COMPUTER NETWORKS - II

(Subject Code: 10CS64)

PART - A

UNIT-1 **6 Hours**

Packet Switching Networks - 1: Network services and internal network operation, Packet network topology, Routing in Packet networks, shortest path routing: Bellman-Ford algorithm.

UNIT–2 **6 Hours**

Packet Switching Networks – 2: Shortest path routing (continued), Traffic management at the Packet level, Traffic management at Flow level, Traffic management at flow aggregate level.

UNIT–3 **6 Hours**

TCP/IP-1: TCP/IP architecture, The Internet Protocol, IPv6, UDP.

UNIT–4 **8 Hours**

TCP/IP-2: TCP, Internet Routing Protocols, Multicast Routing, DHCP, NAT and Mobile IP

PART – B

UNIT-5 **7 Hours**

Applications, Network Management, Network Security: Application layer overview, Domain Name System (DNS), Remote Login Protocols, E-mail, File Transfer and FTP, World Wide Web and HTTP, Network management, Overview of network security, Overview of security methods, Secret-key encryption protocols, Public-key encryption protocols, Authentication, Authentication and digital signature, Firewalls.

UNIT–6 **6 Hours**

QoS, VPNs, Tunneling, Overlay Networks: Overview of QoS, Integrated Services QoS, Differentiated services QoS, Virtual Private Networks, MPLS, Overlay networks.

UNIT-7 **7 Hours**

Multimedia Networking: Overview of data compression, Digital voice and compression, JPEG, MPEG, Limits of compression with loss, Compression methods without loss, Overview of IP Telephony, VoIP signaling protocols, Real-Time Media Transport Protocols, Stream control Transmission Protocol (SCTP)

UNIT–8 **6 Hours**

Mobile AdHoc Networks and Wireless Sensor Neworks: Overview of Wireless Ad-Hoc networks, Routing in AdHOc Networks, Routing protocols for and

Security of AdHoc networks, Sensor Networks and protocol structures, Communication Energy model, Clustering protocols, Routing protocols, ZigBee technology and 802.15.4.

Text Books:

1. Communication Networks – Fundamental Concepts & key architectures, Alberto Leon Garcia & Indra Widjaja, 2nd Edition, Tata McGraw-Hill, India (7 - excluding 7.6, 8)
2. Computer & Communication Networks, Nadir F Mir, Pearson Education, India (9, 10 excluding 10.7, 12.1 to 12.3, 16, 17.1 to 17.6, 18.1 to18.3, 18.5, 19, 20)

Reference Books:

1. Behrouz A. Forouzan: Data Communications and Networking, 4th Edition, Tata McGraw-Hill, 2006.
2. William Stallings: Data and Computer Communication, 8th Edition, Pearson Education, 2007.
3. Larry L Peterson and Bruce S Davie: Computer Networks – A Systems Approach, 4th Edition, Elsevier, 2007.
4. Wayne Tomasi: Introduction to Data Communications and Networking, Pearson Education, 2005.

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PART – B

PART-A

UNIT - 1

Packet Switching Networks - 1: Network services and internal network operation, Packet network topology, Routing in Packet networks, shortest path routing: Bellman-Ford algorithm.

UNIT-I
PACKET-SWITCHING NETWORKS:

In Circuit switching the resources allocated for a particular user can not be used by other users at the same time. This approach is inefficient when the amount of information transferred is small or if information is produced in bursts, as is the case in many computer applications. Networks that transfer blocks of information called packets. Packet-switching networks are better matched to computer applications and can also be designed to support real-time applications such as telephony.

1.1 Packet networks can be viewed by two perspectives:

One perspective involves an external view of the network and is concerned with the services that the network provides to the transport layer that operates above it at the end systems. Ideally the definition of the network services is independent of the underlying network and transmission technologies. This approach allows the transport layer and the applications that operate above it to be designed so that they can function over any network that provides the given services.

A second perspective on packet networks is concerned with the internal operation of a network. Here look at the physical topology of a network, the interconnection of links, switches, and routers. The approach that is used to direct information across the network: datagrams, or virtual circuits. The first perspective, involving the services provided to the layer above, does not differ in a fundamental way between broadcast and switched packet networks.

The second perspective, however, is substantially different.

In the case of LANs, the network is small, addressing is simple, and the frame is transferred in one hop so no routing is required. In the case of packet-switching networks, addressing must accommodate extremely large-scale networks and must work in concert with appropriate routing algorithms. These two challenges, addressing and routing, are the essence of the network layer.

NETWORK SERVICES AND INTERNAL NETWORK OPERATION:

The essential function of a network is to transfer information among the users that are attached to the network or internetwork. In Figure 1.1 we show that this transfer may involve a single block of information or a sequence of blocks that are temporally related. In the case of a single block of information, we are interested in having the block delivered correctly to the destination, and we may also be interested in the delay experienced in traversing the network. In the case of a sequence of blocks, we may be interested not only in receiving the blocks correctly and in the right sequence but also in delivering a relatively unimpaired temporal relation.

Figure 1.2 shows a transport protocol that operates end to end across a network. The transport layer peer processes at the end systems accept messages from their higher layer and transfer these messages by exchanging segments end to end across the network. The figure shows the interface at which the network service is visible to the transport layer. The network service is all that matters to the transport layer, and the manner in which the network operates to provide the service is irrelevant.

The network service can be connection-oriented or connectionless. A con- nectionless service is very simple, with only two basic interactions between the transport layer and the

network layer: a request to the network that it send a packet and an indication from the network that a packet has arrived. The user can request transmission of a packet at any time, and does not need to inform the network layer that the user intends to transmit information ahead of time.

A connection-release procedure may also be required to terminate the connection. It is clear that providing connection-oriented service entails greater complexity than connectionless service in the network layer.

It is also possible for a network layer to provide a choice of services to the user of the network. For example, the network layer could offer:

i) best-effort connectionless service

ii) low-delay connectionless service

iii) connection-oriented reliable stream service

iv) connection-oriented transfer of packet with delay and bandwidth guarantees.

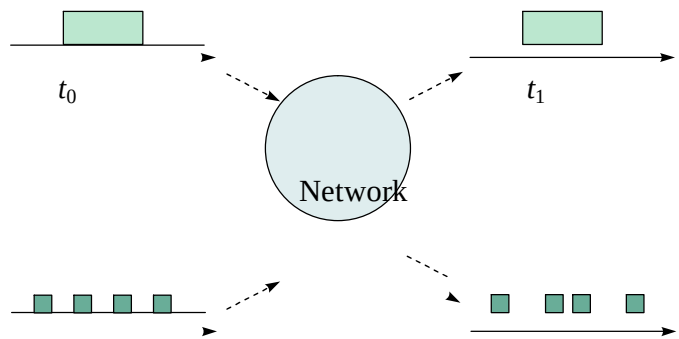


FIGURE 1.1 A network transfers information among user

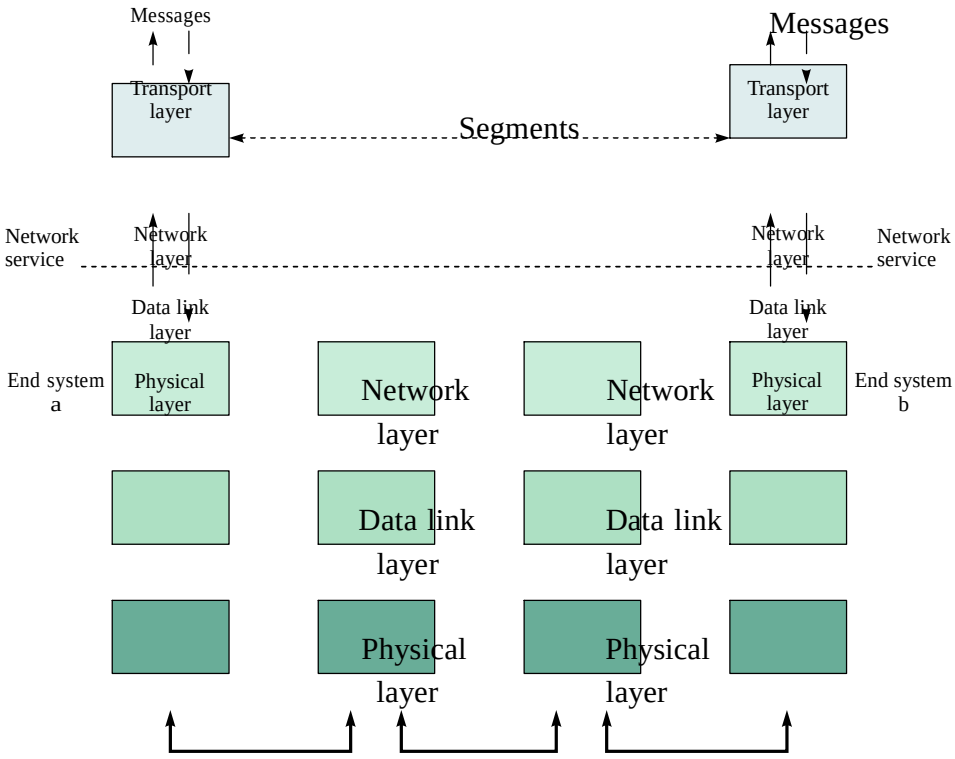


FIGURE 1.2 Peer-to-peer protocols operating end to end across a network protocol stack view

It is easy to come up with examples of applications that can make use of each of these services. However, it does not follow that all the services should be offered by the network layer.

When applied to the issue of choice of network services, the end-to-end argument suggests that functions should be placed as close to the application as possible, since it is the

application that is in the best position to determine whether a function is being carried out completely and correctly. This argument suggests that as much functionality as possible should be located in the transport layer or higher and that the network services should provide the minimum functionality required to meet application performance.

Consider the internal operation of the network. Figure 1 .3 shows the relation between the service offered by the network and the internal operation. T he internal operation of a network is connectionless if packets are transferred within the network as datagrams. Thus in the figure each packet is routed independently. Consequently packets may follow different paths from to and so may arrive out of order. We say that the internal operation of a network is connection-oriented if packets follow virtual circuits that have been established from a source to a destination. Thus to provide communications between and, routing to set up a virtual circuit is done once, and thereafter packets are simply forwarded along the established path. If resources are reserved during connection setup, then bandwidth, delay, and loss guarantees can be provided.

The fact that a network offers connection-oriented service, connectionless service, or both does not dictate how the network must operate internally.

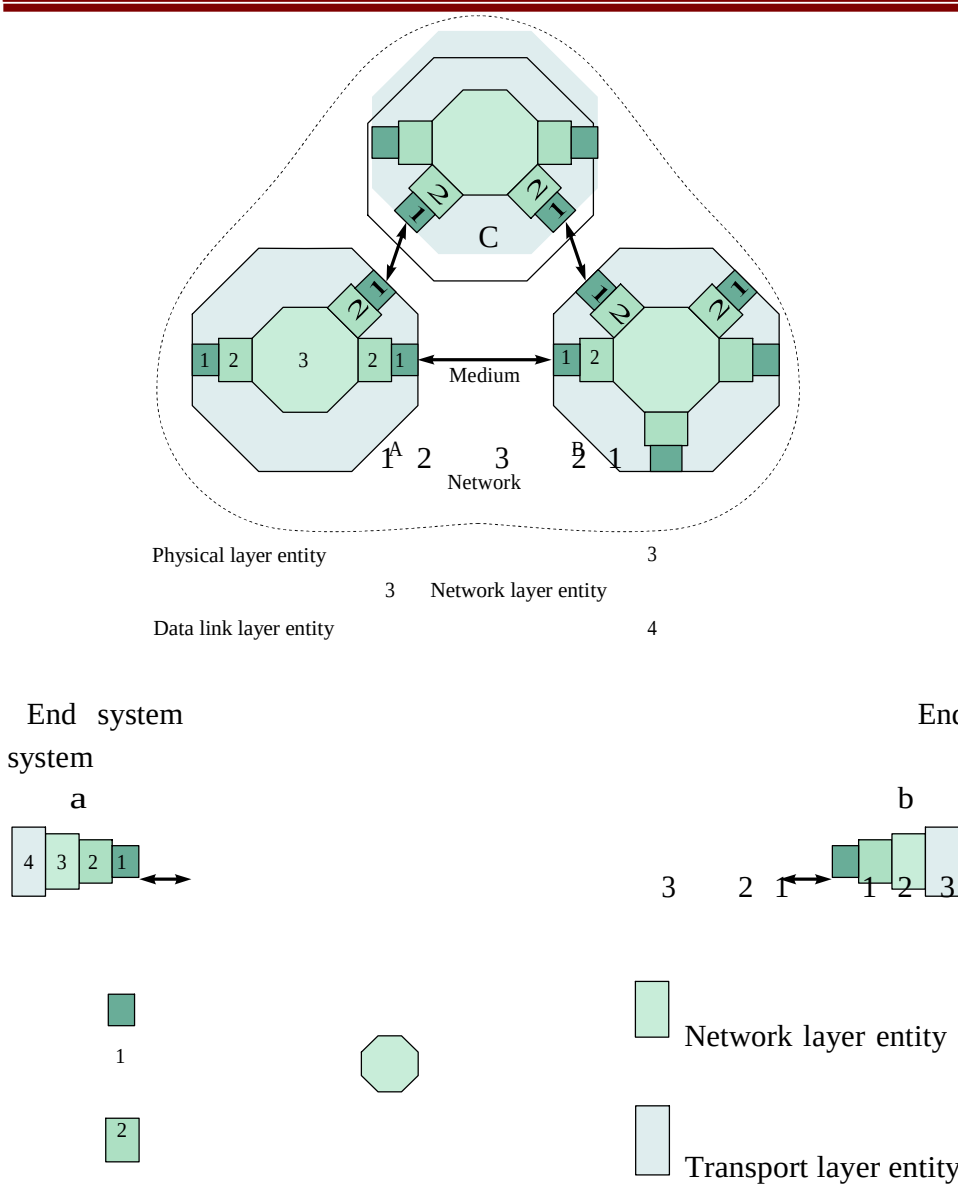


FIGURE 1.3 Layer 3 entities work together to provide network service to layer 4 entities

This reasoning suggests a preference for a connectionless network, which has much lower complexity than a connection-oriented network. The reasoning does allow the possibility for some degree of ``connection orientation'' as a means to ensure that applications can receive the proper level of performance. Indeed current research and standardization efforts can be viewed as an attempt in this direction to determine an appropriate set of network services and an appropriate mode of internal network operation.

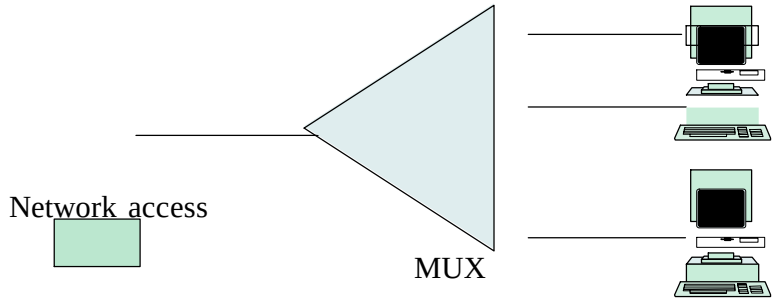
We have concentrated on high-level arguments up to this point. Clearly, functions that need to be carried out at every node in the network must be in the network layer. Thus functions that route and forward packets need to be done in the network layer. Priority and scheduling functions that direct how packets are forwarded so that quality of service is provided also need to be in the network layer. Functions that belong in the edge should, if possible, be imple- mented in the transport layer or higher. A third category of functions can be implemented either at the edge or inside the network. For example, while con- gestion takes place inside the network, the remedy involves reducing input flows at the edge of the network. We will see that congestion control has been imple- mented in the transport layer and in the network layer.

The network layer may therefore be called upon to carry out segmentation inside the network and reassembly at the edge. Alternatively, the network could send error messages to the sending edge, requesting that the packet size be reduced. A more challenging set of functions arises when the ``network'' itself may actually be an internetwork. In this case the network layer must also be concerned not only about differences in the size of the units that the component networks can transfer but also about differences in addressing and in the services that the component networks provide.

1.2 PACKET NETWORK TOPOLOGY:

Let us consider the way in which users access packet networks. Figure 1.4 shows an access multiplexer where the packets from a number of users share a transmission line. This system arises for example, in X.25, frame relay, and ATM networks, where a single transmission line is shared in the access to a wide area packet-switching network. The multiplexer combines the typically bursty f l ows of the individual computers into aggregated flows that make efficient use of the transmission line. Note that different applications within a single computer can generate multiple simultaneous flows to different destinations. From a logical point of view, the link can be viewed as carrying either a single aggregated flow or a number of separate packet flows. The network

access node forwards packets into a backbone packet network.



LANs provide the access to packet-switching networks in many environments. As shown in Figure 1.5a, computers are connected to a shared transmission medium. Transmissions are broadcast to all computers in the network. Each computer is identified by a unique physical address, and so each station listens for its address to receive transmissions. Broadcast and multi- cast transmissions are easily provided in this environment.

Multiple LANs in an organization, in turn, are interconnected into campus networks with a structure such as that shown in Figure 1.6. LANs for a large group of users such as a department are interconnected in an extended LAN through the use of LAN switches, identified by lowercase s in the figure.

Resources such as servers and databases that are primarily of use to this department are kept within the subnetwork. This approach reduces delays in accessing the resources and contains the level of trafic that leaves the subnetwork. Each subnetwork has access to the rest of the organization through a router R that

accesses the campus backbone network. A subnetwork also uses the campus backbone to reach the ``outside world'' such as the Internet or other sites belong- ing to the organization through a gateway router. Depending on the type of organization, the gateway may implement firewall functions to control the traffic that is allowed into and out of the campus network.

Servers containing critical resources that are required by the entire organization are usually located in a data center where they can be easily maintained and

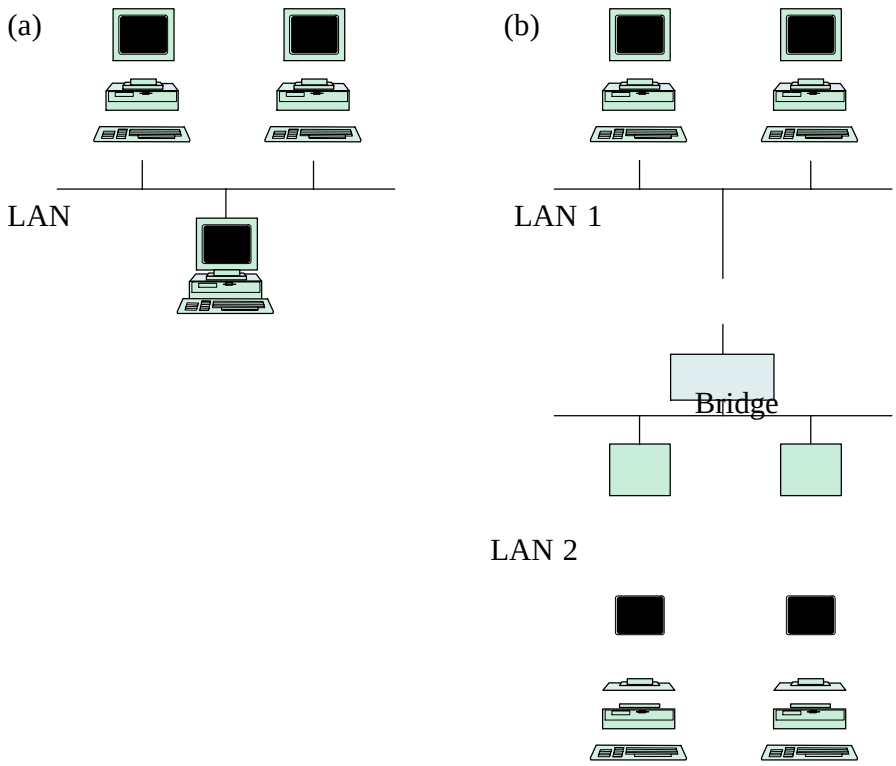


FIGURE 1.5 Local area networks

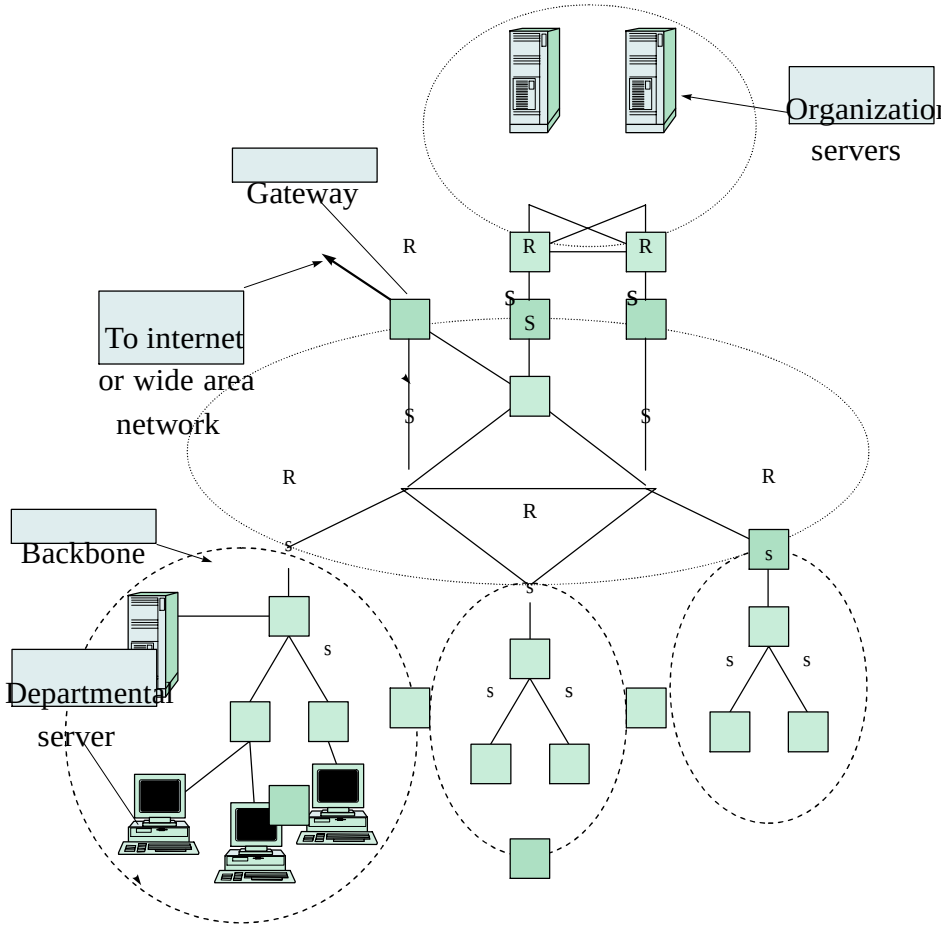


FIGURE 1.6 Campus network

Where security can be enforced. As shown in Figure 1.6, the critical servers may be provided with redundant paths to the campus backbone network. These servers are usually placed near the backbone network to minimize the number of hops required to access them from the rest of the organization.

The trafic within an extended LAN is delivered based on the physical LAN addresses. However, applications in host computers operate on the basis of logical IP addresses. Therefore, the physical address corresponding to an IP address needs to be determined every time an IP packet is to be transmitted over a LAN. This address resolution problem can be solved by using IP address to physical address translation tables.

The routers in the campus network are interconnected to form the campus backbone network, depicted by the mesh of switches, designated S, in Figure 1.6. Typically, for large organizations such as universities these routers are interconnected by using very high speed LANs, for example, Gigabit Ethernet or an ATM network. The routers use the Internet Protocol (IP), which enables them to operate over various data link and network technologies. The routers exchange information about the state of their links to dynamically calculate routing tables that direct packets across the campus network. The routers in the campus network form a domain or autonomous system. The term domain indicates that the routers run the same routing protocol. The term autonomous system is used for one or more domains under a single administration.

Organizations with multiple sites may have their various campus networks interconnected through routers interconnected by leased digital transmission lines or frame relay connections. In this case access to the wide area network may use an access multiplexer such as the one shown in Figure 1.4. In addition the campus network may be connected to an Internet service provider through one or more border routers as shown in Figure 1.7. To communicate with other networks, the autonomous system must

provide information about its network routes in the border routers.

A national ISP provides points of presence in various cities where customers can connect to their network. The ISP has its own national network for interconnecting its POPs. This network could be based on ATM; it might use IP over SONET; or it might use some other network technology. The ISPs in turn exchange traffic as network access points (NAPs), as shown in Figure 1.8a. A NAP is a high-speed LAN or switch at which the routers from different ISPs

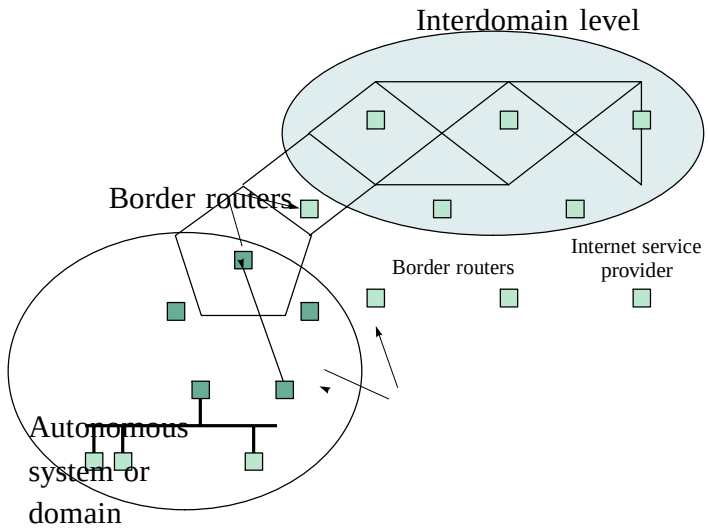


FIGURE 1.7 Intradomain and interdomain levels

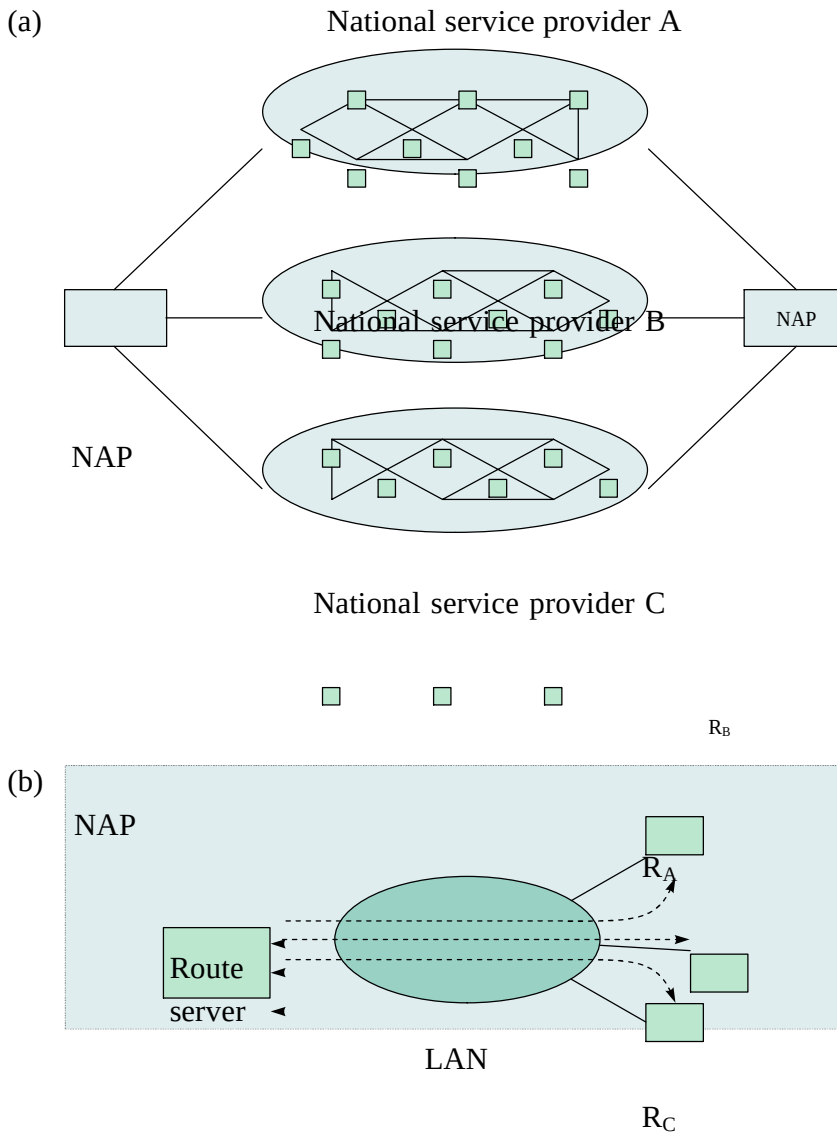


FIGURE 1.8 National ISPs exchange traffic at NAPs

Routing information is exchanged through route servers can exchange trafic, and as such NAPs are crucial to the interconnectivity provided by the Internet.

Thus we see that a multilevel hierarchical network topology arises for the Internet which is much more decentralized than traditional telephone networks. This topology comprises multiple domains consisting of routers interconnected by point-to-point data links, LANs, and wide area networks such as ATM. The principal task of a packet-switching network is to provide connectivity among users. The routing protocols must adapt to changes in network topology due to the introduction of new nodes and links or to failures in equipment. Different routing algorithms are used within a domain and between domains.

1.3 Routing in packet networks:

A packet-switched network consists of nodes (routers or switches) intercom nected by communication links in an arbitrary meshlike fashion as shown in Figure 1.23. As suggested by the figure, a packet could take several possible paths from host A to host B. For example, three possible paths are 1-3-6, 1-4-5-6, and 1-2-5-6. However, which path is the ``best''one?

If the objective is to minimize the number of hops, then path 1-3-6 is the best. If each link incurs a certain delay and the objective function is to minimize the end-to-end delay, then the best path is the one that gives the end-to-end minimum delay. Yet a third objective function involves selecting the path with the greatest available bandwidth. The purpose of the routing algorithm is to identify the set of paths that are best in a sense defined by the network operator. Note that a routing algorithm must have global knowledge about the network state in order to perform its

task.A routing algorithm should seek one or more of the following goals:

1. Rapid and accurate delivery of packets. A routing algorithm must operate correctly; that is, it must be able to find a route to the destination if it exists. In addition, the algorithm should not take an unreasonably long time to find the route to the destination.

2. Adaptability to changes in network topology resulting from node or link failures. In a real network equipment and transmission lines are subject to failures. Thus a routing algorithm must be able to adapt to this situation and reconfigure the routes automatically when equipment fails.

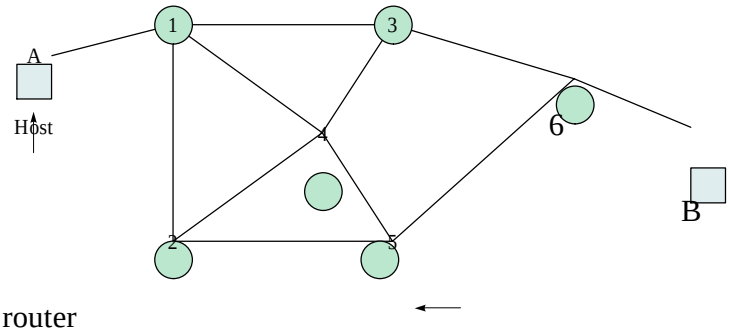


FIGURE 1.23 An example of a packet-switch network

3. Adaptability to varying source-destination traffic loads. Traffic loads are quantities that are changing dynamically. In a period of 24 hours, traffic loads may go into cycles of heavy and light periods. An adaptive routing algorithm would be able to adjust the routes based on the current traffic loads.

4. Ability to route packets away from temporarily congested links. A routing algorithm should try to avoid heavily congested links. Often it is desirable to balance the load on each link.

5. Ability to determine the connectivity of the network. To find optimal routes, the routing system needs to know the connectivity or reachability information.

6. Low overhead. A routing system typically obtains the connectivity information by exchanging control messages with other routing systems. These messages represent an overhead that should be minimized.

Routing Algorithm Classification

One can classify routing algorithms in several ways. Based on their responsive ness, routing can be static or dynamic (or adaptive). In static routing the network topology determines the initial paths. The precomputed paths are then manually loaded to the routing table and remain fixed for a relatively long period of time. Static routing may suffice if the network topology is relatively fixed and the network size is small. Static routing becomes cumbersome as the network size increases. The biggest disadvantage of static routing is its inability to react rapidly to network failures. In dynamic (adaptive) routing each router continu ously learns the state of the network by communicating with its neighbors. Thus a change in a network topology is eventually propagated to all the routers. Based on the information collected, each router can compute the best paths to desired destinations. One disadvantage of dynamic routing is the added complexity in the router.

A routing decision can be made on a per packet basis or during the connec tion setup time. With virtual-circuit packet switching, the path (virtual circuit) is determined during the connection setup phase. Once the virtual circuit is established, all packets belonging to the virtual circuit follow the same route.

Datagram packet switching does not require a connection setup. The route followed by each packet is determined independently.

Shortest-path algorithms:

Network routing is a major component at the network layer and is concerned with the problem of determining feasible paths (or routes) from each source to each destination. A router or a packet-switched node performs two main functions: routing and forwarding.

In the routing function an algorithm finds an optimal path to each destination and stores the result in a routing table. In the forwarding function a router forwards each packet from an input port to the appropriate output port based on the information stored in the routing table.

Shortest-path routing algorithms: the **Bellman-Ford algorithm and Dijkstra's algorithm.**Some other routing approaches are flooding, deflection routing, and source routing.

Most routing algorithms are based on variants of shortest-path algorithms,which try to determine the shortest path for a packet according to some cost criterion. To better understand the purpose of these algorithms, consider a communication network as a graph consisting of a set of nodes (or vertices) and a set of links (or edges, arcs, or branches), where each node represents a router or a packet switch and each link represents a communication channel between two routers. Figure 1.28 shows such an example. Associated with each link is a value that represents the cost of using that link. For simplicity, it is assumed that each link is nondirected. If a link is directed, then the cost must be assigned to each direction. If we define the path cost to be the sum of the link costs along the path, then the shortest path between a pair of nodes is the path with the

least cost. For example, the shortest path from node 2 to node 6 is 2-4-3-6, and the path cost is 4.

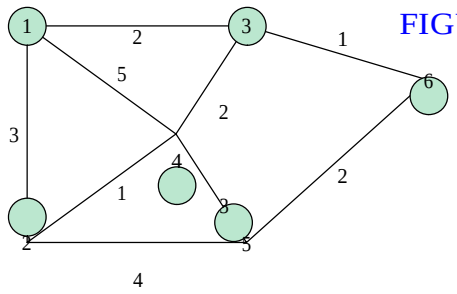


FIGURE 1.28 A sample network with associated link costs

Many metrics can be used to assign a cost to each link, depending on which function is to be optimized. Examples include

1. Cost 1/capacity. The cost is inversely proportional to the link capacity. Here one assigns higher costs to lower-capacity links. The objective is to send a packet through a path with the highest capacity. If each link has equal capacity, then the shortest path is the path with the minimum number of hops.
2. Cost packet delay. The cost is proportional to an average packet delay, which includes queueing delay in the switch buffer and propagation delay in the link. The shortest path represents the fastest path to reach the destination.
3. Cost congestion. The cost is proportional to some congestion measure, for example, traffic loading. Thus the shortest path tries to avoid congested links.

The Bellman-Ford Algorithm

The Bellman-Ford algorithm (also called the Ford-Fulkerson algorithm) is based on a principle that is intuitively easy to understand: If a node is in the shortest path between A and B,

then the path from the node to A must be the shortest path and the path from the node to B must also be the shortest path. As an example, suppose that we want to find the shortest path from node 2 to node 6 (the destination) in Figure 1.28. To reach the destination, a packet from node 2 must first go through node 1, node 4, or node 5. Suppose that someone tells us that the shortest paths from nodes 1, 4, and 5 to the destination (node 6) are 3, 3, and 2, respectively. If the packet first goes through node 1, the total distance (also called total cost) is 3 + 3, which is equal to 6. Through node 4, the total distance is 1 + 3, equal to 4. Through node 5, the total distance is 4 + 2, equal to 6. Thus the shortest path from node 2 to the destination node is achieved if the packet first goes through node 4.

To formalize this idea, let us first fix the destination node. Define D_j to be the current estimate of the minimum cost (or minimum distance) from node j to the destination node and C_{ij} to be the link cost from node i to node j. For example, defined to be zero (that is, C_{ii} fl 0), and the link cost between node i and node k is infinite if node i and node k are not directly connected. For example, C_{15} fl C_{23} fl1 in Figure 1.28. With all these definitions, the minimum cost from node 2 to the destination node (node 6) can be calculated by

$$D_2 \text{ fl } \min\{C_{21} + D_1, C_{24} + D_4, C_{25} + D_5\}$$

$$\text{fl } \min\{3 + 3, 1 + 3, 4 + 2\}$$

$$\text{fl } 4$$

Thus the minimum cost from node 2 to node 6 is equal to 4, and the next node to visit is node 4. One problem in our calculation of the minimum cost from node 2 to node 6 is that we have assumed that the minimum costs from nodes 1, 4, and 5 to the destination were known. In general, these nodes would not know their mini-mum

costs to the destination without performing similar calculations. So let us apply the same principle to obtain the minimum costs for the other nodes. For example,

$$D_1 \text{ fl } \min\{C_{12} + D_2, C_{13} + D_3, C_{14} + D_4\}$$

And $$D_4 \text{ fl } \min\{C_{41} + D_1, C_{42} + D_2, C_{43} + D_3, C_{45} + D_5\}$$

A discerning reader will note immediately that these equations are circular, since D_2 depends on D_1 and D_1 depends on D_2. The magic is that if we keep iterating and updating these equations, the algorithm will eventually converge to the correct result. To see this outcome, assume that initially $D_1 \text{ fl} D_2 \text{ fl} \ldots \text{fl } D_5 \text{ fl} \infty$. Observe that at each iteration, $D_1, D_2, \ldots, D_5$ are nonincreasing. Because the minimum distances are bounded below, eventually $D_1, D_2, \ldots, D_5$ must converge.

Now if we define the destination node, we can summarize the Bellman-Ford algorithm as follows:

1. Initialization

$$D_i \text{ fl} \infty; \text{ for all } i \neq d$$

$$D_d \text{ fl } 0$$

2. Updating: For each $i \neq d$,

$$D_i \text{ fl } \min_j\{C_{ij} + D_j\}, \text{ for all } j \neq i$$

Repeat step 2 until no more changes occur in the iteration.

UNIT – 2

Packet Switching Networks – 2: Shortest path routing (continued), Traffic management at the Packet level, Traffic management at Flow level, Traffic management at flow aggregate level.

UNIT – 2

PACKET SWITCHING NETWORKS – 2:

2.1 Traffic Management at the packet level:

Traffic management is concerned with the delivery of QoS to specific packet flows. Traffic management entails mechanisms for managing the flows in a network to control the load that is applied to various links and switches. Traffic management also involves the setting of priority and scheduling mechanisms at switches, routers, and multiplexers to provide differentiated treatment for packets and cells belonging to different classes, flows, or connections. It also may involve the policing and shaping of traffic flows as they enter the network.

The dashed arrows show packets from other flows that ``interfere"with the packet of interest in the sense of contending for buffers and transmission along the path. We also note that these interfering flows may enter at one multi- plexer and depart at some later multiplexer, since in general they belong to different source-destination pairs and follow different paths through the network.

The performance experienced by a packet along the path is the accumulation of the performance experienced at the N multiplexers. For example, the total end-to-end delay is the sum of the delays experienced at each multiplexer. Therefore, the average end-to-end delay is the sum of the individual average delays. On the other hand, if we can guarantee that the delay at each multiplexer can be kept below some upper bound, then the end-to-end delay can be kept below the sum of the upper bounds at the various multiplexers. The jitter experienced by packets is also of interest. The jitter measures the variability in the packet delays and is typically measured in terms of the difference of the minimum

delay and some maximum value of delay.

Note that the discussion here is not limited solely to connection-oriented packet transfer. In the case of connectionless transfer of packets, each packet will experience the performance along the path traversed. On the other hand, this analysis will hold in connectionless packet switching networks for the period of time during which a single path is used between a source and a destination. If these paths can be ``pinned down'' for certain flows in a connectionless network, then the end-to-end analysis is valid.

Packet-switching networks are called upon to support a wide range of services with diverse QoS requirements. To meet the QoS requirements of multiple services, an ATM or packet multiplexer must implement strategies for managing how cells or packets are placed in the queue or queues, as well as control the transmission bit rates that are provided to the various information flows. We now consider a number of these strategies.

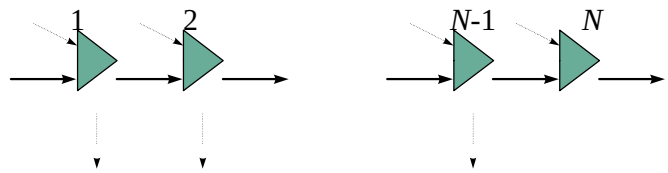


FIGURE 1.41 The end-to-end QoS of a packet along a path traversing N hops

2.2 Traffic management at Flow level:

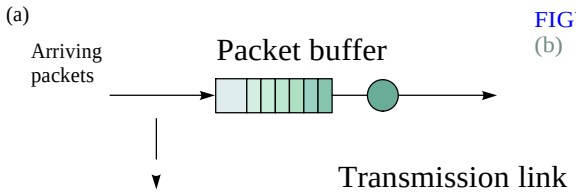
FIFO and Priority Queues: The simplest approach to managing a multiplexer involves first-in, first-out (FIFO) queuing where all arriving packets are placed in a common queue and transmitted in order of arrival, as shown in Figure 1.42a. Packets are discarded when they arrive at a full buffer. The delay and loss experienced by packets in a FIFO system depend on the inter arrival times and on the packet lengths. As inter arrivals become more bursty or packet lengths more variable, performance will deteriorate. Because FIFO queuing treats all packets in the same manner, it is not possible to provide different information flows with different qualities of service. FIFO systems are also subject to hogging, which occurs when a user sends packets at a high rate and fills the buffers in the system, thus depriving other users of access to the multiplexer.

A FIFO queuing system can be modified to provide different packet-loss performance to different traffic types. Figure 1.42b shows an example with two classes of traffic. When the number of packets reaches a certain threshold, arrivals of lower access priority (Class 2) are not allowed into the system. Arrivals of higher access priority (Class 1) are allowed as long as the buffer is not full. As a result, packets of lower access priority will experience a higher packet-loss probability.

Head-of-line (HOL) priority queuing is a second approach that involves defining a number of priority classes. A separate queue is maintained for each priority class. As shown in Figure 1.43, each time the transmission line becomes available the next packet for transmission is selected from the head of the line of the highest priority queue that is not empty.

For example, it does not allow for providing some egree of guaranteed access to transmission bandwidth to the lower

priority classes. Another problem is that it does not discriminate between users of the same priority. Fairness problems can arise here when a certain user hogs the bandwidth by sending an excessive number of packets.



