CS 540
Computer Networks II

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9. TRANSPORT LAYER – TCP/UDP

Topics

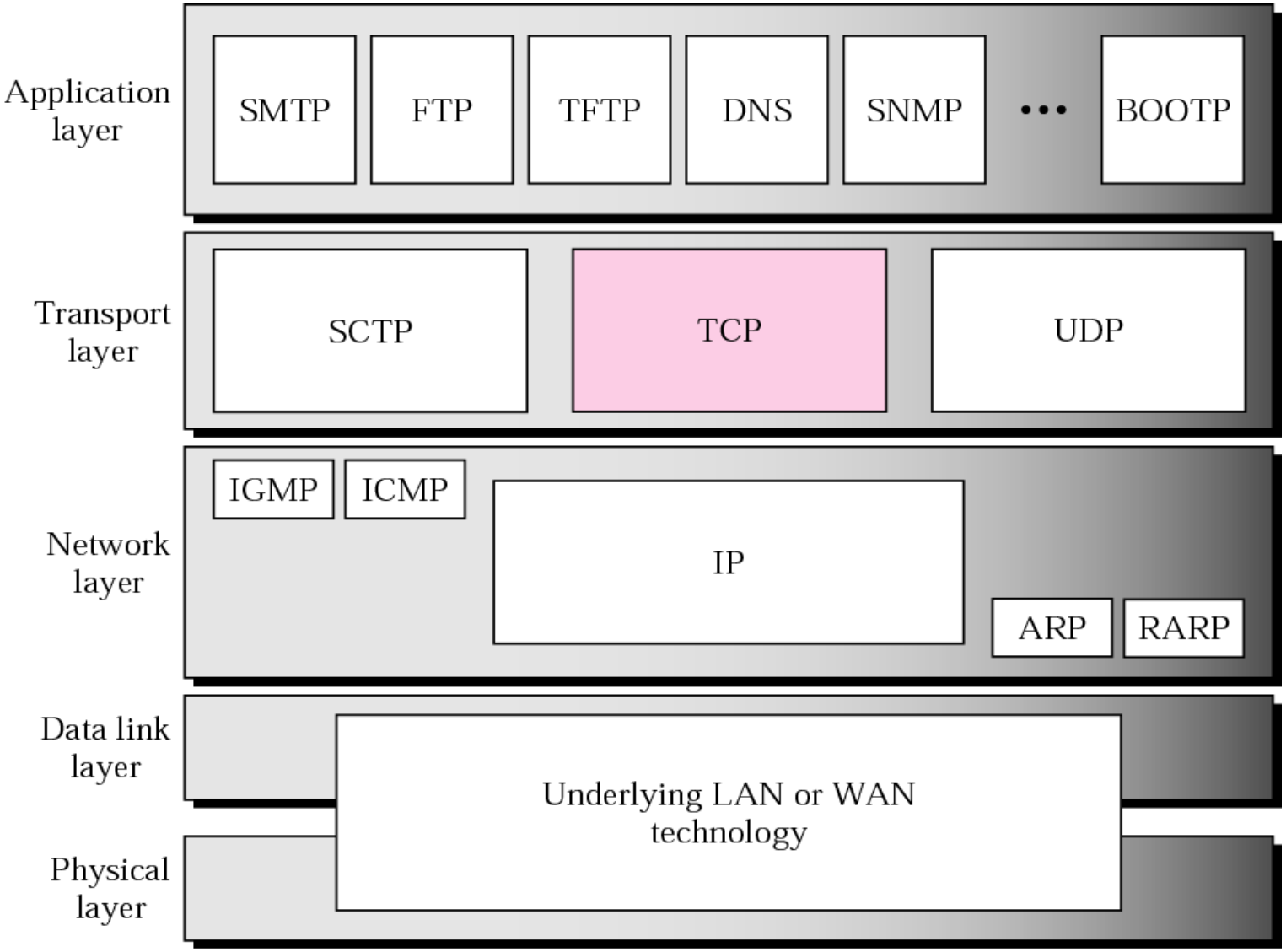
1. Overview
2. LAN Switching
3. IPv4
4. IPv6
5. Routing Protocols -- RIP, RIPng, OSPF
6. Routing Protocols -- ISIS, BGP
7. MPLS
8. Midterm Exam
9. **Transport Layer -- TCP/UDP**
10. Access Control List (ACL)
11. Congestion Control & Quality of Service (QoS)
12. Application Layer Protocols
13. Application Layer Protocols continue
14. Others – Multicast, SDN
15. Final Exam

Reference Books

- **Cisco CCNA Routing and Switching ICND2 200-101 Official Cert Guide, Academic Edition** by Wendel Odom -- July 10, 2013**.** ISBN-13: 978-1587144882
- **The TCP/IP Guide: A Comprehensive, Illustrated Internet Protocols Reference** by Charles M. Kozierok – October 1, 2005. ISBN-13: 978-1593270476
- **Data and Computer Communications (10th Edition) (William Stallings Books on Computer and Data Communications)** by Williams Stallings – September 23, 2013. ISBN-13: 978-0133506488

http://class.svuca.edu/~sandy/class/CS540/

TCP/IP protocol suite



0 → 1024 : wellKnown.

Table 12.1 Well-known ports used by TCP

Port	*Protocol*	*Description*
7	Echo	Echoes a received datagram back to the sender
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	FTP, Data	File Transfer Protocol (data connection)
21	FTP, Control	File Transfer Protocol (control connection)
23	TELNET	Terminal Network
25	SMTP	Simple Mail Transfer Protocol
53	DNS	Domain Name Server
67	BOOTP	Bootstrap Protocol
79	Finger	Finger
80	HTTP	Hypertext Transfer Protocol
111	RPC	Remote Procedure Call

TCP Features

- Numbering System
- Flow Control ← keep the sender / receiver speed match.
- Error Control ← takes care packet loss or order
- Congestion Control ← make sure don't send so much data that the network can't take care of.

The bytes of data being transferred in each connection are numbered by TCP. The numbering starts with a randomly generated number.

EXAMPLE

Suppose a TCP connection is transferring a file of 5000 bytes. The first byte is numbered 10001. What are the sequence numbers for each segment if data is sent in five segments, each carrying 1000 bytes?

Solution

The following shows the sequence number for each segment:

Segment 1 ➞ ***Sequence Number: 10,001 (range: 10,001 to 11,000)***

Segment 2 ➞ ***Sequence Number: 11,001 (range: 11,001 to 12,000)***

Segment 3 ➞ ***Sequence Number: 12,001 (range: 12,001 to 13,000)***

Segment 4 ➞ ***Sequence Number: 13,001 (range: 13,001 to 14,000)***

Segment 5 ➞ ***Sequence Number: 14,001 (range: 14,001 to 15,000)***

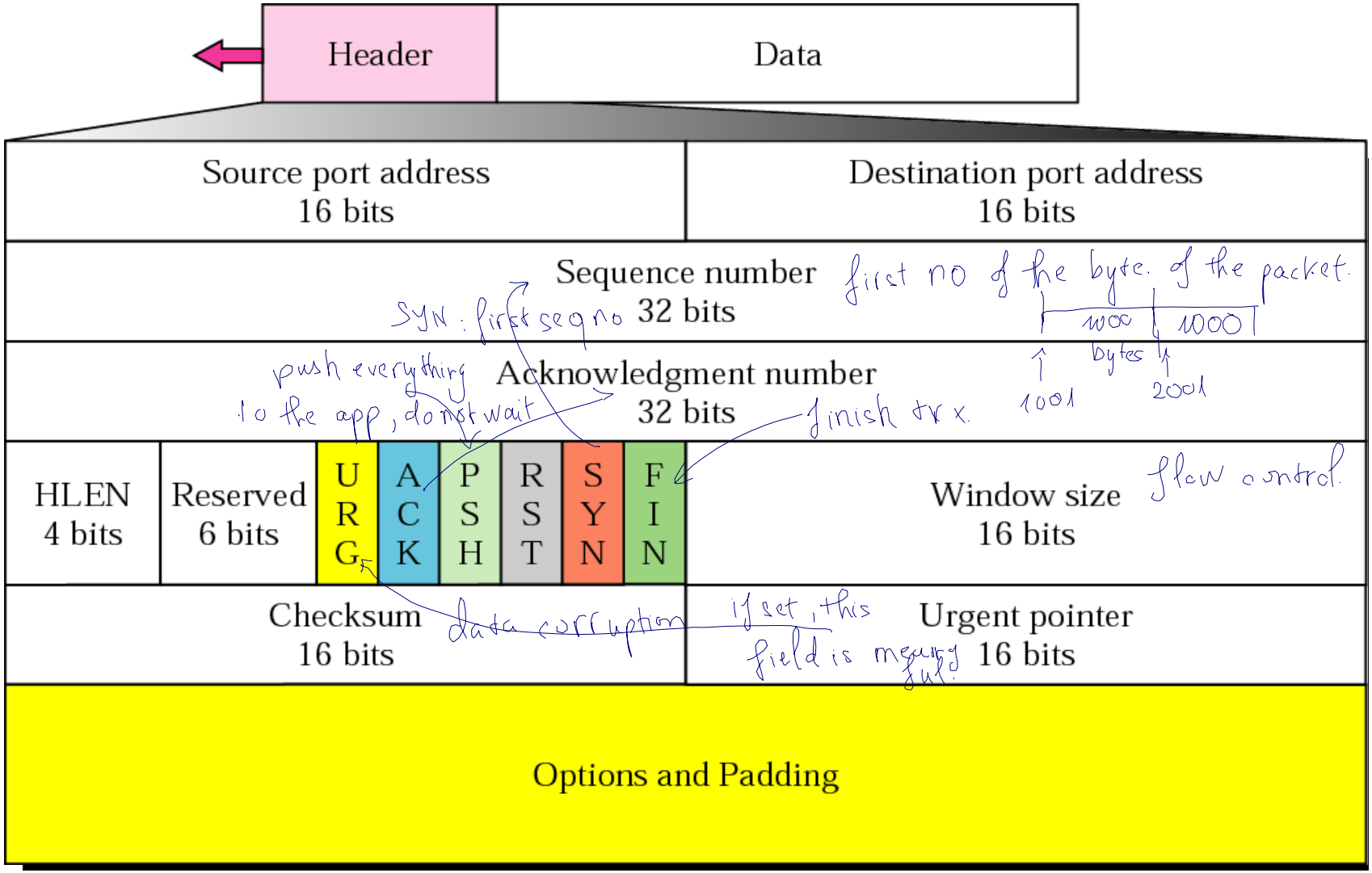
The value in the sequence number field of a segment defines the number of the first data byte contained in that segment.

The value of the acknowledgment field in a segment defines the number of the next byte a party expects to receive.

The acknowledgment number is cumulative.

TCP segment format

header length min = 20 bytes.
HLEN min = 5



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

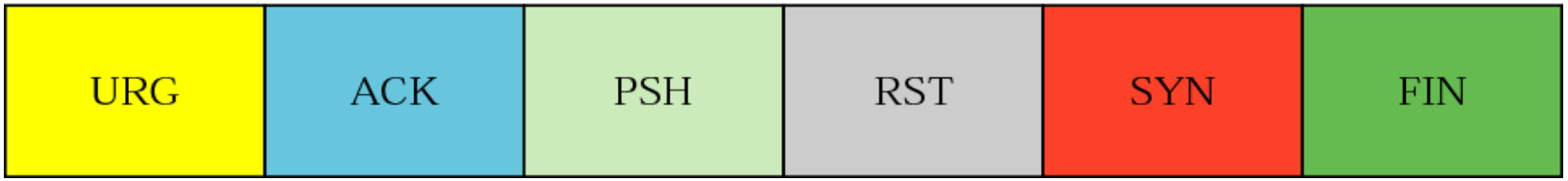
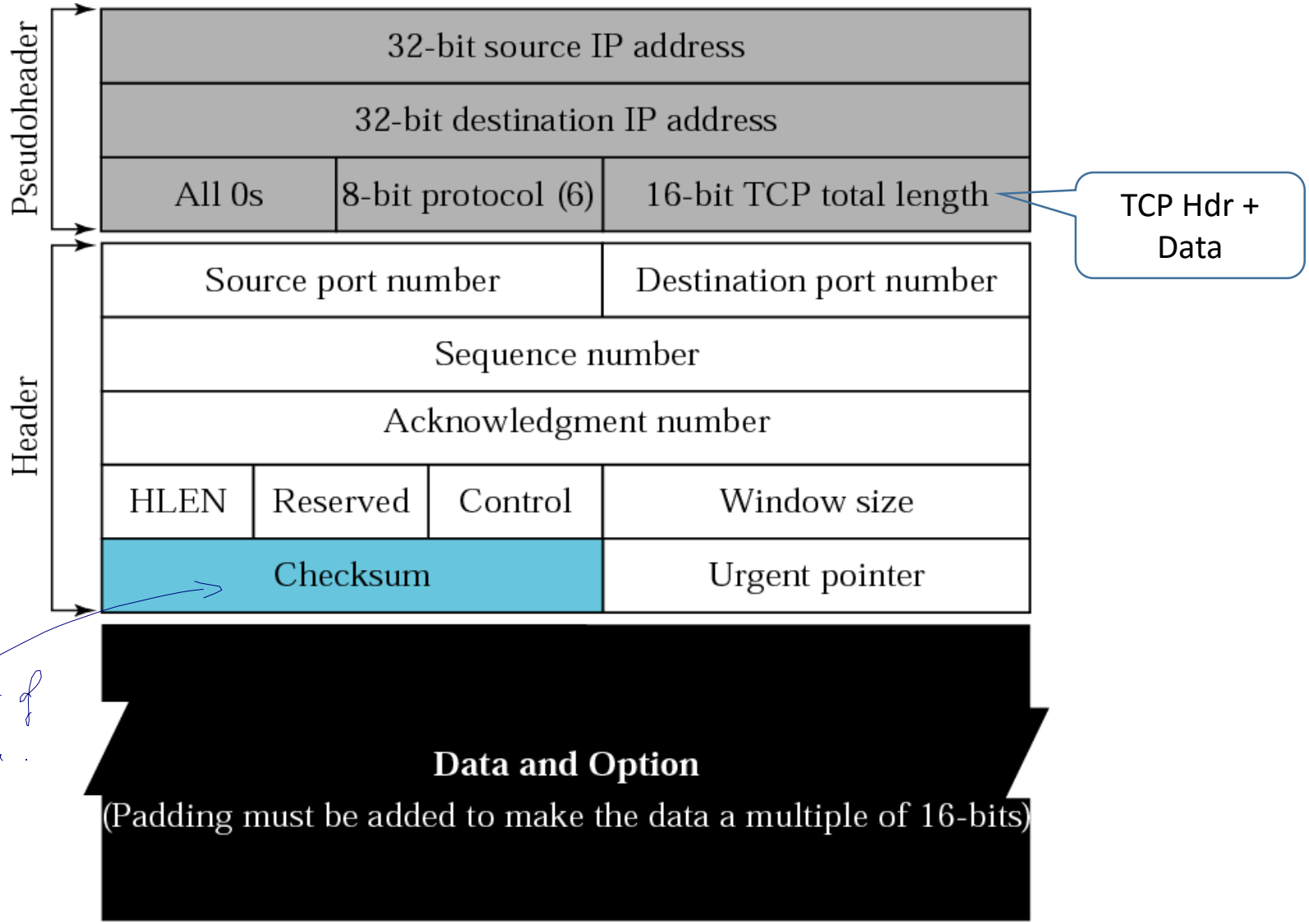


Table 12.2 Description of flags in the control field

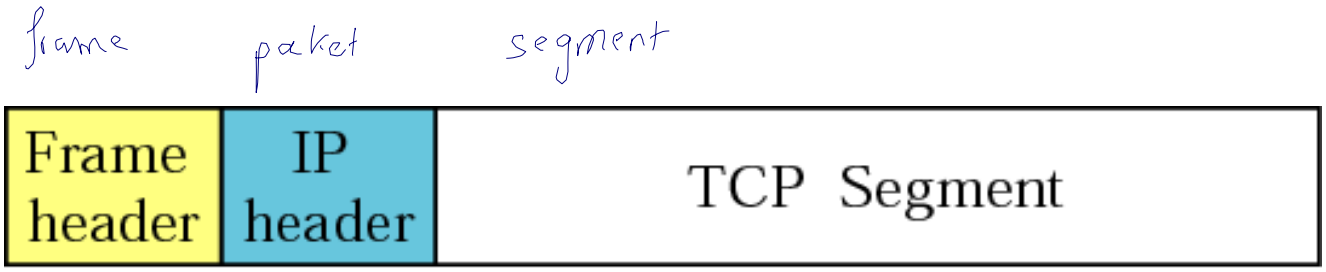
Flag	*Description*
URG	The value of the urgent pointer field is valid
ACK	The value of the acknowledgment field is valid
PSH	Push the data
RST	The connection must be reset
SYN	Synchronize sequence numbers during connection
FIN	Terminate the connection

Pseudoheader added to the TCP datagram



The inclusion of the checksum in TCP is mandatory.

Encapsulation and decapsulation

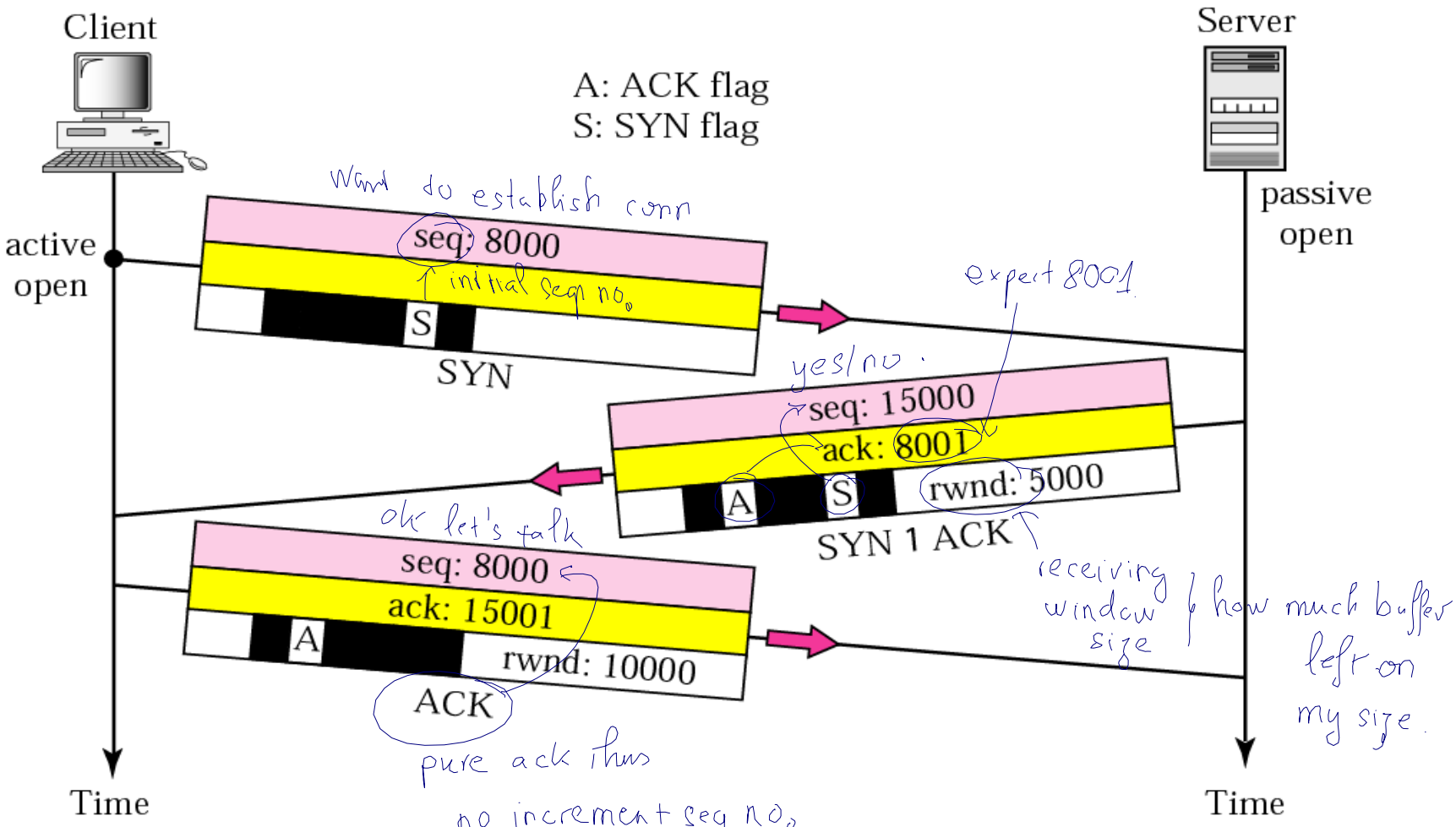


A TCP CONNECTION

TCP is connection-oriented. A connection-oriented transport protocol establishes a virtual path between the source and destination. All of the segments belonging to a message are then sent over this virtual path. A connection-oriented transmission requires three phases: connection establishment, data transfer, and connection termination.

