UNIT III

Network layer – design issues – Routing algorithms - The Optimality Principle - Shortest Path Algorithm – Flooding - Distance Vector Routing - Link State Routing - Hierarchical Routing - Broadcast Routing - Multicast Routing Congestion Control – Approaches - Traffic-Aware Routing - Admission Control - Traffic Throttling - Load Shedding – Internetworking - Tunneling - Internetwork Routing - Packet Fragmentation - IP v4 - IP Addresses – IPv6 - Internet Control Protocols – OSPF - BGP

1. Explain the design issues of the Network layer. /

   Discuss the services provided to the transport layer by the Network layer. /

   How Connection Oriented and Connection Less Services are implemented? Explain.

   / Compare the virtual circuits and datagram within the subnet.

2. Discuss about different routing algorithms in detail. /

   Discuss shortest path routing. /

   What is flooding? Discuss. /

   Differentiate and explain adaptive and nonadaptive routing algorithms. /

   Describe hierarchical Broadcast and Multicasting routing.

3. Discuss about different congestion control algorithms in detail /

   Discuss the reasons for occurrence of congestion. Suggest some algorithms to control congestion.

4. What is meant by fragmentation? Discuss & Write short notes on:

   (a) Tunnelling

   (b) Internetwork routing

   (c) Fragmentation.

5. Explain about the network layer in the internet in detail /

   What is IP? Explain IPV 4 header.

1. Explain the design issues of the Network layer. /

Discuss the services provided to the transport layer by the Network layer. /

How Connection Oriented and Connection Less Services are implemented? Explain. /

Compare the virtual circuits and datagram within the subnet.
(Nov'09, Nov'07, Dec,06)

NETWORK LAYER DESIGN ISSUES

In the following sections we will provide an introduction to some of the issues that the designers of the network layer must grapple with. These issues include the service provided to the transport layer and the internal design of the subnet.

STORE-AND-FORWARD PACKET SWITCHING

- The network layer protocols operation can be seen in Fig. 5-1.
- The major components of the system are the carrier's equipment (routers connected by transmission lines), shown inside the shaded oval.
- The customers' equipment, shown outside the oval. Host *H1* is directly connected to one of the carrier's routers, *A*, by a leased line. In contrast, *H2* is on a LAN with a router, *F*, owned and operated by the customer. This router also has a leased line to the carrier's equipment.
- We have shown *F* as being outside the oval because it does not belong to the carrier, but in terms of construction, software, and protocols, it is probably no different from the carrier's routers. Whether it belongs to the subnet is arguable, but for the purposes of this chapter, routers on customer premises are considered part of the subnet.

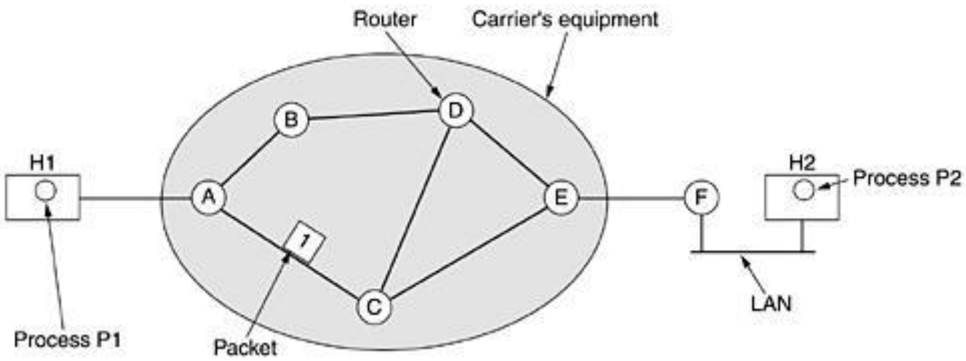


Figure 5-1. The environment of the network layer protocols.

- 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 carrier.

- The packet is stored there until it has fully arrived so the checksum can be verified. 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**.

SERVICES PROVIDED TO THE TRANSPORT LAYER

The network layer provides services to the transport layer at the network layer/transport layer interface. The network layer services have been 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.

- There is a discussion centers on whether the network layer should provide connection-oriented service or connectionless service.
- In their view (based on 30 years of actual experience with a real, working computer network), the subnet is inherently unreliable, no matter how it is designed. Therefore, the hosts should accept the fact that the network is unreliable and do error control (i.e., error detection and correction) and flow control themselves.
- This viewpoint leads quickly to the conclusion that 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, and there is usually little to be gained by doing it twice.
- Furthermore, each packet must carry the full destination address, because each packet sent is carried independently of its predecessors, if any.
- The other camp (represented by the telephone companies) argues that the subnet should provide a reliable, connection-oriented service.
- These two camps are best exemplified by the Internet and ATM. The Internet offers connectionless network-layer service; ATM networks offer connection-oriented network-layer service. However, it is interesting to note that as quality-of-service guarantees are becoming more and more important, the Internet is evolving. In particular, it is starting to acquire properties normally associated with connection-oriented service, as we will see later. Actually, we got an inkling of this evolution during our study of VLANs in Chap. 4.

IMPLEMENTATION OF CONNECTIONLESS SERVICE

- Two different organizations are possible, depending on the type of service offered.
- If connectionless service is offered, packets are injected into the subnet 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 subnet is called a **datagram subnet**.

- If connection-oriented service is used, a path from the source router 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 subnet is called a **virtual-circuit subnet**.
- Let us now see how a datagram subnet 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, probably just another procedure within the operating system.

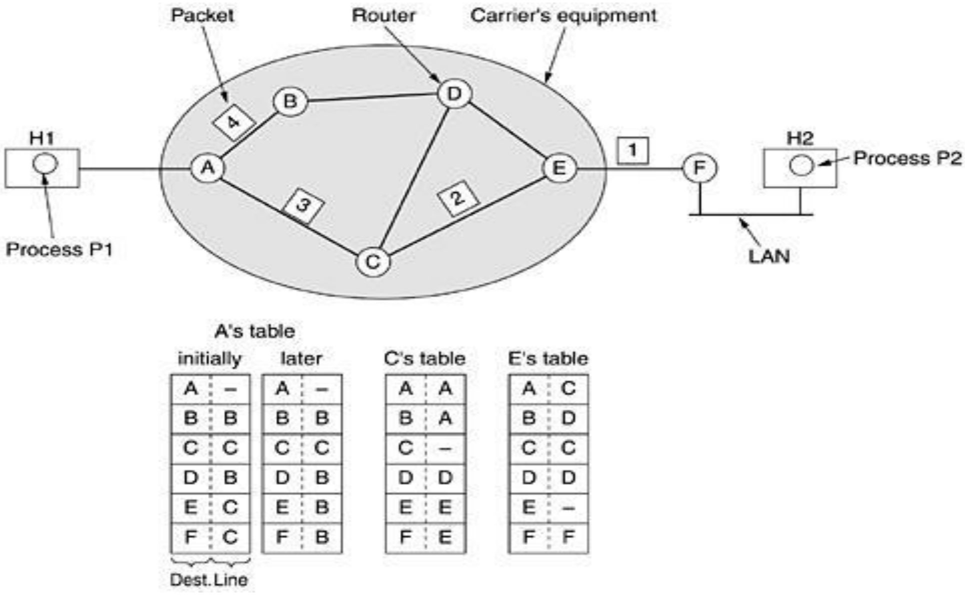


Figure 5-2. Routing within a datagram subnet

- Let us assume that 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 sends each of them in turn to router *A* using some point-to-point protocol,
- For example, PPP. At this point the carrier takes over. Every router has an internal table telling it where to send packets for each possible destination. 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 *C*—so every incoming packet must be sent to one of these routers, even if the ultimate destination is some other router. *A*'s initial routing table is shown in the figure under the label "initially". As they arrived at *A*, packets 1, 2, and 3 were stored briefly (to verify their checksums). Then each was forwarded to *C* according to *A*'s table. Packet 1 was then forwarded to *E* and then to *F*. When it got to *F*, it was encapsulated in a data link layer frame and sent to *H2* over the LAN. Packets 2 and 3 follow the same route.

- However, something different happened to packet 4. When it got to *A* it was 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. Perhaps it 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**.

IMPLEMENTATION OF CONNECTION-ORIENTED SERVICE

- For connection-oriented service, we need a virtual-circuit subnet.
- 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 of **Fig. 5-3**. Here, host *H1* has established connection 1 with host *H2*. It 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.

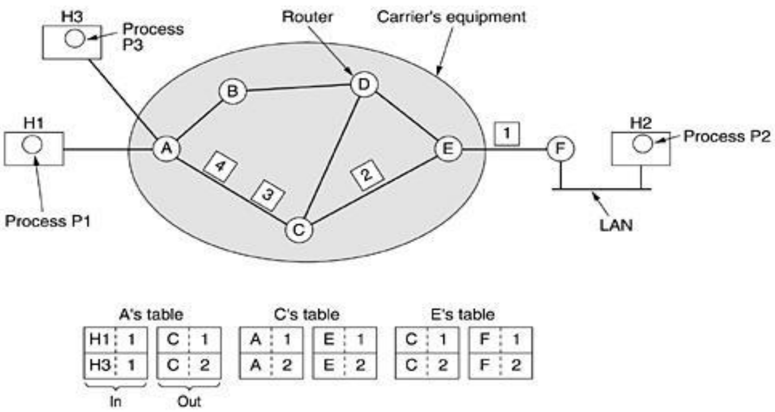


Figure 5-3. Routing within a virtual-circuit subnet.

- 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 subnet 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* from 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 is called label switching.

COMPARISON OF VIRTUAL-CIRCUIT AND DATAGRAM SUBNETS

The major issues are listed in Fig. 5-4, although purists could probably find a counter example for everything in the figure.

Issue	Datagram subnet	Virtual-circuit subnet
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

Figure 5-4. Comparison of datagram and virtual-circuit subnets.

2. Discuss about different routing algorithms in detail. (or)

Discuss shortest path routing. (Or)

What is flooding? Discuss. (Or)

Differentiate and explain adaptive and nonadaptive routing algorithms. (Or)

Describe hierarchical Broadcast and Multicasting routing.

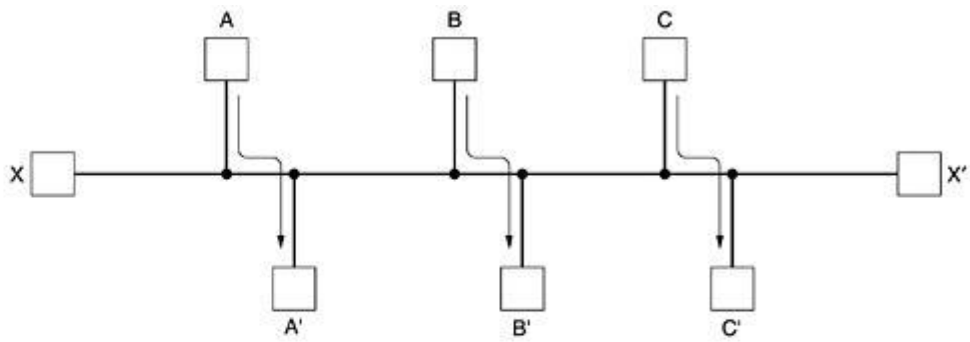
(Nov'11, May'10, Dec'08, Nov'07, Dec'05, Dec'04)

ROUTING ALGORITHMS

The **routing algorithm** is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on.

PROPERTIES OF ROUTING ALGORITHM:
Correctness, simplicity, robustness, stability, fairness, and optimality

FAIRNESS AND OPTIMALITY.



Fairness and optimality may sound obvious, but as it turns out, they are often contradictory goals. 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.

CATEGORY OF ALGORITHM

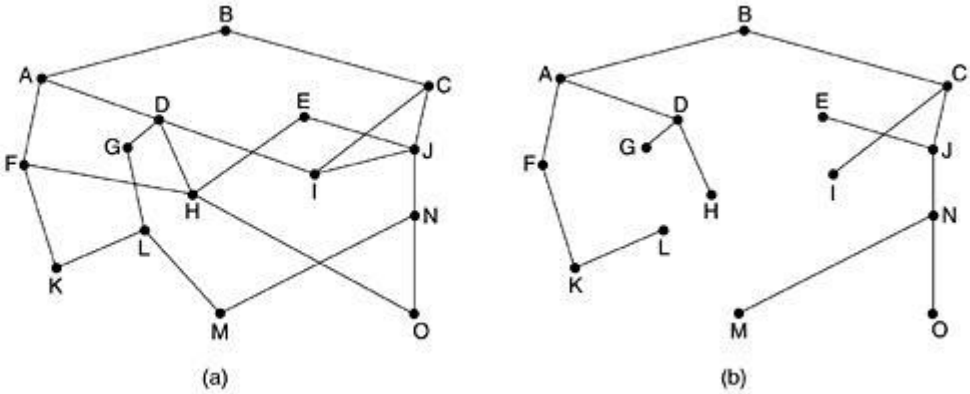
- Routing algorithms can be grouped into two major classes: **nonadaptive and adaptive.**
- **Nonadaptive algorithms** do not base their routing decisions on measurements or estimates of the current traffic and topology. Instead, the choice of the route to use to get from I to J is computed in advance, off-line, and downloaded to the routers when the network is booted.
- This procedure is sometimes called **Static routing**.

- **Adaptive algorithms**, in contrast, change their routing decisions to reflect changes in the topology, and usually the traffic as well
- This procedure is sometimes called **dynamic routing**

THE OPTIMALITY PRINCIPLE

- **I**f 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.

- 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.



