Module – 2

Transport Layer

Introduction and Transport-Layer Services

A transport-layer protocol provides for **logical communication** between application processes running on different hosts. By *logical communication*, we mean that from an application's perspective, it is as if the hosts running the processes were directly connected; in reality, the hosts may be on opposite sides of the planet, connected via numerous routers and a wide range of link types.

Application processes use the logical communication provided by the transport layer to send messages to each other, free from the worry of the details of the physical infrastructure used to carry these messages.

Relationship between Transport and Network Layers

Recall that the transport layer lies just above the network layer in the protocol stack. Whereas a transport-layer protocol provides logical communication between *processes* running on different hosts, a network-layer protocol provides logical communication between *hosts*. This distinction is subtle but important. Let's examine this distinction with the aid of a household analogy.

Consider two houses, one on the East Coast and the other on the West Coast, with each house being home to a dozen kids. The kids in the East Coast household are cousins of the kids in the West Coast household. The kids in the two households love to write to each other—each kid writes each cousin every week, with each letter delivered by the traditional postal service in a separate envelope. Thus, each household sends 144 letters to the other household every week. (These kids would save a lot of money if they had e-mail!) In each of the households there is one kid—Ann in the West Coast house and Bill in the East Coast house—responsible for mail collection and mail distribution. Each week Ann visits all her brothers and sisters, collects the mail, and gives the mail to a postal-service mail carrier, who makes daily visits to the house. When letters arrive at the West Coast house, Ann also has the job of distributing the mail to her brothers and sisters. Bill has a similar job on the East Coast.

In this example, the postal service provides logical communication between the two houses—the postal service moves mail from house to house, not from person to person. On the other hand, Ann and Bill provide logical communication among the cousins—Ann and Bill pick up mail from, and deliver mail to, their brothers and sisters.

Note that from the cousins' perspective, Ann and Bill *are* the mail service, even though Ann and Bill are only a part (the end-system part) of the end-to-end delivery process. This household example serves as a nice analogy for explaining how the transport layer relates to the network layer:

application messages = letters in envelopes processes = cousins
hosts (also called end systems) = houses
transport-layer protocol = Ann and Bill network-layer protocol
= postal service (including mail carriers)

Overview of the Transport Layer in the Internet

Recall that the Internet, and more generally a TCP/IP network, makes two distinct transport-layer protocols available to the application layer. One of these protocols is **UDP** (User Datagram Protocol), which provides an unreliable, connectionless service to the invoking application. The second of these protocols is **TCP** (Transmission Control Protocol), which provides a reliable, connection-oriented service to the invoking application. When designing a network application, the application developer must specify one of these two transport protocols. As we saw in Section 2.7, the application developer selects between UDP and TCP when creating sockets.

To simplify terminology, when in an Internet context, we refer to the transport layer packet as a *segment*. We mention, however, that the Internet literature (for example, the RFCs) also refers to the transport-layer packet for TCP as a segment but often refers to the packet for UDP as a datagram. But this same Internet literature also uses the term *datagram* for the network-layer packet! For an introductory book on computer networking such as this, we believe that it is less confusing to refer to both TCP and UDP packets as segments, and reserve the term *datagram* for the network-layer packet. Before proceeding with our brief introduction of UDP and TCP, it will be useful to say a few words about the Internet's network layer.

The IP service model is a **best-effort delivery service**. This means that IP makes its "best effort" to deliver segments between communicating hosts, *but it makes noguarantees*. In particular, it does not guarantee segment delivery, it does not guarantee orderly delivery of segments, and it does not guarantee the integrity of the datain the segments. For these reasons, IP is said to be an **unreliable service**.

Extending host-to-host delivery to process-to-process delivery is called **transport-layer multiplexing** and **demultiplexing**.

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Multiplexing and Demultiplexing

In this section, we discuss transport-layer multiplexing and demultiplexing, that is, extending the host-to-host delivery service provided by the network layer to a process-to-process delivery service for applications running on the hosts. In order to keep the discussion concrete, we'll discuss this basic transport-layer service in the context of the Internet. We emphasize, however, that a multiplexing/demultiplexing service is needed for all computer networks. At the destination host, the transport layer receives segments from the network layer just below. The transport layer has the responsibility of delivering the data in these segments to the appropriate application process running in the host. Let's take a look at an example. Suppose you are sitting in front of your computer, and you are downloading Web pages while running one FTP session and two Telnet sessions.

Now let's consider how a receiving host directs an incoming transport-layer segment to the appropriate socket. Each transport-layer segment has a set of fields in the segment for this purpose. At the receiving end, the transport layer examines these fields to identify the receiving socket and then directs the segment to that socket. This job of delivering the data in a transport-layer segment to the correct socket is called **demultiplexing**. The job of gathering data chunks at the source host from different sockets, encapsulating each data chunk with header information (that will later be used in demultiplexing) to create segments, and passing the segments to the network layer is called **multiplexing**.

Note that the transport layer in the middle host in Figure 2.2 must demultiplex segments arriving from the network layer below to either process P1 or P2 above; this is done by directing the arriving segment's data to the corresponding process's socket. The transport layer in the middle host must also gather outgoing data from these sockets, form transport-layer segments, and pass these segments down to the network layer.

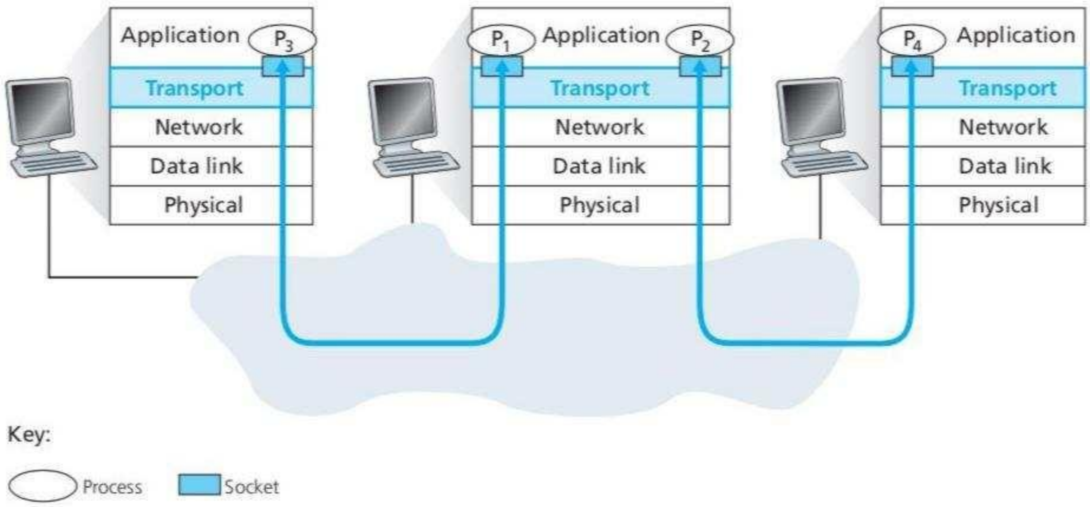


Figure 2.1 Transport-layer multiplexing and demultiplexing

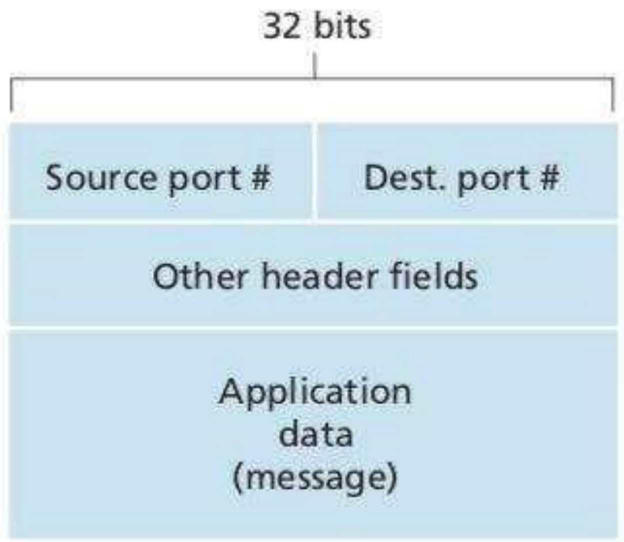


Figure 2.3 Source and destination port-number fields in a transport-layer Segment

Connectionless Multiplexing and Demultiplexing

Recall from Section that the Python program running in a host can create a UDP socket with the line

clientSocket = socket(socket.AF_INET, socket.SOCK_DGRAM)

When a UDP socket is created in this manner, the transport layer automatically assigns a port number to the socket. In particular, the transport layer assigns a port number in the range 1024 to 65535 that is currently not being used by any other UDP port in the host. Alternatively, we can add a line into our Python program after we create the socket to associate a specific port number (say, 19157) to this UDP socket via the socket bind() method:

clientSocket.bind((‘’, 19157))

If the application developer writing the code were implementing the server side of a “well-known protocol,” then the developer would have to assign the corresponding well-known port number. Typically, the client side of the application lets the transport layer automatically (and transparently) assign the port number, whereas the server side of the application assigns a specific port number.

It is important to note that a UDP socket is fully identified by a two-tuple consisting of a destination IP address and a destination port number. As a consequence, if two UDP segments have different source IP addresses and/or source port numbers, but have the same *destination* IP address and *destination* port number, then the two segments will be directed to the same destination process via the same destination socket.

Connection-Oriented Multiplexing and Demultiplexing

In order to understand TCP demultiplexing, we have to take a close look at TCP sockets and TCP connection establishment. One subtle difference between a TCP socket and a UDP socket is that a TCP socket is identified by a four-tuple: (source IP address, source port number, destination IP address, destination port number). Thus, when a TCP segment arrives from the network to a host, the host uses all four values to direct (demultiplex) the segment to the appropriate socket. In particular, and in contrast with UDP, two arriving TCP segments with different source IP addresses or source port numbers will (with the exception of a TCP segment carrying the original connection-establishment request) be directed to two different sockets.

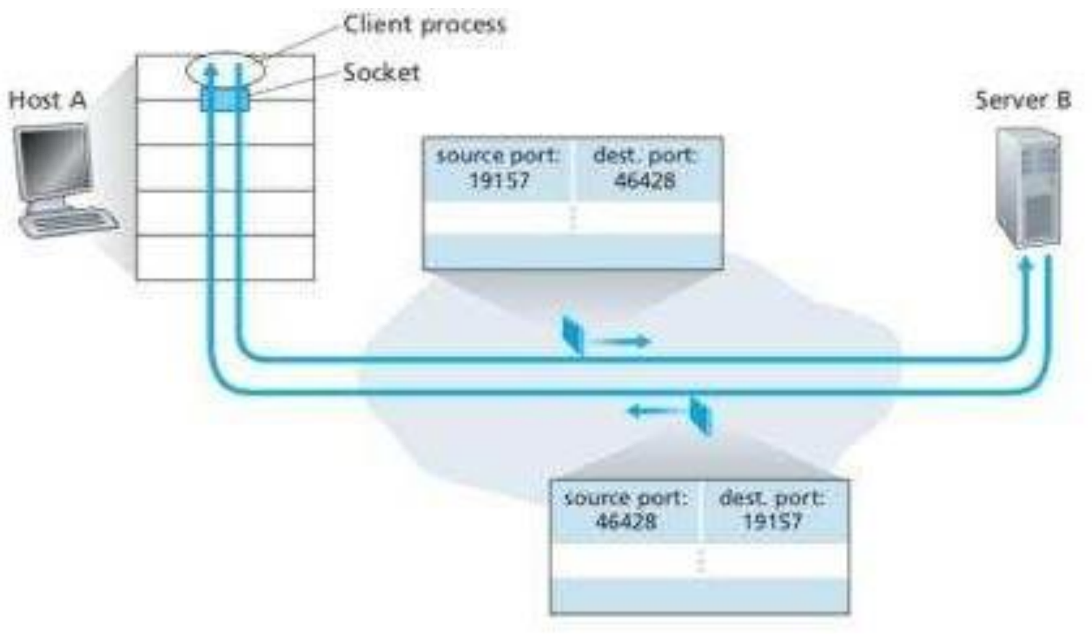


Figure 2.4 The inversion of source and destination port numbers

• The TCP server application has a "welcoming socket," that waits for connection establishment requests from TCP clients on port number 12000.

• The TCP client creates a socket and sends a connection establishment request segment with the lines:

clientSocket = socket(AF_INET, SOCK_STREAM)

clientSocket.connect((serverName,12000))

• Aconnection-establishment request is nothing more than a TCP segment with destination port number 12000 and a special connection-establishment bit set in the TCP header (discussed in Section 3.5). The segment also includes a source port number that was chosen by the client.

• When the host operating system of the computer running the server process receives the incoming connection-request segment with destination port 12000, it locates the server process that is waiting to accept a connection on port number 12000. The server process then creates a new socket: connectionSocket, addr = serverSocket.accept()

• Also, the transport layer at the server notes the following four values in the connection- request segment: (1) the source port number in the segment, (2) the IP address of the source host, (3) the destination port number in the segment, and (4) its own IP address. The newly created connection socket is identified by these four values; all subsequently arriving segments whose source port, source IP address, destination port, and destination IP address match these four values will be demultiplexed to this socket. With the TCP connection now in place, the client and server can now send data to each other.