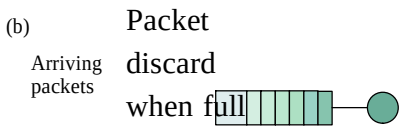
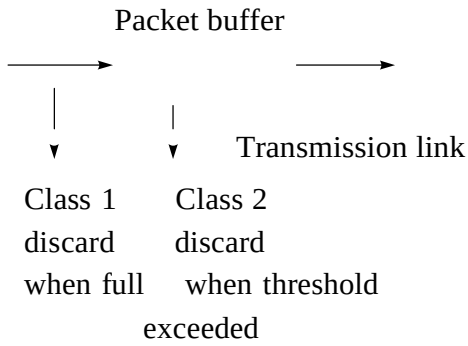


FIGURE 1.42 (a) FIFO queueing; (b)



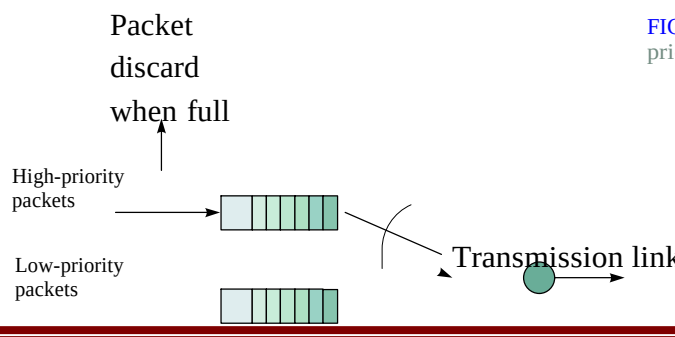
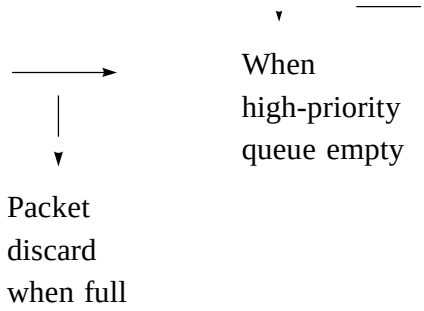


FIGURE 1.43 HOL priority



A third approach to managing a multiplexer, shown in Figure 1.44, involves sorting packets in the queue according to a priority tag that reflects the urgency with which each packet needs to be transmitted. This system is very flexible because the method for defining priority is open and can even be defined dynamically. For example, the priority tag could consist of a priority class followed by the arrival time of a packet to a multiplexer.

A third important example that can be implemented by the approach is fair queueing and weighted fair queueing, which are discussed next.

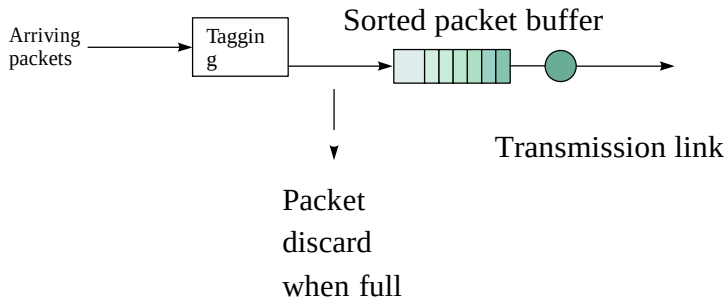


FIGURE 7.44 Sorting packets according to priority tag

Fair Queueing: Fair queueing attempts to provide equitable access to transmission bandwidth. Each user flow has its own logical queue. In an ideal system the transmission bandwidth, say, C bits/second, is divided equally among the queues that have packets to transmit.The contents of each queue can then be viewed as a fluid that is drained continuously. Fair queueing prevents the phenomenon of hogging, which occurs when an information flow receives an unfair share of the bit

Fair queueing is ``fair'' in the following sense. In the ideal fluid flow situation, the transmission bandwidth is divided equally among all nonempty queues. Thus if the total number of flows in the system is n and the transmission capacity is C, then each flow is guaranteed at least Cfln bits/second. In general, the actual transmission rate experienced may be higher because queues will be empty from time to time, so a share larger than Cfln bps is received at those times.

In practice, dividing the transmission capacity exactly equally is not possible. As shown in Figure 1.45 one approach could be to service each nonempty queue one bit at a time in round-robin fashion. However, decomposing the resulting bit stream into the component packets would require the introduction of framing information and extensive processing at the demultiplexer. In the case of ATM, fair queueing can be approximated in a relatively simple way. Because in ATM all packets are the same length, the multiplexer need only service the nonempty queues one packet at a time in round-robin fashion. User flows are then guar- anteed equal access to the transmission bandwidth.

Figure1.46 illustrates the differences between ideal or ``fluid flow'' and packet-by-packet fair queueing. The figure assumes that queue 1 and queue 2 each has a single L-bit packet to transmit at t = 0 and that no subsequent

packets arrive. Assuming a capacity of C = L bits/second fl 1 packet/second,
the fluid-flow system transmits each packet at a rate of 1/2 and therefore.

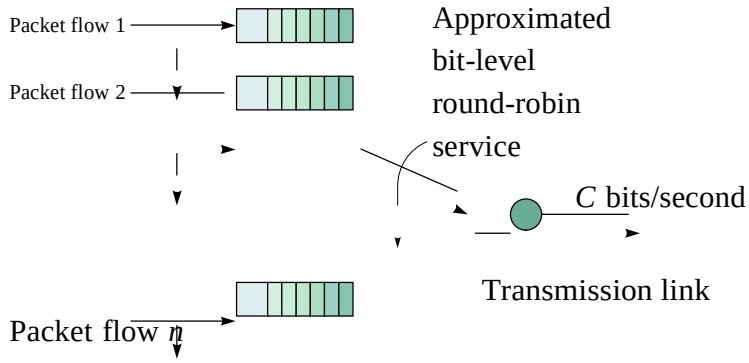


FIGURE 1.45 Fair queueing

completes the transmission of both packets exactly at time t = 2 seconds. The bit-by-bit system (not shown in the figure) would begin by transmitting one bit from queue 1, followed by one bit from queue 2, and so on. On the other hand, the packet-by-packet fair-queueing system transmits the packet from queue 1 first and then transmits the packet from queue 2, so the packet completion times are 1 and 2 seconds. In this case the first packet is 1 second too early relative to the completion time in the fluid system.

Approximating fluid-flow fair queueing is not as straightforward when pack- ets have variable lengths. If the different user queues are serviced one packet at a time in round-robin fashion, we do not necessarily obtain a fair allocation of transmission bandwidth. For example, if the packets of one flow are twice the size of packets in another flow, then in the long run the first flow will obtain twice

the bandwidth of the second flow. A better approach is to transmit packets from the user queues so that the packet completion times approximate those of a fluid- flow fair queueing system. Each time a packet arrives at a user queue, the completion time of the packet is derived from a fluid-flow fair-queueing system. This number is used as a finish tag for the packet. Each time the transmission of a packet is completed, the next packet to be transmitted is the one with the smallest finish tag among all of the user queues. We refer to this system as a packet-by-packet fair-queueing.

Assume that there are n flows, each with its own queue. Suppose for now that each queue is served one bit at a time. Let a round consist of a cycle in which all n queues are offered service as shown in Figure 7.47. The actual duration of a given round is the actual number of queues $n_{active}…t†$ that have information to transmit.
When the number of active queues is large, the duration of a round is large; when the number of active queues is small, the rounds are short in duration.

Now suppose that the queues are served as in a fluid-flow system. Also suppose that the system is started at t ˆ 0. Let R…t† be the number of the rounds at time t, that is, the number of cycles of service to all n queues. However, we let R…t† be a continuous function that increases at a rate that is inversely proportional to the number of active queues; that is:

$$dR…t†fldt = Cfln_{active}…t†$$

where C is the transmission capacity. Note that R…t† is a piecewise linear function that changes in slope each time the number of active queues changes. Each time R…t† reaches a new integer value marks an instant at which all the queues have been given an

equal number of opportunities to transmit a bit.

k Let us see how we can calculate the finish tags to approximate fluid-flow fair queueing. Suppose the kth packet from flow i arrives at an empty queue at time t^i and suppose that the packet has length P…i; k†. This packet will complete its transmission when P…i; k† rounds have elapsed, one round for each bit in the packet. Therefore, the packet completion time will be the value of time t
when the R…t† reaches the value:

$$F\ldots i; k\dagger \hat{} R..{}_k t^i \dagger + P\ldots i; k\dagger$$

We will use F…i; k† as the finish tag of the packet. On the other hand, if the kth packet from the ith flow arrives at a nonempty queue, then the packet will have a finish tag F…i; k† equal to the finish tag of the previous packet in its queue
F…i; k 1† plus its own packet length P…i; k†; that is:

$$F\ldots i; k\dagger \hat{} F\ldots i; k \quad 1\dagger + P\ldots i; k\dagger$$

The two preceding equations can be combined into the following compact
equation:

$$F\ldots i; k\dagger = \max fF\ldots i; k \quad 1\dagger;{}_k R\ldots t^i \dagger g + P\ldots i; k\dagger \quad \text{for fair queueing:}$$

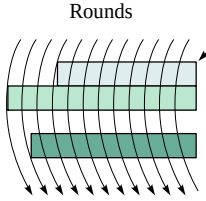


FIGURE 1.47 Computing the finishing time in packet-by-packet fair queueing and weighted fair queueing

We reiterate: The actual packet completion time for the kth packet in flow I in a fluid-flow fair-queueing system is the time t when R…t† reaches the value F…i; k†. The relation between the actual completion time and the finish tag is not straightforward because the time required to transmit each bit varies according to the number of active queues.

As an example, suppose that at time t 0 queue 1 has one packet of length one unit and queue 2 has one packet of length two units. A fluid-flow system services each queue at rate 1/2 as long as both queues remain nonempty. As shown in Figure 1.48, queue 1 empties at time t = 2. Thereafter queue 2 is served at rate 1 until it empties at time t = 3. In the packet-by-packet fair-queueing system, the finish tag of the packet of queue 1 is F…1; 1† = R…0† + 1 = 1. The finish tag of the packet from queue 2 is F…2; 1† = R…0† + 2 = 2. Since the finish tag of the packet of queue 1 is smaller than the finish tag of queue 2, the system will service queue 1 first. Thus the packet of queue 1 completes its transmissions at time t = 1 and the packet of queue 2 completes its transmissions at t = 3.

2.4 Traffic management at flow aggregate level:

WEIGHTED FAIR QUEUEING

Weighted fair queueing addresses the situation in which different users have different requirements. As before, each user flow has its own queue, but each user flow also has a weight that determines its relative share of the bandwidth. Thus if queue 1 has weight 1 and queue 2 has weight 3, then when both queues
are nonempty, queue 1 will receive 1fl…1 + 3† = 1fl4 of the bandwidth and queue 2 will receive 3/4 of the bandwidth. Figure 7.49 shows the completion times for the

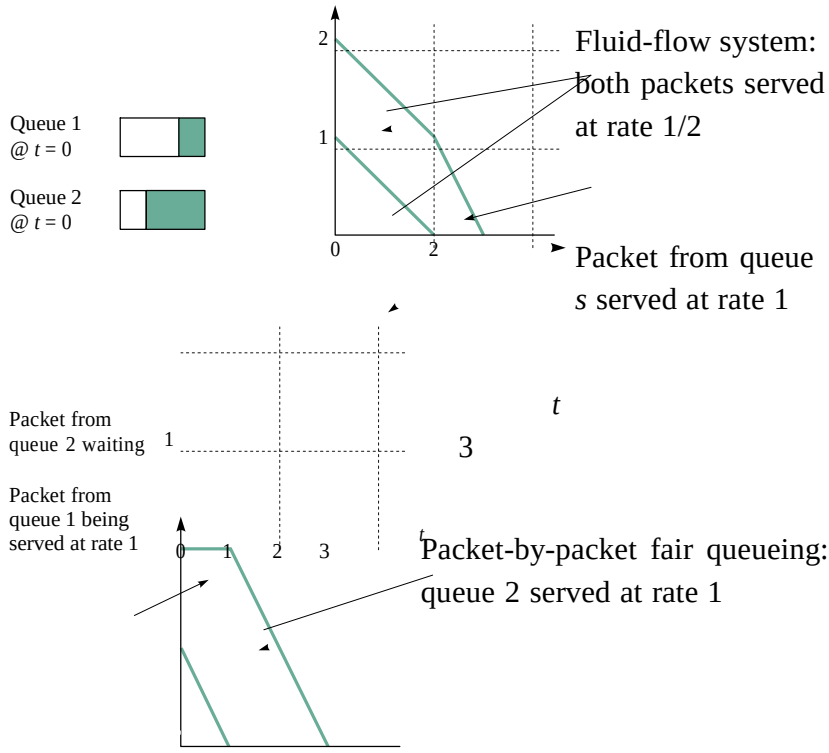


FIGURE 2.48 Fluid flow and packet-by-packet fair queueing (two packets of different lengths)

-flow case where both queues have a one-unit length packet at time t= 0.The transmission of the packet from queue 2 is now completed at time t =4fl3, and the packet from queue 1 is completed at t = 2. The bit-by-bit approximation to weighted fair queuing would operate by allotting each queue a different number of bits/round. In the preceding example, queue 1 would receive 1 bit/round and queue 2 would receive 3 bits/round.

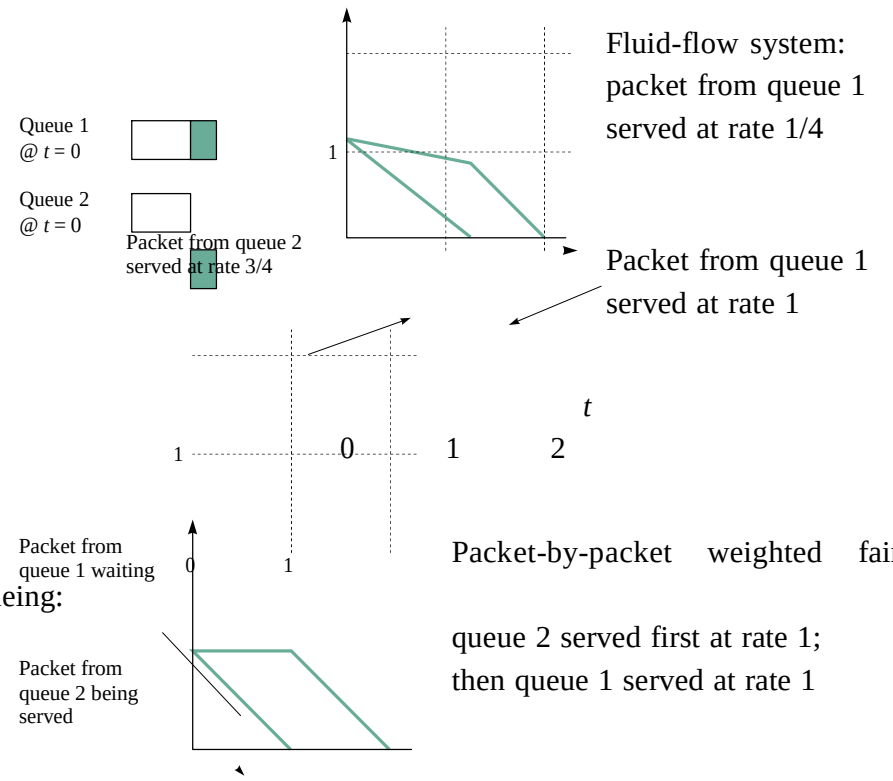


FIGURE2.49 Fluid flow and packetized, weighted fair queueing

Weighted fair-queueing systems are a means for providing QoS guarantees. Suppose a given user flow has weight w_i and suppose that the sum of the weights of all the user flows is W. In the worst case when all the user queues are non- empty, the given user flow will receive a fraction $w_i flW$ of the bandwidth C.

When other user queues are empty, the given user flow will receive a greater

share.

UNIT – 3

TCP/IP-1: TCP/IP architecture, The Internet Protocol, IPv6, UDP.

UNIT – 3

TCP/IP-1:

The Internet Protocol (IP) enables communications across a vast and heterogeneous collection of networks that are based on different technologies. Any host computer that is connected to the Internet can communicate with any other computer that is also connected to the Internet. The Internet therefore offers ubiquitous connectivity and the economies of scale that result from large deployment.

The Internet offers two basic communication services that operate on top of IP: Transmission control protocol (TCP) reliable stream service and user datagram protocol (UDP) datagram service. Any application layer protocol that operates on top of either TCP or UDP automatically operates across the Internet. Therefore the Internet provides a ubiquitous platform for the deployment of network-based services.

3.1 THE TCP/IP ARCHITECTURE

The TCP/IP protocol suite usually refers not only to the two most well-known protocols called the Transmission Control Protocol (TCP) and the Internet Protocol (IP) but also to other related protocols such as the User Datagram Protocol (UDP), the Internet Control Message Protocol (ICMP) and the basic applications such as HTTP, TELNET, and FTP. The basic structure of the TCP/IP protocol suite is shown in Figure 3 .1. Application layer protocols such as SNMP and DNS send their messages using UDP. The PDUs exchanged by the peer TCP protocols are called TCP segments or segments, while those

exchanged by UDP protocols are called UDP datagrams or datagrams. IP multiplexes TCP segments and UDP datagrams and performs fragmentation, if necessary, among other tasks to be discussed below. The protocol data units exchanged by IP protocols are called IP packets or packets.[1] IP packets are sent to the network interface for delivery across the physical network. At the receiver, packets passed up by the network interface are demultiplexed to the appropriate protocol (IP, ARP, or RARP). The receiving IP entity needs to determine whether a packet has to be sent to TCP or UDP. Finally, TCP (UDP) sends each segment (datagram) to the appropriate application based on the port number. The physical network can be implemented by a variety of technologies such as Ethernet, token ring, ATM, PPP over various transmission systems, and others.

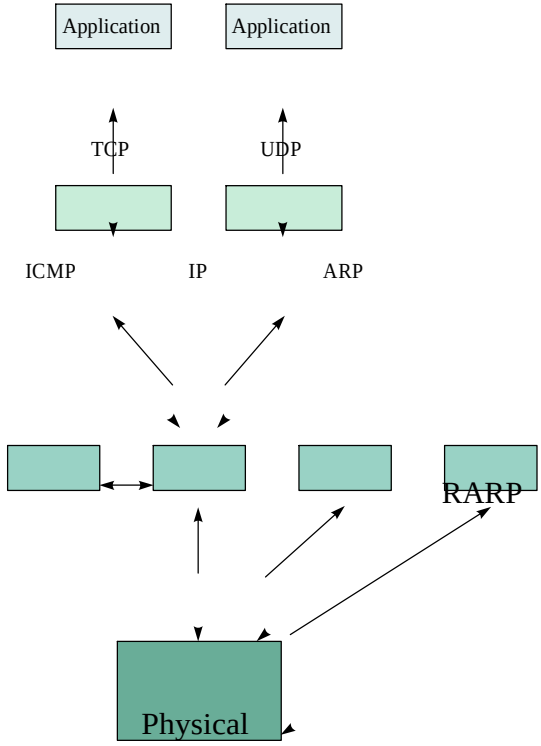


FIGURE 3.1 TCP/IP protocol suite

The PDU of a given layer is encapsulated in a PDU of the layer below as shown in Figure 8.2. In this figure an HTTP GET command is passed to the TCP layer, which encapsulates the message into a TCP segment. The segment header contains an ephemeral port number for the client process and the well-known port 80 for the HTTP server process. The TCP segment in turn is passed to the IP layer where it is encapsulated in an IP packet. The IP packet header contains an IP network address for the sender and an IP network address for the destination. IP network addresses are said to be logical because they are defined in terms of the logical topology of the routers and end systems. The IP packet is then passed through the network interface and encapsulated into a PDU of the underlying network. In Figure 2.2 the IP packet is encapsulated into an Ethernet LAN frame. The frame header contains physical addresses that identify the physical endpoints for the sender and the receiver. The logical IP addresses need to be converted into specific physical addresses to carry out the transfer of bits from one device to the other. This conversion is done by an address resolution protocol.

Each host in the Internet is identified by a globally unique IP address. An IP address is divided into two parts: a network ID and a host ID. The network ID

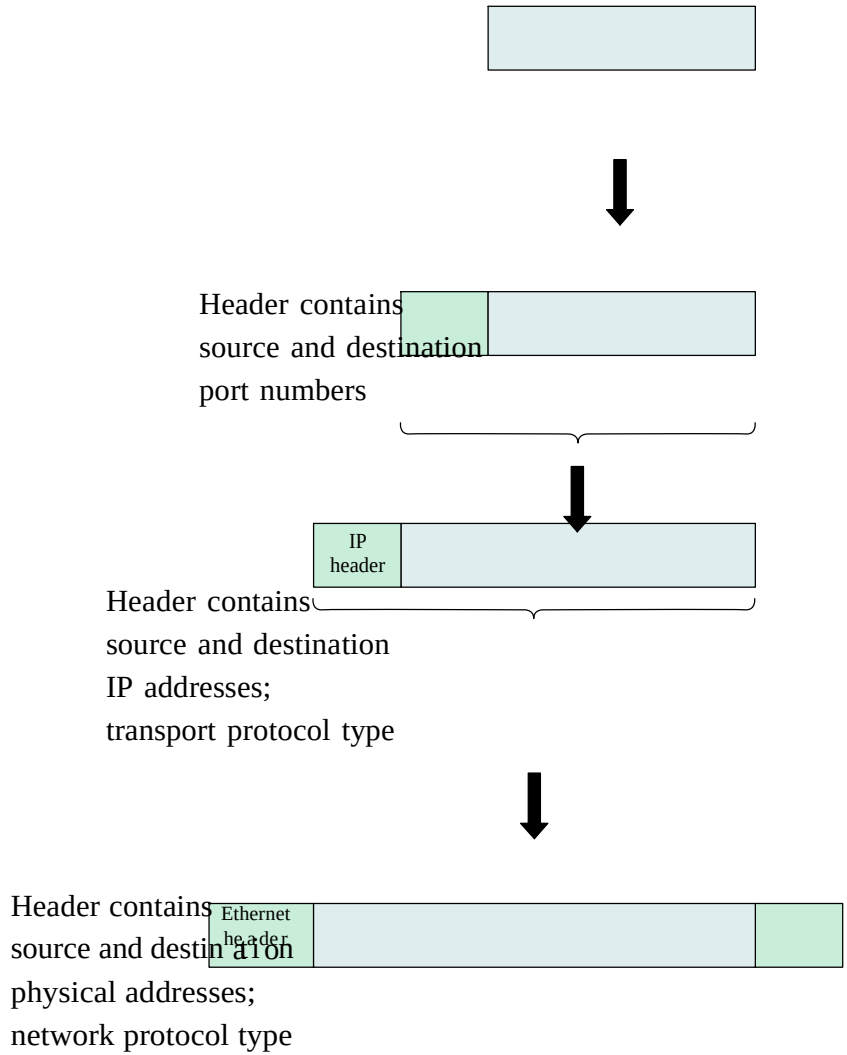


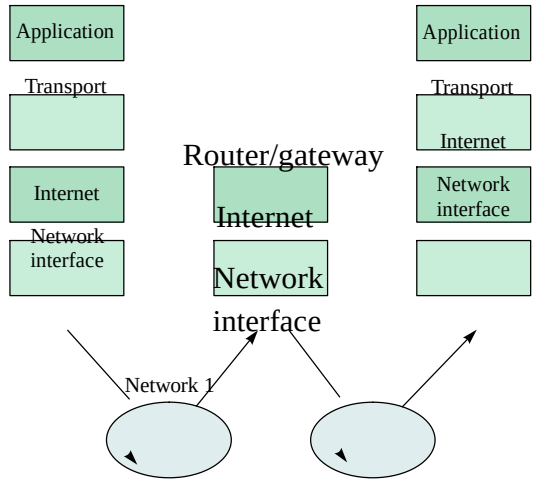
FIGURE 3.2 Encapsulation of PDUs in TCP/IP and addressing information in the headers

must be obtained from an organization authorized to issue IP addresses. The Internet layer provides for the transfer of information across multiple networks through the use of routers, as shown in Figure 2.3. The Internet layer provides a single service, namely, best-effort connectionless packet transfer. IP packets are exchanged between routers without a connection setup; they are routed indepen- dently, and may traverse different paths. The gateways that interconnect the intermediate networks may discard packets when they encounter congestion. The responsibility for recovery from these losses is passed on to the transport layer.

The network interface layer is particularly concerned with the protocols that are used to access the intermediate networks. At each gateway the network protocol is used to encapsulate the IP packet into a packet or frame of the underlying network or link. The IP packet is recovered at the exit router of the given network. This router must determine the next hop in the route to the destination and then encapsulate the IP packet or frame of the type of the next network or link. This approach provides a clear separation of the Internet layer from the technology-dependent network interface layer. This approach also allows the Internet layer to provide a data transfer service that is transparent in the sense of not depending on the details of the underlying networks. Different network technologies impose different limits on the size of the blocks that they can handle. IP must accommodate the maximum transmission unit of an under-
lying network or link by implementing segmentation and reassembly as needed.

To enhance the scalability of the routing algorithms and to control the size of the routing tables, additional levels of hierarchy are introduced in the IP

FIGURE 3.3 The Internet and network interface layers



addresses. Within a domain the host address is further subdivided into a subnet- work part and an associated host part. This system facilitates routing within a domain, yet can remain transparent to the outside world. At the other extreme, addresses of multiple domains can be aggregated to create supernets.

3.2 The Internet protocol:

The Internet Protocol (IP) is the heart of the TCP/IP protocol suite. IP corre- sponds to the network layer in the OSI reference model and provides a connec- tionless and best-effort delivery service to the transport layer. Recall that a connectionless service does not require a virtual circuit to be

established before data transfer can begin. The term best-effort indicates that IP will try its best to forward packets to the destination, but does not guarantee that a packet will be delivered to the destination. The term is also used to indicate that IP does not make any guarantee on the QoS.[2] An application requiring high reliability must implement the reliability function within a higher-layer protocol.

IP Packet

To understand the service provided by the IP entity, it is useful to examine the IP packet format, which contains a header part and a data part. The format of the IP header is shown in Figure 3.4.

The header has a fixed-length component of 20 bytes plus a variable-length component consisting of options that can be up to 40 bytes. IP packets are

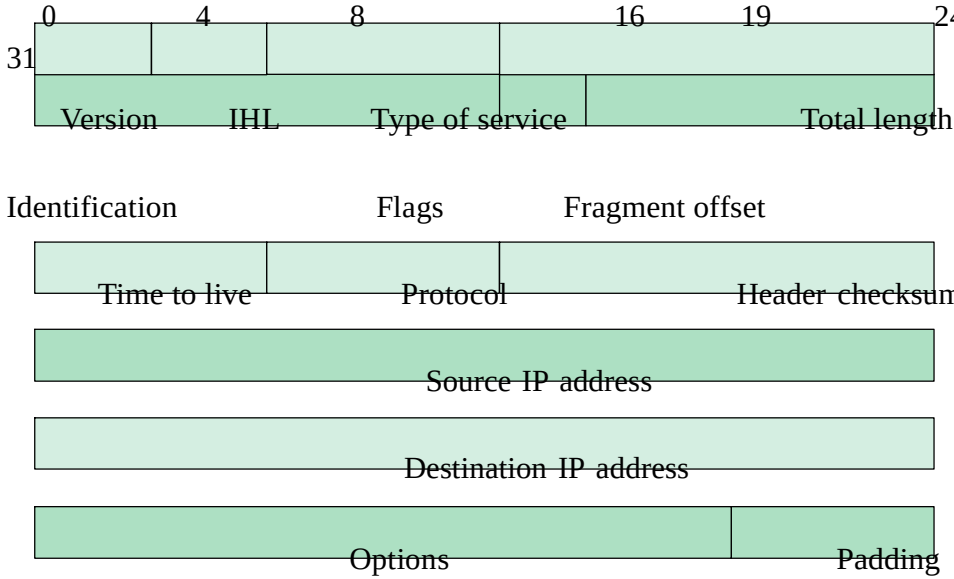


FIGURE 3.4 IP version 4 headers

transmitted according to network byte order: bits 0±7 first, then bits 8±15, then bits 16±23, and finally bits 24±31 for each row. The meaning of each field in the header follows.

Version: The version field indicates the version number used by the IP packet so that revisions can be distinguished from each other. The current IP version is 4. Version 5 is used for a real-time stream protocol called ST2, and version 6 is used for the new generation IP know as IPng or IPv6 (to be discussed in the following section).

Internet header length: The Internet header length (IHL) specifies the length of the header in 32-bit words. If no options are present, IHL will have a value of 5. The length of the options field can be determined from IHL.

Type of service: The type of service (TOS) field specifies the priority of the packet based on delay, throughput, reliability, and cost requirements. Three bits are assigned for priority levels (called ``precedence'') and four bits for the specific requirement (i.e., delay, throughput, reliability, and cost). For example, if a packet needs to be delivered to the destination as soon as possible, the transmitting IP module can set the delay bit to one and use a high-priority level. In practice most routers ignore this field.

Recent work in the Differentiated Service Working Group of IETF tries to redefine the TOS field in order to support other services that are better than the basic best effort.

Total length: The total length specifies the number of bytes of the IP packet including header and data. With 16 bits assigned to this field, the max- imum packet length is 65,535 bytes. In practice the maximum possible length is very rarely used, since most physical networks have their own length limitation. For example, Ethernet limits the payload length to 1500 bytes.

Identification, flags, and fragment offset: These fields are used for fragmentation and reassembly and are discussed below.

Time to live: The time-to-live (TTL) field is defined to indicate the amount of time in seconds the packet is allowed to remain in the network. However, most routers interpret this field to indicate the number of hops the packet is allowed to traverse in the network. Initially, the source host sets this field to some value. Each router decrements this value by one. If the value reaches zero before the packet reaches the destination, the router discards the packet and sends an error message back to the source. With either interpretation, this field prevents packets from wandering aimlessly in the Internet.

Protocol: The protocol field specifies the protocol that is to receive the IP data at the destination host. Examples of the protocols include TCP (pro- tocol fl 6), UDP (protocol fl 17), and ICMP (protocol fl 1). Header checksum: The header checksum field verifies the integrity of the header of the IP packet. The data part is not verified and is left to upper-layer protocols. If the verification process fails, the packet is simply discarded. To compute the header checksum, the sender first sets the header checksum field to 0 and then applies the Internet checksum algorithm discussed in Chapter 3. Note that when a router decrements the TTL field, the router must also recompute the header checksum field. Source IP address and destination IP address: These fields contain the addresses of the source and destination hosts. The format of the IP address is discussed below.

Options: The options field, which is of variable length, allows the packet to request special features such as security level, route to be taken by the packet, and timestamp at each router. The options field is rarely used. Router alert is a new option introduced to alert routers to look inside the IP packet. The option is intended for new protocols that require rela- tively complex processing in

routers along the path [RFC 2113].

Padding: This field is used to make the header a multiple of 32-bit words.

When an IP packet is passed to the router by a network interface, the following processing takes place. First the header checksum is computed and the fields in the header are checked to see if they contain valid values. Next IP fields that need to be changed are updated. For example, the TTL and header checksum fields always require updating. The router then identifies the next loop for the IP packet by consulting its routing tables. The IP packet is then for- warded along the next loop.

3.3 IPv6

IP version 4 has played a central role in the internetworking environment for many years. It has proved flexible enough to work on many different networking technologies. However, it has become a victim of its own success Đexplosive growth! In the early days of the Internet, people using it were typically researchers and scientists working in academia, high-tech companies, and research laboratories, mainly for the purpose of exchanging scientific results through e- mails. In the 1990s the World Wide Web and personal computers shifted the user of the Internet to the general public. This change has created heavy demands for new IP addresses, and the 32 bits of the current IP addresses will be exhausted sooner or later.

In the early1990s the Internet Engineering Task Force (IETF) began to work on the successor of IP version 4 that would solve the address exhaustion problem and other scalability problems. After several proposals were investigated, the new IP version was recommended in late 1994. The new version is called

IPv6 for IP version 6 (also called IP next generation or IPng). IPv6 was designed to interoperate with IPv4 since it would likely take many years to complete the transition from version 4 to version 6. Thus IPv6 should retain the most basic service provided by IPv4Ða connectionless delivery service. On the other hand, IPv6 should also change the IPv4 functions that do not work well and support new emerging applications such as real-time video conferencing, etc. Some of the changes from IPv4 to IPv6 include Longer address fields: The length of address field is extended from 32 bits to128 bits. The address structure also provides more levels of hierarchy.

Theoretically, the address space can support up to 3:4 10^{38} hosts. Simplified header format: The header format of IPv6 is simpler than that of IPv4. Some of the header fields in IPv4 such as checksum, IHL, identifica- tion, flags, and fragment offset do not appear in the IPv6 header.

Flexible support for options: The options in IPv6 appear in optional extension headers that are encoded in a more efficient and flexible fashion than they were in IPv4.

Flow label capability: IPv6 adds a ``flow label" to identify a certain packet ``flow" that requires a certain QoS. Security: IPv6 supports built-in authentication and confidentiality. Large packets: IPv6 supports payloads that are longer than 64 K bytes, called jumbo payloads. Fragmentation at source only: Routers are not allowed to fragment packets. If a packet needs to be fragmented, it must be done at the source. No checksum field: The checksum field has been removed to reduce packet processing time in a router. Packets carried by the physical network such as Ethernet, token ring, X.25, or ATM are typically already checked. Furthermore, higher-layer protocols such as TCP and UDP also perform their own verification. Thus removing the checksum field probably would not introduce a serious problem in most situations.

Header Format

The IPv6 header consists of a required basic header and optional extension headers. The format of the basic header is shown in Figure 8.10. The packet should be transmitted in network byte order. The description of each field in the basic header follows.

Version: The version field specifies the version number of the protocol and should be set to 6 for IPv6. The location and length of the version field stays unchanged so that the protocol software can recognize the version of the packet quickly. Traffic class: The traffic class field specifies the traffic class or priority of the packet. The traffic class field is intended to support differentiated service.

Flow label: The flow label field can be to identify the QoS requested by the packet. In the standard, a flow is defined as ``a sequence of packets sent from a particular source to a particular (unicast or multicast) destination for which the source desires special handling by the intervening routers."

An example of an application that may use a flow label is a packet video system that requires its packets to be delivered to the destination within a certain time constraint. Routers that see these packets will have to process them according to their request. Hosts that do not support flows are required to set this field to 0.

Payload length: The payload length indicates the length of the data (excluding header). With 16 bits allocated to this field, the payload length is limited to 65,535 bytes. As we explain below, it is possible to send larger payloads by using the option in the extension header. Next header: The next header field identifies the type of the extension header that follows the basic header. The extension header is similar to the options field in IPv4 but is more flexible and efficient. Extension headers are further discussed below.

Hop limit: The hop limit field replaces the TTL field in IPv4. The name now says what it means: The value specifies the number of hops the packet can travel before being dropped by a router.

Source address and destination address: The source address and the destination address identify the source host and the destination host, respectively. The address format is discussed below.

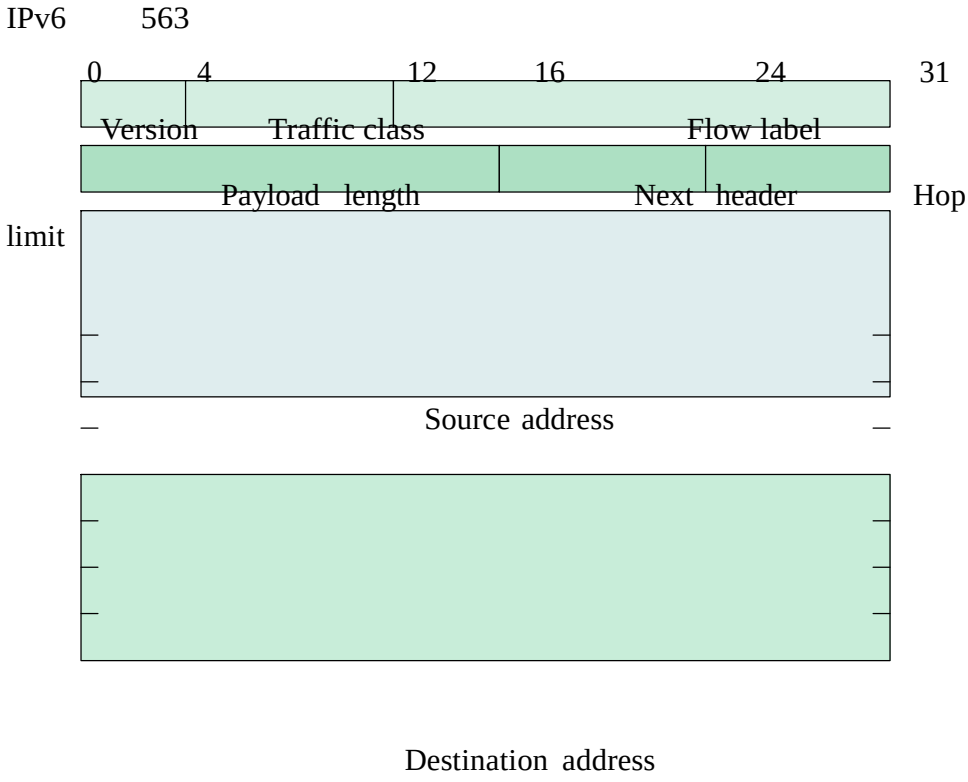


FIGURE 3.10 IPv6 basic headers

3.4 User datagram protocol:

Two transport layer protocols, TCP and UDP, build on the best-effort service provided by IP to support a wide range of applications. In this section we discuss the details of UDP.

The User Datagram Protocol (UDP) is an unreliable, connectionless trans- port layer protocol. It is a very simple protocol that provides only two additional services beyond IP: demultiplexing and error checking on data. Recall that IP knows how to deliver packets to a host, but does not know how to deliver them to the specific application in the host. UDP adds a mechanism that distinguishes among multiple applications in the host. Recall also that IP checks only the integrity of its header. UDP can optionally check the integrity of the entire UDP datagram. Applications that use UDP include Trivial File Transfer Protocol, DNS, SNMP, and Real-Time Protocol (RTP).

The UDP checksum field detects errors in the datagram, and its use is optional. If a source host does not want to compute the checksum, the checksum field should contain all 0s so that the destination host knows that the checksum has not been computed. What if the source host does compute the checksum and finds that the result is 0? The answer is that if a host computes the checksum whose result is 0, it will set the checksum field to all 1s. This is another representation of zero in 1s complement. The checksum computation procedure is similar to that in computing IP checksum except for two new twists. First, if the length of the datagram is not a multiple of 16 bits, the datagram will be padded out with 0s to make it a multiple of 16 bits. In doing so, the actual UDP datagram is not modified. The pad is used only in the checksum computation and is not transmitted. Second, UDP adds a pseudoheader (shown in Figure 8.17) to the beginning of the

datagram when performing the checksum computation. The pseudoheader is also created by the source and destination hosts only during the checksum computation and is not transmitted. The pseudoheader is to ensure

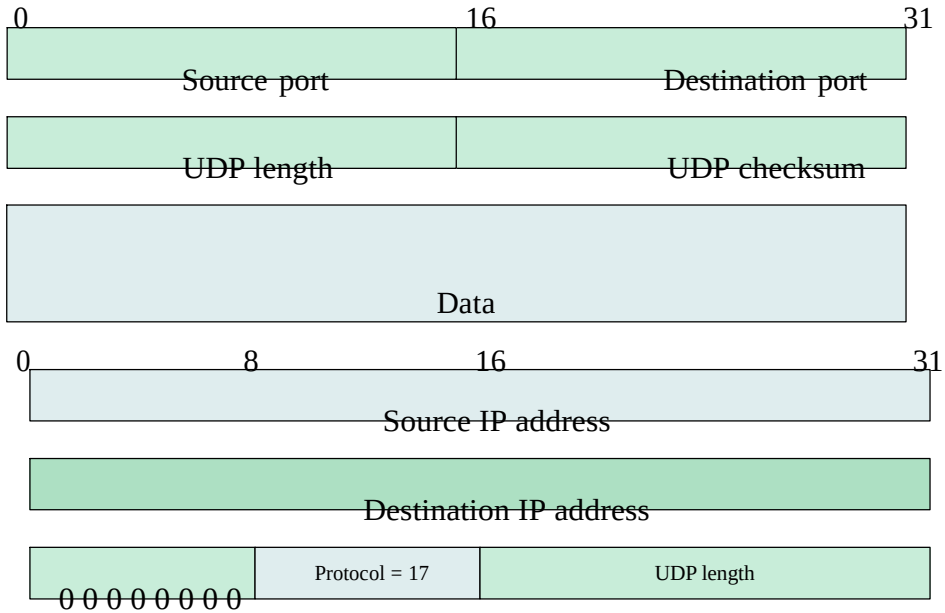


FIGURE 3.17 UDP pseudoheader

that the datagram has indeed reached the correct destination host and port. Finally, if a datagram is found to be corrupted, it is simply discarded and the source UDP entity is not notified.

UNIT – 4

TCP/IP-2: TCP, Internet Routing Protocols, Multicast Routing, DHCP, NAT and Mobile IP.

UNIT – 4

TCP/IP-2:

4.1 Transmission control protocol:

TCP and IP are the workhorses in the Internet. In this section we first discuss how TCP provides reliable, connection-oriented stream service over IP. To do so, TCP implements a version of Selective Repeat ARQ. In addition, TCP implements congestion control through an algorithm that identifies congestion through packet loss and that controls the rate at which information enters the network through a congestion window.

TCP Reliable Stream Service

The Transmission Control Protocol (TCP) provides a logical full-duplex (two- way) connection between two application layer processes across a datagram network. TCP provides these application processes with a connection-oriented,reliable, in-sequence, byte-stream service. TCP also provides flow control that allows receivers to control the rate at which the sender transmits information so that buffers do not overflow. TCP can also support multiple application processes in the same end system.

TCP does not preserve message boundaries and treats the data it gets from the application layer as a byte stream. Thus when a source sends a 1000-byte message in a single chunk (one write), the destination may receive the message in two chunks of 500 bytes each (two reads), in three chunks of 400 bytes, 300 bytes and 300 bytes (three reads), or in any other combination. In other words,

TCP may split or combine the application information in the way it finds most appropriate for the underlying network.

TCP Operation

We now consider the adaptation functions that TCP uses to provide a connection-oriented, reliable, stream service. We are interested in delivering the user information so that it is error free, without duplication, and in the same order that it was produced by the sender. We assume that the user information consists of a stream of bytes as shown in Figure 8.18. For example, in the transfer of a long file the sender is viewed as inserting a byte stream into the transmitter's send buffer. The task of TCP is to ensure the transfer of the byte stream to the receiver and the orderly delivery of the stream to the destination application. TCP was designed to deliver a connection-oriented service in an internet environment, which itself offers connectionless packet transfer service, so different packets can traverse a different path from the same source to the same destination and can therefore arrive out of order. Therefore, in the internet old messages from previous connections may arrive at a receiver, thus potentially complicating the task of eliminating duplicate messages. TCP deals with this problem by using long (32 bit) sequence numbers and by establishing randomly selected initial sequence numbers during connection setup. At any given time the receiver is accepting sequence numbers from a much smaller window, so the likelihood of accepting a very old message is very low. In addition, TCP enforces a time-out period at the end of each connection to allow the network to clear old segments from the network.

TCP uses a sliding-window mechanism, as shown in Figure 4.19. The send window contains three pointers: S_{last} points to the oldest byte that has not yet been acknowledged.

S_{recent} points to the last byte that has been transmitted but not yet acknowledged.

$S_{last} + W_S - 1$ indicates the highest numbered byte that the transmitter is willing to accept from its application.

