payload, the channel capacity from the slave is 64,000 bps as is the channel capacity from the master. This capacity is exactly enough for a single full-duplex PCM voice channel (which is why a hop rate of 1600 hops/sec was chosen). That is, despite a raw bandwidth of 1 Mbps, a single full-duplex uncompressed voice channel can completely saturate the piconet. The efficiency of 13% is the result of spending 41% of the capacity on settling time, 20% on headers, and 26% on repetition coding. This shortcoming highlights the value of the enhanced rates and frames of more than a single slot.

There is much more to be said about Bluetooth, but no more space to say it here. For the curious, the Bluetooth 4.0 specification contains all the details.

4.7 RFID

We have looked at MAC designs from LANs up to MANs and down to PANs. As a last example, we will study a category of low-end wireless devices that people may not recognize as forming a computer network: the **RFID** (**Radio Frequency IDentification**) tags and readers that we described in Sec. 1.5.4.

RFID technology takes many forms, used in smartcards, implants for pets, passports, library books, and more. The form that we will look at was developed in the quest for an **EPC** (**Electronic Product Code**) that started with the Auto-ID Center at the Massachusetts Institute of Technology in 1999. An EPC is a replacement for a barcode that can carry a larger amount of information and is electronically readable over distances up to 10 m, even when it is not visible. It is different technology than, for example, the RFID used in passports,which must be placed quite close to a reader to perform a transaction. The ability to communicate over a distance makes EPCs more relevant to our studies.

EPCglobal was formed in 2003 to commercialize the RFID technology developed by the Auto-ID Center. The effort got a boost in 2005 when Walmart required its top 100 suppliers to label all shipments with RFID tags. Widespread deployment has been hampered by the difficulty of competing with cheap printed barcodes, but new uses, such as in drivers licenses, are now growing. We will describe the second generation of this technology, which is informally called **EPC Gen 2** (EPCglobal, 2008).

4.7.1 EPC Gen 2 Architecture

The architecture of an EPC Gen 2 RFID network is shown in Fig. 4-37. It has two key components: tags and readers. RFID tags are small, inexpensive devices that have a unique 96-bit EPC identifier and a small amount of memory that can be read and written by the RFID reader. The memory might be used to record the location history of an item, for example, as it moves through the supply chain.

Often, the tags look like stickers that can be placed on, for example, pairs of jeans on the shelves in a store. Most of the sticker is taken up by an antenna that is printed onto it. A tiny dot in the middle is the RFID integrated circuit. Alternatively, the RFID tags can be integrated into an object, such as a driver's license. In both cases, the tags have no battery and they must gather power from the radio transmissions of a nearby RFID reader to run. This kind of tag is called a "Class 1" tag to distinguish it from more capable tags that have batteries.

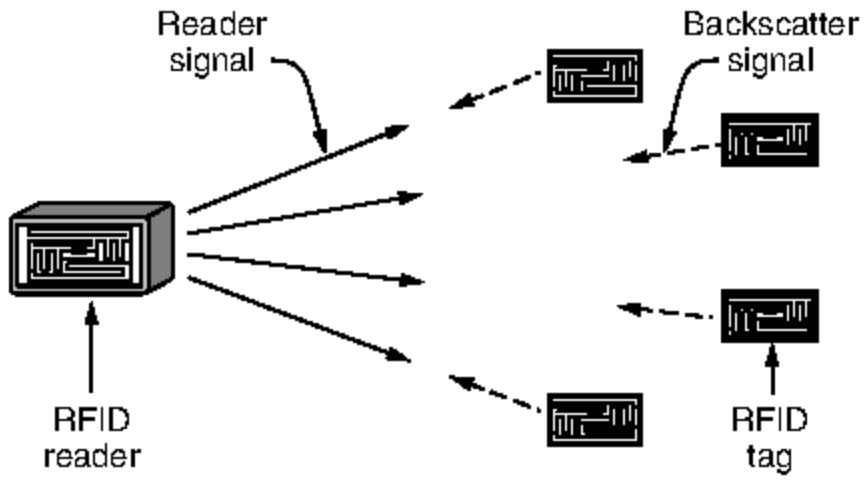


Figure 4-37. RFID architecture.

The readers are the intelligence in the system, analogous to base stations and access points in cellular and WiFi networks. Readers are much more powerful than tags. They have their own power sources, often have multiple antennas, and are in charge of when tags send and receive messages. As there will commonly be multiple tags within the reading range, the readers must solve the multiple access problem. There may be multiple readers that can contend with each other in the same area, too.

The main job of the reader is to inventory the tags in the neighborhood, that is, to discover the identifiers of the nearby tags. The inventory is accomplished with the physical layer protocol and the tag-identification protocol that are outlined in the following sections.

4.7.2 EPC Gen 2 Physical Layer

The physical layer defines how bits are sent between the RFID reader and tags. Much of it uses methods for sending wireless signals that we have seen previously. In the U.S., transmissions are sent in the unlicensed 902–928 MHz ISM band. This band falls in the UHF (Ultra High Frequency) range, so the tags are referred to as UHF RFID tags. The reader performs frequency hopping at least every 400 msec to spread its signal across the channel, to limit interference and satisfy regulatory requirements. The reader and tags use forms of ASK (Amplitude Shift Keying) modulation that we described in Sec. 2.5.2 to encode bits. They take turns to send bits, so the link is half duplex.

There are two main differences from other physical layers that we have studied. The first is that the reader is always transmitting a signal, regardless of whether it is the reader or tag that is communicating. Naturally, the reader transmits a signal to send bits to tags. For the tags to send bits to the reader, the reader transmits a fixed carrier signal that carries no bits. The tags harvest this signal to get the power they need to run; otherwise, a tag would not be able to transmit in the first place. To send data, a tag changes whether it is reflecting the signal from the reader, like a radar signal bouncing off a target, or absorbing it.

This method is called **backscatter**. It differs from all the other wireless situations we have seen so far, in which the sender and receiver never both transmit at the same time. Backscatter is a low-energy way for the tag to create a weak signal of its own that shows up at the reader. For the reader to decode the incoming signal, it must filter out the outgoing signal that it is transmitting. Because the tag signal is weak, tags can only send bits to the reader at a low rate, and tags cannot receive or even sense transmissions from other tags.

The second difference is that very simple forms of modulation are used so that they can be implemented on a tag that runs on very little power and costs only a few cents to make. To send data to the tags, the reader uses two amplitude levels. Bits are determined to be either a 0 or a 1, depending on how long the reader waits before a low-power period. The tag measures the time between low-power periods and compares this time to a reference measured during a preamble. As shown in Fig. 4-38, 1s are longer than 0s.

Tag responses consist of the tag alternating its backscatter state at fixed intervals to create a series of pulses in the signal. Anywhere from one to eight pulse periods can be used to encode each 0 or 1, depending on the need for reliability. 1s have fewer transitions than 0s, as is shown with an example of two-pulse period coding in Fig. 4-38.

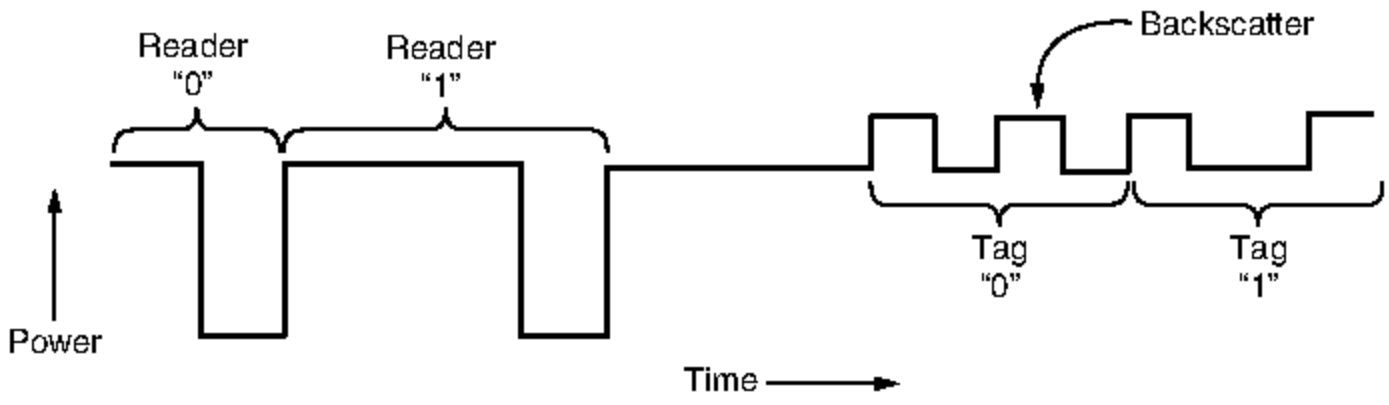


Figure 4-38. Reader and tag backscatter signals.

4.7.3 EPC Gen 2 Tag Identification Layer

To inventory the nearby tags, the reader needs to receive a message from each tag that gives the identifier for the tag. This situation is a multiple access problem for which the number of tags is unknown in the general case. The reader might

broadcast a query to ask all tags to send their identifiers. However, tags that replied right away would then collide in much the same way as stations on a classic Ethernet.

We have seen many ways of tackling the multiple access problem in this chapter. The closest protocol for the current situation, in which the tags cannot hear each others' transmissions, is slotted ALOHA, one of the earliest protocols we studied. This protocol is adapted for use in Gen 2 RFID.

The sequence of messages used to identify a tag is shown in Fig. 4-39. In the first slot (slot 0), the reader sends a *Query* message to start the process. Each *QRepeat* message advances to the next slot. The reader also tells the tags the range of slots over which to randomize transmissions. Using a range is necessary because the reader synchronizes tags when it starts the process; unlike stations on an Ethernet, tags do not wake up with a message at a time of their choosing.

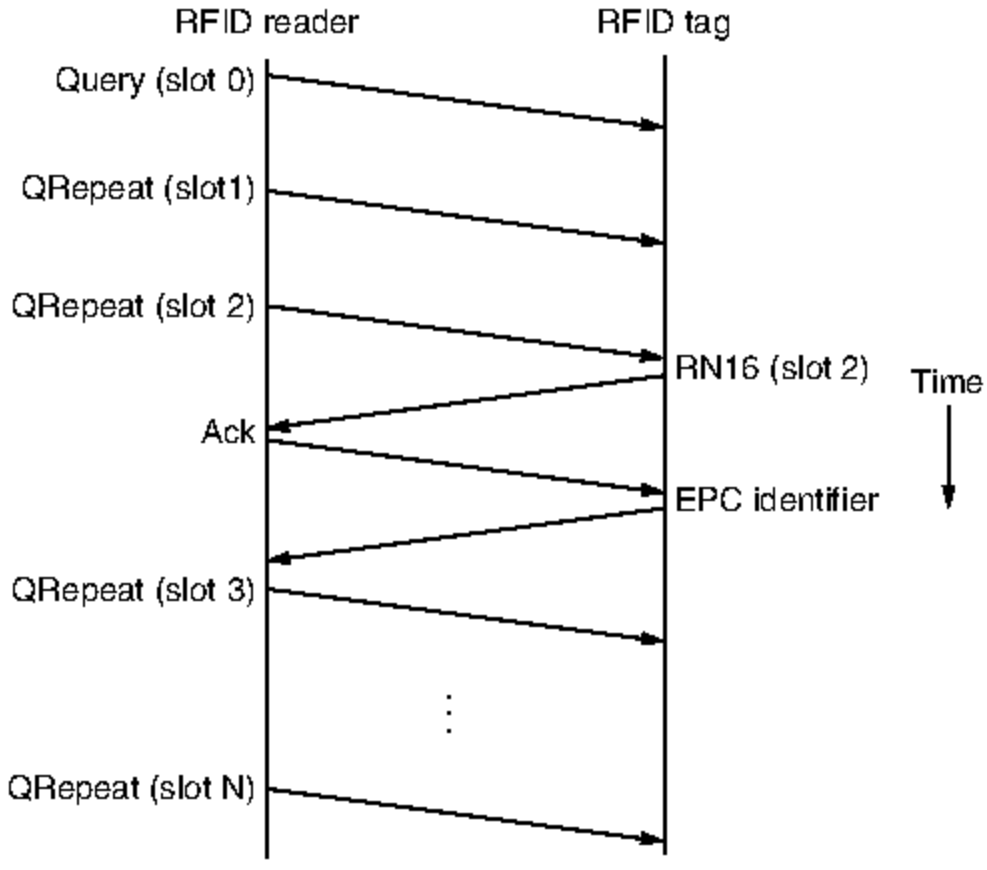


Figure 4-39. Example message exchange to identify a tag.

Tags pick a random slot in which to reply. In Fig. 4-39, the tag replies in slot 2. However, tags do not send their identifiers when they first reply. Instead, a tag sends a short 16-bit random number in an *RN16* message. If there is no collision, the reader receives this message and sends an *ACK* message of its own. At this stage, the tag has acquired the slot and sends its EPC identifier.

The reason for this exchange is that EPC identifiers are long, so collisions on these messages would be expensive. Instead, a short exchange is used to test whether the tag can safely use the slot to send its identifier. Once its identifier has been successfully transmitted, the tag temporarily stops responding to new *Query* messages so that all the remaining tags can be identified.

A key problem is for the reader to adjust the number of slots to avoid collisions, but without using so many slots that performance suffers. This adjustment is analogous to binary exponential backoff in Ethernet. If the reader sees too many slots with no responses or too many slots with collisions, it can send a *QAdjust* message to decrease or increase the range of slots over which the tags are responding.

The RFID reader can perform other operations on the tags. For example, it can select a subset of tags before running an inventory, allowing it to collect responses from, say, tagged jeans but not tagged shirts. The reader can also write data to tags as they are identified. This feature could be used to record the point of sale or other relevant information.

4.7.4 Tag Identification Message Formats

The format of the *Query* message is shown in Fig. 4-40 as an example of a reader-to-tag message. The message is compact because the downlink rates are limited, from 27 kbps up to 128 kbps. The *Command* field carries the code 1000 to identify the message as a *Query*.

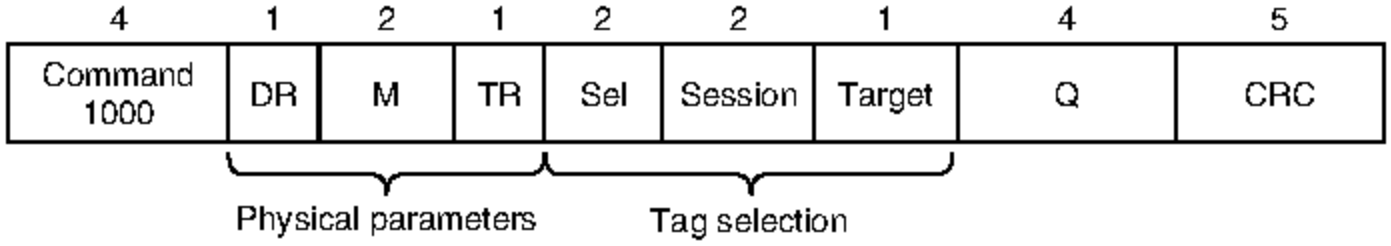


Figure 4-40. Format of the Query message.

