Transport Layer

3.1 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. Application processes use the logical communication provided by the transport layer to send messages to each other.

Transport-layer protocols are implemented in the end systems but not in network routers. On the sending side, the transport layer converts the application-layer messages it receives from a sending application process into transport-layer packets, known as transport-layer segments in Internet terminology. This is done by breaking the application messages into smaller chunks and adding a transport-layer header to each chunk to create the transport-layer segment. The transport layer then passes the segment to the network layer at the sending end system, where the segment is encapsulated within a network-layer packet and sent to the destination.

3.1.1 Relationship Between Transport and Network Layers

A transport-layer protocol provides logical communication between processes running on different hosts, a network-layer protocol provides logical communication between hosts.

Within an end system, a transport protocol moves messages from application processes to the network edge (that is, the network layer) and vice versa, but it doesn't have any say about how the messages are moved within the network core.

If the network-layer protocol cannot provide delay or bandwidth guarantees for transport-layer segments sent between hosts, then the transport-layer protocol cannot provide delay or bandwidth guarantees for application messages sent between processes.

Nevertheless, certain services can be offered by a transport protocol even when the underlying network protocol doesn't offer the corresponding service at the network layer.

3.1.2 Overview of the Transport Layer in the Internet

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.

We refer to the transport-layer packet as a segment. Transport-layer packet for TCP is referred to as a segment but often refers to the the packet for UDP as a datagram.

The Internet's network-layer protocol has a name—IP, for Internet Protocol. IP provides logical communication between hosts. 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 no guarantees. 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 data in the segments. For these reasons, IP is said to be an unreliable service. We also mention here that every host has at least one network-layer address, a so-called IP address.

The most fundamental responsibility of UDP and TCP is to extend IP's delivery service between two end systems to a delivery service between two processes running on the end systems. Extending host-to-host delivery to process-to-process delivery is called transport-layer multiplexing and demultiplexing. UDP and TCP also provide integrity checking by including error-detection fields in their segments' headers. These two minimal transport-layer services—process-to-process data delivery and error checking—are the only two services that UDP provides! UDP is an unreliable service—it does not guarantee that data sent by one process will arrive intact (or at all!) to the destination process.

TCP offers several additional services to applications. First and foremost, it provides reliable data transfer. Using flow control, sequence numbers, acknowledgments, and timers. TCP ensures that data is delivered from sending process to receiving process, correctly and in order. TCP thus converts IP's unreliable service between end systems into a reliable data transport service between processes. TCP also provides congestion control. (Congestion Control) It is a service for the Internet as a whole, a service for the general good. TCP congestion control prevents any one TCP connection from swamping the links and routers between communicating hosts with an excessive amount of traffic. TCP strives to give each connection traversing a congested link an equal share of the link bandwidth. This is done by regulating the rate at which the sending sides of TCP connections can send traffic into the network.

3.2 Multiplexing and Demultiplexing

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. When the transport layer in your computer receives data from the network layer below, it needs to direct the received data to one of these four processes.

The transport layer in the receiving host does not actually deliver data directly to a process, but instead to an intermediary socket. Because at any given time there can be more than one socket in the receiving host, each socket has a unique identifier. The format of the identifier depends on whether the socket is a UDP or a TCP 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 to create segments, and passing the segments to the network layer is called multiplexing. Although we have introduced multiplexing and demultiplexing in the context of the Internet transport protocols, it's important to realize that they are concerns whenever a single protocol at one layer is used by multiple protocols at the next higher layer.

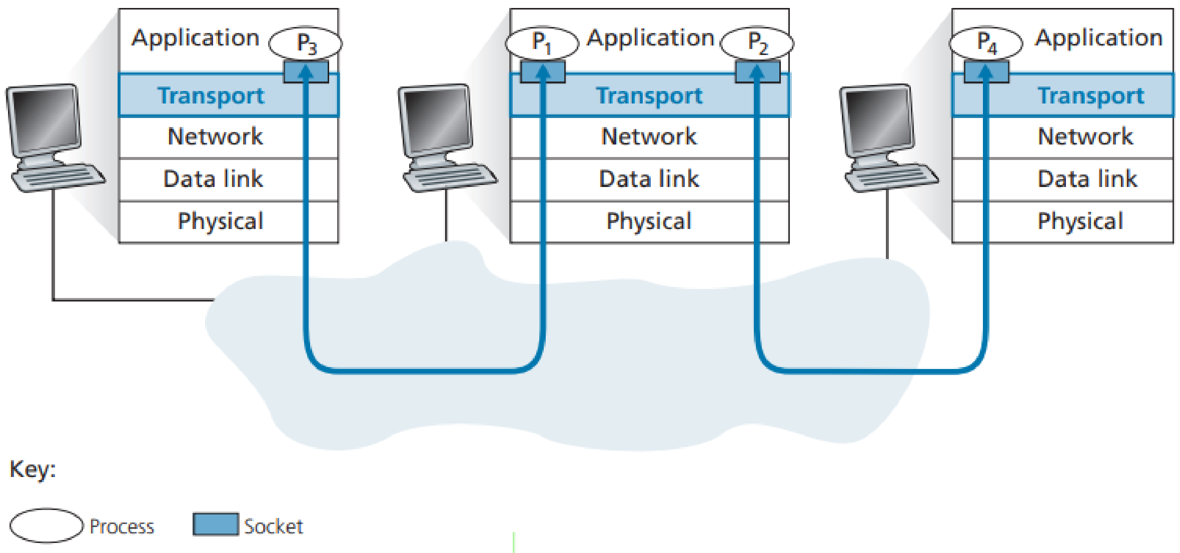


Figure 3.2 ♦ Transport-layer multiplexing and demultiplexing

We know that transport-layer multiplexing requires (1) that sockets have unique identifiers, and (2) that each segment have special fields that indicate the socket to which the segment is to be delivered. The special fields are the source port number field and the destination port number field. Each port number is a 16-bit number, ranging from 0 to 65535. The port numbers ranging from 0 to 1023 are called well-known port numbers and are restricted, which means that they are reserved for use by well-known application protocols.

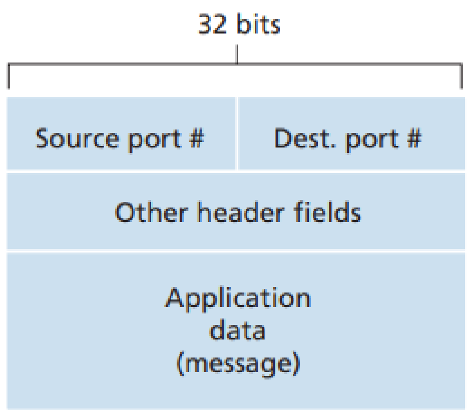


Figure 3.3 ♦ Source and destination port-number fields in a transport-layer segment

It should now be clear how the transport layer could implement the demultiplexing service: Each socket in the host could be assigned a port number, and when a segment arrives at the host, the transport layer examines the destination port number in the segment and directs the segment to the corresponding socket. The segment's data then passes through the socket into the attached process.

Connectionless Multiplexing and Demultiplexing

```
# How to create a UDP socket in Python
clientSocket = socket(AF_INET, SOCK_DGRAM)

# How to associate a specific port number to my UDP socket
clientSocket.bind(('', 19157))
```

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.

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.

Suppose a process in Host A, with UDP port 19157, wants to send a chunk of application data to a process with UDP port 46428 in Host B. The transport layer in Host A creates a transport-layer segment that includes the application data, the source port number (19157), the destination port number (46428), and two other values. The transport layer then passes the resulting segment to the network layer. The network layer encapsulates the segment in an IP datagram and makes a best-effort attempt to deliver the segment to the receiving host. If the segment arrives at the receiving Host B, the transport layer at the receiving host examines the destination port number in the segment (46428) and delivers the segment to its socket identified by port 46428. Note that Host B could be running multiple processes, each with its own UDP socket and associated port number. As UDP segments arrive from the network, Host B directs (demultiplexes) each segment to the appropriate socket by examining the segment's destination 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

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.

Two arriving TCP segments with different source IP addresses or source port numbers will be directed to two different sockets.

- 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 with the lines:

  ```
  clientSocket = socket(AF_INET, SOCK_STREAM)
  clientSocket.connect((serverName, 12000))
  ```

- A connection-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. 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.

The server host may support many simultaneous TCP connection sockets, with each socket attached to a process, and with each socket identified by its own fourtuple. When a TCP segment arrives at the host, all four fields (source IP address, source port, destination IP address, destination port) are used to direct (demultiplex) the segment to the appropriate socket.

3.3 Connectionless Transport: UDP

At the very least, the transport layer has to provide a multiplexing/demultiplexing service in order to pass data between the network layer and the correct application-level process.

Aside from the multiplexing/demultiplexing function and some light error checking, it adds nothing to IP. 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, it might try resending the query, try sending the query to another name server, or inform the invoking application that it can't get a reply.

Some 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.
- **No connection establishment.** 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. Indeed, the QUIC protocol (Quick UDP Internet Connection) uses UDP as its underlying transport protocol and implements reliability in an application-layer protocol on top of UDP.
- **No connection state.** UDP 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.** UDP has only 8 bytes overhead.

UDP is used to carry network management data. UDP is preferred to TCP in this case, since network management applications must often run when the network is in a stressed state—precisely when reliable, congestion-controlled data transfer is difficult to achieve.

When packet loss rates are low, and with some organizations blocking UDP traffic for security reasons, TCP becomes an increasingly attractive protocol for streaming media transport.

Running multimedia applications over UDP needs to be done with care. As we mentioned above, UDP has no congestion control. But congestion control is needed to prevent the network from entering a congested state in which very little useful work is done. Thus, the lack of congestion control in UDP can result in high loss rates between a UDP sender and receiver, and the crowding out of TCP sessions.

We mention that it is possible for an application to have reliable data transfer when using UDP. This can be done if reliability is built into the application itself. We mentioned earlier that the QUIC protocol implements reliability in an application-layer protocol on top of UDP. Nevertheless, building reliability directly into the application allows the application to communicate reliably without being subjected to the transmission-rate constraints imposed by TCP’s congestion-control mechanism.

3.3.1 UDP Segment Structure

The UDP header has only four fields, each consisting of two bytes. The length field specifies the number of bytes in the UDP segment (header plus data). The checksum is used by the receiving host to check whether errors have been introduced into the segment.

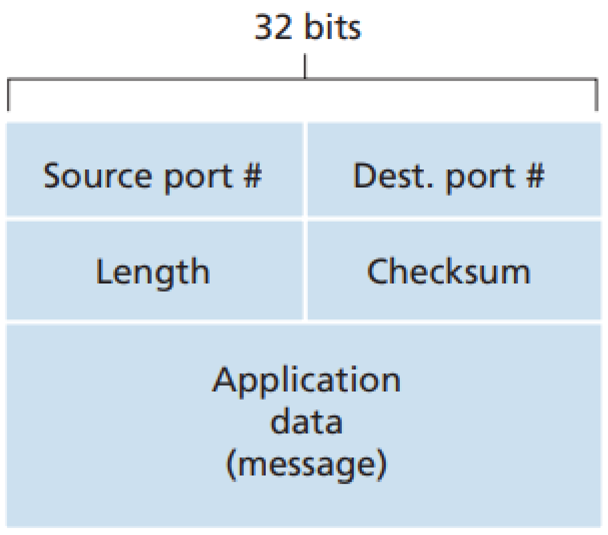


Figure 3.7 ♦ UDP segment structure

3.3.2 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 as it moved from source to destination.

