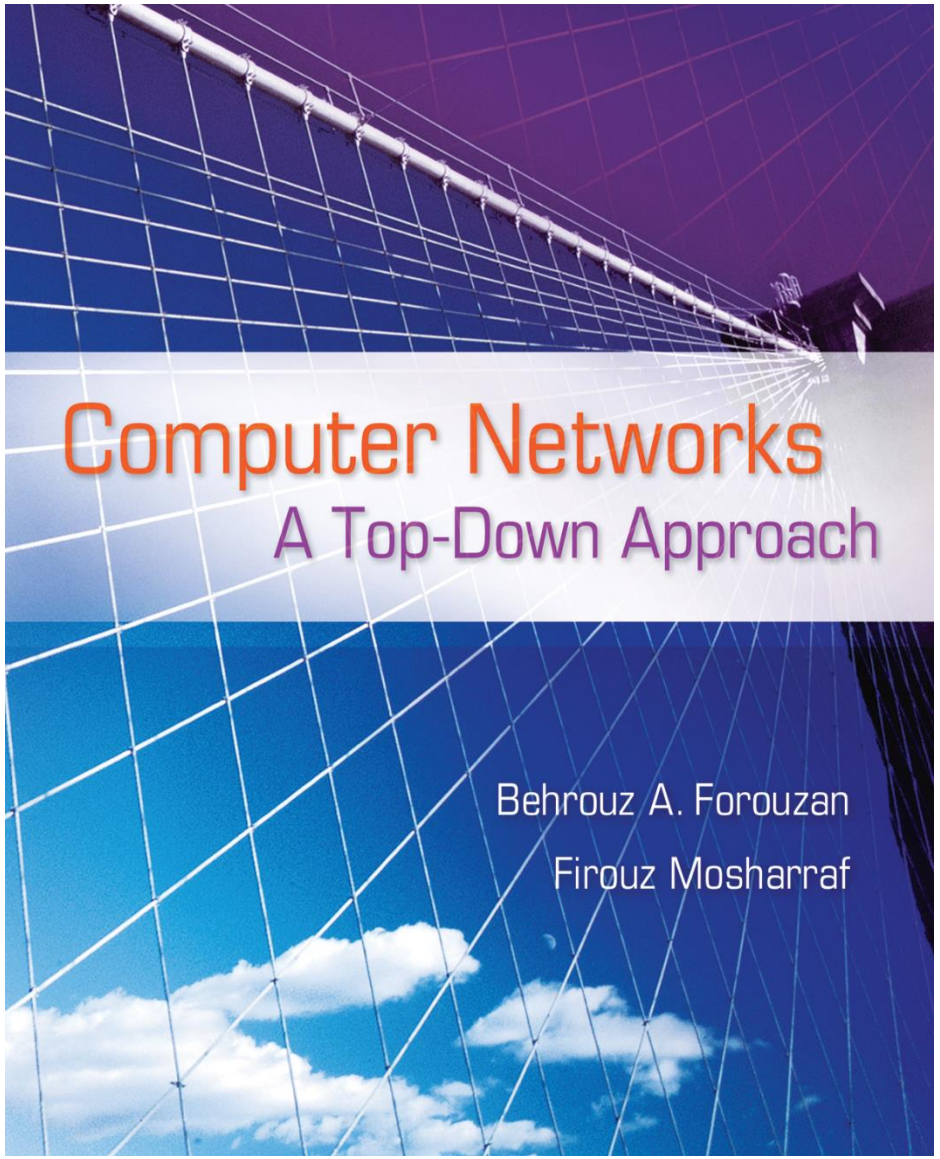
Chapter 3

Transport Layer



Chapter 3: Outline

3.1 INTRODUCTION

3.2 TRANSPORT-LAYER PROTOCOLS

3.3 USER DATAGRAM PROTOCOL

3.4 TRANSMISSION CONTROL PROTOCOL

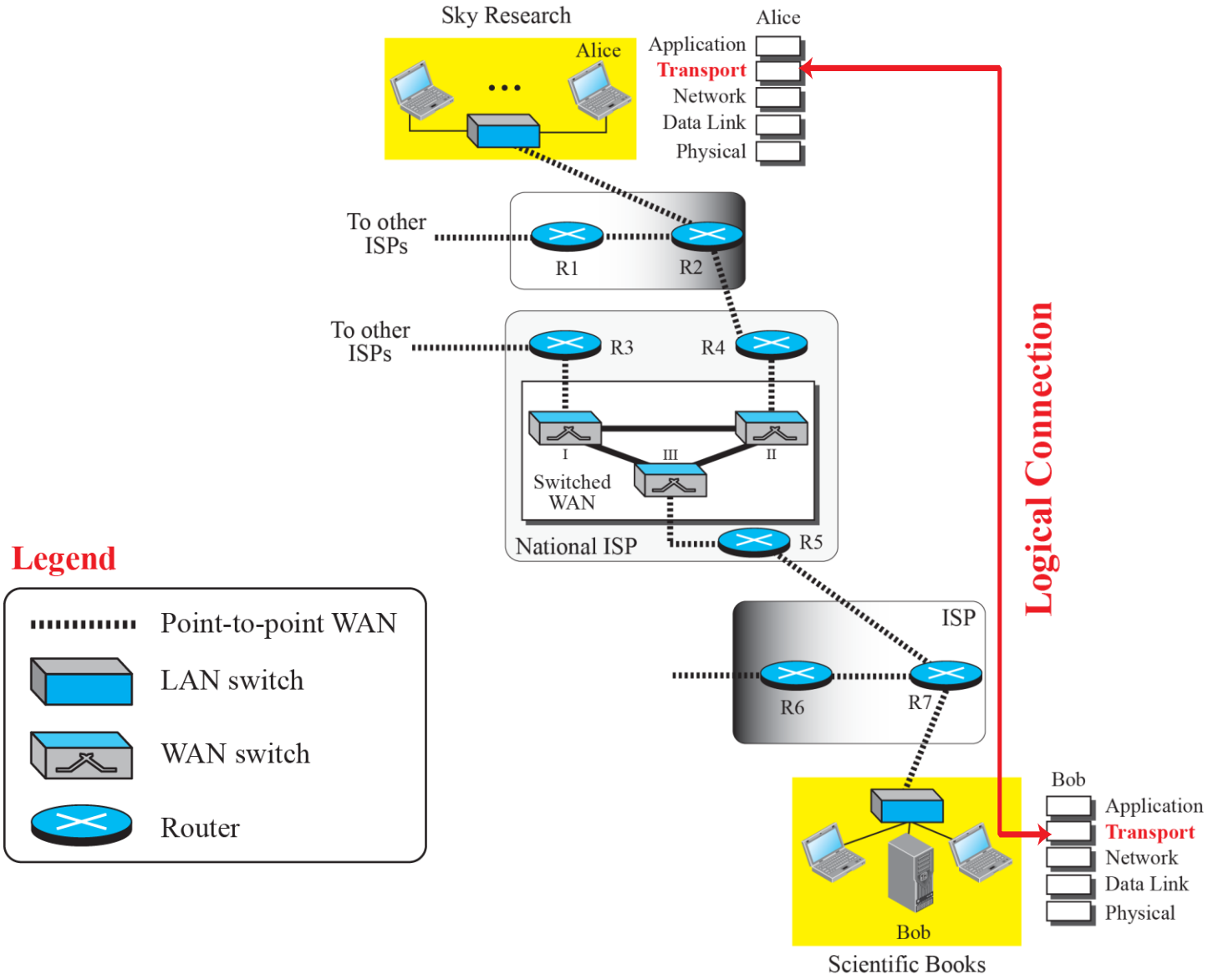
Chapter 3: Objective

- *We introduce general services we normally require from the transport layer, such as*
  - *process-to-process communication,*
  - *addressing,*
  - *Multiplexing*
  - *error, flow, and*
  - *congestion control.*
- *We discuss general transport-layer flow control protocols such as*
  - *Stop-and-Wait,*
  - *Go-Back-N, and*
  - *Selective-Repeat.*
- *We discuss UDP, which is the simpler protocols than TCP*
- *We discuss TCP services and features. We then show how TCP provides a connection-oriented service using a transition diagram.*

3-1 INTRODUCTION

- *The transport layer provides a process-to-process communication between two application layers.*
- *Communication is provided using a logical connection*
- *which means that the two application layers assume that there is an imaginary direct connection through which they can send and receive messages.*

Figure 3.1: Logical connection at the transport layer



3.1.1 Transport-Layer Services

- *As discussed the transport layer is located between the network layer and the application layer.*
- *The transport layer is responsible for providing services to the application layer;*
- *it receives services from the network layer.*
- *In this section, we discuss the services that can be provided by the transport layer;*
- *in the next section, we discuss several transport-layer protocols.*

3.1.1 (continued)

- *Process-to-Process Communication*
- *Addressing: Port Numbers*
- *ICANN Ranges*
  - *Well-known ports*
  - *Registered ports*
  - *Dynamic ports*
- *Encapsulation and Decapsulation*
- *Multiplexing and Demultiplexing*

3.1.1 (continued)

- **Flow Control**
  - Pushing or Pulling
  - Flow Control at Transport Layer
  - Buffers
- **Error Control**
  - Sequence Numbers
  - Acknowledgment
- **Combination of Flow and Error Control**
  - Sliding Window

3.1.1 (continued)

- *Congestion Control*
- *Connectionless and Connection-Oriented*
  - *Connectionless Service*
  - *Connection-Oriented Service*
  - *Finite State Machine*
- *Multiplexing and Demultiplexing*

Figure 3.2: Network layer versus transport layer

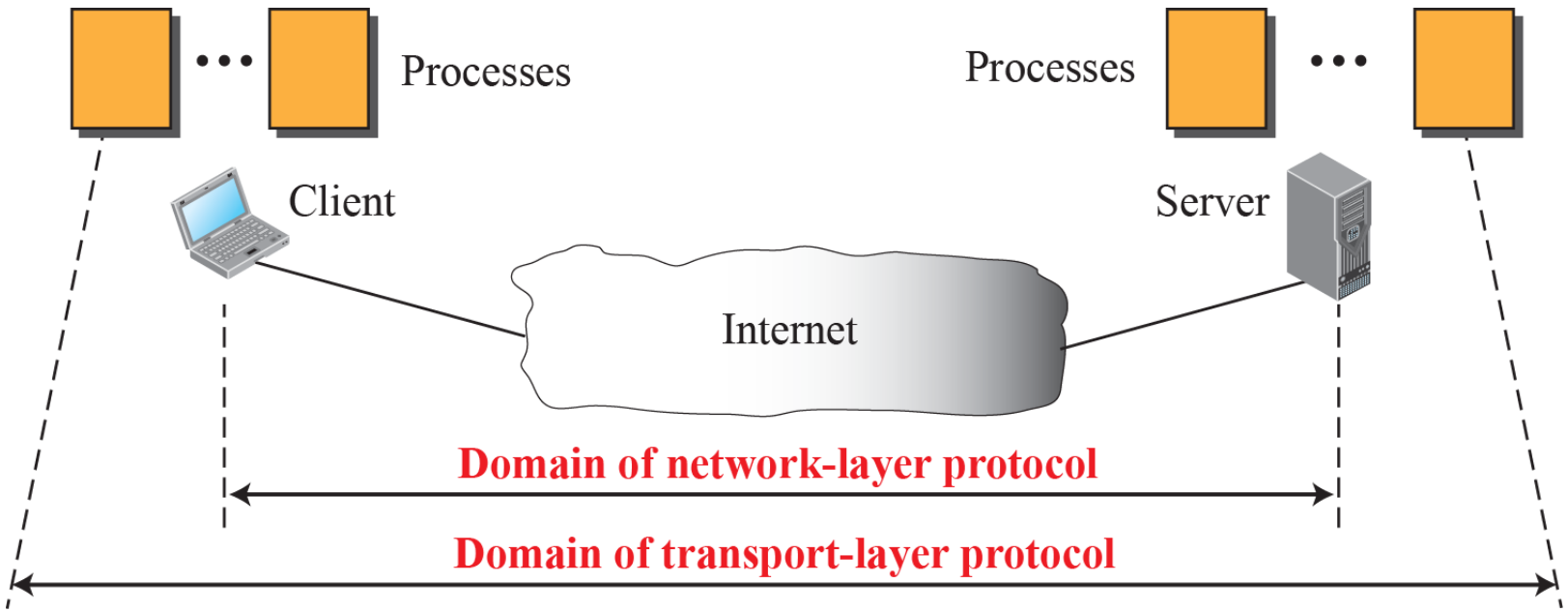


Figure 3.3: Port numbers

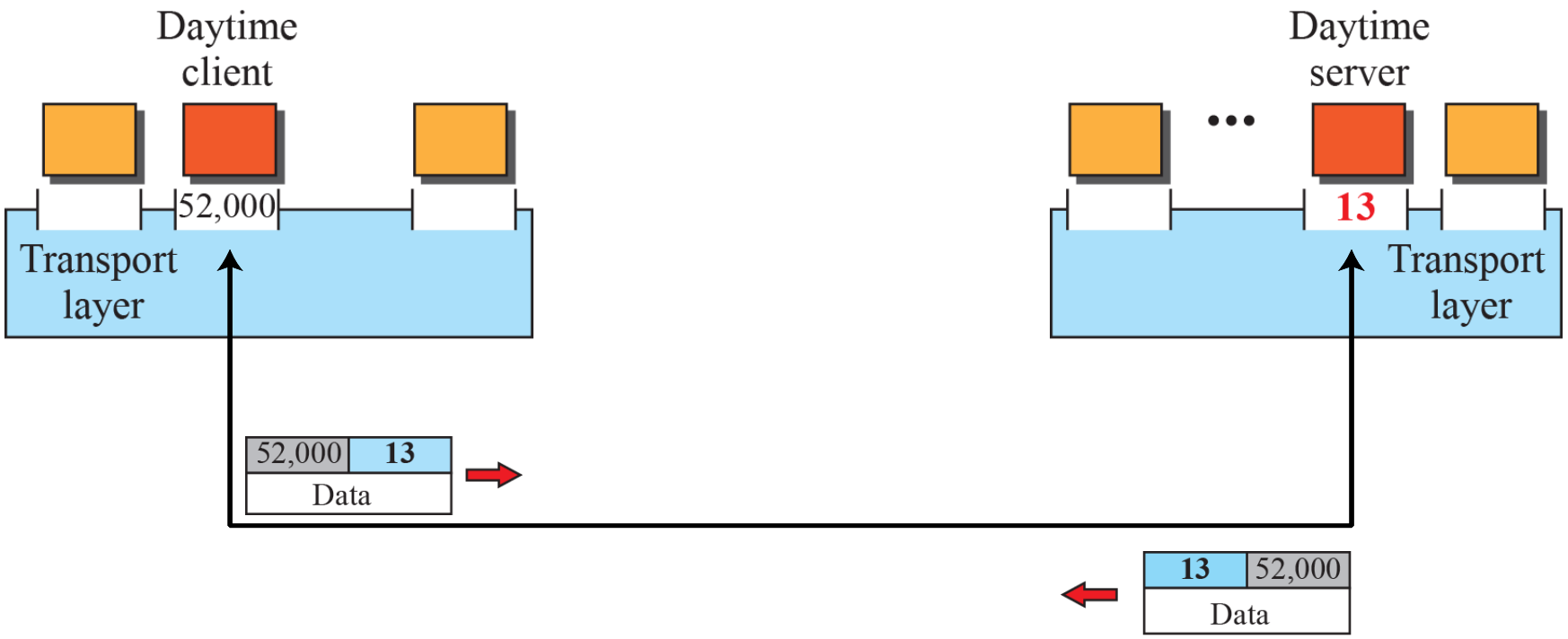


Figure 3.4: IP addresses versus port numbers

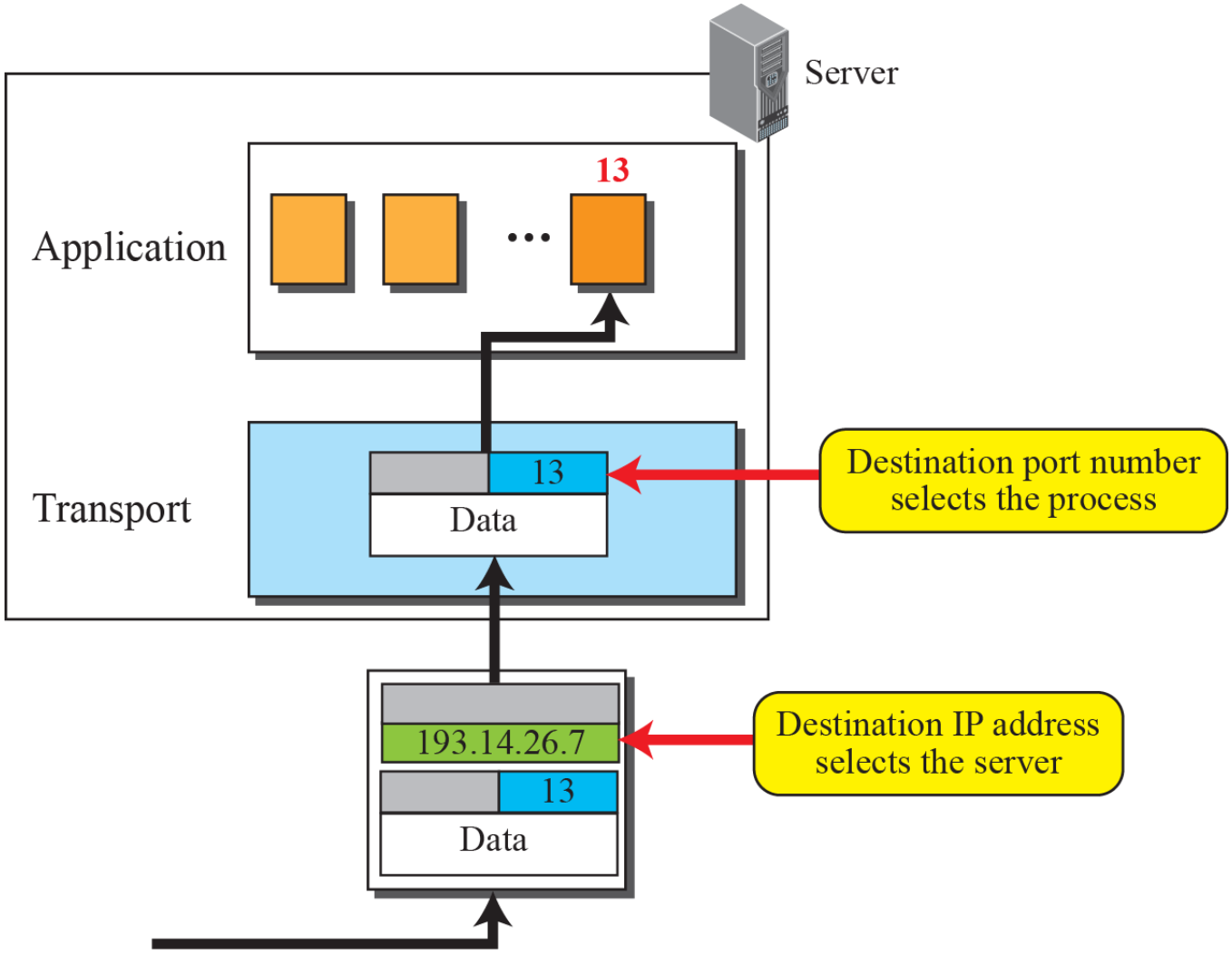
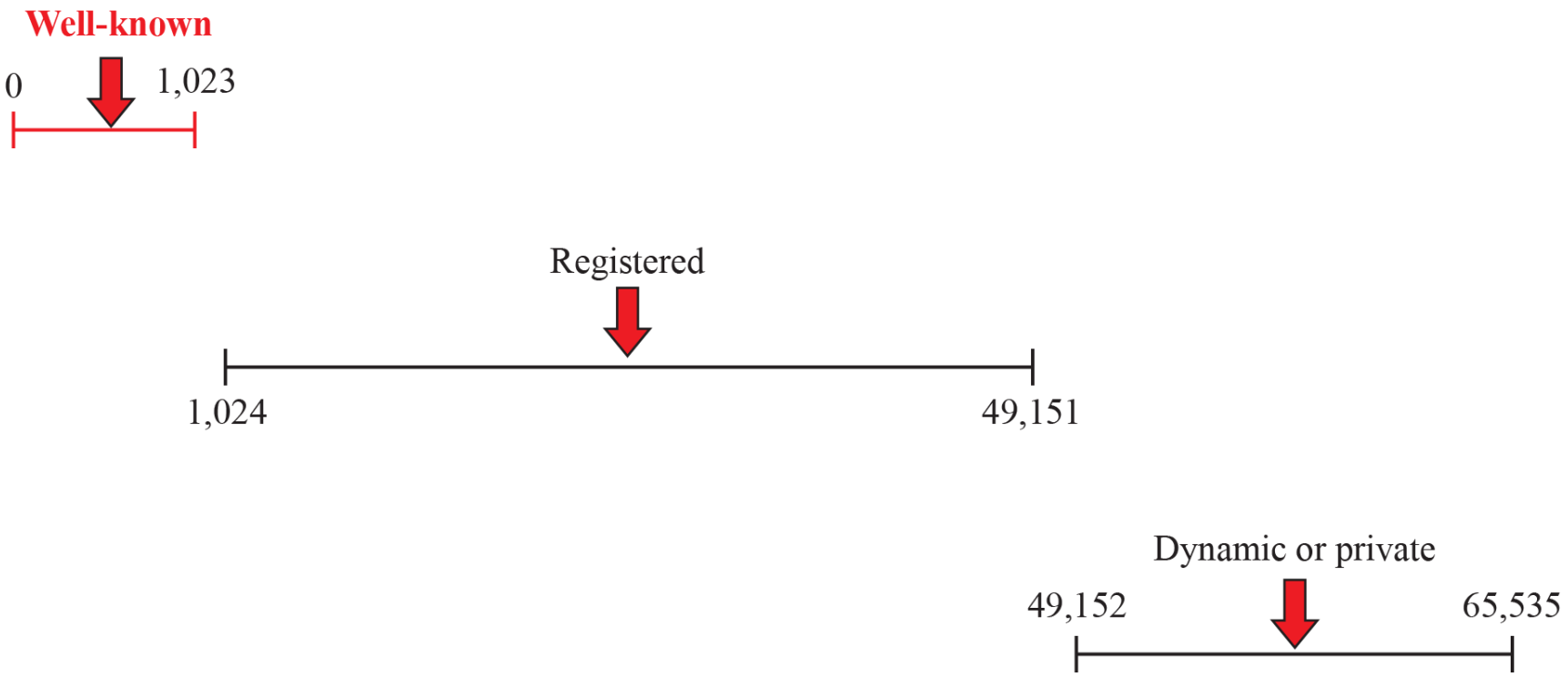


Figure 3.5: ICANN ranges



Example 3.1

In UNIX, the well-known ports are stored in a file called /etc/services. We can use the *grep* utility to extract the line corresponding to the desired application.

```
$grep     tftp/etc/services
tftp 69/tcp
tftp 69/udp
```

SNMP (see Chapter 9) uses two port numbers (161 and 162), each for a different purpose.

```
$grep     snmp/etc/services
snmp161/tcp#Simple Net Mgmt Proto
snmp161/udp#Simple Net Mgmt Proto
snmptrap162/udp#Traps for SNMP
```