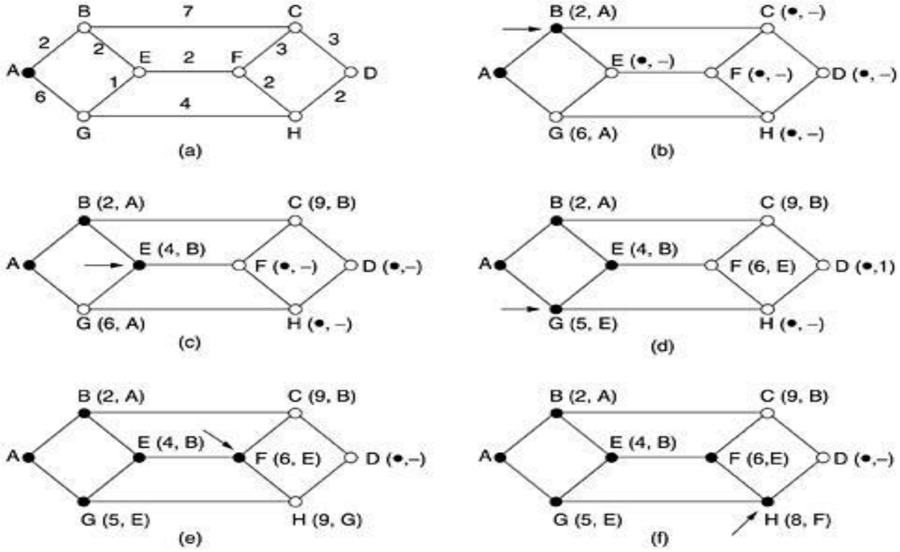
***(a) A subnet. (b) A sink tree for router* B.**

- 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** where the distance metric is the number of hops. Note that a sink tree is not necessarily unique; other trees with the same path lengths may exist.
- The goal of all routing algorithms is to discover and use the sink trees for all routers.

SHORTEST PATH ROUTING

- A technique to study routing algorithms: The idea is to build a graph of the subnet, with each node of the graph representing a router and each arc of the graph representing a communication line (often called a link).
- To choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph.
- One way of measuring path length is the number of hops. Another metric is the geographic distance in kilometers. Many other metrics are also possible. For example, each arc could be labeled with the mean queuing and transmission delay for some standard test packet as determined by hourly test runs.
- In the general case, the labels on the arcs could be computed as a function of the distance, bandwidth, average traffic, communication cost, mean queue length, 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.



The first five steps used in computing the shortest path from* A *to* D*. The arrows indicate the working node.

- 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), relabeling 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.
- 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 five steps of the algorithm.

- To see why the algorithm works, look at Fig. 5-7(c). At that point we have just made *E* permanent. Suppose that there were a shorter path than *ABE*, say *AXYZE*. 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 labelled. Either the label at *Z* is greater than or equal to that at *E*, in which case *AXYZE* cannot be a shorter path than *ABE*, or it is less than that of *E*, in which case *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 path from *t* to *s* in an undirected graph is the same as the shortest path from *s* to *t*, it does not matter at which end we begin (unless there are several shortest paths, in which case reversing the search might discover a different one). 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. By reversing the search, the two effects cancel, and the answer is produced in the correct order.

```
#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);
}
```

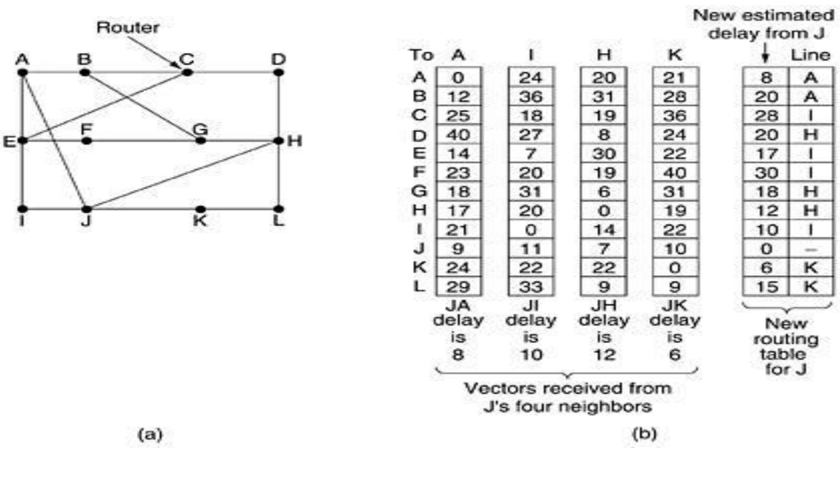
Figure 5-8. Dijkstra's algorithm to compute the shortest path through a graph.

FLOODING

- Another static algorithm 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, which 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 subnet.

DISTANCE VECTOR ROUTING

- **Distance vector routing** algorithms operate by having each router maintain a table (i.e, a vector) giving the best known distance to each destination and which line to use to get there.
- These tables are updated by exchanging information with the neighbors.
- The distance vector routing algorithm is sometimes called by other names, most commonly the distributed **Bellman-Ford** routing algorithm and the **Ford-Fulkerson** 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.



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		

***(a) A subnet. (b) Input from* A, I, H, K, *and the new routing table for* J.**

- Part (a) shows a subnet. 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.

Each node constructs a one-dimensional array containing the "distances"(costs) to all other nodes and distributes that vector to its immediate neighbors.

1. The starting assumption for distance-vector routing is that each node knows the cost of the link to each of its directly connected neighbors.
2. A link that is down is assigned an infinite cost.

Example.

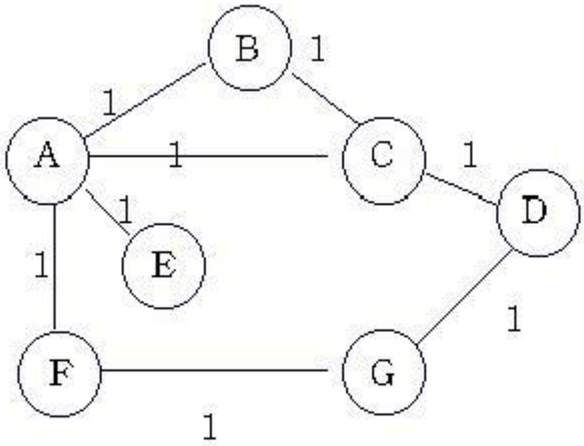


Table 1. Initial distances stored at each node(global view).

Information Stored at Node	**Distance to Reach Node** A	**B**	**C**	**D**	**E**	**F**	**G**
A	0	1	1	∞	1	1	∞
B	1	0	1	∞	∞	∞	∞
C	1	1	0	1	∞	∞	∞
D	∞	∞	1	0	∞	∞	1
E	1	∞	∞	∞	0	∞	∞
F	1	∞	∞	∞	∞	0	1
G	∞	∞	∞	1	∞	1	0

We can represent each node's knowledge about the distances to all other nodes as a table like the one given in Table 1.

Note that each node only knows the information in one row of the table.

1. Every node sends a message to its directly connected neighbors containing its personal list of distance. (for example, **A** sends its information to its neighbors **B,C,E**, and **F**.)
2. If any of the recipients of the information from **A** find that **A** is advertising a path shorter than the one they currently know about, they update their list to give the new path length and note that they should send packets for that destination through **A**. (node **B** learns from **A** that node **E** can be reached at a cost of 1; **B** also knows it can reach **A** at a cost of 1, so it adds these to get the cost of reaching **E** by means of **A**. **B** records that it can reach **E** at a cost of 2 by going through **A**.)
3. After every node has exchanged a few updates with its directly connected neighbors, all nodes will know the least-cost path to all the other nodes.
4. In addition to updating their list of distances when they receive updates, the nodes need to keep track of which node told them about the path that they used to calculate the cost, so that they can create their forwarding table. (for example, **B** knows that it was **A** who said " I can reach **E** in one hop" and so **B** puts an entry in its table that says " To reach **E**, use the link to **A**.)

Table 2. final distances stored at each node (global view).

Information	**Distance to Reach Node**						
Stored at Node	**A**	**B**	**C**	**D**	**E**	**F**	**G**
A	0	1	1	2	1	1	2
B	1	0	1	2	2	2	3
C	1	1	0	1	2	2	2
D	2	2	1	0	3	2	1
E	1	2	2	3	0	2	3
F	1	2	2	2	2	0	1
G	2	3	2	1	3	1	0

In practice, each node's forwarding table consists of a set of triples of the form:

(Destination, Cost, NextHop).

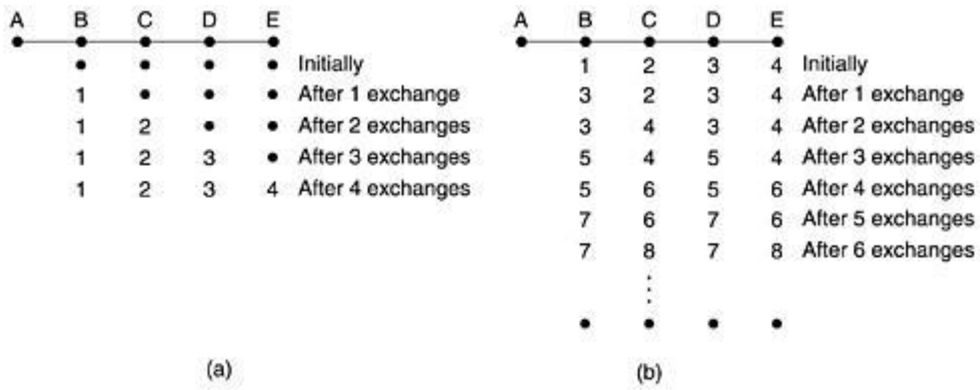
For example, Table 3 shows the complete routing table maintained at node B for the network in figure1.

Table 3. Routing table maintained at node B.

Destination	Cost	NextHop
A	1	A
C	1	C
D	2	C
E	2	A
F	2	A
G	3	A

THE COUNT-TO-INFINITY PROBLEM

The count-to-infinity problem.



- Consider the five-node (linear) subnet 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.
- Now let us consider the situation of Fig. 5-10(b), in which all the lines and routers are initially up. Routers *B*, *C*, *D*, and *E* have distances to *A* of 1, 2, 3, and 4, respectively. Suddenly *A* goes down, or alternatively, the line between *A* and *B* is cut, which is effectively the same thing from *B*'s point of view.

LINK STATE ROUTING

The idea behind link state routing is simple and can be stated as five parts. Each router must do the following:

1. Discover its neighbors and learn their network addresses.

2. Measure the delay or cost to each of its neighbors.

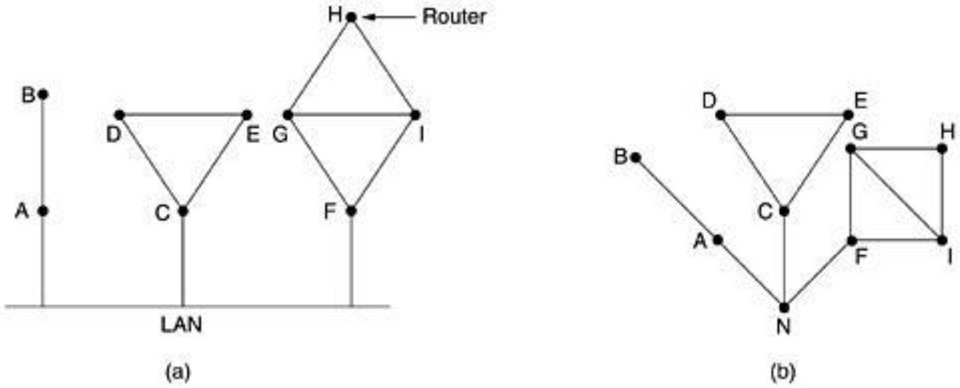
3. Construct a packet telling all it has just learned.

4. Send this packet to all other routers.

5. Compute the shortest path to every other router

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 telling who it is.



(a) Nine routers and a LAN. (b) A graph model of (a).
(b)

Measuring Line Cost

- The link state routing algorithm requires each router to know, or at least have a reasonable estimate of, the delay to each of its neighbors. 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.
- For even better results, the test can be conducted several times, and the average used. Of course, this method implicitly assumes the delays are symmetric, which may not always be the case.

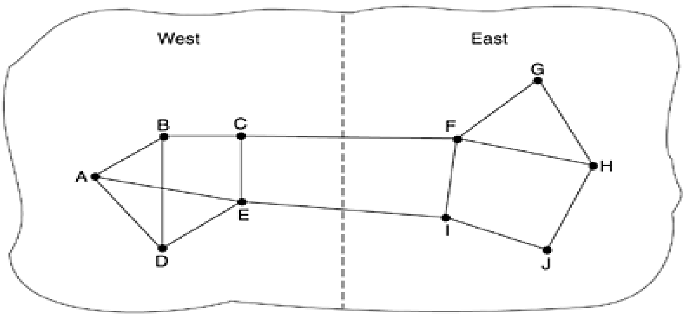
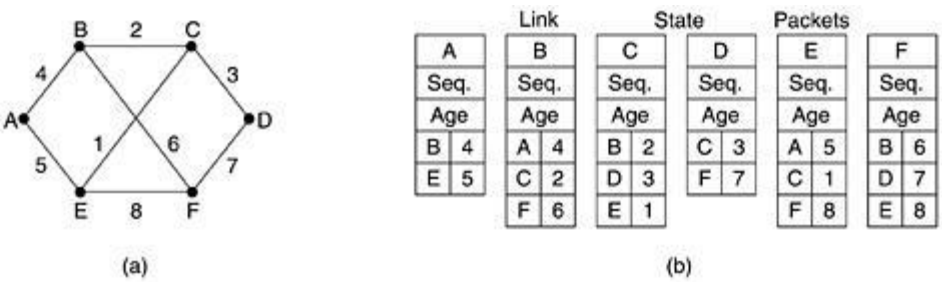


Figure: A subnet in which the East and West parts are connected by two lines.

- Unfortunately, there is also an argument against including the load in the delay calculation. Consider the subnet of Fig. 5-12, which is divided into two parts, East and West, connected by two lines, *CF* and *EI*.

Building Link State Packets



(a) A subnet. (b) The link state packets for this subnet.

- 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 neighbours.
- For each neighbour, the delay to that neighbour is given.
- An example subnet is given in Fig. 5-13(a) with delays shown as labels on the lines. The corresponding link state packets for all six routers are shown in Fig. 5-13(b).

