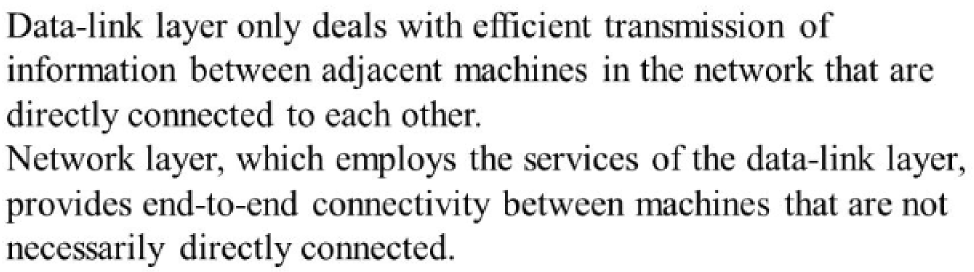
Unit 2

Network layer design issues:

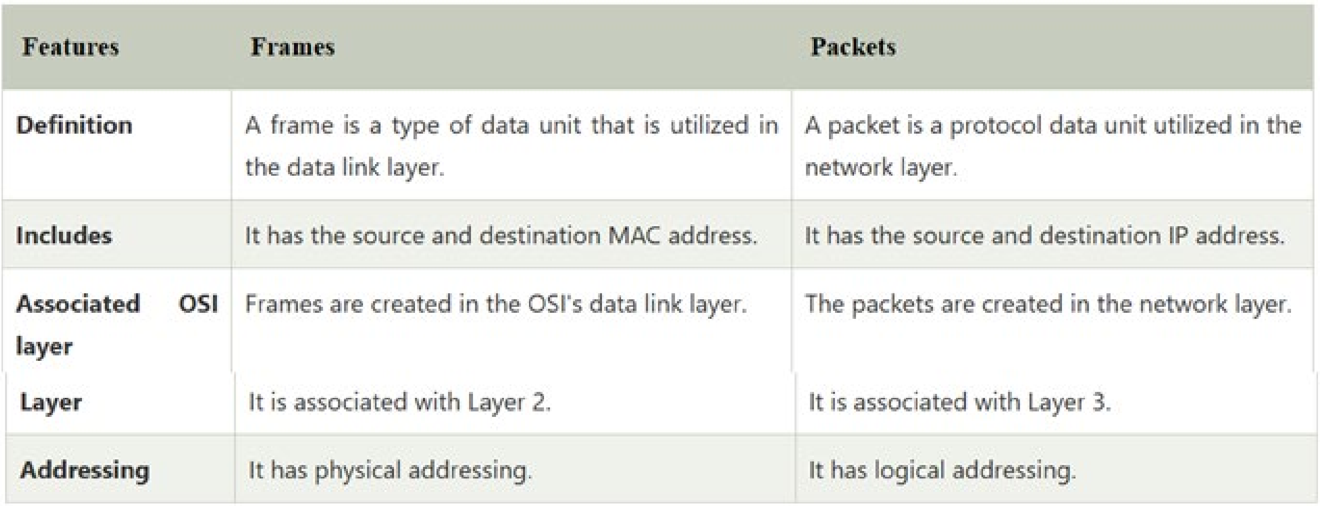
The network layer is concerned with getting packets from the source all the way to the destination. Getting to the destination may require making many hops at intermediate routers along the way. This function clearly contrasts with that of the data link layer, which has the more modest goal of just moving frames from one end of a wire to the other. Thus, the network layer is the lowest layer that deals with end-to-end transmission.

To achieve its goals, the network layer must know about the topology of the network (i.e., the set of all routers and links) and choose appropriate paths through it, even for large networks. It must also take care when choosing routes to avoid overloading some of the communication lines and routers while leaving others idle. Finally, when the source and destination are in different networks, new problems occur. It is up to the network layer to deal with them.

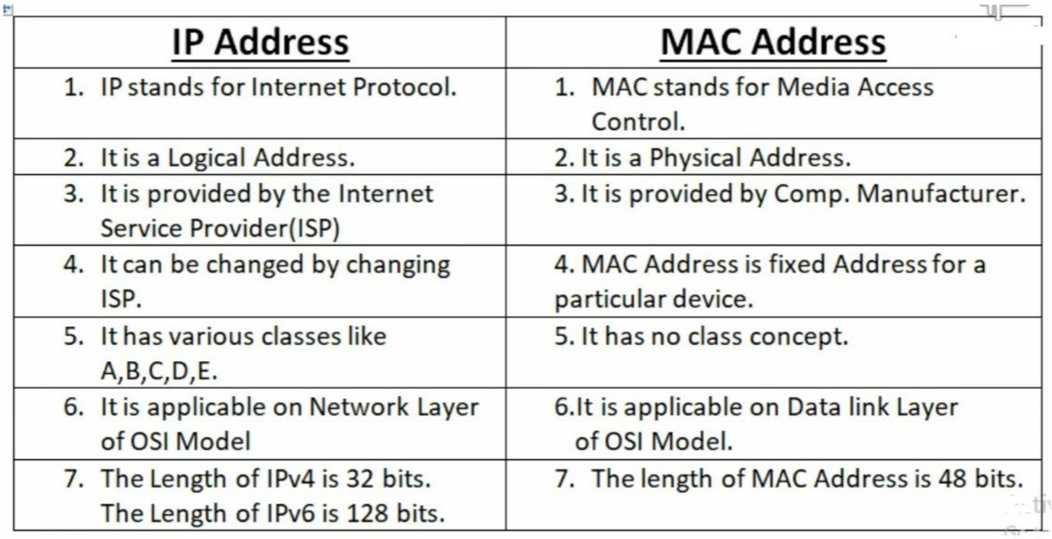


Data-link layer only deals with efficient transmission of information between adjacent machines in the network that are directly connected to each other.

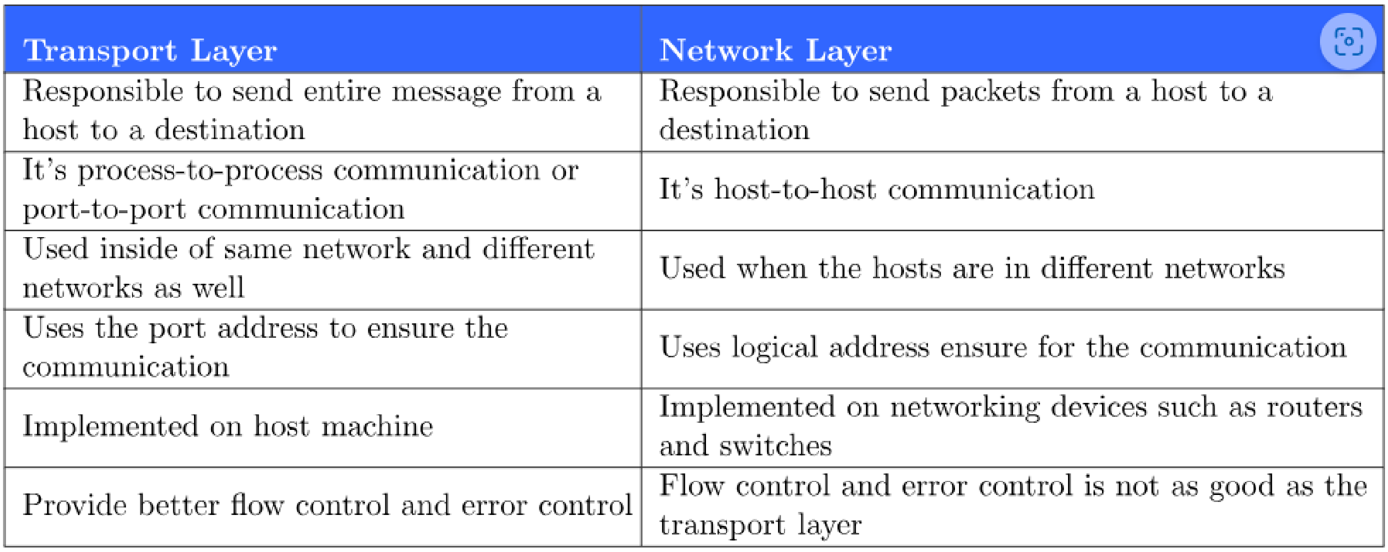
Network layer, which employs the services of the data-link layer, provides end-to-end connectivity between machines that are not necessarily directly connected.



Features	Frames	Packets
Definition	A frame is a type of data unit that is utilized in the data link layer.	A packet is a protocol data unit utilized in the network layer.
Includes	It has the source and destination MAC address.	It has the source and destination IP address.
Associated OSI layer	Frames are created in the OSI's data link layer.	The packets are created in the network layer.
Layer	It is associated with Layer 2.	It is associated with Layer 3.
Addressing	It has physical addressing.	It has logical addressing.



IP Address	MAC Address
1. IP stands for Internet Protocol.	1. MAC stands for Media Access Control.
2. It is a Logical Address.	2. It is a Physical Address.
3. It is provided by the Internet Service Provider(ISP)	3. It is provided by Comp. Manufacturer.
4. It can be changed by changing ISP.	4. MAC Address is fixed Address for a particular device.
5. It has various classes like A,B,C,D,E.	5. It has no class concept.
6. It is applicable on Network Layer of OSI Model	6.It is applicable on Data link Layer of OSI Model.
7. The Length of IPv4 is 32 bits. The Length of IPv6 is 128 bits.	7. The length of MAC Address is 48 bits.