Figure 3.6: Socket address

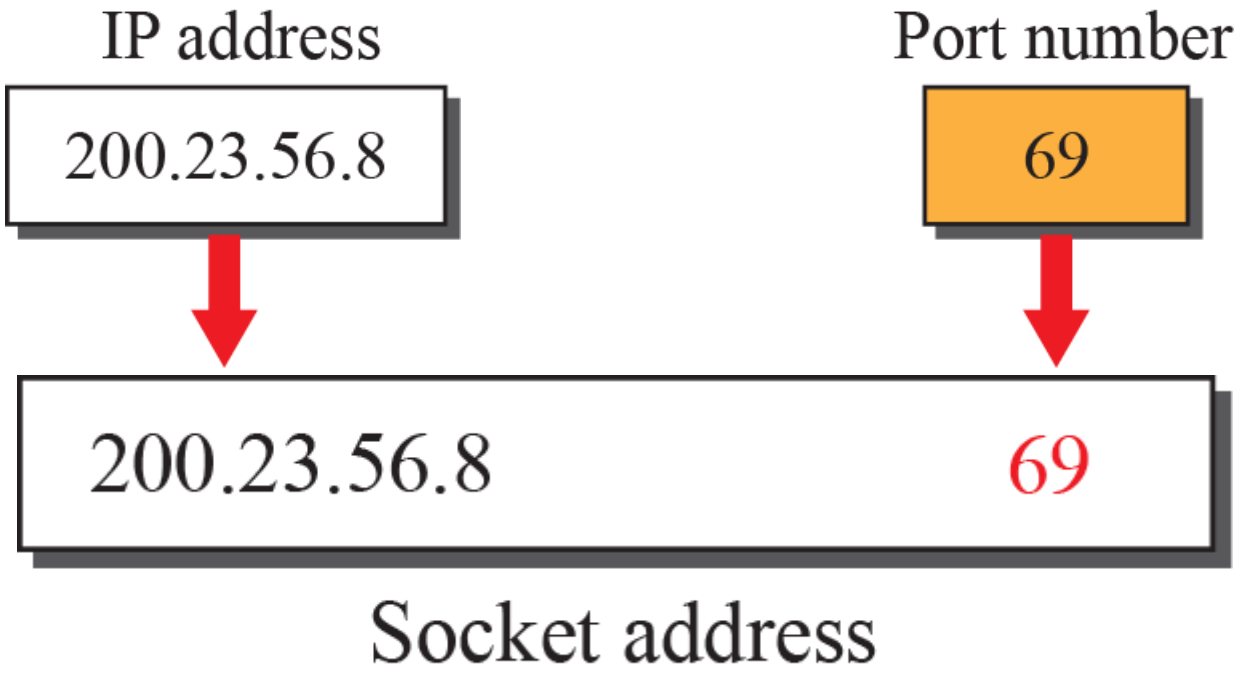


Figure 3.7: Encapsulation and decapsulation

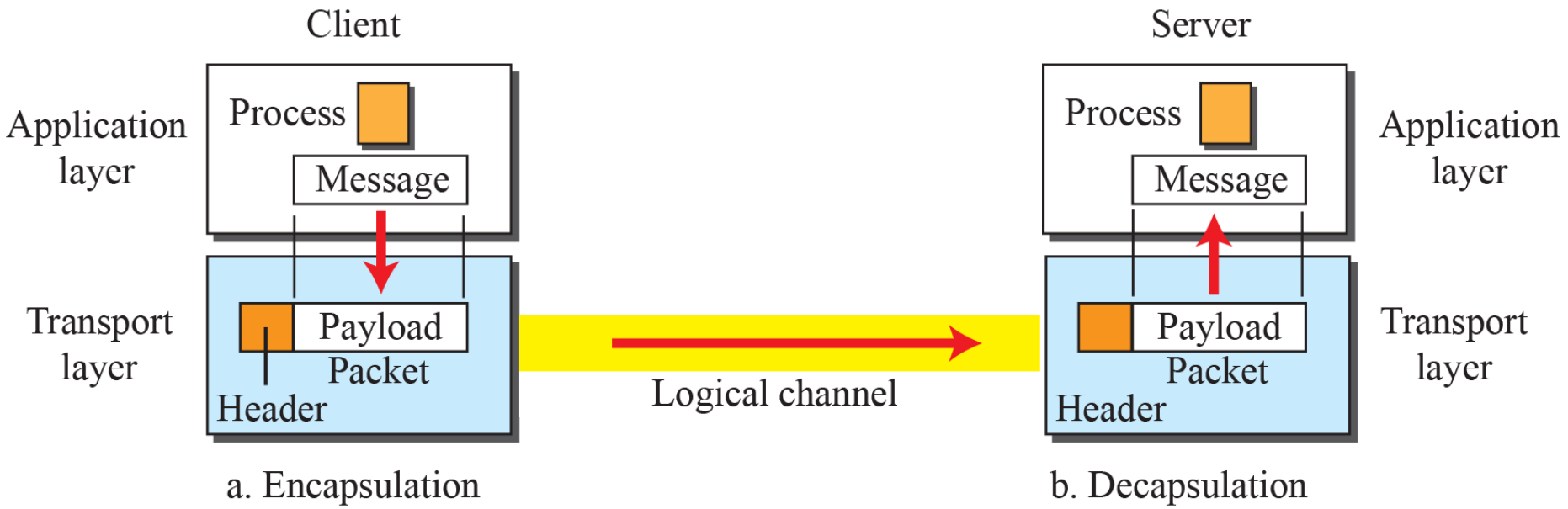


Figure 3.8: Multiplexing and de-multiplexing

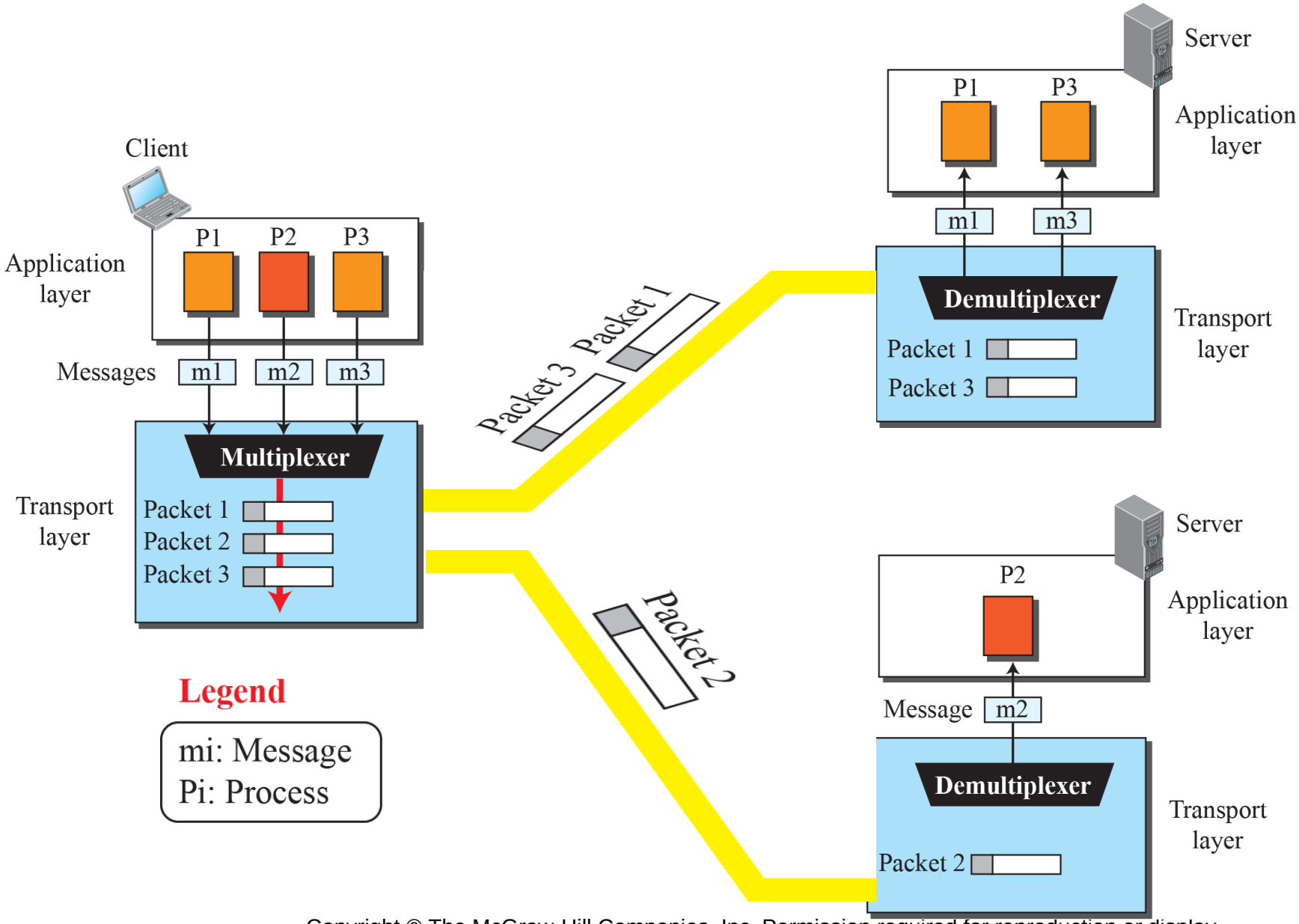
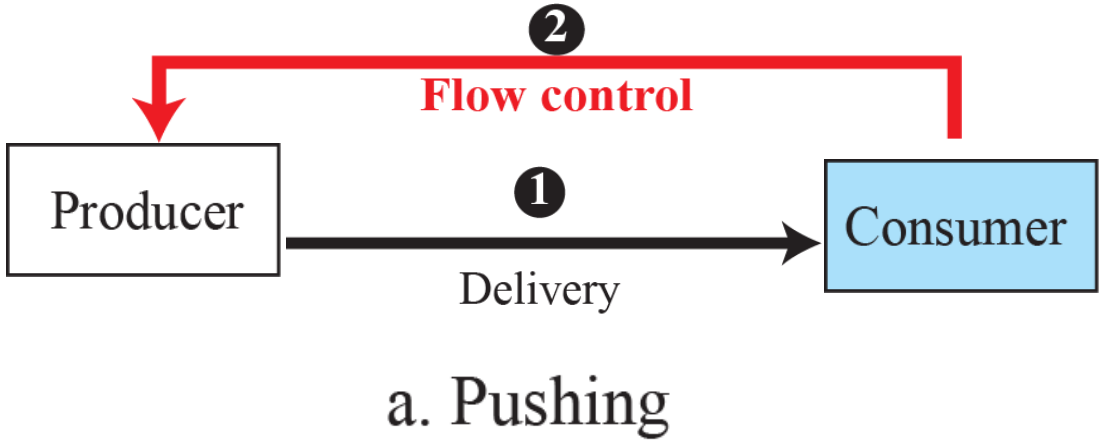
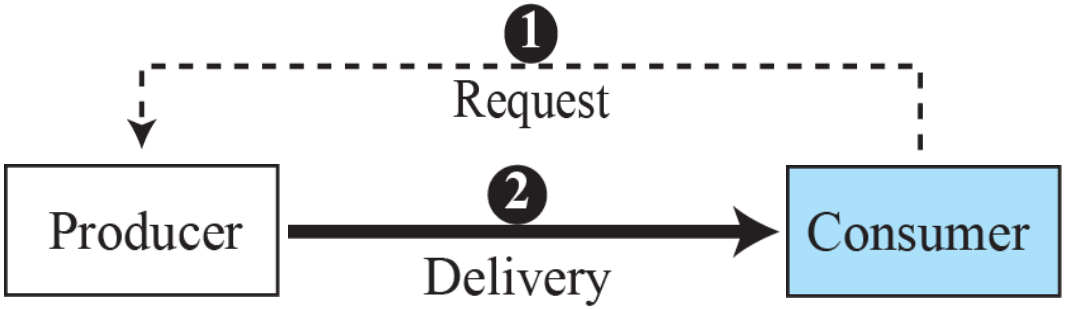


Figure 3.9: *Pushing or pulling*

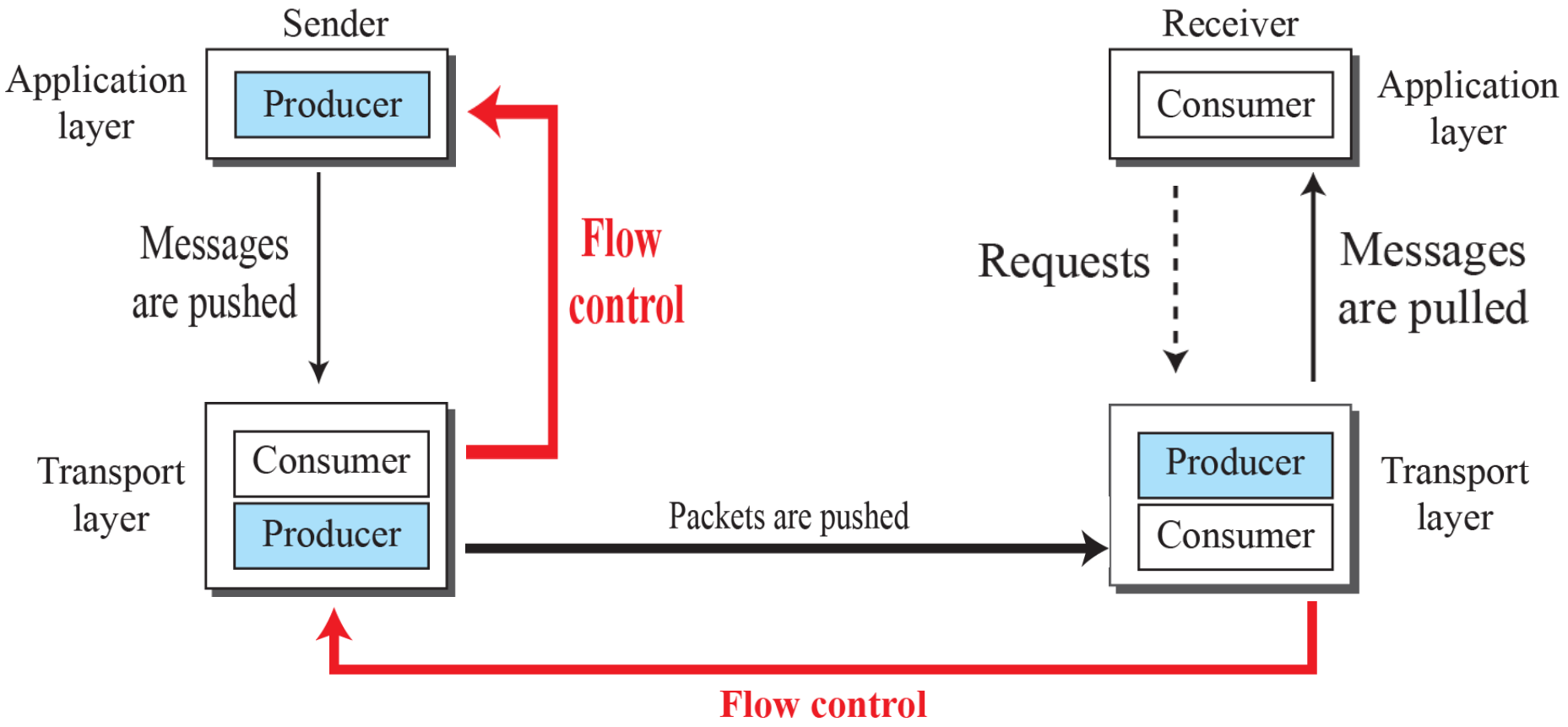


a. Pushing



b. Pulling

Figure 3.10: *Flow control at the transport layer*



Example 3.2

- *The above discussion requires that the consumers communicate with the producers on **two** occasions:*
  - *when the buffer is full and when there are vacancies.*
  - *If the two parties use a buffer with only one slot, the communication can be easier.*
- *Assume that each transport layer uses one single memory location to hold a packet. When this single slot in the sending transport layer is empty, the sending transport layer sends a note to the application layer to send its next chunk; when this single slot in the receiving transport layer is empty, it sends an acknowledgment to the sending transport layer to send its next packet.*
- *As we will see later, however, this type of flow control, using a single-slot buffer at the sender and the receiver, is inefficient.*

Figure 3.11: Error control at the transport layer

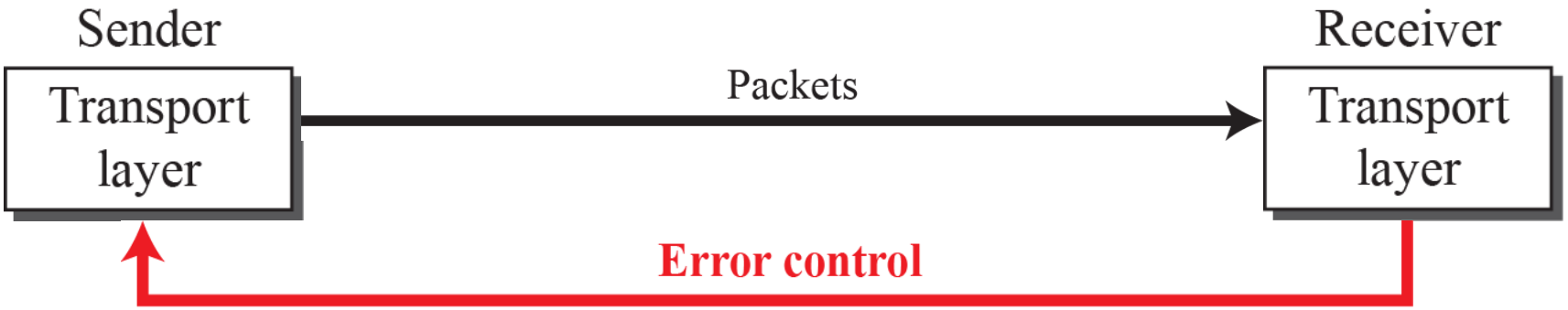
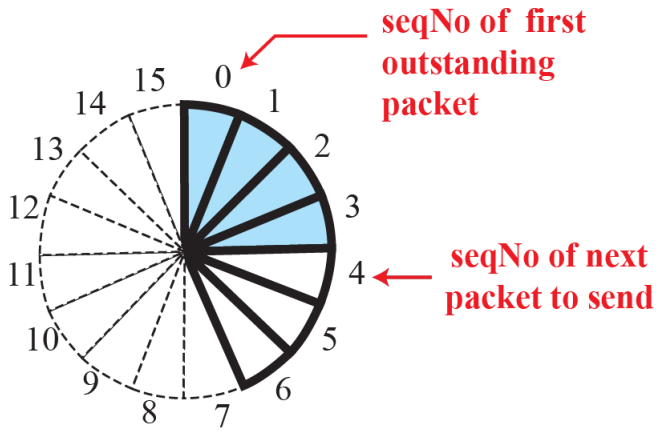
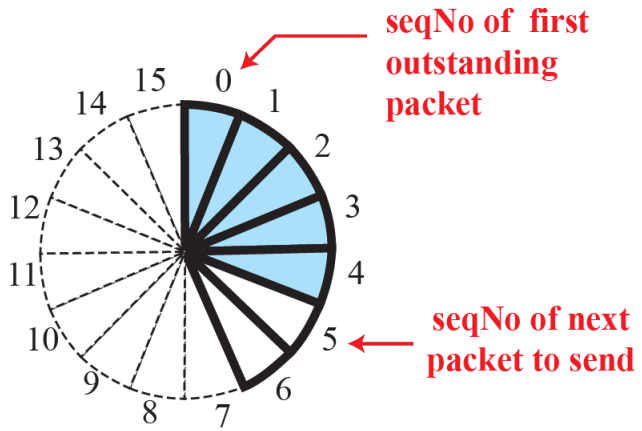


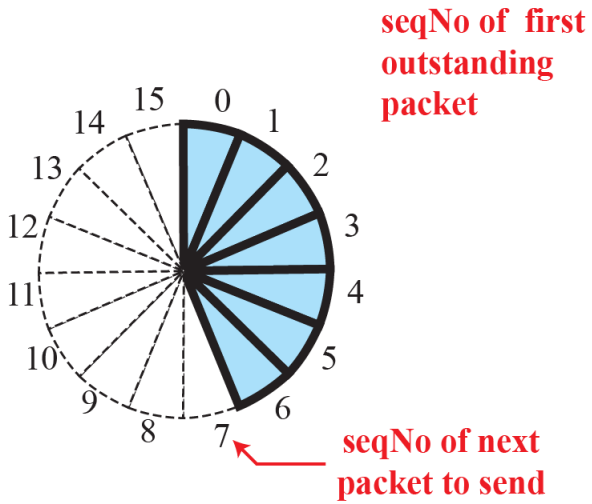
Figure 3.12: Sliding window in circular format



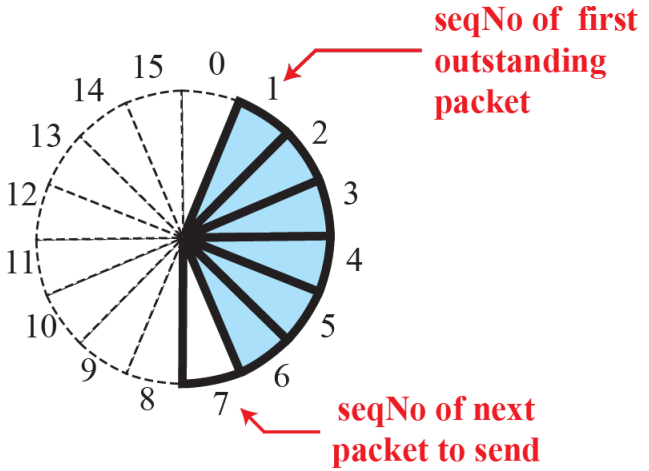
a. Four packets have been sent.



b. Five packets have been sent.

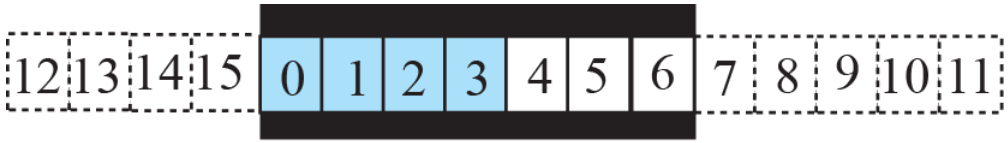


c. Seven packets have been sent; window is full.

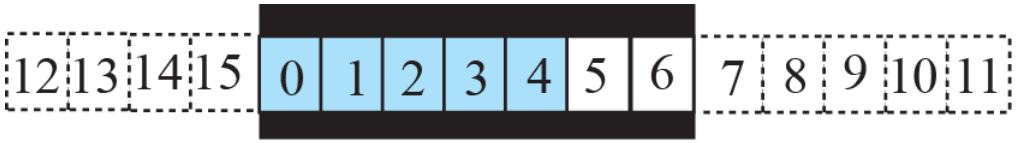


d. Packet 0 has been acknowledged; window slides.

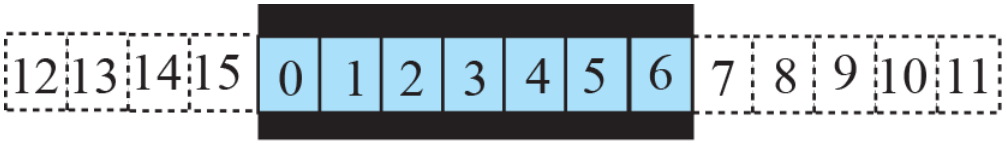
Figure 3.13: Sliding window in linear format



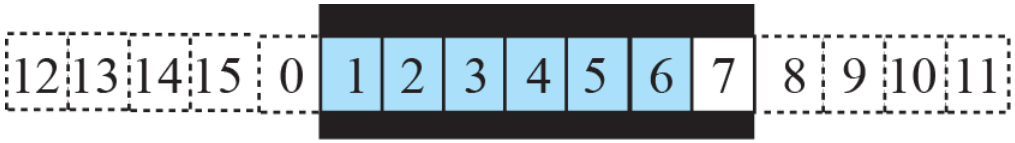
a. Four packets have been sent.



b. Five packets have been sent.



c. Seven packets have been sent;
window is full.



d. Packet 0 has been acknowledged;
window slides.

Figure 3.14: *Connectionless service*

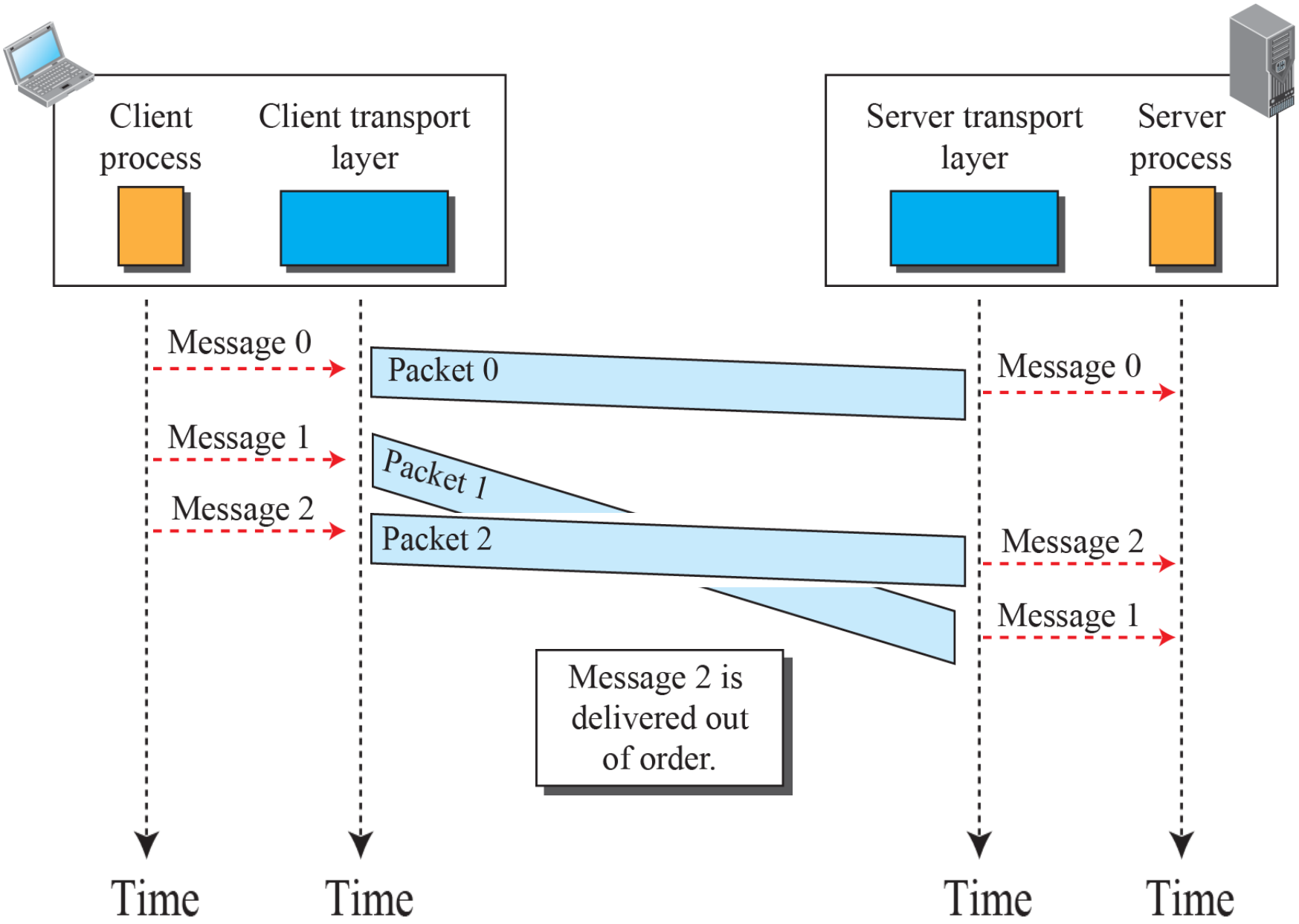
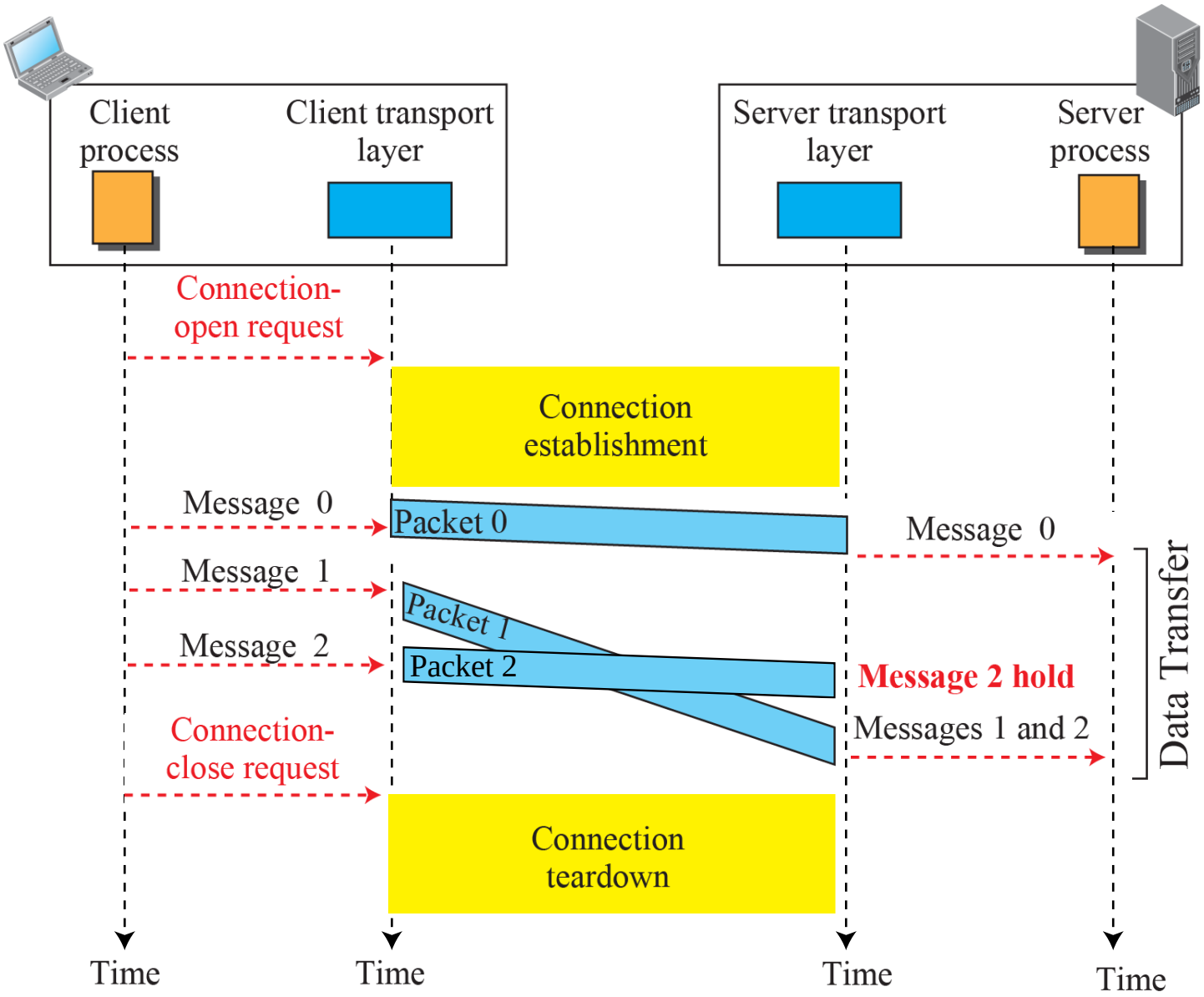


Figure 3.15: Connection-oriented service



3-2 TRANSPORT-LAYER PROTOCOLS

We can create a transport-layer protocol by combining a set of services described in the previous sections.

