advantage is somewhat illusory since connection setup packets have to be routed too, and they use destination addresses, the same as datagrams do.

Virtual circuits have some advantages in guaranteeing quality of service and avoiding congestion within the network because resources (e.g., buffers, bandwidth, and CPU cycles) can be reserved in advance, when the connection is established. Once the packets start arriving, the necessary bandwidth and router capacity will be there. With a datagram network, congestion avoidance is more difficult.

For transaction processing systems (e.g., stores calling up to verify credit card purchases), the overhead required to set up and clear a virtual circuit may easily dwarf the use of the circuit. If the majority of the traffic is expected to be of this kind, the use of virtual circuits inside the network makes little sense. On the other hand, for long-running uses such as VPN traffic between two corporate offices, permanent virtual circuits (that are set up manually and last for months or years) may be useful.

Virtual circuits also have a vulnerability problem. If a router crashes and loses its memory, even if it comes back up a second later, all the virtual circuits passing through it will have to be aborted. In contrast, if a datagram router goes down, only those users whose packets were queued in the router at the time need suffer (and probably not even then since the sender is likely to retransmit them shortly). The loss of a communication line is fatal to virtual circuits using it, but can easily be compensated for if datagrams are used. Datagrams also allow the routers to balance the traffic throughout the network, since routes can be changed partway through a long sequence of packet transmissions.

5.2 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 (e.g., while logged in over a VPN).

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. Correctness and simplicity hardly require comment, but the need for robustness may be less obvious at first. Once a major network comes on the air, it may be expected to run continuously for years without system-wide failures. During that period there will be hardware and software failures of all kinds. Hosts, routers, and lines will fail repeatedly, and the topology will change many times. 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. Imagine the havoc if the network needed to be rebooted every time some router crashed!

Stability is also an important goal for the routing algorithm. There exist routing algorithms that never converge to a fixed set of paths, no matter how long they run. A stable algorithm reaches equilibrium and stays there. It should converge quickly too, since communication may be disrupted until the routing algorithm has reached equilibrium.

Fairness and efficiency may sound obvious—surely no reasonable person would oppose them—but as it turns out, they are often contradictory goals. As a simple example of this conflict, look at Fig. 5-5. Suppose that there is enough traffic between A and A', between B and B', and between C and C' to saturate the horizontal links. To maximize the total flow, the X to X' traffic should be shut off altogether. Unfortunately, X and X' may not see it that way. Evidently, some compromise between global efficiency and fairness to individual connections is needed.

Before we can even attempt to find trade-offs between fairness and efficiency, we must decide what it is we seek to optimize. Minimizing the mean packet delay is an obvious candidate to send traffic through the network effectively, but so is maximizing total network throughput. Furthermore, these two goals are also in conflict, since operating any queueing system near capacity implies a long queueing delay. As a compromise, many networks attempt to minimize the distance a packet must travel, or simply reduce the number of hops a packet must make. Either choice tends to improve the delay and also reduce the amount of bandwidth consumed per packet, which tends to improve the overall network throughput as well.

Routing algorithms can be grouped into two major classes: nonadaptive and adaptive. **Nonadaptive algorithms** do not base their routing decisions on any