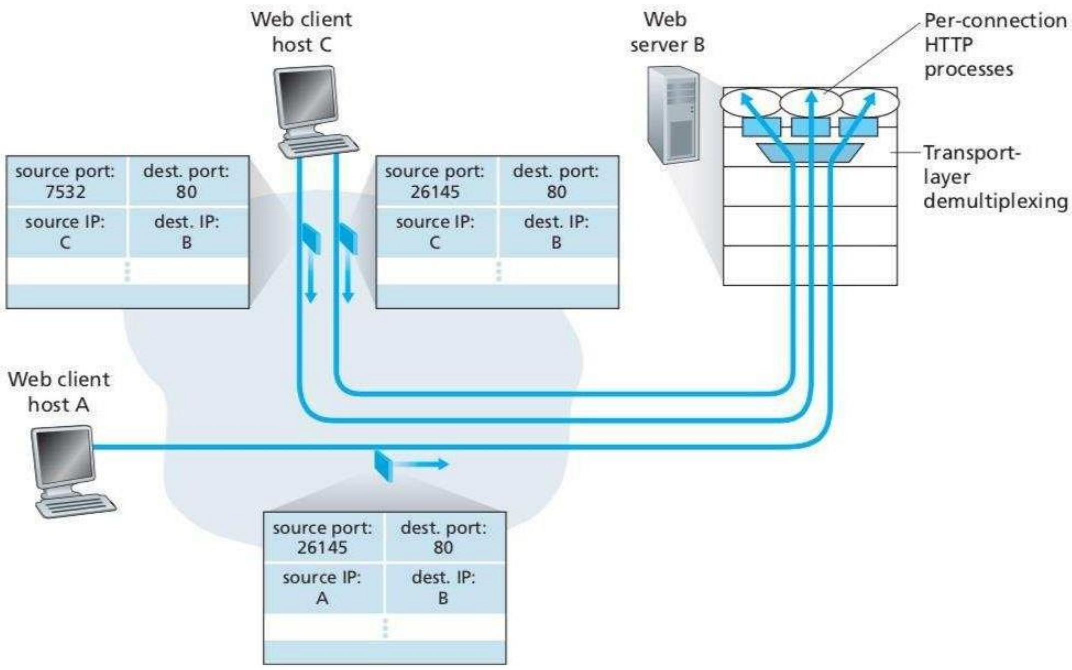


Figure 2.5 Two clients, using the same destination port number (80) to communicate with the same Web server application

Web Servers and TCP

Figure 3.5 shows a Web server that spawns a new process for each connection. As shown in Figure 3.5, each of these processes has its own connection socket through which HTTP requests arrive and HTTP responses are sent. We mention, however, that there is not always a one-to-one correspondence between connection sockets and processes. In fact, today's high-performing Web servers often use only one process, and create a new thread with a new connection socket for each new client connection. (A thread can be viewed as a lightweight subprocess.) If you did the first programming assignment in Chapter 2, you built a Web server that does just this. For such a server, at any given time there may be many connection sockets (with different identifiers) attached to the same process.

If the client and server are using persistent HTTP, then throughout the duration of the persistent connection the client and server exchange HTTP messages via the same server socket. However, if the client and server use non-persistent HTTP, then a new TCP connection is created and closed for every request/response, and hence a new socket is created and later closed for every request/response. This frequent creating and closing of sockets can severely impact the performance of a busy Web server

Connectionless Transport: UDP

UDP, defined in [RFC 768], does just about as little as a transport protocol can do. Aside from the multiplexing/demultiplexing function and some light error checking, it adds nothing to IP. In fact, if the application developer chooses UDP instead of TCP, then the application is almost directly talking with IP. UDP takes messages from the application process, attaches source and destination port number fields for the multiplexing/demultiplexing service, adds two other small fields, and passes the resulting segment to the network layer. The network layer encapsulates the transport-layer segment into an IP datagram and then makes a best-effort attempt to deliver the segment to the receiving host. If the segment arrives at the receiving host, UDP uses the destination port number to deliver the segment's data to the correct application process. Note that with UDP there is no handshaking between sending and receiving transport-layer entities before sending a segment. For this reason, UDP is said to be *connectionless.*

DNS is an example of an application-layer protocol that typically uses UDP. When the DNS application in a host wants to make a query, it constructs a DNS query message and passes the message to UDP. Without performing any handshaking with the UDP entity running on the destination end system, the host-side UDP adds header fields to the message and passes the resulting segment to the network layer. The network layer encapsulates the UDP segment into a datagram and sends the datagram to a name server. The DNS application at the querying host then waits for a reply to its query. If it doesn't receive a reply (possibly because the underlying network lost the query or the reply), either it tries sending the query to another name server, or it informs the invoking application that it can't get a reply.

Many applications are better suited for UDP for the following reasons:

• ***Finer application-level control over what data is sent, and when.*** Under UDP, as soon as an application process passes data to UDP, UDP will package the data inside a UDP segment and immediately pass the segment to the network layer. TCP, on the other hand, has a congestion-control mechanism that throttles the transport-layer TCP sender when one or more links between the source and destination hosts become excessively congested. TCP will also continue to resend a segment until the receipt of the segment has been acknowledged by the destination, regardless of how long reliable delivery takes. Since real-time applications often require a minimum sending rate, do not want to overly delay segment transmission, and can tolerate some data loss, TCP's service model is not particularly well matched to these applications' needs.

• ***No connection establishment.*** As we'll discuss later, TCP uses a three-way handshake before it starts to transfer data. UDP just blasts away without any formal preliminaries. Thus UDP does not introduce any delay to establish a connection. This is probably the principal reason why DNS runs over UDP rather than TCP—DNS would be much slower if it ran over TCP. HTTP uses TCP rather than UDP, since reliability is critical for Web pages with text. But, as we briefly discussed in Section 2.2, the TCP connection-establishment delay in HTTP is an important contributor to the delays associated with downloading Web documents.

• ***No connection state.*** TCP maintains connection state in the end systems. This connection state includes receive and send buffers, congestion-control parameters, and sequence and acknowledgment number parameters. We will see in Section 3.5 that this state information is needed to implement TCP's reliable data transfer service and to provide congestion control. UDP, on the other hand, does not maintain connection state and does not track any of these parameters. For this reason, a server devoted to a particular application can typically support many more active clients when the application runs over UDP rather than TCP.

• ***Small packet header overhead.*** The TCP segment has 20 bytes of header overhead in every segment, whereas UDP has only 8 bytes of overhead.

UDP Segment Structure

The UDP segment structure, shown in Figure 3.7, is defined in RFC 768. The application data occupies the data field of the UDP segment. For example, for DNS, the data field contains either a query message or a response message. For a streaming audio application, audio samples fill the data field. The UDP header has only four fields, each consisting of two bytes. As discussed in the previous section, the port numbers allow the destination host to pass the application data to the correct process running on the destination end system (that is, to perform the demultiplexing function). The length field specifies the number of bytes in the UDP segment (header plus data). An explicit length value is needed since the size of the data field may differ from one UDP segment to the next. The checksum is used by the receiving host to check whether errors have been introduced into the segment. In truth, the checksum is also calculated over a few of the fields in the IP header in addition to the UDP segment. The length field specifies the length of the UDP segment, including the header, in bytes.

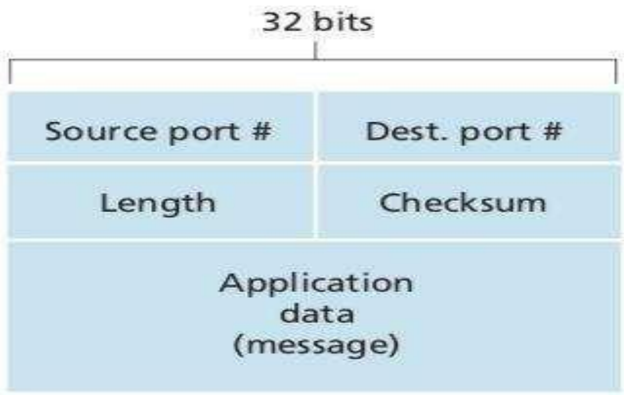


Figure 2.7 _ UDP segment structure

UDP Checksum

The UDP checksum provides for error detection. That is, the checksum is used to determine whether bits within the UDP segment have been altered (for example, by noise in the links or while stored in a router) as it moved from source to destination. UDP at the sender side performs the 1s complement of the sum of all the 16-bit words in the segment, with any overflow encountered during the sum being wrapped around. This result is put in the checksum field of the UDP segment.

Here we give a simple example of the checksum calculation. You can find details about efficient implementation of the calculation in RFC 1071 and performance over real data in [Stone 1998; Stone 2000]. As an example, suppose that we have the following three 16-bit words:

0110011001100000

0101010101010101

1000111100001100

The sum of first two of these 16-bit words is

0110011001100000

0101010101010101

1011101110110101

Adding the third word to the above sum gives

1011101110110101

1000111100001100

0100101011000010

Note that this last addition had overflow, which was wrapped around. The 1s complement is obtained by converting all the 0s to 1s and converting all the 1s to 0s. Thus the 1s complement of the sum 0100101011000010 is 1011010100111101, which becomes the checksum. At the receiver, all four 16-bit words are added, including the checksum. If no errors are introduced into the packet, then clearly the sum at the receiver will be 1111111111111111. If one of the bits is a 0, then we know that errors have been introduced into the packet.

Principles of Reliable Data Transfer

It is the responsibility of a **reliable data transfer protocol** to implement this service abstraction. This task is made difficult by the fact that the layer *below* the reliable data transfer protocol may be unreliable. For example, TCP is a reliable data transfer protocol that is implemented on top of an unreliable (IP) end-to-end network layer. More generally, the layer beneath the two reliably communicating end points might consist of a single physical link (as in the case of a link-level data transfer protocol) or a global internetwork (as in the case of a transport-level protocol). For our purposes, however, we can view this lower layer simply as an unreliable point-to-point channel.

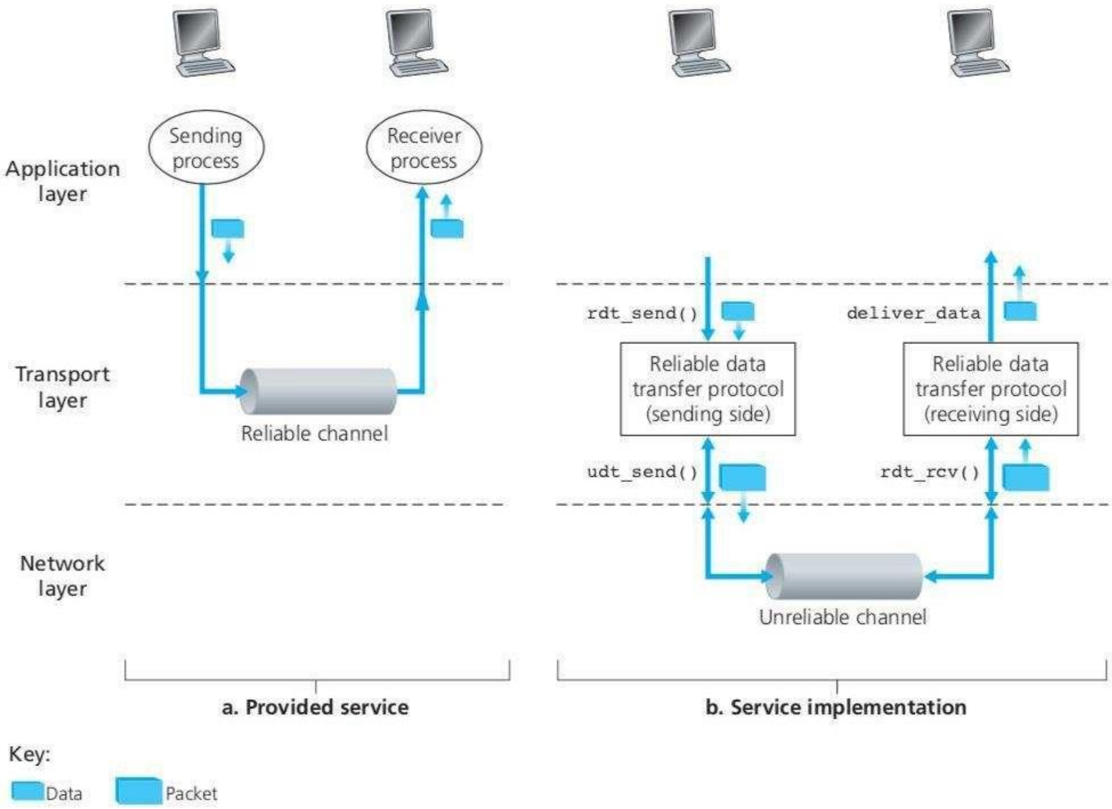


Figure 2.8 _ Reliable data transfer: Service model and service Implementation

Building a Reliable Data Transfer Protocol

Reliable Data Transfer over a Perfectly Reliable Channel: rdt1.0

The **finite-state machine (FSM)** definitions for the rdt1.0 sender and receiver are shown in Figure 2.9. The FSM in Figure 2.9(a) defines the operation of the sender, while the FSM in Figure 2.9(b) defines the operation of the receiver. It is important to note that there are *separate* FSMs for the sender and for the receiver. The sender and receiver FSMs in Figure 2.9 each have just one state. The arrows in the

FSM description indicate the transition of the protocol from one state to another. (Since each FSM in Figure 2.9 has just one state, a transition is necessarily from the one state back to itself; we'll see more complicated state diagrams shortly.) The event causing the transition is shown above the horizontal line labeling the transition, and the actions taken when the event occurs are shown below the horizontal line.