Transport Layer	Network Layer
Responsible to send entire message from a host to a destination	Responsible to send packets from a host to a destination
It's process-to-process communication or port-to-port communication	It's host-to-host communication
Used inside of same network and different networks as well	Used when the hosts are in different networks
Uses the port address to ensure the communication	Uses logical address ensure for the communication
Implemented on host machine	Implemented on networking devices such as routers and switches
Provide better flow control and error control	Flow control and error control is not as good as the transport layer

1. Store and Forward packet Switching

The major components of the network are the ISP's equipment (routers connected by transmission lines), shown inside the shaded oval, and the customers' equipment, shown outside the oval. Host *H1* is directly connected to one of the ISP's routers, *A*, perhaps as a home computer that is plugged into a DSL modem. In contrast, *H2* is on a LAN, which might be an office Ethernet, with a router, *F*, owned and operated by the customer. This router has a leased line to the ISP's equipment. We have shown *F* as being outside the oval because it does not belong to the ISP.

A host with a packet to send transmits it to the nearest router, either on its own LAN or over a point-to-point link to the ISP. The packet is stored there until it has fully arrived and the link has finished its processing by verifying the checksum. Then it is forwarded to the next router along the path until it reaches the destination host, where it is delivered. This mechanism is store-and-forward packet switching.

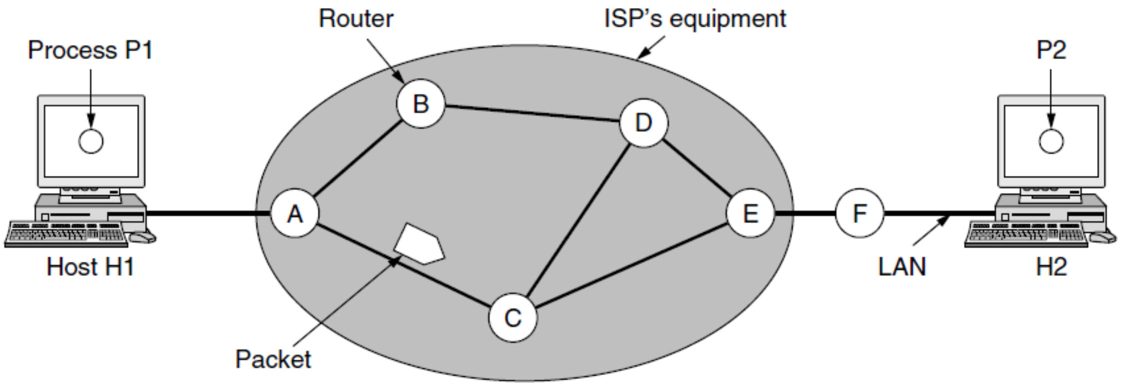


Figure 5-1. The environment of the network layer protocols.

2. Services Provided to the Transport Layer

The network layer provides services to the transport layer at the network layer/transport layer interface. An important question is precisely what kind of services the network layer provides to the transport layer. The services need to be carefully designed with the following goals in mind:

1. The services should be independent of the router technology.

2. The transport layer should be shielded from the number, type, and topology of the routers present.

3. The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.

1) One camp (represented by the Internet community) argues:

the routers' job is moving packets around and nothing else. Therefore, the hosts should accept this fact and do error control (i.e., error detection and correction) and flow control themselves. The network service should be connectionless, with primitives SEND PACKET and RECEIVE PACKET and little else. In particular, no packet ordering and flow control should be done, because the hosts are going to do that anyway. each packet must carry the full destination address, because each packet sent is carried independently of its predecessors, if any.

2) The other camp (represented by the telephone companies) argues:

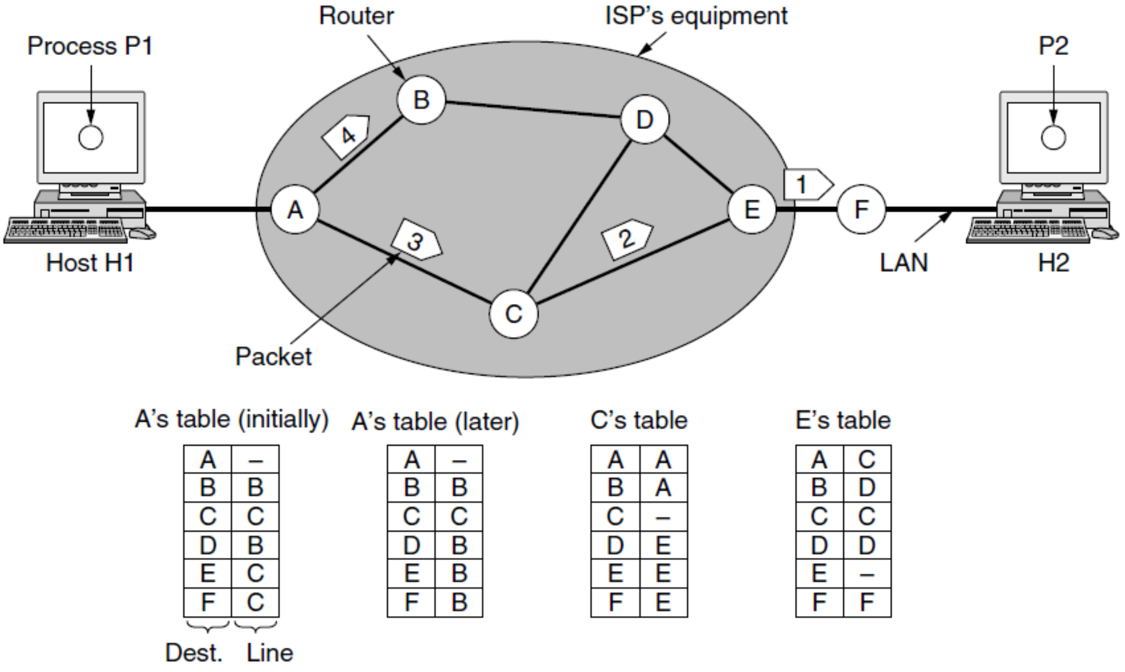
the network should provide a reliable, connection-oriented service. In this view, quality of service is the dominant factor, and without connections in the network, quality of service is very difficult to achieve, especially for real-time traffic such as voice and video.

3. Implementation of Connectionless Service

If **connectionless service** is offered, packets are injected into the network individually and routed independently of each other. No advance setup is needed. In this context, the packets are frequently called **datagrams** (in analogy with telegrams) and the network is called a

datagram network. If **connection-oriented service** is used, a path from the source router all the way to the destination router must be established before any data packets can be sent. This connection is called a **VC** (**virtual circuit**), in analogy with the physical circuits set up by the telephone system, and the network is called a **virtual-circuit network**.

Let us now see how a datagram network works. Suppose that the process *P1* in Fig. 5-2 has a long message for *P2*. It hands the message to the transport layer, with instructions to deliver it to process *P2* on host *H2*. The transport layer code runs on *H1*, typically within the operating system. It prepends a transport header to the front of the message and hands the result to the network layer. Assume the message is four times longer than the maximum packet size, so the network layer has to break it into four packets, 1, 2, 3, and 4, and send each of them in turn to router *A* using some point-to-point protocol, for example, PPP. At this point the ISP takes over. Every router has an internal table telling it where to send packets for each of the possible destinations. Each table entry is a pair consisting of a destination and the outgoing line to use for that destination. Only directly connected lines can be used. For example, in Fig. 5-2, *A* has only two outgoing lines—to *B* and to *C*—so every incoming packet must be sent to one of these routers, even if the ultimate destination is to some other router.



A's table (initially)

Dest.	Line
A	–
B	B
C	C
D	B
E	C
F	C

A's table (later)

A	–
B	B
C	C
D	B
E	B
F	B

C's table

A	A
B	A
C	–
D	E
E	E
F	E

E's table

A	C
B	D
C	C
D	D
E	–
F	F

Figure 5-2. Routing within a datagram network.

and had their checksums verified. Then each packet is forwarded according to *A*'s table, onto the outgoing link to *C* within a new frame. Packet 1 is then forwarded to *E* and then to *F*. When it gets to *F*, it is sent within a frame over the LAN to *H2*. Packets 2 and 3 follow the same route. However, something different happens to packet 4. When it gets to *A* it is sent to router *B*, even though it is also destined for *F*. For some reason, *A* decided to send packet 4 via a different route than that of the first three packets. Perhaps it has learned of a traffic jam somewhere along the *ACE* path and updated its routing table, as shown under the label ‘‘later.’’ The algorithm that manages the tables and makes the routing decisions is called the **routing algorithm.** IP (Internet Protocol), which is the basis for the entire Internet, is the dominant example of a connectionless network service. Each packet carries a destination IP address that routers use to individually forward each packet.

4. Implementation of Connection Oriented Service

For connection-oriented service, we need a virtual-circuit network. A virtual circuit (VC) is a means of transporting data over a data network, based on packet switching where a virtual path is established between the source and the destination systems for data communication to occur. Let us see how that works. The idea behind virtual circuits is to avoid having to choose a new route for every packet sent, as in Fig. 5-2. Instead, when a connection is established, a route from the source machine to the destination machine is chosen as part of the connection setup and stored in tables inside the routers. That route is used for all traffic flowing over the connection, exactly the same way that the telephone system works. When the connection is released, the virtual circuit is also terminated. With connection-oriented service, each packet carries an identifier telling which virtual circuit it belongs to.