To better understand the behavior of these protocols, we start with the simplest one and gradually add more complexity.

3.2.1 Simple Protocol

Our first protocol is a simple connectionless protocol with neither flow nor error control. We assume that the receiver can immediately handle any packet it receives. In other words, the receiver can never be overwhelmed with incoming packets. Figure 3.17 shows the layout for this protocol.

Figure 3.17: Simple protocol

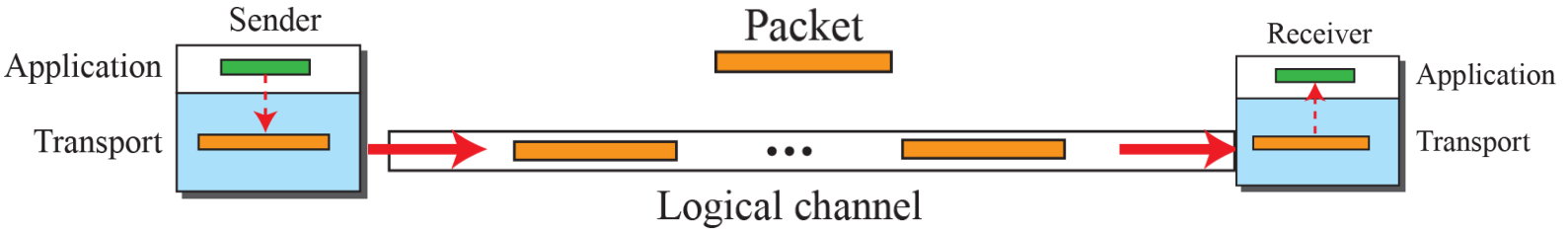
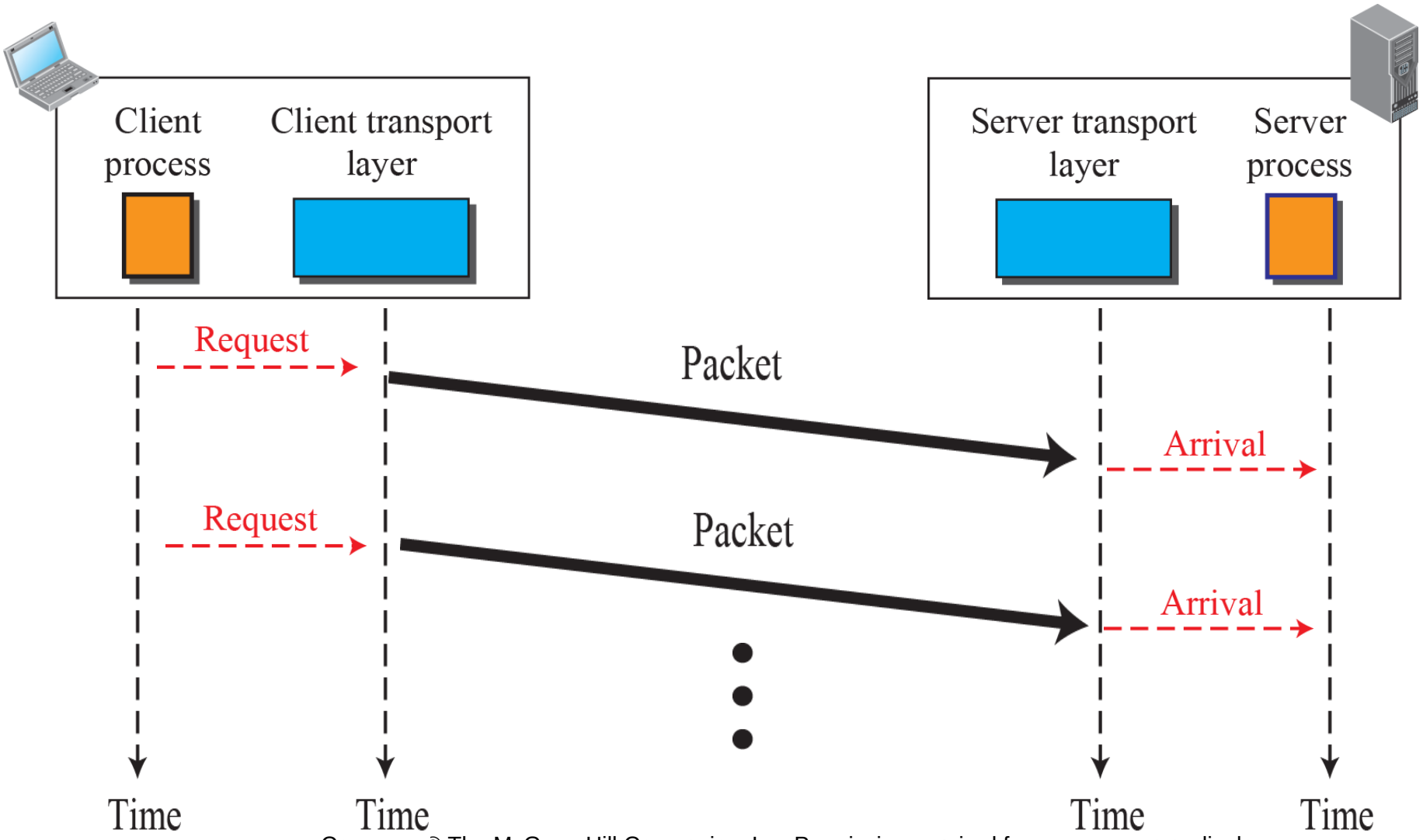


Figure 3.18: FSMs for the simple protocol

Figure 3.19: Flow diagram for Example 3.3

Example 3.3: Figure 3.19 shows an example of communication using this protocol. It is very simple. The ***sender sends packets one after another without even thinking about the receiver****.*



3.2.2 Stop-and-Wait Protocol

- *Our second protocol is a connection-oriented protocol called the Stop-and-Wait protocol, which uses both flow and error control.*
- *Both the sender and the receiver use a sliding window of size 1.*
- *The sender sends one packet at a time and waits for an acknowledgment before sending the next one.*
- *To detect corrupted packets, we need to add a checksum to each data packet. When a packet arrives at the receiver site,*

  - ❑ *Sequence Numbers*
  - ❑ *Acknowledgment Numbers*
  - ❑ *Efficiency*
  - ❑ *Pipelining*

Figure 3.20: Stop-and-Wait protocol

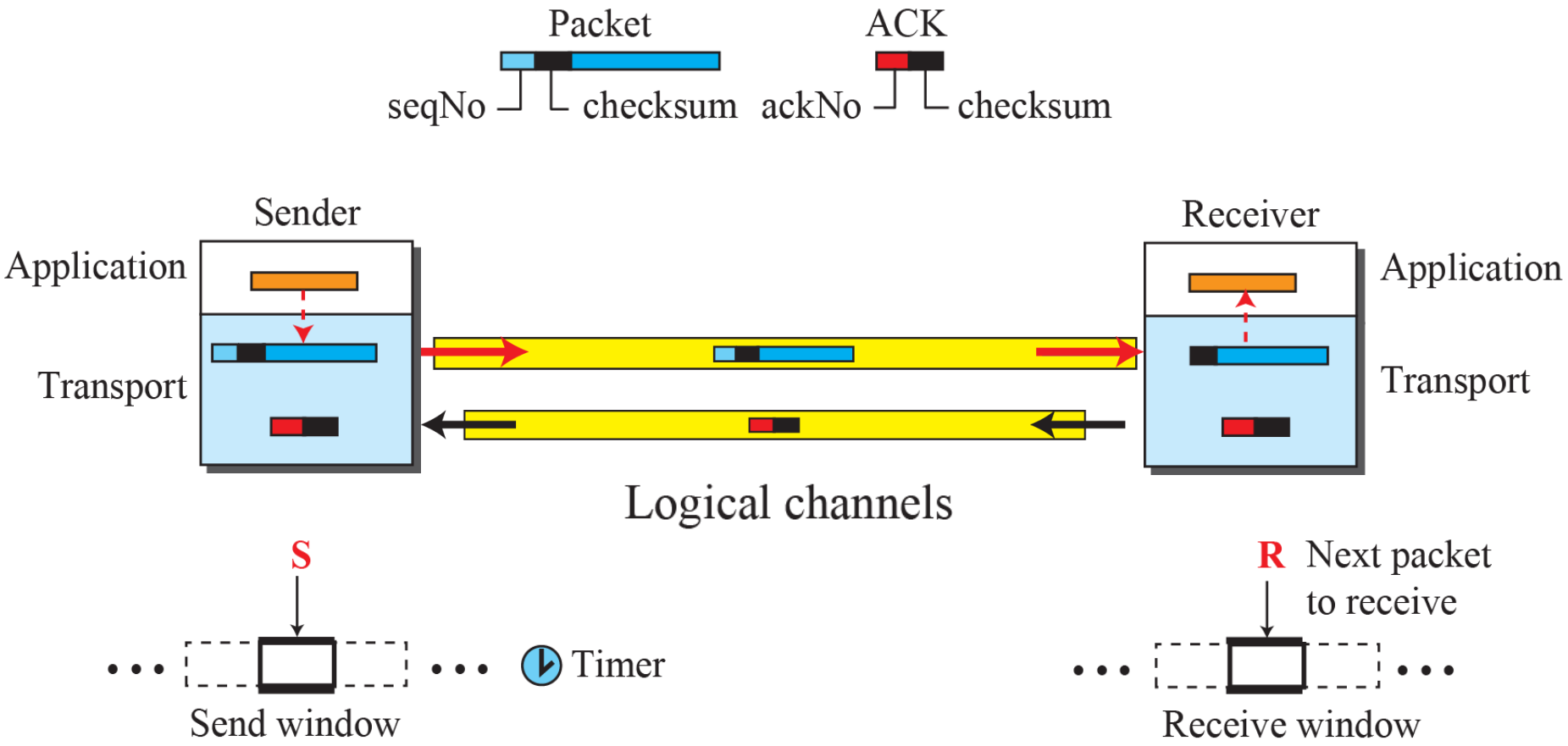
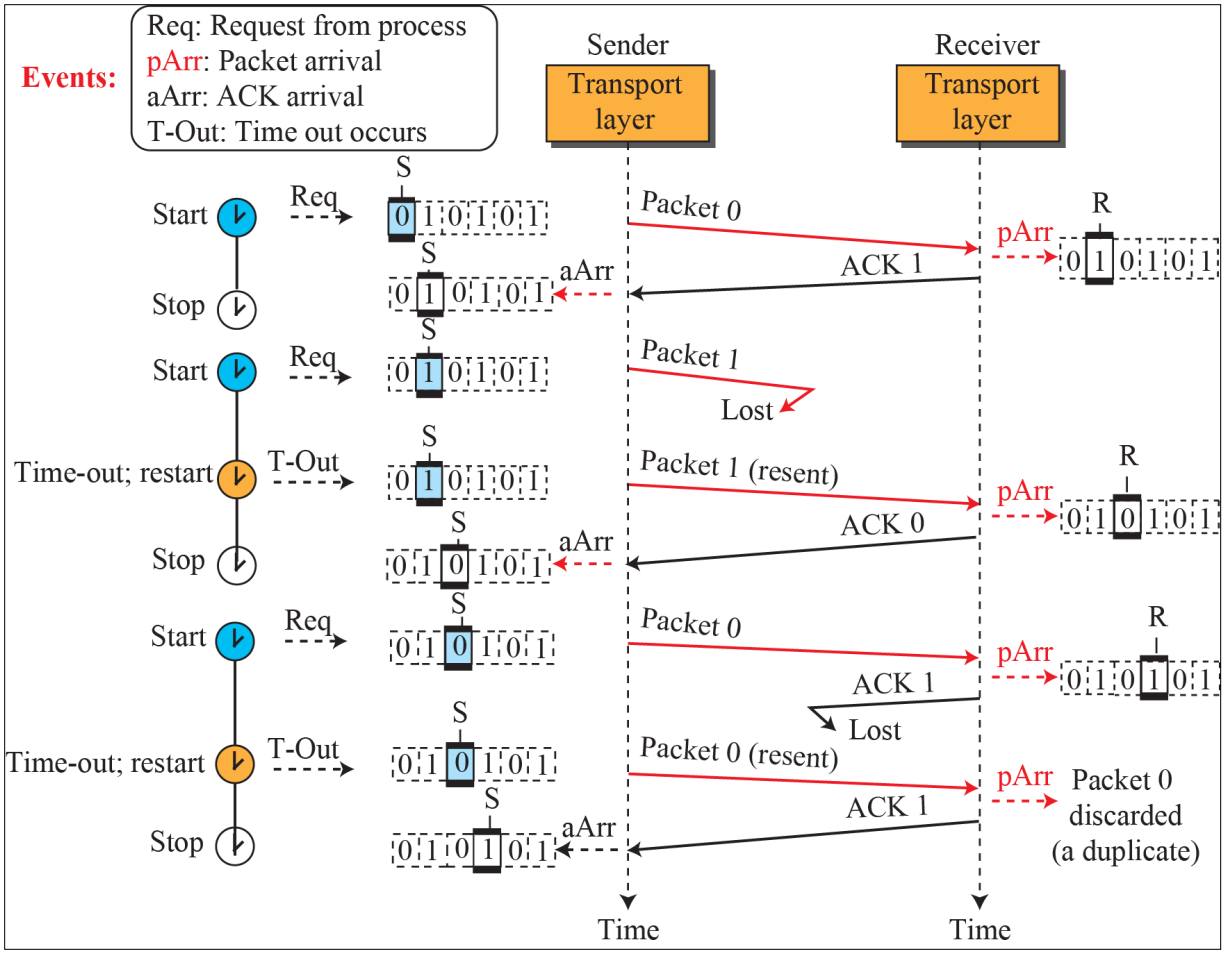


Figure 3.22: Flow diagram for Example 3.4

- *Figure 3.22 shows an example of the Stop-and-Wait protocol.*
- *Packet 0 is sent and acknowledged.*
- *Packet 1 is lost and resent after the time-out.*
- *The resent packet 1 is acknowledged and the timer stops.*
- *Packet 0 is sent and acknowledged, but the acknowledgment is lost.*
- *The sender has no idea if the packet or the acknowledgment is lost, so after the time-out, it resends packet 0, which is acknowledged.*



Example 3.5

Problem: *Assume that, in a Stop-and-Wait system, the bandwidth of the line is 1 Mbps, and 1 bit takes 20 milliseconds to make a round trip. What is the bandwidth-delay product?*

If the system data packets are 1,000 bits in length, what is the utilization percentage of the link?

Solution

The bandwidth-delay product is

$$= BW \times Delay = (1 \times 10^{6}) \times (20 \times 10^{-3}) = 20{,}000 \text{ bits.}$$

The system can send 20,000 bits during the time it takes for the data to go from the sender to the receiver and the acknowledgment to come back.

However, the system sends only 1,000 bits. The link utilization is only 1,000/20,000 == 5 percent.

Example 3.6

Problem: *What is the utilization percentage of the link in Example 3.5 if we have a protocol that can send up to 15 packets before stopping and worrying about the acknowledgments?*

Solution

The bandwidth-delay product is still the same;
= BW × Delay = (1 × 10^6) × (20 × 10^{-3}) = 20,000 bits.

Now the system can send up to 15 packets or 15 x 1000 = 15,000 bits during a round trip.

This means the utilization is now 15,000/20,000, or 75 percent. Of course, if there are damaged packets, the utilization percentage is much less because packets have to be ***resent****.*

3.2.3 Go-Back-N Protocol

To improve the efficiency of transmission multiple packets must be in transition while the sender is waiting for acknowledgment.

In this section, we discuss one protocol that can achieve this goal; in the next section,

we discuss a second. The first is called Go-Back-N (GBN) (the rationale for the name will become clear later).

- ***Sequence Numbers***
- ***Acknowledgment Numbers***
- ***Send Window***
- ***Receive Window***
- ***Timers***
- ***Resending packets***
- ***Send Window Size***
- ***Go-Back-N versus Stop-and-Wait***

Figure 3.24: Send window for Go-Back-N

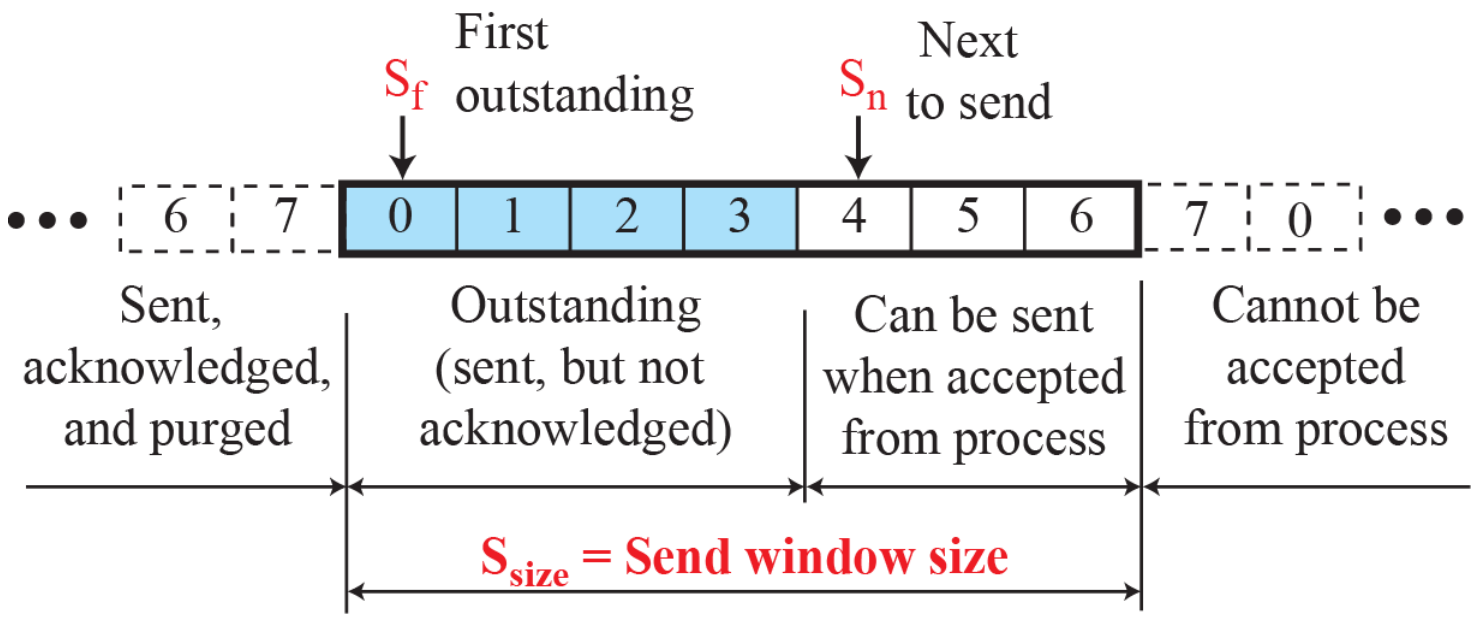


Figure 3.25: Sliding the send window

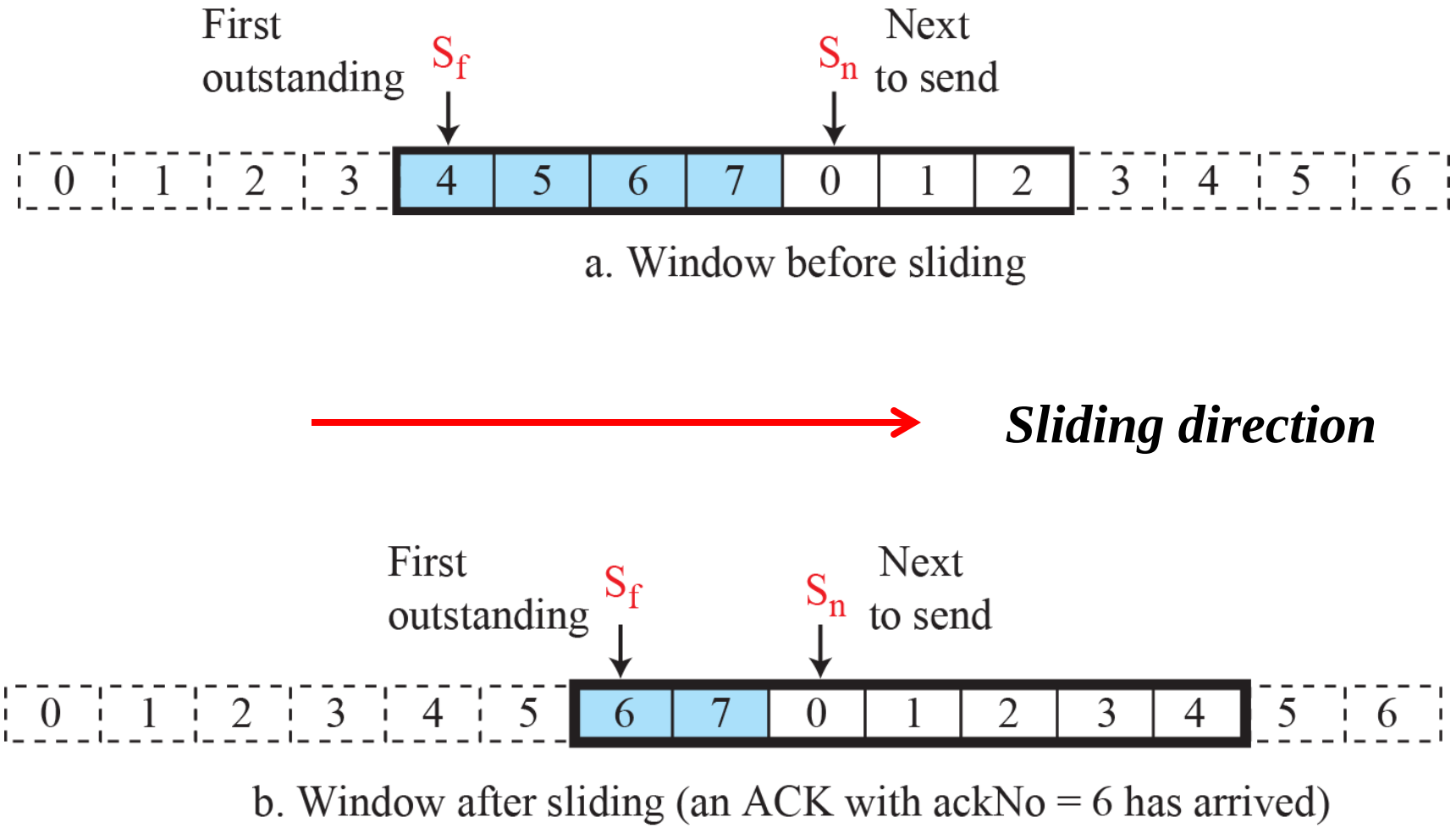
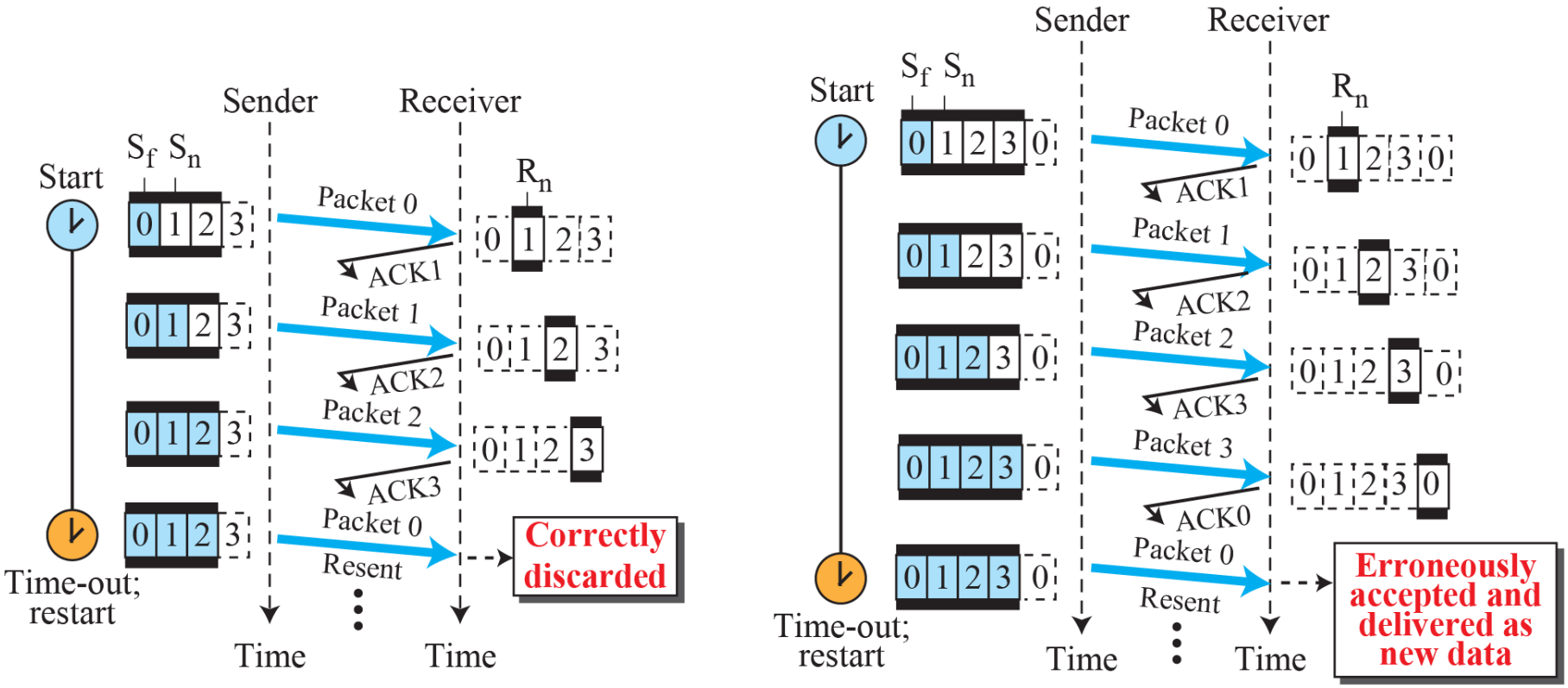