You may wonder why UDP provides a checksum in the first place, as many link-layer protocols also provide error checking. The reason is that there is no guarantee that all the links between source and destination provide error checking; that is, one of the links may use a link-layer protocol that does not provide error checking. Even if segments are correctly transferred across a link, it's possible that bit errors could be introduced when a segment is stored in a router's memory.

Because IP is supposed to run over just about any layer-2 protocol, it is useful for the transport layer to provide error checking as a safety measure. Although UDP provides error checking, it does not do anything to recover from an error. Some implementations of UDP simply discard the damaged segment; others pass the damaged segment to the application with a warning.

3.4 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.

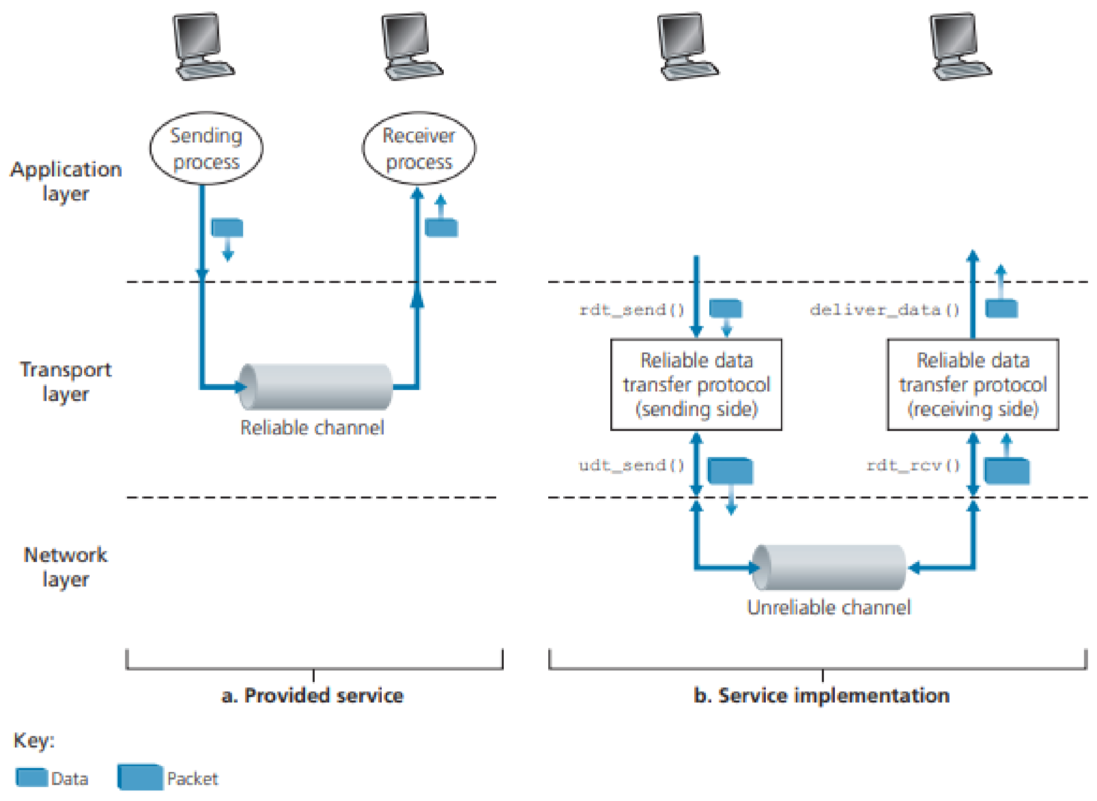


Figure 3.8 ♦ Reliable data transfer: Service model and service implementation

The sending side of the data transfer protocol will be invoked from above by a call to rdt_send(). It will pass the data to be delivered to the upper layer at the receiving side. (Here rdt stands for reliable data transfer protocol and _send indicates that the sending side of rdt is being called). On the receiving side, rdt_rcv() will be called when a packet arrives from the receiving side of the channel. When the rdt protocol wants to deliver data to the upper layer, it will do so by calling deliver_data().

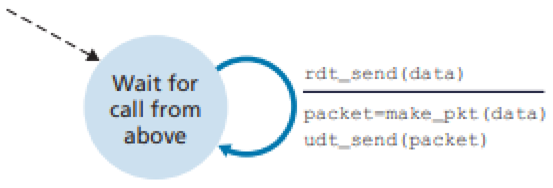
We consider only the case of unidirectional data transfer, that is, data transfer from the sending to the receiving side. The case of reliable bidirectional (that is, full-duplex) data transfer is conceptually no more difficult but considerably
more tedious to explain. We will see shortly that, in addition to exchanging packets containing the data to be transferred,
the sending and receiving sides of rdt will also need to exchange control packets back and forth. Both the send and receive sides of rdt send packets to the other side by a call to udt_send() (where udt stands for unreliable data transfer).

3.4.1 Build a Reliable Data Transfer Protocol

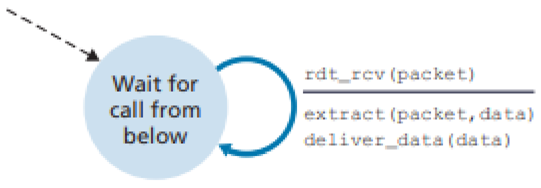
Reliable Data Transfer over a Perfectly Reliable Channel: rdt1.0

We first consider the simplest case, in which the underlying channel is completely reliable. It is important to note that there are separate FSMs for the sender and for the receiver. 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. When no action is taken on an event, or no event occurs and an action is taken, we'll use the symbol Λ below or above the horizontal, respectively, to explicitly denote the lack of an action or event

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.



a. rdt1.0: sending side



b. rdt1.0: receiving side

Figure 3.9 ♦ `rdt1.0`—A protocol for a completely reliable channel

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)).

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

Such bit errors typically occur in the physical components of a network as a packet is transmitted, propagates, or is buffered. We'll continue to assume for the moment that all transmitted packets are received in the order in which they were sent.

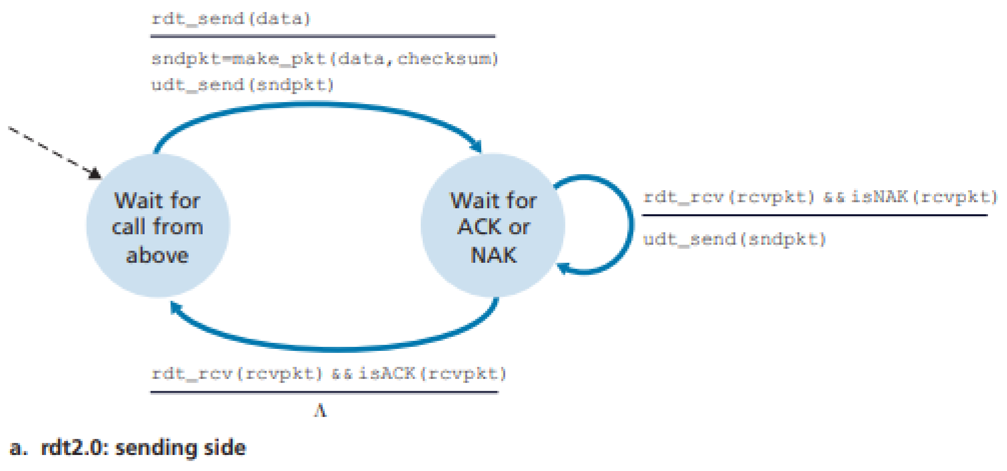
This message-dictation protocol uses both positive acknowledgments ("OK") and negative acknowledgments ("Please repeat that."). These control messages allow the receiver to let the sender know what has been received correctly, and what has been received in error and thus requires repeating. In a computer network setting, reliable data transfer protocols based on such retransmission are known as ARQ (Automatic Repeat reQuest) protocols.

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.

- Receiver feedback. Since the sender and receiver are typically executing on different end systems, the only way for the sender to learn of the receiver’s view of the world is for the receiver to provide explicit feedback
to the sender. The positive (ACK) and negative (NAK) acknowledgment replies in the message-dictation scenario are examples of such feedback.
- Retransmission. A packet that is received in error at the receiver will be retransmitted by the sender.

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.



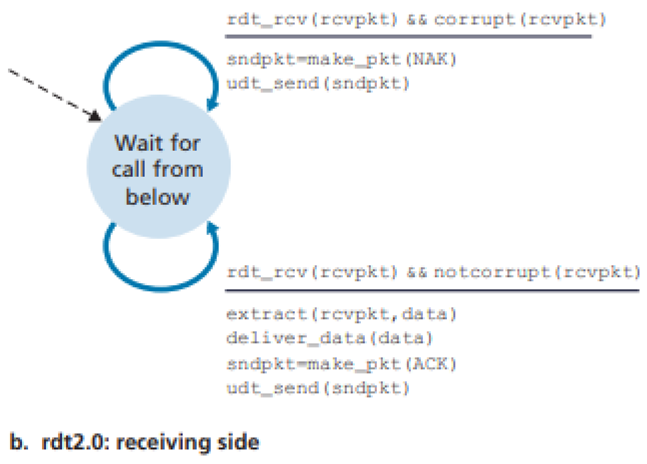


Figure 3.10 ♦ `rdt2.0`—A protocol for a channel with bit errors

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; 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.

It has a fatal flaw. We haven't accounted for the possibility that the ACK or NAK packet could be corrupted! Minimally, we will need to add checksum bits to ACK/NAK packets in order to detect such errors. The more difficult question is how the protocol should recover from errors in ACK or NAK packets. The difficulty here is that if an ACK or NAK is corrupted, the sender has no way of knowing whether or not the receiver has correctly received the last piece of transmitted data.

Consider three possibilities for handling corrupted ACKs or NAKs:

- 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

A simple solution to this new problem is to add a new field to the data packet and have the sender number its data packets by putting a sequence number into this field. The receiver then need only check this sequence number to determine whether or not the received packet is a retransmission. For this simple case of a stop-and-

wait protocol, a 1-bit sequence number will suffice, since it will allow the receiver to know whether the sender is resending the previously transmitted packet or a new packet (the sequence number changes, moving “forward” in modulo-2 arithmetic). Since we are currently assuming a channel that does not lose packets, ACK and NAK packets do not themselves need to indicate the sequence number of the packet they are acknowledging. The sender knows that a
received ACK or NAK packet (whether garbled or not) was generated in response to its most recently transmitted data packet.

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.

