At the same time, each router in the AS keeps track of how to get to each border router using a conventional intradomain protocol with no injected information. By combining these two sets of information, each router in the AS is able to determine the appropriate next hop for all prefixes.

30.IP VERSION 6 (IPV6)

In many respects, the motivation for a new version of IP is the same as the motivation for the techniques described so far in this section: to deal with scaling problems caused by the Internet's massive growth. Subnetting and CIDR have helped to contain the rate at which the Internet address space is being consumed (the address depletion problem) and have also helped to control the growth of routing table information needed in the Internet's routers (the routing information problem).

However, there will come a point at which these techniques are no longer adequate. In particular, it is virtually impossible to achieve 100% address utilization efficiency, so the address space will be exhausted well before the 4 billionth host is connected to the Internet. Even if we were able touse all 4 billion addresses, it's not too hard to imagine ways that that number could be exhausted, such as the assignment of IP addresses to mobile phones, televisions, or other household appliances.

Historical Perspective

The IETF began looking at the problem of expanding the IP address space in 1991, and several lternatives were proposed. Since the IP address is carried in the header of every IP packet, increasing the size of the address dictates a change in the packet header.This means a new version of the Internet Protocol, and as a consequence, a need for new software for every host and router in the Internet. This is clearly not a trivial matter—it is a major change that needs to be thought about very carefully. The effort to define a new version of IP was known as IP Next Generation, or IPng. As the work progressed, an official IP version number was assigned, so IPng is now known as IPv6. Note that the version of IP discussed so far in this chapter is version 4 (IPv4). The apparent discontinuity in numbering is the result of version number 5 being used for an experimental protocol some years ago. The significance of the change to a new version of IP caused a snowball effect.

- Support for real-time services;
- Security support;
- Auto configuration (i.e., the ability of hosts to automatically configure themselves with such information as their own IP address and domain name);
- Enhanced routing functionality, including support for mobile hosts.

Prefix	Use
00. . . 0 (128 bits)	Unspecified
00. . . 1 (128 bits)	Loopback
1111 1111	Multicast addresses
1111 1110 10	Link local unicast
1111 1110 11	Site local unicast
Everything else	Global unicast

Table 4.11 Address prefix assignments for IPv6.

Addresses and Routing

First and foremost, IPv6 provides a 128-bit address space, as opposed to the 32 bitsof version 4. Thus, while version 4 can potentially address 4 billion nodes if addressassignment efficiency reaches 100%, IPv6 can address 3.4×1038 nodes, again assuming100% efficiency. As we have seen, though, 100% efficiency in address assignment is notlikely. Some analysis of other addressing schemes, such as those of the French and U.S.telephone networks, as well as that of IPv4, have turned up some empirical numbers foraddress assignment efficiency.

Address Space Allocation

Drawing on the effectiveness of CIDR in IPv4, IPv6 addresses are also classless, but the address space is still subdivided in various ways based on the leading bits. Rather than specifying different address classes, the leading bits specify different uses of the IPv6 address. This allocation of the address space warrants a little discussion. First, the entire functionality of IPv4's three main address classes (A, B, and C) is contained inside the "everything else" range. Global unicast addresses, as we will see shortly, are a lot like classless IPv4 addresses, only much longer. These are the main ones of interest at this point, with over 99% of the total IPv6 address space available to this important form of address. (At the time of writing, IPv6 unicast addresses are being allocated from the block that begins 001, with the remaining address space—about 87%—being reserved for future use.) The multicast address space is (obviously) for multicast, thereby serving the same role as class D addresses in IPv4. Note that multicast addresses are easy to distinguish— they start with a byte of all 1s.

Address Notation

Just as with IPv4, there is some special notation for writing down IPv6 addresses. The standard representation is x:x:x:x:x:x:x:x where each "x" is a hexadecimal representation of a 16-bit piece of the address. An example would be 7CD:1234:4422:ACO2:0022:1234:A456:0124 Any IPv6 address can be written using this notation. Since there are a few special types of IPv6 addresses, there are some special notations that may be helpful in certain circumstances.

For example, an address with a large number of contiguous 0s can be written more compactly by omitting all the 0 fields. Thus,47CD:0000:0000:0000:0000:0000:A456:0124 could be written 47CD::A456:0124 Clearly, this form of shorthand can only be used for one set of contiguous 0s in an address to avoid ambiguity. Since there are two types of IPv6 addresses that contain an embedded IPv4 address, these have their own special notation that makes extraction of the IPv4 address easier. For example, the IPv4-mapped IPv6 address of a host whose IPv4 address was

128.96.33.81 could be written as ::FFFF:128.96.33.81That is, the last 32 bits are written in IPv4 notation, rather than as a pair of hexadecimal numbers separated by a colon. Note that the double colon at the front indiates the leading 0s.

Figure 4.35 An IPv6 provider-based unicast address.

Global Unicast Addresses

By far the most important sort of addressing that IPv6 must provide is plain old unicast addressing. It must do this in a way that supports the rapid rate of addition of new hosts to the Internet and that allows routing to be done in a scalable way as the number of physical networks in the Internet grows. Thus, at the heart of IPv6 is the unicast address allocation plan that determines how unicast addresses will be assigned to service providers, autonomous systems, networks, hosts, and routers.

Packet Format

Despite the fact that IPv6 extends IPv4 in several ways, its header format is actually simpler. This simplicity is due to a concerted effort to remove unnecessary functionality from the protocolAs with many headers, this one starts with a Version field, which is set to 6 for IPv6. The Version field is in the same place relative to the start of the header as IPv4's Version field so that header-processing software can immediately decide which header format to look for. The TrafficClass and FlowLabel fields both relate to quality of service issues,

The PayloadLen field gives the length of the packet, excluding the IPv6 header,measured in bytes. The NextHeader field cleverly replaces both the IP options and the Protocol field of IPv4. If options are required, then they are carried in one or more special headers following the IP header, and this is indicated by the value of the NextHeader field. If there are no special headers, the NextHeader field is the demux

Figure 4.36 IPv6 packet header.

Auto configuration

While the Internet's growth has been impressive, one factor that has inhibited faster acceptance of the technology is the fact that getting connected to the Internet has typically required a fair amount of system administration expertise. In particular, every host that is connected to the Internet needs to be configured with a certain minimum amount of information, such as a valid IP address, a subnet mask for the link to which it attaches, and the address of a name server.

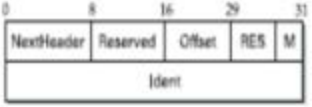


Figure 4.37 IPv6 fragmentation extension header.

Advanced Routing Capabilities

Another of IPv6's extension headers is the routing header. In the absence of this header, routing for IPv6 differs very little from that of IPv4 under CIDR. The routing header contains a list of IPv6 addresses that represent nodes or topological areas that the packet should visit en route to itsdestination.

30.MULTICAST ADDRESSES

IP has a sub range of its address space reserved for multicast addresses. In IPv4, these addresses are assigned in the class D address space, and IPv6 also has a portion of its address space (see Table 4.11) reserved for multicast group addresses. Some sub ranges of the multicast ranges are reserved for intra domain multicast, so they can be reused independently by different domains.

Thus, there are 28 bits of possible multicast addresses in IPv4 when we ignore the prefix shared by all multicast addresses. This presents a problem when attempting to take advantage of hardware multicasting on a LAN. Let's take the case of Ethernet. Ethernet multicast addresses have only 23 bits when we ignore their shared prefix. In other words, to take advantage of Ethernet multicasting, IP has to map 28-bit IP multicast addresses into 23-bit Ethernet multicast addresses. This is implemented by taking the low-order 23 bits of any IP multicast address to use as its Ethernet multicast address, and ignoring the high-order 5 bits. Thus, 32 (25) IP addresses map into each one of the Ethernet addresses.

When a host on an Ethernet joins an IP multicast group, it configures its Ethernet interface to receive any packets with the corresponding Ethernet multicast address. Unfortunately, this causes the receiving host to receive not only the multicast traffic it desired, but also traffic sent to any of the other 31 IP multicast groups that map to the same Ethernet address, if they are routed to that Ethernet. Therefore, IP at the receiving host must examine the IP header of any multicast packet to determine whether the packet really belongs to the desired group. In summary, the mismatch of multicast address sizes means that multicast traffic may place a burden on hosts that are not even interested in the group to which the traffic was sent. Fortunately, in some switched networks (suchas switched Ethernet) this problem can be mitigated by schemes wherein the switches recognize unwanted packets and discard them.

31.MULTICAST ROUTING (DVMRP, PIM)

A router's unicast forwarding tables indicate, for any IP address, which link to use to forward the unicast packet. To support multicast, a router must additionally have multicastforwarding tables that indicate, based on multicast address, which links—possibly more than one—to use to forward the multicast packet (the router duplicates the packet if it is to be forwarded over multiple links). Thus, where unicast forwarding tables collectively specify a set of paths, multicast forwarding tables collectively specify a set of trees: *multicast distribution trees*. Furthermore, to support source-specific multicast (and, it turns out, for some types of any source multicast), the multicast forwarding tables must indicate which links to use based on the combination of multicast address and the (unicast) IP address of the source, again specifying a set of trees.

Multicast routing is the process by which the multicast distribution trees are determined or, more concretely, the process by which the multicast forwarding tables are built. As with unicast routing, it is not enough that a multicast routing protocol "work"; it must also scale reasonably well as the network grows, and it must accommodate the autonomy of different routing domains.

32. DVMRP-DISTANCE VECTOR MULTICAST ROUTING PROTOCOL

Distance-vector routing,for unicast, can be extended to support multicast. The resulting protocol is called *Distance Vector Multicast Routing Protocol*, or DVMRP. DVMRP was the first multicast routing protocol to see widespread use.

Recall that, in the distance-vector algorithm, each router maintains a table of _Destination, Cost, NextHop_ tuples, and exchanges a list of _Destination, Cost_ pairs with its directly connected neighbors.

Extending this algorithm to support multicast is a two-stage process. First, we create a broadcast mechanism that allows a packet to be forwarded to all the networks on the internet. Second, we need to refine this mechanism so that it prunes back networks that do not have hosts that belong to the multicast group.

Consequently, DVMRP is one of several multicast routing protocols described as *flood-and-prune* protocols.

Given a unicast routing table, each router knows that the current shortest path to a given destination goes through NextHop. Thus, whenever it receives a multicast packet from source S, the router forwards the packet on all outgoing links (except the one on which the packet arrived) if and only if the packet arrived over the link that is on the shortest path to S (i.e., the packet came *from* the NextHop associated with S in the routing table). This strategy effectively floods packets outward from S, but does not loop packets back toward S.

There are two major shortcomings to this approach. The first is that it truly floods the network; it has no provision for avoiding LANs that have no members in the multicast group. We address this problem below. The second limitation is that a given packet will be forwarded over a LAN by each of the routers connected to that LAN. This is due to the forwarding strategy of flooding packets on all links other than the one on which the packet arrived, without regard to whether or not those links are part of the shortest-path tree rooted at the source.

The solution to this second limitation is to eliminate the duplicate broadcast packets that are generated when more than one router is connected to a given LAN. One way to do this is to designate one router as the "parent" router for each link, relative to the source, where only the parent router is allowed to forward multicast packets from that source over the LAN. The router

that has the shortest path to source S is selected as the parent; a tie between two routers would be broken according to which router has the smallest address. A given router can learn if it is the parent for the LAN (again relative to each possible source) based upon the distance-vector messages it exchanges with its neighbors.

The second stage is to propagate this "no members of G here" information up the shortest-path tree. This is done by having the router augment the _ Destination, Cost _ pairs it sends to its neighbors with the set of groups for which the leaf network is interested in receiving multicast packets. This information can then be propagated from router to router, so that for each of its links, a given router knows for what groups it should forward multicast packets.

Note that including all of this information in the routing update is a fairly expensive thing to do. In practice, therefore, this information is exchanged only when some source starts sending packets to that group. In other words, the strategy is to use RPB, which adds a small amount of overhead to the basic distance-vector algorithm, until a particular multicast address becomes active. At that time, routers that are not interested in receiving packets addressed to that group speak up, and that information is propagated to the other routers.

33.PIM-PROTOCOL INDEPENDENT MULTICAST

Protocol-independent multicast, or PIM, was developed in response to the scaling problems of earlier multicast routing protocols. In particular, it was recognized that the existing protocols did not scale well in environments where a relatively small proportion of routers want to receive traffic for a certain group.

For example, broadcasting traffic to all routers until they explicitly ask to be removed from the distribution is not a good design choice if most routers don't want to receive the traffic in the first place.

This situation is sufficiently common that PIM divides the problem space into sparse mode and dense mode, where sparse and dense refer to the proportion of routers that will want the multicast. PIM dense mode (PIM-DM) uses a flood-and-prune algorithm like DVMRP, and suffers from the same scalability problem.

PIM sparse mode (PIM-SM) has become the dominant multicast routing protocol. The "protocol-independent" aspect of PIM, by the way, refers to the fact that, unlike earlier protocols such as DVMRP, PIM does not depend on any particular sort of unicast routing—it can be used with any unicast routing protocol. In PIM-SM, routers explicitly join the multicast distribution tree using PIM protocol messages known as Join messages.

The contrast to DVMRP's approach of creating a broadcast tree first and then pruning the uninterested routers. The question that arises is where to send those Join messages because, after all, any host (and any number of hosts) could send to the multicast group. To address this, PIM-SM assigns to each group a special router known as the *rendezvous point (RP)*.

In general, a number of routers in a domain are configured to be candidate RPs, and PIM-SM defines a set of procedures by which all the routers in a domain can agree on the router to use as the RP for a given group. These procedures are rather complex, as they must deal with a wide variety of scenarios, such as the failure of a candidate RP and the partitioning of a domain into two separate networks due to a number of link or node failures. All routers in a domain know the uncast IP address of the RP for a given group. A multicast forwarding tree is built as a result of routers sending Join messages to the RP. PIM-SM allows two types of tree to be constructed: a *shared* tree, which may be used by all senders, and a *source-specific* tree, which may be used only by a specific sending host.

The normal mode of operation creates the shared tree first, followed by one or more source-specific trees if there is enough traffic to warrant it. Because building trees installs state in the routers along the tree, it is important that the default is to have only one tree for a group, not one for every sender to a group.All of its mechanisms for building and maintaining trees take advantage of unicast routing without depending on any particular unicast routing protocol. The formation of trees is entirely determined by the paths that Join messages follow, which is determined by the choice of shortest paths made by unicast routing. Thus, to be precise, PIM is "unicast routing protocol independent," as compared to DVMRP. Note that PIM is very much bound up with the Internet Protocol—it is not protocol independent in terms of network-layer protocols.

The design of PIM-SM again illustrates the challenges in building scalable networks, and how scalability is sometimes pitted against some sort of optimality. The shared tree is certainly more scalable than a source-specific tree, in the sense that it reduces the total state in routers to be on the order of the number of groups rather than the number of senders times the number of groups. However, the source-specific tree is likely to be necessary to achieve efficient routing and effective use of link bandwidth

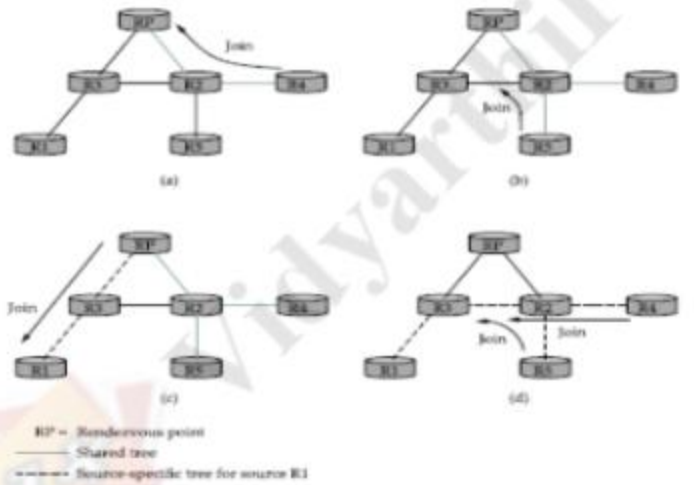


Figure 4.38 PIM operation. (a) R4 sends Join to RP and joins shared tree; (b) R5 joins shared tree; (c) RP builds source-specific tree to R1 by sending Join to R1; (d) R4 and R5 build source-specific tree to R1 by sending Joins to R1.

UNIT IV TRANSPORT LAYER

Overview of Transport layer - UDP - Reliable byte stream (TCP) - Connection management - Flow control - Retransmission – TCP Congestion control - Congestion avoidance (DECbit, RED) – QoS –Application requirements

35. OVERVIEW OF TRANSPORT LAYER

TCP was specifically designed to provide a reliable end to end byte stream over an unreliable internetwork. Each machine supporting TCP has a TCP transport entity either a user process or part of the kernel that manages TCP streams and interface to IP layer. A TCP entity

accepts user data streams from local processes, breaks them up into pieces not exceeding 64KB and sends each piece as a separate IP datagram. Client Server mechanism is not necessary for TCP to behave properly.

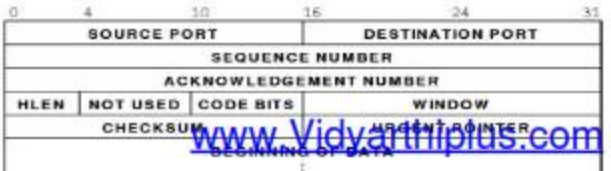
The IP layer gives no guarantee that datagram will be delivered properly, so it is up to TCP to timeout and retransmit, if needed. Duplicate, lost and out of sequence packets are handled using the sequence number, acknowledgements, retransmission, timers, etc to provide a reliable service. Connection is a must for this service.Bit errors are taken care of by the CRC checksum. One difference from usual sequence numbering is that each byte is given a number instead of each packet. This is done so that at the time of transmission in case of loss, data of many small packets can be combined together to get a larger packet, and hence smaller overhead.

TCP connection is a *duplex connection*. That means there is no difference between two sides once the connection is established.

Salient Features of TCP

- **Piggybacking of Acknowledgments:** The ACK for the last received packet need not be sent as a new packet, but gets a free ride on the next outgoing data frame(using the ACK field in the frame header). The technique is temporarily delaying outgoing ACKs so that they can be hooked on the next outgoing data frame is known as piggybacking. But ACK can't be delayed for a long time if receiver(of the packet to be acknowledged) does not have any data to send.
- **Flow and congestion control:** TCP takes care of flow control by ensuring that both ends have enough resources and both can handle the speed of data transfer of each other so that none of them gets overloaded with data. The term congestion control is used in almost the same context except that resources and speed of each router is also taken care of. The main concern is network resources in the latter case.
- **Multiplexing / Demultiplexing:** Many applications can be sending/receiving data at the same time. Data from all of them has to be multiplexed together. On receiving some data from lower layer, TCP has to decide which application is the recipient. This is called demultiplexing. TCP uses the concept of port number to do this.

TCP segment header:



Explanation of header fields:

- **Source and destination port:** These fields identify the local endpoint of the connection. Each host may decide for itself how to allocate its own ports starting at 1024. The source and destination socket numbers together identify the connection.
- **Sequence and ACK number:** This field is used to give a sequence number to each and every byte transferred. This has an advantage over giving the sequence numbers to every packet because data of many small packets can be combined into one at the

time of retransmission, if needed. The ACK signifies the next byte expected from the source and not the last byte received. The ACKs are cumulative instead of selective. Sequence number space is as large as 32-bit although 17 bits would have been enough if the packets were delivered in order. If packets reach in order, then according to the following formula:

(sender's window size) + (receiver's window size) < (sequence number space)

the sequence number space should be 17-bits. But packets may take different routes and reach out of order. So, we need a larger sequence number space. And for optimisation, this is 32-bits.