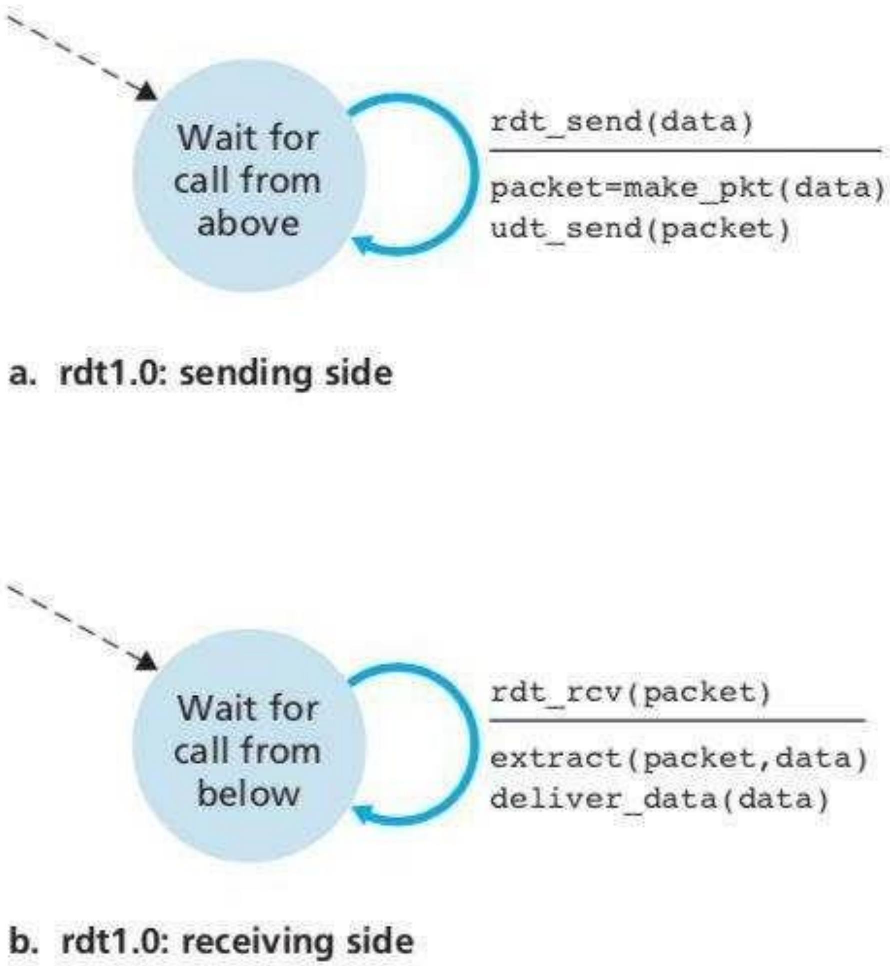


Figure 2.9 _ rdt1.0 – A protocol for a completely reliable channel

When no action is taken on an event, or no event occurs and an action is taken, we'll use the symbol _ below or above the horizontal, respectively, to explicitly denote the lack of an action or event. The initial state of the FSM is indicated by the dashed arrow. Although the FSMs in Figure 3.9 have but one state, the FSMs we will see shortly have multiple states, so it will be important to identify the initial state of each FSM.

The sending side of rdt simply accepts data from the upper layer via the rdt_send(data) event, creates a packet containing the data (via the action make_pkt(data)) and sends the packet into the channel. In practice, the rdt_send(data) event would result from a procedure call (for example, to rdt_send()) by the upper-layer application. On the receiving side, rdt receives a packet from the underlying channel via the rdt_rcv(packet) event, removes the data from the packet (via the action extract (packet, data)) and passes the data up to the upper layer (via the action deliver_data(data)). In practice, the rdt_rcv(packet) event would result from a procedure call (for example, to rdt_rcv()) from the lowerlayer protocol.

Reliable Data Transfer over a Channel with Bit Errors: rdt2.0

Fundamentally, three additional protocol capabilities are required in ARQ protocols to handle the presence of bit errors:

• ***Error detection.*** First, a mechanism is needed to allow the receiver to detect when bit errors have occurred. Recall from the previous section that UDP uses the Internet checksum field for exactly this purpose.

• ***Receiver feedback.*** Since the sender and receiver are typically executing on different end systems, possibly separated by thousands of miles, the only way for the sender to learn of the receiver's view of the World The positive (ACK) and negative (NAK) acknowledgment replies in the message-dictation scenario are examples of such feedback. Our rdt2.0 protocol will similarly send ACK and NAK packets back from the receiver to the sender. In principle, these packets need only be one bit long; for example, a 0 value could indicate a NAK and a value of 1 could indicate an ACK.

• ***Retransmission.*** A packet that is received in error at the receiver will be retransmitted by the sender.

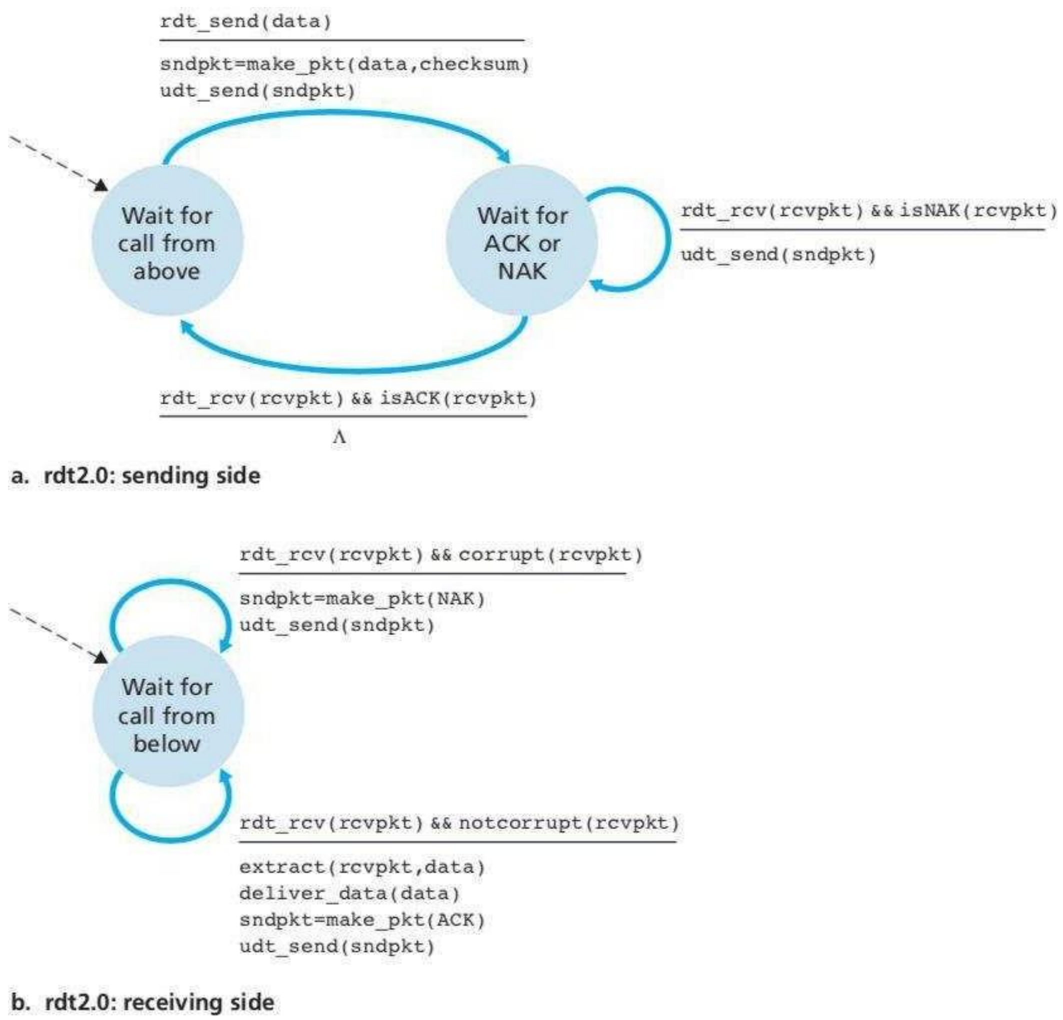


Figure 2.10 _ rdt2.0–A protocol for a channel with bit errors

Figure 2.10 shows the FSM representation of rdt2.0, a data transfer protocol employing error detection, positive acknowledgments, and negative acknowledgments. The send side of rdt2.0 has two states. In the leftmost state, the send-side protocol is waiting for data to be passed down from the upper layer. When the rdt_send(data) event occurs, the sender will create a packet (sndpkt) containing the data to be sent, along with a packet checksum and then send the packet via the udt_send(sndpkt) operation.

In the rightmost state, the sender protocol is waiting for an ACK or a NAK packet from the receiver. If an ACK packet is received (the notation rdt_rcv(rcvpkt) && isACK (rcvpkt) in Figure 2.10 corresponds to this event), the sender knows that the most recently transmitted packet has been received correctly and thus the protocol returns to the state of waiting for data from the Upper layer.

If a NAK is received, the protocol retransmits the last packet and waits for an ACK or NAK to be returned by the receiver in response to the retransmitted data packet. It is important to note that when the sender is in the wait-for-ACK-or-NAK state, it *cannot* get more data from the upper layer; that is, the rdt_send() event can not occur; that will happen only after the sender receives an ACK and leaves this state.

Thus, the sender will not send a new piece of data until it is sure that the receiver has correctly received the current packet. Because of this behavior, protocols such as rdt2.0 are known as **stop-and-wait** protocols. The receiver-side FSM for rdt2.0 still has a single state. On packet arrival, the receiver replies with either an ACK or a NAK, depending on whether or not the received packet is corrupted. In Figure 3.10, the notation rdt_rcv(rcvpkt) && corrupt(rcvpkt) corresponds to the event in which a packet is received and is found to be in error.

Consider three possibilities for handling corrupted ACKs or NAKs:

• For the first possibility, consider what a human might do in the message dictation scenario. If the speaker didn't understand the "OK" or "Please repeat that" reply from the receiver, the speaker would probably ask, "What did you say?" The receiver would then repeat the reply. But what if the speaker's "What did you say?" is corrupted? The receiver, having no idea whether the garbled sentence was part of the dictation or a request to repeat the last reply, would probably then respond with "What did *you* say?" And then, of course, that response might be garbled.

Clearly, we're heading down a difficult path.

• A second alternative is to add enough checksum bits to allow the sender not only to detect, but also to recover from, bit errors. This solves the immediate problem for a channel that can corrupt packets but not lose them.

• A third approach is for the sender simply to resend the current data packet when it receives a garbled ACK or NAK packet. This approach, however, introduces **duplicate packets** into the sender-to-receiver channel.

The fundamental difficulty with duplicate packets is that the receiver doesn't know whether the ACK or NAK it last sent was received correctly at the sender. Thus, it cannot know *a priori* whether an arriving packet contains new data or is a retransmission!

Figures 2.11 and 2.12 show the FSM description for rdt2.1, our fixed version of rdt2.0. The rdt2.1 sender and receiver FSMs each now have twice as many states as before. This is because the protocol state must now reflect whether the packet currently being sent (by the sender) or expected (at the receiver) should have a sequence number of 0 or 1. Note that the actions in those states where a 0-numbered packet is being sent or expected are mirror images of those where a 1-numbered packet is being sent or expected; the only differences have to do with the handling of the sequence number.

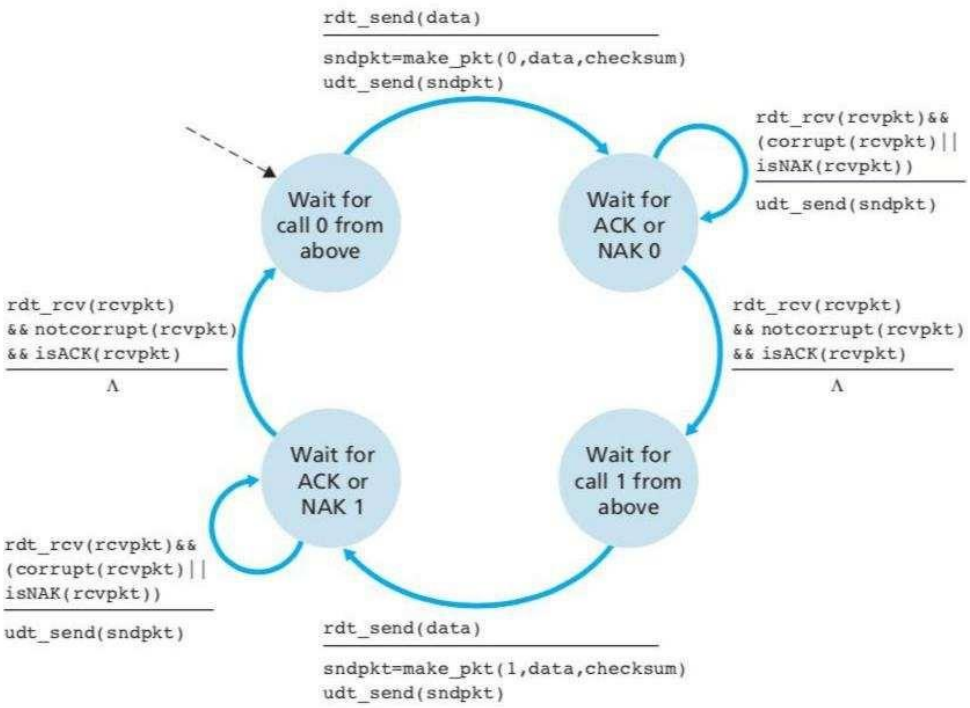


Figure 2.11 _ rdt2.1 sender

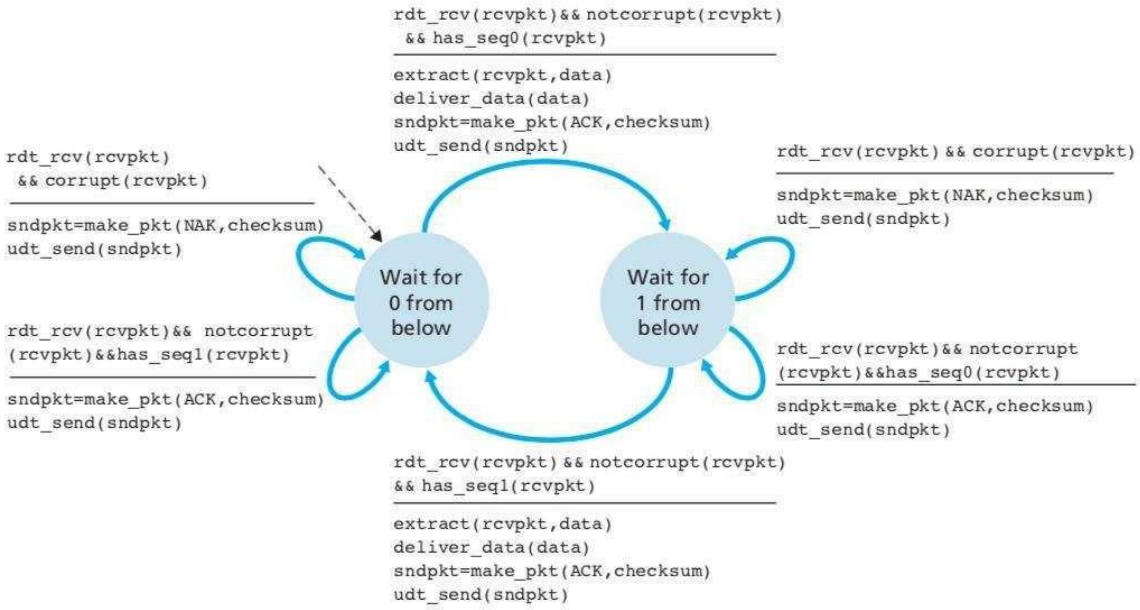


Figure 2.12 _ rdt2.1 receiver

Reliable Data Transfer over a Lossy Channel with Bit Errors: rdt3.0

Two additional concerns must now be addressed by the protocol: how to detect packet loss and what to do when packet loss occurs. The use of checksumming, sequence numbers, ACK packets, and retransmissions—the techniques already developed in rdt2.2—will allow us to answer the latter concern. Handling the first concern will require adding a new protocol mechanism. There are many possible approaches toward dealing with packet loss.

