areas to which it is connected. This summary includes cost information but not all the details of the topology within an area. Passing cost information allows hosts in other areas to find the best area border router to use to enter an area. Not passing topology information reduces traffic and simplifies the shortest-path computations of routers in other areas. However, if there is only one border router out of an area, even the summary does not need to be passed. Routes to destinations out of the area always start with the instruction "Go to the border router." This kind of area is called a **stub area**.

The last kind of router is the **AS boundary router**. It injects routes to external destinations on other ASes into the area. The external routes then appear as destinations that can be reached via the AS boundary router with some cost. An external route can be injected at one or more AS boundary routers. The relationship between ASes, areas, and the various kinds of routers is shown in Fig. 5-65. One router may play multiple roles, for example, a border router is also a backbone router.

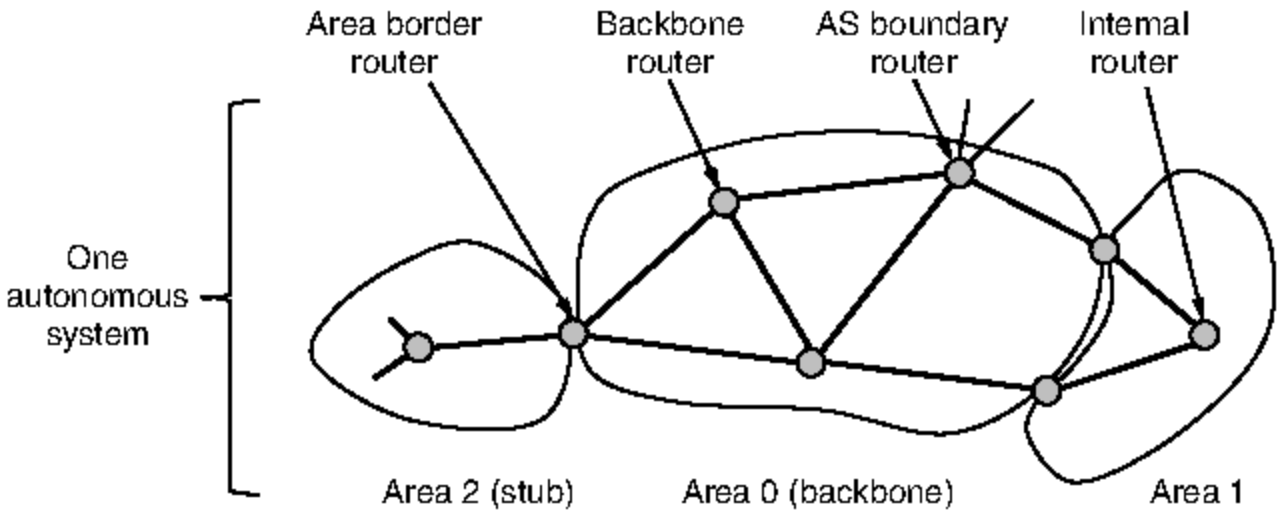


Figure 5-65. The relation between ASes, backbones, and areas in OSPF.

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

For a source and destination in the same area, the best intra-area route (that lies wholly within the area) is chosen. For a source and destination in different areas, the inter-area route must go from the source to the backbone, across the backbone to the destination area, and then to the destination. This algorithm forces a star configuration on OSPF, with the backbone being the hub and the other areas being spokes. Because the route with the lowest cost is chosen, routers in different parts of the network may use different area border routers to enter the backbone and destination area. Packets are routed from source to destination "as is." They are not encapsulated or tunneled (unless going to an area whose

only connection to the backbone is a tunnel). Also, routes to external destinations may include the external cost from the AS boundary router over the external path, if desired, or just the cost internal to the AS.

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

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

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

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

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

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

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

Finally, we can put all the pieces together. Using flooding, each router informs all the other routers in its area of its links to other routers and networks and the cost of these links. This information allows each router to construct the graph for its area(s) and compute the shortest paths. The backbone area does this work, too. In addition, the backbone routers accept information from the area border routers in order to compute the best route from each backbone router to every other router. This information is propagated back to the area border routers, which advertise it within their areas. Using this information, internal routers can select the best route to a destination outside their area, including the best exit router to the backbone.

5.6.7 BGP—The Exterior Gateway Routing Protocol

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

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

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

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

As you might imagine from this list, routing policies can be highly individual. They are often proprietary because they contain sensitive business information.

However, we can describe some patterns that capture the reasoning of the company above and that are often used as a starting point.

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

We can see an example of transit service in Fig. 5-67. There are four ASes that are connected. The connection is often made with a link at **IXPs** (**Internet eXchange Points**), facilities to which many ISPs have a link for the purpose of connecting with other ISPs. *AS2*, *AS3*, and *AS4* are customers of *AS1*. They buy transit service from it. Thus, when source *A* sends to destination *C*, the packets travel from *AS2* to *AS1* and finally to *AS4*. The routing advertisements travel in the opposite direction to the packets. *AS4* advertises *C* as a destination to its transit provider, *AS1*, to let sources reach *C* via *AS1*. Later, *AS1* advertises a route to *C* to its other customers, including *AS2*, to let the customers know that they can send traffic to *C* via *AS1*.

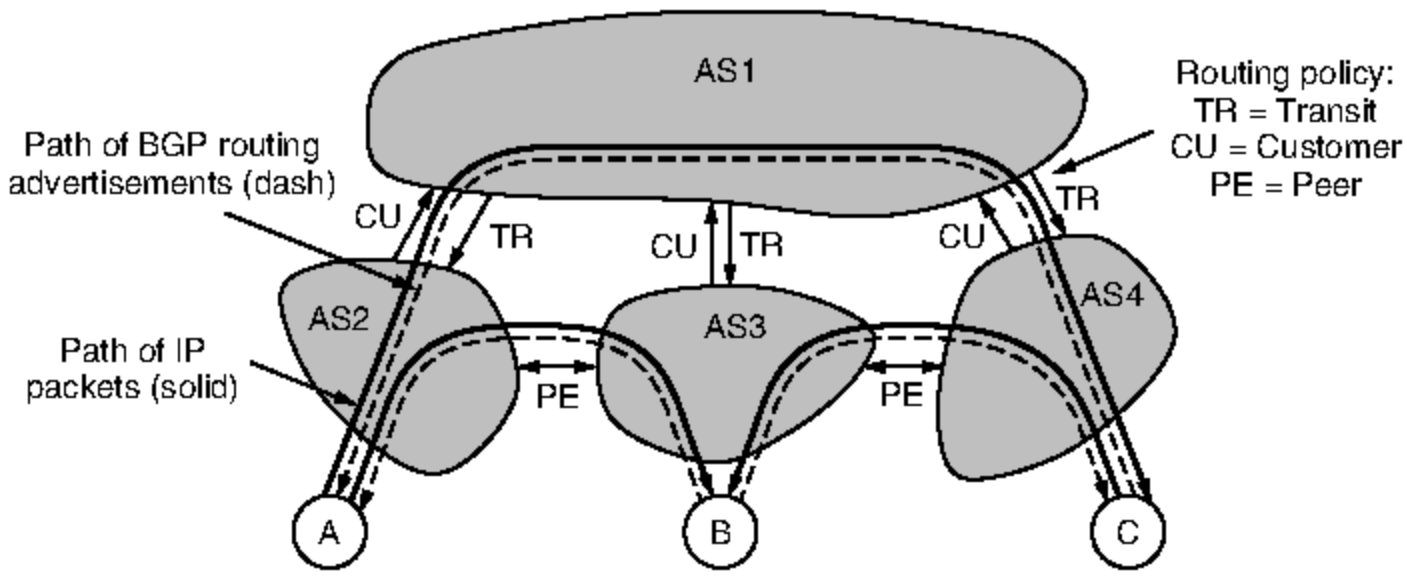


Figure 5-67. Routing policies between four autonomous systems.

In Fig. 5-67, all of the other ASes buy transit service from *AS1*. This provides them with connectivity so they can interact with any host on the Internet. However, they have to pay for this privilege. Suppose that *AS2* and *AS3* exchange a lot of traffic. Given that their networks are connected already, if they want to, they

can use a different policy—they can send traffic directly to each other for free. This will reduce the amount of traffic they must have *AS1* deliver on their behalf, and hopefully it will reduce their bills. This policy is called **peering**.

To implement peering, two ASes send routing advertisements to each other for the addresses that reside in their networks. Doing so makes it possible for *AS2* to send *AS3* packets from *A* destined to *B* and vice versa. However, note that peering is not transitive. In Fig. 5-67, *AS3* and *AS4* also peer with each other. This peering allows traffic from *C* destined for *B* to be sent directly to *AS4*. What happens if *C* sends a packet to *A*? *AS3* is only advertising a route to *B* to *AS4*. It is not advertising a route to *A*. The consequence is that traffic will not pass from *AS4* to *AS3* to *AS2*, even though a physical path exists. This restriction is exactly what *AS3* wants. It peers with *AS4* to exchange traffic, but does not want to carry traffic from *AS4* to other parts of the Internet since it is not being paid to so do. Instead, *AS4* gets transit service from *AS1*. Thus, it is *AS1* who will carry the packet from *C* to *A*.

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

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

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

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

with each other by establishing TCP connections. Operating this way provides reliable communication and also hides all the details of the network being passed through.

An example of how BGP routes are advertised is shown in Fig. 5-68. There are three ASes and the middle one is providing transit to the left and right ISPs. A route advertisement to prefix *C* starts in *AS3*. When it is propagated across the link to *R2c* at the top of the figure, it has the AS path of simply *AS3* and the next hop router of *R3a*. At the bottom, it has the same AS path but a different next hop because it came across a different link. This advertisement continues to propagate and crosses the boundary into *AS1*. At router *R1a*, at the top of the figure, the AS path is *AS2*, *AS3* and the next hop is *R2a*.

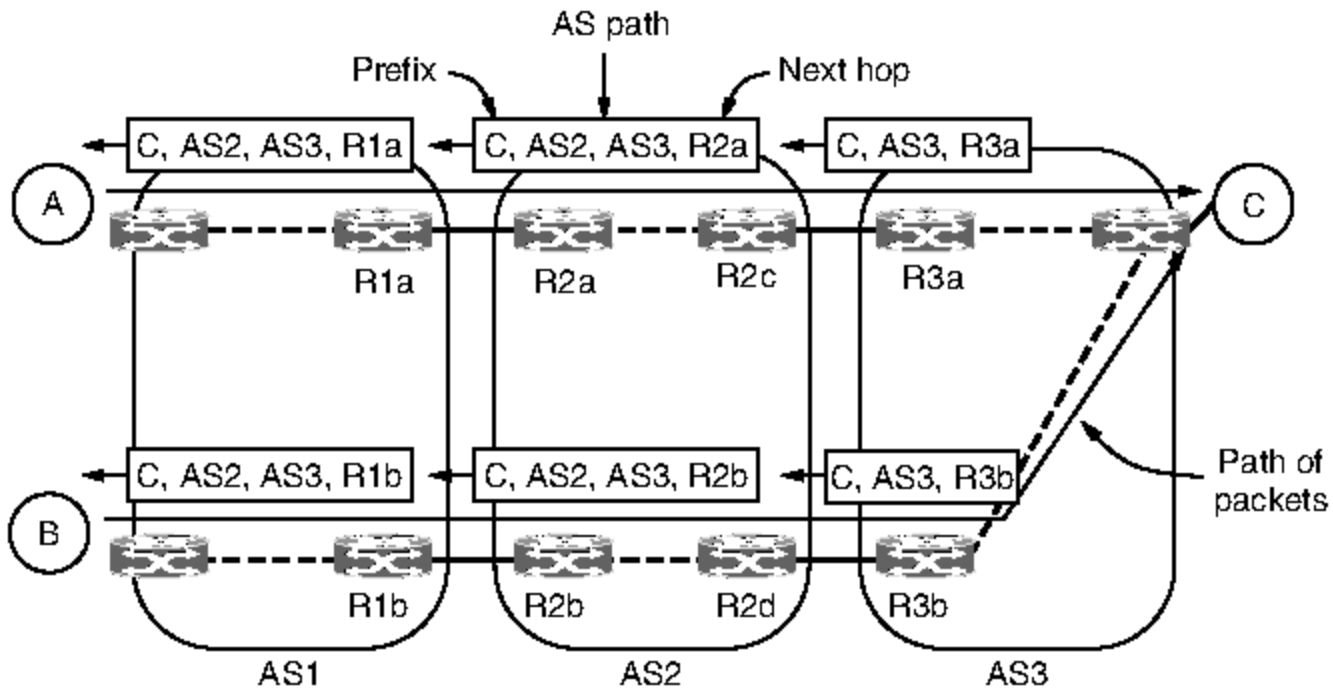


Figure 5-68. Propagation of BGP route advertisements.

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

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

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

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

We have not shown this propagation in Fig. 5-68 to avoid clutter, but, for example, router *R2b* will know that it can reach *C* via either router *R2c* at top or router *R2d* at bottom. The next hop is updated as the route crosses within the ISP so that routers on the far side of the ISP know which router to use to exit the ISP on the other side. This can be seen in the leftmost routes in which the next hop points to a router in the same ISP and not a router in the next ISP.

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

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

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

The final strategy is to prefer the route that has the lowest cost within the ISP. This is the strategy implemented in Fig. 5-68. Packets sent from *A* to *C* exit *AS1* at the top router, *R1a*. Packets sent from *B* exit via the bottom router, *R1b*. The reason is that both *A* and *B* are taking the lowest-cost path or quickest route out of *AS1*. Because they are located in different parts of the ISP, the quickest exit for each one is different. The same thing happens as the packets pass through *AS2*. On the last leg, *AS3* has to carry the packet from *B* through its own network.

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

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

Amazingly, we have only scratched the surface of BGP. For more information, see the BGP version 4 specification in RFC 4271 and related RFCs. However, realize that much of its complexity lies with policies, which are not described in the specification of the BGP protocol.

5.6.8 Internet Multicasting

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

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

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

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

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

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

5.6.9 Mobile IP

Many users of the Internet have mobile computers and want to stay connected when they are away from home and even on the road in between. Unfortunately, the IP addressing system makes working far from home easier said than done, as we will describe shortly. When people began demanding the ability anyway, IETF set up a Working Group to find a solution. The Working Group quickly formulated a number of goals considered desirable in any solution. The major ones were:

1. Each mobile host must be able to use its home IP address anywhere.
2. Software changes to the fixed hosts were not permitted.
3. Changes to the router software and tables were not permitted.
4. Most packets for mobile hosts should not make detours on the way.
5. No overhead should be incurred when a mobile host is at home.

The solution chosen was the one described in Sec. 5.2.10. In brief, every site that wants to allow its users to roam has to create a helper at the site called a **home agent**. When a mobile host shows up at a foreign site, it obtains a new IP address (called a care-of address) at the foreign site. The mobile then tells the home agent where it is now by giving it the care-of address. When a packet for the mobile arrives at the home site and the mobile is elsewhere, the home agent grabs the packet and tunnels it to the mobile at the current care-of address. The mobile can send reply packets directly to whoever it is communicating with, but still using its home address as the source address. This solution meets all the requirements stated above except that packets for mobile hosts do make detours.

