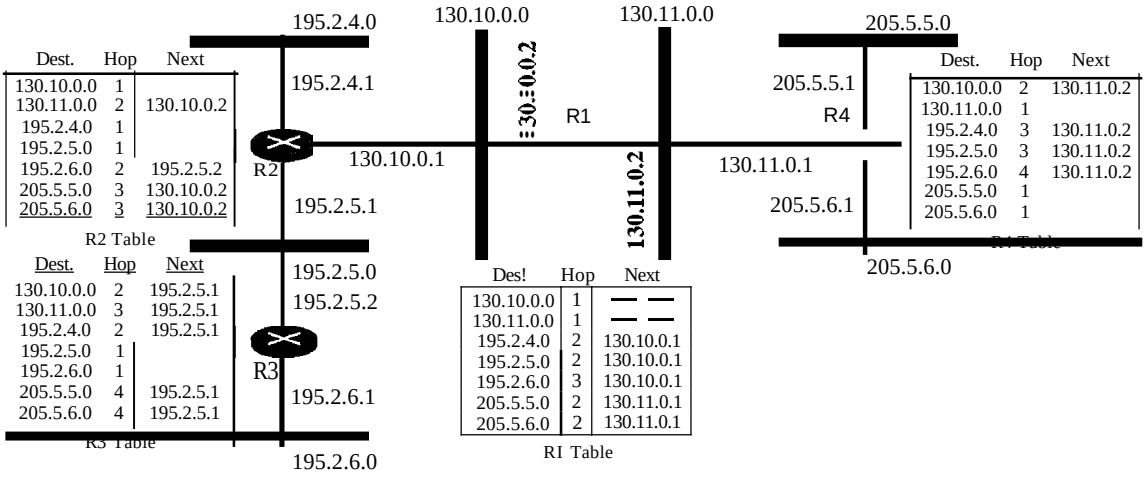
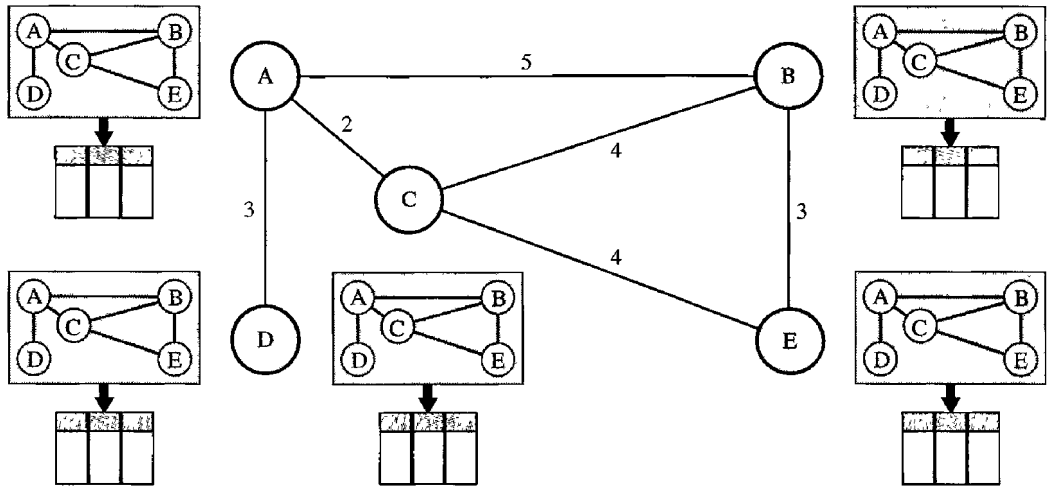
Figure 22.19 *Example of a domain using RIP*



Link State Routing

Link state routing has a different philosophy from that of distance vector routing. In link state routing, if each node in the domain has the entire topology of the domain-the list of nodes and links, how they are connected including the type, cost (metric), and condition of the links (up or down)-the node can use Dijkstra's algorithm to build a routing table. Figure 22.20 shows the concept.

Figure 22.20 *Concept of link state routing*

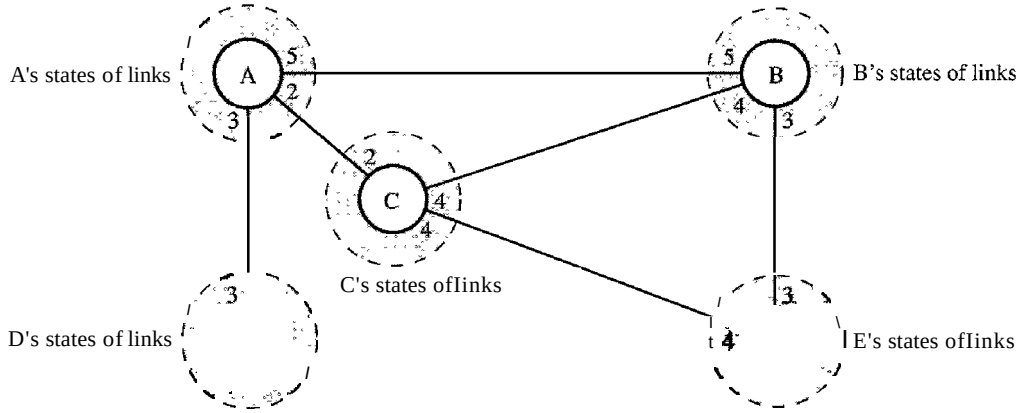


The figure shows a simple domain with five nodes. Each node uses the same topology to create a routing table, but the routing table for each node is unique because the calculations are based on different interpretations of the topology. This is analogous to a city map. While each person may have the same map, each needs to take a different route to reach her specific destination.

The topology must be dynamic, representing the latest state of each node and each link. If there are changes in any point in the network (a link is down, for example), the topology must be updated for each node.

How can a common topology be dynamic and stored in each node? No node can know the topology at the beginning or after a change somewhere in the network. Link state routing is based on the assumption that, although the global knowledge about the topology is not clear, each node has partial knowledge: it knows the state (type, condition, and cost) of its links. In other words, the whole topology can be compiled from the partial knowledge of each node. Figure 22.21 shows the same domain as in Figure 22.20, indicating the part of the knowledge belonging to each node.

Figure 22.21 *Link state knowledge*



Node A knows that it is connected to node B with metric 5, to node C with metric 2, and to node D with metric 3. Node C knows that it is connected to node A with metric 2, to node B with metric 4, and to node E with metric 4. Node D knows that it is connected only to node A with metric 3. And so on. Although there is an overlap in the knowledge, the overlap guarantees the creation of a common topology-a picture of the whole domain for each node.

Building Routing Tables

In link state routing, four sets of actions are required to ensure that each node has the routing table showing the least-cost node to every other node.

1. Creation of the states of the links by each node, called the link state packet (LSP).
2. Dissemination of LSPs to every other router, called **flooding,** in an efficient and reliable way.
3. Formation of a shortest path tree for each node.
4. Calculation of a routing table based on the shortest path tree.

Creation of Link State Packet (LSP) A link state packet can carry a large amount of information. For the moment, however, we assume that it carries a minimum amount

of data: the node identity, the list of links, a sequence number, and age. The first two, node identity and the list of links, are needed to make the topology. The third, sequence number, facilitates flooding and distinguishes new LSPs from old ones. The fourth, age, prevents old LSPs from remaining in the domain for a long time. LSPs are generated on two occasions:

1. *When there is a change in the topology of the domain.* Triggering of LSP dissemination is the main way of quickly informing any node in the domain to update its topology.
2. *On a periodic basis.* The period in this case is much longer compared to distance vector routing. As a matter of fact, there is no actual need for this type of LSP dissemination. It is done to ensure that old information is removed from the domain. The timer set for periodic dissemination is normally in the range of 60 min or 2 h based on the implementation. A longer period ensures that flooding does not create too much traffic on the network.

Flooding of LSPs After a node has prepared an LSP, it must be disseminated to all other nodes, not only to its neighbors. The process is called flooding and based on the following:

1. The creating node sends a copy of the LSP out of each interface.
2. A node that receives an LSP compares it with the copy it may already have. If the newly arrived LSP is older than the one it has (found by checking the sequence number), it discards the LSP. If it is newer, the node does the following:
   a. It discards the old LSP and keeps the new one.
   b. It sends a copy of it out of each interface except the one from which the packet arrived. This guarantees that flooding stops somewhere in the domain (where a node has only one interface).

Formation of Shortest Path Tree: Dijkstra Algorithm After receiving all LSPs, each node will have a copy of the whole topology. However, the topology is not sufficient to find the shortest path to every other node; a shortest path tree is needed.

A tree is a graph of nodes and links; one node is called the root. All other nodes can be reached from the root through only one single route. A shortest path tree is a tree in which the path between the root and every other node is the shortest. What we need for each node is a shortest path tree with that node as the root.

The Dijkstra algorithm creates a shortest path tree from a graph. The algorithm divides the nodes into two sets: tentative and permanent. It finds the neighbors of a current node, makes them tentative, examines them, and if they pass the criteria, makes them permanent. We can informally define the algorithm by using the flowchart in Figure 22.22.

Let us apply the algorithm to node A of our sample graph in Figure 22.23. To find the shortest path in each step, we need the cumulative cost from the root to each node, which is shown next to the node.

The following shows the steps. At the end of each step, we show the permanent (filled circles) and the tentative (open circles) nodes and lists with the cumulative costs.

Figure 22.22 *Dijkstra algorithm*

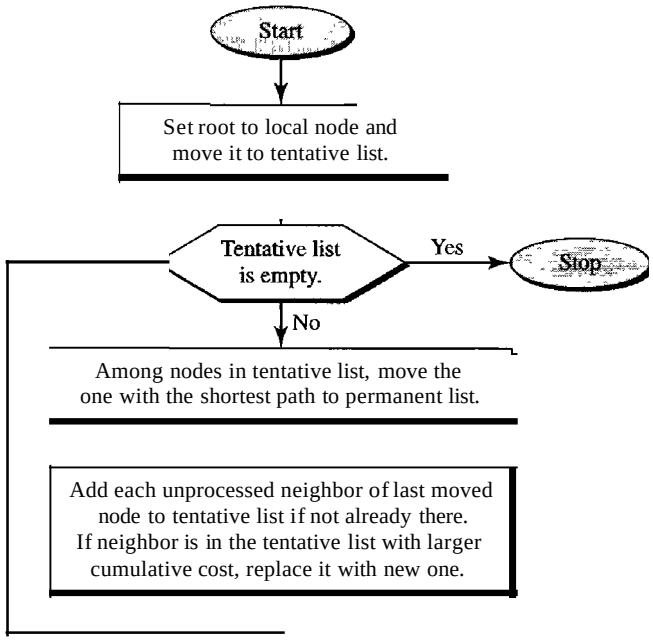
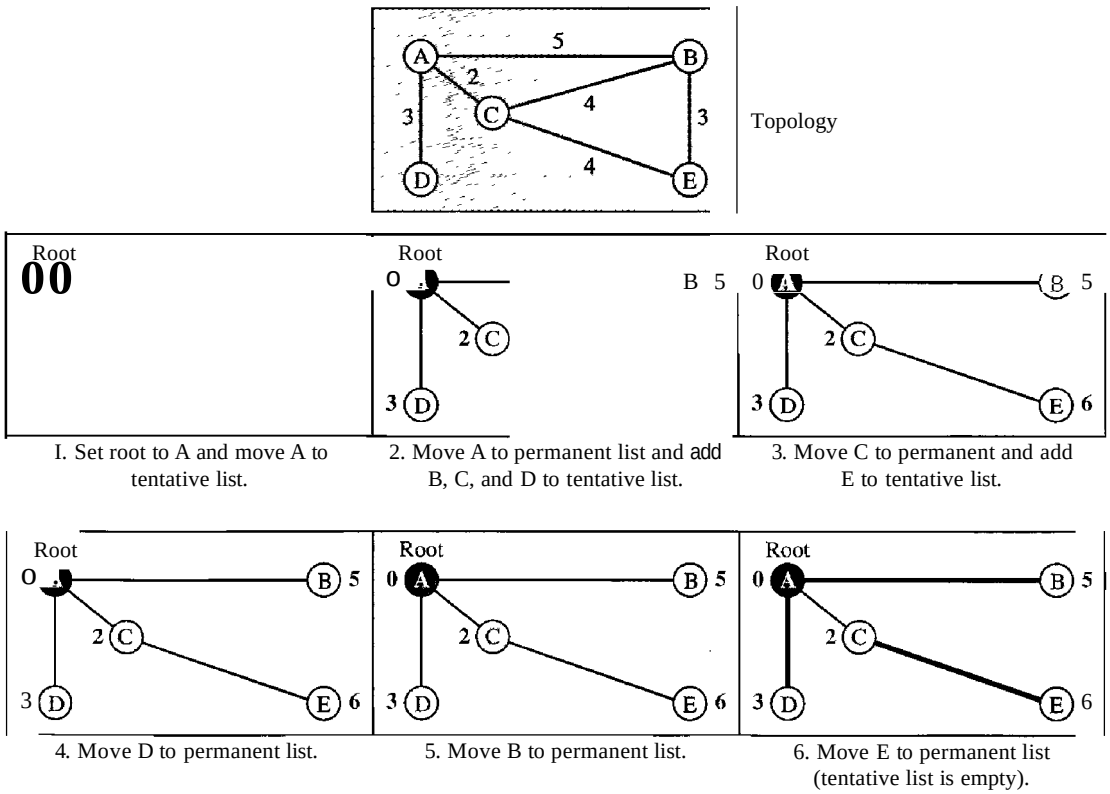


Figure 22.23 *Example of formation of shortest path tree*



1. We make node A the root of the tree and move it to the tentative list. Our two lists are

   Permanent list: empty Tentative list: A(O)

2. Node A has the shortest cumulative cost from all nodes in the tentative list. We move A to the pennanent list and add all neighbors of A to the tentative list. Our new lists are

   Permanent list: A(O) Tentative list: B(5), C(2), D(3)

3. Node C has the shortest cumulative cost from all nodes in the tentative list. We move C to the permanent list. Node C has three neighbors, but node A is already processed, which makes the unprocessed neighbors just B and E. However, B is already in the tentative list with a cumulative cost of 5. Node A could also reach node B through C with a cumulative cost of 6. Since 5 is less than 6, we keep node B with a cumulative cost of 5 in the tentative list and do not replace it. Our new lists are

   Permanent list: A(O), e(2) Tentative list: B(5), 0(3), E(6)

4. Node D has the shortest cumulative cost of all the nodes in the tentative list. We move D to the permanent list. Node D has no unprocessed neighbor to be added to the tentative list. Our new lists are

   Permanent list: A(O), C(2), 0(3) Tentative list: B(5), E(6)

5. Node B has the shortest cumulative cost of all the nodes in the tentative list. We move B to the permanent list. We need to add all unprocessed neighbors of B to the tentative list (this is just node E). However, E(6) is already in the list with a smaller cumulative cost. The cumulative cost to node E, as the neighbor of B, is 8. We keep node E(6) in the tentative list. Our new lists are

   Permanent list: A(O), B(5), C(2), 0(3) Tentative list: E(6)

6. Node E has the shortest cumulative cost from all nodes in the tentative list. We move E to the permanent list. Node E has no neighbor. Now the tentative list is empty. We stop; our shortest path tree is ready. The final lists are

   Permanent list: A(0), B(5), C(2), D(3), E(6) Tentative list: empty

Calculation of Routing Table from Shortest Path Tree Each node uses the shortest path tree protocol to construct its routing table. The routing table shows the cost of reaching each node from the root. Table 22.2 shows the routing table for node A.

