UNIT III

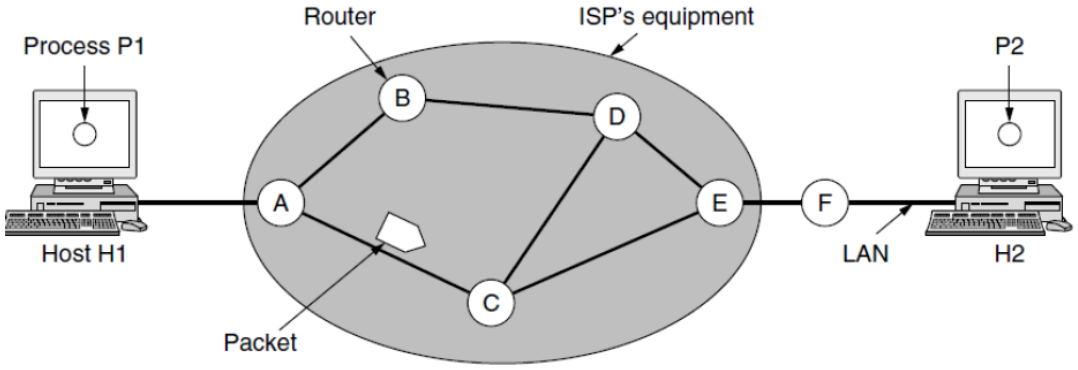
The Network Layer: Network layer design issues, Routing algorithms, Congestion control algorithms, Quality of service, Internetworking, The network layer in the Internet: IPV4 Addresses,IPV6, Internet Control protocol, OSPF, BGP, IP, ICMPv4, IGMP.

NETWORK LAYER DESIGN ISSUES

In the following sections, we will give 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 network.

Store-and-Forward Packet Switching

Before starting to explain the details of the network layer, it is worth restating the context in which the network layer protocols operate. This context can be seen in. The major components of the network are the ISP's equipment (routers connected by transmission lines), shown inside the shaded oval, and the customers' equipment, shown outside the oval. Host *H1* is directly connected to one of the ISP's routers, *A*, perhaps as a home computer that is plugged into a DSL modem. In contrast, *H2* is on a LAN, which might be an office Ethernet, with a router, *F*, owned and operated by the customer. This router has a leased line to the ISP's equipment. We have shown *F* as being outside the oval because it does not belong to the ISP. For the purposes of this chapter, however, routers on customer premises are considered part of the ISP network because they run the same algorithms as the ISP's routers (and our main concern here is algorithms).



The environment of the network layer protocols.

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

Services Provided to the Transport Layer

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

1. The services should be independent of the router technology.
2. The transport layer should be shielded from the number, type, and topology of the routers present.

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

Given these goals, the designers of the network layer have a lot of freedom in writing detailed specifications of the services to be offered to the transport layer.

This freedom often degenerates into a raging battle between two warring factions. The discussion centers on whether the network layer should provide connection-oriented service or connectionless service.

One camp (represented by the Internet community) argues that the routers' job is moving packets around and nothing else. In this view (based on 40 years of experience with a real computer network), the network is inherently unreliable, no matter how it is designed. Therefore, the hosts should accept this fact and do error control (i.e., error detection and correction) and flow control themselves. This viewpoint leads 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. This reasoning is an example of the **end-to-end argument**, a design principle that has been very influential in shaping the Internet (Saltzer et al., 1984). 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 network should provide a reliable, connection-oriented service. They claim that 100 years of successful experience with the worldwide telephone system is an excellent guide. In this view, quality of service is the dominant factor, and without connections in the network, quality of service is very difficult to achieve, especially for real-time traffic such as voice and video. Even after several decades, this controversy is still very much alive. Early, widely used data networks, such as X.25 in the 1970s and its successor Frame Relay in the 1980s, were connection-oriented. However, since the days of the ARPANET and the early Internet, connectionless network layers have grown tremendously in popularity. The IP protocol is now an ever-present symbol of success. It was undeterred by a connection-oriented technology called ATM that was developed to overthrow it in the 1980s; instead, it is ATM that is now found in niche uses and IP that is taking over telephone networks. Under the covers, however, the Internet is evolving connection-oriented features as quality of service becomes more important. Two examples of connection-oriented technologies are MPLS (Multi Protocol Label Switching and VLANs, which we saw in. Both technologies are widely used.

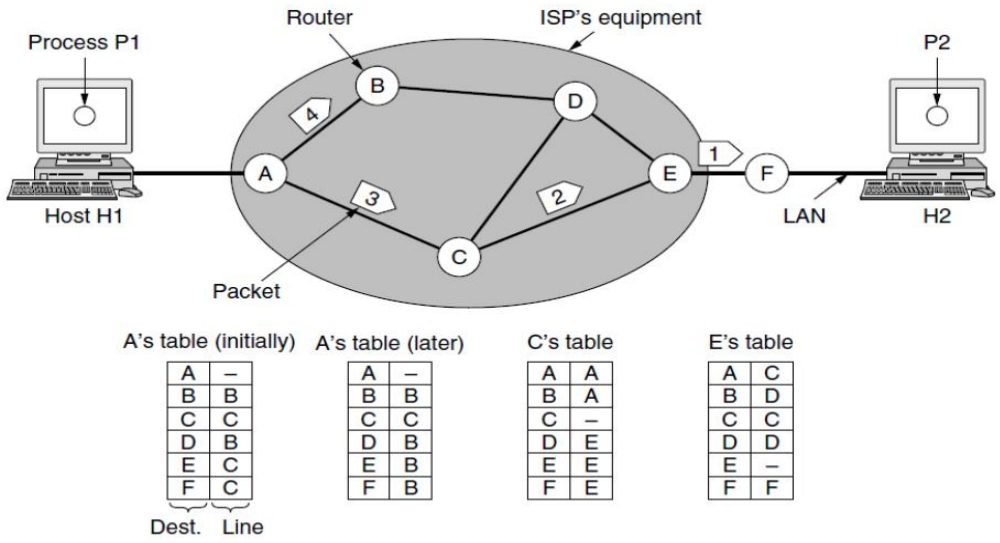
Implementation of Connectionless Service

Having looked at the two classes of service the network layer can provide to its users, it is time to see how this layer works inside. Two different organizations are possible, depending on the type of service offered. If connectionless service is offered, packets are injected into the network individually and routed independently of each other. No advance setup is needed. In this context, the packets are frequently called **datagrams** (in analogy with telegrams) and the network is called a **datagram network**. If connection-oriented service is used, a path from the source router all the way to the destination router must be established before any data packets can be sent. This connection is called a **VC** (**virtual circuit**), in analogy with the physical circuits set up by the

telephone system, and the network is called a **virtual-circuit network**. In this section, we will examine datagram networks; in the next one, we will examine virtual-circuit networks.

Let us now see how a datagram network works. Suppose that the process *P1* in Fig. 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.

Routing within a datagram network.



Routing within a datagram network.

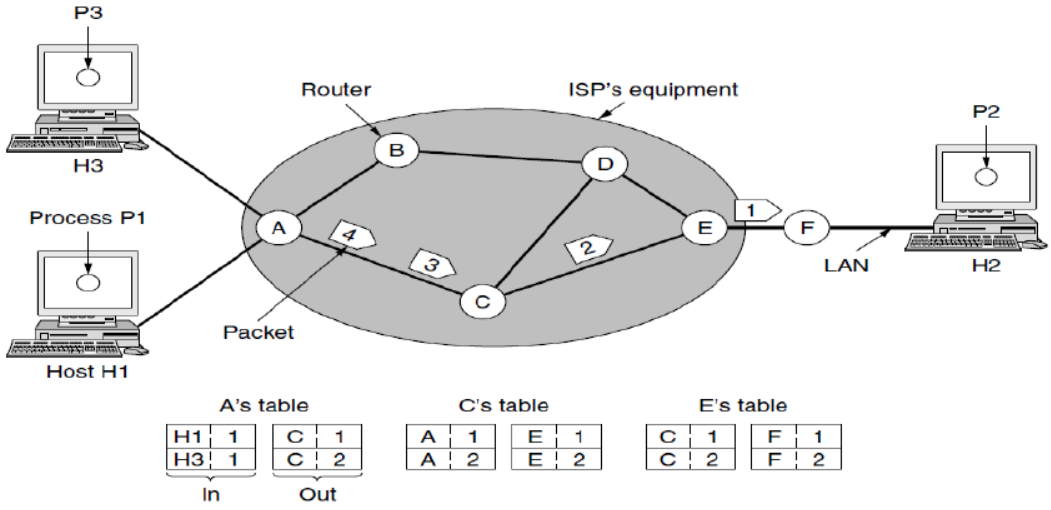
Let us assume for this example that the message is four times longer than the maximum packet size, so the network layer has to break it into four packets, 1, 2,3, and 4, and send each of them in turn to router *A* using some point-to-point protocol, for example, PPP. At this point the ISP takes over. Every router has an internal table telling it where to send packets for each of the possible destinations.

Each table entry is a pair consisting of a destination and the outgoing line to use for that destination. Only directly connected lines can be used. For example, in Fig., *A* has only two outgoing lines—to *B* and to *C*—so every incoming packet must be sent to one of these routers, even if the ultimate destination is to some other router. *A*'s initial routing table is shown in the figure under the label ''initially.'' At *A*, packets 1, 2, and 3 are stored briefly, having arrived on the incoming link and had their checksums verified. Then each packet is forwarded according to *A*'s table, onto the outgoing link to *C* within a new frame. Packet 1 is then forwarded to *E* and then to *F*. When it gets to *F*, it is sent within a frame over the LAN to *H2*. Packets 2 and 3 follow the same route. However, something different happens to packet 4. When it gets to *A* it is sent to router *B*, even though it is also destined for *F*. For some reason, *A* decided to send packet 4 via a different route than that of the first three packets. Perhaps it has learned of a traffic jam somewhere along the *ACE* path and updated its routing table, as shown under the label ''later.'' The algorithm that manages the tables and makes the routing decisions is called the **routing algorithm.** Routing algorithms are one of the main topics we will study in this chapter. There are several different kinds of them, as we will see. IP (Internet Protocol), which is the basis for the entire Internet, is the dominant example of a connectionless network service. Each packet carries

a destination IP address that routers use to individually forward each packet. The addresses are 32 bits in IPv4 packets and 128 bits in IPv6 packets.

Implementation of Connection-Oriented Service

For connection-oriented service, we need a virtual-circuit network. 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. Instead, when a connection is established, a route from the source machine to the destination machine is chosen as part of the connection setup and stored in tables inside the routers. That route is used for all traffic flowing over the connection, exactly the same way that the telephone system works. When the connection is released, the virtual circuit is also terminated. With connection-oriented service, each packet carries an identifier telling which virtual circuit it belongs to. As an example, consider the situation shown in Fig. Here, host *H1* has established connection 1 with host *H2*. This connection is remembered as the first entry in each of the routing tables. The first line of *A*'s table says that if a packet bearing connection identifier 1 comes in from *H1*, it is to be sent to router *C* and given connection identifier 1. Similarly, the first entry at *C* routes the packet to *E*, also with connection identifier 1.



Routing within a virtual-circuit network.

Now let us consider what happens if *H3* also wants to establish a connection to *H2*. It chooses connection identifier 1 (because it is initiating the connection and this is its only connection) and tells the network to establish the virtual circuit. This leads to the second row in the tables. Note that we have a conflict here because although *A* can easily distinguish connection 1 packets from *H1* 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 process is called **label switching**. An example of a connection-oriented network service is **MPLS** (**Multi Protocol Label Switching**). It is used within ISP networks in the Internet, with IP packets wrapped in an MPLS header having a 20-bit connection identifier or label. MPLS is often hidden from customers, with the ISP establishing long-term connections for large amounts of traffic, but it is increasingly being used to help when quality of service is important but also with other ISP traffic management tasks.

Comparison of Virtual-Circuit and Datagram Networks

Both virtual circuits and datagrams have their supporters and their detractors. We will now attempt to summarize both sets of arguments. The major issues are listed in Fig, although purists could probably find a counterexample for everything in the figure.

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

Comparison of datagram and virtual-circuit networks.

