, the number was 213 host computers, with another host connecting approximately every twenty days.

In 1968, two satellite links, traversing the Pacific and Atlantic oceans, to Hawaii and Norway, one, the Norwegian Seismic Array (NORSAR), were connected to the ARPANET. Moreover, from Norway, a terrestrial circuit added a London IMP to the network in 1973.

Given that its primary function was funding research and development, the ARPA, in 1975, transferred ARPANET control to the Defense Communications Agency, a component of the U.S. Department of Defense. In 1983, the U.S. military sub-networks of the ARPANET became the discrete Military Network (MILNET) for unclassified defense department communications; separating the civil and military networks reduced the 113-node ARPANET by 68 nodes.

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1.6 NSFNET

Initially it was decided that NSFNET would be a general-purpose research network, it would be a hub to connect regional networks at supercomputing sites, and that it would make use of the ARPANET's very successful TCP/IP protocol. In 1985, the NSF began funding the creation of five new supercomputer centers: the John von Neumann Center at Princeton University, the San Diego Supercomputer Center on the campus of the University of California at San Diego, the National Center for Supercomputing Applications at the University of Illinois at Urbana-Champaign, the Cornell Theory Center at Cornell University and the Pittsburgh Supercomputing Center. The NSFNET connected these five centers along with the NSF-funded National Center for Atmospheric Research, providing access to their supercomputers over the network at no cost.

The NSFNET went online in 1986, using a TCP/IP-based protocol that was compatible with ARPANET, as a backbone to which regional and academic networks would connect. The six backbone sites were interconnected with leased 56 kbit/s links built by a group including the University of Illinois National Center for Supercomputing Applications (NCSA), Cornell University Theory Center, University of Delaware, and Merit Networks Inc. PDP-11 minicomputers with routing and management software - called Fuzzballs - served as the network routers since they already implemented the TCP/IP standard. As regional networks began to grow the NSFNET backbone experienced exponential growth in its network traffic. As a result of a November 1987 NSF award to a consortium of universities in Michigan, the original 56 kbit/s links were upgraded to 1.5 Mbit/s by July 1988 and again to 45 Mbit/s in 1991.

1.7 THE INTERNET

It is said that Internet is network of networks. These networks are spread over various countries, various continents and are linked through Satellite, via telephone lines. At a given time there are more 20 million users using the Internet throughout the World. Inspite of all this there is nobody who controls Internet. It is a self running process and nobody can claim to be a single owner of it.

To understand this concept of Internet, we have to study, how and when it all began.

History of Internet

In 60s a project was undertaken by the U.S. Defense Advanced Research Projects Agency (DARPA). It was in fact looking for some technology that could enable it to maintain its strategic military-based communication worldwide in case of a nuclear attack. This can be said as the main conception of the Internet.

Later, these developments led to the establishment of the Advanced Research Projects Agency Net (ARPANet). The main interest of this was looking for a technology that could link computers in various locations by using a new technology called Packet Switching Technology. This new technology enabled several users to simultaneously share a single communication line. It also allowed the creation of nets that could automatically route data around downed circuits or computers. This technology was then used by National Science Foundation to create to its own net, and called it

NSDnet. It met with a large of success though its main users were universities and schools for information available over it.

Since the users were mostly scientists and researchers, the demand went on increasing endlessly, NSF found itself unable to cope with it. As a result, a decision was taken to open the Net for use by private organizations as well. This allowed anyone with a modem and a computer to access the Net. The beginning was made. Thus, NSFnet became the backbone of communication service for the Internet and continues to do so even now. At present, NSFNet comprises a set of highspeed data connections that join the major network all over the world.

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This lead to several users using the net for information and then came the phase when its potential for doing business was exploited and companies started marketing services and information over the net. Today in fact, anybody with a computer and modem with dedicated phone line can access Internet with the help of some service provided by local Internet Service Providers.

Internet and Intranet

Internet should not be confused with Intranet. To explain, Intranet is a private network within a company or an organization. An Intranet may use the same kinds of software that you may find on the Internet. Intranet is essentially used to exchange confidential information between the officials at certain levels, information that is not meant to be shared with others in the rest of that organization's overall network.

Information on Internet

To understand how the information is transferred through the Internet, we will have to study more about the Internet technology. After all it is very strange that it allows you share information not in the form of written words but also allows you to have the information in the vocal form.

Remember what I had said earlier, about Packet Switching. This packet switching technology is still used to transfer the data. Digital data made up of a series of 0s (OFFs) and 1s (ONs) are grouped in unique sequences. Each sequences of 0s (OFFs) and 1s (ONs) have a particular meaning, which is translated by the computers to enable you to view the matter on your computer's screen. To understand it more clearly, do remember that computers do not understand languages in the manner in which we speak and write. Instead, they understand languages based on electrical impulses that go 'ON' and 'OFF'. Speaking scientifically, such a system is called the Binary System in which specific combinations of 'ON' and 'OFF".

Thus, a message, sent by you through the Internet, first gets converted (translated) by your computer in to a digital format, made up of a series of ONs (1s) and OFFs (0s) that are grouped in specific and unique sequences. Each sequence of 1s and 0s has a particular meaning. At the final receiving point, these sequences are reconverted (re-translated) by your friend's computer so that your friend can read your message on his computer's screen.

Requirements for connecting to internet

To connect to Internet, you need to have the followings:

1. A computer system with a software like Microsoft Internet Explorer or Netscape Navigator loaded into it.

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2. A telephone line. It will be better to have a dedicated telephone line, i.e., a line exclusively for your Internet connection only. It is because, with the passage of time, you may fine that you are spending most of your time on the Net. This will prevent others from using your telephone or calling you, as they will often get an 'engaged' tone. However, if you wish to use the Internet occasionally, then your existing telephone line will suffice.
3. A modem (modular-demodulator) is a electronic device that converts digital data from computers into signals. These signals can then be transmitted over a normal telephone line. At the receiving end, another modem converts the signals back into digital data understood by computers. Modems can be internal, i.e., inserted in a slot on your computer's motherboard or external, i.e., fitted externally. Irrespective of whether your modem is internal or external, you will need a jack to connect your telephone line to your computer.
4. As mentioned earlier, there are companies who provide you with the services of providing you with the Internet Services. They are called Internet Service Provider (ISP). You have to open an account with ISP to have the connection. An ISP is a company that gives you access to the Internet for a fee. Presently, a number of ISPs are available in India. These include VSNL (Videsh Sanchar Nigam Limited); Satyam Online; Mantra Online; Tata Nova; etc. Each one of these allow you to open an account with them and they would give an e-mail address too.

If you are using Internet more for sending and receiving e-mails. Then you would also be needing the software called Outlook Express.

Basic Internet Terms

Before you start using (browsing/surfing) the Web, it is necessary for you to understand the following terms and their meanings specially in connection with the Internet:

Home page: It is the first page that you would see on the Website. Also known as the Welcome page. It is from here that you would start the navigation of various other pages of the site.

Hypertext and Hyperlinks: Information on the Web is made available in the form of Hypertext. It is a method of presenting information wherein some portion is highlighted. When this highlighted portion is selected, it displays more information on the topic/s that you choose. The highlighted items selected by you are technically called Hyperlinks. In fact, they allow you to navigate from one Web document to another on the same computer or on a different computers in your own city, country or anywhere else in the world.

Internet Information Server: It is a group of Internet servers including the additional capabilities of Windows NT and Windows 2000.

Internet Protocol: It is responsible for the addressing and sending data from one computer to another computer.

Internet Service Provider: It is one gateway to the Internet. As mentioned earlier, you need this service to connect to Internet from your computer.

Multimedia: At the heart of the Web is the ability to display multimedia information, such as images, audio, video, animation, and other multimedia data types.

Transmission Control Protocol: It uses a set of rules to exchange messages with other Internet points at the information packet level.

Web Browser: It is a software application that resides on your PC and can display text, images, and multimedia data found on different Web pages. It allows you to specify a Web page, navigate using links, and bookmark your favourite Web pages, The commonly used Web browsers are Internet Explorer and Netscape Navigator.

Web Server: A Web server refers to a location (computer) on the Internet that contains information in the form of Web pages. Technically speaking, a Web Server means a computer on the Internet having the capability to run software. A page stored on a Web server can be accessed by Web users. It may also be mentioned here that Internet Service Providers (ISPs) normally offer space on their Web servers on which their registered users can publish their Web pages free of charge.

Web Site: A Web site comprises of a collection of Web pages that may be maintained and updated by an organization like a Government or University department, a business house, a research institution etc. even a single individual can also create and maintain his/her own Web site to promote certain ideas. The information on a Web site is stored in the form of a series of files that may be stored on one or more computers. Do remember that the material on a Web site is stored in Hypertext.

Web Page: A Web page refers to a document on the Web. Web pages can be used to display written text, show pictures, play music/sound effects and run video. Do also remember that you need to use Hyper Text Markup Language (HTML) to create Web pages.

Internet Services and Uses

It is not that the Internet is used only for sending and receiving information. It has its commercial aspect too. Internet today offers a number of standardized services to all its users. In fact, it is like a multipurpose tool to communicate with one another, collect information and enhance your knowledge about the subjects that you like, conduct research, play games, invest in shares or just look around (usually called browsing) in search of recipes or even jokes or the latest technological developments. The description that follows will acquaint you with some of the well-known ways in which Internet is used worldwide. Some of the famous usages of Internet are:

Information Retrieval

One of the most commonly used services on the Net comprises retrieval of information about various topics that interest you. This retrieval (getting) of information is made possible through the World Wide Web (popularly called WWW or W3).

Internet and e-mail

The Internet facility used to the maximum is e-mail. It involves writing messages on a computer and transmitting them to another computer so that the addressee can read them, thus saving paper, time, energy and cost. In fact, it can be said that e-mail is the essence of all communications on the Internet. Practically, everyone with Internet access does have an e-mail account.

NOTES

Newsgroups (Usenet)

The Internet provides you a major avenue to communicate with large groups. Popularly, referred to as 'Newsgroups', this service is technically called Usenet and comprises a distributed bulletin board system. The information (news) shared by Usenet groups can be 'unmoderated' (unedited) or 'moderated' (edited.).

Accessing People

Internet would help you in locating your long lost friend, who was probably lost in school or college. That person must have an e-mail account. Rest is easy. For this purpose, you can visit a commonly used site such as whowhere (use your search engine to locate it by typing http://whowhere.lycos.com). Now, to find the e-Mail address, type the name of the person in the name field. Thereafter, type the e-mail server domain name (like hotmail.com) in the domain field. Once you click the Go Get It button, the result will be displayed on your computer's screen. You will also find that the on-screen display serves as a link, wherefrom, you can get the person's e-mail address.

Telnet

It is a text-based Internet service that connects you to a remote host (server). Using a special protocol known as the Network Terminal Protocol, it enables you to log onto another computer on the Internet and use its resources as if they existed on your own machine. To use this service, you need to provide your valid login and user password.

Chat

Everybody likes to chat. Even if it on the Internet. You can in fact chat with totally unknown people. Technically known as Internet Relay Chat (IRC), it is multi-user and multi-channel chatting net that allows users to communicate in real time. Remember! Chatting through IRC is in the written form, i.e., while communicating with some one, you type your message and the receiver responds (types back) with his/her comments. In short, it is form of instant talking, almost like a telephone conversation but in a written form. Residing on many systems on the Net, it is a software, which provides access to a series of interactive services. By availing these services, one can chat or play on-line games with people on the Internet.

FTP (File Transfer Protocol)

It connotes an Internet service that transfers files from one computer to another. It is a common procedure to download and upload files over the Internet. With FTP you can login to another Internet site and transfer (meaning send or receive) files. FTP works on the client/server principle. A client program enables the user to interact with a server in order to access information and services on the server computer.

Anonymous FTP

Some sites have public file archives that you can access by using FTP with the account name 'anonymous' and your e-Mail address as the password. This type of access is called anonymous FTP.

e-commerce

e-commerce means doing business online. It refers to any manner of conducting business on line by an individual/organization. As part of e-commerce, large organizations also send data from the Internet to conduct research and plan their

marketing strategies globally. With credit cards becoming more popular along with computerized banking services, payment for services through Internet is becoming very easy.

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Employment Generation

Job placement agencies and employers have started to increasingly use the Internet as a source of recruitment by advertising on the Net to fill up vacancies. Side by side, people seeking job or better employment opportunities also sue the Internet.

Medicare

Doctors now increasingly use the Internet to know the latest treatment techniques to benefit their patients. Hospitals sometimes use video conferencing to provide on-line guidance for conducting complicated operations.

Shopping Online

Some department stores offer their products, both new and old to Internet users to payment, which is usually made in advance through a credit card. While in the case of software, developers often post the software they have written – a process similar to posting a message. You can also download and use these programs free of cost and give a feedback to the programmer about its usefulness.

Entertainment

Internet now gives you unlimited opportunities to watch latest films, TV programs and listen to music. You can download your favorite movies and music from different sources.

World Wide Web (WWW)

The World Wide Web is a collection of million of files stored in thousand of computers (called Web server) all over world. Using WWW a user can download files, listen to sounds, view video files and jump to other documents on or Net sites by using hypertext links.

Educational Opportunities

Suppose you wish to study in America after passing the Senior Secondary examination. Just surf the Web and you will find that almost every university in the U.S. maintains a web site. Each of these sites provides extensive information, ranging from courses available to credit prices, course fees, etc., details of programs leading to various degrees and career planning services

Lastly I can say that with the help of computer and Internet, you can see the World on your desktop only. And that too anytime of the day.

SUMMARY

1. When more than two computers are connected to each other and sharing information, resources and remote systems then this is called Networking.
2. PAN is used for communication among the personal devices (intrapersonal communication), or for connecting to a higher level network and Internet. The reach of a PAN is typically a few meters.

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3. A network covering a small geographic area, like a home, office, or building is called Local Area Network.
4. WAN technologies generally function at the lower three layers of the OSI reference model: the physical layer, the data link layer, and the network layer.
5. If connected to the Internet, the intranet or extranet is normally protected from being accessed from the Internet without proper authorization
6. All networks are made up of basic hardware building blocks to interconnect network nodes, such as Network Interface Cards (NICs), Bridges, Hubs, Switches, and Routers.
7. A **network card**, **network adapter** or **NIC** (network interface card) is a piece of computer hardware designed to allow computers to communicate over a **computer network**.
8. A **network bridge** connects multiple network segments at the data link layer (layer 2) of the OSI model.
9. A hub is a piece of hardware which provides the connectivity of a segment of a network by directing traffic through the network.
10. Switches are the device of networking that directs traffic to the correct node by filtering and forwarding packets between Nodes.
11. Routers are the networking device that forwards data packets along networks by using headers and forwarding tables to determine the best path to forward the packets.
12. Closely related to the concept of a model is that of an *architecture*.
13. A **MAN** (metropolitan area network) is a larger network that usually spans several buildings in the same city or town.
14. A **WAN** (wide area network), in comparison to a MAN, is not restricted to a geographical location, although it might be confined within the bounds of a state or country.
15. The length of the cable connecting a computer to a LAN also varies depending on the LAN.
16. One of the major benefits of implementation of LAN is sharing expensive resources such as storage devices, printers, etc.
17. Three major categories of services used in LANs are: File Server; Printer Server and Modem Server.
18. Some popular LAN operating systems are : Novel Netware; Ethernet; Corvus; ArcNet LAN; Server Omni Net; PC Net; IBM PC LAN and Etherlink Plus, etc.
19. Features of LAN are: Typically connects computer in a single building or campus; Developed in 1970s; Medium : optical fibres, coaxial cables, twisted pair, wireless; Low latency (except in high traffic periods); High speed networks (0.2 to 100 Mb/sec); Speeds adequate for most distributed systems; Problems : Multi media based applications; Typically buses or rings and Ethernet, Token Ring.
20. Routers is a special type of device that can be used to connect networks that may not be similar.
21. Features of Wide Area Networks are: Developed in 1960s; Generally covers large distances (states, countries, continents); Medium : communication circuits connected by routers; Routers forwards packets from one to another following a route from the sender to the receiver. Store-and-Forward; Hosts are typically connected (or close to) the routers; Typical latencies : 100ms - 500ms; Problems with delays if using satellites; Typical speed : 20 - 2000 Kbits/s; Not (yet) suitable for distributed computing; and New standards are changing the landscape.
22. Features of MAN are: Generally covers towns and cities (50 kms); Developed in 1980s; Medium: optical fibres, cables; Data rates adequate for distributed computing applications; A typical standard is DQDB (Distributed Queue Dual Bus); Typical latencies : < 1 msec and Message routing is fast.
23. The **Open Systems Interconnection Basic Reference Model** (*OSI Reference Model* or *OSI Model* for short) is a layered, abstract description for communications and computer network protocol design, developed as part of Open Systems Interconnection (OSI) initiative.
24. In 1977, the International Organization for Standardization (ISO), began to develop its OSI networking suite.
25. The application layer is the seventh level of the seven-layer OSI model. It interfaces directly to and performs common application services for the application processes; it also issues requests to the presentation layer.

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26. The Presentation layer transforms the data to provide a standard interface for the Application layer.
27. The Session layer controls the dialogues/connections (sessions) between computers.
28. The Transport layer provides transparent transfer of data between end users, providing reliable data transfer services to the upper layers.
29. The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer.
30. The Data Link layer provides the functional and procedural means to transfer data between network entities and to detect and possibly correct errors that may occur in the Physical layer.
31. The Physical layer defines all the electrical and physical specifications for devices.
32. The **physical layer** is level one in the seven level OSI model. It performs services requested by the data link layer.
33. The **data link layer** is layer two of the seven-layer OSI model as well as of the five-layer TCP/IP reference model.
34. The uppermost sublayer is *Logical Link Control* (LLC).
35. The **network layer** is level three of the seven level OSI model as well as of the five layer TCP/IP model.
36. In computing and telecommunications, the **transport layer** is the second highest layer in the four and five layer TCP/IP reference models, where it responds to service requests from the application layer and issues service requests to the network layer.
37. The amount of memory on a computer is limited, and without flow control a larger computer might flood a computer with so much information that it can't hold it all before dealing with it.
38. Network congestion occurs when a queue buffer of a network node is full and starts to drop packets.
39. Ports are essentially ways to address multiple entities in the same location.
40. The **presentation layer** is the sixth level of the seven layer OSI model. It responds to service requests from the application layer and issues service requests to the session layer.
41. The **application layer** is the seventh level of the seven-layer OSI model. It interfaces directly to and performs common application services for the application processes; it also issues requests to the presentation layer.
42. The Internet Protocol (IP), in combination with Transmission Control Protocol (TCP), forms the TCP/IP suite, which is the de facto protocol (i.e., universal computer language) that connects the network of networks – that is, the Internet.
43. The number of valid networks and hosts available is always $2^N - 2$ (where N is the number of bits used, and the 2 adjusts for the invalidity of the first and last addresses). Thus, for a class C address with 8 bits available for hosts, the number of hosts is 254.
44. Internet is network of networks.
45. Nobody controls Internet.
46. NSFnet is the backbone of communication service for the Internet.
47. Intranet is a private network within a company or an organization.
48. A message, sent by through Internet, first gets converted by your computer into a digital format, made up of a series of ONs (1s) and OFFs (0s).
49. For connecting to Internet, you need a computer system with a software like Microsoft Internet Explorer or Netscape Navigator loaded into it; A telephone line. A modem (modular-demodulator); a connection provided by Internet Service Provider (ISP).
50. Home page is the first page that you would see on the Website.
51. Information on the Web is made available in the form of Hypertext.
52. The highlighted items selected by you are technically called Hyperlinks.

NOTES

SELF ASSESSMENT QUESTIONS

1. What is a computer network?
2. Define the various types of networks.
3. Which are the various hardware components required for networking?
4. Describe the various models of network computing.
5. What is Local Area Networks?
6. Describe Wide Area Networks.
7. Describe the various network services.
8. What is OSI model?
9. Describe the various layers.
10. Describe OSI Reference Model.
11. What are OSI physical layer concepts?
12. Describe Data-link layer concepts.
13. What are OSI network layer concepts?
14. Describe Transport layer concepts.
15. What are OSI Session layer concepts?
16. Describe OSI Presentation layer concepts.
17. What are OSI Application layer concepts?
18. What is Internet?
19. How was Internet developed?
20. What all you need for running Internet?

Short Questions with Answers

1. Which are the popular networks?

Ans. Personal Area Network (PAN)
Local Area Network (LAN)
Campus Area Network (CAN)
Metropolitan Area Network (MAN)
Wide Area Network (WAN)
Internetwork
Intranet
Extranet.

2. What are the hardware components of LAN?

Ans. The following are the major hardware components/devices for establishing LAN:
1. Transmission Channel
2. Network Interface Unit or NIU
3. Servers
4. Workstations.

3. What are the various features of LAN?

Ans.
- Typically connects computer in a single building or campus.
- Developed in 1970s.
- Medium : optical fibres, coaxial cables, twisted pair, wireless.
- Low latency (except in high traffic periods).
- High speed networks (0.2 to 100 Mb/sec).
- Speeds adequate for most distributed systems
- Problems : Multi media based applications
- Typically buses or rings.
- Ethernet, Token Ring.

4. What are the features of WAN?

Ans.
- Developed in 1960s.

NOTES

- Generally covers large distances (states, countries, continents).
- Medium : communication circuits connected by routers.
- Routers forwards packets from one to another following a route from the sender to the receiver. Store-and-Forward
- Hosts are typically connected (or close to) the routers.
- Typical latencies : 100ms - 500ms.
- Problems with delays if using satellites.
- Typical speed : 20 - 2000 Kbits/s.
- Not (yet) suitable for distributed computing.
- New standards are changing the landscape.

5. What are the features of MAN?

Ans.
- Generally covers towns and cities (50 kms)
- Developed in 1980s.
- Medium : optical fibres, cables.
- Data rates adequate for distributed computing applications.
- A typical standard is DQDB (Distributed Queue Dual Bus).
- Typical latencies : < 1 msec.
- Message routing is fast.

6. Which are the seven layers?

Ans. Layer 7: Application layer
Layer 6: Presentation layer
Layer 5: Session layer
Layer 4: Transport layer
Layer 3: Network layer
Layer 2: Data Link layer
Layer 1: Physical layer.

7. Which are the major functions and services performed by the physical layer?

Ans. The major functions and services performed by the physical layer are:
- Establishment and termination of a connection to a communications medium.
- Participation in the process whereby the communication resources are effectively shared among multiple users. For example, contention resolution and flow control.
- Modulation, or conversion between the representation of digital data in user equipment and the corresponding signals transmitted over a communications channel. These are signals operating over the physical cabling (such as copper and fiber optic) or over a radio link.

8. What does presentation layer do?

Ans. The Presentation layer transforms the data to provide a standard interface for the Application layer.

9. What does the session layer do?

Ans. The Session layer controls the dialogues/connections (sessions) between computers.

10. What does transport layer provide?

Ans. The Transport layer provides transparent transfer of data between end users, providing reliable data transfer services to the upper layers.

11. What does network layer provide?

Ans. The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer.

12. What does data link layer provide?

Ans. The Data Link layer provides the functional and procedural means to transfer data between network entities and to detect and possibly correct errors that may occur in the Physical layer.

13. What does physical layer define?

Ans. The Physical layer defines all the electrical and physical specifications for devices.

NOTES

14. What is application layer?

Ans. The **application layer** is the seventh level of the seven-layer OSI model. It interfaces directly to and performs common application services for the application processes; it also issues requests to the presentation layer.

The common application layer services provide semantic conversion between associated application processes. Examples of common application services of general interest include the virtual file, virtual terminal, and job transfer and manipulation protocols.

The application layer of the four layer and five layer TCP/IP models corresponds to the application layer, the presentation layer and session layer in the seven layer OSI model.

Further Readings

1. Computer Networks: Ajit Kumar Singh, Firewal Media.
2. Data and Computer Network Communication: Prof. Shashi Banzai, Firewal Media.
3. TCP / IP and Distributed System: Vivek Archarya, Firewal Media.
4. Networking: Balvir Singh, Firewal Media.

UNIT 2

THE PHYSICAL LAYER

NOTES

STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- learn about Transmission Media, which is the main media of transmission of energy.
- understand about Twisted Pair of cables which are used in networking.
- explain about the Baseband and Broadband Coaxial Cable, which are used for networking.
- know about Fiber optics, which has glass core surrounded by several layers of protective materials.
- learn about the various types of Transmissions of Date, e.g., Wireless Transmission, Radio Transmission, Microwave Transmission, Infrared and Light Wave Transmission.
- understand about ISDN services used in networking.
- learn about the differences between Virtual Circuits and Circuit Switching.
- know about Transmission in ATM Networks.
- learn about Paging Systems, Cordless Telephones, Cellular Telephones and Communication Satellite.

NOTES

2.1 TRANSMISSION MEDIA

A transmission line is the material medium or structure that forms all or part of a path from one place to another for directing the transmission of energy, such as electromagnetic waves or acoustic waves, as well as electric power transmission. Components of transmission lines include wires, coaxial cables, dielectric slabs, optical fibers, electric power lines, and waveguides.

History

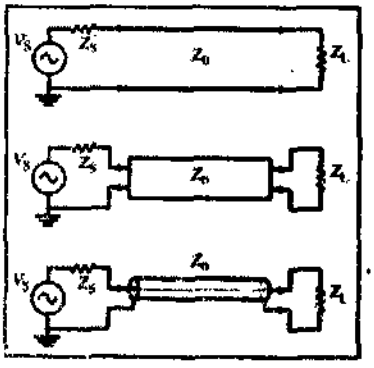
Mathematical analysis of the behaviour of electrical transmission lines grew out of the work of James Clerk Maxwell, Lord Kelvin and Oliver Heaviside. In 1855 Lord Kelvin formulated a diffusion model of the current in a submarine cable. The model correctly predicted the poor performance of the 1858 trans-Atlantic submarine telegraph cable. In 1885 Heaviside published the first papers that described his analysis of propagation in cables and the modern form of the telegrapher's equations.

Transmission Line vs Wire

In many electric circuits, the length of the wires connecting the components can for the most part be ignored. That is, the voltage on the wire at a given time can be assumed to be the same at all points. However, when the voltage changes in a time interval comparable to the time it takes for the signal to travel down the wire, the length becomes important and the wire must be treated as a transmission line. Stated another way, the length of the wire is important when the signal includes frequency components with corresponding wavelengths comparable to the length of the wire.

A common rule of thumb (justified in the input impedance section) is that the cable or wire should be treated as a transmission line if the length is greater than 1/100 of the wavelength.

At this length the phase delay and the interference of any reflections on the line become important and can lead to unpredictable behavior in systems which have not been carefully designed using transmission line theory.



The Four Terminal Model

Variations on the electrical schematic for a transmission line.

For the purposes of analysis, an electrical transmission line can be modelled as a two-port network (also called a quadrupole network), as follows:

In the simplest case, the network is assumed to be linear (i.e. the complex voltage across either port is proportional to the complex current flowing into it when there are no reflections), and the two ports are assumed to be interchangeable. If the transmission line is uniform along its length, then its behaviour

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is largely described by a single parameter called the characteristic impedance, symbol Z0. This is the ratio of the complex voltage of a given wave to the complex current of the same wave at any point on the line. Typical values of Z0 are 50 or 75 ohms for a coaxial cable, about 100 ohms for a twisted pair of wires, and about 300 ohms for a common type of untwisted pair used in radio transmission.

When sending power down a transmission line, it is usually desirable that as much power as possible will be absorbed by the load and as little as possible will be reflected back to the source. This can be ensured by making the source and load impedances equal to Z0, in which case the transmission line is said to be matched.

Some of the power that is fed into a transmission line is lost because of its resistance. This effect is called ohmic or resistive loss. At high frequencies, another effect called dielectric loss becomes significant, adding to the losses caused by resistance. Dielectric loss is caused when the insulating material inside the transmission line absorbs energy from the alternating electric field and converts it to heat

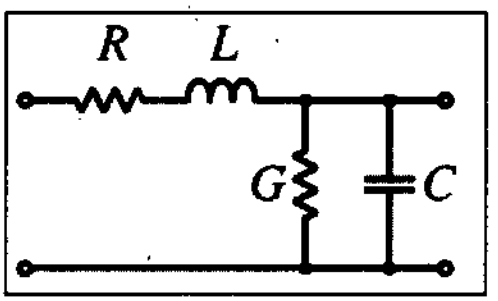
The total loss of power in a transmission line is often specified in decibels per metre, and usually depends on the frequency of the signal.

The manufacturer often supplies a chart showing the loss in dB/m at a range of frequencies. A loss of 3 dB corresponds approximately to a halving of the power.

High-frequency transmission lines can be defined as transmission lines that are designed to carry electromagnetic waves whose wavelengths are shorter than or comparable to the length of the line. Under these conditions, the approximations useful for calculations at lower frequencies are no longer accurate. This often occurs with radio, microwave and optical signals, and with the signals found in high-speed digital circuits.

Telegrapher's Equations

The Telegrapher's Equations (or just Telegraph Equations) are a pair of linear differential equations which describe the voltage and current on an electrical transmission line with distance and time. They were developed by Oliver Heaviside who created the transmission line model, and are based on Maxwell's Equations.



Schematic representation of the elementary component of a transmission line.

The transmission line model represents the transmission line as an infinite series of two-port elementary components, each representing an infinitesimally short segment of the transmission line:

- The distributed resistance R of the conductors is represented by a series resistor (expressed in ohms per unit length).
- The distributed inductance L (due to the magnetic field around the wires, self-inductance, etc.) is represented by a series inductor (henries per unit length).
- The capacitance C between the two conductors is represented by a shunt capacitor C (farads per unit length).
- The conductance G of the dielectric material separating the two conductors is

represented by a conductance G shunted between the signal wire and the return wire (siemens per unit length).

2.2 TWISTED PAIR

NOTES

Cable is the medium through which information usually moves from one network device to another. There are several types of cable which are commonly used with LANs. In some cases, a network will utilize only one type of cable, other networks will use a variety of cable types. The type of cable chosen for a network is related to the network's topology, protocol, and size. Understanding the characteristics of different types of cable and how they relate to other aspects of a network is necessary for the development of a successful network.

The following sections discuss the types of cables used in networks and other related topics.

- Unshielded Twisted Pair (UTP) Cable
- Shielded Twisted Pair (STP) Cable
- Coaxial Cable
- Fiber Optic Cable
- Wireless LANs
- Cable Installation Guides

Unshielded Twisted Pair (UTP) Cable

Twisted pair cabling comes in two varieties: shielded and unshielded. Unshielded twisted pair (UTP) is the most popular and is generally the best option for school networks (See fig. 1).

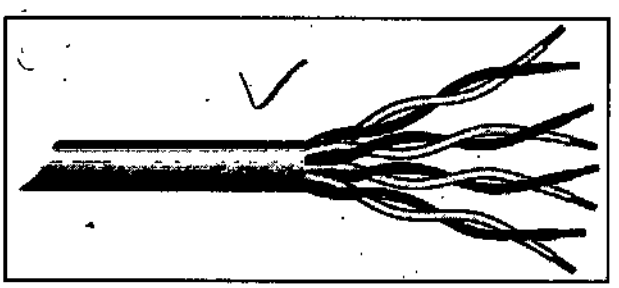


Fig.1. Unshielded twisted pair

The quality of UTP may vary from telephone-grade wire to extremely high-speed cable. The cable has four pairs of wires inside the jacket. Each pair is twisted with a different number of twists per inch to help eliminate interference from adjacent pairs and other electrical devices.

The tighter the twisting, the higher the supported transmission rate and the greater the cost per foot. The EIA/TIA (Electronic Industry Association/Telecommunication Industry Association) has established standards of UTP and rated five categories of wire.

Categories of Unshielded Twisted Pair

Type	Use
Category 1	Voice Only (Telephone Wire)
Category 2	Data to 4 Mbps (LocalTalk)
Category 3	Data to 10 Mbps (Ethernet)
Category 4	Data to 20 Mbps (16 Mbps Token Ring)
Category 5	Data to 100 Mbps (Fast Ethernet)

NOTES

Buy the best cable you can afford; most schools purchase Category 3 or Category 5. If you are designing a 10 Mbps Ethernet network and are considering the cost savings of buying Category 3 wire instead of Category 5, remember that the Category 5 cable will provide more "room to grow" as transmission technologies increase. Both Category 3 and Category 5 UTP have a maximum segment length of 100 meters. In Florida, Category 5 cable is required for retrofit grants. 10BaseT refers to the specifications for unshielded twisted pair cable (Category 3, 4, or 5) carrying Ethernet signals. Category 6 is relatively new and is used for gigabit connections.

Unshielded Twisted Pair Connector

The standard connector for unshielded twisted pair cabling is an RJ-45 connector. This is a plastic connector that looks like a large telephone-style connector (See fig. 2). A slot allows the RJ-45 to be inserted only one way. RJ stands for Registered Jack, implying that the connector follows a standard borrowed from the telephone industry. This standard designates which wire goes with each pin inside the connector.

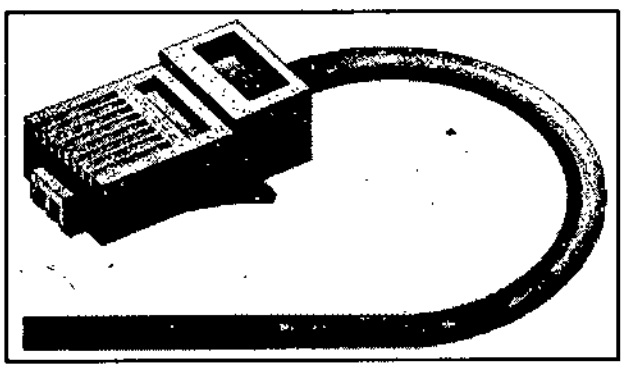


Fig. 2. RJ-45 connector

Shielded Twisted Pair (STP) Cable

A disadvantage of UTP is that it may be susceptible to radio and electrical frequency interference. Shielded twisted pair (STP) is suitable for environments with electrical interference; however, the extra shielding can make the cables quite bulky. Shielded twisted pair is often used on networks using Token Ring topology.

2.3 BASEBAND AND BROADBAND COAXIAL CABLE

Baseband is an adjective that describes signals and systems whose range of frequencies

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is measured from 0 to a maximum bandwidth or highest signal frequency; it is sometimes used as a noun for a band of frequencies starting at 0. It can often be considered as synonym to lowpass, and antonym to passband.

Various uses

- A baseband bandwidth is equal to a highest frequency of a signal or system, or an upper bound on such frequencies. By contrast, a non-baseband (passband) bandwidth is the difference between a highest frequency and a nonzero lowest frequency.
- A baseband signal or lowpass signal is a signal that can include frequencies that are equal to or very near zero, by comparison with its highest frequency (for example, a sound waveform can be considered as a baseband signal, whereas a radio signal is not).
- A baseband channel or lowpass channel (or system, or network) is a channel (e.g. a telecommunications system) that can transfer frequencies that are equal to or very near zero. Examples are serial cables and local area networks (LANs).
- Baseband modulation, also known as line coding, aims at transferring a digital bit stream over an analog baseband channel, as an alternative to carrier-modulated approaches.
- An equivalent baseband signal or equivalent lowpass signal is – in analog and digital modulation methods with constant carrier frequency (for example ASK, PSK and QAM but not FSK) – a complex valued representation of the modulated physical signal (the so called passband signal or RF signal). The equivalent baseband signal is where I(t) is the inphase signal, Q(t) the quadrature phase signal, and j the imaginary unit. In a digital modulation method, the I(t) and Q(t) signals of each modulation symbol are evident from the constellation diagram. The physical passband signal corresponds to where ω is the carrier angular frequency in rad/s.
- A signal "at baseband" is usually considered to include frequencies from near 0 Hz up to the highest frequency in the signal with significant power.

In general, signals can be described as including a whole range of different frequencies added together. In telecommunications in particular, it is often the case that those parts of the signal which are at low frequencies are 'copied' up to higher frequencies for transmission purposes, since there are few communications media that will pass low frequencies without distortion. Then, the original, low frequency components, are referred to as the baseband signal. Typically, the new, high-frequency copy is referred to as the 'RF' (radio-frequency) signal.

The concept of baseband signals is most often applied to real-valued signals, and systems that handle real-value signals. Fourier analysis of such signals includes a negative-frequency band, but the negative-frequency information is just a mirror of the positive-frequency information, not new information.

For complex-valued signals, on the other hand, the negative frequencies carry new information. In that case, the full two-sided bandwidth is generally quoted, rather than just the half measured from zero; the concept of baseband can be applied by treating the real and imaginary parts of the complex-valued signal as two different real signals.

Coaxial Cable

Coaxial cabling has a single copper conductor at its center. A plastic layer provides insulation between the center conductor and a braided metal shield (See fig. 3). The metal shield helps to block any outside interference from fluorescent lights, motors, and other computers.

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Fig. 3. Coaxial cable

Although coaxial cabling is difficult to install, it is highly resistant to signal interference. In addition, it can support greater cable lengths between network devices than twisted pair cable. The two types of coaxial cabling are thick coaxial and thin coaxial.

Thin coaxial cable is also referred to as thinnet. 10Base2 refers to the specifications for thin coaxial cable carrying Ethernet signals. The 2 refers to the approximate maximum segment length being 200 meters. In actual fact the maximum segment length is 185 meters. Thin coaxial cable is popular in school networks, especially linear bus networks.

Thick coaxial cable is also referred to as thicknet. 10Base5 refers to the specifications for thick coaxial cable carrying Ethernet signals. The 5 refers to the maximum segment length being 500 meters. Thick coaxial cable has an extra protective plastic cover that helps keep moisture away from the center conductor. This makes thick coaxial a great choice when running longer lengths in a linear bus network. One disadvantage of thick coaxial is that it does not bend easily and is difficult to install.

Coaxial Cable Connectors

The most common type of connector used with coaxial cables is the Bayone-Neill-Concelman (BNC) connector (See fig. 4). Different types of adapters are available for BNC connectors, including a T-connector, barrel connector, and terminator. Connectors on the cable are the weakest points in any network. To help avoid problems with your network, always use the BNC connectors that crimp, rather than screw, onto the cable.

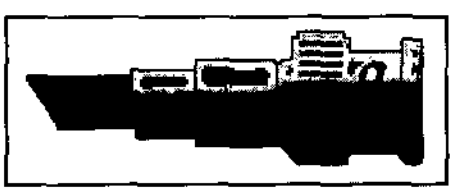

Fig. 4. BNC connector

2.4 FIBER OPTICS

Fiber optic cabling consists of a center glass core surrounded by several layers of protective materials (See fig. 5). It transmits light rather than electronic signals eliminating the problem of electrical interference. This makes it ideal for certain environments that contain a large amount of electrical interference. It has also made

it the standard for connecting networks between buildings, due to its immunity to the effects of moisture and lighting.