Figure 3.28: Send window size for Go-Back-N



a. Send window of size $< 2^m$

b. Send window of size $= 2^m$

Figure 3.29: Flow diagram for Example 3.7

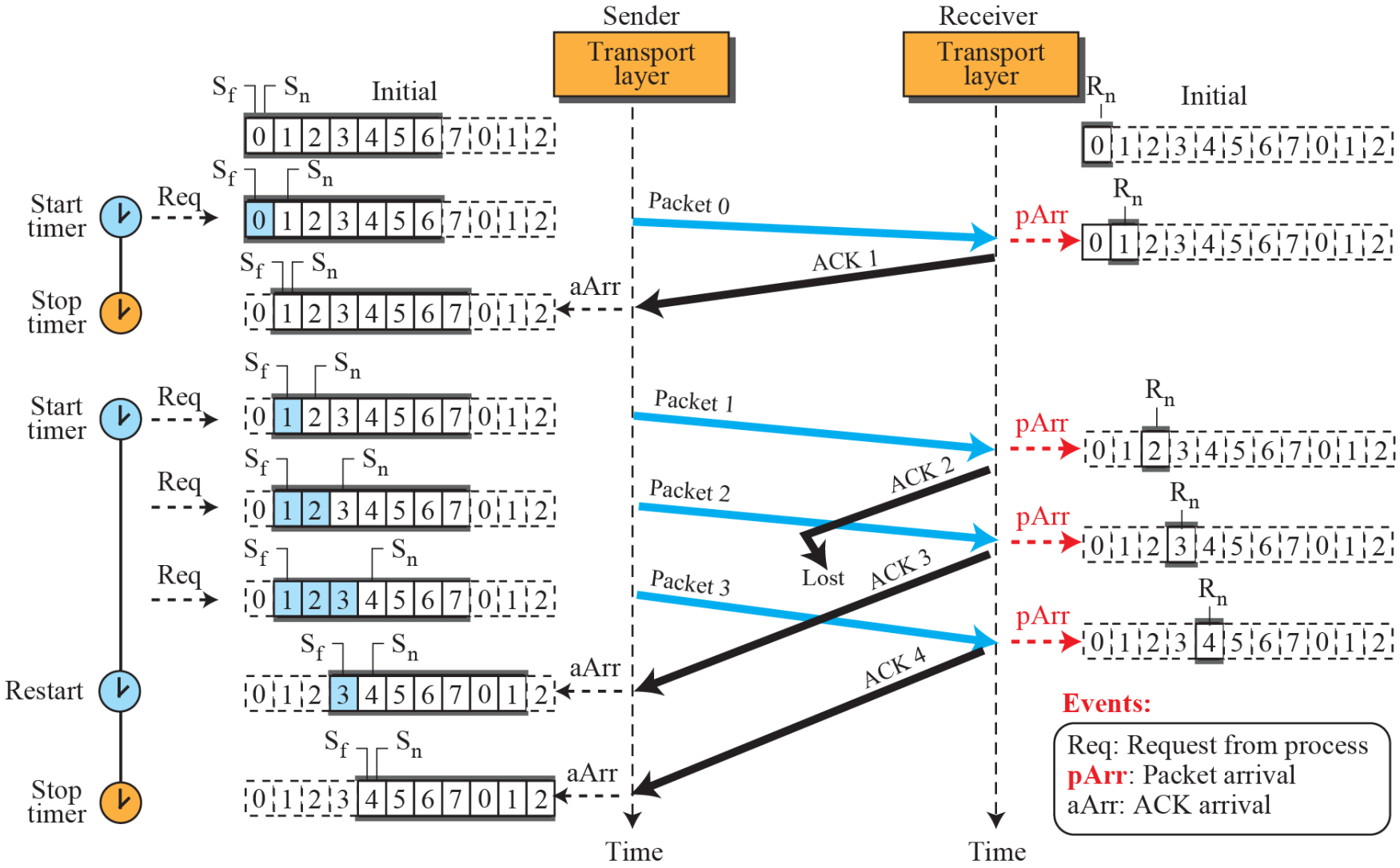
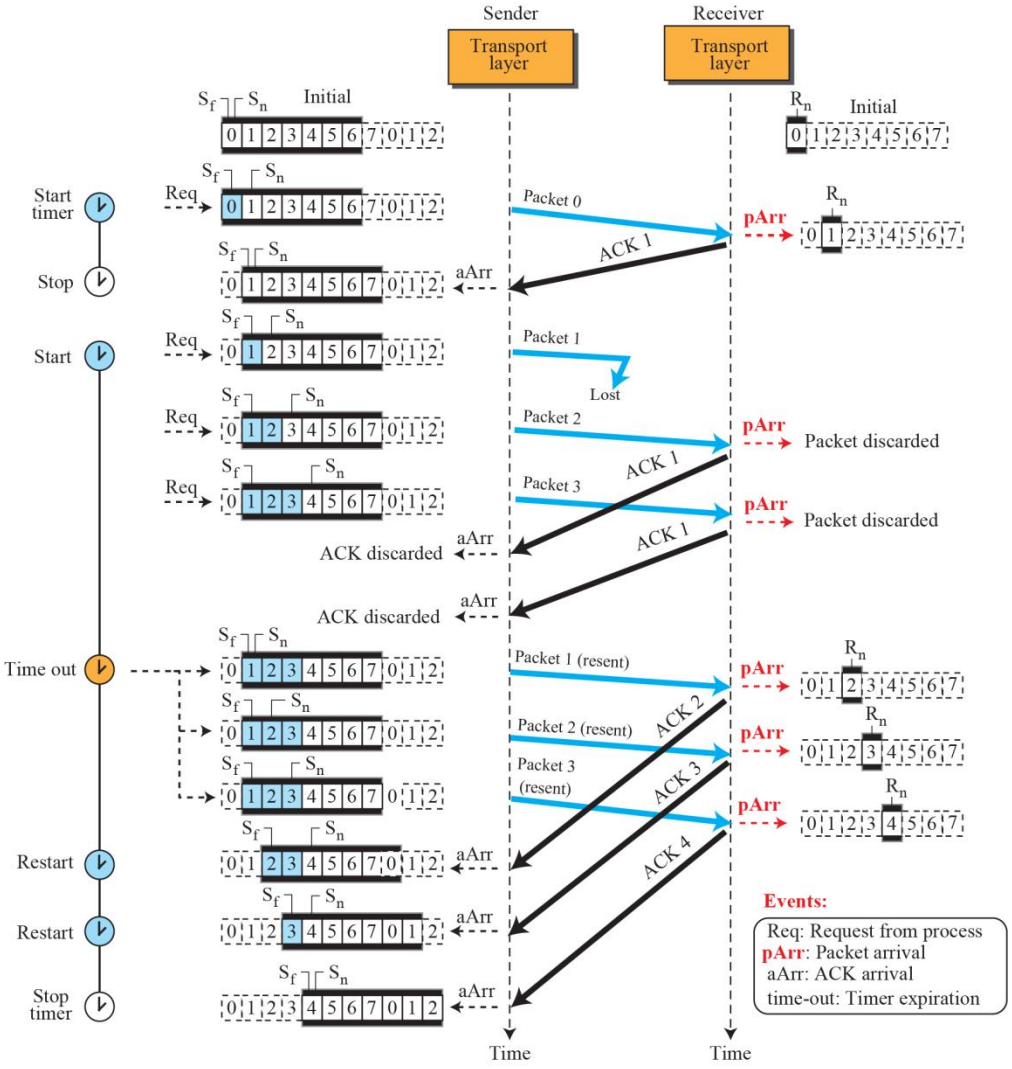


Figure 3.30: Flow diagram for Example 3.8



3.2.4 Selective-Repeat Protocol

- *The Go-Back-N protocol simplifies the process at the receiver.*
- *The receiver keeps track of only one variable, and there is no need to buffer out-of-order packets; they are simply discarded.*
- *Another protocol, called the Selective-Repeat (SR) protocol, has been devised.*
- *Selective-Repeat (SR) protocol, as the name implies, resends only selective packets, those that are actually lost.*
- *The outline of this protocol is shown in Figure 3.31.*

Figure 3.35: Flow diagram for Example 3.10

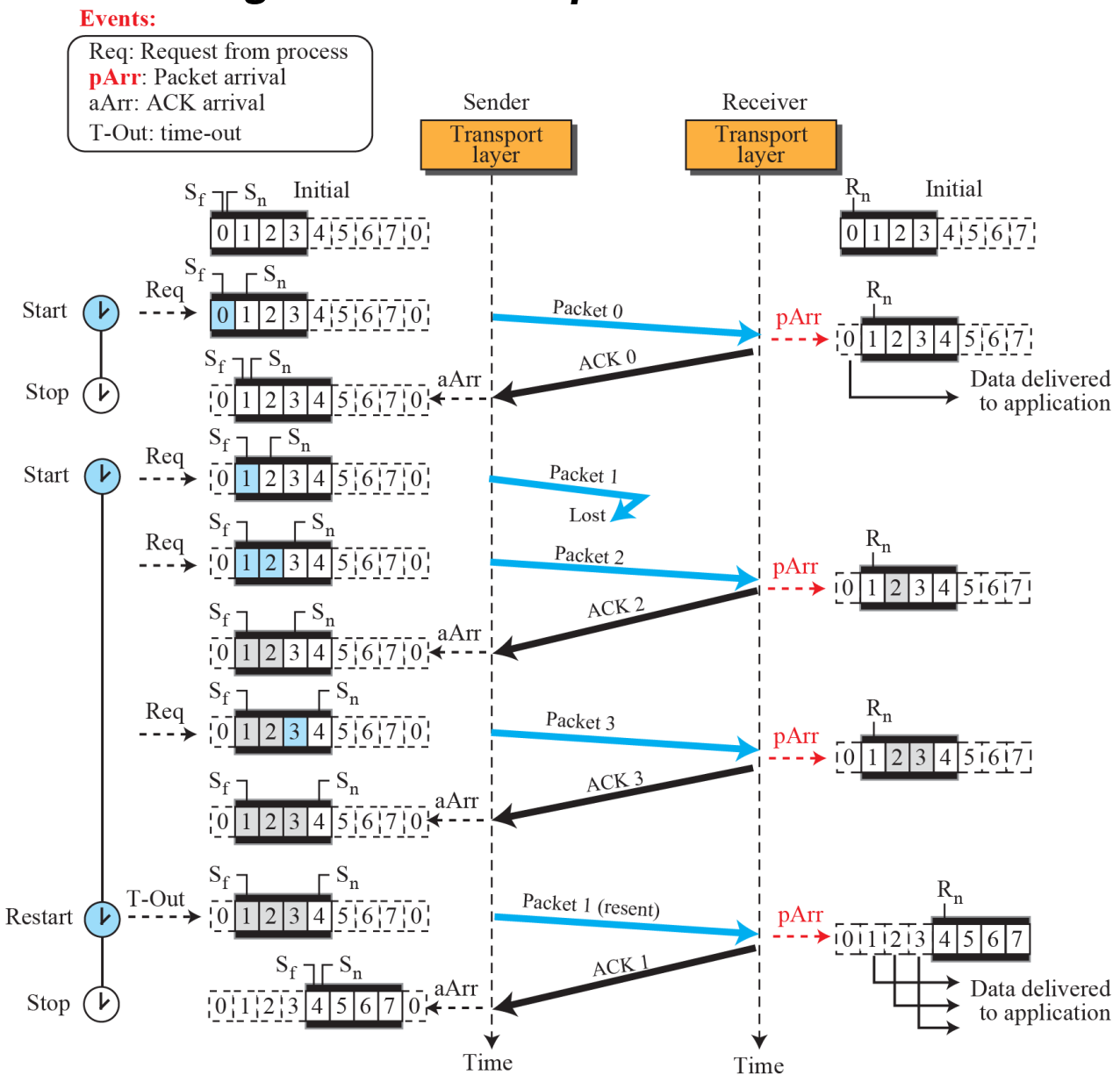
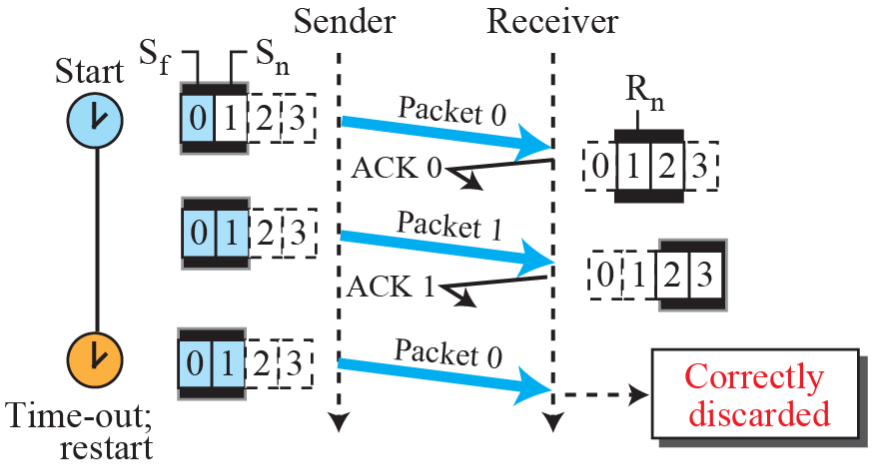
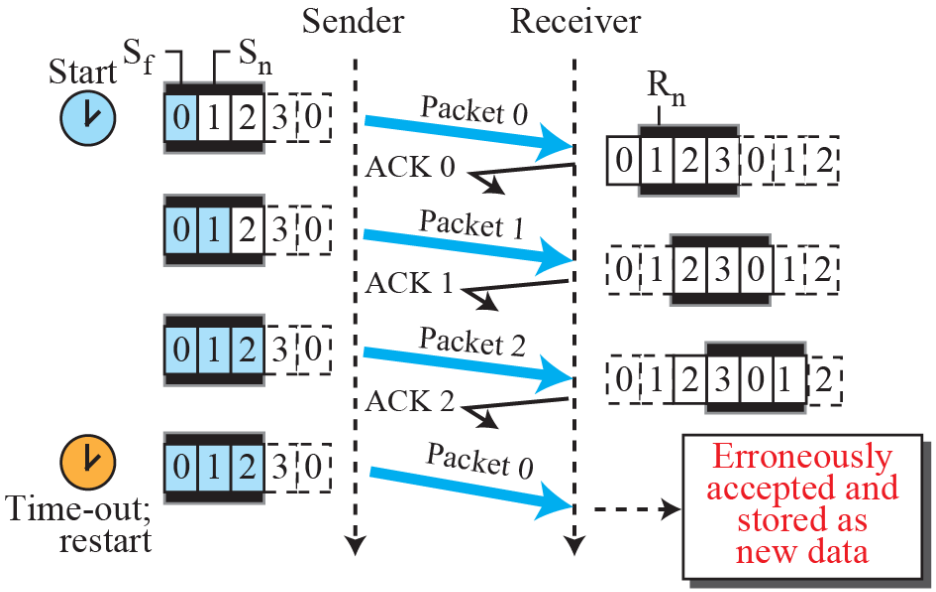


Figure 3.36: Selective-Repeat, window size



a. Send and receive windows of size = 2^{m-1}

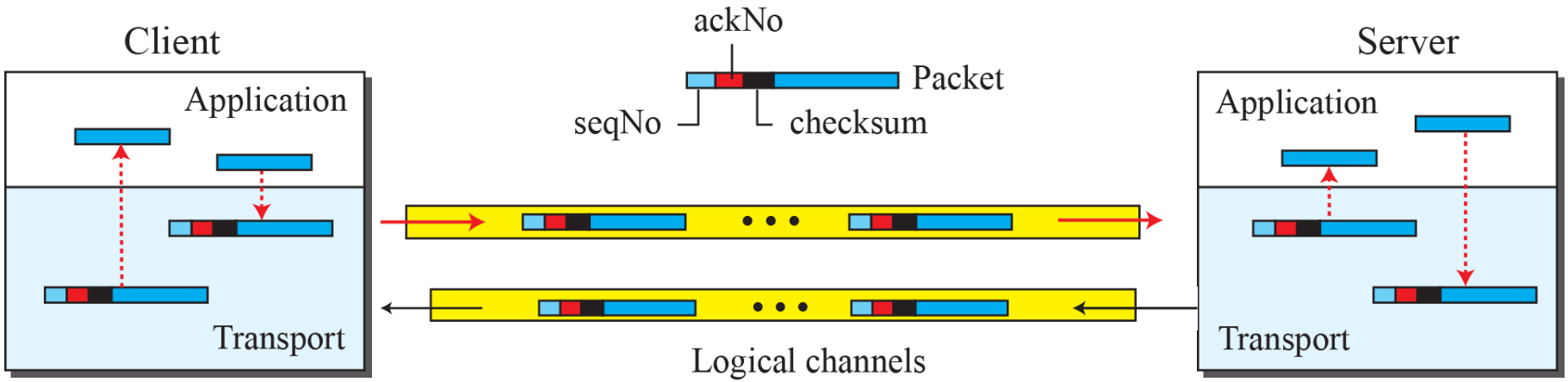


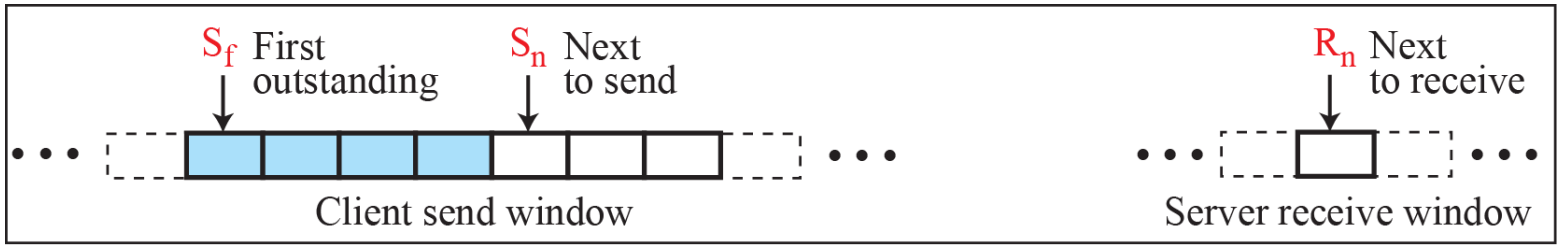
b. Send and receive windows of size > 2^{m-1}

3.2.5 Bidirectional Protocols

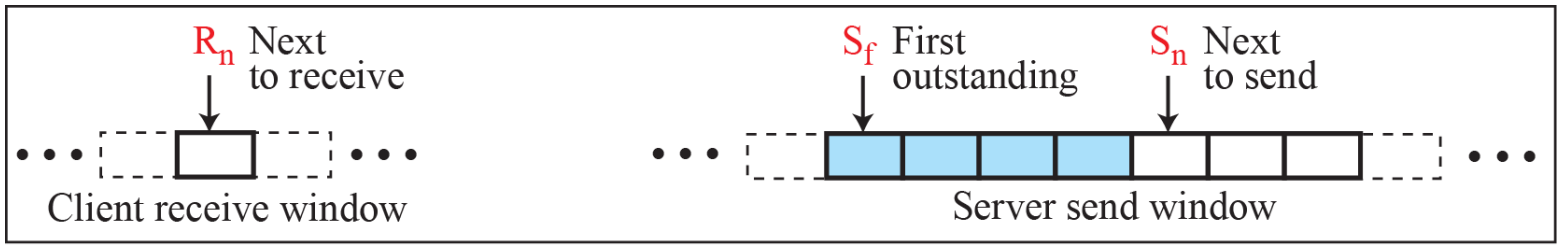
- *The previous four protocols discussed earlier were all unidirectional;*
- *Data packets flow in only one direction and acknowledgments travel in the other direction.*
- *In real systems, data packets are normally flowing in both directions: from client to server and from server to client.*
- *This means that acknowledgments are also need to flow in both directions.*
- *A technique called piggybacking is used to improve the efficiency of the bidirectional protocols.*

Figure 3.37: Design of piggybacking in Go-Back-N





Windows for communication from client to server



Windows for communication from server to client

3.2.6 Internet Transport-Layer Protocols

- *A network is the interconnection of a set of devices capable of communication.*
- *In this definition, a device can be a host such as a large computer, desktop, laptop, workstation, cellular phone, or security system.*
- *A device in this definition can also be a connecting device such as a router a switch, a modem that changes the form of data, and so on.*

