5

THE NETWORK LAYER

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

To achieve its goals, the network layer must learn about the topology of the network (i.e., the set of all routers and links) and compute appropriate paths through it, even for large networks. It must also take care when choosing routes to avoid overloading some of the communication lines and routers while leaving others idle. Finally, when the source and destination are in different independently operated networks, sometimes called autonomous systems, new challenges arise, such as coordinating traffic flows across multiple networks and managing network utilization. These problems are typically handled at the network layer; network operators are often tasked with dealing with these challenges manually. Conventionally, network operators had to reconfigure the network layer manually, through low-level configuration. More recently, however, the advent of software-defined networking and programmable hardware has made it possible to configure the network layer from higher-level software programs, and even to redefine the functions of the network layer entirely. In this chapter, we will study all these issues and illustrate them, focusing in particular on the Internet and its network layer protocol, IP (Internet Protocol).

5.1 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.

5.1.1 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 Fig. 5-1. The major components of the network are the ISP's equipment (routers, switches, and middleboxes 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).

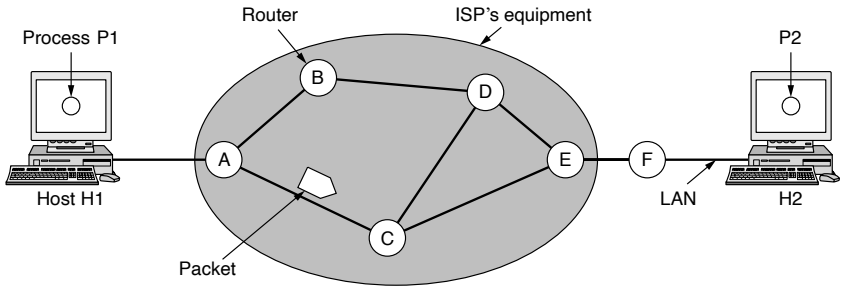


Figure 5-1. The environment of the network layer protocols.

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 (e.g., over an ADSL line or a cable television wire). 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, as we have seen in previous chapters.

5.1.2 Services Provided to the Transport Layer

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

1. The services should be independent of the router technology.
2. The transport layer should be shielded from the number, type, and topology of the routers present.
3. The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.

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 multiprotocol label switching, which we will describe in this chapter, and VLANs, which we saw in Chap. 4. Both technologies are widely used.

5.1.3 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 (old) 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. 5-2 has a long message for *P2*. It hands the message to the transport layer, with instructions to deliver it to process *P2* on host *H2*. The transport layer code runs on *H1*, typically within the operating system. It prepends a transport header to the front of the message and hands the result to the network layer, probably just another procedure within the operating system.

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. 5-2, *A* has only two outgoing lines—to *B* and to *C*—so every incoming packet must be sent to one of these routers, even if the ultimate destination is to some other router. *A*'s 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.

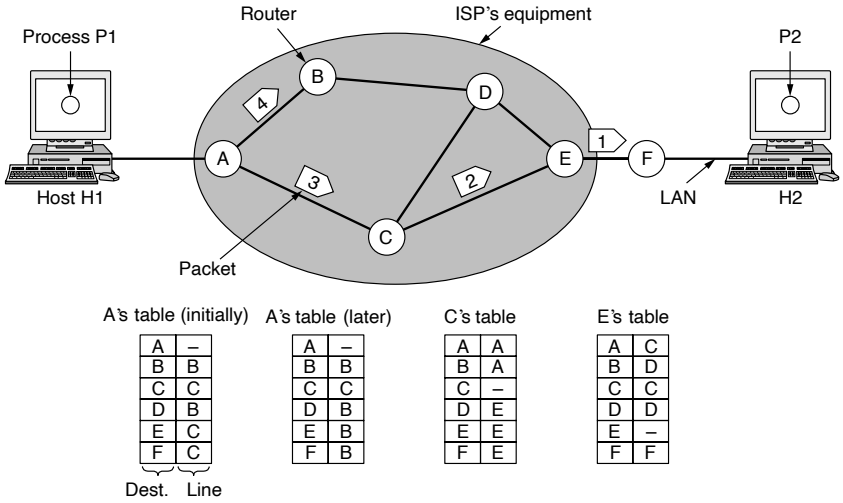


Figure 5-2. Routing within a datagram network.

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, 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. We will describe IP and these two versions in much detail later in this chapter.

5.1.4 Implementation of Connection-Oriented Service

For connection-oriented service, we need to have 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 Fig. 5-2. Instead, when a connection is established, a route from the source machine to the destination machine is chosen as part of the connection setup and stored in tables inside the routers. That route is used for all traffic flowing over the connection, exactly the same way

that the telephone system works. When the connection is released, the virtual circuit is also terminated. With connection-oriented service, each packet carries an identifier telling which virtual circuit it belongs to.

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

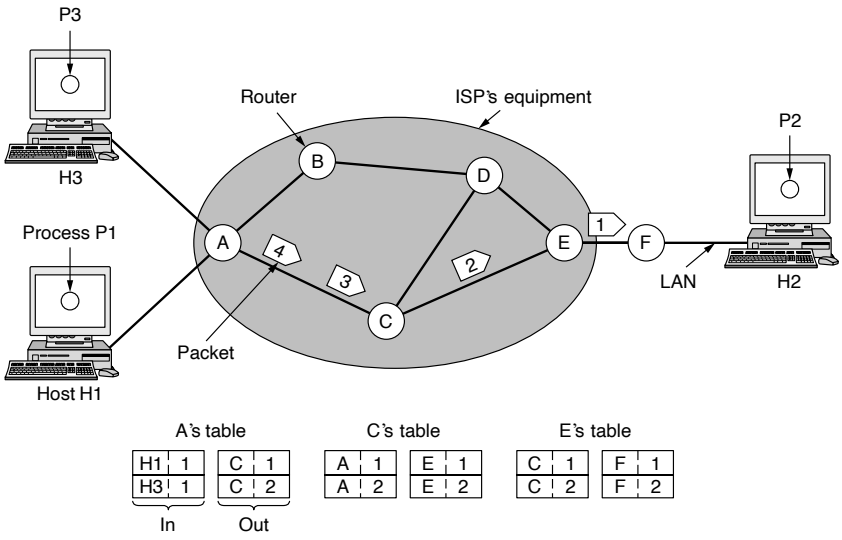


Figure 5-3. Routing within a virtual-circuit network.

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. Please 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.

An example of a connection-oriented network service is MPLS (MultiProtocol Label Switching). It is used within ISP networks in the Internet, with IP packets wrapped in an MPLS header having a 20-bit connection identifier or label. 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. We will have more to say about MPLS later in this chapter.

5.1.5 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. 5-4, 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

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

Inside the network, several trade-offs exist between virtual circuits and datagrams. 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.

5.2 ROUTING ALGORITHMS IN A SINGLE NETWORK

The main function of the network layer is routing packets from the source machine to the destination machine. In this section, we discuss how the network layer achieves this function within a single administrative domain or autonomous system. 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, the routing decision must be made anew for

every arriving data packet since the best route may have changed since last time. If the network uses virtual circuits internally, routing decisions are made only when a new virtual circuit is being set up. Thereafter, data packets just follow the already established route. The latter case is sometimes called **session routing** because a route remains in force for an entire session (e.g., while logged in over a VPN).

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

Regardless of whether routes are chosen independently for each packet sent or only when new connections are established, certain properties are desirable in a routing algorithm: correctness, simplicity, robustness, stability, fairness, and efficiency. Correctness and simplicity hardly require comment, but the need for robustness may be less obvious at first. Once a major network comes on the air, it may be expected to run continuously for years without system-wide failures. During that period there will be hardware and software failures of all kinds. Hosts, routers, and lines will fail repeatedly, and the topology will change many times. The routing algorithm should be able to cope with changes in the topology and traffic without requiring all jobs in all hosts to be aborted. Imagine the havoc if the network needed to be rebooted every time some router crashed.

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

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

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

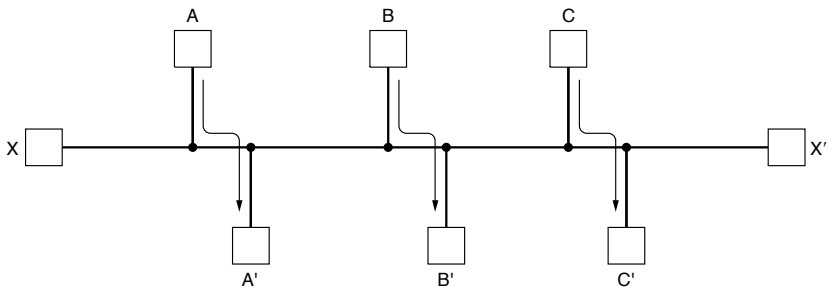


Figure 5-5. Network with a conflict between fairness and efficiency.

bandwidth consumed per packet, which generally tends to improve the overall network throughput as well.

Routing algorithms can be grouped into two major classes: nonadaptive and adaptive. **Nonadaptive algorithms** do not base their routing decisions on any measurements or estimates of the current topology and traffic. Instead, the choice of the route to use to get from I to J (for all I and J) is computed in advance, offline, and downloaded to the routers when the network is booted. This procedure is sometimes called **static routing**. Because it does not respond to failures, static routing is mostly useful for situations in which the routing choice is clear. For example, router F in Fig. 5-3 should send packets headed into the network to router E regardless of the ultimate destination.

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

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

5.2.1 The Optimality Principle

Before we get into specific algorithms, it may be helpful to note that one can make a general statement about optimal routes without regard to network topology or traffic. This statement is known as the **optimality principle** (Bellman, 1957).

It states that if router J is on the optimal path from router I to router K, then the optimal path from J to K also falls along the same route. To see this, call the part of the route from I to J r_1 and the rest of the route r_2. If a route better than r_2 existed from J to K, it could be concatenated with r_1 to improve the route from I to K, contradicting our statement that $r_1 r_2$ is optimal.

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

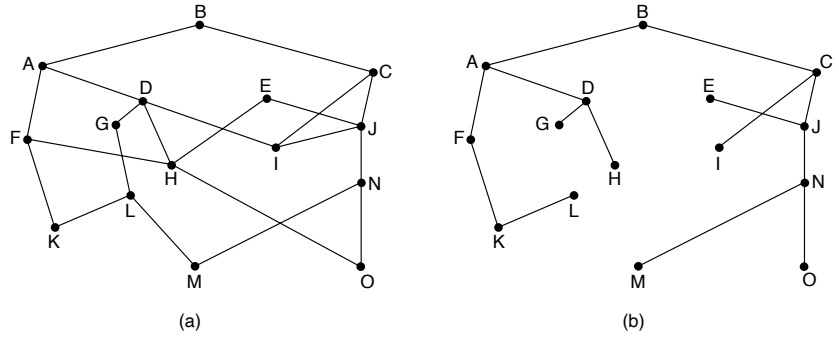


Figure 5-6. (a) A network. (b) A sink tree for router *B*.

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

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

5.2.2 Shortest Path Algorithm

Let us begin our study of routing algorithms with a simple technique for computing optimal paths given a complete picture of the network. These paths are the ones that we want a distributed routing algorithm to find, even though not all routers may know all of the details of the network.

The idea is to build a graph of the network, with each node of the graph representing a router and each edge of the graph representing a communication line, or link. To choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph.

The concept of a **shortest path** deserves some explanation. One way of measuring path length is the number of hops. Using this metric, the paths *ABC* and *ABE* in Fig. 5-7 are equally long. Another metric is the geographic distance in kilometers, in which case *ABC* is clearly much longer than *ABE* (assuming the figure is drawn to scale).

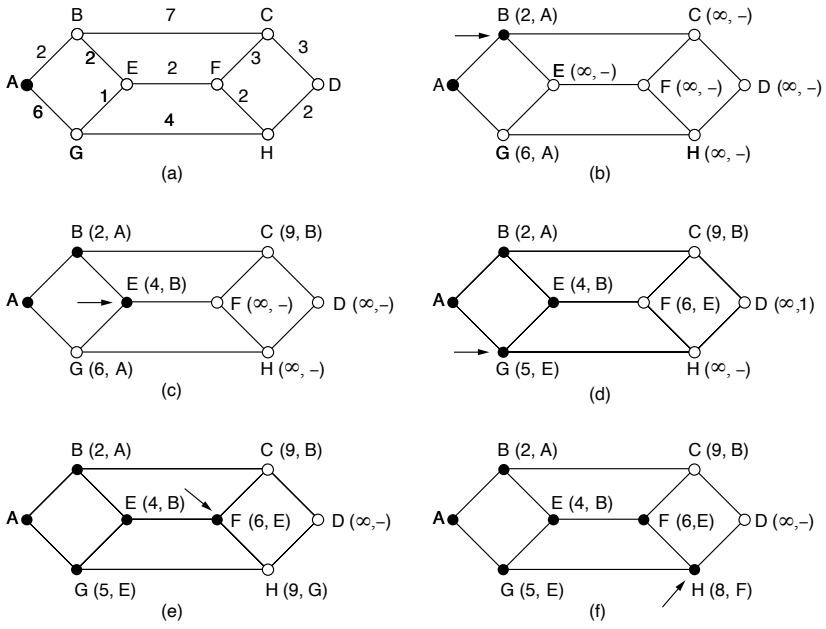


Figure 5-7. The first six steps used in computing the shortest path from *A* to *D*. The arrows indicate the working node.

However, many other metrics besides hops and physical distance are also possible. For example, each edge could be labeled with the mean delay of a standard

test packet, as measured by hourly runs. With this graph labeling, the shortest path is the fastest path rather than the path with the fewest edges or kilometers.

In the general case, the labels on the edges could be computed as a function of the distance, bandwidth, average traffic, communication cost, measured delay, and other factors. By changing the weighting function, the algorithm would then compute the ''shortest'' path measured according to any one of a number of criteria or to a combination of criteria.

Several algorithms for computing the shortest path between two nodes of a graph are known. This one is due to Dijkstra (1959) and finds the shortest paths between a source and all destinations in the network. Each node is labeled (in parentheses) with its distance from the source node along the best known path. The distances must be non-negative, as they will be if they are based on real quantities like bandwidth and delay. Initially, no paths are known, so all nodes are labeled with infinity. As the algorithm proceeds and paths are found, the labels may change, reflecting better paths. A label may be either tentative or permanent. Initially, all labels are tentative. When it is discovered that a label represents the shortest possible path from the source to that node, it is made permanent and never changed thereafter.

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

Having examined each of the nodes adjacent to A, we examine all the tentatively labeled nodes in the whole graph and make the one with the smallest label permanent, as shown in Fig. 5-7(b). This one becomes the new working node.

We now start at B and examine all nodes adjacent to it. If the sum of the label on B and the distance from B to the node being considered is less than the label on that node, we have a shorter path, so the node is relabeled.

After all the nodes adjacent to the working node have been inspected and the tentative labels changed if possible, the entire graph is searched for the tentatively labeled node with the smallest value. This node is made permanent and becomes the working node for the next round. Figure 5-7 shows the first six steps of the algorithm.

To see why the algorithm works, look at Fig. 5-7(c). At this point we have just made E permanent. Suppose that there were a shorter path than ABE, say $AXYZE$ (for some X and Y). There are two possibilities: either node Z has already been made permanent, or it has not been. If it has, then E has already been probed (on

the round following the one when *Z* was made permanent), so the *AXYZE* path has not escaped our attention and thus cannot be a shorter path.

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

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

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

5.2.3 Flooding

When a routing algorithm is implemented, each router must make decisions based on local knowledge, not the complete picture of the network. A simple local technique is **flooding**, in which every incoming packet is sent out on every outgoing line except the one it arrived on.

Flooding obviously generates vast numbers of duplicate packets, in fact, an infinite number unless some measures are taken to damp the process. One such measure is to have a hop counter contained in the header of each packet that is decremented at each hop, with the packet being discarded when the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the network.

Flooding with a hop count can produce an exponential number of duplicate packets as the hop count grows and routers duplicate packets they have seen before. A better technique for damming the flood is to have routers keep track of which packets have been flooded, to avoid sending them out a second time. One way to achieve this goal is to have the source router put a sequence number in each packet it receives from its hosts. Each router then needs a list per source router telling which sequence numbers originating at that source have already been seen. If an incoming packet is on the list, it is not flooded.

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

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

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

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

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

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

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

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

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 smithereens (e.g., in a military network located in a war zone), flooding will find a path if one exists, to get a packet to its destination. Flooding also requires little in the way of setup. The routers only need to know their neighbors. This means that flooding can be used as a building block for other routing algorithms that are more efficient but need more in the way of setup. Flooding can also be used as a metric against which other routing algorithms can be compared. Flooding always chooses the shortest path because it chooses every possible path in parallel. Consequently, no other algorithm can produce a shorter delay (if we ignore the overhead generated by the flooding process itself).

5.2.4 Distance Vector Routing

Computer networks generally use dynamic routing algorithms that are more complex than flooding, but more efficient because they find shortest paths for the current topology. Two dynamic algorithms in particular, distance vector routing and link state routing, are the most popular. In this section, we will look at the former algorithm. In the following section, we will study the latter algorithm.

A **distance vector routing** algorithm operates by having each router maintain a table (i.e., a vector) giving the best known distance to each destination and which link to use to get there. These tables are updated by exchanging information with the neighbors. Eventually, every router knows the best link to reach each destination.

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

In distance vector routing, each router maintains a routing table indexed by, and containing one entry for, each router in the network. This entry has two parts: the preferred outgoing line to use for that destination, and an estimate of the distance to that destination. The distance might be measured as the number of hops or using another metric, as we discussed for computing shortest paths.

The router is assumed to know the ''distance'' to each of its neighbors. If the metric is hops, the distance is just one hop. If the metric is propagation delay, the router can measure it directly with special ECHO packets that the receiver just timestamps and sends back as fast as it can.

As an example, assume that delay is used as a metric and that the router knows the delay to each of its neighbors. Once every T msec, each router sends to each neighbor a list of its estimated delays to each destination. It also receives a similar

list from each neighbor. Imagine that one of these tables has just come in from neighbor X, with X_i being X's estimate of how long it takes to get to router i. If the router knows that the delay to X is m msec, it also knows that it can reach router i via X in $X_i + m$ msec. By performing this calculation for each neighbor, a router can find out which estimate seems the best and use that estimate and the corresponding link in its new routing table. Note that the old routing table is not used in the calculation.

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

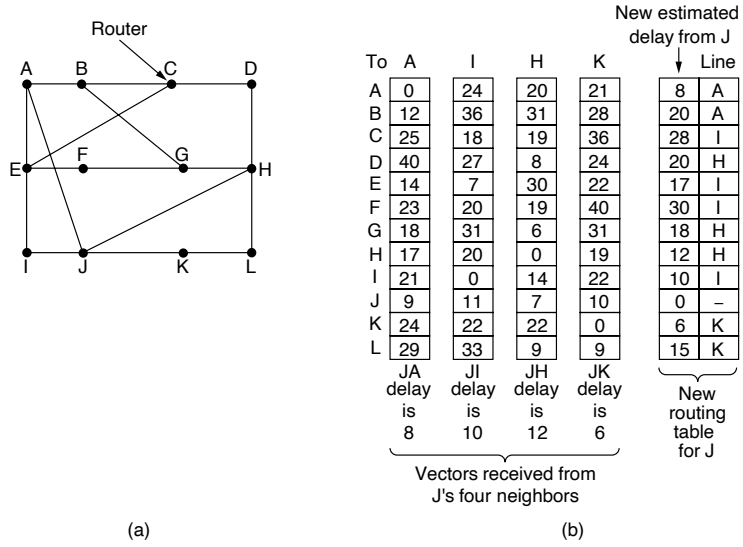


Figure 5-9. (a) A network. (b) Input from A, I, H, K, and the new routing table for J.

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

The Count-to-Infinity Problem

The settling of routes to best paths across the network is called **convergence**. Distance vector routing is useful as a simple technique by which routers can collectively compute shortest paths, but it has a serious drawback in practice: although it converges to the correct answer, it may do so slowly. In particular, it reacts rapidly to good news, but leisurely to bad news. Consider a router whose best route to destination *X* is long. If, on the next exchange, neighbor *A* suddenly reports a short delay to *X*, the router just switches over to using the line to *A* to send traffic to *X*. In one vector exchange, the good news is processed.

To see how fast good news propagates, consider the five-node (linear) network of Fig. 5-10, where the delay metric is the number of hops. Suppose *A* is down initially and all the other routers know this. In other words, they have all recorded the delay to *A* as infinity.

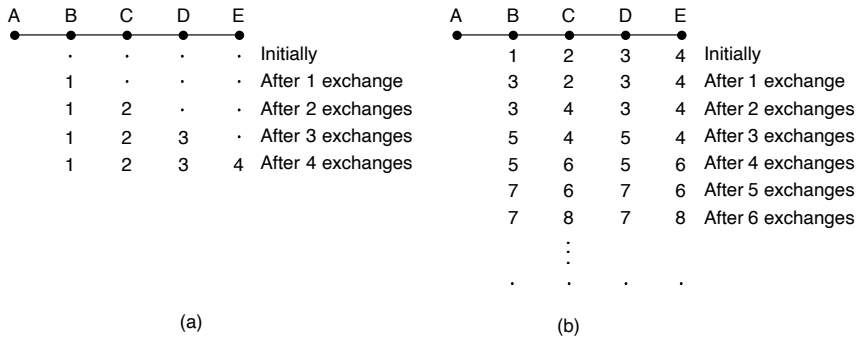


Figure 5-10. The count-to-infinity problem.

When *A* comes up, the other routers learn about it via the vector exchanges. For simplicity, we will assume that there is a gigantic gong somewhere that is struck periodically to initiate a vector exchange at all routers simultaneously. At the time of the first exchange, *B* learns that its left-hand neighbor has zero delay to *A*. *B* now makes an entry in its routing table indicating that *A* is one hop away to the left. All the other routers still think that *A* is down. At this point, the routing table entries for *A* are as shown in the second row of Fig. 5-10(a). On the next exchange, *C* learns that *B* has a path of length 1 to *A*, so it updates its routing table to indicate a path of length 2, but *D* and *E* do not hear the good news until later. Clearly, the good news is spreading at the rate of one hop per exchange. In a network whose longest path is of length *N* hops, within *N* exchanges everyone will know about newly revived links and routers.

Now let us consider the situation of Fig. 5-10(b), in which all the links and routers are initially up. Routers *B*, *C*, *D*, and *E* have distances to *A* of 1, 2, 3, and 4

hops, respectively. Suddenly, either *A* goes down or the link between *A* and *B* is cut (which is effectively the same thing from *B*'s point of view).

At the first packet exchange, *B* does not hear anything from *A*. Fortunately, *C* says ''Do not worry; I have a path to *A* of length 2.'' Little does *B* suspect that *C*'s path runs through *B* itself. For all *B* knows, *C* might have 10 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. Split horizon with poisoned reverse rule are discussed in RFC 1058. However, none of these heuristics work well in practice despite the colorful names. The core of the problem is that when *X* tells *Y* that it has a path somewhere, *Y* has no way of knowing whether it itself is on the path.

5.2.5 Link State Routing

Distance vector routing was used in the ARPANET until 1979, when it was replaced by link state routing. The primary problem that caused its demise was that the algorithm often took too long to converge after the network topology changed (due to the count-to-infinity problem). Consequently, it was replaced by an entirely new algorithm, now called **link state routing**. Variants of link state routing called IS-IS and OSPF are the routing algorithms that are most widely used inside large networks and the Internet today.

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

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

In effect, the complete topology is distributed to every router. Then Dijkstra's algorithm can be run at each router to find the shortest path to every other router. Below we will consider each of these five steps in more detail.

Learning about the Neighbors

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

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

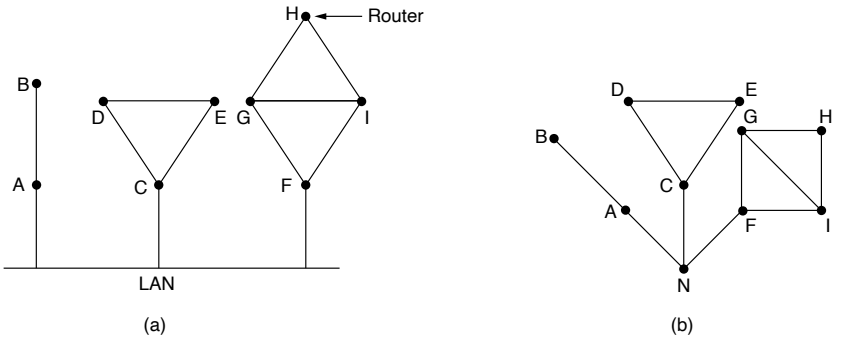


Figure 5-11. (a) Nine routers and a broadcast LAN. (b) A graph model of (a).

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

Setting Link Costs

The link state routing algorithm requires each link to have a distance or cost metric for finding shortest paths. The cost to reach neighbors can be set automatically, or configured by the network operator. A common choice is to make the cost inversely proportional to the bandwidth of the link. For example, 1-Gbps Ethernet may have a cost of 1 and 100-Mbps Ethernet may have a cost of 10. This makes higher-capacity paths better choices.

If the network is geographically spread out, the delay of the links may be factored into the cost so that paths over shorter links are better choices. The most direct way to determine this delay is to send over the line a special ECHO packet that the other side is required to send back immediately. By measuring the round-trip time and dividing it by two, the sending router can get an estimate of the delay.

Building Link State Packets

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

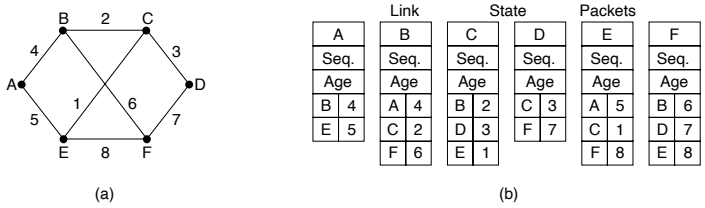


Figure 5-12. (a) A network. (b) The link state packets for this network.

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

Distributing the Link State Packets

The trickiest part of the algorithm is distributing the link state packets. All of the routers must get all of the link state packets quickly and reliably. If different routers are using different versions of the topology, the routes they compute can have inconsistencies, such as loops, unreachable machines, and other problems.

First, we will describe the basic distribution algorithm. After that, we will give some refinements. The fundamental idea is to use flooding to distribute the link state packets to all routers. To keep the flood in check, each packet contains a sequence number that is incremented for each new packet sent. Routers keep track of all the (source router, sequence) pairs they see. When a new link state packet comes in, it is checked against the list of packets already seen. If it is new, it is forwarded on all lines except the one it arrived on. If it is a duplicate, it is discarded. If a packet with a sequence number lower than the highest one seen so far ever arrives, it is rejected as being obsolete as the router has more recent data.

This algorithm has a few problems, but they are manageable. First, if the sequence numbers wrap around, confusion will reign. The solution here is to use a 32-bit sequence number. With one link state packet per second, it would take 137 years to wrap around, so this possibility can be ignored.

Second, if a router ever crashes, it will lose track of its sequence number. If it starts again at 0, the next packet it sends will be rejected as a duplicate.

Third, if a sequence number is ever corrupted and 65,540 is received instead of 4 (a 1-bit error), packets 5 through 65,540 will be rejected as obsolete, since the current sequence number will be thought to be 65,540.

The solution to these problems is to include the age of each packet after the sequence number and decrement it once a second. When the age hits zero, the information from that router is discarded. Normally, a new packet comes in, say, every 10 sec, so router information only times out when a router is down (or six consecutive packets have been lost, an unlikely event). The *Age* field is also decremented by each router during the initial flooding process, to make sure no packet can get lost and live for an indefinite period of time (a packet with age zero is discarded).

Some refinements to this algorithm make it more robust. When a link state packet comes in to a router for flooding, it is not queued for transmission immediately. Instead, it is put in a holding area to wait a short while in case more links are coming up or going down. If another link state packet from the same source comes in before the first packet is transmitted, their sequence numbers are compared. If they are equal, the duplicate is discarded. If they are different, the older one is thrown out. To guard against errors on the links, all link state packets are acknowledged.

The data structure used by router *B* for the network shown in Fig. 5-12(a) is depicted in Fig. 5-13. Each row here corresponds to a recently arrived, but as yet not fully processed, link state packet. The table records where the packet originated, its sequence number and age, and the data. In addition, there are send and acknowledgement flags for each of *B*'s three links (to *A*, *C*, and *F*, respectively). The send flags mean that the packet must be sent on the indicated link. The acknowledgement flags mean that it must be acknowledged there.

In Fig. 5-13, the link state packet from *A* arrives directly, so it must be sent to *C* and *F* and acknowledged to *A*, as indicated by the flag bits. Similarly, the packet from *F* has to be forwarded to *A* and *C* and acknowledged to *F*.

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

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

However, the situation with the third packet, from *E*, is different. It arrives twice, once via *EAB* and once via *EFB*. Consequently, it has to be sent only to *C* but must be acknowledged to both *A* and *F*, as indicated by the bits.

If a duplicate arrives while the original is still in the buffer, bits have to be changed. For example, if a copy of *C*'s state arrives from *F* before the fourth entry in the table has been forwarded, the six bits will be changed to 100011 to indicate that the packet must be acknowledged to *F* but not sent there.

Computing the New Routes

Once a router has accumulated a full set of link state packets, it can construct the entire network graph because every link is represented. Every link is, in fact, represented twice, once for each direction. The different directions may even have different costs. The shortest-path computations may then find different paths from router *A* to *B* than from router *B* to *A*.

Now Dijkstra's algorithm can be run locally to construct the shortest paths to all possible destinations. The results of this algorithm tell the router which link to use to reach each destination. This information is installed in the routing tables, and normal operation is resumed.

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

Link state routing is widely used in actual networks, so a few words about some example protocols are in order. Many ISPs use the **IS-IS** (**Intermediate System-to-Intermediate System**) link state protocol (Oran, 1990). It was designed

for an early network called DECnet, later adopted by ISO for use with the OSI protocols and then modified to handle other protocols as well, most notably, IP. OSPF (Open Shortest Path First), which will be discussed in Sec. 5.7.6, is the other main link state protocol. It was designed by IETF several years after IS-IS and adopted many of the innovations designed for IS-IS. These innovations include a self-stabilizing method of flooding link state updates, the concept of a designated router on a LAN, and the method of computing and supporting path splitting and multiple metrics. As a consequence, there is very little difference between IS-IS and OSPF. The most important difference is that IS-IS can carry information about multiple network layer protocols at the same time (e.g., IP, IPX, and AppleTalk). OSPF does not have this feature, and it is an advantage in large multiprotocol environments.

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

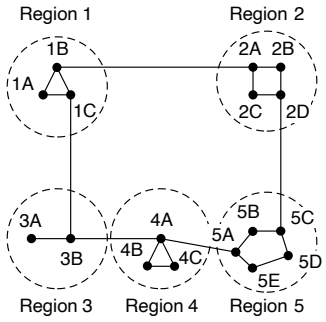
5.2.6 Hierarchical Routing within a Network

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. Additionally, even if every router could store the entire topology, recomputing shortest paths every time the network experienced changes in the topology would be prohibitive; imagine, for example, if a very large network would need to computer shortest paths every time a link in the network failed or recovered. At a certain point, the network may grow to a size 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, through the use of **routing areas**.

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



(a)

Full table for 1A

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

(b)

Hierarchical table for 1A

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

(c)

Figure 5-14. Hierarchical routing.

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.

5.2.7 Broadcast Routing

In some applications, hosts need to send messages to many or all other hosts. For example, a service distributing weather reports, stock market updates, or live radio programs might work best by sending to all machines and letting those that are interested read the data. Sending a packet to all destinations simultaneously is called **broadcasting**. Various methods have been proposed for doing it.

One broadcasting method that requires no special features from the network is for the source to simply send a distinct packet to each destination. Not only is the method wasteful of bandwidth and slow, but it also requires the source to have a complete list of all destinations. This method is not desirable in practice, even though it is widely applicable.

An improvement is **multidestination routing**, in which each packet contains either a list of destinations or a bit map indicating the desired destinations. When a packet arrives at a router, the router checks all the destinations to determine the set of output lines that will be needed. (An output line is needed if it is the best route to at least one of the destinations.) The router generates a new copy of the packet for each output line to be used and includes in each packet only those destinations that are to use the line. In effect, the destination set is partitioned among the output lines. After a sufficient number of hops, each packet will carry only one destination like a normal packet. Multidestination routing is like using separately addressed packets, except that when several packets must follow the same route, one of them pays full fare and the rest ride free. The network bandwidth is therefore used more efficiently. However, this scheme still requires the source to know all the destinations, plus it is as much work for a router to determine where to send one multidestination packet as it is for multiple distinct packets.