Inside the network, several trade-offs exist between virtual circuits and data grams. One trade-off is setup time versus address parsing time. Using virtual circuits requires a setup phase, which takes time and consumes resources. However, once this price is paid, figuring out what to do with a data packet in a virtual-circuit network is easy: the router just uses the circuit number to index into a table to find out where the packet goes. In a datagram network, no setup is needed but a more complicated lookup procedure is required to locate the entry for the destination. A related issue is that the destination addresses used in datagram networks are longer than circuit numbers used in virtual-circuit networks because they have a global meaning. If the packets tend to be fairly short, including a full destination address in every packet may represent a significant amount of overhead , and hence a waste of bandwidth. Yet another issue is the amount of table space required in router memory. A datagram network needs to have an entry for every possible destination, whereas a virtual-circuit network just needs an entry for each virtual circuit. However, this advantage is somewhat illusory since connection setup packets have to be routed too, and they use destination addresses, the same as datagrams do. Virtual circuits have some advantages in guaranteeing quality of service and avoiding congestion within the network because resources (e.g., buffers, bandwidth, and CPU cycles) can be reserved in advance, when the connection is established. Once the packets start arriving, the necessary bandwidth and router capacity will be there. With a datagram network, congestion avoidance is more difficult. For transaction processing systems (e.g., stores calling up to verify credit card purchases), the overhead required to set up and clear a virtual circuit may easily dwarf the use of the circuit. If the majority of the traffic is expected to be of this kind, the use of virtual circuits inside the network makes little sense. On the other hand, for long-running uses such as VPN traffic between two corporate offices, permanent virtual circuits (that are set up manually and last for

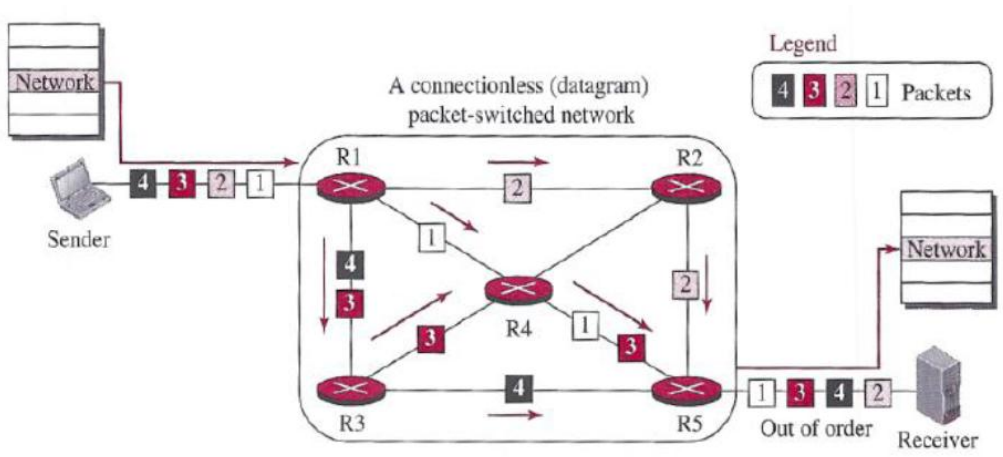
months or years) may be useful. Virtual circuits also have a vulnerability problem. If a router crashes and loses its memory, even if it comes back up a second later, all the virtual circuits passing through it will have to be aborted. In contrast, if a datagram router goes down, only those users whose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

PACKET SWITCHING

From the discussion of routing and forwarding in the previous section, we infer that a kind of *switching* occurs at the network layer. A router, in fact, is a switch that creates a connection between an input port and an output port (or a set of output ports), just as an electrical switch connects the input to the output to let electricity flow. Although in data communication switching techniques are divided into two broad categories, circuit switching and packet switching, only packet switching is used at the network layer because the unit of data at this layer is a packet. Circuit switching is mostly used lat the physical layer; the electrical switch mentioned earlier is a kind of circuit switch. We discussed circuit switching in Chapter 8; we discuss packet switching. At the Network layer, a message from the upper layer is divided into manageable packets and each packet is sent through the network. The source of the message sends the packets one by one; the destination of the message receives the packets one by one. The destination waits for all packets belonging to the same message to arrive before delivering the message to the upper layer. The connecting devices in a packet-switched network still need to decide how to route the packets to the final destination. Today, a packet-switched network can use two different approaches to route the packets: the *datagram approach* and the *virtual circuit approach.* We discuss both approaches in the next section.

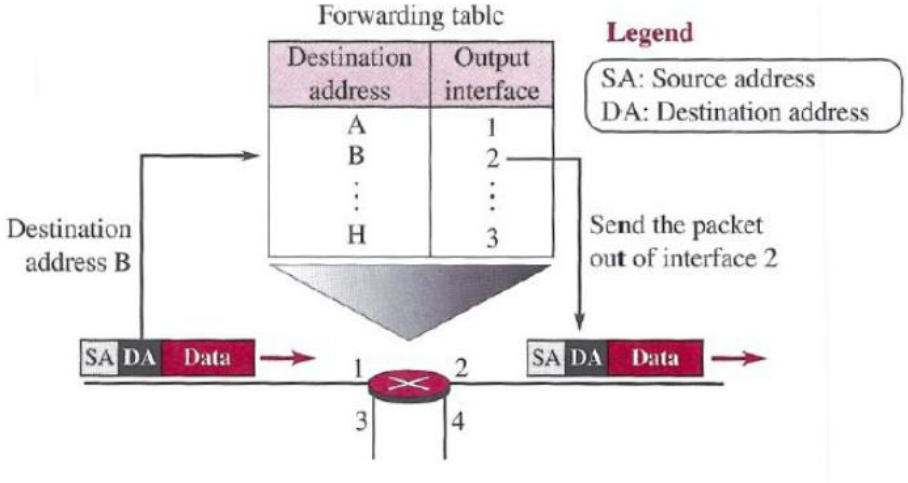
Datagram Approach: Connectionless Service

When the Internet started, to make it simple, the network layer was designed to provide a connectionless service in which the network-layer protocol treats each packet independently, with each packet having no relationship to any other packet. The idea was that the network layer is only responsible for delivery of packets from the source to the destination. Id this approach, the packets in a message mayor may not travel the same path to their destination. Figure shows the idea. When the network layer provides a connectionless service, each packet traveling in the Internet is an independent entity; there is no relationship between packets belonging to the same message. The switches in this type of network are called *routers.* A packet belonging to a message may be followed by a packet belonging to the same message or to a different message. A packet may be followed by a packet coming from the same or from a different source.



A connectionless packet-switched network

Each packet is routed based on the information contained in its header: source and destination addresses. The destination address defines where it should go; the source address defines where it comes from. The router in this case routes the packet based only on the destination address. The source address may be used to send an error message to the source if the packet is discarded. Figure 18.4 shows the forwarding process in a router in this case. We have used symbolic addresses such as A and B.



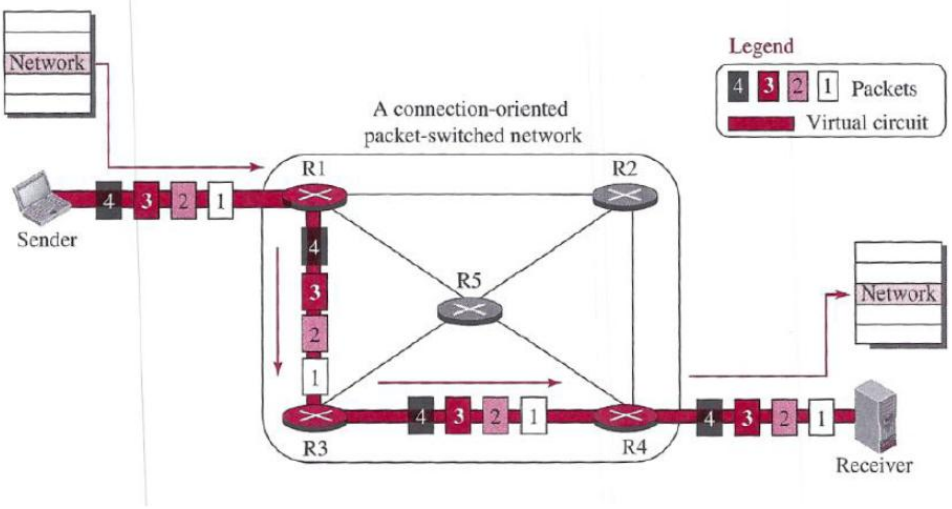
Forwarding process in a muter when used in a connection less network

In the datagram approach, the forwarding decision is based on the destination address of the packet.

Virtual-Circuit Approach: Connection-Oriented Service

In a connection-oriented service (also called *virtual-circuit approach),* there is a relationship between all packets belonging to a message. Before all datagrams in a message can be sent, a virtual connection should be set up to define the path for the datagrams. After connection setup, the datagrams can all follow the same path. In this type of service, not only must the packet contain the source and destination addresses, it must also contain a flow label, a virtual circuit identifier that defines the virtual path the packet should follow. Shortly, we will show how this flow label is determined, but for the moment, we assume that the packet carries this label. Although it looks as though the use of the label may make the source and destination addresses unnecessary during the data transfer phase, parts of the Internet at the network layer still keep

these addresses. One reason is that part of the packet path may still be using the connectionless service. Another reason is that the protocol at the network layer is designed with these addresses, and it may take a while before they can be changed. Figure shows the concept of connection-oriented service.

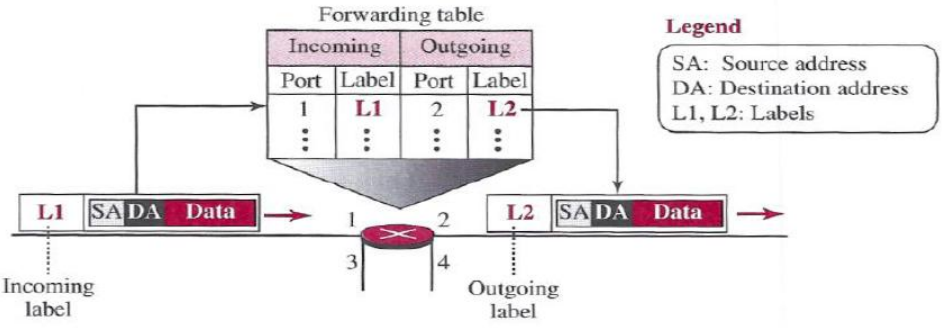


A virtual-circuit packet-switched network

Each packet is forwarded based on the label in the packet. To follow the idea of connection-oriented design to be used in the Internet, we assume that the packet has a label when it reaches the router. Figure 18.6 shows the idea. In this case, the forwarding decision is based on the value of the label, or *virtual circuit identifier,* as it is sometimes called. To create a connection-oriented service, a three-phase process is used: setup, data transfer, and teardown. In the setup phase, the source and destination addresses of the sender and receiver are used to make table entries for the connection-oriented service. In the teardown phase, the source and destination inform the router to delete the corresponding entries. Data transfer occurs between these two phases.

Setup Phase

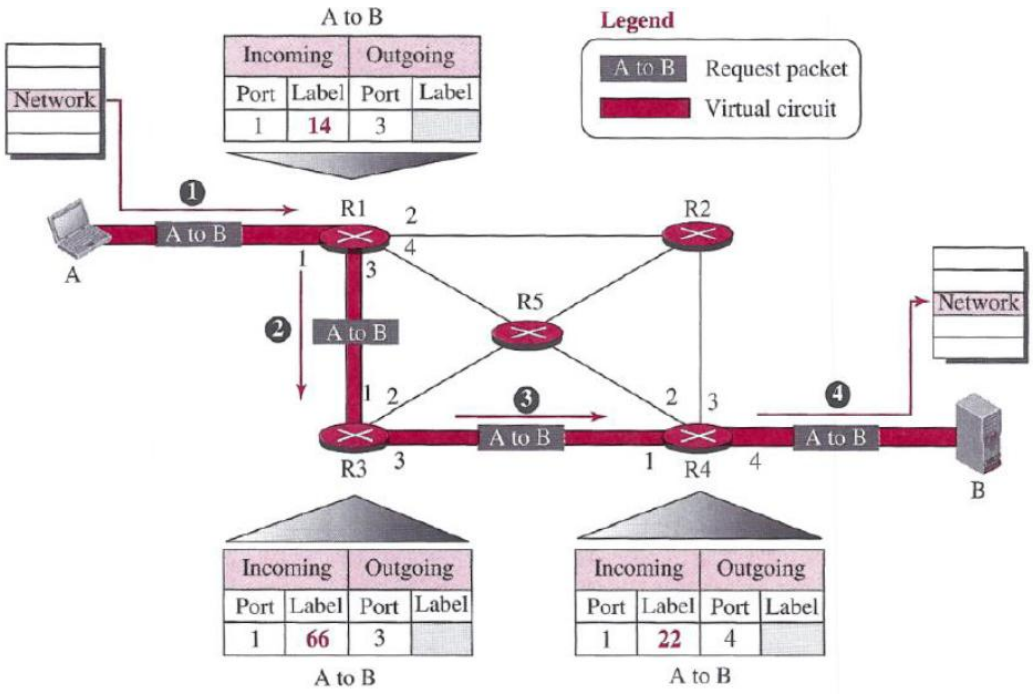
In the setup phase, a router creates an entry for a virtual circuit. For example, suppose source .A needs to create a virtual circuit to destination B. Two auxiliary packets need to be exchanged between the sender and the receiver: the request packet and the acknowledgment packet.



Forwarding process in a router when used in a virtual-circuit network

Request packet

A request packet is sent from the source to the destination. This auxiliary packet carries the source and destination addresses. Figure shows the process.



Sending request packet in a virtual-circuit network

1. Source A sends a request packet to router Rl.

2. Router Rl receives the request packet. It knows that a packet going from A to B goes 0rt through port 3. How the router has obtained this information is a point covered later. For the

moment, assume that it knows the output port. The router creates an entry in its table for this virtual circuit, but it is only able to fill three of the four columns. The router assigns the incoming port (1) and chooses an available in rooming label (14) and the outgoing port (3). It does not yet know the outgoing label, which will be found during the acknowledgment step. The router then forwards the packet through port 3 to router R3.

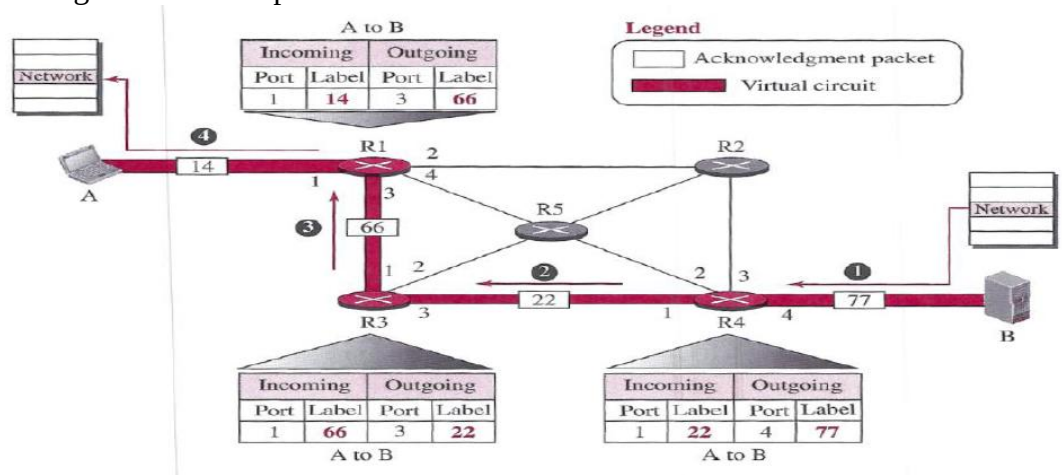
3. Router iR3 receives the setup request packet. The same events happen here as at router Rl; three columns of the table are completed: in this case, incoming port (1)incoming label (66), and outgoing port (3).