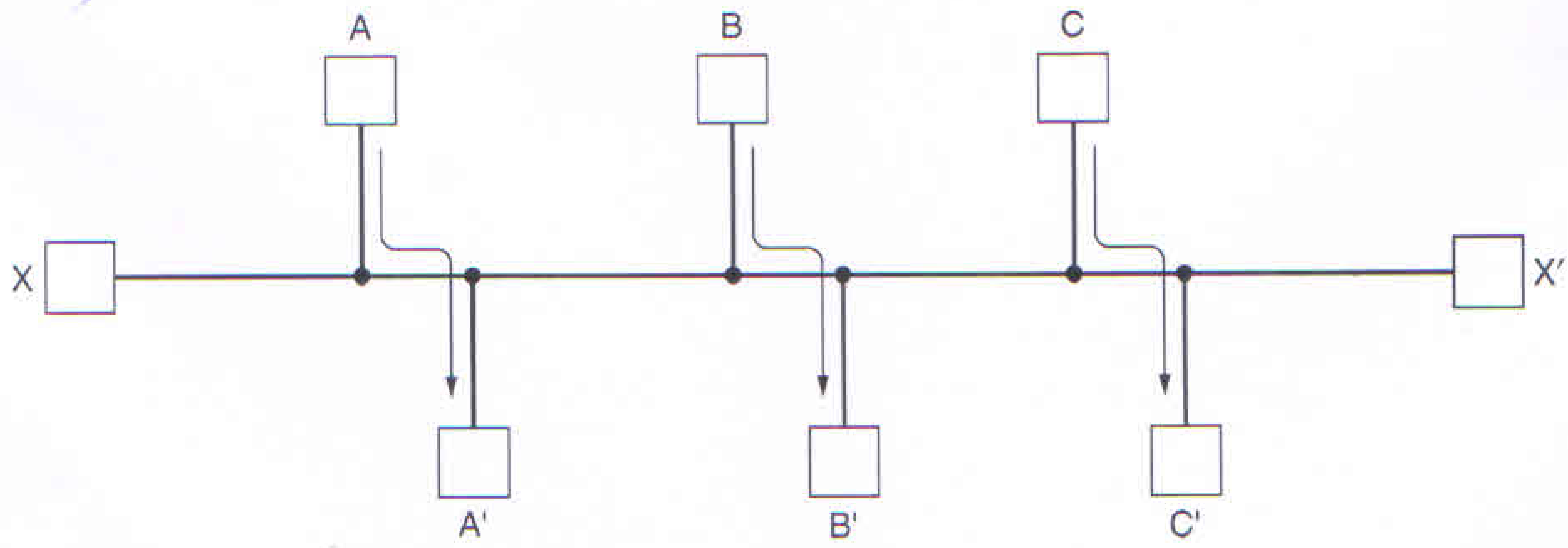


Figure 5-5. Network with a conflict between fairness and efficiency.

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. For example, router *F* in Fig. 5-3 should send packets headed into the network to router *E* regardless of the ultimate destination.

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).

In the following sections, we will discuss a variety of routing algorithms. The algorithms cover delivery models besides sending a packet from a source to a destination. Sometimes the goal is to send the packet to multiple, all, or one of a set of destinations. All of the routing algorithms we describe here make decisions based on the topology; we defer the possibility of decisions based on the traffic levels to Sec 5.3.

5.2.1 The Optimality Principle

Before we get into specific algorithms, it may be helpful to note that one can make a general statement about optimal routes without regard to network topology or traffic. This statement is known as the **optimality principle** (Bellman, 1957). It states that if router *J* is on the optimal path from router *I* to router *K*, then the optimal path from J to K also falls along the same route. To see this, call the part of the route from I to J r_1 and the rest of the route r_2. If a route better than r_2 existed from J to K, it could be concatenated with r_1 to improve the route from I to K, contradicting our statement that $r_1 r_2$ is optimal.

As a direct consequence of the optimality principle, we can see that the set of optimal routes from all sources to a given destination form a tree rooted at the destination. Such a tree is called a **sink tree** and is illustrated in Fig. 5-6(b), where the distance metric is the number of hops. The goal of all routing algorithms is to discover and use the sink trees for all routers.