- Connection Establishment
- Data Transfer
- Connection Termination
- Connection Reset

Connection establishment -- three-way handshaking



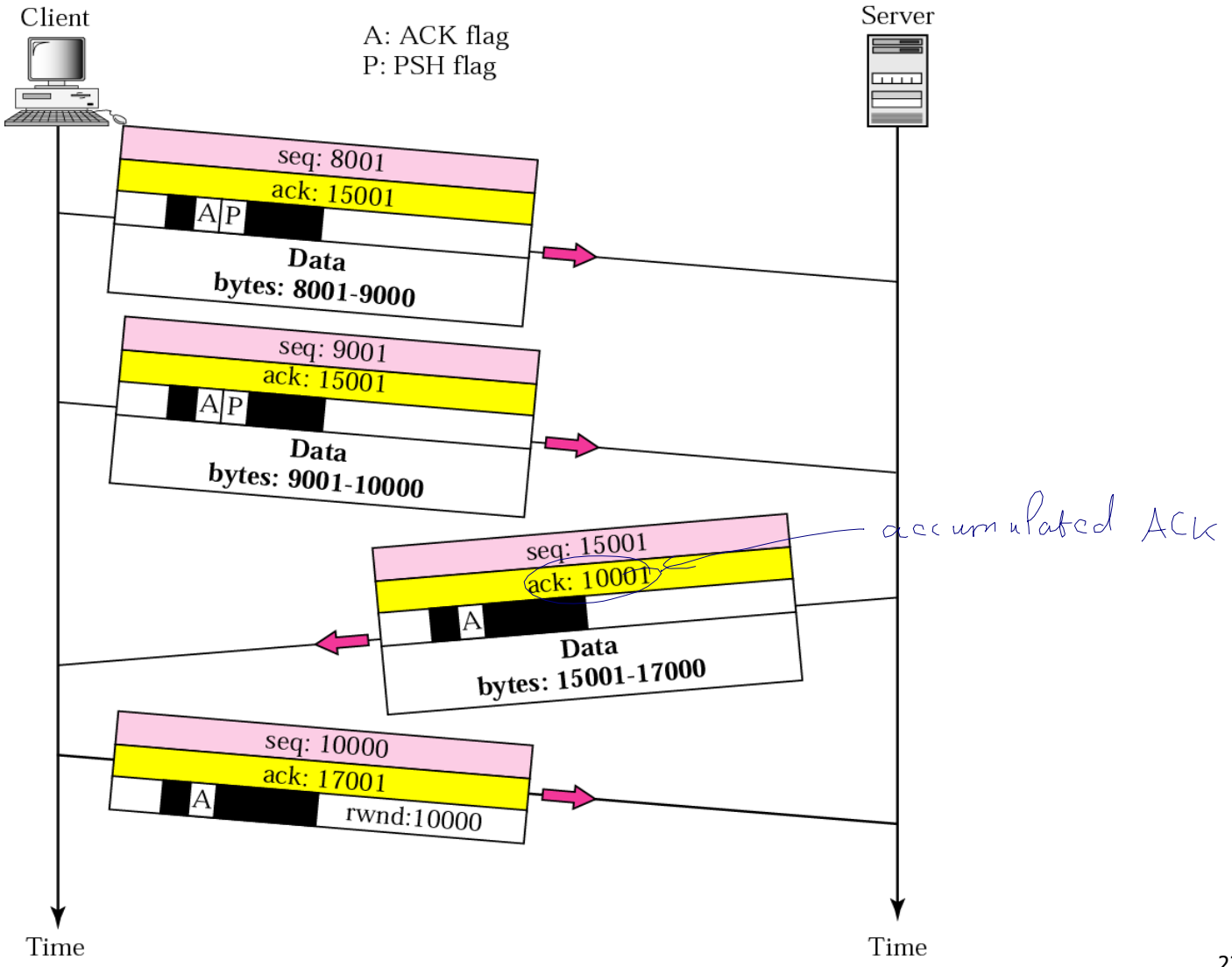
A SYN segment cannot carry data, but it consumes one sequence number.

ignore

A SYN + ACK segment cannot carry data, but does consume one sequence number.

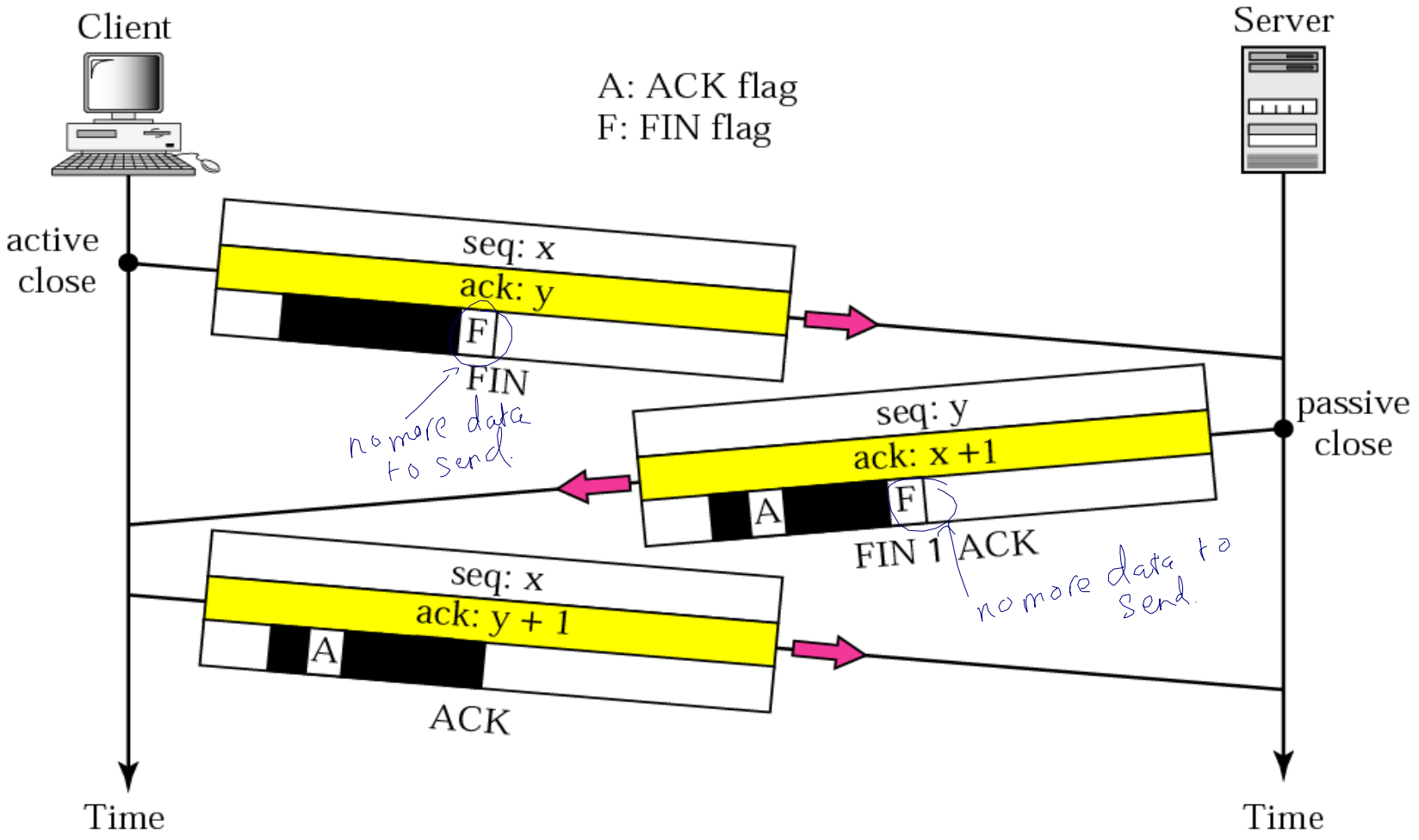
An ACK segment, if carrying no data, consumes no sequence number.

Data transfer



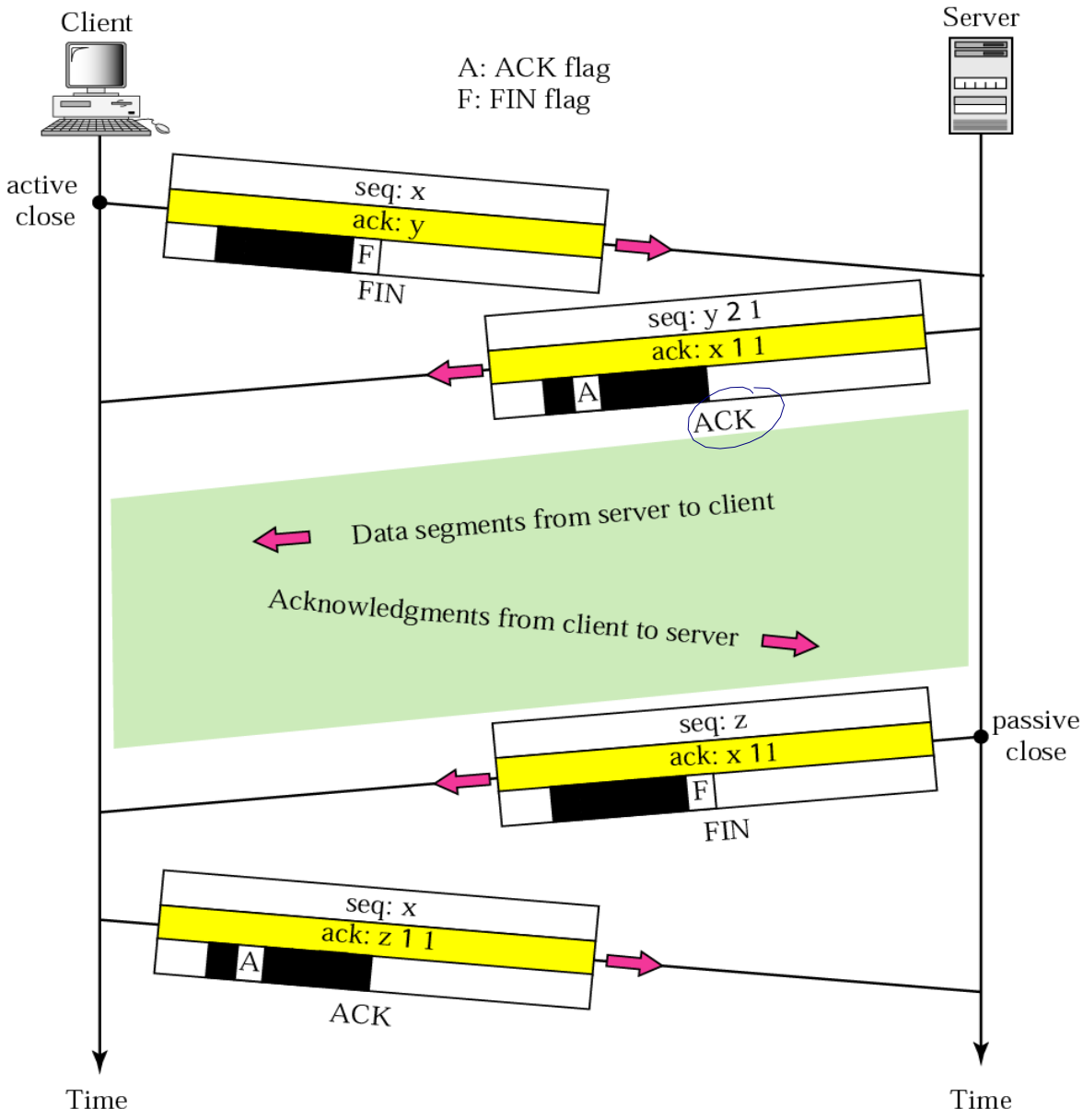
The FIN segment consumes one sequence number if it does not carry data.

Connection termination using three-way handshaking



The FIN + ACK segment consumes one sequence number if it does not carry data.

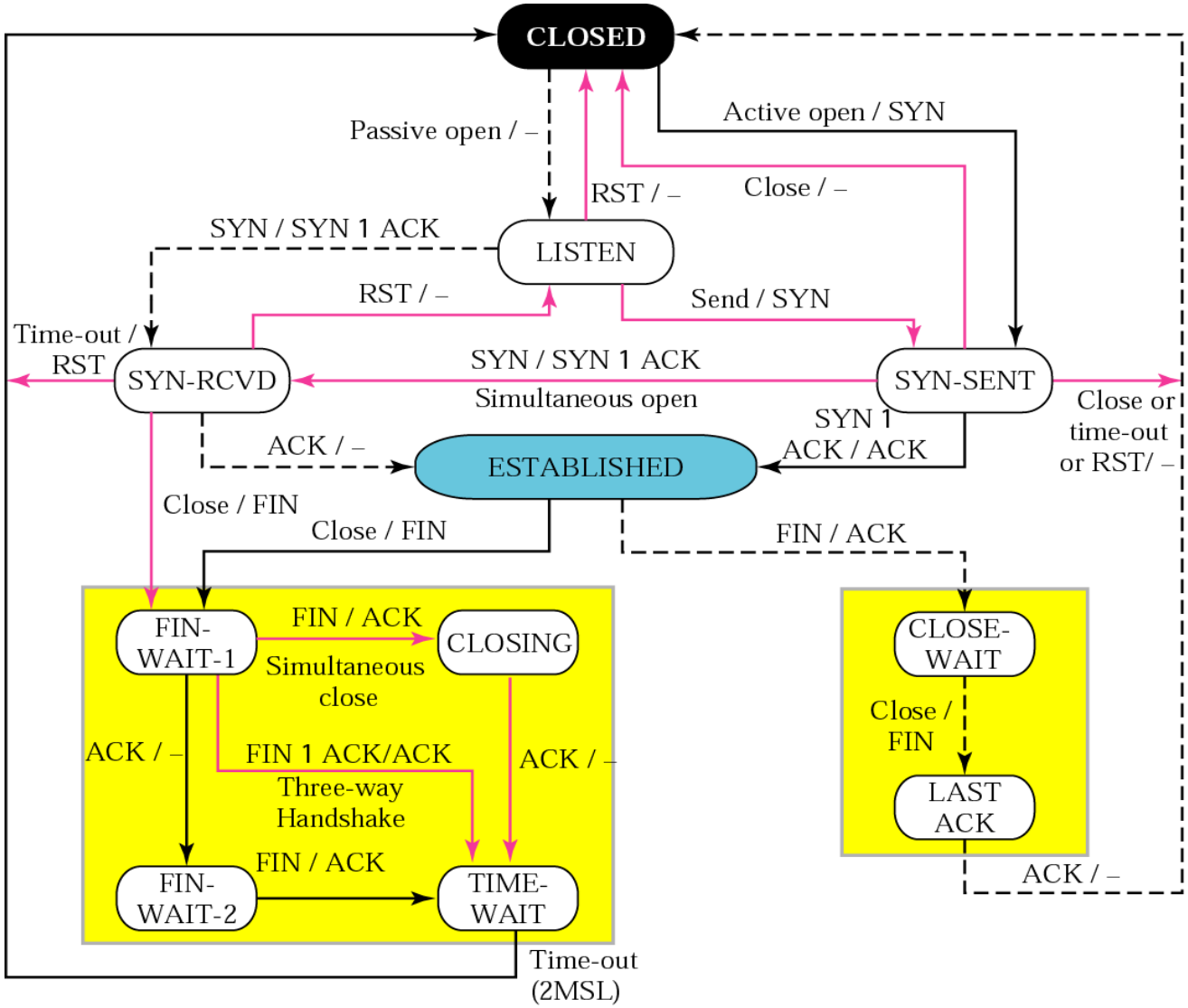
Half-close


Client
Server
A: ACK flag
F: FIN flag
active close
seq: x
ack: y
F
FIN
seq: y 2 1
ack: x 1 1
A
ACK
Data segments from server to client
Acknowledgments from client to server
seq: z
ack: x 11
F
FIN
passive close
seq: x
ack: z 1 1
A
ACK
Time
Time