Table 22.2 *Routing table for node A*

Node	*Cost*	*Next Router*
A	0	-
B	5	-
C	2	-
D	3	-
E	6	C

Compare Table 22.2 with the one in Figure 22.14. Both distance vector routing and link state routing end up with the same routing table for node A.

OSPF

The Open Shortest Path First or OSPF protocol is an intradomain routing protocol based on link state routing. Its domain is also an autonomous system.

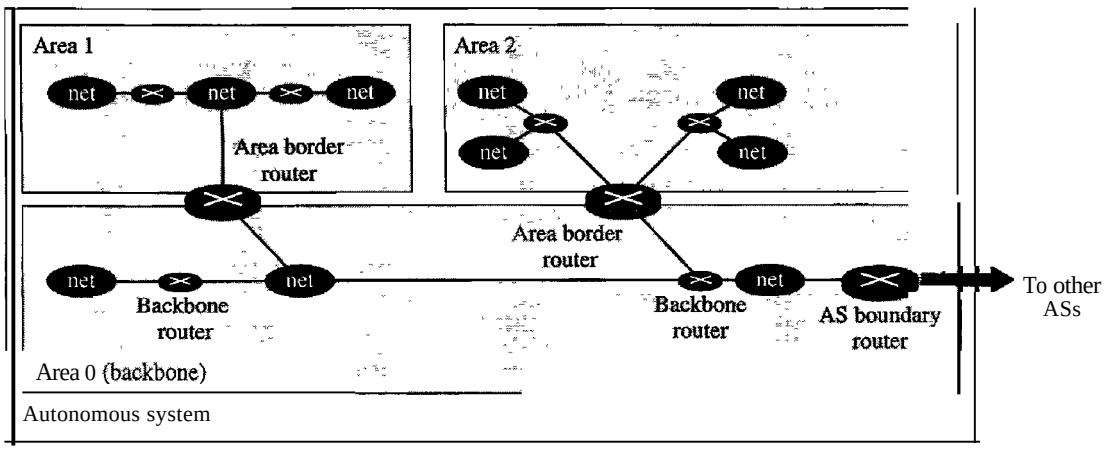
Areas To handle routing efficiently and in a timely manner, OSPF divides an autonomous system into areas. An area is a collection of networks, hosts, and routers all contained within an autonomous system. An autonomous system can be divided into many different areas. All networks inside an area must be connected.

Routers inside an area flood the area with routing information. At the border of an area, special routers called area border routers summarize the information about the area and send it to other areas. Among the areas inside an autonomous system is a special area called the *backbone;* all the areas inside an autonomous system must be connected to the backbone. In other words, the backbone serves as a primary area and the other areas as secondary areas. This does not mean that the routers within areas cannot be connected to each other, however. The routers inside the backbone are called the backbone routers. Note that a backbone router can also be an area border router.

If, because of some problem, the connectivity between a backbone and an area is broken, a virtual link between routers must be created by an administrator to allow continuity of the functions of the backbone as the primary area.

Each area has an area identification. The area identification of the backbone is zero. Figure 22.24 shows an autonomous system and its areas.

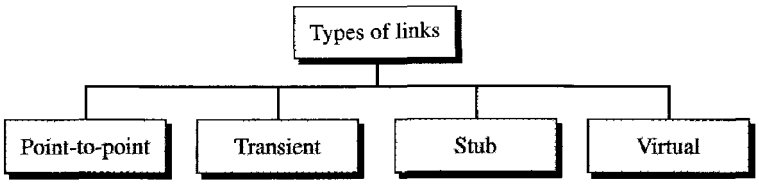
Figure 22.24 *Areas in an autonomous system*



Metric The OSPF protocol allows the administrator to assign a cost, called the metric, to each route. The metric can be based on a type of service (minimum delay, maximum throughput, and so on). As a matter of fact, a router can have multiple routing tables, each based on a different type of service.

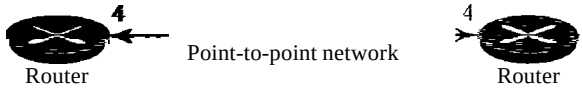
Types of Links In OSPF terminology, a connection is called a *link.* Four types of links have been defined: point-to-point, transient, stub, and virtual (see Figure 22.25).

Figure 22.25 *Types of links*



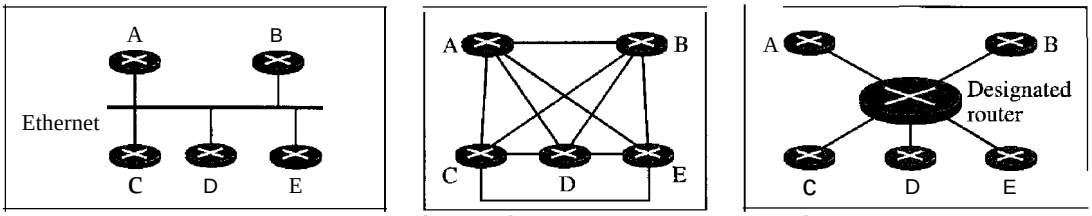
A point-to-point link connects two routers without any other host or router in between. In other words, the purpose of the link (network) is just to connect the two routers. An example of this type of link is two routers connected by a telephone line or a T line. There is no need to assign a network address to this type of link. Graphically, the routers are represented by nodes, and the link is represented by a bidirectional edge connecting the nodes. The metrics, which are usually the same, are shown at the two ends, one for each direction. In other words, each router has only one neighbor at the other side of the link (see Figure 22.26).

Figure 22.26 *Point-to-point link*



A transient link is a network with several routers attached to it. The data can enter through any of the routers and leave through any router. All LANs and some WANs with two or more routers are of this type. In this case, each router has many neighbors. For example, consider the Ethernet in Figure 22.27a. Router A has routers B, C, D, and E as neighbors. Router B has routers A, C, D, and E as neighbors. If we want to show the neighborhood relationship in this situation, we have the graph shown in Figure 22.27b.

Figure 22.27 *Transient link*



a. Transient network b. Unrealistic representation c. Realistic representation

This is neither efficient nor realistic. It is not efficient because each router needs to advertise the neighborhood to four other routers, for a total of 20 advertisements. It is

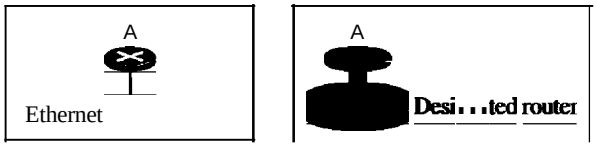
not realistic because there is no single network (link) between each pair of routers; there is only one network that serves as a crossroad between all five routers.

To show that each router is connected to every other router through one single network, the network itself is represented by a node. However, because a network is not a machine, it cannot function as a router. One of the routers in the network takes this responsibility. It is assigned a dual purpose; it is a true router and a designated router. We can use the topology shown in Figure 22.27c to show the connections of a transient network.

Now each router has only one neighbor, the designated router (network). On the other hand, the designated router (the network) has five neighbors. We see that thc number of neighbor announcements is reduced from 20 to 10. Still, the link is represented as a bidirectional edge between the nodes. However, while there is a metric from each node to the designated router, there is no metric from the designated router to any other node. The reason is that the designated router represents the network. We can only assign a cost to a packet that is passing through the network. We cannot charge for this twice. When a packet enters a network, we assign a cost; when a packet leaves the network to go to the router, there is no charge.

A **stub link** is a network that is connected to only one router. The data packets enter the network through this single router and leave the network through this same router. This is a special case of the transient network. We can show this situation using the router as a node and using the designated router for the network. However, the link is only one-directional, from the router to the network (see Figure 22.28).

Figure 22.28 *Stub link*



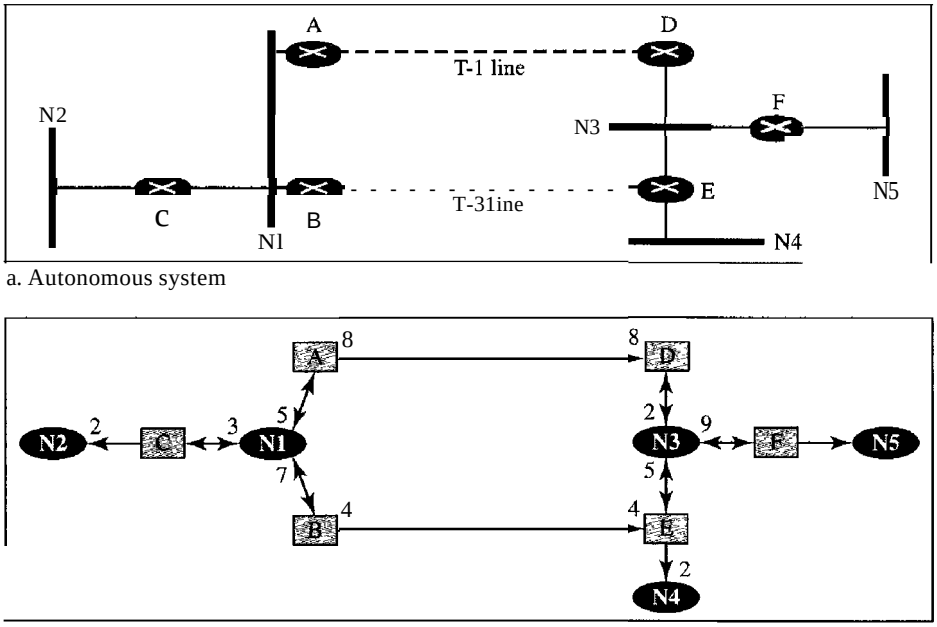
a. Stub network b. Representation

When the link between two routers is broken, the administration may create a **virtual link** between them, using a longer path that probably goes through several routers.

Graphical Representation Let us now examine how an AS can be represented graphically. Figure 22.29 shows a small AS with seven networks and six routers. Two of the networks are point-to-point networks. We use symbols such as Nl and N2 for transient and stub networks. There is no need to assign an identity to a point-to-point network. The figure also shows the graphical representation of the AS as seen by OSPF.

We have used square nodes for the routers and ovals for the networks (represented by designated routers). However, OSPF sees both as nodes. Note that we have three stub networks.

Figure 22.29 *Example of an AS and its graphical representation in OSPF*



a. Autonomous system

b. Graphical representation

Path Vector Routing

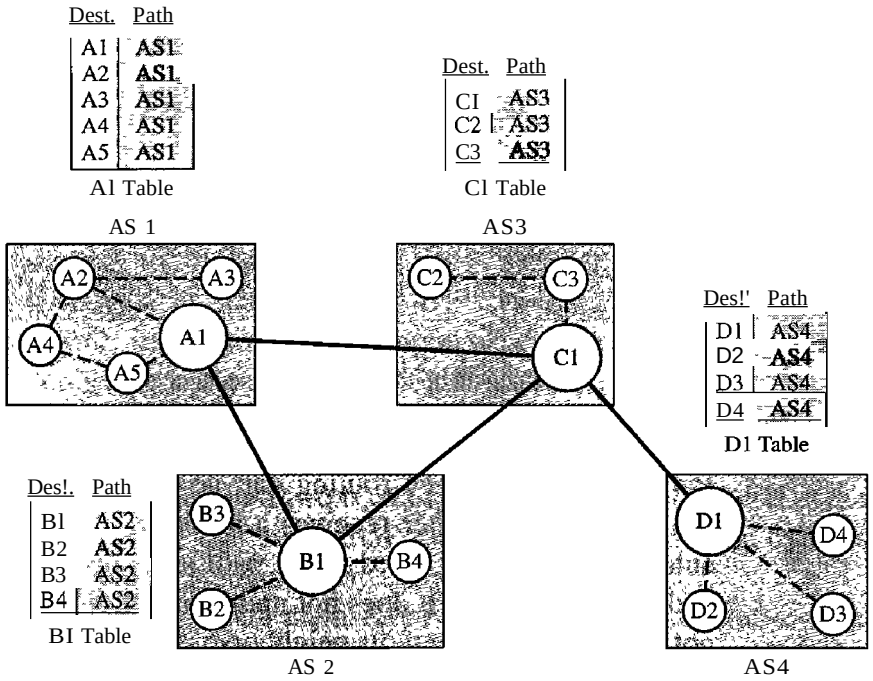
Distance vector and link state routing are both intradomain routing protocols. They can be used inside an autonomous system, but not between autonomous systems. These two protocols are not suitable for interdomain routing mostly because of scalability. Both of these routing protocols become intractable when the domain of operation becomes large. Distance vector routing is subject to instability if there are more than a few hops in the domain of operation. Link state routing needs a huge amount of resources to calculate routing tables. It also creates heavy traffic because of flooding. There is a need for a third routing protocol which we call path vector routing.

Path vector routing proved to be useful for interdomain routing. The principle of path vector routing is similar to that of distance vector routing. In path vector routing, we assume that there is one node (there can be more, but one is enough for our conceptual discussion) in each autonomous system that acts on behalf of the entire autonomous system. Let us call it the speaker node. The speaker node in an AS creates a routing table and advertises it to speaker nodes in the neighboring ASs. The idea is the same as for distance vector routing except that only speaker nodes in each AS can communicate with each other. However, what is advertised is different. A speaker node advertises the path, not the metric of the nodes, in its autonomous system or other autonomous systems.

Initialization

At the beginning, each speaker node can know only the reachability of nodes inside its autonomous system. Figure 22.30 shows the initial tables for each speaker node in a system made of four ASs.