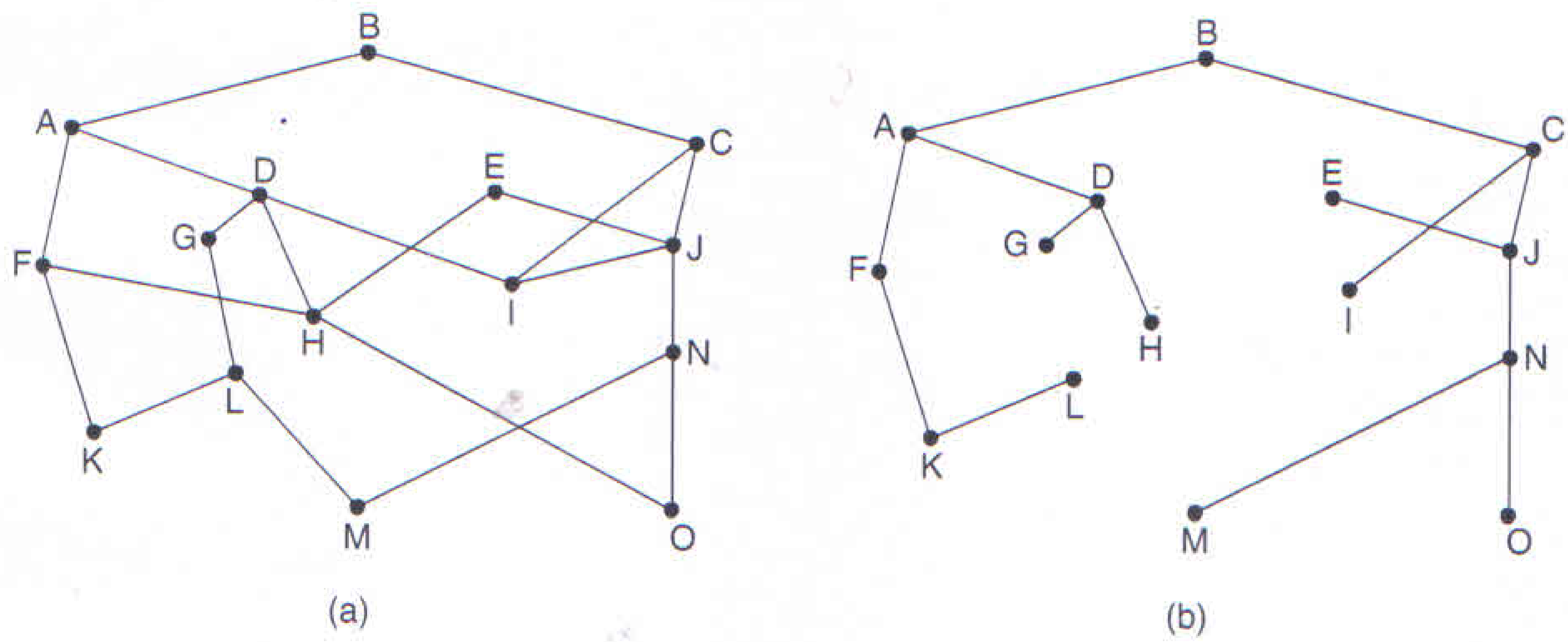


Figure 5-6. (a) A network. (b) A sink tree for router B.

Note that a sink tree is not necessarily unique; other trees with the same path lengths may exist. If we allow all of the possible paths to be chosen, the tree becomes a more general structure called a **DAG** (**Directed Acyclic Graph**). DAGs have no loops. We will use sink trees as a convenient shorthand for both cases. Both cases also depend on the technical assumption that the paths do not interfere with each other so, for example, a traffic jam on one path will not cause another path to divert.

Since a sink tree is indeed a tree, it does not contain any loops, so each packet will be delivered within a finite and bounded number of hops. In practice, life is not quite this easy. Links and routers can go down and come back up during operation, so different routers may have different ideas about the current topology. Also, we have quietly finessed the issue of whether each router has to individually acquire the information on which to base its sink tree computation or whether this information is collected by some other means. We will come back to these issues shortly. Nevertheless, the optimality principle and the sink tree provide a benchmark against which other routing algorithms can be measured.

5.2.2 Shortest Path Algorithm

Let us begin our study of routing algorithms with 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).