Protocol rdt2.1 uses both positive and negative acknowledgments from the receiver to the sender. When an out-of-order packet is received, the receiver sends
a positive acknowledgment for the packet it has received. When a corrupted packet is received, the receiver sends a negative acknowledgment. We can accomplish the same effect as a NAK if, instead of sending a NAK, we send an ACK for the last correctly received packet. A sender that receives two ACKs for the same packet (that is, receives duplicate ACKs) knows that the receiver did not correctly receive the
packet following the packet that is being ACKed twice.

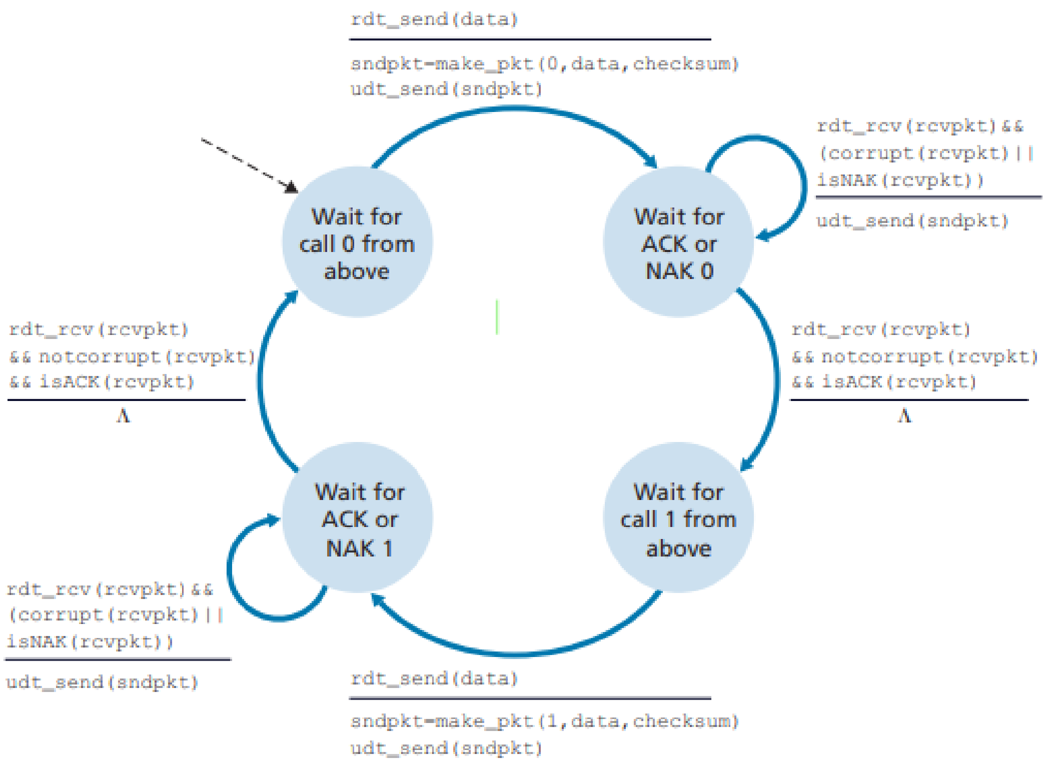


Figure 3.11 ♦ rdt2.1 sender

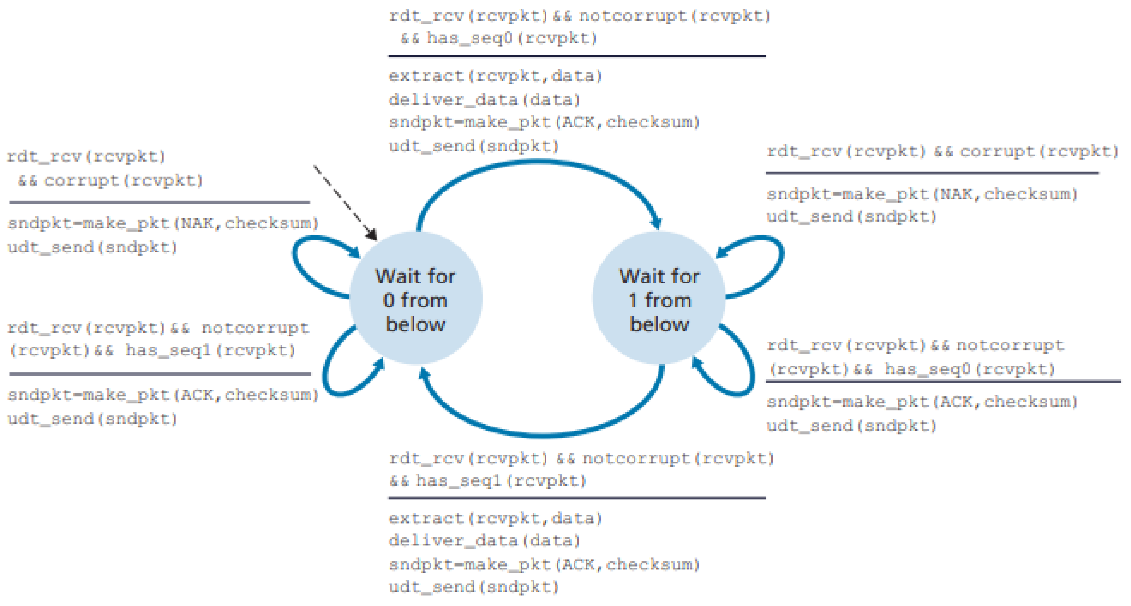

rdt_rcv(rcvpkt)&& notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
sndpkt=make_pkt(ACK,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt)
&& corrupt(rcvpkt)
sndpkt=make_pkt(NAK,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && corrupt(rcvpkt)
sndpkt=make_pkt(NAK,checksum)
udt_send(sndpkt)
Wait for 0 from below
Wait for 1 from below
rdt_rcv(rcvpkt)&& notcorrupt
(rcvpkt)&& has_seq1(rcvpkt)
sndpkt=make_pkt(ACK,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt)&& notcorrupt
(rcvpkt)&& has_seq0(rcvpkt)
sndpkt=make_pkt(ACK,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
sndpkt=make_pkt(ACK,checksum)
udt_send(sndpkt)

Figure 3.12 ♦ rdt2.1 receiver

Our NAK-free reliable data transfer protocol for a channel with bit errors is rdt 2.2.
One subtle change between rtdt2.1 and rdt2.2 is that the receiver must
now include the sequence number of the packet being acknowledged by an ACK

message (this is done by including the ACK, 0 or ACK, 1 and the sender must now check the sequence number of the
packet being acknowledged by a received ACK message (this is done by including the 0 or 1 argument in isACK() in the sender FSM).

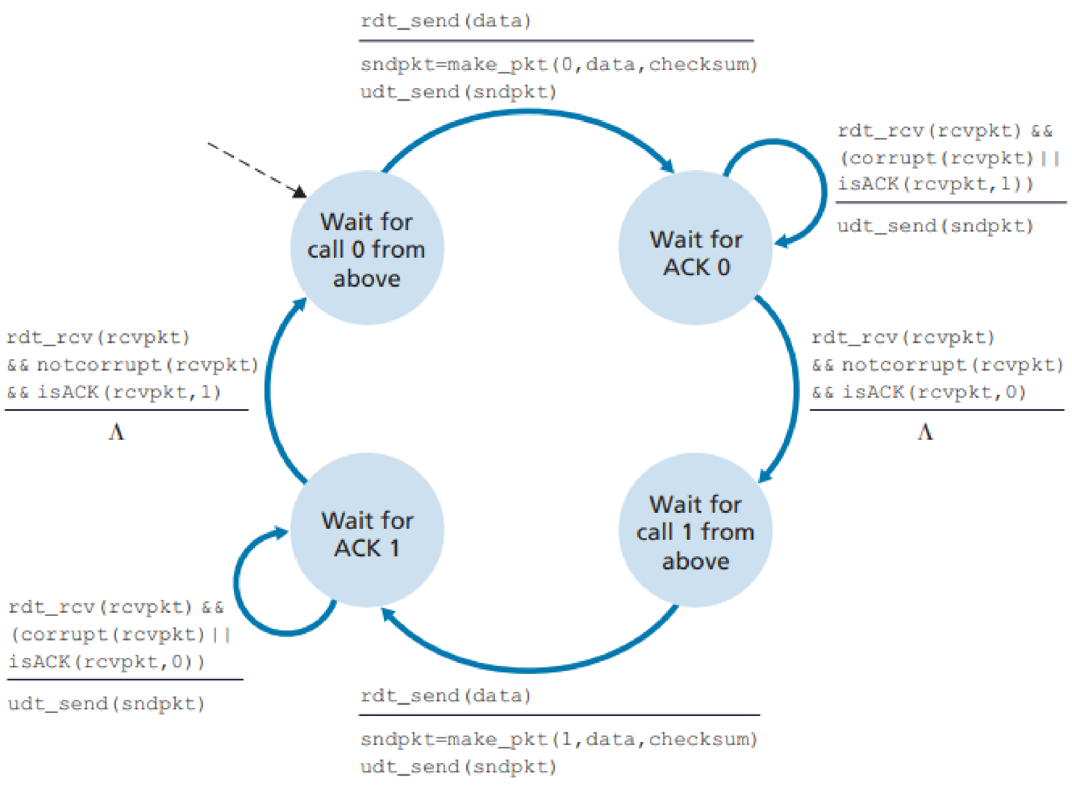
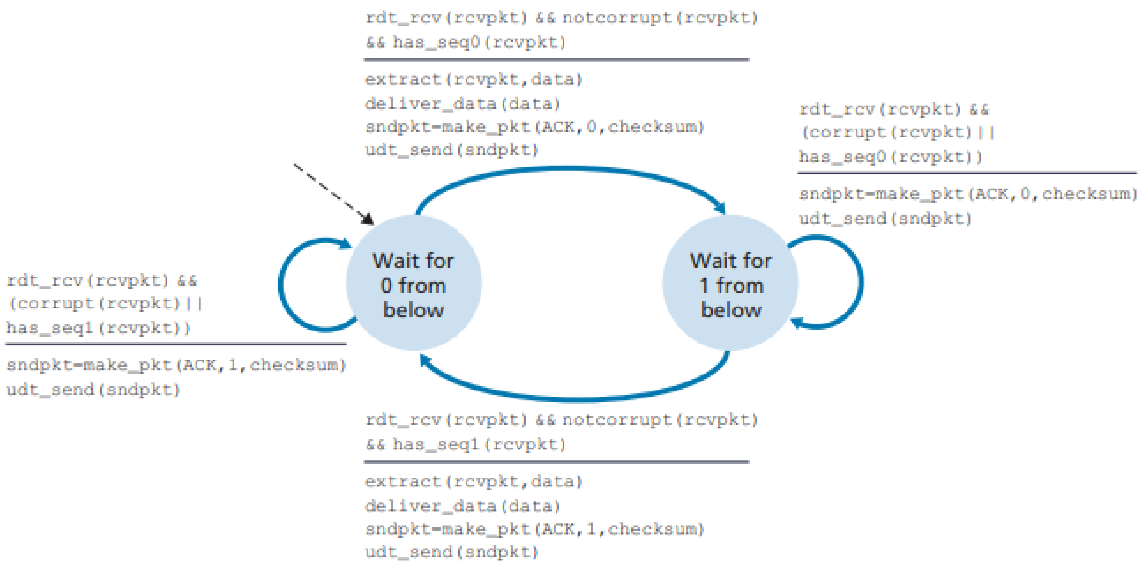

rdt_send(data)
sndpkt=make_pkt(0,data,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isACK(rcvpkt,1))
udt_send(sndpkt)
Wait for call 0 from above
Wait for ACK 0
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,1)
Λ
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,0)
Λ
Wait for ACK 1
Wait for call 1 from above
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isACK(rcvpkt,0))
udt_send(sndpkt)
rdt_send(data)
sndpkt=make_pkt(1,data,checksum)
udt_send(sndpkt)

