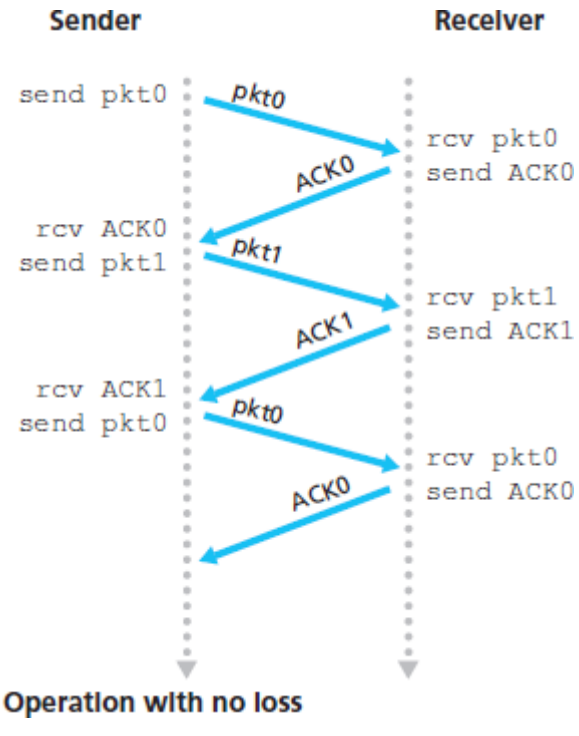
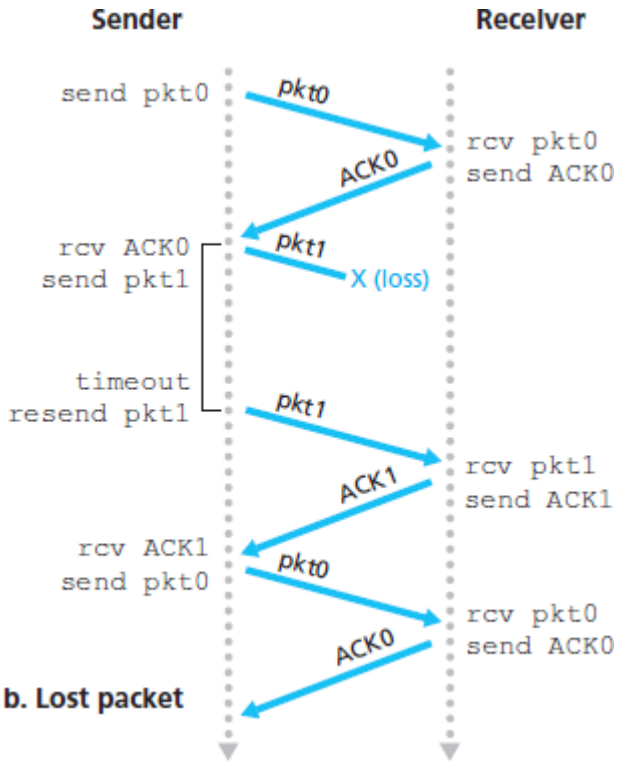
3.4.2 Pipelined Reliable Data Transfer Protocols

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.

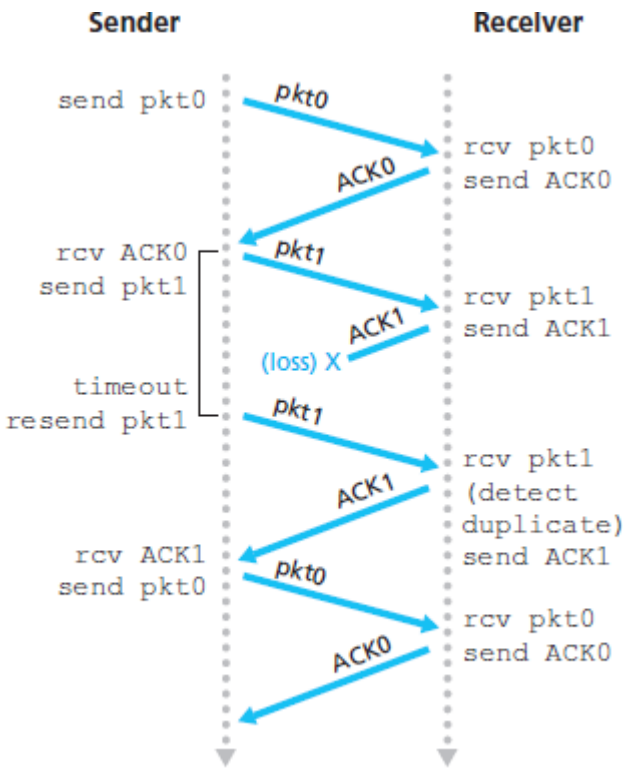


a. Operation with no loss

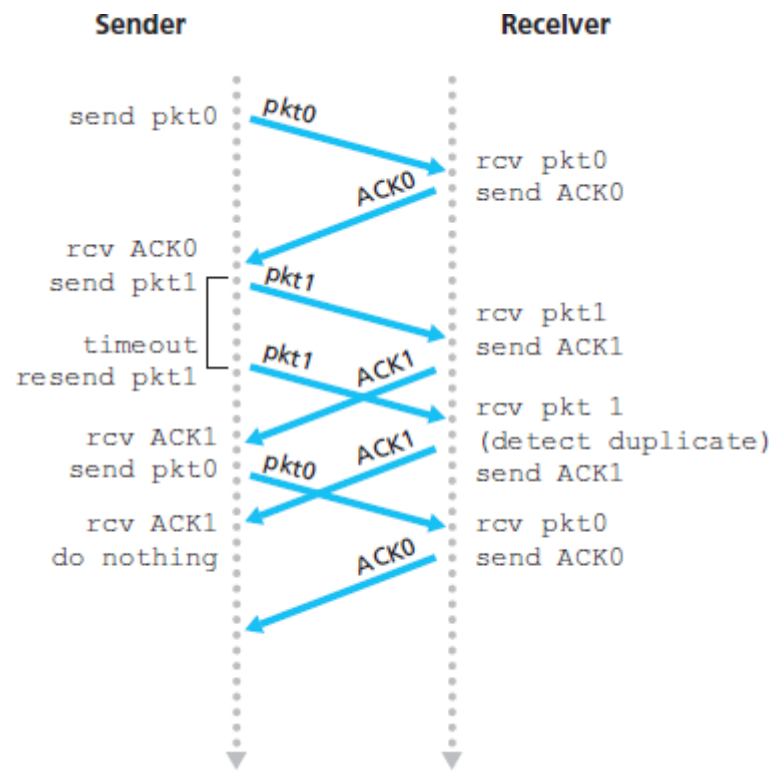
Figure 3.16 Operation of *rdt3.0*, the alternating-bit protocol