The next flags, *DR*, *M*, and *TR*, determine the physical layer parameters for reader transmissions and tag responses. For example, the response rate may be set to between 5 kbps and 640 kbps. We will skip over the details of these flags.

Then come three fields, *Sel*, *Session*, and *Target*, that select the tags to respond. As well as the readers being able to select a subset of identifiers, the tags keep track of up to four concurrent sessions and whether they have been identified in those sessions. In this way, multiple readers can operate in overlapping coverage areas by using different sessions.

Next is the most important parameter for this command, Q. This field defines the range of slots over which tags will respond, from 0 to 2^Q-1. Finally, there is a CRC to protect the message fields. At 5 bits, it is shorter than most CRCs we have seen, but the *Query* message is much shorter than most packets too.

Tag-to-reader messages are simpler. Since the reader is in control, it knows what message to expect in response to each of its transmissions. The tag responses simply carry data, such as the EPC identifier.

Originally the tags were just for identification purposes. However, they have grown over time to resemble very small computers. Some research tags have sensors and are able to run small programs to gather and process data (Sample et al., 2008). One vision for this technology is the "Internet of things" that connects objects in the physical world to the Internet (Welbourne et al., 2009; and Gershenfeld et al., 2004).

4.8 DATA LINK LAYER SWITCHING

Many organizations have multiple LANs and wish to connect them. Would it not be convenient if we could just join the LANs together to make a larger LAN? In fact, we can do this when the connections are made with devices called **bridges**. The Ethernet switches we described in Sec. 4.3.4 are a modern name for bridges; they provide functionality that goes beyond classic Ethernet and Ethernet hubs to make it easy to join multiple LANs into a larger and faster network. We shall use the terms "bridge" and "switch" interchangeably.

Bridges operate in the data link layer, so they examine the data link layer addresses to forward frames. Since they are not supposed to examine the payload field of the frames they forward, they can handle IP packets as well as other kinds of packets, such as AppleTalk packets. In contrast, *routers* examine the addresses in packets and route based on them, so they only work with the protocols that they were designed to handle.

In this section, we will look at how bridges work and are used to join multiple physical LANs into a single logical LAN. We will also look at how to do the reverse and treat one physical LAN as multiple logical LANs, called **VLANs** (**Virtual LANs**). Both technologies provide useful flexibility for managing networks. For a comprehensive treatment of bridges, switches, and related topics, see Seifert and Edwards (2008) and Perlman (2000).

4.8.1 Uses of Bridges

Before getting into the technology of bridges, let us take a look at some common situations in which bridges are used. We will mention three reasons why a single organization may end up with multiple LANs.

First, many university and corporate departments have their own LANs to connect their own personal computers, servers, and devices such as printers. Since the goals of the various departments differ, different departments may set up different LANs, without regard to what other departments are doing. Sooner or later, though, there is a need for interaction, so bridges are needed. In this example, multiple LANs come into existence due to the autonomy of their owners.

Second, the organization may be geographically spread over several buildings separated by considerable distances. It may be cheaper to have separate LANs in each building and connect them with bridges and a few long-distance fiber optic links than to run all the cables to a single central switch. Even if laying the cables is easy to do, there are limits on their lengths (e.g., 200 m for twisted-pair gigabit Ethernet). The network would not work for longer cables due to the excessive signal attenuation or round-trip delay. The only solution is to partition the LAN and install bridges to join the pieces to increase the total physical distance that can be covered.

Third, it may be necessary to split what is logically a single LAN into separate LANs (connected by bridges) to accommodate the load. At many large universities, for example, thousands of workstations are available for student and faculty computing. Companies may also have thousands of employees. The scale of this system precludes putting all the workstations on a single LAN—there are more computers than ports on any Ethernet hub and more stations than allowed on a single classic Ethernet.

Even if it were possible to wire all the workstations together, putting more stations on an Ethernet hub or classic Ethernet would not add capacity. All of the stations share the same, fixed amount of bandwidth. The more stations there are, the less average bandwidth per station.

However, two separate LANs have twice the capacity of a single LAN. Bridges let the LANs be joined together while keeping this capacity. The key is not to send traffic onto ports where it is not needed, so that each LAN can run at full speed. This behavior also increases reliability, since on a single LAN a defective node that keeps outputting a continuous stream of garbage can clog up the entire LAN. By deciding what to forward and what not to forward, bridges act like fire doors in a building, preventing a single node that has gone berserk from bringing down the entire system.

To make these benefits easily available, ideally bridges should be completely transparent. It should be possible to go out and buy bridges, plug the LAN cables into the bridges, and have everything work perfectly, instantly. There should be no hardware changes required, no software changes required, no setting of address switches, no downloading of routing tables or parameters, nothing at all. Just plug in the cables and walk away. Furthermore, the operation of the existing LANs should not be affected by the bridges at all. As far as the stations are concerned, there should be no observable difference whether or not they are part of a bridged LAN. It should be as easy to move stations around the bridged LAN as it is to move them around a single LAN.

Surprisingly enough, it is actually possible to create bridges that are transparent. Two algorithms are used: a backward learning algorithm to stop traffic being sent where it is not needed; and a spanning tree algorithm to break loops that may be formed when switches are cabled together willy-nilly. Let us now take a look at these algorithms in turn to learn how this magic is accomplished.

4.8.2 Learning Bridges

The topology of two LANs bridged together is shown in Fig. 4-41 for two cases. On the left-hand side, two multidrop LANs, such as classic Ethernets, are joined by a special station—the bridge—that sits on both LANs. On the right-hand side, LANs with point-to-point cables, including one hub, are joined together. The bridges are the devices to which the stations and hub are attached. If the LAN technology is Ethernet, the bridges are better known as Ethernet switches.

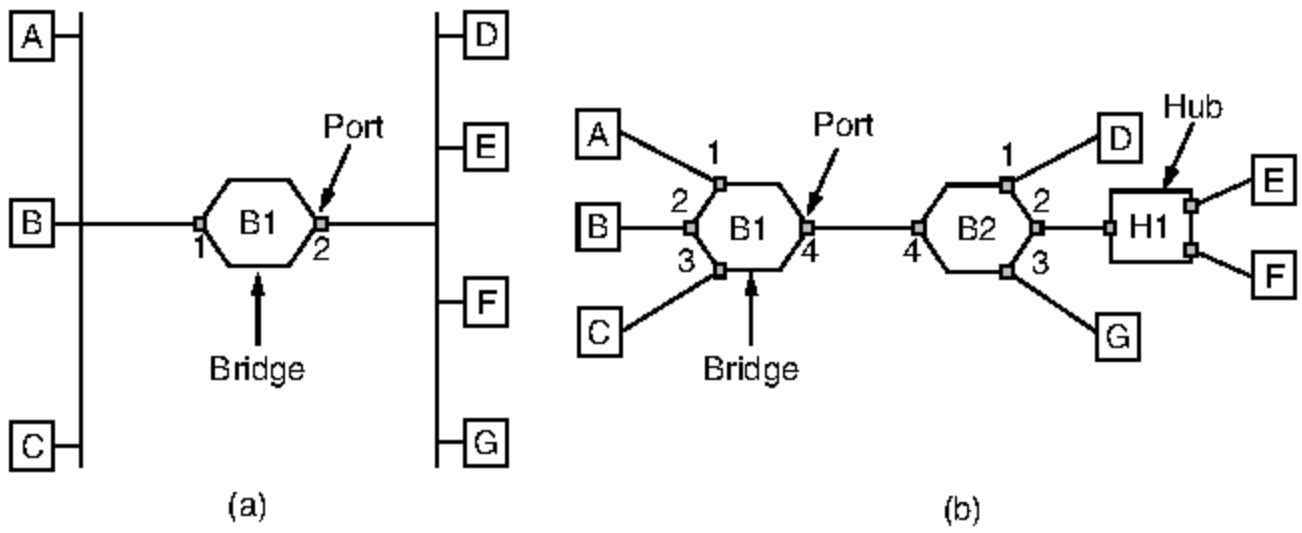


Figure 4-41. (a) Bridge connecting two multidrop LANs. (b) Bridges (and a hub) connecting seven point-to-point stations.

Bridges were developed when classic Ethernets were in use, so they are often shown in topologies with multidrop cables, as in Fig. 4-41(a). However, all the topologies that are encountered today are comprised of point-to-point cables and switches. The bridges work the same way in both settings. All of the stations attached to the same port on a bridge belong to the same collision domain, and this is different than the collision domain for other ports. If there is more than one station, as in a classic Ethernet, a hub, or a half-duplex link, the CSMA/CD protocol is used to send frames.

There is a difference, however, in how the bridged LANs are built. To bridge multidrop LANs, a bridge is added as a new station on each of the multidrop LANs, as in Fig. 4-41(a). To bridge point-to-point LANs, the hubs are either connected to a bridge or, preferably, replaced with a bridge to increase performance. In Fig. 4-41(b), bridges have replaced all but one hub.

Different kinds of cables can also be attached to one bridge. For example, the cable connecting bridge *B1* to bridge *B2* in Fig. 4-41(b) might be a long-distance fiber optic link, while the cable connecting the bridges to stations might be a short-haul twisted-pair line. This arrangement is useful for bridging LANs in different buildings.

Now let us consider what happens inside the bridges. Each bridge operates in promiscuous mode, that is, it accepts every frame transmitted by the stations

attached to each of its ports. The bridge must decide whether to forward or discard each frame, and, if the former, on which port to output the frame. This decision is made by using the destination address. As an example, consider the topology of Fig. 4-41(a). If station *A* sends a frame to station *B*, bridge *B1* will receive the frame on port 1. This frame can be immediately discarded without further ado because it is already on the correct port. However, in the topology of Fig. 4-41(b) suppose that *A* sends a frame to *D*. Bridge *B1* will receive the frame on port 1 and output it on port 4. Bridge *B2* will then receive the frame on its port 4 and output it on its port 1.

A simple way to implement this scheme is to have a big (hash) table inside the bridge. The table can list each possible destination and which output port it belongs on. For example, in Fig. 4-41(b), the table at *B1* would list *D* as belonging to port 4, since all *B1* has to know is which port to put frames on to reach *D*. That, in fact, more forwarding will happen later when the frame hits *B2* is not of interest to *B1*.

When the bridges are first plugged in, all the hash tables are empty. None of the bridges know where any of the destinations are, so they use a flooding algorithm: every incoming frame for an unknown destination is output on all the ports to which the bridge is connected except the one it arrived on. As time goes on, the bridges learn where destinations are. Once a destination is known, frames destined for it are put only on the proper port; they are not flooded.

The algorithm used by the bridges is **backward learning**. As mentioned above, the bridges operate in promiscuous mode, so they see every frame sent on any of their ports. By looking at the source addresses, they can tell which machines are accessible on which ports. For example, if bridge *B1* in Fig. 4-41(b) sees a frame on port 3 coming from *C*, it knows that *C* must be reachable via port 3, so it makes an entry in its hash table. Any subsequent frame addressed to *C* coming in to *B1* on any other port will be forwarded to port 3.

The topology can change as machines and bridges are powered up and down and moved around. To handle dynamic topologies, whenever a hash table entry is made, the arrival time of the frame is noted in the entry. Whenever a frame whose source is already in the table arrives, its entry is updated with the current time. Thus, the time associated with every entry tells the last time a frame from that machine was seen.

Periodically, a process in the bridge scans the hash table and purges all entries more than a few minutes old. In this way, if a computer is unplugged from its LAN, moved around the building, and plugged in again somewhere else, within a few minutes it will be back in normal operation, without any manual intervention. This algorithm also means that if a machine is quiet for a few minutes, any traffic sent to it will have to be flooded until it next sends a frame itself.

The routing procedure for an incoming frame depends on the port it arrives on (the source port) and the address to which it is destined (the destination address). The procedure is as follows.

1. If the port for the destination address is the same as the source port, discard the frame.
2. If the port for the destination address and the source port are different, forward the frame on to the destination port.
3. If the destination port is unknown, use flooding and send the frame on all ports except the source port.

You might wonder whether the first case can occur with point-to-point links. The answer is that it can occur if hubs are used to connect a group of computers to a bridge. An example is shown in Fig. 4-41(b) where stations *E* and *F* are connected to hub *H*1, which is in turn connected to bridge *B2*. If *E* sends a frame to *F*, the hub will relay it to *B2* as well as to *F*. That is what hubs do—they wire all ports together so that a frame input on one port is simply output on all other ports. The frame will arrive at *B2* on port 4, which is already the right output port to reach the destination. Bridge *B2* need only discard the frame.

As each frame arrives, this algorithm must be applied, so it is usually implemented with special-purpose VLSI chips. The chips do the lookup and update the table entry, all in a few microseconds. Because bridges only look at the MAC addresses to decide how to forward frames, it is possible to start forwarding as soon as the destination header field has come in, before the rest of the frame has arrived (provided the output line is available, of course). This design reduces the latency of passing through the bridge, as well as the number of frames that the bridge must be able to buffer. It is referred to as **cut-through switching** or **wormhole routing** and is usually handled in hardware.

We can look at the operation of a bridge in terms of protocol stacks to understand what it means to be a link layer device. Consider a frame sent from station *A* to station *D* in the configuration of Fig. 4-41(a), in which the LANs are Ethernet. The frame will pass through one bridge. The protocol stack view of processing is shown in Fig. 4-42.

The packet comes from a higher layer and descends into the Ethernet MAC layer. It acquires an Ethernet header (and also a trailer, not shown in the figure). This unit is passed to the physical layer, goes out over the cable, and is picked up by the bridge.

In the bridge, the frame is passed up from the physical layer to the Ethernet MAC layer. This layer has extended processing compared to the Ethernet MAC layer at a station. It passes the frame to a relay, still within the MAC layer. The bridge relay function uses only the Ethernet MAC header to determine how to handle the frame. In this case, it passes the frame to the Ethernet MAC layer of the port used to reach station *D*, and the frame continues on its way.