Now that we have covered the network layer of the Internet, we can go into the solution in more detail. The need for mobility support in the first place comes from the IP addressing scheme itself. Every IP address contains a network number and a host number. For example, consider the machine with IP address 160.80.40.20/16. The 160.80 gives the network number; the 40.20 is the host number. Routers all over the world have routing tables telling which link to use to get to network 160.80. Whenever a packet comes in with a destination IP address of the form 160.80.xxx.yyy, it goes out on that line. If all of a sudden, the machine with that address is carted off to some distant site, the packets for it will continue to be routed to its home LAN (or router).

At this stage, there are two options—both unattractive. The first is that we could create a route to a more specific prefix. That is, if the distant site advertises a route to 160.80.40.20/32, packets sent to the destination will start arriving in the right place again. This option depends on the longest matching prefix algorithm that is used at routers. However, we have added a route to an IP prefix with a single IP address in it. All ISPs in the world will learn about this prefix. If everyone changes global IP routes in this way when they move their computer, each router would have millions of table entries, at astronomical cost to the Internet. This option is not workable.

The second option is to change the IP address of the mobile. True, packets sent to the home IP address will no longer be delivered until all the relevant people, programs, and databases are informed of the change. But the mobile can still use the Internet at the new location to browse the Web and run other applications. This option handles mobility at a higher layer. It is what typically happens when a user takes a laptop to a coffee store and uses the Internet via the local wireless network. The disadvantage is that it breaks some applications, and it does not keep connectivity as the mobile moves around.

As an aside, mobility can also be handled at a lower layer, the link layer. This is what happens when using a laptop on a single 802.11 wireless network. The IP address of the mobile does not change and the network path remains the same. It is the wireless link that is providing mobility. However, the degree of mobility is limited. If the laptop moves too far, it will have to connect to the Internet via another network with a different IP address.

The mobile IP solution for IPv4 is given in RFC 3344. It works with the existing Internet routing and allows hosts to stay connected with their own IP addresses as they move about. For it to work, the mobile must be able to discover when it has moved. This is accomplished with ICMP router advertisement and solicitation messages. Mobiles listen for periodic router advertisements or send a solicitation to discover the nearest router. If this router is not the usual address of the router when the mobile is at home, it must be on a foreign network. If this router has changed since last time, the mobile has moved to another foreign network. This same mechanism lets mobile hosts find their home agents.

To get a care-of IP address on the foreign network, a mobile can simply use DHCP. Alternatively, if IPv4 addresses are in short supply, the mobile can send and receive packets via a foreign agent that already has an IP address on the network. The mobile host finds a foreign agent using the same ICMP mechanism used to find the home agent. After the mobile obtains an IP address or finds a foreign agent, it is able to use the network to send a message to its home agent, informing the home agent of its current location.

The home agent needs a way to intercept packets sent to the mobile only when the mobile is not at home. ARP provides a convenient mechanism. To send a packet over an Ethernet to an IP host, the router needs to know the Ethernet address of the host. The usual mechanism is for the router to send an ARP query to ask, for example, what is the Ethernet address of 160.80.40.20. When the mobile is at home, it answers ARP queries for its IP address with its own Ethernet address. When the mobile is away, the home agent responds to this query by giving its Ethernet address. The router then sends packets for 160.80.40.20 to the home agent. Recall that this is called a proxy ARP.

To quickly update ARP mappings back and forth when the mobile leaves home or arrives back home, another ARP technique called a **gratuitous ARP** can be used. Basically, the mobile or home agent send themselves an ARP query for the mobile IP address that supplies the right answer so that the router notices and updates its mapping.

Tunneling to send a packet between the home agent and the mobile host at the care-of address is done by encapsulating the packet with another IP header destined for the care-of address. When the encapsulated packet arrives at the care-of address, the outer IP header is removed to reveal the packet.

As with many Internet protocols, the devil is in the details, and most often the details of compatibility with other protocols that are deployed. There are two complications. First, NAT boxes depend on peeking past the IP header to look at the TCP or UDP header. The original form of tunneling for mobile IP did not use these headers, so it did not work with NAT boxes. The solution was to change the encapsulation to include a UDP header.

The second complication is that some ISPs check the source IP addresses of packets to see that they match where the routing protocol believes the source should be located. This technique is called **ingress filtering**, and it is a security

measure intended to discard traffic with seemingly incorrect addresses that may be malicious. However, packets sent from the mobile to other Internet hosts when it is on a foreign network will have a source IP address that is out of place, so they will be discarded. To get around this problem, the mobile can use the care-of address as a source to tunnel the packets back to the home agent. From here, they are sent into the Internet from what appears to be the right location. The cost is that the route is more roundabout.

Another issue we have not discussed is security. When a home agent gets a message asking it to please forward all of Roberta's packets to some IP address, it had better not comply unless it is convinced that Roberta is the source of this request, and not somebody trying to impersonate her. Cryptographic authentication protocols are used for this purpose. We will study such protocols in Chap. 8.

Mobility protocols for IPv6 build on the IPv4 foundation. The scheme above suffers from the triangle routing problem in which packets sent to the mobile take a dogleg through a distant home agent. In IPv6, route optimization is used to follow a direct path between the mobile and other IP addresses after the initial packets have followed the long route. Mobile IPv6 is defined in RFC 3775.

There is another kind of mobility that is also being defined for the Internet. Some airplanes have built-in wireless networking that passengers can use to connect their laptops to the Internet. The plane has a router that connects to the rest of the Internet via a wireless link. (Did you expect a wired link?) So now we have a flying router, which means that the whole network is mobile. Network mobility designs support this situation without the laptops realizing that the plane is mobile. As far as they are concerned, it is just another network. Of course, some of the laptops may be using mobile IP to keep their home addresses while they are on the plane, so we have two levels of mobility. Network mobility is defined for IPv6 in RFC 3963.

5.7 SUMMARY

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

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

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

The next step beyond just dealing with congestion is to actually try to achieve a promised quality of service. Some applications care more about throughput whereas others care more about delay and jitter. The methods that can be used to provide different qualities of service include a combination of traffic shaping, reserving resources at routers, and admission control. Approaches that have been designed for good quality of service include IETF integrated services (including RSVP) and differentiated services.

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

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

PROBLEMS

1. Give two example computer applications for which connection-oriented service is appropriate. Now give two examples for which connectionless service is best.
2. Datagram networks route each packet as a separate unit, independent of all others. Virtual-circuit networks do not have to do this, since each data packet follows a predetermined route. Does this observation mean that virtual-circuit networks do not need the capability to route isolated packets from an arbitrary source to an arbitrary destination? Explain your answer.
3. Give three examples of protocol parameters that might be negotiated when a connection is set up.
4. Assuming that all routers and hosts are working properly and that all software in both is free of all errors, is there any chance, however small, that a packet will be delivered to the wrong destination?

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

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

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

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

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

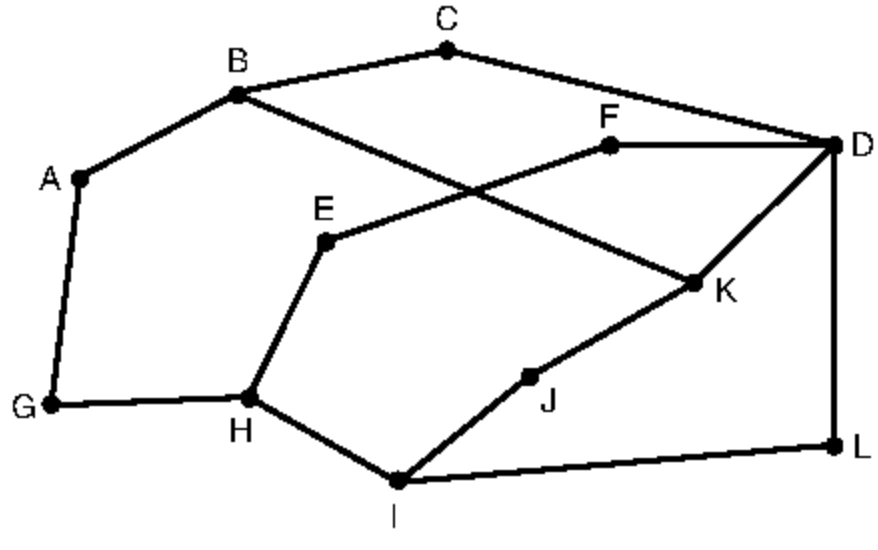
10. In the text it was stated that when a mobile host is not at home, packets sent to its home LAN are intercepted by its home agent on that LAN. For an IP network on an 802.3 LAN, how does the home agent accomplish this interception?

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

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

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

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



14. Suppose that node *B* in Fig. 5-20 has just rebooted and has no routing information in its tables. It suddenly needs a route to *H*. It sends out broadcasts with *TtL* set to 1, 2, 3, and so on. How many rounds does it take to find a route?

15. As a possible congestion control mechanism in a network using virtual circuits internally, a router could refrain from acknowledging a received packet until (1) it knows its last transmission along the virtual circuit was received successfully and (2) it has a free buffer. For simplicity, assume that the routers use a stop-and-wait protocol and that each virtual circuit has one buffer dedicated to it for each direction of traffic. If it takes *T* sec to transmit a packet (data or acknowledgement) and there are *n* routers on the path, what is the rate at which packets are delivered to the destination host? Assume that transmission errors are rare and that the host-router connection is infinitely fast.

16. A datagram network allows routers to drop packets whenever they need to. The probability of a router discarding a packet is *p*. Consider the case of a source host connected to the source router, which is connected to the destination router, and then to the destination host. If either of the routers discards a packet, the source host eventually times out and tries again. If both host-router and router-router lines are counted as hops, what is the mean number of

(a) hops a packet makes per transmission?
(b) transmissions a packet makes?
(c) hops required per received packet?

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

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

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

20. The network of Fig. 5-34 uses RSVP with multicast trees for hosts 1 and 2 as shown. Suppose that host 3 requests a channel of bandwidth 2 MB/sec for a flow from host 1 and another channel of bandwidth 1 MB/sec for a flow from host 2. At the same time, host 4 requests a channel of bandwidth 2 MB/sec for a flow from host 1 and host 5 requests a channel of bandwidth 1 MB/sec for a flow from host 2. How much total bandwidth will be reserved for these requests at routers *A*, *B*, *C*, *E*, *H*, *J*, *K*, and *L*?

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

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

23. Suppose that host *A* is connected to a router *R*1, *R*1 is connected to another router, *R*2, and *R*2 is connected to host *B*. Suppose that a TCP message that contains 900 bytes of data and 20 bytes of TCP header is passed to the IP code at host *A* for delivery to *B*. Show the *Total length*, *Identification*, *DF*, *MF*, and *Fragment offset* fields of the IP header in each packet transmitted over the three links. Assume that link *A-R1* can support a maximum frame size of 1024 bytes including a 14-byte frame header, link *R1-R2* can support a maximum frame size of 512 bytes, including an 8-byte frame header, and link *R2-B* can support a maximum frame size of 512 bytes including a 12-byte frame header.

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

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

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

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

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

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

30. A large number of consecutive IP addresses are available starting at 198.16.0.0. Suppose that four organizations, *A*, *B*, *C*, and *D*, request 4000, 2000, 4000, and 8000 addresses, respectively, and in that order. For each of these, give the first IP address assigned, the last IP address assigned, and the mask in the *w.x.y.z/s* notation.

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

32. The set of IP addresses from 29.18.0.0 to 19.18.128.255 has been aggregated to 29.18.0.0/17. However, there is a gap of 1024 unassigned addresses from 29.18.60.0 to 29.18.63.255 that are now suddenly assigned to a host using a different outgoing line. Is it now necessary to split up the aggregate address into its constituent blocks, add the new block to the table, and then see if any reaggregation is possible? If not, what can be done instead?

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

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

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

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

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

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

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

37. Most IP datagram reassembly algorithms have a timer to avoid having a lost fragment tie up reassembly buffers forever. Suppose that a datagram is fragmented into four fragments. The first three fragments arrive, but the last one is delayed. Eventually, the timer goes off and the three fragments in the receiver's memory are discarded. A little later, the last fragment stumbles in. What should be done with it?

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

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

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

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

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

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

44. Write a program that simulates a computer network using discrete time. The first packet on each router queue makes one hop per time interval. Each router has only a finite number of buffers. If a packet arrives and there is no room for it, it is discarded

and not retransmitted. Instead, there is an end-to-end protocol, complete with time-outs and acknowledgement packets, that eventually regenerates the packet from the source router. Plot the throughput of the network as a function of the end-to-end time-out interval, parameterized by error rate.

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

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

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

6

THE TRANSPORT LAYER

Together with the network layer, the transport layer is the heart of the protocol hierarchy. The network layer provides end-to-end packet delivery using datagrams or virtual circuits. The transport layer builds on the network layer to provide data transport from a process on a source machine to a process on a destination machine with a desired level of reliability that is independent of the physical networks currently in use. It provides the abstractions that applications need to use the network. Without the transport layer, the whole concept of layered protocols would make little sense. In this chapter, we will study the transport layer in detail, including its services and choice of API design to tackle issues of reliability, connections and congestion control, protocols such as TCP and UDP, and performance.

6.1 THE TRANSPORT SERVICE

In the following sections, we will provide an introduction to the transport service. We look at what kind of service is provided to the application layer. To make the issue of transport service more concrete, we will examine two sets of transport layer primitives. First comes a simple (but hypothetical) one to show the basic ideas. Then comes the interface commonly used in the Internet.

6.1.1 Services Provided to the Upper Layers

The ultimate goal of the transport layer is to provide efficient, reliable, and cost-effective data transmission service to its users, normally processes in the application layer. To achieve this, the transport layer makes use of the services provided by the network layer. The software and/or hardware within the transport layer that does the work is called the **transport entity**. The transport entity can be located in the operating system kernel, in a library package bound into network applications, in a separate user process, or even on the network interface card. The first two options are most common on the Internet. The (logical) relationship of the network, transport, and application layers is illustrated in Fig. 6-1.