Figure 22.30 *Initial routing tables in path vector routing*



Node A1 is the speaker node for AS1, B1 for AS2, C1 for AS3, and D1 for AS4. Node A1 creates an initial table that shows A1 to A5 are located in AS1 and can be reached through it. Node B1 advertises that B1 to B4 are located in AS2 and can be reached through B1. And so on.

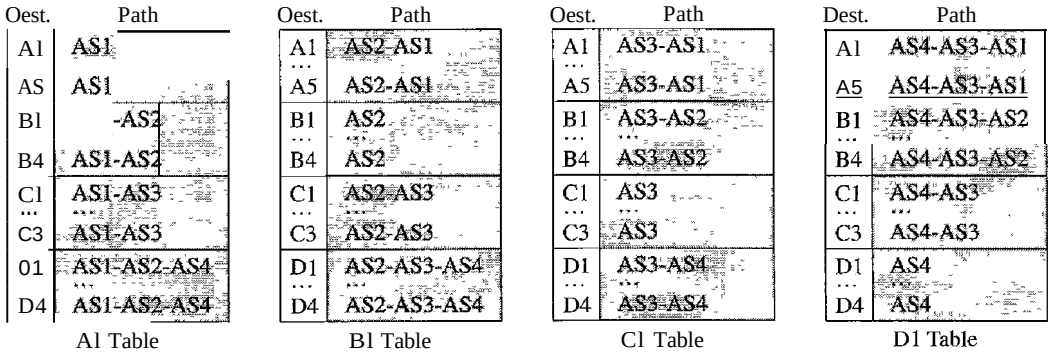
Sharing Just as in distance vector routing, in path vector routing, a speaker in an autonomous system shares its table with immediate neighbors. In Figure 22.30, node A1 shares its table with nodes B1 and C1. Node C1 shares its table with nodes D1, B1, and A1. Node B1 shares its table with C1 and A1. Node D1 shares its table with C1.

Updating When a speaker node receives a two-column table from a neighbor, it updates its own table by adding the nodes that are not in its routing table and adding its own autonomous system and the autonomous system that sent the table. After a while each speaker has a table and knows how to reach each node in other ASs. Figure 22.31 shows the tables for each speaker node after the system is stabilized.

According to the figure, if router A1 receives a packet for nodes A3, it knows that the path is in AS1 (the packet is at home); but if it receives a packet for D1, it knows that the packet should go from AS1, to AS2, and then to AS3. The routing table shows the path completely. On the other hand, if node D1 in AS4 receives a packet for node A2, it knows it should go through AS4, AS3, and AS 1.

- Loop prevention. The instability of distance vector routing and the creation of loops can be avoided in path vector routing. When a router receives a message, it checks to see if its autonomous system is in the path list to the destination. If it is, looping is involved and the message is ignored.

Figure 22.31 *Stabilized tables for three autonomous systems*



- **Policy routing.** Policy routing can be easily implemented through path vector routing. When a router receives a message, it can check the path. If one of the autonomous systems listed in the path is against its policy, it can ignore that path and that destination. It does not update its routing table with this path, and it does not send this message to its neighbors.
- **Optimum path.** What is the optimum path in path vector routing? We are looking for a path to a destination that is the best for the organization that runs the autonomous system. We definitely cannot include metrics in this route because each autonomous system that is included in the path may use a different criterion for the metric. One system may use, internally, RIP, which defines hop count as the metric; another may use OSPF with minimum delay defined as the metric. The optimum path is the path that fits the organization. In our previous figure, each autonomous system may have more than one path to a destination. For example, a path from AS4 to ASI can be AS4-AS3-AS2-AS1, or it can be AS4-AS3-ASI. For the tables, we chose the one that had the smaller number of autonomous systems, but this is not always the case. Other criteria, such as security, safety, and reliability, can also be applied.

BGP

Border Gateway Protocol (BGP) is an interdomain routing protocol using path vector routing. It first appeared in 1989 and has gone through four versions.

Types **of Autonomous** Systems As we said before, the Internet is divided into hierarchical domains called autonomous systems. For example, a large corporation that manages its own network and has full control over it is an autonomous system. A local ISP that provides services to local customers is an autonomous system. We can divide autonomous systems into three categories: stub, multihomed, and transit.

- **Stub** AS. A stub AS has only one connection to another AS. The interdomain data traffic in a stub AS can be either created or terminated in the AS. The hosts in the AS can send data traffic to other ASs. The hosts in the AS can receive data coming from hosts in other ASs. Data traffic, however, cannot pass through a stub AS. A stub AS

is either a source or a sink. A good example of a stub AS is a small corporation or a small local ISP.

- Multihomed AS. A multihomed AS has more than one connection to other ASs, but it is still only a source or sink for data traffic. It can receive data traffic from more than one AS. It can send data traffic to more than one AS, but there is no transient traffic. It does not allow data coming from one AS and going to another AS to pass through. A good example of a multihomed AS is a large corporation that is connected to more than one regional or national AS that does not allow transient traffic.
- Transit AS. A transit AS is a multihomed AS that also allows transient traffic. Good examples of transit ASs are national and international ISPs (Internet backbones).

Path Attributes In our previous example, we discussed a path for a destination network. The path was presented as a list of autonomous systems, but is, in fact, a list of attributes. Each attribute gives some information about the path. The list of attributes helps the receiving router make a more-informed decision when applying its policy.

Attributes are divided into two broad categories: well known and optional. A well-known attribute is one that every BGP router must recognize. An optional attribute is one that needs not be recognized by every router.

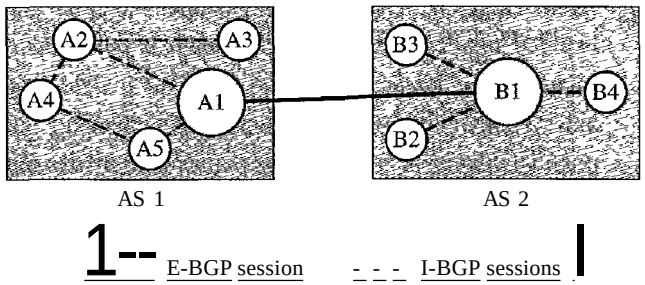
Well-known attributes are themselves divided into two categories: mandatory and discretionary. A *well-known mandatory attribute* is one that must appear in the description of a route. A *well-known discretionary attribute* is one that must be recognized by each router, but is not required to be included in every update message. One well-known mandatory attribute is ORIGIN. This defines the source of the routing information (RIP, OSPF, and so on). Another well-known mandatory attribute is AS_PATH. This defines the list of autonomous systems through which the destination can be reached. Still another well-known mandatory attribute is NEXT-HOP, which defines the next router to which the data packet should be sent.

The optional attributes can also be subdivided into two categories: transitive and nontransitive. An *optional transitive attribute* is one that must be passed to the next router by the router that has not implemented this attribute. An *optional nontransitive attribute* is one that must be discarded if the receiving router has not implemented it.

BGP Sessions The exchange of routing information between two routers using BGP takes place in a session. A session is a connection that is established between two BGP routers only for the sake of exchanging routing information. To create a reliable environment, BGP uses the services of TCP. In other words, a session at the BGP level, as an application program, is a connection at the TCP level. However, there is a subtle difference between a connection in TCP made for BGP and other application programs. When a TCP connection is created for BGP, it can last for a long time, until something unusual happens. For this reason, BGP sessions are sometimes referred to as *semi-pennanent connections.*

External and Internal BGP If we want to be precise, BGP can have two types of sessions: external BGP (E-BGP) and internal BGP (I-BGP) sessions. The E-BGP session is used to exchange information between two speaker nodes belonging to two different autonomous systems. The I-BGP session, on the other hand, is used to exchange routing information between two routers inside an autonomous system. Figure 22.32 shows the idea.

Figure 22.32 *Internal and external BGP sessions*



The session established between AS 1 and AS2 is an E-BOP session. The two speaker routers exchange information they know about networks in the Internet. However, these two routers need to collect information from other routers in the autonomous systems. This is done using I-BOP sessions.

22.4 MULTICAST ROUTING PROTOCOLS

In this section, we discuss multicasting and multicast routing protocols. We first define the term *multicasting* and compare it to unicasting and broadcasting. We also briefly discuss the applications of multicasting. Finally, we move on to multicast routing and the general ideas and goals related to it. We also discuss some common multicast routing protocols used in the Internet today.

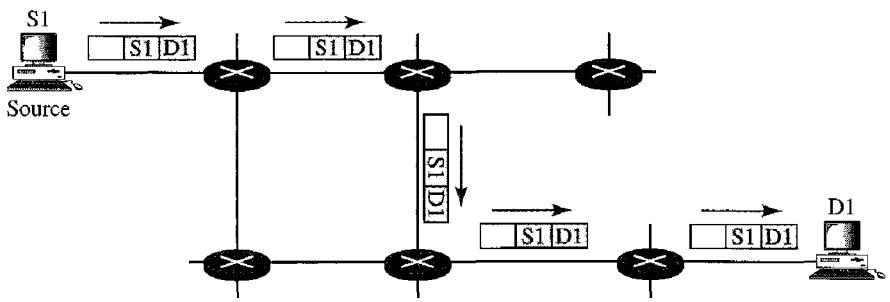
Unicast, Multicast, **and** Broadcast

A message can be unicast, multicast, or broadcast. Let us clarify these terms as they relate to the Internet.

Unicasting

In unicast communication, there is one source and one destination. The relationship between the source and the destination is one-to-one. In this type of communication, both the source and destination addresses, in the IP datagram, are the unicast addresses assigned to the hosts (or host interfaces, to be more exact). In Figure 22.33, a unicast

Figure 22.33 *Unicasting*



packet starts from the source S1 and passes through routers to reach the destination D1. We have shown the networks as a link between the routers to simplify the figure.

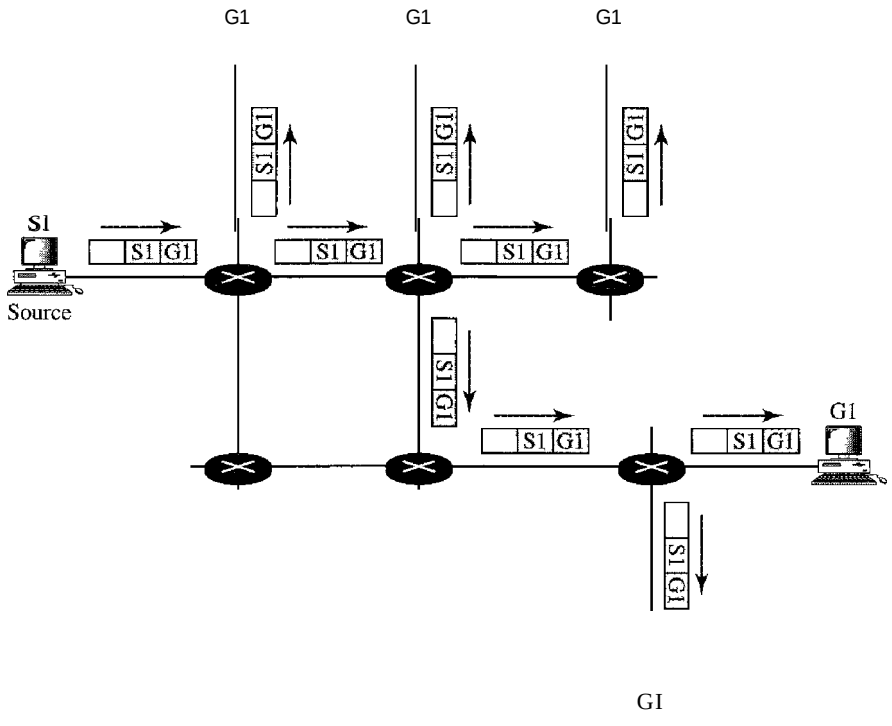
Note that in unicasting, when a router receives a packet, it forwards the packet through only one of its interfaces (the one belonging to the optimum path) as defined in the routing table. The router may discard the packet if it cannot find the destination address in its routing table.

In unicasting, the router forwards the received packet through only one of its interfaces.

Multicasting

In multicast communication, there is one source and a group of destinations. The relationship is one-to-many. In this type of communication, the source address is a unicast address, but the destination address is a group address, which defines one or more destinations. The group address identifies the members of the group. Figure 22.34 shows the idea behind multicasting.

Figure 22.34 *Multicasting*



A multicast packet starts from the source S1 and goes to all destinations that belong to group G1. In multicasting, when a router receives a packet, it may forward it through several of its interfaces.

In multicasting, the router may forward the received packet through several of its interfaces.

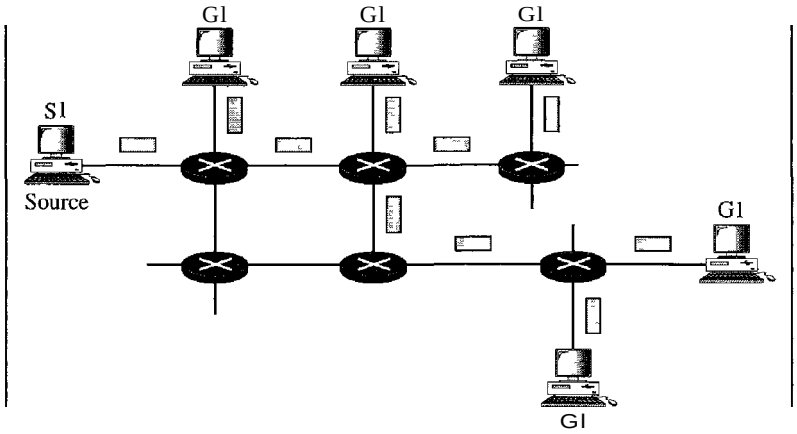
Broadcasting

In broadcast communication, the relationship between the source and the destination is one-to-all. There is only one source, but all the other hosts are the destinations. The Internet does not explicitly support broadcasting because of the huge amount of traffic it would create and because of the bandwidth it would need. Imagine the traffic generated in the Internet if one person wanted to send a message to everyone else connected to the Internet.