We have already seen a better broadcast routing technique: flooding. When implemented with a sequence number per source, flooding uses links efficiently

with a decision rule at routers that is relatively simple. Although flooding is ill-suited for ordinary point-to-point communication, it rates serious consideration for broadcasting. However, it turns out that we can do better still once the shortest path routes for regular packets have been computed.

The idea for **reverse path forwarding** is elegant and remarkably simple once it has been pointed out (Dalal and Metcalfe, 1978). When a broadcast packet arrives at a router, the router checks to see if the packet arrived on the link that is normally used for sending packets *toward* the source of the broadcast. If so, there is an excellent chance that the broadcast packet itself followed the best route from the router and is therefore the first copy to arrive at the router. This being the case, the router forwards copies of it onto all links except the one it arrived on. If, however, the broadcast packet arrived on a link other than the preferred one for reaching the source, the packet is discarded as a likely duplicate.

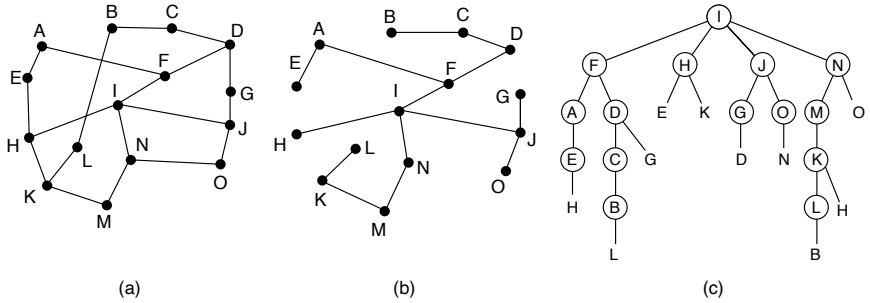


Figure 5-15. Reverse path forwarding. (a) A network. (b) Sink tree for router *I*. (c) The tree built by reverse path forwarding from *I*.

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

The principal advantage of reverse path forwarding is that it is efficient while being easy to implement. It sends the broadcast packet over each link only once in each direction, just as in flooding, yet it requires only that routers know how to

reach all destinations, without needing to remember sequence numbers (or use other mechanisms to stop the flood) or list all destinations in the packet.

Our last broadcast algorithm improves on the behavior of reverse path forwarding. It makes explicit use of the sink tree—or any other convenient spanning tree for that matter—for the router initiating the broadcast. A **spanning tree** is a subset of the network that includes all the routers but contains no loops. Sink trees are spanning trees. If each router knows which of its lines belong to the spanning tree, it can copy an incoming broadcast packet onto all the spanning tree lines except the one it arrived on. This method makes excellent use of bandwidth, generating the absolute minimum number of packets necessary to do the job. In Fig. 5-15, for example, when the sink tree of part (b) is used as the spanning tree, the broadcast packet is sent with the minimum 14 packets. The only problem is that each router must have knowledge of some spanning tree for the method to be applicable. Sometimes this information is available (e.g., with link state routing, all routers know the complete topology, so they can compute a spanning tree) but sometimes it is not (e.g., with distance vector routing).

5.2.8 Multicast Routing

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

Sending a message to such a group is called **multicasting**, and the routing algorithm used is called **multicast routing**. All multicasting schemes require some way to create and destroy groups and to identify which routers are members of a group. How these tasks are accomplished is not of concern to the routing algorithm. For now, we will assume that each group is identified by a multicast address and that routers know the groups to which they belong. We will revisit group membership when we describe Internet multicasting in Sec. 5.7.8.

Multicast routing schemes build on the broadcast routing schemes we have already studied, sending packets along spanning trees to deliver the packets to the members of the group while making efficient use of bandwidth. However, the best spanning tree to use depends on whether the group is dense, with receivers scattered over most of the network, or sparse, with much of the network not belonging to the group. In this section we will consider both cases.

If the group is dense, broadcast is a good start because it efficiently gets the packet to all parts of the network. But broadcast will reach some routers that are

not members of the group, which is wasteful. The solution explored by Deering and Cheriton (1990) is to prune the broadcast spanning tree by removing links that do not lead to members. The result is an efficient multicast spanning tree.

As an example, consider the two groups, 1 and 2, in the network shown in Fig. 5-16(a). Some routers are attached to hosts that belong to none, one or both of these groups, as indicated in the figure. A spanning tree for the leftmost router is shown in Fig. 5-16(b). This tree can be used for broadcast but is overkill for multicast, as can be seen from the two pruned versions that are shown next. In Fig. 5-16(c), all the links that do not lead to hosts that are members of group 1 have been removed. The result is the multicast spanning tree for the leftmost router to send to group 1. Packets are forwarded only along this spanning tree, which is more efficient than the broadcast tree because there are 7 links instead of 10. Fig. 5-16(d) shows the multicast spanning tree after pruning for group 2. It is efficient too, with only five links this time. It also shows that different multicast groups have different spanning trees.

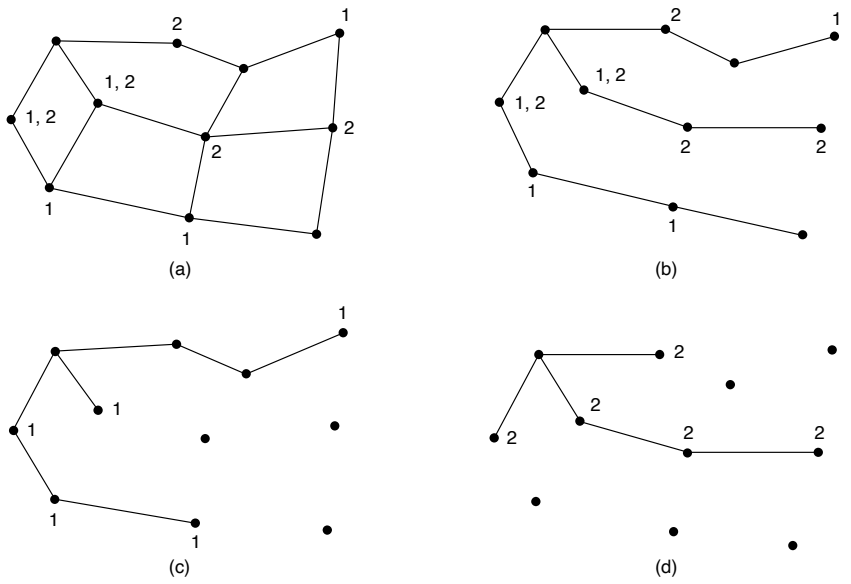


Figure 5-16. (a) A network. (b) A spanning tree for the leftmost router. (c) A multicast tree for group 1. (d) A multicast tree for group 2.

Various ways of pruning the spanning tree are possible. The simplest one can be used if link state routing is used and furthermore each router is aware of the complete topology, including which hosts belong to which groups. Each router can

then construct its own pruned spanning tree for each sender to the group in question by constructing a sink tree for the sender as usual and then removing all links that do not connect group members to the sink node. **MOSPF (Multicast OSPF)** is an example of a link state protocol that works in this way (Moy, 1994).

With distance vector routing, a different pruning strategy can be followed. The basic algorithm is reverse path forwarding. However, whenever a router with no hosts interested in a particular group and no connections to other routers receives a multicast message for that group, it responds with a PRUNE message, telling the neighbor that sent the message not to send it any more multicasts from the sender for that group. When a router with no group members among its own hosts has received such messages on all the lines to which it sends the multicast, it, too, can respond with a PRUNE message. In this way, the spanning tree is recursively pruned. **DVMRP (Distance Vector Multicast Routing Protocol)** is an example of a multicast routing protocol that works this way (Waitzman et al., 1988).

Pruning results in efficient spanning trees that use only the links that are actually needed to reach members of the group and no others. One potential disadvantage is that it is lots of work for routers, especially for very big networks. Suppose that a network has n groups, each with an average of m nodes. At each router and for each group m pruned spanning trees must be stored, for a total of mn trees. For example, Fig. 5-16(c) gives the spanning tree for the leftmost router to send to group 1. The spanning tree for the rightmost router to send to group 1 (not shown in the figure) will look quite different, as packets will head directly for group members rather than via the left side of the graph. This in turn means that routers must forward packets destined to group 1 in different directions depending on which node is sending to the group. When many large groups with many senders exist, considerable storage is needed to store all the trees.

An alternative design uses **core-based trees** to compute a single spanning tree for the group (Ballardie et al., 1993). All of the routers agree on a root (called the **core** or **rendezvous point**) and build the tree by sending a packet from each member to the root. The tree is the union of the paths traced by these packets. Fig. 5-17(a) shows a core-based tree for group 1. To send to this group, a sender sends a packet to the core. When the packet reaches the core, it is forwarded down the tree. This is shown in Fig. 5-17(b) for the sender on the righthand side of the network. As a performance optimization, packets destined for the group do not need to reach the core before they are multicast. As soon as a packet reaches the tree, it can be forwarded up toward the root, as well as down all the other branches. This is the case for the sender at the top of Fig. 5-17(b).

Having a shared tree is not optimal for all sources. For example, in Fig. 5-17(b), the packet from the sender on the righthand side reaches the top-right group member via the core in three hops, instead of directly. The inefficiency depends on where the core and senders are located, but often it is reasonable when the core is in the middle of the senders. When there is only a single sender, as in a video that is streamed to a group, using the sender as the core is optimal.

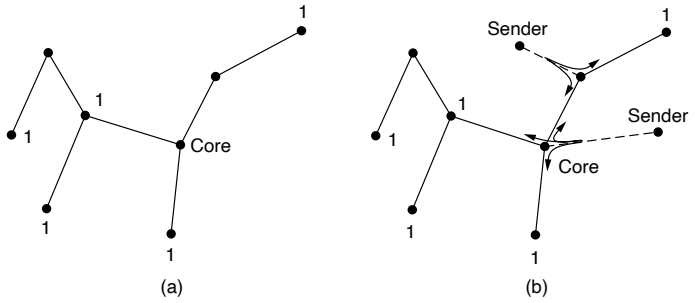


Figure 5-17. (a) Core-based tree for group 1. (b) Sending to group 1.

Also of note is that shared trees can be a major savings in storage costs, messages sent, and computation. Each router has to keep only one tree per group, instead of *m* trees. Further, routers that are not part of the tree do no work at all to support the group. For this reason, shared tree approaches like core-based trees are used for multicasting to sparse groups in the Internet as part of popular protocols such as protocol independent multicast (Fenner et al., 2006).

5.2.9 Anycast Routing

So far, we have covered delivery models in which a source sends to a single destination (called **unicast**), to all destinations (called broadcast), and to a group of destinations (called multicast). Another delivery model, called **anycast** is sometimes also useful. In anycast, a packet is delivered to the nearest member of a group (Partridge et al., 1993). Schemes that find these paths are called **anycast routing**.

Why would we want anycast? Sometimes nodes provide a service, such as time of day or content distribution for which it is getting the right information that matters, not the node that is contacted; any node will do. For example, anycast is used in the Internet as part of DNS, as we will see in Chap. 7.

Fortunately, regular distance vector and link state routing can produce anycast routes, so we do not need to devise a new routing scheme for anycast. Suppose we want to anycast to the members of group 1. They will all be given the address "1," instead of different addresses. Distance vector routing will distribute vectors as usual, and nodes will choose the shortest path to destination 1. This will result in nodes sending to the nearest instance of destination 1. The routes are shown in Fig. 5-18(a). This procedure works because the routing protocol does not realize that there are multiple instances of destination 1. That is, it believes that all the instances of node 1 are the same node, as in the topology shown in Fig. 5-18(b).

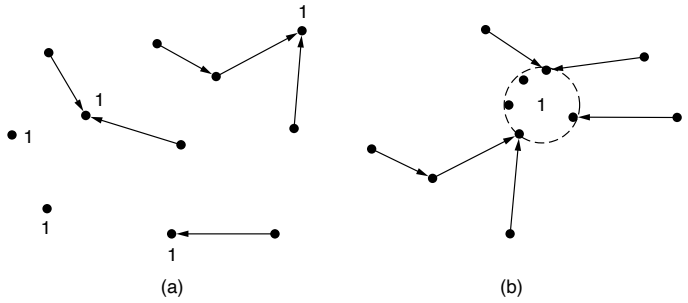


Figure 5-18. (a) Anycast routes to group 1. (b) Topology seen by the routing protocol.

This procedure works for link state routing as well, although there is the added consideration that the routing protocol must not find seemingly short paths that pass through node 1. This would result in jumps through hyperspace, since the instances of node 1 are really nodes located in different parts of the network. However, link state protocols already make this distinction between routers and hosts. We glossed over this fact earlier because it was not needed for our discussion.

5.3 TRAFFIC MANAGEMENT AT THE NETWORK LAYER

Too many packets in any part of the network can ultimately introduce packet delay and loss that degrades performance. This situation is called **congestion**.

5.3.1 The Need for Traffic Management: Congestion

The network and transport layers share the responsibility for managing congestion. Because 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. 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. The network layer does not automatically mitigate congestion, but network operators can configure routers, switches, and other devices at the network layer to mitigate the effects of congestion, typically by taking actions that would encourage a sender to reduce the sending rate, or by sending traffic along different, less-congested paths through the network. In this chapter we will look at the aspects of congestion that concern the network layer, and mechanisms that the network layer uses to control and manage congestion. To avoid confusion with the more common use of the phase ''congestion control,'' which is frequently used by some authors to describe functions of the transport layer, in this chapter we

will discuss practices to manage congestion at the network layer as **congestion management** or **traffic management**. In Chap. 6, we will finish the topic by covering the mechanisms that the transport layer uses to manage congestion control.

Figure 5-19 shows the onset of congestion. When the number of packets that hosts send into the network is well within the network's capacity, the amount of traffic that is delivered is proportional to the amount of traffic that is sent: If twice as much traffic is sent, twice as much is 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. At this point, the network is experiencing congestion.

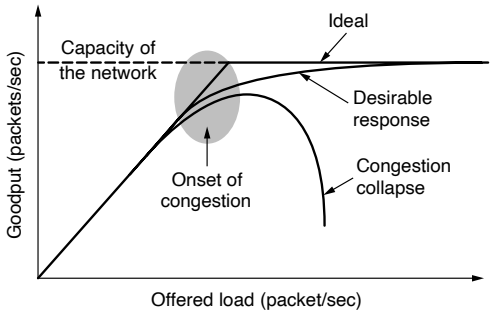


Figure 5-19. Performance drops significantly in the presence of congestion: packet loss rates increase, and latency also increases as router queues fill with packets.

At some point, the network may experience a **congestion collapse**, where performance plummets as the offered load increases beyond the capacity. In short, congestion collapse occurs when increasing load on the network actually results in less traffic being successfully delivered. This situation can occur if packets are 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. 5-19 is given as **goodput**, 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 somehow do become congested. Unfortunately, in a packet-switched network, 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 frequently gets worse, not better. More recently, researchers discovered that many network devices tend to have more memory than they need, a concept that became known as **bufferbloat**. Network devices that have too much memory can degrade network performance for a variety of reasons. First, by the time packets get to the front of the queue, they have already timed out (repeatedly) and duplicates have been sent. Second, as we will discuss in Chap. 6, senders need timely information about network congestion, and if packets are stored in router buffers, rather than dropped, then senders will continue to send traffic that congests the network. All of this makes matters worse, not better—it leads to congestion collapse.

Low-bandwidth links or routers that process packets more slowly than the capacity of a network link can also become congested. In cases where the network has additional capacity in other parts of the network, congestion can be mitigated by directing some of the traffic away from the bottleneck to other (less congested) parts of the network. Ultimately, however, increasing traffic demands may result in congestion being pervasive throughout the network. When this occurs, there are two approaches that operators can take: shedding load (i.e., dropping traffic), or provisioning additional capacity.

It is worth pointing out the difference between **congestion control**, **traffic management**, and **flow control**, as the relationship is a subtle one. Traffic management (sometimes also called traffic engineering) has to do with making sure the network is able to carry the offered traffic; it can be performed by devices in the network, or by the senders of traffic (often through mechanisms in the transport protocol, which are often referred to as congestion control). Congestion management and control concerns the behavior of all the hosts and routers. Flow control, in contrast, relates to the traffic between a particular sender and a particular receiver and is generally concerned with making sure that the sender is not transmitting data faster than the receiver can process it. Its job is to make sure no data is lost because the sender is more powerful than the receiver and can send data faster that the receiver can 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 a 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 management by looking at the approaches that network operators can apply at different time scales. Then we will look at approaches that can prevent congestion from occurring in the first place, followed by approaches for coping with it once it has set in.

5.3.2 Approaches to Traffic Management

The presence of congestion means that the load is (temporarily) greater than the resources (in a part of the network) can handle. There are two approaches to dealing with it: increase the resources or decrease the load. As shown in Fig. 5-20, these solutions are usually applied on different time scales to either prevent congestion or react to it once it has occurred.

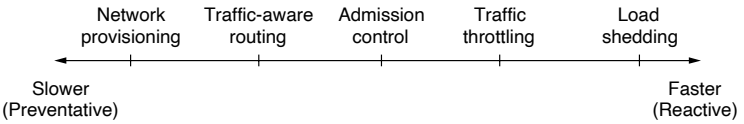


Figure 5-20. Timescales of approaches to traffic and congestion management.

The most straightforward way to avoid congestion is to build a network that is provisioned for the traffic load that it must carry. If there is a low-bandwidth link on the path along which most traffic is directed, congestion is likely. Sometimes resources can be added dynamically when there is serious congestion, for example, turning 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 users 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 can also be helpful.

However, sometimes it is not possible to increase capacity, especially on short time scales. 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 one example of **admission control**, a concept that simply denies senders the ability to send traffic if the network capacity cannot support it.

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 slow down the sending rates, or it can simply slow down the traffic itself, a process sometimes referred to as **throttling**. 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, queueing delay, or packet loss and send feedback to senders, either explicitly or implicitly (e.g., by dropping packets) to tell them to slow down.

In the case where feedback is explicit, 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 μsec, 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.

Another approach is for the network to discard packets that it cannot deliver. The general name for this approach is load shedding, and there are various ways to achieve it, including traffic shaping (restricting the transmission rate for a particular sender) and traffic policing (dropping traffic from a particular sender if it exceeds some rate). A good policy for choosing which packets to discard can help to prevent congestion collapse. We will discuss all of these topics below.

Traffic-Aware Routing

The first approach we will examine is traffic-aware routing. The routing approaches we looked at in Sec. 5.2 used fixed link weights that adapted to changes in topology, but not to changes in traffic 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 queueing 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. 5-21, which is divided into two parts, East and West, connected by two links, *CF* and *EI*. Suppose that most of the East-West traffic is using link *CF*, resulting

in this link being heavily loaded with long delays. Including queueing delay in the weight used for the shortest path calculation will make *EI* more attractive. After the new routing tables have been installed, most of the East-West traffic will now go over *EI*, loading this link. Consequently, in the next update, *CF* will appear to be the shortest path. As a result, the routing tables may oscillate wildly, leading to erratic routing and many potential problems.

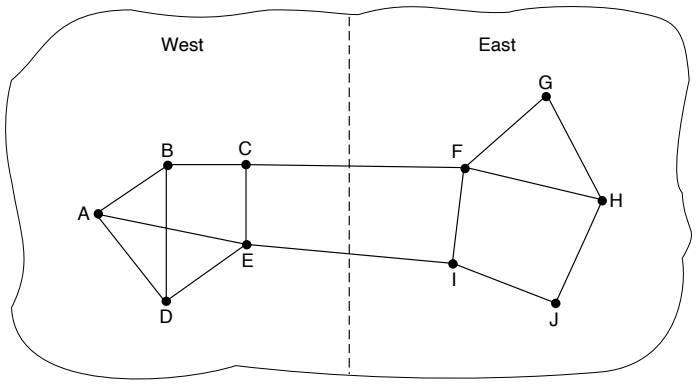


Figure 5-21. A network in which the East and West parts are connected by two links.

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, network operators make adjustments to routing protocols on slower time scales by slowly changing the routing configuration and parameters, a process sometimes called traffic engineering. Traffic engineering has long been a painstaking, manual process, akin to a black art. Some work has attempted to formalize this process, but Internet traffic loads are unpredictable enough, and the protocol configuration parameters are coarse and clunky enough that the process has remained fairly primitive. More recently, however, the advent of software defined networking has made it possible to automate some of these tasks, and the increasing use of certain technologies such as MPLS tunnels across the network has provided operators with more flexibility for a wide range of traffic engineering tasks.

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 approach 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. Because these are two common mechanisms for performing traffic shaping, we will cover these topics in more detail in that section.

Given 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 shortly.

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 be combined with traffic-aware routing by considering routes around traffic hotspots as part of the setup procedure. For example, consider the network of Fig. 5-22(a), in which two routers are congested, as indicated.

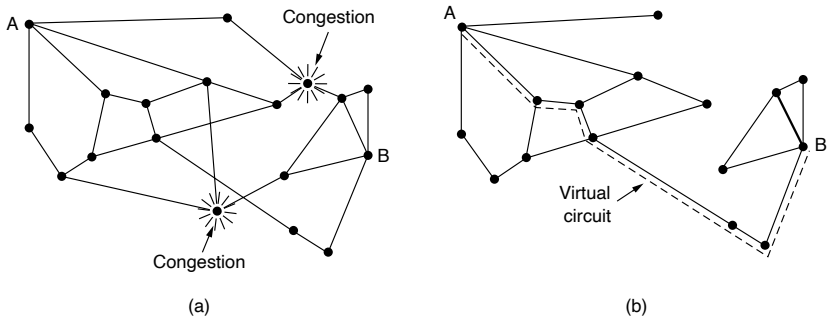


Figure 5-22. (a) A congested network. (b) The portion of the network that is not congested. A virtual circuit from *A* to *B* is also shown.

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. 5-22(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.

Load Shedding

When none of the above methods make the congestion disappear, routers can bring out the heavy artillery: **load shedding**. This is a fancy way of saying that when routers are being inundated by packets that they cannot handle, they just throw them away. The term comes from the world of electrical power generation, where it refers to the practice of utilities intentionally blacking out certain areas to save the entire grid from collapsing on hot summer days when the demand for electricity (to power air conditioners) greatly exceeds the supply.

The key question for a router drowning in packets is which packets to drop. The preferred choice may depend on the type of applications that use the network. For a file transfer, an old packet is worth more than a new one. This is because dropping packet 6 and keeping packets 7 through 10, for example, will only force the receiver to do more work to buffer data that it cannot yet use. In contrast, for real-time media, a new packet is worth more than an old one. This is because packets become useless if they are delayed and miss the time at which they must be played out to the user.

The former policy (old is better than new) is often called **wine** and the latter (new is better than old) is often called **milk** because most people prefer new milk over old milk and old wine over new wine.

More intelligent load shedding requires cooperation from the senders. An example is packets that carry routing information. These packets are more important than regular data packets because they establish routes; if they are lost, the network may lose connectivity. Another example is that algorithms for compressing video, like MPEG, periodically transmit an entire frame and then send subsequent frames as differences from the last full frame. In this case, dropping a packet that is part of a difference is preferable to dropping one that is part of a full frame because future packets depend on the full frame.

To implement an intelligent discard policy, applications must mark their packets to indicate to the network how important they are. Then, when packets have to be discarded, routers can first drop packets from the least important class, then the next most important class, and so on.

Of course, unless there is some significant incentive to avoid marking every packet as VERY IMPORTANT—NEVER, EVER DISCARD, nobody will do it. Often accounting and money are used to discourage frivolous marking. For example, the network might let senders transmit faster than the service they purchased allows if they mark excess packets as low priority. Such a strategy is actually not a bad idea because it makes more efficient use of idle resources, allowing hosts to use them as long as nobody else is interested, but without establishing a right to them when times get tough.

Traffic Shaping

Before the network can make performance guarantees, it must know what traffic is being guaranteed. In the telephone network, this characterization is simple. For example, a voice call (in uncompressed format) needs 64 kbps and consists of one 8-bit sample every 125 μsec. However, traffic in data networks is **bursty**. It typically arrives at nonuniform rates as the traffic rate varies (e.g., videoconferencing with compression), users interact with applications (e.g., browsing a new Web page), and computers switch between tasks. Bursts of traffic are more difficult to handle than constant-rate traffic because they can fill buffers and cause packets to be lost.

Traffic shaping is a technique for regulating the average rate and burstiness of a flow of data that enters the network. The goal is to allow applications to transmit a wide variety of traffic that suits their needs, including some bursts, yet have a simple and useful way to describe the possible traffic patterns to the network. When a flow is set up, the user and the network (i.e., the customer and the provider) agree on a certain traffic pattern (i.e., shape) for that flow. In effect, the customer says to the provider ''My transmission pattern will look like this; can you handle it?''

Sometimes this agreement is called an **SLA** (**Service Level Agreement**), especially when it is made over aggregate flows and long periods of time, such as all of

the traffic for a given customer. As long as the customer fulfills her part of the bargain and only sends packets according to the agreed-on contract, the provider promises to deliver them all in a timely fashion.

Traffic shaping reduces congestion and thus helps the network live up to its promise. However, to make it work, there is also the issue of how the provider can tell if the customer is following the agreement and what to do if the customer is not. Packets in excess of the agreed pattern might be dropped by the network, or they might be marked as having lower priority. Monitoring a traffic flow is called **traffic policing**.

Shaping and policing are not so important for peer-to-peer and other transfers that will consume any and all available bandwidth, but they are of great importance for real-time data, such as audio and video connections, which have stringent quality-of-service requirements. We have already seen one way to limit the amount of data an application sends: the sliding window, which uses one parameter to limit how much data is in transit at any given time, which indirectly limits the rate. Now we will look at a more general way to characterize traffic, with the leaky bucket and token bucket algorithms. The formulations are slightly different but give an equivalent result.

Try to imagine a bucket with a small hole in the bottom, as illustrated in Fig. 5-23(b). No matter the rate at which water enters the bucket, the outflow is at a constant rate, R, when there is any water in the bucket and zero when the bucket is empty. Also, once the bucket is full to capacity B, any additional water entering it spills over the sides and is lost.

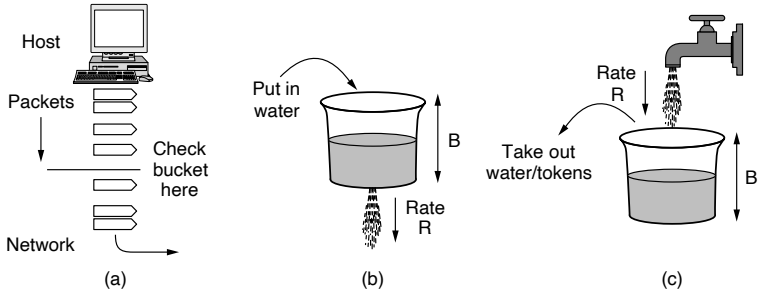


Figure 5-23. (a) Shaping packets. (b) A leaky bucket. (c) A token bucket.

This bucket can be used to shape or police packets entering the network, as shown in Fig. 5-23(a). Conceptually, each host is connected to the network by an interface containing a leaky bucket. To send a packet into the network, it must be possible to put more water into the bucket. If a packet arrives when the bucket is full, the packet must either be queued until enough water leaks out to hold it or be discarded. The former might happen at a host shaping its traffic for the network as part of the operating system. The latter might happen in hardware at a provider

network interface that is policing traffic entering the network. This technique was proposed by Turner (1986) and is called the **leaky bucket algorithm**.

A different but equivalent formulation is to imagine the network interface as a bucket that is being filled, as shown in Fig. 5-23(c). The tap is running at rate R and the bucket has a capacity of B, as before. Now to send a packet we must be able to take water, or tokens, as the contents are commonly called, out of the bucket (rather than putting water into the bucket). No more than a fixed number of tokens, B, can accumulate in the bucket, and if the bucket is empty, we must wait until more tokens arrive before we can send another packet. This algorithm is called the **token bucket algorithm**.

Leaky and token buckets limit the long-term rate of a flow but allow short-term bursts up to a maximum regulated length to pass through unaltered and without suffering any artificial delays. Large bursts will be smoothed by a leaky bucket traffic shaper to reduce congestion in the network. As an example, imagine that a computer can produce data at up to 1000 Mbps (125 million bytes/sec) and that the first link of the network also runs at this speed. The pattern of traffic the host generates is shown in Fig. 5-24(a). This pattern is bursty. The average rate over one second is 200 Mbps, even though the host sends a burst of 16,000 KB at the top speed of 1000 Mbps (for 1/8 of the second).

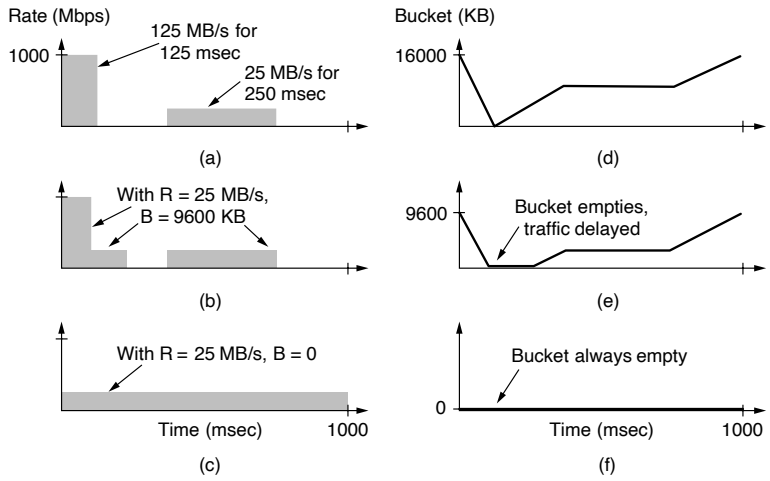


Figure 5-24. (a) Traffic from a host. Output shaped by a token bucket of rate 200 Mbps and capacity (b) 9600 KB and (c) 0 KB. Token bucket level for shaping with rate 200 Mbps and capacity (d) 16,000 KB, (e) 9600 KB, and (f) 0 KB.

Now suppose that the routers can accept data at the top speed only for short intervals, until their buffers fill up. The buffer size is 9600 KB, smaller than the

traffic burst. For long intervals, the routers work best at rates not exceeding 200 Mbps (say, because this is all the bandwidth given to the customer). The implication is that if traffic is sent in this pattern, some of it will be dropped in the network because it does not fit into the buffers at routers.

To avoid this packet loss, we can shape the traffic at the host with a token bucket. If we use a rate, R, of 200 Mbps and a capacity, B, of 9600 KB, the traffic will fall within what the network can handle. The output of this token bucket is shown in Fig. 5-24(b). The host can send full throttle at 1000 Mbps for a short while until it has fully drained the bucket. Then it has to cut back to 200 Mbps until the burst has been sent. The effect is to spread out the burst over time because it was too large to handle all at once. The level of the token bucket is shown in Fig. 5-24(e). It starts off full and is depleted by the initial burst. When it reaches zero, new packets can be sent only at the rate at which the buffer is filling; there can be no more bursts until the bucket has recovered. The bucket fills when no traffic is being sent and stays flat when traffic is being sent at the fill rate.

We can also shape the traffic to be less bursty. Fig. 5-24(c) shows the output of a token bucket with R = 200 Mbps and a capacity of 0. This is the extreme case in which the traffic has been completely smoothed. No bursts are allowed, and the traffic enters the network at a steady rate. The corresponding bucket level, shown in Fig. 5-24(f), is always empty. Traffic is being queued on the host for release into the network and there is always a packet waiting to be sent when it is allowed.

Finally, Fig. 5-24(d) illustrates the bucket level for a token bucket with R = 200 Mbps and a capacity of B = 16,000 KB. This is the smallest token bucket through which the traffic passes unaltered. It might be used at a router in the network to police the traffic that the host sends. However, if the host is sending traffic that conforms to the token bucket on which it has agreed with the network, the traffic will fit through that same token bucket run at the router at the edge of the network. If the host sends at a faster or burstier rate, the token bucket will run out of water. If this happens, a traffic policer will know that the traffic is not as was described. It will then either drop the excess packets or lower their priority, depending on the design of the network. In our example, the bucket empties only momentarily, at the end of the initial burst, then recovers enough for the next burst.

Leaky and token buckets are easy to implement. We will now describe the operation of a token bucket. Even though we have described water flowing continuously into and out of the bucket, real implementations must work with discrete quantities. A token bucket is implemented with a counter for the level of the bucket. The counter is advanced by $R/\Delta T$ units at every clock tick of ΔT seconds. This would be 200 Kbit every 1 msec in our example above. Every time a unit of traffic is sent into the network, the counter is decremented, and traffic may be sent until the counter reaches zero.