- **Header length :**This field tells how many 32-bit words are contained in the TCP header. This is needed because the options field is of variable length.
- **Flags :** There are six one-bit flags.
  1. **URG :** This bit indicates whether the urgent pointer field in this packet is being used.
  2. **ACK :**This bit is set to indicate the ACK number field in this packet is valid.
  3. **PSH :** This bit indicates PUSHed data. The receiver is requested to deliver the data to the application upon arrival and not buffer it until a full buffer has been received.
  4. **RST :** This flag is used to reset a connection that has become confused due to a host crash or some other reason. It is also used to reject an invalid segment or refuse an attempt to open a connection. This causes an abrupt end to the connection, if it existed.
  5. **SYN :** This bit is used to establish connections. The connection request(1st packet in 3-way handshake) has SYN=1 and ACK=0. The connection reply (2nd packet in 3-way handshake) has SYN=1 and ACK=1.

6. **FIN :** This bit is used to release a connection. It specifies that the sender has no more fresh data to transmit. However, it will retransmit any lost or delayed packet. Also, it will continue to receive data from other side. Since SYN and FIN packets have to be acknowledged, they must have a sequence number even if they do not contain any data.

- **Window Size:** Flow control in TCP is handled using a variable-size sliding window. The Window Size field tells how many bytes may be sent starting at the byte acknowledged. Sender can send the bytes with sequence number between (ACK#) to (ACK# + window size - 1) A window size of zero is legal and says that the bytes up to and including ACK# -1 have been received, but the receiver would like no more data for the moment. Permission to send can be granted later by sending a segment with the same ACK number and a nonzero Window Size field.
- **Checksum :** This is provided for extreme reliability. It checksums the header, the data, and the conceptual pseudoheader. The pseudoheader contains the 32-bit IP address of the source and destination machines, the protocol number for TCP(6), and the byte count for the TCP segment (including the header).Including the pseudoheader in TCP checksum computation helps detect misdelivered packets, but doing so violates the protocol hierarchy since the IP addresses in it belong to the IP layer, not the TCP layer.
- **Urgent Pointer:** Indicates a byte offset from the current sequence number at which urgent data are to be found. Urgent data continues till the end of the segment. This is not used in practice. The same effect can be had by using two TCP connections, one for transferring urgent data.
- **Options :** Provides a way to add extra facilities not covered by the regular header. eg, ○ Maximum TCP payload that sender is willing to handle. The maximum size of segment is called MSS (Maximum Segment Size). At the time of handshake, both parties inform each other about their capacity. Minimum of the two is honoured. This information is sent in the options of the SYN packets of the three way handshake. Window scale option can be used to increase the window size. It can be specified by telling the receiver that the window size should be interpreted by shifting it left by specified number of bits. This header option allows window size up to 230.
- **Data:** This can be of variable size. TCP knows its size by looking at the IP size header.

36.UDP (USER DATAGRAM PROTOCOL)

UDP -- like its cousin the Transmission Control Protocol (TCP) -- sits directly on top of the base Internet Protocol (IP). In general, UDP implements a fairly "lightweight" layer above the Internet Protocol. It seems at first site that similar service is provided by both UDP and IP, namely transfer of data.But we need UDP for multiplexing/demultiplexing of addresses.

UDP's main purpose is to abstract network traffic in the form of datagrams. A datagram comprises one single "unit" of binary data; the first eight (8) bytes of a datagram contain the header information and the remaining bytes contain the data itself.

UDP Headers
The UDP header consists of four (4) fields of two bytes each:

Source Port	Destination Port
length	checksum

- source port number
- destination port number
- datagram size
- checksum

UDP port numbers allow different applications to maintain their own "channels" for data; both UDP and TCP use this mechanism to support multiple applications sending and receiving data concurrently. The sending application (that could be a client or a server) sends UDP datagrams through the source port, and the recipient of the packet accepts this datagram through the destination port. Some applications use static port numbers that are reserved for or registered to the application. Other applications use dynamic (unregistered) port numbers. Because the UDP port headers are two bytes long, valid port numbers range from 0 to 65535; by convention, values above 49151 represent dynamic ports.

The datagram size is a simple count of the number of bytes contained in the header and data sections . Because the header length is a fixed size, this field essentially refers to the length of the variable-sized data portion (sometimes called the payload). The maximum size of a datagram varies depending on the operating environment. With a two-byte size field, the theoretical maximum size is 65535 bytes. However, some implementations of UDP restrict the datagram to a smaller number -- sometimes as low as 8192 bytes.

UDP checksums work as a safety feature. The checksum value represents an encoding of the datagram data that is calculated first by the sender and later by the receiver. Should an individual datagram be tampered with (due to a hacker) or get corrupted during transmission (due to line noise, for example), the calculations of the sender and receiver will not match, and the UDP protocol will detect this error. The algorithm is not fool-proof, but it is effective in many cases. In UDP, check summing is optional -- turning it off squeezes a little extra performance from the system -- as opposed to TCP where checksums are mandatory. It should be remembered that

check summing is optional only for the sender, not the receiver. If the sender has used checksum then it is mandatory for the receiver to do so.

Usage of the Checksum in UDP is optional. In case the sender does not use it, it sets the checksum field to all 0's. Now if the sender computes the checksum then the recipient must also compute the checksum an set the field accordingly. If the checksum is calculated and turns out to be all 1's then the sender sends all 1's instead of all 0's. This is since in the algorithm for checksum computation used by UDP, a checksum of all 1's if equivalent to a checksum of all 0's. Now the checksum field is unambiguous for the recipient, if it is all 0's then checksum has not been used, in any other case the checksum has to be computed.

37.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.

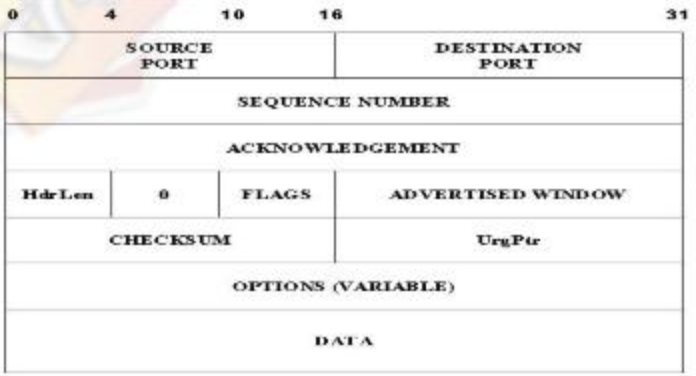
Variations in the RTT are even possible during a single TCP connection. Packets may be reordered as they cross the internet, but this is not possible on a point-to-point link where the first packet put into one end of the link must be the first to appear at the other end. Packets that are slightly out 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.

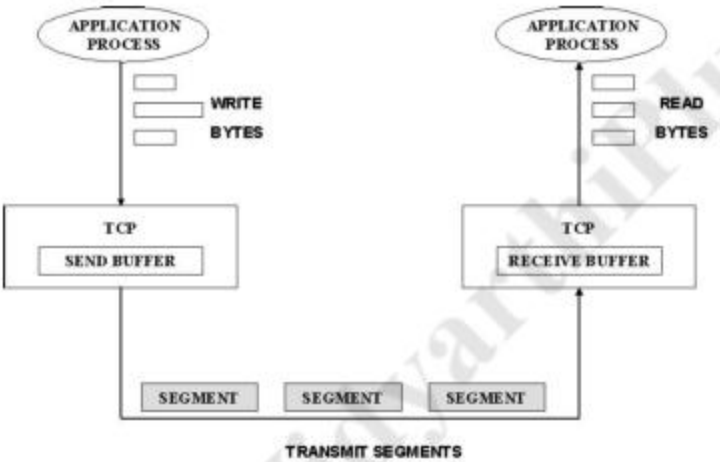
Because the transmitting side of a directly connected link cannot send any faster than the bandwidth of the link allows, and only one host is pumping data into the link, it is not possible to unknowingly congest the link. Said another way, the load on the link is visible in the form of a queue of packets 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:



TCP is a byte oriented protocol, which means that the sender writes bytes into a TCP connection and the receiver reads bytes out of the TCP connection. Although

"byte stream" describes the service TCP offers to application processes, TCP does not itself transmit individual bytes over the internet. Instead, TCP on the source host buffers enough bytes from the sending process to fill a reasonably sized packet and then sends this packet to its peer on the destination host. TCP on the destination host then empties the contents of the packet into a receiving process reads from this buffer at its leisure.



The packets exchanged between TCP peers are called segments, since each one carries a segment of the byte stream. The SrcPort and Distorts fields identify the source and destination ports, respectively, just as in UDP. These two fields, plus the source and destination IP addresses, combine to uniquely identify each TCP connection. That is,

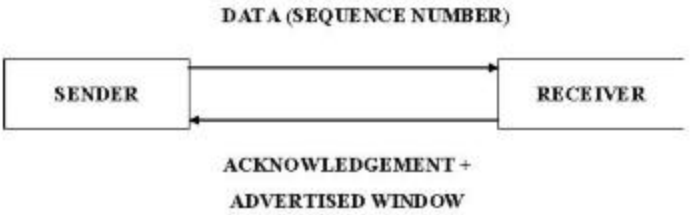
TCP"s demux key is given by the 4-tuple

(SrcPort, SrclPAddr, DstPort, DstlPAddr)

The *acknowledgement, sequence num and advertised window* fields are all involved in TCP"s sliding window algorithm. Because TCP is a byte oriented protocol, each byte of data

has a sequence number, the *sequence num* field contains the sequence

number for the first byte of data carried in that segment. The *acknowledgement and advertisement window* values flowing in the opposite direction.



The 6-bit flags field is used to relay control information between TCP peers. The possible flags include SYN, FIN, RESET, PUSH, URG, and ACK. The SYN and FIN flags are used when establishing and terminating a TCP connection, respectively. The ACK flag is set any time the *Acknowledgement* field is valid, implying that the receiver should pay attention to it. The URG flag signifies that this segment contains urgent data. When this flag is set, the UrgPtr bytes into the segment. The PUSH flag signifies that the sender invoked the push operation which indicates to the receiving side of TCP that it should notify the receiving process of this fact.

The RESET flag signifies that the receiver has become confused for example, because it received a segment it did not except to receive and so wants to abort the connection.

38.CONNECTION MANAGEMENT

A TCP connection begins with a client doing an active open to a server. Assuming that the server had earlier done a passive open, the two sides engage in an exchange of messages to establish the connection. Only after this connection establishment phase is over do the two sides begin sending data. Likewise, as soon as a participant is done sending data, it closes one direction of the connection, which causes TCP to initiate a round of connection termination messages.

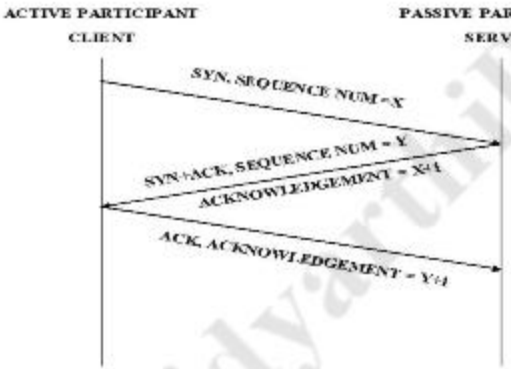
Connection setup is an asymmetric activity (one side does a passive open and the other side does an active open) connection teardown is symmetric (each side has to close the

connection independently). Therefore it is possible for one side to have done a close, meaning that it can no longer send data but for the other side to keep the other half of the bidirectional connection opens and to continue sending data.

THREE WAY HANDSHAKES:

The algorithm used by TCP to establish and terminate a connection is called a *three way handshake*. The client (the active participant) sends a segment to the server(the passive participation) stating the initial sequence number it plans to use(*flag =SYN,SequenceNum =x).*

The server then responds with a single segment that both acknowledges the client"s sequence number (Flags =ACK, Ack=x+1) and states its own beginning sequence number (*Flags=SYN, SequenceNum=y).*



That is, both the SYN and ACK bits are set in the Flags field of this second message. Finally, the client responds with a third segment that acknowledges the server"s sequence number Flags =ACK, Ack=y+1).

The "three-way handshake" is the procedure used to establish a connection. This procedure normally is initiated by one TCP and responded to by another TCP. The procedure also works if two TCP simultaneously initiate the procedure. When simultaneous attempt occurs, each TCP receives a "SYN" segment which carries no acknowledgment after it has sent a "SYN". Of course, the arrival of an old duplicate "SYN" segment can potentially make it appear, to the recipient, that a simultaneous connection initiation is in progress. Proper use of "reset" segments can disambiguate these cases.

The three-way handshake reduces the possibility of false connections. It is the implementation of a trade-off between memory and messages to provide information for this checking.

The simplest three-way handshake is shown in figure below. The figures should be interpreted in the following way. Each line is numbered for reference purposes. Right arrows (-->) indicate departure of a TCP segment from TCP A to TCP B, or arrival of a segment at B from A. Left arrows (<--), indicate the reverse. Ellipsis (...) indicates a segment which is still in the network (delayed). TCP states represent the state AFTER the departure or arrival of the segment (whose contents are shown in the center of each line). Segment contents are shown in abbreviated form, with sequence number, control flags, and ACK field. Other fields such as window, addresses, lengths, and text have been left out in the interest of clarity.

TCP A TCP B

1. CLOSED LISTEN
2. SYN-SENT --> <SEQ=100><CTL=SYN> --> SYN-RECEIVED

3. ESTABLISHED <-- <SEQ=300><ACK=101><CTL=SYN,ACK> <-- SYN-RECEIVED

4. ESTABLISHED --> <SEQ=101><ACK=301><CTL=ACK> --> ESTABLISHED

5. ESTABLISHED --> <SEQ=101><ACK=301><CTL=ACK><DATA> --> ESTABLISHED

Basic 3-Way Handshake for Connection Synchronisation

In line 2 of above figure, TCP A begins by sending a SYN segment indicating that it will use sequence numbers starting with sequence number 100. In line 3, TCP B sends a SYN and acknowledges the SYN it received from TCP A. Note that the acknowledgment field indicates TCP B is now expecting to hear sequence 101, acknowledging the SYN which occupied sequence 100.

At line 4, TCP A responds with an empty segment containing an ACK for TCP B's SYN; and in line 5, TCP A sends some data. Note that the sequence number of the segment in line 5 is the same as in line 4 because the ACK does not occupy sequence number space (if it did, we would wind up ACKing ACK's!).

39.FLOW CONTROL

TCP uses Sliding Window mechanism at octet level. The window size can be variable over time. This is achieved by utilizing the concept of "Window Advertisement" based on :

1. **Buffer availabilty at the receiver**
2. **Network conditions (traffic load etc.)**

In the former case receiver varies its window size depending upon the space available in its buffers. The window is referred as RECEIVE WINDOW (Recv_Win). When receiver buffer begin to fill it advertises a small Recv_Win so that the sender does'nt send more data than it can accept. If all buffers are full receiver sends a "Zero" size advertisement. It stops all transmission. When buffers become available receiver advertises a Non Zero widow to resume retransmission. The sender also periodically probes the "Zero" window to avoid any deadlock if the Non Zero Window advertisement from receiver is lost. The Variable size Recv_Win provides efficient end to end flow control. The second case arises when some intermediate node (e.g. a router) controls the source to reduce transmission rate. Here

another window referred as CONGESTION WINDOW (C_Win) is utilized. Advertisement of C_Win helps to check and avoid congestion.

40.RETRANSMISSION

Following two schemes are used :

1. **Fast Retransmit**
2. **Fast Recovery**

When a source sends a segment TCP sets a timer. If this value is set too low it will result in many unnecessary treransmissions. If set too high it results in wastage of banwidth and hence lower throughput. In Fast Retransmit scheme the timer value is set fairly higher than the RTT. The sender can therefore detect segment loss before the timer expires. This scheme presumes that the sender will get repeated ACK for a lost packet.

Round Trip Time (RTT): In Internet environment the segments may travel across different intermediate networks and through multiple routers. The networks and routers may have different delays, which may vary over time. The RTT therefore is also variable. It makes difficult to set timers. TCP allows varying timers by using an adaptive retransmission algorithm. It works as follows.

1. Note the time (t1) when a segment is sent and the time (t2) when its ACK is received.
2. Compute RTT(sample) = (t 2 - t 1)
3. Again Compute RTT(new) for next segment.
4. Compute Average RTT by weighted average of old and new values of RTT
5. RTT(est) = a *RTT(old) + (1-a) * RTT (new) where 0 < a < 1

   A high value of 'a' makes the estimated RTT insensitive to changes that last for a short time and RTT relies on the history of the network. A low value makes it sensitive to current state of the network. A typical value of 'a' is 0.75
6. Compute Time Out = b * RTT(est) where b> 1 A low value of 'b' will ensure quick detection of a packet loss. Any small delay will however cause unnecessary retransmission. A typical value of 'b' is kept at .2

41.TCP CONGESTION CONTROL

The internet was suffering from congestion collapse-hosts would send their packets into the internet fast as the advertised window would allow, congestion would occur at some router(causing packets to be dropped), & the hosts would time to out & retransmits their packets, resulting in even more congestion.

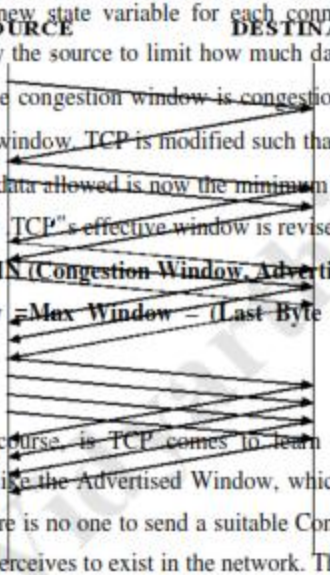
The idea of TCP congestion control is for each source to determine how much capacity is available in the network, so it knows how many packets it can safely have in

transit. Once a given source has this many packets in their transit, it uses the arrival of an ACK as a signal that one of its packets has the left the network, & that it is therefore safe to insert a new packet into the network

without adding to the level of congestion. By using ACKs to pace the transmission of packets, TCP is said to self-clocking.

ADDITIVE INCREASE/MULTIPLICATIVE DECREASE:

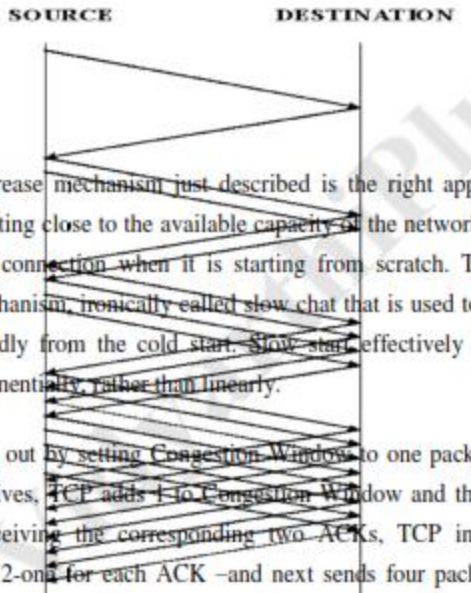
TCP maintains a new state variable for each connection, called Congestion Window, which is used by the source to limit how much data it is allowed to have in transit at a given time .The congestion window is congestion control"s counterpart to flow control"s advertised window. TCP is modified such that the maximum number of bytes of unacknowledged data allowed is now the minimum of the congestion window and the advertised window. TCP"s effective window is revised as follows:



Max Window =MIN (Congestion Window, Advertised Window)

Effective Window =Max Window – (Last Byte Sent- Last Byte Acked)

The problem, of course, is TCP comes to learn an appropriate value for Congestion Window . Unlike the Advertised Window, which is sent by the receiving side of the connection, there is no one to send a suitable Congestion Window based on the level of congestion it perceives to exist in the network. This involves decreasing the congestion goes up & increasing the congestion window when the level of congestion goes down. Taken together, the mechanism is commonly called additive increase/multiplicative decrease (AIMD);



SLOWSTART:

The additive increase mechanism just described is the right approach to use when the source is operating close to the available capacity of the network, but it takes too long to ramp up a connection when it is starting from scratch. TCP therefore provides the second mechanism, ironically called slow chat that is used to increase the congestion window rapidly from the cold start. Slow start effectively increases the congestion window exponentially, rather than linearly.