Fiber optic cable has the ability to transmit signals over much longer distances than coaxial and twisted pair. It also has the capability to carry information at vastly greater speeds. This capacity broadens communication possibilities to include services such as video conferencing and interactive services. The cost of fiber optic cabling is comparable to copper cabling; however, it is more difficult to install and modify. 10BaseF refers to the specifications for fiber optic cable carrying Ethernet signals.

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Fig.5. Fiber optic cable

Facts about fiber optic cables:

- Outer insulating jacket is made of Teflon or PVC.
- Kevlar fiber helps to strengthen the cable and prevent breakage.
- A plastic coating is used to cushion the fiber center.
- Center (core) is made of glass or plastic fibers.

Fiber Optic Connector

The most common connector used with fiber optic cable is an ST connector. It is barrel shaped, similar to a BNC connector. A newer connector, the SC, is becoming more popular. It has a squared face and is easier to connect in a confined space.

Ethernet Cable Summary

Specification	Cable Type	Maximum length
10BaseT	Unshielded Twisted Pair	100 meters
10Base2	Thin Coaxial	185 meters
10Base5	Thick Coaxial	500 meters
10BaseF	Fiber Optic	2000 meters
100BaseT	Unshielded Twisted Pair	100 meters
100BaseTX	Unshielded Twisted Pair	220 meters

2.5 WIRELESS TRANSMISSION

Wireless communication is the transfer of information over a distance without the use of enhanced electrical conductors or "wires". The distances involved may be short (a few meters as in television remote control) or long (thousands or millions of kilometers for radio communications). When the context is clear, the term is often

shortened to "wireless". Wireless communication is generally considered to be a branch of telecommunications.

It encompasses various types of fixed, mobile, and portable two-way radios, cellular telephones, personal digital assistants (PDAs), and wireless networking. Other examples of wireless technology include GPS units, garage door openers and or garage doors, wireless computer mice, keyboards and headsets, satellite television and cordless telephones.

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Applications of Wireless Technology

Security systems

Wireless technology may supplement or replace hard wired implementations in security systems for homes or office buildings.

Television remote control

Modern televisions use wireless (generally infrared) remote control units. Now radio waves are also used.

Cellular telephone (phones and modems)

Perhaps the best known example of wireless technology is the cellular telephone and modems. These instruments use radio waves to enable the operator to make phone calls from many locations worldwide. They can be used anywhere that there is a cellular telephone site to house the equipment that is required to transmit and receive the signal that is used to transfer both voice and data to and from these instruments.

Wi-Fi

Wi-Fi is a wireless local area network that enables portable computing devices to connect easily to the Internet. Standardized as IEEE 802.11 a, b, g, n, Wi-Fi approaches speeds of some types of wired Ethernet. Wi-Fi hot spots have been popular over the past few years. Some businesses charge customers a monthly fee for service, while others have begun offering it for free in an effort to increase the sales of their goods.

Wireless energy transfer

Wireless energy transfer is a process whereby electrical energy is transmitted from a power source to an electrical load that does not have a built-in power source, without the use of interconnecting wires.

Computer Interface Devices

Answering the call of customers frustrated with cord clutter, many manufactures of computer peripherals turned to wireless technology to satisfy their consumer base. Originally these units used bulky, highly limited transceivers to mediate between a computer and a keyboard and mouse, however more recent generations have used small, high quality devices, some even incorporating Bluetooth. These systems have become so ubiquitous that some users have begun complaining about a lack of wired peripherals. Wireless devices tend to have a slightly slower response time than their wired counterparts, however the gap is decreasing. Initial concerns about the security of wireless keyboards have also been addressed with the maturation of the technology.

Many scientists have complained that wireless technology interferes with their

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experiments, forcing them to use less optimal peripherals because the optimum one is not available in a wired version. This has become especially prevalent among scientists who use trackballs as the number of models in production steadily decreases.

2.6 RADIO TRANSMISSION

In telecommunications, transmission is the process of sending, propagating and receiving an analogue or digital information signal over a physical point-to-point or point-to-multipoint transmission medium, either wired, optical fiber or wireless. Transmission technologies and schemes typically refer to physical layer protocol duties such as modulation, demodulation, line coding, equalization, error control, bit synchronization and multiplexing, but the term may also involve higher-layer protocol duties, for example, digitizing an analog message signal, and source coding (compression).

Transmission of a digital message, or of a digitized analog signal, is known as data transmission or digital communication. One transmission is the sending of a signal with limited duration, for example a block or packet of data, or a phone call.

Radio waves are easy to penerate, can travel long distances, and penetrate buildings easily, so they are widely used for communication, both indoor and outdoors. Radio waves also are omnidirectional, meaning that they travel in all directions from the source, so that the transmitter and receiver do not have to carefully aligned physically.

The properties of radio waves are frequency dependent. A low frequencies, radio waves pass through obstacles well, but the power falls off sharply with distance from the source, roughly as $1/r^3$. in air. At high frequencies, radio waves tent to travel in straight lines and bounce off obstacles. They are also absorbed by rain. At all frequencies, radio waves are subject to interference from motors and other electrical equipment.

2.7 MICROWAVE TRANSMISSION

Microwave transmission refers to the technology of transmitting information by the use of the radio waves whose wavelengths are conveniently measured in small numbers of centimeters, by using various electronic technologies. These are called microwaves. This part of the radio spectrum ranges across frequencies of roughly 1.0 gigahertz (GHz) to 30 GHz. Also by using the formula $\lambda = c/f$, these correspond to wavelengths from 30 centimeters down to 1.0 cm. [In the above equation, the Greek letter λ (lambda) is the wavelength in meters; c is the speed of light in meters per second; and f is the frequency in hertz (Hz).]

In the microwave frequency band, antennas are usually of convenient sizes and shapes, and also the use of metal waveguides for carrying the radio power works well. Furthermore, with the use of the modern solid-state electronics and traveling wave tube technologies that have been developed since the early 1960s, the electronics used by microwave radio transmission have been readily used by expert electronics engineers.

Microwave radio transmission is commonly used by communication systems on the surface of the Earth, in satellite communications, and in deep space radio

communications. Other parts of the microwave radio band are used for radars, radio navigation systems, sensor systems, and radio astronomy.

The next higher part of the radio electromagnetic spectrum, where the frequencies are above 30 GHz and below 100 GHz, are called "millimeter waves" because their wavelengths are conveniently measured in millimeters, and their wavelengths range from 10 mm down to 3.0 mm. Radio waves in this band are usually strongly attenuated by the Earthly atmosphere and particles contained in it, especially during wet weather. Also, in wide band of frequencies around 60 GHz, the radio waves are strongly attenuated by molecular oxygen in the atmosphere. The electronic technologies needed in the millimeter wave band are also much more difficult to utilize than those of the microwave band.

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Uses

Backbone or backhaul carriers in cellular networks. Used to link BTS-BSC and BSC-MSC.

Communication with satellites

Microwave radio relay links for television and telephone service providers

A parabolic antenna for Erdfunkstelle Raisting; the biggest facility for satellite communication in the world, based in Raisting, Bavaria, Germany.

Parabolic (microwave) antenna

A parabolic antenna is a high-gain reflector antenna used for radio, television and data communications, and also for radiolocation (radar), on the UHF and SHF parts of the electromagnetic spectrum. The relatively short wavelength of electromagnetic radiation at these frequencies allows reasonably sized reflectors to exhibit the desired highly directional response for both receiving and transmitting.

Microwave link

A microwave link is a communications system that uses a beam of radio waves in the microwave frequency range to transmit video, audio, or data between two locations, which can be from just a few feet or meters to several miles or kilometers apart. Microwave links are commonly used by television broadcasters to transmit programmes across a country, for instance, or from an outside broadcast back to a studio.

Mobile units can be camera mounted, allowing cameras the freedom to move around without trailing cables. These are often seen on the touchlines of sports fields on Steadicam systems.

Properties of microwave links

- Involve line of sight (LOS) communication technology
- Affected greatly by environmental constraints, including rain fade
- Have limited penetration capabilities
- Sensitive to high pollen count
- Signals can be degraded during Solar proton event

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Uses of microwave links

- In communications between satellites and base stations
- As backbone carriers for cellular systems
- In short range indoor communications

Tunable microwave device

A tunable microwave device is a device that works at radio frequency range with the dynamic tunable capabilities, especially an electric field. The material systems for such a device usually have multilayer structure. Usually, magnetic or ferroelectric film on ferrite or superconducting film is adopted. The former two are used as the property tunable component to control the working frequency of the whole system. Devices of this type include tunable varators, tunable microwave filters, tunable phase shifters, and tunable resonators. The main application of them is re-configurable microwave networks, for example, reconfigurable wireless communication, wireless network, and reconfigurable phase array antenna.

2.8 INFRARED TRANSMISSION

It is widely used for short-range communication. The remote controls used on televisions, VCRs, and stereos all use infrared communication. They are relatively directional, cheap, and easy to build, but have a major drawback; they do not pass through solid objects (try standing between your remote control and your television and see if it still works). In general, as we go from long-wave radio toward visible light, the waves behave more and more like light and less and less like radio.

2.9 LIGHT WAVE TRANSMISSION

Its most common use is to connect the LANs in two buildings via lasers mounted on their rooftops. Coherent optical signaing using lasers is inherently unidirectional, so each building needs its own laser and its own photdetector. This scheme offers very high bandwidth and very low cost. It is also relatively easy to install, and, unlike microwave, does not require an FCC license. A disadvantage is that laser beams cannot penetrate rain or thick fog, but they normally work well on sunny days.

2.10 ISDN SERVICES

Integrated Services Digital Network (ISDN) is a set of communications standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network. It was first defined in 1988 in the CCITT red book.

Prior to ISDN, the phone system was viewed as a way to transport voice, with some special services available for data. The key feature of ISDN is that it integrates speech and data on the same lines, adding features that were not available in the classic telephone system. There are several kinds of access interfaces to ISDN defined as Basic Rate Interface (BRI), Primary Rate Interface (PRI) and Broadband ISDN (B-ISDN).

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ISDN is a circuit-switched telephone network system, which also provides access to packet switched networks, designed to allow digital transmission of voice and data over ordinary telephone copper wires, resulting in potentially better voice quality than an analog phone can provide. It offers circuit-switched connections (for either voice or data), and packet-switched connections (for data), in increments of 64 kilobit/s. A major market application for ISDN in some countries is Internet access, where ISDN typically provides a maximum of 128 kbit/s in both upstream and downstream directions. ISDN B-channels can be bonded to achieve a greater data rate, typically 3 or 4 BRIs (6 to 8 64 kbit/s channels) are bonded.

ISDN should not be mistaken for its use with a specific protocol, such as Q.931 whereby ISDN is employed as the network, data-link and physical layers in the context of the OSI model. In a broad sense ISDN can be considered a suite of digital services existing on layers 1, 2, and 3 of the OSI model. ISDN is designed to provide access to voice and data services simultaneously.

However, common use has reduced ISDN to be limited to Q.931 and related protocols, which are a set of protocols for establishing and breaking circuit switched connections, and for advanced call features for the user. They were introduced in 1986. In a videoconference, ISDN provides simultaneous voice, video, and text transmission between individual desktop videoconferencing systems and group (room) videoconferencing systems.

Configurations

In ISDN, there are two types of channels, B (for "bearer") and D (for "delta"). B channels are used for data (which may include voice), and D channels are intended for signaling and control (but can also be used for data).

There are two ISDN implementations. Basic Rate Interface (BRI), also called basic rate access (BRA) — consists of two B channels, each with bandwidth of 64 kbit/s, and one D channel with a bandwidth of 16 kbit/s. Together these three channels can be designated as 2B+D. Primary Rate Interface (PRI), also called primary rate access (PRA) in Europe — contains a greater number of B channels and a D channel with a bandwidth of 64 kbit/s. The number of B channels for PRI varies according to the nation: in North America and Japan it is 23B+1D, with an aggregate bit rate of 1.544 Mbit/s (T1); in Europe, India and Australia it is 30B+1D, with an aggregate bit rate of 2.048 Mbit/s (E1). Broadband Integrated Services Digital Network (BISDN) is another ISDN implementation and it is able to manage different types of services at the same time. It is primarily used within network backbones and employs ATM.

Another alternative ISDN configuration can be used in which the B channels of an ISDN BRI line are bonded to provide a total duplex bandwidth of 128 kbit/s. This precludes use of the line for voice calls while the internet connection is in use. The B channels of several BRIs can be BONDED, a typical use is a 384K videoconferencing channel.

Using bipolar with eight-zero substitution encoding technique, call data is transmitted over the data (B) channels, with the signaling (D) channels used for call setup and management. Once a call is set up, there is a simple 64 kbit/s synchronous bidirectional data channel (actually implemented as two simplex channels, one in each direction) between the end parties, lasting until the call is terminated. There can be as many calls as there are bearer channels, to the same or different end-points. Bearer channels may also be multiplexed into what may be considered single, higher-bandwidth channels

via a process called B channel BONDING, or via use of Multi-Link PPP "bundling" or by using an H0, H11, or H12 channel on a PRI.

The D channel can also be used for sending and receiving X.25 data packets, and connection to X.25 packet network, this is specified in X.31. In practice, X.31 was only commercially implemented in UK, France and Japan.

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Reference points

- A set of reference points are defined in the ISDN standard to refer to certain points between the telco and the end user ISDN equipment.
- R - defines the point between a non-ISDN device and a terminal adapter (TA) which provides translation to and from such a device
- S - defines the point between the ISDN equipment (or TA) and a Network Termination Type 2 (NT-2) device
- T - defines the point between the NT-2 and NT-1 devices1

Types of communications

Among the kinds of data that can be moved over the 64 kbit/s channels are pulse-code modulated voice calls, providing access to the traditional voice PSTN. This information can be passed between the network and the user end-point at call set-up time. In North America, ISDN is now used mostly as an alternative to analog connections, most commonly for Internet access. Some of the services envisioned as being delivered over ISDN are now delivered over the Internet instead. In Europe, and in Germany in particular, ISDN has been successfully marketed as a phone with features, as opposed to a POTS phone with few or no features.

Meanwhile, features that were first available with ISDN (such as Three-Way Call, Call Forwarding, Caller ID, etc.) are now commonly available for ordinary analog phones as well, eliminating this advantage of ISDN. Another advantage of ISDN was the possibility of multiple simultaneous calls (one call per B channel), e.g. for big families, but with the increased popularity and reduced prices of mobile telephony this has become less interesting as well, making ISDN unappealing to the private customer. However, ISDN is typically more reliable than POTS, and has a significantly faster call setup time compared with POTS, and IP connections over ISDN typically have some 30–35ms round trip time, as opposed to 120–180ms (both measured with otherwise unused lines) over 56k or V.34/V.92 modems, making ISDN more reliable and more efficient for telecommuters.

Where an analog connection requires a modem, an ISDN connection requires a terminal adapter (TA). The function of an ISDN terminal adapter is often delivered in the form of a PC card with an S/T interface, and single-chip solutions seem to exist, considering the plethora of combined ISDN- and ADSL-routers.

ISDN is commonly used in radio broadcasting. Since ISDN provides a high quality connection this assists in delivering good quality audio for transmission in radio. Most radio studios are equipped with ISDN lines as their main form of communication with other studios or standard phone lines. Equipment made by companies such as Telos/Omnia (the popular Zephyr codec), Comrex, Tieline and others are used regularly by radio broadcasters. Almost all live sports broadcasts on radio are backhauled to their main studios via ISDN connections.

2.11 VIRTUAL CIRCUITS VERSES CIRCUIT SWITCHING

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In telecommunications and computer networks, a virtual circuit (VC), synonymous with virtual connection and virtual channel, is a connection oriented communication service that is delivered by means of packet mode communication. After a connection or virtual circuit is established between two nodes or application processes, a bit stream or byte stream may be delivered between the nodes; 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 signalling overhead is required during a connection establishment phase. However, circuit switching provides constant bit rate and latency, while these may vary in a virtual circuit service because of reasons 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, by means of data retransmissions because of error detection and automatic repeat request (ARQ).

Examples of protocols that provide virtual circuits

- Transmission Control Protocol (TCP), where a reliable virtual circuit is established on top of the underlying unreliable and connectionless IP protocol. The virtual circuit is identified by the source and destination network socket address pair, i.e. the sender and receiver IP address and port number. Guaranteed QoS is not provided.
- SCTP, where a virtual circuit is established on top of either the IP protocol or the UDP protocol.

Examples of network layer and datalink layer virtual circuit protocols, where data always is delivered over the same path:

- X.25, where the VC is identified by a virtual channel identifier (VCI). X.25 provides reliable node-to-node communication and guaranteed QoS.
- Frame relay, where the VC is identified by a VCI. Frame relay is unreliable, but may provide guaranteed QoS.
- Asynchronous Transfer Mode (ATM), where the circuit is identified by a virtual path identifier (VPI) and virtual channel identifier (VCI) pair. ATM is unreliable, but may provide guaranteed QoS.

General Packet Radio Service (GPRS)

Multiprotocol label switching (MPLS), which can be used for IP over virtual circuits. Each circuit is identified by a label. MPLS is unreliable, but provides eight different QoS classes.

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Permanent and switched virtual circuits in ATM, frame relay, and X.25

Switched virtual circuits (SVCs) are generally set up on a per-call basis and are disconnected when the call is terminated; however, a permanent virtual circuit (PVC) can be established as an option to provide a dedicated circuit link between two facilities. PVC configuration is usually preconfigured by the service provider. Unlike SVCs, PVC are usually very seldom broken/disconnected.

A switched virtual circuit (SVC) is a virtual circuit that is dynamically established on demand and is torn down when transmission is complete, for example after a phone call or a file download. SVCs are used in situations where data transmission is sporadic and/or not always between the same data terminal equipment (DTE) endpoints.

A permanent virtual circuit (PVC) is a virtual circuit established for repeated/continuous use between the same DTE. In a PVC, the long-term association is identical to the data transfer phase of a virtual call. Permanent virtual circuits eliminate the need for repeated call set-up and clearing.

Frame relay is typically used to provide PVCs. ATM provides both switched virtual connections and permanent virtual connections, as they are called in ATM terminology. X.25 provides both SVCs and PVCs, although not all X.25 service providers or DTE implementations support PVCs as their use was much less common than SVCs.

2.12 TRANSMISSION IN ATM NETWORKS

Asynchronous Transfer Mode is a cell-based switching technique that uses asynchronous time division multiplexing. It encodes data into small fixed-sized cells (cell relay) and provides data link layer services that run over OSI Layer 1 physical links. This differs from other technologies based on packet-switched networks (such as the Internet Protocol or Ethernet), in which variable sized packets (known as frames when referencing Layer 2) are used. ATM exposes properties from both circuit switched and small packet switched networking, making it suitable for wide area data networking as well as real-time media transport. ATM uses a connection-oriented model and establishes a virtual circuit between two endpoints before the actual data exchange begins.

ATM is a core protocol used over the SONET/SDH backbone of the Integrated Services Digital Network.

The design of ATM aimed for a low-jitter network interface. However, to be able to provide short queueing delays, but also be able to carry large datagrams, it had to have cells. ATM broke up all packets, data, and voice streams into 48-byte chunks, adding a 5-byte routing header to each one so that they could be reassembled later. The choice of 48 bytes was political rather than technical. When the CCITT was standardizing ATM, parties from the United States wanted a 64-byte payload because this was felt to be a good compromise in larger payloads optimized for data transmission and shorter payloads optimized for real-time applications like voice; parties from Europe wanted 32-byte payloads because the small size (and therefore short transmission times) simplify voice applications with respect to echo cancellation. Most of the European parties eventually came around to the arguments made by the Americans, but France and a few others held out for a shorter cell length. With 32 bytes, France would have been able to implement an ATM-based voice network with

calls from one end of France to the other requiring no echo cancellation. 48 bytes (plus 5 header bytes = 53) was chosen as a compromise between the two sides. 5-byte headers were chosen because it was thought that 10% of the payload was the maximum price to pay for routing information. ATM multiplexed these 53-byte cells instead of packets. Doing so reduced the worst-case jitter due to cell contention by a factor of almost 30, minimizing the need for echo cancellers.

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ATM supports different types of services via ATM Adaptation Layers (AAL). Standardized AALs include AAL1, AAL2, and AAL5; and the rarely used AAL3 and AAL4. AAL1 is used for constant bit rate (CBR) services and circuit emulation. Synchronization is also maintained at AAL1. AAL2 through AAL4 are used for variable bit rate (VBR) services, and AAL5 for data. Which AAL is in use for a given cell is not encoded in the cell. Instead, it is negotiated by or configured at the endpoints on a per-virtual-connection basis.

Following the initial design of ATM, networks have become much faster. A 1500 byte (12000-bit) full-size Ethernet packet takes only 1.2 µs to transmit on a 10 Gbit/s optical network, reducing the need for small cells to reduce jitter due to contention. Some consider that this makes a case for replacing ATM with Ethernet in the network backbone. However, it should be noted that the increased link speeds by themselves do not alleviate jitter due to queuing. Additionally, the hardware for implementing the service adaptation for IP packets is expensive at very high speeds. Specifically, at speeds of OC-3 and above, the cost of segmentation and reassembly (SAR) hardware makes ATM less competitive for IP than Packet Over SONET (POS). SAR performance limits mean that the fastest IP router ATM interfaces are STM16 - STM64 which actually compares, while as of 2004 POS can operate at OC-192 (STM64) with higher speeds expected in the future.

On slower or congested links (622Mbit/s and below), ATM does not makes sense, and for this reason most ADSL systems use ATM as an intermediate layer between the physical link layer and a Layer 2 protocol like PPP or Ethernet.

At these lower speeds, ATM provides a useful ability to carry multiple logical circuits on a single physical or virtual medium, although other techniques exist, such as Multi-link PPP and Ethernet VLANs, which are optional in VDSL implementations. DSL can be used as an access method for an ATM network, allowing a DSL termination point in a telephone central office to connect to many internet service providers across a wide-area ATM network. In the United States, at least, this has allowed DSL providers to provide DSL access to the customers of many internet service providers. Since one DSL termination point can support multiple ISPs, the economic feasibility of DSL is substantially improved.

2.13 PAGING SYSTEMS

A pager (sometimes called a page, beeper, bleep[who?] or bleeper[who?]) is a simple personal telecommunications device for short messages. A one-way numeric pager can only receive a message consisting of a few digits, typically a phone number that the user is then expected to call. Alphanumeric pagers are available, as well as two-way pagers that have the ability to send and receive email, numeric pages, and SMS messages.

Until the popular adoption of mobile phones in the 1990s, pagers filled the role of

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common personal and mobile communications. Today, pagers mainly support the "critical messaging" markets. They are the ideal solution for very quick, very reliable personal or group messaging.

Unlike many other mobile communications networks, they continue to work in times of emergency or disaster as they do not suffer from network overload as has been proven many times. For this reason, they are still very popular with emergency service personnel, medical personnel, and information technology support staff.

Pagers are still in use today in places where mobile phones typically cannot reach users, and also in places where the operation of the radio transmitters contained in mobile phones is problematic or prohibited. One such type of location is a large hospital complex, where cellular coverage is often weak or nonexistent, where radio transmitters are suggested to interfere with sensitive medical equipment and where there is a greater need of assurance for a timely delivery of a message. The U.S. paging industry generated $2.1 billion in revenue in 2008, down from $6.2 billion in 2003.

Some common environments in which pagers are still used are:

- Pagers remain in use to notify emergency personnel. For example, they are required to be used by UK lifeboat crew and retained firefighters.
- Police, coast, local government emergency co-ordinators and other emergency services also carry pagers as a back-up system in the event of civil emergencies when mobile transmitters or networks may be unavailable.
- Security services use pagers (including global satellite pagers) as the signal is broadcast nationally (or across a global region in the case of satellite pagers) and there is thus no way of interceptors tracking the location of the pager-holder. Encrypted messages are also used in this scenario.
- Pagers are mostly carried by staff in medical establishments, allowing them to be summoned to emergencies. This is particularly important as one-way pagers do not interfere with medical equipment.
- Some construction and mining staff have to use one-way 'intrinsically safe' pagers as opposed to mobiles, as these do not risk triggering explosions in certain environments.
- Pagers are also widely used in the IT world, especially in cases where on-call technicians cannot rely on more modern cellular telephone systems. A good example would be in a cellular telephone company, where a service interruption in the cellular network would also mean that it would not be possible to notify a technician due to the outage in the network. Therefore, in these companies, engineers are usually equipped with a pager that uses another telco's mobile network to ensure reachability in case of emergency. Pagers are also frequently used by non-telco IT departments.
- Railway staff (for example those working for rail companies in the UK) use pagers because of their consistency of signal, to supplement mobile usage.
- In certain high-security government and corporate locations, a one-way pager is the only communication device allowed to be brought on-site.
- Deaf people who have no use for mobile voice services sometimes use two-way pagers.

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- Pagers are widely used by rare bird-chasing "twitchers", paying for rare bird information companies to send them messages telling them up-to-the-minute details of the latest rarity sightings across Britain.
- Additionally, some irrigation control systems and traffic signals are now controlled by messages sent via paging networks. Due to energy concerns in the United States and other countries, two way paging networks are being used for power company meter reading and control.
- Another pager technology in wide use today is the restaurant pager, usually from the beeper category. Mainly used in the hospitality industry, customers are given a theft-protected portable receiver which usually vibrates, flashes or beeps when a table becomes free, or when their meal is ready.

2.14 CORDLESS TELEPHONES

A *cordless telephone or portable telephone is a telephone* with a wireless handset that communicates via radio waves with a base station connected to a fixed telephone line, usually within a limited range of its base station (which has the handset cradle). The base station is on the subscriber premises, and attaches to the telephone network the same way a corded telephone does.

The base station on subscriber premises is what differentiates a cordless telephone from a mobile telephone. Current cordless telephone standards, such as PHS and DECT, have blurred the once clear-cut line between cordless and mobile telephones by implementing cell handover, various advanced features, such as data-transfer and even, on a limited scale, international roaming. In these models, base stations are maintained by a commercial mobile network operator and users subscribe to the service.

Unlike a corded telephone, a cordless telephone needs mains electricity to power the base station. The cordless handset is powered by a rechargeable battery, which is charged when the handset sits in its cradle.

Performance

Manufacturers usually advertise that higher frequency systems improve audio quality and range. Higher frequencies actually have worse propagation in the ideal case, as shown by the basic Friis transmission equation, and path loss tends to increase at higher frequencies as well. More important influences on quality and range are signal strength, antenna quality, the method of modulation used, and interference, which varies locally.

"Plain old telephone service" (POTS) landlines are designed to transfer audio with a quality that is just enough for the parties to understand each other. Typical bandwidth is 3.6 kHz; only a fraction of the frequencies that humans can hear, but enough to make the voice intelligible. No phone can improve on this quality, as it is a limitation of the phone system itself.

Higher-quality phones can transfer this signal to the handset with less interference over a greater range, however. Most cordless telephones, though, no matter what frequency band or transmission method is used, will hardly ever exactly match the sound quality of a high-quality wired telephone attached to a good telephone line.

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This constraint is caused by a number of issues, including the following:

- Sidetone: hearing one's own voice echoed in the receiver speaker
- A noticeable amount of constant background noise (This is not interference from outside sources, but noise within the cordless telephone system.)
- Frequency response not being the full frequency response available in a wired landline telephone
- Most manufacturers claim a range of about 30 m (100 ft) for their 2.4 GHz and 5.8 GHz systems, but inexpensive models often fall short of this claim.

However, the higher frequency often brings advantages. The 900 MHz and 2.4 GHz band are increasingly being used for a host of other devices, including baby monitor, microwave oven, Bluetooth, wireless LAN; thus, it is likely that a cordless phone will suffer interference from signals broadcast by those devices. It is also possible for a cordless phone to interfere with the 802.11a wireless standard, as the 802.11a standard can be configured to operate in the 5.8 GHz range. However, this can easily be fixed by configuring the device to work in the 5.180 GHz to 5.320 GHz band.

The recently allocated 1.9 GHz band is reserved for use by phones that use the DECT standard, which should avoid interference issues that are increasingly being seen in the unlicensed 900 MHz, 2.4 GHz, and 5.8 GHz bands.

Many cordless phones in the early 21st century are digital. Digital technology has helped provide clear sound and limit eavesdropping. Many cordless phones have one main base station and can add up to 3 or 4 additional bases. This allows for multiple voice paths that allow 3-way conference calls between the bases. This technology also allows multiple handsets to be used at the same time and up to 2 handsets can have an outside conversation.

2.15 CELLULAR TELEPHONES

Perhaps one of the most well known examples of wireless technology in action is the cellular telephone. These instruments use radio waves to enable the operator to make phone calls from many locations world-wide. They can be used anywhere that there is a cellular telephone site to house the equipment that is required to transmit and receive the signal that is used to transfer both voice and data to and from these instruments.

Wireless is a term used to describe telecommunications in which electromagnetic waves (rather than some form of wire) carry the signal over part or the entire communication path. Common examples of wireless equipment in use today include:

- Cellular phones and pagers: provide connectivity for portable and mobile applications, both personal and business.
- Global Positioning System (GPS): allows drivers of cars and trucks, captains of boats and ships, and pilots of aircraft to ascertain their location anywhere on earth.
- Cordless computer peripherals: the cordless mouse is a common example; keyboards and printers can also be linked to a computer via wireless.

- Cordless telephone sets: these are limited-range devices, not to be confused with cell phones.
- Satellite television: allows viewers in almost any location to select from hundreds of channels.

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Wireless networking is used to meet a variety of needs. Perhaps the most common use is to connect laptop users who travel from location to location. Another common use is for mobile networks that connect via satellite. A wireless transmission method is a logical choice to network a LAN segment that must frequently change locations.

The following situations justify the use of wireless technology:

- To span a distance beyond the capabilities of typical cabling,
- To avoid obstacles such as physical structures, EMI, or RFI,
- To provide a backup communications link in case of normal network failure,
- To link portable or temporary workstations,
- To overcome situations where normal cabling is difficult or financially impractical, or
- To remotely connect mobile users or networks.

2.16 COMMUNICATION SATELLITE

A communications satellite (sometimes abbreviated to COMSAT) is an artificial satellite stationed in space for the purpose of telecommunications. Modern communications satellites use a variety of orbits including geostationary orbits, Molniya orbits, other elliptical orbits and low (polar and non-polar) Earth orbits.

For fixed (point-to-point) services, communications satellites provide a microwave radio relay technology complementary to that of submarine communication cables. They are also used for mobile applications such as communications to ships, vehicles, planes and hand-held terminals, and for TV and radio broadcasting, for which application of other technologies, such as cable, is impractical or impossible.

A Low Earth Orbit (LEO) typically is a circular orbit about 400 kilometres above the earth's surface and, correspondingly, a period (time to revolve around the earth) of about 90 minutes. Because of their low altitude, these satellites are only visible from within a radius of roughly 1000 kilometres from the sub-satellite point. In addition, satellites in low earth orbit change their position relative to the ground position quickly. So even for local applications, a large number of satellites are needed if the mission requires uninterrupted connectivity.

Low earth orbiting satellites are less expensive to launch into orbit than geostationary satellites and, due to proximity to the ground, don't require as high signal strength (Recall that signal strength falls off as the square of the distance from the source, so the effect is dramatic). Thus there is a trade off between the number of satellites and their cost. In addition, there are important differences in the onboard and ground equipment needed to support the two types of missions.

A group of satellites working in concert is known as a satellite constellation. Two such constellations, intended to provide satellite phone services, primarily to remote

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areas, are the Iridium and Globalstar systems. The Iridium system has 66 satellites. Another LEO satellite constellation known as Teledesic, with backing from Microsoft entrepreneur Paul Allen, was to have over 840 satellites. This was later scaled back to 288 and ultimately ended up only launching one test satellite.

It is also possible to offer discontinuous coverage using a low Earth orbit satellite capable of storing data received while passing over one part of Earth and transmitting it later while passing over another part. This will be the case with the CASCADE system of Canada's CASSIOPE communications satellite. Another system using this store and forward method is Orbcomm.

Applications

Telephony

The first and historically most important application for communication satellites was in intercontinental long distance telephony. The fixed Public Switched Telephone Network relays telephone calls from land line telephones to an earth station, where they are then transmitted to a geostationary satellite. The downlink follows an analogous path. Improvements in submarine communications cables, through the use of fiber-optics, caused some decline in the use of satellites for fixed telephony in the late 20th century, but they still serve remote islands such as Ascension Island, Saint Helena, Diego Garcia, and Easter Island, where no submarine cables are in service.

There are also regions of some continents and countries where landline telecommunications are rare to nonexistent, for example large regions of South America, Africa, Canada, China, Russia, and Australia. Satellite communications also provide connection to the edges of Antarctica and Greenland.

Satellite phones connect directly to a constellation of either geostationary or low-earth-orbit satellites. Calls are then forwarded to a satellite teleport connected to the Public Switched Telephone Network or to another satellite phone system.

Satellite television

Television became the main market, its demand for simultaneous delivery of relatively few signals of large bandwidth to many receivers being a more precise match for the capabilities of geosynchronous comsats. Two satellite types are used for North American television and radio: Direct Broadcast Satellite (DBS), and Fixed Service Satellite (FSS)

The definitions of FSS and DBS satellites outside of North America, especially in Europe, are a bit more ambiguous. Most satellites used for direct-to-home television in Europe have the same high power output as DBS-class satellites in North America, but use the same linear polarization as FSS-class satellites. Examples of these are the Astra, Eutelsat, and Hotbird spacecraft in orbit over the European continent. Because of this, the terms FSS and DBS are more so used throughout the North American continent, and are uncommon in Europe.

Fixed Service Satellite

Fixed Service Satellites use the C band, and the lower portions of the Ku bands. They are normally used for broadcast feeds to and from television networks and local affiliate stations (such as program feeds for network and syndicated programming, live shots, and backhauls), as well as being used for distance learning by schools and

universities, business television (BTV), Videoconferencing, and general commercial telecommunications. FSS satellites are also used to distribute national cable channels to cable television headends.

Free-to-air satellite TV channels are also usually distributed on FSS satellites in the Ku band. The Intelsat Americas 5, Galaxy 10R and AMC 3 satellites over North America provide a quite large amount of FTA channels on their Ku band transponders.

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The American Dish Network DBS service has also recently utilized FSS technology as well for their programming packages requiring their SuperDish antenna, due to Dish Network needing more capacity to carry local television stations per the FCC's "must-carry" regulations, and for more bandwidth to carry HDTV channels.

Direct broadcast satellite

A direct broadcast satellite is a communications satellite that transmits to small DBS satellite dishes (usually 18 to 24 inches or 45 to 60 cm in diameter). Direct broadcast satellites generally operate in the upper portion of the microwave Ku band. DBS technology is used for DTH-oriented (Direct-To-Home) satellite TV services, such as DirecTV and DISH Network in the United States, Bell TV and Shaw Direct in Canada, Freesat in the UK and Sky Digital in the UK, the Republic of Ireland, and New Zealand.

Operating at lower frequency and lower power than DBS, FSS satellites require a much larger dish for reception (3 to 8 feet (1 to 2.5m) in diameter for Ku band, and 12 feet (3.6m) or larger for C band). They use linear polarization for each of the transponders' RF input and output (as opposed to circular polarization used by DBS satellites), but this is a minor technical difference that users don't notice.

FSS satellite technology was also originally used for DTH satellite TV from the late 1970s to the early 1990s in the United States in the form of TVRO (TeleVision Receive Only) receivers and dishes. It was also used in its Ku band form for the now-defunct Primestar satellite TV service.

Satellites for communication have now[when?] been launched that have transponders in the Ka band, such as DirecTV's SPACEWAY-1 satellite, and Anik F2. NASA as well has launched experimental satellites using the Ka band recently.

Mobile satellite technologies

Initially available for broadcast to stationary TV receivers, by 2004 popular mobile direct broadcast applications made their appearance with that arrival of two satellite radio systems in the United States: Sirius and XM Satellite Radio Holdings. Some manufacturers have also introduced special antennas for mobile reception of DBS television.

Using GPS technology as a reference, these antennas automatically re-aim to the satellite no matter where or how the vehicle (that the antenna is mounted on) is situated. These mobile satellite antennas are popular with some recreational vehicle owners. Such mobile DBS antennas are also used by JetBlue Airways for DirecTV (supplied by LiveTV, a subsidiary of JetBlue), which passengers can view on-board on LCD screens mounted in the seats.

Satellite radio

Satellite radio offers audio services in some countries, notably the United States.

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Mobile services allow listeners to roam a continent, listening to the same audio programming anywhere.

A satellite radio or subscription radio (SR) is a digital radio signal that is broadcast by a communications satellite, which covers a much wider geographical range than terrestrial radio signals.

Satellite radio offers a meaningful alternative to ground-based radio services in some countries, notably the United States. Mobile services, such as Sirius, XM, and Worldspace, allow listeners to roam across an entire continent, listening to the same audio programming anywhere they go.

Other services, such as Music Choice or Muzak's satellite-delivered content, require a fixed-location receiver and a dish antenna. In all cases, the antenna must have a clear view to the satellites. In areas where tall buildings, bridges, or even parking garages obscure the signal, repeaters can be placed to make the signal available to listeners.

Radio services are usually provided by commercial ventures and are subscription-based. The various services are proprietary signals, requiring specialized hardware for decoding and playback. Providers usually carry a variety of news, weather, sports, and music channels, with the music channels generally being commercial-free.

In areas with a relatively high population density, it is easier and less expensive to reach the bulk of the population with terrestrial broadcasts. Thus in the UK and some other countries, the contemporary evolution of radio services is focused on Digital Audio Broadcasting (DAB) services or HD Radio, rather than satellite radio.

Amateur radio

Amateur radio operators have access to the OSCAR satellites that have been designed specifically to carry amateur radio traffic. Most such satellites operate as spaceborne repeaters, and are generally accessed by amateurs equipped with UHF or VHF radio equipment and highly directional antennas such as Yagis or dish antennas.

Due to launch costs, most current amateur satellites are launched into fairly low Earth orbits, and are designed to deal with only a limited number of brief contacts at any given time. Some satellites also provide data-forwarding services using the AX.25 or similar protocols.

Satellite Internet

After the 1990s, satellite communication technology has been used as a means to connect to the Internet via broadband data connections. This can be very useful for users who are located in very remote areas, and cannot access a broadband connection.