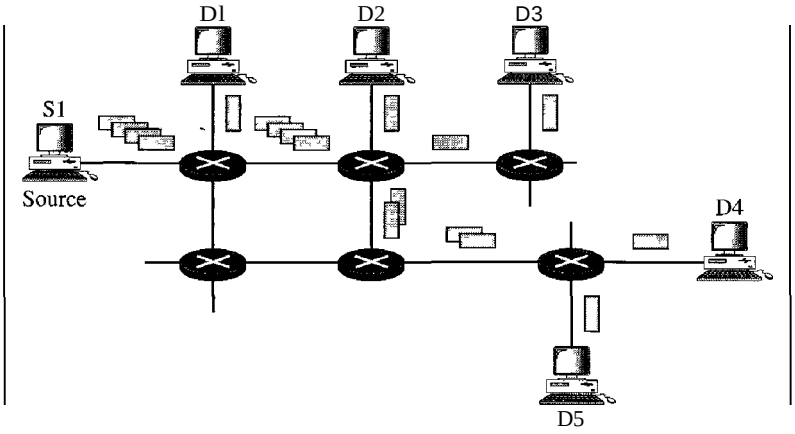
Multicasting Versus Multiple Unicasting

Before we finish this section, we need to distinguish between multicasting and multiple unicasting. Figure 22.35 illustrates both concepts.

Figure 22.35 *Multicasting versus multiple unicasting*



a. Multicasting



b. Multiple nnicasting

Multicasting starts with one single packet from the source that is duplicated by the routers. The destination address in each packet is the same for all duplicates. Note that only one single copy of the packet travels between any two routers.

In multiple unicasting, several packets start from the source. If there are five destinations, for example, the source sends five packets, each with a different unicast destination address. Note that there may be multiple copies traveling between two routers. For example, when a person sends an e-mail message to a group of people, this is multiple unicasting. The e-mail software creates replicas of the message, each with a different destination address and sends them one by one. This is not multicasting; it is multiple unicasting.

Emulation of Multicasting with Unicasting

You might wonder why we have a separate mechanism for multicasting, when it can be emulated with unicasting. There are two obvious reasons for this.

1. Multicasting is more efficient than multiple unicasting. In Figure 22.35, we can see how multicasting requires less bandwidth than does multiple unicasting. In multiple unicasting, some of the links must handle several copies.
2. In multiple unicasting, the packets are created by the source with a relative delay between packets. If there are 1000 destinations, the delay between the first and the last packet may be unacceptable. In multicasting, there is no delay because only one packet is created by the source.

Emulation of multicasting through multiple unicasting is not efficient and may create long delays, particularly with a large group.

Applications

Multicasting has many applications today such as access to distributed databases, information dissemination, teleconferencing, and distance learning.

Access to Distributed Databases

Most of the large databases today are distributed. That is, the information is stored in more than one location, usually at the time of production. The user who needs to access the database does not know the location of the information. A user's request is multicast to all the database locations, and the location that has the information responds.

Information Dissemination

Businesses often need to send information to their customers. If the nature of the information is the same for each customer, it can be multicast. In this way a business can send one message that can reach many customers. For example, a software update can be sent to all purchasers of a particular software package.

Dissemination of News

In a similar manner news can be easily disseminated through multicasting. One single message can be sent to those interested in a particular topic. For example, the statistics of the championship high school basketball tournament can be sent to the sports editors of many newspapers.

Teleconferencing

Teleconferencing involves multicasting. The individuals attending a teleconference all need to receive the same information at the same time. Temporary or permanent groups can be formed for this purpose. For example, an engineering group that holds meetings every Monday morning could have a permanent group while the group that plans the holiday party could form a temporary group.

Distance Learning

One growing area in the use of multicasting is distance learning. Lessons taught by one single professor can be received by a specific group of students. This is especially convenient for those students who find it difficult to attend classes on campus.

Multicast Routing

In this section, we first discuss the idea of optimal routing, common in all multicast protocols. We then give an overview of multicast routing protocols.

Optimal Routing: Shortest Path Trees

The process of optimal interdomain routing eventually results in the finding of the *shortest path tree.* The root of the tree is the source, and the leaves are the potential destinations. The path from the root to each destination is the shortest path. However, the number of trees and the formation of the trees in unicast and multicast routing are different. Let us discuss each separately.

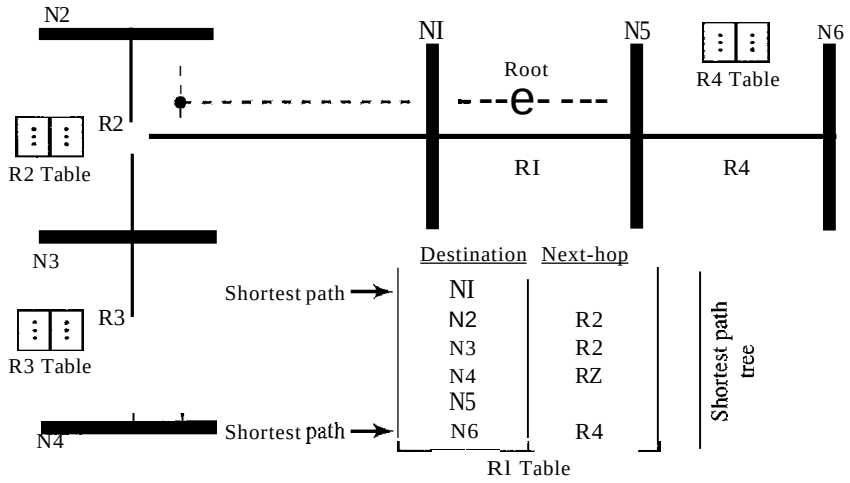
Unicast Routing In unicast routing, when a router receives a packet to forward, it needs to find the shortest path to the destination of the packet. The router consults its routing table for that particular destination. The next-hop entry corresponding to the destination is the start of the shortest path. The router knows the shortest path for each destination, which means that the router has a shortest path tree to optimally reach all destinations. In other words, each line of the routing table is a shortest path; the whole routing table is a shortest path tree. In unicast routing, each router needs only one shortest path tree to forward a packet; however, each router has its own shortest path tree. Figure 22.36 shows the situation.

The figure shows the details of the routing table and the shortest path tree for router Rl. Each line in the routing table corresponds to one path from the root to the corresponding network. The whole table represents the shortest path tree.

In unicast routing, each router in the domain has a table that defines a shortest path tree to possible destinations.

Multicast Routing When a router receives a multicast packet, the situation is different from when it receives a unicast packet. A multicast packet may have destinations in more than one network. Forwarding of a single packet to members of a group requires a shortest path tree. If we have *n* groups, we may need *n* shortest path trees. We can imagine the complexity of multicast routing. Two approaches have been used to solve the problem: source-based trees and group-shared trees.

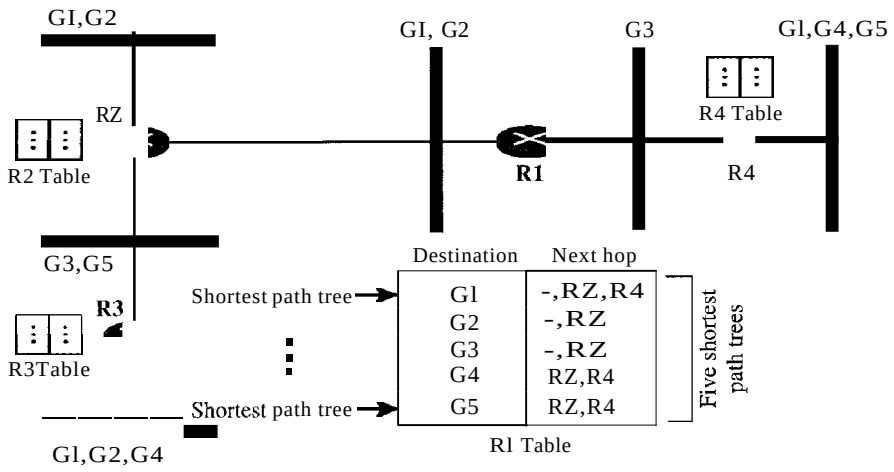
Figure 22.36 *Shortest path tree in unicast routing*



In multicast routing, each involved router needs to construct a shortest path tree for each group.

- Source-Based Tree. In the source-based tree approach, each router needs to have one shortest path tree for each group. The shortest path tree for a group defines the next hop for each network that has loyal member(s) for that group. In Figure 22.37, we assume that we have only five groups in the domain: GI, G2, G3, G4, and G5. At the moment GI has loyal members in four networks, G2 in three, G3 in two, G4 in two, and G5 in two. We have shown the names of the groups with loyal members on each network. Figure 22.37 also shows the multicast routing table for router RI. There is one shortest path tree for each group; therefore there are five shortest path trees for five groups. If router Rl receives a packet with destination

Figure 22.37 *Source-based tree approach*

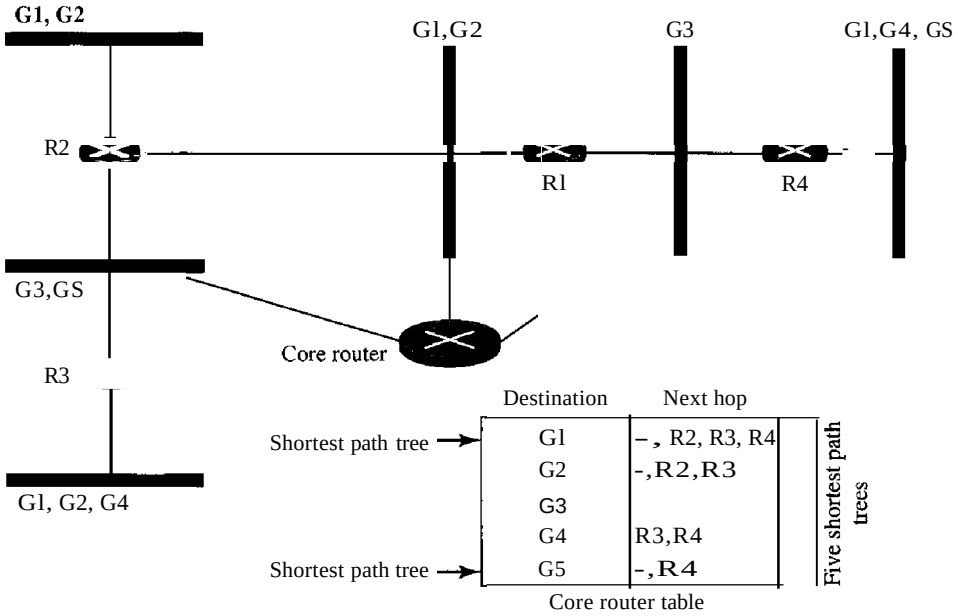


address G1, it needs to send a copy of the packet to the attached network, a copy to router R2, and a copy to router R4 so that all members of G1 can receive a copy. In this approach, if the number of groups is *m,* each router needs to have *m* shortest path trees, one for each group. We can imagine the complexity of the routing table if we have hundreds or thousands of groups. However, we will show how different protocols manage to alleviate the situation.

In the source-based tree approach, each router needs to have one shortest path tree for each group.

- Group-Shared Tree. In the group-shared tree approach, instead of each router having *m* shortest path trees, only one designated router, called the center core, or rendezvous router, takes the responsibility of distributing multicast traffic. The core has *m* shortest path trees in its routing table. The rest of the routers in the domain have none. If a router receives a multicast packet, it encapsulates the packet in a unicast packet and sends it to the core router. The core router removes the multicast packet from its capsule, and consults its routing table to route the packet. Figure 22.38 shows the idea.

Figure 22.38 *Group-shared tree approach*



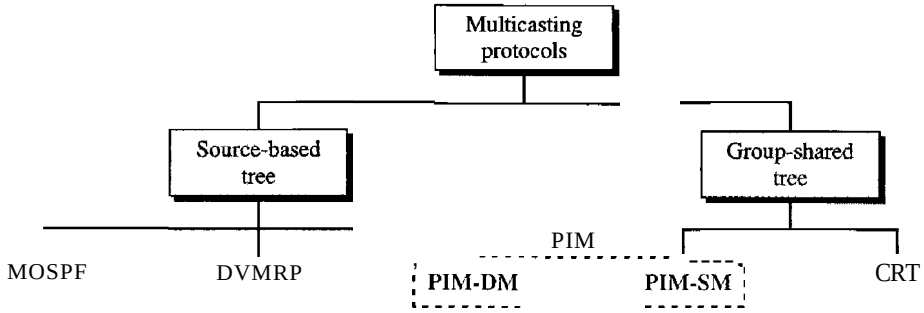
In the group-shared tree approach, only the core router, which has a shortest path tree for each group, is involved in multicasting.

Routing Protocols

During the last few decades, several multicast routing protocols have emerged. Some of these protocols are extensions of unicast routing protocols; others are totally new.

We discuss these protocols in the remainder of this chapter. Figure 22.39 shows the taxonomy of these protocols.

Figure 22.39 *Taxonomy of common multicast protocols*



Multicast Link State Routing: MOSPF

In this section, we briefly discuss multicast link state routing and its implementation in the Internet, MOSPF.

Multicast Link State Routing We discussed unicast link state routing in Section 22.3. We said that each router creates a shortest path tree by using Dijkstra's algorithm. The routing table is a translation of the shortest path tree. Multicast link state routing is a direct extension of unicast routing and uses a source-based tree approach. Although unicast routing is quite involved, the extension to multicast routing is very simple and straightforward.

Multicast link state routing uses the source-based tree approach.

Recall that in unicast routing, each node needs to advertise the state of its links. For multicast routing, a node needs to revise the interpretation of *state.* A node advertises every group which has any loyal member on the link. Here the meaning of state is "what groups are active on this link." The information about the group comes from IGMP (see Chapter 21). Each router running IGMP solicits the hosts on the link to find out the membership status.

When a router receives all these LSPs, it creates *n (n* is the number of groups) topologies, from which *n* shortest path trees are made by using Dijkstra's algorithm. So each router has a routing table that represents as many shortest path trees as there are groups.

The only problem with this protocol is the time and space needed to create and save the many shortest path trees. The solution is to create the trees only when needed. When a router receives a packet with a multicast destination address, it runs the Dijkstra algorithm to calculate the shortest path tree for that group. The result can be cached in case there are additional packets for that destination.

MOSPF Multicast Open Shortest Path First (MOSPF) protocol is an extension of the OSPF protocol that uses multicast link state routing to create source-based trees.

The protocol requires a new link state update packet to associate the unicast address of a host with the group address or addresses the host is sponsoring. This packet is called the group-membership LSA. In this way, we can include in the tree only the hosts (using their unicast addresses) that belong to a particular group. In other words, we make a tree that contains all the hosts belonging to a group, but we use the unicast address of the host in the calculation. For efficiency, the router calculates the shortest path trees on demand (when it receives the first multicast packet). In addition, the tree can be saved in cache memory for future use by the same source/group pair. MOSPF is a data-driven protocol; the first time an MOSPF router sees a datagram with a given source and group address, the router constructs the Dijkstra shortest path tree.