Figure 3.13 ♦ rdt2.2 sender



rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && has_seq0(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
sndpkt=make_pkt(ACK,0,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)|| has_seq0(rcvpkt))
sndpkt=make_pkt(ACK,0,checksum)
udt_send(sndpkt)
Wait for 0 from below
Wait for 1 from below
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)|| has_seq1(rcvpkt))
sndpkt=make_pkt(ACK,1,checksum)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && has_seq1(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
sndpkt=make_pkt(ACK,1,checksum)
udt_send(sndpkt)

Figure 3.14 ♦ `rdt2.2` receiver

Reliable Data Transfer over a Lossy Channel with Bit Errors: rdt 3.0

Suppose now that in addition to corrupting bits, the underlying channel can lose packets as well. 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 rdt 2.2—will allow us to answer the latter concern. Here, we'll put the burden of detecting and recovering from lost packets on the sender. If the sender is willing to wait long enough so that it is certain that a packet has been lost it can simply retransmit the data packet.

But how long must the sender wait to be certain that something has been lost? The sender must clearly wait at least as long as a round-trip between the sender and receiver plus whatever amount of time is needed to process a packet at the receiver. The protocol should ideally recover from packet loss as soon as possible; waiting for the worst-cacse delay could mean a long wait until error recovery is initiated. The approach thus adopted in practice is for the sender to judiciously choose a time value such that packet loss is likely, although not guaranteed, to have happened. If an ACK is not received within this time, the packet is retransmitted. Note that if a packet experiences a particularly large delay, the sender may retransmit the packet even though neither the data packet nor its ACK have been lost. This introduces the possibility of duplicate data packets in the sender-to-receiver channel. Happily, protocol rdt2.2 already has enough functionality to handle the case of duplicate packets.

From the sender’s viewpoint, retransmission is a panacea. The sender does not know whether a data packet was lost, an ACK was lost, or if the packet or ACK was simply overly delayed. 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 is sent, (2) respond to a timer interrupt (taking appropriate actions), and (3) stop the timer.

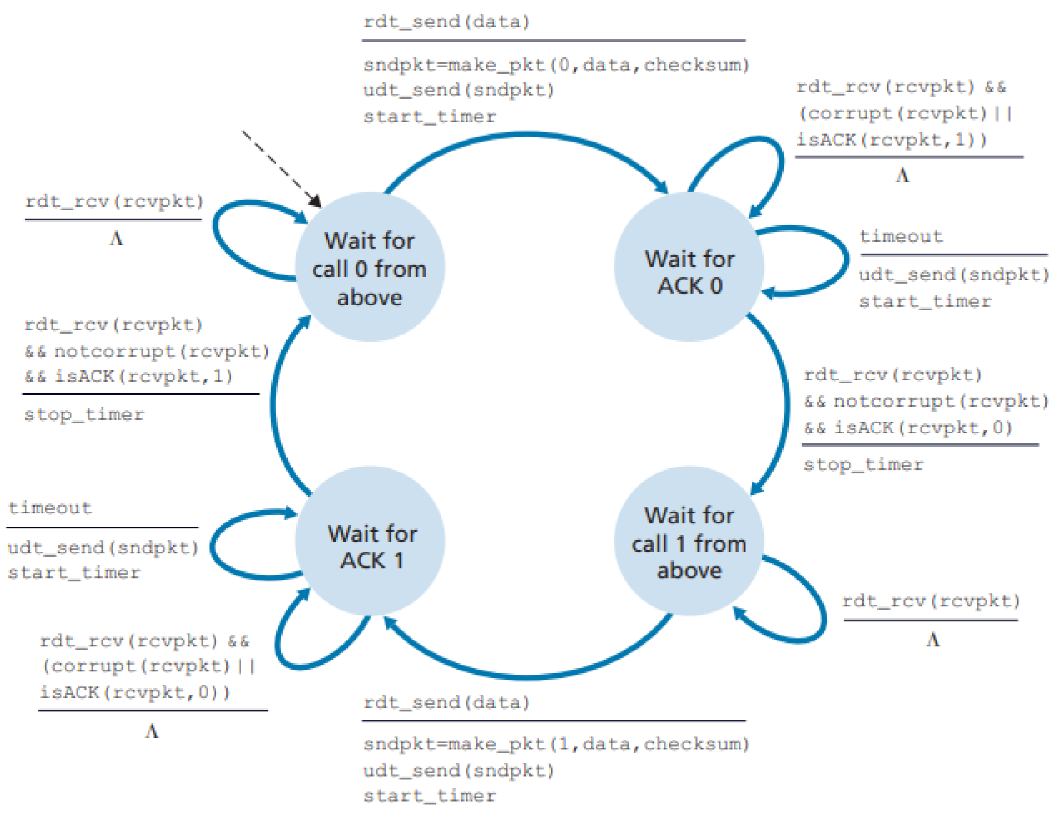

rdt_send(data)
sndpkt=make_pkt(0,data,checksum)
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)|| isACK(rcvpkt,1))
Λ
rdt_rcv(rcvpkt)
Λ
Wait for call 0 from above
Wait for ACK 0
timeout
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,1)
stop_timer
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,0)
stop_timer
timeout
udt_send(sndpkt)
start_timer
Wait for ACK 1
Wait for call 1 from above
rdt_rcv(rcvpkt)
Λ
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)|| isACK(rcvpkt,0))
Λ
rdt_send(data)
sndpkt=make_pkt(1,data,checksum)
udt_send(sndpkt)
start_timer

Figure 3.15 ♦ rdt3.0 sender

Because packet sequence numbers alternate between 0 and 1, protocol rdt3.0 is sometimes known as the alternating-bit protocol.
We have now assembled the key elements of a data transfer protocol. Checksums, sequence numbers, timers, and positive and negative acknowledgment packets each play a crucial and necessary role in the operation of the protocol.

3.4.2 Pipelined Reliable Data Transfer Protocols

At the heart of rdt3.0’s performance problem is the fact that it is a stop-and-wait protocol.

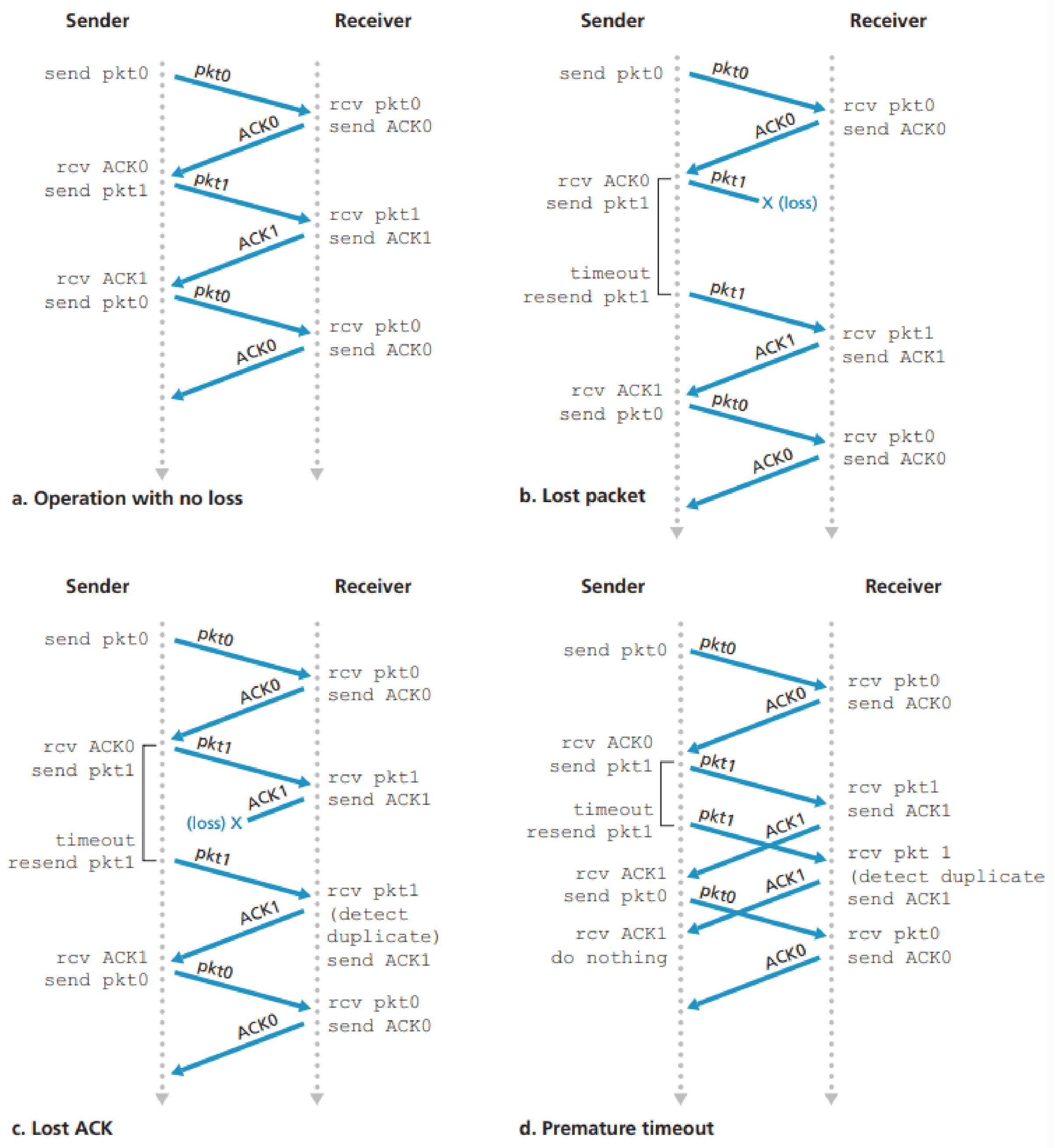


Figure 3.16 ♦ Operation of `rdt3.0`, the alternating-bit protocol

$$dtrans = L/R$$

Assuming for simplicity that ACK packets are extremely small (so that we can ignore their transmission time) and that the receiver can send an ACK as soon as the last bit of a data packet is received, the ACK emerges back at the sender at t = RTT + L/R. At this point, the sender can now transmit the next message.

$$U_{\text{sender}} = \frac{L/R}{RTT + L/R}$$

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.