In the general case, relays at a given layer can rewrite the headers for that layer. VLANs will provide an example shortly. In no case should the bridge look inside the frame and learn that it is carrying an IP packet; that is irrelevant to the

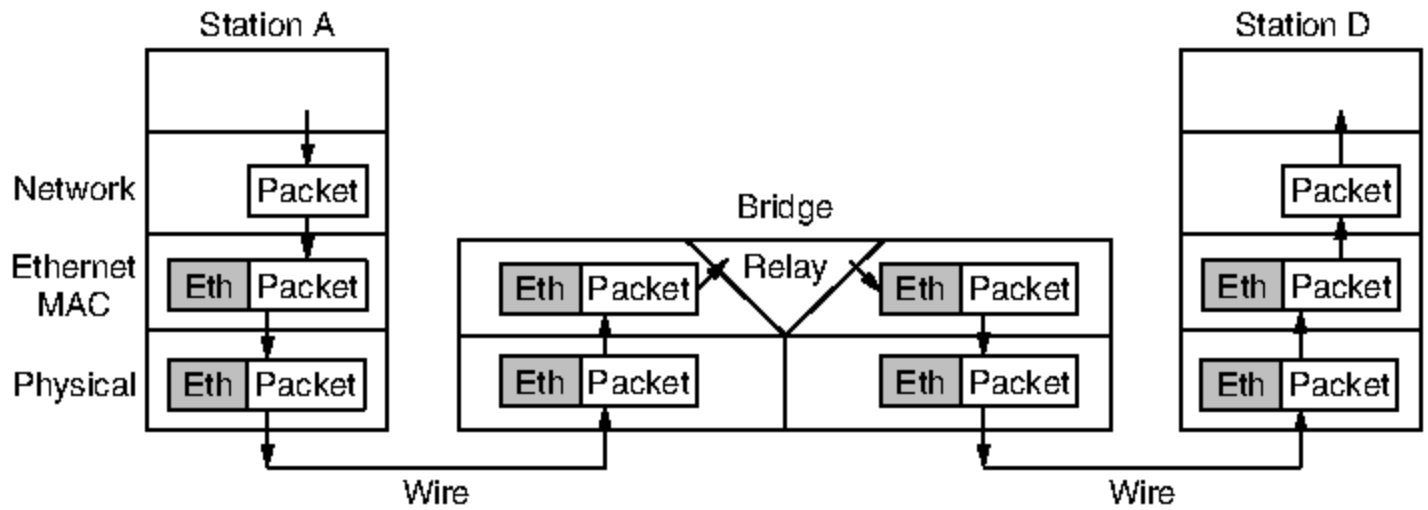


Figure 4-42. Protocol processing at a bridge.

bridge processing and would violate protocol layering. Also note that a bridge with k ports will have k instances of MAC and physical layers. The value of k is 2 for our simple example.

4.8.3 Spanning Tree Bridges

To increase reliability, redundant links can be used between bridges. In the example of Fig. 4-43, there are two links in parallel between a pair of bridges. This design ensures that if one link is cut, the network will not be partitioned into two sets of computers that cannot talk to each other.

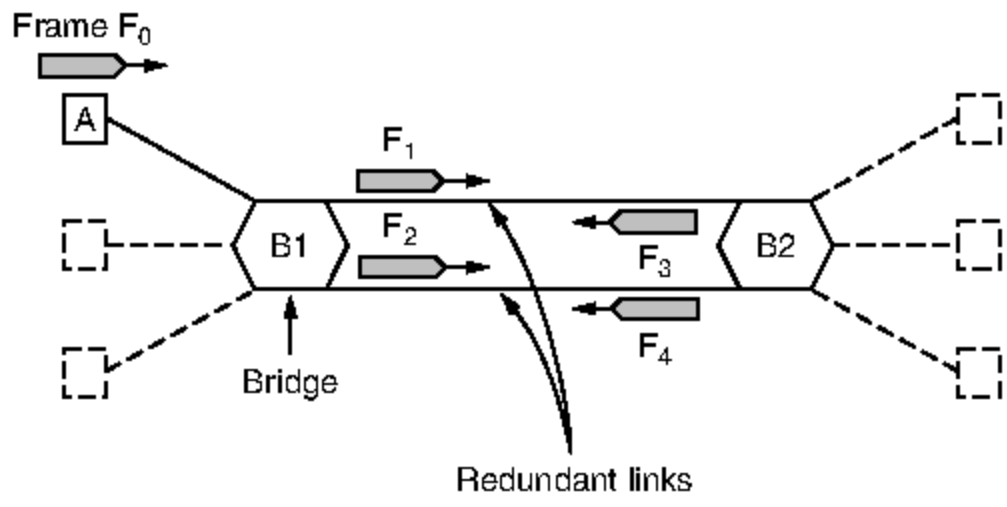


Figure 4-43. Bridges with two parallel links.

However, this redundancy introduces some additional problems because it creates loops in the topology. An example of these problems can be seen by looking at how a frame sent by A to a previously unobserved destination is handled in Fig. 4-43. Each bridge follows the normal rule for handling unknown destinations, which is to flood the frame. Call the frame from A that reaches bridge $B1$ frame F_0. The bridge sends copies of this frame out all of its other ports. We

will only consider the bridge ports that connect *B1* to *B2* (though the frame will be sent out the other ports, too). Since there are two links from *B1* to *B2*, two copies of the frame will reach *B2*. They are shown in Fig. 4-43 as F_1 and F_2.

Shortly thereafter, bridge *B2* receives these frames. However, it does not (and cannot) know that they are copies of the same frame, rather than two different frames sent one after the other. So bridge *B2* takes F_1 and sends copies of it out all the other ports, and it also takes F_2 and sends copies of it out all the other ports. This produces frames F_3 and F_4 that are sent along the two links back to *B1*. Bridge *B1* then sees two new frames with unknown destinations and copies them again. This cycle goes on forever.

The solution to this difficulty is for the bridges to communicate with each other and overlay the actual topology with a spanning tree that reaches every bridge. In effect, some potential connections between bridges are ignored in the interest of constructing a fictitious loop-free topology that is a subset of the actual topology.

For example, in Fig. 4-44 we see five bridges that are interconnected and also have stations connected to them. Each station connects to only one bridge. There are some redundant connections between the bridges so that frames will be forwarded in loops if all of the links are used. This topology can be thought of as a graph in which the bridges are the nodes and the point-to-point links are the edges. The graph can be reduced to a spanning tree, which has no cycles by definition, by dropping the links shown as dashed lines in Fig. 4-44. Using this spanning tree, there is exactly one path from every station to every other station. Once the bridges have agreed on the spanning tree, all forwarding between stations follows the spanning tree. Since there is a unique path from each source to each destination, loops are impossible.

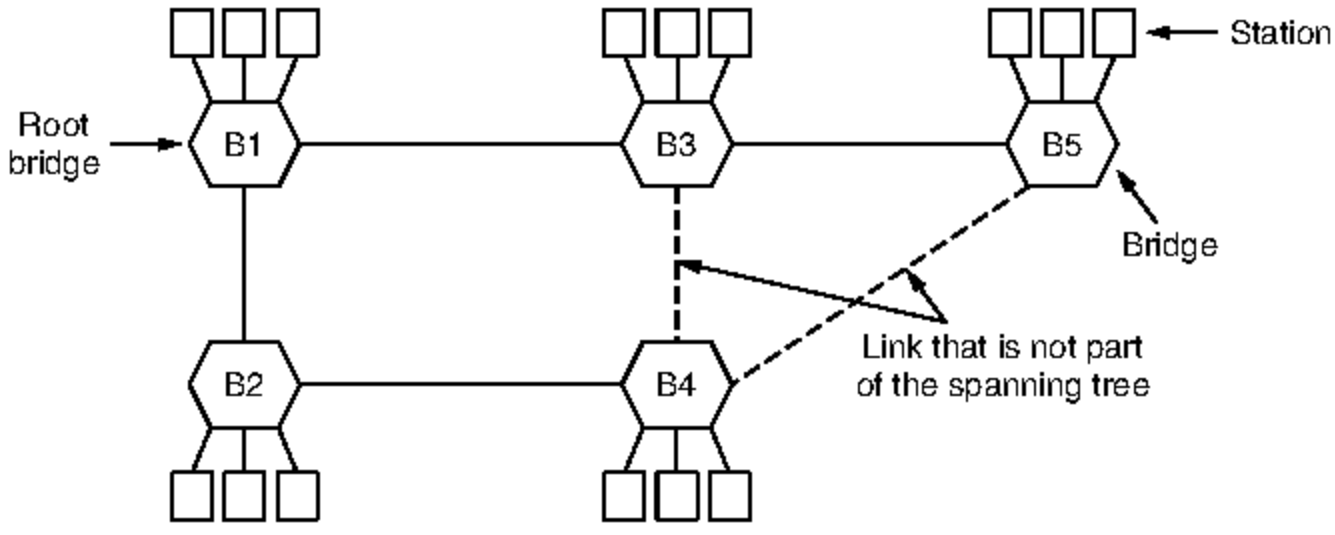


Figure 4-44. A spanning tree connecting five bridges. The dashed lines are links that are not part of the spanning tree.

To build the spanning tree, the bridges run a distributed algorithm. Each bridge periodically broadcasts a configuration message out all of its ports to its

neighbors and processes the messages it receives from other bridges, as described next. These messages are not forwarded, since their purpose is to build the tree, which can then be used for forwarding.

The bridges must first choose one bridge to be the root of the spanning tree. To make this choice, they each include an identifier based on their MAC address in the configuration message, as well as the identifier of the bridge they believe to be the root. MAC addresses are installed by the manufacturer and guaranteed to be unique worldwide, which makes these identifiers convenient and unique. The bridges choose the bridge with the lowest identifier to be the root. After enough messages have been exchanged to spread the news, all bridges will agree on which bridge is the root. In Fig. 4-44, bridge *B1* has the lowest identifier and becomes the root.

Next, a tree of shortest paths from the root to every bridge is constructed. In Fig. 4-44, bridges *B2* and *B3* can each be reached from bridge *B1* directly, in one hop that is a shortest path. Bridge *B4* can be reached in two hops, via either *B2* or *B3*. To break this tie, the path via the bridge with the lowest identifier is chosen, so *B4* is reached via *B2*. Bridge *B5* can be reached in two hops via *B3*.

To find these shortest paths, bridges include the distance from the root in their configuration messages. Each bridge remembers the shortest path it finds to the root. The bridges then turn off ports that are not part of the shortest path.

Although the tree spans all the bridges, not all the links (or even bridges) are necessarily present in the tree. This happens because turning off the ports prunes some links from the network to prevent loops. Even after the spanning tree has been established, the algorithm continues to run during normal operation to automatically detect topology changes and update the tree.

The algorithm for constructing the spanning tree was invented by Radia Perlman. Her job was to solve the problem of joining LANs without loops. She was given a week to do it, but she came up with the idea for the spanning tree algorithm in a day. Fortunately, this left her enough time to write it as a poem (Perlman, 1985):

I think that I shall never see
A graph more lovely than a tree.
A tree whose crucial property
Is loop-free connectivity.
A tree which must be sure to span.
So packets can reach every LAN.
First the Root must be selected
By ID it is elected.
Least cost paths from Root are traced
In the tree these paths are placed.
A mesh is made by folks like me
Then bridges find a spanning tree.

The spanning tree algorithm was then standardized as IEEE 802.1D and used for many years. In 2001, it was revised to more rapidly find a new spanning tree after a topology change. For a detailed treatment of bridges, see Perlman (2000).

4.8.4 Repeaters, Hubs, Bridges, Switches, Routers, and Gateways

So far in this book, we have looked at a variety of ways to get frames and packets from one computer to another. We have mentioned repeaters, hubs, bridges, switches, routers, and gateways. All of these devices are in common use, but they all differ in subtle and not-so-subtle ways. Since there are so many of them, it is probably worth taking a look at them together to see what the similarities and differences are.

The key to understanding these devices is to realize that they operate in different layers, as illustrated in Fig. 4-45(a). The layer matters because different devices use different pieces of information to decide how to switch. In a typical scenario, the user generates some data to be sent to a remote machine. Those data are passed to the transport layer, which then adds a header (for example, a TCP header) and passes the resulting unit down to the network layer. The network layer adds its own header to form a network layer packet (e.g., an IP packet). In Fig. 4-45(b), we see the IP packet shaded in gray. Then the packet goes to the data link layer, which adds its own header and checksum (CRC) and gives the resulting frame to the physical layer for transmission, for example, over a LAN.

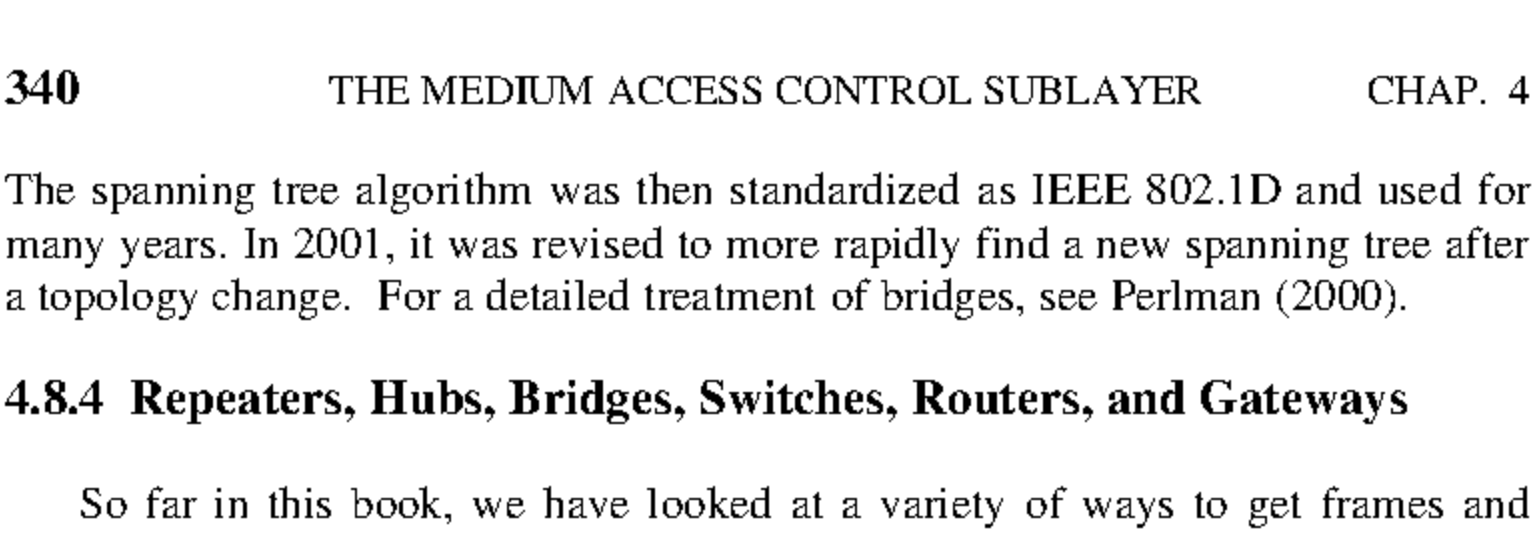


Figure 4-45. (a) Which device is in which layer. (b) Frames, packets, and headers.