Figure 3.38: Position of transport-layer protocols in the TCP/IP protocol suite

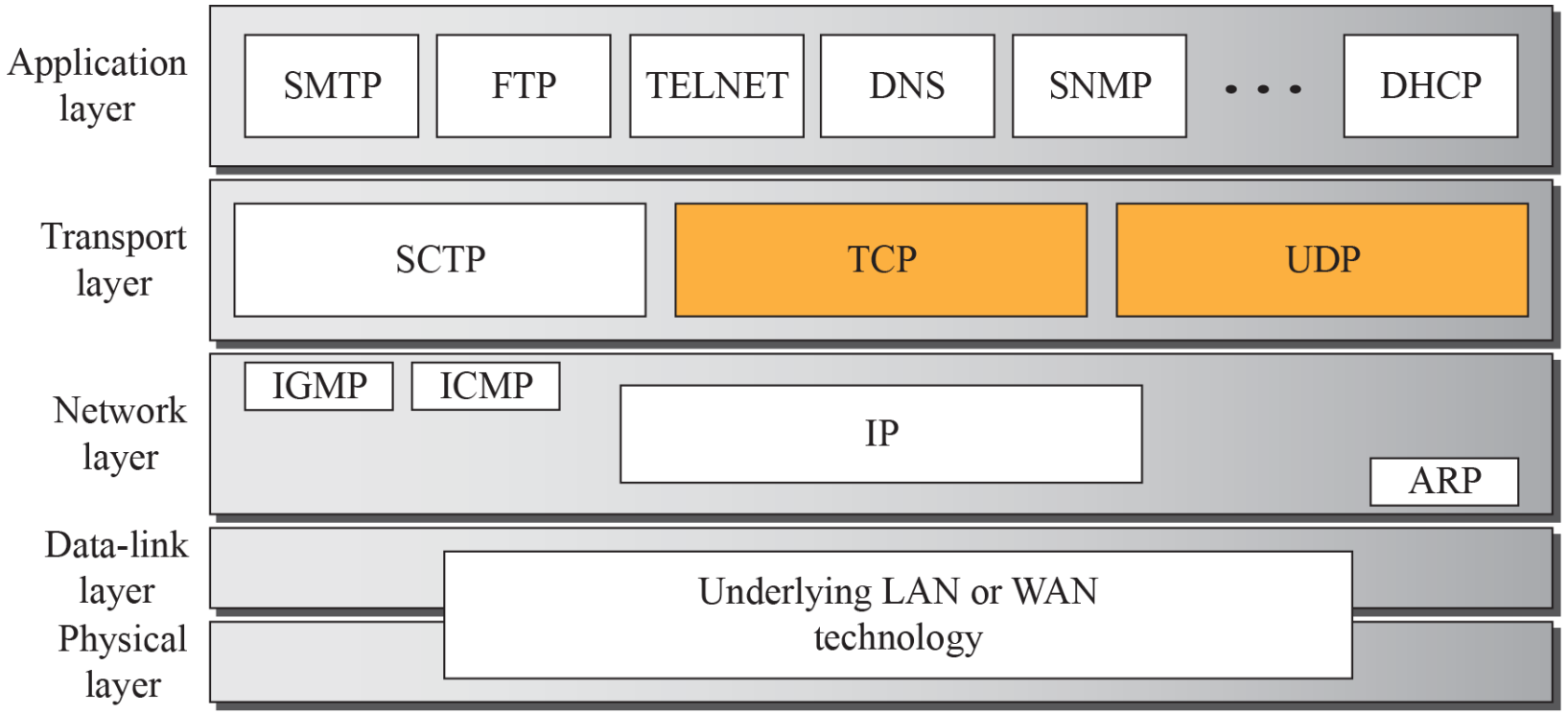


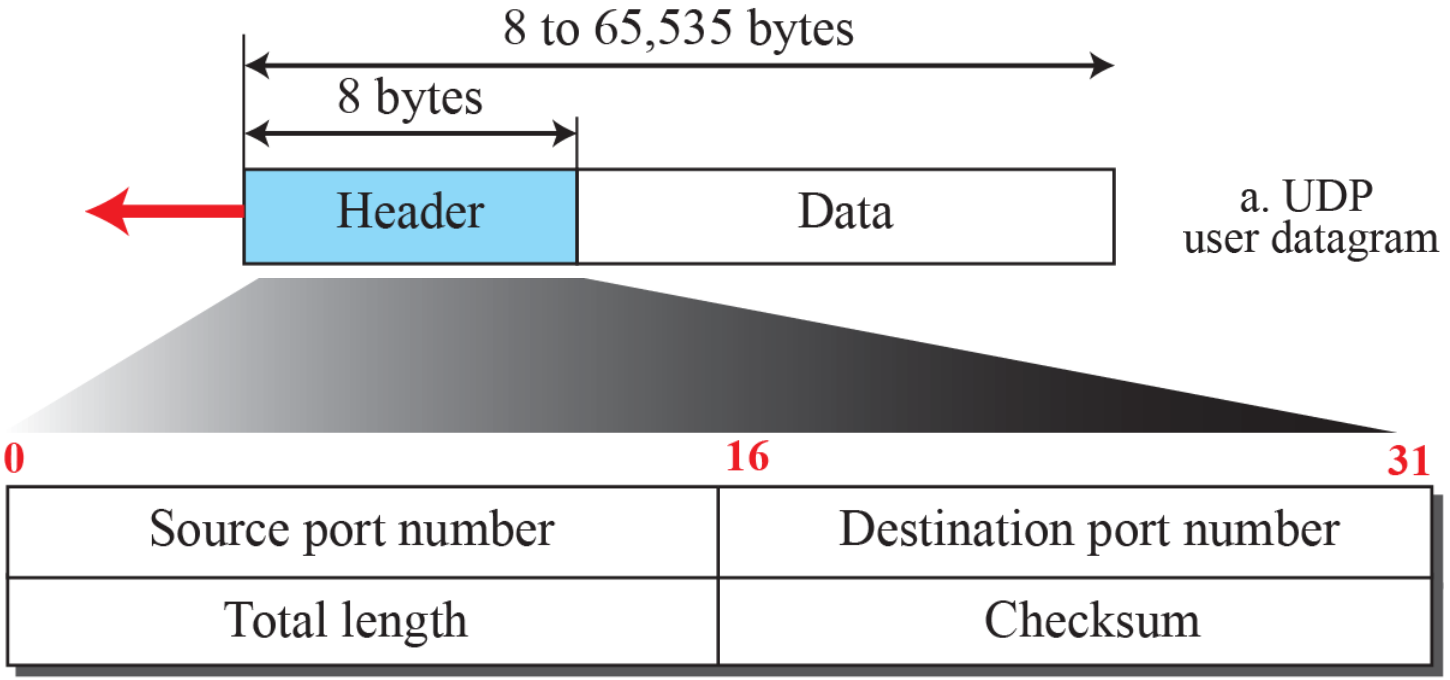
Table 3.1: Some well-known ports used with UDP and TCP

Port	*Protocol*	*UDP*	*TCP*	*Description*
7	Echo	√		Echoes back a received datagram
9	Discard	√		Discards any datagram that is received
11	Users	√	√	Active users
13	Daytime	√	√	Returns the date and the time
17	Quote	√	√	Returns a quote of the day
19	Chargen	√	√	Returns a string of characters
20, 21	FTP		√	File Transfer Protocol
23	TELNET		√	Terminal Network
25	SMTP		√	Simple Mail Transfer Protocol
53	DNS	√	√	Domain Name Service
67	DHCP	√	√	Dynamic Host Configuration Protocol
69	TFTP	√		Trivial File Transfer Protocol
80	HTTP		√	Hypertext Transfer Protocol
111	RPC	√	√	Remote Procedure Call
123	NTP	√	√	Network Time Protocol
161, 162	SNMP		√	Simple Network Management Protocol

3-3 USER DATAGRAM PROTOCOL (UDP)

- ***The User Datagram Protocol (UDP) is a connectionless, unreliable transport protocol.***
- ***It does not add anything to the services of IP except for providing process-to-process instead of host-to-host communication.***
- ***UDP is a very simple protocol using a minimum of overhead.***

Figure 3.39: User datagram packet format

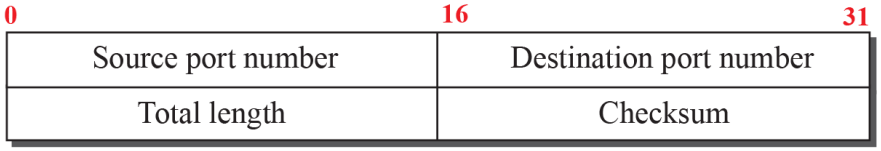


- *UDP packets, called user datagrams, have a fixed size header of **8 bytes made of four fields, each of 2 bytes (16 bits).***
- *Figure 3.39 shows the format of a user datagram. The first two fields define the **source and destination port** numbers.*
- *The **third field defines the total length** of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes.*

Example 3.11

The following is the contents of a UDP header in hexadecimal format.

CB84000D001C001C



b. Header format

a. What is the source port number?

The source port number is the first four hexadecimal digits $(CB84)_{16}$ or 52100

b. What is the destination port number?

The destination port number is the second four hexadecimal digits $(000D)_{16}$ or 13.

c. What is the total length of the user datagram?

The third four hexadecimal digits $(001C)_{16}$ define the length of the whole UDP packet as 28 bytes.

d. What is the length of the data?

The length of the data is the length of the whole packet minus the length of the header, or $28 - 8 = 20$ bytes.

e. Is the packet directed from a client to a server or vice-versa?

Since the destination port number is 13 (well-known port), the packet is from the client to the server.

f. What is the client process?

The client process is the Daytime (see Table 3.1).

3.3.2 UDP Services

Earlier we discussed the general services provided by a transport-layer protocol.

In this section, we discuss what portions of those general services are provided by UDP.

- *Process-to-Process Communication*
- *Connectionless Services*
- *Flow Control*
- *Error Control*

3.3.2 (continued)

- *Checksum*
  - *Optional Inclusion of Checksum*
- *Congestion Control*
- *Encapsulation and Decapsulation*
- *Queuing*
- *Multiplexing and Demultiplexing*
- *Comparison : UDP and Simple Protocol*

Figure 3.40: UDP Pseudoheader for checksum calculation

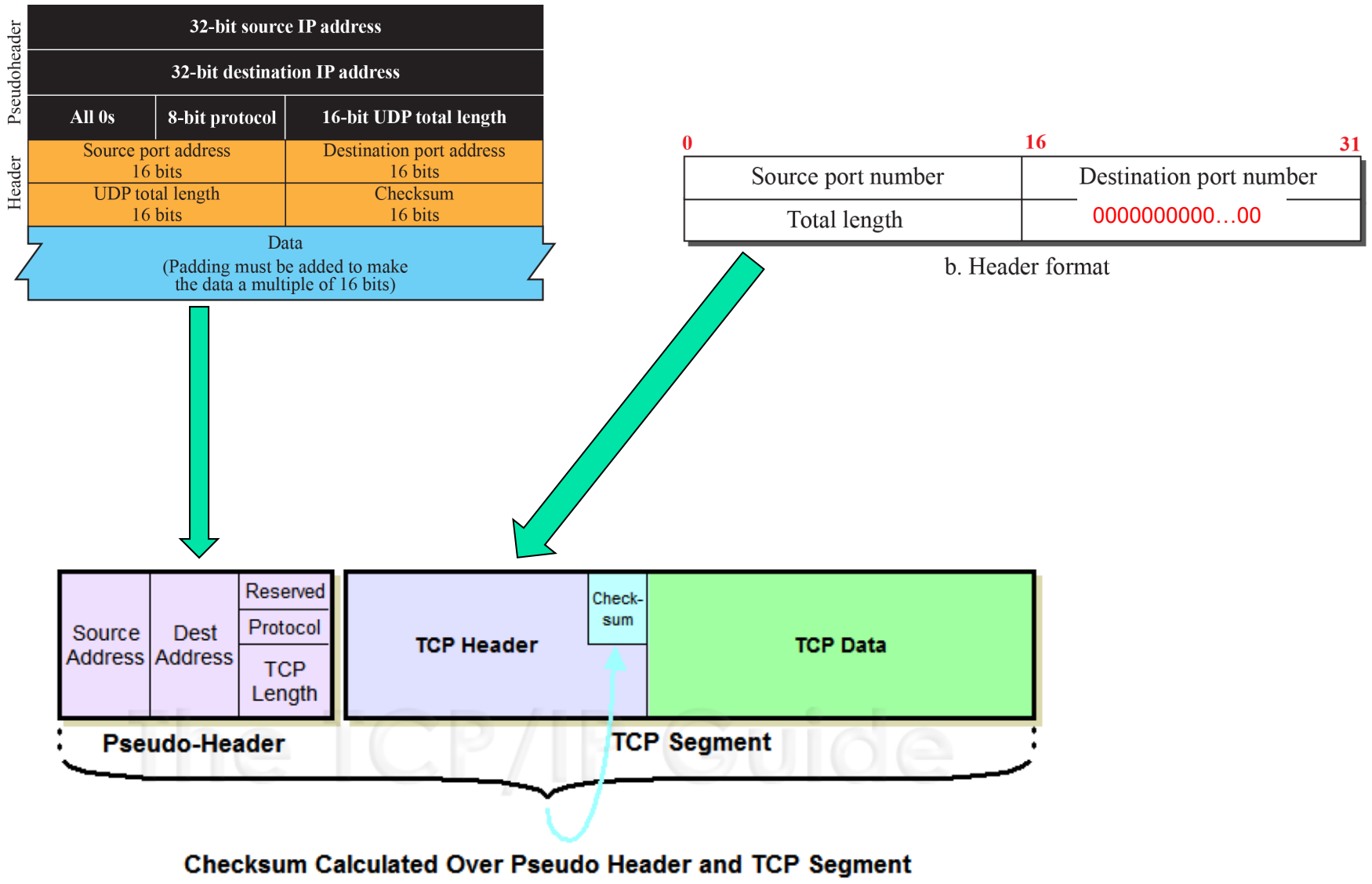
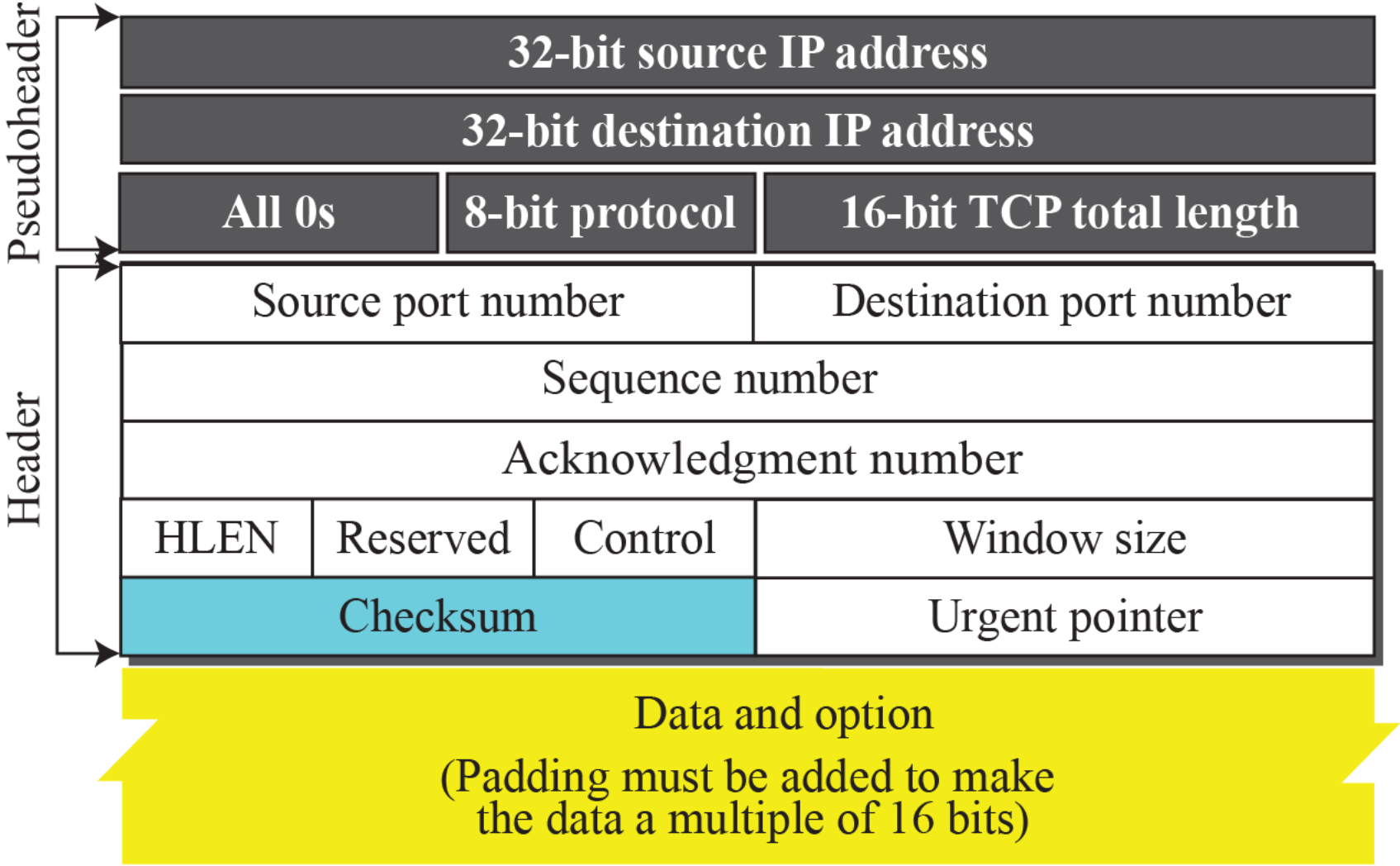


Figure 3.46: Pseudoheader added to the TCP datagram



Example 3.12

What value is sent for the checksum in one of the following hypothetical situations?

a. The sender decides not to include the checksum.

The value sent for the checksum field is all 0s to show that the checksum is not calculated.

b. The sender decides to include the checksum, but the value of the sum is all 1s.

When the sender complements the sum, the result is all 0s; the sender complements the result again before sending. The value sent for the checksum is all 1s. The second complement operation is needed to avoid confusion with the case in part a.

c. The sender decides to include the checksum, but the value of the sum is all 0s.

This situation never happens because it implies that the value of every term included in the calculation of the sum is all 0s, which is impossible; some fields in the pseudoheader have nonzero values.

3.3.3 UDP Applications

- *Although UDP meets almost none of the criteria we mentioned earlier for a reliable transport-layer protocol, UDP is preferable for some applications.*
- *The reason is that some services may have some side effects that are either unacceptable or not preferable. An application designer sometimes needs to compromise to get the optimum.*
- *In this section, we first discuss some features of UDP that may need to be considered when one designs an application program and then show some typical applications.*

3.3.3 (continued)

- *UDP Features*
  - *Connectionless Service*
  - *Lack of Error Control*
  - *Lack of Congestion Control*
- *Typical Applications*

Example 3.13

- *A client-server application such as DNS (see Chapter 2) uses the services of UDP because a client needs to send a short request to a server and to receive a quick response from it.*
- *The request and response can each fit in one user datagram.*
- *Since only one message is exchanged in each direction, the connectionless feature is not an issue;*
- *the client or server does not worry that messages are delivered out of order.*

Example 3.14

- A client-server application such as **SMTP** (see Chapter 2), which is used in electronic mail, **cannot** use the services of UDP because a user can send a long e-mail message, which may include multimedia (images, audio, or video).
- If the application uses UDP and the message does not fit in one single user datagram, the message must be split by the application into different user datagrams. Here the connectionless service may create problems. The user datagrams may arrive and be delivered to the receiver application out of order.
- The receiver application may not be able to reorder the pieces. This means the connectionless service has a disadvantage for an application program that sends long messages.

Example 3.15

- Assume we are downloading a very large text file from the Internet. (**FTP**)
- We definitely need to use a transport layer that provides reliable service. We don't want part of the file to be missing or corrupted when we open the file.
- The delay created between the deliveries of the parts is not an overriding concern for us; we wait until the whole file is composed before looking at it.
- In this case, UDP is **not** a suitable transport layer.

Example 3.16

- Assume we are using a real-time interactive application, such as **Skype**.
- Audio and video are divided into frames and sent one after another.
- If the transport layer is supposed to resend a corrupted or lost frame, the synchronizing of the whole transmission may be lost. The viewer suddenly sees a blank screen and needs to wait until the second transmission arrives. This is not tolerable.
- However, if each small part of the screen is sent using one single user datagram, the receiving UDP can easily ignore the corrupted or lost packet and deliver the rest to the application program.
- That part of the screen is blank for a very short period of time, which most viewers do not even notice.

3-4 TRANSMISSION CONTROL PROTOCOL

- Transmission Control Protocol (TCP) is a connection-oriented, reliable protocol.
- TCP explicitly defines connection establishment, data transfer, and connection teardown phases to provide a connection-oriented service.
- TCP uses a combination of GBN and SR protocols to provide reliability.

3.4.1 TCP Services

A Before discussing TCP in detail, let us explain the services offered by TCP to the processes at the application layer.

- *Process-to-Process Communication*
- *Stream Delivery Service*
  - *Sending and Receiving Buffers*
  - *Segments*
- *Full-Duplex Communication*
- *Multiplexing and Demultiplexing*
- *Connection-Oriented Service*
- *Reliable Service*

Figure 3.41: *Stream delivery*

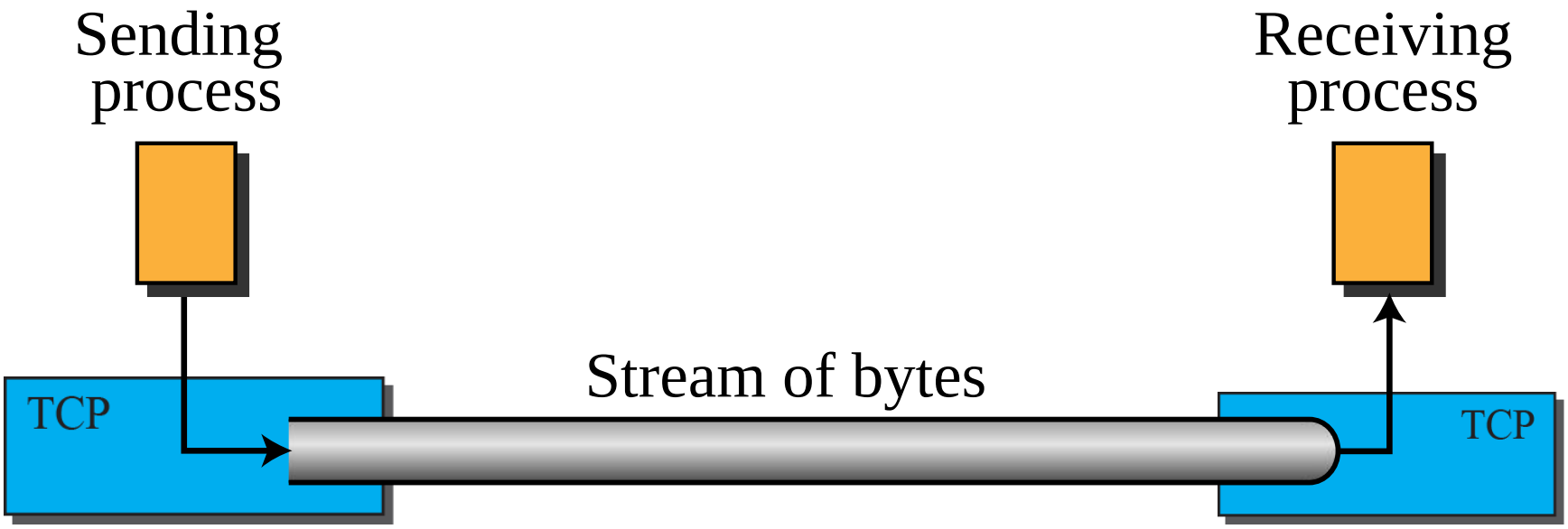


Figure 3.42: Sending and receiving buffers

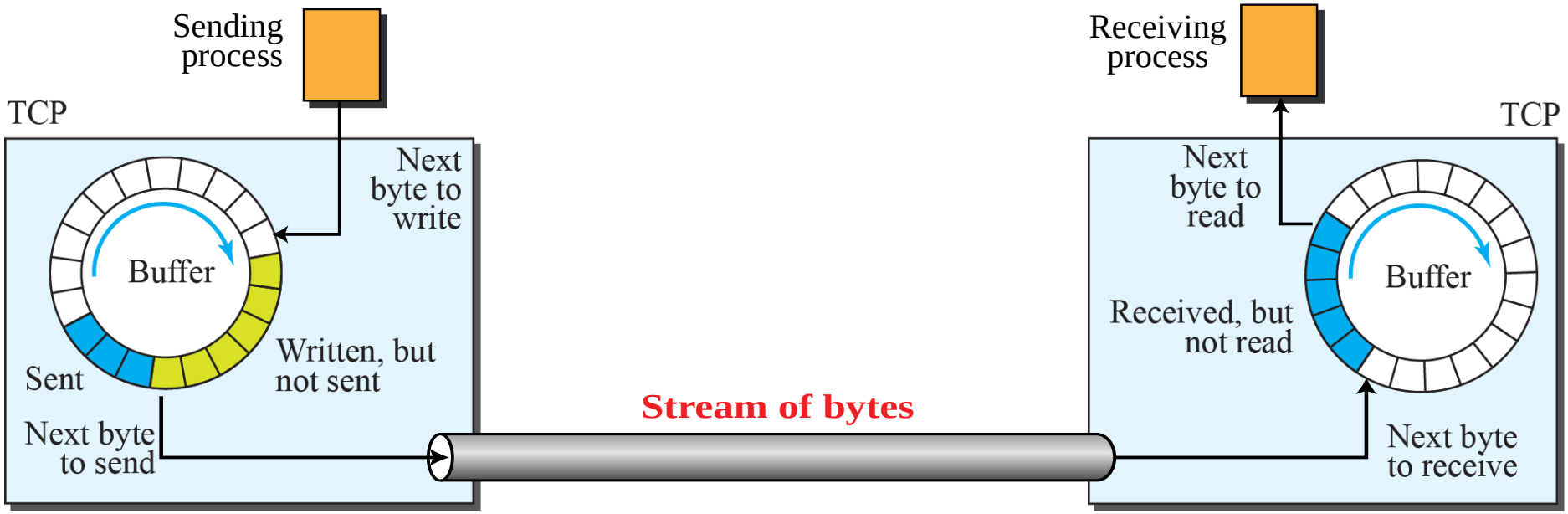
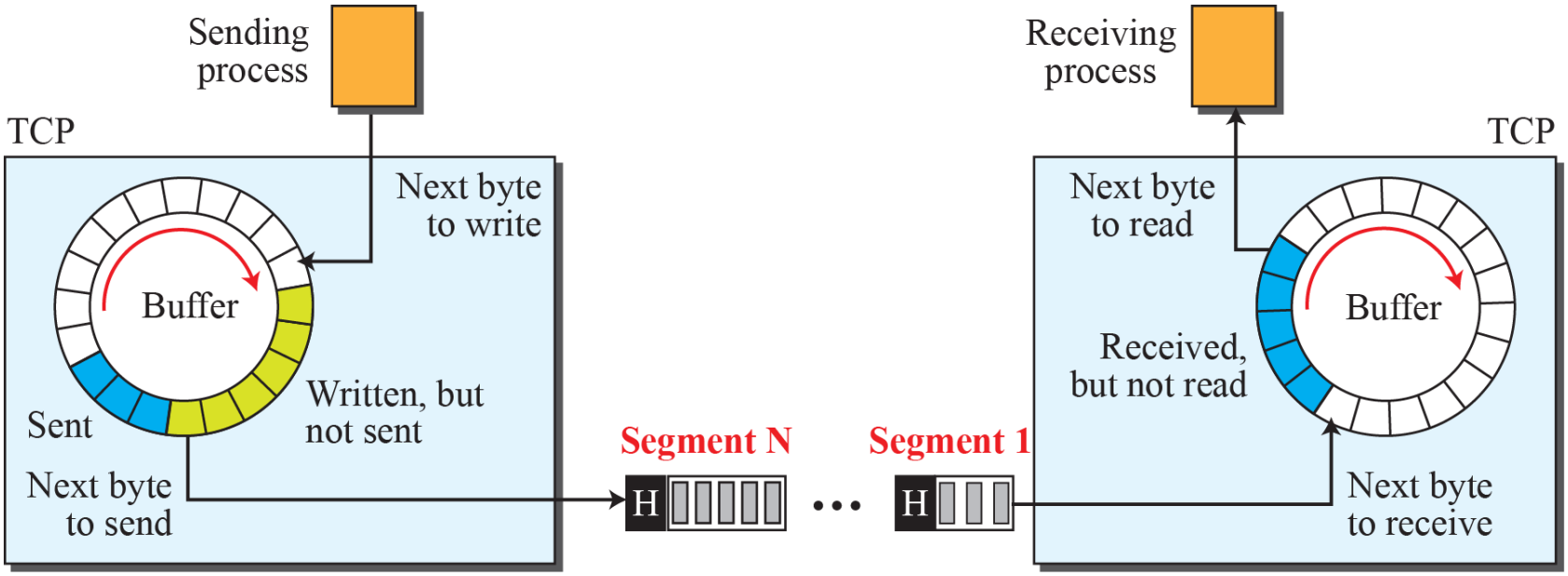


Figure 3.43: TCP segments



3.4.2 TCP Features

To provide the services mentioned in the previous section, TCP has several features that are briefly summarized in this section and discussed later in detail.