Distributing the Link State Packets

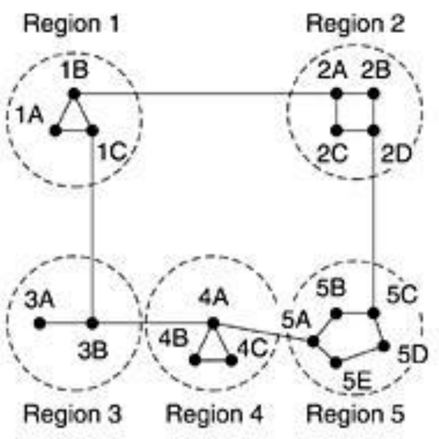
			Send flags			ACK flags			
Source	Seq.	Age	A	C	F	A	C	F	Data
A	21	60	0	1	1	1	0	0	
F	21	60	1	1	0	0	0	1	
E	21	59	0	1	0	1	0	1	
C	20	60	1	0	1	0	1	0	
D	21	59	1	0	0	0	1	1	

The packet buffer for router* B *in Fig. 5-13.

- In Fig. 5-14, 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*.

HIERARCHICAL ROUTING

- The routers are divided into what we will call regions, with each router knowing all the details about how to route packets to destinations within its own region, but knowing nothing about the internal structure of other regions.
- 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, until we run out of names for aggregations.



(a)

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

(b)

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

(c)

- Figure 5-15 gives a quantitative example of routing in a two-level hierarchy with five regions.
- The full routing table for router 1*A* has 17 entries, as shown in Fig. 5-15(b).
- When routing is done hierarchically, as in Fig. 5-15(c), there are entries for all the local routers as before, but all other regions have been condensed into a single router, so all traffic for region 2 goes via the 1*B* -2*A* line, but the rest of the remote traffic goes via the 1*C* -3*B* 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.

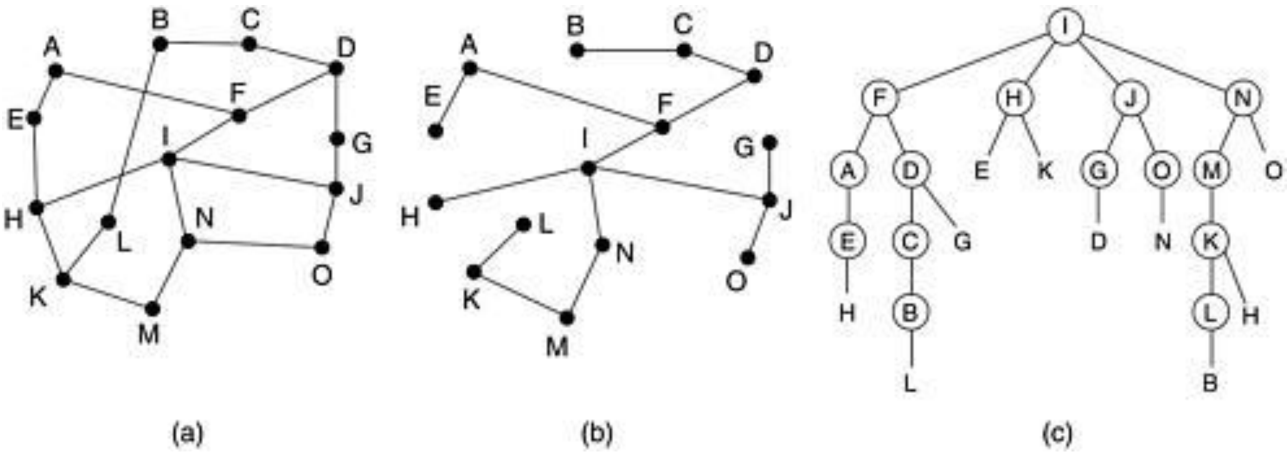
BROADCAST ROUTING

Sending a packet to all destinations simultaneously is called broadcasting.

1) The source simply sends a distinct packet to each destination. Not only is the method wasteful of bandwidth, but it also requires the source to have a complete list of all destinations.

2) Flooding.

The problem with flooding as a broadcast technique is that it generates too many packets and consumes too much bandwidth.



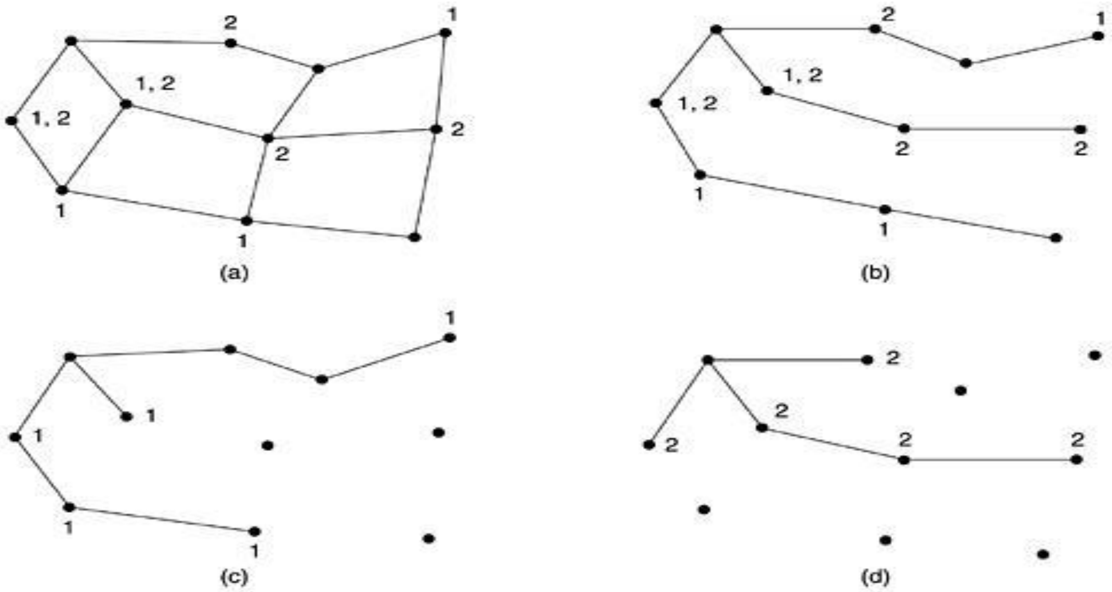
Reverse path forwarding. (a) A subnet. (b) A sink tree. (c) The tree built by reverse path forwarding.

Part (a) shows a subnet, part (b) shows a sink tree for router *I* of that subnet, and part (c) shows how the reverse path algorithm works.

- When a broadcast packet arrives at a router, the router checks to see if the packet arrived on the line that is normally used for sending packets to 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.

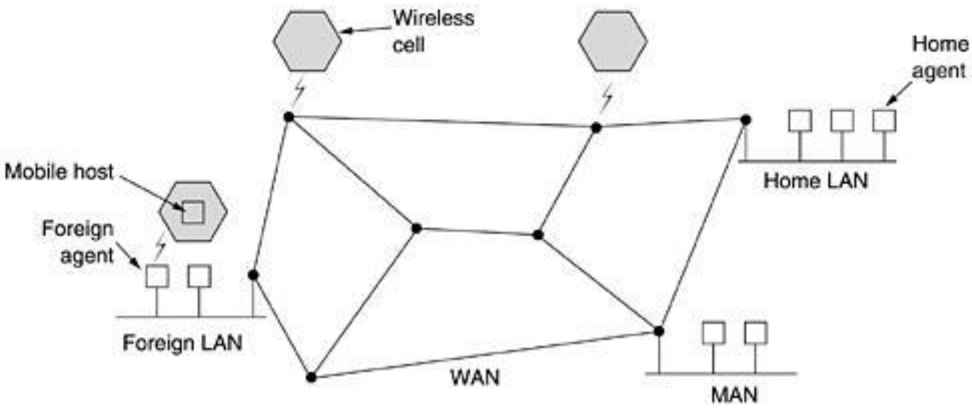
- This being the case, the router forwards copies of it onto all lines except the one it arrived on. If, however, the broadcast packet arrived on a line other than the preferred one for reaching the source, the packet is discarded as a likely duplicate.

MULTICAST ROUTING



- To do multicast routing, each router computes a spanning tree covering all other routers. For example, in Fig. 5-17(a) we have two groups, 1 and 2.
- 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. 5-17(b). When a process sends a multicast packet to a group, the first router examines its spanning tree and prunes it, removing all lines that do not lead to hosts that are members of the group.
- In our example, Fig. 5-17(c) shows the pruned spanning tree for group 1. Similarly, Fig. 5-17(d) shows the pruned spanning tree for group 2. Multicast packets are forwarded only along the appropriate spanning tree.

ROUTING FOR MOBILE HOSTS



- Hosts that never move are said to be stationary.
- They are connected to the network by copper wires or fiber optics. In contrast, we can distinguish two other kinds of hosts.

- Migratory hosts are basically stationary hosts who move from one fixed site to another from time to time but use the network only when they are physically connected to it.
- Roaming hosts actually compute on the run and want to maintain their connections as they move around.
- We will use the term **mobile hosts** to mean either of the latter two categories, that is, all hosts that are away from home and still want to be connected

The registration procedure typically works like this:

1. Periodically, each foreign agent broadcasts a packet announcing its existence and address. A newly-arrived mobile host may wait for one of these messages, but if none arrives quickly enough, the mobile host can broadcast a packet saying: Are there any foreign agents around?
2. The mobile host registers with the foreign agent, giving its home address, current data link layer address, and some security information.
3. The foreign agent contacts the mobile host's home agent and says: One of your hosts is over here. The message from the foreign agent to the home agent contains the foreign agent's network address. It also includes the security information to convince the home agent that the mobile host is really there.
4. The home agent examines the security information, which contains a timestamp, to prove that it was generated within the past few seconds. If it is happy, it tells the foreign agent to proceed.
5. When the foreign agent gets the acknowledgement from the home agent, it makes an entry in its tables and informs the mobile host that it is now registered.

ROUTING IN AD HOC NETWORKS

We have now seen how to do routing when the hosts are mobile but the routers are fixed. An even more extreme case is one in which the routers themselves are mobile. Among the possibilities are:

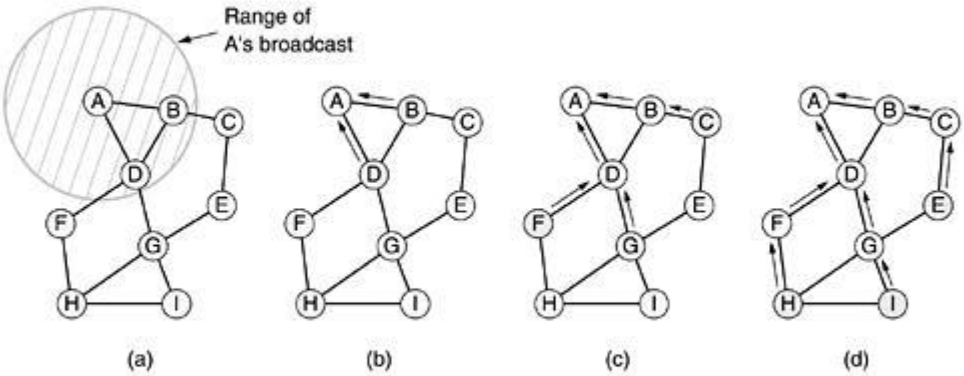
1. Military vehicles on a battlefield with no existing infrastructure.
2. A fleet of ships at sea.
3. Emergency workers at an earthquake that destroyed the infrastructure.
4. A gathering of people with notebook computers in an area lacking 802.11.

In all these cases, and others, each node consists of a router and a host, usually on the same computer. Networks of nodes that just happen to be near each other are called **ad hoc networks** or **MANETs** (**Mobile Ad hoc NETworks**).

- What makes ad hoc networks different from wired networks is that all the usual rules about fixed topologies, fixed and known neighbours, fixed relationship between IP address and location, and more are suddenly tossed out the window.

- Routers can come and go or appear in new places at the drop of a bit. With a wired network, if a router has a valid path to some destination, that path continues to be valid indefinitely (barring a failure somewhere in the system).
- With an ad hoc network, the topology may be changing all the time.
- A variety of routing algorithms for ad hoc networks have been proposed. One of the more interesting ones is the **AODV (Ad hoc On-demand Distance Vector**) routing algorithm (Perkins and Royer, 1999).
- It takes into account the limited bandwidth and low battery life found in environment. Another unusual characteristic is that it is an on-demand algorithm, that is, it determines a route to some destination only when somebody wants to send a packet to that destination. Let us now see what that means**.**

Route Discovery



(a) Range of* A*'s broadcast. (b) After* B *and* D *have received* A*'s broadcast. (c) After* C, F, *and* G *have received* A*'s broadcast. (d) After* E, H, *and* I *have received* A*'s broadcast. The shaded nodes are new recipients. The arrows show the possible reverse routes.

- To locate *I*, *A* constructs a special ROUTE REQUEST packet and broadcasts it. The packet reaches *B* and *D*, as illustrated in Fig. 5-20(a).
- The format of the ROUTE REQUEST packet is shown in Fig. 5-21

Format of a ROUTE REQUEST packet.

Source address	Request ID	Destination address	Source sequence #	Dest. sequence #	Hop count

The format of the ROUTE REQUEST packet is shown in Fig. 5-21. It contains the source and destination addresses, typically their IP addresses, which identify who is looking for whom. It also contains a *Request ID*, which is a local counter maintained separately by each node and incremented each time a ROUTE REQUEST is broadcast. Together, the *Source address* and *Request ID* fields uniquely identify the ROUTE REQUEST packet to allow nodes to discard any duplicates they may receive.

Format of a ROUTE REPLY packet

Source address	Destination address	Destination sequence #	Hop count	Lifetime

In addition to the *Request ID* counter, each node also maintains a second sequence counter incremented whenever a ROUTE REQUEST is sent (or a reply to someone else's ROUTE REQUEST). It functions a little bit like a clock and is used to tell new routes from old routes. The fourth field of Fig. 5-21 is *A*'s sequence counter; the fifth field is the most recent value of *I*'s sequence number that *A* has seen (0 if it has never seen it). The use of these fields will become clear shortly. The final field, *Hop count*, will keep track of how many hops the packet has made. It is initialized to 0.