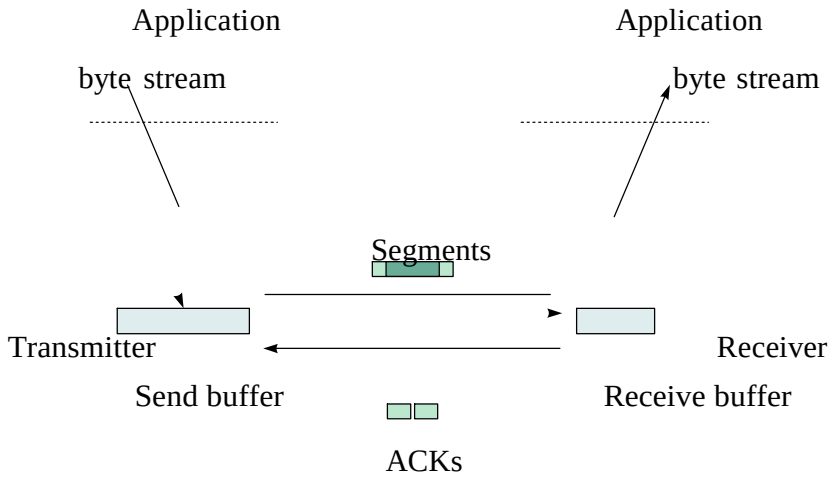


FIGURE 4.18 TCP preview

Enables the network to direct the segment to its destination application process. The segment also contains a sequence number which corresponds to the number of the first byte in the string that is being transmitted. Note that this differs significantly from conventional ARQ. The transmitter decides to transmit a segment when the number of bytes in the send buffer exceeds some specified threshold or when a timer that is set periodically expires. The sending application can also use a push command that forces the transmitter to send a segment.

When a segment arrives, the receiver performs an error check to detect transmission errors. If the segment is error free and is not a duplicate segment, then the bytes are inserted into the

appropriate locations in the receive buffer if the bytes fall within the receive window. Note that the receiver will accept out-of-order but error-free segments. If the received segment contains the byte corresponding to R_{next}, then the R_{next} pointer is moved forward to the location of the next byte that has not yet been received. An acknowledgment with the sequence number R_{next} is sent in a segment that is transmitted in the reverse direction. R_{next} acknowledges the correct receipt of all bytes up to R_{next} 1. This acknowledgment, when received by the transmitter, enables the transmitter to update its parameter S_{last} to R_{next}, thus moving the send window forward. Later in the section we give an example of data transfer in TCP.

TCP separates the flow control function from the acknowledgment function.The flow control function is implemented through an advertised window field in the segment header. Segments that travel in the reverse direction contain the advertised window size that informs the transmitter of the number of buffers currently available at the receiver. The advertised window size is given by

$$W_A \hat{} W_R \quad \ldots R_{new} \quad R_{last} \dagger$$

The transmitter is obliged to keep the number of outstanding bytes below the advertised window size, that is

$$S_{recent} \quad S_{last} \quad W_A$$

To see how the flow control takes effect, suppose that the application at the receiver's side stops reading the bytes from the buffer. Then R_{new} will increase while R_{last} remains fixed, thus leading to a smaller advertised window size.

Eventually the receive window will be exhausted when R_{new} R_{last} $\hat{}$ W_R, and the advertised window size will be zero. This condition will cause the transmitter to stop sending. The transmitter can continue accepting bytes from its application until its buffer

contains W_S bytes. At this point the transmitter blocks its application from inserting any more bytes into the buffer.

Finally, we consider the retransmission procedure that is used in TCP. The transmitter sets a timer each time a segment is transmitted. If the timer expires before any of the bytes in the segment are acknowledged, then the segment is retransmitted.

TCP Protocol

We now discuss the structure of TCP segments and the setting up of a TCP connection, the data transfer phase, and the closing of the connection. Detailed information about TCP can be found in RFC 793 and RFC 1122.

TCP SEGMENT

Figure 8.20 shows the format of the TCP segment. The header consists of a 20 The description of each field in the TCP segment is given below. The term sender refers to the host that sends the segment, and receiver refers to the host that receives the segment.

Source port and destination port: The source and destination ports identify the sending and receiving applications, respectively. Recall from section 2.3.1 that the pair of ports and IP addresses identify a process-to-process connection. byte fixed part plus a variable-size options field.

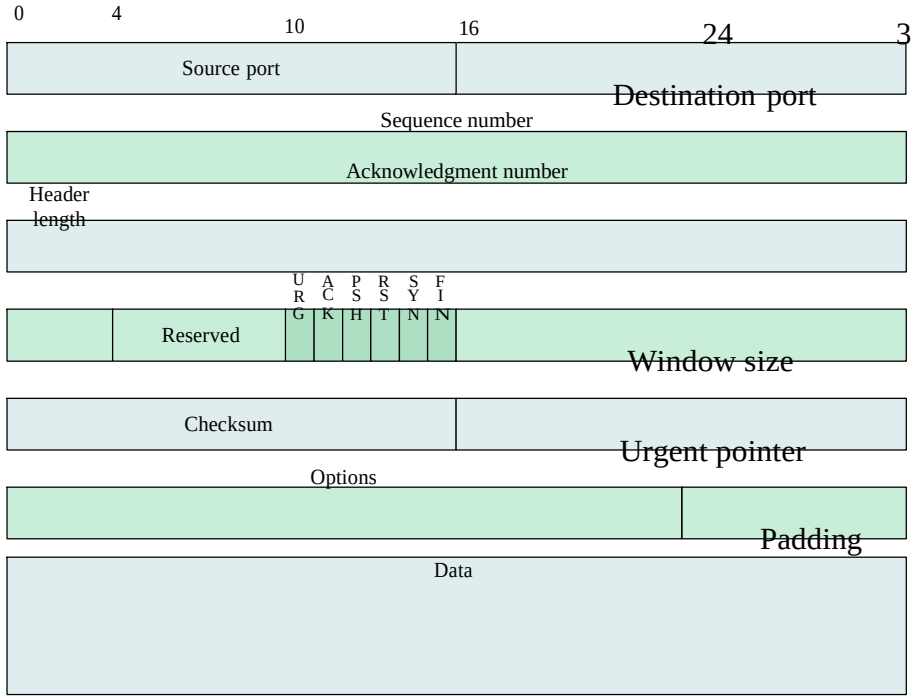


FIGURE 4.20 TCP segment

Sequence number: The 32-bit sequence number field identifies the position of the first data byte of this segment in the sender's byte stream during data transfer (when SYN bit is not set). The sequence number wraps back to 0 after 2^{32} 1. Note that TCP identifies the sequence number for each byte (rather than for each segment). For example, if the value of the sequence number is 100 and the data area contains five bytes, then the next time this

TCP module sends a segment, the sequence number will be 105. If the SYN bit is set to 1 (during connection establishment), the sequence number indicates the initial sequence number (ISN) to be used in the sender's byte stream. The sequence number of the first byte of data for this byte stream will be ISN+1. It is important to note that a TCP connection is full duplex so that each end point independently maintains its own sequence number.

Acknowledgment number: This field identifies the sequence number of the next data byte that the sender expects to receive if the ACK bit is set. This field also indicates that the sender has successfully received all data up to but not including this value. If the ACK bit is not set (during connection establishment), this field is meaningless. Once a connection is established, the ACK bit must be set.

Header length: This field specifies the length of the TCP header in 32-bit words. This information allows the receiver to know the beginning of the data area because the options field is variable length.

Reserved: As the name implies, this field is reserved for future use and must be set to 0.

URG: If this bit is set, the urgent pointer is valid (discussed shortly).

ACK: If this bit is set, the acknowledgment number is valid.

PSH: When this bit is set, it tells the receiving TCP module to

pass the data to the application immediately. Otherwise, the receiving TCP module may choose to buffer the segment until enough data accumulates in its buffer.

RST: When this bit is set, it tells the receiving TCP module to abort the connection because of some abnormal condition.

SYN: This bit requests a connection (discussed later).

FIN: When this bit is set, it tells the receiver that the sender does not have any more data to send. The sender can still receive data from the other direction until it receives a segment with the FIN bit set.

Window size: The window size field specifies the number of bytes the sender is willing to accept. This field can be used to control the flow of data and congestion.

Checksum: This field detects errors on the TCP segment. The procedure is discussed below.

Urgent pointer: When the URG bit is set, the value in the urgent pointer field added to that in the sequence number field points to the last byte of the ``urgent data" (data that needs immediate delivery). However, the first byte of the urgent data is never explicitly defined. Because the receiver's TCP module passes data to the application in sequence, any data in the receiver's buffer up to the last byte of the urgent data may be considered urgent.

Options: The options field may be used to provide other functions that are not covered by the header. If the length of the options field is not a multiple of 32 bits, extra padding bits will be added. The most important option is used by the sender to indicate the maximum segment size (MSS) it can accept. This option is specified during connection setup. Two other options that are negotiated during connection setup are intended to deal with situations that involve large delay-bandwidth products. The

window scale option allows the use of a larger advertised window size. The window can be scaled upward by a factor of up to 2^{14}. Normally the maximum window size is 2^{16} 1 ˆ 65,535. With scaling the maximum advertised window size is 65,535 2^{14} fl 1,073,725,440 bytes. The timestamp option is intended for high-speed connections where the sequence numbers may wrap around during the lifetime of the connection. The timestamp option allows the sender to include a timestamp in every segment. This timestamp can also be used in the RTT calculation.

TCP CHECKSUM

The purpose of the TCP checksum field is to detect errors. The checksum com putation procedure is similar to that used to compute an IP checksum (discussed in Chapter 3) except for two features. First, if the length of the segment is not a multiple of 16 bits, the segment will be padded with zeros to make it a multiple of 16 bits. In doing so, the TCP length field is not modified. Second, a pseudoheader (shown in Figure 8.21) is added to the beginning of the segment when performing the checksum computation.

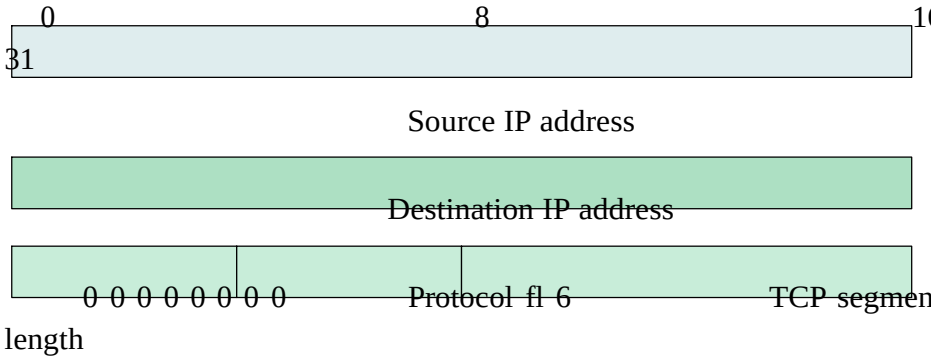


FIGURE 4.21 TCP pseudoheader

CONNECTION ESTABLISHMENT

Before any host can send data, a connection must be established. TCP establishes
the connection using a three-way handshake procedure shown in Figure 8.22. The
handshakes are described in the following steps:

1. Host A sends a connection request to host B by setting the SYN bit. Host A
also registers its initial sequence number to use (Seq_no fl x).
2. Host B acknowledges the request by setting the ACK bit and indicating the next data byte to receive (Ack_no fl x + 1). The ``plus one'' is needed because the SYN bit consumes one sequence number. At the same time, host B also sends a request by setting the SYN bit and registering its initial sequence number to use (Seq_no fl y).
3. Host A acknowledges the request from B by setting the ACK bit and con firming the next data byte to receive (Ack_no fl y + 1). Note that the sequence number is set to x + 1. On receipt at B the connection is established.

If during a connection establishment phase, one of the hosts decides to refuse a connection request, it will send a reset segment by setting the RST bit. Each SYN message can specify options such as maximum segment size, window scal- ing, and timestamps.

Because TCP segments can be delayed, lost, and duplicated, the initial sequence number should be different each time a host requests a connection.

TCP CONNECTION TERMINATION

TCP provides for a graceful close that involves the independent termination of each direction of the connection. A termination is initiated when an application tells TCP that it has no more data to send. The TCP entity completes transmis- sion of its data and, upon receiving acknowledgment from the receiver, issues a segment with the FIN bit set. Upon receiving a FIN segment, a TCP entity informs its application that the other entity has terminated its transmission of data. For example, in Figure 8.27 the TCP entity in host A terminates its trans- mission first by issuing a FIN segment. Host B sends an ACK segment to acknowledge receipt of the FIN segment from A. Note that the FIN segment uses one byte, so the ACK is 5087 in the example.

After B receives the FIN segment, the direction of the flow from B to A is still open. In Figure 8.27 host B sends 150 bytes in one segment, followed by a FIN segment. Host A then sends an acknowledgment. The TCP in host A then enters the TIME_WAIT state and starts the TIME_WAIT timer with an initial value set to twice the maximum segment lifetime (2MSL).

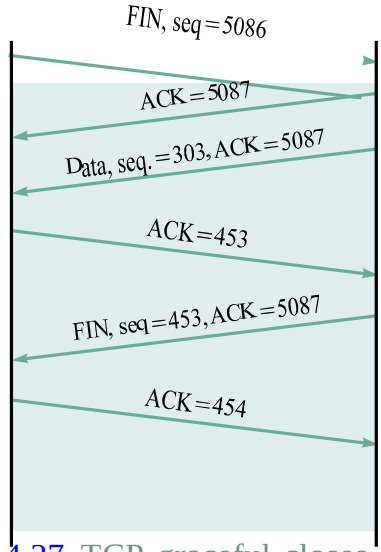


FIGURE 4.27 TCP graceful closes

TIME_WAIT timer is restarted at 2MSL. When the TIME_WAIT timer expires, host A closes the connection and then deletes the record of the connection.

The TIME_WAIT state serves a second purpose. The MSL is the maximum time that a segment can live inside the network before it is discarded. The TIME_WAIT state protects future incarnations of the connection from delayed segments. The TIME_WAIT forces TCP to wait at least two MSLs before setting up an incarnation of the old connection. The first MSL accounts for the maximum time a segment in one direction can remain in the network, and the second MSL allows for the maximum time a reply in the other direction can be in the network. Thus all segments from the old connection will be cleared from the network at the end of the TIME_WAIT state.

TCP provides for an abrupt connection termination through reset (RST) segments. An RST segment is a segment with the

RST bit set. If an application decides to terminate the connection abruptly, it issues an ABORT command, which causes TCP to discard any data that is queued for transmission and to send an RST segment. The TCP that receives the RST segment then notifies its application process that the connection has been terminated.

Tcp state transition diagram:

A TCP connection goes through a series of states during its lifetime. Figure 8.28 shows the state transition diagram. Each state transition is indicated by an arrow, and the associated label indicates associated events and actions. Connection establishment begins in the CLOSED state and proceeds to the ESTABLISHED state. Connection termination goes from the ESTABLISHED state to the CLOSED state. The normal transitions for a client are indicated by thick solid lines, and the normal transitions for a server are denoted by dashed lines. Thus when a client does an active open, it goes from the CLOSED state, to SYN_SENT, and then to ESTABLISHED. The server carry- ing out a passive open goes from the CLOSED state, to LISTEN, SYN_RCVD, and then to ESTABLISHED.

The client normally initiates the termination of the connection by sending a FIN. The associated state trajectory goes from the ESTABLISHED state, to FIN_WAIT_1 while it waits for an ACK, to FIN_WAIT_2 while it waits for the other side's FIN, and then to TIME_WAIT after it sends the final ACK. When the TIME_WAIT 2MSL period expires, the connection is closed and the transmission control block that stores all the TCP connection variables is deleted. Note that the state transition diagram does not show all error conditions that

may arise, especially in relation to the TIME_WAIT state. The server normally goes from the ESTABLISHED state to the CLOSE_WAIT state after it receives a FIN, to the LAST_ACK when it sends its FIN, and finally to CLOSE when it receives the final ACK.

Mobile IP:

Mobile networking is a subject that is becoming increasingly important as por table devices such as personal digital assistants (PDAs) and notebook computers are becoming more powerful and less expensive, coupled with people's need to be connected whenever and wherever they are. The link between the portable device and the fixed communication network can be wireless or wired. Infrared channels are often used in shorter distances. A wireless connection enables a user to maintain its communication session as it roams from one area to another, providing a very powerful communication paradigm. In this section we look at a simple IP solution for mobile computers.

Mobile IP allows portable devices called mobile hosts (MHs) to roam from one area to another while maintaining the communication sessions. One require- ment in mobile IP is that a legacy host communicating with an MH and the intermediate routers should not be modified. This requirement implies that an MH must continuously use its permanent IP address even as it roams to another area. Otherwise, existing sessions will stop working and new sessions should be restarted when an MH moves to another area. The basic mobile IP solution is sketched in Figure 2.29.

The mobile IP routing operates as follows:

When a correspondent host (CH) wants to send a packet to an MH, the CH transmits the standard IP packet with its address as the source IP address and the MH's address as the destination IP address. This packet will be intercepted by the mobile host's router called the home agent (HA), which keeps track of the current location of the MH. The HA manages all MHs in its home network that use the same address prefix. If the MH is located in the home network, the HA simply forwards the packet to its home network.

When an MH moves to a foreign network, the MH obtains a care-of address from the foreign agent (FA) and registers the new address with its HA. The care-of address reflects the MH's current location and is typically the address of the FA. Once the HA knows the care-of address of the MH, the HA can forward the registration packet to the MH via the FA.

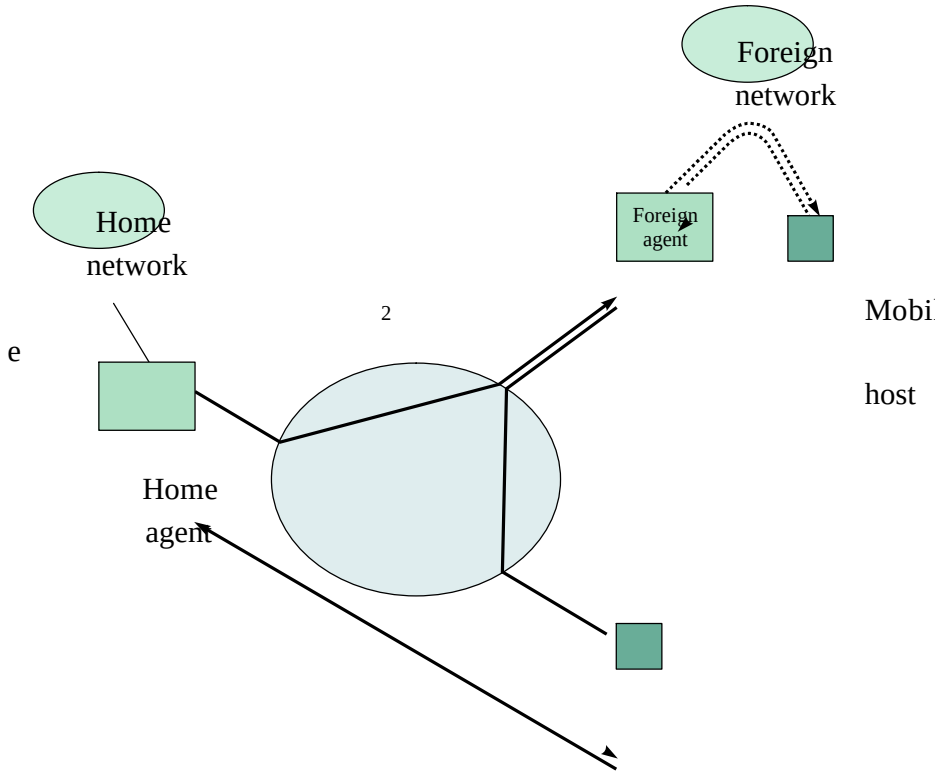


FIGURE 2.29 Routing for mobile hosts

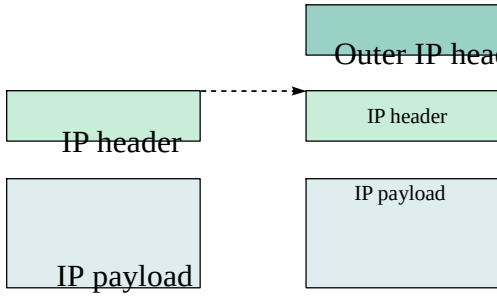


FIGURE 2.30 IP to IP

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Unfortunately, the HA cannot directly send packets to the MH in a foreign network in a conventional way (i.e., by using the care-of address as the destina- tion address of the IP packet, the packet final destination will the FA rather than the MH). The solution is provided by a tunneling mechanism that essentially provides two destination addressesÐthe destination of the other end of the tunnel (i.e., the FA) and the final destination (i.e., the MH). The IP packet tunneled by the HA is encapsulated with an outer IP header (see Figure 8.30) containing the HA's address as the source IP address and the care-of address as the destination IP address. When the FA receives the packet, the FA decapsu-lates the packet that produces the original IP packet with the correspondent host's address as the source IP address and the MH's address as the destination IP address. The FA can then deliver the packet to the MH.

4.2 Internet routing protocols:

The global Internet topology can be viewed as a collection of autonomous sys tems. An autonomous system (AS) is loosely defined as a set of routers or net- works that are technically administered by a single organization such as a corporate network, a campus network, or an ISP network. There are no restrict tions that an AS should run a single routing protocol within the AS. The only important requirement is that to the outside world, an AS should present a consistent picture of which ASs are reachable through it.

There are three categories of ASs:

1. Stub AS has only a single connection to the outside world.

Stub AS is also called single-homed AS.

2. Multihomed AS has multiple connections to the outside world but refuses to carry transit traffic (traffic that originates and terminates in the outside world). Multihomed AS carries only local traffic (traffic that originates or terminates in that AS).

3. Transit AS has multiple connections to the outside world and can carry transit and local traffic.

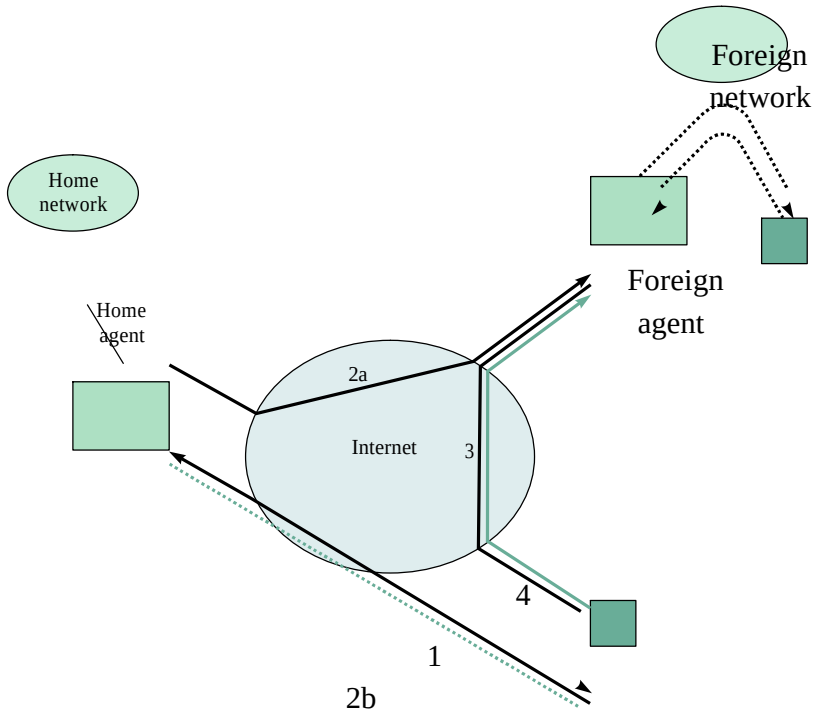


FIGURE 4.31 Route optimization for mobile IP

For the purpose of AS identification, an AS needs to be assigned with a globally unique AS number (ASN) that is represented by a 16-bit integer and thus is limited to about

65,000 numbers. We show later how ASNs are used in exterior routing. Care must be taken not to exhaust the AS space. Currently, there are about 3500 registered ASs in the Internet. Fortunately, a stub AS, which is the most common type, does not need an ASN, since the stub AS prefixes are placed at the provider's routing table. On the other hand, a transit AS needs an ASN. At present, an organization may request an ASN from the American Registry for Internet Numbers (ARIN) in North America. Routing protocols in the Internet are arranged in a hierarchy that involves two types of protocols: Interior Gateway Protocol (IGP) and Exterior Gateway Protocol (EGP). IGP is used for routers to communicate within an AS and relies on IP addresses to construct paths. EGP is used for routers to communicate among different ASs and relies on AS numbers to construct AS paths. In this section we cover two popular IGPs: Routing Information Protocol and Open Shortest Path First.

4.3 Multicast routing:

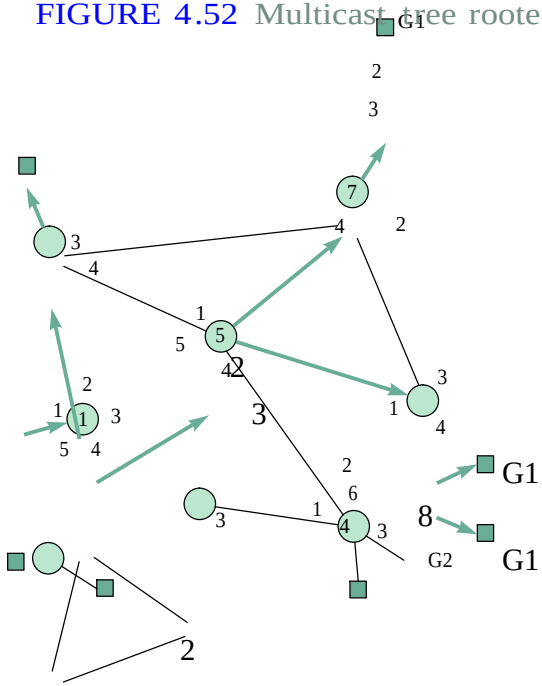
The routing mechanisms discussed so far assume that a given source transmits its packets to a single destination. For some applications such as teleconferencing, a source may want to send packets to multiple destinations simultaneously. This requirement calls for another type of routing called multicast routing as illustrated in Figure 8.52. In the figure, source S wants to transmit to destinations with multicast group G1. Although the source can send each copy of the packet separately to each destination by using conventional unicast routing, a more efficient method would be to minimize the number of copies. For example, when router 1 receives a packet from the source, router 1 copies the packet to routers 2 and 5 simultaneously. Upon receipt of these packets, router 2 forwards

the packet to its local network, and router 5 copies the packet to routers 7 and 8. The packet will eventually be received by each intended destination.

With multicast routing, each packet is transmitted once per link. If unicast routing is used instead, the link between the source and router 1 will carry four copies for each packet transmitted. In general, the bandwidth saving with multi- cast routing becomes more substantial as the number of destinations increases.

There are many ways to generate a multicast tree. One approach that is used in multicast backbone (MBONE) is called reverse-path multicasting. MBONE is basically an overlay packet network on the Internet supporting routing of IP multicast packets (with Class D address

FIGURE 4.52 Multicast tree rooted at source



Reverse-Path Broadcasting:

The easiest way to understand reverse-path multicasting is by first considering a simpler approach called reverse-path broadcasting (RPB). RBP uses the fact that the set of shortest paths to a node forms a tree that spans the network. See Figure 1.29. The operation of RBP is very simple:

Upon receipt of a multicast packet, a router records the source address of the packet and the port the packet arrives on.

If the shortest path from the router back to the source is through the port the packet arrived on, the router forwards the packet to all ports except the one the packet arrived on; otherwise, the router drops the packet. The port over which the router expects to receive multicast packets from a given source is referred to as the parent port.

The advantages of RPB are that routing loops are automatically suppressed and each packet is forwarded by a router exactly once. The basic assumption underlying the RPB algorithm is that the shortest path from the source to a given router should be the same as the shortest path from the router to the source. This assumption requires each link to be symmetric (each direction has the same cost). If links are not symmetric, then the router must compute the shortest path from the source to the router. This step is possible only if link-state protocols are used.

To see how RPB works for the network shown in Figure 8.53,

assume that source S transmits a multicast packet with group G1. The parent ports for this multicast group are identifed in bold numbers, as shown in Figure 8.53.

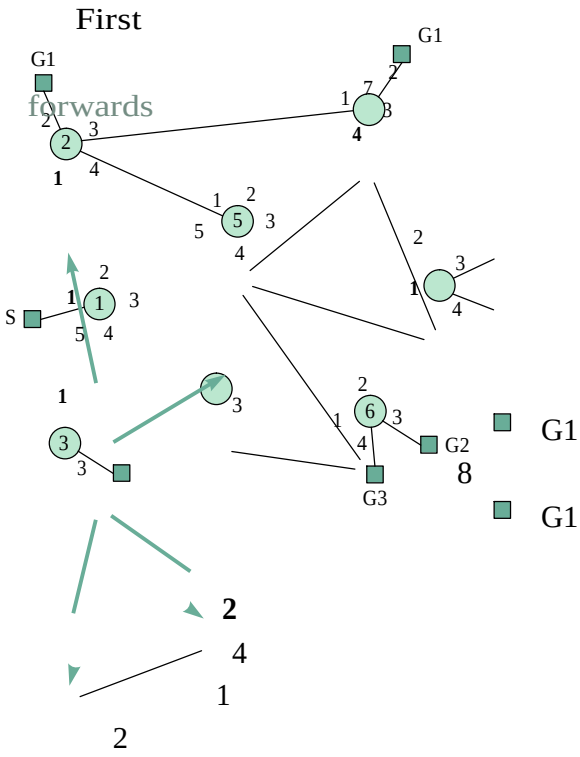


FIGURE 4.53 Router 1 forwards the packet

router 1 receives a packet on port 1 from source S. Because the packet comes from the parent port, the router forwards the packet to all other ports.

Assume that these packets reach routers 2, 3, 4, and 5 some time later. Router 2 computes the shortest path from itself to S and finds that the parent port is port 1. It then forwards the packet through all other ports. Similarly, routers 3, 4, and 5

find that the packet arrives on the parent ports. As a result, each router forwards the packet to its other ports, as shown in Figure 8.54. Note that the bidirectional link indicates that each router forwards the packet to the other.

Next assume that the packets arrive at routers 2 through 8. Because router 2 receives the packet from port 4, which is not the parent port, router 2 drops the packet. Routers 3, 4, and 5 also drop the packet for the same reason. Router 6,however, finds that the parent port is port 1, so router 6 forwards the packet coming from router 4 to its other ports. The packet that came from router 5 to router 6 is dropped. Routers 7 and 8 forward the packet that came from router 5. Figure 8.55 illustrates the process pictorially. The reader should verify that the packet will no longer be propagated after this point.

Note that although the hosts connected to routers 3 and 6 belong to different multicast groups, the packets for group G1 are still forwarded by these routers. Typically, these hosts are attached to the router via a shared medium LAN.

Therefore, unnecessary bandwidth is wasted regardless of the multicast group.

This problem can be solved by having the router truncate its transmission to the local network if none of the hosts attached to the network belong to the multicast group. This refinement is called truncated reverse-path broadcasting (TRPB).

Note that TRPB performs truncation only at the ``leaf'' routers (routers at the edge of the network). The next section describes a protocol that allows a router to determine which hosts belong to which multicast groups.

Internet Group Management Protocol:

The Internet Group Management Protocol (IGMP) allows a host to signal its multicast group membership to its attached router. IGMP runs directly on IP using protocol type 2. However, IGMP is usually considered as part of IP. The IGMP message format is very simple, as can be seen from Figure 8.56.

A description of each field follows.

Version: This field identifies the version number.

Type: This field identifies the message type. There are two message types:

Type 1 indicates a query message sent by a router, and type 2 indicates a report sent by a host.

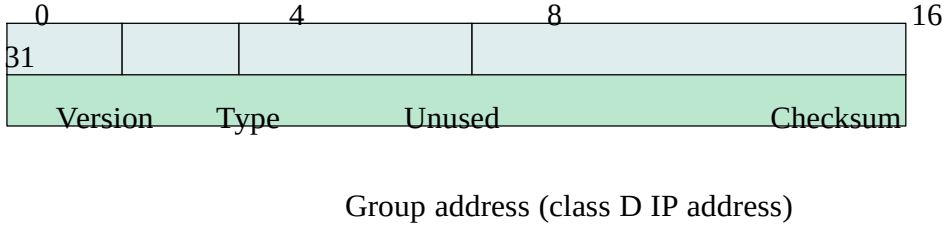


FIGURE 4.56 IGMP message format

Unused: This field must be set to zero.

Checksum: This field contains a checksum for all eight bytes of the IGMP message.

Group Address: This address is the class D IPv4 address. This field is set to zero in a query message, and is set to a valid group address in the response.

When a host wants to join a new multicast group, the host sends an IGMP report specifying the group address to join. The host needs to issue only one IGMP report, even though multiple

applications are joining the same group. The host does not issue a report if it wants to leave the group.

To make sure that multiple hosts do not send a report at the same time when receiving a query that would lead to a collision, the scheduled transmission time should be randomized. Because the router only has to know that at least one host belongs to a particular group, efficiency can be improved by having a host monitor if another host sends the same report earlier than its scheduled transmission. If another host has already sent the same report, the first host cancels its report.

Reverse-Path Multicasting:

As section 8.8.2 just explained IGMP allows hosts to indicate their group members. We now continue with selective multicast routing called reverse-path multicasting (RPM), which is an enhancement of TRPB. Unlike TRPB, RPM forwards a multicast packet only to a router that will lead to a leaf router with group members. Each router forwards the first packet for a given (source, group) according to TRPB. All leaf routers receive at least the first multicast packet. If a leaf router does not find a member for this group on any of its ports, the leaf router will send a prune message to its upstream router instructing the upstream router not to forward subsequent packets belonging to this (source, group). If the upstream router receives the prune messages from all of its downstream neighbors, it will in turn generate a prune message to its further upstream router.

A host may later decide to join a multicast group after a prune message has been sent by its leaf router. In this case the leaf router would send a graft message to its upstream router to cancel its earlier prune message. Upon receiving the graft message, the first

upstream router will forward subsequent multicast pack- ets to this leaf router. If the first upstream router has also sent a prune message earlier, the first upstream router will send a graft message to the second upstream router. Eventually, a router in the multicast tree will reactivate its affected port so that the multicast packets will travel downstream toward the host. Figure 8.57 shows the grafting message flow when a host attached to router 6 wants to join the group. Subsequently, router 1 will forward the multicast packets to router 4, which will forward the multicast packets to router 6. Eventually, the multicast packets arrive at the host.

4.4 DHCP:

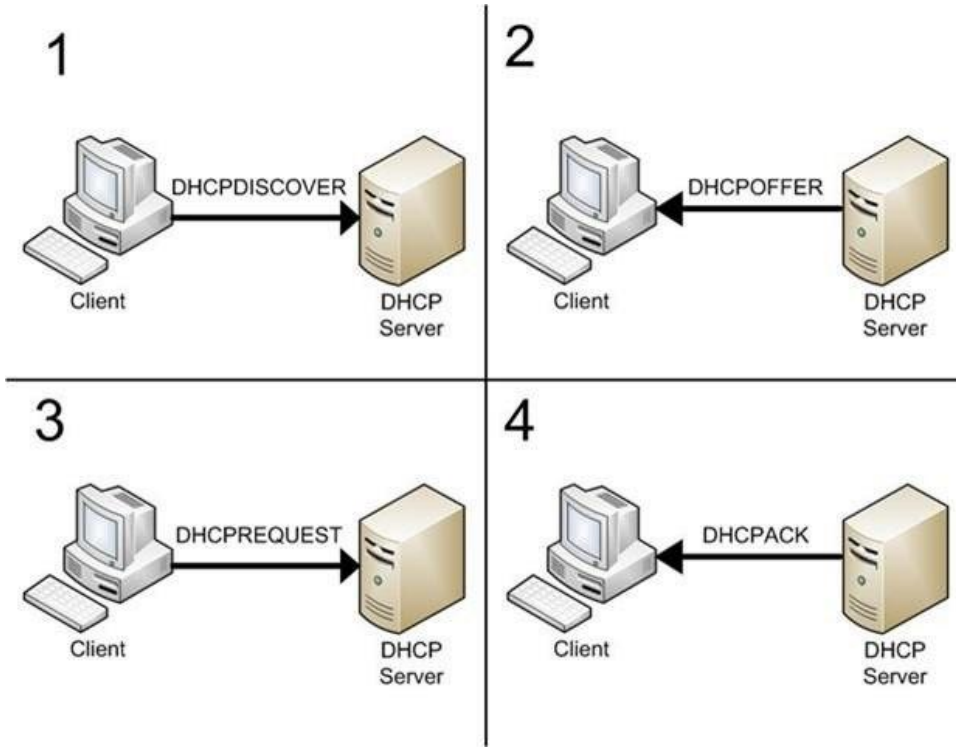
The Dynamic Host Configuration Protocol (DHCP) is a network protocol that is used to configure network devices so that they can communicate on an IP network. A DHCP client uses the DHCP protocol to acquire configuration information, such as an IP address, a default route and one or more DNS server addresses from a DHCP server. The DHCP client then uses this information to configure its host. Once the configuration process is complete, the host is able to communicate on that internet. The DHCP server maintains a database of available IP addresses and configuration information. When it receives a request from a client, the DHCP server determines the network to which the DHCP client is connected, and then allocates an IP address or prefix that is appropriate for the client, and sends configuration information appropriate for that client.

Because the DHCP protocol must work correctly even before DHCP

clients have been configured, the DHCP server and DHCP client must be connected to the same network link. In larger networks, this is not practical. On such networks, each network link contains one or more DHCP relay agents. These DHCP relay agents receive messages from DHCP clients and forward them to DHCP servers. DHCP servers send responses back to the relay agent, and the relay agent then sends these responses to the DHCP client on the local network link.

DHCP servers typically grant IP addresses to clients only for a limited interval. DHCP clients are responsible for renewing their IP address before that interval has expired, and must stop using the address once the interval has expired, if they have not been able to renew it. DHCP is used for IPv4 and IPv6. While both versions serve much the same purpose, the details of the protocol for IPv4 and IPv6 are sufficiently different that they may be considered separate protocols.

Hosts that do not use DHCP for address configuration may still use it to obtain other configuration information. Alternatively, IPv6 hosts may use stateless address auto configuration. IPv4 hosts may use link-local addressing to achieve limited local connectivity.



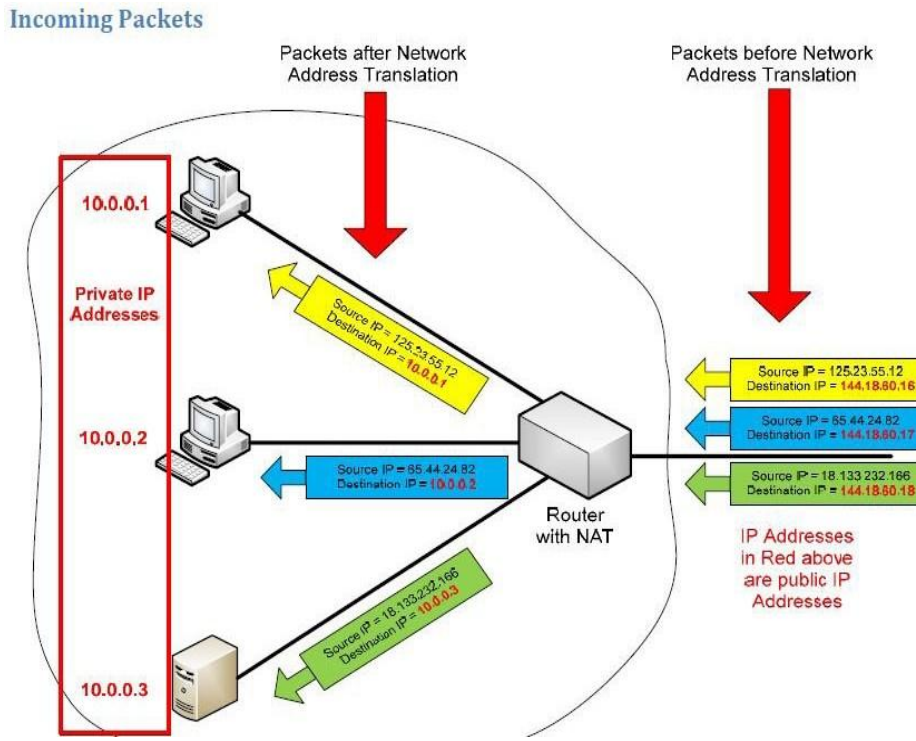
4.5 NAT and Mobile IP:

The concept of NAT is a very powerful concept for several reasons:

It shields computers in a private LAN from the Internet and therefore reduces the risks that are associated with connecting a computer to the Internet (hacking attacks).More importantly, Internet service providers usually assignone IP address to a home network or multiple IP addresses to an organization. However, the number of computers on the home network. What NAT does is that local

addresses (in one of the 3 ranges of private IP addresses that start with 10,172, or 192) are translated to one public IP address assigned to the home network (in the case of DSL service) or multiple public IP addresses assigned to the organization by the Internet service provider (in the case of organizations such as KFUPM). The NAT system also translates from the public IP address(es) to the corresponding private IP addresses as the packets arrive from the Internet to the private network. In fact, all computers in a network that uses NAT appear to the outside world as having only few IP addresses.

For the case of a home network, all computers in your home network will appear to the outside world as having a single IP address. If you visit a website that records your IP address from one of your home network computers and then try to visit the same website from another computer, the website will not be able to distinguish between the two computers. The following are two examples that show how NAT works. In the first case, the network is assigned multiple public IP addresses equal to the number of machines in the network. All that the NAT does is translate each private IP address into one of the public IP addresses and vice versa. The two situations for outgoing packets (packets going from the private network to the Internet) and incoming packets (packets going from the Internet to the private network) are shown below. In the second case, the network is assigned a single public IP address that will be used by all computers in the private network. The two situations for outgoing packets and incoming packets are shown afterwards.



Mobile IP:

Mobile IP (or IP mobility) is an Internet Engineering Task Force (IETF) standard communications protocol that is designed to allow mobile device users to move from one network to another while maintaining a permanent IP address. Mobile IP for IPv4 is described in IETF RFC 5944, and extensions are defined in IETF RFC 4721. Mobile IPv6, the IP mobility implementation for the next generation of the Internet Protocol, IPv6, is described in RFC 6275.

The Mobile IP protocol allows location-independent routing of IP datagrams on the Internet. Each mobile node is identified by its home address disregarding its current location in the Internet. While away from its home network, a mobile node

is associated with a care-of address which identifies its current location and its home address is associated with the local endpoint of a tunnel to its home agent. Mobile IP specifies how a mobile node registers with its home agent and how the home agent routes datagrams to the mobile node through the tunnel.

Applications:

In many applications (e.g., VPN, VoIP), sudden changes in network connectivity and IP address can cause problems. Mobile IP was designed to support seamless and continuous Internet connectivity.

Mobile IP is most often found in wired and wireless environments where users need to carry their mobile devices across multiple LAN subnets. Examples of use are in roaming between overlapping wireless systems, e.g., IP over DVB, WLAN, WiMAX and BWA.

Changes in IPv6 for Mobile IPv6

- A set of mobility options to include in mobility messages
- A new Home Address option for the Destination Options header
- A new Type 2 Routing header
- New Internet Control Message Protocol for IPv6 (ICMPv6) messages to discover the set of home agents and to obtain the prefix of the home link
- Changes to router discovery messages and options and additional Neighbor Discovery options.

PART – B

UNIT - 5

Applications, Network Management, Network Security:

Application layer overview, Domain Name System (DNS), Remote Login Protocols, E-mail, File Transfer and FTP, World Wide Web and HTTP, Network management, Overview of network security, Overview of security methods, Secret-key encryption protocols, Public-key encryption protocols, Authentication, Authentication and digital signature, Firewalls.

UNIT-5

Applications, Network Management, Network Security

5.1. Application-Layer Overview

The application layer is built on the transport layer and provides network services to user applications. The application layer defines and performs such applications as electronic mail (e-mail), remote access to computers, file transfers, newsgroups, and the Web, as well as streaming video, Internet radio and telephony, P2P file sharing, multiuser networked games, streaming stored video clips, and real-time video conferencing.

The application layer has its own software dependencies. When a new application is developed, its software must be able to run on multiple machines, so that it does not need to be rewritten for networking devices, such as routers, that function at the network layer. In a client/server architecture for example, a client end host requests services from a server host. A client host can be on sometimes or always. Figure 5.1 shows an example of application-layer communication.