Multicast Distance Vector: DVMRP

In this section, we briefly discuss multicast distance vector routing and its implementation in the Internet, DVMRP.

Multicast Distance Vector Routing Unicast distance vector routing is very simple; extending it to support multicast routing is complicated. Multicast routing does not allow a router to send its routing table to its neighbors. The idea is to create a table from scratch by using the information from the unicast distance vector tables.

Multicast distance vector routing uses source-based trees, but the router never actually makes a routing table. When a router receives a multicast packet, it forwards the packet as though it is consulting a routing table. We can say that the shortest path tree is evanescent. After its use (after a packet is forwarded) the table is destroyed.

To accomplish this, the multicast distance vector algorithm uses a process based on four decision-making strategies. Each strategy is built on its predecessor. We explain them one by one and see how each strategy can improve the shortcomings of the previous one.

- Flooding. Flooding is the first strategy that comes to mind. A router receives a packet and, without even looking at the destination group address, sends it out from every interlace except the one from which it was received. Flooding accomplishes the first goal of multicasting: every network with active members receives the packet. However, so will networks without active members. This is a broadcast, not a multicast. There is another problem: it creates loops. A packet that has left the router may come back again from another interlace or the same interlace and be forwarded again. Some flooding protocols keep a copy of the packet for a while and discard any duplicates to avoid loops. The next strategy, reverse path forwarding, corrects this defect.

Flooding broadcasts packets, but creates loops in the systems.

- Reverse Path Forwarding (RPF). RPF is a modified flooding strategy. To prevent loops, only one copy is forwarded; the other copies are dropped. In RPF, a router forwards only the copy that has traveled the shortest path from the source to the router. To find this copy, RPF uses the unicast routing table. The router receives a packet and extracts the source address (a unicast address). It consults its unicast routing table as though it wants to send a packet to the source address. The routing table tells the

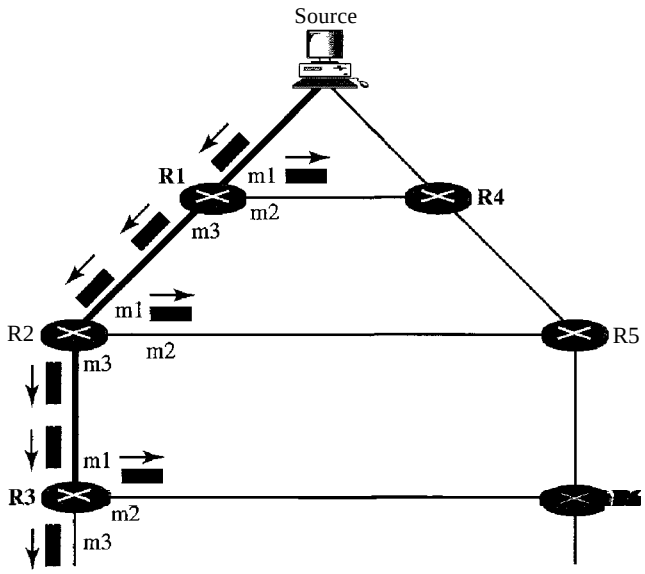
router the next hop. If the multicast packet has just come from the hop defined in the table, the packet has traveled the shortest path from the source to the router because the shortest path is reciprocal in unicast distance vector routing protocols. If the path from A to B is the shortest, then it is also the shortest from B to A. The router forwards the packet if it has traveled from the shortest path; it discards it otherwise.

This strategy prevents loops because there is always one shortest path from the source to the router. If a packet leaves the router and comes back again, it has not traveled the shortest path. To make the point clear, let us look at Figure 22.40.

Figure 22.40 shows part of a domain and a source. The shortest path tree as calculated by routers RI, R2, and R3 is shown by a thick line. When RI receives a packet from the source through the interface rnl, it consults its routing table and finds that the shortest path from RI to the source is through interface mI. The packet is forwarded. However, if a copy of the packet has arrived through interface m2, it is discarded because m2 does not define the shortest path from RI to the source. The story is the same with R2 and R3. You may wonder what happens if a copy of a packet that arrives at the ml interface of R3, travels through R6, R5, R2, and then enters R3 through interface ml. This interface is the correct interface for R3. Is the copy of the packet forwarded? The answer is that this scenario never happens because when the packet goes from R5 to R2, it will be discarded by R2 and never reaches R3. The upstream routers toward the source always discard a packet that has not gone through the shortest path, thus preventing confusion for the downstream routers.

RPF eliminates the loop in the flooding process.

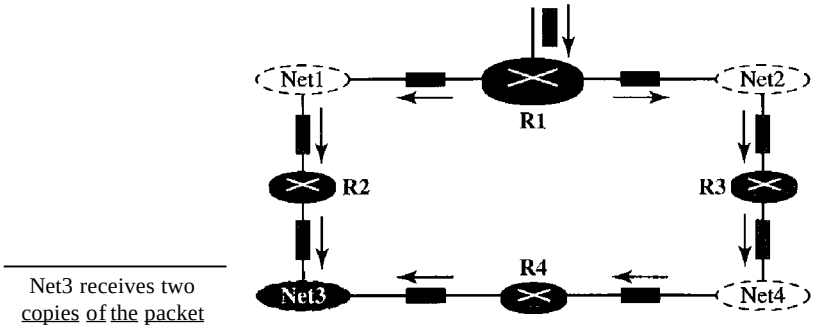
Figure 22.40 *Reverse path forwarding (RPF)*



- Reverse Path Broadcasting (RPB). RPF guarantees that each network receives a copy of the multicast packet without formation of loops. However, RPF does not

guarantee that each network receives only one copy; a network may receive two or more copies. The reason is that RPF is not based on the destination address (a group address); forwarding is based on the source address. To visualize the problem, let us look at Figure 22.41.

Figure 22.41 *Problem with RPF*

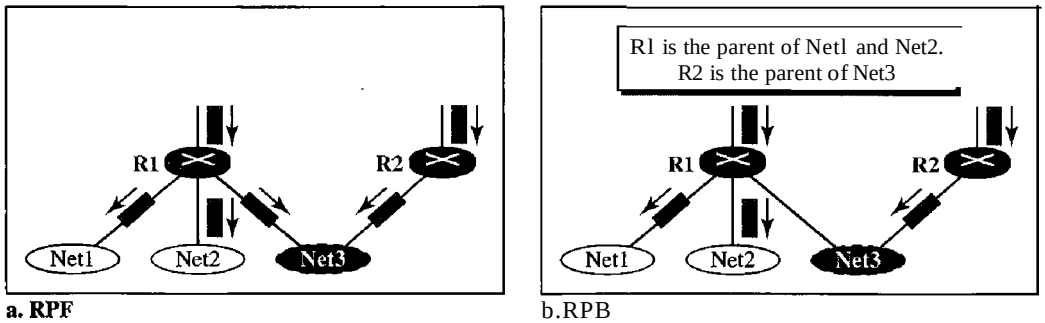


Net3 in this figure receives two copies of the packet even though each router just sends out one copy from each interface. There is duplication because a tree has not been made; instead of a tree we have a graph. Net3 has two parents: routers R2 andR4.

To eliminate duplication, we must define only one parent router for each network. We must have this restriction: A network can receive a multicast packet from a particular source only through a designated parent router.

Now the policy is clear. For each source, the router sends the packet only out of those interfaces for which it is the designated parent. This policy is called reverse path broadcasting (RPB). RPB guarantees that the packet reaches every network and that every network receives only one copy. Figure 22.42 shows the difference between RPF and RPB.

Figure 22.42 *RPF Versus RPB*



a. RPF b.RPB

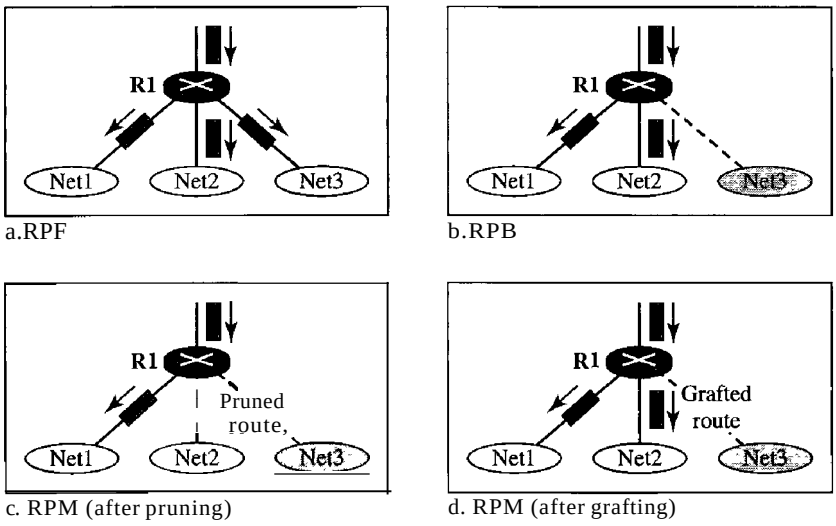
The reader may ask how the designated parent is determined. The designated parent router can be the router with the shortest path to the source. Because routers periodically

send updating packets to each other (in RIP), they can easily determine which router in the neighborhood has the shortest path to the source (when interpreting the source as the destination), If more than one router qualifies, the router with the smallest IP address is selected.

RPB creates a shortest path broadcast tree from the source to each destination. It guarantees that each destination receives one and only one copy of the packet.

- Reverse Path Multicasting (RPM). As you may have noticed, RPB does not multicast the packet, it broadcasts it. This is not efficient. To increase efficiency, the multicast packet must reach only those networks that have active members for that particular group. This is called reverse path multicasting (RPM). To convert broadcasting to multicasting, the protocol uses two procedures, pruning and grafting. Figure 22.43 shows the idea of pruning and grafting.

Figure 22.43 *RPF, RPB, and RPM*



The designated parent router of each network is responsible for holding the membership information. This is done through the IGMP protocol described in Chapter 21. The process starts when a router connected to a network finds that there is no interest in a multicast packet. The router sends a prune message to the upstream router so that it can exclude the corresponding interface. That is, the upstream router can stop sending multicast messages for this group through that interface. Now if this router receives prune messages from all downstream routers, it, in turn, sends a prune message to its upstream router.

What if a leaf router (a router at the bottom of the tree) has sent a prune message but suddenly realizes, through IGMP, that one of its networks is again interested in receiving the multicast packet? It can send a graft message. The graft message forces the upstream router to resume sending the multicast messages.

RPM adds pruning and grafting to RPB to create a multicast shortest path tree that supports dynamic membership changes.

DVMRP The Distance Vector Multicast Routing Protocol (DVMRP) is an implementation of multicast distance vector routing. It is a source-based routing protocol, based on RIP.

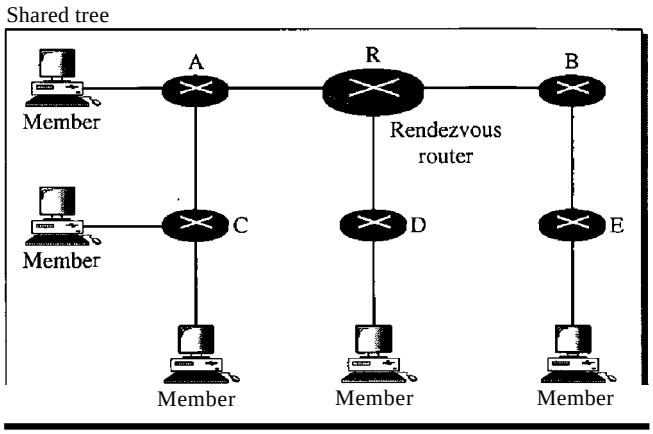
CRT

The Core-Based Tree (CBT) protocol is a group-shared protocol that uses a core as the root of the tree. The autonomous system is divided into regions, and a core (center router or rendezvous router) is chosen for each region.

Formation of the Tree After the rendezvous point is selected, every router is infonned of the unicast address of the selected router. Each router then sends a unicast join message (similar to a grafting message) to show that it wants to join the group. This message passes through all routers that are located between the sender and the rendezvous router. Each intermediate router extracts the necessary infonnation from the message, such as the unicast address of the sender and the interface through which the packet has arrived, and forwards the message to the next router in the path. When the rendezvous router has received all join messages from every member of the group, the tree is formed. Now every router knows its upstream router (the router that leads to the root) and the downstream router (the router that leads to the leaf).

If a router wants to leave the group, it sends a leave message to its upstream router. The upstream router removes the link to that router from the tree and forwards the message to its upstream router, and so on. Figure 22.44 shows a group-shared tree with its rendezvous router.

Figure 22.44 *Group-shared tree with rendezvous router*

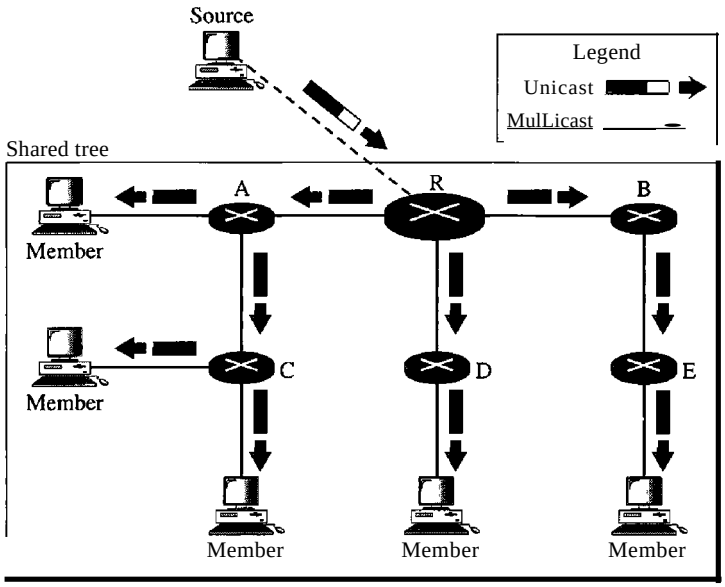


The reader may have noticed two differences between DVMRP and MOSPF, on one hand, and CBT, on the other. First, the tree for the first two is made from the root up; the tree for CBT is fonned from the leaves down. Second, in DVMRP, the tree is

first made (broadcasting) and then pruned; in CBT, there is no tree at the beginning; the joining (grafting) gradually makes the tree.

