MODULE VI

APPLICATION LAYER

4.1. DOMAIN NAME SYSTEM

The Domain Name System (DNS) is the method by which Internet addresses in mnemonic form such as cse**.cusat.ac.in** are converted into the equivalent numeric IP address such as **134.220.4.1**. To the user and application process this translation is a service provided either by the local host or from a remote host via the Internet. The DNS server (or resolver) may communicate with other Internet DNS servers if it cannot translate the address itself.

The system accesses the DNS through a resolver. The resolver gets the hostname and returns the IP address or gets an IP address and looks up a hostname. The resolver returns the IP address before asking the TCP to open a connection or sending a datagram using UDP.

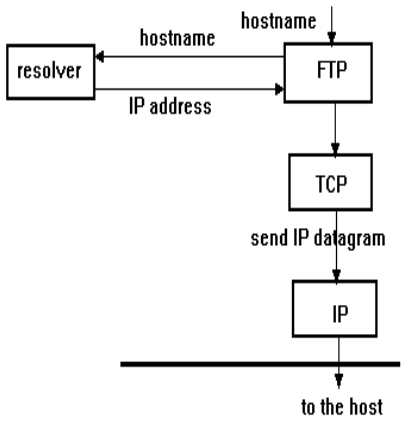


Fig1. DNS working scheme.

4.1.1. DNS Name Structure

DNS names are constructed hierarchically. The highest level of the hierarchy being the **last** component or **label** of the DNS address. Labels can be up to 63 characters long and are case insensitive. A maximum length of 255 characters is allowed. Labels must start with a letter and can only consist of letters, digits and hyphens. The root of the DNS tree is a special node with a null label. In the DNS we start from the leaf and "go up" till the root.

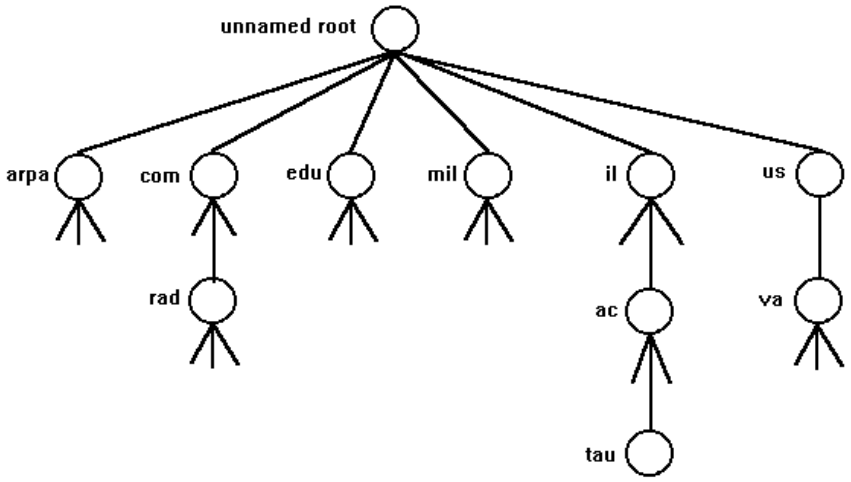


Fig.2 Hierachical organization of the DNS .

Top-level domains can be classified as

- Generic
- Country

The seven 3-character generic domain names.

code	meaning
com	Commercial. Now international.
edu	Educational.
gov	Government.
int	International Organization.
mil	Military.
net	Network related.
org	Miscellaneous Organization.

The Internet scheme can accommodate a wide variety of organizations, and allows each group to choose between geographical or organizational naming hierarchies. Most sites follow the Internet scheme so they can attach their TCP/IP installations to the connected Internet without changing names. The zone is a sub tree of the DNS that is administered separately. A common zone is a second-level domain, "ac.il" for example. Thus a lot of second-level domains divide their zone into smaller zones.

Whenever a new system is installed in a zone, the DNS administrator for the zone allocates a name and an IP address for the new system and enters these into the name server's database. A name server is said to have authority for one zone or multiple zones. Often, server software executes on a dedicated processor, and this computing machine is called the name Server.

The person responsible for a zone must provide a primary name server for that zone and one or more secondary name servers. The main difference between a primary and a secondary is that the primary loads all the information for the zone from disk files, while the secondary obtain all the information from the primary. When a secondary obtains the information from its primary it is called a zone transfer.

When a new host is added to the zone, the administrator adds the appropriate information (name and IP address) to a disk file on the system running the primary. The primary name server is then notified to reread its configuration files. The secondary query the primary on regular basis (normally every 3 hours) and if the primary contains newer data, the secondary obtains the new data using a zone transfer.

If the name server doesn't contain the information requested, it must contact another name server. The root servers then know the name and location (i.e. IP address) of each authoritative name servers for all the second-level domains. There are six root servers in the world and every primary name server has to know the address of one of root server.

The country domains are of three characters. The country domain for India is ***in***, United Kingdom is ***uk*** etc. Every country in the world has a two letter domain name.

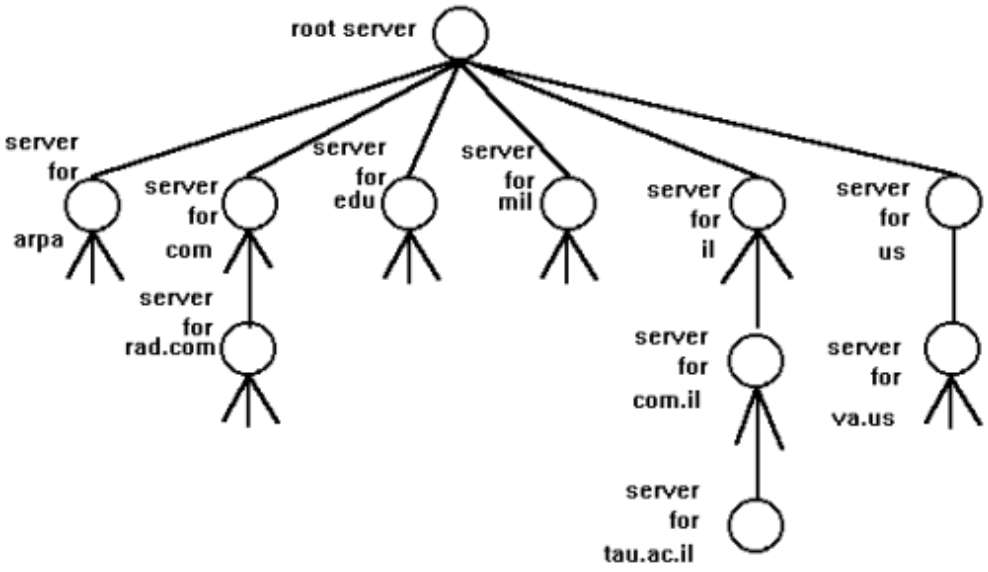


Fig. 3 The conceptual arrangement of domain name servers in the tree depicted in Fig.2

4.1.2. Resource Records

The Domain Name System provides a consistent name space that is used for referring to resources. It is a distributed hierarchical system that can be used in many applications and across multiple communications systems. It consists of three kinds of entities: name servers, name resolvers, and resource records. Name servers answer queries using data available to them. Name resolvers serve as an interface between user programs and the Domain Name System. Resource records are the data associated with the names. Resource records are used to store data about domain names and IP addresses.

Every domain has a set of resource records. Resource record is a five tuple. Format is

<domain_name> ***<time_to_live>*** ***<class>*** ***<type>*** ***<value>***

If the domain name is only a host name, the default domain is appended. If it is left empty, then the last specific name is used. If it is the @ symbol, the domain name of the DNS server is used.

Time to live is an optional entry that specifies how long the record can be cached.

Class is the class of the record, and it is IN for Internet.

Type is the type of the record.

Value is the resource record data.

Some common resource record type

Type	Description
A	Host address, maps host name to IP address
NS	Authoritative name server for this zone
CNAME	Canonical name, used to define an alias for a host name
SOA	Start of Authority, starts DNS entries in zone file, specifies name server for domain and other features like server contact and serial number
PTR	Pointer record, for performing reverse domain name lookups, maps IP address to host name
MX	Mail exchanger, informs remote site of your zone's mail server
TXT	Text strings, usually information about a host

4.2. ELECTRONIC MAIL

Email is the most widely used *Internet* application. For some people, it is their most frequent form of communication. Electronic mail is a natural use of networked communication technology that developed along with the evolution of the Internet. Message exchange in one form or another has existed from the early days of timesharing computers. Network email was developed for the *ARPANET* shortly after it's creation, and has now evolved into the powerful email technology that is the most used application on the Internet today.

Email servers exchange messages using the SMTP protocol. Client applications log into the servers to send and receive user's email with one of several protocols, including the leading POP3, IMAP, and MAPI protocols.

Each internet *domain* has an associated *email server* that manages all addresses at that domain. Each email address is expressed in the form "name@domain", and is unique at that domain, as in " jane@twenty.net".

The key to this elegant architecture is its simple structure, where each domain has an associated mail server that maintains the accounts for all users with addresses at that domain. Therefore, any email server can easily use the Internet's *domain name system* to find the *IP address* of any other mail server, connect to that server, and transfer email to recipients at that domain using the standard *SMTP* protocol.

4.3. MIME

In the early days of the ARPANET, e-mail consisted exclusively of text messages written in English and expressed in ASCII. For this environment, RFC 822 did the job completely: it specified the headers but left the content entirely up to the users. Nowadays, on the worldwide Internet, this approach is no longer adequate. The problems include sending and receiving

- Messages in languages with accents (e.g., French and German).
- Messages in non-Latin alphabets (e.g., Hebrew and Russian).
- Messages in languages without alphabets (e.g., Chinese and Japanese).
- Messages not containing text at all (e.g., audio or images).

A solution was proposed in RFC 1341 and updated in RFCs 2045–2049. This solution, called MIME (Multipurpose Internet Mail Extensions) is now widely used.

The basic idea of MIME is to continue to use the RFC 822 format, but to add structure to the message body and define encoding rules for non-ASCII messages. By not deviating from RFC 822, MIME messages can be sent using the existing mail programs and protocols. All that has to be changed are the sending and receiving programs, which users can do for themselves.

MIME is an Internet Standard for the format of e-mail. Virtually all human written Internet e-mail and a fairly large proportion of automated e-mail is transmitted via SMTP in MIME format. Internet e-mail is so closely associated with the SMTP and MIME standards that it is sometimes called SMTP/MIME e-mail.

To insure that email messages containing images or other non-text information will be delivered with maximum protection against corruption, MIME provides a way for non-text information to be encoded as text. This encoding is known as base64, and appears to be a source of frustration for many email users.

MIME defines five new message headers, as shown in figure. 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.

Header	Meaning
MIME version	Identifies the MIME version
Content description	Human readable string telling what is in the message
Content-Id	Unique identifier
Content-Transfer-Encoding:	How the body is wrapped for transmission
Content-Type	Type and format of the content

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. If the string says: "Photo of Barbara's hamster" and the person getting the message is not a big hamster fan, the message will probably be discarded rather than decoded into a high-resolution color photograph.

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. Five schemes (plus an escape to new schemes) are provided. The simplest scheme is just ASCII text. ASCII characters use 7 bits and can be carried directly by the e-mail protocol provided that no line exceeds 1000 characters.

The Content-Type: It specifies the nature of the message body.

4.4. MOBILE TELEPHONE SYSTEMS

4.4.1. Evolution of mobile telephone systems

Cellular is one of the fastest growing and most demanding telecommunications applications. Today, it represents a continuously increasing percentage of all new telephone subscriptions around the world. Currently there are more than 45 million cellular subscribers worldwide, and nearly 50 percent of those subscribers are located in the United States. It is forecasted that cellular systems using a digital technology will become the universal method of telecommunications.

The concept of cellular service is the use of low-power transmitters where frequencies can be reused within a geographic area. The idea of cell-based mobile radio service was formulated in the United States at Bell Labs in the early 1970s. However, the Nordic

countries were the first to introduce cellular services for commercial use with the introduction of the Nordic Mobile Telephone (NMT) in 1981.

Cellular systems began in the United States with the release of the advanced mobile phone service (AMPS) system in 1983. The AMPS standard was adopted by Asia, Latin America, and Oceanic countries, creating the largest potential market in the world for cellular.

In the early 1980s, most mobile telephone systems were analog rather than digital, like today's newer systems. One challenge facing analog systems was the inability to handle the growing capacity needs in a cost-efficient manner. As a result, digital technology was welcomed. The advantages of digital systems over analog systems include ease of signaling, lower levels of interference, integration of transmission and switching, and increased ability to meet capacity demands.

4.4.2. Global System for Mobile Communication (GSM)

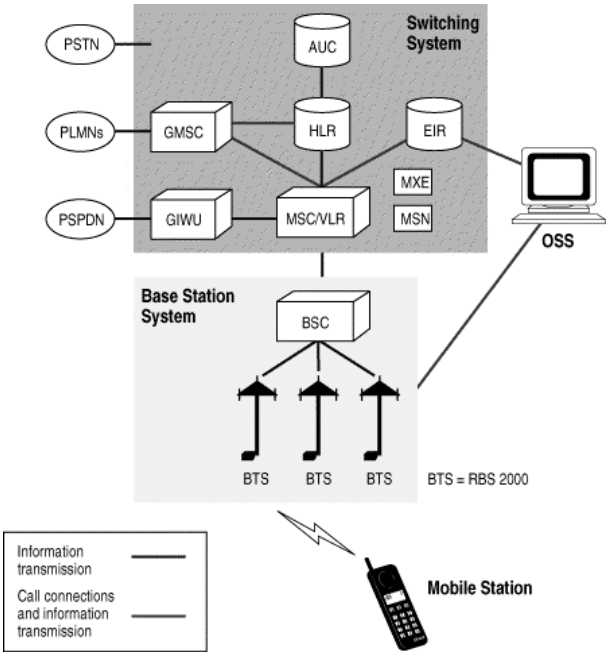
Throughout the evolution of cellular telecommunications, various systems have been developed without the benefit of standardized specifications. This presented many problems directly related to compatibility, especially with the development of digital radio technology. The GSM standard is intended to address these problems.

From 1982 to 1985 discussions were held to decide between building an analog or digital system. After multiple field tests, a digital system was adopted for GSM. The next task was to decide between a narrow or broadband solution. In May 1987, the narrowband time division multiple access (TDMA) solution was chosen.

The GSM Network

GSM provides recommendations, not requirements. The GSM specifications define the functions and interface requirements in detail but do not address the hardware. The reason for this is to limit the designers as little as possible but still to make it possible for the operators to buy equipment from different suppliers. The GSM network is divided into three major systems: the switching system (SS), the base station system (BSS), and the operation and support system (OSS).

The basic GSM network is given below.



The Switching System

The switching system (SS) is responsible for performing call processing and subscriber-related functions. The switching system includes the following functional units.

Home Location Register (HLR)—The HLR is a database used for storage and management of subscriptions. The HLR is considered the most important database, as it stores permanent data about subscribers, including a subscriber's service profile, location information, and activity status. When an individual buys a subscription from one of the PCS operators, he or she is registered in the HLR of that operator.

Mobile Services Switching Center (MSC)—The MSC performs the telephony switching functions of the system. It controls calls to and from other telephone and data systems. It also performs such functions as toll ticketing, network interfacing, common channel signaling, and others.

Visitor Location Register (VLR)—The VLR is a database that contains temporary information about subscribers that is needed by the MSC in order to service visiting subscribers. The VLR is always integrated with the MSC. When a mobile station roams into a new MSC area, the VLR connected to that MSC will request data about the mobile station from the HLR. Later, if the mobile station makes a call, the VLR will have the information needed for call setup without having to interrogate the HLR each time.

Authentication Center (AUC)—A unit called the AUC provides authentication and encryption parameters that verify the user's identity and ensure the confidentiality of each

call. The AUC protects network operators from different types of fraud found in today's cellular world.

Equipment Identity Register (EIR)—The EIR is a database that contains information about the identity of mobile equipment that prevents calls from stolen, unauthorized, or defective mobile stations. The AUC and EIR are implemented as stand-alone nodes or as a combined AUC/EIR node.

The Base Station System (BSS)

All radio-related functions are performed in the BSS, which consists of base station controllers (BSCs) and the base transceiver stations (BTSs).

BSC—The BSC provides all the control functions and physical links between the MSC and BTS. It is a high-capacity switch that provides functions such as handover, cell configuration data, and control of radio frequency (RF) power levels in base transceiver stations. A number of BSCs are served by an MSC.

BTS—The BTS handles the radio interface to the mobile station. The BTS is the radio equipment (transceivers and antennas) needed to service each cell in the network. A group of BTSs are controlled by a BSC.

The Operation and Support System

The operations and maintenance center (OMC) is connected to all equipment in the switching system and to the BSC. The implementation of OMC is called the operation and support system (OSS). The OSS is the functional entity from which the network operator monitors and controls the system. The purpose of OSS is to offer the customer cost-effective support for centralized, regional and local operational and maintenance activities that are required for a GSM network. An important function of OSS is to provide a network overview and support the maintenance activities of different operation and maintenance organizations.

- **Message Center (MXE)**—The MXE is a node that provides integrated voice, fax, and data messaging. Specifically, the MXE handles short message service, cell broadcast, voice mail, fax mail, e-mail, and notification.
- **Mobile Service Node (MSN)**—The MSN is the node that handles the mobile intelligent network (IN) services.

- **Gateway Mobile Services Switching Center (GMSC)**—A gateway is a node used to interconnect two networks. The gateway is often implemented in an MSC. The MSC is then referred to as the GMSC.
- **GSM Interworking Unit (GIWU)**—The GIWU consists of both hardware and software that provides an interface to various networks for data communications. Through the GIWU, users can alternate between speech and data during the same call. The GIWU hardware equipment is physically located at the MSC/VLR.

4.5. BLUETOOTH

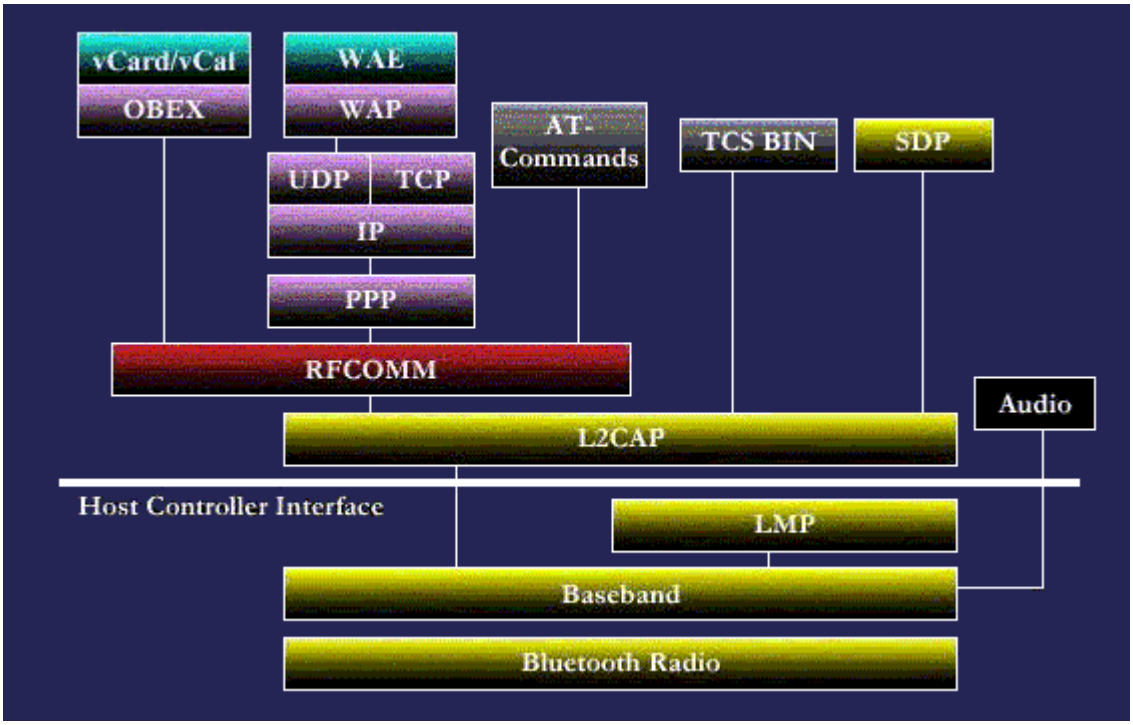
Bluetooth is a global standard for wireless connectivity. Bluetooth technology facilitates the replacement of the cables used to connect one device to another, with one universal short-range radio link operating in the unlicensed 2.45 GHz ISM band. The main objectives of Bluetooth technology can be described as follows,

- Cable replacement: Getting rid of the various types of cables and wires required for interconnectivity between various devices would enable the lay man to use all electronic devices without wasting time and money.
- Small size: the Bluetooth device is very small so that it can be attached to any device required like the cell phones without adding much to the weight of the system.
- Low cost: Bluetooth is aimed to be a low cost device approximately $5 in the near future.
- Low power: The utilization of power is very less (within 100 mW) as it is short range equipment and so it facilitates the use of small batteries for its usage.

4.5.1. Working of BLUETOOTH

Basically, Bluetooth is the term used to describe the protocol of a short range (10 meter) frequency-hopping radio link between devices. These devices implementing the Bluetooth technology are termed Bluetooth - enabled. Bluetooth can imitate a universal bridge to attach the existing data networks, and also as a mechanism for forming ad-hoc networks. Designed to operate in noisy frequency environments, the Bluetooth radio uses a fast acknowledgement and frequency hopping scheme to make the link robust.

4.5.2. Bluetooth Architecture



Different layers are:

Bluetooth Radio: The Bluetooth Radio (layer) is the lowest defined layer of the Bluetooth specification. It defines the requirements of the Bluetooth transceiver device operating in the 2.4GHz ISM band. The Bluetooth air interface is based on three power classes,

- Power Class 1: designed for long range (~100m), max output power of 20 dBm.
- Power Class 2: ordinary range devices (~10m), max output power of 4 dBm.
- Power Class 3 short range devices (~10cm), with a max output power of 0 dBm.

Baseband: The Base band is the physical layer of the Bluetooth. It manages physical channels and links apart from other services like error correction, data whitening, hop selection and Bluetooth security. As mentioned previously, the basic radio is a hybrid spread spectrum radio. Typically, the radio operates in a frequency-hopping manner in which the 2.4GHz ISM band is broken into 79 1MHz channels that the radio randomly hops through while transmitting and receiving data. A Pico net is formed when one Bluetooth radio connects to another Bluetooth radio.

LMP: The Link Manager Protocol is used by the Link Managers (on either side) for link set-up and control.

HCI: The Host Controller Interface provides a command interface to the Baseband Link Controller and Link Manager, and access to hardware status and control registers.

L2CAP: Logical Link Control and Adaptation Protocol supports higher level protocol multiplexing, packet segmentation and reassembly, and the conveying of quality of service information.

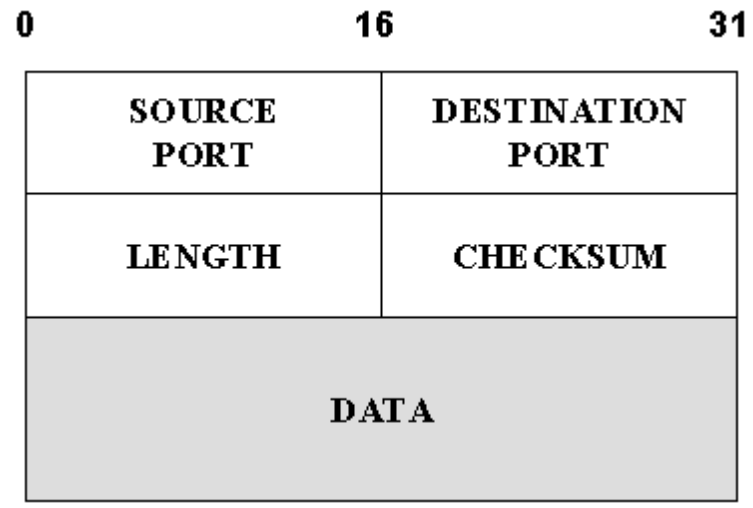
RFCOMM: The RFCOMM protocol provides emulation of serial ports over the L2CAPprotocol. The protocol is based on the ETSI standard TS 07.10.

SDP: The Service Discovery Protocol provides a means for applications to discover which services are provided by or available through a Bluetooth device. It also allows applications to determine the characteristics of those available services.

MODULE 5

UDP - SIMPLE DEMULTIPLEXER (UDP)

The simplest transport protocol is one that extends the host-to-host delivery service of the underlying network into a process-to-process communication service. There are likely to be many processes running on any given host, so the protocol needs to add a level of demultiplexing, thereby allowing multiple application process on each host to share the network. Aside from this requirement, the transport protocol adds no other functionality to the best-effort service provided by the underlying network. The Internet‟s Users Datagram protocol (UDP) is an example of such a transport protocol. The only interesting issue in such a protocol is the form of the address used to identify the target process. Although it is possible for a process to directly identify each other with an OS-assigned process ID (pid), such an approach is only practical in a closed distributed system. in which a single OS runs on all hosts and assigns each process a unique ID.A more common approach, and the one used by UDP, is for process to indirectly identify each other using an abstract locater, often called a port or mail box. The basic idea is for a source process to send a message to a port and for the destination process to receive the message from a port.

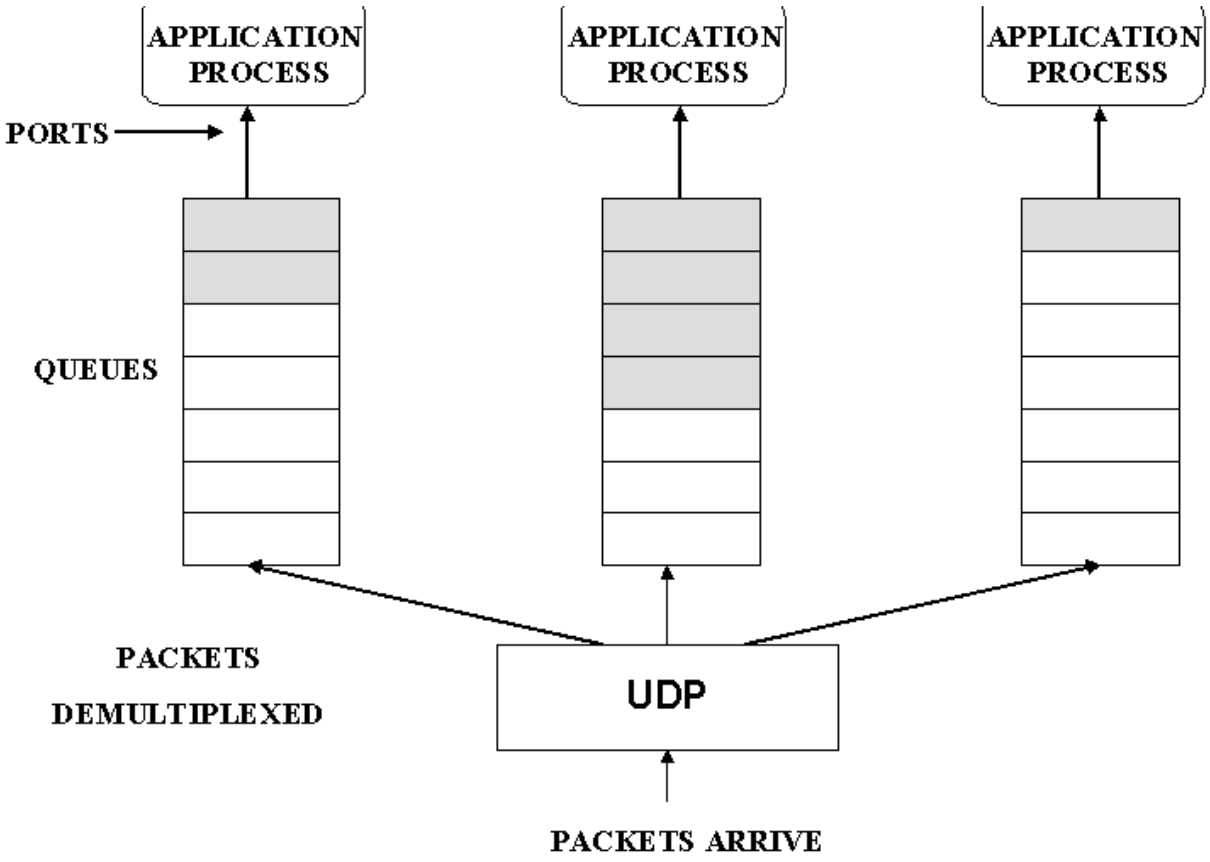


The header for an end-to-end protocol that implements this demultiplexing function typically contains an identifier (port) for both the sender (source)and the receiver(destination)of the message. The UDP port field is only 16bits long. This means that there are up to 64K possible ports, clearly not enough to identify all the processes on all the hosts in the Internet. Fortunately, ports are not interpreted across the entire Internet, but only on a single host. That is, a process is really identified by a port on some particular host-<port, host >pair. In fact, this pair constitutes the demultiplexing key for the UDP protocol.

The next issue is how a process learns the port for the process to which it wants to send the message. Typically a client process initiates a message exchange with a server process. Once a client has contacted a server, the server knows the client‟s port (it was contained in the message header) and can reply to it. The real problem, therefore, is how the client learns the server‟s port in the first place. A common approach is for the server to accept messages at a well known port. That is, each server receives its messages at some fixed port that is widely published, much like the emergency telephone

service available at the well-known number 911.In the Internet, for example, the domain name server (DNS) receives messages at well-known port 53 on each host, the mail service listens of messages at port 25,and the Unix talk program accepts messages at well known port 517,and so on. This mapping is published periodically in an RFC and is available on the most Unix systems in file /etc/services. Sometimes a well-known port to agree on some other port that they will use for subsequent communication leaving the well-known port free for other clients.

An alternative strategy is to generalize this idea, so that there is only a single well-known port- the one at which the "port mapper" service accepts messages. A client would send a message to the port mapper‟s well known port asking for the port it should use to talk to the "whatever" service and the port associated with different services over time, and for each host to use a different port for the same service.



As just mentioned, a port is purely an abstraction. Exactly how it is implemented differs from system to system, or more precisely, from OS to OS. For example, the socket API is an example implementation of ports. Typically, a port is implemented by a message queue. When a message arrives, the protocol l(eg.UDP) appends the message to the end of the queue. Should the queue be full, the message is discarded. There is no flow-control mechanism that tells the sender to slow down. When an application process wants to receive a message, one is removed from the front of the queue. If the queue is empty, the process blocks until a message becomes available.

Finally, although UDP does not implement flow control or reliable/ordered delivery, it does a little more work than to simplify demultiplexer messages to some application process- it also ensures the correctness of the message by the use of a checksum.(The UDP checksum is optional in the current

Internet, but it will become mandatory with IPv6.) UDP computes its checksum over the UDP header, the contents of the message body, and something called the pseudo header. The pseudo header consists of three fields from IP header- protocol number , source IP address and destination IP address- plus the UDP length field.(Yes, the UDP length field is included twice in the checksum calculation.) UDP uses the same checksum algorithm as IP, as defined. The motivation behind having the pseudo header is to verify that this message has been delivered between the correct two endpoints For example, if the destination IP address was modified while the packet was in transit, causing the packet to be misdelivered, or, this fact would be detected by the UDP checksum.

TCP - RELIABLE BYTE STREAM:

TCP is a more sophisticated transport protocol is one that offers a reliable, connection oriented byte stream service. Such a service has proven useful to a wide assortment of application because it frees the application from having to worry about missing or reordered data.

TCP guarantees the reliable in order delivery of a stream of bytes. It is a full duplex protocol meaning that each TCP connection supports a pair of byte streams, one flowing each direction. It also includes a flow control mechanism for each of these byte streams that allow the receiver to limit how much data the sender can transmit at a given time.

Finally, like UDP, TCP supports a demultiplexing mechanism that allows multiple application programs on any given host to simultaneously carry on a conversation with their peers. In addition to the above features, TCP also implements a highly tuned congestion control mechanism.

END TO END ISSUES:

At the heart of TCP is sliding window algorithm. TCP supports logical connections between processes that are running on any two computers in the internet. This means that TCP needs an explicit connection establishment phase during which the two sides of the connection agree to exchange data with each other. This difference is analogous to having a dedicated phone line. TCP also has an explicit connection teardown phase.

One of the things that happen during connection establishment is that the two parties establish some shared state to enable the sliding window algorithm to begin. Connection teardown is needed so each host known it is OK to free this state.

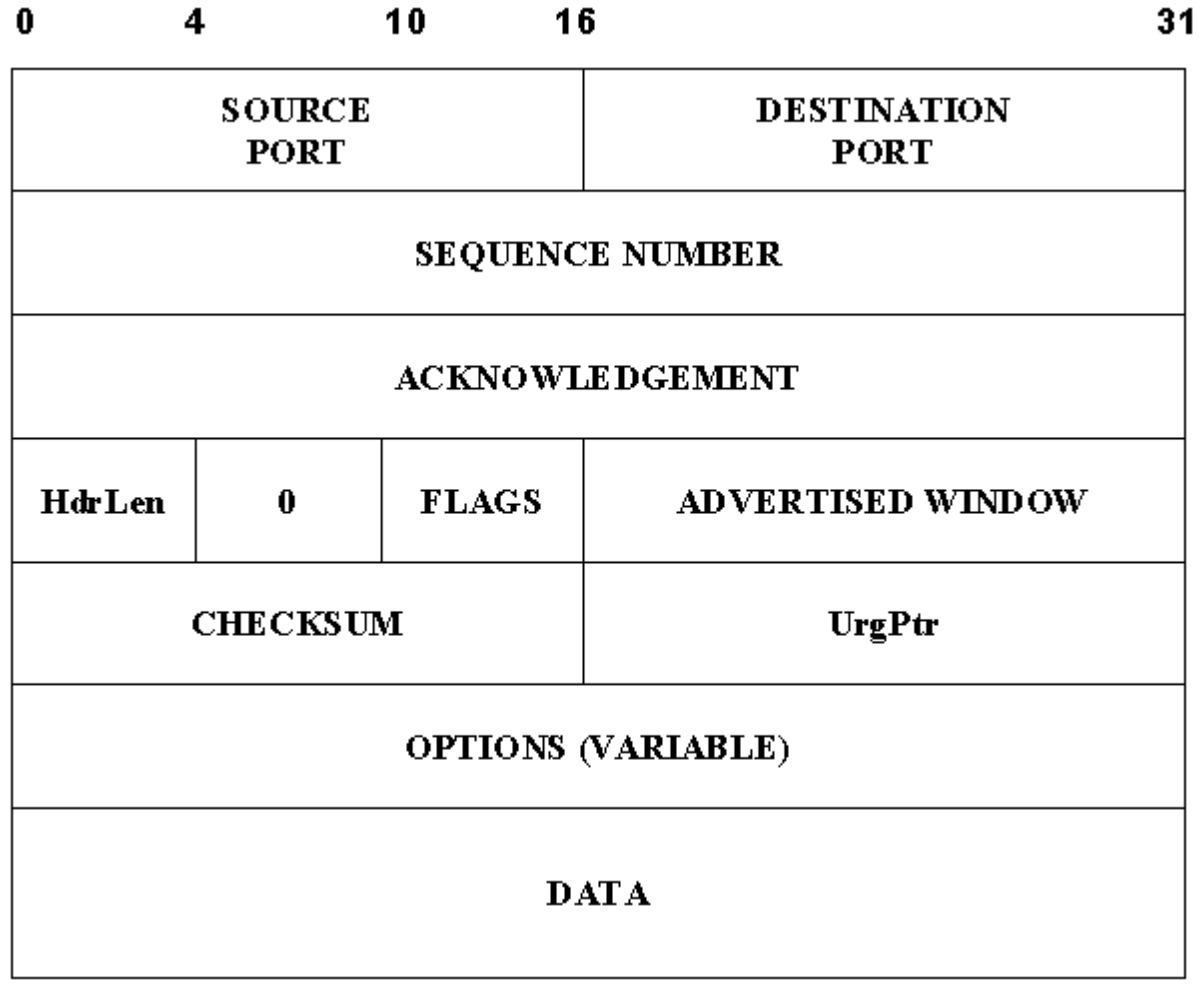
Whereas, a single physical link that always connects the same two computers has a fixed RTT, TCP connection are likely to have widely different round trip times.