4. Router R4 receives the setup request packet. Again, three columns are completed: incoming port (1), incoming label (22), and outgoing port (4).

5. Destination B receives the setup packet, and if it is ready to receive packets from A, it assigns a label to the incoming packets that come from A, in this case 77, as shown in Figure. This label lets the destination know that the packets come from A,I and not from other sources.

Acknowledgment Packet

A special Packet, called the acknowledgment packet, completes the entries in the switching tables. Figure shows the process.



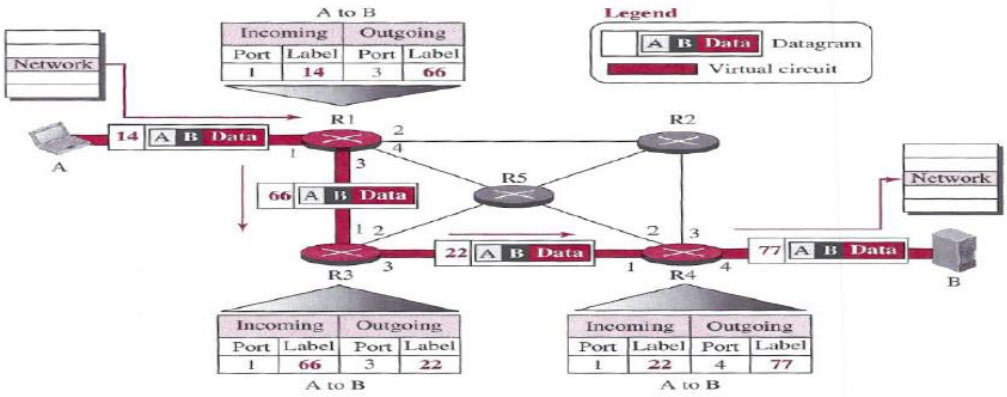
Sending acknowledgments in a virtual-circuit network

1. The destination sends an acknowledgment to router R4. The acknowledgment carries the global source and destination addresses so the router knows which entry in the table is to be completed. The packet also carries label 77, chosen by the destination as the incoming label for packets from A. Router R4 uses this label to complete the outgoing label column for this entry. Note that 77 is the incoming label for destination B, but the outgoing label for router R4.
2. Router R4 sends an acknowledgment to router R3 that contains its incoming label in the table, chosen in the setup phase. Router R3 uses this as the outgoing label in the table.
3. Router R3 sends an acknowledgment to router Rl that contains its incoming label in the table, chosen in the setup phase. Router RI uses this as the outgoing label in the table.
4. Finally router Rl sends an acknowledgment to source A that contains its incoming label in the table, chosen in the setup phase.
5. The source uses this as the outgoing label for the data packets to be sent to destination B.

Data- Transfer Phase

The second phase is called the data-transfer phase. After all routers have created their forwarding table for a specific virtual circuit, then the network-layer packets belonging to one message can be sent one after another. In Figure, we show the flow of a single packet, but the process is the

same for 1, 2, or 100 packets. The source computer uses the label 14, which it has received from router Rl in the setup.



Flow of one packet in an established virtual circuit

phase. Router Rl forwards the packet to router R3, but changes the label to 66.Router R3I forwards the packet to router R4, but changes the label to 22. Finally, router R4 delivers the packet to its final destination with the label 77. All the packets in the message follow the same sequence of labels, and the packets arrive in order at the destination. *Teardown Phase* In the teardown phase, source A, after sending all packets to B, sends a special packet called a teardown packet. Destination B responds with a confirmation packet. All routers delete the corresponding entries from their tables.

ROUTING ALGORITHMS

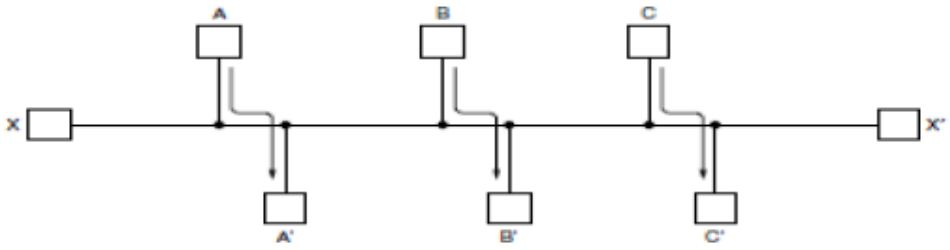
The main function of the network layer is routing packets from the source machine to the destination machine. In most networks, packets will require multiple hops to make the journey. The only notable exception is for broadcast networks, but even here routing is an issue if the source and destination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

The **routing algorithm** is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on. If the network uses datagrams internally, this decision must be made anew for every arriving data packet since the best route may have changed since last time. If the network uses virtual circuits internally, routing decisions are made only when a new virtual circuit is being set up. Thereafter, data packets just follow the already established route. The latter case is sometimes called **session routing** because a route remains in force for an entire session (e.g., while logged in over a VPN). It is sometimes useful to make a distinction between routing, which is making the decision which routes to use, and forwarding, which is what happens when a packet arrives. One can think of a router as having two processes inside it. One of them handles each packet as it arrives, looking up the outgoing line to use for it in the routing tables. This process is **forwarding**. The other process is responsible for filling in and updating the routing tables. That is where the routing algorithm comes into play.

Regardless of whether routes are chosen independently for each packet sent or only when new connections are established, certain properties are desirable in a routing algorithm: correctness, simplicity, robustness, stability, fairness, and efficiency. Correctness and simplicity hardly require comment, but the need for robustness may be less obvious at first. Once a major network comes on the air, it may be expected to run continuously for years without system-wide failures. During that period there will be hardware and software failures of all kinds. Hosts, routers, and

lines will fail repeatedly, and the topology will change many times. The routing algorithm should be able to cope with changes in the topology and traffic without requiring all jobs in all hosts to be aborted. Imagine the havoc if the network needed to be rebooted every time some router crashed! Stability is also an important goal for the routing algorithm. There exist routing algorithms that never converge to a fixed set of paths, no matter how long they run. A stable algorithm reaches equilibrium and stays there. It should converge quickly too, since communication may be disrupted until the routing algorithm has reached equilibrium. Fairness and efficiency may sound obvious—surely no reasonable person would oppose them—but as it turns out, they are often contradictory goals. As a simple example of this conflict, look at Fig. Suppose that there is enough traffic between *A* and *A′*, between *B* and *B′*, and between *C* and *C′* to saturate the horizontal links. To maximize the total flow, the *X* to *X′* traffic should be shut off altogether. Unfortunately, *X* and *X′* may not see it that way. Evidently, some compromise between global efficiency and fairness to individual connections is needed. Before we can even attempt to find trade-offs between fairness and efficiency, we must decide what it is we seek to optimize. Minimizing the mean packet delay is an obvious candidate to send traffic through the network effectively, but so is maximizing total network throughput. Furthermore, these two goals are also in conflict, since operating any queuing system near capacity implies a long queuing delay. As a compromise, many networks attempt to minimize the distance a packet must travel, or simply reduce the number of hops a packet must make. Either choice tends to improve the delay and also reduce the amount of bandwidth consumed per packet, which tends to improve the overall network throughput as well.

Routing algorithms can be grouped into two major classes: non adaptive and adaptive. **Non adaptive algorithms** do not base their routing decisions on any measurements or estimates of the current topology



Network with a conflict between fairness and efficiency.

and traffic. Instead, the choice of the route to use to get from *I* to *J* (for all *I* and *J*) is computed in advance, offline, and downloaded to the routers when the network is booted. This procedure is sometimes called **static routing**. Because it does not respond to failures, static routing is mostly useful for situations in which the routing choice is clear. For example, router *F* in Fig. should send packets headed into the network to router *E* regardless of the ultimate destination. **Adaptive algorithms**, in contrast, change their routing decisions to reflect changes in the topology, and sometimes changes in the traffic as well.

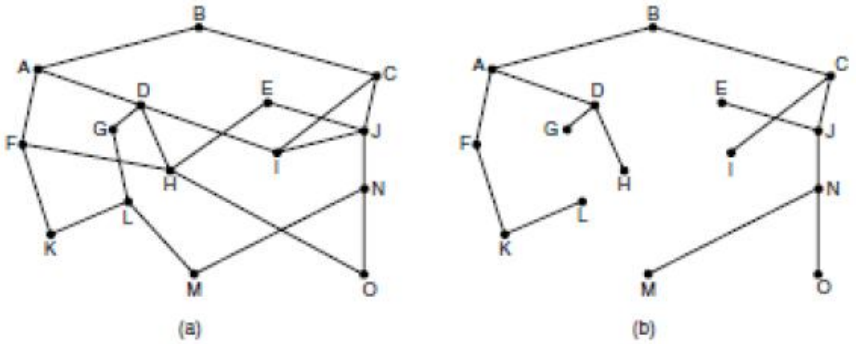
These **dynamic routing** algorithms differ in where they get their information (e.g., locally, from adjacent routers, or from all routers), when they change the routes(e.g., when the topology *T* seconds as the load changes), and what metric is used for optimization (e.g., distance, number of hops, or estimated transit time).In the following sections, we will discuss a variety of routing algorithms. The algorithms cover delivery models besides sending a packet from a source to a destination. Sometimes the goal is to send the packet to multiple, all, or one of a set of

destinations. All of the routing algorithms we describe here make decisions based on the topology; we defer the possibility of decisions based on the traffic levels to Sec.

The Optimality Principle

Before we get into specific algorithms, it may be helpful to note that one can make a general statement about optimal routes without regard to network topology or traffic. This statement is known as the **optimality principle** (Bellman, 1957). It states that if router *J* is on the optimal path from router *I* to router *K*, then the optimal path from *J* to *K* also falls along the same route. To see this, call the part of the route from *I* to *J r*1 and the rest of the route *r* 2. If a route better than *r* 2 existed from *J* to *K*, it could be concatenated with *r* 1 to improve the route from *I* to *K*, contradicting our statement that *r* 1*r* 2 is optimal. As a direct consequence of the optimality principle, we can see that the set of optimal routes from all sources to a given destination form a tree rooted at the destination. Such a tree is called a **sink tree** and is illustrated in Fig. where the distance metric is the number of hops. The goal of all routing algorithms is to discover and use the sink trees for all routers.

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



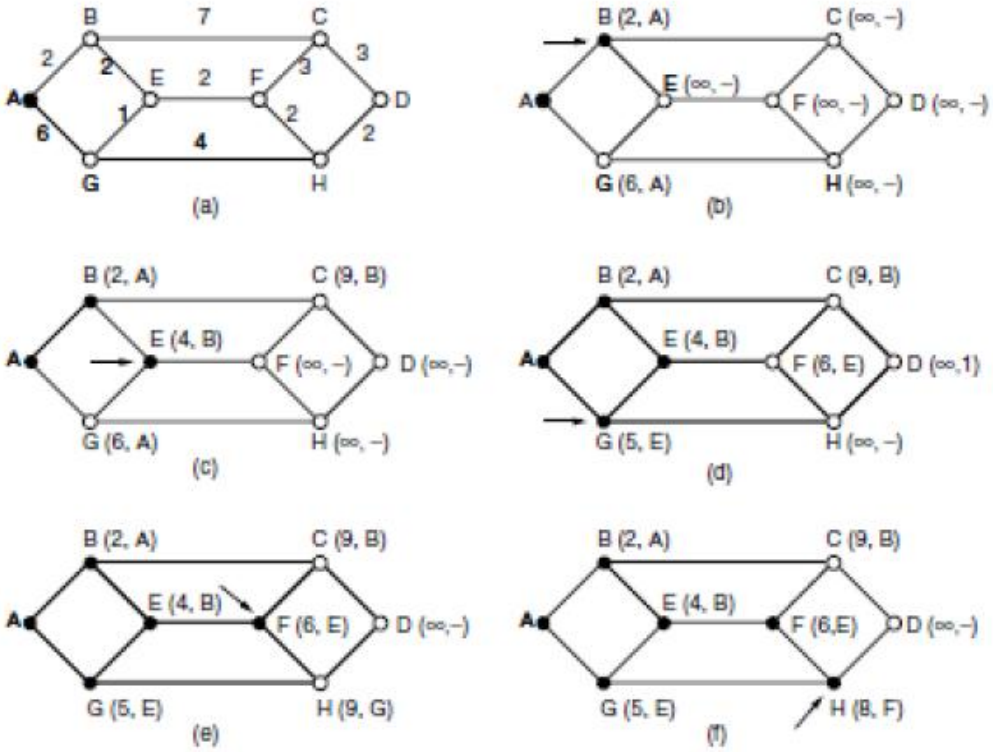
Note that a sink tree is not necessarily unique; other trees with the same path lengths may exist. If we allow all of the possible paths to be chosen, the tree becomes a more general structure called a **DAG** (**Directed Acyclic Graph**). DAGs have no loops. We will use sink trees as convenient shorthand for both cases. Both cases also depend on the technical assumption that the paths do not interfere with each other so, for example, a traffic jam on one path will not cause another path to divert. Since a sink tree is indeed a tree, it does not contain any loops, so each packet will be delivered within a finite and bounded number of hops. In practice, life is not quite this easy. Links and routers can go down and come back up during operation, so different routers may have different ideas about the current topology. Also, we have quietly finessed the issue of whether each router has to individually acquire the information on which to base its sink tree computation or whether this information is collected by some other means. We will come back to these issues shortly. Nevertheless, the optimality principle and the sink tree provide a benchmark against which other routing algorithms can be measured.

Shortest Path Algorithm

Let us begin our study of routing algorithms with a simple technique for computing optimal paths given a complete picture of the network. These paths are the ones that we want a distributed routing algorithm to find, even though not all routers may know all of the details of the network. The idea is to build a graph of the network, with each node of the graph representing a router and each edge of the graph representing a communication line, or link. To

choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph.