The source starts out by setting Congestion Window to one packet. When the ACK for this packet arrives, TCP adds 1 to Congestion Window and then sends two packets. Upon their receiving the corresponding two ACKs, TCP increments the Congestion Window by 2-one for each ACK –and next sends four packets. The end result is that TCP effectively doubles the number of packets it has in transit every RTT.

There are actually two different situations in which slow start runs. The first is at the very beginning of a connection, at which time the source has no idea how many packets it is going to be able to have in transit at and given time. In this situation, slow start continues to double Congestion Window each RTT until there is and loss, at which time a timeout cause"s multiplicative decrease to divide Congestion Window by 2.

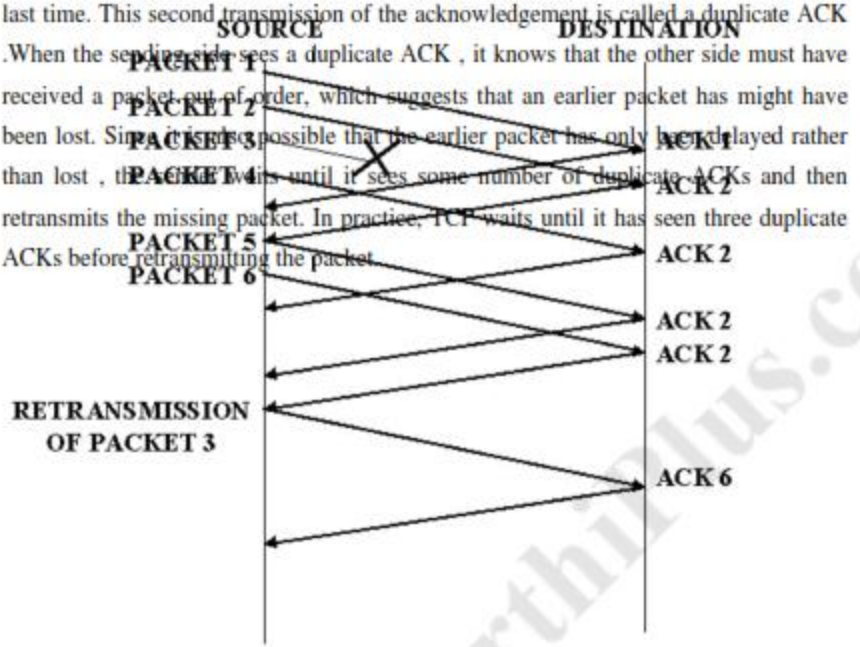
The second situation in which slow start is a bit more subtle; it occurs when the connection goes dead while waiting for a timeout to occur.

Recall how TCP „s sliding window algorithm works – when the packet is lost, the source eventually reaches a point where it is sent as much data as the advertised window allows, and so it blocks while waiting for an ACK that will not arrive. Eventually, a timeout happens, but by this time there are no packets in transit; meaning that the source will receive no ACKs to "clock" the transmission of new packets. The source will instead receive a single commutative ACK that reopens the entire advertised window, but as explained above, the source then uses slow start to restart the flow of data rather than dumping a whole window"s worth of data on the network all at once.

FAST RETRANSMIT AND FAST RECOVERY:

The idea of fast retransmit is straight forward. Every time a data packet arrives at the receiving side, the receiver responds with an acknowledgement, even if this sequence number has already been acknowledged. Thus, when a packet arrives out of order – that is, TCP cannot yet acknowledge the data the packet contains because earlier data has not yet arrived – TCP resends the same acknowledgement it sent the

last time. This second transmission of the acknowledgement is called a duplicate ACK .When the sending side sees a duplicate ACK , it knows that the other side must have received a packet out of order, which suggests that an earlier packet has might have been lost. Since it is also possible that the earlier packet has only been delayed rather than lost , the sender waits until it sees some number of duplicate ACKs and then retransmits the missing packet. In practice, TCP waits until it has seen three duplicate ACKs before retransmitting the packet.



42. CONGESTION AVOIDANCE –DECBit

TCP repeatedly increases the load it imposes on the network in an effort to find the point at which Congestion occurs, and then it backs off from this point. Said another way, TCP *needs* to create losses to find the available bandwidth of the connection. An appealing alternative, but one that has not yet been widely adopted, is to predict when congestion is about to happen and then to reduce the rate at which hosts send data just before packets start being discarded. We call such a strategy *congestion avoidance*, to distinguish it from *congestion control*.

This section describes three different congestion-avoidance mechanisms. The first two take a similar approach: They put a small amount of additional functionality into the router to assist the end node in the anticipation of congestion. The third mechanism is very different from the first two: It attempts to avoid congestion purely from the end nodes.

The first mechanism was developed for use on the Digital Network Architecture (DNA),a connectionless network with a connection-oriented transport protocol. This mechanism could, therefore, also be applied to TCP and IP. This notification is implemented by setting a binary congestion bit in the packets that flow through the router; hence the name DECbit. The destination host then copies this congestion bit into the ACK it sends backto the source. Finally, the source adjusts its sending rate so as to avoid congestion.

A single congestion bit is added to the packet header. A router sets this bit in a packet if its Average queue length is greater than or equal to 1 at the time the packet arrives. This average queue length is measured over a time interval that spans the last busy+idle cycle, plus the current busy cycle. (The router is *busy* when it is transmitting and *idle* when it is not.) Figure 6.14 shows the queue length at a router as a function of time. Essentially, the router calculates the area under the curve and divides this value by the time interval to compute the average queue length. Using a queue length of 1 as the trigger for setting the congestion bit is a trade-off between significant queuing(and hence higher throughput) and increased idle time (and hence lower delay). In other words, a queue length of 1 seems to optimize the power function.

43.RANDOM EARLY DETECTION (RED)

A second mechanism, called *random early detection (RED)*, is similar to the DECbit scheme in that each router is programmed to monitor its own queue length, and when it detects that congestion is imminent, to notify the source to adjust its congestion window. RED, invented by Sally Floyd and Van Jacobson in the early 1990s, differs from the DECbit scheme in two major ways.

The first is that rather than explicitly sending a congestion notification message to the source, RED is most commonly implemented such that it *implicitly* notifies the source of congestion by dropping one of its packets. The source is, therefore, effectively notified by the subsequent timeout or duplicate ACK. In case you haven't already guessed, RED is designed to be used in conjunction with TCP, which currently detects congestion by means of timeouts (or some other means of detecting packet loss such asduplicate ACKs). As the "early" part of the RED acronym suggests, the gateway drops the packet earlier than it would have to, so as to notify the source that it should decrease its congestion window sooner than it would normally have.

44.QUALITY OF SERVICES (QoS):

Network should support multimedia applications that are those combine audio, video, and data. For that it should provide sufficient bandwidth. The timeliness of delivery can be very important. The applications that is sensitive to the timeliness of data as real time applications. The data should be delivered correctly. A network that can provide these different levels of services is often said to be support quality of services.

45.APPLICATION REQUIREMENTS:

Applications are divided into two classes. They are

- real time
- non real time – they are called as traditional data applications. Since they have traditionally been the major applications found on data networks. Examples are, Telnet, FTP, email, web browsing etc.

TAXONOMY OF REAL TIME APPLICATIONS:

The characteristics used to categorize the applications are,

1. tolerance of loss of data
2. adaptability

APPROACHES TO QoS SUPPORT:

The approaches are divided into two broad categories. They are,

1. Fine-grained approaches, which provide QoS to individual applications of flows.
2. Coarse-grained approaches, which provides QoS to large class of data or aggregated traffic.

In the first category, integrated services are used and in the second category differentiated services are used.

INTEGRATED SERVICES (RSVP)

The term "Integrated Services" refers to a body of work that was produced by the

IETF around 1995-97.The IntServ working group developed the specifications of a number of service classes designed to meet the needs of some of the application types described above. It also defined how RSVP could be used to make reservations using these service classes.

SERVICE CLASSES:

One of the service classes is designed for intolerant applications. These applications require that a packet never arrive late. The network should guarantee that the maximum delay that any packet will experience has some specified value; the application can then set its playback point so that no packet will ever arrive after its playback time.

The aim of the controlled load service is to emulate a lightly loaded network for those applications that request service, even though the network as a whole may in fact be heavily loaded. The trick to this is to use a queuing mechanism such as WFQ to isolate

the controlled load traffic from the other traffic and some form of admission control to limit the total amount of controlled load traffic on a link such that the load is kept reasonably low.

OVERVIEW OFMECHANISMS:

The set of information that we provide to the network is referred to as a flow spec. When we ask the network to provide us with a particular service, the network needs

to decide if it can in fact provide that service.

The process of deciding when it says no is called admission control. We need a mechanism by which the users of the network and the components of the network itself exchange the information such requests for service, flow specs, and admission control decisions. This is called signaling in the ATM world, but since this word has several meanings, we refer to this process as resource reservation, and it is achieved using a Resource Reservation Protocol.

When flows and their requirements have been described, and admission control decisions have been made, the network switches and routers need to meet the requirements of flows. A key part of meeting these requirements is managing the way packets are queued and scheduled for transmission in the switches and routers. This last mechanism is packet scheduling.

FLOWSPECS:

There are two separable parts to the flow spec: the part that describes the flow"s traffic characteristics and the part that describes the service requested from the network. The

RSpec is very service specific and relatively easy to describe.

The TSpec is a little more complicated.

ADMISSION CONTROL:

When some new flow wants to receive a particular level of service, admission control looks at the TSpec and RSpec of the flow and tries to decide if the desired service can be provided to that amount of traffic, given the currently available resources, without causing any previously admitted flow to receive worse service it had requested. If it can provide the service, the flow is admitted; if not then denied. The hard part is figuring out when to say yes and when to say no.

Admission control is very dependent on the type of requested service and on the queuing discipline employed in the routers; when discuss the latter topic later in this section. For a guaranteed service, you need to have a good algorithm to make a definitive yes/no decision.

RESERVATION PROTOCOL:

While connection oriented networks have always needed some sort of setup protocol to establish the necessary virtual circuit state in the switches, connectionless networks like the internet have had no such protocols. While there have been a number of setup protocols p[proposed for the internet, the one on which most current attention is focused is called resource reservation protocol (RSVP).

The characteristics of RSVP are,

It tries to maintain the robustness by using the idea of soft state in the routers. It aims to support multicast flows just as effectively unicast flows.

PACKET CLASSIFYING AND SCHEDULING:

Once we have described our traffic and our desired network service and have installed a suitable reservation at all the routers on the path, the only thing that remains is for the routers to actually deliver the requested service to the data packets. There are two things that need to be done:

- Associate each packet with the appropriate reservation so that it can be handled correctly, a process known as **classifying packets**. It is done by examining five

fields in the packet: the source address, the destination address, protocol number, source port, destination port.

- Manage the packets in the queues so that they receive the service that has been requested, a process known as **packet scheduling**.

UNIT V APPLICATION LAYER

Traditional applications -Electronic Mail (SMTP, POP3, IMAP, MIME) – HTTP – Web Services – DNS - SNMP

46. ELECTRONIC MAIL (SMTP, POP3, MIME, IMAP)

Email is one of the oldest network applications. How email works is to (1) distinguish the user interface(i.e your mail reader) from the underlying message transfer protocol (in this case,SMTP),and(2)to distinguish between this transfer protocol and a companion protocol(RFC 822 and

Message Format

RFC 822 defines messages to have two parts: a header and a body. Both parts are represented in ASCII text. Originally, the body was assumed to be simple text. This is still the case, although RFC 822 has been augmented by MIME to allow the message body to carry all sorts of data. This data is still represented as ASCII text, but because it may be an encoded version of, say a JPEG image, it"s not necessarily readable by human users.
More on MIME in a moment.

The message header is a series of <CRLF> terminated lines.(<CRLF> stands for carriage-return + line-feed, which are a pair of ASCII control characters often used to indicate the end of a line of text.) The header is separated from the message body by a blank line. Each header line contains a type and value separated by a colon. Many of these header lines are familiar to users since they are asked to fill them out when they compose an email message. For example ,the **To**:header identifies the message recipient , and the **Subject**:header says something about the purpose of the message. Other headers are filled in by the underlying mail delivery system.Examples include **Date**: (when the message was transmitted).**From**: (what user sent the message),and **Received:** (each mail server that handled this message).There are, of course ,many other header lines;the interested reader is referred to RFC 822.

These header lines describe, in various ways ,the data being carried in the message body. They include **MIME-Version**: (the version of MIME being used),

Content-Description: (a human –readable description of what"s in the message,analogous to the **Subject:**line),**Content-Type**:the type of data contained in the message),and **Content-Transfer-Encoding**(how the message body is encoded)

The second piece is definitions for a set of content types(and subtypes).For example ,MIME defines two different still-image types, denoted image/gif and image/jpeg,each with the obvious meaning. As another example ,text/plain refers to simple text you might find in a vanilla 822-style message ,while text/richtext denotes a message that contains "marked up" text (text using special fonts , italics, etc).As a third example, MIME defines an application type , where the subtypes correspond to the output of different application programs(eg.,application/postscript and application/msword).

MIME also defines a multipart type that says how a message carrying more than one data type is structured. This is like a programming language that defines both base types(eg.,integers and floats) and compound types (eg.,. structures and arrays).One possible multipart subtype is mixed ,which says that the message contains a set of independent data pieces in a specified order. Each piece then has its own header line that describes the type of that piece.

The third piece is a way to encode the various data types so they can be shipped in an ASCII email message. The problem is that for some data types(a JPEG image, for example),any given 8-bit byte in the image might contain one of 256 different values. Only a subset of these values are valid ASCII characters .It is important that email messages contain only ASCII ,because they might pass through a number of intermediate systems(gateways ,as described below) that assume all email is ASCII and would corrupt the message if it contained non-ASCII characters .To address this issue ,MIME uses a straightforward encoding of binary data into the ASCII character.The encoding is called base64.The idea is to map every three bytes of the original binary data into four ASCII characters .This is done by grouping the binary data into 24-bit units ,and breaking each such unit into four 6-bit pieces .Each 6-bit piece maps onto one of 64 valid ASCII character;for example ,0maps onto A,1 maps onto B ,and so on.If you look at a message that has been encoded using the base 64 encoding scheme,you will notice only the 52 uppercase and lowercase letters ,the 10 digits through 0 to9 ,and the special characters + and /.These are the first 64 values in the ASCII character set.

47.SMTP –SIMPLE MAIL TRANSFER PROTOCOL

Next we look at SMTP- the protocol used to transfer messages from one host to another. To place SMTP in the right context, we need to identify the key players. First, users interact with a mail reader when they compose ,file ,search, and read their email. There are countless mail readers available ,just like there are many web browsers now include a mail reader. Second ,there is a mail daemon running on each host. You can think of this process as playing the role of a post office :mail readers give the daemon messages they want to send to others users, the daemon uses SMTP running over TCP to transmit the message into a daemon running on another machine, and the daemon puts incoming messages into the user"s mailbox. Since SMTP is a protocol that anyone could implement , in theory there could be many different implementations of the mail daemon. It runs out, though that the mail daemon running on most

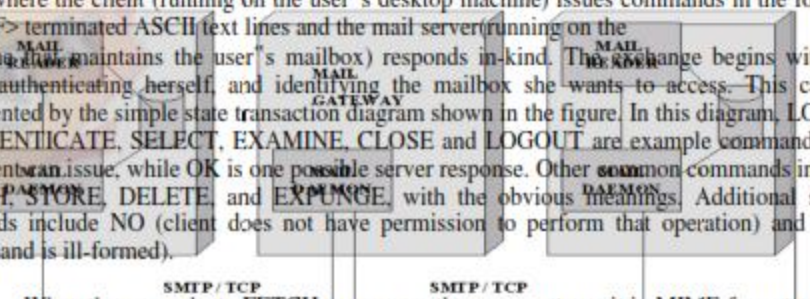
hosts is derived from the sendmail program originally implemented on berkely unix.

While it is certainly possible that the sendmail program on a sender"s machine establishes an SMTP/TCP connection to the sendmail program on the recipient"s machine, in many cases the mail traverses one or more mail gateways on its route from the sender"s host to the receiver"s host. Like the end hosts, these gateways also run a send-mail process. It"s not an accident that these intermediate nodes are called "gateways" since their job is to store and forward email messages.

Mail Reader:

The final step is for the user to actually receive her messages from the mail box, read them ,reply to them, and possibly save a copy for future reference .The user performs all the actions by interacting with a mail reader. In many cases ,this reader is just a program running on the same machine as the user"s mailbox resides, in which case it simply reads and writes the file that implements the mailbox .In other cases ,the user accesses her mailbox from a remote machine using yet another protocol, such as the Post Office Protocol(POP) or the Internet Message Access Control(IMAP).It is beyond the scope of this book to discuss the user interface aspects of the mail reader but it is definitely within our scope to talk about the access protocol. We consider IMAP, in particular.

IMAP is similar to SMTP in many ways .It is a client/server protocol running over TCP, where the client (running on the user"s desktop machine) issues commands in the form of <CRLF> terminated ASCII text lines and the mail server(running on the machine that maintains the user"s mailbox) responds in-kind. The exchange begins with the client authenticating herself, and identifying the mailbox she wants to access. This can be represented by the simple state transaction diagram shown in the figure. In this diagram, LOGIN, AUTHENTICATE, SELECT, EXAMINE, CLOSE and LOGOUT are example commands that the client can issue, while OK is one possible server response. Other common commands include FETCH, STORE, DELETE, and EXPUNGE, with the obvious meanings. Additional server responds include NO (client does not have permission to perform that operation) and BAD (command is ill-formed).



When the user asks to FETCH a message, the server returns it in MIME format and the mail reader decodes it. In addition to the message itself, IMAP also defines a set of message attributes that are exchanged as part of other commands, independent of transferring the message

itself. Message attributes include information like the size of the message, but more interestingly, various flags associated with a message, such as Seen, Answered, Deleted and Recent. These flags are used to keep the client and server synchronized, that is, when the user deletes a message in the mail reader, the client needs to report this fact to the mail server. Later, should the user decide to expunge all deleted messages, the client issues an EXPUNGE command to the server, which knows to actually remove all earlier deleted messages from the mail box.

Finally, note that when the user replies to a message,or sends a new message, the mail reader does not forward the message from the client to the mail server using IMAP ,but it instead uses SMTP .This means that the user"s mail server is effectively the first mail gateway traversed along the path from the desktop to the recipient"s mail box.

TCP/IP protocol suite specifies a standard for the exchange of mail between machines. It was derived from the (MTP) Mail Transfer Protocol. it deals with how the underlying mail delivery system passes messages across a link from one machine to another. The mail is enclosed in what is called an **envelope.** The envelope contains t he To and From fields and these are followed by the mail . The mail consists of two parts namely the Header and the Data. The Header has the To and From fields. If Headers are defined by us they should start with X. The standard headers do not start with X. In SMTP data portion can contain only printable ASCII characters The old method of sending a binary file was to send it in uuencoded form but there was no way to distinguish between the many types of binary files possible eg. .tar , .gz , .dvi etc.