Variations in the RTT are even possible during a single TCP connection. Packets may be reordered as they cross the internet, but this is not possible on a point-to-point link where the first packet put into one end of the link must be the first to appear at the other end. Packets that are slightly out of order don‟t cause a problem since the sliding window algorithm can reorder packets correctly using the sequence number.

TCP assumes that each packet has a maximum lifetime. The exact lifetime, known as the maximum segment lifetime (MSL), is an engineering choice. The current recommended setting is 120seconds. The computers connected to a point to point link are generally engineered to support the link. For example, if a link‟s delay X bandwidth product is computed to be 8KB –meaning that a window size is selected to allow up to 8kb of data to be unacknowledgement at a given time then it is likely that the computers at either end of the link have the ability to buffer up to 8kb of data.

Because the transmitting side of a directly connected link cannot send any faster than the bandwidth of the link allows, and only one host is pumping data into the link, it is not possible to unknowingly congest the link. Said another way, the load on the link is visible in the form of a queue of packets at the sender. In contrast, the sending side of a TCP connection has no idea what links will be traversed to reach the destination.

SEGMENT FORMAT



MODULE IV

TRANSPORT LAYER

4.1. FUNCTIONS OF THE TRANSPORT LAYER

Transport layer is the fourth layer in the OSI and TCP/IP models. The layer above it is the Session layer in the OSI model and the Application layer in the TCP/IP model. Protocols in the transport layer are responsible for providing a reliable and cost effective data transport from an application program running on one machine to a program running on another machine. Application layer programs interact with each other using the services provided by the transport layer without being aware of the existence of lower layers. For this the transport layer makes use of the services provided by the network layer. The hardware or software within the transport layer that is used to implement the services of the transport layer is called the transport entity.

To achieve these the transport layer may need to incorporate the following functions:

(a) Packetizing: Long messages sent by the application programs are divided into smaller units. These units form the TPDU payload. Headers are added to these to form TPDU.

(b) Addressing

(c) 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.

(d) Providing reliability

- Sequencing
- Flow Control
- Error Control

4.2. NEED FOR TRANSPORT LAYER

- Transport layer is part of the subnet and is run by the carrier. Users have no control over the subnet. Transport layer is put on top of the network layer to improve the quality of service.
- Since the network layer employs connectionless protocol, lost packets and mangled data are detected and compensated for by the transport layer.
- There are a variety of networks. So the network service primitive varies from network to network. Transport layer makes it possible for the application

programs to be written using a standard set of primitives and these programs will run on a variety of networks.

4.3. FUNCTIONS OF THE TRANSPORT LAYER

4.3.1. Service provided to the upper layer.

The Hardware /Software in the transport protocol provide the service to the upper layer is called transport entity. The transport entity can be operating system kernel on network interface card.

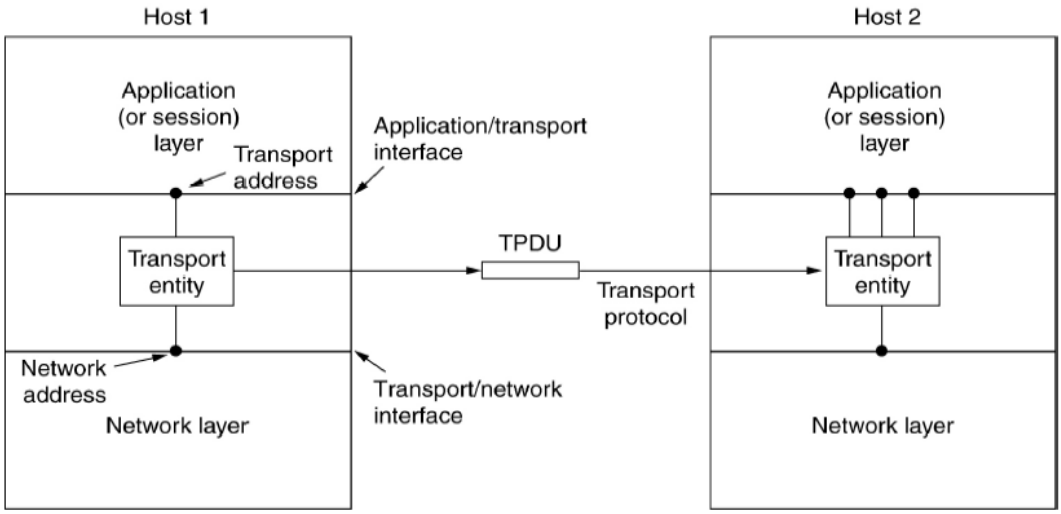


Figure 4.1. Network, transport and application layers

4.3.2. Quality of Service

The OSI transport service allows the user to specify preferred, acceptable, and unacceptable values for QOS parameters at the time a connection is set up. Some of the parameters also apply for connectionless transport. It is up to the transport layer to examine these parameters, and depending on the kind of network service or services available to it, determine whether it can provide the required service. The common transport layer QOS parameters are

- **Connection establishment delay** - The amount of time elapsing between a transport connection being requested and connection confirmation being received by the user of the transport service.
- **Connection establishment failure probability** - This is chance of connection not being established within the maximum establishment delay time.
- **Throughput** - This parameter measures the number of bytes of user data transferred per second, measured over some time interval.

- **Transit delay** - This measures the time between a message being sent by the transport user on the source machine and it being received by the transport user on the destination machine.
- **Residual error rate** - This parameter measures the number of lost messages as a fraction of total sent.
- **Protection** -This parameter provides a way for the transport user to specify interest in having the transport layer provide protection against unauthorized third parties reading or modifying the transmitted data.
- **Priority** - This parameter provides a way for transport user to indicate that some of its connection is more important than other ones.
- **Resilience** - This gives the probability of the transport layer terminating a connection due to internal problems or congestion.

4.3.3. Transport Service Primitives

Transport service primitives allow the transport users to access the transport service. Each transport service has its own access primitives. The primitives for simple transport service are given below.

Primitive	Packet sent	Meaning
LISTEN	(none)	Block until some process tries to connect
CONNECT	CONNECTION REQ.	Actively attempt to establish a connection
SEND	DATA	Send information
RECEIVE	(none)	Block until a DATA packet arrives
DISCONNECT	DISCONNECTION REQ.	This side wants to release the connection

Figure 4.2. Primitives for a simple transport service

To start with the server executes LISTEN primitive. When the client executes a CONNECT primitive, the transport entity at the client sends a CONNECTION REQUEST TPDU to the server. When the server receives this TPDU, it sends back a CONNECTION ACCEPTED TPDU to the client. The connection is now established and data is exchanged using SEND and RECEIVE primitives. When data transfer is over a DISCONNECT primitive is executed. There are two types of disconnection.

- ***Asymmetric***: Either transport user issues a DISCONNECT primitive. A DISCONNECT TPDU is sent to the remote transport entity and when this TPDU arrives the connection is released.

- ***Symmetric***: Each direction is closed separately. When one side does a DISCONNECT, it means that it has no data to send. But it can still accept data from its partner. A connection is released only when both sides have done a DISCONNECT.

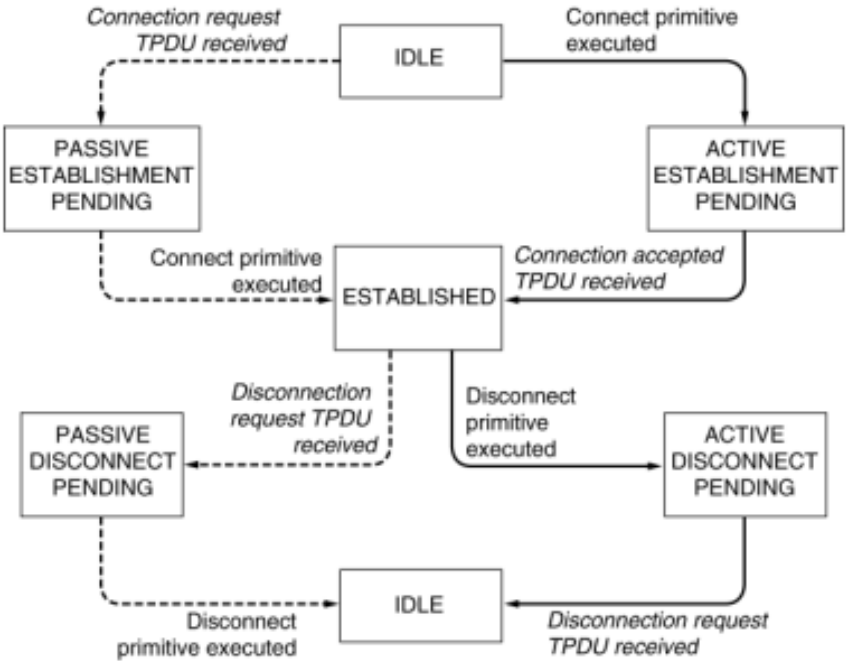


Figure 4.3. State diagram for simple connection management

The message sent by transport entity to transport entity is called Transport Protocol Data Unit (TPDU). TPDUs are contained in packets. Packets are contained in frames. When a frame arrives, the DLL process 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.

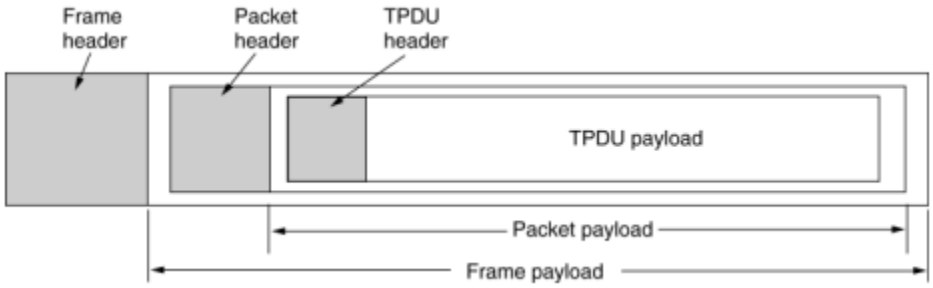


Figure 4.4. Nesting of TPDUs, packets, and frames

Berkeley Sockets

Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Attach a local address to a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Block the caller until a connection attempt arrives
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

Figure 4.5. Socket primitives used in Berkeley UNIX for TCP

Servers execute the first four primitives. The SOCKET primitive create a new end point within the transport entity. Newly created socket do not have addresses. These are assigned using BIND primitive. In this socket model LISTEN is not a blocking call. To block waiting for an incoming connection, the server executes an ACCEPT primitive.

Client side also create socket by executing SOCKET primitive. Establish a connection using CONNECT primitive. When it completes the client process is unblocked and the connection is established. Both sides can use SEND and RECEIVE to transmit and receive data. When both sides have executed CLOSE primitive, the connection is released.

4.4. ELEMENTS OF THE TANSPORT PROTOCOL

The transport service is implemented by a **transport protocol** used between two transport entities.

Transport protocols resemble the data link protocols: error control, sequencing, and flow control.

There are major dissimilarities between DLL protocol and transport layer protocol.

- In the data link layer, it is not necessary for a router to specify which router it wants to talk to. In the transport layer, explicit addressing of destination is required.
- Initial connection setup is much more complicated in transport layer.
- The potential existence of storage capacity in the subnet requires special transport protocols.

- Buffering and flow control are needed in both layers, but the presence of a large and dynamically varying number of connections in the transport layer may require a different approach than the data link layer approach (e.g., sliding window buffer management).

4.4.1. Addressing

The method normally used is to define transport addresses to which processes can listen for connection requests. In Internet, these end points are **(IP address, local port)** pairs. The term **TSAP (Transport Service Access Point)** is used to refer to these end points.

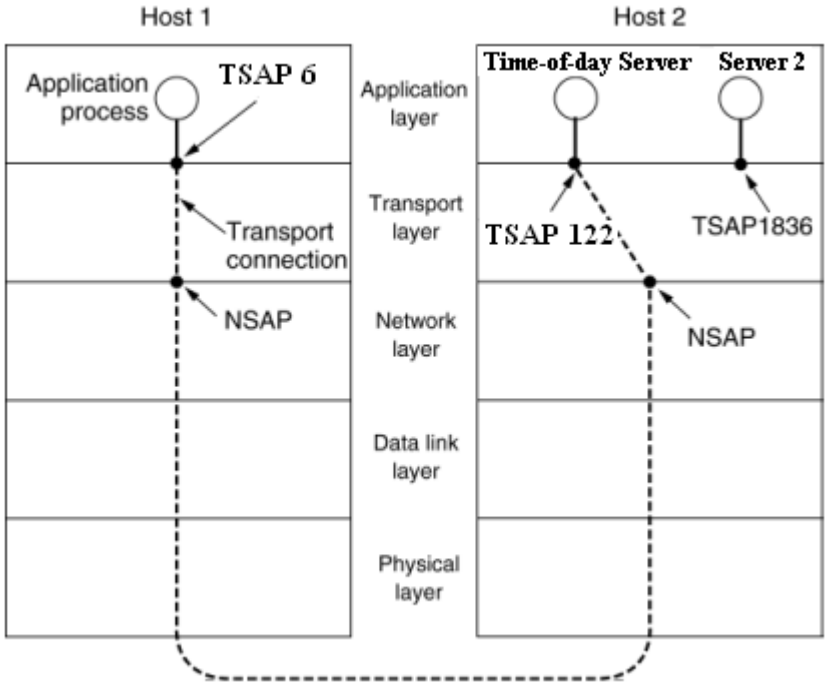
A possible connection scenario:

- A time-of-day server process on Host 2 attaches itself to TSAP 122 to wait for an incoming call.
- A process on Host 1 wants to find out the time-of-day, so it issues a CONNECT request specifying TASP 6 as the source and TSAP 122 as the destination.
- The transport entity on Host 1 selects an NSAP on its machine (if it has more than one) and on the destination machine, and sets up a network connection (e.g., virtual circuit) between them (with a connectionless subnet, establishing this network layer connection would not be done). Using this network connection, it can talk to the transport entity on Host 2.
- The first thing the transport entity on Host 1 says to its peer on Host 2 is: "I would like to establish a transport connection between my TSAP 6 and your TSAP 122. What do you say?"
- The transport entity on Host 2 then asks the time-of-day server at TSAP 122 if it is willing to accept a new connection. If it agrees, the transport connection is established.

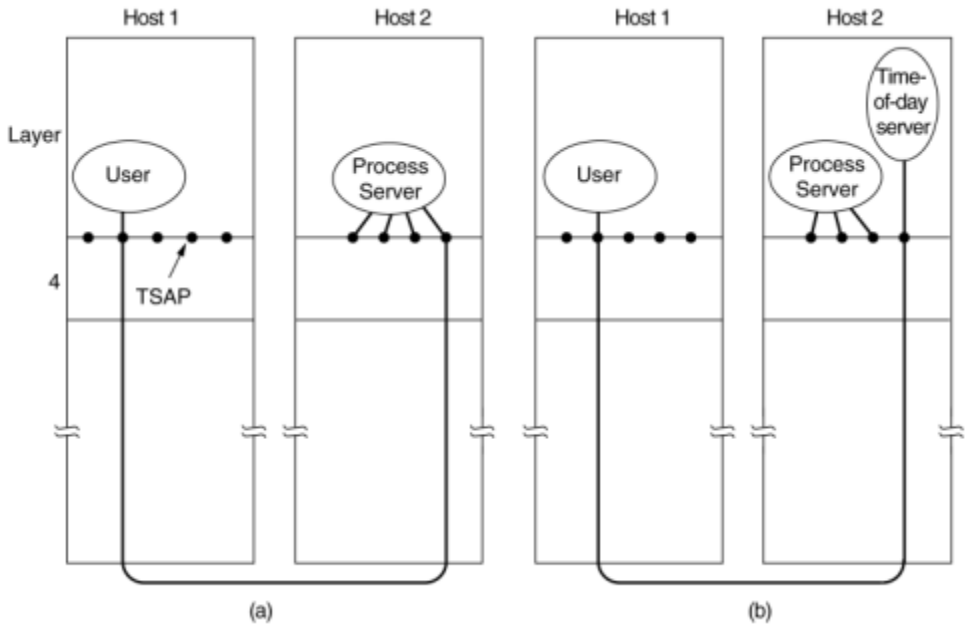
How does a process learn the port address of the process to which it wants to send a message? It is the client process that initiates a message exchange with the server process. Once the client has contacted the server, the server knows the client's port address and can reply to it. But how does the client learn the server's port?

Server accepts messages at a well-known port. Each server will receive its messages at some fixed port that is widely published. Ports ranging from 0-1024 are well known ports. These ports are assigned and controlled by IANA (Internet Assigned Number Authority).

A port mapper service accepts messages at a well-known port. Client sends a message to the port mapper specifying the service name. Port mapper returns back the appropriate TSAP.



In this model, service has stable TSAP address, which can be printed on paper and given to new users when they join the network. Instead of every server listening at a well-known TSAP, each machine that wishes to offer a service has a special server **process server** that acts as a proxy. It listen a set of ports at the same time, waiting for a TCP connection request. User begin by doing CONNECT request specifying the TSAP address. If no server is waiting for them, they get a connection to the process server. After it gets the incoming request the process server give this to the requested server. The new server does the requested work.



In this model there exists a special process called a **name server** or **directory server.** To find the TSAP address corresponding to a given service name, the user send a message specifying the service name and the name server sends back the TSAP address. Then the user releases the connection with the name server and establishes a new connection with the desired service. In this model, when a new service is created, it must register itself with the name server, giving both its service name and the address of its TSAP.

4.4.2. Establishing a Connection

Connection establishment begins when a Transport Service user **(TS-user)** issues a T.CONNECT request primitive to the transport entity. The transport entity issues a CR-TPDU (Connection Request Transport Protocol Data Unit) to its peer transport entity who informs its user with a T.CONNECT indication primitive. The user can respond with a T.CONNECT response primitive if the user is prepared to except the connection or a T.DISCONNECT request primitive .The peer transport entity then issues a CC-TPDU (Connection Confirm) or a DR-TPDU (Disconnect Request) respectively. This response is conveyed to the initiating user by means of a T.CONNECT confirm primitive (for success) or T.DISCONNECT indication primitive (for rejection) contain the reason for the failure.

The CR-TPDU and the CC-TPDUs both contain information relating to the connection being established.

The purpose of CC is to establish a connection with agreed-upon characteristics. These characteristics are defined in the CC-TPDU.

A transport connection has three types of Identifiers:

- **TSAP** - Transport Service Access Point
- **NSAP** - Network Service Access Point
- **Transport Connection I.D.**

As there may be more than one user of the transport entity, a TSAP allows the transport entity to multiplex data between users. This identifier must be passed down from the transport user, and included in CC- and CR-TPDUs. The NSAP identifies the system on which the transport entity is located. Each transport connection is given a unique identifier used in all TPDUs. It allows the transport entity to multiplex multiple transport connections on a single network.

The issue here is how to deal with the problem of delayed duplicates and establish connections in a reliable way. The methods suggested are

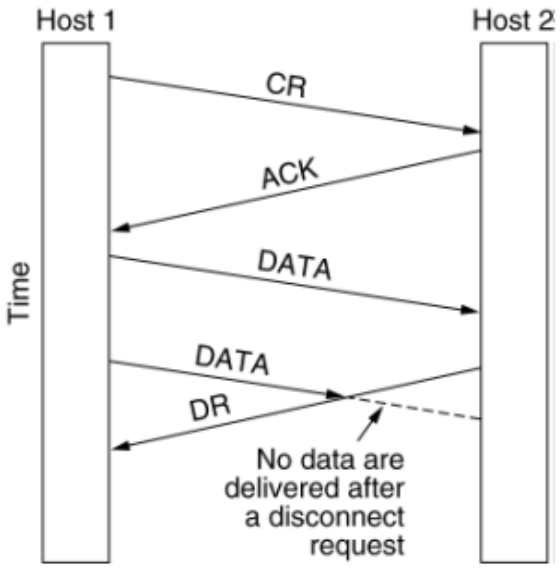
a. Use throwaway TSAP addresses - Each time a TSAP address is needed, a new, unique address is generated, typically based on the current time. When a connection is released, the addresses are discarded forever.
b. Each connection is assigned a connection identifier (i.e., a sequence number incremented for each connection established), chosen by the initiating party, and put in each TPDU, including the one requesting the connection. After each connection is released, each transport entity could update a table listing obsolete connections as *(peer transport entity, connection identifier)* pair. Whenever a connection request comes in, it could be checked against the table, to see if it belonged to a previously released connection.
c. Kill off aged packets - here we ensure that no packet lives longer than some known time. So we need is a mechanism to kill off very old packets that are still wandering about. Methods are
    - Restricted subnet design - prevents packets from looping
    - Putting a hop counter in each packet.
    - Time stamping each packet
d. When a host crashes and comes up again, the transport entity will remain idle for T seconds (where T is the life time of a packet) so that all old TPDUs will die off.

4.4.3. Releasing a Connection

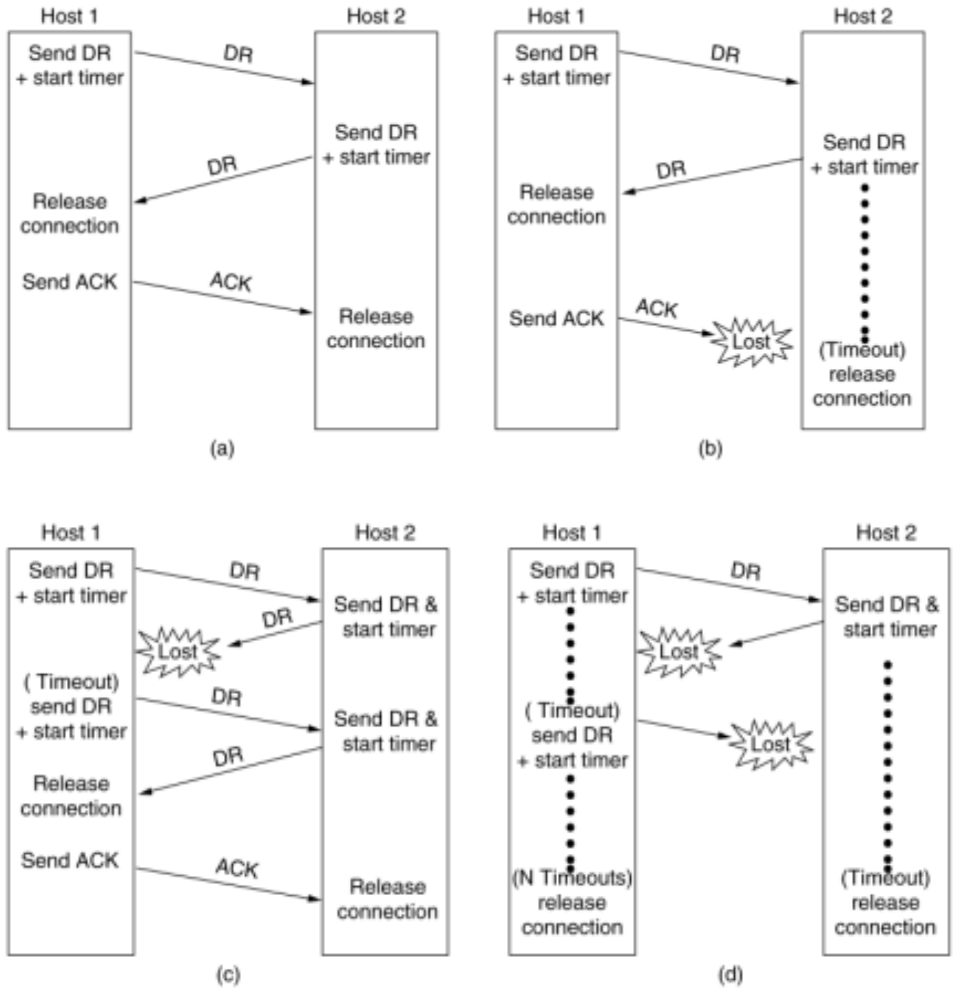
Releasing a connection is easier than establishing one. There are two types of connection termination.

- Asymmetric - Either party issues a DISCONNECT, which results in a DISCONNECT TPDU being sent and the transmission ends in both directions.
- Symmetric - Both parties issue DISCONNECT, closing only one direction at a time.

A user requests connection release by sending a T.DISCONNECT request primitive to the transport entity with the reason for closing the connection. The transport entity then sends a Disconnect Request (DR) -TPDU to its peer transport entity. The transport entity then ignores all subsequent TPDUs until receipt of a Disconnect Confirm (DC) -TPDU. The peer transport entity on receipt of a DR-TPDU returns a DC-TPDU and issues a T.DISCONNECT to the transport user. The transport connection is now closed.



Four protocol scenario for releasing a connection.



(a) Normal case of a three-way handshake. (b) Final ACK lost.

(c) Response lost. (d) Response lost and subsequent DRs lost.

4.4.4. Flow Control and Buffering

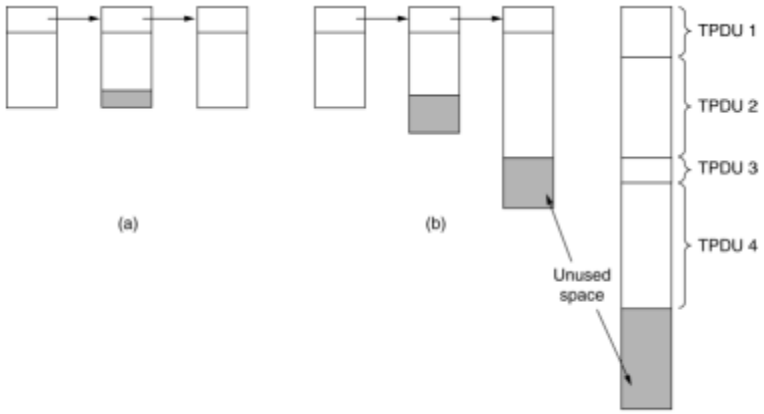
For flow control, a sliding window is needed on each connection to keep a fast transmitter from overrunning a slow receiver (the same as the data link layer).

Since a host may have numerous connections, it is impractical to implement the same data link buffering strategy (using dedicated buffers for each line).

The sender should always buffer outgoing TPDUs until they are acknowledged.

The receiver may not dedicate specific buffers to specific connections. Instead, a single buffer pool may be maintained for all connections. When a TPDU comes in, if there is a free buffer available, the TPDU is accepted, otherwise it is discarded.

However, for high-bandwidth traffic (e.g., file transfers), it is better if the receiver dedicate a full window of buffers, to allow the data to flow at maximum speed.

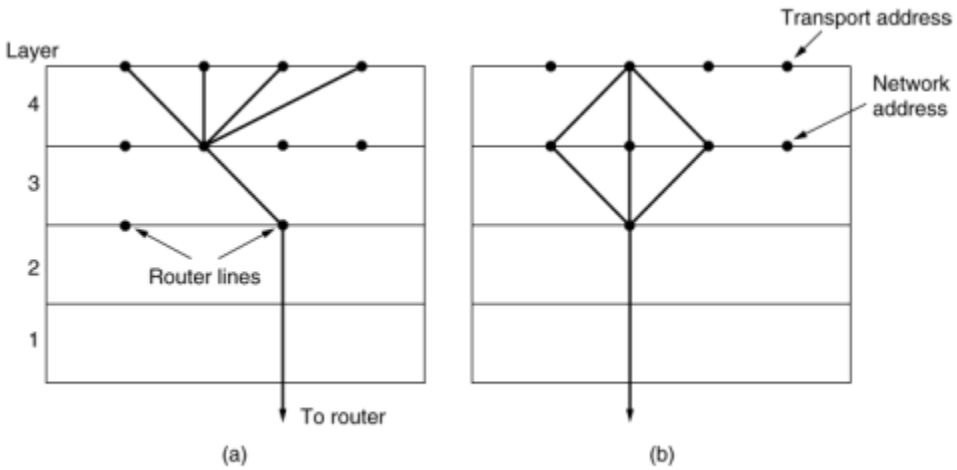


chained fixed size buffers

4.4.5. Multiplexing

The reasons for multiplexing:

- To share the price of a virtual circuit connection: mapping multiple transport connections to a single network connection (**upward multiplexing**).
- To provide a high bandwidth: mapping a single transport connection to multiple network connections (**downward multiplexing**).



Upward Multiplexing **Downward Multiplexing**

4.4.6. Crash Recovery

In case of a **router crash**, the two transport entities must exchange information after the crash to determine which TPDUs were received and which were not. The crash can be recovered by retransmitting the lost ones.

It is very difficulty to recover from a **host crash**.

Suppose that the sender is sending a long file to the receiver using a simple stop-and-wait protocol. Part way through the transmission the receiver crashes.

When the receiver comes back up, it might send a broadcast TPDU to all other hosts, requesting the other hosts to inform it of the status of all open connections before the crash.

The sender can be in one of two states: one TPDU outstanding or no TPDUs outstanding.

4.5. TRANSPORT LAYER PROTOCOLS

The two primary protocols in this layer are

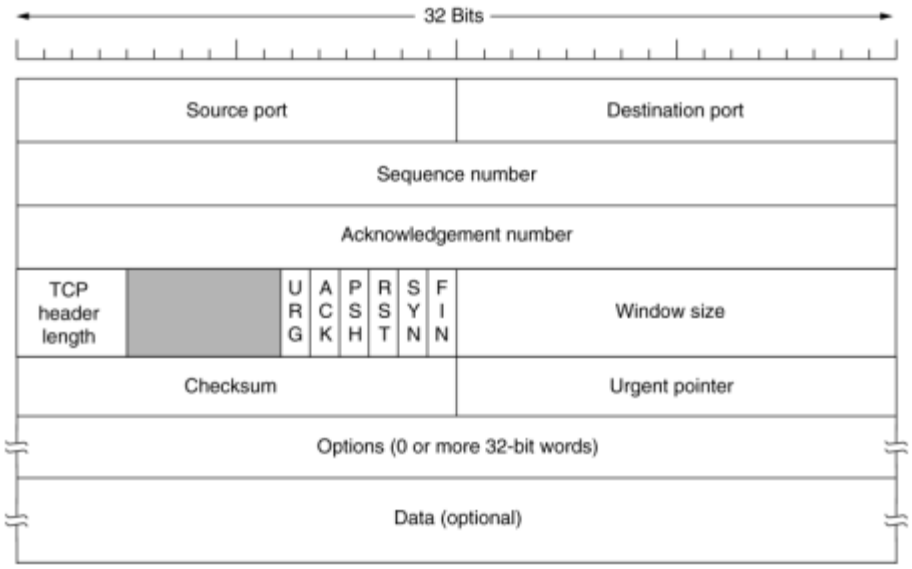
- Transmission Control Protocol (TCP)
- User Datagram Protocol (UDP)

4.5.1. Transmission Control Protocol (TCP)

4.5.1.1. TCP Header

Applications that require the transport protocol to provide reliable data delivery use TCP because it verifies that data is delivered across the network accurately and in the proper sequence. TCP is a *reliable*, *connection-oriented*, *byte-stream* protocol. TCP provides reliability with a mechanism called *Positive Acknowledgment with Re-transmission* (PAR). A system using PAR sends the data again, unless it hears from the remote system that the data arrived okay. The unit of data exchanged between cooperating TCP modules is called a ***segment.*** Each segment contains a checksum that the recipient uses to verify that the data is undamaged. If the data segment is received undamaged, the receiver sends a *positive acknowledgment* back to the sender. If the data segment is damaged, the receiver discards it. After an appropriate time-out period, the sending TCP module re-transmits any segment for which no positive acknowledgment has been received.

Every TCP segment begins with a fixed format 20-byte header followed by header options. After the options data bytes may follow.



TCP Segment Header

TCP is responsible for delivering data received from IP to the correct application. The application that the data is bound for is identified by a 16-bit number called the *port number*. The ***Source Port*** **and** ***Destination Port*** is contained in the first word of the segment header. These 16-bit *Source Port* and 16-bit *Destination Port* is used to identify the application.

Sequence number and Acknowledge number perform their usual functions.

Header length tells how many 32-bit words are contained in the TCP header.

URG is set to 1 if urgent point is in use. The urgent pointer is used to indicate a byte offset from the current sequence number at which the urgent data is to be found.

ACK bit is set to 1 to indicate that the acknowledge number is valid.

PSH indicate the pushed data. This field is used for requesting to deliver the data immediately and not to buffer.

RST is used to reset the connection.

SYN bit is used to establish the connection.

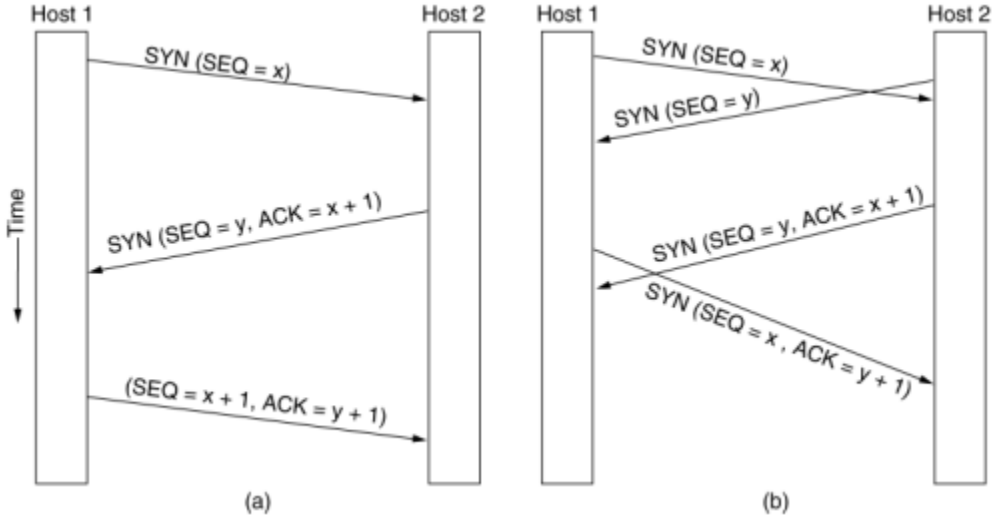
FIN is used to release the connection.

Window size tells how many bytes may be sent starting at the byte acknowledged.