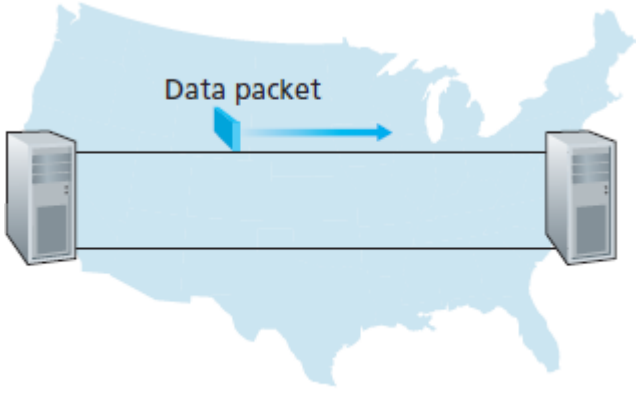
b. Lost packet



c. Lost ACK

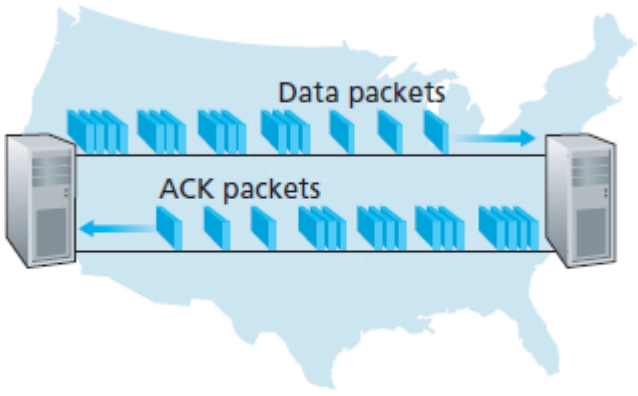


d. Premature timeout



a. A stop-and-wait protocol in operation

Figure 3.17 Stop-and-wait versus pipelined protocol



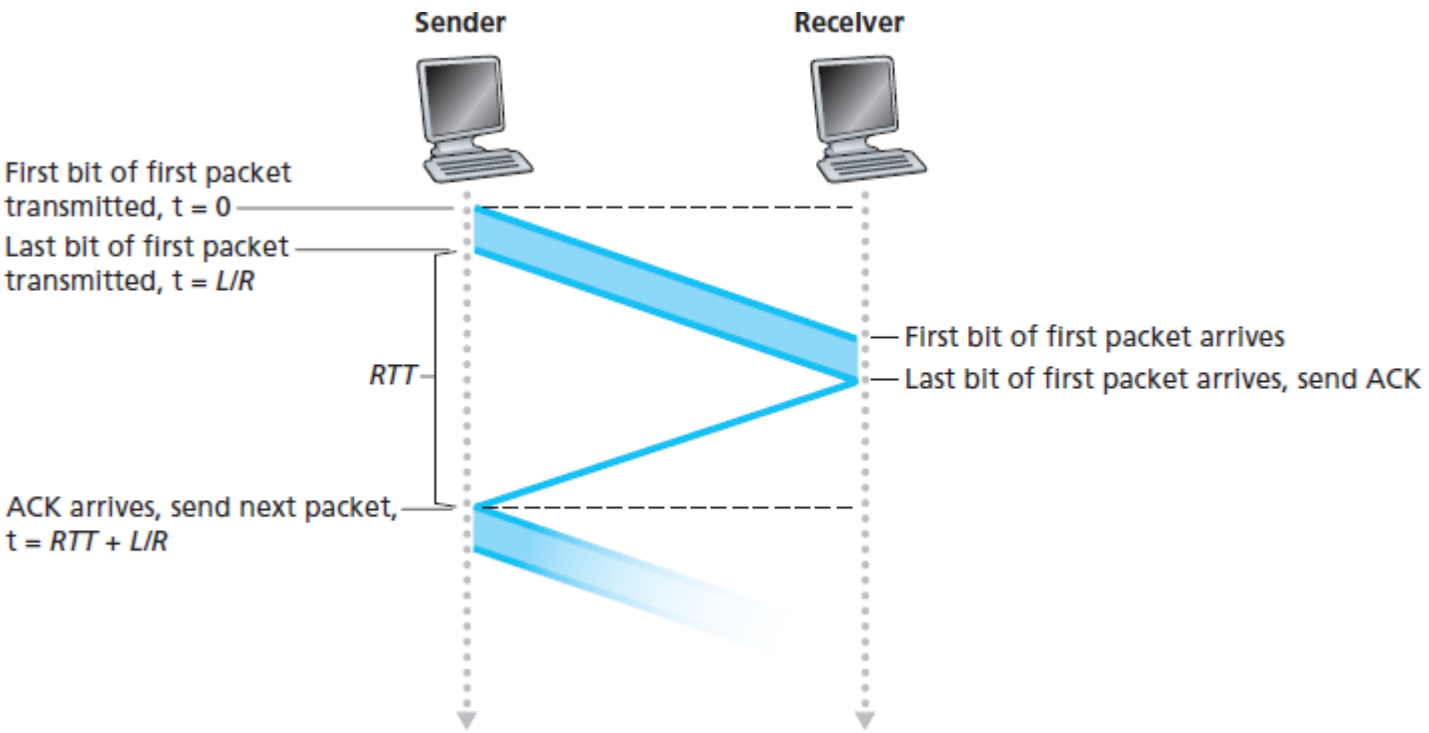
b. A pipelined protocol in operation

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 3.17**. 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 (10^9 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=LR=8000 bits/packet109 bits/sec=8 microseconds