Military uses

Communications satellites are used for military communications applications, such as Global Command and Control Systems. Examples of military systems that use communication satellites are the MILSTAR, the DSCS, and the FLTSATCOM of the United States, NATO satellites, United Kingdom satellites, and satellites of the former Soviet Union. Many military satellites operate in the X-band, and some also use UHF radio links, while MILSTAR also utilizes Ka band.

Navigation

One of the fascinating applications of satellites is GPS (Global Positioning System). Its primary application is navigation. There is a network composed of 24 to 32 satellites in Medium Earth Orbit spaced equally around the world in overlapping pattern for this purpose.

They use low microwave frequencies such as 1.57542Ghz and 1.2276 GHz for transmission. Receivers on the earth pick up transmissions from four satellites simultaneously. The receiver uses the microprocessor to compute and display the exact position, in terms of latitude and longitude.

NOTES

SUMMARY

1. A transmission line is the material medium or structure that forms all or part of a path from one place to another for directing the transmission of energy, such as electromagnetic waves or acoustic waves, as well as electric power transmission.
2. In many electric circuits, the length of the wires connecting the components can for the most part be ignored.
3. High-frequency transmission lines can be defined as transmission lines that are designed to carry electromagnetic waves whose wavelengths are shorter than or comparable to the length of the line.
4. The Telegrapher's Equations (or just Telegraph Equations) are a pair of linear differential equations which describe the voltage and current on an electrical transmission line with distance and time.
5. Cable is the medium through which information usually moves from one network device to another.
6. The standard connector for unshielded twisted pair cabling is an RJ-45 connector.
7. Baseband is an adjective that describes signals and systems whose range of frequencies is measured from 0 to a maximum bandwidth or highest signal frequency; it is sometimes used as a noun for a band of frequencies starting at 0.
8. Coaxial cabling has a single copper conductor at its center. A plastic layer provides insulation between the center conductor and a braided metal shield.
9. Fiber optic cabling consists of a center glass core surrounded by several layers of protective materials.
10. Wireless communication is the transfer of information over a distance without the use of enhanced electrical conductors or "wires".
11. Radio waves are easy to penerate, can travel long distances, and penetrate buildings easily, so they are widely used for communication, both indoor and outdoors.
12. Microwave transmission refers to the technology of transmitting information by the use of the radio waves whose wavelengths are conveniently measured in small numbers of centimeters, by using various electronic technologies.
13. A microwave link is a communications system that uses a beam of radio waves in the microwave frequency range to transmit video, audio, or data between two locations, which can be from just a few feet or meters to several miles or kilometers apart.
14. The remote controls used on televisions, VCRs, and stereos all use infrared communication.
15. Integrated Services Digital Network (ISDN) is a set of communications standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network.
16. In ISDN, there are two types of channels, B (for "bearer") and D (for "delta"). B channels are used for data (which may include voice), and D channels are intended for signaling and control (but can also be used for data).

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17. In telecommunications and computer networks, a virtual circuit (VC), synonymous with virtual connection and virtual channel, is a connection oriented communication service that is delivered by means of packet mode communication.
18. Asynchronous Transfer Mode is a cell-based switching technique that uses asynchronous time division multiplexing.
19. Pagers are still in use today in places where mobile phones typically cannot reach users, and also in places where the operation of the radio transmitters contained in mobile phones is problematic or prohibited.
20. A cordless telephone or portable telephone is a telephone with a wireless handset that communicates via radio waves with a base station connected to a fixed telephone line, usually within a limited range of its base station (which has the handset cradle).
21. Perhaps one of the most well known examples of wireless technology in action is the cellular telephone.
22. A communications satellite is an artificial satellite stationed in space for the purpose of telecommunications.

SELF ASSESSMENT QUESTIONS

1. Describe Unshielded Twisted Pair.
2. What is Unshielded Twisted Pair Connector?
3. What is Coaxial Cable?
4. Describe Fiber optics.
5. What do you understand by Wireless Transmission?
6. Describe Radio Transmission.
7. Describe Microwave Transmission.
8. What do you understand by Infrared Transmission?
9. Describe Light Wave Transmission.
10. What is ISDN?
11. What are Virtual Circuits?
12. Describe the method of transmission in ATM networks.
13. What are paging systems?
14. Describe the working of Cordless Telephones.
15. What are the features of Communication Satellite?

Short Questions with Answers

1. What is the transmission line?

Ans. A transmission line is the material medium or structure that forms all or part of a path from one place to another for directing the transmission of energy, such as electromagnetic waves or acoustic waves, as well as electric power transmission. Components of transmission lines include wires, coaxial cables, dielectric slabs, optical fibers, electric power lines, and waveguides.

2. What are twisted cables?

Ans. Cable is the medium through which information usually moves from one network device to another. There are several types of cable which are commonly used with LANs. In some cases, a network will utilize only one type of cable, other networks will use a variety of cable types. The type of cable chosen for a network is related to the network's topology, protocol, and size. Understanding the characteristics of different types of cable and how they relate to other aspects of a network is necessary for the development of a successful network.

3. What is baseband transmission?

Ans. Baseband is an adjective that describes signals and systems whose range of frequencies is

measured from 0 to a maximum bandwidth or highest signal frequency; it is sometimes used as a noun for a band of frequencies starting at 0. It can often be considered as synonym to lowpass, and antonym to passband.

4. What are the fibre optic cables?

Ans. Fiber optic cabling consists of a center glass core surrounded by several layers of protective materials. It transmits light rather than electronic signals eliminating the problem of electrical interference. This makes it ideal for certain environments that contain a large amount of electrical interference. It has also made it the standard for connecting networks between buildings, due to its immunity to the effects of moisture and lighting.

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5. *What is wireless transmission?*

Ans. Wireless communication is the transfer of information over a distance without the use of enhanced electrical conductors or "wires". The distances involved may be short (a few meters as in television remote control) or long (thousands or millions of kilometers for radio communications). When the context is clear, the term is often shortened to "wireless". Wireless communication is generally considered to be a branch of telecommunications.

6. Which are Microwave transmissions?

Ans. Microwave transmission refers to the technology of transmitting information by the use of the radio waves whose wavelengths are conveniently measured in small numbers of centimeters, by using various electronic technologies. These are called microwaves. This part of the radio spectrum ranges across frequencies of roughly 1.0 gigahertz (GHz) to 30 GHz. Also by using the formula ? = c/f , these correspond to wavelengths from 30 centimeters down to 1.0 cm. [In the above equation, the Greek letter ? (lambda) is the wavelength in meters; c is the speed of light in meters per second; and f is the frequency in hertz (Hz).]

7. What is ISDN?

Ans. Integrated Services Digital Network (ISDN) is a set of communications standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network. It was first defined in 1988 in the CCITT red book. Prior to ISDN, the phone system was viewed as a way to transport voice, with some special services available for data. The key feature of ISDN is that it integrates speech and data on the same lines, adding features that were not available in the classic telephone system. There are several kinds of access interfaces to ISDN defined as Basic Rate Interface (BRI), Primary Rate Interface (PRI) and Broadband ISDN (B-ISDN).

8. What is Virtual Circuit?

Ans. In telecommunications and computer networks, a virtual circuit (VC), synonymous with *virtual connection* and *virtual channel*, *is a connection oriented communication service* that is delivered by means of packet mode communication. After a connection or virtual circuit is established between two nodes or application processes, a bit stream or byte stream may be delivered between the nodes; a virtual circuit protocol allows higher level protocols to avoid dealing with the division of data into segments, packets, or frames.

9. What is ATM transmission?

Ans. Asynchronous Transfer Mode is a cell-based switching technique that uses asynchronous time division multiplexing. It encodes data into small fixed-sized cells (cell relay) and provides data link layer services that run over OSI Layer 1 physical links. This differs from other technologies based on packet-switched networks (such as the Internet Protocol or Ethernet), in which variable sized packets (known as frames when referencing Layer 2) are used. ATM exposes properties from both circuit switched and small packet switched networking, making it suitable for wide area data networking as well as real-time media transport. ATM uses a connection-oriented model and establishes a virtual circuit between two endpoints before the actual data exchange begins.

10. What is a cordless telephone?

Ans. A cordless telephone or portable telephone is a telephone with a wireless handset that communicates via radio waves with a base station connected to a fixed telephone line, usually within a limited range of its base station (which has the handset cradle). The base station is on the subscriber premises, and attaches to the telephone network the same way a corded telephone does.

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Ans. Perhaps one of the most well known examples of wireless technology in action is the cellular telephone. These instruments use radio waves to enable the operator to make phone calls from many locations world-wide. They can be used anywhere that there is a cellular telephone site to house the equipment that is required to transmit and receive the signal that is used to transfer both voice and data to and from these instruments.

Wireless is a term used to describe telecommunications in which electromagnetic waves (rather than some form of wire) carry the signal over part or the entire communication path.

12. What is a communication satellite?

Ans. A communications satellite (sometimes abbreviated to COMSAT) is an artificial satellite stationed in space for the purpose of telecommunications. Modern communications satellites use a variety of orbits including geostationary orbits, Molniya orbits, other elliptical orbits and low (polar and non-polar) Earth orbits.

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Further Readings

1. Elements of Data Communication and Networks: S. A. Amutha Jeevakumari, University Science Press.
2. Wide Area Networks: Navneet Sharma, Firewal Media.
3. Data Communication System: Monika Khuaran, Firewal Media.
4. Computer Network: Bharat Bhushan Agarwal and Sumit Prakash Tayal, University Science Press.
5. Computer Network: Rachna Sharma, University Science Press.
6. Computer Network: Sumit Raj Chauhan and Er. Punit Soni, University Science Press.

UNIT 3

THE DATA LINK LAYER

NOTES

STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- explain about the Data Link Layer, which is the Layer 2 of the seven-layer OSI model of computer networking.
- learn about Framing, which is why we split the bit stream into frames.
- know about Error controls which we take to make sure that there is no error in transmission.
- describe about Flow-controls which we take to make sure the data flows smoothly.
- know about the Error detection and correction in the transmission.
- explain about the various Protocols which are used.
- describe about Simplex Stop, Wait Protocols and One bit sliding window protocols.
- know about Go-Back n, an Automatic Request Protocol.
- explain about the Example; The Data Link in the Internet.

NOTES

3.1 THE DATA LINK LAYER

The Data Link Layer is Layer 2 of the seven-layer OSI model of computer networking. It corresponds to, or is part of the link layer of the TCP/IP reference model.

The Data Link Layer is the protocol layer which transfers data between adjacent network nodes in a wide area network or between nodes on the same local area network segment. The Data Link Layer provides the functional and procedural means to transfer data between network entities and might provide the means to detect and possibly correct errors that may occur in the Physical Layer. Examples of data link protocols are Ethernet for local area networks (multi-node), the Point-to-Point Protocol (PPP), HDLC and ADCCP for point-to-point (dual-node) connections.

The Data Link Layer is concerned with local delivery of frames between devices on the same LAN. Data Link frames, as these protocol data units are called, do not cross the boundaries of a local network. Inter-network routing and global addressing are higher layer functions, allowing Data Link protocols to focus on local delivery, addressing, and media arbitration. In this way, the Data Link layer is analogous to a neighborhood traffic cop; it endeavors to arbitrate between parties contending for access to a medium.

When devices attempt to use a medium simultaneously, frame collisions occur. Data Link protocols specify how devices detect and recover from such collisions, but it does not prevent them from happening.

Delivery of frames by layer 2 devices is affected through the use of unambiguous hardware addresses. A frame's header contains source and destination addresses that indicate which device originated the frame and which device is expected to receive and process it. In contrast to the hierarchical and routable addresses of the network layer, layer 2 addresses are flat, meaning that no part of the address can be used to identify the logical or physical group to which the address belongs.

The data link thus provides data transfer across the physical link. That transfer can be reliable or unreliable; many data link protocols do not have acknowledgments of successful frame reception and acceptance, and some data link protocols might not even have any form of checksum to check for transmission errors. In those cases, higher-level protocols must provide flow control, error checking, and acknowledgments and retransmission.

In some networks, such as IEEE 802 local area networks, the Data Link Layer is described in more detail with Media Access Control (MAC) and Logical Link Control (LLC) sublayers; this means that the IEEE 802.2 LLC protocol can be used with all of the IEEE 802 MAC layers, such as Ethernet, token ring, IEEE 802.11, etc., as well as with some non-802 MAC layers such as FDDI. Other Data Link Layer protocols, such as HDLC, are specified to include both sublayers, although some other protocols, such as Cisco HDLC, use HDLC's low-level framing as a MAC layer in combination with a different LLC layer. In the ITU-T G.hn standard, which provides a way to create a high-speed (up to 1 Gigabit/s) Local area network using existing home wiring (power lines, phone lines and coaxial cables), the Data Link Layer is divided into three sub-layers (Application Protocol Convergence, Logical Link Control and Medium Access Control).

Within the semantics of the OSI network architecture, the Data Link Layer protocols

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respond to service requests from the Network Layer and they perform their function by issuing service requests to the Physical Layer.

3.2 FRAMING

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The usual approach for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame. When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it, e.g., discarding the bad frame and sending an error report. Breaking the bit stream up into frames is more difficult than it at first appears. One way to achieve this framing is to insert time gaps between frames, much like the space between words in ordinary text.

However, networks rarely make any guarantees about timing, so it is possible these gaps might be squeezed out, or other gaps might be inserted during transmission. Since it is too risky to count on timing to mark the start and end of each frame, other methods have been devised.

Some of the methods, which are commonly used are:

1. Character count.
2. Starting and ending characters, with character stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

A major disadvantage of using the framing method is that it is closely tied to 8-bit characters in general and the ASCII character code in particular. As networks deeloped, the disadvantages of embedding the character code in the framing mechanism became more and more obvious so a new technique had to be developed to allow arbitary sized characters. The new technique allows data frames to contain an arbitary number of bits and allows character codes with an arbitary number of bits per character.

The last method of framing is only applicable to networks in which the encoding on the physical medium contains some rdundancy. For example, encode 1 bit of data by using 2 physical bits. Normally, a 1 bit is a high-low pair and 0 bit is a low-high pair. The combination high-high and low-low are not used for data. The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. This use of invalid physical codes is part of the 802 LAN standard.

As a final note on framing, many data link protocols use a combination of a characer count with one of the other methods for extra safety. When a frame arrives, the count field is used to locate the end of the frame. Only, if the appropriate swlimirwe is pesent at that position and the checksum is correct, is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter.

3.3 ERROR CONTROL

The usual way to ensure reliable delivery of the frames is to provide the sender with

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some feedback about what is happening at the other end of the line. Typically the protocol calls for the receiver to send back special control frames bearing positive or negative acknowledgements about the incoming frames. If the sender receives a positive acknowledgement about a frame, it knows the frame has arrived safely. On the other hand, a negative acknowledgement means that something has gone wrong, and the frame must be transmitted again.

An additional complication comes from the possibility that hardware troubles may cause a frame to vanish completely, e.g., in a noise burst. In this case, the receiver will not react at all, since it has no reason to react. It should be clear that a protocol in which the sender transmitted a frame and then waited for an achcknowledgement, positive or negative, would hand forever if a frame were ever completely lost due to malfunctioning hardware.

This possibility is dealth with by introducing timers into the data link layer. When the sender transmits a frame, it generally also starts a timer. The timer is set to go off after an interval long enough for the frame to reach the destination, to be processed there, and have the acknowledgement propagate back to the sender. Normally, the frame wil be correctly received and the acknowledgement will get back before the time runs out, in which case it will be canceled.

3.4 FLOW-CONTROL

Take for example, the sender keeps pumping the frames out at a high rate until the receiver is completely swamped. Even if the transmission is error free, at a certain point the receiver will simply not be able to handle the frames as they arrive and will start to lose some. Clearly, somehting has to be done to prevent this situation. The usual solution is to introduce flow control to throttle the sener into sending no faster than the receiver can handle the traffic.

This throttling generally requires some kind of a feedback mechanism, so the sender can be made aware of whether or not the receiver is able to keep up. Various flow control schemes are known, but most of them use the same basic principle. The protocol contains well-defined rules about when a sender may transmit the next frame. These rules often prohibit frames from being sent until the receiver has granted permission, either implicity or explicity.

3.5 ERROR DETECTION AND CORRECTION PROTOCOLS

As you know, the telephone system has three parts: the switches, the interoffice trunks, and the local loops. The first two are now almost entirely digital in the United States and some other countries. The local loops are still analog twisted copper pairs everywhere and will continue to be so for decades due to the enormous expense of replacing them. While errors are rare on the digital part, they are still common on the local loops.

Furthermore, wireless communication is becoming more common, and the error rates are order of magnitude worse than on the interoffice fiber trunks. As a result of the physical processes that generate them, errors on some media, e.g., radio, tend to

come in bursts rather than singly. Having the errors come in bursts has bothe advantages and isadvantages over isolated single-bit errors. On the advantage side, computer data are always sent in blocks of bits. Suppose that the block size is 1000 bits, and the error rate is 0.001 per bit.

If errors were independent, most blcoks would contain an error. If the errors came in bursts of 100 however, only one or two blocks in 100 would be affected, on the average. The disadvantage of burst errors is that they are much harder to detect and correct than are isolated errors.

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3.6 SIMPLEX STOP AND WAIT PROTOCOLS

Now we come to the most unrealistic restriction used in protocol 1;the ability of the receiving network layer to process incoming data infinitely fast (or equivalently, the presence in the receiving data link layer of an infinite amount of buffer space in which to store all incoming frames while they are waiting respective turns). The communication channel is still assumed to be error free however, and the data traffic is still simplex.

The main problem we have to deal with here is how to prevent the sender from flooding the receiver with data faster than the latter is able to process it. In essence, if the receiver requires a time x to execute from_physical_layer plus to_network_layer, the sender must trnsmit at an average rate less than one fram per time x. Moreover, if we assume that there no automatic buffering and queuing doen within the receiver's hardware, the sender must never transmit a new frame until the old one has been fetched by from_physical_layer, lest the new one overwrite the old one.

A more general solution to this dilemma is to have the receiver provide feedback to the sender. After having pased a packet up its network layer, the receiver sends a little dummy frame back to the sender which, in effect, gives the sender permission to transmit the next frame. After having sent a frame, the sender is required by the protocol to bide its time until the little dummy, i.e., acknwledgement, frame arrives. Protocols in which the sender sends one frame and then waits for an acknowledgement before proceeding are caled stop-and-wait.

3.7 ONE BIT SLIDING WINDOW

Sliding Window Protocols are a feature of packet-based data transmission protocols. They are used where reliable in-order delivery of packets is required, such as in the data link layer (OSI model) as well as in TCP (transport layer of the OSI model).

Conceptually, each portion of the transmission (packets in most data link layers, but bytes in TCP) is assigned a unique consecutive sequence number, and the receiver uses the numbers to place received packets in the correct order, discarding duplicate packets and identifying missing ones. The problem with this is that there is no limit of the size of the sequence numbers that can be required.

By placing limits on the number of packets that can be transmitted or received at any given time, a sliding window protocol allows an unlimited number of packets to be communicated using fixed-size sequence numbers.

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For the highest possible throughput, it is important that the transmitter is not forced to stop sending by the sliding window protocol earlier than one round-trip delay time (RTT).

The limit on the amount of data that it can send before stopping to wait for an acknowledgment should be larger than the bandwidth-delay product of the communications link. If it is not, the protocol will limit the effective bandwidth of the link.

Examples

The simplest sliding window: Stop-and-wait

Although commonly distinguished from the sliding-window protocol, the stop-and-wait ARQ protocol is actually the simplest possible implementation of it. The transmit window is 1 packet, and the receive window is 1 packet. Thus, N=1+1=2 possible sequence numbers (conveniently represented by a single bit) are required.

Ambiguity example

The transmitter alternately sends packets marked "odd" and "even". The acknowledgments likewise say "odd" and "even". Suppose that the transmitter, having sent an odd packet, did not wait for an odd acknowledgment, and instead immediately sent the following even packet. It might then receive an acknowledgment saying "expecting an odd packet next". This would leave the transmitter in a quandary: has the receiver received both of the packets, or neither?

Go-Back-N

Go-Back-N ARQ is the sliding window protocol with wt>1, but a fixed wr=1. The receiver refuses to accept any packet but the next one in sequence. If a packet is lost in transit, following packets are ignored until the missing packet is retransmitted, a minimum loss of one round trip time. For this reason, it is inefficient on links that suffer frequent packet loss.

Ambiguity example

Suppose that we are using a 3-bit sequence number, such as is typical for HDLC. This gives $N=2^3=8$. Since wr=1, we must limit wt=7. This is because, after transmitting 7 packets, there are 8 possible results: Anywhere from 0 to 7 packets could have been received successfully. This is 8 possibilities, and the transmitter needs enough information in the acknowledgment to distinguish them all.

If the transmitter sent 8 packets without waiting for acknowledgment, it could find itself in a quandary similar to the stop-and-wait case: does the acknowledgment mean that all 8 packets were received successfully, or none of them?

Selective Repeat

The most general case of the sliding window protocol is Selective Repeat ARQ. This requires a much more capable receiver, which can accept packets with sequence numbers higher than the current nr and store them until the gap is filled in.

The advantage, however, is that it is not necessary to discard following correct data for one round-trip time before the transmitter can be informed that a retransmission is required. This is therefore preferred for links with low reliability and/or a high bandwidth-delay product.

The window size wr need only be larger than the number of consecutive lost packets that can be tolerated. Thus, small values are popular; wr=2 is common.

3.8 USING GO-BACK N

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Go-Back-N ARQ is a specific instance of the Automatic Repeat-reQuest (ARQ) Protocol, in which the sending process continues to send a number of frames specified by a window size even without receiving an ACK packet from the receiver. It is a special case of the general sliding window protocol with the transmit window size of N and receive window size of 1.

The receiver process keeps track of the sequence number of the next frame it expects to receive, and sends that number with every ACK it sends. The receiver will ignore any frame that does not have the exact sequence number it expects – whether that frame is a "past" duplicate of a frame it has already ACK'ed or whether that frame is a "future" frame past the last packet it is waiting for.

Once the sender has sent all of the frames in its window, it will detect that all of the frames since the first lost frame are outstanding, and will go back to sequence number of the last ACK it received from the receiver process and fill its window starting with that frame and continue the process over again.

Go-Back-N ARQ is a more efficient use of a connection than Stop-and-wait ARQ, since unlike waiting for an acknowledgement for each packet, the connection is still being utilized as packets are being sent. In other words, during the time that would otherwise be spent waiting, more packets are being sent.

However, this method also results in sending frames multiple times – if any frame was lost or damaged, or the ACK acknowledging them was lost or damaged, then that frame and all following frames in the window (even if they were received without error) will be re-sent. To avoid this, Selective Repeat ARQ can be used.

3.9 THE DATA LINK LAYER IN THE INTERNET

The Internet consits of individual machines, hosts and routers, and the communication infrastructure that connects them. Within a single building, LANs are widely used for interconnection, but most of the wide area infrastructure is built up from point-to-point leased lines.

In practice, point-to-point communication is primarily used in two situations. First, thousands of organizations have one or more LANs, each with some number of hosts (or a bridge, which is functionally similar). Often, the routers are interconnected by a backbone LAN.

Typically, all connections to the outside would go through one or two routers that have point-to-point leased lines to distant routers. It is these routers and their leased lines that make up the communication subnets on which the Internet is built.

The second situation where point-to-point lines play a major role in the Internet is the millions of individuals who have home connections to the Internet using modems and dial-up telephone lines. Usually, what happens is that the user's home PC calls up an Internet provider, whichincludes commercial companies like America Online,

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CompuServe, and the Microsoft Network, but also many universities and companies that provide home Internet connectivity to their students and employees.

Sometimes the home PC just functions as a character-oriented terminal logged into the Internet service provider's timesharing system. In this mode, the user can type commands and run programs, but the graphical Internet services, such as World Wide Web, are not available. This way of working is called having a shell account.

SUMMARY

1. The Data Link Layer is the protocol layer which transfers data between adjacent network nodes in a wide area network or between nodes on the same local area network segment.
2. Within the semantics of the OSI network architecture, the Data Link Layer protocols respond to service requests from the Network Layer and they perform their function by issuing service requests to the Physical Layer.
3. The usual approach for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame.
4. The usual way to ensure reliable delivery of the frames is to provide the sender with some feedback about what is happening at the other end of the line.
5. While errors are rare on the digital part, they are still common on the local loops.
6. The main problem we have to deal with here is how to prevent the sender from flooding the receiver with data faster than the latter is able to process it.
7. Go-Back-N ARQ is a specific instance of the Automatic Repeat-reQuest (ARQ) Protocol, in which the sending process continues to send a number of frames specified by a window size even without receiving an ACK packet from the receiver.
8. The Internet consits of individual machines, hosts and routers, and the communication infrastructure that connects them.

SELF ASSESSMENT QUESTIONS

1. Describe The Data Link Layer.
2. What is Framing?
3. What is Error control?
4. Describe Flow control.
5. Describe Error detection and correction protocol.
6. What is Simplex Stop and Wait Protocols?
7. Describe One bit sliding window.
8. What is Go-Back n?
9. Describe Data Link Layer in the Internet.

Short Questions with Answers

1. What is the Data Link Layer?

Ans. The Data Link Layer is Layer 2 of the seven-layer OSI model of computer networking. It corresponds to, or is part of the link layer of the TCP/IP reference model. The Data Link Layer is the protocol layer which transfers data between adjacent network nodes in a wide area network or between nodes on the same local area network segment.

The Data Link Layer provides the functional and procedural means to transfer data

between network entities and might provide the means to detect and possibly correct errors that may occur in the Physical Layer. Examples of data link protocols are Ethernet for local area networks (multi-node), the Point-to-Point Protocol (PPP), HDLC and ADCCP for point-to-point (dual-node) connections.

2. What is framing?

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Ans. The usual approach for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame. When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it, e.g., discarding the bad frame and sending an error report.

Breaking the bit stream up into frames is more difficult than it at first appears. One way to achieve this framing is to insert time gaps between frames, much like the space between words in ordinary text.

3. What is Sliding Window Protocol?

Ans. Sliding Window Protocols are a feature of packet-based data transmission protocols. They are used where reliable in-order delivery of packets is required, such as in the data link layer (OSI model) as well as in TCP (transport layer of the OSI model).

Conceptually, each portion of the transmission (packets in most data link layers, but bytes in TCP) is assigned a unique consecutive sequence number, and the receiver uses the numbers to place received packets in the correct order, discarding duplicate packets and identifying missing ones. The problem with this is that there is no limit of the size of the sequence numbers that can be required.

4. What is Go-Back-N?

Ans. Go-Back-N ARQ is a specific instance of the Automatic Repeat-reQuest (ARQ) Protocol, in which the sending process continues to send a number of frames specified by a window size even without receiving an ACK packet from the receiver. It is a special case of the general sliding window protocol with the transmit window size of N and receive window size of 1.

The receiver process keeps track of the sequence number of the next frame it expects to receive, and sends that number with every ACK it sends. The receiver will ignore any frame that does not have the exact sequence number it expects – whether that frame is a "past" duplicate of a frame it has already ACK'ed or whether that frame is a "future" frame past the last packet it is waiting for.

5. What is wireless transmission?

Ans. Wireless communication is the transfer of information over a distance without the use of enhanced electrical conductors or "wires". The distances involved may be short (a few meters as in television remote control) or long (thousands or millions of kilometers for radio communications). When the context is clear, the term is often shortened to "wireless". Wireless communication is generally considered to be a branch of telecommunications.

6. What is the role of Internet in communication?

Ans. The Internet consits of individual machines, hosts and routers, and the communication infrastructure that connects them. Within a single building, LANs are widely used for interconnection, but most of the wide area infrastructure is built up from point-to-point leased lines.

In practice, point-to-point communication is primarily used in two situations. First, thousands of organizations have one or more LANs, each with some number of hosts (or a bridge, which is functionally similar). Often, the routers are interconnected by a backbone LAN. Typically, all connections to the outside would go through one or two routers that have point-to-point leased lines to distant routers. It is these routers and their leased lines that make up the communication subnets on which the Internet is built.

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Further Readings

1. Computer Networks: Ajit Kumar Singh, Firewal Media.
2. TCP / IP and Distributed System: Vivek Archarya, Firewal Media.
3. Elements of Data Communication and Networks: S. A. Amutha Jeevakumari, University Science Press.
4. Data Communication System: Monika Khuaran, Firewal Media.
5. Computer Network: Rachna Sharma, University Science Press.

UNIT 4

THE MEDIUM ACCESS SUB-LAYER

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STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- understand about the Medium Access Sub Layer used in networking.
- know about Framing Static and Dynamic Channel Allocation in LANs and MANs.
- learn about the various IEEE standard 802.3 and Ethernet.
- understand about IEEE standard 802.4 and Token Bus.
- know about IEEE 802.4 and token Ring; Bridges.
- describe about Bridges from 802 x to 802 y.
- understand about the various Transparent Bridges.
- know about the Source Routing Bridges.

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4.1 THE MEDIUM ACCESS SUB LAYER

In any broadcast network, the key issue is how to determine who gets to use the channel when there is competition for it. To make this point clearer, consider a conference call in which six people, on six different telephones, are all connected together so that each one can hear and talk to all the others. It is very likely that when one of the stops speaking, two or more will start talking at once, leadint to chaos. In a face-to-face meeting, chaos is avoided by external means, for example, at a meeting, people raiseheir hands to request permission to speak. When only a single channel is available, determining who should go next is much harder. The protocols used to determine who goes next on a multiaccesschannel belong to a sublayer of the data link layer called Medium Access Control Sub Layer.

4.2 FRAMING STATIC AND DYNAMIC CHANNEL ALLOCATION IN LANS AND MANS

Let us first formulate the allocation problem of channel allocation. Underlying all the work done in this area are five key assumptions, described below:

1. **Station model.** The model consists of N independent stations (computers, telephones, personal communications, etc.), each with a program or user that generates frames for transmission. The probality of a frame being generated in an interval of of length Δt is $\lambda\Delta t$, where λ is a constant (the arrival rate of new frames). Once a frame has been generated, the station is blocked and does nothing until the frame has been successfully transmitted.
2. **Single Channel Assumption.** A single channel is available for all communication. All stations can transmit on it and all can receive from it. As far as the hardware is concerned, all stations are equivalent, although protocol software may assign priorities to them.
3. **Collision Assumption.** If two frames are trnsmitted simultaneously, they overlap in time and the resulting signal is garbled. This even is called a collsion. All stations can detect collisions. A collided frame must be trnsmitted again later. There are no errors other than those generated by collisions.
4. **Continuous Time.** Frame transmission can begin at any instant. There is no master clock dividing time into discrete intervals.
5. **Slotted Time.** Time is divided into discrete intervals (slots). Frame transissions always begin at the start of a slot. A slot may contain 0, 1, or more frames, corresponding to an idel slot, a successful transmission, or a collision, respectively.
6. **Carrier Sense.** Stations can tell if the channel is in use before trying to use it. If the channel is sensed as busy, no station will attempt to use it until it goes idle.
7. **No Carrier Sense.** Stations cannot sense the channel before trying to use. They just go ahead and transit. Only later can they determine whether or not the transmission was successful.

4.3 IEEE STANDARD 802.3 AND ETHERNET

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IEEE 802.3 is a collection of IEEE standards defining the Physical Layer and Data Link Layer's media access control (MAC) sublayer of wired Ethernet. This is generally a LAN technology with some WAN applications. Physical connections are made between nodes and/or infrastructure devices (hubs, switches, routers) by various types of copper or fiber cable.

802.3 is a technology that supports the IEEE 802.1 network architecture.

The maximum packet size is 1518 bytes, although to allow the Q-tag for Virtual LAN and priority data in 802.3ac it is extended to 1522 bytes. If the upper layer protocol submits a protocol data unit (PDU) less than 64 bytes, 802.3 will pad the data field to achieve the minimum 64 bytes. The minimum Frame size will then always be of 64 bytes.

Although it is not technically correct, the terms packet and frame are often used interchangeably. The ISO/IEC 8802-3 and ANSI/IEEE 802.3 standards refer to MAC sub-layer frames consisting of the destination address, the source address, length/type, data payload, and frame check sequence (FCS) fields. The preamble and Start Frame Delimiter (SFD) are (usually) together considered a header to the MAC frame. This header and the MAC frame constitute a packet.

Ethernet Communication Standards

Ethernet Standard	Date	Description
	Experimental	
Ethernet	1972	2.94 Mbit/s (367 kB/s) over coaxial cable (coax) cable bus
Ethernet II		
(DIX v2.0)	1982	10 Mbit/s (1.25 MB/s) over thick coax. Frames have a Type field. This frame format is used on all forms of Ethernet by protocols in the Internet protocol suite.
IEEE 802.3	1983	10BASE5 10 Mbit/s (1.25 MB/s) over thick coax. Same as Ethernet II (above) except Type field is replaced by Length, and an 802.2 LLC header follows the 802.3 header
802.3a	1985	10BASE2 10 Mbit/s (1.25 MB/s) over thin Coax (a.k.a. thinnet or cheapernet)
802.3b	1985	10BROAD36
802.3c	1985	10 Mbit/s (1.25 MB/s) repeater specs
802.3d	1987	FOIRL (Fiber-Optic Inter-Repeater Link)
802.3e	1987	1BASE5 or StarLAN
802.3i	1990	10BASE-T 10 Mbit/s (1.25 MB/s) over twisted pair

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802.3j	1993	10BASE-F 10 Mbit/s (1.25 MB/s) over Fiber-Optic
802.3u	1995	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbit/s (12.5 MB/s) w/ autonegotiation
802.3x	1997	Full Duplex and flow control; also incorporates DIX framing, so there's no longer a DIX/802.3 split
802.3y	1998	100BASE-T2 100 Mbit/s (12.5 MB/s) over low quality twisted pair
802.3z	1998	1000BASE-X Gbit/s Ethernet over Fiber-Optic at 1 Gbit/s (125 MB/s)
802.3-1998	1998	A revision of base standard incorporating the above amendments and errata
802.3ab	1999	1000BASE-T Gbit/s Ethernet over twisted pair at 1 Gbit/s (125 MB/s)
802.3ac	1998	Max frame size extended to 1522 bytes (to allow "Q-tag") The Q-tag includes 802.1Q VLAN information and 802.1p priority information.
802.3ad	2000	Link aggregation for parallel links, since moved to IEEE 802.1AX
802.3-2002	2002	A revision of base standard incorporating the three prior amendments and errata
802.3ae	2003	10 Gbit/s (1,250 MB/s) Ethernet over fiber; 10GBASE-SR, 10GBASE-LR, 10GBASE-ER, 10GBASE-SW, 10GBASE-LW, 10GBASE-EW
802.3af	2003	Power over Ethernet (12.95 W)
802.3ah	2004	Ethernet in the First Mile
802.3ak	2004	10GBASE-CX4 10 Gbit/s (1,250 MB/s) Ethernet over twin-axial cable
802.3-2005	2005	A revision of base standard incorporating the four prior amendments and errata.
802.3an	2006	10GBASE-T 10 Gbit/s (1,250 MB/s) Ethernet over unshielded twisted pair(UTP)
802.3ap	2007	Backplane Ethernet (1 and 10 Gbit/s (125 and 1,250 MB/s) over printed circuit boards)
802.3aq	2006	10GBASE-LRM 10 Gbit/s (1,250 MB/s) Ethernet over multimode fiber
P802.3ar	Cancelled	Congestion management (withdrawn)

802.3as	2006	Frame expansion
802.3at	2009	Power over Ethernet enhancements (25.5 W)
802.3au	2006	Isolation requirements for Power Over Ethernet (802.3-2005/Cor 1)
802.3av	2009	10 Gbit/s EPON
802.3aw	2007	Fixed an equation in the publication of 10GBASE-T (released as 802.3-2005/Cor 2)
802.3-2008	2008	A revision of base standard incorporating the 802.3an/ap/aq/as amendments, two corrigenda and errata. Link aggregation was moved to 802.1AX.
P802.3az	~ Sep 2010[1]	Energy Efficient Ethernet
802.3ba	2010	40 Gbit/s and 100 Gbit/s Ethernet. 40 Gbit/s over 1m backplane, 10m Cu cable assembly (4x25 Gbit or 10x10 Gbit lanes) and 100 m of MMF and 100 Gbit/s up to 10 m of Cu cable assembly, 100 m of MMF, or 40 km of SMF respectively
802.3bb	2009	Increase Pause Reaction Delay timings which are insufficient for 10G/sec (released as 802.3-2008/Cor 1)
802.3bc	2009	Move and update Ethernet related TLVs (type, length, values), previously specified in Annex F of IEEE 802.1AB (LLDP) to 802.3.
P802.3bd	~July 2010[2]	Priority-based Flow Control. A amendment by the IEEE 802.1 Data Center Bridging Task Group (802.1Qbb) to develop an amendment to IEEE Std 802.3 to add a MAC Control Frame to support IEEE 802.1Qbb Priority-based Flow Control.
P802.3be	~ Feb 2011	Creates an IEEE 802.3.1 MIB definitions for Ethernet that consolidates the Ethernet related MIBs present in Annex 30A&B, various IETF RFCs, and 802.1AB annex F into one master document with a machine readable extract.
P802.3bf	~ Jun 2011	Provide an accurate indication of the transmission and reception initiation times of certain packets as required to support IEEE P802.1AS.
P802.3bg	~ Sep 2011	Provide a 40 Gbit/s PMD which is optically compatible with existing carrier SMF 40Gb/s client interfaces (OTU3/STM-256/OC-768/40G POS).

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What is defined in earlier IEEE 802.3 standards is often confused for what is used in

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practice: most network frames you will find on an Ethernet will be DIX frames, since the Internet protocol suite will use this format, with the type field set to the corresponding IETF protocol type. IEEE 802.3x-1997 allows the 16-bit field after the MAC addresses to be used as a type field or a length field, so that DIX frames are also valid 802.3 frames in 802.3x-1997 and later versions of the IEEE 802.3 Ethernet standard.

4.4 IEEE STANDARD 802.4 AND TOKEN BUS

Token bus is a network implementing the token ring protocol over a "virtual ring" on a coaxial cable. A token is passed around the network nodes and only the node possessing the token may transmit. If a node doesn't have anything to send, the token is passed on to the next node on the virtual ring. Each node must know the address of its neighbour in the ring, so a special protocol is needed to notify the other nodes of connections to, and disconnections from, the ring.