- *Numbering System*
  - *Byte Number*
  - *Sequence Number*
  - *Acknowledgment Number*

3.4.3 TCP Segment

Before discussing TCP in more detail, let us discuss the TCP packets themselves. A packet in TCP is called a segment.

- *Format*
- *Encapsulation*

Figure 3.44: TCP segment format

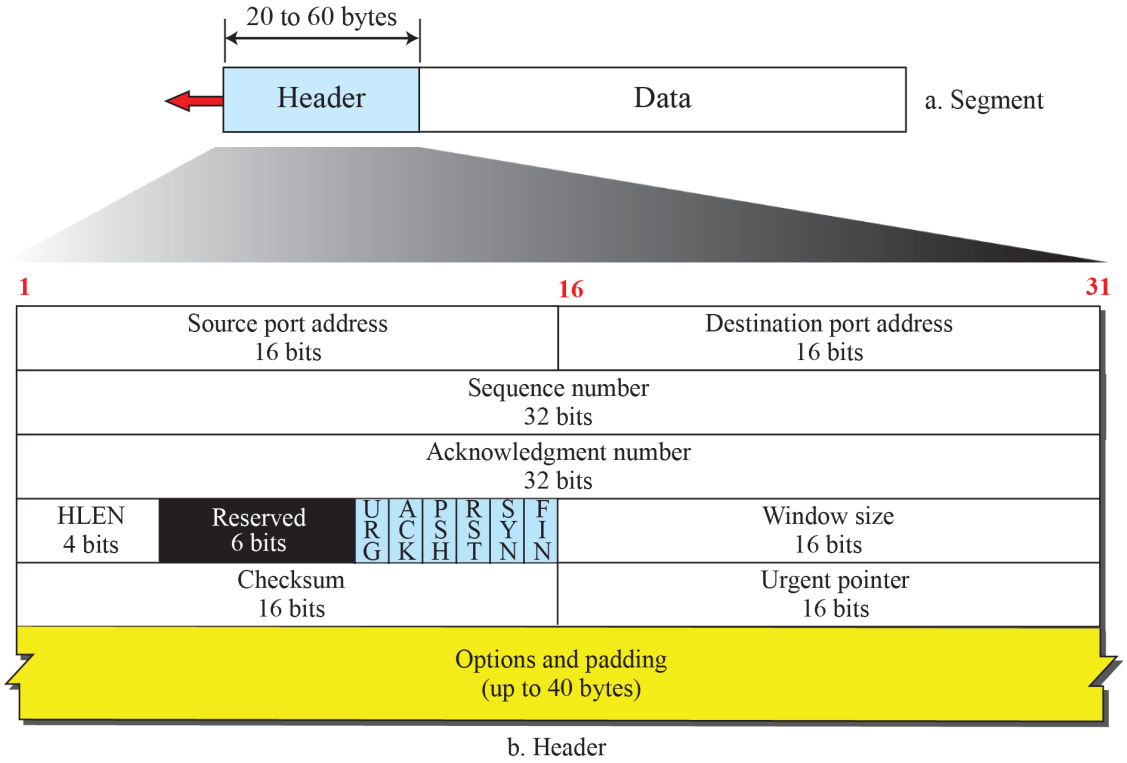
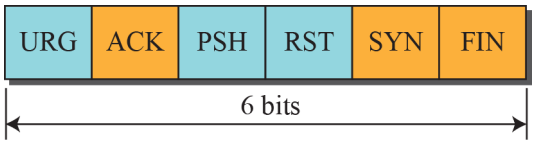


Figure 3.45: Control field



URG: Urgent pointer is valid
ACK: Acknowledgment is valid
PSH: Request for push
RST: Reset the connection
SYN: Synchronize sequence numbers
FIN: Terminate the connection

3.4.4 A TCP Connection

- *TCP is connection-oriented.*
- *Connection-oriented transport protocol establishes a logical path between the source and destination.*
- *All of the segments belonging to a message are then sent over this logical path.*
- *Using a single logical pathway for the entire message facilitates the acknowledgment process as well as retransmission of damaged or lost frames.*

3.4.4 (continued)

- **Connection Establishment**
  - Three-Way TCP Handshaking
  - SYN Flooding Attack
- **Data Transfer**
  - Pushing Data
  - Urgent Data
- **Connection Termination**
  - Three-Way Handshaking
  - Half-Close
- **Connection Reset**

Figure 3.47: Connection establishment using three-way handshaking

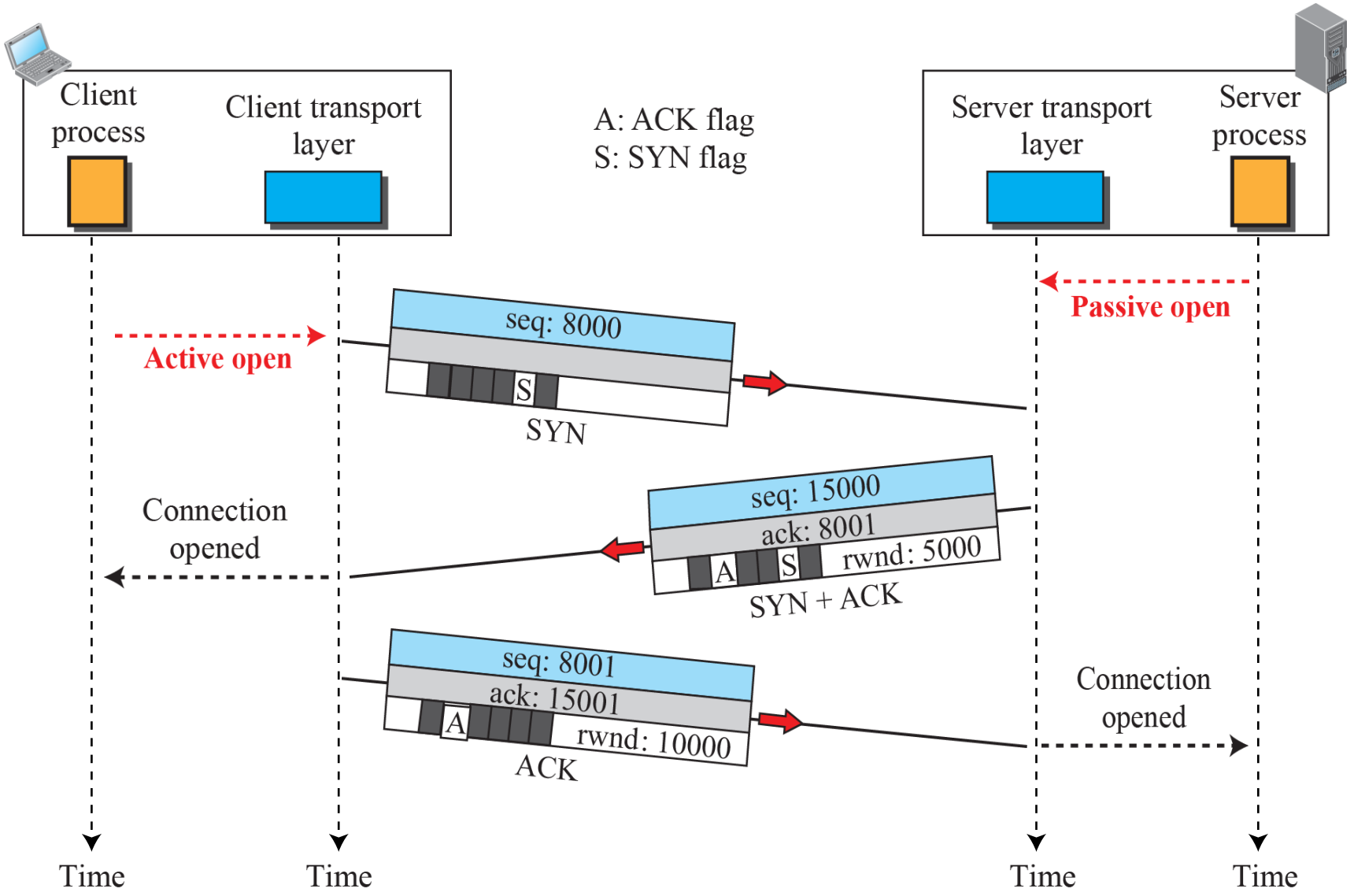


Figure 3.48: Data transfer

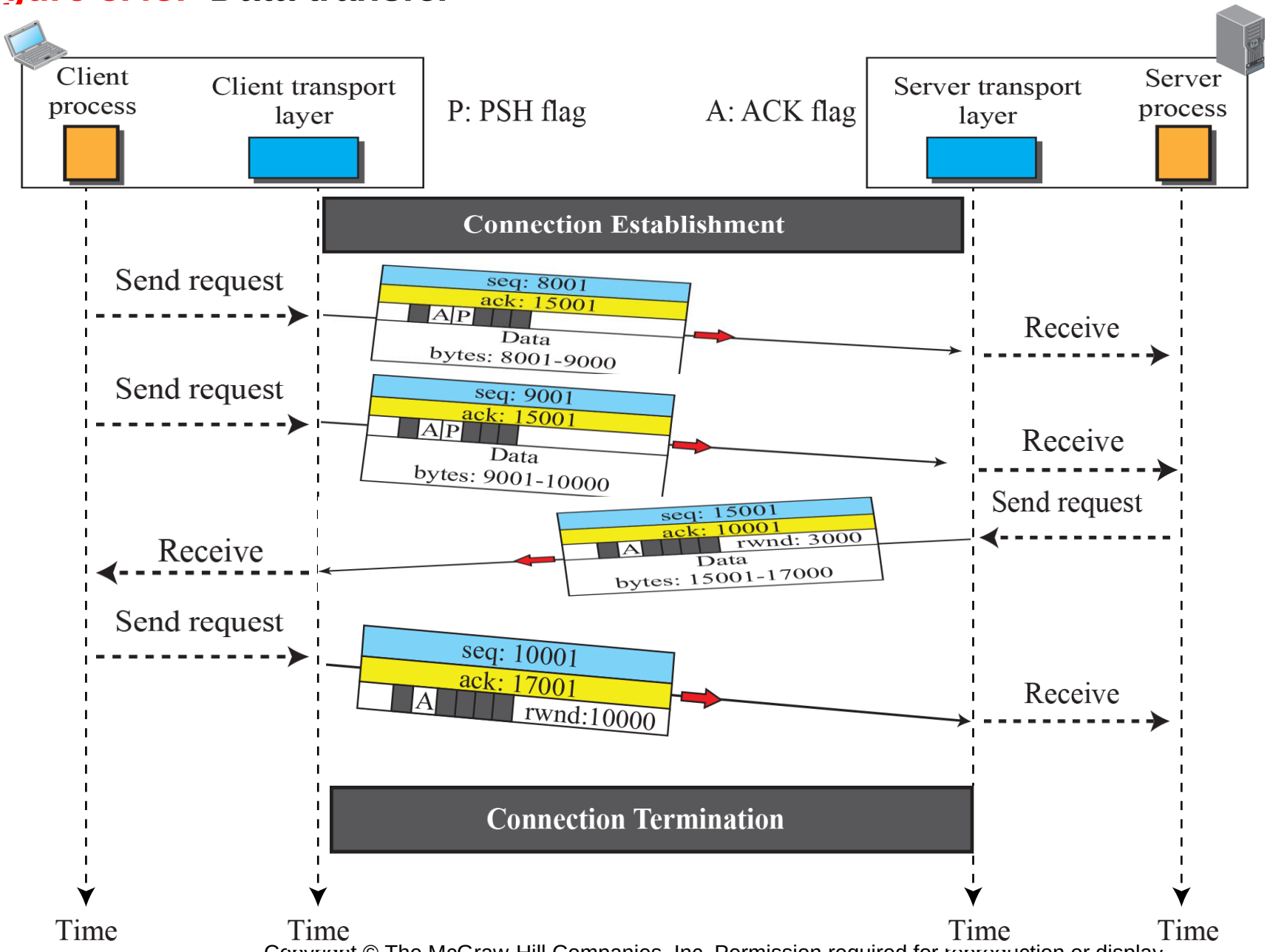


Figure 3.49: Connection termination using three-way handshaking

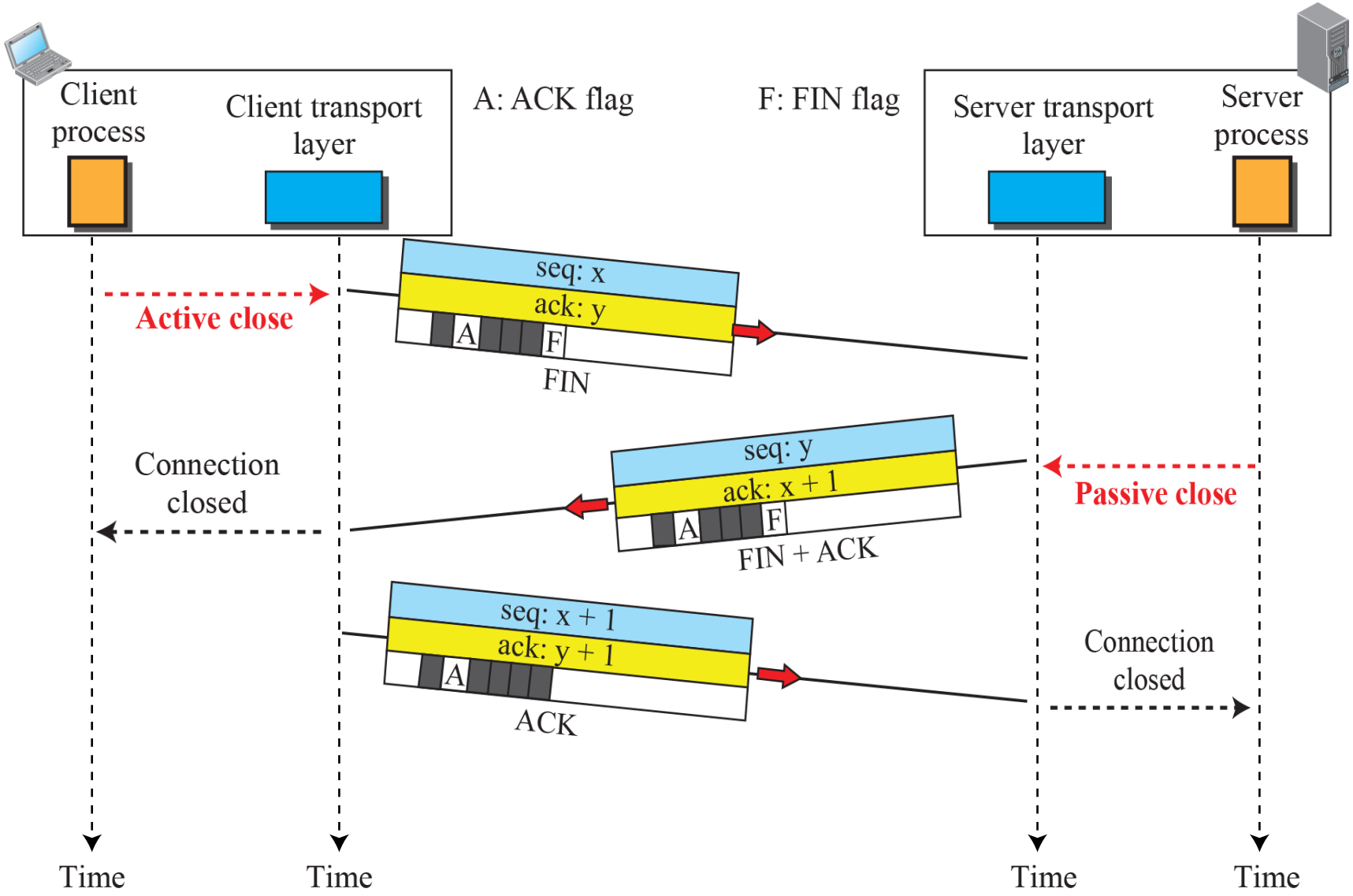


Figure 3.50: Half-close

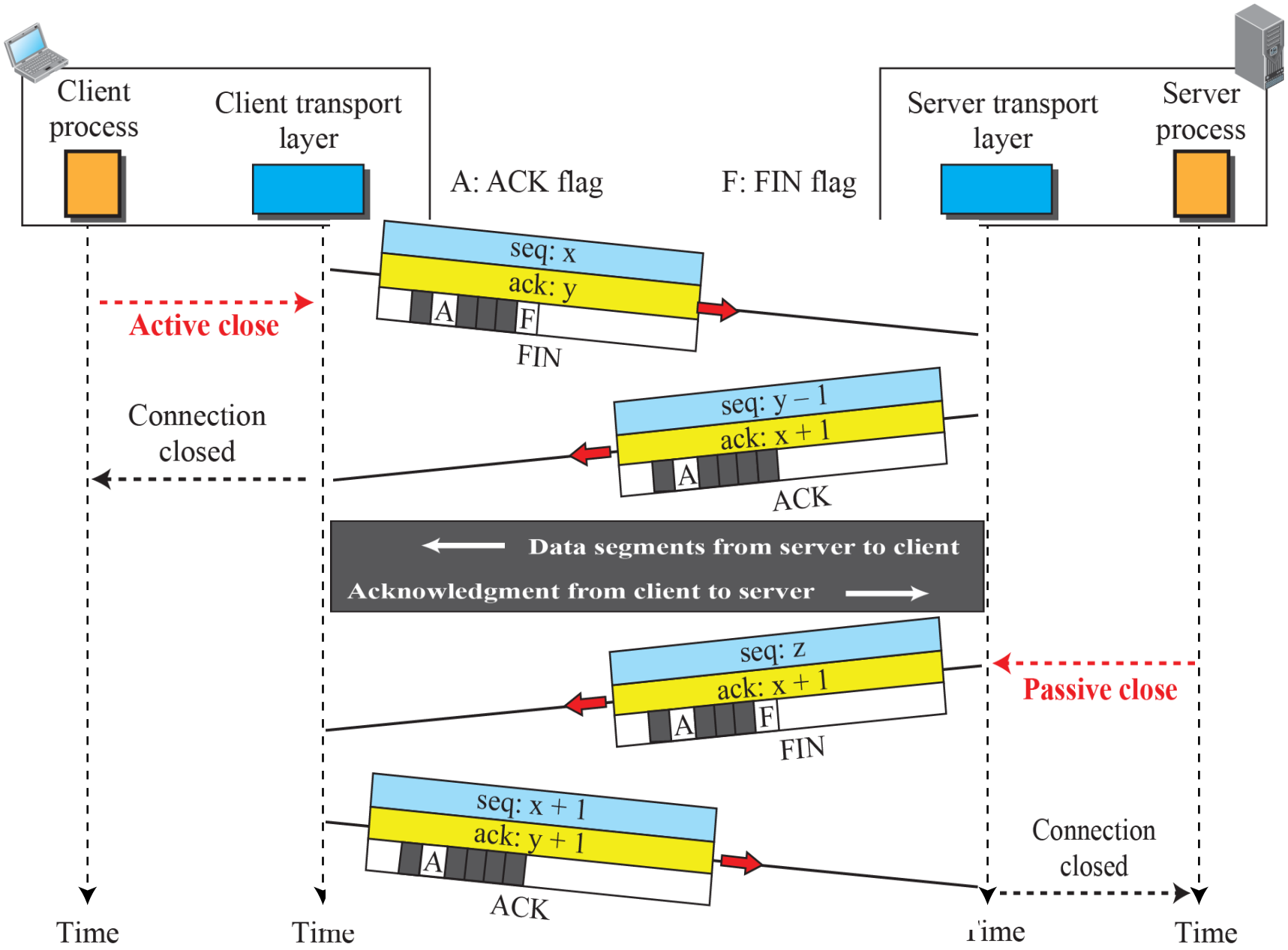
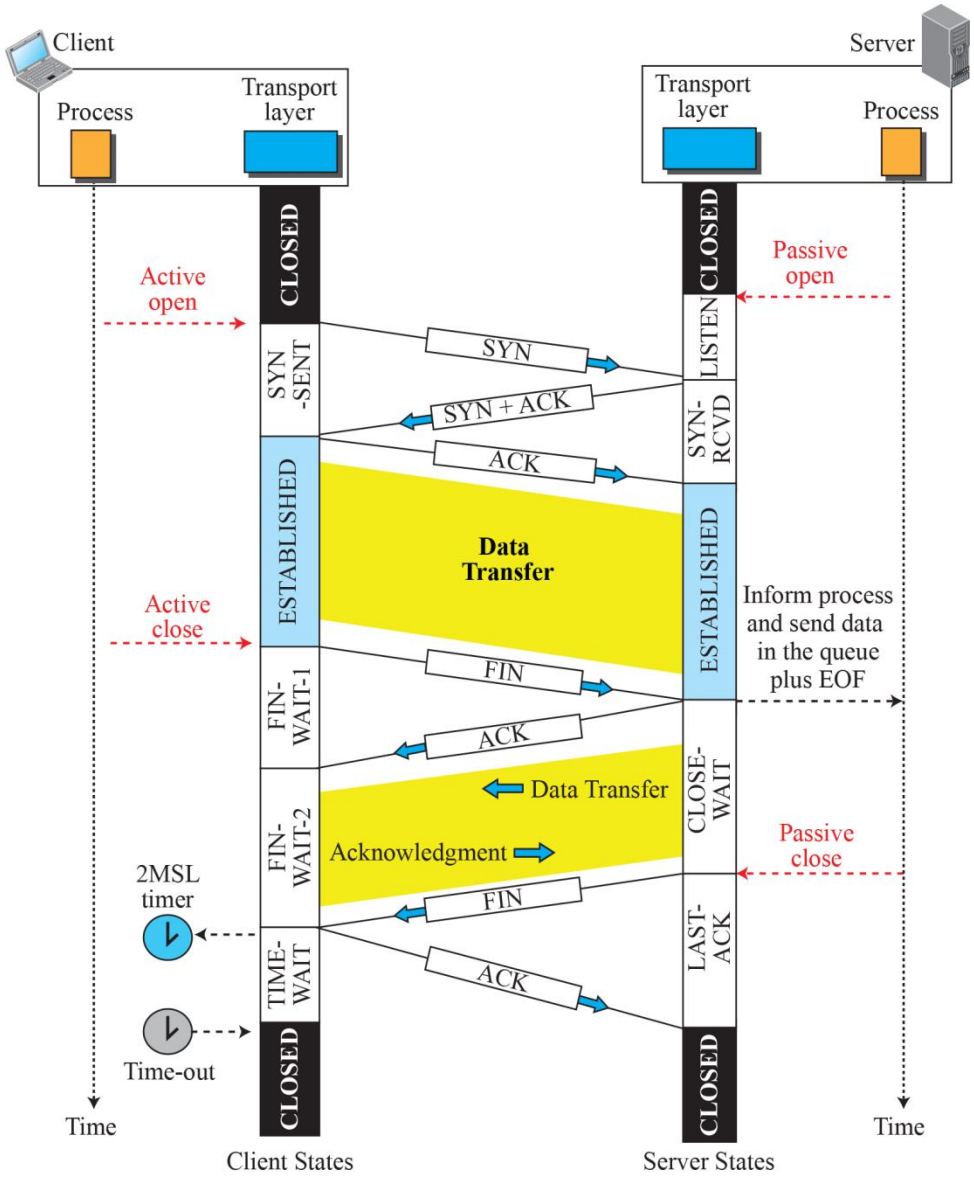


Figure 3.53: Time-line diagram for a common scenario



3.4.6 Windows in TCP

- *TCP uses two windows (send window and receive window) for each direction of data transfer, which means four windows for a bidirectional communication.*
- *To make the discussion simple, we make an unrealistic unidirectional; the bidirectional communication can be inferred using two unidirectional communications with piggybacking.*