Sending Multicast Packets After formation of the tree, any source (belonging to the group or not) can send a multicast packet to all members of the group. It simply sends the packet to the rendezvous router, using the unicast address of the rendezvous router; the rendezvous router distributes the packet to all members of the group. Figure 22.45 shows how a host can send a multicast packet to all members of the group. Note that the source host can be any of the hosts inside the shared tree or any host outside the shared tree. In Figure 22.45 we show one located outside the shared tree.

Figure 22.45 *Sending a multicast packet to the rendezvous router*



Selecting the Rendezvous Router This approach is simple except for one point. How do we select a rendezvous router to optimize the process and multicasting as well? Several methods have been implemented. However, this topic is beyond the scope of this book, and we leave it to more advanced books.

In summary, the Core-Based Tree (CBT) is a group-shared tree, center-based protocol using one tree per group. One of the routers in the tree is called the core. A packet is sent from the source to members of the group following this procedure:

1. The source, which mayor may not be part of the tree, encapsulates the multicast packet inside a unicast packet with the unicast destination address of the core and sends it to the core. This part of delivery is done using a unicast address; the only recipient is the core router.
2. The core decapsulates the unicast packet and forwards it to all interested intetfaces.
3. Each router that receives the multicast packet, in tum, forwards it to all interested interfaces.

In CRT, the source sends the multicast packet (encapsulated in a unicast packet) to the core router. The core router decapsulates the packet and forwards it to all interested interfaces.

PIM

Protocol Independent Multicast (PIM) is the name given to two independent multicast routing protocols: Protocol Independent Multicast, Dense Mode (PIM-DM) and Protocol Independent Multicast, Sparse Mode (PIM-SM). Both protocols are unicast-protocol-dependent, but the similarity ends here. We discuss each separately.

PIM-DM PIM-DM is used when there is a possibility that each router is involved in multicasting (dense mode). In this environment, the use of a protocol that broadcasts the packet is justified because almost all routers are involved in the process.

PIM-DM is used in a dense multicast environment, such as a LAN.

PIM-DM is a source-based tree routing protocol that uses RPF and pruning and grafting strategies for multicasting. Its operation is like that of DVMRP; however, unlike DVMRP, it does not depend on a specific unicasting protocol. It assumes that the autonomous system is using a unicast protocol and each router has a table that can find the outgoing interface that has an optimal path to a destination. This unicast protocol can be a distance vector protocol (RIP) or link state protocol (OSPF).

PIM-DM uses RPF and pruning and grafting strategies to handle multicasting. However, it is independent of the underlying unicast protocol.

PIM-SM PIM-SM is used when there is a slight possibility that each router is involved in multicasting (sparse mode). In this environment, the use of a protocol that broadcasts the packet is not justified; a protocol such as CBT that uses a group-shared tree is more appropriate.

PIM-SM is used in a sparse multicast environment such as a WAN.

PIM-SM is a group-shared tree routing protocol that has a rendezvous point (RP) as the source of the tree. Its operation is like CBT; however, it is simpler because it does not require acknowledgment from a join message. In addition, it creates a backup set of RPs for each region to cover RP failures.

One of the characteristics of PIM-SM is that it can switch from a group-shared tree strategy to a source-based tree strategy when necessary. This can happen if there is a dense area of activity far from the RP. That area can be more efficiently handled with a source-based tree strategy instead of a group-shared tree strategy.

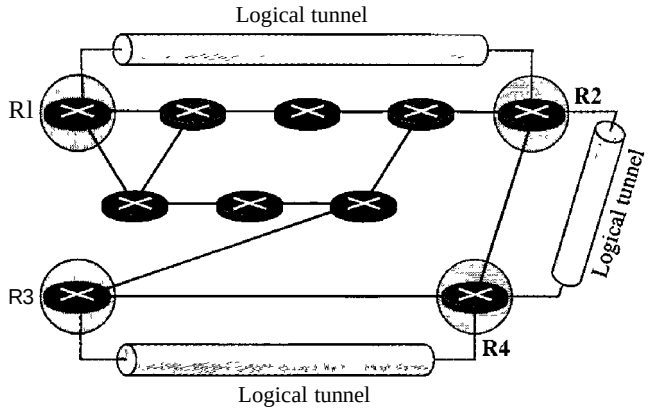
PIM-SM is similar to CRT but uses a simpler procedure.

MBONE

Multimedia and real-time communication have increased the need for multicasting in the Internet. However, only a small fraction of Internet routers are multicast routers. In

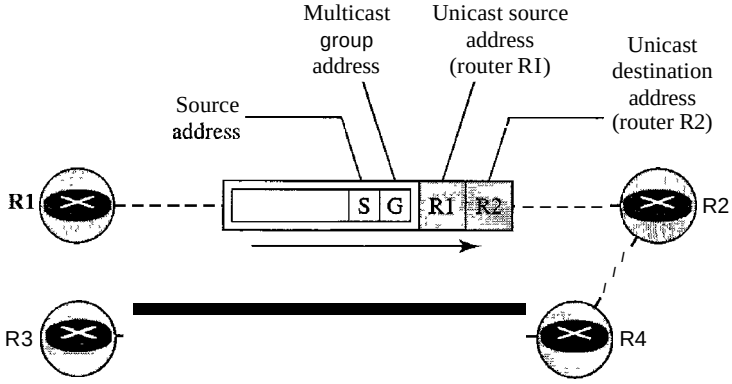
other words, a multicast router may not find another multicast router in the neighborhood to forward the multicast packet. Although this problem may be solved in the next few years by adding more and more multicast routers, there is another solution to this problem. The solution is tunneling. The multicast routers are seen as a group of routers on top of unicast routers. The multicast routers may not be connected directly, but they are connected logically. Figure 22.46 shows the idea. In Figure 22.46, only the routers enclosed in the shaded circles are capable of multicasting. Without tunneling, these routers are isolated islands. To enable multicasting, we make a multicast backbone (MBONE) out of these isolated routers by using the concept of tunneling.

Figure 22.46 *Logical tunneling*



A logical tunnel is established by encapsulating the multicast packet inside a unicast packet. The multicast packet becomes the payload (data) of the unicast packet. The intermediate (nonmulticast) routers forward the packet as unicast routers and deliver the packet from one island to another. It's as if the unicast routers do not exist and the two multicast routers are neighbors. Figure 22.47 shows the concept. So far the only protocol that supports MBONE and tunneling is DVMRP.

Figure 22.47 *MBONE*



22.5 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books and sites. The items in brackets [...] refer to the reference list at the end of the text.

Books

Delivery and forwarding are discussed in Chapter 6 of [For06]. Unicast routing protocols are discussed in Chapter 14 of [For06]. Multicasting and multicast routing are discussed in Chapter 15 of [For06]. For a complete discussion of multicasting see [WZOl]. For routing protocols see [HuiOO]. OSPF is discussed in [Moy98].

Sites

- www.ietf.org/rfc.html Information about RFCs

RFCs

A discussion of RIP can be found in the following RFCs:

1131,1245,1246,1247,1370,1583,1584,1585, 1586,1587, 1722,1723,2082,2453

A discussion of OSPF can be found in the following RFCs:

1131,1245,1246,1247,1370,1583,1584,1585, 1586, 1587,2178,2328,2329,2370

A discussion of BGP can be found in the following RFCs:

1092,1105,1163,1265,1266,1267,1364,1392,1403, 1565,1654,1655,1665,1771,1772, 1745,1774,2283

22.6 KEY TERMS

address aggregation
area
area border routers
area identification
autonomous system (AS)
backbone router
Border Gateway Protocol (BGP)
broadcasting
Core-Based Tree (CBT) protocol
data-driven
default method
delivery
designated parent router
Dijkstra's algorithm
direct delivery
distance learning
Distance Vector Multicast Routing Protocol (DVMRP)
distance vector routing
distributed database
dynamic routing method
dynamic routing table
flooding
forwarding
graft message
group-shared tree

hierarchical routing
hop count
host-specific method
ifconfig
immediate neighbors
indirect delivery
interdomain routing
intradomain routing
least-cost tree
link state routing
logical tunnel
longest mask matching
metric
multicast backbone (MBONE)
Multicast Open Shortest Path First (MOSPF)
multicast router
multicast routing
multicasting
multiple unicasting
netstat
network-specific method
next-hop address
next-hop method
Open Shortest Path First (OSPF)
optional attribute
OSPF protocol
path vector routing
point-to-point link
poison reverse
policy routing
Protocol Independent Multicast (PIM)
Protocol Independent Multicast, Dense Mode (PIM-DM)
Protocol Independent Multicast, Sparse Mode (PIM-SM)
prune message
rendezvous router
rendezvous-point tree
reverse path broadcasting (RPB)
reverse path forwarding (RPF)
reverse path multicasting (RPM)
route method
routing
Routing Information Protocol (RIP)
routing protocols
shortest path tree
slow convergence
source-based tree
speaker node
split horizon
static routing table
stub link
switching fabric
teleconferencing
transient link
triggered update
tunneling
unicasting
update message
virtual link
well-known attribute

22.7 SUMMARY

- The delivery of a packet is called direct if the deliverer (host or router) and the destination are on the same network; the delivery of a packet is called indirect if the deliverer (host or router) and the destination are on different networks.
- In the next-hop method, instead of a complete list of the stops the packet must make, only the address of the next hop is listed in the routing table; in the network-specific method, all hosts on a network share one entry in the routing table.

- In the host-specific method, the full IP address of a host is given in the routing table.
- In the default method, a router is assigned to receive all packets with no match in the routing table.
- The routing table for classless addressing needs at least four columns.
- Address aggregation simplifies the forwarding process in classless addressing.
- Longest mask matching is required in classless addressing.
- Classless addressing requires hierarchical and geographical routing to prevent immense routing tables.
- A static routing table's entries are updated manually by an administrator; a dynamic routing table's entries are updated automatically by a routing protocol.
- A metric is the cost assigned for passage of a packet through a network.
- An autonomous system (AS) is a group of networks and routers under the authority of a single administration.
- RIP is based on distance vector routing, in which each router shares, at regular intervals, its knowledge about the entire AS with its neighbors.
- Two shortcomings associated with the RIP protocol are slow convergence and instability. Procedures to remedy RIP instability include triggered update, split horizons, and poison reverse.
- OSPF divides an AS into areas, defined as collections of networks, hosts, and routers.
- OSPF is based on link state routing, in which each router sends the state of its neighborhood to every other router in the area. A packet is sent only if there is a change in the neighborhood.
- OSPF routing tables are calculated by using Dijkstra's algorithm.
- BGP is an interautonomous system routing protocol used to update routing tables.
- BGP is based on a routing protocol called path vector routing. In this protocol, the ASs through which a packet must pass are explicitly listed.
- In a source-based tree approach to multicast routing, the source/group combination determines the tree; in a group-shared tree approach to multicast routing, the group determines the tree.
- MOSPF is a multicast routing protocol that uses multicast link state routing to create a source-based least-cost tree.
- In reverse path forwarding (RPF), the router forwards only the packets that have traveled the shortest path from the source to the router.
- Reverse path broadcasting (RPB) creates a shortest path broadcast tree from the source to each destination. It guarantees that each destination receives one and only one copy of the packet.
- Reverse path multicasting (RPM) adds pruning and grafting to RPB to create a multicast shortest path tree that supports dynamic membership changes.
- DVMRP is a multicast routing protocol that uses the distance routing protocol to create a source-based tree.
- The Core-Based Tree (CBT) protocol is a multicast routing protocol that uses a router as the root of the tree.

- PIM-DM is a source-based tree routing protocol that uses RPF and pruning and grafting strategies to handle multicasting.
- PIM-SM is a group-shared tree routing protocol that is similar to CBT and uses a rendezvous router as the source of the tree.
- For multicasting between two noncontiguous multicast routers, we make a multicast backbone (MBONE) to enable tunneling.

22.8 PRACTICE SET

Review Questions

1. What is the difference between a direct and an indirect delivery?
2. List three forwarding techniques discussed in this chapter and give a brief description of each.
3. Contrast two different routing tables discussed in this chapter.
4. What is the purpose of RIP?
5. What are the functions of a RIP message?
6. Why is the expiration timer value 6 times that of the periodic timer value?
7. How does the hop count limit alleviate RIP's problems?
8. List RIP shortcomings and their corresponding fixes.
9. What is the basis of classification for the four types of links defined by OSPF?
10. Why do OSPF messages propagate faster than RIP messages?
11. What is the purpose of BGP?
12. Give a brief description of two groups of multicast routing protocols discussed in this chapter.

Exercises

13. Show a routing table for a host that is totally isolated.
14. Show a routing table for a host that is connected to a LAN without being connected to the Internet.
15. Find the topology of the network if Table 22.3 is the routing table for router R1.

Table 22.3 *Routing table for Exercise 15*

Mask	*Network Address*	*Next-Hop Address*	*Interface*
/27	202.14.17.224	-	m1
/18	145.23.192.0	-	mO
Default	Default	130.56.12.4	m2

16. Can router R1 in Figure 22.8 receive a packet with destination address 140.24.7.194? Explain your answer.

17. Can router Rl in Figure 22.8 receive a packet with destination address l40.24.7.42? Explain your answer.
18. Show the routing table for the regional ISP in Figure 22.9.
19. Show the routing table for local ISP 1 in Figure 22.9.
20. Show the routing table for local ISP 2 in Figure 22.9.
21. Show the routing table for local ISP 3 in Figure 22.9.
22. Show the routing table for small ISP 1 in Figure 22.9.
23. Contrast and compare distance vector routing with link state routing.
24. A router has the following RIP routing table:

Net!	4	B
Net2	2	C
Net3	!	F
Net4	5	G

What would be the contents of the table if the router received the following RIP message from router C?

Net!	2
Net2	!
Net3	3
Net4	7

25. How many bytes are empty in a RIP message that advertises *N* networks?
26. A router has the following RIP routing table:

Net!	4	B
Net2	2	C
Net3	1	F
Net4	5	G

Show the response message sent by this router.

27. Show the autonomous system with the following specifications:
    a. There are eight networks (Nl to N8).
    b. There are eight routers (Rl to R8).
    c. Nl, N2, N3, N4, NS, and N6 are Ethernet LANs.
    d. N7 and N8 are point-to-point WANs.
    e. Rl connects Nl and N2.
    f. R2 connects Nl and N7.
    g. R3 connects N2 and N8.
    h. R4 connects N7 and N6.
    i. RS connects N6 and N3.
    j. R6 connects N6 and N4.
    k. R7 connects N6 and NS.
    1. R8 connects N8 and N5.