The concept of a **shortest path** deserves some explanation. One way of measuring path length is the number of hops. Using this metric, the paths *ABC* and *ABE* in Fig. are equally long. Another metric is the geographic distance in kilometers, in which case *ABC* is clearly much longer than *ABE* (assuming the figure is drawn to scale).



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

However, many other metrics besides hops and physical distance are also possible. For example, each edge could be labeled with the mean delay of a standard test packet, as measured by hourly runs. With this graph labeling, the shortest path is the fastest path rather than the path with the fewest edges or kilometers. In the general case, the labels on the edges could be computed as a function of the distance, bandwidth, average traffic, communication cost, measured delay, and other factors. By changing the weighting function, the algorithm would then compute the ‘‘shortest’’ path measured according to any one of a number of criteria or to a combination of criteria. Several algorithms for computing the shortest path between two nodes of a graph are known. This one is due to Dijkstra (1959) and finds the shortest paths between a source and all destinations in the network. Each node is labeled (in parentheses) with its distance from the source node along the best known path. The distances must be non-negative, as they will be if they are based on real quantities like bandwidth and delay. Initially, no paths are known, so all nodes are labeled with infinity. As the algorithm proceeds and paths are found, the labels may change, reflecting better paths. A label may be either tentative or permanent. Initially, all labels are tentative. When it is discovered that a label represents the shortest possible path from the source to that node, it is made permanent and never changed thereafter. To illustrate how the labeling algorithm works, look at the weighted, undirected graph of Fig., where the weights

represent, for example, distance. We want to find the shortest path from *A* to *D*. We start out by marking node *A* as permanent, indicated by a filled-in circle. Then we examine, in turn, each of the nodes adjacent to *A* (the working node), relabeling each one with the distance to *A*. Whenever a node is relabeled, we also label it with the node from which the probe was made so that we can reconstruct the final path later. If the network had more than one shortest path from *A* to *D* and we wanted to find all of them, we would need to remember all of the probe nodes that could reach a node with the same distance. Having examined each of the nodes adjacent to *A*, we examine all the tentatively labeled nodes in the whole graph and make the one with the smallest label permanent, as shown in Fig. (b). this one becomes the new working node. We now start at *B* and examine all nodes adjacent to it. If the sum of the label on *B* and the distance from *B* to the node being considered is less than the label on that node, we have a shorter path, so the node is relabeled. After all the nodes adjacent to the working node have been inspected and the tentative labels changed if possible, the entire graph is searched for the tentatively labeled node with the smallest value. This node is made permanent and becomes the working node for the next round. Figure shows the first six steps of the algorithm. To see why the algorithm works, look at Fig. (c). At this point we have just made *E* permanent. Suppose that there were a shorter path than *ABE*, say *AXYZE* (for some *X* and *Y*). There are two possibilities: either node *Z* has already been made permanent, or it has not been. If it has, then *E* has already been probed (on the round following the one when *Z* was made permanent), so the *AXYZE* path has not escaped our attention and thus cannot be a shorter path. Now consider the case where *Z* is still tentatively labeled. If the label at *Z* is greater than or equal to that at *E*, then *AXYZE* cannot be a shorter path than *ABE*. If the label is less than that of *E*, then *Z* and not *E* will become permanent first, allowing *E* to be probed from *Z*. This algorithm is given in Fig. 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., we compute the shortest path starting at the terminal node, *t*, rather than at the source node, *s*. Since the shortest paths from *t* to *s* in an undirected graph are the same as the shortest paths from *s* to *t*, it does not matter at which end we begin. The reason for searching backward is that each node is labeled with its predecessor rather than its successor. When the final path is copied into the output variable, *path*, the path is thus reversed. The two reversal effects cancel, and the answer is produced in the correct order.

Flooding

When a routing algorithm is implemented, each router must make decisions based on local knowledge, not the complete picture of the network. A simple local technique is **flooding**, in which every incoming packet is sent out on every outgoing line except the one it arrived on. Flooding obviously generates vast numbers of duplicate packets, in fact, an infinite number unless some measures are taken to damp the process. One such measure is to have a hop counter contained in the header of each packet that is decremented at each hop, with the packet being discarded when the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the network.

Flooding with a hop count can produce an exponential number of duplicate packets as the hop count grows and routers duplicate packets they have seen before. A better technique for damming the flood is to have routers keep track of which packets have been flooded, to avoid sending them out a second time. One way to achieve this goal is to have the source router put a sequence number in each packet it receives from its hosts. Each router then needs a list per

source router telling which sequence numbers originating at that source have already been seen. If an incoming packet is on the list, it is not flooded.

```
#define MAX NODES 1024 /* maximum number of nodes */
#define INFINITY 1000000000 /* a number larger than every maximum path */
int n, dist[MAX NODES][MAX NODES]; /* dist[i][j] is the distance from i to j */
void shortest path(int s, int t, int path[])
{ struct state { /* the path being worked on */
int predecessor; /* previous node */
int length; /* length from source to this node */
enum {permanent, tentative} label; /* label state */
} state[MAX NODES];
int i, k, min;
struct state *p;
for (p = &state[0]; p < &state[n]; p++) { /* initialize state */
p->predecessor = −1;
p->length = INFINITY;
p->label = tentative;
}
state[t].length = 0; state[t].label = permanent;
k = t; /* k is the initial working node */
do { /* Is there a better path from k? */
for (i = 0; i < n; i++) /* this graph has n nodes */
if (dist[k][i] != 0 && state[i].label == tentative) {
if (state[k].length + dist[k][i] < state[i].length) {
state[i].predecessor = k;
state[i].length = state[k].length + dist[k][i];
}
}
/* Find the tentatively labeled node with the smallest label. */
k = 0; min = INFINITY;
for (i = 0; i < n; i++)
if (state[i].label == tentative && state[i].length < min) {
min = state[i].length; k = i;
}
state[k].label = permanent;
} while (k != s);
/* Copy the path into the output array. */
i = 0; k = s;
do {path[i++] = k; k = state[k].predecessor; } while (k >= 0);
}
```

Dijkstra’s algorithm to compute the shortest path through a graph

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

ensures that a packet is delivered to every node in the network. This may be wasteful if there is a single destination that needs the packet, but it is effective for broadcasting information. In wireless networks, all messages transmitted by a station can be received by all other stations within its radio range, which is, in fact, flooding, and some algorithms utilize this property. Second, flooding is tremendously robust. Even if large numbers of routers are blown to bits (e.g., in a military network located in a war zone), flooding will find a path if one exists, to get a packet to its destination. Flooding also requires little in the way of setup. The routers only need to know their neighbors. This means that flooding can be used as a building block for other routing algorithms that are
more efficient but need more in the way of setup. Flooding can also be used as a metric against which other routing algorithms can be compared. Flooding always chooses the shortest path because it chooses every possible path in parallel. Consequently, no other algorithm can produce a shorter delay (if we ignore the overhead generated by the flooding process itself).

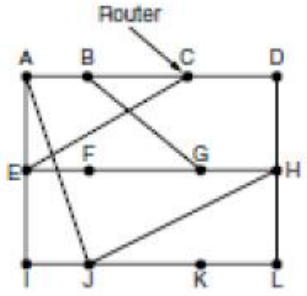
Distance Vector Routing

Computer networks generally use dynamic routing algorithms that are more complex than flooding, but more efficient because they find shortest paths for the current topology. Two dynamic algorithms in particular, distance vector routing and link state routing, are the most popular. In this section, we will look at the former algorithm. In the following section, we will study the latter algorithm. A **distance vector routing** algorithm operates by having each router maintain a table (i.e., a vector) giving the best known distance to each destination and which link to use to get there. These tables are updated by exchanging information with the neighbors. Eventually, every router knows the best link to reach each destination. The distance vector routing algorithm is sometimes called by other names, most commonly the distributed **Bellman-Ford** routing algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962). It was the original ARPANET routing algorithm and was also used in the Internet under the name RIP. In distance vector routing, each router maintains a routing table indexed by, and containing one entry for each router in the network. This entry has two parts: the preferred outgoing line to use for that destination and an estimate of the distance to that destination. The distance might be measured as the number of hops or using another metric, as we discussed for computing shortest paths. The router is assumed to know the ‘‘distance’’ to each of its neighbors. If the metric is hops, the distance is just one hop. If the metric is propagation delay, the router can measure it directly with special ECHO packets that the receiver just timestamps and sends back as fast as it can. As an example, assume that delay is used as a metric and that the router knows the delay to each of its neighbors. Once every *T* m sec, each router sends to each neighbor a list of its estimated delays to each destination. It also receives a similar list from each neighbor. Imagine that one of these tables has just come in from neighbor *X*, with *Xi* being *X*’s estimate of how long it takes to get to router *i*.If the router knows that the delay to *X* is *m* m sec, it also knows that it can reach router *i* via *X* in *Xi m* msec. By performing this calculation for each neighbor, a router can find out which estimate seems the best and use that estimate and the corresponding link in its new routing table. Note that the old routing table is not used in the calculation. This updating process is illustrated in Fig. 5-9. Part (a) shows a network. The first four columns of part (b) show the delay vectors received from the neighbors of router *J*. *A* claims to have a 12-msec delay to *B*, a 25-msec delay to *C*, a 40-msec delay to *D*,

etc. Suppose that *J* has measured or estimated its delay to its neighbors, *A*, *I*, *H*, and *K*, as 8, 10, 12, and 6 m sec, respectively.

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

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



(a)

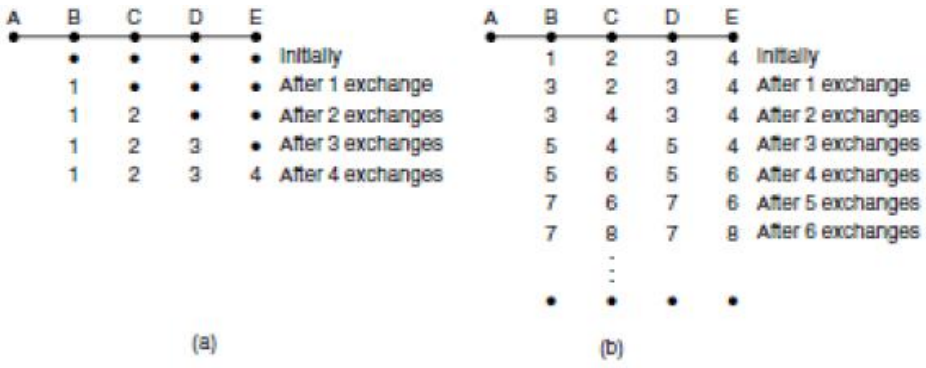
To	A	I	H	K	New estimated delay from J	Line
A	0	24	20	21	8	A
B	12	36	31	28	20	A
C	25	18	19	36	28	I
D	40	27	8	24	20	H
E	14	7	30	22	17	I
F	23	20	19	40	30	I
G	18	31	6	31	18	H
H	17	20	0	19	12	H
I	21	0	14	22	10	I
J	9	11	7	10	0	–
K	24	22	22	0	6	K
L	29	33	9	9	15	K
	JA delay is 8	JI delay is 10	JH delay is 12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

Vectors received from J's four neighbors

(b)

The Count-to-Infinity Problem

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



The count-to-infinity problem.

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

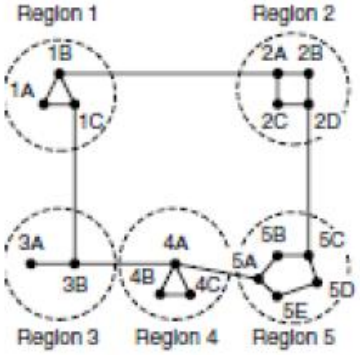
On the second exchange, *C* notices that each of its neighbors claims to have a path to *A* of length 3. It picks one of them at random and makes its new distance to *A* 4, as shown in the third row of Fig. 5-10(b). Subsequent exchanges produce the history shown in the rest of Fig. 5-10(b).From this figure, it should be clear why bad news travels Slowly: no router ever has a value more than one higher than the minimum of all its neighbors. Gradually, all routers work their way up to infinity, but the number of exchanges required depends on the numerical value used for infinity. For this reason, it is wise to set infinity to the longest path plus 1.Not entirely surprisingly, this problem is known as the **count-to-infinity** problem. There have been many attempts to solve it, for example, preventing routers from advertising their best paths back to the neighbors from which they heard them with the split horizon with poisoned reverse rule discussed in RFC 1058.However, none of these heuristics work well in practice despite the colorful names. The core of the problem is that when *X* tells *Y* that it has a path somewhere, *Y* has no way of knowing whether it itself is on the path.