1. No route to *I* is known.

2. The sequence number for *I* in the ROUTE REPLY packet is greater than the value in the routing table.

3. The sequence numbers are equal but the new route is shorter.

CONGESTION CONTROL ALGORITHMS

- When too many packets are present in (a part of) the subnet, performance degrades. This situation is called **congestion**.
- Figure 5-25 depicts the symptom. When the number of packets dumped into the subnet by the hosts is within its carrying capacity, they are all delivered (except for a few that are afflicted with transmission errors) and the number delivered is proportional to the number sent.
- However, as traffic increases too far, the routers are no longer able to cope and they begin losing packets. This tends to make matters worse. At very high traffic, performance collapses completely and almost no packets are delivered.

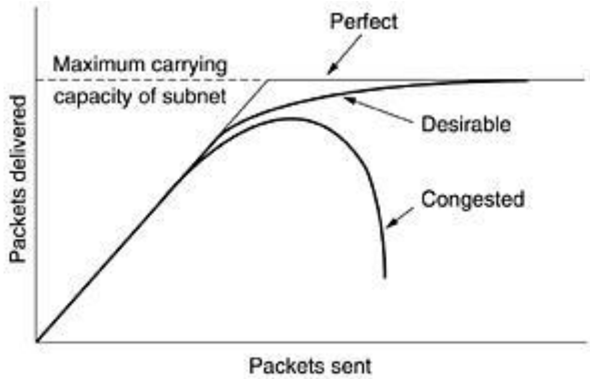


Figure 5-25. When too much traffic is offered, congestion sets in and performance degrades sharply.

- Congestion can be brought on by several factors. If all of a sudden, streams of packets begin arriving on three or four input lines and all need the same output line, a queue will build up.
- If there is insufficient memory to hold all of them, packets will be lost.
- Slow processors can also cause congestion. If the routers' CPUs are slow at performing the bookkeeping tasks required of them (queuing buffers, updating tables, etc.), queues can build up, even though there is excess line capacity. Similarly, low-bandwidth lines can also cause congestion.

APPROACHES TO CONGESTION CONTROL

- Many problems in complex systems, such as computer networks, can be viewed from a control theory point of view. This approach leads to dividing all solutions into two groups: open loop and closed loop.

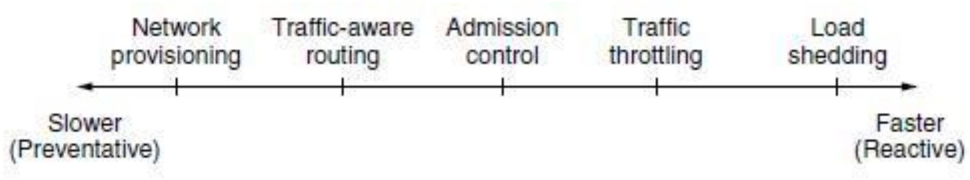


Figure: Timescales Of Approaches To Congestion Control

- Open loop solutions attempt to solve the problem by good design.
- Tools for doing open-loop control include deciding when to accept new traffic, deciding when to discard packets and which ones, and making scheduling decisions at various points in the network.
- Closed loop solutions are based on the concept of a feedback loop.
- This approach has three parts when applied to congestion control:
  1. Monitor the system to detect when and where congestion occurs.
  2. Pass this information to places where action can be taken.
  3. Adjust system operation to correct the problem.
- A variety of metrics can be used to monitor the subnet for congestion. Chief among these are the percentage of all packets discarded for lack of buffer space, the average queue lengths, the number of packets that time out and are retransmitted, the average packet delay, and the standard deviation of packet delay. In all cases, rising numbers indicate growing congestion.
- The second step in the feedback loop is to transfer the information about the congestion from the point where it is detected to the point where something can be done about it.

- In all feedback schemes, the hope is that knowledge of congestion will cause the hosts to take appropriate action to reduce the congestion.
- The presence of congestion means that the load is (temporarily) greater than the resources (in part of the system) can handle. Two solutions come to mind: increase the resources or decrease the load.

CONGESTION PREVENTION POLICIES

The methods to control congestion by looking at open loop systems. These systems are designed to minimize congestion in the first place, rather than letting it happen and reacting after the fact. They try to achieve their goal by using appropriate policies at various levels. In Fig. 5-26 we see different data link, network, and transport policies that can affect congestion (Jain, 1990).

Layer	Policies
Transport	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy • Timeout determination
Network	• Virtual circuits versus datagram inside the subnet • Packet queueing and service policy • Packet discard policy • Routing algorithm • Packet lifetime management
Data link	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy

Figure 5-26. Policies that affect congestion.

The **data link layer Policies.**

- The **retransmission policy** is concerned with how fast a sender times out and what it transmits upon timeout. A jumpy sender that times out quickly and retransmits all outstanding packets using go back n will put a heavier load on the system than will a leisurely sender that uses selective repeat.
- Closely related to this is the **buffering policy**. If receivers routinely discard all out-of-order packets, these packets will have to be transmitted again later, creating extra load. With respect to congestion control, selective repeat is clearly better than go back n.
- **Acknowledgement policy** also affects congestion. If each packet is acknowledged immediately, the acknowledgement packets generate extra traffic. However, if acknowledgements are saved up to piggyback onto reverse traffic, extra timeouts and retransmissions may result. A tight flow control scheme (e.g., a small window) reduces the data rate and thus helps fight congestion.

The **network layer Policies.**

- The choice between using **virtual circuits and using datagrams** affects congestion since many congestion control algorithms work only with virtual-circuit subnets.
- **Packet queueing and service policy** relates to whether routers have one queue per input line, one queue per output line, or both. It also relates to the order in which packets are processed (e.g., round robin or priority based).
- **Discard policy** is the rule telling which packet is dropped when there is no space.
- A good **routing algorithm** can help avoid congestion by spreading the traffic over all the lines, whereas a bad one can send too much traffic over already congested lines.
- **Packet lifetime management** deals with how long a packet may live before being discarded. If it is too long, lost packets may clog up the works for a long time, but if it is too short, packets may sometimes time out before reaching their destination, thus inducing retransmissions.

The **transport layer Policies**,

- The **same issues occur as in the data link layer**, but in addition, determining the **timeout interval** is harder because the transit time across the network is less predictable than the transit time over a wire between two routers. If the timeout interval is too short, extra packets will be sent unnecessarily. If it is too long, congestion will be reduced but the response time will suffer whenever a packet is lost.

ADMISSION CONTROL

- One technique that is widely used to keep congestion that has already started from getting worse is **admission control**.
- Once congestion has been signaled, no more virtual circuits are set up until the problem has gone away.
- An alternative approach is to allow new virtual circuits but carefully route all new virtual circuits around problem areas. For example, consider the subnet of Fig. 5-27(a), in which two routers are congested, as indicated.

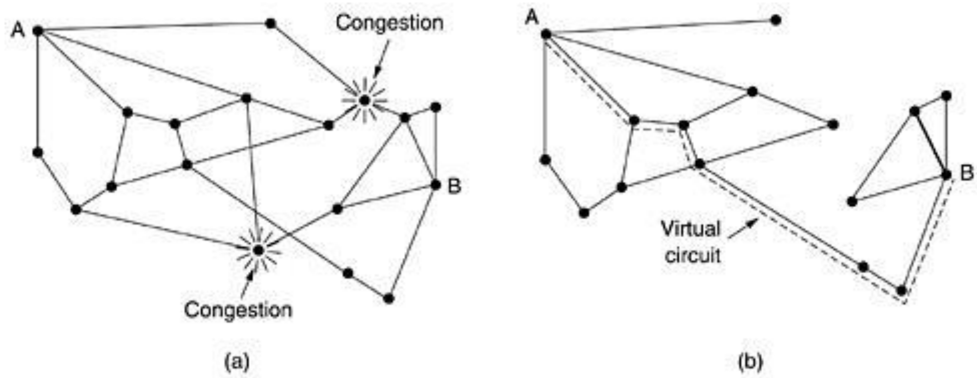


Figure 5-27. (a) A congested subnet. (b) A redrawn subnet that eliminates the congestion. A virtual circuit from* A *to* B *is also shown.

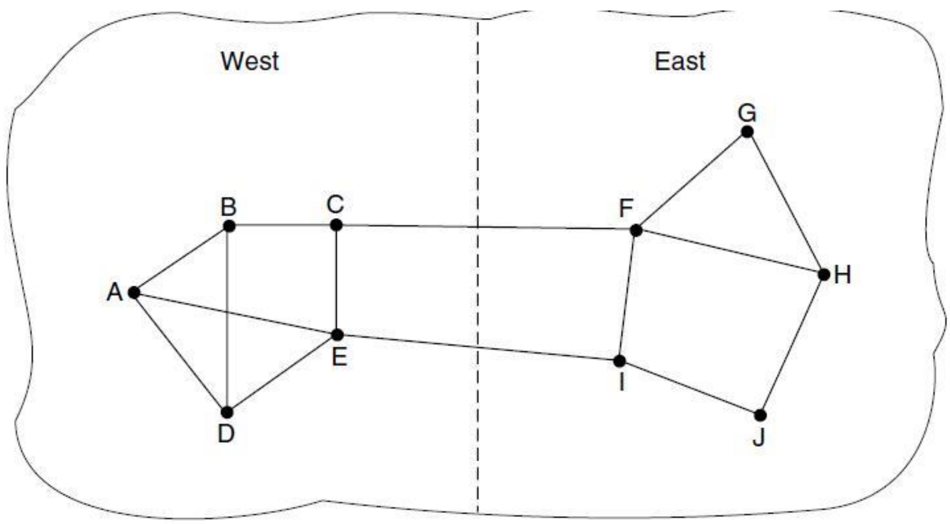
Suppose that a host attached to router *A* wants to set up a connection to a host attached to router *B*. Normally, this connection would pass through one of the congested routers. To avoid this situation, we can redraw the subnet as shown in Fig. 5-27(b), omitting the congested routers and all of their lines. The dashed line shows a possible route for the virtual circuit that avoids the congested routers.

TRAFFIC AWARE ROUTING

These schemes adapted to changes in topology, but not to changes in load. The goal in taking load into account when computing routes is to shift traffic away from hotspots that will be the first places in the network to experience congestion.

The most direct way to do this is to set the link weight to be a function of the (fixed) link bandwidth and propagation delay plus the (variable) measured load or average queuing delay. Least-weight paths will then favor paths that are more lightly loaded, all else being equal.

Consider the network of Fig. 5-23, which is divided into two parts, East and West, connected by two links, *CF* and *EI*. Suppose that most of the traffic between East and West is using link *CF*, and, as a result, this link is heavily loaded with long delays. Including queueing delay in the weight used for the shortest path calculation will make *EI* more attractive. After the new routing tables have been installed, most of the East-West traffic will now go over *EI*, loading this link. Consequently, in the next update, *CF* will appear to be the shortest path. As a result, the routing tables may oscillate wildly, leading to erratic routing and many potential problems.



If load is ignored and only bandwidth and propagation delay are considered, this problem does not occur. Attempts to include load but change weights within a narrow range only slow down routing oscillations. Two techniques can contribute to a successful solution. The first is multipath routing, in which there can be multiple paths from a source to a destination. In our example this means that the traffic can be spread across both of the East to West links. The second one is for the routing scheme to shift traffic across routes slowly enough that it is able to converge.

TRAFFIC THROTTLING

- Each router can easily monitor the utilization of its output lines and other resources. For example, it can associate with each line a real variable, u, whose value, between 0.0 and 1.0, reflects the recent utilization of that line. To maintain a good estimate of u, a sample of the instantaneous line utilization, f (either 0 or 1), can be made periodically and u updated according to

$$u_{\text{new}} = au_{\text{old}} + (1-a)f$$

where the constant a determines how fast the router forgets recent history.

Whenever u moves above the threshold, the output line enters a "warning" state. Each newly-arriving packet is checked to see if its output line is in warning state. If it is, some action is taken. The action taken can be one of several alternatives, which we will now discuss.

THE WARNING BIT

- The old DECNET architecture signaled the warning state by setting a special bit in the packet's header.
- When the packet arrived at its destination, the transport entity copied the bit into the next acknowledgement sent back to the source. The source then cut back on traffic.
- As long as the router was in the warning state, it continued to set the warning bit, which meant that the source continued to get acknowledgements with it set.
- The source monitored the fraction of acknowledgements with the bit set and adjusted its transmission rate accordingly. As long as the warning bits continued to flow in, the source continued to decrease its transmission rate. When they slowed to a trickle, it increased its transmission rate.
- Note that since every router along the path could set the warning bit, traffic increased only when no router was in trouble.

CHOKE PACKETS

- In this approach, the router sends a **choke packet** back to the source host, giving it the destination found in the packet.
- The original packet is tagged (a header bit is turned on) so that it will not generate any more choke packets farther along the path and is then forwarded in the usual way.
- When the source host gets the choke packet, it is required to reduce the traffic sent to the specified destination by *X* percent. Since other packets aimed at the same destination are probably already under way and will generate yet more choke packets, the host should ignore choke packets referring to that destination for a fixed time interval. After that period has expired, the host listens for more choke packets for another interval. If one

arrives, the line is still congested, so the host reduces the flow still more and begins ignoring choke packets again. If no choke packets arrive during the listening period, the host may increase the flow again.

- The feedback implicit in this protocol can help prevent congestion yet not throttle any flow unless trouble occurs.
- Hosts can reduce traffic by adjusting their policy parameters.
- Increases are done in smaller increments to prevent congestion from reoccurring quickly.
- Routers can maintain several thresholds. Depending on which threshold has been crossed, the choke packet can contain a mild warning, a stern warning, or an ultimatum.

HOP-BY-HOP BACK PRESSURE

- At high speeds or over long distances, sending a choke packet to the source hosts does not work well because the reaction is so slow.

Consider, for example, a host in San Francisco (router *A* in Fig. 5-28) that is sending traffic to a host in New York (router *D* in Fig. 5-28) at 155 Mbps. If the New York host begins to run out of buffers, it will take about 30 msec for a choke packet to get back to San Francisco to tell it to slow down. The choke packet propagation is shown as the second, third, and fourth steps in Fig. 5-28(a). In those 30 msec, another 4.6 megabits will have been sent. Even if the host in San Francisco completely shuts down immediately, the 4.6 megabits in the pipe will continue to pour in and have to be dealt with. Only in the seventh diagram in Fig. 5-28(a) will the New York router notice a slower flow.