When the packets are all the same size, the bucket level can just be counted in packets (e.g., 200 Kbit is 20 packets of 1250 bytes). However, often variable-sized packets are used. In this case, the bucket level can be counted in bytes. If the

residual byte count is too low to send a large packet, the packet must wait until the next tick (or even longer, if the fill rate is small).

Calculating the length of the maximum burst (until the bucket empties) is slightly tricky. It is longer than just 9600 KB divided by 125 MB/sec because while the burst is being output, more tokens arrive. If we call the burst length S sec., the maximum output rate M bytes/sec, the token bucket capacity B bytes, and the token arrival rate R bytes/sec, we can see that an output burst contains a maximum of $B + RS$ bytes. We also know that the number of bytes in a maximum-speed burst of length S seconds is MS. Hence, we have

$$B + RS = MS$$

We can solve this equation to get $S = B/(M - R)$. For our parameters of B = 9600 KB, M = 125 MB/sec, and R = 25 MB/sec, we get a burst time of about 94 msec.

A potential problem with the token bucket algorithm is that it reduces large bursts down to the long-term rate R. It is frequently desirable to reduce the peak rate, but without going down to the long-term rate (and also without raising the long-term rate to allow more traffic into the network). One way to get smoother traffic is to insert a second token bucket after the first one. The rate of the second bucket should be much higher than the first one. Basically, the first bucket characterizes the traffic, fixing its average rate but allowing some bursts. The second bucket reduces the peak rate at which the bursts are sent into the network. For example, if the rate of the second token bucket is set to be 500 Mbps and the capacity is set to 0, the initial burst will enter the network at a peak rate of 500 Mbps, which is lower than the 1000 Mbps rate we had previously.

Using all of these buckets can be a bit tricky. When token buckets are used for traffic shaping at hosts, packets are queued and delayed until the buckets permit them to be sent. When token buckets are used for traffic policing at routers in the network, the algorithm is simulated to make sure that no more packets are sent than permitted. Nevertheless, these tools provide ways to shape the network traffic into more manageable forms to assist in meeting quality-of-service requirements.

Active Queue Management

In the Internet and many other computer networks, senders adjust their transmissions to send as much traffic as the network can readily deliver. In this setting, the network aims to operate just before the onset of congestion. When congestion is imminent, it must tell the senders to throttle back their transmissions and slow down. This feedback is business as usual rather than an exceptional situation. The term **congestion avoidance** is sometimes used to contrast this operating point with the one in which the network has become (overly) congested.

Let us now look at some approaches to throttling traffic that can be used in both datagram networks and virtual-circuit networks alike. Each approach must solve two problems. First, routers must determine when congestion is approaching,

ideally before it has arrived. To do so, each router can continuously monitor the resources it is using. Three possibilities are the utilization of the output links, the buffering of queued packets inside the router, and the number of packets that are lost due to insufficient buffering. Of these possibilities, the second one is the most useful. Averages of utilization do not directly account for the burstiness of most traffic—a utilization of 50% may be low for smooth traffic and too high for highly variable traffic. Counts of packet losses come too late. Congestion has already set in by the time that packets are lost.

The queueing delay inside routers directly captures any congestion experienced by packets. It should be low most of time, but will jump when there is a burst of traffic that generates a backlog. To maintain a good estimate of the queueing delay, d, a sample of the instantaneous queue length, s, can be made periodically and d updated according to

$$d_{\text{new}} = \alpha d_{\text{old}} + (1 - \alpha)s$$

where the constant α determines how fast the router forgets recent history. This is called an **EWMA** (**Exponentially Weighted Moving Average**). It smoothes out fluctuations and is equivalent to a low-pass filter. Whenever d moves above some predefined threshold, the router notes the onset of congestion.

The second problem is that routers must deliver timely feedback to the senders that are causing the congestion. Congestion is experienced in the network, but relieving congestion requires action on behalf of the senders that are using the network. To deliver feedback, the router must identify the appropriate senders. It must then warn them carefully, without sending many more packets into the already congested network. Different schemes use different feedback mechanisms, as we will now describe.

Random Early Detection

Dealing with congestion when it first starts is more effective than letting it gum up the works and then trying to deal with it. This observation leads to an interesting twist on load shedding, which is to discard packets before all the buffer space is really exhausted.

The motivation for this idea is that most Internet hosts do not yet get congestion signals from routers in the form of an explicit notification. Instead, the only reliable indication of congestion that hosts get from the network is packet loss. After all, it is difficult to build a router that does not drop packets when it is completely overloaded. Transport protocols such as TCP are thus hardwired to react to loss as congestion, slowing down the source in response. The reasoning behind this logic is that TCP was designed for wired networks and wired networks are very reliable, so lost packets are mostly due to buffer overruns rather than transmission errors. Wireless links must recover transmission errors at the link layer (so they are not seen at the network layer) to work well with TCP.

This situation can be exploited to help reduce congestion. By having routers drop packets early, before the situation has become hopeless, there is time for the source to take action before it is too late. A popular algorithm for doing this is called **RED** (**Random Early Detection**) (Floyd and Jacobson, 1993). To determine when to start discarding, routers maintain a running average of their queue lengths. When the average queue length on some link exceeds a threshold, the link is said to be congested and a small fraction of the packets are dropped at random. Picking packets at random makes it more likely that the fastest senders will see a packet drop; this is the best option since the router cannot tell which source is causing the most trouble in a datagram network. The affected sender will notice the loss when there is no acknowledgement, and then the transport protocol will slow down. The lost packet is thus delivering the same message as a notification packet, but implicitly, without the router sending any explicit signal.

RED routers improve performance compared to routers that drop packets only when their buffers are full, though they require tuning to work well. For example, the ideal number of packets to drop depends on how many senders need to be notified of congestion. However, explicit notification is the better option if it is available. It works in exactly the same manner, but delivers a congestion signal explicitly rather than as a loss; RED is used when hosts cannot receive explicit signals.

Choke Packets

The most direct way to notify a sender of congestion is to tell it directly. In this approach, the router selects a congested packet and sends a **choke packet** back to the source host, giving it the destination found in the packet. The original packet may be tagged (a header bit is turned on) so that it will not generate any more choke packets farther along the path and then forwarded in the usual way. To avoid increasing load on the network during a time of congestion, the router may only send choke packets at a low rate.

When the source host gets the choke packet, it is required to reduce the traffic sent to the specified destination, for example, by 50%. In a datagram network, simply picking packets at random when there is congestion is likely to cause choke packets to be sent to fast senders, because they will have the most packets in the queue. The feedback created by this protocol can help prevent congestion yet not throttle any sender unless it causes trouble. For the same reason, it is likely that multiple choke packets will be sent to a given host and destination. The host should ignore these additional chokes for the fixed time interval until its reduction in traffic takes effect. After that period, further choke packets indicate that the network is still congested.

A choke packet used in the early Internet is the SOURCE QUENCH message (Postel, 1981). It never caught on, though, partly because the circumstances in which it was generated and the effect it had were not well specified. The modern Internet uses a different notification design that we will describe next.

Explicit Congestion Notification

Instead of generating additional packets to warn of congestion, a router can tag any packet it forwards (by setting a bit in the packet's header) to signal that it is experiencing congestion. When the network delivers the packet, the destination can note that there is congestion and inform the sender when it sends a reply packet. The sender can then throttle its transmissions as before.

This design is called **ECN** (**Explicit Congestion Notification**) and is used in the Internet (Ramakrishnan et al., 2001). It is a refinement of early congestion signaling protocols, notably the binary feedback scheme of Ramakrishnan and Jain (1988) that was used in the DECnet architecture. Two bits in the IP packet header are used to record whether the packet has experienced congestion. Packets are unmarked when they are sent, as illustrated in Fig. 5-25. If any of the routers they pass through is congested, that router will then mark the packet as having experienced congestion as it is forwarded. The destination will then echo any marks it has received back to the sender as an explicit congestion signal in its next reply packet. This is shown with a dashed line in the figure to indicate that it happens above the IP level (e.g., in TCP). The sender must then throttle its transmissions, as in the case of choke packets.

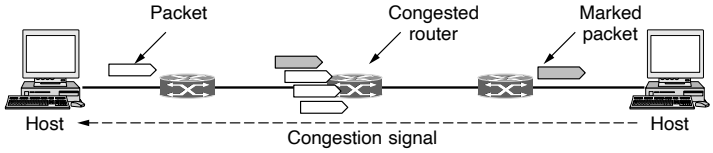


Figure 5-25. Explicit congestion notification

Hop-by-Hop Backpressure

At high speeds or over long distances, many new packets may be transmitted after congestion has been signaled because of the delay before the signal takes effect. Consider, for example, a host in San Francisco (router *A* in Fig. 5-26) that is sending traffic to a host in New York (router *D* in Fig. 5-26) at the OC-3 speed of 155 Mbps. If the New York host begins to run out of buffers, it will take about 40 msec for a choke packet to get back to San Francisco to tell it to slow down. An ECN indication will take even longer because it is delivered via the destination. Choke packet propagation is illustrated as the second, third, and fourth steps in Fig. 5-26(a). In those 40 msec, another 6.2 megabits will have been sent. Even if the host in San Francisco completely shuts down immediately, the 6.2 megabits in the pipe will continue to pour in and have to be dealt with. Only in the seventh diagram in Fig. 5-26(a) will the New York router notice a slower flow.

An alternative approach is to have the choke packet take effect at every hop it passes through, as shown in the sequence of Fig. 5-26(b). Here, as soon as the choke packet reaches *F*, *F* is required to reduce the flow to *D*. Doing so will require *F* to devote more buffers to the connection, since the source is still sending away at full blast, but it gives *D* immediate relief, like a headache remedy in a television commercial. In the next step, the choke packet reaches *E*, which tells *E* to reduce the flow to *F*. This action puts a greater demand on *E*'s buffers but gives *F* immediate relief. Finally, the choke packet reaches *A* and the flow genuinely slows down.

The net effect of this hop-by-hop scheme is to provide quick relief at the point of congestion, at the price of using up more buffers upstream. In this way, congestion can be nipped in the bud without losing any packets. The idea is discussed in detail by Mishra et al. (1996).

5.4 QUALITY OF SERVICE AND APPLICATION QOE

The techniques we looked at in the previous sections are designed to reduce congestion and improve network performance. However, there are applications (and customers) that demand stronger performance guarantees from the network than ''the best that could be done under the circumstances,'' sometimes referred to as **best effort**. Yet, many applications often require some minimum level of throughput to function and also do not perform well when latency exceeds some threshold.. In this section, we will continue our study of network performance, with a sharper focus on ways to provide quality of service that can meet application needs. This is an area in which the Internet is undergoing a long-term upgrade. More recently, there has also been increased focus on user (**QoE**) **Quality of Experience**, which recognizes that ultimately the user experience matters, and different applications have very different requirements and thresholds, as far as network performance goes. An increasing area of focus pertains to estimating user QoE given the ability to observe only encrypted network traffic.

5.4.1 Application QoS Requirements

A stream of packets from a source to a destination is called a **flow** (Clark, 1988). A flow might be all the packets of a connection in a connection-oriented network, or all the packets sent from one process to another process in a connectionless network. The needs of each flow can be characterized by four primary parameters: bandwidth, delay, jitter, and loss. Together, these determine the **QoS** (**Quality of Service**) the flow requires.

Several common applications and the stringency of their network requirements are listed in Fig. 5-27. Note that network requirements are less demanding than application requirements in those cases that the application can improve on the

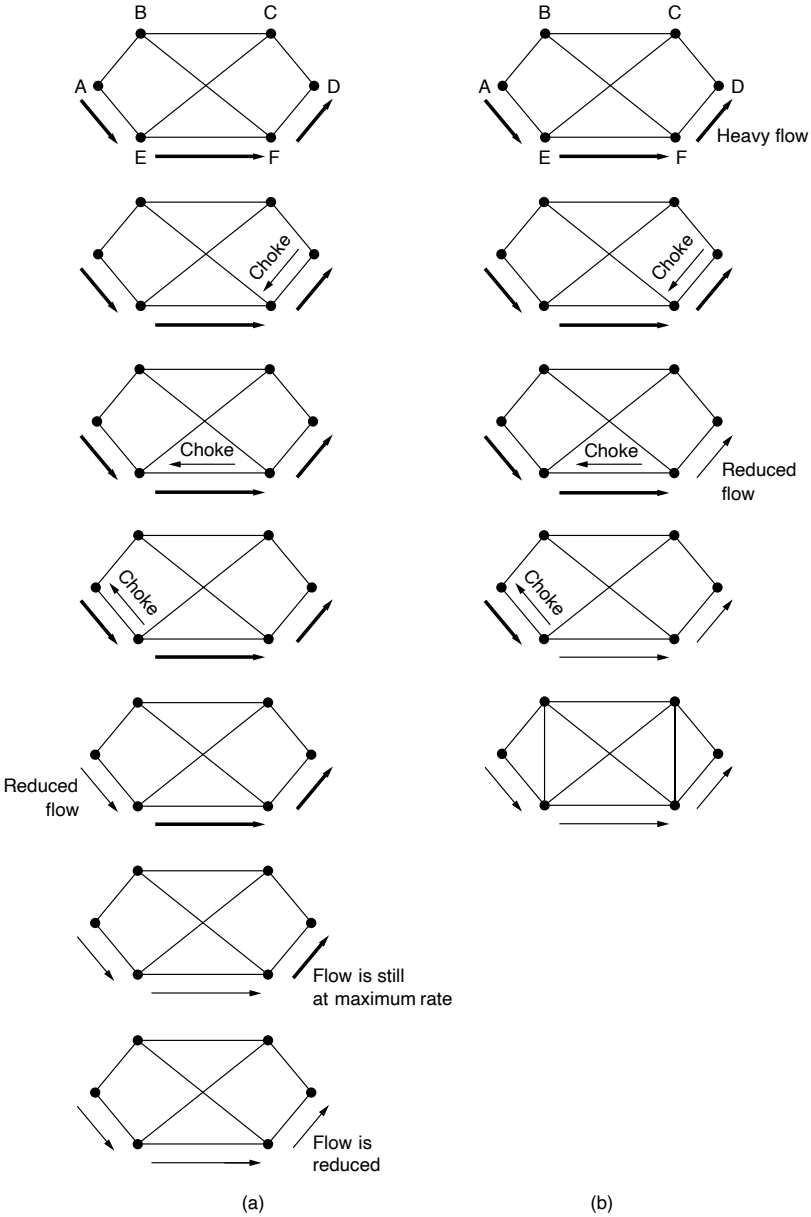


Figure 5-26. (a) A choke packet that affects only the source. (b) A choke packet that affects each hop it passes through.

service provided by the network. In particular, networks do not need to be lossless for reliable file transfer, and they do not need to deliver packets with identical delays for audio and video playout. Some amount of loss can be repaired with retransmissions, and some amount of jitter can be smoothed by buffering packets at the receiver. However, there is nothing applications can do to remedy the situation if the network provides too little bandwidth or too much delay.

Application	Bandwidth	Delay	Jitter	Loss
Email	Low	Low	Low	Medium
File sharing	High	Low	Low	Medium
Web access	Medium	Medium	Low	Medium
Remote login	Low	Medium	Medium	Medium
Audio on demand	Low	Low	High	Low
Video on demand	High	Low	High	Low
Telephony	Low	High	High	Low
Videoconferencing	High	High	High	Low

Figure 5-27. Stringency of applications' quality-of-service requirements.

The applications differ in their bandwidth needs, with email, audio in all forms, and remote login not needing much, but file sharing and video in all forms needing a great deal.

More interesting are the delay requirements. File transfer applications, including email and video, are not delay sensitive. If all packets are delayed uniformly by a few seconds, no harm is done. Interactive applications, such as Web surfing and remote login, are more delay sensitive. Real-time applications, such as telephony and videoconferencing, have strict delay requirements. If all the words in a telephone call are each delayed by too long, the users will find the connection unacceptable. On the other hand, playing audio or video files from a server does not require low delay.

The variation (i.e., standard deviation) in the delay or packet arrival times is called **jitter**. The first three applications in Fig. 5-27 are not sensitive to the packets arriving with irregular time intervals between them. Remote login is somewhat sensitive to that, since updates on the screen will appear in little bursts if the connection suffers much jitter. Video and especially audio are extremely sensitive to jitter. If a user is watching a video over the network and the frames are all delayed by exactly 2.000 seconds, no harm is done. But if the transmission time varies randomly between 1 and 2 seconds, the result will be terrible unless the application hides the jitter. For audio, a jitter of even a few milliseconds is clearly audible.

The first four applications have more stringent requirements on loss than audio and video because all bits must be delivered correctly. This goal is usually achieved with retransmissions of packets that are lost in the network by the transport layer. This is wasted work; it would be better if the network refused packets

it was likely to lose in the first place. Audio and video applications can tolerate some lost packets without retransmission because people do not notice short pauses or occasional skipped frames.

To accommodate a variety of applications, networks may support different categories of QoS. An influential example comes from ATM networks, which were once part of a grand vision for networking but have since become a niche technology. They support:

1. Constant bit rate (e.g., telephony).
2. Real-time variable bit rate (e.g., compressed videoconferencing).
3. Non-real-time variable bit rate (e.g., watching a movie on demand).
4. Available bit rate (e.g., file transfer).

These categories are also useful for other purposes and other networks. Constant bit rate is an attempt to simulate a wire by providing a uniform bandwidth and a uniform delay. Variable bit rate occurs when video is compressed, with some frames compressing more than others. Sending a frame with a lot of detail in it may require sending many bits, whereas a shot of a white wall may compress extremely well. Movies on demand are not actually real time because a few seconds of video can easily be buffered at the receiver before playback starts, so jitter on the network merely causes the amount of stored-but-not-played video to vary. Available bit rate is for applications such as email that are not sensitive to delay or jitter and will take what bandwidth they can get.

5.4.2 Overprovisioning

An easy solution to provide good quality of service is to build a network with enough capacity for whatever traffic will be thrown at it. The name for this solution is **overprovisioning**. The resulting network will carry application traffic without significant loss and, assuming a decent routing scheme, will deliver packets with low latency. Performance doesn't get any better than this. To some extent, the telephone system is overprovisioned because it is rare to pick up a telephone and not get a dial tone instantly. There is simply so much capacity available that demand can almost always be met.

The trouble with this solution is that it is expensive. It is basically solving a problem by throwing money at it. Quality of service mechanisms let a network with less capacity meet application requirements just as well at a lower cost. Moreover, overprovisioning is based on expected traffic. All bets are off if the traffic pattern changes too much. With quality of service mechanisms, the network can honor the performance guarantees that it makes even when traffic spikes, at the cost of turning down some requests.

Four issues must be addressed to ensure quality of service:

1. What applications need from the network.
2. How to regulate the traffic that enters the network.
3. How to reserve resources at routers to guarantee performance.
4. Whether the network can safely accept more traffic.

No single technique deals efficiently with all these issues. Instead, a variety of techniques have been developed for use at the network (and transport) layer. Practical quality-of-service solutions combine multiple techniques. To this end, we will describe two versions of quality of service for the Internet called Integrated Services and Differentiated Services.

5.4.3 Packet Scheduling

Being able to regulate the shape of the offered traffic is a good start. However, to provide a performance guarantee, we must reserve sufficient resources along the route that the packets take through the network. To do this, we are assuming that the packets of a flow follow the same route. Spraying them over routers at random makes it hard to guarantee anything. As a consequence, something similar to a virtual circuit has to be set up from the source to the destination, and all the packets that belong to the flow must follow this route.

Algorithms that allocate router resources among the packets of a flow and between competing flows are called **packet scheduling algorithms**. Three different kinds of resources can potentially be reserved for different flows:

1. Bandwidth.
2. Buffer space.
3. CPU cycles.

The first one, bandwidth, is the most obvious. If a flow requires 1 Mbps and the outgoing line has a capacity of 2 Mbps, trying to direct three flows through that line is not going to work. Thus, reserving bandwidth means not oversubscribing any output line.

A second resource that is often in short supply is buffer space. When a packet arrives, it is buffered inside the router until it can be transmitted on the chosen outgoing line. The purpose of the buffer is to absorb small bursts of traffic as the flows contend with each other. If no buffer is available, the packet has to be discarded since there is no place to put it. For good quality of service, some buffers might be reserved for a specific flow so that flow does not have to compete for buffers with other flows. Up to some maximum value, there will always be a buffer available when the flow needs one.

Finally, CPU cycles may also be a scarce resource. It takes router CPU time to process a packet, so a router can process only a certain number of packets per second. While modern routers are able to process most packets quickly, some kinds of packets require greater CPU processing, such as the ICMP packets we will describe in Sec. 5.7.4. Making sure that the CPU is not overloaded is needed to ensure timely processing of these packets.

First-In First-Out (FIFO) Scheduling

Packet scheduling algorithms allocate bandwidth and other router resources by determining which of the buffered packets to send on the output line next. We already described the most straightforward scheduler when explaining how routers work. Each router buffers packets in a queue for each output line until they can be sent, and they are sent in the same order that they arrived. This algorithm is known as **FIFO** (**First-In First-Out**), or equivalently **FCFS** (**First-Come First-Served**).

FIFO routers usually drop newly arriving packets when the queue is full. Since the newly arrived packet would have been placed at the end of the queue, this behavior is called **tail drop**. It is intuitive, and you may be wondering what alternatives exist. In fact, the RED algorithm we described in Sec. 5.3.2 chose a newly arriving packet to drop at random when the average queue length grew large. The other scheduling algorithms that we will describe also create other opportunities for deciding which packet to drop when the buffers are full.

Fair Queueing

FIFO scheduling is simple to implement, but it is not suited to providing good quality of service because when there are multiple flows, one flow can easily affect the performance of the other flows. If the first flow is aggressive and sends large bursts of packets, they will lodge in the queue. Processing packets in the order of their arrival means that the aggressive sender can hog most of the capacity of the routers its packets traverse, starving the other flows and reducing their quality of service. To add insult to injury, the packets of the other flows that do get through are likely to be delayed because they had to sit in the queue behind many packets from the aggressive sender.

Many packet scheduling algorithms have been devised that provide stronger isolation between flows and thwart attempts at interference (Bhatti and Crowcroft, 2000). One of the first ones was the **fair queueing** algorithm devised by Nagle (1987). The essence of this algorithm is that routers have separate queues, one for each flow for a given output line. When the line becomes idle, the router scans the queues round robin, as shown in Fig. 5-28. It then takes the first packet on the next queue. In this way, with n hosts competing for the output line, each host gets to send one out of every n packets. It is fair in the sense that all flows get to send packets at the same rate. Sending more packets will not improve this rate.

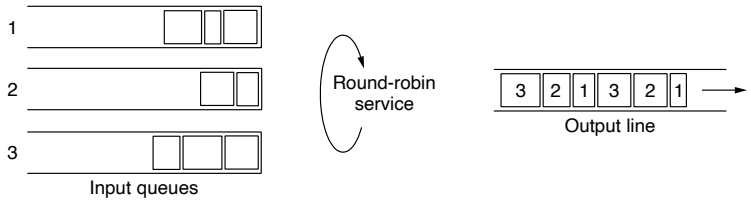
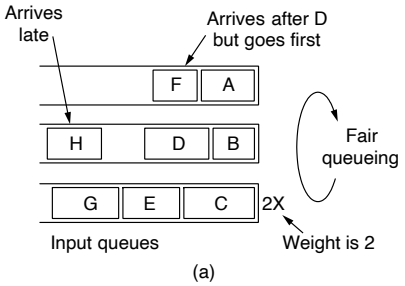


Figure 5-28. Round-robin fair queueing.

Although a start, the algorithm has a flaw: it gives more bandwidth to hosts that use large packets than to hosts that use small packets. Demers et al. (1990) suggested an improvement in which the round robin is done in such a way as to simulate a byte-by-byte round robin, instead of a packet-by-packet round robin. The trick is to compute a virtual time that is the number of the round at which each packet would finish being sent. Each round drains a byte from all of the queues that have data to send. The packets are then sorted in order of their finishing times and sent in that order.

This algorithm and an example of finish times for packets arriving in three flows are illustrated in Fig. 5-29. If a packet has length L, the round at which it will finish is simply L rounds after the start time. The start time is either the finish time of the previous packet, or the arrival time of the packet, if the queue is empty when it arrives.



Packet	Arrival time	Length	Finish time	Output order
A	0	8	8	1
B	5	6	11	3
C	5	10	10	2
D	8	9	20	7
E	8	8	14	4
F	10	6	16	5
G	11	10	19	6
H	20	8	28	8

(b)

Figure 5-29. (a) Weighted Fair Queueing. (b) Finishing times for the packets.

From the table in Fig. 5-29(b), and looking only at the first two packets in the top two queues, packets arrive in the order A, B, D, and F. Packet A arrives at round 0 and is 8 bytes long, so its finish time is round 8. Similarly the finish time for packet B is 11. Packet D arrives while B is being sent. Its finish time is 9 byte-rounds after it starts when B finishes, or 20. Similarly, the finish time for F is 16. In the absence of new arrivals, the relative sending order is A, B, F, D, even though F arrived after D. It is possible that another small packet will arrive on the top flow and obtain a finish time before D. It will only jump ahead of D if the

transmission of that packet has not started. Fair queueing does not preempt packets that are currently being transmitted. Because packets are sent in their entirety, fair queueing is only an approximation of the ideal byte-by-byte scheme. But it is a very good approximation, staying within one packet transmission of the ideal scheme at all times.

Weighted Fair Queueing

One shortcoming of this algorithm in practice is that it gives all hosts the same priority. In many situations, it is desirable to give, for example, video servers more bandwidth than, say, file servers. This is easily possible by giving the video server two or more bytes per round. This modified algorithm is called **WFQ** (**Weighted Fair Queueing**). Letting the number of bytes per round be the weight of a flow, W, we can now give the formula for computing the finish time:

$$F_i = \max(A_i, F_{i-1}) + L_i/W$$

where A_i is the arrival time, F_i is the finish time, and L_i is the length of packet i. The bottom queue of Fig. 5-29(a) has a weight of 2, so its packets are sent more quickly as you can see in the finish times given in Fig. 5-29(b).

Another practical consideration is implementation complexity. WFQ requires that packets be inserted by their finish time into a sorted queue. With N flows, this is at best an $O(\log N)$ operation per packet, which is difficult to achieve for many flows in high-speed routers. Shreedhar and Varghese (1995) describe an approximation called **deficit round robin** that can be implemented very efficiently, with only $O(1)$ operations per packet. WFQ is widely used given this approximation.

Other kinds of scheduling algorithms exist, too. A simple example is priority scheduling, in which each packet is marked with a priority. High-priority packets are always sent before any low-priority packets that are buffered. Within a priority, packets are sent in FIFO order. However, priority scheduling has the disadvantage that a burst of high-priority packets can starve low-priority packets, which may have to wait indefinitely. WFQ often provides a better alternative. By giving the high-priority queue a large weight, say 3, high-priority packets will often go through a short line (as relatively few packets should be high priority) yet some fraction of low-priority packets will continue to be sent even when there is high priority traffic. A high- and low-priority system is essentially a two-queue WFQ system in which the high priority has infinite weight.

As a final example of a scheduler, packets might carry timestamps and be sent in timestamp order. Clark et al. (1992) describe a design in which the timestamp records how far the packet is behind or ahead of schedule as it is sent through a sequence of routers on the path. Packets that have been queued behind other packets at a router will tend to be behind schedule, and the packets that have been serviced first will tend to be ahead of schedule. Sending packets in order of their timestamps has the beneficial effect of speeding up slow packets while at the same time

slowing down fast packets. The result is that all packets are delivered by the network with a more consistent delay, which is obviously a good thing.

Putting it Together

We have now seen all the necessary elements for QoS, so it is time to put them together to actually provide it. QoS guarantees are established through the process of admission control. We first saw admission control used to control congestion, which is a performance guarantee, albeit a weak one. The guarantees we are considering now are stronger, but the model is the same. The user offers a flow with an accompanying QoS requirement to the network. The network then decides whether to accept or reject the flow based on its capacity and the commitments it has made to other flows. If it accepts, the network reserves capacity in advance at routers to guarantee QoS when traffic is sent on the new flow.

The reservations must be made at all of the routers along the route that the packets take through the network. Any routers on the path without reservations might become congested, and a single congested router can break the QoS guarantee. Many routing algorithms find the single best path between each source and each destination and send all traffic over that path. This may cause some flows to be rejected if there is not enough spare capacity along the best path. QoS guarantees for new flows may still be accommodated by choosing a different route for the flow that has excess capacity. This is called **QoS routing**. Chen and Nahrstedt (1998) give an overview of these techniques. It is also possible to split the traffic for each destination over multiple paths to more easily find excess capacity. A simple method is for routers to choose equal-cost paths and to divide the traffic equally or in proportion to the capacity of the outgoing links. However, more sophisticated algorithms are also available (Nelakuditi and Zhang, 2002).

Given a path, the decision to accept or reject a flow is not a simple matter of comparing the resources (bandwidth, buffers, and cycles) requested by the flow with the router's excess capacity in those three dimensions. It is a little more complicated than that. To start with, although some applications may know about their bandwidth requirements, few know about buffers or CPU cycles, so at the minimum, a different way is needed to describe flows and translate this description to router resources. We will get to this shortly.

Next, some applications are far more tolerant of an occasional missed deadline than others. The applications must choose from the type of guarantees that the network can make, whether hard guarantees or behavior that will hold most of the time. All else being equal, everyone would like hard guarantees, but the difficulty is that they are expensive because they constrain worst case behavior. Guarantees for most of the packets are often sufficient for applications, and more flows with this guarantee can be supported for a fixed capacity.

Finally, some applications may be willing to haggle about the flow parameters and others may not be willing to do so. For example, a movie viewer that normally

runs at 30 frames/sec may be willing to drop back to 25 frames/sec if there is not enough free bandwidth to support 30 frames/sec. Similarly, the number of pixels per frame, audio bandwidth, and other properties may be adjustable.

Because many parties may be involved in the flow negotiation (the sender, the receiver, and all the routers along the path between them), flows must be described accurately in terms of specific parameters that can be negotiated. A set of such parameters is called a **flow specification**. Typically, the sender (e.g., the video server) produces a flow specification proposing the parameters it would like to use. As the specification propagates along the route, each router examines it and modifies the parameters as need be. The modifications can only reduce the flow, not increase it (e.g., a lower data rate, not a higher one). When it gets to the other end, the parameters can be established.

As an example of what can be in a flow specification, consider the example of Fig. 5-30. This is based on RFC 2210 and RFC 2211 for Integrated Services, a QoS design we will cover in the next section. It has five parameters. The first two parameters, the *token bucket rate* and *token bucket size*, use a token bucket to give the maximum sustained rate the sender may transmit, averaged over a long time interval, and the largest burst it can send over a short time interval.

Parameter	Unit
Token bucket rate	Bytes/sec
Token bucket size	Bytes
Peak data rate	Bytes/sec
Minimum packet size	Bytes
Maximum packet size	Bytes

Figure 5-30. An example flow specification.

The third parameter, the *peak data rate*, is the maximum transmission rate tolerated, even for brief time intervals. The sender must never exceed this rate even for short bursts.

The last two parameters specify the minimum and maximum packet sizes, including the transport and network layer headers (e.g., TCP and IP). The minimum size is useful because processing each packet takes some fixed time, no matter how short. A router may be prepared to handle 10,000 packets/sec of 1 KB each, but not be prepared to handle 100,000 packets/sec of 50 bytes each, even though this represents a lower data rate. The maximum packet size is important due to internal network limitations that may not be exceeded. For example, if part of the path goes over an Ethernet, the maximum packet size will be restricted to no more than 1500 bytes no matter what the rest of the network can handle.

An interesting question is how a router turns a flow specification into a set of specific resource reservations. At first glance, it might appear that if a router has a link that runs at, say, 1 Gbps and the average packet is 1000 bits, it can process 1

million packets/sec. This observation is not the case, though, because there will always be idle periods on the link due to statistical fluctuations in the load. If the link needs every bit of capacity to get its work done, idling for even a few bits creates a backlog it can never get rid of.