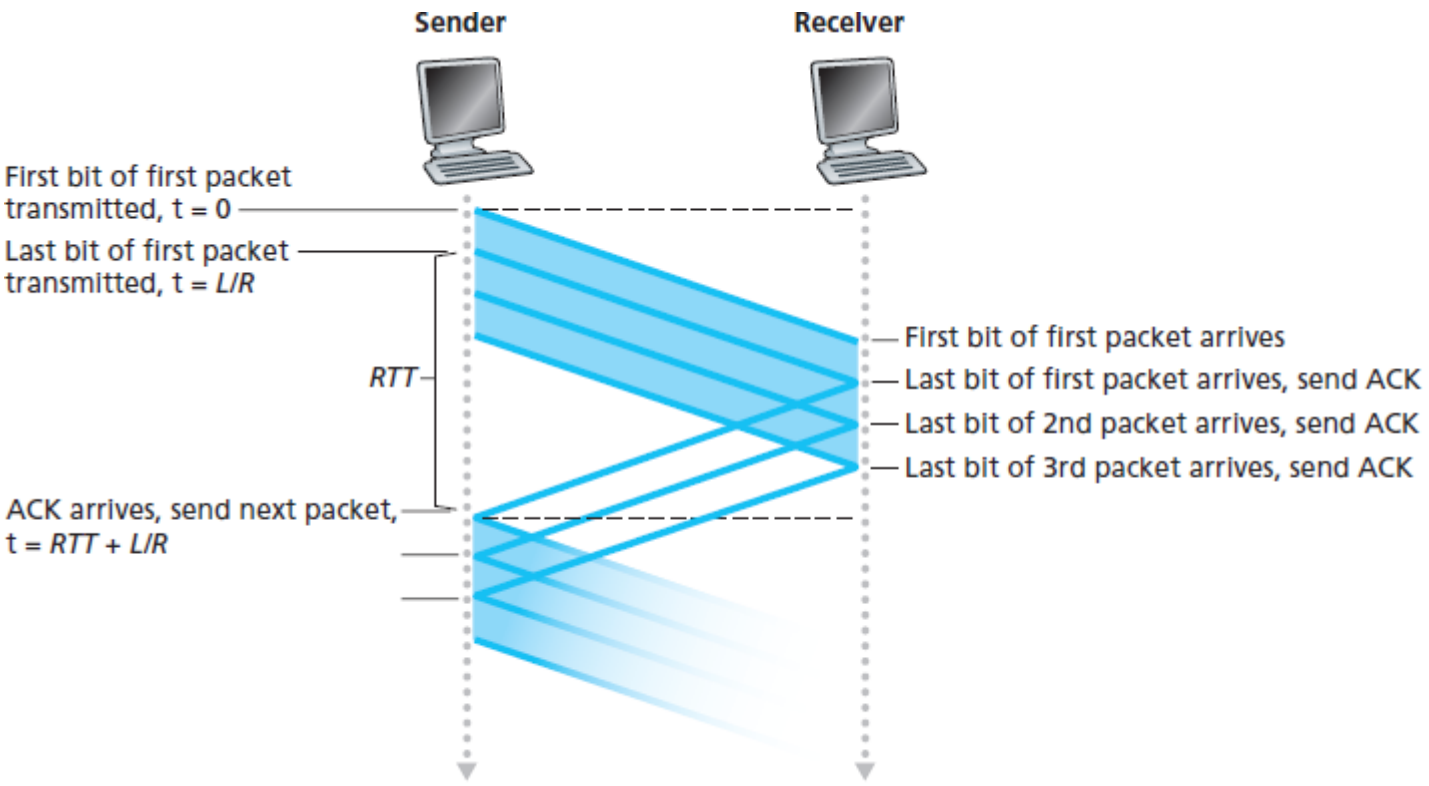
Figure 3.18(a) shows that with our stop-and-wait protocol, if the sender begins sending the packet at t=0, then at t=L/R=8 microseconds, the last bit enters the channel at the sender side. The packet then makes its 15-msec cross-country journey, with the last bit of the packet emerging at the receiver at t=RTT/2+L/R= 15.008 msec. 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=30.008 msec. At this point, the sender can now transmit the next message. Thus, in 30.008 msec, the sender was sending for only 0.008 msec. If we define the **utilization** of the sender (or the channel) as the fraction of time the sender is actually busy sending bits into the channel, the analysis in **Figure 3.18(a)** shows that the stop-and-wait protocol has a rather dismal sender utilization, U_{sender}, of

Usender=L/RRTT+L/R =.00830.008=0.00027



a. Stop-and-wait operation

Figure 3.18 Stop-and-wait and pipelined sending



b. Pipelined operation

That is, the sender was busy only 2.7 hundredths of one percent of the time! Viewed another way, the sender was able to send only 1,000 bytes in 30.008 milliseconds, an effective throughput of only 267 kbps—even though a 1 Gbps link was available! Imagine the unhappy network manager who just paid a fortune for a gigabit capacity link but manages to get a throughput of only 267 kilobits per second! This is a graphic example of how network protocols can limit the capabilities provided by the underlying network hardware. Also, we have neglected lower-layer protocol-processing times at the sender and receiver, as well as the processing and queuing delays that would occur at any intermediate routers

between the sender and receiver. Including these effects would serve only to further increase the delay and further accentuate the poor performance.

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**.

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. We describe the GBN protocol in some detail in this section. But before reading on, you are encouraged to play with the GBN applet (an awesome applet!) at the companion Web site.