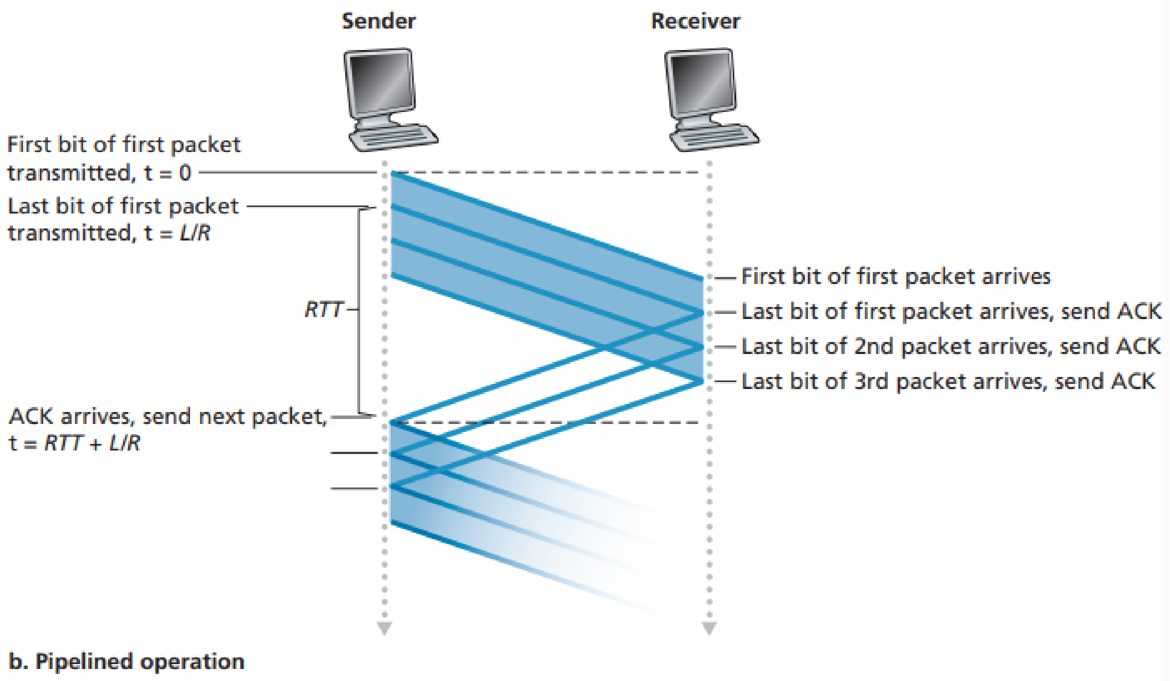

Sender
Receiver
First bit of first packet transmitted, t = 0
Last bit of first packet transmitted, t = L/R
RTT
ACK arrives, send next packet, t = RTT + L/R
First bit of first packet arrives
Last bit of first packet arrives, send ACK
Last bit of 2nd packet arrives, send ACK
Last bit of 3rd packet arrives, send ACK
b. Pipelined operation

Figure 3.18 ♦ Stop-and-wait and pipelined sending

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.

3.4.3 Go-Back-N (GBN)

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

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.

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. You might be wondering why we would even limit the number of outstanding, unacknowledged packets to a value of N in the first place. Flow control is one reason to impose a limit on the sender.

In practice, a packet’s sequence number is carried in a fixed-length field in the packet header. If k is the number of bits in the packet sequence number field, the range of sequence numbers is thus [0,2k - 1]. With a finite range of sequence numbers, all arithmetic involving sequence numbers must then be done using modulo 2^k arithmetic.

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.
- *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.
- *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. 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.

The receiver's actions in GBN are also simple. If a packet with sequence number n is received correctly and is in order the receiver sends an ACK for packet n and delivers the data portion of the packet to the upper layer. In all other cases, the receiver discards the packet and resends an ACK for the most recently received in-order packet. Note that since packets are delivered one at a time to the upper layer, if packet k has been received and delivered, then all packets with a sequence number lower than k have also been delivered. Thus, the use of cumulative acknowledgments is a natural choice for GBN.

In our GBN protocol, the receiver discards out-of-order packets. Recall that the receiver must deliver data in order to the upper-layer. Recall that the receiver must deliver data in order to the upper layer. Suppose now that packet n is expected, but packet n + 1 arrives. Because data must be delivered in order, the receiver could buffer (save) packet n + 1 and then deliver this packet to the upper layer after it had later received and delivered packet n. However, if packet n is lost, both it and packet n + 1 will eventually be retransmitted as a result of the GBN retransmission rule at the sender. Thus, the receiver can simply discard packet n + 1. The
advantage of this approach is the simplicity of receiver buffering—the receiver need not buffer any out-of-order packets. Thus, while the sender must maintain the upper

and lower bounds of its window and the position of nextseqnum within this window, the only piece of information the receiver need maintain is the sequence number of the next in-order packet. This value is held in the variable expectedseqnum. Of course, the disadvantage of throwing away a correctly received packet is that the subsequent retransmission of that packet might be lost or garbled and thus even more retransmissions would be required.

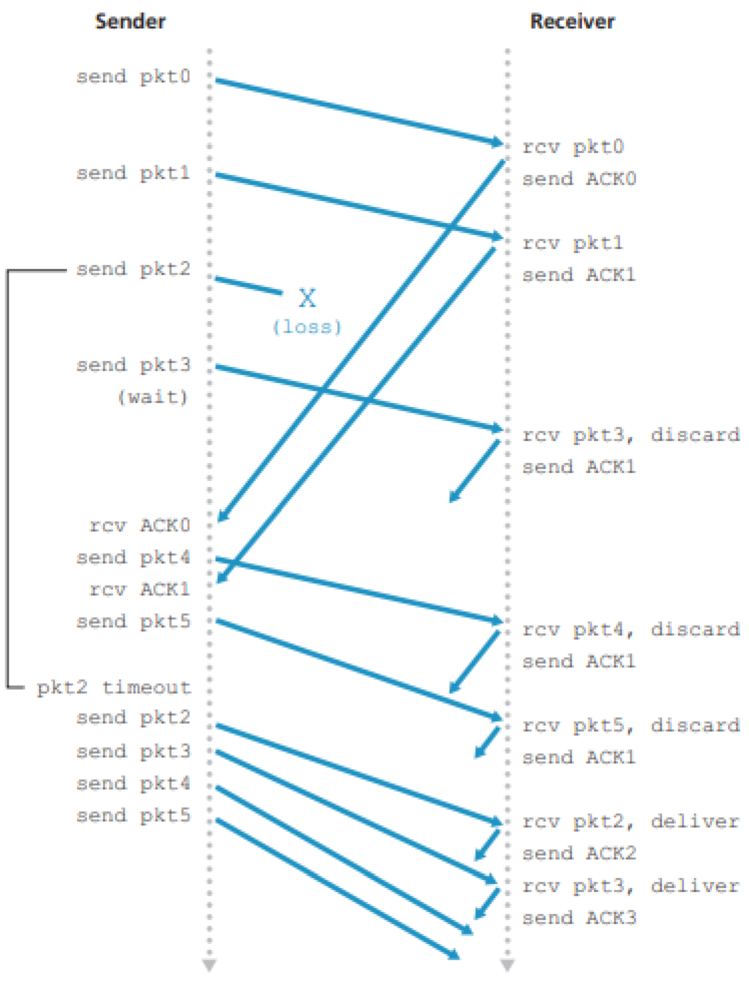


Figure 3.22 ♦ Go-Back-N in operation

3.4.4 Selective Repeat (SR)

There are, however, scenarios in which GBN itself suffers from performance problems. In particular, when the window size and bandwidth-delay product are both large, many packets can be in the pipeline. A single packet error can thus cause GBN to retransmit a large number of packets, many unnecessarily.

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 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 number of outstanding, unacknowledged packets in the pipeline. However, unlike GBN, the sender will have already received ACKs for some of the packets in the window.

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 are received.

SR sender events and actions:

1. Data received from above. When data is received from above, the SR sender checks the next available sequence number for the packet. If the sequence number is within the sender's window, the data is packetized and sent; otherwise it is either buffered or returned to the upper layer for later transmission, as in GBN.
2. *Timeout*. Timers are again used to protect against lost packets. However, each packet must now have its own logical timer, since only a single packet will be transmitted on timeout.
3. ACK received. If an ACK is received, the SR sender marks that packet as having been received, provided it is in the window. If the packet's sequence number is equal to send_base, the window base is moved forward to the unacknowledged packet with the smallest sequence number. If the window moves and there are untransmitted packets with sequence numbers that now fall within the window, these packets are transmitted.

SR receiver events and actions:

1. Packet with sequence number in [rcv_base, rcv_base+N-1] is correctly received. In this case, the received packet falls within the receiver's window and a selective ACK packet is returned to the sender. If the packet was not previously received, it is buered. If this packet has a sequence number equal to the base of the receive window (rcv_base in Figure 3.22), then this packet, and any previously buered and consecutively numbered (beginning with rcv_base) packets are delivered to the upper layer. The receive window is then moved forward by the number of packets delivered to the upper layer.
2. Packet with sequence number in [rcv_base-N, rcv_base-1] is correctly received. In this case, an ACK must be generated, even though this is a packet that the

receiver has previously acknowledged.

3. Otherwise. Ignore the packet

It is important to note that in Step 2 in Figure 3.25, the receiver reacknowledges (rather than ignores) already received packets with certain sequence numbers below the current window base.