Even with a load slightly below the theoretical capacity, queues can build up and delays can occur. Consider a situation in which packets arrive at random with a mean arrival rate of λ packets/sec. The packets have random lengths and can be sent on the link with a mean service rate of μ packets/sec. Under the assumption that both the arrival and service distributions are Poisson distributions (what is called an M/M/1 queueing system, where "M" stands for Markov, i.e., Poisson), it can be proven using queueing theory that the mean delay experienced by a packet, T, is

$$T = \frac{1}{\mu} \times \frac{1}{1 - \lambda/\mu} = \frac{1}{\mu} \times \frac{1}{1 - \rho}$$

where $\rho = \lambda/\mu$ is the CPU utilization. The first factor, $1/\mu$, is what the service time would be in the absence of competition. The second factor is the slowdown due to competition with other flows. For example, if $\lambda = 950{,}000$ packets/sec and $\mu = 1{,}000{,}000$ packets/sec, then $\rho = 0.95$ and the mean delay experienced by each packet will be 20 μsec instead of 1 μsec. This time accounts for both the queueing time and the service time, as can be seen when the load is very low ($\lambda/\mu \approx 0$). If there are, say, 30 routers along the flow's route, queueing delay alone will account for 600 μsec of delay.

One method of relating flow specifications to router resources that correspond to bandwidth and delay performance guarantees is given by Parekh and Gallagher (1993, 1994). It is based on traffic sources shaped by (R, B) token buckets and WFQ at routers. Each flow is given a WFQ weight W large enough to drain its token bucket rate R as shown in Fig. 5-31. For example, if the flow has a rate of 1 Mbps and the router and output link have a capacity of 1 Gbps, the weight for the flow must be greater than 1/1000th of the total of the weights for all of the flows at that router for the output link. This guarantees the flow a minimum bandwidth. If it cannot be given a large enough rate, the flow cannot be admitted.

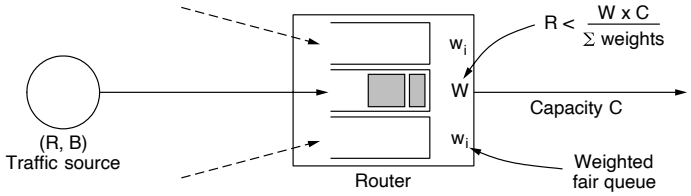


Figure 5-31. Bandwidth and delay guarantees with token buckets and WFQ.

The largest queueing delay the flow will see is a function of the burst size of the token bucket. Consider the two extreme cases. If the traffic is smooth, without

any bursts, packets will be drained from the router just as quickly as they arrive. There will be no queueing delay (ignoring packetization effects). On the other hand, if the traffic is saved up in bursts, then a maximum-size burst, B, may arrive at the router all at once. In this case, the maximum queueing delay, D, will be the time taken to drain this burst at the guaranteed bandwidth, or B/R (again, ignoring packetization effects). If this delay is too large, the flow must request more bandwidth from the network.

These guarantees are hard. The token buckets bound the burstiness of the source, and fair queueing isolates the bandwidth given to different flows. This means that the flow will meet its bandwidth and delay guarantees regardless of how the other competing flows behave at the router. Those other flows cannot break the guarantee even by saving up traffic and all sending at once.

Moreover, the result holds for a path through multiple routers in any network topology. Each flow gets a minimum bandwidth because that bandwidth is guaranteed at each router. The reason each flow gets a maximum delay is more subtle. In the worst case that a burst of traffic hits the first router and competes with the traffic of other flows, it will be delayed up to the maximum delay of D. However, this delay will also smooth the burst. In turn, this means that the burst will incur no further queueing delays at later routers. The overall queueing delay will be at most D.

5.4.4 Integrated Services

Between 1995 and 1997, IETF put a lot of effort into devising an architecture for streaming multimedia. This work resulted in over two dozen RFCs, starting with RFC 2205 through RFC 2212. The generic name for this work is **integrated services**. It was aimed at both unicast and multicast applications. An example of the former is a single user streaming a video clip from a news site. An example of the latter is a collection of digital television stations broadcasting their programs as streams of IP packets to many receivers at various locations. Below we will concentrate on multicast, since unicast is a special case of multicast.

In many multicast applications, groups can change membership dynamically, for example, as people enter a video conference and then get bored and switch to a soap opera or the croquet channel. Under these conditions, the approach of having the senders reserve bandwidth in advance does not work well, since it would require each sender to track all entries and exits of its audience. For a system designed to transmit television with millions of subscribers, it would not work at all.

RSVP—The Resource reSerVation Protocol

The main part of the integrated services architecture that is visible to the users of the network is **RSVP** (**Resource reSerVation Protocol**). It is described in RFC 2205 through RFC 2210. This protocol is used for making the reservations; other

protocols are used for sending the data. RSVP allows multiple senders to transmit to multiple groups of receivers, permits individual receivers to switch channels freely, and also optimizes bandwidth use while at the same time eliminating congestion.

In its simplest form, the protocol uses multicast routing using spanning trees, as discussed earlier. Each group is assigned a group address. To send to a group, a sender puts the group's address in its packets. The standard multicast routing algorithm then builds a spanning tree covering all group members. The routing algorithm is not part of RSVP. The only difference from normal multicasting is a little extra information that is multicast to the group periodically to tell the routers along the tree to maintain certain data structures in their memories.

As an example, consider the network of Fig. 5-32(a). Hosts 1 and 2 are multicast senders, and hosts 3, 4, and 5 are multicast receivers. In this example, the senders and receivers are disjoint, but in general, the two sets may overlap. The multicast trees for hosts 1 and 2 are shown in Fig. 5-32(b) and Fig. 5-32(c), respectively.

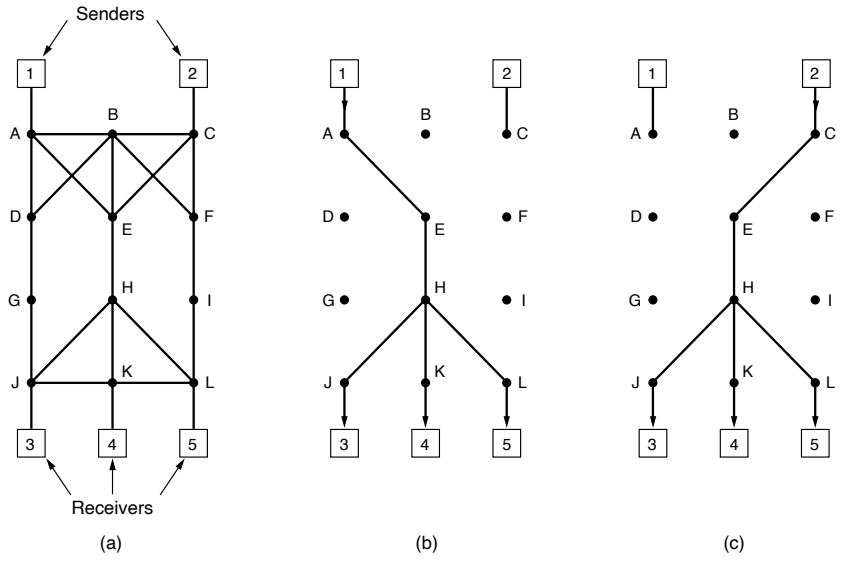


Figure 5-32. (a) A network. (b) The multicast spanning tree for host 1. (c) The multicast spanning tree for host 2.

To get better reception and eliminate congestion, any of the receivers in a group can send a reservation message up the tree to the sender. The message is

propagated using the reverse path forwarding algorithm discussed earlier. At each hop, the router notes the reservation and reserves the necessary bandwidth. We saw in the previous section how a weighted fair queueing scheduler can be used to make this reservation. If insufficient bandwidth is available, it reports back failure. By the time the message gets back to the source, bandwidth has been reserved all the way from the sender to the receiver making the reservation request along the spanning tree.

An example of such a reservation is shown in Fig. 5-33(a). Here host 3 has requested a channel to host 1. Once it has been established, packets can flow from 1 to 3 without congestion. Now consider what happens if host 3 next reserves a channel to the other sender, host 2, so the user can watch two television programs at once. A second path is reserved, as illustrated in Fig. 5-33(b). Note that two separate channels are needed from host 3 to router *E* because two independent streams are being transmitted.

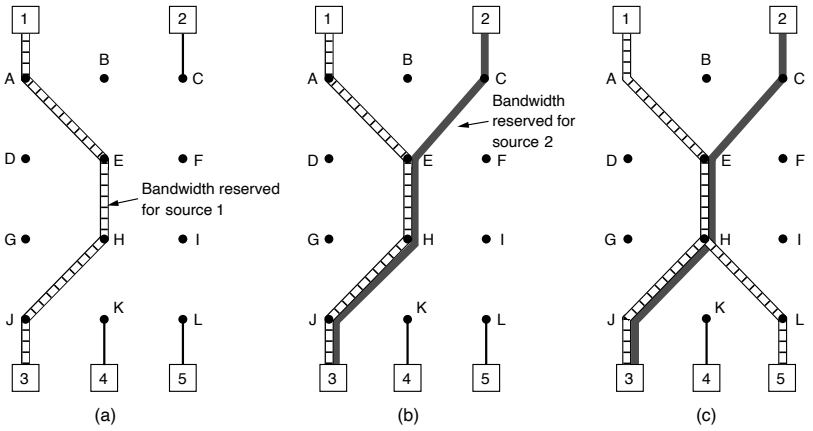


Figure 5-33. (a) Host 3 requests a channel to host 1. (b) Host 3 then requests a second channel, to host 2. (c) Host 5 requests a channel to host 1.

Finally, in Fig. 5-33(c), host 5 decides to watch the program being transmitted by host 1 and also makes a reservation. First, dedicated bandwidth is reserved as far as router *H*. However, this router sees that it already has a feed from host 1, so if the necessary bandwidth has already been reserved, it does not have to reserve any more. Note that hosts 3 and 5 might have asked for different amounts of bandwidth (e.g., if host 3 is playing on a small screen and only wants the low-resolution information), so the capacity reserved must be large enough to satisfy the greediest receiver.

When making a reservation, a receiver can (optionally) specify one or more sources that it wants to receive from. It can also specify whether these choices are

fixed for the duration of the reservation or whether the receiver wants to keep open the option of changing sources later. The routers use this information to optimize bandwidth planning. In particular, two receivers are only set up to share a path if they both agree not to change sources later on.

The reason for this strategy in the fully dynamic case is that reserved bandwidth is decoupled from the choice of source. Once a receiver has reserved bandwidth, it can switch to another source and keep that portion of the existing path that is valid for the new source. If host 2 is transmitting several video streams in real time, for example a TV broadcaster with multiple channels, host 3 may switch between them at will without changing its reservation: the routers do not care what program the receiver is watching.

5.4.5 Differentiated Services

Flow-based algorithms have the potential to offer good quality of service to one or more flows because they reserve whatever resources are needed along the route. However, they also have a downside. They require an advance setup to establish each flow, something that does not scale well when there are thousands or millions of flows. Also, they maintain internal per-flow state in the routers, making them vulnerable to router crashes. Finally, the changes required to the router code are substantial and involve complex router-to-router exchanges for setting up the flows. As a consequence, while work continues to advance integrated services, few deployments of it or anything like it exist yet.

For these reasons, IETF has also devised a simpler approach to quality of service, one that can be largely implemented locally in each router without advance setup and without having the whole path involved. This approach is known as **class-based** (as opposed to flow-based) quality of service. IETF has standardized an architecture for it, called **differentiated services**, which is described in RFC 2474, RFC 2475, and numerous others. We will now describe it.

Differentiated services can be offered by a set of routers forming an administrative domain (e.g., an ISP or a telco). The administration defines a set of service classes with corresponding forwarding rules. If a customer subscribes to differentiated services, customer packets entering the domain are marked with the class to which they belong. This information is carried in the *Differentiated services* field of IPv4 and IPv6 packets (described in Sec. 5.7.1). The classes are defined as **per-hop behaviors** because they correspond to the treatment the packet will receive at each router, not a guarantee across the network. Better service is provided to packets with some per-hop behaviors (e.g., premium service) than to others (e.g., regular service). Traffic within a class may be required to conform to some specific shape, such as a leaky bucket with some specified drain rate. An operator with a good nose for business might charge extra for each premium packet transported or might allow up to N premium packets per month for a fixed additional monthly fee. Note that this scheme requires no advance setup, no resource

reservation, and no time-consuming end-to-end negotiation for each flow, as with integrated services. This makes differentiated services relatively easy to implement.

Class-based service also occurs in other industries. For example, package delivery companies often offer overnight, two-day, and three-day service. Airlines offer first class, business class, and cattle-class service. Long-distance trains have multiple service classes. The Paris subway even had two service classes for the same quality of seating. For packets, the classes may differ in terms of delay, jitter, and probability of being discarded in the event of congestion, among other possibilities (but probably not roomier Ethernet frames).

To make the difference between flow-based quality of service and class-based quality of service clearer, consider an example: Internet telephony. With a flow-based scheme, each telephone call gets its own resources and guarantees. With a class-based scheme, all the telephone calls together get the resources reserved for the class telephony. These resources cannot be taken away by packets from the Web browsing class or other classes, but no telephone call gets any private resources reserved for it alone.

Expedited Forwarding

The choice of service classes is up to each operator, but since packets are often forwarded between networks run by different operators, IETF has defined some network-independent service classes. The simplest class is **expedited forwarding**, so let us start with that one. It is described in RFC 3246.

The idea behind expedited forwarding is very simple. Two classes of service are available: regular and expedited. The vast majority of the traffic is expected to be regular, but a limited fraction of the packets are expedited. The expedited packets should be able to transit the network as though no other packets were present. In this way, they will get low loss, low delay and low jitter service—just what is needed for VoIP. A symbolic representation of this ''two-tube'' system is given in Fig. 5-34. Note that there is still just one physical line. The two logical pipes shown in the figure represent a way to reserve bandwidth for different classes of service, not a second physical line.

One way to implement this strategy is as follows. Packets are classified as expedited or regular and marked accordingly. This step might be done on the sending host or in the ingress (first) router. The advantage of doing classification on the sending host is that more information is available about which packets belong to which flows. This task may be performed by networking software or even the operating system, to avoid having to change existing applications. For example, it is becoming common for VoIP packets to be marked for expedited service by hosts. If the packets pass through a corporate network or ISP that supports expedited service, they will receive preferential treatment. If the network does not support expedited service, no harm is done. In that case, it makes sense to at least try.

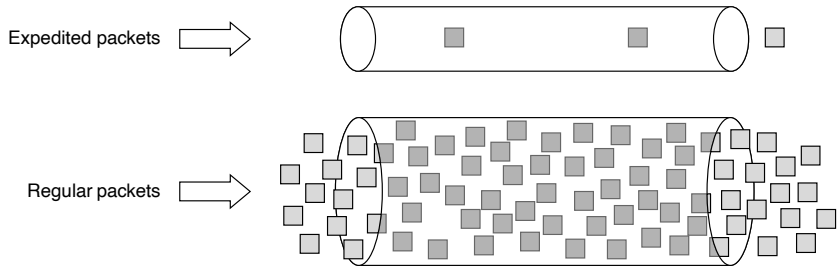


Figure 5-34. Expedited packets experience a traffic-free network.

Of course, if the marking is done by the host, the ingress router is likely to police the traffic to make sure that customers are not sending more expedited traffic than they have paid for. Within the network, the routers may have two output queues for each outgoing line, one for expedited packets and one for regular packets. When a packet arrives, it is queued accordingly. The expedited queue is given priority over the regular one, for example, by using a priority scheduler. In this way, expedited packets see an unloaded network, even when there is, in fact, a heavy load of regular traffic.

Assured Forwarding

A somewhat more elaborate scheme for managing the service classes is called **assured forwarding**. It is described in RFC 2597. Assured forwarding specifies that there shall be four priority classes, each class having its own resources. The top three classes might be called gold, silver, and bronze. In addition, it defines three discard classes for packets that are experiencing congestion: low, medium, and high. Taken together, these factors define 12 service classes.

Figure 5-35 shows one way packets might be processed under assured forwarding. The first step is to classify the packets into one of the four priority classes. As before, this step might be done on the sending host (as shown in the figure) or in the ingress router, and the rate of higher-priority packets may be limited by the operator as part of the service offering.

The next step is to determine the discard class for each packet. This is done by passing the packets of each priority class through a traffic policer such as a token bucket. The policer lets all of the traffic through, but it identifies packets that fit within small bursts as low discard, packets that exceed small bursts as medium discard, and packets that exceed large bursts as high discard. The combination of priority and discard class is then encoded in each packet.

Finally, the packets are processed by routers in the network with a packet scheduler that carefully distinguishes the different classes. A common choice is to

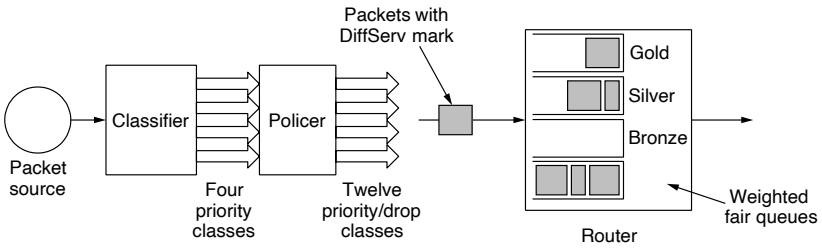


Figure 5-35. A possible implementation of assured forwarding.

use weighted fair queueing for the four priority classes, with higher classes given higher weights. In this way, the higher classes will get most of the bandwidth, but the lower classes will not be starved of bandwidth entirely. For example, if the weights double from one class to the next higher class, the higher class will get twice the bandwidth. Within a priority class, packets with a higher discard class can be preferentially dropped by running an algorithm such as RED. RED will start to drop packets as congestion builds but before the router has run out of buffer space. At this stage, there is still buffer space with which to accept low discard packets while dropping high discard packets.

5.5 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, and more. Numerous protocols are in widespread use across these networks in every layer.

5.5.1 Internetworks: An Overview

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 very 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 far 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, the network layer protocol of the Internet, including techniques for tunneling through networks, routing in internetworks, and packet fragmentation.

5.5.2 How Networks Differ

Networks can differ in many ways. Some of the differences, such as different modulation techniques or frame formats, are internal to the physical and data link layers. These differences will not concern us here. Instead, in Fig. 5-36 we list some of the differences that can be exposed to the network layer. It is papering over these differences that makes internetworking more difficult than operating within a single network.

When packets sent by a source on one network must transit one or more foreign networks before reaching the destination network, many problems can occur at the interfaces between networks. To start with, the source needs to be able to address the destination. What do we do if the source is on an Ethernet network and the destination is on the cellular telephone network? Assuming we can even specify a cellular destination from an Ethernet network, packets would cross from a connectionless network to a connection-oriented one. This may require that a new connection be set up on short notice, which injects a delay, and much overhead if the connection is not used for many more packets.

Item	Some Possibilities
Service offered	Connectionless versus connection oriented
Addressing	Different sizes, flat or hierarchical
Broadcasting	Present or absent (also multicast)
Packet size	Every network has its own maximum
Ordering	Ordered and unordered delivery
Quality of service	Present or absent; many different kinds
Reliability	Different levels of loss
Security	Privacy rules, encryption, etc.
Parameters	Different timeouts, flow specifications, etc.
Accounting	By connect time, packet, byte, or not at all

Figure 5-36. Some of the many ways networks can differ.

Many specific differences may have to be accommodated as well. How do we multicast a packet to a group with some members on a network that does not support multicast? The differing max packet sizes used by different networks can be a major nuisance, too. How do you pass an 8000-byte packet through a network whose maximum size is 1500 bytes? If packets on a connection-oriented network transit a connectionless network, they may arrive in a different order than they were sent. That is something the sender likely did not expect, and it might come as an (unpleasant) surprise to the receiver as well.

With effort, these kinds of differences can be papered over. For example, a gateway joining two networks might generate separate packets for each destination to simulate multicast. A large packet might be broken up, sent in pieces, and then joined back together. Receivers might buffer packets and deliver them in order.

Networks also can differ in large respects that are more difficult to reconcile. The clearest example is quality of service. If one network has strong QoS and the other offers best effort service, it will be impossible to make bandwidth and delay guarantees for real-time traffic end to end. In fact, they can likely only be made while the best-effort network is operated at a low utilization, or hardly used, which is unlikely to be the goal of most ISPs. Security mechanisms are problematic, but at least encryption for confidentiality and data integrity can be layered on top of networks that do not already include it. Finally, differences in accounting can lead to unwelcome bills when normal usage suddenly becomes expensive, as roaming mobile phone users with data plans have discovered.

5.5.3 Connecting Heterogeneous Networks

There are two basic choices for connecting different networks: we can build devices that translate or convert packets from each kind of network into packets for each other network, or as computer scientists often do, we can try to solve the

problem by adding a layer of indirection and building a common layer on top of the different networks. In either case, the devices are placed at the boundaries between networks; initially, these devices were called **gateways**.

Early on, Cerf and Kahn (1974) argued for a common layer to hide the differences of existing networks. This approach has been tremendously successful, and the layer they proposed was eventually separated into the TCP and IP protocols. Almost four decades later, IP is the foundation of the modern Internet. For this accomplishment, Cerf and Kahn were awarded the 2004 Turing Award, informally known as the Nobel Prize of computer science. IP provides a universal packet format that all routers recognize and that can be passed through almost every network. IP has extended its reach from computer networks to take over the telephone network. It also runs on sensor networks and other tiny devices that were once presumed too resource-constrained to support it.

We have discussed several different devices that connect networks, including repeaters, hubs, switches, bridges, routers, and gateways. Repeaters and hubs just move bits from one wire to another. They are mostly analog devices and do not understand anything about higher layer protocols. Bridges and switches operate at the link layer. They can be used to build networks, but only with minor protocol translation in the process, for example, among 10-, 100-, and 1000-Mbps Ethernet switches. Our focus in this section is interconnection devices that operate at the network layer, namely the routers. We will leave gateways, which are higher-layer interconnection devices, until later.

Let us first explore at a high level how interconnection with a common network layer can be used to interconnect dissimilar networks. An internet comprised of 802.11, MPLS, and Ethernet networks is shown in Fig. 5-37(a). Suppose that the source machine on the 802.11 network wants to send a packet to the destination machine on the Ethernet network. Since these technologies are different, and they are further separated by another kind of network (MPLS), some added processing is needed at the boundaries between the networks.

Because different networks may, in general, have different forms of addressing, the packet carries a network layer address that can identify any host across the three networks. The first boundary the packet reaches is when it transitions from an 802.11 network to an MPLS network. Remember, 802.11 provides a connectionless service, but MPLS provides a connection-oriented service. This means that a virtual circuit must be set up to cross that network. Once the packet has traveled along the virtual circuit, it will reach the Ethernet network. At this boundary, the packet may be too large to be carried, since 802.11 can work with larger frames than Ethernet. To handle this problem, the packet is divided into fragments, and each fragment is sent separately. When the fragments reach the destination, they are reassembled. Then the packet has completed its journey.

The protocol processing for this journey is shown in Fig. 5-37(b). The source accepts data from the transport layer and generates a packet with the common network layer header, which is IP in this example. The network header contains the

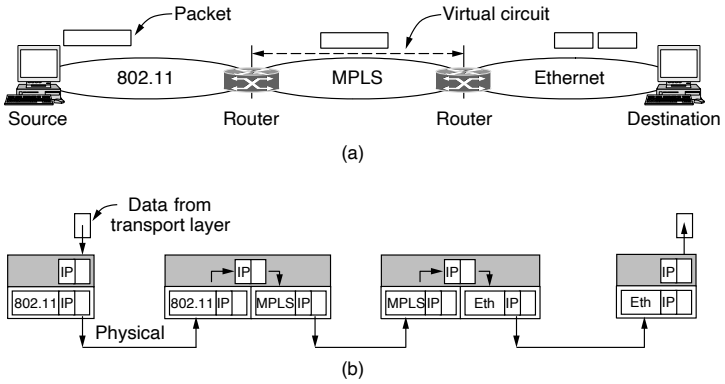


Figure 5-37. (a) A packet crossing different networks. (b) Network and link layer protocol processing.

ultimate destination address, which is used to determine that the packet should be sent via the first router. So the packet is encapsulated in an 802.11 frame whose destination is the first router and transmitted. At the router, the packet is removed from the frame's data field and the 802.11 frame header is discarded. The router now examines the IP address in the packet and looks up this address in its routing table. Based on this address, it decides to send the packet to the second router next. For this part of the path, an MPLS virtual circuit must be established to the second router and the packet must be encapsulated with MPLS headers that travel this circuit. At the far end, the MPLS header is discarded and the network address is again consulted to find the next network layer hop. It is the destination itself. When a packet is too long to be sent over Ethernet, it is split into two portions. Each of these portions is put into the data field of an Ethernet frame and sent to the Ethernet address of the destination. At the destination, the Ethernet header is stripped from each of the frames, and the contents are reassembled. The packet has finally reached its destination.

Observe that there is an essential difference between the routed case and the switched (or bridged) case. With a router, the packet is extracted from the frame and the network address in the packet is used for deciding where to send it. With a switch (or bridge), the entire frame is transported on the basis of its MAC address. Switches do not have to understand the network layer protocol being used to switch packets. Routers do.

Unfortunately, internetworking is not nearly as easy as we have made it sound. In fact, when bridges were introduced, it was intended that they would join different types of networks, or at least different types of LANs. They were to do this by translating frames from one LAN into frames from another LAN. However, this did not work well, for exactly the same reason that internetworking is difficult:

the differences in the features of LANs, such as different maximum packet sizes and LANs with and without priority classes, are hard to mask. Today, bridges are predominantly used to connect the same kind of network at the link layer, and routers connect different networks at the network layer.

Internetworking has been very successful at building large networks, but it only works when there is a common network layer. There have, in fact, been many network protocols over time. Getting everybody to agree on a single format is difficult when companies perceive it to their commercial advantage to have a proprietary format that they control. Examples besides IP, which is now the near-universal network protocol, were IPX, SNA, and AppleTalk. None of these protocols are still in widespread use, but there will always be other protocols. The most relevant example now is probably IPv4 and IPv6. While these are both versions of IP, they are not compatible (or it would not have been necessary to create IPv6).

A router that can handle multiple network protocols is called a **multiprotocol router**. It must either translate the protocols, or leave connection for a higher protocol layer. Neither approach is entirely satisfactory. Connection at a higher layer, say, by using TCP, requires that all the networks implement TCP (which may not be the case). Then it limits usage across the networks to applications that use TCP (which does not include many real-time applications).

The alternative is to translate packets between the networks. However, unless the packet formats are close relatives with the same information fields, such conversions will always be incomplete and often doomed to failure. For example, IPv6 addresses are 128 bits long. They will not fit in a 32-bit IPv4 address field, no matter how hard the router tries. Getting IPv4 and IPv6 to run in the same network has proven to be a major obstacle to the deployment of IPv6. (To be fair, so has getting customers to understand why they should want IPv6 in the first place.) Greater problems can be expected when translating between very different protocols, such as connectionless and connection-oriented network protocols. Given these difficulties, conversion is only rarely attempted. Arguably, even IP has only worked so well by serving as a kind of lowest common denominator. It requires little of the networks on which it runs, but offers only best-effort service as a result.

5.5.4 Connecting Endpoints Across Heterogeneous Networks

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. 5-38.

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

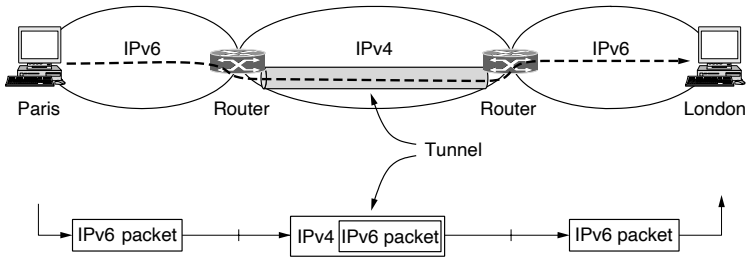


Figure 5-38. Tunneling a packet from Paris to London.

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. 5-39. 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 is tunneled over can be reached because the packets cannot escape in the middle of the tunnel. 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. We will explore VPNs when we get to Chap. 8.

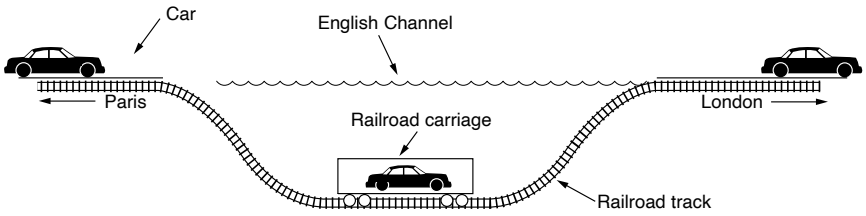


Figure 5-39. Tunneling a car from France to England.

5.5.5 Internetwork Routing: Routing Across Multiple Networks

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 **intradomain** 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 **interdomain** or exterior gateway protocol is used. The networks may all use different intradomain protocols, but they must use the same interdomain protocol. In the Internet, the interdomain routing protocol is called Border Gateway Protocol (BGP). We will describe it in Sec. 5.7.7

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 or 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.

5.5.6 Supporting Different Packet Sizes: 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 a packet has to 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 all the fragmentation caused by a ''small-packet'' network transparent to any subsequent networks through which the packet must pass on its way to the ultimate destination. This option is shown in Fig. 5-40(a). In this approach, when an oversized packet arrives at G_1, the router breaks it up into fragments. Each fragment is addressed to the same exit router, G_2, 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 treated as though it were an original packet. The routers pass the fragments, as shown in Fig. 5-40(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

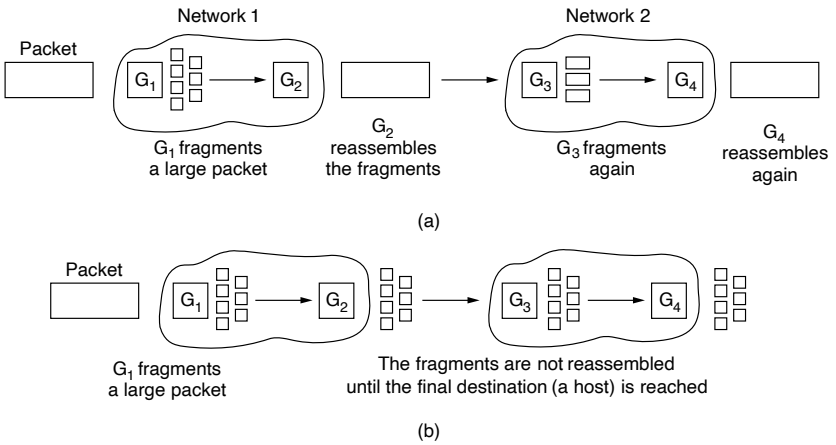


Figure 5-40. (a) Transparent fragmentation. (b) Nontransparent fragmentation.

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. 5-41. 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. 5-41(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.

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 like this. 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,

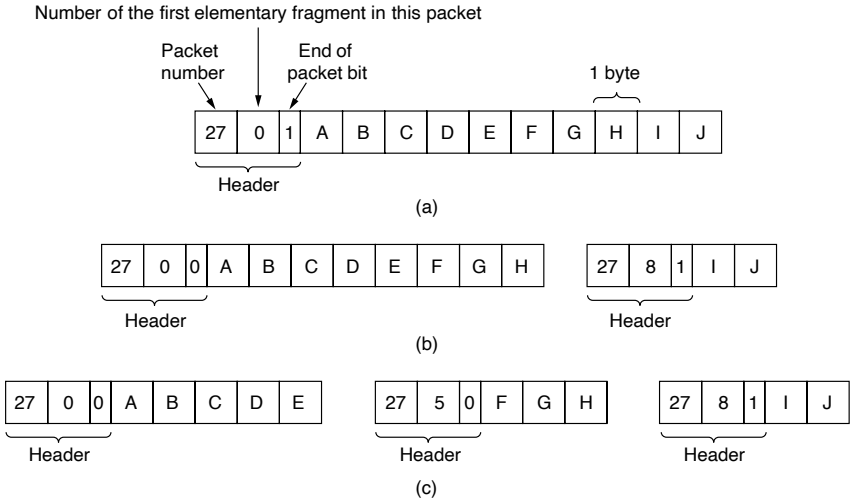


Figure 5-41. Fragmentation when the elementary data size is 1 byte. (a) Original packet, containing 10 data bytes. (b) Fragments after passing through a network with maximum packet size of 8 payload bytes plus header. (c) Fragments after passing through a size 5 gateway.

returns it to the source, and drops the packet. This is shown in Fig. 5-42. 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.

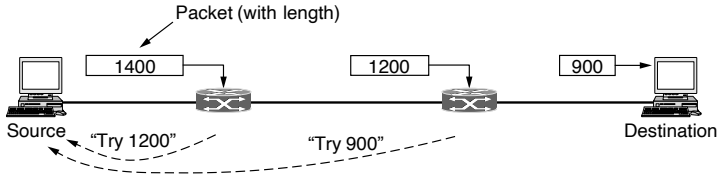


Figure 5-42. Path MTU discovery.