4.5.1.2. TCP Connection Establishment

TCP uses a ***three-way handshaking*** procedure to set-up a connection. A connection is set up by the initiating side sending a segment with the SYN flag set and the proposed initial sequence number in the sequence number field (seq = X). The receiver then returns

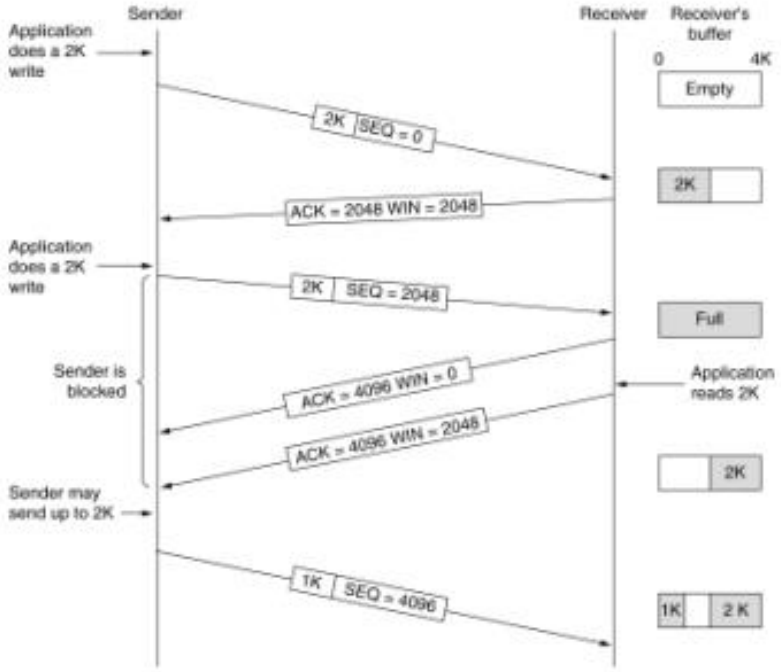
a segment with both the SYN and ACK flags set with the sequence number field set to its own assigned value for the reverse direction (seq = Y) and and acknowledge field of X + 1 (ack = X + 1). On receipt of this, the initiating side makes a note of Y and returns a segment with just the ACK flag set and an acknowledgement field of Y + 1.



4.5.1.3. TCP Connection Termination

When the user has sent all its data and wishes to close the connection it sends a CLOSE primitive to **TCP**, which sends a segment with the FIN flag set. On receipt of this, the peer TCP issues a CLOSING primitive flag to the user and returns an ACK segment. If the peer TCP user has finished send all its data it sends a CLOSE primitive. If the user still has some data to send it sends the data in a segment with the SEQ and FIN flags set. On receipt of this segment the initiating TCP issues a TERMINATE primitive to the user and returns an ACK for the data just received. When the peer TCP receives the ACK, it returns a TERMINATE primitive to the user.

4.5.1.4. TCP Window Management

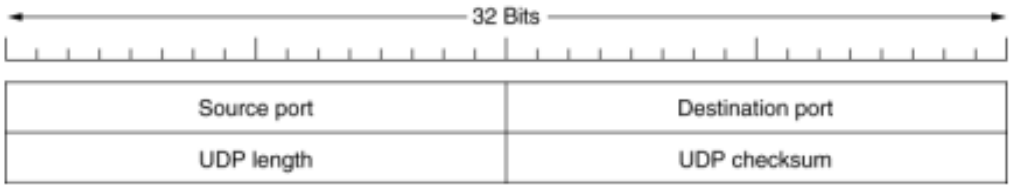


Suppose the receiver has 4096 byte buffer. IF the sender transmits a 2048 byte segment that is correctly received, the receiver will acknowledge the segment. Now it has only 2048 of buffer space. It will send a message with the new window size indicated. Now sender transmits another 2048 bytes, which are acknowledged, but the window size is 0 now. The sender must stop sending. When the receiver has removed some data from the buffer and sends the new window update to the sender.

4.5.2. User Datagram Protocol (UDP)

The User Datagram Protocol gives application programs direct access to a datagram delivery service, like the delivery service that IP provides. This allows applications to exchange messages over the network with a minimum of protocol overhead.

UDP is an unreliable, connectionless datagram protocol. “Unreliable" merely means that there are no techniques in the protocol for verifying that the data reached the other end of the network correctly. Within your computer, UDP will deliver data correctly. UDP uses 16-bit *Source Port* and *Destination Port* numbers in word 1 of the message header, to deliver data to the correct applications process. Figure shows the UDP message format



UDP Header

The amount of data being transmitted is small; the overhead of creating connections and ensuring reliable delivery may be greater than the work of re-transmitting the entire data set. In this case, UDP is the most efficient choice for a Transport Layer protocol. UDP can be used in applications that fit a *query-response* model. The response can be used as a positive acknowledgment to the query. If a response isn't received within a certain time period, the application just sends another query. Still other applications provide their own techniques for reliable data delivery, and don't require that service from the transport layer protocol. Imposing another layer of acknowledgment on any of these types of applications is inefficient.

4.6. ASYNCHRONOUS TRANSFER MODE (ATM)

Asynchronous Transfer Mode (ATM) is a connection-oriented, unreliable, virtual circuit packet switching technology. ATM technology includes:

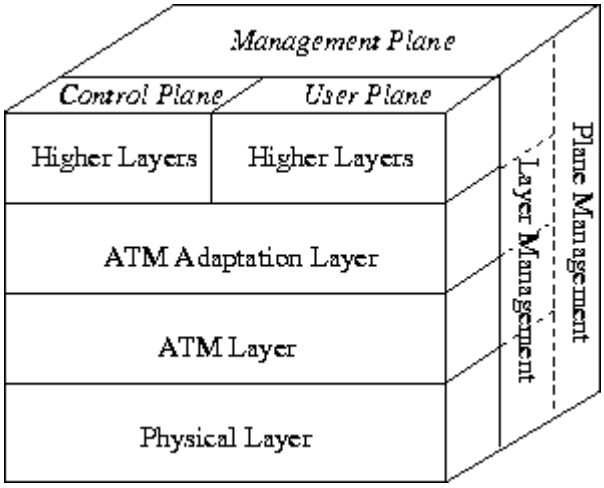
- Scalable performance — ATM can send data across a network quickly and accurately, regardless of the size of the network. ATM works well on both very low and very high-speed media.
- Flexible, guaranteed Quality of Service (QoS) — ATM allows the accuracy and speed of data transfer to be specified by the client. This feature distinguishes ATM from other high-speed LAN technologies such as gigabit Ethernet. The QoS feature of ATM also supports time dependent traffic. Traffic management at the hardware level ensures that the level of service exists end-to-end. Each virtual circuit in an ATM network is unaffected by traffic on other virtual circuits. Small packet size and a simple header structure ensure that switching is done quickly and that bottlenecks are minimized.
- Speed — ATM imposes no architectural speed limitations. Its pre-negotiated virtual circuits, fixed-length cells, message segmentation and re-assembly in hardware, and hardware-level switching all help support extremely fast forwarding of data.

- Integration of different traffic types — ATM supports integration of voice, video, and data services on a single network.

The ATM protocols are used in high-speed switching to local area networking. ATM is used by the telecommunications community as a broadband carrier for Integrated Services Digital Network (ISDN) networks and by the computer industry for high-speed LAN.

4.6.1. The Protocol Reference Model

In a similar way to the OSI 7-layer model, ATM has also developed a protocol reference model. Above the Physical Layer rests the ATM Layer and the ATM Adaptation Layer (AAL).



The AAL and Physical layers are further divided into sublayers, and the functions of these layers are shown below.

Layer/Sublayer	Function
ATM Adaptation Layer Convergence Sublayer	Convergence
Segmentation & Reassembly Sublayer	Segmentation & Reassembly
ATM Layer	Generic Flow Control Cell header generation/extraction Cell VPI/VCI translation Cell multiplex & demultiplex
Physical Layer Transmission Convergence Sublayer	Cell-rate decoupling HEC header generation/check Cell delineation
Physical Medium Sublayer	Bit timing Physical medium

Responsibilities of ATM Layer

The ATM layer is responsible for transporting information across the network. ATM uses virtual connections for information transport. The connections are deemed virtual because although the users can connect end-to-end, connection is only made when a cell needs to be sent. The connection is not dedicated to the use of one conversation.

4.6.2. ATM Cell Structure

At the core of the ATM architecture is a fixed length "cell." An ATM cell is a short, fixed length block of data that contains a short header (5 bytes) with addressing information, followed by the upper layer traffic, or "payload" (48 bytes).

Figure 1: Structure of ATM cell

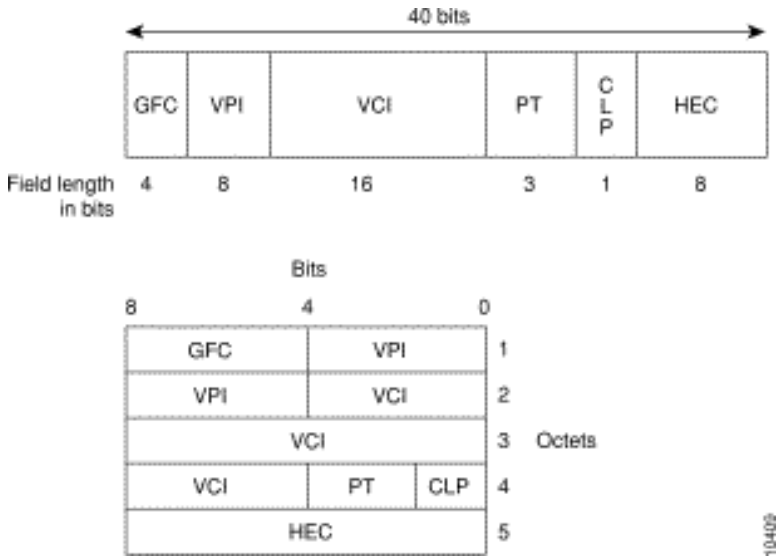
Header	**Payload**
(5 bytes)	**(48 bytes)**

Formats of the ATM Cell Header

The ATM standards groups have defined two header formats. The User-Network Interface (UNI) header format is defined by the UNI specification, and the Network-to-Network Interface (NNI) header format is defined by the NNI specification.

The UNI specification defines communications between ATM endpoints (such as workstations and routers) and switch routers in private ATM networks. The format of the UNI cell header is shown in Figure 2.

Figure 2: UNI Header Format



The UNI header consists of the following fields:

GFC—4 bits of *generic flow control* that are used to provide local functions, such as identifying multiple stations that share a single ATM interface. The GFC field is typically not used and is set to a default value.

VPI—8 bits of *virtual path identifier* that is used, in conjunction with the VCI, to identify the next destination of a cell as it passes through a series of switch routers on its way to its destination.

VCI—16 bits of *virtual channel identifier* that is used, in conjunction with the VPI, to identify the next destination of a cell as it passes through a series of switch routers on its way to its destination.

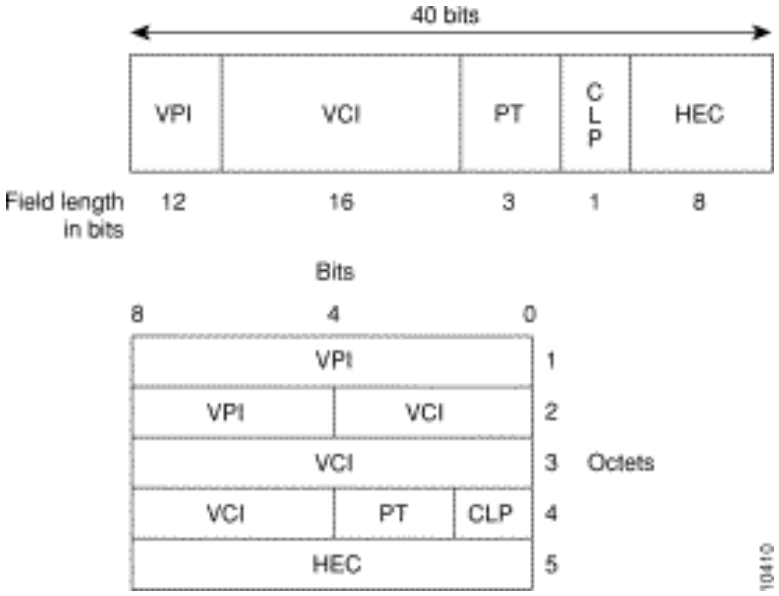
PT—3 bits of *payload type*. The first bit indicates whether the cell contains user data or control data. If the cell contains user data, the second bit indicates congestion, and the third bit indicates whether the cell is the last in a series of cells that represent a single AAL5 frame.

CLP—1 bit of *congestion loss priority* (cell loss priority) that indicates whether the cell should be discarded if it encounters extreme congestion as it moves through the network.

HEC—8 bits of *header error control* that are a checksum calculated only on the header itself.

The NNI specification defines communications between switch routers. The format of the NNI header is shown in Figure 3.

Figure 3: NNI Header Format



The GFC field is not present in the format of the NNI header. Instead, the VPI field occupies the first 12 bits, which allows switch routers to assign larger VPI values. With that exception, the format of the NNI header is identical to the format of the UNI header.

4.6.3. ATM Adaptation Layer (AAL)

The ATM adaptation layer lies between the ATM layer and the higher layers which use the ATM service. Its main purpose is to resolve any disparity between a service required by the user and services available at the ATM layer. The ATM adaptation layer maps user information into ATM cells and accounts for transmission errors. It also may transport timing information so the destination can regenerate time dependent signals.

The goal of AAL is to provide useful services to application programs and to shield them from the mechanics of chopping data up into cells at the source and reassembling them at the destination.

The AAL service space is organized along three axes:

- Real-time service versus non-real-time service.
- Constant bit rate service versus variable bit rate service.
- Connection-oriented service versus connectionless service.

Different AALs were defined in supporting different traffic or service expected to be used. The service classes and the corresponding types of AALs were as follows:

Class A - Constant Bit Rate (CBR) service: AAL1 supports a connection-oriented service in which the bit rate is constant. Examples of this service include 64 Kbps voice, fixed-rate uncompressed video and leased lines for private data networks.

Class B - Variable Bit Rate (VBR) service: AAL2 supports a connection-oriented service in which the bit rate is variable but requires a bounded delay for delivery. Examples of this service include compressed pocketsize voice or video. The requirement on bounded delay for delivery is necessary for the receiver to reconstruct the original uncompressed voice or video.

Class C - Connection-oriented data service: For connection-oriented file transfer and in general, data network applications where a connection is set up before data is transferred, this type of service has variable bit rate and does not require bounded delay for delivery. Two AAL protocols were defined to support this service class, and have been merged into a single type, called AAL3/4. But with its high complexity, the AAL5 protocol is often used to support this class of service.

Class D - Connectionless data service: Examples of this service include datagram traffic and in general, data network applications where no connection is set up before data is transferred. Either AAL3/4 or AAL5 can be used to support this class of service.

The ATM adaptation layer is divided into two sub layers:

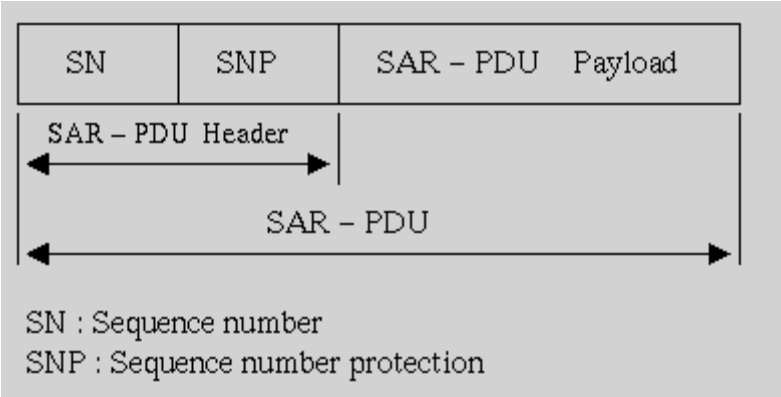
- Convergence Sub layer: This layer wraps the user-service data units in a header and trailer that contain information used to provide the services required. The information in the header and trailer depends on the class of information to be transported but will usually contain error handling and data priority preservation information.
- Segmentation and reassembly sub layer: This layer receives the convergence sublayer protocol data unit and divides it up into pieces which it can place in an ATM cell. It adds to each piece a header that contains information used to reassemble the pieces at the destination.

AAL1

AAL-1 is used to transfer constant bit rate data that is time dependent. It must therefore send timing information with the data so that the time dependency maybe recovered. AAL-1 provides error recovery and indicates error information that could not be recovered.

Convergence Sub layer: The functions provided at this layer differ depending on the service provided. It provides bit error correction and may use explicit time stamps to transfer timing information.

Segmentation and reassembly sub layer: At this layer the convergence sub layer - protocol data unit is segmented and a header added. The header contains 3 fields (see diagram below).



- Sequence Number used to detect cell insertion and cell loss.
- Sequence Number protection used to correct and errors that occur in the sequence number.
- Convergence sub layer indication used to indicate the presence of the convergence sub layer function.

AAL2

AAL-2 is used to transfer variable bit rate data that is time dependant. It sends timing information along with the data so that the timing dependency may be recovered at the destination. AAL-2 provides error recovery and indicates error information that could not be recovered. As the source generates a variable bit rate some of the cells transferred maybe useful and therefore additional features are required at the segmentation and recovery layer.

Convergence sublayer: This layer provides for error correction and transports the timing information from source to destination. Inserting time stamps or timing information into the convergence sublayer - protocol data unit, achieves this.

Segmentation and recovery sublayer: The CS-PDU is segmented at this layer and a header and trailer added to each piece. The header contains two fields.

- Sequence number is used to detect inserted or lost cells.

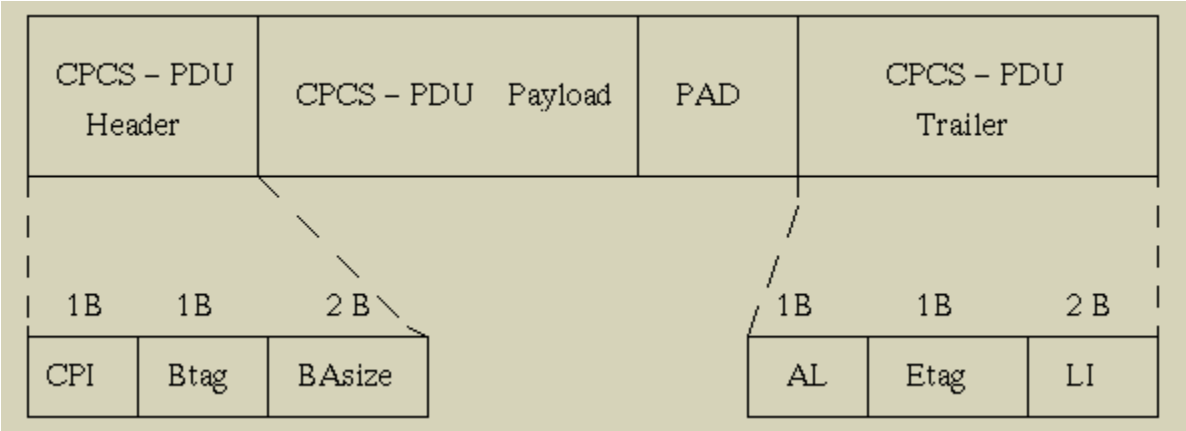
- Information type is one of the following:
    - BOM, beginning of message
    - COM, continuation of message
    - EOM, end of message

The trailer also contains two fields

- Length indicator indicated the number of true data bytes in a partially full cell.
- CRC is a cyclic redundancy check used by the segmentation and reassembly sublayer to correct errors.

AAL 3-4

AAL-3 is designed to transfer variable rate data that is time independent. It supports both message mode and streaming mode services. Message mode services are transported in a single ATM adaptation layer interface data unit, while streaming mode services require one or more AAL-IDUs. AAL-3 can be further divided into two modes of operation:
Assured operation: Corrupted or lost convergence sublayer - protocol data units are retransmitted and flow control is supported.
Non-assured operation: Error recovery is left to higher layers and flow control is optional
Convergence sublayer: The ATM adaptation layer 3 convergence sublayer is similar to the ATM adaptation layer 2 convergence sublayer as both handle non real time data. The ATM adaptation layer 3 convergence sublayer is therefore subdivided into two sections
1. The common part convergence sublayer. The AAL-2 CS also provides this. It appends a header and trailer to the common part convergence sublayer - protocol data unit payload.



The header contains 3 fields:

- Common part indicator indicates that the payload is part of the common part.

- Begin tag marks the start of the common part convergence sublayer - protocol data unit.
- Buffer allocation size tells the receiver how much buffer space is required to accommodate the message.

The trailer also contains 3 fields:

- Alignment is byte filler used to make the header and trailer the same length. i.e. 4bytes.
- End tag marks the end of the common part convergence sublayer - protocol data unit.
- The length field holds the length of the common part convergence sublayer - protocol data unit payload.

The service specific part. The functions provided at this layer depend on the services requested. They generally include functions for error detection and recovery and may also include special functions such as transparent delivery.

2. Segmentation and reassembly sublayer: At this layer the convergence sublayer - protocol data unit is segmented into pieces which can be placed in ATM cells. A header and trailer that contain information necessary for reassembly and error recovery are appended to each piece. The header contains 3 fields:

Segment type indicates what part of a message is contained in the payload. It has one of the following values:

- BOM: Begining of message
- COM: continuation of message
- EOM: End of message
- SSM: Single segment message

Sequence number used to detect cell insertion and cell loss.

Multiplexing identifier. This field is used to distinguish between data from different connection that have been multiplexed onto a single ATM connection. The trailer contains two fields:

Length indicator holds the number of useful data bytes in a partially full cell.

CYC is a cyclic redundancy check used for error detection and recovery.


ST
SN
RES / MID
SAR – SDU
LI
CRC
2
4
10
44 octets
6
10
48 octets

MODULE III
NETWORK LAYER

NETWORK LAYER DESIGN ISSUES

The network layer has been designed with the following goals:

- The services provided should be independent of the routing technology. Users of the service need not be aware of the physical implementation of the network. 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 that in the countries like the US or Ireland. The design of the layer must not disable us from connecting to networks of different technologies.
- The transport layer (that is the host computer) should be shielded from the number, type and different topologies of the routers present. That is, all that the transport layer wants is a communication link; it need not know how that link is made.
- Finally, there is a need for some uniform addressing scheme for network addresses.

Services provided to the Transport Layer

The network layer provides two different types of service,

- Connection oriented
- Connectionless.

A connection-oriented service is one in which the user is given a "reliable" end to end connection. 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 somewhat identical to the postal system. A letter sent, 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.

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, 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. A telephone call is the classic example of a connection oriented service.

Internal organization of the network layer

The subnet is organized in two different ways

- Virtual circuits: used in subnets whose primary service is connection-oriented.
- Datagrams correspond to the independent packets of the connectionless organization.

Datagrams

Each datagram must contain the full destination address. The routers have a table telling which outgoing line to use for each possible destination router. (These tables are also needed in virtual circuit subnets for determining the route for a SETUP packet). Routes are not worked out in advance. Successive packets may follow different routes.

Virtual Circuits

- Each router must maintain a table with one entry per open virtual circuit passing through it.
- Each packet traveling through the subnet must contain a virtual circuit number field in its header, in addition to sequence numbers, checksums, and the like.
- Whenever a host wants to create a new outbound virtual circuit, it chooses the lowest circuit number not currently in use (locally), and put this number in the header of a SETUP packet.
- The router does not forward this SETUP packet to the next router as is. Instead, router chooses the lowest free VC number and substitutes that number for the one chosen by the host, and then forwards the SETUP packet to router.

- Similarly, router chooses the lowest free number between it and the next router, substitutes that number for the one chosen by router, and then forwards the setup packet to the next router.
- When this SETUP packet finally arrives at the destination, the router there chooses the lowest free inbound circuit number, substitutes that number for the one in the packet, and pass it to the host.

Comparison between datagrams and virtual circuits

Issue	Datagram	Virtual Circuit
Addressing	Each packet contains the full source and destination addresses	Each packet contains a virtual circuit number
Circuit Setup	Not needed	Required
Routing	Each packet is routed independently	Route is chosen when the virtual circuit is setup, all packets follow this route
State Information	Routers do not hold information about connections	Each VC requires router table space per connection
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
Effect of router failures	None, except for packets lost during the crash	All virtual circuits that are passed through the failed router are terminated

ROUTING ALGORITHMS

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 datalink layer) of the OSI reference model, whereas routing occurs at Layer 3 (the network layer).

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.

The **routing algorithm** is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on.

Routing algorithms often have one or more of the following design goals:

- Correctness – are the packets getting to the right destination?
- Simplicity – the simpler the algorithm and steps, the easier to understand, to extend, modify and improve. Routing algorithms also are designed to be as simple as possible.
- Robustness – If something happens to the network (e.g. some lines go down, some user in a machine starts sending out a huge amount of data, etc), does the algorithm cater for that? This is good if a network is very dynamic and changes occur frequently, which means that they should perform correctly in the face of unusual or unforeseen circumstances, such as hardware failures, high load conditions, and incorrect implementations. Because routers are located at network junction points, they can cause considerable problems when they fail. The best routing algorithms are often those that have withstood the test of time and that have proven stable under a variety of network conditions.
- Stability – Does the routes change all the time? If the routes are unstable, even when there are little changes to the network, the routers will incur the overhead of having to keep changing their routing tables. This must be balanced with robustness of the algorithm.

- Fairness – How well does the algorithm ensure all routers have the same share of network utilization, and the same consideration when finding routes for them?
- Optimality – How well does the algorithm ensure that the cost (time, hops, distance, etc) is kept to a minimal? This is balanced against fairness

Optimality principle: Optimality refers to the capability of the routing algorithm to select the best route, which depends on the metrics and metric weightings used to make the calculation. If a router k is on the optimal path from router i to router j, then the optimal path from k to j is also falls along the same route.
The set of optimal routes from all sources to a given destination form a tree (called **sink tree**) rooted at the destination.

In addition, routing algorithms must converge rapidly. Convergence is the process of agreement, by all routers, on optimal routes. When a network event causes routes to either go down or become available, routers distribute routing update messages, stimulating recalculation of optimal routes and eventually causing all routers to agree on these routes. Routing algorithms that converge slowly can cause routing loops.

Routing algorithms can be divided into two major classes:

- Nonadaptive algorithms or Static Routing algorithms
- Adaptive algorithms or Dynamic Routing algorithms

Static Routing Algorithms

Nonadaptive algorithms do not base their routing decisions on measurements or estimates of the current traffic and topology. The choice of the route to use is computed in advance, off-line, and downloaded to the routers when the network is booted (Static Routing).

1. Shortest Path Routing

This is the simplest and widely used technique. The basic idea:

- Build a graph of the subnet, with each node of the graph representing a router and each arc of the graph representing a communication line.
- To choose a route between a given pair of routers, the algorithm finds the shortest path between them.

Metrics for measuring the shortest path can be any one of the following,

- The number of hops.
- The geographic distance in kilometers.
- The mean queuing and transmission delay.
- A function of the distance, bandwidth, average traffic, communication cost, mean queue length, measured delay, etc.

The **Dijkstra's algorithm** is used to find the shortest path. The following steps are performed:

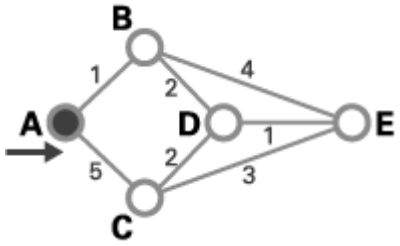
1. The router builds a status record set for every node on the network. The record contains three fields:
   - Predecessor field - This field shows the previous node.
   - Length field - This field shows the sum of the weights from the source to that node.
   - Label field - This field shows the status of node. Each node can have one status mode: "permanent" or "tentative."
2. The router initializes the parameters of the status record set (for all nodes) and sets their length to "infinity" and their label to "tentative."
3. The router sets a T-node. For example, if V1 is to be the source T-node, the router changes V1's label to "permanent." When a label changes to "permanent," it never changes again. A T-node is an agent and nothing more.

4. The router updates the status record set for all tentative nodes that are directly linked to the T-node.
5. The router looks at all of the tentative nodes in the entire network and chooses the one whose weight to V1 is lowest. That node is then the destination T-node.
6. If this node is not V2 (the intended destination), the router goes back to step 4.
7. If this node is V2, the router extracts its previous node from the status record set and does this until it arrives at V1. This list of nodes shows the best route from V1 to V2.

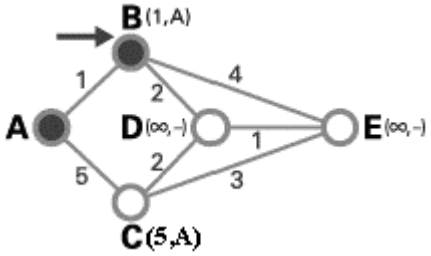
Example: Dijkstra Algorithm

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.

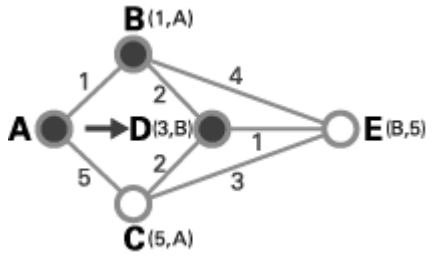
1. As you see in the image below, the source node (A) has been chosen as T-node, and so its label is permanent (we show permanent nodes with filled circles and T-nodes with the --> symbol).



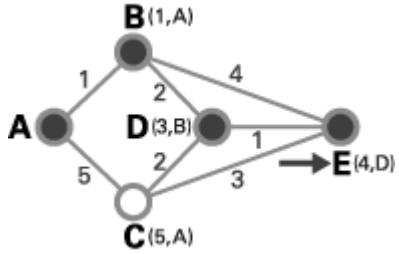
2. In this step, you see that the status record set of tentative nodes directly linked to T-node (B, C) has been changed. Also, since B has less weight, it has been chosen as T-node and its label has changed to permanent (see below).



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.

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 weight is **4** (1+2+1).

2. Flooding

Another static routing algorithm is flooding. In flooding every incoming packet is sent out on every outgoing line except the one it arrived on.

Measures for damming the flood:

- A hop counter is included in the header of each packet, which is decremented at each hop. A packet is discarded when the counter reaches zero.
- A sequence number is included in each packet. Each router maintains a list per source router telling which sequence numbers originating at that source have already been seen. A packet is discarded when it contains a sequence number that is in the list.

Selective flooding - an incoming packet is sent on those lines that are going approximately in the right direction.

Possible applications of flooding:

- In military applications, to withstand large numbers of routers crashes at any instant.
- As a metric (always choose the shortest path) against which other routing algorithms can be compared.
- In distributed database flooding used to update the databases concurrently.

Flow Based Routing

It takes into account of the load. It is suitable in some networks (e.g., a corporate network for a retail store chain), in which the mean data flow between each pair of nodes is relatively stable (approximately constant in time) and predictable.

Information known in advance:

- The subnet topology.
- The capacity for each line in the subnet.
- The average traffic (pkts/sec) between any pair of nodes.

Main calculations involved:

- Tentatively choose a routing algorithm.
- Apply the tentatively chosen routing algorithm to the known subnet topology to select a path from each node to all other nodes.
- For each line, calculate the average flow according to the selected path and the known average traffic between any pair of nodes.
- For each line, compute the mean packet delay on that line from queuing theory (assuming a mean packet delay)
- For each line, calculate the flow-weight - the fraction of the total traffic using that line.
- For the whole subnet, calculate mean packet delay according to the flow-weight and the mean packet delay for each line.

The routing problem reduces to finding the routing algorithm (from a collection of known single path routing algorithms) that produces the minimum average delay for the entire subnet. Since the calculations can be done off-line, the fact that it may be time consuming is not necessarily a serious problem.

Dynamic Routing Algorithms

Adaptive algorithms attempt to change their routing decisions to reflect changes in topology and the current traffic. (Dynamic Routing).

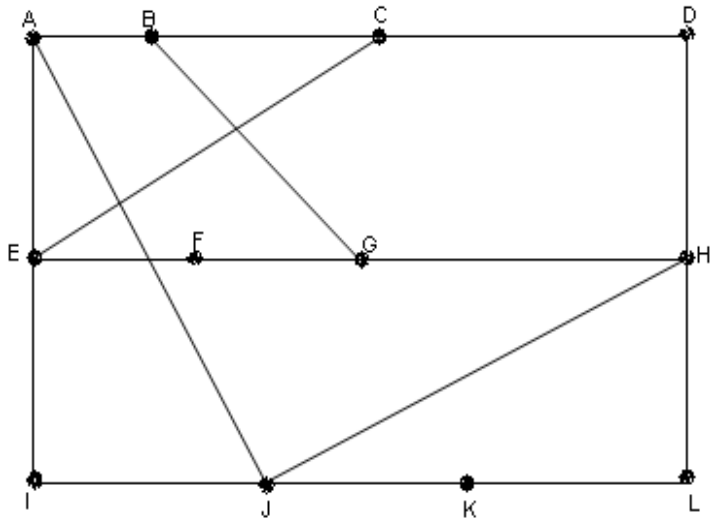
1. Distance Vector Routing

A *distance vector* routing protocol advertises the number of hops to a network destination (the *distance*) and the direction in which a packet can reach a network destination (the *vector*). The distance vector algorithm, also known as the *Bellman-Ford algorithm*, enables a router to pass route updates to its neighbors at regularly scheduled intervals. Each neighbor then adds its own distance value and forwards the routing information on to its immediate neighbors. The result of this process is a table containing the cumulative distance to each network destination.

Distance vector routing protocols, the earliest dynamic routing protocols, are an improvement over static routing, but have some limitations. When the topology of the internetwork changes, distance vector routing protocols can take several minutes to detect the change and make the appropriate corrections.

One advantage of distance vector routing protocols is simplicity. Distance vector routing protocols are easy to configure and administer. They are well suited for small networks with relatively low performance requirements. Most distance vector routing protocols use a hop count as a routing metric. A *routing metric* is a number associated with a route that a router uses to select the best of several matching routes in the IP routing table. The *hop count* is the number of routers that a packet must cross to reach a destination.

Consider the following subnet



Router J receives the following delay vectors from its neighboring routers A, I, H and K. Router J has calculated its delay to A, I, H and K as 8, 10, 12 and 6 respectively. This is done by sending an ECHO packet to its neighbors and finding the round trip time.

	A	I	H	K
A	0	24	20	21
B	12	36	31	28
C	25	18	19	36
D	40	27	8	24
E	14	7	30	22
F	23	20	19	40
G	18	31	6	31
H	17	20	0	19
I	21	0	14	22
J	9	11	7	10
K	24	22	22	0
L	29	33	9	9

From this information, router J computes its new routing table by finding the path with the shortest delay to each router in the subnet.

For example:

For router B, if J takes the path through A the delay is 8+12 = 20

If J takes the path through I the delay is 10+36 = 46

If J takes the path through H the delay is 12+31 = 43

If J takes the path through K the delay is 6+28 = 34

The path with shortest delay for J to reach B is through its neighbor A. Similarly for the other routers also J calculates the path with minimum delay and updates its routing table with the delay and the path to reach each router.

Now the routing table for J is

	Delay	**Path**
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

Routing Information Protocol (RIP) is the best known and most widely used of the distance vector routing protocols. RIP version 1 (RIP v1), which is now outmoded, was the first routing protocol accepted as a standard for TCP/IP. RIP version 2 (RIP v2) provides authentication support, multicast announcing, and better support for classless networks. The Windows Server 2003 Routing and Remote Access service supports both RIP v1 and RIP v2 (for IPv4 only).

Using RIP, the maximum hop count from the first router to the destination is 15. Any destination greater than 15 hops away are considered unreachable. This limits the diameter of a RIP internetwork to 15. However, if you place your routers in a hierarchical structure, 15 hops can cover a large number of destinations.