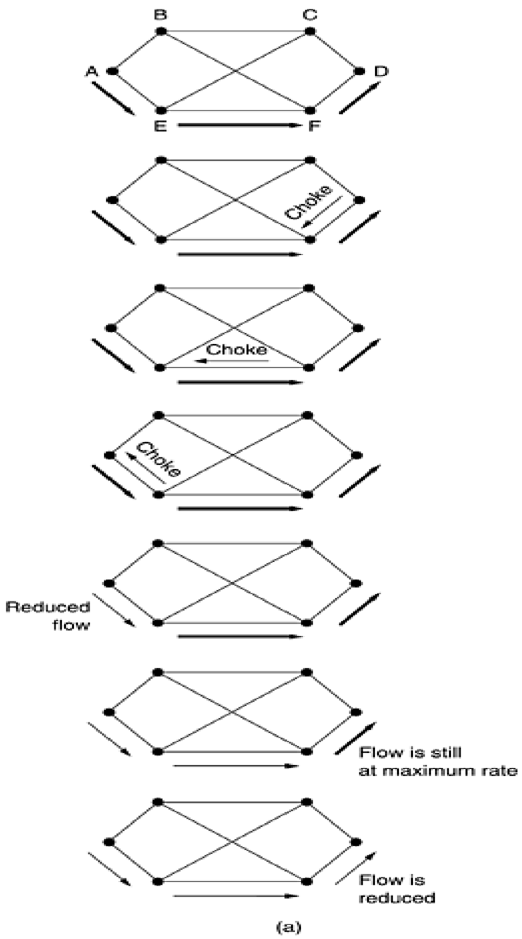


Figure 5-28. (a) A choke packet that affects only the source. (b) A choke packet that affects each hop it passes through.

An alternative approach is to have the choke packet take effect at every hop it passes through, as shown in the sequence of Fig. 5-28(b). Here, as soon as the choke packet reaches *F*, *F* is required to reduce the flow to *D*. Doing so will require *F* to devote more buffers to the flow, since the source is still sending away at full blast, but it gives *D* immediate relief, like a headache remedy in a television commercial. In the next step, the choke packet reaches *E*, which tells *E* to reduce the flow to *F*. This action puts a greater demand on *E*'s buffers but gives *F* immediate relief. Finally, the choke packet reaches *A* and the flow genuinely slows down.

The net effect of this hop-by-hop scheme is to provide quick relief at the point of congestion at the price of using up more buffers upstream. In this way, congestion can be nipped in the bud without losing any packets.

LOAD SHEDDING

- When none of the above methods make the congestion disappear, routers can bring out the heavy artillery: load shedding.
- **Load shedding** is a fancy way of saying that when routers are being in undated by packets that they cannot handle, they just throw them away.

- A router drowning in packets can just pick packets at random to drop, but usually it can do better than that.
- Which packet to discard may depend on the applications running.
- To implement an intelligent discard policy, applications must mark their packets in priority classes to indicate how important they are. If they do this, then when packets have to be discarded, routers can first drop packets from the lowest class, then the next lowest class, and so on.

RANDOM EARLY DETECTION

- It is well known that dealing with congestion after it is first detected is more effective than letting it gum up the works and then trying to deal with it. This observation leads to the idea of discarding packets before all the buffer space is really exhausted. A popular algorithm for doing this is called **RED** (**Random Early Detection)**.
- In some transport protocols (including TCP), the response to lost packets is for the source to slow down. The reasoning behind this logic is that TCP was designed for wired networks and wired networks are very reliable, so lost packets are mostly due to buffer overruns rather than transmission errors. This fact can be exploited to help **reduce congestion**.
- By having routers drop packets before the situation has become hopeless (hence the "early" in the name), the idea is that there is time for action to be taken before it is too late. To determine when to start discarding, routers maintain a running average of their queue lengths. When the average queue length on some line exceeds a threshold, the line is said to be congested and action is taken.

JITTER CONTROL

- The variation (i.e., standard deviation) in the packet arrival times is called **jitter**.
- High jitter, for example, having some packets taking 20 msec and others taking 30 msec to arrive will give an uneven quality to the sound or movie. Jitter is illustrated in Fig. 5-29. In contrast, an agreement that 99 percent of the packets be delivered with a delay in the range of 24.5 msec to 25.5 msec might be acceptable.

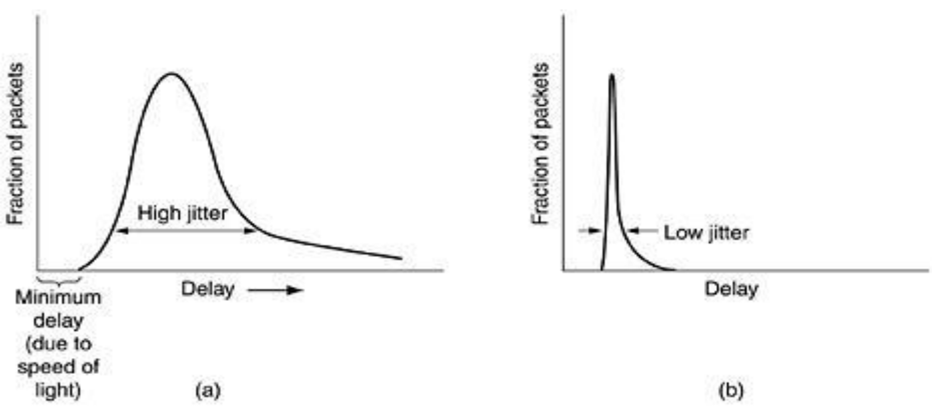
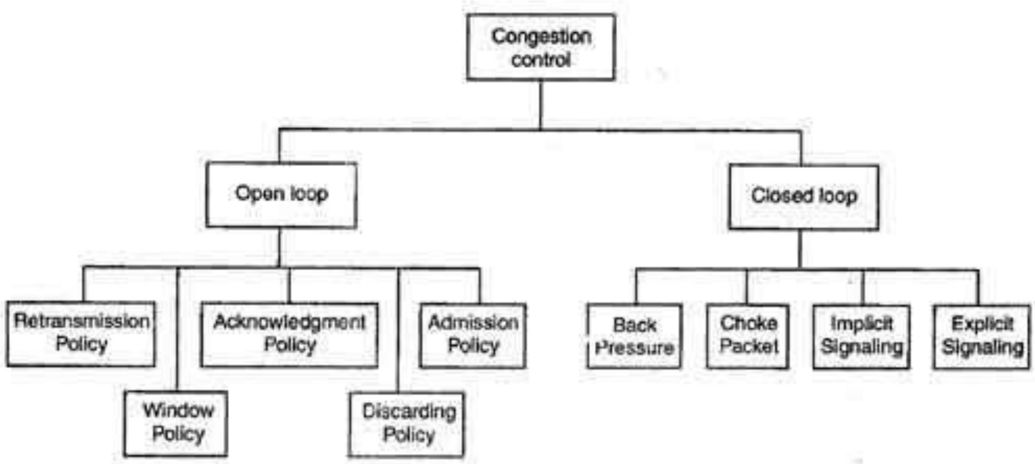


Figure 5-29. (a) High jitter. (b) Low jitter.

- The jitter can be bounded by computing the expected transit time for each hop along the path. When a packet arrives at a router, the router checks to see how much the packet is behind or ahead of its schedule. This information is stored in the packet and updated at each hop. If the packet is ahead of schedule, it is held just long enough to get it back on schedule. If it is behind schedule, the router tries to get it out the door quickly.
- In fact, the algorithm for determining which of several packets competing for an output line should go next can always choose the packet furthest behind in its schedule.
- In this way, packets that are ahead of schedule get slowed down and packets that are behind schedule get speeded up, in both cases reducing the amount of jitter.
- In some applications, such as video on demand, jitter can be eliminated by buffering at the receiver and then fetching data for display from the buffer instead of from the network in real time. However, for other applications, especially those that require real-time interaction between people such as Internet telephony and videoconferencing, the delay inherent in buffering is not acceptable.

How to correct the Congestion Problem:

Congestion Control refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. Congestion control mechanisms are divided into two categories, one category prevents the congestion from happening and the other category removes congestion after it has taken place.



Types of Congestion Control Methods

These two categories are:

1. Open loop
2. Closed loop

Open Loop Congestion Control

• In this method, policies are used to prevent the congestion before it happens.

• Congestion control is handled either by the source or by the destination.

• The various methods used for open loop congestion control are:

1. Retransmission Policy

• The sender retransmits a packet, if it feels that the packet it has sent is lost or corrupted.

• However retransmission in general may increase the congestion in the network. But we need to implement good retransmission policy to prevent congestion.

• The retransmission policy and the retransmission timers need to be designed to optimize efficiency and at the same time prevent the congestion.

2. Window Policy

• To implement window policy, selective reject window method is used for congestion control.

• Selective Reject method is preferred over Go-back-n window as in Go-back-n method, when timer for a packet times out, several packets are resent, although some may have arrived safely at the receiver. Thus, this duplication may make congestion worse.

• Selective reject method sends only the specific lost or damaged packets.

3. Acknowledgement Policy

• The acknowledgement policy imposed by the receiver may also affect congestion.

• If the receiver does not acknowledge every packet it receives it may slow down the sender and help prevent congestion.

• Acknowledgments also add to the traffic load on the network. Thus, by sending fewer acknowledgements we can reduce load on the network.

• To implement it, several approaches can be used:

1. A receiver may send an acknowledgement only if it has a packet to be sent.

2. A receiver may send an acknowledgement when a timer expires.

3. A receiver may also decide to acknowledge only *N* packets at a time.

4. Discarding Policy

• A router may discard less sensitive packets when congestion is likely to happen.

• Such a discarding policy may prevent congestion and at the same time may not harm the integrity of the transmission.

5. Admission Policy

• An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual circuit networks.

• Switches in a flow first check the resource requirement of a flow before admitting it to the network.

• A router can deny establishing a virtual circuit connection if there is congestion in the "network or if there is a possibility of future congestion.

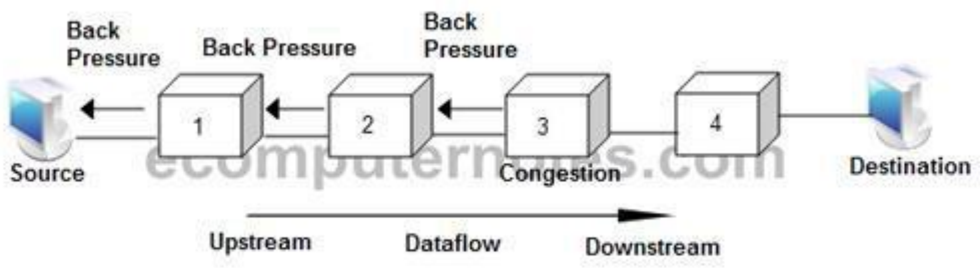
Closed Loop Congestion Control

• Closed loop congestion control mechanisms try to remove the congestion after it happens.

• The various methods used for closed loop congestion control are:

1. Backpressure

• Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow.



Backpressure Method

• The backpressure technique can be applied only to virtual circuit networks. In such virtual circuit each node knows the upstream node from which a data flow is coming.

• In this method of congestion control, the congested node stops receiving data from the immediate upstream node or nodes.

• This may cause the upstream node on nodes to become congested, and they, in turn, reject data from their upstream node or nodes.

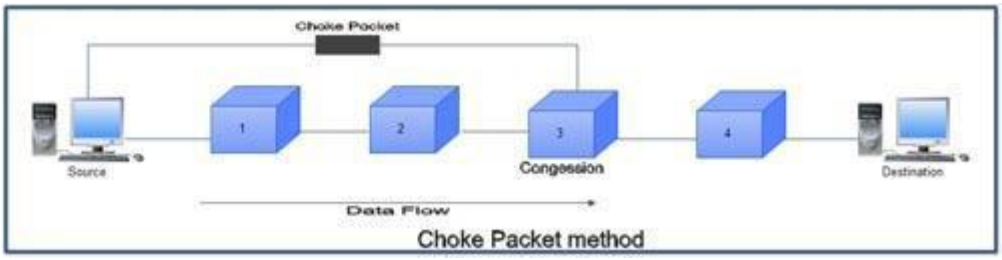
• As shown in fig node 3 is congested and it stops receiving packets and informs its upstream node 2 to slow down. Node 2 in turns may be congested and informs node 1 to slow down. Now node 1 may create congestion and informs the source node to slow down. In this way the congestion is alleviated. Thus, the pressure on node 3 is moved backward to the source to remove the congestion.

2. Choke Packet

• In this method of congestion control, congested router or node sends a special type of packet called choke packet to the source to inform it about the congestion.

• Here, congested node does not inform its upstream node about the congestion as in backpressure method.

• In choke packet method, congested node sends a warning directly to the source station *i.e.* the intermediate nodes through which the packet has traveled are not warned.



3. Implicit Signaling

• In implicit signaling, there is no communication between the congested node or nodes and the source.

• The source guesses that there is congestion somewhere in the network when it does not receive any acknowledgment. Therefore the delay in receiving an acknowledgment is interpreted as congestion in the network.

• On sensing this congestion, the source slows down.

• This type of congestion control policy is used by TCP.

4. Explicit Signaling

• In this method, the congested nodes explicitly send a signal to the source or destination to inform about the congestion.

• Explicit signaling is different from the choke packet method. In choke packed method, a separate packet is used for this purpose whereas in explicit signaling method, the signal is included in the packets that carry data .

• Explicit signaling can occur in either the forward direction or the backward direction .

• In backward signaling, a bit is set in a packet moving in the direction opposite to the congestion. This bit warns the source about the congestion and informs the source to slow down.

• In forward signaling, a bit is set in a packet moving in the direction of congestion. This bit warns the destination about the congestion. The receiver in this case uses policies such as slowing down the acknowledgements to remove the congestion.