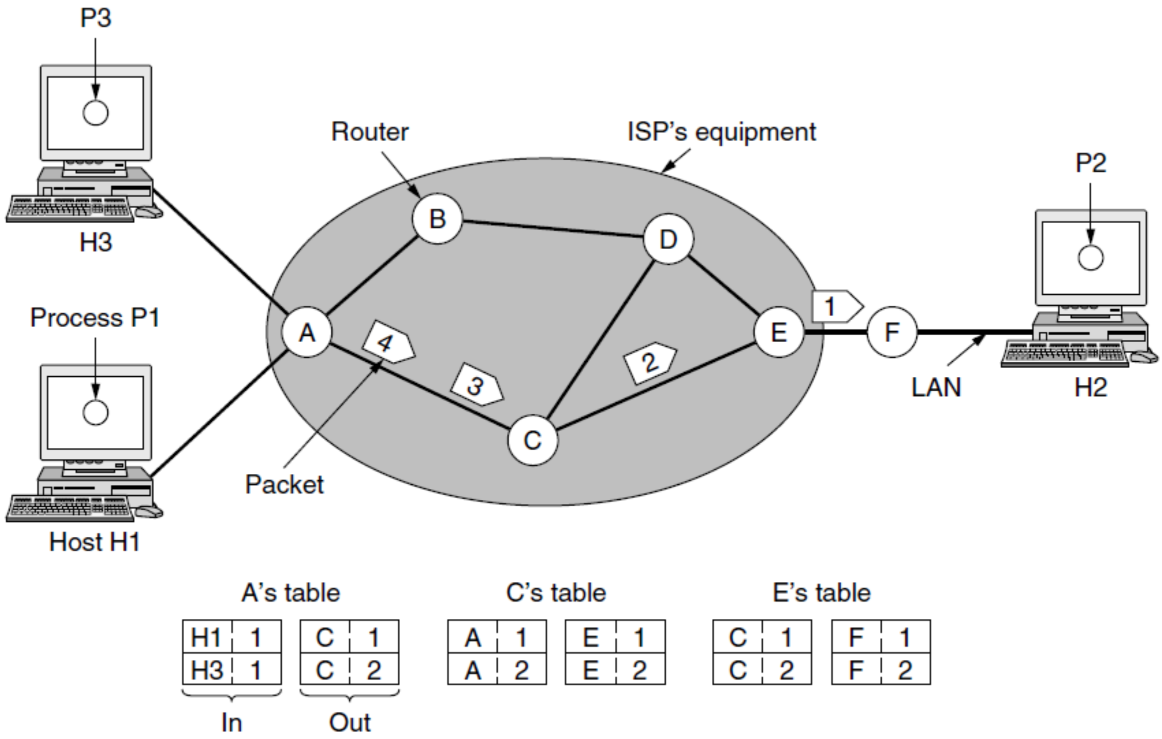
As an example, consider the situation shown in Fig. 5-3. Here, host *H1* has established connection 1 with host *H2*. This connection is remembered as the first entry in each of the routing tables. The first line of *A*’s table says that if a packet bearing connection identifier 1 comes in from *H1*, it is to be sent to router *C* and given connection identifier 1. Similarly, the first entry at *C* routes the packet to *E*, also with connection identifier 1.

Now let us consider what happens if *H3* also wants to establish a connection to *H2*. It chooses connection identifier 1 (because it is initiating the connection and this is its only connection) and tells the network to establish the virtual circuit. This leads to the second row in the tables. Note that we have a conflict here because although *A* can easily distinguish connection 1 packets from *H1* and connection 1 packets from *H3*, *C* cannot do this.

For this reason, *A* assigns a different connection identifier to the outgoing traffic for the second connection. Avoiding conflicts of this kind is why routers need the ability to replace connection identifiers in outgoing packets. In some contexts, this process is called **label switching**.

The ability of the router to replace connection identifiers in outgoing packets is called label switching.

An example of a connection-oriented network service is **MPLS** (**MultiProtocol Label Switching**). It is used within ISP networks in the Internet, with IP packets wrapped in an MPLS header having a 20-bit connection identifier or label.



A's table

In		Out	
H1	1	C	1
H3	1	C	2

C's table

A	1	E	1
A	2	E	2

E's table

C	1	F	1
C	2	F	2

Figure 5-3. Routing within a virtual-circuit network.

5. Comparison of Virtual Circuit and Datagram Subnets

Issue	Datagram network	Virtual-circuit network
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Routers do not hold state information about connections	Each VC requires router table space per connection
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow it
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Quality of service	Difficult	Easy if enough resources can be allocated in advance for each VC
Congestion control	Difficult	Easy if enough resources can be allocated in advance for each VC

Routing algorithms:

The main function of the network layer is routing packets from the source machine to the destination machine. In most networks, packets will require multiple hops to make the journey. The only notable exception is for broadcast networks, but even here routing is an issue if the source and destination are not on the same network segment. The algorithms that choose the routes and the data structures that they use are a major area of network layer design.

The **routing algorithm** is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on. If the network uses datagrams internally, this decision must be made anew for every arriving data packet since the best route may have changed since last time. If the network uses virtual circuits internally, routing decisions are made only when a new virtual circuit is being set up. Thereafter, data packets just follow the already established route. The latter case is sometimes called **session routing** because a route remains in force for an entire session.

It is sometimes useful to make a distinction between routing, which is making the decision which routes to use, and forwarding, which is what happens when a packet arrives. One can think of a router as having two processes inside it. One of them handles each packet as it arrives, looking up the outgoing line to use for it in the routing tables. This process is **forwarding**. The other process is responsible for filling in and updating the routing tables. That is where the routing algorithm comes into play.

Regardless of whether routes are chosen independently for each packet sent or only when new connections are established, certain properties are desirable in a routing algorithm: correctness, simplicity, robustness, stability, fairness, and efficiency.

The routing algorithm should be able to cope with changes in the topology and traffic without requiring all jobs in all hosts to be aborted. Routing algorithms can be grouped into two major classes: nonadaptive and adaptive. **Nonadaptive algorithms** do not base their routing decisions on any measurements or estimates of the current topology and traffic. Instead, the choice of the route to use to get from *I* to *J* (for all *I* and *J*) is computed in advance, offline, and downloaded to the routers when the network is booted. This procedure is sometimes called **static routing**. Because it does not respond to failures, static routing is mostly useful for situations in which the routing choice is clear.

Adaptive algorithms, in contrast, change their routing decisions to reflect changes in the topology, and sometimes changes in the traffic as well. These **dynamic routing** algorithms differ in where they get their information (e.g., locally, from adjacent routers, or from all routers), when they change the routes (e.g., when the topology changes, or every ΔT seconds as the load changes), and what metric is used for optimization (e.g., distance, number of hops, or estimated transit time).

1. Shortest Path Routing

It is a simple technique for computing optimal paths given a complete picture of the network. These paths are the ones that we want a distributed routing algorithm to find, even though not all routers may know all of the details of the network.

The idea is to build a graph of the network, with each node of the graph representing a router and each edge of the graph representing a communication line, or link. To choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph. The concept of a **shortest path** deserves some explanation. One way of measuring path length is the number of hops. Using this metric, the paths *ABC* and *ABE* in Fig. 5-7 are equally long. Another metric is the geographic distance in kilometers, in which case *ABC* is clearly much longer than *ABE* (assuming the figure is drawn to scale).

In the general case, the labels on the edges could be computed as a function of the distance, bandwidth, average traffic, communication cost, measured delay, and other factors. By changing the weighting function, the algorithm would then compute the ''shortest'' path measured according to any one of a number of criteria or to a combination of criteria. Several algorithms for computing the shortest path between two nodes of a graph are known. This one is due to Dijkstra (1959) and finds the shortest paths between a source and all destinations in the network. Each node is labelled (in parentheses) with its distance from the source node along the best known path. Initially, no paths are known, so all nodes are labelled with infinity. As the algorithm proceeds and paths are found, the labels may change, reflecting better paths. A label may be either tentative or permanent. Initially, all labels are tentative. When it is discovered that a label represents the shortest possible path from the source to that node, it is made permanent and never changed thereafter.

To illustrate how the labelling algorithm works, look at the weighted, undirected graph of Fig. 5-7(a), where the weights represent, for example, distance. We want to find the shortest path from *A* to *D*. We start out by marking node *A* as permanent, indicated by a filled-in circle. Then we examine, in turn, each of the nodes adjacent to *A* (the working node), relabelling each one with the distance to *A*. Whenever a node is relabelled, we also label it with the node from which the probe was made so that we can reconstruct the final path later. If the network had more than one shortest path from *A* to *D* and we wanted to find all of them, we would need to remember all of the probe nodes that could reach a node with the same distance.

Having examined each of the nodes adjacent to *A*, we examine all the tentatively labelled nodes in the whole graph and make the one with the smallest label permanent, as shown in Fig. 5-7(b). This one becomes the new working node. We now start at *B* and examine all nodes adjacent to it. If the sum of the label on *B* and the distance from *B* to the node being considered is less than the label on that node, we have a shorter path, so the node is relabelled. After all the nodes adjacent to the working node have been inspected and the tentative labels changed if possible, the entire graph is searched for the tentatively labelled node with the smallest value. This node is made permanent and becomes the working node for the next round. Figure 5-7 shows the first six steps of the algorithm.