48. POP3 (POST OFFICE PROTOCOL)

Here the mail person accesses the mail box from say a PC and the mail gets accumulated on a server. So in POP3 the mail is downloaded to the PC at a time interval which can be specified by the user. POP3 is used when the mail is always read from the same machine, so it helps to download the mail to it in advance.

49.IMAP(INTERMEDIATE MAIL ACCESS PROTOCOL)

Here the user may access the mail box on the server from different machines so there is no point in downloading the mail beforehand. Instead when the mail has to be read one has to log on to the server. (IMAP thus provides **authentication**) The mailbox on the server can be looked upon as a **relational database.**

50.MIME(MULTIPURPOSE INTERNET MAIL EXTENSION)

This allows the transmission of Non ASCII data through the email, MIME allows arbitrary data to be encoded in ASCII and sent in a standard email message. Each MIME message includes information that tells the recipient the type of data and the type of encoding used and this information along with the MIME version resides in the MIME header. Typical MIME header looks like,

MIME-Version: 1.0

Content-Description:

Content-Id:

Content-Type: image/gif

Content-Transfer-Encoding: base64

Content Description: contains the file name of the file that is being sent. Content -Type : is an important field that specifies the data format ie. tells what kind of data is being sent. It contains two identifiers a content type and a subtype separated by a slash. for e.g. image/gif There are 7 Content Types -

1. text
2. image
3. video
4. audio
5. application

The delivery protocols determine how the mail is transferred by the mail transfer agent to the user agent which provides an interface for reading mails.

Ensuring Network Security

1. How to ensure that nobody else reads your mail?
2. How to be sure that the mail has not been seen by someone else in your name?
3. Integrity ie. mail has not been tampered with
4. Non-Repudiability- means once I send a mail I cannot deny it, and this fact can be proved to a third person
5. Authentication

51.HTTP (HYPERTEXT TRANSFER PROTOCOL)

The Hypertext Transfer Protocol (HTTP) is a protocol used mainly to access data on the World Wide Web. The protocol transfer all data in the form of plain text, hypertext, audio, video, and so on. However it is called the hypertext transfer protocol because its efficiency allows its use in a hypertext environment where there are rapid jumps from one document to another.

HTTP functions like a combination of FTP and SMTP. It is similar to FTP because it transfers files and uses the services of TCP. However, it is much simpler than FTP because it uses only data are transferred between the client and the server.

HTTP is like SMTP because the data transferred between the client and server look like SMTP messages. In addition, the format of the messages is controlled by MIME-like headers.

However, HTTP differs from SMTP in the way the messages are sent from the client to the server and from the server to the client. Unlike SMTP, the HTTP messages are not destined to be read by humans; they are read and interpreted by the HTTP server and HTTP client (browser). SMTP messages are stored and forwarded, but HTTP messages are delivered immediately.

The idea of HTTP is very simple. A client sends a request, which looks like mail, to the server. The server sends the response, which looks like a mail reply, to the client. The request and response messages carry data in the form of a letter with MIME-like format.

The commands from the client to the server are embedded in a letter like request message. The contents of the requested file or other information are embedded in a letter like response message.

HTTP Transaction

Figure illustrates the HTTP transaction between the client and server. The client initializes the transaction by sending a request message. The server replies by sending a response.

Messages

There are two general types of HTTP messages, shown in figure request and response. Both message types follow almost the same format.

Request Messages

A request message consists of a request line, headers, and sometimes a body.

Response Message

A response message consists of a status line, headers, and sometimes a body.

Uniform Resource Locator (URL)

A client that wants to access a document needs an address. To facilitate the access of documents distributed throughout the world, HTTP uses the concept of locators. The uniform resource locator (URL) is a standard for specifying any kind of information on the Internet.

The URL defines four things:

- Method

- Host computer
- Port
- Path

The method is the protocol used to retrieve the document, for example HTTP. The host is the computer where the information is located, although the name can be an alias.
Web pages are usually stored in computers, and computers are given alias names that usually begin with the characters "www". This is not mandatory, however, as the host can be any name given to the computer that hosts the web page.

The URL optionally can contain the port number of the server. If the port is included, it should be inserted between the host and the path, and it should be separated from the host by a colon.

Path is the pathname of the file where the information is located. Note that the path can itself contain slashes that, in the UNIX operating system, separate the directories from subdirectories and files.

52.WEB SERVICES

Web Services, taking their name from the term for the individual applications that offer a remotely accessible service to client applications to form network applications.The two Web Services architectures are *SOAP* and *REST* discuss the technical meanings of those terms shortly. The SOAP architecture's approach to the problem is to make it feasible, at least in theory, to generate protocols that are customized to each network application—a kind of mass customization. The key elements of that approach are a framework for protocol specification, software toolkits for automatically generating protocol implementations from the specifications, and modular partial specifications that can be reused across protocols.

The REST architecture's approach to the problem is to regard individual Web Services as World Wide Web resources—identified by URIs and accessed via HTTP. Essentially, the REST architecture is just the web architecture. The web architecture's strengths include stability and a demonstrated scalability (in the network-size sense). It could be considered a weakness that HTTP is not well suited to the usual procedural or operation-oriented style of invoking a remote service. REST advocates argue, however, that rich services can nonetheless be exposed using a more data-oriented or documentpassing style to which HTTP is well-suited.

Custom Application Protocols (WSDL, SOAP)
The architecture informally referred to as SOAP is based on *Web Services Description Language (WSDL)* and *SOAP*.2 Both of these standards are issued by the World Wide Web Consortium (W3C). This is the architecture that people usually mean when they use the term Web Services.WSDL and SOAP are frameworks for specifying and implementing application protocols and transport protocols, respectively. They are generally used together, although

WSDL can be used to specify an application protocol that uses a transport protocol not specified using SOAP, and a SOAP-based protocol can transport a non-WSDL application protocol. WSDL is used to specify application-specific details suchas what operations are supported, the formats of the application data to invoke or respond to those operations, and whether an operation involves a response. SOAP's role is to make it easy to define a transport protocol with exactly the desired semantics regarding protocol features such as reliability and security.

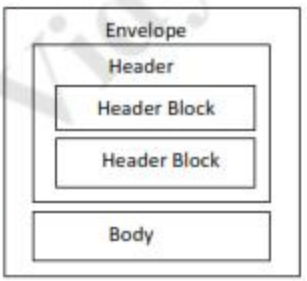
Defining Application Protocols

WSDL has chosen a procedural *operation* model of application protocols. An abstract web service interface consists of a set of named operations, each representing a simple interaction between a client and the web service. An operation is analogous to a remotely callable procedure in an RPC system. An example from W3C's WSDL Primer is a hotel reservation web service with two operations, CheckAvailability and MakeReservation.

Defining Transport Protocols

Although SOAP is often called a protocol, it is better thought of as the foundation of a family of protocols, or a framework for defining protocols. As the SOAP 1.2 specification explains, "SOAP provides a simple messaging framework whose core functionality is concerned with providing extensibility." SOAP uses many of the same strategies as WSDL, including message formats defined using XML Schema, bindings to underlying protocols, MEPs, and reusable specification elements identified using XML namespaces.

- A URI that identifies the feature;
- The state information and processing, abstractly described, that is required at each SOAP node to implement the feature;
- The information to be relayed to the next node;
- If the feature is a MEP, the life cycle and temporal/causal relationships of the messages exchanged (e.g., responses follow requests and are sent to the originator of the request).



SOAP message Structure

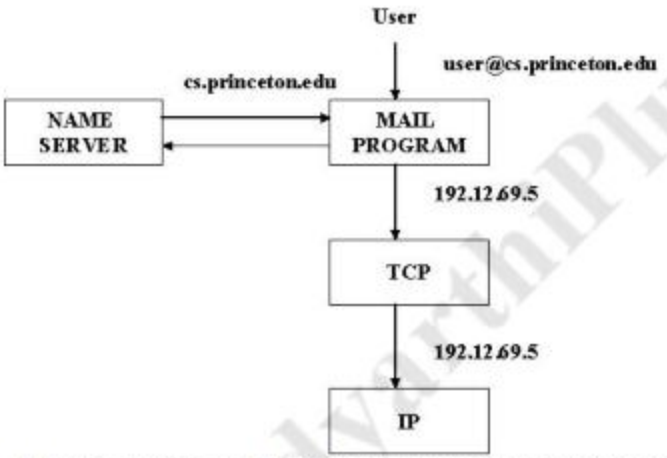
A Generic Application Protocol (REST)

The WSDL/SOAP Web Services architecture is based on the assumption that the best way to integrate applications across networks is via protocols that are customized to each application. That architecture is designed to make it practical to specify and implement all those protocols. In contrast, the REST Web Services architecture is based on the assumption that the best way to

integrate applications across networks is by applying the model underlying the World Wide Web architecture (Section 9.1.2). This model, articulated by Web architect Roy Fielding, is known as *REpresentational State Transfer(REST)*.

53.DOMAIN NAME SERVICE (DNS):

We have been using address to identify hosts. While perfectly suited for processing by routers, addresses are not exactly user friendly. It is for this reason that a unique name is also typically assigned to each host in a network.



A naming service can be developed to map user-friendly names into router-friendly addresses. Name services are sometimes called middleware because they fill a gap between applications and the underlying network.

Host names differ from host addresses in two important ways. First, they are usually of variable length and mnemonic, thereby making them easier for humans to remember. (In contrast, fixed-length numeric addresses are easier for routers to process).Second, names typically contain no information that helps the network locate (route packets toward) the host. Addresses, in contrast, sometimes have routing information embedded in them; flat addresses (those not divisible into component parts) are the exception.

A namespace defines the set of possible names. A namespace can be either flat (names are not divisible into components), or it can be hierarchical.The naming system maintains a collection of bindings of names to values. The value can be anything we want the naming system to return when presented with a name; in many cases it is an address.

A resolution mechanism is a procedure that, when invoked with a name, returns the corresponding value. A name server is a specific implementation of a resolution mechanism that is available on a network and that can be queried by sending it a message.

DNS employs a hierarchical namespace rather than a flat namespace, and the"table" of

bindings that implements this namespace is partitioned into disjoint pieces and distributed throughout the Internet. These sub tables are made available in name servers that can be queried over the network.

What happens in the Internet is that a user presents a host name to an application program, and this program encages the naming system to translate this name into a host address. The application then opens a connection to this host by presenting some transport protocol with the host"s IP address.

DOMAIN HIERARCHY:

DNS names are processed from right to left and use periods as the separator. An example domain name for a host is cicada.cs.princeton.edu.There are domains for each country, plus the "big six" domains: .edu, .com,.gov, .mil, .org, and .net.

NAME SERVERS:

The first step is to partition the hierarchy into sub trees called zones. Each zone can be thought of as corresponding to some administrative authority that is responsible for that portion of the hierarchy.

Within this zone, some departments is a zone want the responsibility of managing the hierarchy (and so they remain in the university-level zone), while others, like the Department of Computer science, manage their own department-level zone.The relevance of a zone is that it corresponds to the fundamental unit of implementation in DNS-the name server. Specifically, the information contained in each zone is implemented in two or more name servers.

Each name server, in turn, is a program that can be accessed over the Internet. Clients send queries to name servers, and name servers respond with the requested information. Sometimes the response contains the final answer that the client wants, and sometimes the response contains a pointer to another that the client should query next.

Each name server implements the zone information as a collection of resource records. In essence, a resource record is a name-to-value binding, or more specifically, a 5-tuple that contains the following fields:

< Name, Value, Type, Class, TTL >

The Name and Value fields are exactly what you would expect, while the Type field specifies how the Value should be interpreted. For example, Type=A indicates that the Value is in IP address. Thus, A records implement the name-to-address mapping we have been assuming. Other record types include

- NS: The Value field gives the domain name for a host is running a name server that knows how to resolve names within the specified domain.

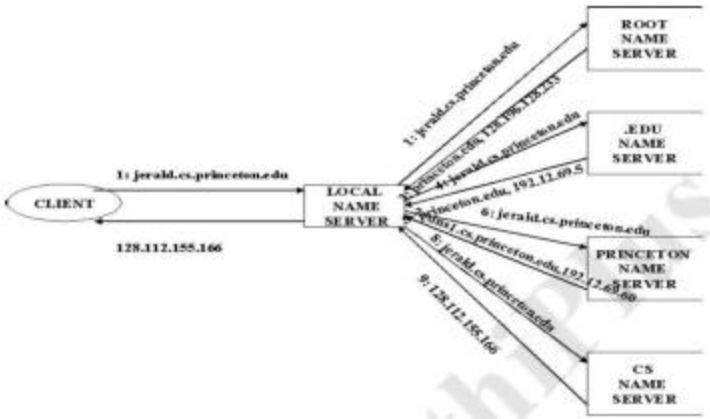
- CNAME: the Value field gives the canonical name for a particular host; it is used to define aliases.

- MX: The Value field gives the domain name for a host that is running a mail server that accepts the messages for the specified domain.

The Class field was included to allow entities other than the NIC to define useful record types. To date, the only widely used Class is the one used by the Internet; it is denoted

IN. Finally, the TTL field shows how long this resource record is valid. It is used by servers that cache resource records from other servers; when the TTL expires, the server must evict the record from its cache.

NAME RESOLUTION



54.SNMP(SIMPLE NETWORK MANAGEMENT PROTOCOL)

A large network can often get into various kinds of trouble due to routers (dropping too many packets), hosts(going down) etc. One has to keep track of all these occurence and adapt to such situations. A protocol has been defined. Under this scheme all entities in the network belong to 4 classes:

1. Managed Nodes
2. Management Stations
3. Management Information (called Object)
4. A management protocol

The managed nodes can be hosts,routers,bridges,printers or any other device capable of communicating status information to others. To be managed directly by SNMP, a node must be capable of running am SNMP management process, called SNMP agent. Network management is done by management stations by exchanging information with the nodes. These are basically general purpose computers running special management software. The management stations polls the stations periodically. Since SNMP uses unreliable service of UDP the polling is

essential to keep in touch with the nodes. Often the nodes send a trap message indicating that it is going to go down. The management stations then periodically checks (with an increased frequency) . This type of polling is called trap directed polling. Often a group of nodes are represented by a single node which communicates with the management stations. This type of node is called proxy agent. The proxy agent can also server as a security arrangement. All the variables in these schemes are called Objects. Each variable can be referenced by a specific addressing scheme adopted by this system. The entire collection of all objects is called Management Information Base (MIB). The addressing is hierarchical as seen in the picture.
Internet is addressed as 1.3.61. All the objects under this domain has this string at the beginning. The information are exchanged in a standard and vendor-neutral way . All the data are represented in Abstract Syntax Notation 1 (ASN.1). It is similar to XDR as in RPC but it have widely different representation scheme. A part of it actually adopted in SNMP and modified to form Structure Of Information Base. The Protocol specifies various kinds of messages that can be exchanged between the managed nodes and the management station.

Message	Description
1.Get_Request	Request the value for a variable
2. Get_Response	Returns the value of the variable asked for
3.Get_Next_Request	Request a variable next to the previous one
4.Set_Request	Set the value of an Object.
5.Trap	Agent to manager Trap report
6.Get_bulk_request	Request a set of variable of same type
7.Inform_Request	Exchange of MIB among Management stations

The last two options have been actually added in the SNMPv2. The fourth option need some kind of authentication from the management station.

Addressing Example :

Following is an Example of the kind of address one can refer to when fetching a value in the table :-

(20) IP-Addr-Table = Sequence of IPAddr-Entry (1)

IPAddrEntry = SEQUENCE {
IPADDENTRYADDR : IPADDR (1)

Index : integer (2)
Netmask : IPAddr (3) }

So when accessing the netmask of some IP-entity the variable name would be :
1.3.6.1.2.4.20 .1.3.key-value

Here since Ip-address the unique key to index any member of the array the address can be like :- 1.3.6.1.2.4.20.1.3.128.10.2.3

GLOSSARY

3DES: Triple DES, a version of DES that uses three keys, effectively increasing the key size and robustness of the encryption.
3G: Third-generation mobile wireless, a class of cellular wireless technologies based on CDMA.
4B/5B: A type of bit-encoding scheme used in FDDI, in which every 4 bits of data are transmitted as a 5-bit sequence.
802.3: IEEE Ethernet standard.
802.5: IEEE token ring standard.
802.11: IEEE wireless network standard.
802.17 IEEE resilient packet ring standard.
822: Refers to RFC 822, which defines the format of Internet email messages. See *SMTP*.
AAL: ATM Adaptation Layer. A protocol layer, configured over ATM. Two AALs are defined for data communications, AAL3/4 and AAL5. Each protocol layer provides a mechanism to segment large packets into cells at the sender and to reassemble the cells back together at the receiver.
ABR: (1) Available bit rate. A rate-based congestion-control scheme being developed for use on ATM networks. ABR is intended to allow a source to increase or decrease its allotted rate, based on feedback from switches within the network. Contrast with *CBR*, *UBR*, and *VBR*. (2) Area border router. Router at the edge of an *area* in a link-state protocol.
ACK: An abbreviation for *acknowledgment*. An acknowledgment is sent by a receiver of data to indicate to the sender that the data transmission was successful.
additive increase/multiplicative decrease: Congestion window strategy used by TCP. TCP opens the congestion window at a linear rate, but halves it when losses are experi enced due to congestion. It has been shown that additive increase/multiplicative decrease is a necessary condition for a congestion-control mechanism to be stable.
AES: Advanced Encryption Standard. A cryptographic cipher that has been proposed to supersede DES.
AF: Assured forwarding. One of the per-hop behaviors proposed for Differentiated Services.
ALF: Application Level Framing. A protocol design principle that says that application programs better understand their communication needs than do general-purpose transport protocols.
AMPS: Advanced mobile phone system. Analog-based cell phone system. Currently being replaced by digital system, known as PCS.
ANSI: American National Standards Institute. Private U.S. standardization body that commonly participates in the ISO standardization process. Responsible for SONET.
API: Application programming interface. Interface that application programs use to access the network subsystem (usually the transport protocol). Usually OS-specific. The socket API from Berkeley Unix is a widely used example.
area: In the context of link-state routing, a collection of adjacent routers that share full routing information with each other. A routing domain is divided into areas to improve

scalability.
ARP: Address Resolution Protocol. Protocol of the Internet architecture, used to translate high-level protocol addresses into physical hardware addresses. Commonly used on the Internet to map IP addresses into Ethernet addresses.
ARPA: Advanced Research Projects Agency. One of the research and development organizations
within the Department of Defense. Responsible for funding the ARPANET as well as the research that led to the development of the TCP/IP Internet. Also known as DARPA, the *D* standing for Defense.
ARPANET: An experimental wide-area packet-switched network funded by ARPA and begun in the late 1960s, which became the backbone of the developing Internet.
ARQ: Automatic repeat request. General strategy for reliably sending packets over an unreliable link. If the sender does not receive an ACK for a packet after a certain time period, it assumes that the packet did not arrive (or was delivered with bit errors) and retransmits it. Stop-and-wait and sliding window are two example ARQ protocols. Contrast with *FEC*.