Link State Routing

- To replace Distance Vector Routing
- Problem with Distance Vector Routing – uses only delay at a node – doesn't consider line capacity

Each router

1. Discover its neighbors and addresses – this is done by sending out HELLO packets on start-up and receiving replies from neighbors.
2. Measure the cost to each neighbor – measure the round-trip time taken to send an ECHO packet and to receive a reply.
3. Construct link state packets and send it to all routers – these link state packets contains what each router knows about the cost to each of its neighbors.
4. Compute the shortest path to each router based on received link state packets from other routers.

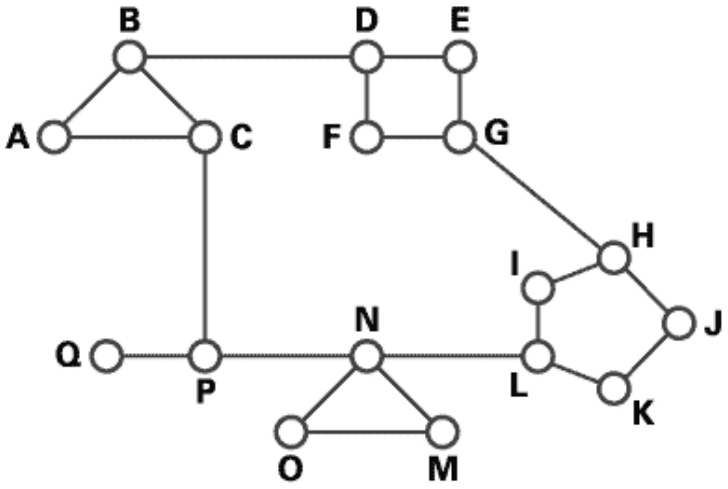
HIERARCHICAL ROUTING

In both link state and distance vector algorithms, every router has to save some information about other routers. When the network size grows, the number of routers in the network increases. Consequently, the size of routing table increases, and routers can't handle network traffic efficiently. Hierarchical routing solves this problem.

In hierarchical routing network is divided into regions. Each region has a main router.

All routers need to know only about the routers in their region and the main routers in the other regions. So routers just save one record in their table for every other region.

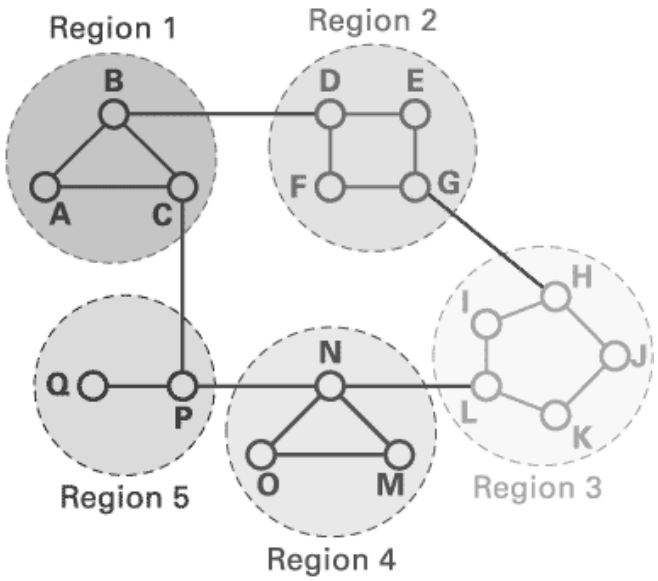
Suppose we use distance vector algorithms to find best routes between nodes. In the situation depicted below, every node of the network has to save a routing table with 17 records. Here is a typical graph and routing table for router A.



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

Routing Table for Router A

In this example, we have classified our network into five regions.



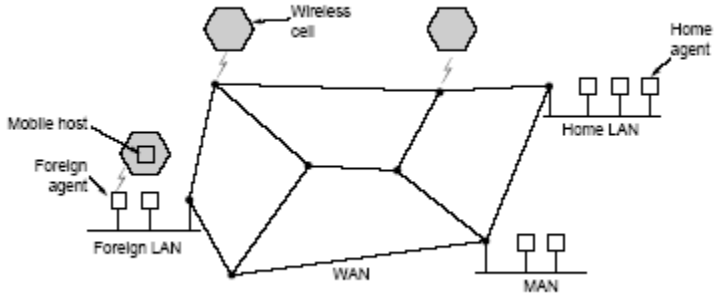
Destination	Line	Weight
A	---	---
B	B	1
C	C	1
Region 2	B	2
Region 5	C	2
Region 4	C	3
Region 3	C	4

Routing table for A in Hierarchical Routing

If A wants to send packets to any router in region 2 (D, E, F or G), it sends them to B. 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.

ROUTING FOR MOBILE HOSTS



In this routing, nodes can move amongst regions or areas.

There are three kinds of nodes

- Stationary
- Migratory
- Mobile

The challenge is how to find a portable host that could potentially be anywhere. Each mobile host has a home location, with a home agent. When the mobile host moves to a new location, it registers with a foreign agent. The foreign agent lets home agent know when a sender sends a packet to the mobile host. Home agent forwards packet to foreign agent to pass to mobile host. Home agent also notifies sender to send subsequent packets to mobile host's foreign agent.

BROADCAST ROUTING

Broadcasting is sending packets to more than one host in the subnet. The possible ways are,

- Sender sends one packet to each possible destination
- Flooding
- Multi-destination Routing
- Reverse path forwarding – router will forward the packet to its outgoing links only if the packet arrived on the shortest path back to the source.
- Using spanning tree (sink tree) – a node will forward the packet to all its neighbors in the spanning tree.

MULTICAST ROUTING

Some hosts spread over a large network can be put into "multicast" groups – packets sent out to a group is sent to every member of a group. E.g. news updates service where news is sent to every user who subscribes. It can't use broadcast since some machines are not meant to receive the packets. In a LAN, we are less security conscious and assume that the supporting protocols for each machine will ignore all packets not meant for it. Here we have a set of routers from very different networks. We can't trust that all routers (or their administrators) will not illegally read the packets.

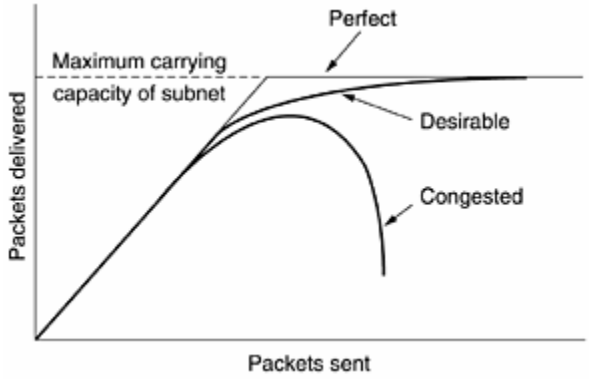
One way of doing multi-cast routing to a particular group:

- Compute a spanning tree joining all nodes – there may be more than one.
- Delete the links that do not lead to members of the group.
- Using this pruned spanning tree forward the packet to all the neighbors in the pruned spanning tree.

CONGESTION CONTROL

What happens when there are more packets in a path than the routers can handle? This situation is called congestion. At very high traffic, performance collapses completely and almost no packets are delivered.

When too much traffic is offered, congestion sets in and performance degrades sharply



Congestion can be brought on by several factors. If all of a sudden, streams of packets begin arriving on three or four input lines and all need the same output line, a queue will build up. If there is insufficient memory to hold all of them, packets will be lost. Regulating the average rate of data transmission is called traffic shaping.

Causes of congestion

- The input traffic rate exceeds the capacity of the output lines (Routers receiving packets from multiple incoming lines, all meant for the same outgoing line).
- Slow processors for the routers - in this case, the packets from the multiple incoming lines are coming in faster than the routers can process them and send them back out.
- Low bandwidth lines
- The routers' buffer is too limited.

Difference between congestion control and flow control

Flow control is between sender and receiver. In flow control, it is about not having a fast sender flood a slow receiver, when the slow receiver is not able to process the packets. An example when we have a flow control problem but no congestion is a fast supercomputer sends packets to a slow workstation. The packets are not delayed at any of the routers, and so get to the workstation very quickly. Unfortunately, the workstation can't process the packets fast enough. In this case, there is no congestion (in the network), but there is a flow problem (between the sender and receiver).

Congestion control makes sure that the subnet is able to carry the offered traffic. It is a global issue that involves the behavior of all hosts, all routers, store and forwarding processing within the routers etc.

CONGESTION CONTROL ALGORITHMS

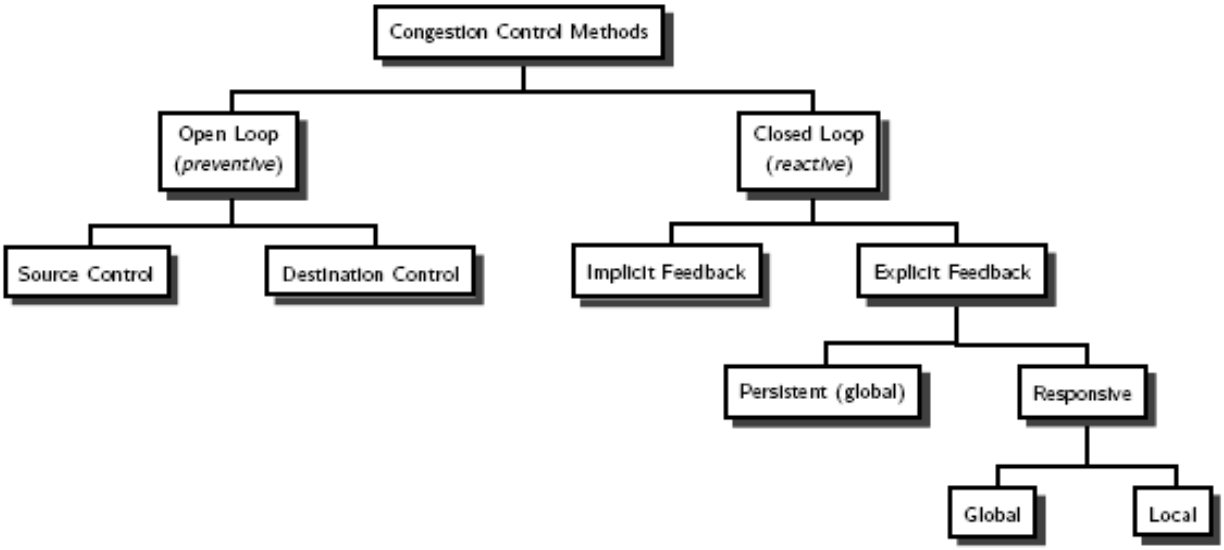
Congestion control algorithms can be classified as

a. **Open loop**
    - Attempts to solve the problem by good design
    - Makes sure that congestion does not occur
    - Once the system is up and running, midcourse corrections are not made
b. **Closed loop**

    Have feedback on congestion in real-time, and respond. The different steps are:

1. Monitor for congestion - Explicit monitors detects congestions by having the congested nodes send packets to notify others. Implicit monitors detect congestion by inferring from local data, such as how long acknowledgments take to get back. Chief metrics for monitoring the subnet for congestion are:
    - percentage of all packets discarded for lack of buffer space
    - average queue lengths
    - number of packets that time out and are retransmitted
    - average packet delay
    - standard deviation of packet delay
2. Pass the information to appropriate places where actions can be taken. Usually the best place to pass congestion information to is the source of the packets. The source is most able to adjust to congestion (by re-routing, slowing down, etc). The different methods to propagate the monitored congestion information are
    - The router detecting the congestion sends a separate warning packet to the traffic source.
    - A bit or field can be reserved in each packet. When a router detects a congested state, it fills in the field in all outgoing packets to warn the neighbors.
    - Hosts or routers send probe packets out periodically to explicitly ask about congestion and to route traffic around problem areas.
3. Adjust the system operations to correct the problem. The different methods to correct congestion problem are:
    - Increase the resource:
        - Using an additional line to temporarily increase the bandwidth between certain points.
        - Splitting traffic over multiple routes.
        - Using spare routers.
    - Decrease the load:
        - denying service to some users,
        - degrading service to some or all users
        - having users schedule their demands in a more predictable way.

Congestion Control Categories



Policies Affecting Network Congestion

Layer	Policies
Transport	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy • Timeout determination
Network	• Virtual circuits versus datagram inside the subnet • Packet queueing and service policy • Packet discard policy • Routing algorithm • Packet lifetime management
Data link	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy

Open Loop Solutions

Traffic Shaping

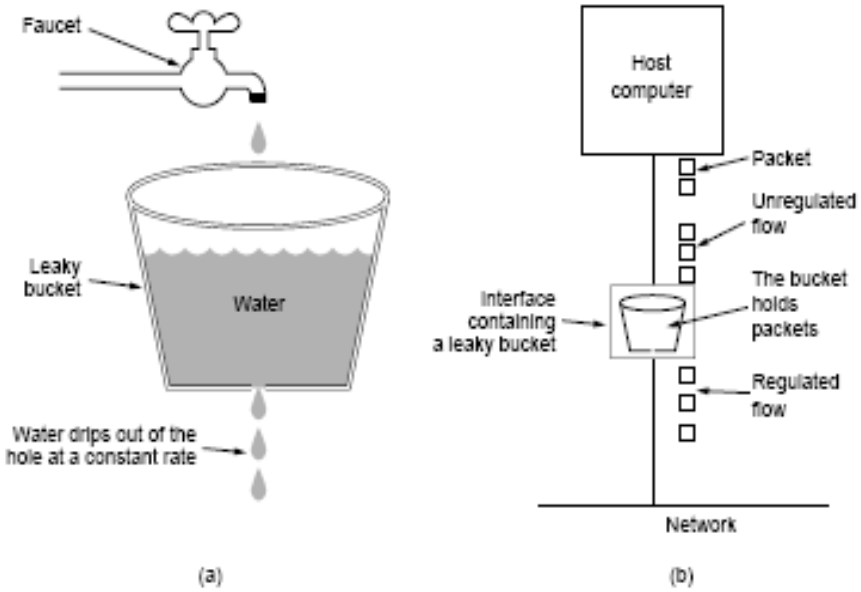
- Controlling congestion by controlling the rate of data transmission mainly at the source of the packets.
- Having host transmit at uniform rate usually cause less congestion – network less affected by bursts of packets.
- Requires an agreement between all senders, receivers and subnet on how the network flow should be like - if for example, most hosts agrees to transmit at a slow rate, but one decides it will transmit at the fast rate, then the fast host will end up hogging the lines it uses.

Algorithms used for traffic shaping are

- Leaky Bucket Algorithm
- Token Bucket Algorithm

Leaky Bucket Algorithm

The leaky bucket algorithm is similar to how a leaky bucket behaves under a tap. The drip from the leak is always constant regardless of how fast water is coming from a tap, and if the bucket is full, then water will overflow and be discarded.



In a network using the leaky bucket algorithm, data coming from a host goes through an interface. The interface stores up the packets from the source in memory and sends them out in a constant rate. If the memory fills up, the packets will be discarded (the host will re-transmit at a later time when it times out).

To handle variable size packets, instead of sending one packet per clock tick, a fixed number of bytes are transmitted.

Token Bucket Algorithm

Here also we use an interface (bucket) to regulate the flow, but we want to allow a slightly faster rate when the bucket is relatively full and slower when relatively empty. We do this by having the interface generate tokens. A packet being sent through the interface must consume a token. If there are no tokens, the packets must wait. By having the interface generate the tokens at a regular interval, we can have the tokens saved up at idle times, to be used for a burst when the packets come. And at busy times, the rate of transmission is still constant since the tokens are generated at a constant rate.

Token bucket algorithm design

– Leaky bucket holds **tokens**

– Tokens generated by a clock, one token every ΔT seconds

– Bucket has a maximum size (capacity) of c tokens

– For a cell to be transmitted, it must remove one token from the bucket

Performance of the token bucket algorithm

Let s = burst length (seconds)

c = bucket capacity (bytes)

ρ = token arrival rate (bytes/second)

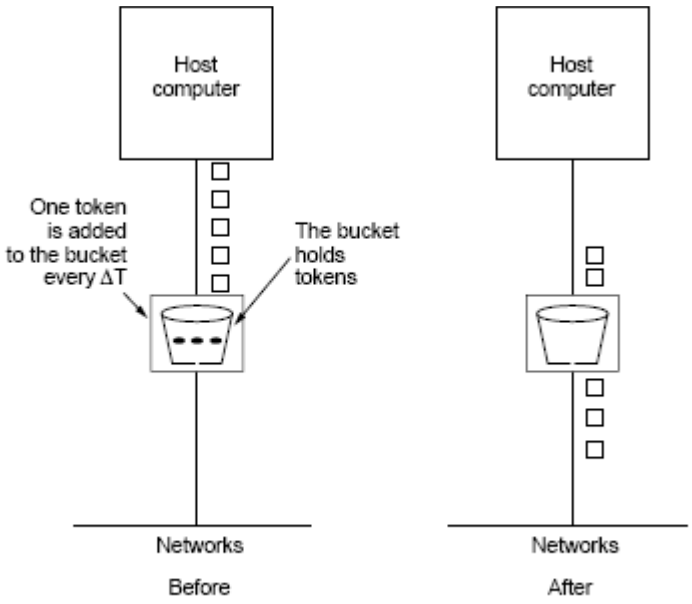
m = maximum source rate (bytes/second)

Then the maximum bytes sent from the token bucket during a burst is $\mathbf{c + \rho \times s}$

Maximum bytes the source can send during a burst is $\mathbf{m \times s}$

Setting the two equal and solving for s

$$\boldsymbol{s = c \div (m - \rho)}$$



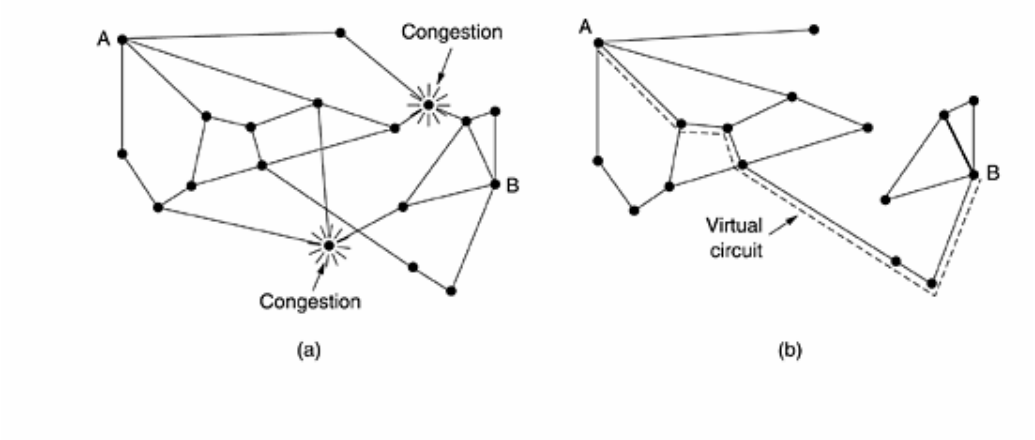
Closed Loop Solutions

The different closed loop solutions are

- Congestion control in virtual circuit subnets
- Choke packets
- Load shedding

Congestion Control in Virtual Circuit Subnets

- Admission control: once congestion has been signaled, no more virtual circuits are set up until the problem has gone away.
- Allow new VC, but choose an alternate path: Suppose that a host attached to router A wants to set up a connection to a host attached to router B. Normally, this connection would pass through one of the congested routers. To avoid this situation, we can redraw the subnet omitting the congested routers and all of their lines. The dashed line shows a possible route for the virtual circuit that avoids the congested routers.



- Negotiate an agreement between the host and subnet when a virtual circuit is set up: This agreement specifies the volume and shape of the traffic, quality of service required, and other parameters. The subnet will reserve resources along the path when the circuit is set up. These resources can include table and buffer space in the routers and bandwidth on the lines. In this way, congestion is unlikely to occur on the new virtual circuits because all the necessary resources are guaranteed to be available.

Choke Packets

This method can be used in either VCs or datagram subnets. Every router maintains a variable for every outgoing line. The variable indicates how often the line is used. The router will obviously know how often a particular outgoing line is being used. If the variable goes above a threshold value, the line goes into a warning state. If an incoming packet requires an outgoing line that is in warning state, a **choke packet** is sent back to the source and the packet is till forwarded as normal. Source receiving choke packets adjust their sending patterns by either slow down, or re-route through somewhere else.

Load Shedding

The strategy adopted is that, when routers can't handle the load, just dump incoming packets. But decision has to be made on which packets to dump, new or old packets. This depends on what type the packets are. In some applications, it is better that new packets get dumped. E.g. in interlaced image packets, packets sent out first (old packets) contain

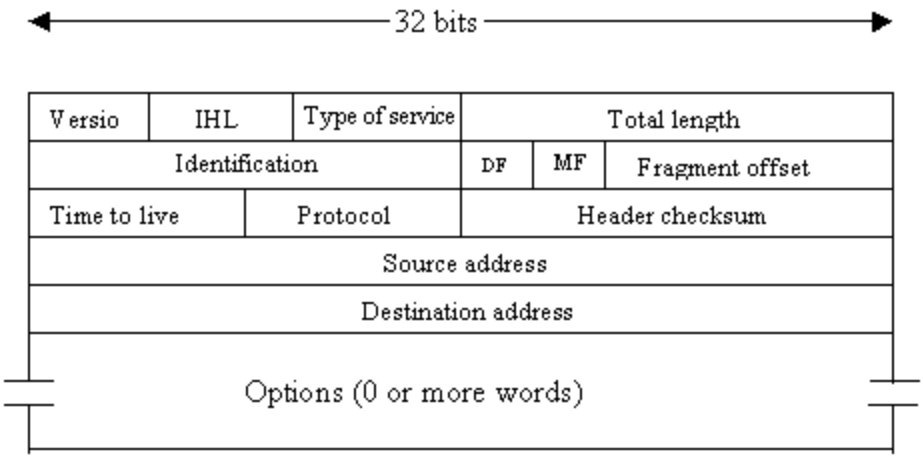
the coarse-grain image, and then the later packets (new packets) fill in the details. So getting the coarse-grain image without the details may still be usable, but having the details without the coarse-grain image to work with is not. On the other hand, in some applications, it is better that old packets get dumped. E.g. in file transfers, if any packets are to be discarded, it would be better that the one sent out earlier (old packets). This is so that the sender gets time out a lot sooner and re-transmits the file, rather than falsely believing that since the earlier packets are getting acknowledged, that everything is OK.
Another strategy is to dump packets with low priority. Source label the priority of the packets and put them in priority fields in the packet headers and routers can use these priority fields and dump the low priority packets first.

Jitter Control

In some audio and video transmission, speed for each packet is not as important as having a constant rate for all the packets. Routers can calculate if these packets are ahead or behind the scheduled rate, and during congestion hold off packets currently ahead of the schedule. This involves the packets having a time-stamp, and the routers using this time-stamp to determine if the packet is ahead of the schedule or not.

INTERNET PROTOCOL

The Internet is a collection of many different networks. The central protocol, which holds all the different networks together, is the **Internet Protocol** (or IP). The aim of IP is to transport datagrams from source to destination inside and outside the same network.



The IPv4 (Internet Protocol) header

Fields in an IPv4 header:

- Version: specifies which version of IP.
- IHL: this header's length - minimum 160 bits.
- Type of Service: what type of service this packet requires (e.g. reliability, speed, etc). Each router can try and satisfy this.
- Total length: this header's length - maximum 64Kb.
- Identification, DF, MF, Fragment Offset: for fragmenting packets that are too large.
- Time to live: value to limit packet lifetime, so that it doesn't float around forever - set by host and decreased by each router.
- Protocol: the protocol, which pass IP the data (e.g. TCP).
- Checksum: checksum for the header, to ensure there are no errors.
- Source and Destination Addresses: sender and receiver of the data.
- Options: other useful header fields not included above.

MODULE II

Lecture # 6

1. Synopsis

Data link layer – Design Issues

2. Target

At the completion of this lecture you should be able to answer questions like

1. What are the functions of the data link layer?
2. What are the design issues of the data link layer?
3. What is framing?
4. What are the different framing methods?
5. What is error control?
6. What are the requirements for error control?
7. What is flow control?

3. Introduction

In lecture 2 we discussed the different layers in the OSI reference model. In lecture 4 & 5 we discussed the different aspects of the physical layer. As stated above, in this lecture, we plan to explore more on the next layer i.e. the data link layer. In particular, we will emphasize the following

- Functions of the data link layer
- Services provided by the data link layer to the network layer
- Framing
- Error Control
- Flow Control

4. Revision / Prerequisites

Please refer lecture 2 for an introduction to the data link layer.

5.1. Data link Layer

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. It accomplishes this task by having the sender break the input data up into data frames (typically a few hundred bytes), 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.

The data link layer should provide error control between adjacent nodes. The data link layer provides the functional and procedural means to transfer data between network entities and to detect and possibly correct errors that may occur in the Physical layer. Examples of data link protocols are Ethernet for local area networks and PPP, HDLC and ADCCP for point-to-point connections.

This layer is made up of two components. The first component is **Logical Link Control**. This component determines where one frame of data ends and the next one starts.

The second component is **Media Access Control.** This component determines who is allowed to access the media at any one time. There are generally two forms of media access control: distributed and centralized.

5.2. Data Link Layer Design Issues

5.2.1 Services provided to the network layer

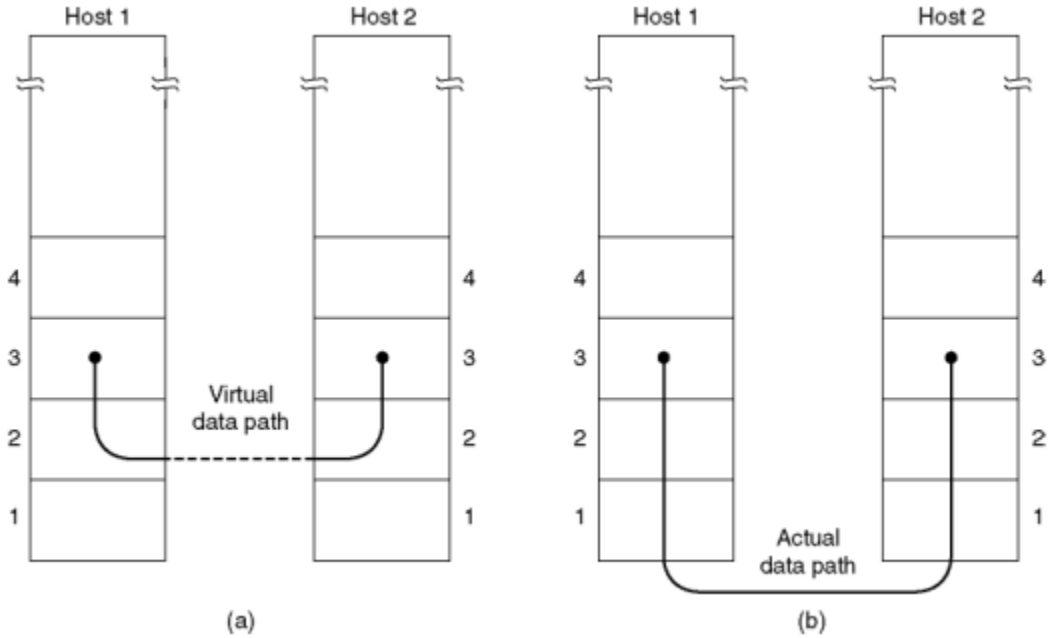


Figure 6.5.1. (a) Virtual Communication (b) Actual Communication

Functions of DLL

- Group the physical layer bit stream into units called ***frames***

- Sender checksums the frame and sends checksum together with data. The checksum allows the receiver to determine when a frame has been damaged in transit
- Receiver recomputes the checksum and compares it with the received value. If they differ, an error has occurred and the frame is discarded
- Perhaps return a *positive* or *negative acknowledgment* to the sender. A positive acknowledgment indicates that the frame was received without errors, while a negative acknowledgment indicates the opposite
- Flow control. Prevent a fast sender from overwhelming a slower receiver. For example, a supercomputer can easily generate data faster than a PC can consume it
- Provide service to the network layer

5.2.2. Framing

The DLL translates the physical layer's raw bit stream into discrete units (messages) called ***frames*.**

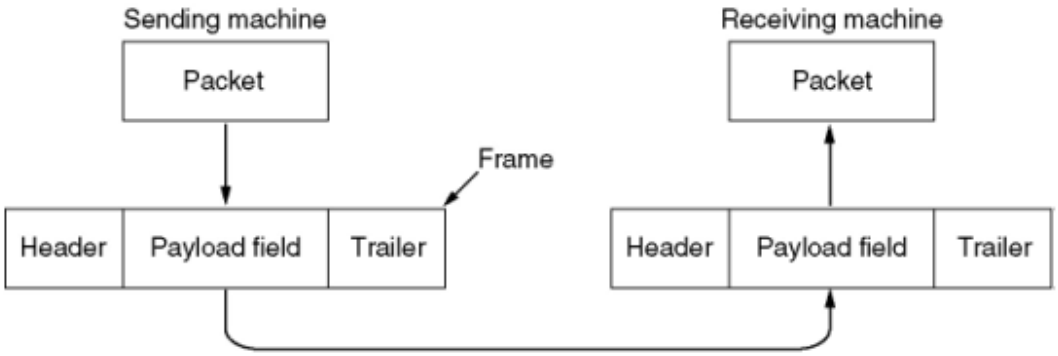


Figure 6.5.2 Relationship between packets and frames

Breaking the bit stream up to frame is difficult. One way to achieve this framing is to insert the time gap between frames. Network rarely make any guarantee about timing, so it is possible to these time gap might be squeezed out, or other time gap can be inserted during transmission. Different methods are used to mark the start and end of the frame.

Framing Methods

a) Length Count (Character Count):

Make the first field in the frame's header be the length of the frame. That way the receiver knows how big the current frame is and can determine where the next frame ends.

Disadvantage: Receiver loses synchronization when bits become garbled. If the bits in the count become corrupted during transmission, the receiver will think that the frame contains fewer (or more) bits than it actually does. Although checksum will detect the incorrect frames, the receiver will have difficulty resynchronizing to the start of a new frame. This technique is not used anymore, since better techniques are available.

b) Bit Stuffing:

Use reserved bit patterns to indicate the start and end of a frame. For instance, use the 4-bit sequence of 0111 to delimit consecutive frames. A frame consists of everything between two delimiters.

Problem: What happens if the reserved delimiter happens to appear in the frame itself? If we don't remove it from the data, the receiver will think that the incoming frame is actually two smaller frames!

Solution: Use ***bit stuffing***. Within the frame, replace every occurrence of two consecutive 1's with 110. E.g., append a zero bit after each pair of 1's in the data. This prevents 3 consecutive 1's from ever appearing in the frame.

Likewise, the receiver converts two consecutive 1's followed by a 0 into two 1's, but recognizes the 0111 sequence as the end of the frame.

Example1: The frame ``1011101" would be transmitted over the physical layer as ``0111101101010111".

Example2:

(a) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0

(b) 0 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 0 0 1 0

Stuffed bits

(c) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0

Note: When using bit stuffing, locating the start/end of a frame is easy, even when frames are damaged. The receiver simply scans arriving data for the reserved patterns. Moreover,

the receiver will resynchronize quickly with the sender as to where frames begin and end, even when bits in the frame get garbled.

The main disadvantage with bit stuffing is the insertion of additional bits into the data stream, wasting bandwidth. How much expansion? The precise amount depends on the frequency in which the reserved patterns appear as user data.

c) Character stuffing:

Same idea as bit-stuffing, but operates on bytes instead of bits.

Use reserved characters to indicate the start and end of a frame. For instance, use the two-character sequence DLE STX (Data-Link Escape, Start of Text) to signal the beginning of a frame, and the sequence DLE ETX (End of Text) to flag the frame's end.

Problem: What happens if the two-character sequence DLE ETX happens to appear in the frame itself?

Solution: Use ***character stuffing***. Within the frame, replace every occurrence of DLE with the two-character sequence DLE DLE. The receiver reverses the processes, replacing every occurrence of DLE DLE with a single DLE.

Example: If the frame contained ``A B DLE D E DLE", the characters transmitted over the channel would be ``DLE STX A B DLE DLE D E DLE DLE DLE ETX".

Disadvantage: character is the smallest unit that can be operated on; not all architectures are byte oriented.

d) Encoding Violations:

Send a signal that doesn't conform to any legal bit representation. In Manchester encoding, for instance, 1-bits are represented by a high-low sequence, and 0-bits by low-high sequences. The start/end of a frame could be represented by the signal low-low or high-high.

The advantage of encoding violations is that no extra bandwidth is required as in bit-stuffing. The IEEE 802.4 standard uses this approach.

5.2.3. Error Control

Error control is concerned with insuring that all frames are eventually delivered (possibly in order) to a destination.

Three Items are required for this.

Acknowledgements:

Typically, reliable delivery is achieved using the ``acknowledgments with retransmission" paradigm, whereby the receiver returns a special *acknowledgment* (ACK) frame to the sender indicating the correct receipt of a frame.

In some systems, the receiver also returns a *negative acknowledgment* (NACK) for incorrectly-received frames. This is nothing more than a hint to the sender so that it can retransmit a frame right away without waiting for a timer to expire.

Timers:

One problem that simple ACK/NACK schemes fail to address is recovering from a frame that is lost, and as a result, fails to solicit an ACK or NACK. What happens if an ACK or NACK becomes lost?

Retransmission timers are used to resend frames that don't produce an ACK. When sending a frame, schedule a timer to expire at some time after the ACK should have been returned. If the timer goes off, retransmit the frame.

Sequence Numbers:

Retransmissions introduce the possibility of duplicate frames. To suppress duplicates, add sequence numbers to each frame, so that a receiver can distinguish between new frames and old copies.

5.2.4. Flow Control

Flow control deals with throttling the speed of the sender to match that of the receiver. Usually, this is a dynamic process, as the receiving speed depends on such changing factors as the load, and availability of buffer space.

One solution is to have the receiver extend *credits* to the sender. For each credit, the sender may send one frame. Thus, the receiver controls the transmission rate by handing out credits.

6. Summary

1. 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.
2. The DLL translates the physical layer's raw bit stream into discrete units (messages) called ***frames*.**
3. Bit stuffing is to use reserved bit patterns to indicate the start and end of a frame.

4. Character stuffing uses reserved characters to indicate the start and end of a frame.
5. Error control is concerned with insuring that all frames are eventually delivered (possibly in order) to a destination.
6. *Flow control* deals with throttling the speed of the sender to match that of the receiver.

University Questions

1. What are the three major service types provided by the LLC layer?

7. Exercise questions

1. What are the major functions of data link layer?
2. One way of detecting errors is to transmit data as a block of n rows of k bits per row and adding parity bits to each row and each column. Will this scheme detect all single errors? Double errors? Triple errors?

8. Programming / Computational assignments

1. The following data fragment occurs in the middle of a data stream for which the character stuffing algorithm is used: DLE, STX, A, DLE, B, DLE, ETX. What is the output after stuffing?
2. If the bit string 0111101111101111110 is bit stuffed, what is the output string?

Lecture # 7

1. Synopsis

Error Detection and Correction

2. Target

At the completion of this lecture you should be able to answer questions like

1. What is hamming distance?
2. What is the difference between error detection and correction?
3. How is error correction done using Hamming code?
4. What is Cyclic Redundancy Codes?
5. How is the checksum calculated in CRC?