In all cases, the action is the same: retransmit. Implementing a time-based retransmission mechanism requires a **countdown timer** that can interrupt the sender after a given amount of time has expired. The sender will thus need to be able to

(1) start the timer each time a packet (either a first-time packet or a retransmission) is sent,

(2) respond to a timer interrupt (taking appropriate actions), and

(3) stop the timer.

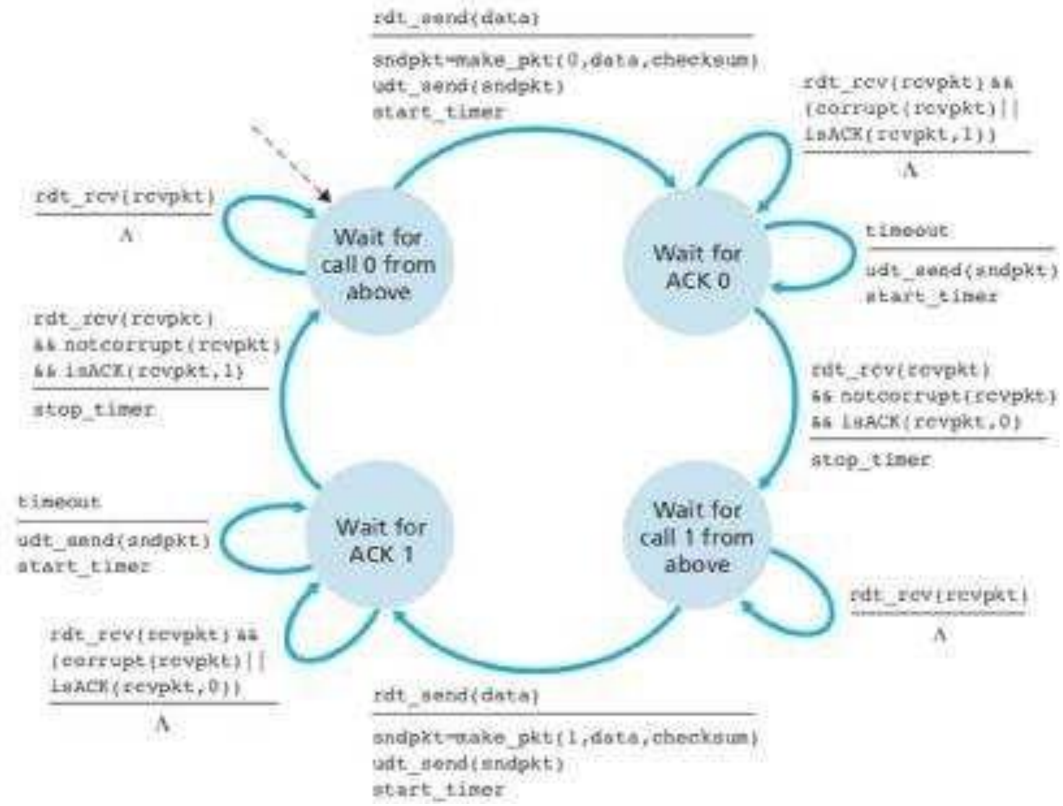


Figure 2.15 _ rdt3.0 sender

Figure 2.15 shows the sender FSM for rdt3.0, a protocol that reliably transfers data over a channel that can corrupt or lose packets; in the homework problems, you’ll be asked to provide the receiver FSM for rdt3.0. Figure 2.16 shows how the protocol operates with no lost or delayed packets and how it handles lost data packets. In Figure 2.16, time moves forward from the top of the diagram toward the bottom of the diagram; note that a receive time for a packet is necessarily later than the send time for a packet as a result of transmission and propagation delays. In Figures 2.16(b)–(d), the send-side brackets indicate the times at which a timer is set and later times out.

Several of the more subtle aspects of this protocol are explored in the exercises at the end of this chapter. Because packet sequence numbers alternate between 0 and 1, protocol rdt3.0 is sometimes known as the **alternating-bit protocol**.

Pipelined Reliable Data Transfer Protocols

Protocol rdt3.0 is a functionally correct protocol, but it is unlikely that anyone would be happy with its performance, particularly in today's high-speed networks. At the heart of rdt3.0's performance problem is the fact that it is a stop-and-wait protocol. To appreciate the performance impact of this stop-and-wait behavior, consider an idealized case of two hosts, one located on the West Coast of the United States and the other located on the East Coast, as shown in Figure 2.17.

The speed-of-light round-trip propagation delay between these two end systems, RTT, is approximately 30 milliseconds. Suppose that they are connected by a channel with a transmission rate, *R*, of 1 Gbps (109 bits per second). With a packet size, *L*, of 1,000 bytes (8,000 bits) per packet, including both header fields and data, the time needed to actually transmit the packet into the 1 Gbps link is *dtrans*

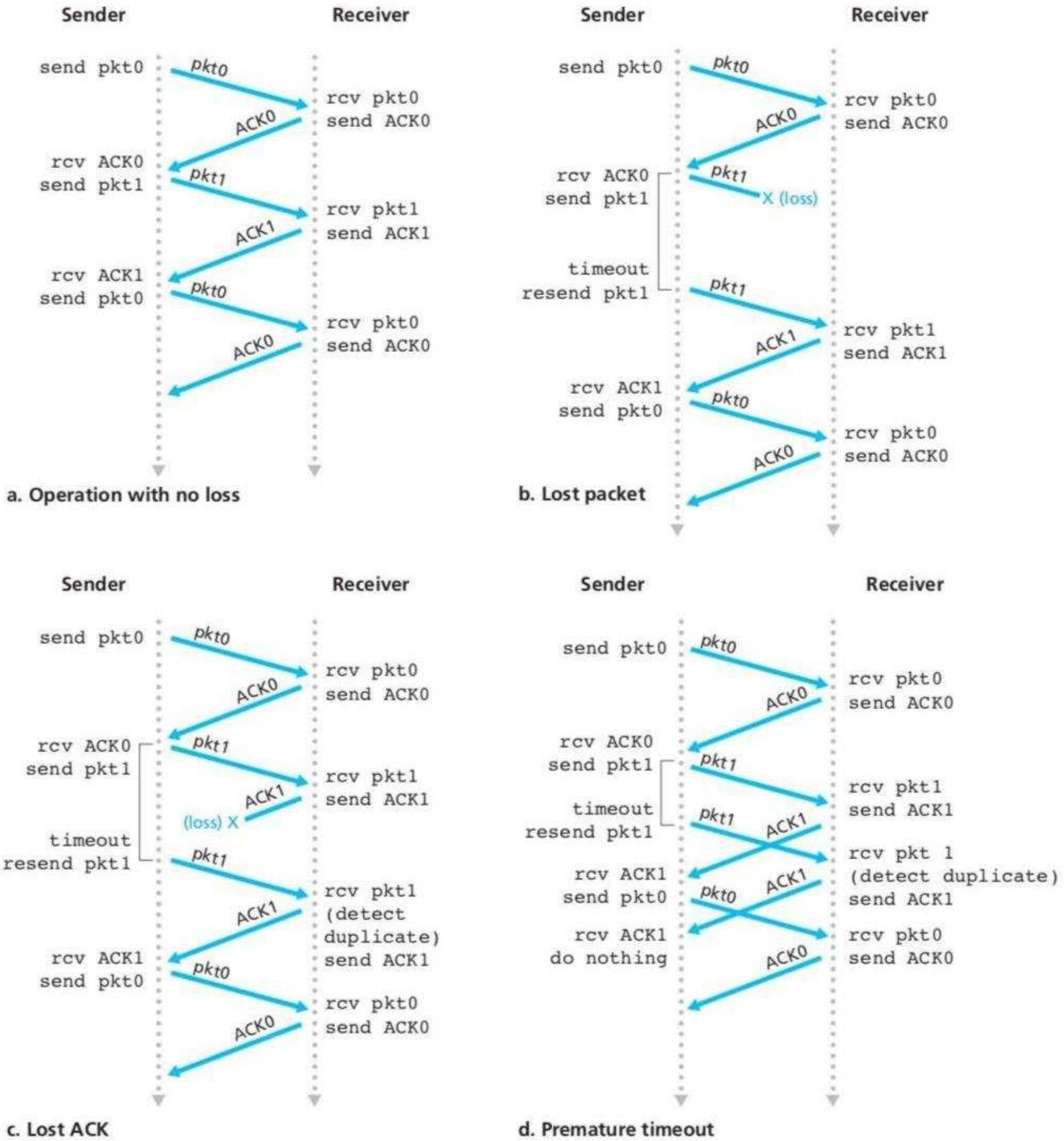


Figure 2.16 _ Operation of rdt3.0, the alternating-bit protocol

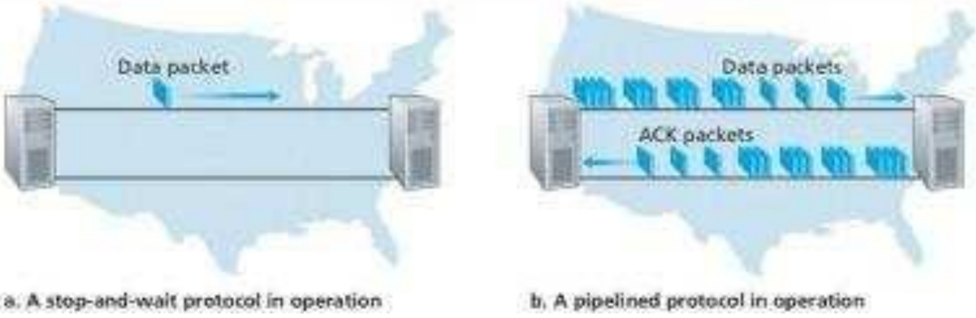


Figure 2.17 _ Stop-and-wait versus pipelined protocol

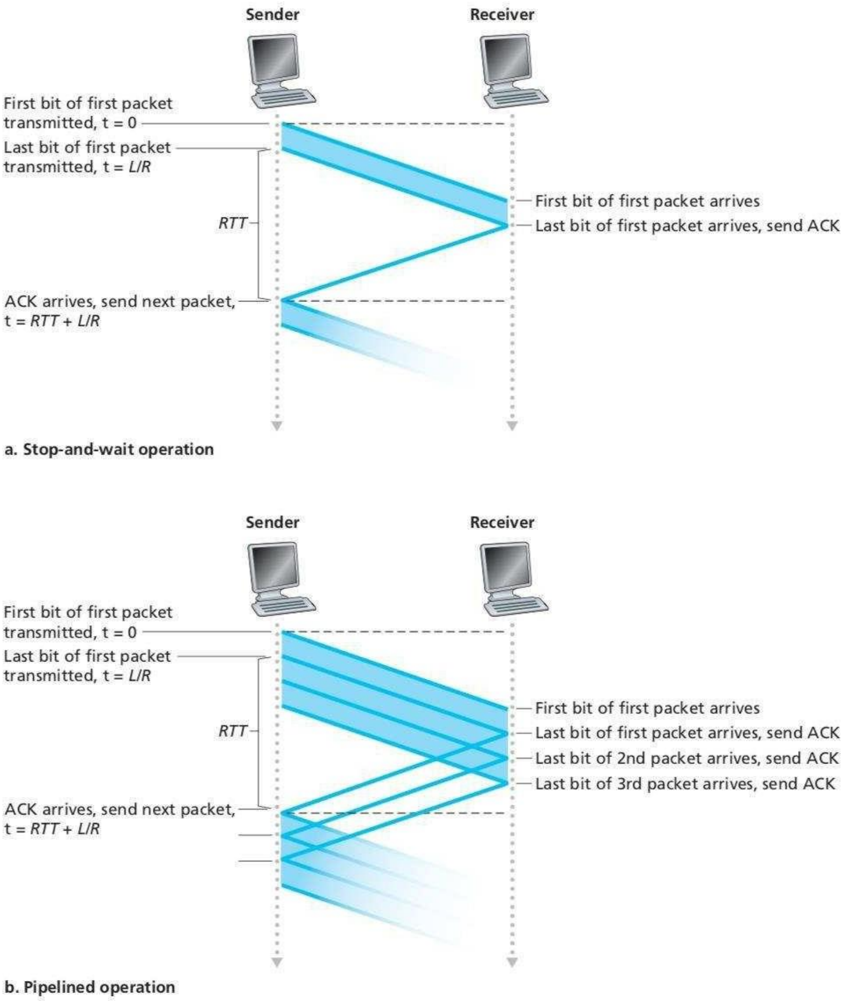


Figure 2.18 _ Stop-and-wait and pipelined sending

Figure 2.18(a) shows that with our stop-and-wait protocol, if the sender begins sending the packet at $t = 0$, then at $t = L/R = 8$ microseconds, the last bit enters the channel at the sender side. The packet then makes its 15-msec cross-country journey, with the last bit of the packet emerging at the receiver at $t = \text{RTT}/2 + L/R = 15.008$ msec.