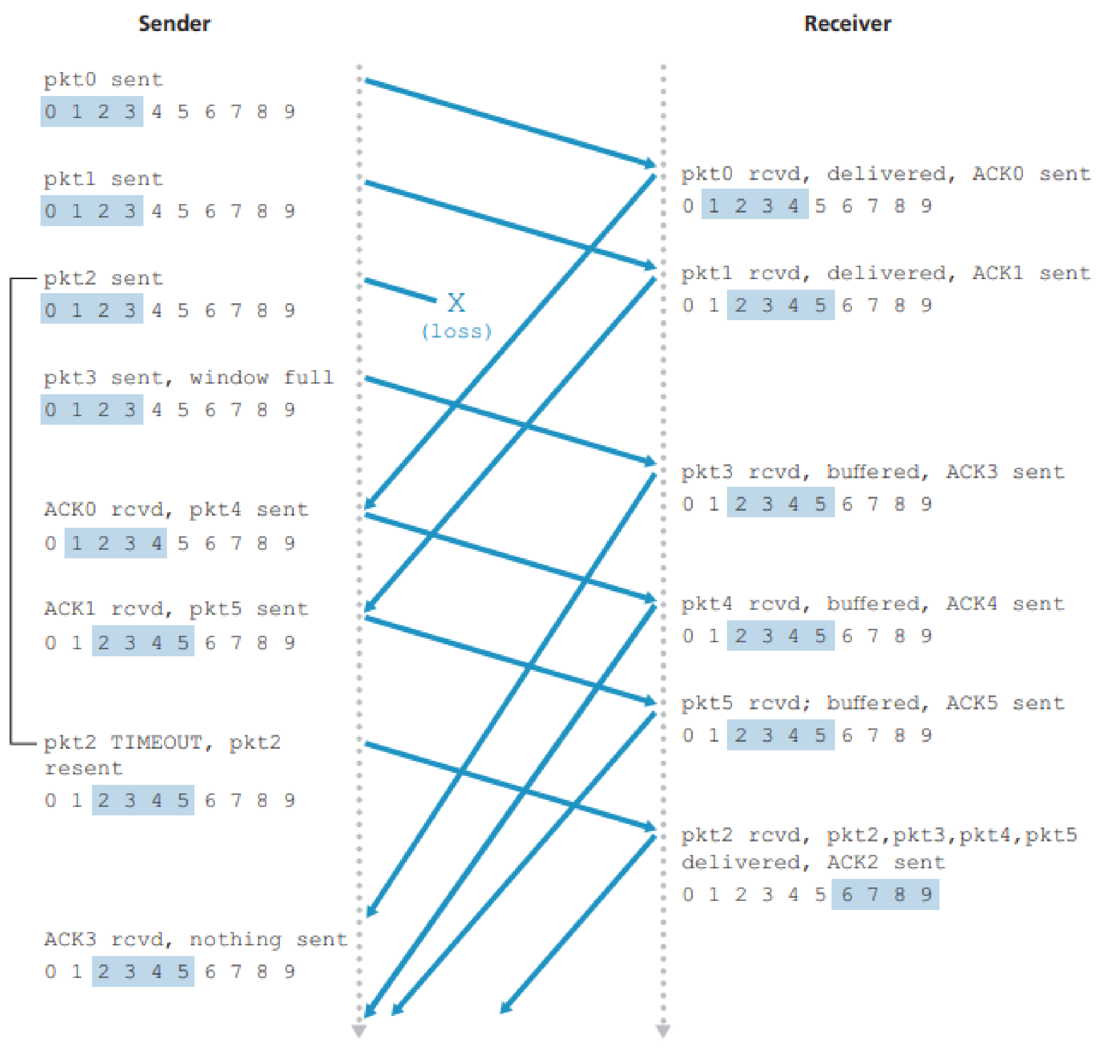
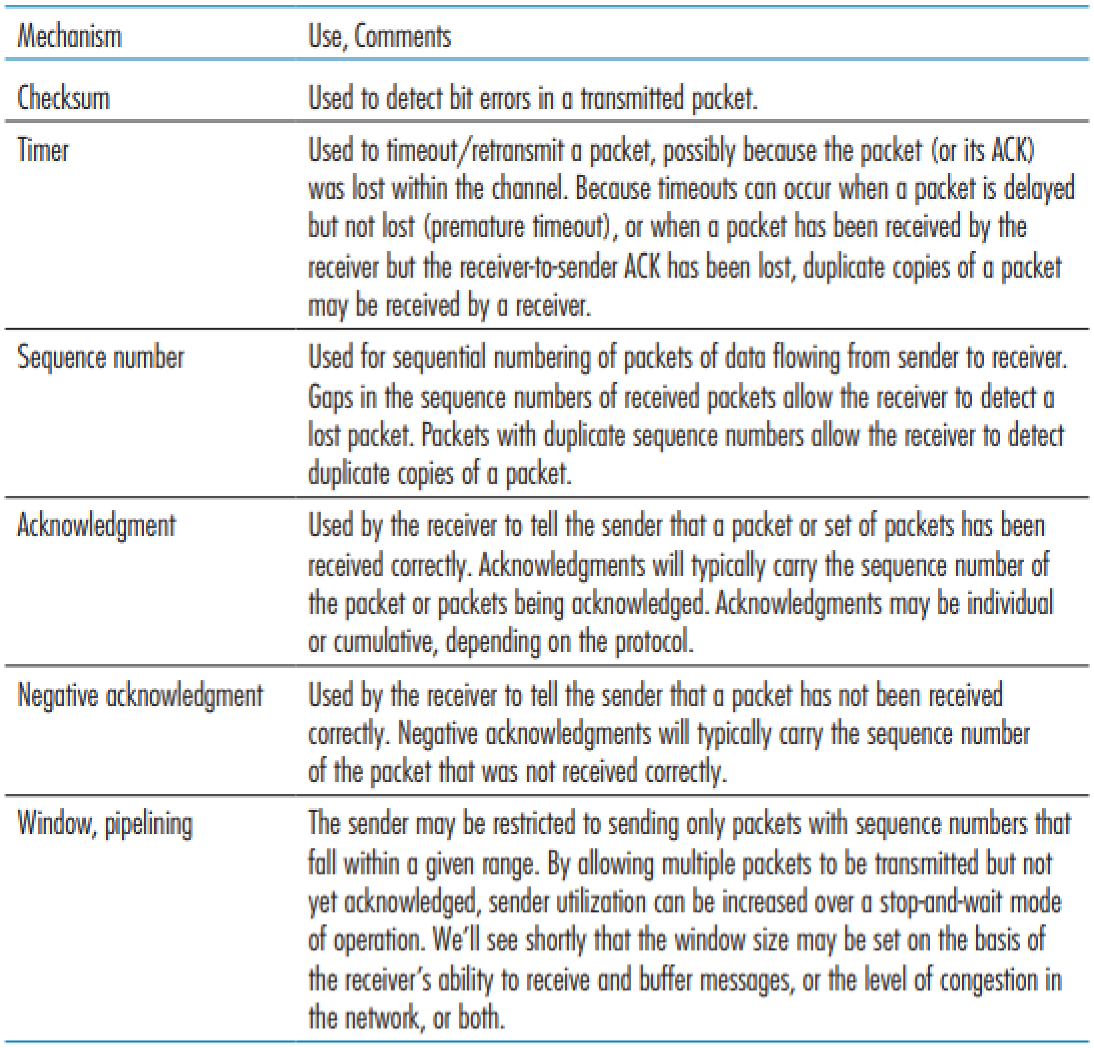


Figure 3.26 ♦ SR operation

the receiver to the sender, the sender will eventually retransmit packet send_base, even though it is clear that the receiver has already received that packet. If the receiver were not to acknowledge this packet, the sender's window would never move forward!

The lack of synchronization between sender and receiver windows has important consequences when we are faced with the reality of a finite range of sequence numbers.



Mechanism	Use, Comments
Checksum	Used to detect bit errors in a transmitted packet.
Timer	Used to timeout/retransmit a packet, possibly because the packet (or its ACK) was lost within the channel. Because timeouts can occur when a packet is delayed but not lost (premature timeout), or when a packet has been received by the receiver but the receiver-to-sender ACK has been lost, duplicate copies of a packet may be received by a receiver.
Sequence number	Used for sequential numbering of packets of data flowing from sender to receiver. Gaps in the sequence numbers of received packets allow the receiver to detect a lost packet. Packets with duplicate sequence numbers allow the receiver to detect duplicate copies of a packet.
Acknowledgment	Used by the receiver to tell the sender that a packet or set of packets has been received correctly. Acknowledgments will typically carry the sequence number of the packet or packets being acknowledged. Acknowledgments may be individual or cumulative, depending on the protocol.
Negative acknowledgment	Used by the receiver to tell the sender that a packet has not been received correctly. Negative acknowledgments will typically carry the sequence number of the packet that was not received correctly.
Window, pipelining	The sender may be restricted to sending only packets with sequence numbers that fall within a given range. By allowing multiple packets to be transmitted but not yet acknowledged, sender utilization can be increased over a stop-and-wait mode of operation. We'll see shortly that the window size may be set on the basis of the receiver's ability to receive and buffer messages, or the level of congestion in the network, or both.

Table 3.1 ♦ Summary of reliable data transfer mechanisms and their use

3.5 Connection-Oriented Transport: TCP

3.5.1 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 a logical one, with common state
residing only in the TCPs in the two communicating end systems. Recall that 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. In fact, the intermediate routers are completely oblivious to TCP connections; they see datagrams, not connections.

