Paper Name :		**Computer Network**
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Chapter Title :		**Introduction**

Objective : To understand networking concepts, connection oriented and connection less communication, network topologies, concept of LAN, MAN, WAN ,lastly and Analysis & comparison of OSI and TCP/IP reference model for data communication.

1.1 INTRODUCTION

Earlier, computer networks consisted of mainframes in an enclosure. Input was in the form of punch cards read by card readers and output was in the form of printed results via local printers. Local terminals were mostly used for control and programming input. All processing was on a batch basis, rather than being interactive. In other words, the input was transmitted in a batch from a card reader over a short circuit to the processor, the processor processed the program in a batch and the output to the printer was in a batch. The first true mainframe was the IBM 360, introduced in 1964. Over time, input to the

mainframe was extended to multiple users at dumb terminals that connected to ports on the mainframe through terminal controllers, or cluster controllers.

In parallel to the development of data networking, the computers began to change. Computers became more powerful as processor speeds increased with the development of faster microprocessors on silicon chips. Memory became more available as chip technology and hard drive technology improved. Additionally, computers became smaller and less expensive, to the point that the typical desktop PC is equivalent to an early mainframe that would have filled a moderate-size office building. As a result, all of this computing power and storage capability on all of these desktops would lead to a need to network those devices within the workplace. It has been estimated that majority of data transfer is confined to the workplace, while only small percentage travels to remote places. Therefore, it is clear that PC users need to share access to hosts, databases, printers, etc. LANs provide a solution to that requirement.

Robert M. Metcalfe and his associates at the Xerox Palo Alto Research Center (Xerox PARC) first conceived LAN technology. Later on, Xerox commercialized the technology and named it The Xerox Wire. When Digital Equipment Corporation (DEC), Intel and Xerox corporate to standardize the technology in 1979, they named it to Ethernet. Ethernet quickly became a de facto standard. Ethernet and LANs were officially recognized when the IEEE established Project 802 at the request of members. In the end of 1982, the first standard was published and circulated. Ethernet, clearly, is still the most popular LAN standard.

1.1.1 What is Network ?

Tenenbaum defines a network as an interconnected collection of autonomous computers. Two computers are said to be interconnected if they are capable of exchanging information. Central to this definition is the fact that the computers are autonomous. This means that no computer on the network can start, stop, or control another.

Advantages

Many organizations. already have a substantial number of computers, often located far apart. For example, a company .with many offices may have a computer at each location to keep track of customer orders, monitor sales, and do the local payroll. Previously, each of these computers may have worked in isolation from others but at some point, management decided to connect them together information about entire company. In general we can refer to it as.

(I) RESOURCE SHARING. The aim is to make all programs, data and peripherals available to anyone on the network irrespective of the physical location of the resources and the user.

(ii) RELIABILITY. A file can have copies on two or three different machines, so if one of . them is unavailable (hardware crash), the other copies could be used. For military, banking, air reservation and many other applications it is of great importance.

(iii) COST FACTOR. Personal computers have better price/performance ratio than micro computers. So it is better to have PC's, one per user, with data stored on one shared file server machine.

(iv) COMMUNICATION MEDIUM. Using a network, it is possible for managers, working far apart, to prepare financial report of the company. The

changes at one end can be immediately noticed at another and hence it speeds up co-operation among them.

1.1.2 ELEMENTARY TERMINOLOGY OF NETWORKS

It is some time to learn about the components/terms mostly used in networking. Whenever we talk about a network it includes. the hardware and the software that make up the network. Now let us have a look at some typical hardware components of network.

Nodes (Workstations)

The term nodes refers to the computers that are attached to a network and are seeking to share the resources of the network. Of course, if there were no nodes (also called workstations), there would be no network at all.

A computer becomes a workstation of a network as soon as it is attached to a network.

Server

Def:. A computer that facilitates "the sharing of data" software" and hardware - resources (e.g. "printers" modems etc,) on the network" is termed as a SERVER.

On small networks, sometimes, all the shareable stuff (like files, data, software etc.) is stored on the server. A network can have more than one server also. Each server has a unique name on the network and all users of network identify the server by its unique name. Servers can be of two types: non-dedicated and dedicated servers.

Non-dedicated Servers:. On small networks, a workstation that can double up as a server, is known as non-dedicated server since it is not

completely dedicated to the cause of serving. Such servers can facilitate the resource-sharing among workstations on a proportionately smaller scale. Since one computer works as a workstation as well as a server, it is .slower and requires. more memory. The (small) networks using such a server are known as peer-to-peer networks.

Dedicated Servers:. On bigger network installations, there is a computer reserved for server's job and its only job is to help workstations access data, software and hardware resources. It does not double-up as a workstation and such a server is known as dedicated server. The networks using such a server are known as master-slave networks.

On a network, there may be several servers that allow workstations to share specific resources. For example, there may be a server exc1usivelyfor serving files-related requests like storing files, deciding about their access privileges and regulating the amount of space allowed for each user. This server is known as file server. Similarly, there may be printer server and modem server. The printer server takes care of the printing requirements of a number of workstations and the modem server helps a group of network users use a modem to transmit long distance messages.

Network Interface Unit (NIU)

Def: A NETWORK INTERFACE UNIT is an interpreter that helps to establish communication between the server and workstations.

A standalone computer (a computer that is not attached to a network) lives in its own world and carries out its tasks with its own inbuilt resources. But as soon as it becomes a workstation, it needs an interface to help establish a

connection with the network because without this, the workstations will not be able to share network resources.

The network-interface-unit is a device that is attached to each of the workstations and the server, and helps the workstation to establish the all-important connection with the network. Each network-interface-unit that is attached to a workstation has a unique number identifying it which is known as the node address. The NIU is also called Terminal Access Point (TAP). Different manufacturers have different names for the interface.

Computer networks can be used for numerous services, both for companies and for individuals. For companies, networks of personal computers using shared servers often provide access to corporate information. Typically they follow the client-server model, with client workstations on employee desktops accessing powerful servers in the machine room. For individuals, networks offer access to a variety of information and entertainment resources. Individuals often access the Internet by calling up an ISP using a modem, although increasingly many people have a fixed connection at home. An up-and-coming area is wireless networking with new application such as mobile e-mail access and m-commerce.

1.1.3 Applications & Uses of Networks

In the short time they have been around, data communication networks have become an indispensable part of business, industry, and entertainment. Some of the network applications in different fields are the following:

- **Marketing and sales.** Computer networks are used extensively in both marketing and sales organizations. Marketing professionals use them to collect, exchange, and analyze data relating to customer needs and product

development cycles. Sales applications include teleshopping, which uses order-entry computers or telephones connected to an order-processing network, and on-line reservation services for hotels, airlines, and so on.

- **Financial services.** Today's financial services are totally dependent on computer networks. Applications include credit history searches, foreign exchange and investment services, and electronic funds transfer (EFT), which allows a user, to transfer money without going into a bank (an automated teller machine is a kind of electronic funds transfer; automatic paycheck deposit is another).
- **Manufacturing.** Computer networks are used today in, many aspects of manufacturing, including the manufacturing process itself. Two applications that use networks to provide essential services are computer-assisted design (CAD) and computer-assisted manufacturing (CAM), both of which allow multiple users to work on a project simultaneously.
- **Electronic messaging**: Probably the most widely used network application is electronic mail (e-mail).
- **Directory services:** Directory services allow lists of files to be stored in a central location to speed worldwide search operations.
- **Information services:** Network information services include bulletin boards and data banks. A World Wide Web site offering the technical specifications for a new product is an information service.
- **Electronic data interchange (EDI):** EDI allows business information (including documents such as purchase orders and invoices) to be transferred without using paper.
- **Teleconferencing:** Teleconferencing allows conferences to occur

without the participants being in the same place. Applications include simple text conferencing (where participants communicate through their keyboards and computer monitors). voice conferencing (where participants at a number of locations communicate simultaneously over the phone) and video conferencing (where participants can see as well as talk to one another).

- **Cellular telephone:** In the past two parties wishing to use the services of the telephone company had to be linked by a fixed physical connection. Today's cellular networks make it possible to maintain wireless phone connections even while traveling over large distances.
- **Cable television**: Future services provided by cable television networks may include video on request, as well as the same information, financial, and communications services currently provided by the telephone companies and computer networks.

1.2 History (Development) of Computer Networks

Each of the past three centuries has been dominated by a single technology. The 18th century was the era of the great mechanical systems accompanying the Industrial Revolution. The 19th century was the age' of the steam engine. During the 20th century, the key technology was information gathering, processing, and distribution. Among other developments, we saw the installation of worldwide telephone networks, the invention of radio and television, the birth and unprecedented growth of the computer industry, and the launching of communication satellites.

As a result of rapid technological progress, these areas are rapidly converging and the differences between collecting, transporting, storing, and processing information are quickly disappearing. Organizations with hundreds

of offices spread over a wide geographical area routinely expect to be able to examine the current status of even their most remote output at the push of a button. As our ability together, process and distribute information grows, the demand forever more sophisticated information processing grows even faster.

Although the computer industry is still young compared to other industries (e.g., automobiles and air transportation), computers have made spectacular progress in a short time. During the first two decades of their existence, computer systems were highly centralized, usually within a single large room. Not infrequently, this room had glass walls, through which visitors could gawk at the great electronic wonder inside. A medium-sized company or university might have had one or two computers, while. large institutions had at most a few dozen. The idea that within twenty years equally powerful computers smaller than postage stamps would be mass produced by the millions was pure science fiction.

The merging of computers and communications has had a profound influence on the way computer systems are organized. The concept of the "computer center" as a room with a large computer to which users bring their work for processing is now totally obsolete. The old model of a single computer serving all of the organization's computational needs has been replaced by one. in which a large, number of separate but interconnected computers do the job. These systems are called computer networks. The design and organization of these networks are the subjects of this book.

Throughout the book we will use the term "computer network" to mean a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange

information. The connection need not be via a copper wire; fiber optics, microwaves, infrared, and communication satellites can also be used. Networks come in many sizes, shapes and forms, as we will see later. Although it may sound strange to some people, neither the Internet nor the World Wide Web is a computer network. By the end of this book, it should be clear why. The quick answer is- the Internet is not a single network but a network of networks and the Web is a distributed system that runs on top of the Internet.

There is considerable confusion in the literature, between a computer network and a distributed system. The key distinction is that in a distributed system, a collection of independent computers appears to its users as a single coherent system. Usually, it has a single model or paradigm that it presents to the users. Often a layer of software on top of the operating system, called middleware, is responsible for implementing this model. A well-known example of a distributed system is the World Wide Web, in which everything looks like a document (Web page).

In a computer network, this coherence, model, and software are absent. Users are exposed to the actual machines, without any attempt by the system to make the machines look and act in a coherent way. If the machines have different hardware and different operating systems, that is fully visible to the users. If a user wants to run a program on a remote machine, he t has to log onto that machine and run it there.

In effect, a distributed system is a software system built on top of a network. The software gives it a high degree of cohesiveness and transparency. Thus, the distinction between a network and a distributed system lies with the software (especially the operating system), rather than with the hardware.

1.3 Network Topologies

Def: The pattern of interconnection of nodes in a network is called the TOPOLOGY.

The selection of a topology for a network cannot be done in isolation as it affects the choice of media and the access method used. There are a number of factors to consider in making this choice, the most important of which are setout below:

1. Cost: For a network to be cost effective, one would try to minimize installation cost. This may be achieved by using well understood media and also, to a lesser extent, by minimizing the distances involved. .

2. Flexibility: Because the arrangement of furniture, internal walls etc. in offices is often subject to change, the topology should allow for easy reconfiguration of the network. This involves moving existing nodes and adding new ones.

3. Reliability: Failure in a network can take two forms. Firstly, an individual node can malfunction. This is not nearly as serious as the second type default where the network itself fails to operate. The topology chosen for the network can help by allowing the location of the fault to be detected and to provide some means of isolating it.

1.3.1 The Star Topology

This topology consists of a central node to which all other nodes are connected by a single path. It is the topology used in most existing information networks involving data processing or voice communications. The most common example of this is IBM 370 installations. In this case multiple 3270 terminals are connected to either a host system or a terminal controller.

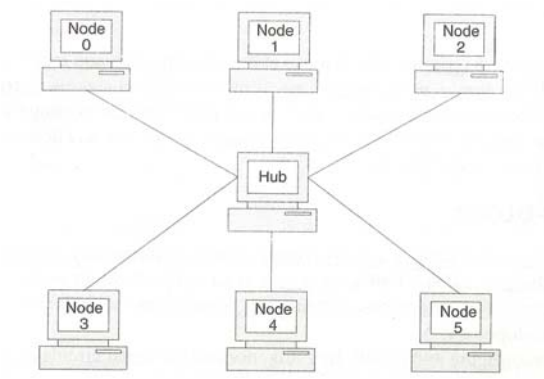


Fig 1.1 Star Topology

Advantages of the Star Topology

1. Ease of service: The star topology has a number of concentration points (where connections are joined). These provide easy access for service or reconfiguration of the network.

2. One device per connection: Connection points in any network are inherently prone to failure in the star topology, failure of a single connection typically involves disconnecting one node from an otherwise fully functional network.

3. Centralized control/problem diagnosis: The fact that the 'central node is connected directly to every other node in the network means that faults are easily detected and isolated. It is a simple matter to disconnect failing nodes from the system.

4. Simple access protocols: Any given connection in a star network involves only the central node. In this situation, contention for who has control of the medium for the transmission purposes is easily solved. Thus in a star network, access protocols are very simple.

Disadvantages of the Star Topology.

1. Long cable length: Because each node is directly connected to the center, the star topology necessitates a large quantity of cable as the cost of cable is often small, congestion in cable ducts and maintenance and installation problems can increase cost considerably.

2. Difficult to expand: The addition of a new node to a star network involves a connection all the way to the central node.

3. Central node dependency: If the central node in a star network fails, the entire network is rendered inoperable. This introduces heavy reliability and redundancy constraints on this node.

The star topology has found extensive application in areas where intelligence in the network is concentrated at the central node.

Examples of Star Topology

Asynchronous Transmission Mode (ATM)

Asynchronous Transfer Mode (ATM) is an International Telecommunication Union - Telecommunication Standardization Sector (ITU- T) standard for cell relay wherein information for multiple service types, such as voice, video, or data, is conveyed in small, fixed-size cells. ATM networks are connection-oriented.

ATM is the emerging standard for communications. It can provide medium to high bandwidth and a virtual dedicated link between ends for the delivery of real-time voice, data and video. Today, in most instances, separate networks are used to carry voice, data and video information mostly because these traffic types have different characteristics. For instance, data traffic tends to be "bursty". Bursty means that data traffic does not need to communicate for

an extended period of time and whenever it needs, it communicates large quantities of information as fast as possible. Voice and video, on the other hand, tend to be more even in the amount of information required but are very sensitive to when and in what order the information arrives. With ATM, separate networks will not be required. ATM is the only standards-based technology that has been designed from the beginning to accommodate the simultaneous transmission of data, voice and video. Using ATM, information to be sent is segmented into fixed length cells, transported to and re-assembled at the destination. The ATM cell has a fixed length of 53 bytes. Being fixed length allows the information to be transported in a predictable manner. This predictability accommodates different traffic types on the same network. The cell is broken into two main sections, the header and the payload. The payload (48 bytes) is the portion which carries the actual information, i.e. voice, data, or video. The header (5 bytes) is the addressing mechanism.

1.3.2 The Bus or Linear Topology

Another popular topology for data networks is the linear. This consists of a single length of the transmission medium (normally coaxial cable) onto which the various nodes are attached. The topology is used in traditional data communication network where the host at one end of the bus communicates with several terminals attached along its length.

The transmission from any station travels the length of the bus, in both directions, and can be received by another stations. The bus has terminators at either end which absorb the signal, removing it from the bus.

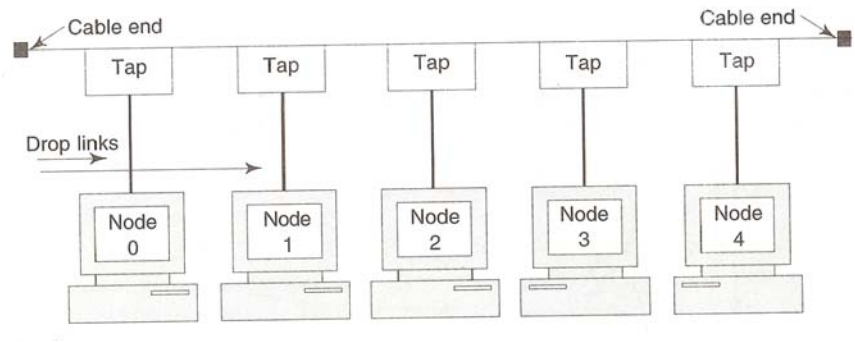


Fig. 1.2 Bus Topology

Data is transmitted in small blocks, known as packets. Each packet has some data bits, plus a header containing its destination address. A station wanting to transmit some data sends it in packets along the bus. The destination device, on identifying the address on the packets, copies the data onto its disk.

Advantages of the Linear Topology

1. **Short cable length and simple wiring layout:** Because there is a single common data path connecting all nodes, the linear topology allows a very short cable length to be used. This decreases the installation cost, and also leads to a simple, easy to maintain wiring layout.

2. **Resilient Architecture:** The LINEAR architecture has an inherent simplicity that makes it very reliable from a hardware point of view. There is a single cable through which all the data propogates and to which all nodes are connected.

3. **Easy to extend:** Additional nodes can be connected to an existing bus network at any point along its length. More extensive additions can be achieved by adding extra segments connected by a type of signal amplifier known as

repeater.

Disadvantages of the Bus (Linear) Topology

1. **Fault diagnosis is difficult:** Although simplicity of the bus topology means that there is very little to go wrong, fault detection is not a simple matter. Control of the network is not centralized in any particular node. This means that detection of a fault may have to .be performed from many points in the network.

2. **Fault isolation is difficult**: In the star topology, a defective node can easily be isolated from the network by removing its connection at the center. If a node is faulty on the bus, it must be rectified at the point where the node is connected to the network.

3. **Repeater configuration:** When BUS type network has its backbone extended using repeaters, reconfiguration may be necessary.

4. **Nodes must be intelligent:** Each node on the network is directly connected to the central bus. This means that some way of deciding who can use the network at any given time must be performed in each node.

Examples of Bus Topology

Ethernet: Ethernet is the least expensive high-speed LAN alternative. It transmits and receives data at a speed of 10 million bits per second. Data is transferred between wiring closets using either a heavy coaxial cable (thick net) or fiber optic cable. Thick net coaxial is still used for medium-long distances where medium levels of reliability are needed. Fiber goes farther and has greater reliability but a higher cost. To connect a number of workstations within the same room, a light duty

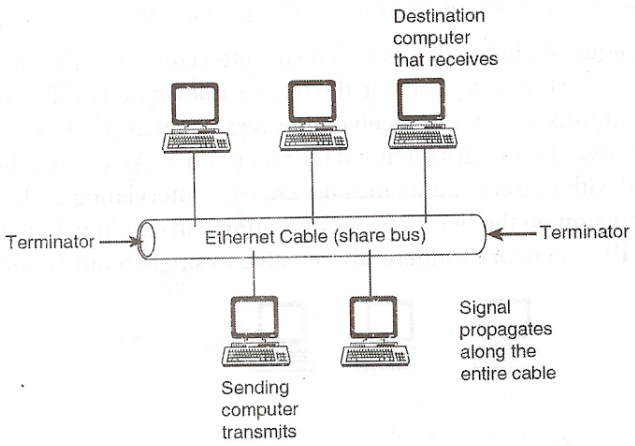


Fig 1.3 Signal flow across an Ethernet

coaxial cable called thin net is commonly used. These other media reflect an older view of workstation computers in a laboratory environment. Figure shows the scheme of Ethernet where a sender transmits a modulated carrier wave that propagates from the sender toward both ends of the cable.

Ethernet was first designed and installed by Xerox Corporation at its Palo Atto Research Centers (PARC) in the mid-1970. In the year 1980, DEC Intel and Xerox came out with a joint specification which has become the de facto standard. Ethernet from this period is often called DIX after its corporate sponsors Digital, Intel, and Xerox. Ethernet as the most popular protocol for LAN technology has been further discussed in chapter on Network Protocols.

1.3.3 The Ring or Circular Topology

The third topology that we will consider is the ring or circular. In this case, each node is connected to two and only two neighbouring nodes. Data is

accepted from one of the neighbouring nodes and is transmitted onwards to another. Thus data travels in one direction only, from node to node around the ring. After passing through each node, it returns to the sending node, which removes it.

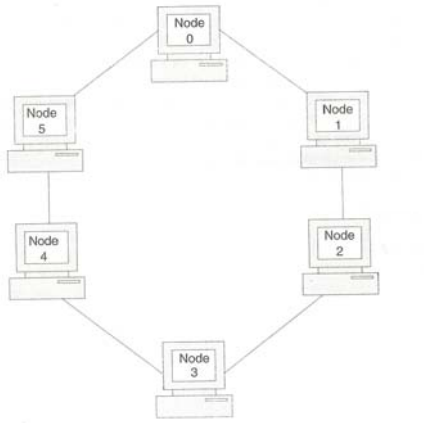


Fig. 1.4 Ring Topology

It is important to note that data 'passed through' rather than 'travels past' each node. This means that the signal may be amplified before being 'repeated' on the out ward channel.

Advantages of the Ring Topology

1. Short cable length. The amount of cabling involved in a ring topology is comparable to that of a bus and is small relative to that of a star. This means that less connections will be needed, which will in turn increase network reliability.

2. No wiring closet space required. Since there is only one cable connecting each node to its immediate neighbours, it is not necessary to allocate space in the building for wiring closets.

3. Suitable for optical fibers. Using optical fibers offers the possibility of very high speed transmission. Because traffic on a ring travels in one direction, it is easy to use optical fibers as a medium of transmission.

Disadvantages of the Ring Topology

1. Node failure causes network failure. The transmission of data on a ring goes through every connected node on the ring before returning to the sender. If one node fails to pass data through itself, the entire network has failed and. no traffic can. flow until the defective node has been removed from the ring.

2. Difficult to diagnose faults. The fact that failure of one node will affect all others has serious implications for fault diagnosis. It may be necessary to examine a series of adjacent nodes to determine the faulty one. This operation may also require diagnostic facilities to be built into each node.

3. Network reconfiguration is difficult. It is not possible to shut down a small section of the ring while keeping the majority of it working normally.

Examples of Token Ring Topology

IBM Token Ring: A local area network access mechanism and topology in which all stations actively attached to the bus listen for a broadcast token or supervisory frame. Stations wishing to transmit must receive the token before doing so. After a station finishes transmission, it passes the token to the next node in the ring. It operates at 16 Mbps and can be used with computers from IBM, computers from other vendors and peripheral devices such as printers.

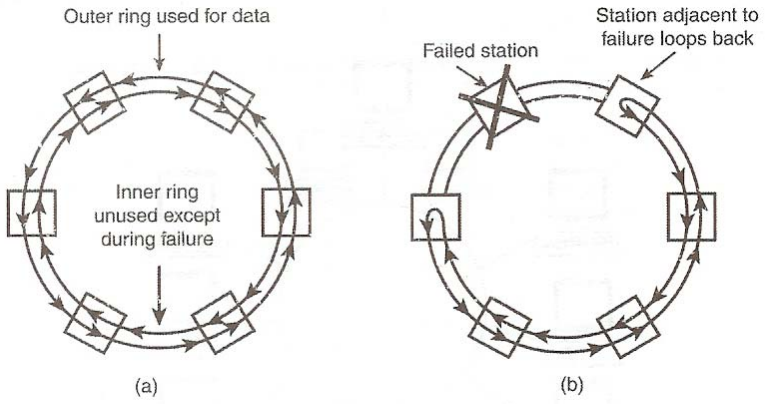


Figure 1.5 (a) FDDI network with counter ring
(b) the same network after a station has failed

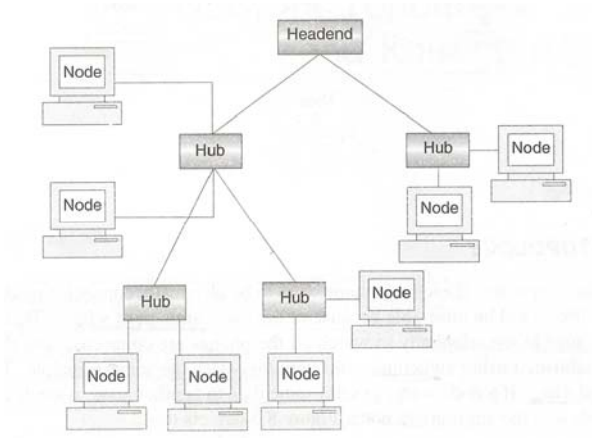
The IEEE published its standard as IEEE 802.5 for token ring in 1984 after IBM Token Ring specifications. Therefore, the IEEE 802.5 specification is almost identical to, and completely compatible with, IBM Token Ring network. The objective of Token Ring is to provide reliability at all functional levels of LAN. This topology addresses issues like systematic wiring, ease in configuration and maintenance, fault tolerance and redundancy. It is further discussed in chapter on Network Protocols.

FDDI is a reliable, high-speed network for high traffic. It can transport data at a rate of 100 Mbps and can support up to 500 stations on a single network. FDDI was designed to run through fiber cables, transmitting light pulses to convey information back and forth between stations, but it can also run on copper using electrical signals. A related technology, Copper Distributed Data Interface (CDDI) works FDDI using copper cables instead of fiber cables. FDDI maintains a high reliability because FDDI networks consist

of two counter-rotating rings as shown in Figure 1.5(a). These rings work to back each other up, so should something go wrong on the network, an alternate way to get the data can be found. Figure 1.5(b) illustrates the data flow when one station is failed. After a station fails, adjacent stations use the reverse path to form a closed ring. FDDI is also considered reliable because it has mechanisms to fix its own problems.

1.3.4 Tree Topology

Tree topology can be derived from the star topology. Tree has a hierarchy of various bubs, like you have branches in a tree; hence the name. Figure 1.6 in this case every node is connected to some hub. However, only a few nodes are connected directly to the central hub.



The central hub contains a repeater, which looks at the incoming bits and regenerates them afresh as the full blown signals for 0 or 1 as required. This allows the digital signals to traverse over longer distances. Therefore, the

central hub is also called active hubs. The tree topology also contains many secondary hubs, which may be active hubs or passive hubs. The merits and demerits of tree topology are almost similar to those of the star topology.

1.3.5 Hybrid Topology

Hybrid topology is one that uses two or more of the topologies mentioned above together, Figure 1.7 depicts this. In this case, the bus, star and ring topologies are used to create this hybrid topology. There are multiple ways in which this can be created. In practice, many networks are quite complex but they can be reduced to some form of hybrid topology.

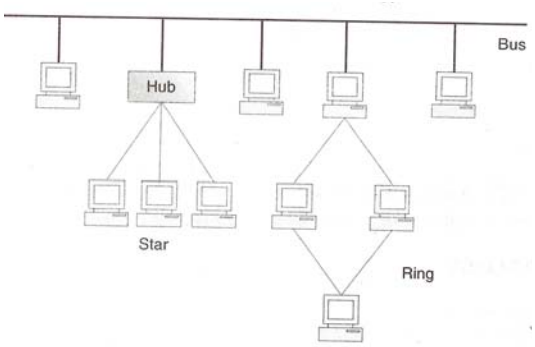


Fig. 1.7 Hybrid Topology

1.4 NETWORK HARDWARE

It is now time to turn our attention from the applications and social aspects of networking (the fun stuff) to the technical issues involved in network design (the pi work stuff). There is no generally accepted taxonomy into which all computer networks fit, but two dimensions stand out as important: transmission technology and scale. We will now examine each of these in turn.

Broadly speaking, there are two types of transmission technology that

are in widespread use. They are as follows:

1. Broadcast links.
2. Point-to-point links.

Broadcast networks have a single communication channel that is shared by all the machines on the network. Short messages, called packets in certain contexts, sent by any machine are received by all the others. An address field within the packet specifies the intended recipient. Upon receiving a packet, a machine checks the address field. If the packet is intended for the receiving machine, that machine processes the packet; if the packet is intended for some other machine, it is just ignored.

As an analogy, consider someone standing at the end of a corridor with many rooms off it and shouting "Watson, come here. I want you." Although the packet, may actually be received (heard) by many people, only Watson responds. The others just ignore it. Another analogy is an airport announcement asking all flight 644 passengers to report to gate 12 for immediate boarding.

Broadcast systems generally also allow the possibility of addressing a packet to all destinations by using a special code in the address field.. When a packet with this code is transmitted, it is received and processed by every machine on the network. This mode of operation is called broadcasting. Some broadcast systems also support transmission to a subset of the machines, something known as multicasting. One possible scheme is to reserve one bit to indicate multicasting. The remaining (n – 1) address bits can hold a group number. Each machine can "subscribe" to any or all of the groups. When a packet is sent to a certain group, it is delivered to all machines subscribing to that group.

In contrast, point-to-point networks consist of many connections between individual pairs of machines. To go from the source to the destination, a packet on this type of network may have to, first visit one or more intermediate machines. Often multiple routes, of different lengths, are possible, so finding good ones is important in point-to-point networks. As a general rule (although there are many exceptions), smaller, geographically localized networks tend to use broadcasting, whereas larger networks usually are point-to-point. Point-to-point transmission with one sender and one receiver is sometimes called unicasting.

An alternative criterion for classifying networks is their scale. In Fig. 1-6 we classify multiple processor systems by their physical size. At the top are the personal area networks, networks that are meant for one person. For example, a wireless network connecting a computer with its mouse, keyboard, and printer is a personal area network. Also, a PDA that controls the user's hearing aid or pacemaker fits in this category. Beyond the personal area networks come longer range networks. These can be divided into local, metropolitan, and wide area networks. Finally, the connection of two or more networks is called an internetwork.

1m	Square meter	Personal area network
10m	Room	Local area network
100 m	Building	
1 km	Campus	
10km	City	Metropolitan area network
100 km	Country	wide area network
1000 km	Continent	
10,000 km	Planet	the internet

Fig.1.8 Classification of interconnected processors by scale.

The worldwide Internet is a well-known example of an internetwork. Distance is important as a classification metric because different techniques are used at different scales. In this chapter we will be concerned with networks at all these scales. Below we give a brief introduction to network hardware.

1.4.1 Local Area Network (LAN)

In the mainframe and minicomputer environment each user is connected to the main system through a dumb terminal that is unable to perform any of its own processing tasks. In this computing environment, processing and memory are centralized. However, this type of computerization has its merits but the major disadvantage is that the system could get easily overloaded as the number of users and consequently terminals increase. Second, most of the information is centralized to one group of people, the systems professionals rather than the end users. This type of centralized processing system differs from the distributed processing system used by LANs. In distributed processing system, most of the processing is done in the memory of the individual PCs or workstations besides sharing expensive computer resources like software, disk files, printers and plotters, etc.

There may arise a question why PCs cannot be connected together in point-to-point manner. The point-to-point scheme provides separate communication channels for each pair of computers. When more than two computers need to communicate with one another, the number of connections grows very quickly as number of computers increases. Figure 1.9 illustrates that two computers need only one connection, three computers need three connections and four computers need six connections.

The Figure 1.9 illustrates that the total number of connections grows more rapidly than the total number of computers. Mathematically, the number of connections needed for N computers is proportional to the square of N:

Point-to-point connections required = (W-N)/2

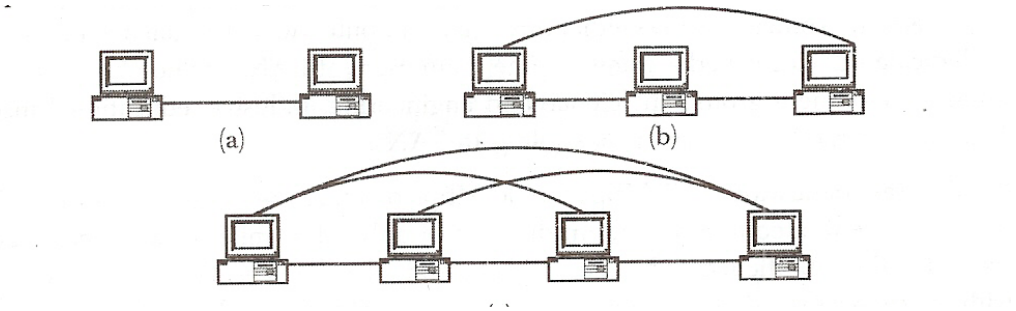


Figure 1.9 (a), (b), (c) Number of connections for 2,3,4 computers, respectively

Adding the Nth computer requires N-l new connections, which becomes a very expensive option. Moreover, many connections may follow the same physical path. Figure 1.10 shows a point-to-point connection for five computers located at two different locations, say, ground and first floor of a building.

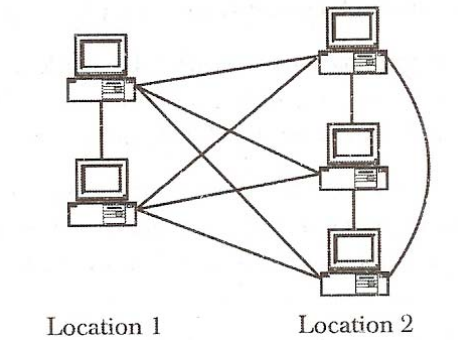


Figure 1.10 Five PCs at two different locations

As there are five PCs, therefore, total ten connections will be required for point-to-point connection. Out of these ten connections, six are passing through the same location thereby making point-to-point connection an expensive one. By increasing the PC by one in the above configuration at location 2 as shown in Figure 1.10 will increase the total number of connections to fifteen. Out of these connections, eight connections will pass through the same area.

Definition:

Local Area Networks (LANs) are most often described as privately owned networks that offer reliable high speed communication channels optimized for connecting information processing equipment in a limited geographical area, namely, an office building, complex of buildings, or campus.

A LAN is a form of local (limited-distance), shared packet network for computer communications. LANs interconnect computers and peripherals over a common medium in order that users might share access to host computers, databases, files, applications, and peripherals.

LANs in addition to linking the computer equipment available in a particular premises can also provide a connection to other networks either through a computer, which is attached to both networks, or through a, dedicated device called a gateway. The main users of LANs include business organizations, research and development groups in science and engineering, industry, educational institution. The electronic or paperless office concept is possible with LANs.

LANs offer raw bandwidth of 1 Mbps to 100 Mbps or more, although

actual throughput often is much less. LANs are limited to a maximum distance of only a few miles or kilometers, although they may be extended through the use of bridges, routers, and other devices. Data are transmitted in packet format, with packet sizes ranging up to 1500 bytes and more. Mostly, IEEE develops LAN specifications, although ANSI and other standards bodies are also involved.

LAN HARDWARE

In addition to the attached devices also referred to as nodes or stations, LANs may make use of other devices to control physical access to the shared medium to extend the maximum reach of the LAN, and to switch traffic. Such hardware is in the form of NIC/NIU, transceivers, MAU, hubs, bridges, routers, and gateways.

Network Interface Card (NIC)

This is also known as Network Interface Unit (NIU). NIC is printed circuit boards that provide physical access from the node to the LAN medium. The NIC can be fitted into the expansion slot of a PC, or it can exist as a separate box. A standalone, multiport NIC can serve a number of devices, thereby providing an additional level of contention control. A standard IEEE NIC contains a unique hard-coded logical address. Transceivers are embedded in NIC/NIU and MAD. MAU (Media Access Unit, or Multistation Access Unit) are standalone devices that contain NIC in support of one or more nodes.)

The other devices will be explained subsequently when their applications will appear in the respective sections.

1.4.2 Metropolitan Area Network (MAN)

A metropolitan area network (MAN) is designed to extend over an entire city. It may be a single network such as a cable television network, or it may be a means of connecting a number of LANs into a larger network so that resources may be shared LAN-to-LAN as well as device-to-device. For example, a company can use a MAN to connect the LANs in all of its offices throughout a city.

A MAN may be wholly owned and operated by a private company, or it may be a service provided by a public company, such as a local telephone company. Many telephone companies vide a popular MAN service called Switched Multi-megabit Data Services (SMDS).

1.4.3 Wide Area Network (WAN)

A wide area network (WAN) provides long-distance transmission of data, voice, image, and video information over large geographical areas that may comprise a country, a continent, or even the whole world (see Figure 2.18).

In contrast to LANs (which depend on their own hardware for transmission), WANs may utilize public, leased, or private communication devices, usually in combinations, and can therefore span an unlimited number of miles.

A WAN that is wholly owned and used by a single company is often referred to as an enterprise network.

1.4.4 Wireless Networks

Digital wireless communication is not a new idea. As early as 1901, the Italian physicist Guglielmo Marconi demonstrated a ship-to-shore wireless

telegraph, using Morse Code (dots and dashes are binary, after all). Modem digital wireless systems have better performance, but the basic idea is the same.

To a first approximation, wireless networks can be divided into three main categories:

1. System interconnection.
2. Wireless LANs.
3. Wireless WANs.

System interconnection is all about interconnecting the components of a computer using short-range radio. Almost every computer has a monitor, keyboard, mouse, and printer connected to the main unit by cables. So many new users have a hard time plugging all the cables into the right little holes (even though they are usually color coded) that most computer vendors offer the option of sending a technician to the user's home to do it. Consequently, some companies got together to design a short-range wireless network called **Bluetooth** to connect these components without wires. Bluetooth also allows digital cameras, headsets, scanners, and other devices to connect to a computer by merely being brought within range. No cables, no driver installation, just put them down, turn them on, and they work. For many people, this ease of operation is a big plus.

1.4.5 Home Networks

Home networking is on the horizon. The fundamental idea is that in the future most homes will be set up for networking. Every device in the home will be capable of communicating with every other device, and all of them will be accessible over the Internet. This is one of those visionary concepts that nobody asked for (like TV remote controls or mobile phones), but once they

arrived nobody can imagine how they lived without them.

Many devices are capable of being networked. Some. of the more obvious categories (with examples) are as follows:

1. Computers (desktop PC, notebook PC, PDA, shared peripherals).
2. Entertainment (TV, DVD, VCR, camcorder,-camera, stereo, MP3).
3. Telecommunications (telephone, mobile telephone, intercom, fax).
4. Appliances (microwave, refrigerator, clock, furnace, airco, lights).
5. Telemetry (utility meter, smokelburglar alarm, thermostat, babycam).

Home computer networking is already here in a limited way. Many homes already have a device to connect multiple computers to a fast Internet connection. Networked entertainment is not quite here, but as more and more music and movies can be downloaded from the Internet, there will be a demand to connect stereos and televisions to it. Also, people will want to share their own videos with friends and family, so the connection will need to go both ways. Telecommunications gear is already connected to the outside world, but soon it will be digital and go over the Internet. The average home probably has a dozen clocks (e.g., in appliances), all of which have to be reset twice a year when daylight saving time (summer time) comes and goes. If all the clocks were on the Internet, that resetting could be done automatically. Finally, remote monitoring of the home and its contents is a likely winner. Probably many parents would be willing to spend some money to monitor their sleeping babies on their PDAs when they are eating put, even with a rented teenager in the house. While one can imagine a separate network for each application area, integrating all of them into a single network is probably a better idea.

1.4.6 Inter-networks

Many networks exist in the world, often with different hardware and software. People connected to one network often want to communicate with people attached to a different one. The fulfillment of this desire requires that different, and frequently incompatible networks, be connected, sometimes by means of machines called gateways to make the connection and provide the necessary translation, both in terms of hardware and software. A collection of interconnected networks is called an internetwork or internet. These terms will be used in a generic sense, in contrast to the worldwide Internet (which is one specific internet), which we will always capitalize.

An internetwork is formed when distinct networks are interconnected. In our view, connecting a LAN and a WAN or connecting two LAN s forms an internetwork, but there is little agreement in the industry over terminology in this area. One rule of thumb is that if different organizations paid to construct different parts of the network and each maintains its part, we have an internetwork rather than a single network. Also, if the underlying technology is different in different parts (e.g., broadcast versus point-to-point), we probably have two networks.

1.5 Network Software

The first computer networks were designed with the hardware as the main concern and the software as an after thought. This strategy no longer works. Network software s now highly structured. In the following sections we examine the software structuring technique in some details. The method described here forms the keystone of the entire literature and will occur repeatedly later.

1.5.1 Protocol Hierarchies

To reduce their design complexity, most networks are organized as a stack of layers or levels, each one built upon the one below it. The number of layers, the name of each layer, the contents of each layer, and the function of each layer differ from network to network. The purpose of each layer is to offer certain services to the higher layers, shielding those layers from the details of how the offered services

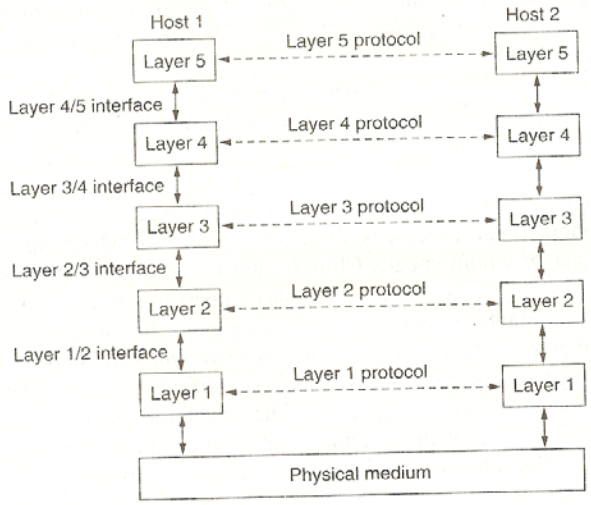


Fig. 1.11 Layers, protocols, and interfaces.

are actually implemented. In a sense, each layer is a kind of virtual machine, offering certain services to the layer above it.

This concept is actually a familiar one and used throughout computer science, where it is variously known as information hiding, abstract data types, data encapsulation, and object-oriented programming. The fundamental idea is that a particular piece of software (or hardware) provides a service to its users but keeps the details of its internal state and algorithms hidden from them.

Layer on one machine carries on a conversation with layer n on another machine. The rules and conventions used in this conversation are collectively known as the layer n protocol. 'Basically, a protocol is an agreement between the communicating parties on how communication is to proceed. As an analogy, When a woman is introduced to a man, she may choose to stick out her hand. He, in turn, may decide either to shake it or kiss it, depending, for example, on whether she is an American lawyer at a business meeting or a European princess at a formal ball. Violating the protocol will make communication more difficult, if not completely impossible. .

A five-layer network is illustrated in Fig. 1.11. The entities comprising the corresponding layers on different machines are called peers. The peers may be processes, hardware devices, or even human beings. In other words, it is the peers that communicate by using the protocol.

In reality, no data are directly transferred from layer n on one machine to layer n on another machine. Instead, each layer passes data and control information to the layer immediately below it, until the lowest layer is reached. Below layer I is the physical medium through which actual communication occurs. In Fig. 1-11, virtual communication is shown by dotted lines and physical communication by solid lines.

Between each pair of adjacent layers is an **interface.** The interface defines which primitive operations and services the lower layer makes available to the upper one. When network designers decide how many layers to include in a network and what each one should do, one of the most important considerations is defining clean interfaces between the layers. Doing so, in turn, requires that each layer perform a specific collection of well-understood

functions. In addition to minimizing the amount of information that must be passed between layers, clear cut interfaces also make it simpler to replace the implementation of one layer with a completely different implementation (e.g., all the telephone lines are replaced by satellite channels) because all that is required of the new implementation is that "it offer exactly the same set of services to its upstairs neighbour as the old implementation did. In fact, it is common that different hosts use different implementations.

A set of layers and protocols is called a network architecture. The specification of architecture must contain enough information to allow an implementer to write the program or build the hardware for each layer so that it will correctly obey the appropriate protocol. Neither the details of the implementation nor the specification of the interfaces is part of the architecture because these are hidden away inside the machines and not visible from the outside. It is not even necessary that' the interfaces on all machines in a network be the same, provided that each machine can correctly use all the protocols. A list of protocols used by a certain system, one protocol per layer, is called a protocol stack.

1.5.2 Connection-Oriented and Connectionless Services

Layers can offer two different types of service to the layers above them: connection-oriented and connectionless. In this section we will look at these two types and examine the differences between them.

Connection-oriented service is modeled after the telephone system. To talk to someone, you pick up the phone, dial the number, talk, and then hang up. Similarly, to use a connection-oriented network service, the service user

first establishes a connection, uses the connection, and then releases the connection. The essential aspect of a connection is that it acts like a tube: the sender pushes objects (bits) in at one end, and the receiver takes them out at the other end. In most cases the order is preserved so that the bits arrive in the order they were sent.

In some cases when a connection is established, the sender, receiver, and subnet conduct a negotiation about parameters to be used, such as maximum message size, quality of service required, and other issues. Typically, one side makes a proposal and the other side can accept it, reject it, or make a counterproposal.

In contrast, **connectionless service** is modeled after the postal system. Each message (letter) carries the full destination address, and each one is routed through the system independent of all the others. Normally, when two messages are sent to the same destination, the first one sent will be the first one to arrive. However, it is possible that the first one sent can be delayed so that the second one arrives first.

Each service can be characterized by a quality of service. Some services are reliable in the sense that they never lose data. Usually, a reliable service is implemented by having the receiver acknowledge the receipt of each message so the sender is sure that it arrived. The acknowledgement process introduces overhead and delays, which are often worth it but are sometimes undesirable.

A typical situation in which a reliable connection-oriented service is appropriate is file transfer. The owner of the file wants to be sure that all the bits arrive, correctly and in the same order they were sent. Very few file transfer customers would prefer a service that occasionally scrambles or loses a

few bits, even if it is much faster.

1.5.3 Service Primitives

A service is formally specified by a set of primitives (operations) available to a user process to access the service. These primitives tell the service to perform some action or report on an action taken by a peer entity. If the protocol stack is located in the operating system, as it often is, the primitives are normally system calls. These calls cause a trap to kernel mode, which then turns control of the machine over to the operating system to send the necessary packets.

The set of primitives available depends on the nature of the service being provided. The primitives for connection-oriented service are different from those of connection less service. As a minimal example of the service primitives that might be provided to implement a reliable byte stream in a client-server environment, consider the primitives listed in Fig. 1-12.

Primitive	Meaning
LISTEN	Block waiting for an incoming connection
CONNECT	Establish a connection with a waiting peer
RECEIVE	Block waiting for an incoming message
SEND	Send a message to the peer
DISCONNECT	Terminate a connection

Figure 1-12. Five service primitives for implementing a simple connection oriented service.

These primitives might be used as follows. First, the server executes LISTEN to indicate that it is prepared to accept incoming connections. A common way to implement LISTEN is to make it a blocking system call. After executing the primitive. the server process is blocked until a request for connection appears.

Next, the client process executes CONNECT to establish a connection with the server. The CONNECT call needs to specify who to connect to, so it might have a parameter giving the server's address. The operating system then typically sends a packet to the peer asking it to connect, as shown by (1) in Fig. 1-13. The client process is suspended until there is a response. When the packet arrives at the server, it is processed by the operating system there. When the system sees that the packet is requesting a connection, it checks to see if there is a listener. If so, it does two things: unblocks the listener and sends back an acknowledgement (2). The arrival of this acknowledgement then releases the client. At this point the client and server are both running and they have a connection established. It is important to note that the acknowledgement (2) is generated by the protocol code itself, not in response to a user-level primitive. If a connection request arrives and there is no listener, the result is undefined. In some systems the packet may be queued for a short time in anticipation of a LISTEN.

The obvious analogy between this protocol and real life is a customer (client) calling a company's customer service manager, The service manager starts out by being near the telephone in case it rings. Then the client places the call. When the manager picks up the phone, the connection is established.

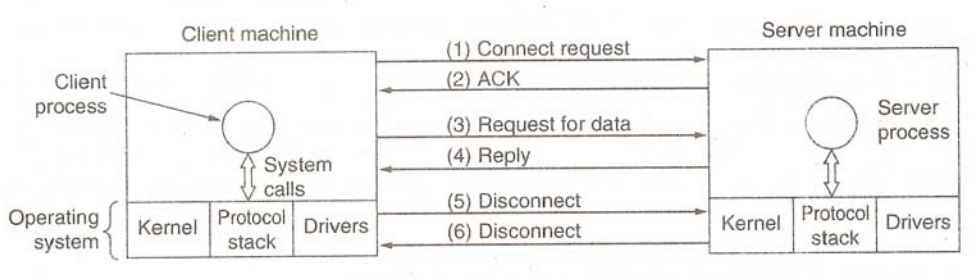


Figure 1-14. Packets sent in a simple client-server interaction on a connection oriented network.

The next step is for the server to execute RECEIVE, to prepare to accept the first request. Normally, the server does this immediately upon being released from the LISTEN, before the acknowledgement can get back to the client. The RECEIVE call blocks the server.

Then the client executes SEND to transmit its request (3) followed by the execution of RECEIVE to get the reply.

The arrival of the request packet at the server machine unblocks the server process so it can process the request. After it has done the work, it uses SEND to return the answer to the client (4). The arrival of this packet unblocks the client, which can now inspect the answer. If the client has additional requests, it can make them now. If it is done, it can use DISCONNECT to terminate the connection. Usually, an initial DISCONNECT is a blocking call, suspending the client and sending a packet to the server saying that the connection is no longer needed (5). When the server gets the packet, it also issues a DISCONNECT of its own, acknowledging the client' and releasing the connection. When the server's packet (6) gets back to the client machine, the

client process is released and the connection is broken. In a nutshell, this is how connection-oriented communication works.

Of course, life is not so simple. Many things can go wrong here. The timing can be wrong (e.g., the CONNECT is done before the LISTEN), packets can get lost, and much more. We will look at these issues in great detail later, but for the moment, Fig. 1-14 briefly summarizes how client-server communication might work over a connection-oriented network. .

Given that six packets are required to complete this protocol, one might wonder why a connection less protocol is not used instead. The answer is that in a perfect world it could be, in which case only two packets would be needed: one for the request and one. for the reply. However, in the face of large messages in either direction (e.g., a megabyte file), transmission errors, and lost packets, the situation changes. If the reply consisted of hundreds of packets, some of which could be lost during transmission, how would the client know if some pieces were missing? How would the client know whether the last packet actually received was really the last packet sent? Suppose that the client wanted a second file. How could it tell packet 1 from the second file from a lost packet I from the first file that suddenly found its way to the client? In short, in the real world, a simple request-reply protocol over an unreliable network is often inadequate.

The Relationship of Services to Protocols

Services and protocols are distinct concepts, although they are frequently confused. This distinction is so important, however, that we emphasize it again here. A service is a set of primitives (operations) that a layer provides to the layer above it. The service defines what operations the layer is

prepared to perform on behalf of its users, but it says nothing at all about how these operations are implemented. A service relates to an interface between two layers, with the lower layer being the service provider and the upper layer being the service user.

A protocol in contrast, is a set of rules governing the format and meaning of the packets, or messages that are exchanged by the peer entities within a layer. Entities use protocols. to implement their service definitions. They are free to change their protocols at will, provided they do not change the service visible to their users. In this way, the service and the protocol are completely decoupled.

In other words, services relate to the interfaces between layers, as illustrated in Fig. 1-15. In contrast, protocols relate to the packets sent between peer entities on different machines. It is important not to confuse the two concepts.

An analogy with programming languages is worth making. A service is like an abstract data type or an object in an object-oriented language. It defines operations that can be performed on an object but does not specify how these operations are implemented. A protocol relates to the implementation of the service and as such is not visible to the user of the service.

Many older protocols did not distinguish the service from the protocol. In

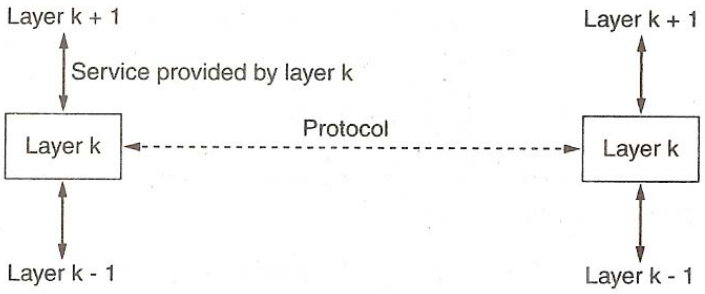


Figure 1-15. The relationship between a service and a protocol.

user providing a pointer to a fully assembled packet. This arrangement meant that. all changes to the protocol were immediately visible to the users. Most network designers now regard such a design as a serious blunder.

REFERENCE MODELS

Now that we have discussed layered networks in the abstract, it is time to look at some examples. In the next two sections we will discuss two important network architectures, the OSI reference model and the TCP/IP reference model. Although the protocols associated with the OSI model are rarely used any more, the model itself is actually quite general and still valid, and the features discussed at each layer are still very important. The TCP/IP model has the opposite properties: the model itself is not of much use but the protocols are widely used. For this reason we will look at both of them in detail. Also, sometimes you can learn more from failures than from successes.

1.6 The OSI Reference Model

The OSI model (minus the physical medium) is shown in Fig. 1-16. This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). It was revised in 1995. The model is called the ISO OSI (Open Systems Interconnection) Reference Model because it deals with connecting open systems-that is, systems that are open for communication with other systems. We will just call it the OSI model for short.

The OSI model has seven layers. The principles that were applied to arrive at the seven layers can be briefly summarized as follows:

1. A layer should be created where a different abstraction is needed.
2. Each layer should perform a well-defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that the architecture does not become unwieldy.

Below we will discuss each layer of the model in turn, starting at the bottom layer. Note that the OSI model itself is not a network architecture because it does not specify the exact services and protocols to be used in each layer. It just tells what each layer should do. However, ISO has also produced" standards for all the layers, although these are not part of the reference model

itself. Each one has been published as a separate international standard.

The Physical Layer

The physical layer is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit. Typical questions here are how many volts should be used to represent a 1 and how many for a 0, how many nanoseconds a bit lasts, whether transmission may proceed simultaneously in both directions, how the initial connection is established and how it is tom down when both sides are finished, and how many pins the network connector has and what each pin is used for. The design issues here largely deal with mechanical, electrical, and timing interfaces, and the physical transmission medium, which lies below the physical layer.

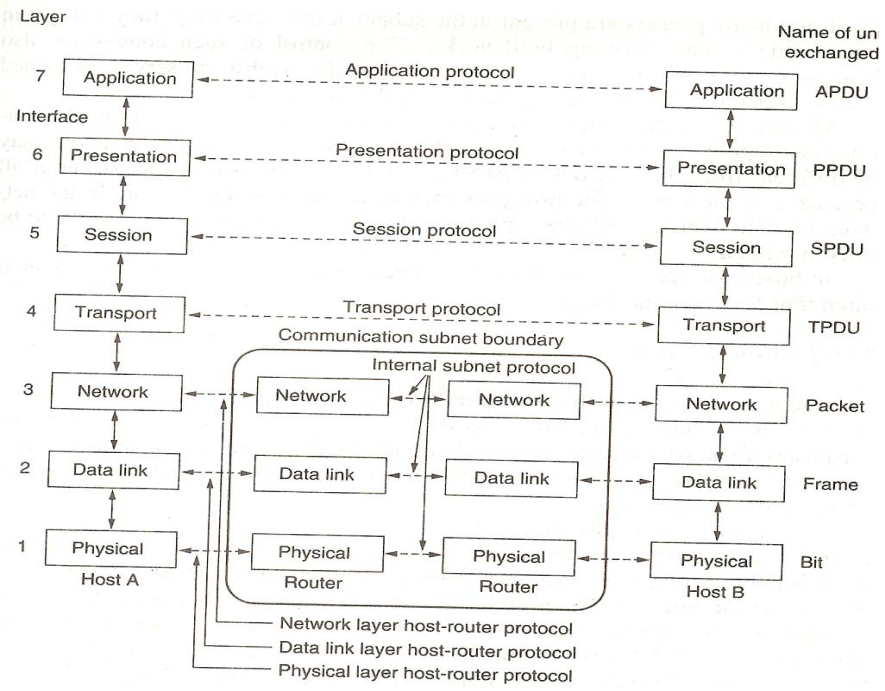


Fig 1.16 The OSI reference Model

To summarize, the physical layer has to take into account the following factors:

- **Signal encoding** : How are the bits 0 and 1 to be represented ?
- **Medium:** What is the medium used, and what are its properties ?
- **Bit synchronization:** Is the transmission asynchronous or synchronous?
- **Transmission type**: Is the transmission serial or parallel ?
- **Transmission mode** : Is the transmission simplex, half-duplex or full duplex ?
- **Topology:** What is the topology (mesh, star ring, bus or hybrid) used?
- **Multiplexing:** Is multiplexing used, and if so, what is its type (FDM, TDM)?
- **Interface:** How are the two closely linked devices connected ?
- **Bandwidth:** Which of the two baseband or broadband communication is being used?
- **Signal type:** Are analog signals or digital ?

The Data Link Layer

The main task of the data link layer is to transform a raw transmission facility into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break up the input data into data frames (typically a few hundred or a few thousand bytes) and transmit the frames sequentially. If the service is reliable, the receiver confirms correct receipt of each frame by sending back an acknowledgement frame.

Another issue that arises in the data link layer (and most of the higher layers as well) is how to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanism is often needed to let the transmitter know that

how much buffer space the receiver has at .the moment. Frequently, this flow regulation and the error handling are integrated.

Broadcast networks have an additional issue in the data link layer: how to control access to the shared channel. A special sublayer of the data link layer, the medium access control sublayer, deals with this problem.

- **Addressing:** Headers and trailers are added, containing the physical addresses of the adjacent nodes, and removed upon a successful delivery.
- **Flow Control:** This avoids overwriting on the receiver's buffer by regulating the amount of data that can be sent.
- **Media Access Control (MAC):** In LANs it decides who can send data, when and how much.
- **Synchronization:** Headers have bits, which tell the receiver when a frame is arriving. It also contains bits to synchronize its timing to know the bit interval to recognize the bit correctly. Trailers mark the end of a frame, apart from containing the error control bits.
- **Error control:** It checks the CRC to ensure the correctness of the frame. If incorrect, it asks for retransmission. Again, here there are multiple schemes (positive acknowledgement, negative acknowledgement, go-back-n, sliding window, etc.).

- **Node-to-node delivery:** Finally, it is responsible for error free delivery of the entire frame/ packet to the next adjacent node (node to node delivery).

The Network Layer

The network layer controls the operation of the subnet. A key design issue is determining how packets are routed from source to destination. Routes can be based on static tables that are "wired into" the network and rarely changed. They can also be determined at the start of each conversation, for example, a terminal session (e.g., a login to a remote machine). Finally, they can be highly dynamic, being determined a new for each packet, to reflect the current network load.

If too many packets are present in the subnet at the same time, they will get in one another's way, forming bottlenecks. The control of such congestion also belongs to the network layer. More generally, the quality of service provided (delay, transit time, jitter, etc.) is also a network layer issue.

When a packet has to travel from one network to another to get to its destination, many problems can arise. The addressing used by the second network may be different from the first one. The second one may not accept the packet at all because it is too large. The protocols may differ, and so on. It is up to the network layer to overcome all these problems to allow heterogeneous networks to be interconnected.

In broadcast networks, the routing problem is simple, so the network layer is often thin or even nonexistent.

To summarize, the network layer performs the following functions :

- Routing As discussed earlier.
- Congestion Control As discussed before.
- Logical addressing Source and destination logical addresses (e.g. IP addresses)
- Address transformations Interpreting logical addresses to get their physical equivalent (e.g. ARP protocol.)
- Source to destination error – free delivery of a packet.

The Transport Layer

The basic 'function' of the transport layer is to accept data from above, split it up into smaller units if need be, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done, efficiently and in a way that isolates the upper layers from the inevitable changes in the hardware technology.

The transport layer also determines what type of service to provide to the session layer, and, ultimately, to the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are the transporting of isolated messages, with no guarantee about the order of delivery, and the broadcasting of messages to multiple destinations. The type' of service is determined when the connection is established. (As an aside, an error-free channel is impossible to achieve; what people really mean by this term is that the error rate is low enough to ignore in practice.

The transport layer is a true end-to-end layer, all the way from the source to the destination. In other words, a program on the source machine

carries on a conversation with a similar program on the destination machine, using the message headers and control messages. In the lower layers, the protocols are between each machine and its immediate neighbors, and not between the ultimate source and destination machines, which may be separated by many routers. The difference between layers 1 through 3, which are chained, and layers 4 through 7, which are end-to-end, is illustrated in Fig. 1-16.

To summarize, the responsibilities of the transport layer are as follows :

- **Host-to-host message delivery:** Ensuring that all the packets of a message sent by a source node arrive at the intended destination.
- **Application to application communication:** The transport layer enables communication between two applications running on different computers.
- **Segmentation and reassembly:** The transport layer breaks a message into packets, numbers them by adding sequence numbers at the destination to reassemble the original message.
- **Connection:** The transport layer might create a logical connection between the source and the destination for the duration of the complete message transfer for better control over the message transfer.

The Session Layer

The session layer allows users on different machines to establish sessions between them. Sessions offer various services, including dialog control (keeping track of whose turn it is to transmit), token management

(preventing two parties from attempting the same critical. operation at the same time), and synchronization (check pointing long transmissions to allow them to continue from where they were after a crash).

- **Session and sub sub-sessions:** The layer divides a session into sub-sessions for avoiding retransmission of entire messages by adding check pointing feature.
- **Synchronization :** The session layer decides the order in which data need to be passed to the transport layer.
- **Dialog control:** The session layer also decides which user/ application send data, and at what point of time, and whether the communication is simplex, half duplex or full duplex.
- **Session closure:** The session layer ensure that the session between the hosts is closed grace fully.

The Presentation Layer

Unlike lower layers, which are mostly concerned with moving bits around, the presentation layer is concerned with the syntax and semantics of the information transmitted. In order to make it possible for computers with different data representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and allows higher-level data structures (e.g., banking records); to be defined and exchanged.

To summarize, the responsibilities of the presentation layer are as follows :

- **Translation:** The translation between the sender's and the receiver's massage formats is done by the presentation layer if the two formats are different.
- **Encryption:** The presentation layer performs data encryption and decryption for security.
- **Compression:** For the efficient transmission, the presentation layer performs data compression before sending and decompression at the destination.

The Application Layer

The application layer contains a variety of protocols that are commonly needed by users. One widely used application protocol is HTTP (Hyper Text Transfer Protocol), which is the basis for the World Wide Web. When a browser wants a Web page, it sends the name of the page it wants to the server using HTTP. The server then sends the page back other application protocols are used for file transfer, electronic mail, and network news.

To summarize the responsibilities of the application layer are as follows :

- **Network abstraction:** The application layer provides an abstraction of the underlaying network to an end user and an application.
- **File access and transfer:** It allows a use to access, download or upload files from/to a remote host.
- **Mail services:** It allows the users to use the mail services.
- **Remote login:** It allows logging into a host which is remote
- **World Wide Web (WWW):** Accessing the Web pages is also a part of this layer.

1.7 The TCP/IP Reference Model

Let us now turn from the OSI reference model to the reference model used in the grandparent of all wide area computer networks, the ARPANET, and its successor, the worldwide Internet. Although we will give a brief history of the ARPANET later, it is useful to mention a few key aspects of it now. The ARPANET was a research network sponsored by the DoD (U.S. Department of Defense). It eventually connected hundreds of universities and government installations, using leased telephone lines. When satellite and radio networks were added later, the existing protocols had trouble inter-working with them, so a new reference architecture was needed. Thus, the ability to connect multiple networks in a seamless way was one of the major design goals from the very beginning. This architecture later became known as the TCP/IP Reference Model, after its two primary protocols. It was first defined in (Cerf and Kahn, 1974). A later perspective is given in (Leiner et aI., 1985). The design philosophy behind the model is discussed in (Clark, 1988).

Given the DoD's worry that some of its precious hosts, routers, and internetwork gateways might get blown to pieces at a moment's notice, another major goal was that the network be able to survive loss of subnet hardware, with existing conversations not being broken off. . In other words, DoD wanted connections to remain intact as long as the source and destination machines were functioning, even if some of the machines or transmission lines in between were suddenly put out of operation.. Furthermore, a flexible architecture was needed since applications with divergent requirements were envisioned, ranging from transferring files to real-time speech transmission.

The Internet Layer

All these requirements led to the choice of a packet-switching network based on a connectionless internetwork layer. This layer, called the internet layer, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have them travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The analogy here is with the (snail) mail system. A person can drop a sequence of international letters into a mail box in one country, and with a little luck, most of them will be delivered to the correct address in the destination country. Probably the letters will travel through one or more international mail gateways along the way, but this is transparent to the users. Furthermore, that each country (i.e., each network) has its own stamps, preferred envelope sizes, and delivery rules is hidden from the users.

The internet layer defines an official packet format and protocol called IP **(Internet Protocol).** The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to, say that the TCP/IP internet layer is similar in functionality to the OSI network layer.