Figure 3.19 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

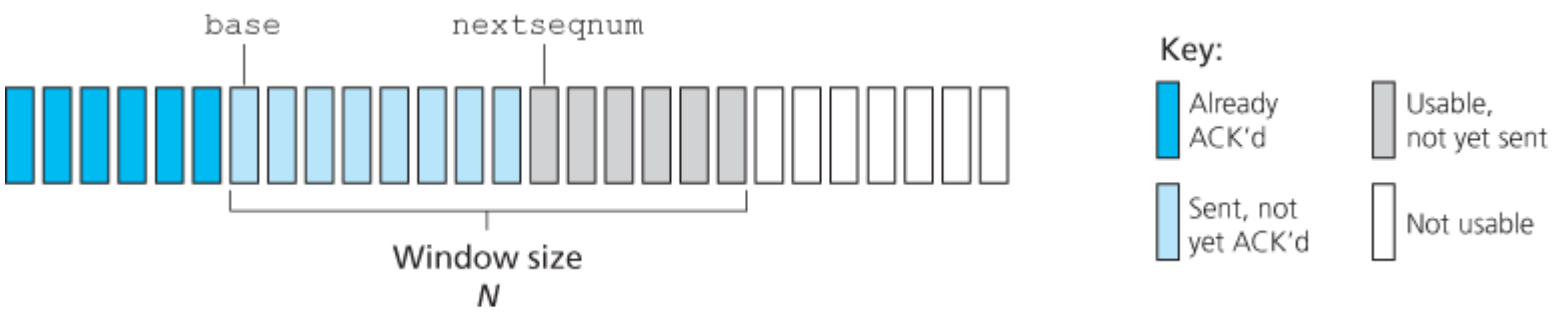


Figure 3.19 Sender's view of sequence numbers in Go-Back-N

packet and `nextseqnum` to be the smallest unused sequence number (that is, the sequence number of the next packet to be sent), 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 3.19**, 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**. You might be wondering why we would even limit the number of outstanding, unacknowledged packets to a value of *N* in the first place. Why not allow an unlimited number of such packets? We'll see in **Section 3.5** that flow control is one reason to impose a limit on the sender. We'll examine another reason to do so in **Section 3.7**, when we study TCP congestion control.

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. (That is, the sequence number space can be thought of as a ring of size 2^k, where sequence number $2k-1$ is immediately followed by sequence number 0.) Recall that `rdt3.0` had a 1-bit sequence number and a range of sequence numbers of [0,1]. Several of the problems at the end of this chapter explore the consequences of a finite range of sequence numbers. We will see in **Section 3.5** that TCP has a 32-bit sequence number field, where TCP sequence numbers count bytes in the byte stream rather than packets.

Figures 3.20 and **3.21** give an extended FSM description of the sender and receiver sides of an ACK-based, NAK-free, GBN protocol. We refer to this FSM