Hierarchical Routing

As networks grow in size, the router routing tables grow proportionally. Not only is router memory consumed by ever-increasing tables, but more CPU time is needed to scan them and more bandwidth is needed to send status reports about them. At a certain point, the network may grow to the point where it is no longer feasible for every router to have an entry for every other

router, so the routing will have to be done hierarchically, as it is in the telephone network. When hierarchical routing is used, the routers are divided into what we will call **regions**. Each router knows all the details about how to route packets to destinations within its own region but knows nothing about the internal structure of other regions. When different networks are interconnected, it is natural to regard each one as a separate region to free the routers in one network from having to know the topological structure of the other ones. For huge networks, a two-level hierarchy may be insufficient; it may be necessary to group the regions into clusters, the clusters into zones, the zones into groups, and so on, until we run out of names for aggregations. As an example of a multilevel hierarchy, consider how a packet might be routed from Berkeley, California, to Malindi, Kenya. The Berkeley router would know the detailed topology within California but would send all out-of-state traffic to the Los Angeles router. The Los Angeles router would be able to route traffic directly to other domestic routers but would send all foreign traffic to New York. The New York router would be programmed to direct all traffic to the router in the destination country responsible for handling foreign traffic, say, in Nairobi. Finally, the packet would work its way down the tree in Kenya until it got to Malindi. Figure gives a quantitative example of routing in a two-level hierarchy with five regions. The full routing table for router *1A* has 17 entries, as shown in Fig. (b). When routing is done hierarchically, as in Fig. 5-14(c), there are entries for all the local routers, as before, but all other regions are condensed into a single router, so all traffic for region 2 goes via the *1B-2A* line, but the rest of the remote traffic goes via the *1C-3B* line. Hierarchical routing has reduced the table from 17 to 7 entries. As the ratio of the number of regions to the number of routers per region grows, the savings in table space increase. Unfortunately, these gains in space are not free. There is a penalty to be paid: increased path length. For example, the best route from *1A* to *5C* is via region 2,but with hierarchical routing all traffic to region 5 goes via region 3, because that is better for most destinations in region 5.When a single network becomes very large, an interesting question is ‘‘how many levels should the hierarchy have?’’ For example, consider a network with 720 routers. If there is no hierarchy, each router needs 720 routing table entries. If the network is partitioned into 24 regions of 30 routers each, each router needs 30 local entries plus 23 remote entries for a total of 53 entries. If a three-level hierarchy is chosen, with 8 clusters each containing 9 regions of 10 routers, each router needs 10 entries for local routers, 8 entries for routing to other regions within its own cluster, and 7 entries for distant clusters, for a total of 25 entries.Kamoun and Kleinrock (1979) discovered that the optimal number of levels for an N router network is ln N, requiring a total of e ln N entries per router. They have also shown that the increase in effective mean path length caused by hierarchical routing is sufficiently small that it is usually acceptable.

Hierarchical routing.



(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)

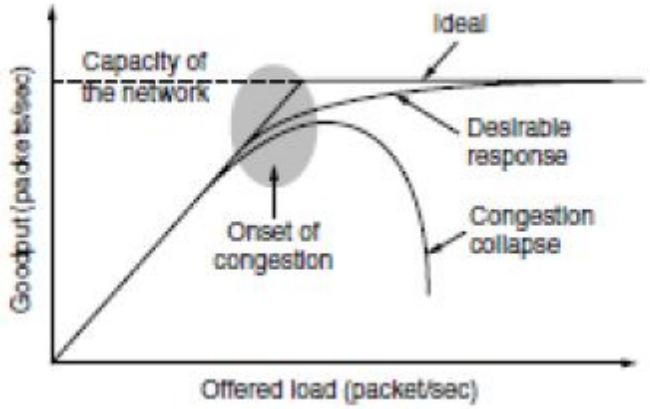
Hierarchical table for 1A

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

(c)

CONGESTION CONTROL ALGORITHMS

Too many packets present in (a part of) the network causes packet delay and loss that degrades performance. This situation is called **congestion**. The network and transport layers share the responsibility for handling congestion. Since congestion occurs within the network, it is the network layer that directly experiences it and must ultimately determine what to do with the excess packets. However, the most effective way to control congestion is to reduce the load that the transport layer is placing on the network. This requires the network and transport layers to work together. In this chapter we will look at the network aspects of congestion. In Chap. 6, we will complete the topic by covering the transport aspects of congestion. Figure depicts the onset of congestion. When the number of packets hosts send into the network is well within its carrying capacity, the number delivered is proportional to the number sent. If twice as many are sent, twice as many are delivered. However, as the offered load approaches the carrying capacity, bursts of traffic occasionally fill up the buffers inside routers and some packets are lost. These lost packets consume some of the capacity, so the number of delivered packets falls below the ideal curve. The network is now congested.



With too much traffic, performance drops sharply.

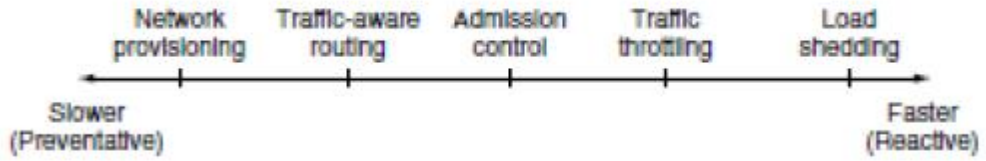
Unless the network is well designed, it may experience a **congestion collapse**, in which performance plummets as the offered load increases beyond the capacity. This can happen because packets can be sufficiently delayed inside the network that they are no longer useful when they leave the network. For example, in the early Internet, the time a packet spent waiting for a backlog of packets ahead of it to be sent over a slow 56-kbps link could reach the maximum time it was allowed to remain in the network. It then had to be thrown away. A different failure mode occurs when senders retransmit packets that are greatly delayed, thinking that they have been lost. In this case, copies of the same packet will be delivered by the network, again wasting its capacity. To capture these factors, the y-axis of Fig. is given as **good put**, which is the rate at which *useful* packets are delivered by the network. We would like to design networks that avoid congestion where possible and do not suffer from congestion collapse if they do become congested. Unfortunately, congestion cannot wholly be avoided. 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. Adding more memory may help up to a point, but Nagle (1987) realized that if routers have an infinite amount of memory, congestion gets worse, not better. This is because by the time packets get to the front of the queue, they have already timed out (repeatedly) and duplicates have been sent. This makes matters worse, not better—it leads to congestion collapse. Low-bandwidth links or routers that process packets more slowly than the line rate can also become congested. In this case, the situation can be improved by directing some of the traffic away from the bottleneck to other parts of the network. Eventually, however, all regions of the network will be congested. In this situation, there is no alternative but to shed load or build a faster network. It is worth pointing out the difference between congestion control and flow control, as the relationship is a very subtle one. Congestion control has to do with making sure the network is able to carry the offered traffic. It is a global issue, involving the behavior of all the hosts and routers. Flow control, in contrast, relates to the traffic between a particular sender and a particular receiver. Its job is to make sure that a fast sender cannot continually transmit data faster than the receiver is able to absorb it. To see the difference between these two concepts, consider a network made up of 100-Gbps fiber optic links on which a supercomputer is trying to force feed a large file to a personal computer that is capable of handling only 1 Gbps. Although there is no congestion (the network itself is not in trouble), flow control is needed to force the supercomputer to stop frequently to give the personal computer chance to breathe. At the other extreme, consider a network with 1-Mbps lines and 1000 large computers, half of which are trying to transfer files at 100 kbps to the

other half. Here, the problem is not that of fast senders overpowering slow receivers, but that the total offered traffic exceeds what the network can handle.

The reason congestion control and flow control are often confused is that the best way to handle both problems is to get the host to slow down. Thus, a host can get a ‘‘slow down’’ message either because the receiver cannot handle the load or because the network cannot handle it. We will come back to this point in Chap. 6. We will start our study of congestion control by looking at the approaches that can be used at different time scales. Then we will look at approaches to preventing congestion from occurring in the first place, followed by approaches for coping with it once it has set in.

Approaches to Congestion Control

The presence of congestion means that the load is (temporarily) greater than the resources (in a part of the network) can handle. Two solutions come to mind: increase the resources or decrease the load. As shown in Fig., these solutions are usually applied on different time scales to either prevent congestion or react to it once it has occurred.



Timescales of approaches to congestion control.

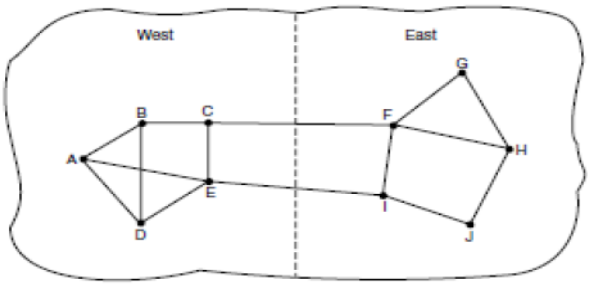
The most basic way to avoid congestion is to build a network that is well matched to the traffic that it carries. If there is a low-bandwidth link on the path along which most traffic is directed, congestion is likely. Sometimes resources on spare routers or enabling lines that are normally used only as backups (to make the system fault tolerant) or purchasing bandwidth on the open market. More often, links and routers that are regularly heavily utilized are upgraded at the earliest opportunity. This is called **provisioning** and happens on a time scale of months, driven by long-term traffic trends. To make the most of the existing network capacity, routes can be tailored to traffic patterns that change during the day as network user’s wake and sleep in different time zones. For example, routes may be changed to shift traffic away from heavily used paths by changing the shortest path weights. Some local radio stations have helicopters flying around their cities to report on road congestion to make it possible for their mobile listeners to route their packets (cars) around hotspots. This is called **traffic-aware routing**. Splitting traffic across multiple paths is also helpful. However, sometimes it is not possible to increase capacity. The only way then to beat back the congestion is to decrease the load. In a virtual-circuit network, new connections can be refused if they would cause the network to become congested. This is called **admission control**. At a finer granularity, when congestion is imminent the network can deliver feedback to the sources whose traffic flows are responsible for the problem. The network can request these sources to throttle their traffic, or it can slow down the traffic itself. Two difficulties with this approach are how to identify the onset of congestion, and how to inform the source that needs to slow down. To tackle the first issue, routers can monitor the average load, queuing delay, or packet loss. In all cases, rising numbers indicate growing congestion. To tackle the second issue, routers must participate in a feedback loop with the sources. For a scheme to work correctly, the time scale must be adjusted carefully. If every time two packets arrive in a row, a router yells STOP and every time a router is idle for 20 sec, it yells GO, the system will oscillate wildly and never converge. On the other hand, if it waits 30

minutes to make sure before saying anything, the congestion-control mechanism will react too sluggishly to be of any use. Delivering timely feedback is a nontrivial matter. An added concern is having routers send more messages when the network is already congested.

Finally, when all else fails, the network is forced to discard packets that it cannot deliver. The general name for this is **load shedding**. A good policy for choosing which packets to discard can help to prevent congestion collapse.

Traffic-Aware Routing

The first approach we will examine is traffic-aware routing. The routing schemes we looked at in Sec used fixed link weights. 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. Traffic-aware routing was used in the early Internet according to this model (Khanna and Zinky, 1989). However, there is a peril. Consider the network of Fig., 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 queuing 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.



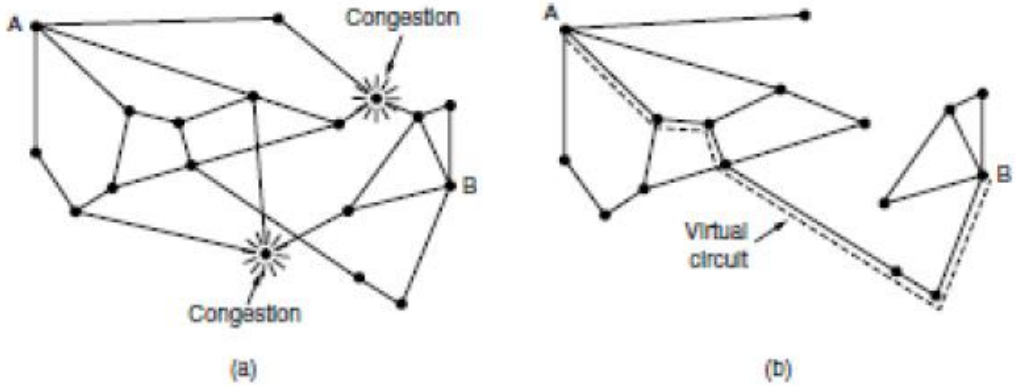
A network in which the East and West parts are connected by two links.

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, as in the scheme of Gallagher (1977).Given these difficulties, in the Internet routing protocols do not generally adjust their routes depending on the load. Instead, adjustments are made outside the routing protocol by slowly changing its inputs. This is called **traffic engineering**.

Admission Control

One technique that is widely used in virtual-circuit networks to keep congestion at bay is **admission control**. The idea is simple: do not set up a new virtual circuit unless the network can carry the added traffic without becoming congested. Thus, attempts to set up a virtual circuit may fail. This is better than the alternative, as letting more people in when the network is busy just

makes matters worse. By analogy, in the telephone system, when a switch gets overloaded it practices admission control by not giving dial tones. The trick with this approach is working out when a new virtual circuit will lead to congestion. The task is straightforward in the telephone network because of the fixed bandwidth of calls (64 kbps for uncompressed audio). However, virtual circuits in computer networks come in all shapes and sizes. Thus, the circuit must come with some characterization of its traffic if we are to apply admission control. Traffic is often described in terms of its rate and shape. The problem of how to describe it in a simple yet meaningful way is difficult because traffic is typically bursty—the average rate is only half the story. For example, traffic that varies while browsing the Web is more difficult to handle than a streaming movie with the same long-term throughput because the bursts of Web traffic are more likely to congest routers in the network. A commonly used descriptor that captures this effect is the **leaky bucket** or **token bucket**. A leaky bucket has two parameters that bound the average rate and the instantaneous burst size of traffic. Since leaky buckets are widely used for quality of service, we will go over them in detail in Sec. Armed with traffic descriptions, the network can decide whether to admit the new virtual circuit. One possibility is for the network to reserve enough capacity along the paths of each of its virtual circuits that congestion will not occur. In this case, the traffic description is a service agreement for what the network will guarantee its users. We have prevented congestion but veered into the related topic of quality of service a little too early; we will return to it in the next section. Even without making guarantees, the network can use traffic descriptions for admission control. The task is then to estimate how many circuits will fit within the carrying capacity of the network without congestion. Suppose that virtual circuits that may blast traffic at rates up to 10 Mbps all pass through the same 100-Mbps physical link. How many circuits should be admitted? Clearly, 10 circuits can be admitted without risking congestion, but this is wasteful in the normal case since it may rarely happen that all 10 are transmitting full blast at the same time. In real networks, measurements of past behavior that capture the statistics of transmissions can be used to estimate the number of circuits to admit, to trade better performance for acceptable risk. Admission control can also be combined with traffic-aware routing by considering routes around traffic hotspots as part of the setup procedure. For example, consider the network illustrated in Fig (a), in which two routers are congested, as indicated.