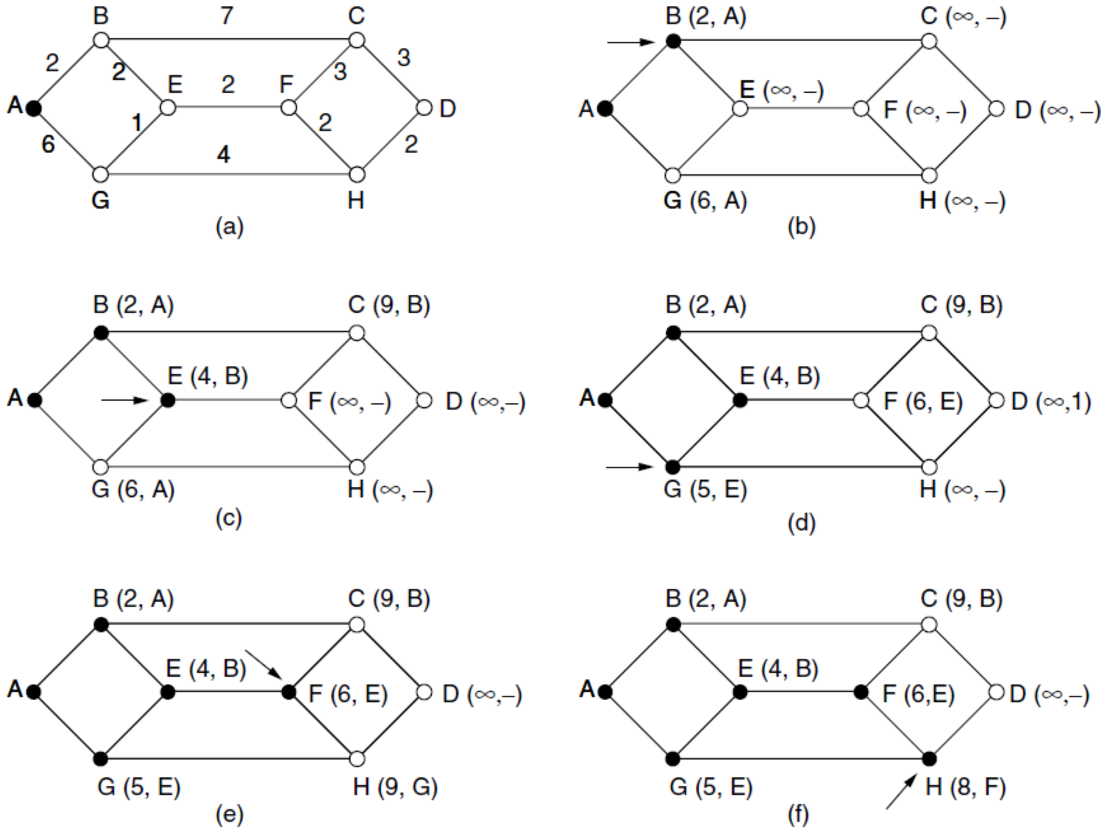


Figure 5-7. The first six steps used in computing the shortest path from *A* to *D*. The arrows indicate the working node.

Since the shortest paths from *t* to *s* in an undirected graph are the same as the shortest paths from *s* to *t*, it does not matter at which end we begin. The reason for searching backward is that each node is labelled with its predecessor rather than its successor. When the final path is copied into the output variable, *path*, the path is thus reversed. The two reversal effects cancel, and the answer is produced in the correct order.

2. Flooding

When a routing algorithm is implemented, each router must make decisions based on local knowledge, not the complete picture of the network. A simple local technique is **flooding**, in which every incoming packet is sent out on ever outgoing line except the one it arrived on. Flooding obviously generates vast numbers of duplicate packets, in fact, an infinite number unless some measures are taken to damp the process. One such measure is to have a hop counter contained in the header of each packet that is decremented at each hop, with the packet being discarded when the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the network. Flooding with a hop count can produce an exponential number of duplicate packets as the hop count grows and routers duplicate packets they have seen before. A better technique for damming the flood is to have routers keep track of which packets have been flooded, to avoid sending them out a second time. One way to achieve this goal is to have the source router put a sequence number in each packet it receives from its hosts. Each router then needs a list per source router telling which sequence numbers originating at that source have already been seen. If an incoming packet is on the list, it is not flooded. Flooding is not practical for sending most packets, but it does have some important uses. First, it ensures that a packet is delivered to every node in the network. This may be wasteful if there is a single destination that needs the packet, but it is effective for broadcasting information. In wireless networks, all messages transmitted by a station can be received by all other stations within its radio range, which is, in fact, flooding, and some algorithms utilize this property.

Second, flooding is tremendously robust. Even if large numbers of routers are blown to bits (e.g., in a military network located in a war zone), flooding will find a path if one exists, to get a packet to its destination. Flooding also requires little in the way of setup. The routers only need to know their neighbours. This means that flooding can be used as a building block for other routing algorithms that are more efficient but need more in the way of setup. Flooding can also be used as a metric against which other routing algorithms can be compared. Flooding always chooses the shortest path because it chooses every possible path in parallel.

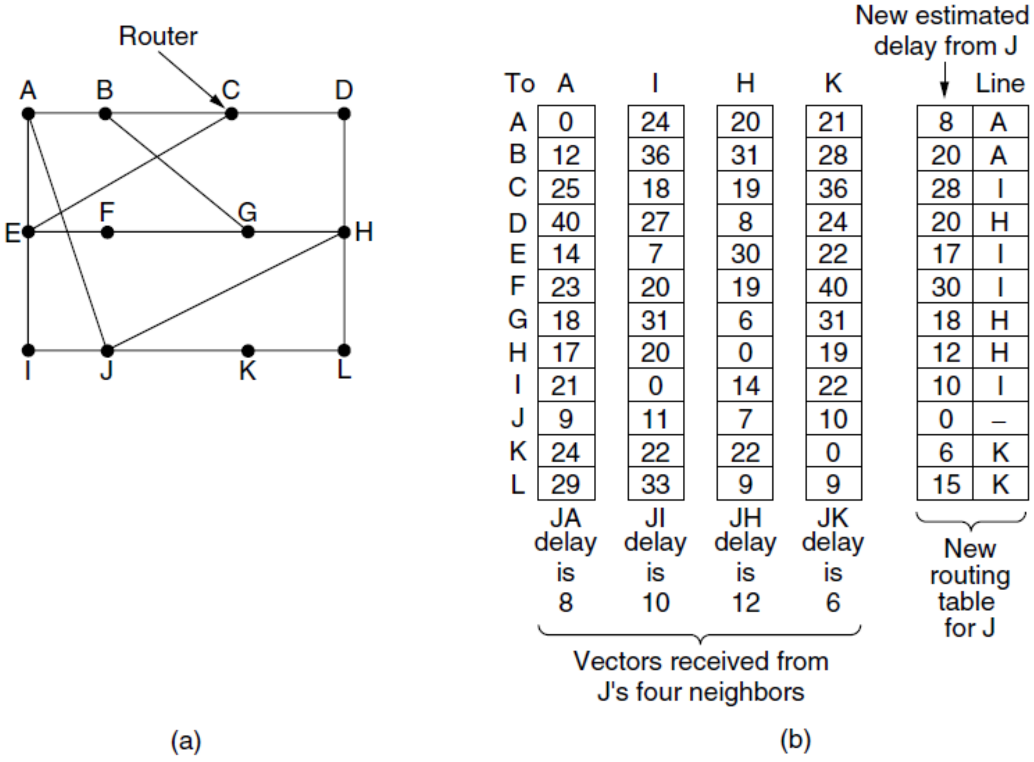
3. Distance Vector Routing

Computer networks generally use dynamic routing algorithms that are more complex than flooding, but more efficient because they find shortest paths for the current topology. Two dynamic algorithms in particular, distance vector routing and link state routing, are the most popular. A **distance vector routing** algorithm operates by having each router maintain a table (i.e., a vector) giving the best known distance to each destination and which link to use to get there. These tables are updated by exchanging information with the neighbours. Eventually, every router knows the best link to reach each destination. The distance vector routing algorithm is also called as distributed **Bellman-Ford** routing algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962). It was the original ARPANET routing algorithm. In distance vector routing, each router maintains a routing table indexed by, and containing one entry for each router in the network. This entry has two parts:

the preferred outgoing line to use for that destination and an estimate of the distance to that destination. The distance might be measured as the number of hops or using another metric. The router is assumed to know the ‘‘distance’’ to each of its neighbours. If the metric is hops, the distance is just one hop. If the metric is propagation delay, the router can measure it directly with special ECHO packets that the receiver just timestamps and sends back as fast as it can.

As an example, assume that delay is used as a metric and that the router knows the delay to each of its neighbours. Once every *T* msec, each router sends to each neighbour a list of its estimated delays to each destination. It also receives a similar list from each neighbour. Imagine that one of these tables has just come in from neighbour *X*, with *Xi* being *X*’s estimate of how long it takes to get to router *i*. If the router knows that the delay to *X* is *m* msec, it also knows that it can reach router *i* via *X* in *Xi* + *m* msec. By performing this calculation for each neighbour, a router can find out which estimate seems the best and use that estimate and the corresponding link in its new routing table. Note that the old routing table is not used in the calculation.

This updating process is illustrated in Fig. 5-9. Part (a) shows a network. The first four columns of part (b) show the delay vectors received from the neighbours of router *J*. *A* claims to have a 12-msec delay to *B*, a 25-msec delay to *C*, a 40- msec delay to *D*, etc. Suppose that *J* has measured or estimated its delay to its neighbours, *A*, *I, H*, and *K*, as 8, 10, 12, and 6 msec, respectively.



To	A	I	H	K	New estimated delay from J	Line
A	0	24	20	21	8	A
B	12	36	31	28	20	A
C	25	18	19	36	28	I
D	40	27	8	24	20	H
E	14	7	30	22	17	I
F	23	20	19	40	30	I
G	18	31	6	31	18	H
H	17	20	0	19	12	H
I	21	0	14	22	10	I
J	9	11	7	10	0	–
K	24	22	22	0	6	K
L	29	33	9	9	15	K
	JA delay is 8	JI delay is 10	JH delay is 12	JK delay is 6	New routing table for J	

Vectors received from J's four neighbors

Figure 5-9. (a) A network. (b) Input from *A*, *I*, *H*, *K*, and the new routing table for *J*.