ASN.1: Abstract Syntax Notation One. In conjunction with BER, a presentationformatting standard devised by the ISO as part of the OSI architecture.
ATM: Asynchronous transfer mode. A connection-oriented network technology that uses small, fixed-size packets (called *cells*) to carry data.
ATMARP: Address Resolution Protocol as enhanced for ATM networks.
ATM Forum: A key ATM standards-setting body.
authentication: Security protocol by which two suspicious parties prove to each other that they are who they claim to be.
autonomous system (AS): A group of networks and routers, subject to a common authority and using the same intradomain routing protocol.
bandwidth: A measure of the capacity of a link or connection, usually given in units of bits per second.
Bellman-Ford: A name for the distance-vector routing algorithm, from the names of the inventors.
BER: Basic encoding rules. Rules for encoding data types defined by ASN.1.
best-effort delivery: The service model of the current Internet architecture. Delivery of a message is attempted but is not guaranteed.
BGP: Border Gateway Protocol. An interdomain routing protocol by which autonomous systems exchange reachability information. The most recent version is BGP-4.
BISYNC: Binary Synchronous Communication. A byte-oriented link-level protocol developed in the late 1960s by IBM.
bit stuffing: A technique used to distinguish control sequences and data on the bit level. Used by the HDLC protocol.
block: An OS term used to describe a situation in which a process suspends execution while awaiting some event, such as a change in the state of a *semaphore*.
Bluetooth: A short-range wireless standard used to connect computers, mobile phones, and peripheral devices, among other things.
bridge: A device that forwards link-level frames from one physical network to another, sometimes called a LAN switch. Contrast with *repeater* and *router*.

broadcast: A method of delivering a packet to every host on a particular network or internet. May be implemented in hardware (e.g., Ethernet) or software (e.g., IP broadcast).
CA: Certification authority (also known as certificate authority). An entity that signs security certificates, thereby promising that the public key contained in the certificate belongs to the entity named in the certificate.
CBC: Cipher block chaining. A cryptographic mode in which each plaintext block is XORed with the previous block of ciphertext before encryption.
CBR: Constant bit rate. A class of service in ATM that guarantees transmission of data at a constant bit rate, thus emulating a dedicated transmission link. Contrast with *ABR*, *UBR*, and *VBR*.
CCITT: The now defunct *Comité Consultif International de Telegraphique et Telephonique*, a unit of the International Telecommunications Union (ITU) of the United Nations. Now replaced by ITU-T.
CDMA: Code Division Multiple Access, a form of multiplexing used in wireless networks.
CDN: Content distribution network. A collection of surrogate web servers, distributed across the Internet, that respond to web HTTP requests in place of the server. The goal of widely distributing the surrogate servers is to have a surrogate close to the client, making it possible to respond to requests more quickly.
cell: A 53-byte ATM packet, capable of carrying up to 48 bytes of data.
certificate: A document digitally signed by one entity that contains the name and public key of another entity. Used to distribute public keys. Also see *CA*.
channel: A generic communication term used in this book to denote a logical processto-process connection.
checksum: Typically a ones complement sum over some or all of the bytes of a packet, computed and appended to the packet by the sender. The receiver recomputes the checksum and compares it to the one carried in the message. Checksums are used to detect errors in a packet and may also be used to verify that the packet has been delivered to the correct host. The term *checksum* is also sometimes (imprecisely) used to refer generically to error-detecting codes.
chipping code: Random sequence of bits that is XORed with the data stream to implement the direct sequence technique of spread spectrum.

CIDR: Classless interdomain routing. A method of aggregating routes that treats a block of contiguous Class C IP addresses as a single network.
circuit switching: A general strategy for switching data through a network. It involves establishing a dedicated path (circuit) between the source and destination. Contrast with *packet switching*.
client: The requester of a service in a distributed system.
CLNP: Connectionless Network Protocol. The ISO counterpart to the Internet's IP.
clock recovery: The process of deriving a valid clock from a serially transmitted digital signal.
concurrent logical channels: Multiplexing several stop-and-wait logical channels onto a single point-to-point link. No delivery order is enforced. This mechanism was used by the IMP-IMP protocol of the ARPANET.
congestion: A state resulting from too many packets contending for limited resources (e.g., link bandwidth and buffer space on routers or switches), which may force the

router (switch) to discard packets.
congestion control: Any network resource management strategy that has, as its goal, the alleviation or avoidance of congestion. A congestion-control mechanism may be implemented on the routers (switches) inside the network, by the hosts at the edges of the network, or by a combination of both.
connection: In general, a channel that must be established prior to use (e.g., by the transmission of some setup information). For example, TCP provides a connection abstraction that offers reliable, ordered delivery of a byte stream. Connection-oriented networks, such as ATM, are often said to provide a *virtual circuit* abstraction.
connectionless protocol: A protocol in which data may be sent without any advance setup. IP is an example of such a protocol.
context switch: An operation in which an operating system suspends the execution of one process and begins the execution of another. A context switch involves saving the state of the former process (e.g., the contents of all registers) and loading the state of the latter process.
controlled load: One of the service classes available in the Internet's Integrated Services architecture.

CRC: Cyclic redundancy check. An error-detecting code computed over the bytes composing a packet and then appended to the packet by the network hardware (e.g., Ethernet adaptor). CRC provides stronger error detection than a simple checksum.
crossbar switch: A simple switch design in which every input is directly connected to every output and the output port is responsible for resolving contention.
CSMA/CD: Carrier Sense Multiple Access with Collision Detect. CSMA/CD is a functionality of network hardware. "Carrier sense multiple access" means that multiple stations can listen to the link and detect when it is in use or idle; "collision detect" indicates that if two or more stations are transmitting on the link simultaneously, they will detect the collision of their signals. Ethernet is the best-known technology that uses CSMA/CD.
cut-through: A form of switching or forwarding in which a packet starts to be transferred to an output before it has been completely received by the switching node, thus reducing latency through the node.
datagram: The basic transmission unit in the Internet architecture. A datagram contains all of the information needed to deliver it to its destination, analogous to a letter in the U.S. postal system. Datagram networks are connectionless.
DCE: Distributed Computing Environment. An RPC-based suite of protocols and standards that support distributed computing. Defined by OSF.
DDCMP: DigitalData CommunicationMessage Protocol. A byte-oriented link-level protocol used in Digital Equipment Corporation's DECNET.
DDoS: Distributed denial of service. A DoS attack in which the attack originates at a set of nodes. Each attacking node may put only a marginal load on the target machine, but the aggregate load from all the attacking nodes swamps the target machine.
DECbit: A congestion-control scheme in which routers notify the endpoints of imminent congestion by setting a bit in the header of routed packets. The endpoints decrease their sending rates when a certain percentage of received packets have the bit set.
decryption: The act of reversing an *encryption* process to recover the data from an encrypted message.

delay bandwidth product: The product of a network's RTT and bandwidth. Gives a measure of how much data can be in transit on the network.
demultiplexing: Using information contained in a packet header to direct it upward through a protocol stack. For example, IP uses the ProtNum field in the IP header to decide which higher protocol (i.e., TCP, UDP) a packet belongs to, and TCP uses the port number to demultiplex a TCP packet to the correct application process. Contrast with *multiplexing*.
demultiplexing key: A field in a packet header that enables demultiplexing to take place (e.g., the ProtNum field of IP).
dense mode multicast: PIM mode used when most routers or hosts need to receive multicast packets.
DES: Data Encryption Standard. An algorithm for data encryption based on a 64-bit secret key.
DHCP: Dynamic Host Configuration Protocol. A protocol used by a host as it boots or when it is connected to a network, to learn various network information, such as its IP address.
DHT: Distributed hash table. A technique by which a message is routed toward a machine that supports a particular object, based on the object's name. The object is hashed to a unique identifier, with each intermediate node along the route forwarding the message to a node that is able to interpret a larger prefix of this ID. DHTs are often used in peer-to-peer networks.
Differentiated Services: A new architecture for providing better than best-effort service on the Internet. It has been proposed as an alternative to Integrated Services.
direct sequence: A spread spectrum technique that involves XORing the data stream with a random bit sequence known as a chipping code.
distance vector: A lowest-cost-path algorithm used in routing. Each node advertises reachability information and associated costs to its immediate neighbors, and uses the updates it receives to construct its forwarding table. The routing protocol RIP uses a distance-vector algorithm. Contrast with *link state*.
DMA: Direct memory access. An approach to connecting hosts to I/O devices, in which the device directly reads data from and writes data to the host's memory. Also see *PIO*.
DNA/DECNET: Digital Network Architecture. An OSI-based architecture that supports a connectionless network model and a connection-oriented transport protocol.
DNS: Domain name system. The distributed naming system of the Internet, used to resolve host names (e.g., cicada.cs.princeton.edu) into IP addresses (e.g., 192.12.69.35). The DNS is implemented by a hierarchy of name servers.

domain: Can refer either to a context in the hierarchical DNS namespace (e.g., the "edu" domain) or to a region of the Internet that is treated as a single entity for the purpose of hierarchical routing. The latter is equivalent to *autonomous system*.
DoS: Denial of service. A situation in which an attacking node floods a target node with so much work (so many packets) that it effectively keeps legitimate users from accessing the node, hence, they are denied service.
DS3: A 44.7-Mbps transmission link service offered by the phone company. Also called T3.
DSL: Digital subscriber line. A family of standards for transmitting data over twisted pair

telephone lines at multimegabit-per-second speeds.
duplicate ACK: A retransmission of a TCP acknowledgment. The duplicate ACK does not acknowledge any new data. The receipt of multiple duplicate ACKs triggers the TCP *fast retransmit* mechanism.
DVMRP: Distance Vector Multicast Routing Protocol. Multicast routing protocol originally used in the MBone.
DWDM: Dense wavelength division multiplexing. Multiplexing multiple light waves (colors) onto a single physical fiber. The technique is "dense" in the sense that a large number of optical wavelengths can be supported.
ECN: Explicit congestion notification. A technique by which routers inform end hosts about congestion by setting a flag in packets they are forwarding. Used in conjunction with active queue management algorithms like RED.
EF: Expedited forwarding. One of the per-hop behaviors proposed for Differentiated Services.
EGP: Exterior Gateway Protocol. An early interdomain routing protocol of the Internet, which was used by exterior gateways (routers) of autonomous systems to exchange routing information with other ASs. Replaced by BGP.
encapsulation: The operation, performed by a lower-level protocol, of attaching a protocol-specific header and/or trailer to a message passed down by a higher-level protocol. As a message travels down the protocol stack, it gathers a sequence of headers, of which the outermost corresponds to the protocol at the bottom of the stack.
encryption: The act of applying a transforming function to data, with the intention that only the receiver of the data will be able to read it (after applying the inverse function, *decryption*). Encryption generally depends on either a secret shared by the sender and receiver or on a public/private key pair.
Ethernet: A popular local area network technology that uses CSMA/CD and has a bandwidth of 10 Mbps. An Ethernet itself is just a passive wire; all aspects of Ethernet transmission are completely implemented by the host adaptors.
exponential backoff: A retransmission strategy that doubles the timeout value each time a packet is retransmitted.
exposed node problem: Situation that occurs on a wireless network where two nodes receive signals from a common source, but each is able to reach other nodes that do not receive this signal.
extended LAN: A collection of LANs connected by bridges.
fabric: The part of a switch that actually does the switching, that is, moves packets from input to output. Contrast with *port*.
fair queuing (FQ): A round-robin-based queuing algorithm that prevents a badly behaved process from capturing an arbitrarily large portion of the network capacity.
fast retransmit: A strategy used by TCP that attempts to avoid timeouts in the presence of lost packets. TCP retransmits a segment after receiving three consecutive duplicate ACKs, acknowledging the data up to (but not including) that segment.
FDDI: Fiber Distributed Data Interface. A token ring networking technology designed to run over optical fiber.
FEC: 1 Forward error correction. A general strategy for recovering from bit errors introduced into data packets without having to retransmit the packet. Redundant information is included with each packet that can be used by the receiver to

determine which bits in a packet are incorrect. Contrast with *ARQ*.
2 Forwarding equivalence class. A set of packets that are to receive the same forwarding treatment at a router. MPLS labels are normally associated with FECs.
Fibre Channel: A bidirectional link protocol commonly used to connect computers, peripherals, and storage devices. Originally had a bandwidth of 100 MBps but since enhanced to GBps speeds.
firewall: A router that has been configured to filter (not forward) packets from certain sources. Used to enforce a security policy.

flow control: A mechanism by which the receiver of data throttles the transmission rate of the sender, so that data will not arrive too quickly to be processed. Contrast with *congestion control*.
flowspec: Specification of a flow's bandwidth and delay requirements presented to the network to establish a reservation. Used with RSVP.
forwarding: The operation performed by a router on every packet: receiving it on an input, deciding what output to send it to, and sending it there.
forwarding table: The table maintained in a router that lets it make decisions on how to forward packets. The process of building up the forwarding table is called *routing*, and thus the forwarding table is sometimes called a *routing table*. In some implementations, the routing and forwarding tables are separate data structures.
fragmentation/reassembly: A method for transmission of messages larger than the network's MTU. Messages are fragmented into small pieces by the sender and reassembled by the receiver.
frame: Another name for a packet, typically used in reference to packets sent over a single link rather than a whole network. An important problem is how the receiver detects the beginning and ending of a frame, a problem known as framing.
Frame Relay: A connection-oriented public packet-switched service offered by the phone company.
frequency hopping: A spread spectrum technique that involves transmitting data over a random sequence of frequencies.
FTP: File Transfer Protocol. The standard protocol of the Internet architecture for transferring files between hosts. Built on top of TCP.
GMPLS: Generalized MPLS. Allows IP to run natively over optically-switched networks.
GPRS: General Packet Radio Service. A packet transmission service provided by cellular wireless networks.
GSM: Global System for Mobile communication. Digital cellular phone system being deployed throughout the world (less so in the United States and Canada). Similar to PCS, which is being deployed throughout the United States and Canada.
gopher: An Internet information service.
H.323: Session control protocol often used for Internet telephony.

handle: In programming, an identifier or pointer that is used to access an object.
hardware address: The link-level address used to identify the host adaptor on the local network.
HDLC: High-Level Data Link Control protocol. An ISO-standard link-level protocol. It uses bit stuffing to solve the framing problem.
hidden node problem: Situation that occurs on a wireless network where two nodes are

sending to a common destination, but are unaware that the other exists.
hierarchical routing: A multilevel routing scheme that uses the hierarchical structure of the address space as the basis for making forwarding decisions. For example, packets might first be routed to a destination network and then to a specific host on that network.
HiPPI: High Performance Parallel Interface. An ANSI-standard network technology capable of Gbps transmission rates, typically used to connect supercomputers to peripheral devices. Used in same way as *Fibre Channel*.
host: A computer attached to one or more networks that supports users and runs application programs.
HTML: HyperText Markup Language. A language used to construct World Wide Web pages.
HTTP: HyperText Transport Protocol. An application-level protocol based on a request/reply paradigm and used in the World Wide Web. HTTP uses TCP connections to transfer data.
IAB: Internet Architecture Board. The main body that oversees the development of the Internet architecture.
IBGP: Interior BGP. The protocol used to exchange interdomain routing information among routers in the same domain.
ICMP: Internet ControlMessage Protocol. This protocol is an integral part of IP. It allows a router or destination host to communicate with the source, typically to report an error in IP datagram processing.
IEEE: Institute for Electrical and Electronics Engineers. A professional society for engineers that also defines network standards, including the 802 series of LAN standards.
IETF: Internet Engineering Task Force. The body responsible for the specification of standards and protocols related to the Internet.

IMAP: InternetMessage Access Protocol. An application layer protocol that allows a user to retrieve her email from a mail server.
IMP-IMP: A byte-oriented link-level protocol used in the original ARPANET.
Integrated Services: Usually taken to mean a packet-switched network that can effectively support both conventional computer data and real-time audio and video. Also, a name given to a proposed Internet service model that was designed to supplement the current best-effort service model.
integrity: In the context of network security, a service that ensures that a received message is the same one that was sent.
interdomain routing: The process of exchanging routing among different routing domains. BGP is an example of an interdomain protocol.
internet: A collection of (possibly heterogeneous) packet-switching networks interconnected by routers. Also called an internetwork.
Internet: The global internet based on the Internet (TCP/IP) architecture, connecting millions of hosts worldwide.
interoperability: The ability of heterogeneous hardware and multivendor software to communicate by correctly exchanging messages.
interrupt: An event (typically generated by a hardware device) that tells the operating system to stop its current activity and take some action. For example, an interrupt is used to notify the OS that a packet has arrived from the network.
intradomain routing: The exchange of routing information within a single domain or

autonomous system. RIP and OSPF are example intradomain protocols.
IP: Internet Protocol (also known as IPv4). A protocol that provides a connectionless, best-effort delivery service of datagrams across the Internet.
IPng: Internet Protocol—Next Generation (also known as IPv6). Proposed version of IP that provides a larger, more hierarchical address space and other new features.
IPSEC: IP Security. An architecture for authentication, privacy, and message integrity, among other security services to the Internet architecture.
IRTF: Internet Research Task Force. A sibling body to the IETF, responsible for charting direction in research and development for the Internet.
IS-IS: A link-state routing protocol, similar to OSPF.

ISDN: Integrated Services Digital Network. A digital communication service offered by telephone carriers and standardized by ITU-T. ISDN combines voice connection and digital data services in a single physical medium.
ISO: International Standards Organization. The international body that drafted the seven-layer OSI architecture and a suite of protocols that has not enjoyed commercial success.
ITU-T: A subcommittee of the International Telecommunications Union, a global body that drafts technical standards for all areas of international analog and digital communication. ITU-T deals with standards for telecommunications, notably ATM.
jitter: Variation in network latency. Large jitter has a negative impact on the quality of video and audio applications.
JPEG: Joint Photographic Experts Group. Typically used to refer to a widely used algorithm for compressing still images that was developed by the JPEG.
Kerberos: A TCP/IP-based authentication system developed atMIT, in which two hosts use a trusted third party to authenticate each other.
key distribution: Mechanism by which users learn each others' public keys through the exchange of digitally signed certificates.
LAN: Local area network. A network based on any physical network technology that is designed to span distances of up to a few thousand meters (e.g., Ethernet or FDDI). Contrast with *SAN, MAN,* and *WAN.*
LANE: Local area network emulation. Adding functionality to ATM to make it behave like a shared-media (i.e., Ethernet-like) LAN.
LAN switch: Another term for a *bridge*, usually applied to a bridge with many ports. Also called an Ethernet switch if the link technology it supports is Ethernet.
latency: A measure of how long it takes a single bit to propagate from one end of a link or channel to the other. Latency is measured strictly in terms of time.
LDAP: Lightweight Directory Access Protocol. A subset of the X.500 directory service that has recently become a popular directory service for information about users.
LER: Label edge router. A router at the edge of an MPLS cloud. Performs a complete IP lookup on arriving IP packets, and then applies labels to them as a result of the lookup.
link: A physical connection between two nodes of a network. It may be implemented over copper or fiber-optic cable or it may be a wireless link (e.g., a satellite).

link-level protocol: A protocol that is responsible for delivering frames over a directly connected network (e.g., an Ethernet, token ring, or point-to-point link). Also called link-layer protocol.

link state: A lowest-cost-path algorithm used in routing. Information on directly connected neighbors and current link costs are flooded to all routers; each router uses this information to build a view of the network on which to base forwarding decisions. The OSPF routing protocol uses a link-state algorithm. Contrast with *distance vector*.
LSR: Label-switching router. A router that runs IP control protocols, but uses the label switching forwarding algorithm of MPLS.
MAC: Media access control. Algorithms used to control access to shared-media networks like Ethernet and FDDI.
MACA: Multiple access with collision avoidance. Distributed algorithm used to mediate access to a shared media.
MACAW: Multiple access with collision avoidance for wireless. Enhancement of the general MACA algorithm to better support wireless networks. Used by 802.11.
MAN: Metropolitan area network. A network based on any of several new network technologies that operate at high speeds (up to several Gbps) and across distances wide enough to span a metropolitan area. Contrast with *SAN*, *LAN*, and *WAN*.
Manchester: A bit-encoding scheme that transmits the exclusive-OR of the clock and the NRZ-encoded data. Used on the Ethernet.
MBone: Multicast backbone. A logical network imposed over the top of the Internet, in which multicast-enhanced routers use tunneling to forward multicast datagrams across the Internet.
MD5: Message Digest version 5. An efficient cryptographic checksum algorithm commonly used to verify that the contents of a message are unaltered.
MIB: Management information base. Defines the set of network-related variables that may be read or written on a network node. The MIB is used in conjunctionwith SNMP.
MIME: Multipurpose InternetMail Extensions. Specifications for converting binary data (such as image files) to ASCII text, which allows it to be sent via email.
Mosaic: A once-popular and free graphical World Wide Web browser developed at the National Center for Supercomputing Applications at the University of Illinois.
MP3: MPEG Layer 3. Audio compression standard used with MPEG.