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.

5.6 SOFTWARE-DEFINED NETWORKING

Traffic management and engineering is historically very challenging: it requires network operators to tune the configuration parameters of routing protocols, which then re-compute routes. Traffic flows along the new paths and results in a re-balancing of traffic. Unfortunately, the mechanisms for traffic control in this manner are indirect: changes to routing configuration result in changes to routing both in the network and between networks, and these protocols can interact in unpredictable ways. **SDN** (**Software-Defined Networking**) aims to fix many of these problems. We will discuss it below.

5.6.1 Overview

In a certain way, networks have always been ''software defined,'' in the sense that configurable software running on routers is responsible for looking up information in packets and making forwarding decisions about them. Yet, the software that runs the routing algorithms and implements other logic about packet forwarding was historically vertically integrated with the networking hardware. An operator who bought a Cisco or Juniper router was, in some sense, stuck with the software technology that the vendor shipped with the hardware. For example, making changes to the way OSPF or BGP work was simply not possible. One of the main concepts driving SDN was to recognize that the **control plane**, the software and logic that select routes and decide what to do with forwarding traffic, runs in software and can operate completely separately from the **data plane**, the hardware-based technology that is responsible for actually performing lookups on packets and deciding what to do with them. The two planes are shown in Fig. 5-43.

Given the architectural separation of the control plane and the data plane, the next natural logical step is to recognize that the control plane need not run on the network hardware at all! In fact, one common instantiation of SDN involves a

logically centralized program, often written in a high-level language (e.g., Python, Java, Golang, C) making logical decisions about forwarding and communicating those decisions to every forwarding device in the network. That communication channel between the high-level software program and the underlying hardware could be anything that the network device understands. One of the first SDN controllers used BGP itself as a control plane (Feamster et al., 2003); subsequently, technologies such as OpenFlow, NETCONF, and YANG have emerged as more flexible ways to communicate control-plane information with network devices. In some sense, SDN was a re-incarnation of a well-established idea (i.e., centralized control) at a time when various enablers (open chipset APIs, software control of distributed systems) were also at a level of maturity to enable the architectural ideas to finally gain a foothold.

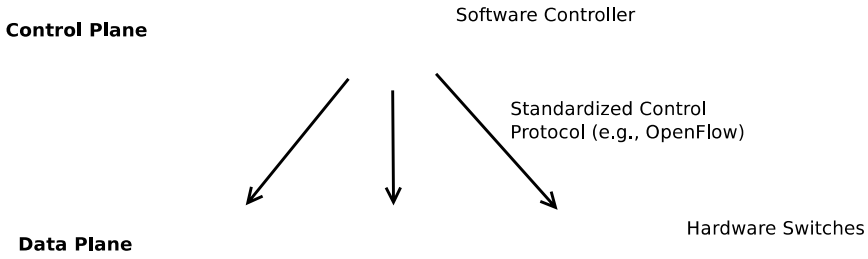


Figure 5-43. Control and data plane separation in SDN.

While the technology of SDN continues to rapidly evolve, the central tenet of the separation of the data and control planes remains invariant. SDN technology has evolved over a number of years; readers who wish to appreciate a complete history of SDN can read further to appreciate the genesis of this increasingly popular technology (Feamster et al., 2013). Below, we survey several of the major trends in SDN: (1) control over routing and forwarding (i.e., the technology behind the control plane); (2) programmable hardware and customizable forwarding (i.e., the technology that makes the data plane more programmable), and (3) programmable network telemetry (a network management application that puts the two pieces together and in many ways may be the ‘‘killer app’’ for SDN).

5.6.2 The SDN Control Plane: Logically Centralized Software Control

One of the main technical ideas that underlies SDN is a control plane that runs separately from the routers, often as a single, logically centralized program. In some sense, SDN has always really existed: routers are configurable, and many

large networks would often even auto-generate their router configuration from a centralized database, keep it in version control, and push those configurations to the routers with scripts. While, in a pedantic sense, this kind of setup could be called an SDN, technically speaking this type of setup only gives operators limited control over how traffic is forwarded through the network. More typically, SDN control programs (sometimes called ''controllers'') are responsible for more of the control logic, such as computing the paths through the network on behalf of the routers, and simply updating the resulting forwarding tables remotely.

Early work in software-defined networking aimed to make it easier for network operators to perform traffic engineering tasks by directly controlling the routes that each router in the network selects, rather than relying on indirect tuning of network configuration parameters. Early incarnations of SDN thus aimed to work within the constraints of existing Internet routing protocols to use them to directly control the routes. One such example was the **RCP** (**Routing Control Platform**) (Feamster et al., 2003), which was subsequently deployed in backbone networks to perform traffic load balancing and defend against denial-of-service attacks. Subsequent developments included a system called Ethane (Casado et al., 2007), which used centralized software control to authenticate hosts within a network. One of the problems with Ethane, however, was that it required customized switches to operate, which limited its deployment in practice.

After demonstrating these benefits of SDN to network management, network operators and vendors began to take notice. Additionally, there was a convenient back door to making the switches even more flexible through a programmable control plane: many network switches relied on a common Broadcom chipset, which had an interface that allowed direct writes into switch memory. A team of researchers worked with switch vendors to expose this interface to software programs, ultimately developing a protocol called **OpenFlow** (McKeown et al, 2008). The OpenFlow protocol was exposed by many switch vendors who were trying to compete with the dominant incumbent switch vendor, Cisco. Initially, the protocol supported a very simple interface: writes into a content-addressable memory that acted as a simple **match-action table**. This match-action table allowed a switch to identify packets that matched one or more fields in the packet header (e.g., MAC address, IP address) and perform one of a set of possible actions, including forwarding the packet to a specific port, dropping it, or sending it to an off-path software controller.

There were multiple versions of the OpenFlow protocol standard. An early version of OpenFlow, version 1.0, had a *single* match-action table, where entries in the table could refer to either exact matches on combinations of packet header fields (e.g., MAC address, IP address) or wild-card entries (e.g., an IP address or MAC address prefix). Later versions of OpenFlow (the most prominent version being OpenFlow 1.3) added more complex operations, including *chains* of tables, but very few vendors ever implemented these standards. Expressing AND and OR conjunctions on these types of matches turned out to be a bit tricky, especially for

programmers, so some technologies emerged to make it easier for programmers to express more complex combinations of conditionals (Foster et al., 2011), and even to incorporate temporal and other aspects into the forwarding decisions (Kim et al., 2015). In the end, adoption of some of these technologies was limited: the OpenFlow protocol gained some traction in large data centers where operators could have complete control over the network. Yet, widespread adoption in wide-area and enterprise networks proved more limited because the operations one could perform in the forward table were so limited. Additionally, many switch vendors never fully implemented later versions of the standard, making it difficult to deploy solutions that depended on these standards in practice. Ultimately, however, the OpenFlow protocol left several important legacies: (1) control over a network with a single, centralized software program, permitting coordination across network devices and forwarding elements, and (2) the ability to express such control over the entire network from a single high-level programming language (e.g., Python, Java).

Ultimately, OpenFlow turned out to be a very limiting interface. It was not designed with flexible network control in mind, but rather was a product of convenience: network devices already had TCAM-based lookup tables in their switches and OpenFlow was, more than anything, a market-driven initiative to open the interface to these tables so that external software programs could write to it. It wasn't long before networking researchers started to think about whether there was a better way to design the hardware as well, to allow for more flexible types of control in the data plane. The next section discusses the developments in programmable hardware that have ultimately made the switches themselves more programmable.

Meanwhile, programmable software control, mostly initially focused on transit and data center networks, is beginning to find its way into cellular networks as well. For example, the Central Office Re-Architected as a Datacenter (CORD) project aims to develop a 5G network from disaggregated commodity hardware and open-source software components (Peterson et al., 2019).

5.6.3 The SDN Data Plane: Programmable Hardware

Recognizing the limitations of the OpenFlow chipset, a subsequent development in SDN was to make the hardware itself programmable. A number of developments in programmable hardware, in both network interface cards (NICs) and switches have made it possible to customize everything from packet format to forwarding behavior.

The general architecture is sometimes called a **protocol-independent switch architecture**. The architecture involves a fixed set of processing pipelines, each with memory for match-action tables, some amount of register memory, and simple operations such as addition (Bosshart et al., 2013). The forwarding model is often referred to as **RMT** (**Reconfigurable Match Tables**), a pipeline architecture that was inspired by RISC architectures. Each stage of the processing pipeline can read information from the packet headers, make modifications to the values in the

header based on simple arithmetic operations, and write back the values to the packets. The processing pipeline is as shown in Fig. 5-44. The chip architecture includes a programmable parser, a set of match stages, which have state and can perform arithmetic computations on packets, as well as perform simple forwarding and dropping decisions, and a ''deparser,'' which writes resulting values back into the packets. Each of the read/modify stages can modify both the state that is maintained at each stage, plus any packet metadata (e.g., information about the queue depth that an individual packet sees).

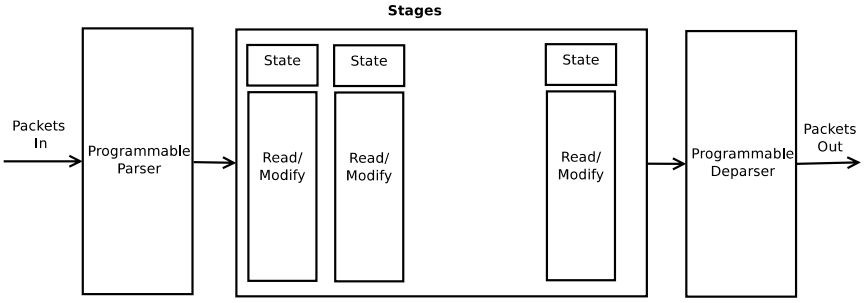


Figure 5-44. Reconfigurable match-action pipeline for a programmable data plane.

The RMT model also allows for custom packet header formats, thus making it possible to store additional information, beyond simply that which is in standard protocol headers, in each packet. RMT makes it possible for a programmer to change aspects of the hardware data plane, without modifying the hardware itself. The programmer can specify multiple match tables of arbitrary size, subject to an overall resource limit. It also gives an operator sufficient flexibility to modify arbitrary header fields.

Modern chipsets, such as the Barefoot Tofino chipset, make it possible to perform protocol-independent custom packet processing on both packet ingress and egress. as shown in Fig. 5-45. The ability to perform customized processing on both ingress and egress makes it possible to perform analytics on queue timings (e.g., how long individual packets spend in queues), as well as customized encapsulation and de-encapsulation. It also makes it possible to perform active queue management (e.g., RED) on egress queues, based on metadata that would be available from ingress queues. Ongoing work is investigating ways to exploit this architecture for traffic and congestion management purposes, such as performing fine-grained queue measurements (Chen et al., 2019).

This level of programmability has generally proved most useful in data-center networks, whose architectures can benefit from high degrees of customizability. On the other hand, the model does also allow for some general improvements and features. For example, the model makes it possible for packets to carry information about the state of the network itself, allowing for such applications as so-called

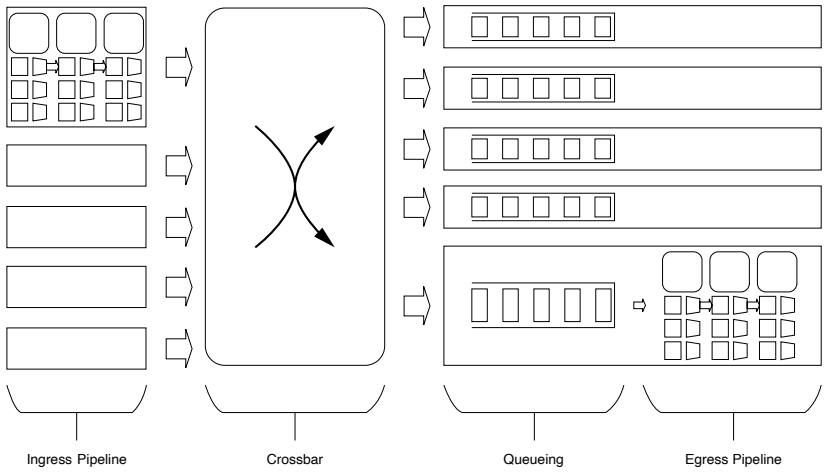


Figure 5-45. Reconfigurable match-action pipelines on both ingress and egress.

INT (**In-band Network Telemetry**), a technology that allows packets to carry information about,for example, the latency along each hop in a network path.

Programmable NICs, libraries such as Intel's Data Plane Development Kit (DPDK), and the emergence of more flexible processing pipelines, such as the Barefoot Tofino chipset, which is programmable with a language called P4 (Bosshart et al., 2014), now make it possible for network operators to develop custom protocols and more extensive packet processing in the switch hardware itself. P4 is a high-level language for programming protocol-independent packet processors such as the RMT chip. Programmable data planes have emerged for software switches, as well (in fact, long before programmable hardware switches). Along these lines, an important development in programmable control over switches was the development of Open vSwitch (OVS), an open-source implementation of a switch that processes packets at multiple layers, operating as a module in the Linux kernel. The software switch offers a range of features, from VLANs to IPv6. The emergence of OVS made it possible for network operators to customize forwarding in data centers, in particular, with OVS running as a switch in the hypervisor of servers in data centers.

5.6.4 Programmable Network Telemetry

One of the more important benefits of SDN is its ability to support programmable network measurement. For many years, network hardware has only exposed a limited amount of information about network traffic, such as aggregate

statistics about traffic flows that the network switch sees (e.g., through standards such as IPFIX). On the other hand, support for the capture of every network packet can also be prohibitive, given the amount of storage and bandwidth that would be required to capture the traffic, as well as the amount of processing that would be required to analyze the data at a later point. For many applications, there is a need to strike a balance between the granularity of packet traces with the scalability of IPFIX aggregates This balance is needed to support network management tasks such as application performance measurement, and for the congestion management tasks that we discussed earlier.

Programmable switch hardware such as that which we discussed in the previous section can enable more flexible telemetry. One trend, for example, is enabling operators to express queries about network traffic in high-level programming languages using frameworks such as MapReduce (Dean and Ghemawat, 2008). Such a paradigm, originally designed for data processing on large clusters, also naturally lends itself to queries about network traffic, for example, how many bytes or packets are destined to a given address or port, within a specified time window? Unfortunately, programmable switch hardware is not (yet) sophisticated enough to support complex queries, and as a result, the query may need to be partitioned across the stream processor and the network switch. Various technologies aim to make it possible to support this type of query partitioning (Gupta et al., 2019). Open research problems involve figuring out how to efficiently map high-level query constructs and abstractions to lower-level switch hardware and software.

One of the final challenges for programmable network telemetry in the coming years is the increasing pervasiveness of encrypted traffic on the Internet. On the one hand, encryption improves privacy by making it difficult for network eavesdroppers to see the contents of user traffic. On the other hand, however, it is also more difficult for network operators to manage their networks when they cannot see the contents of the traffic. One such example concerns tracking the quality of Internet video streams. In the absence of encryption, the contents of the traffic make details such as the video bitrate and resolution apparent. When the traffic is encrypted, these properties must be indirectly inferred, based on properties of the network traffic that can be directly observed (e.g., packet interarrival times, bytes transferred). Recent work has explored ways to automatically infer the higher-level properties of network application traffic from low-level statistics (Bronzino et al., 2020). Network operators will ultimately need better models to help infer how conditions such as congestion affect application performance.

5.7 THE NETWORK LAYER IN THE INTERNET

It is now time to discuss the network layer of the Internet in detail. But before getting into specifics, it is worth taking a look at the principles that drove its design in the past and made it the success that it is today. All too often, nowadays, people

seem to have forgotten them. These principles are enumerated and discussed in RFC 1958, which is well worth reading (and should be mandatory for all protocol designers—with a final exam at the end). This RFC draws heavily on ideas put forth by Clark (1988) and Saltzer et al. (1984). We will now summarize what we consider to be the top 10 principles (from most important to least important).

1. **Make sure it works.** Do not finalize the design or standard until multiple prototypes have successfully communicated with each other. All too often, designers first write a 1000-page standard, get it approved, then discover it is deeply flawed and does not work. Then they write version 1.1 of the standard. This is not the way to go.
2. **Keep it simple.** When in doubt, use the simplest solution. William of Occam stated this principle (Occam's razor) in the 14th century. Put in modern terms: fight features. If a feature is not absolutely essential, leave it out, especially if the same effect can be achieved by combining other features.
3. **Make clear choices.** If there are several ways of doing the same thing, choose one. Having two or more ways to do the same thing is looking for trouble. Standards often have multiple options or modes or parameters because several powerful parties insist that their way is best. Designers should strongly resist this tendency. Just say no.
4. **Exploit modularity.** This principle leads directly to the idea of having protocol stacks, each of whose layers is independent of all the other ones. In this way, if circumstances require one module or layer to be changed, the other ones will not be affected.
5. **Expect heterogeneity.** Different types of hardware, transmission facilities, and applications will occur on any large network. To handle them, the network design must be simple, general, and flexible.
6. **Avoid static options and parameters.** If parameters are unavoidable (e.g., maximum packet size), it is best to have the sender and receiver negotiate a value rather than defining fixed choices.
7. **Look for a good design; it need not be perfect.** Often, the designers have a good design but it cannot handle some weird special case. Rather than messing up the design, the designers should go with the good design and put the burden of working around it on the people with the strange requirements.
8. **Be strict when sending and tolerant when receiving.** In other words, send only packets that rigorously comply with the standards, but expect incoming packets that may not be fully conformant and try to deal with them.

9. **Think about scalability.** If the system is to handle millions of hosts and billions of users effectively, no centralized databases of any kind are tolerable and load must be spread as evenly as possible over the available resources.

10. **Consider performance and cost.** If a network has poor performance or outrageous costs, nobody will use it.

Let us now leave the general principles and start looking at the details of the Internet's network layer. In the network layer, the Internet can be viewed as a collection of networks or Autonomous Systems (ASes) that are interconnected. There is no real structure, but several major backbones exist. These are constructed from high-bandwidth lines and fast routers.

The biggest of these backbones, to which everyone else connects to reach the rest of the Internet, are called **Tier 1 networks**. Attached to the backbones are ISPs (Internet Service Providers) that provide Internet access to homes and businesses, data centers and colocation facilities full of server machines, and regional (mid-level) networks. The data centers serve much of the content that is sent over the Internet. Attached to the regional networks are more ISPs, LANs at many universities and companies, and other edge networks. A sketch of this quasihierarchical organization is given in Fig. 5-46.

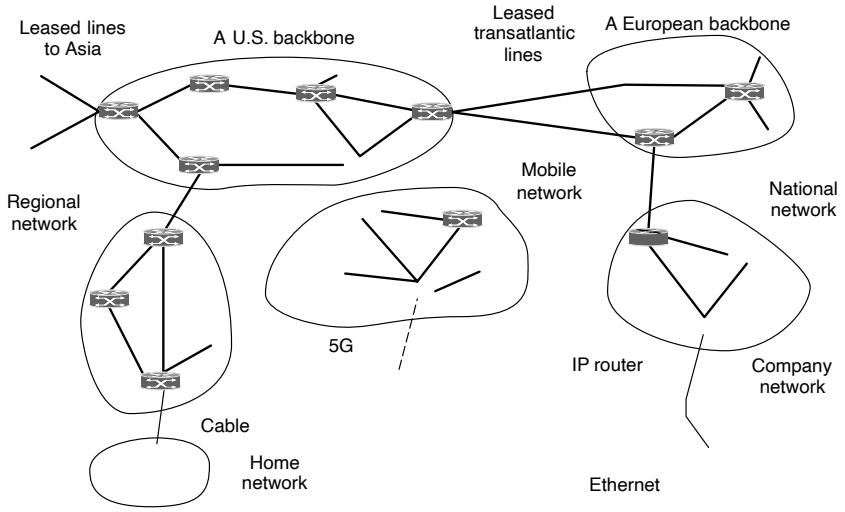


Figure 5-46. The Internet is an interconnected collection of many networks.

The glue that holds the whole Internet together is the network layer protocol, **IP** (**Internet Protocol**). Unlike almost all older network layer protocols, IP was

designed from the beginning with internetworking in mind. A good way to think of the network layer is this: its job is to provide a best-effort (i.e., not guaranteed) way to transport packets from source to destination, without regard to whether these machines are on the same network or whether there are other networks in between them.

Communication in the Internet works as follows. The transport layer takes data streams and breaks them up so that they may be sent as IP packets. In theory, packets can be up to 64 KB each, but in practice they are usually not more than 1500 bytes (so they fit in one Ethernet frame). IP routers forward each packet through the Internet, along a path from one router to the next, until the destination is reached. At the destination, the network layer hands the data to the transport layer, which gives it to the receiving process. When all the pieces finally get to the destination machine, they are reassembled by the network layer into the original datagram. This datagram is then handed to the transport layer.

In the example of Fig. 5-46, a packet originating at a host on the home network has to traverse four networks and a large number of IP routers before even getting to the company network on which the destination host is located. This is not unusual in practice, and there are many longer paths. There is also much redundant connectivity in the Internet, with backbones and ISPs connecting to each other in multiple locations. This means that there are many possible paths between two hosts. It is the job of the IP routing protocols to decide which paths to use.

5.7.1 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. 5-47. 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 *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 2^{31} 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.

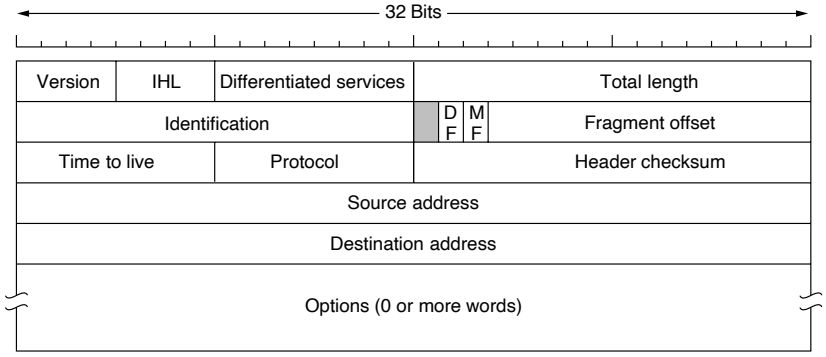


Figure 5-47. The IPv4 (Internet Protocol version 4) header.

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, but it was a nice try.

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. 5.5.6.

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 halfwords 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. We will discuss Internet addresses in the next section.

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. 5-48 were defined.

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

Figure 5-48. Some of the IP options.

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 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 or performance 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, like: ''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.

5.7.2 IP Addresses

A defining feature of IPv4 is its 32-bit addresses. Every host and router on the Internet has an IP address that can be used in the *Source address* and *Destination address* fields of IP packets. It is important to note that an IP address does not actually refer to a host. It really refers to a network interface, so if a host is on two networks, it must have two IP addresses. However, in practice, most hosts are on one network and thus have one IP address. In contrast, routers have multiple interfaces and thus multiple IP addresses.

Prefixes

IP addresses are hierarchical, unlike Ethernet addresses. Each 32-bit address is comprised of a variable-length network portion in the top bits and a host portion in the bottom bits. The network portion has the same value for all hosts on a single network, such as an Ethernet LAN. This means that a network corresponds to a contiguous block of IP address space. This block is called a **prefix**.

IP addresses are written in **dotted decimal notation**. In this format, each of the 4 bytes is written in decimal, from 0 to 255. For example, the 32-bit hexadecimal address 80D00297 is written as 128.208.2.151. Prefixes are written by giving the lowest IP address in the block and the size of the block. The size is determined by the number of bits in the network portion; the remaining bits in the host portion can vary. This means that the size must be a power of two. By convention, it is written after the prefix IP address as a slash followed by the length in bits of the network portion. In our example, if the prefix contains 2^8 addresses and so leaves 24 bits for the network portion, it is written as 128.208.2.0/24.

Since the prefix length cannot be inferred from the IP address alone, routing protocols must carry the prefixes to routers. Sometimes prefixes are simply described by their length, as in a ''/16'' which is pronounced ''slash 16.'' The length of the prefix corresponds to a binary mask of 1s in the network portion. When written out this way, it is called a **subnet mask**. It can be ANDed with the IP address to extract only the network portion. For our example, the subnet mask is 255.255.255.0. Fig. 5-49 shows a prefix and a subnet mask.

Hierarchical addresses have significant advantages and disadvantages. The key advantage of prefixes is that routers can forward packets based on only the network portion of the address, as long as each of the networks has a unique address block. The host portion does not matter at all to the routers because all hosts on

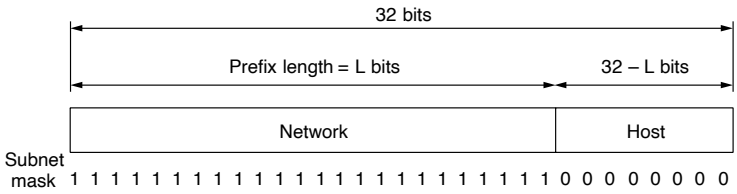


Figure 5-49. An IP prefix and a subnet mask.

the same network will be sent in the same direction. It is only when the packets reach the network for which they are destined that they are forwarded to the correct host. This makes the routing tables much smaller than they would otherwise be. Consider that the number of hosts on the Internet is approaching one billion. That would be a very large table for every router to keep. However, by using a hierarchy, routers need to keep routes for only around 300,000 prefixes.

While using a hierarchy lets Internet routing scale, it has two disadvantages. First, the IP address of a host depends on where it is located in the network. An Ethernet address can be used anywhere in the world, but every IP address belongs to a specific network, and routers will only be able to deliver packets destined to that address to the network. Designs such as mobile IP are needed to support hosts that move between networks but want to keep the same IP addresses.

The second disadvantage is that the hierarchy is wasteful of addresses unless it is carefully managed. If addresses are assigned to networks in (too) large blocks, there will be (many) addresses that are allocated but not in use. This allocation would not matter much if there were plenty of addresses to go around. However, it was realized more than two decades ago that the tremendous growth of the Internet was rapidly depleting the free address space. IPv6 is the solution to this shortage, but until it is widely deployed there will be great pressure to allocate IP addresses so that they are used very efficiently.

Subnets

Network numbers are managed by a nonprofit corporation called **ICANN** (**Internet Corporation for Assigned Names and Numbers**), to avoid conflicts. In turn, ICANN has delegated parts of the address space to various regional authorities, which dole out IP addresses to ISPs and other companies. This is the process by which a company is allocated a block of IP addresses.

However, this process is only the start of the story, as IP address assignment is ongoing as companies grow. We have said that routing by prefix requires all the hosts in a network to have the same network number. This property can cause problems as networks grow. For example, let us consider a university that started out with our example /16 prefix for use by the Computer Science Dept. for the

computers on its Ethernet. A year later, the Electrical Engineering Dept. wants to get on the Internet. The Art Dept. soon follows suit. What IP addresses should these departments use? Getting further blocks requires going outside the university and may be expensive or inconvenient. Moreover, the /16 already allocated has enough addresses for over 60,000 hosts. It might be intended to allow for significant growth, but until that happens, it is wasteful to allocate further blocks of IP addresses to the same university. A different organization is required.

The solution is to allow the block of addresses to be split into several parts for internal use as multiple networks, while still acting like a single network to the outside world. This is called **subnetting** and the networks (such as Ethernet LANs) that result from dividing up a larger network are called **subnets**. As we mentioned in Chap. 1, you should be aware that this new usage of the term conflicts with older usage of ''subnet'' to mean the set of all routers and communication lines in a network.

Figure 5-50 shows how subnets can help with our example. The single /16 has been split into pieces. This split does not need to be even, but each piece must be aligned so that any bits can be used in the lower host portion. In this case, half of the block (a /17) is allocated to the Computer Science Dept., a quarter is allocated to the Electrical Engineering Dept. (a /18), and one-eighth (a /19) to the Art Dept. The remaining eighth is unallocated. A different way to see how the block was divided is to look at the resulting prefixes when written in binary notation:

Computer Science:	10000000	11010000	1lxxxxxxx	xxxxxxxx
Electrical Eng.:	10000000	11010000	00lxxxxxx	xxxxxxxx
Art:	10000000	11010000	011lxxxxx	xxxxxxxx

Here, the vertical bar (l) shows the boundary between the subnet number and the host portion.

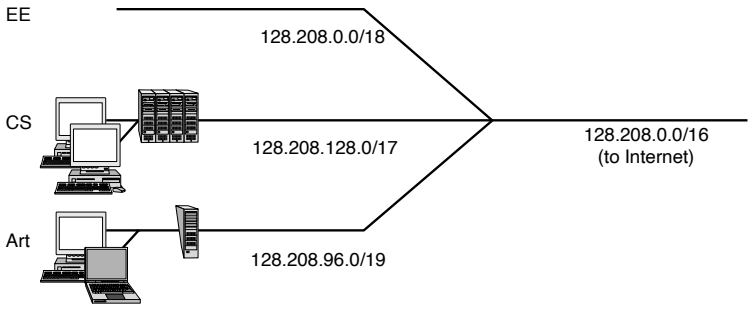


Figure 5-50. Splitting an IP prefix into separate networks with subnetting.

When a packet comes into the main router, how does the router know which subnet to give it to? This is where the details of our prefixes come in. One way

would be for each router to have a table with 65,536 entries telling it which outgoing line to use for each host on campus. But this would undermine the main scaling benefit we get from using a hierarchy. Instead, the routers simply need to know the subnet masks for the networks on campus.

When a packet arrives, the router looks at the destination address of the packet and checks which subnet it belongs to. The router can do this by ANDing the destination address with the mask for each subnet and checking to see if the result is the corresponding prefix. For example, consider a packet destined for IP address 128.208.2.151. To see if it is for the Computer Science Dept., we AND with 255.255.128.0 to take the first 17 bits (which is 128.208.0.0) and see if they match the prefix address (which is 128.208.128.0). They do not match. Checking the first 18 bits for the Electrical Engineering Dept., we get 128.208.0.0 when ANDing with the subnet mask. This does match the prefix address, so the packet is forwarded onto the interface that leads to the Electrical Engineering network.

The subnet divisions can be changed later if necessary, by updating all subnet masks at routers inside the university. Outside the network, the subnetting is not visible, so allocating a new subnet does not require contacting ICANN or changing any external databases.

CIDR—Classless InterDomain Routing

Even if blocks of IP addresses are allocated so that the addresses are used efficiently, there is still a problem that remains: routing table explosion.

Routers in organizations at the edge of a network, such as a university, need to have an entry for each of their subnets, telling the router which line to use to get to that network. For routes to destinations outside of the organization, they can use the simple default rule of sending the packets on the line toward the ISP that connects the organization to the rest of the Internet. The other destination addresses must all be out there somewhere.

Routers in ISPs and backbones in the middle of the Internet have no such luxury. They must know which way to go to get to every network and no simple default will work. These core routers are said to be in the **default-free zone** of the Internet. No one really knows how many networks are connected to the Internet any more, but it is a large number, probably at least a million. This can make for a very large table. It may not sound large by computer standards, but realize that routers must perform a lookup in this table to forward every packet, and routers at large ISPs may forward up to millions of packets per second. Specialized hardware and fast memory are needed to process packets at these rates, not a general-purpose computer.

In addition, routing algorithms require each router to exchange information about the addresses it can reach with other routers. The larger the tables, the more information needs to be communicated and processed. The processing grows at least linearly with the table size. Greater communication increases the likelihood

that some parts will get lost, at least temporarily, possibly leading to routing instabilities.

The routing table problem could have been solved by going to a deeper hierarchy, like the telephone network. For example, having each IP address contain a country, state/province, city, network, and host field might work. Then each router would only need to know how to get to each country, the states or provinces in its own country, the cities in its state or province, and the networks in its city. Unfortunately, this solution would require considerably more than 32 bits for IP addresses and would use addresses inefficiently (and Liechtenstein would have as many bits in its addresses as the United States).

Fortunately, there is something we can do to reduce routing table sizes. We can apply the same insight as subnetting: routers at different locations can know about a given IP address as belonging to prefixes of different sizes. However, instead of splitting an address block into subnets, here we combine multiple small prefixes into a single larger prefix. This process is called **route aggregation**. The resulting larger prefix is sometimes called a **supernet**, to contrast with subnets as the division of blocks of addresses.

With aggregation, IP addresses are contained in prefixes of varying sizes. The same IP address that one router treats as part of a /22 (a block containing 2^{10} addresses) may be treated by another router as part of a larger /20 (which contains 2^{12} addresses). It is up to each router to have the corresponding prefix information. This design works with subnetting and is called **CIDR** (**Classless InterDomain Routing**), which is pronounced ''cider,'' as in the drink. The most recent version of it is specified in RFC 4632 (Fuller and Li, 2006). The name highlights the contrast with addresses that encode hierarchy with classes, which we will describe shortly.