Now let us look at the switching devices and see how they relate to the packets and frames. At the bottom, in the physical layer, we find the repeaters. These are analog devices that work with signals on the cables to which they are connected. A signal appearing on one cable is cleaned up, amplified, and put out on another cable. Repeaters do not understand frames, packets, or headers. They understand the symbols that encode bits as volts. Classic Ethernet, for example, was

designed to allow four repeaters that would boost the signal to extend the maximum cable length from 500 meters to 2500 meters.

Next we come to the hubs. A hub has a number of input lines that it joins electrically. Frames arriving on any of the lines are sent out on all the others. If two frames arrive at the same time, they will collide, just as on a coaxial cable. All the lines coming into a hub must operate at the same speed. Hubs differ from repeaters in that they do not (usually) amplify the incoming signals and are designed for multiple input lines, but the differences are slight. Like repeaters, hubs are physical layer devices that do not examine the link layer addresses or use them in any way.

Now let us move up to the data link layer, where we find bridges and switches. We just studied bridges at some length. A bridge connects two or more LANs. Like a hub, a modern bridge has multiple ports, usually enough for 4 to 48 input lines of a certain type. Unlike in a hub, each port is isolated to be its own collision domain; if the port has a full-duplex point-to-point line, the CSMA/CD algorithm is not needed. When a frame arrives, the bridge extracts the destination address from the frame header and looks it up in a table to see where to send the frame. For Ethernet, this address is the 48-bit destination address shown in Fig. 4-14. The bridge only outputs the frame on the port where it is needed and can forward multiple frames at the same time.

Bridges offer much better performance than hubs, and the isolation between bridge ports also means that the input lines may run at different speeds, possibly even with different network types. A common example is a bridge with ports that connect to 10-, 100-, and 1000-Mbps Ethernet. Buffering within the bridge is needed to accept a frame on one port and transmit the frame out on a different port. If frames come in faster than they can be retransmitted, the bridge may run out of buffer space and have to start discarding frames. For example, if a gigabit Ethernet is pouring bits into a 10-Mbps Ethernet at top speed, the bridge will have to buffer them, hoping not to run out of memory. This problem still exists even if all the ports run at the same speed because more than one port may be sending frames to a given destination port.

Bridges were originally intended to be able to join different kinds of LANs, for example, an Ethernet and a Token Ring LAN. However, this never worked well because of differences between the LANs. Different frame formats require copying and reformatting, which takes CPU time, requires a new checksum calculation, and introduces the possibility of undetected errors due to bad bits in the bridge's memory. Different maximum frame lengths are also a serious problem with no good solution. Basically, frames that are too large to be forwarded must be discarded. So much for transparency.

Two other areas where LANs can differ are security and quality of service. Some LANs have link-layer encryption, for example 802.11, and some do not, for example Ethernet. Some LANs have quality of service features such as priorities, for example 802.11, and some do not, for example Ethernet. Consequently, when

a frame must travel between these LANs, the security or quality of service expected by the sender may not be able to be provided. For all of these reasons, modern bridges usually work for one network type, and routers, which we will come to soon, are used instead to join networks of different types.

Switches are modern bridges by another name. The differences are more to do with marketing than technical issues, but there are a few points worth knowing. Bridges were developed when classic Ethernet was in use, so they tend to join relatively few LANs and thus have relatively few ports. The term "switch" is more popular nowadays. Also, modern installations all use point-to-point links, such as twisted-pair cables, so individual computers plug directly into a switch and thus the switch will tend to have many ports. Finally, "switch" is also used as a general term. With a bridge, the functionality is clear. On the other hand, a switch may refer to an Ethernet switch or a completely different kind of device that makes forwarding decisions, such as a telephone switch.

So far, we have seen repeaters and hubs, which are actually quite similar, as well as bridges and switches, which are even more similar to each other. Now we move up to routers, which are different from all of the above. When a packet comes into a router, the frame header and trailer are stripped off and the packet located in the frame's payload field (shaded in Fig. 4-45) is passed to the routing software. This software uses the packet header to choose an output line. For an IP packet, the packet header will contain a 32-bit (IPv4) or 128-bit (IPv6) address, but not a 48-bit IEEE 802 address. The routing software does not see the frame addresses and does not even know whether the packet came in on a LAN or a point-to-point line. We will study routers and routing in Chap. 5.

Up another layer, we find transport gateways. These connect two computers that use different connection-oriented transport protocols. For example, suppose a computer using the connection-oriented TCP/IP protocol needs to talk to a computer using a different connection-oriented transport protocol called SCTP. The transport gateway can copy the packets from one connection to the other, reformatting them as need be.

Finally, application gateways understand the format and contents of the data and can translate messages from one format to another. An email gateway could translate Internet messages into SMS messages for mobile phones, for example. Like "switch," "gateway" is somewhat of a general term. It refers to a forwarding process that runs at a high layer.

4.8.5 Virtual LANs

In the early days of local area networking, thick yellow cables snaked through the cable ducts of many office buildings. Every computer they passed was plugged in. No thought was given to which computer belonged on which LAN. All the people in adjacent offices were put on the same LAN, whether they belonged together or not. Geography trumped corporate organization charts.

With the advent of twisted pair and hubs in the 1990s, all that changed. Buildings were rewired (at considerable expense) to rip out all the yellow garden hoses and install twisted pairs from every office to central wiring closets at the end of each corridor or in a central machine room, as illustrated in Fig. 4-46. If the Vice President in Charge of Wiring was a visionary, Category 5 twisted pairs were installed; if he was a bean counter, the existing (Category 3) telephone wiring was used (only to be replaced a few years later, when fast Ethernet emerged).

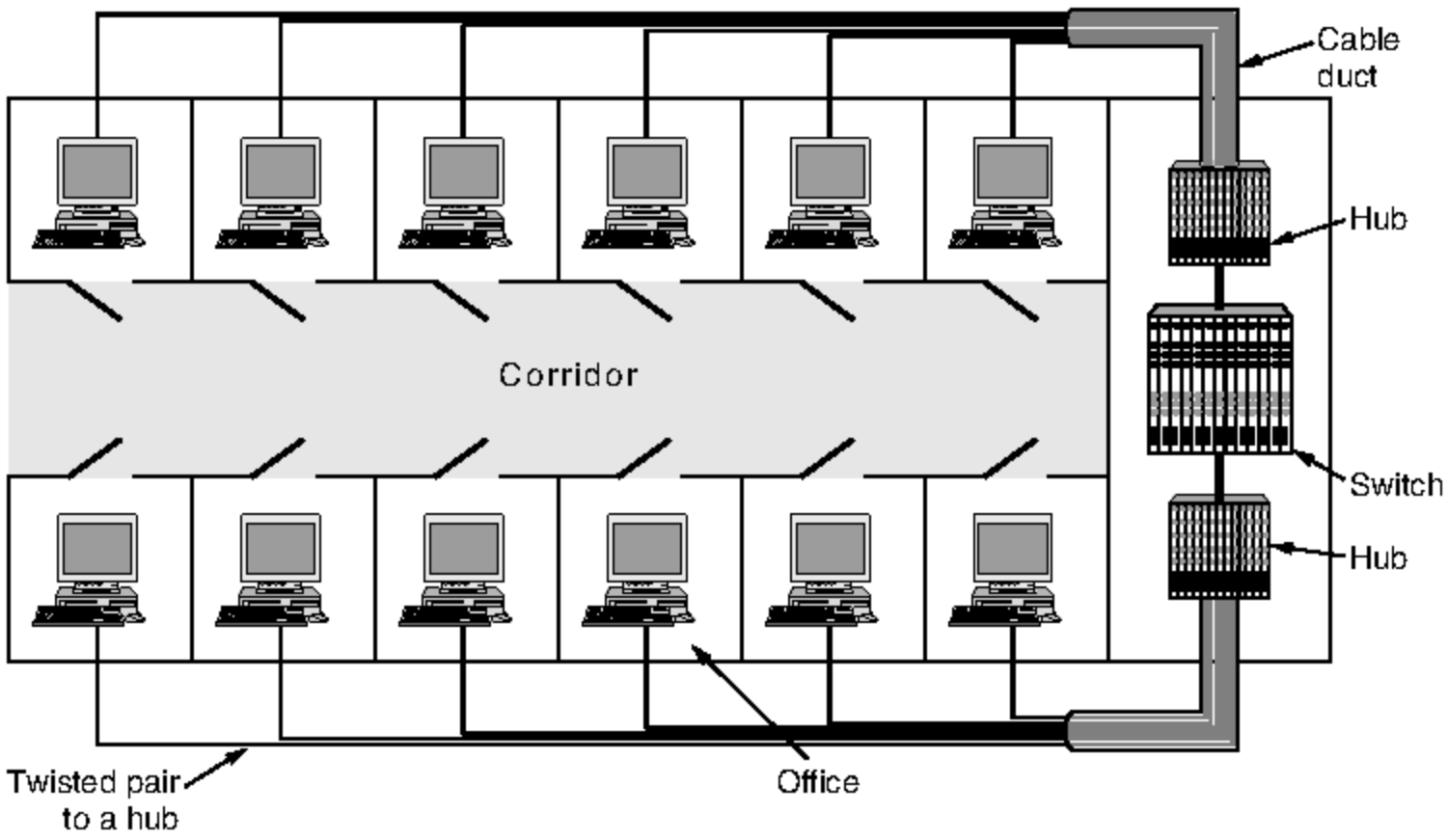


Figure 4-46. A building with centralized wiring using hubs and a switch.

Today, the cables have changed and hubs have become switches, but the wiring pattern is still the same. This pattern makes it possible to configure LANs logically rather than physically. For example, if a company wants k LANs, it could buy k switches. By carefully choosing which connectors to plug into which switches, the occupants of a LAN can be chosen in a way that makes organizational sense, without too much regard to geography.

Does it matter who is on which LAN? After all, in nearly all organizations, all the LANs are interconnected. In short, yes, it often matters. Network administrators like to group users on LANs to reflect the organizational structure rather than the physical layout of the building, for a variety of reasons. One issue is security. One LAN might host Web servers and other computers intended for public use. Another LAN might host computers containing the records of the Human Resources department that are not to be passed outside of the department. In such a situation, putting all the computers on a single LAN and not letting any of the servers be accessed from off the LAN makes sense. Management tends to frown when hearing that such an arrangement is impossible.

A second issue is load. Some LANs are more heavily used than others and it may be desirable to separate them. For example, if the folks in research are running all kinds of nifty experiments that sometimes get out of hand and saturate their LAN, the folks in management may not be enthusiastic about donating some of the capacity they were using for videoconferencing to help out. Then again, this might impress on management the need to install a faster network.

A third issue is broadcast traffic. Bridges broadcast traffic when the location of the destination is unknown, and upper-layer protocols use broadcasting as well. For example, when a user wants to send a packet to an IP address x, how does it know which MAC address to put in the frame? We will study this question in Chap. 5, but briefly summarized, the answer is that it broadcasts a frame containing the question "who owns IP address x?" Then it waits for an answer. As the number of computers in a LAN grows, so does the number of broadcasts. Each broadcast consumes more of the LAN capacity than a regular frame because it is delivered to every computer on the LAN. By keeping LANs no larger than they need to be, the impact of broadcast traffic is reduced.

Related to broadcasts is the problem that once in a while a network interface will break down or be misconfigured and begin generating an endless stream of broadcast frames. If the network is really unlucky, some of these frames will elicit responses that lead to ever more traffic. The result of this **broadcast storm** is that (1) the entire LAN capacity is occupied by these frames, and (2) all the machines on all the interconnected LANs are crippled just processing and discarding all the frames being broadcast.

At first it might appear that broadcast storms could be limited in scope by separating the LANs with bridges or switches, but if the goal is to achieve transparency (i.e., a machine can be moved to a different LAN across the bridge without anyone noticing it), then bridges have to forward broadcast frames.

Having seen why companies might want multiple LANs with restricted scopes, let us get back to the problem of decoupling the logical topology from the physical topology. Building a physical topology to reflect the organizational structure can add work and cost, even with centralized wiring and switches. For example, if two people in the same department work in different buildings, it may be easier to wire them to different switches that are part of different LANs. Even if this is not the case, a user might be shifted within the company from one department to another without changing offices, or might change offices without changing departments. This might result in the user being on the wrong LAN until an administrator changes the user's connector from one switch to another. Furthermore, the number of computers that belong to different departments may not be a good match for the number of ports on switches; some departments may be too small and others so big that they require multiple switches. This results in wasted switch ports that are not used.

In many companies, organizational changes occur all the time, meaning that system administrators spend a lot of time pulling out plugs and pushing them back

in somewhere else. Also, in some cases, the change cannot be made at all because the twisted pair from the user's machine is too far from the correct switch (e.g., in the wrong building), or the available switch ports are on the wrong LAN.

In response to customer requests for more flexibility, network vendors began working on a way to rewire buildings entirely in software. The resulting concept is called a **VLAN** (**Virtual LAN**). It has been standardized by the IEEE 802 committee and is now widely deployed in many organizations. Let us now take a look at it. For additional information about VLANs, see Seifert and Edwards (2008).

VLANs are based on VLAN-aware switches. To set up a VLAN-based network, the network administrator decides how many VLANs there will be, which computers will be on which VLAN, and what the VLANs will be called. Often the VLANs are (informally) named by colors, since it is then possible to print color diagrams showing the physical layout of the machines, with the members of the red LAN in red, members of the green LAN in green, and so on. In this way, both the physical and logical layouts are visible in a single view.

As an example, consider the bridged LAN of Fig. 4-47, in which nine of the machines belong to the G (gray) VLAN and five belong to the W (white) VLAN. Machines from the gray VLAN are spread across two switches, including two machines that connect to a switch via a hub.

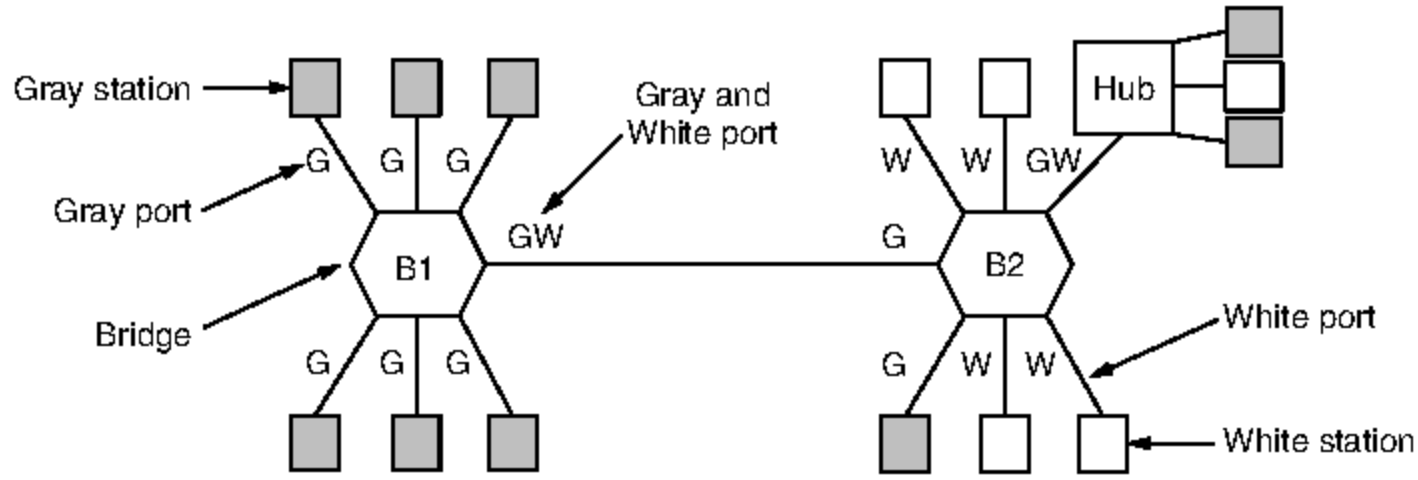


Figure 4-47. Two VLANs, gray and white, on a bridged LAN.