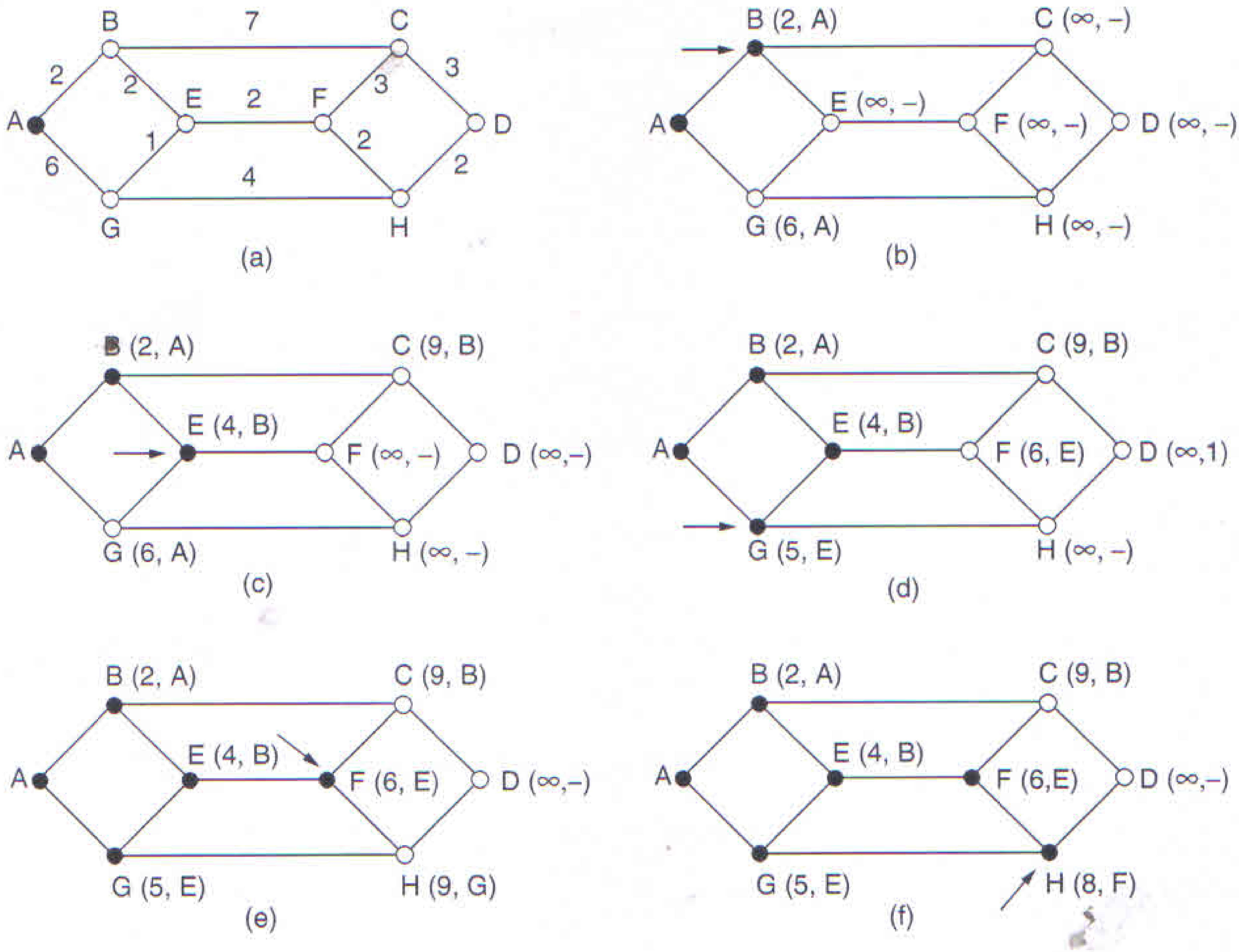


Figure 5-7. The first six steps used in computing the shortest path from *A* to *D*. The arrows indicate the working node.

However, many other metrics besides hops and physical distance are also possible. For example, each edge could be labeled with the mean delay of a standard test packet, as measured by hourly runs. With this graph labeling, the shortest path is the fastest path rather than the path with the fewest edges or kilometers.

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 labeled (in parentheses) with its distance from the source node along the best known path. The distances must be non-negative, as they will be if they are based on real quantities like bandwidth and delay. Initially, no paths are known, so all nodes are labeled 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 labeling 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), relabeling each one with the distance to *A*. Whenever a node is relabeled, 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 labeled 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 relabeled.

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 labeled 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.

To see why the algorithm works, look at Fig. 5-7(c). At this point we have just made *E* permanent. Suppose that there were a shorter path than *ABE*, say

AXYZE (for some *X* and *Y*). There are two possibilities: either node *Z* has already been made permanent, or it has not been. If it has, then *E* has already been probed (on the round following the one when *Z* was made permanent), so the *AXYZE* path has not escaped our attention and thus cannot be a shorter path.