28. Draw the graphical representation of the autonomous system of Exercise 27 as seen by OSPF.
29. Which of the networks in Exercise 27 is a transient network? Which is a stub network?
30. A router using DVMRP receives a packet with source address 10.14.17.2 from interface 2. If the router forwards the packet, what are the contents of the entry related to this address in the unicast routing table?
31. Does RPF actually create a shortest path tree? Explain.
32. Does RPB actually create a shortest path tree? Explain. What are the leaves of the tree?
33. Does RPM actually create a shortest path tree? Explain. What are the leaves of the tree?

Research Activities

34. If you have access to UNIX (or LINUX), use *netstat* and *ifconfig* to find the routing table for the server to which you are connected.
35. Find out how your ISP uses address aggregation and longest mask match principles.
36. Find out whether your IP address is part of the geographical address allocation.
37. If you are using a router, find the number and names of the columns in the routing table.

PART 5

Transport Layer

Objectives

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level.

The transport layer is responsible for the delivery of a message from one process to another.

Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process on one computer to a specific process on the other. The transport layer header must therefore include a type of address called a *service-point address* in the OSI model and port number or port addresses in the Internet and TCP/IP protocol suite.

A transport layer protocol can be either connectionless or connection-oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection-oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data is transferred, the connection is terminated.

In the transport layer, a message is normally divided into transmittable segments. A connectionless protocol, such as UDP, treats each segment separately. A connection-oriented protocol, such as TCP and SCTP, creates a relationship between the segments using sequence numbers.

Like the data link layer, the transport layer may be responsible for flow and error control. However, flow and error control at this layer is performed end to end rather than across a single link. We will see that one of the protocols discussed in this part of

the book, UDP, is not involved in flow or error control. On the other hand, the other two protocols, TCP and SCTP, use sliding windows for flow control and an acknowledgment system for error control.

Part 5 of the book is devoted to the transport layer and the services provided by this layer.

Chapters

This part consists of two chapters: Chapters 23 and 24.

Chapter 23

Chapter 23 discusses three transport layer protocols in the Internet: UDP, TCP, and SCTP. The first, User Datagram Protocol (UDP), is a connectionless, unreliable protocol that is used for its efficiency. The second, Transmission Control Protocol (TCP), is a connection-oriented, reliable protocol that is a good choice for data transfer. The third, Stream Control Transport Protocol (SCTP) is a new transport-layer protocol designed for multimedia applications.

Chapter 24

Chapter 24 discuss two related topics: congestion control and quality of service. Although these two issues can be related to any layer, we discuss them here with some references to other layers.

CHAPTER 23

Process-la-Process Delivery: UDp, TCp, and SCTP

We begin this chapter by giving the rationale for the existence of the transport layer-the need for process-to-process delivery. We discuss the issues arising from this type of delivery, and we discuss methods to handle them.

The Internet model has three protocols at the transport layer: UDP, TCP, and SCTP. First we discuss UDP, which is the simplest of the three. We see how we can use this very simple transport layer protocol that lacks some of the features of the other two.

We then discuss TCP, a complex transport layer protocol. We see how our previously presented concepts are applied to TCP. We postpone the discussion of congestion control and quality of service in TCP until Chapter 24 because these two topics apply to the data link layer and network layer as well.

We finally discuss SCTP, the new transport layer protocol that is designed for multihomed, multistream applications such as multimedia.

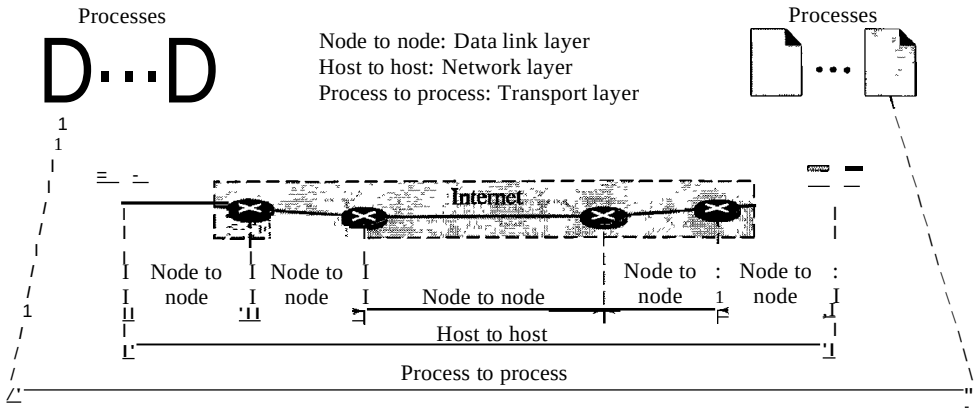
23.1 PROCESS-TO-PROCESS DELIVERY

The data link layer is responsible for delivery of frames between two neighboring nodes over a link. This is called *node-to-node delivery.* The network layer is responsible for delivery of datagrams between two hosts. This is called *host-to-host delivery.* Communication on the Internet is not defined as the exchange of data between two nodes or between two hosts. Real communication takes place between two processes (application programs). We need process-to-process delivery. However, at any moment, several processes may be running on the source host and several on the destination host. To complete the delivery, we need a mechanism to deliver data from one of these processes running on the source host to the corresponding process running on the destination host.

The transport layer is responsible for process-to-process delivery-the delivery of a packet, part of a message, from one process to another. Two processes communicate in a client/server relationship, as we will see later. Figure 23.1 shows these three types of deliveries and their domains.

The transport layer is responsible for process-to-process delivery.

Figure 23.1 *Types ofdata deliveries*



Client/Server Paradigm

Although there are several ways to achieve process-to-process communication, the most common one is through the client/server paradigm. A process on the local host, called a client, needs services from a process usually on the remote host, called a server.

Both processes (client and server) have the same name. For example, to get the day and time from a remote machine, we need a Daytime client process running on the local host and a Daytime server process running on a remote machine.

Operating systems today support both multiuser and multiprogramming environments. A remote computer can run several server programs at the same time, just as local computers can run one or more client programs at the same time. For communication, we must define the following:

1. Local host
2. Local process
3. Remote host
4. Remote process

Addressing

Whenever we need to deliver something to one specific destination among many, we need an address. At the data link layer, we need a MAC address to choose one node among several nodes if the connection is not point-to-point. A frame in the data link layer needs a destination MAC address for delivery and a source address for the next node's reply.

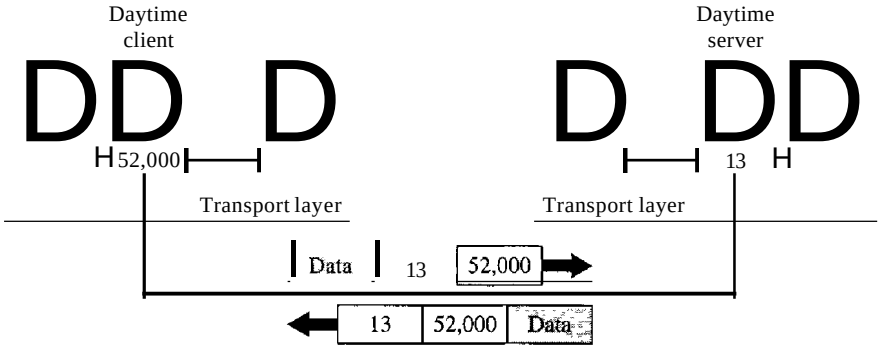
At the network layer, we need an IP address to choose one host among millions. A datagram in the network layer needs a destination IP address for delivery and a source IP address for the destination's reply.

At the transport layer, we need a transport layer address, called a port number, to choose among multiple processes running on the destination host. The destination port number is needed for delivery; the source port number is needed for the reply.

In the Internet model, the port numbers are 16-bit integers between 0 and 65,535. The client program defines itself with a port number, chosen randomly by the transport layer software running on the client host. This is the ephemeral port number.

The server process must also define itself with a port number. This port number, however, cannot be chosen randomly. **If** the computer at the server site runs a server process and assigns a random number as the port number, the process at the client site that wants to access that server and use its services will not know the port number. Of course, one solution would be to send a special packet and request the port number of a specific server, but this requires more overhead. The Internet has decided to use universal port numbers for servers; these are called well-known port numbers. There are some exceptions to this rule; for example, there are clients that are assigned well-known port numbers. Every client process knows the well-known port number of the corresponding server process. For example, while the Daytime client process, discussed above, can use an ephemeral (temporary) port number 52,000 to identify itself, the Daytime server process must use the well-known (permanent) port number 13. Figure 23.2 shows this concept.

Figure 23.2 *Port numbers*



It should be clear by now that the IP addresses and port numbers play different roles in selecting the final destination of data. The destination IP address defines the host among the different hosts in the world. After the host has been selected, the port number defines one of the processes on this particular host (see Figure 23.3).

lANA Ranges

The lANA (Internet Assigned Number Authority) has divided the port numbers into three ranges: well known, registered, and dynamic (or private), as shown in Figure 23.4.

- Well-known ports. The ports ranging from 0 to 1023 are assigned and controlled by lANA. These are the well-known ports.
- Registered ports. The ports ranging from 1024 to 49,151 are not assigned or controlled by lANA. They can only be registered with lANA to prevent duplication.
- Dynamic ports. The ports ranging from 49,152 to 65,535 are neither controlled nor registered. They can be used by any process. These are the ephemeral ports.

Figure 23.3 *IP addresses versus port numbers*

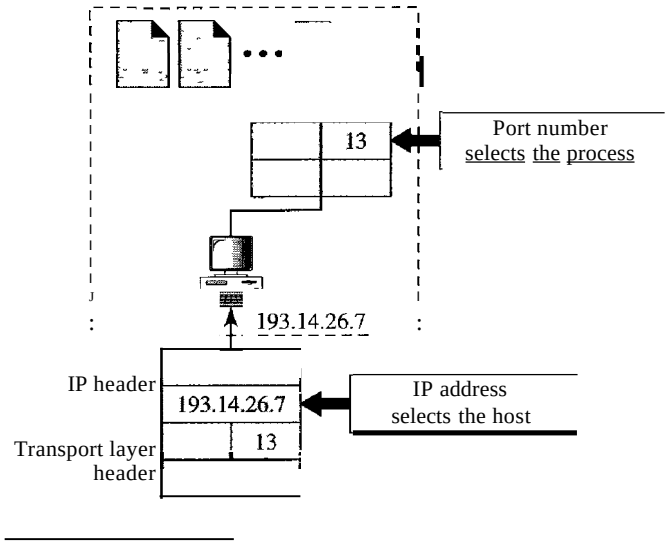
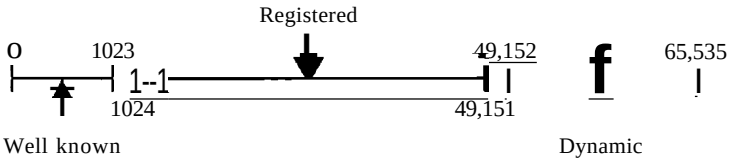


Figure 23.4 *lANA ranges*

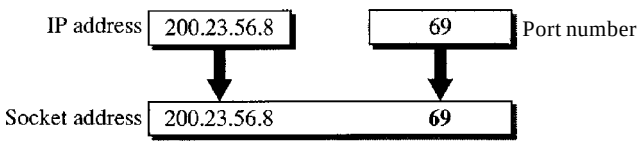


Socket Addresses

Process-to-process delivery needs two identifiers, IP address and the port number, at each end to make a connection. The combination of an IP address and a port number is called a socket address. The client socket address defines the client process uniquely just as the server socket address defines the server process uniquely (see Figure 23.5).

A transport layer protocol needs a pair of socket addresses: the client socket address and the server socket address. These four pieces of information are part of the IP header and the transport layer protocol header. The IP header contains the IP addresses; the UDP or TCP header contains the port numbers.

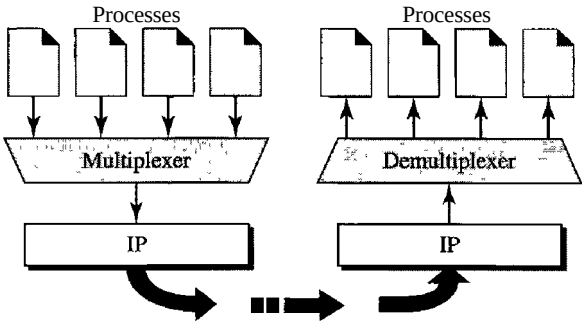
Figure 23.5 *Socket address*



Multiplexing and Demultiplexing

The addressing mechanism allows multiplexing and demultiplexing by the transport layer, as shown in Figure 23.6.

Figure 23.6 *Multiplexing and demultiplexing*



Multiplexing

At the sender site, there may be several processes that need to send packets. However, there is only one transport layer protocol at any time. This is a many-to-one relationship and requires multiplexing. The protocol accepts messages from different processes, differentiated by their assigned port numbers. After adding the header, the transport layer passes the packet to the network layer.

Demultiplexing

At the receiver site, the relationship is one-to-many and requires demultiplexing. The transport layer receives datagrams from the network layer. After error checking and dropping of the header, the transport layer delivers each message to the appropriate process based on the port number.

Connectionless Versus Connection-Oriented Service

A transport layer protocol can either be connectionless or connection-oriented.

Connectionless Service

In a connectionless service, the packets are sent from one party to another with no need for connection establishment or connection release. The packets are not numbered; they may be delayed or lost or may arrive out of sequence. There is no acknowledgment either. We will see shortly that one of the transport layer protocols in the Internet model, UDP, is connectionless.

Connection-Oriented Service

In a connection-oriented service, a connection is first established between the sender and the receiver. Data are transferred. At the end, the connection is released. We will see shortly that TCP and SCTP are connection-oriented protocols.