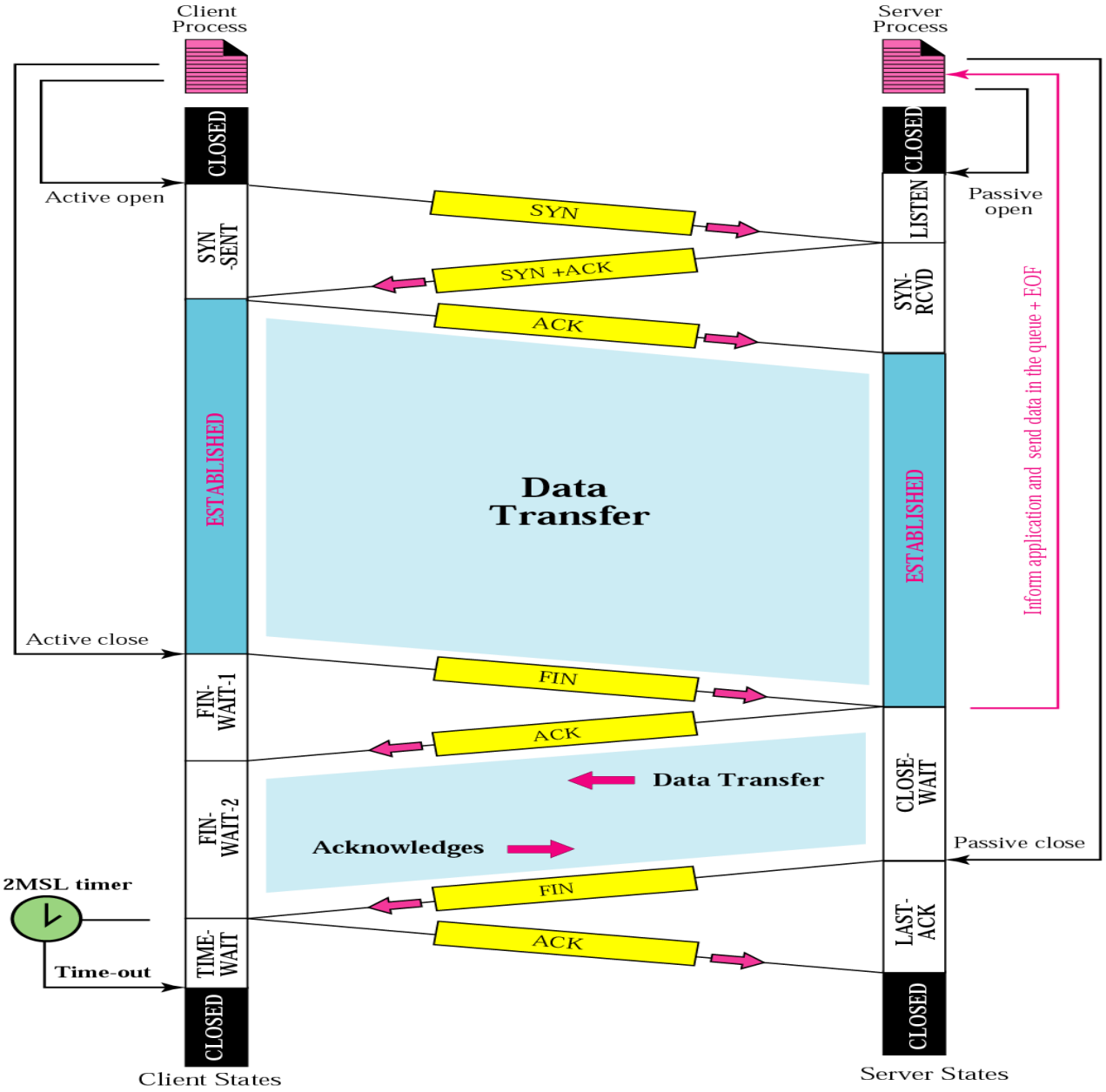
States for TCP

State	*Description*
CLOSED	There is no connection
LISTEN	Passive open received; waiting for SYN
SYN-SENT	SYN sent; waiting for ACK
SYN-RCVD	SYN+ACK sent; waiting for ACK
ESTABLISHED	Connection established; data transfer in progress
FIN-WAIT-1	First FIN sent; waiting for ACK
FIN-WAIT-2	ACK to first FIN received; waiting for second FIN
CLOSE-WAIT	First FIN received, ACK sent; waiting for application to close
TIME-WAIT	Second FIN received, ACK sent; waiting for 2MSL time-out
LAST-ACK	Second FIN sent; waiting for ACK
CLOSING	Both sides have decided to close simultaneously

State transition diagram



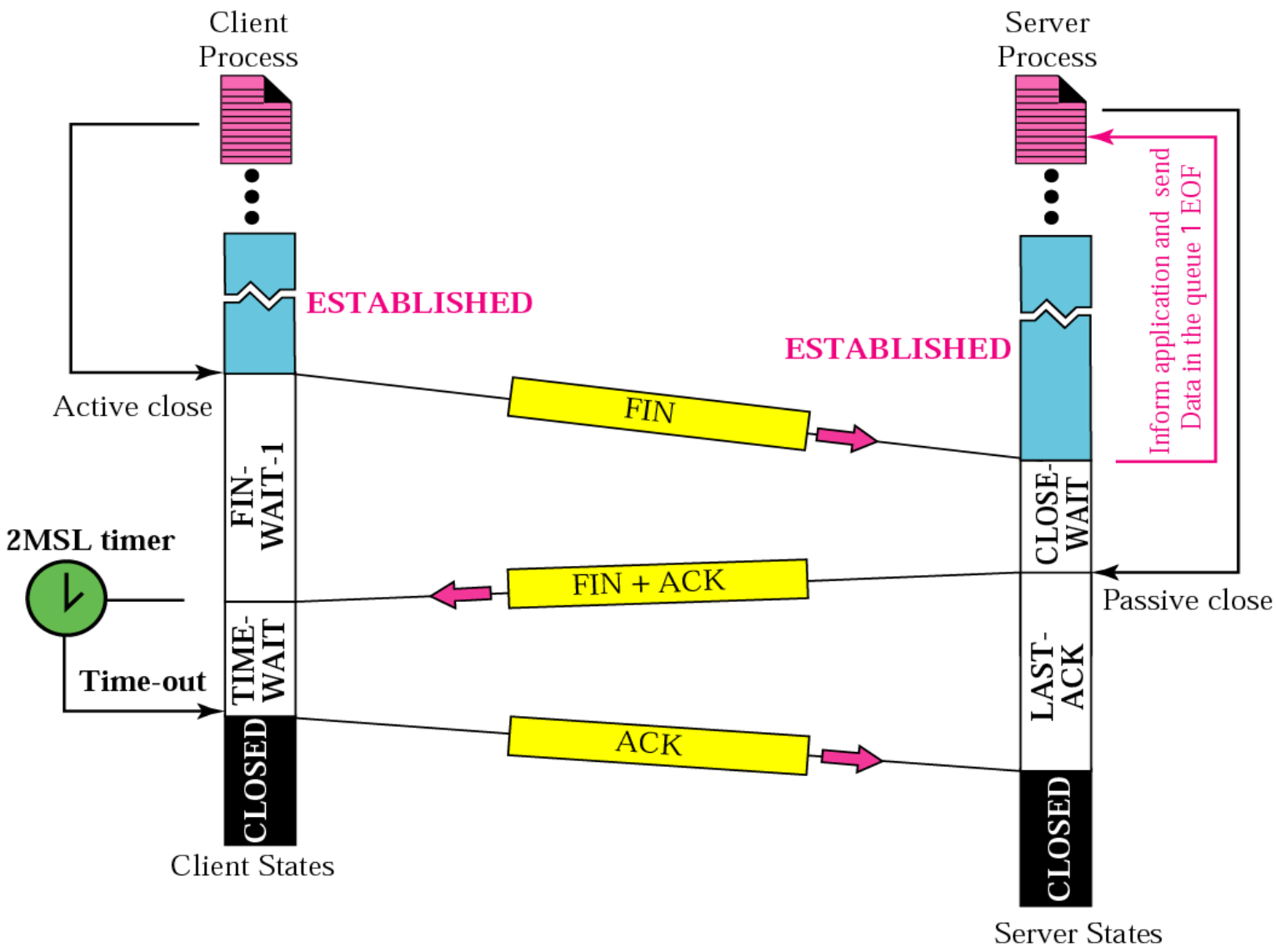
Common scenario



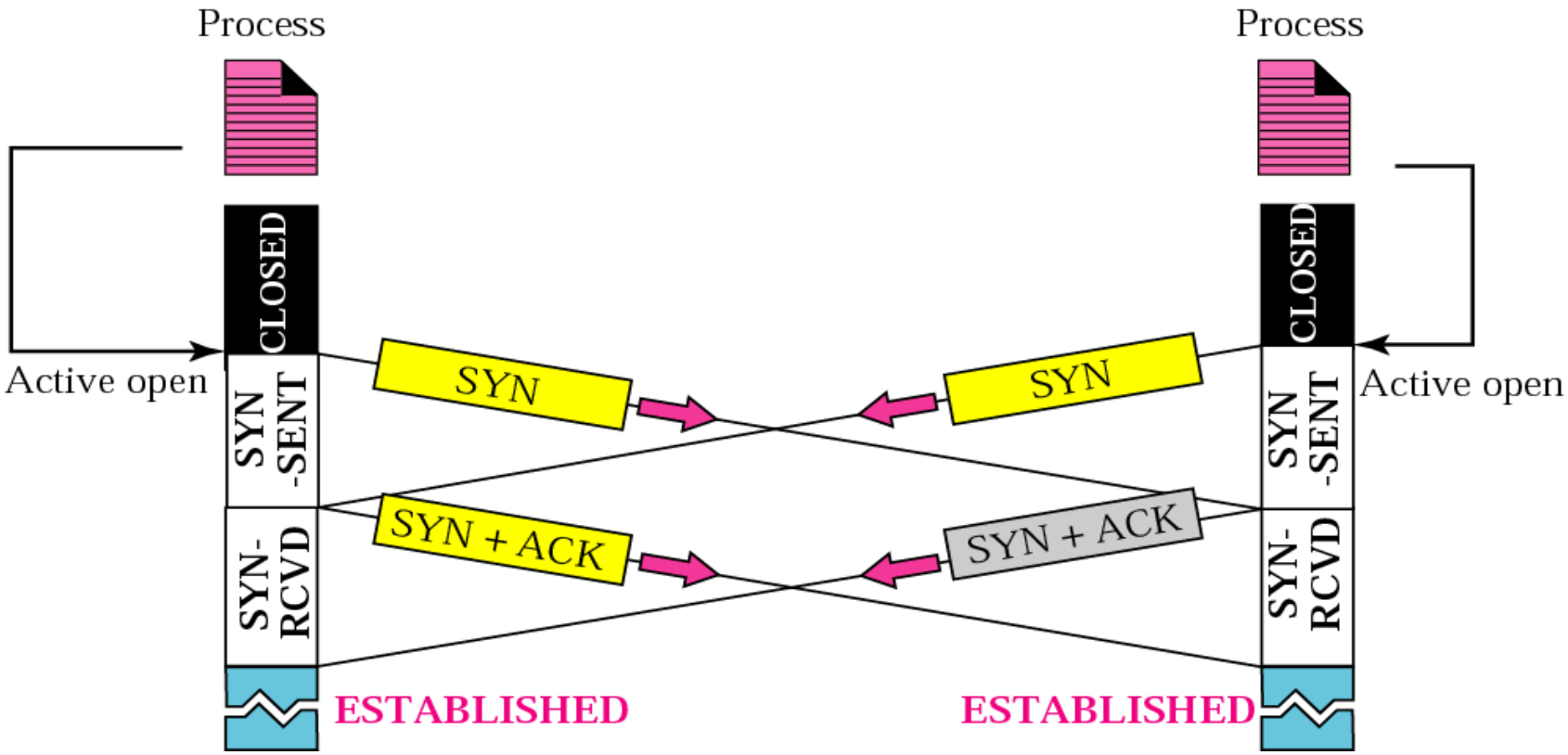
The common value for MSL is between 30 seconds and 1 minute.