Now consider the case where *Z* is still tentatively labeled. If the label at *Z* is greater than or equal to that at *E*, then *AXYZE* cannot be a shorter path than *ABE*. If the label is less than that of *E*, then *Z* and not *E* will become permanent first, allowing *E* to be probed from *Z*.

This algorithm is given in Fig. 5-8. The global variables *n* and *dist* describe the graph and are initialized before *shortest_path* is called. The only difference between the program and the algorithm described above is that in Fig. 5-8, we compute the shortest path starting at the terminal node, *t*, rather than at the source node, *s*.

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 labeled 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.

5.2.3 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 every 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.

```
#define MAX_NODES 1024                      /* maximum number of nodes */
#define INFINITY 1000000000                 /* a number larger than every maximum path */
int n, dist[MAX_NODES][MAX_NODES];          /* dist[i][j] is the distance from i to j */

void shortest_path(int s, int t, int path[])
{ struct state {                            /* the path being worked on */
      int predecessor;                      /* previous node */
      int length;                           /* length from source to this node */
      enum {permanent, tentative} label;    /* label state */
  } state[MAX_NODES];

  int i, k, min;
  struct state *p;

  for (p = &state[0]; p < &state[n]; p++) { /* initialize state */
      p->predecessor = -1;
      p->length = INFINITY;
      p->label = tentative;
  }
  state[t].length = 0; state[t].label = permanent;
  k = t;                                    /* k is the initial working node */
  do {                                      /* Is there a better path from k? */
      for (i = 0; i < n; i++)               /* this graph has n nodes */
           if (dist[k][i] != 0 && state[i].label == tentative) {
                 if (state[k].length + dist[k][i] < state[i].length) {
                      state[i].predecessor = k;
                      state[i].length = state[k].length + dist[k][i];
                 }
           }

      /* Find the tentatively labeled node with the smallest label. */
      k = 0; min = INFINITY;
      for (i = 0; i < n; i++)
           if (state[i].label == tentative && state[i].length < min) {
                 min = state[i].length;
                 k = i;
           }
      state[k].label = permanent;
  } while (k != s);

  /* Copy the path into the output array. */
  i = 0; k = s;
  do {path[i++] = k; k = state[k].predecessor; } while (k >= 0);
}
```

Figure 5-8. Dijkstra's algorithm to compute the shortest path through a graph.

To prevent the list from growing without bound, each list should be augmented by a counter, k, meaning that all sequence numbers through k have been seen. When a packet comes in, it is easy to check if the packet has already been flooded (by comparing its sequence number to k; if so, it is discarded. Furthermore, the full list below k is not needed, since k effectively summarizes it.

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 neighbors. 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. Consequently, no other algorithm can produce a shorter delay (if we ignore the overhead generated by the flooding process itself).

5.2.4 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. In this section, we will look at the former algorithm. In the following section, we will study the latter algorithm.

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 neighbors. Eventually, every router knows the best link to reach each destination.

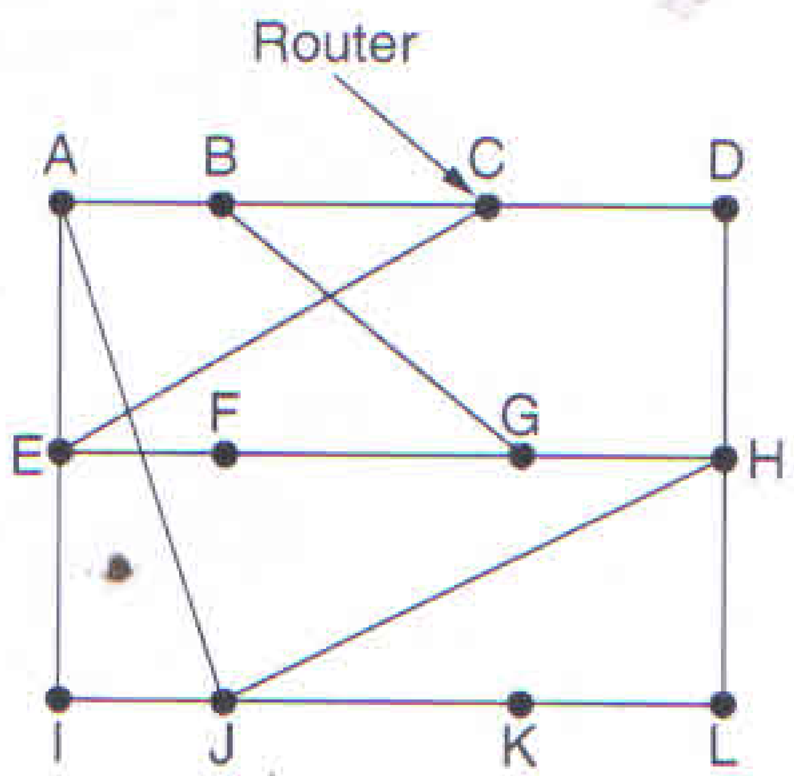
The distance vector routing algorithm is sometimes called by other names, most commonly the distributed **Bellman-Ford** routing algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962). It was the original ARPANET routing algorithm and was also used in the Internet under the name RIP.