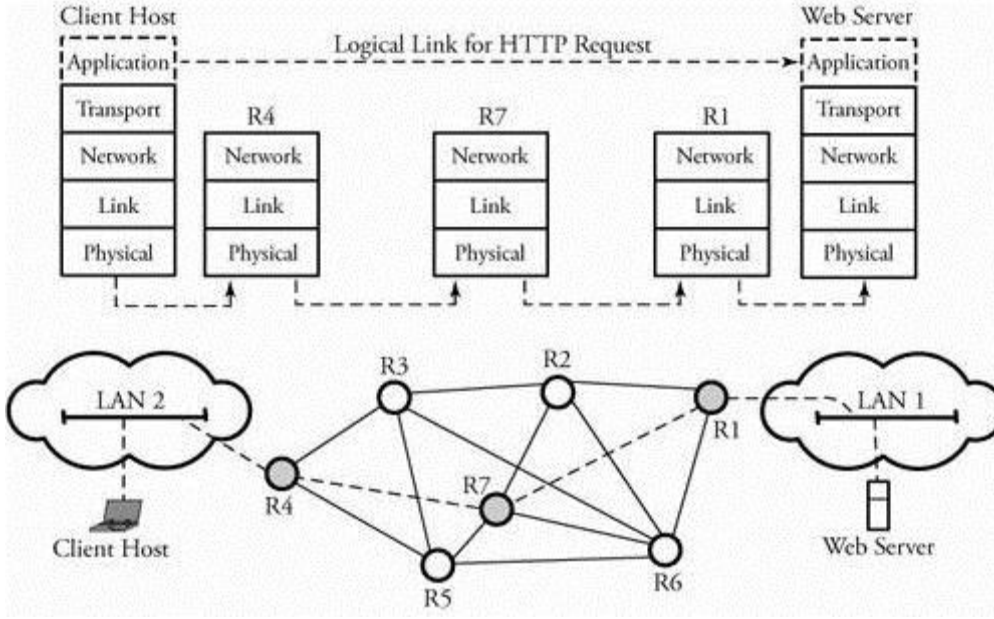


Figure 5.1. Web communication between two end systems

5.1.1. Client and Server Model

A client/server model provides specific computational services, such as partial-time usage services to multiple machines. Reliable communication protocols, such as TCP, allow interactive use of remote servers as well. For example, we can build a server that provides remote image-processing services to clients. Implementing such a communication service requires a server loaded with the application protocol to accept requests and a client to make such requests. To invoke remote image processing, a user first executes a client program establishing a TCP connection to a server. Then, the client begins transmitting pieces of a raw image to the server. The server processes the received objects and sends the results back.

5.2. Domain Name System (DNS)

One of the most important components of the application layer is the Domain Name System (DNS) server. DNS is a distributed hierarchical and global directory that translates machine or domain names to numerical IP addresses.DNS can run over either UDP or TCP. Some of the information-processing functions a DNS server handles are

- Finding the address of a particular host
- Delegating a subtree of server names to another server
- Denoting the start of the subtree that contains cache and configuration parameters, and giving corresponding addresses
- Naming a host that processes incoming mail for the designated target
- Finding the host type and the operating system information
- Finding an alias for the real name of a host
- Mapping IP addresses to host names

5.2.1. Domain Name Space

Any entity in the TCP/IP environment is identified by an IP address, which thereby identifies the connection of the corresponding host to the Internet. An IP address can also be assigned a domain name. Unique domain names assigned to hosts must be selected from a name space and are generally organized in a hierarchical fashion.

Domain names are defined in a tree-based structure with the root at the top, as shown in Figure 5.2. A tree is structured with a maximum of 128 levels, starting at level 0 (root). Each level consists of nodes. A node on a tree is identified by a label, with a string of up to 63 characters, except for the root label, which has empty string.

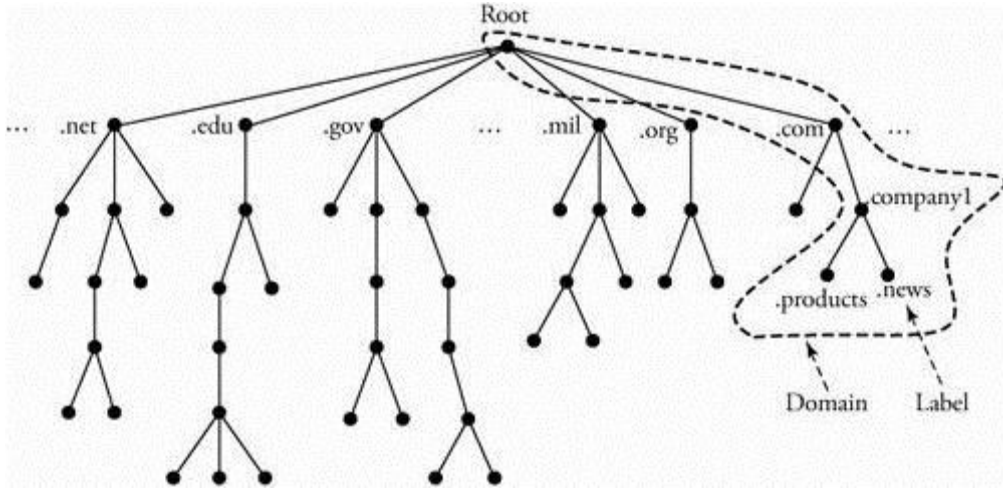


Figure 5.2. Hierarchy of domain name space, labels, and domain names

The last label of a domain name expresses the type of organization; other parts of the domain name indicate the hierarchy of the departments within the organization. Thus, an organization can add any suffix or prefix to its name to define its host or resources. A domain name is a sequence of labels separated by dots and is read from the node up to the root. For example, moving from right to left, we can parse as follows: domain name news.company1.com, a commercial organization (.com) and the "news" section of "company1" (news.company1). Domain names can also be partial. For example, company1.com is a partial domain name.

Domain-Name Servers

The domain name space is divided into sub domains, and each domain or sub domain is assigned a domain name server. This way, we can form a hierarchy of servers, as shown in Figure 5.3, just as we did for the hierarchy of domain names. A domain name server has a database consisting of all the information for every node under that domain..

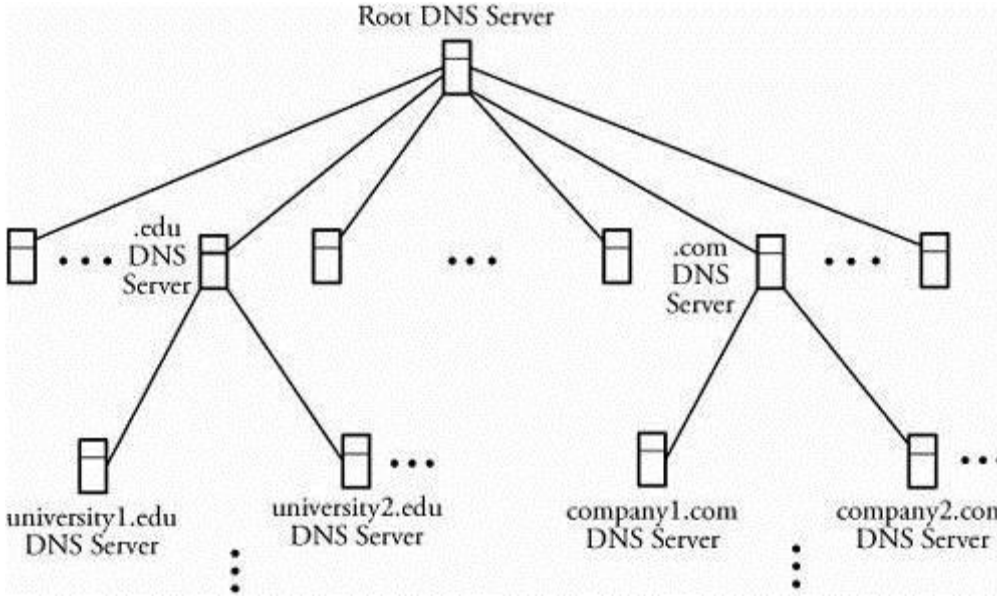


Figure 5.3. Hierarchy of DNS domain name servers

5.2.2. Name/Address Mapping

DNS operates based on the client/server application. Any client host can send an IP address to a domain name server to be mapped to a domain name. Each host that needs to map an address to a name or vice versa should access the closest DNS server with its request.

Mapping can be of either recursive or iterative. In recursive mapping (Figure 5.4), the client host makes the request to its corresponding DNS server. The DNS server is responsible for finding the answer recursively. The requesting client host asks for the answer through its local DNS server, news.company1.com

Finally, .com server sends the query to the local DNS server of the requested place, as

dns.company2.com, and finds the answer. The answer to a query in this method is routed back to the origin, as shown in the figure. The local DNS server of the requested place is called the authoritative server and adds information to the mapping, called time to live (TTL).

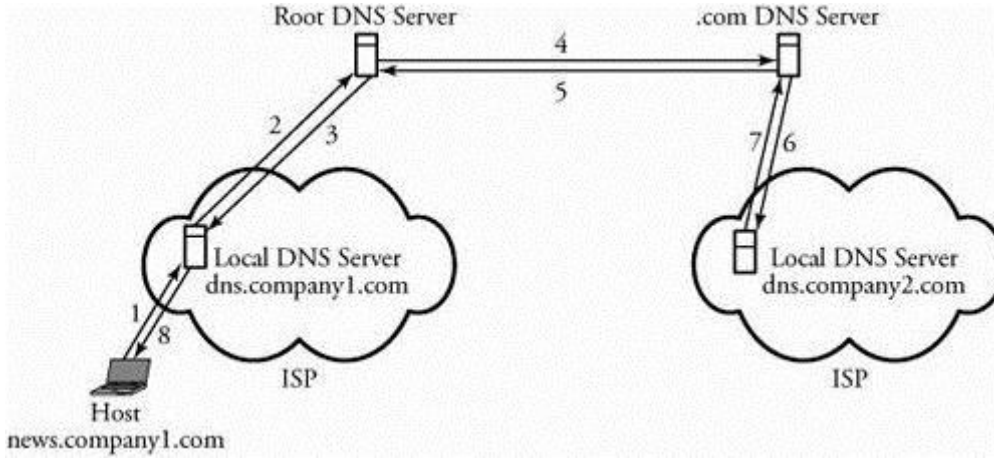


Figure 5.4. Recursive mapping

In the iterative approach, the mapping function is as shown in Figure 5.5. In this case, if it does not have the name to provide, the server returns to the client host.

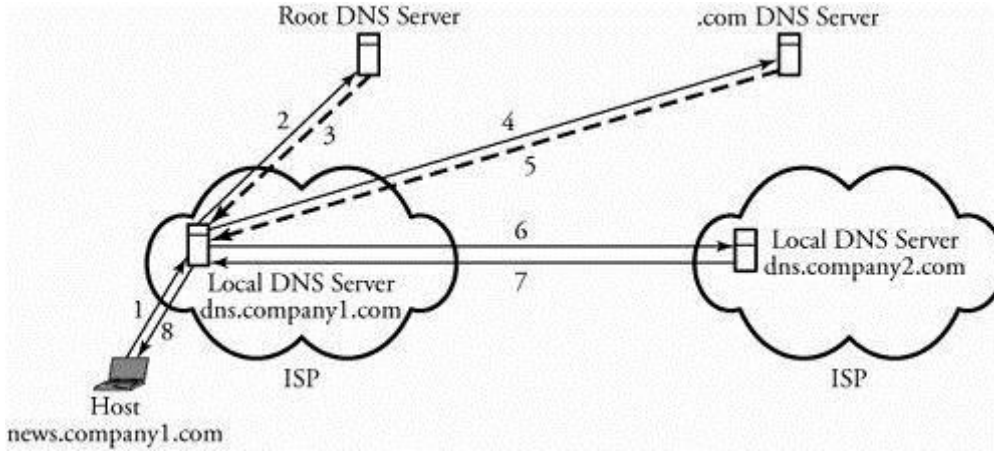


Figure 5.5. Iterative mapping

In Figure 5.5, the news.company1.com host sends the query to its own local DNS server, dns.company1.comthus trying the root DNS server firstand then tries .com server, finally ending up with the local DNS server of the requested place: dns.company2.com.

5.2.3. DNS Message Format

DNS communication is made possible through query and reply messages. Both message types have the 12-byte header format shown in Figure 5.6.

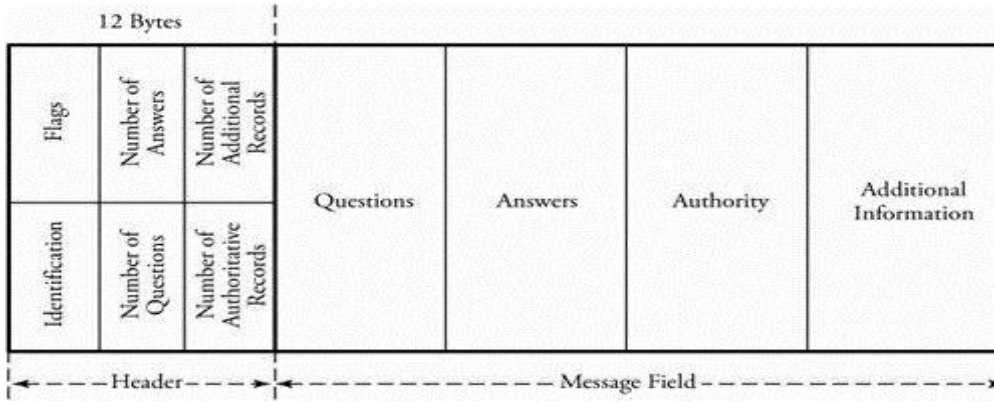


Figure 5.6. DNS message format

The header has six fields as follows. A client uses the identification field to match the reply with the query. This field may appear with a different number each time a client transmits a query. The server copies this number in its reply. The flags field contains subfields that represent the type of the message, such as the type of answer requested or requested DNS recursive or iterative mapping. The number of questions field indicates how many queries are in the question portion of the message. The number of answers shows how many answers are in the answer field. For the query message, this field contains all zeros. The number of authoritative records field consists of the number of authoritative records in the authority portion of a reply message. Similarly, this field is filled by zeros for a query message. Finally, the number of additional records field records are in the additional information portion of a reply message and is similarly filled by zeros in a query message.

5.3. Remote Login Protocols

A client/server model can create a mechanism that allows a user to establish a session on the remote machine and then run its applications. This application is known as remote login. This can be done by a client/server application program for the desired service. Two remote login protocols are TELNET and SSH.

5.3.1. TELNET Protocol

TELNET (terminal network) is a TCP/IP standard for establishing a connection to a remote system. TELNET allows a user to log in to a remote machine across the Internet by first making a TCP connection and then pass the detail of the application

from the user to the remote machine..

Logging to Remote Servers

With TELNET, an application program on the user's machine becomes the client. The user's keyboard and its monitor also attach directly to the remote server. The remote-logging operation is based on timesharing, whereby an authorized user has a login name and a password. TELNET has the following properties.

- Client programs are built to use the standard client/server interfaces without knowing the details of server programs.
- A client and a server can negotiate data format options.
- Once a connection is established through TELNET, both ends of the connection are treated symmetrically.

When a user logs in to a remote server, the client's terminal driver accepts the keystrokes and interprets them as characters by its operating system. Characters are typically transformed to a universal character set called network virtual terminal (NVT), which uses 7-bit USASCII representation for data. The client then establishes a TCP connection to the server. Texts in the NVT format are transmitted using a TCP session and are delivered to the operating system of the remote server. The server converts the characters back from NVT to the local client machine's format.

5.3.2. Secure Shell (SSH) Protocol

Secure Shell (SSH), another remote login protocol, is based on UNIX programs. SSH uses TCP for communications but is more powerful and flexible than TELNET and allows the user to more easily execute a single command on a remote client. SSH has the following advantages over TELNET.

- SSH provides a secure communication by encrypting and authenticating messages.

- SSH provides several additional data transfers over the same connection by multiplexing multiple channels that are used for remote login.

  SSH security is implemented by using public-key encryption between the client and remote servers. When a user establishes a connection to a remote server, the data being transmitted remains confidential even if an intruder obtains a copy of the packets sent over an SSH connection. SSH also implements an authentication process on messages so that a server can find out and verify the host attempting to form a connection. Normally, SSH requires users to enter a private password.

  The advantage of port forwarding is that application data can be passed between two sites the client and the second server without requiring a second client and server the first server as a client and the second server.Figure 5.7 shows the format of an SSH packet.

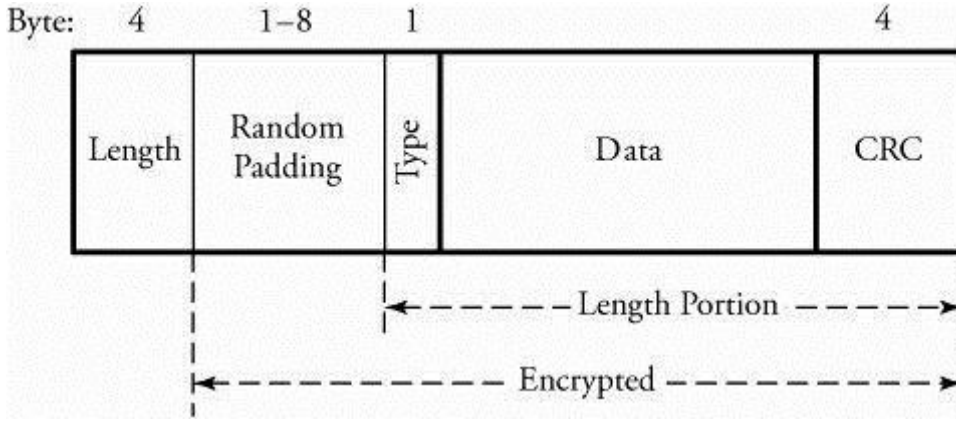


Figure 5.7. SSH packet format

- Length indicates the size of the packet, not including the length field or the variable-length random padding field that follows it.

- Padding causes an intrusion to be more difficult.
- Type identifies the type of message.
- CRC, or cyclic redundancy check, is an error-detection field.

5.4. Electronic Mail (E-mail)

5.4.1. Simple Mail Transfer Protocol (SMTP) and E-mail

The Simple Mail Transfer Protocol (SMTP) plays a major role in transferring Internet electronic mail. This protocol transfers electronic mail (e-mail) from the mail server of a source to the mail servers of destinations. SMTP is older than the Hypertext Transfer Protocol (HTTP), the Web communication protocol, and imposes certain restrictions, such as limits on the size of e-mail content.

In Figure 5.8, user 1 is in a residential area, has an Internet service provider (ISP), and is sending an e-mail to user 2, working in an organization. Suppose that the mail servers are isp.com and organization.com, respectively. Thus, user 1 and user 2 have e-mail addresses of user1@isp.com and user2@organization.com, respectively. The procedure for an e-mail exchange between user 1 and user 2 is as follows.

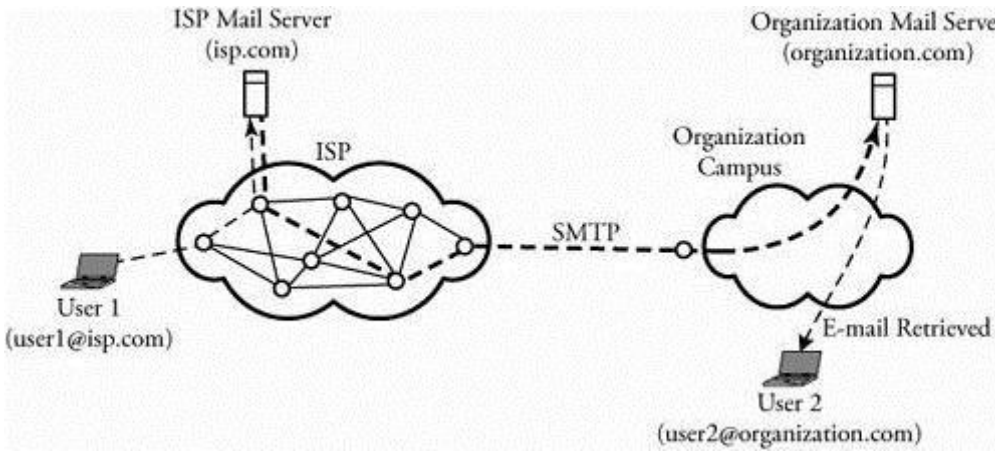


Figure 5.8. Two users exchanging e-mail through SMTP

Begin SMTP Between Two Users

1. User 1 provides user 2's e-mail address (user2@organization.com) and composes its message.
2. User 1 sends the message to its mail server (isp.com).
3. Server isp.com places the message in its queue.
4. SMTP on user 1's mail server notices the message in the queue and opens a TCP connection with the organization mail server (organization.com).
5. Initial SMTP handshaking takes place between the two servers.
6. The message is sent to organization.com's mail server, using the established TCP connection.
7. User 2's mail server receives the message and then puts it in user 2's mailbox, ready to be retrieved by user 2.

5.5. File Transfer and FTP

File transfer is another computer networking application. It is always essential that files and information geographically distributed over different locations be shared among the members of a working group. In a certain application, files are typically saved in a server. A user then uses a file transfer protocol to access the server and transfer the desired file. Two file transfer protocols are FTP and SCP.

5.5.1. File Transfer Protocol (FTP)

File Transfer Protocol (FTP) is part of the TCP/IP suite and is very

similar to TELNET. Both FTP and TELNET are built on the client/server paradigm, and both allow a user to establish a remote connection. However, TELNET provides a broader access to a user, whereas FTP allows access only to certain files. The essence of this protocol is as follows.

Begin File Transfer Protocol

1. A user requests a connection to a remote server.
2. The user waits for an acknowledgment.
3. Once connected, the user must enter a user ID, followed by a password.
4. The connection is established over a TCP session.
5. The desired file is transferred.
6. The user closes the FTP connection.

FTP can also run through a Web browser.

5.5.2. Secure Copy Protocol (SCP)

The Secure Copy Protocol (SCP) is similar to TELNET but is secure. Incorporated in the SCP structure are a number of encryption and authentication features that are similar to those in SSH. Also similar is the exchange of commands between local and remote hosts. SCP commands automatically prompt the user for the password information when it is time to access a remote machine. SCP cannot handle file transfer between machines of significantly different architectures.

5.6. World Wide Web (WWW) and HTTP

Application-layer software is the intelligence built for end servers. The World Wide Web (WWW), or simply Web, is a global network of servers linked by a common protocol allowing access to all connected hypertext resources. When a client host requests an object, a Web server responds by sending the requested object through browsing tools. A browser is a user agent displaying the requested Web page. The Hyper Text Transfer Protocol (HTTP) transfers that page at the application layer. HTTP uses TCP rather than UDP, since reliability of delivery is important for Web pages with text. The TCP connection-establishment delay in HTTP is one of the main contributing delay factors associated with downloading Web documents.

HTTP is based on the client/server idea, having a client and a server program, both of which can be executed on different end systems. The communication is carried out through an exchange of HTTP messages. This protocol specifies the structure of these messages. For example, HTTP defines how a pair of client/server hosts should exchange messages. In this context, a Web page consists of files, such as Hypertext Mark-up Language (HTML) file or an image that can be addressed by a single uniform resource locator (URL). A URL is a global address of an HTML document and has two parts. The first part indicates what protocol is used, and the second part determines the IP address of the associated resource.

5.6.1. Web Caching (Proxy Server)

An HTTP request from a user is first directed to the network proxy server, or Web cache. Once configured by the network, a browser's request for an

object is directed to the Web cache, which must contain updated copies of all objects in its defined proximity. The main reason for Web caching is to reduce the response time for a user request. This benefit is much more obvious when the bandwidth to a requested server is limited because of traffic at certain hours of the day. Normally, each organization or ISP should have its own cache providing a high-speed link to its users. Consequently, it is to users' advantages that this rapid method of finding objects be available. This method of Internet access also reduces traffic on an organization's access link to the Internet. The details of Web caching are as follows:

Begin Web Caching Algorithm

1. The source browser makes a TCP connection to the Web cache.
2. The user browser transmits its HTTP request to the Web cache.
3. If it has a copy of the requested object, the Web cache forwards the object to the user browser.

   Otherwise the Web cache establishes a TCP connection to the requested server and asks for the object. Once it receives the requested object, the Web cache stores a copy of it and forwards another copy to the requesting user browser over the existing TCP connection.

   Figure 5.9 shows three Internet service providers (ISPs). A user in ISP domain 3 is browsing to find and watch an object named http://www.filmmaker.com in ISP domain 1. The request for this object is directed to the Web cache, shown by dashed lines. In this example, the Web cache has no record of the requested object and therefore is establishing another TCP connection to update its record.

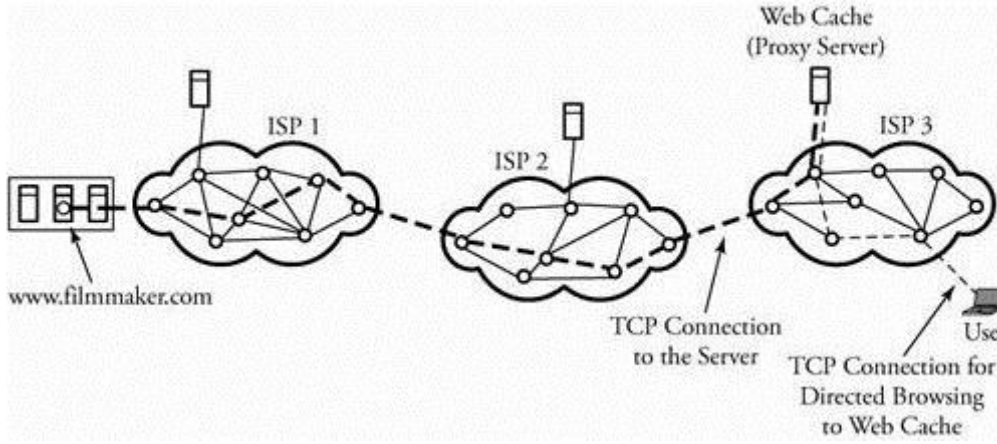


Figure 5.9. A user's browser requesting an object through the Web cache

5.7. Network Management

The main purpose of network management is to monitor, manage, and control a network. A network can be structured with many links, routers, servers, and other physical-layer devices, which can be equipped with many network protocols that coordinate them. Imagine when thousands of such devices or protocols are tied together by an ISP and how drastic their management can become to avoid any interruptions in routine services. In this context the purpose of network management is to monitor, test, and analyze the hardware, software, and human elements of a network and then to configure and control those elements to meet the operational performance requirements

of the network.

Figure 5.10 illustrates a simple network management scenario in which LANs connect to the Internet. LAN 1 is dedicated to the network administrator facilities. The network administrator can periodically send management packets to communicate with a certain network entity. A malfunctioning component in a network can also initiate communication of its problem to the network administrator.

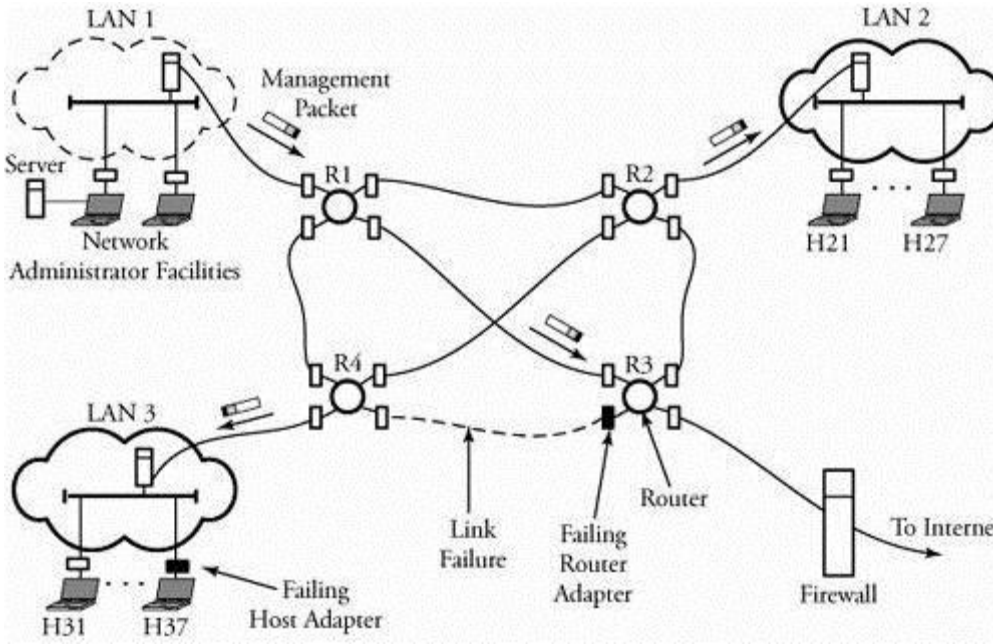


Figure 5.10. Simple network management in a scenario of LANs connecting to the Internet

Network management tasks can be characterized as follows:

- QoS and performance management. A network administrator periodically

monitors and analyzes routers, hosts, and utilization of links and then redirect traffic flow to avoid any overloaded spots. Certain tools are available to detect rapid changes in traffic flow.

- Network failure management. Any fault in a network, such as link, host, or router hardware or software outages, must be detected, located, and responded to by the network. Typically, increased checksum errors in frames is an indication of possible error. Figure 5.10 shows adapter failures at router R3 and host H37; these failures can be detected through network management.
- Configuration management. This task involves tracking all the devices under management and ensuring that all devices are connected and operate properly. If there is an unexpected change in routing tables, a network administrator wants to discover the misconfigured spot and reconfigure the network before the error affects the network substantially.
- Security management. A network administrator is responsible for the security of its network. This task is handled mainly through firewalls. A firewall can monitor and control access points. In such cases, the network administrator wants to know about any intrusion from a suspicious source to the network. For example, a host in a network can be attacked by receiving a large number of SYN packets.
- Billing and accounting management. The network administrator specifies user access or restrictions to network resources and issues all billing and charges, if any, to users.

Locating a failing point, such as an adapter failure at a host or a router, can be

done by appropriate network management tools. Normally, a standard packet format is specified for network management.

5.7.1. Elements of Network Management

Network management has three main components: network management: a managing center, a managed device, and a network management protocol. The managing center consists of the network administrator and his or her facilities. Typically, the managing center comprises a substantial human network. A managed device is the network equipment, including its software, that is controlled by the managing center. Any hub, bridge, router, server, printer, or modem can be a managed device. The network management protocol is a policy between the managing center and the managed devices. The protocol in this context allows the managing center to obtain the status of managed devices. In network management, an agent is a managed device, such as a router, hub, or bridge. A manager is a network administrative device, as a management host. An agent can use the network management protocol to inform the managing center of an unexpected event.

5.7.2. Structure of Management Information (SMI)

The structure of management information (SMI) language is used to define the rules for naming objects and to encode objects in a managed network center. In other words, SMI is a language by which a specific instance of the data in a managed network center is defined. For example, Integer32 means a 32-bit integer with a value between -2^{31} and $-2^{31} - 1$. The SMI language also provides higher-level language constructs, which typically specify the data type, status, and semantics of managed objects containing the management

data. For example, the STATUS clause specifies whether the object definition is current or obsolete, ipInDelivers defines a 32-bit counter to trace the number of IP datagrams received at a managed device and then received at an upper-layer protocol.

5.7.3. Management Information Base (MIB)

Management information base (MIB) is an information storage medium that contains managed objects reflecting the current status of the network. Because managed objects have associated pieces of information that are stored in a MIB, the MIB forms a collection of named objects, including their relationships to one another in a management center. The information pieces can be obtained by directing the managing center to do so.

Objects are organized in a hierarchical manner and are identified by the abstract syntax notation one (ASN.1) object definition language. The hierarchy of object names, known as ASN.1 object identifier, is an object identifier tree in which each branch has both a name and a number, as shown in Figure 5.11. Network management can then identify an object by a sequence of names or numbers from the root to that object.

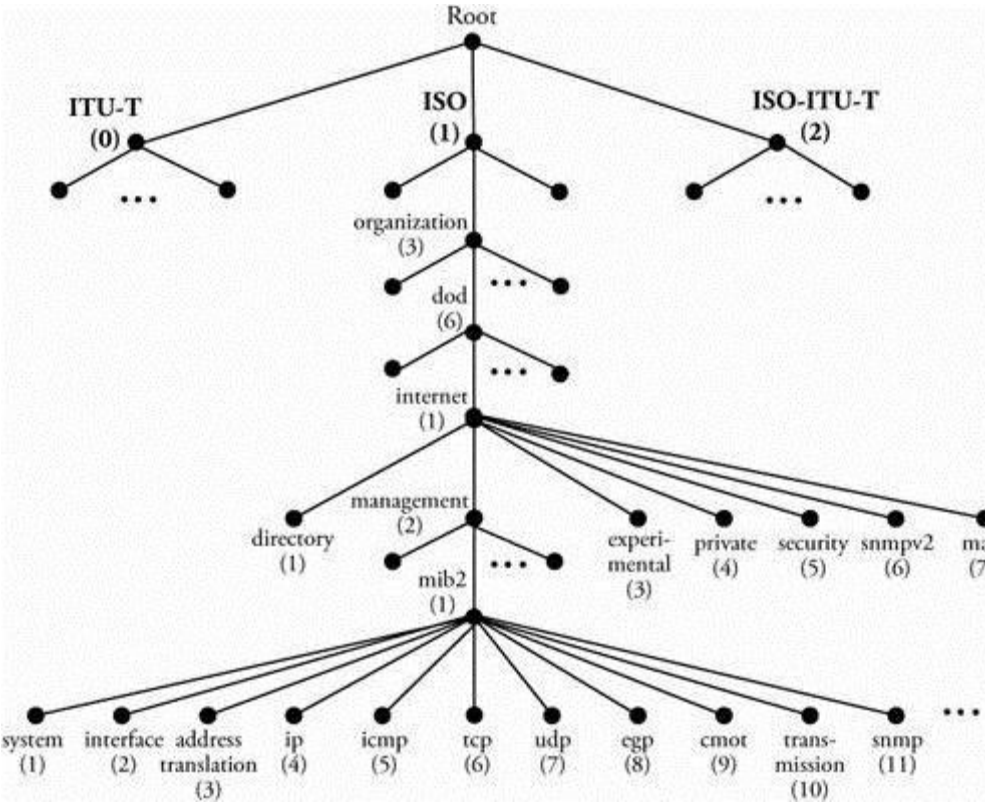


Figure 5.11. ASN.1 object identifier organized hierarchically

On the root of the object identifier hierarchy are three entries: ISO (International Standardization Organization), ITU-T (International Telecommunication Union-Telecommunication) standardization sector, and ISO-ITU-T, the joint branch of these two organizations. Figure 5.11 shows only part of the hierarchy. Under the ISO entry are other branches. For example, the organization (3) branch is labeled sequentially from the root as 1.3. If we continue to follow the entries on this branch, we see a path over

dod (6), Internet (1), management (2), mib-2(1), and ip (4). This path is identified by (1.3.6.1.2.1.4) to indicate all the labeled numbers from the root to the ip (4) entry. Besides that entry, MIB module represents a number of network interfaces and well-known Internet protocols at the bottom of this tree. This path clearly shows all the standards of "IP" associated with the "MIB-2" computer networking "management."

5.7.4. Simple Network Management Protocol (SNMP)

The Simple Network Management Protocol (SNMP) is designed to monitor the performance of network protocols and devices. SNMP protocol data units (PDUs) can be carried in the payload of a UDP datagram, and so its delivery to a destination is not guaranteed. Managed devices, such as routers and hosts, are objects, and each object has a formal ASN.1 definition. For each object, MIB accommodates a database of information that describes its characteristics. With this protocol, a network manager can find the location of a fault. SNMP runs on top of UDP and uses client/server configurations. The commands of this protocol define how to query information from a server and forward information to a server or a client.

The task of SNMP is to transport MIB information among managing centers and agents executing on behalf of managing centers. For each managed MIB object, an SNMP request is used to retrieve or change its associated value. If an unsolicited message is received by an agent, or when an interface or device goes down, the protocol can also inform the managing center. The second version of this protocol, SNMPv2, runs on top of more protocols and has more messaging options, resulting in more effective network

management. SNMPv3 has more security options.

SNMPv2 has seven PDUs, or messages, as follows.

1. GetRequest is used to obtain a MIB object value.
2. GetNextRequest is used to obtain the next value of a MIB object.
3. GetBulkRequest gets multiple values, equivalent to multiple GetRequests but without using multiple overheads.
4. InformRequest is a manager-to-manager message that two communicating management centers are remote to each other.
5. SetRequest is used by a managing center to initiate the value of a MIB object.
6. Response is a reply message to a request-type PDU.
7. Trap notifies a managing center that an unexpected event has occurred.

Figure 5.12 shows the format of SNMP PDUs. Two types of PDUs are depicted: Get or Set and Trap. The Get or Set PDU format is as follows:

- PDU type indicates one of the seven PDU types.
- Request ID is an ID used to verify the response of a request. Thus, a managing center can detect lost requests or replies.
- Error status is used only by Response PDUs to indicate types of errors reported by an agent.
- Error index is a parameter indicating to a network administrator which name has caused an error.

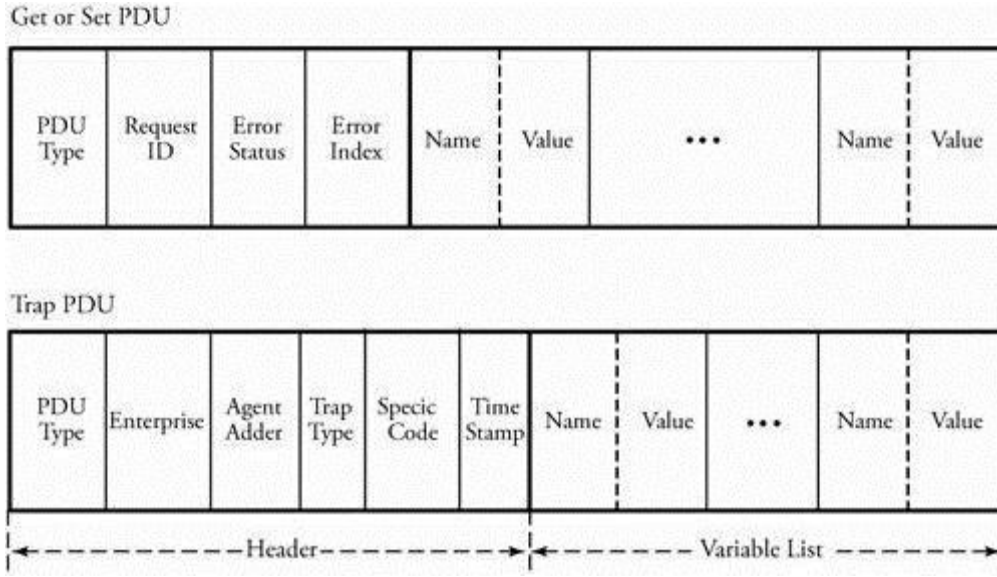


Figure 5.12. SNMP PDU format

If requests or replies are lost, SNMP does not mandate any method for retransmission. Error status and Error index fields are all zeros except for the one in a GetBulkRequest PDU. Figure 5.12 also shows the format of the Trap PDU, whereby the enterprise field is for use in multiple networks; the timestamp field, for measuring up time; and the agent address field, for indicating that the address of the managed agent is included in the PDU header.

5.8. Network Security

We begin by identifying and classifying types of network threats, hackers, and attacks, including DNS hacking attacks and router attacks. Network security can be divided into two broad categories:

cryptographic techniques and authentication techniques (verification). Both secret-key and public-key encryption protocols are presented. We then discuss message authentication and digital signature methods, through which a receiver can be assured that an incoming message is from whom it says it is.

We then consider several standardized security techniques, such as IPsec, and the security of wireless networks and IEEE 802.11 standards, as well as firewalls, or devices designed to protect a network. Finally, we present a case study on the security of wireless networks.

5.8.1. Overview of Network Security

Network security is a top-priority issue in data networks. As communication networks are growing rapidly, security issues have pushed to the forefront of concern for end users, administrators, and equipment suppliers. Despite enormous joint efforts by various groups to develop effective security solutions for networks, hackers continue to pose new, serious threats by taking advantage of weaknesses present in the Internet infrastructure.

Elements of Network Security

Network security is concerned mainly with the following two elements:

1. Confidentiality. Information should be available only to those who have rightful access to it.
2. Authenticity and integrity. The sender of a message and the message itself should be verified at the receiving point.

In Figure 5.13, user 1 sends a message ("I am user 1") to user 2. In part (a) of the figure, the network lacks any security system, so an intruder can receive the message, change its content to a different message ("Hi! I am user 1") and send it to user 2. User 2 may not know that this falsified message is really from user 1 (authentication) and that the content of the message is what user 1 (confidentiality). In part (b) of the figure, a security block is added to each side of the communication, and a secret key that only users 1 and 2 would know about is included. Therefore, the message is changed to a form that cannot be altered by the intruder, who would be disabled in this communication transaction.

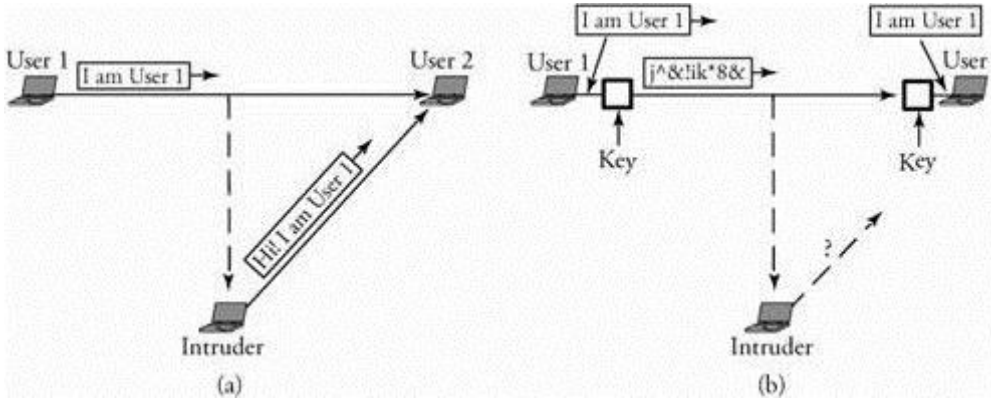


Figure 5.13 (a) Message content and sender identity falsified by intruder; (b) a method of applied security

In general, no protocol or network architecture can ensure full security. Internet routing is based on a distributed system of many routers, switches, and protocols. These protocols have a number of points of vulnerabilities that can be exploited to cause such problems as misdelivery or no delivery of user traffic, misuse of network resources, network congestion and packet delays, and the violation of local routing policies.

5.8.2. Threats to Network Security

Internet infrastructure attacks are broadly classified into four categories, as follows:

1. DNS hacking
2. Routing table poisoning
3. Packet mistreatment

4. Denial of service

Among these threats, the first three attacks are related to network infrastructure; denial-of-service attacks are related to end systems.

DNS Hacking Attacks

The Domain Name System (DNS) server is a distributed hierarchical and global directory that translates domain names into numerical IP address. DNS is a critical infrastructure, and all hosts contact DNS to access servers and start connections. In the normal mode of operation, hosts send UDP queries to the DNS server. Servers reply with a proper answer, or direct the queries to smarter servers. A DNS server also stores information other than host addresses.