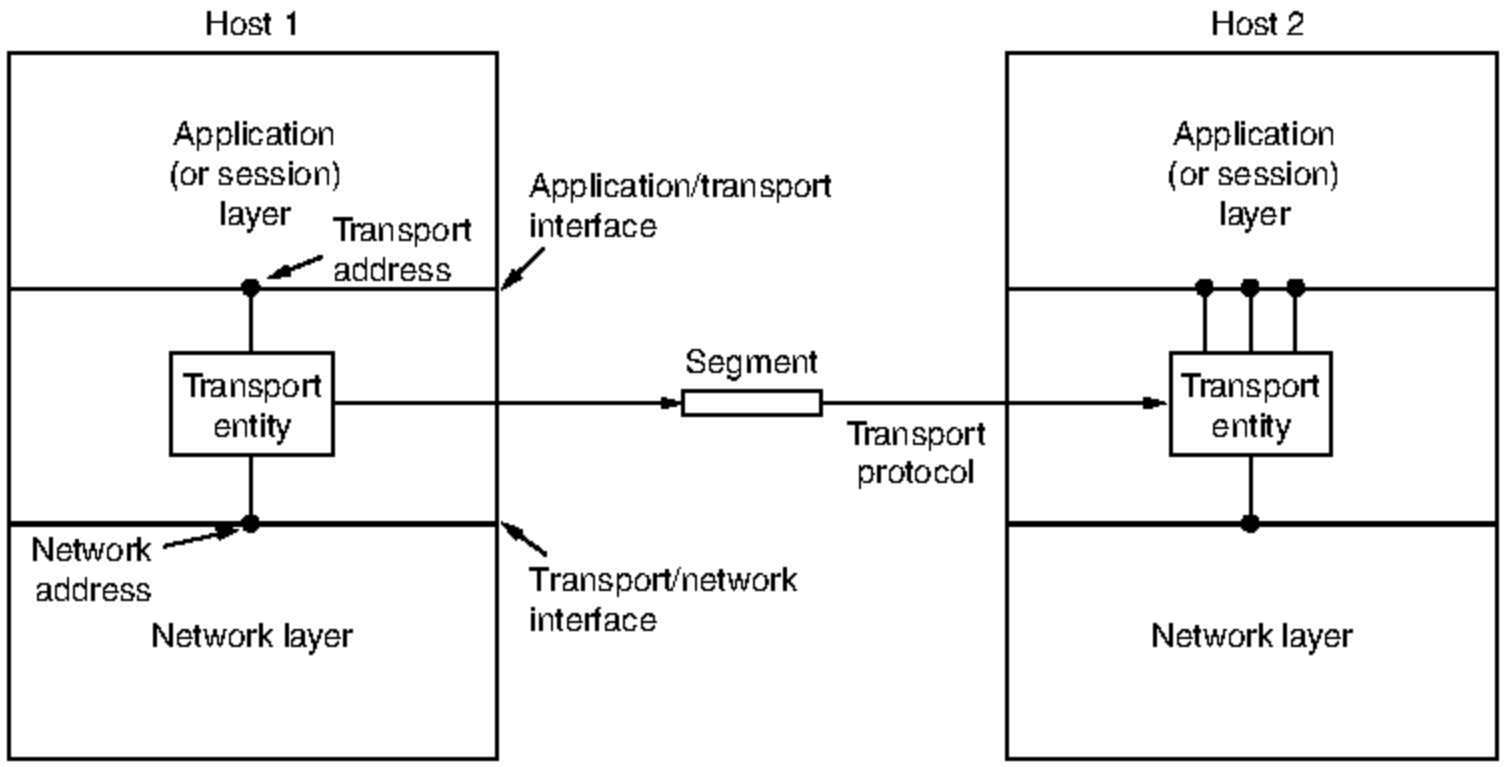


Figure 6-1. The network, transport, and application layers.

Just as there are two types of network service, connection-oriented and connectionless, there are also two types of transport service. The connection-oriented transport service is similar to the connection-oriented network service in many ways. In both cases, connections have three phases: establishment, data transfer, and release. Addressing and flow control are also similar in both layers. Furthermore, the connectionless transport service is also very similar to the connectionless network service. However, note that it can be difficult to provide a connectionless transport service on top of a connection-oriented network service, since it is inefficient to set up a connection to send a single packet and then tear it down immediately afterwards.

The obvious question is this: if the transport layer service is so similar to the network layer service, why are there two distinct layers? Why is one layer not

adequate? The answer is subtle, but crucial. The transport code runs entirely on the users' machines, but the network layer mostly runs on the routers, which are operated by the carrier (at least for a wide area network). What happens if the network layer offers inadequate service? What if it frequently loses packets? What happens if routers crash from time to time?

Problems occur, that's what. The users have no real control over the network layer, so they cannot solve the problem of poor service by using better routers or putting more error handling in the data link layer because they don't own the routers. The only possibility is to put on top of the network layer another layer that improves the quality of the service. If, in a connectionless network, packets are lost or mangled, the transport entity can detect the problem and compensate for it by using retransmissions. If, in a connection-oriented network, a transport entity is informed halfway through a long transmission that its network connection has been abruptly terminated, with no indication of what has happened to the data currently in transit, it can set up a new network connection to the remote transport entity. Using this new network connection, it can send a query to its peer asking which data arrived and which did not, and knowing where it was, pick up from where it left off.

In essence, the existence of the transport layer makes it possible for the transport service to be more reliable than the underlying network. Furthermore, the transport primitives can be implemented as calls to library procedures to make them independent of the network primitives. The network service calls may vary considerably from one network to another (e.g., calls based on a connectionless Ethernet may be quite different from calls on a connection-oriented WiMAX network). Hiding the network service behind a set of transport service primitives ensures that changing the network merely requires replacing one set of library procedures with another one that does the same thing with a different underlying service.

Thanks to the transport layer, application programmers can write code according to a standard set of primitives and have these programs work on a wide variety of networks, without having to worry about dealing with different network interfaces and levels of reliability. If all real networks were flawless and all had the same service primitives and were guaranteed never, ever to change, the transport layer might not be needed. However, in the real world it fulfills the key function of isolating the upper layers from the technology, design, and imperfections of the network.

For this reason, many people have made a qualitative distinction between layers 1 through 4 on the one hand and layer(s) above 4 on the other. The bottom four layers can be seen as the **transport service provider**, whereas the upper layer(s) are the **transport service user**. This distinction of provider versus user has a considerable impact on the design of the layers and puts the transport layer in a key position, since it forms the major boundary between the provider and user of the reliable data transmission service. It is the level that applications see.

6.1.2 Transport Service Primitives

To allow users to access the transport service, the transport layer must provide some operations to application programs, that is, a transport service interface. Each transport service has its own interface. In this section, we will first examine a simple (hypothetical) transport service and its interface to see the bare essentials. In the following section, we will look at a real example.

The transport service is similar to the network service, but there are also some important differences. The main difference is that the network service is intended to model the service offered by real networks, warts and all. Real networks can lose packets, so the network service is generally unreliable.

The connection-oriented transport service, in contrast, is reliable. Of course, real networks are not error-free, but that is precisely the purpose of the transport layer—to provide a reliable service on top of an unreliable network.

As an example, consider two processes on a single machine connected by a pipe in UNIX (or any other interprocess communication facility). They assume the connection between them is 100% perfect. They do not want to know about acknowledgements, lost packets, congestion, or anything at all like that. What they want is a 100% reliable connection. Process *A* puts data into one end of the pipe, and process *B* takes it out of the other. This is what the connection-oriented transport service is all about—hiding the imperfections of the network service so that user processes can just assume the existence of an error-free bit stream even when they are on different machines.

As an aside, the transport layer can also provide unreliable (datagram) service. However, there is relatively little to say about that besides "it's datagrams," so we will mainly concentrate on the connection-oriented transport service in this chapter. Nevertheless, there are some applications, such as client-server computing and streaming multimedia, that build on a connectionless transport service, and we will say a little bit about that later on.

A second difference between the network service and transport service is whom the services are intended for. The network service is used only by the transport entities. Few users write their own transport entities, and thus few users or programs ever see the bare network service. In contrast, many programs (and thus programmers) see the transport primitives. Consequently, the transport service must be convenient and easy to use.

To get an idea of what a transport service might be like, consider the five primitives listed in Fig. 6-2. This transport interface is truly bare bones, but it gives the essential flavor of what a connection-oriented transport interface has to do. It allows application programs to establish, use, and then release connections, which is sufficient for many applications.

To see how these primitives might be used, consider an application with a server and a number of remote clients. To start with, the server executes a LISTEN primitive, typically by calling a library procedure that makes a system call that

Primitive	Packet sent	Meaning
LISTEN	(none)	Block until some process tries to connect
CONNECT	CONNECTION REQ.	Actively attempt to establish a connection
SEND	DATA	Send information
RECEIVE	(none)	Block until a DATA packet arrives
DISCONNECT	DISCONNECTION REQ.	Request a release of the connection

Figure 6-2. The primitives for a simple transport service.

blocks the server until a client turns up. When a client wants to talk to the server, it executes a CONNECT primitive. The transport entity carries out this primitive by blocking the caller and sending a packet to the server. Encapsulated in the payload of this packet is a transport layer message for the server's transport entity.

A quick note on terminology is now in order. For lack of a better term, we will use the term **segment** for messages sent from transport entity to transport entity. TCP, UDP and other Internet protocols use this term. Some older protocols used the ungainly name **TPDU** (**Transport Protocol Data Unit**). That term is not used much any more now but you may see it in older papers and books.

Thus, segments (exchanged by the transport layer) are contained in packets (exchanged by the network layer). In turn, these packets are contained in frames (exchanged by the data link layer). When a frame arrives, the data link layer processes the frame header and, if the destination address matches for local delivery, passes the contents of the frame payload field up to the network entity. The network entity similarly processes the packet header and then passes the contents of the packet payload up to the transport entity. This nesting is illustrated in Fig. 6-3.

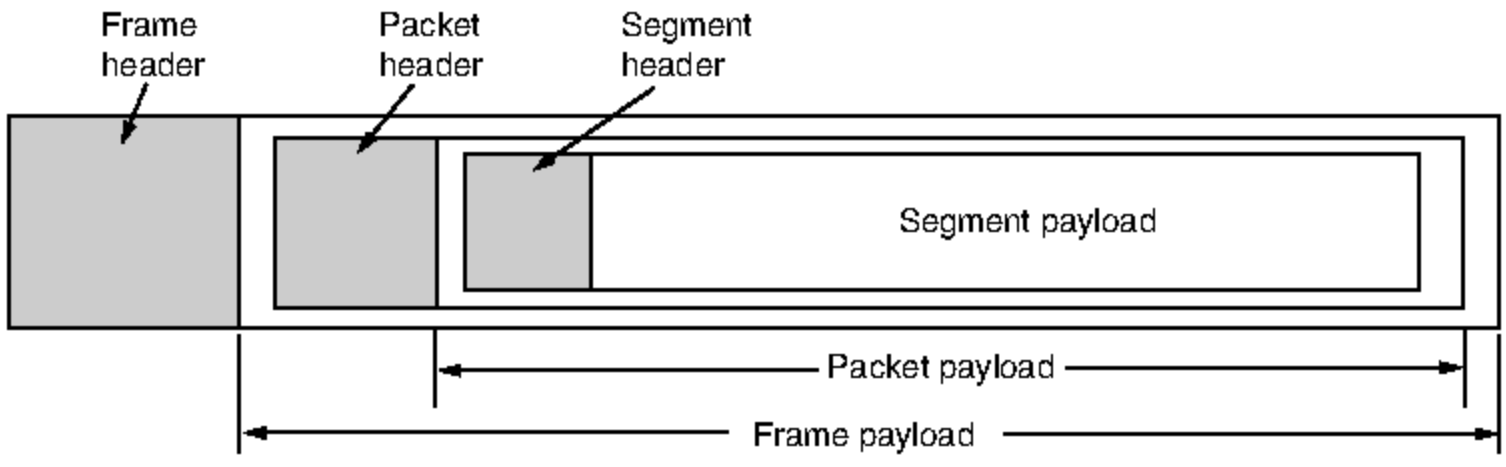


Figure 6-3. Nesting of segments, packets, and frames.

Getting back to our client-server example, the client's CONNECT call causes a CONNECTION REQUEST segment to be sent to the server. When it arrives, the

transport entity checks to see that the server is blocked on a LISTEN (i.e., is interested in handling requests). If so, it then unblocks the server and sends a CONNECTION ACCEPTED segment back to the client. When this segment arrives, the client is unblocked and the connection is established.

Data can now be exchanged using the SEND and RECEIVE primitives. In the simplest form, either party can do a (blocking) RECEIVE to wait for the other party to do a SEND. When the segment arrives, the receiver is unblocked. It can then process the segment and send a reply. As long as both sides can keep track of whose turn it is to send, this scheme works fine.

Note that in the transport layer, even a simple unidirectional data exchange is more complicated than at the network layer. Every data packet sent will also be acknowledged (eventually). The packets bearing control segments are also acknowledged, implicitly or explicitly. These acknowledgements are managed by the transport entities, using the network layer protocol, and are not visible to the transport users. Similarly, the transport entities need to worry about timers and retransmissions. None of this machinery is visible to the transport users. To the transport users, a connection is a reliable bit pipe: one user stuffs bits in and they magically appear in the same order at the other end. This ability to hide complexity is the reason that layered protocols are such a powerful tool.

When a connection is no longer needed, it must be released to free up table space within the two transport entities. Disconnection has two variants: asymmetric and symmetric. In the asymmetric variant, either transport user can issue a DISCONNECT primitive, which results in a DISCONNECT segment being sent to the remote transport entity. Upon its arrival, the connection is released.

In the symmetric variant, each direction is closed separately, independently of the other one. When one side does a DISCONNECT, that means it has no more data to send but it is still willing to accept data from its partner. In this model, a connection is released when both sides have done a DISCONNECT.

A state diagram for connection establishment and release for these simple primitives is given in Fig. 6-4. Each transition is triggered by some event, either a primitive executed by the local transport user or an incoming packet. For simplicity, we assume here that each segment is separately acknowledged. We also assume that a symmetric disconnection model is used, with the client going first. Please note that this model is quite unsophisticated. We will look at more realistic models later on when we describe how TCP works.

6.1.3 Berkeley Sockets

Let us now briefly inspect another set of transport primitives, the socket primitives as they are used for TCP. Sockets were first released as part of the Berkeley UNIX 4.2BSD software distribution in 1983. They quickly became popular. The primitives are now widely used for Internet programming on many operating

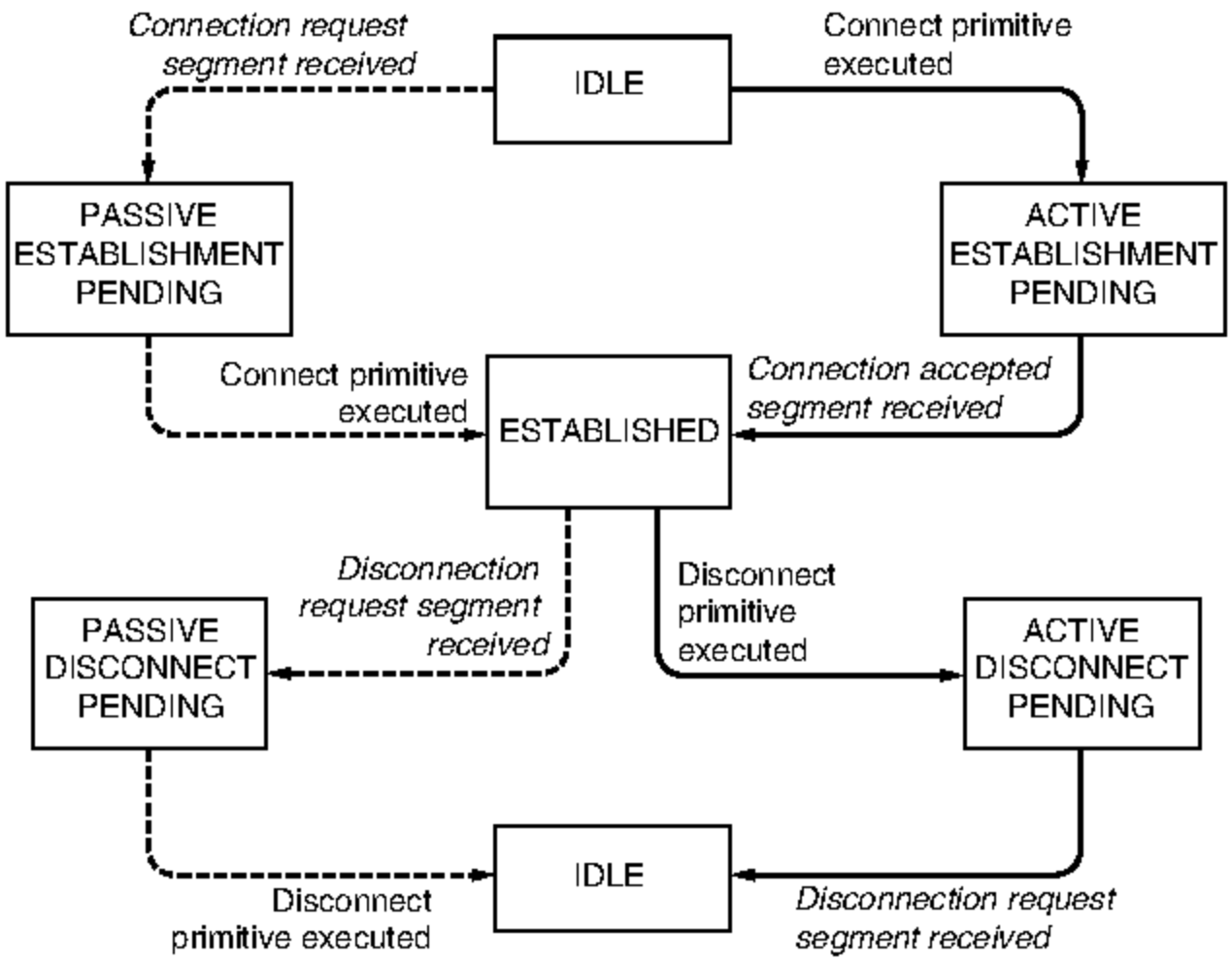


Figure 6-4. A state diagram for a simple connection management scheme. Transitions labeled in italics are caused by packet arrivals. The solid lines show the client's state sequence. The dashed lines show the server's state sequence.

systems, especially UNIX-based systems, and there is a socket-style API for Windows called "winsock."