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, as we discussed for computing shortest paths.

The router is assumed to know the "distance" to each of its neighbors. 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 neighbors. Once every T msec, each router sends to each neighbor a list of its estimated delays to each destination. It also receives a similar list from each neighbor. Imagine that one of these tables has just come in from neighbor X, with X_i 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 $X_i + m$ msec. By performing this calculation for each neighbor, 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 neighbors 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 neighbors, A, I, H, and K, as 8, 10, 12, and 6 msec, respectively.



(a)

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

(b)

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 settling of routes to best paths across the network is called **convergence**. 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, neighbor 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 neighbor 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 neighbors 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).

From this figure, it should be clear why bad news travels slowly: no router ever has a value more than one higher than the minimum of all its neighbors. Gradually, all routers work their way up to infinity, but the number of exchanges required depends on the numerical value used for infinity. For this reason, it is wise to set infinity to the longest path plus 1.

Not entirely surprisingly, this problem is known as the **count-to-infinity** problem. There have been many attempts to solve it, for example, preventing routers from advertising their best paths back to the neighbors from which they heard them with the split horizon with poisoned reverse rule discussed in RFC 1058. However, none of these heuristics work well in practice despite the colorful names. The core of the problem is that when X tells Y that it has a path somewhere, Y has no way of knowing whether it itself is on the path.

5.2.5 Link State Routing

Distance vector routing was used in the ARPANET until 1979, when it was replaced by link state routing. 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.

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 neighbors and learn their network addresses.
2. Set the distance or cost metric to each of its neighbors.
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. Below we will consider each of these five steps in more detail.

Learning about the Neighbors

When a router is booted, its first task is to learn who its neighbors 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.