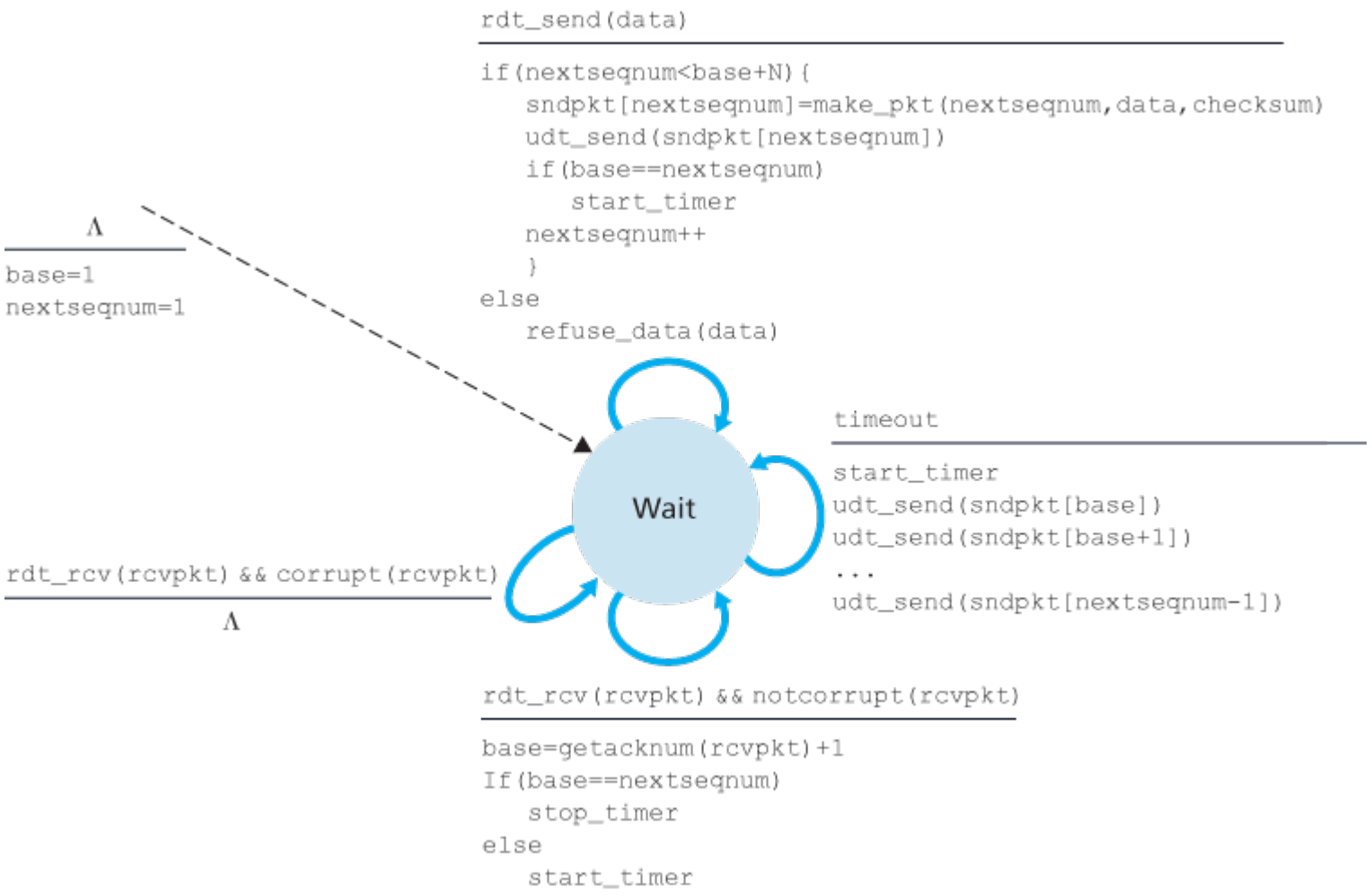


Figure 3.20 Extended FSM description of the GBN sender

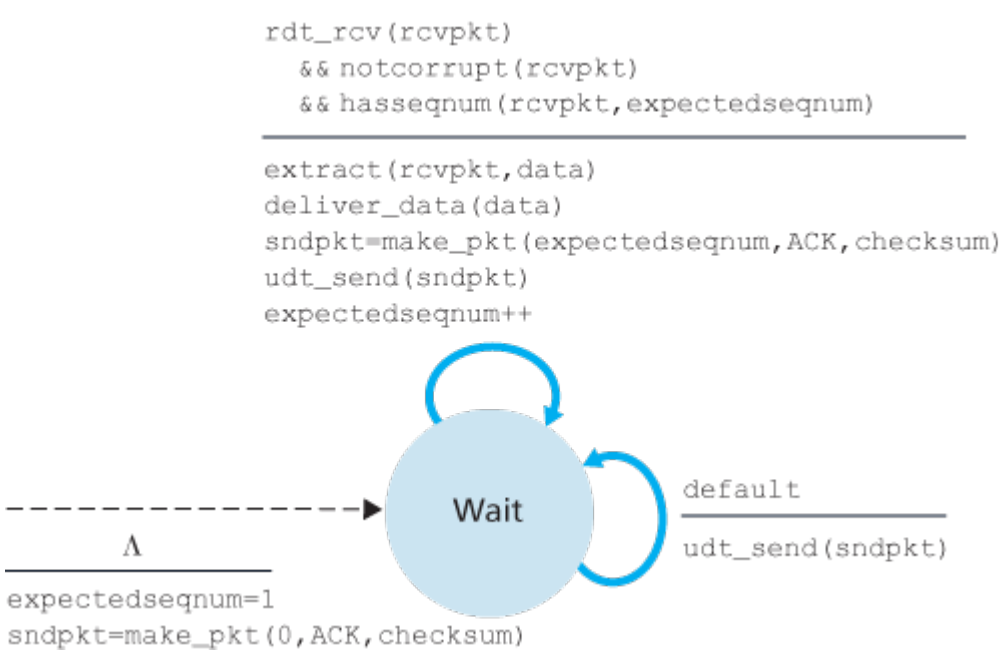


Figure 3.21 Extended FSM description of the GBN receiver

description as an *extended FSM* because we have added variables (similar to programming-language variables) for `base` and `nextseqnum`, and added operations on these variables and conditional actions involving these variables. Note that the extended FSM specification is now beginning to look somewhat like a programming-language specification. **[Bochman 1984]** provides an excellent survey of

additional extensions to FSM techniques as well as other programming-language-based techniques for specifying protocols.

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 3.20** 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.

The receiver's actions in GBN are also simple. If a packet with sequence number *n* is received correctly and is in order (that is, the data last delivered to the upper layer came from a packet with sequence number $n-1$), 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. Although it may seem silly and wasteful to discard a correctly received (but out-of-order) packet, there is some justification for doing so. 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`, shown in the receiver FSM in **Figure 3.21**. 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 shows the operation of the GBN protocol for the case of a window size of four packets. Because of this window size limitation, the sender sends packets 0 through 3 but then must wait for one or more of these packets to be acknowledged before proceeding. As each successive ACK (for example, `ACK0` and `ACK1`) is received, the window slides forward and the sender can transmit one new packet (pkt4 and pkt5, respectively). On the receiver side, packet 2 is lost and thus packets 3, 4, and 5 are found to be out of order and are discarded.

Before closing our discussion of GBN, it is worth noting that an implementation of this protocol in a protocol stack would likely have a structure similar to that of the extended FSM in **Figure 3.20**. The implementation would also likely be in the form of various procedures that implement the actions to be taken in response to the various events that can occur. In such **event-based programming**, the various procedures are called (invoked) either by other procedures in the protocol stack, or as the result of an interrupt. In the sender, these events would be (1) a call from the upper-layer entity to invoke `rdt_send()`, (2) a timer interrupt, and (3) a call from the lower layer to invoke `rdt_rcv()` when a packet arrives. The programming exercises at the end of this chapter will give you a chance to actually implement these routines in a simulated, but realistic, network setting.

We note here that the GBN protocol incorporates almost all of the techniques that we will encounter when we study the reliable data transfer components of TCP in **Section 3.5**. These techniques include the use of sequence numbers, cumulative acknowledgments, checksums, and a timeout/retransmit operation.