The primitives are listed in Fig. 6-5. Roughly speaking, they follow the model of our first example but offer more features and flexibility. We will not look at the corresponding segments here. That discussion will come later.

Primitive	Meaning
SOCKET	Create a new communication endpoint
BIND	Associate a local address with a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Passively establish an incoming connection
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

Figure 6-5. The socket primitives for TCP.

The first four primitives in the list are executed in that order by servers. The SOCKET primitive creates a new endpoint and allocates table space for it within the transport entity. The parameters of the call specify the addressing format to be used, the type of service desired (e.g., reliable byte stream), and the protocol. A successful SOCKET call returns an ordinary file descriptor for use in succeeding calls, the same way an OPEN call on a file does.

Newly created sockets do not have network addresses. These are assigned using the BIND primitive. Once a server has bound an address to a socket, remote clients can connect to it. The reason for not having the SOCKET call create an address directly is that some processes care about their addresses (e.g., they have been using the same address for years and everyone knows this address), whereas others do not.

Next comes the LISTEN call, which allocates space to queue incoming calls for the case that several clients try to connect at the same time. In contrast to LISTEN in our first example, in the socket model LISTEN is not a blocking call.

To block waiting for an incoming connection, the server executes an ACCEPT primitive. When a segment asking for a connection arrives, the transport entity creates a new socket with the same properties as the original one and returns a file descriptor for it. The server can then fork off a process or thread to handle the connection on the new socket and go back to waiting for the next connection on the original socket. ACCEPT returns a file descriptor, which can be used for reading and writing in the standard way, the same as for files.

Now let us look at the client side. Here, too, a socket must first be created using the SOCKET primitive, but BIND is not required since the address used does not matter to the server. The CONNECT primitive blocks the caller and actively starts the connection process. When it completes (i.e., when the appropriate segment is received from the server), the client process is unblocked and the connection is established. Both sides can now use SEND and RECEIVE to transmit and receive data over the full-duplex connection. The standard UNIX READ and WRITE system calls can also be used if none of the special options of SEND and RECEIVE are required.

Connection release with sockets is symmetric. When both sides have executed a CLOSE primitive, the connection is released.

Sockets have proved tremendously popular and are the de facto standard for abstracting transport services to applications. The socket API is often used with the TCP protocol to provide a connection-oriented service called a **reliable byte stream**, which is simply the reliable bit pipe that we described. However, other protocols could be used to implement this service using the same API. It should all be the same to the transport service users.

A strength of the socket API is that is can be used by an application for other transport services. For instance, sockets can be used with a connectionless transport service. In this case, CONNECT sets the address of the remote transport peer and SEND and RECEIVE send and receive datagrams to and from the remote peer.

(It is also common to use an expanded set of calls, for example, SENDTO and RECEIVEFROM, that emphasize messages and do not limit an application to a single transport peer.) Sockets can also be used with transport protocols that provide a message stream rather than a byte stream and that do or do not have congestion control. For example, **DCCP** (**Datagram Congestion Controlled Protocol**) is a version of UDP with congestion control (Kohler et al., 2006). It is up to the transport users to understand what service they are getting.

However, sockets are not likely to be the final word on transport interfaces. For example, applications often work with a group of related streams, such as a Web browser that requests several objects from the same server. With sockets, the most natural fit is for application programs to use one stream per object. This structure means that congestion control is applied separately for each stream, not across the group, which is suboptimal. It punts to the application the burden of managing the set. Newer protocols and interfaces have been devised that support groups of related streams more effectively and simply for the application. Two examples are **SCTP** (**Stream Control Transmission Protocol**) defined in RFC 4960 and **SST** (**Structured Stream Transport**) (Ford, 2007). These protocols must change the socket API slightly to get the benefits of groups of related streams, and they also support features such as a mix of connection-oriented and connectionless traffic and even multiple network paths. Time will tell if they are successful.

6.1.4 An Example of Socket Programming: An Internet File Server

As an example of the nitty-gritty of how real socket calls are made, consider the client and server code of Fig. 6-6. Here we have a very primitive Internet file server along with an example client that uses it. The code has many limitations (discussed below), but in principle the server code can be compiled and run on any UNIX system connected to the Internet. The client code can be compiled and run on any other UNIX machine on the Internet, anywhere in the world. The client code can be executed with appropriate parameters to fetch any file to which the server has access on its machine. The file is written to standard output, which, of course, can be redirected to a file or pipe.

Let us look at the server code first. It starts out by including some standard headers, the last three of which contain the main Internet-related definitions and data structures. Next comes a definition of *SERVER_PORT* as 12345. This number was chosen arbitrarily. Any number between 1024 and 65535 will work just as well, as long as it is not in use by some other process; ports below 1023 are reserved for privileged users.

The next two lines in the server define two constants needed. The first one determines the chunk size in bytes used for the file transfer. The second one determines how many pending connections can be held before additional ones are discarded upon arrival.

```
/* This page contains a client program that can request a file from the server program
 * on the next page. The server responds by sending the whole file.
 */

#include <sys/types.h>
#include <sys/socket.h>
#include <netinet/in.h>
#include <netdb.h>

#define SERVER_PORT 12345                       /* arbitrary, but client & server must agree */
#define BUF_SIZE 4096                           /* block transfer size */

int main(int argc, char **argv)
{
  int c, s, bytes;
  char buf[BUF_SIZE];                           /* buffer for incoming file */
  struct hostent *h;                            /* info about server */
  struct sockaddr_in channel;                   /* holds IP address */

  if (argc != 3) fatal("Usage: client server-name file-name");
  h = gethostbyname(argv[1]);                   /* look up host's IP address */
  if (!h) fatal("gethostbyname failed");

  s = socket(PF_INET, SOCK_STREAM, IPPROTO_TCP);
  if (s <0) fatal("socket");
  memset(&channel, 0, sizeof(channel));
  channel.sin_family= AF_INET;
  memcpy(&channel.sin_addr.s_addr, h->h_addr, h->h_length);
  channel.sin_port= htons(SERVER_PORT);

  c = connect(s, (struct sockaddr *) &channel, sizeof(channel));
  if (c < 0) fatal("connect failed");

  /* Connection is now established. Send file name including 0 byte at end. */
  write(s, argv[2], strlen(argv[2])+1);

  /* Go get the file and write it to standard output. */
  while (1) {
      bytes = read(s, buf, BUF_SIZE);           /* read from socket */
      if (bytes <= 0) exit(0);                  /* check for end of file */
      write(1, buf, bytes);                     /* write to standard output */
  }
}

fatal(char *string)
{
  printf("%s\n", string);
  exit(1);
}
```

Figure 6-6. Client code using sockets. The server code is on the next page.

```
#include <sys/types.h>                          /* This is the server code */
#include <sys/fcntl.h>
#include <sys/socket.h>
#include <netinet/in.h>
#include <netdb.h>

#define SERVER_PORT 12345                       /* arbitrary, but client & server must agree */
#define BUF_SIZE 4096                           /* block transfer size */
#define QUEUE_SIZE 10

int main(int argc, char *argv[])
{
  int s, b, l, fd, sa, bytes, on = 1;
  char buf[BUF_SIZE];                           /* buffer for outgoing file */
  struct sockaddr_in channel;                   /* holds IP address */

  /* Build address structure to bind to socket. */
  memset(&channel, 0, sizeof(channel));         /* zero channel */
  channel.sin_family = AF_INET;
  channel.sin_addr.s_addr = htonl(INADDR_ANY);
  channel.sin_port = htons(SERVER_PORT);

  /* Passive open. Wait for connection. */
  s = socket(AF_INET, SOCK_STREAM, IPPROTO_TCP);    /* create socket */
  if (s < 0) fatal("socket failed");
  setsockopt(s, SOL_SOCKET, SO_REUSEADDR, (char *) &on, sizeof(on));
  b = bind(s, (struct sockaddr *) &channel, sizeof(channel));
  if (b < 0) fatal("bind failed");

  l = listen(s, QUEUE_SIZE);                    /* specify queue size */
  if (l < 0) fatal("listen failed");

  /* Socket is now set up and bound. Wait for connection and process it. */
  while (1) {
      sa = accept(s, 0, 0);                     /* block for connection request */
      if (sa < 0) fatal("accept failed");

      read(sa, buf, BUF_SIZE);                  /* read file name from socket */

      /* Get and return the file. */
      fd = open(buf, O_RDONLY);                 /* open the file to be sent back */
      if (fd < 0) fatal("open failed");

      while (1) {
          bytes = read(fd, buf, BUF_SIZE); /* read from file */
          if (bytes <= 0) break;                /* check for end of file */
          write(sa, buf, bytes);                /* write bytes to socket */
      }
      close(fd);                                /* close file */
      close(sa);                                /* close connection */
  }
}
```

After the declarations of local variables, the server code begins. It starts out by initializing a data structure that will hold the server's IP address. This data structure will soon be bound to the server's socket. The call to *memset* sets the data structure to all 0s. The three assignments following it fill in three of its fields. The last of these contains the server's port. The functions *htonl* and *htons* have to do with converting values to a standard format so the code runs correctly on both little-endian machines (e.g., Intel x86) and big-endian machines (e.g., the SPARC). Their exact semantics are not relevant here.

Next, the server creates a socket and checks for errors (indicated by $s < 0$). In a production version of the code, the error message could be a trifle more explanatory. The call to *setsockopt* is needed to allow the port to be reused so the server can run indefinitely, fielding request after request. Now the IP address is bound to the socket and a check is made to see if the call to *bind* succeeded. The final step in the initialization is the call to *listen* to announce the server's willingness to accept incoming calls and tell the system to hold up to *QUEUE_SIZE* of them in case new requests arrive while the server is still processing the current one. If the queue is full and additional requests arrive, they are quietly discarded.

At this point, the server enters its main loop, which it never leaves. The only way to stop it is to kill it from outside. The call to *accept* blocks the server until some client tries to establish a connection with it. If the *accept* call succeeds, it returns a socket descriptor that can be used for reading and writing, analogous to how file descriptors can be used to read from and write to pipes. However, unlike pipes, which are unidirectional, sockets are bidirectional, so *sa* (the accepted socket) can be used for reading from the connection and also for writing to it. A pipe file descriptor is for reading or writing but not both.

After the connection is established, the server reads the file name from it. If the name is not yet available, the server blocks waiting for it. After getting the file name, the server opens the file and enters a loop that alternately reads blocks from the file and writes them to the socket until the entire file has been copied. Then the server closes the file and the connection and waits for the next connection to show up. It repeats this loop forever.

Now let us look at the client code. To understand how it works, it is necessary to understand how it is invoked. Assuming it is called *client*, a typical call is

```
client flits.cs.vu.nl /usr/tom/filename >f
```

This call only works if the server is already running on *flits.cs.vu.nl* and the file */usr/tom/filename* exists and the server has read access to it. If the call is successful, the file is transferred over the Internet and written to *f*, after which the client program exits. Since the server continues after a transfer, the client can be started again and again to get other files.

The client code starts with some includes and declarations. Execution begins by checking to see if it has been called with the right number of arguments (*argc* = 3 means the program name plus two arguments). Note that *argv*[1] contains the

name of the server (e.g., *flits.cs.vu.nl*) and is converted to an IP address by *gethostbyname*. This function uses DNS to look up the name. We will study DNS in Chap. 7.

Next, a socket is created and initialized. After that, the client attempts to establish a TCP connection to the server, using *connect*. If the server is up and running on the named machine and attached to *SERVER_PORT* and is either idle or has room in its *listen* queue, the connection will (eventually) be established. Using the connection, the client sends the name of the file by writing on the socket. The number of bytes sent is one larger than the name proper, since the 0 byte terminating the name must also be sent to tell the server where the name ends.

Now the client enters a loop, reading the file block by block from the socket and copying it to standard output. When it is done, it just exits.

The procedure *fatal* prints an error message and exits. The server needs the same procedure, but it was omitted due to lack of space on the page. Since the client and server are compiled separately and normally run on different computers, they cannot share the code of *fatal*.

These two programs (as well as other material related to this book) can be fetched from the book's Web site

http://www.pearsonhighered.com/tanenbaum

Just for the record, this server is not the last word in serverdom. Its error checking is meager and its error reporting is mediocre. Since it handles all requests strictly sequentially (because it has only a single thread), its performance is poor. It has clearly never heard about security, and using bare UNIX system calls is not the way to gain platform independence. It also makes some assumptions that are technically illegal, such as assuming that the file name fits in the buffer and is transmitted atomically. These shortcomings notwithstanding, it is a working Internet file server. In the exercises, the reader is invited to improve it. For more information about programming with sockets, see Donahoo and Calvert (2008, 2009).

6.2 ELEMENTS OF TRANSPORT PROTOCOLS

The transport service is implemented by a **transport protocol** used between the two transport entities. In some ways, transport protocols resemble the data link protocols we studied in detail in Chap. 3. Both have to deal with error control, sequencing, and flow control, among other issues.

However, significant differences between the two also exist. These differences are due to major dissimilarities between the environments in which the two protocols operate, as shown in Fig. 6-7. At the data link layer, two routers

communicate directly via a physical channel, whether wired or wireless, whereas at the transport layer, this physical channel is replaced by the entire network. This difference has many important implications for the protocols.