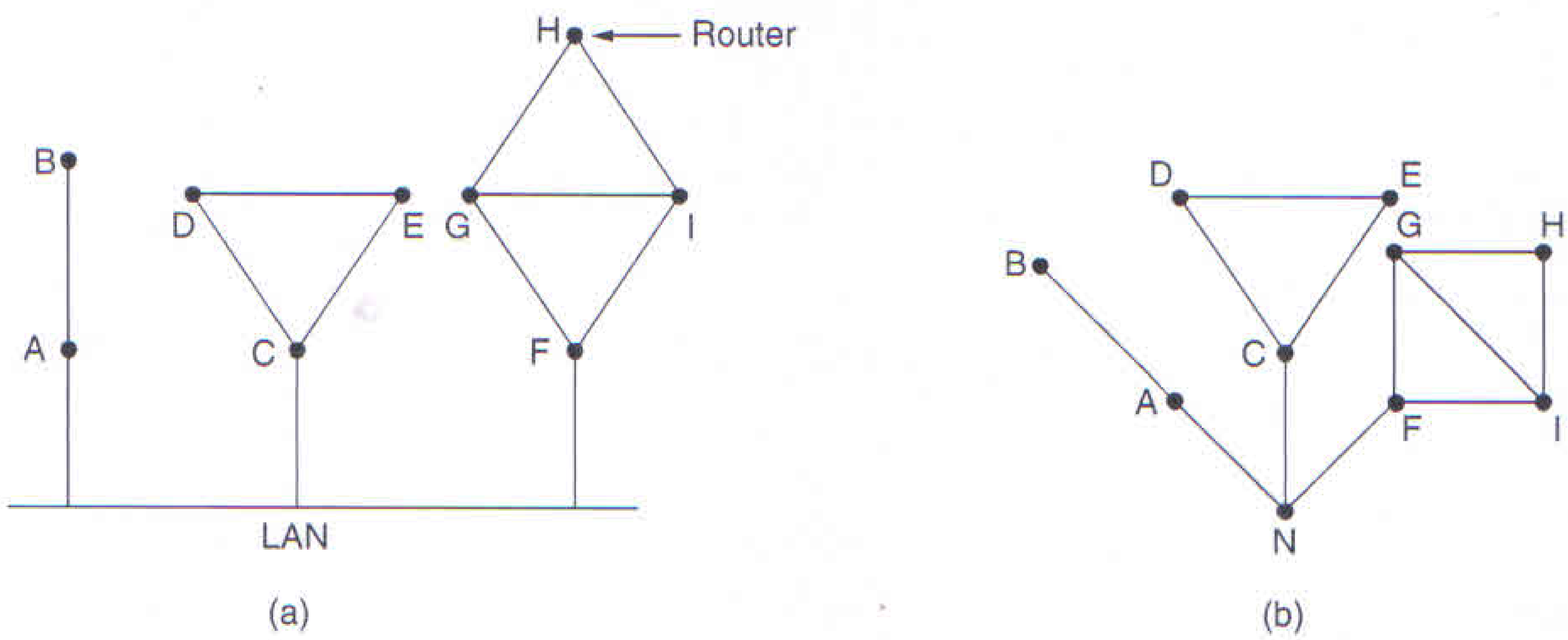


Figure 5-11. (a) Nine routers and a broadcast LAN. (b) A graph model of (a).

The broadcast LAN provides connectivity between each pair of attached routers. However, modeling 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.

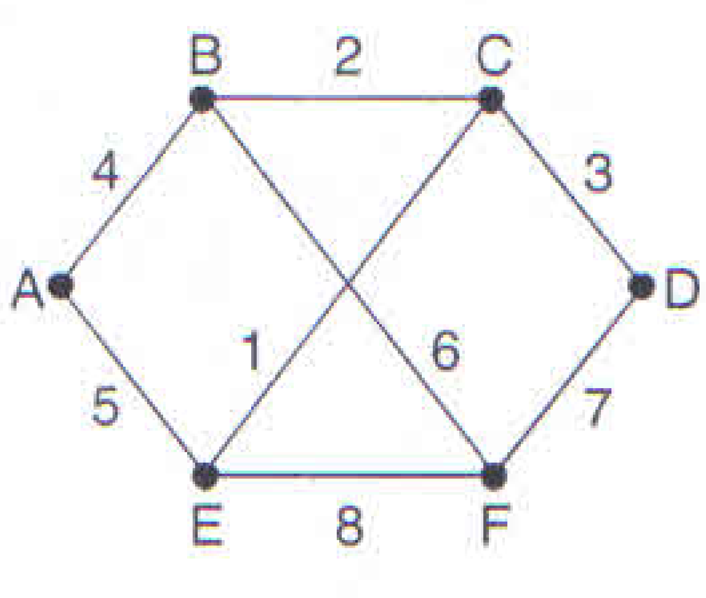
Setting Link Costs

The link state routing algorithm requires each link to have a distance or cost metric for finding shortest paths. The cost to reach neighbors 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.

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 (to be described later) and a list of neighbors. The cost to each neighbor 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).



(a)

Link State Packets

A		B		C		D		E		F	
Seq.		Seq.		Seq.		Seq.		Seq.		Seq.	
Age		Age		Age		Age		Age		Age	
B	4	A	4	B	2	C	3	A	5	B	6
E	5	C	2	D	3	F	7	C	1	D	7
		F	6	E	1			F	8	E	8

(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 neighbor going down or coming back up again or changing its properties appreciably.

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.

First, we will describe the basic distribution algorithm. After that we will give some refinements. 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. 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.

			Send flags			ACK flags			
Source	Seq.	Age	A	C	F	A	C	F	Data
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).

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.

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 neighbors, 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. We will go over OSPF in Sec. 5.6.6.

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.

5.2.6 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