To make CIDR easier to understand, let us consider an example in which a block of 8192 IP addresses is available starting at 194.24.0.0. Suppose that Cambridge University needs 2048 addresses and is assigned the addresses 194.24.0.0 through 194.24.7.255, along with mask 255.255.248.0. This is a /21 prefix. Next, Oxford University asks for 4096 addresses. Since a block of 4096 addresses must lie exactly on a 4096-byte boundary, Oxford cannot be given addresses starting at 194.24.8.0. Instead, it gets 194.24.16.0 through 194.24.31.255, along with subnet mask 255.255.240.0. Finally, the University of Edinburgh asks for 1024 addresses and is then assigned addresses 194.24.8.0 through 194.24.11.255 and also mask 255.255.252.0. These assignments are summarized in Fig. 5-51.

All of the routers in the default-free zone are now told about the IP addresses in the three networks. Routers close to the universities may need to send on a different outgoing line for each of the prefixes, so they need an entry for each of the prefixes in their routing tables. An example is the router in London in Fig. 5-52.

Now let us look at these three universities from the point of view of a distant router in New York. All of the IP addresses in the three prefixes should be sent from New York (or the U.S. in general) to London. The routing process in London

University	First address	Last address	How many	Prefix
Cambridge	194.24.0.0	194.24.7.255	2048	194.24.0.0/21
Edinburgh	194.24.8.0	194.24.11.255	1024	194.24.8.0/22
(Available)	194.24.12.0	194.24.15.255	1024	194.24.12.0/22
Oxford	194.24.16.0	194.24.31.255	4096	194.24.16.0/20

Figure 5-51. A set of IP address assignments.

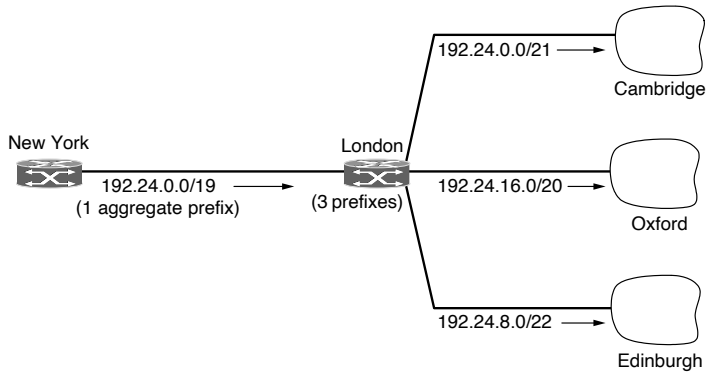


Figure 5-52. Aggregation of IP prefixes.

notices this and combines the three prefixes into a single aggregate entry for the prefix 194.24.0.0/19 that it passes to the New York router. This prefix contains 8K addresses and covers the three universities and the otherwise unallocated 1024 addresses. By using aggregation, three prefixes have been reduced to one, reducing the prefixes that the New York router must be told about and the routing table entries in the New York router.

When aggregation is turned on, it is an automatic process. It depends on which prefixes are located where in the Internet not on the actions of an administrator assigning addresses to networks. Aggregation is heavily used throughout the Internet and can reduce the size of router tables to around 200,000 prefixes.

As a further twist, prefixes are allowed to overlap. The rule is that packets are sent in the direction of the most specific route, or the **longest matching prefix** that has the fewest IP addresses. Longest matching prefix routing provides a useful degree of flexibility, as seen in the behavior of the router at New York in Fig. 5-53. This router still uses a single aggregate prefix to send traffic for the three universities to London. However, the previously available block of addresses within this prefix has now been allocated to a network in San Francisco. One possibility is for the New York router to keep four prefixes, sending packets for three of them to

London and packets for the fourth to San Francisco. Instead, longest matching prefix routing can handle this forwarding with the two prefixes that are shown. One overall prefix is used to direct traffic for the entire block to London. One more specific prefix is also used to direct a portion of the larger prefix to San Francisco. With the longest matching prefix rule, IP addresses within the San Francisco network will be sent on the outgoing line to San Francisco, and all other IP addresses in the larger prefix will be sent to London.

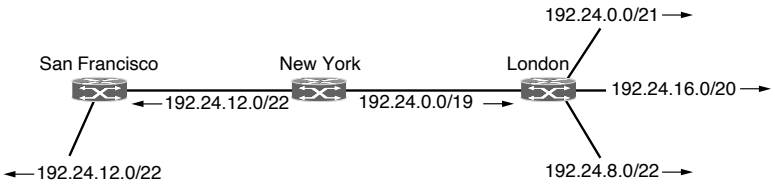


Figure 5-53. Longest matching prefix routing at the New York router.

Conceptually, CIDR works as follows. When a packet comes in, the routing table is scanned to determine if the destination lies within the prefix. It is possible that multiple entries with different prefix lengths will match, in which case the entry with the longest prefix is used. Thus, if there is a match for a /20 mask and a /24 mask, the /24 entry is used to look up the outgoing line for the packet. However, this process would be tedious if the table were really scanned entry by entry. Instead, complex algorithms have been devised to speed up the address matching process (Ruiz-Sanchez et al., 2001). Commercial routers use custom VLSI chips with these algorithms embedded in hardware.

Classful and Special Addressing

To help you better appreciate why CIDR is so useful, we will briefly relate the design that predated it. Before 1993, IP addresses were divided into the five categories listed in Fig. 5-54. This allocation has come to be called **classful addressing**.

The class A, B, and C formats allow for up to 128 networks with 16 million hosts each, 16,384 networks with up to 65,536 hosts each, and 2 million networks (e.g., LANs) with up to 256 hosts each (although a few of these are special). Also supported is multicast (the class D format), in which a datagram is directed to multiple hosts. Addresses beginning with 1111 are reserved for use in the future. They would be valuable to use now given the depletion of the IPv4 address space. Unfortunately, many hosts will not accept these addresses as valid because they have been off-limits for so long and it is hard to teach old hosts new tricks.

This is a hierarchical design, but unlike CIDR the sizes of the address blocks are fixed. Over 2 billion 21-bit addresses exist, but organizing the address space

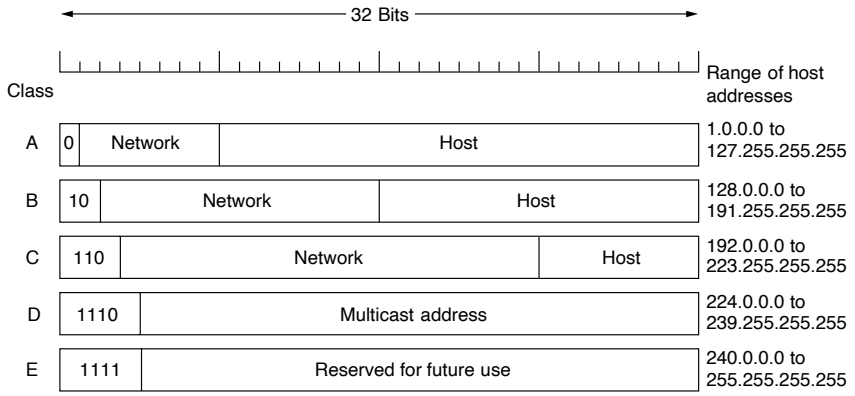


Figure 5-54. IP address formats.

by classes wastes millions of them. In particular, the real villain is the class B network. For most organizations, a class A network, with 16 million addresses, is too big, and a class C network, with 256 addresses is too small. A class B network, with 65,536, is just right. In Internet folklore, this situation is known as the **three bears problem** as in *Goldilocks and the Three Bears* (Southey, 1848).

In reality, though, a class B address is far too large for most organizations. Studies have shown that more than half of all class B networks have fewer than 50 hosts. A class C network would have done the job, but no doubt every organization that asked for a class B address thought that one day it would outgrow the 8-bit host field. In retrospect, it might have been better to have had class C networks use 10 bits instead of 8 for the host number, allowing 1022 hosts per network. Had this been the case, most organizations would probably have settled for a class C network, and there would have been half a million of them (versus only 16,384 class B networks).

It is hard to fault the Internet's designers for not having provided more (and smaller) class B addresses. At the time the decision was made to create the three classes, the Internet was a research network connecting the major research universities in the U.S. (plus a very small number of companies and military sites doing networking research). No one then perceived the Internet becoming a mass-market communication system rivaling the telephone network. At the time, someone no doubt said: ''The U.S. has about 2000 colleges and universities. Even if all of them connect to the Internet and many universities in other countries join, too, we are never going to hit 16,000, since there are not that many universities in the whole world. Furthermore, having the host number be an integral number of bytes speeds up packet processing'' (which was then done entirely in software). Perhaps some day people will look back and fault the folks who designed the telephone

number scheme and say: ''What idiots. Why didn't they include the planet number in the phone number?'' But at the time, it did not seem necessary.

To handle these problems, subnets were introduced to flexibly assign blocks of addresses within an organization. Later, CIDR was added to reduce the size of the global routing table. Today, the bits that indicate whether an IP address belongs to class A, B, or C network are no longer used, though references to these classes in the literature are still common.

To see how dropping the classes made forwarding more complicated, consider how simple it was in the old classful system. When a packet arrived at a router, a copy of the IP address was shifted right 28 bits to yield a 4-bit class number. A 16-way branch then sorted packets into A, B, C (and D and E) classes, with eight of the cases for class A, four of the cases for class B, and two of the cases for class C. The code for each class then masked off the 8-, 16-, or 24-bit network number and right aligned it in a 32-bit word. The network number was then looked up in the A, B, or C table, usually by indexing for A and B networks and hashing for C networks. Once the entry was found, the outgoing line could be looked up and the packet forwarded. This is much simpler than the longest matching prefix operation, which can no longer use a simple table lookup because an IP address may have any length prefix.

Class D addresses continue to be used in the Internet for multicast. Actually, it might be more accurate to say that they are starting to be used for multicast, since Internet multicast has not been widely deployed in the past.

There are also several other addresses that have special meanings, as shown in Fig. 5-55. The IP address 0.0.0.0, the lowest address, is used by hosts when they are being booted. It means ''this network'' or ''this host.'' IP addresses with 0 as the network number refer to the current network. These addresses allow machines to refer to their own network without knowing its number (but they have to know the network mask to know how many 0s to include). The address consisting of all 1s, or 255.255.255.255—the highest address—is used to mean all hosts on the indicated network. It allows broadcasting on the local network, typically a LAN. The addresses with a proper network number and all 1s in the host field allow machines to send broadcast packets to distant LANs anywhere in the Internet. However, many network administrators disable this feature as it is mostly a security hazard. Finally, all addresses of the form 127.*xx.yy.zz* are reserved for loopback testing. Packets sent to that address are not put out onto the wire; they are processed locally and treated as incoming packets. This allows packets to be sent to the host without the sender knowing its number, which is useful for testing.

NAT—Network Address Translation

IP addresses are scarce. An ISP might have a /16 address, giving it 65,534 usable host numbers. If it has more customers than that, it has a problem. In fact, with 32-bit addresses, there are only 2^{32} of them and they are all gone.

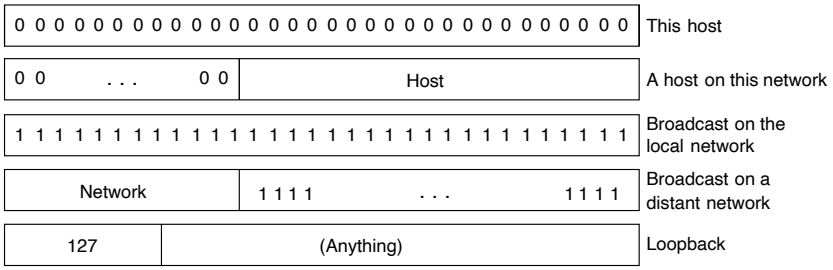


Figure 5-55. Special IP addresses.

This scarcity has led to techniques to use IP addresses sparingly. One approach is to dynamically assign an IP address to a computer when it is on and using the network, and to take the IP address back when the host becomes inactive. The IP address can then be assigned to another computer that becomes active. In this way, a single /16 address can handle up to 65,534 active users.

This strategy works well in some cases, for example, for dialup networking and mobile and other computers that may be temporarily absent or powered off. However, it does not work very well for business customers. Many PCs in businesses are expected to be on continuously. Some are employee machines, backed up at night, and some are servers that may have to serve a remote request at a moment's notice. These businesses have an access line that always provides connectivity to the rest of the Internet.

Increasingly, this situation also applies to home users subscribing to ADSL or Internet over cable, since there is no hourly connection charge (as there once was), just a monthly flat rate charge). Many of these users have two or more computers at home, often one for each family member, and they all want to be online all the time. The solution is to connect all the computers into a home network via a LAN and put a (wireless) router on it. The router then connects to the ISP. From the ISP's point of view, the family is now the same as a small business with a handful of computers. Welcome to Jones, Inc. With the techniques we have seen so far, each computer must have its own IP address all day long. For an ISP with many thousands of customers, particularly business customers and families that are just like small businesses, the demand for IP addresses can quickly exceed the block that is available.

The problem of running out of IP addresses is not a theoretical one that might occur at some point in the distant future. It is happening right here and right now. The long-term solution is for the whole Internet to migrate to IPv6, which has 128-bit addresses. This transition is slowly occurring, but it will be years before the process is complete. To get by in the meantime, a quick fix was needed. The quick fix that is widely used today came in the form of **NAT** (**Network Address**

Translation), which is described in RFC 3022 and which we will summarize below. For additional information, see Dutcher (2001).

The basic idea behind NAT is for the ISP to assign each home or business a single IP address (or at most, a small number of them) for Internet traffic. *Within* the customer network, every computer gets a unique IP address, which is used for routing intramural traffic. However, just before a packet exits the customer network and goes to the ISP, an address translation from the unique internal IP address to the shared public IP address takes place. This translation makes use of three ranges of IP addresses that have been declared as private. Networks may use them internally as they wish. The only rule is that no packets containing these addresses may appear on the Internet itself. The three reserved ranges are:

10.0.0.0	– 10.255.255.255/8	(16,777,216 hosts)
172.16.0.0	– 172.31.255.255/12	(1,048,576 hosts)
192.168.0.0	– 192.168.255.255/16	(65,536 hosts)

The first range provides for 16,777,216 addresses (except for all 0s and all 1s, as usual) and is the usual choice, even if the network is not large.

The operation of NAT is shown in Fig. 5-56. Within the customer premises, every machine has a unique address of the form 10.*x.y.z*. However, before a packet leaves the customer premises, it passes through a **NAT box** that converts the internal IP source address, 10.0.0.1 in the figure, to the customer's true IP address, 198.60.42.12 in this example. The NAT box is often combined in a single device with a firewall, which provides security by carefully controlling what goes into the customer network and what comes out of it. We will study firewalls in Chap. 8. It is also possible to integrate the NAT box into a router or ADSL modem.

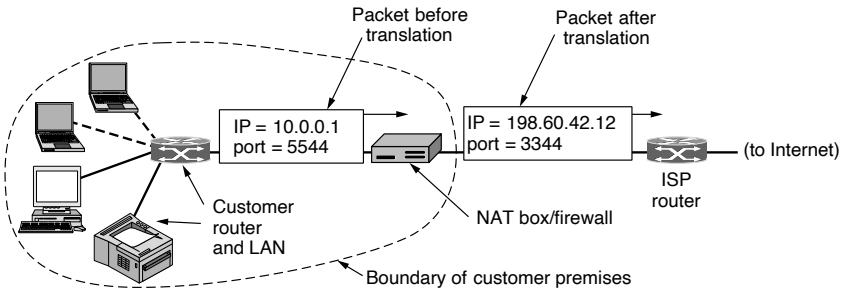


Figure 5-56. Placement and operation of a NAT box.

So far, we have glossed over one tiny but crucial detail: when the reply comes back (e.g., from a Web server), it is naturally addressed to 198.60.42.12, so how does the NAT box know which internal address to replace it with? Herein lies the problem with NAT. If there were a spare field in the IP header, that field could be used to keep track of who the real sender was, but only 1 bit is still unused. In

principle, a new option could be created to hold the true source address, but doing so would require changing the IP code on all the machines on the entire Internet to handle the new option. This is not a promising alternative for a quick fix.

What actually happens is as follows. The NAT designers observed that most IP packets carry either TCP or UDP payloads. When we study TCP and UDP in Chap. 6, we will see that both of these have headers containing a source port and a destination port. Below we will just discuss TCP ports, but exactly the same story holds for UDP ports. The ports are 16-bit integers that indicate where the TCP connection begins and ends. These ports provide the field needed to make NAT work.

When a process wants to establish a TCP connection with a remote process, it attaches itself to an unused TCP port on its own machine. This is called the **source port** and tells the TCP code where to send incoming packets belonging to this connection. The process also supplies a **destination port** to tell who to give the packets to on the remote side. Ports 0–1023 are reserved for well-known services. For example, port 80 is the port used by Web servers, so remote clients can locate them. Each outgoing TCP message contains both a source port and a destination port. Together, these ports serve to identify the processes using the connection on both ends.

An analogy may make the use of ports clearer. Imagine a company with a single main telephone number. When people call the main number, they reach an operator who asks which extension they want and then puts them through to that extension. The main number is analogous to the customer's IP address and the extensions on both ends are analogous to the ports. Ports are effectively an extra 16 bits of addressing that identify which process gets which incoming packet.

Using the *Source port* field, we can solve our mapping problem. Whenever an outgoing packet enters the NAT box, the 10.*x.y.z* source address is replaced by the customer's true IP address. In addition, the TCP *Source port* field is replaced by an index into the NAT box's 65,536-entry translation table. This table entry contains the original IP address and the original source port. Finally, both the IP and TCP header checksums are recomputed and inserted into the packet. It is necessary to replace the *Source port* because connections from machines 10.0.0.1 and 10.0.0.2 may both happen to use port 5000, for example, so the *Source port* alone is not enough to identify the sending process.

When an incoming packet arrives at the NAT box from the ISP, the *Destination port* in the TCP header is extracted and used as an index into the NAT box's mapping table. From the entry located, the internal IP address and original TCP port are extracted and inserted into the packet. Then both the IP and TCP checksums are recomputed and inserted into the packet. The packet is then passed to the customer router for normal delivery using the 10.*x.y.z* address.

Although this scheme sort of solves the problem, networking purists in the IP community have a tendency to regard it as an abomination-on-the-face-of-the-earth. Briefly summarized, here are some of the objections. First, NAT violates

the architectural model of IP, which states that every IP address uniquely identifies a single machine worldwide. The whole software structure of the Internet is built on this fact. With NAT, thousands of machines may (and do) use address 10.0.0.1.

Second, NAT breaks the end-to-end connectivity model of the Internet, which says that any host can send a packet to any other host at any time. Since the mapping in the NAT box is set up by outgoing packets, incoming packets cannot be accepted until after an outgoing one is sent. In practice, this means that a home user with NAT can make TCP/IP connections to a remote Web server, but a remote user cannot make connections to a game server on the home network. Special configuration or **NAT traversal** techniques are needed to support this situation.

Third, NAT changes the Internet from a connectionless network to a very strange kind of connection-oriented network. The problem is that the NAT box must maintain state (i.e., the mapping) for each connection passing through it. Having the network maintain connection state is a property of connection-oriented networks, not a connectionless one. If the NAT box crashes and its mapping table is lost, all its TCP connections are destroyed. In the absence of NAT, a router can crash and restart with no long-term effect on TCP connections. The sending process just times out within a few seconds and retransmits all unacknowledged packets. With NAT, the Internet becomes as vulnerable as a circuit-switched network.

Fourth, NAT violates the most fundamental rule of protocol layering: layer k may not make any assumptions about what layer $k + 1$ has put into the payload field. This basic principle is there to keep the layers independent. If TCP is later upgraded to TCP-2, with a different header layout (e.g., 32-bit ports), NAT will fail. The whole idea of layered protocols is to ensure that changes in one layer do not require changes in other layers. NAT destroys this independence.

Fifth, processes on the Internet are not required to use TCP or UDP. If a user on machine *A* decides to use some new transport protocol to talk to a user on machine *B* (e.g., for a multimedia application), introduction of a NAT box will cause the application to fail because the NAT box will not be able to locate the TCP *Source port* correctly.

A sixth and related problem is that some applications use multiple TCP/IP connections or UDP ports in prescribed ways. For example, **FTP**, the standard **File Transfer Protocol**, inserts IP addresses in the body of packet for the receiver to extract and use. Since NAT knows nothing about these arrangements, it cannot rewrite the IP addresses or otherwise account for them. This lack of understanding means that FTP and other applications such as the H.323 Internet telephony protocol (which we will study in Chap. 7) will fail in the presence of NAT unless special precautions are taken. It is often possible to patch NAT for these cases, but having to patch the code in the NAT box for every new application is not a good idea.

Finally, since the TCP *Source port* field is 16 bits, at most 65,536 machines can be mapped onto an IP address. Actually, the number is slightly less because the first 4096 ports are reserved for special uses. However, if multiple IP addresses are available, each one can handle up to 61,440 machines.

A view of these and other problems with NAT is given in RFC 2993. Despite the issues, NAT is widely used in practice, especially for home and small business networks, as the only expedient technique to deal with the IP address shortage. It has become wrapped up with firewalls and privacy because it blocks unsolicited incoming packets by default. For this reason, it is unlikely to go away even when IPv6 is widely deployed.

5.7.3 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 were allocated on Nov. 25, 2019. 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 in only a fraction of the Internet (estimates are 25%) despite having been an Internet Standard since 1998. The next several years will be an interesting time. Each IPv4 address is now worth as much as $19. In 2019, a man was convicted of stockpiling 750,000 IP addresses (worth about $14 million) and selling them on the black market.

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, especially 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 2^5) 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** (**Internet Protocol version 6**).

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 RFC 2460 through RFC 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 half-hearted 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. 5-57. 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 demultiplexing, 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 continue to be vigorous and lengthy for years to come.

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

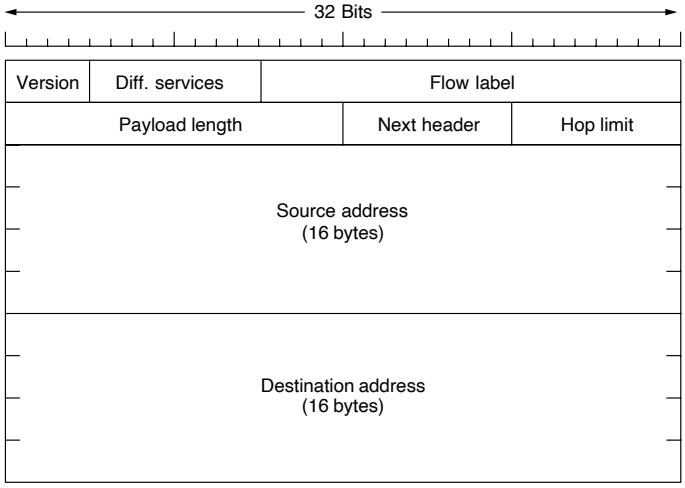


Figure 5-57. The IPv6 fixed header (required).

same way by the network, forming a pseudoconnection. 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 2^{20} 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. 5-57. 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-byte addresses. Specifically, there are 2^{128} of them, which is approximately 3×10^{38}. If the entire earth, land and water, were covered with computers, IPv6 would allow 7×10^{23} IP 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. 5-47) with the IPv6 header (Fig. 5-57) 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. 5.5.6. 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. 5-58. 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.

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.

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

Figure 5-58. IPv6 extension headers.

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 Fig. 5-59. When it is used, the *Payload length* field in the fixed header is set to 0.

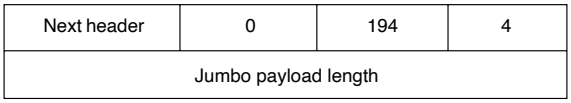


Figure 5-59. The hop-by-hop extension header for large datagrams (jumbograms).

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 **jumbograms**. The use of jumbograms 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. 5-60.

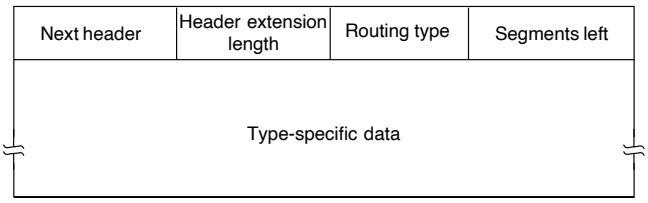


Figure 5-60. The extension header for routing.

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.

The fragment header deals with fragmentation similarly to the way IPv4 does. The header holds the datagram identifier, fragment number, and a bit telling whether more fragments will follow. In IPv6, unlike in IPv4, only the source host can fragment a packet. Routers along the way may not do this. This change is a major philosophical break with the original IP, but in keeping with current practice for IPv4. Plus, it simplifies the routers' work and makes routing go faster. As mentioned above, if a router is confronted with a packet that is too big, it discards the packet and sends an ICMP error packet back to the source. This information allows the source host to fragment the packet into smaller pieces using this header and try again.

The authentication header provides a mechanism by which the receiver of a packet can be sure of who sent it. The encrypted security payload makes it possible to encrypt the contents of a packet so that only the intended recipient can read it. These headers use the cryptographic techniques that we will describe in Chap. 8 to accomplish their missions.

Controversies

Given the open design process and the strongly held opinions of many of the people involved, it should come as no surprise that many choices made for IPv6 were highly controversial, to say the least. We will summarize a few of these briefly below. For all the gory details, see the RFCs.

We have already mentioned the argument about the address length. The result was a compromise: 16-byte fixed-length addresses.

Another fight developed over the length of the *Hop limit* field. One camp felt strongly that limiting the maximum number of hops to 255 (implicit in using an 8-bit field) was a gross mistake. After all, paths of 32 hops are common now, and 10 years from now much longer paths may be common. These people argued that using a huge address size was farsighted but using a tiny hop count was shortsighted. In their view, the greatest sin a computer scientist can commit is to provide too few bits somewhere.

The response was that arguments could be made to increase every field, leading to a bloated header. Also, the function of the *Hop limit* field is to keep packets from wandering around for too long a time and 65,535 hops is far, far too long. Finally, as the Internet grows, more and more long-distance links will be built, making it possible to get from any country to any other country in half a dozen hops at most. If it takes more than 125 hops to get from the source and the destination to their respective international gateways, something is wrong with the national backbones. The 8-bitters won this one.

Another hot potato was the maximum packet size. The supercomputer community wanted packets in excess of 64 KB. When a supercomputer gets started transferring, it really means business and does not want to be interrupted every 64 KB. The argument against large packets is that if a 1-MB packet hits a 1.5-Mbps T1 line, that packet will tie the line up for over 5 seconds, producing a very noticeable delay for interactive users sharing the line. A compromise was reached here: normal packets are limited to 64 KB, but the hop-by-hop extension header can be used to permit jumbograms.

A third hot topic was removing the IPv4 checksum. Some people likened this move to removing the brakes from a car. Doing so makes the car lighter so it can go faster, but if an unexpected event happens, you have a problem.

The argument against checksums was that any application that really cares about data integrity has to have a transport layer checksum anyway, so having another one in IP (in addition to the data link layer checksum) is overkill. Furthermore, experience showed that computing the IP checksum was a major expense in IPv4. The antichecksum camp won this one, and IPv6 does not have a checksum.

Mobile hosts were also a point of contention. If a portable computer flies halfway around the world, can it continue operating there with the same IPv6 address, or does it have to use a scheme with home agents? Some people wanted to build explicit support for mobile hosts into IPv6. That effort failed when no consensus could be found for any specific proposal.

Probably the biggest battle was about security. Everyone agreed it was essential. The war was about where to put it. The argument for putting it in the network layer is that it then becomes a standard service that all applications can use without any advance planning. The argument against it is that really secure applications generally want nothing less than end-to-end encryption, where the source application does the encryption and the destination application undoes it. With anything

less, the user is at the mercy of potentially buggy network layer implementations over which he has no control. The response to this argument is that these applications can just refrain from using the IP security features and do the job themselves. The rejoinder to that is that the people who do not trust the network to do it right do not want to pay the price of slow, bulky IP implementations that have this capability, even if it is disabled.

Another aspect of where to put security relates to the fact that many (but not by no means all) countries have very stringent export laws concerning cryptography and encrypted data. especially personal data. Some, notably France and Iraq, also restrict its use domestically, so that people cannot have secrets from the government. As a result, any IP implementation that used a cryptographic system strong enough to be of much value could not be exported from the United States (and many other countries) to customers worldwide. Having to maintain two sets of software, one for domestic use and one for export, is something most computer vendors vigorously oppose.

One point on which there was no controversy is that no one expects the IPv4 Internet to be turned off on a Sunday evening and come back up as an IPv6 Internet Monday morning. Instead, isolated ''islands'' of IPv6 will be converted, initially communicating via tunnels, as we showed in Sec. 5.5.4. As the IPv6 islands grow, they will merge into bigger islands. Eventually, all the islands will merge, and the Internet will be fully converted.

At least, that was the plan. Deployment has proved the Achilles heel of IPv6. It's use is still far from universal, though all major operating systems fully support it and have supported it for over a decade. Most deployments are new situations in which a network operator—for example, a mobile phone operator— needs a large number of IP addresses. Nevertheless, it is slowly taking over. On Comcast, most traffic is now IPv6 and a quarter of Google's is also IPv6, so there is progress.

Many strategies have been defined to help ease the transition. Among them are ways to automatically configure the tunnels that carry IPv6 over the IPv4 Internet, and ways for hosts to automatically find the tunnel endpoints. Dual-stack hosts have an IPv4 and an IPv6 implementation so that they can select which protocol to use depending on the destination of the packet. These strategies will streamline the substantial deployment that seems inevitable when IPv4 addresses are exhausted. For more information about IPv6, see Davies (2008).

5.7.4 Internet Control Protocols

In addition to IP, which is used for data transfer, the Internet has several companion control protocols that are used in the network layer. They include ICMP, ARP, and DHCP. In this section, we will look at each of these in turn, describing the versions that correspond to IPv4 because they are the protocols that are in common use. ICMP and DHCP have similar versions for IPv6; the equivalent of ARP is called NDP (Neighbor Discovery Protocol) for IPv6.

ICMP—The Internet Control Message Protocol

The operation of the Internet is monitored closely by the routers. When something unexpected occurs during packet processing at a router, the event is reported to the sender by the **ICMP** (**Internet Control Message Protocol**). ICMP is also used to test the Internet. About a dozen types of ICMP messages are defined. Each ICMP message type is carried encapsulated in an IP packet. The most important ones are listed in Fig. 5-61.

Message type	Description
Destination unreachable	Packet could not be delivered
Time exceeded	Time to live field hit 0
Parameter problem	Invalid header field
Source quench	Choke packet
Redirect	Teach a router about geography
Echo and echo reply	Check if a machine is alive
Timestamp request/reply	Same as Echo, but with timestamp
Router advertisement/solicitation	Find a nearby router

Figure 5-61. The principal ICMP message types.

The DESTINATION UNREACHABLE message is used when the router cannot locate the destination or when a packet with the *DF* bit cannot be delivered because a ''small-packet'' network stands in the way.

The TIME EXCEEDED message is sent when a packet is dropped because its *TtL (Time to live)* counter has reached zero. This event is a symptom that packets are looping, or that the counter values are being set too low.

One clever use of this error message is the **traceroute** utility that was developed by Van Jacobson in 1987. Traceroute finds the routers along the path from the host to a destination IP address. It finds this information without any kind of privileged network support. The method is simply to send a sequence of packets to the destination, first with a TtL of 1, then a TtL of 2, 3, and so on. The counters on these packets will reach zero at successive routers along the path. These routers will each obediently send a TIME EXCEEDED message back to the host. From those messages, the host can determine the IP addresses of the routers along the path, as well as keep statistics and timings on parts of the path. It is not what the TIME EXCEEDED message was intended for, but it is perhaps the most useful network debugging tool of all time.

The PARAMETER PROBLEM message indicates that an illegal value has been detected in a header field. This problem indicates a bug in the sending host's IP software or possibly in the software of a router transited.

The SOURCE QUENCH message was long ago used to throttle hosts that were sending too many packets. When a host received this message, it was expected to

slow down. It is rarely used anymore because when congestion occurs, these packets tend to add more fuel to the fire and it is unclear how to respond to them. Congestion control in the Internet is now done largely by taking action in the transport layer, using packet losses as a congestion signal; we will study how this is done in detail in Chap. 6.

The REDIRECT message is used when a router notices that a packet seems to be routed incorrectly. It is used by the router to tell the sending host to update to a better route.