Token bus was standardized by IEEE standard 802.4. It is mainly used for industrial applications. Token bus was used by GM (General Motors) for their Manufacturing Automation Protocol (MAP) standardization effort. This is an application of the concepts used in token ring networks. The main difference is that the endpoints of the bus do not meet to form a physical ring. The IEEE 802.4 Working Group is disbanded. In order to guarantee the packet delay and transmission in Token bus protocol, a modified Token bus was proposed in Manufacturing Automation Systems and flexible manufacturing system (FMS).

4.5 IEEE 802.4 AND TOKEN RING; BRIDGES

Ring networks have been around for many years and have long been used for both local and wide area networks. Among their many attractive features is the fact that a ring is not really a broadcast medium, but a collection of individual point-to-point links that happen to form a circle. Point-to-point links involve a well understood and field-proven technology and can run on twisted pair, coaxial cable, or fibre optics. Ring engineering is also almost entirely digital, whereas 802.3, for example, has a substantial analog component for collision detection. A ring is also fair and has a known upper bound on channel access. For three reasons, IBM chose the ring as its LAN and IEEE has included the token ring standard as 802.5.

In a token ring a special bit pattern, called the token, circulates around the ring whenever all stations are idle. When a station wants to transmit a frame, it is required to seize the token and remove it rfrom the ring before transmitting. This action is done by inverting a single bit in the 3-byte token, which instantly changes it into the first 3 bytes of a normal data frame. Because there is only one token, only station can trnsmit at a given instant, thus solving the channel access problem the same way the token bus solves it.

4.6 BRIDGES FROM 802 X TO 802 Y

You might think that a bridge from one 802 LAN to another one would be completely trival. Such is not the case. Each of the nine combinations of 802.x and 802.y has its

own unique set of problems. However, before dealing with these one at a time, let us look at some general problems common to all the bridges. To start with, each of he LANs uses a different frame format. There is no valid technical reason for this incompatibility. It is just that none of the corporations supporting the three standards (Xerox, GM, and IBM) wanted to change theirs.

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As a result any copying between different LANs requires reformatting, which takes CPU time, requires a new checksum calculation, and introduces the possibility of undetected errors due to bad bits in the bridge's memory. None of this would have been necessary if the three committees has been able to agree on a single format. A second problem is that interconnected LANs do not necessarily run at the same data rate.

When forwarding a long run of back-to-back frames from a fast LAN to a slower one, the bridge will not be able to get rid of the frames as fast as they come in. It will have to buffer them, hoping not to run out of memory. The problem also exists from 802.4 to 802.3 at 10 Mbps to some extent because some of 802.3 bandwidth is lost to collisions. It does not really have 10 Mbps, whereas 802.4 really does. Bridges that connect three or more LANs have a similar problem when several LANs are trying to feed the same output LAN at the same time.

A third and potentially most serious problem of all, is that, all three 802 LANs have a different maximum frame length. For 802.3 it depends on the parameters of the configuration, but for the standard 10-Mbps system the payload is a maximum of 1500 bytes. For 802.4 it is fixed at 8191 bytes. For 802.5 there is not upper limit except that a station may not transmit longer than the token-holding time. With the default value of 10 mses, the maximum frame length is 5000 bytes.

From 802.4 to 802.3 two problems exist. First, 802.4 frames carry priority bits that 802.3 framess do not have. As a result, if two 802.4 LANs communicate via an 802.3 LAN, the priority will be lost by the intermediate LAN.

4.7 TRANSPARENT BRIDGES

Bridging is a forwarding technique used in packet-switched computer networks. Unlike routing, bridging makes no assumptions about where in a network a particular address is located. Instead, it depends on flooding and examination of source addresses in received packet headers to locate unknown devices. Once a device has been located, its location is recorded in a table where the MAC address is stored so as to preclude the need for further broadcasting. The utility of bridging is limited by its dependence on flooding, and is thus only used in local area networks.

Bridging generally refers to Transparent bridging or Learning bridge operation which predominates in Ethernet. Another form of bridging, Source route bridging, was developed for token ring networks.

A network bridge connects multiple network segments at the data link layer (Layer 2) of the OSI model. In Ethernet networks, the term bridge formally means a device that behaves according to the IEEE 802.1D standard. A bridge and switch are very much alike; a switch being a bridge with numerous ports. Switch or Layer 2 switch is often used interchangeably with bridge.

Bridges are similar to repeaters or network hubs, devices that connect network

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segments at the physical layer; however, with bridging, traffic from one network is managed rather than simply rebroadcast to adjacent network segments. Bridges are more complex than hubs or repeaters. Bridges can analyze incoming data packets to determine if the bridge is able to send the given packet to another segment of the network.

Transparent bridging operation

A bridge uses a forwarding database to send frames across network segments. The forwarding database is initially empty and entries in the database are built as the bridge receives frames. If an address entry is not found in the forwarding database, the frame is flooded to all other ports of the bridge, forwarding the frame to all segments except the source address. By means of these broadcast frames, the destination network will respond and forwarding database entry will be created.

As an example, consider three hosts, A, B and C and a bridge. The bridge has three ports. A is connected to bridge port 1, B is connected bridge port 2, C is connected to bridge port 3. A sends a frame addressed to B to the bridge. The bridge examines the source address of the frame and creates an address and port number entry for A in its forwarding table. The bridge examines the destination address of the frame and does not find it in its forwarding table so it floods it to all other ports: 2 and 3. The frame is received by hosts B and C. Host C examines the destination address and ignores the frame. Host B recognizes a destination address match and generates a response to A. On the return path, the bridge adds an address and port number entry for B to its forwarding table. The bridge already has A's address in its forwarding table so it forwards the response only to port 1. Host C or any other hosts on port 3 are not burdened with the response. Two-way communication is now possible between A and B without any further flooding.

Note that both source and destination addresses are used in this algorithm. Source addresses are recorded in entries in the table, while destination addresses are looked up in the table and matched to the proper segment to send the frame to.

The technology was originally developed by the Digital Equipment Corp. in the 1980s.

Filtering database

To translate between two segments, a bridge reads a frame's destination MAC address and decides to either forward or filter. If the bridge determines that the destination node is on another segment on the network, it forwards it (retransmits) the packet to that segment. If the destination address belongs to the same segment as the source address, the bridge filters (discards) the frame. As nodes transmit data through the bridge, the bridge establishes a filtering database (also known as a forwarding table) of known MAC addresses and their locations on the network. The bridge uses its filtering database to determine whether a packet should be forwarded or filtered.

Advantages of network bridges

- Self-configuring
- Simple bridges are inexpensive
- Isolate collision domain
- Reduce the size of collision domain by microsegmentation in non-switched networks

- Transparent to protocols above the MAC layer
- Allows the introduction of management/performance information and access control
- LANs interconnected are separate, and physical constraints such as number of stations, repeaters and segment length don't apply
- Helps minimize bandwidth usage

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Disadvantages of network bridges

- Does not limit the scope of broadcasts [broadcast domain cannot be controlled]
- Does not scale to extremely large networks
- Buffering and processing introduces delays
- Bridges are more expensive than repeaters or hubs

A complex network topology can pose a problem for transparent bridges. For example, multiple paths between transparent bridges and LANs can result in bridge loops. The spanning tree protocol helps to reduce problems with complex topologies.

4.8 SOURCE ROUTING BRIDGES

Transparent bridges have the advantages of being easy to install. You just plug them in and walk away. On the other hand, they do not make optimal use of the bandwith, since they only use a subset of the topology (the spanning tree). The relative importance of these two (and other) factors led to a split within the 802 committees. Thee ring CSMA/CD and token bus people chose the transparent bridge. The ring people (with encouragement from IBM) preferred a scheme, called source routing, which is explained next.

Source route bridging is used on token ring networks, and is standardized in Section 9 of the IEEE 802.2 standard. The operation of the bridge is simpler (spanning tree protocol is not necessary) and much of the bridging functions are performed by the end systems, particularly the sources, giving rise to its name.

Source-route transparent bridging, abbreviated SRT bridging, is a hybrid of source routing and transparent bridging, standardized in Section 9 of the IEEE 802.2 standard. It allows source routing and transparent bridging to coexist on the same bridged network by using source routing with hosts that support it and transparent bridging otherwise.

A field in the token ring header, the routing information field (RIF), is used to support source-route bridging. Upon sending a packet, a host attaches a RIF to the packet indicating the series of bridges and network segments to be used for delivering the packet to its destination. The bridges merely follow the list given in the RIF - if a given bridge is next in the list, it forwards the packet, otherwise it ignores it.

When a host wishes to send a packet to a destination for the first time, it needs to determine an appropriate RIF. A special type of broadcast packet is used, which instructs the network bridges to append their bridge number and network segment number to each packet as it is forwarded. Loops are avoided by requiring each bridge to ignore packets which already contain its bridge number in the RIF field. At the

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destination, these broadcast packets are modified to be standard unicast packets and returned to the source along the reverse path listed in the RIF. Thus, for each route discovery packet broadcast, the source receives back a set of packets, one for each possible path through the network to the destination. It is then up to the source to choose one of these paths (probably the shortest one) for further communications with the destination.

Two frame types are used in order to find the route to the destination network segment. Single-Route (SR) frames make up most of the network traffic and have set destinations, while All-Route (AR) frames are used to find routes. Bridges send AR frames by broadcasting on all network branches; each step of the followed route is registered by the bridge performing it. Each frame has a maximum hop count, which is determined to be greater than the diameter of the network graph, and is decremented by each bridge. to avoid indefinite looping of AR frames, frames are dropped when this hop count reaches zero. The first AR frame that reaches its destination is considered to have followed the best route, and the route can be used for subsequent SR frames; the other AR frames are discarded.

This method of locating a destination network can allow for indirect load balancing among multiple bridges connecting two networks. The more a bridge is loaded, the less likely it is to take part in the route finding process for a new destination as it will be slow to forward packets. A new AR packet will find a different route over a less busy path if one exists. This method is very different from transparent bridge usage, where redundant bridges will be inactivated; however, more overhead is introduced to find routes, and space is wasted to store them in frames. A switch with a faster backplane can be just as good for performance, if not for fault tolerance. They are primarily found in Token Ring networks.

SUMMARY

1. The protocols used to determine who goes next on a multiaccesschannel belong to a sublayer of the data link layer called Medium Access Control Sub Layer.
2. IEEE 802.3 is a collection of IEEE standards defining the Physical Layer and Data Link Layer's media access control (MAC) sublayer of wired Ethernet.
3. Token bus is a network implementing the token ring protocol over a "virtual ring" on a coaxial cable.
4. Each of the nine combinations of 802.x and 802.y has its own unique set of problems.
5. Bridging is a forwarding technique used in packet-switched computer networks. Unlike routing, bridging makes no assumptions about where in a network a particular address is located.
6. Bridging is a forwarding technique used in packet-switched computer networks. Unlike routing, bridging makes no assumptions about where in a network a particular address is located.
7. A bridge uses a forwarding database to send frames across network segments. The forwarding database is initially empty and entries in the database are built as the bridge receives frames.
8. Source-route transparent bridging, abbreviated SRT bridging, is a hybrid of source routing and transparent bridging, standardized in Section 9 of the IEEE 802.2 standard.

SELF ASSESSMENT QUESTIONS

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1. Describe The Medium Access Sub Layer.
2. What is Framing Static and Dynamic Challen Allocation in LANs and MANs?
3. Describe IEEE standard 802.3 and Ethernet.
4. Describe IEEE standard 802.4 and Token Bus.
5. Describe IEEE standard 802.4 and Token Ring, Bridges.
6. What is difference between Bridges 802.x and 802.y?
7. What are transparent bridges?
8. Describe Source Routing Bridges.

Short Questions with Answers

1. What is the framing static and dynamic channel allocation in LANs and MANs?

Ans. Underlying all the work done in this area are five key assumptions, described below:

1. **Station model.** The model consists of N independent stations (computers, telephones, personal communications, etc.), each with a program or user that generates frames for transmission. The probality of a frame being generated in an interval of of length Δt is $\lambda \Delta t$, where λ is a constant (the arrival rate of new frames). Once a frame has been generated, the station is blocked and does nothing until the frame has been successfully transmitted.
2. **Single Channel Assumption.** A single channel is available for all communication. All stations can transmit on it and all can receive from it. As far as the hardware is concerned, all stations are equivalent, although protocol software may assign priorities to them.
3. **Collision Assumption.** If two frames are trnsmitted simultaneously, they overlap in time and the resulting signal is garbled. This even is called a collsion. All stations can detect collisions. A collided frame must be trnsmitted again later. There are no errors other than those generated by collisions.
4. **Continuous Time.** Frame transmission can begin at any instant. There is no master clock dividing time into discrete intervals.
5. **Slotted Time.** Time is divided into discrete intervals (slots). Frame transissions always begin at the start of a slot. A slot may contain 0, 1, or more frames, corresponding to an idel slot, a successful transmission, or a collision, respectively.
6. **Carrier Sense.** Stations can tell if the channel is in use before trying to use it. If the channel is sensed as busy, no station will attempt to use it until it goes idle.
7. **No Carrier Sense.** Stations cannot sense the channel before trying to use. They just go ahead and transit. Only later can they determine whether or not the transmission was successful.

2. What is IEEE 802.3 standard?

Ans. IEEE 802.3 is a collection of IEEE standards defining the Physical Layer and Data Link Layer's media access control (MAC) sublayer of wired Ethernet. This is generally a LAN technology with some WAN applications. Physical connections are made between nodes and/or infrastructure devices (hubs, switches, routers) by various types of copper or fiber cable.

3. What is IEEE 802.4 standard?

Ans. Token bus was standardized by IEEE standard 802.4. It is mainly used for industrial applications. Token bus was used by GM (General Motors) for their Manufacturing Automation Protocol (MAP) standardization effort. This is an application of the concepts used in token ring networks. The main difference is that the endpoints of the bus do not meet to form a physical ring. The IEEE 802.4 Working Group is disbanded. In order to

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guarantee the packet delay and transmission in Token bus protocol, a modified Token bus was proposed in Manufacturing Automation Systems and flexible manufacturing system (FMS).

4. What is IEEE 802.4 and Token Bridge?

Ans. In a token ring a special bit pattern, called the token, circulates around the ring whenever all stations are idle. When a station wants to transmit a frame, it is required to seize the token and remove it rfrom the ring before transmitting. This action is done by inverting a single bit in the 3-byte token, which instantly changes it into the first 3 bytes of a normal data frame. Because there is only one token, only station can trnsmit at a given instant, thus solving the channel access problem the same way the token bus solves it.

Further Readings

1. Data and Computer Network Communication: Prof. Shashi Banzai, Firewal Media.
2. Networking: Balvir Singh, Firewal Media.
3. Wide Area Networks: Navneet Sharma, Firewal Media.
4. Computer Network: Bharat Bhushan Agarwal and Sumit Prakash Tayal, University Science Press.

UNIT 5

THE NETWORK LAYER

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STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- understand about the Network Layer of the networking.
- know about Network Layer Design Issues.
- learn the Shortest Path Routing.
- describe about the Flooding.
- understand about the Flow Based Routine.
- know about the Broadcast Routine.
- learn about Congestion Control and Prevention Policies.
- understand about the Internet Working.
- learn about Connectionless Internet Working.
- know about Tunneling Internet Work Routing.
- understand about Fragmentation, Firewalls, IP address and Internet Control Protocols.

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5.1 THE NETWORK LAYER

The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer. The Network layer performs network routing functions, and might also perform fragmentation and reassembly, and report delivery errors. Routers operate at this layer—sending data throughout the extended network and making the Internet possible. This is a logical addressing scheme – values are chosen by the network engineer.

Perhaps it's easier to visualize this layer as managing the sequence of human carriers taking a letter from the sender to the local post office, trucks that carry sacks of mail to other post offices or airports, airplanes that carry airmail between major cities, trucks that distribute mail sacks in a city, and carriers that take a letter to its destinations. Think of fragmentation as splitting a large document into smaller envelopes for shipping, or, in the case of the new.

5.2 NETWORK LAYER DESIGN ISSUES

Some of the design issues which need to tackled are:

1. Services Provided to the Transport Layer.
2. Internal Organization of the Network Layer.
3. Comparision of Virtual Circuit and Ditagram Subnets.

Services Provided to the Transport Layer

The network layer provides services to the Transport Layer at the network layer/ transport layer interface. This interface is often especially important for another reason; the boundary of the subnet. The carrier often has control of the protocols and interfaces up to and including the network layer. Its job is to deliver packets given to it by its customers. For this reason, this interface must be especially well defined.

The network layer services have been designed with the following goals in mind.

1. The services should be independent of the subnet technology.
2. The transport layer should be shielded from the number, type, and topology of the subnets present.
3. The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.

Some people argue that the subnet should provide a (reasonably) reliable, connection-oriented service. They claim 100 years of successful experience with the worldwide telephone system is a good guide. In this view, conections should have the following properties:

1. Before sending data, a network layer process on the sending side must set up a connection to its peer on the receiving side. This connection, which is given a special identifier, is then used until all the data have been sent, at which time it is explicity released.

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2. When a connection is set up, the two processes can enter into a negotiation about the parameters, quality, and cost of the service to the provided.
3. Communication is in both directions, and packets are delivered in sequence.
4. Flow control is provided automatically to prevent a fast sender from dumping packets into the pipe at a higher rate than the receiver can take them out, thus leading to overflow.

Internet Organization of the Network Layer

There are basically two ways of looking at the organizing the subnet, one using connections and the other working connectionless. In the context of the internal operation of the subnet, a connection is usually called a virtual circuit, in analogy with the physical circuits set up by the telephone system. The independent packets of the connectionless organization are called datagrams, in analogy with telegrams.

As compared to datagram, for subnet no routes are worked out in advance, even if the service is connection-oriented. Each packet sent is routed independently of its predecessors. Successive packets may follow different routes. While datagram subnets have to do more work, they are also generally more roboust and adapt to failures and congestion more easily than virtual circuit subnets.

When a network connection is set up, a virtual circuit number not already in use on that machine is chosen as connection identifier. Since each machine chooses virtual curcuit numbers independently, these numbers have only local significance. If they wer globally significant over the whole network, it is likely through some intermediate router, leasing to ambiguities.

Comparision of Virtual Circuit and Datagram Subnets

Both virtual circuits and datagrams have their supporters and their detractors. These can be summarized in the following manner.

Issue	Datagram subnet	VC subnet
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination addresses	Each packet contains a short VC number
State information	Subnet does not hold state information	Each VC requires subnet table space
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow this route
Effect of route failures	None, except for packetss lost during the crash	All VCs that passed through the failed router are terminated
Congestion *control*	Difficult	Easy if enough buffers can be allocated in advance for each VC.

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Inside the subnet, several trade-off exist between virual circuits and datagrams. One trade-off is between router memory space and bandwidth. Virtual circuits allow packets to contain cicuit numbers instead of full destination addresses. If the packets tend to be fairly short, a full destination address in every packet may represent a significant amout of overhead, and hence wasted bandwidth. The price paid for using virtual circuits internally is the table space within the routers. Depending upon the relative cost of communication circuits versus router memory, one or the other may be cheaper.

5.3 SHORTEST PATH ROUTING

Open **Shortest Path** First (OSPF) is a dynamic routing protocol for use in Internet Protocol (IP) networks. Specifically, it is a link-state routing protocol and falls into the group of interior gateway protocols, operating within a single autonomous system (AS). It is defined as OSPF Version 2 in RFC 2328 (1998) for IPv4. The updates for IPv6 are specified as OSPF Version 3 in RFC 5340 (2008).

OSPF is perhaps the most widely-used interior gateway protocol (IGP) in large enterprise networks; IS-IS, another link-state routing protocol, is more common in large service provider networks. The most widely-used exterior gateway protocol is the Border Gateway Protocol (BGP), the principal routing protocol between autonomous systems on the Internet.

OSPF is an interior gateway protocol that routes Internet Protocol (IP) packets solely within a single routing domain (autonomous system). It gathers link state information from available routers and constructs a topology map of the network. The topology determines the routing table presented to the Internet Layer which makes routing decisions based solely on the destination IP address found in IP datagrams. OSPF was designed to support variable-length subnet masking (VLSM) or Classless Inter-Domain Routing (CIDR) addressing models.

OSPF detects changes in the topology, such as link failures, very quickly and converges on a new loop-free routing structure within seconds. It computes the shortest path tree for each route using a method based on Dijkstra's algorithm, a shortest path first algorithm.

The link-state information is maintained on each router as a link-state database (LSDB) which is a tree-image of the entire network topology. Identical copies of the LSDB are periodically updated through flooding on all OSPF routers.

The OSPF routing policies to construct a route table are governed by link cost factors (external metrics) associated with each routing interface. Cost factors may be the distance of a router (round-trip time), network throughput of a link, or link availability and reliability, expressed as simple unitless numbers. This provides a dynamic process of traffic load balancing between routes of equal cost.

An OSPF network may be structured, or subdivided, into routing areas to simplify administration and optimize traffic and resource utilization. Areas are identified by 32-bit numbers, expressed either simply in decimal, or often in octet-based dot-decimal notation, familiar from IPv4 address notation.

By convention, area 0 (zero) or 0.0.0.0 represents the core or backbone region of an OSPF network. The identifications of other areas may be chosen at will, often, administrators select the IP address of a main router in an area as the area's

identification. Each additional area must have a direct or virtual connection to the backbone OSPF area. Such connections are maintained by an interconnecting router, known as area border router (ABR). An ABR maintains separate link state databases for each area it serves and maintains summarized routes for all areas in the network.

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OSPF does not use a TCP/IP transport protocol (UDP, TCP), but is encapsulated directly in IP datagrams with protocol number 89. This is in contrast to other routing protocols, such as the Routing Information Protocol (RIP), or the Border Gateway Protocol (BGP). OSPF handles its own error detection and correction functions.

OSPF uses multicast addressing for route flooding on a broadcast network link. For non-broadcast networks special provisions for configuration facilitate neighbor discovery. OSPF multicast IP packets never traverse IP routers, they never travel more than one hop. OSPF reserves the multicast addresses 224.0.0.5 for IPv4 or FF02::5 for IPv6 (all SPF/link state routers, also known as AllSPFRouters) and 224.0.0.6 for IPv4 or FF02::6 for IPv6 (all Designated Routers, AllDRouters), as specified in RFC 2328 and RFC 5340.

For routing multicast IP traffic, OSPF supports the Multicast Open Shortest Path First protocol (MOSPF) as defined in RFC 1584. Neither Cisco nor Juniper Networks include MOSPF in their OSPF implementations. PIM (Protocol Independent Multicast) in conjunction with OSPF or other IGPs, (Interior Gateway Protocol), is widely deployed.

The OSPF protocol, when running on IPv4, can operate securely between routers, optionally using a variety of authentication methods to allow only trusted routers to participate in routing. OSPFv3, running on IPv6, no longer supports protocol-internal authentication. Instead, it relies on IPv6 protocol security (IPsec).

OSPF version 3 introduces modifications to the IPv4 implementation of the protocol. Except for virtual links, all neighbor exchanges use IPv6 link-local addressing exclusively. The IPv6 protocol runs per link, rather than based on the subnet. All IP prefix information has been removed from the link-state advertisements and from the Hello discovery packet making OSPFv3 essentially protocol-independent. Despite the expanded IP addressing to 128-bits in IPv6, area and router identifications are still based on 32-bit values.

5.4 FLOODING

Flooding is a static algorithm in which every incoming packet is sent out on every *outgoing line except the one it arrived on.* Flooding obviously generates vast numbers of duplicate packets. In fact, an infinite number unless some measures are taken to damp the prrocess. One such measure is to have a hop, counter contained in the header of each packet, which is decremented at each hop, with the packet being discarded when.the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the subnet.

An alternative technique for damming the floop is to keep track of which packets have been flooded, to avoid sending them out a second tiem. One way to achieve this goad is to have the source router put a sequence number in each packet it receives

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from its hosts. Each router then needs a list per source router telling which sequence numbers originating at that source have already been seen. If an incoming packet is on the list, it is not flooded.

To prevent the list from growing without bound, each list should be augmented by a counte, k, meaning all sequence numbers through k have been seen. When a packet comes in, it is easy to check if the packet is a duplicate, if so, it is descarded. Furthermore, the full list below k is not needed, since k effectively summarizes it.

5.5 FLOW BASED ROUTING

A static algorithm that uses both topology and load for routing, is called flow-based routing. In some networks, the mean data flow between each pair of nodes is relatively stable and predictable. For example, in a coorporate network for a retail store chain, each store might send orders, sales reports, inventory updates, and other well-defined types of messages to known sites in a predefined pattern, so that the total volume of traffic various little from day to day. Under conditions in which he average traffic from i to j is known in advance and, to a reasonable approximation, constant in time; it is possible to analyze the flows mathematically to optimize the routing.

The basic idea behind the analysis is that for a given line, if the capacity and avarage flow are know, it is possible to compute the mean packet delay on the line from queueing theory. From the mean delays on all the lines, it is straightforward to calculate a flow weighted average to get the mean packet delay for the whole subnet. The routing problem then reduces to finding the routing algorithm that produces he minimim avarage delay for the subnet.

To use this technique, certain information must be known in advance. First the subnet topology must be known. Second, the traffic matrix, Fy, must be given. Third, the line capacity matrix, Cy, specifying the capacity of each line in bps must be available. Finally, a (possibly tentative) routing algorithm must be chosen.

5.6 BROADCAST ROUTING

For some applications, hosts need to send mesages to many or all other hosts. For example, a service distributing wather reports, stock market updates, or live radio programs might work best by broadcasting to all machines and letting those that are interested read the data. Sending a packet to all destinations simultaneously is called Broadcasting; various methods have been proposed for doing it.

One broadcasting method that requires no special features from the subnet is for the source to simply send a distinct packet to each destination. Not only is the method wasteful of bandwidth, but it also requires the source to have a complete list of all destinations. In practice this may be the only possibility, but is the least desirable of the methods.

Flooding is another obvious choice. Although flooding is ill-suited for ordinary point-to-point communcation, for broadcasting it might rate serious consideration, especially if none of the methods described below are applicable. The problem with flooding as a broadcast technique is the same problem it has as a point-to-point routing algorithm; it generates too many packets and consumes too much bandwith.

5.7 CONGESTION CONTROL AND PREVENTION POLICIES

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Congestion control concerns controlling traffic entry into a telecommunications network, so as to avoid congestive collapse by attempting to avoid oversubscription of any of the processing or link capabilities of the intermediate nodes and networks and taking resource reducing steps, such as reducing the rate of sending packets. It should not be confused with flow control, which prevents the sender from overwhelming the receiver.

Classification of congestion control algorithms

- There are many ways to classify congestion control algorithms:
- By the type and amount of feedback received from the network: Loss; delay; single-bit or multi-bit explicit signals
- By incremental deployability on the current Internet: Only sender needs modification; sender and receiver need modification; only router needs modification; sender, receiver and routers need modification.
- By the aspect of performance it aims to improve: high bandwidth-delay product networks; lossy links; fairness; advantage to short flows; variable-rate links
- By the fairness criterion it uses: max-min, proportional, "minimum potential delay"

5.8 INTERNET WORKING

Up until now, we have implicitly assumed that there is a single homogenous network, with each machine using the same protocol in each layer. Unfortunately, this assumtion is widly optimistic. Many differnt networks exist, including LANs, WANs, and MANs. Numerous protocols are in widespread use in every layer. Now we look at the issues that arise when two or more networks are together to form an internet.

First of all, the installed base of different networks is large and growing. Nearly al UNIX shops run TCP/IP. Many large businesses still have mainframes running SNA. DEC is still developing DECnet. Personal computer LANs often use Novell, NCP/IPX or AppleTalk ATM systems are starting to be widespread. Finally, specialized protocols are often used on satellite, cellular, and infrared networks. This trend will continue for years due to the large number of existing networks and because not all vendors perceive it in their interest for their customers to be able to easily migrate to another vendor's system.

Second, as computers and networks get cheaper, the place where decisions get made moves downward. Many companies have a policy to the effect that purchases costing over a million dollars have to be approved by top management, purchases costing over 100,000 dollars have to be approved by middle management, but purchases under 100,000 dollars can be made by department hads without any higher approval. Thsi can easily lead to the accounting department instaling an Ethernet, the engineering department installing a token bus, and the personnel department installing a token ring.

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Third, different networks, (e.g., ATM and wireless) have radically different technology, so it should not be surprising that as new hardware developments occur, new software will be created to fit eh new hardware. For example, the average home now is like the average office ten years ago; it is full of computers that do not talk to one another. In the future, it may be commonplace for the telephone, the television set, and other applicances all to be networked together, so they can be controlled remotely. This new technology will undoubtedly bring new protocols.

5.9 CONNECTIONLESS INTERNET WORKING

In this case, the only service the network layer offers to the transport layer is the ability to inject daagrams into the subnet and hope for the best. There is no motion of a virtual circuit at all in the network layer, let alone a concatenation of them. This model does not require all packets belonging to one connection to traverse the same sequence of gateways. A routing decision is made separately for each packet, possibly depending on the traffic at the moment the packet is sent. This strategy can use multiple routes and thus achieve a higher bandwidth than the concatenated virtual circuit mode. On the other hand, there is no guarantee that the packets arrive at the destination in order, assuming that they arrive at all.

A second and more serious problem, is addressing. Imagine a simple case; a host on the Internet is trying to send an IP packet to a host on an adjoining OSI network. The OSI datagram protocol, CLNP, was based on IP and is close enough to it that a conversion might well work. The trouble is that IP packets all carry the 32-bit Internet address of the destination host in a header field. OSI hosts do not have 32-bit Internet addresses. They use decimal addresses similar to telephone numbers.

To make it possible for the multiprotocol router to convert between formats, someone wold have to assign a 32-bit Interest address to each OSI host. Taken to the limit, this approach would mean assigning an Internet address to every mahine in the world that an Internet host might want to talk to. It would also mean assigning an OSI address to every machine in the world that an OSI host might want to talk to. The same problem occurs with every other address space (SNA, AppleTalk, etc.) The problems here are insurmountable. In addition, someone would have to maintain a database mapping everything to everything.

5.10 TUNNELING INTERNET WORK ROUTING

Handling the general case of making two different networks interwork is exceedingly difficult. However, there is a common special case that is manageable. This case is where the source and destination hosts are on the same type of network, but there is a different network in between. As an example, think of an international bank wih a TCP/IP based Ethernet in Paris, a TCP/IP based Ethernet in London, and a PTT WAN in between.

The solution to this problem is a technique called tunneling. To send an IP packet to host 2, host 1 constructs the packet containing the IP address of host 2, inserts it not an Ethernet frame addressed to the Paris multiprotocol router, and puts it on the Ethernet. When the multiprotocol router gets the frame, it removes the IP packet, inserts it in the payload field of the WAN network layer packet and addresses the

latter to the WAN address of the London multiprotocol router. When it gets there, the London router removes the IP packet and sends it to host 2 inside an Ethernet frame.

The WAN can be seen as a big tunnel extending from one multiprotocol router to the other. The IP packet just travesl from one end of the tunnel to the other, snug in its nice box. It does not have to worry about dealing with the WAN at all. Neither do the hosts on either Ethernet. Only the multiprotocol router has to understand IP and WAN packets. In effect, the entire distance from the middle of one multiprotocol router to the middle of the other acts like a serial line.

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5.11 FRAGMENTATION

In computer storage, fragmentation is a phenomenon in which storage space is used inefficiently, reducing storage capacity and in most cases performance. The term is also used to denote the wasted space itself.

There are three different but related forms of fragmentation: external fragmentation, internal fragmentation, and data fragmentation. Various storage allocation schemes exhibit one or more of these weaknesses. Fragmentation can be accepted in return for increase in speed or simplicity.

Internal fragmentation

Internal fragmentation occurs when storage is allocated without intention to use it. This space is wasted. While this seems foolish, it is often accepted in return for increased efficiency or simplicity. The term "internal" refers to the fact that the unusable storage is inside the allocated region but is not being used.

For example, in many file systems, each file always starts at the beginning of a cluster, because this simplifies organization and makes it easier to grow files. Any space left over between the last byte of the file and the first byte of the next cluster is a form of internal fragmentation called file slack or slack space. Slack space is a very important source of evidence in computer forensic investigation. These clips describe an analogy of a slack space.

Similarly, a program which allocates a single byte of data is often allocated many additional bytes for metadata and alignment. This extra space is also internal fragmentation.

Another common example: English text is often stored with one character in each 8-bit byte even though in standard ASCII encoding the most significant bit of each byte is always zero. The unused bits are a form of internal fragmentation.

Similar problems with leaving reserved resources unused appear in many other areas. For example, IP addresses can only be reserved in blocks of certain sizes, resulting in many IPs that are reserved but not actively used. This is contributing to the IPv4 address shortage.

Unlike other types of fragmentation, internal fragmentation is difficult to reclaim; usually the best way to remove it is with a design change. For example, in dynamic memory allocation, memory pools drastically cut internal fragmentation by spreading the space overhead over a larger number of objects.

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External fragmentation

External fragmentation is the phenomenon in which free storage becomes divided into many small pieces over time. It is a weakness of certain storage allocation algorithms, occurring when an application allocates and deallocates ("frees") regions of storage of varying sizes, and the allocation algorithm responds by leaving the allocated and deallocated regions interspersed. The result is that although free storage is available, it is effectively unusable because it is divided into pieces that are too small to satisfy the demands of the application. The term "external" refers to the fact that the unusable storage is outside the allocated regions.

For example, in dynamic memory allocation, a block of 1000 bytes might be requested, but the largest contiguous block of free space has only 300 bytes. Even if there are ten blocks of 300 bytes of free space, separated by allocated regions, one still cannot allocate the requested block of 1000 bytes, and the allocation request will fail.

External fragmentation also occurs in file systems as many files of different sizes are created, change size, and are deleted. The effect is even worse if a file which is divided into many small pieces is deleted, because this leaves similarly small regions of free spaces.

Data fragmentation

Data fragmentation occurs when a piece of data in memory is broken up into many pieces that are not close together. It is typically the result of attempting to insert a large object into storage that has already suffered external fragmentation.

For example, files in a file system are usually managed in units called blocks or clusters. When a file system is created, there is free space to store file blocks together contiguously. This allows for rapid sequential file reads and writes. However, as files are added, removed, and changed in size, the free space becomes externally fragmented, leaving only small holes in which to place new data. When a new file is written, or when an existing file is extended, the new data blocks are necessarily scattered, slowing access due to seek time and rotational delay of the read/write head, and incurring additional overhead to manage additional locations. This is called file system fragmentation.

As another example, if the nodes of a linked list are allocated consecutively in memory, this improves locality of reference and enhances data cache performance during traversal of the list. If the memory pool's free space is fragmented, new nodes will be spread throughout memory, increasing the number of cache misses.

Just as compaction can eliminate external fragmentation, data fragmentation can be eliminated by rearranging data storage so that related pieces are close together. For example, the primary job of a defragmentation tool is to rearrange blocks on disk so that the blocks of each file are contiguous. Most defragmenting utilities also attempt to reduce or eliminate free space fragmentation. Some moving garbage collectors will also move related objects close together (this is called compacting) to improve cache performance.

5.12 FIREWALLS

A firewall is a part of a computer system or network that is designed to block

unauthorized access while permitting authorized communications. It is a device or set of devices which is configured to permit or deny computer applications based upon a set of rules and other criteria.

Firewalls can be implemented in either hardware or software, or a combination of both. Firewalls are frequently used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

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There are several types of firewall techniques:

Packet filter: Packet filtering inspects each packet passing through the network and accepts or rejects it based on user-defined rules. Although difficult to configure, it is fairly effective and mostly transparent to its users. It is susceptible to IP spoofing.

Application gateway: Applies security mechanisms to specific applications, such as FTP and Telnet servers. This is very effective, but can impose a performance degradation.

Circuit-level gateway: Applies security mechanisms when a TCP or UDP connection is established. Once the connection has been made, packets can flow between the hosts without further checking.

Proxy server: Intercepts all messages entering and leaving the network. The proxy server effectively hides the true network addresses.

First generation: packet filters

The first paper published on firewall technology was in 1988, when engineers from Digital Equipment Corporation (DEC) developed filter systems known as packet filter firewalls. This fairly basic system was the first generation of what became a highly evolved and technical internet security feature. At AT&T Bell Labs, Bill Cheswick and Steve Bellovin were continuing their research in packet filtering and developed a working model for their own company based upon their original first generation architecture.

This type of packet filtering pays no attention to whether a packet is part of an existing stream of traffic (it stores no information on connection "state"). Instead, it filters each packet based only on information contained in the packet itself (most commonly using a combination of the packet's source and destination address, its protocol, and, for TCP and UDP traffic, the port number).

TCP and UDP protocols comprise most communication over the Internet, and because TCP and UDP traffic by convention uses well known ports for particular types of traffic, a "stateless" packet filter can distinguish between, and thus control, those types of traffic (such as web browsing, remote printing, email transmission, file transfer), unless the machines on each side of the packet filter are both using the same non-standard ports.

Packet filtering firewalls work on the first three layers of the OSI reference model, which means all the work is done between the network and physical layers. When a packet originates from the sender and filters through a firewall, the device checks for matches to any of the packet filtering rules that are configured in the firewall and drops or rejects the packet accordingly. When the packet passes through the firewall it filters the packet on a protocol/port number basis (GSS). For example if a rule in

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the firewall exists to block telnet access, then the firewall will block the IP protocol for port number 23.

Second generation: application layer

The key benefit of application layer filtering is that it can "understand" certain applications and protocols (such as File Transfer Protocol, DNS, or web browsing), and it can detect if an unwanted protocol is sneaking through on a non-standard port or if a protocol is being abused in any harmful way.

An application firewall is much more secure and reliable compared to packet filter firewalls because it works on all seven layers of the OSI reference model, from the application down to the physical Layer. This is similar to a packet filter firewall but here we can also filter information on the basis of content. The best example of an application firewall is ISA (Internet Security and Acceleration) server. An application firewall can filter higher-layer protocols such as FTP, Telnet, DNS, DHCP, HTTP, TCP, UDP and TFTP (GSS). For example, if an organization wants to block all the information related to "foo" then content filtering can be enabled on the firewall to block that particular word. Software-based firewalls are thus much slower than stateful firewalls.

Third generation: "stateful" filters

From 1989-1990 three colleagues from AT&T Bell Laboratories, Dave Presetto, Janardan Sharma, and Kshitij Nigam, developed the third generation of firewalls, calling them circuit level firewalls.

Third-generation firewalls, in addition to what first- and second-generation look for, regard placement of each individual packet within the packet series. This technology is generally referred to as a stateful packet inspection as it maintains records of all connections passing through the firewall and is able to determine whether a packet is the start of a new connection, a part of an existing connection, or is an invalid packet. Though there is still a set of static rules in such a firewall, the state of a connection can itself be one of the criteria which trigger specific rules.