To make the VLANs function correctly, configuration tables have to be set up in the bridges. These tables tell which VLANs are accessible via which ports. When a frame comes in from, say, the gray VLAN, it must be forwarded on all the ports marked with a G. This holds for ordinary (i.e., unicast) traffic for which the bridges have not learned the location of the destination, as well as for multicast and broadcast traffic. Note that a port may be labeled with multiple VLAN colors.

As an example, suppose that one of the gray stations plugged into bridge *B1* in Fig. 4-47 sends a frame to a destination that has not been observed beforehand. Bridge *B1* will receive the frame and see that it came from a machine on the gray

VLAN, so it will flood the frame on all ports labeled G (except the incoming port). The frame will be sent to the five other gray stations attached to *B1* as well as over the link from *B1* to bridge *B2*. At bridge *B2*, the frame is similarly forwarded on all ports labeled G. This sends the frame to one further station and the hub (which will transmit the frame to all of its stations). The hub has both labels because it connects to machines from both VLANs. The frame is not sent on other ports without G in the label because the bridge knows that there are no machines on the gray VLAN that can be reached via these ports.

In our example, the frame is only sent from bridge *B1* to bridge *B2* because there are machines on the gray VLAN that are connected to *B2*. Looking at the white VLAN, we can see that the bridge *B2* port that connects to bridge *B1* is *not* labeled W. This means that a frame on the white VLAN will not be forwarded from bridge *B2* to bridge *B1*. This behavior is correct because no stations on the white VLAN are connected to *B1*.

The IEEE 802.1Q Standard

To implement this scheme, bridges need to know to which VLAN an incoming frame belongs. Without this information, for example, when bridge *B2* gets a frame from bridge *B1* in Fig. 4-47, it cannot know whether to forward the frame on the gray or white VLAN. If we were designing a new type of LAN, it would be easy enough to just add a VLAN field in the header. But what to do about Ethernet, which is the dominant LAN, and did not have any spare fields lying around for the VLAN identifier?

The IEEE 802 committee had this problem thrown into its lap in 1995. After much discussion, it did the unthinkable and changed the Ethernet header. The new format was published in IEEE standard **802.1Q**, issued in 1998. The new format contains a VLAN tag; we will examine it shortly. Not surprisingly, changing something as well established as the Ethernet header was not entirely trivial. A few questions that come to mind are:

1. Need we throw out several hundred million existing Ethernet cards?
2. If not, who generates the new fields?
3. What happens to frames that are already the maximum size?

Of course, the 802 committee was (only too painfully) aware of these problems and had to come up with solutions, which it did.

The key to the solution is to realize that the VLAN fields are only actually used by the bridges and switches and *not* by the user machines. Thus, in Fig. 4-47, it is not really essential that they are present on the lines going out to the end stations as long as they are on the line between the bridges. Also, to use VLANs, the bridges have to be VLAN aware. This fact makes the design feasible.

As to throwing out all existing Ethernet cards, the answer is no. Remember that the 802.3 committee could not even get people to change the *Type* field into a *Length* field. You can imagine the reaction to an announcement that all existing Ethernet cards had to be thrown out. However, new Ethernet cards are 802.1Q compliant and can correctly fill in the VLAN fields.

Because there can be computers (and switches) that are not VLAN aware, the first VLAN-aware bridge to touch a frame adds VLAN fields and the last one down the road removes them. An example of a mixed topology is shown in Fig. 4-48. In this figure, VLAN-aware computers generate tagged (i.e., 802.1Q) frames directly, and further switching uses these tags. The shaded symbols are VLAN-aware and the empty ones are not.

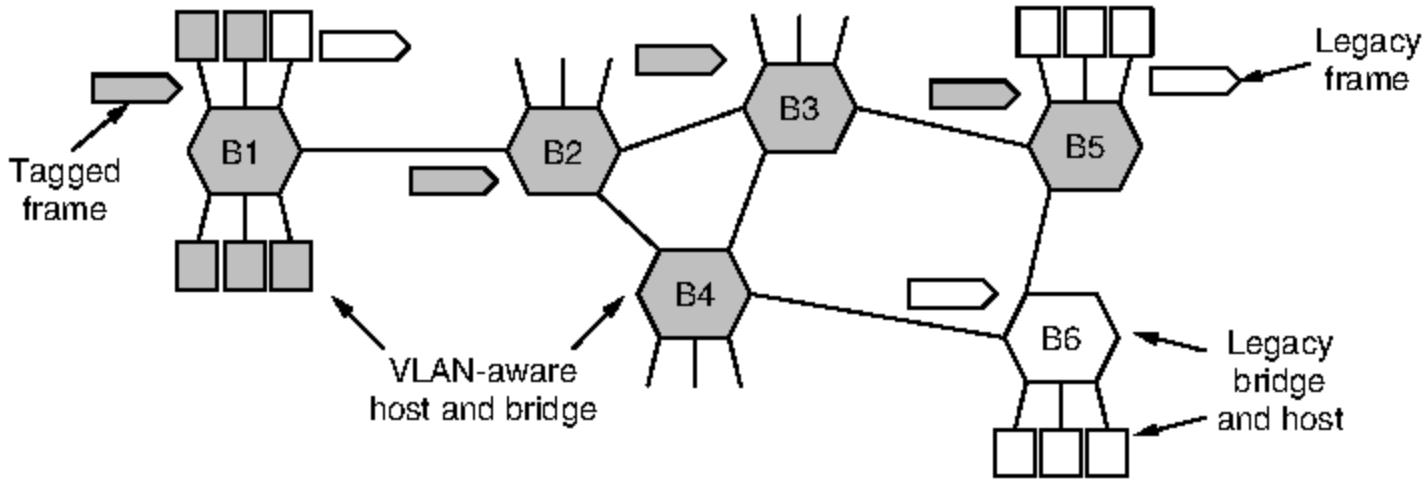


Figure 4-48. Bridged LAN that is only partly VLAN aware. The shaded symbols are VLAN aware. The empty ones are not.

With 802.1Q, frames are colored depending on the port on which they are received. For this method to work, all machines on a port must belong to the same VLAN, which reduces flexibility. For example, in Fig. 4-48, this property holds for all ports where an individual computer connects to a bridge, but not for the port where the hub connects to bridge *B2*.

Additionally, the bridge can use the higher-layer protocol to select the color. In this way, frames arriving on a port might be placed in different VLANs depending on whether they carry IP packets or PPP frames.

Other methods are possible, but they are not supported by 802.1Q. As one example, the MAC address can be used to select the VLAN color. This might be useful for frames coming in from a nearby 802.11 LAN in which laptops send frames via different ports as they move. One MAC address would then be mapped to a fixed VLAN regardless of which port it entered the LAN on.

As to the problem of frames longer than 1518 bytes, 802.1Q just raised the limit to 1522 bytes. Luckily, only VLAN-aware computers and switches must support these longer frames.

Now let us take a look at the 802.1Q frame format. It is shown in Fig. 4-49. The only change is the addition of a pair of 2-byte fields. The first one is the

VLAN protocol ID. It always has the value 0x8100. Since this number is greater than 1500, all Ethernet cards interpret it as a type rather than a length. What a legacy card does with such a frame is moot since such frames are not supposed to be sent to legacy cards.

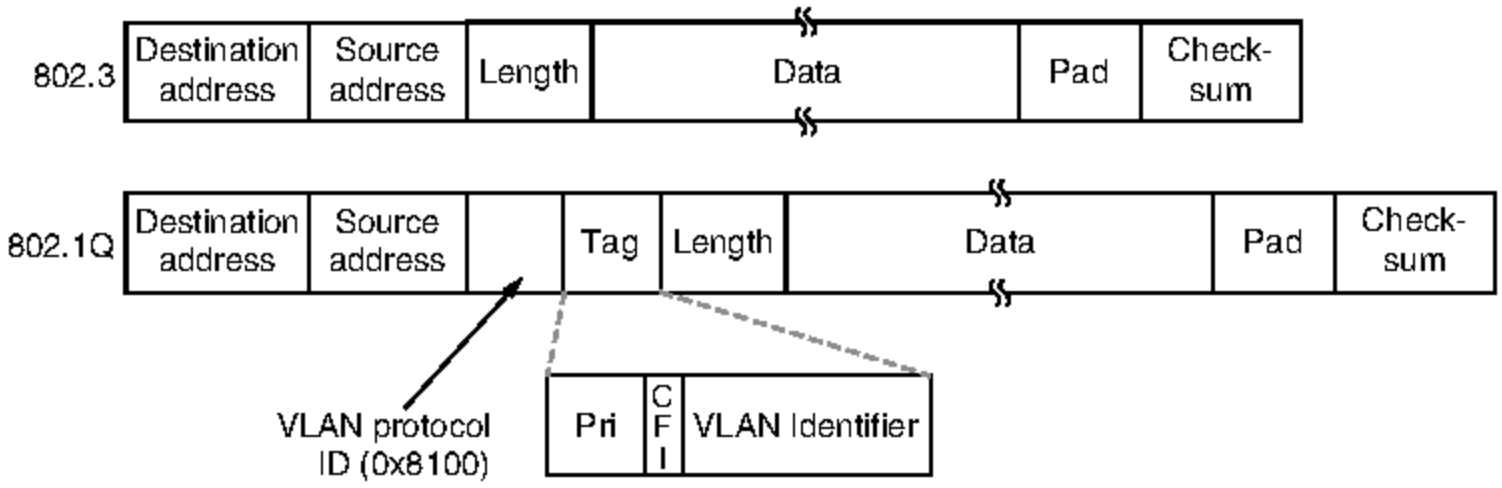


Figure 4-49. The 802.3 (legacy) and 802.1Q Ethernet frame formats.

The second 2-byte field contains three subfields. The main one is the *VLAN identifier*, occupying the low-order 12 bits. This is what the whole thing is about—the color of the VLAN to which the frame belongs. The 3-bit *Priority* field has nothing to do with VLANs at all, but since changing the Ethernet header is a once-in-a-decade event taking three years and featuring a hundred people, why not put in some other good things while you are at it? This field makes it possible to distinguish hard real-time traffic from soft real-time traffic from time-insensitive traffic in order to provide better quality of service over Ethernet. It is needed for voice over Ethernet (although in all fairness, IP has had a similar field for a quarter of a century and nobody ever used it).

The last field, *CFI* (*Canonical format indicator*), should have been called the *CEI* (*Corporate ego indicator*). It was originally intended to indicate the order of the bits in the MAC addresses (little-endian versus big-endian), but that use got lost in other controversies. Its presence now indicates that the payload contains a freeze-dried 802.5 frame that is hoping to find another 802.5 LAN at the destination while being carried by Ethernet in between. This whole arrangement, of course, has nothing whatsoever to do with VLANs. But standards' committee politics are not unlike regular politics: if you vote for my bit, I will vote for your bit.

As we mentioned above, when a tagged frame arrives at a VLAN-aware switch, the switch uses the VLAN identifier as an index into a table to find out which ports to send it on. But where does the table come from? If it is manually constructed, we are back to square zero: manual configuration of bridges. The beauty of the transparent bridge is that it is plug-and-play and does not require any manual configuration. It would be a terrible shame to lose that property. Fortunately, VLAN-aware bridges can also autoconfigure themselves based on observing the tags that come by. If a frame tagged as VLAN 4 comes in on port

3, apparently some machine on port 3 is on VLAN 4. The 802.1Q standard explains how to build the tables dynamically, mostly by referencing appropriate portions of the 802.1D standard.

Before leaving the subject of VLAN routing, it is worth making one last observation. Many people in the Internet and Ethernet worlds are fanatically in favor of connectionless networking and violently opposed to anything smacking of connections in the data link or network layers. Yet VLANs introduce something that is surprisingly similar to a connection. To use VLANs properly, each frame carries a new special identifier that is used as an index into a table inside the switch to look up where the frame is supposed to be sent. That is precisely what happens in connection-oriented networks. In connectionless networks, it is the destination address that is used for routing, not some kind of connection identifier. We will see more of this creeping connectionism in Chap. 5.

4.9 SUMMARY

Some networks have a single channel that is used for all communication. In these networks, the key design issue is the allocation of this channel among the competing stations wishing to use it. FDM and TDM are simple, efficient allocation schemes when the number of stations is small and fixed and the traffic is continuous. Both are widely used under these circumstances, for example, for dividing up the bandwidth on telephone trunks. However, when the number of stations is large and variable or the traffic is fairly bursty—the common case in computer networks—FDM and TDM are poor choices.

Numerous dynamic channel allocation algorithms have been devised. The ALOHA protocol, with and without slotting, is used in many derivatives in real systems, for example, cable modems and RFID. As an improvement when the state of the channel can be sensed, stations can avoid starting a transmission while another station is transmitting. This technique, carrier sensing, has led to a variety of CSMA protocols for LANs and MANs. It is the basis for classic Ethernet and 802.11 networks.

A class of protocols that eliminates contention altogether, or at least reduces it considerably, is well known. The bitmap protocol, topologies such as rings, and the binary countdown protocol completely eliminate contention. The tree walk protocol reduces it by dynamically dividing the stations into two disjoint groups of different sizes and allowing contention only within one group; ideally that group is chosen so that only one station is ready to send when it is permitted to do so.

Wireless LANs have the added problems that it is difficult to sense colliding transmissions, and that the coverage regions of stations may differ. In the dominant wireless LAN, IEEE 802.11, stations use CSMA/CA to mitigate the first problem by leaving small gaps to avoid collisions. The stations can also use the RTS/CTS protocol to combat hidden terminals that arise because of the second

problem. IEEE 802.11 is commonly used to connect laptops and other devices to wireless access points, but it can also be used between devices. Any of several physical layers can be used, including multichannel FDM with and without multiple antennas, and spread spectrum.