MSL – Maximum Segment Lifetime

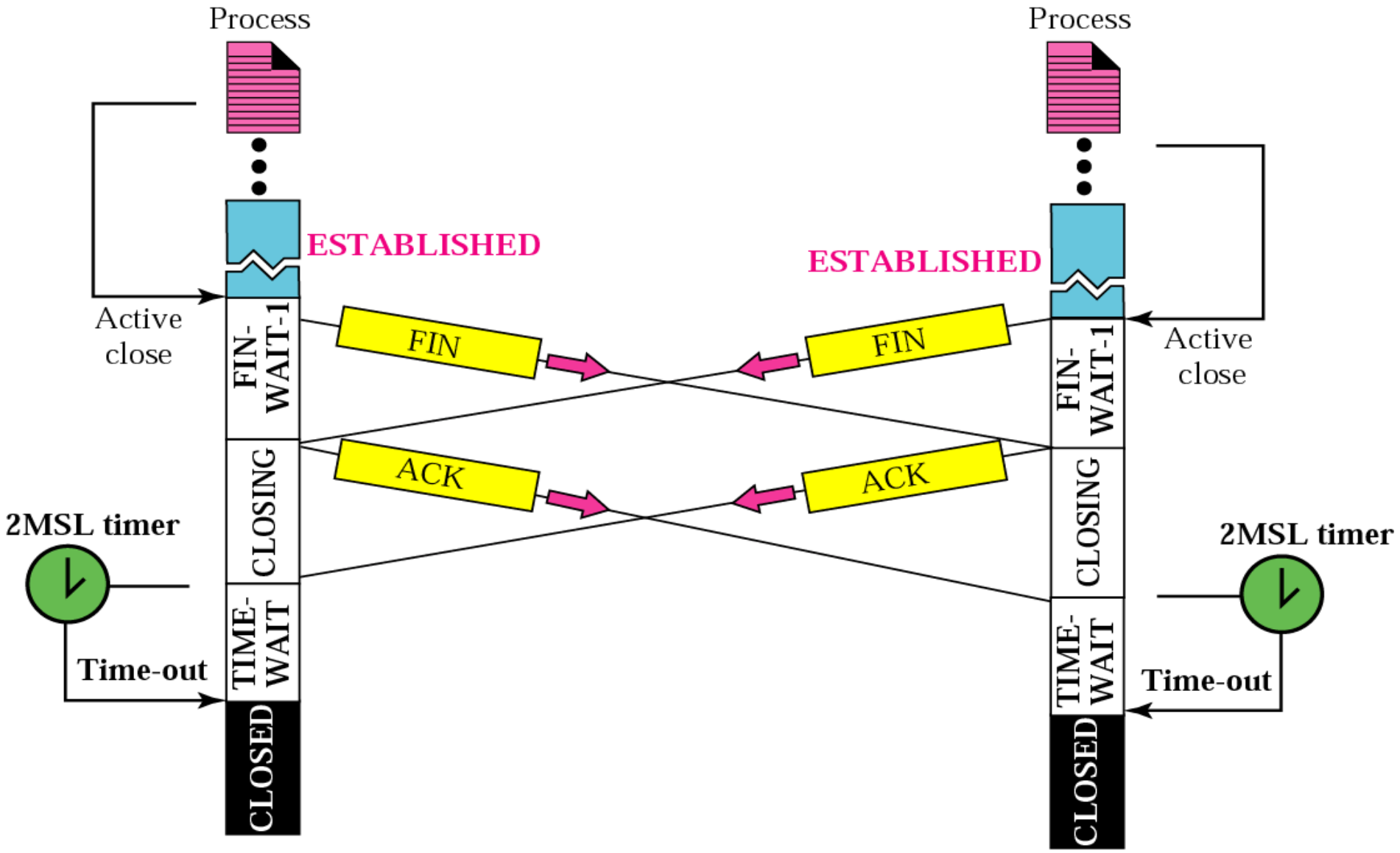
Three-way handshake



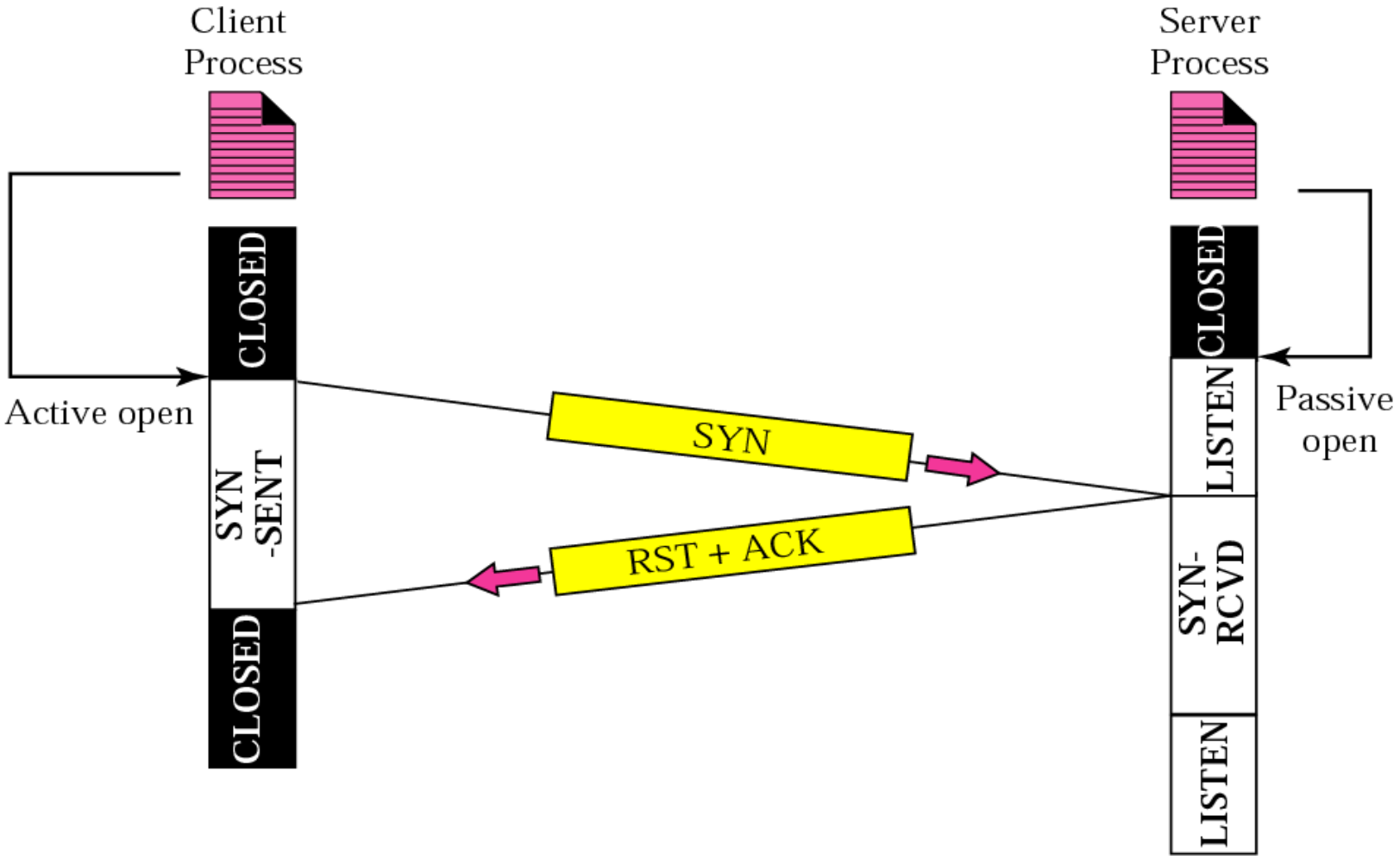
Simultaneous open



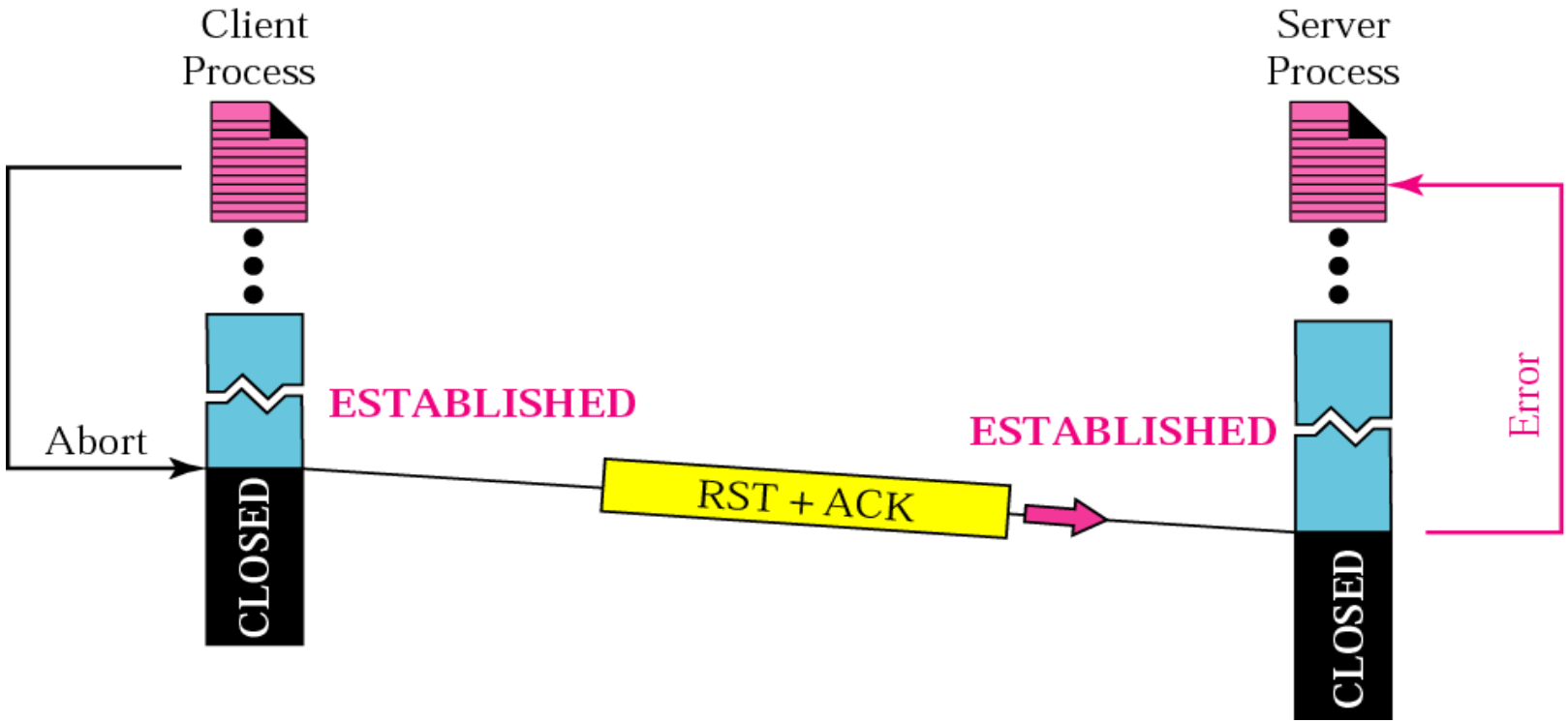
Simultaneous close



Denying a connection



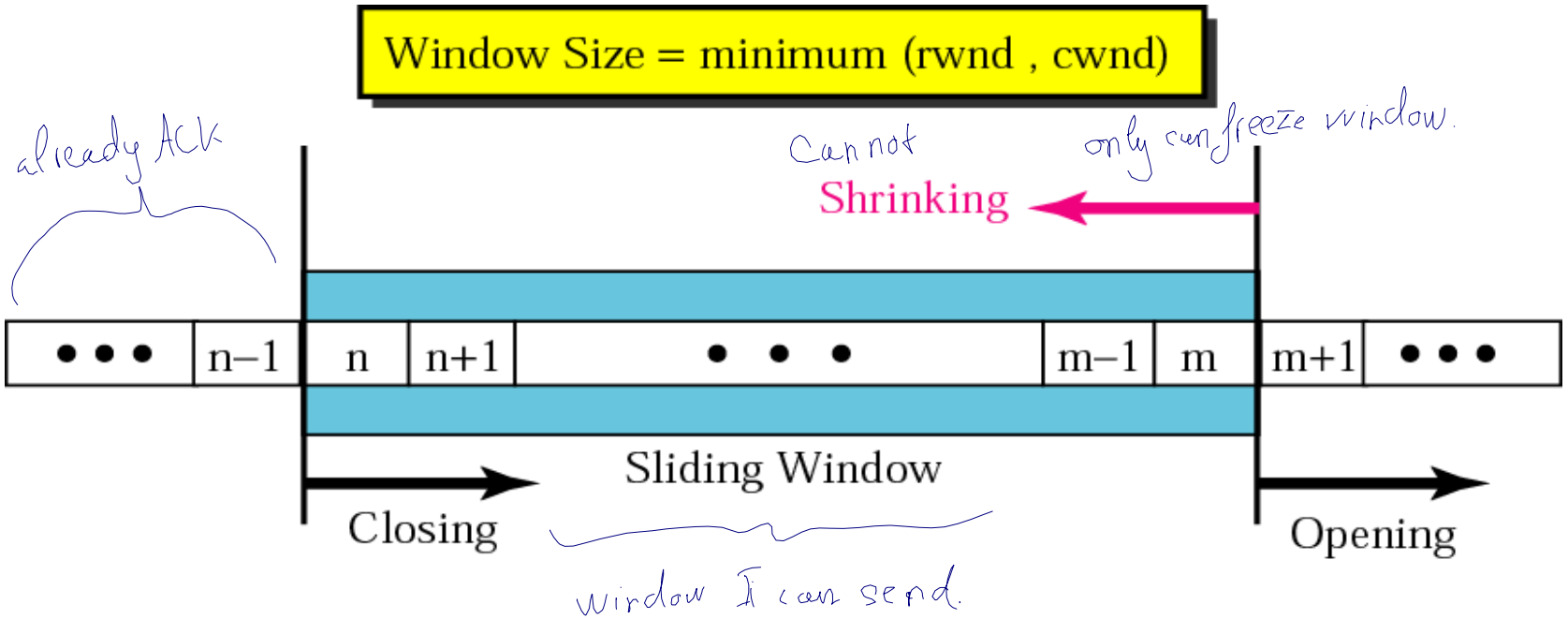
Aborting a connection



Flow Contrl

- Flow control regulates the amount of data a source can send before receiving an acknowledgment from the destination. TCP defines a window that is imposed on the buffer of data delivered from the application program.
  - Sliding Window Protocol
  - Silly Window Syndrome

Sliding Window



A sliding window is used to make transmission more efficient as well as to control the flow of data so that the destination does not become overwhelmed with data.

TCP's sliding windows are byte oriented.

EXAMPLE

What is the value of the receiver window (rwnd) for host A if the receiver, host B, has a buffer size of 5,000 bytes and 1,000 bytes of received and unprocessed data?

Solution

The value of rwnd = 5,000 − 1,000 = 4,000. Host B can receive only 4,000 bytes of data before overflowing its buffer. Host B advertises this value in its next segment to A.

EXAMPLE 4

What is the size of the window for host A if the value of rwnd is 3,000 bytes and the value of cwnd is 3,500 bytes?

congestion window (on the sender size)
always use the smaller one.

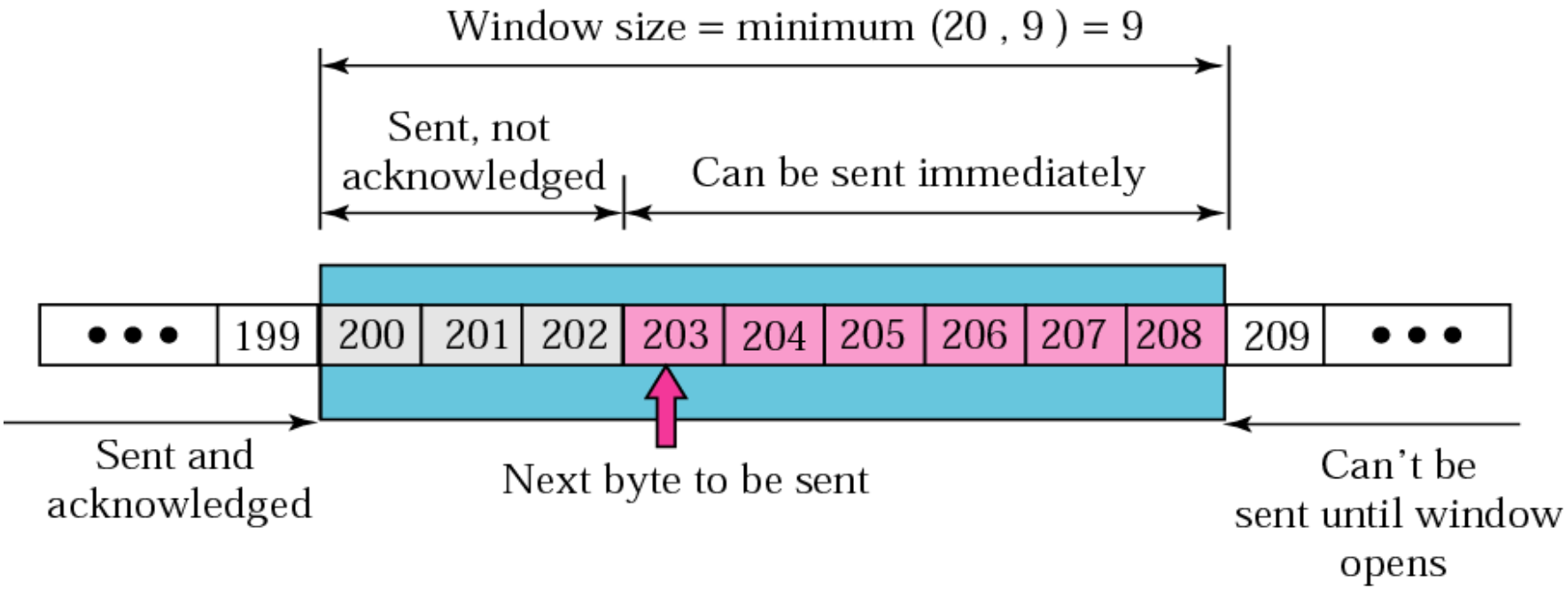
Solution

The size of the window is the smaller of rwnd and cwnd, which is 3,000 bytes.

EXAMPLE

The sender has sent bytes up to 202. We assume that cwnd is 20 (in reality this value is thousands of bytes). The receiver has sent an acknowledgment number of 200 with an rwnd of 9 bytes (in reality this value is thousands of bytes). The size of the sender window is the minimum of rwnd and cwnd or 9 bytes. Bytes 200 to 202 are sent, but not acknowledged. Bytes 203 to 208 can be sent without worrying about acknowledgment. Bytes 209 and above cannot be sent.

Example 5


Window size = minimum (20 , 9) = 9
Sent, not acknowledged
Can be sent immediately
199
200
201
202
203
204
205
206
207
208
209
Sent and acknowledged
Next byte to be sent
Can't be sent until window opens

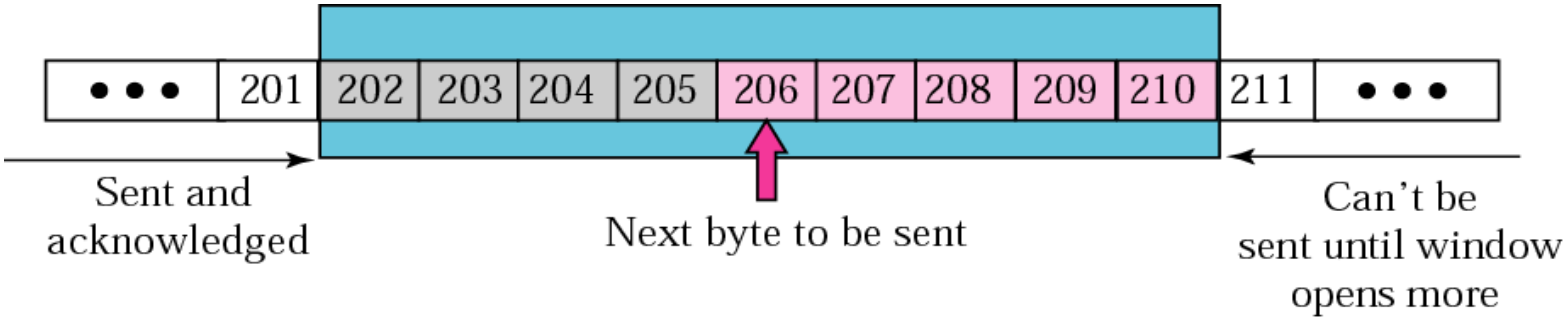
EXAMPLE

In Figure 12.21 the server receives a packet with an acknowledgment value of 202 and an rwnd of 9. The host has already sent bytes 203, 204, and 205. The value of cwnd is still 20. Show the new window.

Solution

Figure 12.22 shows the new window. Note that this is a case in which the window closes from the left and opens from the right by an equal number of bytes; the size of the window has not been changed. The acknowledgment value, 202, declares that bytes 200 and 201 have been received and the sender needs not worry about them; the window can slide over them.

Example 6

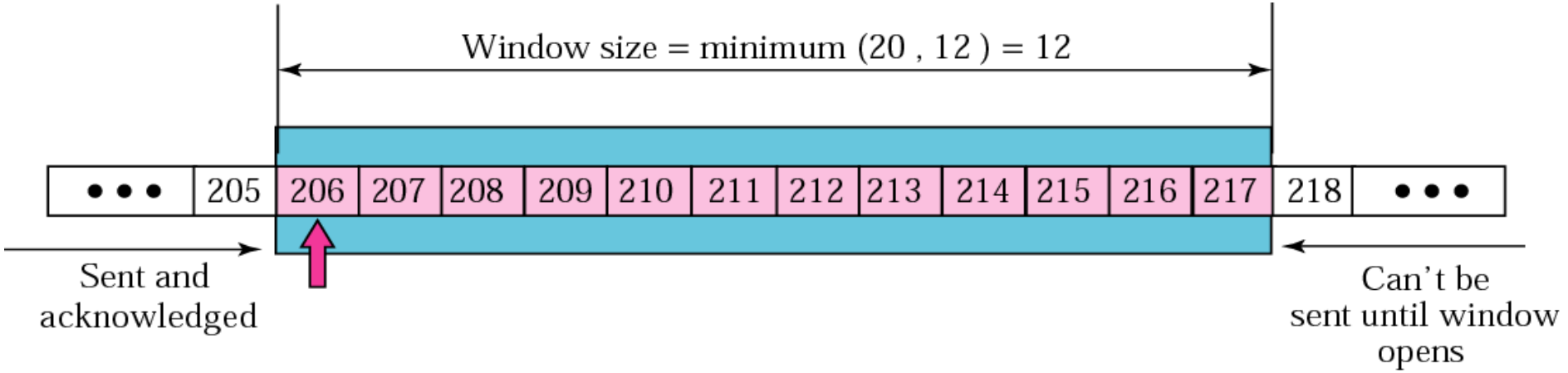

201
202
203
204
205
206
207
208
209
210
211
Sent and acknowledged
Next byte to be sent
Can't be sent until window opens more

EXAMPLE

In Figure 12.22 the sender receives a packet with an acknowledgment value of 206 and an rwnd of 12. The host has not sent any new bytes. The value of cwnd is still 20. Show the new window.

Solution

The value of rwnd is less than cwnd, so the size of the window is 12. Figure 12.23 shows the new window. Note that the window has been opened from the right by 7 and closed from the left by 4; the size of the window has increased.


Window size = minimum (20 , 12) = 12
205
206
207
208
209
210
211
212
213
214
215
216
217
218
Sent and acknowledged
Can't be sent until window opens

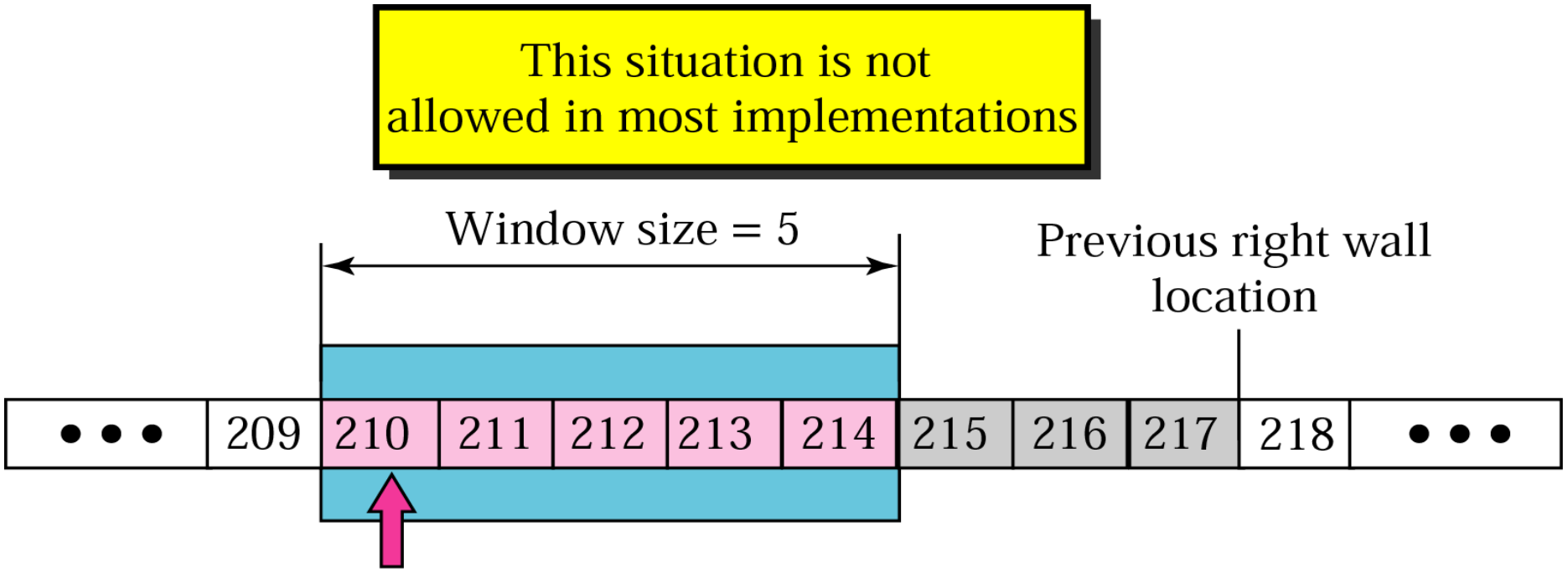
EXAMPLE

In Figure 12.23 the host receives a packet with an acknowledgment value of 210 and an rwnd of 5. The host has sent bytes 206, 207, 208, and 209. The value of cwnd is still 20. Show the new window.

210, 211, 212, 213, 214

Solution

The value of rwnd is less than cwnd, so the size of the window is 5. Figure 12.24 shows the situation. Note that this is a case not allowed by most implementations. Although the sender has not sent bytes 215 to 217, the receiver does not know this.


This situation is not
allowed in most implementations
Window size = 5
Previous right wall
location
209
210
211
212
213
214
215
216
217
218

EXAMPLE

How can the receiver avoid shrinking the window in the previous example?

Solution

The receiver needs to keep track of the last acknowledgment number and the last rwnd. If we add the acknowledgment number to rwnd we get the byte number following the right wall. If we want to prevent the right wall from moving to the left (shrinking), we must always have the following relationship.

$$\textit{new ack} + \textit{new rwnd} \geq \textit{last ack} + \textit{last rwnd}$$

or

$$\textit{new rwnd} \geq (\textit{last ack} + \textit{last rwnd}) - \textit{new ack}$$

To avoid shrinking the sender window, the receiver must wait** until **more space is available in its buffer.

Note:

Some points about TCP's sliding windows:

- *The size of the window is the lesser of rwnd and cwnd.*
- *The source does not have to send a full window's worth of data.*
- *The window can be opened or closed by the receiver, but should not be shrunk.*
- *The destination can send an acknowledgment at any time as long as it does not result in a shrinking window.*
- *The receiver can temporarily shut down the window; the sender, however, can always send a segment of one byte after the window is shut down.*

Error Control

TCP provides reliability using error control, which detects corrupted, lost, out-of-order, and duplicated segments. Error control in TCP is achieved through the use of the checksum, acknowledgment, and time-out.