- ❑ *Send Window*
- ❑ *Receive Window*

Figure 3.54: Send window in TCP

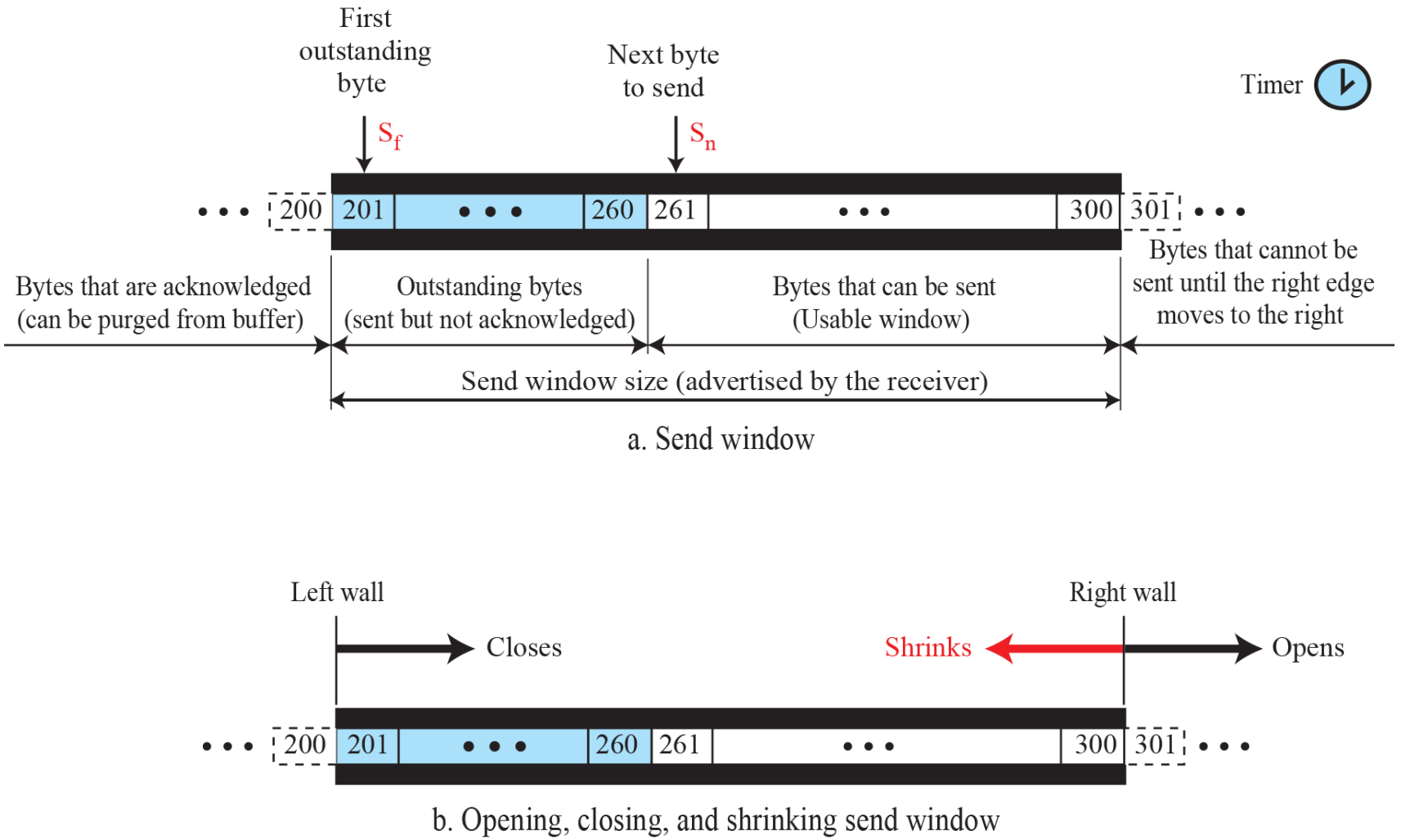
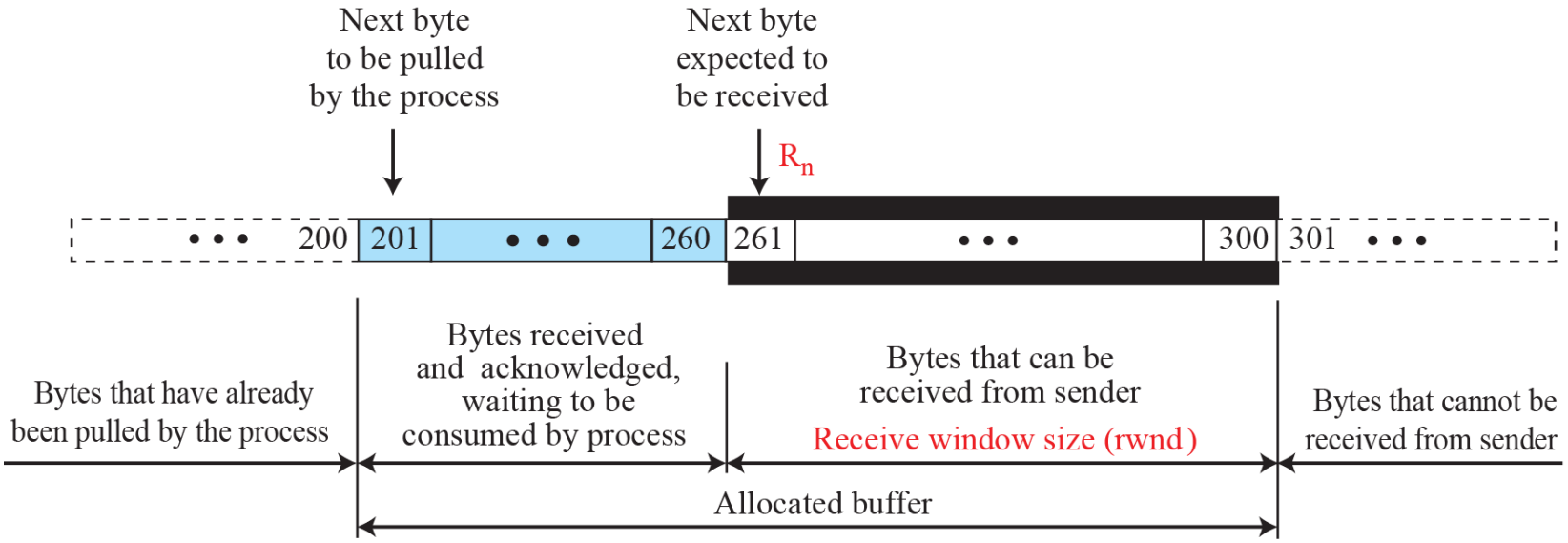
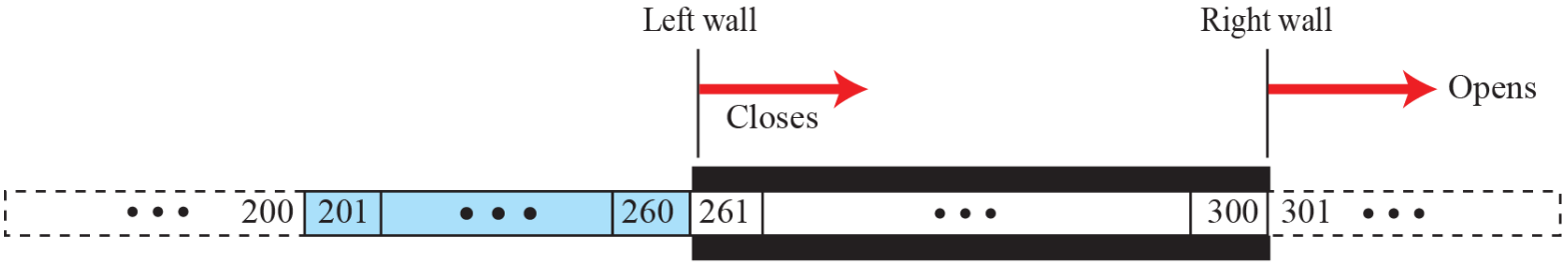


Figure 3.55: Receive window in TCP



a. Receive window and allocated buffer



b. Opening and closing of receive window

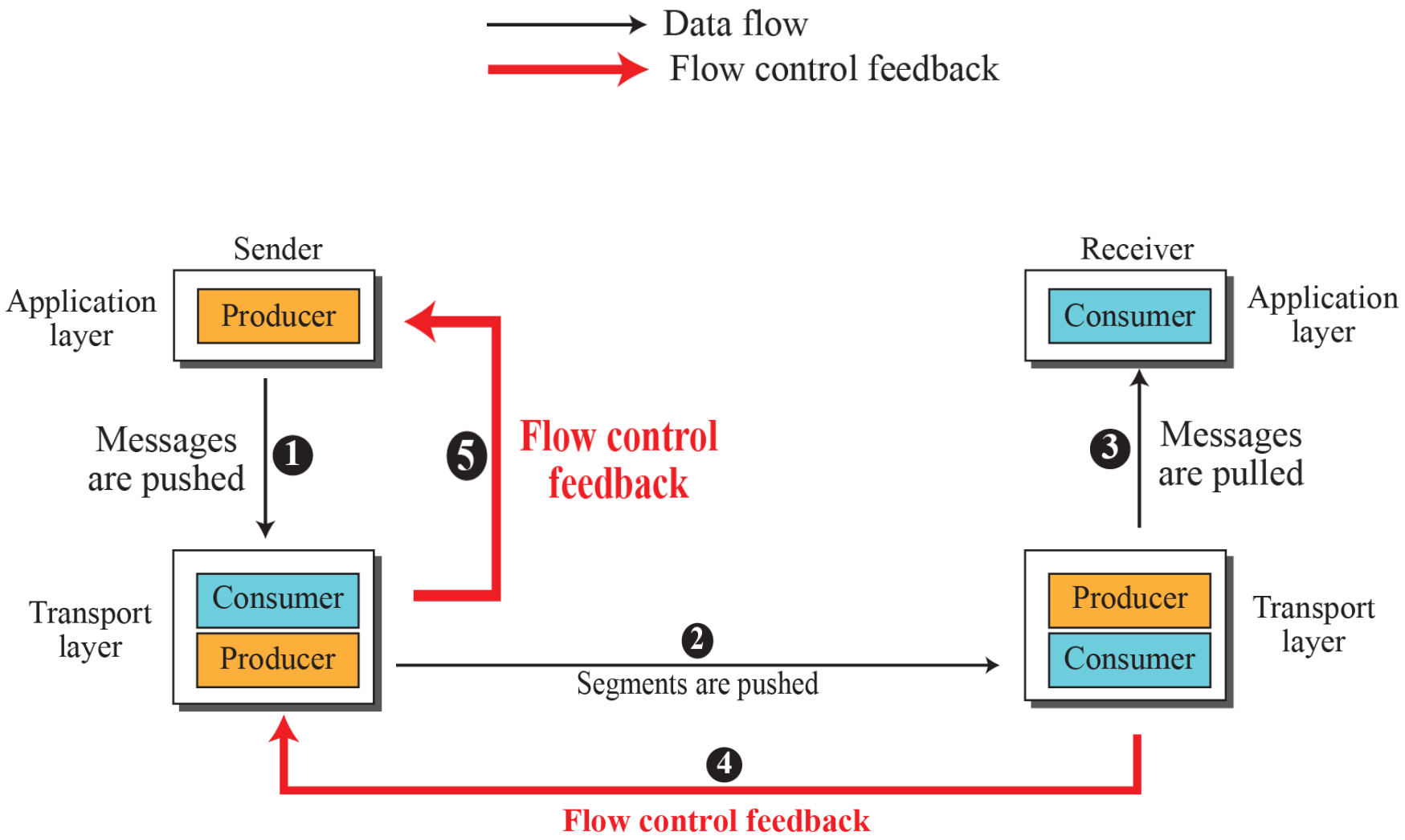
3.4.7 Flow Control

- *As discussed before, flow control balances the rate a producer creates data with the rate a consumer can use the data.*
- *TCP separates flow control from error control.*
- *In this section we discuss flow control, ignoring error control.*
- *We assume that the logical channel between the sending and receiving TCP is error-free.*

3.4.7 (continued)

- **Opening and Closing Windows**
  - A Scenario
- **Shrinking of Windows**
  - Window Shutdown
- **Silly Window Syndrome**
  - Syndrome Created by the Sender
  - Syndrome Created by the Receiver

Figure 3.56: Data flow and flow control feedbacks in TCP



3.4.8 Error Control

- *TCP is a reliable transport-layer protocol.*
- *This means that an application program that delivers a stream of data to TCP relies on TCP to deliver the entire stream to the application program on the other end in order, without error, and without any part lost or duplicated.*

3.4.8 (continued)

- *Checksum*
- *Acknowledgment*
  - *Cumulative Acknowledgment (ACK)*
  - *Selective Acknowledgment (SACK)*
- *Generating Acknowledgments*
- *Retransmission*
  - *Retransmission after RTO*
  - *Retransmission after Three Duplicate ACK*
- *Out-of-Order Segments*

3.4.8 (continued)

- ***Some Scenarios***
  - ***Normal Operation***
  - ***Lost Segment***
  - ***Fast Retransmission***
  - ***Delayed Segment***
  - ***Duplicate Segment***
  - ***Automatically Corrected Lost ACK***
  - ***Correction by Resending a Segment***
  - ***Deadlock Created by Lost Acknowledgment***