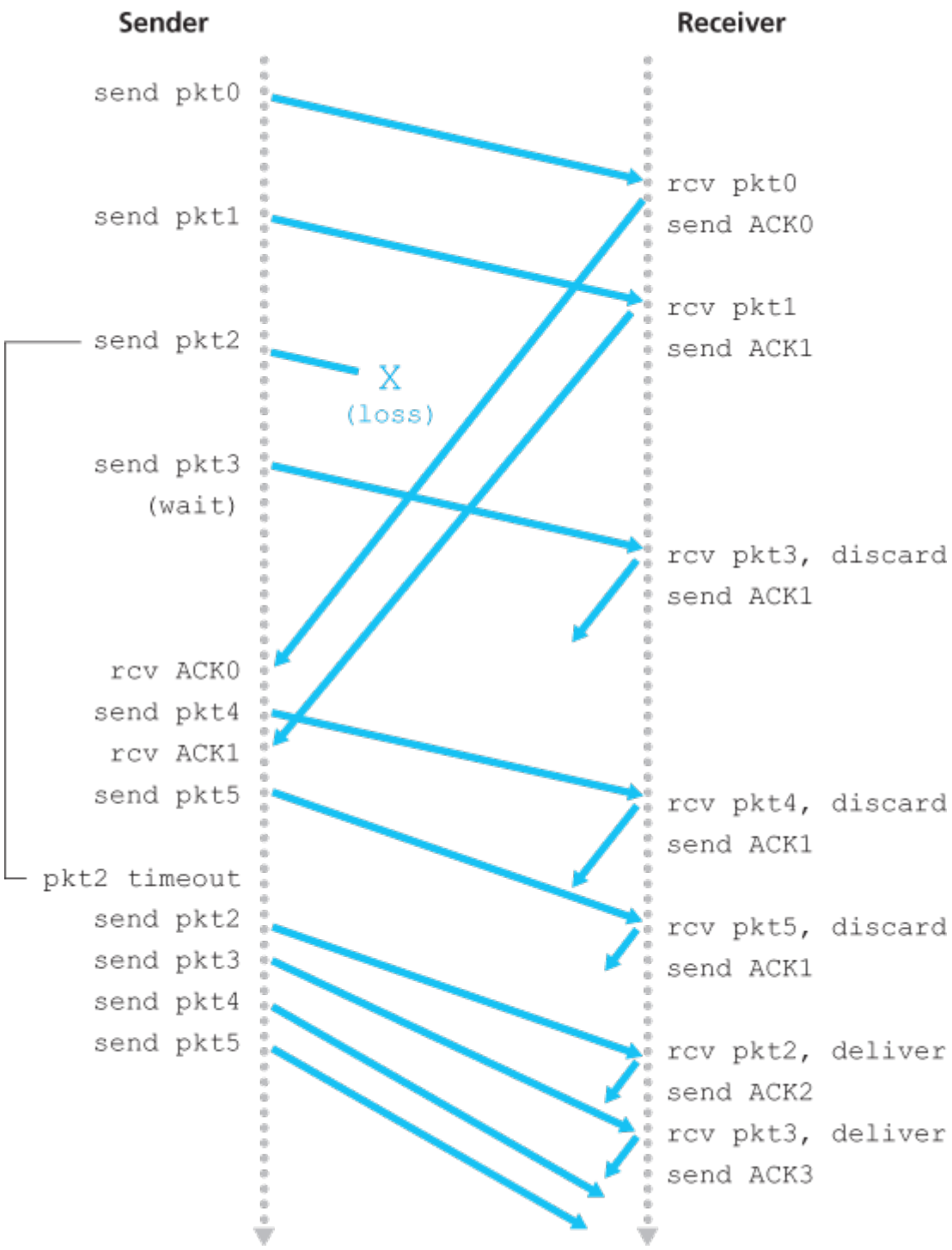


Figure 3.22 Go-Back-N in operation

3.4.4 Selective Repeat (SR)

The GBN protocol allows the sender to potentially “fill the pipeline” in **Figure 3.17** with packets, thus avoiding the channel utilization problems we noted with stop-and-wait protocols. 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 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 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. **Figure 3.23** shows the SR sender's view of the sequence number space. **Figure 3.24** 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 3.25** itemizes the various actions taken by the SR receiver. **Figure 3.26** shows an example of SR operation in the presence of lost packets. Note that in **Figure 3.26**, 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.

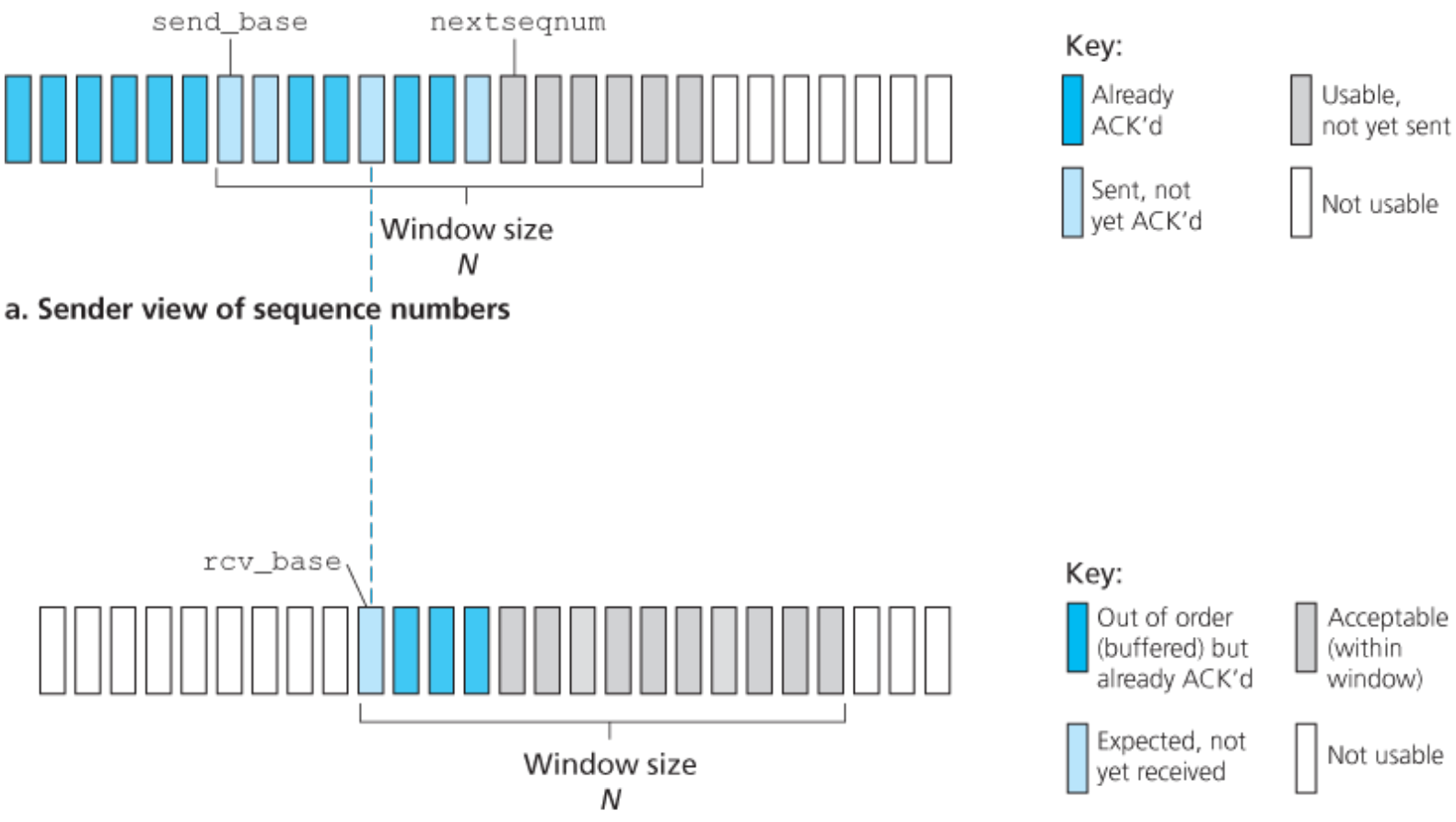


Figure 3.23 Selective-repeat (SR) sender and receiver views of sequence-number space

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. A single hardware timer can be used to mimic the operation of multiple logical timers [Varghese 1997].
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.

Figure 3.24 SR sender 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 buffered. 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 buffered 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. As an example, consider Figure 3.26. When a packet with a sequence number of `rcv_base=2` is received, it and packets 3, 4, and 5 can be 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.