- Checksum
- Acknowledgment
- Acknowledgment Type
- Retransmission
- Out-of-Order Segments
- Some Scenarios

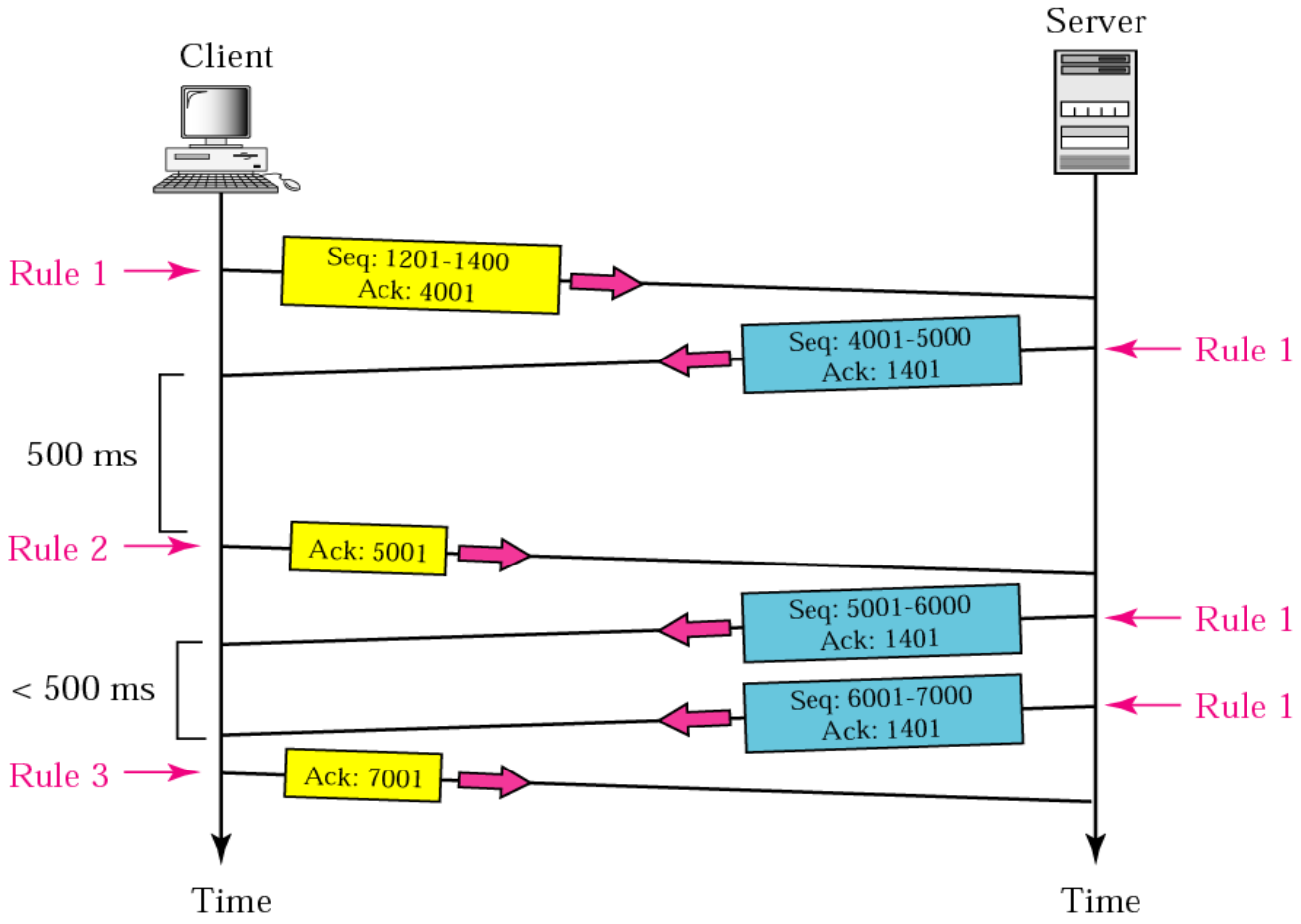
ACK segments do not consume sequence numbers and are not acknowledged.

In modern implementations, a retransmission occurs if the retransmission timer expires or three duplicate ACK segments have arrived.

No retransmission timer is set for an ACK segment.

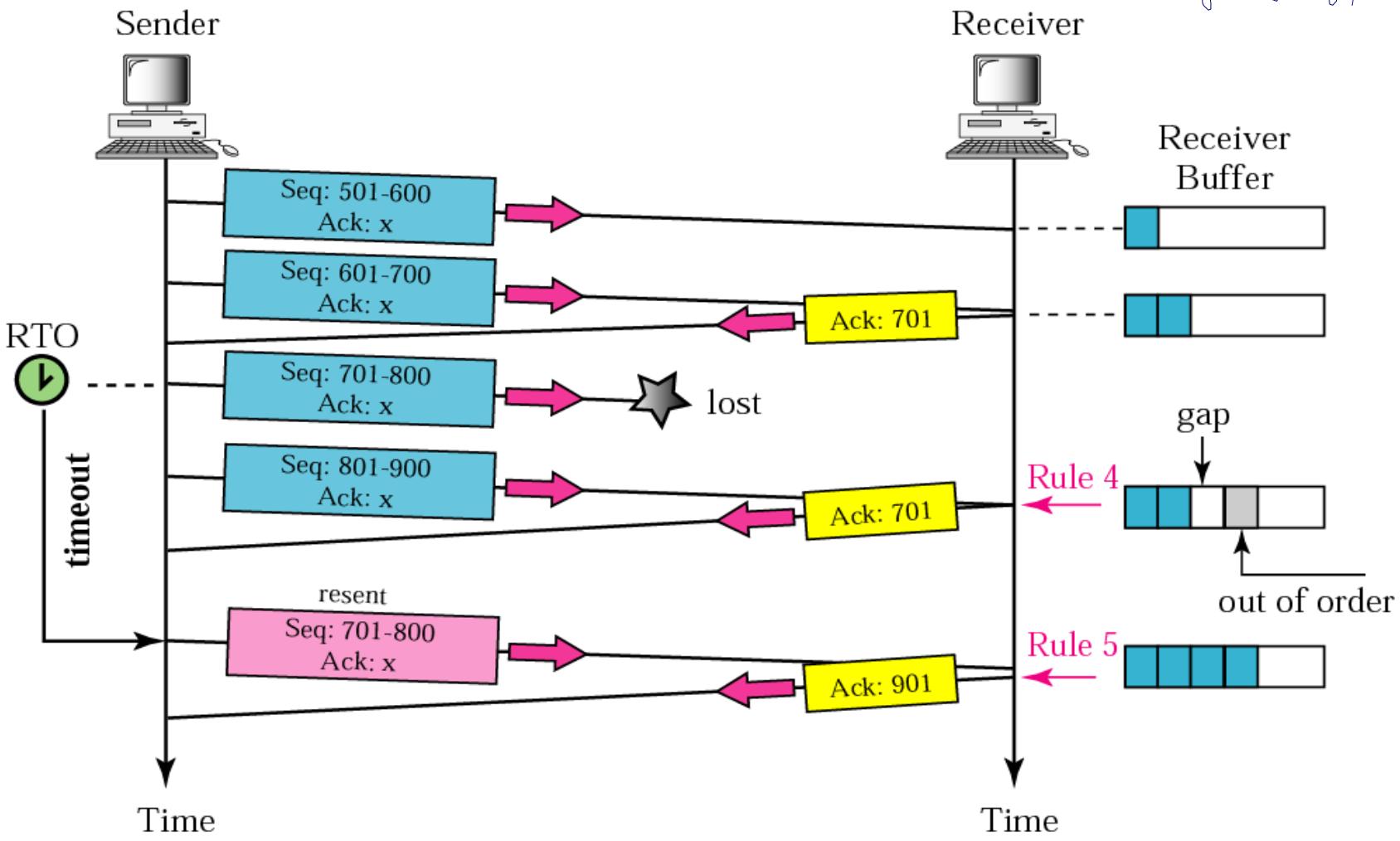
Data may arrive out of order and be temporarily stored by the receiving TCP, but TCP guarantees that no out-of-order segment is delivered to the process.

Normal Operation



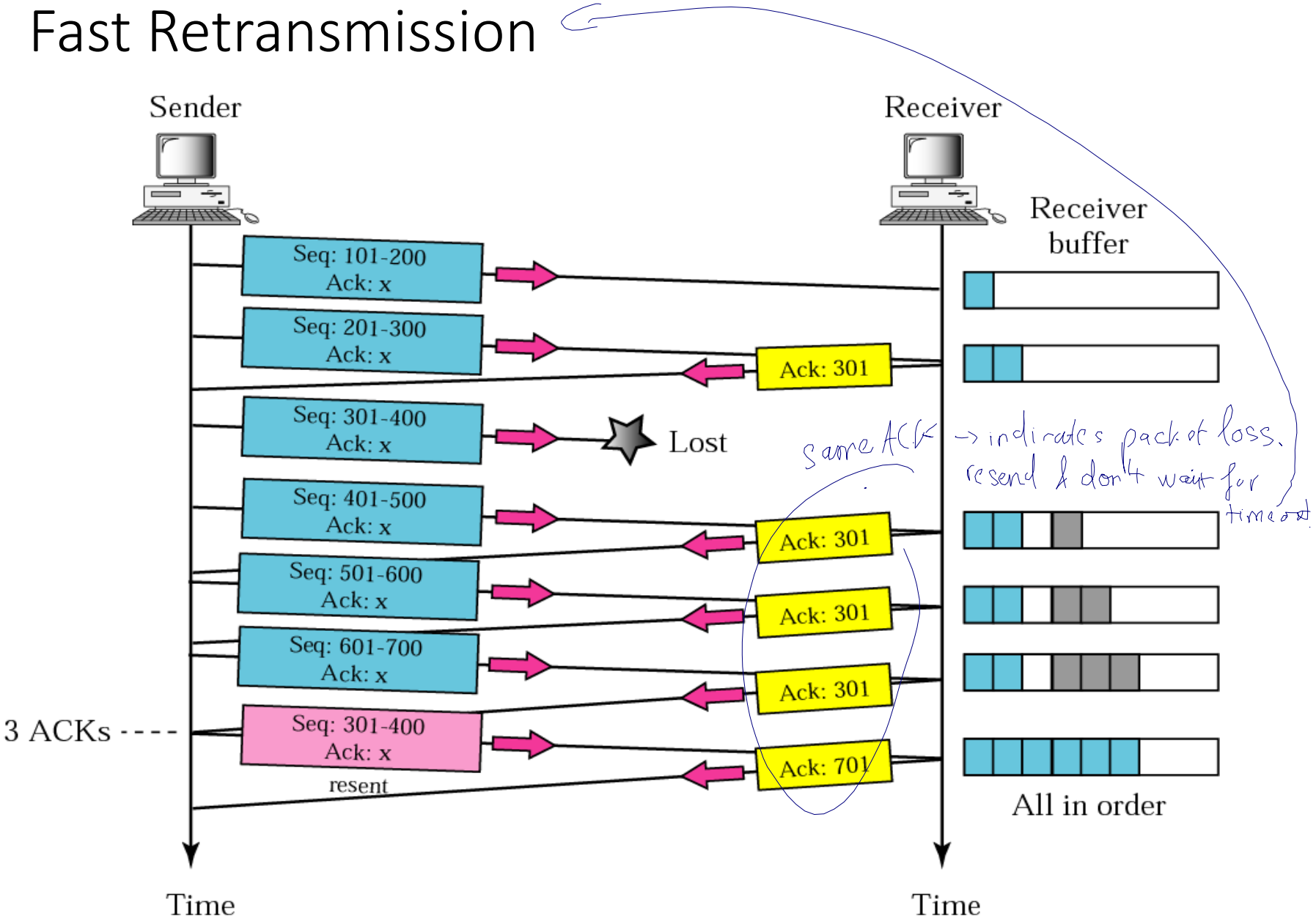
Lost Segment

Reasonable retransmission time is Round Trip Delay (slightly bigger)

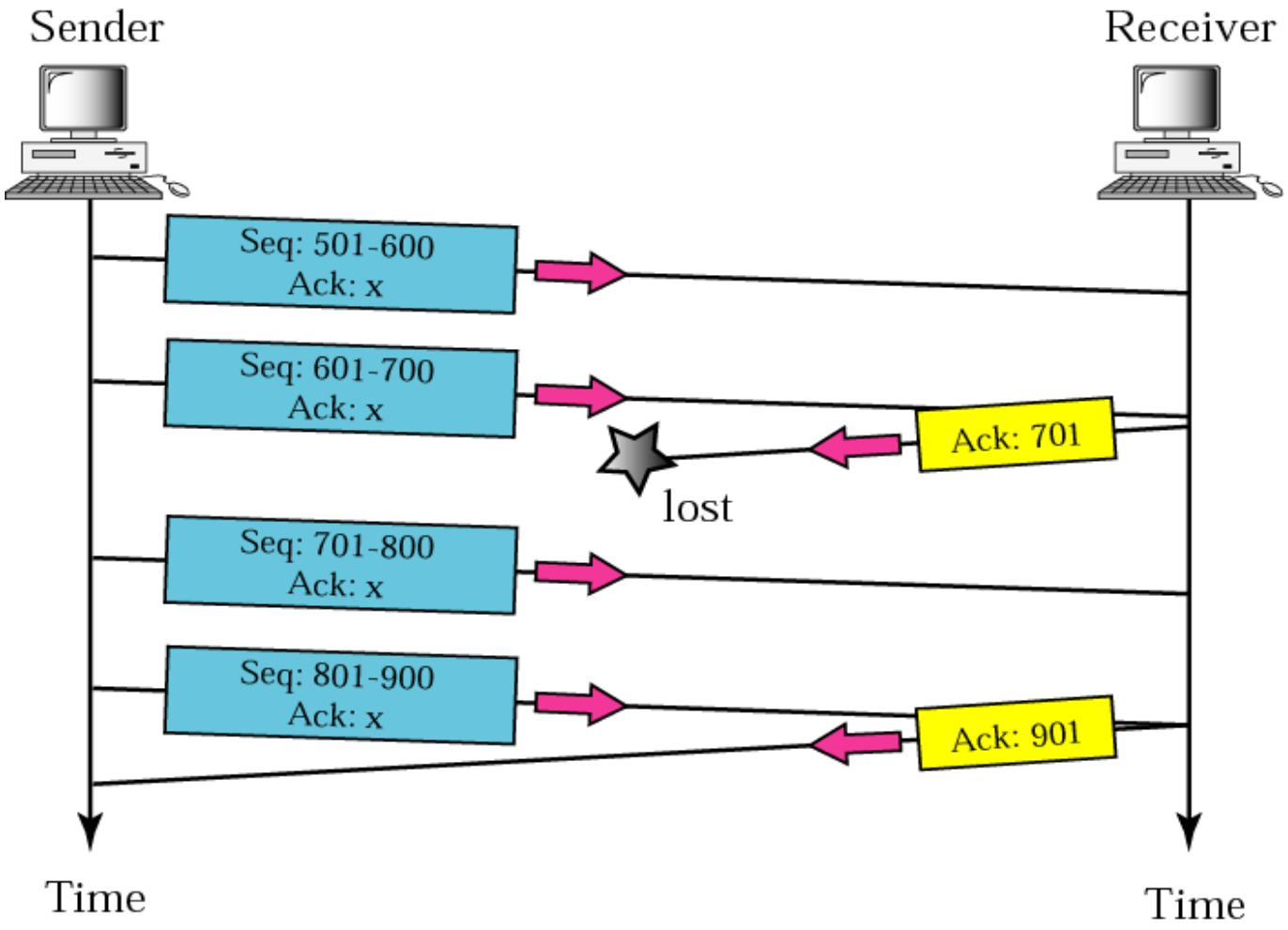


The receiver TCP delivers only ordered data to the process.

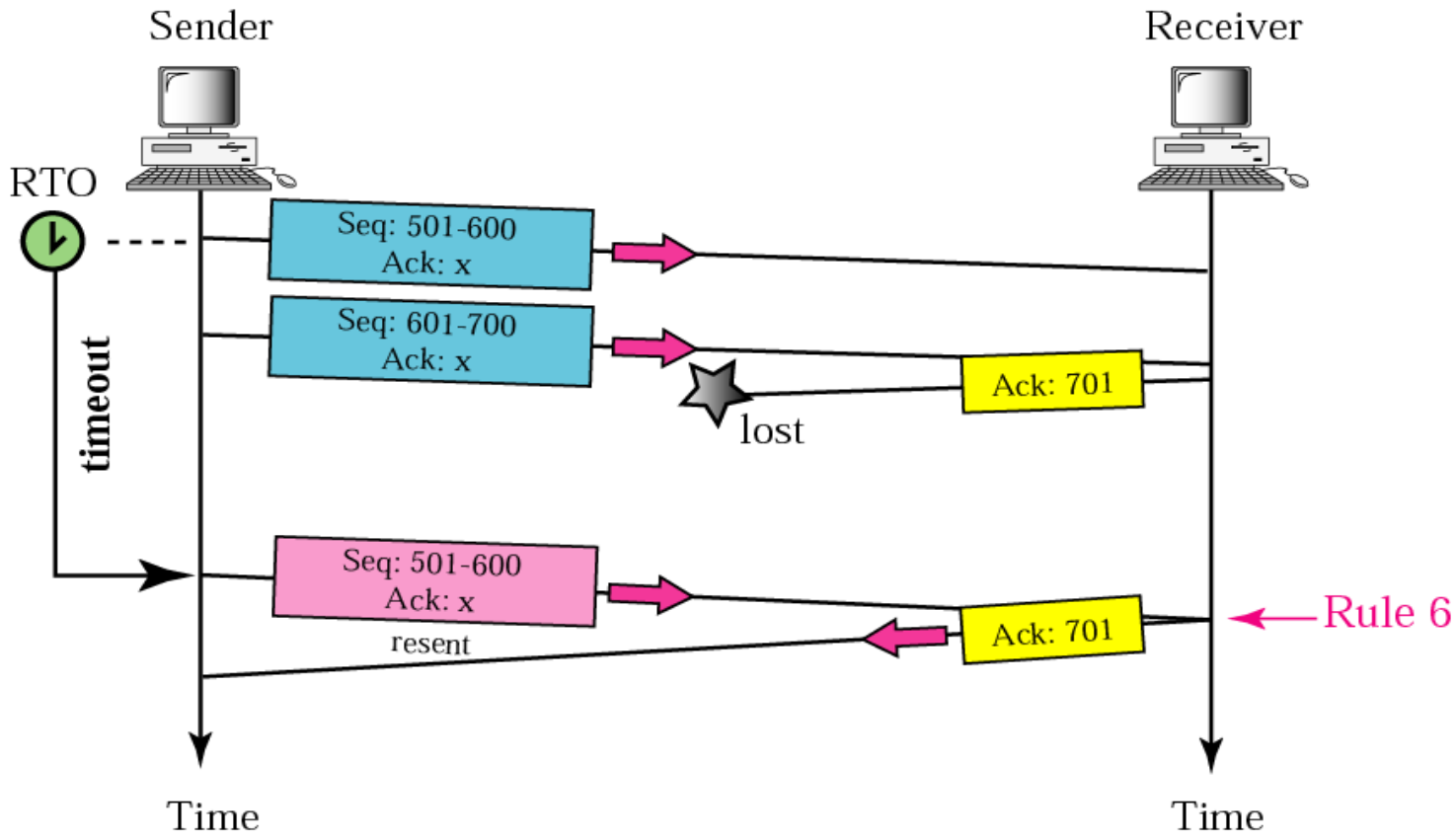
Fast Retransmission



Lost Acknowledgement



Lost acknowledgment corrected by resending a segment



Lost acknowledgments may create deadlock if they are not properly handled.