The solution to this particular performance problem is simple: Rather than operate in a stop-and-wait manner, the sender is allowed to send multiple packets without waiting for acknowledgments, as illustrated in Figure 3.17(b). Figure 3.18(b) shows that if the sender is allowed to transmit three packets before having to wait for acknowledgments, the utilization of the sender is essentially tripled. Since the many in-transit sender-to-receiver packets can be visualized as filling a pipeline, this technique is known as **pipelining**.

Pipelining has the following consequences for reliable data transfer protocols:

• The range of sequence numbers must be increased, since each in-transit packet (not counting retransmissions) must have a unique sequence number and there may be multiple, in-transit, unacknowledged packets.

• The sender and receiver sides of the protocols may have to buffer more than one packet. Minimally, the sender will have to buffer packets that have been transmitted but not yet acknowledged. Buffering of correctly received packets may also be needed at the receiver, as discussed below.

• The range of sequence numbers needed and the buffering requirements will depend on the manner in which a data transfer protocol responds to lost, corrupted, and overly delayed packets. Two basic approaches toward pipelined error recovery can be identified: **Go-Back-N** and **selective repeat**.

Go-Back-N (GBN)

In a **Go-Back-N (GBN) protocol**, the sender is allowed to transmit multiple packets without waiting for an acknowledgment, but is constrained to have no more than some maximum allowable number, *N*, of unacknowledged packets in the pipeline.

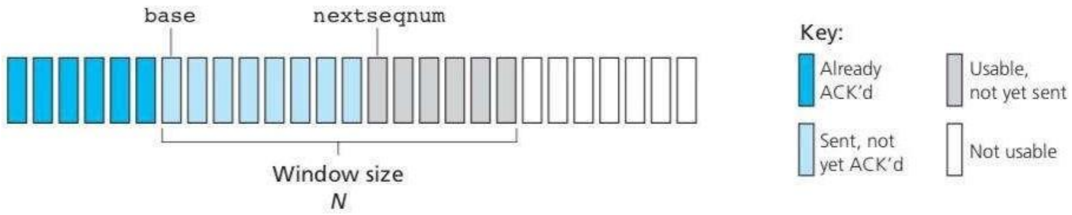
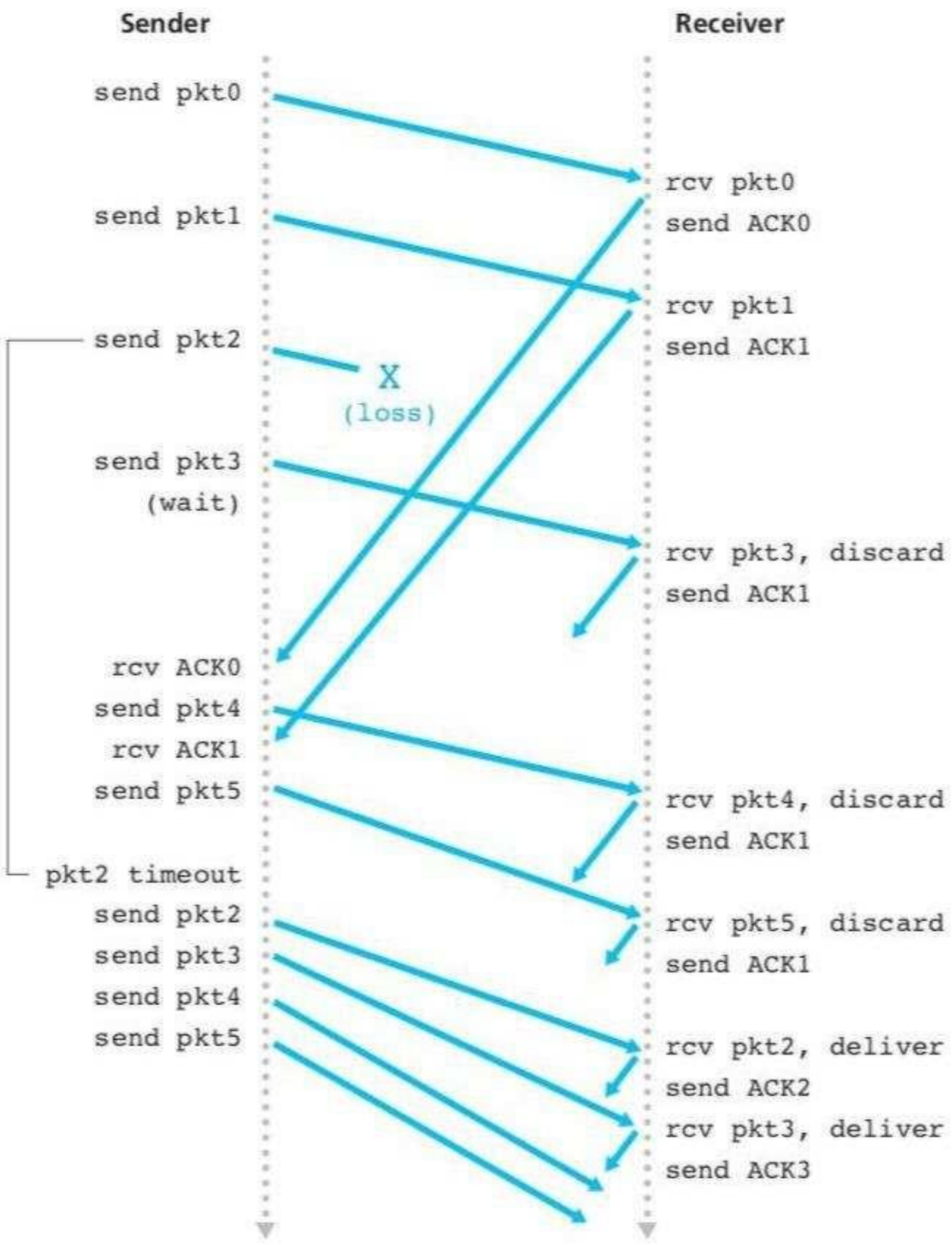


Figure 2.19 _ Sender's view of sequence numbers in Go-Back-N

Figure 2.19 shows the sender's view of the range of sequence numbers in a GBN protocol. If we define base to be the sequence number of the oldest unacknowledged packet and nextseqnum to be the smallest unused sequence number then four intervals in the range of sequence numbers can be identified. Sequence numbers in the interval [0,base-1] correspond to packets that have already been transmitted and acknowledged. The interval [base,nextseqnum-1] corresponds to packets that have been sent but not yet acknowledged. Sequence numbers in the interval [nextseqnum,base+N-1] can be used for packets that can be sent immediately, should data arrive from the upper layer.

Finally, sequence numbers greater than or equal to base+N cannot be used until an unacknowledged packet currently in the pipeline (specifically, the packet with sequence number base) has been acknowledged. As suggested by Figure 2.19, the range of permissible sequence numbers for transmitted but not yet acknowledged packets can be viewed as a window of size *N* over the range of sequence numbers. As the protocol operates, this window slides forward over the sequence number space. For this reason, *N* is often referred to as the **window size** and the GBN protocol itself as a **sliding-window protocol**.


Sender
Receiver
send pkt0
rcv pkt0
send ACK0
send pkt1
rcv pkt1
send ACK1
send pkt2
X
(loss)
send pkt3
(wait)
rcv pkt3, discard
send ACK1
rcv ACK0
send pkt4
rcv ACK1
send pkt5
rcv pkt4, discard
send ACK1
pkt2 timeout
send pkt2
send pkt3
send pkt4
send pkt5
rcv pkt5, discard
send ACK1
rcv pkt2, deliver
send ACK2
rcv pkt3, deliver
send ACK3

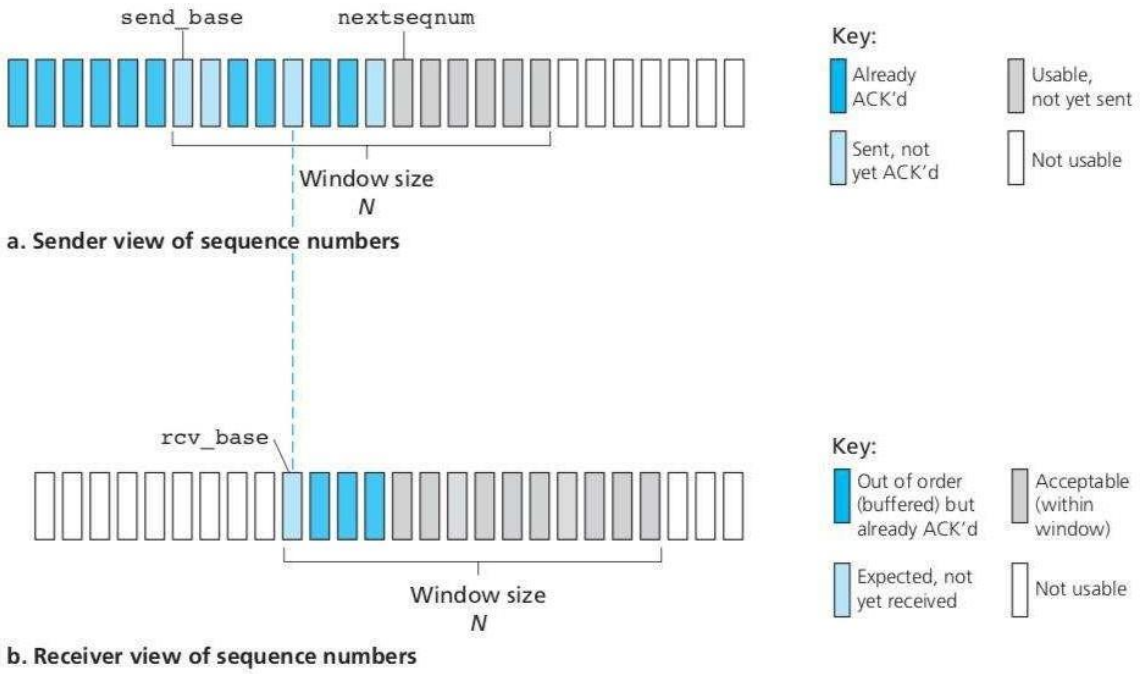


Figure 2.20 _ Extended FSM description of GBN sender

The GBN sender must respond to three types of events:

• ***Invocation from above.*** When rdt_send() is called from above, the sender first checks to see if the window is full, that is, whether there are *N* outstanding, unacknowledged packets. If the window is not full, a packet is created and sent, and variables are appropriately updated. If the window is full, the sender simply returns the data back to the upper layer, an implicit indication that the window is full. The upper layer would presumably then have to try again later.

•

In a real implementation, the sender would more likely have either buffered (but not immediately sent) this data, or would have a synchronization mechanism (for example, a semaphore or a flag) that would allow the upper layer to call rdt_send() only when the window is not full.

• ***Receipt of an ACK.*** In our GBN protocol, an acknowledgment for a packet with sequence number *n* will be taken to be a **cumulative acknowledgment**, indicating that all packets with a sequence number up to and including *n* have been correctly received at the receiver. We'll come back to this issue shortly when we examine the receiver side of GBN.

• ***A timeout event.*** The protocol's name, "Go-Back-N," is derived from the sender's behavior in the presence of lost or overly delayed packets. As in the stop-and-wait protocol, a timer will again be used to recover from lost data or acknowledgment packets. If a timeout occurs, the sender resends *all* packets that have been previously sent but that have not yet been acknowledged. Our sender in Figure 2.20 uses only a single timer, which can be thought of as a timer for the oldest transmitted but not yet acknowledged packet.

• If an ACK is received but there are still additional transmitted but not yet acknowledged packets, the timer is restarted. If there are no outstanding, unacknowledged packets, the timer is stopped.

Selective Repeat (SR)

A single packet error can thus cause GBN to retransmit a large number of packets, many unnecessarily. As the probability of channel errors increases, the pipeline can become filled with these unnecessary retransmissions.

Imagine, in our message-dictation scenario, that if every time a word was garbled, the surrounding 1,000 words (for example, a window size of 1,000 words) had to be repeated. The dictation would be slowed by all of the reiterated words. As the name suggests, selective-repeat protocols avoid unnecessary retransmissions by having the sender retransmit only those packets that it suspects were received in error (that is, were lost or corrupted) at the receiver.

This individual, as needed, retransmission will require that the receiver *individually* acknowledge correctly received packets. A window size of *N* will again be used to limit the numberof outstanding, unacknowledged packets in the pipeline. However, unlike GBN, the sender will have already received ACKs for some of the packets in the window.