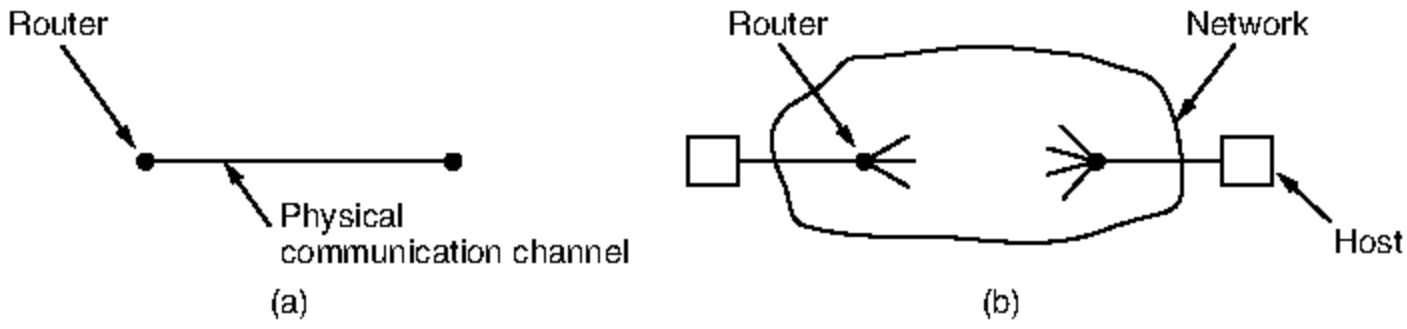


Figure 6-7. (a) Environment of the data link layer. (b) Environment of the transport layer.

For one thing, over point-to-point links such as wires or optical fiber, it is usually not necessary for a router to specify which router it wants to talk to—each outgoing line leads directly to a particular router. In the transport layer, explicit addressing of destinations is required.

For another thing, the process of establishing a connection over the wire of Fig. 6-7(a) is simple: the other end is always there (unless it has crashed, in which case it is not there). Either way, there is not much to do. Even on wireless links, the process is not much different. Just sending a message is sufficient to have it reach all other destinations. If the message is not acknowledged due to an error, it can be resent. In the transport layer, initial connection establishment is complicated, as we will see.

Another (exceedingly annoying) difference between the data link layer and the transport layer is the potential existence of storage capacity in the network. When a router sends a packet over a link, it may arrive or be lost, but it cannot bounce around for a while, go into hiding in a far corner of the world, and suddenly emerge after other packets that were sent much later. If the network uses datagrams, which are independently routed inside, there is a nonnegligible probability that a packet may take the scenic route and arrive late and out of the expected order, or even that duplicates of the packet will arrive. The consequences of the network's ability to delay and duplicate packets can sometimes be disastrous and can require the use of special protocols to correctly transport information.

A final difference between the data link and transport layers is one of degree rather than of kind. Buffering and flow control are needed in both layers, but the presence in the transport layer of a large and varying number of connections with bandwidth that fluctuates as the connections compete with each other may require a different approach than we used in the data link layer. Some of the protocols discussed in Chap. 3 allocate a fixed number of buffers to each line, so that when a frame arrives a buffer is always available. In the transport layer, the larger number of connections that must be managed and variations in the bandwidth each

connection may receive make the idea of dedicating many buffers to each one less attractive. In the following sections, we will examine all of these important issues, and others.

6.2.1 Addressing

When an application (e.g., a user) process wishes to set up a connection to a remote application process, it must specify which one to connect to. (Connectionless transport has the same problem: to whom should each message be sent?) The method normally used is to define transport addresses to which processes can listen for connection requests. In the Internet, these endpoints are called **ports**. We will use the generic term **TSAP** (**Transport Service Access Point**) to mean a specific endpoint in the transport layer. The analogous endpoints in the network layer (i.e., network layer addresses) are not-surprisingly called **NSAPs** (**Network Service Access Points**). IP addresses are examples of NSAPs.

Figure 6-8 illustrates the relationship between the NSAPs, the TSAPs, and a transport connection. Application processes, both clients and servers, can attach themselves to a local TSAP to establish a connection to a remote TSAP. These connections run through NSAPs on each host, as shown. The purpose of having TSAPs is that in some networks, each computer has a single NSAP, so some way is needed to distinguish multiple transport endpoints that share that NSAP.

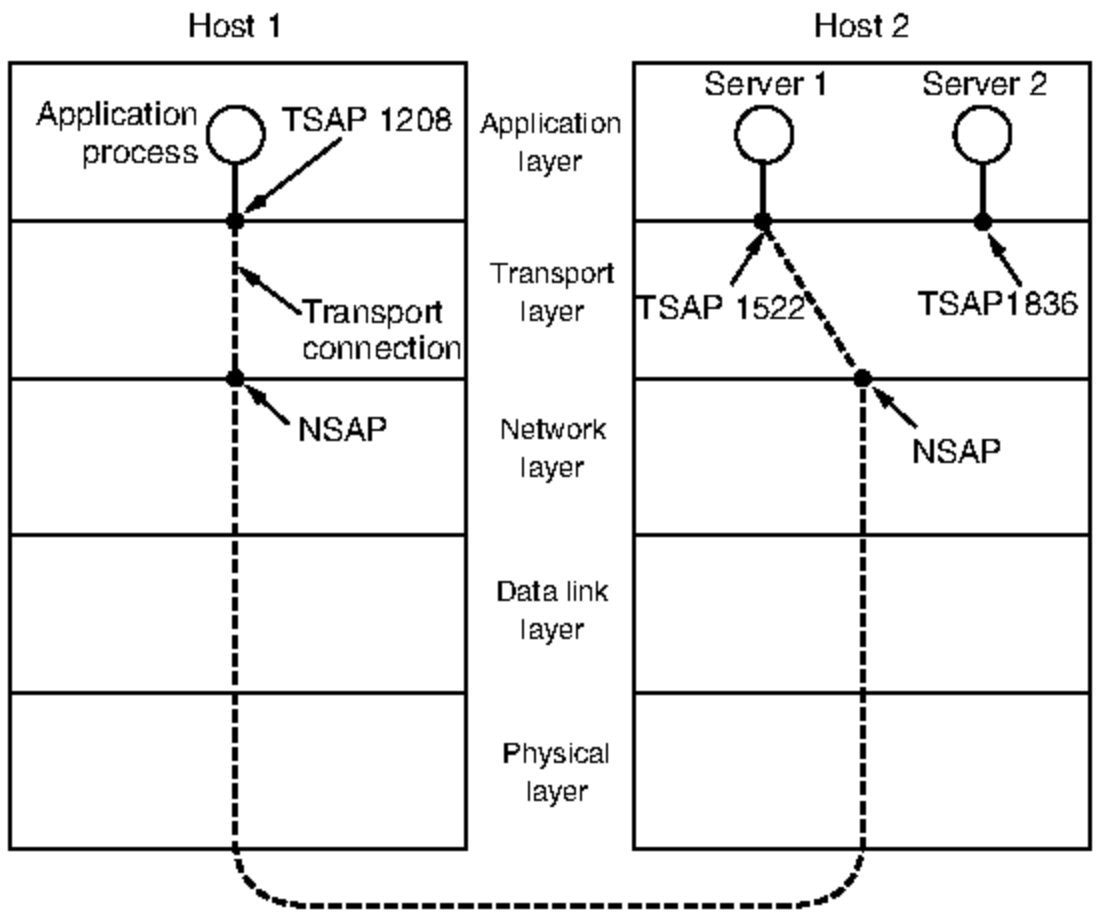


Figure 6-8. TSAPs, NSAPs, and transport connections.

A possible scenario for a transport connection is as follows:

1. A mail server process attaches itself to TSAP 1522 on host 2 to wait for an incoming call. How a process attaches itself to a TSAP is outside the networking model and depends entirely on the local operating system. A call such as our LISTEN might be used, for example.
2. An application process on host 1 wants to send an email message, so it attaches itself to TSAP 1208 and issues a CONNECT request. The request specifies TSAP 1208 on host 1 as the source and TSAP 1522 on host 2 as the destination. This action ultimately results in a transport connection being established between the application process and the server.
3. The application process sends over the mail message.
4. The mail server responds to say that it will deliver the message.
5. The transport connection is released.

Note that there may well be other servers on host 2 that are attached to other TSAPs and are waiting for incoming connections that arrive over the same NSAP.

The picture painted above is fine, except we have swept one little problem under the rug: how does the user process on host 1 know that the mail server is attached to TSAP 1522? One possibility is that the mail server has been attaching itself to TSAP 1522 for years and gradually all the network users have learned this. In this model, services have stable TSAP addresses that are listed in files in well-known places. For example, the */etc/services* file on UNIX systems lists which servers are permanently attached to which ports, including the fact that the mail server is found on TCP port 25.

While stable TSAP addresses work for a small number of key services that never change (e.g., the Web server), user processes, in general, often want to talk to other user processes that do not have TSAP addresses that are known in advance, or that may exist for only a short time.

To handle this situation, an alternative scheme can be used. In this scheme, there exists a special process called a **portmapper**. To find the TSAP address corresponding to a given service name, such as "BitTorrent," a user sets up a connection to the portmapper (which listens to a well-known TSAP). The user then sends a message specifying the service name, and the portmapper sends back the TSAP address. Then the user releases the connection with the portmapper and establishes a new one with the desired service.

In this model, when a new service is created, it must register itself with the portmapper, giving both its service name (typically, an ASCII string) and its TSAP. The portmapper records this information in its internal database so that when queries come in later, it will know the answers.

The function of the portmapper is analogous to that of a directory assistance operator in the telephone system—it provides a mapping of names onto numbers. Just as in the telephone system, it is essential that the address of the well-known TSAP used by the portmapper is indeed well known. If you do not know the number of the information operator, you cannot call the information operator to find it out. If you think the number you dial for information is obvious, try it in a foreign country sometime.

Many of the server processes that can exist on a machine will be used only rarely. It is wasteful to have each of them active and listening to a stable TSAP address all day long. An alternative scheme is shown in Fig. 6-9 in a simplified form. It is known as the **initial connection protocol**. Instead of every conceivable server listening at a well-known TSAP, each machine that wishes to offer services to remote users has a special **process server** that acts as a proxy for less heavily used servers. This server is called *inetd* on UNIX systems. It listens to a set of ports at the same time, waiting for a connection request. Potential users of a service begin by doing a CONNECT request, specifying the TSAP address of the service they want. If no server is waiting for them, they get a connection to the process server, as shown in Fig. 6-9(a).

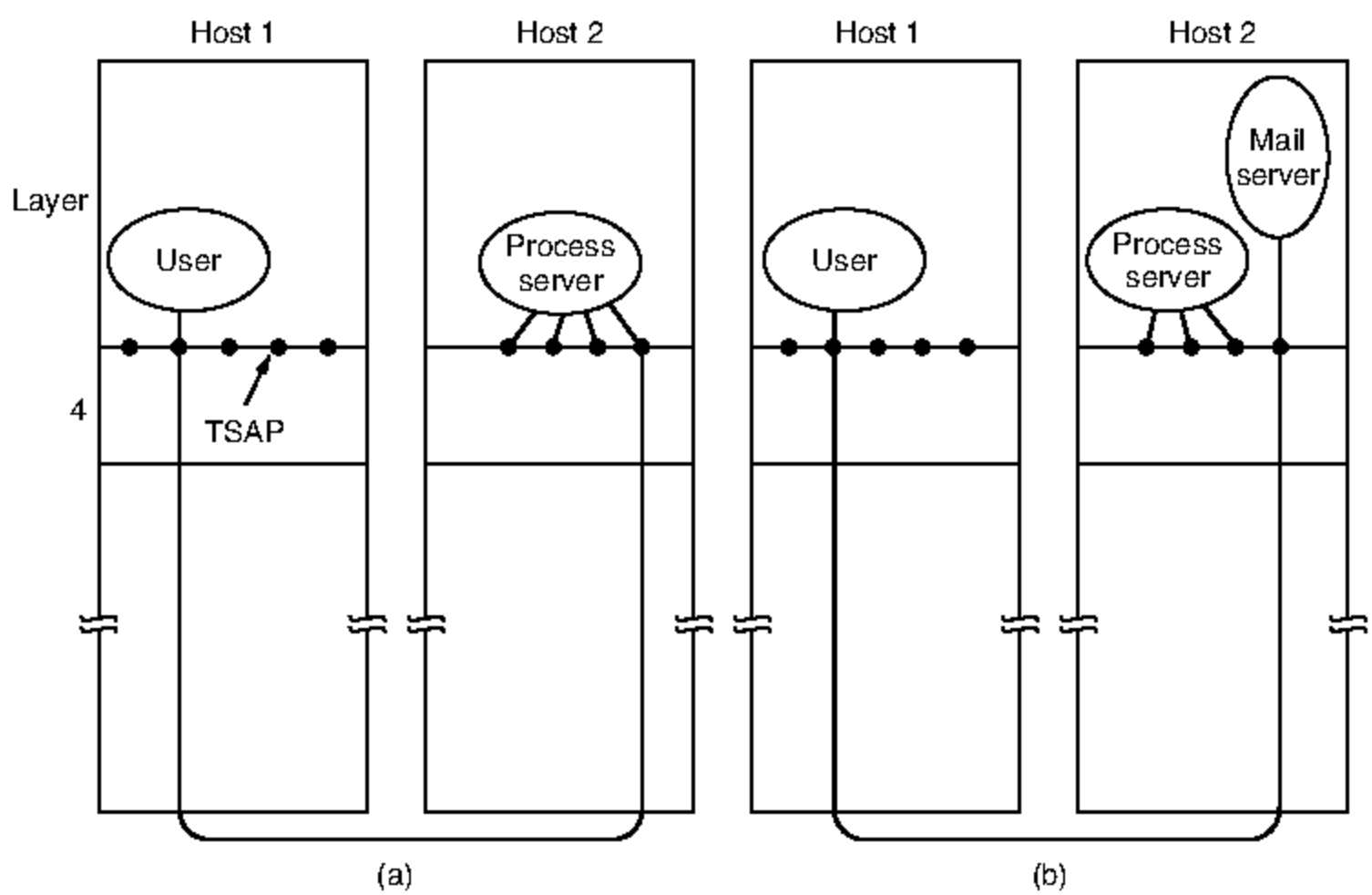


Figure 6-9. How a user process in host 1 establishes a connection with a mail server in host 2 via a process server.

After it gets the incoming request, the process server spawns the requested server, allowing it to inherit the existing connection with the user. The new server

does the requested work, while the process server goes back to listening for new requests, as shown in Fig. 6-9(b). This method is only applicable when servers can be created on demand.

6.2.2 Connection Establishment

Establishing a connection sounds easy, but it is actually surprisingly tricky. At first glance, it would seem sufficient for one transport entity to just send a CONNECTION REQUEST segment to the destination and wait for a CONNECTION ACCEPTED reply. The problem occurs when the network can lose, delay, corrupt, and duplicate packets. This behavior causes serious complications.

Imagine a network that is so congested that acknowledgements hardly ever get back in time and each packet times out and is retransmitted two or three times. Suppose that the network uses datagrams inside and that every packet follows a different route. Some of the packets might get stuck in a traffic jam inside the network and take a long time to arrive. That is, they may be delayed in the network and pop out much later, when the sender thought that they had been lost.