TUNNELING

- Handling the general case of making two different networks interwork is exceedingly difficult. However, there is a common special case that is manageable.
- This case is where the source and destination hosts are on the same type of network, but there is a different network in between.
- As an example, think of an international bank with a TCP/IP-based Ethernet in Paris, a TCP/IP-based Ethernet in London, and a non-IP wide area network (e.g., ATM) in between, as shown in Fig. 5-47.

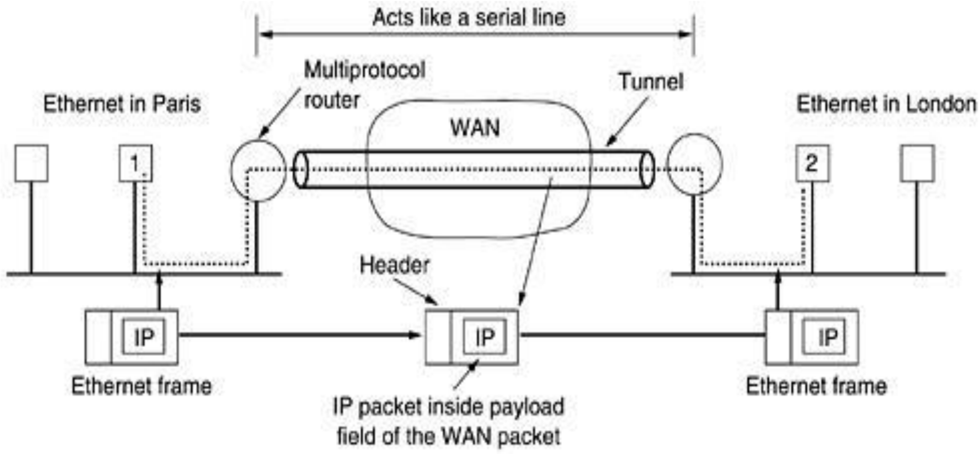


Figure 5-47. Tunneling a packet from Paris to London.

- The solution to this problem is a technique called **tunneling**.
- To send an IP packet to host 2, host 1 constructs the packet containing the IP address of host 2, inserts it into an Ethernet frame addressed to the Paris multiprotocol router, and puts it on the Ethernet. When the multiprotocol router gets the frame, it removes the IP packet, inserts it in the payload field of the WAN network layer packet, and addresses the latter to the WAN address of the London multiprotocol router. When it gets there, the London router removes the IP packet and sends it to host 2 inside an Ethernet frame.
- The WAN can be seen as a big tunnel extending from one multiprotocol router to the other. The IP packet just travels from one end of the tunnel to the other, snug in its nice box. Neither do the hosts on either Ethernet. Only the multiprotocol router has to

understand IP and WAN packets. In effect, the entire distance from the middle of one multiprotocol router to the middle of the other acts like a serial line.

- Consider a person driving her car from Paris to London. Within France, the car moves under its own power, but when it hits the English Channel, it is loaded into a high-speed train and transported to England through the Chunnel (cars are not permitted to drive through the Chunnel). Effectively, the car is being carried as freight, as depicted in Fig. 5-48. At the far end, the car is let loose on the English roads and once again continues to move under its own power. Tunneling of packets through a foreign network works the same way.

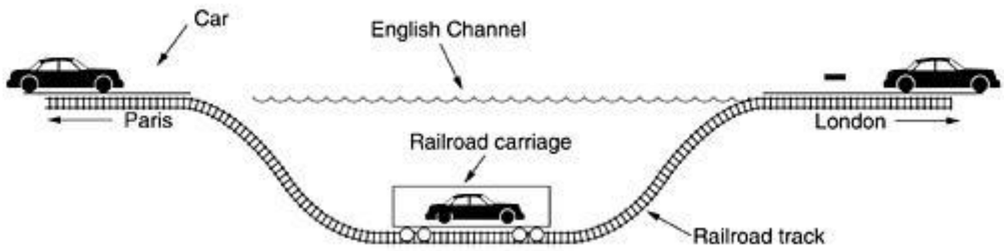


Figure 5-48. Tunneling a car from France to England.

INTERNETWORK ROUTING

- Routing through an internetwork is similar to routing within a single subnet, but with some added complications.
- Consider, for example, the internetwork of Fig. 5-49(a) in which five networks are connected by six (possibly multiprotocol) routers. Making a graph model of this situation is complicated by the fact that every router can directly access (i.e., send packets to) every other router connected to any network to which it is connected. For example, *B* in Fig. 5-49(a) can directly access *A* and *C* via network 2 and also *D* via network 3. This leads to the graph of Fig. 5-49(b).

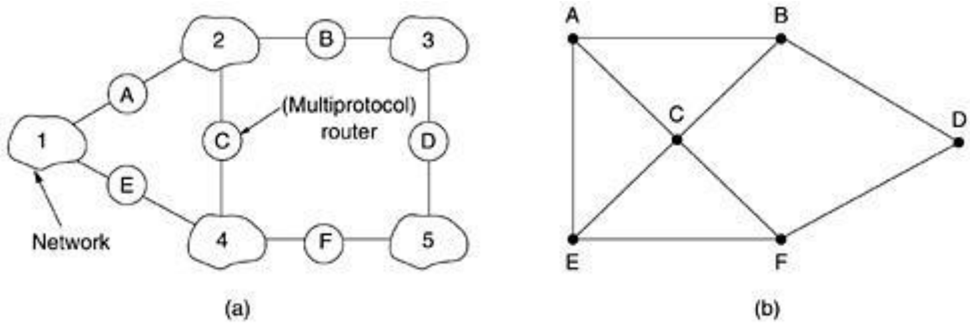


Figure 5-49. (a) An internetwork. (b) A graph of the internetwork.

- Once the graph has been constructed, known routing algorithms, such as the distance vector and link state algorithms, can be applied to the set of multiprotocol routers.
- This gives a two-level routing algorithm: within each network an **interior gateway protocol** is used, but between the networks, an **exterior gateway protocol** is used ("gateway" is an older term for "router").

- Network in an internetwork is independent of all the others, it is often referred to as an **Autonomous System** (AS).
- A typical internet packet starts out on its LAN addressed to the local multiprotocol router (in the MAC layer header). After it gets there, the network layer code decides which multiprotocol router to forward the packet to, using its own routing tables. If that router can be reached using the packet's native network protocol, the packet is forwarded there directly. Otherwise it is tunneled there, encapsulated in the protocol required by the intervening network. This process is repeated until the packet reaches the destination network.
- One of the differences between internetwork routing and intranet work routing is that internetwork routing may require crossing international boundaries. Various laws suddenly come into play, such as Sweden's strict privacy laws about exporting personal data about Swedish citizens from Sweden. Another example is the Canadian law saying that data traffic originating in Canada and ending in Canada may not leave the country. This law means that traffic from Windsor, Ontario to Vancouver may not be routed via nearby Detroit, even if that route is the fastest and cheapest.

Another difference between interior and exterior routing is the cost. Within a single network, a single charging algorithm normally applies. However, different networks may be under different managements, and one route may be less expensive than another. Similarly, the quality of service offered by different networks may be different, and this may be a reason to choose one route over another.

FRAGMENTATION

Each network imposes some maximum size on its packets. These limits have various causes, among them:

1. Hardware (e.g., the size of an Ethernet frame).
2. Operating system (e.g., all buffers are 512 bytes).
3. Protocols (e.g., the number of bits in the packet length field).
4. Compliance with some (inter)national standard.
5. Desire to reduce error-induced retransmissions to some level.
6. Desire to prevent one packet from occupying the channel too long.

- From the above factors maximum payloads range from 48 bytes (ATM cells) to 65,515 bytes (IP packets), although the payload size in higher layers is often larger.
- If the original source packet is too large to be handled by the destination network? The routing algorithm can hardly bypass the destination.
- The only solution to the problem is to allow gateways to break up packets into **fragments**, sending each fragment as a separate internet packet. Packet-switching networks, too, have trouble putting the fragments back together again.

Two opposing strategies exist for recombining the fragments back into the original packet.

The **first strategy** is to make fragmentation caused by a "small-packet" network transparent to any subsequent networks through which the packet must pass on its way to the ultimate destination. This option is shown in Fig. 5-50(a). In this approach, the small-packet network has gateways (most likely, specialized routers) that interface to other networks. When an oversized packet arrives at a gateway, the gateway breaks it up into fragments. Each fragment is addressed to the same exit gateway, where the pieces are recombined. In this way passage through the small-packet network has been made transparent. Subsequent networks are not even aware that fragmentation has occurred.

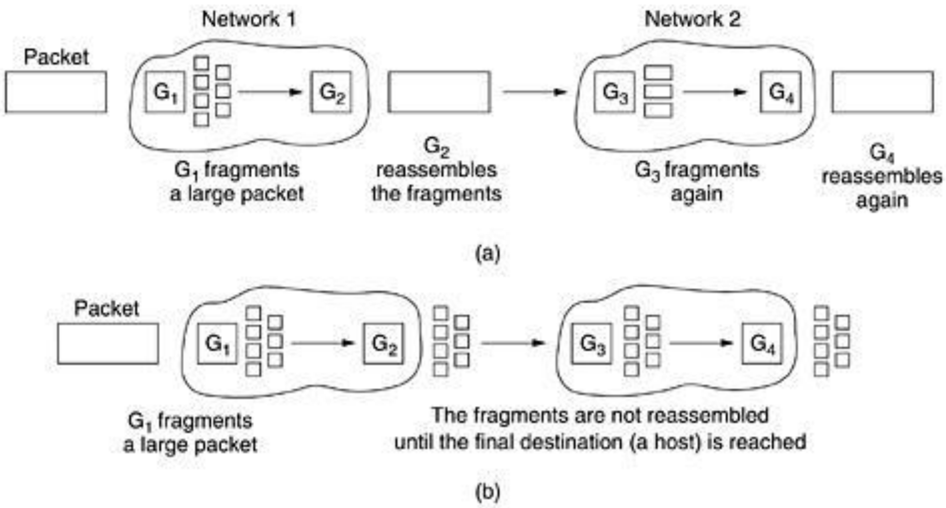


Figure 5-50. (a) Transparent fragmentation. (b) Nontransparent fragmentation.

- Transparent fragmentation is straightforward but has some problems.
- For one thing, the exit gateway must know when it has received all the pieces, so either a count field or an "end of packet" bit must be provided.
- For another thing, all packets must exit via the same gateway. By not allowing some fragments to follow one route to the ultimate destination and other fragments a disjoint route, some performance may be lost.
- A last problem is the overhead required to repeatedly reassemble and then refragment a large packet passing through a series of small-packet networks. ATM requires transparent fragmentation.

The **other fragmentation strategy** is to refrain from recombining fragments at any intermediate gateways. Once a packet has been fragmented, each fragment is treated as though it were an original packet. All fragments are passed through the exit gateway (or gateways), as shown in Fig. 5-50(b). Recombination occurs only at the destination host. IP works this way.

- Nontransparent fragmentation also has some problems.
- For example, it requires *every* host to be able to do reassembly.

- Yet another problem is that when a large packet is fragmented, the total overhead increases because each fragment must have a header. Whereas in the first method this overhead disappears as soon as the small-packet network is exited, in this method the overhead remains for the rest of the journey.
- An advantage of nontransparent fragmentation, however, is that multiple exit gateways can now be used and higher performance can be achieved.
- When a packet is fragmented, the fragments must be numbered in such a way that the original data stream can be reconstructed.
- One way of numbering the fragments is to use a tree. If packet 0 must be split up, the pieces are called 0.0, 0.1, 0.2, etc. If these fragments themselves must be fragmented later on, the pieces are numbered 0.0.0, 0.0.1, 0.0.2, . . . , 0.1.0, 0.1.1, 0.1.2, etc.
- No duplicates are generated anywhere, this scheme is sufficient to ensure that all the pieces can be correctly reassembled at the destination, no matter what order they arrive in.
- However, if even one network loses or discards packets, end-to-end retransmissions are needed, with unfortunate effects for the numbering system.
- Suppose that a 1024-bit packet is initially fragmented into four equal-sized fragments, 0.0, 0.1, 0.2, and 0.3. Fragment 0.1 is lost, but the other parts arrive at the destination. Eventually, the source time out and retransmits the original packet again.
- When a packet is fragmented, all the pieces are equal to the elementary fragment size except the last one, which may be shorter. An internet packet may contain several fragments, for efficiency reasons.
- The internet header must provide the original packet number and the number of the (first) elementary fragment contained in the packet.
- A bit indicating that the last elementary fragment contained within the internet packet is the last one of the original packet.

This approach requires two sequence fields in the internet header:

- The original packet number and the fragment number. There is clearly a trade-off between the size of the elementary fragment and the number of bits in the fragment number. Because the elementary fragment size is presumed to be acceptable to every network, subsequent fragmentation of an internet packet containing several fragments causes no problem. The ultimate limit here is to have the elementary fragment be a single bit or byte, with the fragment number then being the bit or byte offset within the original packet, as shown in Fig. 5-51.

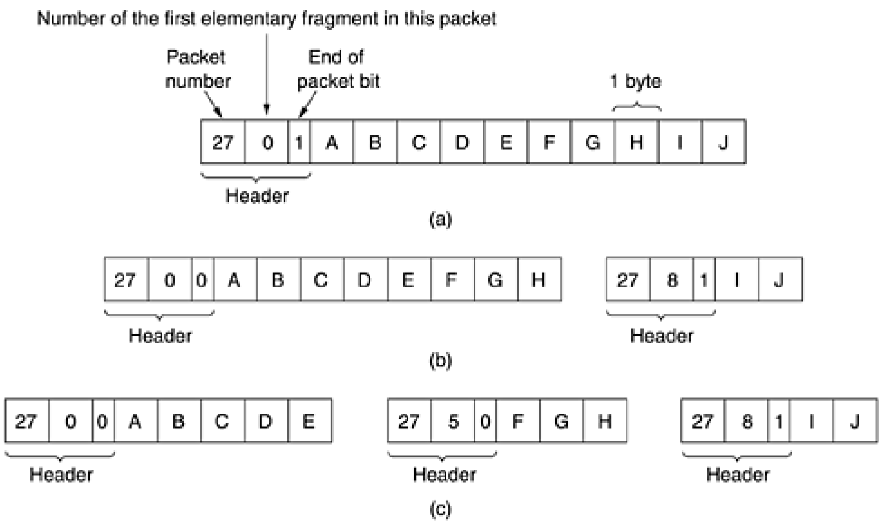


Figure 5-51. Fragmentation when the elementary data size is 1 byte. (a) Original packet, containing 10 data bytes. (b) Fragments after passing through a network with maximum packet size of 8 payload bytes plus header. (c) Fragments after passing through a size 5 gateway.