The Transport Layer

The layer above the internet layer in the TCP/IP model is 'now usually called the transport layer. It is designed to allow peer entities on the source and

destination hosts to carry on a conversation, just as in the OSI transport layer. Two end-to-end transport protocols have been defined here. The first one, **TCP (Transmission Control Protocol),** is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one on to the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

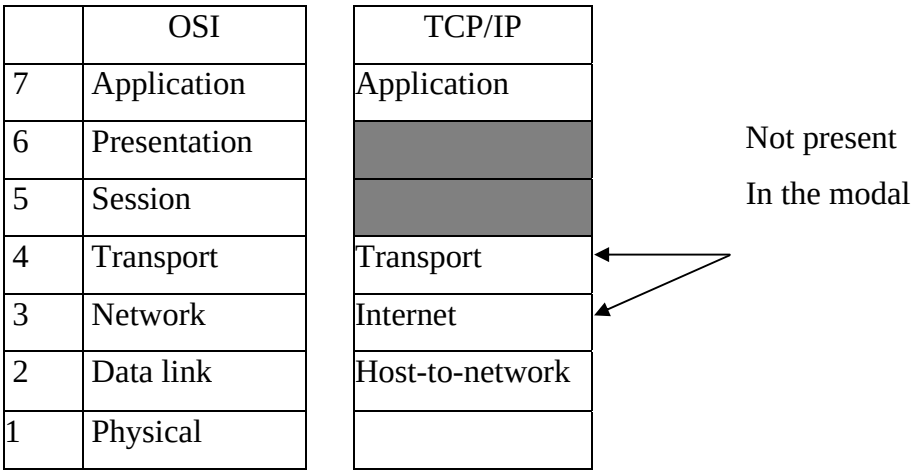


Figure 1-17. The TCP/IP reference model.

The second protocol in this layer, **UDP (User Datagram Protocol),** is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one shot, client-server-type request-reply queries and applications in which prompt delivery is more important -than accurate delivery, such as transmitting

speech or video. The relation of IP, TCP, and UDP is shown in Fig. 1-18. Since the model was developed, IP has been implemented on many other networks.

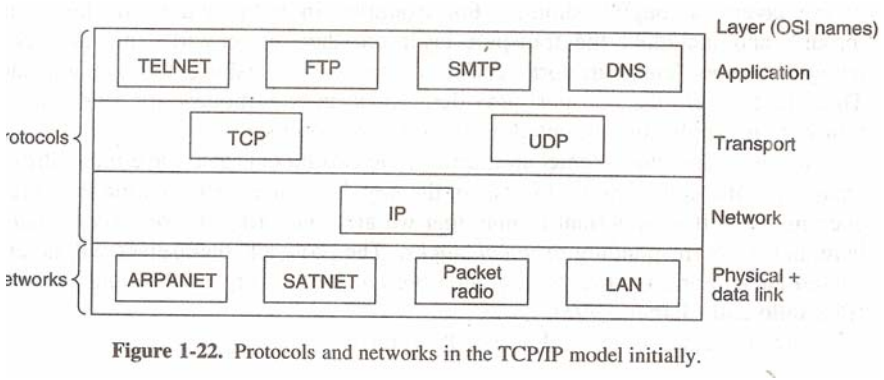


Figure 1-22. Protocols and networks in the TCP/IP model initially.

Figure 1-18. Protocols and networks in the TCP/IP model initially**.**

The Application Layer

The TCP/IP model does not have session or presentation layers. No need for them was perceived, so they were not included. Experience with the OSI model has proven this view correct: they are of little use to most applications.

On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig. 1-18. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it. Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the protocol for moving USENET news articles around, and HTTP, the

protocol for fetching pages on the World Wide Web, and many others.

Host-to-Network Layer

Below the Internet layer is a great void. The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network .to network. Books and papers about the TCP/IP model rarely discuss it.

1.8 A Comparison of the OSI and TCPI / IP Reference Models

The OSI and TCP/IP reference models have much in common. Both are based on the concept of a stack of independent protocols. Also, the functionality of the layers is roughly similar. For example, in both models the layers up through and including the transport layer are there to provide an end-to-end, network-independent transport service to processes wishing to communicate. These layers form the transport provider. Again in both models, the layers above transport are application-oriented users of the transport service.

Despite these fundamental similarities, the two models also have many differences. In this section we will focus on the key differences between the two reference models. It is important to note that we are comparing the reference models here, not the corresponding protocol stacks. The protocols themselves will be discussed later.Three concepts are central to the OSI model:

1. Services.
2. Interfaces.
3. Protocols.

Probably the biggest contribution of the OSI model is to make the distinction between these three concepts explicit. Each layer performs some services for the layer above it. The service definition tells what the layer does, not how entities above it access it or how the layer works. It defines the layer's semantics.

A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in. higher layers.

These ideas fit very nicely with modern ideas about object-oriented programming. An object, like a layer, has a set of methods (operations) that processes out side the object can invoke. The semantics of these methods define the set of services that the object offers. The methods parameters and results form the object's interface. The code internal to the object is its protocol and is not visible or of any concern outside the object.

The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like. For example, the only real services offered by the internet layer are As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place.

The OSI reference model was devised before the corresponding protocols were invented. This ordering means that the model was not biased toward one particular set of protocols, a fact that made it quite general. The downside of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.

For example, the data link layer originally dealt only with point-to-point networks. When broadcast networks came around, a new sublayer had to be hacked into the model. When people started to build real networks using the OSI model and existing protocols, it was discovered that these networks did not match the required service specifications (wonder of wonders), so convergence sublayers had to be grafted onto the model to provide a place for papering over the differences. Finally, the committee originally expected that each country would have one network, run. by the government and using the OSI protocols, so no thought was given to internetworking. To make a long story short, things did not turn out that way.

With TCP/IP the reverse was true: the protocols came first, and the model was really just a description of the existing protocols. There was no problem with the protocols fitting the model. They fit perfectly. The only trouble was that the model did not fit any other protocol stacks. Consequently, it was not especially useful for describing other, non- TCP/IP networks.

Turning from philosophical matters to more specific ones, an obvious difference between the two models is the number of layers: the OSI model has seven layers and the TCP/IP has four layers. Both have (inter) network, transport, and application layers, but the other layers are different.

Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because-the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connection less) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for simple request-response protocols)

1.9 A Critique of the OSI Model and Protocols

Neither the OSI model and its protocols nor the TCP/IP model and its protocols are perfect. Quite a bit of criticism can be, and has been, directed at both of them. In this section and the next one, we will look at some of these criticisms. We will begin with OSI and examine TCP/IP afterward.

At the time when the concepts were revised by the other experts, it appeared to many experts in the field that the OSI model and its protocols were going to take over the world and push everything else out of their way. This did not happen. Why? A look back at some of the concepts may be useful. These concepts can be summarized as:

1. Bad timing.
2. Bad technology.
3. Bad implementations.
4. Bad politics.

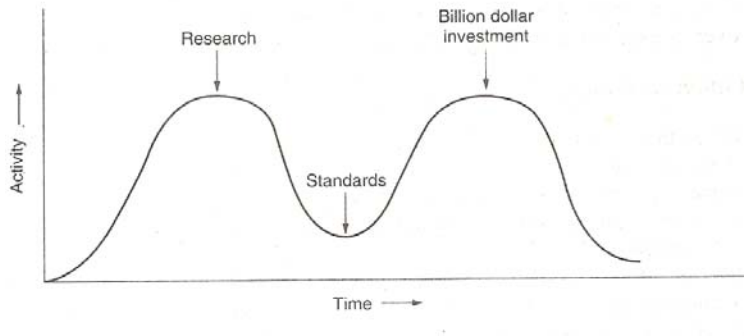


Fig. 1.19 The apocalypse of the two elephants.

Bad Timing

First let us look at reason one: bad timing. The time at which a standard is established is absolutely critical to its success. David Clark of M.I.T. has a theory of standards that he calls the apocalypse of the two elephants, which is illustrated in Fig. 1-19.

This figure shows the amount of activity surrounding a new subject. When the subject is first discovered, there is a burst of research activity in the form of discussions, papers, and meetings. After a while this activity subsides, corporations discover the subject, and the billion-dollar wave of investment hits.

It is essential that the standards be written in the trough in between the two "elephants." If the standards are written too early, before the research is finished, the subject may still be poorly understood; the result is bad standards. If they are written too late, so many companies may have already made major investments in different ways of doing things that the standards are effectively ignored. If the interval between the two elephants is very short (because everyone is in a hurry to get started), the people developing the standards may

get crushed.

It now appears that the standard OSI protocols got crushed. The competing TCP/IP protocols were already in widespread use by research universities by the time the OSI protocols appeared. While the billion-dollar wave of investment had not yet hit, the academic market was large enough that many vendors had begun cautiously offering TCP/IP products. When OSI came around, they did not want to support a second protocol stack until they were forced to, so there were no initial offerings. With every company waiting for every. other company to go first, no company went first and OSI never happened.

Bad Technology

The second reason that OSI never caught on is that both the model and the protocols are flawed. The choice of seven layers was more political than technical, and two of the layers (session and presentation) are nearly empty, whereas two other ones (data link and network) are overfull.

The OSI model, along with the associated service definitions and protocols, is extraordinarily complex. When piled up, the printed standards occupy a significant fraction of a meter of paper. They are also difficult to implement and inefficient in operation.

In addition to being incomprehensible, another problem with OSI is that some functions, such .as addressing, flow control, and error control, reappear again and again in each layer. Saltzer et al. (1984), for example, have pointed out that to be effective, error control must be done in the highest layer, so that repeating it over and over in each of the lower layers is often unnecessary and inefficient.

Bad Implementations

Given the enormous complexity of the model and the protocols, it will come as no surprise that the initial implementations were huge, unwieldy, and slow. Everyone who tried them got burned. It did not take long for people to associate "OSI" with "poor quality." Although the products improved in the course of time, the image stuck.

In contrast, one of the first implementations of TCP/IP was part of Berkeley UNIX and was quite good (not to mention, free). People began using it quickly, which led to a large user community, which led to improvements, which led to an even larger community. Here the spiral was upward instead of downward.

Bad Politics

On account of the initial implementation, many people, especially in academia, thought of TCP/IP as part of UNIX, and UNIX in the 1980s in academia was not unlike parenthood (then incorrectly called motherhood) and apple pie.

OSI, on the other hand, was widely thought to be the creature of the European telecommunication ministries, the European Community, and later the U.S.

Government. This belief was only partly true, but the very idea of a bunch of government bureaucrats trying to shove a technically inferior standard down the throats of the poor researchers and programmers down in the trenches actually developing computer networks did not help much. Some people viewed this development in the same light as IBM announcing in the 1960s that

PL/I was the language of the future, or DoD correcting this layer by announcing that it was actually Ada.

1.4.5 A Critique of the TCP/IP Reference Model

The TCPI/IP model and protocols have, their problems too. First, the model does not clearly distinguish the concepts of service, interface, and protocol. Good software engineering practice requires differentiating between the specification and the implementation, something that OSI does very carefully, and TCPI/IP does not. Consequently, the TCPI/IP model is not much of a guide for designing new networks using new technologies.

Second, the TCPI/IP model is not at all general and is poorly suited to describing any protocol stack other than TCPI/IP. Trying to use the TCPI/IP model to describe Bluetooth, for example, is completely impossible.

Third, the, host-to-network layer is not really a layer at all in the normal sense of the term as used in the context of layered protocols. It is an interface (between the network and data link layers). The distinction between an interface and a layer is crucial, and one should not be sloppy about it.

Fourth, the TCP/IP model does not distinguish (or even mention) the physical and data link layers. These are completely different. The physical layer has to do with the transmission characteristics of copper wire, fiber optics, and wireless communication. The data link layer's job is to delimit the start and end of frames and get them from one side to the other with the desired degree of reliability. A proper model should include both as separate layers. The TCP/IP model does not do this.

Finally, although the IP and TCP protocols were carefully thought out

and well implemented, many of the other protocols were ad hoc, generally produced by a couple of graduate students hacking away until they got tired. The protocol implementations were then distributed free, which resulted in their becoming widely used, deeply entrenched, and thus hard to replace. Some of them are a bit of an embarrassment now: The virtual terminal protocol, TELNET, for example, was designed for a ten-character per second mechanical Teletype terminal. It knows nothing of graphical user interfaces and mice. Nevertheless, 25 years later, it is still in widespread use.

In summary, despite its problems, the OSI model (minus the session and presentation layers) has proven to be exceptionally useful for discussing computer networks. In contrast, the OSI protocols have not become popular. The reverse is true of TCP/IP: the model is practically nonexistent, but the protocols are widely used. Since computer scientists like to have their cake and eat it, too, in this book we will use a modified OSI model but concentrate primarily on the TCP/IP and related protocols, as well as newer ones such as 802, SONET, and Bluetooth. In effect, we will use the hybrid model of Fig. 1-20.

5	Application layer
4	Transport layer
3	Network layer
2	Data link layer
1	Physical layer

Figure 1-20. The hybrid reference model

SUMMARY

Computer networks can be used for numerous services, both for companies and for individuals. For companies, networks of personal computers using shared servers often provide access to corporate information. Typically they follow the client-server model, with client workstations on employee desktops accessing powerful servers in the machine room. For individuals, networks offer access to a variety of information and entertainment resources. Individuals often access the Internet by calling up an ISP using a modem, although increasingly many people have a fixed connection at home. An up-and-coming area is wireless networking with new applications such as mobile e-mail access and m-commerce.

Roughly speaking, networks can be divided up into LANs, MANs, WANs, and internetworks, with their own characteristics, technologies; speeds, and niches. LANs cover a building and operate at high speeds. MANs cover a city, for example, the cable television system, which is now used by many people to access the Internet. WANs cover a country or continent. LANs and MANs are unswitched (i.e., do not have routers); WANs are switched. Wireless networks are becoming extremely popular, especially wireless LANs. Networks can be interconnected to form internetworks.

Network software consists of protocols, which are rules by which processes communicate. Protocols are either connectionless or connection-oriented. Most networks support protocol hierarchies, with each layer providing services to the layers above it and insulating them from the details of the protocols used in the lower layer. *A* model known as Open Systems Interconnect (OSI) was devised by the International Standards organization

(ISO). This model would allow the sending and receiving of data between two computers.

It works on a layer approach, where each layer is responsible for performing certain jobs and functions. The need of layered architecture arises because when data is sent from one computer to another, there are many different things involved. These are network adapters, voltages and signals on the cable, how the data is packaged, error control in case something goes wrong, and many other issues. By segmenting these issues into separate layers, it makes the task of designing network much easier. The OSI model, which allows dissimilar computers to transfer data between themselves, is divided into seven distinct layers.

These layers are application, presentation, session, transport, network, data link and physical layer with their distinct functions. These functions are to provide applications with access to network, determine the format used to exchange data among computer over the network, allow connection disconnection to be established between applications, allow error free delivery of data and send data at correct destination.

TCP/IP is the protocol used by computers on the internet and may be considered as two separate protocols such as TCP and IP.

Each computer has an IP address. A protocol is a set of rules that govern how computers talk to each other. *TCP/IP* is a widely used and very popular protocol. With *TCP/IP*, different computer systems can reliably exchange data on an interconnected network. It also provides a consistent set of application programming interfaces (APIs) to support application development. This means that software programs can use TCP/IP to exchange data. An example of

this is web servers and web browsers, software applications that use TCP/IP to exchange data.

Protocol stacks are typically based either on the OSI model or on the TCP/IP model. Both have network, transport, and application layers, but they differ on the other layers. Design issues include multiplexing, flow control, error control, and others.

Networks provide services to their users. These services can be connection oriented or connectionless. In some networks, connectionless service is provided in one layer and connection-oriented service is provided in the layer above it. Well-known networks include the Internet, ATM networks, Ethernet, and the IEEE 802.11 wireless LAN. The Internet evolved from the ARPANET, to which other networks were added to form an internetwork. The present Internet is actual ally a collection of many thousands of networks, rather than a single network. What characterizes it is the use of the TCP/IP protocol stack throughout. ATM is widely used inside the telephone system for long-haul data traffic. Ethernet is the most popular LAN and is present in most large companies and universities. Finally, wireless LANs at surprisingly high speeds (up to 54 Mbps) are beginning to be widely deployed.

1.11 Self Assessment Questions

1. Define the following
   (a) Workstation
   (b) Server
   (c) Network interface Unit
2. What is network topology ? Explain the following with suitable example and their relative advantages and disadvantages.

(a) Star topology

(b) Bus topology

(c) Ring topology

(d) Tree topology

(e) Hybrid topology

3. (a) Differentiate between LAN, MAN and WAN.

   (b) Differentiate between connection oriented and connectionless services.

4. Describe the OSI reference model and explain the function of each layer.
5. Critically compare OSI reference model and TCP/IP model.
6. Explain the functions of different layers of TCP/IP reference model and various protocols working in these layers.

Suggested Readings

1. Tannenbaum,"Computer Network", Pearson Education
2. Godbole,"Data Communications and Networking", Tata Mc Graw-Hill
3. Forouzan," Communications and Networking", Tata Mc Graw-Hill

Subject :		**Computer Network**
Paper Code :	**301**	**Author : Dharmender Kumar**
Lesson No. :	**02**	**Vetter : Dr. Yogesh Chaba**
Chapter Title :		**Physical Layer**

Topics

2.0 Objective

2.1 Theoretical Coverage of Basis of Data Communication

2.1.1 Introduction to Analogue and Digital Signal

2.1.2 Amplitude, Period, Frequency and phase

2.1.3 Fourier Analysis & Concept of Bandwidth of a Signal

2.1.4 Maximum Data rate of a channel

2.2. Transmission Media

2.2.1 Guided Media

Twisted Pair Cable

Coaxial Cable

Optical Fibre

2.2.2 Unguided Media

Radio Communication

Microwave Communication

Satellite Communication

Cellular (Mobile) Communication

2.3. Types of Transmission

2.0 Objective : To Learn about the various concepts related to data communication, Transmission Media available and various types of data transmission.

2.1 Theoretical Coverage of Basis of Data Communication

At the core of all computer to computer data communications is the concept of signal propagation. The simple idea in signal propagation is that when a signal is applied in any form (such as heat, voltage or current) to one end of a conducting material, after some time the signal reaches the other end of the material. Thus, a wire can also be used for carrying data between the two points that it connects.

2.1.1 Introduction to Analog and Digital Signals

Any Signal can be classified into one of the two types : analog and digital. An analog signal is a continuously varying signal, similar to a sinusoidal waveform. For instance, if we measure the room temperature

continuously and plot its graph with the time on the x-axis and the temperature on the y axis, we would get a continuous waveform, which is an analog signal. Analog is always continuous.

In contrast, a digital signal takes the form of pulses, where we have something or nothing, (i.e. 1 or o in digital terms). The absence or presence of something (i.e. 0 or 1) can be used to plot a digital signal. A typical digital signal is shown in Fig. (2.2)

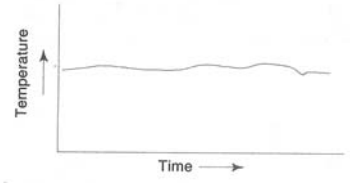


Fig. 2.1

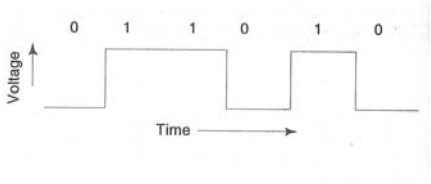


Fig. 2.2

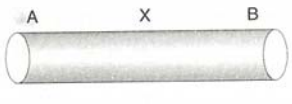


Fig 2.3 Digital Signal

Thus, we can say that an analog signal is smooth and continuous in nature. In contrast, a digital signal represents a sudden jump from one value to another. A digital signal for bits 0110 10 is shown in Fig. 2.2 where 0 means no voltage and 1 means some +ve voltage.

Again, regardless of whether the signal is analog or digital, if we apply this signal at one end of the wire, the signal form will travel at the other end (with some distortions caused due to noise, which we shall study later). For instance let us imagine that we have a wire A-X-B (like the one shown in (Fig. 2.3). Let us imagine that we apply 0 voltage to point A for some time

(representing 0), then 5 volts for a longer duration (representing 11), then again 0 voltage for some time (representing 0) and so on. It will mean that we are applying a digital signal as shown, in Fig. 2.2, at point A. Due to the principle of signal propagation, the same signal (i.e. 011010 in this case) will traverse to point X and then to point B with some distortion and after a finite propagation delay.

2.1.2 Amplitude, Period, Frequency and Phase

At any point, the strength of the signal or amplitude is given by the Y-axis. Thus, the amplitude of a signal is equal to its vertical distance from the horizontal X-axis. Fig 2.4. shows that the signal has maximum value, called amplitude (A) shown as A at point K. Amplitude is measured in volts (representing voltage), amperes (representing current) or watts (representing power), depending on the type of the signal.

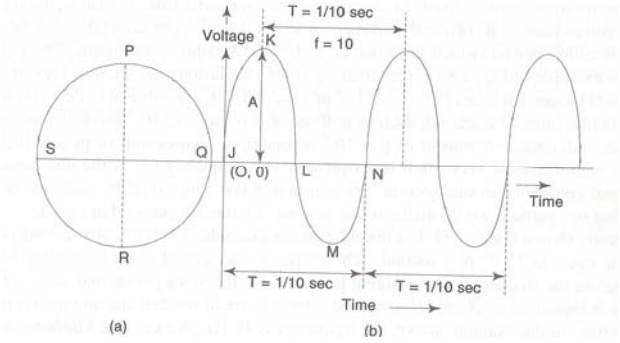


Fig. 2.4 Sinusoidal Signal

We see that the signal repeats itself; i.e. completes one cycle after time T. That is, the signal reaches the peak and comes back to its original starting position in time T. The time taken for the completion of one cycle is called period, which is shown as 1/10th of a second in the figure. This is because the particle in our example in Fig.2.4(a) was rotating at ten revolutions per second.

This means that the signal value will be same at times t and t + 1/10th of a second, for any value of t. Period is expressed in seconds, or smaller units of a second, such as milliseconds or ms (i.e. 10^{-3} seconds), microseconds or MS (i.e. 10^{-6} seconds), nanoseconds or ns (i.e. 10^{-9} seconds), or pico-seconds or ps (i.e. 10^{-12} seconds). Another term which is used very often is frequency. The frequency (f) is the number of cycles or

periods a signal completes in one second. We can notice that this would be same as the number of revolutions that our particle would make in one second. Figure 2.4 shows that f = 10.

It can be easily shown that f = 1/T. For instance, in our example, T = $1/10^{th}$ of a second, i.e. the signal completes one cycle in 1/l0th of a second. Therefore, in one second, it will complete 1/(1/10) = 10 cycles. Therefore the frequency of this signal is 10, i.e. f = 10 cycles per second and f= l/T.

Frequency is measured in Hertz (Hz), named after a German mathematician. One Hz is one cycle/ second Therefore, in the example above, the frequency is 10 Hz. We use one Kilohertz or one kHz to mean 1000 Hz, and one Megahertz or one MHz to mean 1000 kHz or 1000000 Hz..

2.1.3 Fourier analysis and the concept of bandwidth of a signal

Now we return to Fourier's theory. How can we decompose a random signal, such as generated by a telephone conversation into various sinusoidal signals of varying amplitudes. frequencies and phases? We do not want to give a mathematical proof of this here. We will just take an example to illustrate this concept.

Figure 2.5 (a), (b) and (c) show three sinusoidal signals with different

amplitudes and frequencies (e.g. 10Hz, 30Hz and 50Hz). In Fig.(a), the signal completes one cycle in $l/10^{th}$ of a second, i.e. its frequency = 10 Hz (10 cycles/sec). Similarly. it is easy to see that the signals in Fig. 2.5(b) and (c) have frequencies 30 and 50, respectively.

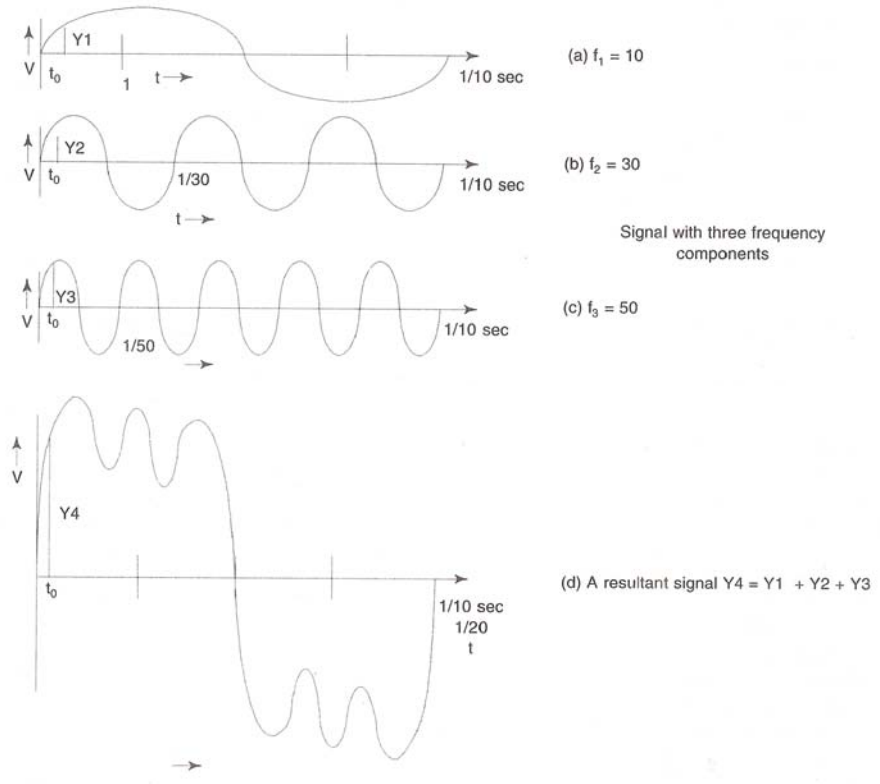


Fig. 2.5 Fourier Analysis Illustrated

For every value of time t on the X-axis, if we add the signal strength (amplitude) of these three signals and plot the result as the signal strength of the resultant signal on the Y-axis, we get the resultant analog signal as shown in Fig. 2.5(d). For instance, if we add Y I, Y2 and Y3, we will get Y 4, all at

time = 10. If we do that for all values of t, we will get the resultant signal shown in Fig. 2.5(d). Conversely, we can show that the analog signal shown in Fig. 2.5(d) is derived out of superimposing the three sinusoidal signals in Figs 2.5(a), (b) and (c). Similarly, it can be shown that any signal can be decomposed into different sinusoidal signals. This process is called Fourier analysis.

We can now understand that if we carry out a Fourier analysis on any complex signal, we will get a set of sinusoidal signals of different frequencies. Now, if we find out the difference between the highest and the lowest frequencies from within the sinusoidal signals making up that complex signal, we get a very useful concept. We call this difference bandwidth of the complex signal. For instance, the bandwidth of the signal in Fig. 2.5(d) is 50 Hz-10Hz = 40 Hz. The simplest way to describe bandwidth is the fastest continually oscillating signal that can be sent across a wire.

Example

A periodic signal has been decomposed using Fourier analysis to yield four sine waves of frequencies 100,400,600 and 800 Hz. What is the bandwidth of the resulting periodic signal?

Solution

We know that the bandwidth of a signal is the difference between the highest and the lowest frequencies. Therefore, the bandwidth of the periodic signal = 800-100 = 700 Hz.

Example

A signal has a bandwidth of 20 Hz and its highest frequency is 60 Hz. What is its lowest frequency?

Solution

Since the bandwidth of a signal is the difference between its highest and lowest frequencies, in this case, we have the following solution:

Lowest Frequency = Highest Frequency - Bandwidth = 60 - 20 = 40 Hz.

Similarly, we can find out the bandwidth of any signal of any shape and size. It is clear that if the signal changes very rapidly (as in a video signal), the higher frequency will be very high, i.e. when we decompose this signal into various sinusoidal signals, there will be some with very high frequencies.

Therefore, the bandwidth of such signals will be very high. If the signal changes very slowly, its bandwidth will be very small. At an extreme, a digital signal has both horizontal and vertical lines, and therefore, signals with 0 and 00 frequencies. Therefore, the bandwidth of a digital signal is 00 - 0 = 00. We will return to this theme shortly.

We must point out that similar to the bandwidth of a signal, there is a bandwidth of a medium. What does this mean? Suppose that we have a thin water pipe and a very forceful water flow. The thinness of the water pipe (i.e. the medium) in such a case represents the fact that its bandwidth is quite small as compared to the bandwidth of the water (i.e. the signal) that needs to be carried through it. Thus, both. a signal and the medium that is to carry the signal have their own bandwidths.

As we can imagine, even in the case of a medium, its bandwidth is the difference between the maximum and the minimum frequencies that it can carry. For many media (except the radio), the minimum frequency is 0. Therefore, generally, the bandwidth of a medium is also the same as the maximum frequency that it can carry. Why should there be a limit on this

highest frequency? The reason lies in the nature of the medium. Depending on the metal or the material used for the medium (i.e. the free electrons, etc), changes in the signals at one end are transmitted to the other end faithfully only up to a particular rate at which the signal changes (i.e. frequency). If this rate increases more than a certain limit, before the change in the signal is transmitted from points A to B, the signal at point A would change, and therefore, faithful transmission would not be possible. The signal would get distorted beyond repair. In simple terms, if the sender attempts to transmit signals that are faster than the bandwidth, the underlying hardware would not be able to keep up with the pace, because it would not have sufficient time to complete one change before the sender attempts to send another. This defines the maximum rate at which hardware can change a signal, i.e. the bandwidth of a medium. As we shall see, the bandwidth of optical fiber is far more than that of coaxial cable, which, in turn, is far higher than that of a twisted wire pair.

2.1.4 The Maximum Data Rate of a Channel

As early as 1924, an AT&T engineer, Henry Nyquist, realized that even a perfect channel has a finite transmission capacity. He derived an equation expressing the maximum data rate for a finite bandwidth noiseless channel. In 1948, Claude Shannon carried Nyquist's work further and extended "it to the case of a channel subject to random (that is, thermodynamic) noise (Shannon, 1948). We will just briefly summarize their now classical results here.

Nyquist proved that if an arbitrary signal. has been run through a low-pass filter of bandwidth H, the filtered signal can be completely reconstructed by making only 2H (exact) samples per second. Sampling the line faster than 2H times per second is pointless because the higher frequency components that

such sampling could recover have already been filtered out. If the signal consists of V discrete levels, Nyquist's theorem states: .

maximum data rate = 2H log2 V bits/ sec

For example, a noiseless 3-kHz channel cannot transmit binary (i.e., ,two-level) signals at a rate exceeding 6000 bps.

So far we have considered only noiseless channels. If random noise is present, the situation deteriorates rapidly. And there is always random (thermal) noise present due to the motion of the molecules in the system. The amount of thermal noise present is measured by the ratio of the signal power to. the noise power, called the signal-to-noise ratio. If we denote' the signal power by S and the noise power by N, the signal-to-noise ratio is S/N. Usually, the ratio itself is not quoted; instead, the quantity 10 log 10 S/N is given. These units are called decibels (dB). An S/N ratio of 10 is 10 dB, a ratio of 100 is 20 dB, a ratio of 1000 is 30 dB, and so on. The manufacturers of stereo amplifiers often characterize the bandwidth (frequency range) over which their product is linear by giving the 3-dB frequency

on each end. These are the points at which the amplification factor has been approximately halved (because $\log_{10} 3 \approx 0.5$).

Shannon's major result is that the maximum data rate of a noisy channel whose bandwidth is H Hz, and whose signal-to-noise ratio is SIN, is given by

maximum number of bits/ sec = H log2 (I + S/N)

For example, a channel of 3000-Hz bandwidth with a signal to thermal noise ratio of 30 dB (typical parameters of the analog part of the telephone system) can never transmit much more than 30,000 bps, no matter how many or how few signal levels are used and no matter how often or how infrequently

samples are taken. Shannon's result was derived from information-.theory arguments and applies to any channel subject to thermal noise. Counterexamples should be treated in the same category as perpetual motion machines. It should be noted that this is only an upper bound and real systems rarely achieve it.

2.2 Transmission media

Transmission media are the physical infrastructure components that carry data from one computer to another. They are at the basis of data communications. Examples of simple forms of transmission media are telephone wires that connect telephones to the central offices (i.e. telephone exchanges), and coaxial cables that carry the cable television transmission to homes. Transmission media need not always be in the form of a physical wire-they can be invisible as well We shall study these different types of transmission media in this chapter.

Broadly, all transmission media can be divided into two main categories: Guided and Unguided, as shown in Fig. 2.6.

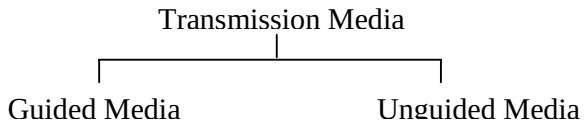


Fig2.6 Categories of Transmission Media

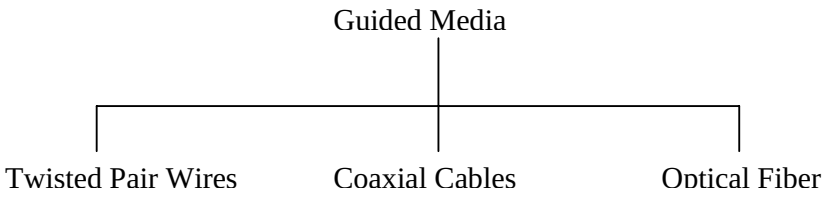


Fig 2.7 Types of Guided Media

2.2.1 GUIDED MEDIA

Guided Media can be further subdivided into three main types as shown in Fig.2.8 Guided media are typically based on some physical cable. Twisted pair and coaxial cables carry signals in the form of electrical current, whereas optical fiber carries signals in the form of light. We will now study these one by one.

1. Twisted Pair Cable

There are two classes of twisted pair cables as shown in fig. 2.8

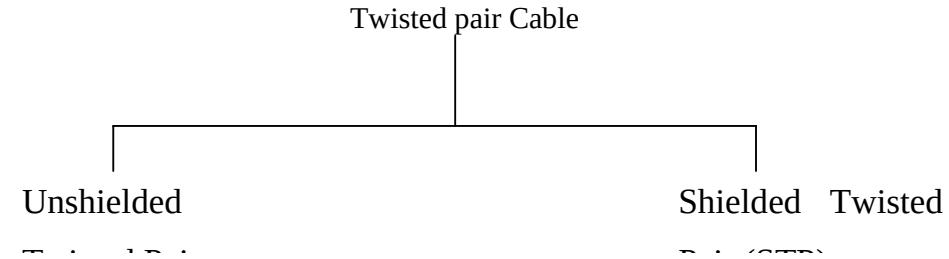


Fig 2.8 Categories of Twisted Pair Cable

a. Unshielded Twisted Pair (UTP)

This is the most commonly used medium today, because of its usage in the telephone system. This cable can carry both voice as well as data. It consists of two conductors (usually copper). In the beginning, the wires used to be kept parallel. However, this results in far greater levels of noise. Hence, the wires are normally twisted as shown Fig. 2.9. This results in the reduction of noise to a great extent, although it is not eliminated completely. The copper conductors are covered by Fig. 2.9 PVC. or other insulator.

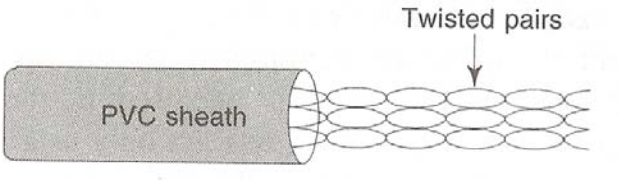


Fig. 2.9 Unshielded Twisted Pair (UTP)

UTP is flexible, cheap and easy to install. The Electronic Industries Association (EIA) has developed standards for UTP cables. They are given in Fig. 2.10. Each one is manufactured differently for a specific purpose.

Category	**Usage**
1.	The basic cable used in the telephone system. This is fine for voice communication, but is unsuitable for data communication, except at very low speed.
2.	Suitable for voice and data communication up to the speed of 4 Mbps.
3.	Can carry voice and data up to 10 Mbps. It requires minimum three twists per foot. Today, these are more regularly used in telephone networks.
4.	These are similar to the category 3, but can handle data up to 16 Mbps.
5.	Can handle data speed of 100 Mbps.

Fig.2.10 Categories of UTP

Today, UTPs of higher categories are also used in computer networks given the higher speeds and reliability.

b. Shielded Twisted Pair (STP)

In this case, apart from the insulator, the twisted wire pair itself is covered by metal shield and finally by the plastic cover as shown in Fig. 2.11. The metal shield prevents penetration of electromagnetic noise. It also helps eliminate crosstalk, an effect wherein one wire picks up some of the signals travelling on the other wire. This effect can be felt sometimes during telephone conversations, when we hear other conversations in the background during our call. The shield prevents such unwanted sounds.

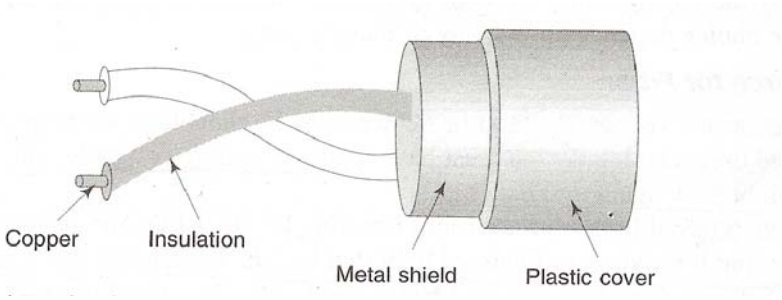


Fig 2.11. Shielded Twisted Pair (STP)

As we have seen, the shield reduces the noise and cross talk (noticed in the telephone calls sometimes1'Substantially. However, STP is more expensive than UTP.

2 Coaxial Cable

Coaxial cable (also called as coax) has an inner central conductor surrounded by an insulating sheath, which in turn is enclosed in the outer conductor (shield). This acts not only as a second conductor for completing the circuit but also acts as a shield against noise. This outer conductor is covered by a plastic cover. This whole arrangement is shown in Fig. 2.12

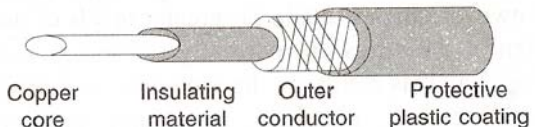


Fig. 2.12 Coaxial Cable

Compared to UTP or STP, coaxial cable is more expensive, less flexible and more difficult to install in a building where a number of twists and turns are required. However, it is much more reliable and can carry far higher data rates. Coaxial cables are divided into various categories depending upon the thickness and size of the shields, insulators and the outer coating and other considerations. They are commonly used by cable companies to carry cable transmissions.

The various coaxial cable standards are RG-8, RG-9, RG-11, RG-58 and RG-59.

3 Optical Fiber

Optical Fiber Structure

Optical fibers use light instead of electrical signals as a means of signal propagation. They are made of glass fibers that are enclosed in a plastic jacket. This allows the fibers to bend and not break. A transmitter at the sender's end of the optical fiber sends a light emitting diode (LED) or laser to send pulses of light across the fiber. A receiver at the other end makes use of a light-sensitive transistor to detect the absence or presence of light to indicate a 0 or a 1.

In the fiber, the cladding covers the core (fiber) depending upon the size. Commonly, the fiber is covered by a buffer layer, which protects it from moisture. Finally, an outer jacket surrounds the entire cable. This is shown in Fig. 2.13

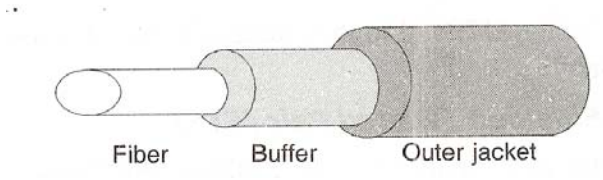


Fig 2.13. Optical Fiber

The core and cladding must be extremely pure, and should not vary in size or shape. The outer jacket, however, can be made of Teflon, plastic or metal. Teflon is more expensive than plastic, but both do not provide the structural strength to the fiber. Metal can provide that, but is very heavy and expensive. The choice depends upon many of these factors.

Light Source for Fiber

For data transmission. to occur, the sending device must be capable of inducing data bits 0 to I into the light source and the receiving device must have a photosensitive cell called photodiode, which would translate this light back again into data bits.

There are two types of light sources: Light Emitting Diode (LED) and Injection Laser Diode (ILD). LED is cheaper but it provides unfocussed light that hits the boundaries of the core at different angles and diffuses over distance. This is why LED is used only for a short distance. Laser, however, can provide a very focused beam, which retains its character over a long distance.

Light travels at a speed of 300,000 km/second or 186,000 miles/second in a straight direction as long as it moves through a uniform substance. However, when the medium through which the light passes changes, its speed

and therefore, the direction, also changes suddenly. A spoon half immersed in the glass of water appears bent due to the same reason. This phenomenon is called as refraction.

If we change the direction of the incoming light, the direction of the refracted light also changes as shown in Figs. 2.14(a), (b) and (c). Finally, we reach a stage where the refracted light becomes horizontal as shown in Fig. 2.14(d). After this stage, if the angle still changes, the light is not refracted, instead, it is reflected, as shown in Fig. 2.14(e).

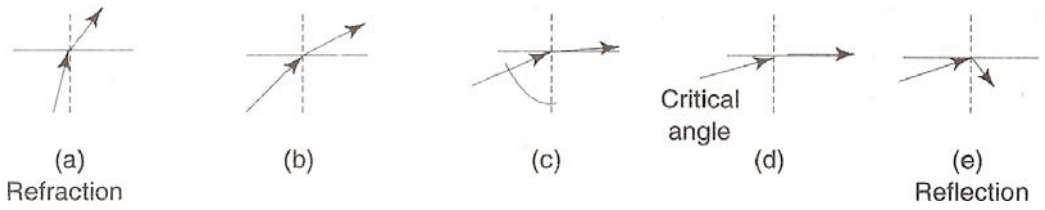


Fig 2.14 Reflection and Refraction

Optical fibers use reflection to guide the light through the optical fiber. A glass or plastic core is surrounded by a cover of less dense glass or plastic. The difference in the densities of the two is adjusted such that reflection, instead of refraction, occurs.

Propagation Modes

There are different modes of propagation, depending upon the physical characteristics of the fiber and the light source. These are shown in Fig. 2.15.

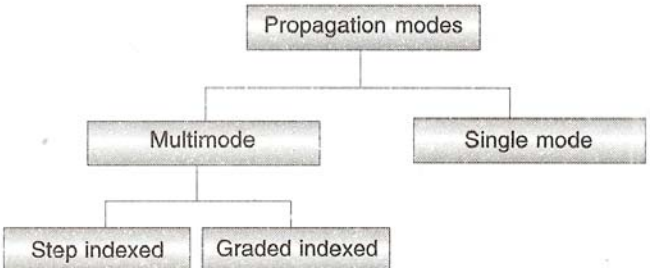


Fig 2.15. Propagation Modes

Multimode

In this case, LED is mostly used as a light source. Therefore, multiple beams pass through the core in different paths. The structure of the core decides the way they travel.

Step Index

In this case. the core has one density and the cladding has another. Therefore, at the interface, there is a sudden change. That is why it is called step index. Multiple beams take different paths on

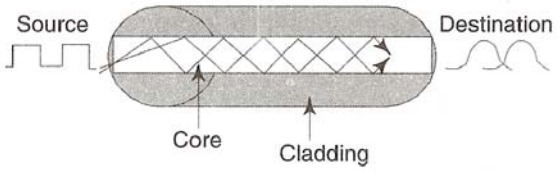


Fig 2.16 Multimode Step Index Fiber

reflection as shown in the figure 2.16. The beam that strikes the core at a smaller angle has to be reflected many more times than the beam that shifted the core at a larger angle to reach the other end. This means that at the destination, all beams do not reach simultaneously, creating diffusion and in fact confusion in terms of interpretation. This scheme is therefore used only for

short distances.

Graded Index

In this case, the core itself is made of a material of varying densities. The density is the highest at the core and gradually decreases towards the edge. Therefore, a beam goes through gradual refraction giving rise to a curve as shown in Fig. except that the horizontal beam travels unchanged.

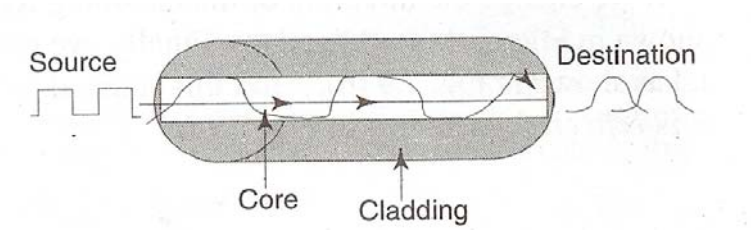


Fig. Multiple graded index Fibre

In this case also, different beams result in different curves or waveforms. Therefore, if we place the receiver at one of these intersection points, the accuracy of the signal reconstruction is enhanced substantially.

Single Mode

This uses a highly focused light beam and travels more or less horizontally. The fiber core diameter in this case is much smaller than multimode and also has far lower density. This decrease results in a critical angle close to 90 degrees to make the propagation of different beams very close to horizontal. As the propagation of different beams is more or less similar, the delays are negligible and the reconstruction of the signal that much easier as shown in figure 2.17.

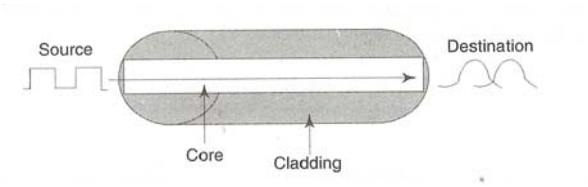


Fig 2.17. Single Mode Fiber

Advantages of Optical Fiber

Optical fiber provides the following advantages over twisted-pair and coaxial cable:

- ✓ **Resistance to noise** Optical fiber uses light rays for communication, rather than electricity. Therefore, noise is not an issue here. The only possible source of interference is external light, which is also blocked by the outer jacket of the optical fiber, thus providing resistance to interference.
- ✓ **Huge bandwidth** The bandwidth offered by optical fiber is huge as compared to twisted-pair and coaxial cable.
- ✓ **Higher signal carrying capacity** Unlike twisted-pair and coaxial cable, the signals carried by optical fiber travel longer distances (several kilometers) without needing regeneration.

Disadvantages of Optical Fiber

Along with all the advantages that it offers, optical fiber also has some disadvantages, as summarized

- ✓ **Fragility** Optical fiber is more delicate than twisted-pair and coaxial cable, so the chances of it breaking are higher.
- ✓ **Cost** Optical fiber is quite expensive because it has to be extremely pure to be able to carry signals without damage over longer distances. The

laser light source itself is also very costly, as compared to the electrical signal generators.

- ✓ **Maintenance** overhead Optical fibers need more maintenance overheads as compared to twisted-pair and coaxial cable.

2.2.2 UNGUIDED MEDIA (Wireless Communication)

Unguided media, also called as wireless communication, transport electromagnetic waves without using a physical conductor. The signals propagate through air (or sometimes water). The communication band for unguided media is as shown in Fig. 2.18. We shall concentrate on the radio communications.

(a) Radio Communication

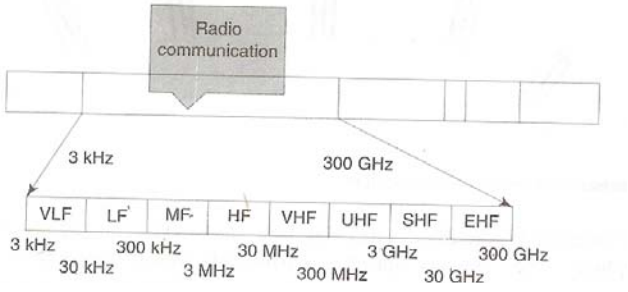


Fig 2.18. Radio Communications Band

- ✓ **Very Low Frequency (VLF)** waves propagate as surface waves, usually through air, but sometimes also through water. VLF waves do not suffer attenuation, but atmospheric noise affects them. VLF waves are usually used for long-range radio navigation and submarine communication.
- ✓ **Low Frequency (LF)** waves also propagate as surface waves. Attenuation is higher during the daytime. These wave are used for long

range radio navigation or navigation locators.

- ✓ **Middle Frequency (MF)** waves rely on line-of-sight antennas to increase and control absorption problems. These waves are used for AM radio, radio direction finding and emergency frequencies.
- ✓ **High Frequency (HF)** waves are used for amateur radio, citizen's band radio, international broadcasting, military communication, telephone, telegraph and facsimile communication.
- ✓ **Very High Frequency (VHF)** waves use line-of-sight propagation, and are used for VHF television, FM radio, aircraft AM radio and aircraft navigation.
- ✓ **Ultra High Frequency (UHF)** waves use line-of-sight propagation. They are used for television, mobile phone, cellular radio, paging and microwave links.
- ✓ **Super High Frequency (SHF)** waves are transmitted as either line-of-sight or into the space. They are used for terrestrial and satellite microwave and radar communication.
- ✓ **Extreme high Frequency (EHF)** waves are transmitted into space, and are used for scientific applications such as radar, satellite and experimental communication.

(b) Microwave Communication

As shown in Fig.2.19, microwaves use the line of sight method of propagation. as the signals do not travel along the surface of the earth. Therefore, the two antennas must be in a straight line, able to look at each other without any obstacle in between. The taller the antennas, the more is the distance that these waves travel. Usually, the antennas are positioned on

mountain tops to avoid obstacles. Microwave signals travel only in one direction at a time. This means that for two-wave communication such as in telephony, two frequencies need to be allocated. At both ends, a transceiver is' used which is a combination of a transmitter and a receiver operating at the two respective frequencies. Therefore, only one antenna can serve both the functions and cover both the frequencies.. Repeaters are used along with. The antennas to enhance the signa1. The data rates offered are 1 Mbps -10 Gbps.

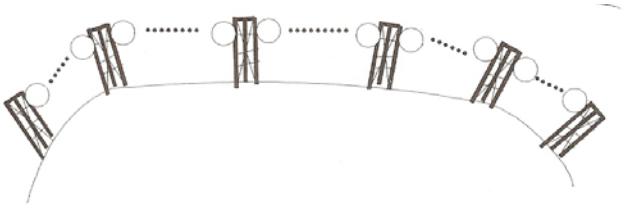


Fig.2.19 Terrestrial Microwave Communication

For many decades, before fiber optics was used, microwave communication formed an important part of telephony. The company Sprint was based on optical fiber communication, whereas the company MCI was completely based on the microwave communication initially. The name MCI, in fact, stood for Microwave Communications Inc. Lately, MCI is also changing over to optical fiber. However, microwave communication will continue to be useful as it allows you to communicate from anywhere. Microwave is also relatively inexpensive.

(c) Satellite Communication

Satellite communications is similar to the terrestrial microwave, except that the satellite acts as one of the stations. Figure 2.20 illustrates this. The satellite

does the functions of an antenna and the repeater together. For instance, as Fig. 2.20 illustrates, ground station A can send the information to ground station B via the satellite.

This, however, poses a problem. If the earth along with its ground stations is revolving and the satellite is stationery, the sending and receiving earth stations and the satellite can be out of sync over time. Therefore, normally Geosynchronous satellites are used, which move at the same Revolutions Per Minute (RPM) as that of the earth in the same direction, exactly like the earth. Thus, both the earth and the satellite complete one revolution

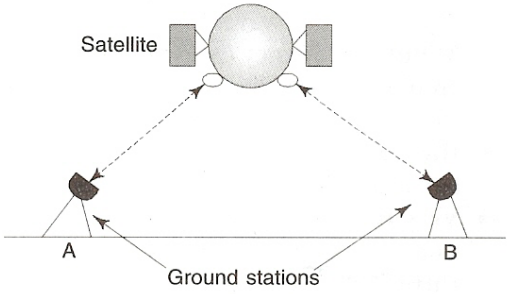


Fig.2.20 Satellite Microwave Communication

exactly in the same time: the relative position of the ground station with respect to the- satellite never changes. Therefore, the movement of the earth does not matter for the communicating nodes based on the earth. Using satellites, we can communicate from any part of the world to any other. However, a minimum of three satellite needed to cover the earth's surface entirely as shown in the Fig. 2.21.

Normally, SHF, which covers the frequency range of 3 GHz to 30 GHz

used for satellite communications. Two frequency bands are used for the signals from the earth to the satellite (called uplink), and from the satellite to the earth (called downlink).

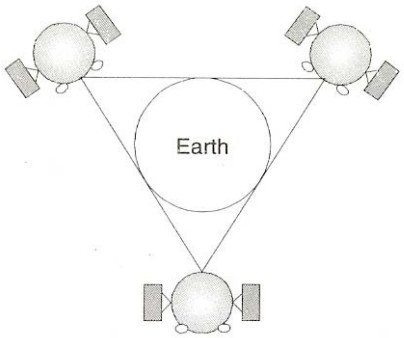


Fig.2.21 Three Satellites to Cover the Planet

Access Methods

There are three methods for communication using satellites. These three methods use principles that are similar in concept to normal wired communication. Like the wired world, satellite communication is also based on modulation techniques. The three primary modulation techniques are: (a) Frequency Division Multiple Access (FDMA), (b) Time Division Multiple Access (TDMA) and (c) Code Division Multiple Access (CDMA).

These technologies can be explained at a broad level by breaking down the title of each one.

(a) The first word tells us what the access method is and the second word, division, lets you know that it splits transmissions based on that access method.

- FDMA puts each transmission on a separate frequency
- TDMA assigns each transmission a certain portion of time on a designated frequency

- CDMA gives a unique code to each transmission and spreads it over the available set of frequencies .

(b) The last words of each category are multiple access. This simply means that more than one user (multiple) can use (access) each cell. We shall now discuss these technologies in brief.

Frequency Division Multiple Access (FDMA)

FDMA splits the total bandwidth into multiple channels. Each ground station on the earth is allocated a particular frequency group (or a range of frequencies). Within each group, the ground station can allocate different frequencies to individual channels, which are used by different stations connected to that ground station. Before the transmission begins, the transmitting ground station looks for an empty channel within the frequency range that is allocated to it and once it finds an empty channel, it allocates it to the particular transmitting station.

This is the most popular method for communication using satellites. The transmission station on earth combines multiple signals to be sent into a single carrier signal of a unique frequency by multiplexing them, similar to the way a TV transmission works. The satellite receives this single multiplexed signal. The satellite, in turn, broadcasts the received signal to the receiving earth station. The receiving station is also supposed to agree to this carrier frequency, so that it can demultiplex the received signals. The basic idea of FDMA is shown in Fig. 2.22

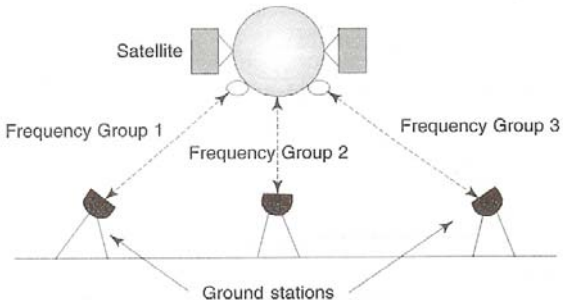


Fig.2.22 Frequency Division Multiple Access (FDMA)

Time Division Multiplexing (TDMA)

Unlike FDMA, TDMA allows access to the full bandwidth of the frequency spectrum. In TDMA, each transmitter is allocated a predefined time slot. Each transmitter receives the time slot in turn and it is allowed to transmit data for the duration of the time slot. This is shown in Fig. 2.23.

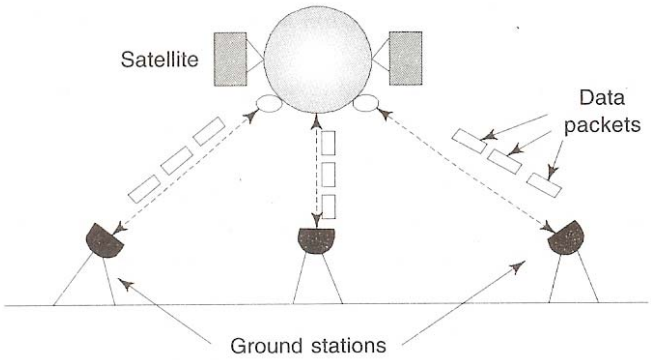


Fig.2.23 Time Division Multiple Access (TDMA)

After FDMA, TDMA is the second most popular mechanism for communication using satellites. In case of TDMA, there is no modulation of frequencies (unlike FDMA). Instead, the transmitting earth station transmits data in the form of packets of data. These data packets arrive at

the satellite one by one (hence the name TDMA). By the same logic that was discussed during TDM, TDMA is also a digital form of data transmission; TDMA operates in time domain, rather than frequency domain. Bit rates of 10-100 Mbps are common for TDMA transmissions. This can be translated into roughly 1800 simultaneous voice calls using 64 Kbps PCM.

Code Division Multiple Access (CDMA)

As we have seen, in FDMA, a fixed frequency channel is allocated to a transmission pair (i.e. the transmitter and the receiver). In the case of TDMA, transmission happens by using predefined time slots. Nothing of this sort happens in the case of CDMA. CDMA allows any transmitter to transmit in any frequency, and at any time. Thus, it is different from both FDMA and TDMA.

An obvious question is: Would this not lead to confusion and mixing of different signals? It would. However, to counter this, the transmitter and the receiver agree upon a unique coding scheme (similar to encryption) before the start of the transmission. The analogy often used to explain CDMA is an international party, where there are dozens of people. All are talking at once, and all talking in different languages that you do not understand. Suddenly, from across the room, you hear a voice speaking in your own familiar language, and your brain tunes out all the background gibberish and locks onto to that one person. Your brain understands the code being used by the other person, and vice versa.

Similarly, CDMA uses coding that is unique to a particular transmission and allows the receiver to disregard other transmissions on the same frequency.

In this case, the coding scheme is a unique frequency with which the original signal is modulated to form the codified signal that is to be transmitted. The transmitter then codifies all its transmissions in this manner, before they are sent. Since only the receiver knows how to decode this (as agreed upon before the start of the transmission), there is no scope for confusion. This is shown in Fig.2.24.

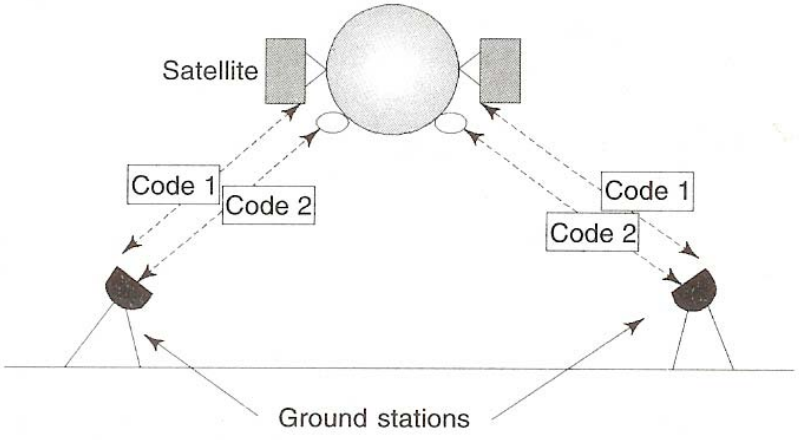


Fig.2.24 Code Division Multiple Access (CDMA)

CDMA is the newest and the least used access method using satellites. It is popular in military installations, but not so much in the case of commercial applications.

d. Cellular (Mobile) Communication

The city of St. Louis in the USA saw the emergence of the first mobile telephone as early as 1946. This system was difficult to operate and had a single channel for both sending and receiving. You had to push a button to enable the transmitter and disable the receiver. This half-duplex system, known as push-to-talk-system, was installed in the big cities in 1950s. Even today,

taxis, CB-radio etc. use the same technology.

The second development took place in 1960s. This was called Improved Mobile Telephone System (IMTS). It had a strong transmitter, and used two frequencies. One frequency was used for sending and the other for receiving. This full-duplex system had 23 channels and was a great improvement over the push-to-talk-system. However, in IMTS too, users had to wait for a long time to get a dial tone.

The third step was the development of Advanced Mobile Phone System (AMPS). In England, it is called TACS and in Japan, MCS-Ll. In this scheme, the area covered is conceptually divided in small regions known as cells, thus the name cellular phones. Though the cells are actually circular, they are shown as hexagonal for conceptual clarity in the figure that follows. Each cell has an antenna and a cell office to control that cell. A Mobile Telephone Switching Office (MTSO) controls various such cell offices and coordinates the communication between them and the Telephone Central Office (TCO) or a telephone exchange. The TCO is a part of the wired land telephone system. The computer at MTSO is responsible for not only the connections but also for the information and billing of the calls. The typical cell radius size is 0 to 12 miles. However, it can be changed, depending upon the population in the area. The whole cellular phone systme is described in the fig. 2.25 shown below.

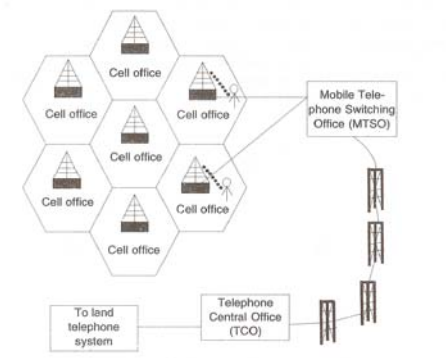


Fig.2.25 Cellular Phone System

Bands in Cellular Telephony

Classically, analog transmission is used for cellular telephony. Frequency modulation is used for communication between the mobile phone and the cell office. Normally, two frequency bands are allocated for this purpose. One of them is for the communication that is initiated by the mobile phone and the other for the land phone. However, each channel requires a full-duplex dialog. For preventing interference, adjacent channels are rarely allocated. Some channels are also required for control purposes. This reduces the number of channels available for each cell. This number is 40 in USA today. However, the same frequency band can be used for multiple non-adjacent cells as shown in Fig. 2.26. In USA, FCC controls these bands. The two bands are typically 824-849 MHz and 869-894 MHz.

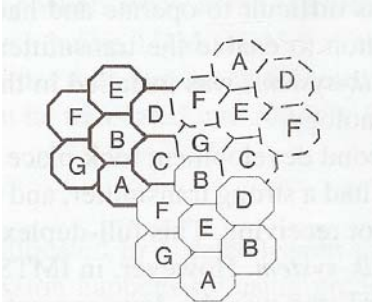


Fig. 2.26 Frequency Reuse

Calls Using Mobile Phones

When a call is made from the mobile phone by entering a 7-,8- or lO-digit phone number, the mobile phone itself scans the band and seeks a channel for setting up the call. After this, it sends this number to the closest cell office, which, in turn, sends it to MTSO. The MTSO, in turn, sends it to the CTO. If

the called party is available, CTO lets MTSO know. At this juncture, MTSO allocates an empty voice channel to the cell to establish the connection. The mobile phone adjusts its tuning to the new channel and the dialog begins.

When a land phone places a call to a mobile phone, the telephone central office sends the number to the MTSO. The MTSO performs a lookup to see where the mobile phone is currently placed by sending appropriate query signals to all the cells. This process is called paging. The cell where the mobile phone is currently located responds to the MTSO. The MTSO then transmits the incoming call signal to that mobile phone, and when the mobile phone is answered, the MTSO assigns a voice channel to the call, thus enabling the conversation.

Transmitting/Receiving/Handoff Operations

During the conversation, if the mobile phone crosses the cell, the signal can become weak. The MTSO constantly checks the signal level, and if it finds it low, it immediately seeks a new cell that can look after the communication better. The MTSO then changes the cell-carrying channel so smoothly that the user hardly notices. This process of handling the signal off from the old channel to the new one is called handoff. This process is described below.

Step 1: Cell A senses that the user of cell 50 is moving to cell B, and that its signal is becoming weaker. So, it alerts the MTSO. This is shown in Fig. 2.27

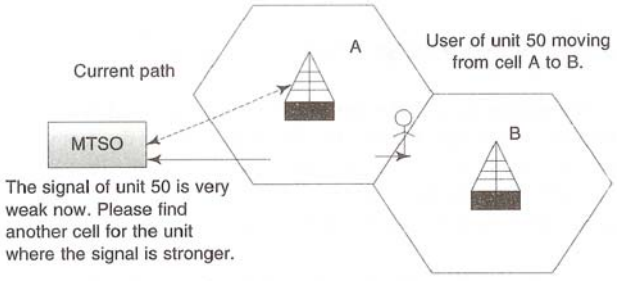


Fig.2.27 Handoff Part l-A Unit Becomes

Weak in Cell A

Step 2: The MTSO wants to check if any of the adjacent units (B, C, etc.) can take up the responsibility of unit 50. Cell C responds by saying that it cannot do so, as unit 50 is weak in cell C, too. However, cell B says that it is ready to take up the responsibility of unit 50 as it is receiving a strong signal from unit 50. This is shown in Fig. 2.28.