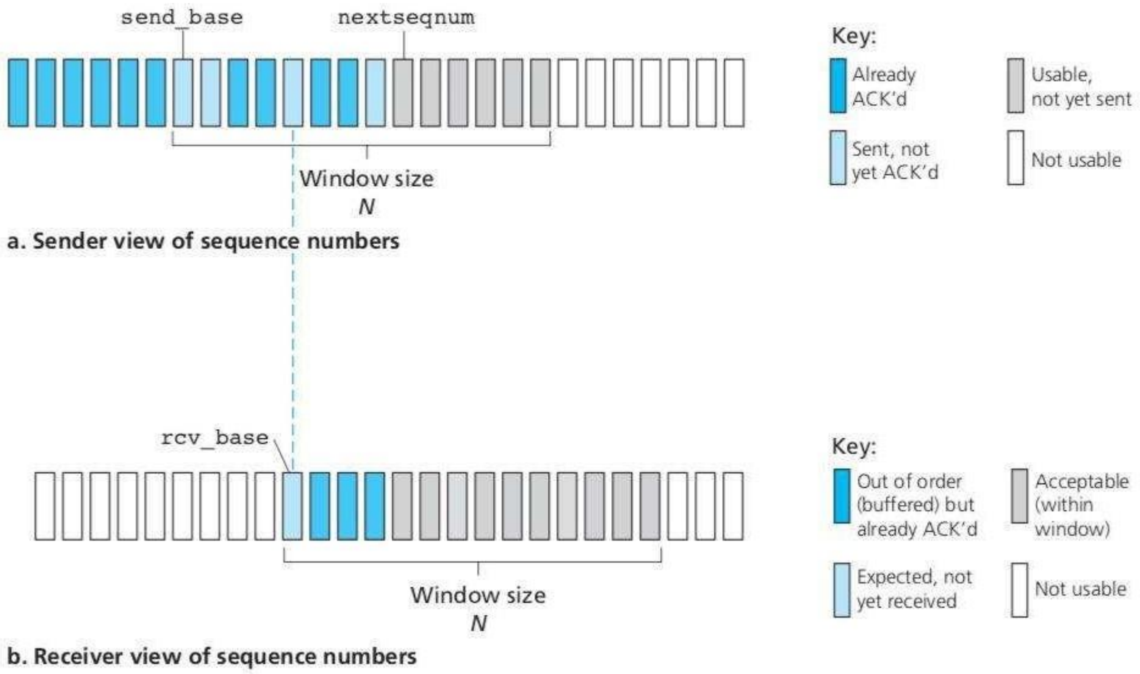


Figure 2.23 _ Selective-repeat (SR) sender and receiver views of sequence-number space

Figure 3.23 shows the SR sender's view of the sequence number space. Figure 2.24 details the various actions taken by the SR sender. The SR receiver will acknowledge a correctly received packet whether or not it is in order. Out-of-order packets are buffered until any missing packets (that is, packets with lower sequence numbers) are received, at which point a batch of packets can be delivered in order to the upper layer. Figure 2.25 itemizes the various actions taken by the SR receiver. Figure 2.26 shows an example of SR operation in the presence of lost packets. Note that in Figure 2.26, the receiver initially buffers packets 3, 4, and 5, and delivers them together with packet 2 to the upper layer when packet 2 is finally received.

It is important to note that in Step 2 in Figure 2.25, the receiver reacknowledges (rather than ignores) already received packets with certain sequence numbers *below* the current window base. You should convince yourself that this reacknowledgment is indeed needed. Given the sender and receiver sequence number spaces in Figure 2.23, for example, if there is no ACK for packet send_base propagating from the receiver to the sender, the sender will eventually retransmit packet send_base, even though it is clear (to us, not the sender!) that the receiver has already received.

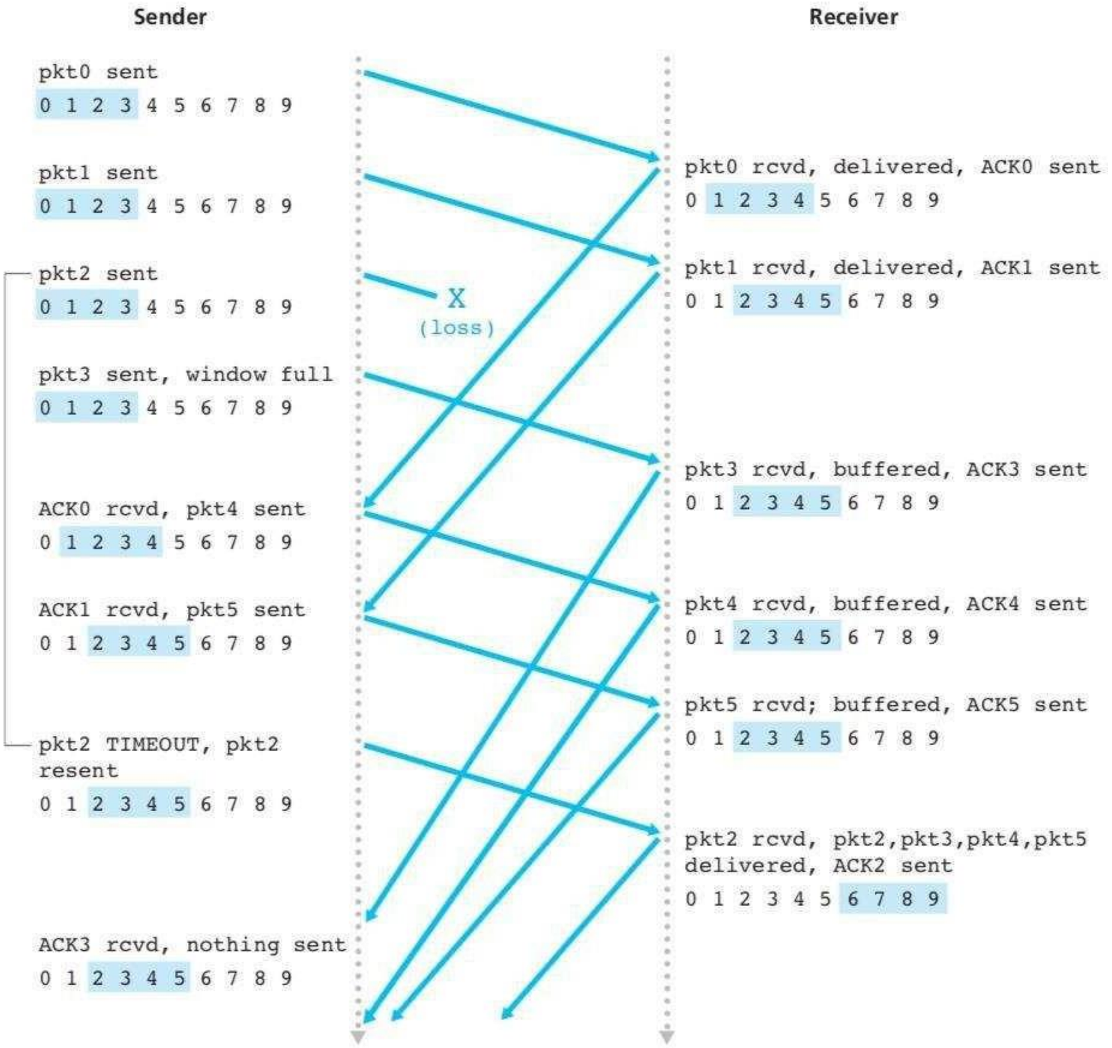


Figure 2.22 _ Go-Back-N in operation

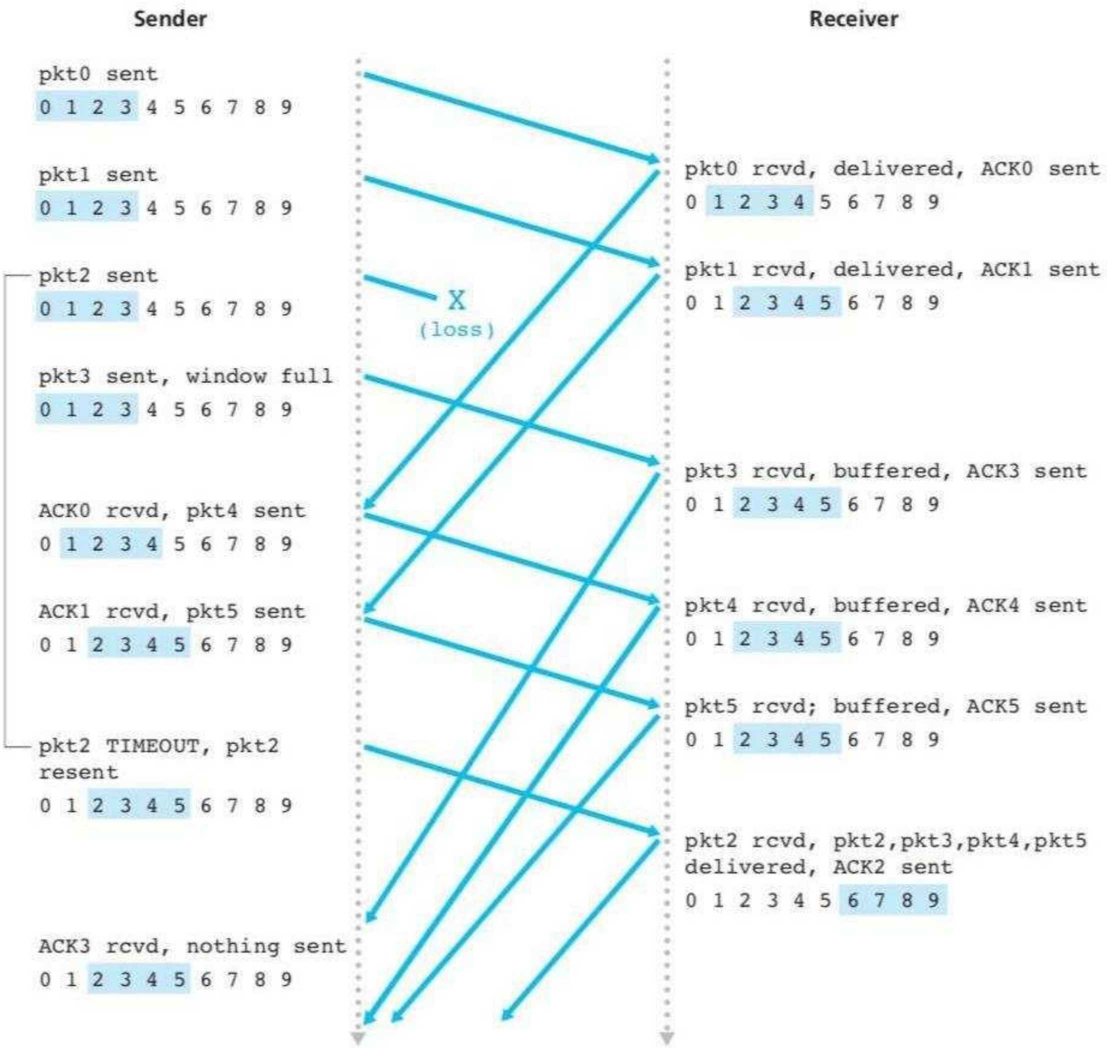


Figure 2.23 _ Selective-repeat (SR) sender and receiver views of sequence-number space

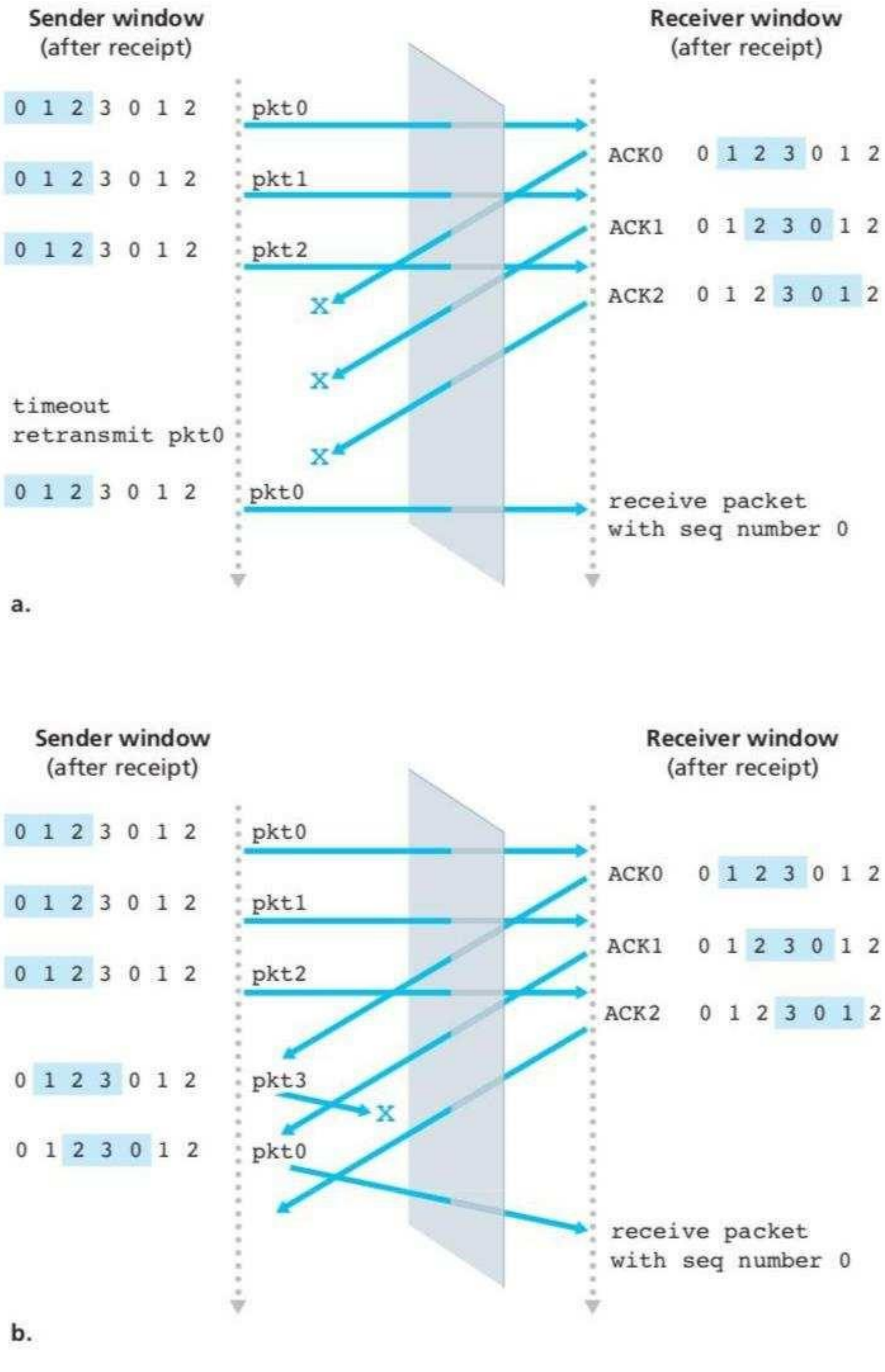


Figure 2.14 _ rdt2.2 receiver

Connection-Oriented Transport: TCP

The TCP Connection

TCP is said to be **connection-oriented** because before one application process can begin to send data to another, the two processes must first "handshake" with each other that is, they must send some preliminary segments to each other to establish the parameters of the ensuing data transfer. As part of TCP connection establishment, both sides of the connection will initialize many TCP state variables associated with the TCP connection.