Some internet protocols take this method even further and consider the entire transmission on a virtual circuit to be one giant packet, so that each fragment contains the absolute byte number of the first byte within the fragment.

5. Explain about the network layer in the internet in detail /

What is IP? Explain IPV 4 header. (Nov'10, May'10, Nov'07)

Top 10 principles

1. **Make sure it works.** Do not finalize the design or standard until multiple prototypes have successfully communicated with each other. All too often designers first write a

   1000-page standard, get it approved, then discover it is deeply flawed and does not work. Then they write version 1.1 of the standard. This is not the way to go.

2. **Keep it simple.** When in doubt, use the simplest solution. William of Occam stated this principle (Occam's razor) in the 14th century. Put in modern terms: fight features. If a feature is not absolutely essential, leave it out, especially if the same effect can be achieved by combining other features.

3. **Make clear choices.** If there are several ways of doing the same thing, choose one. Having two or more ways to do the same thing is looking for trouble. Standards often have multiple options or modes or parameters because several powerful parties insist that their way is best. Designers should strongly resist this tendency. Just say no.

4. **Exploit modularity.** This principle leads directly to the idea of having protocol stacks, each of whose layers is independent of all the other ones. In this way, if circumstances that require one module or layer to be changed, the other ones will not be affected.
5. **Expect heterogeneity.** Different types of hardware, transmission facilities, and applications will occur on any large network. To handle them, the network design must be simple, general, and flexible.
6. **Avoid static options and parameters.** If parameters are unavoidable (e.g., maximum packet size), it is best to have the sender and receiver negotiate a value than defining fixed choices.
7. **Look for a good design; it need not be perfect.** Often the designers have a good design but it cannot handle some weird special case. Rather than messing up the design, the designers should go with the good design and put the burden of working around it on the people with the strange requirements.
8. **Be strict when sending and tolerant when receiving.** In other words, only send packets that rigorously comply with the standards, but expect incoming packets that may not be fully conformant and try to deal with them.
9. **Think about scalability.** If the system is to handle millions of hosts and billions of users effectively, no centralized databases of any kind are tolerable and load must be spread as evenly as possible over the available resources.
10. **Consider performance and cost.** If a network has poor performance or outrageous costs, nobody will use it.

- At the network layer, the Internet can be viewed as a collection of subnetworks or **Autonomous Systems** (**ASes**) that are interconnected.
- There is no real structure, but several major backbones exist. These are constructed from high-bandwidth lines and fast routers. Attached to the backbones are regional (midlevel) networks, and attached to these regional networks are the LANs at many universities, companies, and Internet service providers.
- A sketch of this quasi-hierarchical organization is given in Fig. 5-52.

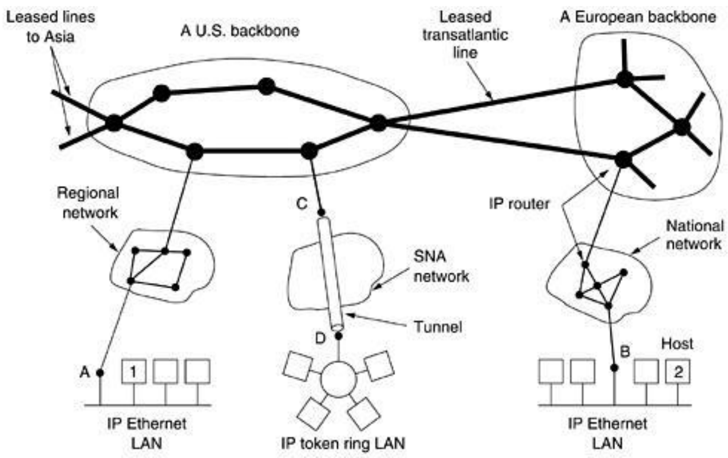
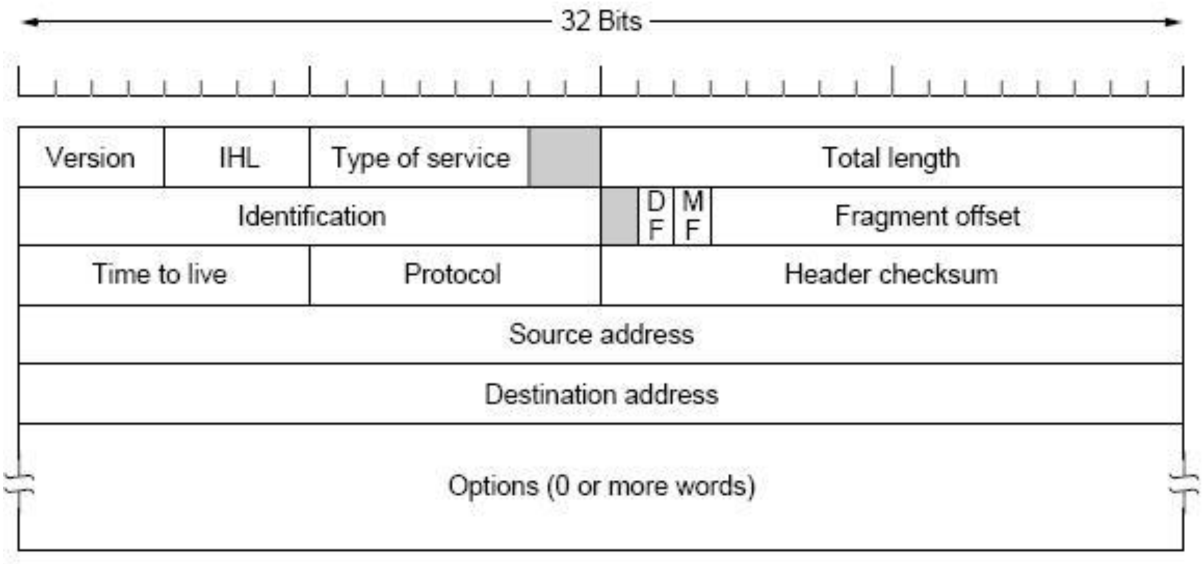


Figure 5-52. The Internet is an interconnected collection of many networks.

- The glue that holds the whole Internet together is the network layer protocol, **IP** (**Internet Protocol**). Unlike most older network layer protocols, it was designed from the beginning with internetworking in mind.
- The network layer job is to provide a best-efforts (i.e., not guaranteed) way to transport datagram from source to destination, without regard to whether these machines are on the same network or whether there are other networks in between them.
- The transport layer takes data streams and breaks them up into datagram's.
- Each datagram is transmitted through the Internet, possibly being fragmented into smaller units as it goes.
- When all the pieces finally get to the destination machine, they are reassembled by the network layer into the original datagram.
- This datagram is then handed to the transport layer, which inserts it into the receiving process' input stream.

<u>THE IP PROTOCOL</u>

- An IP datagram consists of a header part and a text part. The header has a 20-byte fixed part and a variable length optional part.

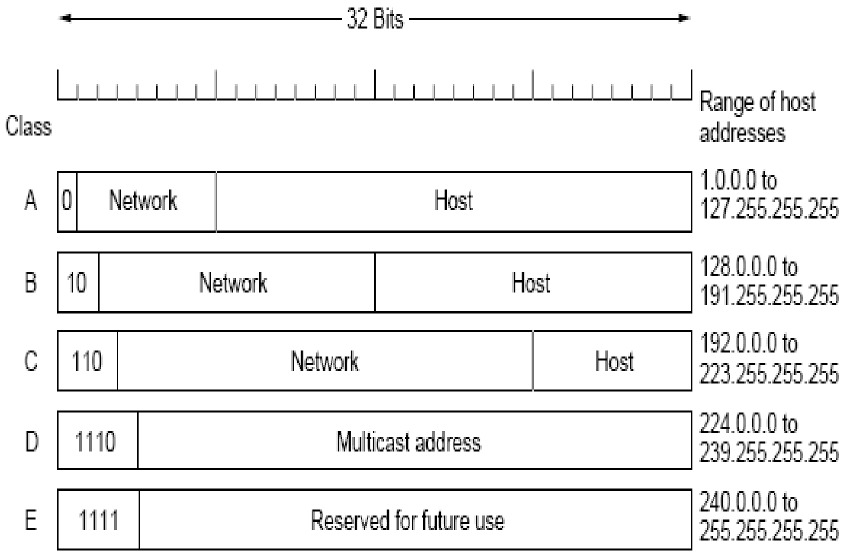


- The ***Version*** field keeps track of which version of the protocol the datagram belongs to.
- The header length is not constant, a field in the header, ***IHL***, is provided to tell how long the header is, in 32-bit words.
- The ***Type of service*** field is one of the few fields that have changed its meaning (slightly) over the years. It was and is still intended to distinguish between different classes of service.
- The ***Total length*** includes everything in the datagram—both header and data. The maximum length is 65,535 bytes.

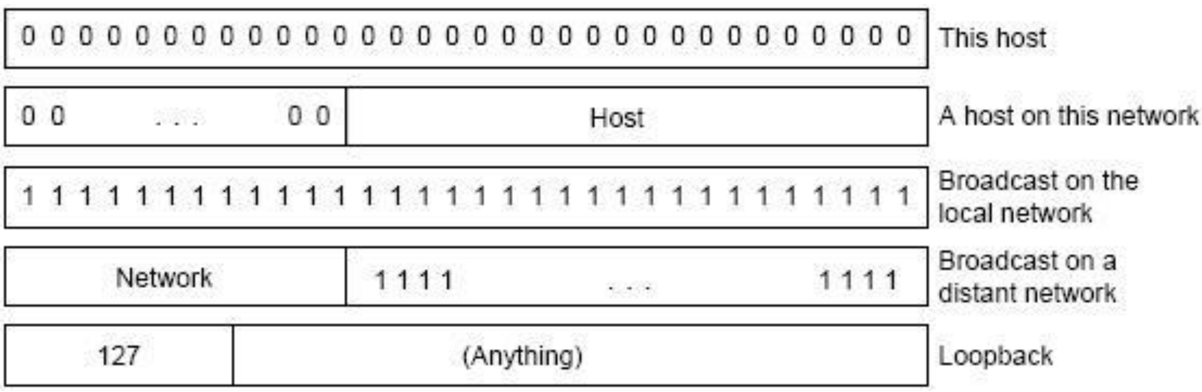
- The ***Identification*** field is needed to allow the destination host to determine which datagram a newly arrived fragment belongs to. All the fragments of a datagram contain the same *Identification* value. Next comes an unused bit and then two 1-bit fields.
- ***DF*** stands for Don't Fragment. It is an order to the routers not to fragment the datagram because the destination is incapable of putting the pieces back together again.
- ***MF*** stands for More Fragments. All fragments except the last one have this bit set. It is needed to know when all fragments of a datagram have arrived.
- The ***Fragment offset*** tells where in the current datagram this fragment belongs. All fragments except the last one in a datagram must be a multiple of 8 bytes, the elementary fragment unit. Since 13 bits are provided, there is a maximum of 8192 fragments per datagram, giving a maximum datagram length of 65,536 bytes, one more than the ***Total length*** **field.**
- The ***Time to live*** field is a counter used to limit packet lifetimes. It is supposed to count time in seconds, allowing a maximum lifetime of 255 sec.
- When the network layer has assembled a complete datagram, it needs to know what to do with it. The ***Protocol*** field tells it which transport process to give it to. TCP is one possibility, but so are UDP and some others. The numbering of protocols is global across the entire Internet.
- The ***Header checksum*** verifies the header only.
- The ***Source address*** **and** ***Destination address*** indicate the network number and host number.
- The ***Options*** field was designed to provide an escape to allow subsequent versions of the protocol to include information not present in the original design, to permit experimenters to try out new ideas, and to avoid allocating header bits to information that is rarely needed.
- The options are variable length.

<u>IP ADDRESS</u>

- Every host and router on the Internet has an IP address, which encodes its network number and host number.
- All IP addresses are 32 bits long. It is important to note that an IP address does not actually refer to a host. It really refers to a network interface, so if a host is on two networks, it must have two IP addresses.
- IP addresses were divided into the five categories

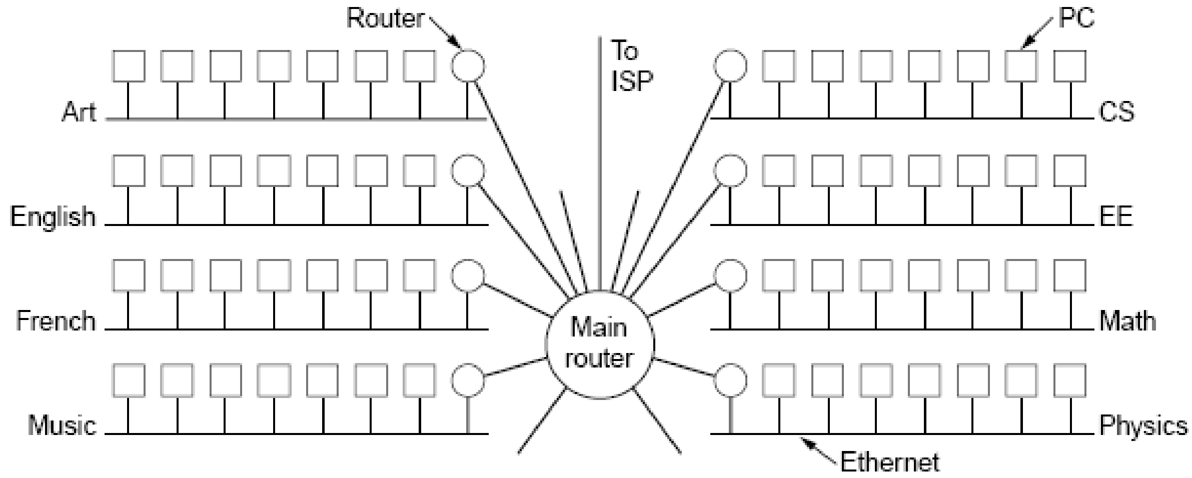


- The values 0 and -1 (all 1s) have special meanings. The value 0 means this network or this host. The value of -1 is used as a broadcast address to mean all hosts on the indicated network.