This type of firewall can help prevent attacks which exploit existing connections, or certain Denial-of-service attacks.

Types

There are several classifications of firewalls depending on where the communication is taking place, where the communication is intercepted and the state that is being traced.

Network layer and packet filters

Network layer firewalls, also called packet filters, operate at a relatively low level of the TCP/IP protocol stack, not allowing packets to pass through the firewall unless they match the established rule set. The firewall administrator may define the rules; or default rules may apply. The term "packet filter" originated in the context of BSD operating systems.

Network layer firewalls generally fall into two sub-categories, stateful and stateless. Stateful firewalls maintain context about active sessions, and use that "state information" to speed packet processing. Any existing network connection can be

described by several properties, including source and destination IP address, UDP or TCP ports, and the current stage of the connection's lifetime (including session initiation, handshaking, data transfer, or completion connection). If a packet does not match an existing connection, it will be evaluated according to the ruleset for new connections. If a packet matches an existing connection based on comparison with the firewall's state table, it will be allowed to pass without further processing.

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Stateless firewalls require less memory, and can be faster for simple filters that require less time to filter than to look up a session. They may also be necessary for filtering stateless network protocols that have no concept of a session. However, they cannot make more complex decisions based on what stage communications between hosts *have reached.*

Modern firewalls can filter traffic based on many packet attributes like source IP address, source port, destination IP address or port, destination service like WWW or FTP. They can filter based on protocols, TTL values, netblock of originator, of the source, and many other attributes.

Commonly used packet filters on various versions of Unix are ipf (various), ipfw (FreeBSD/Mac OS X), pf (OpenBSD, and all other BSDs), iptables/ipchains (Linux).

Application-layer

Application-layer firewalls work on the application level of the TCP/IP stack (i.e., all browser traffic, or all telnet or ftp traffic), and may intercept all packets traveling to or from an application. *They block other packets (usually dropping them without* acknowledgment to the sender). In principle, application firewalls can prevent all unwanted outside traffic from reaching protected machines.

On inspecting all packets for improper content, firewalls can restrict or prevent outright the spread of networked computer worms and trojans. The additional inspection criteria can add extra latency to the forwarding of packets to their destination.

Proxies

A proxy device (running either on dedicated hardware or as software on a general-purpose machine) may act as a firewall by responding to input packets (connection requests, for example) in the manner of an application, whilst blocking other packets.

Proxies make tampering with an internal system from the external network more difficult and misuse of one internal system would not necessarily cause a security breach exploitable from outside the firewall (as long as the application proxy remains intact and properly configured). Conversely, intruders may hijack a publicly-reachable system and use it as a proxy for their own purposes; the proxy then masquerades as that system to other internal machines. While use of internal address spaces enhances security, crackers may still employ methods such as IP spoofing to attempt to pass packets to a target network.

Network address translation

Firewalls often have network address translation (NAT) functionality, and the hosts protected behind a firewall commonly have addresses in the "private address range", as defined in RFC 1918. Firewalls often have such functionality to hide the true address of protected hosts. Originally, the NAT function was developed to address the limited number of IPv4 routable addresses that could be used or assigned to

companies or individuals as well as reduce both the amount and therefore cost of obtaining enough public addresses for every computer in an organization. Hiding the addresses of protected devices has become an increasingly important defense against network reconnaissance.

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5.13 IP ADDRESS

An Internet Protocol (IP) address is a numerical label that is assigned to devices participating in a computer network, that uses the Internet Protocol for communication between its nodes. 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 TCP/IP defined an IP address as a 32-bit number and this system, known as Internet Protocol Version 4 or IPv4, is still in use today. However, due to the enormous growth of the Internet and the predicted depletion of available addresses, a new addressing system (IPv6), using 128 bits for the address, was developed in 1995 and standardized by RFC 2460 in 1998. Although IP addresses are stored as binary numbers, they are usually displayed in human-readable notations, such as 208.77.188.166 (for IPv4), and 2001:db8:0:1234:0:567:1:1 (for IPv6).

The Internet Protocol is used to route data packets between networks; IP addresses specify the locations of the source and destination nodes in the topology of the routing system. For this purpose, some of the bits in an IP address are used to designate a subnetwork. The number of these bits is indicated in CIDR notation, appended to the IP address; e.g., 208.77.188.166/24.

As the development of private networks raised the threat of IPv4 address exhaustion, RFC 1918 set aside a group of private address spaces that may be used by anyone on private networks. They are often used with network address translators to connect to the global public Internet.

The Internet Assigned Numbers Authority (IANA), which manages the IP address space allocations globally, cooperates with five Regional Internet Registries (RIRs) to allocate IP address blocks to Local Internet Registries (Internet service providers) and other entities.

Static vs dynamic IP addresses

When a computer is configured to use the same IP address each time it powers up, this is known as a static IP address. In contrast, in situations when the computer's IP address is assigned automatically, it is known as a dynamic IP address.

Method of assignment

Static IP addresses are manually assigned to a computer by an administrator. The exact procedure varies according to platform. This contrasts with dynamic IP addresses, which are assigned either by the computer interface or host software itself, as in Zeroconf, or assigned by a server using Dynamic Host Configuration Protocol (DHCP). Even though IP addresses assigned using DHCP may stay the same for long periods of time, they can generally change. In some cases, a network administrator may implement dynamically assigned static IP addresses. In this case, a DHCP server

is used, but it is specifically configured to always assign the same IP address to a particular computer. This allows static IP addresses to be configured centrally, without having to specifically configure each computer on the network in a manual procedure.

In the absence or failure of static or stateful (DHCP) address configurations, an operating system may assign an IP address to a network interface using state-less autoconfiguration methods, such as Zeroconf.

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Uses of dynamic addressing

Dynamic IP addresses are most frequently assigned on LANs and broadband networks by Dynamic Host Configuration Protocol (DHCP) servers. They are used because it avoids the administrative burden of assigning specific static addresses to each device on a network. It also allows many devices to share limited address space on a network if only some of them will be online at a particular time. In most current desktop operating systems, dynamic IP configuration is enabled by default so that a user does not need to manually enter any settings to connect to a network with a DHCP server. DHCP is not the only technology used to assign dynamic IP addresses. Dialup and some broadband networks use dynamic address features of the Point-to-Point Protocol.

Sticky dynamic IP address

A sticky dynamic IP address or sticky IP is an informal term used by cable and DSL Internet access subscribers to describe a dynamically assigned IP address that does not change often. The addresses are usually assigned with the DHCP protocol. Since the modems are usually powered-on for extended periods of time, the address leases are usually set to long periods and simply renewed upon expiration. If a modem is turned off and powered up again before the next expiration of the address lease, it will most likely receive the same IP address.

Address autoconfiguration

RFC 3330 defines an address block, 169.254.0.0/16, for the special use in link-local addressing for IPv4 networks. In IPv6, every interface, whether using static or dynamic address assignments, also receives a local-link address automatically in the fe80::/10 subnet.

These addresses are only valid on the link, such as a local network segment or point-to-point connection, that a host is connected to. These addresses are not routable and like private addresses cannot be the source or destination of packets traversing the Internet.

When the link-local IPv4 address block was reserved, no standards existed for mechanisms of address autoconfiguration. Filling the void, Microsoft created an implementation that called Automatic Private IP Addressing (APIPA). Due to Microsoft's market power, APIPA has been deployed on millions of machines and has, thus, become a de facto standard in the industry. Many years later, the IETF defined a formal standard for this functionality, RFC 3927, entitled Dynamic Configuration of IPv4 Link-Local Addresses.

Uses of static addressing

Some infrastructure situations have to use static addressing, such as when finding the Domain Name System host that will translate domain names to IP addresses. Static addresses are also convenient, but not absolutely necessary, to locate servers inside

an enterprise. An address obtained from a DNS server comes with a time to live, or caching time, after which it should be looked up to confirm that it has not changed. Even static IP addresses do change as a result of network administration (RFC 2072)

Modifications to IP addressing

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Firewalls are common on today's Internet. For increased network security, they control access to private networks based on the public IP of the client. Whether using a blacklist or a whitelist, the IP address that is blocked is the perceived public IP address of the client, meaning that if the client is using a proxy server or NAT, blocking one IP address might block many individual people.

IP address translation

Multiple client devices can appear to share IP addresses: either because they are part of a shared hosting web server environment or because an IPv4 network address translator (NAT) or proxy server acts as an intermediary agent on behalf of its customers, in which case the real originating IP addresses might be hidden from the server receiving a request. A common practice is to have a NAT hide a large number of IP addresses in a private network. Only the "outside" interface(s) of the NAT need to have Internet-routable addresses.

Most commonly, the NAT device maps TCP or UDP port numbers on the outside to individual private addresses on the inside. Just as a telephone number may have site-specific extensions, the port numbers are site-specific extensions to an IP address.

In small home networks, NAT functions usually take place in a residential gateway device, typically one marketed as a "router". In this scenario, the computers connected to the router would have 'private' IP addresses and the router would have a 'public' address to communicate with the Internet. This type of router allows several computers to share one public IP address.

5.14 INTERNET CONTROL PROTOCOLS

The Internet Control Message Protocol (ICMP) is one of the core protocols of the Internet Protocol Suite. It is chiefly used by the operating systems of networked computers to send error messages—indicating, for instance, that a requested service is not available or that a host or router could not be reached.

ICMP relies on IP to perform its tasks, and it is an integral part of IP. It differs in purpose from transport protocols such as TCP and UDP in that it is typically not used to send and receive data between end systems. It is usually not used directly by user network applications, with some notable exceptions being the ping tool and traceroute.

ICMP for Internet Protocol version 4 (IPv4) is also known as ICMPv4. IPv6 has a similar protocol, ICMPv6.

Technical details

Internet Control Message Protocol is part of the Internet Protocol Suite as defined in RFC 792. ICMP messages are typically generated in response to errors in IP datagrams (as specified in RFC 1122) or for diagnostic or routing purposes.

ICMP messages are constructed at the IP layer, usually from a normal IP datagram that has generated an ICMP response. IP encapsulates the appropriate ICMP message with a new IP header (to get the ICMP message back to the original sending host) and transmits the resulting datagram in the usual manner.

For example, every machine (such as an intermediate router) that forwards an IP datagram has to decrement the time to live (TTL) field of the IP header by one; if the TTL reaches 0, an ICMP Time to live exceeded in transit message is sent to the source of the datagram.

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Each ICMP message is encapsulated directly within a single IP datagram, and thus, like UDP, ICMP is unreliable.

Although ICMP messages are contained within standard IP datagrams, ICMP messages are usually processed as a special case, distinguished from normal IP processing, rather than processed as a normal sub-protocol of IP. In many cases, it is necessary to inspect the contents of the ICMP message and deliver the appropriate error message to the application that generated the original IP packet, the one that prompted the sending of the ICMP message.

Many commonly-used network utilities are based on ICMP messages. The traceroute command is implemented by transmitting UDP datagrams with specially set IP TTL header fields, and looking for ICMP Time to live exceeded in transit (above) and "Destination unreachable" messages generated in response. The related ping utility is implemented using the ICMP "Echo request" and "Echo reply" messages.

SUMMARY

1. The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer.
2. The network layer provides services to the Transport Layer at the network layer/transport layer interface.
3. There are basically two ways of looking at the organizing the subnet, one using connections and the other working connectionless.
4. Open **Shortest Path** First (OSPF) is a dynamic routing protocol for use in Internet Protocol (IP) networks.
5. Flooding is a static algorithm in which every incoming packet is sent out on every outgoing line except the one it arrived on. Flooding obviously generates vast numbers of duplicate packets.
6. A static algorithm that uses both topology and load for routing, is called flow-based routing. In some networks, the mean data flow between each pair of nodes is relatively stable and predictable.
7. Sending a packet to all destinations simultaneously is called Broadcasting; various methods have been proposed for doing it.
8. Congestion control concerns controlling traffic entry into a telecommunications network, so as to avoid congestive collapse by attempting to avoid oversubscription of any of the processing or link capabilities of the intermediate nodes and networks and taking resource reducing steps, such as reducing the rate of sending packets.
9. There is no motion of a virtual circuit at all in the network layer, let alone a concatenation of them.
10. The WAN can be seen as a big tunnel extending from one multiprotocol router to the other. The IP packet just travesl from one end of the tunnel to the other, snug in its nice box. It does not have to worry about dealing with the WAN at all.

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11. In computer storage, fragmentation is a phenomenon in which storage space is used inefficiently, reducing storage capacity and in most cases performance.
12. External fragmentation is the phenomenon in which free storage becomes divided into many small pieces over time.
13. Data fragmentation occurs when a piece of data in memory is broken up into many pieces that are not close together. It is typically the result of attempting to insert a large object into storage that has already suffered external fragmentation.
14. A firewall is a part of a computer system or network that is designed to block unauthorized access while permitting authorized communications. It is a device or set of devices which is configured to permit or deny computer applications based upon a set of rules and other criteria.
15. Network layer firewalls, also called packet filters, operate at a relatively low level of the TCP/IP protocol stack, not allowing packets to pass through the firewall unless they match the established rule set.
16. Application-layer firewalls work on the application level of the TCP/IP stack (i.e., all browser traffic, or all telnet or ftp traffic), and may intercept all packets traveling to or from an application.
17. An Internet Protocol (IP) address is a numerical label that is assigned to devices participating in a computer network, that uses the Internet Protocol for communication between its nodes.
18. Dynamic IP addresses are most frequently assigned on LANs and broadband networks by Dynamic Host Configuration Protocol (DHCP) servers.
19. Some infrastructure situations have to use static addressing, such as when finding the Domain Name System host that will translate domain names to IP addresses.
20. The Internet Control Message Protocol (ICMP) is one of the core protocols of the Internet Protocol Suite. It is chiefly used by the operating systems of networked computers to send error messages—indicating, for instance, that a requested service is not available or that a host or router could not be reached.

SELF ASSESSMENT QUESTIONS

1. Describe The Network Layer.
2. What is the Network Layer Design issue?
3. Describe Path Routing.
4. Describe Flooding.
5. Describe Flow Based and Broadcast Routing.
6. What do you understand by Congestion Control and Prevention Policies?
7. Describe the working of Interent.
8. What do you understand by Connectionless Internet Working?
9. What is Fragmentation?
10. Describe Firewalls.
11. What is IP address?
12. Describe the various Internet Control Protocols.

Short Questions with Answers

1. What is Network Layer?

Ans. The Network layer provides the functional and procedural means of transferring variable length data sequences from a source to a destination via one or more networks while maintaining the quality of service requested by the Transport layer. The Network layer performs network routing functions, and might also perform fragmentation and reassembly, and report delivery errors. Routers operate at this layer—sending data throughout the extended network and making the Internet possible. This is a logical addressing scheme – values are chosen by the network engineer.

2. What sort of services are provided by the network layer?

Ans. The network layer services have been designed with the following goals in mind.

1. The services should be independent of the subnet technology.
2. The transport layer should be shielded from the number, type, and topology of the subnets present.
3. The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.

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3. What is Open Shortes Path First?

Ans. Open **Shortest Path** First (OSPF) is a dynamic routing protocol for use in Internet Protocol (IP) networks. Specifically, it is a link-state routing protocol and falls into the group of interior gateway protocols, operating within a single autonomous system (AS). It is defined as OSPF Version 2 in RFC 2328 (1998) for IPv4. The updates for IPv6 are specified as OSPF Version 3 in RFC 5340 (2008).

OSPF is perhaps the most widely-used interior gateway protocol (IGP) in large enterprise networks; IS-IS, another link-state routing protocol, is more common in large service provider networks. The most widely-used exterior gateway protocol is the Border Gateway Protocol (BGP), the principal routing protocol between autonomous systems on the Internet.

4. What is Flooding?

Ans. Flooding is a static algorithm in which every incoming packet is sent out on every outgoing line except the one it arrived on. Flooding obviously generates vast numbers of duplicate packets. In fact, an infinite number unless some measures are taken to damp the prrocess. One such measure is to have a hop, counter contained in the header of each packet, which is decremented at each hop, with the packet being discarded when the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the subnet.

5. Describe the classification of congestion control algorithms.

Ans. Classification of congestion control algorithms

- There are many ways to classify congestion control algorithms:
- By the type and amount of feedback received from the network: Loss; delay; single-bit or multi-bit explicit signals
- By incremental deployability on the current Internet: Only sender needs modification; sender and receiver need modification; only router needs modification; sender, receiver and routers need modification.
- By the aspect of performance it aims to improve: high bandwidth-delay product networks; lossy links; fairness; advantage to short flows; variable-rate links
- By the fairness criterion it uses: max-min, proportional, "minimum potential delay"

6. What is fragmentation?

Ans. In computer storage, fragmentation is a phenomenon in which storage space is used inefficiently, reducing storage capacity and in most cases performance. The term is also used to denote the wasted space itself.

There are three different but related forms of fragmentation: external fragmentation, internal fragmentation, and data fragmentation. Various storage allocation schemes exhibit one or more of these weaknesses. Fragmentation can be accepted in return for increase in speed or simplicity.

7. What is Data Fragmentation?

Ans. Data fragmentation occurs when a piece of data in memory is broken up into many pieces that are not close together. It is typically the result of attempting to insert a large object into storage that has already suffered external fragmentation.

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8. What is a Firewall?

Ans. A firewall is a part of a computer system or network that is designed to block unauthorized access while permitting authorized communications. It is a device or set of devices which is configured to permit or deny computer applications based upon a set of rules and other criteria.

Firewalls can be implemented in either hardware or software, or a combination of both. Firewalls are frequently used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

9. What are Network layer and packet filters?

Ans. Network layer firewalls, also called packet filters, operate at a relatively low level of the TCP/IP protocol stack, not allowing packets to pass through the firewall unless they match the established rule set. The firewall administrator may define the rules; or default rules may apply. The term "packet filter" originated in the context of BSD operating systems.

Further Readings

1. Computer Networks: Ajit Kumar Singh, Firewal Media.
2. Data and Computer Network Communication: Prof. Shashi Banzai, Firewal Media.
3. TCP / IP and Distributed System: Vivek Archarya, Firewal Media.
4. Networking: Balvir Singh, Firewal Media.

UNIT 6

THE TRANSPORTATION LAYER

NOTES

STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- understand about the Transportation Layer of networking.
- *explain about the Transport Services it provides.*
- know about the various Transport Protocols.
- understand what is meant by Addressing.
- explain about Establishing a Connection.
- know about Establishing and Releasing a Connection.
- explain about the Internet Transport Protocols.
- understand about Transmission Control Protocol (TCP).

6.1 THE TRANSPORTATION LAYER

In computer networking, the Transport Layer provides "end-to-end communication services for applications." It is a group of methods and protocols within a layered

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architecture of network components and protocols, providing such services as connection-oriented data stream support, reliability, flow-control, and error-correction.

Transport layers are contained in both the TCP/IP model (RFC 1122), which is the foundation of the Internet, and the Open Systems Interconnection (OSI) model of general networking.

The definitions of the Transport Layer are slightly different in these two models. This article primarily refers to the TCP/IP model, in which TCP is largely for a convenient application programming interface to internet hosts. See also the OSI model definition of the Transport Layer.

The most well-known transport protocol is the Transmission Control Protocol (TCP). It lent its name to the title of the entire Internet Protocol Suite, TCP/IP. It is used for connection-oriented transmissions, whereas the connectionless

User Datagram Protocol (UDP) is used for simpler messaging transmissions. TCP is the more complex protocol, due to its stateful design incorporating reliable transmission services. Other prominent protocols in this group are the Datagram Congestion Control Protocol (DCCP) and the Stream Control Transmission Protocol (SCTP).

The Transport Layer is responsible for delivering data to the appropriate application process on the host computers. This involves statistical multiplexing of data from different application processes, i.e. forming data packets, and adding source and destination port numbers in the header of each Transport Layer data packet.

Together with the source and destination IP address, the port numbers constitutes a network socket, i.e. an identification address of the process-to-process communication. In the OSI model, this function is supported by the Session Layer.

Some Transport Layer protocols, for example TCP, but not UDP, support virtual circuits, i.e. provide connection oriented communication over an underlying packet oriented datagram network.

A byte-stream is delivered while hiding the packet mode communication for the application processes. This involves connection establishment, dividing of the data stream into packets called segments, segment numbering and reordering of out-of order data.

Finally, some Transport Layer protocols, for example TCP, but not UDP, provide end-to-end reliable communication, i.e. error recovery by means of error detecting code and automatic repeat request (ARQ) protocol. The ARQ protocol also provides flow control, which may be combined with congestion avoidance.

UDP is a very simple protocol, and does not provide virtual circuits, nor reliable communication, delegating these functions to the application program. UDP packets are called datagrams, rather than segments.

TCP is used for many protocols, including HTTP web browsing and email transfer. UDP may be used for multicasting and broadcasting, since retransmissions are not possible to a large amount of hosts.

UDP typically gives higher throughput and shorter latency, and is therefore often used for real-time multimedia communication where packet loss occasionally can be accepted, for example IP-TV and IP-telephony, and for online computer games.

In many non-IP-based networks, for example X.25, Frame Relay and ATM, the connection oriented communication is implemented at network layer or data link layer rather than the Transport Layer. In X.25, in telephone network modems and in wireless communication systems, reliable node-to-node communication is implemented at lower protocol layers.

The OSI model defines five classes of transport protocols: TP0, providing the least error recovery, to TP4, which is designed for less reliable networks.

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6.2 THE TRANSPORT SERVICES

There is a long list of services that can be optionally provided by the Transport Layer. None of them are compulsory, because not all applications require all available services.

Connection-oriented: Interpreting the connection as a data stream can provide many benefits. It is normally easier to deal with than connection-less models, so where the Network layer only provides a connection-less service, often a connection-oriented service is built on top of that in the Transport Layer.

Same Order Delivery: The Network layer doesn't generally guarantee that packets of data will arrive in the same order that they were sent, but often this is a desirable feature, so the Transport Layer provides it. The simplest way of doing this is to give each packet a number, and allow the receiver to reorder the packets.

Reliability: Packets may be lost in routers, switches, bridges and hosts due to network congestion, when the packet queues are filled and the network nodes have to delete packets. Packets may be lost or corrupted in Ethernet due to interference and noise, since Ethernet does not retransmit corrupted packets. Packets may be delivered in the wrong order by an underlying network. Some Transport Layer protocols, for example TCP, can fix this. By means of an error detection code, for example a checksum, the transport protocol may check that the data is not corrupted, and verify that by sending an ACK message to the sender. Automatic repeat request schemes may be used to retransmit lost or corrupted data. By introducing segment numbering in the Transport Layer packet headers, the packets can be sorted in order. Of course, error free is impossible, but it is possible to substantially reduce the numbers of undetected errors.

Flow control: The amount of memory on a computer is limited, and without flow control a larger computer might flood a computer with so much information that it can't hold it all before dealing with it. Nowadays, this is not a big issue, as memory is cheap while bandwidth is comparatively expensive, but in earlier times it was more important. Flow control allows the receiver to respond before it is overwhelmed. Sometimes this is already provided by the network, but where it is not, the Transport Layer may add it on.

Congestion avoidance: Network congestion occurs when a queue buffer of a network node is full and starts to drop packets. Automatic repeat request may keep the network in a congested state. This situation can be avoided by adding congestion avoidance to the flow control, including slow-start. This keeps the bandwidth consumption at a low level in the beginning of the transmission, or after packet retransmission.

Byte orientation: Rather than dealing with things on a packet-by-packet basis, the

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Transport Layer may add the ability to view communication just as a stream of bytes. This is nicer to deal with than random packet sizes, however, it rarely matches the communication model which will normally be a sequence of messages of user defined sizes.

Ports: Ports provide multiplexing. For example, the first line of a postal address is a kind of port, and distinguishes between different occupants of the same house. Computer applications will each listen for information on their own ports, which is why you can use more than one network-based application at the same time. It is part of the Transport Layer in the TCP/IP model, but of the Session Layer in the OSI model.

6.3 TRANSPORT PROTOCOLS

The transport service is implemented by a transport protocol used between the two transport entities. For one thing, in the data link layer, it is not necessary for a router to specify which router it wants to talk to — each outgoing line uniquely specifies a particular router. In the transport layer, explicit addressing of destinations is required.

Another major difference between the data link layer and the transport layer is the potential existence of storage capacity in the subnet. When a router sends aframe, it may arrive or to be lost, but it cannot bounce around for a while, go into hiding in a far corner of the world, and then suddenly emerge at an inopprtune moment 30 secs later. If the subnet uses datagrams and adaptive routing sides, there is a nonneglibible probability that a packet may be stored for a number of seconds and then delivered later. The consequences of this ability of the subnet to store packets can sometimes be disastrous and require the use of special protocols.

A final difference between the data link and transport layers is one of amount rather than of kind. Buffering and flow control are needed in both layers, but the presence of a large and dynamically varying number of connections in the transport layer may require a different approach than we used in the data link layer.

6.4 ADDRESSING

When an application process wishes to set up a connection to a remote application process, it must specify which one to connect to. The method normally used is to define transport addresses to which processes can listen for connection requests. In the Internet, these end points are (IP address, local port) pairs. In ATM networks, they are AAL-SAPs. A possible connection scenario for a transport connection over a connection-oriented network layer is as follows:

1. A time-of-day server process on host 2 attaches itself to TSAP 122 to wait for incoming call. How a process attaches itself to a TSAP is outside the networking model and depends entirely on the local operating system. A call such as our LISTEN might be used, for example.
2. An application process on host 1 want to find out the time-of-day, so it issues a CONNECT request specifying TSAP 6 as the source and TSAP 122 as the destination.
3. The transport entity on host 1 selects a network address on its machine (if it

has more than one) and sets up a network connection between them. (With a connectionless subent, establishing this network layer connection would not be done.) Using this network connection, host 1's transport entity can talk to the transport entity on host 2.

4. The first thing the transport entity on 1 says to its peer on 2 is: "Good morning, I would like to establish a transport connection between any TSAP 6 and your TSAP 122. What do you say"
5. The transport entity on 2 then asks the time-of-day server at TSAP 122 if it wiling ato accept a new connection. If it agres, the transport connection is established.

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6.5 ESTABLISHING A CONNECTION

Imagine a subnet that is so congested that acknowledgement hardly ever get back in time, and each packet times out and is retransmitted two or three times. Suppose that the subnet uses datagrams inside, and every packet follows a different route. Some of the packets might get stuck in a traffic jam inside the subnet and take a long time to arrive, that is, they are stored in the subnet and pop out much later. The main problem is the existence of delayed duplicates. It can be attacked in various ways, none of them very satisfactory. One way is to use throwaway transport addresses. In this approach, each time a transport address is needed; a new one is generated. When a connecion is released, the address is discarded.

Another possibility is to give each connection a connection identifier (i.e., a sequence number incremented for each connection established), chosen by the initiating party, and put in each TPDU, including the one requesting the connection. After each connection is released, each transport entity could update a table listing obsolete connections as (peer transport entity, connection identifier) pairs. Whenever a connection reuqest came in, it could be checked against the table, to see if it belonged to a previously released connection. But, it has a flaw: it requires each transport entity to maintain a certain amount of history information indefinitely. If a machine crashes and loses its memory, it will no longer know which connection identifiers have already been used.

6.6 RELEASING A CONNECTION

It is easier than establishing one. Nevertheless, there are more pitfalls than one can expect. There are two styles of terminating a connection; asymmetric release and symmetric lease. Asymmetric relase is the way the telephone system works: when on party hangs up, the connection is broken. Symmetric release treats the connection as two separate unidirectional connections and requires each one to be released separately. Asymmetric relase is abrupt and may result in data loss. After the connection is established, host 1 sends a TPDU that arrives properly at host 2. Then host 1 sends another TPDU. Unfortunately, host 2 issues a DISCONNECT before the second TPDU arrives. The result is that the connection is released and data are lost.

Clearly, a more sophisticated release protocol is required to avoid data loss. One way is to use symmetric release, in which each direction is released independently of the

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other line. Gere a host can continue to receive data even after it has sent a DISCONNECT TPDU. Symmetric release does the job when each process has a fixed amount of data to send and clearly knows when it has sent it. In other siuations, determining that al the work has been done and the connection should be terminated is not so obvious. One can envision a protocol in which host 1 says; "I am done. Are you don too?" If host 2 responds: "I am done too. Goodbye." the connection can be safely released.

6.7 THE INTERNET TRANSPORT PROTOCOLS

The Internet has two main protocols in the transport layer, a connection-oriented protocol and a connectionless one. In the following section, we would study the most important one, i.e., Transmission Control Protocol (TCP).

6.8 TCP

The Transmission Control Protocol (TCP) is one of the core protocols of the Internet Protocol Suite. TCP is one of the two original components of the suite (the other being Internet Protocol, or IP), so the entire suite is commonly referred to as TCP/IP. Whereas IP handles lower-level transmissions from computer to computer as a message makes its way across the Internet, TCP operates at a higher level, concerned only with the two end systems, for example a Web browser and a Web server. In particular, TCP provides reliable, ordered delivery of a stream of bytes from a program on one computer to another program on another computer. Besides the Web, other common applications of TCP include e-mail and file transfer. Among other management tasks, TCP controls segment size, flow control, and data exchange rate.

One interesting aspect of the history of the OSI Reference Model is that the original objective was not to create a model primarily for educational purposes—even though many people today think that this was the case. The OSI Reference Model was intended to serve as the foundation for the establishment of a widely-adopted suite of protocols that would be used by international internetworks—basically, what the Internet became. This was called, unsurprisingly, the OSI Protocol Suite.

However, things didn't quite work out as planned. The rise in popularity of the Internet and its TCP/IP protocols met the OSI suite head on, and in a nutshell, TCP/IP won. Some of the OSI protocols were implemented, but as a whole, the OSI protocols lost out to TCP/IP when the Internet started to grow.

TCP provides a communication service at an intermediate level between an application program and the Internet Protocol (IP). That is, when an application program desires to send a large chunk of data across the Internet using IP, instead of breaking the data into IP-sized pieces and issuing a series of IP requests, the software can issue a single request to TCP and let TCP handle the IP details.

IP works by exchanging pieces of information called packets. A packet is a sequence of bytes and consists of a header followed by a body. The header describes the packet's destination and, optionally, the routers to use for forwarding until it arrives at its final destination. The body contains the data IP is transmitting.

Due to network congestion, traffic load balancing, or other unpredictable network

behavior, IP packets can be lost, duplicated, or delivered out of order. TCP detects these problems, requests retransmission of lost packets, rearranges out-of-order packets, and even helps minimize network congestion to reduce the occurrence of the other problems. Once the TCP receiver has finally reassembled a perfect copy of the data originally transmitted, it passes that datagram to the application program. *Thus, TCP abstracts the application's communication from the underlying networking* details.

TCP is used extensively by many of the Internet's most popular applications, including the World Wide Web (WWW), E-mail, File Transfer Protocol, Secure Shell, peer-to-peer file sharing, and some streaming media applications.

TCP is optimized for accurate delivery rather than timely delivery, and therefore, TCP sometimes incurs relatively long delays (in the order of seconds) while waiting for out-of-order messages or retransmissions of lost messages. It is not particularly suitable for real-time applications such as Voice over IP. For such applications, protocols like the Real-time Transport Protocol (RTP) running over the User Datagram Protocol (UDP) are usually recommended instead.

TCP is a reliable stream delivery service that guarantees delivery of a data stream sent from one host to another without duplication or losing data. Since packet transfer is not reliable, a technique known as positive acknowledgment with retransmission is used to guarantee reliability of packet transfers. This fundamental technique requires the receiver to respond with an acknowledgment message as it receives the data. The sender keeps a record of each packet it sends, and waits for acknowledgment before sending the next packet. The sender also keeps a timer from when the packet was sent, and retransmits a packet if the timer expires. The timer is needed in case a packet gets lost or corrupted.

TCP consists of a set of rules: for the protocol, that are used with the Internet Protocol, and for the IP, to send data "in a form of message units" between computers over the Internet. At the same time that IP takes care of handling the actual delivery of the data, TCP takes care of keeping track of the individual units of data transmission, called segments, that a message is divided into for efficient routing through the network. For example, when an HTML file is sent from a Web server, the TCP software layer of that server divides the sequence of bytes of the file into segments and forwards them individually to the IP software layer (Internet Layer). The Internet Layer encapsulates each TCP segment into an IP packet by adding a header that includes (among other data) the destination IP address. Even though every packet has the same destination address, they can be routed on different paths through the *network. When the client program on the destination computer receives them, the* TCP layer (Transport Layer) reassembles the individual segments and ensures they are correctly ordered and error free as it streams them to an application.

Connection establishment

To establish a connection, TCP uses a three-way handshake. Before a client attempts to connect with a server, the server must first bind to a port to open it up for connections: this is called a passive open. Once the passive open is established, a client may initiate an active open. To establish a connection, the three-way (or 3-step) handshake occurs:

The active open is performed by the client sending a SYN to the server. It sets the segment's sequence number to a random value A.

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In response, the server replies with a SYN-ACK. The acknowledgment number is set to one more than the received sequence number (A + 1), and the sequence number that the server chooses for the packet is another random number, B.

Finally, the client sends an ACK back to the server. The sequence number is set to the received acknowledgement value i.e. A + 1, and the acknowledgement number is set to one more than the received sequence number i.e. B + 1.

At this point, both the client and server have received an acknowledgment of the connection.

Data transfer

There are a few key features that set TCP apart from User Datagram Protocol:

- Ordered data transfer - the destination host rearranges according to sequence number
- Retransmission of lost packets - any cumulative stream not acknowledged is retransmitted
- Discarding duplicate packets
- Error-free data transfer (The checksum in UDP is optional)

Flow control - limits the rate a sender transfers data to guarantee reliable delivery. The receiver continually hints the sender on how much data can be received (controlled by the sliding window). When the receiving host's buffer fills, the next acknowledgment contains a 0 in the window size, to stop transfer and allow the data in the buffer to be processed.

Reliable transmission

TCP uses a sequence number to identify each byte of data. The sequence number identifies the order of the bytes sent from each computer so that the data can be reconstructed in order, regardless of any fragmentation, disordering, or packet loss that may occur during transmission. For every payload byte transmitted the sequence number must be incremented. In the first two steps of the 3-way handshake, both computers exchange an initial sequence number (ISN). This number can be arbitrary, and should in fact be unpredictable to defend against TCP Sequence Prediction Attacks.

TCP primarily uses a cumulative acknowledgment scheme, where the receiver sends an acknowledgment signifying that the receiver has received all data preceding the acknowledged sequence number. Essentially, the first byte in a segment's data field is assigned a sequence number, which is inserted in the sequence number field, and the receiver sends an acknowledgment specifying the sequence number of the next byte they expect to receive. For example, if computer A sends 4 bytes with a sequence number of 100 (conceptually, the four bytes would have a sequence number of 100, 101, 102, & 103 assigned) then the receiver would send back an acknowledgment of 104 since that is the next byte it expects to receive in the next packet.

In addition to cumulative acknowledgments, TCP receivers can also send selective acknowledgments to provide further information (see selective acknowledgments).

If the sender infers that data has been lost in the network, it retransmits the data.

Error detection

Sequence numbers and acknowledgments cover discarding duplicate packets, retransmission of lost packets, and ordered-data transfer. To assure correctness a checksum field is included (see TCP segment structure for details on checksumming).

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The TCP checksum is a weak check by modern standards. Data Link Layers with high bit error rates may require additional link error correction/detection capabilities. The weak checksum is partially compensated for by the common use of a CRC or better integrity check at layer 2, below both TCP and IP, such as is used in PPP or the Ethernet frame. However, this does not mean that the 16-bit TCP checksum is redundant: remarkably, introduction of errors in packets between CRC-protected hops is common, but the end-to-end 16-bit TCP checksum catches most of these simple errors. This is the end-to-end principle at work.

Flow control

TCP uses an end-to-end flow control protocol to avoid having the sender send data too fast for the TCP receiver to receive and process it reliably. Having a mechanism for flow control is essential in an environment where machines of diverse network speeds communicate. For example, if a PC sends data to a hand-held PDA that is slowly processing received data, the PDA must regulate data flow so as not to be overwhelmed.

TCP uses a sliding window flow control protocol. In each TCP segment, the receiver specifies in the receive window field the amount of additional received data (in bytes) that it is willing to buffer for the connection. The sending host can send only up to that amount of data before it must wait for an acknowledgment and window update from the receiving host.

TCP sequence numbers and receive windows behave very much like a clock. The receive window shifts each time the receiver receives and acknowledges a new segment of data. Once it runs out of sequence numbers, the sequence number loops back to 0.

When a receiver advertises a window size of 0, the sender stops sending data and starts the persist timer. The persist timer is used to protect TCP from a deadlock situation that could arise if the window size update from the receiver is lost and the sender has no more data to send while the receiver is waiting for the new window size update. When the persist timer expires, the TCP sender sends a small packet so that the receiver sends an acknowledgement with the new window size.

If a receiver is processing incoming data in small increments, it may repeatedly advertise a small receive window. This is referred to as the silly window syndrome, since it is inefficient to send only a few bytes of data in a TCP segment, given the relatively large overhead of the TCP header. TCP senders and receivers typically employ flow control logic to specifically avoid repeatedly sending small segments. The sender-side silly window syndrome avoidance logic is referred to as Nagle's algorithm.

Congestion control

The final main aspect of TCP is congestion control. TCP uses a number of mechanisms to achieve high performance and avoid 'congestion collapse', where

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network performance can fall by several orders of magnitude. These mechanisms control the rate of data entering the network, keeping the data flow below a rate that would trigger collapse.

Acknowledgments for data sent, or lack of acknowledgments, are used by senders to infer network conditions between the TCP sender and receiver. Coupled with timers, TCP senders and receivers can alter the behavior of the flow of data. This is more generally referred to as congestion control and/or network congestion avoidance.

Modern implementations of TCP contain four intertwined algorithms: Slow-start, congestion avoidance, fast retransmit, and fast recovery (RFC 2581).

In addition, senders employ a retransmission timeout (RTO) that is based on the estimated round-trip time (or RTT) between the sender and receiver, as well as the variance in this round trip time. The behavior of this timer is specified in RFC 2988. There are subtleties in the estimation of RTT. For example, senders must be careful when calculating RTT samples for retransmitted packets; typically they use Karn's Algorithm or TCP timestamps (see RFC 1323). These individual RTT samples are then averaged over time to create a Smoothed Round Trip Time (SRTT) using Jacobson's algorithm. This SRTT value is what is finally used as the round-trip time estimate.

Enhancing TCP to reliably handle loss, minimize errors, manage congestion and go fast in very high-speed environments are ongoing areas of research and standards development. As a result, there are a number of TCP congestion avoidance algorithm variations.