The TCP "connection" is not an end-to-end TDM or FDM circuit as in a circuit switched network. Nor is it a virtual circuit (see Chapter 1), as the connection state resides entirely in the two end systems. Because the TCP protocol runs only in the end systems and not in the intermediate network elements (routers and link-layer switches), the intermediate network elements do not maintain TCP connection state. A TCP connection provides a **full-duplex service**: If there is a TCP connection between Process A on one host and Process B on another host, then applicationlayer data can flow from Process A to Process B at the same time as application layer data flows from Process B to Process A. A TCP connection is also always **point-to-point**, that is, between a single sender and a single receiver. So-called "multicasting" the transfer of data from one sender to many receivers in a single send operation is not possible with TCP.

Let's now take a look at how a TCP connection is established. Suppose a process running in one host wants to initiate a connection with another process in another host. Recall that the process that is initiating the connection is called the *client process*, while the other process is called the *server process*. The client application process first informs the client transport layer that it wants to establish a connection to a process in the server. Recall from earlier, a Python client program does this by issuing the command

clientSocket.connect((serverName,serverPort))

where serverName is the name of the server and serverPort identifies the process on the server. TCP in the client then proceeds to establish a TCP connection with TCP in the server. At the end of this section we discuss in some detail the connection establishment procedure. For now it suffices to know that the client first sends a special TCP segment; the server responds with a second special TCP segment; and finally the client responds again with a third special segment. The first two segments carry no payload, that is, no application-layer data; the third of these segments may carry a payload. Because three segments are sent between the two hosts, this connection establishment procedure is often referred to as a **three-way handshake**.

Once a TCP connection is established, the two application processes can send data to each other. Let's consider the sending of data from the client process to the server process. The client process passes a stream of data through the socket (the door of the process), as described. Once the data passes through the door, the data is in the hands of TCP running in the client. As shown in Figure 2.28, TCP directs this data to the connection's **send buffer**, which is one of the buffers that is set aside during the initial three-way handshake. From time to time, TCP will grab chunks of data from the send buffer and pass the data to the network layer. Interestingly, the TCP specification [RFC 793] is very laid back about specifying when TCP should actually send buffered data, stating that TCP should "send that data in segments at its own convenience."

The maximum amount of data that can be grabbed and placed in a segment is limited by the **maximum segment size (MSS)**. The MSS is typically set by first determining the length of the largest link-layer frame that can be sent by the local sending host (the so-called **maximumtransmission unit**, **MTU**), and then setting the MSS to ensure that a TCP segment (when encapsulated in an IP datagram) plus the TCP/IP header length (typically 40 bytes) will fit into a single link-layer frame. Both Ethernet and PPP link-layer protocols have an MSS of 1,500 bytes. Approaches have also been proposed for discovering the path MTU—the largest link-layer frame that can be sent on all links from source to destination [RFC 1191]—and setting the MSS based on the path MTU value. Note that the MSS is the maximum amount of application-layer data in the segment, not the maximum size of the TCP segment including headers.

TCP Segment Structure

As with UDP, the header includes **source and destination port numbers**, which are used for multiplexing/demultiplexing data from/to upper-layer applications. Also, as with UDP, the header includes a **checksum field**. A TCP segment header also contains the following fields:

• The 32-bit **sequence number field** and the 32-bit **acknowledgment number field** are used by the TCP sender and receiver in implementing a reliable data transfer service, as discussed below.

• The 16-bit **receive window** field is used for flow control. We will see shortly that it is used to indicate the number of bytes that a receiver is willing to accept.

• The 4-bit **header length field** specifies the length of the TCP header in 32-bit words. The TCP header can be of variable length due to the TCP options field.

• The optional and variable-length **options field** is used when a sender and receiver negotiate the maximum segment size (MSS) or as a window scaling factor for use in high-speed networks. A time-stamping option is also defined.

• The **flag field** contains 6 bits. The **ACK bit** is used to indicate that the value carried in the acknowledgment field is valid; that is, the segment contains an acknowledgment for a segment that has been successfully received. The **RST**, **SYN**, and **FIN** bits are used for connection setup and teardown, as we will discuss at the end of this section. Setting the **PSH** bit indicates that the receiver should pass the data to the upper layer immediately. Finally, the **URG** bit is used to indicate that there is data in this segment that the sending-side upper-layer entity has marked as "urgent." The location of the last byte of this urgent data is indicated by the 16-bit **urgent data pointer field**. TCP must inform the receiving side upper-layer entity when urgent data exists and pass it a pointer to the end of the urgent data.

Round-Trip Time Estimation and Timeout

Estimating the Round-Trip Time

Let's begin our study of TCP timer management by considering how TCP estimates the round-trip time between sender and receiver. This is accomplished as follows. The sample RTT, denoted SampleRTT, for a segment is the amount of time between when the segment is sent (that is, passed to IP) and when an acknowledgment for the segment is received. Instead of measuring a SampleRTT for every transmitted segment, most TCP implementations take only one SampleRTT measurement at a time. That is, at any point in time, the SampleRTT is being estimated for only one of the transmitted but currently unacknowledged segments, leading to a new value of SampleRTT approximately once every RTT. Also, TCP never computes a SampleRTT for a segment that has been retransmitted; it only measures SampleRTT for segments that have been transmitted once. Obviously, the SampleRTT values will fluctuate from segment to segment due to congestion in the routers and to the varying load on the end systems. Because of this fluctuation, any given SampleRTT value may be atypical. In order to estimate a typical RTT, it is therefore natural to take some sort of average of the SampleRTT values. TCP maintains an average, called EstimatedRTT, of the SampleRTT values. Upon obtaining a new SampleRTT, TCP updates EstimatedRTT according to the following formula:

EstimatedRTT = (1 – _) • EstimatedRTT + _ • SampleRTT

The formula above is written in the form of a programming-language statement the new value of EstimatedRTT is a weighted combination of the previous value of EstimatedRTT and the new value for SampleRTT. The recommended value of _ is _ = 0.125 (that is, 1/8), in which case the formula above becomes:

EstimatedRTT = 0.875 • EstimatedRTT + 0.125 • SampleRTT

Note that EstimatedRTT is a weighted average of the SampleRTT values. more recent samples better reflect the current congestion in the network. In statistics, such an average is called an **exponential weighted moving average (EWMA)**. The word “exponential” appears in EWMA because the weight of a given SampleRTT decays exponentially fast as the updates proceed. In the homework problems you will be asked to derive the exponential term in EstimatedRTT. Figure 3.32 shows the SampleRTT values and EstimatedRTT for a value of = 1/8 for a TCP connection between gaia.cs.umass.edu (in Amherst, Massachusetts) to fantasia.eurecom.fr (in the south of France).

Clearly, the variations in the SampleRTT are smoothed out in the computation of the EstimatedRTT. In addition to having an estimate of the RTT, it is also valuable to have a measure of the variability of the RTT. [RFC 6298] defines the RTT variation, DevRTT, as an estimate of how much SampleRTT typically deviates from

EstimatedRTT: DevRTT = (1 – _) • DevRTT + _•| SampleRTT – EstimatedRTT |

Note that DevRTT is an EWMA of the difference between SampleRTT and EstimatedRTT. If the SampleRTT values have little fluctuation, then DevRTT will be small; on the other hand, if there is a lot of fluctuation, DevRTT will be large. The recommended value of β is 0.25.

Setting and Managing the Retransmission Timeout Interval

TimeoutInterval = EstimatedRTT + 4 • DevRTT

Reliable Data Transfer

TCP creates a **reliable data transfer service** on top of IP’s unreliable best effort service. TCP’s reliable data transfer service ensures that the data stream that a process reads out of its TCP receive buffer is uncorrupted, without gaps, without duplication, and in sequence; that is, the byte stream is exactly the same byte stream that was sent by the end system on the other side of the connection. How TCP provides a reliable data transfer involves many of the principles.

In our earlier development of reliable data transfer techniques, it was conceptually easiest to assume that an individual timer is associated with each transmitted but not yet acknowledged segment. While this is great in theory, timer management can require considerable overhead. Thus, the recommended TCP timer management procedures [RFC 6298] use only a *single* retransmission timer, even if there are multiple transmitted but not yet acknowledged segments. The TCP protocol described in this section follows this single-timer recommendation.

```
NextSeqNum=InitialSeqNumber
SendBase=InitialSeqNumber
loop (forever) {
switch(event)
event: data received from application above
create TCP segment with sequence number NextSeqNum
if (timer currently not running) start timer

pass segment to IP
NextSeqNum=NextSeqNum+length(data)
break;
event: timer timeout
retransmit not-yet-acknowledged segment with
smallest sequence number
start timer
break;
event: ACK received, with ACK field value of y
if (y > SendBase) {
SendBase=y
if (there are currently any not-yet-acknowledged segments)
start timer
}
break;
} /* end of loop forever */
```

Figure 2.33 _ Simplified TCP sender

The second major event is the timeout. TCP responds to the timeout event by retransmitting the segment that caused the timeout. TCP then restarts the timer.

Flow Control

TCP provides a **flow-control service** to its applications to eliminate the possibility of the sender overflowing the receiver's buffer. Flow control is thus a speed-matching service matching the rate at which the sender is sending against the rate at which the receiving application is reading. As noted earlier, a TCP sender can also be throttled due to congestion within the IP network; this form of sender control is referred to as **congestion control**, a topic we will explore. Even though the actions taken by flow and congestion control are similar (the throttling of the sender), they are obviously taken for very different reasons. Unfortunately, many authors use the terms interchangeably, and the savvy reader would be wise to distinguish between them. Let's now discuss how TCP provides its flow-control service. In order to see the forest for the trees, we suppose throughout this section that the TCP implementation is such that the TCP receiver discards out-of-order segments.

TCP provides flow control by having the *sender* maintain a variable called the **receive window**. Informally, the receive window is used to give the sender an idea of how much free buffer space is available at the receiver. Because TCP is full-duplex, the sender at each side of the connection maintains a distinct receive window. Let's investigate the receive window in the context of a file transfer. Suppose that Host Ais sending a large file to Host B over a TCP connection. Host B allocates a receive buffer to this connection; denote its size by RcvBuffer. From time to time, the application process in Host B reads from the buffer. Define the following variables:

• LastByteRead: the number of the last byte in the data stream read from the buffer by the application process in B.

• LastByteRcvd: the number of the last byte in the data stream that has arrived from the network and has been placed in the receive buffer at B.

Because TCP is not permitted to overflow the allocated buffer, we must have

LastByteRcvd – LastByteRead _ RcvBuffer

The receive window, denoted rwnd is set to the amount of spare room in the buffer:

rwnd = RcvBuffer – [LastByteRcvd – LastByteRead]

Because the spare room changes with time, rwnd is dynamic. The variable rwnd is illustrated in Figure

3.38. TCP Connection Management

The TCP in the client then proceeds to establish a TCP connection with the TCP in the server in the following manner:

• ***Step 1.*** The client-side TCP first sends a special TCP segment to the server-side TCP. This special segment contains no application-layer data. But one of the flag bits in the segment's header, the SYN bit, is set to 1. For this reason, this special segment is referred to as a SYN segment. In addition, the client randomly chooses an initial sequence number (client_isn) and puts
this number in the sequence number field of the initial TCP SYN segment. This segment is encapsulated within an IP datagram and sent to the server. There has been considerable interest in properly randomizing the choice of the client_isn in order to avoid certain security attacks.

• ***Step 2.*** Once the IP datagram containing the TCP SYN segment arrives at the server host (assuming it does arrive!), the server extracts the TCP SYN segment from the datagram, allocates the TCP buffers and variables to the connection, and sends a connection-granted segment to the client TCP. This connection-granted segment also contains no application layer data. However, it does contain three important pieces of information in the segment header. First, the SYN bit is set to 1. Second, the acknowledgment field of the TCP segment header is set to client_isn+1. Finally, the server chooses its own initial sequence number (server_isn) and puts this value in the sequence number field of the TCP segment header. This connection-granted segment is saying, in effect, "I received your SYN packet to start a connection with your initial sequence number, client_isn. I agree to establish this connection. My own initial sequence number is
server_isn." The connectiongranted segment is referred to as a **SYNACK segment**.

• ***Step 3.*** Upon receiving the SYNACK segment, the client also allocates buffers and variables to the connection. The client host then sends the server yet another segment; this last segment acknowledges the server's connection-granted segment (the client does so by putting the value server_isn+1 in the acknowledgment field of the TCP segment header). The SYN bit is set to zero, since the connection is established. This third stage of the three-way handshake may carry client-to-server data in the segment payload.