SUBNET

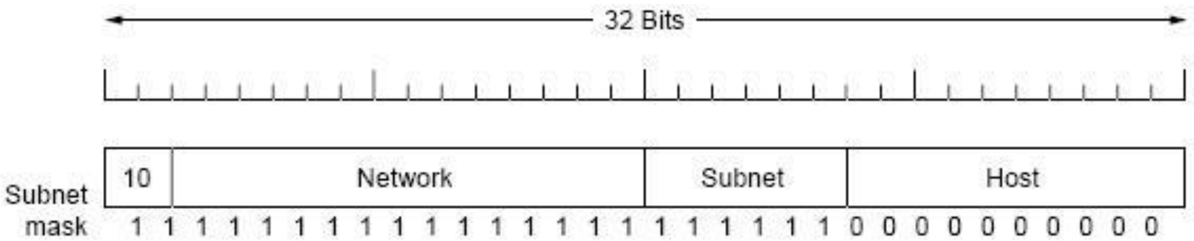
- All the hosts in a network must have the same network number. This property of IP addressing can cause problems as networks grow. For example……
- The problem is the rule that a single class A, B, or C address refers to one network, not to a collection of LANs.
- The solution is to allow a network to be split into several parts for internal use but still act like a single network to the outside world.



- To implement subnetting, the main router needs a subnet mask that indicates the split between network + subnet number and host.

- For example, if the university has a B address(130.50.0.0) and 35 departments, it could use a 6-bit subnet number and a 10-bit host number, allowing for up to 64 Ethernets, each with a maximum of 1022 hosts.

- The subnet mask can be written as 255.255.252.0. An alternative notation is /22 to indicate that the subnet mask is 22 bits long.



OSPF—THE INTERIOR GATEWAY ROUTING PROTOCOL

- The Internet is made up of a large number of autonomous systems. Each AS is operated by a different organization and can use its own routing algorithm inside.
- For example, the internal networks of companies *X*, *Y*, and *Z* are usually seen as three ASes if all three are on the Internet.
- All three may use different routing algorithms internally. Nevertheless, having standards, even for internal routing, simplifies the implementation at the boundaries between ASes and allows reuse of code.
- In this section we will study routing within an AS. In the next one, we will look at routing between ASes.
- A routing algorithm within an AS is called an **interior gateway protocol**; an algorithm for routing between ASes is called an **exterior gateway protocol**.
- The original Internet interior gateway protocol was a distance vector protocol (RIP) based on the Bellman-Ford algorithm inherited from the ARPANET.

- In 1988, the Internet Engineering Task Force began work on a successor. That successor, called **OSPF** (**Open Shortest Path First**), became a standard in 1990.
- Most router vendors now support it, and it has become the main interior gateway protocol. Below we will give a sketch of how OSPF works. For the complete story, see RFC 2328.
- Given the long experience with other routing protocols, the group designing the new protocol had a long list of requirements that had to be met. First, the algorithm had to be published in the open literature, hence the "O" in OSPF. A proprietary solution owned by one company would not do. Second, the new protocol had to support a variety of distance metrics, including physical distance, delay, and so on. Third, it had to be a dynamic algorithm, one that adapted to changes in the topology automatically and quickly.

- Fourth, and new for OSPF, it had to support routing based on type of service. The new protocol had to be able to route real-time traffic one way and other traffic a different way. The IP protocol has a *Type of Service* field, but no existing routing protocol used it. This field was included in OSPF but still nobody used it, and it was eventually removed.
- Fifth, and related to the above, the new protocol had to do load balancing, splitting the load over multiple lines. Most previous protocols sent all packets over the best route. The second-best route was not used at all. In many cases, splitting the load over multiple lines gives better performance.
- Sixth, support for hierarchical systems was needed. By 1988, the Internet had grown so large that no router could be expected to know the entire topology. The new routing protocol had to be designed so that no router would have to.

- Seventh, some modicum of security was required to prevent fun-loving students from spoofing routers by sending them false routing information. Finally, provision was needed for dealing with routers that were connected to the Internet via a tunnel. Previous protocols did not handle this well.

OSPF supports three kinds of connections and networks:

1. Point-to-point lines between exactly two routers.
2. Multiaccess networks with broadcasting (e.g., most LANs).
3. Multiaccess networks without broadcasting (e.g., most packet-switched WANs).

- A **multiaccess** network is one that can have multiple routers on it, each of which can directly communicate with all the others. All LANs and WANs have this property. Figure 5-64(a) shows an AS containing all three kinds of networks. Note that hosts do not generally play a role in OSPF.

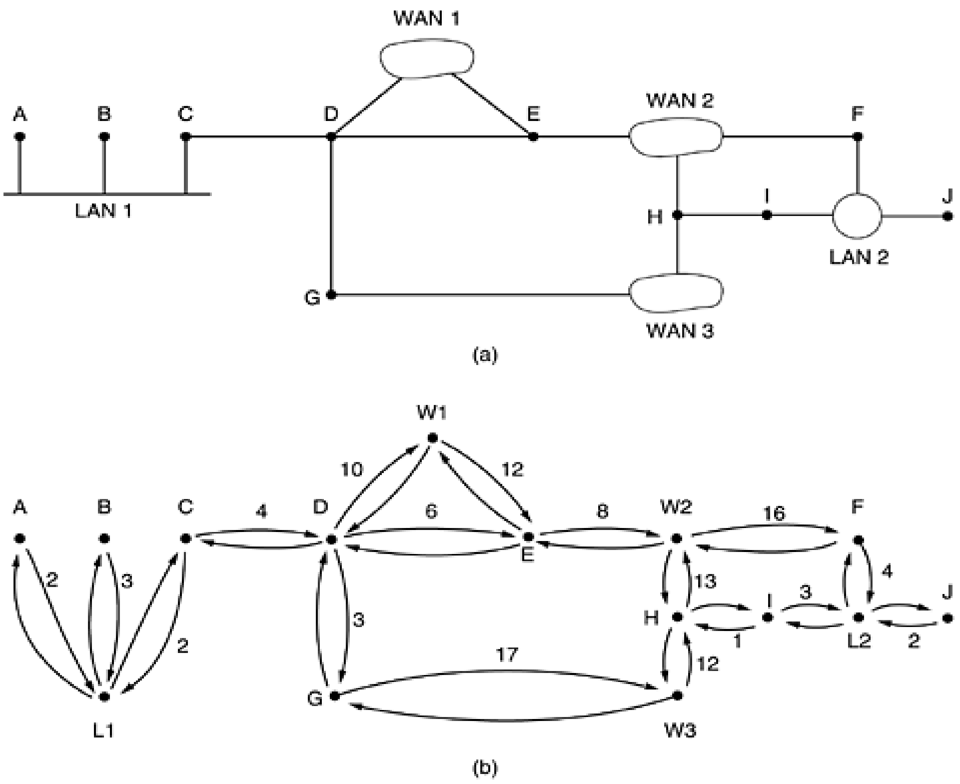


Figure 5-64. (a) An autonomous system. (b) A graph representation of (a).

- OSPF operates by abstracting the collection of actual networks, routers, and lines into a directed graph in which each arc is assigned a cost (distance, delay, etc.). It then computes the shortest path based on the weights on the arcs. A serial connection between two routers is represented by a pair of arcs, one in each direction. Their weights may be different. A multiaccess network is represented by a node for the network itself plus a node for each router. The arcs from the network node to the routers have weight 0 and are omitted from the graph.
- Figure 5-64(b) shows the graph representation of the network of Fig. 5-64(a). Weights are symmetric, unless marked otherwise. What OSPF fundamentally does is represent the actual network as a graph like this and then compute the shortest path from every router to every other router.

- Many of the ASes in the Internet are themselves large and nontrivial to manage. OSPF allows them to be divided into numbered **areas**, where an area is a network or a set of contiguous networks. Areas do not overlap but need not be exhaustive, that is, some routers may belong to no area. An area is a generalization of a subnet. Outside an area, its topology and details are not visible.
- Every AS has a **backbone** area, called area 0. All areas are connected to the backbone, possibly by tunnels, so it is possible to go from any area in the AS to any other area in the AS via the backbone. A tunnel is represented in the graph as an arc and has a cost. Each router that is connected to two or more areas is part of the backbone. As with other areas, the topology of the backbone is not visible outside the backbone.
- Within an area, each router has the same link state database and runs the same shortest path algorithm. Its main job is to calculate the shortest path from itself to every other

router in the area, including the router that is connected to the backbone, of which there must be at least one. A router that connects to two areas needs the databases for both areas and must run the shortest path algorithm for each one separately.

- During normal operation, three kinds of routes may be needed: intra-area, interarea, and inter-AS. Intra-area routes are the easiest, since the source router already knows the shortest path to the destination router. Interarea routing always proceeds in three steps: go from the source to the backbone; go across the backbone to the destination area; go to the destination. This algorithm forces a star configuration on OSPF with the backbone being the hub and the other areas being spokes. Packets are routed from source to destination "as is." They are not encapsulated or tunneled, unless going to an area whose only connection to the backbone is a tunnel. Figure 5-65 shows part of the Internet with ASes and areas.

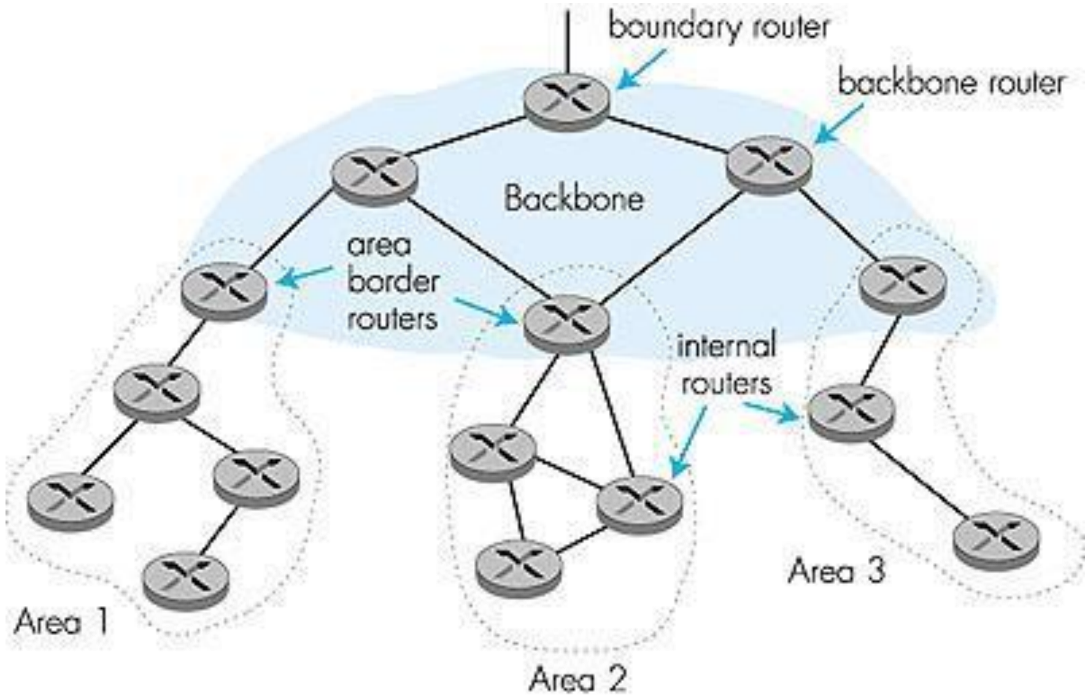


Figure 5-65. The relation between ASes, backbones, and areas in OSPF

OSPF distinguishes four classes of routers:

1. Internal routers are wholly within one area.
2. Area border routers connect two or more areas.
3. Backbone routers are on the backbone.
4. AS boundary routers talk to routers in other ASes.

- These classes are allowed to overlap. For example, all the border routers are automatically part of the backbone. In addition, a router that is in the backbone but not part of any other area is also an internal router. Examples of all four classes of routers are illustrated in Fig. 5-65.
- When a router boots, it sends HELLO messages on all of its point-to-point lines and multicasts them on LANs to the group consisting of all the other routers. On WANs, it

needs some configuration information to know who to contact. From the responses, each router learns who its neighbors are. Routers on the same LAN are all neighbors.

- OSPF works by exchanging information between adjacent routers, which is not the same as between neighboring routers. In particular, it is inefficient to have every router on a LAN talk to every other router on the LAN. To avoid this situation, one router is elected as the **designated router**. It is said to be **adjacent** to all the other routers on its LAN, and exchanges information with them. Neighboring routers that are not adjacent do not exchange information with each other. A backup designated router is always kept up to date to ease the transition should the primary designated router crash and need to replaced immediately.

- During normal operation, each router periodically floods LINK STATE UPDATE messages to each of its adjacent routers. This message gives its state and provides the costs used in the topological database. The flooding messages are acknowledged, to make them reliable. Each message has a sequence number, so a router can see whether an incoming LINK STATE UPDATE is older or newer than what it currently has. Routers also send these messages when a line goes up or down or its cost changes.

- DATABASE DESCRIPTION messages give the sequence numbers of all the link state entries currently held by the sender. By comparing its own values with those of the sender, the receiver can determine who has the most recent values. These messages are used when a line is brought up.

- Either partner can request link state information from the other one by using LINK STATE REQUEST messages. The result of this algorithm is that each pair of adjacent routers checks to see who has the most recent data, and new information is spread throughout the area this way. All these messages are sent as raw IP packets. The five kinds of messages are summarized in Fig. 5-66.

Message type	Description
Hello	Used to discover who the neighbors are
Link state update	Provides the sender's costs to its neighbors
Link state ack	Acknowledges link state update
Database description	Announces which updates the sender has
Link state request	Requests information from the partner

Figure 5-66. The five types of OSPF messages.