Like 802.11, RFID readers and tags use a random access protocol to communicate identifiers. Other wireless PANs and MANs have different designs. The Bluetooth system connects headsets and many kinds of peripherals to computers without wires. IEEE 802.16 provides a wide area wireless Internet data service for stationary and mobile computers. Both of these networks use a centralized, connection-oriented design in which the Bluetooth master and the WiMAX base station decide when each station may send or receive data. For 802.16, this design supports different quality of service for real-time traffic like telephone calls and interactive traffic like Web browsing. For Bluetooth, placing the complexity in the master leads to inexpensive slave devices.

Ethernet is the dominant form of wired LAN. Classic Ethernet used CSMA/CD for channel allocation on a yellow cable the size of a garden hose that snaked from machine to machine. The architecture has changed as speeds have risen from 10 Mbps to 10 Gbps and continue to climb. Now, point-to-point links such as twisted pair are attached to hubs and switches. With modern switches and full-duplex links, there is no contention on the links and the switch can forward frames between different ports in parallel.

With buildings full of LANs, a way is needed to interconnect them all. Plug-and-play bridges are used for this purpose. The bridges are built with a backward learning algorithm and a spanning tree algorithm. Since this functionality is built into modern switches, the terms "bridge" and "switch" are used interchangeably. To help with the management of bridged LANs, VLANs let the physical topology be divided into different logical topologies. The VLAN standard, IEEE 802.1Q, introduces a new format for Ethernet frames.

PROBLEMS

1. For this problem, use a formula from this chapter, but first state the formula. Frames arrive randomly at a 100-Mbps channel for transmission. If the channel is busy when a frame arrives, it waits its turn in a queue. Frame length is exponentially distributed with a mean of 10,000 bits/frame. For each of the following frame arrival rates, give the delay experienced by the average frame, including both queueing time and transmission time.

(a) 90 frames/sec.
(b) 900 frames/sec.
(c) 9000 frames/sec.

2. A group of N stations share a 56-kbps pure ALOHA channel. Each station outputs a 1000-bit frame on average once every 100 sec, even if the previous one has not yet been sent (e.g., the stations can buffer outgoing frames). What is the maximum value of N?

3. Consider the delay of pure ALOHA versus slotted ALOHA at low load. Which one is less? Explain your answer.

4. A large population of ALOHA users manages to generate 50 requests/sec, including both originals and retransmissions. Time is slotted in units of 40 msec.
(a) What is the chance of success on the first attempt?
(b) What is the probability of exactly k collisions and then a success?
(c) What is the expected number of transmission attempts needed?

5. In an infinite-population slotted ALOHA system, the mean number of slots a station waits between a collision and a retransmission is 4. Plot the delay versus throughput curve for this system.

6. What is the length of a contention slot in CSMA/CD for (a) a 2-km twin-lead cable (signal propagation speed is 82% of the signal propagation speed in vacuum)?, and (b) a 40-km multimode fiber optic cable (signal propagation speed is 65% of the signal propagation speed in vacuum)?

7. How long does a station, s, have to wait in the worst case before it can start transmitting its frame over a LAN that uses the basic bit-map protocol?

8. In the binary countdown protocol, explain how a lower-numbered station may be starved from sending a packet.

9. Sixteen stations, numbered 1 through 16, are contending for the use of a shared channel by using the adaptive tree walk protocol. If all the stations whose addresses are prime numbers suddenly become ready at once, how many bit slots are needed to resolve the contention?

10. Consider five wireless stations, A, B, C, D, and E. Station A can communicate with all other stations. B can communicate with A, C and E. C can communicate with A, B and D. D can communicate with A, C and E. E can communicate A, D and B.

(a) When A is sending to B, what other communications are possible?
(b) When B is sending to A, what other communications are possible?
(c) When B is sending to C, what other communications are possible?

11. Six stations, A through F, communicate using the MACA protocol. Is it possible for two transmissions to take place simultaneously? Explain your answer.

12. A seven-story office building has 15 adjacent offices per floor. Each office contains a wall socket for a terminal in the front wall, so the sockets form a rectangular grid in the vertical plane, with a separation of 4 m between sockets, both horizontally and vertically. Assuming that it is feasible to run a straight cable between any pair of sockets, horizontally, vertically, or diagonally, how many meters of cable are needed to connect all sockets using

(a) A star configuration with a single router in the middle?
(b) A classic 802.3 LAN?

13. What is the baud rate of classic 10-Mbps Ethernet?

14. Sketch the Manchester encoding on a classic Ethernet for the bit stream 0001110101.

15. A 1-km-long, 10-Mbps CSMA/CD LAN (not 802.3) has a propagation speed of 200 m/μsec. Repeaters are not allowed in this system. Data frames are 256 bits long, including 32 bits of header, checksum, and other overhead. The first bit slot after a successful transmission is reserved for the receiver to capture the channel in order to send a 32-bit acknowledgement frame. What is the effective data rate, excluding overhead, assuming that there are no collisions?

16. Two CSMA/CD stations are each trying to transmit long (multiframe) files. After each frame is sent, they contend for the channel, using the binary exponential backoff algorithm. What is the probability that the contention ends on round k, and what is the mean number of rounds per contention period?

17. An IP packet to be transmitted by Ethernet is 60 bytes long, including all its headers. If LLC is not in use, is padding needed in the Ethernet frame, and if so, how many bytes?

18. Ethernet frames must be at least 64 bytes long to ensure that the transmitter is still going in the event of a collision at the far end of the cable. Fast Ethernet has the same 64-byte minimum frame size but can get the bits out ten times faster. How is it possible to maintain the same minimum frame size?

19. Some books quote the maximum size of an Ethernet frame as 1522 bytes instead of 1500 bytes. Are they wrong? Explain your answer.

20. How many frames per second can gigabit Ethernet handle? Think carefully and take into account all the relevant cases. *Hint*: the fact that it is *gigabit* Ethernet matters.

21. Name two networks that allow frames to be packed back-to-back. Why is this feature worth having?

22. In Fig. 4-27, four stations, A, B, C, and D, are shown. Which of the last two stations do you think is closest to A and why?

23. Give an example to show that the RTS/CTS in the 802.11 protocol is a little different than in the MACA protocol.

24. A wireless LAN with one AP has 10 client stations. Four stations have data rates of 6 Mbps, four stations have data rates of 18 Mbps, and the last two stations have data rates of 54 Mbps. What is the data rate experienced by each station when all ten stations are sending data together, and

(a) TXOP is not used?
(b) TXOP is used?

25. Suppose that an 11-Mbps 802.11b LAN is transmitting 64-byte frames back-to-back over a radio channel with a bit error rate of 10^{-7}. How many frames per second will be damaged on average?

26. An 802.16 network has a channel width of 20 MHz. How many bits/sec can be sent to a subscriber station?

27. Give two reasons why networks might use an error-correcting code instead of error detection and retransmission.

28. List two ways in which WiMAX is similar to 802.11, and two ways in which it is different from 802.11.

29. From Fig. 4-34, we see that a Bluetooth device can be in two piconets at the same time. Is there any reason why one device cannot be the master in both of them at the same time?

30. What is the maximum size of the data field for a 3-slot Bluetooth frame at basic rate? Explain your answer.

31. Figure 4-24 shows several physical layer protocols. Which of these is closest to the Bluetooth physical layer protocol? What is the biggest difference between the two?

32. It is mentioned in Section 4.6.6 that the efficiency of a 1-slot frame with repetition encoding is about 13% at basic data rate. What will the efficiency be if a 5-slot frame with repetition encoding is used at basic data rate instead?

33. Beacon frames in the frequency hopping spread spectrum variant of 802.11 contain the dwell time. Do you think the analogous beacon frames in Bluetooth also contain the dwell time? Discuss your answer.

34. Suppose that there are 10 RFID tags around an RFID reader. What is the best value of Q? How likely is it that one tag responds with no collision in a given slot?

35. List some of the security concerns of an RFID system.

36. A switch designed for use with fast Ethernet has a backplane that can move 10 Gbps. How many frames/sec can it handle in the worst case?

37. Briefly describe the difference between store-and-forward and cut-through switches.

38. Consider the extended LAN connected using bridges *B1* and *B2* in Fig. 4-41(b). Suppose the hash tables in the two bridges are empty. List all ports on which a packet will be forwarded for the following sequence of data transmissions:

(a) *A* sends a packet to *C*.
(b) *E* sends a packet to *F*.
(c) *F* sends a packet to *E*.
(d) *G* sends a packet to *E*.
(e) *D* sends a packet to *A*.
(f) *B* sends a packet to *F*.

39. Store-and-forward switches have an advantage over cut-through switches with respect to damaged frames. Explain what it is.

40. It is mentioned in Section 4.8.3 that some bridges may not even be present in the spanning tree. Outline a scenario where a bridge may not be present in the spanning tree.

41. To make VLANs work, configuration tables are needed in the bridges. What if the VLANs of Fig. 4-47 used hubs rather than switches? Do the hubs need configuration tables, too? Why or why not?

42. In Fig. 4-48, the switch in the legacy end domain on the right is a VLAN-aware switch. Would it be possible to use a legacy switch there? If so, how would that work? If not, why not?

43. Write a program to simulate the behavior of the CSMA/CD protocol over Ethernet when there are N stations ready to transmit while a frame is being transmitted. Your program should report the times when each station successfully starts sending its frame. Assume that a clock tick occurs once every slot time (51.2 μsec) and a collision detection and sending of a jamming sequence takes one slot time. All frames are the maximum length allowed.

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 wire to the other. Thus, the network layer is the lowest layer that deals with end-to-end transmission.

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

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 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).

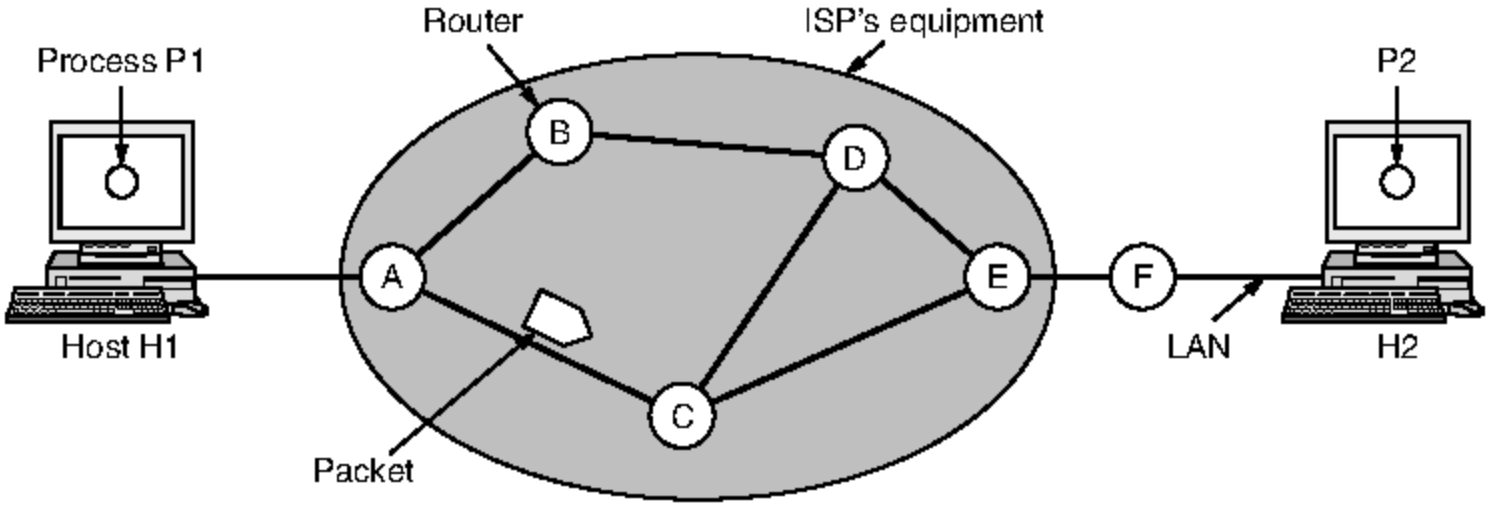


Figure 5-1. The environment of the network layer protocols.

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. 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.

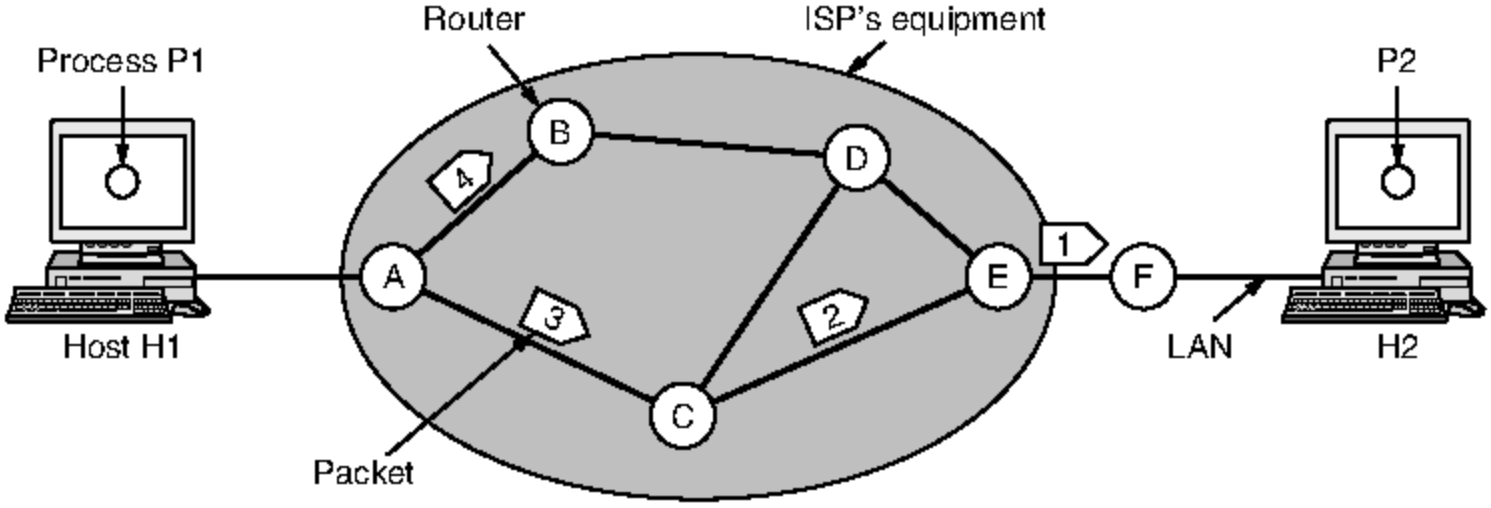
The other camp (represented by the telephone companies) argues that the network should provide a reliable, connection-oriented service. They claim that 100 years of successful experience with the worldwide telephone system is an excellent guide. In this view, quality of service is the dominant factor, 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 MPLS (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 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.