The worst possible nightmare is as follows. A user establishes a connection with a bank, sends messages telling the bank to transfer a large amount of money to the account of a not-entirely-trustworthy person. Unfortunately, the packets decide to take the scenic route to the destination and go off exploring a remote corner of the network. The sender then times out and sends them all again. This time the packets take the shortest route and are delivered quickly so the sender releases the connection.

Unfortunately, eventually the initial batch of packets finally come out of hiding and arrive at the destination in order, asking the bank to establish a new connection and transfer money (again). The bank has no way of telling that these are duplicates. It must assume that this is a second, independent transaction, and transfers the money again.

This scenario may sound unlikely, or even implausible but the point is this: protocols must be designed to be correct in all cases. Only the common cases need be implemented efficiently to obtain good network performance, but the protocol must be able to cope with the uncommon cases without breaking. If it cannot, we have built a fair-weather network that can fail without warning when the conditions get tough.

For the remainder of this section, we will study the problem of delayed duplicates, with emphasis on algorithms for establishing connections in a reliable way, so that nightmares like the one above cannot happen. The crux of the problem is that the delayed duplicates are thought to be new packets. We cannot prevent packets from being duplicated and delayed. But if and when this happens, the packets must be rejected as duplicates and not processed as fresh packets.

The problem can be attacked in various ways, none of them very satisfactory. One way is to use throwaway transport addresses. In this approach, each time a

transport address is needed, a new one is generated. When a connection is released, the address is discarded and never used again. Delayed duplicate packets then never find their way to a transport process and can do no damage. However, this approach makes it more difficult to connect with a process in the first place.

Another possibility is to give each connection a unique identifier (i.e., a sequence number incremented for each connection established) chosen by the initiating party and put in each segment, including the one requesting the connection. After each connection is released, each transport entity can update a table listing obsolete connections as (peer transport entity, connection identifier) pairs. Whenever a connection request comes in, it can be checked against the table to see if it belongs to a previously released connection.

Unfortunately, this scheme has a basic flaw: it requires each transport entity to maintain a certain amount of history information indefinitely. This history must persist at both the source and destination machines. Otherwise, if a machine crashes and loses its memory, it will no longer know which connection identifiers have already been used by its peers.

Instead, we need to take a different tack to simplify the problem. Rather than allowing packets to live forever within the network, we devise a mechanism to kill off aged packets that are still hobbling about. With this restriction, the problem becomes somewhat more manageable.

Packet lifetime can be restricted to a known maximum using one (or more) of the following techniques:

1. Restricted network design.
2. Putting a hop counter in each packet.
3. Timestamping each packet.

The first technique includes any method that prevents packets from looping, combined with some way of bounding delay including congestion over the (now known) longest possible path. It is difficult, given that internets may range from a single city to international in scope. The second method consists of having the hop count initialized to some appropriate value and decremented each time the packet is forwarded. The network protocol simply discards any packet whose hop counter becomes zero. The third method requires each packet to bear the time it was created, with the routers agreeing to discard any packet older than some agreed-upon time. This latter method requires the router clocks to be synchronized, which itself is a nontrivial task, and in practice a hop counter is a close enough approximation to age.

In practice, we will need to guarantee not only that a packet is dead, but also that all acknowledgements to it are dead, too, so we will now introduce a period T, which is some small multiple of the true maximum packet lifetime. The maximum packet lifetime is a conservative constant for a network; for the Internet, it is somewhat arbitrarily taken to be 120 seconds. The multiple is protocol dependent

and simply has the effect of making T longer. If we wait a time T secs after a packet has been sent, we can be sure that all traces of it are now gone and that neither it nor its acknowledgements will suddenly appear out of the blue to complicate matters.

With packet lifetimes bounded, it is possible to devise a practical and foolproof way to reject delayed duplicate segments. The method described below is due to Tomlinson (1975), as refined by Sunshine and Dalal (1978). Variants of it are widely used in practice, including in TCP.

The heart of the method is for the source to label segments with sequence numbers that will not be reused within T secs. The period, T, and the rate of packets per second determine the size of the sequence numbers. In this way, only one packet with a given sequence number may be outstanding at any given time. Duplicates of this packet may still occur, and they must be discarded by the destination. However, it is no longer the case that a delayed duplicate of an old packet may beat a new packet with the same sequence number and be accepted by the destination in its stead.

To get around the problem of a machine losing all memory of where it was after a crash, one possibility is to require transport entities to be idle for T secs after a recovery. The idle period will let all old segments die off, so the sender can start again with any sequence number. However, in a complex internetwork, T may be large, so this strategy is unattractive.

Instead, Tomlinson proposed equipping each host with a time-of-day clock. The clocks at different hosts need not be synchronized. Each clock is assumed to take the form of a binary counter that increments itself at uniform intervals. Furthermore, the number of bits in the counter must equal or exceed the number of bits in the sequence numbers. Last, and most important, the clock is assumed to continue running even if the host goes down.

When a connection is set up, the low-order k bits of the clock are used as the k-bit initial sequence number. Thus, unlike our protocols of Chap. 3, each connection starts numbering its segments with a different initial sequence number. The sequence space should be so large that by the time sequence numbers wrap around, old segments with the same sequence number are long gone. This linear relation between time and initial sequence numbers is shown in Fig. 6-10(a). The forbidden region shows the times for which segment sequence numbers are illegal leading up to their use. If any segment is sent with a sequence number in this region, it could be delayed and impersonate a different packet with the same sequence number that will be issued slightly later. For example, if the host crashes and restarts at time 70 seconds, it will use initial sequence numbers based on the clock to pick up after it left off; the host does not start with a lower sequence number in the forbidden region.

Once both transport entities have agreed on the initial sequence number, any sliding window protocol can be used for data flow control. This window protocol will correctly find and discard duplicates of packets after they have already been

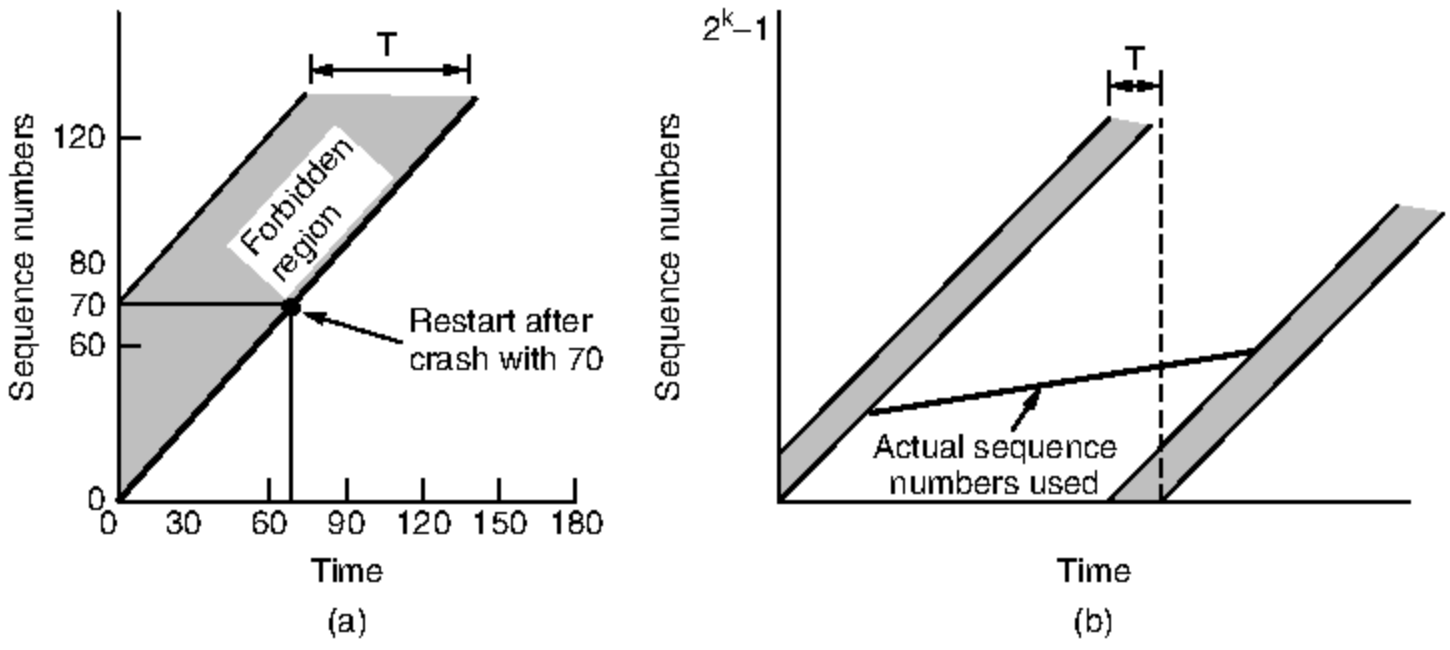


Figure 6-10. (a) Segments may not enter the forbidden region. (b) The resynchronization problem.

accepted. In reality, the initial sequence number curve (shown by the heavy line) is not linear, but a staircase, since the clock advances in discrete steps. For simplicity, we will ignore this detail.

To keep packet sequence numbers out of the forbidden region, we need to take care in two respects. We can get into trouble in two distinct ways. If a host sends too much data too fast on a newly opened connection, the actual sequence number versus time curve may rise more steeply than the initial sequence number versus time curve, causing the sequence number to enter the forbidden region. To prevent this from happening, the maximum data rate on any connection is one segment per clock tick. This also means that the transport entity must wait until the clock ticks before opening a new connection after a crash restart, lest the same number be used twice. Both of these points argue in favor of a short clock tick (1 μsec or less). But the clock cannot tick too fast relative to the sequence number. For a clock rate of C and a sequence number space of size S, we must have $S/C > T$ so that the sequence numbers cannot wrap around too quickly.

Entering the forbidden region from underneath by sending too fast is not the only way to get into trouble. From Fig. 6-10(b), we see that at any data rate less than the clock rate, the curve of actual sequence numbers used versus time will eventually run into the forbidden region from the left as the sequence numbers wrap around. The greater the slope of the actual sequence numbers, the longer this event will be delayed. Avoiding this situation limits how slowly sequence numbers can advance on a connection (or how long the connections may last).

The clock-based method solves the problem of not being able to distinguish delayed duplicate segments from new segments. However, there is a practical snag for using it for establishing connections. Since we do not normally remember sequence numbers across connections at the destination, we still have no way of

knowing if a CONNECTION REQUEST segment containing an initial sequence number is a duplicate of a recent connection. This snag does not exist during a connection because the sliding window protocol does remember the current sequence number.

To solve this specific problem, Tomlinson (1975) introduced the **three-way handshake**. This establishment protocol involves one peer checking with the other that the connection request is indeed current. The normal setup procedure when host 1 initiates is shown in Fig. 6-11(a). Host 1 chooses a sequence number, x, and sends a CONNECTION REQUEST segment containing it to host 2. Host 2 replies with an ACK segment acknowledging x and announcing its own initial sequence number, y. Finally, host 1 acknowledges host 2's choice of an initial sequence number in the first data segment that it sends.

Now let us see how the three-way handshake works in the presence of delayed duplicate control segments. In Fig. 6-11(b), the first segment is a delayed duplicate CONNECTION REQUEST from an old connection. This segment arrives at host 2 without host 1's knowledge. Host 2 reacts to this segment by sending host 1 an ACK segment, in effect asking for verification that host 1 was indeed trying to set up a new connection. When host 1 rejects host 2's attempt to establish a connection, host 2 realizes that it was tricked by a delayed duplicate and abandons the connection. In this way, a delayed duplicate does no damage.

The worst case is when both a delayed CONNECTION REQUEST and an ACK are floating around in the subnet. This case is shown in Fig. 6-11(c). As in the previous example, host 2 gets a delayed CONNECTION REQUEST and replies to it. At this point, it is crucial to realize that host 2 has proposed using y as the initial sequence number for host 2 to host 1 traffic, knowing full well that no segments containing sequence number y or acknowledgements to y are still in existence. When the second delayed segment arrives at host 2, the fact that z has been acknowledged rather than y tells host 2 that this, too, is an old duplicate. The important thing to realize here is that there is no combination of old segments that can cause the protocol to fail and have a connection set up by accident when no one wants it.

TCP uses this three-way handshake to establish connections. Within a connection, a timestamp is used to extend the 32-bit sequence number so that it will not wrap within the maximum packet lifetime, even for gigabit-per-second connections. This mechanism is a fix to TCP that was needed as it was used on faster and faster links. It is described in RFC 1323 and called **PAWS** (**Protection Against Wrapped Sequence numbers**). Across connections, for the initial sequence numbers and before PAWS can come into play, TCP originally used the clock-based scheme just described. However, this turned out to have a security vulnerability. The clock made it easy for an attacker to predict the next initial sequence number and send packets that tricked the three-way handshake and established a forged connection. To close this hole, pseudorandom initial sequence numbers are used for connections in practice. However, it remains important that

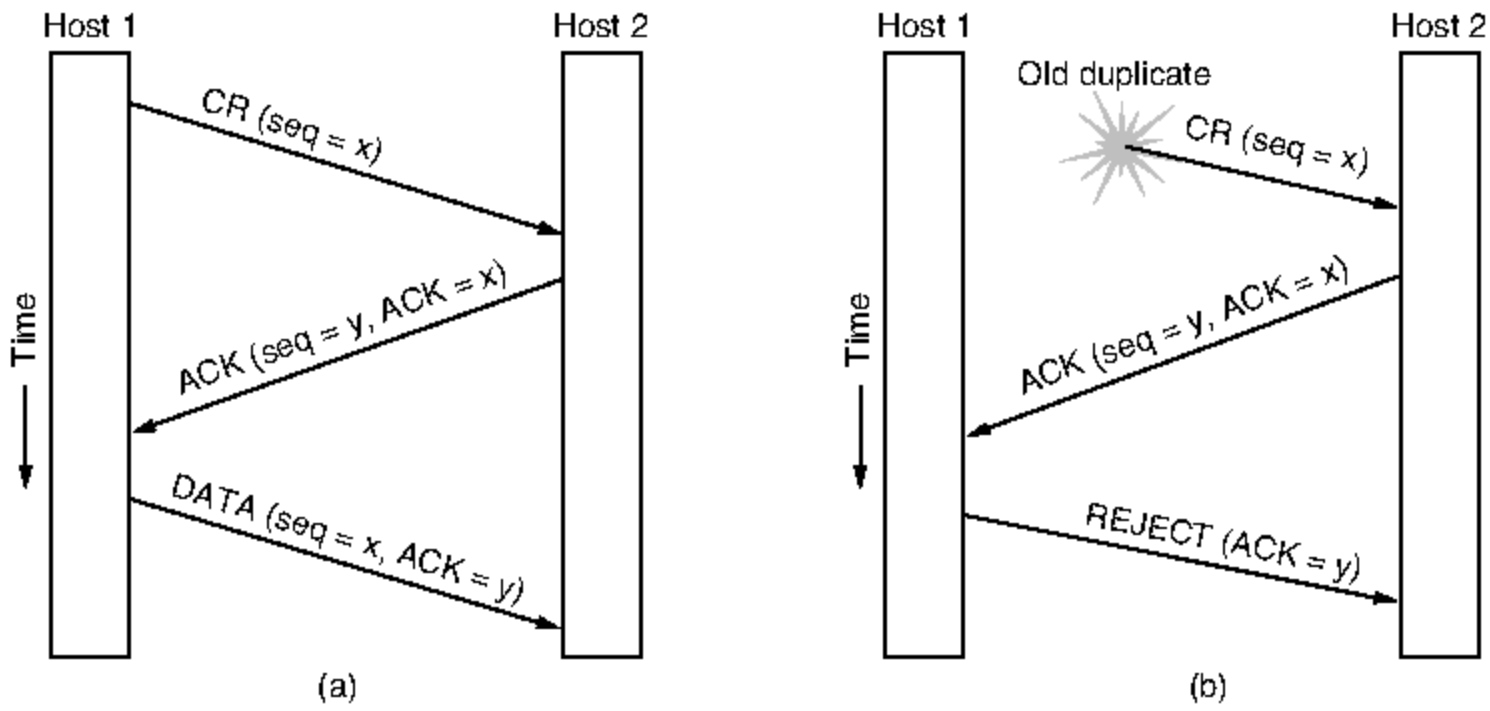


Figure 6-11. Three protocol scenarios for establishing a connection using a three-way handshake. CR denotes CONNECTION REQUEST. (a) Normal operation. (b) Old duplicate CONNECTION REQUEST appearing out of nowhere. (c) Duplicate CONNECTION REQUEST and duplicate ACK.

the initial sequence numbers not repeat for an interval even though they appear random to an observer. Otherwise, delayed duplicates can wreak havoc.