Maximum segment size

The Maximum segment size (MSS) is the largest amount of data, specified in bytes, that TCP is willing to send in a single segment. For best performance, the MSS should be set small enough to avoid IP fragmentation, which can lead to excessive retransmissions if there is packet loss. To try to accomplish this, typically the MSS is negotiated using the MSS option when the TCP connection is established, in which case it is determined by the maximum transmission unit (MTU) size of the data link layer of the networks to which the sender and receiver are directly attached. Furthermore, TCP senders can use Path MTU discovery to infer the minimum MTU along the network path between the sender and receiver, and use this to dynamically adjust the MSS to avoid IP fragmentation within the network.

Selective acknowledgments

Relying purely on the cumulative acknowledgment scheme employed by the original TCP protocol can lead to inefficiencies when packets are lost. For example, suppose 10,000 bytes are sent in 10 different TCP packets, and the first packet is lost during transmission. In a pure cumulative acknowledgment protocol, the receiver cannot say that it received bytes 1,000 to 9,999 successfully, but failed to receive the first packet, containing bytes 0 to 999. Thus the sender may then have to resend all 10,000 bytes.

To solve this problem TCP employs the selective acknowledgment (SACK) option, defined in RFC 2018, which allows the receiver to acknowledge discontinuous blocks of packets that were received correctly, in addition to the sequence number of the last contiguous byte received successively, as in the basic TCP acknowledgment.

The acknowledgement can specify a number of SACK blocks, where each SACK block is conveyed by the starting and ending sequence numbers of a contiguous range that the receiver correctly received. In the example above, the receiver would send SACK with sequence numbers 1,000 and 9,999. The sender thus retransmits only the first packet, bytes 0 to 999.

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An extension to the SACK option is the "duplicate-SACK" option, defined in RFC 2883. An out-of-order packet delivery can often falsely indicate the TCP sender of lost packet and, in turn, the TCP sender retransmits the suspected-to-be-lost packet and slow down the data delivery to prevent network congestion. The TCP sender undoes the action of slow-down, that is a recovery of the original pace of data transmission, upon receiving a D-SACK that indicates the retransmitted packet is duplicate.

The SACK option is not mandatory and it is used only if both parties support it. This is negotiated when connection is established. SACK uses the optional part of the TCP header (see TCP segment structure for details). The use of SACK is widespread - all popular TCP stacks support it. Selective acknowledgment is also used in Stream Control Transmission Protocol (SCTP).

Window scaling

For more efficient use of high bandwidth networks, a larger TCP window size may be used. The TCP window size field controls the flow of data and its value is limited to between 2 and 65,535 bytes.

Since the size field cannot be expanded, a scaling factor is used. The TCP window scale option, as defined in RFC 1323, is an option used to increase the maximum window size from 65,535 bytes to 1 Gigabyte. Scaling up to larger window sizes is a part of what is necessary for TCP Tuning.

The window scale option is used only during the TCP 3-way handshake. The window scale value represents the number of bits to left-shift the 16-bit window size field. The window scale value can be set from 0 (no shift) to 14 for each direction independently. Both sides must send the option in their SYN segments to enable window scaling in either direction.

Some routers and packet firewalls rewrite the window scaling factor during a transmission. This causes sending and receiving sides to assume different TCP window sizes. The result is non-stable traffic that may be very slow. The problem is visible on some sending and receiving sites behind the path of defective routers.

TCP Timestamps

TCP timestamps, defined in RFC 1323, help TCP compute the round-trip time between the sender and receiver. Timestamp options include a 4-byte timestamp value, where the sender inserts its current value of its timestamp clock, and a 4-byte echo reply timestamp value, where the receiver generally inserts the most recent timestamp value that it has received. The sender uses the echo reply timestamp in an acknowledgment to compute the total elapsed time since the acknowledged segment was sent.

TCP timestamps are also used to help in the case where TCP sequence numbers encounter their 232 bound and "wrap around" the sequence number space. This scheme is known as Protect Against Wrapped Sequence numbers, or PAWS (see RFC 1323

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for details). Furthermore, the Eifel detection algorithm, defined in RFC 3522, which detects unnecessary loss recovery requires TCP timestamps.

Out of band data

One is able to interrupt or abort the queued stream instead of waiting for the stream to finish. This is done by specifying the data as urgent. This tells the receiving program to process it immediately, along with the rest of the urgent data. When finished, TCP informs the application and resumes back to the stream queue. An example is when TCP is used for a remote login session, the user can send a keyboard sequence that interrupts or aborts the program at the other end. These signals are most often needed when a program on the remote machine fails to operate correctly. The signals must be sent without waiting for the program to finish its current transfer.

TCP OOB data was not designed for the modern Internet. The urgent pointer only alters the processing on the remote host and doesn't expedite any processing on the network itself. When it gets to the remote host there are two slightly different interpretations of the protocol, which means only single bytes of OOB data are reliable. This is assuming it's reliable at all as it's one of the least commonly used protocol elements and tends to be poorly implemented.

Forcing data delivery

Normally, TCP waits for the buffer to exceed the maximum segment size before sending any data. This creates serious delays when the two sides of the connection are exchanging short messages and need to receive the response before continuing. For example, the login sequence at the beginning of a telnet session begins with the short message "Login", and the session cannot make any progress until these five characters have been transmitted and the response has been received. This process can be seriously delayed by TCP's normal behavior when the message is provided to TCP in several send calls.

However, an application can force delivery of segments to the output stream using a push operation provided by TCP to the application layer. This operation also causes TCP to set the PSH flag or control bit to ensure that data is delivered immediately to the application layer by the receiving transport layer.

In the most extreme cases, for example when a user expects each keystroke to be echoed by the receiving application, the push operation can be used each time a keystroke occurs. More generally, application programs use this function to force output to be sent after writing a character or line of characters. By forcing the data to be sent immediately, delays and wait time are reduced.

Connection termination

The connection termination phase uses, at most, a four-way handshake, with each side of the connection terminating independently. When an endpoint wishes to stop its half of the connection, it transmits a FIN packet, which the other end acknowledges with an ACK. Therefore, a typical tear-down requires a pair of FIN and ACK segments from each TCP endpoint.

A connection can be "half-open", in which case one side has terminated its end, but the other has not. The side that has terminated can no longer send any data into or receive any data from the connection, but the other side can (but generally if it tries,

this should result in no acknowledgment and therefore a timeout, or else result in a positive RST, and either way thereby the destruction of the half-open socket).

It is also possible to terminate the connection by a 3-way handshake, when host A sends a FIN and host B replies with a FIN & ACK (merely combines 2 steps into one) and host A replies with an ACK. This is perhaps the most common method.

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It is possible for both hosts to send FINs simultaneously then both just have to ACK. This could possibly be considered a 2-way handshake since the FIN/ACK sequence is done in parallel for both directions.

Some host TCP stacks may implement a "half-duplex" close sequence, as Linux or HP-UX do. If such a host actively closes a connection but still has not read all the incoming data the stack already received from the link, this host sends a RST instead of a FIN (Section 4.2.2.13 in RFC 1122). This allows a TCP application to be sure the remote application has read all the data the former sent—waiting the FIN from the remote side, when it actively closes the connection. However, the remote TCP stack cannot distinguish between a Connection Aborting RST and this Data Loss RST. Both cause the remote stack to throw away all the data it received, but that the application still didn't read.

Some application protocols may violate the OSI model layers, using the TCP open/close handshaking for the application protocol open/close handshaking - these may find the RST problem on active close. As an example:

```
s = connect(remote);
send(s, data);
close(s);
```

For a usual program flow like above, a TCP/IP stack like that described above does not guarantee that all the data arrives to the other application unless the programmer is sure that the remote side will not send anything.

Denial of service

By using a spoofed IP address and repeatedly sending purposely assembled SYN packets, attackers can cause the server to consume large amounts of resources keeping track of the bogus connections. This is known as a SYN flood attack. Proposed solutions to this problem include SYN cookies and Cryptographic puzzles. Sockstress is a similar attack, against which no defense is yet known. An advanced DoS attack involving the exploitation of the TCP Persist Timer was analyzed at Phrack #66.

Connection hijacking

An attacker who is able to eavesdrop a TCP session and redirect packets can hijack a TCP connection. To do so, the attacker learns the sequence number from the ongoing communication and forges a false segment that looks like the next segment in the stream. Such a simple hijack can result in one packet being erroneously accepted at one end. When the receiving host acknowledges the extra segment to the other side of the connection, synchronization is lost. Hijacking might be combined with ARP or routing attacks that allow taking control of the packet flow, so as to get permanent control of the hijacked TCP connection.

Impersonating a different IP address was possible prior to RFC 1948, when the initial sequence number was easily guessable. That allowed an attacker to blindly send a

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sequence of packets that the receiver would believe to come from a different IP address, without the need to deploy ARP or routing attacks: it is enough to ensure that the legitimate host of the impersonated IP address is down, or bring it to that condition using denial of service attacks. This is why the initial sequence number is chosen at random.

TCP ports

TCP uses the notion of port numbers to identify sending and receiving application end-points on a host, or Internet sockets. Each side of a TCP connection has an associated 16-bit unsigned port number (0-65535) reserved by the sending or receiving application. Arriving TCP data packets are identified as belonging to a specific TCP connection by its sockets, that is, the combination of source host address, source port, destination host address, and destination port. This means that a server computer can provide several clients with several services simultaneously, as long as a client takes care of initiating any simultaneous connections to one destination port from different source ports.

Port numbers are categorized into three basic categories: well-known, registered, and dynamic/private. The well-known ports are assigned by the Internet Assigned Numbers Authority (IANA) and are typically used by system-level or root processes. Well-known applications running as servers and passively listening for connections typically use these ports. Some examples include: FTP (21), SSH (22), TELNET (23), SMTP (25) and HTTP (80). Registered ports are typically used by end user applications as ephemeral source ports when contacting servers, but they can also identify named services that have been registered by a third party. Dynamic/private ports can also be used by end user applications, but are less commonly so. Dynamic/private ports do not contain any meaning outside of any particular TCP connection.

Development

TCP is a complex protocol. However, while significant enhancements have been made and proposed over the years, its most basic operation has not changed significantly since its first specification RFC 675 in 1974, and the v4 specification RFC 793, published in September 1981. RFC 1122, Host Requirements for Internet Hosts, clarified a number of TCP protocol implementation requirements. RFC 2581, TCP Congestion Control, one of the most important TCP-related RFCs in recent years, describes updated algorithms that avoid undue congestion. In 2001, RFC 3168 was written to describe explicit congestion notification (ECN), a congestion avoidance signalling mechanism.

The original TCP congestion avoidance algorithm was known as "TCP Tahoe", but many alternative algorithms have since been proposed (including TCP Reno, TCP Vegas, FAST TCP, TCP New Reno, and TCP Hybla).

TCP Interactive (iTCP) is a research effort into TCP extensions that allows applications to subscribe to TCP events and register handler components that can launch applications for various purposes, including application assisted congestion control.

Multipath TCP (MPTCP) is a another research effort attempting to utilize multiple path for one TCP connection, thus maximizing resource usage and increasing redundancy.

TCP over wireless networks

TCP has been optimized for wired networks. Any packet loss is considered to be the result of congestion and the congestion window size is reduced dramatically as a precaution. However, wireless links are known to experience sporadic and usually temporary losses due to fading, shadowing, hand off, and other radio effects, that cannot be considered congestion. After the (erroneous) back-off of the congestion window size, due to wireless packet loss, there can be a congestion avoidance phase with a conservative decrease in window size. This causes the radio link to be underutilized. Extensive research has been done on the subject of how to combat these harmful effects. Suggested solutions can be categorized as end-to-end solutions (which require modifications at the client and/or server), link layer solutions (such as RLP in CDMA2000), or proxy based solutions (which require some changes in the network without modifying end nodes).

Hardware implementations

One way to overcome the processing power requirements of TCP is to build hardware implementations of it, widely known as TCP Offload Engines (TOE). The main problem of TOEs is that they are hard to integrate into computing systems, requiring extensive changes in the operating system of the computer or device. One company to develop such a device was Alacritech.

Debugging

A packet sniffer, which intercepts TCP traffic on a network link, can be useful in debugging networks, network stacks and applications that use TCP by showing the user what packets are passing through a link. Some networking stacks support the SO_DEBUG socket option, which can be enabled on the socket using setsockopt. That option dumps all the packets, TCP states, and events on that socket, which is helpful in debugging. Netstat is another utility that can be used for debugging.

Alternatives

For many applications TCP is not appropriate. One big problem (at least with normal implementations) is that the application cannot get at the packets coming after a lost packet until the retransmitted copy of the lost packet is received. This causes problems for real-time applications such as streaming multimedia (such as Internet radio), real-time multiplayer games and voice over IP (VoIP) where it is sometimes more useful to get most of the data in a timely fashion than it is to get all of the data in order.

For both historical and performance reasons, most storage area networks (SANs) prefer to use Fibre Channel protocol (FCP) instead of TCP/IP.

Also for embedded systems, network booting and servers that serve simple requests from huge numbers of clients (e.g. DNS servers) the complexity of TCP can be a problem. Finally some tricks such as transmitting data between two hosts that are both behind NAT (using STUN or similar systems) are far simpler without a relatively complex protocol like TCP in the way.

Generally where TCP is unsuitable the User Datagram Protocol (UDP) is used. This provides the application multiplexing and checksums that TCP does, but does not handle building streams or retransmission giving the application developer the ability to code those in a way suitable for the situation and/or to replace them with other methods like forward error correction or interpolation.

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SCTP is another IP protocol that provides reliable stream oriented services not so dissimilar from TCP. It is newer and considerably more complex than TCP, and has not yet seen widespread deployment. However, it is especially designed to be used in situations where reliability and near-real-time considerations are important.

Venturi Transport Protocol (VTP) is a patented proprietary protocol that is designed to replace TCP transparently to overcome perceived inefficiencies related to wireless data transport.

TCP also has issues in high bandwidth environments. The TCP congestion avoidance algorithm works very well for ad-hoc environments where the data sender is not known in advance, but if the environment is predictable, a timing based protocol such as Asynchronous Transfer Mode (ATM) can avoid TCP's retransmits overhead.

Multipurpose Transaction Protocol (MTP/IP) is patented proprietary software that is designed to adaptively achieve high throughput and transaction performance in a wide variety of network conditions, particularly those where TCP is perceived to be inefficient.

SUMMARY

1. In computer networking, the Transport Layer provides "end-to-end communication services for applications."
2. The transport service is implemented by a transport protocol used between the two transport entities.
3. When an application process wishes to set up a connection to a remote application process, it must specify which one to connect to.
4. Asymmetric relase is the way the telephone system works: when on party hangs up, the connection is broken. Symmetric release treats the connection as two separate unidirectional connections and requires each one to be released separately.
5. The Transmission Control Protocol (TCP) is one of the core protocols of the Internet Protocol Suite.
6. TCP is used extensively by many of the Internet's most popular applications, including the World Wide Web (WWW), E-mail, File Transfer Protocol, Secure Shell, peer-to-peer file sharing, and some streaming media applications.
7. TCP uses a sequence number to identify each byte of data.
8. The TCP checksum is a weak check by modern standards.
9. TCP uses a number of mechanisms to achieve high performance and avoid 'congestion collapse', where network performance can fall by several orders of magnitude.
10. The Maximum segment size (MSS) is the largest amount of data, specified in bytes, that TCP is willing to send in a single segment.
11. The window scale option is used only during the TCP 3-way handshake. The window scale value represents the number of bits to left-shift the 16-bit window size field.
12. TCP timestamps are also used to help in the case where TCP sequence numbers encounter their 232 bound and "wrap around" the sequence number space.
13. Normally, TCP waits for the buffer to exceed the maximum segment size before sending any data.
14. The connection termination phase uses, at most, a four-way handshake, with each side of the connection terminating independently.
15. TCP has been optimized for wired networks. Any packet loss is considered to be the result of congestion and the congestion window size is reduced dramatically as a precaution.

SELF ASSESSMENT QUESTIONS

NOTES

1. Describe The Transportation Layer.
2. What sort of services are provided by Transport Layers?
3. Describe the various Transport protocols..
4. Describe Addressing..
5. How would you establish a connection?
6. How would you release a connection?
7. Describe the various Internet Transport Protocols.
8. Describe TCP in details.

Short Questions with Answers

1. What is Transport Layer?

Ans. In computer networking, the Transport Layer provides "end-to-end communication services for applications." It is a group of methods and protocols within a layered architecture of network components and protocols, providing such services as connection-oriented data stream support, reliability, flow-control, and error-correction.

2. What is Flow Control?

Ans. The amount of memory on a computer is limited, and without flow control a larger computer might flood a computer with so much information that it can't hold it all before dealing with it. Nowadays, this is not a big issue, as memory is cheap while bandwidth is comparatively expensive, but in earlier times it was more important. Flow control allows the receiver to respond before it is overwhelmed. Sometimes this is already provided by the network, but where it is not, the Transport Layer may add it on.

3. What are Ports?

Ans. Ports provide multiplexing. For example, the first line of a postal address is a kind of port, and distinguishes between different occupants of the same house. Computer applications will each listen for information on their own ports, which is why you can use more than one network-based application at the same time. It is part of the Transport Layer in the TCP/IP model, but of the Session Layer in the OSI model.

4. How would you establish a connection?

Ans. Imagine a subnet that is so congested that acknowledgement hardly ever get back in time, and each packet times out and is retransmitted two or three times. Suppose that the subnet uses datagrams inside, and every packet follows a different route. Some of the packets might get stuck in a traffic jam inside the subnet and take a long time to arrive, that is, they are stored in the subnet and pop out much later.

The main problem is the existence of delayed duplicates. It can be attacked in various ways, none of them very satisfactory. One way is to use throwaway transport addresses. In this approach, each time a transport address is needed; a new one is generated. When a connecion is released, the address is discarded.

5. Describe TCP.

Ans. The Transmission Control Protocol (TCP) is one of the core protocols of the Internet Protocol Suite. TCP is one of the two original components of the suite (the other being Internet Protocol, or IP), so the entire suite is commonly referred to as TCP/IP. Whereas IP handles lower-level transmissions from computer to computer as a message makes its way across the Internet, TCP operates at a higher level, concerned only with the two end systems, for example a Web browser and a Web server. In particular, TCP provides reliable, ordered delivery of a stream of bytes from a program on one computer to another program on another computer. Besides the Web, other common applications of TCP include e-mail and file transfer. Among other management tasks, TCP controls segment size, flow control, and data exchange rate.

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6. What is the sequence order of TCP?

Ans. TCP uses a sequence number to identify each byte of data. The sequence number identifies the order of the bytes sent from each computer so that the data can be reconstructed in order, regardless of any fragmentation, disordering, or packet loss that may occur during transmission. For every payload byte transmitted the sequence number must be incremented. In the first two steps of the 3-way handshake, both computers exchange an initial sequence number (ISN). This number can be arbitrary, and should in fact be unpredictable to defend against TCP Sequence Prediction Attacks.

7. What is Maximum Segment Size?

Ans. The Maximum segment size (MSS) is the largest amount of data, specified in bytes, that TCP is willing to send in a single segment. For best performance, the MSS should be set small enough to avoid IP fragmentation, which can lead to excessive retransmissions if there is packet loss. To try to accomplish this, typically the MSS is negotiated using the MSS option when the TCP connection is established, in which case it is determined by the maximum transmission unit (MTU) size of the data link layer of the networks to which the sender and receiver are directly attached. Furthermore, TCP senders can use Path MTU discovery to infer the minimum MTU along the network path between the sender and receiver, and use this to dynamically adjust the MSS to avoid IP fragmentation within the network.

8. What is a TCP timestapms?

Ans. TCP timestamps, defined in RFC 1323, help TCP compute the round-trip time between the sender and receiver. Timestamp options include a 4-byte timestamp value, where the sender inserts its current value of its timestamp clock, and a 4-byte echo reply timestamp value, where the receiver generally inserts the most recent timestamp value that it has received. The sender uses the echo reply timestamp in an acknowledgment to compute the total elapsed time since the acknowledged segment was sent.

9. What do you understand by connection termination?

Ans. The connection termination phase uses, at most, a four-way handshake, with each side of the connection terminating independently. When an endpoint wishes to stop its half of the connection, it transmits a FIN packet, which the other end acknowledges with an ACK. Therefore, a typical tear-down requires a pair of FIN and ACK segments from each TCP endpoint.

10. Describe the various TCP ports.

Ans. TCP uses the notion of port numbers to identify sending and receiving application end-points on a host, or Internet sockets. Each side of a TCP connection has an associated 16-bit unsigned port number (0-65535) reserved by the sending or receiving application. Arriving TCP data packets are identified as belonging to a specific TCP connection by its sockets, that is, the combination of source host address, source port, destination host address, and destination port. This means that a server computer can provide several clients with several services simultaneously, as long as a client takes care of initiating any simultaneous connections to one destination port from different source ports.

11. What is TCP debugging?

Ans. A packet sniffer, which intercepts TCP traffic on a network link, can be useful in debugging networks, network stacks and applications that use TCP by showing the user what packets are passing through a link. Some networking stacks support the SO_DEBUG socket option, which can be enabled on the socket using setsockopt. That option dumps all the packets, TCP states, and events on that socket, which is helpful in debugging. Netstat is another utility that can be used for debugging.

Further Reading

1. Elements of Data Communication and Networks: S. A. Amutha Jeevakumari, University Science Press.
2. Wide Area Networks: Navneet Sharma, Firewal Media.
3. Data Communication System: Monika Khuaran, Firewal Media.

UNIT 7

THE APPLICATION LAYER

NOTES

STRUCTURE

Learning Objectives

After going through this unit, students will be able to:

- understand about the Application Layer of the networking.
- know about the Network Security in brief.
- learn about sending and receiving Electronic Mail.

7.1 THE APPLICATION LAYER

The application layer is the seventh level of the seven-layer OSI model. It interfaces directly to and performs common application services for the application processes; it also issues requests to the presentation layer. Note carefully that this layer provides services to user-defined application processes, and not to the end user. For example, it defines a file transfer protocol, but the end user must go through an application process to invoke file transfer. The OSI model does not include human interfaces.

The common application services sublayer provides functional elements including the Remote Operations Service Element (comparable to Internet Remote Procedure Call), Association Control, and Transaction Processing (according to the ACID requirements).

Above the common application service sublayer are functions meaningful to user application programs, such as messaging (X.400), directory (X.500), file transfer (FTAM), virtual terminal (VTAM), and batch job manipulation (JTAM).

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7.2 NETWORK SECURITY

Computer or network security has been violated when unauthorized access by any party occurs. So it becomes your duty as the Network Administrator to work on the security to guard against any such incidence taking place.

Need of Network Security

Computer security is required because most organizations can be damaged by hostile software or intruders. There may be several forms of damage which are obviously interrelated. These include:

- Damage or destruction of computer systems.
- Damage or destruction of internal data.
- Loss of sensitive information to hostile parties.
- Use of sensitive information to steal items of monitary value.
- Use of sensitive information against the organization's customers which may result in legal action by customers against the organization and loss of customers.
- Damage to the reputation of an organization.
- Monitary damage due to loss of sensitive information, destruction of data, hostile use of sensitive data, or damage to the organization's reputation.

The methods used to accomplish these unscrupulous objectives are many and varied depending on the circumstances. This guide will help administrators understand some of these methods and explain some countermeasures.

Security Issues

Computer security can be very complex and may be very confusing to many people. It can even be a controversial subject. Network administrators like to believe that their network is secure and those who break into networks may like to believe that they can break into any network.

There are many fallacies that network administrators may fall victim to. These fallacies may allow administrators to wrongfully believe that their network is more secure than it really is. This guide will attempt to clarify many issues related to security by doing the following:

- Help you determine what you are protecting.
- Break computer security into categories.
- Explain security terms and methods.
- Point out some common fallacies that may allow administrators to be overconfident.

- Categorize many common attacks against networks and computers.
- Explain some attack methods.
- Describe tools that can be used to help make a network more secure.

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Security Interdependence

There are many different aspects to computer and network security as you will read in this book. These different areas of computer security are interdependent on each other in order for a network to be secure. If one or more areas of computer security are ignored, then the entire security integrity of the organization's network may be compromised. A clear example of this is in the area of computer virus or worm protection.

Computer virus protection programs can only filter known viruses or worms. There are viruses or worms that are not yet recognized as virus programs immediately after their release.

Some of these vulnerabilities are operating system and application program errors. When security patches are created for software, they should be quickly applied. In this way the vulnerability to viruses is minimized but not eliminated. There are other steps which may further reduce this vulnerability, but it can never be completely eliminated.

Security Limitations and Applications

If you are reading this document and are thinking that you can get all the information required to make your network completely secure, then you are sadly mistaken. In many ways, computer security is almost a statistical game.

You can reduce but not eliminate the chance that you may be penetrated by an intruder or virus. This is mainly for one reason.

This is why even those who consider themselves hackers will say that the number one computer security threat is the lack of quality in the applications and operating systems. At this point, I could talk about the various corporate entities that write software and why software lacks the quality that many of us believe that it should possess, but that subject is not only way beyond the scope of this document, but also way beyond the scope of this project.

Additionally the users on your network are potentially a greater security risk than any programs. Obviously removing all vulnerabilities is impossible and will not secure your network against user errors. I have even considered the possibility that an operating system without a network interface can be completely secure, but even this cannot be guaranteed. Unknown viruses or trojan programs can creep in with applications on CDs or floppies. This has been known to happen. Although an attacker may not be able to get data from the system, they can damage or destroy data.

Layered Security

The fact that complete security is impossible is the reason security experts recommend "layered security". The idea is to have multiple ways of preventing an intrusion to decrease the chance that intrusions will be successful. For example, you should have virus protection on your client computers. To help layer this security you should also filter viruses at your email server.

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Another good defense layer would also include educating your users about viruses, how they spread, and how to avoid them.

Hackers

There are many documents that attempt to define the term hacker. I believe that the term hacker is a connotative term. This means that it is more defined by people's beliefs rather than by a dictionary.

Others believe that hackers are those that perform unauthorized break-ins to computer systems. The media and many sources have caused many uninformed people to believe that a hacker is a threat to computer and network security while this is not the case. A hacker is no more likely to break the law than anyone else. It would be wise to use the more accurate descriptive term, "intruder" to describe those who intrude into networks or systems without authorization.

Physical Security

This book will not talk about physical computer security beyond this paragraph. Your organization should be aware how physically secure every aspect of its network is because if an intruder gets physical access, they can get your data.

Be sure that your organization properly secures locations and consider the following:

- **Servers** - Contain your data and information about how to access that data.
- **Workstations** - May contain some sensitive data and can be used to attack other computers.
- **Routers, switches, bridges, hubs** and any other network equipment may be used as an access point to your network.
- **Network wiring and media** and where they pass through may be used to access your network or place a wireless access point to your network.
- **External media** which may be used between organizational sites or to other sites the organization does business with.
- **Locations of staff** who may have information that a hostile party can use.
- **Some employees** may take data home or may take laptops home or use laptops on the internet from home then bring them to work. Any information on these laptops should be considered to be at risk and these laptops should be secure according to proper policy when connected externally on the network (more on this later).

Some Technical Terms

This paragaph describes some commonly used computer security terms.

- **Protocol** - Well defined specification allowing computer communication.
- **Confidentiality** - Information is available only to people with rightful access.
- **Integrity** - Information can only be changed by authorized personnel. Integrity - The receiver of the message should be able to tell that the message was not modified. Requires key exchange.
- **Availability** - Information is available to only those who need it.

- **Verification - nonrepudiation** - There is proof that the sender sent the message.
- **Authentification** - The receiver of the message should be able to be sure of the origin of the message. Requires a digital signature (One way hash, public key algorithm, and symmetric algorithm) or a public key algorithm.
- **Spyware** - A computer program whose purpose is to spy on your internet activities usually for marketing purposes and usually done by a shady corporate entity.
- **Malware** - A computer program with some evil intent. It may on the surface have a good or useful intent, but may be a trojan (with a hidden purpose) which can be used to gain unauthorized access to your computer.

Attackers vs Hackers: attackers from Within and External

In a security context, a Hacker is someone who is involved in computer security/ insecurity, specializing in the discovery of exploits in systems (for exploitation or prevention), or in obtaining or preventing unauthorized access to systems through skills, tactics and detailed knowledge. In the most common general form of this usage, "hacker" refers to a black-hat hacker (a malicious or criminal hacker); many who use the term in any other sense insist on using the term cracker to refer to such hackers.

The context of computer security hacking forms a subculture which is often referred to as the network hacker subculture or simply the computer underground. According to its adherents, cultural values center around the idea of creative and extraordinary computer usage. Proponents claim to be motivated by artistic and political ends, but are often unconcerned about the use of criminal means to achieve them.

Artifacts and Customs

Contrary to the academic hacker subculture, networking hackers have no inherently close connection to the academic world. They have a tendency to work anonymously and in private. It is common among them to use aliases for the purpose of concealing identity, rather than revealing their real names.

This practice is uncommon within and even frowned upon by the academic hacker subculture. Members of the network hacking scene are often being stereotypically described as crackers by the academic hacker subculture, yet see themselves as hackers and even try to include academic hackers in what they see as one wider hacker culture, a view harshly rejected by the academic hacker subculture itself. Instead of a hacker–cracker dichotomy, they give more emphasis to a spectrum of different categories, such as white hat ("ethical hacking"), grey hat, black hat and script kiddie.

The network hacking subculture is supported by regular gatherings, so called Hacker cons. These have drawn more and more people every year including SummerCon (Summer), DEF CON, HoHoCon (Christmas), PumpCon (Halloween), H.O.P.E. (Hackers on Planet Earth) and HEU (Hacking at the End of the Universe). They have helped expand the definition and solidify the importance of the network hacker subculture. In Germany, members of the subculture are organized mainly around the Chaos Computer Club.

The subculture has given birth to what its members consider to be novel forms of art, most notably ascii art. It has also produced its own slang and various forms of unusual alphabet use, for example leetspeak. Both things are usually seen as an especially

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silly aspect by the academic hacker subculture. In part due to this, the slangs of the two subcultures differ substantially.

Political attitude usually includes views for freedom of information, freedom of speech, a right for anonymity and most have a strong opposition against copyright. Writing programs and performing other activities to support these views is referred to as hacktivism by the subculture. Some go as far as seeing illegal computer cracking ethically justified for this goal; the most common form is website defacement.

The security hackers have also edited some publications, most notably

- "2600: The Hacker Quarterly"
- "Hakin9"
- "Binary Revolution Magazine 2006"
- "Blacklisted 411"

Academic hackers

In the academic hacker culture, a computer hacker is a person who enjoys designing software and building programs with a sense for aesthetics and playful cleverness.

According to Eric S. Raymond, the academic hacker subculture developed in the 1960s among hackers working on early minicomputers in academic computer science environments. After 1969 it fused with the technical culture of the pioneers of the Internet. One PDP-10 machine at MIT connected to the Internet provided an early hacker meeting point.

It was called AI and ran ITS. After 1980 the subculture coalesced with the culture of Unix, and after 1987 with elements of the early microcomputer hobbyists that themselves had connections to radio amateurs in the 1920s. Since the mid-1990s, it has been largely coincident with what is now called the free software movement and the open source movement.

Many programmers have been labeled "great hackers," but the specifics of who that label applies to is a matter of opinion. Certainly major contributors to computer science such as Edsger Dijkstra and Donald Knuth, as well as the inventors of popular software such as Linus Torvalds (Linux), and Dennis Ritchie and Ken Thompson (the C programming language) are likely to be included in any such list.

People primarily known for their contributions to the consciousness of the academic hacker culture include Richard Stallman, the founder of the free software movement and the GNU project, president of the Free Software Foundation and author of the famous Emacs text editor as well as the GNU Compiler Collection (GCC), and Eric S. Raymond, one of the founders of the Open Source Initiative and writer of the famous text The Cathedral and the Bazaar and many other essays, maintainer of the Jargon File (which was previously maintained by Guy L. Steele, Jr.).

Within the academic hacker culture, the term hacker is also used for a programmer who reaches a goal by employing a series of modifications to extend existing code or resources. In this sense, it can have a negative connotation of using kludges to accomplish programming tasks that are ugly, inelegant, and inefficient.

In a very universal sense, a hacker also means someone who makes things work beyond perceived limits in a clever way in general, for example reality hackers.

Hobby Hackers

The hobby hacking subculture relates to hobbyist home computing of the late 1970s, beginning with the availability of MITS Altair. An influential organization was the Homebrew Computer Club.

The parts that didn't fuse with the academic hacker subculture focus mainly on commercial computer and video games, software cracking and exceptional computer programming (demo scene), but also to the modification of computer hardware and other electronic devices.

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Overlaps and differences

The main basic difference between academic and computer security hackers is their separate historical origin and development. The Jargon File reports that although considerable overlap existed for the early phreaking at the beginning of the 1970s, it quickly started to break when people joined in the activity who did it in a less responsible way.

Also, their activities in practice are largely distinct. The former focus on creating new and improving existing infrastructure (especially the software environment they work with), while the latter primarily and strongly emphasize the general act of circumvention of security measures, with the effective use of the knowledge (which can be to report and help fixing the security bugs, or exploitation for criminal purpose) being only rather secondary.

The most visible difference in these views was in the design of the MIT hackers' Incompatible Timesharing System, which deliberately didn't have any security measures.

It sees secondary circumvention of security mechanisms as legitimate if it is done to get practical barriers out of the way for doing actual work. In special forms, that can even be an expression of playful cleverness. However, the systematic and primary engagement in such activities is not one of the actual interests of the academic hacker subculture and it doesn't have significance in its actual activities, either.

Since the mid-1980s, there are some overlaps in ideas and members with the computer security hacking community. The most prominent case is Robert T. Morris, who was a user of MIT-AI, yet wrote the Morris worm. The Jargon File hence calls him "a true hacker who blundered".

Nevertheless, members of the academic subculture have a tendency to look down and disassociate from these overlaps. They commonly refer disparagingly to people in the computer security subculture as crackers, and refuse to accept any definition of hacker that encompasses such activities.

The computer security hacking subculture on the other hand tends not to distinguish between the two subcultures as harshly, instead acknowledging that they have much in common including many members, political and social goals, and a love of learning about technology. They restrict the use of the term cracker to their categories of script kiddies and black hat hackers instead. There is also overlap into the other direction. Since the mid-1990s, with home computers that could run Unix-like operating systems and with inexpensive internet home access being available for the first time, many people from outside of the academic world started to take part in the academic hacking subculture.

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All three subcultures have relations to hardware modifications. In the early days of network hacking, phreaks were building blue boxes and various variants. The academic hacker culture has stories about several hardware hacks in its folklore, such as a mysterious 'magic' switch attached to a PDP-10 computer in MIT's AI lab, that, when turned off, crashed the computer.

However, all these activities have died out during the 1980s, when the phone network switched to digitally controlled switchboards, causing network hacking to shift to dialling remote computers with modems, when preassembled inexpensive home computers were available, and when academic institutions started to give individual mass-produced workstation computers to scientists instead of using a central timesharing system. The only kind of widespread hardware modification nowadays is case modding.

An encounter of the academic and the computer security hacker subculture occured at the end of the 1980s, when a group of hackers, sympathizing with the Chaos Computer Club (who disclaimed any knowledge in these activities), broke into computers of American military organizations and academic institutions.

They sold data from these machines to the Soviet secret service, one of them in order to fund his drug addiction. The case could be solved when scientists from the environment of the academic hacker subculture found ways to log the attacks and to trace them back.

7.3 ELECTRONIC MAIL

Simple Mail Transfer Protocol (SMTP) is an Internet standard for electronic mail (e-mail) transmission across Internet Protocol (IP) networks. SMTP was first defined in RFC 821 (STD 10), and last updated by RFC 5321 (2008), which describes the protocol in widespread use today, also known as extended SMTP (ESMTP).

While electronic mail server software uses SMTP to send and receive mail messages, user-level client mail applications typically only use SMTP for sending messages to a mail server for relaying. For receiving messages, client applications usually use either the Post Office Protocol (POP) or the Internet Message Access Protocol (IMAP) to access their mail box accounts on a mail server.

SMTP is a relatively simple, text-based protocol, in which one or more recipients of a message are specified (and in most cases verified to exist) along with the message text and possibly other encoded objects. The message is then transferred to a remote server using a series of queries and responses between the client and server. Either an end-user's e-mail client, a.k.a. MUA (Mail User Agent), or a relaying server's MTA (Mail Transport Agents) can act as an SMTP client.

An e-mail client knows the outgoing mail SMTP server from its configuration. A relaying server typically determines which SMTP server to connect to by looking up the MX (Mail eXchange) DNS record for each recipient's domain name. Conformant MTAs (not all) fall back to a simple.

A record in the case of no MX (relaying servers can also be configured to use a smart host). The SMTP client initiates a TCP connection to server's port 25 (unless overridden by configuration). It is quite easy to test an SMTP server using the netcat program (see on next page).

SMTP is a "push" protocol that cannot "pull" messages from a remote server on demand. To retrieve messages only on demand, which is the most common requirement on a single-user computer, a mail client must use POP3 or IMAP.

Another SMTP server can trigger a delivery in SMTP using ETRN. It is possible to receive mail by running an SMTP server. POP3 became popular when single-user computers connected to the Internet only intermittently; SMTP is more suitable for a machine permanently connected to the Internet.

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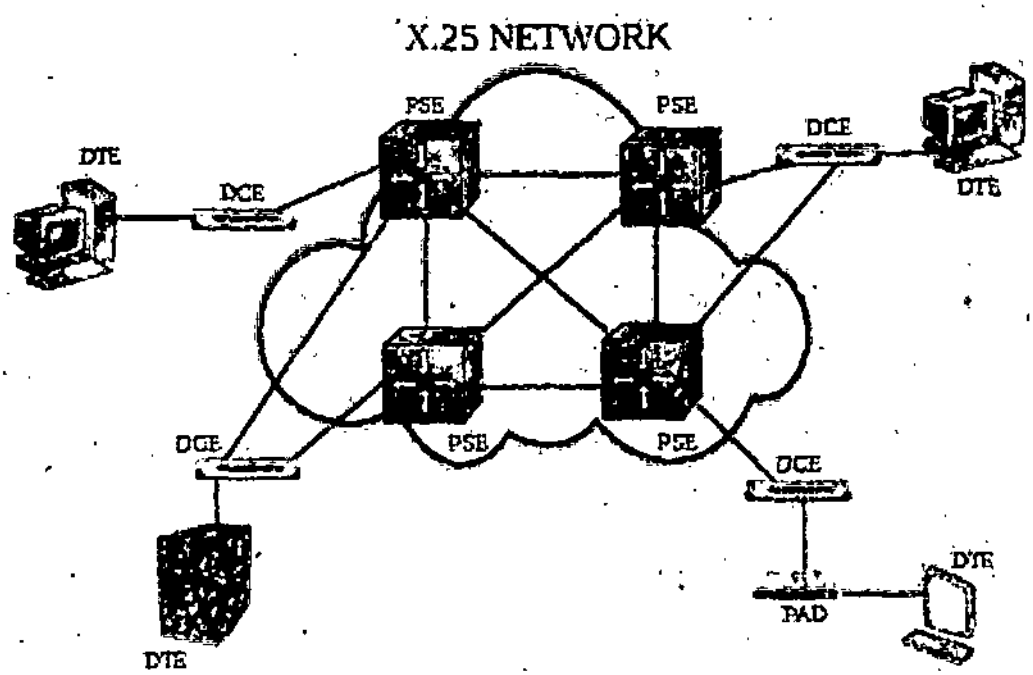
A simple aid to memory is "Send Mail To People."