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 application-layer 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” is not possible with TCP. With TCP, two hosts are company and three are a crowd!

```
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. 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. Once the data passes through the door, the data is in the hands of TCP running in the client. 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.

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 maximum transmission 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.

TCP pairs each chunk of client data with a TCP header, thereby forming TCP segments. The segments are passed down to the network layer, where they are separately encapsulated within network-layer IP datagrams. The IP datagrams are then sent into the network. When TCP receives a segment at the other end, the segment's
data is placed in the TCP connection's receive buffer. The application reads the stream of data from this buffer. Each side of the connection has its own send buffer and its own receive buffer.

3.5.2 TCP Segment Structure

The TCP segment consists of header fields and a data field. The data field contains a chunk of application data. The MSS limits the maximum size of a segment's data field. When TCP sends a large file, such as an image as part of a Web page, it typically breaks the file into chunks of size MSS.

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:

1. 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.
2. The 16-bit **receive window** field is used for flow control.
3. 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.
4. 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.
5. The **flag field** contains 6 bits. The **ACK** bit is used to indicate that the value caried 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. The **CWR** and **ECE** bits are used in explicit congestion notification. 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 upperlayer 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**.

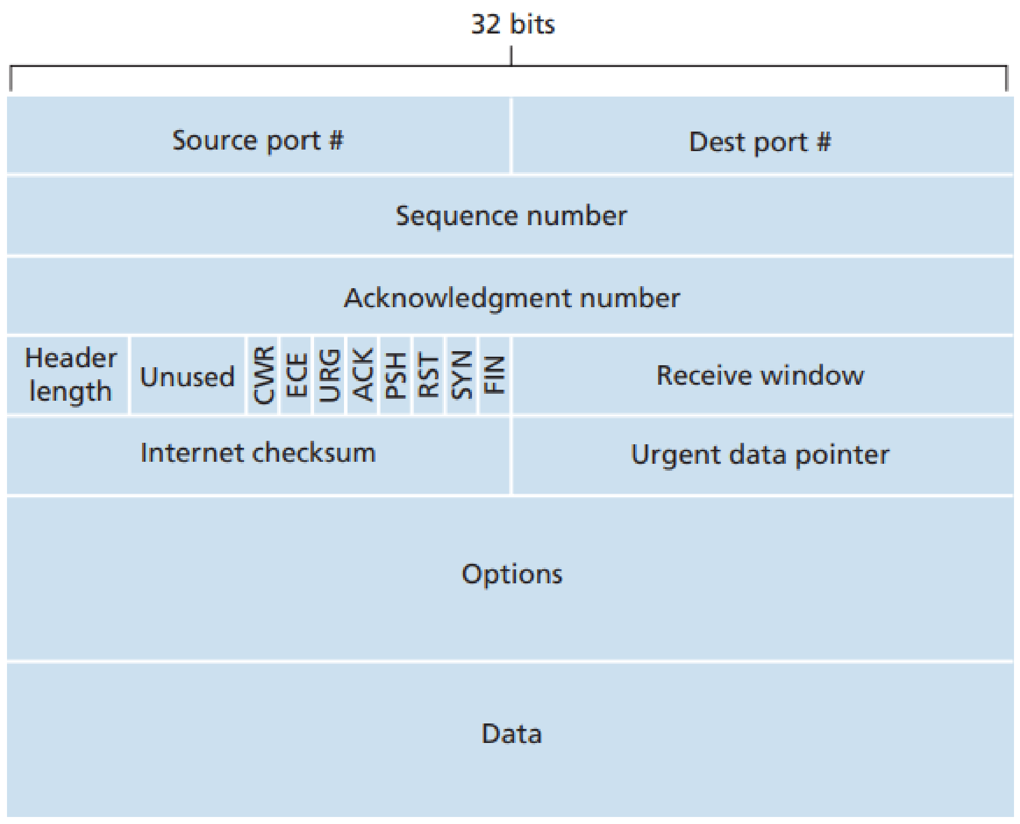


Figure 3.29 ♦ TCP segment structure

Sequence Numbers and Acknowledgment Numbers

Two of the most important fields in the TCP segment header are the sequence number field and the acknowledgment number field. These fields are a critical part of TCP's reliable data transfer service.

TCP views data as an unstructured, but ordered, stream of bytes. TCP's use of sequence numbers reflects this view in that sequence numbers are over the stream of transmitted bytes and not over the series of transmitted segments. **The sequence**

number for a segment is therefore the byte-stream number of the first byte in the segment.

Example given:

> The TCP in Host A will
> implicitly number each byte in the data stream. Suppose that the data stream consists of a file consisting of 500,000 bytes, that the MSS is 1,000 bytes, and that the first byte of the data stream is numbered 0. The first segment gets assigned sequence number 0, the second segment gets assigned sequence number 1,000, the third segment gets assigned sequence number 2,000, and so on.

Because TCP only acknowledges bytes up to the first missing byte in the stream, TCP is said to provide cumulative acknowledgments.

> Example given:
>
> As another example, suppose that Host A has received one segment from Host B containing bytes 0 through 535 and another segment containing bytes 900 through 1,000. For some reason Host A has not yet received bytes 536 through 899. In this example, Host A is still waiting for byte 536 (and beyond) in order to re-create B's
> data stream. Thus, A's next segment to B will contain 536 in the acknowledgment number field.

3.5.3 Round-Trip Time Estimation and Timeout

TCP uses a timeout/retransmit mechanism to recover from lost segments. Perhaps the most obvious question is the length of the timeout intervals. Clearly, the timeout should be larger than the connection's round-trip time (RTT),
that is, the time from when a segment is sent until it is acknowledged. Otherwise, unnecessary retransmissions would be sent. But how much larger?

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. 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.

$$\texttt{EstimatedRTT} = (1 - \alpha) \cdot \texttt{EstimatedRTT} + \alpha \cdot \texttt{SampleRTT}$$

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 **α is α = 0.125**

Note that EstimatedRTT is a weighted average of the SampleRTT values. This weighted average puts more weight on recent samples than on old samples. This is natural, as the more recent samples better reflect the current congestion in the network. In statistics, such an average is called an exponential weighted moving average (EWMA).

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:

$$\texttt{DevRTT} = (1 - \beta) \cdot \texttt{DevRTT} + \beta \cdot | \texttt{SampleRTT} - \texttt{EstimatedRTT} |$$

Setting and Managing the Retrnasmission Timeout Interval

Given values of EstimatedRTT and DevRTT, what value should be used for TCP’s timeout interval? Clearly, the interval should be greater than or equal to EstimatedRTT, or unnecessary retransmissions would be sent. But the timeout interval should not be too much larger than EstimatedRTT; otherwise, when a segment is lost, TCP would not quickly retransmit the segment, leading to large data transfer delays. It is therefore desirable to set the timeout equal to the EstimatedRTT plus some margin. The margin should be large when there is a lot of fluctuation in the SampleRTT values; it should be small when there is little fluctuation.

$$\texttt{TimeoutInterval} = \texttt{EstimatedRTT} + 4 \cdot \texttt{DevRTT}$$

Also, when a timeout occurs, the value of TimeoutInterval is doubled to avoid a premature timeout occurring for a subsequent segment that will soon be acknowledged. However, as soon as a segment is received and EstimatedRTT is updated, the TimeoutInterval is again computed using the formula above.

3.5.4 Reliable Data Transfer

IP does not guarantee datagram delivery, does not guarantee in-order delivery of datagrams, and does not guarantee the integrity of the data in the datagrams. With IP service, datagrams can overflow router buffers and never reach their destination, datagrams can arrive out of order, and bits in the datagram can get corrupted.

TCP creates a reliable data transfer service on top of IP’s unreliable besteffort 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.

We will discuss how TCP provides reliable data transfer in two incremental steps. We first present a highly simplified description of a TCP sender that uses only timeouts to recover from lost segments; we then present a more complete description that uses duplicate acknowledgments in addition to timeouts.

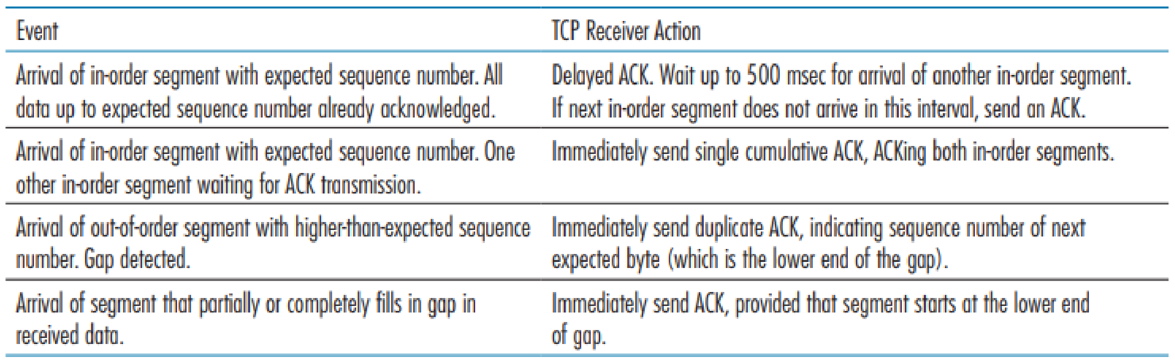
```
/* Assume sender is not constrained by TCP flow or congestion control, that data from above is less
than MSS in size, and that data transfer is in one direction only. */

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 3.33 ♦ Simplified TCP sender

Figure 3.33 presents a highly simplified description of a TCP sender. We see that there are three major events related to data transmission and retransmission in the TCP sender: data received from application above; timer timeout; and ACK receipt. Upon the occurrence of the first major event, TCP receives data from the application, encapsulates the data in a segment, and passes the segment to IP. Note that each segment includes a sequence number that is the byte-stream number of the first data byte in the segment.

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.

The third major event that must be handled by the TCP sender is the arrival of an acknowledgment segment (ACK) from the receiver. On the occurrence of this event, TCP compares the ACK value y with its variable SendBase. The TCP state variable SendBase is the sequence number of the oldest unacknowledged byte. (Thus SendBase–1 is the sequence number of the last byte that is known to have been received correctly and in order at the receiver.) As indicated earlier, TCP uses cumulative acknowledgments, so that y acknowledges the receipt of all bytes before byte number y. If y > SendBase, then the ACK is acknowledging one or more previously unacknowledged segments. Thus the sender updates its SendBase variable; it also restarts the timer if there currently are any not-yet-acknowledged segments.



Event	TCP Receiver Action
Arrival of in-order segment with expected sequence number. All data up to expected sequence number already acknowledged.	Delayed ACK. Wait up to 500 msec for arrival of another in-order segment. If next in-order segment does not arrive in this interval, send an ACK.
Arrival of in-order segment with expected sequence number. One other in-order segment waiting for ACK transmission.	Immediately send single cumulative ACK, ACKing both in-order segments.
Arrival of out-of-order segment with higher-than-expected sequence number. Gap detected.	Immediately send duplicate ACK, indicating sequence number of next expected byte (which is the lower end of the gap).
Arrival of segment that partially or completely fills in gap in received data.	Immediately send ACK, provided that segment starts at the lower end of gap.

Table 3.2 ♦ TCP ACK Generation Recommendation [RFC 5681]

Fast Retransmit

One of the problems with timeout-triggered retransmissions is that the timeout period can be relatively long. The sender can often detect packet loss well before the timeout event
occurs by noting so-called duplicate ACKs. A duplicate ACK is an ACK that reacknowledges a segment for which the sender has already received an earlier acknowledgment. To understand the sender's response to a duplicate ACK, we must look at why the receiver sends a duplicate ACK in the first place.

Since TCP does not use negative acknowledgments, the receiver cannot send an explicit negative acknowledgment back to the sender. Instead, it simply reacknowledges the last in-order byte of data it has received.

Because a sender often sends a large number of segments back to back, if one segment is lost, there will likely be many back-to-back duplicate ACKs. If the TCP sender receives three duplicate ACKs for the same data, it takes this as an indication that the segment following the segment that has been ACKed three times has been lost. In the case that three duplicate ACKs are received, the TCP sender performs a fast retransmit, retransmiting the missing segment before that segment's timer expires.

Go-Back-B or Selective Repeat?

Is TCP a GBN or an SR protocol? The TCP sender need only maintain the smallest sequence number of a transmitted but unacknowledged byte (SendBase) and the sequence number
of the next byte to be sent (NextSeqNum). In this sense, TCP looks a lot like a GBN-style protocol. But there are some striking differences between TCP and GoBack-N. Many TCP implementations will buffer correctly received but out-of-order segments. Further suppose that the acknowledgment for packet n 6 N gets lost, but the remaining N - 1 acknowledgments arrive at the sender before their respective timeouts. In this example, GBN would retransmit not only packet n, but also all of the subsequent packets n + 1, n + 2, . . . , N. TCP, on the other hand, would retransmit at most one segment, namely, segment n. Moreover, TCP would not even retransmit segment n if the acknowledgment for segment n + 1 arrived before the timeout for segment n.

A proposed modification to TCP, the so-called selective acknowledgment [RFC 2018], allows a TCP receiver to acknowledge out-of-order segments selectively rather than just cumulatively acknowledging the last correctly received, in-order segment. Thus, TCP's error-recovery mechanism
is probably best categorized as a hybrid of GBN and SR protocols.

3.5.5 Flow Control

Recall that the hosts on each side of a TCP connection set aside a receive buffer for the connection. When the TCP connection receives bytes that are correct and in sequence, it places the data in the receive buffer. The associated application process will read data from this buffer, but not necessarily at the instant the data arrives. If the application is relatively slow at reading the data, the sender can very easily overflow the connection's receive buffer by sending too much data too quickly.