A's table (initially)

Dest.	Line
A	–
B	B
C	C
D	B
E	C
F	C

A's table (later)

A	–
B	B
C	C
D	B
E	B
F	B

C's table

A	A
B	A
C	–
D	E
E	E
F	E

E's table

A	C
B	D
C	C
D	D
E	–
F	F

Figure 5-2. Routing within a datagram network.

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.

However, something different happens to packet 4. When it gets to *A* it is sent to router *B*, even though it is also destined for *F*. For some reason, *A* decided to send packet 4 via a different route than that of the first three packets. Perhaps it has learned of a traffic jam somewhere along the *ACE* path and updated its routing table, as shown under the label "later." The algorithm that manages the tables and makes the routing decisions is called the **routing algorithm**. Routing algorithms are one of the main topics we will study in this chapter. There are several different kinds of them, as we will see.

IP (Internet Protocol), which is the basis for the entire Internet, is the dominant example of a connectionless network service. Each packet carries a destination IP address that routers use to individually forward each packet. The addresses are 32 bits in IPv4 packets and 128 bits in IPv6 packets. We will describe IP in much detail later in this chapter.

5.1.4 Implementation of Connection-Oriented Service

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

As an example, consider the situation shown in Fig. 5-3. Here, host *H1* has established connection 1 with host *H2*. This connection is remembered as the first entry in each of the routing tables. The first line of *A*'s table says that if a packet

bearing connection identifier 1 comes in from *H1*, it is to be sent to router *C* and given connection identifier 1. Similarly, the first entry at *C* routes the packet to *E*, also with connection identifier 1.

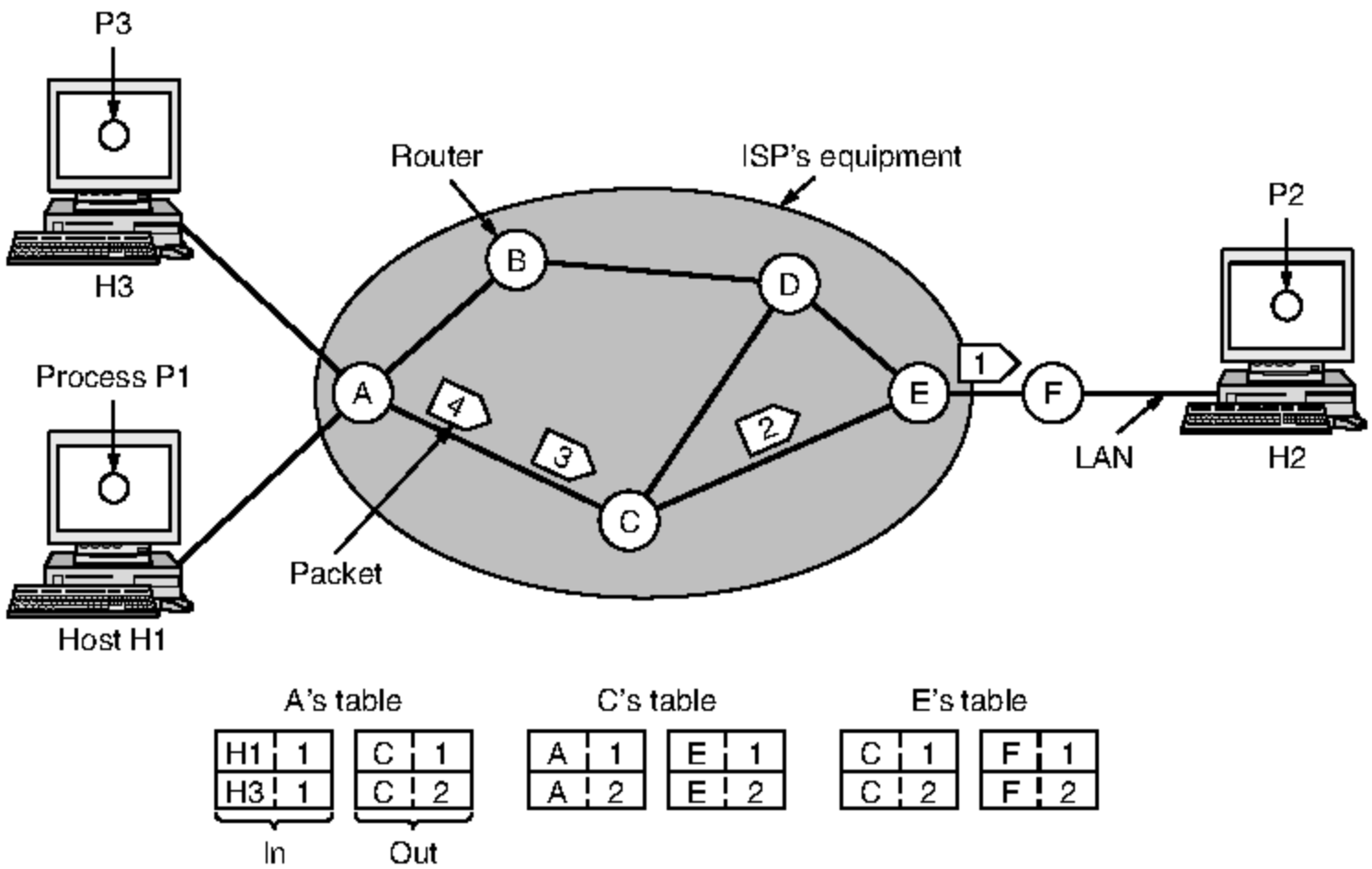


Figure 5-3. Routing within a virtual-circuit network.

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. Note that we have a conflict here because although *A* can easily distinguish connection 1 packets from *H1* from connection 1 packets from *H3*, *C* cannot do this. For this reason, *A* assigns a different connection identifier to the outgoing traffic for the second connection. Avoiding conflicts of this kind is why routers need the ability to replace connection identifiers in outgoing packets.

In some contexts, this process is called **label switching**. An example of a connection-oriented network service is **MPLS** (**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

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

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

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

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

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

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

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

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

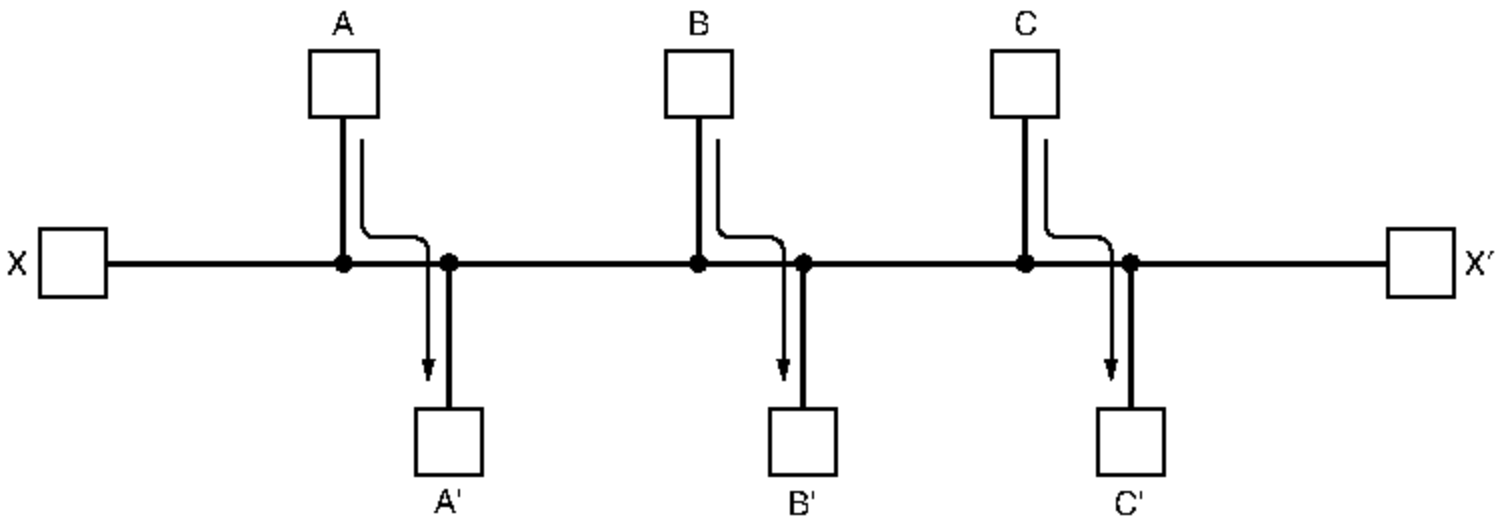


Figure 5-5. Network with a conflict between fairness and efficiency.

measurements or estimates of the current topology and traffic. Instead, the choice of the route to use to get from *I* to *J* (for all *I* and *J*) is computed in advance, off-line, and downloaded to the routers when the network is booted. This procedure is sometimes called **static routing**. Because it does not respond to failures, static routing is mostly useful for situations in which the routing choice is clear. For example, router *F* in Fig. 5-3 should send packets headed into the network to router *E* regardless of the ultimate destination.

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

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

5.2.1 The Optimality Principle

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

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

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

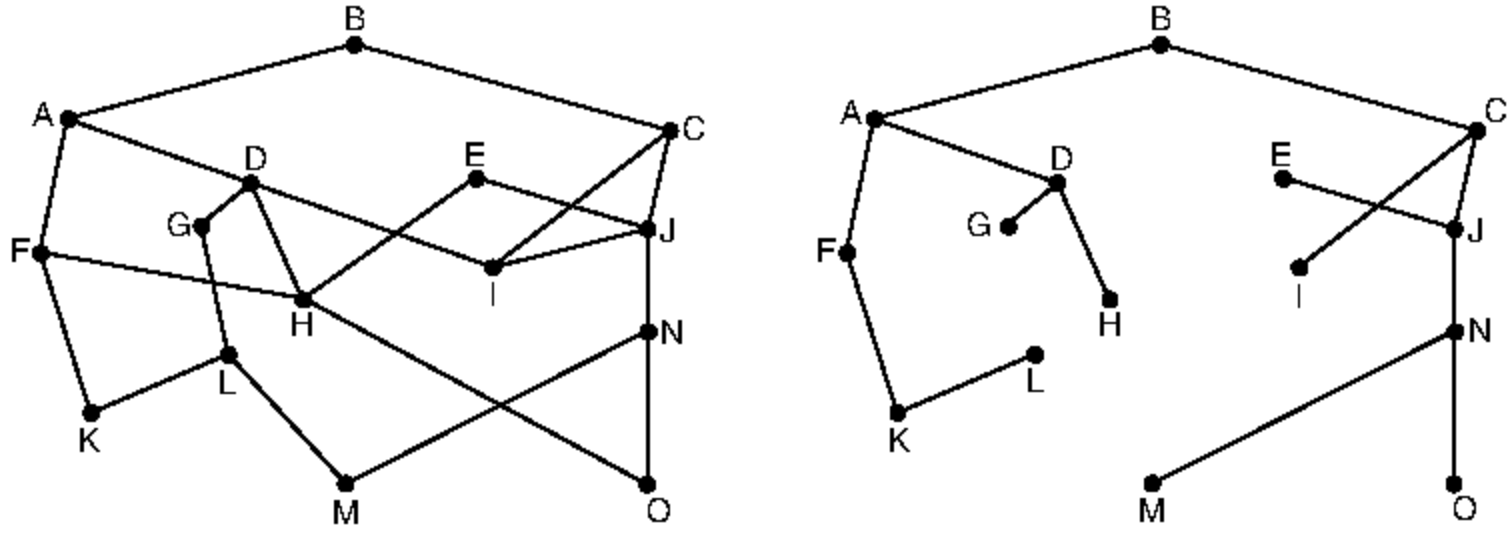


Figure 5-6. (a) A network. (b) A sink tree for router *B*.

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

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

5.2.2 Shortest Path Algorithm

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

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

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

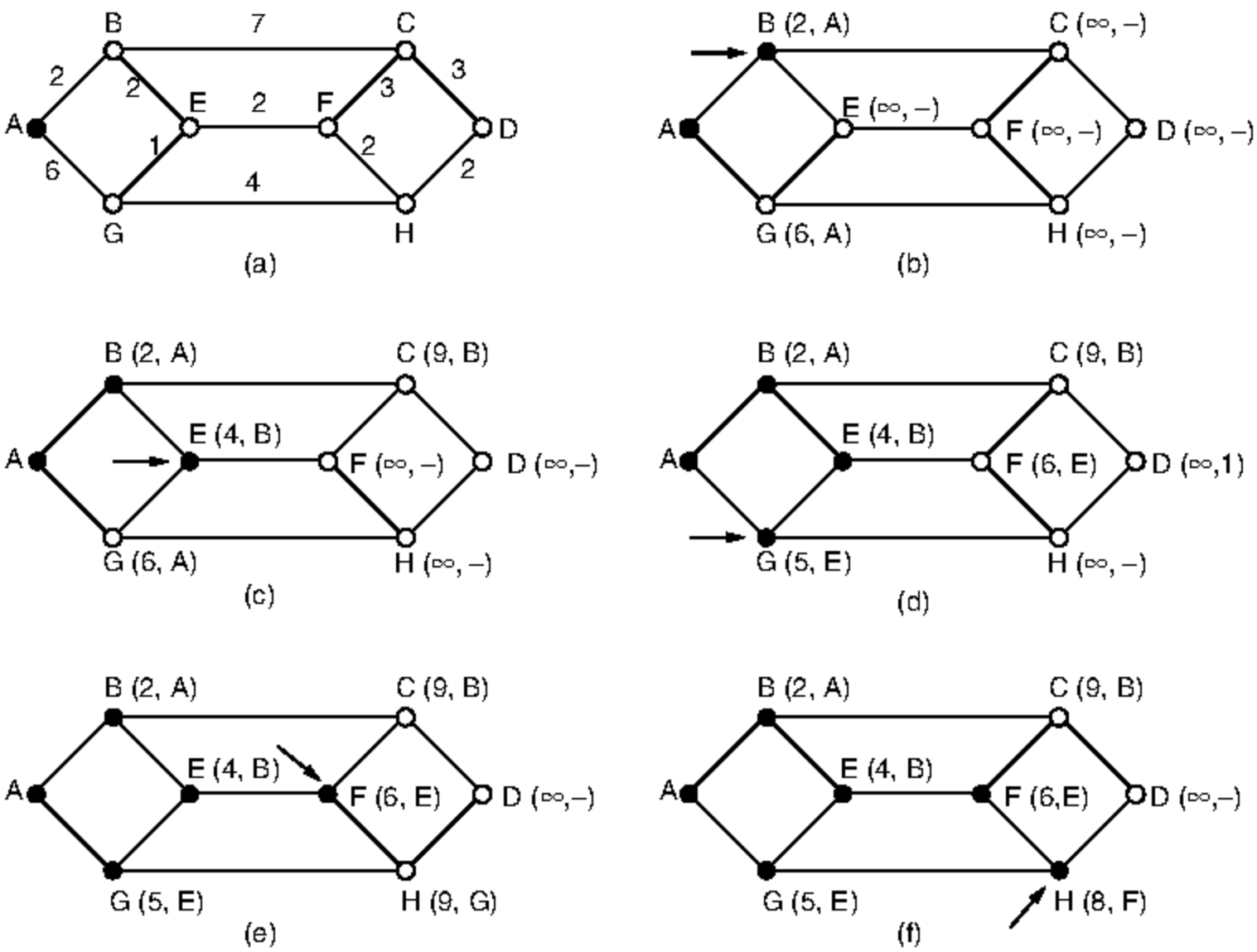


Figure 5-7. The first six steps used in computing the shortest path from *A* to *D*. The arrows indicate the working node.