Name-resolution services in the modern Internet environment are essential for e-mail transmission, navigation to Web sites, or data transfer. Thus, an attack on DNS can potentially affect a large portion of the Internet. A DNS hacking attack may result in the lack of data authenticity and integrity and can appear in any of the following forms:

1. An information-level attack forces a server to correspond with other than the correct answer. With cache poisoning, a hacker tricks a remote name server into caching the answer for a third-party domain by providing malicious information for the domain's authorized servers. Hackers can then redirect traffic to a preselected site.
2. In a masquerading attack, the adversary poses as a trusted entity and obtains all the secret information. In this guise, the attacker can stop any message from being transmitted further or can change the content or redirect the

packet to bogus servers. This action is also known as a middle-man attack.

3. The attacker normally sends queries to each host and receives in reply the DNS host name. In an information leakage attack, the attacker sends queries to all hosts and identifies which IP addresses are not used. Later on, the intruder can use those IP addresses to make other types of attacks.
4. Once a domain name is selected, it has to be registered. Various tools are available to register domain names over the Internet. If the tools are not smart enough, an invader might obtain secure information and use it to highjack the domain later. In the domain high jacking attack, whenever a user enters a domain address, she/he is forced to enter into the attacker's Web site. This can be very irritating and can cause a great loss of Internet usage ability.

Routing Table Poisoning Attacks

A routing table poisoning attack is the undesired modification of routing tables. An attacker can do this by maliciously modifying the routing information update packets sent by routers. This is a challenging and important problem, as a routing table is the basis of routing in the Internet. Any false entry in a routing table could lead to significant consequences, such as congestion, an overwhelmed host, looping, illegal access to data, and network partition. Two types of routing table poisoning attacks are the link attack and the router attack.

A link attack occurs when a hacker gets access to a link and thereby intercepts, interrupts, or modifies routing messages on packets. Link attacks act similarly on both the link-state and the distance-vector protocols. If an attacker succeeds in placing an attack in a link-state routing protocol, a router

may send incorrect updates about its neighbors or remain silent even if the link state of its neighbor has changed. The attack through a link can be so severe that the attacker can program a router to either drop packets from a victim or readdress packets to a victim, resulting in a lower throughput of the network. Sometimes, a router can stop an intended packet from being forwarded further. However, since more than one path to any destination exists, the packet ultimately reaches its destination.

Router attacks may affect the link-state protocol or even the distance-vector protocol. If link-state protocol routers are attacked, they become malicious. They may add a nonexisting link to a routing table, delete an existing link, or even change the cost of a link. This attack may cause a router to simply ignore the updates sent by its neighbors, leading to a serious impact on the operability of the network traffic flow.

In the distance-vector protocol, an attacker may cause routers to send wrong updates about any node in the network, thereby misleading a router and resulting in network problems.

Most unprotected routers have no way to validate updates. Therefore, both link-state and distance-vector router attacks are very effective. In the distance-vector protocol, for example, a malicious router can send wrong information in the form of a distance vector to all its neighbors. A neighbor may not be able to detect this kind of attack and thus proceeds to update its routing table, based on wrong distance vectors. The error can in turn be propagated to a great portion of the network before being detected.

Packet-Mistreatment Attacks

A packet-mistreatment attack can occur during any data transmission. A hacker may capture certain data packets and mistreat them. This type of attack is very difficult to detect. The attack may result in congestion, lowering throughput, and denial-of-service attacks. Similar to routing table poisoning attacks, packet-mistreatment attacks can also be subclassified into link attacks and router attacks. The link attack causes interruption, modification, or replication of data packets. A router attack can misroute all packets and may result in congestion or denial of service. Following are some examples of a packet-mistreatment attack:

- Interruption. If an attacker intercepts packets, they may not be allowed to be propagated to their destinations, resulting in a lower throughput of the network. This kind of attack cannot be detected easily, as even in normal operations, routers can drop some packets, for various reasons.
- Modification. Attackers may succeed in accessing the content of a packet while in transit and change its content. They can then change the address of the packet or even change its data. To solve this kind of problem, a digital signature mechanism, discussed later in this chapter, can be used.
- Replication. An attacker might trap a packet and replay it. This kind of attack can be detected by using the sequence number for each packet.
- Ping of death. An attacker may send a ping message, which is large and therefore must be fragmented for transport. The receiver then starts to reassemble the fragments as the ping fragments arrive. The total packet length becomes too large and might cause a system crash.
- Malicious misrouting of packets. A hacker may attack a router and change its

routing table, resulting in misrouting of data packets, causing a denial of service.

Denial-of-Service Attacks

A denial-of-service attack is a type of security breach that prohibits a user from accessing normally provided services. The denial of service does not result in information theft or any kind of information loss but can nonetheless be very dangerous, as it can cost the target person a large amount of time and money. Denial-of-service attacks affect the destination rather than a data packet or router.

Usually, a denial-of-service attack affects a specific network service, such as e-mail or DNS. For example, such an attack may overwhelm the DNS server in various ways and make it inoperable. One way of initiating this attack is by causing buffer overflow. Inserting an executable code inside memory can potentially cause a buffer overflow. Or, an adversary may use various tools to send large numbers of queries to a DNS server, which then is not able to provide services in a timely manner.

Denial-of-service attacks are easy to generate but difficult to detect. They take important servers out of action for few hours, thereby denying service to all users. There are yet a few other situations that can cause this kind of attack, such as UDP flood, a TCP flood and ICMP flood. In all these attacks, the hacker's main aim is to overwhelm victims and disrupt services provided to them.

Denial-of-service attacks are two types:

1. Single-source. An attacker sends a large number of packets to a target system

to overwhelm and disable it. These packets are designed such that their real sources cannot be identified.

2. Distributed. In this type of attack, a large number of hosts are used to flood unwanted traffic to a single target. The target cannot then be accessible to other users in the network, as it is processing the flood of traffic.

   The flood may be either a UDP flood or a TCP SYN flood. UDP flooding is used against two target systems and can stop the services offered by either system. Hackers link the UDP character-generating services of a system to another one by sending UDP packets with spoofed return addresses. This may create an infinite looping between the two systems, leading to system uselessness.

   Normally, a SYN packet is sent by a host to a user who intends to establish a connection. The user then sends back an acknowledgment. In the TCP SYN flood, a hacker sends a large number of SYN packets to a target user. Since the return addresses are spoofed, the target user queues up a SYN/ACK packet and never processes it. Therefore, the target system keeps on waiting. The result may be a hard disk crash or reboot.

5.8.2. Overview of Security Methods

Common solutions that can protect computer communication networks from attacks are classified as cryptographic techniques or authentication techniques (verification).

Cryptographic Techniques

Cryptography has a long and fascinating history. Centuries ago, cryptography was used as a tool to protect national secrets and strategies. Today, network engineers focus on cryptography methods for computer communication networks. Cryptography is the process of tranforming a piece of information or message shared by two parties into some sort of code. The message is scrambled before transmission so that it is undetectable by outside watchers. This kind of message needs to be decoded at the receiving end before any further processing.

The main tool that network security experts are using to encrypt a message M is a secret key K; the fundamental operation often used to encrypt a message is the Exclusive-OR ($\oplus$). Suppose that we have one bit, M, and a secret bit, K. A simple encryption is carried out using M $\oplus$K. To decrypt this message, the second partyif he/she has the key, Kcan easily detect M by performing the following:

Equation 10.1

$$(M \oplus K) \oplus K = M.$$

In computer communication networks, data can travel between two users while it is encrypted. In Figure 5.14, two servers are exchanging data while two types of encryption devices are installed in their communication network. The first encryption device is end-to-end encryption, whereby secret coding is carried out at both end systems.

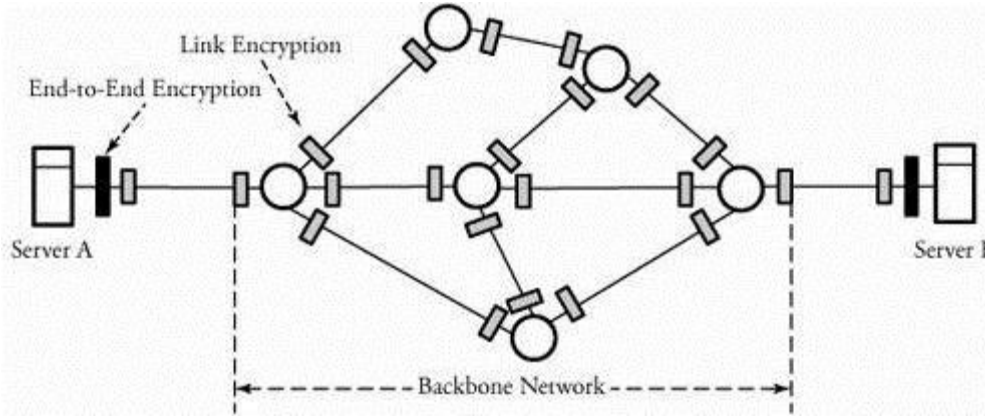


Figure 5.14. Overview of encryption points in a communication network

In this figure, server A encodes its data, which can be decoded only at the other end server. The second phase of encryption is link encryption, which secures all the traffic passing over that link.

The two types of encryption techniques are secret-key encryption and public-key encryption. In a secret-key model, both sender and receiver conventionally use the same key for an encryption process. In a public-key model, a sender and a receiver each use a different key. The public-key system is more powerful than the secret-key system and provides better security and message privacy. But the biggest drawback of public-key encryption is speed. The public-key system is significantly more complex computationally and may not be practical in many cases. Hence, the public-key system is used only to establish a session to exchange a session key. Then, this session key is used in a secret-key system for encrypting messages for the duration of the session.

Authentication Techniques

Encryption methods offer the assurance of message confidentiality. However, a networking system must be able to verify the authenticity of the message and the sender of the message. These forms of security techniques in computer networks are known as authentication techniques and are categorized as authentication with message digest and authentication with digital signature. Message authentication protects a user in a network against data falsification and ensures data integrity. These methods do not necessarily use keys.

5.9. Secret-Key Encryption Protocols

Secret-key encryption protocols, sometimes known as symmetric encryption, or single-key encryption protocols, are conventional encryption models. They typically consist of an encryption algorithm, a key, and a decryption algorithm. At the end point, the encrypted message is called ciphertext. Several standard mechanisms can be used to implement a secret-key encryption algorithm. Here, we focus on two protocols: Data Encryption Standard (DES) and Advanced Encryption Standard (AES).

In these algorithms, a shared secret key between a transmitter and a receiver is assigned at the transmitter and receiver points. The encryption algorithm produces a different key at any time for a specific transmission. Changing the key changes the output of the algorithm. At the receiving end, the encrypted information can be transformed back to the original data by using a

decryption algorithm and the same key that was used for encryption. The security of conventional encryption depends on the secrecy of the key, not on the secrecy of the encryption algorithm. Consequently, the algorithm need not be kept secret; only the key has to be secret.

5.9.1. Data Encryption Standard (DES)

With the Data Encryption Standard (DES), plaintext messages are converted into 64-bit blocks, each encrypted using a key. The key length is 64 bits but contains only 56 usable bits; thus, the last bit of each 8 byte in the key is a parity bit for the corresponding byte. DES consists of 16 identical rounds of an operation, as shown in Figure 5.15. The details of the algorithm on each 64-bit block of message at each round i of operation are as follows.

Begin DES Algorithm

1. Initialize. Before round 1 begins, all 64 bits of an incoming message and all 56 bits of the secret key are separately permuted (shuffled).
2. Each incoming 64-bit message is broken into two 32-bit halves denoted by L_i and R_i, respectively.
3. The 56 bits of the key are also broken into two 28-bit halves, and each half is rotated one or two bit positions, depending on the round.
4. All 56 bits of the key are permuted, producing version k_i of the key on round i.
5. In this step, $\oplus$ is a logic Exclusive-OR, and the description of function F() appears next. Then, L_i and R_i are determined by

Equation 10.2

$$L_i = R_{i-1}$$

and

Equation 10.3

$$R_i = L_{i-1} \oplus F(R_{i-1}, k_i).$$

6. All 64 bits of a message are permuted.

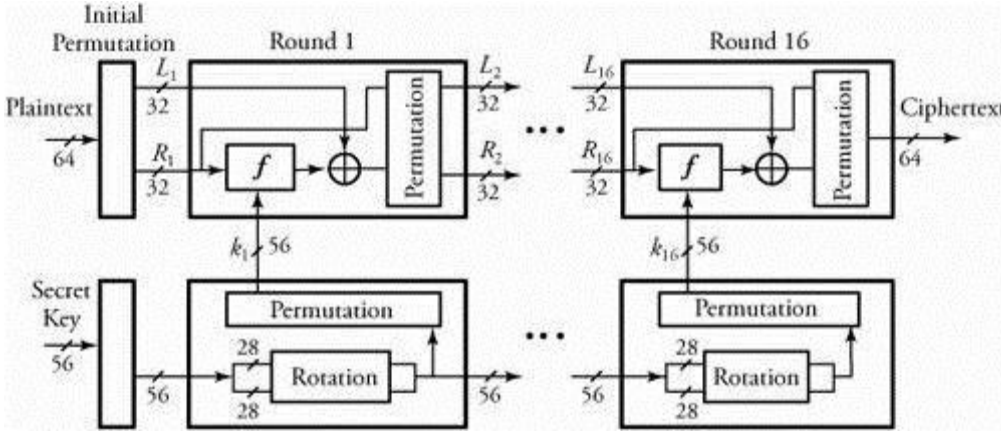


Figure 5.3. The Data Encryption Standard (DES)

The operation of function F() at any round i of DES is as follows.

1. Out of 52 bits of k_i, function F() chooses 48 bits.
2. The 32-bit R_{i-1} is expanded from 32 bits to 48 bits so that it can be combined with 48-bit k_i. The expansion of R_{i-1} is carried out by first breaking R_{i-1} into eight 4-bit chunks and then expanding each chunk by copying the leftmost bit and the rightmost bit from left and right adjacent chunks, respectively.

3. Function F() also partitions the 48 bits of k_i into eight 6-bit chunks.
4. The corresponding eight chunks of R_{i-1} and eight chunks of k_i are combined as follows:

   Equation 10.4

$$R_{i-1} = R_{i-1} \oplus k_i .$$

At the receiver, the same steps and the same key are used to reverse the encryption. It is now apparent that the 56-bit key length may not be sufficient to provide full security. This argument is still controversial. Triple DES provides a solution for this controversy: three keys are used, for a total of 168 bits. It should also be mentioned that DES can be implemented more efficiently in hardware than in software.

5.9.2. Advanced Encryption Standard (AES)

The Advanced Encryption Standard (AES) protocol has a better security strength than DES. AES supports 128-bit symmetric block messages and uses 128-, 192-, or 256-bit keys. The number of rounds in AES is variable from 10 to 14 rounds, depending on the key and block sizes. Figure 5.16 illustrates the encryption overview of this protocol, using a 128-bit key. There are ten rounds of encryptions for the key size of 128 bits. All rounds are identical except for the last round, which has no mix-column stage.

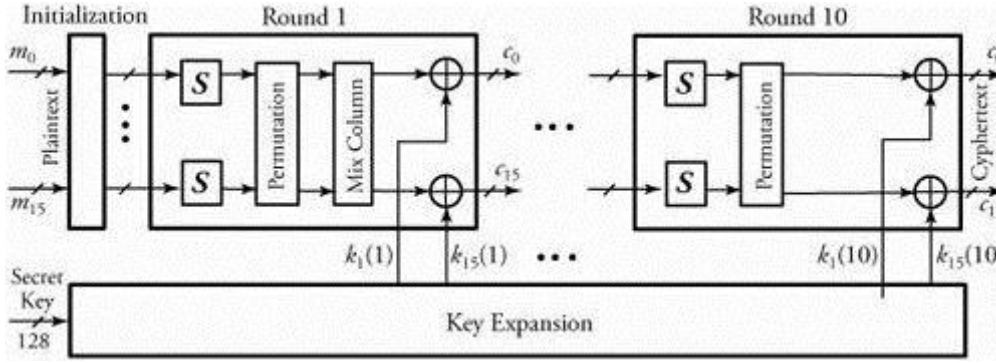


Figure 5.16. Overview of Advanced Encryption Standard (AES) protocol

A single block of 128-bit plaintext (16 bytes) as an input arrives from the left. The plaintext is formed as 16 bytes m_0 through m_{15} and is fed into round 1 after an initialization stage. In this round, substitute unitsindicated by S in the figure perform a byte-by-byte substitution of blocks. The ciphers, in the form of rows and columns, move through a permutation stage to shift rows to mix columns. At the end of this round, all 16 blocks of ciphers are Exclusive-ORed with the 16 bytes of round 1 key $k_0(1)$ through $k_{15}(1)$. The 128-bit key is expanded for ten rounds. The AES decryption algorithm is fairly simple and is basically the reverse of the encryption algorithm at each stage of a round. All stages of each round are reversible.

5.10 Public-Key Encryption Protocols

The introduction of public-key encryption brought a revolution to the field of cryptography. Public-key cryptography provided a very clever method for key exchange. In the public-key encryption model, a sender/receiver pair use

different keys. This model is sometimes known as asymmetric, or two-key, encryption.

Public-key algorithm is based on mathematical functions rather than on substitution or permutation, although the security of any encryption scheme indeed depends on the length of the key and the computational work involved in breaking an encrypted message. Several public-key encryption protocols can be implemented. Among them, the following two protocols are the focus of our study:

- Rivert, Shamir, and Aldeman (RSA) protocol
- Diffie-Hillman key-exchange protocol.

In the public-key encryption methods, either of the two related keys can be used for encryption; the other one, for decryption. It is computationally infeasible to determine the decryption key given only the algorithm and the encryption key. Each system using this encryption method generates a pair of keys to be used for encryption and decryption of a message that it will receive. Each system publishes its encryption key by placing it in a public register or file and sorts out the key as a public one.

The companion key is kept private. If A wishes to send a message to B, A encrypts the message by using B's public key. On receiving the message, B decrypts it with the private B key. No other recipients can decrypt the message, since only B knows its private key. This way, public-key encryption secures an incoming communication as long as a system controls its private key. However, public-key encryption has extra computational overhead and is more complex than the conventional one.

5.10.1. RSA Algorithm

Rivert, Shamir, and Aldeman developed the RSA public-key encryption and signature scheme. This was the first practical public-key encryption algorithm. RSA is based on the intractability of factoring large integers. Assume that a plaintext m must be encrypted to a ciphertext c. The RSA algorithm has three phases for this: key generation, encryption, and decryption.

Key Generation

In the RSA scheme, the key length is typically 512 bits, which requires an enormous computational power. A plaintext is encrypted in blocks, with each block having a binary value less than some number n. Encryption and decryption are done as follows, beginning with the generation of a public key and a private key.

Begin Key Generation Algorithm

1. Choose two roughly 256-bit prime numbers, a and b, and derive n = ab. (A number is prime if it has factors of 1 and itself.)
2. Find x. Select encryption key x such that x and (a - 1)(b - 1) are relatively prime. (Two numbers are relatively prime if they have no common factor greater than 1.)
3. Find y. Calculate decryption key y:

   Equation 10.5

   $$xy \bmod (a-1)(b-1) = 1.$$

4. At this point, a and b can be discarded.

5. The public key = {x, n}.
6. The private key = {y, n}.

In this algorithm, x and n are known to both sender and receiver, but only the receiver must know y. Also, a and b must be large and about the same size and both greater than 1,024 bits. The larger these two values, the more secure the encryption.

Encryption

Both sender and receiver must know the value of n. The sender knows the value of x, and only the receiver knows the value of y. Thus, this is a public-key encryption, with the public key {x, n} and the private key {y, n}. Given m<n, ciphertext c is constructed by

Equation 5.6

$$c = m^x \bmod n.$$

Note here that if a and b are chosen to be on the order of 1,024 bits, n ≈2, 048. Thus, we are not able to encrypt a message longer than 256 characters.

Decryption

Given the ciphertext, c, the plaintext, m, is extracted by

Equation 5.7

$$m = c^y \bmod n.$$

In reality, the calculations require a math library, as numbers are typically huge. One can see easily how Equations (5.6) and (5.7) work.

Example. For an RSA encryption of a 4-bit message of 1,000, or m = 9, we choose a = 3 and b = 11. Find the public and the private keys for this security action, and show the ciphertext.

Solution. Clearly, n = ab = 33. We select x = 3, which is relatively prime to (a - 1)(b - 1) = 20. Then, from xy mod (a - 1)(b - 1) = 3y mod 20 = 1, we can get y = 7. Consequently, the public key and the private key should be {3, 33} and {7, 33}, respectively. If we encrypt the message, we get c = m^x mod n = 9^3 mod 33 = 3. The decryption process is the reverse of this action, as m = c^y mod n = 3^7 mod 33 = 9.

5.10.2. Diffie-Hillman Key-Exchange Protocol

In the Diffie-Hillman key-exchange protocol, two end users can agree on a shared secret code without any information shared in advance. Thus, intruders would not be able to access the transmitted communication between the two users or discover the shared secret code. This protocol is normally used for virtual private networks (VPNs). The essence of this protocol for two users, 1 and 2, is as follows. Suppose that user 1 selects a prime a, a random integer number x_1, and a generator g and creates $y_1 \in \{1, 2, ..., a - 1\}$ such that

Equation 5.8

$$y_1 = g^{x_1} \bmod a.$$

In practice, the two end users agree on a and g ahead of time. User 2 performs the same function and creates y_2:

Equation 5.9

$$y_2 = g^{x_2} \bmod a.$$

User 1 then sends y_1 to user 2. Now, user 1 forms its key, k_1, using the information its partner sent as

Equation 5.10

$$k_1 = y_2^{x_1} \bmod a,$$

and user 2 forms its key, k_2, using the information its partner sent it as

Equation 05.11

$$k_2 = y_1^{x_2} \bmod a.$$

It can easily be proved that the two Keys k_1 and k_2 are equal. Therefore, the two users can now encrypt their messages, each using its own key created by the other one's information.

5.11. Authentication

Authentication techniques are used to verify identity. Message authentication verifies the authenticity of both the message content and the message sender. Message content is authenticated through implementation of a hash function

and encryption of the resulting message digest. The sender's authenticity can be implemented by use of a digital signature.

A common technique for authenticating a message is to implement a hash function, which is used to produce a "fingerprint" of a message. The hash value is added at the end of message before transmission. The receiver recomputed the hash value from the received message and compares it to the received hash value. If the two hash values are the same, the message was not altered during transmission. Once a hash function is applied on a message, m, the result is known as a message digest, or h(m). The hash function has the following properties.

- Unlike the encryption algorithm, the authentication algorithm is not required to be reversible.
- Given a message digest h(m), it is computationally infeasible to find m.
- It is computationally infeasible to find two different messages m_1 and m_2 such that $h(m_1) = h(m_2)$.

Message authentication can be implemented by two methods. In the first method, as shown in Figure 05.17 (a), a hash function is applied on a message, and then a process of encryption is implemented. Thus, a message digest can also be encrypted in this method. At this stage, the encryption can be a public key or a secret key. The authenticity of a message in this method is assured only if the sender and the receiver share the encryption key. At the receiver site, the receiving user 2 has to decrypt the received message digest and compare it with the one made locally at its site for any judgments on the integrity of the message.

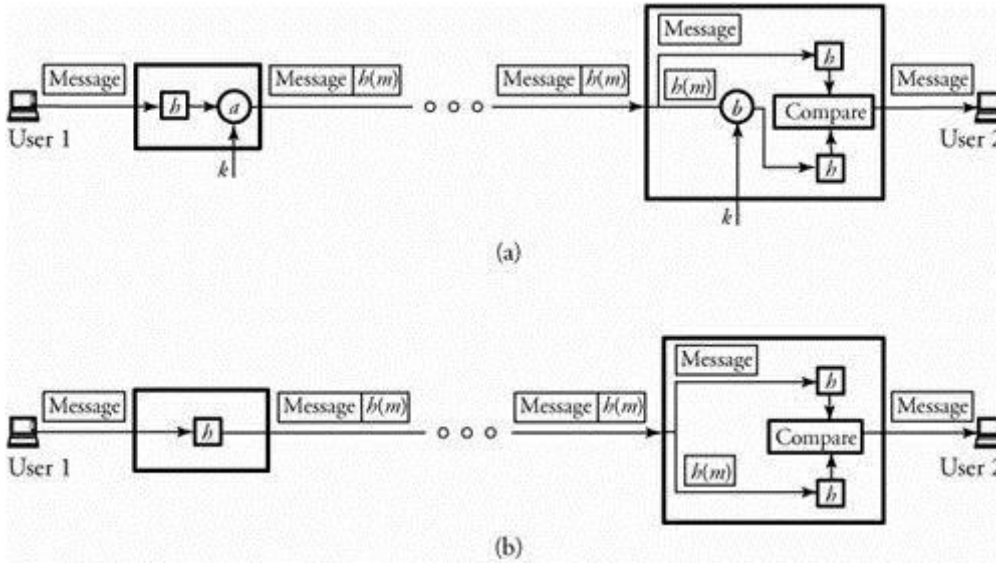


Figure 5.17. Message authentication: (a) combined with encryption; (b) use of the hash function

In the second method, as shown in Figure 5.17 (b), no encryption is involved in the process of message authentication. This method assumes that the two parties share a secret key. Hence, at the receiving site, the comparison is made between the received h(m) and the message digest made locally from the received message. This technique is more popular in the security infrastructure of the Internet Protocol. Among the message authentication protocols are the MD5 hash algorithm and the Secure Hash Algorithm (SHA). SHA is the focus of our discussion.

5.11.1. Secure Hash Algorithm (SHA)

The Secure Hash Algorithm (SHA) was proposed as part of the digital signature standard. SHA-1, the first version of this standard, takes messages with a maximum length of 2^{24} and produces a 160-bit digest. With this algorithm, SHA-1 uses five registers, R_1 through R_5, to maintain a "state" of 20 bytes.

The first step is to pad a message m with length l_m. The message length is forced to $l_m = 448 \bmod 512$. In other words, the length of the padded message becomes 64 bits less than the multiple of 512 bits. The number of padding bits can be as low as 1 bit and as high as 512 bits. The padding includes a 1 bit and as many 0 bits as required. Therefore, the least-significant 64 bits of the message length are appended to convert the padded message to a word with a multiple of 512 bits.

After padding, the second step is to expand each block of 512-bit (16 32 bits) words $\{m_0, m_1, ..., m_{15}\}$ to words of 80 32 bits using:

Equation 5.12

$$w_i = m_i \text{ for } 0 \leq i \leq 15$$

and

Equation 05.13

$$w_i = w_{i-3} \oplus w_{i-8} \oplus w_{i-14} \oplus w_{i-16} \hookleftarrow 1 \text{ for } 16 \leq i \leq 79,$$

where $\hookleftarrow$j means left rotation by j bits. This way, bits are shifted several times if the incoming block is mixed with the state. Next, bits from each

block of w_i are mixed into the state in four steps, each maintaining 20 rounds. For any values of a, b, and c, and bit number i, we define a function $F_i(a, b, c)$ as follows:

Equation 05.14

$$F_i(a,b,c) = \begin{cases} (a \cap b) \cup (\bar{a} \cap c) & 0 \le i \le 19 \\ a \oplus b \oplus c & 20 \le i \le 39 \\ (a \cap b) \cup (a \cap c) \cup (b \cap c) & 40 \le i \le 59 \\ a \oplus b \oplus c & 60 \le i \le 79 \end{cases}.$$

Then, the 80 steps (i = 0, 1, 2, ..., 79) of the four rounds are described as follows:

Equation 05.15

$$\delta = (R_1 \leftarrow\!\!\rightarrow 5) + F_i(R_2, R_3, R_4) + R_5 + w_i + C_i$$

Equation 05.16

$$R_5 = R_4$$

Equation 05.17

$$R_4 = R_3$$

Equation 05.18

$$R_3 = R_2 \leftarrow\!\!\rightarrow 30$$

Equation 05.19

$$R_2 = R_1$$

Equation 05.20

$$R_1 = \delta,$$

where C_i is a constant value specified by the standard for round i. The message digest is produced by concatenation of the values in R_1 through R_5.

5.12. Authentication and Digital Signature

A digital signature is one of the most important required security measures. Much like a person's signature on a document, a digital signature on a message is required for the authentication and identification of the right sender. The digital signature is supposed to be unique to an individual and serves as a means of identifying the sender. An electronic signature is not as easy as it was with the paper-based system. The digital signature requires a great deal of study, research, and skill. Even if these requirements are met, there is no guarantee that all the security requirements have been met.

The technical method of providing a sender's authentication is performed through cryptography. Many cryptographic mechanisms have been developed. Among them, the RSA algorithm implements both encryption and digital signature. When RSA is applied, the message is encrypted with the sender's

private key. Thus, the entire encrypted message serves as a digital signature. This means that at the receiving end, the receiver can decrypt it, using the public key. This authenticates that the packet comes from the right user.

5.13. Firewalls

As the name suggests, a firewall protects data from the outside world. A firewall can be a software program or a hardware device. A firewall a popular security mechanism for networks. A firewall is a simple router implemented with a special program. This unit is placed between hosts of a certain network and the outside world, as shown in Figure 5.18, and the rest of the network. The security issues faced by a smaller network like the one used at home are similar to larger networks. A firewall is used to protect the network from unwanted Web sites and potential hackers.

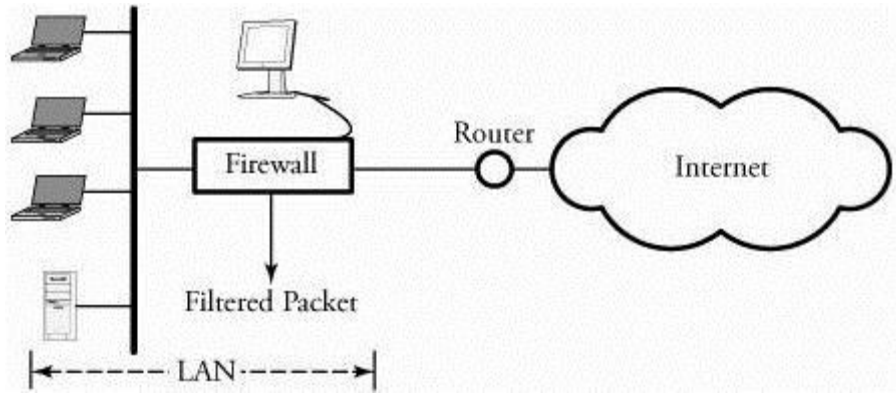


Figure 05.18. A simple configuration of a secured network using a firewall

A firewall is placed on the link between a network router and the Internet or between a user and a router. The objective of such a configuration is to monitor and filter packets coming from unknown sources.

Software firewall programs can be installed in home computers by using an Internet connection with these so-called gateways, the computer with such software can access Web servers only through this software firewall. But hardware firewalls are more secure than software firewalls. Moreover, hardware firewalls are not expensive. Some firewalls also offer virus protection. The biggest security advantage of installing a firewall in a business network is to protect from any outsider logging on to the network under protection.

UNIT – 6

QoS, VPNs, Tunneling, and Overlay Networks: Overview of QoS, Integrated Services QoS, and Differentiated services QoS, Virtual Private Networks, MPLS, Overlay networks.

UNIT – 6 QOS, VPNS, TUNNELING, OVERLAY NETWORKS:

Two broad approaches to QoS are integrated services and differentiated services. Integrated services provide QoS to individual applications and flow records. Providing QoS requires certain features to be maintained in switching nodes. QoS protocols govern traffic shaping and packet scheduling. Traffic shaping regulates the spacing between incoming packets. Other advanced QoS protocols covered are admission control and RSVP.

The differentiated services provide QoS support to a broad class of applications. The basics of resource allocation for packet-switched networks are reviewed, as is resource allocation for all possible layers of the protocol stack. Resource-allocation algorithms can be used to avoid possible ATM congestion.

6.1 Overview of QoS:

Communication networks face a variety of quality-of-service demands. The main motivation for a QoS unit in a data network port processor is to control access to available bandwidth and to regulate traffic. Traffic regulation traffic is always necessary in WANs in order to avoid congestion. A network must be designed to support both real-time and non-real-time applications. Voice and video transmissions over IP must be able to request a higher degree of assurance from the network. A network that can provide these various levels of services requires a more complex structure.

The provision of QoS to a network either does or does not come with a guarantee. Nonguaranteed QoS is typically based on the best-effort model, whereby a network provides no guarantees on the delivery of packets but makes its best effort to do so. In a non-real-time application, a network can use the retransmit strategy for successful data delivery. However, in a real-time application, such as voice or video networking, where timely delivery of packets is required, the application requires a low-latency communication. Consequently, the network must be able to handle packets of such applications more carefully.

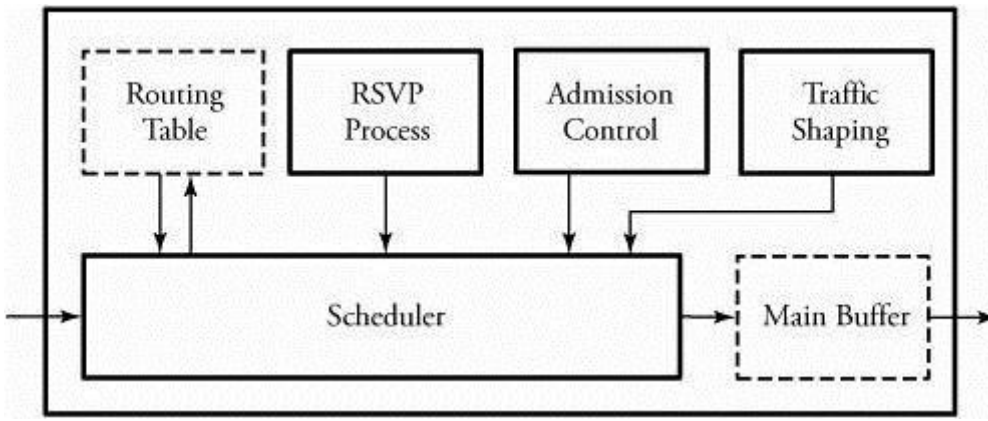
6.2 Integrated Services QoS:

The integrated services approach, consisting of two service classes, defines both the service class and mechanisms that need to be used in routers to provide the services associated with the class. The first service class, the guaranteed service class, is defined for applications that cannot tolerate a delay beyond a particular value. This type of service class can be used for real-time applications, such as voice or video communications. The second, controlled-load service class, is used for applications that can tolerate some delay and loss. Controlled-load service is designed such that applications run very well when the network is not heavily loaded or congested. These two service classes cover the wide range of applications on the Internet. A network needs to obtain as much information as possible about the flow of traffic to provide optimum services to the flow. This is true especially when real-time services to an application are requested. Any application request to a network has to first specify the type of service required, such as controlled load or guaranteed service. An application

that requires guaranteed service also needs to specify the maximum delay that it can tolerate. Once the delay factor is known to a service provider, the QoS unit of the node must determine the necessary processes to be applied on incoming flows. Figure 6.2. shows four common categories of processes providing quality of service.

1. Traffic shaping regulates turbulent traffic.
2. Admission control governs whether the network, given information about an application's flow, can admit or reject the flow.
3. Resource allocation lets network users reserve bandwidth on neighboring routers.
4. Packet scheduling sets the timetable for the transmission of packet flows. Any involving router needs to queue and transmit packets for each flow appropriately.

Figure 6.2. Overview of QoS methods in integrated services



The widespread deployment of the integrated services approach has been deterred owning to scalability issues. As the network size increases, routers need

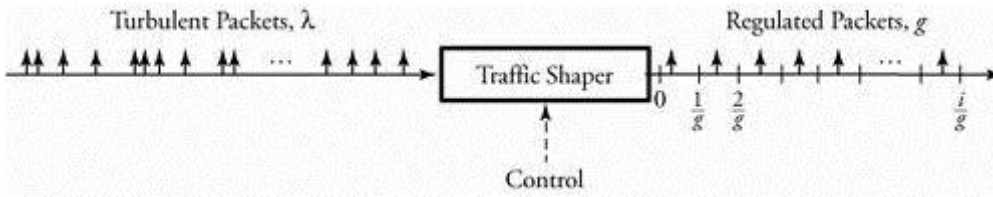
to handle larger routing tables and switch larger numbers of bits per second. In such situations, routers need to refresh information periodically. Routers also have to be able to make admission-control decisions and queue each incoming flow. This action leads to scalability concerns, especially when networks scale larger.

6.2.1. Traffic Shaping

Realistically, spacing between incoming packets has an irregular pattern, which in many cases causes congestion. The goal of traffic shaping in a communication network is to control access to available bandwidth to regulate incoming data to avoid congestion, and to control the delay incurred by packets (see Figure 12.2). Turbulent packets at rate λ and with irregular arrival patterns are regulated in a traffic shaper over equal-sized 1/g intervals.

Figure 6.2.2. Traffic shaping to regulate any incoming turbulent traffic

[View full size image]



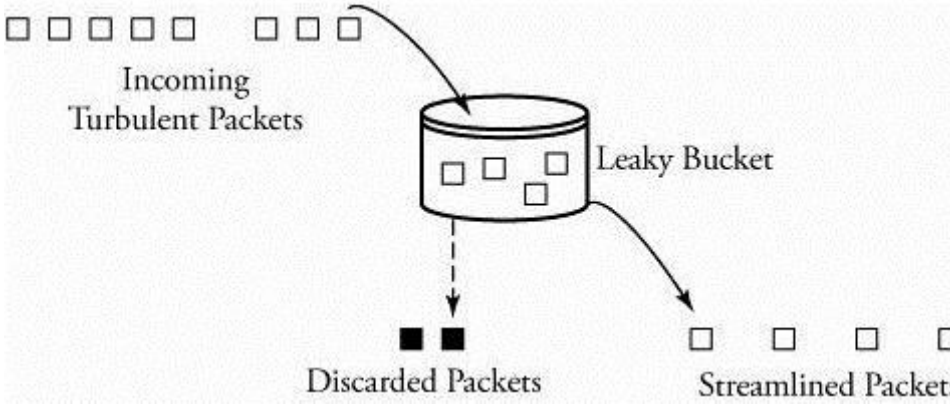
If a policy dictates that the packet rate cannot exceed a specified rate even though the network node's access rates might be higher, a mechanism is needed to smooth out the rate of traffic flow. If different traffic rates are applied to a network node, the traffic flow needs to be regulated. (Monitoring the traffic flow is called traffic policing.) Traffic shaping also prevents packet loss by preventing

the sudden increased usage of system bandwidth. The stochastic model of a traffic shaper consists of a system that converts any form of traffic to a deterministic one. Two of the most popular traffic-shaping algorithms are leaky bucket and token bucket.

Leaky-Bucket Traffic Shaping

This algorithm converts any turbulent incoming traffic into a smooth, regular stream of packets. Figure 12.3 shows how this algorithm works. A leaky-bucket interface is connected between a packet transmitter and the network. No matter at what rate packets enter the traffic shaper, the outflow is regulated at a constant rate, much like the flow of water from a leaky bucket. The implementation of a leaky-bucket algorithm is not difficult.

Figure 6.3. The leaky-bucket traffic-shaping algorithm



At the heart of this scheme is a finite queue. When a packet arrives, the interface decides whether that packet should be queued or discarded, depending on the capacity of the buffer. The number of packets that leave the interface depends on

the protocol. The packet-departure rate expresses the specified behavior of traffic and makes the incoming bursts conform to this behavior. Incoming packets are discarded once the bucket becomes full.

This method directly restricts the maximum size of a burst coming into the system. Packets are transmitted as either fixed-size packets or variable-size packets. In the fixed-size packet environment, a packet is transmitted at each clock tick. In the variable-size packet environment, a fixed-sized block of a packet is transmitted. Thus, this algorithm is used for networks with variable-length packets and also equal-sized packet protocols, such as ATM.

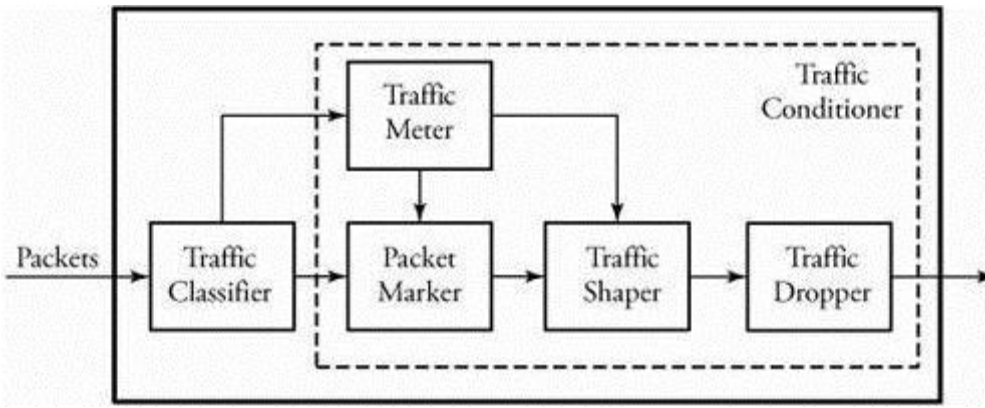
The leaky-bucket scheme is modeled by two main buffers, as shown in Figure 6.4. One buffer forms a queue of incoming packets, and the other one receives authorizations. The leaky-bucket traffic-shaper algorithm is summarized as follows.

6.3 Differentiated Services QoS:

The differentiated services (DS), or DiffServ, approach provides a simpler and more scalable QoS. DS minimizes the amount of storage needed in a router by processing traffic flows in an aggregate manner, moving all the complex procedures from the core to the edge of the network. A traffic conditioner is one of the main features of a DiffServ node to protect the DiffServ domain. As shown in Figure 6.5, the traffic conditioner includes four major components: meter, marker, shaper, and dropper. A meter measures the traffic to make sure that packets do not exceed their traffic profiles. A marker marks or unmarks packets in order to keep track of their situations in the DS node. A shaper delays

any packet that is not compliant with the traffic profile. Finally, a dropper discards any packet that violates its traffic profile.

Figure 6.5 Overview of DiffServ operation



When users request a certain type of service, a service provider must satisfy the user's need. In order to allocate and control the available bandwidth within the DS domain, a bandwidth broker is needed to manage the traffic. The bandwidth broker operates in its own DS domain and maintains contact with other bandwidth brokers at neighboring domains. This is a way to confirm that a packet's requested service is valid throughout all domains within the routing path of the packet.

In order to process traffic flows in an aggregate manner, a packet must go through a service-level agreement (SLA) that includes a traffic-conditioning agreement (TCA). An SLA indicates the type of forwarding service, and a TCA presents all the detailed parameters that a customer receives. An SLA can be

either static or dynamic. A static SLA is a long-term agreement, and a dynamic SLA uses the bandwidth broker that allows users to make changes more frequently. A user preferring packet-dependent quality-of-service can simply mark different values in the type of service (ToS) field at either the host or its access router. Routers in the DS model then detect the value in the DS field in per hop behaviors (PHBs). The quality-of-service can then be performed in accordance with the PHB.

In order to establish a traffic-policing scheme, a service provider uses a traffic classifier and a traffic conditioner at the domain's edge router when packets enter the service provider's network. The traffic classifier routes packets to specific outputs, based on the values found inside multiple fields of a packet header. The traffic conditioner detects and responds if any packet has violated any of the rules specified in the TCA. The DiffServ field value is set at the network boundaries. A DiffServ router uses the traffic classifier to select packets and then uses buffer management and a scheduling mechanism to deliver the specific PHB. The 8-bit DiffServ field is intended to replace the IPv4 ToS field and the IPv6 traffic class field. Six bits are used as a differentiated services code point (DSCP) to specify its PHB. The last 2 bits are unused and are ignored by the DS node.