3. Introduction

In lecture 6 we have discussed that one of the functions of the data link layer is to perform error detection and correction. In this lecture we plan to explore some of the techniques used for error detection and correction. Before we begin on this topic, a revision of the topics that you have learned in your Data Communication paper may be helpful. In this lecture, we will emphasize the following

- Error Correction using Hamming code
- Error Detection using Cyclic Redundancy Code

4. Revision / Prerequisites

Please refer 243 to 260 of Data Communication & Networking by Behrouz A. Forouzan.

5.1. Error Detection and Correction

In data communication, line noise is a fact of life (e.g., signal attenuation, natural phenomenon such as lightning, and the telephone repairman). Moreover, noise usually occurs as bursts rather than independent, single bit errors. For example, a burst of lightning will affect a set of bits for a short time after the lightning strike.

There are two types of attacks against errors:

Error Detecting Codes:

Include enough redundancy bits to *detect* errors and use ACKs and retransmissions to recover from the errors.

Error Correcting Codes:

Include enough redundancy to detect *and* correct errors.

To understand errors, consider the following:

1. Messages (frames) consist of *m* data (message) bits and *r* redundancy bits, yielding an *n* = (*m*+*r*)-bit *codeword*.
2. *Hamming Distance*. Given any two codeword, we can determine how many of the bits differ. Simply exclusive or (XOR) the two words, and count the number of 1 bits in the result.
3. If two codewords are *d* bits apart, *d* errors are required to convert one to the other.
4. A code's *Hamming Distance* is defined as the minimum Hamming Distance between any two of its legal codeword (from all possible codewords).
5. In general, all 2^m possible data words are legal. However, by choosing check bits carefully, the resulting codeword will have a large Hamming Distance. The larger the Hamming distance, the better able the code can detect errors.

To detect *d* 1-bit errors requires having a Hamming Distance of at least *d*+1 bits.

To correct *d* errors requires 2*d*+1 bits. Intuitively, after *d* errors, the garbled messages is still closer to the original message than any other legal codeword.

Parity Bits

A single *parity bit* is appended to each data block (e.g. each character in ASCII systems) so that the number of 1 bits always adds up to an even (odd) number. 1000000(1) 1111101(0). The Hamming Distance for parity is 2, and it cannot correct even single-bit errors (but can detect single-bit errors). As another example, consider a 10-bit code used to represent 4 possible values: ``00000 00000", ``00000 11111", ``11111 00000", and ``11111 11111". Its Hamming distance is 5, and we can correct 2 single-bit errors:

For instance, ``10111 00010" becomes ``11111 00000" by changing only two bits.

However, if the sender transmits ``11111 00000" and the receiver sees ``00011 00000", the receiver will not correct the error properly. Finally, in this example we are guaranteed to catch all 2-bit errors, but we might do better: if ``00111 00111" contains 4 single-bit errors, we will reconstruct the block correctly.

Error correction is most useful in three contexts:

1. Simplex links (e.g., those that provide only one-way communication).
2. Long delay paths, where retransmitting data leads to long delays (e.g., satellites).

3. Links with very high error rates, where there is often one or two errors in each frame. Without forward error correction, most frames would be damaged, and retransmitting them would result in the frames becoming garbled again.

Error Correction using Hamming Code

Using Hamming code provides 100% certainty of detecting a single bit error in a code. Additionally, the use of Hamming code allows the erroneous bit to be identified. Hamming code has 11-bits comprising 7-bits of an ASCII code and an additional 4-bits which are parity bits.

A 7-bit ASCII character may be represented by an 11-bit Hamming code in which the additional 4 parity bits take the positions 1, 2, 4 and 8 in the binary sequence and are 0 or 1 according to specified criteria which allow not only a single error to be detected, but also identified.

As an example take the ASCII code for the $ sign. This is 0100100. In an 11-bit Hamming code sequence this bits fill the following positions:

_ _ 0 _ 1 0 0 _ 1 0 0

Bit #: 1 2 3 4 5 6 7 8 9 10 11

The additional bits are filled according to the following criteria

- bit 1, 0 or 1 so that the total number of 1s in bits 1,3,5,7,9,11 is *even*
- bit 2, 0 or 1, so that the total number of 1s in bits 2,3,6,7,10,11 is *even*
- bit 4, 0 or 1, so that the total number of 1s in bits 4,5,6,7 is *even*
- bit 8, 0 or 1, so that the total number of 1s in bits 8,9,10,11 is *even*

The Hamming code for the $ sign is therefore: 00011001100

If a bit is changed during communication then checking if the conditions above are met allows identification of the erroneous bit. For example, suppose the *6th* bit is changed from 0 to 1 so that the received code is 00011101100. Checking the parity bits:

- bit 1 - total number of 1s in bits 1,3,5,7,9,11 is even - passes test
- bit 2 - total number of 1s in bits 2,3,6,7,10,11 is odd - *fails* test
- bit 4 - total number of 1s in bits 4,5,6,7 is odd - *fails* test
- bit 8 - total number of 1s in bits 8,9,10,11 is even - *passes* test

Adding up the bit numbers of the failed tests gives 2 + 4 = 6 - correctly identifies the erroneous bit.

Error Detection

Error correction is relatively expensive (computationally and in bandwidth). For example, 10 redundancy bits are required to correct 1 single-bit error in a 1000-bit message. Detection? In contrast, detecting a single bit error requires only a single-bit, no matter how large the message. The most popular error detection codes are based on *polynomial codes* or *cyclic redundancy codes* (CRCs). Allows us to acknowledge correctly received frames and to discard incorrect ones.

CRC (Cyclic Redundancy Code)Checksums

The most popular error detection codes are based on *polynomial codes* or *cyclic redundancy codes.* Idea:

- Represent a *k*-bit frame as coefficients of a polynomial expansion ranging from $x^{k-1}$to x^0, with the high-order bit corresponding to the coefficient of x^{k-1}.
  For example, represent the string ``11011" as the polynomial: $x^4 + x^3 + x + 1$
- Perform modulo 2 arithmetic (e.g. XOR of the bits)
- Sender and receiver agree on a *generator polynomial*: *G*(*x*). (*G*(*x*) must be smaller than the number of bits in the message.)
- Append a *checksum* to message; let's call the message *M*(*x*), and the combination *T*(*x*). The checksum is computed as follows:
  1. Let *r* be the degree of *G*(*x*), append *r* zeros to M(x). Our new polynomial becomes $x^r M(x)$
  2. Divide $x^r M(x)$ by *G*(*x*) using modulo 2 arithmetic.
  3. Subtract the remainder from $x^r M(x)$ giving us T(x).
- When receiver gets *T*(*x*), it divides *T*(*x*) by *G*(*x*); if *T*(*x*) divides cleanly (e.g., no remainder), no error has occurred.

The presence of a remainder indicates an error.

Assume:

- the receiver gets *T*(*x*) + *E*(*x*), where each bit in *E*(*x*) corresponds to an error bit.
- *k* 1 bits indicate *k* single-bit errors.

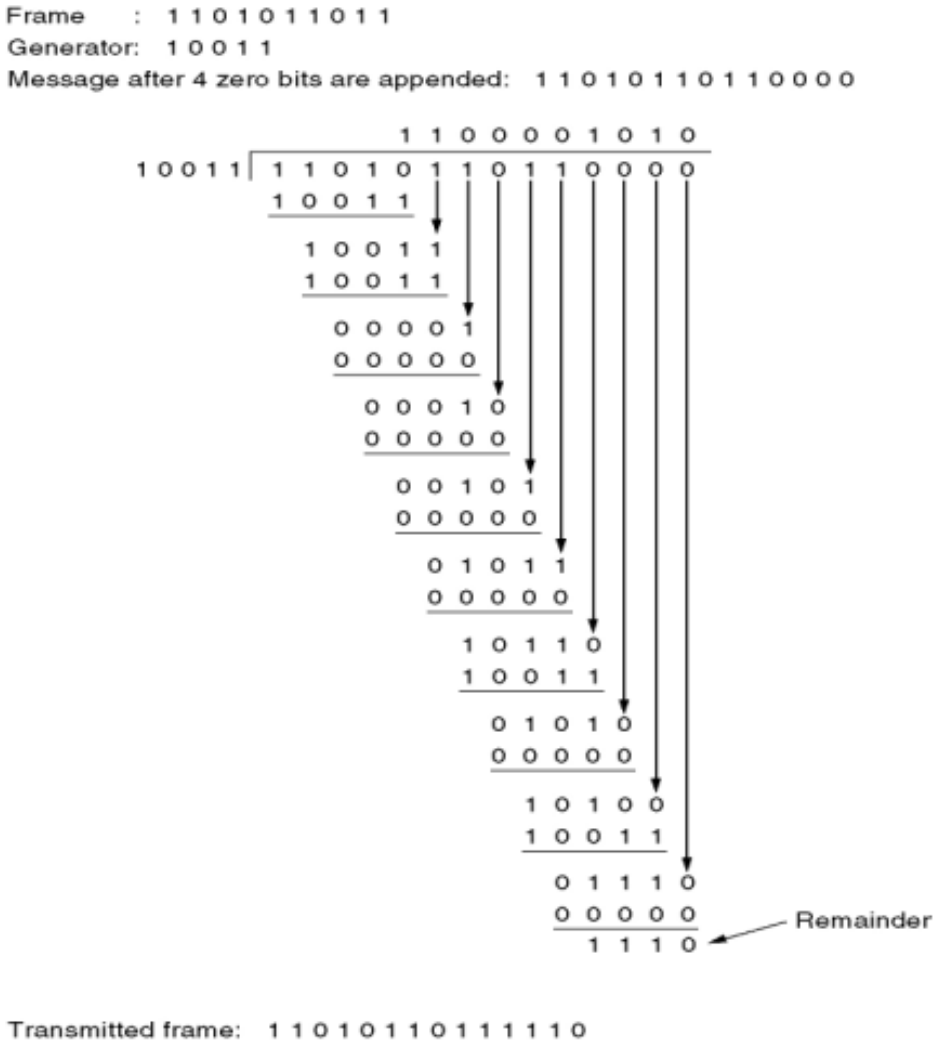
- Receiver computes $[T(x) + E(x)]/G(x) = E(x)/G(x)$.

Will detect:

- *single bit errors*. If a single-bit error occurs, $G(x)$ will detect it if it contains more than one term. If it contains only one term, it may or may not detect the error, depending on the $E(x)$ and $G(x)$.
- *two isolated single-bit errors*. Consider two single-bit errors:

$$E(x) = x^i + x^j = x^i(1 + x^{j-i})$$

Example:



Transmitted frame: 1 1 0 1 0 1 1 0 1 1 1 1 1 0

CRC Standards

There are currently three international standards:

- CRC-12: $x^{12} + x^{11} + x^3 + x^2 + x + 1$
- CRC-16: $x^{16} + x^{15} + x^2 + 1$
- CRC-CCITT: $x^{16} + x^{12} + x^5 + 1$

University Questions

1. Describe any one method of error detection.

6. Summary

1. Hamming Distance is calculaed by exclusive or (XOR) the two words, and count the number of 1 bits in the result.
2. If two codewords are *d* bits apart, *d* errors are required to convert one to the other.
3. The most popular error detection codes are based on *polynomial codes* or *cyclic redundancy codes* (CRCs).

7. Exercise questions

1. Explain the difference between error detection and correction.
2. What is hamming distance?
3. What is cyclic redundancy code?
4. How is the checksum calculated in CRC?
5. Explain how error correction is done using hamming code.

8. Programming / Computational assignments

1. Assume that the Frame polynomial is x^7+x^5+1 and Generator Polynomial: x^3+1. When the polynomial code method is employed, what is the check summed frame to be transmitted?

Lecture # 8

1. Synopsis:

Elementary Data Link Protocols

2. Target

At the completion of this lecture you should be able to answer questions like

1. Which are the elementary data link protocols?
2. Explain unrestricted simplex protocol.
3. Explain simplex Stop-and-wait protocol.
4. Explain Simplex protocol for noisy channel.

3. Introduction

In this lecture we will be discussing some elementary protocols in the data link layer. In particular, we will emphasize the following

- An unrestricted simplex protocol
- Simplex stop-and-wait protocol
- Simplex protocol for noisy channel

4. Revision / Prerequisites

Prerequisite for this lecture is that you should have some knowledge of C programming.

5.1. Elementary Data Link Protocols

As far as the data link layer concerned, the packet passed across the interface to it from the network layer. While the DLL accepts a packet, it encapsulates the packet into frame by adding header and trailer to it. Thus frame consists of an embedded packet and some control (header) information. The frame is then transmitted to the other DLL.

We will assume that there exist suitable library functions. The transmitting hardware computes and appends the checksum, so that DLL software need not worry about it. Initially the receiver has nothing to do. It sit simply and waiting for the frame arrival.

Library functions and predefined data structures are given below.

```
#define MAX_PKT 4 /* determines packet size in bytes */
typedef enum {false, true} boolean; /* boolean type */
typedef unsigned int seq_nr;  /* sequence or ack numbers */
typedef struct {unsigned char data[MAX_PKT];} packet; /* packet
definition */
```

```
typedef enum {data, ack, nak} frame_kind; /* frame_kind definition */

typedef struct
{                  /* frames are transported in this layer */
  frame_kind kind;      /* what kind of a frame is it? */
  seq_nr seq;           /* sequence number */
  seq_nr ack;           /* acknowledgement number */
  packet info;          /* the network layer packet */
} frame;

/* Wait for an event to happen; return its type in event. */
void wait_for_event(event_type *event);

/* Fetch a packet from the network layer for transmission on the
channel. */
void from_network_layer(packet *p);

/* Deliver information from an inbound frame to the network layer. */
void to_network_layer(packet *p);

/* Go get an inbound frame from the physical layer and copy it to r. */
void from_physical_layer(frame *r);

/* Pass the frame to the physical layer for transmission. */
void to_physical_layer(frame *s);

/* Start the clock running and enable the timeout event. */
void start_timer(seq_nr k);

/* Stop the clock and disable the timeout event. */
void stop_timer(seq_nr k);

/* Start an auxiliary timer and enable the ack_timeout event. */
void start_ack_timer(void);

/* Stop the auxiliary timer and disable the ack_timeout event. */
void stop_ack_timer(void);
```

```
/* Allow the network layer to cause a network_layer_ready event. */
void enable_network_layer(void);

/* Forbid the network layer from causing a network_layer_ready event.
*/
void disable_network_layer(void);

/* Macro inc is expanded in-line: Increment k circularly. */
#define inc(k) if (k < MAX_SEQ) k = k + 1; else k = 0
```

5.1.1. An Unrestricted Simplex Protocol

This is the simplest protocol in which data are transmitted in one direction only. Both the transmitting and receiving network layers are always ready. Processing time is ignored and infinite buffer space is available. Also it is assumed that the communication channel between the data link layers never loses or damages 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.

```
/* Protocol 1 provides for data transmission in one direction only,
from    sender to receiver.  The communication channel is assumed to be
error free, and the receiver is assumed to be able to process all the
input infinitely fast. Consequently, the sender just sits in a loop
pumping data out onto the line as    fast as it can. */

typedef enum {frame_arrival} event_type;
#include "protocol.h"

void sender1(void)
{
  frame s;  /* buffer for an outbound frame */
  packet buffer;  /* buffer for an outbound packet */

  while (true)
      {
        from_network_layer(&buffer);      /* go get something to send
*/
        s.info = buffer;      /* copy it into s for transmission */
```

```
        to_physical_layer(&s);      /* send it on its way */
      }
}

void receiver1(void)
{
  frame r;
  event_type event;     /* filled in by wait, but not used here */

  while (true)
{
        wait_for_event(&event);   /*only possibility is frame_arrival*/
        from_physical_layer(&r);    /* go get the inbound frame */
        to_network_layer(&r.info);  /* pass the data to the network
                                       layer */
  }
}
```

5.1.2. A Simplex Stop-and-Wait Protocol

Stop-and-wait protocol provides for a one-directional flow of data from sender to receiver. The communication channel is once again assumed to be error free, as in protocol 1. However, this time, the receiver has only a finite buffer capacity and a finite processing speed, so the protocol must explicitly prevent the sender from flooding the receiver with data faster than it can be handled.

```
/* Protocol 2 (stop-and-wait) also provides for a one-directional flow
of data from    sender to receiver. The communication channel is once
again assumed to be error    free, as in protocol 1. However, this
time, the receiver has only a finite buffer    capacity and a finite
procesing speed, so the protocol must explicitly prevent the sender
from flooding the receiver with data faster than it can be handled. */

typedef enum {frame_arrival} event_type;
#include "protocol.h"

void sender2(void)
{
  frame s;              /* buffer for an outbound frame */
  packet buffer;        /* buffer for an outbound packet */
```

```
  event_type event;     /* frame_arrival is the only possibility */

  while (true)
 {
        from_network_layer(&buffer);  /* go get something to send */
        s.info = buffer;          /* copy it into s for transmission */
        to_physical_layer(&s);      /* bye bye little frame */
        wait_for_event(&event);     /* do not proceed until given the
                                       go ahead */
  }
}

void receiver2(void)
{
  frame r, s;           /* buffers for frames */
  event_type event;     /* frame_arrival is the only possibility */
  while (true)
  {
        wait_for_event(&event); /* only possibility is frame_arrival */
        from_physical_layer(&r);    /* go get the inbound frame */
        to_network_layer(&r.info);  /* pass the data to the network
                                    layer */
        to_physical_layer(&s);      /* send a dummy frame to awaken
                                    sender */
  }
}
```

5.1.3. A Simplex Protocol for a Noisy Channel

This protocol allows unidirectional data flow over an unreliable channel.

```
/* Protocol 3 (par) allows unidirectional data flow over an unreliable
channel. */

#define MAX_SEQ 1 /* must be 1 for protocol 3 */
typedef enum  {frame_arrival, cksum_err, timeout} event_type;
#include "protocol.h"

void sender3(void)
{
  seq_nr next_frame_to_send;    /* seq number of next outgoing frame */
```

```
  frame s;                        /* scratch variable */
  packet buffer;                  /* buffer for an outbound packet */
  event_type event;

  next_frame_to_send = 0;    /* initialize outbound sequence numbers */
  from_network_layer(&buffer);      /* fetch first packet */
  while (true)
  {
        s.info = buffer;      /* construct a frame for transmission */
        s.seq = next_frame_to_send; /*insert sequence number in frame*/
        to_physical_layer(&s);      /* send it on its way */
        start_timer(s.seq);   /* if answer takes too long, time out */
        wait_for_event(&event);     /* frame_arrival, cksum_err,
                                    timeout */
        if (event == frame_arrival)
            {
                from_physical_layer(&s);  /* get the acknowledgement */
                if (s.ack == next_frame_to_send)
                  {
                   from_network_layer(&buffer);   /*get the next one
                                                  to send*/
                        inc(next_frame_to_send);   /* invert
                                                next_frame_to_send */
                  }
            }
  }
}

void receiver3(void)
{
  seq_nr frame_expected;
  frame r, s;
  event_type event;

  frame_expected = 0;
  while (true)
  {
```

```
        wait_for_event(&event);  /* possibilities: frame_arrival,
                                    cksum_err */
        if (event == frame_arrival)
            {
                        /* A valid frame has arrived. */
                from_physical_layer(&r);  /* go get the newly arrived
                                             frame */
                if (r.seq == frame_expected)
                 {
                        /* This is what we have been waiting for. */
                  to_network_layer(&r.info);    /* pass the data to the
                                                   network layer */
                        inc(frame_expected);    /* next time expect the
                                                   other sequence nr */
                 }
                s.ack = 1 - frame_expected;     /* tell which frame is
                                                   being acked */
                to_physical_layer(&s);    /*only the ack field is use*/
           }
  }
}
```

6. Summary

1. Unrestricted Simplex Protocol is the simplest protocol in which data are transmitted in one direction only and processing time is ignored and infinite buffer space is available.
2. Stop-and-wait protocol provides for a one-directional flow of data from sender to receiver but the receiver has only a finite buffer capacity and a finite processing speed.
3. Simplex protocol for noisy channel allows unidirectional data flow over an unreliable channel.

7. Exercise questions

Given below is a psuedo code of a data link layer protocol. Identify the protocol

```
void sender (void)
{
        frame s;
```

```
        packet buffer;
        event_type event;
        while (true)
        {
                from_network_layer(&buffer);
                s.info =  buffer;
                to_physical_layer(&s);
                wait_for_event(&event);
        }
}
void receiver(void)
{
        frame r,s;
        event_type event;
        while (true)
        {
                wait_for_event(&event);
                from_physical_layer(&r);
                to_network_layer(&r.info);
                to_physical_layer(&s);
        }
}
```

Lecture # 9

1. Synopsis:

Sliding Window Protocols

2. Target: At the completion of this lecture you should be able to answer questions like

1. Explain one bit sliding window protocol (Stop and wait)
2. Explain Go-back n protocol
3. Explain selective repeat protocol

3. Introduction

In lecture 8 we discussed the elementary data link protocols like unrestricted simplex protocol, stop and wait protocol and simplex protocol for a noisy channel. In all these protocols data frames were transmitted in one direction only. As stated above, in this lecture, we plan to explore three protocols that are more robust. These protocols belong to a class of protocols called sliding window protocols. In particular, we will emphasize the following

1. One Bit Sliding Window Protocol
2. Go Back n Protocol
3. Selective Repeat Protocol

4. Revision / Prerequisites

Please refer to pages 202 to 206 of your textbook.

5.1. Sliding Window Protocols

Sliding window is used by most connection oriented network protocol, among others, the Point-to-Point protocol (PPP) which many people use to establish their home PC as temporary Internet node via a phone-line connection to an existing node. Sliding Window Protocol assumes two-way communication (full duplex). It uses two types of frames:

1. Data
2. Ack (sequence number of last correctly received frame)

The basic idea of sliding window protocol is that both sender and receiver keep a ``window" of acknowledgment. The sender keeps the value of expected acknowledgment; while the receiver keeps the value of expected receiving frame. When it receives an acknowledgment from the receiver, the sender advances the window. When it receives the expected frame, the receiver advances the window. When communication links are

noisy and loss rate high, ACK schemes are preferred: a sender has to receive an acknowledgment from each receiver of a message. If ACKs are not received, the message is resent until an ACK has been received.

The idea of sliding windows is to keep track of the acknowledgements for each ID. However, a scheme in which a sender send a single message (e.g. to multiple receivers in a group) and then waits for all ACKs is to slow: a sender should be able to send a number of messages and a separate thread should receive ACKs, and resend messages with ACKs missing.

The senders and receivers each maintain a window of messages for which no ACKs have been received: a window is essentially a sequence of message IDs, starting with a low water mark and bounded by a high water mark. Whenever an ACK is received, the low and high water marks are advanced by 1, this allows 1 more ACK to be received, therefore sliding the window 1 to the right. When the window is full, an ACK is either discarded, or some kind of flow control is used to throttle the sender until there is more space available.

Sliding windows usually start out with a given size, however, more sophisticated protocols will dynamically adapt the window size, trying to find an agreed-upon size between sender and receiver.

The characteristics of sliding windows used at the sender and receiver usually involve

a. Error correction (by retransmission)
b. Flow control
c. Message ordering by sender (FIFO).

Example:

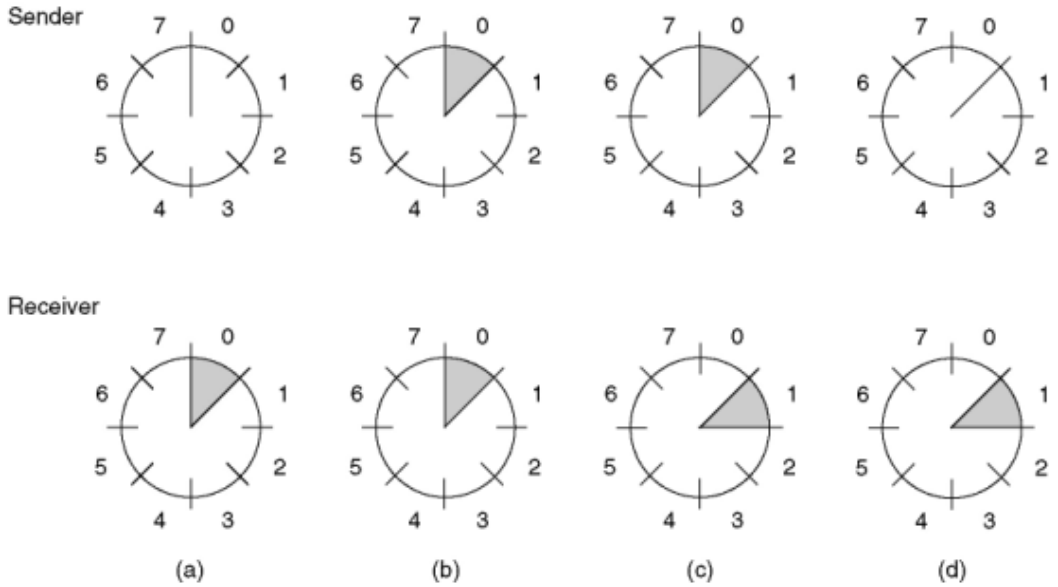


Figure 9.5.1. A sliding window of size 1, with a 3-bit sequence number (a) Initially (b) After the first frame has been sent (c) After the first frame has been received (d) After the first acknowledgement has been received

5.1.2. Protocol 4: One Bit Sliding Window Protocol (Stop And Wait)

One bit sliding window protocol is also called Stop-And-Wait protocol. In this protocol, the sender sends out one frame, waits for acknowledgment before sending next frame, thus the name Stop-And-Wait.

Problem with Stop-And-Wait protocol is that it is very inefficient. At any one moment, only in frame is in transition. The sender will have to wait at least one round trip time before sending next. The waiting can be long for a slow network such as satellite link.

```
/* Protocol 4 (sliding window) is bidirectional and is more robust than
protocol 3. */

#define MAX_SEQ 1 /* must be 1 for protocol 4 */
typedef enum {frame_arrival, cksum_err, timeout} event_type;
#include "protocol.h"

void protocol4 (void)
{
  seq_nr next_frame_to_send;  /* 0 or 1 only */
  seq_nr frame_expected;      /* 0 or 1 only */
  frame r, s;     /* scratch variables */
  packet buffer;  /* current packet being sent */
```

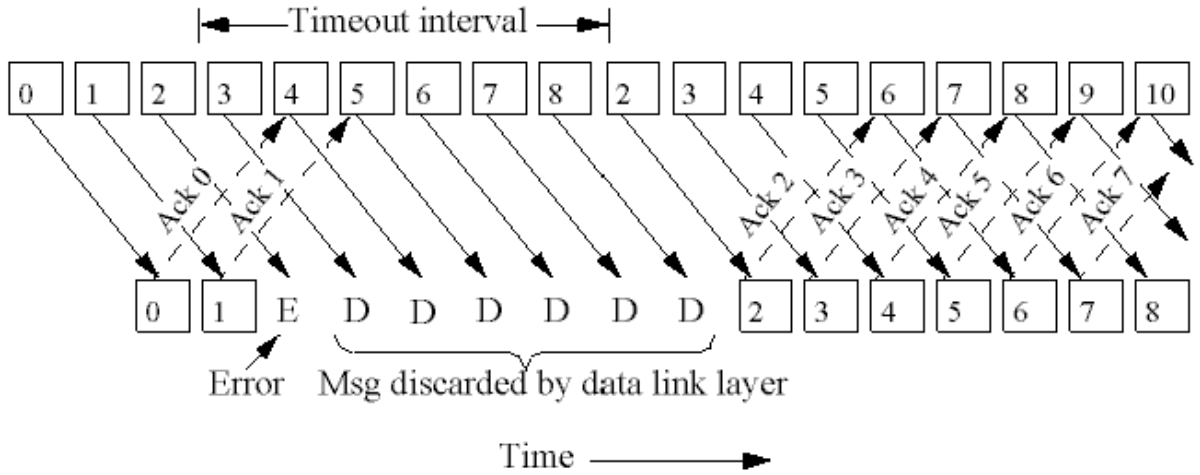
```
event_type event;
next_frame_to_send = 0;     /* next frame on the outbound stream */
frame_expected = 0;   /* number of frame arriving frame expected */
from_network_layer(&buffer);      /* fetch a packet from the network
                                  layer */
s.info = buffer;      /* prepare to send the initial frame */
s.seq = next_frame_to_send; /* insert sequence number into frame */
s.ack = 1 - frame_expected; /* piggybacked ack */
to_physical_layer(&s);      /* transmit the frame */
start_timer(s.seq);   /* start the timer running */
while (true)
{
      wait_for_event(&event); /* could be: frame_arrival, cksum_err,
                                  timeout */
      if (event == frame_arrival)
      {                     /* a frame has arrived undamaged. */
              from_physical_layer(&r);  /* go get it */

              if (r.seq == frame_expected)
                {                 /* Handle inbound frame stream. */
                   to_network_layer(&r.info); /* pass packet to
                                                 network layer */
                   inc(frame_expected); /* invert sequence number
                                           expected next */
                }
              if (r.ack == next_frame_to_send)
                {     /* handle outbound frame stream. */
                  from_network_layer(&buffer);  /* fetch new packet
                                                   from network layer */
                  inc(next_frame_to_send);    /* invert sender's
                                                 sequence number */
                }
      }
      s.info = buffer;      /* construct outbound frame */
      s.seq = next_frame_to_send; /*insert sequence number into it */
      s.ack = 1 - frame_expected; /* seq number of last received
                                     frame */
      to_physical_layer(&s);      /* transmit a frame */
```

```
        start_timer(s.seq);   /* start the timer running */
  }
}
```

5.1.3. Protocol 5: Go back n

In Go back n protocol if there is one frame k missing, the receiver simply discard all subsequent frames k+1, k+2, ..., sending no acknowledgments. So the sender will retransmit frames from k onwards. Figure 3-15(a) on page 208. This effectively sets the receiver window size to be 1. This can be a waste of bandwidth.




```
/* Protocol 5 (pipelining) allows multiple outstanding frames. The
sender may transmit up    to MAX_SEQ frames without waiting for an ack.
In addition, unlike the previous protocols, the network layer is not
assumed to have a new packet all the time. Instead, the network layer
causes a network_layer_ready event when there is a packet to send. */

#define MAX_SEQ 7 /* should be 2^n - 1 */
typedef enum {frame_arrival, cksum_err, timeout, network_layer_ready}
event_type;
#include "protocol.h"

static boolean between(seq_nr a, seq_nr b, seq_nr c)
{
/* Return true if (a <=b < c circularly; false otherwise. */
  if (((a <= b) && (b < c)) || ((c < a) && (a <= b)) || ((b < c) && (c
< a)))
        return(true);
    else
```

```
        return(false);
}
static void send_data(seq_nr frame_nr, seq_nr frame_expected, packet
buffer[])
{
            /* Construct and send a data frame. */
            frame s;    /* scratch variable */

  s.info = buffer[frame_nr];  /* insert packet into frame */
  s.seq = frame_nr;     /* insert sequence number into frame */
  s.ack = (frame_expected + MAX_SEQ) % (MAX_SEQ + 1); /*piggyback ack*/
  to_physical_layer(&s);      /* transmit the frame */
  start_timer(frame_nr);      /* start the timer running */
}

void protocol5(void)
{
  seq_nr next_frame_to_send;  /*MAX_SEQ > 1; used for outbound stream*/
  seq_nr ack_expected;  /* oldest frame as yet unacknowledged */
  seq_nr frame_expected;    /* next frame expected on inbound stream */
  frame r;                    /* scratch variable */
  packet buffer[MAX_SEQ+1];   /* buffers for the outbound stream */
  seq_nr nbuffered;           /* # output buffers currently in use */
  seq_nr i;                   /* used to index into the buffer array */
  event_type event;

  enable_network_layer();     /* allow network_layer_ready events */
  ack_expected = 0;           /* next ack expected inbound */
  next_frame_to_send = 0;     /* next frame going out */
  frame_expected = 0;         /* number of frame expected inbound */
  nbuffered = 0;              /* initially no packets are buffered */
  while (true)
  {
     wait_for_event(&event);  /* four possibilities: see event_type
                                 above */

     switch(event)
      {
```

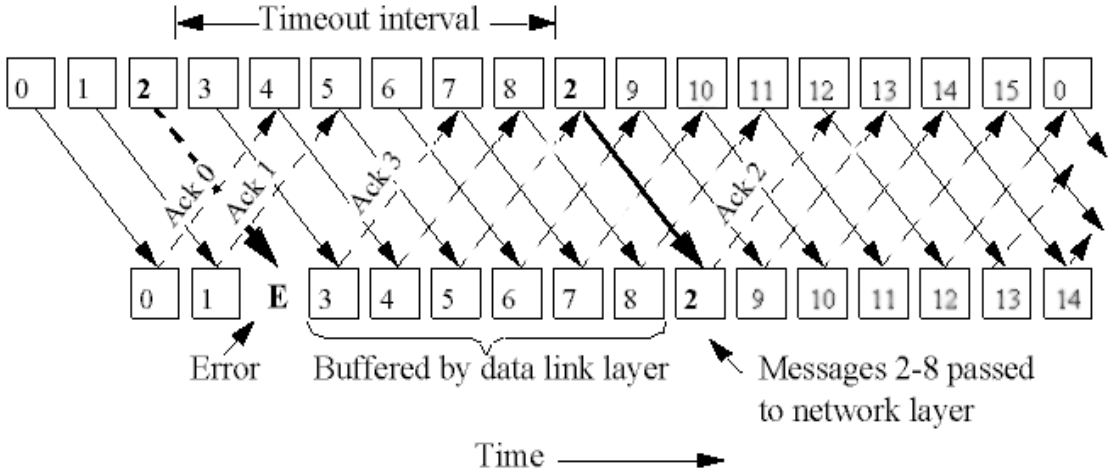
```
case network_layer_ready:   /* the network layer has a packet
                            to send */
                /* Accept, save, and transmit a new frame. */
from_network_layer(&buffer[next_frame_to_send]); /* fetch new
                                                    packet */
nbuffered = nbuffered + 1;  /* expand the sender's window */
send_data(next_frame_to_send, frame_expected, buffer);
                                  /* transmit the frame */
inc(next_frame_to_send);    /* advance sender's upper window
                            edge */
 break;
case frame_arrival:   /* a data or control frame has arrived */
from_physical_layer(&r);    /* get incoming frame from physical
                            layer */

if (r.seq == frame_expected)
    {                  /* Frames are accepted only in order. */
       to_network_layer(&r.info); /* pass packet to network
                                  layer */
       inc(frame_expected); /* advance lower edge of receiver's
                            window */
     }
        /* Ack n implies n - 1, n - 2, etc.  Check for this. */
while (between(ack_expected, r.ack, next_frame_to_send))
     {             /* Handle piggybacked ack. */
       nbuffered = nbuffered - 1; /*one frame fewer buffered */
       stop_timer(ack_expected);  /* frame arrived intact; stop
                                  timer */
       inc(ack_expected);   /* contract sender's window */
     }
        break;
case cksum_err: ;           /* just ignore bad frames */
        break;
case timeout:   /*trouble; retransmit all outstanding frames */
  next_frame_to_send = ack_expected;    /* start retransmitting
                                        here */
  for (i = 1; i <= nbuffered; i++)
    {
```

```
                send_data(next_frame_to_send, frame_expected, buffer);
                                    /* resend 1 frame */
                inc(next_frame_to_send);      /* prepare to send the next
                                                 one */
              }
        }
        if (nbuffered < MAX_SEQ)
           enable_network_layer();
        else
           disable_network_layer();
     }
}
```

5.1.4. Protocol 6: Selective Repeat

Another strategy is to re-send only the ones that are actually lost or damaged. The receiver buffers all the frames after the lost one. When the sender finally noticed the problem (e.g. no ack for the lost frame is received within time-out limit), the sender retransmits the frame in question.