Step 3: The MTSO redirects unit 50 from cell A to cell B. This is shown in Fig. 2.29

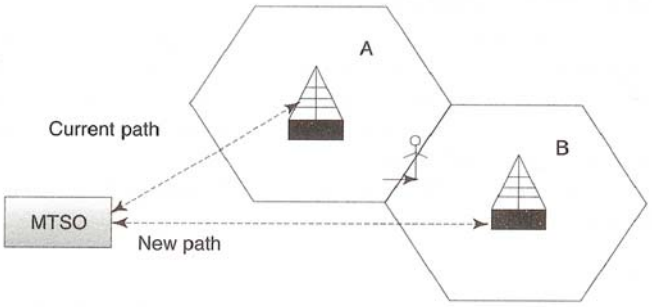


Fig. 2.28 Handoff Part 2 – MTSO Enquires to see if Anybody can take up unit 50

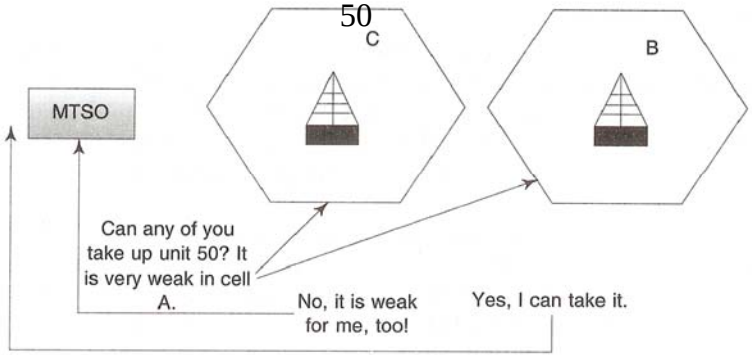


Fig. 2.29 Handoff Part 3 – MTSO Hands over Unit 50 to Cell B

New Developments

Three developments will take place in the near future. One is digital cellular telephone. The second is the integration of cellular phones with

satellite communication. This will enable a person to have a unique but same telephone number throughout the world, unlike today, when we have different numbers for land phones and mobile phones. The third is the integration of the mobile telephony with the PC. The scheme is called Mobile Personnel Communication, which will allow people to use a small, mobile PC to send and receive multimedia information in all forms: data, voice, image and video.

Figure 2.30 summarizes the various transmission media in terms of their characteristics and other features.

Medium	**Speed**	**Cost**	**Security**	**Attenuation**
UTP	1-100 Mbps	Low	Low	High
STP	1-150 Mbps	Medium	Low	High
COAX	1 Mbps-1 Gbps	Medium	Low	High
Fiber	10 Mbps-2 Gbps	High	High	Low
Radio	1-10 Mbps	Medium	Low	High-Low
Terrestrial Microwave	1 Mbps-10 Gbps	High	Medium	Variable
Satellite Microwave	1 Mbps-10 Gbps	High	Medium	Variable
Cellular	1.6-1.2 Kbps	High	Low	Low

Fig.2.30 Transmission Media Characteristics

Shannon Capacity

Regardless of whether it is a guided or unguided medium, every transmission medium has a finite data carrying capacity. We are normally very interested in knowing how much data a particular transmission medium can carry. Claude Shannon solved this mystery in 1944 by introducing a formula

that determines the maximum data rate of a channel. This is called as the Shannon capacity.

Shannon capacity essentially depends on one basic factor of a transmission medium: its signal-to noise ratio. Understanding how to calculate signal-to-noise ratio would take us to physics, therefore, we shall avoid it. Instead, we shall have a look at the formula to determine the maximum data rate of a transmission medium (called as Shannon capacity). The formula reads:

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

C is the Shannon capacity in bps, B is the bandwidth of the transmission medium, and SIN is the signal-to-noise ration of the transmission medium.

For instance, let us calculate the theoretical upper limit of transmission rate through a normal telephone channel. Normally, a telephone channel has a bandwidth of 3000 Hz (300 to 3300 Hz). The SIN ration is usually around 3162. Thus, its capacity C will be:

$$C = B \log_2 (1 + SIN)$$
$$= 3000 \log_2 (1 + 3162)$$
$$= 3000 \log_2 (3163)$$
$$= 3000 \times 11.62$$
$$= 34{,}860 \text{ bps or } 34.860 \text{ Kbps}$$

This signifies that no matter how hard you try, you cannot transmit more than 34.860 Kbps data through a telephone line. To achieve higher transmission rates, you must try and improve the S/N ratio.

Thus, Shannon capacity formula informs engineers that no amount of intelligent encoding can break the laws of physics that place a basic limit on the number of bits that can be transmitted per second over a communications

system.

2.3 Types of Transmissions

Digital data can be transmitted in a number of ways from the source to the destination. These modes of data transmission can be outlined as follows:

- Parallel and serial communication
- Asynchronous, synchronous and isochronous communication
- Simplex, half-duplex and full-duplex communication

These techniques will be discussed first followed by the concept of multiplexing. Multiplexing allows more than one signal to be sent on a single transmission path using techniques that help a more effective use of the transmission medium than is possible otherwise.

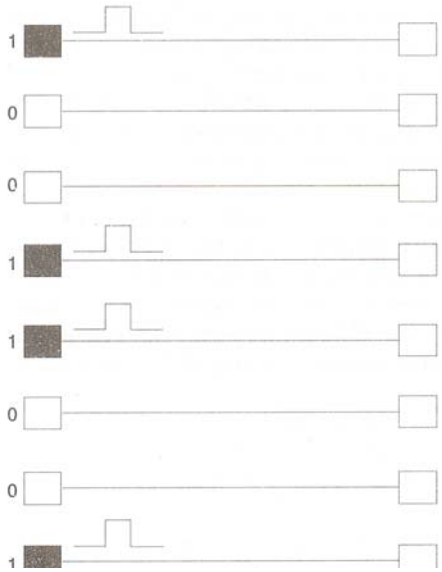
2.3.1 PARALLEL COMMUNICATION

Imagine that we want to send digital data stored as bytes of 8 bits each, and in words of may be 16 or 32 bits, depending upon the word length;

In parallel communication, we transfer a word or a byte at a time. Therefore, we need to have those many wires parallel to each other, each carrying a single bit. This is shown in Fig.2.31, where eight wires are transmitting a byte at a time. In this example, the byte contains 10011001.

This is a very fast method of transmitting data from one place to another. However, there are problems in this scheme. In the first place, it a very expensive method because it requires several wires in sending as well as receiving equipment. In addition, it demands extraordinary accuracy, which cannot be guaranteed over long distances. The digital pulses ma not traverse at the same seed, because the underlying media through which the data passes from one computer system to another may not be exactly identical. This gives

rise



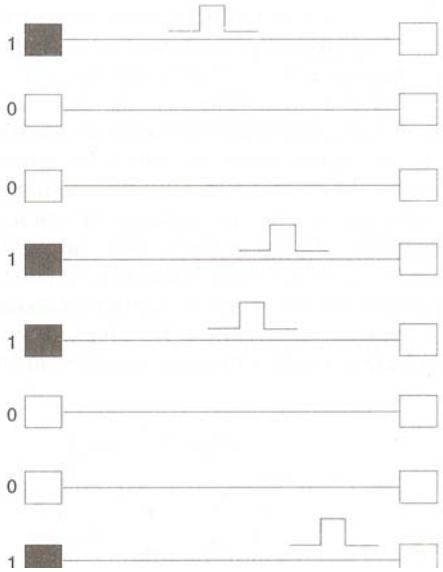


Fig. 2.31 Parallel Transmission **Fig.2.32** The Problem of Skew

When a message is skewed, the bits 10011001 are sent from the source to the destination, but they traverse at different speeds At the...destination, the measurement of signal values to determine whether it was a bit 0 or 1 have to be done at the same time for all the bits. Therefore, the problem of skew can result In an inaccurate interpretation of bits.

To avoid this problem, parallel transmission is used only for a very short distance and there too, all the parallel wires have to De absolutely identical. This method is used for data transmission within the computer system, such as from the CPU registers to the memory or vice versa through the data bus. The data bus essentially implements the parallel transmission of data. This is done because speed, rather than cost, is of paramount importance in this case. However, in view of the problems outlined above, this method cannot be used over long distances.

2.3.3 Serial Communication

Over long distances, serial communication is used. While sending the data serially, characters or bytes have to be separated and sent bit by bit. Thus, some hardware is needed to convert the data from parallel to serial. At the destination, the measurement of the signal values is done in the middle of the bit duration as shown in Fig. This is because, if the values are taken at the edge or a point where the pit value changes (shown as point A in the figure), the reading will be indeterminate.

At the destination, all the bits are collected, measured and put together as bytes in the memory of the destination. This requires conversion from serial to parallel.

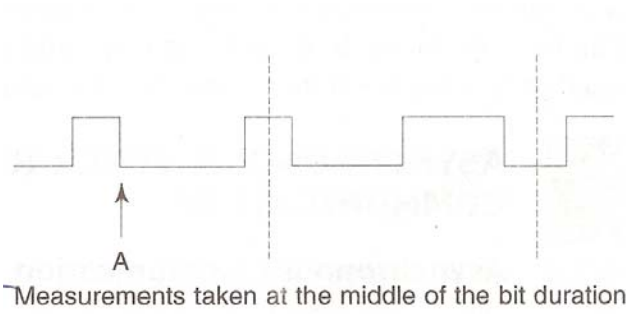


Fig 2.33. Serial Transmission

In serial data transmission, we have to identify where the character starts. We have to identify where the character starts. We have to also identify the middle position of each bit interval so that measurement can be taken. Normally, the transmitter and the receiver have two different clocks. The point is to synchronize the clock of the receiver exactly with that of the transmitter, so that correct readings will result and the bit values are understood correctly. This is essentially a problem of synchronization. We can take an example to illustrate this.

Let us say that we have to send 8 bits b_0 to b_7 from point X to point Y as shown in Fig. 2.34 We have arranged the bits in a circular fashion to illustrate the point better. Let us say that arms Al and A2 at point X and Y rotate in clockwise directions. When both Al and A2 point to the bit 1'0 at X and Y respectively, the connection is made and the bit b_0 is transferred from X to Y. Now due to rotation, after some time, Al will point towards b_l at point X. At that time A2 also has to point to b_l. at point Y. Otherwise the bit will not be transferred correctly. The same is true for the transfer of b_2 through b_7 as A I and A2 rotate clockwise. Therefore, the two arms A I and A2 have to be perfectly synchronized and they have to rotate at the same speed in order to succeed in sending/receiving all the bits accurately.

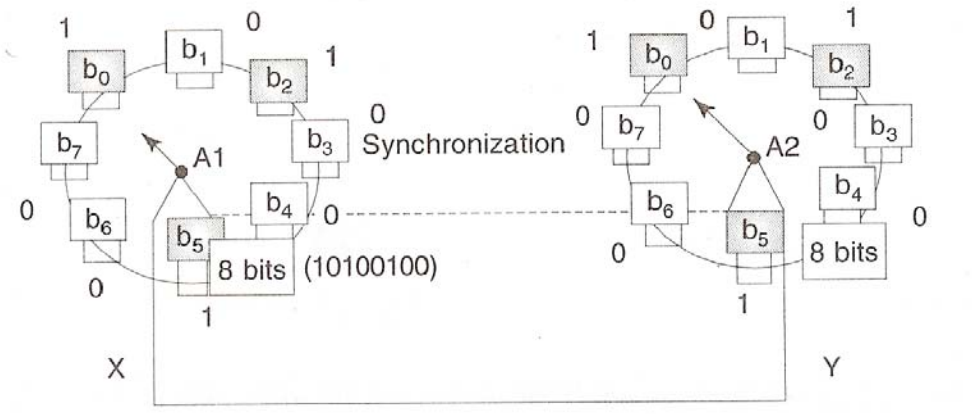


Fig 2.34. Serial Transmission and Synchronization

In order to achieve this, initially, we have to adjust and synchronize the clocks in the electronics of 'the source and the destination. There are three ways in which this can be achieved. These are: asynchronous, synchronous and isochronous transmission. We will look at them very shortly.

A line adapter or interface generally achieves this task of synchronization. It achieves a few other things also. We will call it only serial interface in this text.

Electronics devices called Universal Asynchronous Receiver Transmitter (UART) and Universal Synchronous Asynchronous Receiver Transmitter (USART) are examples of such an interface. USART can be used for both the asynchranous and synchronous transmission. On the other hand, UART is used only when the transmission is asynchronous.

2.3.3 ASYNCHRONOUS COMMUNICATION

Asynchronous Communication

We have learnt why serial communication is preferred to parallel communication over a long distance. However, we have only one problem in serial communication. How do we identify individual bits? Essentially, the problem is that of synchronizing the sender (source) and the receiver (destination) as outlined above.

In asynchronous communication, the time when a character will be sent cannot be predicted. Imagine, for instance, data entry operator. She may key in a few characters, then may consult her colleague, then key in a few more characters before she goes off for lunch. In such a scenario, where the data is transmitted as characters and at an unpredictable pace, we have to synchronize the source and destination for each character. This is called asynchronous communication.

In this case, each character is preceded with a start bit and succeeded with 1, 1.5 or 2 stop bits. Figure 2.35 depicts this. Normally NRZ-L signaling is used for asynchronous transmission. According to this convention, a negative

voltage denotes a binary 1 bit and a positive voltage denotes a binary 0 bit. When the line is idle i.e. when no character is being sent over the line, a constant negative voltage signifying a binary 'l' is generated.

- ✓ When the character is to be sent, bit 0 is sent first. This positive voltage is called the start bit. All the bits of the character, according to the coding scheme used (ASCII/EBCDIC), then follow, each of which could have a 0 or 1 value.
- ✓ A parity bit, if used, then follows
- ✓ In the end, 1, 1.5 or 2 stop bits are added. The stop bit again corresponds to the idle state, which is bit 1. In this context 1.5 bits only means that the signal with negative voltage denoting 1 is generated for 1.5 times the normal bit interval.

The sender and receiver have to exactly agree on the number of bits that represent a character (7 or 8), the parity bit scheme if used, the bit interval (or bit rate) and also what represents 0 and 1 (e.g. NRZ-L).

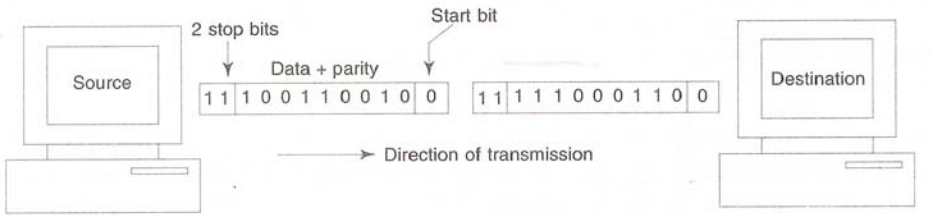


Fig. 2.35 Asynchronous Communication

Essentially, when the start bit is recognized by the destination, it knows that a character is being sent. It then adjusts and starts its clock so that measurement of signal values can now start at the middle of every bit that follows for the expected number of bits. After all the bits in the character

including the parity bit are measured and stored, it expected a stop bit, which is encountered if everything is OK. This is to recognize the bits being sent. This continues until the stop bit is received. At this time, the, line becomes idle, and the destination clock also goes to sleep. On receiving the start bit, it wakes up for the duration of that character, and after the full character is received, again goes to sleep until the next character arrives. This makes sense where (a) the rate at which the characters arrive is unpredictable and (b) the speed at which the characters are keyed in is far less than the speed with which they are transmitted or processed.

There has to be a proper understanding between the source and destination about the bit rate, in bps, from which the bit interval can computed to determine when the signal measurements have to be taken. Using this. the destination waits for the start bit and then starts its clock to measure values to recognize the bits including the parity bits.

2.3.4 Synchronous Communication

In synchronous transmission, the whole block of data bits is transferred at once, instead of one character at a time. The block of bits mayor may not consist of different characters. It could well be a digitized image. It could be anything. In the latter case, we need a bit-oriented protocol between the sender and the receiver. If the block of bits consists of different characters, the receiver needs to know the beginning of the first bit of the first character. It can then start its clock and start sampling to determine the bit values. Knowing the number of bits per character, it can now recognize the character and store them at the destination. This is shown in Fig. 2.36. In this case, we need to have a byte-oriented protocol between the sender and the receiver.

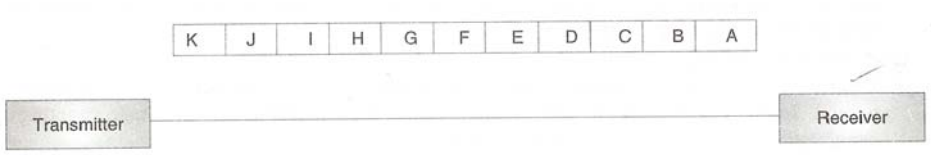


Fig. 2.36 Synchronous Communication

The point is: How does the receiver know the first bit of the first character, or when to start the clock for sampling? In order to perform this synchronization, each data block is preceded with a unique synchronizing bit pattern. We use the SYN (abbreviation of synchronize) transmission control character for this. It has a bit pattern of 0010110. This bit pattern is generally not used in any normal communication. Therefore, this SYN character can be reserved for indicating the start of the block, i.e. for synchronization.

However, there is a problem in this scheme. The transmitter will not send a SYN character as a part of the data. However mistake, the bit pattern of two characters could he such that if sent one after the other. They can constitute a SYN character, thereby fooling the receiver. Figure 2.37 illustrates this case. In this case, if the ASCII characters b and a are sent one after the other, we get a bit combination of 4 bits from the ASCII character band 3 bits from the ASCII character a to constitute the SYN character.

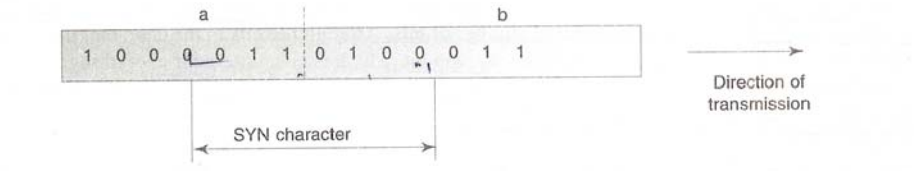


Fig. 2.37 A Misleading SYN Character

At the receiving end, if the receiver is all the time looking for SYN characters, it can get fooled. It is for this reason that normally two SYN bytes are sent consecutively. The bit combination of two SYN bytes-Leo 00101100010110--cannot be obtained by concatenating any characters. Therefore, the receiver is asked to look for two SYN characters. If not found, it is asked to throw them off. In some systems even three or four SYN characters are sent to make sure that proper synchronization takes place. The receiver gets ready after receiving the required SYN character(s) and then starts its clock for measuring the bit values for the actual data. This is shown in Fig. 2.38.

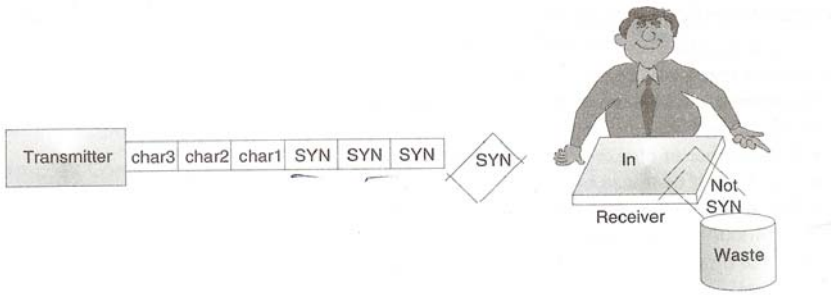


Fig. 2.38 Discarding Non-SYN Characters Until SYN is Received for Synchronization

When the clock starts measuring the bit values, the counter within a receiver is increment every bit receive and Measured and pushed into the character assembler within the receiver. This is as shown in Fig. 2.39.

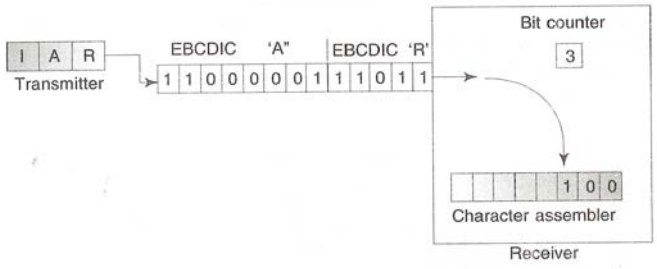


Fig 2.39 Characters Received

After a character is assembled, the character is moved to a separate buffer and the bit counter is set to 0 to prepare for the next character. In the case of synchronous transmission. the equipment used at both the ends is called Universal Synchronous/Asynchronous Receiver Transmitter (USART). This is responsible for converting the data from parallel to serial at the transmitter end .and from serial to parallel at the receiver end. It is also responsible for generating the SYN characters at the beginning of the data blocks, before transmission and recognizing them at the receiving end as shown m Flg. 2.39.

Synchronous communication is used w en large amount of data is to be sent from one place to the other. Syfithronous transmission has obviously a much higher efficiency because it is a continuous transmission, without any start/stop bits. For example, if we have to transmit a file of 100,000 characters at a rate of 9.6 kbps, the overheads (extra bits) for asynchronous and synchronous communication will be as follows:

Asynchronous Communication

Let us assume that for each character of 8 bits, 1 start bit and 1 stop bit are sent. Thus, at 2 bits/ character as overhead, we will have to send an overhead of:

- 2 bits/character * 100,000 character = 200,000 bits
- At 9600 bits/seconds, it will take 200,000/9600 = 20.83 seconds more for the overhead bits

Synchronous Communication

We assume that the file of 100,000 characters is broken into blocks of 1200 characters = 9600 bits. We will assume that 48 overhead bits are sent along with each block. (For SYN (synchronize), STX (start of transmission), ETX

(end of transmission) and other characters). The overhead is now computed as follows:

- The number of blocks = 100,000/1200 = 250/3
- Numbers of overhead bits = 250/3 * 48 = 4000 bits
- The time taken to send overhead bits at 9600 bps;
  = 4000/9600 seconds = 0.4167 seconds

Thus, the time taken for asynchronous communication is much higher: in this case, 50 times higher

2.3.5 Isochronous Communication

This method combines the approaches of asynchronous and synchronous communications. In this method, as in the asynchronous method, each character has both the start and stop bits. However the idle period between the two characters is not random. For instance, all idle periods of no transmission c9nsist of an exact multiple of one-character time interval. Therefore, if the time to transmit a character (including its parity, start, stop bits) is t, the time interval between characters cannot be random as in asynchronous method. It is also not 0 as in synchronous method. It has to be t, 2t, 3t ... nt where n is a positive integer in isochronous communication. Figure 2.40 depicts the differences between the three approaches.

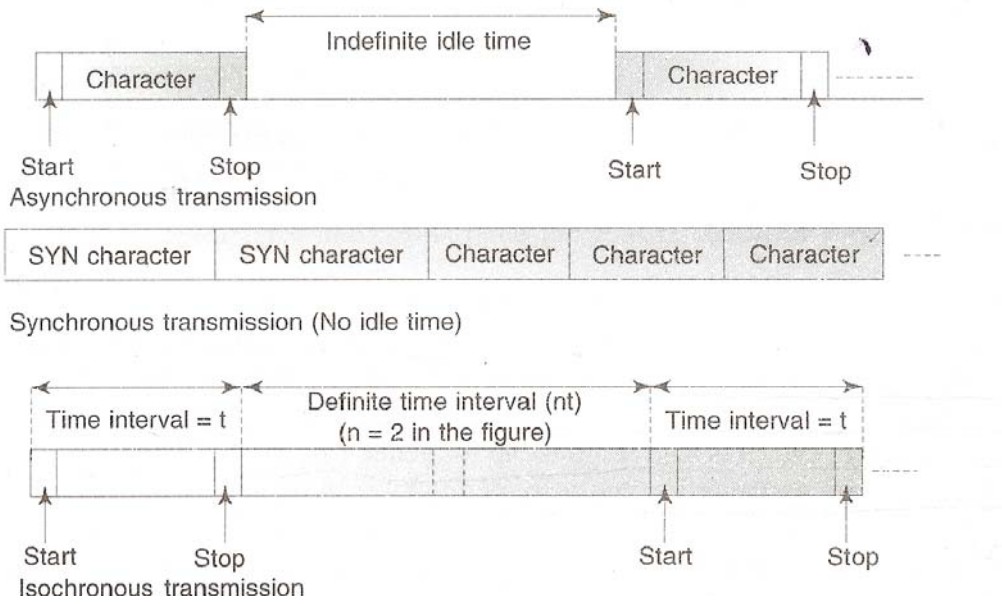


Fig.2.40 Asynchronous, Synchronous and Isochronous Transmissions

The main reason for using isochronous method to asynchronous method is speed. In practice, asynchronous transmission is limited to the data rate of 2,400 bits per second according to the precision of timing in the transmitting and receiving modems. By contrast, isochronous transmission can achieve the data rates of up to 19,200 bits per second.

2.3.6 Simplex, half-duplex and full-duplex Communication

This classification of data transmission is based on which of the communicating devices can send data, and at what point of time. There are basically three ways in which this can be done Simplex, Half-Duplex and Duplex. These are very simple to understand, as explained in the next section.

Simplex Communication

In simplex mode, the communication is unidirectional. This is similar to a one-way street, where vehicles are allowed to drive only in a single direction, as shown in Fig. 2.41

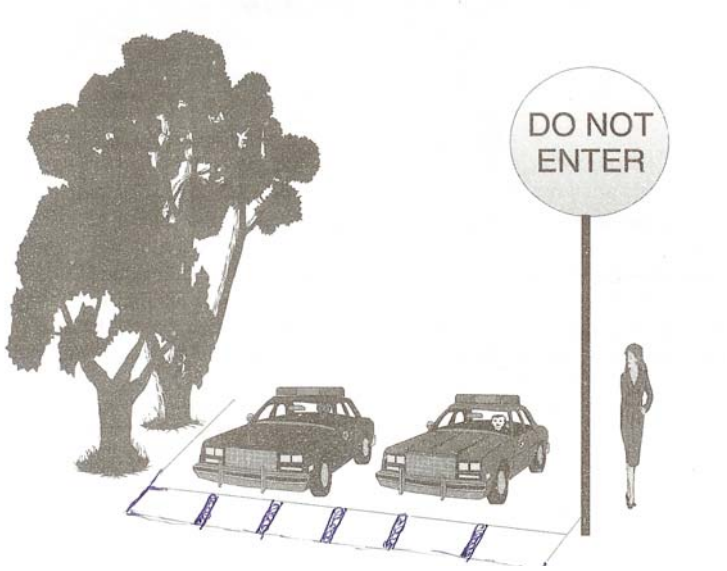


Fig.2.41 A Case for Simplex Communication

Here, one of the communicating devices can only send data, whereas the other can only receive it like in a radio or a TV today. Keyboard to computer monitor data transmission is a simple example of this. Another example is shown in Fig. 2.41, where one host can only send data, whereas the other can only receive it.

Half-duplex Communication

Unlike what happens in the simplex mode, in the half-duplex mode, both devices can transmit data, though not at the same time. When one device is sending data, the other must only receive it, and vice versa. This is

conceptually similar to a street that has a single lane for vehicle traffic. When vehicles from one side are coming, the vehicles from the other side must wait. This is shown in Fig. 2.42.

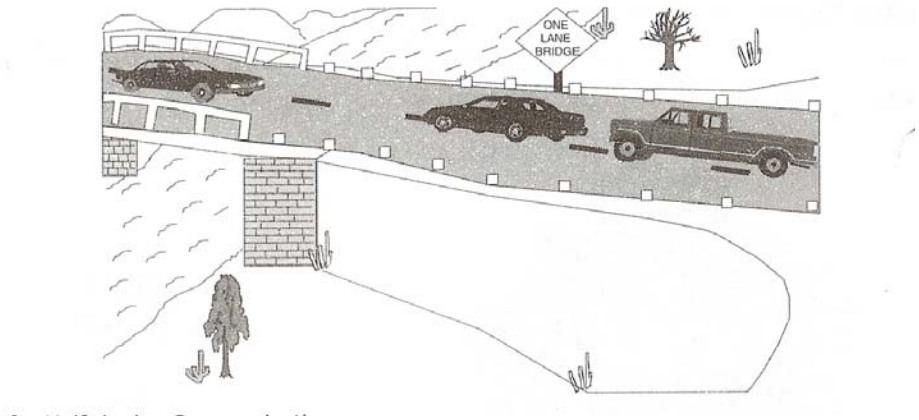


Fig. 2.42 A Case for Half-duplex Communication

Thus, both sides take turns to send data as shown in Fig. 2.43. This requires a definite turn around time during which the device changes from the receiving mode to the transmitting mode. Due to this delay, half-duplex communication is slower than simplex communication. However, it is more convenient than simplex communication, as both the devices can send and receive the data.

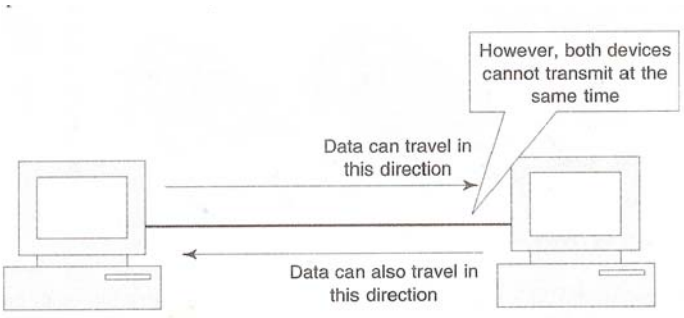


Fig.2.43 Half-duplex Communication

Half-duplex communication is normally implemented by using a two-wire circuit: one for data, and one for ground. In this case, the full bandwidth of the wire is used while sending the data in either direction. Examples of half-duplex communication are conversations over walkie-talkie.

Full-duplex Communication

In full-duplex (or simply duplex) communication mode, both the devices can transmit data at the same time. It means that both devices are capable of sending as well as receiving data at the same time. This is like a two-way street with traffic flowing in both directions at the same time. This is shown in Fig. 2.44. It is also similar to a telephone conversation, where both parties can talk to each other simultaneously.

This can be done using a two-wire circuit or a four-wire circuit. In a two-wire circuit, one wire is used for data and one for ground as in half-duplex. However, the bandwidth of the wire for data is divided in two channels for carrying data in either direction. Thus, each channel can use only half the bandwidth normally. This is shown in Fig.

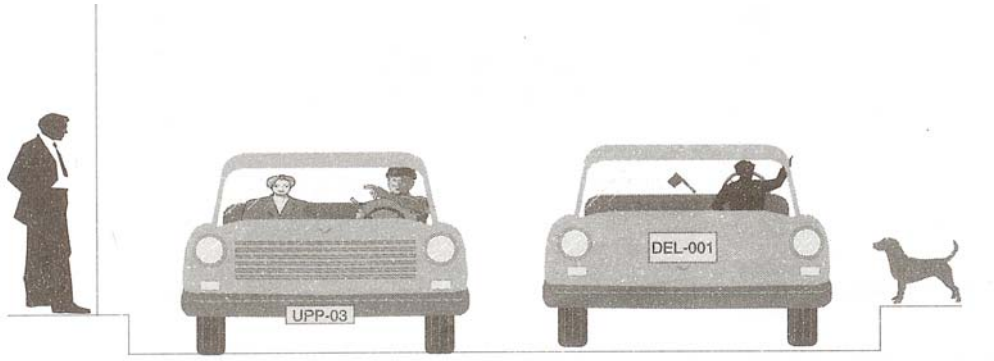


Fig.2.44 Full-duplex Communication

In four-wire circuit, there are two wires for data and two for ground. Thus, we can have one data wire for the transmission in each direction, increasing the bandwidth and therefore data rate.

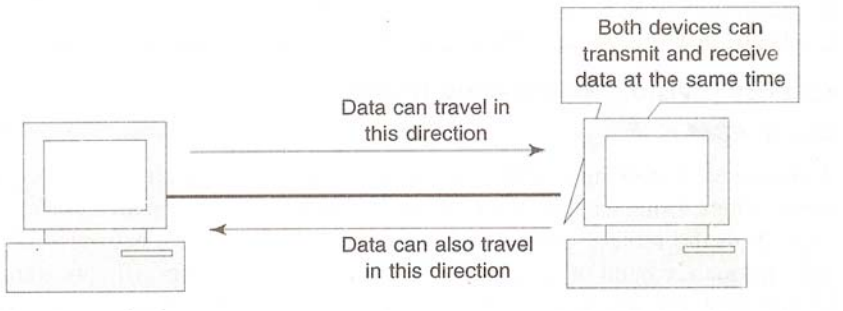


Fig 2.46 (Full duplex communication)

2.4 Summary

At the core of all computer to computer data communications is the concept of signal propagation. The simple idea in signal propagation is that when a signal is applied in any form (such as heat, voltage or current) to one end of a conducting material, after some time the signal reaches the other end of the material. Thus, a wire can also be used for carrying data between the two points that it connects.

Fourier proved that any signal could be shown to be equivalent to the addition of one or many sinusoidal signals; and that this was true about analog as well as digital signals. Any sinusoidal signal has three properties: amplitude (the strength of the signal), frequency (the number of cycles completed in one second) and the phase. The bandwidth of a signal is the difference between its highest and lowest frequencies. frequency can be calculated *as* f = I/T, where f is the frequency and *T* is the time period.

An analog signal is a continuous waveform. A digital signal is a discrete

waveform. The bandwidth of a digital signal is infinite.

Signal, Digital Transmission, (c) Digital Signal, Analog Transmission and (d) Analog Signal, Digital Transmission. That is, a signal can be transmitted as it is, or can be transformed into the other form (from analog to digital or vice versa) before it is transmitted.

The telephone is an example of an analog signal being transmitted as it is. The human voice is an analog sound, which can be sent as an analog electrical signal over telephone wires.

When computer data is sent over telephone wires as digital pulses, it is a case of digital transmission of digital data. Here, repeaters are used for reproducing the signal at periodic distances, so that it does not become distorted beyond repair as it reaches the destination.

Creating digital lines is a very expensive and time-consuming process. Therefore, engineers use the existing analog telephone wires to send digital computer data.

In parallel communication, we transfer a word or a byte at a time. Therefore, we need to have those many wires parallel to each other, each carrying a single bit. Over long distances, serial communication is used. While sending the data serially, characters or bytes have to be separated and sent out bit by bit. Thus, some hardware is necessary to convert the data from parallel to serial. At the destination, the measurement of the signal values is done at the middle of the bit durations.

In asynchronous communication, the time at which a data operator will send a character cannot be predicted. Thus, each character is preceded by a start bit and succeeded by I, 1.5 or 2 stop bits. In synchronous transmission, the

whole block of data bits is transferred at once, instead of a character at a time. Synchronous communication is used when large amount of data, such as a file, is to be sent from one place to the other. Synchronous transmission has obviously a much higher efficiency because it is a continuous transmission, without any start/stop bits. The isochronous method combines the approaches of asynchronous and synchronous communications. In this method, as in the asynchronous method, each character has both the start and stop bits. However in the isochronous method, the idle time between characters is not random.

In simplex mode, communication is unidirectional. This is similar to a one-way street, where vehicles are allowed to drive only in a single direction. In case of half-duplex mode, both devices can transmit data though not at the same time. When one device is sending data, the other must only receive it, and vice versa. This is conceptually similar to a street that has a single lane for vehicle traffic. In full duplex (or simply duplex) communication mode, both the devices can transmit data at the same time. It means that both devices are capable of sending as well as receiving data at the same time. This is like a two-way street with traffic flowing in both directions at the same time.

Transmission media can be classified into two main categories: guided media and unguided media. The difference between the two is that guided media use physical cable for data transmission, whereas in the case of unguided media, air is the transmission medium.

Twisted copper wire, coaxial cable and optical fiber are the three main types of guided media. They differ in their physical characteristics as well as transmission capacities. Optical fiber is the latest and the best of them all. However, it is the most expensive as well. The traditional transmission medium

is twisted pair.

Unguided media can be radio, microwave, satellite or cellular. Satellite communications take place using one of the three techniques: (a) Frequency Division Multiple Access (FDMA), (b) Time Division Multiple Access (TDMA) and (c) Code Division Multiple Access (CDMA). FDMA puts each transmission on a separate frequency; TDMA assigns each transmission a certain portion of time on a designated frequency; CDMA gives a unique code to each transmission and spreads it over the available set of frequencies.

Security is a major concern in the case of unguided media. Of these, microwave and satellite offer the highest data transmission rates. Cellular phones, which are becoming popular all over the world, offer the most modem way of communication using unguided media.

2.4 Self Assessment Questions

1. What are analog and digital signals? Explain the advantages of analog and digital signals.
2. Explain the following with suitable diagram
   (a) Amplitude
   (b) Period
   (c) Frequency
   (d) Phase
3. Explain the Nyquist's theorem for maximum data rate of the channel.
4. Give a brief description of the application and limitations of the following
   (a) Co-axial cable
   (b) Twisted pair cable

(c) Optical fiber

(d) Microwaves

5. Explain the structure and data transmission mechanism in optical fiber. Highlight the various advantages and disadvantages of optical fiber.
6. Write short note on the following

(a) Radio communication

(b) Microwave communication

(c) Satellite communication

(d) Cellular communication

Suggested Readings

1. Tannenbaum,"Computer Network", Pearson Education
2. Godbole,"Data Communications and Networking", Tata Mc Graw-Hill
3. Forouzan," Communications and Networking", Tata Mc Graw-Hill

Subject: Computer Networks **Subject Code:**MCA-301

Lesson No.: 3 **Author:** Mr. Yudhvir Singh

Lesson Title: Medium Access Sublayer **Vetter: Dr. Sandeep Arya**

Structure

3.0 Objective

This lesson concerns the medium access sub-layer, which is part of the data link layer. The basic question it deals with is how to determine that who may use the network next when the network consists of a single shared channel, as in most networks. In this chapter we will discuss Channel Allocation Problem, Access Methods, Multiple Access Protocols, ALOHA, Carrier Sense Multiple Access Protocols, Collision-Free Protocols.

3.1 Introduction

The medium access sub layer, which is part of the data link layer, it deals how to determine that who may use the network next when the network consists of a single shared channel, as in most networks. This layer is also known as the Medium Access Control Sub-layer. Networks can be divided into two categories: those using point-to-point connections and those using broadcast channels. 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 so that each one can hear and talk to all the others. It is very likely that when one of them stops speaking, two or more will start talking at once, leading to chaos. 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 multi-access channel belong to a sub-layer of the data link layer called the MAC (Medium Access Control) sub-layer. The MAC sub-layer is especially important in LANs, many of which use a multi-access channel as the basis for communication. For most

people, understanding protocols involving multiple parties is easier after two-party protocols are well understood. For that reason we have deviated slightly from a strict bottom-up order of presentation.

3.2 The Channel Allocation Problem

The central theme of this chapter is how to allocate a single broadcast channel among competing users.

Static Channel Allocation

The traditional way of allocating a single channel, eg. a telephone line, among multiple competing users is Frequency Division Multiplexing (FDM). If there are N users, the bandwidth is divided into N equal-sized portions, each user being assigned one portion. Since each user has a private frequency band, there is no interference between users. When there are only a small and constant number of users, each of which has a heavy load of traffic, FDM is a simple and efficient allocation mechanism.

When the number of senders is large and continuously varying or the traffic is bursty, FDM presents some problems. If the spectrum is cut up into N regions and fewer than N users are currently interested in communicating, a large piece of valuable spectrum will be wasted. If more than N users want to communicate, some of them will be denied permission for lack of bandwidth, even if some of the users who have been assigned a frequency band hardly ever transmit or receive anything.

Even assuming that the number of users could somehow be held constant at N, dividing the single available channel into static sub-channels is inherently inefficient. The basic problem is that when some users are quiescent, their

bandwidth is simply lost. They are not using it, and no one else is allowed to use it either. The poor performance of static FDM can easily be seen from a simple queuing theory calculation. Let us start with the mean time delay, T, for a channel of capacity C bps, with an arrival rate of F frames/sec, each frame having a length drawn from an exponential probability density function with mean 1/μ bits/frame. With these parameters the arrival rate is F frames/sec and the service rate is μC frames/sec. From queuing theory it can be shown that for Poisson arrival and service times.

$$T=1/(\mu C-F) \qquad \text{----------(i)}$$

For example, if C is 100 Mbps, the mean frame length, 1/μ, is 10,000 bits, and the frame arrival rate, F, is 5000 frames/sec, then T = 200 μsec. Note that if we ignored the queuing delay and just asked how long it takes to send a 10,000 bit frame on a 100-Mbps network, we would get the (incorrect) answer of 100 μsec. That result only holds when there is no contention for the channel.

Now let us divide the single channel into N independent sub-channels, each with capacity C/N bps. The mean input rate on each of the sub-channels will now be F/N. Re-computing T we get, Equation (ii)

$$T_{FDM} = 1/(\mu(C/N) - (F/N) = N/(\mu C-F) = NT \qquad \text{----------(ii)}$$

The mean delay using FDM is N times worse than if all the frames were somehow magically arranged orderly in a big central queue.

Precisely the same arguments that apply to FDM also apply to time division multiplexing (TDM). Each user is statically allocated every Nth time slot. If a user does not use the allocated slot, it just lies fallow. The same holds if we split up the networks physically. Using our previous example again, if we were to replace the 100-Mbps networks with 10 networks of 10 Mbps each and

statically allocate each user to one of them, the mean delay would jump from 200 μsec to 2 msec.

Dynamic Channel Allocation

Some key assumptions about this are described below.

Station Model. The model consists of N independent stations, each with a program or user that generates frames for transmission. Stations are sometimes called terminals. The probability of a frame being generated in an interval of length dt is F dt, where F 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.

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.

Collision Assumption. If two frames are transmitted simultaneously, they overlap in time and the resulting signal is garbled. This event is called a collision. All stations can detect collisions. A collided frame must be transmitted again later. There are no errors other than those generated by collisions.

Continuous Time. Frame transmission can begin at any instant. There is no master clock dividing time into discrete intervals.

Slotted Time. Time is divided into discrete intervals (slots). Frame transmissions always begin at the start of a slot. A slot may contain 0, 1, or more frames, corresponding to an idle slot, a successful transmission, or a collision, respectively.

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.

No Carrier Sense. Stations cannot sense the channel before trying to use it. They just go ahead and transmit. Only later can they determine whether the transmission was successful.

3.3 Access Methods

Access method is the term given to the set of rules by which networks arbitrate the use of a common medium. It is the way the LAN keeps different streams of data from crashing into each other as they share the network.

Networks need access methods for the same reason streets need traffic lights-to keep people from hitting each other. Think of the access method as traffic law. The network cable is the street. Traffic law (or the access method) regulates the use of the street (or cable), determining who can drive (or send data) where and at what time. On a network, if two or more people try to send data at exactly the same time, their signals will interfere with each other, ruining the data being transmitted. The access method prevents this.

The access method works at the data-link layer (layer 2) because it is concerned with the use of the medium that connects users. The access method doesn't care what is being sent over the network, just like the traffic law doesn't stipulate what you can carry. It just says you have to drive on the right side of the road and obey the traffic lights and signs.

Three traditional access methods are used today, although others exist and may become increasingly important. They are Ethernet, Token Ring, and ARCnet. Actually, these technologies encompass wider-ranging standards than their access methods. They also define other features of network transmission, such as the electrical characteristics of signals, and the size of

data packets sent. Nevertheless, these standards are best known by the access methods they employ these in accessing channels.

3.4 Multiple Access Protocols

Many algorithms for allocating a multiple access channel are known, we will describe a small sample of the more interesting ones and give some examples of their use.

ALOHA

There are two versions of ALOHA here: pure and slotted. They differ with respect to whether time is divided into discrete slots into which all frames must fit. Pure ALOHA does not require global time synchronization; slotted ALOHA does.

Pure ALOHA

The basic idea of an ALOHA system is simple: let users transmit whenever they have data to be sent. There will be collisions, of course, and the colliding frames will be damaged. However, due to the feedback property of broadcasting, a sender can always find out whether its frame was destroyed by listening to the channel, the same way other users do. If the frame was destroyed, the sender just waits a random amount of time and sends it again. The waiting time must be random or the same frames will collide over and over, in lockstep. Systems in which multiple users share a common channel in a way that can lead to conflicts are widely known as contention systems.

A sketch of frame generation in an ALOHA system is given in Figure 3.1 given below. We have made the frames all the same length because the throughput

of ALOHA systems is maximized by having a uniform frame size rather than by allowing variable length frames.

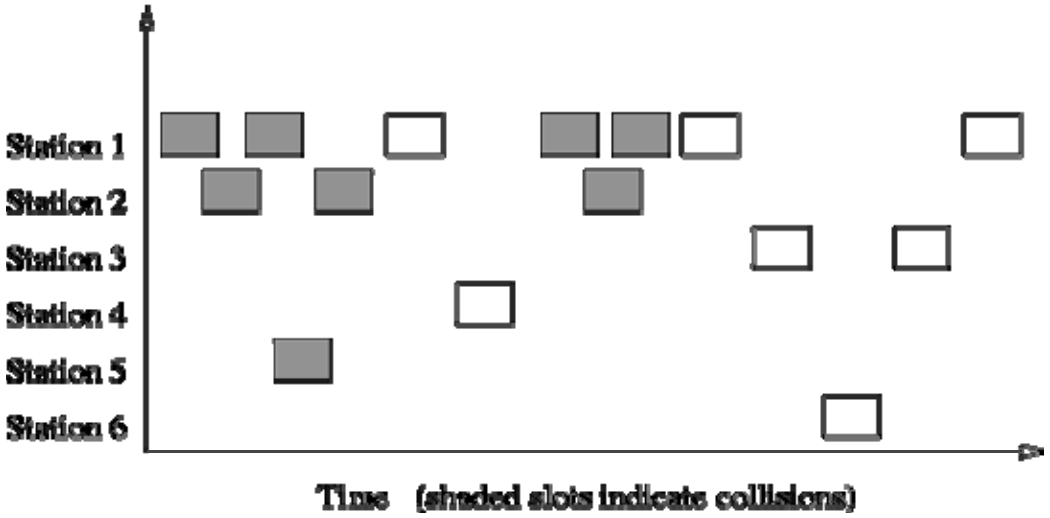


Figure 3.1. In pure ALOHA, frames are transmitted at completely arbitrary times.

Whenever two frames try to occupy the channel at the same time, there will be a collision and both will be garbled. If the first bit of a new frame overlaps with just the last bit of a frame almost finished, both frames will be totally destroyed and both will have to be retransmitted later. The checksum cannot (and should not) distinguish between a total loss and a near miss. Bad is bad.

The station then transmits a frame containing the line and checks the channel to see if it was successful. If so, the user sees the reply and goes back to typing. If not, the user continues to wait and the frame is retransmitted over and over until it has been successfully sent. Let the "frame time" denote the amount of time needed to transmit the standard, fixed-length frame. At this point we assume that the infinite population of users generates new frames according to a Poisson distribution with mean N frames per frame time. If N >

1, the user community is generating frames at a higher rate than the channel can handle, and nearly every frame will suffer a collision. For reasonable throughput we would expect 0 < N < 1. In addition to the new frames, the stations also generate retransmissions of frames that previously suffered collisions. Let us further assume that the probability of k transmission attempts per frame time, old and new combined, is also Poisson, with mean G per frame time. Clearly, G >=N. At low load (i.e., N =0), there will be few collisions, hence few retransmissions, so G >N. At high load there will be many collisions, so G > N. Under all loads, the throughput, S, is just the offered load, G, times the probability, P_0, of a transmission succeeding that is, $S = GP_0$, where P_0 is the probability that a frame does not suffer a collision. The probability that k frames are generated during a given frame the Poisson distribution gives time: Equation (ii)

$$Pr[k] = G^k e^{-G}/k! \qquad \text{----------(iii)}$$

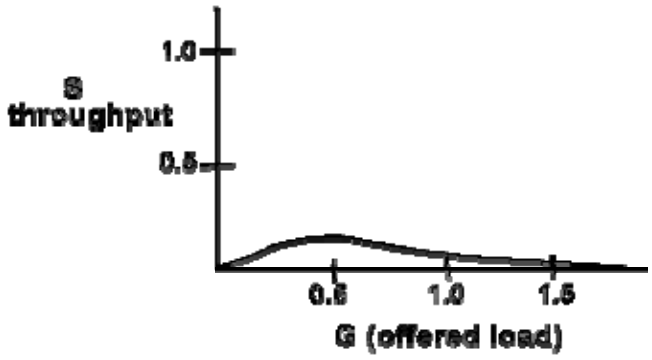


Figure 3.2. Performance in case of pure ALOHA

so the probability of zero frames is just e^{-G}. In an interval two frame times long, the mean number of frames generated is 2G. The probability of no other traffic being initiated during the entire vulnerable period is thus given by $P_0 = e^{-2G}$. Using $S = GP_0$, we get

$$S=G e^{-2G} \qquad \text{----------(iv)}$$

The maximum throughput occurs at G = 0.5, with S = 1/2e, which is about 0.184 as shown in figure 3.2. In other words, the best we can hope for is a channel utilization of 18 percent. This result is not very encouraging, but with everyone transmitting at will, we could hardly have expected a 100 percent success rate.

Advantages:

Superior to fixed assignment when there are large number of bursty stations.

Adapts to varying number of stations.

Disadvantages:

Theoretically proven throughput maximum of 18.4%.

Requires queuing buffers for retransmission of packets.

Slotted ALOHA

This is to divide time into discrete intervals, each interval corresponding to one frame. This approach requires the users to agree on slot boundaries as shown in figure 3.3. One way to achieve synchronization would be to have one special station emit a pip at the start of each interval, like a clock. It is required to wait for the beginning of the next slot. Thus, the continuous pure ALOHA is turned into a discrete one. Since the vulnerable period is now halved, the probability of no other traffic during the same slot as our test frame is e^{-G} which leads to Equation (iv) $S=G\, e^{-2G}$

The slotted ALOHA peaks at G = 1, with a throughput of S =1/e or about 0.368, twice that of pure ALOHA as shown in figure 3.4. If the system is operating at G = 1, the probability of an empty slot is 0.368. The best we can hope for using slotted ALOHA is 37 percent of the slots empty, 37 percent successes, and 26 percent collisions. Operating at higher values of G reduces

the number of empties but increases the number of collisions exponentially. To see how this rapid growth of collisions with G comes about, consider the transmission of a test frame.

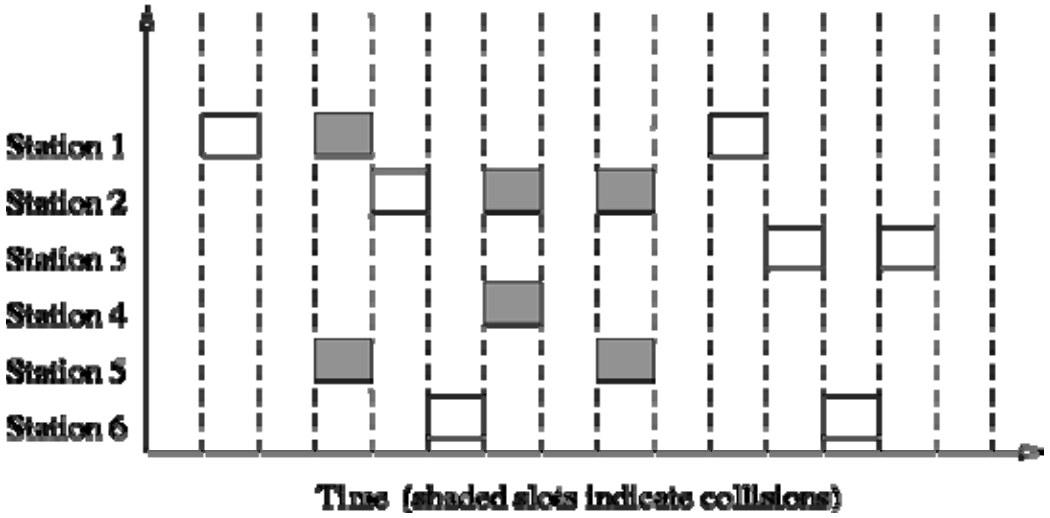


Figure 3.3. In slotted ALOHA, frames are transmitted at discrete intervals of times

The probability that it will avoid a collision is e^{-G}, the probability that all the other users are silent in that slot. The probability of a collision is then just $1 - e^{-G}$. The probability of a transmission requiring exactly k attempts, is

$$P_k = e^{-G}(1 - e^{-G})^{k-1}$$

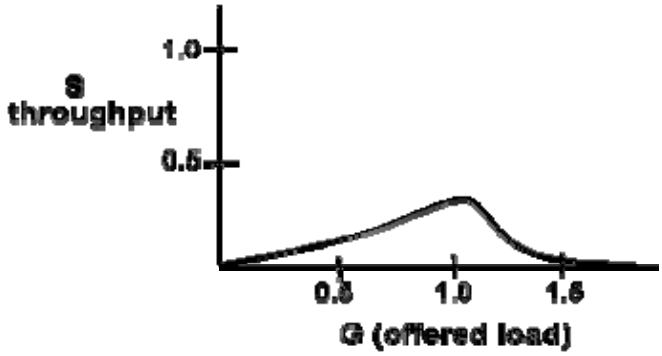


Figure 3.4 Performance in case of slotted ALOHA

Advantages:

Doubles the efficiency of Aloha.

Adaptable to a changing station population.

Disadvantages:

Theoretically proven throughput maximum of 36.8%.

Requires queuing buffers for retransmission of packets.

Synchronization required.

Synchronous system: time divided into slots

Slot size equals fixed packet transmission time

When Packet ready for transmission, wait until start of next slot

Packets overlap completely or not at all

Carrier Sense Multiple Access Protocols

Protocols in which stations listen for a carrier (i.e., a transmission) and act accordingly are called carrier sense protocols. With slotted ALOHA the best channel utilization that can be achieved is 1/e. This is hardly surprising, since with stations transmitting at will, without paying attention to what the other stations are doing, there are bound to be many collisions. In local area networks, however, it is possible for stations to detect what other stations are doing, and adapt their behavior accordingly. These networks can achieve a much better utilization than 1/e. In this section we will discuss some protocols for improving performance.

Persistent and Non-persistent CSMA

The first carrier sense protocol that we will study here is called 1-persistent

CSMA (Carrier Sense Multiple Access). When a station has data to send, it first listens to the channel to see if anyone else is transmitting at that moment. If the channel is busy, the station waits until it becomes idle. When the station detects an idle channel, it transmits a frame. If a collision occurs, the station waits a random amount of time and starts all over again. The protocol is called 1-persistent because the station transmits with a probability of 1 when it finds the channel idle. The figure 3.5 given below shows performance comparisons of different protocols.

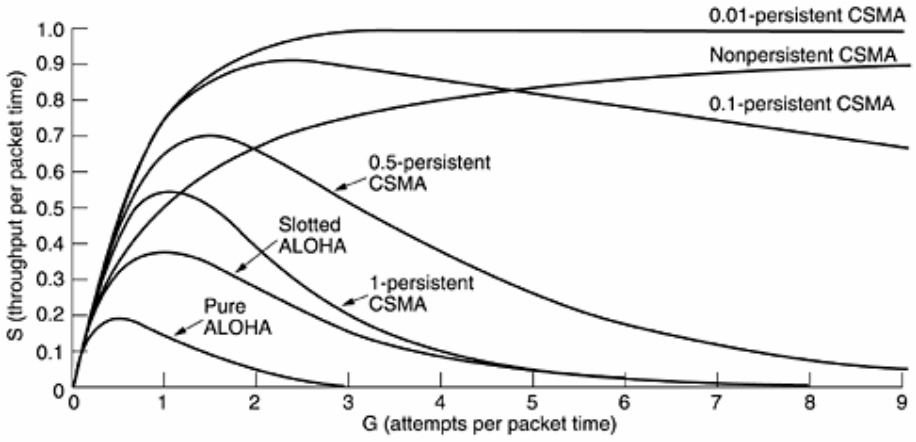


Figure 3.5 Performance of different Protocols

The propagation delay has an important effect on the performance of the protocol. There is a small chance that just after a station begins sending, another station will become second one, the latter will sense an idle channel and will also begin sending, resulting in a collision. The longer the propagation delay, the more important this effect becomes, and the worse the performance of the protocol.

Even if the propagation delay is zero, still there will be collisions. If two stations become ready in the middle of a third station's transmission, both will wait politely until the transmission ends and then both will begin transmitting exactly simultaneously, resulting in a collision. If they were not so impatient,

there would be fewer collisions. Even so, this protocol is far better than pure ALOHA because both stations have the decency to desist from interfering with the third station's frame. Intuitively, this approach will lead to a higher performance than pure ALOHA. Exactly the same holds for slotted ALOHA.

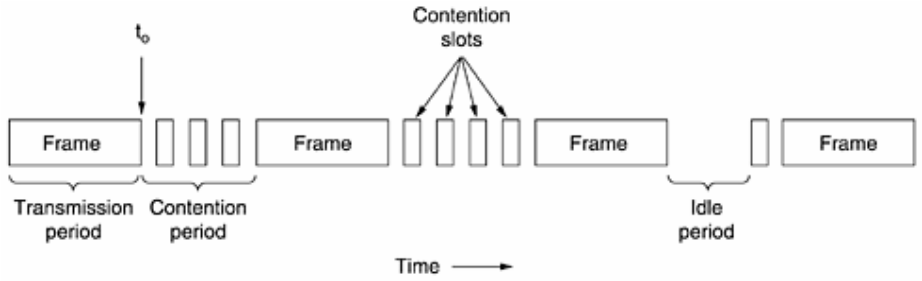
A second carrier sense protocol is non-persistent CSMA. In this protocol, a conscious attempt is made to be less greedy than in the previous one. Before sending, a station senses the channel. If no one else is sending, the station begins doing so itself. However, if the channel is already in use, the station does not continually sense it for the purpose of seizing it immediately upon detecting the end of the previous transmission. Instead, it waits a random period of time and then repeats the algorithm. Consequently, this algorithm leads to better channel utilization but longer delays than 1-persistent CSMA.

The last protocol is p-persistent CSMA. It applies to slotted channels and works as follows. When a station becomes ready to send, it senses the channel. If it is idle, it transmits with a probability p. With a probability q = 1 - p, it defers until the next slot. If that slot is also idle, it either transmits or defers again, with probabilities p and q. This process is repeated until either the frame has been transmitted or another station has begun transmitting. In the latter case, the unlucky station acts as if there had been a collision (i.e., it waits a random time and starts again). If the station initially senses the channel busy, it waits until the next slot and applies the above algorithm.

CSMA with Collision Detection

Persistent and non-persistent CSMA protocols are clearly an improvement over ALOHA because they ensure that no station begins to transmit when it senses the channel busy. Another improvement is for stations to abort their transmissions as soon as they detect a collision. In other words, if two stations sense the channel to be idle and begin transmitting simultaneously, they

detect the collision almost immediately. Rather than finish transmitting their frames, which are irretrievably garbled anyway, they should abruptly stop transmitting as soon as the collision is detected. Quickly terminating damaged



frames saves time and bandwidth. This protocol, known as CSMA/CD (CSMA with Collision Detection) is widely used on LANs in the MAC sub-layer. At the point marked t_0, a station has finished transmitting its frame. Any other station having a frame to send may now attempt to do so. If two or more stations decide to transmit simultaneously, there will be a collision. Collisions can be detected by looking at the power or pulse width of the received signal and comparing it to the transmitted signal.

Figure 3.6 Different situations in CSMA with Collision Detection

After a station detects a collision, it aborts its transmission, waits for a random period of time, and then tries again, assuming that no other station has started transmitting in the meantime. Therefore, our model for CSMA/CD will consist of alternating contention and transmission periods, with idle periods occurring when all stations are quiet (e.g., for lack of work), as shown in figure 3.6 also.

Now let us look closely at the details of the contention algorithm. Suppose that two stations both begin transmitting at exactly time t_0. How long will it take them to realize that there has been a collision? The answer to this question is vital to determining the length of the contention period and hence what the delay and throughput will be. The minimum time to detect the collision is then just the time it takes the signal to propagate from one station to the other. It

worth's noting that a sending station must continually monitor the channel, listening for noise bursts that might indicate a collision. For this reason, CSMA/CD with a single channel is inherently a half-duplex system. It is impossible for a station to transmit and receive frames at the same time because the receiving logic is in use, looking for collisions during every transmission. To avoid any misunderstanding, it is worth noting that no MAC-sublayer protocol guarantees reliable delivery. Even in the absence of collisions, the receiver may not have copied the frame correctly for various reasons.

Collision-Free Protocols

Although collisions do not occur with CSMA/CD once a station has unambiguously captured the channel, they can still occur during the contention period. These collisions adversely affect the system performance, especially when the cable is long and the frames are short. And CSMA/CD is not universally applicable. In this section, we will examine some protocols that resolve the contention for the channel without any collisions at all, not even during the contention period. Most of these are not currently used in major systems, but in a rapidly changing field, having some protocols with excellent properties available for future systems is often a good thing. In the protocols to be described, we assume that there are exactly N stations, each with a unique address from 0 to N - 1 "wired" into it. It does not matter that some stations may be inactive part of the time. We also assume that propagation delay is negligible.

A Bit-Map Protocol

In this collision-free protocol, the basic bit-map method, each contention period consists of exactly N slots. If station 0 has a frame to send, it transmits

a 1 bit during the zeroth slot. No other station is allowed to transmit during this slot. Regardless of what station 0 does, station 1 gets the opportunity to transmit a 1 during slot 1, but only if it has a frame queued. In general, station j may announce that it has a frame to send by inserting a 1 bit into slot j. After all N slots have passed by, each station has complete knowledge of which stations wish to transmit. At that point, they begin transmitting in numerical order as shown in figure 3.7.

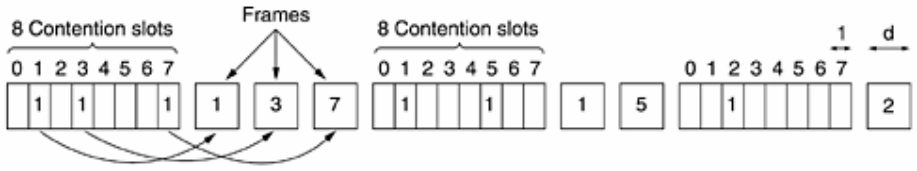


Figure 3.7 Transmission of frames in Bit-Map Protocol

Since everyone agrees on who goes next, there will never be any collisions. After the last ready station has transmitted its frame, an event all stations can easily monitor, another N bit contention period is begun. If a station becomes ready just after its bit slot has passed by, it is out of luck and must remain silent until every station has had a chance and the bit map has come around again. Protocols like this in which the desire to transmit is broadcast before the actual transmission are called reservation protocols.

Binary Countdown

A problem with the basic bit-map protocol is that the overhead is 1 bit per station, so it does not scale well to networks with thousands of stations. We can do better than that by using binary station addresses. A station wanting to use the channel now broadcasts its address as a binary bit string, starting with the high-order bit. All addresses are assumed to be the same length. The bits in each address position from different stations are BOOLEAN ORed together. We will call this protocol binary countdown. It implicitly assumes that the transmission delays are negligible so that all stations see asserted bits essentially instantaneously. To avoid conflicts, an arbitration rule must be applied: as soon as a station sees that a high-order bit position that is 0 in its address has been overwritten with a 1, it gives up as shown in figure 3.8.

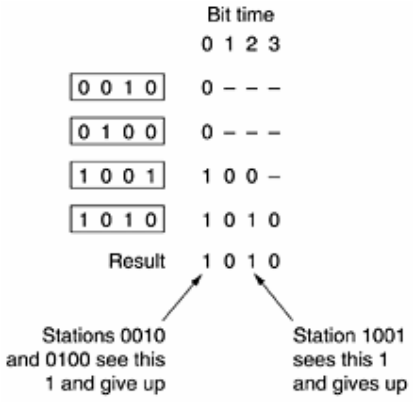
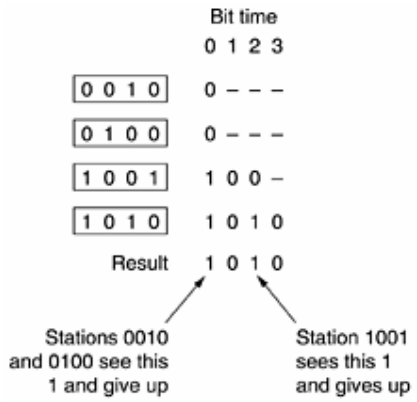


Figure 3.8 Station give ups in Binary Countdown



3.5 Summary

Some networks have a single channel that is used for all communication. In these networks, the key design issue is the allocation of this channel among the competing stations wishing to use it. Numerous channel allocation algorithms have been devised. The simplest allocation schemes are FDM and TDM. These are efficient when the number of stations is small and fixed and the traffic is continuous. Both are widely used under these circumstances, for example, for dividing up the bandwidth on telephone trunks.

When the number of stations is large and variable or the traffic is fairly bursty, FDM and TDM are poor choices. The ALOHA protocol, with and without slotting, has been proposed as an alternative. ALOHA and its many variants and derivatives have been widely discussed, analyzed. When the state of the channel can be sensed, stations can avoid starting a transmission while another station is transmitting. This technique, carrier sensing, has led to a variety of protocols that can be used on LANs and MANs.

Method	Description
FDM	Dedicate a frequency band to each station
WDM	A dynamic FDM scheme for fiber
TDM	Dedicate a time slot to each station
Pure ALOHA	Unsynchronized transmission at any instant
Slotted ALOHA	Random transmission in well-defined time slots
1-persistent CSMA	Standard carrier sense multiple access
Nonpersistent CSMA	Random delay when channel is sensed busy
P-persistent CSMA	CSMA, but with a probability of p of persisting
CSMA/CD	CSMA, but abort on detecting a collision
Bit map	Round-robin scheduling using a bit map
Binary countdown	Highest-numbered ready station goes next

Table 3.9 Descriptions of MAC Layer Protocols

A class of protocols that eliminates contention altogether, or at least reduce it considerably, is well known. Binary countdown completely eliminates contention. The tree walk protocol reduces it by dynamically dividing the stations into two disjoint groups, one of which is permitted to transmit and one of which is not. It tries to make the division in such a way that only one station that is ready to send is permitted to do so.

A brief description of MAC Layer Protocols is given in table 3.1, which shows the basic characteristics of different protocols.