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

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

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

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

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

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

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

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

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

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

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

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

5.2.3 Flooding

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

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

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

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

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

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

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

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

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

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

To prevent the list from growing without bound, each list should be augmented by a counter, k, meaning that all sequence numbers through k have been seen. When a packet comes in, it is easy to check if the packet has already been

flooded (by comparing its sequence number to k; if so, it is discarded. Furthermore, the full list below k is not needed, since k effectively summarizes it.

Flooding is not practical for sending most packets, but it does have some important uses. First, it ensures that a packet is delivered to every node in the network. This may be wasteful if there is a single destination that needs the packet, but it is effective for broadcasting information. In wireless networks, all messages transmitted by a station can be received by all other stations within its radio range, which is, in fact, flooding, and some algorithms utilize this property.

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

5.2.4 Distance Vector Routing

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

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

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

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

The router is assumed to know the "distance" to each of its neighbors. If the metric is hops, the distance is just one hop. If the metric is propagation delay, the

router can measure it directly with special ECHO packets that the receiver just timestamps and sends back as fast as it can.

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

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

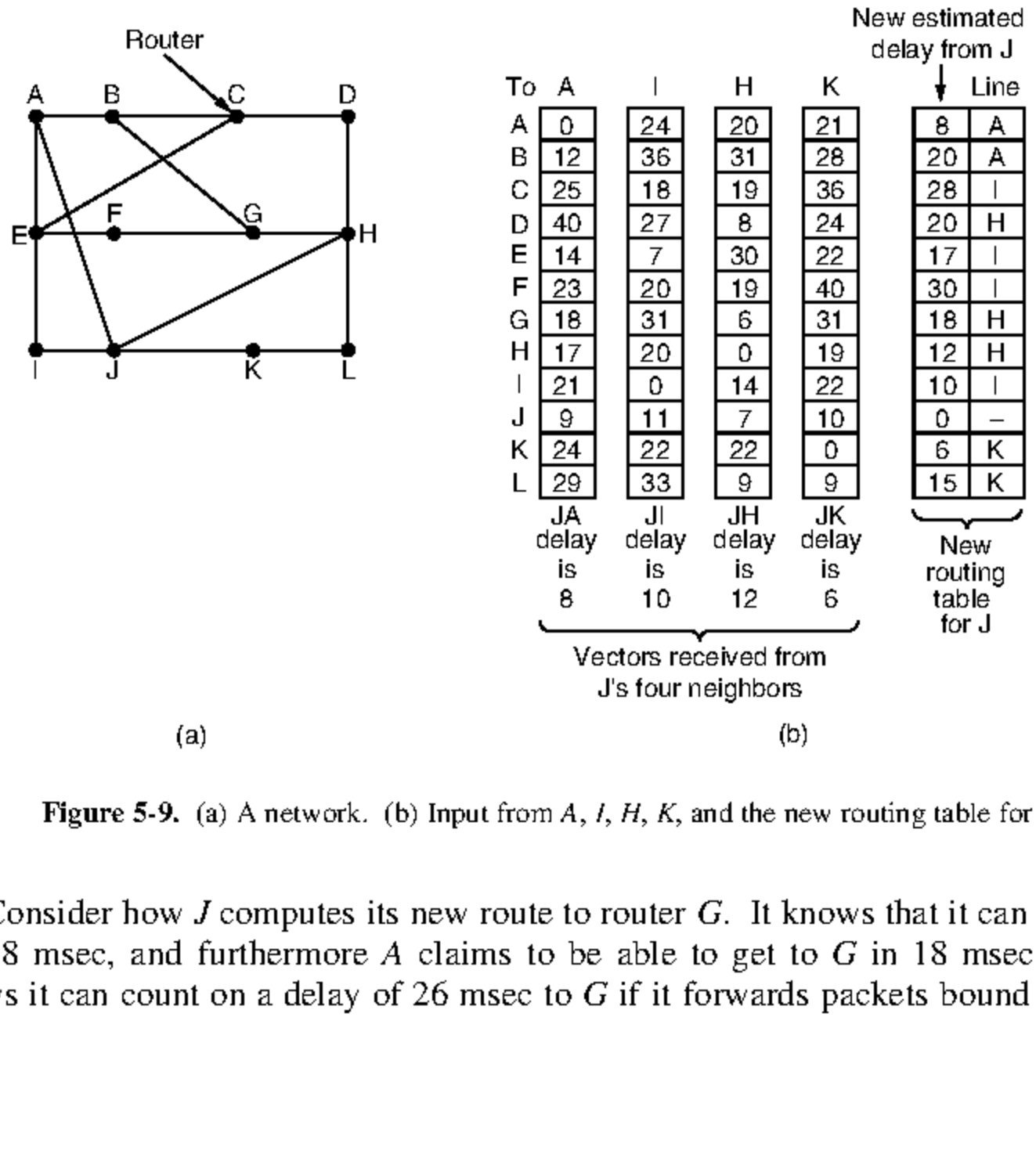


Figure 5-9. (a) A network. (b) Input from A, I, H, K, and the new routing table for J.

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.

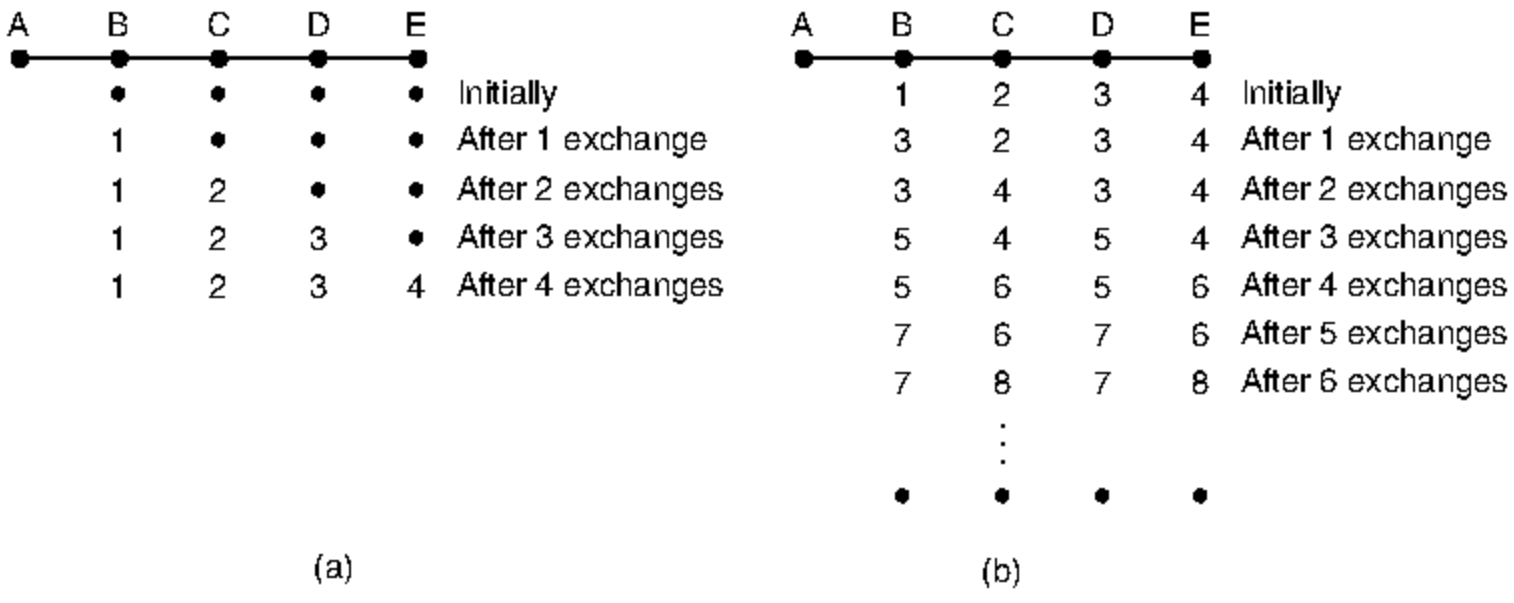


Figure 5-10. The count-to-infinity problem.

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 ten links all with separate paths to *A* of length 2. As a result, *B* thinks it can reach *A* via *C*, with a path length of 3. *D* and *E* do not update their entries for *A* on the first exchange.

On the second exchange, *C* notices that each of its neighbors claims to have a path to *A* of length 3. It picks one of them at random and makes its new distance to *A* 4, as shown in the third row of Fig. 5-10(b). Subsequent exchanges produce the history shown in the rest of Fig. 5-10(b).

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

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

5.2.5 Link State Routing

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

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

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

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

Learning about the Neighbors

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

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

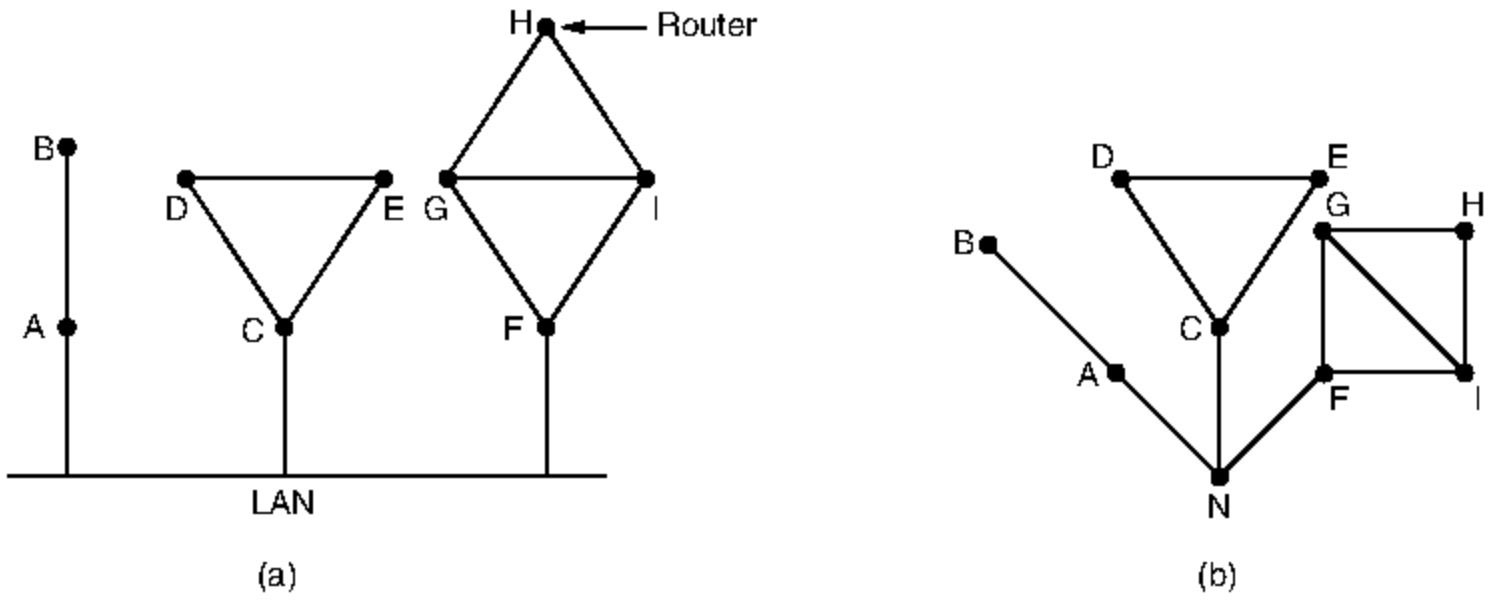


Figure 5-11. (a) Nine routers and a broadcast LAN. (b) A graph model of (a).

The broadcast LAN provides connectivity between each pair of attached routers. However, modeling the LAN as many point-to-point links increases the size

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

Setting Link Costs

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

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

Building Link State Packets

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

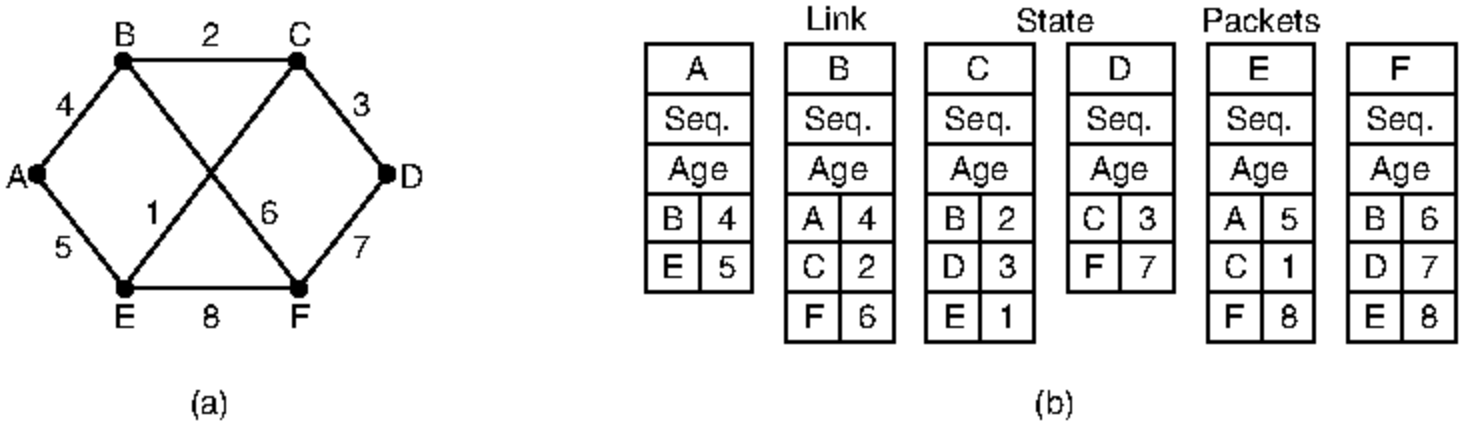


Figure 5-12. (a) A network. (b) The link state packets for this network.

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

Distributing the Link State Packets

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

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

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

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

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

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

Some refinements to this algorithm make it more robust. When a link state packet comes in to a router for flooding, it is not queued for transmission immediately. Instead, it is put in a holding area to wait a short while in case more links are coming up or going down. If another link state packet from the same source comes in before the first packet is transmitted, their sequence numbers are