The ECHO and ECHO REPLY messages are sent by hosts to see if a given destination is reachable and currently alive. Upon receiving the ECHO message, the destination is expected to send back an ECHO REPLY message. These messages are used in the **ping** utility that checks if a host is up and on the Internet.

The TIMESTAMP REQUEST and TIMESTAMP REPLY messages are similar, except that the arrival time of the message and the departure time of the reply are recorded in the reply. This facility can be used to measure network performance.

The ROUTER ADVERTISEMENT and ROUTER SOLICITATION messages are used to let hosts find nearby routers. A host needs to learn the IP address of at least one router to be able to send packets off the local network.

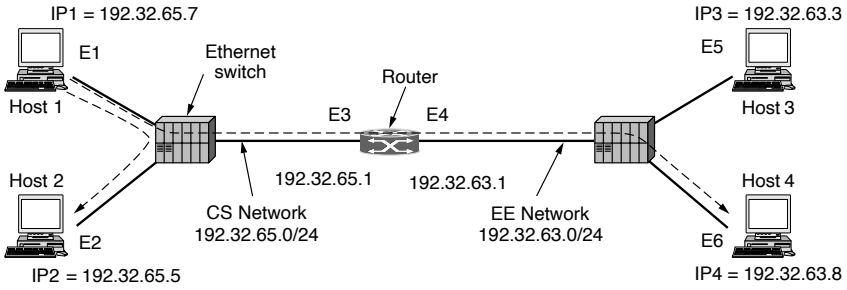
In addition to these messages, others have been defined. The online list is now kept at *www.iana.org/assignments/icmp-parameters*.

ARP—The Address Resolution Protocol

Although every machine on the Internet has one or more IP addresses, these addresses are not sufficient for sending packets. Data link layer NICs (Network Interface Cards) such as Ethernet cards do not understand Internet addresses. In the case of Ethernet, every NIC ever manufactured comes equipped with a unique 48-bit Ethernet address. Manufacturers of Ethernet NICs request a block of Ethernet addresses from IEEE to ensure that no two NICs have the same address (to avoid conflicts should the two NICs ever appear on the same LAN). The NICs send and receive frames based on 48-bit Ethernet addresses. They know nothing at all about 32-bit IP addresses.

The question now arises, how do IP addresses get mapped onto data link layer addresses, such as Ethernet? To explain how this works, let us use the example of Fig. 5-62, in which a small university with two /24 networks is illustrated. One network (CS) is a switched Ethernet in the Computer Science Dept. It has the prefix 192.32.65.0/24. The other LAN (EE), also switched Ethernet, is in Electrical Engineering and has the prefix 192.32.63.0/24. The two LANs are connected by an IP router. Each machine on an Ethernet and each interface on the router has a unique Ethernet address, labeled *E1* through *E6*, and a unique IP address on the CS or EE network.

Let us start out by seeing how a user on host 1 sends a packet to a user on host 2 on the CS network. Let us assume the sender knows the name of the intended



Frame	Source IP	Source Eth.	Destination IP	Destination Eth.
Host 1 to 2, on CS net	IP1	E1	IP2	E2
Host 1 to 4, on CS net	IP1	E1	IP4	E3
Host 1 to 4, on EE net	IP1	E4	IP4	E6

Figure 5-62. Two switched Ethernet LANs joined by a router.

receiver, possibly something like *eagle.cs.uni.edu*. The first step is to find the IP address for host 2. This lookup is performed by DNS, which we will study in Chap. 7. For the moment, we will just assume that DNS returns the IP address for host 2 (192.32.65.5).

The upper layer software on host 1 now builds a packet with 192.32.65.5 in the *Destination address* field and gives it to the IP software to transmit. The IP software can look at the address and see that the destination is on the CS network, (i.e., its own network). However, it still needs some way to find the destination's Ethernet address to send the frame. One solution is to have a configuration file somewhere in the system that maps IP addresses onto Ethernet addresses. While this solution is certainly possible, for organizations with thousands of machines keeping all these files up to date is an error-prone, time-consuming job.

A better solution is for host 1 to output a broadcast packet onto the Ethernet asking who owns IP address 192.32.65.5. The broadcast will arrive at every machine on the CS Ethernet, and each one will check its IP address. Host 2 alone will respond with its Ethernet address (*E2*). In this way host 1 learns that IP address 192.32.65.5 is on the host with Ethernet address *E2*. The protocol used for asking this question and getting the reply is called **ARP** (**Address Resolution Protocol**). Almost every machine on the Internet runs it. ARP is defined in RFC 826.

The advantage of using ARP over configuration files is the simplicity. The system manager does not have to do much except assign each machine an IP address and decide about subnet masks. ARP does the rest.

At this point, the IP software on host 1 builds an Ethernet frame addressed to *E2*, puts the IP packet (addressed to 192.32.65.5) in the payload field, and dumps it

onto the Ethernet. The IP and Ethernet addresses of this packet are given in Fig. 5-62. The Ethernet NIC of host 2 detects this frame, recognizes it as a frame for itself, scoops it up, and causes an interrupt. The Ethernet driver extracts the IP packet from the payload and passes it to the IP software, which sees that it is correctly addressed and processes it.

Various optimizations are possible to make ARP work more efficiently. To start with, once a machine has run ARP, it caches the result in case it needs to contact the same machine shortly. Next time it will find the mapping in its own cache, thus eliminating the need for a second broadcast. In many cases, host 2 will need to send back a reply, forcing it, too, to run ARP to determine the sender's Ethernet address. This ARP broadcast can be avoided by having host 1 include its IP-to-Ethernet mapping in the ARP packet. When the ARP broadcast arrives at host 2, the pair (192.32.65.7, *E1*) is entered into host 2's ARP cache. In fact, all machines on the Ethernet can enter this mapping into their ARP caches.

To allow mappings to change, for example, when a host is configured to use a new IP address (but keeps its old Ethernet address), entries in the ARP cache should time out after a few minutes. A clever way to help keep the cached information current and to optimize performance is to have every machine broadcast its mapping when it is configured. This broadcast is generally done in the form of an ARP looking for its own IP address. There should not be a response, but a side effect of the broadcast is to make or update an entry in everyone's ARP cache. This is known as a **gratuitous ARP**. If a response does (unexpectedly) arrive, two machines have been assigned the same IP address. The error must be resolved by the network manager before both machines can use the network.

Now let us look at Fig. 5-62 again, only this time assume that host 1 wants to send a packet to host 4 (192.32.63.8) on the EE network. Host 1 will see that the destination IP address is not on the CS network. It knows to send all such off-network traffic to the router, which is also known as the **default gateway**. By convention, the default gateway is the lowest address on the network (198.32.65.1). To send a frame to the router, host 1 must still know the Ethernet address of the router interface on the CS network. It discovers this by sending an ARP broadcast for 198.32.65.1, from which it learns *E3*. It then sends the frame. The same lookup mechanisms are used to send a packet from one router to the next over a sequence of routers in an Internet path.

When the Ethernet NIC of the router gets this frame, it gives the packet to the IP software. It knows from the network masks that the packet should be sent onto the EE network where it will reach host 4. If the router does not know the Ethernet address for host 4, then it will use ARP again to find out. The table in Fig. 5-62 lists the source and destination Ethernet and IP addresses that are present in the frames as observed on the CS and EE networks. Please observe that the Ethernet addresses change with the frame on each network while the IP addresses remain constant (because they indicate the endpoints across all of the interconnected networks).

It is also possible to send a packet from host 1 to host 4 without host 1 knowing that host 4 is on a different network. The solution is to have the router answer ARPs on the CS network for host 4 and give its Ethernet address, *E3*, as the response. It is not possible to have host 4 reply directly because it will not see the ARP request (as routers do not forward Ethernet-level broadcasts). The router will then receive frames sent to 192.32.63.8 and forward them onto the EE network. This solution is called **proxy ARP**. It is used in special cases in which a host wants to appear on a network even though it actually resides on another network. A common situation, for example, is a mobile computer that wants some other node to pick up packets for it when it is not on its home network.

DHCP—The Dynamic Host Configuration Protocol

ARP (as well as other Internet protocols) makes the assumption that hosts are configured with some basic information, such as their own IP addresses. How do hosts get this information? It is possible to manually configure each computer, but that is tedious and error-prone. There is a better way, and it is called **DHCP** (**Dynamic Host Configuration Protocol**).

With DHCP, every network must have a DHCP server that is responsible for configuration. When a computer is started, it has a built-in Ethernet or other link layer address embedded in the NIC, but no IP address. Much like ARP, the computer broadcasts a request for an IP address on its network. It does this by using a DHCP DISCOVER packet. This packet must reach the DHCP server. If that server is not directly attached to the network, the router will be configured to receive DHCP broadcasts and relay them to the DHCP server, wherever it is located.

When the server receives the request, it allocates a free IP address and sends it to the host in a DHCP OFFER packet (which again may be relayed via the router). To be able to do this work even when hosts do not have IP addresses, the server identifies a host using its Ethernet address (which is carried in the DHCP DISCOVER packet)

An issue that arises with automatic assignment of IP addresses from a pool is for how long an IP address should be allocated. If a host leaves the network and does not return its IP address to the DHCP server, that address will be permanently lost. After a period of time, many addresses may be lost. To prevent that from happening, IP address assignment may be for a fixed period of time, a technique called **leasing**. Just before the lease expires, the host must ask for a DHCP renewal. If it fails to make a request or the request is denied, the host may no longer use the IP address it was given earlier.

DHCP is described in RFC 2131 and RFC 2132. It is widely used in the Internet to configure all sorts of parameters in addition to providing hosts with IP addresses. As well as in business and home networks, DHCP is used by ISPs to set the parameters of devices over the Internet access link, so that customers do not need to phone their ISPs to get this information. Common examples of the kind of

information that is configured include the network mask, the IP address of the default gateway, and the IP addresses of DNS and time servers. DHCP has largely replaced earlier protocols (called RARP and BOOTP) with more limited functionality.

5.7.5 Label Switching and MPLS

So far, on our tour of the network layer of the Internet, we have focused exclusively on packets as datagrams that are forwarded by IP routers. There is also another kind of technology that is starting to be widely used, especially by ISPs, in order to move Internet traffic across their networks. This technology is called **MPLS** (**MultiProtocol Label Switching**) and it is perilously close to circuit switching. Despite the fact that many people in the Internet community have an intense dislike for connection-oriented networking, the idea seems to keep coming back. As Yogi Berra once put it, it is like déjà vu all over again. However, there are essential differences between the way the Internet handles route construction and the way connection-oriented networks do it, so the technique is certainly not traditional circuit switching.

MPLS adds a label in front of each packet, and forwarding is based on the label rather than on the destination address. Making the label an index into an internal table makes finding the correct output line just a matter of table lookup. Using this technique, forwarding can be done very quickly. This advantage was the original motivation behind MPLS, which began as proprietary technology known by various names including **tag switching**. Eventually, IETF began to standardize the idea. It is described in RFC 3031 and many other RFCs. The main benefits over time have come to be routing that is flexible and forwarding that is suited to quality of service as well as fast.

The first question to ask is where does the label go? Since IP packets were not designed for virtual circuits, there is no field available for virtual-circuit numbers within the IP header. For this reason, a new MPLS header had to be added in front of the IP header. On a router-to-router line using PPP as the framing protocol, the frame format, including the PPP, MPLS, IP, and TCP headers, is as shown in Fig. 5-63.

The generic MPLS header is 4 bytes long and has four fields. Most important is the *Label* field, which holds the index. The *QoS* field indicates the class of service. The *S* field relates to stacking multiple labels (which is discussed below). The *TtL* field indicates how many more times the packet may be forwarded. It is decremented at each router, and if it hits 0, the packet is discarded. This feature prevents infinite looping in the case of routing instability.

MPLS falls between the IP network layer protocol and the PPP link layer protocol. It is not really a layer 3 protocol because it depends on IP or other network layer addresses to set up label paths. It is not really a layer 2 protocol either because it forwards packets across multiple hops, not a single link. For this reason,

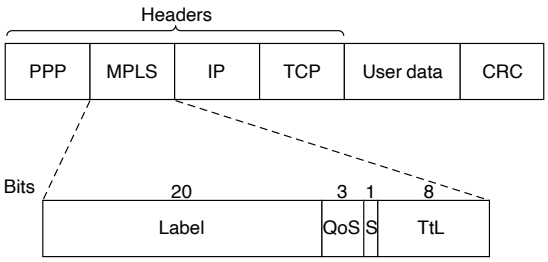


Figure 5-63. Transmitting a TCP segment using IP, MPLS, and PPP.

MPLS is sometimes described as a layer 2.5 protocol. It is an illustration that real protocols do not always fit neatly into our ideal layered protocol model.

On the brighter side, because the MPLS headers are not part of the network layer packet or the data link layer frame, MPLS is to a large extent independent of both layers. Among other things, this property means it is possible to build MPLS switches that can forward both IP packets and non-IP packets, depending on what shows up. This feature is where the ''multiprotocol'' in the name MPLS came from. MPLS can also carry IP packets over non-IP networks.

When an MPLS-enhanced packet arrives at a **LSR** (**Label Switched Router**), the label is used as an index into a table to determine the outgoing line to use and also the new label to use. This label swapping is used in all virtual-circuit networks. Labels have only local significance and two different routers can feed unrelated packets with the same label into another router for transmission on the same outgoing line. To be distinguishable at the other end, labels have to be remapped at every hop. We saw this mechanism in action in Fig. 5-3. MPLS uses the same technique.

As an aside, some people distinguish between *forwarding* and *switching*. Forwarding is the process of finding the best match for a destination address in a table to decide where to send packets. An example is the longest matching prefix algorithm used for IP forwarding. In contrast, switching uses a label taken from the packet as an index into a forwarding table. It is simpler and faster. These definitions are far from universal, however.

Since most hosts and routers do not understand MPLS, we should also ask when and how the labels are attached to packets. This happens when an IP packet reaches the edge of an MPLS network. The **LER** (**Label Edge Router**) inspects the destination IP address and other fields to see which MPLS path the packet should follow, and puts the right label on the front of the packet. Within the MPLS network, this label is used to forward the packet. At the other edge of the MPLS network, the label has served its purpose and is removed, revealing the IP packet again for the next network. This process is shown in Fig. 5-64. One difference from traditional virtual circuits is the level of aggregation. It is certainly possible

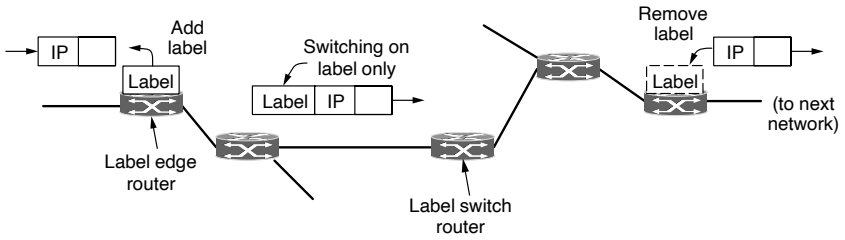


Figure 5-64. Forwarding an IP packet through an MPLS network.

for each flow to have its own set of labels through the MPLS network. However, it is more common for routers to group multiple flows that end at a particular router or LAN and use a single label for them. The flows that are grouped together under a single label are said to belong to the same **FEC** (**Forwarding Equivalence Class**). This class covers not only where the packets are going, but also their service class (in the differentiated services sense) because all the packets are treated the same way for forwarding purposes.

With traditional virtual-circuit routing, it is not possible to group several distinct paths with different endpoints onto the same virtual-circuit identifier because there would be no way to distinguish them at the final destination. With MPLS, the packets still contain their final destination address, in addition to the label. At the end of the labeled route, the label header can be removed and forwarding can continue the usual way, using the network layer destination address.

Actually, MPLS goes even further. It can operate at multiple levels at once by adding more than one label to the front of a packet. For example, suppose that there are many packets that already have different labels (because we want to treat the packets differently somewhere in the network) that should follow a common path to some destination. Instead of setting up many label switching paths, one for each of the different labels, we can set up a single path. When the already-labeled packets reach the start of this path, another label is added to the front. This is called a stack of labels. The outermost label guides the packets along the path. It is removed at the end of the path, and the labels revealed, if any, are used to forward the packet further. The *S* bit in Fig. 5-63 allows a router removing a label to know if there are any additional labels left. It is set to 1 for the bottom label and 0 for all the other labels.

The final question we will ask is how the label forwarding tables are set up so that packets follow them. This is one area of major difference between MPLS and conventional virtual-circuit designs. In traditional virtual-circuit networks, when a user wants to establish a connection, a setup packet is launched into the network to create the path and make the forwarding table entries. MPLS does not involve users in the setup phase. Requiring users to do anything other than send a datagram would break too much existing Internet software.

Instead, the forwarding information is set up by protocols that are a combination of routing protocols and connection setup protocols. These control protocols are separated from label forwarding, which allows multiple, different control protocols to be used. One of the variants works like this. When a router is booted, it checks to see which routes it is the final destination for (e.g., which prefixes belong to its interfaces). It then creates one or more FECs for them, allocates a label for each one, and passes the labels to its neighbors. They, in turn, enter the labels in their forwarding tables and send new labels to their neighbors, until all the routers have acquired the path. Resources can also be reserved as the path is constructed to guarantee an appropriate quality of service. Other variants can set up different paths, such as traffic engineering paths that take unused capacity into account, and create paths on-demand to support service offerings such as quality of service.

Although the basic ideas behind MPLS are straightforward, the details are complicated, with many variations and use cases that are being actively developed. For more information, see Davie and Farrel (2008) and Davie and Rekhter (2000).

5.7.6 OSPF—An Interior Gateway Routing Protocol

We have now finished our study of how packets are forwarded in the Internet. It is time to move on to the next topic: routing in the Internet. As we mentioned earlier, the Internet is made up of a large number of independent networks or **ASes** (**Autonomous Systems**) that are operated by different organizations, usually a company, university, or ISP. Inside of its own network, an organization can use its own algorithm for internal routing, or **intradomain routing**, as it is more commonly known. Nevertheless, there are only a handful of standard protocols that are popular. In this section, we will study the problem of intradomain routing and look at the OSPF protocol that is widely used in practice. An intradomain routing protocol is also called an **IGP** (**Interior Gateway Protocol**). In the next section, we will study the problem of routing between independently operated networks, or **interdomain routing**. For that case, all networks must use the same interdomain routing protocol or **exterior gateway protocol**. The protocol that is used in the Internet is BGP (Border Gateway Protocol). It will be discussed in Sec. 5.7.7.

Early intradomain routing protocols used a distance vector design, based on the distributed Bellman-Ford algorithm inherited from the ARPANET. **RIP** (**Routing Information Protocol**) is the main example that is used to this day. It works well in small systems, but less well as networks get larger. It also suffers from the count-to-infinity problem and generally slow convergence. The ARPANET switched over to a link state protocol in May 1979 because of these problems, and in 1988 IETF began work on a link state protocol for intradomain routing. That protocol, called **OSPF** (**Open Shortest Path First**), became a standard in 1990. It drew on a protocol called **IS-IS** (**Intermediate-System to Intermediate-System**), which became an ISO standard. Because of their shared heritage, the two protocols are much more alike than different. For the complete story, see RFC 2328.

They are the dominant intradomain routing protocols, and most router vendors now support both of them. OSPF is more widely used in company networks, and IS-IS is more widely used in ISP networks. Of the two, we will give a sketch of how OSPF works.

Given the long experience with other routing protocols, the group designing OSPF had a long list of requirements that had to be met. First, the algorithm had to be published in the open literature, hence the ''O'' in OSPF. A proprietary solution owned by one company would not do. Second, the new protocol had to support a variety of distance metrics, including physical distance, delay, and so on. Third, it had to be a dynamic algorithm, one that adapted to changes in the topology automatically and quickly.

Fourth, and new for OSPF, it had to support routing based on type of service. The new protocol had to be able to route real-time traffic one way and other traffic a different way. At the time, IP had a *Type of service* field, but no existing routing protocol used it. This field was included in OSPF but still nobody used it, and it was eventually removed. Perhaps this requirement was ahead of its time, as it preceded IETF's work on differentiated services, which has rejuvenated classes of service.

Fifth, and related to the above, OSPF had to do load balancing, splitting the load over multiple lines. Most previous protocols sent all packets over a single best route, even if there were two routes that were equally good. The other route was not used at all. In many cases, splitting the load over multiple routes gives better performance.

Sixth, support for hierarchical systems was needed. By 1988, some networks had grown so large that no router could be expected to know the entire topology. OSPF had to be designed so that no router would have to.

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

OSPF supports both point-to-point links (e.g., SONET) and broadcast networks (e.g., most LANs). Actually, it is able to support networks with multiple routers, each of which can communicate directly with the others (called **multiaccess networks**) even if they do not have broadcast capability. Earlier protocols did not handle this case well.

An example of an autonomous system network is given in Fig. 5-65(a). Hosts are omitted because they do not generally play a role in OSPF, while routers and networks (which may contain hosts) do. Most of the routers in Fig. 5-65(a) are connected to other routers by point-to-point links, and to networks to reach the hosts on those networks. However, routers *R3*, *R4*, and *R5* are connected by a broadcast LAN such as switched Ethernet.

OSPF operates by abstracting the collection of actual networks, routers, and links into a directed graph in which each arc is assigned a weight (distance, delay,

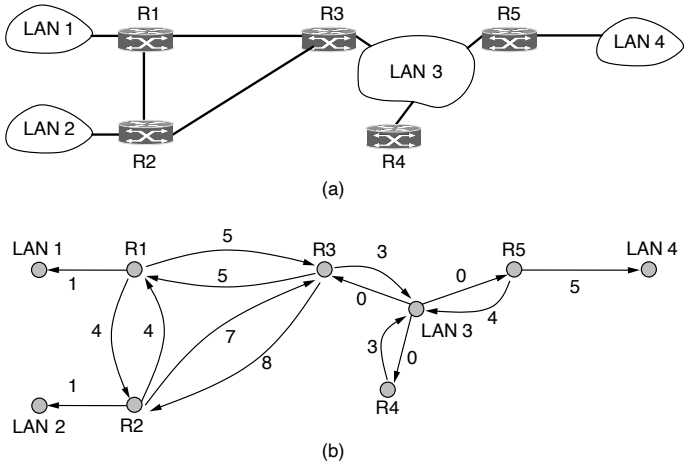


Figure 5-65. (a) An autonomous system. (b) A graph representation of (a).

etc.). A point-to-point connection between two routers is represented by a pair of arcs, one in each direction. Their weights may be different. A broadcast network is represented by a node for the network itself, plus a node for each router. The arcs from that network node to the routers have weight 0. They are important nonetheless, as without them there is no path through the network. Other networks, which have only hosts, have only an arc reaching them and not one returning. This structure gives routes to hosts, but not through them.

Figure 5-65(b) shows the graph representation of the network of Fig. 5-65(a). What OSPF fundamentally does is represent the actual network as a graph like this and then use the link state method to have every router compute the shortest path from itself to all other nodes. Multiple paths may be found that are equally short. In this case, OSPF remembers the set of shortest paths and during packet forwarding, traffic is split across them. This helps to balance load. It is called **ECMP** (**Equal Cost MultiPath**).

Many of the ASes in the Internet are themselves large and nontrivial to manage. To work at this scale, OSPF allows an AS to be divided into numbered **areas**, where an area is a network or a set of contiguous networks. Areas do not overlap but need not be exhaustive, that is, some routers may belong to no area. Routers that lie wholly within an area are called **internal routers**. An area is a generalization of an individual network. Outside an area, its destinations are visible but not its topology. This characteristic helps routing to scale.

Every AS has a **backbone area**, called area 0. The routers in this area are called **backbone routers**. All areas are connected to the backbone, possibly by tunnels, so it is possible to go from any area in the AS to any other area in the AS via

the backbone. A tunnel is represented in the graph as just another arc with a cost. As with other areas, the topology of the backbone is not visible outside the backbone.

Each router that is connected to two or more areas is called an **area border router**. It must also be part of the backbone. The job of an area border router is to summarize the destinations in one area and to inject this summary into the other areas to which it is connected. This summary includes cost information but not all the details of the topology within an area. Passing cost information allows hosts in other areas to find the best area border router to use to enter an area. Not passing topology information reduces traffic and simplifies the shortest-path computations of routers in other areas. However, if there is only one border router out of an area, even the summary does not need to be passed. Routes to destinations out of the area always start with the instruction ''Go to the border router.'' This kind of area is called a **stub area**.

The last kind of router is the **AS boundary router**. It injects routes to external destinations on other ASes into the area. The external routes then appear as destinations that can be reached via the AS boundary router with some cost. An external route can be injected at one or more AS boundary routers. The relationship between ASes, areas, and the various kinds of routers is shown in Fig. 5-66. One router may play multiple roles, for example, a border router is also a backbone router.

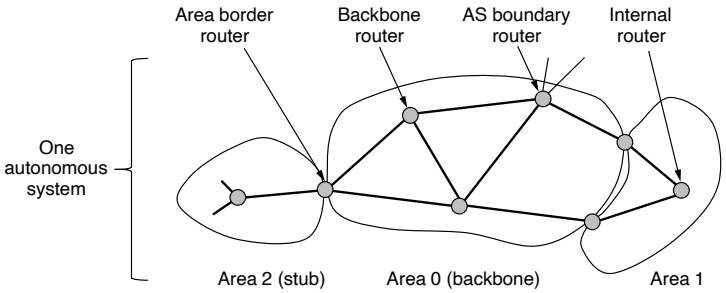


Figure 5-66. The relation between ASes, backbones, and areas in OSPF.

During normal operation, each router within an area has the same link state database and runs the same shortest path algorithm. Its main job is to calculate the shortest path from itself to every other router and network in the entire AS. An area border router needs the databases for all the areas to which it is connected and must run the shortest path algorithm for each area separately.

For a source and destination in the same area, the best intra-area route (that lies wholly within the area) is chosen. For a source and destination in different areas, the inter-area route must go from the source to the backbone, across the backbone to the destination area, and then to the destination. This algorithm forces a star

configuration on OSPF, with the backbone being the hub and the other areas being spokes. Because the route with the lowest cost is chosen, routers in different parts of the network may use different area border routers to enter the backbone and destination area. Packets are routed from source to destination ''as is.'' They are not encapsulated or tunneled (unless going to an area whose only connection to the backbone is a tunnel). Also, routes to external destinations may include the external cost from the AS boundary router over the external path, if desired, or just the cost internal to the AS.

When a router boots, it sends HELLO messages on all of its point-to-point lines and multicasts them on LANs to the group consisting of all the other routers. From the responses, each router learns who its neighbors are. Routers on the same LAN are all neighbors.

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

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

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

Either partner can request link state information from the other one by using LINK STATE REQUEST messages. The result of this algorithm is that each pair of adjacent routers checks to see who has the most recent data, and new information is spread throughout the area this way. All these messages are sent directly in IP packets. The five kinds of messages are summarized in Fig. 5-67.

Finally, we can put all the pieces together. Using flooding, each router informs all the other routers in its area of its links to other routers and networks and the cost of these links. This information allows each router to construct the graph for its area(s) and compute the shortest paths. The backbone area does this work, too. In addition, the backbone routers accept information from the area border routers in order to compute the best route from each backbone router to every other router.

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

Figure 5-67. The five types of OSPF messages.

This information is propagated back to the area border routers, which advertise it within their areas. Using this information, internal routers can select the best route to a destination outside their area, including the best exit router to the backbone.

5.7.7 BGP—The Exterior Gateway Routing Protocol

Within a single AS, OSPF and IS-IS are the protocols that are commonly used. Between ASes, a different protocol, called **BGP** (**Border Gateway Protocol**), is used. A different protocol is needed because the goals of an intradomain protocol and an interdomain protocol are not the same. All an intradomain protocol has to do is move packets as efficiently as possible from the source to the destination. It does not have to worry about politics.

In contrast, interdomain routing protocols have to worry about politics a great deal (Metz, 2001). For example, a corporate AS might want the ability to send packets to any Internet site and receive packets from any Internet site. However, it might be unwilling to carry transit packets originating in a foreign AS and ending in a different foreign AS, even if its own AS is on the shortest path between the two foreign ASes (''That's their problem, not ours''). On the other hand, it might be willing to carry transit traffic for its neighbors, or even for specific other ASes that paid it for this service. Telephone companies, for example, might be happy to act as carriers for their customers, but not for others. Exterior gateway protocols in general, and BGP in particular, have been designed to allow many kinds of routing policies to be enforced in the interAS traffic.

Typical policies involve political, security, or economic considerations. A few examples of possible routing constraints are:

1. Do not carry commercial traffic on the educational network.
2. Never send traffic from the Pentagon on a route through Iraq.
3. Use TeliaSonera instead of Verizon because it is cheaper.
4. Don't use AT&T in Australia because performance is poor.
5. Traffic starting or ending at Apple should not transit Google.

As you might imagine from this list, routing policies can be highly individual. They are often proprietary because they contain sensitive business information. However, we can describe some patterns that capture the reasoning of the companies above and that are often used as a starting point.

A routing policy is implemented by deciding what traffic can flow over which of the links between ASes. One common policy is that a customer ISP pays another provider ISP to deliver packets to any other destination on the Internet and receive packets sent from any other destination. The customer ISP is said to buy **transit service** from the provider ISP. This is very similar a customer at home buying Internet access service from an ISP. To make it work, the provider should advertise routes to all destinations on the Internet to the customer over the link that connects them. In this way, the customer will have a route to use to send packets anywhere. Conversely, the customer should advertise routes only to the destinations on its network to the provider. This will let the provider send traffic to the customer only for those addresses; the customer does not want to handle traffic intended for other destinations.

We can see an example of transit service in Fig. 5-68. There are four ASes that are connected. The connection is often made with a link at **IXPs** (**Internet eXchange Points**), facilities to which many ISPs have a link for the purpose of connecting with other ISPs. *AS2*, *AS3*, and *AS4* are customers of *AS1*. They buy transit service from it. Thus, when source *A* sends to destination *C*, the packets travel from *AS2* to *AS1* and finally to *AS4*. The routing advertisements travel in the opposite direction to the packets. *AS4* advertises *C* as a destination to its transit provider, *AS1*, to let sources reach *C* via *AS1*. Later, *AS1* advertises a route to *C* to its other customers, including *AS2*, to let the customers know that they can send traffic to *C* via *AS1*.

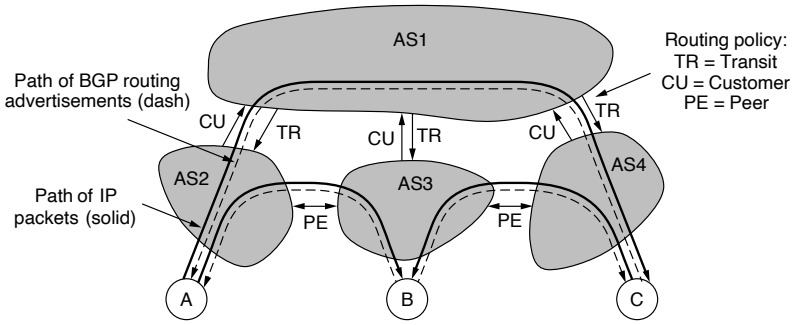


Figure 5-68. Routing policies between four autonomous systems.

In Fig. 5-68, all of the other ASes buy transit service from *AS1*. This provides them with connectivity so they can interact with any host on the Internet. However,

they have to pay for this privilege. Suppose that *AS2* and *AS3* exchange a lot of traffic. Given that their networks are connected already, if they want to, they can use a different policy—they can send traffic directly to each other for free. This will reduce the amount of traffic they must have *AS1* deliver on their behalf, and hopefully it will reduce their bills. This policy is called **settlement-free peering** or **settlement-free interconnection**.

To implement settlement-free peering, two ASes send routing advertisements to each other for the addresses that reside in their networks. Doing so makes it possible for *AS2* to send *AS3* packets from *A* destined to *B* and vice versa. However, note that settlement-free peering is not transitive. In Fig. 5-68, *AS3* and *AS4* also peer with each other. This arrangement allows traffic from *C* destined for *B* to be sent directly to *AS4*. What happens if *C* sends a packet to *A*? *AS3* is only advertising a route to *B* to *AS4*. It is not advertising a route to *A*. The consequence is that traffic will not pass from *AS4* to *AS3* to *AS2*, even though a physical path exists. This restriction is exactly what *AS3* wants. It peers with *AS4* to exchange traffic, but does not want to carry traffic from *AS4* to other parts of the Internet since it is not being paid to do so. Instead, *AS4* gets transit service from *AS1*. Thus, it is *AS1* that will carry the packet from *C* to *A*.