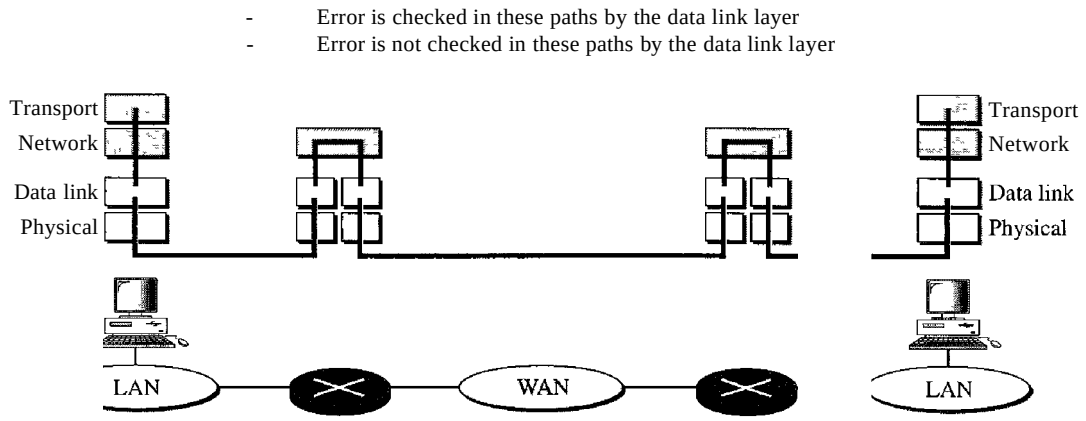
Reliable Versus Unreliable

The transport layer service can be reliable or unreliable. If the application layer program needs reliability, we use a reliable transport layer protocol by implementing flow and error control at the transport layer. This means a slower and more complex service. On the other hand, if the application program does not need reliability because it uses its own flow and error control mechanism or it needs fast service or the nature of the service does not demand flow and error control (real-time applications), then an unreliable protocol can be used.

In the Internet, there are three common different transport layer protocols, as we have already mentioned. UDP is connectionless and unreliable; TCP and SCTP are connection-oriented and reliable. These three can respond to the demands of the application layer programs.

One question often comes to the mind. If the data link layer is reliable and has flow and error control, do we need this at the transport layer, too? The answer is yes. Reliability at the data link layer is between two nodes; we need reliability between two ends. Because the network layer in the Internet is unreliable (best-effort delivery), we need to implement reliability at the transport layer. To understand that error control at the data link layer does not guarantee error control at the transport layer, let us look at Figure 23.7.

Figure 23.7 *Error control*

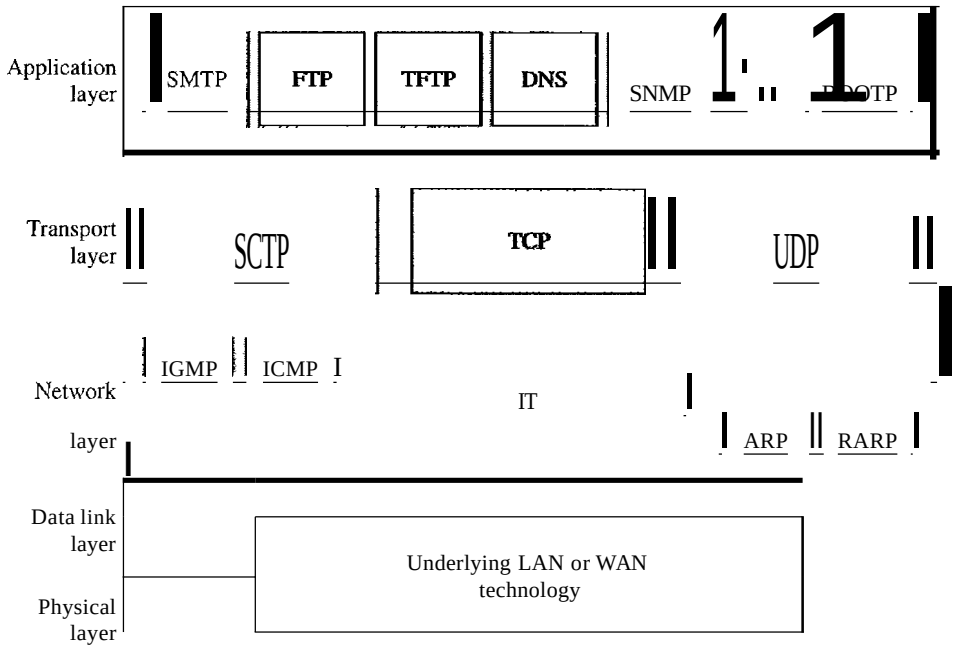


As we will see, flow and error control in TCP is implemented by the sliding window protocol, as discussed in Chapter 11. The window, however, is character-oriented, instead of frame-oriented.

Three Protocols

The original TCP/IP protocol suite specifies two protocols for the transport layer: UDP and TCP. We first focus on UDP, the simpler of the two, before discussing TCP. A new transport layer protocol, SCTP, has been designed, which we also discuss in this chapter. Figure 23.8 shows the position of these protocols in the TCP/IP protocol suite.

Figure 23.8 *Position of UDp, TCp, and SCTP in TCPIIP suite*



23.2 USER DATAGRAM PROTOCOL (UDP)

The User Datagram Protocol (UDP) is called a connectionless, unreliable transport protocol. It does not add anything to the services of IP except to provide process-to-process communication instead of host-to-host communication. Also, it performs very limited error checking.

If UDP is so powerless, why would a process want to use it? With the disadvantages come some advantages. UDP is a very simple protocol using a minimum of overhead. If a process wants to send a small message and does not care much about reliability, it can use UDP. Sending a small message by using UDP takes much less interaction between the sender and receiver than using TCP or SCTP.

Well-Known Ports for UDP

Table 23.1 shows some well-known port numbers used by UDP. Some port numbers can be used by both UDP and TCP. We discuss them when we talk about TCP later in the chapter.

Table 23.1 *Well-known ports used with UDP*

Port	*Protocol*	*Description*
7	Echo	Echoes a received datagram back to the sender
9	Discard	Discards any datagram that is received
11	Users	Active users

Table 23.1 *Well-known ports used with UDP (continued)*

Port	*Protocol*	*Description*
13	Daytime	Returns the date and the time
17	Quote	Returns a quote of the day
19	Chargen	Returns a string of characters
53	Nameserver	Domain Name Service
67	BOOTPs	Server port to download bootstrap information
68	BOOTPc	Client port to download bootstrap information
69	TFTP	Trivial File Transfer Protocol
III	RPC	Remote Procedure Call
123	NTP	Network Time Protocol
161	SNMP	Simple Network Management Protocol
162	SNMP	Simple Network Management Protocol (trap)

Example 23.1

In UNIX, the well-known ports are stored in a file called fetcfservices. Each line in this file gives the name of the server and the well-known port number. We can use the *grep* utility to extract the line corresponding to the desired application. The following shows the port for FTP. Note that FTP can use port 21 with either UDP or TCP.

```
$grep    ftp    fetclservices
ftp          21ftcp
fip          211udp
```

SNMP uses two port numbers (161 and 162), each for a different purpose, as we will see in Chapter 28.

```
$grep      snmp fetclservices
snmp            161ftcp          #Simple Net  Mgmt Proto
snmp            1611udp          #Simple Net  Mgmt Proto
snmptrap        162/udp          #Traps for SNMP
```

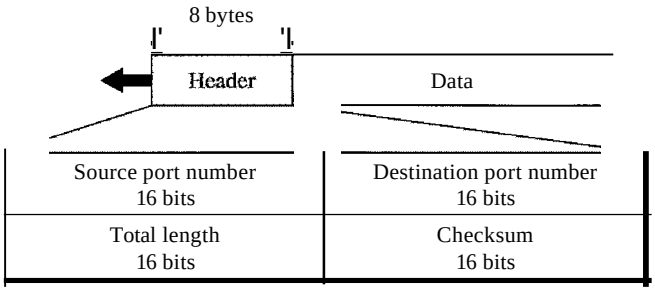
User Datagram

UDP packets, called user datagrams, have a fixed-size header of 8 bytes. Figure 23.9 shows the format of a user datagram.

The fields are as follows:

- Source port number. This is the port number used by the process running on the source host. It is 16 bits long, which means that the port number can range from 0 to 65,535. If the source host is the client (a client sending a request), the port number, in most cases, is an ephemeral port number requested by the process and chosen by the UDP software running on the source host. If the source host is the server (a server sending a response), the port number, in most cases, is a well-known port number.

Figure 23.9 *User datagram format*



- Destination port number. This is the port number used by the process running on the destination host. It is also 16 bits long. If the destination host is the server (a client sending a request), the port number, in most cases, is a well-known port number. If the destination host is the client (a server sending a response), the port number, in most cases, is an ephemeral port number. In this case, the server copies the ephemeral port number it has received in the request packet.
- Length. This is a 16-bit field that defines the total length of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes. However, the total length needs to be much less because a UDP user datagram is stored in an IP datagram with a total length of 65,535 bytes.

  The length field in a UDP user datagram is actually not necessary. A user datagram is encapsulated in an IP datagram. There is a field in the IP datagram that defines the total length. There is another field in the IP datagram that defines the length of the header. So if we subtract the value of the second field from the first, we can deduce the length of a UDP datagram that is encapsulated in an IP datagram.

$$\text{UDP length} = \text{IP length} - \text{IP header's length}$$

  However, the designers of the UDP protocol felt that it was more efficient for the destination UDP to calculate the length of the data from the information provided in the UDP user datagram rather than ask the IP software to supply this information. We should remember that when the IP software delivers the UDP user datagram to the UDP layer, it has already dropped the IP header.
- Checksum. This field is used to detect errors over the entire user datagram (header plus data). The checksum is discussed next.

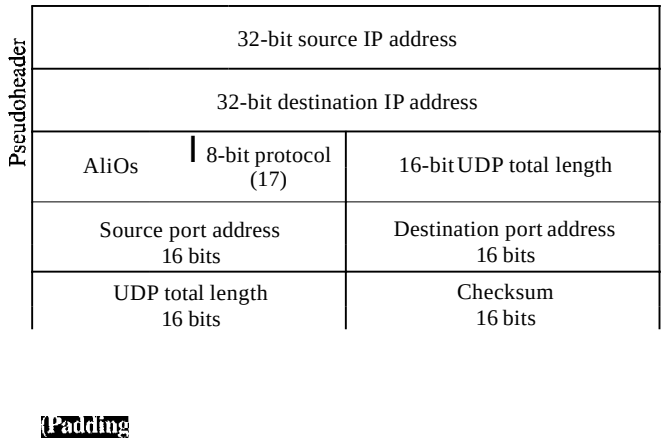
Checksum

We have already talked about the concept of the checksum and the way it is calculated in Chapter 10. We have also shown how to calculate the checksum for the IP and ICMP packet. We now show how this is done for UDP.

The UDP checksum calculation is different from the one for IP and ICMP. Here the checksum includes three sections: a pseudoheader, the UDP header, and the data coming from the application layer.

The pseudoheader is the part of the header of the IP packet in which the user datagram is to be encapsulated with some fields filled with Os (see Figure 23.10).

Figure 23.10 *Pseudoheader for checksum calculation*



If the checksum does not include the pseudoheader, a user datagram may arrive safe and sound. However, if the IP header is corrupted, it may be delivered to the wrong host.

The protocol field is added to ensure that the packet belongs to UDP, and not to other transport-layer protocols. We will see later that if a process can use either UDP or TCP, the destination port number can be the same. The value of the protocol field for UDP is 17. If this value is changed during transmission, the checksum calculation at the receiver will detect it and UDP drops the packet. It is not delivered to the wrong protocol.

Note the similarities between the pseudoheader fields and the last 12 bytes of the IP header.

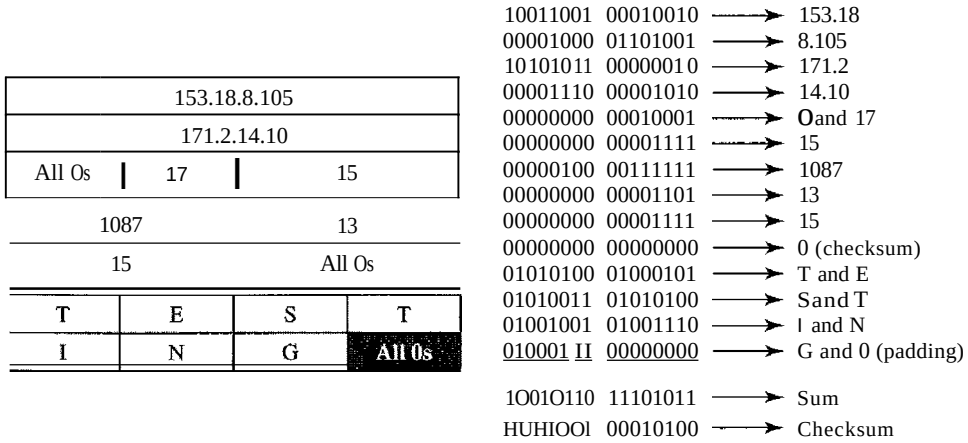
Example 23.2

Figure 23.11 shows the checksum calculation for a very small user datagram with only 7 bytes of data. Because the number of bytes of data is odd, padding is added for checksum calcUlation. The pseudoheader as well as the padding will be dropped when the user datagram is delivered to IP.

Optional Use of the Checksum

The calculation of the checksum and its inclusion in a user datagram are optional. If the checksum is not calculated, the field is filled with 1s. Note that a calculated checksum can never be allIs because this implies that the sum is all Os, which is impossible because it requires that the value of fields to be Os.

Figure 23.11 *Checksum calculation of a simple UDP user datagram*



UDP Operation

UDP uses concepts common to the transport layer. These concepts will be discussed here briefly, and then expanded in the next section on the TCP protocol.

Connectionless Services

As mentioned previously, UDP provides a connectionless service. This means that each user datagram sent by UDP is an independent datagram. There is no relationship between the different user datagrams even if they are coming from the same source process and going to the same destination program. The user datagrams are not numbered. Also, there is no connection establishment and no connection termination, as is the case for TCP. This means that each user datagram can travel on a different path.

One of the ramifications of being connectionless is that the process that uses UDP cannot send a stream of data to UDP and expect UDP to chop them into different related user datagrams. Instead each request must be small enough to fit into one user datagram. Only those processes sending short messages should use UDP.

Flow and Error Control

UDP is a very simple, unreliable transport protocol. There is no flow control and hence no window mechanism. The receiver may overflow with incoming messages.

There is no error control mechanism in UDP except for the checksum. This means that the sender does not know if a message has been lost or duplicated. When the receiver detects an error through the checksum, the user datagram is silently discarded.

The lack of flow control and error control means that the process using UDP should provide these mechanisms.

Encapsulation and Decapsulation

To send a message from one process to another, the UDP protocol encapsulates and decapsulates messages in an IP datagram.