6.2.3 Connection Release

Releasing a connection is easier than establishing one. Nevertheless, there are more pitfalls than one might expect here. As we mentioned earlier, there are two styles of terminating a connection: asymmetric release and symmetric release.

Asymmetric release is the way the telephone system works: when one party hangs up, the connection is broken. Symmetric release treats the connection as two separate unidirectional connections and requires each one to be released separately.

Asymmetric release is abrupt and may result in data loss. Consider the scenario of Fig. 6-12. After the connection is established, host 1 sends a segment that arrives properly at host 2. Then host 1 sends another segment. Unfortunately, host 2 issues a DISCONNECT before the second segment arrives. The result is that the connection is released and data are lost.

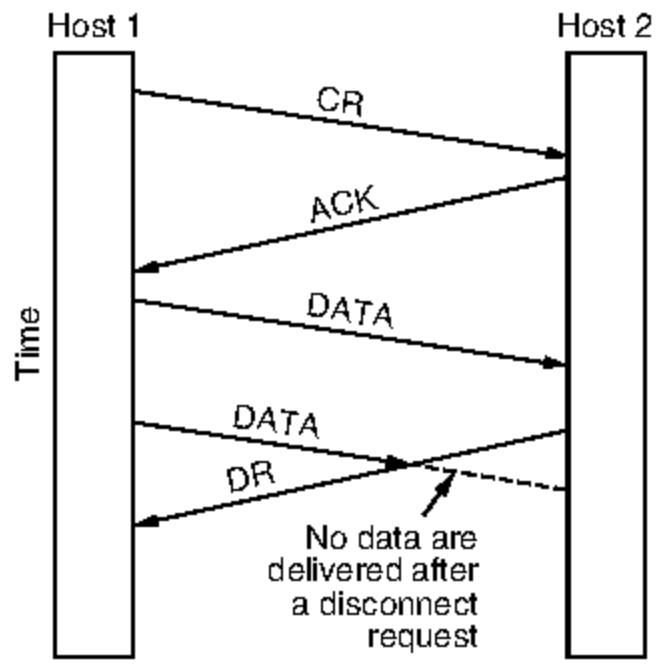


Figure 6-12. Abrupt disconnection with loss of data.

Clearly, a more sophisticated release protocol is needed to avoid data loss. One way is to use symmetric release, in which each direction is released independently of the other one. Here, a host can continue to receive data even after it has sent a DISCONNECT segment.

Symmetric release does the job when each process has a fixed amount of data to send and clearly knows when it has sent it. In other situations, determining that all the work has been done and the connection should be terminated is not so obvious. One can envision a protocol in which host 1 says "I am done. Are you done too?" If host 2 responds: "I am done too. Goodbye, the connection can be safely released."

Unfortunately, this protocol does not always work. There is a famous problem that illustrates this issue. It is called the **two-army problem**. Imagine that a white army is encamped in a valley, as shown in Fig. 6-13. On both of the surrounding hillsides are blue armies. The white army is larger than either of the blue armies alone, but together the blue armies are larger than the white army. If either blue army attacks by itself, it will be defeated, but if the two blue armies attack simultaneously, they will be victorious.

The blue armies want to synchronize their attacks. However, their only communication medium is to send messengers on foot down into the valley, where

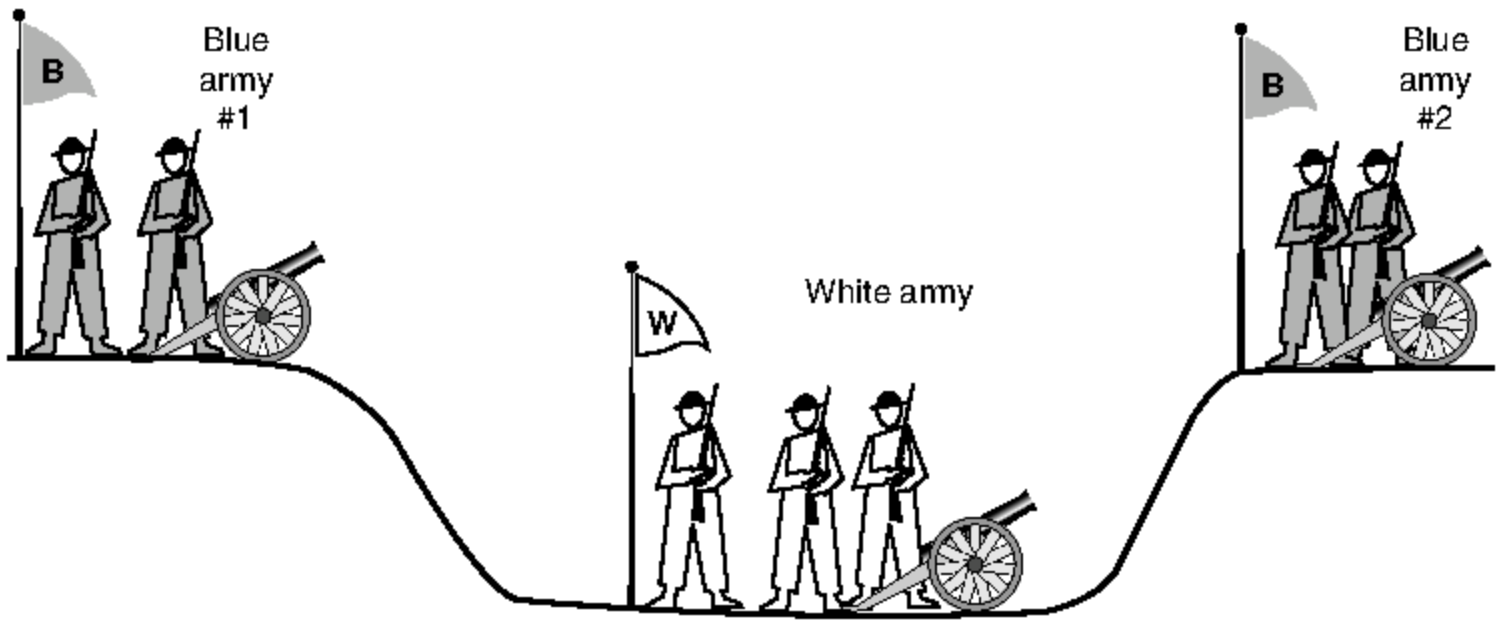


Figure 6-13. The two-army problem.

they might be captured and the message lost (i.e., they have to use an unreliable communication channel). The question is: does a protocol exist that allows the blue armies to win?

Suppose that the commander of blue army #1 sends a message reading: "I propose we attack at dawn on March 29. How about it?" Now suppose that the message arrives, the commander of blue army #2 agrees, and his reply gets safely back to blue army #1. Will the attack happen? Probably not, because commander #2 does not know if his reply got through. If it did not, blue army #1 will not attack, so it would be foolish for him to charge into battle.

Now let us improve the protocol by making it a three-way handshake. The initiator of the original proposal must acknowledge the response. Assuming no messages are lost, blue army #2 will get the acknowledgement, but the commander of blue army #1 will now hesitate. After all, he does not know if his acknowledgement got through, and if it did not, he knows that blue army #2 will not attack. We could now make a four-way handshake protocol, but that does not help either.

In fact, it can be proven that no protocol exists that works. Suppose that some protocol did exist. Either the last message of the protocol is essential, or it is not. If it is not, we can remove it (and any other unessential messages) until we are left with a protocol in which every message is essential. What happens if the final message does not get through? We just said that it was essential, so if it is lost, the attack does not take place. Since the sender of the final message can never be sure of its arrival, he will not risk attacking. Worse yet, the other blue army knows this, so it will not attack either.

To see the relevance of the two-army problem to releasing connections, rather than to military affairs, just substitute "disconnect" for "attack." If neither side is

prepared to disconnect until it is convinced that the other side is prepared to disconnect too, the disconnection will never happen.

In practice, we can avoid this quandary by foregoing the need for agreement and pushing the problem up to the transport user, letting each side independently decide when it is done. This is an easier problem to solve. Figure 6-14 illustrates four scenarios of releasing using a three-way handshake. While this protocol is not infallible, it is usually adequate.

In Fig. 6-14(a), we see the normal case in which one of the users sends a DR (DISCONNECTION REQUEST) segment to initiate the connection release. When it arrives, the recipient sends back a DR segment and starts a timer, just in case its DR is lost. When this DR arrives, the original sender sends back an ACK segment and releases the connection. Finally, when the ACK segment arrives, the receiver also releases the connection. Releasing a connection means that the transport entity removes the information about the connection from its table of currently open connections and signals the connection's owner (the transport user) somehow. This action is different from a transport user issuing a DISCONNECT primitive.

If the final ACK segment is lost, as shown in Fig. 6-14(b), the situation is saved by the timer. When the timer expires, the connection is released anyway.

Now consider the case of the second DR being lost. The user initiating the disconnection will not receive the expected response, will time out, and will start all over again. In Fig. 6-14(c), we see how this works, assuming that the second time no segments are lost and all segments are delivered correctly and on time.

Our last scenario, Fig. 6-14(d), is the same as Fig. 6-14(c) except that now we assume all the repeated attempts to retransmit the DR also fail due to lost segments. After N retries, the sender just gives up and releases the connection. Meanwhile, the receiver times out and also exits.

While this protocol usually suffices, in theory it can fail if the initial DR and N retransmissions are all lost. The sender will give up and release the connection, while the other side knows nothing at all about the attempts to disconnect and is still fully active. This situation results in a half-open connection.

We could have avoided this problem by not allowing the sender to give up after N retries and forcing it to go on forever until it gets a response. However, if the other side is allowed to time out, the sender will indeed go on forever, because no response will ever be forthcoming. If we do not allow the receiving side to time out, the protocol hangs in Fig. 6-14(d).

One way to kill off half-open connections is to have a rule saying that if no segments have arrived for a certain number of seconds, the connection is automatically disconnected. That way, if one side ever disconnects, the other side will detect the lack of activity and also disconnect. This rule also takes care of the case where the connection is broken (because the network can no longer deliver packets between the hosts) without either end disconnecting first. Of course, if this rule is introduced, it is necessary for each transport entity to have a timer that is stopped and then restarted whenever a segment is sent. If this timer expires, a

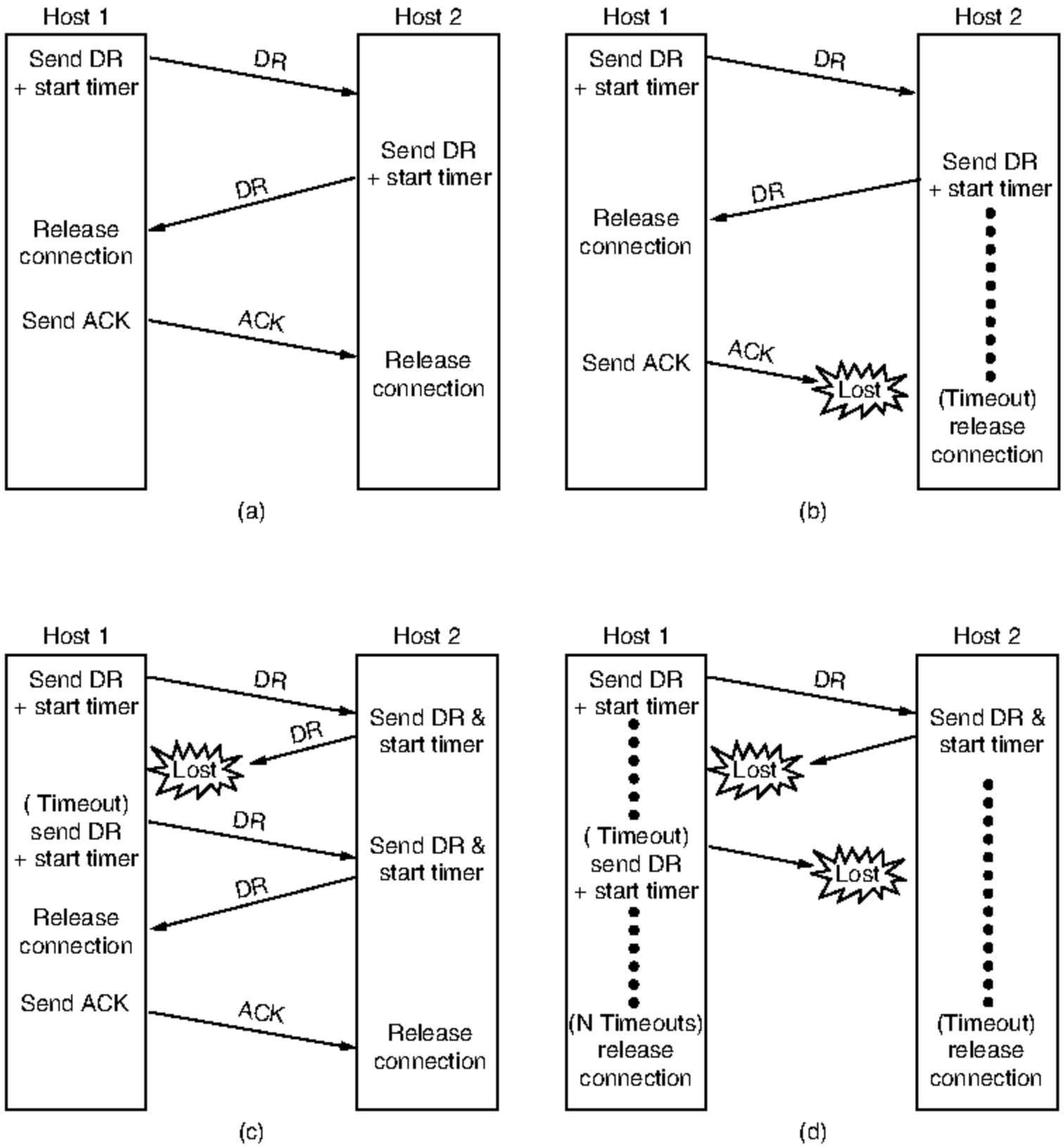


Figure 6-14. Four protocol scenarios for releasing a connection. (a) Normal case of three-way handshake. (b) Final ACK lost. (c) Response lost. (d) Response lost and subsequent DRs lost.

dummy segment is transmitted, just to keep the other side from disconnecting. On the other hand, if the automatic disconnect rule is used and too many dummy segments in a row are lost on an otherwise idle connection, first one side, then the other will automatically disconnect.