X.25

X.25 is an ITU-T standard network layer protocol for packet switched wide area network (WAN) communication. An X.25 WAN consists of packet-switching exchange (PSE) nodes as the networking hardware, and leased lines, Plain old telephone service connections or ISDN connections as physical links.

X.25 is part of the OSI protocol suite, a family of protocols that was used especially during the 1980s by telecommunications companies and in financial transaction systems such as automated teller machines.

X.25 is today to a large extent replaced by less complex and less secure protocols, especially the Internet protocol (IP) although some telephone operators offer X.25-based communication via the signalling (D) channel of ISDN lines.



Architecture

The general concept of X.25 was to create a universal and global packet-switched network. Much of the X.25 system is a description of the rigorous error correction

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needed to achieve this, as well as more efficient sharing of capital-intensive physical resources.

The X.25 specification defines only the interface between a subscriber (DTE) and an X.25 network (DCE). X.75, a very similar protocol to X.25, defines the interface between two X.25 networks to allow connections to traverse two or more networks. X.25 does not specify how the network operates internally—many X.25 network implementations used something very similar to X.25 or X.75 internally, but others used quite different protocols internally. The ISO equivalent protocol to X.25, ISO 8208, is the compatible with X.25, but additionally includes provision for two X.25 DTEs to be directly connected to each other with no network in between.

The X.25 model was based on the traditional telephony concept of establishing reliable circuits through a shared network, but using software to create "virtual calls" through the network. These calls interconnect "data terminal equipment" (DTE) providing endpoints to users, which looked like point-to-point connections. Each endpoint can establish many separate virtual calls to different endpoints.

For a brief period, the specification also included a connectionless datagram service, but this was dropped in the next revision. The "fast select with restricted response facility" is intermediate between full call establishment and connectionless communication. It is widely used in query-response transaction applications involving a single request and response limited to 128 bytes of data carried each way. The data is carried in an extended call request packet and the response is carried in an extended field of the call reject packet, with a connection never being fully established.

Closely related to the X.25 protocol are the protocols to connect asynchronous devices (such as dumb terminals and printers) to an X.25 network: X.3, X.28 and X.29. This functionality was performed using a Packet Assembler/Disassembler or PAD (also known as a Triple-X device, referring to the three protocols used).

Relation to the OSI Reference Model

Although X.25 predates the OSI Reference Model (OSIRM), the physical layer of the model corresponds to the X.25 physical level; the link layer, the X.25 link level; and network layer, the X.25 packet level. The X.25 link-layer, LAPB, provides a reliable data path across a data link (or multiple parallel data links, multilink) which may not be reliable itself. The X.25 packet-layer, provides the virtual call mechanisms, running over X.25 LAPB. As long as the link layer does provide reliable data transmission, the packet-layer will provide error-free virtual calls.[citation needed] However, the packet-layer also includes mechanisms to maintain virtual calls and to signal data errors in the event that the link-layer does not provide reliable data transmission. All but the earliest versions of X.25 include facilities which provide for OSI network layer Addressing.

User Device Support

X.25 was developed in the era of dumb terminals connecting to host computers, although it also can be used for communications between computers. Instead of dialing directly "into" the host computer — which would require the host to have its own pool of modems and phone lines, and require non-local callers to make long-distance calls — the host could have an X.25 connection to a network service provider. Now dumb-terminal users could dial into the network's local "PAD" (Packet

Assembly/Disassembly facility), a gateway device connecting modems and serial lines to the X.25 link as defined by the X.29 and X.3 standards.

Having connected to the PAD, the dumb-terminal user tells the PAD which host to connect to, by giving a phone-number-like address in the X.121 address format (or by giving a host name, if the service provider allows for names that map to X.121 addresses). The PAD then places an X.25 call to the host, establishing a virtual circuit. Note that X.25 provides for virtual circuits, so appears to be a circuit switched network, even though in fact the data itself is packet switched internally, similar to the way TCP provides virtual circuits even though the underlying data is packet switched. Two X.25 hosts could, of course, call one another directly; no PAD is involved in this case. In theory, it doesn't matter whether the X.25 caller and X.25 destination are both connected to the same carrier, but in practice it was not always possible to make calls from one carrier to another.

NOTES

For the purpose of flow-control, a sliding window protocol is used with the default window size of 2. The acknowledgements may have either local or end to end significance. A D bit (Data Delivery bit) in each data packet indicates if the sender requires end to end acknowledgement. When D=1, it means that the acknowledgement has end to end significance and must take place only after the remote DTE has acknowledged receipt of the data. When D=0, the network is permitted (but not required) to acknowledge before the remote DTE has acknowledged or even received the data. While the PAD function defined by X.28 and X.29 specifically supported asynchronous character terminals, PAD equivalents were developed to support a wide range of proprietary intelligent communications devices, such as those for IBM System Network Architecture (SNA).

7.4 WORKING OF E-MAIL

Since the facility of e-mail provided by Internet, people have more or less stopped using the post. It has been in fact, named as Snail mail, or the mail which moves at the snail speed. Whereas electronic mail is fast and moves over from one computer to another electronically at an amazing speed. Most of the Internet users are very happy with this facility. In fact, you can send not only the text, but pictures and graphics too.

Most of the offices are sending their mail through computers. No more fussy papers or carbons. In fact, no need to file them too. All is stored in computers and can be accessed whenever needed. All you need is to open an e-mail account and start mailing your letters even if you do not have a computer. You can send the mail using your account number from any Cyber cafe too.

So to work with e-mail you must have an account of your own. For this, as already mentioned, each popular search engine allows you to open an e-mail account. Here, in our case, I am using the account opening process of Hotmail.com.

Opening of e-mail account

The following are the popular sites which allow you to open an e-mail account:

Yahoo	http://www.mail.yahoo.com
Hotmail	http://www.hotmail.com

NOTES

Rediff	http://www.rediff.com
Mailcity	http://www.mailcity.com

Besides these most of your ISPs would allow you to open an account. Please remember that these e-mail services are all free and the sites do not charge any amount for using the same.

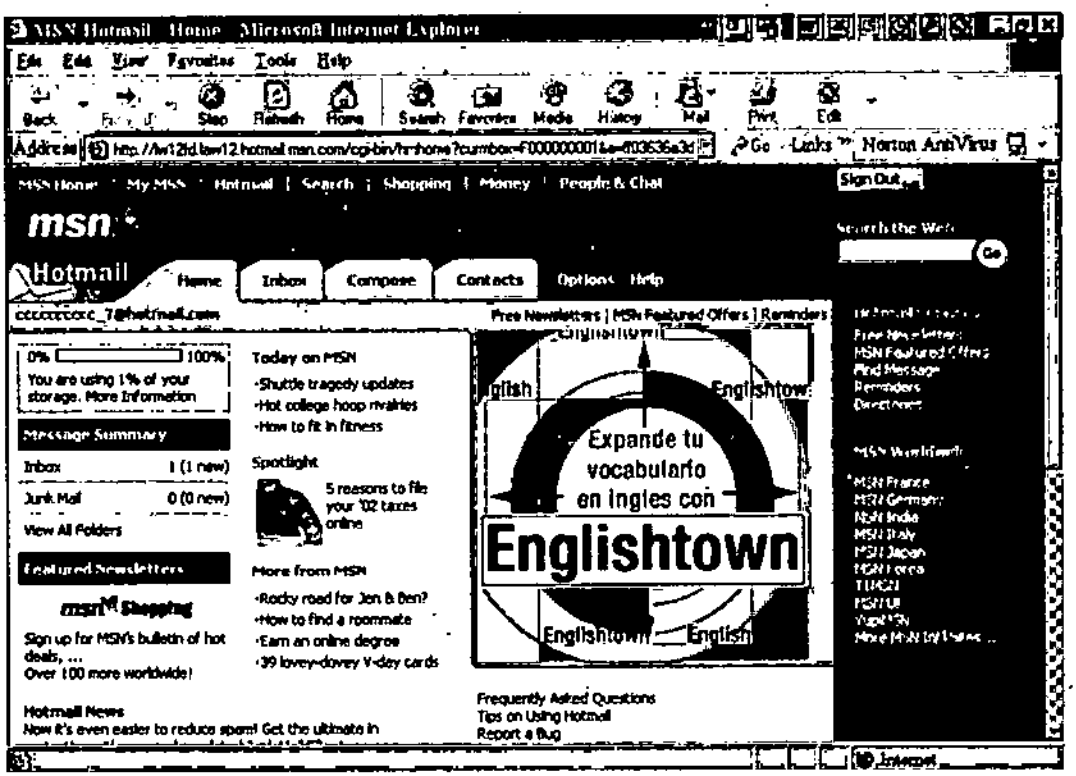
The following dialog boxes would show you how to create your own e-mail account with hotmail.

E-mail Organization

With reference to the e-mail program named Outlook Express, the messages in the software can be roughly organized in the following folders:

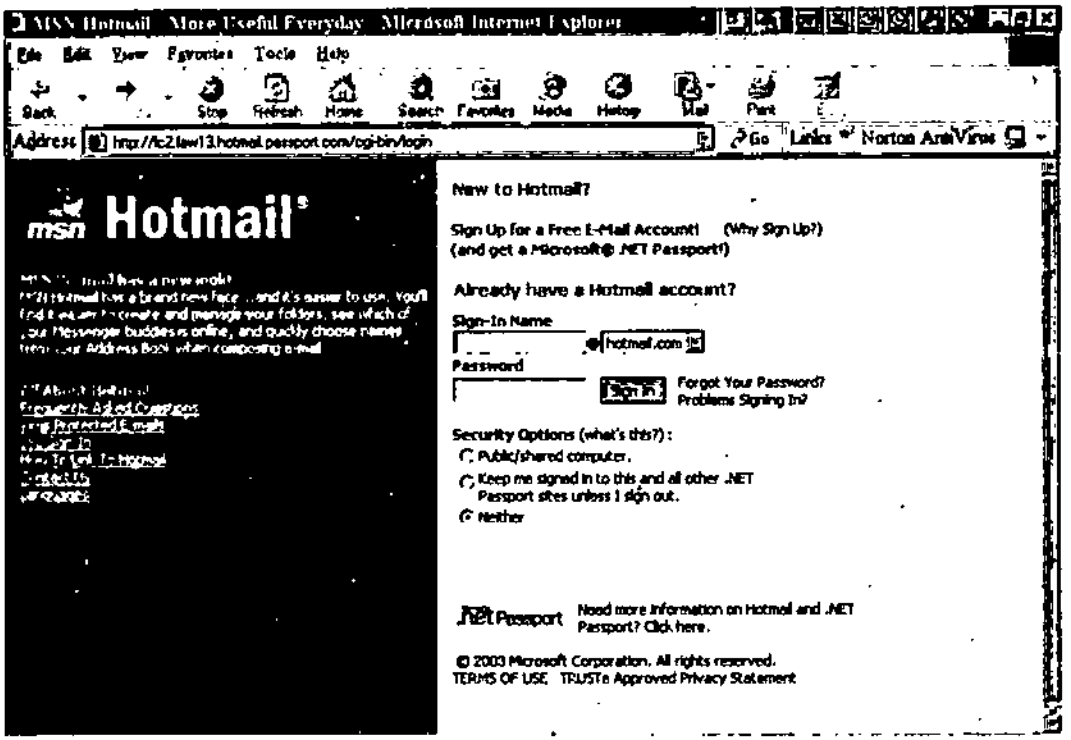
Inbox	For all incoming messages
Outbox	For all messages queued for sending
Sent Items	For all messages previously sent
Deleted items	For all messages marked for deletion
Drafts	For all messages which are pending completion

All these can be seen in the figure on the next page.



Member of the appropriate sex	MOTAS
Member of the opposite sex	MOTOS

NOTES



Before we start reading and sending the e-mails, we must understand few e-mail terms which are quite useful in reading and sending messages.

Using Abbreviations

An abbreviation would save both time and space. It is important that the abbreviations used by you are understood equally well by the receiver of your e-Mail. So I give here a list of abbrevations which have by now become the universally accepted abbrevations around the world.

Full Word/Term/Phrase	*Abbreviation*
By The Way	BTW
Frequently Asked Questions	FAQ
Friend of a friend	FOAF
For what it is worth	FWTW
For Your Information	FYI
In My Humble Opinion	IMHO
In My Opinion	IMO
In my opinion	IMO
In other words	IOW
In Regard To	IRT
Later	LR
Laughing out loud	LOL

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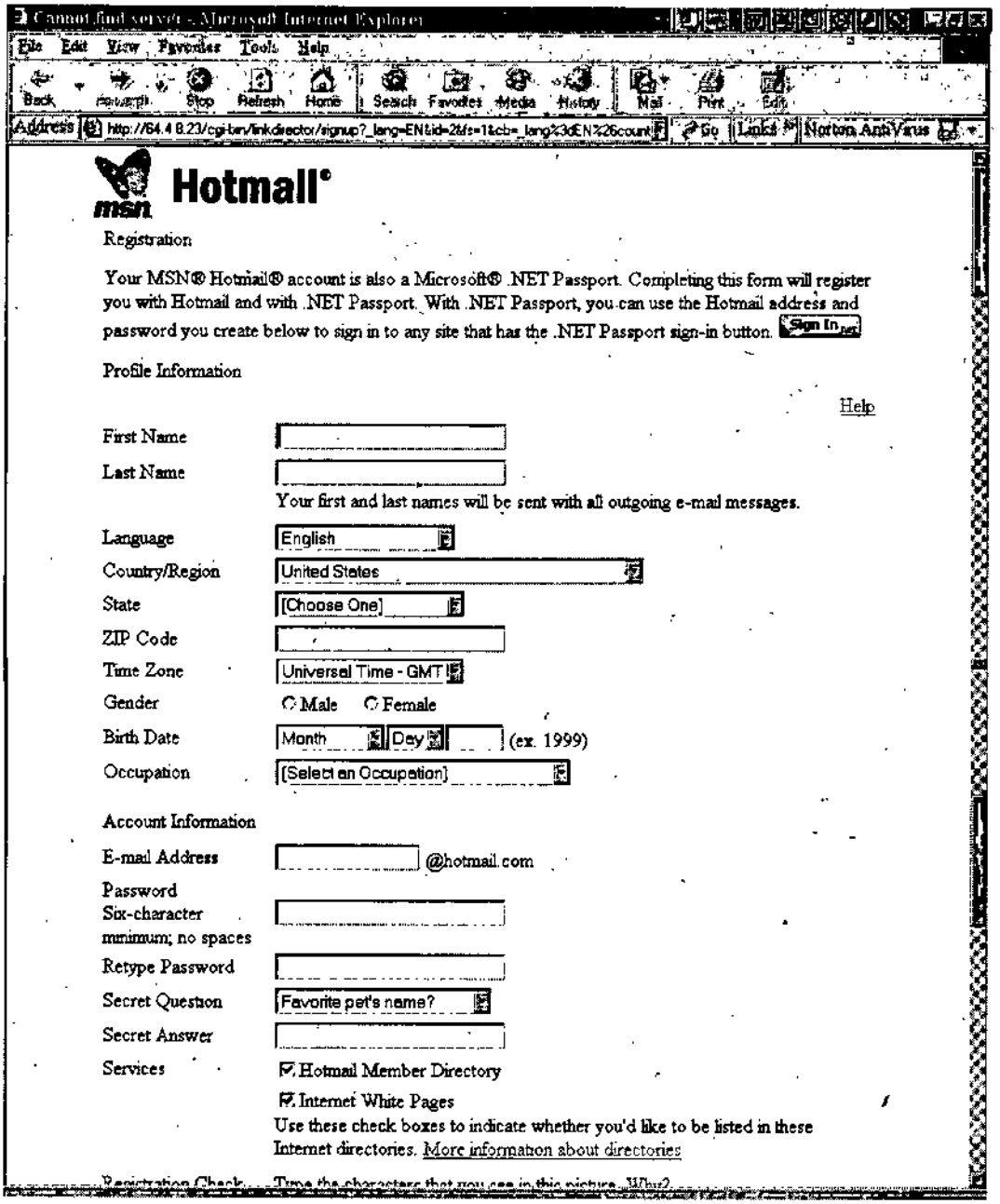

On The Other Hand	OTOH
Refer to/Regarding	RE:
Rolling on the floor and laughing	ROFTAL
Significant other	SO
You	U
With respect to	WRT

Using Smilies

There are lots of ways that you can convey your messages across. Not, only in terms of abbrevations but in terms of small similies too. These are also like abbrevations

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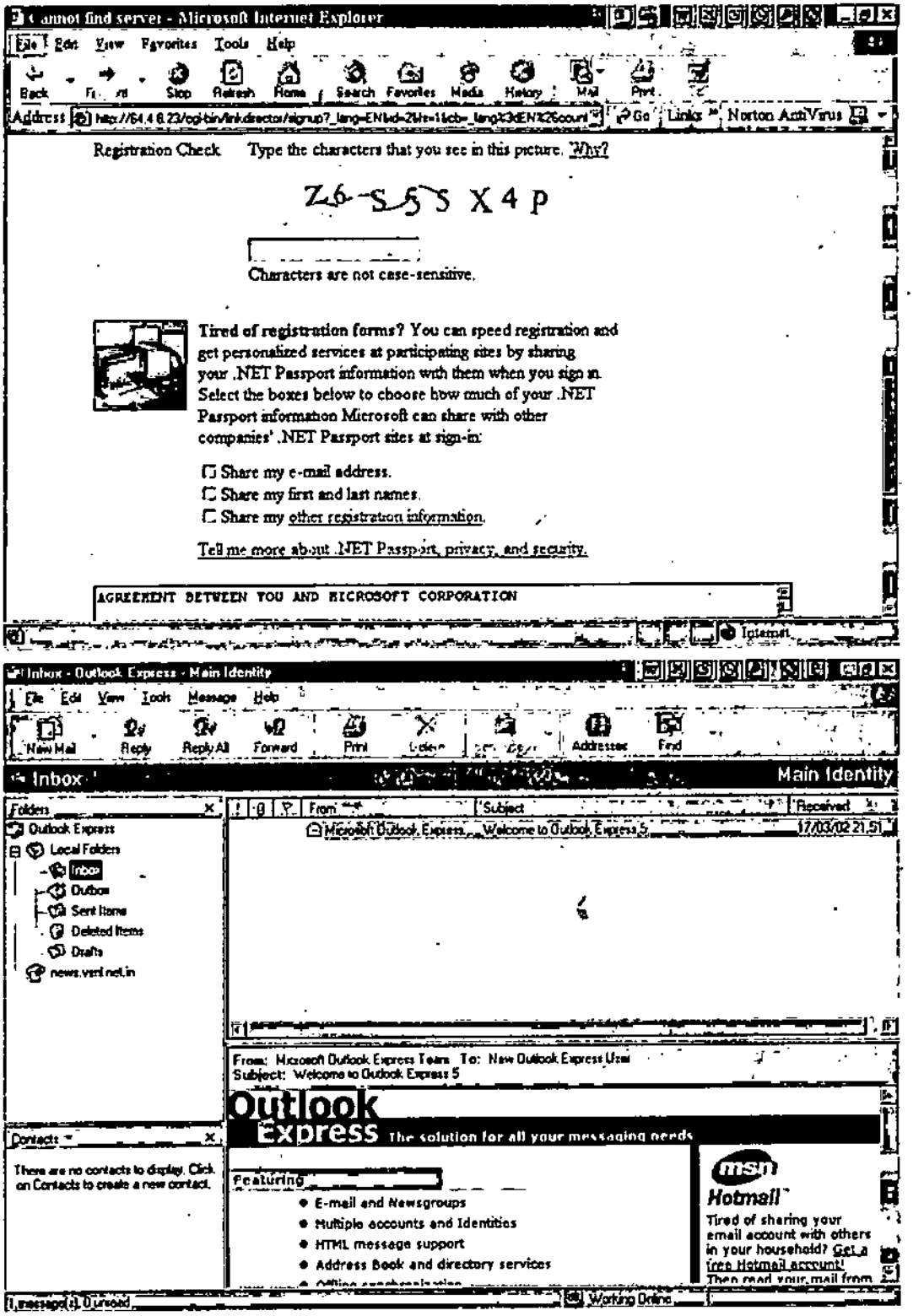


are finding their way into most of the computers around the world. A smiley is a sequence of characters on your computer's keyboard. For example, the colon represents the eyes, the dash represents the nose and the right parenthesis represents the mouth.

Given below a list of the most common smileys.

NOTES

Emotions	*Meaning*
8:-)	Little girl
X-(	Mad
&-L	Makes me cry
:-(*)	Makes me sick
:-S	Makes no sense
:-{	Moustache
:-#	My lips are seaed
8-O	Omigod!
:^)	Personality
:-C	Real unhappy
:-)^<	Big boy
(:-)	Big face
:-D	Said with a smile
:-)8<	Big girl
:-@	Screening
:-V	Shouting
:-(<	Standing firm
:-0	Talkative
H-)	Cross-eyed
:-@!	Cursing
:-E	Disappointed
(:-(	Very unhappy
:-6	Eating something spicy
]-I	Wearing sunglasses
:")	Embrassed
'-)	Winking
L-O	Yawning

As you become proficient in e-Mailing, you will be able to learn many more of these fascinating emoticons, which exist in thousands.

Parts of E-mail Text

Like a formal letter e-mail also is divided into various sections. In a good business letter, you would have, date, the name and address of the person to whom the letter is addressed, salutation, text matter, sender's name, etc. Most of these you would find here too but with some difference. Lets see what an e-mail is supposed to have.

NOTES

Headers

It is very important that your header must be loud and clear for it can be seen and read from a distance too.

To

The To: field contains the e-mail address of the person to whom you are sending the e-mail. If you are sending e-mail to someone in your own domain, you do not need to include @ domain.

From

In most of the cases, you would not be typing this, since it would be taken from the system. It in fact includes your e-mail address.

Subject

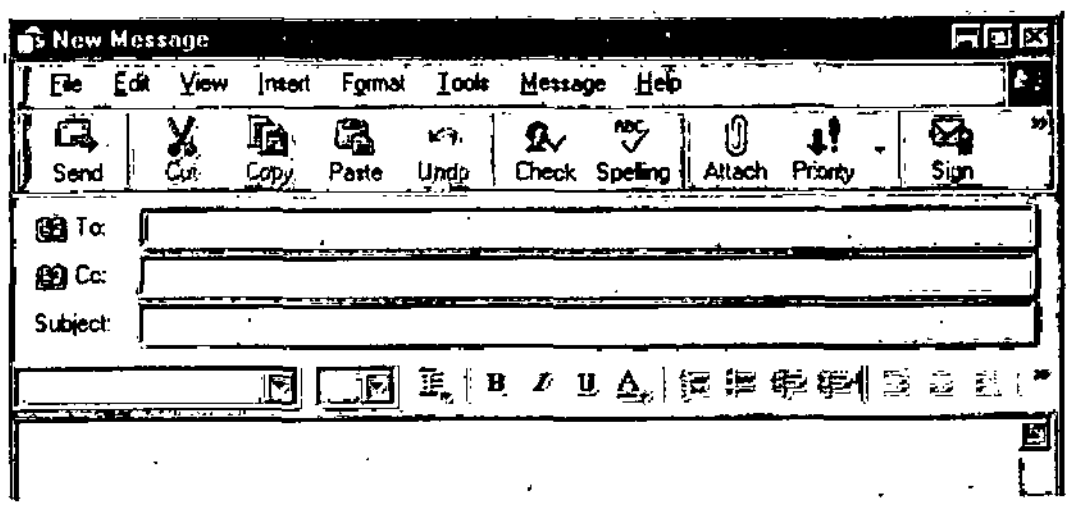
The Subject: field should contain a very short description (20-30 characters) of what your message is about. This field can also be called Subject of the Message: or simply message. It is not compulsary to have a subject.

Cc

You can send the copy of the same e-mail to another person. For this you have to mention the e-mail address of the second receipt. The software would take care that the same message is sent to the second recepient too.

Bcc

It is the short form of Blind Carbon Copy, which gives you a way of copying and transmitting an e-mail message to another person, without the first person (the main addressee), knowing about it.



Attachments

Using this option you can attach a file created in other software along with this e-mail. In most of the cases while applying for a job, the letter is typed in the e-mail text and the bio-data which is mostly in some word processing mode, attached to it.

Date

Date and time are usually taken from the system. You need not type it.

NOTES

Body

The body of an e-mail message is simply the text that you want to send to the other person.

Working with Messages

The first thing that you would do on opening the software would be receive the messages. For this you will have to click on Send/Receive button.

You can also set Outlook to automatically check for messages at regular intervals, or you can have Outlook check for message whenever you want.

1. Click the Send/Receive button on the Standard toolbar, as shown below.
2. The Intenet connection will be made depending upon the type of connection you have. Its progress will be shown to you in the dialog box in the lower right corner of the window.
3. When messages arrive, Outlook asks you whether you want to start reading them. Click Yes in the Microsoft Outlook dialog box to starting reading messages, or click No to read them later.

Reading a Message

Once the messages have been received in the Inbox, they will be appended to the other ones, i.e., in the last. The last few messages are always new. There are various type of messages which are there in the mailbox.

The messages in the Inbox can have the following symbols attached to them.

- Unread message
- High importance message
- Low importance message
- Message that has been read
- Message with attachment
- Message that has been forwarded
- Message that has been replied to
- Meeting for which response is requested
- Follow-up flag

All these have a symbol attached to them. .

Reading the Message

1. Double-click a message in the Inbox list.

   Or

   Select the message to open and press Ctrl + O

   Or

   Select the message to open and from the File menu, choose Open. Then choose Selected Items from the submenu.

Each message opens in a separate window.

Replying to a Message

Once read, a message can either be replied or trashed.

To reply to a message:

1. With a message selected in the Inbox window or with the message open, click the Reply button on the Standard toolbar. The reply has the sender in the To box and the original subject in the Subject box preceded by RE.
2. In the text area, type your reply. The reply appears above the original message.

NOTES

Replying in a Phased Manner

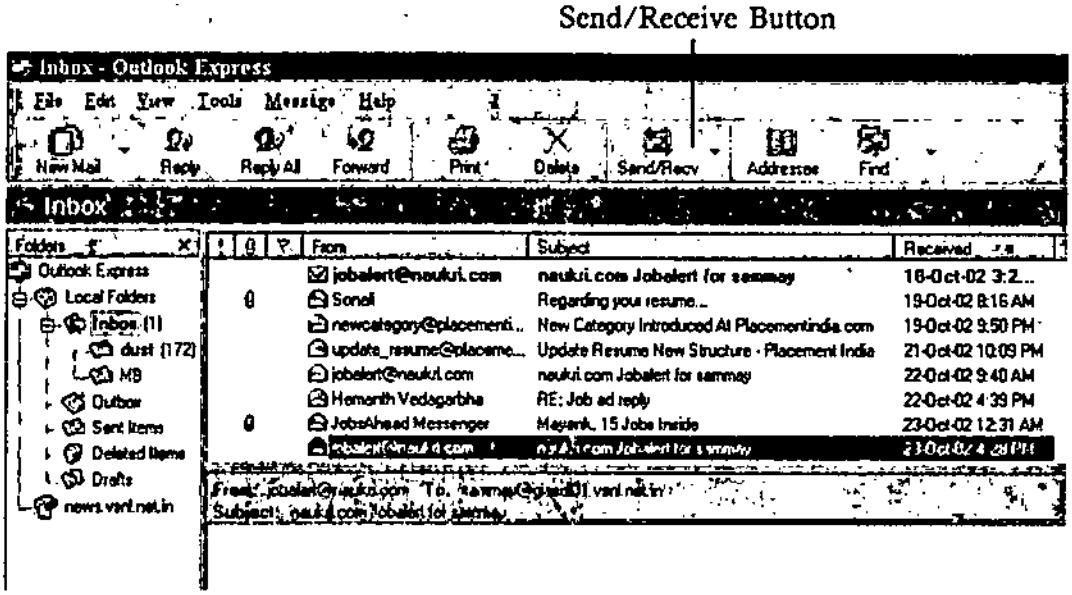
You can reply in phases. You can type half a reply now, save it and then complete it later. In this case, since the reply is half baked, it is stored in drafts folder.

For this do the following:

1. Click Save on the Standard toolbar.

   Or

   From the File menu, choose Save.
2. Close the window.
3. Outlook places the reply in the Drafts folder. You can open it there to continue working on it.
4. To reply to all of them, click the Reply to All button on the Standard toolbar.
5. You can still edit a message that is waiting in the Outbox. Just double-click the message, make your changes, and then click Send again.



Forwarding a Message

The message received from A can be sent to B by using Forward option. You can also add your own comments to it before sending it.

1. With a message selected in the Inbox window or with the message open, click the Forward button on the Standard toolbar.

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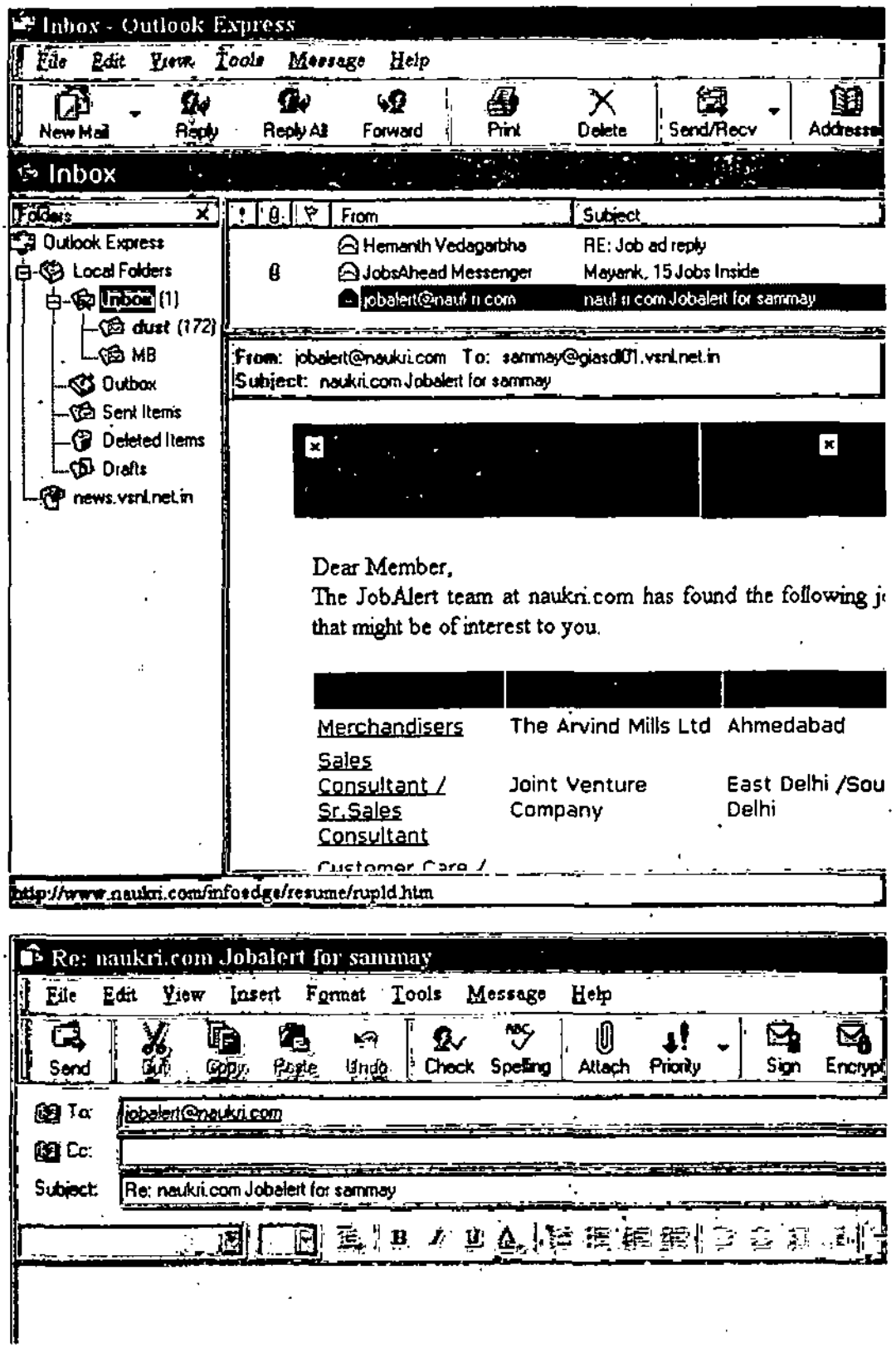

The new message shows the original subject in the Subject box preceded by FW, as shown here.

2. Type the recipient's e-mail address in the To box.

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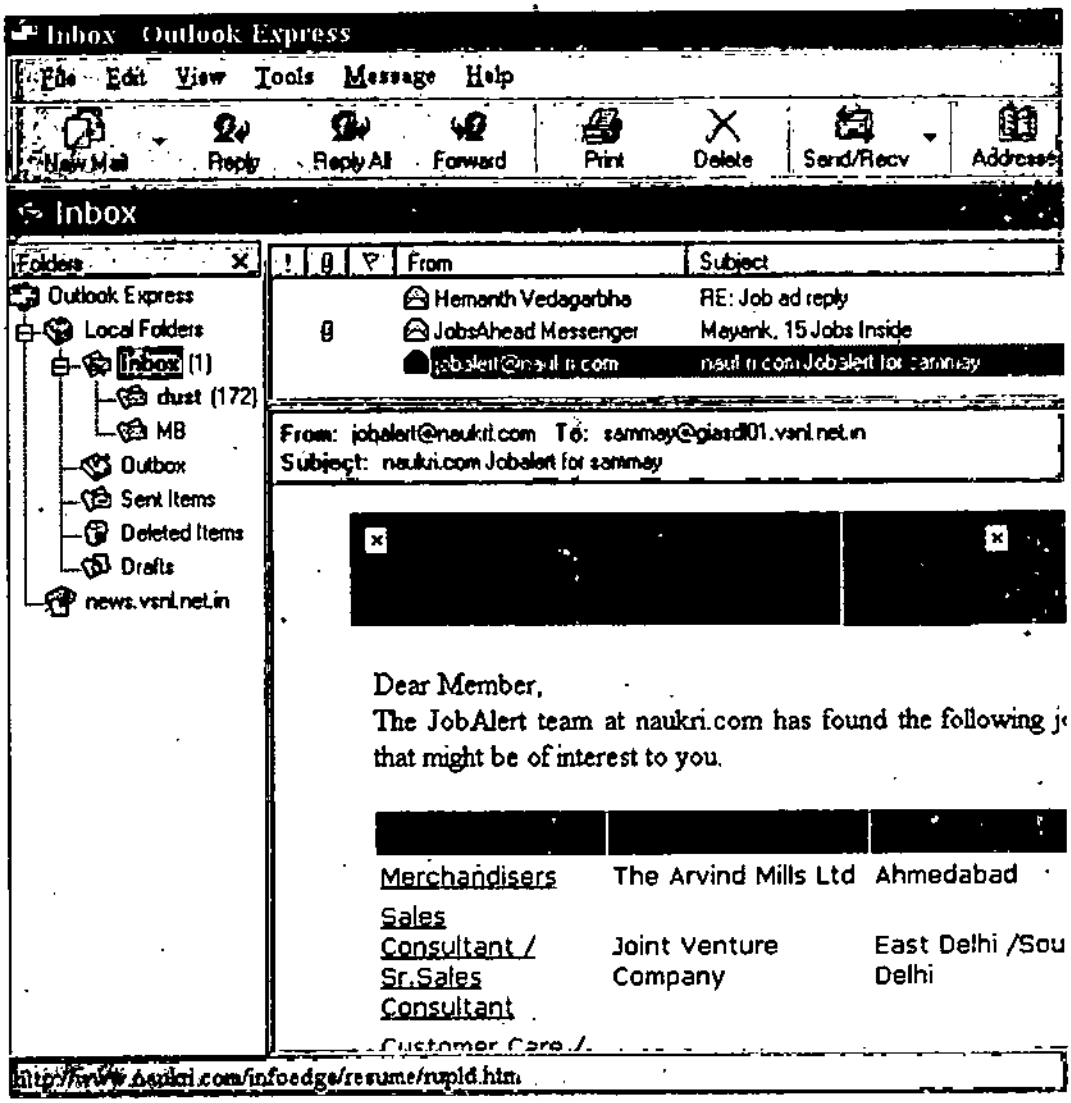
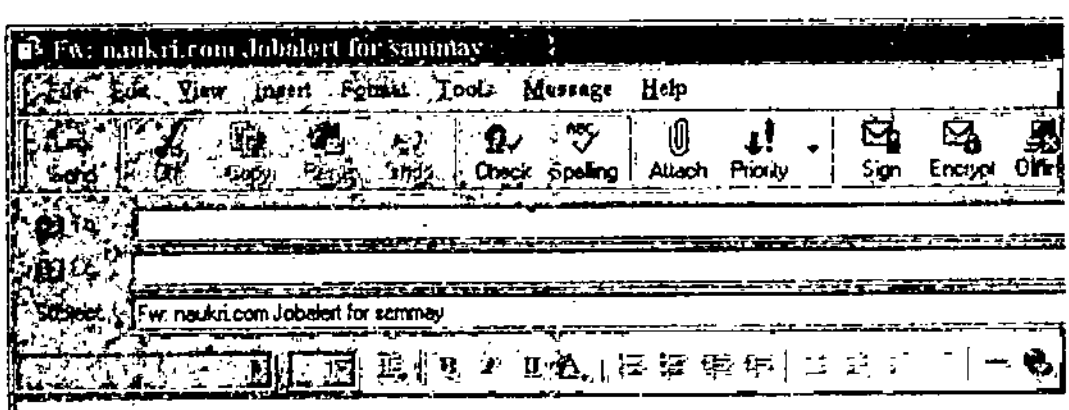


Or

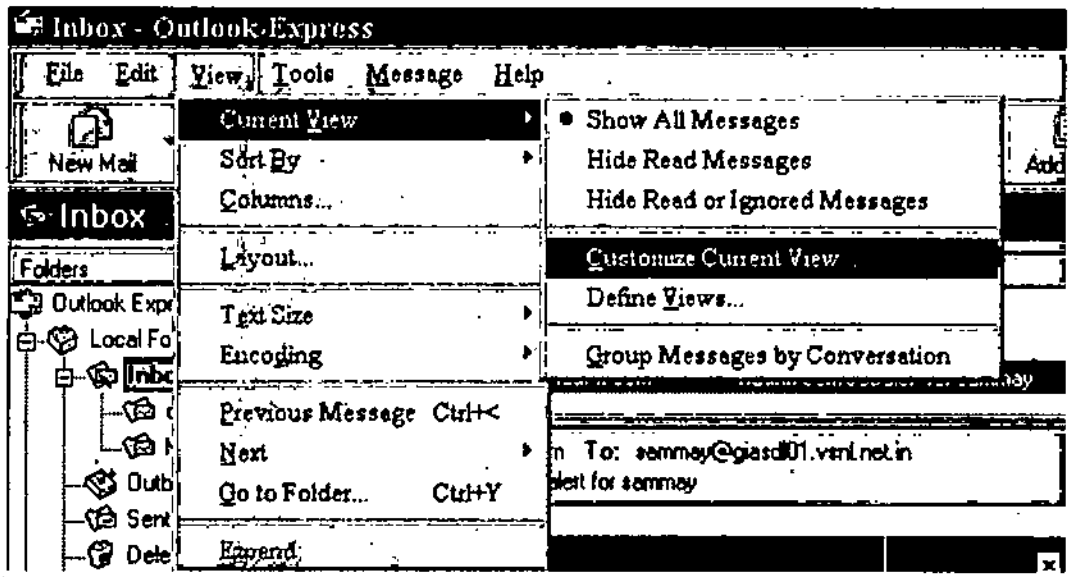
To select an address from the Address Book, click the To button, select one or more names in the Select Names dialog box, click the To button in the dialog box, and click Ok.