(a) A congested network. (b) The portion of the network that is not congested. 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 network as shown in Fig. 5-24(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. Shaikh et al. (1999) give a design for this kind of load-sensitive routing.

.

INTERNETWORKING

Until now, we have implicitly assumed that there is a single homogeneous network, with each machine using the same protocol in each layer. Unfortunately, this assumption is wildly optimistic. Many different networks exist, including PANs, LANs, MANs, and WANs. We have described Ethernet, Internet over cable, the fixed and mobile telephone networks, 802.11, 802.16, and more. Numerous protocols are in widespread use across these networks in every layer. In the following sections, we will take a careful look at the issues that arise when two or more networks are connected to form an **internetwork**, or more simply an **internet**.

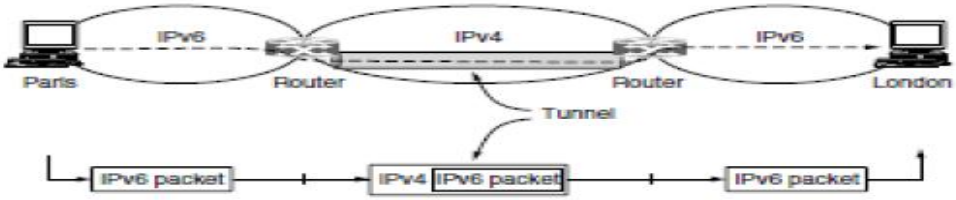
It would be much simpler to join networks together if everyone used a single networking technology, and it is often the case that there is a dominant kind of network, such as Ethernet. Some pundits speculate that the multiplicity of technologies will go away as soon as everyone realizes how wonderful [fill in your favorite network] is. Do not count on it. History shows this to be wishful thinking. Different kinds of networks grapple with different problems, so, for example, Ethernet and satellite networks are always likely to differ. Reusing existing systems, such as running data networks on top of cable, the telephone network, and power lines, adds constraints that cause the features of the networks to diverge. Heterogeneity is here to stay. If there will always be different networks, it would be simpler if we did not need to interconnect them. This also is unlikely. Bob Metcalfe postulated that the value of a network with N nodes is the number of connections that may be made between the nodes, or N^2 (Gilder, 1993).

This means that large networks are much more valuable than small networks because they allow many more connections, so there always will be an incentive to combine smaller networks. The Internet is the prime example of this interconnection. (We will write Internet with a capital ‘‘I’’ to distinguish it from other internets, or connected networks.)The purpose of joining all these networks is to allow users on any of them to communicate with users on all the other ones. When you pay an ISP for Internet service, you may be charged depending on the bandwidth of your line, but what you are really paying for is the ability to exchange packets with any other host that is also connected to the Internet. After all, the Internet would not be very popular if you could only send packets to other hosts in the same city. Since networks often differ in important ways, getting packets from one network to another is not always so easy. We must address problems of heterogeneity, and also problems of scale as the resulting internet grows very large. We will begin by looking at how networks can differ to see what we are up against. Then we shall see the approach used so successfully by IP (Internet Protocol), the network layer protocol of the Internet, including techniques for tunneling through networks, routing in internetworks, and packet fragmentation.

Tunneling

Handling the general case of making two different networks interwork is exceedingly difficult. However, there is a common special case that is manageable even for different network protocols. 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 an IPv6 network in Paris, an IPv6 network in London and connectivity between the offices via the IPv4 Internet. This situation is shown in Fig.



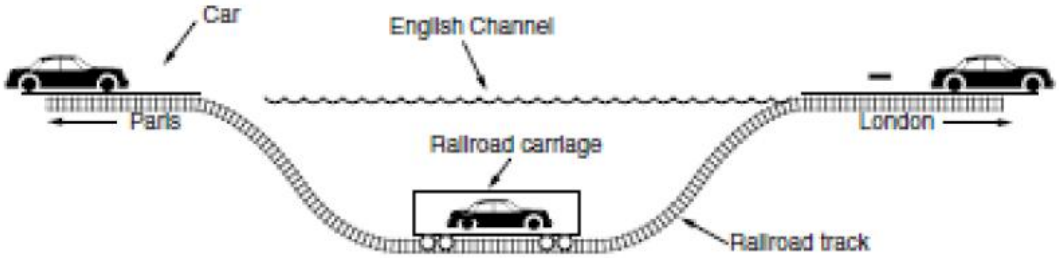
Tunneling a packet from Paris to London.

The solution to this problem is a technique called **tunneling**. To send an IP packet to a host in the London office, a host in the Paris office constructs the packet containing an IPv6 address in London, and sends it to the multiprotocol router that connects the Paris IPv6 network to the IPv4 Internet. When this router gets the IPv6 packet, it encapsulates the packet with an IPv4 header addressed to the IPv4 side of the multiprotocol router that connects to the London IPv6 network. That is, the router puts a (IPv6) packet inside a (IPv4) packet. When this wrapped packet arrives, the London router removes the original IPv6 packet and sends it onward to the destination host.

The path through the IPv4 Internet can be seen as a big tunnel extending from one multiprotocol router to the other. The IPv6 packet just travels from one end of the tunnel to the other, snug in its nice box. It does not have to worry about dealing with IPv4 at all. Neither do the hosts in Paris or London. Only the multiprotocol routers have to understand both IPv4 and IPv6 packets. In effect, the entire trip from one multiprotocol router to the other is like a hop over a single link.

An analogy may make tunneling clearer. 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 onto 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. 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. Tunneling is widely used to connect isolated hosts and networks using other networks. The network that results is called an **overlay** since it has effectively been overlaid on the base network. Deployment of a network protocol with a new feature is a common reason, as our ‘‘IPv6 over IPv4’’ example shows. The disadvantage of tunneling is that none of the hosts on the network that are tunneled over can be reached because the packets cannot escape in the middle of the tunnel.

Tunneling a car from France to England.



However, this limitation of tunnels is turned into an advantage with **VPNs** (**Virtual Private Networks**). A VPN is simply an overlay that is used to provide a measure of security.

Internetwork Routing

Routing through an internet poses the same basic problem as routing within a single network, but with some added complications. To start, the networks may internally use different routing algorithms. For example, one network may use link state routing and another distance vector routing. Since link state algorithms need to know the topology but distance vector algorithms do not, this difference alone would make it unclear how to find the shortest paths across the internet. Networks run by different operators lead to bigger problems. First, the operators may have different ideas about what is a good path through the network. One operator may want the route with the least delay, while another may want the most inexpensive route. This will lead the operators to use different quantities to set the shortest-path costs (e.g., milliseconds of delay vs.

monetary cost). The weights will not be comparable across networks, so shortest paths on the internet will not be well defined. Worse yet, one operator may not want another operator to even know the details of the paths in its network, perhaps because the weights and paths may reflect sensitive information (such as the monetary cost) that represents a competitive business advantage. Finally, the internet may be much larger than any of the networks that comprise it. It may therefore require routing algorithms that scale well by using a hierarchy; even if none of the individual networks need to use a hierarchy. All of these considerations lead to a two-level routing algorithm. Within each network, an **intra domain** or **interior gateway protocol** is used for routing.(‘‘Gateway’’ is an older term for ‘‘router.’’) It might be a link state protocol of the kind we have already described. Across the networks that make up the internet, an **inter domain** or **exterior gateway protocol** is used. The networks may all use different intra domain protocols, but they must use the same inter domain protocol. In the Internet, the inter domain routing protocol is called **BGP** (**Border Gateway Protocol**).

We will there is one more important term to introduce. Since each network is operated independently of all the others, it is often referred to as an **AS** (**Autonomous System**). A good mental model for an AS is an ISP network. In fact, an ISP network may be comprised of more than one AS, if it is managed, or, has been acquired, as multiple networks. But the difference is usually not significant. The two levels are usually not strictly hierarchical, as highly suboptimal paths might result if a large international network and a small regional network were both abstracted to be a single network. However, relatively little information about routes within the networks is exposed to find routes across the internetwork.

This helps to address all of the complications. It improves scaling and lets operators freely select routes within their own networks using a protocol of their choosing. It also does not require weights to be compared across networks or expose sensitive information outside of networks. However, we have said little so far about how the routes across the networks of the internet are determined. In the Internet, a large determining factor is the business arrangements between ISPs. Each ISP may charge or receive money from the other ISPs for carrying traffic. Another factor is that if internetwork routing requires crossing international boundaries, various laws may suddenly come into play, such as Sweden’s strict privacy laws about exporting personal data about Swedish citizens from Sweden. All of these nontechnical factors are wrapped up in the concept of a **routing policy** that governs the way autonomous networks select the routes that they use. We will return to routing policies when we describe BGP.

Packet Fragmentation

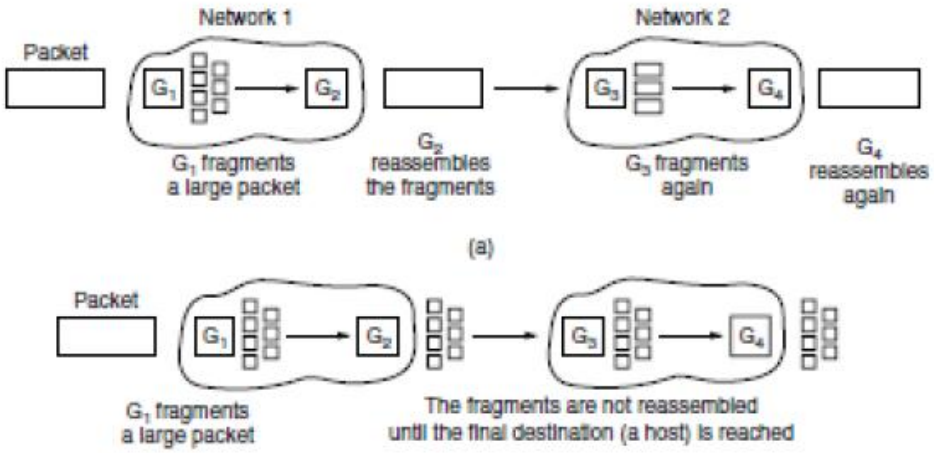
Each network or link imposes some maximum size on its packets. These limits have various causes, among them

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.

The result of all these factors is that the network designers are not free to choose any old maximum packet size they wish. Maximum payloads for some common technologies are 1500 bytes for Ethernet and 2272 bytes for 802.11. IP is more generous, allows for packets as big as 65,515 bytes. Hosts usually prefer to transmit large packets because this reduces packet overheads such as bandwidth wasted on header bytes. An obvious internetworking problem

appears when a large packet wants to travel through a network whose maximum packet size is too small. This nuisance has been a persistent issue, and solutions to it have evolved along with much experience gained on the Internet. One solution is to make sure the problem does not occur in the first place. However, this is easier said than done. A source does not usually know the path a packet will take through the network to a destination, so it certainly does not know how small packets must be to get there. This packet size is called the **Path MTU** (**Path Maximum Transmission Unit**). Even if the source did know the path MTU, packets are routed independently in a connectionless network such as the Internet. This routing means that paths may suddenly change, which can unexpectedly change the path MTU. The alternative solution to the problem is to allow routers to break up packets into **fragments**, sending each fragment as a separate network layer packet. However, as every parent of a small child knows, converting a large object into small fragments is considerably easier than the reverse process. (Physicists have even given this effect a name: the second law of thermodynamics.) 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 (a). In this approach, when an oversized packet arrives at *G1*, the router breaks it up into fragments. Each fragment is addressed to the same exit router, *G2*, where the pieces are recombined. In this way, passage through the small-packet network is made transparent. Subsequent networks are not even aware that fragmentation has occurred.

Transparent fragmentation is straightforward but has some problems. For one thing, the exit router must know when it has received all the pieces, so either a count field or an ‘‘end of packet’’ bit must be provided. Also, because all packets must exit via the same router so that they can be reassembled, the routes are constrained. By not allowing some fragments to follow one route to the ultimate destination and other fragments a disjoint route, some performance may be lost. More significant is the amount of work that the router may have to do. It may need to buffer the fragments as they arrive, and decide when to throw them away if not all of the fragments arrive. Some of this work may be wasteful, too, as the packet may pass through a series of small packet networks and need to be repeatedly fragmented and reassembled. The other fragmentation strategy is to refrain from recombining fragments at any intermediate routers. Once a packet has been fragmented, each fragment is

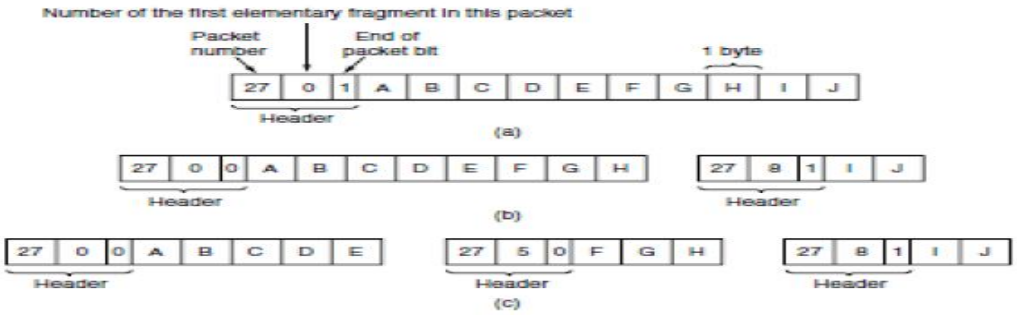


(a) Transparent fragmentation. (b) Nontransparent fragmentation.

treated as though it were an original packet. The routers pass the fragments, as shown in Fig. (b), and reassembly is performed only at the destination host. The main advantage of nontransparent fragmentation is that it requires routers to do less work. IP works this way. A complete design requires that the fragments be numbered in such a way that the original data stream can be reconstructed.