3.6 Self Assessment Questions

1. Consider the delay of pure ALOHA versus slotted ALOHA at low load. Which one is less? Explain your answer.

2. A large population of ALOHA users manage to generate 50 requests/sec, including both originals and retransmissions. Time is slotted in units of 40 msec.

(a) What is the chance of success on the first attempt?

(b) What is the probability of exactly k collisions and then a success?

(c) What is the expected number of transmission attempts needed?

3. Measurements of a slotted ALOHA channel with an infinite number of users show that 10 percent of the slots are idle.
(a) What is the channel load, G?
(b) What is the throughput?
(c) Is the channel underloaded or overloaded?

4. In an infinite-population slotted ALOHA system, the mean number of slots a station waits between a collision and its retransmission is 4. Plot the delay versus throughput curve for this system.

5. Sixteen stations, numbered 1 through 16, are contending for the use of a shared channel by using the adaptive tree walk protocol. If all the stations whose addresses are prime numbers suddenly become ready at once, how many bit slots are needed to resolve the contention?

3.8 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
- Computer Network, ED Tittle, Tata MacGraw Hills Publications

Subject: Computer Networks **Subject Code:** MCA-301

Lesson No.: 4

Author: Mr. Yudhvir Singh

Lesson Title: Local Area Network **Vetter:** **Dr. Sandeep Arya**

Structure

4.0 Objective

In this chapter we will study the Local Area Network (LAN). This lesson includes uses and attributes of LAN. This chapter also with the IEEE LAN standards: 802.3, 802.4, 802.5, 802.6 (MAN), FDDI. The frame format of different standards with all header are described here.

4.1 Introduction

One of the limiting factors of personal computers is the primary use of such devices as isolated workstations. This means that in most organizations it is difficult for personal computer users to share data and the use of peripheral devices, since such sharing normally requires physical activity to occur. Both the direct economical benefits of sharing data and the use of peripheral devices as well as productivity gains resulting from the integration of personal computers into a common network have contributed to an increasing demand for local area network. Local area networks, generally called LANs, are privately owned networks within a single building or campus of up to a few kilometers in size. They are widely used to connect personal computers and workstations in company offices and factories to share resources and exchange information.

4.2 Uses of LAN

A local area network is a cable that provides an electronic highway for the transportation of information to and from different devices connected to the network. Because a LAN provides the capability to route data between devices connected to a common network within a relatively limited distance, numerous benefits can accrue to users of the network.

Peripheral sharing: Peripheral sharing allows network users to access color laser printers, CD-ROM jukebox systems, and other devices. Users of a LAN can obtain access to resources that would be too expensive to justify for each individual workstation user.

Common software access: The ability to access data files and programs from multiple workstations can substantially reduce the cost of software and shared access to database information allows network users to obtain access to updated files on a real-time basis.

Electronic mail: One popular type of application program used on LANs enables users to transfer messages electronically. E-mail application program can be used to supplement and, in many cases, eliminate the need for paper.

Gateway access to mainframes: For organizations with mainframe or minicomputers, a local area network gateway can provide a common method of access to those computers. Without the use of a LAN gateway, each personal computer requiring access to a mainframe or minicomputer would require a separate method of access. This might increase both the complexity and the cost of providing access.

Internet Access: Instead of supporting individual dial access to the Internet you could install a router and leased line to an Internet Service Provider (ISP) that is a fraction of the cost associated with a large number of individual Internet dial network accounts.

Virtual private network operations: A virtual private network (VPN) represents a network created via the routing of data between two or more locations over a public packet network. If an organization has two

or more geographically separated LANs it may be less expensive to connect them to the Internet and use the Internet as a VPN than to interconnect the LANs via a leased line. However, since the Internet is a public network you would also need to consider the cost of security equipment, such as a firewall, authentication server, and encryption equipment to secure communications between organizational locations via the Internet.

4.3 Attributes of LAN

Although a local area network is a limited distance transmission system the variety of options available for constructing such networks is anything but limited. Many of the options available or the construction of local area networks are based on the technological characteristics that govern their operation. These characteristics include different size, transmission technology, topologies, signaling methods, transmission media, access methods used to transmit data on the network, and the hardware and software required to make the network operate.

Topology

The topology of a local area network is the structure or geometric layout of the cable used to connect stations on the network. Unlike conventional data communications networks, which can be configured in a variety of ways with the addition of hardware and software, most local area networks are designed to operate based upon the interconnection of stations that follow a specific topology. The most

common topologies used in LANs include the loop, bus, ring, star and tree.

Signaling methods

The signaling method used by a local area network refers to both the way data are encoded for transmission and the frequency spectrum of the media. To a large degree, the signaling method is related to the use of the frequency spectrum of the media.

Two signaling methods used by LANs are broadband and base-band. In broadband signaling, the bandwidth of the transmission medium is subdivided by frequency to form two or more sub-channels, with each sub-channel permitting data transfer to occur independently of data transfer on another sub-channel. In base-band signaling only one signal is transmitted on the medium at any point in time. Broadband is more complex than base-band, because it requires information to be transmitted via the modulation of a carrier signal, thus requiring the use of special types of modems.

Transmission medium

The transmission medium used in a local area network can range in scope from twisted-pair wire, such as is used in conventional telephone lines, to coaxial cable, fiber optic cable, and electromagnetic waves such as those used by FM radio and infrared.

Each transmission medium has a number of advantages and disadvantages. The primary differences between media are their cost and ease of installation; the bandwidth of the cable, which may or may not permit several transmission sessions to occur simultaneously; the

maximum speed of communications permitted; and the geographical scope of the network that the medium supports.

Access methods

If the topology of a local area network can be compared to a data highway, then the access method might be viewed as the set of rules that enable data from one workstation to successfully reach the destination via the data highway. Without such rules, it is quite possible for two messages sent by two different workstations to collide, with the result that neither message reaches its destination.

Two common access methods primarily employed in local area networks are carrier-sense multiple access/collision detection (CSMA/CD) and token passing. Each of these access methods is uniquely structured to address the previously mentioned collision and data destination problems.

4.3 IEEE LAN STANDARDS

IEEE has produced several standards for LANS, these standards collectively known as IEEE 802, including CSMA/CD, Token Bus and Token Ring. These standards have been adopted by ANSI and by ISO. IEEE LAN standards are described in Table 4.1.

IEEE LAN Standards	Description
802.1	High Level Interface
802.2	Logical Link Control
802.3	Ethernet CSMA/CD

802.4	Token-Passing Bus
802.5	Token-Passing Ring
802.6	Metropolitan Area Networks
802.7	Broadband Technical Advisory Group
802.8	Fiber Optic Technical Advisory Group
802.9	Integrated Voice and Data Networks
802.10	Network Security
802.11	Wireless LANs
802.12	100VG-AnyLAN

Table 4.1: Different LAN Standards with Descriptions

4.5 IEEE LAN Standard 802.3

IEEE has produced 802.3 standards for Ethernet CSMA/CD Local Area Networks.

Ethernet Network Elements

Ethernet LANs consist of network nodes and interconnecting media. The network nodes fall into two major classes:

Data terminal equipment (DTE)—Devices that are either the source or the destination of data frames. DTEs are typically devices such as PCs, workstations, file servers, or print servers that, as a group, are all often referred to as end stations.

Data communication equipment (DCE)—Intermediate network devices that receive and forward frames across the network. DCEs may be either

standalone devices such as repeaters, network switches, and routers, or communications interface units such as interface cards and modems.

Ethernet Network Topologies and Structures

LANs take on many topological configurations, but regardless of their size or complexity, all will be a combination of only three basic interconnection structures or network building blocks. The simplest structure is the point-to-point interconnection. Only two network units are involved, and the connection may be DTE-to-DTE, DTE-to-DCE, or DCE-to-DCE. The cable in point-to-point interconnections is known as a network link.

The IEEE 802.3 Logical Relationship to the ISO Reference Model

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

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

Logical Link Control (LLC)- This sub-layer provides the interface between the Ethernet MAC and the upper layers in the protocol stack of the end station. The LLC sublayer is defined by IEEE 802.2 standards.

Bridge entity- Bridge entities provide LAN-to-LAN interfaces between LANs that use the same protocol and also between different protocols, Bridge entities are defined by IEEE 802.1 standards.

The Ethernet MAC Sub-layer

The MAC sub-layer has two primary responsibilities: Data encapsulation, including frame assembly before transmission, and frame checking/parsing - error detection during and after reception. Media access control, including initiation of frame transmission and recovery from transmission failure.

The Basic Ethernet Frame Format

The IEEE 802.3 standard defines a basic data frame format that is required for all MAC implementations, plus several additional optional formats that are used to extend the protocol's basic capability shown in figure 4.1.

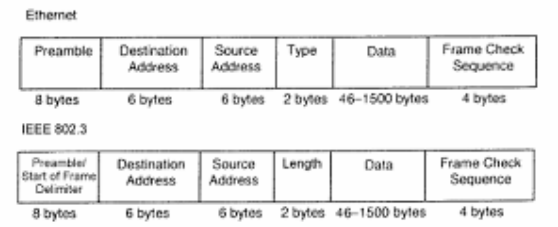


Figure 4.1 Ethernet and IEEE 802.3 frame format

Preamble (PRE)—Consists of 7 bytes. The PRE is an alternating pattern of ones and zeros that tells receiving stations that a frame is coming, and that provides a means to synchronize the frame-reception portions of receiving physical layers with the incoming bit stream.

Start-of-frame delimiter (SOF)—Consists of 1 byte. The SOF is an alternating pattern of ones and zeros, ending with two consecutive 1-bits indicating that the next bit is the left-most bit in the left-most byte of the destination address.

Destination address (DA)—Consists of 6 bytes. The DA field identifies which station(s) should receive the frame. The left-most bit in the DA field indicates whether the address is an individual address (indicated by a 0) or a group address (indicated by a 1). The second bit from the left indicates whether the DA is globally administered (indicated by a 0) or locally administered (indicated by a 1). The remaining 46 bits are a uniquely assigned value that identifies a single station, a defined group of stations, or all stations on the network.

Source addresses (SA)—Consists of 6 bytes. The SA field identifies the sending station. The SA is always an individual address and the left-most bit in the SA field is always 0.

Length/Type—Consists of 2 bytes. This field indicates either the number of MAC-client data bytes that are contained in the data field of the frame, or the frame type ID if the frame is assembled using an optional format. If the Length/Type field value is less than or equal to 1500, the number of LLC bytes in the Data field is equal to the Length/Type field value. If the Length/Type field value is greater than 1536, the frame is an optional type frame, and the Length/Type field value identifies the particular type of frame being sent or received.

Data—Is a sequence of *n* bytes of any value, where *n* is less than or equal to 1500. If the length of the Data field is less than 46, the Data field must be extended by adding a filler (a pad) sufficient to bring the Data field length to 46 bytes.

Frame check sequence (FCS)—Consists of 4 bytes. This sequence contains a 32-bit cyclic redundancy check (CRC) value, which is created by the sending MAC and is recalculated by the receiving MAC to check for damaged frames. The FCS is generated over the DA, SA, Length/Type, and Data fields.

The CSMA/CD Access Method

The CSMA/CD protocol was originally developed as a means by which two or more stations could share a common media in a switch-less environment when the protocol does not require central arbitration, access tokens, or assigned time slots to indicate when a station will be allowed to transmit. Each Ethernet MAC determines for itself when it will be allowed to send a frame.

Carrier sense—Each station continuously listens for traffic on the medium to determine when gaps between frame transmissions occur.

Multiple access—Stations may begin transmitting any time they detect that the network is quiet (there is no traffic).

Collision detect—If two or more stations in the same CSMA/CD network (collision domain) begin transmitting at approximately the same time, the bit streams from the transmitting stations will interfere (collide) with each other, and both transmissions will be unreadable. If that happens, each transmitting station must be capable of detecting that a collision has occurred before it has finished sending its frame.

Each must stop transmitting as soon as it has detected the collision and then must wait a quasi-random length of time (determined by a back-off algorithm) before attempting to retransmit the frame.

4.6 IEEE LAN Standard 802.4

IEEE has produced 802.4 standards for Token Bus Local Area Networks. IEEE 802.4 Token bus uses a broadband coaxial cable with analog transmission as shown in figure 4.2. Stations physically on a bus, but logically in a ring, with left and right neighbors. The stations on the bus or tree form a logical ring. It determines the logical ring of the physical bus by the numeric value of the address. Each station knows the identity of the stations preceding and following it. The TOKEN regulates the access rights (transmissions), the station received TOKEN can transmission time slot is granted, after transmission, pass the token to the next DTE. When a token is passed to a successor, the sender waits for evidence that it has been transmitted properly (i.e. the successor transmits the next frame). Otherwise it may establish a new successor (the next address to the problematic station).

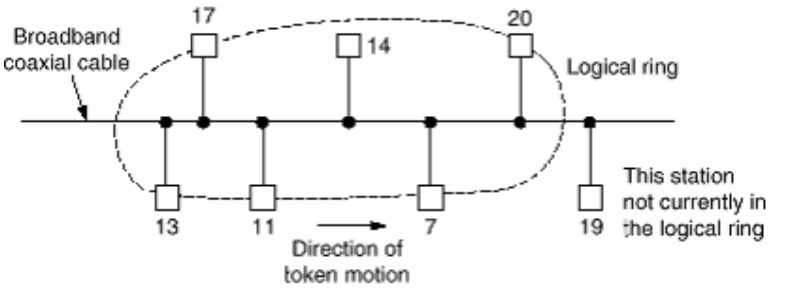


Figure 4.2: Token Bus

The access control byte contains a bit called the "token" bit, which is set to 0 if this frame is the token. When a station wants to send, it seizes the token and changes that bit to a 1. A station wishing to transmit waits until it detects a free token passing by.

The operations are similar to token ring except more flexible for logical ring definition (can put in essentially any order) can recover single-node errors better cover phase-coherent, frequency shift keying - uses two frequencies, no phase mismatch. It can avoid much noise by filtering everything but these two frequencies similar to token ring mechanism.

It has a complex maintenance scheme, which have to required: Ring initialization, Add/ delete station algorithm, Fault management, Based on predecessor and successor scheme, Recovery and Priority. Token Bus has major functions in MAC are Interface, Access Control, Receive, Transmit, and RRM Regenerative Repeater machine.

Under light loads there is some inefficiency. Under heavy loads, the ring functions in a round-robin fashion. Provides a regulated access. Must provide for token maintenance.

Token Bus has a special frame called the "token". When you have the token, you can transmit; when you don't have it, you can't. The token format for token bus is given in figure 4.3.

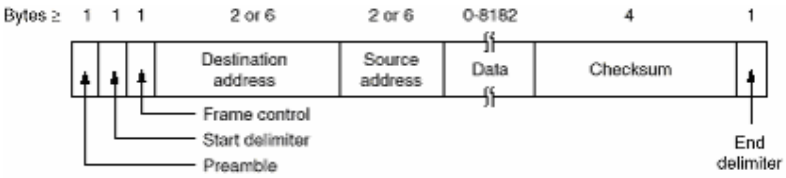


Figure 4.3: Frame Format IEEE 802.4

Token Frame Fields

The three token frame fields illustrated in Figure are summarized in the descriptions that follow:

- **Preamble (PRE)**—The PRE is an alternating pattern of ones and zeros that tells receiving stations that a frame is coming, and that provides a means to synchronize the frame-reception portions of receiving physical layers with the incoming bit stream
- **Start delimiter**—Alerts each station of the arrival of a token (or data/command frame). This field includes signals that distinguish the byte from the rest of the frame by violating the encoding scheme used elsewhere in the frame.
- **Frame-control bytes**—Indicates whether the frame contains data or control information. In control frames, this byte specifies the type of control information.

- **Destination and source addresses**—Consists of two 6-byte address fields that identify the destination and source station addresses.
- **Data**—Indicates that the length of field is limited by the ring token holding time, which defines the maximum time a station can hold the token and data limit is 0 to 8182 bytes.
- **Frame-Checksum (FCS)**—Is filed by the source station with a calculated value dependent on the frame contents. The destination station recalculates the value to determine whether the frame was damaged in transit. If so, the frame is discarded.
- **End Delimiter**—Signals the end of the token or data/command frame. The end delimiter also contains bits to indicate a damaged frame and identify the frame that is the last in a logical sequence.

As with 802.4, the Start and End delimiter fields contain signals that are "illegal"; that is, they are neither 0 nor 1. A control frame known as a TOKEN regulates right of access. The station in control of the token may transmit data frames, poll stations and receive responses, When finished the station passes the token to the next station in logical sequence. Non-token using stations are allowed onto the bus in response to polls or selections.

Frame-control field indicates whether the frame contains data or control information. In control frames, this byte specifies the type of control information. The major control token are given in table 4.2.

Frame control field	Name	Meaning
00000000	Claim_token	Claim token during ring initialization
00000001	Solicit_successor_1	Allow stations to enter the ring
00000010	Solicit_successor_2	Allow stations to enter the ring
00000011	Who_follows	Recover from lost token
00000100	Resolve_contention	Used when multiple stations want to enter
00001000	Token	Pass the token
00001100	Set_successor	Allow station to leave the ring

Table 4.2: Token Bus Control Frame

Token Bus has following pros and cons.

– pros: reliable equipment, more deterministic, priorities

– cons: complex protocols, hard to implement in fiber, not popular

4.6 IEEE LAN Standard 802.5

IEEE 802.5 networks are basically compatible with Token Ring, although the specifications differ in minor ways. Token Ring network specifies a star, with all end stations attached to a device called a multi-station access unit (MSAU). In contrast, IEEE 802.5 does not specify a topology, although virtually all IEEE 802.5 implementations are based on a star. Other differences exist, including media type (IEEE 802.5 does not specify a media type, although Token Ring networks use twisted-pair wire) and routing information field size. Table 4.3 summarizes Token Ring network and IEEE 802.5 specifications. Although Dissimilar in Some Respects, Token Ring Network and IEEE 802.5 Are Generally Compatible

	IBM Token Ring network	IEEE 802.5
Data rates	4.16 Mbps	4.16 Mbps
Stations/segment	260 (shielded twisted pair) 72 (unshielded twisted pair)	250
Topology	Star	Not specified
Media	Twisted pair	Not specified
Signaling	Baseband	Baseband
Access method	Token passing	Token passing
Encoding	Differential manchester	Differential manchester

Table 4.3 Token Ring and IEEE 802.5 Specifications

Physical Connections

Token Ring network stations are directly connected to MSAUs, which can be wired together to form one large ring. Patch cables connect MSAUs to adjacent MSAUs, while lobe cables connect MSAUs to stations. MSAUs include bypass relays for removing stations from the ring.

Token Ring Operation

Token Ring and IEEE 802.5 are two principal examples of token-passing networks. *Token-passing networks* move a small frame, called a token, around the network. Possession of the token grants the right to transmit. If a node receiving the token has no information to send, it passes the token to the next end station. Each station can hold the token for a maximum period of allowed time.

If a station possessing the token does have information to transmit, it seizes the token, alters 1 bit of the token (which turns the token into a start-of-frame sequence), appends the information that it wants to transmit, and sends this information to the next station on the ring. While the information frame is circling the ring, no token is on the network (unless the ring supports early token release), which means that other stations wanting to transmit must wait. Therefore, collisions cannot occur in Token Ring networks. If early token release is supported, a new token can be released when frame transmission is complete. The information frame circulates the ring until it reaches the intended destination station, which copies the information for further processing. The information frame continues to circle the ring and is finally removed when it reaches the sending station. The sending station can check the returning frame to see whether the frame was seen and subsequently copied by the destination. Unlike CSMA/CD networks (such as Ethernet), token-passing networks are *deterministic*, which means that it is possible to

calculate the maximum time that will pass before any end station will be capable of transmitting. This feature and several reliability features make Token Ring networks ideal for applications in which delay must be predictable and robust network operation is important.

Priority System

Token Ring networks use a sophisticated priority system that permits certain user-designated, high-priority stations to use the network more frequently. Token Ring frames have two fields that control priority: the priority field and the reservation field. Only stations with a priority equal to or higher than the priority value contained in a token can seize that token. After the token is seized and changed to an information frame, only stations with a priority value higher than that of the transmitting station can reserve the token for the next pass around the network. When the next token is generated, it includes the higher priority of the reserving station. Stations that raise a token's priority level must reinstate the previous priority after their transmission is complete.

Fault-Management Mechanisms

Token Ring networks employ several mechanisms for detecting and compensating for network faults. For example, one station in the Token Ring network is selected to be the *active monitor*. This station, which potentially can be any station on the network, acts as a centralized source of timing information for other ring stations and performs a variety of ring-maintenance functions. One of these functions is the removal of continuously circulating frames from the ring. When a sending device fails, its frame may continue to circle the ring. This can prevent other stations from transmitting their own frames and essentially can lock up the network. The active monitor can detect such frames, remove them from the ring, and generate a new token.

The Token Ring network's star topology also contributes to overall network reliability. Because all information in a Token Ring network is seen by active MSAUs, these devices can be programmed to check for problems and selectively remove stations from the ring, if necessary. A Token Ring algorithm called *beaconing* detects and tries to repair certain network faults. Whenever a station detects a serious problem with the network (such as a cable break), it sends a beacon frame, which defines a failure domain. This domain includes the station reporting the failure, its nearest active upstream neighbor (NAUN), and everything in between. Beaconing initiates a process called *autoreconfiguration*, in which nodes within the failure domain automatically perform diagnostics in an attempt to reconfigure the network around the failed areas. Physically, the MSAU can accomplish this through electrical reconfiguration.

Frame Format

Token Ring and IEEE 802.5 support two basic frame types: tokens and data/command frames. Tokens are 3 bytes in length and consist of a start delimiter, an access control byte, and an end delimiter. Data/command frames vary in size, depending on the size of the Information field. Data frames carry information for upper-layer protocols, while command frames contain control information and have no data for upper-layer protocols. Both formats are shown in figure 4.5.

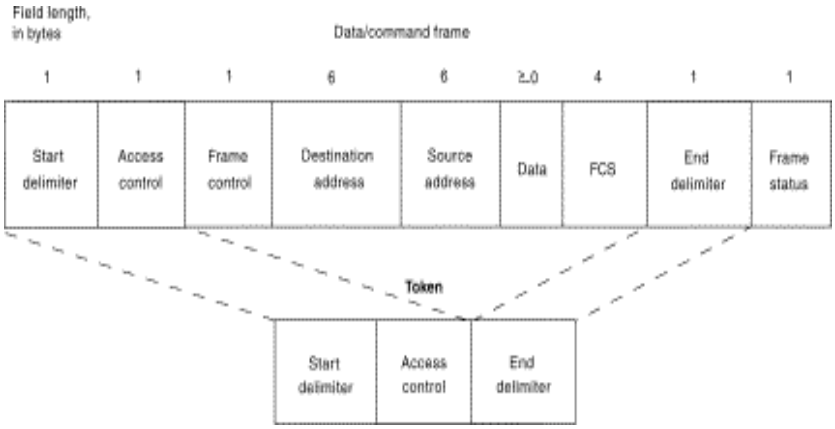


Figure 4.5 IEEE 802.5 and Token Ring Specify Tokens and Data/Command Frames

Token Frame Fields

The three token frame fields illustrated in Figure 4.3 are summarized in the descriptions that follow:

- **Start delimiter**—Alerts each station of the arrival of a token (or data/command frame). This field includes signals that distinguish the byte from the rest of the frame by violating the encoding scheme used elsewhere in the frame.
- **Access-control byte**—Contains the Priority field (the most significant 3 bits) and the Reservation field (the least significant 3 bits), as well as a token bit (used to differentiate a token from a data/command frame) and a monitor bit (used by the active monitor to determine whether a frame is circling the ring endlessly).
- **End delimiter**—Signals the end of the token or data/command frame. This field also contains bits to indicate a damaged frame and identify the frame that is the last in a logical sequence.

Data/Command Frame Fields

Data/command frames have the same three fields as Token Frames, plus several others.

The Data/command frame fields illustrated are described in the following summaries:

- **Start delimiter**—Alerts each station of the arrival of a token (or data/command frame). This field includes signals that distinguish the byte from the rest of the frame by violating the encoding scheme used elsewhere in the frame.
- **Access-control byte**—Contains the Priority field (the most significant 3 bits) and the Reservation field (the least significant 3 bits), as well as a token bit (used to differentiate a token from a data/command frame) and a monitor bit (used by the active monitor to determine whether a frame is circling the ring endlessly).
- **Frame-control bytes**—Indicates whether the frame contains data or control information. In control frames, this byte specifies the type of control information.
- **Destination and source addresses**—Consists of two 6-byte address fields that identify the destination and source station addresses.
- **Data**—Indicates that the length of field is limited by the ring token holding time, which defines the maximum time a station can hold the token.

- **Frame-check sequence (FCS)**—Is filed by the source station with a calculated value dependent on the frame contents. The destination station recalculates the value to determine whether the frame was damaged in transit. If so, the frame is discarded.
- **End Delimiter**—Signals the end of the token or data/command frame. The end delimiter also contains bits to indicate a damaged frame and identify the frame that is the last in a logical sequence.
- **Frame Status**—Is a 1-byte field terminating a command/data frame. The Frame Status field includes the address-recognized indicator and frame-copied indicator.

4.8 IEEE 802.6 (MAN)

Data Over Cable Service Interface Distributed Queue Dual Bus (DQDB) is a Data-link layer communication protocol for Metropolitan Area Networks (MANs), specified in the IEEE 802.6 standard, designed for use in MANs. DQDB is designed for data as well as voice and video transmission based on cell switching technology. DQDB, which permits multiple systems to interconnect using two unidirectional logical buses, is an open standard that is designed for compatibility with carrier transmission standards such as SMDS, which is based on the DQDB standards.

For a MAN to be effective it requires a system that can function across long distances of several miles, have a low susceptibility to error, adapt to the number of nodes attached and have variable bandwidth distribution. Using DQDB, networks can be thirty miles long and function in the range of 34 Mbps to 155 Mbps. The data rate fluctuates due to many hosts sharing a dual bus as well as the location of a single host in relation to the frame generator, but

there are schemes to compensate for this problem making DQDB function reliably and fairly for all hosts.

The DQDB is composed of a two bus lines with stations attached to both and a frame generator at the end of each bus. The buses run in parallel in such a fashion as to allow the frames generated to travel across the stations in opposite directions as described the basic DQDB architecture in figure 4.6.

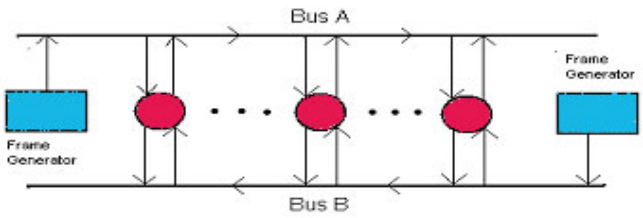


Figure 4.6: Basic DQDB architecture

DQDB cell has the similar format as the ATM defined in IEEE 802.6, as described in figure 4.7

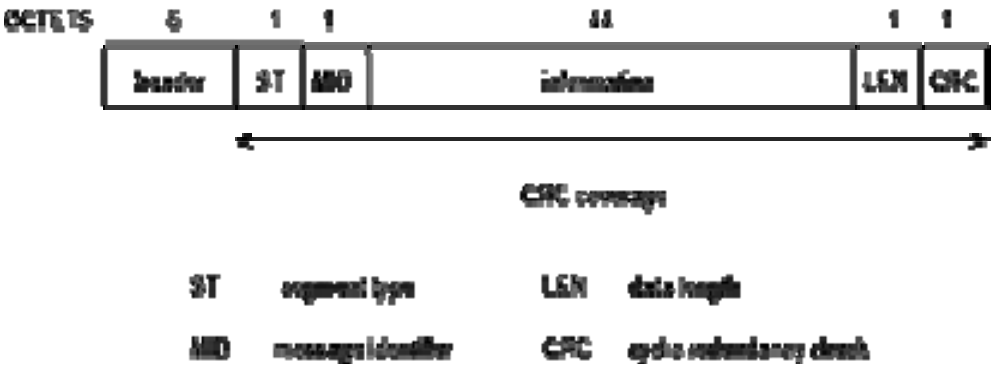


Figure 4.7: Protocol Structure - DQDB: Distributed Queue Dual Bus Defined in IEEE 802.6

DQDB cell header defined by IEEE 802.6, as shown in figure 4.8 :

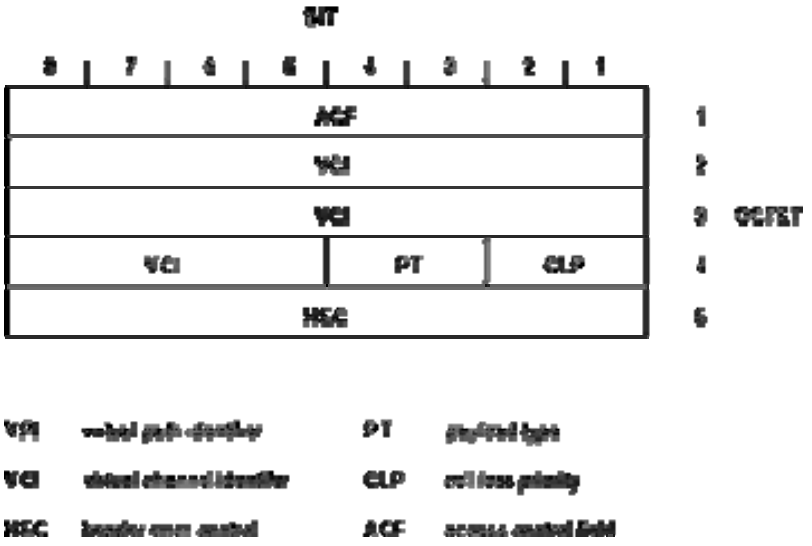


Figure 4.8: DQDB Cell Headers as Defined in IEEE 802.6

4.9 FDDI

The full form of FDDI is Fiber Distributed Data Interface. It specifies a 100 Mbps token passing dual ring LAN using fiber optical cable. FDDI is frequently used as high-speed backbone technology because of its support for high bandwidth and greater distance than copper. It should be noted that relatively recently a related copper specification called Copper Distributed Data Interface has emerged to provide 100 mbps service over copper. CDDI is the implementation of FDDI protocol over twisted pair copper wire.

FDDI uses dual ring architecture with traffic on each ring flowing in opposite direction. The dual ring data is flowing in opposite directions. The dual rings consist of a primary and a secondary ring. During normal operation the primary ring is used for data transmission and the

secondary ring remain idle. The primary purpose of the dual rings is to provide superior reliability and robustness.

FDDI Transmission Media

FDDI uses optical fiber as the primary transmission medium but it also can run over copper cabling. FDDI over copper is referred to as Copper Distributed Data Interface. Optical fiber has several advantages over copper media .In particular security reliability and performance all are enhanced with optical fiber media because fiber does not emit electrical signals therefore would permit unauthorized access to the data that is transiting the medium. IN addition fiber is immune to electrical interference radio frequency interference and (RFI) and electromagnetic interference (EMI). Fiber historically has supported much higher bandwidth than copper, although recent technological advances have made copper of transmitting at 100 Mbps. Finally FDDI allows 2 km between stations using multi mode fiber and even longer distances using a single mode.

FDDI define two types of optical fiber: single mode and multimode. A mode is a ray of light that enters the fiber at a particular angel. Multimode fiber uses LED as the light generating devices while single mode fiber is generally uses Laser.

Multimode fiber allow multiple modes of light to propagate through the fiber . Because these modes of light enter the fiber at different angles they will arrive at the end of the fiber at different times. This is known as model dispersion. The model dispersion limit the bandwidth and distances that can be accomplished using multi mode fibers, for this

reason multimode fiber is generally used for connectivity within a building or a relatively geographically contained environment.

Single mode fiber allow only one mode of light to propagate through the fiber. Because only single mode of light is used. Model dispersion is used modal dispersion is not present with in the single mode fiber. Therefore single mode fiber is capable of performing higher performance connectivity between building and within environment that are geographically dispersed.

FDDI Specifications

FDDI specifies the physical and media-access portions of the OSI reference model FDDI is a collection of four separate specification each with specific function combined these specification have the capability to provide high speed connectivity between the upper layer protocols such as TCP/IP and IPX and media such as fiber- optical cabling.

FDDI four specifications are the media access control (MAC), Physical Layer Protocol (PHY), Physical medium Dependent (PMD), station Mgt (SMT). The MAC specification define how the medium is accessed including frame token handling, addressing, algorithms for calculating cyclic redundancy check (CRC) values and error recovery mechanisms .The PHY define data encoding /decoding procedures, clocking requirements and framing among other functions. The PMD specification define the characteristics of the transmission medium, including fiber optic links, power levels bit error rates, optical components and connector. The SMT specification define the characteristics of the FDDI station configuration, ring configuration and

ring control features including station insertion and removal, initialization, fault isolation and recovery, scheduling and statistics collection.

FDDI is similar to IEEE802.3 Ethernet and IEEE802.5 Token Ring in its relationship with the OSI model. Its primary purpose is to provide connectivity between upper OSI layers of common protocols and the media used to connect network devices.

FDDI Station Attachment Types

There are four way in which FDDI device can be connected: -

- Single attachment station (SAS)
- Dual attachment station (DAS)
- Single attached concentrator (SAC) and
- Dual attached concentrator (DAC)

An SAS attaches to only one ring through a concentrator. One of the primary advantages of connecting devices with SAS attachments is that the devices will not have any effect on the FDDI ring if they are disconnected or powered off. Concentrators will be covered in more detail in the following discussion.

Each FDDI DAS has two ports, designated A and B. These ports connect the DAS to the dual FDDI ring. Therefore each port provides a connection for the primary and secondary rings. AS we will see in the next section devices using DAS connections will affect the rings if they are disconnected or powered off.

AN FDDI concentrator is the building block of an FDDI network. IT attached directly to both the primary and secondary rings and ensures

that the failure or power down of any SAS does not bring down the ring. This is particularly useful when PCs or similar devices that are frequently powered on and off, connect to the ring.

FDDI Fault Tolerance

FDDI provides a number of fault tolerant features. In particular, FDDI's dual ring environment the implementation of the optical bypass switch and dual homing support make FDDI a resilient media technology.

FDDI Frame Format

The FDDI frame format is similar to the format of a token Ring frame. This is one of the areas in which FDDI similar to earlier LAN technologies, such as token ring, FDDI frames can be as large as 4500 bytes,

FDDI Frame Fields: -

- Preamble: Give a unique sequence that prepares each station for an upcoming frame.
- Start delimiter: Indicate the beginning of a frame by employing a signaling pattern that differentiates it from the rest of the frame.
- Frame control: Indicates the size of the address fields and whether the frame contains asynchronous or synchronous data among other control information.
- Destination address: Contains a unicast, multicast or broadcast address .AS with Ethernet and Token Ring addresses, FDDI destination addresses are 6 bytes long.

- Source address: Identifies the single station that sent the frame. As with Ethernet and token ring addresses, FDDI source addresses are 6 bytes long.
- Data: Contains either information destined for upper layer protocol or control information.
- Frame check sequence (FCS): Is field by the source station with a calculated cyclic redundancy check value dependent on frame contents. The destination address recalculates the value to determine whether the frame was damaged in transit. If so, then frame is discarded.
- End delimiter: contains unique symbols; cannot be data symbols that indicate the end of the frame.
- Frame status: allows the source station to determine whether an error occurred; identifies whether the frame was recognized and copied by a receiving station.

Advantages of FDDI: -

- High bandwidth:- It has bandwidth as high as a 250gbps. High bandwidth allows or tremendous speed. FDDI implementation can handle data rates of 100Mbps.
- Security: - It is difficult to eavesdrop on the fiber optical cable transmission.
- Physical durability: - Fiber optical does not break as easily as do other kind of cable.
- Resistance to EMI: - Fiber optical cables aren't susceptible to electromagnetic interference.

- Cable distance: - Fiber optic can transmit signals over 2km.
- Weight: - Fiber optic cables weigh a lot less than a copper wire less than a copper wire with similar bandwidth.
- Use of multiple token: - FDDI uses multiple token to improve network speed.
- System fault tolerance: - FDDI can ISO late faulty nodes with the use of wiring concentrators function as centralized cable connection device for workstations. The other advantage is wrapping.

Disadvantages: -

- FDDI is a complex technology. Installation and maintenance require a great deal of expertise.
- FDDI is costly IN addition to the fiber optic cost; the adapters and concentrators are also very expensive.

4.10 Summary

In this lesson we have discussed Local Area Network (LAN), uses, attributes of LAN, FDDI and also IEEE LAN standards: 802.3, 802.4, 802.5, 802.6(MAN).

4.11 Self Assessment Questions

1. Describe about features of LAN and discuss five possible LAN applications.
2. Describe the advantages and disadvantages associated with five LAN topologies.
3. What is the difference between broadband and baseband signaling?

4. How can software determine a particular type of Ethernet frame as the frame flows on a network?
5. How is a MAC frame distinguished from an LLC frame on a Token-Ring network?
6. What is the purpose of the E bit in a Token-Ring frame?
7. Describe about FDDI.
8. Describe about Token Bus.
9. Describe about Ethernet.
10. Describe about IEEE LAN Standards 802.3, 802.4, 802.5, 802.6.

4.12 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
- Computer Network, ED Tittle, Tata MacGraw Hills Publications

Subject: Computer Networks **Subject Code:** MCA-301

Lesson No.: 5 **Author:** Mr. Yudhvir Singh

Lesson Title: Data Link Layer **Vettor:** **Dr. Manoj Dhun**

Structure

5.0 Objective

In this lesson we will study the design principles for layer 2, the data link layer. This study deals with the algorithms for achieving reliable and efficient communication between two adjacent machines at the data link layer. They have only a finite data rate, and there is a nonzero propagation delay between the time a bit is sent and the time it is received. These limitations have important implications on the efficiency of the data transfer. The protocols used for communications must take all these factors into consideration. Here we have discussed are Error Detection and Correction, Elementary and Sliding window protocols.

5.1 Introduction

In this lesson we will study the design principles for layer 2, the data link layer. This study deals with the algorithms for achieving reliable, efficient communication between two adjacent machines at the data link layer. At first you might think this problem is so trivial that there is no software to study—machine X just puts the bits on the wire, and machine Y just takes them off. Unfortunately, communication circuits make errors occasionally. Furthermore, they have only a finite data rate, and there is a nonzero propagation delay between the time a bit is sent and the time it is received. These limitations have important implications for the efficiency of the data transfer. The protocols used for communications must take all these factors into consideration.

The main task of the data link layer is to take a raw transmission facility and transform it into a line that appears free of transmission errors in the network layer. Here the sender break the input data into data frames, transmit the frames sequentially, and process the acknowledgment frames sent back by

the receiver. Since the physical layer merely accepts and transmits a stream of bits without any regard to meaning of structure, it is up to the data link layer to create and recognize frame boundaries. This can be accomplished by attaching special bit patterns to the beginning and end of the frame. If there is a chance that these bit patterns might occur in the data, special care must be taken to avoid confusion.

The data link layer should provide error control between adjacent nodes. Another issue that arises in the data link layer is how to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanism must be employed in order to let the transmitter know how much buffer space the receiver has at the moment. Frequently, flow regulation and error handling are integrated, for convenience. If the line can be used to transmit data in both directions, this introduces a new complication that the data link layer software must deal with. The problem is that the acknowledgment frames for A to B traffic compete for the use of the line with data frames for the B to A traffic. A clever solution piggybacking has been devised.

5.2 Data Link Layer Design Issues

The data link layer has a number of specific functions it can carry out. These functions include

1. Providing a well-defined service interface to the network layer.
2. Dealing with transmission errors.
3. Regulating the flow of data so that slow receivers are not swamped by fast senders.

To accomplish these goals, the data link layer takes the packets it gets from the network layer and encapsulates them into frames for transmission. Each

frame contains a frame header, a payload field for holding the packet, and a frame trailer.

In data link layer these are the requirements and objectives for effective data communication between two directly connected transmitting-receiving stations:

Frame synchronization-Data are sent in blocks called frames. The beginning and end of each frame must be recognizable.

Flow control-The sending station must not send frames at a rate faster then the receiving station can absorb them.

Error control-Any bit errors introduced by the transmission system must be corrected. Addressing-On a multipoint line, such as a local area network (LAN), the identity of the two stations involved in a transmission must be specified.

Control and data on same link- It is usually not desirable to have a physically separate communications path for control information. Accordingly, the receiver must be able to distinguish control information from the data being transmitted.

Link management-The initiation, maintenance, and termination of a sustained data exchange requires a fair amount of coordination and cooperation among stations. Procedures for the management of this exchange are required.

5.2.1 Services Provided to the Network Layer

The function of the data link layer is to provide services to the network layer. The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine. On the source machine is an entity, call it a process, in the network layer that hands some bits to the data link layer for transmission to the destination. The job of the

data link layer is to transmit the bits to the destination machine so they can be handed over to the network layer there.

The data link layer can be designed to offer various services. The actual services offered can vary from system to system. Three reasonable possibilities that are commonly provided are

1. Unacknowledged connectionless service.
2. Acknowledged connectionless service.
3. Acknowledged connection-oriented service.

Unacknowledged connectionless service consists of having the source machine send independent frames to the destination machine without having the destination machine acknowledge them. No logical connection is established beforehand or released afterward. Acknowledged connectionless service, there are still no logical connections used, but each frame sent is individually acknowledged. In this way, the sender knows whether a frame has arrived correctly. If it has not arrived within a specified time interval, it can be sent again.

Connection-oriented service, the source and destination machines establish a connection before any data is transferred. Each frame sent over the connection is numbered, and the data link layer guarantees that each frame sent is indeed received. Furthermore, it guarantees that each frame is received exactly once and that all frames are received in the right order. With connectionless service, in contrast, it is conceivable that a lost acknowledgement causes a packet to be sent several times and thus received several times. Connection-oriented service, in contrast, provides the network layer processes with the equivalent of a reliable bit stream.

When connection-oriented service is used, transfers go through three distinct phases. In the first phase, the connection is established by having both sides initialize variables and counters needed to keep track of which frames have been received and which ones have not. In the second phase, one or more frames are actually transmitted. In the third and final phase, the connection is released, freeing up the variables, buffers, and other resources used to maintain the connection.

5.2.2 Framing

To provide service to the network layer, the data link layer must use the service provided to it by the physical layer. What the physical layer does is accept a raw bit stream and attempt to deliver it to the destination. This bit stream is not guaranteed to be error free. The number of bits received may be less than, equal to, or more than the number of bits transmitted, and they may have different values. It is up to the data link layer to detect and, if necessary, correct errors.

The usual approach is 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. Breaking the bit stream up into frames is more difficult than it at first appears. In this section we will look at four methods:

1. Character count.
2. Flag bytes with byte stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

The first framing method uses a field in the header to specify the number of characters in the frame. When the data link layer at the destination sees the character count, it knows how many characters follow and hence where the end of the frame is.

The second framing method gets around the problem of resynchronization after an error by having each frame start and end with special bytes. In the past, the starting and ending bytes were different, but in recent years most protocols have used the same byte, called a flag byte, as both the starting and ending delimiter as FLAG. In this way, if the receiver ever loses synchronization, it can just search for the flag byte to find the end of the current frame. Two consecutive flag bytes indicate the end of one frame and start of the next one. A serious problem occurs with this method when binary data, such as object programs or floating-point numbers, are being transmitted. It may easily happen that the flag byte's bit pattern occurs in the data. This situation will usually interfere with the framing. One way to solve this problem is to have the sender's data link layer insert a special escape byte (ESC) just before each "accidental" flag byte in the data. The data link layer on the receiving end removes the escape byte before the data are given to the network layer. This technique is called byte stuffing or character stuffing. Thus, a framing flag byte can be distinguished from one in the data by the absence or presence of an escape byte before it.

The new technique allows data frames to contain an arbitrary number of bits and allows character codes with an arbitrary number of bits per character. It works like this. Each frame begins and ends with a special bit pattern. Whenever the sender's data link layer encounters five consecutive 1s in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This bit stuffing

is analogous to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs the 0 bit. Just as byte stuffing is completely transparent to the network layer in both computers, so is bit stuffing. With bit stuffing, the boundary between two frames can be unambiguously recognized by the flag pattern. Thus, if the receiver loses track of where it is, all it has to do is scan the input for flag sequences, since they can only occur at frame boundaries and never within the data.

The last method of framing is only applicable to networks in which the encoding on the physical medium contains some redundancy. The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. The combinations high-high and low-low are not used for data but are used for delimiting frames in some protocols. As a final note on framing, many data link protocols use a combination of a character 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 delimiter is present at that position and the checksum is correct is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter.

5.2.3 Error Control

The next problem: how to make sure that all frames are eventually delivered to the network layer at the destination and in the proper order. The usual way to ensure reliable delivery is to provide the sender with 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. The obvious solution is to just transmit the frame again. However, when frames may be transmitted multiple times there is a danger that the receiver will accept the same frame two or more times and pass it to the network layer more than once. To prevent this from happening, it is generally necessary to assign sequence numbers to outgoing frames, so that the receiver can distinguish retransmissions from originals. The whole issue of managing the timers and sequence numbers so as to ensure that each frame is ultimately passed to the network layer at the destination exactly once, no more and no less, is an important part of the data link layer's duties.

5.2.4 Flow Control

Another important design issue that occurs in the data link layer is what to do with a sender that systematically wants to transmit frames faster than the receiver can accept them. This situation can easily occur when the sender is running on a fast computer and the receiver is running on a slow machine. 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 be unable to handle the frames as they arrive and will start to lose some. Clearly, something has to be done to prevent this situation. Two approaches are commonly used. In the first one, feedback-based flow control, the receiver sends back information to the sender giving it permission to send more data or at least telling the sender how the receiver is doing. In the second one, rate-based flow control, the protocol has a built-in

mechanism that limits the rate at which senders may transmit data, without using feedback from the receiver

5.3 Error Detection and Correction

Transmission errors are going to be with us for many years to come. The physical processes that generate them, errors on some media tend to come in bursts rather than singly. Having the errors come in bursts has both advantages and disadvantages over isolated single-bit errors. On the advantage side, computer data are always sent in blocks of bits. If errors were independent, most blocks would contain an error. The disadvantage of burst errors is that they are much harder to correct than are isolated errors.

5.3.1 Error-Correcting Codes

Network designers have developed two basic strategies for dealing with errors. One way is to include enough redundant information along with each block of data sent, to enable the receiver to deduce what the transmitted data must have been. The other way is to include only enough redundancy to allow the receiver to deduce that an error occurred, but not which error, and have it request a retransmission. The former strategy uses error-correcting codes and the latter uses error-detecting codes. The use of error-correcting codes is often referred to as forward error correction. On channels that are highly reliable, it is cheaper to use an error detecting code and just retransmit the occasional block found to be faulty. However, on channels that make many errors, it is better to add enough redundancy to each block for the receiver to be able to figure out what the original block was, rather than relying on a retransmission, which itself may be in error. A frame consists of m data bits

and r redundant, or check, bits. Let the total length be n (n = m + r). An n-bit unit containing data and check bits is often referred to as an n-bit codeword.
Given any two codewords, say, 10001001 and 10110001, it is possible to determine how many corresponding bits differ. In this case, 3 bits differ. To determine how many bits differ, just exclusive OR the two codewords and count the number of 1 bits in the result. The number of bit positions in which two codewords differ is called the Hamming distance. Its significance is that if two codewords are a Hamming distance d apart, it will require d single-bit errors to convert one into the other.

In most data transmission applications, all 2^m possible data messages are legal, but due to the way the check bits are computed, not all of the 2^n possible codewords are used. Given the algorithm for computing the check bits, it is possible to construct a complete list of the legal codewords, and from this list find the two codewords whose Hamming distance is minimum. This distance is the Hamming distance of the complete code.
The error-detecting and error-correcting properties of a code depend on its Hamming distance. To detect d errors, you need a distance d + 1 code because with such a code there is no way that d single-bit errors can change a valid codeword into another valid codeword. When the receiver sees an invalid codeword, it can tell that a transmission error has occurred. Similarly, to correct d errors, you need a distance 2d + 1 code because that way the legal codewords are so far apart that even with d changes, the original codeword is still closer than any other codeword, so it can be uniquely determined.
As a simple example of an error-detecting code, consider a code in which a single parity bit is appended to the data. The parity bit is chosen so that the

number of 1 bits in the codeword is even (or odd). For example, when 1011010 is sent in even parity, a bit is added to the end to make it 10110100. With odd parity 1011010 becomes 10110101. A code with a single parity bit has a distance 2, since any single-bit error produces a codeword with the wrong parity. It can be used to detect single errors.

As a simple example of an error-correcting code, consider a code with only four valid codewords:

0000000000, 0000011111, 1111100000, and 1111111111

This code has a distance 5, which means that it can correct double errors. If the codeword 0000000111 arrives, the receiver knows that the original must have been 0000011111. If, however, a triple error changes 0000000000 into 0000000111, the error will not be corrected properly.

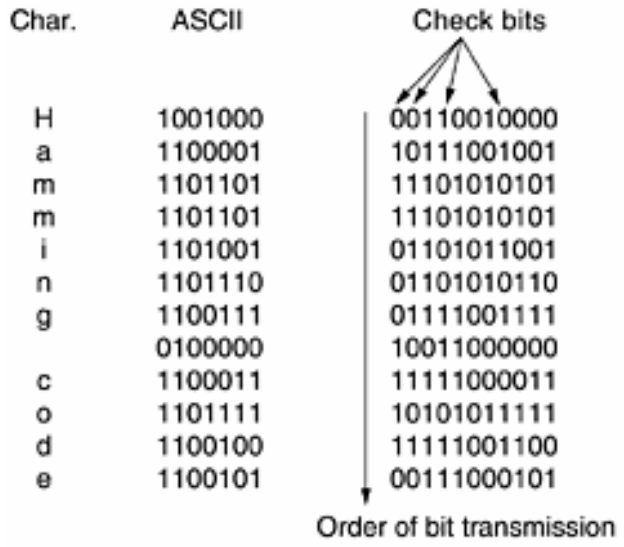


Figure 5.1: Use of a Hamming code to correct burst errors

When a codeword arrives, the receiver initializes a counter to zero. It then examines each check bit, k (k = 1, 2, 4, 8, ...), to see if it has the correct parity. If not, the receiver adds k to the counter. If the counter is zero after all the check bits have been examined (i.e., if they were all correct), the codeword is accepted as valid. If the counter is nonzero, it contains the number of the incorrect bit. For example, if check bits 1, 2, and 8 are in error, the inverted bit is 11, because it is the only one checked by bits 1, 2, and 8. Figure5.1 shows some 7-bit ASCII characters encoded as 11-bit codewords using a Hamming code. Remember that the data are found in bit positions 3, 5, 6, 7, 9, 10, and 11.

Hamming codes can only correct single errors. However, there is a trick that can be used to permit Hamming codes to correct burst errors. A sequence of k consecutive codewords are arranged as a matrix, one codeword per row. Normally, the data would be transmitted one codeword at a time, from left to right. To correct burst errors, the data should be transmitted one column at a time, starting with the leftmost column. When all k bits have been sent, the second column is sent, and so on. When the frame arrives at the receiver, the matrix is reconstructed, one column at a time. If a burst error of length k occurs, at most 1 bit in each of the k codewords will have been affected, but the Hamming code can correct one error per codeword, so the entire block can be restored. This method uses kr check bits to make blocks of km data bits immune to a single burst error of length k or less.

5.3.2 Error-Detecting Codes

As a simple example, consider a channel on which errors are isolated and the error rate is 10^{-6} per bit. Let the block size be 1000 bits. To provide error

correction for 1000-bit blocks, 10 check bits are needed; a megabit of data would require 10,000 check bits. To merely detect a block with a single 1-bit error, one parity bit per block will suffice. Once every 1000 blocks, an extra block (1001 bits) will have to be transmitted. The total overhead for the error detection + retransmission method is only 2001 bits per megabit of data, versus 10,000 bits for a Hamming code.

The theoretical limitations of coding are placed by the results of information theory. These results are frustrating in that they offer little clue as to how the coding should be performed. Error detection coding is designed to permit the detection of errors. Once detected, the receiver may ask for a re-transmission of the erroneous bits, or it may simply inform the recipient that the transmission was corrupted. In a binary channel, error checking codes are called parity check codes.

If a single parity bit is added to a block and the block is badly garbled by a long burst error, the probability that the error will be detected is only 0.5, which is hardly acceptable. The odds can be improved considerably if each block to be sent is regarded as a rectangular matrix n bits wide and k bits high, as described above. A parity bit is computed separately for each column and affixed to the matrix as the last row. The matrix is then transmitted one row at a time. When the block arrives, the receiver checks all the parity bits. If any one of them is wrong, the receiver requests a retransmission of the block. Additional retransmissions are requested as needed until an entire block is received without any parity errors.

Practical codes are normally block codes. A block code converts a fixed length of K data bits to a fixed length N codeword, where N > K. The rate of the code is the ratio ***K/N***, and the redundancy of the code is ***1-(K/N)*** . Our

ability to detect errors depends on the rate. A low rate has a high detection probability, but a high redundancy.

The receiver will assign to the received codeword the preassigned codeword that minimises the Hamming distance between the two words. If we wish to identify any pattern of n or less errors, the Hamming distance between the preassigned codewords must be n + 1 or greater. A very common code is the single parity check code. This code appends to each K data bits an additional bit whose value is taken to make the K + 1 word even (or odd). Such a choice is said to have even (odd) parity. With even (odd) parity, a single bit error will make the received word odd (even). The preassigned code words are always even (odd), and hence are separated by a Hamming distance of 2 or more. To see how the addition of a parity bit can improve error performance, consider the following example. A common choice of K is eight. Suppose the BER is $p=10^{-4}$. Then:

$$
\begin{aligned}
P\{\text{single bit error}\} &= p \\
P\{\text{no error in single bit}\} &= (1-p) \\
P\{\text{no error in 8 bits}\} &= (1-p)^8 \\
P\{\text{unseen error in 8 bits}\} &= 1-(1-p)^8 \\
&= 7.9 \times 10^{-4}
\end{aligned}
$$

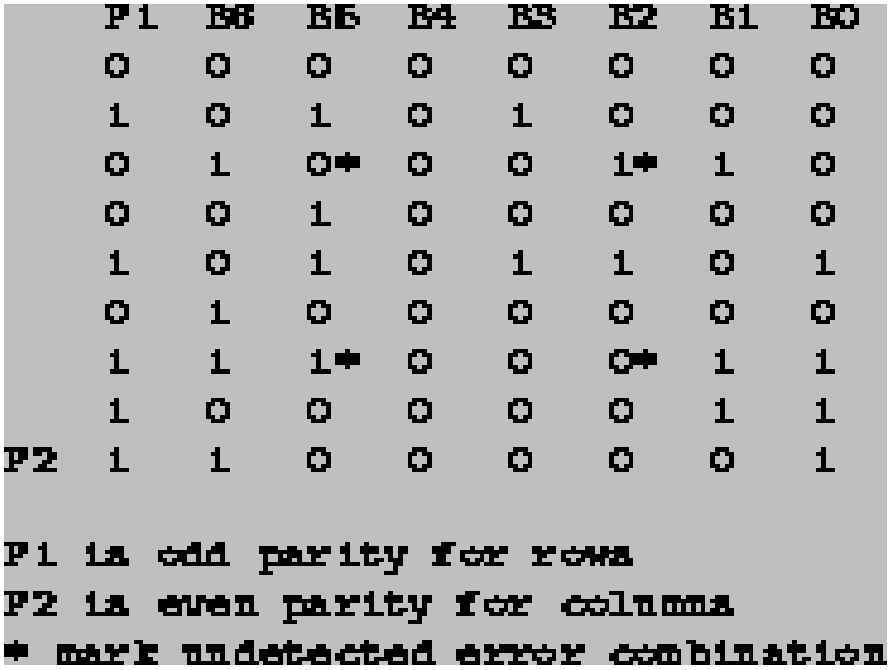
So, the probability of a transmission with an error is $7.9 * 10^{-4}$. With the addition of a parity error bit we can detect any single bit error. So:

$$
\begin{aligned}
P\{\text{no error in single bit}\} &= (1-p) \\
P\{\text{no error in 9 bits}\} &= (1-p)^9 \\
P\{\text{single error in 9 bits}\} &= 9(P\{\text{single bit error}\}P\{\text{no error in other 8 bits}\}) \\
&= 9p(1-p)^8 \\
P\{\text{unseen error in 9 bits}\} &= 1-P\{\text{no error in 9 bits}\}-P\{\text{single error in 9 bits}\} \\
&= 1-(1-p)^9-9p(1-p)^8 \\
&= 3.6 \times 10^{-7}
\end{aligned}
$$

As it can be seen, the addition of a parity bit has reduced the uncorrected error rate by three orders of magnitude. Single parity bits are common in asynchronous, character oriented transmission. Where synchronous transmission is used, additional parity symbols are added that checks not only the parity of each 8 bit *row*, but also the parity of each 8 bit *column*. The column is formed by listing each successive 8 bit word one beneath the other. This type of parity checking is called block sum checking, and it can correct any single 2 bit error in the transmitted block of rows and columns. However, there are some combinations of errors that will go undetected in such a scheme in figure 5.2.

Parity checking in this way provides good protection against single and multiple bit errors when the probability of the errors are independent. However, in many circumstances, errors occur in groups, or bursts. Parity checking of the kind just described then provides little protection. In these circumstances, a polynomial code is used.

This method can detect a single burst of length n, since only 1 bit per column will be changed. A burst of length n + 1 will pass undetected, however, if the first bit is inverted, the last bit is inverted, and all the other bits are correct. If the block is badly garbled by a long burst or by multiple shorter bursts, the probability that any of the n columns will have the correct parity, by accident, is 0.5, so the probability of a bad block being accepted when it should not be is 2^{-n}.



	P1	B7	B6	B4	B3	B2	B1	B0
	0	0	0	0	0	0	0	0
	1	0	1	0	1	0	0	0
	0	1	0*	0	0	1*	1	0
	0	0	1	0	0	0	0	0
	1	0	1	0	1	1	0	1
	0	1	0	0	0	0	0	0
	1	1	1*	0	0	0*	1	1
	1	0	0	0	0	0	1	1
P2	1	1	0	0	0	0	0	1

P1 is odd parity for rows
P2 is even parity for columns
* mark undetected error combination

Figure 5.2 : **Example of block sum check showing undetected errors**

Polynomial codes work on each frame. Additional digits are added to the end of each frame. These digits depend on the contents of the frame. The number of added digits depends on the length of the expected error burst. Typically 16 or 32 digits are added. The computed digits are called the frame check sequence (FCS) or cyclic redundancy check (CRC). Before transmission, each frame is divided by a generator polynomial. The remainder of this division is added to the frame. On reception, the division is repeated. Since the remainder has been added, the result should be zero. A non-zero result indicates that an error has occurred. A polynomial code can detect any error burst of length less than or equal to the length of the generator polynomial. The technique requires the addition of hardware to perform the division. However, with modern integrated circuitry, this hardware is now available

inexpensively. CRC error checking is now quite common, and its use will increase.

CRC (Cyclic Redundancy Check)- Polynomial codes are based upon treating bit strings as representations of polynomials with coefficients of 0 and 1 only. A k-bit frame is regarded as the coefficient list for a polynomial with k terms, ranging from x^{k-1} to x^0. Such a polynomial is said to be of degree k - 1. The high-order (leftmost) bit is the coefficient of x^{k-1}; the next bit is the coefficient of x^{k-2}, and so on. For example, 110001 has 6 bits and thus represents a six-term polynomial with coefficients 1, 1, 0, 0, 0, and 1: $x^5 + x^4 + x^0$. Polynomial arithmetic is done modulo 2, according to the rules of algebraic field theory. There are no carries for addition or borrows for subtraction. Both addition and subtraction are identical to exclusive OR. Long division is carried out the same way as it is in binary except that the subtraction is done modulo 2, as above. A divisor is said "to go into" a dividend if the dividend has as many bits as the divisor.

When the polynomial code method is employed, the sender and receiver must agree upon a generator polynomial, G(x), in advance. Both the high- and low-order bits of the generator must be 1. To compute the checksum for some frame with m bits, corresponding to the polynomial M(x), the frame must be longer than the generator polynomial. The idea is to append a checksum to the end of the frame in such a way that the polynomial represented by the checksummed frame is divisible by G(x). When the receiver gets the checksummed frame, it tries dividing it by G(x). If there is a remainder, there has been a transmission error.

The algorithm for computing the checksum is as follows:

1. Let r be the degree of G(x). Append r zero bits to the low-order end of the frame so it now contains m + r bits and corresponds to the polynomial $x^rM(x)$.
2. Divide the bit string corresponding to G(x) into the bit string corresponding to $x^rM(x)$, using modulo 2 division.
3. Subtract the remainder (which is always r or fewer bits) from the bit string corresponding to $x^rM(x)$ using modulo 2 subtraction.

Figure 5.3 - illustrates the calculation for a frame 1101011011 using the generator $G(x) = x^4 + x + 1$.checksummed frame to be transmitted. Call its polynomial T(x).

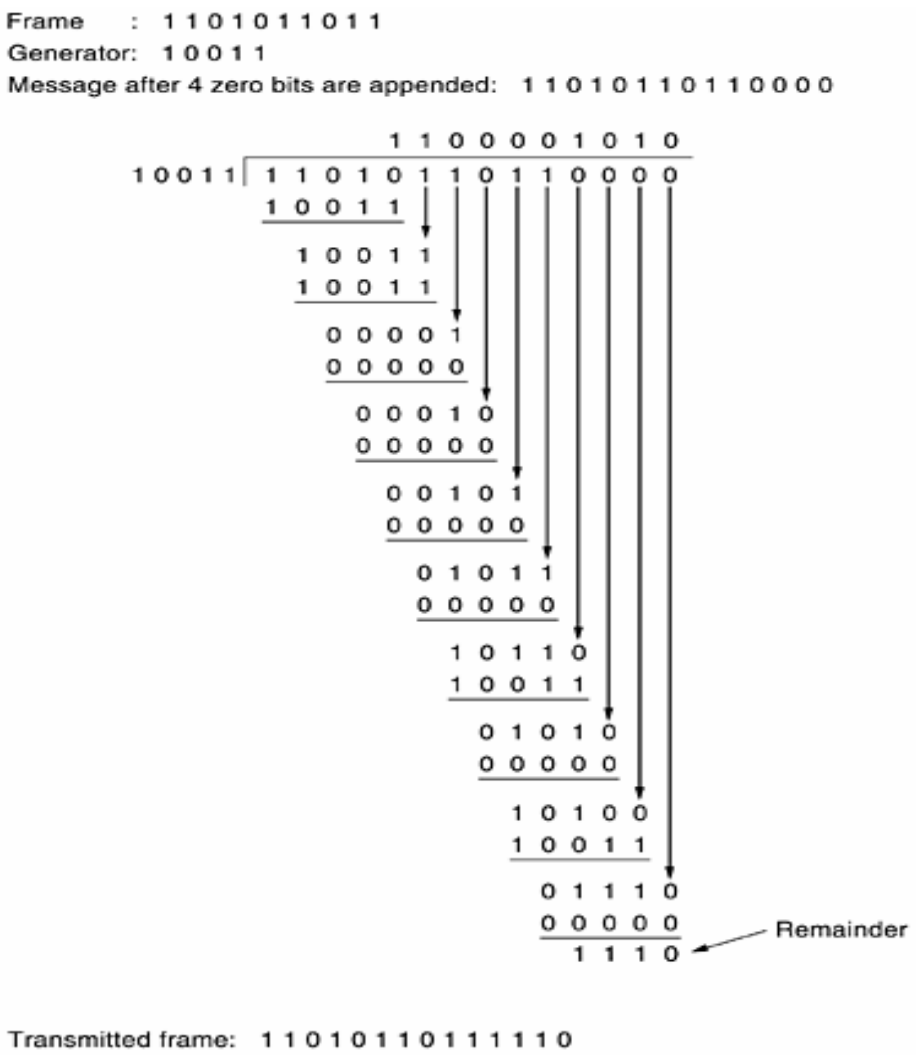


Figure 5.3: Calculation of the polynomial code checksum.

It should be clear that T(x) is divisible (modulo 2) by G(x). In any division problem, if you diminish the dividend by the remainder, what is left over is divisible by the divisor. Imagine that a transmission error occurs, so that

instead of the bit string for T(x) arriving, T(x) + E(x) arrives. Each 1 bit in E(x) corresponds to a bit that has been inverted. If there are k1 bits in E(x), k single-bit errors have occurred. A single burst error is characterized by an initial 1, a mixture of 0s and 1s, and a final 1, with all other bits being 0. Upon receiving the checksummed frame, the receiver divides it by G(x); that is, it computes [T(x) + E(x)]/G(x). T(x)/G(x) is 0, so the result of the computation is simply E(x)/G(x). Those errors that happen to correspond to polynomials containing G(x) as a factor will slip by; all other errors will be caught.

5.4 Elementary Data Link Protocols

We assume that in the physical layer, data link layer, and network layer are independent processes that communicate by passing messages back and forth. As far as the data link layer is concerned, the packet passed across the interface to it from the network layer is pure data, whose every bit is to be delivered to the destination's network layer. The fact that the destination's network layer may interpret part of the packet as a header is of no concern to the data link layer. When the data link layer accepts a packet, it encapsulates the packet in a frame by adding a data link header and trailer to it. Thus, a frame consists of an embedded packet, some control information and a checksum. The frame is then transmitted to the data link layer on the other machine. When a frame arrives at the receiver, the hardware computes the checksum. If the checksum is incorrect, the data link layer is so informed. If the inbound frame arrived undamaged, the data link layer is also informed. As soon as the receiving data link layer has acquired an undamaged frame, it checks the control information in the header, and if everything is all right, passes the packet portion to the network layer. Under no circumstances is a frame header ever given to a network layer. A frame is composed of four

fields: kind, seq, ack, and info, the first three of which contain control information and the last of which may contain actual data to be transferred. These control fields are collectively called the frame header.