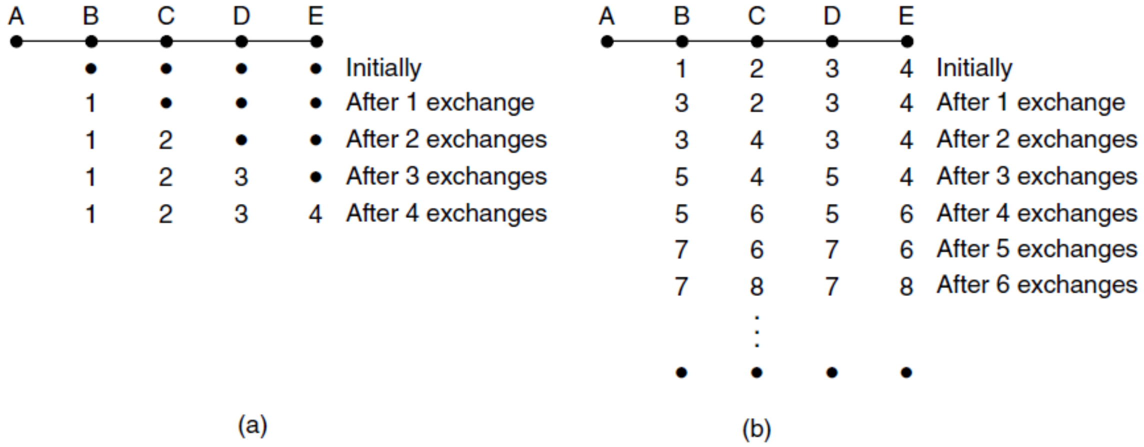
Consider how *J* computes its new route to router *G*. It knows that it can get to *A* in 8 msec, and furthermore *A* claims to be able to get to *G* in 18 msec, so *J* knows it can count on a delay of 26 msec to *G* if it forwards packets bound for *G* to *A*. Similarly, it computes the delay to *G* via *I*, *H*, and *K* as 41 (31 + 10), 18 (6 + 12), and 37 (31 + 6) msec, respectively. The best of these values is 18, so it makes an entry in its routing table that the delay to *G* is 18 msec and that the route to use is via *H*. The same calculation is performed for all the other destinations, with the new routing table shown in the last column of the figure.

The Count-to-Infinity Problem

The "Count-to-Infinity" problem is a scenario that can occur in computer networking protocols, particularly in distance-vector routing algorithms such as the Routing Information Protocol (RIP). It arises when there is a network topology change, but the routing information does not propagate quickly or efficiently throughout the network. This can result in routers incorrectly believing that they have found the shortest path to a destination and creating routing loops.

The Count-to-Infinity problem can result in significant network instability, increased network traffic, and delayed convergence. It is a fundamental limitation of distance-vector routing algorithms that do not have mechanisms to detect and prevent routing loops.

The settling of routes to best paths across the network is called **convergence**. Convergence or routing convergence is a state in which a set of routers in a network share the same topological information. Distance vector routing is useful as a simple technique by which routers can collectively compute shortest paths, but it has a serious drawback in practice: although it converges to the correct answer, it may do so slowly. In particular, it reacts rapidly to good news, but leisurely to bad news. Consider a router whose best route to destination *X* is long. If, on the next exchange, neighbour *A* suddenly reports a short delay to *X*, the router just switches over to using the line to *A* to send traffic to *X*. In one vector exchange, the good news is processed. To see how fast good news propagates, consider the five-node (linear) network of Fig. 5-10, where the delay metric is the number of hops. Suppose *A* is down initially and all the other routers know this. In other words, they have all recorded the delay to *A* as infinity.



A	B	C	D	E	
	•	•	•	•	Initially
	1	•	•	•	After 1 exchange
	1	2	•	•	After 2 exchanges
	1	2	3	•	After 3 exchanges
	1	2	3	4	After 4 exchanges

(a)

A	B	C	D	E	
	1	2	3	4	Initially
	3	2	3	4	After 1 exchange
	3	4	3	4	After 2 exchanges
	5	4	5	4	After 3 exchanges
	5	6	5	6	After 4 exchanges
	7	6	7	6	After 5 exchanges
	7	8	7	8	After 6 exchanges
		⋮			
	•	•	•	•	

(b)

Figure 5-10. The count-to-infinity problem.

When A comes up, the other routers learn about it via the vector exchanges. For simplicity, we will assume that there is a gigantic gong somewhere that is struck periodically to initiate a vector exchange at all routers simultaneously. At the time of the first exchange, *B* learns that its left-hand neighbour has zero delay to *A*. *B* now makes an entry in its routing table indicating that *A* is one hop away to the left. All the other routers still think that *A* is down. At this point, the routing table entries for *A* are as shown in the second row of Fig. 5-10(a). On the next exchange, *C* learns that *B* has a path of length 1 to *A*, so it updates its routing table to indicate a path of length 2, but *D* and *E* do not hear the good news until later. Clearly, the **good news is spreading at the rate of one hop per exchange.** In a network whose longest path is of length *N* hops, within *N* exchanges everyone will know about newly revived links and routers. Now let us consider the situation of Fig. 5-10(b), in which all the links and routers are initially up. Routers *B*, *C*, *D*, and *E* have distances to *A* of 1, 2, 3, and 4 hops, respectively. Suddenly, either *A* goes down or the link between *A* and *B* is cut (which is effectively the same thing from *B*'s point of view). At the first packet exchange, *B* does not hear anything from *A*. Fortunately, *C* says ''Do not worry; I have a path to *A* of length 2''. Little does *B* suspect that *C*'s path runs through *B* itself. For all *B* knows, *C* might have ten links all with separate paths to *A* of length 2. As a result, *B* thinks it can reach *A* via *C*, with a path length of 3. *D* and *E* do not update their entries for *A* on the first exchange. On the second exchange, *C* notices that each of its neighbours claims to have a path to *A* of length 3. It picks one of them at random and makes its new distance to *A* 4, as shown in the third row of Fig. 5-10(b). Subsequent exchanges produce the history shown in the rest of Fig. 5-10(b).

4. Link state Routing

Distance vector routing was used in the ARPANET until 1979, when it was replaced by link state routing. (The Advanced Research Projects Agency Network was the first wide-area packet-switched network with distributed control and one of the first networks to implement the TCP/IP protocol suite)

The primary problem that caused its demise was that the algorithm often took too long to converge after the network topology changed (due to the count-to-infinity problem). Consequently, it was replaced by an entirely new algorithm, now called **link state routing**. Variants of link state routing called IS-IS and OSPF are the routing algorithms that are most widely used inside large networks and the Internet today.

IS-IS (Intermediate System to Intermediate System)

OSPF (Open Shortest Path First) are two popular link-state routing protocols

The idea behind link state routing is fairly simple and can be stated as five parts. Each router must do the following things to make it work:

1. Discover its neighbours and learn their network addresses.
2. Set the distance or cost metric to each of its neighbours.
3. Construct a packet telling all it has just learned.
4. Send this packet to and receive packets from all other routers.
5. Compute the shortest path to every other router.

In effect, the complete topology is distributed to every router. Then Dijkstra’s algorithm can be run at each router to find the shortest path to every other router.

Dijkstra’s – Shortest Path Algorithm (SPT)- Animation - YouTube

1. Learning about the Neighbours

When a router is booted, its first task is to learn who its neighbours are. It accomplishes this goal by sending a special HELLO packet on each point-to-point line. The router on the other end is expected to send back a reply giving its name. These names must be globally unique because when a distant router later hears that three routers are all connected to *F*, it is essential that it can determine whether all three mean the same *F*.

When two or more routers are connected by a broadcast link (e.g., a switch, ring, or classic Ethernet), the situation is slightly more complicated. Fig. 5-11(a) illustrates a broadcast LAN to which three routers, *A*, *C*, and *F*, are directly connected. Each of these routers is connected to one or more additional routers, as shown.

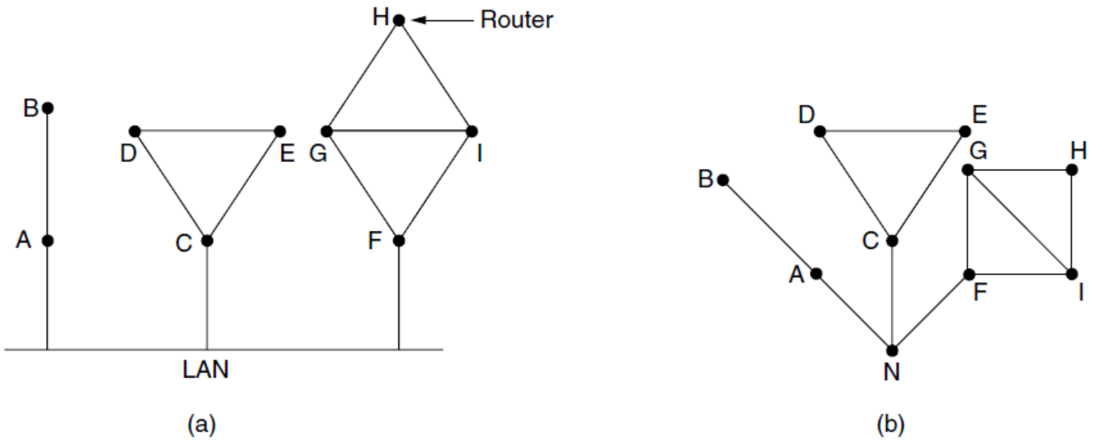


Figure 5-11. (a) Nine routers and a broadcast LAN. (b) A graph model of (a).

The broadcast LAN provides connectivity between each pair of attached routers. However, modelling the LAN as many point-to-point links increases the size of the topology and leads to wasteful messages. A better way to model the LAN is to consider it as a node itself, as shown in Fig. 5-11(b). Here, we have introduced a new, artificial node, *N*, to which *A*, *C*, and *F* are connected. One **designated router** on the LAN is selected to play the role of *N* in the routing protocol. The fact that it is possible to go from *A* to *C* on the LAN is represented by the path *ANC* here.