The design used by IP is to give every fragment a packet number (carried on all packets), an absolute byte offset within the packet, and a flag indicating whether it is the end of the packet. An example is shown in Fig. While simple, this design has some attractive properties. Fragments can be placed in a buffer at the destination in the right place for reassembly, even if they arrive out of order.

Fragments can also be fragmented if they pass over a network with a yet smaller MTU. This is shown in Fig. (c). Retransmissions of the packet (if all fragments were not received) can be fragmented into different pieces. Finally, fragments can be of arbitrary size, down to a single byte plus the packet header. In all cases, the destination simply uses the packet number and fragment offset to place the data in the right position, and the end-of-packet flag to determine when it has the complete packet. Unfortunately, this design still has problems. The overhead can be higher than with transparent fragmentation because fragment headers are now carried over some links where they may not be needed. But the real problem is the existence of fragments in the first place. Kent and Mogul (1987) argued that fragmentation is detrimental to performance because, as well as the header overheads, a whole packet is lost if any of its fragments are lost and because fragmentation is more of a burden for hosts than was originally realized.



Fragmentation when the elementary data size is 1 byte.

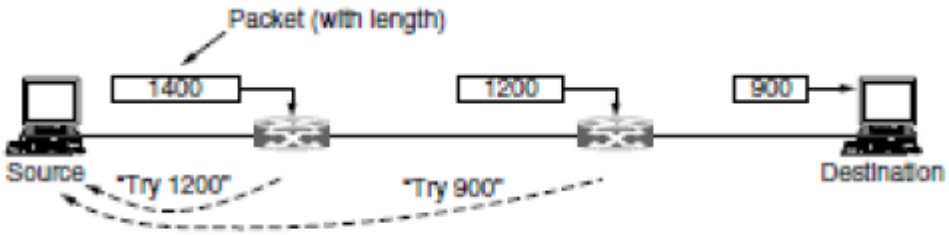
(a) Original packet, containing 10 data bytes.

(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.

This leads us back to the original solution of getting rid of fragmentation in the network, the strategy used in the modern Internet. The process is called **path MTU discovery** (Mogul and Deering, 1990). It works as follows. Each IP packet is sent with its header bits set to indicate that no fragmentation is allowed to be performed. If a router receives a packet that is too large, it generates an error packet, returns it to the source, and drops the packet. This is shown in Fig. When the source receives the error packet, it uses the information inside to refragment the packet into pieces that are small enough for the router to handle. If a router further down the path has an even smaller MTU, the process is repeated.

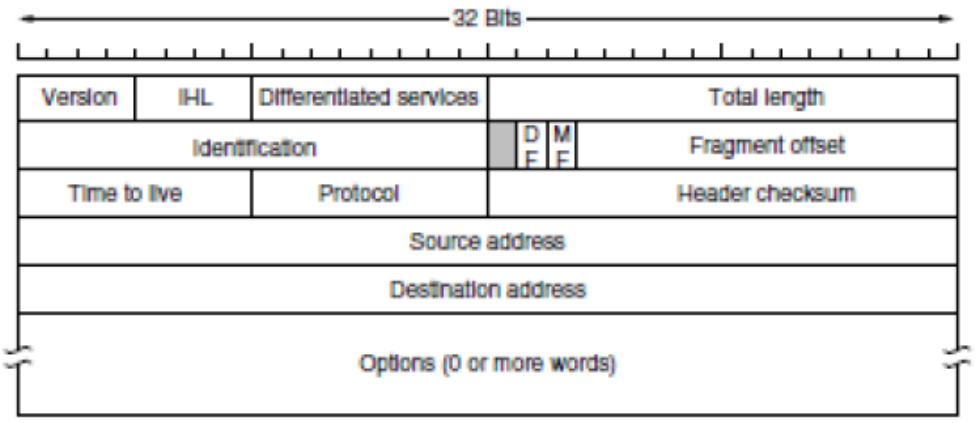
Path MTU discovery.



The advantage of path MTU discovery is that the source now knows what length packet to send. If the routes and path MTU change, new error packets will be triggered and the source will adapt to the new path. However, fragmentation is still needed between the source and the destination unless the higher layers learn the path MTU and pass the right amount of data to IP. TCP and IP are typically implemented together (as ‘‘TCP/IP’’) to be able to pass this sort of information. Even if this is not done for other protocols, fragmentation has still been moved out of the network and into the hosts. The disadvantage of path MTU discovery is that there may be added startup delays simply to send a packet. More than one round-trip delay may be needed to probe the path and find the MTU before any data is delivered to the destination. This begs the question of whether there are better designs. The answer is probably ‘‘Yes.’’ Consider the design in which each router simply truncates packets that exceed its MTU. This would ensure that the destination learns the MTU as rapidly as possible (from the amount of data that was delivered) and receives some of the data.

The IP Version 4 Protocol

An appropriate place to start our study of the network layer in the Internet is with the format of the IP datagrams themselves. An IPv4 datagram consists of a header part and a body or payload part. The header has a 20-byte fixed part and a variable-length optional part. The header format is shown in Fig. 5-46. The bits are transmitted from left to right and top to bottom, with the high-order bit of the *Version* field going first. (This is a ‘‘big-endian’’ network byte order. On little endian machines, such as Intel x86 computers, a software conversion is required on both transmission and reception.) In retrospect, little endian would have been a better choice, but at the time IP was designed, no one knew it would come to dominate computing.



The IPv4 (Internet Protocol) header.

The *Version* field keeps track of which version of the protocol the datagram belongs to. Version 4 dominates the Internet today, and that is where we have started our discussion. By including the version at the start of each datagram, it becomes possible to have a transition between

versions over a long period of time. In fact, IPv6, the next version of IP, was defined more than a decade ago, yet is only just beginning to be deployed. We will describe it later in this section. Its use will eventually be forced when each of China's almost 231 people has a desktop PC, a laptop, and an IP phone. As an aside on numbering, IPv5 was an experimental real-time stream protocol that was never widely used.

Since 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 minimum value is 5, which applies when no options are present. The maximum value of this 4-bit field is 15, which limits the header to 60 bytes, and thus the *Options* field to 40 bytes. For some options, such as one that records the route a packet has taken, 40 bytes is far too small, making those options useless. The *Differentiated services* field is one of the few fields that has changed its meaning (slightly) over the years. Originally, it was called the *Type of service* field. It was and still is intended to distinguish between different classes of service.

Various combinations of reliability and speed are possible. For digitized voice, fast delivery beats accurate delivery. For file transfer, error-free transmission is more important than fast transmission. The *Type of service* field provided 3 bits to signal priority and 3 bits to signal whether a host cared more about delay, throughput, or reliability. However, no one really knew what to do with these bits at routers, so they were left unused for many years. When differentiated services were designed, IETF threw in the towel and reused this field. Now, the top 6 bits are used to mark the packet with its service class; we described the expedited and assured services earlier in this chapter. The bottom 2 bits are used to carry explicit congestion notification information, such as whether the packet has experienced congestion; we described explicit congestion notification as part of congestion control earlier in this chapter. The *Total length* includes everything in the datagram—both header and data. The maximum length is 65,535 bytes. At present, this upper limit is tolerable, but with future networks, larger datagrams may be needed. The *Identification* field is needed to allow the destination host to determine which packet a newly arrived fragment belongs to. All the fragments of a packet contain the same *Identification* value.

Next comes an unused bit, which is surprising, as available real estate in the IP header is extremely scarce. As an April fool's joke, Bellovin (2003) proposed using this bit to detect malicious traffic. This would greatly simplify security, as packets with the ''evil'' bit set would be known to have been sent by attackers and could just be discarded. Unfortunately, network security is not this simple. Then come two 1-bit fields related to fragmentation. *DF* stands for Don't Fragment. It is an order to the routers not to fragment the packet. Originally, it was intended to support hosts incapable of putting the pieces back together again. Now it is used as part of the process to discover the path MTU, which is the largest packet that can travel along a path without being fragmented. By marking the datagram with the *DF* bit, the sender knows it will either arrive in one piece, or an error message will be returned to the sender. *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 packet 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, supporting a maximum packet length up to the limit of the *Total length* field. Working together, the *Identification*, *MF*, and *Fragment offset* fields are used to implement fragmentation as described in Sec. 5.5.5.The *TtL (Time to live)* field is a counter used to limit packet lifetimes. It

was originally supposed to count time in seconds, allowing a maximum lifetime of 255 sec. It must be decremented on each hop and is supposed to be decremented multiple times when a packet is queued for a long time in a router. In practice, it just counts hops. When it hits zero, the packet is discarded and a warning packet is sent back to the source host. This feature prevents packets from wandering around forever, something that otherwise might happen if the routing tables ever become corrupted.

When the network layer has assembled a complete packet, it needs to know what to do with it. The *Protocol* field tells it which transport process to give the packet to. TCP is one possibility, but so are UDP and some others. The numbering of protocols is global across the entire Internet. Protocols and other assigned numbers were formerly listed in RFC 1700, but nowadays they are contained in an online database located at *www.iana.org*. Since the header carries vital information such as addresses, it rates its own checksum for protection, the *Header checksum*. The algorithm is to add up all the 16-bit half words of the header as they arrive, using one's complement arithmetic, and then take the one's complement of the result. For purposes of this algorithm, the *Header checksum* is assumed to be zero upon arrival. Such a checksum is useful for detecting errors while the packet travels through the network. Note that it must be recomputed at each hop because at least one field always changes (the *Time to live* field), but tricks can be used to speed up the computation. The *Source address* and *Destination address* indicate the IP address of the source and destination network interfaces. 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 of variable length. Each begins with a 1-byte code identifying the option. Some options are followed by a 1-byte option length field, and then one or more data bytes. The *Options* field is padded out to a multiple of 4 bytes. Originally, the five options listed in Fig. were defined.

The *Security* option tells how secret the information is. In theory, a military router might use this field to specify not to route packets through certain countries the military considers to be ''bad guys.'' In practice, all routers ignore it, so its only practical function is to help spies find the good stuff more easily. The *Strict source routing* option gives the complete path from source to destination as a sequence of IP addresses. The datagram is required to follow that

Option	Description
Security	Specifies how secret the datagram is
Strict source routing	Gives the complete path to be followed
Loose source routing	Gives a list of routers not to be missed
Record route	Makes each router append its IP address
Timestamp	Makes each router append its address and timestamp

Some of the IP options

exact route. It is most useful for system managers who need to send emergency packets when the routing tables have been corrupted, or for making timing measurements. The *Loose source routing* option requires the packet to traverse the list of routers specified, in the order specified, but it is allowed to pass through other routers on the way. Normally, this option will provide only a few routers, to force a particular path. For example, to force a packet from London to Sydney to go west instead of east, this option might specify routers in New York, Los Angeles, and Honolulu. This option is most useful when political or economic considerations dictate passing through or avoiding certain countries.

The *Record route* option tells each router along the path to append its IP address to the *Options* field. This allows system managers to track down bugs in the routing algorithms (‘‘Why are packets from Houston to Dallas visiting Tokyo first?’’). When the ARPANET was first set up, no packet ever passed through more than nine routers, so 40 bytes of options was plenty. As mentioned above, now it is too small. Finally, the *Timestamp* option is like the *Record route* option, except that in addition to recording its 32-bit IP address, each router also records a 32-bit timestamp. This option, too, is mostly useful for network measurement. Today, IP options have fallen out of favor. Many routers ignore them or do not process them efficiently, shunting them to the side as an uncommon case. That is, they are only partly supported and they are rarely used.

IP Version 6

IP has been in heavy use for decades. It has worked extremely well, as demonstrated by the exponential growth of the Internet. Unfortunately, IP has become a victim of its own popularity: it is close to running out of addresses. Even with CIDR and NAT using addresses more sparingly, the last IPv4 addresses are expected to be assigned by ICANN before the end of 2012. This looming disaster was recognized almost two decades ago, and it sparked a great deal of discussion and controversy within the Internet community about what to do about it. In this section, we will describe both the problem and several proposed solutions. The only long-term solution is to move to larger addresses. **IPv6** (**IP version 6**) is a replacement design that does just that. It uses 128-bit addresses; a shortage of these addresses is not likely any time in the foreseeable future. However, IPv6 has proved very difficult to deploy. It is a different network layer protocol that does not really interwork with IPv4, despite many similarities. Also, companies and users are not really sure why they should want IPv6 in any case. The result is that IPv6 is deployed and used on only a tiny fraction of the Internet (estimates are 1%) despite having been an Internet Standard since 1998. The next several years will be an interesting time, as the few remaining IPv4 addresses are allocated. Will people start to auction off their IPv4 addresses on eBay? Will a black market in them spring up? Who knows? In addition to the address problems, other issues loom in the background. In its early years, the Internet was largely used by universities, high-tech industries, and the U.S. Government (especially the Dept. of Defense). With the explosion of interest in the Internet starting in the mid-1990s, it began to be used by a different group of people, often with different requirements. For one thing, numerous people with smart phones use it to keep in contact with their home bases. For another, with the impending convergence of the computer, communication, and entertainment industries, it may not be that long before every telephone and television set in the world is an Internet node, resulting in a billion machines being used for audio and video on demand. Under these circumstances, it became apparent that IP had to evolve and become more flexible. Seeing these problems on the horizon, in 1990 IETF started work on a new version of IP, one that would never run out of addresses, would solve a variety of other problems, and be more flexible and efficient as well. Its major goals were:

1. Support billions of hosts, even with inefficient address allocation.
2. Reduce the size of the routing tables.
3. Simplify the protocol, to allow routers to process packets faster.
4. Provide better security (authentication and privacy).
5. Pay more attention to the type of service, particularly for real-time data.
6. Aid multicasting by allowing scopes to be specified.
7. Make it possible for a host to roam without changing its address.
8. Allow the protocol to evolve in the future.