/* Protocol 6 (nonsequential receive) accepts frames out of order, but passes packets to the network layer in order. Associated with each outstanding frame is a timer. When the timer goes off, only that frame is retransmitted, not all the outstanding frames, as in protocol 5. */

```
#define MAX_SEQ 7 /* should be 2^n - 1 */
#define NR_BUFS ((MAX_SEQ + 1)/2)
```

```
typedef enum {frame_arrival, cksum_err, timeout, network_layer_ready,
ack_timeout} event_type;
#include "protocol.h"
boolean no_nak = true;  /* no nak has been sent yet */
seq_nr oldest_frame = MAX_SEQ+1;  /* init value is for the simulator */

static boolean between(seq_nr a, seq_nr b, seq_nr c)
{
/* Same as between in protocol5, but shorter and more obscure. */
  return ((a <= b) && (b < c)) || ((c < a) && (a <= b)) || ((b < c) &&
(c < a));
}

static void send_frame(frame_kind fk, seq_nr frame_nr, seq_nr
frame_expected, packet buffer[])
{
                  /* Construct and send a data, ack, or nak frame. */
  frame s;        /* scratch variable */

  s.kind = fk;    /* kind == data, ack, or nak */
  if (fk == data) s.info = buffer[frame_nr % NR_BUFS];
  s.seq = frame_nr;           /* only meaningful for data frames */
  s.ack = (frame_expected + MAX_SEQ) % (MAX_SEQ + 1);
  if (fk == nak) no_nak = false;    /* one nak per frame, please */
  to_physical_layer(&s);            /* transmit the frame */
  if (fk == data) start_timer(frame_nr % NR_BUFS);
  stop_ack_timer();                 /*no need for separate ack frame */
}

void protocol6(void)
{
  seq_nr ack_expected;        /* lower edge of sender's window */
  seq_nr next_frame_to_send;  /* upper edge of sender's window + 1 */
  seq_nr frame_expected;      /* lower edge of receiver's window */
  seq_nr too_far;             /* upper edge of receiver's window + 1 */
  int i;                      /* index into buffer pool */
  frame r;                    /* scratch variable */
  packet out_buf[NR_BUFS];    /* buffers for the outbound stream */
```

```
packet in_buf[NR_BUFS];     /* buffers for the inbound stream */
boolean arrived[NR_BUFS];   /* inbound bit map */
seq_nr nbuffered;          /*how many output buffers currently used*/
event_type event;

enable_network_layer();     /* initialize */
ack_expected = 0;          /*next ack expected on the inbound stream*/
next_frame_to_send = 0;     /* number of next outgoing frame */
frame_expected = 0;         /* frame number expected */
too_far = NR_BUFS;          /* receiver's upper window + 1 */
nbuffered = 0;              /* initially no packets are buffered */

for (i = 0; i < NR_BUFS; i++) arrived[i] = false;
while (true)
 {
   wait_for_event(&event);  /* five possibilities: see event_type
                            above */
   switch(event)
    {
      case network_layer_ready:   /* accept, save, and transmit a new
                                  frame */
       nbuffered = nbuffered + 1; /* expand the window */
       from_network_layer(&out_buf[next_frame_to_send % NR_BUFS]);
                                  /*fetch new packet */
       send_frame(data, next_frame_to_send, frame_expected, out_buf);
                                  /* transmit the frame */
      inc(next_frame_to_send);    /* advance upper window edge */
      break;

      case frame_arrival:   /* a data or control frame has arrived */
      from_physical_layer(&r);    /* fetch incoming frame from
                                  physical layer */
      if (r.kind == data)
          {
                     /* An undamaged frame has arrived. */
            if ((r.seq != frame_expected) && no_nak)
             send_frame(nak, 0, frame_expected, out_buf);
            else
```

```
                start_ack_timer();
               if (between(frame_expected, r.seq, too_far) &&
                   (arrived[r.seq%NR_BUFS] == false))
                {
                              /* Frames may be accepted in any order. */
                arrived[r.seq % NR_BUFS] = true; /*mark buffer as full*/
                in_buf[r.seq % NR_BUFS] = r.info;  /* insert data into
                                                      buffer */
                while (arrived[frame_expected % NR_BUFS])
                   {
                                   /* Pass frames and advance window. */

                    to_network_layer(&in_buf[frame_expected % NR_BUFS]);
                    no_nak = true;
                    arrived[frame_expected % NR_BUFS] = false;
                    inc(frame_expected);   /* advance lower edge of
                                              receiver's window */
                    inc(too_far);    /* advance upper edge of receiver's
                                        window */
                     start_ack_timer();    /* to see if (a separate ack
                                              is needed */
                      }
                }
              }
                if((r.kind==nak) &&
etween(ack_expected,(r.ack+1)%(MAX_SEQ+1),next_frame_to_send))
                send_frame(data, (r.ack+1) % (MAX_SEQ + 1),
                frame_expected, out_buf);

                while (between(ack_expected, r.ack,
                next_frame_to_send))
                  {
                   nbuffered = nbuffered - 1;/*handle piggybacked ack*/
                   stop_timer(ack_expected % NR_BUFS);  /* frame
                                                arrived intact */
                   inc(ack_expected);     /* advance lower edge of
                                             sender's window */
                   }
```

```
                break;

            case cksum_err: if (no_nak) send_frame(nak, 0,
              frame_expected,out_buf); break;     /* damaged frame */
              case timeout: send_frame(data, oldest_frame,
              frame_expected, out_buf); break;    /* we timed out */
              case ack_timeout: send_frame(ack,0,frame_expected,
              out_buf);   /* ack timer expired; send ack */
        }

        if (nbuffered < NR_BUFS) enable_network_layer(); else
        disable_network_layer();
    }
}
```

University Questions

1. How is error control achieved using Sliding Window Protocol? Explain "Go-Back-N" ARQ.
2. Under what circumstances can Selective Repeat ARQ be gainfully employed? Why?
3. Discuss about one bit sliding window protocol.

6. Summary

1. The technique of temporarily delaying outgoing acknowledgements so that they can be hooked onto the next outgoing data frame is called piggy backing.
2. One bit sliding window protocol is also called Stop-And-Wait protocol.
3. In Stop-And-Wait protocol protocol, the sender sends out one frame and waits for acknowledgment before sending next frame.
4. In Go back n protocol if there is one frame k missing, the receiver simply discards all subsequent frames k+1, k+2,..., sending no acknowledgments. So the sender will retransmit frames from k onwards.
5. In Selective Repeat protocol the sender re-send only the ones that are actually lost or damaged.

7. Exercise questions

1. Explain Selective repeat protocol.
2. What is piggybacking?

Lecture # 10

1. Synopsis:

LAN Protocols: Static & Dynamic channel allocation in LAN's and WAN's,

Multiple access protocols – ALOHA – Pure ALOHA – Slotted ALOHA

2. Target: At the completion of this lecture you should be able to answer questions like

1. Explain Medium Access Sublayer
2. Explain ALOHA systems
3. Compare Pure ALOHA and Slotted ALOHA

3. Introduction

In lecture 6 we discussed that the data link layer consists of Media Access Control. This component determines who is allowed to access the media at any time.As stated above, in this lecture, we plan to explore the MAC layer, the Static and Dynamic Channel Allocation in LANs and MANs and the Multi Access Protocols like ALOHA system. Before we begin on this topic, a revision of the Frequency Division Multiplexing and Time Division Multiplexing that you have learned in your Data Communications paper may be helpful. Once we finish this aspect, we will proceed towards exposition of items listed in the synopsis. In particular, we will emphasize the following

- Medium Access Sublayer
- Static Channel allocation in LAN and MAN
- Dynamic Channel allocation in LAN and MAN
- Aloha System

4. Revision / Prerequisites

Multiplexing

Multiplexing is a transmission of multiple signals over a single communications or computer channel. The multiplexer brings together several low speed communications lines, transforms them into one high speed channel and reverses the operation at the other end. The multiplexers also perform the concentration functions. Multiplexers are less expensive than concentrators, but many of the earlier versions were not programmable and thus did not have the flexibility of concentrators.

However today's microprocessor equipped multiplexers perform much like concentrators.

There are many applications in which several terminals are connected to a computer. If each terminal is operating at 300 bits per second over a communications line (channel) that can operate at 9600 bits per second, then we see a very inefficient operation. It has been found that the capacity of a channel exceeds that required for a single signal. A channel is an expensive resource. Hence, for its optimal utilization, the channel can be shared in such a way so as to simultaneously transmit multiple signals over it. The method of dividing a physical channel into many logical channels so

that a number of independent signals may be simultaneously transmitted on it is known as *multiplexing.*

The electronic device that performs this task is known as a multiplexer.

A multiplexer takes several data communications lines or signals and converts them into one data communications line or signal at the sending location. For example, as shown in figure there may be 4 terminals connected to a multiplexer. The multiplexer takes the signals from the 4 terminals and converts them into 1 large signal, which can be transmitted over 1 communications line. Then, at the receiving location, a multiplexer takes the 1 large signal and breaks it into the original 4 signals. Without multiplexers, you would have to have 4 separate communications lines.

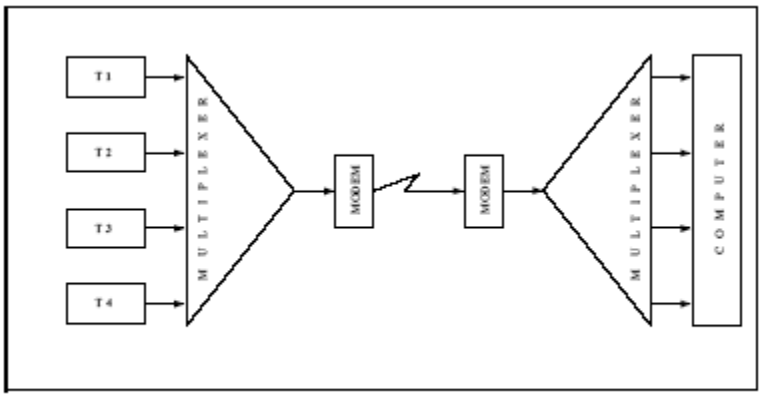


Figure 10.4.1

Thus, with multiplexing it is possible for a single transmission medium to concurrently

transmit data between several transmitters and receivers. There are two basic methods of multiplexing channels. They are frequency division multiplexing and time division multiplexing.

Frequency Division Multiple Access (FDMA)

With Frequency Division Multiple Access (FDMA) the entire available frequency band is divided into bands each of which serves a single user. Every user is therefore equipped with a transmitter for a given, predetermined, frequency band, and a receiver for each band. The main advantage of FDMA is its simplicity--it does not require any coordination or synchronization among the users since each can use its own frequency band without interference. This, however, is also the cause of waste especially when the load is momentarily uneven, since when one user is idle his share of the bandwidth cannot be used by other users. It should be noted that if the users have uneven long term demands, it is possible to divide the frequency range unevenly, i.e., proportional to the demands. FDMA is also not flexible; adding a new user to the network requires equipment modification (in every other user. The best example of FDM is the way we receive various stations in a radio. Each radio station is assigned a frequency range within a bandwidth of radio frequencies. Several radio stations may be transmitting speech signals simultaneously over the physical channel that is "ether" in this case. A radio receiver's antenna receives signals transmitted by all the stations. Finally, the tuning dial in the radio is used to isolate the speech signal of the station tuned. In FDM, the signals to be transmitted must be analog signals. Thus, digital signals must be converted to analog form if they are to use FDM.

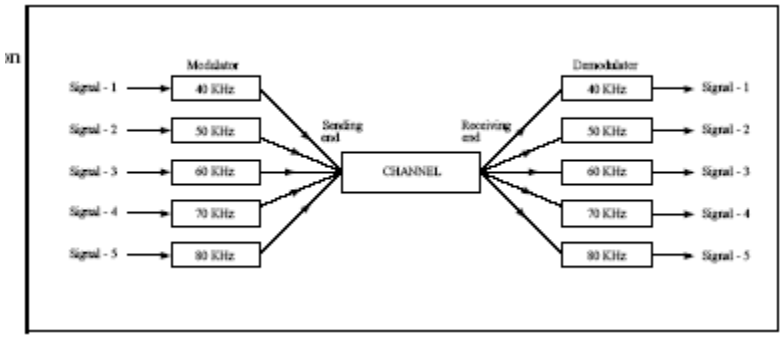


Figure 10.4.2

Time Division Multiple Access (TDMA)

In the time division multiple access (TDMA) scheme the time axis is divided into time

slots, pre assigned to the different users. Every user is allowed to transmit freely during the slot assigned to it, that is, during the assigned slot the entire system resources are devoted to that user. The slot assignments follow a predetermined pattern that repeats itself periodically; each such period is called a *cycle* or a *frame*. Whenever a node has a frame to send, it transmits the frame's bits during its assigned time slot in the revolving TDM frame.

In the most basic TDMA scheme every user has exactly one slot in every frame.

Moreover, communication between computers occurs in short, fast bursts. Each burst would thus need the full channel bandwidth, which is available to a signal in TDM. Besides this, TDM is generally more efficient as more subchannels can be derived because it is upto the network designers to allocate time slots to different channels. It is common to have 32 low speed terminals connected to one high speed line.

Whether or not to use multiplexing usually depends upon economics. The cost of high-speed modems and multiplexers is very high compared to the cost of low-speed modems. However, if line costs are high due to long distances, then multiplexing is cost effective. One serious disadvantage with multiplexing relates to transmission line failure. If the line goes out, everything is dead. With individual lines only one terminal is likely to be lost.

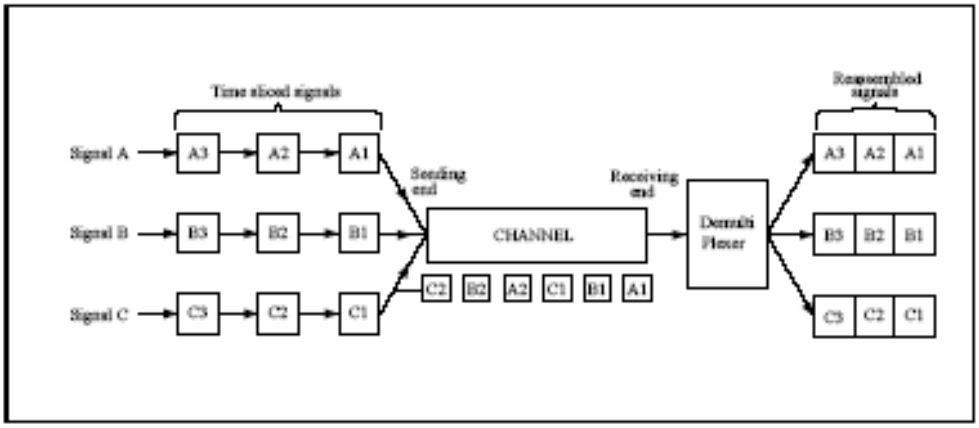


Figure 10.4.3.

Problems with FDMA & TDMA

Static conflict-free protocols such as the FDMA and TDMA protocols do not utilize the shared channel very efficiently, especially when the system is lightly loaded or when the

loads of different users are asymmetric. The static and fixed assignment in these protocols, cause the channel (or part of it) to be idle even though some users have data to transmit. Dynamic channel allocation protocols are designed to overcome this drawback.

5.1. The Medium Access Sub layer

There are two types of network links: point-to-point links, and broadcast links. A point-to-point link consists of a single sender on one end of the link, and a single receiver at the other end of the link. Second type of link, a broadcast link, can have multiple sending and receiving nodes all connected to the same, single, shared broadcast channel.

Traditional networks make use of point-to-point channels, that is, channels that are dedicated to a (ordered) pair of users. These channels, beyond being very economical, are advantageous due to their noninterference feature namely, that transmission between a pair of nodes has no effect on the transmission between another pair of nodes even if these two pairs have a common node. Point-to-point channels, however, require the topology to be fixed, mostly determined at network design time. Subsequent topological changes are quite hard (and costly) to implement.

When point-to-point channels are not economical, not available, or when dynamic topologies are required broadcast channels can be used. Informally stated, a broadcast channel is one in which more than a single receiver can potentially receive every transmitted message.

Broadcast channels appear naturally in radio, satellite, and some local area networks.

This basic property has its advantages and disadvantages. If, indeed, a message is destined to a large number of destinations then a broadcast channel is clearly superior. However, in a typical case a message is destined to a single or a very small number of destinations and wasteful processing results in all those switches for whom the message is not intended. The transmissions over a broadcast channel interfere, in the sense that one transmission coinciding in time with another may cause none of them to be received.

To make a transmission successful interference must be avoided or at least controlled.

The channel then becomes the shared resource whose allocation is critical for proper operation of the network. This is done by ***Multiple Access Protocols***. These protocols are nothing but channel allocation schemes that posses desirable performance characteristics. In terms of known networking models, such as the OSI reference model, these protocols

reside mostly within a special layer called the ***Medium Access Control (MAC)* l**ayer. The *MAC* layer is between the *Data Link Control (DLC)* layer and the Physical Layer.

5.2. Static Channel Allocation in LANs and MANs

The different methods for static channel allocation are

- Frequency Division Multiplexing
- Time Division Multiplexing

5.3. Dynamic Channel Allocation in LANs and MANs

Dynamic allocation strategies, the channel allocation changes with time and is based on current (and possibly changing) demands of the various users.

5.4. Multi access Protocols

Protocols that solve the resolution problem dynamically are called Multiple Access (Multi access) Protocols.

Different types of multi-access protocols:

- **Contention protocols** resolve a collision after it occurs. These protocols execute a collision resolution protocol after each collision
- **Collision-free protocols** ensure that a collision can never occur.

5.5. Contention Protocols

5.5.1. ALOHA

Aloha, also called the Aloha method refers to a simple communications scheme in which each source (transmitter) in a network sends data whenever there is a frame to send. If the frame successfully reaches the destination (receiver), the next frame is sent. If the frame fails to be received at the destination, it is sent again. This protocol was originally developed at the University of Hawaii for use with satellite communication systems in the Pacific.

In a wireless broadcast system or a half-duplex two-way link, Aloha works perfectly. But as networks become more complex, for example in an Ethernet system involving multiple sources and destinations in which data travels many paths at once, trouble occurs because data frames collide (conflict). The heavier the communications volume, the worse the collision problems become. The result is degradation of system efficiency, because when two frames collide, the data contained in both frames is lost.

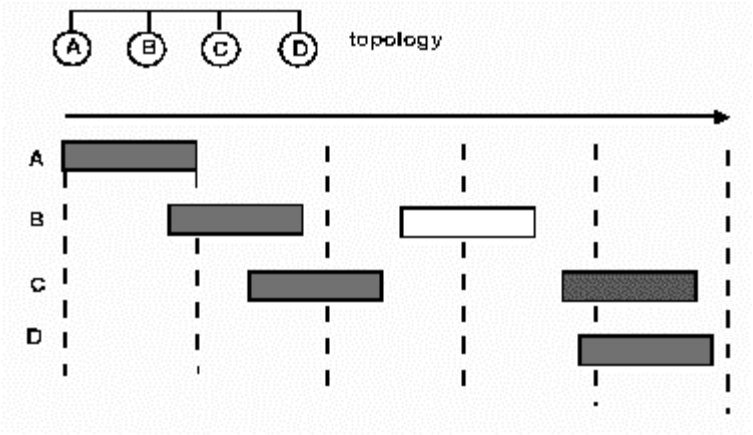


Figure 10.5.1

5.5.2. Pure ALOHA

With Pure Aloha, stations are allowed access to the channel whenever they have data to transmit. Because the threat of data collision exists, each station must either monitor its transmission on the rebroadcast or await an acknowledgment from the destination station. By comparing the transmitted packet with the received packet or by the lack of an acknowledgement, the transmitting station can determine the success of the transmitted packet. If the transmission was unsuccessful it is resent after a random amount of time to reduce the probability of re-collision.

- Whenever a station has data, it transmits
- Sender finds out whether transmission was successful or experienced a collision by listening to the broadcast from the central node
- Sender retransmits after some random time if there is a collision

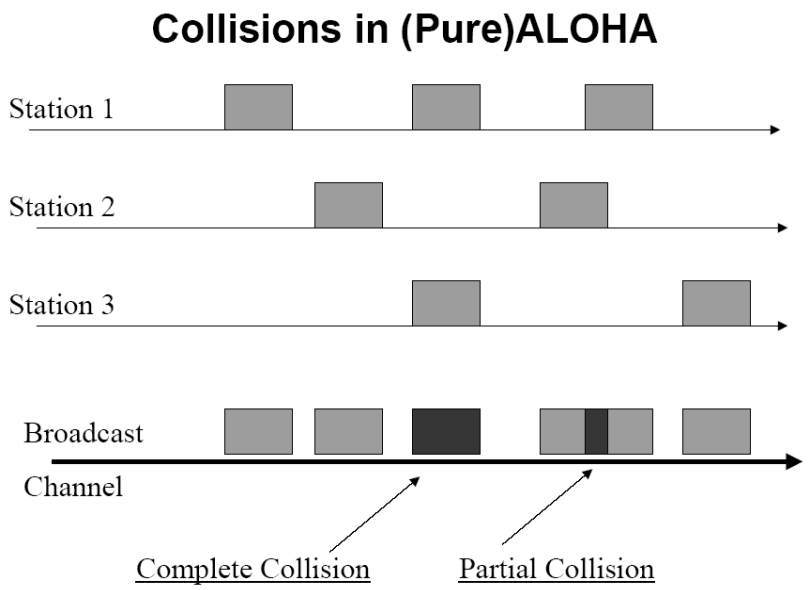


Figure 10.5.2

Advantages:

- Superior to fixed assignment when there are a 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.

Performance of Pure ALOHA

Let S –Throughput: Expected number of successful transmissions per time unit

Normalization: Frame transmission time is 1

Maximum throughput is 1

G -Offered Load: Expected number of transmission and retransmission attempts (from all users) per time unit

- Frames have fixed length of one time unit normalized
- Infinite user population

Prob [k frames generated in 1 frame time] = $(G^k e^{-G})/k!$

Prob [frame suffers no collision]=

= Prob [no other frame is generated during the vulnerable period for this frame]

= Prob [no frame is generated during a 2-frame period]

$$\frac{(2G)^0}{0!} \cdot e^{-2G}$$

Throughput in ALOHA

$$S = G \times e^{-2G}$$

Throughput will be maximum for G=0.5. So Effiency of Pure ALOHA is 18%.

5.5.3. Slotted ALOHA

- Aloha with an additional constraint
- Time is divided into discrete time intervals (=slot)
- Slot size equals fixed packet transmission time
- A station can transmit only at the beginning of a slot.
- When Packet ready for transmission, *wait* until start of *next* slot
- Packets overlap completely or not at all

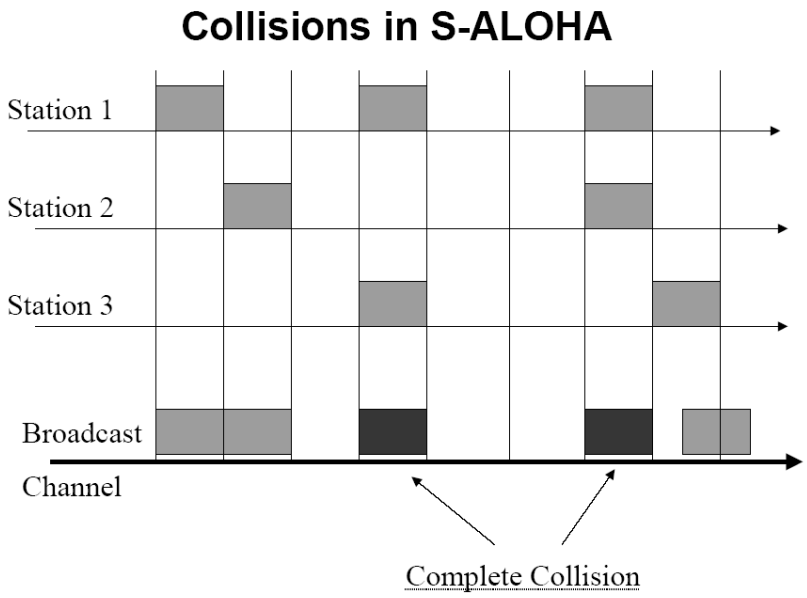


Figure 10.5.3

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

Performance of Slotted ALOHA

Prob [frame suffers no collision] = Prob [no other frame is generated during the vulnerable period for this frame]

= Prob [no frame is generated during a 1-frame period]

$$\frac{(G)^0}{0!} \cdot e^{-G}$$

Throughput in S-ALOHA:

$$S = G \times e^{-G}$$

Throughput will be maximum for G=1.0. So Effiency of Slotted ALOHA is 36%.

5.5.4. Comparison between Pure ALOHA & Slotted ALOHA.

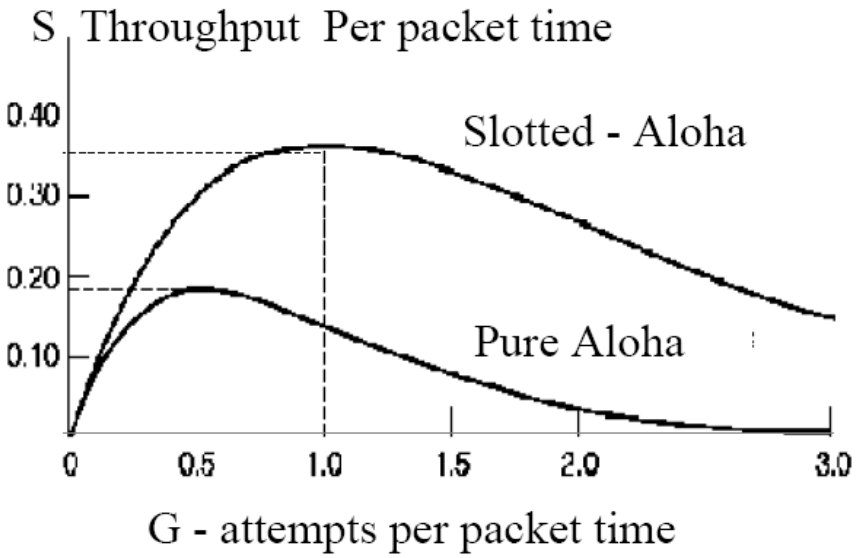


Figure 10.5.4

University Questions

1. How is collision avoidance achieved in Slotted ALOHA?
2. How is Finite Population ALOHA different from pure ALOHA?

3. Draw the curve relating throughput with offered traffic for ALOHA system. What does the curve convey?
4. Explain the improvements obtained in slotted ALOHA with suitable diagrams.

6. Summary

1. Multiple Access Protocols are channel allocation schemes that posses desirable performance characteristics.
2. Multiple Access Protocols reside within a special layer called the **Medium Access Control (MAC)** layer.
3. MAC layer is between the Data Link Control (DLC) layer and the Physical Layer.
4. Aloha method refers to a simple communications scheme in which each source in a network sends data whenever there is a frame to send.
5. In Pure Aloha, stations are allowed access to the channel whenever they have data to transmit.

7. Exercise questions

1. Describe ALOHA system
2. Describe MAC sub layer.
3. Compare Pure ALOHA and Slotted ALOHA

Lecture # 11

1. Synopsis

Carrier Sense Multiple Access protocols – persistent and non-persistent CSMA – CSMA with collision detection – IEEE 802.3 standards for LAN

2. Target

At the completion of this lecture you should be able to answer questions like

1. Explain Carrier Sense Multiple Access protocol
2. Explain the different CSMA protocols
3. Explain CSMA with collision detection
4. Explain IEEE 802.3 (Ethernet) standards for LAN

3. Introduction

In lecture 10 we discussed the ALOHA method. But the best channel utilization that can be achieved with slotted ALOHA is 1/e. In this lecture we will be discussing some protocols for improving performance. In particular, we will emphasize the following

- Non-Persistent CSMA
- 1-Persistent CSMA
- p-Persistent CSMA
- CSMA with collision detection (CSMA/CD)
- IEEE 802.3 Standard for LAN

4. Revision / Prerequisites

Please refer lecture 10 for the contention protocols.

5.1. Carrier Sense Multiple Access Protocol (CSMA)

In both slotted and pure ALOHA, a node's decision to transmit is made independently of the activity of the other nodes attached to the broadcast channel. Ethernet uses a refinement of ALOHA, known as Carrier Sense Multiple Access (CSMA), which improves performance when there is a higher medium utilization. When a Station has data to transmit, the station first listens to the cable (using a transceiver) to see if a carrier (signal) is being transmitted by another node. This may be achieved by monitoring whether a current is flowing in the cable (each bit corresponds to 18-20 milliamps (mA)).

If a sender does not receive an acknowledgment after some period, it assumes that a collision has occurred. After a collision a station backs off for a certain (random) time and retransmits.

There are a number of variations of CSMA protocols:

- Non-Persistent CSMA
- 1-Persistent CSMA
- p-Persistent CSMA

5.2. Non-Persistent CSMA Protocol:

1. If the medium is idle, transmit immediately
2. If the medium is busy, wait a random amount of time and repeat Step 1
    - Random back-off reduces probability of collisions
    - Wasted idle time if the back-off time is too long
    - May result in long access delays
    - Better Channel utilization than 1 persistent.

5.3. 1-persistent CSMA Protocol:

When a station has to send data it first listens to the channel to see if anyone else is transmitting. 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.

1. Check the medium.
2. if idle, send
3. else, wait until idle (continuously sense), then send
4. if collision, wait random amount of time, goto 1

There will always be a collision if two stations want to transmit simultaneously.

1-persistent CSMA transmits with probability 1 when it finds channel idle.

5.4. p-Persistent CSMA Protocol:

This protocol applies to slotted channels. When a station becomes ready to send it sends the channel. If it is idle it transmits with a probability p.

1. If the medium is idle, transmit with probability p, and delay for one time unit with probability (1 - p) (time unit = length of propagation delay).
2. If the medium is busy, continue to listen until medium becomes idle, then go to Step 1

3. If transmission is delayed by one time unit, continue with Step 1
4. If p=1, station can transmit the frame whenever the channel is idle. If p=0 station always wait.

If a collision has occurred, the channel is unstable until colliding packets have been fully transmitted. CSMA/CD overcome this difficulty.

5.5. Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

While transmitting, the sender is listening to medium for collisions.

Sender stops if collision has occurred.

- Use one of the CDMA persistence algorithm (non-persistent, 1-persistent, p-persistent) for transmission.
- If a collision is detected during transmission, cease transmission and transmit a jam signal to notify other stations of collision.
- After sending the jam signal, back off for a random amount of time, then start to transmit again.

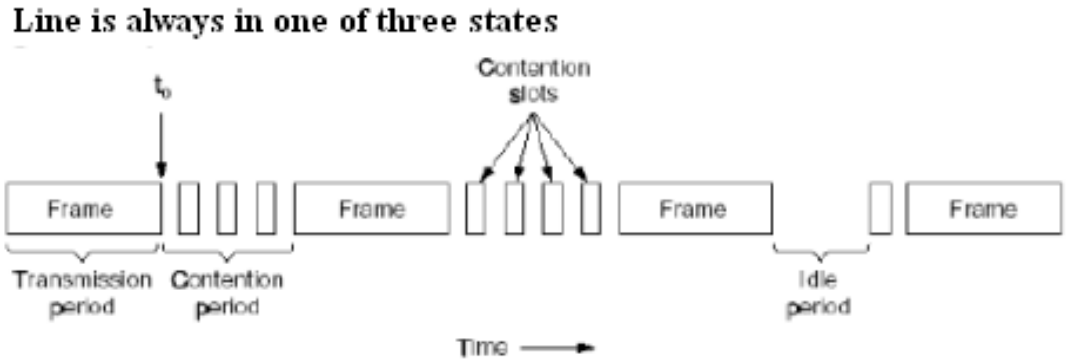


Figure 12.5.1 CSMA/CD can be in one of three states: contention, transmission, or idle

5.6. IEEE 802.3 standard for LAN (Ethernet)

Ethernet is a frame-based computer networking technology for local area networks (LANs). The name comes from the physical concept of ether. It defines wiring and signaling for the physical layer, and frame formats and protocols for the media access control (MAC)/data link layer of the OSI model. Ethernet is mostly standardized as IEEEs 802.3. Ethernet was originally developed as one of the many pioneering projects at Xerox PARC.

Ethernet uses 1-persistent carrier sense multiple access with collision detection (CSMA/CD) for channel allocation. Originally developed in the 1960s for the

ALOHAnet in Hawaii using radio, the scheme is relatively simple compared to token ring or master controlled networks. When one station wants to send some information, it obeys the following algorithm:

Step 1: Start - If the wire is idle, starts transmitting, else go to step 4

Step 2: Transmitting - If detecting a collision, continue transmitting until the minimum packet time is reached (to ensure that all other transmitters and receivers detect the collision) then go to step 4.

Step 3: End successful transmission - Report success to higher network layers; exit transmit mode.

Step 4: Wire is busy - Wait until wire becomes idle

Step 5: Wire just became idle - Wait a random time, then go to step 1, unless maximum number of transmission attempts has been exceeded

Step 6: Maximum number of transmission attempt exceeded - Report failure to higher network layers; exit transmit mode.

University Questions

1. Compare Non Persistent, 1-Persistent and p-Persistent CSMA.
2. What does part 802.3 of IEEE 802 standard describe?
3. Explain CSMA.
4. Explain different multiple access protocols.

6. Summary

1. Carrier Sense Protocols are protocols in which stations listen for a carrier and act accordingly.
2. In 1-persistent CSMA the station transmits with a probability of 1 whenever it finds the channel idle.
3. The p-persistent protocol applies to slotted channels.
4. CSMA/CD is used on LANs in the MAC layer.
5. Ethernet is a frame-based computer networking technology for local area networks

7. Exercise questions

1. Name some of the multiple access protocols
2. Explain various contention protocols.

MODULE I
Lecture # 1

1. Synopsis

Introduction

2. Target

At the completion of this lecture you should be able to answer questions like

1. What is a computer network?
2. What is the need for a network?
3. What are the applications of computer network?
4. What is the need for a network standard?

3. Introduction

This being the first class of this course you will get a very basic idea of computer networks. Before we begin on this topic, a revision of the concepts developed earlier (subjects taught earlier etc or previous lectures) may be helpful. Once we finish this aspect, we will proceed towards exposition of items listed in the synopsis. In particular, we will emphasize the following

1. What is a computer network?
2. What is the need for a network?
3. What are the applications of computer network?
4. What is the need for a network standard?

4. Revision / Prerequisites

In your Data Communication paper you have come across multiplexing, the different types of multiplexing like FDM, STDM etc. You have also learned about the different techniques for transmitting digital data, the different transmission mode and different kinds of switching like packet switching and circuit switching. The Data Communication course RT506 is demanded as a prerequisite for this course.

5.1. What is a computer network?

Computer network is a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange information. It is not necessary that the connection should be via a copper

wire. Fiber optics, microwaves, infrared, and communication satellites can also be used to establish the connection.