6.3.1. Per-Hop Behavior (PHB)

We define two PHBs: expedited forwarding and assured forwarding. As for DiffServ domains, the expedited forwarding PHB provides low-loss, low-

latency, low-jitter, ensured-bandwidth, and end-to-end services. Low latency and ensured bandwidth can be provided with a few configurations on the DiffServ node. Both the aggregate arrival rate for expedited-forwarding PHB packets and the aggregate arrival rate should be less than the aggregate minimum departure rate. Several types of queue-scheduling mechanisms may be used to implement expedited-forwarding PHB. Ensured-forwarding PHHB delivers packets with high assurance and high throughput, as long as the aggregate traffic does not exceed TCA. However, users are allowed to violate TCA, but the traffic beyond TCA is not given high assurance. Unlike the expedited forwarding PHB, the ensured-forwarding PHB does not provide low-latency and low-jitter application. The ensured forwarding PHB group can be classified into three service types: good, average, and poor. Three possible drop-precedence values are then assigned to packets within each class, determining the priority of the corresponding packet.

6.4 Virtual Private Networks:

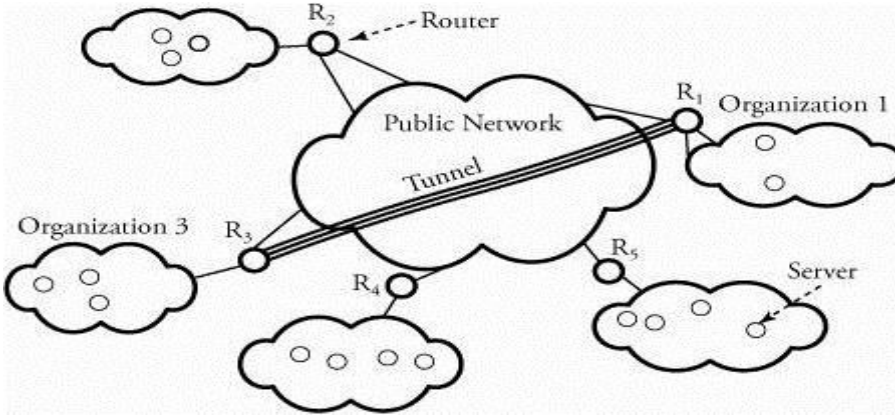
A virtual private network (VPN) is a data network having connections that make use of public networking facilities. The (VPN) part of public network is set up "virtually" by a private-sector entity to provide public networking services to small entities. With the globalization of businesses, many companies have facilities across the world and use VPNs to maintain fast, secure, and reliable communications across their branches.

VPNs are deployed with privacy through the use of a tunneling protocol and security procedures. Figure 16.1 shows two organizations, 1 and 3, connected

through their corresponding routers, forming a tunnel in the public network, such as the Internet. Such a structure gives both private organizations the same capabilities they have on their own networks but at much lower cost. They can do this by using the shared public infrastructure. Creating a VPN benefits an organization benefits by providing

- Extended geographical communication
- Reduced operational cost
- Enhanced organizational management
- Enhanced network management with simplified local area networks
- Improved productivity and globalization

Figure 6.6. Two organizations connected through a tunnel using public facilities



But since each user has no control over wires and routers, one of the issues with the Internet is still its lack of security, especially when a tunnel is exposed to the public. Thus, VPNs remain susceptible to security issues when they try to

connect between two private networks using a public resource. The challenge in making a practical VPN, therefore, is finding the best security for it. Before discussing VPN security, we focus on types of VPNs. There are two types of VPNs each determined by its method of tunneling, remote-access and site-to-site. We will explain these two approaches in the next two sections.

6.4.1. Remote-Access VPN

Remote-access VPN is a user-to-LAN connection that an organization uses to connect its users to a private network from various remote locations. Large remote-access VPNs are normally outsourced to an Internet service provider to set up a network-access server. Other users, working off campus, can then reach the network-access server and use the VPN software to access the corporate network. Remote-access VPNs allow encrypted connections between an organization's private network and remote users through a third-party service provider. Tunneling in a remote-access VPN uses mainly the Point-to-Point Protocol (PPP). PPP is the carrier for other Internet protocols when communicating over the network between a host computer and a remote point. Besides IPsec, other types of protocols associated with PPP are L2F, PPTP, and L2TP. The Layer 2 Forwarding (L2F) protocol uses the authentication scheme supported by PPP. The Point-to-Point Tunneling Protocol (PPTP) supports 40-bit and 128-bit encryption and uses the authentication scheme supported by PPP. The Layer 2 Tunneling Protocol (L2TP) combines features of both PPTP and L2F.

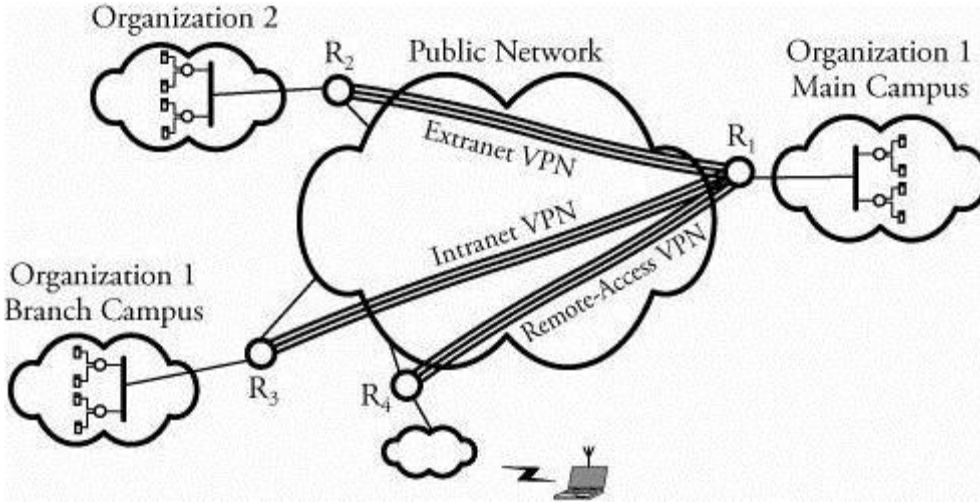
6.4.2. Site-to-Site VPN

By using effective security techniques, an organization can connect multiple fixed sites over a public network. Site-to-site VPNs can be classified as either intranets or extranets.

- Intranet VPNs connect an organization's remote-site LANs into a single private network.
- Extranet VPNs allow two organizations to work in a shared environment through a tunnel built to connect their LANs.

Figure 6.4 shows the three types VPNs discussed so far. Organization 1's main campus and branch campus are connected through an intranet VPN tunnel. The main campus can also be connected to organization 2 through an extranet VPN tunnel. The employees of organization 1 can also access their corporation through a remote-access VPN. Each remote-access member must communicate in a secure medium. The main benefit of using a VPN is scalability with a reasonable cost. However, the physical and virtual distances of two communicating organizations have a great impact on the overall cost of building a VPN.

Figure 6.4. Three types of VPNs to and from a headquarter organization



In a site-to-site VPN, generic routing encapsulation (GRE) is normally the encapsulating protocol. GRE provides the framework for the encapsulation over an IP-based protocol. IPsec in tunnel mode is sometimes used as the encapsulating protocol. IPsec works well on both remote-access and site-to-site VPNs but must be supported at both tunnel interfaces. The Layer 2 Tunneling Protocol (L2TP) can be used in site-to-site VPNs. L2TP fully supports IPsec regulations and can be used as a tunneling protocol for remote-access VPNs.

6.4.3. Tunneling and Point-to-Point Protocol (PPP)

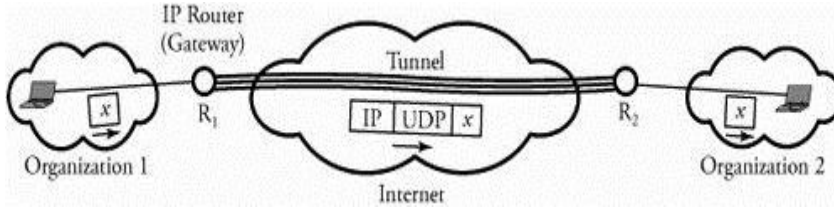
A tunnel is a connection that forms a virtual network on top of a physical network. In computer networking, a tunnel resembles a telephone line in a public switched telephone network. VPNs typically rely on tunneling to create a private network that reaches across a public network. Tunneling is a process of encapsulating packets and sending them over the public network. Employees who are located outside an organization's main building can use point-to-point

connections to create tunnels through the Internet. Since tunneling connections normally run over the Internet, they need to be secure. A tunnel is a relatively inexpensive connection, since it uses the Internet as its primary form of communication. Besides Internet protocols, tunneling requires two other types of protocols:

1. Carrier protocols, through which information travels over the public network
2. Encapsulating protocols, through which data is wrapped, encapsulated, and secured One of the amazing implications of VPNs is that packets that use a protocol not supported on the Internet, such as NetBeui, can be placed inside an IP packet and sent safely over the Internet. VPNs can put a packet that uses a nonroutable IP address inside a packet to extend a private network over the Internet.

Consider the two LANs of the two organizations shown in Figure 6.7. We want to connect these two LANs through the Internet by using tunnels. Assume that the two LANs, as organization 1 and organization 2, want to use their own customized networking protocols, denoted by x, using connectionless datagram IP services. The IP resources can be at the scale of the Internet. Therefore, x-type packets cannot run over the Internet directly. The IP gateway R_1 listens for x-type packets on organization 1, encapsulates x-type packets in the transport-layer UDP datagrams, and transmits them over the Internet to R_2. When R_2 receives the encapsulates x packets, it decapsulates and feeds them into organization 2. This connectionin fact, a tunnel made through the Internetresembles a direct physical link between the two LANs.

Figure 6.7. A customized protocol packet tunneling through the Internet



6.5 Multiprotocol Label Switching (MPLS):

Multiprotocol label switching (MPLS) improves the overall performance and delay characteristics of the Internet. MPLS transmission is a special case of tunneling and is an efficient routing mechanism. Its connection-oriented forwarding mechanism, together with layer 2 label-based lookups, enables traffic engineering to implement peer-to-peer VPNs effectively.

MPLS adds some traditional layer 2 capabilities and services, such as traffic engineering, to the IP layer. The separation of the MPLS control and forwarding components has led to multilayer, multiprotocol interoperability between layer 2 and layer 3 protocols. MPLS uses a small label or stack of labels appended to packets and typically makes efficient routing decisions. Another benefit is flexibility in merging IP-based networks with fast-switching capabilities. This technology adds new capabilities to IP-based networks:

- Connection-oriented QoS support
- Traffic engineering
- VPN support
- Multiprotocol support

Traditional IP routing has several limitations, ranging from scalability issues to

poor support for traffic engineering. The IP backbone also presents a poor integration with layer 2 existing in large service provider networks. For example, a VPN must use a service provider's IP network and build a private network and run its own traffic shielded from prying eyes. In this case, VPN membership may not be well engineered in ordinary IP networks and can therefore result in an inefficient establishment of tunnels.

MPLS network architectures also support other applications, such as IP multicast routing and QoS extensions. The power of MPLS lies in the number of applications made possible with simple label switching, ranging from traffic engineering to peer-to-peer VPNs. One of the major advantages of MPLS is integration of the routing and switching layers. The development of the label-switched protocol running over all the existing layer 2 and layer 3 architectures is a major networking development.

6.5.1. MPLS Operation

MPLS is based on the assignment of labels to packets. Assigning labels to each packet makes a label-swapping scheme perform its routing process much more efficiently. An MPLS network consists of nodes called label switch routers (LSR). An LSR switches labeled packets according to particular switching tables. An LSR has two distinct functional components: a control component and a forwarding component. The control component uses routing protocols, such as OSPF and the border gateway protocol (BGP). The control component also facilitates the exchange of information with other LSRs to build and maintain the forwarding table.

A label is a header used by an LSR to forward packets. The header format

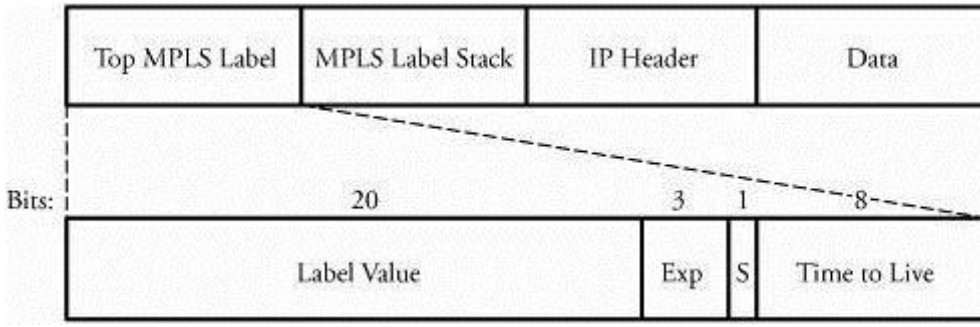
depends on the network characteristics. LSRs read only labels and do not engage in the network-layer packet headers. One key to the scalability of MPLS is that labels have only local significance between two devices that communicate. When a packet arrives, the forwarding component uses the label of the packet as an index to search the forwarding table for a match. The forwarding component then directs the packet from the input interface to the output interface through the switching fabric.

MPSL Packet Format

MPLS uses label stacking to become capable of multilevel hierarchical routing. A label enables the network to perform faster by using smaller forwarding tables, a property that ensures a convenient scalability of the network. Figure 6.6 shows the MPLS header encapsulation for an IP packet. An MPLS label is a 32-bit field consisting of several fields as follows.

- Label value is a 20-bit field label and is significant only locally.
- Exp is a 3-bit field reserved for future experimental use.
- S is set to 1 for the oldest entry in the stack and to 0 for all other entries.
- Time to live is an 8-bit field used to encode a hop-count value to prevent packets from looping forever in the network.

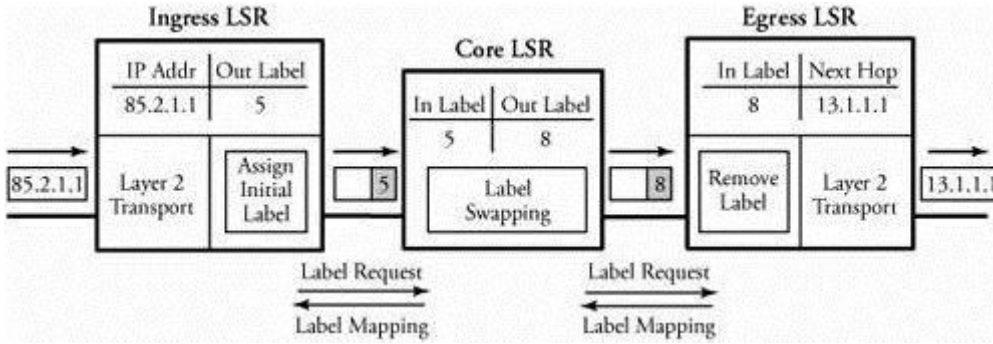
Figure 6.6. MPLS header encapsulation for an IP packet



6.5.2. Routing in MPLS Domains

Figure 6.7 shows the label-switching paradigm in an MPLS network. An ingress LSR is an edge device that performs the initial packet processing and classification and applies the first label. An ingress LSR creates a new label. A core LSR swaps the incoming label with a corresponding next-hop label found from a forwarding table. At the other end of the network, another edge router, the egress LSR, is an outbound edge router and pops the label from the packet. It should be noted that multiple labels may be attached to a packet, forming a stack of labels. Label stacking enables multilevel hierarchical routing. For example, BGP labels are used for higher-level hierarchical packet forwarding from one BGP speaker to the other, whereas Interior Gateway Protocol (IGP) labels are used for packet forwarding within an autonomous system. Only the label at the top of the stack determines the forwarding decision.

Figure 6.5. Multiple layer 2 switching example in MPLS



Once an IP packet enters an MPLS domain, the ingress LSR processes its header information and maps that packet to a forward equivalence class (FEC). At this point, a label switch path (LSP) through the network must be defined, and the QoS parameters along that path must be established. The QoS parameters define how many resources are to be used for the path and what queueing and discarding policy are to be used. For these functions, two protocols are used to exchange necessary information among routers: An intradomain routing protocol, such as OSPF, is used to exchange routing information, and the Label Distribution Protocol (LDP) assigns labels. At the end of the process, the router appends an appropriate label for FEC purposes and forwards the packet through.

Packet forwarding at the core LSR is based on a label-swapping mechanism. Once it receives a labeled packet, the core LSR reads the label as an index to search in the incoming label map table for the corresponding next-hop label. The label in the MPLS header is swapped with the out-label and sent on the next hop. This method of packet forwarding simplifies the routing process by replacing the longest-prefix match of IP routing with simple short-label exact-match forwarding. The real benefit of this method is that instead of processing IP

headers for forwarding packets, routers process a short label. Once a packet arrives at the egress LSR, its MPLS header is decapsulated, and the stripped packet is routed to its destination.

In summary, an MPLS domain has three label manipulation instructions: An ingress LSR creates a new label and pushes it to the label stack of a packet, a core LSR swaps the incoming label with a corresponding next-hop label found from the forwarding table, and an egress LSR (outbound edge router) pops a label from the label stack. Only the label at the top of the stack determines the forwarding decision. The egress LSR strips the label, reads the IP packet header, and forwards the packet to its final destination.

6.5.3. Tunneling and Use of FEC

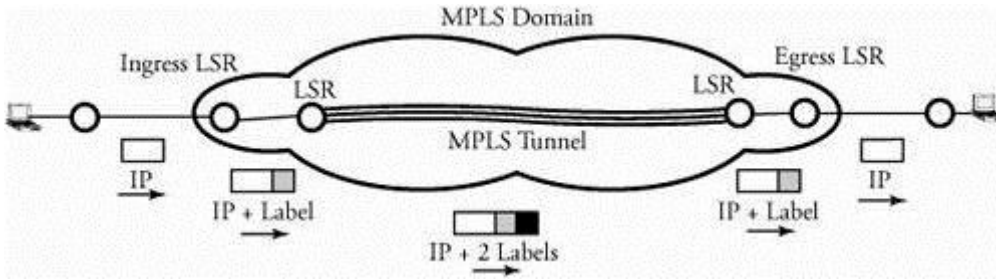
In an MPLS operation, any traffic is grouped into FECs. FEC implies that a group of IP packets are forwarded in the same mannerfor example, over the same path or with the same forwarding treatment. A packet can be mapped to a particular FEC, based on the following criteria:

- Source and/or destination IP address or IP network addresses
- TCP/UDP port numbers
- Class of service
- Applications

As mentioned earlier, labels have only local significance. This fact removes a considerable amount of the network-management burden. An MPLS packet may carry as many labels as required by a network sender. The process of labeled packets can always be performed based on the top label. The feature of label stack allows the aggregation of LSPs into a single LSP for a portion of the route,

creating an MPLS tunnel. Figure 6.8 shows an IP packet moving through an MPLS domain. When the labeled packet reaches the ingress LSR, each incoming IP packet is analyzed and classified into different FECs. This traffic-classification scheme provides the capability to partition the traffic for service differentiation.

Figure 6.8. An IP packet labeled in an MPLS domain and tunneled to reach the other end of the domain



Route selection can be done either hop by hop or by explicit routing. With hop-byhop routing, each LSR can independently choose the next hop for each FEC. Hop-byhop routing does not support traffic engineering, owing to limited available resources. Explicit routing can provide all the benefits of traffic engineering. With explicit routing, a single LSR determines the LSP for a given FEC. For explicit routing, LSRs in the LSP are identified, whereas in an explicit routing, only some of the LSRs in an LSP are specified.

With the introduction of constraint-based routing, FEC can segregate the traffic into different levels of QoS, each with different service constraints, to support a variety of services, such as latency-based voice traffic and security-based VPN. At the beginning of the tunnel, an LSR assigns the same label to packets from a

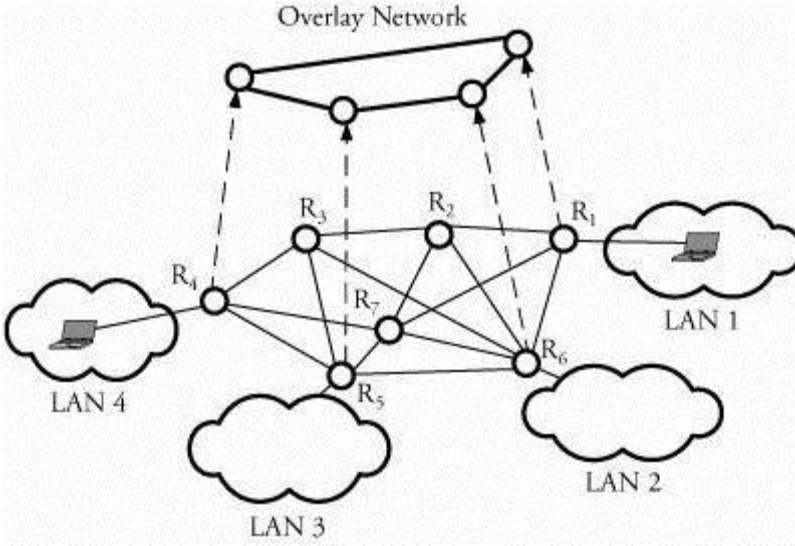
number of LSPs by pushing the label onto each packet's stack. At the other side of the tunnel, another LSR pops the top element from the label stack, revealing the inner label.

6.6 Overlay Networks:

An overlay network is an application-specific computer network built on top of another network. In other words, an overlay network creates a virtual topology on top of the physical topology. This type of network is created to protect the existing network structure from new protocols whose testing phases require Internet use. Such networks protect packets under test while isolating them from the main networking infrastructure in a test bed.

Figure 6.11 shows an overlay network configured over a wide area network. Nodes in an overlay network can be thought of as being connected by logical links. In Figure 6.11, for example, routers R_4, R_5, R_6, and R_1 are participating in creating an overlay network where the interconnection links are realized as overlay logical links. Such a logical link corresponds to a path in the underlying network. An obvious example of these networks is the peer-to-peer network, which runs on top of the Internet. Overlay networks have no control over how packets are routed in the underlying network between a pair of overlay source/destination nodes. However, these networks can control a sequence of overlay nodes through a message-passing function before reaching the destination.

Figure 6.11. An overlay network for connections between two LANs associated with routers R_1 and R_4



For various reasons, an overlay network might be needed in a communication system. An overlay network permits routing messages to destinations when the IP address is not known in advance. Sometimes, an overlay network is proposed as a method to improve Internet routing as implemented in order to achieve higher-quality streaming media. Sometimes, for the implementation of such techniques as DiffServ and IP multicast, modification of all routers in the network is required. In such cases, an overlay network can be deployed on end hosts running the overlay protocol software, without cooperation from Internet service providers.

Overlay networks are self-organized. When a node fails, the overlay network algorithm should provide solutions that let the network recover and recreate an appropriate network structure. Another fundamental difference between an overlay network and an unstructured network is that overlays' look-up routing

information is on the basis of identifiers derived from the content of moving frames.

6.6.1. Peer-to-Peer (P2P) Connection

As an overlay network resembles a system consisting of various applications running on a single operating system, it could also resemble a set of tunnels that interconnect resources and users. The interconnects are carried out by peer-to-peer (P2P) protocols.

Let δ be the time required to establish a connection and t_f be the time to finish the service as soon as the connection establishes. Assuming that the requests arrive at random to a peer node, the service time s is

Equation 16.1

$$s = \begin{cases} t_f & \text{if the peer is connected} \\ t_f + \delta & \text{if the peer is not connected} \end{cases}$$

When a request arrives, the peer can be connected with probability p_c and not connected with probability 1 - p_c. Realizing that s is a continuous random variable (see Appendix C) and is discrete for its two cases, discussed in Equation (6.1), the expected value of average service time is derived by

Equation 16.2

$$E[S] = (1 - p_c)t_f + p_c(t_f + \delta) = t_f + p_c\delta.$$

Let $\rho = \frac{\lambda}{\mu}$ be the utilization of the peer node, where λ is the request arrival rate, and μ is the average service rate. Thus, a fraction ρ of any given time that either

"the peer uses for connecting' or "the connection is used for service' is expressed by

Equation 16.3

$$u_s = \rho \left(\frac{(1 - p_c)\delta)}{E[S]} \right).$$

Similarly, the fraction 1 - ρ that the same mentioned given time is idle can be derived by

Equation 16.4

$$u_i = (1 - \rho)p_c,$$

where the idle time occurs when the peer is either disconnected or connected but not using the connection. We can now derive an expression for the peer connection efficiency, u, as follows:

Equation 16.5

$$u = 1 - (u_s + u_i).$$

The connection efficiency of the peer can be used to determine the overall efficiency of the P2P connection.

UNIT - 7

Multimedia Networking: Overview of data compression, Digital voice and compression, JPEG, MPEG, Limits of compression with loss, Compression methods without loss, Overview of IP Telephony, VoIP signaling protocols, Real-Time Media Transport Protocols, Stream control Transmission Protocol (SCTP)

UNIT - 7

MULTIMEDIA NETWORKING:

7.1 Overview of Data Compression:

The benefits of data compression in high-speed networks are obvious. Following are those that are especially important for the compressed version of data.

- Less transmission power is required.
- Less communication bandwidth is required.
- System efficiency is increased.

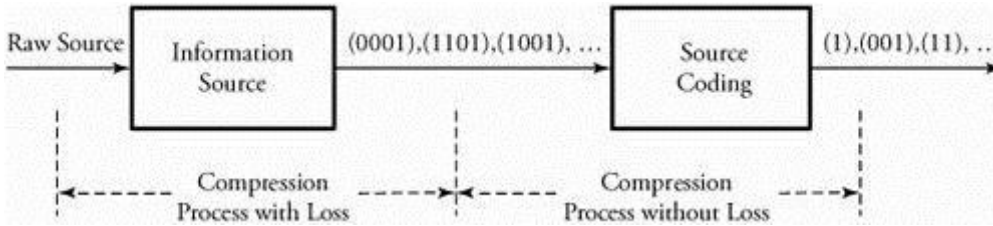
There are, however, certain trade-offs with data compression. For example, the encoding and decoding processes of data compression increase the cost, complexity, and delay of data transmission. Both of the two processes of data compression are required for producing multimedia networking information: compression with loss and compression without loss.

In the first category of data compression, some less valuable or almost similar data must be eliminated permanently. The most notable case of compression with loss is the process of signal sampling. In this category, for example, is voice sampling (Section 7.2). With data compression without data loss, the compressed data can be recovered and converted back to its original form when received. This method of compression is typically applied to digital bits after sampling.

Figure 7.1 shows the basic information process in high-speed communication systems. Any type of "source" data is converted to digital form in a long

information-source process. The outcome is the generation of digital words. Words are encoded in the source coding system to result in a compressed form of the data.

Figure 7.1. Overview of information process and compression in multimedia networks



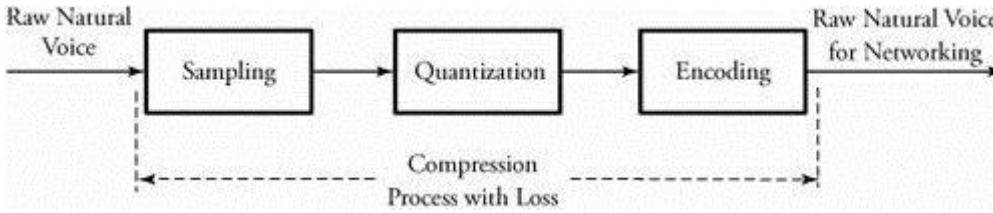
7.2 Digital Voice and Compression:

Our discussion starts with the voice as a simple real-time signal. We first review the fundamentals of voice digitization and sampling.

7.2.1. Signal Sampling

In the process of digitalizing a signal, analog signals first go through a sampling process, as shown in Figure 7.2. The sampling function is required in the process of converting an analog signal to digital bits. However, acquiring samples from an analog signal and eliminating the unsampled portions of the signal may result in some permanent loss of information. In other words, the sampling resembles an information-compression process with loss.

Figure 7.2. Overview of digital voice process



Sampling techniques are of several types:

- Pulse amplitude modulation (PAM), which translates sampled values to pulses with corresponding amplitudes
- Pulse width modulation (PWM), which translates sampled values to pulses with corresponding widths
- Pulse position modulation (PPM), which translates sampled values to identical pulses but with corresponding positions to sampling points

PAM is a practical and commonly used sampling method; PPM is the best modulation technique but is expensive. PWM is normally used in analog remote-control systems. The sampling rate in any of these schemes obeys the Nyquist theorem, according to which at least two samples on all components of the spectrum are needed in order to reconstruct a spectrum:

Equation 17.1

$$f_S \geq 2f_H,$$

where f_H is the highest-frequency component of a signal, and f_S is the sampling

rate.

7.2.2. Quantization and Distortion

Samples are real numbersdecimal-point values and integer valuesand, thus, up to infinite bits are required for transmission of a raw sample. The transmission of infinite bits occupies infinite bandwidth and is not practical for implementation. In practice, sampled values are rounded off to available quantized levels. However, rounding off the values loses data and generates distortion. A measure is needed to analyze this distortion. The distortion measure should show how far apart a signal denoted by x(t) is to its reproduced version, denoted by $\hat{x}(t)$. The distortion measure of a single source is the difference between source sample X_i and its corresponding quantized value $\hat{X}_i$, denoted by $d(X, \hat{X})$, and is widely known as squared-error distortion:

Equation 17.2

$$d(X, \hat{X}) = (x - \hat{x})^2.$$

Note that $\hat{X}_i$ is noninvertible, since lost information cannot be recovered. The distortion measure of n samples is based on the values of source samples obtained at the sampler output. As a result, the collection of n samples forms a random process:

Equation 17.3

$$\mathrm{X}_n = \{X_1, X_2, \cdots, X_n\}.$$

Similarly, the reconstructed signal at the receiver can be viewed as a random process:

Equation 17.4

$$\hat{\mathbf{X}}_n = \{\hat{X}_1, \hat{X}_2, \cdots, \hat{X}_n\}.$$

The distortion between these two sequences is the average between their components:

Equation 17.5

$$d(\mathbf{X}_n, \hat{\mathbf{X}}_n) = \frac{1}{n} \sum_{i=1}^{n} d(X_i, \hat{X}_i).$$

Note that $d(\mathbf{X}_n, \hat{\mathbf{X}}_n)$ itself is a random variable, since it takes on random numbers. Thus, the total distortion between the two sequences is defined as the expected value of $d(\mathbf{X}_n, \hat{\mathbf{X}}_n)$:

Equation 17.6

$$D = E[d(\mathbf{X}_n, \hat{\mathbf{X}}_n)] = E\left[\frac{1}{n} \sum_{i=1}^{n} d(X_i, \hat{X}_i)\right]$$
$$= \frac{1}{n} E[d(X_1, \hat{X}_1) + d(X_2, \hat{X}_2) + \cdots + d(X_n, \hat{X}_n)].$$

If all samples are expected to have approximately the same distortion denoted by

$d(X, \hat{X}), d(X_1, \hat{X}_1) = d(X_2, \hat{X}_2) = \cdots = d(X_n, \hat{X}_n) = d(X, \hat{X})$. By using squared-error distortion, we obtain the total distortion:

Equation 17.7

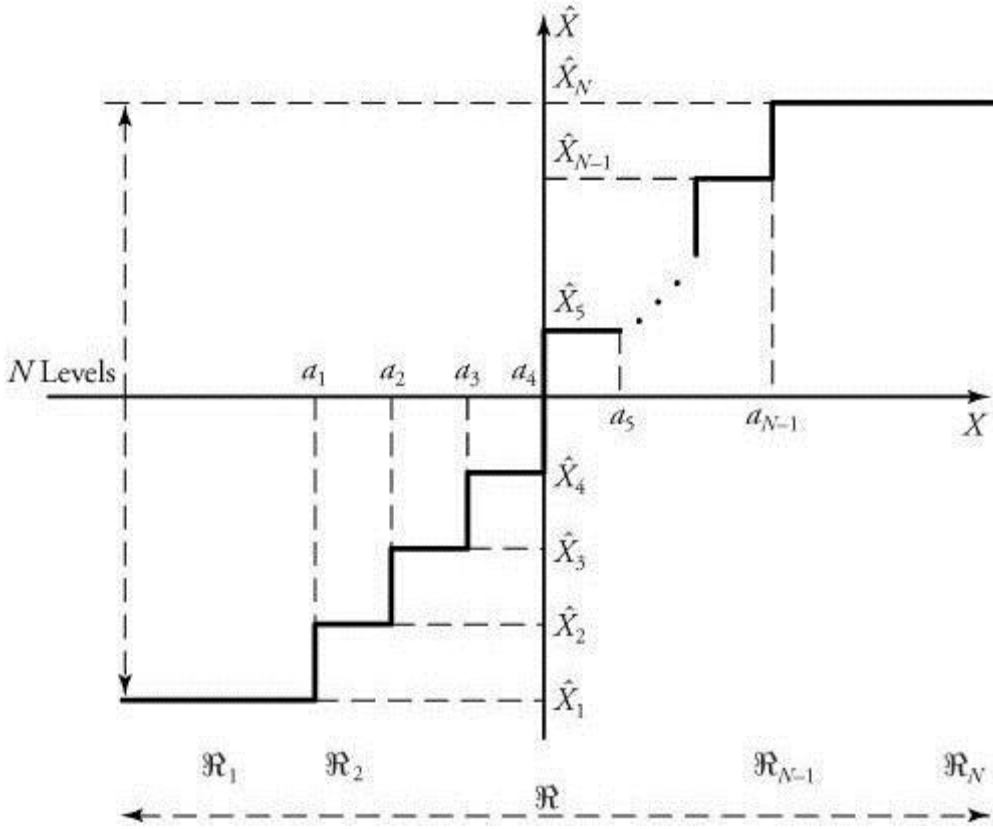
$$D = \frac{1}{n}\left(nE[d(X, \hat{X})]\right) = E[d(X, \hat{X})] = E[(X - \hat{X})^2].$$

Let R be the minimum number of bits required to reproduce a source and guarantee that the distortion be less than a certain distortion bound D_b. Clearly, if D decreases, R must increase. If X is represented by R bits, the total number of different values X_i takes is 2^R. Each single-source output is quantized into N levels. Each level 1, 2, ..., N is encoded into a binary sequence. Let $\Re$ be the set of real numbers $\Re_1$, ..., $\Re_k$, ..., $\Re_N$, and let $\hat{X}_k$ be the quantized value belonging to subset $\Re_k$. Note that $\hat{X}_k$ is a quantized version of X_k. Apparently, R = $\log_2$ N bits are required to encode N quantized levels. Figure 7.3 shows a model of N-level quantization: For the subsets $\Re_1$=[∞ a_1], $\Re_2$ =[a_1, a_2], ..., $\Re_8$ = [a_{N-1}, ∞], the quantized values are $\hat{X}_1$, $\hat{X}_2$,..., $\hat{X}_n$, respectively. We can use the definition of expected value to obtain D, as follows:

Equation 17.8

$$D = \int_{-\infty}^{+\infty} (x - \hat{x})^2 f_X(x) dx = \sum_{i=1}^{N} \int_{\Re_i} (x - \hat{x})^2 f_X(x) dx.$$

Figure 7.3. N-level quantization



Typically, a distortion bound denoted by D_b is defined by designers to ensure that $D_b \leq D$.

Example. Consider that each sample of a sampled source is a Gaussian random variable with a given probability distribution function $f_X(x) = 0.01e^{-\frac{1}{800}x^2}$. We want eight levels of quantization over the regions $\{a_1 = -60, a_2 = -40, ..., a_7 = 60\}$ and $\{\hat{x}_1 = -70, \hat{x}_2 = -50, \cdots, \hat{x}_8 = 70\}$. Assuming that the distortion bound for this signal is D_b = 7.2, find out how far the real distortion, D, is to D_b.

Solution. Since N = 8, the number of bits required per sample is R = $\log_2$ 8 = 3. Using this, D can be developed further:

$$D = \sum_{i=1}^{8} \int_{\Re_i} (X - \hat{X})^2 f_X(x) dx$$

$$= \int_{-\infty}^{a_1} (x - \hat{x})^2 (0.01e^{-\frac{1}{800}x^2}) dx + \sum_{i=2}^{7} \int_{a_{i-1}}^{a_7} (x - \hat{x})^2 (0.01e^{-\frac{1}{800}x^2}) dx$$

$$+ \int_{a_7}^{\infty} (x - \hat{x}_8)^2 (0.01e^{-\frac{1}{800}x^2}) dx = 16.64.$$

We note here that the total distortion as a result of eight-level quantization is

16.64, which is considerably greater than the given distortion bound of 7.2.

The conclusion in the example implies that the quantization chosen for that source may not be optimal. A possible reason may be inappropriate choices of (a_1, ..., a_7) and/or $(\hat{x}_1, \cdots, \hat{x}_8)$. This in turn means that R is not optimal.

Optimal Quantizers

Let Δ be the length of each region equal to $a_{i+1} - a_i$. Thus, regions can be restated as ($-\infty$, a_1) ... (a_{N-1}, $+\infty$). Clearly, the limits of the upper region can also be shown by ($a_1 + (N - 2)\Delta$, $+\infty$). Then, the total distortion can be rewritten as

Equation 17.9

$$D = \int_{-\infty}^{a_1} (x - \hat{x})^2 f_X(x)dx + \sum_{i=1}^{N-2} \int_{a_1+(i-1)\Delta}^{a_1+i\Delta} (x - \hat{x}_{i+1})^2 f_X(x)dx + \int_{a_1+(N-2)\Delta}^{\infty} (x - \hat{x}_N)^2 f_X(x)dx.$$

For D to be optimized, we must have $\frac{\partial D}{\partial a_1} = 0$, $\frac{\partial D}{\partial \Delta} = 0$ and also $\frac{\partial D}{\partial \hat{x}_1} = 0$, ..., $\frac{\partial D}{\partial x_N} = 0$.

The resu1t of solving the N + 2 equations can be summarized as follows. For N = even:

Equation 17.10

$$a_i = -a_{N-i} = -(\frac{N}{2} - i)\Delta \quad 1 \le i \le \frac{N}{2}$$

and

Equation 7.11

$$\hat{x}_i = \hat{x}_{N+1-i} = -(\frac{N}{2} - i + \frac{1}{2})\Delta.$$

For N = odd:

Equation 7.12

$$a_i = -a_{N-i} = -(\frac{N}{2} + i)\Delta \quad 1 \le i \le \frac{N+1}{2}$$

and

Equation 7.13

$$\hat{x}_i = -\hat{x}_{N+1-i} = -(\frac{N}{2} + i + \frac{1}{2})\Delta.$$

7.3 JPEG & MPEG Files:

Color images are based on the fact that any color can be represented to the human eye by using a particular combination of the base colors red, green, and blue (RGB). Computer monitor screens, digital camera images, or any other still color images are formed by varying the intensity of the three primary colors at pixel level, resulting in the creation of virtually any corresponding color from the real raw image. Each intensity created on any of the three pixels is represented by 8 bits, as shown in Figure 17.6. The intensities of each 3-unit pixel are adjusted, affecting the value of the 8-bit word to produce the desired color. Thus, each pixel can be represented by using 24 bits, allowing 2^{24} different colors. However, the human eye cannot distinguish all colors among the 2^{24} possible

colors. The number of pixels in a typical image varies with the image size.

GIF Files

JPEG is designed to work with full-color images up to 2^{24} colors. The graphics interchange format (GIF) is an image file format that reduces the number of colors to 256. This reduction in the number of possible colors is a trade-off between the quality of the image and the transmission bandwidth. GIF stores up to $2^8 = 256$ colors in a table and covers the range of colors in an image as closely as possible. Therefore, 8 bits are used to represent a single pixel. GIF uses a variation of Lempel-Ziv encoding (Section 17.6.3) for compression of an image. This technology is used for images whose color detailing is not important, such as cartoons and charts.

DCT Process

The discrete cosine transform (DCT) is a lossy compression process that begins by dividing a raw image into a series of standard N x N pixel blocks. For now, consider a monochrome block of pixels. With a standard size of N = 8, N is the number of pixels per row or column of each block. Let x, y be the position of a particular pixel in a block where $0 \leq x \leq N - 1$ and $0 \leq y \leq N - 1$. Hence, a gray scale of a given pixel x, y can get an integer value in {0, 1, 2, ..., 255}. For an N x N pixel block, the DCT process is summarized in two steps

1. Form a P[x][y] matrix to represent the collection of light-intensity values taken from various points of a real raw image.

2. Convert the values of P[x][y]matrix to matrix with normalized and reduced values denoted by T[i][j] obtained as follows.

The objective of matrix T[i][j] is to create as many 0s as possible instead of small numbers in the P[x][y] matrix in order to reduce the overall bandwidth required to transmit the image. Similarly, matrix T[i][j] is a two-dimensional array with N rows and N columns, where 0 ≤i ≤N - 1 and 0 ≤j ≤N - 1. The elements of T[i][j] are known as spatial frequencies and are obtained from Equation 17.14

$$T[i][j] = \frac{2}{N} C(i)C(j) \sum_{x=0}^{N-1} \sum_{y=0}^{N-1} \cos\left(\frac{\pi i(2x+1)}{2N}\right) \cos\left(\frac{\pi i(2y+1)}{2N}\right),$$

where

$$C(i) = \begin{cases} \frac{1}{\sqrt{2}} & \text{for } i = 0 \\ 0 & \text{otherwise} \end{cases}$$

$$C(j) = \begin{cases} \frac{1}{\sqrt{2}} & \text{for } j = 0 \\ 0 & \text{otherwise} \end{cases}$$

Moving Images and MPEG Compression

A motion image, or video is a rapid display of still images. Moving from one image to another must be fast enough to fool the human eye. There are different standards on the number of still images comprising a video clip. One common standard produces a motion image by displaying still images at a rate of 30 frames per second. The common standard that defines the video compression is the Moving Pictures Expert Group (MPEG), which has several branch standards:

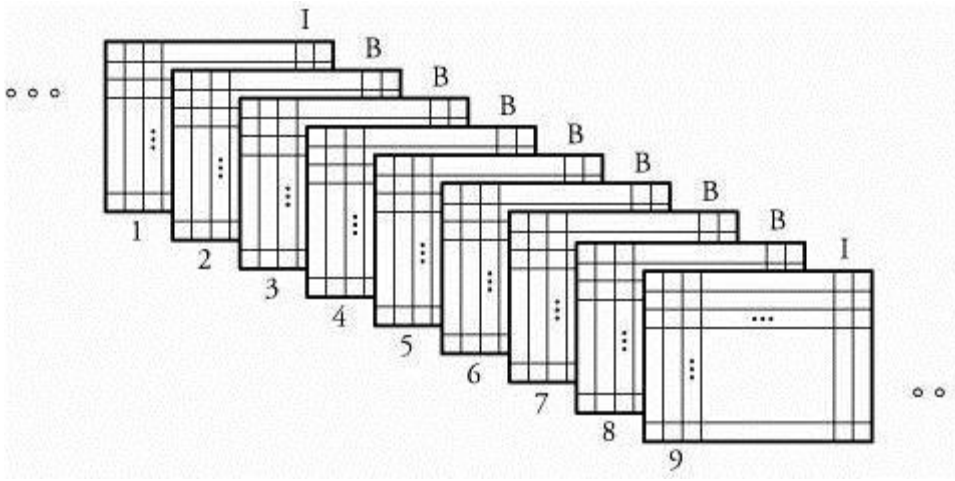
- MPEG-1, primarily for video on CD-ROM
- MPEG-2, for multimedia entertainment and high-definition television (HDTV) and the satellite broadcasting industry
- MPEG-4, for object-oriented video compression and videoconferencing over low-bandwidth channels
- MPEG-7, for a broad range of demands requiring large bandwidths providing multimedia tools
- MPEG-21 for interaction among the various MPEG groups

Logically, using JPEG compression for each still picture does not provide sufficient compression for video as it occupies a large bandwidth. MPEG deploys additional compression. Normally, the difference between two consecutive frames is small. With MPEG, a base frame is sent first, and successive frames are encoded by computing the differences. The receiver can reconstruct frames based on the first base frame and the submitted differences. However, frames of a completely new scene in a video may not be compressed this way, as the difference between the two scenes is substantial. Depending on the relative position of a frame in a sequence, it can be compressed through one of the following types of frames:

- Interimage (I) frames. An I frame is treated as a JPEG still image and compressed using DCT.
- Predictive (P) frames. These frames are produced by computing differences between a current and a previous I or P frame.
- Bidirectional (B) frames. A B frame is similar to a P frame, but the P frame considers differences between a previous, current, and future frames.