Figure 3.61: Normal operation

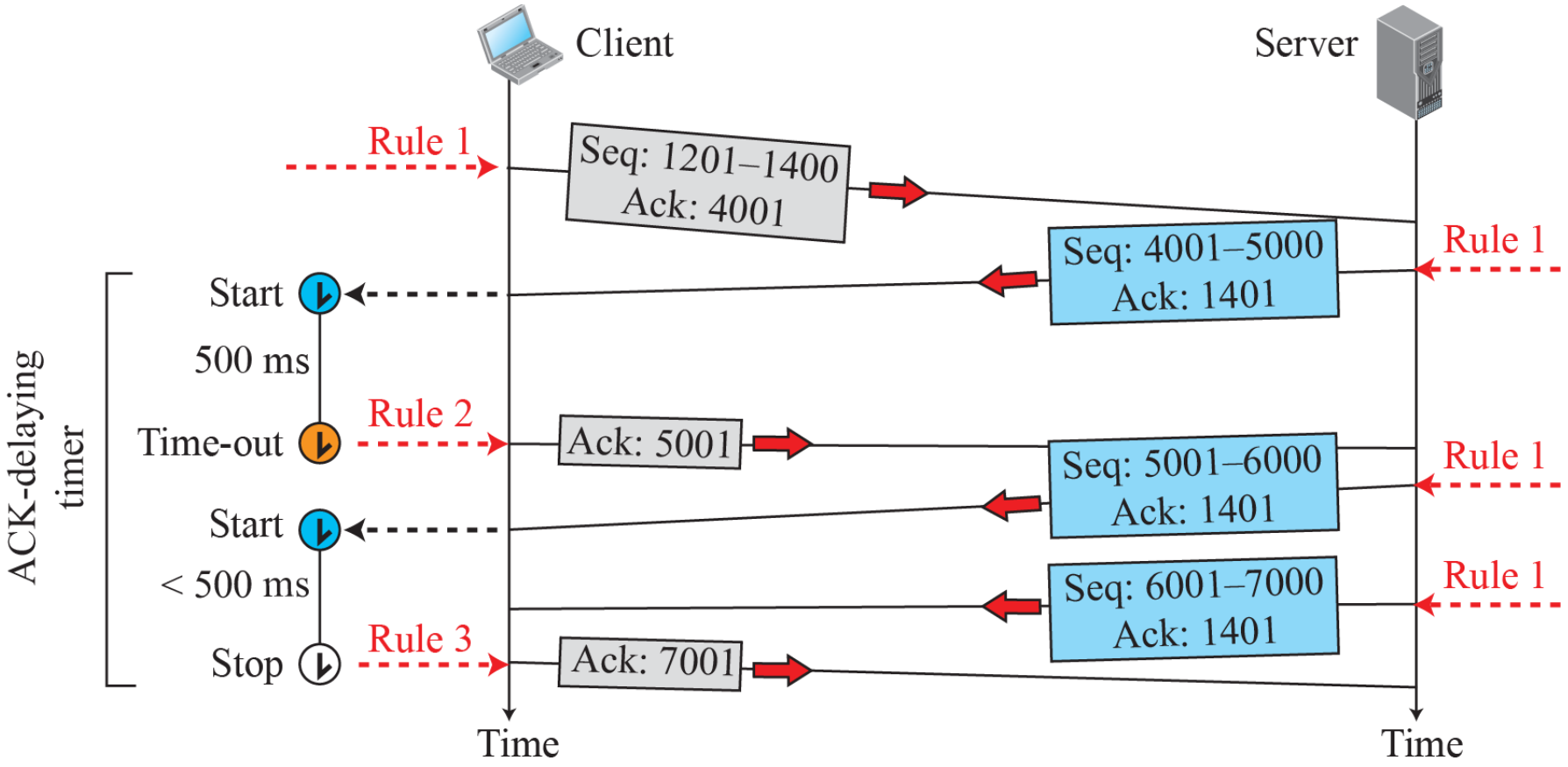


Figure 3.62: Lost segment

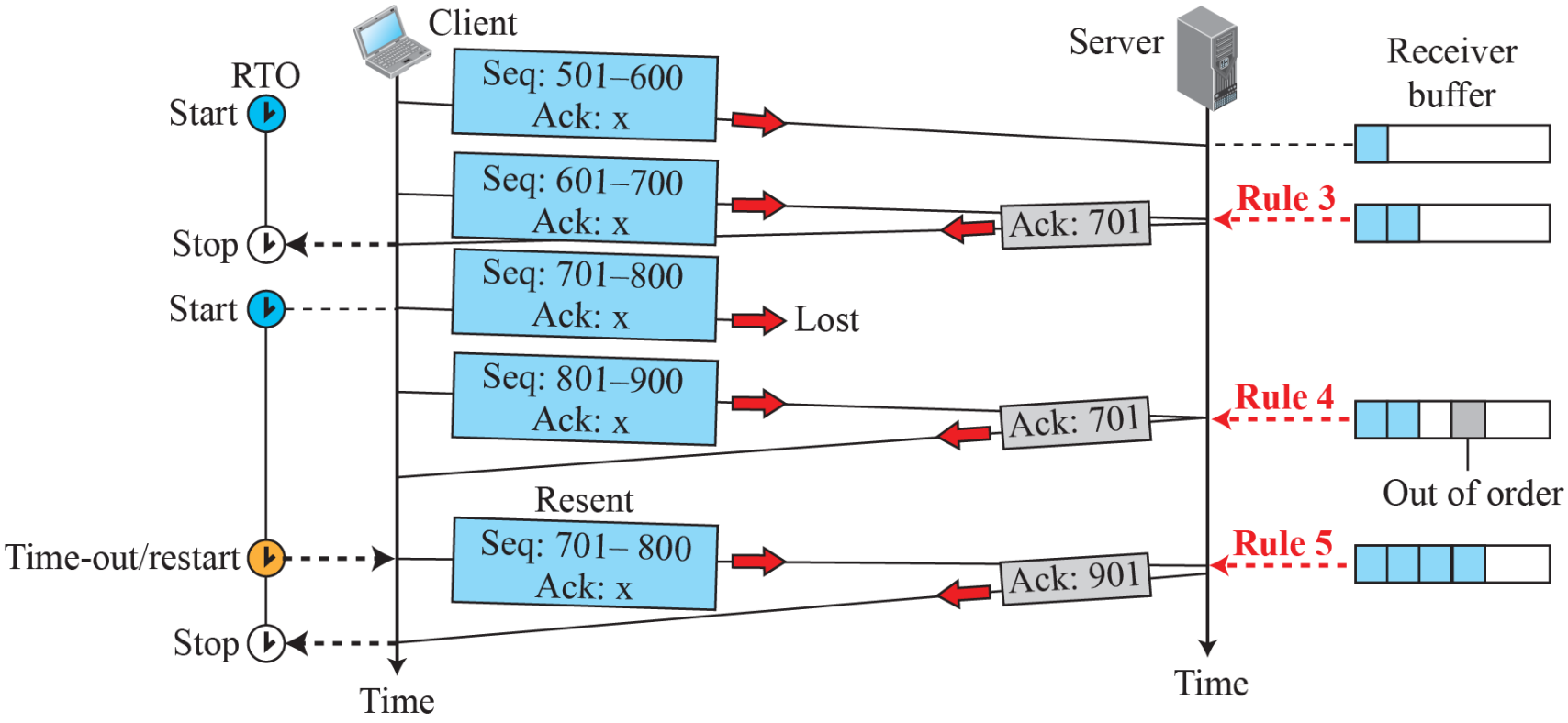


Figure 3.63: Fast retransmission

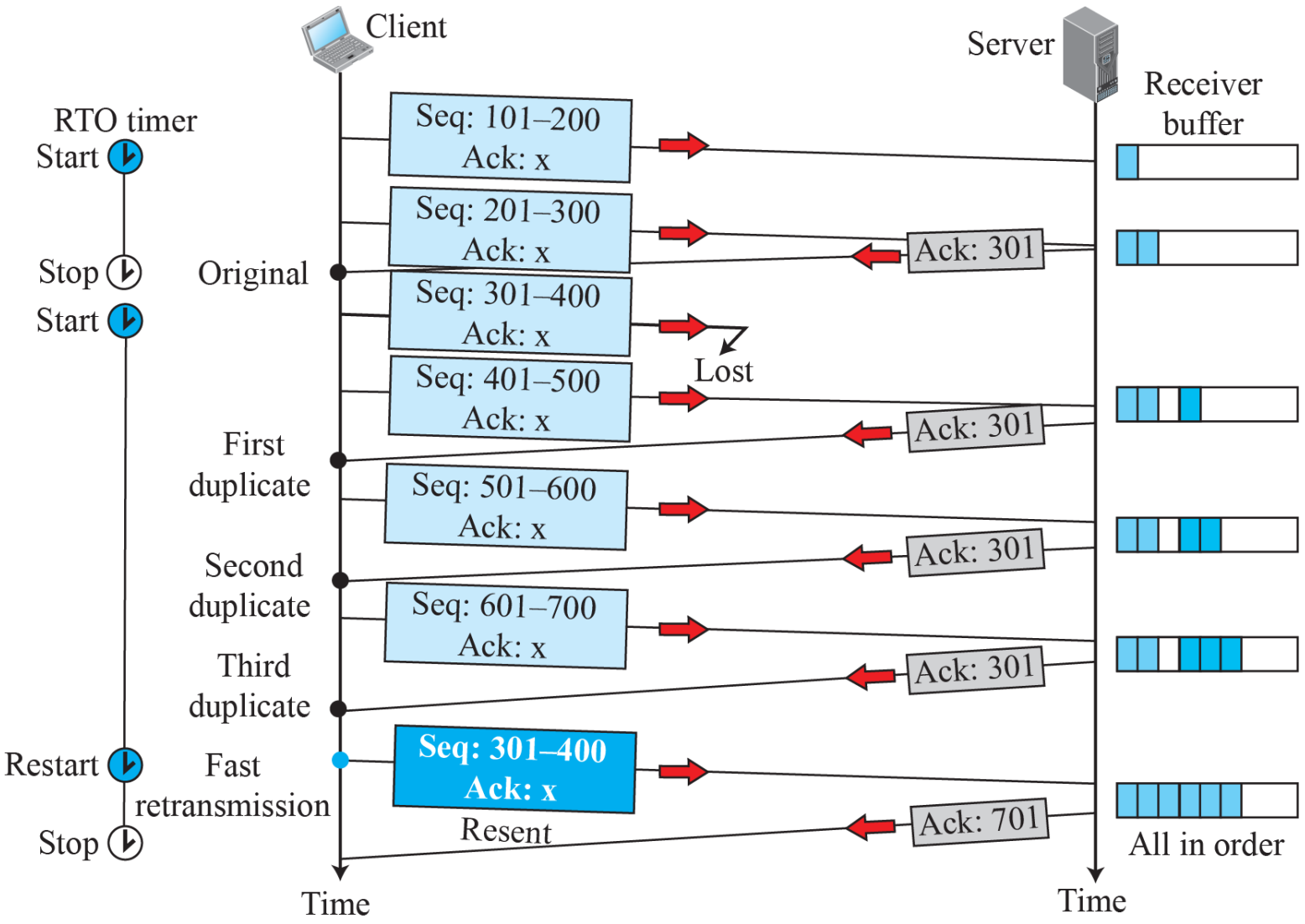


Figure 3.64: Lost acknowledgment

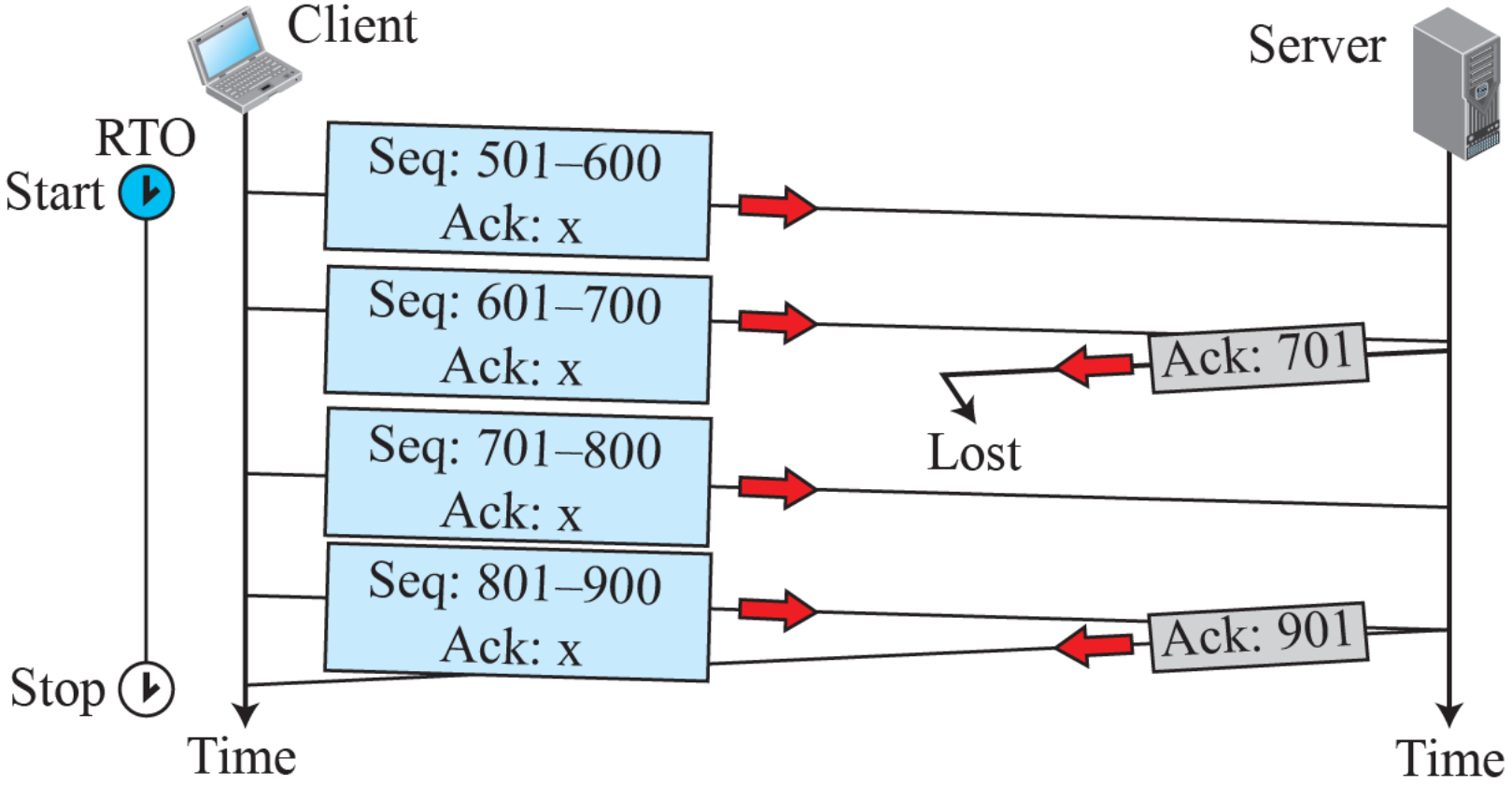
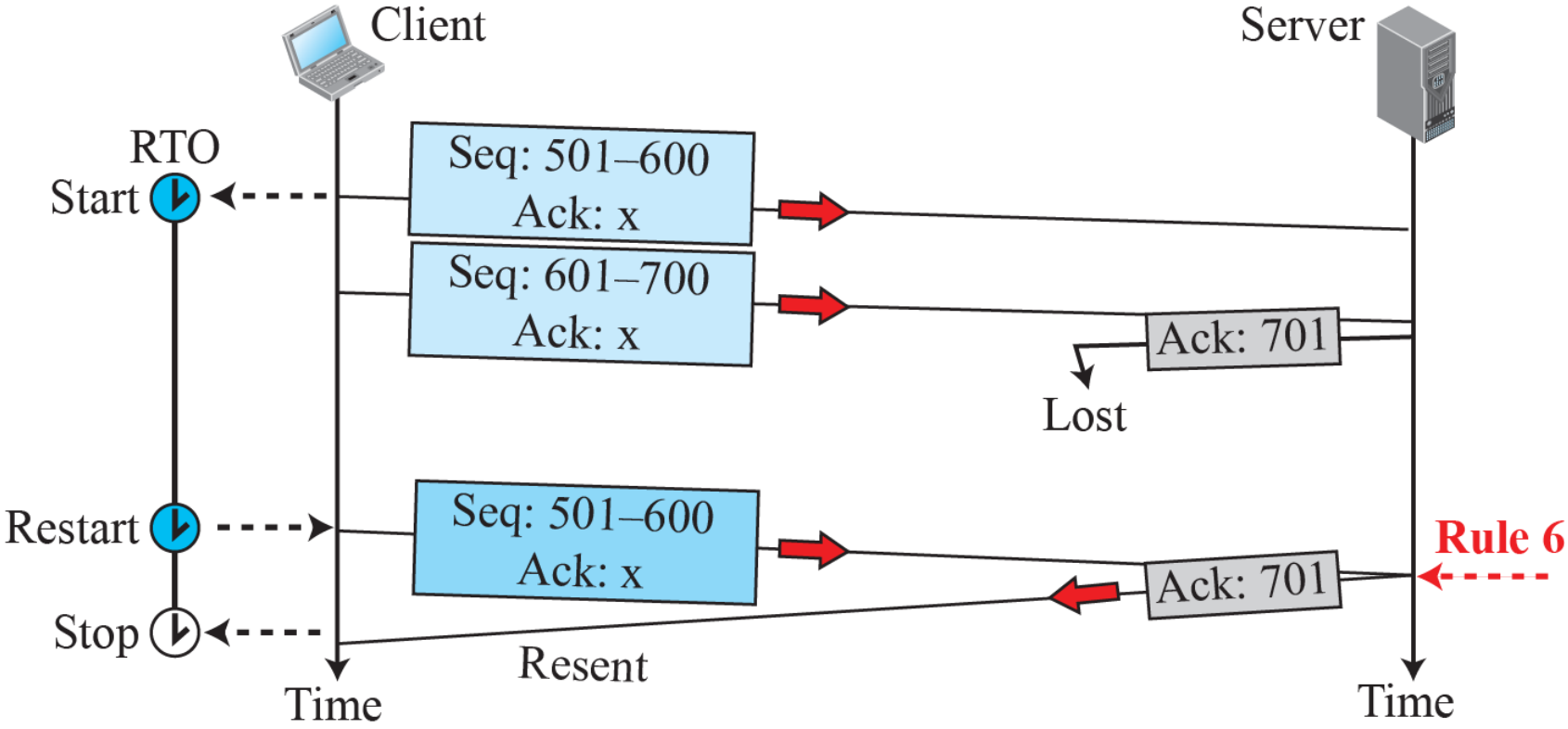


Figure 3.65: *Lost acknowledgment corrected by resending a segment*



3.4.9 TCP Congestion Control

- *TCP uses different policies to handle the congestion in the network.*
- *We describe these policies in this section.*

  - *Congestion Window*
  - *Congestion Detection*
  - *Congestion Policies*

    - *Slow Start: Exponential Increase*
    - *Congestion Avoidance: Additive Increase*

  - *Policy Transition*

    - *Taho TCP*
    - *Reno TCP*
    - *NewReno TCP*

  - *Additive Increase, Multiplicative Decrease*
  - *TCP Throughput*

Figure 3.66: Slow start, exponential increase

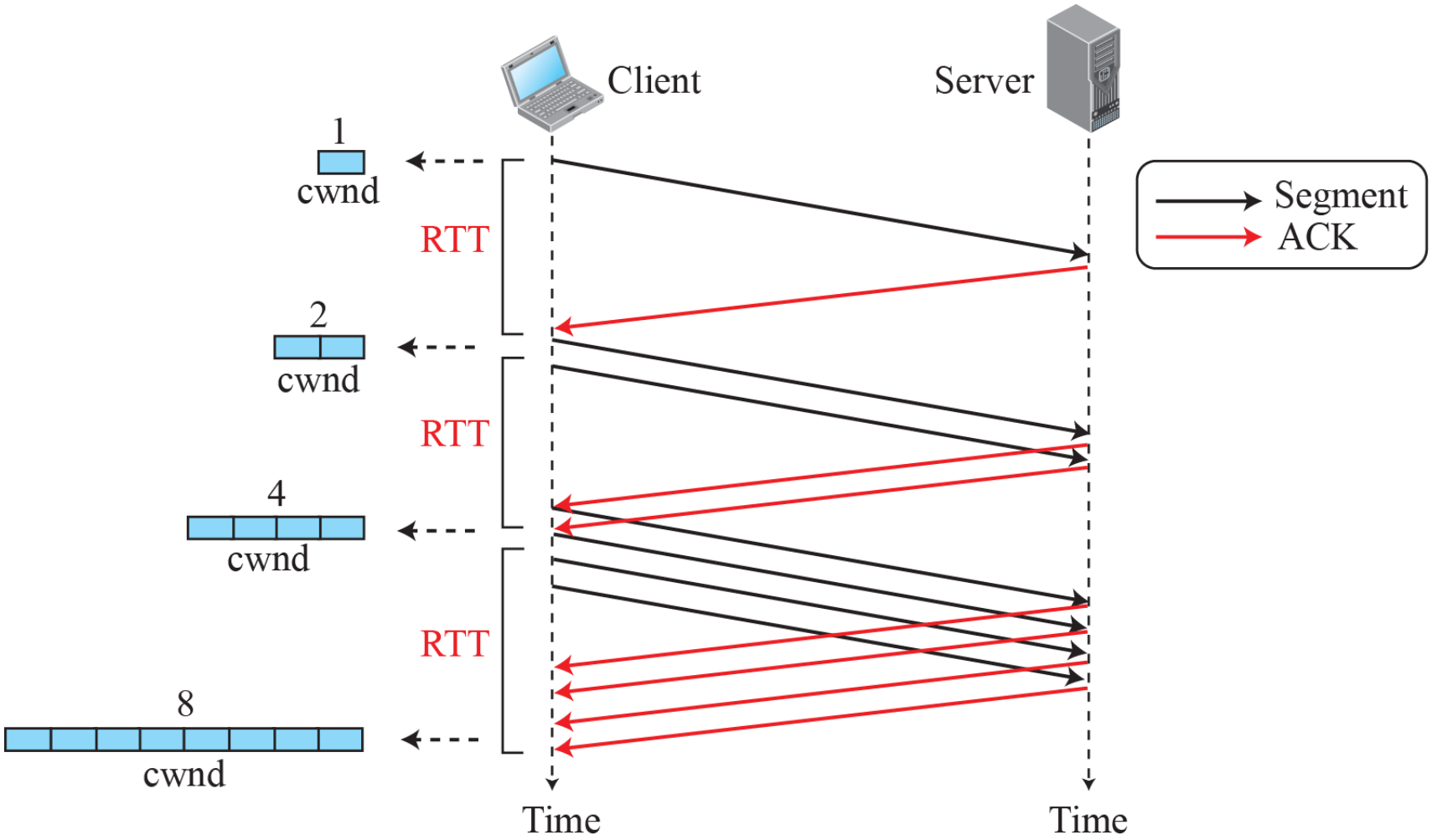


Figure 3.67: Congestion avoidance, additive increase

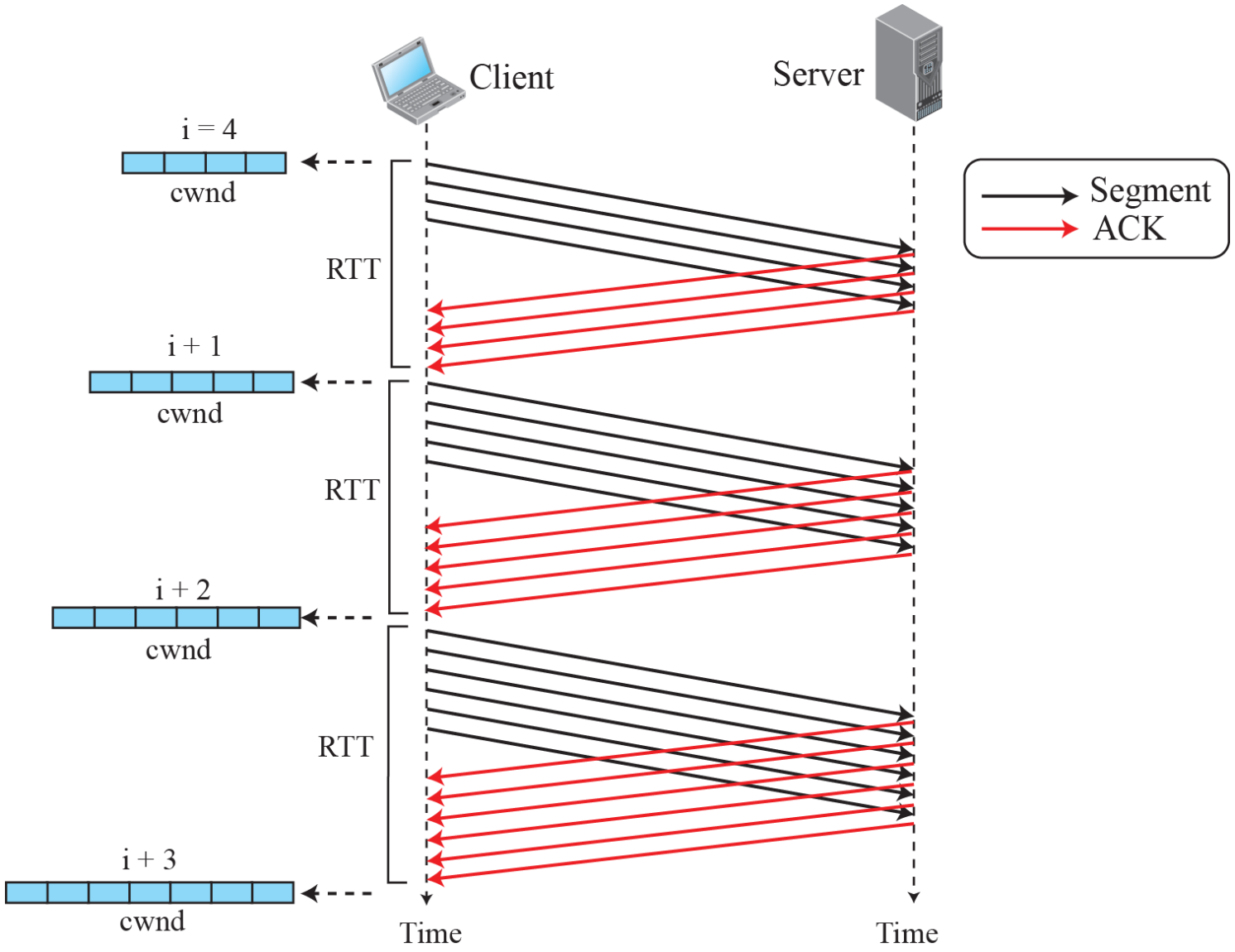


Figure 3.69: Example of Taho TCP

1. *A fast retransmit algorithm is used when duplicate ACKs are detected as packet loss.*
2. *TCP Tahoe is the simplest one out of all variants. It doesn't have fast recovery. At congestion avoidance phase, it treats the triple(3) duplicate ACKs same as timeout. When timeout or triple duplicate ACKs is received, it will perform fast retransmit, reduce congestion window to 1, and enters slow-start phase.*

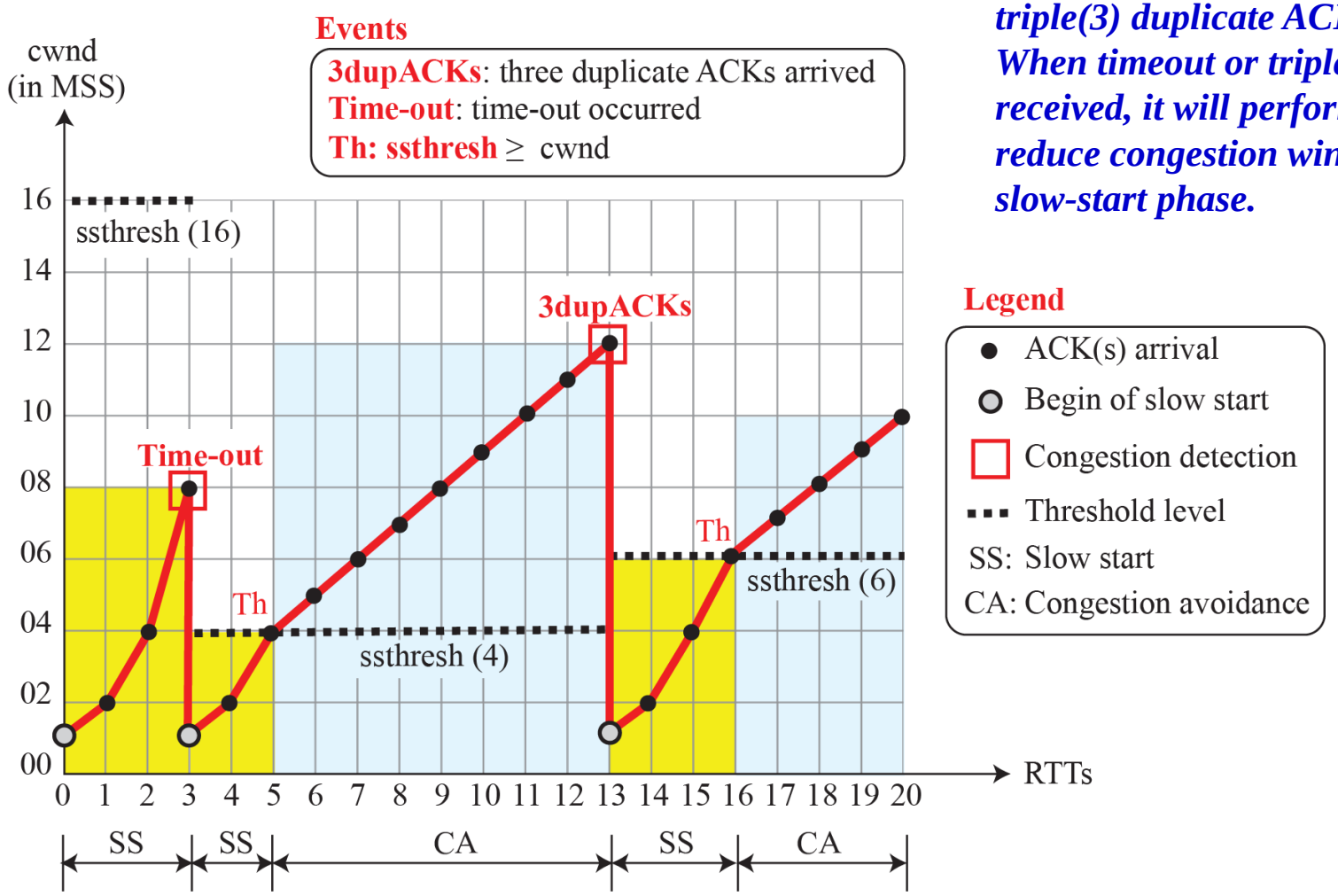


Figure 3.71: Example of a Reno TCP

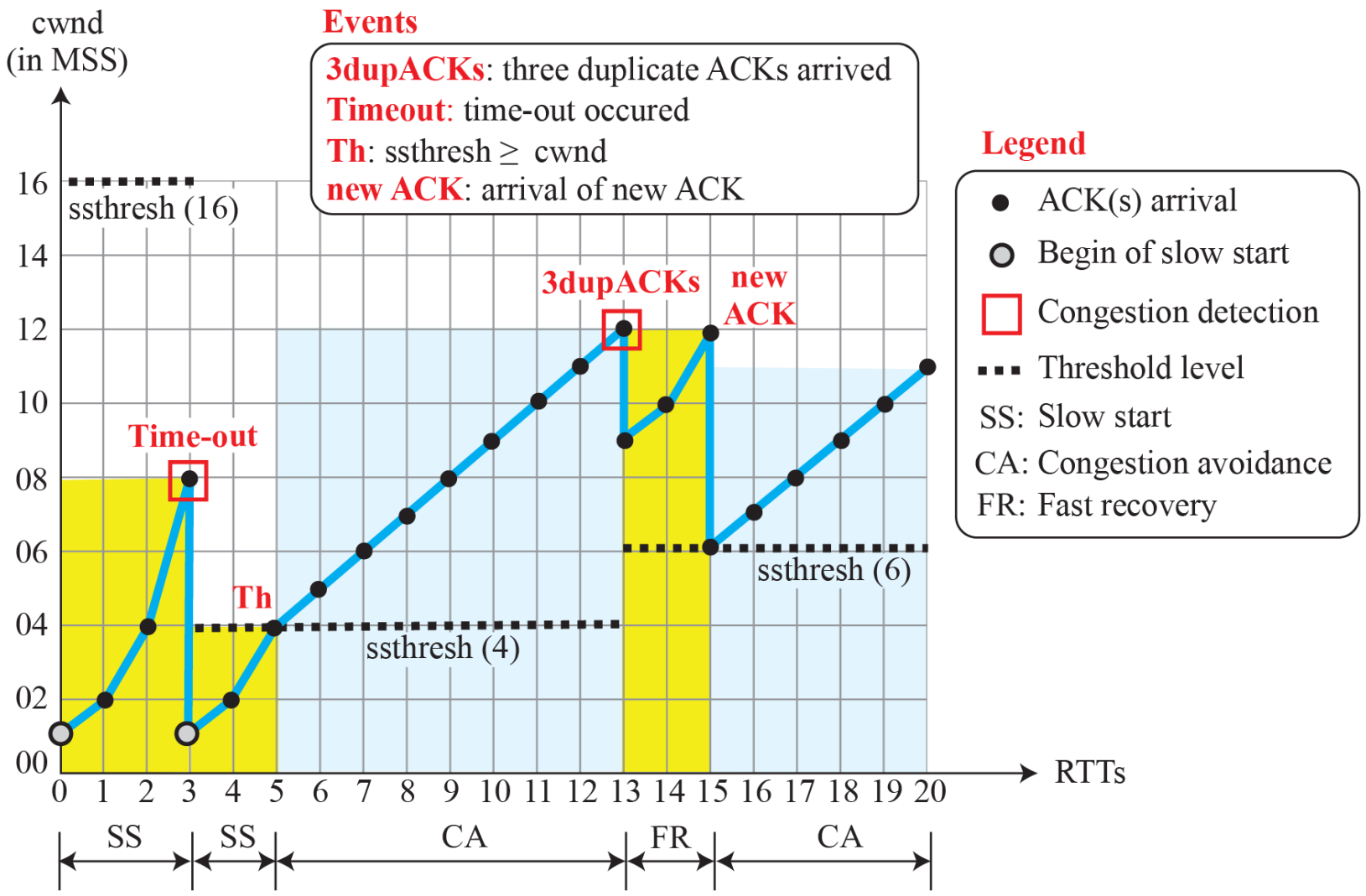
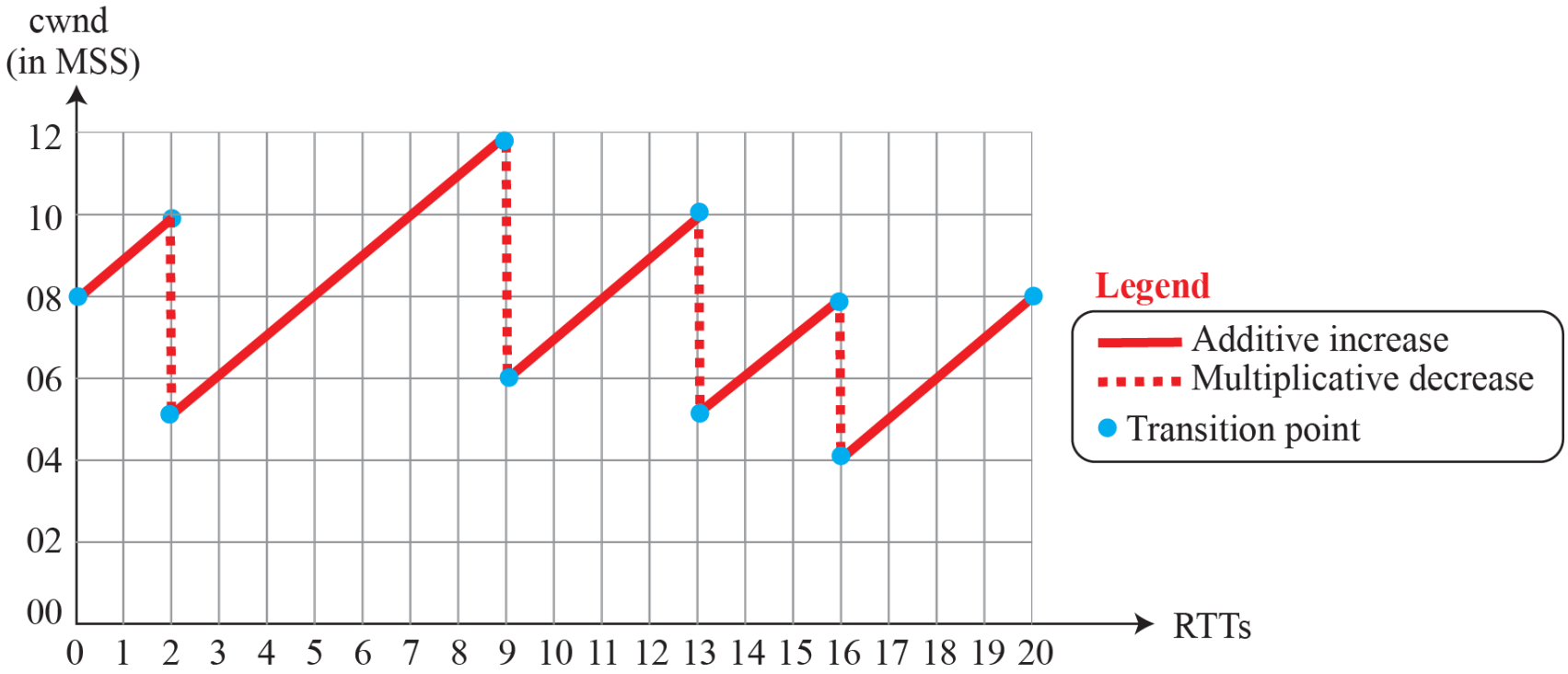


Figure 3.72: Additive increase, multiplicative decrease (AIMD)



Chapter 3: Summary

- ***The main duty of a transport-layer protocol is to provide process-to-process communication. To define the processes, we need port numbers. The client program defines itself with an ephemeral port number. The server defines itself with a well-known port number. To send a message from one process to another, the transport-layer protocol encapsulates and decapsulates messages. Flow control balances the exchange of data items between a producer and a consumer. A transport-layer protocol can provide two types of services: connectionless and connection-oriented. In a connectionless service, the sender sends packets to the receiver without any connection establishment. In a connection-oriented service, the client and the server first need to establish a connection between themselves.***

Chapter 3: Summary (continued)

- *We have discussed several common transport-layer protocols in this chapter. The Stop-and-Wait protocol provides both flow and error control, but is inefficient. The Go-Back-N protocol is the more efficient version of the Stop-and-Wait protocol and takes advantage of pipelining. The Selective-Repeat protocol, a modification of the Go-Back-N protocol, is better suited to handle packet loss. All of these protocols can be implemented bidirectionally using piggybacking.*

- *UDP is a transport protocol that creates a process-to-process communication. UDP is a (mostly) unreliable and connectionless protocol that requires little overhead and offers fast delivery. The UDP packet is called a user datagram.*

Chapter 3: Summary (continued)

- ***Transmission Control Protocol (TCP) is another transport-layer protocol in the TCP/IP protocol suite. TCP provides process-to-process, full-duplex, and connection-oriented service. The unit of data transfer between two devices using TCP software is called a segment. A TCP connection consists of three phases: connection establishment, data transfer, and connection termination.***