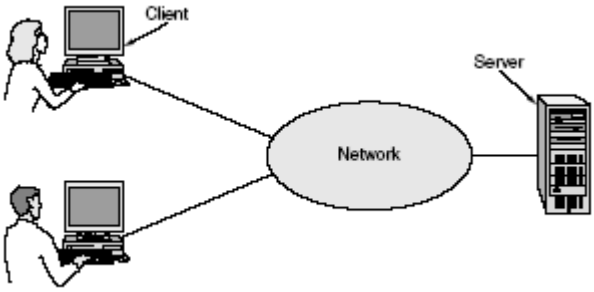
5.2. Why Networks?

- Distribute computation among nodes
- Coordination between processes running on different nodes
- Remote I/O Devices
- Remote Data/File Access
- Personal communications (e-mail, chat, A/V)
- World Wide Web

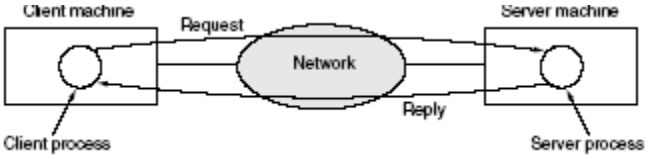
5.3. Applications of networks

Business applications

Resource sharing by using client server model



In this model the data are stored on powerful computers called **servers**. Users have simpler machines, called **clients**, on their desks, with which they access remote data.



A computer network can provide a powerful **communication medium** among employees. Virtually every company that has two or more computers now has **e-mail** (**electronic mail**), which employees generally use for a great deal of daily communication.

Home Applications

Access to remote information.

Person-to-person communication.

Interactive entertainment.

Electronic commerce.

5.4. Why Do We Need A Standard?

- Many types of connection media: telephone lines, optical fibers, cables, radios, etc.
- Many different types of machines and operating systems
- Many different network applications

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

University Questions

1. Compare the advantages and disadvantages of Analog transmission with its digital counterpart.

6. Summary

1. Computer network is a collection of autonomous computers interconnected by a single technology.
2. Since many types of connection media can be used for communication between many different types of machines that may have different operating systems and may support different network applications a standard is essential for the network of computers.

7. Exercise questions

1. What are the applications of computer network?
2. What are two reasons for using layered protocols?

8. Programming / Computational assignments

Lecture # 2

1. **Synopsis**:

   ISO-OSI Reference Model

2. **Target:** At the completion of this lecture you should be able to answer questions like
   1. What are the layers in the OSI model?
   2. What is the function of the physical layer?
   3. What is the function of the data link layer?
   4. What is the function of the network layer?
   5. What is the function of the transport layer?
   6. What is the function of the session layer?
   7. What is the function of the presentation layer?
   8. What is the function of the application layer?

3. Introduction

In this lecture, we will emphasize the following

a. ISO-OSI Model

b. The functions of the seven layers in the OSI model

4. Revision / Prerequisites

In lecture 1 we have discussed the need for a standard for establishing a network connection. In this lecture we will be discussing an important reference model that was developed by the **International Standards Organization (ISO).** This model is called the **ISO OSI (Open Systems Interconnection)** Reference model and was the first step towards the international standardization of the protocols used in the various layers.

5.1. ISO - OSI Model

One of the most important concepts in networking is the **open-systems interconnection (OSI)** reference model. It serves as a framework within which communication protocol standards are developed.

In 1977, **International Standards Organization (ISO)** began an ambitious project to develop a single international standard set of communications protocols. By 1984,

ISO had defined an overall model of computer communications called the Reference Model for Open Systems Interconnection, or OSI Model.

ISO also developed a comprehensive set of standards for the various layers of the OSI model. The standards making up the OSI architecture were not widely implemented in commercial products for computer networking. However, the OSI model is still important. The concepts and terminology associated with the OSI model have become widely accepted as a basis for discussing and describing network architectures. The OSI model is often used in categorizing the various communications protocols that are in common use today and also used for comparing one network architecture to another.

- Model: It means that it is only theory. In fact the OSI model is not yet fully implemented in real networks.
- Open System: It can communicate with any other system that follows the specified standards, formats, and semantics.
- Protocols give rules that specify how the communication parties may communicate.
- Supports two general types of protocols. They are
    - Connection-Oriented:
        - Sender and receiver first establish a connection, possibly negotiate on a protocol (virtual circuit)
        - Transmit the stream of data
        - Release the connection when done
        - e.g. Telephone connection
    - Connectionless
        - No advance setup is needed.
        - Transmit the message (datagram) when sender is ready.

e.g. surface mail

5.2. The Seven Layers

- The OSI model defines the seven functional layers. Each layer performs a different set of functions, and the intent was to make each layer as independent as possible from all others.

- Each layer deals with a specific aspect of communication.
- Each layer provides an interface to the layer above. The set of operations define the service provided by that layer.
- As a message sent by the top layer is passed on to the next lower layer until the most bottom layer.
- At each level a header may be appended to the message. Some layers add both a header and a trailer.
- The lowest layer transmits the message over the network to the receiving machine. It communicates with the most bottom layer of the receiver.
- Each layer then strips the header (trailer), handles the message using the protocol provided by the layer and passes it on to the next higher layer. Finally to the highest layer in the receiver.

Layer 7	Application Layer
Layer 6	Presentation Layer
Layer 5	Session Layer
Layer 4	Transport Layer
Layer 3	Network Layer
Layer 2	Data-Link Layer
Layer 1	Physical Layer

Layer-1: Physical Layer

The physical layer defines the physical characteristics of the interface, such as mechanical components and connectors, electrical aspects such as voltage levels representing binary values, and functional aspects such as setting up, maintaining, and taking down the physical link. Well-known physical layer interfaces for local area networks (LANs) include Ethernet, Token-Ring, and Fiber Distributed Data Interface (FDDI). Physical Layer

- Concerned with the transmission of *bits*.
- How many volts for 0, how many for 1?
- Number of bits of second to be transmitted.
- Two way or one-way transmission
- Standardized protocol dealing with electrical, mechanical and signaling interfaces.

- Many standards have been developed, e.g. RS-232 (for serial communication lines).
- Example: X.21

Layer-2: Data-Link Layer

The data link layer defines the rules for sending and receiving information across the physical connection between two systems. This layer encodes and frames data for transmission, in addition to providing error detection and control. Because the data link layer can provide error control, higher layers may not need to handle such services. However, when reliable media is used, there is a performance advantage by not handling error control in this layer, but in higher layers. Bridges operate at this layer in the protocol stack.

- Handles errors in the physical layer.
- Groups bits into *frames* and ensures their correct delivery.
- Adds some bits at the beginning and end of each frame plus the checksum.
- Receiver verifies the checksum.
- If the checksum is not correct, it asks for retransmission. (Send a control message).
- Consists of two sub layers:
    - **Logical Link Control (LLC)** defines how data is transferred over the cable and provides data link service to the higher layers.
    - **Medium Access Control (MAC)** defines who can use the network when multiple computers are trying to access it simultaneously (i.e. Token passing, Ethernet [CSMA/CD]).

Layer-3: Network Layer

The network layer defines protocols for opening and maintaining a path on the network between systems. It is concerned with data transmission and switching procedures, and hides such procedures from upper layers. Routers operate at the network layer. The network layer can look at packet addresses to determine routing methods. If a packet is addressed to a workstation on the local network, it is sent directly there. If it is addressed to a network on another segment, the packet is sent to a routing device, which forwards it on the network.

- Concerned with the transmission of *packets*.
- Choose the best path to send a packet (*routing).*
- It may be complex in a large network (e.g. Internet).
- Shortest (distance) route vs. route with least delay.
- Static (long term average) vs. dynamic (current load) routing.
- Two protocols are most widely used.
- **X.25**
  - Connection Oriented
  - Public networks, telephone, European PTT
  - Send a call request at the outset to the destination
  - If destination accepts the connection, it sends an connection identifier
- **IP (Internet Protocol)**
  - Connectionless
  - Part of Internet protocol suite.
  - An IP **packet** can be sent without a connection being established.
  - Each packet is routed to its destination independently.

Layer-4: Transport Layer

The transport layer provides a high level of control for moving information between systems, including more sophisticated error handling, prioritization, and security features. The transport layer provides quality service and accurate delivery by providing connection-oriented services between two end systems. It controls the sequence of packets, regulates traffic flow, and recognizes duplicate packets. The transport layer assigns pocketsize information a traffic number that is checked at the destination. If data is missing from the packet, the transport layer protocol at the receiving end arranges with the transport layer of the sending system to have packets re-transmitted. This layer ensures that all data is received and in the proper order.

- Network layer does not deal with lost messages.
- Transport layer ensures reliable service.
- Breaks the message (from sessions layer) into smaller packets, assigns sequence number and sends them.
- Reliable transport connections are built on top of X.25 or IP.

- In case IP, lost packets arriving out of order must be reordered.
- TCP: (Transport Control Protocol) Internet transport protocol.
- TCP/IP Widely used for network/transport layer (UNIX).
- UDP (Universal Datagram Protocol) : Internet connectionless transport layer protocol.
- Application programs that do not need connection-oriented protocol generally use UDP.

Layer-5: Session Layer

The session layer coordinates the exchange of information between systems by using conversational techniques, or dialogues. Dialogues are not always required, but some applications may require a way of knowing where to restart the transmission of data if a connection is temporarily lost, or may require a periodic dialog to indicate the end of one data set and the start of a new one.

- Enhanced version of transport layer.
- Dialog control, synchronization facilities.
- Rarely supported (Internet suite does not).

Layer-6: Presentation Layer

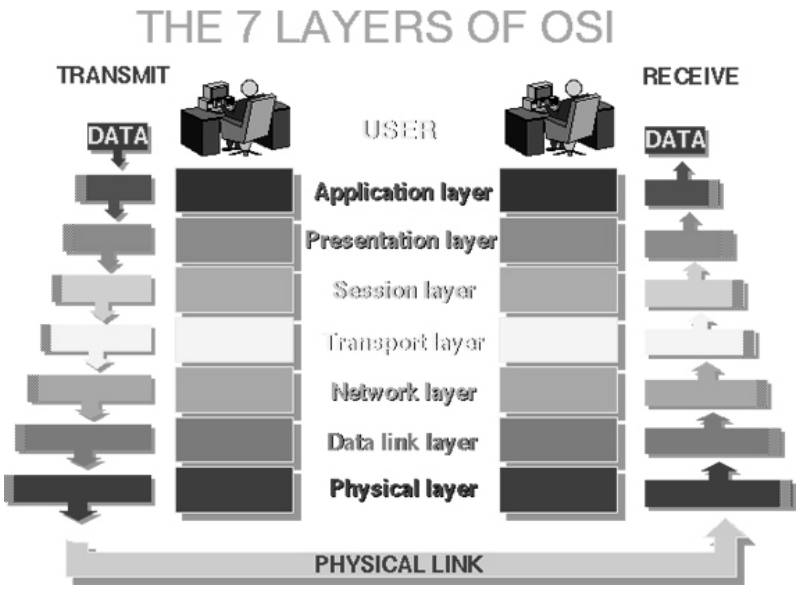
Protocols at the presentation layer are part of the operating system and application the user runs in a workstation. Information is formatted for display or printing in this layer. Codes within the data, such as tabs or special graphics sequences, are interpreted. Data encryption and the translation of other character sets are also handled in this layer.

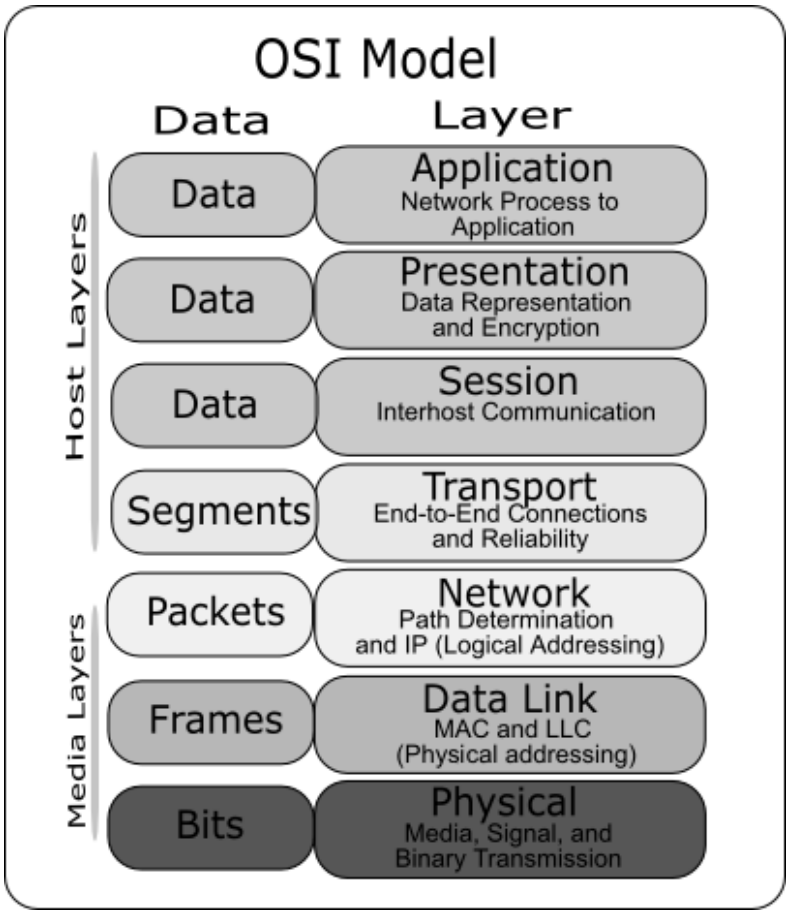
- Concerned with the semantics of the bits.
- Define records and fields in them.
- Sender can tell the receiver of the format.
- Makes machines with different internal representations to communicate.
- If implemented, the best layer for cryptography.

Layer-7: Application Layer

Applications access the underlying network services using defined procedures in this layer. The application layer is used to define a range of applications that handle file transfers, terminal sessions, network management, and message exchange.

- Collection of miscellaneous protocols for high level applications
- Electronic mail, file transfer, connecting remote terminals, etc.
- E.g. SMTP, FTP, Telnet, HTTP, etc.





5.3. Performance Parameters

Latency: Time required to transfer an empty message between relevant computers. Sum total of

1. Delay introduced by the sender software.
2. Delay introduced by the receiver software.
3. Delay in accessing the network.
4. Delay introduced by the network.

Data transfer rate: is the speed at which data can be transferred between sender and receiver in a network, once transmission has begun. (bit/sec)

Message transfer time = latency + (length of message) / (Data transfer rate).

Bandwidth: is the total volume of traffic that can be transferred across the network.

Max. Data rate (bit/sec) = carrier BW $\cdot \log_2 (1 + (\text{signal/noise}))$

e.g. phone line BW = 3 kHz, S/N = 30 dB = 1000 Max. Data rate = 30 kbit/sec

University Questions

1. Draw the block diagram of an OSI reference model and briefly explain the functions of each. (Nov 2005)
2. What are the advantages of layered architecture?

6. Summary

1. **Open Systems Interconnection (OSI)** reference model serves as a framework within which communication protocol standards are developed.
2. Open System can communicate with any other system that follows the specified standards, formats, and semantics.
3. Protocols are rules that specify how the communication parties may communicate.
4. The physical layer defines the physical characteristics of the interface, such as mechanical components and connectors, electrical aspects such as voltage levels representing binary values, and functional aspects such as setting up, maintaining, and taking down the physical link.
5. Examples of physical layer interfaces for local area networks (LANs) include Ethernet, Token-Ring, and Fiber Distributed Data Interface (FDDI).
6. The data link layer encodes and frames data for transmission and also provides error detection and control.
7. Bridges operate at the data link layer in the protocol stack.
8. Data link layer consists of two sub layers, **Logical Link Control (LLC)** and **Medium Access Control (MAC)**
9. **Logical Link Control (LLC)** defines how data is transferred over the cable and provides data link service to the higher layers.
10. **Medium Access Control (MAC)** defines who can use the network when multiple computers are trying to access it simultaneously (i.e. Token passing, Ethernet [CSMA/CD]).
11. The network layer defines protocols for opening and maintaining a path on the network between systems.

12. The transport layer provides a high level of control for moving information between systems, including more sophisticated error handling, prioritization, and security features.
13. The transport layer provides quality service and accurate delivery by providing connection-oriented services between two end systems.
14. The session layer coordinates the exchange of information between systems by using conversational techniques, or dialogues.
15. Data encryption and the translation of other character sets are also handled in the presentation layer.
16. Applications access the underlying network services using defined procedures in the application layer.

7. Exercise questions

1. Describe OSI reference model.
2. Which of the OSI layers handles each of the following:
    a. Breaking the transmitted bit stream into frames
    b. Determining which route through the subnet to use
3. What is the principal difference between connectionless communication and connection-oriented communication?

8. Programming / Computational assignments

1. Calculate the latency (from first bit sent to the last bit received) for a 10 Mbps Ethernet with a single store-and –forward switch in the path, and a packet size of 5000 bits. Assume that each link introduces a propagation delay of 10 μs and that the switch begins retransmitting immediately after it has finished receiving the packet.

Lecture # 3

1. Synopsis

TCP/IP Reference Model

2. Target

At the completion of this lecture you should be able to answer questions like

1. Which are the different layers in the TCP/IP Reference model?
2. Which are the application level protocols in the TCP/IP?
3. What is Domain Name Server?
4. What is the function of FTP?
5. What is the function of SMTP?
6. What is Hyper Text Transfer Protocol?
7. What is the function of the transport layer in the TCP/IP?
8. Which are the transport level protocols in TCP/IP?
9. What is the function of the Internet layer in TCP/IP?
10. Why is IP referred to as unreliable protocol?
11. What is the function of the Network Access layer?
12. What are the similarities of OSI and TCP/IP model?
13. How do the OSI and TCP/IP models differ?

3. Introduction

In lecture 1 we discussed about the need for a standard for establishing a network of computers. Also in lecture 2 we discussed on such reference model, the ISO-OSI reference model. As stated above, in this lecture, we plan to explore another important reference model, the TCP/IP reference model. In this lecture we will emphasize the following

a) TCP/IP Reference Model
b) Comparison of the OSI and TCP/IP models

4. Revision / Prerequisites

Before beginning this lecture you should go through the lecture 2, which explains the OSI reference model. This will help you in comparing the two reference models.

5.1. TCP/IP Reference Model

The TCP/IP reference model is the network model used in the current Internet architecture. It has its origins back in the 1960's with the grandfather of the Internet, the ARPANET. The TCP/IP model does not exactly match the OSI model. There is no universal agreement regarding how to describe TCP/IP with a layered model but it is generally agreed that there are fewer levels than the seven layers of the OSI model.

Layers of the TCP/IP model are defined as follows:

Application layer

Like OSI Model, it contains all the higher-level protocols. In TCP/IP the Application Layer also includes the OSI Presentation Layer and Session Layer. This includes all of the processes that involve user interaction. The application determines the presentation of the data and controls the session. In TCP/IP the terms **socket** and **port** are used to describe the path over which applications communicate. There are numerous application level protocols in TCP/IP, including Simple Mail Transfer Protocol (SMTP) and Post Office Protocol (POP) used for e-mail, Hyper Text Transfer Protocol (HTTP) used for the World-Wide-Web, and File Transfer Protocol (FTP). Most application level protocols are associated with one or more port number.

The most widely known and implemented Application Layer protocols are:

Network Terminal Protocol (Telnet): Provides text communication for remote login and communication across the network

File Transfer Protocol (FTP): Used to download and upload files across the network

Simple Mail Transfer Protocol (SMTP): Delivers electronic mail messages across the network

Post Office Protocol, Version 3 (POP-3): Allows users to pick up e-mail across the network from a central server

Domain Name Service (DNS): Maps IP addresses to Internet domain names

Hyper Text Transfer Protocol (HTTP): The protocol used by the World-Wide-Web to exchange text, pictures, sounds, and other multi-media information via a graphical user interface (GUI)

Routing Information Protocol (RIP): Used by network devices to exchange routing information

Some protocols are used directly by users as applications, such as FTP and Telnet. Other protocols are directly behind applications, such as SMTP and HTTP. Others, such as RIP and DNS, happen indirectly or are used by the programs and operating system routines. A system administrator must be aware of all of the protocols and how they interact with each other and the lower TCP/IP layers.

Most of the application level protocols in TCP/IP are implemented as 7-bit ASCII stream conversations. This means that the communication that takes place across the network is formatted using upper and lower case alphabetic characters, numeric digits, and standard punctuation. This was partly done because, historically, as a packet of information passed across the Internet there was no predicting what type of system it would pass through. 7-bit ASCII was chosen as a coding scheme that almost any computer or piece of equipment would be able to handle. Any information that is not actually 7-bit ASCII is first converted to ASCII, then transmitted across the network, and converted back to its original form at the destination.

To really understand the rest of the reasons behind why application protocols are simple text conversations it is important to understand a little of the history of TCP/IP, the Internet, and UNIX.

The Internet and TCP/IP originally grew up on networks of UNIX computer systems. On a classic UNIX system there was one central computer system with a number of "Dumb", text display terminals connected to it. A user on one of the terminals typed in text commands to run programs, which displayed text as output.

One of the first real network applications allowed a user on one UNIX computer to communicate across a connection to a remote UNIX computer and run programs as if directly connected to the remote system. This type of connection used Network Terminal Protocol, which is now commonly called Telnet. The local system that the user is connected to is called the client, and the remote computer is called the server. This was one of the original Client-Server applications.

When a user wanted to communicate from their computer to a remote system, they would Telnet to the remote and run a program on the remote system. For example, if they wanted to send mail to someone on a remote computer, they could Telnet to the remote system, run the mail receiving program, and type in a message to a particular user. If they

wanted to send a file from computer to computer, they would Telnet to the remote system, start a Receive program running, then drop back to the local computer and start a Send program to transmit the file.

Eventually there were certain programs that were left running all the time on the server computers, E-Mail, File Transfer, and other common applications, so that a remote user did not have to log onto the remote system and start them each time they wanted to use them. These programs were left running in the background, using only a small amount of resources, while they waited to be contacted. These are what is known in the UNIX world as Daemons, and in the Windows world as Services. Just so that people could find the correct Daemon to talk to, each common or well known program was given a specific Port on the computer that it listened to while it was waiting for someone to contact it.

Programmers, being the basically lazy people they are, grew tired of all the Telnetting from system to system. They began writing programs to do the Telnetting about for them. These programs at first were barely more than a modified Telnet program that followed a script to talk to a remote server. But these client applications grew in complexity and sophistication, becoming much more popular than the manual method of Telnetting around and typing in long text messages. Eventually there were mostly computers talking to computers, but the conversations were still the original text messages developed when it was people doing the conversing.

Now, the above description is not exactly the order of events, or exactly the way it all happened, but it does give a rough idea of how the application protocols ended up the way they are. The important thing to remember is that most application protocols are text conversations that can be manually operated using a Telnet client program.

The text conversation aspect of application level protocols comes in very handy when setting up, testing, and troubleshooting TCP/IP applications. It is usually possible, and often quite easy, to use a simple Telnet client to contact a remote TCP/IP application and send it commands and test data. For example, if you are having trouble sending an E-Mail message to a particular person, it is quite easy to Telnet directly to the recipients mail server and manually compose a message for delivery. Then the responses of the remote system to each step of the procedure can be observed. This usually makes it much easier to identify where and how a problem is occurring.

One downside of the text conversation aspect of TCP/IP application level protocols is that it is fairly easy to eavesdrop on a conversation or to write a program to generate the correct text messages so that one program can impersonate another. There have been many solutions developed to address this problem. Pretty Good Privacy (PGP) is a publicly available application that encrypts text messages before they are passed across unsecured channels, and then un encrypts them at the destination. Secure Hyper Text Transfer Protocol (SHTTP) and the Secure Sockets Layer (SSL) are designed to verify the identities of computer systems at both ends of a World Wide Web conversation and to encrypt information in transit between them. These are just a few of a broad range of available applications that address the security issues.

Secure communications across TCP/IP is a broad topic and is beyond the scope of this technical reference. Usually it is best to approach it on an application by application basis, or as an individual solution such as a Firewall system.

Transport layer

In TCP/IP there are two Transport Layer protocols. The Transmission Control Protocol (TCP) guarantees that information is received as it was sent. The User Datagram Protocol (UDP) performs no end-to-end reliability checks.

Between the Internet layer and Application layer of the TCP/IP architecture model is the Transport Layer. This layer has two primary protocols, the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP).

TCP is a connection based protocol that provides error detection and correction with reliable delivery of data packets. UDP is a connectionless protocol with low overhead.

When writing application software a developer normally chooses TCP or UDP based on whether it is more important to have a reliable connection with bi-directional communication and error management, or if it is more important to develop a low overhead, streamlined application.

Internet Layer

In the OSI Reference Model the Network Layer isolates the upper layer protocols from the details of the underlying network and manages the connections across the network. The Internet Protocol (IP) is normally described as the TCP/IP Network Layer. Because of the Inter-Networking emphasis of TCP/IP this is commonly referred to as the Internet

Layer. All upper and lower layer communications travel through IP as they are passed through the TCP/IP protocol stack.

The Internet Layer of the TCP/IP architecture model resides just above the Network Access Layer and below the Transport Layer. The primary concern of the protocol at this layer is to manage the connections across networks as information is passed from source to destination. The Internet Protocol (IP) is the primary protocol at this layer of the TCP/IP architecture model.

IP is a **connectionless protocol**. This means it does not use a handshake to provide end-to-end control of communications flow. It relies on other layers to provide this function if it is required. IP also relies on other layers to provide error detection and correction. Because of this IP is sometimes referred to as an **unreliable protocol**. This does not mean that IP cannot be relied upon to accurately deliver data across a network; it simply means that IP itself does not perform the error checking and correcting functions.

The functions that IP performs include:

- Defining a datagram and an addressing scheme
- Moving data between transport layer and network access layer protocols
- Routing datagram to remote hosts
- The fragmentation and reassembly of datagram

The only other protocol that is generally described as being at the Internet Layer of the TCP/IP model is the Internet Control Message Protocol (ICMP), a protocol used to communicate control messages between IP systems.

Network Access Layer (Host to Network Layer)

In TCP/IP the Data Link Layer and Physical Layer are normally grouped together. TCP/IP makes use of existing Data Link and Physical Layer standards rather than defining its own. Most RFCs that refer to the Data Link Layer describe how IP utilizes existing data link protocols such as Ethernet, Token Ring, FDDI, HSSI, and ATM. The characteristics of the hardware that carries the communication signal are typically defined by the Physical Layer. This describes attributes such as pin configurations, voltage levels, and cable requirements. Examples of Physical Layer standards are RS-232C, V.35, and IEEE 802.3.

The Network Access layer is the lowest level of the TCP/IP protocol hierarchy. It is often ignored by users as it is well hidden by the mid-level protocols such as IP, TCP, and UDP, and higher level protocols such as SMTP, HTTP, and FTP. Functions performed at the network access layer include encapsulation of IP datagram into frames to be transmitted by the network, and mapping IP addresses to physical hardware addresses.

Much of the work that takes place at the network access layer is handled by software applications and drivers that are unique to individual pieces of hardware. Configuration often consists of simply selecting the appropriate driver for loading, and selecting TCP/IP as the protocol for use. Many computers come with this driver software pre-loaded and configured, or can automatically configure themselves via "plug-and-play" applications.

5.2. Comparison of the OSI and TCP/IP reference models

- Both are based on the concept of a stack of independent protocols.
- Functionality of the layers is similar.
- The layers up through and including transport layer are providing an end-to-end transport service.
- Concepts, Interfaces, Protocols are the three concepts central to the OSI model. Each layer provides some service for the layer above to it. A layer's interface tells the process above it how to access it.
- The TCP/IP model did not originally clearly distinguish between service, interface, and protocol.
- The OSI reference model was devised before the protocols were invented.
- With the TCP/IP, protocol came first, and the model was really the description of the existing protocol.

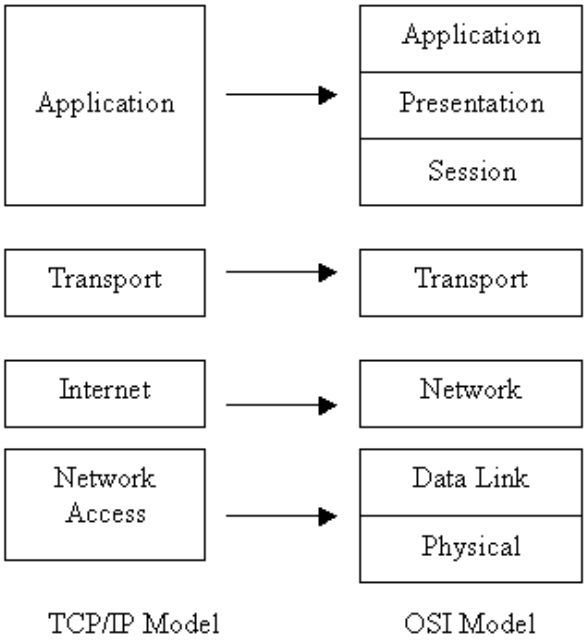


Figure: 3.1 TCP/IP and OSI Models

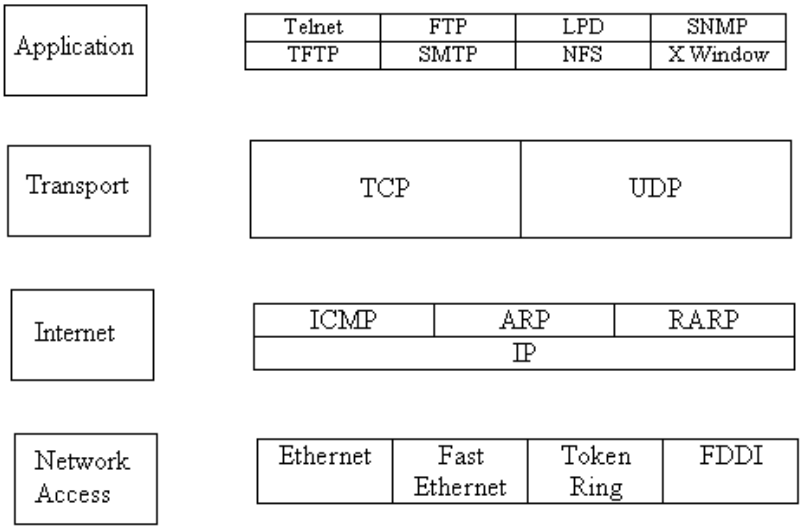


Figure 3.2. TCP/IP Protocol Suite

University Questions

1. Compare OSI and TCP/IP Reference Models.

6. Summary

1. The TCP/IP reference model is the network model used in the current Internet architecture.
2. In TCP/IP the Application Layer also includes the OSI Presentation Layer and Session Layer.
3. The numerous application level protocols in TCP/IP, includes Simple Mail Transfer Protocol (SMTP) and Post Office Protocol (POP) used for e-mail, Hyper Text Transfer Protocol (HTTP) used for the World-Wide-Web, and File Transfer Protocol (FTP).
4. **Network Terminal Protocol (Telnet**) provides text communication for remote login and communication across the network
5. **File Transfer Protocol (FTP)** is used to download and upload files across the network
6. **Simple Mail Transfer Protocol (SMTP)** delivers electronic mail messages across the network
7. **Post Office Protocol, Version 3 (POP-3)** allows users to pick up e-mail across the network from a central server
8. **Domain Name Service (DNS)** is used to map IP addresses to Internet domain names
9. **Hyper Text Transfer Protocol (HTTP)** is the protocol used by the World-Wide-Web to exchange text, pictures, sounds, and other multi-media information via a graphical user interface (GUI)
10. **Routing Information Protocol (RIP)** is used by network devices to exchange routing information
11. In TCP/IP there are two Transport Layer protocols, the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP).
12. TCP is a connection-based protocol that provides error detection and correction with reliable delivery of data packets.
13. UDP is a connectionless protocol with low overhead.
14. The Internet Protocol (IP) is described as the TCP/IP Network Layer.
15. IP is a **connectionless protocol**.
16. IP is sometimes referred to as an **unreliable protocol**.

17. The Network Access layer is the lowest level of the TCP/IP protocol hierarchy.
18. Functions performed at the network access layer include encapsulation of IP datagram into frames to be transmitted by the network, and mapping IP addresses to physical hardware addresses.

7. Exercise questions

1. List two ways in which the OSI reference model and the TCP/IP reference model are the same. List two ways in which they differ.
2. What is the main difference between TCP and UDP?
3. List two advantages and disadvantages of having international standards for network protocols.
4. How many Layers does TCP/IP model consist of? Explain the function of each layer.

Lecture # 4

1. Synopsis

Comparison Network hardware-Repeaters, Routers, Bridges, Gateways, Hub, Cable Modem

2. Target

At the completion of this lecture you should be able to answer questions like

1. What are the different network topologies?
2. What is a repeater?
3. What is a router?
4. What is a bridge?
5. What is a gateway?
6. What is a Hub?
7. What is a Cable Modem?
8. Compare the different network hardware elements.

3. Introduction

In your previous paper on Data Communication you have come across the different types of networks. As stated above, in this lecture, we plan to explore the different network topologies and the network hardware elements. Before we begin on this topic, a revision of the concepts like the different transmission technologies and the different types of networks etc. will be helpful. Once we finish this aspect, we will proceed towards exposition of items listed in the synopsis. In particular, we will emphasize the following

- Network Topologies
  - Bus
  - Star
  - Ring
  - Mesh
- Network Hardware
  - Repeaters
  - Router
  - Bridge
  - Gateway

- Hub
- Cable Modem

4. Revision / Prerequisites

4.1. Transmission Technology

There are two types of transmission technology.

- **Broadcast links (Multiple Access).**
- **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. 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**.

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**.

4.2. Types of Networks

Networks can be divided into three types based on geographical areas covered:

- **LAN**
- **MAN**
- **WAN**

LAN (Local Area Networks): LANs may use a transmission technology consisting of a cable, to which all the machines are attached,

- Typically connects computer in a single building or campus.
- Developed in 1970s.
- Medium: optical fibers, coaxial cables, twisted pair, wireless.
- Low latency (except in high traffic periods).
- High-speed networks (10 to 100 Mb/sec).
- Speeds adequate for most distributed systems

Problems:

- Multi media based applications
- Typically buses or rings.
- Ethernet, Token Ring

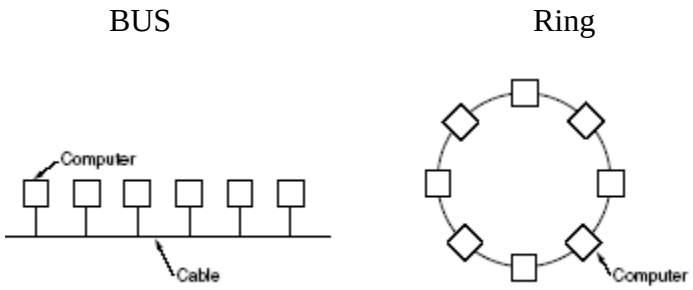


Figure 4.4.1.

MAN (Metropolitan Area Networks): The best-known example of a MAN is the cable television network available in many cities.

- Generally covers towns and cities (50 kms)
- Developed in 1980s.
- Medium: optical fibers, cables.
- Data rates adequate for distributed computing applications.
- A typical standard is DQDB (Distributed Queue Dual Bus).

- Typical latencies < 1 msec.
- Message routing is fast.