both stops talking

if ACK lost

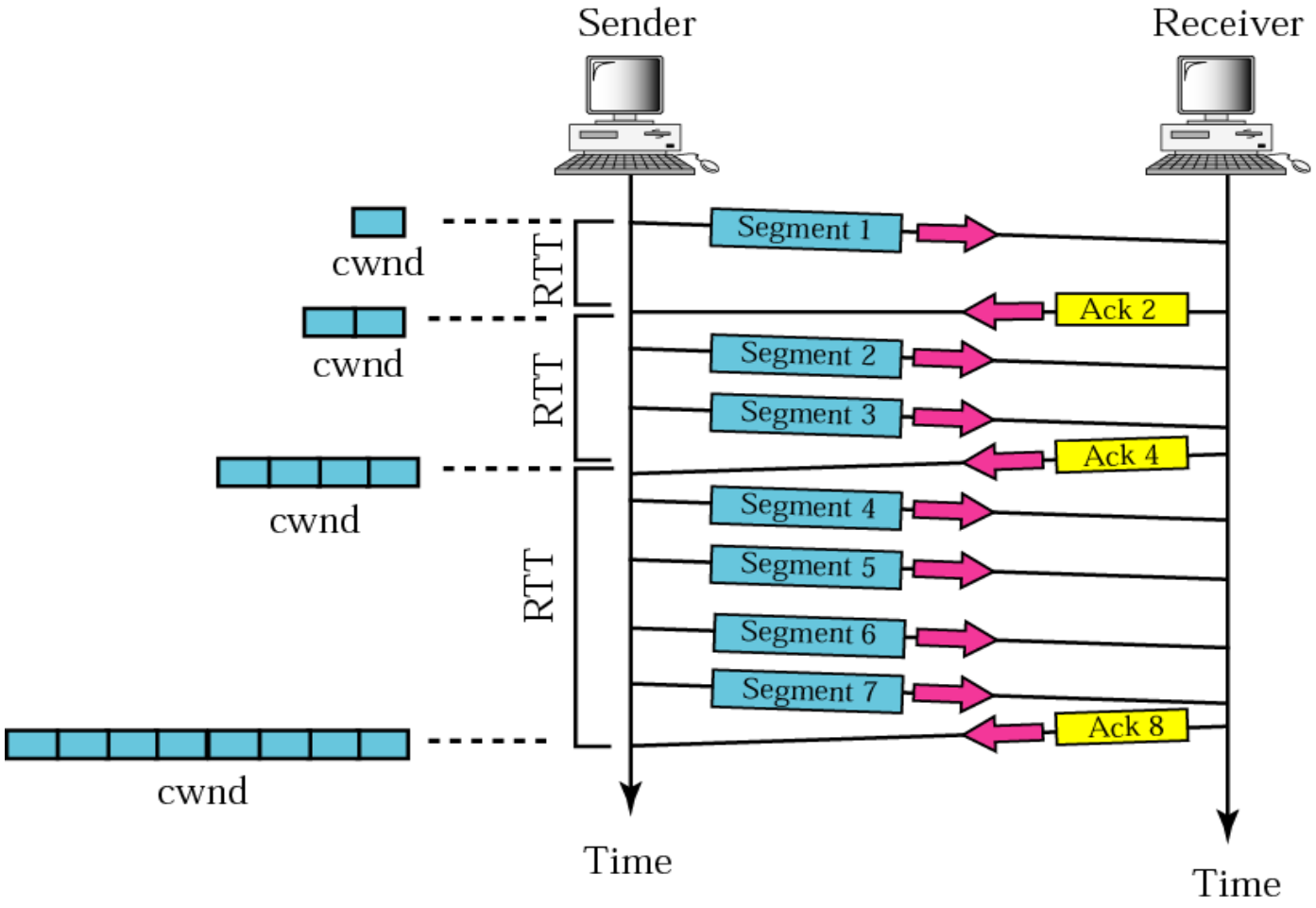
TCP persistent timer: sender sends 1 byte out to test the water

Congestion Control

Congestion control refers to the mechanisms and techniques to keep the load below the capacity.

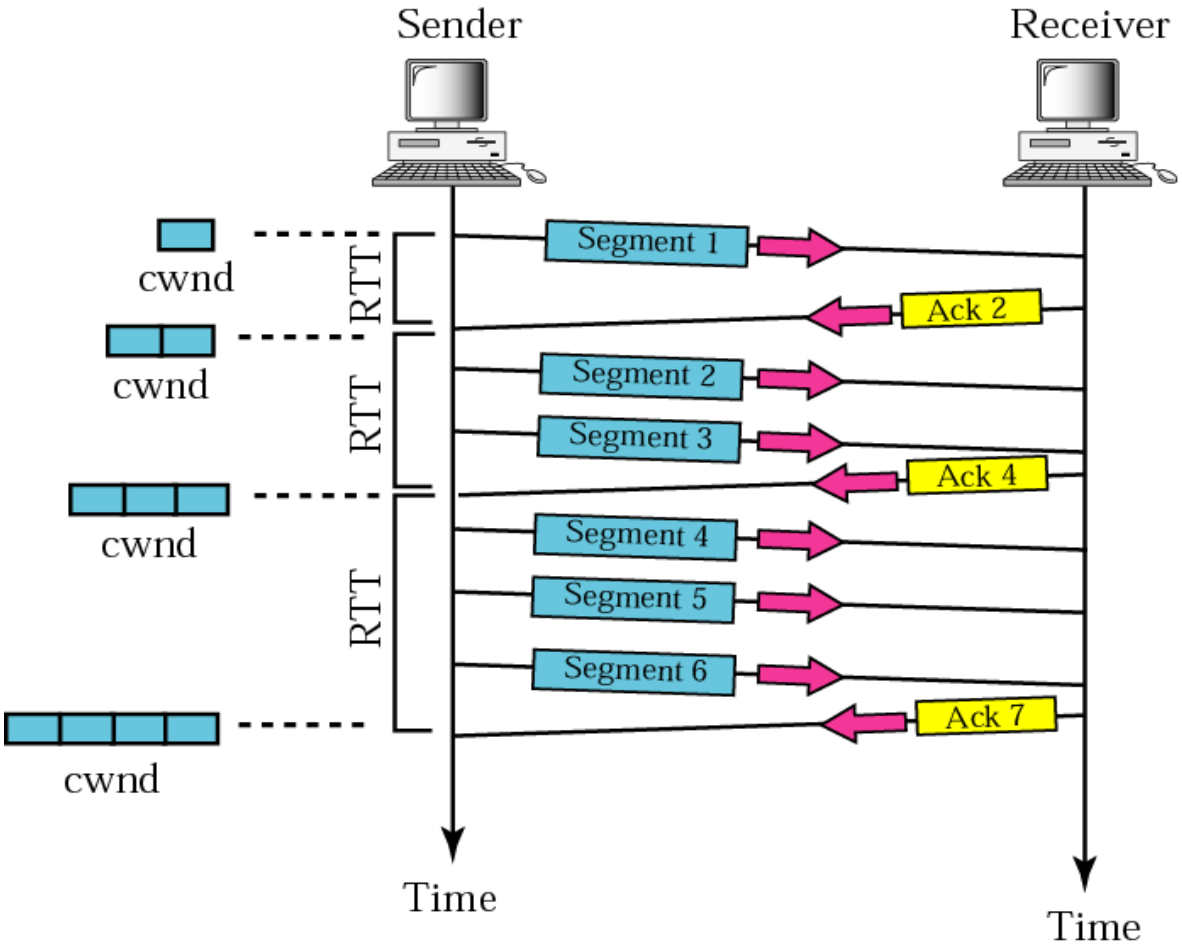
- Network Performance
- Congestion Control Mechanisms
- Congestion Control in TCP

Slow start, exponential increase



In the slow start algorithm, the size of the congestion window increases exponentially until it reaches a threshold.

Congestion avoidance, additive increase

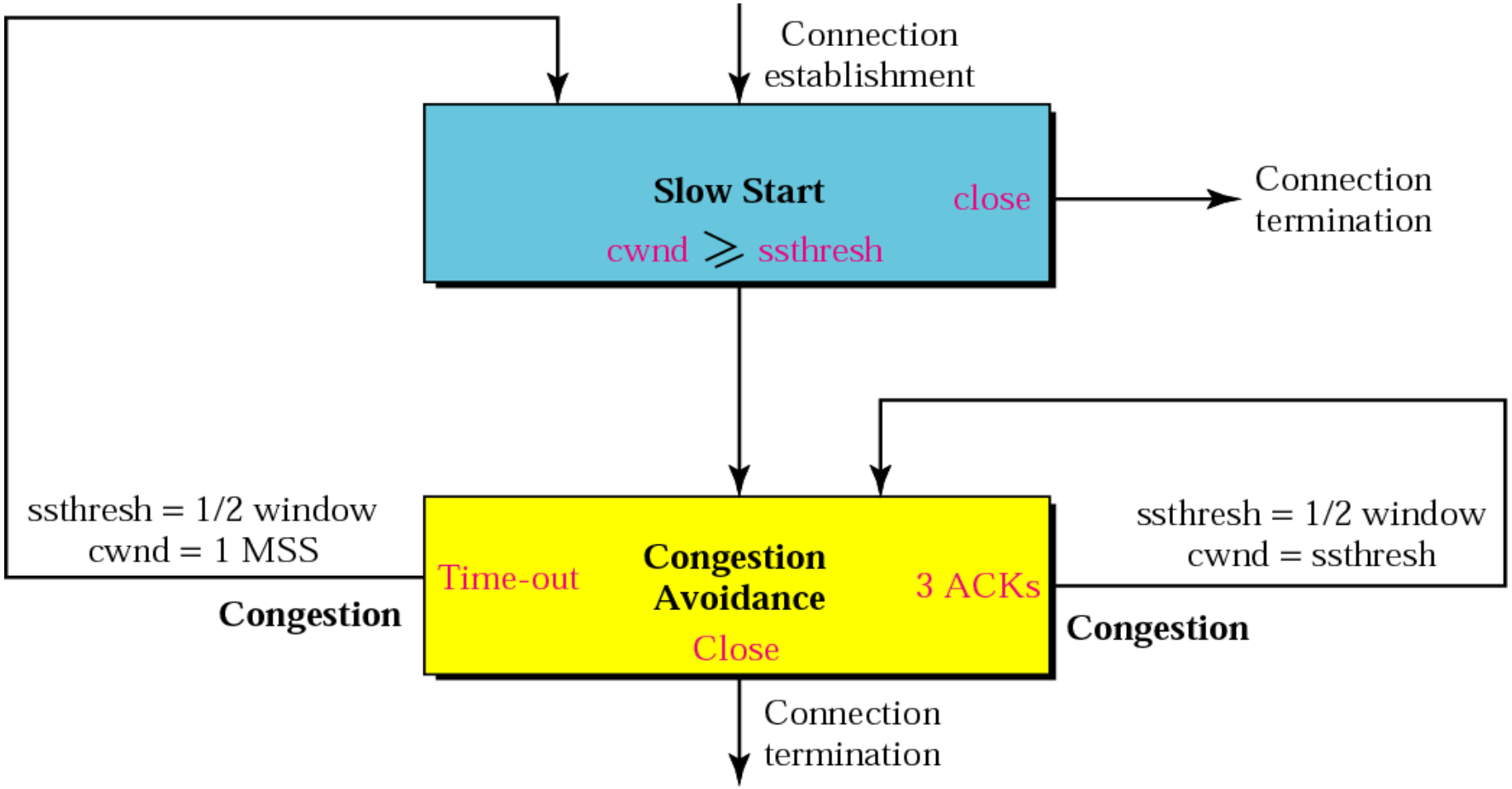


In the congestion avoidance algorithm the size of the congestion window increases additively until congestion is detected.

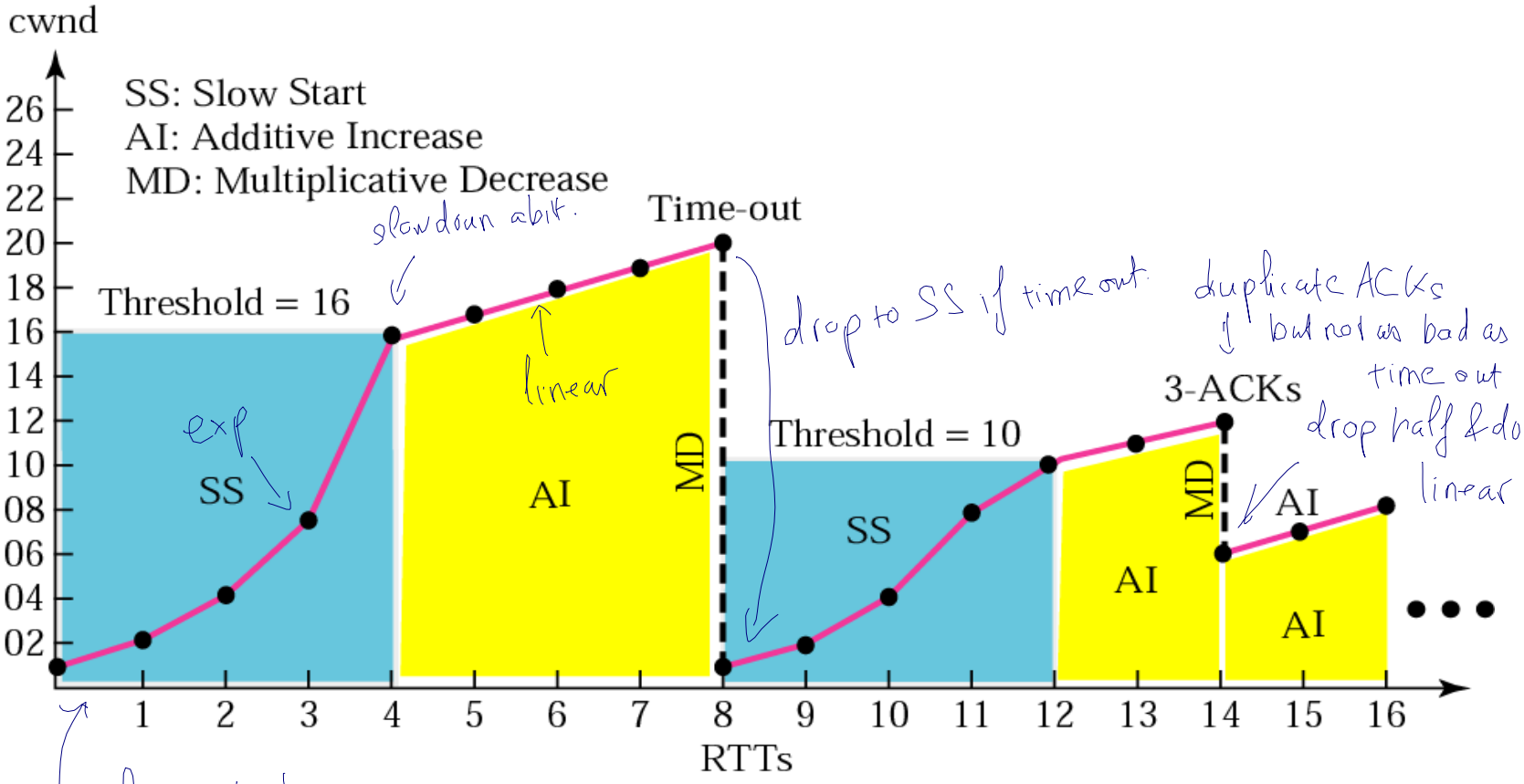
Most implementations react differently to congestion detection:

- *If detection is by time-out, a new slow start phase starts.*
- *If detection is by three ACKs, a new congestion avoidance phase starts.*

TCP congestion policy summary



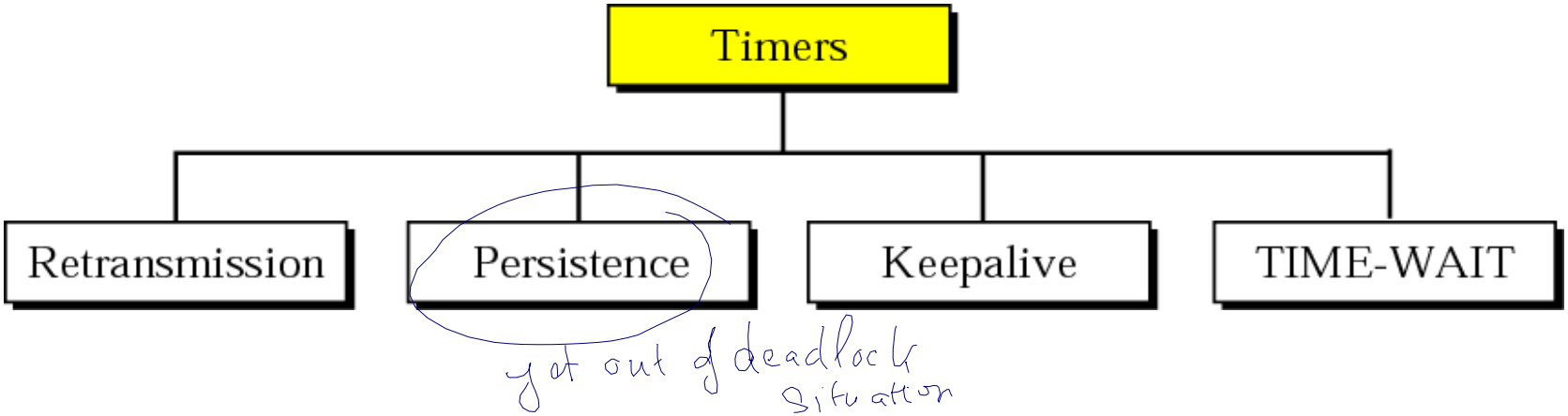
Congestion example



TCP Timers

To perform its operation smoothly, most TCP implementations use at least four timers.

- Retransmission Timer
- Persistence Timer
- Keepalive Timer
- TIME-WAIT Timer



TCP Timers

- TCP maintains four (4) timers for each connection:
  - **Retransmission Timer:**
    - The timer is started during a transmission. A timeout causes a retransmission
  - **Persist Timer**
    - Ensures that window size information is transmitted even if no data is transmitted
  - **Keepalive Timer**
    - Detects crashes on the other end of the connection
  - **2MSL Timer**
    - Measures the time that a connection has been in the TIME_WAIT state
    - **MSL – Maximum Segment Lifetime**

Retransmission Timer (RT Timer)

Setting the RT timer

- When a segment is sent and RT timer is not running, start RT timer with RTO value
- Turn off RT timer, when all data is acknowledged
- When an ACK is received for new data, reset the RT timer to RTO value

RT timer expires

- Retransmit the earliest segment that has not been acknowledged
- Double value of RTO (see Karn's rule)
- Start the RT timer with RTO value

TCP Round Trip Time Calculation

- RFC 2988 " Computing TCP's Retransmission Timer"

$$RTT_{new} = (\alpha * RTT_{old}) + ((1- \alpha) * RTT_{measured})$$

	RTT					
	0	1	2	3	4	5
α	1.500	2.000	2.500	3.000	3.500	4.000
1	1.500	1.500	1.500	1.500	1.500	1.500
2	1.500	1.625	1.844	2.133	2.475	2.856
3	1.500	1.750	2.125	2.563	3.031	3.516
4	1.500	1.875	2.344	2.836	3.334	3.833
5	1.500	2.000	2.500	3.000	3.500	4.000

Acknowledgement Ambiguity

- Problem: Cannot distinguish original segment from retransmitted segments when receiving an acknowledgement.
- This affects the correctness of RTT calculation.
- Karn's Algorithm
  - Do not measure RTT for any segments that are retransmitted
  - Timer Backoff – Increase retransmission time for every retransmitted packet
  - Keep the longer retransmission time until a valid RTT can be measured on a segment that is sent and acked without retransmission

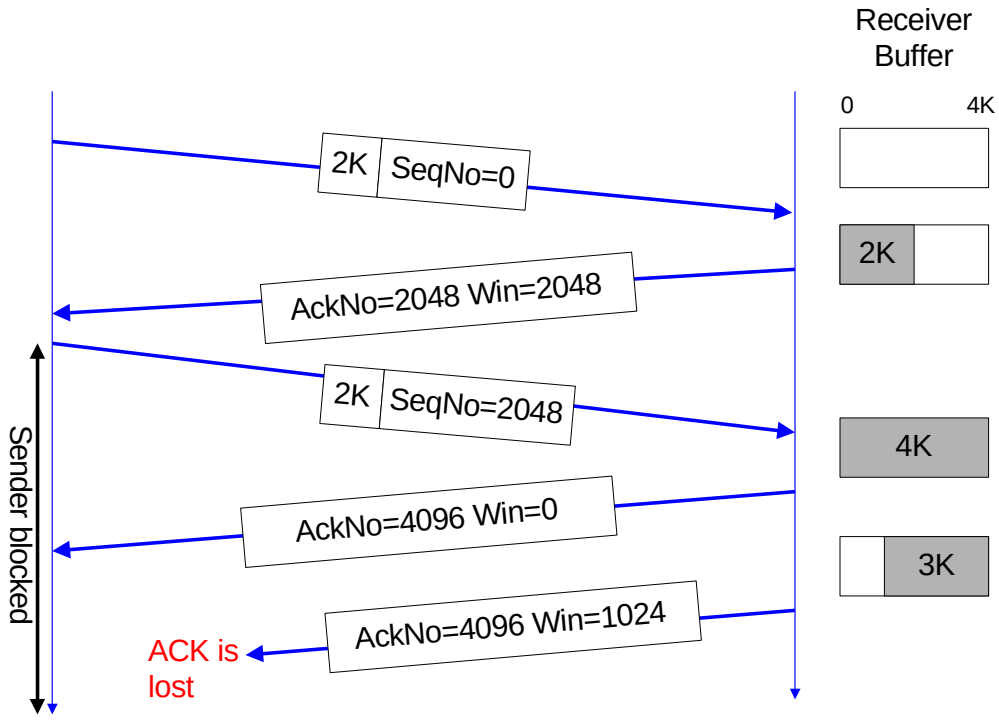
TCP Persist Timer

- Assume the window size goes down to zero and the ACK that opens the window gets lost

If ACK (see figure) is lost, both sides are blocked.