Once these three steps have been completed, the client and server hosts can send segments containing data to each other. In each of these future segments, the SYN bit will be set to zero. Note that in order to establish the connection, three packets are sent between the two hosts. For this reason, this connection establishment procedure is often referred to as a **three-way handshake**.

There are three possible outcomes:

• ***The source host receives a TCP SYNACK segment from the target host.*** Since this means that an application is running with TCP port 6789 on the target post, nmap returns "open."

• ***The source host receives a TCP RST segment from the target host.*** This means that the SYN segment reached the target host, but the target host is not running an application with TCP port 6789. But the attacker at least knows that the segments destined to the host at port 6789 are not blocked by any firewall on the path between source and target hosts.

• ***The source receives nothing.*** This likely means that the SYN segment was blocked by an intervening firewall and never reached the target host.

Principles of Congestion Control

The Causes and the Costs of Congestion

Scenario 1: Two Senders, a Router with Infinite Buffers

We begin by considering perhaps the simplest congestion scenario possible: Two hosts (A and B) each have a connection that shares a single hop between source and destination. Let's assume that the application in Host A is sending data into the connection (for example, passing data to the transport-level protocol via a socket) at an average rate of _in bytes/sec. These data are original in the sense that each unit of data is sent into the socket only once. The underlying transport-level protocol is a simple one.

Data is encapsulated and sent; no error recovery (for example, retransmission), flow control, or congestion control is performed. Ignoring the additional overhead due to adding transport- and lower-layer header information, the rate at which Host A offers traffic to the router in this first scenario is thus _in bytes/sec. Host B operates in a similar manner, and we assume for simplicity that it too is sending at a rate of _in bytes/sec. Packets from Hosts A and B pass through a router and over a shared outgoing link of capacity
R. The router has buffers that allow it to store incoming packets when the packet-arrival rate exceeds the outgoing link's capacity.

Scenario 2: Two Senders and a Router with Finite Buffers

The performance realized under scenario 2 will now depend strongly on how retransmission is performed. First, consider the unrealistic case that Host A is able to somehow (magically!) determine whether or not a buffer is free in the router and thus sends a packet only when a buffer is free.

Scenario 3: Four Senders, Routers with Finite Buffers, and
Multihop Paths

Let's consider the connection from Host A to Host C, passing through routers R1 and R2. The A–C connection shares router R1 with the D–B connection and shares router R2 with the B–D connection. For extremely small values of _in, buffer overflows are rare (as in congestion scenarios 1 and 2), and the throughput approximately equals the offered load. For slightly larger values of _in, the corresponding throughput is also larger, since more original data is being transmitted into thenetwork and delivered to the destination, and overflows are still rare. Thus, for small values of _in, an increase in _in results in an increase in _out.

Approaches to Congestion Control

At the broadest level, we can distinguish among congestion-control approaches by whether the network layer provides any explicit assistance to the transport layer for congestion-control purposes:

• ***End-to-end congestion control.*** In an end-to-end approach to congestion control, the network layer provides *no explicit support* to the transport layer for congestion control purposes. Even the presence of congestion in the network must be inferred by the end systems based only on observed network behavior (for example, packet loss and delay). TCP segment loss (as indicated by a timeout or a triple duplicate acknowledgment) is taken as an indication of network congestion and TCP decreases its window size accordingly. We will also see a more recent proposal for TCP congestion control that uses increasing round-trip delay values as indicators of increased network congestion.

• ***Network-assisted congestion control.*** With network-assisted congestion control, network-layer components (that is, routers) provide explicit feedback to the sender regarding the congestion state in the network. This feedback may be as simple as a single bit indicating congestion at a link. Recently proposed for TCP/IP networks is used in ATM available bit-rate (ABR) congestion control as well, as discussed below. More sophisticated network feedback is also possible. For example, one form of ATM ABR congestion control that we will study shortly allows a router to inform the sender explicitly of the transmission rate it (the router) can support on an outgoing link. The XCP protocol provides router-computed feedback to each source, carried in the packet header, regarding how that source should increase or decrease its transmission rate.

Network-Assisted Congestion-Control Example:

ATM ABR Congestion Control

Fundamentally ATM takes a virtual-circuit (VC) oriented approach toward packet switching. Recall from our discussion in Chapter 1, this means that each switch on the source-to-destination path will maintain state about the source-to destination VC. This per-VC state allows a switch to track the behavior of individual senders (e.g., tracking their average transmission rate) and to take source-specific congestion-control actions (such as explicitly signaling to the sender to reduce its rate when the switch becomes congested). This per-VC state at network switches makes ATM ideally suited to perform network-assisted congestion control.

ABR has been designed as an elastic data transfer service in a manner reminiscent of TCP. When the network is underloaded, ABR service should be able to take advantage of the spare available bandwidth; when the network is congested, ABR service should throttle its transmission rate to some predetermined minimum transmission rate.

In our discussion we adopt ATM terminology (for example, using the term *switch* rather than *router*, and the term *cell* rather than *packet*). With ATM ABR service, data cells are transmitted from a source to a destination through a series of intermediate switches. Interspersed with the data cells are **resource-management cells(RM cells)**; these RM cells can be used to convey congestion-related information among the hosts and switches. When an RM cell arrives at a destination, it will be turned around and sent back to the sender (possibly after the destination has modified the contents of the RM cell). It is also possible for a switch to generate an RM cell itself and send this RM cell directly to a source. RM cells can thus be used to provide both direct network feedback and network feedback via the receiver.

ATM ABR congestion control is a rate-based approach. That is, the sender explicitly computes a maximum rate at which it can send and regulates itself accordingly. ABR provides three mechanisms for signaling congestion-related information from the switches to the receiver:

• *EFCI bit.* Each *data cell* contains an **explicit forward congestion indication (EFCI) bit**. A congested network switch can set the EFCI bit in a data cell to 1 to signal congestion to the destination host. The destination must check the EFCI bit in all received data cells. When an RM cell arrives at the destination, if the most recently received data cell had the EFCI bit set to 1, then the destination sets the congestion indication bit (the CI bit) of the RM cell to 1 and sends the RM cell back to the sender. Using the EFCI in data cells and the CI bit in RM cells, a sender can thus be notified about congestion at a network switch.

• ***CI and NI bits.*** As noted above, sender-to-receiver RM cells are interspersed with data cells. The rate of RM cell interspersion is a tunable parameter, with the default value being one RM cell every 32 data cells. These RM cells have a **congestion indication (CI) bit** and a **no increase (NI) bit** that can be set by a congested network switch. Specifically, a switch can set the NI bit in a passing RM cell to 1 under mild congestion and can set the CI bit to 1 under severe congestion conditions. When a destination host receives an RM cell, it will send the RM cell back to the sender with its CI and NI bits intact (except that CI may be set to 1 by the destination as a result of the EFCI mechanism described above).

• ***ER setting.*** Each RM cell also contains a 2-byte **explicit rate (ER) field**. A congested switch may lower the value contained in the ER field in a passing RM cell. In this manner, the ER field will be set to the minimum supportable rate of all switches on the source-to-destination path.

TCP Congestion Control

TCP answers these questions using the following guiding principles:

• ***A lost segment implies congestion, and hence, the TCP sender's rate should be decreased when a segment is lost.*** Recall from our discussion in 4, that a timeout event or the receipt of four acknowledgments for a given segment (one original ACK and then three duplicate ACKs) is interpreted as an implicit "loss event" indication of the segment following the quadruply ACKed segment, triggering a retransmission of the lost segment. From a congestion control standpoint, the question is how the TCP sender should decrease its congestion window size, and hence its sending rate, in response to this inferred loss event.

• ***An acknowledged segment indicates that the network is delivering the sender's segments to the receiver, and hence, the sender's rate can be increased when an ACK arrives for a previously unacknowledged segment.*** The arrival of acknowledgments is taken as an implicit indication that all is well—segments are being successfully delivered from sender to receiver, and the network is thus not congested. The congestion window size can thus be increased.

• ***Bandwidth probing.*** Given ACKs indicating a congestion-free source-to-destination path and loss events indicating a congested path, TCP's strategy for adjusting its transmission rate is to increase its rate in response to arriving ACKs until a loss event occurs, at which point, the transmission rate is decreased. The TCP sender thus increases its transmission rate to probe for the rate that at which congestion onset begins, backs off from that rate, and then to begins probing again to see if the congestion onset rate has changed. The TCP sender's behavior is perhaps analogous to the child who requests (and gets) more and more goodies until finally he/she is finally told "No!", backs off a bit, but then begins making requestsagain shortly afterwards. Note that there is no explicit signaling of congestion state by the network—ACKs and loss events serve as implicit signals—and that each TCP sender acts on local information asynchronously from other TCP senders.

Given this overview of TCP congestion control, we're now in a position to consider the details of the celebrated **TCP congestion-control algorithm**

Slow Start

When a TCP connection begins, the value of cwnd is typically initialized to a small value of 1 MSS [RFC 3390], resulting in an initial sending rate of roughly MSS/RTT. For example, if MSS = 500 bytes and RTT = 200 msec, the resulting initial sending rate is only about 20 kbps. Since the available bandwidth to the TCP sender may be much larger than MSS/RTT, the TCP sender would like to find the amount of available bandwidth quickly. Thus, in the **slow-start** state, the value of cwnd begins at 1 MSS and increases by 1 MSS every time a transmitted segment is first acknowledged.

In the example TCP sends the first segment into the network and waits for an acknowledgment. When this acknowledgment arrives, the TCP sender increases the congestion window by one MSS and sends out two maximum-sized segments. These segments are then acknowledged, with the sender increasing the congestion window by 1 MSS for each of the acknowledged segments, giving a congestion window of 4 MSS, and so on. This process results in a doubling of the sending rate every RTT. Thus, the TCP send rate starts slow but grows exponentially during the slow start phase.

Congestion Avoidance

On entry to the congestion-avoidance state, the value of cwnd is approximately half its value when congestion was last encountered—congestion could be just around the corner! Thus, rather than doubling the value of cwnd every RTT, TCP adopts a more conservative approach and increases the value of cwnd by just a single MSS every RTT [RFC 5681]. This can be accomplished in several ways. A common approach is for the TCP sender to increase cwnd by MSS bytes (MSS/cwnd) whenever a new acknowledgment arrives. For example, if MSS is 1,460 bytes and cwnd is 14,600 bytes, then 10 segments are being sent within an RTT. Each arriving ACK (assuming one ACK per segment) increases the congestion window size by 1/10 MSS, and thus, the value of the congestion window will have increased by one MSS after ACKs when all 10 segments have been received.

Fast Recovery

In fast recovery, the value of cwnd is increased by 1 MSS for every duplicate ACK received for the missing segment that caused TCP to enter the fast-recovery state. Eventually, when an ACK arrives for the missing segment, TCP enters the congestion-avoidance state after deflating cwnd. If a timeout event occurs, fast recovery transitions to the slow-start state after performing the same actions as in slow start and congestion avoidance: The value of cwnd is set to 1 MSS, and the value of ssthresh is set to half the value of cwnd when the loss event occurred.

Fast recovery is a recommended, but not required, component of TCP [RFC 5681]. It is interesting that an early version of TCP, known as **TCP Tahoe**, unconditionally cut its congestion window to 1 MSS and entered the slow-start phase after either a timeout-indicated or triple-duplicate-ACK-indicated loss event. The newer version of TCP, **TCP Reno**, incorporated fast recovery. The evolution of TCP's congestion window for both Reno and Tahoe. In this figure, the threshold is initially equal to 8 MSS. For the first eight transmission rounds, Tahoe and Reno take identical actions. The congestion window climbs exponentially fast during slow start and hits the threshold at the fourth round of transmission. The congestion window then climbs linearly until a triple duplicate- ACK event occurs, just after transmission round 8. Note that the congestion window is 12 • *MSS* when this loss event occurs. The value of ssthresh is then set to 0.5 • cwnd = 6 • MSS. Under TCP Reno, the congestion window is set to cwnd = 6 • MSS and then grows linearly. Under TCP Tahoe, the congestion window is set to 1 MSS and grows exponentially until it reaches the value of ssthresh, at which point it grows linearly.

Fairness

Fairness and UDP

We have just seen how TCP congestion control regulates an application's transmission rate via the congestion window mechanism. Many multimedia applications, such as Internet phone and video conferencing, often do not run over TCP for this very reason—they do not want their transmission rate throttled, even if the network is very congested.

Instead, these applications prefer to run over UDP, which does not have built-in congestion control. When running over UDP, applications can pump their audio and video into the network at a constant rate and occasionally lose packets, rather than reduce their rates to "fair" levels at times of congestion and not lose any packets. From the perspective of TCP, the multimedia applications running over UDP are not being fair—they do not cooperate with the other connections nor adjust their transmission rates appropriately. Because TCP congestion control will decrease its transmission rate in the face of increasing congestion (loss), while UDP sources need not, it is possible for UDP sources to crowd out TCP traffic.

Fairness and Parallel TCP Connections

But even if we could force UDP traffic to behave fairly, the fairness problem would still not be completely solved. This is because there is nothing to stop a TCP-based application from using multiple parallel connections. For example, Web browsers often use multiple parallel TCP connections to transfer the multiple objects within a Web page. (The exact number of multiple connections is configurable in most browsers.)

When an application uses multiple parallel connections, it gets a larger fraction of the bandwidth in a congested link. As an example, consider a link of rate *R* supporting nine ongoing client-server applications, with each of the applications using one TCP connection. If a new application comes along and also uses one TCP connection, then each application gets approximately the same transmission rate of *R*/10.

But if this new application instead uses 11 parallel TCP connections, then the new application gets an unfair allocation of more than *R*/2. Because Web traffic is so pervasive in the Internet, multiple parallel connections are not uncommon.