The kind field tells whether there are any data in the frame, because some of the protocols distinguish frames containing only control information from those containing data as well. The seq and ack fields are used for sequence numbers and acknowledgements, respectively; their use will be described in more detail later. The info field of a data frame contains a single packet; the info. field of a control frame is not used.

The network layer builds a packet by taking a message from the transport layer and adding the network layer header to it. This packet is passed to the data link layer for inclusion in the info. field of an outgoing frame. When the frame arrives at the destination, the data link layer extracts the packet from the frame and passes the packet to the network layer. In this manner, the network layer can act as though machines can exchange packets directly.

5.4.1 An Unrestricted Simplex Protocol

We will consider a protocol that is as simple as it can be. Data are transmitted in one direction only. Both the transmitting and receiving network layers are always ready. Processing time can be ignored. Infinite buffer space is available. And best of all, the communication channel between the data link layers never damages or loses frames. The protocol consists of two distinct procedures, a sender and a receiver. The sender runs in the data link layer of the source machine, and the receiver runs in the data link layer of the destination machine. No sequence numbers or acknowledgements are used here, so MAX_SEQ is not needed. The only event type possible is frame_arrival (i.e., the arrival of an undamaged frame).

The sender is in an infinite while loop just pumping data out onto the line as fast as it can. The body of the loop consists of three actions: go fetch a packet from the (always obliging) network layer, construct an outbound frame using the variable s, and send the frame on its way. Only the info field of the frame is used by this protocol, because the other fields have to do with error and flow control and there are no errors or flow control restrictions here. The receiver is equally simple. Initially, it waits for something to happen, the only possibility being the arrival of an undamaged frame. The data portion is passed on to the network layer, and the data link layer settles back to wait for the next frame, effectively suspending itself until the frame arrives.

5.4.2 A Simplex Stop-and-Wait Protocol

Now we will drop the most unrealistic restriction used in protocol 1: the ability of the receiving network layer to process incoming data infinitely quickly (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 their 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 them. In essence, if the receiver requires a time dt to execute from_physical_layer plus to_network_layer, the sender must transmit at an average rate less than one frame per time (i.e. dt). Moreover, if we assume that no automatic buffering and queueing are done 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.

In certain restricted, it might be possible for the sender to simply insert a delay into protocol 1 to slow it down sufficiently to keep from swamping the receiver. However, more usually, each data link layer will have several lines to attend to, and the time interval between a frame arriving and its being processed may vary considerably. If the network designers can calculate the worst-case behavior of the receiver, they can program the sender to transmit so slowly that even if every frame suffers the maximum delay, there will be no overruns. It leads to a bandwidth utilization that is far below the optimum, unless the best and worst cases are almost the same (i.e., the variation in the data link layer's reaction time is small).

A more general solution to this dilemma is to have the receiver provide feedback to the sender. After having passed a packet to 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., acknowledgement) frame arrives. Protocols in which the sender sends one frame and then waits for an acknowledgement before proceeding are called stop-and-wait.

This protocol entails a strict alternation of flow: first the sender sends a frame, then the receiver sends a frame, then the sender sends another frame, then the receiver sends another one, and so on. A half- duplex physical channel would suffice here.

As in protocol 1, the sender starts out by fetching a packet from the network layer, using it to construct a frame, and sending it on its way. But now, unlike in protocol 1, the sender must wait until an acknowledgement frame arrives before looping back and fetching the next packet from the network layer.

5.4.3 A Simplex Protocol for a Noisy Channel

Now let us consider the normal situation of a communication channel that makes errors. Frames may be either damaged or lost completely. However, we assume that if a frame is damaged in transit, the receiver hardware will detect this when it computes the checksum. If the frame is damaged in such a way that the checksum is nevertheless correct, an unlikely occurrence, this protocol can fail. At first glance it might seem that a variation of protocol 2 would work: adding a timer. The sender could send a frame, but the receiver would only send an acknowledgement frame if the data were correctly received. If a damaged frame arrived at the receiver, it would be discarded. After a while the sender would time out and send the frame again. This process would be repeated until the frame finally arrived intact.

To see what might go wrong, remember that it is the task of the data link layer processes to provide error-free, transparent communication between network layer processes. The network layer on machine A gives a series of packets to its data link layer, which must ensure that an identical series of packets are delivered to the network layer on machine B by its data link layer. In particular, the network layer on B has no way of knowing that a packet has been lost or duplicated, so the data link layer must guarantee that no combination of transmission errors, however unlikely, can cause a duplicate packet to be delivered to a network layer.

Consider the following scenario:

1. The network layer on A gives packet 1 to its data link layer. The packet is correctly received at B and passed to the network layer on B. B sends an acknowledgement frame back to A.

2. The acknowledgement frame gets lost completely. It just never arrives at all. Life would be a great deal simpler if the channel mangled and lost only data frames and not control frames, but sad to say, the channel is not very discriminating.
3. The data link layer on A eventually times out. Not having received an acknowledgement, it (incorrectly) assumes that its data frame was lost or damaged and sends the frame containing packet 1 again.
4. The duplicate frame also arrives at the data link layer on B perfectly and is unwittingly passed to the network layer there. If A is sending a file to B, part of the file will be duplicated (i.e., the copy of the file made by B will be incorrect and the error will not have been detected). In other words, the protocol will fail.

Clearly, what is needed is some way for the receiver to be able to distinguish a frame that it is seeing for the first time from a retransmission. The obvious way to achieve this is to have the sender put a sequence number in the header of each frame it sends. Then the receiver can check the sequence number of each arriving frame to see if it is a new frame or a duplicate to be discarded.

This protocol differs from its predecessors in that both sender and receiver have a variable whose value is remembered while the data link layer is in the wait state. The sender remembers the sequence number of the next frame to send; the receiver remembers the sequence number of the next frame expected. Each protocol has a short initialization phase before entering the infinite loop. After transmitting a frame and starting the timer, the sender waits for something exciting to happen. Only three possibilities exist: an acknowledgement frame arrives undamaged, a damaged acknowledgement frame staggers in, or the timer expires. If a valid acknowledgement comes in,

the sender fetches the next packet from its network layer and puts it in the buffer, overwriting the previous packet. It also advances the sequence number. If a damaged frame arrives or no frame at all arrives, neither the buffer nor the sequence number is changed so that a duplicate can be sent. When a valid frame arrives at the receiver, its sequence number is checked to see if it is a duplicate. If not, it is accepted, passed to the network layer, and an acknowledgement is generated. Duplicates and damaged frames are not passed to the network layer.

5.5 Sliding Window Protocols

In most practical situations, there is a need for transmitting data in both directions. One way of achieving full-duplex data transmission is to have two separate communication channels and use each one for simplex data traffic. If this is done, we have two separate physical circuits, each with a "forward" channel (for data) and a "reverse" channel (for acknowledgements). In both cases the bandwidth of the reverse channel is almost entirely wasted. In effect, the user is paying for two circuits but using only the capacity of one.

A better idea is to use the same circuit for data in both directions. After all, in protocols 2 and 3 it was already being used to transmit frames both ways, and the reverse channel has the same capacity as the forward channel. In this model the data frames from A to B are intermixed with the acknowledgement frames from A to B. By looking at the kind field in the header of an incoming frame, the receiver can tell whether the frame is data or acknowledgement.

Although interleaving data and control frames on the same circuit is an improvement over having two separate physical circuits, yet another improvement is possible. When a data frame arrives, instead of immediately sending a separate control frame, the receiver restrains itself and waits until

the network layer passes it the next packet. The acknowledgement is attached to the outgoing data frame (using the ack field in the frame header). In effect, the acknowledgement gets a free ride on the next outgoing data frame. The technique of temporarily delaying outgoing acknowledgements so that they can be hooked onto the next outgoing data frame is known as piggybacking.

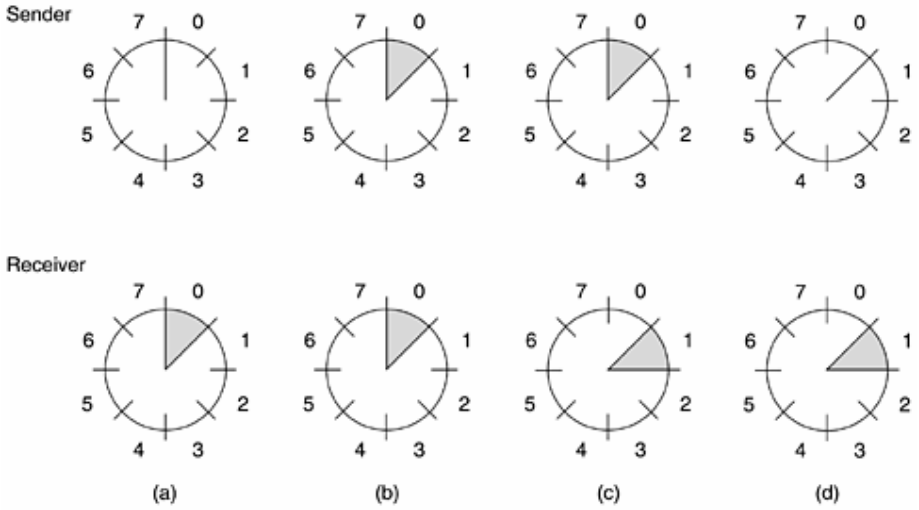
The principal advantage of using piggybacking over having distinct acknowledgement frames is a better use of the available channel bandwidth. The ack field in the frame header costs only a few bits, whereas a separate frame would need a header, the acknowledgement, and a checksum. It rarely costs more than a few bits.

The data link layer cannot foretell the future, so it must resort to some ad hoc scheme, such as waiting a fixed number of milliseconds. If a new packet arrives quickly, the acknowledgement is piggybacked onto it; otherwise, if no new packet has arrived by the end of this time period, the data link layer just sends a separate acknowledgement frame.

The next three protocols are bidirectional protocols that belong to a class called sliding window protocols. The three differ among themselves in terms of efficiency, complexity, and buffer requirements. In these, as in all sliding window protocols, each outbound frame contains a sequence number, ranging from 0 up to some maximum. The maximum is usually $2^n - 1$ so the sequence number fits exactly in an n-bit field. The stop-and-wait sliding window protocol uses n = 1, restricting the sequence numbers to 0 and 1, but more sophisticated versions can use arbitrary n.

The essence of all sliding window protocols is that at any instant of time, the sender maintains a set of sequence numbers corresponding to frames it is permitted to send. The sequence numbers within the sender's window

represent frames that have been sent or can be sent but are as yet not acknowledged. Whenever a new packet arrives from the network layer, it is



given the next highest sequence number, and the upper edge of the window is advanced by one. When an acknowledgement comes in, one advances the lower edge. In this way the window continuously maintains a list of unacknowledged frames. Figure 5.4 shows an example of 8-windows.

Figure 5.4: 8-Windows for Sliding Window Protocols

Since frames currently within the sender's window may ultimately be lost or damaged in transit, the sender must keep all these frames in its memory for possible retransmission. Thus, if the maximum window size is n, the sender needs n buffers to hold the unacknowledged frames. If the window ever grows to its maximum size, the sending data link layer must forcibly shut off the network layer until another buffer becomes free.

The receiving data link layer's window corresponds to the frames it may accept. Any frame falling outside the window is discarded without comment. When a frame whose sequence number is equal to the lower edge of the

window is received, it is passed to the network layer, an acknowledgement is generated, and the window is rotated by one. Unlike the sender's window, the receiver's window always remains at its initial size. Note that a window size of 1 means that the data link layer only accepts frames in order, but for larger windows this is not so. The network layer, in contrast, is always fed data in the proper order, regardless of the data link layer's window size.

5.5.1 A One-Bit Sliding Window Protocol

Let us first examine a sliding window protocol with a maximum window size of 1. Such a protocol uses stop-and-wait since the sender transmits a frame and waits for its acknowledgement before sending the next one.

Under normal circumstances, one of the two data link layers goes first and transmits the first frame. In the event that both data link layers start off simultaneously, a peculiar situation arises. The starting machine fetches the first packet from its network layer, builds a frame from it, and sends it. When this frame arrives, the receiving data link layer checks to see if it is a duplicate, just as in protocol 3. If the frame is the one expected, it is passed to the network layer and the receiver's window is slid up. The acknowledgement field contains the number of the last frame received without error. If this number agrees with the sequence number of the frame the sender is trying to send, the sender knows it is done with the frame stored in buffer and can fetch the next packet from its network layer. If the sequence number disagrees, it must continue trying to send the same frame. Whenever a frame is received, a frame is also sent back.

5.5.2 A Protocol Using Go Back N

Until now we have made the tacit assumption that the transmission time required for a frame to arrive at the receiver plus the transmission time for the acknowledgement to come back is negligible. Sometimes this assumption is clearly false. In these situations the long round-trip time can have important implications for the efficiency of the bandwidth utilization. Clearly, the combination of a long transit time, high bandwidth, and short frame length is disastrous in terms of efficiency.

The problem described above can be viewed as a consequence of the rule requiring a sender to wait for an acknowledgement before sending another frame. If we relax that restriction, much better efficiency can be achieved. Basically, the solution lies in allowing the sender to transmit up to w frames before blocking, instead of just 1. With an appropriate choice of w the sender will be able to continuously transmit frames for a time equal to the round-trip transit time without filling up the window.

The need for a large window on the sending side occurs whenever the product of bandwidth x round-trip-delay is large. If the bandwidth is high, even for a moderate delay, the sender will exhaust its window quickly unless it has a large window. If the delay is high, the sender will exhaust its window even for a moderate bandwidth. The product of these two factors basically tells what the capacity of the pipe is, and the sender needs the ability to fill it without stopping in order to operate at peak efficiency. This technique is known as pipelining. If the channel capacity is b bits/sec, the frame size L bits, and the round-trip propagation time T sec, the time required to transmit a single frame is L/b sec. After the last bit of a data frame has been sent, there is a delay of T/2 before that bit arrives at the receiver and another delay of at

least T/2 for the acknowledgement to come back, for a total delay of R. In stop-and-wait the line is busy for L/band idle for T, giving

Line Utilization = (L/L+bT)

If L < bT, the efficiency will be less than 50 percent. Since there is always a nonzero delay for the acknowledgement to propagate back, pipelining can, in principle, be used to keep the line busy during this interval, but if the interval is small, the additional complexity is not worth. Pipelining frames over an unreliable communication channel raises some serious issues. What happens if a frame in the middle of a long stream is damaged or lost? Large numbers of succeeding frames will arrive at the receiver before the sender even finds out that anything is wrong. When a damaged frame arrives at the receiver, it obviously should be discarded, but what should the receiver do with all the correct frames following it? The receiving data link layer is obligated to hand packets to the network layer in sequence.

Two basic approaches are available for dealing with errors in the presence of pipelining. One way, called go back n, is for the receiver simply to discard all subsequent frames, sending no acknowledgements for the discarded frames. This strategy corresponds to a receive window of size 1. In other words, the data link layer refuses to accept any frame except the next one it must give to the network layer. If the sender's window fills up before the timer runs out, the pipeline will begin to empty. Eventually, the sender will time out and retransmit all unacknowledged frames in order, starting with the damaged or lost one. This approach can waste a lot of bandwidth if the error rate is high.

The other general strategy for handling errors when frames are pipelined is called selective repeat. When it is used, a bad frame that is received is discarded, but good frames received after it are buffered. When the sender

times out, only the oldest unacknowledged frame is retransmitted. If that frame arrives correctly, the receiver can deliver to the network layer, in sequence, all the frames it has buffered. Selective repeat is often combined with having the receiver send a negative acknowledgement (NAK) when it detects an error, for example, when it receives a checksum error or a frame out of sequence. NAKs stimulate retransmission before the corresponding timer expires and thus improve performance. These two alternative approaches are trade-offs between bandwidth and data link layer buffer space. Depending on which resource is scarcer, one or the other can be used.

5.5.3 A Protocol Using Selective Repeat

Earlier protocol works well if errors are rare, but if the line is poor, it wastes a lot of bandwidth on retransmitted frames. An alternative strategy for handling errors is to allow the receiver to accept and buffer the frames following a damaged or lost one. Such a protocol does not discard frames merely because an earlier frame was damaged or lost.

In this protocol, both sender and receiver maintain a window of acceptable sequence numbers. The sender's window size starts out at 0 and grows to some predefined maximum, MAX_SEQ. The receiver's window, in contrast, is always fixed in size and equal to MAX_SEQ. The receiver has a buffer reserved for each sequence number within its fixed window. Associated with each buffer is a bit (arrived) telling whether the buffer is full or empty. Whenever a frame arrives, its sequence number is checked by the function between to see if it falls within the window. If so and if it has not already been received, it is accepted and stored. This action is taken without regard to whether or not it contains the next packet expected by the network layer. Of

course, it must be kept within the data link layer and not passed to the network layer until all the lower-numbered frames have already been delivered to the network layer in the correct order.

Non-sequential receive introduces certain problems not present in protocols in which frames are only accepted in order. For the same reason, the number of timers needed is equal to the number of buffers, not to the size of the sequence space. Effectively, a timer is associated with each buffer. When the timer runs out, the contents of the buffer are retransmitted. It is essential that the timeout associated with the auxiliary timer be appreciably shorter than the timer used for timing out data frames. This condition is required to make sure a correctly received frame is acknowledged early enough that the frame's retransmission timer does not expire and retransmit the frame.

5.6 Summary

The task of the data link layer is to convert the raw bit stream offered by the physical layer into a stream of frames for use by the network layer. Various framing methods are used, including character count, byte stuffing, and bit stuffing. Data link protocols can provide error control to retransmit damaged or lost frames. To prevent a fast sender from overrunning a slow receiver, the data link protocol can also provide flow control. The sliding window mechanism is widely used to integrate error control and flow control in a convenient way. Sliding window protocols can be categorized by the size of the sender's window and the size of the receiver's window. When both are equal to 1, the protocol is stop-and-wait. When the sender's window is greater than 1, for example, to prevent the sender from blocking on a circuit with a long propagation delay, the receiver can be programmed either to discard all

frames other than the next one in sequence or to buffer out-of-order frames until they are needed.

5.7 Self Assessment Questions

1. The following character encoding is used in a data link protocol: A: 01000111; B: 11100011; FLAG: 01111110; ESC: 11100000 Show the bit sequence transmitted (in binary) for the four-character frame: A B ESC FLAG when each of the following framing methods are used:
    a. Character count.
    b. Flag bytes with byte stuffing.
    c. Starting and ending flag bytes, with bit stuffing.
2. When bit stuffing is used, is it possible for the loss, insertion, or modification of a single bit to cause an error not detected by the checksum? If not, why not? If so, how? Does the checksum length play a role here?
3. To provide more reliability than a single parity bit can give, an error-detecting coding scheme uses one parity bit for checking all the odd-numbered bits and a second parity bit for all the even-numbered bits. What is the Hamming distance of this code?
4. An 8-bit byte with binary value 10101111 is to be encoded using an even-parity Hamming code. What is the binary value after encoding?
5. What is the remainder obtained by dividing $x^7 + x^5 + 1$ by the generator polynomial $x^3 + 1$?
6. A bit stream 10011101 is transmitted using the standard CRC method described in the text. The generator polynomial is $x^3 + 1$. Show the actual bit string transmitted. Suppose the third bit from the left is

inverted during transmission. Show that this error is detected at the receiver's end.

7. Imagine a sliding window protocol using so many bits for sequence numbers that wraparound never occurs. What relations must hold among the four window edges and the window size, which is constant and the same for both the sender and the receiver.

5.8 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
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Subject Name: Computer Networks **Subject Code:** MCA-301

Lesson No.: 6 **Author:** Mr. Yudhvir Singh

Lesson Title: Network Layer **Vetter: Dr. Manoj Dhun**

Structure

6.0 Objective

In this lesson we will study the design principles for layer 3, the network layer. This study deals with the services, design, protocols for routing and congestion, internetworking concepts and protocols at the network layer. This lesson deals with the network layer, especially routing, with many routing algorithms, both static and dynamic, being covered. Even with good routing algorithms though, if more traffic is offered than the network can handle, congestion can develop, so we discuss congestion and how to prevent it. Connecting heterogeneous networks with inter-networking concepts.

6.1 Introduction

The network layer is concerned with getting packets from the source all the way to the destination. Getting to the destination may require making many hops at intermediate routers along the way. This function clearly contrasts with that of the data link layer, which has the more modest goal of just moving frames from one end of a wire to the other. Thus, the network layer is the lowest layer that deals with end-to-end transmission. To achieve the goal, the network layer must know about the topology of the communication subnet (i.e., the set of all routers) and choose appropriate paths through it. It must also take care to choose routes to avoid overloading some of the communication lines and routers while leaving others idle. Finally, when the source and destination are in different networks, new problems occur. It is up to the network layer to deal with them.

6.2 Design Issues

The network layer has been designed with the following goals:

1. The services should be independent of the subnet technology. Users of the service need not be aware of the physical implementation of the network - for all they know, they're which messages could be transported via carrier pigeon. This design goal has great importance when we consider the great variety of networks in operation. In the area of Public networks, networks in underdeveloped countries are nowhere near the technological prowess of those in the developed countries. The design of the layer must not disable us from connecting to networks of different technologies.

2. The transport layer should be shielded from the number, type, and topology of the subnets presents i.e. all the transport layers want a communication link; it need not to know how that link is made.
3. Finally, there is a need for some uniform addressing scheme for network addresses. The network addresses made available to the transport layer should use a uniform numbering plan even across LANs and WANs.

The design issues also include the service provided to the transport layer and the internal design of the subnet.

- **Store-and-Forward Packet Switching**
- **Services Provided to the Transport Layer**
    - The services should be independent of the router technology.
    - The transport layer should be shielded from the number, type, and topology of the routers present.
    - The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.
- **Implementation of Connectionless Service (Datagram)**
- **Implementation of Connection-Oriented Service (virtual-circuit subnets)**

Comparison of Datagram and virtual-circuit subnets.

With these goals in mind, two different types of service have emerged: (i) Connection oriented and (ii) connectionless. A connection-oriented service is one in which the user is given a "reliable" end-to-end connection. To communicate, the user requests a connection, then uses the connection to his

hearts content, and then closes the connection. A telephone call is the classic example of a connection-oriented service. In a connection-less service, the user simply bundles his information together, puts an address on it, and then sends it off, in the hope that it will reach its destination. There is no guarantee that the bundle will arrive. So - a connection less service is one reminiscent of the postal system. A letter is sent, that is, put in the post box. It is then in the "postal network" where it gets bounced around and hopefully will leave the network in the correct place, that is, in the addressee's letter box. We can never be totally sure that the letter will arrive, but we know that there is a high probability that it will, and so we place our trust in the postal network.

Issue	Datagram subnet	Virtual-circuit subnet
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Routers do not hold state information about connections	Each VC requires router table space per connection
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow it
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Quality of service	Difficult	Easy if enough resources can be allocated in advance for each VC
Congestion control	Difficult	Easy if enough resources can be allocated in advance for each VC

With a connection-oriented service, the user must pay for the length (i.e. the duration) of his connection. Usually this will involve a fixed start up fee. Now, if the user intends to send a constant stream of data down the line, this is great - he is given a reliable service for as long as he wants. However, say the user

wished to send only a packet or two of data - now the cost of setting up the connection greatly overpowers the cost of sending that one packet. Consider also the case where the user wishes to send a packet once every 3 minutes. In a connection-oriented service, the line will thus be idle for the majority of the time, thus wasting bandwidth. So, connection-oriented services seem to be useful only when the user wishes to send a constant stream of data. One would therefore think that the reliable nature of the connection-oriented service would prompt people to choose it over the connectionless service - this is in fact not the case. One can never ensure that the network is 100% reliable; in fact for many applications, we must assume that the network is not reliable at all. With this in mind, many applications perform their own error detection, flow and congestion control at a higher level in the protocol stack, that is, on their own machine, in the transport layer. So, if the sender and the receiver are going to engage in their own control mechanisms, why to put this functionality into the network layer? This is the argument for the connectionless service: the network layer should provide a raw means of sending packets from point A to B, and that is all. Proponents of this argument are quick to point out that the standard of our networks have increased greatly in the past years, that packets of information rarely get lost, so much of the correction facilities in the network layer are redundant and serve only to complicate the layer and slow down transfer.

It is interesting to note here that it is easy to provide a connection oriented service over an inherently connectionless service, so in fact defining the service of the network layer as connectionless is the general solution. However, at the time of defining the network layer, the controversy between the two camps was unresolved, and so instead of deciding on one service, the ISO allows both.

6.3 Internetworking

Multiple networks are a fact of life:

- **Growth**. Individual departments in a university buy LANs for their own machines and eventually want to interconnect with other campus LANs.
- **Fault isolation, geography, and security**. Even when feasible to use one network, an organization can obtain exclusive control over a single local network.

Internetworking deals with the issues of interconnecting multiple networks.

Physical networks can be connected at several levels:

1. Repeaters operate at the physical layer, copying signals from one LAN to another. They operate at the bit level, and have no notion of what the bits mean. Repeaters operating at the physical layer operate at level 1.
2. Bridges operate at the data link layer, copying frames from one LAN to another. They perform store-and-forward packet switching, but use only level-2 (e.g. frame fields) information. Bridges operate at level 2.
3. Routers or gateways operate at the network layer. Because they operate at the network layer, they are fully aware of different network technologies, and can overcome such problems as interconnecting different network technologies. Also known as level 3 routers.

4. Protocol converters operate at higher levels (e.g., ``level 4''). For instance, protocol converters can translate between OSI mail and SMTP (Internet) mail formats.

Bridges

The idea behind bridges is to transparently interconnect LANs. By transparently, we mean that a host treats all other hosts as if they were attached to the local LAN, when in fact, they may be connected to remote LANs.

Bridges operate in promiscuous mode, reading every frame on a LAN, regardless of who it is addressed to. The decision as to where to forward a frame (if at all) is made by looking only at the frame's header i.e. a bridge does not look inside the data portion of the frame (hence, level 2).

Bridges run into immediate difficulties. In particular, frame format and service provided differ for each LAN type:

Maximum frame size:

802.3: 1518 bytes; 802.4: 8191 bytes; 802.5: unlimited. It is simply not possible for an ethernet to carry large frames originating from token ring LANs.

Different operating speeds:

Token rings operate at 4 Mbps, while 802.3 operates at 10Mbps. Bridges will have to buffer frames. When congested, bridges must discard frames.

Differing semantics:

What should the 802.5 ``copy'' bit mean in bridged environments? The sending host may declare a destination down if the copy bit is not set, but is it correct for the bridge to set it? No, because the bridge may be up, although the actual destination host may not be.

In addition, host software that assumes hosts are on the directly connected LAN may fail. For example, if a destination is actually behind several bridges, the round trip delay to the destination may be higher than the sender assumes, and the sender may time out and retransmit too soon.

Priorities:

How should one interpret the priorities of frames on one type of LAN when forwarding across another? CSMA/CD LANs do not support priorities, while priorities mean different things for token ring and token bus networks.

There are two types of bridges on the market today: spanning tree bridges and source routing bridges.

Spanning Tree Bridges

Spanning tree bridges were designed with transparency as a primary goal. A customer should be able to buy a bridge, insert it between two networks, and have everything to work correctly with hardware, software, or configuration changes on either hosts or existing bridges.

1. Each bridge maintains a table that maps destination addresses to the outgoing interface. (Analogous to routing tables in routers.)

2. Bridge operates in promiscuous mode, reading every frame on each of its connected LANs, and the routing decision is made as follows:
    a) Extract the source and destination address from the frame, and find the corresponding table entries for each address.
    b) If the two table entries point to the same interface, discard the frame. Why? If both pointers point to the same interface, both the sender and recipient are on the same local network (as far as we can tell), and the frame doesn't need to be forwarded.
    c) Otherwise, if the two pointers are different, send the frame out on the LAN given by the routing entry for the destination address.
    d) If the destination is not in the table, flood the frame on all interfaces (except the one on which it arrived). We don't know where the destination is, so let's be conservative and send it everywhere. That way we can be sure that the packet traverses the LAN on which the destination resides.
3. Bridges use backward learning to build tables. They determine which LAN to use to reach destination X by recording the interface on which frames having source address X arrive on.
4. The table is a cache; unreferenced entries are periodically flushed, allowing machines to be conneted from one LAN to another.

Use of Spanning Tree Bridges

The above approach works only for tree-structured topologies. Why? Frames will loop forever (there is no time-to-live field in LAN frames) if there are multiple distinct paths between any two bridges.

To handle arbitrary topologies, bridges use a special protocol to build a spanning tree:

1. Bridges that are not part of the spanning tree are unused. That is, they are specifically excluded from the tree and do not forward packets. They are available for backup, however, should one of the other bridges or LANs fail.
2. Bridges periodically rebuild tables. They regularly exchange topology information, allow them to detect the failure of a bridge or LAN. When a bridge or link that is part of the spanning fails, a new spanning tree is constructed.

Advantages:

Easy to use. Just install the bridges. No software changes are needed for hosts.

Disadvantages:

1. It does not support multipath routing. By definition, only the bridges that belong to the spanning tree are used.
2. The path between any two hosts may not be the optimal path. An optimal path may traverse a bridge that is not part of the spanning tree and hence cannot be used.
3. Broadcast and multicast frames must be flooded in all cases.

Source Routing Bridges

Source routing bridges take a completely opposite approach from spanning tree bridges:

1. They are not transparent. Hosts treat frames sent locally differently from those sent through bridges. Conceptually, the sending host specifies a road map saying that through, which of bridges the frame must go to reach its destination.
2. Each LAN is assigned a 16-bit LAN number, and each bridge on a LAN is assigned a 4-bit bridge number. The numbers must be unique and are set by the network administrator.
3. Each frame carries a source route listing the path the frame is to take. The path consists of a sequence of [LAN number, bridge number] pairs.
4. Sending hosts (rather than bridges) are responsible which chooses the source route. Host selects paths by broadcasting (flooding) special discovery frames. A discovery frame includes space for each bridge to add its number to the recorded path.
5. Eventually, a discovery frame reaches the destination host, which returns it to the sender. Each returned message contains a viable path, and the sending host chooses the shortest one.
6. How many frames are generated? Unfortunately, the discovery process leads to frame explosion. The destination may receive an exponential number of copies of the original frame.

Use of Source Routing Bridges

Advantages:

It uses the optimal route.

It also can make use of multiple paths to same destination. Because paths aren't required to always lie along the spanning tree, better use of resources.

Disadvantages:

1. It is not transparent to hosts; i.e. hosts must participate in source routing. This is a significant disadvantage.
2. Installing new bridges are non-trivial. Specifically, a system administrator must assign LAN numbers and bridge numbers. Improper configuration leads to disaster.
3. Each host must detect bridge failure on its own (e.g., using timeouts). With spanning tree bridges, the bridges hold the responsibility, and once they have reconfigured, all hosts start using the new path at the same time.

Gateways/Routers

It can use terms interchangeably or think of routers as within a subnet (same network) versus gateways (between subnets). Text calls gateways multi-protocol routers. Gateways are packet switches that operate at the network layer (level 3). Operating at the network level gives gateways increased flexibility compared to bridges in terms of:

1. Translating addresses between dissimilar networks.
2. Fragmenting large packets for transmission across networks that carry only small maximum packet lengths.
3. Selecting an appropriate path through the subnet.
4. Enforcing policies (e.g., don't forward any local packets off of this network).

Gateways do more work than bridges, however they generally run slower than bridges.

Gateway Ownership

One issue arising with gateways is who owns them. Typically, bridges connect LANs of one organization, and the issue does not arise there. The ownership question is important because someone has to be responsible for the gateway's operation and dual ownership frequently leads to finger pointing when something goes wrong.

One solution is to use half gateways. If two countries are involved, for instance, each country owns its half of the gateway, with a wire separating the two. A special protocol operates over the wire, and each half of the gateway is responsible for implementing the protocol.

Connection vs. Connectionless

Internetworking in a connection-oriented environment operates essentially as in the single network case:

1. The sending host opens a virtual circuit, but a circuit goes through gateway hops.
2. Any two neighboring gateways at the internetworking level must be connected to a common network.
3. Regular router-based virtual circuits connect neighboring gateways on the same physical network).
4. The end-to-end virtual circuit is a concatenation of individual virtual circuits through each of the networks along the path.

Connectionless internets operate just as connectionless networks. A host sends a packet to a neighboring gateway, which forwards it the next gateway, and so forth. Just as with connectionless networks, gateways make only a best-effort attempt at delivering the packet. When a gateway receives a packet, it selects the interface to send the packet out on and encapsulates the

packet using the local data link layer format. As a packet moves from gateway to gateway, it is repeatedly encapsulated and unencapsulated as it travels across each network.

Special case is tunneling between same type of networks across a foreign network(s).

Fragmentation

How does one transmit packets across networks whose Maximum Transmission Unit (MTU) is smaller than the packet being transmitted:

1. Connection-oriented internets avoid this problem. How? By selecting a maximum packet size at connection set up time. Once the connection is established, the path never changes, so the sender can select a packet size and again never worry that it will be too large.
2. In connectionless internets, the appropriate packet size depends on the path used. Thus, it can change at any time.

In general case, setting a minimum MTU for all networks is impractical. Why? A minimum MTU would have to be small, yet sending larger packets should be encouraged for efficiency reasons. Solutions:

1. Have gateway drop packets that are too large to send across a network and return an error message to the sender. The transmitter host could then retransmit the data in a smaller packet.
2. Have gateway fragment large packets into several fragments, each small enough to traverse the network.

Transparent Fragmentation

With transparent fragmentation end hosts i.e. sender and receiver hosts are unaware that fragmentation has taken place. A gateway fragments a packet, and the next-hop gateway on the same network reassembles the fragments back into the original packet.

Drawbacks

1. All fragments must travel through the same gateway. Why? So that they can be reassembled by the next-hop gateway.
2. Gateways must be careful to avoid reassembly lockup. (A gateway has used up all of its buffer space to hold fragments and can no longer accept new ones).
3. Reassembling fragments uses precious gateway resources that could otherwise be used for forwarding packets.

Another approach is to have gateways fragment packets, while hosts perform reassembly (if needed). However, now every host must be prepared to do reassembly.

Problems associated with fragmenting:

1. Fragmenting increases waste: the sum of the bits of the individual fragments exceeds the number of bits in the original message.
2. Loss of a single fragment requires an end-to-end retransmission, i.e., the loss of a single fragment has the same effect as losing the entire packet.
3. More work to forward three small packets than one large one. The cost of forwarding packets includes a fixed per-packet cost, that includes doing the route lookup, fielding interrupts, etc.

Switching Architectures

Gateways (and bridges) are usually special-purpose or dedicated machines as they are used to switch many packets. Consider a bridge between two Ethernets:

1. An Ethernet can carry ≈20,000 (64 byte) frames per second, leaving only 51 μsec to process each packet. That's not a lot of time.
2. If both LANs are operating at maximum capacity, the time drops to 26 μsec, only 1300 instructions on a 50-MIP machine. Of course, a bridge may connect to three of four LANs, making the problem even worse.

   Packet switches quickly become CPU-bound.
3. The need to process packets quickly leads to interesting operating system design issues: minimizing data copying, minimizing interrupt processing and context-switch time, etc.
4. Leads to interesting architecture design issues: dual-ported memory, multiprocessors, etc.

Network Differs

These are some ways that makes the network1 differ from network2.

Item	Some Possibilities
Service offered	Connection oriented versus connectionless
Protocols	IP, IPX, SNA, ATM, MPLS, AppleTalk, etc.
Addressing	Flat (802) versus hierarchical (IP)
Multicasting	Present or absent (also broadcasting)
Packet size	Every network has its own maximum
Quality of service	Present or absent; many different kinds
Error handling	Reliable, ordered, and unordered delivery
Flow control	Sliding window, rate control, other, or none
Congestion control	Leaky bucket, token bucket, RED, choke packets, etc.
Security	Privacy rules, encryption, etc.
Parameters	Different timeouts, flow specifications, etc.
Accounting	By connect time, by packet, by byte, or not at all

The Network Layer in the Internet

The principles that drove Internet design in the past and made it the successful. The basic principles are here.

- Make sure it works.
- Keep it simple.
- Make clear choices.
- Exploit modularity.
- Expect heterogeneity.
- Avoid static options and parameters.
- Think about scalability.
- Look for a good design; it need not be perfect.
- Consider performance and cost.
- Be strict when sending and tolerant when receiving.

The IP Protocol

An IP datagram consists of a header part and a text part. The header has a 20-byte fixed part and a variable length optional part. The header format is shown in figure 6.1.

The Version field keeps track of which version of the protocol the datagram belongs to. Since the header length is not constant, a field in the header, IHL, is provided to tell how long the header is, in 32-bit words. The Type of service field is one of the few fields that intended to distinguish between different classes of service. The Total length includes everything in the datagram—both header and data. The maximum length is 65,535 bytes.

The Identification field is needed to allow the destination host to determine which datagram a newly arrived fragment belongs to. Next comes an unused bit and then two 1-bit fields. DF stands for Don't Fragment. MF stands for More Fragments. The Fragment offset tells where in the current datagram this fragment belongs. The Time to live field is a counter used to limit packet lifetimes. The Protocol field tells it which transport process to give it to. The

Header checksum verifies the header only. The Source address and Destination address indicate the network number and host number.

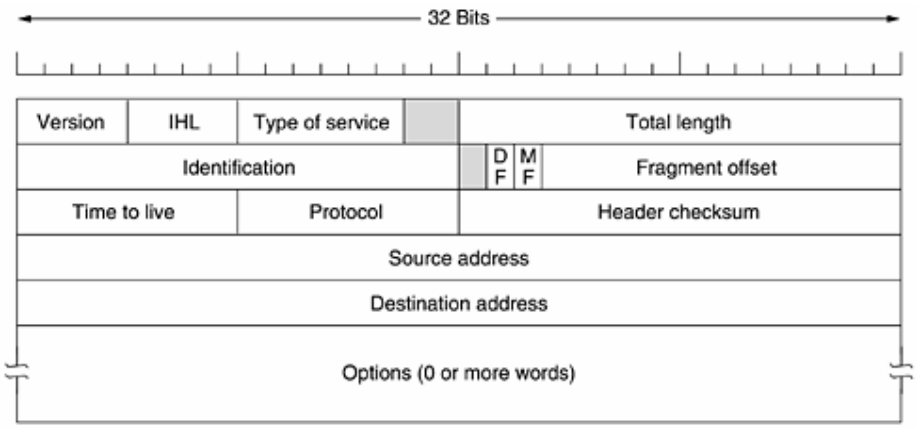


Figure 6.1: The header format of IP Protocol

Some of the IP options

Option	Description
Security	Specifies how secret the datagram is
Strict source routing	Gives the complete path to be followed
Loose source routing	Gives a list of routers not to be missed
Record route	Makes each router append its IP address
Timestamp	Makes each router append its address and timestamp

IP Addressing

Every computer that communicates over the Internet is assigned an IP address that uniquely identifies the device and distinguishes it from other computers on the Internet. An IP address consists of 32 bits, often shown as 4 octets of numbers from 0-255 represented in decimal form instead of binary form. For example, the IP address

168.212.226.204

in binary form is

10101000.11010100.11100010.11001100.

But it is easier for us to remember decimals than it to remember binary numbers, so we use decimals to represent the IP addresses when describing them. However, the binary number is important because that will determine which class of network the IP address belongs to. An IP address consists of two parts, one identifying the network and one identifying the node, or host. The Class of the address determines which part belongs to the network address and which part belongs to the node address. All nodes on a given network share the same network prefix but must have a unique host number as shown in figure 6.2.

Class A Network - binary address start with 0, therefore the decimal number can be anywhere from 1 to 126. The first 8 bits (the first octet) identify the network and the remaining 24 bits indicate the host within the network. An example of a Class A IP address is 102.168.212.226, where "102" identifies the network and "168.212.226" identifies the host on that network.

Class B Network - binary addresses start with 10, therefore the decimal number can be anywhere from 128 to 191. (The number 127 is reserved for loopback and is used for internal testing on the local machine.) The first 16 bits (the first two octets) identify the network and the remaining 16 bits indicate the host within the network. An example of a Class B IP address is 168.212.226.204 where "168.212" identifies the network and "226.204" identifies the host on that network.

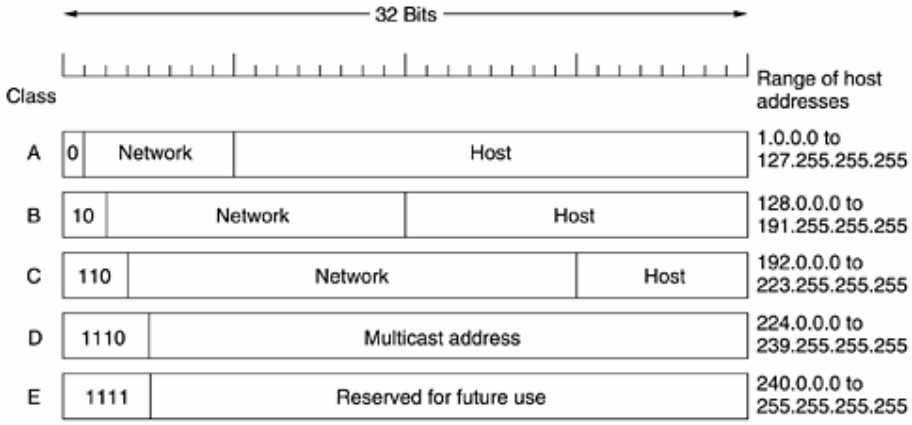


Figure 6.2: Classification of Networks

Class C Network - binary addresses start with 110, therefore the decimal number can be anywhere from 192 to 223. The first 24 bits (the first three octets) identify the network and the remaining 8 bits indicate the host within the network. An example of a Class C IP address is 200.168.212.226 where "200.168.212" identifies the network and "226" identifies the host on that network.

Class D Network - binary addresses start with 1110, therefore the decimal number can be anywhere from 224 to 239. Class D networks are used to support multicasting.

Class E Network -- binary addresses start with 1111, therefore the decimal number can be anywhere from 240 to 255. Class E networks are used for experimentation. They have never been documented or utilized in a standard way.

The IPv6 Protocol

The Version field is always 6 for IPv6 (and 4 for IPv4). During the transition period from IPv4, which will probably take a decade, routers will be able to

examine this field to tell what kind of packet they have. As an aside, making this test wastes a few instructions in the critical path, so many implementations are likely to try to avoid it by using some field in the data link header to distinguish IPv4 packets from IPv6 packets. IPv6 protocol header is shown in figure 6.3. In this way, packets can be passed to the correct network layer handler directly. However, having the data link layer be aware of network packet types completely violates the design principle that each layer should not be aware of the meaning of the bits given to it from the layer above. The discussions between the "Do it right" and "Make it fast" camps will no doubt be lengthy and vigorous.

The Traffic class field is used to distinguish between packets with different real-time delivery requirements.

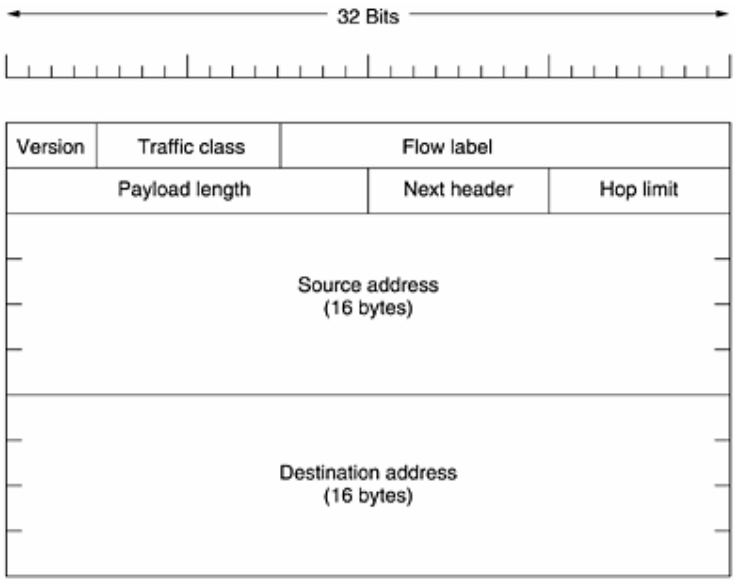


Figure 6.3: The header format of Ipv6 Protocol

The Flow label field is also still experimental but will be used to allow a source and destination to set up a pseudo-connection with particular properties and requirements.

The Payload length field tells how many bytes follow the 40-byte header.

The Next header field lets the cat out of the bag.

The Hop limit field is used to keep packets from living forever.

The Source address and Destination address fields.

IPv6 extension headers.

Extension header	Description
Hop-by-hop options	Miscellaneous information for routers
Destination options	Additional information for the destination
Routing	Loose list of routers to visit
Fragmentation	Management of datagram fragments
Authentication	Verification of the sender's identity
Encrypted security payload	Information about the encrypted contents

6.4 Routing

Routing is the process by which an item gets from one location to another. Many items get routed: for example, mail, telephone calls etc. In networking, a router is the device used to route traffic. Routing is the act of moving information across an internetwork from a source to a destination. Along the way, at least one intermediate node typically is encountered. Routing is often contrasted with bridging, which might seem to accomplish precisely the same thing to the casual observer. The primary difference between the two is that bridging occurs at Layer 2 (the link layer) of the OSI reference model, whereas routing occurs at Layer 3 (the network layer). This distinction provides routing and bridging with different information to use in the process of moving information from source to destination, so the two functions

accomplish their tasks in different ways. The router takes packets (data) from the source and delivers them to destination. The software at network layer is responsible for determining paths from source to destination. This software consists of routing algorithms for determining paths.

Key Information a Router Needs

To be able to route anything, a router, or any other entity that performs the routing, needs to know the following key information:

- **Destination Address** - What is the destination (or address) of the item that needs to be routed?
- **Identifying sources of information** - From which source (other routers) can the router learn the paths to given destinations?
- **Discovering routes** - What are the initial possible routes, or paths, to the intended destinations?
- **Selecting routes** - What is the best path to the intended destination?
- **Maintaining routing information** - A way of verifying that the known paths to destinations are the most current.

The two methods for learning routes through the network are as follows:

Static routes - Routes learned by the router when an administrator manually establishes the route. The administrator must manually update this static route entry whenever an internetwork topology change requires an update. Static routes are user-defined routes that specify the specific path packets moving between a source and a destination take. A default route is a special type of static route. A default route is a route to use for situations when the

route from a source to a destination is not known or when it is unfeasible for the routing table to store sufficient information about the route.

Dynamic Routes - Routes dynamically learned by the router after an administrator configures a routing protocol that helps determine routes. Unlike static routes, once the network administrator enables dynamic routing, route knowledge is automatically updated by a routing process whenever new topology information is received from the internetwork.

The basic Routers use **routing algorithms** to find the best route to a destination. When we say "best route," we consider parameters like the number of **hops** (the trip a packet takes from one router or intermediate point to another in the network), time delay and communication cost of packet transmission. Based on how routers gather information about the structure of a network and their analysis of information to specify the best route, we have two major routing algorithms: global routing algorithms and decentralized routing algorithms. In decentralized routing algorithms, each router has information about the routers it is directly connected to -- it doesn't know about every router in the network. These algorithms are also known as DV (distance vector) algorithms. In global routing algorithms, every router has complete information about all other routers in the network and the traffic status of the network. These algorithms are also known as LS (link state) algorithms.

Routing Components

Routing involves two basic activities: determining optimal routing paths and transporting information groups (typically called packets) through an internetwork. In the context of the routing process, the latter of these is referred to as packet switching. Although packet switching is relatively straightforward, path determination can be very complex.

Path Determination

Routing protocols use metrics to evaluate what path will be the best for a packet to travel. A metric is a standard of measurement, such as path bandwidth, that is used by routing algorithms to determine the optimal path to a destination. To aid the process of path determination, routing algorithms initialize and maintain routing tables, which contain route information. Route information varies depending on the routing algorithm used. Routing algorithms fill routing tables with a variety of information. Destination/next hop associations tell a router that a particular destination can be reached optimally by sending the packet to a particular router representing the "next hop" on the way to the final destination. When a router receives an incoming packet, it checks the destination address and attempts to associate this address with a next hop. Figure 6.4 depicts a sample destination/next hop routing table.

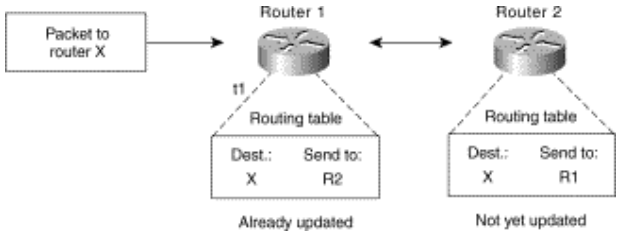


Figure 6.4: Destination/Next Hop Associations Determine the Data's Optimal Path

Routers communicate with one another and maintain their routing tables through the transmission of a variety of messages. The routing update message is one such message that generally consists of all or a portion of a routing table. By analyzing routing updates from all other routers, a router can build a detailed picture of network topology.

Switching

Switching algorithms is relatively simple; it is same for most routing protocols. In most cases, a host determines that it must send a packet to another host. Having acquired a router's address by some means, the source host sends a packet addressed specifically to a router's physical (Media Access Control [MAC]-layer) address, this time with the protocol (network layer) address of the destination host.

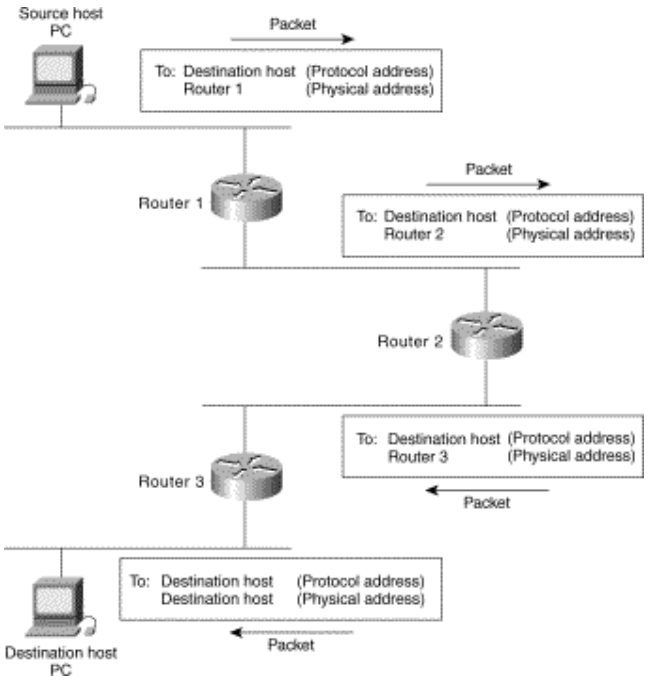


Figure 6.5: Numerous Routers May Come into Play during the Switching Process

As it examines the packet's destination protocol address, the router determines that it either knows or does not know how to forward the packet to the next hop. If the router does not know how to forward the packet, it typically drops the packet. If the router knows how to forward the packet, however, it changes the destination physical address to that of the next hop and transmits the packet. As the packet moves through the internetwork, its physical

address changes, but its protocol address remains constant, as illustrated in figure 6.5.

Routing Algorithms

Routing algorithms can be differentiated based on several key characteristics. First, the particular goals of the algorithm designer affect the operation of the resulting routing protocol. Second, various types of routing algorithms exist, and each algorithm has a different impact on network and router resources. Finally, routing algorithms use a variety of metrics that affect calculation of optimal routes. The following sections analyze these routing algorithm attributes.

Design Goals

Routing algorithms often have one or more of the following design goals:

- Optimality
- Simplicity and low overhead
- Robustness and stability
- Rapid convergence
- Flexibility

Algorithm Types

Routing algorithms can be classified by type. Key differentiators include these:

- Static versus dynamic
- Single-path versus multipath
- Flat versus hierarchical
- Host-intelligent versus router-intelligent
- Intradomain versus interdomain
- Link-state versus distance vector

Static Versus Dynamic

Static routing algorithms and dynamic routing algorithms are already discussed in the above sections.

Single-Path Versus Multipath

Some sophisticated routing protocols support multiple paths to the same destination. Unlike single-path algorithms, these multipath algorithms permit traffic multiplexing over multiple lines. The advantages of multipath algorithms are obvious: They can provide substantially better throughput and reliability. This is generally called load sharing.

Flat Versus Hierarchical

Some routing algorithms operate in a flat space, while others use routing hierarchies. In a flat routing system, the routers are peers of all others. In a hierarchical routing system, some routers form what amounts to a routing backbone. Packets from nonbackbone routers travel to the backbone routers, where they are sent through the backbone until they reach the general area of the destination. At this point, they travel from the last backbone router through one or more nonbackbone routers to the final destination.

Routing systems often designate logical groups of nodes, called domains, autonomous systems, or areas. In hierarchical systems, some routers in a domain can communicate with routers in other domains, while others can communicate only with routers within their domain. In very large networks, additional hierarchical levels may exist, with routers at the highest hierarchical level forming the routing backbone. The primary advantage of hierarchical routing is that it mimics the organization of most companies and therefore supports their traffic patterns well.

Most network communication occurs within small company groups (domains). Because intradomain routers need to know only about other routers within

their domain, their routing algorithms can be simplified, and, depending on the routing algorithm being used, routing update traffic can be reduced accordingly.

Host-Intelligent Versus Router-Intelligent

Some routing algorithms assume that the source end node will determine the entire route. This is usually referred to as source routing. In source-routing systems, routers merely act as store-and-forward devices, mindlessly sending the packet to the next stop. Other algorithms assume that hosts know nothing about routes. In these algorithms, routers determine the path through the internetwork based on their own calculations. In the first system, the hosts have the routing intelligence. In the latter system, routers have the routing intelligence.

Intradomain Versus Interdomain

Some routing algorithms work only within domains; others work within and between domains. The nature of these two algorithm types is different. It stands to reason, therefore, that an optimal intradomain-routing algorithm would not necessarily be an optimal interdomain-routing algorithm.

Link-State Versus Distance Vector

Link-state algorithms (also known as shortest path first algorithms) flood routing information to all nodes in the internetwork. Each router, however, sends only the portion of the routing table that describes the state of its own links. In link-state algorithms, each router builds a picture of the entire network in its routing tables.

Distance vector algorithms (also known as Bellman-Ford algorithms) call for each router to send all or some portion of its routing table, but only to its neighbors. In essence, link-state algorithms send small updates everywhere, while distance vector algorithms send larger updates only to neighboring

routers. Distance vector algorithms know only about their neighbors. Because they converge more quickly, link-state algorithms are somewhat less prone to routing loops than distance vector algorithms. On the other hand, link-state algorithms require more CPU power and memory than distance vector algorithms. Link-state algorithms, therefore, can be more expensive to implement and support. Link-state protocols are generally more scalable than distance vector protocols.

Various Routing Algorithms:

Shortest Path Routing

In Shortest Path Routing, there are two ways in which we can find the shortest path from the source to destination. These are:

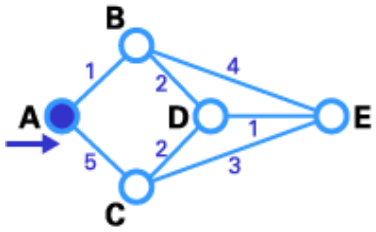
1. On the basis of number of hops used in data transfer.
2. On the basis of distance (in kilometers) among the different hops to reach the destination.

In shortest path Dijkstra algorithm is implemented. In this algorithm, all the nodes are labeled with infinity at the starting. As the algorithm precedes the corresponding node value replaces the infinity with the new value. This label may be changed or made permanent depending on the calculation of shortest path from the initial position to final position.. In starting, these labels are not permanent but when it is termed as the shortest path then this label is made as permanent. This algorithm can be better explained with the help of following example using weighted graph where weight represents the distance from one hop to another.

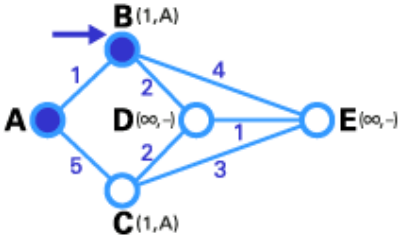
Example: Dijkstra Algorithm

Here we want to find the best route between A and E (see below). You can see that there are six possible routes between A and E (ABE, ACE, ABDE, ACDE, ABDCE, ACDBE), and it's obvious that ABDE is the best route because its weight is the lowest. But life is not always so easy, and there are some complicated cases in which we have to use algorithms to find the best route.

1. As you see in the image below, the source node (A) has been chosen as T-node, and so its label is permanent (we show permanent nodes with filled circles and T-nodes with the --> symbol).

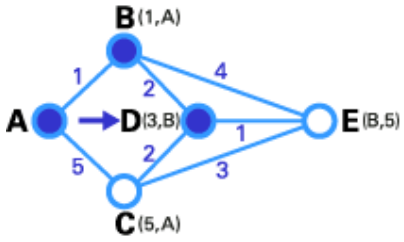


2. In this step, you see that the status record set of tentative nodes directly linked to T-node (B, C) has been changed. Also, since B has less weight, it has been chosen as T-node and its label has changed to permanent (see below).

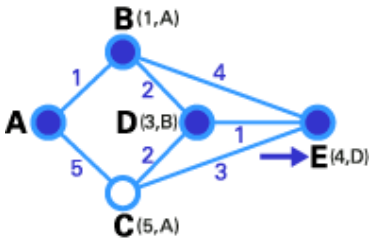


3. In this step, like in step 2, the status record set of tentative nodes that have a direct link to T-node (D, E), has been changed. Also,

since D has less weight, it has been chosen as T-node and its label has changed to permanent (see below).



4. In this step, we don't have any tentative nodes, so we just identify the next T-node. Since E has the least weight, it has been chosen as T-node.



5. E is the destination, so we stop here.

In the end, now we have to identify the route. The previous node of E is D, and the previous node of D is B, and B's previous node is A. So the best route is ABDE. In this case, the total weigh is 4 (1+2+1). Although this algorithm works well, it's so complicated that it may take a long time for routers to process it, and the efficiency of the network fails. Also, if a router gives the wrong information to other routers, all routing decisions will be ineffective.

Flooding

Flooding is a static algorithm, in which every incoming packet is sent to each outgoing line but the packet is not sent on the line from which it comes. Flooding algorithm generates many duplicates of the packet that is to be sent to the destination. A hop counter is placed in the header of each packet that tells the number of copies generated at each hop. The value of hop counter is initialized equal to the length of the path from source to destination. The hop counter is decremented by 1 at each hop. A variation of flooding is selective flooding which is more practical than flooding. In selective flooding the router does not send every incoming packet on every line. It sends only on those lines that are going in right direction. Some uses of flooding are explained as below:

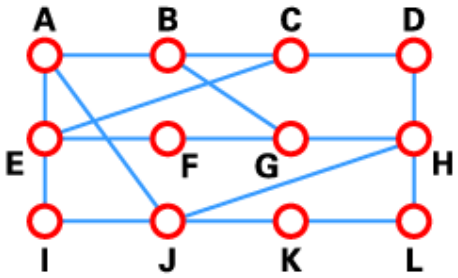
1. In military applications, where large numbers of routers may be blown to bits at any instant, the tremendous robustness of flooding is highly desirable.
2. In distributed database applications, it is sometimes necessary to update all the databases concurrently, in which flooding can be useful.
3. In wireless networks, all messages transmitted by a station can be received by all other stations within the radio range.
4. A fourth possible use of flooding is as a metric against which other routing algorithms can be compared.

Distance Vector Routing

In this each router maintains a Distance Vector to every other router in its network. This vector contains, among other things, the cost of reaching the destination router and the next hop to forward to. Each router starts by exchanging a distance vector to its neighbors and eventually ends of exchanging a distance vector to

all routers in its domain. Obviously, this routing protocol is preferred for small network sizes. Distance vector routing is a dynamic algorithm. In this each router maintains its own table giving the best known path. These tables are updated by exchanging the information from the neighbors. The router is assumed to know the distance to each of its neighbors. If the metric is hops, the distance is just one hop. If the metric is queue length, the router simply examines each queue.

Distance Vector(DV) Algorithm: DV algorithms are also known as Bellman-Ford routing algorithms and Ford-Fulkerson routing algorithms. In these algorithms, every router has a routing table that shows it the best route for any destination. A typical graph and routing table for router J is shown below.



Destination	Weight	Line
A	8	A
B	20	A

C	28	I
D	20	H
E	17	I
F	30	I
G	18	H
H	12	H
I	10	I
J	0	---
K	6	K
L	15	K

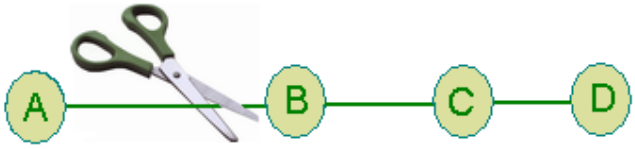
A typical network graph and routing table for router J

As the table shows, if router J wants to get packets to router D, it should send them to router H. When packets arrive at router H, it checks its own table and decides how to send the packets to D.

In DV algorithms, each router has to follow these steps:

1. It counts the weight of the links directly connected to it and saves the information to its table.
2. In a specific period of time, it sends its table to its neighbor routers (not to all routers) and receives the routing table of each of its neighbors.
3. Based on the information in its neighbors' routing tables, it updates its own.

One of the most important problems with DV algorithms is called "count to infinity." Let's examine this problem with an example: Imagine a network with a graph as shown below. As you see in this graph, there is only one link between A and the other parts of the network. Here you can see the graph and routing table of all nodes:



	A	B	C	D
A	0,-	1,A	2,B	3,C
B	1,B	0,-	2,C	3,D
C	2,B	1,C	0,-	1,C
D	3,B	2,C	1,D	0,-

Network graph and routing tables

Now imagine that the link between A and B is cut. At this time, B corrects its table. After a specific amount of time, routers exchange their tables, and so B receives C's routing table. Since C doesn't know what has happened to the link between A and B, it says that it has a link to A with the weight of 2 (1 for C to B, and 1 for B to A -- it doesn't know B has no link to A). B receives this table and thinks there is a separate link between C and A, so it corrects its table and changes infinity to 3 (1 for B to C, and 2 for C to A, as C said). Once again, routers exchange their

tables. When C receives B's routing table, it sees that B has changed the weight of its link to A from 1 to 3, so C updates its table and changes the weight of the link to A to 4 (1 for C to B, and 3 for B to A, as B said).

This process loops until all nodes find out that the weight of link to A is infinity. This situation is shown in the table below. In this way, experts say DV algorithms have a slow convergence rate.

	B	C	D
Sum of weight to A after link cut	∞ ,A	2,B	3,C
Sum of weight to B after 1st updating	3,C	2,B	3,C
Sum of weight to A after 2nd updating	3,C	4,B	3,C
Sum of weight to A after 3rd updating	5,C	4,B	5,C
Sum of weight to A after 4th updating	5,C	6,B	5,C
Sum of weight to A after 5th updating	7,C	6,B	7,C
Sum of weight to A after nth updating	...	...	...
∞	∞	∞	∞

The "count to infinity" problem

One way to solve this problem is for routers to send information only to the neighbors that are not exclusive links to the destination. For example, in this case, C shouldn't send any information to B about A, because B is the only way to A.

Link State Routing

Link state protocols use characteristics of the route such as speed and cost as well as current congestion to determine the best path, which is typically computed by the Dijkstra algorithm. Link state routers are updated from all the routers in the entire network by passing information from router to nearest router. Rather than continuously broadcast its routing tables as does a distance vector protocol, a link state protocol router only notifies its neighboring routers when it detects a change. The basic concept of link-state routing is that every node receives a map of the connectivity of the network, in the form of a graph showing which nodes are connected to which other nodes. Each node then independently calculates the best next hop from it for every possible destination in the network. (It does this using only its local copy of the map, and without communicating in any other way with any other node.) The collection of best next hops forms the routing table for the node. This contrasts with distance-vector routing protocols, which work by having each node share its routing table with its neighbors. In a link-state protocol, the only information passed between the nodes is information used to construct the connectivity maps.