9. Permit the old and new protocols to coexist for years.

The design of IPv6 presented a major opportunity to improve all of the features in IPv4 that fall short of what is now wanted. To develop a protocol that met all these requirements, IETF issued a call for proposals and discussion in RFC 1550. Twenty-one responses were initially received. By December 1992, seven serious proposals were on the table. They ranged from making minor patches to IP, to throwing it out altogether and replacing it with a completely different protocol. One proposal was to run TCP over CLNP, the network layer protocol designed for OSI. With its 160-bit addresses, CLNP would have provided enough address space forever as it could give every molecule of water in the oceans enough addresses (roughly 25) to set up a small network. This choice would also have unified two major network layer protocols. However, many people felt that this would have been an admission that something in the OSI world was actually done right, a statement considered Politically Incorrect in Internet circles. CLNP was patterned closely on IP, so the two are not really that different. In fact, the protocol ultimately chosen differs from IP far more than CLNP does. Another strike against CLNP was its poor support for service types, something required to transmit multimedia efficiently.

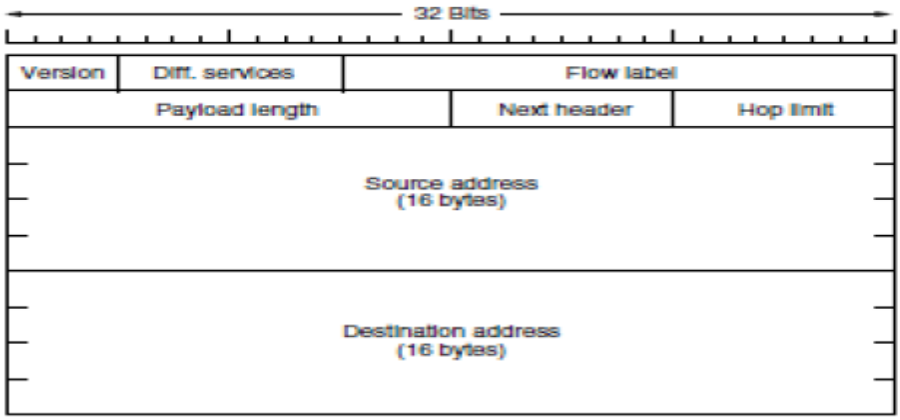
Three of the better proposals were published in *IEEE Network* (Deering, 1993; Francis, 1993; and Katz and Ford, 1993). After much discussion, revision, and jockeying for position, a modified combined version of the Deering and Francis proposals, by now called **SIPP** (**Simple Internet Protocol Plus**) was selected and given the designation **IPv6.** IPv6 meets IETF's goals fairly well. It maintains the good features of IP, discards or deemphasizes the bad ones, and adds new ones where needed. In general, IPv6 is not compatible with IPv4, but it is compatible with the other auxiliary Internet protocols, including TCP, UDP, ICMP, IGMP, OSPF, BGP, and DNS, with small modifications being required to deal with longer addresses. The main features of IPv6 are discussed below. More information about it can be found in RFCs 2460 through 2466.First and foremost, IPv6 has longer addresses than IPv4. They are 128 bits long, which solves the problem that IPv6 set out to solve: providing an effectively unlimited supply of Internet addresses. We will have more to say about addresses shortly. The second major improvement of IPv6 is the simplification of the header. It contains only seven fields (versus 13 in IPv4). This change allows routers to process packets faster and thus improves throughput and delay. We will discuss the header shortly, too. The third major improvement is better support for options. This change was essential with the new header because fields that previously were required are now optional (because they are not used so often). In addition, the way options are represented is different, making it simple for routers to skip over options not intended for them. This feature speeds up packet processing time.

A fourth area in which IPv6 represents a big advance is in security. IETF had its fill of newspaper stories about precocious 12-year-olds using their personal computers to break into banks and military bases all over the Internet. There was a strong feeling that something had to be done to improve security. Authentication and privacy are key features of the new IP. These were later retrofitted to IPv4, however, so in the area of security the differences are not so great any more. Finally, more attention has been paid to quality of service. Various halfhearted efforts to improve QoS have been made in the past, but now, with the growth of multimedia on the Internet, the sense of urgency is greater.

The Main IPv6 Header

The IPv6 header is shown in Fig. 5-56. The *Version* field is always 6 for IPv6 (and 4 for IPv4). During the transition period from IPv4, which has already taken more than a decade, routers will be able to examine this field to tell what kind of packet they have. As an aside, making this test

wastes a few instructions in the critical path, given that the data link header usually indicates the network protocol for de-multiplexing, so some routers may skip the check. For example, the Ethernet *Type* field has different values to indicate an IPv4 or an IPv6 payload. The discussions between the ‘‘Do it right’’ and ‘‘Make it fast’’ camps will no doubt be lengthy and vigorous.



The IPv6 fixed header (required).

The ***Differentiated services*** field (originally called *Traffic class*) is used to distinguish the class of service for packets with different real-time delivery requirements. It is used with the Differentiated service architecture for quality of service in the same manner as the field of the same name in the IPv4 packet. Also, the low-order 2 bits are used to signal explicit congestion indications, again in the same way as with IPv4.

The ***Flow label*** field provides a way for a source and destination to mark groups of packets that have the same requirements and should be treated in the same way by the network, forming a pseudo connection. For example, a stream of packets from one process on a certain source host to a process on a specific destination host might have stringent delay requirements and thus need reserved bandwidth. The flow can be set up in advance and given an identifier. When a packet with a nonzero *Flow label* shows up, all the routers can look it up in internal tables to see what kind of special treatment it requires. In effect, flows are an attempt to have it both ways: the flexibility of a datagram network and the guarantees of a virtual-circuit network. Each flow for quality of service purposes is designated by the source address, destination address, and flow number. This design means that up to 220 flows may be active at the same time between a given pair of IP addresses. It also means that even if two flows coming from different hosts but with the same flow label pass through the same router, the router will be able to tell them apart using the source and destination addresses. It is expected that flow labels will be chosen randomly, rather than assigned sequentially starting at 1, so routers are expected to hash them. The *Payload length* field tells how many bytes follow the 40-byte header of Fig. The name was changed from the IPv4 *Total length* field because the meaning was changed slightly: the 40 header bytes are no longer counted as part of the length (as they used to be). This change means the payload can now be 65,535 bytes instead of a mere 65,515 bytes.

The *Next header* field lets the cat out of the bag. The reason the header could be simplified is that there can be additional (optional) extension headers. This field tells which of the (currently) six extension headers, if any, follow this one.

If this header is the last IP header, the *Next header* field tells which transport protocol handler (e.g., TCP, UDP) to pass the packet to. The *Hop limit* field is used to keep packets from living forever. It is, in practice, the same as the *Time to live* field in IPv4, namely, a field that is decremented on each hop. In theory, in IPv4 it was a time in seconds, but no router used it that way, so the name was changed to reflect the way it is actually used. Next come the *Source address* and *Destination address* fields. Deering's original proposal, SIP, used 8-byte addresses, but during the review process many people felt that with 8-byte addresses IPv6 would run out of addresses within a few decades, whereas with 16-byte addresses it would never run out. Other people argued that 16 bytes was overkill, whereas still others favored using 20-byte addresses to be compatible with the OSI datagram protocol. Still another faction wanted variable-sized addresses. After much debate and more than a few words unprintable in an academic textbook, it was decided that fixed-length 16-byte addresses were the best compromise.

A new notation has been devised for writing 16-byte addresses. They are written as eight groups of four hexadecimal digits with colons between the groups, like this: 8000:0000:0000:0000:0123:4567:89AB: CDEF Since many addresses will have many zeros inside them, three optimizations have been authorized. First, leading zeros within a group can be omitted, so 0123 can be written as 123. Second, one or more groups of 16 zero bits can be replaced by a pair of colons. Thus, the above address now becomes 8000::123:4567:89AB:CDEF Finally, IPv4 addresses can be written as a pair of colons and an old dotted decimal number, for

example: ::192.31.20.46 Perhaps it is unnecessary to be so explicit about it, but there are a lot of 16- addresses per square meter. Students of chemistry will notice that this number is larger than Avogadro's number. While it was not the intention to give every molecule on the surface of the earth its own IP address, we are not that far off. In practice, the address space will not be used efficiently, just as the telephone number address space is not (the area code for Manhattan, 212, is nearly full, but that for Wyoming, 307, is nearly empty). In RFC 3194, Durand and Huitema calculated that, using the allocation of telephone numbers as a guide, even in the most pessimistic scenario there will still be well over 1000 IP addresses per square meter of the entire earth's surface (land and water). In any likely scenario, there will be trillions of them per square meter. In short, it seems unlikely that we will run out in the foreseeable future. It is instructive to compare the IPv4 header (Fig.) with the IPv6 header (Fig) to see what has been left out in IPv6. The *IHL* field is gone because the IPv6 header has a fixed length. The *Protocol* field was taken out because the *Next header* field tells what follows the last IP header (e.g., a UDP or TCP segment).All the fields relating to fragmentation were removed because IPv6 takes a different approach to fragmentation. To start with, all IPv6-conformant hosts are expected to dynamically determine the packet size to use. They do this using the path MTU discovery procedure we described in Sec. 5.5.5. In brief, when a host sends an IPv6 packet that is too large, instead of fragmenting it, the router that is unable to forward it drops the packet and sends an error message back to the sending host. This message tells the host to break up all future packets to that destination. Having the host send packets that are the right size in the first place is ultimately much more efficient than having the routers fragment them on the fly. Also, the minimum-size packet that routers must be able to forward has been raised from 576 to 1280 bytes to allow 1024 bytes of data and many headers. Finally, the *Checksum* field is gone because calculating it greatly reduces performance. With the reliable networks now used, combined with the fact that the data link layer and transport layers normally have their own checksums, the value of yet another checksum was deemed not worth the performance price it extracted. Removing all these features has resulted in a lean and mean network layer protocol. Thus, the goal of IPv6—a fast, yet flexible, protocol with plenty of address space—is met by this design.

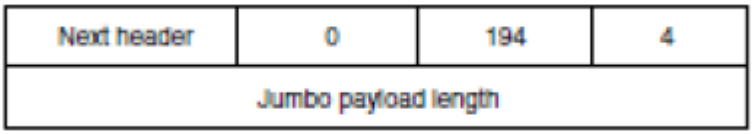
Extension Headers

Some of the missing IPv4 fields are occasionally still needed, so IPv6 introduces the concept of (optional) **extension headers**. These headers can be supplied to provide extra information, but encoded in an efficient way. Six kinds of extension headers are defined at present, as listed in Fig. Each one is optional, but if more than one is present they must appear directly after the fixed header, and preferably in the order listed.

Extension header	Description
Hop-by-hop options	Miscellaneous information for routers
Destination options	Additional information for the destination
Routing	Loose list of routers to visit
Fragmentation	Management of datagram fragments
Authentication	Verification of the sender's identity
Encrypted security payload	Information about the encrypted contents

IPv6 extension headers

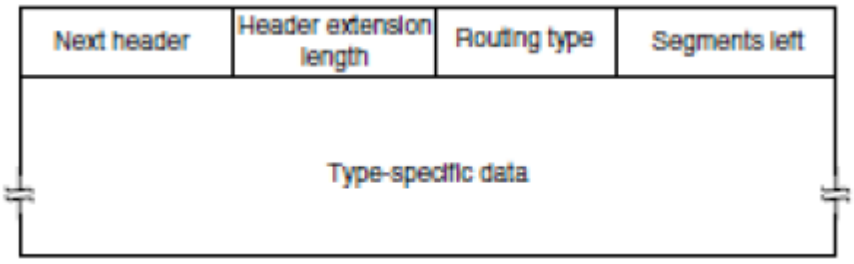
Some of the headers have a fixed format; others contain a variable number of variable-length options. For these, each item is encoded as a (*Type, Length, Value*) tuple. The *Type* is a 1-byte field telling which option this is. The *Type* values have been chosen so that the first 2 bits tell routers that do not know how to process the option what to do. The choices are: skip the option; discard the packet; discard the packet and send back an ICMP packet; and discard the packet but do not send ICMP packets for multicast addresses (to prevent one bad multicast packet from generating millions of ICMP reports). The *Length* is also a 1-byte field. It tells how long the value is (0 to 255 bytes). The *Value* is any information required, up to 255 bytes. The hop-by-hop header is used for information that all routers along the path must examine. So far, one option has been defined: support of datagrams exceeding 64 KB. The format of this header is shown in. When it is used, the *Payload length* field in the fixed header is set to 0.



The hop-by-hop extension header for large data grams (jumbo grams)

As with all extension headers, this one starts with a byte telling what kind of header comes next. This byte is followed by one telling how long the hop-by-hop header is in bytes, excluding the first 8 bytes, which are mandatory. All extensions begin this way. The next 2 bytes indicate that this option defines the datagram size (code 194) and that the size is a 4-byte number. The last 4 bytes give the size of the datagram. Sizes less than 65,536 bytes are not permitted and will result in the first router discarding the packet and sending back an ICMP error message. Datagrams using this header extension are called **jumbo grams**. The use of jumbo grams is important for supercomputer applications that must transfer gigabytes of data efficiently across the Internet.

The destination options header is intended for fields that need only be interpreted at the destination host. In the initial version of IPv6, the only options defined are null options for padding this header out to a multiple of 8 bytes, so initially it will not be used. It was included to make sure that new routing and host software can handle it, in case someone thinks of a destination option some day. The routing header lists one or more routers that must be visited on the way to the destination. It is very similar to the IPv4 loose source routing in that all addresses listed must be visited in order, but other routers not listed may be visited in between. The format of the routing header is shown in Fig.



The extension header for routing

The first 4 bytes of the routing extension header contain four 1-byte integers. The *Next header* and *Header extension length* fields were described above. The *Routing type* field gives the format of the rest of the header. Type 0 says that a reserved 32-bit word follows the first word, followed by some number of IPv6 addresses. Other types may be invented in the future, as

needed. Finally, the *Segments left* field keeps track of how many of the addresses in the list have not yet been visited. It is decremented every time one is visited. When it hits 0, the packet is on its own with no more guidance about what route to follow. Usually, at this point it is so close to the destination that the best route is obvious.