MPEG: Moving Picture Experts Group. Typically used to refer to an algorithm for compressing video streams developed by the MPEG.
MPLS: Multiprotocol Label Switching. A collection of techniques used to effectively implement IP routers on top of level 2 (e.g., ATM) switches.
MSAU: Multistation access unit. A device used in token ring networks to connect several stations to the ring and remove them in the event of failure.
MSDP: Multicast Source Discovery Protocol. A protocol used to facilitate interdomain multicast.
MTU: Maximum transmission unit. The size of the largest packet that can be sent over a physical network.
multicast: A special form of broadcast in which packets are delivered to a specified subgroup of network hosts.
multiplexing: Combining distinct channels into a single, lower-level channel. For example, separate TCP and UDP channels are multiplexed into a single host-to-host IP channel. The inverse operation, *demultiplexing*, takes place on the receiving host.
name resolution: The action of resolving host names (which are easy for humans to read) into their corresponding addresses (which machines can read). See *DNS*.

NAT: Network address translation. A technique for extending the IP address space that involves translating between globally understood IP addresses and local-only addresses at the edge of a network or site.
NDR: Network Data Representation. The data-encoding standard used in the Distributed Computing Environment (DCE), as defined by the Open Software Foundation. NDR uses a receiver-makes-right strategy and inserts an architecture tag at the front of each message.
network-level protocol: A protocol that runs over switched networks, directly above the link level.
NFS: Network File System. A popular distributed file system developed by SunMicrosystems. NFS is based on SunRPC, an RPC protocol developed by Sun.
NIST: National Institute for Standards and Technology. The official U.S. standardization body.
node: A generic term used for individual computers that make up a network. Nodes include general-purpose computers, switches, and routers.

NRZ: Nonreturn to zero. A bit-encoding scheme that encodes a 1 as the high signal and a 0 as the low signal.
NRZI: Nonreturn to zero inverted. A bit-encoding scheme that makes a transition from the current signal to encode a 1 and stays at the current signal to encode a 0.
NSF: National Science Foundation. An agency of the U.S. government that funds scientific research in the United States, including research on networks and on the Internet infrastructure.
nv: Network video. A videoconferencing application.
OC: Optical carrier. The prefix for various rates of SONET optical transmission. For example, OC-1 refers to the SONET standard for 51.84-Mbps transmission over fiber. An OC-*n* signal differs from an STS-*n* signal only in that the OC-*n* signal is scrambled for optical transmission.
ONC: Open Network Computing. A version of SunRPC that is being standardized for the Internet.
optical switch: A switching device that forwards optical lightwaves from input port to output port without converting to electrical format.
OSF: Open Software Foundation. A consortium of computer vendors that have defined standards for distributed computing, including the NDR presentation format.
OSI: Open Systems Interconnection. The seven-layer network reference model developed by the ISO. Guides the design of ISO and ITU-T protocol standards.
OSPF: Open Shortest Path First. A routing protocol developed by the IETF for the Internet architecture. OSPF is based on a *link-state* algorithm, in which every node constructs a topography of the Internet and uses it to make forwarding decisions. Today known as Open Group.
overlay: A virtual (logical) network running on top of an existing physical network. Overlay nodes communicate with each other through tunnels rather than over physical links. Overlays are often used to deploy new network services since they do not require the cooperation of the existing network infrastructure.
packet: A data unit sent over a packet-switched network. Also see *frame* and *segment*.
packet switching: A general strategy for switching data through a network. Packet switching uses store-and-forward switching of discrete data units called packets, and implies

statistical multiplexing.

participants: A generic term used to denote the processes, protocols, or hosts that are sending messages to each other.
PAWS: Protection against wrapped sequence numbers. Engineering transport protocol with a large enough sequence number space to protect against the numbers wrapping around on a network where packets can be delayed for a long period of time.
PCS: Personal Communication Services. New digital cellular phone system being deployed throughout the United States and Canada. Similar to GSM, which is being deployed throughout the rest of the world.
PDU: Protocol data unit. Another name for a packet or frame.
peer: A counterpart on another machine that a protocol module interoperates with to implement some communication service.
peer-to-peer networks: A general class of applications that integrate application logic (e.g., file storage) with routing. Popular examples include Napster and Gnutella. Research prototypes often use distributed hash tables.
PEM: Privacy Enhanced Mail. Extensions to Internet email that support privacy and integrity protection. See also *PGP.*
PGP: Pretty Good Privacy. A collection of public domain software that provides privacy and authentication capabilities using RSA and that uses a mesh of trust for public key distribution.
PHB: Per-hop behavior. Behavior of individual routers in the Differentiated Services architecture.
AF and EF are two proposed PHBs.
physical-level protocol: The lowest layer of the OSI protocol stack. Its main function is to encode bits onto the signals that are propagated across the physical transmission media.
piconet: Wireless network spanning short distances (e.g., 10m). Used to connect office computers (laptops, printers, PDAs, workstations, etc.) without cables.
PIM: Protocol Independent Multicast. A multicast routing protocol that can be built on top of different unicast routing protocols.
Ping: A Unix utility used to test the RTT to various hosts over the Internet. Ping sends an ICMP ECHO_REQUEST message, and the remote host sends an ECHO_ RESPONSE message back.

PIO: Programmed input/output. An approach to connecting hosts to I/O devices, in which the CPU reads data from and writes data to the I/O device. Also see *DMA.*
poison reverse: Used in conjunction with *split horizon.* A heuristic technique to avoid routing loops in distance-vector routing protocols.
port: A generic term usually used to mean the point at which a network user attaches to the network. On a switch, a port denotes the input or output on which packets are received and sent.
POTS: Plain old telephone service. Used to specify the existing phone service, in contrast to ISDN, ATM, or other technologies that the telephone companies offer now or may offer in the future.
PPP: Point-to-Point Protocol. Data link protocol typically used to connect computers over a dial-up line.

process: An abstraction provided by an operating system to enable different operations to take place concurrently. For example, each user application usually runs inside its own process, while various operating system functions take place in other processes.
promiscuous mode: A mode of operation for a network adaptor in which it receives all frames transmitted on the network, not just those addressed to it.
protocol: A specification of an interface between modules running on different machines, as well as the communication service that those modules implement. The term is also used to refer to an implementation of the module that meets this specification. To distinguish between these two uses, the interface is often called a *protocol specification*.
proxy: An agent sitting between a client and server that intercepts messages and provides some service. For example, a proxy can "stand in" for a server by responding to client requests, perhaps using data it has cached, without contacting the server.
pseudoheader: A subset of fields fromthe IP header that are passed up to transport protocols TCP and UDP for use in their checksum calculation. The pseudoheader contains source and destination IP addresses and IP datagram length, thus enabling detection of corruption of these fields or delivery of a packet to an incorrect address.
public key encryption: Any of several encryption algorithms (e.g., RSA) in which each participant has a private key (shared with no one else) and a public key (available to everyone). A secure message is sent to a user by encrypting the data with that user's public key; possession of the private key is required to decrypt the message, and so only the receiver can read it.

QoS: Quality of service. Packet delivery guarantees provided by a network architecture. Usually related to performance guarantees, such as bandwidth and delay. The Internet offers a best-effort delivery service, meaning that every effort is made to deliver a packet but delivery is not guaranteed.
RED: Random early detection. A queuing discipline for routers in which, when congestion is anticipated, packets are randomly dropped to alert the senders to slow down.
rendezvous point: A router used by PIM to allow receivers to learn about senders.
repeater: A device that propagates electrical signals from one Ethernet cable to another. There can be a maximum of two repeaters between any two hosts in an Ethernet. Repeaters forward signals, whereas *bridges* forward *frames*, and *routers* and *switches* forward *packets*.
REST: Representational State Transfer. An approach to building web services that uses HTTP as the generic application protocol.
reverse-path broadcast (RPB): A technique used to eliminate duplicate broadcast packets.
RFC: Request for Comments. Internet reports that contain, among other things, specifications for protocols like TCP and IP.
RIO: RED with In and Out. A packet drop policy based on RED, but involving two drop curves: one for packets that have been marked as being "in" profile and one for packets that have been marked "out" of profile. Designed to be used to implement differentiated services.
RIP: Routing Information Protocol. An intradomain routing protocol supplied with Berkeley Unix. Each router running RIP dynamically builds its forwarding table based on a *distance-vector* algorithm.
router: A network node connected to two or more networks that forwards packets from

one network to another. Contrast with *bridge*, *repeater*, and *switch*.
routing: The process by which nodes exchange topological information to build correct forwarding tables. See *forwarding*, *link state*, and *distance vector*.
routing table: See *forwarding table*.
RPC: Remote Procedure Call. Synchronous request/reply transport protocol used in many client/server interactions.

RPR: Resilient Packet Ring. A type of ring network that is mostly used in metropolitan area networks. See *802.17*.
RSA: A public-key encryption algorithm named after its inventors: Rivest, Shamir, and Adleman.
RSVP: Resource Reservation Protocol. A protocol for reserving resources in the network. RSVP uses the concept of *soft state* in routers and puts responsibility for making reservations on receivers instead of on senders.
RTCP: Real-time Transport Control Protocol. Control protocol associated with RTP.
RTP: Real-time Transport Protocol. An end-to-end protocol used by multimedia applications that have real-time constraints.
RTT: Round-trip time. The time it takes for a bit of information to propagate from one end of a link or channel to the other and back again; in other words, double the latency of the channel.
SAN: Storage area network. A network that spans the components of a computer system (e.g., display, camera, disk). Includes interfaces like HiPPI and Fibre Channel. Contrast with *LAN*, *MAN*, and *WAN*.
schema: A specification of how to structure and interpret a set of data. Schema are defined for XML documents.
scrambling: The process of XORing a signal with a pseudorandom bitstream before transmission to cause enough signal transitions to allow clock recovery. Scrambling is used in SONET.
SDP: Session Description Protocol. An application layer protocol used to learn about the available audio/video channels. It reports the name and purpose of the session, start and end times for the session, the media types (e.g., audio, video) that comprise the session, and detailed information needed to receive the session (e.g., the multicast address, transport protocol, and port numbers to be used).
segment: A TCP packet. A segment contains a portion of the byte stream that is being sent by means of TCP.
semaphore: A variable used to support synchronization between processes. Typically a process *blocks* on a semaphore while it waits for some other process to signal the semaphore.
server: The provider of a service in a client/server distributed system.

SHA: Secure Hash Algorithm. A family of cryptographic hash algorithms.
signalling: At the physical level, denotes the transmission of a signal over some physical medium. In ATM, signalling refers to the process of establishing a virtual circuit.
silly window syndrome: A condition occurring in TCP that may arise if each time the receiver opens its receive window a small amount, the sender sends a small segment to fill the window. The result is many small segments and an inefficient use of bandwidth.
SIP: Session Initiation Protocol. An application layer protocol used in multimedia applications.

It determines the correct device with which to communicate to reach a particular user, determines if the user is willing or able to take part in a particular communication, determines the choice of media and coding scheme to use, and establishes session parameters (e.g., port numbers).

sliding window: An algorithm that allows the sender to transmit multiple packets (up to the size of the window) before receiving an acknowledgment. As acknowledgments are returned for those packets in the window that were sent first, the window "slides" and more packets may be sent. The sliding window algorithm combines reliable delivery with a high throughput. See *ARQ*.

slow start: A congestion-avoidance algorithm for TCP that attempts to pace outgoing segments. For each ACK that is returned, two additional packets are sent, resulting in an exponential increase in the number of outstanding segments.

SMDS: Switched Multimegabit Data Service. A service supporting LAN-to-WAN connectivity, offered by some telephone companies.

SMTP: Simple Mail Transfer Protocol. The electronic mail protocol of the Internet. See *822*.

SNA: System Network Architecture. The proprietary network architecture of IBM.

SNMP: Simple Network Management Protocol. An Internet protocol that allows the monitoring of hosts, networks, and routers.

SOAP: A component of the web services framework for specifying and implementing application protocols.

socket: The abstraction provided by Unix that provides the application programming interface (API) to TCP/IP.

soft state: Connection-related information contained in a router that is cached for a limited period of time rather than being explicitly established (and requiring explicit teardown) through a connection setup.

SONET: SynchronousOpticalNetwork. A clock-based framing standard for digital transmission over optical fiber. It defines how telephone companies transmit data over optical networks.

source routing: Routing decisions performed at the source before the packet is sent. The route consists of the list of nodes that the packet should traverse on the way to the destination.

source-specific multicast: A mode of multicast in which a group may have only a single sender.

sparse mode multicast: A mode used in PIM when relatively few hosts or routers need to receive multicast data for a certain group.

split horizon: A method of breaking routing loops in a distance-vector routing algorithm. When a node sends a routing update to its neighbors, it does not send those routes it learned from each neighbor back to that neighbor. Split horizon is used with *poison reverse*.

spread spectrum: Encoding technique that involves spreading a signal over a wider frequency than necessary, so as to minimize the impact of interference.

SSL: Secure Socket Layer. A protocol layer that runs over TCP to provide authentication and encryption of connections. Also known as Transport Layer Security (TLS).

statistical multiplexing: Demand-based multiplexing of multiple data sources over a shared link or channel.

stop-and-wait: A reliable transmission algorithm in which the sender transmits a packet and waits for an acknowledgment before sending the next packet. Compare with *sliding window* and *concurrent logical channels*. See also *ARQ*.
STS: Synchronous Transport Signal. The prefix for various rates of SONET transmission. For example, STS-1 refers to the SONET standard for 51.84-Mbps transmission.
subnetting: The use of a single IP network address to denote multiple physical networks. Routers within the subnetwork use a subnet mask to discover the physical network to which a packet should be forwarded. Subnetting effectively introduces a third level to the two-level hierarchical IP address.
SunRPC: Remote procedure call protocol developed by Sun Microsystems. SunRPC is used to support NFS. See also *ONC*.

switch: A network node that forwards packets from inputs to outputs based on header information in each packet. Differs from a *router* mainly in that it typically does not interconnect networks of different types.
switching fabric: The component of a switch that directs packets from their inputs to the correct outputs.
T1: A standard telephone carrier service equal to 24 ISDN circuits, or 1.544 Mbps. Also called DS1.
T3: A standard telephone carrier service equal to 24 T1 circuits, or 44.736 Mbps. Also called DS3.
TCP: Transmission Control Protocol. Connection-oriented transport protocol of the Internet architecture. TCP provides a reliable, byte-stream delivery service.
TDMA: Time Division Multiple Access. A form of multiplexing used in cellular wireless networks. Also the name of a particular wireless standard.
Telnet: Remote terminal protocol of the Internet architecture. Telnet allows you to interact with a remote system as if your terminal is directly connected to that machine.
throughput: The observed rate at which data is sent through a channel. The term is often used interchangeably with *bandwidth*.
TLS: Transport Layer Security. Security services that can be layered on top of a transport protocol like TCP. It is often used by HTTP to perform secure transactions on theWorld Wide Web. Derived from *SSL*.
token bucket: A way to characterize or police the bandwidth used by a flow. Conceptually, processes accumulate tokens over time, and they must spend a token to transmit a byte of data and then must stop sending when they have no tokens left. Thus, overall bandwidth is limited, with the accommodation of some burstiness.
token ring: A physical network technology in which hosts are connected in a ring. A token (bit pattern) circulates around the ring. A given node must possess the token before it is allowed to transmit. 802.5 and FDDI are examples of token ring networks.
TP4: OSI Transport Protocol Class 4. The most powerful OSI transport protocol. TP4 is the ISO equivalent of TCP.
transport protocol: An end-to-end protocol that enables processes on different hosts to communicate. TCP is the canonical example.

TTL: Time to live. Usually a measure of the number of hops (routers) an IP datagram can visit before it is discarded.
tunneling: Encapsulating a packet using a protocol that operates at the same layer as the

packet. For example, multicast IP packets are encapsulated inside unicast IP packets to tunnel across the Internet to implement the MBone. Tunneling will also be used during the transition from IPv4 to IPv6.

two-dimensional parity: A parity scheme in which bytes are conceptually stacked as a matrix, and parity is calculated for both rows and columns.

Tymnet: An early network in which a *virtual circuit* abstraction was maintained across a set of routers.

UBR: Unspecified bit rate. The "no frills" service class in ATM, offering best-effort cell delivery. Contrast with *ABR*, *CBR*, and *VBR*.

UDP: User Datagram Protocol. Transport protocol of the Internet architecture that provides a connectionless datagram service to application-level processes.

UMTS: Universal Mobile Telecommunications System. Cellular wireless standard based on wideband CDMA that offers relatively high data rates.

unicast: Sending a packet to a single destination host. Contrast with *broadcast* and *multicast*.

URI: Uniform Resource Identifier. A generalization of the URL. Used for example, in conjunction with SIP to set up audio/visual sessions.

URL: Uniform Resource Locator. A text string used to identify the location of Internet resources. A typical URL looks like http://www.cisco.com. In this URL, http is the protocol to use to access the resource located on host www.cisco.com.

vat: Audioconferencing tool used on the Internet that runs over RTP.

VBR: Variable bit rate. One of the classes of service in ATM, intended for applications with bandwidth requirements that vary with time, such as compressed video. Contrast with *ABR*, *CBR*, and *UBR*.

VCI: Virtual circuit identifier. An identifier in the header of a packet that is used for virtual circuit switching. In the case of ATM, the VPI and VCI together identify the end-to-end connection.

vic: Unix-based videoconferencing tool that uses RTP.

virtual circuit: The abstraction provided by connection-oriented networks such as ATM.

Messages must usually be exchanged between participants to establish a virtual circuit (and perhaps to allocate resources to the circuit) before data can be sent. Contrast with *datagram*.

virtual clock: A service model that allows the source to reserve resources on routers using a rate-based description of its needs. Virtual clock goes beyond the best-effort delivery service of the current Internet.

VPI: Virtual path identifier. An 8-bit or 12-bit field in the ATM header. VPI can be used to hide multiple virtual connections across a network inside a single virtual "path," thus decreasing the amount of connection state that the switches must maintain. See also *VCI*.

VPN: Virtual private network. A logical network overlaid on top of some existing network. For example, a company with sites around the world may build a virtual network on top of the Internet rather than lease lines between each site.