In LS algorithms, every router has to follow these steps:

1. Identify the routers that are physically connected to them and get their IP addresses. When a router starts working, it first sends a "HELLO" packet over network. Each router that receives this packet replies with a message that contains its IP address.
2. Measure the delay time for neighbor routers. In order to do that, routers send echo packets over the network. Every router that receives these packets replies with an echo reply packet. By dividing round trip time by 2, routers can count the delay time. (Round trip time is a measure of the current delay on a network, found by timing a packet bounced off some remote host.) Note that this time includes both transmission and processing times -- the time it takes the packets to reach the destination and the time it takes the receiver to process it and reply.
3. Broadcast its information over the network for other routers and receive the other routers' information. In this step, all routers share their knowledge and broadcast their information to each other. In this way, every router can know the structure and status of the network.
4. Using an appropriate algorithm, identify the best route between two nodes of the network. In this step, routers choose the best route to every node. They do this using an algorithm, such as the Dijkstra shortest path algorithm. In this algorithm, a router, based on information that has been collected from other routers, builds a graph of the network. This graph shows the location of routers in the network and their links to each other. Every link is labeled

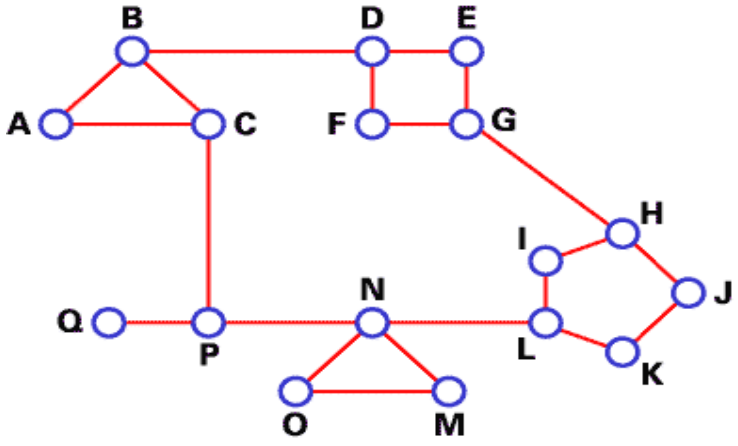
with a number called the weight or cost. This number is a function of delay time, average traffic, and sometimes simply the number of hops between nodes. For example, if there are two links between a node and a destination, the router chooses the link with the lowest weight.

Advantages and disadvantages of link-state routing

The primary advantage of link-state routing is that it reacts more quickly, and in a bounded amount of time, to connectivity changes. The primary disadvantage of link-state routing is that it requires more storage and more computing to run than does distance-vector routing.

Hierarchical Routing

As you see, in both LS and DV algorithms, every router has to save some information about other routers. When the network size grows, the number of routers in the network increases. Here is a typical graph and routing table for A:

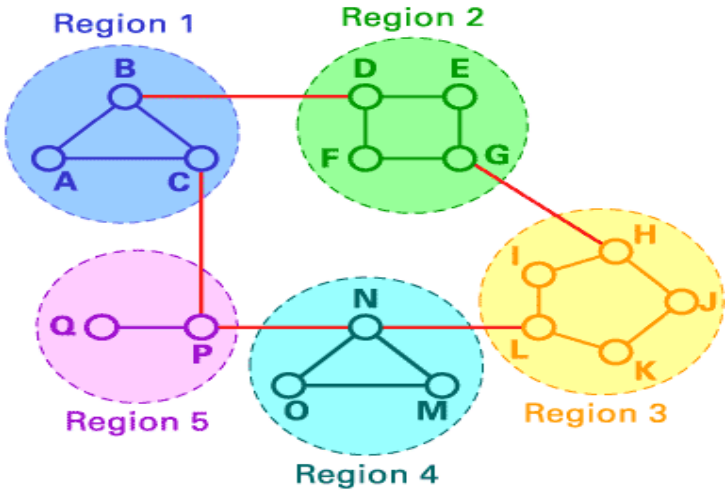


Destination	Line	Weight
A	---	---
B	B	1
C	C	1
D	B	2
E	B	3
F	B	3
G	B	4
H	B	5
I	C	5
J	C	6
K	C	5
L	C	4
M	C	4
N	C	3
O	C	4
P	C	2
Q	C	3

Network graph and A's routing table

Consequently, the size of routing tables increases, as well, and routers can't handle network traffic as efficiently. We use hierarchical routing to overcome this problem. Let's examine this subject with an example: We use DV algorithms to find best routes between nodes. In the situation depicted below, every node of the network has to save a routing table with 17 records.

In hierarchical routing, routers are classified in groups known as regions. Each router has only the information about the routers in its own region and has no



information about routers in other regions. So routers just save one record in their table for every other region. In this example, we have classified our network into five regions (see below).

Destination	Line	Weight
A	---	---
B	B	1

C	C	1
Region 2	B	2
Region 3	C	2
Region 4	C	3
Region 5	C	4

Hierarchical routing

If A wants to send packets to any router in region 2 (D, E, F or G), it sends them to B, and so on. As you can see, in this type of routing, the tables can be summarized, so network efficiency improves. The above example shows two-level hierarchical routing. We can also use three- or four-level hierarchical routing. In three-level hierarchical routing, the network is classified into a number of clusters. Each cluster is made up of a number of regions, and each region contains a number or routers. Hierarchical routing is widely used in Internet routing and makes use of several routing protocols.

6.5 Congestion Control

In a network vulnerable to congestion, the more the total amount of sent packets reaches the maximum capacity, the more the total amount of delivered packets decreases. This can result in a situation where nearly no packets get delivered successfully. The goal of congestion control is to keep network capacity at its maximum level. The following factors responsible for congestion:

• Slow network links

• Shortage of buffer space
• Slow processors

Congestion Control in computer networks covers several mechanisms getting the performance - capacity desirable one and thus prevent the network from congestion. The generic term Congestion Control covers preventive as well as reactive mechanisms.

Congestion Control: Congestion Control means to react on situations when the network is congested.

Congestion Avoidance: Congestion Avoidance covers mechanisms to prevent congestion from happening.

The term congestion always applies to network resources that are congested. A network resource is congested, when the following condition is true:

Total Demand > Available Resources

Congestion Control can be realized by working on either side of the equation, thus by decreasing demand or by increasing available resources. To control congestion in computer networks, it is important to take a look on the architecture and implementation of the network, because any design decision affects the congestion control strategy of the network and thus the demand or the available resources in the previous equation. Those design decisions are called Policies.

One important policy is the connection mechanism. There are two fundamental distinct network types, connection-oriented and connectionless networks. In connection-oriented network,all communication of a connection between two endpoints is routed over the same gateways. In connectionless networks, single packets are routed independently. Therefore, packets belonging to a single connection can be routed differently.

Admission Control as described in below is part of the connection-oriented networks concept, but it is conceptually not a part of connectionless networks. However, state of the art network architectures combine the robustness of the cheaper connectionless networks with the service quality advantages of connection-oriented networks. The Quality of Service (QoS) research area deals with this topic. Congestion Control and resource allocation are two sides of the same coin.

In a congested network, where Total Demand > Available Resources is true.

It is often desirable to allocate the rare resources fair, so that every participant gets an equal share of the networks capacity. Packet queuing and service policies affect this subject. We will introduce Fair Queuing below as a mechanism to ensure fairness for every source. Under the side effect of greater overhead, another concept ensures fairness for every source-destination pair. Packet drop policies are related to the issue of dropping packets when buffer space is to short to

queue more incoming packets. Examples for this are Load Shedding,. and Active Queue Management (AQM).

Transport layer	Round trip delay estimation algorithm Timeout algorithm Retransmission policy Out-of-order packet caching policy Acknowledgement policy Flow control policy Buffer management policy
Network layer	Connection mechanism Packet queuing and service policy Packet drop policy Packet routing policy Lifetime control policy
Data link layer	Data link level retransmission policy Data link level queuing and service policy Data link level packet drop policy Data link level acknowledgement policy Data link level flow control policy

Policies that affects congestion control

Table summarizes policies that affect congestion control mechanisms on different network layers. In this topic we only consider policies from the network and the transport layer.

Congestion can be controlled by increasing available resources. This becomes necessary when network resources are congested over a long period of time. Stronger resources have to be build up, but that is a manual task and not subject to Congestion Control algorithms implemented in network software or hardware.

Three examples to increase available resources:

1. Dial-up links can be added during high usage.

2. Power increases on satellite links to increase their bandwidth.

3. Network paths can be split, so that extra traffic is sent via routes that may not be considered as optimal under low load. This solution is not yet researched well and therefore is not used in practice today. This approach breaks with the concept of connection-oriented networks, because data is no longer routed over the same network path.

Here focus is on Congestion Control algorithms that decrease demand or service. The mechanisms that try to handle congestion by degradation of demand or service can be classified in various ways. The following classification approach:

Open-loop Open-loop solutions for Congestion Control try to solve the problem of congestion by a good design of themselves, thus a once started system does not need to be modified because congestion does not occur anyway. Open loop solutions also includes decisions about admission for connection establishment or packet drops. Open loop solutions do not take the actual network state into account.

Closed-loop In contrast to the open-loop solutions, mechanisms based on the closed-loop use feedback mechanisms to control congestion.

Three steps necessary to control congestion can be distinguished:

1. Observing the system to detect congestion.

2. Informing agents that are able to resolve congestion.

3. Taking action to resolve the problem.

The Open-loop/Closed-loop classification of the Congestion Control mechanisms does not appear to be optimal in our case, because it only allows us to divide the presented algorithms into two categories.

The algorithms are categorized by two dimensions:

1. The network layer on which the algorithms work according to the OSI/ISO model. These can be transport or network layer as well as algorithms working between these layers.
2. The involved network resources. Those can be single endpoints or gateways as well as pairs or groups of them. To give a good introduction into the Congestion Control research area, it is noted for each and every mechanism presented here, if it is an open- or a closed-loop solution.

Traffic Shaping

All network nodes like hosts or gateways, can be sending agents, because they are able to send data to other network nodes. In computer networks, one of the main reasons for congestion is that data is not sent in constant rates but rather in bursts called traffic peaks. Handling these traffic peaks is quite simple. Controlling the sending rate on network participants must flatten traffic peaks. This kind of traffic control is a preventive Congestion Avoidance mechanism called Traffic Shaping.

In networks based on virtual circuits, the transmission rates and quality are negotiated with receiving network nodes when a connection is initialized. A flow specification is the result of this negotiation. Sending agents have to adhere to the flow specification and shape their traffic accordingly. In packet switched networks the volume of traffic is changing quickly over time. This makes periodic renegotiations necessary in those networks. Quality of service negotiation is called Traffic Policing. To control the sending rate of agents and prevent the

receiving agents from data overflow, one of the most established mechanisms on the network layer is the Leaky Bucket algorithm.

Leaky Bucket Algorithm

The Leaky Bucket algorithm is a Traffic Shaping solution, categorized by as an open loop approach. Just imagine a leaky bucket that has a hole on its ground. As you can see in figure 6.6 given below.

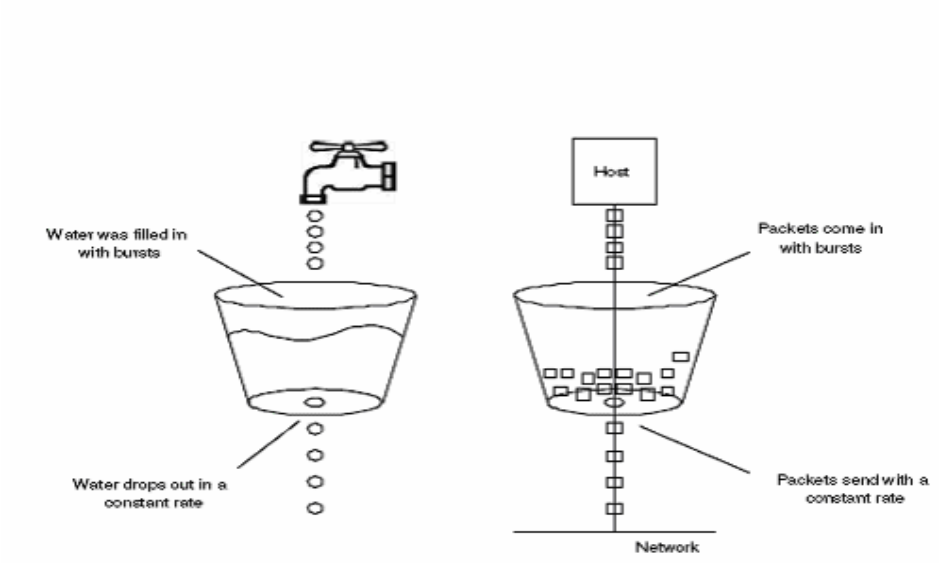


Figure 6.6: Leaky Bucket Algorithm

If the bucket is empty, no water drops out. If there is some water inside, water drops out with a constant rate. If the bucket is full and additional water is filled in, the water overflows the bucket. This can be implemented as follows. On a sender, a time-dependent queue ensures that an equal amount of data units is send per time interval. On the one hand, data can be put fast into the queue until it is full. On the other hand, data always leave the queue at the same rate. If the queue is full,

no more packets can be stored in it and packet loss occurs. If the queue is empty, no data leaves the queue. The Leaky Bucket algorithm can be implemented for packets or a constant amount of bytes, send within each time interval. Using this algorithm, transmission peaks are flattened to a constant transmission rate on a sending agent.

Token Bucket Algorithm

A more flexible approach to control the sending rate on agents is the Token Bucket algorithm, also categorized as an open-loop approach. This algorithm allows bursts for short transmission while making sure that no data is lost. In contrast to the Leaky Bucket algorithm, not the data that is to be send but tokens are queued in a time-depended queue. One token is needed to send a single portion of data.

Implementations contain a token counter that is incremented on every time interval, so that the counter grows over time up until a maximum counter value is reached. The token counter is decremented by one for every data portion sent. When the token counter is zero, no data can be transmitted. An example would be a token counter with a maximum of 50 tokens. When no data is sent for a longer period of time, the counter will reach and stay at its maximum. In this situation, when an application sends a series of 100 data portions through the network interface, 50 portions can be send immediately because there are 50 tokens available. After that, the remaining 50 portions are stored and transmitted one by one on each time interval the token counter gets incremented.

Token Bucket's characteristic influence on the traffic shape is displayed on diagram (c) in figure 6.7, where the sample incoming burst displayed on diagram (a) is processed. First, there are enough buckets available to allow a burst over a while. After that, the traffic is sent with a constant lower rate, because new tokens get available at a constant rate of time.

Token Bucket as well as Leaky Bucket algorithms can be implemented for packets or constant byte amounts. In the case of Token Bucket with a constant amount of bytes for each token, it is possible to store fractions of tokens for later transmissions, when they have not be fully consumed by sending packets,. For example when a 100 byte token is used to send a 50 byte packet, it is possible to store the rest of the token (50 bytes) for later transmissions.

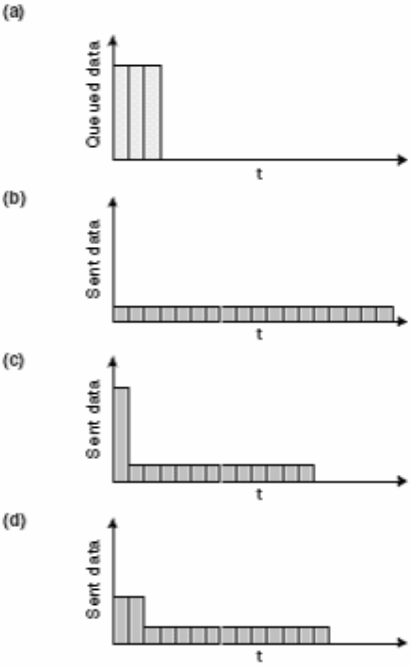


Figure 6.7: Traffic Shaping with Leaky and Token Bucket Algorithms

Congestion Control on gateways

The algorithms in the previous section can work on every node in the network. Now we will describe some algorithms applied on routers or other gateways to avoid congestion.

Load Shedding

An open-loop approach to control congestion on routers is the Load Shedding mechanism. Load Shedding is a way to instruct a router to drop packets from its queue, if it is congested. Note that the Load Shedding mechanism only reacts on situations, where the router is already congested and no more packets can be stored in the queue.

Other queue management solutions like Active Queue Management (AQM) to monitor the queue state and take actions to prevent the router from congestion. When dropping packets, the router has to decide which packets should be dropped. There are two concepts for dropping packets. On the one hand the router can drop new incoming packets. This approach can be optimal for file transmissions. Consider a router that has packets 7, 8 and 9 of a 10 packets long file in its full queue. Now packet 10 arrives. Assuming a receiving host drops packets that arrive out of sequential order, the worst decision would be to drop packet 7, as it would lead to retransmission of packets 7 to 10. Dropping packet 10 would be optimal, as it would be the only packet that would have to be retransmitted. On the other hand for some multimedia and realtime applications, the value of new packets is higher than the value of old packets. In those cases the router can be instructed to drop old packets. The concept of dropping new packets is called wine, dropping old packets is called milk. Wine is implemented as tail-drop on queues, milk is implemented as head-drop.

Fair Queuing

As they say Congestion Control and resource allocation are two sides of the same coin. To enforce hosts and routers to implement and use congestion control and avoidance algorithms described here, an open-loop mechanisms called Fair Queuing was introduced. Imagine hosts and routers connected to a router that reduces their data rates according to congestion control algorithms. A connected misbehaving sending host that does not reduce its data rates according to congestion

control algorithms will overflow the router and use more resources than well behaving hosts. To resolve this issue, a FIFO queue is introduced on every incoming line as shown in figure 6.8.

Packets to deliver are chosen from all queues by a round robin algorithm. This ensures, that hosts or routers that misbehave do not use more resources and get a poorer service than they would, if they behaved well. Packets from misbehaving hosts will be dropped when they overflow their incoming FIFO queue on a router.

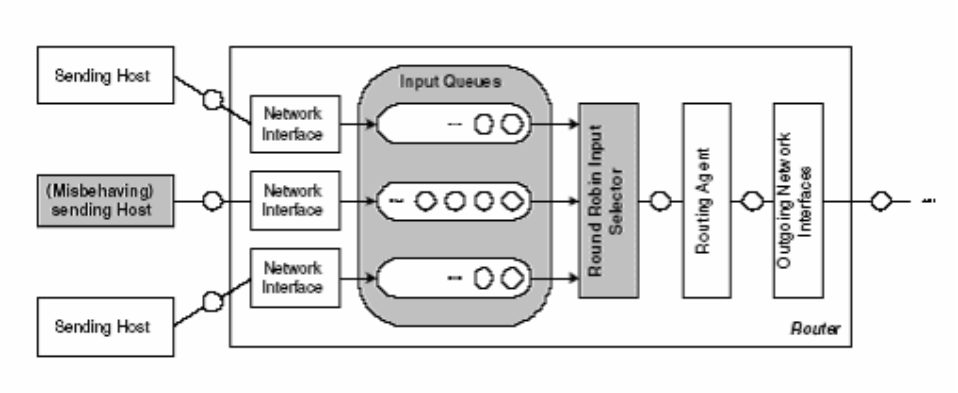


Figure 6.8: Fair Queuing Example

Congestion Control between network nodes

This section shows some algorithms, working on the network layer between network nodes. Therefore communication is needed, that is not provided by the IP protocol.

Admission Control

Generally used in closed-loop, Congestion Avoidance mechanism is Admission Control. Admission Control manages the establishment of connections between connection endpoints in connection-oriented

networks. When a connection is established, the connected endpoints can rely on the expected service quality, as congestion cannot occur. On the Internet, Admission Control cannot be applied as it is working connectionless.

In connection-oriented networks, the transmissions belonging to a connection are always routed over the same network nodes. When a host tries to establish a connection, the virtual circuit has to be build up by all involved network nodes that have the chance to decide, if theycan handle an additional connection or not. Answering this kind of question is not trivial, since it requires the knowledge of the current load on nodes as well as the expected additional load that results from the additional connection. Clients must declare their demand of service quality to aid this decision on the involved network nodes. For clients, this approach seems to be a bit restrictive. However, it is a very stable solution. In networks where connection establishment is only possible when the network load allows it, clients get a guarantee that the requested quality of service is fulfilled by the network. In such networks, there is no congestion during normal operation. A problem is to ensure that all participants follow the policies they advertised after the connections are established. If they use more service than they requested, the network can be congested. In connection-oriented networks, the network nodes have to save information about every connection, so it is possible for them to check, if clients do not exceed their contracts. If not, they should decrease their service to the bad-behaving clients accordingly.

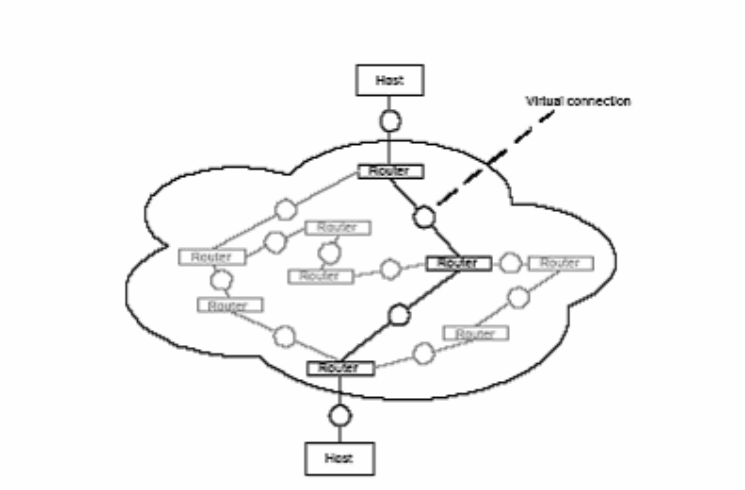


Figure 6.9: Admission control in computer networks

Examples of connection-oriented networks using admission control to avoid congestion are telephone networks or Asynchronous Transfer Mode (ATM) networks. As Figure 6.9 shows admission control in computer networks.

Resource Reservation Protocol (RSVP)

On the connectionless working internet, based on the IP protocol, some application require a certainty about the Quality of Service (QoS) they get from the network. A video stream, for example, needs a constant data rate as well as constant delivery time of packets. The Resource reSerVation Protocol (RSVP) is an extension protocol, conceptually working on the network layer that allows clients to reserve resource on the network path for later transmission of data and thus it is often used for multicast applications, but it can also be used in unicast scenarios.

RSVP is used by receiving hosts of downstream-only simplex connections to reserve network resources along the route. Unicast connections are one to one connections where multicast connections can be either one to many or many to many connections. Multicast is a method to optimize traffic volume for scenarios where senders must send the same data to many receivers.

In these scenarios, every router along the path can forward a single stream to many receiving clients. This ensures that the data is transmitted only once along every sender-receiver route. This minimizes network traffic in contrast to one to one connections. The routes are shaped as trees with the sending host as the root of the tree, routers being internal nodes and the leafs being receiving clients. RSVP works as follows:

• When a client wants to establish a connection, it first issues an RSVP service request in the upstream direction.

• The first router checks, if the requested service quality can be provided. The requested service can be provided if the router has got enough capacity to forward a stream to the receiver. If not, the router sends a negative acknowledgement to the receiver. If yes, the router must distinguish multicast and unicast cases as described next.

• In multicast scenarios, a router must check if he has already got a reservation for the same stream. If yes, he can signal free admission to the sender. If the router does not have a reservation for the same stream yet, he must issue an RSVP request to the next router on the

route to the sender. If this RSVP request is positively acknowledged, the router sends an acknowledgement to the client.

• In unicast scenarios, a router must always issue a new RSVP request in direction of the sender. If this RSVP request is positively acknowledged, the router sends an acknowledgement to the client.

• Every router on the route works the same as the first router being contacted by the receiving client. As internet traffic volume is changing very quickly over time, this reservation process must be repeated in intervals. When the client does not re-request a reservation for a while, the reservation times out.

Choke Packets

Choke Packets are a closed-loop solution that is used to directly inform a sending host about congestion on a network resource along the network path. The receiver should then take action to reduce load. When deciding about issuing Choke Packets, relevant resources must be monitored. It is also important not to issue too many choke packets, to give the receiver the chance to react. Sending of Choke Packets is often criticized because additional packets may increase the problem when the network is already congested. In IP-based networks there is a possibility to send Choke Packets via Internet Control Message Protocol (ICMP) source quench messages.

Hop-by-Hop Choke Packets

One problem with Choke Packets that are addressed to a sending host is the long time that it takes to resolve the congestion. When the Choke Packet is issued, there may be still a lot of packets on the way from the

sender to the congested resource. To resolve the situation on the congested resource more quickly, all network nodes along the path should interpret and react immediately on a Choke Packet that passes their way.

6.6 Summary

The network layer provides services to the transport layer. It can be based on either virtual circuits or datagrams. In both cases, its main job is routing packets from the source to the destination. In virtual-circuit subnets, a routing decision is made when the virtual circuit is set up. In datagram subnets, it is made on every packet. Many routing algorithms are used in computer networks. Static algorithms include shortest path routing and flooding. Dynamic algorithms include distance vector routing and link state routing. Most actual networks use one of these. Subnets can easily become congested, increasing the delay and lowering the throughput for packets. Network designers attempt to avoid congestion by proper design. Networks differ in various ways, so when multiple networks are interconnected problems can occur. Sometimes the problems can be finessed by tunneling a packet through a hostile network, but if the source and destination networks are different, this approach fails. Protocols described are IP, a new version of IP i.e. IPv6.

6.7 Self Assessment Questions

1. Give two example of computer applications for which connection-oriented service is appropriate. Now give two examples for which connectionless service is best.

2. For hierarchical routing with 4800 routers, what region and cluster sizes should be chosen to minimize the size of the routing table for a three-layer hierarchy? A good starting place is the hypothesis that a solution with k clusters of k regions of k routers is close to optimal, which means that k is about the cube root of 4800 (around 16). Use trial and error to check out combinations where all three parameters are in the general vicinity of 16.
3. A datagram subnet allows routers to drop packets whenever they are needed to. The probability of a router discarding a packet is p. Consider the case of a source host connected to the source router, which is connected to the destination router, and then to the destination host. If either of the routers discards a packet, the source host eventually times out and tries again. If both host-router and router-router lines are counted as hops, what is the mean number of
    a. hops a packet makes per transmission?
    b. transmissions a packet makes?
    c. hops required per received packet?
4. Give an argument why the leaky bucket algorithm should allow just one packet per tick, independent of how large the packet is.
5. The byte-counting variant of the leaky bucket algorithm is used in a particular system. The rule is that one 1024-byte packet, or two 512-byte packets, etc., may be sent on each tick. Give a serious restriction of this system that was not mentioned in the text.
6. A computer on a 6-Mbps network is regulated by a token bucket. The token bucket is filled at a rate of 1 Mbps. It is initially filled to capacity with 8 megabits. How long can the computer transmit at the full 6 Mbps?

7. Imagine a flow specification that has a maximum packet size of 1000 bytes, a token bucket rate of 10 million bytes/sec, a token bucket size of 1 million bytes, and a maximum transmission rate of 50 million bytes/sec. How long can a burst at maximum speed last?
8. Describe a way to reassemble IP fragments at the destination.
9. IPv6 uses 16-byte addresses. If a block of 1 million addresses is allocated every picosecond, how long will the addresses last?
10. The Protocol field used in the IPv4 header is not present in the fixed IPv6 header. Explain why?

6.8 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
- Computer Network, ED Tittle, Tata MacGraw Hills Publications

Subject Name: Computer Networks **Subject Code:**MCA-301

Lesson No.: 7 **Author:** Mr. Yudhvir Singh

Title of the Lesson: Transport Layer **Vetter: Dr. Manoj Dhun**

Structure

7.0 Objective

In this lesson we will study the design principles for layer 4, the transport layer. This study deals with the services, design, and elements protocols for achieving reliable, efficient communication between two adjacent machines/peers at the transport layer. Protocols discussed are UDP, TCP, connection management etc.

7.1 Introduction

Protocols at the transport layer oversee the delivery of data from a process (a running application program) on one computer to a process on another computer. These transport layer protocols act as liaison or

interface between the application layer protocols and the services provided by lower layers (network, data link, and physical). The application layer programs interact with each other, using the services of the transport layer, without even being aware of the existence of the lower layers. In effect, the application layer programs are oblivious to the intricacies of the physical network and are not dependent on the physical network type.

The figure 7.1 shows the position of the transport layer in the OSI model. The upper three layers (application, transport, and session) have been grouped together as one application layer, as in the Transmission Control Protocol/Internet Protocol (TCP/IP) model. The transport layer receives services from the network layer and provides services for the application layer:

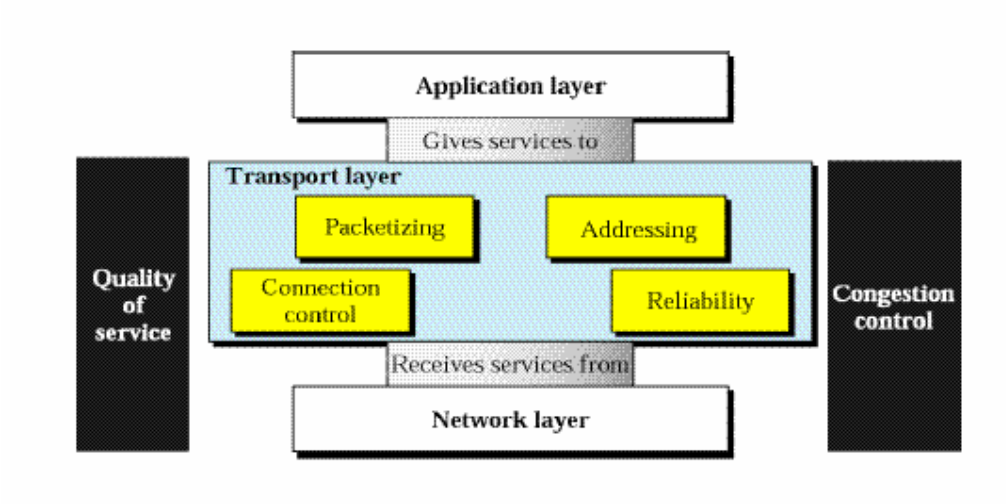


Figure 7.1: Transport Layer in ISO-OSI Reference Model

The transport layer hides details of any network-dependent information from the higher layers by providing transparent data transfer. The transport layer is concerned with machine-to-machine transfer rather than process-to-process. The transport layer will split data into smaller packets dispatch those packets and at the receiver reassemble the data in the correct sequence.

7.2 Services

The ultimate goal of the transport layer is to provide efficient, reliable, and cost-effective service to its users, normally processes in the application layer. To achieve this goal, the transport layer makes use of the services provided by the network layer. The hardware and/or software within the transport layer that does the work is called the transport entity. The transport entity can be located in the operating system kernel, in a separate user process, in a library package bound into network applications, or conceivably on the network interface card. The (logical) relationship of the network, transport, and application layers is illustrated in figure 7.2.

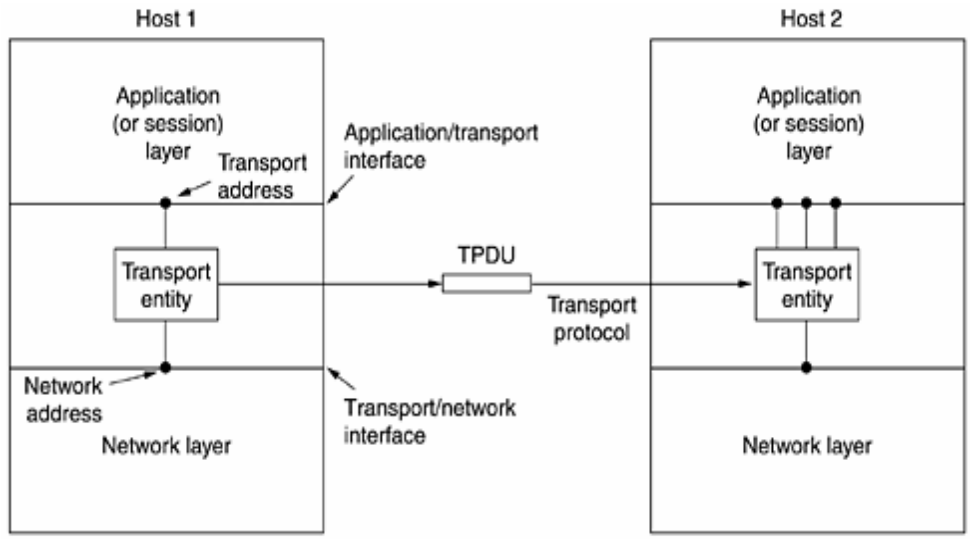


Figure 7.2: Logical – Relationship between network, transport and application layers

Just as there are two types of network service, connection-oriented and connectionless, there are also two types of transport services.

Connectionless versus Connection-Oriented Service

A transport-layer protocol can be either connectionless or connection-oriented.

Connectionless Service

In a connectionless service, the packets are sent from one party to another with no need for connection establishment or connection release. The packets are not numbered, so they may be delayed, lost, or arrive out of sequence. There is no acknowledgement either. One of the transport-layer protocols, User Datagram Protocol (UDP), is connectionless.

Connection-Oriented Service

In a connection-oriented service, a connection is first established between the sender and the receiver. Data are then transferred. At the end, the connection is released. Transmission Control Protocol (TCP) is a connection-oriented service.

Reliable versus Unreliable

The transport-layer service can be reliable or unreliable. If the application-layer protocol needs reliability, a reliable transport-layer protocol is used to implement flow and error control. This means a slower and more complex service. However, if the application program does not need reliability because it uses its own flow and error

control mechanism or if it needs fast service or the nature of the service does not demand flow and error control (e.g. real time applications), an unreliable protocol can be used. There are two different transport-layer protocols. UDP is connectionless and unreliable; TCP is connection-oriented and reliable.

7.3 Elements of Transport Protocols

TPDU (Transport Protocol Data Unit) for messages sent from transport entity to transport entity. Thus, TPDUs (exchanged by the transport layer) are contained in packets (exchanged by the network layer) as shown in figure 7.3. In turn, packets are contained in frames (exchanged by the data link layer). When a frame arrives, the data link layer processes the frame header and passes the contents of the frame payload field up to the network entity. The network entity processes the packet header and passes the contents of the packet payload up to the transport entity.

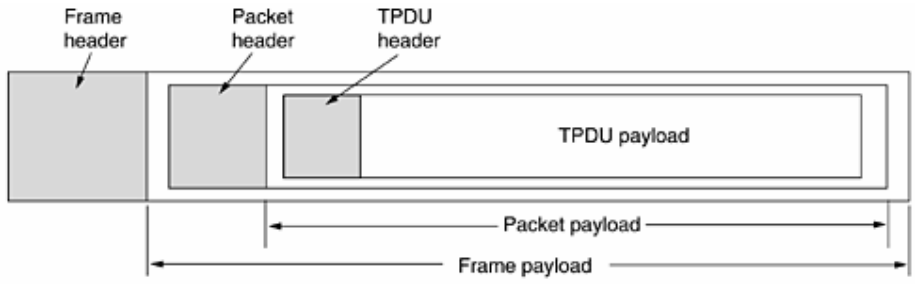


Figure 7.3: Nesting of TPDUs, packets, and frames

The transport layer may provide the following services to the session layer:

The establishment and release of transport connections

This is done by means of transport addresses and transport service data units (TSDUs). Each session entity has a transport address

associated with it and provide connections between transport addresses. There may be more than one transport connection between two endpoints. One transport connection may support more than one session- entity.

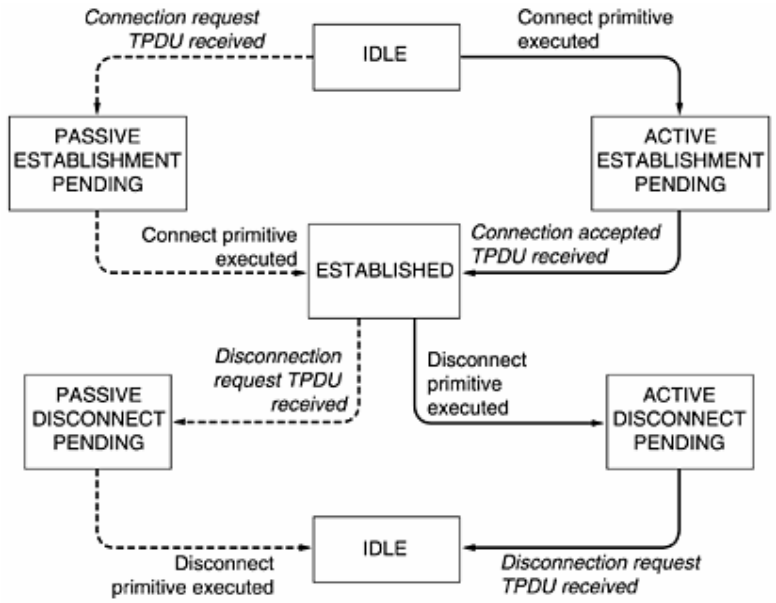


Figure 7.4: State diagram of Connection establishment & release.

A state diagram for connection establishment and release for these simple primitives is given in figure 7.4. Each transition is triggered by some event, either a primitive executed by the local transport user or an incoming packet. For simplicity, we assume here that each TPDU is separately acknowledged. We also assume that a symmetric disconnection model is used, with the client going first. Please note that this model is quite complicated.

Data transfer is in accordance with the required quality of service. The transport layer attempts to provide the quality of service required by its associated session-entity using the service provided by the network layer. To achieve these the transport layer need to incorporate the following functions:

Three-way handshake Connection Establishment

This establishment protocol does not require both sides to begin sending with the same sequence number, so it can be used with synchronization methods other than the global clock method. The normal setup procedure when a host 1 initiate is shown in figure 7.5. Host 1 chooses a sequence number, x, and sends a CONNECTION REQUEST TPDU containing it to host 2. Host 2 replies with an ACK TPDU acknowledging x and announcing its own initial sequence number, y. Finally, host 1 acknowledges host 2's choice of an initial sequence number in the first data TPDU that it sends.

Connection Release

There are two styles of terminating a connection: asymmetric release and symmetric release. Asymmetric release is the way similar to the telephone system works: when one 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 release is abrupt and may result in data loss. 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 situations, determining that all the work has been done and the connection should be terminated is not so obvious.

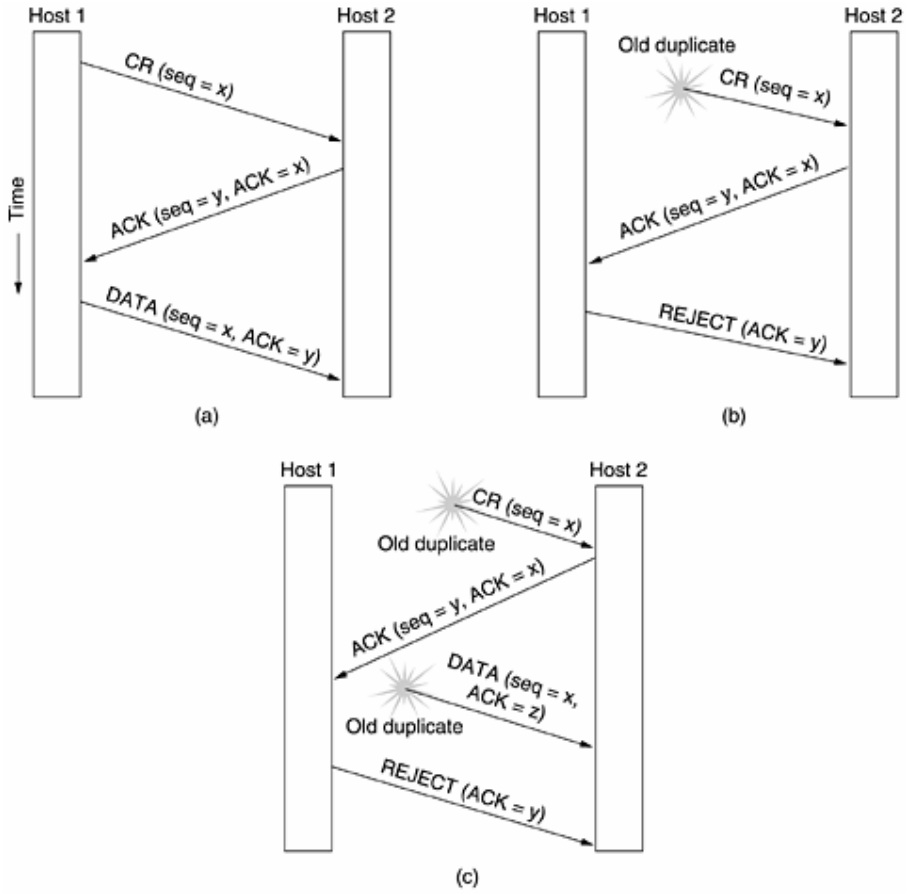


Figure 7.5: Three protocol scenarios for establishing a connection using a three-way handshake. CR denotes CONNECTION REQUEST. (a) Normal operation. (b) Old duplicate CONNECTION REQUEST appearing out of nowhere. (c) Duplicate CONNECTION REQUEST and duplicate ACK.

Flow Control and Buffering

In some ways the flow control problem in the transport layer is the same as in the data link layer, but in other way it is different. The basic similarity is that in both layers a sliding window or other scheme is needed on each connection to keep a fast transmitter from overrunning slow receiver. The main difference is that a router usually has relatively few lines, whereas a host has numerous connections. This difference makes it impractical to implement the data link buffering strategy in the transport layer.

The subnet provides datagram service, the sending transport entity must also buffer. The receiver knows that the sender buffers all TPDUs until they are acknowledged, the receiver may or may not dedicate specific buffers to specific connections, as it sees fit. The receiver may maintain a single buffer pool shared by all connections. When a TPDU comes in, an attempt is made to dynamically acquire a new buffer. If one is available, the TPDU is accepted; otherwise, it is discarded. Since the sender is prepared to retransmit TPDUs lost by the subnet, having the receiver drop TPDUs, although some resources are wasted, does no harm. The sender just keeps trying until it gets an acknowledgement.

Even if the receiver has agreed to do the buffering, there still remains the question of the buffer size. If most TPDUs are nearly the same size, it is natural to organize the buffers as a pool of identically sized buffers, with one TPDU per buffer. However, if there is wide variation in TPDU size, from a few characters typed at a terminal to thousands of characters from file transfers, a pool of fixed-sized buffers presents problems. If the buffer size is chosen equal to the largest possible TPDU, space will be wasted whenever a short

TPDU arrives. If the buffer size is chosen less than the maximum TPDU size, multiple buffers will be needed for long TPDUs, with the attendant complexity.

The optimum trade-off between source buffering and destination buffering depends on the type of traffic carried by the connection. For low-bandwidth bursty traffic, such as that produced by an interactive terminal, it is better not to dedicate any buffers, but rather to acquire them dynamically at both ends. Since the sender cannot be sure the receiver will be able to acquire a buffer, the sender must retain a copy of the TPDU until it is acknowledged. On the other hand, for file transfer and other high-bandwidth traffic, it is better if the receiver dedicate a full window of buffers, to flow the data at maximum speed. Thus, for low-bandwidth bursty traffic, it is better to use buffer at the sender end, and for high bandwidth smooth traffic, it is better to use buffer at the receiver end.

Multiplexing

Multiplexing is used for several conversations on one connection, virtual circuits, and physical links plays a role in several layers of the network architecture. If only one network address is available on a host, all transport connections on that machine have to use it. When a TPDU comes, some way is needed to process it, the way for it to process is given, this situation, called upward multiplexing. If a user needs more bandwidth than one virtual circuit, a way out is to open multiple network connections and to distribute the traffic among them on a round-robin basis; this modus operandi is called downward multiplexing.

Crash Recovery

If hosts and routers subject to crashes, recovery from these crashes becomes an issue. If the transport entity is entirely within the hosts, recovery from

network and router crashes is straightforward. If the network layer provides datagram service, the transport entities expect lost TPDUs all the time and know how to cope with them. If the network layer provides connection-oriented service, then loss of a virtual circuit is handled by establishing a new one and then probing the remote transport entity to ask it which TPDUs it has received and which ones it has not received. The latter ones can be retransmitted.

Transport address to network address mapping: Transport connections are end-to-end. These are mapped onto network addresses that identify the endpoints. One transport entity may serve more than one session entity and several transport addresses may be associated with one network address.

Multiplexing and splitting of transport connections: A number of transport connections may be carried on one network connection and a transport connection may be split between a numbers of different network connections.

Also the layer will provide the following functions -

- Sequencing
- Flow Control
- Error detection and recover

Process-to-process delivery

Process to process delivery is achieved through a set of functions performed by the transport layer. The most important are packetizing, connection control, addressing, and providing reliable. These duties are shown in the figure 7.6:

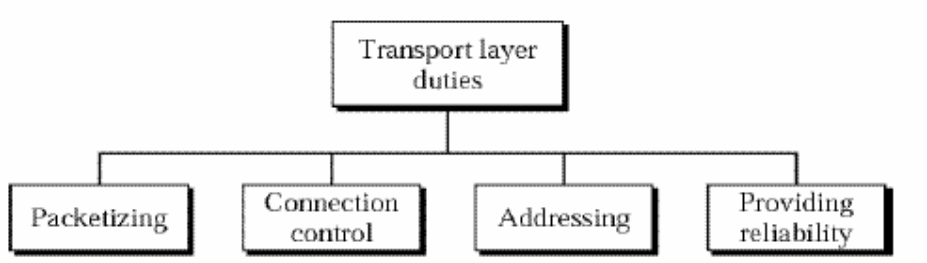


Figure 7.6: Functions / Duties of Transport Layer

Packetizing: Packetizing divides a long message into smaller ones. These are then encapsulated into the data field of the transport-layer packet and after that headers are added.

Connection Control: Transport-layer protocols are divided into two categories: connection-oriented and connectionless.

Connection-Oriented Delivery: A connection-oriented transport layer protocol first establishes a connection (a virtual path) between the sender and receiver. The connection is virtual in that the application layer perceives that a single path has been created. In reality the packets may travel through different physical paths. A session is said to exist between sender and receiver and the session remains intact until broken by one of the parties. Once the session is established, the two parties can send multiple packets related to each other, travelling one after another through the virtual path. The packets may travel out of order, but the transport layer ensures that this is transparent to the application program. The packets are numbered consecutively and communication can take place in both directions.

Connectionless Delivery: A connectionless transport protocol treats each packet independently without any connection between them.

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 called ports. In Asynchronous Transfer Mode (ATM) networks, they are called ATM adaptation Layer – Service Access Points (AAL-SAPs). We will use the generic term TSAP, (Transport Service Access Point). The analogous end points in the network layer are then called Network Service Access Points (NSAPs). IP addresses are examples of NSAPs.

Consider, for example, that an HTTP client (a browser) on a local computer needs to send a request to an HTTP server on a remote computer. First, the client needs a unique address of the remote computer and addressing is implemented at the network layer. The remote computer may though be running several programs, such as HTTP, SMTP, and TELNET, at the same time. When the request arrives at the remote computer it must be delivered to the correct server program, in our example to the HTTP server program. The packet carrying the request must therefore specify to which server program the request must be delivered. The request packet must also specify the client program that sends the packet and the server uses this when it responds to the request. The reason for this is that the local computer might also be running several clients, since most people usually open several client programs at the same time.

Providing Reliability: The transport layer can provide reliability for the application program that uses its services. Reliability involves flow control and error control.

The data-link layer is responsible for delivery of frames between two neighboring nodes over a link. This is called node-to-node delivery. The network layer is responsible for delivery of datagrams between two hosts. This is called host-to-host delivery. However, real communication on the Internet actually takes place between two processes (application programs) and for this we need process-to-process delivery. At any time there may be several processes running at both the source host and the destination host. To complete the delivery, a mechanism is needed to deliver data from one of the processes running on the source host to the corresponding process running on the destination host. The transport layer is responsible for process-to-process delivery, the delivery of a packet, part of a message, from one process to another. Two processes communicate in a client-server relationship.

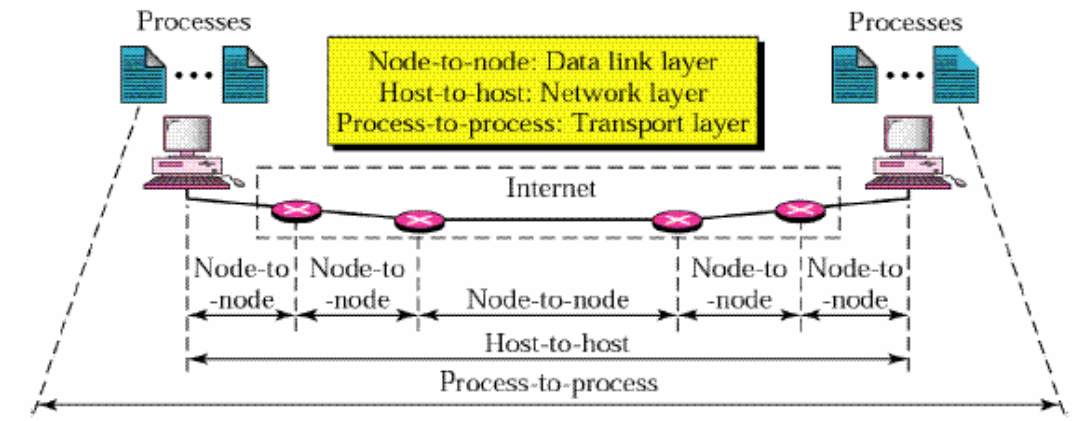


Figure 7.7: Deliveries Type and their Domains

The figure 7.7 shows the three types of deliveries and their domains:

Client-Server Paradigm

Although there are several ways to achieve process-to-process communication, the most common one is through the client-server paradigm. A process on the local host, called a client, needs services from a process usually on a remote host, called a server. A remote computer can run several server programs simultaneously as can a local computer can run several client programs. For communication, the following are therefore defined:

- Local host
- Local process
- Remote host
- Remote process

Addressing

Whenever something needs to be delivered to one specific address amongst many an address is needed. At the data link layer, a MAC address is needed to choose between nodes. A frame in the data link layer needs a destination MAC address for delivery and a source MAC address for the next node's reply. At the network layer, an IP address is needed to choose one host from among millions. A datagram in the network layer needs a destination IP address for delivery and a source IP address for the destination's reply. At the transport layer, a transport-layer address is needed, called a port number, to choose among multiple processes running on the destination host. The destination port number is needed for delivery and the source port number is needed for the reply. In TCP/IP, port numbers are 16-bit integers between 0 and 65,535. The

client program defines itself with a port number, chosen randomly by the transport layer software running on the client host. This is the ephemeral port number. The server program must also define itself a port number, but this cannot be chosen randomly. The Internet has decided to use universal port numbers for servers, called well-know port numbers. Every client knows the well-known port number of the corresponding server process. Consider an example of a Daytime client process running on a local host and a Daytime server process running on a remote host. The Daytime client process can use an ephemeral (temporary) port number 52,000 to identify itself and the daytime server process must use the well-know port number 13. The figure 7.8 illustrates this concept:

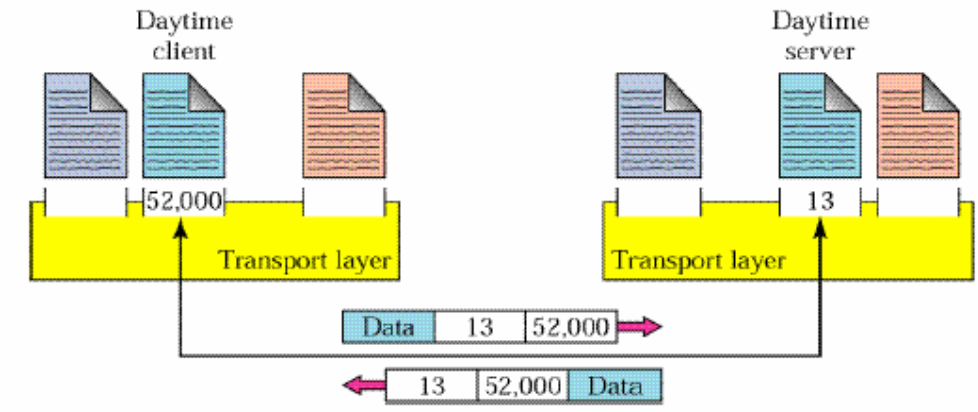


Figure 7.8: Addressing at Transport Layer

IP addresses and port addresses play different roles in selecting the final destination of data. The destination IP address defines the host among the different hosts in the world. After the host has been selected, the port number defines one of the processes on this particular host. This is illustrated in the figure 7.9:

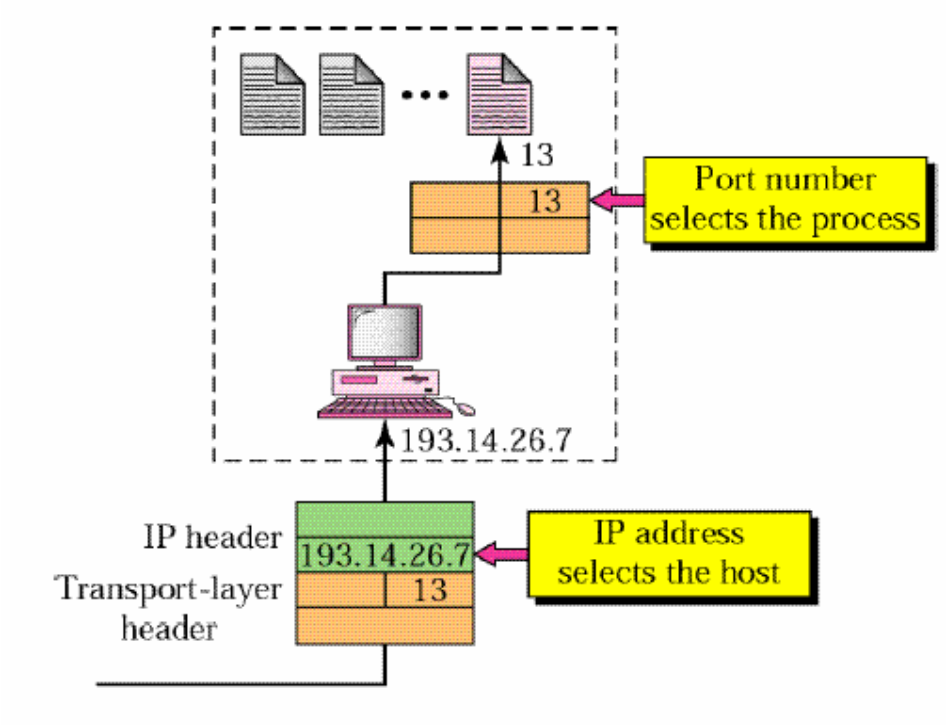


Figure 7.9 IP & Port Addressing Concept

IANA Ranges

The IANA (Internet Assigned Number Authority) has divided the port numbers into three ranges: well known, registered, and dynamic (or private), as show in the figure 7.10:

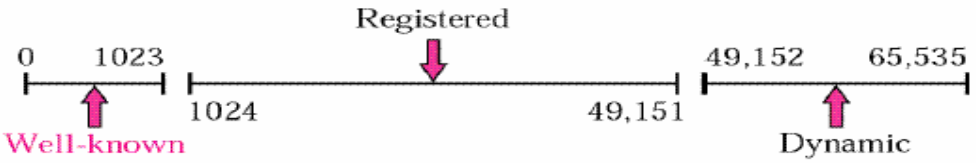


Figure 7.10: IANA Ranges for Port Numbers.

Socket Addresses

Process-to-process delivery needs two identifiers, IP address and the port number, at ach end-to-end connection. The combination of an IP address and a port number is called a socket address. The client socket address defines the client process uniquely just as the server socket address defines the server process uniquely as shown in figure 7.11.

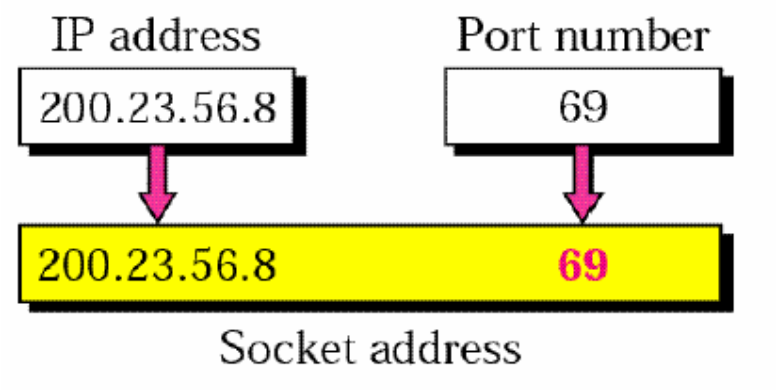


Figure 7.11: Socket Addressing

A transport layer protocol needs a pair of socket addresses: the client socket address and the server socket address. These four pieces of information are part of the IP header and the transport-layer protocol header. The IP header contains the IP address and the UDP or TCP header contains the port number.

7.4 User Datagram Protocol (UDP)

The simple unreliable transport layer protocol is called the User Datagram Protocol (UDP). UDP is a connectionless, unreliable transport protocol. It does not add anything to the services of IP except for providing process-to-process communication instead of host-to-host communication. Also, it performs very little error checking. UDP is a very simple protocol with a minimum of overhead. If a process wants to send a small message and does not care much about reliability, it can use UDP. Sending a small message using UDP takes much less time than using TCP. In addition, UDP is a convenient protocol for multimedia and multicasting applications.

The service provided by UDP is an unreliable service that provides no guarantees for delivery and no protection from duplication (e.g. if this arises due to software errors within an Intermediate System (IS)). The simplicity of UDP reduces the overhead from using the protocol and the services may be adequate in many cases.

A computer may send UDP packets without first establishing a connection to the recipient. The computer completes the appropriate fields in the UDP header (PCI) and forwards the data together with the header for transmission by the IP network layer as shown in figure 7.12.

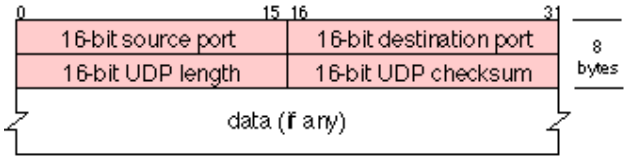


Figure 7.12: UDP Header

The UDP protocol header consists of 8 bytes of Protocol Control Information (PCI)

The UDP header consists of four fields each of 2 bytes in length:

- Source Port (UDP packets from a client use this as a service access point (SAP) to indicate the session on the local client that originated the packet. UDP packets from a server carry the server SAP in this field)
- Destination Port (UDP packets from a client use this as a service access point (SAP) to indicate the service required from the remote server. UDP packets from a server carry the client SAP in this field)
- UDP length (The number of bytes comprising the combined UDP header information and payload data)
- UDP Checksum (A checksum to verify that the end to end data has not been corrupted by routers or bridges in the network or by the processing in an end system. The algorithm to compute the checksum is the Standard Internet Checksum algorithm. If this check is not required, the value of 0x0000 is placed in this field, in which case the data is not checked by the receiver.)

Like for other transport protocols, the UDP header and data are not processed by Intermediate Systems (IS) in the network, and are delivered to the final destination in the same form as originally transmitted. At the final destination, the UDP protocol layer receives packets from the IP network layer as shown in figure 7.13.

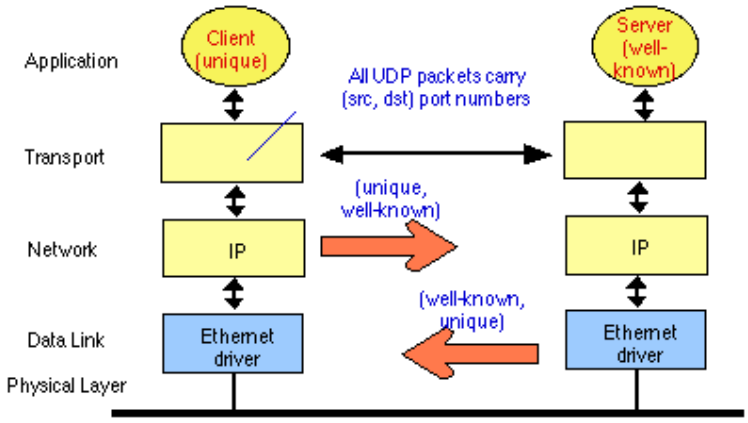


Figure 7.13: Packet Transmission – UDP at Transport Layer

These are checked using the checksum (when >0, this checks correct end-to-end operation of the network service) and all invalid PDUs are discarded. UDP does not make any provision for error reporting if the packets are not delivered. Valid data are passed to the appropriate session layer protocol identified by the source and destination port numbers (i.e. the session service access points).

Generally, clients set the source port number to a unique number that they choose themselves - usually based on the program that started the connection. Since this number is returned by the server in response, this lets the sender know which "conversation" incoming packets are to be sent to. The destination port of packets sent by the client is usually set to one of a number of well-known ports. These usually correspond to one of a number of different applications, e.g. port 23 is used for telnet, and port 80 is used for web servers.

A server process (program), listens for UDP packets received with a particular well-known port number and tells its local UDP layer to send packets matching this destination port number to the server program. It determines which client these packets come from by examining the received IP source address and the received unique UDP source port number. Any responses which the server needs to send to back to a client are sent with the source port number of the server (the well-known port number) and the destination port selected by the client.

UDP uses port numbers as the addressing mechanism in the transport layer. Some well-known port numbers used by UDP are listed in table 7.1.

Port	Protocol	Description
7	Echo	Echoes a received datagram back to the sender
9	Discard	Discards any datagram that is received
11	Users	Active users
13	Daytime	Returns the date and the time
17	Quote	Returns a quote of the day
19	Chargen	Returns a string of characters
53	Nameserver	Domain Name Service
67	Bootps	Server port to download bootstrap information
68	Bootpc	Client port to download bootstrap information
69	TFTP	Trivial File Transfer Protocol
111	RPC	Remote Procedure Call
123	NTP	Network Time Protocol
161	SNMP	Simple Network Management Protocol
162	SNMP	Simple Network Management Protocol (trap)

Table 7.1: Well-known ports for UDP Protocols

User Datagram

UDP packets are also called user datagrams.

Applications

UDP is used by applications such as Trivial File Transfer Protocol (TFTP) since this

application includes flow and error control.

7.5 Transmission Control Protocol (TCP)

The reliable, but complex transport-layer protocol is called Transmission Control Protocol (TCP). TCP is called a stream connection-oriented and reliable transmission protocol. It adds connection-oriented and reliability features to IP. TCP applies to the Internet, the end-to-end argument says that by putting excessive intelligence in physical and link layers to handle error control, encryption or flow control you unnecessarily complicate the system. This is because these functions will usually need to be done at the endpoints anyway, so why duplicate the effort along the way? The result of an end-to-end network then, is to provide minimal functionality on a hop-by-hop basis and maximal control between end-to-end communicating systems.

The end-to-end argument helped determine how the characteristics of TCP operate; performance and error handling. TCP performance is often dependent on a subset of algorithms and techniques such as flow control and congestion control. Flow control determines the rate at which data is transmitted between a sender and receiver. Congestion control defines the methods for implicitly interpreting signals from the network in order for a sender to adjust its rate of transmission. The term congestion control is a bit of a misnomer. Congestion avoidance would be a better term since TCP cannot control congestion per se. Ultimately intermediate devices, such as IP routers would only be able to control congestion. Congestion control is currently a large area of research and concern in the network community. A companion study on congestion control examines the current state of activity in that area.

Timeouts and retransmissions handle error control in TCP. Although delay could be substantial, particularly if you were to implement real-time

applications, the use of both techniques offer error detection and error correction thereby guaranteeing that data will eventually be sent successfully. The nature of TCP and the underlying packet switched network provide formidable challenges for managers, designers and researchers of networks. Once regulated to low speed data communication applications, the Internet and in part TCP are being used to support very high speed communications of voice, video and data. It is unlikely that the Internet protocols will remain static as the applications change and expand. Understanding the current state of affairs will assist us in understanding protocol changes made to support future applications.

Transmission Control Protocol

TCP is often described as a byte stream, connection-oriented, reliable delivery transport layer protocol. In turn, we will discuss the meaning for each of these descriptive terms.

Byte Stream Delivery

TCP interfaces between the application layer above and the network layer below. When an application sends data to TCP, it does so in 8-bit byte streams. It is then up to the sending TCP to segment or delineate the byte stream in order to transmit data in manageable pieces to the receiver. It is this lack of 'record boundaries" which give it the name "byte stream delivery service".

Connection-Oriented

Before the two communicating TCPs exchange data, they must first agree upon the willingness to communicate. Analogous to a telephone call, a connection must first be made before two parties exchange information.

Reliability

A number of mechanisms help provide the reliability TCP guarantees. Each of these is described briefly below.

Checksums: All TCP segments carry a checksum, which is used by the receiver to detect errors with either the TCP header or data.

Duplicate data detection: It is possible for packets to be duplicated in packet switched network; therefore TCP keeps track of bytes received in order to discard duplicate copies of data that has already been received.[2]

Retransmissions: In order to guaranteed delivery of data, TCP must implement retransmission schemes for data that may be lost or damaged. The use of positive acknowledgements by the receiver to the sender confirms successful reception of data. The lack of positive acknowledgements, coupled with a timeout period (see timers below) calls for a retransmission.

Sequencing: In packet switched networks, it is possible for packets to be delivered out of order. It is TCP's job to properly sequence segments; it receives, so that it can deliver the byte stream data to an application in order.

Timers: TCP maintains various static and dynamic timers on data sent. The sending TCP waits for the receiver to reply with an acknowledgement within a bounded length of time. If the timer expires before receiving an acknowledgement, the sender can retransmit the segment.

TCP Header Format

Remember that the combination of TCP header and TCP in one packet is called a TCP segment.