Figure 7.9 illustrates a typical grouping of frames, with I, P, and B frames forming a sequence. In any frame sequence, I frames appear periodically as the base of the scene. Normally, there is a P frame between each two groups of B frames.

Figure 7.9. Snapshot of moving frames for MPEG compression



7.4.1. MP3 and Streaming Audio

Section 7.2 explained how an audible sound or a human voice ranging between 20 Hz and 20 KHz can be converted into digital bits and eventually into packets for networking. A signal is sampled, quantized, and encoded, a process called PCM. A variety of methods of compressing such encoded products at the output of the PCM are available. However, Huffman compression of the processed signal may not be sufficient for transmission over IP networks.

The MPEG-1 layer 3 (MP3) technology compresses audio for networking and

producing CD-quality sound. The sampling part of PCM is performed at a rate of 44.1 KHz to cover the maximum of 20 KHz of audible signals. Using the commonly used 16-bit encoding for each sample, the maximum total bits required for audio is 16 x 44.1 = 700 kilobits and 1.4 megabits for two channels if the sound is processed in a stereo fashion. For example a 60-minute CD (3,600 seconds) requires about 1.4 x 3,600 = 5,040 megabits, or 630 megabytes. This amount may be acceptable for recording on a CD but is considered extremely large for networking, and thus a carefully designed compression technique is needed.

MP3 combines the advantages of MPEG with "three" layers of audio compressions. MP3 removes from a piece of sound all portions that an average ear may not be able to hear, such as weak background sounds. On any audio streaming, MP3 specifies what humans are not able to hear, removes those components, and digitizes the remaining. By filtering some part of an audio signal, the quality of compressed MP3 is obviously degraded to lower than the original one. Nonetheless, the compression achievement of this technology is remarkable.

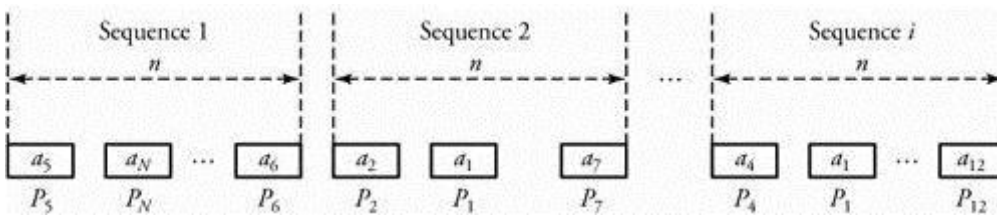
7.4 Limits of Compression with Loss:

Hartely, Nyquist, and Shannon are the founders of information theory, which has resulted in the mathematical modeling of information sources. Consider a communication system in which a source signal is processed to produce sequences of n words, as shown in Figure 7.10. These sequences of digital bits at the output of the information source can be compressed in the

source encoder unit to save the transmission link bandwidth. An information source can be modeled by a random process $X_n = (X_1, ..., X_n)$, where X_i is a random variable taking on values from a set of values as $\{a_1, ...,a_N\}$, called alphabet. We use this model in our analysis to show the information process in high-speed networks.

Figure 7.10. A model of data sequences

[View full size image]



7.5.1. Basics of Information Theory

The challenge of data compression is to find the output that conveys the most information. Consider a single source with random variable X, choosing values in $\{a_1, ... a_N\}$.

If a_i is the most likely output and a_j is the least likely output, clearly, a_j conveys the most information and a_i conveys the least information. This observation can be rephrased as an important conclusion: The measure of information for an output is a decreasing and continuous function of the probability of source output. To formulate this statement, let P_{k1} and P_{k2} be the probabilities of an information source's outputs a_{k1} and a_{k2}, respectively. Let $I(P_{k1})$ and $I(P_{k2})$ be the information content of a_{k1} and a_{k2}, respectively. The following four facts apply.

1. As discussed, $I(P_k)$ depends on P_k.
2. $I(P_k)$ = a continuous function of P_k.
3. $I(P_k)$ = a decreasing function of P_k.
4. $P_k = P_{k1}.P_{k2}$ (probability of two outputs happen in the same time).
5. $I(P_k) = I(P_{k1}) + I(P_{k2})$ (sum of two pieces of information).

These facts lead to an important conclusion that can relate the probability of a certain data to its information content:

Equation 17.16

$$I(P_k) = -\log_2 P_k = \log_2 \left(\frac{1}{P_k}\right).$$

The log function has a base 2, an indication of incorporating the binary concept of digital data.

7.4.2. Entropy of Information

In general, entropy is a measure of uncertainty. Consider an information source producing random numbers, X, from a possible collection of $\{a_1, ..., a_N\}$ with corresponding probabilities of $\{P_1,...,P_N\}$ and information content of $\{I(P_1),..., I(P_N)\}$, respectively. In particular, the entropy, H(x), is defined as the average information content of a source:

Equation 17.17

$$H_X(x) = \sum_{k=1}^{N} P_k I(P_k)$$
$$= \sum_{k=1}^{N} -P_k \log_2 P_k = \sum_{k=1}^{N} P_k \log_2 \left(\frac{1}{P_k}\right).$$

Example. A source with bandwidth 8 KHz is sampled at the Nyquist rate. If the result is modeled using any value from {-2,-1,0,1,2} and corresponding probabilities {0.05, 0.05, 0.08, 0.30, 0.52}, find the entropy.

Solution.

$$H_X(x) = -\sum_{k=1}^{5} P_k \log_2 P_k = 0.522 \text{ bits/sample.}$$

The information rate in samples/sec = 8,000 x 2 = 16,000, and the rate of information produced by the source = 16,000 x 0.522 = 8,352 bits.

Joint Entropy

The joint entropy of two discrete random variables X and Y is defined by

Equation 17.18

$$H_{X,Y}(x, y) = -\sum_{x,y} P_{X,Y}(x, y) \log_2 P_{X,Y}(x, y),$$

where $P_{X,Y}$ (x, y) = Prob[X = x and the same time Y = y] and is called the joint probability mass function of two random variables. In general, for a random process $X_n = (X_1,..., X_n)$ with n random variables:

Equation 17.19

$$H_{Xn}(\mathbf{x_n}) = -\sum_{X_1,\ldots,X_n} P_{X_1,\ldots,X_n}(x_1,\ldots,x_n)\log_2 P_{X_1,\ldots,X_n}(x_1,\ldots,x_n),$$

where $P_{X1, ..., Xn}$ (x_1, ..., x_n) is the joint probability mass function (J-PMF) of the random process X_n. For more information on J-PMF,

7.4.3. Shannon's Coding Theorem

This theorem limits the rate of data compression. Consider again Figure 7.10, which shows a discrete source modeled by a random process that generates sequences of length n using values in set {a_1, ..., a_N} with probabilities {P_1, ..., P_N}, respectively. If n is large enough, the number of times a value a_i is repeated in a given sequence = nP_i, and the number of values in a typical sequence is therefore $n(P_1 + ... + P_N)$.

We define the typical sequence as one in which any value a_i is repeated nP_i times. Accordingly, the probability that a_i is repeated nP_i times is obviously $P_i P_i \cdots P_i = P_i^{nP_i}$, resulting in a more general statement: The probability of a typical sequence is the probability [(a_1 is repeated np_1)] x the probability [(a_2 is repeated np_2] x .... This can be shown by $P_1^{np_1} P_2^{np_2} \cdots P_N^{np_N}$, or

Equation 17.20

$$\text{Prob(Typical Sequence)} = \prod_{i=1}^{N} P_i^{nP_i}.$$

Knowing $P_i^{nP_i} = 2^{nP_i \log_2 P_i}$, we can obtain the probability of a typical

sequence P_t as follow:

Equation 17.21

$$P_t = \prod_{i=1}^{N} 2^{nP_i \log_2 P_i}$$
$$= 2^{(nP_1 \log_2 P_1 + \cdots nP_N \log_2 P_N)}$$
$$= 2^{n(P_1 \log_2 P_1 + \cdots P_N \log_2 P_N)}$$
$$= 2^{n(\sum_{i=1}^{N} P_i \log_2 P_i)}.$$

This last expression results in the well-known Shannon's theorem, which expresses the probability that a typical sequence of length n with entropy $H_X(x)$ is equal to

Equation 17.22

$$P_t = 2^{-nH_X(x)}.$$

Example. Assume that a sequence size of 200 of an information source chooses values from the set $\{a_1, ..., a_5\}$ with corresponding probabilities {0.05, 0.05, 0.08, 0.30, 0.52}. Find the probability of a typical sequence.

Solution. In the previous example, we calculated the

entropy to be $H_X(x) = 0.522$ for the same situation. With n = 200, N = 5, the probability of a typical sequence is the probability of a sequence in which a_1, a_2, a_3, a_4, and a_5 are repeated, respectively, 200 x 0.05 = 10 times, 10 times, 16 times, 60 times, and 104 times. Thus, the probability of a typical sequence is $P_t = 2^{-nH_X(x)} = 2^{-200(0.52)}$.

The fundamental Shannon's theorem leads to an important conclusion. As the probability of a typical sequence is $2^{-nH_X(x)}$ and the sum of probabilities of all typical sequences is 1, the number of typical sequences is obtained by

$$= \frac{1}{2^{-nH_X(x)}} = 2^{nH(x)}.$$

Knowing that the total number of all sequences, including typical and nontypical ones, is N^n, it is sufficient, in all practical cases when n is large enough, to transmit only the set of typical sequences rather than the set of all sequences. This is the essence of data compression: The total number of bits required to represent $2^{nH}X(x)$ sequences is $nH_X(x)$ bits, and the average number of bits for each source = $H_X(x)$.

Example. Following the previous example, in which the sequence size for an information source is 200, find the ratio of the number of typical sequences to the number of all types of sequences.

Solution. We had n = 200 and N = 5; thus, the number of typical sequences is $2^{nH}(x)=2^{200}x0.522$, and the total number of all sequences is 5^{200}. This ratio is almost zero, which may cause a significant loss of data if it is compressed, based on Shannon's theorem. It is worth mentioning that the number of bits required to represent 2^{nH} $X^{(x)}$ sequences is $nH_X(x) = 104$ bits.

7.4.4. Compression Ratio and Code Efficiency

Let $\bar{R}$ be the average length of codes, ℓ_i be the length of code word i, and P_i be the probability of code word i:

Equation 17.23

$$\bar{R}_i = \sum_{i=0}^{N} P_i \ell_i .$$

A compression ratio is defined as

Equation 17.24

$$C_r = \frac{\bar{R}}{\bar{R}_x},$$

where $\bar{R}_{x_x}$ is the length of a source output before coding. It can also be shown that

Equation 7.25

$$H_X(x) \leq \bar{R} < H_X(x) + 1.$$

Code efficiency is a measure for understanding how close code lengths are to the corresponding decoded data and is defined by

Equation 7.26

$$\eta_{code} = \frac{H_X(x)}{\bar{R}}.$$

When data is compressed, part of the data may need to be removed in the process.

7.5 Compression Methods Without Loss:

Some types of data, including text, image, and video, might contain redundant or repeated elements. If so, those elements can be eliminated and some sort of codes substituted for future decoding. In this section, we focus on techniques that do not incur any loss during compression:

- Arithmetic encoding
- Run-length encoding
- Huffman encoding
- Lempel-Ziv encoding

Here, we ignore arithmetic encoding and consider only the last three encoding

techniques.

7.5.1. Run-Length Encoding

One of the simplest data-compression techniques is run-length encoding. This technique is fairly effective for compression of plaintext and numbers, especially for facsimile systems. With run-length code, repeated letters can be replaced by a run length, beginning with C_c to express the compression letter count.

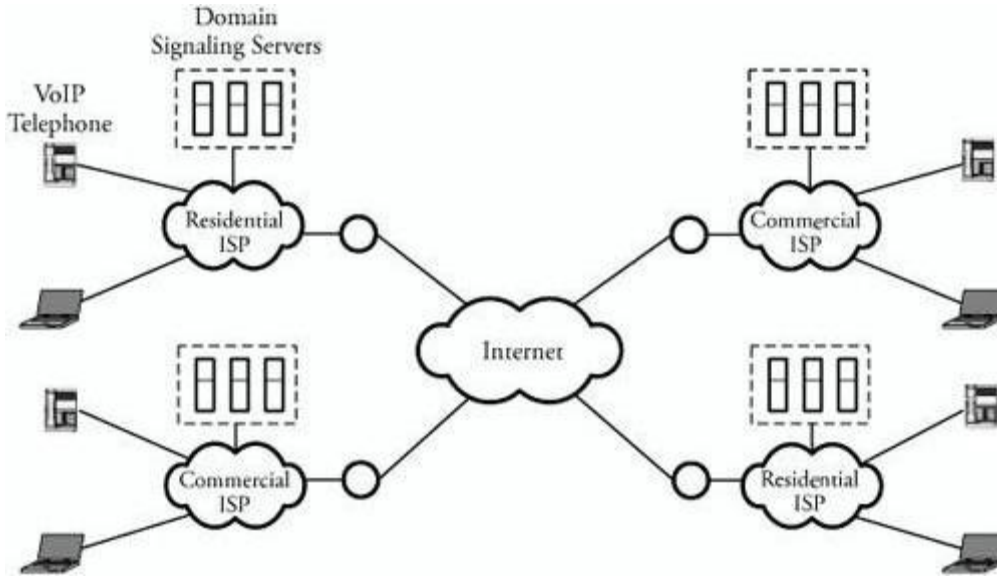
7.7 Overview of IP Telephony:

An IP telephone can be used to make telephone calls over IP networks. Voice over IP (VoIP), or IP telephony, uses packet-switched networks to carry voice traffic in addition to data traffic. The basic scheme of IP telephony starts with pulse code modulation, discussed in Chapter 7. The encoded data is transmitted as packets over packet-switched networks. At a receiver, the data is decoded and converted back to analog form. The packet size must be properly chosen to prevent large delays. The IP telephone system must also be able to handle the signaling function of the call setup, mapping of phone number to IP address, and proper call termination.

Basic components of an IP telephone system include IP telephones, the Internet backbone, and signaling servers, as shown in Figure 7.1. The IP telephone can also be a laptop or a computer with the appropriate software. An IP telephone connects to the Internet through a wired or a wireless medium. The signaling servers in each domain are analogous to the central processing unit in a computer and are responsible for the coordination between IP phones. The hardware devices required to deploy packet-switched networks are less expensive than

those required for the connection-oriented public-switched telephone networks. On a VoIP network, network resources are shared between voice and data traffic, resulting in some savings and efficient use of the available network resources.

Figure 7.1. Voice over IP system



A VoIP network is operated through two sets of protocols: signaling protocols and real-time packet-transport protocols. Signaling protocols handle call setups and are controlled by the signaling servers. Once a connection is set, RTP transfers voice data in real-time fashion to destinations. RTP runs over UDP because TCP has a very high overhead. RTP builds some reliability into the UDP scheme and contains a sequence number and a real-time clock value. The sequence number helps RTP recover packets from out-of-order delivery. Two RTP sessions are associated with each phone conversation. Thus, the IP telephone plays a dual role: an RTP sender for outgoing data and an RTP

receiver for incoming data.

7.1.1. VoIP Quality-of-Service

A common issue that affects the QoS of packetized audio is jitter. Voice data requires a constant packet interarrival rate at receivers to convert data into a proper analog signal for playback. The variations in the packet interarrival rate lead to jitter, which results in improper signal reconstruction at the receiver. Typically, an unstable sine wave reproduced at the receiver results from the jitter in the signal. Buffering packets can help control the interarrival rate. The buffering scheme can be used to output the data packets at a fixed rate. The buffering scheme works well when the arrival time of the next packet is not very long. Buffering can also introduce a certain amount of delay.

Another issue having a great impact on real-time transmission quality is network latency, or delay, which is a measure of the time required for a data packet to travel from a sender to a receiver. For telephone networks, a round-trip delay that is too large can result in an echo in the earpiece. Delay can be controlled in networks by assigning a higher priority for voice packets. In such cases, routers and intermediate switches in the network transport these high-priority packets before processing lower-priority data packets.

Congestion in networks can be a major disruption for IP telephony. Congestion can be controlled to a certain extent by implementing weighted random early discard, whereby routers begin to intelligently discard lower-priority packets before congestion occurs. The drop in packets results in a subsequent decrease in the window size in TCP, which relieves congestion to a certain extent.

A VoIP connection has several QoS factors:

- Packet loss is accepted to a certain extent.
- Packet delay is normally unacceptable.
- Jitter, as the variation in packet arrival time, is not acceptable after a certain limit.

Packet loss is a direct result of the queueing scheme used in routers. VoIP can use priority queueing, weighted fair queuing, or class-based weighted fair queuing, whereby traffic amounts are also assigned to classes of data traffic. Besides these well-known queueing schemes, voice traffic can be handled by a custom queuing, in which a certain amount of channel bandwidth for voice traffic is reserved.

Although the packet loss is tolerable to a certain extent, packet delay may not be tolerable in most cases. The variability in the time of arrival of packets in packet-switched networks gives rise to jitter variations. This and other QoS issues have to be handled differently than in conventional packet-switched networks. QoS must also consider connectivity of packet-voice environment when it is combined with traditional telephone networks.

7.6.2. VoIP Signaling Protocols

The IP telephone system must be able to handle signalings for call setup, conversion of phone number to IP address mapping, and proper call termination. Signaling is required for call setup, call management, and call termination. In the standard telephone network, signaling involves identifying the user's location

given a phone number, finding a route between a calling and a called party, and handling the issue of call forwarding and other call features.

IP telephone systems can use either a distributed or a centralized signaling scheme. The distributed approach enables two IP telephones to communicate using a client/ server model, as most Internet applications do. The distributed approach works well with VoIP networks within a single company. The centralized approach uses the conventional model and can provide some level of guarantee. Three well-known signaling protocols are

1. Session Initiation Protocol (SIP)
2. H.323 protocols
3. Media Gateway Control Protocol (MGCP)

Figure 7.2 shows the placement of VoIP in the five-layer TCP/IP model. SIP, H.323, and MGCP run over TCP or UDP; real-time data transmission protocols, such as RTP, typically run over UDP. Real-time transmissions of audio and video traffic are implemented over UDP, since the real-time data requires lower delay and less overhead. Our focus in this chapter is on the two signaling protocols SIP and H.323 and on RTP and RTCP.

Figure 7.2. Main protocols for VoIP and corresponding layers of operation

<table>
<tr><td rowspan="2">Layer</td><td colspan="8">Protocol</td></tr>
<tr><td colspan="2">SIP</td><td colspan="6">H.323</td></tr>
<tr><td rowspan="3">5</td><td rowspan="2">Other Signals</td><td rowspan="2">Media Transport</td><td rowspan="2">Registration</td><td colspan="2">Media Transport</td><td rowspan="2">Security</td><td rowspan="2">Signaling</td><td rowspan="2">Data</td></tr>
<tr><td>Voice Codec</td><td>Video Codec</td></tr>
<tr><td></td><td></td><td>H.225.0-RAS</td><td>G.711
H.263
G.722
G.723
G.728</td><td>H.261
H.323</td><td>H.235</td><td>H.225.9-Q.931
H.250
H.245
H.250</td><td>T.120</td></tr>
<tr><td></td><td>RTP, RTCP</td><td colspan="3">RTP, RTCP</td><td></td><td></td></tr>
<tr><td>4</td><td colspan="5">UDP</td><td colspan="3">TCP</td></tr>
<tr><td>3</td><td colspan="8">IP, RSVP, and IGMP</td></tr>
</table>

7.2.1. Session Initiation Protocol (SIP)

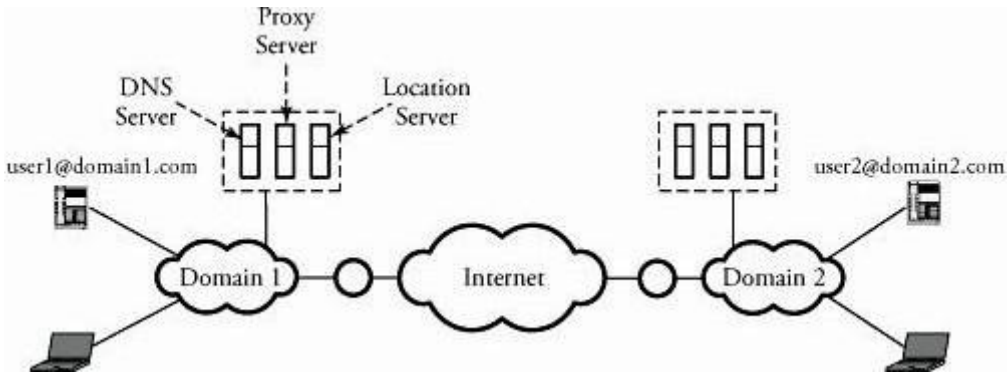
The Session Initiation Protocol (SIP) is one of the most important VoIP signaling protocols operating in the application layer in the five-layer TCP/IP model. SIP can perform both unicast and multicast sessions and supports user mobility. SIP handles signals and identifies user location, call setup, call termination, and busy signals. SIP can use multicast to support conference calls and uses the Session Description Protocol (SDP) to negotiate parameters.

SIP Components

Figure 7.3 shows an overview of SIP. A call is initiated from a user agent: the user's IP telephone system, which is similar to a conventional phone. A user agent assists in initiating or terminating a phone call in VoIP networks. A user agent can be implemented in a standard telephone or in a laptop with a microphone that runs some software. A user agent is identified using its

associated domain. For example, user1@domain1.com refers to user 1, who is associated with the domain1.com network.

Figure 7.3. Overview of SIP



7.3. Real-Time Media Transport Protocols

In real-time applications, a stream of data is sent at a constant rate. This data must be delivered to the appropriate application on the destination system, using real-time protocols. The most widely applied protocol for real-time transmission is the Real-Time Transport Protocol (RTP), including its companion version: Real-Time Control Protocol (RTCP).

UDP cannot provide any timing information. RTP is built on top of the existing UDP stack. Problems with using TCP for real-time applications can be identified easily. Real-time applications may use multicasting for data delivery. As an end-toend protocol, TCP is not suited for multicast distribution. TCP uses a retransmission strategy for lost packets, which then arrive out of order. Real-time applications cannot afford these delays. TCP does not maintain timing information for packets. In real-time applications, this would be a requirement.

7.8 Real-Time Transport Protocol (RTP):

The real-time transport protocol (RTP) provides some basic functionalities to real-time applications and includes some specific functions to each application. RTP runs on top of the transport protocol as UDP. As noted in Chapter 7, UDP is used for port addressing in the transport layer and for providing such transport-layer functionalities as reordering. RTP provides application-level framing by adding application-layer headers to datagrams. The application breaks the data into smaller units, called application data units (ADUs). Lower layers in the protocol stack, such as the transport layer, preserve the structure of the ADU.

Real-time applications, such as voice and video, can tolerate a certain amount of packet loss and do not always require data retransmission. The mechanism RTP uses typically informs a source about the quality of delivery. The source then adapts its sending rate accordingly. If the rate of packet loss is very high, the source might switch to a lower-quality transmission, thereby placing less load on the network. A real-time application can also provide the data required for retransmission. Thus, recent data can be sent instead of retransmitted old data. This approach is more practical in voice and video applications. If a portion of an ADU is lost, the application is unable to process the data, and the entire ADU would have to be retransmitted.

Real-Time Session and Data Transfer

The TCP/IP and OSI models divide the network functionalities, based on a layered architecture. Each layer performs distinct functions, and the data flows sequentially between layers. The layered architecture may restrict the

implementation on certain functions out of the layered order. Integrated layer processing dictates a tighter coupling among layers. RTP is used to transfer data among sessions in real time. A session is a logical connection between an active client and an active server and is defined by the following entities:

- RTP port number, which represents the destination port address of the RTP session. Since RTP runs over UDP, the destination port address is available on the UDP header.
- IP address of the RTP entity, which involves an RTP session. This address can be either a unicast or a multicast address.

RTP uses two relays for data transmission. A relay is an intermediate system that acts as both a sender and a receiver. Suppose that two systems are separated by a firewall that prevents them from direct communication. A relay in this context is used to handle data flow between the two systems. A relay can also be used to convert the data format from a system into a form that the other system can process easily. Relays are of two types: mixer and translator.

A mixer relay is an RTP relay that combines the data from two or more RTP entities into a single stream of data. A mixer can either retain or change the data format. The mixer provides timing information for the combined stream of data and acts as the source of timing synchronization. Mixers can be used to combine audio streams in real-time applications and can be used to service systems that may not be able to handle multiple RTP streams.

The translator is a device that generates one or more RTP packets for each incoming RTP packet. The format of the outgoing packet may be different from that of the incoming packet. A translator relay can be used in video applications

in which a high-quality video signal is converted to a lower-quality in order to service receivers that support a lower data rate. Such a relay can also be used to transfer packets between RTP entities separated by an application-level firewall. Translators can sometimes be used to transfer an incoming multicast packet to multiple destinations.

RTP Packet Header

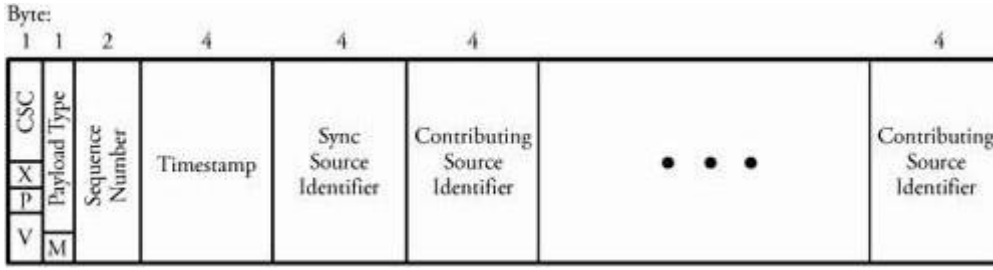
RTP contains a fixed header and an application-specific variable-length header field. Figure 7.8 shows the RTP header format. The RTP header fields are:

- Version (V), a 2-bit field indicating the protocol version.
- Padding (P), a 1-bit field that indicates the existence of a padding field at the end of the payload. Padding is required in applications that require the payload to be a multiple of some length.
- Extension (X), a 1-bit field indicating the use of an extension header for RTP.
- Contributing source count (CSC), a 4-bit field that indicates the number of contributing source identifiers.
- Marker (M), a 1-bit field indicating boundaries in a stream of data traffic. For video applications, this field can be used to indicate the end of a frame.
- Payload type, A 7-bit field specifying the type of RTP payload. This field also contains information on the use of compression or encryption.
- Sequence number, a 16-bit field that a sender uses to identify a particular packet within a sequence of packets. This field is used to detect packet loss and for packet reordering.
- Timestamp, a 32-bit field enabling the receiver to recover timing information. This field indicates the timestamp when the first byte of data in the payload was

generated.

- Synchronization source identifier, a randomly generated field used to identify the RTP source in an RTP session.
- Contributing source identifier, an optional field in the header to indicate the contributing sources for the data.

Figure 7.7. Packet format for the real-time transport protocol



Overall, the main segment of an RTP header includes 12 bytes and is appended to a packet being prepared for multimedia application.

7.7.2. Real-Time Control Protocol (RTCP)

The Real-Time Transport Protocol (RTCP) also runs on top of UDP. RTCP performs several functions, using multicasting to provide feedback about the data quality to all session members. The session multicast members can thus get an estimate of the performance of other members in the current active session. Senders can send reports about data rates and the quality of data transmission. Receivers can send information about packet-loss rate, jitter variations, and any other problems they might encounter. Feedback from a receiver can also enable a sender to diagnose a fault. A sender can isolate a problem to a single RTP entity or a global problem by looking at the reports from all receivers.

RTCP performs source identification. RTCP packets contain some information to identify the source of the control packet. The rate of RTCP packets must also be kept to less than 5 percent of the total session traffic. Thus, this protocol carries out "rate control" of RTCP packets. At the same time, all session members must be able to evaluate the performance of all other session members. As the number of active members in a session increases, the transmission rates of the control packets must be reduced. RTCP is also responsible for session control and can provide some session-control information, if necessary.

Packet Type and Format

RTCP transmits control information by combining a number of RTCP packets in a single UDP datagram. The RTCP packet types are sender reports (SR), receiver reports (RR), source descriptor (SDES), goodbye (BYE), and application-specific types. Figure 7.9 shows some RTCP packet formats. The fields common to all packet types are as follows:

- Version, a 2-bit field that indicates the current version.
- Padding, a 1-bit field that indicates the existence of padding bytes at the end of the control data.
- Count, a 5-bit field that indicates the number of SR or RR reports or the number of source items in the SDES packet.
- Packet type, an 8-bit field that indicates the type of RTCP packet. (Four RTCP packet types were specified earlier.)
- Length, a 16-bit field that represents the length of packet in 32-bit words minus 1.
- Synchronization source identifier, a field common to the SR and RR packet

types; it indicates the source of the RTCP packets.

Figure 7.9. Format of the SR packet in RCTP

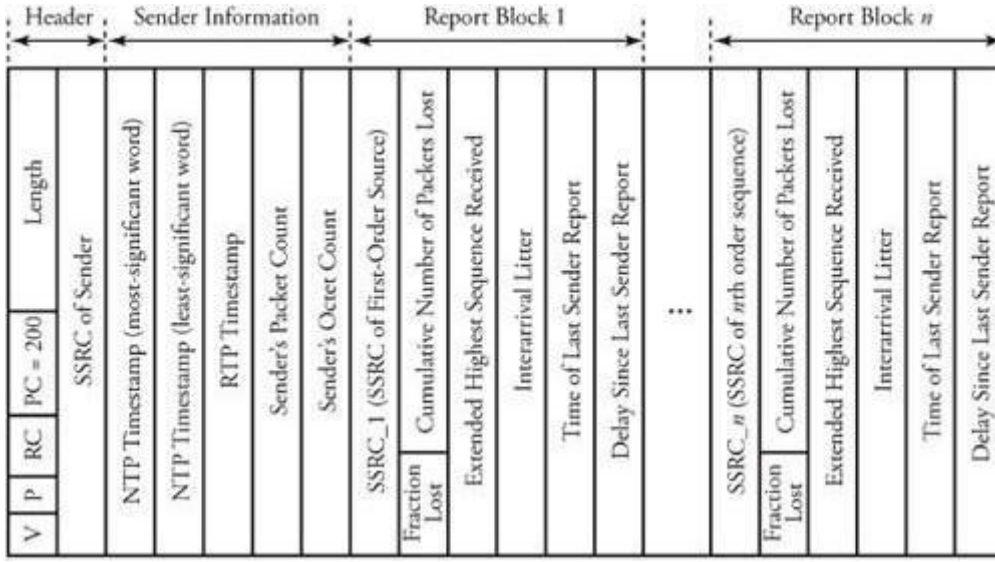


Figure 7.9 also shows a typical format of a sender report. The report consists of the common header fields and a block of sender information. The sender report may also contain zero or more receiver report blocks, as shown in the figure. The fields in the sender information block are:

- NTP timestamp, a 64-bit field that indicates when a sender report was sent. The sender can use this field in combination with the timestamp field returned in receiver reports to estimate the round-trip delay to receivers.
- RTP timestamp, a 32-bit field used by a receiver to sequence RTP data packets from a particular source.
- Sender's packet count, a 32-bit field that represents the number of RTP data packets transmitted by the sender in the current session.

- Sender's byte count, a 32-bit field that represents the number of RTP data octets transmitted by the sender in the current session.

The SR packet includes zeros or more RR blocks. One receiver block is included for each sender from which the member has received data during the session. The RR block includes the following fields:

- SSRC_n, a 32-bit field that identifies the source in the report block, where n is the number of sources.
- Fraction lost, an 8-bit field indicating the fraction of data packet loss from source SSRC_n since the last SR or RR report was sent.
- Cumulative number of packets lost, a 24-bit field that represents the total number of RTP data packets lost from the source in the current active session identified by SSRC_n.
- Extended highest sequence number received, the first 16 least-significant bits, used to indicate the highest sequence number for packets received from source SSRC_n. The first 16 most-significant bits indicate the number of times that the sequence number has been wrapped back to zero.
- Interarrival jitter, a 32-bit field used to indicate the jitter variations at the receiver for the source SSRC_n.
- Last SR timestamp, a 32-bit field indicating the timestamp for the last SR packet received from the source SSRC_n.
- Delay since last SR, a 32-bit field indicating the delay between the arrival time of the last SR packet from the source SSRC_n and the transmission of the current report block.

Receivers in RTCP can provide feedback about the quality of reception through

a receiver report. A receiver that is also a sender during a session, it also sends the sender reports.

7.3.3. Estimation of Jitter in Real-Time Traffic

The jitter factor is a measure of the delay experienced by RTP packets in a given session. The average jitter can be estimated at the receiver. Let us define the following parameters at the receiver:

t_i	Timestamp of the RTP data packet i indicated by the source.
α_i	Arrival time of the RTP data packet i at the receiver.
d_i	Measure of difference between interarrival time of RTP packets at receiver and the one for packet departure from the source. This value represents the difference in packet spacing at source and receiver.
E[i]	Estimate of average jitter until the time of packet i arrival.

The difference interval d_i is given by

Equation 18.1

$$d_i = (a_i - a_{i-1}) - (t_i - t_{i-1}).$$

The estimated average jitter until the time of packet i arrival is given by

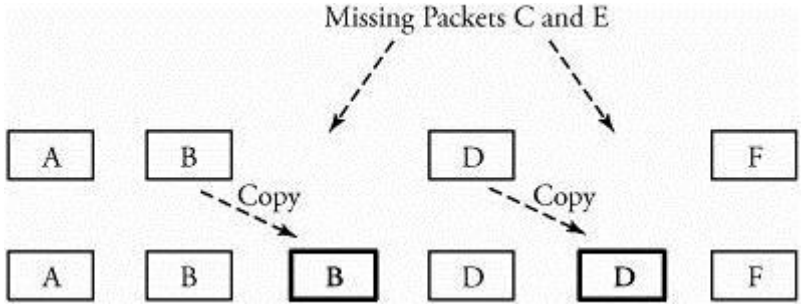
Equation 18.2

$$E[i] = k\left(E[i-1] + |d_i|\right),$$

where k is a normalizing coefficient. The interarrival jitter value indicated in the sender report provides useful information on network conditions to the sender and the receiver. The jitter value can be used as an estimate to calculate the variation of network congestion.

RTP packet-sequence numbers are used to help a receiver sequence packets in order to recover lost packets. When packets arrive out of order at the receiver, the sequence number can be used to assemble data packets. Consider Figure 7.10. When certain packets are lost, the gaps are filled in with previously received data packets. As shown in the figure, packet D is replayed twice, since packet C is lost. This mechanism can help reduce the pips and clicks in voice signals owing to lost packets. This reconstructed data stream is sent to the receiver, with the lost packets replaced by previously received packets. This can significantly improve the latency.

Figure 7.10. Missing voice packets and reconstructing the data stream



7.9 Stream Control Transmission Protocol (SCTP):

The Stream Control Transmission Protocol (SCTP) provides a general-purpose transport protocol for message-oriented applications. It is a reliable transport protocol for transporting stream traffic, can operate on top of unreliable connectionless networks, and offers acknowledged and nonduplicated transmission data on connectionless networks (datagrams). SCTP has the following features.

- The protocol is error free. A retransmission scheme is applied to compensate for loss or corruption of the datagram, using checksums and sequence numbers.
- It has ordered and unordered delivery modes.
- SCTP has effective methods to avoid flooding congestion and masquerade attacks.
- This protocol is multipoint and allows several streams within a connection.

In TCP, a stream is a sequence of bytes; in SCTP, a sequence of variable-sized messages. SCTP services are placed at the same layer as TCP or UDP services. Streaming data is first encapsulated into packets, and each packet carries several correlated chunks of streaming details. If an MPEG movie is displayed live over the Internet, a careful assignment of data per packet is required. An MPEG video consists of frames, each consisting of n x m blocks of pixels, with each pixel normally an 8 x 8 matrix. In this case, each block of pixels can be encapsulated into a chunk, where each row of the block is formatted as a packet.

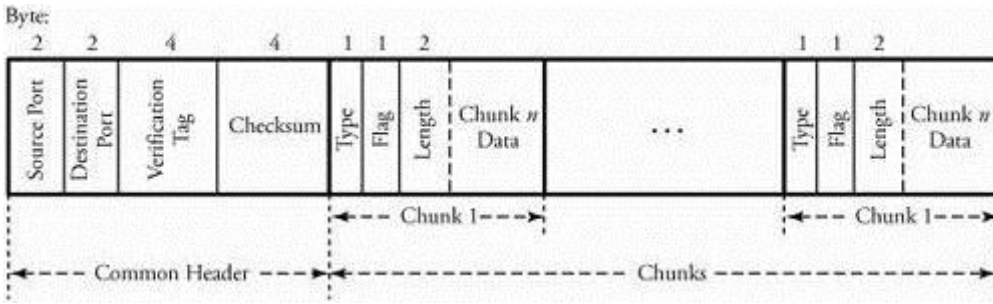
7.5.1. SCTP Packet Structure

Figure 18.14 shows the structure of streaming packets used in SCTP. An SCTP

packet is also called a protocol data unit (PDU). As soon as the streaming data is ready to be transmitted over IP, an SCTP packet forms the payload of an IP packet. Each packet consists of a common header and chunks. The streaming data is distributed over packets, and each packet carries correlated "chunks" of streaming data. Multiple chunks representing multiple portions of streaming information are in fact multiplexed into one packet up to the path-maximum packet size.

Figure 7.14. The structure of packets in the stream control transmission protocol (SCTP). Streaming data is encapsulated into packets and each packet carries several correlated chunks of streaming details.

[View full size image]



A chunk header starts with a chunk type field used to distinguish data chunks and any other types of control chunks. The type field is followed by a flag field and a chunk length field to indicate the chunk size. A chunk, and therefore a packet, may contain either control information or user data. The common header begins with the source port number followed by the destination port number. SCTP uses the same port concept as TCP or UDP does. A 32-bit verification tag field is exchanged between the end-point servers at startup to verify the two

servers involved. Thus, two tag values are used in a connection. The common header consists of 12 bytes. SCTP packets are protected by a 32-bit checksum. The level of protection is more robust than the 16-bit checksum of TCP and UDP.

Each packet has n chunks, and each chunk is of two types: payload data chunk for transmitting actual streaming data and control chunks for signaling and control. Signaling and control chunks are of several different types, as follows:

- Initiation, to initiate an SCTP session between two end points
- Initiation acknowledgment, to acknowledge the initiation of an SCTP session
- Selective acknowledgment, to be transmitted to a peer end point to acknowledge received data chunks
- Heartbeat request, to probe the reachability of a particular destination transport address defined in the session
- Heartbeat acknowledgment, to respond to the heartbeat request chunk
- Abort, to close a session
- Shutdown, to initiate a graceful close of a session
- Shutdown acknowledgment, to acknowledge receipt of the shutdown chunk once the shutdown process is completed
- Operation error, to notify the other party of a certain error
- State cookie, sent by the source to its peer to complete the initialization process
- Cookie acknowledgment, to acknowledge receipt of a state cookie chunk
- Shutdown complete, to acknowledge receipt of the shutdown acknowledgment chunk at the completion of the shutdown process

SCTP can easily and effectively be used to broadcast live video clips or full-

color video movies. The SCTP exercises at the end of this chapter explore SCTP further.

UNIT – 8

Mobile AdHoc Networks and Wireless Sensor Neworks: Overview of Wireless Ad-Hoc networks, Routing in AdHOc Networks, Routing protocols for and Security of AdHoc networks, Sensor Networks and protocol structures, Communication Energy model, Clustering protocols, Routing protocols, ZigBee technology and 802.15.4.

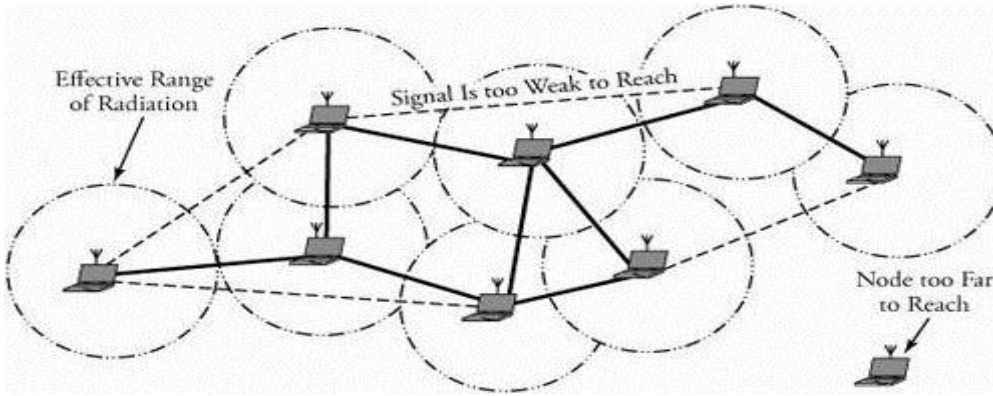
UNIT – 8

MOBILE ADHOC NETWORKS AND WIRELESS SENSOR NEWORKS:

8.1. Overview of Wireless Ad-Hoc Networks

Wireless mobile ad-hoc network (MANET) technology is designed for the establishment of a network anywhere and anytime, without any fixed infrastructure to support the mobility of the users in the network. In other words, a wireless ad-hoc network is a collection of mobile nodes with a dynamic network infrastructure forming a temporary network. Such networks no central server or base station for providing connectivity, and all network intelligence must be placed inside the mobile user devices. Figure 8.1 gives overview of an ad-hoc network, where wireless mobile nodes have formed a network, with one node too far to reach.

Figure 8.1. Overview of an ad-hoc network



In such an environment, each mobile user acts as a routing node, and a packet is routed from a source to its destination by incorporating of other network users. Since the topology of an ad-hoc network can change quickly and unpredictably, it should be adaptable to changes, such as when a link breaks, a node leaves the network, or a new node is attached to the network. Thus, unlike intradomain routing algorithm for regular networks, if a node leaves an ad-hoc network, all affected nodes can discover new routes. Ad-hoc networks have several types of applications

- Rescue operations. In an emergency public disaster, such as an earthquake, ad-hoc networks can be set up at a location where no infrastructure is present. Ad-hoc networks can be used to support network connectivity when no fixed network is available.
- Military. Ad-hoc networks can be used in a battle zone, for a military command and mobile units.
- Law enforcement and security operations. An ad-hoc network can be used in a temporary security operation, acting as a mobile surveillance network.
- Home networks. An ad-hoc network can be used to support seamless connectivity among various devices.
- Conferencing. Ad-hoc networks can be set up for a presentation. An audience can download a presentation, browse the slides on a portable device, print them on the local printer, or e-mail the presentation to an absent colleague.

Ad-hoc networks must possess several unique features. One is automatic discovery of available services. Each time a new service becomes available, an ad hoc networking device has to configure use of the new service. As an

ad-hoc network lacks centralized administration, the network must be able to prevent network collapse when one of the mobile nodes moves out of transmitter range. In general, nodes should be able to enter or leave the network as they wish. Thus, every node acts as both a host and a router, and the network must be intelligent enough to handle network dynamics. This property is called self-stabilization.

One of the most common tasks of an ad-hoc network is to multicast a message to many users efficiently. In such an environment, networks are subject to severe blocking. Thus, the performance of an ad hoc system depends on the stability of the network architecture. The inclusion of all these features in ad-hoc networks requires considerable architectue sophistication.