Now that we know about transit and settlement-free peering, we can also see that *A*, *B*, and *C* have transit arrangements. For example, *A* must buy Internet access from *AS2*. *A* might be a single home computer or a company network with many LANs. However, it does not need to run BGP because it is a **stub network** that is connected to the rest of the Internet by only one link. So the only place for it to send packets destined outside of the network is over the link to *AS2*. There is nowhere else to go. This path can be arranged simply by setting up a default route. For this reason, we have not shown *A*, *B*, and *C* as ASes that participate in interdomain routing.

Transit and settlement-free peering business arrangements are implemented through a combination of routing policies that implement (1) preference among multiple routes to a destination, (2) filtering of how routes are advertised to neighboring networks. Generally speaking, preference works as follows: a router will prefer routes learned from paying customers first, followed by routes learned from settlement-free peers, and finally routes learned from provider networks. The rationale is simple: an AS would prefer to send traffic along routes where it is paid, as opposed to sending traffic on routes where it has to pay for use. For similar reasons, an AS will advertise all of its routes to customers, but it will not re-advertise routes learned from a settlement-free peer or transit provider to other peers or providers. In addition to these two business arrangements, ASes have other arrangements, including **paid peering**, whereby one AS pays another for access to routes learned from that ASes customers. Paid peering is similar to settlement-free peering, except that money changes hands. Finally, there can also be **partial transit** arrangements, whereby an AS might pay another AS for routes to some subset of all Internet destinations.

Some company networks are connected to multiple ISPs. This technique is used to improve reliability, since if the path through one ISP fails, the company can use the path via the other ISP. This technique is called **multihoming**. In this case, the company network is likely to run an interdomain routing protocol (e.g., BGP) to tell other ASes which addresses should be reached via which ISP links.

Many variations on these transit and peering policies are possible, but they already illustrate how business relationships and control over where route advertisements go can implement different kinds of policies. Now we will consider in more detail how routers running BGP advertise routes to each other and select paths over which to forward packets.

BGP is a form of distance vector protocol, but it is quite unlike intradomain distance vector protocols such as RIP. We have already seen that policy, instead of minimum distance, is used to pick which routes to use. Another large difference is that instead of maintaining just the cost of the route to each destination, each BGP router keeps track of the path used. This approach is called a **path vector protocol**. The path consists of the next hop router (which may be on the other side of the ISP, not adjacent) and the sequence of ASes, or **AS path**, that the route has followed (given in reverse order). Finally, pairs of BGP routers communicate with each other by establishing TCP connections. Operating this way provides reliable communication and also hides all the details of the network being passed through.

An example of how BGP routes are advertised is shown in Fig. 5-69. There are three ASes and the middle one is providing transit to the left and right ISPs. A route advertisement to prefix *C* starts in *AS3*. When it is propagated across the link to *R2c* at the top of the figure, it has the AS path of simply *AS3* and the next hop router of *R3a*. At the bottom, it has the same AS path but a different next hop because it came across a different link. This advertisement continues to propagate and crosses the boundary into *AS1*. At router *R1a*, at the top of the figure, the AS path is *AS2*, *AS3* and the next hop is *R2a*.

Carrying the complete path with the route makes it easy for the receiving router to detect and break routing loops. The rule is that each router that sends a route outside of the AS prepends its own AS number to the route. (This is why the list is in reverse order.) When a router receives a route, it checks to see if its own AS number is already in the AS path. If it is, a loop has been detected and the advertisement is discarded. However, and somewhat ironically, it was realized in the late 1990s that despite this precaution BGP suffers from a version of the count-to-infinity problem (Labovitz et al., 2001). There are no long-lived loops, but routes can sometimes be slow to converge and have transient loops.

Giving a list of ASes is a very coarse way to specify a path. An AS might be a small company, or an international backbone network. There is no way of telling from the route. BGP does not even try because different ASes may use different intradomain protocols whose costs cannot be compared. Even if they could be compared, an AS may not want to reveal its internal metrics. This is one of the ways that interdomain routing protocols differ from intradomain protocols.

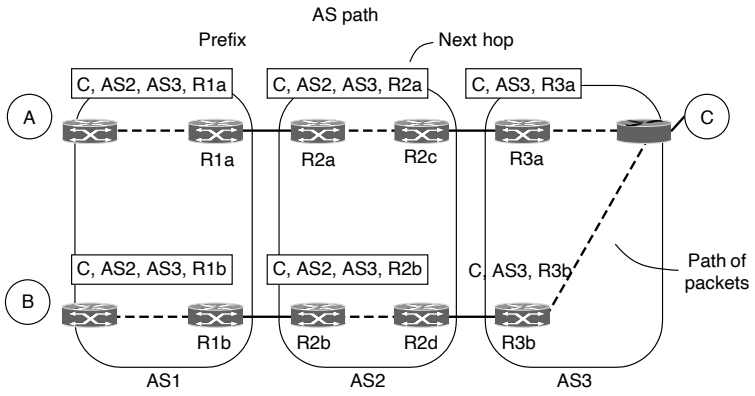


Figure 5-69. Propagation of BGP route advertisements.

So far we have seen how a route advertisement is sent across the link between two ISPs. We still need some way to propagate BGP routes from one side of the ISP to the other, so they can be sent on to the next ISP. This task could be handled by the intradomain protocol, but because BGP is very good at scaling to large networks, a variant of BGP is often used. It is called **iBGP** (**internal BGP**) to distinguish it from the regular use of BGP as **eBGP** (**external BGP**).

The rule for propagating routes inside an ISP is that every router at the boundary of the ISP learns of all the routes seen by all the other boundary routers, for consistency. If one boundary router on the ISP learns of a prefix to IP 128.208.0.0/16, all the other routers will learn of this prefix. The prefix will then be reachable from all parts of the ISP, no matter how packets enter the ISP from other ASes.

We have not shown this propagation in Fig. 5-69 to avoid clutter, but, for example, router *R2b* will know that it can reach *C* via either router *R2c* at top or router *R2d* at bottom. The next hop is updated as the route crosses within the ISP so that routers on the far side of the ISP know which router to use to exit the ISP on the other side. This can be seen in the leftmost routes in which the next hop points to a router in the same ISP and not a router in the next ISP.

We can now describe the key missing piece, which is how BGP routers choose which route to use for each destination. Each BGP router may learn a route for a given destination from the router it is connected to in the next ISP and from all of the other boundary routers (which have heard different routes from the routers they are connected to in other ISPs). Each router must decide which route in this set of routes is the best one to use. Ultimately the answer is that it is up to the ISP to write some policy to pick the preferred route. However, this explanation is very general and not so satisfying, so we can at least describe some common strategies.

The first strategy is that routes via peered networks are chosen in preference to routes via transit providers. The former are free; the latter cost money. A similar strategy is that customer routes are given the highest preference. It is only good business to send traffic directly to the paying customers.

A different kind of strategy is the default rule that shorter AS paths are better. This is debatable since an AS could be a network of any size, so a path through three small ASes could actually be shorter than a path through one big AS. However, shorter tends to be better on average, and this rule is a common tiebreaker.

The final strategy is to prefer the route that has the lowest cost within the ISP. This is the strategy implemented in Fig. 5-69. Packets sent from *A* to *C* exit *AS1* at the top router, *R1a*. Packets sent from *B* exit via the bottom router, *R1b*. The reason is that both *A* and *B* are taking the lowest-cost path or quickest route out of *AS1*. Because they are located in different parts of the ISP, the quickest exit for each one is different. The same thing happens as the packets pass through *AS2*. On the last leg, *AS3* has to carry the packet from *B* through its own network.

This strategy is known as **early exit** or **hot-potato routing**. It has the curious side effect of tending to make routes asymmetric. For example, consider the path taken when *C* sends a packet back to *B*. The packet will exit *AS3* quickly, at the top router, to avoid wasting its resources. Similarly, it will stay at the top when *AS2* passes it to *AS1* as quickly as possible. Then the packet will have a longer journey in *AS1*. This is a mirror image of the path taken from *B* to *C*.

The above discussion should make clear that each BGP router chooses its own best route from the known possibilities. It is not the case, as might naively be expected, that BGP chooses a path to follow at the AS level and OSPF chooses paths within each of the ASes. BGP and the interior gateway protocol are integrated much more deeply. This means that, for example, BGP can find the best exit point from one ISP to the next and this point will vary across the ISP, as in the case of the hot-potato policy. It also means that BGP routers in different parts of one AS may choose different AS paths to reach the same destination. Care must be exercised by the ISP to configure all of the BGP routers to make compatible choices given all of this freedom, but this can be done in practice.

The above policies are implemented with a variety of protocol configurations and settings. The main aspect of the mechanics that is worth understanding is the route selection process, which allows a router to select a route to an Internet destination, given multiple options. Route selection proceeds in the following steps:

1. Prefer the route with the highest local preference value.
2. Prefer the route with the shortest AS path length.
3. Prefer routes learned via external connections (i.e., via eBGP) over those learned from internal connections (i.e., via iBGP).
4. Among routes learned from the same neighboring AS, prefer routes with the lowest multiple exit discriminator (MED) value.

5. Prefer routes with the shortest IGP path cost to the next-hope IP address in the BGP route (where the next-hop IP address is typically that of the border router).

These route selection steps proceed in sequence until the router chooses a single route for each IP prefix. The router performs the above process for each IP prefix in its routing table. Although this ordering seems lengthy and complicated, it is fairly intuitive. The **local preference** value for each route is a value that the local network operator can set, and it remains internal to that AS. Because it has the highest precedence among route selection rules, it allows an operator to implement the types of route preferences and priorities that we discussed earlier in this section (e.g., preferring a route learned from a customer over a settlement-free route). After that rule, the others generally involve selection of short routes, as well as a way to implement early exit routing, as previously described. For example, the preference for a route learned from an external AS over an internal router is an attempt to implement early exit. Similarly, a preference for a route with a shortest IGP path cost to the border router is also an attempt to implement early exit.

Amazingly, we have only scratched the surface of BGP. For more information, see the BGP version 4 specification in RFC 427 and related RFCs. However, realize that much of its complexity lies with policies, which are not described in the specification of the BGP protocol.

Interdomain Traffic Engineering

As previously described in this chapter, network operators often need to tune the parameters and configuration of network protocols to manage utilization and congestion. Such traffic engineering practices are common with BGP, where an operator may want to control how BGP selects routes to control how traffic enters the network (**inbound traffic engineering**) or how it leaves the network (**outbound traffic engineering**)

The most common way to perform inbound traffic engineering is by adjusting how routers set the local preference attribute for individual routes. By setting a higher local preference value for all routes learned from a particular customer AS, for example, an operator can ensure that that customer's routes are picked over, say, a transit route whenever the customer route exists. Inbound traffic engineering is trickier, because BGP does not let one AS tell another AS how to select routes (hence the name, autonomous). Nevertheless, an operator can send indirect signals to routers in neighboring networks to control how these routers select routes. One common way to do this is to artificially inflate the length of the AS path by repeating the network's own AS multiple times in the route announcement, a practice called **AS path prepending**. Another approach is to leverage longest prefix match and simply split a prefix into multiple smaller (longer) prefixes, so that upstream routers prefer the routes with longer prefixes. For example, a route for a /20 prefix

could be split into routes for two /21 prefixes, four /22 prefixes, and so forth. This approach has some cost, however, as it can make the routing tables larger, and beyond a certain length, routers will filter the announcements.

5.7.8 Internet Multicasting

Normal IP communication is between one sender and one receiver. However, for some applications, it is useful for a process to be able to send to a large number of receivers simultaneously. Examples are streaming a live sports event to many viewers, delivering program updates to a pool of replicated servers, and handling digital conference (i.e., multiparty) telephone calls.

IP supports one-to-many communication, or multicasting, using class D IP addresses. Each class D address identifies a group of hosts. Twenty-eight bits are available for identifying groups, so over 250 million groups can exist at the same time. When a process sends a packet to a class D address, a best-effort attempt is made to deliver it to all the members of the group addressed, but no guarantees are given. Some members may not get the packet.

The range of IP addresses 224.0.0.0/24 is reserved for multicast on the local network. In this case, no routing protocol is needed. The packets are multicast by simply broadcasting them on the LAN with a multicast address. All hosts on the LAN receive the broadcasts, and hosts that are members of the group process the packet. Routers do not forward the packet off the LAN. Some examples of local multicast addresses are:

224.0.0.1 All systems on a LAN
224.0.0.2 All routers on a LAN
224.0.0.5 All OSPF routers on a LAN
224.0.0.251 All DNS servers on a LAN

Other multicast addresses may have members on different networks. In this case, a routing protocol is needed. But first, the multicast routers need to know which hosts are members of a group. A process asks its host to join in a specific group. It can also ask its host to leave the group. Each host keeps track of which groups its processes currently belong to. When the last process on a host leaves a group, the host is no longer a member of that group. About once a minute or so, each multicast router sends a query packet to all the hosts on its LAN (using the local multicast address of 224.0.0.1, of course) asking them to report back on the groups to which they currently belong. The multicast routers may or may not be colocated with the standard routers. Each host sends back responses for all the class D addresses it is interested in. These query and response packets use a protocol called **IGMP** (**Internet Group Management Protocol**). It is described in RFC 3376.

Any of several multicast routing protocols may be used to build multicast spanning trees that give paths from senders to all of the members of the group.

The algorithms that are used are the ones we described in Sec. 5.2.8. Within an AS, the main protocol used is **PIM** (**Protocol Independent Multicast**). PIM comes in several flavors. In Dense Mode PIM, a pruned reverse path forwarding tree is created. This is suited to situations in which members are everywhere in the network, such as distributing files to many servers within a data center network. In Sparse Mode PIM, spanning trees that are built are similar to core-based trees. This is suited to situations such as a content provider multicasting TV to subscribers on its IP network. A variant of this design, called Source-Specific Multicast PIM, is optimized for the case that there is only one sender to the group. Finally, multicast extensions to BGP or tunnels need to be used to create multicast routes when the group members are in more than one AS.

5.8 POLICY AT THE NETWORK LAYER

Traffic management has become a topic related to policy in recent years, as streaming video traffic has become a dominant fraction of overall traffic and Internet interconnection has increasingly become direct between content providers and access networks. Two aspects of the network layer that relate to policy are peering disputes and traffic prioritization (sometimes associated with **net neutrality**). We will discuss each of these aspects below.

5.8.1 Peering Disputes

Although BGP is a technical standard, ultimately interconnection amounts to routing money. Traffic flows along paths that make service provider and transit networks the most money; paying for transit is considered a last resort. Settlement-free peering of course depends on both parties agreeing that interconnection is mutually beneficial. When one network feels it is getting the short end of the bargain, it can ask the other network to pay. The other connecting network might agree, or refuse, but if negotiations break down, this results in a so-called **peering dispute**.

A very high-profile peering dispute occurred a few years ago. In recent years, large content providers have been serving enough traffic to congest any interconnect link. In 2013, large video providers were congesting interconnect links between transit providers and residential access networks. Ultimately, the streaming video traffic filled the capacity of these links, creating high utilization on interconnection links that was difficult for access networks to mitigate without provisioning extra capacity. The question then became one of who should pay for augmenting the network capacity. In the end, in many cases, the large content providers ended up paying the access networks for direct interconnection, effectively a paid peering arrangement as discussed earlier in this chapter. Many wrongly construed

these circumstances as somehow relating to unfair de-prioritization or blocking of video traffic. In fact, the incidents resulted from business disputes concerning which network should be responsible for paying to provision interconnection points. For more information on peering disputes and how they are handled, see The Peering Playbook (Norton, 2012).

Peering disputes are as old as the commercial Internet. As a higher fraction of traffic on the Internet goes over private interconnects, however, the nature of these disputes is likely to evolve. For example, residential access networks now send a very high fraction of their own traffic to the same distributed clouds where other content is hosted. Thus, it is not in their interests to let the interconnects to those distributed cloud platforms experience high utilization. Recently, some operators have gone so far as to predict the death of transit connections entirely (Huston, 2018). Whether that comes to pass remains to be seen, but needless to say the dynamics of peering, interconnection, and transit continue to evolve rapidly.

5.8.2 Traffic Prioritization

Traffic prioritization, of the types that we have discussed earlier in this chapter, is a complicated topic that sometimes crosses over into the policy realm. On the one hand, a core aspect of traffic management is the prioritization of latency-sensitive traffic (e.g., gaming and interactive video) so that high utilization for other types of traffic (e.g., a large file transfer) does not result in poor overall user experience. Some applications such as file transfers do not require interactivity, whereas interactive applications often require low latency and jitter.

To achieve good performance for a mix of application traffic, network operators often institute various forms of traffic prioritization, including methods such as the weighted fair queueing approaches described earlier in this chapter. Additionally, as previously discussed, newer versions of DOCSIS will have support for placing interactive application traffic in low-latency queues. Differentiated treatment across different types of application traffic can in fact result in improved quality of experience for certain applications without negatively affecting the quality of experience for other classes of applications.

Prioritization starts to get messier, however, if and when money changes hands. The third rail in Internet policy is **paid prioritization**, whereby one party might pay an Internet service provider so that its traffic would receive higher priority than other competing traffic of the same application type. Such paid prioritization might be viewed as anti-competitive behavior. In other cases, a transit network with a particular service offering (e.g., video, or voice over IP) could prioritize its own service with respect to services from competitors. For example, in one instance, AT&T, was found to be blocking FaceTime video calls. For these reasons, prioritization can often be a sensitive flash point in discussions about **network neutrality** or **net neutrality**. The concept of net neutrality has complex legal and policy

implications beyond the scope of a technical networking textbook, but the generally agreed upon **bright-line rules** are:

1. No blocking.
2. No throttling.
3. No paid prioritization.
4. Disclosure of any prioritization practices.

Any net neutrality policy also generally allows exceptions for reasonable network management practices (e.g., prioritization to improve network efficiency, blocking or filtering for network security reasons). What constitutes ''reasonable'' is often left up to lawyers to decide. Another policy and legal question is who (i.e., what government agency) gets to decide what the rules are, and what the penalties should be for breaking them. Some aspects of the net neutrality policy debates in the United States, for example, are about whether an Internet service provider is more similar to a telephone utility company (e.g., AT&T) or to an information and content provider (e.g., Google). Depending on the answer to that question, different government agencies get to set the rules on everything from prioritization to privacy.

5.9 SUMMARY

The network layer provides services to the transport layer. It can be based on either datagrams or virtual circuits. In both cases, its main job is routing packets from the source to the destination. In datagram networks, a routing decision is made on every packet. In virtual-circuit networks, it is made when the virtual circuit is set up.

Many routing algorithms are used in computer networks. Flooding is a simple algorithm to send a packet along all paths. Most algorithms find the shortest path and adapt to changes in the network topology. The main algorithms are distance vector routing and link state routing. Most actual networks use one of these. Other important routing topics are the use of hierarchy in large networks, routing for mobile hosts, and broadcast, multicast, and anycast routing.

Networks can easily become congested, leading to increased delay and lost packets. Network designers attempt to avoid congestion by designing the network to have enough capacity, configuring the protocols to prefer uncongested routes, refusing to accept more traffic, signaling sources to slow down, and shedding load.

The next step beyond just dealing with congestion is to actually try to achieve a promised quality of service. Some applications care more about throughput whereas others care more about delay and jitter. The methods that can be used to provide different qualities of service include a combination of traffic shaping,

reserving resources at routers, and admission control. Approaches that have been designed for good quality of service include IETF integrated services (including RSVP) and differentiated services.

Networks differ in various ways, so when multiple networks are interconnected, problems can occur. When different networks have different maximum packet sizes, fragmentation may be needed. Different networks may run different routing protocols internally but need to run a common protocol externally. Sometimes the problems can be finessed by tunneling a packet through a hostile network, but if the source and destination networks use different technologies, this approach fails.

The Internet has a rich variety of protocols related to the network layer. These include the datagram protocol, IP, and associated control protocols such as ICMP, ARP, and DHCP. A connection-oriented protocol called MPLS carries IP packets across some networks. One of the main routing protocols used within networks is OSPF, and the routing protocol used across networks is BGP. The Internet is rapidly running out of IP addresses, so a new version of IP, IPv6, has been developed and is ever-so-slowly being deployed.

Some aspects of traffic engineering and management touch on policy-related issues. Two common issues are peering disputes, where networks cannot agree on the business terms of interconnection; and traffic prioritization, which is generally applied to mitigate adverse effects of congestion but can touch on issues related to network neutrality if it is applied in anti-competitive ways.

PROBLEMS

1. Give two example computer applications for which connection-oriented service is appropriate. Now give two examples for which connectionless service is best.

2. Datagram networks route each packet as a separate unit, independent of all others. Virtual-circuit networks do not have to do this, since each data packet follows a predetermined route. Does this observation mean that virtual-circuit networks do not need the capability to route isolated packets from an arbitrary source to an arbitrary destination? Explain your answer.

3. Give three examples of protocol parameters that might be negotiated when a connection is set up.

4. Assuming that all routers and hosts are working properly and that all software in both is free of all errors, is there any chance, however small, that a packet will be delivered to the wrong destination?

5. Show that the count-to-infinity problem shown in Fig. 5-10(b) can be solved by having routers add to their distance vectors the outgoing link for every destination and cost pair. For example, In Fig. 5-10(a), node *C* not only advertises a route to *A* with distance 2, it also communicates that this path goes through node *B*. Show the distances

from all routers to *A* after every distance vector exchange, until all routers realize *A* is no longer reachable.

6. Sketch a network topology different from the one in Fig. 5-10 for which including the next hop does not solve the count-to-infinity problem if node *A* fails.

7. Consider the network of Fig. 5-12(a). Distance vector routing is used, and the following link state packets have just come in at router *D*: from *A*: (*B*: 5, *E*: 4); from *B*: (*A*: 4, *C*: 1, *F*: 5); from *C*: (*B*: 3, *D*: 4, *E*: 3); from *E*: (*A*: 2, *C*: 2, *F*: 2); from *F*: (*B*: 1, *D*: 2, *E*: 3). The cost of the links from *D* to *C* and *F* are 3 and 4 respectively. What is *D*'s new routing table? Give both the outgoing line to use and the cost.

8. Give a simple heuristic for finding two paths through a network from a given source to a given destination that can survive the loss of any communication line (assuming two such paths exist). The routers are considered reliable enough, so it is not necessary to worry about the possibility of router crashes.

9. Consider the network of Fig. 5-12(a). Distance vector routing is used, and the following vectors have just come in to router *C*: from *B*: (5, 0, 8, 12, 6, 2); from *D*: (16, 12, 6, 0, 9, 10); and from *E*: (7, 6, 3, 9, 0, 4). The cost of the links from *C* to *B*, *D*, and *E*, are 6, 3, and 5, respectively. What is *C*'s new routing table? Give both the outgoing line to use and the cost.

10. If costs are recorded as 8-bit numbers in a 50-router network, and distance vectors are exchanged twice a second, how much bandwidth per (full-duplex) line is chewed up by the distributed routing algorithm? Assume that each router has three lines to other routers.

11. Explain the difference between routing, forwarding, and switching.

12. In Fig. 5-13, the Boolean OR of the two sets of *ACF* bits are 111 in every row. Is this just an accident here, or does it hold for all networks under all circumstances?

13. Consider the network and link costs shown in Fig. 5-12. This network uses link state routing. Node *F* broadcasts a message using reverse path forwarding. Sketch the broadcast tree used in this scenario.

14. For hierarchical routing with 4800 routers, what region and cluster sizes should be chosen to minimize the size of the routing table for a three-layer hierarchy? A good starting place is the hypothesis that a solution with k clusters of k regions of k routers is close to optimal, which means that k is about the cube root of 4800 (around 16). Use trial and error to check out combinations where all three parameters are in the general vicinity of 16.

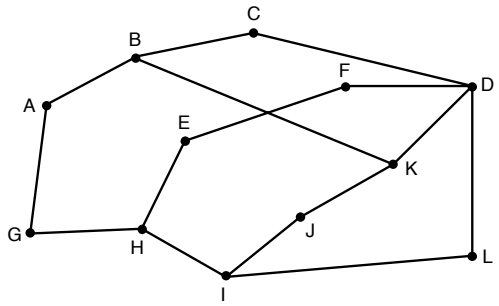
15. In the text it was stated that when a mobile host is not at home, packets sent to its home LAN are intercepted by its home agent on that LAN. For an IP network on an 802.3 LAN, how does the home agent accomplish this interception?

16. Looking at the network of Fig. 5-6, how many packets are generated by a broadcast from *B*, using

(a) reverse path forwarding?
(b) the sink tree?

17. Consider the network of Fig. 5-15(a). Imagine that one new line is added, between *F* and *G*, but the sink tree of Fig. 5-15(b) remains unchanged. What changes occur to Fig. 5-15(c)?

18. Compute a multicast spanning tree for router *C* in the following network for a group with members at routers *A*, *B*, *C*, *D*, *E*, *F*, *I*, and *K*.



19. Consider two hosts connected via a router. Explain how congestion can occur, even when both hosts and the router use flow control, but no congestion control. Then explain how the receiver can be overwhelmed, even when using congestion control, but no flow control.

20. As a possible congestion control mechanism in a network using virtual circuits internally, a router could refrain from acknowledging a received packet until (1) it knows its last transmission along the virtual circuit was received successfully and (2) it has a free buffer. For simplicity, assume that the routers use a stop-and-wait protocol and that each virtual circuit has one buffer dedicated to it for each direction of traffic. If it takes *T* sec to transmit a packet (data or acknowledgement) and there are *n* routers on the path, what is the rate at which packets are delivered to the destination host? Assume that transmission errors are rare and that the host-router connection is infinitely fast so it is not a bottleneck.

21. Describe two major differences between the ECN method and the RED method of congestion avoidance.

22. A token bucket scheme is used for traffic shaping. A new token is put into the bucket every 5 μsec. Each token is good for one short packet, which contains 48 bytes of data. What is the maximum sustainable data rate?

23. Explain how large file transfers could degrade the latency observed by both a gaming application and small file transfers.

24. A possible solution to the problem above involves shaping the file transfer traffic so that it never exceeds a certain rate. You decide to shape the traffic so that the sending rate never exceeds 20 Mbps. Should you use a token bucket or a leaky bucket to implement this shaping, or will neither work? What should the drain rate of the bucket be?

25. Given a sender who is sending at 100 Mbps, you would also like to automatically drop (police) traffic from the sender after 1 second. How large should you make the bucket in bytes?

26. A computer on a 6-Mbps network is regulated by a token bucket. The token bucket is filled at a rate of 1 Mbps. It is initially filled to capacity with 8 megabits. How long can the computer transmit at the full 6 Mbps?

27. A computer uses a token bucket with a capacity of 500 megabytes (MB), and a rate of 5 MB per second. The machine starts generating 15 MB per second when the bucket contains 300 MB. How long will it take to send 1000 MB?

28. Consider the packet queues shown in Fig. 5-29. What is the finish time and output order of the packets if the middle queue, instead of the bottom queue, has a weight of 2? Order packets with the same finish time alphabetically.

29. The network of Fig. 5-32 uses RSVP with multicast trees for hosts 1 and 2 as shown. Suppose that host 3 requests a channel of bandwidth 2 MB/sec for a flow from host 1 and another channel of bandwidth 1 MB/sec for a flow from host 2. At the same time, host 4 requests a channel of bandwidth 2 MB/sec for a flow from host 1 and host 5 requests a channel of bandwidth 1 MB/sec for a flow from host 2. How much total bandwidth will be reserved for these requests at routers A, B, C, E, H, J, K, and L?

30. A router can process 2 million packets/sec. The load offered to it is 1.5 million packets/sec on average. If a route from source to destination contains 10 routers, how much time is spent being queued and serviced by the router?

31. Consider the user of differentiated services with expedited forwarding. Is there a guarantee that expedited packets experience a shorter delay than regular packets? Why or why not?

32. A router is blasting out IP packets whose total length (data plus header) is 1024 bytes. Assuming that packets live for 10 sec, what is the maximum line speed the router can operate at without danger of cycling through the IP datagram ID number space?

33. An IP datagram using the *Strict source routing* option has to be fragmented. Do you think the option is copied into each fragment, or is it sufficient to just put it in the first fragment? Explain your answer.

34. Suppose that instead of using 16 bits for the network part of a class B address originally, 20 bits had been used. How many class B networks would there have been?

35. Convert the IP address whose hexadecimal representation is C22F1582 to dotted decimal notation.

36. Two IPv6-enabled devices wish to communicate across the Internet. Unfortunately, the path between these two devices includes a network that has not yet deployed IPv6. Design a way for the two devices to communicate.

37. A network on the Internet has a subnet mask of 255.255.240.0. What is the maximum number of hosts it can handle?

38. While IP addresses are tried to specific networks, Ethernet addresses are not. Can you think of a good reason why they are not?

39. A router has just received the following new IP addresses: 57.6.96.0/21, 57.6.104.0/21, 57.6.112.0/21, and 57.6.120.0/21. If all of them use the same outgoing line, can they be aggregated? If so, to what? If not, why not?

40. A router has the following (CIDR) entries in its routing table:

Address/mask	Next hop
135.46.56.0/22	Interface 0
135.46.60.0/22	Interface 1
192.53.40.0/23	Router 1
default	Router 2

For each of the following IP addresses, what does the router do if a packet with that address arrives?

(a) 135.46.63.10
(b) 135.46.57.14
(c) 135.46.52.2
(d) 192.53.40.7
(e) 192.53.56.7

41. Aggregate these three address ranges:
37.60.64.0/18
37.60.96.0/19
37.60.128.0/17

42. Many companies have a policy of having two (or more) routers connecting the company to the Internet to provide some redundancy in case one of them goes down. Is this policy still possible with NAT? Explain your answer.

43. Two machines on the same network try to use the same port number to communicate with a server on another network. Is this possible? Explain why (not). What changes if these machines are separated from other networks by a NAT box?

44. You have just explained the ARP protocol to a friend. When you are all done, he says: ''I've got it. ARP provides a service to the network layer, so it is part of the data link layer.'' What do you say to him?

45. You connect your phone to the wireless network at your home. This wireless network is created by the modem obtained from your ISP. Using DHCP, your phone obtains IP address 192.168.0.103. What is the likely source IP address of the DHCP OFFER message?

46. Describe a way to reassemble IP fragments at the destination.

47. In IP, the checksum covers only the header and not the data. Why do you suppose this design was chosen?

48. A person who lives in Boston travels to Minneapolis, taking her portable computer with her. To her surprise, the LAN at her destination in Minneapolis is a wireless IP LAN, so she does not have to plug in. Is it still necessary to go through the entire business with home agents and foreign agents to make email and other traffic arrive correctly?

49. IPv6 uses 16-byte addresses. If a block of 1 million addresses is allocated every picosecond, how long will the addresses last?

50. One of the solutions ISPs use to deal with the shortage of IPv4 addresses is to dynamically allocate them to their clients. Once IPv6 is fully deployed, the address space is large enough to give every device a unique address. To reduce system complexity, IPv6 addresses could be assigned to devices permanently. Explain why this is not a good idea.

51. The *Protocol* field used in the IPv4 header is not present in the fixed IPv6 header. Why not?

52. When the IPv6 protocol is introduced, does the ARP protocol have to be changed? If so, are the changes conceptual or technical?

53. Write a program to simulate routing using flooding. Each packet should contain a counter that is decremented on each hop. When the counter gets to zero, the packet is discarded. Time is discrete, with each line handling one packet per time interval. Make three versions of the program: all lines are flooded, all lines except the input line are flooded, and only the (statically chosen) best k lines are flooded. Compare flooding with deterministic routing ($k = 1$) in terms of both delay and the bandwidth used.

54. Write a program that simulates a computer network using discrete time. The first packet on each router queue makes one hop per time interval. Each router has only a finite number of buffers. If a packet arrives and there is no room for it, it is discarded and not retransmitted. Instead, there is an end-to-end protocol, complete with timeouts and acknowledgement packets, that eventually regenerates the packet from the source router. Plot the throughput of the network as a function of the end-to-end timeout interval, parameterized by error rate.

55. Write a function to do forwarding in an IP router. The procedure has one parameter, an IP address. It also has access to a global table consisting of an array of triples. Each triple contains three integers: an IP address, a subnet mask, and the outline line to use. The function looks up the IP address in the table using CIDR and returns the line to use as its value.

56. Use the *traceroute* (UNIX) or *tracert* (Windows) programs to trace the route from your computer to various universities on other continents. Make a list of transoceanic links you have discovered. Some sites to try are

www.berkeley.edu (California)
www.mit.edu (Massachusetts)
www.vu.nl (Amsterdam)
www.ucl.ac.uk (London)
www.usyd.edu.au (Sydney)
www.u-tokyo.ac.jp (Tokyo)
www.uct.ac.za (Cape Town)