Figure 3.25 SR receiver events and actions

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. You should convince yourself that this reacknowledgment is indeed needed. Given the sender and receiver sequence number spaces in **Figure 3.23**, for example, if there is no ACK for packet ***`send_base`*** propagating from the

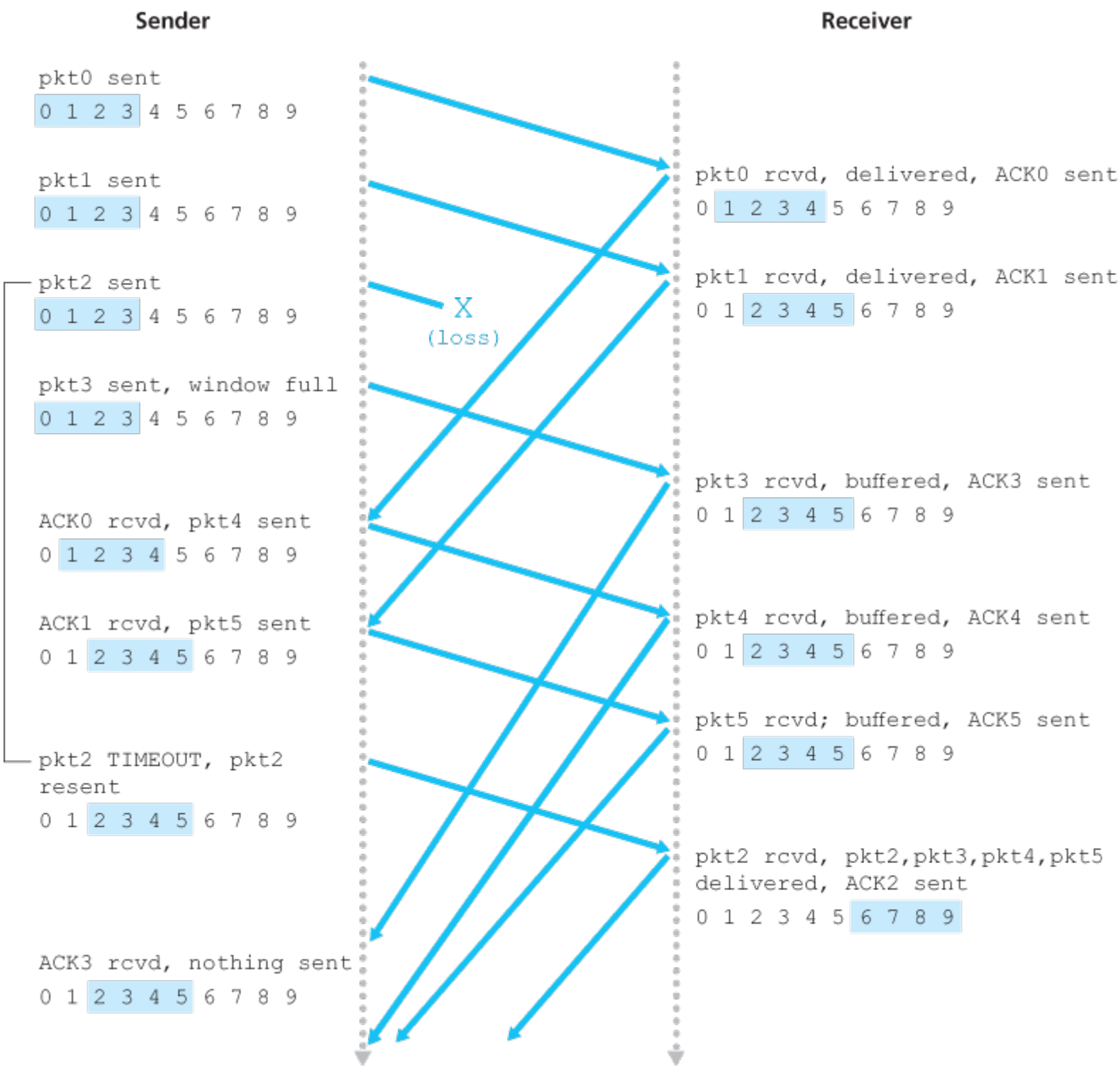


Figure 3.26 SR operation

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 that packet. If the receiver were not to acknowledge this packet, the sender's window would never move forward! This example illustrates an important aspect of SR protocols (and many other protocols as well). The sender and receiver will not always have an identical view of what has been received correctly and what has not. For SR protocols, this means that the sender and receiver windows will not always coincide.

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. Consider what could happen, for example, with a finite range of four packet sequence numbers, 0, 1, 2, 3, and a window size of three.

Suppose packets 0 through 2 are transmitted and correctly received and acknowledged at the receiver. At this point, the receiver's window is over the fourth, fifth, and sixth packets, which have sequence numbers 3, 0, and 1, respectively. Now consider two scenarios. In the first scenario, shown in **Figure 3.27(a)**, the ACKs for the first three packets are lost and the sender retransmits these packets. The receiver thus next receives a packet with sequence number 0—a copy of the first packet sent.

In the second scenario, shown in **Figure 3.27(b)**, the ACKs for the first three packets are all delivered correctly. The sender thus moves its window forward and sends the fourth, fifth, and sixth packets, with sequence numbers 3, 0, and 1, respectively. The packet with sequence number 3 is lost, but the packet with sequence number 0 arrives—a packet containing *new* data.

Now consider the receiver's viewpoint in **Figure 3.27**, which has a figurative curtain between the sender and the receiver, since the receiver cannot "see" the actions taken by the sender. All the receiver observes is the sequence of messages it receives from the channel and sends into the channel. As far as it is concerned, the two scenarios in **Figure 3.27** are *identical.* There is no way of distinguishing the retransmission of the first packet from an original transmission of the fifth packet. Clearly, a window size that is 1 less than the size of the sequence number space won't work. But how small must the window size be? A problem at the end of the chapter asks you to show that the window size must be less than or equal to half the size of the sequence number space for SR protocols.

At the companion Web site, you will find an applet that animates the operation of the SR protocol. Try performing the same experiments that you did with the GBN applet. Do the results agree with what you expect?

This completes our discussion of reliable data transfer protocols. We've covered a *lot* of ground and introduced numerous mechanisms that together provide for reliable data transfer. **Table 3.1** summarizes these mechanisms. Now that we have seen all of these mechanisms in operation and can see the "big picture," we encourage you to review this section again to see how these mechanisms were incrementally added to cover increasingly complex (and realistic) models of the channel connecting the sender and receiver, or to improve the performance of the protocols.