Persist Timer:

Forces that the sender periodically queries the receiver about its window size (window probes)



TCP Persist Timer

- The persist timer is started by the sender when the sliding window is zero

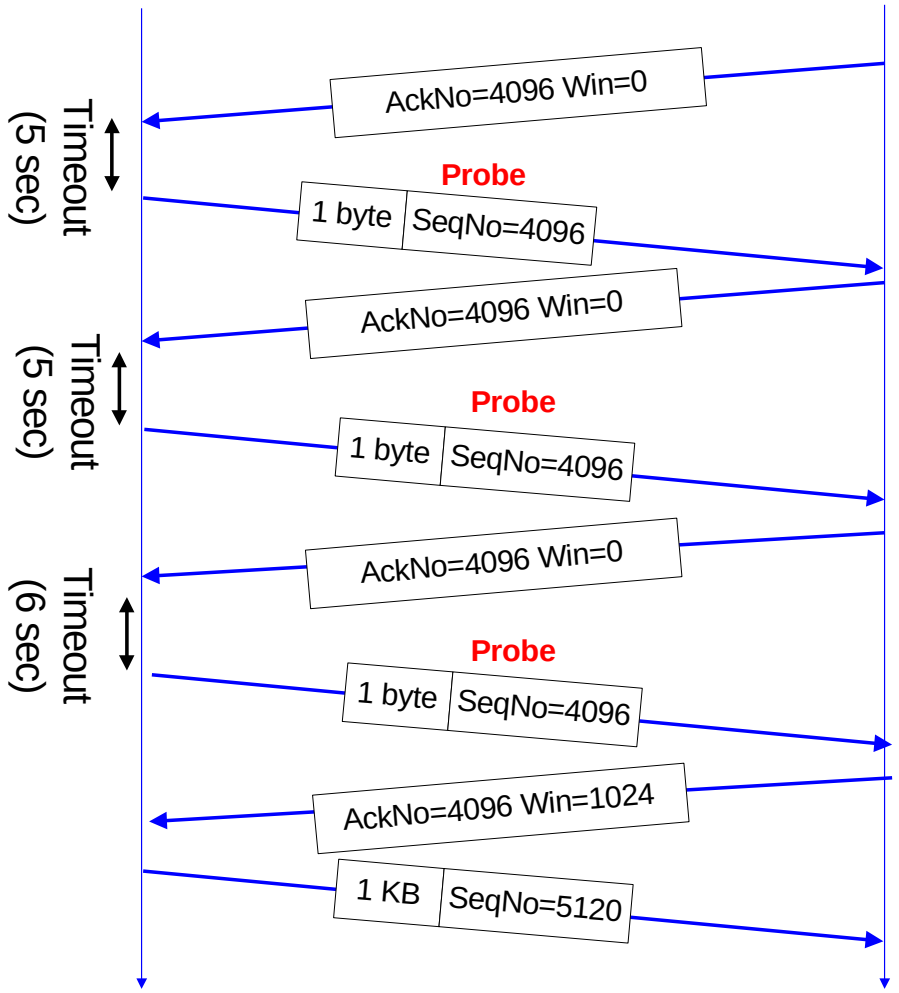
Persist timer uses exponential backoff (initial value is 1.5 seconds) rounded to the range [5 sec, 60sec]
So the time interval between timeouts are at:

5, 5, 6, 12, 24, 48, 60, 60, ...

The window probe packet contains one byte of data (TCP can do this even if the window size is zero)

TCP Persist Timer

deadlock



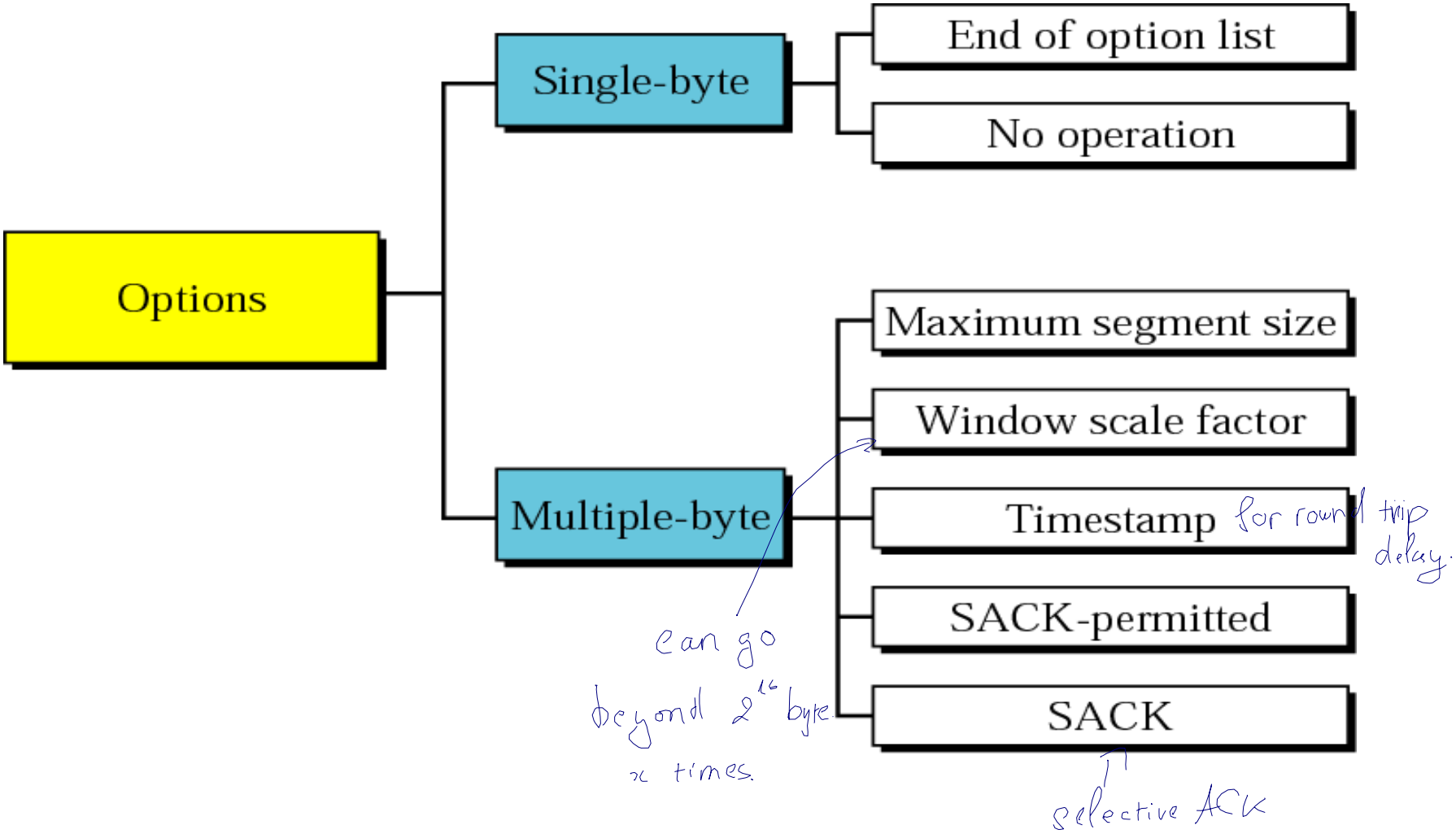
TCP Keepalive Timer

- When a TCP connection has been idle for a long time (1 min – 2 hours), a **Keepalive timer** reminds a station to check if the other side is still there.
- A segment wihtout data is sent if the connection has been idle for 2 hours
- Assume a probe has been sent from **A** to **B**:

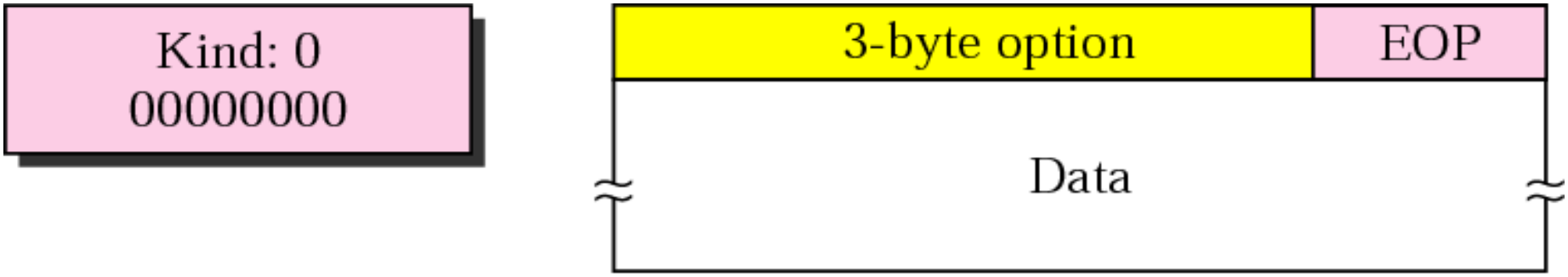
(1) **B** is up and running:	**B** responds with an ACK
(2) **B** has crashed and is down:	**A** will send 10 more probes, each 75 seconds apart. If **A** does not get a response, it will close the connection
(3) **B** has rebooted:	**B** will send a RST segment
(4) **B** is up, but unreachable:	Looks to **A** the same as (2)

TCP Options

The TCP header can have up to 40 bytes of optional information.



End-of-option option

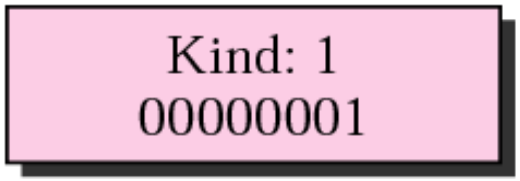


a. End of option list

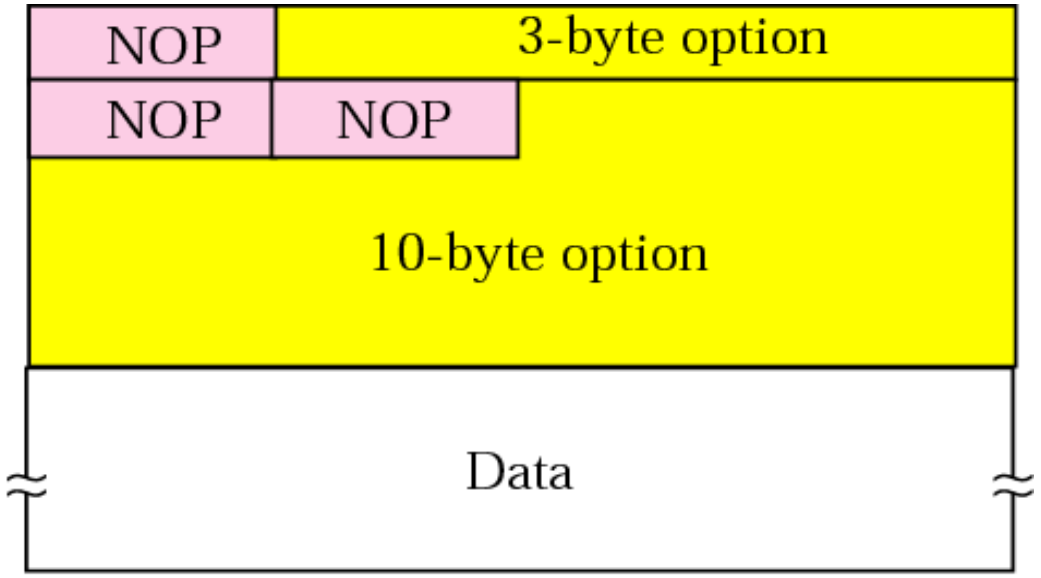
b. Used for padding

EOP can be used only once.

No-operation option



a. No operation option

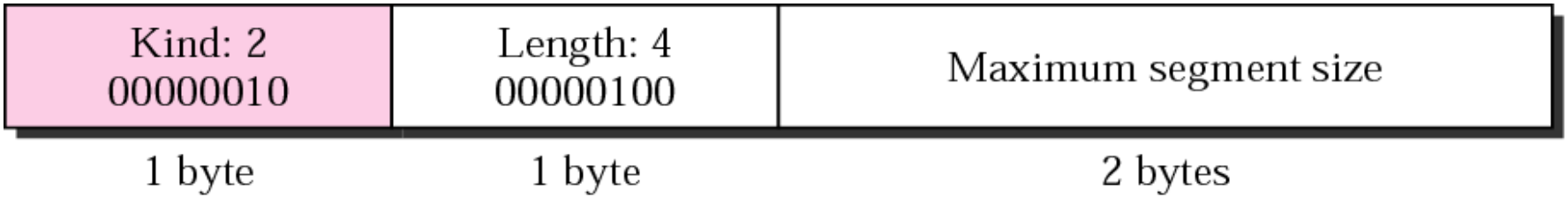


b. Used to align beginning of an option

An option may begin on any byte boundary. The TCP header must be padded with zeros to make the header length a multiple of 32 bits.

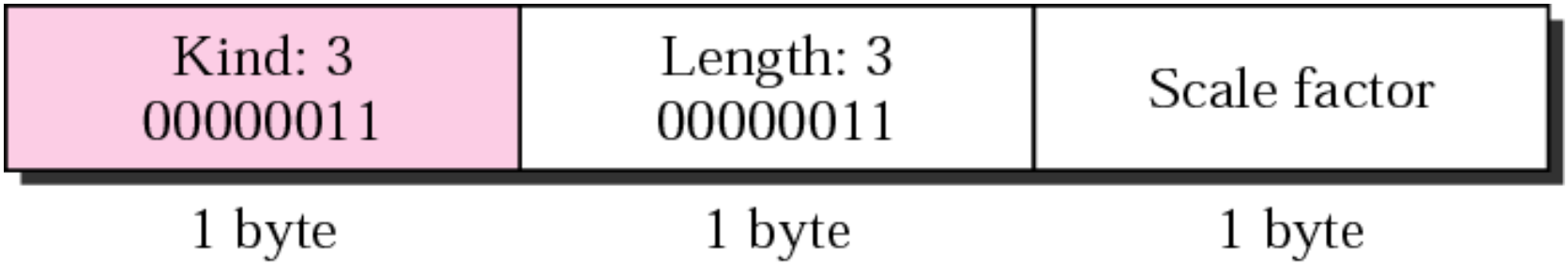
NOP can be used more than once.

Maximum-segment-size option



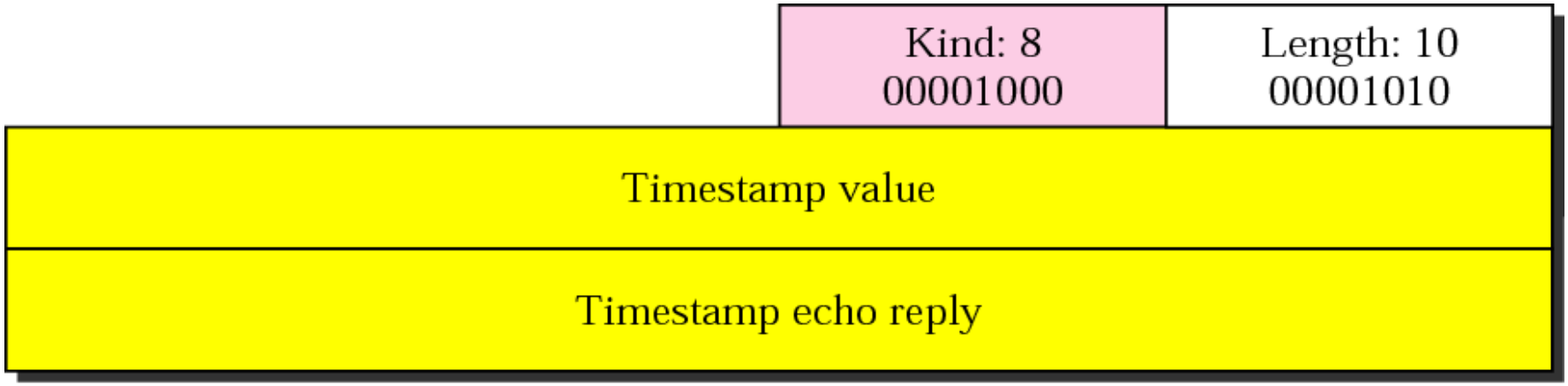
The value of MSS is determined during connection establishment and does not change during the connection.