We will not belabor this point any more, but by now it should be clear that releasing a connection without data loss is not nearly as simple as it first appears. The lesson here is that the transport user must be involved in deciding when to

disconnect—the problem cannot be cleanly solved by the transport entities themselves. To see the importance of the application, consider that while TCP normally does a symmetric close (with each side independently closing its half of the connection with a FIN packet when it has sent its data), many Web servers send the client a RST packet that causes an abrupt close of the connection that is more like an asymmetric close. This works only because the Web server knows the pattern of data exchange. First it receives a request from the client, which is all the data the client will send, and then it sends a response to the client. When the Web server is finished with its response, all of the data has been sent in either direction. The server can send the client a warning and abruptly shut the connection. If the client gets this warning, it will release its connection state then and there. If the client does not get the warning, it will eventually realize that the server is no longer talking to it and release the connection state. The data has been successfully transferred in either case.

6.2.4 Error Control and Flow Control

Having examined connection establishment and release in some detail, let us now look at how connections are managed while they are in use. The key issues are error control and flow control. Error control is ensuring that the data is delivered with the desired level of reliability, usually that all of the data is delivered without any errors. Flow control is keeping a fast transmitter from overrunning a slow receiver.

Both of these issues have come up before, when we studied the data link layer. The solutions that are used at the transport layer are the same mechanisms that we studied in Chap. 3. As a very brief recap:

1. A frame carries an error-detecting code (e.g., a CRC or checksum) that is used to check if the information was correctly received.
2. A frame carries a sequence number to identify itself and is retransmitted by the sender until it receives an acknowledgement of successful receipt from the receiver. This is called **ARQ** (**Automatic Repeat reQuest**).
3. There is a maximum number of frames that the sender will allow to be outstanding at any time, pausing if the receiver is not acknowledging frames quickly enough. If this maximum is one packet the protocol is called **stop-and-wait**. Larger windows enable pipelining and improve performance on long, fast links.
4. The **sliding window** protocol combines these features and is also used to support bidirectional data transfer.

Given that these mechanisms are used on frames at the link layer, it is natural to wonder why they would be used on segments at the transport layer as well.

However, there is little duplication between the link and transport layers in practice. Even though the same mechanisms are used, there are differences in function and degree.

For a difference in function, consider error detection. The link layer checksum protects a frame while it crosses a single link. The transport layer checksum protects a segment while it crosses an entire network path. It is an end-to-end check, which is not the same as having a check on every link. Saltzer et al. (1984) describe a situation in which packets were corrupted inside a router. The link layer checksums protected the packets only while they traveled across a link, not while they were inside the router. Thus, packets were delivered incorrectly even though they were correct according to the checks on every link.

This and other examples led Saltzer et al. to articulate the **end-to-end argument**. According to this argument, the transport layer check that runs end-to-end is essential for correctness, and the link layer checks are not essential but nonetheless valuable for improving performance (since without them a corrupted packet can be sent along the entire path unnecessarily).

As a difference in degree, consider retransmissions and the sliding window protocol. Most wireless links, other than satellite links, can have only a single frame outstanding from the sender at a time. That is, the bandwidth-delay product for the link is small enough that not even a whole frame can be stored inside the link. In this case, a small window size is sufficient for good performance. For example, 802.11 uses a stop-and-wait protocol, transmitting or retransmitting each frame and waiting for it to be acknowledged before moving on to the next frame. Having a window size larger than one frame would add complexity without improving performance. For wired and optical fiber links, such as (switched) Ethernet or ISP backbones, the error-rate is low enough that link-layer retransmissions can be omitted because the end-to-end retransmissions will repair the residual frame loss.

On the other hand, many TCP connections have a bandwidth-delay product that is much larger than a single segment. Consider a connection sending data across the U.S. at 1 Mbps with a round-trip time of 100 msec. Even for this slow connection, 200 Kbit of data will be stored at the receiver in the time it takes to send a segment and receive an acknowledgement. For these situations, a large sliding window must be used. Stop-and-wait will cripple performance. In our example it would limit performance to one segment every 200 msec, or 5 segments/sec no matter how fast the network really is.

Given that transport protocols generally use larger sliding windows, we will look at the issue of buffering data more carefully. Since a host may have many connections, each of which is treated separately, it may need a substantial amount of buffering for the sliding windows. The buffers are needed at both the sender and the receiver. Certainly they are needed at the sender to hold all transmitted but as yet unacknowledged segments. They are needed there because these segments may be lost and need to be retransmitted.

However, since the sender is buffering, the receiver may or may not dedicate specific buffers to specific connections, as it sees fit. The receiver may, for example, maintain a single buffer pool shared by all connections. When a segment comes in, an attempt is made to dynamically acquire a new buffer. If one is available, the segment is accepted; otherwise, it is discarded. Since the sender is prepared to retransmit segments lost by the network, no permanent harm is done by having the receiver drop segments, although some resources are wasted. The sender just keeps trying until it gets an acknowledgement.

The best trade-off between source buffering and destination buffering depends on the type of traffic carried by the connection. For low-bandwidth bursty traffic, such as that produced by an interactive terminal, it is reasonable not to dedicate any buffers, but rather to acquire them dynamically at both ends, relying on buffering at the sender if segments must occasionally be discarded. On the other hand, for file transfer and other high-bandwidth traffic, it is better if the receiver does dedicate a full window of buffers, to allow the data to flow at maximum speed. This is the strategy that TCP uses.

There still remains the question of how to organize the buffer pool. If most segments are nearly the same size, it is natural to organize the buffers as a pool of identically sized buffers, with one segment per buffer, as in Fig. 6-15(a). However, if there is wide variation in segment size, from short requests for Web pages to large packets in peer-to-peer file transfers, a pool of fixed-sized buffers presents problems. If the buffer size is chosen to be equal to the largest possible segment, space will be wasted whenever a short segment arrives. If the buffer size is chosen to be less than the maximum segment size, multiple buffers will be needed for long segments, with the attendant complexity.

Another approach to the buffer size problem is to use variable-sized buffers, as in Fig. 6-15(b). The advantage here is better memory utilization, at the price of more complicated buffer management. A third possibility is to dedicate a single large circular buffer per connection, as in Fig. 6-15(c). This system is simple and elegant and does not depend on segment sizes, but makes good use of memory only when the connections are heavily loaded.

As connections are opened and closed and as the traffic pattern changes, the sender and receiver need to dynamically adjust their buffer allocations. Consequently, the transport protocol should allow a sending host to request buffer space at the other end. Buffers could be allocated per connection, or collectively, for all the connections running between the two hosts. Alternatively, the receiver, knowing its buffer situation (but not knowing the offered traffic) could tell the sender "I have reserved *X* buffers for you." If the number of open connections should increase, it may be necessary for an allocation to be reduced, so the protocol should provide for this possibility.

A reasonably general way to manage dynamic buffer allocation is to decouple the buffering from the acknowledgements, in contrast to the sliding window protocols of Chap. 3. Dynamic buffer management means, in effect, a variable-sized

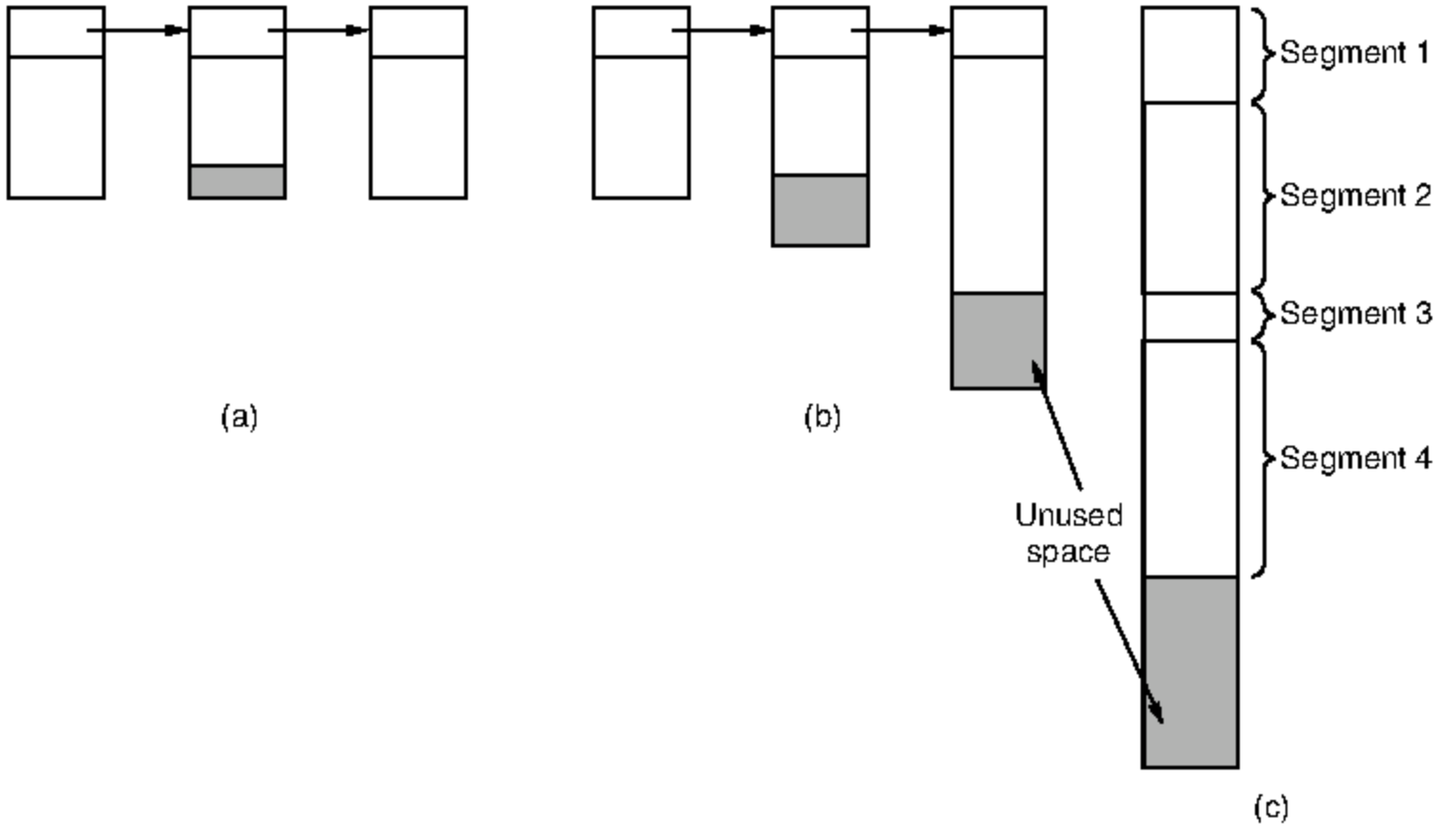


Figure 6-15. (a) Chained fixed-size buffers. (b) Chained variable-sized buffers. (c) One large circular buffer per connection.

window. Initially, the sender requests a certain number of buffers, based on its expected needs. The receiver then grants as many of these as it can afford. Every time the sender transmits a segment, it must decrement its allocation, stopping altogether when the allocation reaches zero. The receiver separately piggybacks both acknowledgements and buffer allocations onto the reverse traffic. TCP uses this scheme, carrying buffer allocations in a header field called *Window size*.

Figure 6-16 shows an example of how dynamic window management might work in a datagram network with 4-bit sequence numbers. In this example, data flows in segments from host *A* to host *B* and acknowledgements and buffer allocations flow in segments in the reverse direction. Initially, *A* wants eight buffers, but it is granted only four of these. It then sends three segments, of which the third is lost. Segment 6 acknowledges receipt of all segments up to and including sequence number 1, thus allowing *A* to release those buffers, and furthermore informs *A* that it has permission to send three more segments starting beyond 1 (i.e., segments 2, 3, and 4). *A* knows that it has already sent number 2, so it thinks that it may send segments 3 and 4, which it proceeds to do. At this point it is blocked and must wait for more buffer allocation. Timeout-induced retransmissions (line 9), however, may occur while blocked, since they use buffers that have already been allocated. In line 10, *B* acknowledges receipt of all segments up to and including 4 but refuses to let *A* continue. Such a situation is impossible with the fixed-window protocols of Chap. 3. The next segment from *B* to *A* allocates

another buffer and allows *A* to continue. This will happen when *B* has buffer space, likely because the transport user has accepted more segment data.

	A	Message	B	Comments
1	→	< request 8 buffers>	→	A wants 8 buffers
2	←	<ack = 15, buf = 4>	←	B grants messages 0-3 only
3	→	<seq = 0, data = m0>	→	A has 3 buffers left now
4	→	<seq = 1, data = m1>	→	A has 2 buffers left now
5	→	<seq = 2, data = m2>	• • •	Message lost but A thinks it has 1 left
6	←	<ack = 1, buf = 3>	←	B acknowledges 0 and 1, permits 2-4
7	→	<seq = 3, data = m3>	→	A has 1 buffer left
8	→	<seq = 4, data = m4>	→	A has 0 buffers left, and must stop
9	→	<seq = 2, data = m2>	→	A times out and retransmits
10	←	<ack = 4, buf = 0>	←	Everything acknowledged, but A still blocked
11	←	<ack = 4, buf = 1>	←	A may now send 5
12	←	<ack = 4, buf = 2>	←	B found a new buffer somewhere
13	→	<seq = 5, data = m5>	→	A has 1 buffer left
14	→	<seq = 6, data = m6>	→	A is now blocked again
15	←	<ack = 6, buf = 0>	←	A is still blocked
16	• • •	<ack = 6, buf = 4>	←	Potential deadlock

Figure 6-16. Dynamic buffer allocation. The arrows show the direction of transmission. An ellipsis (...) indicates a lost segment.

Problems with buffer allocation schemes of this kind can arise in datagram networks if control segments can get lost—which they most certainly can. Look at line 16. *B* has now allocated more buffers to *A*, but the allocation segment was lost. Oops. Since control segments are not sequenced or timed out, *A* is now deadlocked. To prevent this situation, each host should periodically send control segments giving the acknowledgement and buffer status on each connection. That way, the deadlock will be broken, sooner or later.

Until now we have tacitly assumed that the only limit imposed on the sender's data rate is the amount of buffer space available in the receiver. This is often not the case. Memory was once expensive but prices have fallen dramatically. Hosts may be equipped with sufficient memory that the lack of buffers is rarely, if ever, a problem, even for wide area connections. Of course, this depends on the buffer size being set to be large enough, which has not always been the case for TCP (Zhang et al., 2002).

When buffer space no longer limits the maximum flow, another bottleneck will appear: the carrying capacity of the network. If adjacent routers can exchange at most x packets/sec and there are k disjoint paths between a pair of hosts, there is no way that those hosts can exchange more than kx segments/sec, no matter how much buffer space is available at each end. If the sender pushes too hard