3. Type comments which you want to add in the space above the message.



NOTES

4. Click Send on the Standard toolbar to move the message to the Outlook folder.
5. If a message has files attached, the files are forwarded as well.



Deleting a Message

If you do not require any message, you can delete it or others too.

1. Click the Delete button on the toolbar in the message window.

   Or

   Press Ctrl + D

The message is moved to the Deleted Items folder.

Changing View

Outlook lets you view the contents of a message folder in a number of ways.

1. From the View menu, choose Current View, and then choose a view from the submenu, as shown on next page.
2. The view of the current folder is changed.

Using your Own Stationery

When you send a message to somebody, Outlook sees to it that it should have your distinctive features on it. It provides you with the stationery which you can add to your message page. What's more it even gives you an option of putting your signature. You can also set flags to call attention of important messages and to request follow-up actions such as asking the recipient to reply to the message before a certain date.

If you want to send file, such as documents or pictures, you can attach them to e-mail messages, too, using the attachment command.

Mail Options

You can use the mail format options to specify the appearance of your outgoing messages. You can override these default settings for individual messages.

NOTES

Setting Mail Format Options

1. From the Tools menu, choose Options.
2. On the Send tab of the Options dialog box, choose a format from the Send in this Message Format drop-down list.

   Choose HTML to send the message in the format used for Web pages, which allows text formatting. Most popular mail programs can display HTML e-mail messages.

   Choose Microsoft Outlook Rich Text if you know that your recipient uses Outlook too. This setting allows you to use boldfacing and other formatting in the message.
3. Click Ok when you have finished.

Choosing Default Fonts

1. On the Mail Format tab of the Options dialog box, choose the Plain Text or Microsoft Outlook Rich Text format option.
2. Click the Fonts button.
3. In the Fonts dialog box, choose the default fonts for messages you type.
4. Click Ok to return to the Options dialog box.
5. Click Ok when you have finished setting options.
6. If you choose HTML or Microsoft Outlook Rich Text as the mail format, a formatting toolbar appears in a new message window. You can use the buttons on this toolbar to set text formatting for the message text.
7. In the Fonts dialog box, you can indicate when you want your default fonts to override stationery fonts.

Starting and Addressing a Message

You can start a message from any Outlook e-mail folder, address the message to as many recipients as you want, and send copies to anyone who might want to be informed. You can also send a blind copy (hidden from other recipients) to someone.

Starting Message

1. In the Inbox or Outlook Today window (or any other e-mail folder), click the New button on the Standard toolbar.
2. If your e-mail editor is not Word, choose whether you want Word to be default editor.
3. A message window with the default format opens.

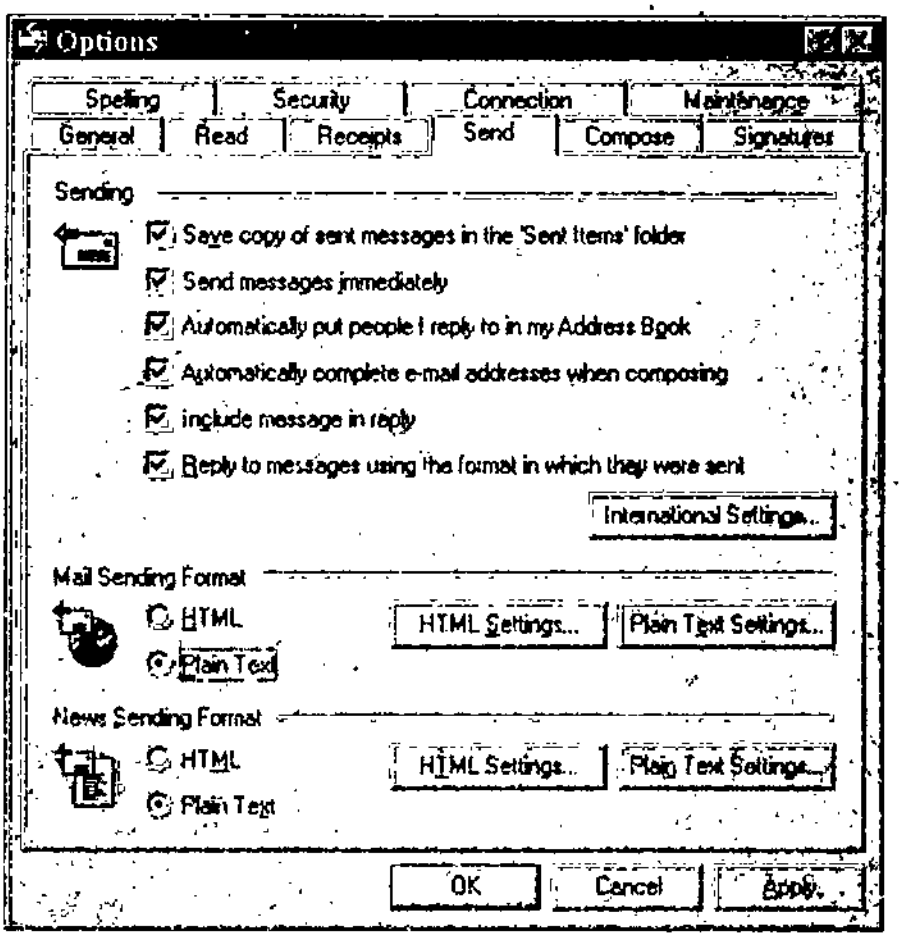
Typing Addresses

1. In the To box, type a single e-mail address.

   Or

   Type several addresses, separating them by commas or semicolons.

NOTES



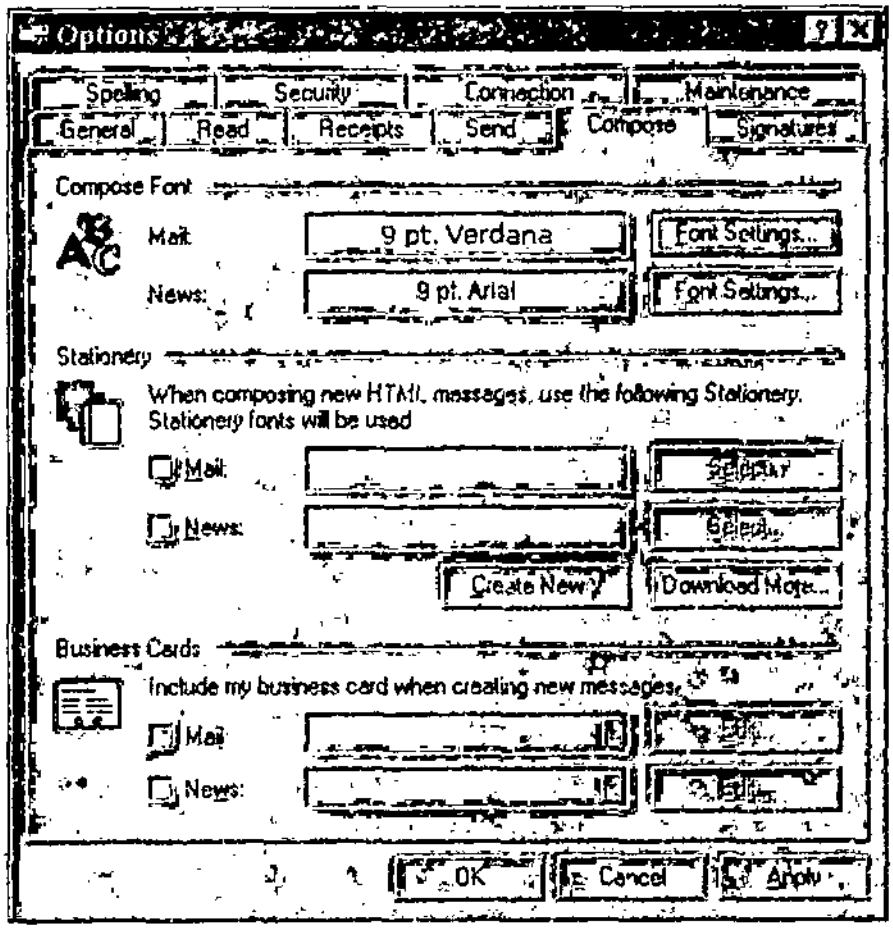
Using Address Book

1. Click the To or the Cc button to choose from the Address Book.
2. Choose a name in the Select Names dialog box and click the To, Cc, or Bcc button to add the name.
3. When you have finished adding names, click Ok to return to the new message.
4. If you want others to receive a copy of this message, you can add e-mail addresses in the Cc box as well.
5. To verify that you have typed an e-mail address properly, click the Check Name button on the Standard toolbar. Outlook checks the address against the Address Book and warns you if no match is found.
6. If you want to display the Bcc field, choose Bcc field from the View menu.
7. For help finding a name, click the Find button in the Select Names dialog box to open the Find People dialog box.
8. To open the Address Book quickly, click the Address Book button on the Standard toolbar.

Creating Stationery

For e-mail in HTML format, you can choose predesigned stationery, which includes a background; graphical elements such as bullets, pictures, and horizontal lines; text

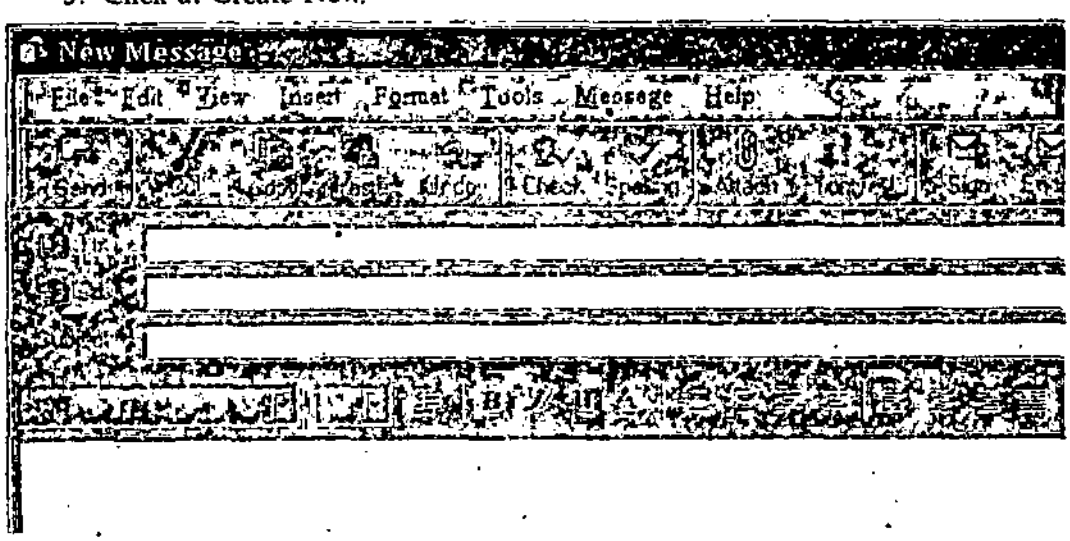
NOTES



fonts; and colours that blend with the background design. If Word is specified as your e-mail editor and HTML is selected format, the first time you create a new message you can choose a theme.

Choosing Default Outlook Stationery:

1. From the Tools menu, choose Options.
2. On the Send tab of the Options dialog box, choose HTML as the format.
3. Click at Create New.



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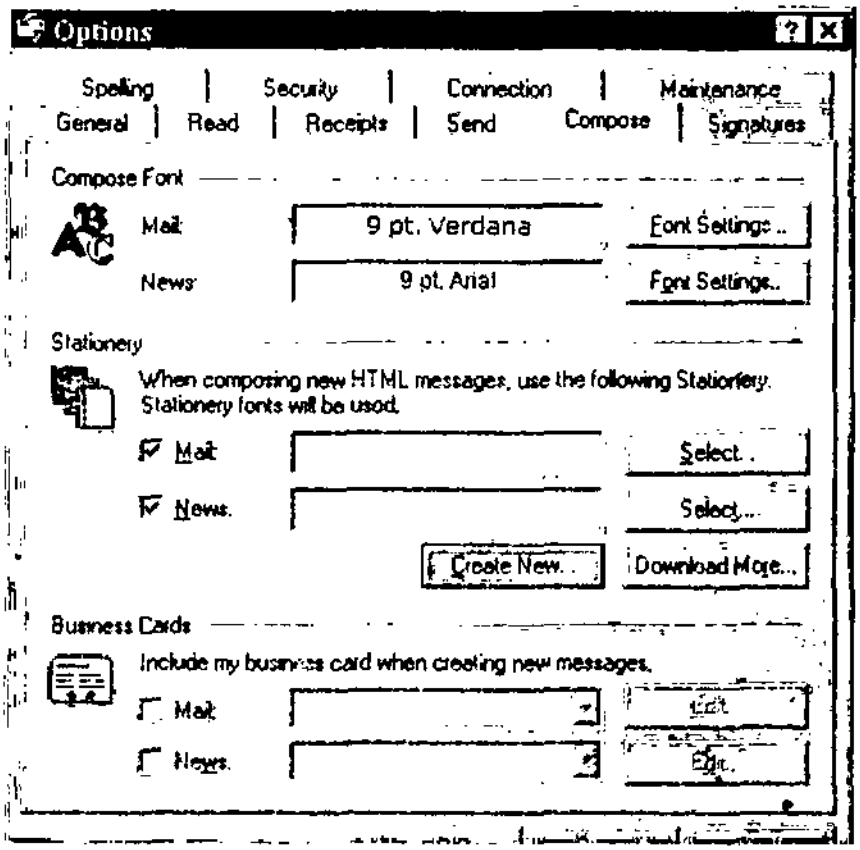
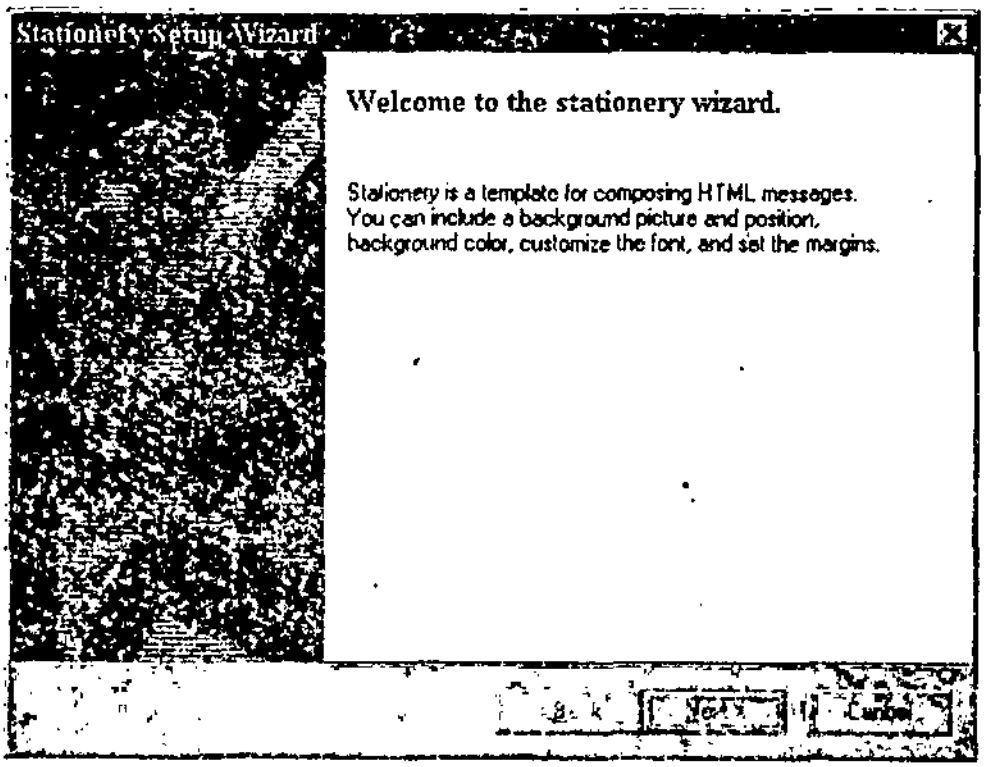


4. This would lead you to a Stationery Picker Wizard, which will guide you to select the stationery of your choice.
5. On this and next pages, you will see the various options of the wizard and their selection.



6. You can use the same options or can have some of your own.

More Stationery Options:

1. In the Options dialog box, choose Download More.
2. From the net, this would give you more options of stationery to choose from.

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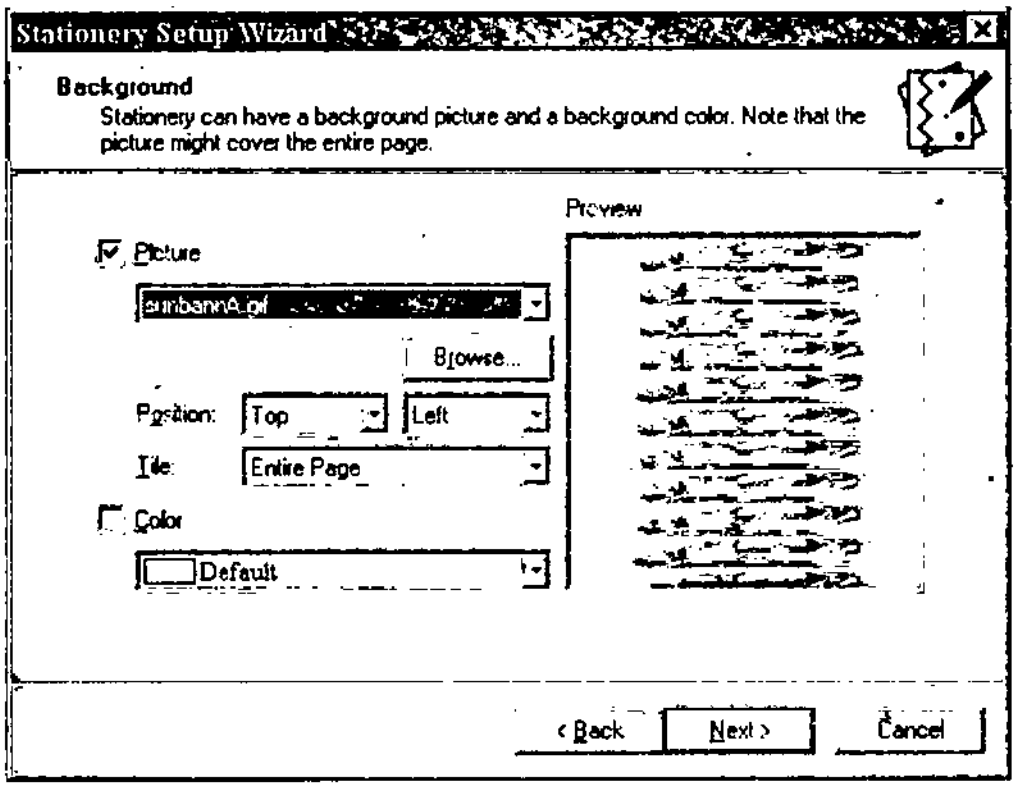

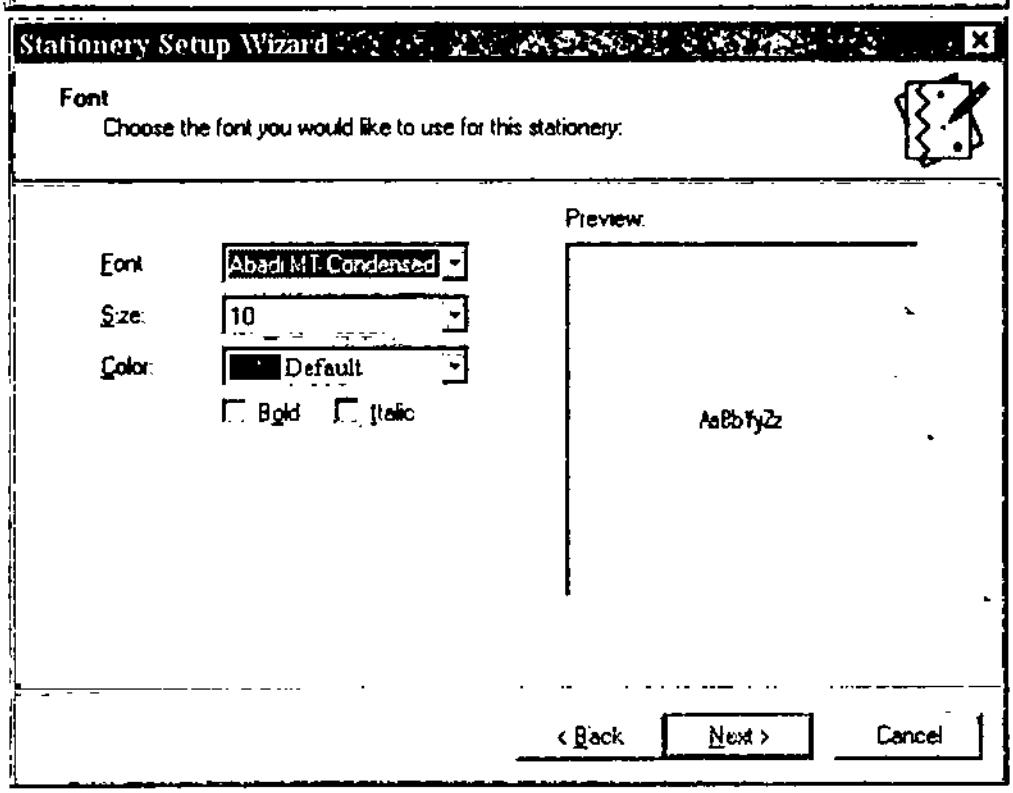

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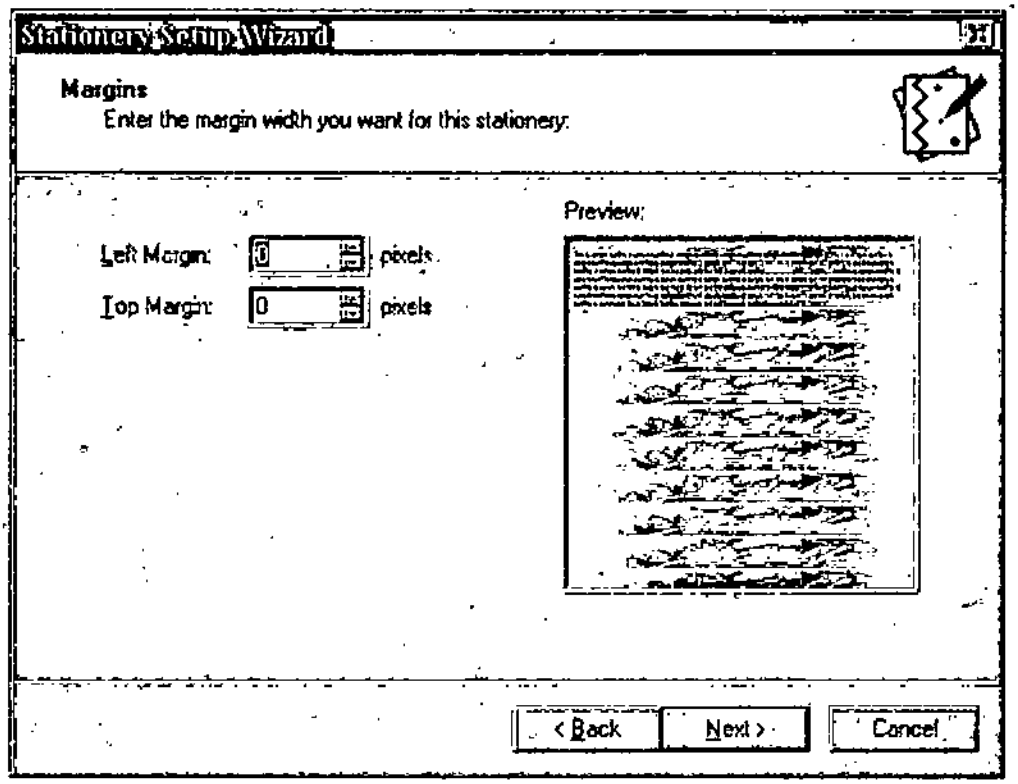
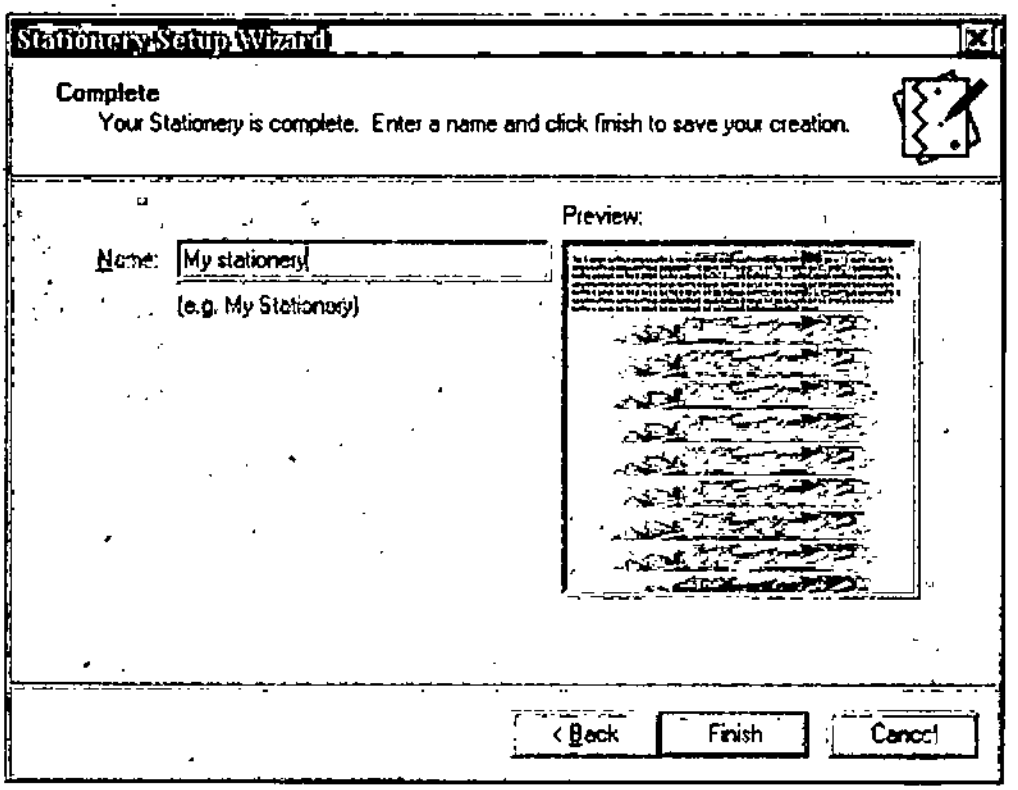




Creating a Signature

You can have Outlook add a signature, a special tag line, to each message you send.

Creating Default Signature

1. From the Tools menu, choose Options.

NOTES

2. On the Signature tab of the Options dialog box, click at New.
3. In the Create New Signature dialog box, enter à name for the new signature and click Next.
4. In the Edit Signature dialog box, type the signature that will appear at the end of your message if this signature is selected.
5. You can make this signature as the default signature to appear at all messages sent by you.
6. Click Ok.

Attaching a File or an Item to a Message

You can attach files to your message, and you also attach or include as text the items in your Outlook folders.

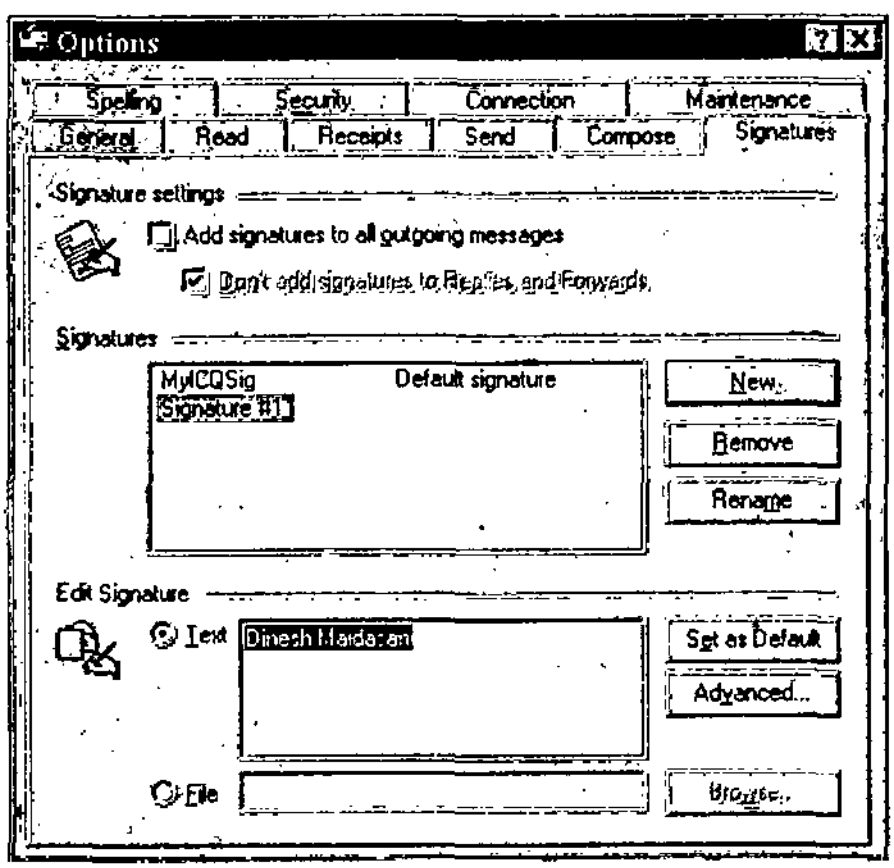
Attaching a File

1. From the Insert menu, choose Attachment or Picture whichever the case maybe.

   Or

   Click the Insert File button on the Standard toolbar.
2. In the Insert File dialog box, select the file you want to attach and click Insert.

The file icon appears at the bottom of the message. The recipient can view the attachment by double-clicking its icon.



NOTES

SUMMARY

1. Computer or network security has been violated when unauthorized access by any party occurs.
2. Network administrators like to believe that their network is secure and those who break into networks may like to believe that they can break into any network.
3. These different areas of computer security are interdependent on each other in order for a network to be secure.
4. The bottom line here is that unless you can remove all the application and operating system problems that allow viruses and intruders to penetrate networks, you can never secure your network.
5. A hacker is no more likely to break the law than anyone else.
6. Spyware is a computer program whose purpose is to spy on your internet activities usually for marketing purposes and usually done by a shady corporate entity.
7. Confidentiality is the information available only to people with rightful access.
8. Contrary to the academic hacker subculture, networking hackers have no inherently close connection to the academic world.
9. In the academic hacker culture, a computer hacker is a person who enjoys designing software and building programs with a sense for aesthetics and playful cleverness.
10. Academic hackers usually work openly and use their real name, while computer security hackers prefer secretive groups and identity-concealing aliases.
11. The early hobbyist hackers built their home computers themselves, from construction kits.
12. All Internet Service Provider allow you to open an e-mail account.
13. In Outlook Express, Inbox is used for incoming messages..
14. Outbox is used for sending messages.
15. Sent items stores all messages previously sent.
16. Deleted items has the messages which are marked for deletion.
17. Drafts has all the half completed messages.
18. To cut down the time of writing a message, lots of abbrevations are used.
19. You can even use similes in your message.
20. Various parts of the e-mail text are: Headers, To, From, Subject, Cc, Bcc, Attachments, Date and Body.
21. Using Send/Receive button on the standard toolbar, you can send and receive mail.
22. Various type of messages are: Unread message, High importance message, Low importance message, Message that has been read, Message with attachment, Message that has been forwarded, Message that has been replied to, Meeting for which response is requested and Follow-up flag.
23. For reading the message you can use Ctrl + O.
24. You can click the RE button for replying to a message.
25. By using FW you can forward a message.
26. You can delete a message by Ctrl+D.
27. You can create your own stationery for sending message.
28. You can create and put your own signature in messages.
29. You can attach a file to your message.
30. You can even attach a picture to your message.

SELF ASSESSMENT QUESTIONS

1. What is a Network Security?
2. What is the need for network security?
3. Who are hackers?

4. What is meant by physical security?
5. Who are academic hackers?
6. What is the difference between networking hackers and academic hackers?
7. What is e-mail?
8. How would create your own e-mail account?
9. Which are the different folders in Outlook Express?
10. Which type of messages you have in Outlook Express?
11. Describe the process of replying to a message.
12. How would you forward a message?
13. How would you create your own stationery?
14. How would you create your own signature?
15. Describe the process of attaching a file to your message.
16. Define the following terms:

Protocol	Confidentiality
Integrity	Availability
Verification - nonrepudiation	Authentification
Spyware	Malware

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Short Questions with Answers

1. What are the needs for network security?

Ans. These include:

- Damage or destruction of computer systems.
- Damage or destruction of internal data.
- Loss of sensitive information to hostile parties.
- Use of sensitive information to steal items of monitary value.
- Use of sensitive information against the organization's customers which may result in legal action by customers against the organization and loss of customers.
- Damage to the reputation of an organization.
- Monitary damage due to loss of sensitive information, destruction of data, hostile use of sensitive data, or damage to the organization's reputation.

2. Who is a hacker?

Ans. In a security context, a Hacker is someone who is involved in computer security/insecurity, specializing in the discovery of exploits in systems (for exploitation or prevention), or in obtaining or preventing unauthorized access to systems through skills, tactics and detailed knowledge.

3. Describe the various components of an e-mail organization.

Ans. With reference to the e-mail program named Outlook Express, the messages in the software can be roughly organized in the following folders:

Inbox	For all incoming messages
Outbox	For all messages queued for sending
Sent Items	For all messages previously sent
Deleted items	For all messages marked for deletion
Drafts	For all messages which are pending completion

4. Describe the various parts of e-mail message.

Ans. Like a formal letter e-mail also is divided into various sections. In a good business letter, you would have, date, the name and address of the person to whom the letter is addressed, salutation, text matter, sender's name, etc. Most of these you would find here too but with some difference. Lets see what an e-mail is supposed to have.

Headers It is very important that your header must be loud and clear for it can be seen and read from a distance too.

To The To: field contains the e-mail address of the person to whom you are sending the

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e-mail. If you are sending e-mail to someone in your own domain, you do not need to include @ domain.

From In most of the cases, you would not be typing this, since it would be taken from the system. It in fact includes your e-mail address.

Subject The Subject: field should contain a very short description (20-30 characters) of what your message is about. This field can also be called Subject of the Message: or simply message. It is not compulsary to have a subject.

Cc You can send the copy of the same e-mail to another person. For this you have to mention the e-mail address of the second receipt. The software would take care that the same message is sent to the second recepient too.

Bcc It is the short form of Blind Carbon Copy, which gives you a way of copying and transmitting an e-mail message to another person, without the first person (the main addressee), knowing about it.

Attachments Using this option you can attach a file created in other software along with this e-mail. In most of the cases while applying for a job, the letter is typed in the e-mail text and the bio-data which is mostly in some word processing mode, attached to it.

Date Date and time are usually taken from the system. You need not type it.

Body The body of an e-mail message is simply the text that you want to send to the other person.

5. Which flags are associated with an e-mail message?

Ans. The messages in the Inbox can have the following symbols attached to them.

- Unread message
- High importance message
- Low importance message
- Message that has been read
- Message with attachment
- Message that has been forwarded
- Message that has been replied to
- Meeting for which response is requested
- Follow-up flag

6. What is stationery used for in an e-mail message?

Ans. When you send a message to somebody, Outlook sees to it that it should have your distinctive features on it. It provides you with the stationery which you can add to your message page. What's more it even gives you an option of putting your signature. You can also set flags to call attention of important messages and to request follow-up actions such as asking the recipient to reply to the message before a certain date.

Further Readings

1. Computer Networks: Ajit Kumar Singh, Firewal Media.
2. Data and Computer Network Communication: Prof. Shashi Banzai, Firewal Media.
3. TCP / IP and Distributed System: Vivek Archarya, Firewal Media.
4. Networking: Balvir Singh, Firewal Media.
5. Elements of Data Communication and Networks: S. A. Amutha Jeevakumari, University Science Press.
6. Wide Area Networks: Navneet Sharma, Firewal Media.
7. Data Communication System: Monika Khuaran, Firewal Media.
8. Computer Network: Bharat Bhushan Agarwal and Sumit Prakash Tayal, University Science Press.