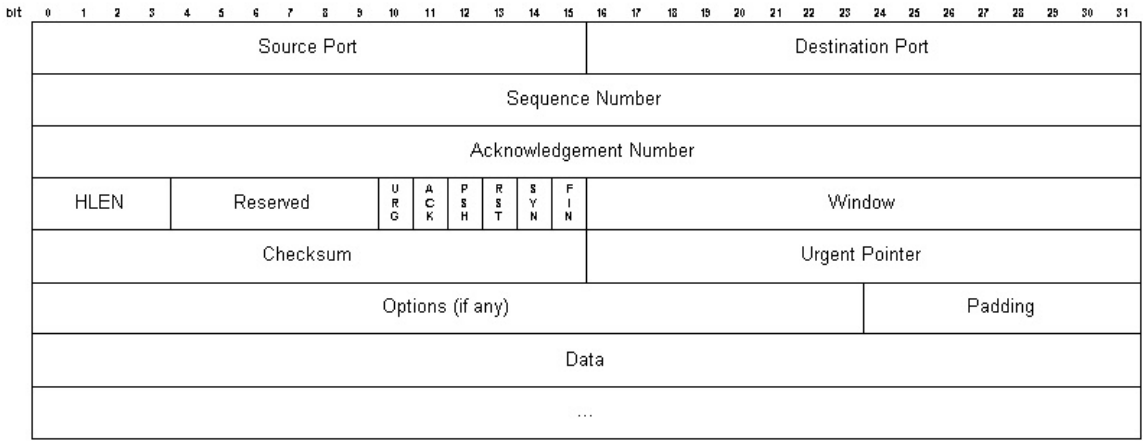


Figure 7.14: TCP Header Format

Figure 7.14 depicts the format of all valid TCP segments. The size of the header without options is 20 bytes. We will briefly define each field of the TCP header in figure 7.14.

Source Port A 16-bit number identifying the application the TCP segment originated from within the sending host. The port numbers are divided into three ranges, well-known ports (0 through 1023), registered ports (1024 through 49151) and private ports (49152 through 65535). Port assignments are used by TCP as an interface to the application layer. For example, the TELNET server is always assigned to the well-known port 23 by default on TCP hosts. A complete pair of IP addresses (source and destination) plus a complete pair of TCP ports (source and destination) define a single TCP connection that is globally unique as given in table 7.2.

Port	Protocol	Description
7	Echo	Echoes a received datagram back to the sender
9	Discard	Discards any datagram that is received
11	Users	Active users
13	Daytime	Returns the date and the time
17	Quote	Returns a quote of the day
19	Chargen	Returns a string of characters
20	FTP, Data	File Transfer Protocol (data connection)
21	FTP, Control	File Transfer Protocol (control connection)
23	TELNET	Terminal Network
25	SMTP	Simple Mail Transfer Protocol
53	DNS	Domain Name Server
67	BOOTP	Bootstrap Protocol
79	Finger	Finger
80	HTTP	Hypertext Transfer Protocol
111	RPC	Remote Procedure Call

Table 7.2: Some ports for TCP Protocols

Destination Port is a 16-bit number, identifying the application the TCP segment. It is destined for on a receiving host. Destination ports use the same port number assignments as those set aside for source ports.

Sequence Number is a 32-bit number, identifying the current position of the first data byte in the segment within the entire byte stream for the TCP connection. After reaching 2^{32} -1, this number will wrap around to 0.

Acknowledgement Number is a 32-bit number, identifying the next data byte the sender expects from the receiver. Therefore, the number will be one greater than the most recently received data byte. This field is only used when the ACK control bit is turned on.

Header Length is a 4-bit field that specifies the total TCP header length in 32-bit words (or in multiples of 4 bytes if you prefer). Without options, a TCP header is always 20 bytes in length. The largest a TCP header may be is 60 bytes. This field is required because the size of the options field(s) cannot be determined in advance. Note that this field is called "data offset" in the official TCP standard, but header length is more commonly used.

Reserved is a 6-bit field currently unused and reserved for future use.

Control Bits Urgent Pointer (URG). If this bit field is set, the receiving TCP should interpret the urgent pointer field.

Acknowledgement (ACK). If this bit field is set, the acknowledgement field described earlier is valid.

Push Function (PSH). If this bit field is set, the receiver should deliver this segment to the receiving application as soon as possible. An example of its

use may be to send a Control-BREAK request to an application, which can jump ahead of queued data.

Reset the Connection (RST). If this bit is present, it signals the receiver that the sender is aborting the connection and all queued data and allocated buffers for the connection can be freely relinquished.

Synchronize (SYN). When present, this bit field signifies that sender is attempting to "synchronize" sequence numbers. This bit is used during the initial stages of connection establishment between a sender and receiver.

No More Data from Sender/Final (FIN). If set, this bit field tells the receiver that the sender has reached the end of its byte stream for the current TCP connection.

Window A 16-bit integer used by TCP for flow control in the form of a data transmission window size. This number tells the sender how much data the receiver is willing to accept. The maximum value for this field would limit the window size to 65,535 bytes, however a "window scale" option can be used to make use of even larger windows.

Checksum A TCP sender computes a value based on the contents of the TCP header and data fields. This 16-bit value will be compared with the value the receiver generates using the same computation. If the values match, the receiver can be very confident that the segment arrived intact.

Urgent Pointer In certain circumstances, it may be necessary for a TCP sender to notify the receiver of urgent data that should be processed by the receiving application as soon as possible. This 16-bit field tells the receiver when the last byte of urgent data in the segment ends.

Options In order to provide additional functionality, several optional parameters may be used between a TCP sender and receiver. Depending on the option(s) used, the length of this field will vary in size, but it cannot be larger than 40 bytes due to the size of the header length field (4 bits). The most common option is the maximum segment size (MSS) option. A TCP receiver tells the TCP sender the maximum segment size it is willing to accept through the use of this option. Other options are often used for various flow control and congestion control techniques.

Padding as the options may vary in size, it may be necessary to "pad"

the TCP header with zeroes so that the segment ends on a 32-bit word boundary as defined by the standard.

Data may not used in some circumstances (e.g. acknowledgement segments with no data in the reverse direction), this variable length field carries the application data from TCP sender to receiver. This field coupled with the TCP header fields constitutes a TCP segment.

Connection Establishment and Termination

TCP provides a connection-oriented service over packet switched networks. Connection-oriented implies that there is a virtual connection between two endpoints. There are three phases in any virtual connection. These are the connection establishment, data transfer and connection termination phases.

Three-Way Handshake

In order for two hosts to communicate using TCP they must first establish a connection by exchanging messages in what is known as the three-way handshake. The figure 7.15 depicts the process of the three-way handshake.

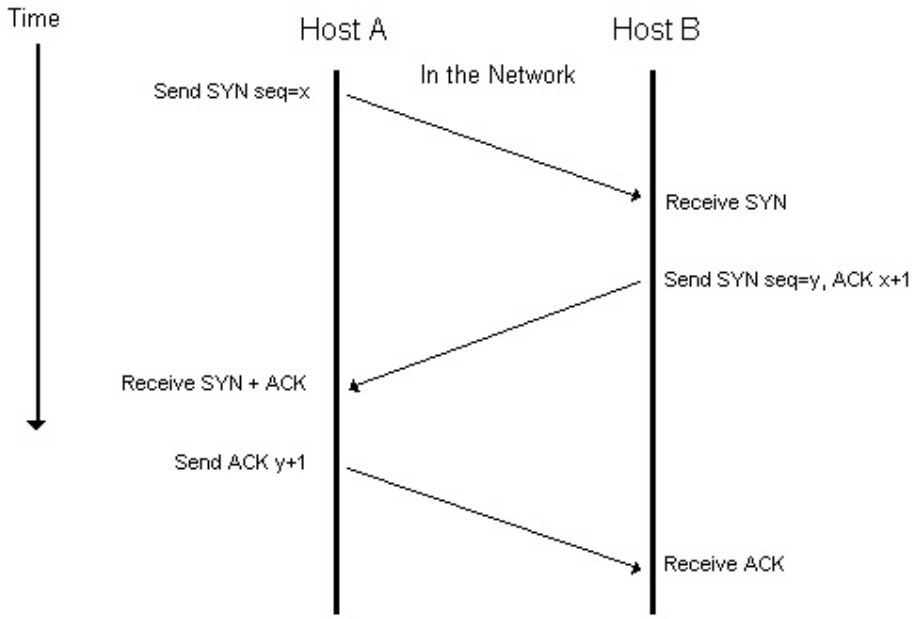


Figure 7.15 - TCP Connection Establishment

From figure 7.15, it can be seen that there are three TCP segments exchanged between two hosts, host A and host B. Reading down the diagram depicts events in time. To start, host A, initiates the connection by sending a TCP segment with the SYN control bit set and an initial sequence number (ISN), we represent as the variable x in the sequence number field.

At some moment later in time, host B, receives this SYN segment, processes it and responds with a TCP segment of its own. The response from Host B contains the SYN control bit set and its own ISN represented as variable y. Host B also sets the ACK control bit to indicate the next expected byte from Host A should contain data starting with sequence number x+1. When Host A receives Host B's ISN and ACK, it finishes the connection establishment phase by sending a final acknowledgement segment to Host B. In this case, Host A sets the ACK control bit and indicates the next expected byte from

Host B by placing acknowledgement number y+1 in the acknowledgement field.

In addition to the information shown in the figure above, an exchange of source and destination ports to use for this connection are also included in each senders' segments.

Data Transfer

Once ISNs have been exchanged, communicating applications can transmit data between each other. Most of the discussion surrounding data transfer requires us to look at flow control and congestion control techniques

A simple TCP implementation will place segments into the network for a receiver as long as there is data to send and as long as the sender does not exceed the window advertised by the receiver. As the receiver accepts and processes TCP segments, it sends back positive acknowledgements, indicating where is the byte stream. These acknowledgements also contain the "window" which determines how many bytes the receiver is currently willing to accept. If data is duplicated or lost, a "hole" may exist in the byte stream. A receiver will continue to acknowledge the most current contiguous place in the byte stream it has accepted. If there is no data to send, the sending TCP will simply sit idly by waiting for the application to put data into the byte stream or to receive data from the other end of the connection.

If data queued by the sender reaches a point where data sent will exceed the receiver's advertised window size, the sender must halt transmission and wait for further acknowledgements and an advertised window size that is greater than zero before resuming. Timers are used to avoid deadlock and unresponsive connections. Delayed transmissions are used to make more

efficient use of network bandwidth by sending larger "chunks" of data at once rather than in smaller individual pieces.

Connection Termination

In order for a connection to be released, four segments are required to completely close a connection. Four segments are necessary due to the fact that TCP is a full-duplex protocol, meaning that each end must shut down independently.

Sliding Window and Flow Control

Flow control is a technique whose primary purpose is to properly match the transmission rate of sender to that of the receiver and the network. It is important for the transmission to be at a high enough rate to ensure good performance, but also to protect against overwhelming the network or receiving host. We note that flow control is not the same as congestion control. Congestion control is primarily concerned with a sustained overload of network intermediate devices such as IP routers. TCP uses the window field, briefly described previously, as the primary means for flow control. During the data transfer phase, the window field is used to adjust the rate of flow of the byte stream between communicating TCPs.

Congestion Control

TCP congestion control and Internet traffic management issues in general is an active area of research and experimentation. This final section is a very brief summary of the standard congestion control algorithms widely used in TCP implementations today.

Slow Start

Slow Start, a requirement for TCP software implementations is a mechanism used by the sender to control the transmission rate, otherwise known as sender-based flow control. This is accomplished through the return rate of

acknowledgements from the receiver. In other words, the rate of acknowledgements returned by the receiver determine the rate at which the sender can transmit data.

When a TCP connection first begins, the Slow Start algorithm initializes a congestion window to one segment, which is the maximum segment size (MSS) initialized by the receiver during the connection establishment phase. When acknowledgements are returned by the receiver, the congestion window increases by one segment for each acknowledgement returned. Thus, the sender can transmit the minimum of the congestion window and the advertised window of the receiver, which is simply called the transmission window.

Slow Start is actually not very slow when the network is not congested and network response time is good. For example, the first successful transmission and acknowledgement of a TCP segment increases the window to two segments. After successful transmission of these two segments and acknowledgements completes, the window is increased to four segments. Then eight segments, then sixteen segments and so on, doubling from there on out up to the maximum window size advertised by the receiver or until congestion finally does occur.

Congestion Avoidance

During the initial data transfer phase of a TCP connection the Slow Start algorithm is used. However, there may be a point during Slow Start that the network is forced to drop one or more packets due to overload or congestion. If this happens, congestion avoidance is used to slow the transmission rate. However, Slow Start is used in conjunction with congestion avoidance as the means to get the data transfer going again so it doesn't slow down and stay slow.

In the Congestion Avoidance algorithm a retransmission timer expiring or the reception of duplicate ACKs can implicitly signal the sender that a network congestion situation is occurring. The sender immediately sets its transmission window to one half of the current window size (the minimum of the congestion window and the receiver's advertised window size), but to at least two segments. If congestion was indicated by a timeout, the congestion window is reset to one segment, which automatically puts the sender into Slow Start mode. If congestion was indicated by duplicate ACKs, the fast retransmit and fast recovery algorithms are invoked. As data is received during Congestion Avoidance, the congestion window is increased. However, Slow Start is only used up to the halfway point where congestion originally occurred. This halfway point was recorded earlier as the new transmission window. After this halfway point, the congestion window is increased by one segment for all segments in the transmission window that are acknowledged. This mechanism will force the sender to more slowly grow its transmission rate, as it will approach the point where congestion had previously been detected.

Fast Retransmit

When a duplicate ACK is received, the sender does not know, it is because that a TCP segment was lost or simply that a segment was delayed and received out of order at the receiver. If the receiver can re-order segments, it should not be long before the receiver sends the latest expected acknowledgement. Typically no more than one or two duplicate ACKs should be received when simple out of order conditions exist. If however more than two duplicate ACKs are received by the sender, it is a strong indication that at least one segment has been lost. The TCP sender will assume enough time has lapsed for all segments to be properly re-ordered by the fact that the

receiver had enough time to send three duplicate ACKs. When three or more duplicate ACKs are received, the sender does not even wait for a retransmission timer to expire before retransmitting the segment (as indicated by the position of the duplicate ACK in the byte stream). This process is called the Fast Retransmit algorithm. Immediately following Fast Retransmit is the Fast Recovery algorithm.

Fast Recovery

Since the Fast Retransmit algorithm is used when duplicate ACKs are being received, the TCP sender has implicit knowledge that there is data still flowing to the receiver. The reason is because duplicate ACKs can only be generated when a segment is received. This is a strong indication that serious network congestion may not exist and that the lost segment was a rare event. So instead of reducing the flow of data abruptly by going all the way into Slow Start, the sender only enters Congestion Avoidance mode.

Rather than start at a window of one segment as in Slow Start mode, the sender resumes transmission with a larger window, incrementing as if in Congestion Avoidance mode.

7.6 Summary

The transport layer is the key to understanding layered protocols. It provides various services, the most important of which is an end-to-end, reliable, connection-oriented byte stream from sender to receiver. It is accessed through service primitives that permit the establishment, use, and release of connections.

Transport protocols must be able to do connection management over unreliable networks. Connection establishment is complicated by the

existence of delayed duplicate packets that can reappear at inopportune moments. It must handle all the service primitives, manage connections and timers, and allocate and utilize credits. The Internet has two main transport protocols: UDP and TCP.

7.7 Self Assessment Questions

1. Why does the maximum packet lifetime, T, have to be large enough to ensure that not only the packet but also its acknowledgements have vanished?
2. Imagine that a two-way handshake rather than a three-way handshake were used to set up connections. In other words, the third message was not required. Are deadlocks now possible? Give an example or show that none exist.
3. Why does UDP exist? Would it not have been enough to just let user processes send raw IP packets?
4. Consider a simple application-level protocol built on top of UDP that allows a client to retrieve a file from a remote server residing at a well-known address. The client first sends a request with file name, and the server responds with a sequence of data packets containing different parts of the requested file. To ensure reliability and sequenced delivery, client and server use a stop-and-wait protocol. Ignoring the obvious performance issue, do you see a problem with this protocol? Think carefully about the possibility of processes crashing.

5. Both UDP and TCP use port numbers to identify the destination entity when delivering a message. Give two reasons for why these protocols

invented a new abstract ID (port numbers), instead of using process IDs, which already existed when these protocols were designed.

6. What is the total size of the minimum TCP MTU, including TCP and IP overhead but not including data link layer overhead?
7. Datagram fragmentation and reassembly are handled by IP and are invisible to TCP. Does this mean that TCP does not have to worry about data arriving in the wrong order?
8. A process on host 1 has been assigned port p, and a process on host 2 has been assigned port q. Is it possible for there to be two or more TCP connections between these two ports at the same time?
9. The maximum payload of a TCP segment is 65,495 bytes. Why was such a strange number chosen?

7.8 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
- Computer Network, ED Tittle, Tata MacGraw Hills Publications

Subject Name: Computer Networks **Subject Code:** MCA-301

Lesson No.: 8 **Author:** Mr. Yudhvir Singh

Lesson Title: Application Layer **Vetter: Dr. Manoj Dhun**

Structure

8.0 Objective

In this lesson we will study the design principles for layer 7, the application layer of layer of network model (ISO-OSI Reference). This study deals with the services, protocols at the network layer. This lesson deals with the application layer; especially network security, DNS, E-mail, SNMP, USENET,

World Wide Web, and Multimedia. This lesson also covers the basic concepts of ISDN and ATM.

8.1 Introduction

In this lesson we will study some real network applications. In the application layer there is a need for support protocols, to allow the applications to function. Accordingly, we will look at one of these before starting with the applications themselves. The item in question is DNS, which handles naming within the Internet. After that, we will examine other real applications: electronic mail, the USENET, the World Wide Web, and finally, multimedia. In this lesson we also discuss the network security techniques and to manage network we also discuss SNMP protocol. We have also covered the brief overview of ISDN and ATM.

8.2Network Security

Prevention is the key when it comes to network security. Identifying and stopping intrusion—in all its forms—is what security is all about. Physical theft, electronic tampering, and unauthorized access are just three of the more obvious threats to network equipment and data. Networks seriously increase access to your information, and with access comes the responsibility of restriction and control. In addition to the usual sources of security breaches—people taping passwords to their monitors and using scanners to electronically eavesdrop—networks invite a whole host of other vulnerabilities. Your network can be a danger to itself. Your network is physically vulnerable.

Your first line of defense is the simplest: Use locks, guards, and alarms to protect against these physical vulnerabilities. Lock servers in a room and lock

wiring closets, permitting access to only those with a key. Sensitive data must be completely wiped off the media when deleted.

If your security needs are extreme, you might employ biometric devices. Biometric devices use a physical aspect of people, such as their fingerprints, to verify their identity. A secure method to stop unauthorized access is an add-in card for each workstation. This card forces the workstation to boot up from a particular drive every time. It can also enforce some kind of user validation, like a password. If the card is removed, the workstation is automatically disabled.

Forms Of Data Security

Information security entails making sure the right people have access to the right information, that the information is correct, and that the system is available. These aspects are referred to as confidentiality, integrity, and availability.

Cryptography & Encryption

Passwords, locks, access privileges, and even biometric devices do not always deter the determined intruder. A common tool like a protocol analyzer can be hooked up to the network and the intruder can watch all data, including passwords, pass by. Data encryption is the answer. With encryption, data is scrambled before transmission, making it unreadable as it passes over the wire, even if it is intercepted. To encrypt this data, its bits must be transformed according to an algorithm. The data is transmitted, and at the receiving end, a system of keys is used to decode the bits into intelligible information. Keys are necessary for encoding and decoding.

Encryption usually requires extra hardware because of the processing power required. Hardware-based encryption schemes are more difficult to crack than software-based methods. A cipher is a character-for-character or bit-for-bit

transformation, without regard to the linguistic structure of the message. In contrast, a code replaces one word with another word or symbol.

The messages to be encrypted, known as the plaintext, are transformed by a function that is parameterized by a key. The output of the encryption process, known as the ciphertext, is then transmitted, often by messenger or radio. The enemy, or intruder, hears and accurately copies down the complete ciphertext. Unlike, the intended recipient, he does not know what the decryption key is and so cannot decrypt the ciphertext easily. Sometimes the intruder can not only listen to the communication channel but can also record messages and play them back later, inject his own messages, or modify legitimate messages before they get to the receiver . The art of breaking ciphers, called cryptanalysis, and the art devising them (cryptography) is collectively known as cryptology.

It will often be useful to have a notation for relating plaintext, ciphertext, and keys. We will use $C = E_K(P)$ to mean that the encryption of the plaintext P using key K gives the ciphertext C. Similarly, $P = D_K(C)$ represents the decryption of C to get the plaintext again. It then follows that

$$D_K(E_K(P)) = P$$

This notation suggests that E and D are just mathematical functions, which they are. The only tricky part is that both are functions of two parameters, and we have written one of the parameters – key as a subscript, rather than as an argument, to distinguish it from the message.

Substitution Ciphers

In a substitution cipher each letter or group of letters is replaced by another letter or group of letters to disguise it. In this method, a becomes D, b becomes E, c becomes F, ... , and z becomes C. For example, attack

becomes DWWDFN. A slight generalization of the cipher allows the ciphertext alphabet to be shifted by k letters, instead of always 3. In this case k becomes a key to the general method of circularly shifted alphabets. The next improvement is to have each of the symbols in the plaintext, say, the 26 letters for simplicity, map onto some other letter. For example,

plaintext: a b c d e

ciphertext: Q W E R T

The general system of symbol-for-symbol substitution is called a monoalphabetic substitution, with the key being the 26-letter string corresponding to the full alphabet.

Transposition Ciphers

Substitution ciphers preserve the order of the plaintext symbols but disguise them. Transposition ciphers, in contrast, reorder the letters but do not disguise them.The cipher is keyed by a word or phrase not containing any repeated letters. In this example, NETWORKS is the key. The purpose of the key is to number the columns, column 1 being under the key letter closest to the start of the alphabet, and so on. The plaintext is written horizontally, in rows, padded to fill the matrix if need be. The ciphertext is read out by columns, starting with the column whose key letter is the lowest.

To break a transposition cipher, the cryptanalyst must first be aware that he is dealing with a transposition cipher. By looking at the frequency of E, T, A, O, I, N, etc., it is easy to see if they fit the normal pattern for plaintext. If so, the cipher is clearly a transposition cipher, because in such a cipher every letter represents itself, keeping the frequency distribution intact.

One-Time Pads

Constructing an unbreakable cipher is actually quite easy; the technique has been known for decades. First choose a random bit string as the key. Then

convert the plaintext into a bit string, for example by using its ASCII representation. Finally, compute the XOR (eXclusive OR) of these two strings, bit by bit. The resulting ciphertext cannot be broken, because in a sufficiently large sample of ciphertext, each letter will occur equally often, as will every digram, every trigram, and so on. This method, known as the one-time pad, is immune to all present and future attacks no matter how much computational power the intruder has. The reason derives from information theory: there is simply no information in the message because all possible plaintexts of the given length are equally likely.

DES

A common data encryption standard specified is Data Encryption Standard (DES). DES defines how the data should be encrypted and the specifications for an electronic key. It uses one 64-bit key for encryption and decryption. This can cause problems because the key must be in the hands of the sender and receiver. The only way to get it from place to place is to transmit it. Transmitting the key introduces a security threat.

An outline of DES is shown in Figure 8.1, Plaintext is encrypted in blocks of 64 bits, yielding 64 bits of ciphertext. The algorithm, which is parameterized by a 56-bit key, has 19 distinct stages. The first stage is a key-independent transposition on the 64-bit plaintext. The last stage is the exact inverse of this transposition. The stage prior to the last one exchanges the leftmost 32 bits with the rightmost 32 bits. The remaining 16 stages are functionally identical but are parameterized by different functions of the key. The algorithm has been designed to allow decryption to be done with the same key as encryption, a property needed in any symmetric-key algorithm. The steps are just run in the reverse order.

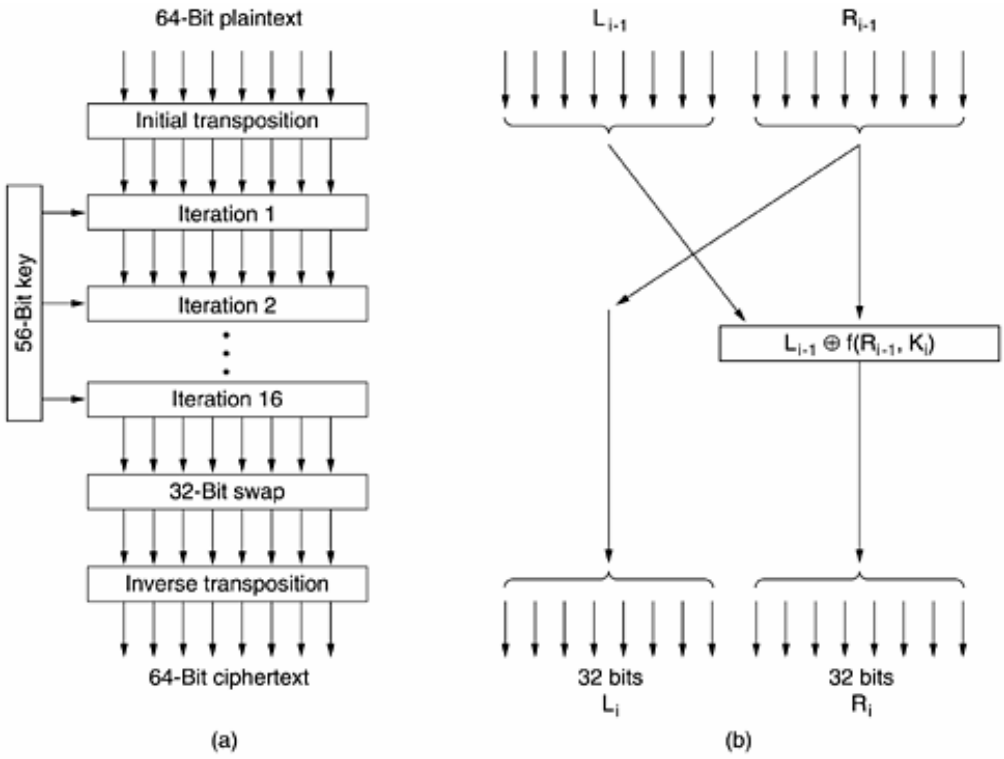


Figure 8.1: The data encryption standard. (a) General outline. (b) Detail of one iteration. The circled + means exclusive OR.

The operation of one of these intermediate stages is illustrated in here. Each stage takes two 32-bit inputs and produces two 32-bit outputs. The left output is simply a copy of the right input. The right output is the bitwise XOR of the left input and a function of the right input and the key for this stage, K_i. All the complexity lies in this function. The function consists of four steps, carried out in sequence. First, a 48-bit number, E, is constructed by expanding the 32-bit R_{i-1} according to a fixed transposition and duplication rule. Second, E and K_i are XORed together. This output is then partitioned into eight groups of 6 bits each, each of which is fed into a different S-box. Each of the 64 possible

inputs to an S-box is mapped onto a 4-bit output. Finally, these 8 x 4 bits are passed through a P-box.

In each of the 16 iterations, a different key is used. Before the algorithm starts, a 56-bit transposition is applied to the key. Just before each iteration, the key is partitioned into two 28-bit units, each of which is rotated left by a number of bits dependent on the iteration number. K_i is derived from this rotated key by applying yet another 56-bit transposition to it. A different 48-bit subset of the 56 bits is extracted and permuted on each round.

IP Security

Security extensions to IP bring authentication and privacy to the Internet. This layer 3 protocol oversees packet forwarding through different types of networks. But, IP's ubiquity and wide acceptance as the basis for both public and private networks has led to great concern over the issue of security. Several types of attacks have been known to take place over IP networks. One is called IP spoofing, in which an intruder tries to gain access by changing a packet's IP address to make it appear that the packet came from somewhere else. Other attacks include eavesdropping on an IP transmission. Using a protocol analyzer to record network traffic can do this. Another type of IP attack involves taking over a session and masquerading as one of the parties involved in the communication. The IP security problem will probably only get worse as companies rely on the protocol more and more for remote communications.

Virtual Private Networks (VPNs), which allow companies to create a private connection over the public Internet, require strong safeguards. Although VPNs do exist to a certain extent today, most industry watchers will tell you that for this concept to really fly, authentication, encryption, and other security measures need to be in place. The IP Security (IPSec), an extension to IP,

will be the catalyst for private and secure communications over the Internet. The suite includes the Authentication Header (AH), which addresses authentication for IP traffic, and the Encapsulating Security Payload (ESP), which defines encryption for IP data. The Authentication Header ensures that the packet has not been altered or tampered with during transmission. It can be used in combination with the ESP, or it can simply be used to verify the authenticity of a regular IP packet. The AH also allows the receiver to verify the identity of the sender.

Key Management

Together, the IPSec ESP and AH protocols provide privacy, integrity, and authentication of IP packets, but they are not the complete package.

The Internet Security Association and Key Management Protocol (ISAKMP)/Oakley key exchange protocol automatically handles exchange of secret symmetric keys between sender and receiver. The protocol integrates ISAKMP with the Oakley method of key exchange. ISAKMP is based on the Diffie-Hellman model of key generation, in which the two parties share information beforehand to ensure the identity of the other party. Under Diffie-Hellman, two parties generate their own public values, which they send to the other party. The two parties communicate through UDP messages. Each party then takes the public key they received and combines it with a private key. The result should be the same for both parties, but no one else can generate the same value.

Although ISAKMP is an automated method, it does allow for the level of trust in keys to be controlled, much like administrators currently control network passwords and how often they are changed. With ISAKMP, the SPI (that 32-bit number that contains security protocol information for a packet) can be reformatted at specified intervals. ISAKMP/Oakley supports three methods of

key information exchange: main mode, aggressive mode, and quick mode. Main mode establishes what's known as the first phase of ISAKMP SA. SA or Security Association, is the method of keeping track of all the details of keys and algorithms in an IPSec-compliant session. The SA includes a wide range of information, including the AH authentication algorithm and keys, the ESP encryption algorithm and keys, how often keys are to be changed, how communications are authenticated, and information about the SA's lifespan.

Securely Speaking

Once IPSec finds its way into the security policy of most companies, you can expect to see it used in a variety of applications. It can be used in instances where a network contains information that needs to be kept secure, as well as in cases where only authorized users should be allowed access to network resources. Perhaps IPSec's most profound impact will be on secure VPNs, which promise to provide a less expensive, yet still secure, method of communication than dedicated private networks. However you look at it, the encryption, authentication, and management capabilities of IPSec make it an encouraging way to extend a private network through the public Internet.

Biometrics

Biometrics technology verifies or identifies a person based on physical characteristics. A biometrics system uses hardware to capture the biometric information, and software to maintain and manage it. The system translates these measurements into a mathematical, computer-readable format. When a user first creates a biometric profile, known as a template, that template is stored in a database. The biometrics system then compares this template to the new image created every time a user accesses the system. A biometric device automates entry into secure locations, relieving or at least reducing the need for full-time monitoring by personnel and when rolled into an

authentication scheme, biometrics adds a strong layer of verification for user names and passwords. Biometrics adds a unique identifier to network authentication, one that's extremely difficult to duplicate.

Identify Vs. Verify

It's important to distinguish whether a biometrics system is used to verify or identify a person. These are separate goals, and some biometrics systems are more appropriate for one than the other, though no biometric system is limited to one or the other. The needs of the environment will dictate which system is chosen.

The most common use of biometrics is verification. As the name suggests, the biometric system verifies the user based on information provided by the user. Identification seeks to determine who the subject is without information from, or participation of, the subject. For instance, face recognition systems are commonly used for identification; a device captures an image of the subject's face and looks for a match in its database. Identification is complicated and resource-intensive because the system must perform a one-to-many comparison of images, rather than a one-to-one comparison performed by a verification system.

Biometric Errors

All biometrics systems suffer from two forms of error: false acceptance and false rejection. False acceptance happens when the biometric system authenticates an impostor. False rejection means that the system has rejected a valid user. A biometric system's accuracy is determined by combining the rates of false acceptance and rejection.

Each error presents a unique administrative challenge. For instance, if you're protecting sensitive data with a biometric system, you may want to tune the system to reduce the number of false acceptances. However, a system that's

highly calibrated to reduce false acceptances may also increase false rejections, resulting in more help desk calls and administrator intervention. Therefore, administrators must clearly understand the value of the information or systems to be protected, and then find a balance between acceptance and rejection rates appropriate to that value. A poorly created enrollment template can compound false acceptance and rejection. Natural changes in a user's physical traits may also lead to errors.

Biometric Types

Finger Scan: Fingerprint scanning is the most common biometric system used today. The human fingerprint is made up of ridges that take the shape of loops, arches, and whorls. Rather than scan each ridge, fingerprint-based biometrics look for minutia, which are the points on a fingerprint where a ridge ends or splits into two.

Hand Geometry: Hand scanners use an optical device to measure biometric data such as the length, width, thickness, and surface area of a person's hand and fingers. While accurate, hand scanners don't gather as much biometric data as a fingerprint or eye scan. They aren't suitable for identification (as opposed to verification) because similarities among hands don't allow for a one-to-many database search. Hand injuries may also result in false rejections.

Iris and Retina: The iris is the colored ring that surrounds the pupil. A camera using visible and infrared light scans the iris and creates a biometric template based on characteristics of the iris tissue, such as rings, furrows, and freckles. Iris scanning is remarkably accurate, making it suitable for both identification and verification. Peripheral cameras are available for desktop-based network authentication, but iris scanning has traditionally been used for physical access to secure locations. The retina is a nerve in the back of the eye that

senses light. A retina scan creates a template from blood vessels in the retina, the patterns of which are unique to each person. Unlike other biometric characteristics, a person's retinal blood vessel pattern changes very little throughout a person's lifetime, making retina scanning a robust solution. A retina scan is also highly accurate.

Face Scan: Facial scans capture physical characteristics such as the upper outlines of the eye sockets, the areas around the cheekbones, and the sides of the mouth. Face scanning is appropriate for both identification and verification. For network authentication, PC video cameras are sufficient, but higher quality cameras are necessary for capturing images from greater distances and in more variable lighting conditions.

Voice and Signature: Voice scanning captures characteristics such as the pitch, tone, and frequency of a subject's voice. Voice biometrics may win a role in network authentication because many PCs include a microphone or can be easily fitted with one. However, background noise or poor-quality microphones can interfere with authentication. Signature scans capture both the image of the signature and the physical characteristics of the signer, such as the speed and pressure he or she exerts. Signature scanning requires a specialized tablet to capture the biometric data. At this point, voice and signature biometrics aren't widely deployed, though they may be successful in certain niches, such as point-of-sale transactions. On the positive side, user resistance to these biometrics is low because of their ease of use and familiarity.

Issues And Standards: Administrators must keep several points in mind when considering a biometric solution. Aside from scanner hardware and software, a biometric system has other costs, including the template repository. This repository must be at least as well secured as a password

database. Administrators must also consider availability, backup, and general maintenance costs. Biometric enrollment-when the subject first creates a template-must be handled carefully. A poor enrollment scan can lead to false rejection rates later on. Users also need to be properly trained in the use of the biometric system to prevent unnecessary rejections. Users may also have privacy concerns that will need to be addressed.

Firewalls

When your network is connected to a public network, it is exposed to spies, thieves, hackers, thrill seekers, and various other threats. As the public Internet has come to play a critical role in most everyone's networking strategy, protection from undesirable Internet inhabitants is a necessity. One of the key elements of safe networking is a properly designed and configured firewall system.

There are three basic designs for firewall operation. The first, and simplest, is the packet filter. Most routers can be configured to filter packets based on straightforward comparison of packet contents with filter specifications. In most modern routers, adding a security-based filter step to the packet forwarding process will add little or no overhead, so packet-filtering firewalls can have very high performance. Unfortunately, would-be intruders have a number of options for defeating simple filtering. The second basic firewall design enhances packet filtering so that it can't be circumvented by these measures. Stateful inspection extends the packet-by-packet filtering process to include multipacket flows. A connection table tracks individual flows, enabling policy checks that extend across series of packets.

The third firewall design is the application proxy. With a pure application proxy, no traffic at all goes through the firewall. Instead, the application proxy behaves like a server to clients on the trusted network and like a client to

servers outside the trusted network. Application proxy firewalls and circuit-level gateways transparently perform Network Address Translation (NAT). The network interface that connects to the untrusted network is all that the untrusted network sees, so the addresses and the structure of the trusted network are shielded from view. In themselves, packet filters and stateful inspection firewalls don't translate IP addresses and port numbers, though there's no reason that NAT can't be implemented separately.

Intrusion Detection

The idea behind an IDS is simple: an agent monitors file activity on a host or traffic on a network, and reports strange behavior to an administrator. The IDS market is divided into two primary groups: host-based and network-based systems. This tutorial examines the basics of host- and network-based detection systems, discusses how to implement an IDS, and outlines a proper incident-response plan.

Host-Based IDS: Host-based IDSs add a targeted layer of security to particularly vulnerable or essential systems. An agent sits on an individual system and monitors audit trails and system logs for anomalous behavior, such as repeated login attempts or changes to file permissions. The agent may also employ a checksum at regular intervals to look for changes to system files. In some cases, an agent can halt an "attack" on a system, though a host agent's primary function is to log events and send alerts.

The primary benefit of a host-based system is that it can detect both external and internal misuse, something that network monitors and firewalls can't do. The appeal of such a tool is obvious, as security breaches are more likely to come from an internal user than from a hacker outside the network. Host agents are powerful tools for addressing the authorization and access issues that make internal security so complex. Agents install directly on the host to

be monitored, so they must be compatible with the host's OS. Memory requirements and CPU utilization will vary from vendor to vendor, so be sure to learn ahead of time the demands the agent will place on the system.

Network-Based IDS: A network-based IDS sits on the LAN (or a LAN segment) and monitors network traffic packet by packet in real time (or as near to real time as possible), to see if that traffic conforms to predetermined attack signatures. Attack signatures are activities that match known attack patterns. The IDS vendor provides a database of attack signatures, and administrators can also add customized signatures. If the IDS recognizes an attack, it alerts an administrator. In addition to its monitoring and alarm functions, the IDS also records attack sessions for later analysis. Network IDSs can also be linked to other security features, such as firewalls, to make sure those systems haven't been breached.

A network monitor has two main benefits. The first is the real-time nature of the alarm, which can give administrators an opportunity to halt or contain an attack before it does significant harm. This is especially valuable for DoS attacks, which must be dealt with immediately to mitigate damages. The second benefit is evidence collection. Not only can administrators analyze the attack to determine what damage might have been done, the attack session itself can point out security flaws that need addressing. Because many hackers first scan a target network for known vulnerabilities, a hacker's choice of attack may indicate that such vulnerabilities exist on your network. A simple example is an operating system that has yet to be secured with the latest vendor patch.

Network monitors are OS-independent. Basic requirements include a dedicated node that sits on the segment to be monitored and a NIC set to

promiscuous mode. You may also want to set up a secure communications link between the monitor and its management console.

E-Mail Security

Security problems hinder the growth of e-mail as a business tool.

Identity Crises: E-mail has several inherent security problems. When placed in the context of business communications, these problems limit e-mail's potential as a serious business tool. Normally, e-mail is sent "in the clear," meaning the message is sent in plaintext. So, anyone who can access the e-mail, whether in transit or in storage, can read the message. Clearly, this is a security problem that may prevent companies from using e-mail to convey confidential business information.

Here's a list of the main security issues that affect e-mail.

Lack of privacy E-mail is sent in plaintext and can be read by anyone who can access it.

Lack of integrity There is no safeguard to prevent someone from changing the contents of an e-mail message while it's in storage or in transit.

Lack of authenticity Anyone can forge an e-mail message that claims it was written by another individual.

Lack of nonrepudiation Any particular e-mail message can't be bound to its sender, so a sender can deny ever having sent a message to you.

Viruses E-mail messages can contain attachments that are actually viruses in disguise; when you open the attachment, the virus spreads to your PC.

Spam An e-mail account is an open home for spam, those annoying mass e-mail rants and advertisements

Another basic problem revolves around the integrity of a particular piece of e-mail. A more complex security problem is one of authenticity. A related

security problem is a lack of non-repudiation, in which a sender can deny that he or she ever sent a message.

E-Mailer ID: Currently, several schemes seek to address e-mail's security woes, specifically those of privacy, integrity, authentication, and non-repudiation. These solutions are all rooted in public key cryptography technology.

However, for this system to work, and this is a critical point, users must be able to trust that a user's key is valid. To provide this verification, a trusted entity, generically known as a Certificate Authority (CA), assigns each user a unique digital certificate that assures that a certain public key belongs to a certain user.

Public Key Cryptography

A secure and simple method of encrypting and decrypting electronic information.

Public-Key Algorithms: Diffie and Hellman, proposed a radically new kind of cryptosystem, one in which the encryption and decryption keys were different, and the decryption key could not feasibly be derived from the encryption key. In their proposal, the (keyed) encryption algorithm, E, and the (keyed) decryption algorithm, D, had to meet three requirements. These requirements can be stated simply as follows:

1. D(E(P)) = P.
2. It is exceedingly difficult to deduce D from E.
3. E cannot be broken by a chosen plaintext attack.

The first requirement says that if we apply D to an encrypted message, E(P), we get the original plaintext message, P, back. Without this property, the legitimate receiver could not decrypt the ciphertext. The second requirement speaks for itself. The third requirement is needed because, as we shall see in

a moment, intruders may experiment with the algorithm to their hearts' content. Under these conditions, there is no reason that the encryption key cannot be made public.

RSA: The only catch is that we need to find algorithms that indeed satisfy all requirements. It is known by the initials of the three discoverers (Rivest, Shamir, Adleman): RSA. The RSA method is based on some principles from number theory. We will now summarize how to use the method:

1. Choose two large primes, p and q (typically 1024 bits).
2. Compute n = p x q and z = (p - 1) x (q - 1).
3. Choose a number relatively prime to z and call it d.
4. Find e such that e x d = 1 mod z.

With these parameters computed in advance, we are ready to begin encryption. Divide the plaintext (regarded as a bit string) into blocks, so that each plaintext message, P, falls in the interval $0 < P < n$. Do that by grouping the plaintext into blocks of k bits, where k is the largest integer for which $2^k < n$ is true. To encrypt a message, P, compute $C = P^e \pmod{n}$. To decrypt C, compute $P = C^d \pmod{n}$. It can be proven that for all P in the specified range, the encryption and decryption functions are inverses. To perform the encryption, you need e and n. To perform the decryption, you need d and n. Therefore, the public key consists of the pair (e, n), and the private key consists of (d, n).

The security of the method is based on the difficulty of factoring large numbers. If the cryptanalyst could factor the (publicly known) n, he could then

find p and q, and from these z. Equipped with knowledge of z and e, d can be found using Euclid's algorithm. Fortunately, mathematicians have been trying to factor large numbers for at least 300 years, and the accumulated evidence suggests that it is an exceedingly difficult problem. According to Rivest and colleagues, factoring a 500-digit number requires 10^{25} years using brute force. In both cases, they assume the best known algorithm and a computer with a 1-µsec instruction time. Even if computers continue to get faster by an order of magnitude per decade, it will be centuries before factoring a 500-digit number becomes feasible, at which time our descendants can simply choose p and q still larger.

The Virus Threat

Although many viruses are labeled benign, a virus usually causes at least some inconvenience, some loss of system-access time and, at the worst, and loss of data. A virus is a program that has the ability to reproduce by modifying other programs to include a copy of itself. It may contain destructive code that moves into multiple programs, data files, or devices on a system and spreads through multiple systems in a network. Viral code may execute immediately or wait for a specific set of circumstances. Viruses are not distinct programs; they need a host program executed to activate their code. Many distinct programmed threats have been lumped together as viruses; however, most are not actually viruses at all. To better understand the threat, we will identify and define these other techniques.

Bacteria, also known as rabbits, are programs that do not explicitly damage any files. Their sole purpose is to reproduce themselves. A typical bacteria program does nothing more than reproduce itself exponentially, eventually eating up all the processor capacity, memory, or disk space, denying the user

access to those resources. This kind of programming attack is one of the oldest forms of programmed threat.

A *logic bomb* is a program or a section of code built into a program that lies dormant until a predefined condition is met. When that condition occurs, the bomb goes off with a result that is neither expected nor desired. Time bombs explode frequently. Depending on who authored the bomb and how many generations of backups it contaminated, the recovery effort ranges from mildly inconvenient to nearly impossible.

A *password catcher* is a program that mimics the actions of a normal sign-on screen but stores supplied ID and password combinations, usually in a hidden file accessible only to the author. The collected ID/password combinations are later used to attack the system. Running for three or four days will generally yield more than 90 percent of a site's active system users. Truly clever ones cause no perceptible delay and force no repeated sign-ons.

A *repeat dialer* is a program that continually calls the same number, thus placing it virtually out of service for any other caller. This technique has been used against TV preachers during fund drives to stop pledge calls. As would-be donors keep receiving busy signals, they tend to get frustrated and stop trying.

Trapdoors, also known as backdoors, are undocumented entry points into an otherwise secure system. During software development, programmers often create entry points into "their" programs to aid in debugging and adding final enhancements. These trapdoors are supposed to be closed before the program goes through the promotion to production process. Many times, however, they are not. This breach leaves an unadvertised but very real hole in the system's security.

A *Trojan horse* is a program that appears to perform a useful function, and sometimes does so quite well, but also includes an unadvertised feature that is usually malicious in nature. Viruses and logic bombs can be hidden in Trojan horses. The code in a well-constructed Trojan horse can perform its apparent function long and admirably before encountering the triggering condition that prompts it to let loose its secret agenda.

A *war or demon dialer* is a program run from outside an organization's environment and control, usually by hackers, to find dial-up ports into computer systems. Such a program identifies numbers in a given range that connect to a computer. From a modem-equipped PC, the hacker enters a starting phone number and an ending value to the resident dialer program. The program then sequentially dials each number in the range, seeking a computer tone. Copies of this type of program are found on bulletin boards everywhere.

A *worm* is a program that scans a system or an entire network for available, unused disk space in which to run. Originally, worms were developed by systems programmers searching for fragments of core in which to run segments of large programs. They tend to tie up all computing resources in a system or on a network and effectively shut it down. Worms can be activated at boot-up or submitted separately.

A computer virus, like its human counterpart, does not spread through the air. Humans become infected by a virus by coming in contact with someone who is infected. So it is with computers. They must come in contact with some contaminated source. The virus infects any form of writable storage, including hard drives, diskettes, magnetic tapes and cartridges, optical media, and memory. The most frequent sources of contamination include:

- physical or communication contact with an infected computer,
- copying an unknown disk containing a carrier program,
- downloading a file from a bulletin board system,
- running an infected network program,
- booting with an infected disk,
- infected software from a vendor,
- overt action by individuals, and
- e-mail attachments.

Virus Protection

A number of clues can indicate that a virus has infected or attempted to infect a system, even before any damage is done. Unexplained system crashes, programs that suddenly don't seem to work properly, data files or programs mysteriously erased, disks becoming unreadable—all could be caused by a virus. Here are some indicators that may confirm the presence of a virus. Most viruses use information provided by the command DIR, which lists a disk's directory, and CHKDSK, which snapshots disk and memory usage.

File size increase. A file's size may increase when a virus attaches itself to the file.

Change in update timestamp. When a virus modifies another program—even one such as COMMAND.COM (which is part of the operating system)—the "last-update" date and time are often changed. Since most programs are normally never modified (except when they're upgraded), periodically checking the last-update timestamp, with the DIR command, can alert the user to the presence of a virus. Another danger sign is when many programs list the same date and/or time in their last-update field. This occurrence indicates that all have been modified together, possibly by a virus.

Sudden decrease of free space. When running a new program, particularly if it is freeware or shareware, be alert for a sudden, unexpected decrease in disk space or memory.

Numerous unexpected disk accesses. Unless a program is exceptionally large or uses huge data files, it should not conduct a high number of disk accesses. Unexpected disk activity might signal a virus.

Preventing Infection

Preventing a virus infection is the best way to protect your organization against damage. If a virus cannot establish itself within your systems, then it cannot damage your programs or data. The following steps can help keep a clean system from becoming infected with a virus.

Awareness training. All employees having access to computer systems should be required to attend a training session on the virus threat. It is crucial that employees realize how much damage a virus can inflict.

Policies and procedures. The organization should prepare a policy on virus control to address the following issues: tight control of freeware and shareware; a control process that includes running anti-virus software regularly by each department; a virus response team and methods for contacting the team; control of the infection once it is detected; and recovery from the virus, including backup and dump policies.

For two very important reasons, the user community should be made aware of the risks of sharing software. The primary cause of the spread of virus infections is through the uncontrolled use of diskettes being introduced into computer systems. The other reason is the possibility of the illegal use of copyrighted software. If your organization lets employees transport diskettes out of the work facility, a quarantined system to test diskettes and software

before their introduction into the system should be in effect. This quarantine system should test all diskettes for the possibility of virus contamination.
In the network environment, avoid placing shareware in a common file server directory, thereby making it accessible to any PC in the network. Only allow the network administrator to sign on to the file server node. The most prudent precaution is to carefully make, store, and routinely check backup copies of files and programs—all on an established schedule. And control access to backups to guarantee integrity.

8.1 DNS — The Domain Name System

Theoretically programs refer to hosts, mailboxes, and other resources by their network addresses (IP), these addresses are difficult for people to remember. Also, sending e-mail to ys@192.168.4.241 means that if YS's ISP or organization changes the mail server to a different machine with a different IP address, his e-mail address has to change. So, ASCII names were introduced to decouple machine names from machine addresses. In this way, YS's address might be something like ys@gju.ernet.in. The network itself understands only numerical addresses, so some mechanism is required to convert the ASCII strings to network addresses.

If host name conflicts would occur constantly unless names were centrally managed, something unthinkable in a huge international network due to the load and latency. To solve these problems, DNS (the Domain Name System) was designed. The essence of DNS is hierarchical, domain-based naming scheme and a distributed database system for implementing this naming scheme. It is primarily used for mapping host names and e-mail destinations to IP addresses but can also be used for other purposes. To map a name

onto an IP address, an application program calls a library procedure called the resolver, passing it the name as a parameter. The resolver sends a UDP packet to a local DNS server, which then looks up the name and returns the IP address to the resolver, which then returns it to the caller. DNS with the IP address, the program can then establish a TCP connection with the destination or send it UDP packets.

The DNS Name Space

In the postal system, name management is done by requiring letters to specify the country, state or province, city, and street address of the addressee. By using this kind of hierarchical addressing, there is no confusion between the two addresses of persons. Conceptually, the Internet is divided into over 200 top-level domains, where each domain covers many hosts. Each domain is partitioned into subdomains, and these are further partitioned, and so on. All these domains can be represented as a tree. The leaves of the tree represent domains that have no subdomains. A leaf domain may contain a single host, or it may represent a company and contain thousands of hosts.

The top-level domains come in two flavors: generic and countries. The original generic domains were com (commercial), edu (educational institutions), gov (the U.S. Federal Government), int (certain international organizations), mil (the U.S. armed forces), net (network providers), and org (nonprofit organizations). And some new general-purpose, top-level domains, namely, biz (businesses), info (information), name (people's names), and pro (professions, such as doctors and lawyers). In addition, three more specialized top-level domains were introduced at the request of certain industries. These are aero (aerospace industry), coop (co-operatives), and museum (museums).

In general, getting a second-level domain, such as name-of-company.com, is easy. It merely requires going to a registrar for the corresponding top-level domain to check if the desired name is available and not somebody else's trademark. If there are no problems, the requester pays a small annual fee and gets the name. By now, virtually every common word has been taken in the com domain. Try household articles, animals, plants, body parts, etc. Nearly all are taken. Domain names can be either absolute or relative. An absolute domain name always ends with a period whereas a relative one does not. Relative names have to be interpreted in some context to uniquely determine their true meaning. In both cases, a named domain refers to a specific node in the tree and all the nodes under it. Domain names are case insensitive, so edu, Edu, and EDU mean the same thing. Component names can be up to 63 characters long, and full path names must not exceed 255 characters.

Resource Records

Every domain, whether it is a single host or a top-level domain, can have a set of resource records associated with it. For a single host, the most common resource record is just its IP address, but many other kinds of resource records also exist. When a resolver gives a domain name to DNS, what it gets back are the resource records associated with that name. Thus, the primary function of DNS is to map domain names onto resource records. A resource record is a five-tuple. Although they are encoded in binary for efficiency, in most expositions, resource records are presented as ASCII text, one line per resource record. The format we will use is as follows:

Domain_name Time_to_live Class Type Value

The Domain_name tells the domain to which this record applies.

The Time_to_live field gives an indication of how stable the record is.

The third field of every resource record is the Class.

Name Servers

A single name server could contain the entire DNS database and respond to all queries about it. In practice, this server would be so overloaded as to be useless. Furthermore, if it ever went down, the entire Internet would be crippled. To avoid the problems associated with having only a single source of information, the DNS name space is divided into nonoverlapping zones. Each zone contains some part of the tree and also contains name servers holding the information about that zone. Normally, a zone will have one primary name server, which gets its information from a file on its disk, and one or more secondary name servers, which get their information from the primary name server. To improve reliability, some servers for a zone can be located outside the zone. Where the zone boundaries are placed within a zone is up to that zone's administrator. This decision is made in large part based on how many name servers are desired, and where. If, however, the domain is remote and no information about the requested domain is available locally, the name server sends a query message to the top-level name server for the domain requested.

8.2 E-mail Electronic Mail

Electronic mail, or e-mail, has been around for over two decades. It became known to the public at large and grew exponentially to the point where the number of e-mails sent per day now is vastly more than the number of snail mail letters. E-mail is full of jargon such as BTW (By The Way). Many people also use little ASCII symbols called smileys or emoticons in their e-mail.

The first e-mail systems simply consisted of file transfer protocols, with the convention that the first line of each message contained the recipient's

address. As time went on, the limitations of this approach became more obvious. Some of the complaints were as follows:

1. Sending a message to a group of people was inconvenient. Managers often need this facility to send memos to all their subordinates.
2. Messages had no internal structure, making computer processing difficult. For example, if a forwarded message was included in the body of another message, extracting the forwarded part from the received message was difficult.
3. The originator (sender) never knew if a message arrived or not.
4. If someone was planning to be away on business for several weeks and wanted all incoming e-mail to be handled by his secretary, this was not easy to arrange.
5. The user interface was poorly integrated with the transmission system requiring users first to edit a file, then leave the editor and invoke the file transfer program.
6. It was not possible to create and send messages containing a mixture of text, drawings, facsimile, and voice.

Architecture and Services

E-mail normally consist of two subsystems: the user agents, which allow people to read and send e-mail, and the message transfer agents, which move the messages from the source to the destination. The user agents are local programs that provide a command-based, menu-based, or graphical method for interacting with the e-mail system. The message transfer agents are typically system daemons, that is, processes that run in the background. Their job is to move e-mail through the system.

Typically, e-mail systems support five basic functions. Composition refers to the process of creating messages and answers. Although any text editor can

be used for the body of the message, the system itself can provide assistance with addressing and the numerous header fields attached to each message. Transfer refers to moving messages from the originator to the recipient. In large part, this requires establishing a connection to the destination or some intermediate machine, outputting the message, and releasing the connection. The e-mail system should do this automatically, without bothering the user.

Reporting has to do with telling the originator what happened to the message. Was it delivered? Was it rejected? Was it lost? Numerous applications exist in which confirmation of delivery is important and may even have legal significance.

Displaying incoming messages is needed so people can read their e-mail. Sometimes conversion is required or a special viewer must be invoked. Disposition is the final step and concerns what the recipient does with the message after receiving it. Possibilities include throwing it away before reading, throwing it away after reading, saving it, and so on. It should also be possible to retrieve and reread saved messages, forward them, or process them in other ways. Most systems allow users to create mailboxes to store incoming e-mail. Commands are needed to create and destroy mailboxes, inspect the contents of mailboxes, insert and delete messages from mailboxes, and so on.

Corporate managers often need to send a message to each of their subordinates, customers, or suppliers. This gives rise to the idea of a mailing list, which is a list of e-mail addresses. When a message is sent to the mailing list, identical copies are delivered to everyone on the list. Other advanced features are carbon copies, blind carbon copies, high-priority e-mail, secret e-mail etc. The message inside the envelope consists of two parts: the header

and the body. The header contains control information for the user agents. The body is entirely for the human recipient.

The User Agent

E-mail systems have two basic parts, as we have seen: the user agents and the message transfer agents. A user agent is normally a program that accepts a variety of commands for composing, receiving, and replying to messages, as well as for manipulating mailboxes. Some user agents have a fancy menu- or icon-driven interface that requires a mouse, whereas others expect 1-character commands from the keyboard. Functionally, these are the same. Some systems are menu- or icon-driven but also have keyboard shortcuts.

Sending E-mail

To send an e-mail message, a user must provide the message, the destination address, and possibly some other parameters. The message can be produced with a free-standing text editor, a word processing program, or possibly with a specialized text editor built into the user agent. The destination address must be in a format that the user agent can deal with. Most e-mail systems support mailing lists, so that a user can send the same message to a list of people with a single command. If the mailing list is maintained locally, the user agent can just send a separate message to each intended recipient. However, if the list is maintained remotely, then messages will be expanded there.

Reading E-mail

Typically, when a user agent is started up, it looks at the user's mailbox for incoming e-mail before displaying anything on the screen. Then it may announce the number of messages in the mailbox or display a one-line summary of each one and wait for a command. Each line of the display contains several fields extracted from the envelope or header of the

corresponding message. In a simple e-mail system, the choice of fields displayed is built into the program. In a more sophisticated system, the user can specify which fields are to be displayed by providing a user profile, a file describing the display format. The third field tells how long the message is, and the fourth one tells who sent the message. Since this field is simply extracted from the message, this field may contain first names, full names, initials, login names, or whatever else the sender chooses to put there. Finally, the Subject field gives a brief summary of what the message is about. People who fail to include a Subject field often discover that responses to their e-mail tend not to get the highest priority.

After the headers have been displayed, the user can perform any of several actions, such as displaying a message, deleting a message, and so on. The older systems were text based and typically used one-character commands for performing these tasks, such as T (type message), A (answer message), D (delete message), and F (forward message). An argument specified the message in question. More recent systems use graphical interfaces. Usually, the user selects a message with the mouse and then clicks on an icon to type, answer, delete, or forward it.

Message Formats

Messages consist of a primitive envelope some number of header fields, a blank line, and then the message body. Each header field consists of a single line of ASCII text containing the field name, a colon, and, for most fields, a value. The user agent builds a message and passes it to the message transfer agent, which then uses some of the header fields to construct the actual envelope, a somewhat old-fashioned mixing of message and envelope. The basic header fields related to message transport are listed in here in table 8.1.

Header	Meaning
To:	E-mail address(es) of primary recipient(s)
Cc:	E-mail address(es) of secondary recipient(s)
Bcc:	E-mail address(es) for blind carbon copies
From:	Person or people who created the message
Sender:	E-mail address of the actual sender
Received:	Line added by each transfer agent along the route
Return-Path:	Can be used to identify a path back to the sender

Table 8.1: Header fields of E-mail

The To: field gives the DNS address of the primary recipient. Having multiple recipients is also allowed.

The Cc: field gives the addresses of any secondary recipients. In terms of delivery, there is no distinction between the primary and secondary recipients. It is entirely a psychological difference that may be important to the people involved but is not important to the mail system. The term Cc: (Carbon copy) is a bit dated, since computers do not use carbon paper, but it is well established.

The Bcc: (Blind carbon copy) field is like the Cc: field, except that this line is deleted from all the copies sent to the primary and secondary recipients. This feature allows people to send copies to third parties without the primary and secondary recipients knowing this.

The next two fields, From: and Sender:, tell who wrote and sent the message, respectively.

A line containing Received: is added by each message transfer agent along the way. The line contains the agent's identity, the date and time the message was received, and other information that can be used for finding bugs in the routing system.

The Return-Path: field is added by the final message transfer agent and was intended to tell how to get back to the sender

MIME—The Multipurpose Internet Mail Extensions

MIME defines five new message headers. The first of these simply tells the user agent receiving the message that it is dealing with a MIME message, and which version of MIME it uses. Any message not containing a MIME-Version: header is assumed to be an English plaintext message and is processed as such.

The Content-Description: header is an ASCII string telling what is in the message. This header is needed so the recipient will know whether it is worth decoding and reading the message.

The Content-Id: header identifies the content. It uses the same format as the standard Message-Id: header.

The Content-Transfer-Encoding: tells how the body is wrapped for transmission through a network that may object to most characters other than letters, numbers, and punctuation marks.

Message Transfer

The message transfer system is concerned with relaying messages from the originator to the recipient. The simplest way to do this is to establish a transport connection from the source machine to the destination machine and then just transfer the message. After examining how this is normally done, we will examine some situations in which this does not work and what can be done about them.

SMTP—The Simple Mail Transfer Protocol

Within the Internet, e-mail is delivered by having the source machine establish a TCP connection to port 25 of the destination machine. Listening to this port is an e-mail daemon that speaks SMTP (Simple Mail Transfer Protocol). This daemon accepts incoming connections and copies messages from them into the appropriate mailboxes. If a message cannot be delivered, an error report

containing the first part of the undeliverable message is returned to the sender.

SMTP is a simple ASCII protocol. After establishing the TCP connection to port 25, the sending machine, operating as the client, waits for the receiving machine, operating as the server, to talk first. The server starts by sending a line of text giving its identity and telling whether it is prepared to receive mail. If it is not, the client releases the connection and tries again later. If the server is willing to accept e-mail, the client announces whom the e-mail is coming from and whom it is going to. If such a recipient exists at the destination, the server gives the client the go-ahead to send the message. Then the client sends the message and the server acknowledges it. No checksums are needed because TCP provides a reliable byte stream. If there is more e-mail, that is now sent. When all the e-mail has been exchanged in both directions, the connection is released.

Final Delivery

The solution is to have a message transfer agent on an ISP machine accept e-mail for its customers and store it in their mailboxes on an ISP machine. Since this agent can be on-line all the time, e-mail can be sent to it 24 hours a day.

POP3

The situation that used to hold (both sender and receiver having a permanent connection to the Internet. A situation in which the sender is (currently) on-line but the receiver is not. POP3 begins when the user starts the mail reader. The mail reader calls up the ISP (unless there is already a connection) and establishes a TCP connection with the message transfer agent at port 110. Once the connection has been established, the POP3 protocol goes through three states in sequence:

1. Authorization.
2. Transactions.
3. Update.

The authorization state deals with having the user log in. The transaction state deals with the user collecting the e-mails and marking them for deletion from the mailbox. The update state actually causes the e-mails to be deleted.

Feature	POP3	IMAP
Where is protocol defined	RFC 1939	RFC 2060
TCP port used	110	143
Where is e-mail stored	User's PC	Server
Where is e-mail read	Off-line	On-line
Connect time required	Little	Much
Use of server resources	Minimal	Extensive
Multiple mailboxes	No	Yes
Who backs up mailboxes	User	ISP
Good for mobile users	No	Yes
User control over downloading	Little	Great
Partial message downloads	No	Yes
Are disk quotas a problem	No	Could be in time
Simple to implement	Yes	No
Widespread support	Yes	Growing

Table 8.2: A comparison of POP3 and IMAP.

IMAP

Internet Message Access Protocol, Unlike POP3, which basically assumes that the user will clear out the mailbox on every contact and work off-line after that, IMAP assumes that all the e-mail will remain on the server indefinitely in multiple mailboxes. IMAP provides extensive mechanisms for reading messages or even parts of messages, a feature useful when using a slow modem to read the text part of a multipart message with large audio and video attachments. Since the working assumption is that messages will not be

transferred to the user's computer for permanent storage, IMAP provides mechanisms for creating, destroying, and manipulating multiple mailboxes on the server. In this way a user can maintain a mailbox for each correspondent and move messages there from the inbox after they have been read. A comparison of POP3 and IMAP is given in table 8.2.

Delivery Features

Independently of whether POP3 or IMAP is used, many systems provide hooks for additional processing of incoming e-mail. An especially valuable feature for many e-mail users is the ability to set up filters. These are rules that are checked when e-mail comes in or when the user agent is started. Each rule specifies a condition and an action. Some ISPs provide a filter that automatically categorizes incoming e-mail as either important or spam (junk e-mail) and stores each message in the corresponding mailbox. Such filters typically work by first checking to see if the source is a known spammer. Then they usually examine the subject line. If hundreds of users have just received a message with the same subject line, it is probably spam. Other techniques are also used for spam detection.

8.3 SNMP Simple Network Management Protocol

The most widely adopted standard provides a way for devices and consoles to communicate. Manageability requires instrumentation in many devices from numerous vendors; it requires accessibility or transportability across the network, as well as on dedicated out-of-band channels; and it requires programming interfaces for a multitude of general purpose, as well as specialized, applications. It's all too evident that multivendor network management is impossible without a standard. In developing such a standard, the ISO bet on a horse named CMIP (common management information

protocol), while the Internet world bet on the Simple Network Management Protocol, or SNMP.

SNMP management has three components. The first component is the Structure of Management Information (SMI), which is a sort of toolkit for creating a management information base(MIB). The SMI identifies the permissible data types and spells out the rules for naming and identifying MIB components. It defines the structure of the SNMP naming mechanism. The MIB, which represents the second component of SNMP, is a layout or schema for information relevant to managing networks and their component devices. The third component, which the Simple Network Management Protocol itself defines, is the different possible payloads or Protocol Data Units (PDUs) that form legitimate management messages. SNMP is most often implemented over IP. In fact, UDP port numbers 161 and 162 identify SNMP agents and SNMP managers, respectively. But nothing in the standard prevents SNMP messages from being delivered with TCP, HTTP, or nonInternet protocols, such as IPX or AppleTalk's datagram delivery protocol.