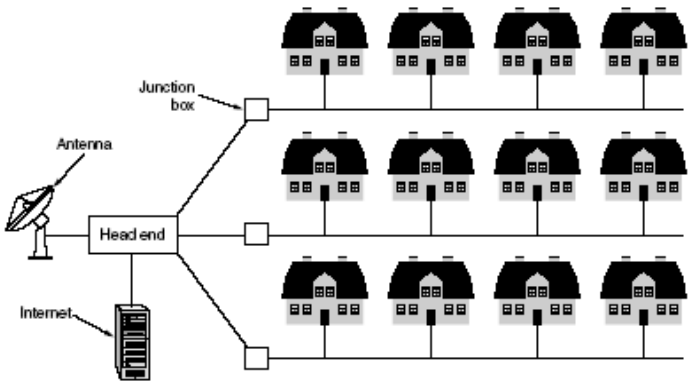


Figure 4.4.2. Man based on Cable TV

WAN (Wide Area Networks): It contains a collection of machines intended for running user (i.e., application) programs. In most wide area networks, the subnet consists of two distinct components: transmission lines and switching elements. **Transmission lines** move bits between machines. They can be made of copper wire, optical fiber, or even radio links. **Switching elements** are specialized computers that connect three or more transmission lines. When data arrive on an incoming line, the switching element must choose an outgoing line on which to forward them. These switching computers have been called by various names in the past; the name **router** is now most commonly used.

- Developed in 1960s.
- Generally covers large distances (states, countries, continents).
- Medium: communication circuits connected by routers.
- Routers forward packets from one to another following a route from the sender to the receiver. Store-and-Forward
- Hosts are typically connected (or close to) the routers.
- Typical latencies: 100ms - 500ms.
- Problems with delays if using satellites.
- Typical speed: 20 - 2000 Kbits/s.

- Not (yet) suitable for distributed computing.
- New standards are changing the landscape.

In this model, each host is frequently connected to a LAN on which a router is present, although in some cases a host can be connected directly to a router. The collection of communication lines and routers (but not the hosts) form the subnet.

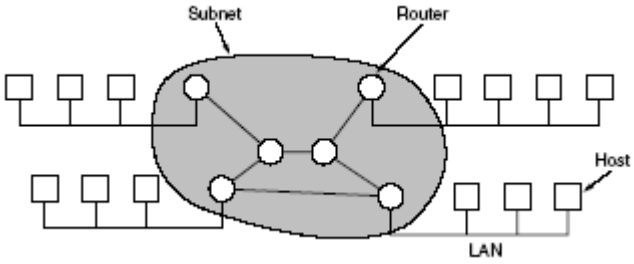


Figure 4.2.3. Network Components

Common requirements are:

- Connect networks of different types, different vendors.
- Provide common communication facilities and hide different hardware and protocols of constituent networks.
- Needed for extensible open distributed systems

5.1. Network Topology

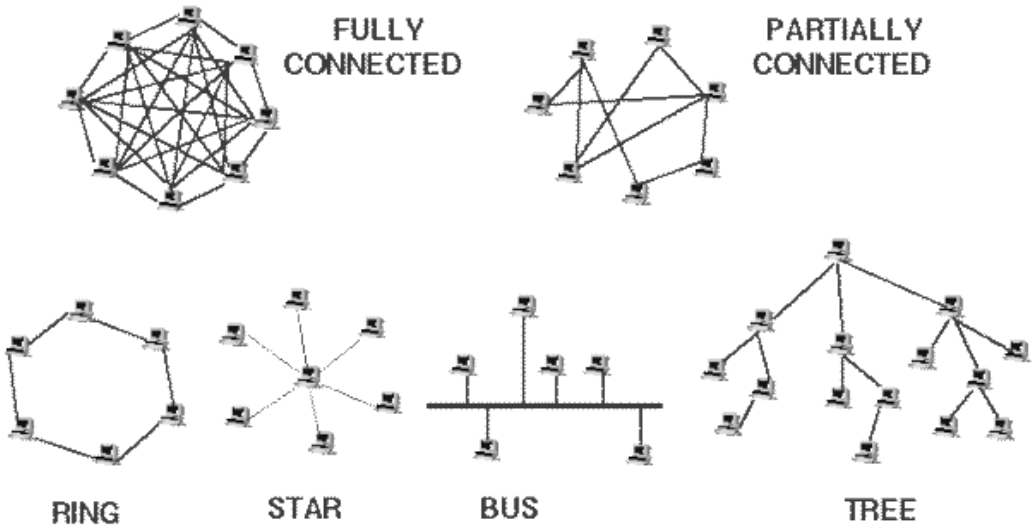


Figure 4.5.1. Network Topology

BUS Network: Bus network is a network architecture in which a set of clients is connected via a shared communications line, called a bus. Bus networks are the simplest way to connect multiple clients, but often have problems when two clients want to communicate at the same time on the same bus.

Advantages

Easy to implement and extend

Well suited for temporary networks (quick setup)

Typically the cheapest topology to implement

Failure of one station does not affect others

Disadvantages

Difficult to administer/troubleshoot

Limited cable length and number of stations

A cable break can disable the entire network

Maintenance costs may be higher in the long run

Performance degrades as additional computers are added

Low security (all computers on the bus can see all data transmissions on the bus)

One virus in the network will affect all of them (but not as badly as a star or ring network)

Star network: In its simplest form, star network consists of one central or hub computer which acts as a router to transmit messages.

Advantages

Easy to implement and extend

Well suited for temporary networks

The failure of a non-central node will not affect the functionality of the network

Disadvantages

Limited cable length and number of stations

Maintenance costs may be higher in the long run

Failure of the central node can disable the entire network

One virus in the network will affect them all

Ring network: Ring network is a topology where each user is connected to two other users. Popular example for such a network is the token ring network.

Advantages

All stations have equal access

Each node on the ring acts as a repeater, allowing ring networks to span greater distances than other topologies

When a coaxial cable is used to create the ring network the service becomes much faster

Disadvantages

The most expensive topology

Mesh Network: Mesh network is a way to route data, voice and instructions between nodes. It allows for continous connections and reconfiguration around blocked paths by hopping from node to node until a connection can be established.

Mesh networks are self healing, i.e. the network can still operate even when a node breaks down or a connection goes bad. Thus it a very reliable network. This is applicable to wireless networks, wired networks and software interaction.

This network allows inexpensive peer network nodes to supply services to other nodes in the same network.

Star-Bus Network: It is a combination of a star network and a bus network. A hub is used to connect the nodes to the network.

5.2. Network Hardware

5.2.1. Network adapter: Interfaces a computer board with the network medium.

5.2.2.Repeaters

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

Figure 4.5.2. Repeater

5.2.3. Bridges

A bridge is a simple device that split the physical network being shared by many computers into smaller segments. A bridge generally has only two ports; bridges with more than two ports are usually called switches. Bridges are normally used to connect LAN segments within a limited geographic area (local bridges), like a building or a campus. Bridges can be programmed to reject packets from particular networks.

Ethernet is the most commonly used physical network. On an Ethernet network, all the connected computers share the same piece of "wire" (it's not physically the same piece, but it is electrically). When two computers attempt to talk at the same time, they drown each other out and create what's called a collision. The more computers you have on one Ethernet, the bigger chance you'll have a collision.

Bridges split an Ethernet into multiple collision-domains. All the data on one side of the bridge stays there, unless it's destined for a computer on the other side of the bridge, lessening the overall load on each segment. Bridges forward all broadcast messages. Only a special bridge called a **translation bridge** will allow two networks of different architectures to be connected. Bridges do not normally allow connection of networks with different architectures. It does not propagate noise signals and defective frames as it was the case for repeaters (at the physical layer). It adaptively recognizes which machines are reachable from a port. It reduces traffic on each port and it improves security since each port will only transmit frames directed to nodes reachable from that port. All the nodes reachable from a node through segments and bridges will receive broadcast messages sent by that node.

Bridges don't care what protocol is being used on the network (TCP/IP, IPX, AppleTalk, etc.) since they operate at the data-link level. This is both a benefit and a curse; since they work at such a simple level, bridges are able to operate at blindingly fast speeds, but since they will indiscriminately forward data, one has little control over their operation. This is where routers come in.

5.2.4. HUB

Hubs can also be called either Multi port Repeaters or Concentrators. They are physical hardware devices. Hubs are used to provide a Physical Star Topology. At the center of

the star is the Hub, with the network nodes located on the tips of the star. Hubs are a crucial element to all star topology LANs. Hubs serve as a central device through which data bound for a workstation travels. Hubs do not read any of the data passing through them and are not aware of their source or destination. Essentially, a hub simply receives incoming packets, possibly amplifies the electrical signal, and broadcasts these packets out to all devices on the network - including the one that originally sent the packet. The data may be distributed, amplified, regenerated, screened or cut off. The hierarchical use of hubs removes the length limitation of the cables.

Hub joins multiple computers (or other network devices) together to form a single network segment. On this network segment, all computers can communicate directly with each other.

A hub includes a series of ports that each accepts a network cable. Small hubs network four computers.

Hubs classify as Layer 1 devices in the OSI model. (Physical layer)

Three different types of hubs exist:

- Passive
- Active
- Intelligent

Passive hubs do not amplify the electrical signal of incoming packets before broadcasting them out to the network. Active hubs, on the other hand, do perform this amplification, as does a different type of dedicated network device called a repeater. They can also be treated as concentrator when referring to a passive hub and multiport repeater when referring to an active hub.

Intelligent hubs add extra features to an active hub that are of particular importance to businesses. An intelligent hub typically is stackable (built in such a way that multiple units can be placed one on top of the other to conserve space). It also typically includes remote management capabilities

5.2.5. Router

A router is a box or a regular computer with at least two ports, used to connect also dissimilar networks. A router is used to route data packets between two networks. It reads the information in each packet to tell where it is going. It differs from bridges since it

operates at the network level. It will also use different addresses. For example a bridge may use Ethernet addresses while a router uses IP addresses. Routers work at the network layer - they actually understand the protocols being used to carry the data over the network. And since they understand the protocols, they can use rules to decide what to do with a specific piece of data. Because of this, routers are useful in linking networks that are used for different purposes or by different organizations. One can apply rules or filters to let certain data in, but keep other data out. Also to route data serving one purpose over a certain set of network connections, while routing other data over other connections. This convenience comes at a price. The more detail a router must acquire about a specific piece of data before forwarding it on, the longer that piece of data is delayed before being sent on to its destination. Also, the greater configurability of routers requires faster, more expensive hardware.

It does all the transformations that may be required by the transfer of packets across the networks it connects. Routing involves two basic activities: running routing algorithms to determine routes, as expressed by **routing tables**, and using the routing tables to move packets across the network. The latter activity is easier and it is called **switching**. Routing tables contain information that will indicate for each packet on the basis of its final destination (usually an IP address) where to go next (**next-hop forwarding)** as to the port to be used and the physical address of the next router.

5.2.6. Gateways

A gateway can translate information between different network data formats or network architectures. It can translate TCP/IP to AppleTalk so computers supporting TCP/IP can communicate with Apple brand computers. Most gateways operate at the application layer, but can operate at the network or session layer of the OSI model. Gateways will start at the lower level and strip information until it gets to the required level and repackage the information and work its way back toward the hardware layer of the OSI model.

5.2.7. Cable Modem

The cable television companies have the high speed bandwidth to the homes. Cable modems use the existing cable TV line to provide the high speed bandwidth. It is an asymmetrical transfer rates with the upstream data transfer rate at 2 Mbps. The

downstream data transfer rate is a maximum of 30 Mbps. Most users connect the cable modem to their 10 Mbps Ethernet NIC, and don't utilize the cable modems full bandwidth. Switching to a 100 Mbps Ethernet NIC would give them full bandwidth. Most cable companies use dynamic IP addressing: each time the user connects; they are assigned a new IP address. Most cable TV companies are placing high performance web proxy servers at the head end. These servers store the most commonly accessed web pages and files locally at the head end. The user's web browser first checks the proxy server to see if the file has been downloaded there. If it hasn't, then it goes out on the Internet to download it. The storing of the web pages and files on the local proxy server reduces the load on the communication servers (to the Internet), and gives the impression of extremely fast Internet access.

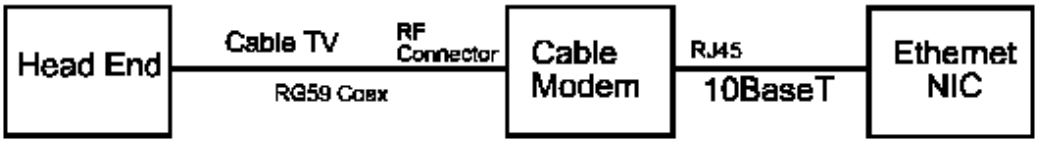


Figure 4.5.3.

The cable modem is connected to the existing cable TV RG59 coax line, using a standard RF connector. The output of the cable modem is a 10BaseT or 100BaseT Ethernet connection to your NIC.

Cable Modem Advantages

- Fast data transfers, up to 30 Mbps if using a 100BaseT NIC
- Competitive pricing against competing technologies
- Easy to install - home prewired

Cable Modem Disadvantages

- The available bandwidth depends on the number of users on the local cable TV line segment.
- There is an asymmetrical transfer rate. Upstream is slower than downstream.
- There can be a bottleneck at the communication server at the head end.

University Questions

1. What is the need for gateways?
2. List the advantages and disadvantages of BUS topology.

3. What are the various network topologies? Discuss with diagram.
4. Write the features of LAN and WAN.
5. Explain briefly the network components used in computer data communication
6. Distinguish clearly between Hubs, Bridges and Routers.

6. Summary

1. Bus network is a network architecture in which a set of clients is connected via a shared communications line, called a bus.
2. Star network consists of one central or hub computer which acts as a router to transmit messages.
3. Ring network is a topology where each user is connected to two other users.
4. Mesh network allows for continuous connections and reconfiguration around blocked paths by hopping from node to node until a connection can be established.
5. A repeater connects two segments of your network cable.
6. A bridge is a simple device that split the physical network being shared by many computers into smaller segments.
7. A bridge generally has only two ports; bridges with more than two ports are usually called switches.
8. Bridges are normally used to connect LAN segments within a limited geographic area (local bridges), like a building or a campus. Bridges can be programmed to reject packets from particular networks.
9. Hubs can also be called either Multi port Repeaters or Concentrators.
10. Hubs are physical hardware devices used to provide a Physical Star Topology.
11. A router is a box or a regular computer with at least two ports, used to connect dissimilar networks and is used to route data packets between two networks.
12. A gateway can translate information between different network data formats or network architectures.
13. Cable modems use the existing cable TV line to provide the high-speed bandwidth.

7. Exercise questions

1. Differentiate between bridges and routers.
2. Write short notes on Gateways.

3. Write short note on Network Repeater.
4. What is unicasting?
5. What is multicasting?

Lecture # 5

1. Synopsis:

Transmission Media– ISDN system Architecture – Communication Satellites – geostationary satellites - Medium Earth Orbit Satellites- Low earth orbit satellites– Satellite v/s Fiber

2. Target

At the completion of this lecture you should be able to answer questions like

1. What are the different transmission media available?
2. Explain the different media for transmission.
3. What is ISDN?
4. Explain ISDN architecture.

3. Introduction

In lecture 4 we discussed about the different network topologies and the network hardware elements. In your previous paper you have come across some media for transmission. As stated above, in this lecture, we plan to explore the bounded, unbounded and wireless transmission media. Also we will be discussing the ISDN architecture. In this lecture, we will emphasize the following

- Transmission media
- ISDN
- ISDN Architecture

4. Revision / Prerequisites

Please refer to Data Communications by Forouzan for notes on different transmission media.

5.1. Transmission Media

The media for transmission can be classified as

- Bound Media
- Unbound Media

- Wireless Media

5.1.1. Bound Media

Media is the path used for transferring data in a network. Bound Media consists of physical substances used to transfer data. The following is a list of the types of bound media and their susceptibility to EMI - (Electrical Magnetic Interference).

- Coaxial Cable - copper core, shielding, used in LANs, EMI
- Fiber Optic - light signal, glass core, no shielding (not required) No EMI
- Unshielded Twisted Pair (UTP) - No shielding, high EMI, very common, cheap
- Shielded Twisted Pair (STP) - Shielding, less EMI than UTP, IBM networks

Coaxial Cable

Coaxial cable consists of two conductors. The inner conductor is held inside an insulator with the other conductor woven around it providing a shield. An insulating protective coating called a jacket covers the outer conductor.

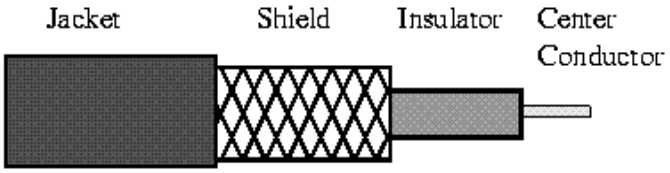


Figure 5.5.1 Coaxial Cable

The outer shield protects the inner conductor from outside electrical signals. The distance between the outer conductor (shield) and inner conductor plus the type of material used for insulating the inner conductor determine the cable properties or impedance. Typical impedances for coaxial cables are 75 ohms for Cable TV, 50 ohms for Ethernet. The excellent control of the impedance characteristics of the cable allow higher data rates to be transferred than with twisted pair cable.

- Used extensively in LANs
- Single central conductor surrounded by a circular insulation layer and a conductive shield
- High bandwidth: Upto 400 Mhz
- High quality of data transmission

- Max. Used data rates: 100 Mbits/s
- Problems: Signal loss at high frequencies

Twisted Pair Cable

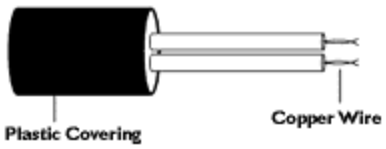


Figure 5.5.2 Twisted Pair Cable

- Extensively used in telephone circuits, where several wires are insulated and put together.
- Bandwidth: 250 KHz.
- Low signal to noise ratio (cross talk) -> Low data rate.
- Good for short-distance communications.
- Used in LAN (UTP or 10baseT).

Optical Fiber

Fiber optic cable has the ability to transmit more information, more quickly and over longer distances. Fiber optic cable offers almost unlimited bandwidth and unique advantages over all previously developed transmission media. The basic point-to-point fiber optic transmission system consists of three basic elements: the optical transmitter, the fiber optic cable and the optical receiver.

The Optical Transmitter: The transmitter converts an electrical analog or digital signal into a corresponding optical signal. The source of the optical signal can be either a light emitting diode, or a solid-state laser diode. The most popular wavelengths of operation for optical transmitters are 850, 1300, or 1550 nanometers.

The Fiber Optic Cable: The cable consists of one or more glass fibers, which act as waveguides for the optical signal. Fiber optic cable is similar to electrical cable in its construction, but provides special protection for the optical fiber within. For systems requiring transmission over distances of many kilometers, or where two or more fiber optic cables must be joined together, an optical splice is commonly used.

The Optical Receiver: The receiver converts the optical signal back into a replica of the original electrical signal. The detector of the optical signal is either a PIN-type photodiode or avalanche-type photodiode.

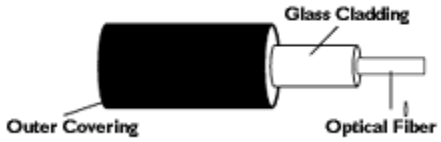


Figure 5.5.3 Fiber Optic Cable

- High quality and high bandwidth data transmission applications.
- Use light instead of electric pulses for message transmission.
- Very high frequency ranges (20,000 Mhz).
- Single fiber can support over 30,000 telephone lines.
- Data transmission rates of 400 Mbits/s and more.
- Becoming very popular for MAN and LAN, also used for intercontinental links.
- High signal to noise ratio, difficulty in tapping (security).
- Cost is the single biggest drawback (currently).

Advantages of Fiber Optic communication

Fiber optic transmission systems – a fiber optic transmitter and receiver, connected by fiber optic cable – offer a wide range of benefits not offered by traditional copper wire or coaxial cable. These include:

1. The ability to carry much more information and deliver it with greater fidelity than either copper wire or coaxial cable.

2. Fiber optic cable can support much higher data rates, and at greater distances, than coaxial cable, making it ideal for transmission of serial digital data.

3. The fiber is totally immune to virtually all kinds of interference, including lightning, and will not conduct electricity. It can therefore come in direct contact with high voltage electrical equipment and power lines. It will also not create ground loops of any kind.

4. As the basic fiber is made of glass, it will not corrode and is unaffected by most chemicals. It can be buried directly in most kinds of soil or exposed to most corrosive atmospheres in chemical plants without significant concern.

5. Since the only carrier in the fiber is light, there is no possibility of a spark from a broken fiber. Even in the most explosive of atmospheres, there is no fire hazard, and no danger of electrical shock to personnel repairing broken fibers.

6. Fiber optic cables are virtually unaffected by outdoor atmospheric conditions, allowing them to be lashed directly to telephone poles or existing electrical cables without concern for extraneous signal pickup.

7. A fiber optic cable, even one that contains many fibers, is usually much smaller and lighter in weight than a wire or coaxial cable with similar information carrying capacity. It is easier to handle and install, and uses less duct space. (It can frequently be installed without ducts.)

8. Fiber optic cable is ideal for secure communications systems because it is very difficult to tap but very easy to monitor. In addition, there is absolutely no electrical radiation from a fiber

5.1.2. Unbound Media

Media is the path used for transferring data in a network. Unbound Media consists of the wireless path used to transfer data. The following is a list of the types of unbound media and their suseptability to EMI - (Electrical Magnetic Interference).

- Radio Waves
- Micro Waves - Terrestrial and Satellite
- Infrared

Microwave Transmission

Another popular transmission medium is microwave. This is a popular way of transmitting data since it does not require the expense of laying cables. Microwave systems use very high frequency radio signals to transmit data through space. However, at microwave frequencies the electromagnetic waves cannot bend or pass through obstacles like hills, etc. Hence microwave transmission is a line-of-sight method of

communication. In other words, the transmitter and receiver of a microwave system, which are mounted on very high towers, should be in a line-of-sight. This may not be possible for very long distance transmission due to physical constraints. Moreover, the signals become weak after traveling a certain distance and require power amplification.

In order to overcome the problems of line-of-sight and power amplification of weak signals, microwave systems use repeaters at intervals of about 25 to 30 km in between the transmitting and receiving stations. The first repeater is placed in line-of-sight of the transmitting station and the last repeater is placed in line-of-sight of the receiving station. Two consecutive repeaters are also placed in line-of-sight of each other. The data signals are received, amplified, and re-transmitted by each of these stations.

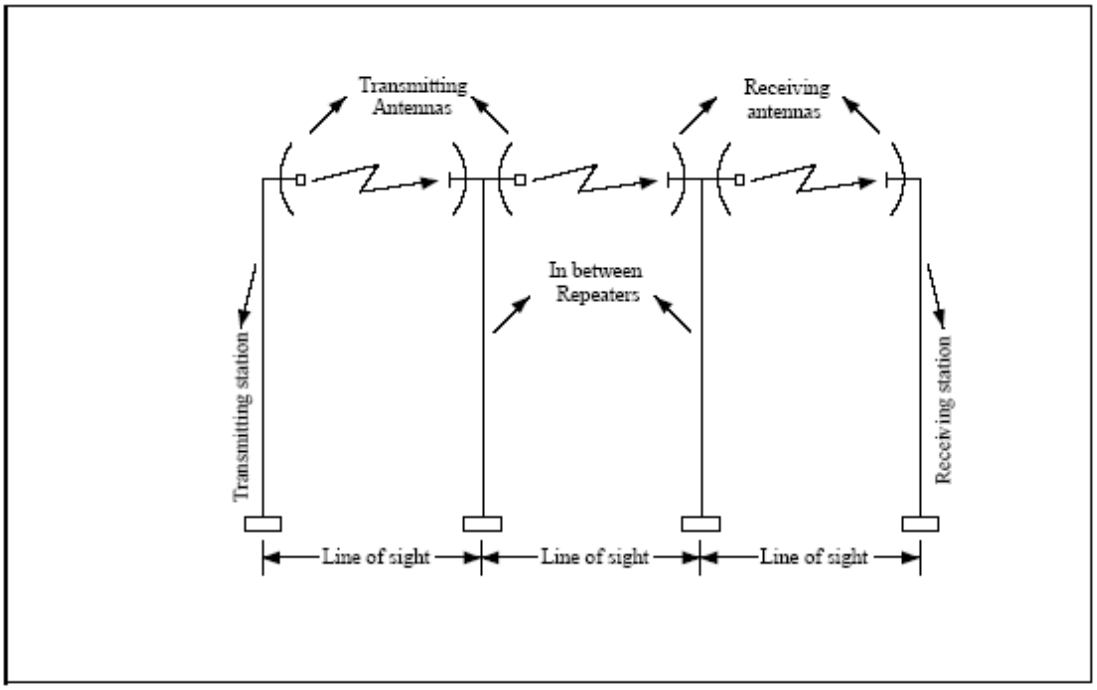


Figure 5.5.4. Microwave Transmission

Advantages and Limitations of Microwave Transmission

Microwave systems permit data transmission rates of about 16 Giga (1 giga = 109) bits per second. At such high frequency, a microwave system can carry 250,000 voice channels at the same time.

However, the capital investment needed to install microwave links is very high and hence they are mostly used to link big metropolitan cities that have heavy telephone traffic between them.

5.1.3. Wireless Media

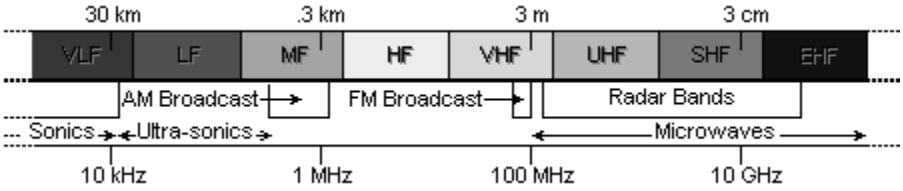


Figure 5.5.5.

- For WANs satellites provide global communication over the world, receiving signals from transmitters and relaying them back to the receivers.
- With geostationary satellites senders and receivers always points the same direction.
- High communication capacity. Big latency : 0.25 secs.
- For MANs microwave radio technology is widely used (2 to 24 Mbit/s).
- For LANs Spread Spectrum radio technology is becoming very popular (up to 2 Mbit/s).
- Infrared: Line of sight limitation.

Communication Satellite

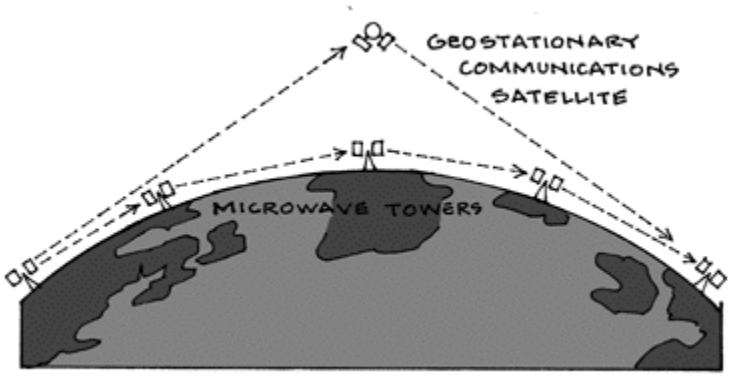


Figure 5.5.6.

The main problem with microwave communications is that the curvature of the earth, mountains, and other structures often block the line-of-sight. Due to this reason, several

repeater stations are normally required for long distance transmission, which increases the cost of data transmission between two points. This problem is overcome by using satellites which are relatively newer and more promising data transmission media.

A communication satellite is basically a microwave relay station placed precisely at 36,000 km above the equator where its orbit speed exactly matches the earth's rotation speed. Since a satellite is positioned in a *geo-synchronous* orbit, (i.e. the orbit where the speed of the satellite matches the earth's rotation speed), it appears to be stationary relative to earth and always stays over the same point with respect to earth. This allows a ground station to aim its antenna at a fixed point in the sky. The Indian satellite, INSAT-1B, is positioned in such a way that it is accessible from any place in India.

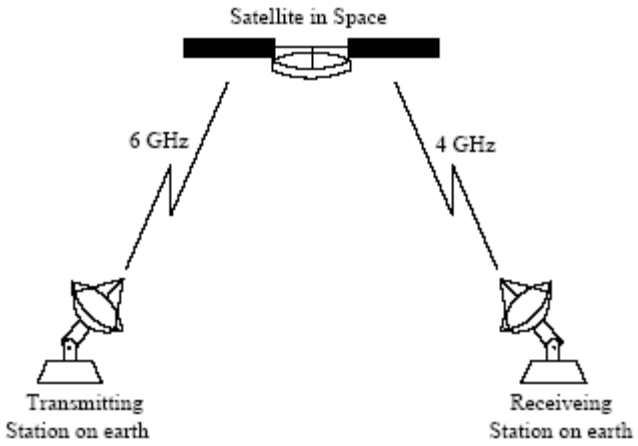


Figure 5.5.7.

Communications through satellites are either passive or active. The first communications satellites were passive. Signals from Earth were merely reflected from the orbiting metallic sphere. Later types of satellites are active. Active communication satellites receive signals from Earth, electronically strengthen the signals, and transmit the signals to Earth.

This relaying of signals from one Earth Station to another is done through the satellite's transponder. Most communications satellites have more than one transponder and antenna so that they can relay several users of radio waves or signals at the same time.

Advantages and Limitations

The main advantage of satellite communication is that it is a single microwave relay station visible from any point of a very large area on the earth. For example, satellites

used for national transmission are visible from all parts of the country. Thus transmission and reception can be between any two randomly chosen places in that area. Moreover, transmission and reception costs are independent of the distance between the two points. In addition to this, a transmitting station can receive back its own transmission and check whether the satellite has transmitted the information correctly. If an error is detected, the data would be retransmitted.

A major drawback of satellite communications has been the high cost of placing the satellite into its orbit. Moreover, a signal sent to a satellite is broadcasted to all receivers within the satellite's range. Hence necessary security measures need to be taken to prevent unauthorized tampering of information.

Communications satellites are launched by rockets or carried into space by the Space Shuttle. Once in space, small engines on the satellites guide the satellite into orbit and help keep them there. Most communications satellites are placed in orbit at an altitude of 22,300 miles above the Earth. This is known as a geostationary or synchronous orbit. This allows the satellite to orbit the Earth at the same speed as the rotation of the Earth. As a result, the satellite appears to be stationary above the same location on Earth.

Communications satellites will be used to link all the regions and people of the world.

This is a giant step from the early uses of communication satellites. This global system will consist of many satellites, positioned in geostationary orbit, providing high bandwidth capacity; interconnect many highly specialized Earth Stations operating in more than thirty countries. This network, already in progress by consortiums headed by Motorola (Iridium) will provide the framework and capability for anyone in the world to communicate with anyone else, regardless of location.

5.2. ISDN

ISDN stands for Integrated Services Digital Network. It is a design for a completely digital telephone/telecommunications network. It is designed to carry voice, data, images, and video, everything you could ever need. It is also designed to provide a single interface (in terms of both hardware and communication protocols) for hooking up your phone, your fax machine, your computer, your videophone, your video-on-demand system (someday), and your microwave.

B-ISDN is Broadband ISDN. (The older ISDN is often called Narrowband ISDN.) This is *not* simply faster ISDN, or ISDN with the copper to your home finally upgraded to fiber. B-ISDN is a *complete* redesign. It is still capable of providing all the integrated services (voice, data, video, etc.) through a single interface just like ISDN was supposed to.

5.2.1 ISDN Architecture

ISDN components include terminals, terminal adapters (TA), network-termination devices, line-termination equipment and exchange-termination equipment.

- **ISDN** terminals come in two types:
- Specialized **ISDN** terminals are referred to as **terminal equipment type 1 (TE1)**.
- Non - **ISDN** terminals such as **DTE** that predate the **ISDN** standards are referred to as **terminal equipment type 2 (TE2)**.

TE1 are connected to the **ISDN** network through a four-wired twisted-pair digital link.

TE2 are connected to the **ISDN** network through a terminal adapter.

The **ISDN TA** can either be a stand-alone device or a board inside TE2. If implemented as a stand-alone device, the TE2 is connected to the TA via a standard physical layer interface.

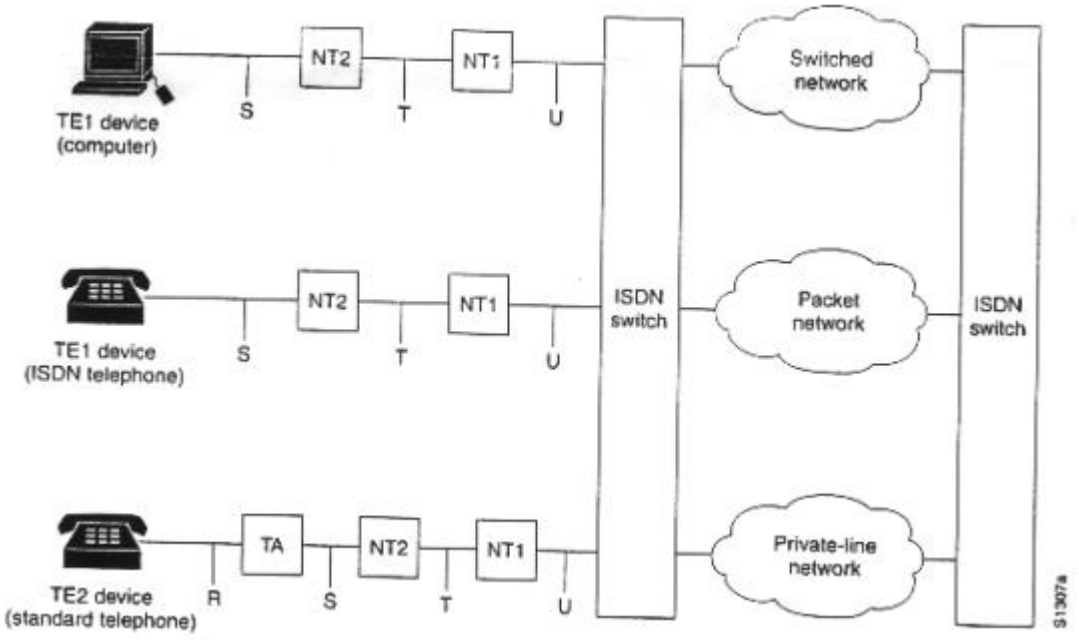


Figure 5.5.8. Sample ISDN Configuration

University Questions

1. What is a low earth orbit satellite?
2. Explain the different types of satellites used as transmission media. (Nov 2005)
3. Explain in detail the ISDN system architecture.
4. Write notes on ISDN service.
5. What are the major differences between Basic Rate and Primary Rate interfaces of ISDN?
6. Explain Ethernet Networks.
7. Explain the various physical media's used in computer networks.
8. Compare analog and digital transmission with special reference to ISDN.
9. Write notes on fiber optic communication.
10. What are some of the physical media that Ethernet can run over?
11. What are the various IEEE standards? Briefly explain.

6. Summary

1. The different Bound media for transmission are co-axial cable, twisted pair cable and optical fiber.
2. Unbound media for communication are radio waves, microwaves and infrared
3. Wireless media of transmission is with the help of communication satellites.
4. ISDN (Integrated Services Digital Network) is a design for a completely digital telephone/telecommunications network to carry voice, data, images, and video, to provide a single interface for hooking up your phone, your fax machine, your computer, your videophone, your video-on-demand system (someday), and your microwave

7. Exercise questions

1. Explain various transmission medias.
2. Write notes on fiber optic communication.
3. Write notes on coaxial cable.