WAN: Wide area network. Any physical network technology that is capable of spanning long distances (e.g., cross-country). Compare with *SAN*, *LAN*, and *MAN*.

weighted fair queuing (WFQ): A variation of *fair queuing* in which each flow can be given a different proportion of the network capacity.

well-known port: A port number that is, by convention, dedicated for use by a particular server. For instance, the Domain Name Server receives messages at well-known UDP and TCP port 53 on every host.
WSDL: Web Services Description Language. A component of the web services framework for specifying and implementing application protocols.
WWW: World Wide Web. A hypermedia information service on the Internet.
X.25: The ITU packet-switching protocol standard.
X.400: The ITU electronic mail standard. The counterpart to SMTP in the Internet architecture.
X.500: The ITU directory services standard, which defines an attribute-based naming service.
X.509: An ITU standard for digital certificates.
XDR: External Data Representation. SunMicrosystems' standard for machine-independent data structures. Contrast with *ASN.1* and *NDR*.

XML: Extensible Markup Language. Defines a syntax for describing data that may be

passed between Internet applications.
XSD: XML Schema Definition. A schema language for defining the format and interpretation of XML objects.
zone: A partition of the domain name hierarchy, corresponding to an administrative authority that is responsible for that portion of the hierarchy. Each zone must have at least two name servers to field DNS requests for the zone.

WORKED OUT PROBLEMS

1.Calculate the total time required to transfer a 1.5MB file in the following cases,assuming a RTT of 80 ms, a packet size of 1 KB data, and an initial 2×RTT of"handshaking" before data is sent.
(a)The bandwidth is 10Mbps, and data packets can be sent continuously.
(b) The bandwidth is 10Mbps, but after we finish sending each data packet we must wait one RTT before sending the next.
(c) The link allows infinitely fast transmit, but limits bandwidth such that only 20 packets can be sent per RTT.
(d) Zero transmit time as in (c), but during the first RTT we can send one packet, during the second RTT we can send two packets, during the third we can send four = 23−1, and so on.

SOLUTION
We will count the transfer as completed when the last data bit arrives at its destination.

(a) 1.5 MB = 12,582,912 bits. 2 initial RTT's (160 ms) + 12,582,912/10,000,000 bps (transmit)+ RTT/2 (propagation) ≈ 1.458 seconds.
(b) Number of packets required = 1.5 MB/1 KB = 1,536. To the above we add the time for 1,535 RTTs (the number of RTTs between when packet 1 arrives and packet 1,536 arrives), for a total of 1.458+122.8 = 124.258 seconds.
(c) Dividing the 1,536 packets by 20 gives 76.8. This will take 76.5 RTTs (half an RTT for the first batch to arrive, plus 76 RTTs between the first batch and the 77th partial batch), plus the initial 2 RTTs, for 6.28 seconds.
(d) Right after the handshaking is done we send one packet. One RTT after the handshaking we send two packets. At n RTTs past the initial handshaking we have sent $1+2+4+\cdots+2n = 2n+1 - 1$ packets. At $n = 10$ we have thus been able to send all 1,536 packets; the last batch arrives 0.5 RTT later. Total time is 2 +10.5 RTTs, or 1 second.

2.Consider a point-to-point link 50 km in length. At what bandwidth would propagation delay (at a speed of 2 × 108 m/sec) equal transmit delay for 100- byte packets? What about 512-byte packets?

SOLUTION :

Propagation delay is 50 × 103 m/(2 × 108 m/sec) = 250 μs 800 bits/250 μs is 3.2 Mbits/sec. For 512-byte packets, this rises to 16.4 Mbit/sec.

3.Suppose a 128-Kbps point-to-point link is set up between Earth and a rover on Mars. The distance from Earth to Mars (when they are closest together) is approximately 55 Gm, and data travels over the link at the speed of light—3×108 m/sec.
(a) Calculate the minimum RTT for the link.
(b) Calculate the delay × bandwidth product for the link.
(c) A camera on the rover takes pictures of its surroundings and sends these to Earth. How quickly after a picture is taken can it reach Mission Control on Earth? Assume that each image is 5 MB in size.

SOLUTION :

(a) Propagation delay on the link is (55 × 109)/(3 × 108) = 184 seconds. Thus the RTT is 368 seconds.
(b) The delay × bandwidth product for the link is = 184 × 128 × 103 = 2.81 MB.
(c) After a picture is taken it must be transmitted on the link, and be completely propagated before Mission Control can interpret it. Transmit delay
for 5 MB of data is 41,943,040 bits/128×103 = 328 seconds. Thus, the total time required is transmit delay + propagation delay = 328 + 184 = 512 seconds.

4.Calculate the latency (from first bit sent to last bit received) for:

(a) A 1-Gbps Ethernet with a single store-and-forward switch in the path, and a packet size of 5,000 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
(b) Same as (a) but with three switches.

(c) Same as (b) but assume the switch implements cut-through switching: it is able to begin retransmitting the packet after the first 128 bits have been received.

SOLUTION :

(a) For each link, it takes 1 Gbps/5 kb = 5 μs to transmit the packet on the link, after which it takes an additional 10 μs for the last bit to propagate across the link. Thus, for a LAN with only one switch that starts forwarding only after receiving the whole packet, the total transfer delay is two transmit delays + two propagation delays = 30 μs.

(b) For three switched and thus four links, the total delay is four transmit delays + four propagation delays = 60 μs.

(c) For "cut-through," a switch need only decode the first 128 bits before beginning to forward. This takes 128 ns. This delay replaces the switch transmit delays in the previous answer for a total delay of one Transmit delay + three cut-through decoding delays + four propagation delays =45.384 μs.

5. For the following, as in the previous problem, assume that no data compression is done. Calculate the bandwidth necessary for transmitting in real time:

(a) HDTV high-definition video at a resolution of 1,920 × 1,080, 24 bits/pixel, 30 frames/sec.
(b) Plain old telephone service (POTS) voice audio of 8-bit samples at 8 KHz.
(c) GSM mobile voice audio of 260-bit samples at 50 Hz.
(d) HDCD high-definition audio of 24-bit samples at 88.2 kHz.

SOLUTION :

(a) 1,920×1,080×24×30 = 1,492,992,000 ≈ 1.5 Gbps.
(b) 8×8,000 = 64 Kbps.
(c) 260 ×50 = 13 Kbps.
(d) 24×88,200 = 216,800 ≈ 2.1 Mbps.

6. Show the 4B/5B encoding, and the resulting NRZI signal, for the following bit sequence:

1101 1110 1010 1101 1011 1110 1110 1111

SOLUTION :

The 4B/5B encoding of the given bit sequence is the following.
11011 11100 10110 11011 10111 11100 11100 11101

7 Suppose the following sequence of bits arrive over a link:
0110101111101010011111111011001111110
Show the resulting frame after any stuffed bits have been removed. Indicate any errors that might have been introduced into the frame.

SOLUTION :

Let ∧ mark each position where a stuffed 0 bit was removed. There was one error where the sever consecutive 1s are detected (***err***). At the end of the bit sequence, the end of frame was detected (***eof***).

01101011111∧1010011111111*err*0 110 01111110*eof*

8. Suppose we want to transmit the message 1011 0010 0100 1011 and protect it from errors using the CRC8 polynomial $x8 + x2 + x1 + 1$.
(a) Use polynomial long division to determine the message that should be transmitted.
(b) Suppose the leftmost bit of the message is inverted due to noise on the transmission link. What is the result of the receiver's CRC calculation? How does the receiver know that an error has occurred?

SOLUTION :

(a) We take the message 1011 0010 0100 1011, append 8 zeros and divide by 1 0000 0111 ($x8 + x2 + x1 + 1$). The remainder is 1001 0011. We transmit the original message with this remainder appended, resulting in 1011 0010 0100 0011 1001 0011.
(b) Inverting the first bit gives 0011 0010 0100 1011 1001 0011. Dividing by 1 0000 0111 ($x8 + x2 + x1 + 1$) gives a a remainder of 1011 0110.

9. Suppose you are designing a sliding window protocol for a 1-Mbps pointto-point link to the stationary satellite evolving around Earth at 3 × 104 km altitude. Assuming that each frame carries 1 KB of data, what is the minimum number of bits you need for the sequence number in the following cases? Assume
the speed of light is 3×108 meters per second.
(a) RWS=1.
(b) RWS=SWS.

SOLUTION :

One-way latency of the link is 100 msec. (Bandwidth) × (roundtrip delay) is about 125 pps×0.2 sec, or 25 packets. SWS should be this large.
(a) If RWS = 1, the necessary sequence number space is 26. Therefore, 5 bitsare needed.
(b) IfRWS = SWS, the sequence number space must cover twice the SWS, or up to 50. Therefore, 6 bits are needed.

10.Given the extended LAN shown in Figure 3.34, assume that bridge B1 suffers catastrophic failure. Indicate which ports are not selected by the spanning tree algorithm after the recovery process and a new tree has been formed.

SOLUTION :

The following list shows the mapping between LANs and their designated bridges.
B1 dead[B7]
B2 A,B,D
B3 E,F,G,H
B4 I
B5 idle
B6 J
B7 C

11.Suppose we have the forwarding tables shown in Table 4.13 for nodes A and F, in a network where all links have cost 1. Give a diagram of the smallest network consistent with these tables.

A

Node	Cost	NextHop
B	1	B
C	1	C
D	2	B
E	3	C
F	2	C

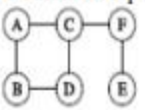
F

Node	Cost	NextHop
A	2	C
B	3	C
C	1	C
D	2	C
E	1	E

Table 4.13 Forwarding tables for Exercise 19.

SOLUTION :

The following is an example network topology.



QUESTION BANK

UNIT-I

PART-A (2 MARKS)

1. What are the three criteria necessary for an effective and efficient network?

The most important criteria are performance, reliability and security.

Performance of the network depends on number of users, type of transmission medium, and the capabilities of the connected h/w and the efficiency of the s/w.

Reliability is measured by frequency of failure, the time it takes a link to recover from the failure and the network's robustness in a catastrophe.

Security issues include protecting data from unauthorized access and viruses.

2. Group the OSI layers by function?

The seven layers of the OSI model belonging to three subgroups.

Physical, data link and network layers are the *network support layers*; they deal with the physical aspects of moving data from one device to another.

Session, presentation and application layers are the *user support layers*; they allow interoperability among unrelated software systems.

The transport layer ensures *end-to-end reliable data transmission*.

3. What are header and trailers and how do they get added and removed?

Each layer in the sending machine adds its own information to the message it receives from the layer just above it and passes the whole package to the layer just below it. This information is added in the form of headers or trailers. Headers are added to the message at the layers 6,5,4,3, and 2. A trailer is added at layer2. At the receiving machine, the headers or trailers attached to the data unit at the corresponding sending layers are removed, and actions appropriate to that layer are taken.

4. What are the features provided by layering?

Two nice features:

- It decomposes the problem of building a network into more manageable components.
- It provides a more modular design.

5. Why are protocols needed?

In networks, communication occurs between the entities in different systems. Two entities cannot just send bit streams to each other and expect to be understood. For communication, the entities must agree on a protocol. A protocol is a set of rules that govern data communication.

6. What are the two interfaces provided by protocols?

- Service interface
- Peer interface

Service interface- defines the operations that local objects can perform on the protocol.

Peer interface- defines the form and meaning of messages exchanged between protocol peers to implement the communication service.

7. Mention the different physical media?

- Twisted pair(the wire that your phone connects to)
- Coaxial cable(the wire that your TV connects to)
- Optical fiber(the medium most commonly used for high-bandwidth, long-distance links)
- Space(the stuff that radio waves, microwaves and infra red beams propagate through)

8. Define Signals?

Signals are actually electromagnetic waves traveling at the speed of light. The speed of light is, however, medium dependent-electromagnetic waves traveling through copper and fiber do so at about two-thirds the speed of light in vacuum.

9. What is wave's wavelength?

The distance between a pair of adjacent maxima or minima of a wave, typically measured in meters, is called wave's wavelength.

10. Define Modulation?

Modulation -varying the frequency, amplitude or phase of the signal to effect the transmission of information. A simple example of modulation is to vary the power (amplitude) of a single wavelength.

11. Explain the two types of duplex?

- *Full duplex*-two bit streams can be simultaneously transmitted over the links at the same time, one going in each direction.
- *Half duplex*-it supports data flowing in only one direction at a time.

12. What is CODEC?

A device that encodes analog voice into a digital ISDN link is called a CODEC, for *coder/decoder.*

13. What is spread spectrum and explain the two types of spread spectrum?

Spread spectrum is to spread the signal over a wider frequency band than normal in such a way as to minimize the impact of interference from other devices.

- Frequency Hopping
- Direct sequence

14. What are the different encoding techniques?

- NRZ
- NRZI
- Manchester
- 4B/5B

15. How does NRZ-L differ from NRZ-I?

In the NRZ-L sequence, positive and negative voltages have specific meanings: positive for 0 and negative for 1. in the NRZ-I sequence, the voltages are meaningless.

Instead, the receiver looks for changes from one level to another as its basis for recognition of 1s.

16. What are the responsibilities of data link layer?

Specific responsibilities of data link layer include the following. a) Framing b) Physical addressing c) Flow control d) Error control e) Access control.

17. What are the ways to address the framing problem?

- Byte-Oriented Protocols(PPP)
- Bit-Oriented Protocols(HDLC)
- Clock-Based Framing(SONET)

18. Distinguish between peer-to-peer relationship and a primary-secondary relationship. peer -to- peer relationship?

All the devices share the link equally.

Primary-secondary relationship: One device controls traffic and the others must transmit through it.

19. Mention the types of errors and define the terms?

There are 2 types of errors

- Single-bit error.
- Burst-bit error.

Single bit error: The term single bit error means that only one bit of a given data unit (such as byte character/data unit or packet) is changed from 1 to 0 or from 0 to 1.

Burst error: Means that 2 or more bits in the data unit have changed from 1 to 0 from 0 to 1.

20. List out the available detection methods.

There are 4 types of redundancy checks are used in data communication.

- Vertical redundancy checks (VRC).
- Longitudinal redundancy checks (LRC).
- Cyclic redundancy checks (CRC).
- Checksum.

21. Write short notes on VRC.

The most common and least expensive mechanism for error detection is the vertical redundancy check (VRC) often called a parity check. In this technique a redundant bit called a parity bit, is appended to every data unit so, that the total number of 0's in the unit (including the parity bit) becomes even.

22. Write short notes on LRC.

In longitudinal redundancy check (LRC), a block of bits is divided into rows and a redundant row of bits is added to the whole block.

23. Write short notes on CRC.

The third and most powerful of the redundancy checking techniques is the cyclic redundancy checks (CRC) CRC is based on binary division. Here a sequence of redundant bits, called the CRC remainder is appended to the end of data unit.

24. Write short notes on CRC checker.

A CRC checker functions exactly like a generator. After receiving the data appended with the CRC it does the same modulo-2 division. If the remainder is all 0's the CRC is dropped and the data accepted. Otherwise, the received stream of bits is discarded and the dates are resent.

25. Define checksum.

The error detection method used by the higher layer protocol is called checksum. Checksum is based on the concept of redundancy.

26. What are the steps followed in checksum generator?

The sender follows these steps a) the units are divided into k sections each of n bits. b) All sections are added together using 2's complement to get the sum. c) The sum is complemented and become the checksum. d) The checksum is sent with the data.

27. Mention the types of error correcting methods.

There are 2 error-correcting methods.

- Single bit error correction
- Burst error correction.

28. Write short notes on error correction?

It is the mechanism to correct the errors and it can be handled in 2 ways.

- When an error is discovered, the receiver can have the sender retransmit the entire data unit.
- A receiver can use an error correcting coder, which automatically corrects certain errors.

29. What is the purpose of hamming code?

A hamming code can be designed to correct burst errors of certain lengths. So the simple strategy used by the hamming code to correct single bit errors must be redesigned to be applicable for multiple bit correction.

30. What is redundancy?

It is the error detecting mechanism, which means a shorter group of bits or extra bits may be appended at the destination of each unit.

31. Define flow control?

Flow control refers to a set of procedures used to restrict the amount of data. The sender can send before waiting for acknowledgment.

32. Mention the categories of flow control?

There are 2 methods have been developed to control flow of data across communication links. a) Stop and wait- send one from at a time. b) Sliding window- send several frames at a time.

33. What is a buffer?

Each receiving device has a block of memory called a buffer, reserved for storing incoming data until they are processed.

UNIT I

PART-B (16 MARKS)

1. Explain the OSI-ISO model I of computer with neat diagram. (16)

2. Distinguish between Point to Point links and multi-point links with relevant diagram. (16)

3. (i) Compare connection oriented and connection less service. (8)

(ii) Explain the various topologies. (8)

4. (i) Write a short notes on various types of transmission media, highlighting their merits and demerits ? (8)

(ii) Describe the categories of network.(8)

5. Explain Error correction and detection ?(16)

6.Explain framing with its header diagram?(16)

7. (i) Discuss about stop and wait protocol with an example (8)

(ii) Explain sliding window flow control mechanism with an example (8)

UNIT-II

PART-A (2 MARKS)

1. What are the functions of MAC?

MAC sub layer resolves the contention for the shared media. It contains synchronization, flag, flow and error control specifications necessary to move information from one place to another, as well as the physical address of the next station to receive and route a packet.

2. What are the functions of LLC?

The IEEE project 802 models take the structure of an HDLC frame and divides it into 2 sets of functions. One set contains the end user portion of the HDLC frame – the logical address, control information, and data. These functions are handled by the IEEE 802.2 logical link control (LLC) protocol.

3. What is Ethernet?

Ethernet is a multiple-access network, meaning that a set of nodes send and receive frames over a shared link.

4. Define the term carrier sense in CSMA/CD?

All the nodes can distinguish between idle and a busy-link and "collision detect" means that a node listens as it transmits and can therefore detect when a frame it is transmitting has interfered (collided) with a frame transmitted by another node.

5. Define Repeater?

A repeater is a device that forwards digital signals, much like an amplifier forwards analog signals. However, no more than four repeaters may be positioned between any pairs of hosts, meaning that an Ethernet has a total reach of only 2,500m.

6. Define collision detection?

In Ethernet, all these hosts are competing for access to the same link, and as a consequence, they are said to be in the same collision detection.

7. Why Ethernet is said to be a I-*persistent* protocol?

An adaptor with a frame to send transmits with probability '1 'whenever a busy line goes idle.

8. What is exponential back off?

Once an adaptor has detected a collision and stopped its transmission, it waits a certain amount of time and tries again. Each time it tries to transmit but fails, the adaptor doubles the amount of time it waits before trying again. This strategy of doubling the delay interval between each transmission attempt is a general technique known as exponential back off.

9. What is token holding time (THT)?

It defines that how much data a given node is allowed to transmit each time it possesses the token or equivalently, how long a given node is allowed to hold the token.

10. What are the two classes of traffic in FDDI?

- Synchronous
- Asynchronous

11. What are the four prominent wireless technologies?

- Bluetooth
- Wi-Fi(formally known as 802.11)
- WiMAX(802.16)
- Third generation or 3G cellular wireless.

12. Define Bluetooth?

Bluetooth fills the niche of very short-range communication between mobile phones, PDAs, notebook computers, and other personal or peripheral devices. For example, Bluetooth can be used to connect mobile phones to a headset, or a notebook computer to a printer.

13. What are the four steps involves in scanning?

1. The node sends a Probe frame.
2. All APs within reach reply with a Probe Response frame.
3. The node selects one of the access points, and sends that AP an Association Request frame.
4. The AP replies with an Association Response frame.

14. Explain the term handoff?

If the phone is involved in a call at the time , the call must be transferred to the new base station in what is called a hand off.

15. Define satphones?

Satphones use communication satellites as base stations, communicating on frequency bands that have been reserved internationally for satellite use.

16. How to mediate access to a shared link?

Ethernet,token ring, and several wireless protocols. Ethernet and token ring media access protocols have no central arbitrator of access. Media access in wireless networks is made more complicated by the fact that some nodes may be hidden from each other due to range limitations of radio transmission.