8.2. Routing in Ad-Hoc Networks:

The lack of a backbone infrastructure makes packet routing in ad-hoc networks a challenging task. A routing protocol should be able to automatically recover from any problem in a finite amount of time without human intervention. Conventional routing protocols are designed for nonmoving infrastructures and assume that routes are bidirectional, which is not always the case for ad-hoc networks. Identification of mobile terminals and correct routing of packets to and from each terminal while moving are certainly challenging.

Since the topology of an ad-hoc network is dynamic, reserving resources and sustaining QoS are difficult. In an ad-hoc medium, it may not be possible to communicate bidirectionally, so ad-hoc routing protocols should assume that

links are unidirectional. The power of a wireless device is another important factor. The routing protocol also has to support node stand-by mode. Devices such as laptops are very limited in battery power; hence, the use of stand-by mode to save power is important.

8.2.1. Classification of Routing Protocols

Ad-hoc routing protocols can be classified into two broad categories:

1. Centralized versus distributed. In centralized routing protocols, the routing decision is made at a central node. In distributed routing protocols, the routing decision is made by all the network nodes. Routing protocols in most efficiently designed ad-hoc networks are distributed to increase the reliability of the network. In such cases, nodes can enter or leave the network easily, and each node can make routing decisions, using other collaborative nodes.
2. Static versus adaptive. In static routing protocols, a route of a source/destination pair does not change because of any traffic condition or link failure. In adaptive routing protocols, routes may change because of any congestion.

Whether a protocol is centralized, distributed, static, or adaptive, it can generally be categorized as either table driven or source initiated.

Table-Driven Routing Protocols

Table-driven, or proactive, routing protocols find routes to all possible destinations ahead of time. are needed. The routes are recorded in the nodes' routing tables and are updated within the predefined intervals. Proactive protocols are faster in decision making but need more time to converge to a steady state, causing problems if the topology of the network continually changes. However, maintaining routes can lead to a large overhead. Table-driven

protocols require every node to maintain one or more tables to store updated routing information from every node to all other nodes. Nodes propagate updated tables all over the network such that the routing information in each table corresponds to topological changes in the network.

Source-Initiated Routing Protocols

Source-initiated, or reactive, routing protocols, are on-demand procedures and create routes only when requested to do so by source nodes. A route request initiates a route-discovery process in the network and is completed once a route is discovered. If it exists at the time of a request, a route is maintained by a route-maintenance procedure until either the destination node becomes irrelevant to the source or the route is no longer needed. On-demand protocols are more suitable for ad-hoc networks. In most cases, reactive protocols are desired. This means that the network reacts only when needed and does not broadcast information periodically. However, the control overhead of packets is smaller than for proactive protocols.

Routing Protocols for Ad-Hoc Networks

This section discusses three table-driven protocols and four source-initiated protocols. The table-driven protocols are the Destination-Sequenced Distance Vector (DSDV) protocol, the Cluster-Head Gateway Switch Routing (CGSR) protocol, and the Wireless Routing Protocol (WRP). The source-initiated protocols are the Dynamic Source Routing (DSR) protocol, the Associative-Based Routing (ABR) protocol, Temporally Ordered Routing Algorithm (TORA), and Ad-Hoc On-Demand Distance Vector (AODV) protocol.

8.2.1. Destination-Sequenced Distance Vector (DSDV) Protocol

The Destination-Sequenced Distance Vector (DSDV) protocol is a table-driven routing protocol based on the improved version of classical Bellman-Ford routing algorithm. DSDV is based on the Routing Information Protocol (RIP), explained in Chapter 7. With RIP, a node holds a routing table containing all the possible destinations within the network and the number of hops to each destination. DSDV is also based on distance vector routing and thus uses bidirectional links. A limitation of DSDV is that it provides only one route for a source/destination pair.

Routing Tables

The structure of the routing table for this protocol is simple. Each table entry has a sequence number that is incremented every time a node sends an updated message. Routing tables are periodically updated when the topology of the network changes and are propagated throughout the network to keep consistent information throughout the network.

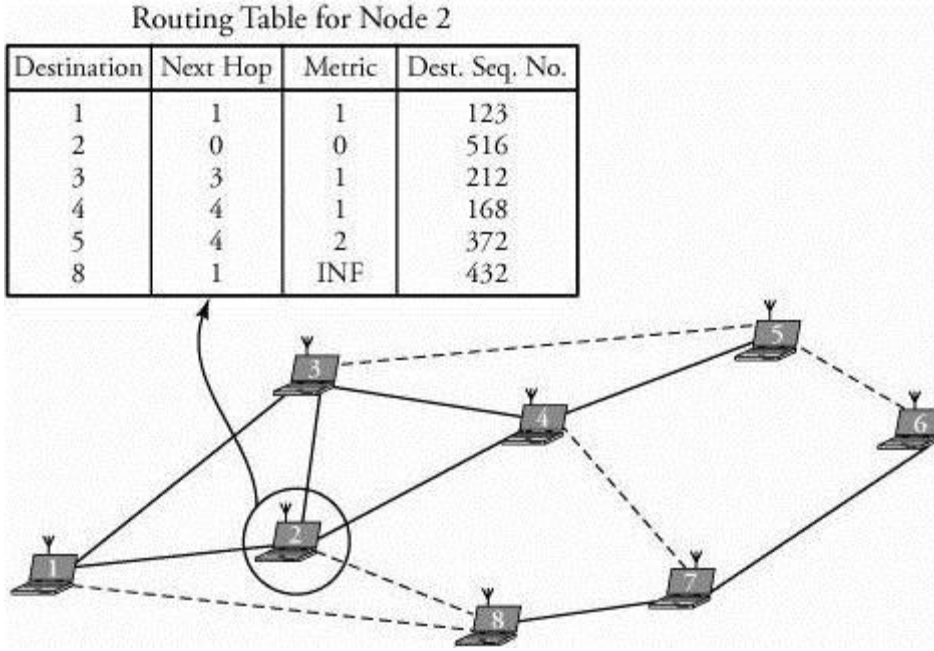
Each DSDV node maintains two routing tables: one for forwarding packets and one for advertising incremental routing packets. The routing information sent periodically by a node contains a new sequence number, the destination address, the number of hops to the destination node, and the sequence number of the destination. When the topology of a network changes, a detecting node sends an update packet to its neighboring nodes. On receipt of an update packet from a neighboring node, a node extracts the information from the packet and updates its routing table as follows

DSDV Packet Process Algorithm

1. If the new address has a higher sequence number, the node chooses the route with the higher sequence number and discards the old sequence number.
2. If the incoming sequence number is identical to the one belonging to the existing route, a route with the least cost is chosen.
3. All the metrics chosen from the new routing information are incremented.
4. This process continues until all the nodes are updated. If there are duplicate updated packets, the node considers keeping the one with the least-cost metric and discards the rest.

   In case of a broken link, a cost of ∞metric with a new sequence number (incremented) is assigned to it to ensure that the sequence number of that metric is always greater than or equal to the sequence number of that node. Figure 8.2 shows a routing table for node 2, whose neighbors are nodes 1, 3, 4, and 8. The dashed lines indicate no communications between any corresponding pair of nodes. Therefore, node 2 has no information about node 8.

Figure 8.2. A DSDV routing table



The packet overhead of the DSDV protocol increases the total number of nodes in the ad-hoc network. This fact makes DSDV suitable for small networks. In large adhoc networks, the mobility rateand therefore the overheadincrease, making the network unstable to the point that updated packets might not reach nodes on time.

8.2.2. Cluster-Head Gateway Switch Routing Protocol

The Cluster-Head Gateway Switch Routing (CGSR) protocol is a table-driven routing protocol. In a clustering system. each predefined number of nodes are formed into a cluster controlled by a cluster head, which is assigned using a distributed clustering algorithm. However, with the clustering scheme, a cluster

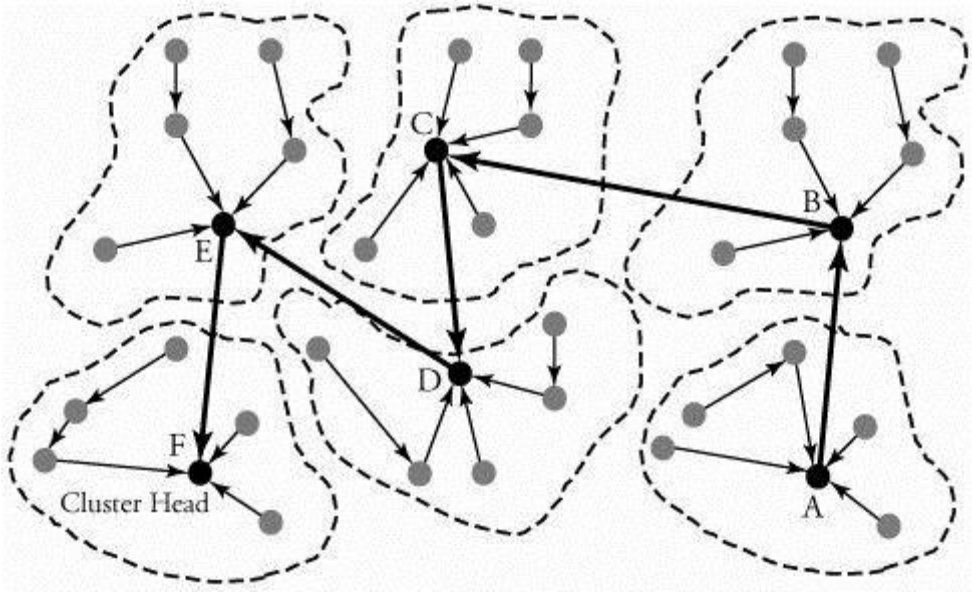
head can be replaced frequently by another node, for several reasons, such as lower-level energy left in the node or a node moves out of contact.

With this protocol, each node maintains two tables: a cluster-member table and a routing table. The cluster-member table records the cluster head for each destination node, and the routing table contains the next hop to reach the destination. As with the DSDV protocol, each node updates its cluster-member table on receiving a new update from its neighbors.

Clustering and Routing Algorithms

CGSR routing involves cluster routing, whereby a node is required to find the best route over cluster heads from the cluster-member table. Figure 19.3 shows an example of routing in an area in which six clusters have been formed. A node in cluster A is transmitting a packet to a node in cluster F. Nodes within each cluster route their packets to their own associated clusters. The transmitting node then sends its packet to the next hop, according to the routing table entry associated with that cluster head. The cluster head transmits the packet to another cluster head until the cluster head of the destination node is reached. The routing is made through a series of available cluster heads from A to F. Packets are then transmitted to the destination.

Figure 8.2. Communication with cluster-head gateway switch routing (CGSR) protocol



8.2.3. Wireless Routing Protocol (WRP)

The Wireless Routing Protocol (WRP) is a table-based protocol maintaining routing information among all nodes in the network. This protocol is based on the distributed Bellman-Ford algorithm. The main advantage of WRP is that it reduces the number of routing loops. With this protocol, each node in a network maintains four tables, as follows:

1. Distance table, which holds the destination, next hop, distance, and predecessors of each destination and each neighbor.
2. Routing table, which saves the destination address, next hop, distance, predecessor, and a marker for each destination, specifying whether that entry corresponds to a simple path.

3. Link-cost table, which provides the link cost to each neighbor and also the number of periodic update periods elapsed since the node received any error-free message from it.
4. Message transmission-list table, which records which updates in an update message are to be retransmitted and which neighbors need to acknowledge the retransmission. The table provides the sequence number of the update message, a retransmission counter, acknowledgments, and a list of updates sent in the update message.

   Nodes should either send a message including the update message or a HELLO message to their neighbors. If a node has no message to send, it should send a HELLO message to ensure connectivity. If the sending node is new, it is added to the node's routing table, and the current node sends the new node a copy of its routing table content.

   Once it detects a change in a route, a node sends the update message to its neighbors. The neighboring nodes then change their distance entries and look for new possible paths through other nodes. This protocol avoids the count-to-infinity issue present in most ad-hoc network protocols. This issue is resolved by making each node perform consistency checks of predecessor information reported by all its neighbors in order to remove looping and make a faster route convergence in the presence of any link or node failure.

8.3 Routing protocols for and Security of AdHoc networks:

This section discusses three table-driven protocols and four source-initiated protocols. The table-driven protocols are the Destination-Sequenced Distance Vector (DSDV) protocol, the Cluster-Head Gateway Switch Routing (CGSR) protocol, and the Wireless Routing Protocol (WRP). The source-initiated protocols are the Dynamic Source Routing (DSR) protocol, the Associative-Based Routing (ABR) protocol, Temporally Ordered Routing Algorithm (TORA), and Ad-Hoc On-Demand Distance Vector (AODV) protocol.

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Routing Tables

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DSDV Packet Process Algorithm

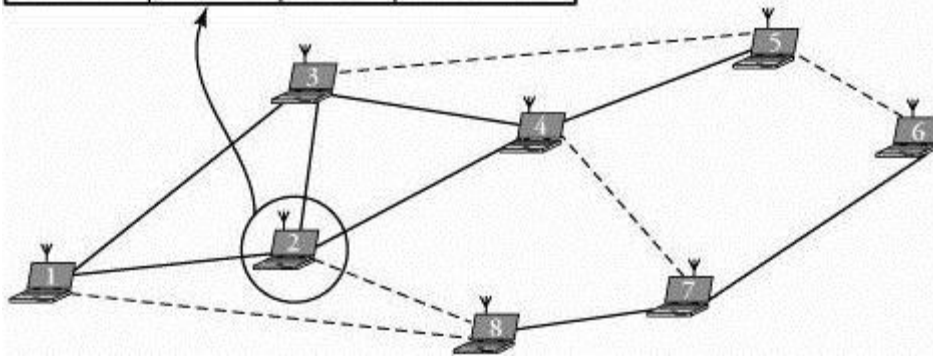
5. If the new address has a higher sequence number, the node chooses the route with the higher sequence number and discards the old sequence number.
6. If the incoming sequence number is identical to the one belonging to the existing route, a route with the least cost is chosen.
7. All the metrics chosen from the new routing information are incremented.
8. This process continues until all the nodes are updated. If there are duplicate updated packets, the node considers keeping the one with the least-cost metric and discards the rest.

In case of a broken link, a cost of ∞metric with a new sequence number (incremented) is assigned to it to ensure that the sequence number of that metric is always greater than or equal to the sequence number of that node. Figure 8.2 shows a routing table for node 2, whose neighbors are nodes 1, 3, 4, and 8. The dashed lines indicate no communications between any corresponding pair of nodes. Therefore, node 2 has no information about node 8.

Figure 8.2. A DSDV routing table

Routing Table for Node 2

Destination	Next Hop	Metric	Dest. Seq. No.
1	1	1	123
2	0	0	516
3	3	1	212
4	4	1	168
5	4	2	372
8	1	INF	432



The packet overhead of the DSDV protocol increases the total number of nodes in the ad-hoc network. This fact makes DSDV suitable for small networks. In large adhoc networks, the mobility rateand therefore the overheadincrease, making the network unstable to the point that updated packets might not reach nodes on time.

8.3.2. Cluster-Head Gateway Switch Routing Protocol

The Cluster-Head Gateway Switch Routing (CGSR) protocol is a table-driven routing protocol. In a clustering system. each predefined number of nodes are formed into a cluster controlled by a cluster head, which is assigned using a distributed clustering algorithm. However, with the clustering scheme, a cluster

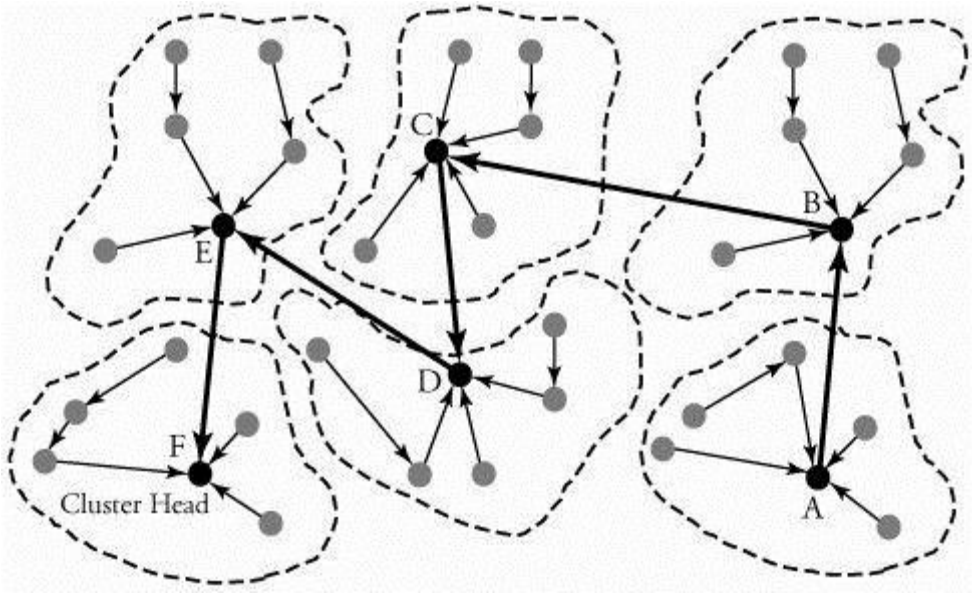
head can be replaced frequently by another node, for several reasons, such as lower-level energy left in the node or a node moves out of contact.

With this protocol, each node maintains two tables: a cluster-member table and a routing table. The cluster-member table records the cluster head for each destination node, and the routing table contains the next hop to reach the destination. As with the DSDV protocol, each node updates its cluster-member table on receiving a new update from its neighbors.

Clustering and Routing Algorithms

CGSR routing involves cluster routing, whereby a node is required to find the best route over cluster heads from the cluster-member table. Figure 8.3 shows an example of routing in an area in which six clusters have been formed. A node in cluster A is transmitting a packet to a node in cluster F. Nodes within each cluster route their packets to their own associated clusters. The transmitting node then sends its packet to the next hop, according to the routing table entry associated with that cluster head. The cluster head transmits the packet to another cluster head until the cluster head of the destination node is reached. The routing is made through a series of available cluster heads from A to F. Packets are then transmitted to the destination.

Figure 8.3. Communication with cluster-head gateway switch routing (CGSR) protocol



8.3.3. Wireless Routing Protocol (WRP)

The Wireless Routing Protocol (WRP) is a table-based protocol maintaining routing information among all nodes in the network. This protocol is based on the distributed Bellman-Ford algorithm. The main advantage of WRP is that it reduces the number of routing loops. With this protocol, each node in a network maintains four tables, as follows:

5. Distance table, which holds the destination, next hop, distance, and predecessors of each destination and each neighbor.

6. Routing table, which saves the destination address, next hop, distance, predecessor, and a marker for each destination, specifying whether that entry corresponds to a simple path.
7. Link-cost table, which provides the link cost to each neighbor and also the number of periodic update periods elapsed since the node received any error-free message from it.
8. Message transmission-list table, which records which updates in an update message are to be retransmitted and which neighbors need to acknowledge the retransmission. The table provides the sequence number of the update message, a retransmission counter, acknowledgments, and a list of updates sent in the update message.

   Nodes should either send a message including the update message or a HELLO message to their neighbors. If a node has no message to send, it should send a HELLO message to ensure connectivity. If the sending node is new, it is added to the node's routing table, and the current node sends the new node a copy of its routing table content.

   Once it detects a change in a route, a node sends the update message to its neighbors. The neighboring nodes then change their distance entries and look for new possible paths through other nodes. This protocol avoids the count-to-infinity issue present in most ad-hoc network protocols. This issue is resolved by making each node perform consistency checks of predecessor information reported by all its neighbors in order to remove looping and make a faster route convergence in the presence of any link or node failure.

8.3.4. Dynamic Source Routing (DSR) Protocol

The Dynamic Source Routing (DSR) protocol is an on-demand, or source-initiated, routing protocol in which a source node finds an unexpired route to a destination to send the packet. DSR quickly adapts to topological changes and is typically used for networks in which mobile nodes move with moderate speed. Overhead is significantly reduced with this protocol, since nodes do not exchange routing table information when there are no changes in the topology. DSR creates multiple paths from a source to a destination, eliminating route discovery when the topology changes. Similar to most ad-hoc networks, DSR has two phases: route discovery and route maintenance.

Route Discovery and Maintenance

Route discovery is initiated when a node wants to send packets to another node and no unexpired route to the destination is in its routing table. In such circumstances, the node first broadcasts a route-request packet, including the destination address, source address, and a unique identification number. When it receives a route-request packet, the neighboring node it looks at its table; if any route to the requested destination address is already present in the node's route record, the packet is discarded to avoid the looping issue. Otherwise, the node adds its own address to the preassigned field of the route-request packet and forwards it to its neighboring nodes.

When the route-request packet reaches a destination node or another intermediate node that has an unexpired route to the destination, this node generates a route-reply packet, which contains a route record of the sequence of nodes taken from the source to this node. Once the source receives all the route-reply packets, it updates its routing table with the best path to the destination and

sends its packets through that selected route.

8.4 Sensor Networks and protocol structures:

Chemical, biological, or solar sensors can be networked together as a sensor network to strengthen the power of sensing. A sensor network is controlled through a software core engine. The network is typically wireless but may also be wired. Sensor networks are designed to be self-configuring such that they can gather information about a large geographical area or about movements of an object for surveillance purposes.

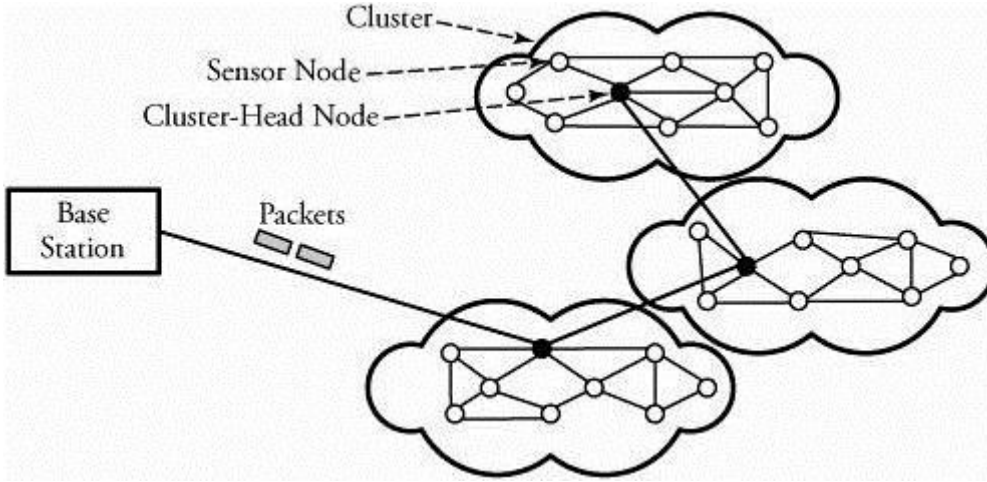
Sensor networks can be used for target tracking, environmental monitoring, system control, and chemical or biological detection. In military applications, sensor networks can enable soldiers to see around corners and to detect chemical and biological weapons long before they get close enough to cause harm. Civilian uses include environmental monitoring, traffic control, and providing health care monitoring for the elderly while allowing them more freedom to move about.

8.4.1. Clustering in Sensor Networks

The region being sensed is normally partitioned into equally loaded clusters of sensor nodes, as shown in Figure 8.1. A cluster in a sensor network resembles a domain in a computer network. In other words, nodes are inserted in the vicinity of a certain predefined region, forming a cluster. Different types of sensors can also be deployed in a region. Thus, a sensor network is typically cluster based and has irregular topology. The most effective routing scheme in sensor networks is normally based on the energy (battery level) of nodes. In such

routing schemes, the best path has the highest amount of total energy. The network of such sensing nodes is constructed with identical sensor nodes, regardless of the size of the network. In Figure 8.1, three clusters are interconnected to the main base station, each cluster contains a cluster head responsible for routing data from its corresponding cluster to a base station.

Figure 8.1. A sensor network and its clusters



Communicating nodes are normally linked by a wireless medium, such as radio. The wireless sensor node is equipped with a limited power source, such as a battery or even a solar cell, where there is enough sunlight exposure on the node. However, a solar cell may not be the best choice as a power supply, owing to its weight, volume, and expense. In some application scenarios, sensor-node lifetime depends on the battery lifetime. Removal of dead nodes can cause significant topological changes and may require packet rerouting. As a result, power management is a key issue in system design, node design, and

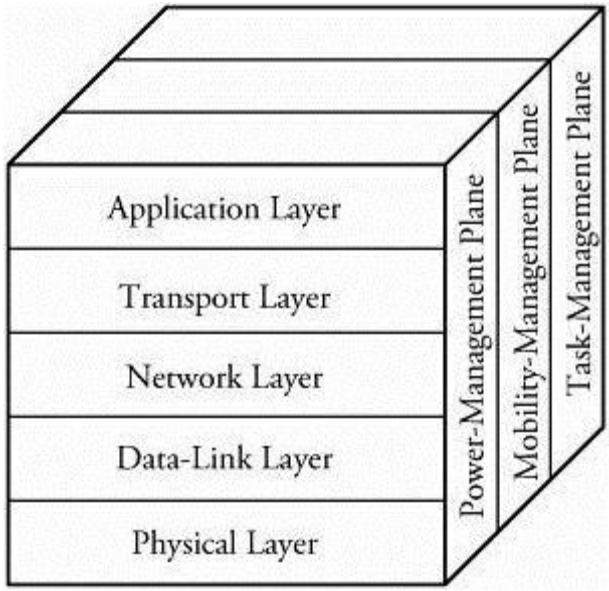
communication protocol development. In summary, efficient energy-conscious clustering and routing algorithms can potentially prolong the network lifetime.

8.4.2. Protocol Stack

The algorithms developed for wireless ad-hoc networks cannot be used for sensor networks, for several reasons. One is that the number of sensor nodes is typically much more than in a typical ad-hoc network, and sensor nodes, unlike ad-hoc nodes, are prone to permanent failure. In addition, sensor nodes normally use broadcast rather than point-to-point communication with its limited power and memory. Unlike computer networks, sensor nodes do not have global ID, since a typical packet overhead can be too large for them.

Figure 8.2 shows protocol architecture for sensor networks. The protocol stack combines power efficiency and least-cost-path routing. This protocol architecture integrates networking protocols and power through the wireless medium and promotes cooperative efforts of sensor nodes. The protocol stack consists of the physical layer, data-link layer, network layer, transport layer, and application layer, backed by a power-management plane, mobility-management plane, and task-management plane. The physical layer is responsible for robust modulation, transmission, and receiving signals. Media access control (MAC) at the data-link layer must minimize packet collision with neighboring nodes, as power is a restricted factor. The network layer routes packets provided by the transport layer. The application layer uses software for preparation of data on an event. The power-management plane monitors the sensor's power level among the sensor nodes and manages the amount of power a sensor node has used.

Figure 8.2. Sensor network protocol stack architecture



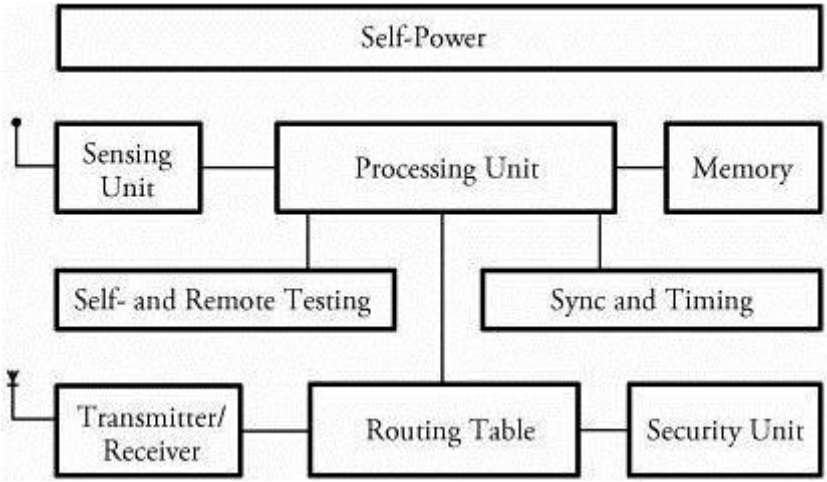
Most of the sensor network routing techniques and sensing tasks require an accurate knowledge of location. Thus, a sensor node commonly has a location-finding system. A mobilizer may sometimes be needed to move sensor nodes to carry out assigned tasks. Sensor network routing protocols must be capable of self-organizing. For these purposes, a series of energy-aware MAC, routing, and clustering protocols have been developed for wireless sensor networks. Most of the energy-aware MAC protocols aim to either adjust the transmission power or keep transceivers off as long as possible.

8.4.3. Sensor Node Structure

Figure 8.3 shows a typical sensor node. A node consists mainly of a sensing unit, a processing unit and memory, a self-power unit, and a wireless transceiver

component, as well as a self- and remote-testing unit, a synchronizing and timing unit, a routing table, and security units. Since nodes in a network are not physically accessible once they are deployed in the field, they are not worth being brought under test. An option is an on-board remote self-testing unit for the node on a routine basis.

Figure 8.3. A typical wireless sensor node



Each node must determine its location. This task is carried out by a location-finding system based on the global positioning system (GPS). All the processes within the sensor node are synchronized by a local clocking and synchronizing system. The communication and security protocol units are in fact part of the processing unit. These two units are responsible for computing the best path for networking and security of the data being transmitted. The three main blocks of the sensor nodesensing unit, processing and memory unit, and power unitare described in more detail in the following subsections.

Sensing Unit

The sensing unit consists of a sensor and an analog-to-digital converter. A smart sensor node consists of a combination of multiple sensors. The analog signals produced by the sensors, based on the observed event, are converted to digital signals by the converter and then fed into the processing unit. The sensing unit collects data externally and interacts with the central processor at the heart of the node.

Processing and Memory Unit

The processing unit performs certain computations on the data and, depending on how it is programmed, may send the resulting information out to the network. The processing unit, which is generally associated with memory, manages the procedures that make the sensor node collaborate with the other nodes to carry out the assigned sensing task. The central processor determines what data needs to be analyzed, stored, or compared with the data stored in memory. The streamed data from the sensor input is processed as it arrives. The database in memory stores an indexed data list to be used as a reference to detect an event. Since sensing nodes are typically tiny and many nodes are engaged in a network, the communication structure makes use of a hierarchically arranged self-routing network through cluster heads.

In smart wireless sensor networks, a tiny processor and a tiny database are used in a node. Thousands of such nodes are spread on fields to power up the sensing task, as in the deployment of numerous small intelligent sensor nodes in a battlefield to monitor enemy movements. By inserting self-organizing capability into a sensor network, a smart node can extract data, compare it with

the data stored in its memory database, and process it for relevance before relaying it to its central base station.

Self-Power Unit

A sensor node is supposed to be mounted in a small physical unit, limiting space for the battery. Moreover, the random distribution of sensors makes it impossible to periodically recharge or exchange batteries. In most types of sensor networks, the power unit in a sensor node is the most important unit of the node because the liveliness and existence of a node depend on the energy left in the node, and the routing in the sensor network is based on the algorithm that finds a path with the most energy. Thus, it is essential to use energy-efficient algorithms to prolong the life of sensor networks. The main task of the sensor node is to identify events, to process data, and then to transmit the data. The power of a node is consumed mainly in the transmitter and receiver unit. The sensor node can be supplied by a self-power unit, self-power unitbattery, or solar cells.

8.5 Communication Energy model:

IEEE standardsas 802.11a, b, and g provide a wide range of data rates: 54, 48, 36, 24, 18, 12, 9, and 6 Mb/s. This range reflects the trade-off between the transmission range and data rate intrinsic in a wireless communication channel. An accurate energy model is crucial for the development of energy-efficient clustering and routing protocols. The energy consumption, E, for all components of the transceiver in watts is summarized as

Equation 8.1

$$E = \theta + \eta\omega d^n,$$

where θ is the distance-independent term that accounts for the overhead of the radio electronics and digital processing, and $\eta\omega d^n$, is the distance-dependent term, in which η represents the amplifier inefficiency factor, ω is the free-space path loss, d is the distance, and n is the environmental factor. Based on an environmental condition, n can be a number between 2 and 4, and η specifies the inefficiency of the transmitter when generating maximum power ωd^n at the antenna. Clearly, the distance-dependent portion of total energy consumption depends on the real-world transceiver parameters, θ, η, and the path attenuation ωd^n. If the value of θ overshadows $\eta\omega d^n$, the reduction in the transmission distance through the use of multihop communication is not effective.

In theory, the maximum efficiency of a power amplifier is 48.4 percent. However, practical implementations show that the power-amplifier efficiency is less than 40 percent. Therefore, θ is calculated assuming that η = 1/0.4 = 2.5. Using Equation (8.1), we can express the energy consumption of a transmitter and a receiver, E_T and E_R, respectively, by

Equation 20.2

$$E_T = \theta_T + \eta\omega d^n$$

and

Equation 20.3

$$E_R = \theta_R,$$

where θ_T and θ_R are the distance-dependent terms for the transmitter and the receiver, respectively. Although maximum output power and total power consumption are provided in the manufacturer's data sheet, θ can be calculated the following formula:

Equation 20.4

$$\theta = \theta_{TX} + \theta_{RX} = (E_T - \eta\omega d^n) + E_R.$$

8.6 Clustering protocols:

Clustering protocols specify the topology of the hierarchical nonoverlapping clusters of sensor nodes. A robust clustering technique is essential for self-organizing sensor networks. An efficient clustering protocol ensures the creation of clusters with almost the same radius and cluster heads that are best positioned in the clusters. Since every node in a clustered network is connected to a cluster head, route discovery among cluster heads is sufficient to establish a feasible route in the network. For a large sensor network, clustering can simplify multihop route discovery and limit the number of transmissions compared to a flat, nonclustered network.

8.6.1. Classification of Clustering Protocols

Clustering techniques can be either centralized or decentralized. Centralized clustering algorithms require each sensor node to send its individual information, such as energy level and geographical position, to the central base station. Based on a predefined algorithm, a base station calculates the number of clusters, their sizes, and the cluster heads' positions and then provides each node with its newly

assigned duty.

Given the assumption that sensor networks might consist of thousands of nodes, it is impractical, if not impossible, for a base station to collect information about every node in the network prior to route setup. Therefore, centralized clustering is not an option for large sensor networks. Since a sensor node begins a clustering process without any knowledge about its location relative to the corresponding base station, a clustering algorithm should be able to form clusters without the help of the base station and knowledge of node positioning. Although location-finder devices can also be deployed to perform this task, they are often either costly or add too much overhead on the network.

Decentralized clustering techniques create clusters without the help of any centralized base station. An energy-efficient and hierarchical clustering algorithm can be such a way whereby each sensor node becomes a cluster head with a probability of p and advertises its candidacy to nodes that are no more than k hops away from the cluster head. Given the limited transmission range of wireless sensor nodes, a hierarchical structure with an arbitrary number of levels has its limitations. As the number of hierarchical levels grows, the distance between upper-level cluster heads may increase to the point that they are no longer able to communicate with one another. The Low-Energy Adaptive Clustering Hierarchy (LEACH) algorithm and the Decentralized Energy-Efficient Cluster Propagation (DEEP) protocol are two examples of the decentralized clustering protocols and are explained next.

8.6.2. LEACH Clustering Protocol

The Low-Energy Adaptive Clustering Hierarchy (LEACH) protocol is a decentralized clustering algorithm that does not offer a complete energy-optimization solution, as it has no strategy for specifying cluster-head positioning and distribution. LEACH is an application-specific protocol architecture that aims to prolong network lifetime by periodic reclustering and change of the network topology.

LEACH is divided into rounds consisting of a clustering phase and a steady-state phase for data collection. At the start of each round, a sensor node randomly chooses a number between 0 and 1 and then compares this number to a calculated threshold called T(n). If T(n) is larger than the chosen number, the node becomes a cluster head for the current round. The value T(n) is calculated using the following formula:

Equation 20.11

$$T(n) = \begin{cases} \frac{p}{1-p(r \bmod (1/p))} & \text{for } n \in G \\ 0 & \text{otherwise} \end{cases},$$

where p is the ratio of the total number of cluster heads to the total number of nodes, r is the number of rounds, and G is a set of nodes that have not been chosen as cluster heads for the last 1/p rounds. For the first round (r=0), T(n) is equal to p, and nodes have an equal chance to become cluster head. As r gets closer to 1/p, T(n) increases, and nodes that have not been selected as cluster head in the last 1/p rounds have more chance to become cluster head. After 1/p - 1 rounds, T(n) is equal to 1, meaning that all the remaining nodes have been

selected as cluster head. Thus, after 1/p rounds, all the nodes have had a chance to become a cluster head once. Since being the cluster head puts a substantial burden on the sensor nodes, this ensures that the network has no overloaded node that runs out of energy sooner than the others.

After cluster heads are self-selected, they start to advertise their candidacy to the other sensor nodes. When it receives advertisements from more than one cluster-head candidate, a sensor node starts to make a decision about its corresponding cluster head. Each node listens to the advertisement signals and chooses the candidate whose associated signal is received with higher power. This ensures that each sensor node chooses the closest candidate as cluster head. The LEACH algorithm is distributed, as it can be accomplished by local computation and communication at each node, rather than the transmission of all the nodes' energy level and geographical position to a centralized point. However, cluster heads are chosen randomly, and there is no optimization in terms of energy consumption.

8.7 Routing protocols:

After clusters with well-distributed cluster heads have been established in a network, energy-conscious routing is essential in order to set communication routes among cluster heads in a two-level hierarchical system. Similar to computer networks, routing protocols in sensor networks can be classified as either intracluster or intercluster. This section looks at both categories.

The fundamental concept behind them is much the same as the concept behind intradomain and interdomain routings Assuming that every node in a cluster can

act as a relay node, there could be a large number of possible routes from a source to a sink. Because of the limited transmission range associated with low-power wireless technologies cluster-head packets cannot reach the base station unless other cluster heads act as relay nodes. Two major approaches can be used for routing and selecting the best path in a sensor network, as follows:

1. Centralized routing, whereby the routing decision is made by a single command center
2. Distributed routing, whereby the routing decision is made in a distributed fashion by several entities

Distributed routing algorithms are further classified as proactive or reactive. With proactive routing algorithms, such as link-state routing and distance-vector routing, nodes keep a routing table that contains next-hop information to every node in the network. Reactive routing protocols set a route to the desirable destination only when it is needed. Note that none of the ad hoc network protocols explained earlier considers energy consumption.

Another group of on-demand reactive routing protocols address the exclusive issues of wireless sensor network. For example, directed diffusion introduces a concept of "interest" propagation whenever a node wants to send data or a source needs to ask for it. With this type of protocol, flooding the network with interest signals establishes a path from a sink to every possible source (spanning tree).

8.7.1. Intracluster Routing Protocols

A routing algorithm within a cluster can be either direct or multihop. In a direct routing algorithm, the cluster head as the destination for all cluster nodes is located in the center of the cluster, so all nodes can communicate with the cluster

head directly, as shown in Figure 8.8. Note that in this figure, two nodes cannot reach the destination, as they are located far from it. The number shown in each node indicates the level of energy the corresponding node has.

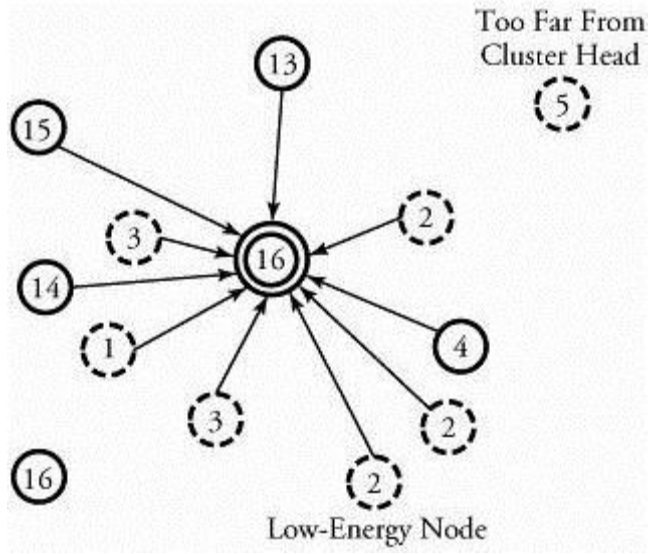
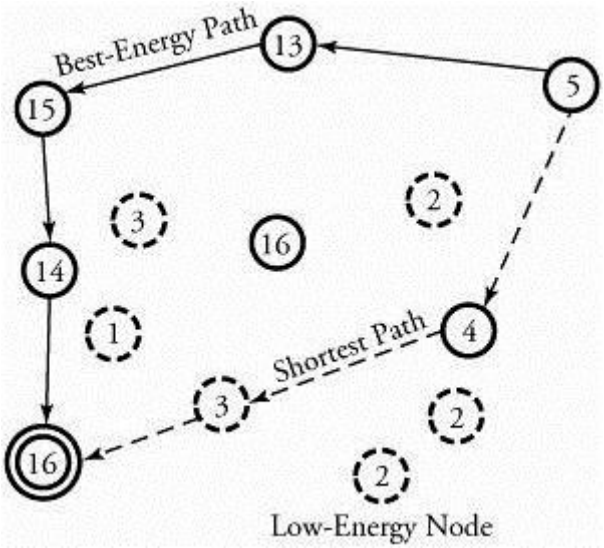


Figure 8.8. Direct routing in a cluster. The number associated with each node indicates a normalized value of the remaining energy in that node.

In a multihop routing algorithm, a node can face multiple hops in order to reach the destination. If a multihop algorithm is used for the centralized clustering procedure, the algorithm aims to choose the appropriate next neighbor for each node, using a central command node. Typically, a central command node collects the information about direct paths' costs and geographical positions of the nodes and finds the best path.

Figure 8.9 shows a routing implementation. Sensor nodes are usually scattered in the field. A packet from a node is routed to a neighboring node that exhibits the

highest amount of energy. The energy is an indication of the node's battery level. The number associated with each node indicates a normalized value of the remaining energy in that node. Figure 8.9 shows two paths from a node to a cluster-head node. One path involves the shortest distance in terms of hop counts; the other one uses the highest-energy route. The challenge here is to find the best path that suits the rapid and secure deployment of data. Data is routed to the cluster head as shown in Figure 8.1, where the cluster head may communicate with the base station via radio frequencies.



8.8 ZigBee technology and 802.15.4:

The ZigBee technology is a communication standard that provides a short-range low-cost networking capability that allows low-cost devices to quickly transmit small amounts of data, such as temperature readings for thermostats, on/off requests for light switches, or keystrokes for a wireless keyboard. Other ZigBee applications are in professional installation kits for lighting controls, heating, ventilation, air conditioning, and security. Even the Bluetooth short-range wireless technology found in laptops and cellphones lacks the affordability, power savings, and mesh-networking capabilities of ZigBee.

ZigBee comes from higher-layer enhancements by a multivendor consortium called the Zigbee Alliance. IEEE standard 802.15.4/ZigBee specifies the MAC and physical layers. The 802.15.4 standard specifies 128-bit AES encryption; ZigBee specifies how to handle encryption key exchange. The 802.15.4/ZigBee networks run in the unlicensed frequencies, 900 MHz and 2.4 GHz band, based on a packet radio standard and support many cordless telephones, allowing data to be sent over distances up to 20 meters.

ZigBee devices, typically battery powered, can transmit information much farther than 20 meters, because each device within listening distance passes the message along to any other device within range. Only the intended device acts on the message. By instructing nodes to wake up only for those split-second intervals when they are needed, ZigBee device batteries might last for years. Although this technology is targeting for manufacturing, health care, shipping, and homeland defense, the ZigBee Alliance is initially keeping its focus small.