2. Setting Link Costs

The link state routing algorithm requires each link to have a distance or cost metric for finding shortest paths. The cost to reach neighbours can be set automatically, or configured by the network operator. A common choice is to make the cost inversely proportional to the bandwidth of the link. For example, 1-Gbps Ethernet may have a cost of 1 and 100-Mbps Ethernet a cost of 10. This makes higher-capacity paths better choices. If the network is geographically spread out, the delay of the links may be factored into the cost so that paths over shorter links are better choices. The most direct way to determine this delay is to send over the line a special ECHO packet that the other side is required to send back immediately. By measuring the round-trip time and dividing it by two, the sending router can get a reasonable estimate of the delay.

3. Building Link State Packets

Once the information needed for the exchange has been collected, the next step is for each router to build a packet containing all the data. The packet starts with the identity of the sender, followed by a sequence number and age and a list of neighbours. The cost to each neighbour is also given. An example network is presented in Fig. 5-12(a) with costs shown as labels on the lines. The corresponding link state packets for all six routers are shown in Fig. 5- 12(b).

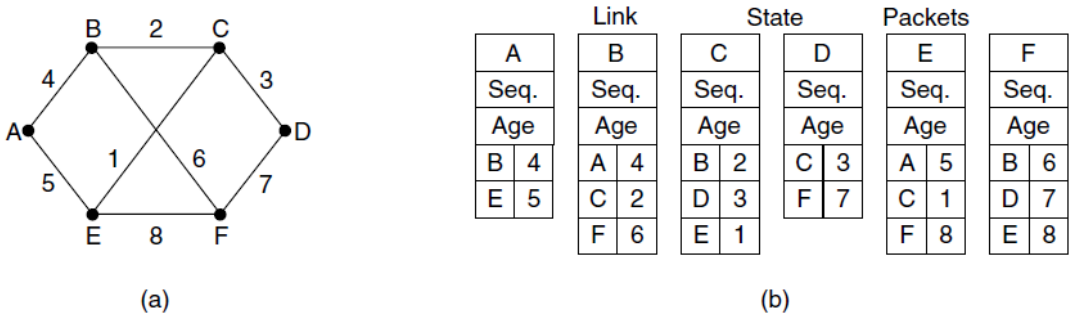


Figure 5-12. (a) A network. (b) The link state packets for this network.

Building the link state packets is easy. The hard part is determining when to build them. One possibility is to build them periodically, that is, at regular intervals. Another possibility is to build them when some significant event occurs, such as a line or neighbour going down or coming back up again or changing its properties appreciably.

4. Distributing the Link State Packets

The trickiest part of the algorithm is distributing the link state packets. All of the routers must get all of the link state packets quickly and reliably. If different routers are using different versions of the topology, the routes they compute can have inconsistencies such as loops, unreachable machines, and other problems. The fundamental idea is to use flooding to distribute the link state packets to all routers. To keep the flood in check, each packet contains a sequence number that is incremented for each new packet sent. Routers keep track of all the (source router, sequence) pairs they see. When a new link state packet comes in, it is checked against the list of packets already seen. If it is new, it is forwarded on all lines except the one it arrived on. If it is a duplicate, it is discarded. If a packet with a sequence number lower than the highest one seen so far ever arrives, it is rejected as being obsolete as the router has more recent data.

This algorithm has a few problems, but they are manageable. First, if the sequence numbers wrap around, confusion will reign. The solution here is to use a 32-bit sequence number. With one link state packet per second, it would take 137 years to wrap around, so this possibility can be ignored. Second, if a router ever crashes, it will lose track of its sequence number. If it starts again at 0, the next packet it sends will be rejected as a duplicate. Third, if a sequence number is ever corrupted and 65,540 is received instead of 4 (a 1-bit error), packets 5 through 65,540 will be rejected as obsolete, since the current sequence number will be thought to be 65,540.

The solution to all these problems is to include the age of each packet after the sequence number and decrement it once per second. When the age hits zero, the information from that router is discarded. Normally, a new packet comes in, say, every 10 sec, so router information only times out when a router is down (or six consecutive packets have been lost, an unlikely event). The *Age* field is also decremented by each router during the initial flooding process, to make sure no packet can get lost and live for an indefinite period of time (a packet whose age is zero is discarded). Some refinements to this algorithm make it more robust. When a link state packet comes in to a router for flooding, it is not queued for transmission immediately. Instead, it is put in a holding area to wait a short while in case more links are coming up or going down. If another link state packet from the same source comes in before the first packet is transmitted, their sequence numbers are compared. If they are equal, the duplicate is discarded. If they are different, the older one is thrown out. To guard against errors on the links, all link state packets are acknowledged. The data structure used by router *B* for the network shown in Fig. 5-12(a) is depicted in Fig. 5-13.

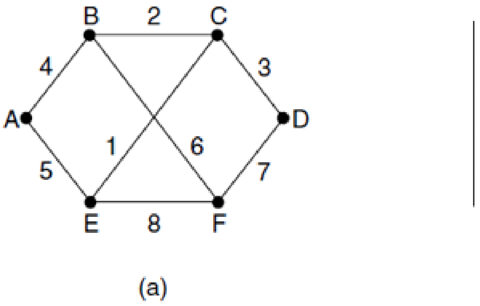


Figure 5-12. (a) A network

Source	Seq.	Age	Send flags			ACK flags			Data
			A	C	F	A	C	F	
A	21	60	0	1	1	1	0	0	
F	21	60	1	1	0	0	0	1	
E	21	59	0	1	0	1	0	1	
C	20	60	1	0	1	0	1	0	
D	21	59	1	0	0	0	1	1	

Figure 5-13. The packet buffer for router *B* in Fig. 5-12(a).

Each row here corresponds to a recently arrived, but as yet not fully processed, link state packet. The table records where the packet originated, its sequence number and age, and the data. In addition, there are send and acknowledgement flags for each of ***B***'s three links (to *A*, *C*, and *F*, respectively). The send flags mean that the packet must be sent on the indicated link. The acknowledgement flags mean that it must be acknowledged there. In Fig. 5-13, the link state packet from *A* arrives directly, so it must be sent to *C* and *F* and acknowledged to *A*, as indicated by the flag bits. Similarly, the packet from *F* has to be forwarded to *A* and *C* and acknowledged to *F*. However, the situation with the third packet, from *E*, is different. It arrives twice, once via *EAB* and once via *EFB*. Consequently, it has to be sent only to *C* but must be acknowledged to both *A* and *F*, as indicated by the bits. If a duplicate arrives while the original is still in the buffer, bits have to be changed. For example, if a copy of *C*'s state arrives from *F* before the fourth entry in the table has been forwarded, the six bits will be changed to 100011 to indicate that the packet must be acknowledged to *F* but not sent there.

5. Computing the New Routes

Once a router has accumulated a full set of link state packets, it can construct the entire network graph because every link is represented. Every link is, in fact, represented twice, once for each direction. The different directions may even have different costs. The shortest-path computations may then find different paths from router *A* to *B* than from router *B* to *A*. Now Dijkstra's algorithm can be run locally to construct the shortest paths to all possible destinations. The results of this algorithm tell the router which link to use to reach each destination. This information is installed in the routing tables, and normal operation is resumed.

Compared to distance vector routing, link state routing requires more memory and computation. For a network with *n* routers, each of which has *k* neighbours, the memory required to store the input data is proportional to *kn*, which is at least as large as a routing table listing all the destinations. Also, the computation time grows faster than *kn*, even with the most efficient data structures, an issue in large networks. Nevertheless, in many practical situations, link state routing works well because it does not suffer from slow convergence problems.

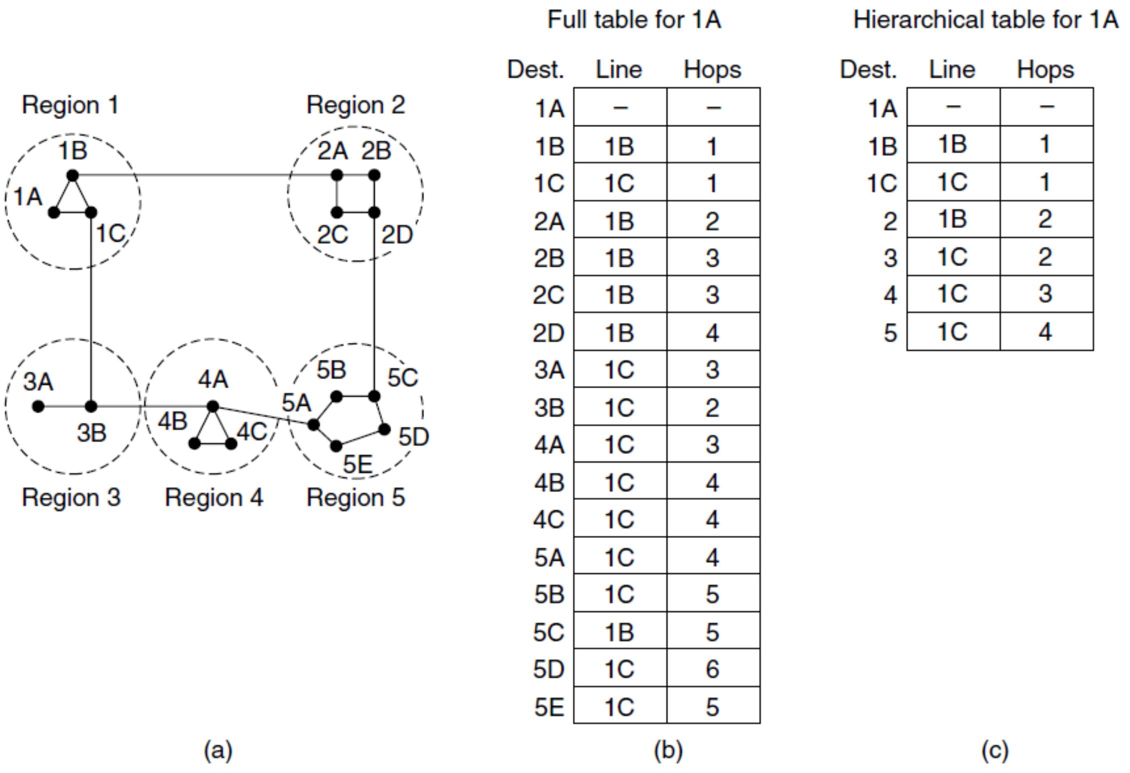
Link state routing is widely used in actual networks, so a few words about some example protocols are in order. Many ISPs use the **IS-IS** (**Intermediate System-Intermediate System**) link state protocol (Oran, 1990). It was designed for an early network called DECnet, later adopted by ISO for use with the OSI protocols and then modified to handle other protocols as well, most notably, IP.