Let's conclude our discussion of reliable data transfer protocols by considering one remaining assumption in our underlying channel model. Recall that we have assumed that packets cannot be reordered within the channel between the sender and receiver. This is generally a reasonable assumption when the sender and receiver are connected by a single physical wire. However, when the "channel" connecting the two is a network, packet reordering can occur. One manifestation of packet reordering is that old copies of a packet with a sequence or acknowledgment

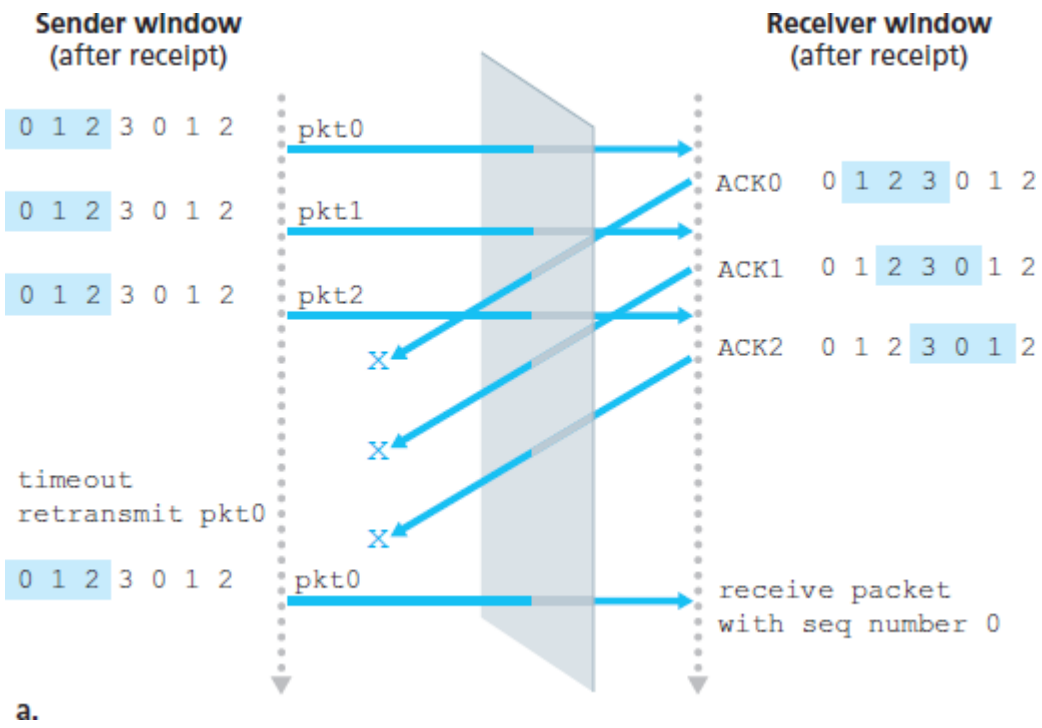


Figure 3.27 SR receiver dilemma with too-large windows: A new packet or a retransmission?

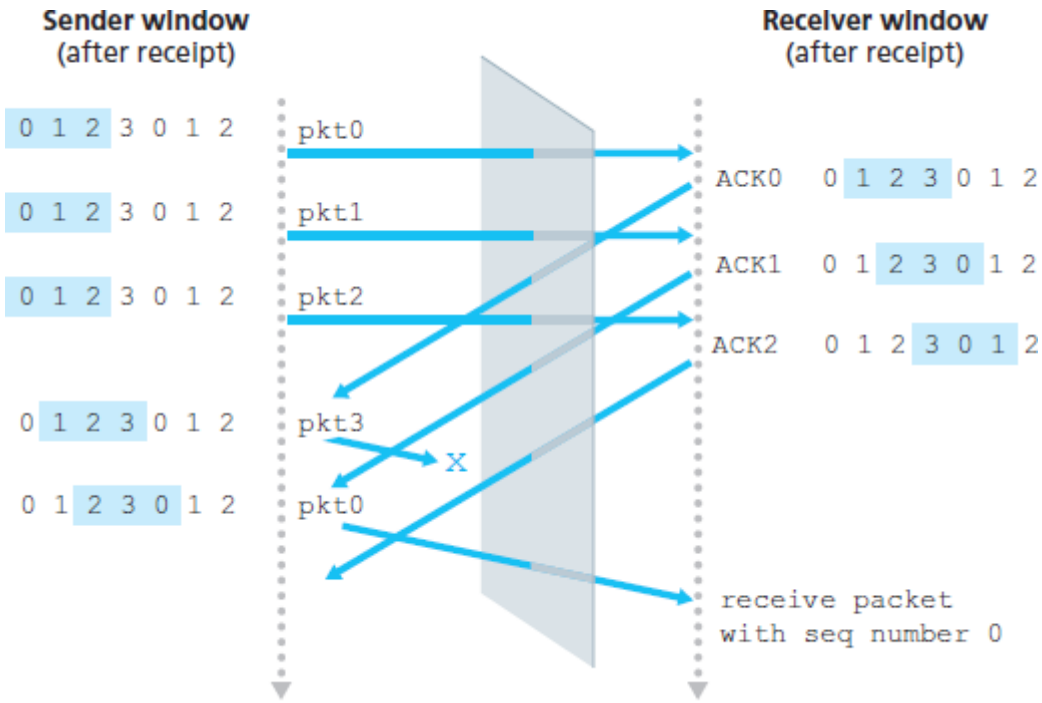


Table 3.1 Summary of reliable data transfer mechanisms and their use

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.

number of *x* can appear, even though neither the sender's nor the receiver's window contains *x*. With packet reordering, the channel can be thought of as essentially buffering packets and spontaneously emitting these packets at *any* point in the future. Because sequence numbers may be reused, some care must be taken to guard against such duplicate packets. The approach taken in practice is to ensure that a sequence number is not reused until the sender is "sure" that any previously sent packets with sequence number *x* are no longer in the network. This is done by assuming that a packet cannot "live" in the network for longer than some fixed maximum amount of time. A maximum packet lifetime of approximately three minutes is assumed in the TCP extensions for high-speed networks **[RFC 1323]**. **[Sunshine 1978]** describes a method for using sequence numbers such that reordering problems can be completely avoided.