TCP provides a flow-control service to its applications to eliminate the possibility of the sender overflowing the receiver's buffer. 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. 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 outof-order segments.

TCP provides flow control by having the sender maintain a variable called the receive window. 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.

- 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.

LastByteSent - LastByteAcked ≤ rwnd

There is one minor technical problem with this scheme. To see this, suppose Host B’s receive buffer becomes full so that rwnd = 0. After advertising rwnd = 0 to Host A, also suppose that B has nothing to send to A. Now consider what happens. As the application process at B empties the buffer, TCP does not send new segments with new rwnd values to Host A; indeed, TCP sends a segment to Host A only if it has data to send or if it has an acknowledgment to send. Therefore, Host A is never informed that some space has opened up in Host B’s receive buffer—Host A is blocked and can transmit no more data! To solve this problem, the TCP specification requires Host A to continue to send segments with one data byte when B’s receive window is zero. These segments will be acknowledged by the receiver. Eventually the buffer will begin to empty and the acknowledgments will contain a nonzero rwnd value.

3.5.6 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:

1. **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.

2. **Step 2.** Once the IP datagram containing the TCP SYN segment arrives at the server host the server extract 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. The connection-granted segment is referred to as a SYNACK segment.
3. 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. The SYN bit is set to zero, since the connection is established.

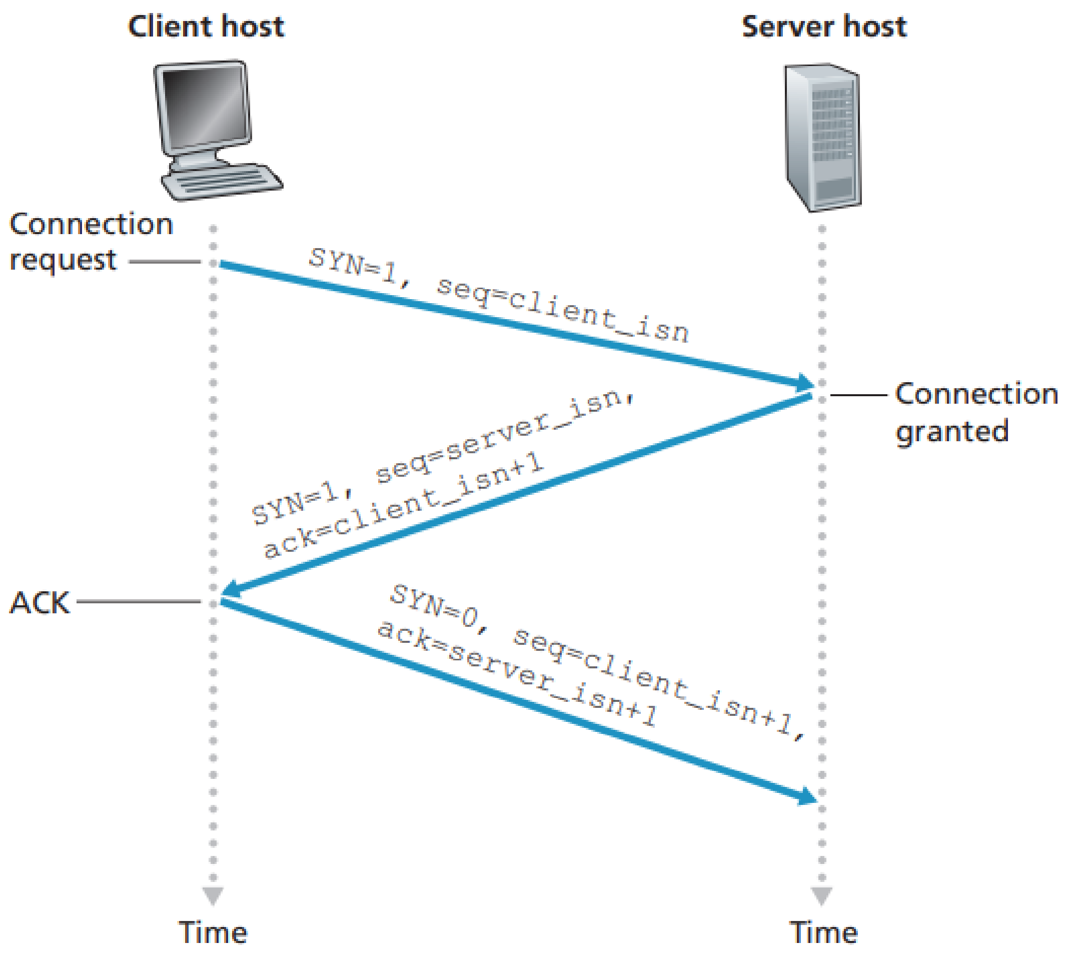


Figure 3.39 ♦ TCP three-way handshake: segment exchange

During the life of a TCP connection, the TCP protocol running in each host makes transitions through various TCP states.

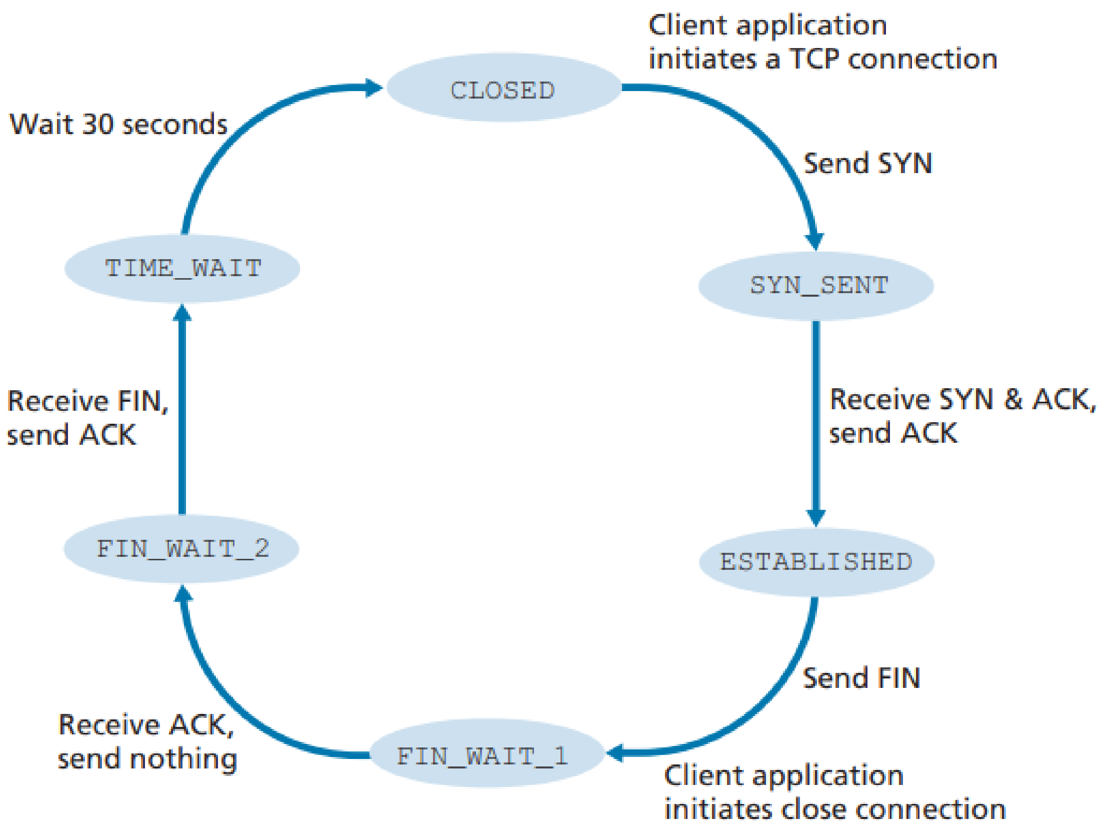

Client application initiates a TCP connection
CLOSED
Wait 30 seconds
Send SYN
TIME_WAIT
SYN_SENT
Receive FIN, send ACK
Receive SYN & ACK, send ACK
FIN_WAIT_2
ESTABLISHED
Send FIN
Receive ACK, send nothing
FIN_WAIT_1
Client application initiates close connection

Figure 3.41 ♦ A typical sequence of TCP states visited by a client TCP

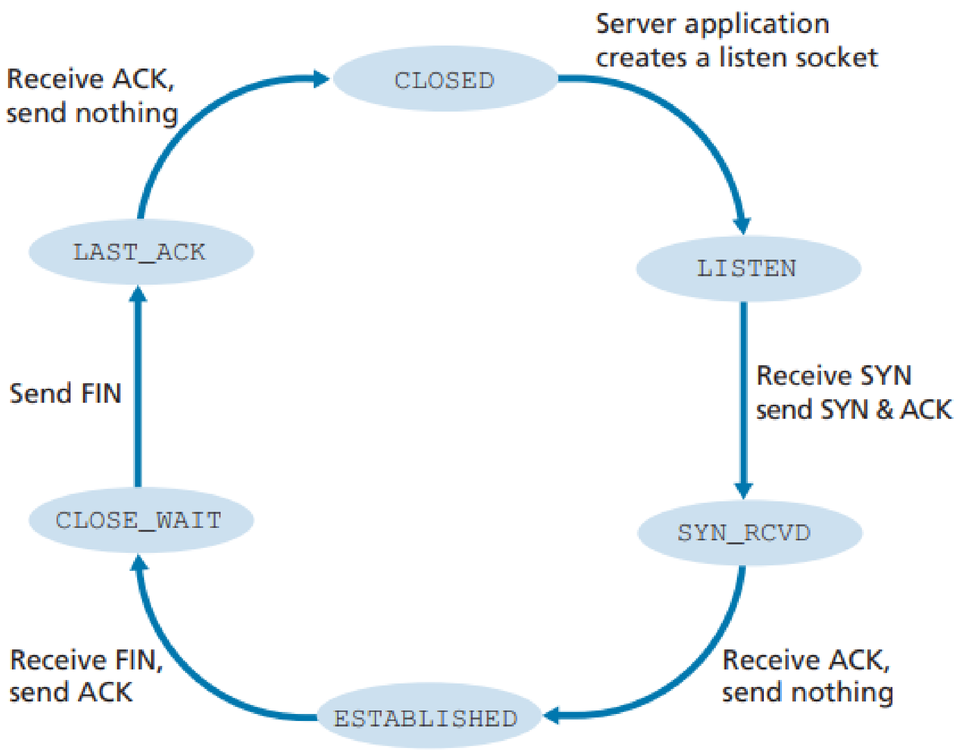

Server application creates a listen socket
CLOSED
Receive ACK, send nothing
LAST_ACK
LISTEN
Receive SYN send SYN & ACK
Send FIN
CLOSE_WAIT
SYN_RCVD
Receive FIN, send ACK
ESTABLISHED
Receive ACK, send nothing

Figure 3.42 ♦ A typical sequence of TCP states visited by a server-side TCP