OSPF (**Open Shortest Path First**) is the other main link state protocol. It was designed by IETF several years after IS-IS and adopted many of the innovations designed for IS-IS. These innovations include a self-stabilizing method of flooding link state updates, the concept of a designated router on a LAN, and the method of computing and supporting path splitting and multiple metrics. As a consequence, there is very little difference between IS-IS and OSPF. The most important difference is that IS-IS can carry information about multiple network layer protocols at the same time (e.g., IP, IPX, and AppleTalk). OSPF does not have this feature, and it is an advantage in large multiprotocol environments.

A general comment on routing algorithms is also in order. Link state, distance vector, and other algorithms rely on processing at all the routers to compute routes. Problems with the hardware or software at even a small number of routers can wreak havoc across the network. For example, if a router claims to have a link it does not have or forgets a link it does have, the network graph will be incorrect. If a router fails to forward packets or corrupts them while forwarding them, the route will not work as expected. Finally, if it runs out of memory or does the routing calculation wrong, bad things will happen. As the network grows into the range of tens or hundreds of thousands of nodes, the probability of some router failing occasionally becomes nonnegligible. The trick is to try to arrange to limit the damage when the inevitable happens. Perlman (1988) discusses these problems and their possible solutions in detail.

5. Hierarchical Routing

As networks grow in size, the router routing tables grow proportionally. Not only is router memory consumed by ever-increasing tables, but more CPU time is needed to scan them and more bandwidth is needed to send status reports about them. At a certain point, the network may grow to the point where it is no longer feasible for every router to have an entry for every other router, so the routing will have to be done hierarchically, as it is in the telephone network. When hierarchical routing is used, the routers are divided into **regions**. Each router knows all the details about how to route packets to destinations within its own region but knows nothing about the internal structure of other regions. When different networks are interconnected, it is natural to regard each one as a separate region to free the routers in one network from having to know the topological structure of the other ones. For huge networks, a two-level hierarchy may be insufficient; it may be necessary to group the regions into clusters, the clusters into zones, the zones into groups, and so on.



Full table for 1A

Dest.	Line	Hops
1A	–	–
1B	1B	1
1C	1C	1
2A	1B	2
2B	1B	3
2C	1B	3
2D	1B	4
3A	1C	3
3B	1C	2
4A	1C	3
4B	1C	4
4C	1C	4
5A	1C	4
5B	1C	5
5C	1B	5
5D	1C	6
5E	1C	5

Hierarchical table for 1A

Dest.	Line	Hops
1A	–	–
1B	1B	1
1C	1C	1
2	1B	2
3	1C	2
4	1C	3
5	1C	4

Figure 5-14. Hierarchical routing.

Figure 5-14 gives a quantitative example of routing in a two-level hierarchy with five regions. The full routing table for router *1A* has 17 entries, as shown in Fig. 5-14(b). When routing is done hierarchically, as in Fig. 5-14(c), there are entries for all the local routers, as before, but all other regions are condensed into a single router, so all traffic for region 2 goes via the *1B-2A* line, but the rest of the remote traffic goes via the *1C-3B* line. Hierarchical routing has reduced the table from 17 to 7 entries. As the ratio of the number of regions to the number of routers per region grows, the savings in table space increase.

Unfortunately, these gains in space are not free. There is a penalty to be paid: increased path length. For example, the best route from *1A* to *5C* is via region 2, but with hierarchical routing all traffic to region 5 goes via region 3, because that is better for most destinations in region 5. When a single network becomes very large, an interesting question is ‘‘how many levels should the hierarchy have?’’ For example, consider a network with 720 routers. If there is no hierarchy, each router needs 720 routing table entries. If the network is partitioned into 24 regions of 30 routers each, each router needs 30 local entries plus 23 remote entries for a total of 53 entries. If a three-level hierarchy is chosen, with 8 clusters each containing 9 regions of 10 routers, each router needs 10 entries for local routers, 8 entries for routing to other regions within its own cluster, and 7 entries for distant clusters, for a total of 25 entries. Kamoun and Kleinrock (1979) discovered that the optimal number of levels for an N router network is ln N, requiring a total of e ln N entries per router. They have also shown that the increase in effective mean path length caused by hierarchical routing is sufficiently small that it is usually acceptable.

6. Broadcast Routing

In some applications, hosts need to send messages to many or all other hosts. For example, a service distributing weather reports, stock market updates, or live radio programs might work best by sending to all machines and letting those that are interested read the data. Sending a packet to all destinations simultaneously is called **broadcasting**. Various methods have been proposed for doing it. One broadcasting method that requires no special features from the network is for the source to simply send a distinct packet to each destination. Not only is the method wasteful of bandwidth and slow, but it also requires the source to have a complete list of all destinations. This method is not desirable in practice, even though it is widely applicable. An improvement is **multi-destination routing**, in which each packet contains either a list of destinations or a bit map indicating the desired destinations. When a packet arrives at a router, the router checks all the destinations to determine the set of output lines that will be needed. (An output line is needed if it is the best route to at least one of the destinations.) The router generates a new copy of the packet for each output line to be used and includes in each packet only those destinations that are to use the line. In effect, the destination set is partitioned among the output lines. After a sufficient number of hops, each packet will carry only one destination like a normal packet. Multi-destination routing is like using separately addressed packets, except that when several packets must follow the same route, one of them pays full fare and the rest ride free. The network bandwidth is therefore used more efficiently. However, this scheme still requires the source to know all the destinations, plus it is as much work for a router to determine where to send one multi-destination packet as it is for multiple distinct packets.

The idea for **reverse path forwarding** is elegant and remarkably simple. When a broadcast packet arrives at a router, the router checks to see if the packet arrived on the link that is normally used for sending packets *toward* the source of the broadcast. If so, there is an excellent chance that the broadcast packet itself followed the best route from the router and is therefore the first copy to arrive at the router. This being the case, the router forwards copies of it onto all links except the one it arrived on. If, however, the broadcast packet arrived on a link other than the preferred one for reaching the source, the packet is discarded as a likely duplicate.

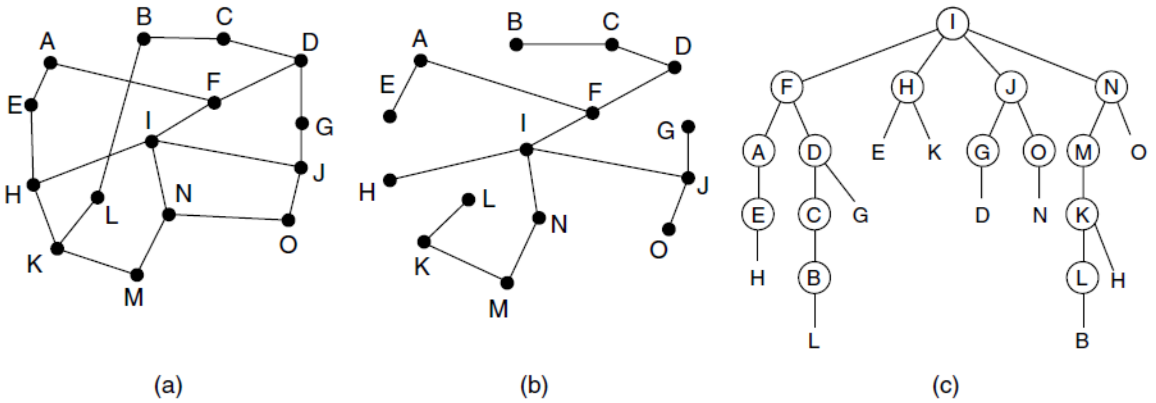


Figure 5-15. Reverse path forwarding. (a) A network. (b) A sink tree. (c) The tree built by reverse path forwarding.

An example of reverse path forwarding is shown in Fig. 5-15. Part (a) shows a network, part (b) shows a sink tree for router *I* of that network, and part (c) shows how the reverse path algorithm works. On the first hop, *I* sends packets to *F*, *H*, *J*, and *N*, as indicated by the second row of the tree. Each of these packets arrives on the preferred path to *I* (assuming that the preferred path falls along the sink tree) and is so indicated by a circle around the letter. On the second hop, eight packets are generated, two by each of the routers that received a packet on the first hop. As it turns out, all eight of these arrive at previously unvisited routers, and five of these arrive along the preferred line. Of the six packets generated on the third hop, only three arrive on the preferred path (at *C*, *E*, and *K*); the others are duplicates. After five hops and 24 packets, the broadcasting terminates, compared with four hops and 14 packets had the sink tree been followed exactly.