17. Define Aggregation points?

It collects and processes the data they receive from neighboring nodes, and then transmit the processed data. By processing the data incrementally, instead of forwarding all the raw data to the base station, the amount of traffic in the network is reduced.

18. Define Beacons?

Beacon to determine their own absolute locations based on GPS or manual configuration. The majority of nodes can then derive their absolute location by combining an estimate of their position relative to the beacons with the absolute location information provided by the beacons.

19. What is the use of Switch?

It is used to forward the packets between shared media LANs such as Ethernet. Such switches are sometimes known by the obvious name of LAN switches.

20. Explain Bridge?

It is a collection of LANs connected by one or more bridges is usually said to form an extended LAN. In their simplest variants, bridges simply accept LAN frames on their inputs and forward them out on all other outputs.

21. What is Spanning tree?

It is for the bridges to select the ports over which they will forward frames.

22. What are the three pieces of information in the configuration messages?

1. The ID for the bridge that is sending the message.
2. The ID for what the sending bridge believes to the root bridge.
3. The distance, measured in hops, from the sending bridge to the root bridge.

23. What is broadcast?

Broadcast is simple – each bridge forwards a frame with a destination broadcast address out on each active (selected) port other than the one on which the frame was received.

24. What is multicast?

It can be implemented with each host deciding for itself whether or not to accept the message.

25. How does a given bridge learn whether it should forward a multicast frame over a given port?

It learns exactly the same way that a bridge learns whether it should forward a unicast frame over a particular port- by observing the source addresses that it receives over that port.

26. What are the limitations of bridges?

- scale
- heterogeneity

UNIT II

PART-B (16 MARKS)

1. Explain the frame format for token ring and token bus. **(16)**

2. Explain Ethernet protocol . (16)

4. Explain the following Inter connection devices also discuss their uses

a. Repeater (4)

b. Bridge (4)

c. Switch (4)

d. Gateway (4)

5. Explain any one of the protocols used for flow control in noisy channel fiber distributed data interface operations. (16)

6.(i) Explain about SONET **(8)**

(ii)Explain the CSMA/CD algorithm in detail (8)

7. (i) Explain the token passing mechanism of Token ring network (8)

(ii) Discuss the ring maintenance in Token ring network (8)

8. Discuss in detail about the wireless LAN (16)

UNIT-III

PART-A (2 MARKS)

1. Define packet switching?

A packet switch is a device with several inputs and outputs leading to and from the hosts that the switch interconnects.

2. What is a virtual circuit?

A logical circuit made between the sending and receiving computers. The connection is made after both computers do handshaking. After the connection, all packets follow the same route and arrive in sequence.

3. What are data grams?

In datagram approach, each packet is treated independently from all others. Even when one packet represents just a place of a multi packet transmission, the network treats it although it existed alone. Packets in this technology are referred to as datagram.

4. What is meant by switched virtual circuit?

Switched virtual circuit format is comparable conceptually to dial-up line in circuit switching. In this method, a virtual circuit is created whenever it is needed and exits only for the duration of specific exchange.

5. What is meant by Permanent virtual circuit?

Permanent virtual circuits are comparable to leased lines in circuit switching. In this method, the same virtual circuit is provided between two uses on a continuous basis. The circuit is dedicated to the specific uses.

6. What are the properties in star topology?

- Even though a switch has a fixed number of inputs and outputs, which limits the number of hosts that can be connected to a single switch , large networks can be built by interconnecting a number of switches.
- We can connect switches to each other and to hosts using point-to point links, which typically means that we can build networks of large geographic scope.

7. What is VCI?

A Virtual Circuit Identifier that uniquely identifies the connection at this switch, and which will be carried inside the header of the packets that belongs to this connection.

8. What is hop-by-hop flow control?

Each node is ensured of having the buffers it needs to queue the packets that arrive on that circuit. This basic strategy is usually called hop-by-hop flow control.

9. Explain the term best-effort?

If something goes wrong and the packet gets lost, corrupted, misdelivered, or in any way fails to reach its intended destination, the network does nothing.

10. What is maximum transmission unit?

MTU- which is the largest IP datagram that it can carry in a frame .

11. Define Routing?

It is the process of building up the tables that allow thwe collect output for a packet to be determined.

12. Define ICMP?

Internet Control Message Protocol is a collection of error messages that are sent back to the source host whenever a router or host is unable to process an IP datagram successfully

13. Write the keys for understanding the distance vector routing?

The three keys for understanding the algorithm are,

- Knowledge about the whole networks
- Routing only to neighbors
- Information sharing at regular intervals

14. Write the keys for understanding the link state routing?

The three keys for understanding the algorithm are,

- Knowledge about the neighborhood.
- Routing to all neighbors.
- Information sharing when there is a range.

15. How the packet cost referred in distance vector and link state routing?

In distance vector routing, cost refer to hop count while in case of link state routing, cost is a weighted value based on a variety of factors such as security levels, traffic or the state of the link.

16. Define Reliable flooding?

It is the process of making sure that all the nodes participating in the routing protocol get a copy of the link state information from all the other nodes.

17. What are the features in OSPF?

- Authentication of routing messages.
- Additional hierarchy.
- Load balancing.

18. Define Subnetting?

Subnetting provides an elegantly simple way to reduce the total number of network numbers that are assigned. The idea is to take a single IP network number and allocate the IP address with that network to several physical networks, which are now referred to as subnets.

19. What are the different types of AS?

- Stub AS
- Multi homed AS
- Transit AS

20. What is an Area?

An Area is a set of routers that are administratively configured to exchange link-state information with each other. There is one special area- the backbone area, also known as area 0.

21. What is Source Specific Multicast?

SSM , a receiving host specifies both a multicast group and a specific host .the receiving host would then receive multicast addressed to the specified group, but only if they are from the special sender.

22. What is meant by congestion?

Congestion in a network occurs if user sends data into the network at a rate greater than that allowed by network resources.

23. Why the congestion occurs in network?

Congestion occurs because the switches in a network have a limited buffer size to store arrived packets.

24. What are the rules of non boundary-level masking?

- The bytes in the IP address that corresponds to 255 in the mask will be repeated in the sub network address
- The bytes in the IP address that corresponds to 0 in the mask will change to 0 in the sub network address
- For other bytes, use the bit-wise AND operator.

25. What is LSP?

In link state routing, a small packet containing routing information sent by a router to all other router by a packet called link state packet.

UNIT III

PART-B (16 MARKS)

1. Discuss how these routing and link state routing techniques work with example (16)

2. (i) Explain the two approaches of packet switching techniques and circuit switching techniques (16)

3. State the major difference between RIP and OSPF. (16)

4. Explain IP in detail .(16)

5. (i) What is the purpose of subneting ?explain the various subnet mask?(8)

(ii) Compare ARP and RARP ? (8)

6. Explain IPv6 in detail (16)

7. (i) Write notes on BGP and CIDR (10)

(ii) What is multicasting (2)

(ii) What is interdomain routing (4)

NIT-IV

PART-A (2 MARKS)

1. Explain the main idea of UDP?

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.

2. What are the different fields in pseudo header?

- Protocol number
- Source IP address
- Destination IP addresses.

3. Define TCP?

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 in each direction.

4. Define Congestion Control?

It involves preventing too much data from being injected into the network, thereby causing switches or links to become overloaded. Thus flow control is an end to an end issue, while congestion control is concerned with how hosts and networks interact.

5. State the two kinds of events trigger a state transition?

- A segment arrives from the peer.
- The local application process invokes an operation on TCP.

6. What is meant by segment?

At the sending and receiving end of the transmission, TCP divides long transmissions into smaller data units and packages each into a frame called a segment.

7. What is meant by segmentation?

When the size of the data unit received from the upper layer is too long for the network layer datagram or data link layer frame to handle, the transport protocol divides it into smaller usable blocks. The dividing process is called segmentation.

8. What is meant by Concatenation?

The size of the data unit belonging to single sessions are so small that several can fit together into a single datagram or frame, the transport protocol combines them into a single data unit. The combining process is called concatenation.

9. What is rate based design?

Rate- based design, in which the receiver tells the sender the rate-expressed in either bytes or packets per second – at which it is willing to accept incoming data.

10. Define Gateway.

A device used to connect two separate networks that use different communication protocols.

11. What is meant by quality of service?

The quality of service defines a set of attributes related to the performance of the connection. For each connection, the user can request a particular attribute each service class is associated with a set of attributes.

12. What are the two categories of QoS attributes?

The two main categories are,

- User Oriented
- Network Oriented

13. List out the user related attributes?

User related attributes are SCR – Sustainable Cell Rate PCR – Peak Cell Rate MCR- Minimum Cell Rate CVDT – Cell Variation Delay Tolerance.

14. What are the networks related attributes?

The network related attributes are, Cell loss ratio (CLR) Cell transfer delay (CTD) Cell delay variation (CDV) Cell error ratio (CER).

15. What is RED?

Random Early Detection in each router is programmed to monitor its own queue length and when it detects that congestion is imminent, to notify the source to adjust its congestion window.

16. What are the three events involved in the connection?

For security, the transport layer may create a connection between the two end ports. A connection is a single logical path between the source and destination that is associated with all packets in a message. Creating a connection involves three steps:

- Connection establishment
- Data transfer
- Connection release

17.What is Silly Window Syndrome?

If the sender or the receiver application program processes slowly and can send only 1 byte of data at a time, then the overhead is high.This is because to send one byte of data, 20 bytes of TCP header and 20 bytes of IP header are sent. This is called as silly window syndrome.

UNIT IV

PART-B (16 MARKS)

1. Explain congestion control and congestion avoidance in detail. (16)
2. Compare TCP and UDP with neat diagram (16)
3. i) Discuss about quality of services. (8)

   ii) Explain the three way handshake protocol to establish the transport level connection.(8)

4.(i) Explain the working of TCP using the state diagram(12)

(ii) What is adaptive retransmission and mention the algorithm used (4)

UNIT-V

PART-A (2 MARKS)

1. What is the function of SMTP?

The TCP/IP protocol supports electronic mail on the Internet is called Simple Mail Transfer (SMTP). It is a system for sending messages to other computer users based

on e-mail addresses. SMTP provides mail exchange between users on the same or different computers.

2. What is the difference between a user agent (UA) and a mail transfer agent (MTA)?

The UA prepares the message, creates the envelope, and puts the message in the envelope. The MTA transfers the mail across the Internet.

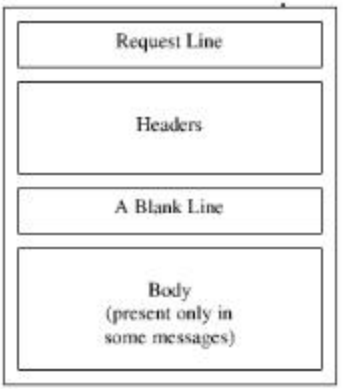
3. How does MIME enhance SMTP?

MIME is a supplementary protocol that allows non-ASCII data to be sent through SMTP. MIME transforms non-ASCII data at the sender site to NVT ASCII data and deliverers it to the client SMTP to be sent through the Internet. The server SMTP at the receiving side receives the NVT ASCII data and delivers it to MIME to be transformed back to the original data.

4. **Why is an application such as POP needed for electronic messaging?**

Workstations interact with the SMTP host, which receives the mail on behalf of every host in the organization, to retrieve messages by using a client-server protocol such as Post Office Protocol, version 3(POP3). Although POP3 is used to download messages from the server, the SMTP client still needed on the desktop to forward messages from the workstation user to its SMTP mail server.

5. Give the format of HTTP request message?



6. What is the purpose of Domain Name System?

Domain Name System can map a name to an address and conversely an address to name.

7. Discuss the three main division of the domain name space.

Domain name space is divided into three different sections: generic domains, country domains & inverse domain.

Generic domain: Define registered hosts according to their generic behavior, uses generic suffixes.

Country domain: Uses two characters to identify a country as the last suffix.

Inverse domain: Finds the domain name given the IP address.

8. Discuss the TCP connections needed in FTP.

FTP establishes two connections between the hosts. One connection is used for data transfer, the other for control information. The control connection uses very simple rules of communication. The data connection needs more complex rules due to the variety of data types transferred.

9. Discuss the basic model of FTP.

The client has three components: the user interface, the client control process, and the client data transfer process. The server has two components: the server control process and the server data transfer process. The control connection is made between the control processes. The data connection is made between the data transfer processes.

10. Name four factors needed for a secure network?

Privacy: The sender and the receiver expect confidentiality.

Authentication: The receiver is sure of the sender's identity and that an imposter has not sent the message.

Integrity: The data must arrive at the receiver exactly as it was sent.

Non-Reputation: The receiver must able to prove that a received message came from a specific sender.

11. How is a secret key different from public key?

In secret key, the same key is used by both parties. The sender uses this key and an encryption algorithm to encrypt data; the receiver uses the same key and the corresponding decryption algorithm to decrypt the data. In public key, there are two keys: a private key and a public key. The private key is kept by the receiver. The public key is announced to the public.

12. What is a digital signature?

Digital signature is a method to authenticate the sender of a message. It is similar to that of signing transactions documents when you do business with a bank. In network transactions, you can create an equivalent of an electronic or digital signature by the way you send data.

13. What are the advantages & disadvantages of public key encryption?

Advantages:

a) Remove the restriction of a shared secret key between two entities. Here each entity can create a pair of keys, keep the private one, and publicly distribute the other one.

b) The no. of keys needed is reduced tremendously. For one million users to communicate, only two million keys are needed.

Disadvantage:

If you use large numbers the method to be effective. Calculating the cipher text using the long keys takes a lot of time. So it is not recommended for large amounts of text.

14. What are the advantages & disadvantages of secret key encryption?

Advantage:

Secret Key algorithms are efficient: it takes less time to encrypt a message. The reason is that the key is usually smaller. So it is used to encrypt or decrypt long messages.

Disadvantages:

a) Each pair of users must have a secret key. If N people in world want to use this method, there needs to be N (N-1)/2 secret keys. For one million people to communicate, a half-billion secret keys are needed.

b) The distribution of the keys between two parties can be difficult.

15. Define permutation.

Permutation is transposition in bit level.

Straight permutation: The no. of bits in the input and output are preserved.

Compressed permutation: The no. of bits is reduced (some of the bits are dropped).

Expanded permutation: The no. of bits is increased (some bits are repeated).

16. Define substitution & transposition encryption?

. *Substitution:* A character level encryption in which each character is replaced by another character in the set.

Transposition: A Character level encryption in which the characters retain their plaintext but the position of the character changes.

17. Define CGI?

. CGI is a standard for communication between HTTP servers and executable programs. It is used in crating dynamic documents.

18. What are the requests messages support SNMP and explain it?

- GET
- SET

The former is used to retrieve a piece of state from some node and the latter is used to store a new piece of state in some node.

19. Define PGP?

Pretty Good Privacy is used to provide security for electronic mail. It provides authentication, confidentiality, data integrity, and non repudiation.

20. Define SSH?

Secure Shell is used to provide a remote login, and used to remotely execute commands and transfer files and also provide strong client/server authentication / message integrity.

21. Give the format of HTTP response message?

22. What is the difference between service point address, logical address and physical address? Service point addressing Logical addressing Physical addressing

Service point addressing	Logical addressing	Physical addressing

The transport layer header includes a type of address called a service point address or port address, which makes a data delivery from a specific process on one computer to a specific process on another computer.	If a packet passes the network boundary we need another addressing to differentiate the source and destination systems. The network layer adds a header, which indicate the logical address of the sender and receiver.	If the frames are to be distributed to different systems on the network, the data link layer adds the header, which defines the source machine' s address and the destination Machine' s address.

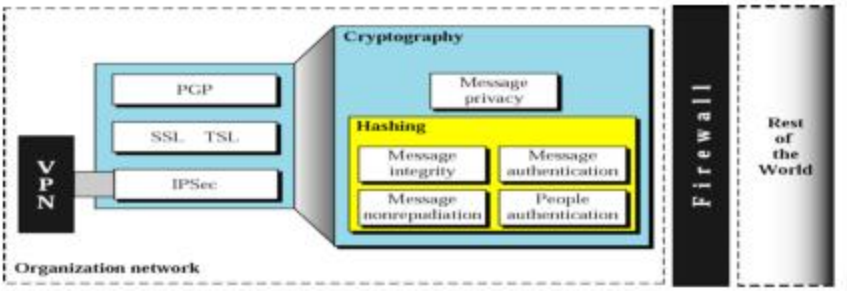
21. Discuss the basic model of FTP.

The client has three components: the user interface, the client control process, and the client data transfer process. The server has two components: the server control process and the server data transfer process. The control connection is made between the control processes. The data connection is made between the data transfer processes.

22.What is a digital signature?

Digital signature is a method to authenticate the sender of a message. It is similar to that of signing transactions documents when you do business with a bank. In network transactions, you can create an equivalent of an electronic or digital signature by the way you send data.

23.Draw the diagram foe explain security



24. Define Cryptography

- Cryptography refers to the science and art of transforming messages to make them secure and immune to attacks.
- Original message before being transformed is called **plaintext**.
- After the message is transformed, is called **ciphertext**.
- An encryption algorithm transforms the plaintext to ciphertext; a decryption algorithm transforms the ciphertext back to plaintext.
- The term cipher is used to refer to encryption and decryption algorithms.

25. What are the types of DNS Message

- Two types of messages

  Query: header and question records

  - Response: Header, question records, answer records, authoritative records, and additional records.

26. What is TELNET PROTOCOL?

A TELNET connection is a Transmission Control Protocol (TCP) connection used to transmit data with interspersed TELNET control information.

The TELNET Protocol is built upon three main ideas: first, the concept of a "Network Virtual Terminal"; second, the principle of negotiated options; and third, a symmetric view of terminals and processes.

27. What is PGP?

Pretty Good Privacy. A program using public key encryption popularly used with email

A high security RSA public-key encryption application for MS-DOS, Unix, VAX/VMS, and other computers. It was written by Philip R. Zimmermann of Phil's Pretty Good(tm) Software and later augmented by a cast of thousands, especially including Hal Finney, Branko Lankester, and Peter Guttmann.

28. What is POP3?

POP3 (Post Office Protocol 3) is the most recent version of a standard protocol for receiving e-mail. POP3 is a client/server protocol in which e-mail is received and held for you by your Internet server. POP and IMAP deal with the receiving of e-mail and are not to be confused with the Simple Mail Transfer Protocol (SMTP), a protocol for transferring e-mail across the Internet.

29. What is IMAP.

IMAP (Internet Message Access Protocol) is a standard protocol for accessing e-mail from your local server. IMAP (the latest version is IMAP Version 4) is a client/server protocol in which e-mail is received and held for you by your Internet server.

IMAP can be thought of as a remote file server. POP3 can be thought of as a "store-and-forward" service.

30. What is SSH?

(**S**ecure **Sh**ell) A security protocol for logging into a remote server. SSH provides an encrypted session for transferring files and executing server programs. Also serving as a secure client/server connection for applications such as database access and e-mail SSH supports a variety of authentication methods.

UNIT V

PART-B (16 MARKS)

1.. i) Explain in detail a protocol for electronic mail. (8)

ii) Explain in detail any one SNMP protocol. (8)

2. Describe the main aspects of hyper text transfer protocol(HTTP) in accessing data on world wide web.(16)

3. Explain DNS with reference to its components and working.(16)

4. Write notes on Security protocols PGP & SSH (16)

5. (i) Write notes on IMAP, POP3 (8)

(ii) Discuss in detail about FTP (8)

6. (i) Explain the salient features of the SMTP protocol (12)

(ii) Explain telnet in detail (4)