The SMI

The SMI specifies the allowable data types in the MIB, and it spells out the ways data can be represented. Also, it defines a hierarchical naming structure that ensures unique, unambiguous names for managed objects, which are the components of a MIB. Compared with the objects that general purpose programmers use to build applications, SNMP-managed objects are very simple and stripped down. MIB objects typically have six or so attributes.

MIB

The MIB is a hierarchical name space, with the registration of its nodes administered by the Internet Assigned Numbers Authority (IANA). According to the hierarchy in the figure, the object ID for every relevant

MIB object must begin with either 1.3.6.1.2.1 (which represents the mib-2 node) or 1.3.6.1.4.1 (the enterprises node). The term MIB is also used to refer to specific collections of objects used for particular purposes. Thus, MIB objects under the mib-2 node include the RMON (remote monitoring) MIB objects and other generic MIB objects, while those under the enterprises node include all proprietary MIB objects. It has been estimated that there are 10 times as many proprietary MIB objects as there are generic ones.

Each MIB object has a value associated with it according to the syntax part of its specification—for example, the number of packets that have come in since the last system reset, the number of clock ticks since the last reset, or the administrative state of a router. When a MIB object is instantiated on a device or other entity, the value associated with the object is sometimes called a MIB variable. SNMP agents store MIB variables and send them to SNMP managing processes (or consoles) when they are asked to. The value of a MIB variable is the basic unit of useful management information.

MIBs are often divided into groups of related objects. MIB-2 has 10 groups, including "system," "interfaces," "ip," "tcp," "udp," and "snmp." The RMON MIB for Ethernet segments has nine groups, including "statistics," "history," "alarm," "event," and "capture." While RMON is an extension of the MIB, traditional SNMP agents are not capable of capturing most RMON data. Special RMON probe devices or built-in probe functions that can see an entire segment are necessary to collect and forward RMON information. An RMON probe not only captures more data and processes it more than ordinary

device agents, it can reduce traffic by storing intermediate results locally and forwarding them to an application on demand.

Agents And Managing Processes

SNMP managed entities—hubs, routers, database managers, toasters—must be outfitted with an agent that implements all the MIB objects relevant to them. The managing process must also be aware in advance of all the MIB objects implemented on entities it is expected to manage. Management consoles, such as HP OpenView, that communicate with many kinds of devices must allocate incremental resources for each different type of entity—it's no wonder these platforms are resource intensive.

SNMP proper describes the packet layout (or protocol data unit) for messages between management agents and managing processes. In the first version of SNMP, there are only five possible messages.

The GetRequest message with one or more object instances asks for the values of those object instances.

The GetResponse message returns the object instances with their values.

The GetNext-Request message provides an efficient method for the managing process to search tables of values.

The SetRequest message writes a value to the agent, providing SNMP with the ability to configure and control entities remotely. (It's probably not necessary to mention that managed objects with an access attribute of "read-only" can't be Set.) SetRequests are acknowledged with GetResponse messages. Finally, Traps are messages initiated by the SNMP agent and sent to the managing process. They are not acknowledged.

SNMP has a crude and insecure mechanism for access control and authentication in the form of a "password" called a community name or community string. Among other problems, the community name is generally

visible in unencrypted form in each SNMP packet—short work for any hacker with a protocol analyzer. With such severe potential threats, many network managers have been reluctant to implement the Set or write parts of SNMP, using it only to monitor devices and collect statistics. Of course, SNMP is a much less valuable technology as a result, and the potential for creating self-healing networks and other such marvels is cut off.

8.4 USENET

USENET Messages

The standard format for interchange of Network News articles among USER Network (USENET) sites is described here. It describes the format for articles themselves, and gives partial standards for transmission of news. The news transmission is not entirely standardized in order to give a good deal of flexibility to the individual hosts to choose transmission hardware and software, whether to batch news, and so on.

The primary consideration in choosing an article format is that it fit in with existing tools as well as possible. Existing tools include both implementations of mail and news. A standard format for mail messages has existed for many years on the ARPANET, and this format meets most of the needs of USENET. Since the ARPANET format is extensible, extensions to meet the additional needs of USENET are easily made within the ARPANET standard. Therefore, the rule is adopted that all USENET news articles must be formatted as valid ARPANET mail messages, according to the ARPANET standard RFC 822.

A message is included the following header or fields: *From, Newsgroups, Title, Article-I.D., Posted, Received, Expires, The body of the article etc*. Certain headers are required and certain headers are optional. Any unrecognized headers are allowed, and will be passed through unchanged. The required headers are *Relay-Version, Posting-Version, From, Date, Newsgroups, Subject, Message-ID, and Path*. The optional headers are *Followup-To, Date-Received, Expires, Reply-To, Sender, References, Control, Distribution, and Organization*.

Control Messages

The body of the Control header is the control message. Messages are a sequence of zero or more words, separated by white space (blanks or tabs). The first word is the name of the control message; remaining words are parameters to the message. The remainder of the header and the body of the message are also potential parameters. Some of the control headers are *Cancel, Ihave/Sendme, Newgroup, Rmgroup, Sendsys, Senduuname, and Version.*

Transmission Methods

USENET is not a physical network, but rather a logical network resting on top of several existing physical networks. These networks include, but are not limited to, UUCP, the ARPANET, an Ethernet, the BLICN network, an NSC Hyperchannel, and a Berknet. What is important is that two neighboring systems on USENET have some method to get a new article, in the format listed here, from one system to the other, and once on the receiving system, processed by the netnews software on that system.

It is not a requirement that USENET sites have mail systems capable of understanding the ARPA Internet mail syntax, but it is strongly recommended. Since From, Reply-To, and Sender lines use the Internet syntax, replies will be difficult or impossible without an Internet mailer. A site without an Internet mailer can attempt to use the Path header line for replies, but this field is not guaranteed to be a working path for replies. In any event, any site generating or forwarding news messages must have an internet address that allows them to receive mail from sites with internet mailers, and they must include their internet address on their From line.

Remote Execution

Some networks permit direct remote command execution. On these networks, news may be forwarded by spooling the rnews command with the article on the standard input. For example, if the remote system is called "remote", news would be sent over a UUCP link with the command "uux - remote!rnews", and on a Berknet, "net -mremote rnews". It is important that the article be sent via a reliable mechansim, normally involving the possibility of spooling, rather than direct real-time remote execution. This is because, if the remote system is down, a direct execution command will fail, and the article will never be delivered. If the article is spooled, it will eventually be delivered when both systems are up.

Transfer by Mail

On some systems, direct remote spooled execution is not possible. However, most systems support electronic mail, and a news article can

be sent as mail. One approach is to send a mail message, which is identical to the news message: the mail headers are the news headers, and the mail body is the news body. By convention, this mail is sent to the user "newsmail" on the remote machine. One problem with this method is that it may not be possible to convince the mail system that the From line of the message is valid, since the mail message was generated by a program on a system different from the source of the news article. Another problem is that error messages caused by the mail transmission would be sent to the originator of the news article, who has no control over news transmission between two cooperating hosts and does not know who to contact. Transmission error messages should be directed to a responsible contact person on the sending machine.

A solution to this problem is to encapsulate the news article into a mail message, such that the entire article (headers and body) is part of the body of the mail message. The convention here is that such mail is sent to user "rnews" on the remote system. A mail message body is generated by prepending the letter "N" to each line of the news article, and then attaching whatever mail headers are convenient to generate. The N's are attached to prevent any special lines in the news article from interfering with mail transmission, and to prevent any extra lines inserted by the mailer (headers, blank lines, etc.) from becoming part of the news article. A program on the receiving machine receives mail to "rnews", extracting the article itself and invoking the "rnews" program.

Batching

Since news articles are usually short, and since a large number of messages are often sent between two sites in a day, it may make sense to batch news articles. Several articles can be combined into one large article, using conventions agreed upon in advance by the two sites. One such batching scheme is described here; its use is still considered experimental.

The News Propagation Algorithm

This describes the overall scheme of USENET and the algorithm followed by sites in propagating news to the entire network. Since all sites are affected by incorrectly formatted articles and by propagation errors, it is important for the method to be standardized. USENET is a directed graph. Each node in the graph is a host computer; each arc in the graph is a transmission path from one host to another host. Each arc is labeled with a newsgroup pattern, specifying which newsgroup classes are forwarded along that link. Most arcs are bi-directional, that is, if site A sends a class of newsgroups to site B, then site B usually sends the same class of newsgroups to site A. This bi-directionality is not, however, required. USENET is made up of many sub-networks. Each subnet has a name, such as "net" or "btl". The special subnet "net" is defined to be USENET, although the union of all subnets may be a superset of USENET. Each subnet is a connected graph, that is, a path exists from every node to every other node in the subnet. In addition, the entire graph is connected.

An article is posted on one machine to a list of newsgroups. That machine accepts it locally, and then forwards it to all its neighbors that are interested in at least one of the newsgroups of the message. (Site A deems site B to be "interested" in a newsgroup if the newsgroup matches the pattern on the arc from A to B. This pattern is stored in a file on the A machine.) The sites receiving the incoming article examine it to make sure they really want the article, accept it locally, and then in turn forward the article to all their interest neighbors. This process continues until the entire network has seen the article.

An important part of the algorithm is the prevention of loops. The above process would cause a message to loop along a cycle forever. In particular, when site A sends an article to site B, site B will send it back to site A, which will send it to site B, and so on. One solution to this is the history mechanism. Each site keeps track of all articles it has seen (by their message ID) and whenever an article comes in that it has already seen, the incoming article is discarded immediately. This solution is sufficient to prevent loops, but additional optimizations can be made to avoid sending articles to sites that will simply throw them away.

One optimization is that an article should never be sent to a machine listed in the Path line of the header. When a machine name is in the Path line, the message is known to have passed through the machine. Another optimization is that, if the article originated on site A, then site A has already seen the article.

Thus, if an article is posted to newsgroup "net.misc", it will match the pattern "net.all" (where "all" is a metasymbol that matches any string), and will be forwarded to all sites that subscribe to net.all (as determined by what their neighbors send them). These sites make up the "net" subnetwork. An article posted to "btl.general" will reach all sites receiving "btl.all", but will not reach sites that do not get "btl.all". In effect, the article reaches the "btl" subnetwork. An article posted to newsgroups "net.micro, btl.general" will reach all sites subscribing to either of the two classes.

8.5 WWW - The World Wide Web

The World Wide Web is an architectural framework for accessing linked documents spread out over millions of machines all over the Internet.

Architectural Overview

The Web consists of a vast, worldwide collection of documents or Web pages, often just called pages for short. Each page may contain links to other pages anywhere in the world. Users can follow a link by clicking on it, which then takes them to the page pointed to. This process can be repeated indefinitely. The idea of having one page point to another, now called hypertext. Pages are viewed with a program called a browser, the browser fetches the page requested, interprets the text and formatting commands on it, and displays the page, properly formatted, on the screen. Like many Web pages, this one starts with a title, contains some information, and ends with the e-mail address of the page's maintainer. Strings of text that are links to other pages, called hyperlinks, are often highlighted, by underlining, displaying them in a

special color, or both. To follow a link, the user places the mouse cursor on the highlighted area, which causes the cursor to change, and clicks on it.

The basic model of how the Web works is Here the browser is displaying a Web page on the client machine. When the user clicks on a line of text that is linked to a page on the server, the browser follows the hyperlink by sending a message to the server asking it for the page. When the page arrives, it is displayed. If this page contains a hyperlink to a page on another server that is clicked on, the browser then sends a request to that machine for the page, and so on indefinitely.

The Client Side

A browser is a program that can display a Web page and catch mouse clicks to items on the displayed page. When an item is selected, the browser follows the hyperlink and fetches the page selected. Therefore, the embedded hyperlink needs a way to name any other page on the Web. Pages are named using URLs (Uniform Resource Locators). URL has three parts: the name of the protocol (http), the DNS name of the machine where the page is located and the name of the file containing the page. When a user clicks on a hyperlink, the browser carries out a series of steps in order to fetch the page pointed to. To be able to display the new page (or any page), the browser has to understand its format. To allow all browsers to understand all Web pages, Web pages are written in a standardized language called HTML, which describes Web pages.

The Server Side

When the user types in a URL or clicks on a line of hypertext, the browser parses the URL and interprets the part between http:// and the next slash as a DNS name to look up. Armed with the IP address of the server, the browser establishes a TCP connection to port 80 on that server. Then it sends over a

command containing the rest of the URL, which is the name of a file on that server. The server then returns the file for the browser to display. That server, like a real Web server, is given the name of a file to look up and return. In both cases, the steps that the server performs in its main loop are:

1. Accept a TCP connection from a client (a browser).
2. Get the name of the file requested.
3. Get the file (from disk).
4. Return the file to the client.
5. Release the TCP connection.

Modern Web servers have more features, but in essence, this is what a Web server does. Modern Web servers do more than just accept file names and return files. In fact, the actual processing of each request can get quite complicated. For this reason, in many servers each processing module performs a series of steps. The front end passes each incoming request to the first available module, which then carries it out using some subset of the following steps, depending on which ones are needed for that particular request.

1. Resolve the name of the Web page requested.
2. Authenticate the client.
3. Perform access control on the client.
4. Perform access control on the Web page.
5. Check the cache.
6. Fetch the requested page from disk.
7. Determine the MIME type to include in the response.
8. Take care of miscellaneous odds and ends.
9. Return the reply to the client.
10. Make an entry in the server log.

URLs—Uniform Resource Locators

Each page is assigned a URL (Uniform Resource Locator) that effectively serves as the page's worldwide name. URLs have three parts: the protocol (also known as the scheme), the DNS name of the machine on which the page is located, and a local name uniquely indicating the specific page (usually just a file name on the machine where it resides). This URL consists of three parts: the protocol, the DNS name of the host, and the file name, with certain punctuation separating the pieces.

Cookies

When a client requests a Web page, the server can supply additional information along with the requested page. This information may include a cookie, which is a small file. Browsers store offered cookies in a cookie directory on the client's hard disk unless the user has disabled cookies. Cookies are just files or strings, not executable programs. In principle, a cookie could contain a virus, but since cookies are treated as data, there is no official way for the virus to actually run and do damage. However, it is always possible for some hacker to exploit a browser bug to cause activation.

A cookie may contain up to five fields, the Domain tells where the cookie came from. Browsers are supposed to check that servers are not lying about their domain. Each domain may store no more than 20 cookies per client. The Path is a path in the server's directory structure that identifies which parts of the server's file tree may use the cookie. It is often /, which means the whole tree. The Content field takes the form name = value. Both name and value can be anything the server wants. This field is where the cookie's content is stored. The Expires field specifies when the cookie expires. If this field is absent, the browser discards the cookie when it exits. Such a cookie is called

a nonpersistent cookie. If a time and date are supplied, the cookie is said to be persistent and is kept until it expires. Finally, the Secure field can be set to indicate that the browser may only return the cookie to a secure server. This feature is used for e-commerce, banking, and other secure applications.

Static Web Documents

The basis of the Web is transferring Web pages from server to client. In the simplest form, Web pages are static, that is, are just files sitting on some server waiting to be retrieved. In this context, even a video is a static Web page because it is just a file.

Dynamic Web Documents

In recent years, more and more content has become dynamic, that is, generated on demand, rather than stored on disk. Content generation can take place either on the server side or on the client side

HTTP—The HyperText Transfer Protocol

The transfer protocol used throughout the World Wide Web is HTTP. It specifies what messages clients may send to servers and what responses they get back in return. Each interaction consists of one ASCII request, followed by one MIME-like response. All clients and all servers must obey this protocol.

Connections

The usual way for a browser to contact a server is to establish a TCP connection to port 80 on the server's machine, although this procedure is not formally required. The value of using TCP is that neither browsers nor servers have to worry about lost messages, duplicate messages, long messages, or acknowledgements. All of these matters are handled by the TCP implementation.

Methods

Each request consists of one or more lines of ASCII text, with the first word on the first line being the name of the method requested. The built-in methods are listed in table 8.3.

Method	Description
GET	Request to read a Web page
HEAD	Request to read a Web page's header
PUT	Request to store a Web page
POST	Append to a named resource (e.g., a Web page)
DELETE	Remove the Web page
TRACE	Echo the incoming request
CONNECT	Reserved for future use
OPTIONS	Query certain options

Table 8.3: Built in Methods

For accessing general objects, additional object-specific methods may also be available. The names are case sensitive, so GET is a legal method but get is not.

Message Headers

The request line may be followed by additional lines with more information. They are called request headers. This information can be compared to the parameters of a procedure call. Responses may also have response headers. Some headers can be used in either direction.

The User-Agent header allows the client to inform the server about its browser, operating system, and other properties.

The Accept headertell the server what the client is willing to accept in the event that it has a limited repertoire of what is acceptable.

The Host header names the server. It is taken from the URL. It is used because some IP addresses may serve multiple DNS names and the server needs some way to tell which host to hand the request to.

The Authorization header is needed for pages that are protected.

The Cookie header is used by clients to return to the server a cookie that was previously sent by some machine in the server's domain.

The Date header can be used in both directions and contains the time and date the message was sent.

The Upgrade header is used to make it easier to make the transition to a future version of the HTTP protocol.

The next headers, starting with Content-, allow the server to describe properties of the page it is sending.

The Last-Modified header tells when the page was last modified. This header plays an important role in page caching.

The Location header is used by the server to inform the client that it should try a different URL

The Accept-Ranges header announces the server's willingness to handle this type of partial page request.

The cookie header, Set-Cookie, is how servers send cookies to clients. The client is expected to save the cookie and return it on subsequent requests to the server.

8.8 Multimedia

Multimedia is just two or more media. If the publisher of this book wanted to join the current hype about multimedia, it could advertise the book as using multimedia technology. After all, it contains two media: text and graphics (the figures). Nevertheless, when most people refer to multimedia, they generally

mean the combination of two or more continuous media, that is, media that have to be played during some well-defined time interval, usually with some user interaction. In practice, the two media are normally audio and video, that is, sound plus moving pictures.

Introduction to Digital Audio

An audio wave is a one-dimensional acoustic wave. When an acoustic wave enters the ear, the eardrum vibrates, causing the tiny bones of the inner ear to vibrate along with it, sending nerve pulses to the brain. These pulses are perceived as sound by the listener. In a similar way, when an acoustic wave strikes a microphone, the microphone generates an electrical signal, representing the sound amplitude as a function of time. The representation, processing, storage, and transmission of such audio signals are a major part of the study of multimedia systems. The frequency range of the human ear runs from 20 Hz to 20,000 Hz. Digitized sound can be easily processed by computers in software. Dozens of programs exist for personal computers to allow users to record, display, edit, mix, and store sound waves from multiple sources. Virtually all professional sound recording and editing are digital nowadays.

Audio Compression

CD-quality audio requires a transmission bandwidth of 1.411 Mbps. Clearly, substantial compression is needed to make transmission over the Internet practical. For this reason, various audio compression algorithms have been developed Audio compression can be done in one of two ways. In waveform coding the signal is transformed mathematically by a Fourier transform into its frequency components. The amplitude of each component is then encoded in a minimal way. The goal is to reproduce the waveform accurately at the other

end in as few bits as possible. The audio compression is done by sampling the waveform at 32 kHz, 44.1 kHz, or 48 kHz. Sampling can be done on one or two channels, in any of four configurations:

1. Monophonic (a single input stream).
2. Dual monophonic (e.g., an English and a Japanese soundtrack).
3. Disjoint stereo (each channel compressed separately).
4. Joint stereo (interchannel redundancy fully exploited).

Streaming Audio

It is, listening to sound over the Internet. This is also called music on demand. The Internet is full of music Web sites, many of which list song titles that users can click on to play the songs. To get around this problem without changing how the browser works, music sites have come up with the following scheme. The file linked to the song title is not the actual music file. Instead, it is what is called a metafile, a very short file just naming the music.

In most cases, the server named in the metafile is not the same as the Web server. In fact, it is generally not even an HTTP server, but a specialized media server. The media player has four major jobs to do:

1. Manage the user interface.
2. Handle transmission errors.
3. Decompress the music.
4. Eliminate jitter.

Internet Radio

Commercial radio stations got the idea of broadcasting their content over the Internet as well as over the air. With current technology, virtually anyone can start a radio station. The whole area of Internet radio is very new and in a state of flux, but it is worth saying a little bit about. There are two general approaches to Internet radio. In the first one, the programs are prerecorded

and stored on disk. Listeners can connect to the radio station's archives and pull up any program and download it for listening. It is also possible to store each program just after it is broadcast live, so the archive is only running, say, half an hour, or less behind the live feed. The advantages of this approach are that it is easy to do, all the techniques we have discussed work here too, and listeners can pick and choose among all the programs in the archive. The other approach is to broadcast live over the Internet. Some stations broadcast over the air and over the Internet simultaneously, but there are increasingly many radio stations that are Internet only.

Voice over IP

Once upon a time, the public switched telephone system was primarily used for voice traffic with a little bit of data traffic here and there. Many packet-switching network operators suddenly became interested in carrying voice over their data networks. The amount of additional bandwidth required for voice is minuscule since the packet networks are dimensioned for the data traffic. However, the average person's phone bill is probably larger than his Internet bill, so the data network operators saw Internet telephony as a way to earn a large amount of additional money without having to put any new fiber in the ground. Thus Internet telephony (also known as voice over IP).

Introduction to Video

If a sequence of images is drawn line by line at 50 images/sec, the eye does not notice that it is looking at discrete images. All video television systems exploit this principle to produce moving pictures. The simplest representation of digital video is a sequence of frames, each consisting of a rectangular grid of picture elements, or pixels. Each pixel can be a single bit, to represent

either black or white. The quality of such a system is similar to what you get by sending a color photograph by fax—awful.

To produce smooth motion, digital video, like analog video, must display at least 25 frames/sec. However, since good-quality computer monitors often rescan the screen from images stored in memory at 75 times per second or more, interlacing is not needed and consequently is not normally used. Just repainting the same frame three times in a row is enough to eliminate flicker. In other words, smoothness of motion is determined by the number of different images per second, whereas flicker is determined by the number of times the screen is painted per second. These two parameters are different. A still image painted at 20 frames/sec will not show jerky motion, but it will flicker because one frame will decay from the retina before the next one appears. A movie with 20 different frames per second, each of which is painted four times in a row, will not flicker, but the motion will appear jerky.

Video Compression

All compression systems require two algorithms: one for compressing the data at the source, and another for decompressing it at the destination. These algorithms are referred to as the encoding and decoding algorithms, respectively. These algorithms exhibit certain asymmetries that are important to understand. First, for many applications, a multimedia document, say, a movie will only be encoded once but will be decoded thousands of times. This asymmetry means that it is acceptable for the encoding algorithm to be slow and require expensive hardware provided that the decoding algorithm is fast and does not require expensive hardware. After all, the operator of a multimedia server might be quite willing to rent a parallel supercomputer for a few weeks to encode its entire video library, but requiring consumers to rent a supercomputer for 2 hours to view a video is not likely to be a big success.

Many practical compression systems go to great lengths to make decoding fast and simple, even at the price of making encoding slow and complicated.

On the other hand, for real-time multimedia, such as video conferencing, slow encoding is unacceptable. Encoding must happen on-the-fly, in real time. Consequently, real-time multimedia uses different algorithms or parameters than storing videos on disk, often with appreciably less compression. A second asymmetry is that the encode/decode process need not be invertible. That is, when compressing a file, transmitting it, and then decompressing it, the user expects to get the original back, accurate down to the last bit. With multimedia, this requirement does not exist. It is usually acceptable to have the video signal after encoding and then decoding be slightly different from the original. When the decoded output is not exactly equal to the original input, the system is said to be lossy. If the input and output are identical, the system is lossless. Lossy systems are important because accepting a small amount of information loss can give a huge payoff in terms of the compression ratio possible.

The JPEG Standard

The JPEG (Joint Photographic Experts Group) standard for compressing continuous-tone still pictures was developed by photographic experts working under the joint auspices of ITU, ISO, and IEC, another standards body. It is important for multimedia because, to a first approximation, the multimedia standard for moving pictures, MPEG, is just the JPEG encoding of each frame separately, plus some extra features for interframe compression and motion detection. JPEG is defined in International Standard 10918.

The MPEG Standard

MPEG (Motion Picture Experts Group) standards. These are the main algorithms used to compress videos and have been international standards

since 1993. Because movies contain both images and sound, MPEG can compress both audio and video. MPEG-1 has three parts: audio, video, and system, which integrates the other two. The audio and video encoders work independently, which raises the issue of how the two streams get synchronized at the receiver. This problem is solved by having a 90-kHz system clock that outputs the current time value to both encoders. These values are 33 bits, to allow films to run for 24 hours without wrapping around. These timestamps are included in the encoded output and propagated all the way to the receiver, which can use them to synchronize the audio and video streams.

Video on Demand

Video on demand is sometimes compared to an electronic video rental store. The user selects any one of a large number of available videos and takes it home to view. Only with video on demand, the selection is made at home using the television set's remote control, and the video starts immediately. No trip to the store is needed. Needless to say, implementing video on demand is a wee bit more complicated than describing it. In particular, video rental users are used to the idea of being able to stop a video, make a quick trip to the kitchen or bathroom, and then resume from where the video stopped. Television viewers do not expect to put programs on pause. If video on demand is going to compete successfully with video rental stores, it may be necessary to allow users to stop, start, and rewind videos at will. Giving users this ability virtually forces the video provider to transmit a separate copy to each one.

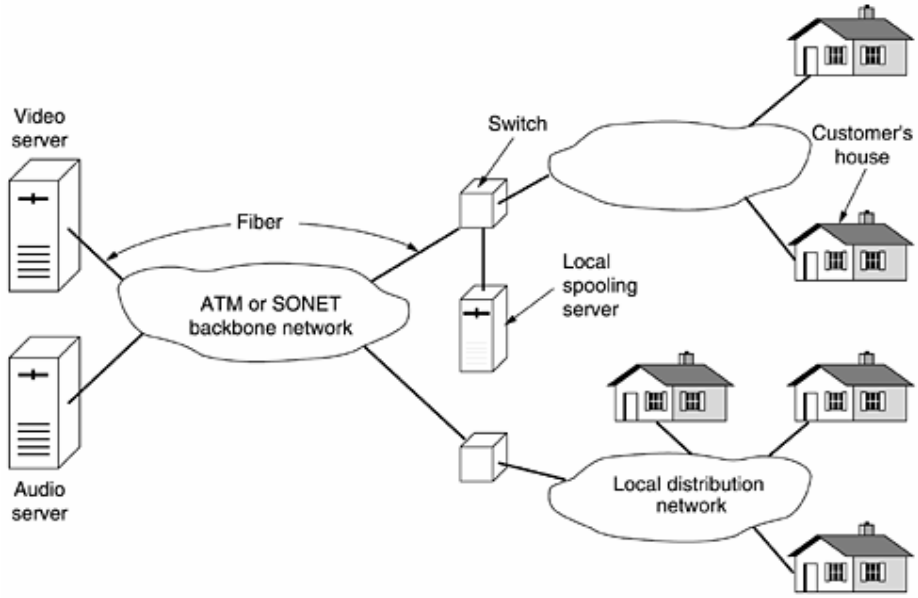


Figure 8.2: Overview of a video-on-demand system.

Attached to the backbone by high-bandwidth optical fibers are numerous information providers as shown in figure 8.2.Video Servers capable of storing and outputting a large number of movies simultaneously. The CPU software also defines the nature of the interface that the server presents to the clients. The distribution network is the set of switches and lines between the source and destination. It consists of a backbone, connected to a local distribution network. Usually, the backbone is switched and the local distribution network is not. The main requirement imposed on the backbone is high bandwidth. It used to be that low jitter was also a requirement, but with even the smallest PC now able to buffer 10 sec of high-quality MPEG-2 video, low jitter is not a requirement anymore. Local distribution is highly chaotic, with different companies trying out different networks in different regions.

8.9 ISDN

Rapid advances in computer and communication technologies have resulted in the increasing merging of these two fields. The lines have blurred among computing, switching, and digital transmission equipment, and the same digital techniques are being used for data, voice, and image transmission. Merging and evolving technologies, coupled with increasing demands for efficient and timely collection, processing, and dissemination of information, are leading to the development of integrated systems that transmit and process all types of data. The ultimate goal of this evolution is the integrated services digital network (ISDN). The ISDN is intended to be a worldwide public telecommunications network to replace existing public telecommunications networks and deliver a wide variety of services. The ISDN is defined by the standardization of user interfaces and is implemented as a set of digital switches and paths supporting a broad range of traffic types and providing value-added processing services.

ISDN Concept

The concept of ISDN is best introduced by considering it from several different viewpoints:

- Principles of ISDN
- The user interface
- Objectives
- Services

Principles of ISDN

Standards for ISDN have been defined by ITU-T (formerly CCITT. ISDN-related standards, states the principles of ISDN from the point of view of CCITT. Let us look at each of these points in turn:

- Support of voice and non-voice applications using a limited set of standardized facilities.
- Support for switched and non-switched applications.
- Reliance on 64-kbps connections.
- Intelligence in the network.
- Layered protocol architecture.
- Variety of configurations.

The User Interface

Figure 8.3 given below is a conceptual view of the lSDN from a user, or customer, point of view. The user has access to the ISDN by means of a local interface to a "digital pipe" of a certain bit rate. Pipes of various sizes are available to satisfy differing needs. At any given point in time, the pipe to the user's premises has a fixed capacity, but the traffic on the pipe may be a variable mix up to the capacity limit. Thus, a user may access circuit-switched and packet-switched services, as well as other services, in a dynamic mix of signal types and bit rates. To provide these services, the ISDN requires rather complex control signals to instruct it how to sort out the time multiplexed data and provide the required services. These control signals are also multiplexed onto the same digital pipe.

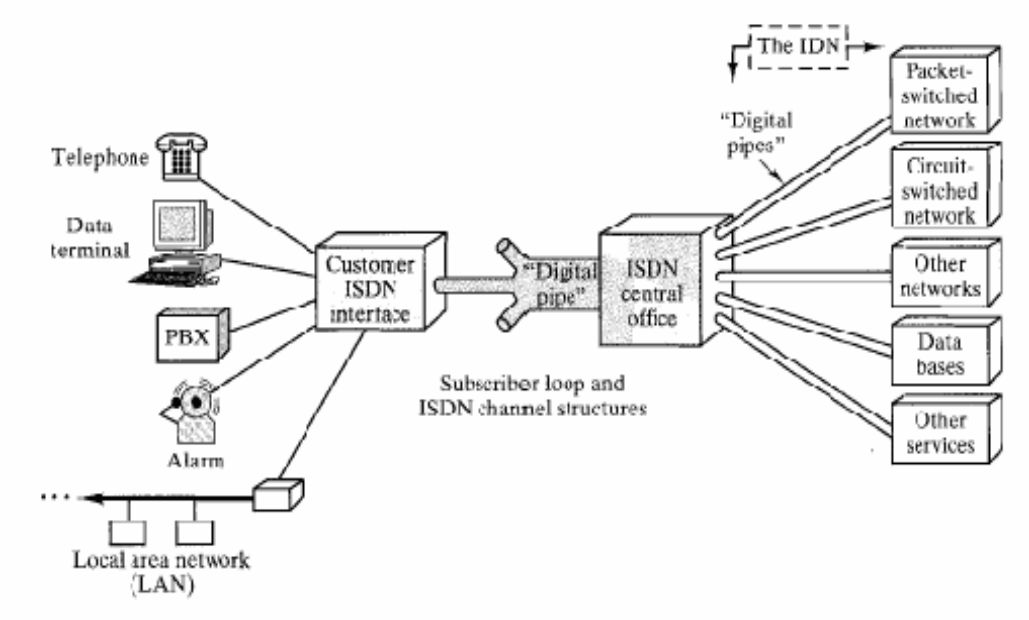


Figure 8.3: Conceptual view of ISDN connections

An important aspect of the interface is that the user may, at any time, employ less than the maximum capacity of the pipe, and will be charged according to the capacity used rather than "connect time." This characteristic significantly diminishes the value of current user design efforts that are geared to optimize circuit utilization by use of concentrators, multiplexers, packet switches, and other line sharing arrangements.

Objectives

Activities currently under way are leading to the development of a worldwide ISDN. Certain common objectives are as given below:

Standardization. It is essential that a single set of ISDN standards be provided to permit universal access and to permit the development of cost-effective equipment.

Transparency. The most important service to be provided is a transparent transmission service, thereby permitting users to develop applications and protocols with the confidence that they will not be affected by the underlying ISDN.

Separation of competitive functions. It must be possible to separate out functions that could be provided competitively as opposed to those that are fundamentally part of the ISDN. In many countries, a single, government-owned entity provides all services. Some countries desire that certain enhanced services be offered competitively (e.g., videotex, electronic mail).

Leased and switched services. The ISDN should provide dedicated point-to point services as well as switched services, thereby allowing the user to optimize implementation of switching and routing techniques.

Cost-related tariffs. The price for ISDN service should be related to cost, and should be independent of the type of data being carried. One type of service should not be in the position of subsidizing others.

Smooth migration. The conversion to ISDN will be gradual, and the evolving network must coexist with existing equipment and services. Thus, ISDN interfaces should evolve from current interfaces, and provide a migration path for users.

Multiplexed support. In addition to providing low-capacity support to individual users, multiplexed support must be provided to accommodate user owned PBX and local network equipment.

Architecture

Figure 8.4 given below is a block diagram of ISDN. ISDN supports a new physical connecter for users, a digital subscriber loop (link from end user to central or end office), and modifications to all central office equipment. The area to which most attention has been paid by standards organizations is that of user access. A common physical interface has been defined to provide, in essence, a DTE-DCE connection. The same interface should be usable for telephone, computer terminal, and video-tex terminal. Protocols are needed for the exchange of control information between user device and the network. Under the ISDN, one or two twisted pairs are used to provide a basic full-duplex digital communications link. The digital central office connects the numerous ISDN subscriber loop signals to the IDN.

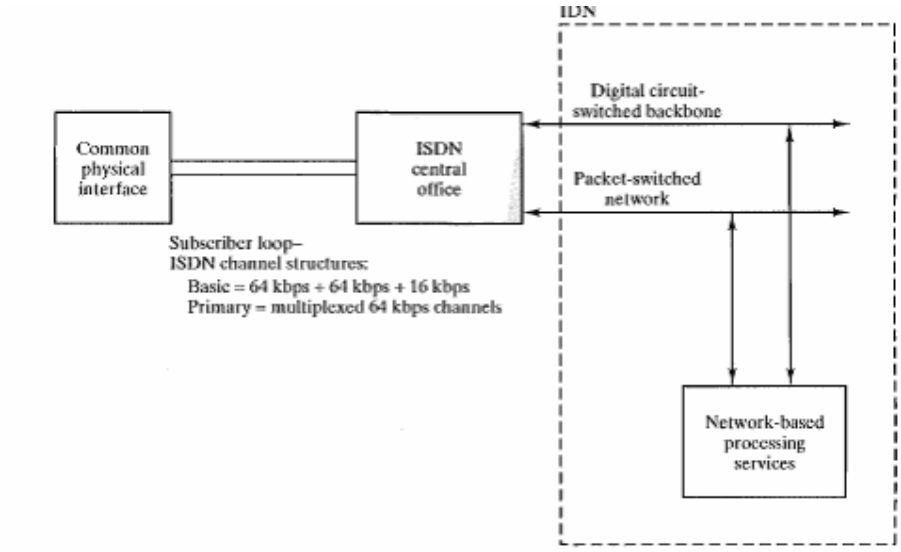


FIGURE 8.4: Block diagram of ISDN functions.

Standards

The development of ISDN is governed by a set of recommendations issued by ISDN, called the I-series Recommendations. The characterization of ISDN contained in these Recommendations is centered on three main areas:

1. The standardization of services offered to users, so as to enable services to be internationally compatible.

2. The standardization of user-network interfaces, so as to enable terminal equipment to be portable, and to assist in (1).

3. The standardization of ISDN capabilities to the degree necessary to allow user-network and network-network interworking, and thus achieve (1) and (2).

The I-series Recommendations are broken up into six main groupings, labeled 1.100 through 1.600.

- 1.100 Series-General Concepts
- 1.200 Series-Service Capabilities
- 1.300 Series-Network Aspects
- 1.400 Series-User-Network Interfaces
    - Physical configurations.
    - Transmission rates.
    - Protocol specifications.
- 1.500 Series-Internetwork Interfaces
- 1.600 Series-Maintenance Principles

ISDN CHANNELS

The digital pipe between the central office and the ISDN user is used to carry a number of communication channels. The capacity of the pipe, and therefore the number of channels carried, may vary from user to user. The transmission structure of any access link is constructed from the following types of channels:

- B channel: 64 kbps
- D channel: 16 or 64 kbps
- H channel: 384(H0), 1536(H11), and 1920 (H12) kbps

The B channel is the basic user channel. It can be used to carry digital data, PCM-encoded digital voice, or a mixture of lower-rate traffic, including digital data and digitized voice encoded at a fraction of 64 kbps. In the case of mixed traffic, all traffic must be destined for the

same endpoint. Four kinds of connections can be set up over a B channel:

Circuit-switched. This is equivalent to switched digital service available today. The user places a call, and a circuit-switched connection is established with another network user. An interesting feature is that call-establishment dialogue does not take place over the B channel, but is done over the D, as explained below.

Packet-switched. The user is connected to a packet-switching node, and data are exchanged with other users via X.25.

Frame mode. The user is connected to a frame relay node, and data are exchanged with other users via LAPF.

Semipermanent. This is a connection to another user set up by prior arrangement, and not requiring a call-establishment protocol; this is equivalent to a leased line.

The D *channel* serves two purposes. First, it carries signaling information to control circuit-switched calls on associated B channels at the user interface. In addition, the D channel may be used for packet-switching or low-speed telemetry at times when no signaling information is waiting. The channel function are given in table 8.4

ISDN Channel functions.

B Channel (64 kbps)	D Channel (16 kbps)
Digital voice	Signaling
64 kbps PCM	Basic
Low bit rate (32 kbps)	Enhanced
High-speed data	Low-speed data
Circuit-switched	Videotex
Packet-switched	Teletex
Other	Terminal
Facsimile	Telemetry
Slow-scan video	Emergency services
	Energy management

Table 8.4: ISDN Channel Functions

H *channels* are provided for user information at higher bit rates. The user may employ such a channel as a high-speed trunk, or the channel may be subdivided according to the user's own TDM scheme. Examples of applications include fast facsimile, video, high-speed data, high-quality audio, and multiple information streams at lower data rates.

These channel types are grouped into transmission structures that are offered as a package to the user. The best-defined structures at this time are the basic channel structure (basic access) and the primary channel structure (primary access), which are depicted in Figure 8.5.

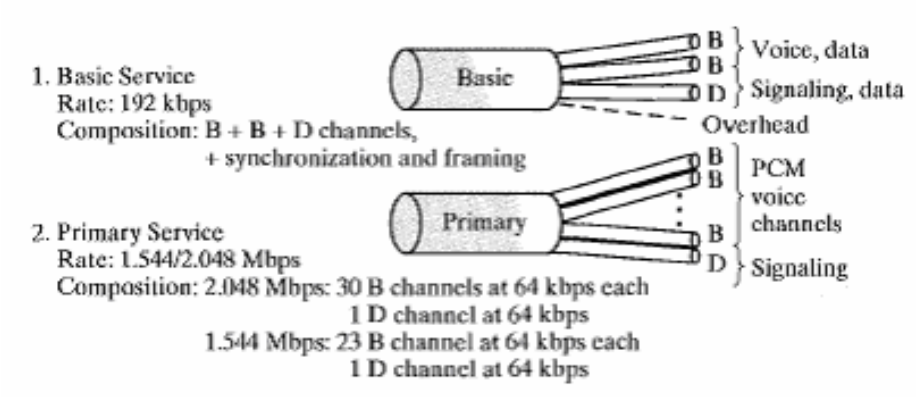


Figure 8.5: ISDN channel structure

Basic access consists of two full-duplex 64-kbps B channels and a full-duplex 16-kbps D channel. The total bit rate, by simple arithmetic, is 144 kbps. However, framing, synchronization, and other overhead bits bring the total bit rate on a basic access link to 192 kbps. Each frame of 48 bits includes 16 bits from each of the B channels and 4 bits from the D channel.

The basic service is intended to meet the needs of most individual users, including residential and very small offices. It allows the simultaneous use of voice and several data applications, such as packet-switched access, a link to a central alarm service, facsimile, videotex, and so on. These services could be accessed through a single multifunction terminal or several separate terminals. In either case, a single physical interface is provided. Most existing two-wire local loops can support this interface.

In some cases, one or both of the B channels remain unused; this results in a B+D or D interface, rather than the 2B+D interface. However, to simplify the network implementation, the data rate at the interface remains at 192 kbps. Nevertheless, for those subscribers with more modest transmission requirements, there may be a cost savings in using a reduced basic interface.

Primary access is intended for users with greater capacity requirements, such as offices with a digital PBX or a local network. Because of differences in the digital transmission hierarchies used in

different countries, it was not possible to get agreement on a single data rate. The primary interface may also be used to support H channels. Some of these structures include a 64-kbps D channel for control signaling. When no D channel is present, it is assumed that a D channel on another primary interface at the same subscriber location will provide any required signaling. The following structures are recognized:

Primary rate interface HO channel structures. This interface supports multiple 384-kbps HO channels. The structures are 3H0 + D and 4H0 for the 1.544-Mbps interface, and 5H0 + D for the 2.048-Mbps interface.

Primary rate interface H1 channel structures. The HI1 channel structure consists of one 1536-kbps HI1 channel. The H12 channel structure consists of one 1920-kbps H12 channel and one D channel.

Primary rate interface structures for mixtures of B and HO channels. This interface consists of 0 or 1 D channels plus any possible combination of B and HO channels, up to the capacity of the physical interface (e.g., 3H0 + 5B + D and 3H0 + 6B).

USER ACCESS

To define the requirements for ISDN user access, an understanding of the anticipated configuration of user premises equipment and of the necessary standard interfaces is critical. The first step is to group functions that may exist on the user's premises.

Functional groupings. Certain finite arrangements of physical equipment or combinations of equipment.

Reference points. Conceptual points used to separate groups of functions.

The architecture on the subscriber's premises is broken up functionally into groupings separated by reference points. This separation permits interface standards to be developed at each reference point; this effectively organizes the standards work and provides guidance to the equipment providers. Once stable interface standards exist, technical improvements on either side of an interface can be made without impacting adjacent functional groupings. Finally, with stable interfaces, the subscriber is free to procure equipment from different suppliers for the various functional groupings, so long as the equipment conforms to the relevant interface standards.

Network termination 1 (NT1) includes functions associated with the physical and electrical termination of the ISDN on the user's premises; these correspond to OSI layer1.

Network termination 2 (NT2) is an intelligent device that can perform switching and concentration functions; it may include functionality up through layer 3 of the OSI model.

Network termination 1, 2 (NT12) is a single piece of equipment that contains the combined functions of NT1 and NT2; this points out one of the regulatory issues associated with ISDN interface development.

Terminal equipment type 1 (TE1) refers to devices that support the standard ISDN interface. Examples are digital telephones, integrated voice-data terminals, and digital facsimile equipment. Terminal equipment type 2 (TE2) encompasses existing non-ISDN equipment.

Reference point T (terminal) corresponds to a minimal ISDN network termination at the customer's premises; it separates the network provider's equipment from the user's equipment.

Reference point S (system) corresponds to the interface of individual ISDN terminals and separates user terminal equipment from network-related communications functions. ***Reference point R*** (rate) provides a non-ISDN interface between user equipment that is not ISDN-compatible and adapter equipment.

ISDN Protocol Architecture

Figure 8.6 illustrates, in the context of the OSI model, the protocols defined or referenced in the ISDN documents. As a network, ISDN is essentially unconcerned with user layers 4-7. These are end-to-end layers employed by the user for the exchange of information. Network access is concerned only with layers 1-3. Layer 1, specifies the physical interface for both basic and primary access. Because B and D channels are multiplexed over the same physical interface, these standards apply to both types of channels. Above this layer, the protocol structure differs for the two channels.

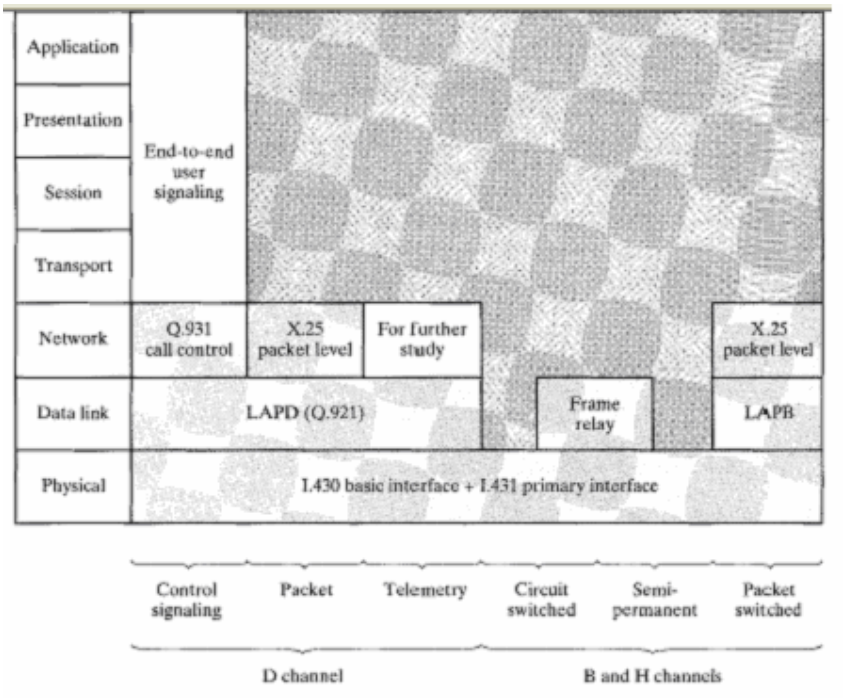


Figure 8.6: ISDN protocol at the user-network interface

For the D channel, a new data link layer standard, LAPD (Link Access Protocol, D channel) has been defined. This standard is based on HDLC, modified to meet ISDN requirements. All transmission on the D channel is in the form of LAPD frames that are exchanged between the subscriber equipment and an ISDN switching element. Three applications are supported: control signaling, packet switching, and telemetry. For control signaling, a call control protocol has been defined. This protocol is used to establish, maintain, and terminate connections on B channels; thus, it is a protocol between the user and the network. Above

layer 3, there is the possibility for higher-layer functions associated with user-to-user control signaling.

The D channel can also be used to provide *packet-switching* services to the subscriber. In this case, the X.25 level-3 protocol is used, and X.25 packets are transmitted in LAPD frames. The X.25 level-3 protocol is used to establish virtual circuits on the D channel to other users and to exchange packetized data. The B channel can be used for circuit switching, semipermanent circuits, and packet switching. For *circuit switching,* a circuit is set up on a B channel, on demand. The D-channel call control protocol is used for this purpose. Once the circuit is set up, it may be used for data transfer between the users. A *semipermanent circuit* is a B-channel circuit that is set up by prior agreement between the connected users and the network. As with a circuit-switched connection, it provides a transparent data path between end systems. With either a circuit-switched connection or a semipermanent circuit, it appears to the connected stations that they have a direct, full-duplex link with each other. They are free to use their own formats, protocols, and frame synchronization. Hence, from the point of view of ISDN, layers 2 through 7 are not visible or specified. In the case of *packet-switching,* a circuit-switched connection is set up on a B channel between the user and a packet-switched node using the D-channel control protocol. Once the circuit is set up on the B channel, the user may employ Frame relay or X.25 layers 2 and 3 to establish a virtual

circuit to another user over that channel and to exchange packetized data.

ISDN Connections

ISDN provides four types of service for end-to-end communication:

- Circuit-switched calls over a B channel.
- Semipermanent connections over a B channel.
- Packet-switched calls over a B channel.
- Packet-switched calls over the D channel.

Circuit-Switched Calls

The network configuration and protocols for circuit switching involve both the B and D channels. The B channel is used for the transparent exchange of user data. The communicating users may employ any protocols they wish for end-to-end communication. The D channel is used to exchange control information between the user and the network for call establishment and termination, as well as to gain access to network facilities.

Semipermanent Connections

A semipermanent connection between agreed points may be provided for an indefinite period of time after subscription, for a fixed period, or for agreed-upon periods during a day, a week, or some other interval. As with circuit-switched connections, only Layer-1 functionality is provided by the network interface. The call-control protocol is not needed because the connection already exists.

Packet-Switched Calls over a B Channel

The ISDN must also permit user access to packet-switched services for data traffic that is best serviced by packet switching. There are two possibilities for implementing this service: Either the packet-switching capability is furnished by a separate network, referred to as a packet-switched public data network (PSPDN), or the packet-switching capability is integrated into ISDN.

Packet-Switched Calls over a D Channel

When the packet-switching service is provided internal to the ISDN, it can also be accessed on the D channel. For this access, ISDN provides a semipermanent connection to a packet-switching node within the ISDN. The user employs the X.25 level-3 protocol, as is done in the case of a B-channel virtual call.

LAPD

All traffic over the D channel employs a link-layer protocol known as LAPD (Link Access Protocol-D Channel).

LAPD Services

The LAPD standard provides two forms of service to LAPD users: the unacknowledged information-transfer service and the acknowledged information-transfer service. The *unacknowledged information-transfer service* simply provides for the transfer of frames containing user data with no acknowledgment. The *acknowledged information-transfer* is the more common service, and is similar to that offered by LAP-B and

HDLC. With this service, a logical connection is established between two LAPD users prior to the exchange of data.

LAPD Protocol

The LAPD protocol is based on HDLC. Both user information and protocol control information and parameters are transmitted in frames. Corresponding to the two types of service offered by LAPD, there are two types of operation:

Unacknowledged operation. Layer-3 information is transferred in unnumbered frames. Error detection is used to discard damaged frames, but there is no error control or flow control.

Acknowledged operation. Layer-3 information is transferred in frames that include acknowledged sequence numbers. Error control and flow control procedures are included in the protocol. This type is also referred to in the standard as multiple-frame operation.

Physical Layer

The ISDN physical layer is presented to the user at either reference point S or T. The electrical specification depends on the specific interface. For the basic access interface, pseudoternary coding is used At the relatively modest data rate of the basic-access interface, this is a suitable code. For the higher-speed, primary-access interface, a more efficient coding scheme is needed. For the 1.544-Mbps data rate, the B8ZS code is used, while for the 2.048-Mbps data rate, the HDB3 code is used. There is no particular advantage of one over the other; the specification reflects historical usage. The functional specification for the physical layer includes the following functions:

- Full-duplex transmission of B-channel data
- Full-duplex transmission of D-channel data
- Full-duplex transmission of timing signals
- Activation and deactivation of physical circuit
- Power feeding from the network termination to the terminal
- Terminal identification
- Faulty-terminal isolation
- D-channel-contention access

The final function is required when multiple TE1 terminals share a single physical interface (i.e., a multipoint line). In that case, no additional functionality is needed to control access in the B channels, as each channel is dedicated to a particular circuit at any given time. However, the D channel is available for use by all the devices for both control signaling and for packet transmission. For incoming data, the LAPD addressing scheme is sufficient to sort out the proper destination for each data unit. For outgoing data, some sort of contention resolution protocol is needed to assure that only one device at a time attempts to transmit.

8.10 ATM

The OSI Model was developed for protocols used over unreliable facilities. Copper cable at fairly low speeds was the only option available, and data transmission was very unreliable. ATM does not suffer from these problems because today the transmission medium is highly reliable. Error detection/correction is not necessarily within the network nodes to transmit

data quickly with very little processing. The endpoints are left with the chore of checking the user data for errors and managing retransmissions.

ATM is divided into layers as shown in figure 8.7. The physical layer is divided into two parts. The ATM physical medium sublayer is responsible for transmission of data over the physical medium, regardless of the type of medium used. ATM was originally designed to operate over fiber optics but because of the slow deployment of fiber, was later modified to operate over copper and coaxial facilities as well.

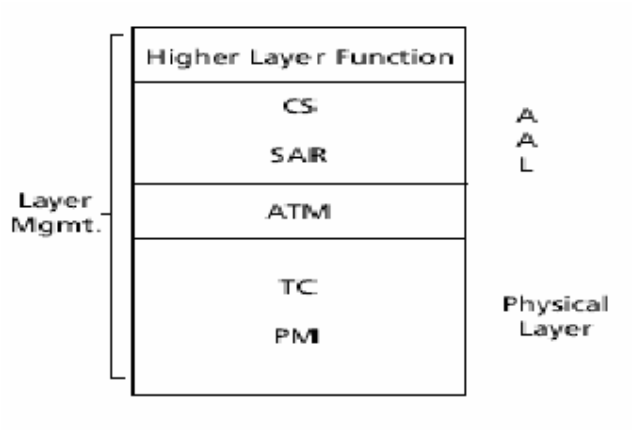


Figure 8.7: ATM Layers

The physical medium sublayer is responsible for receiving and transmitting bit streams in a continuous method. This is important to channelized services with rely on constant bit streams to maintain synchronization. When the bit stream stops, channelized equipment interprets the condition as an error and releases the virtual connection. bit synchronization is also maintained by this sublayer.

The transmission convergence sublayer is responsible for the transmission and reception of frames over framed transmission facility, such as T-3. ATM

cells are packed into these frames and unpacked at the remote end. This sublayer also performs error detection/correction but only on the ATM header. This prevents the cells from being sent to the wrong destination. Cell rate decoupling is used when a continuous data stream is required at the physical layer, as in SONET and channelized facilities such as DS1. Cell rate decoupling sends special "idle" cells over the framed facility and discards any idle cells it receives. Idle cells are necessary to maintain a connection in channelized facilities because the channel bank equipment must always see a constant bit rate transmission, or it disconnects the channel. When nothing is being sent over a channelized facility, idle flags are transmitted (this is also used to maintain clock synchronization between two endpoints). Idle cells are not recognized by the ATM layer.

The functions of the transmission Convergence Sublayer (CS) differ depending on the medium being used. For instance, if SONET is the medium, the physical layer requires a different set of functions than a DS-3 medium would require. This sublayer provides whatever services are needed by each type of medium. There are some specific functions required for DS3 and 100-Mbps interfaces. The ATM physical layer provides a convergence protocol (Physical Layer Convergence Protocol, PLCP), which maps ATM cells onto a DS3. The interface supports 44.736 Mbps. ATM cells are mapped into a DS3 PLCP data unit, which is then mapped into the DS3 payload. The DS3 PLCP is not aligned to the DS3 framing bits.

The 100-Mbps access was intended for private UNIs. Private UNIs are not as complex as public UNIs, which must provide higher reliability and complex monitoring. The specification is based on the FDDI physical layer. There are two 155-Mbps interfaces were defined. One is for the public UNI, while the other is for the private UNI. The difference lies in the distances supported by

each interface. The 155-Mbps private UNI interface can be used over fiber optics or twisted-pair copper. The public UNI requires fiber optics using single-mode fiber.

The ATM layer is responsible for multiplexing cells over the interface. ATM must read the VPI/VCI of incoming cells, determine which link cells are to be transmitted over, and place new VPI/VCI values into the header. At endpoints, the ATM layer generates and interprets cell headers (endpoints do not route cells). The ATM layer supports the following connection types:

- Point-to-point Virtual Channel Connection (VCC)
- Point-to-multipoint VCC
- Point-to-point Virtual Path Connection (VPC)
- Point-to-multipoint VPC

A VCC is a single connection between two endpoints. A VPC is a bundle (or group) of VCCs carried transparently between two end points. The AAL is used mostly by endpoints. It is divided into two sublayers: SAR and the Convergence Sublayer (CS). The SAR reconstructs data that has been segmented into different cells (reassembly). It is also responsible for segmenting data that cannot fit within a 48-byte payload of an ATM cell (segmentation). The CS determines the class of service to be used for a transmission. This will depend on the bit rate (constant or variable bit rate), the type of data, and the type of service to be provided (connection-oriented or connectionless). The quality of service parameters necessary for the transmission are determined by the class of service assigned

The ATM cell is 53 bytes in length. The first 5 bytes are used for the header. The payload portion of the cell is 48 bytes. Keep in mind as we discuss the protocol that the payload is also used for the AAL overhead and any other overhead from upper layers. There are two formats for the ATM header as shown in figure 8.8. One header is used for the UNI, while the other is used for the NNI. The difference lies in the Generic Flow Control (GFC) parameter found in the UNI header. The GFC is not supported in the public network, nor was it intended to be.

The header is placed in front of the payload (it arrives first). There is no trailer used in ATM. Figure shows the two-header formats. In the UNI header, the GFC can be used to throttle traffic from a specific destination. The GFC values are not currently defined, but the intent is that the GFC could be used to provide flow control from the user to the network (and vice versa). This parameter is not used out in the public network and is overwritten by network nodes.

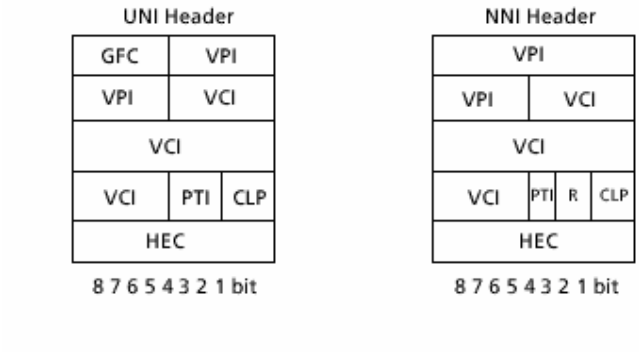


Figure 8.8: ATM Header (UNI & NNI)

Two modes are defined for GFC, controlled and uncontrolled. Controlled GFC allows subscriber equipment to control flow of ATM cells, however, the values for this have not been defined. They are of local significance only, which means they are related to a link connection at a switch and used to

communicate with the remote end of a link. Uncontrolled GFC simply means that GFC is not supported, and the parameter is set to all zeroes. The VPI is used to identify a group of virtual channels within the same endpoint. This is the form of addressing supported in ATM. Rather than identifying millions of unique nodes, ATM addressing identifies a connection. A virtual channel is used for a communication link. Each virtual channel is identified by the VCI.

The meta signaling channel is a dedicated channel used to establish virtual channel connections between two endpoints. The virtual paths are predetermined at the time of installation (either through administration or by the service provider). Virtual paths can be negotiated by the user or the network using meta signaling. This means the establishment or release of virtual paths can be controlled by the user or the network. Following the VPI and the VCI is the Payload Type Indicator (PTI). This parameter indicates the type of data found in the payload portion of the cell. Remember that the payload is not always data. There could be signaling information, network management messages, and other forms of data. These are identified by the PTI. The PTI is followed by the Cell Loss Priority (CLP) parameter. This is used to prioritize cells. In the event of congestion or some other trouble, a node can discard cells that have a CLP value of 1 (considered low priority). If the CLP value is 0, the cell has a high priority and should only be discarded if it cannot be delivered.

The last byte in the header is the Header Error Control (HEC) parameter, which is used for error checking and cell delineation. Only the header is checked for errors. The HEC works like other error checking methods, where an algorithm is run on the header and the value placed in the HEC. ATM is capable of fixing single bit errors but not multiple bit errors. An ATM node places itself in error correction mode during normal operation. If a single bit

error is detected in the header, the data in the header is small enough that the error correctio algorithm can determine which bit is in error and correct it. If a multibit error is detected, the node places itself in error detection mode. Errors are not corrected while in this mode. The node remains in error detection mode as long as cells are received in error. When cells are received without error, the node places itself back into error correction mode.

8.11 Summary

Cryptography is a tool that can be used to keep information confidential and to ensure its integrity and authenticity. Cryptographic algorithms can be divided into symmetric-key algorithms and public-key algorithms. Symmetric-key algorithms mangle the bits in a series of rounds parameterized by the key to turn the plaintext into the ciphertext. Triple DES and (AES) are the most popular symmetric-key algorithms at present. These algorithms can be used in electronic code book mode, cipher block chaining mode, stream cipher mode, counter mode, and others.

Public-key algorithms have the property that different keys are used for encryption and decryption and that the decryption key cannot be derived from the encryption key. Public-key management can be done using certificates, which are documents that bind a principal to a public key. Certificates are signed by a trusted authority or by someone approved by a trusted authority. E-mail security can be achieved by a combination of the techniques we have studied in this chapter. PGP, Web security is also an important topic, starting with secure naming. DNSsec provides a way to prevent DNS spoofing, as do self-certifying names. Most e-commerce Web sites use SSL to establish secure, authenticated sessions between the client and server.

Naming in the Internet uses a hierarchical scheme called the domain name system (DNS). DNS is implemented as a distributed database system with servers all over the world. DNS holds records with IP addresses, mail exchanges, and other information. The Web is a system for linking hypertext documents. Originally, each document was a page written in HTML with hyperlinks to other documents. A browser can display a document by establishing a TCP connection to its server, asking for the document, and then closing the connection. Multimedia allows audio and video to be digitized and transported electronically for display. Audio requires less bandwidth, so it is further along. Streaming audio, Internet radio, and voice over IP are a reality now, with new applications coming along all the time. Video on demand is an up-and-coming area in which there is great interest.

8.12 Self Assessment Questions

1. DNS uses UDP instead of TCP. If a DNS packet is lost, there is no automatic recovery. Does this cause a problem, and if so, how is it solved?
2. Can a machine with a single DNS name have multiple IP addresses? How could this occur?
3. Consider the quoted-printable MIME encoding scheme. Mention a problem not discussed in the text and propose a solution.
4. From an ISP's point of view, POP3 and IMAP differ in an important way. POP3 users generally empty their mailboxes every day. IMAP users keep their mail on the server indefinitely. Imagine that you were called in to advise an ISP on which protocol it should support. What considerations would you bring up?

5. When are external viewers needed? How does a browser know which one to use?
6. Does voice over IP have the same problems with firewalls that streaming audio does? Discuss your answer.
7. Design an attack on DES based on the knowledge that the plaintext consists exclusively of upper case ASCII letters, plus space, comma, period, semicolon, carriage return, and line feed. Nothing is known about the plaintext parity bits.
8. Compare cipher block chaining with cipher feedback mode in terms of the number of encryption operations needed to transmit a large file. Which one is more efficient and by how much?
9. In addition to being subject to loss, UDP packets have a maximum length, potentially as low as 576 bytes. What happens when a DNS name to be looked up exceeds this length? Can it be sent in two packets?
10. Can a computer have two DNS names that fall in different top-level domains? If so, give a plausible example. If not, explain why not.
11. The number of companies with a Web site has grown explosively in recent years. As a result, thousands of companies are registered in the com domain, causing a heavy load on the top-level server for this domain. Suggest a way to alleviate this problem without changing the naming scheme. It is permitted that your solution requires changes to the client code.
12. Some e-mail systems support a header field Content Return:. It specifies whether the body of a message is to be returned in the event of nondelivery. Does this field belong to the envelope or to the header?

13. Consider the quoted-printable MIME encoding scheme. Mention a problem not discussed in the text and propose a solution.
14. Name five MIME types not listed in the book. You can check your browser or the Internet for information.
15. POP3 allows users to fetch and download e-mail from a remote mailbox. Does this mean that the internal format of mailboxes has to be standardized so any POP3 program on the client side can read the mailbox on any mail server? Discuss your answer.
16. Does Webmail use POP3, IMAP, or neither? If one of these, why was that one chosen? If neither, which one is it closer to in spirit?
17. When Web pages are sent out, they are prefixed by MIME headers. Why?
18. When are external viewers needed? How does a browser know which one to use?
19. Is it possible that when a user clicks on a link with Netscape, a particular helper is started, but clicking on the same link in Internet Explorer causes a completely different helper to be started, even though the MIME type returned in both cases is identical? Explain your answer.
20. The standard http URL assumes that the Web server is listening on port 80. However, it is possible for a Web server to listen to some other port. Devise a reasonable syntax for a URL accessing a file on a nonstandard port.
21. Write a program in JavaScript that accepts an integer greater than 2 and tells whether it is a prime number. Note that JavaScript has `if` and `while` statements with the same syntax as C and Java. The modulo operator is %. If you need the square root of x, use Math.sqrt (x).

22. An audio streaming server has a one-way distance of 50 msec with a media player. It outputs at 1 Mbps. If the media player has a 1-MB buffer, what can you say about the position of the low-water mark and the high-water mark?
23. Does voice over IP have the same problems with firewalls that streaming audio does? Discuss your answer.
24. What is the bit rate for transmitting uncompressed 800 x 600 pixel color frames with 8 bits/pixel at 40 frames/sec?
25. Can a 1-bit error in an MPEG frame affect more than the frame in which the error occurs? Explain your answer.
26. Design an attack on DES based on the knowledge that the plaintext consists exclusively of upper case ASCII letters, plus space, comma, period, semicolon, carriage return, and line feed. Nothing is known about the plaintext parity bits.
27. Using the RSA public key cryptosystem, with a = 1, b = 2, etc.,
    a. If p = 7 and q = 11, list five legal values for d.
    b. If p = 13, q = 31, and d = 7, find e.
    c. Using p = 5, q = 11, and d = 27, find e and encrypt ''abcdefghij''.

8.13 References / Suggested Readings

- Computer Networks (3rd & 4th Edition), Andrew S. Tannenbaum, PHI / Pearson's Publications
- Computer Networks (2nd edition), Uyless Black, PHI Publication
- Computer Network, ED Tittle, Tata MacGraw Hills Publications