The principal advantage of reverse path forwarding is that it is efficient while bei ng easy to implement. It sends the broadcast packet over each link only once in each direction, just as in flooding, yet it requires only that routers know how to reach all destinations, without needing to remember sequence numbers (or use other mechanisms to stop the flood) or list all destinations in the packet. A **spanning tree** is a subset of the network that includes all the routers but contains no loops. Sink trees are spanning trees. If each router knows which of its lines belong to the spanning tree, it can copy an incoming broadcast packet onto all the spanning tree lines except the one it arrived on. This method makes excellent use of bandwidth, generating the absolute minimum number of packets necessary to do the job.

7. Multicast Routing.

Some applications, such as a multiplayer game or live video of a sports event streamed to many viewing locations, send packets to multiple receivers. Unless the group is very small, sending a distinct packet to each receiver is expensive. On the other hand, broadcasting a packet is wasteful if the group consists of, say, 1000 machines on a million-node network, so that most receivers are not interested in the message (or worse yet, they are definitely interested but are not supposed to see it). Thus, we need a way to send messages to well-defined groups that are numerically large in size but small compared to the network as a whole.

Sending a message to such a group is called **multicasting**, and the routing algorithm used is called **multicast routing**. All multicasting schemes require some way to create and destroy groups and to identify which routers are members of a group. How these tasks are accomplished is not of concern to the routing algorithm. For now, we will assume that each group is identified by a multicast address and that routers know the groups to which they belong. If the group is dense, broadcast is a good start because it efficiently gets the packet to all parts of the network. But broadcast will reach some routers that are not members of the group, which is wasteful. The solution explored by Deering and Cheriton (1990) is to prune the broadcast spanning tree by removing links that do not lead to members. The result is an efficient multicast spanning tree.

As an example, consider the two groups, 1 and 2, in the network shown in Fig. 5-16(a). Some routers are attached to hosts that belong to one or both of these groups, as indicated in the figure. A spanning tree for the leftmost router is shown in Fig. 5-16(b). This tree can be used for broadcast but is overkill for multicast, as can be seen from the two pruned versions that are shown next. In Fig. 5-16(c), all the links that do not lead to hosts that are members of group 1 have been removed. The result is the multicast spanning tree for the leftmost router to send to group 1. Packets are forwarded only along this spanning tree, which is more efficient than the broadcast tree because there are 7 links instead of 10. Fig. 5-16(d) shows the multicast spanning tree after pruning for group 2. It is efficient too, with only five links this time. It also shows that different multicast groups have different spanning trees.

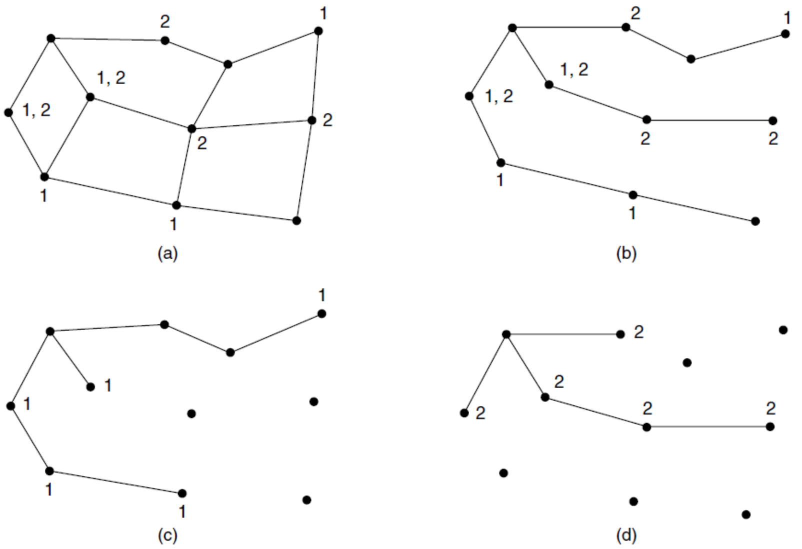
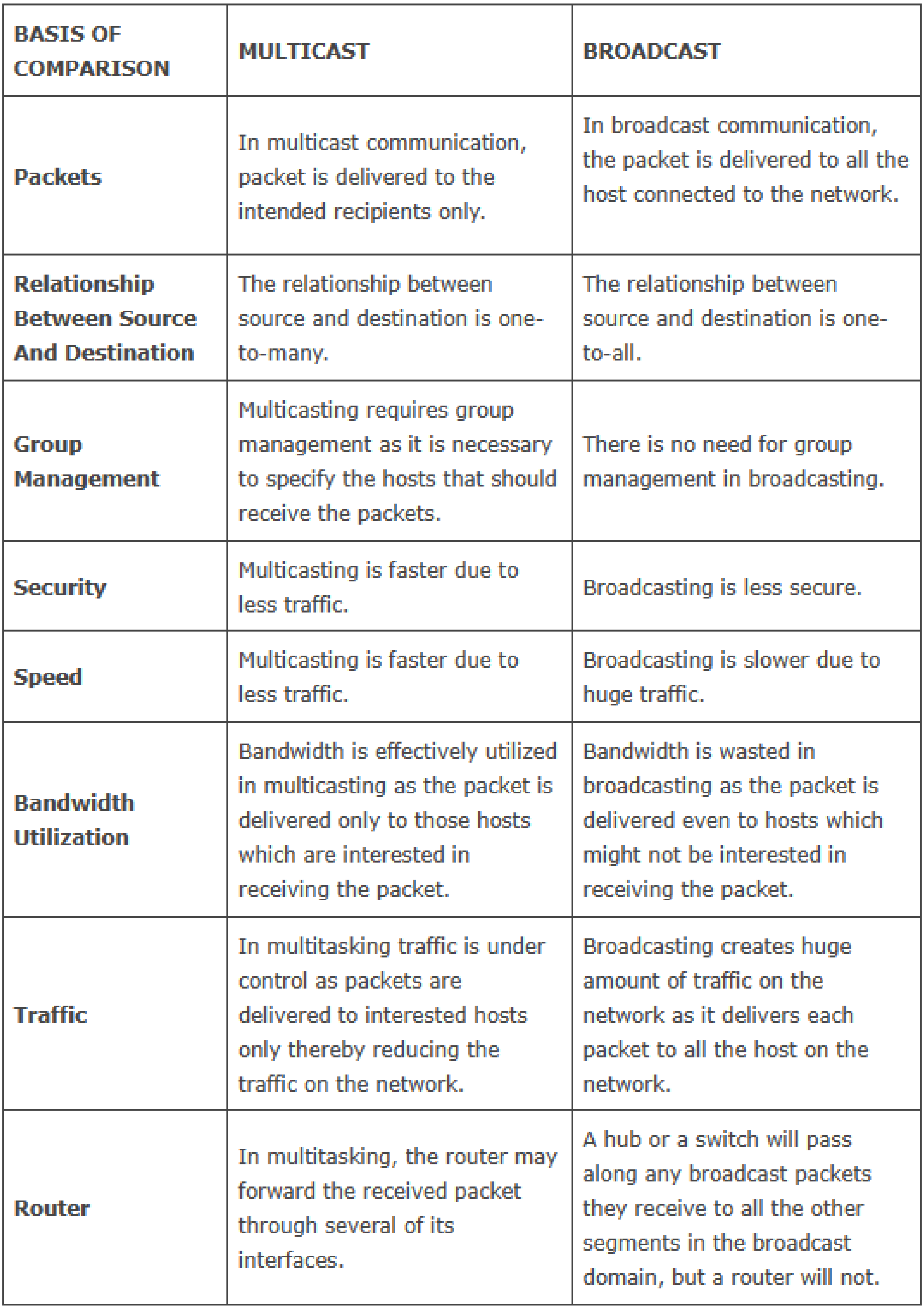


Figure 5-16. (a) A network. (b) A spanning tree for the leftmost router. (c) A multicast tree for group 1. (d) A multicast tree for group 2.

With distance vector routing, a different pruning strategy can be followed. The basic algorithm is reverse path forwarding. However, whenever a router with no hosts interested in a particular group and no connections to other routers receives a multicast message for that group, it responds with a PRUNE message, telling the neighbour that sent the message not to send it any more multicasts from the sender for that group. When a router with no group members among its own hosts has received such messages on all the lines to which it sends the multicast, it, too, can respond with a PRUNE message. In this way, the spanning tree is recursively pruned. **DVMRP** (**Distance Vector Multicast Routing Protocol**) is an example of a multicast routing protocol that works this way. Pruning results in efficient spanning trees that use only the links that are actually needed to reach members of the group. One potential disadvantage is that it is lots of work for routers, especially for large networks.



BASIS OF COMPARISON	MULTICAST	BROADCAST
Packets	In multicast communication, packet is delivered to the intended recipients only.	In broadcast communication, the packet is delivered to all the host connected to the network.
Relationship Between Source And Destination	The relationship between source and destination is one-to-many.	The relationship between source and destination is one-to-all.
Group Management	Multicasting requires group management as it is necessary to specify the hosts that should receive the packets.	There is no need for group management in broadcasting.
Security	Multicasting is faster due to less traffic.	Broadcasting is less secure.
Speed	Multicasting is faster due to less traffic.	Broadcasting is slower due to huge traffic.
Bandwidth Utilization	Bandwidth is effectively utilized in multicasting as the packet is delivered only to those hosts which are interested in receiving the packet.	Bandwidth is wasted in broadcasting as the packet is delivered even to hosts which might not be interested in receiving the packet.
Traffic	In multitasking traffic is under control as packets are delivered to interested hosts only thereby reducing the traffic on the network.	Broadcasting creates huge amount of traffic on the network as it delivers each packet to all the host on the network.
Router	In multitasking, the router may forward the received packet through several of its interfaces.	A hub or a switch will pass along any broadcast packets they receive to all the other segments in the broadcast domain, but a router will not.

Understanding Unicast Multicast Broadcast - YouTube