Window-scale-factor option



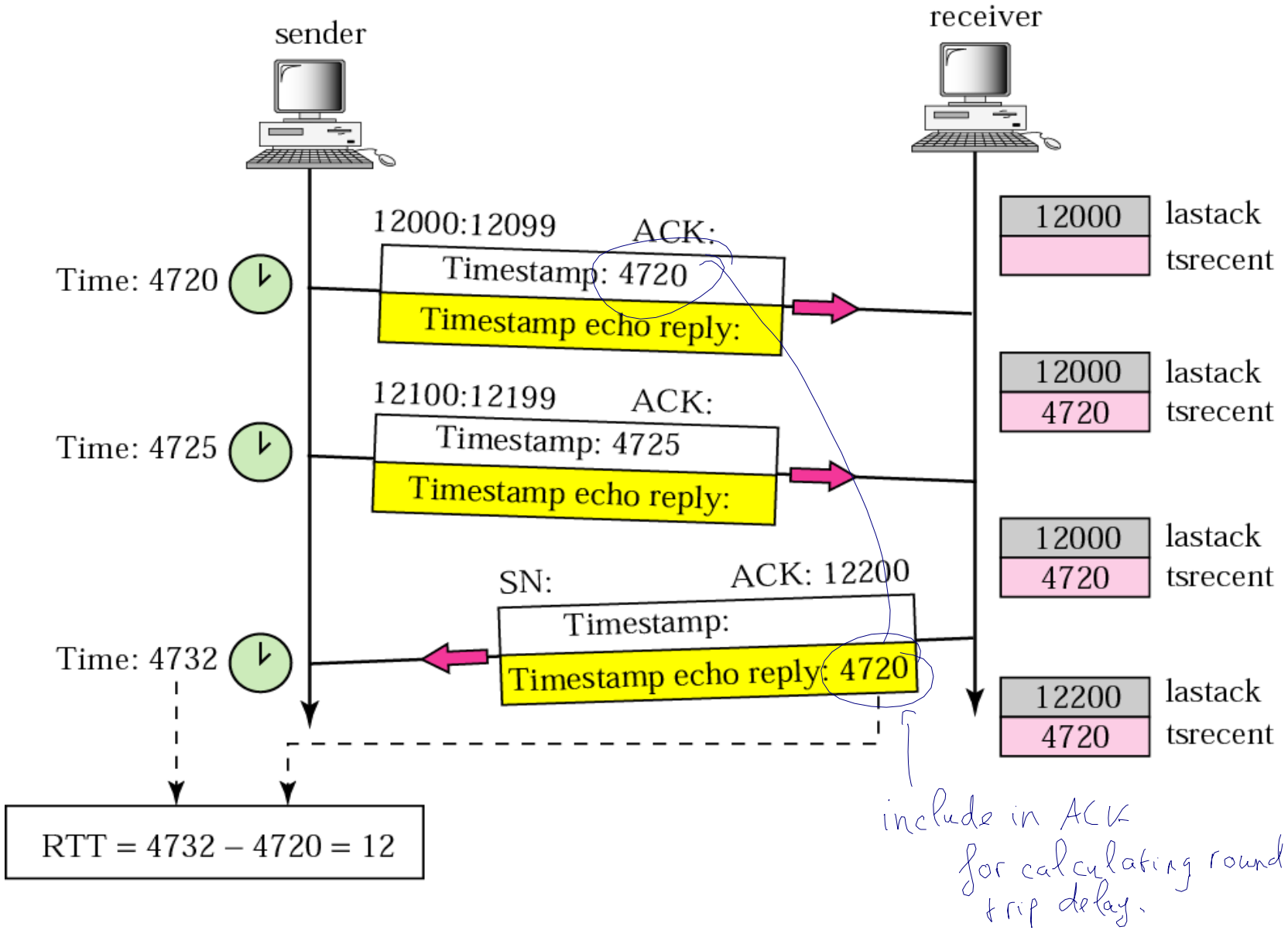
The value of the window scale factor can be determined only during connection establishment; it does not change during the connection.

Timestamp option



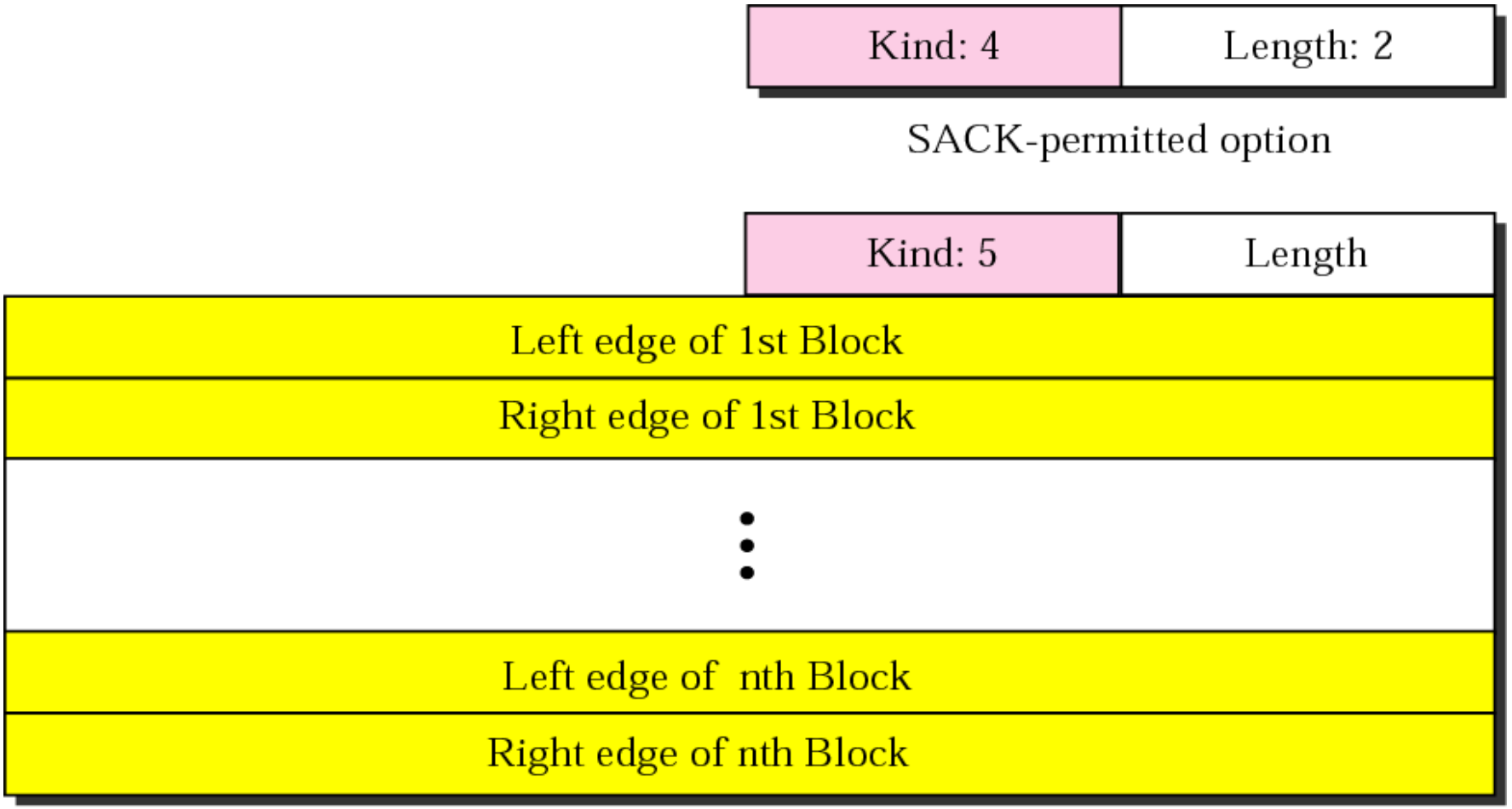
One application of the timestamp option is the calculation of round trip time (RTT).

Example 12


sender
receiver
12000:12099
ACK:
Timestamp: 4720
Timestamp echo reply:
Time: 4720
12100:12199
ACK:
Timestamp: 4725
Timestamp echo reply:
Time: 4725
SN:
ACK: 12200
Timestamp:
Timestamp echo reply: 4720
Time: 4732
RTT = 4732 – 4720 = 12
12000
lastack
tsrecent
12000
4720
lastack
tsrecent
12000
4720
lastack
tsrecent
12200
4720
lastack
tsrecent
include in ACK
for calculating round
trip delay.

The timestamp option can also be used for PAWS – Protection Against Wrapping Sequence Number

SACK

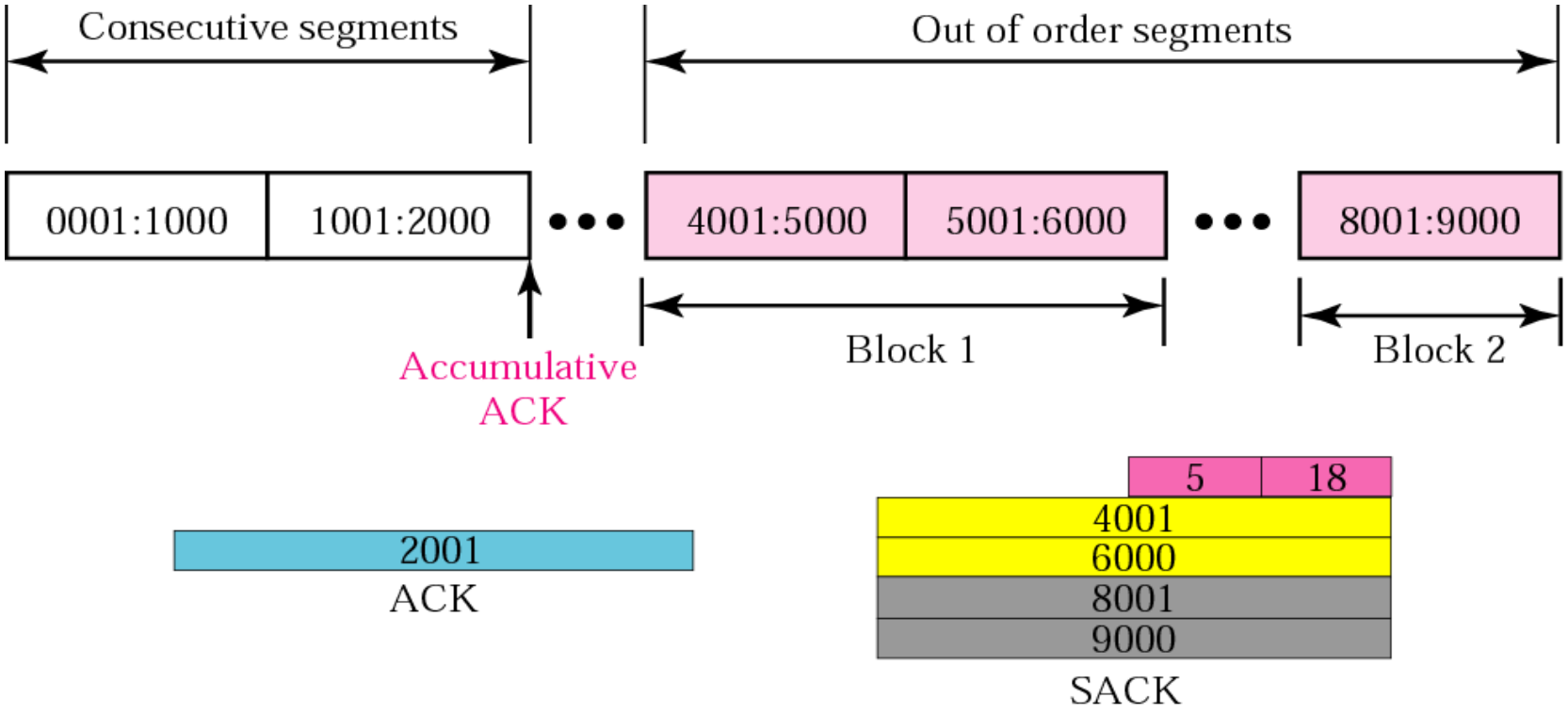

Kind: 4
Length: 2
SACK-permitted option
Kind: 5
Length
Left edge of 1st Block
Right edge of 1st Block
Left edge of nth Block
Right edge of nth Block
SACK option

EXAMPLE 13

Let us see how the SACK option is used to list out-of-order blocks. In Figure 12.48 an end has received five segments of data.

The first and second segments are in consecutive order. An accumulative acknowledgment can be sent to report the reception of these two segments. Segments 3, 4, and 5, however, are out of order with a gap between the second and third and a gap between the fourth and the fifth. An ACK and a SACK together can easily clear the situation for the sender. The value of ACK is2001, which means that the sender need not worry about bytes 1 to 2000. The SACK has two blocks. The first block announces that bytes 4001 to 6000 have arrived out of order. The second block shows that bytes 8001 to 9000 have also arrived out of order. This means that bytes 2001 to 4000 and bytes 6001 to 8000 are lost or discarded. The sender can resend only these bytes.

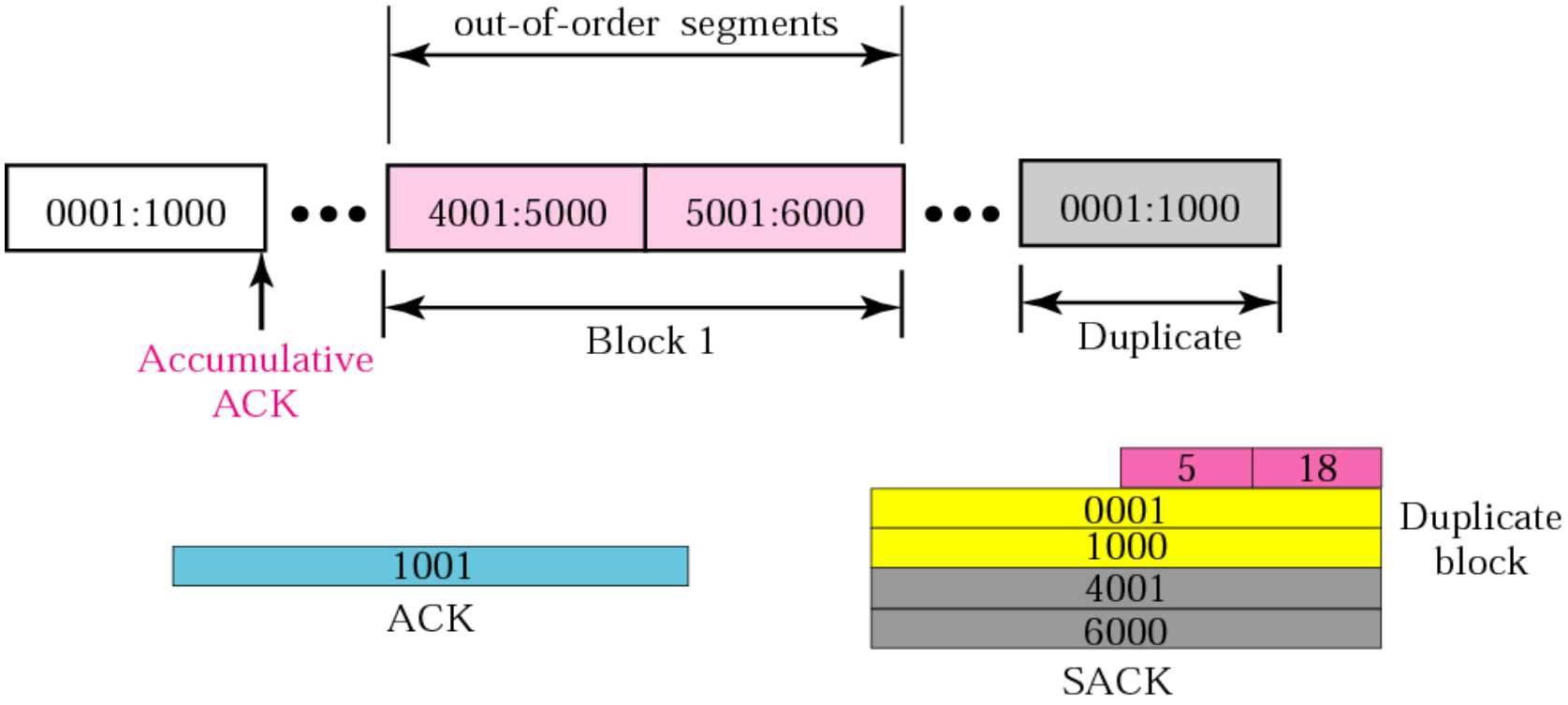
Example 13


Consecutive segments
Out of order segments
0001:1000
1001:2000
4001:5000
5001:6000
8001:9000
Accumulative ACK
Block 1
Block 2
2001
ACK
5
18
4001
6000
8001
9000
SACK

EXAMPLE 14

The example in Figure 12.49 shows how a duplicate segment can be detected with a combination of ACK and SACK. In this case, we have some out-of-order segments (in one block) and one duplicate segment. To show both out-of-order and duplicate data, SACK uses the first block, in this case, to show the duplicate data and other blocks to show out-of-order data. Note that only the first block can be used for duplicate data. The natural question is how the sender, when it receives these ACK and SACK values knows that the first block is for duplicate data (compare this example with the previous example). The answer is that the bytes in the first block are already acknowledged in the ACK field; therefore, this block must be a duplicate.

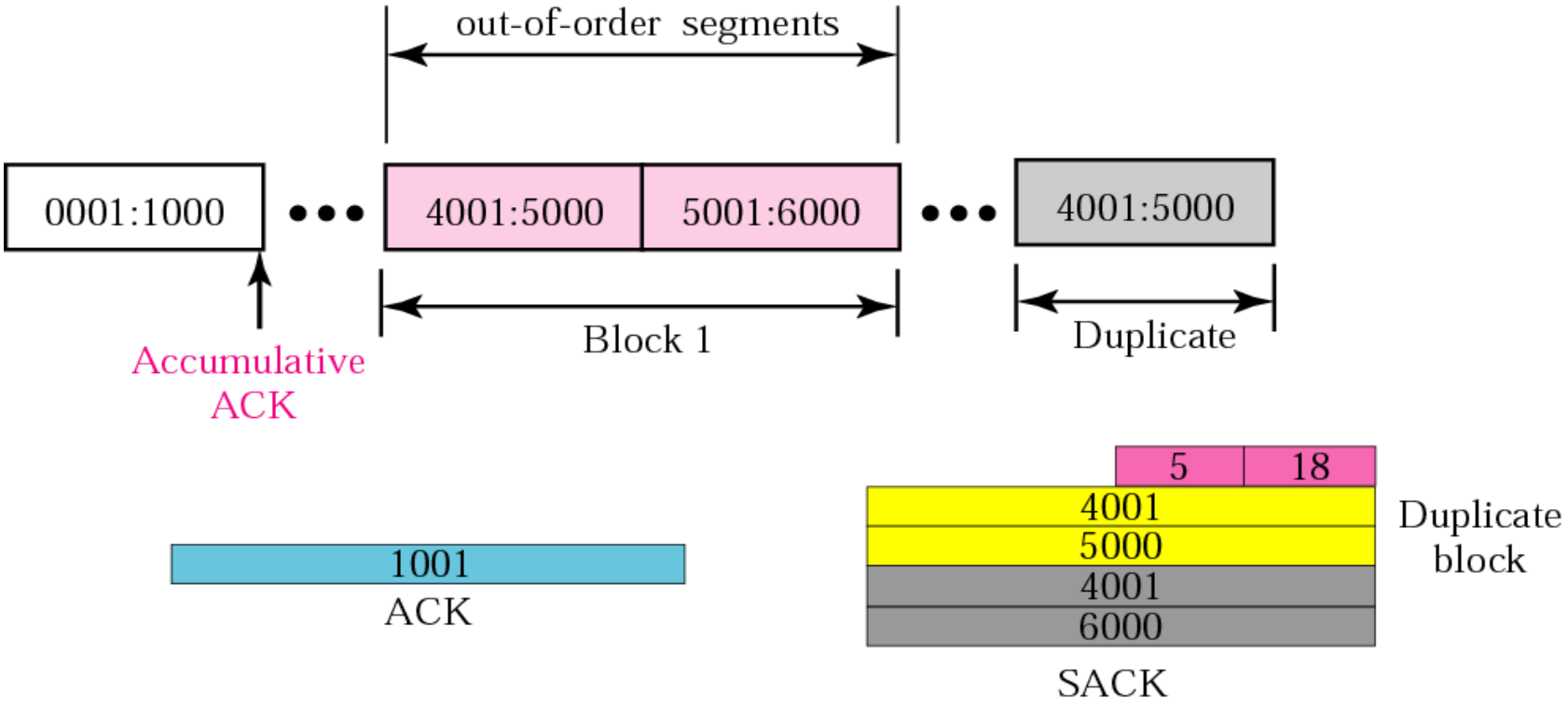
Example 14


out-of-order segments
0001:1000
4001:5000
5001:6000
0001:1000
Accumulative ACK
Block 1
Duplicate
1001
ACK
5
18
0001
1000
4001
6000
SACK
Duplicate block

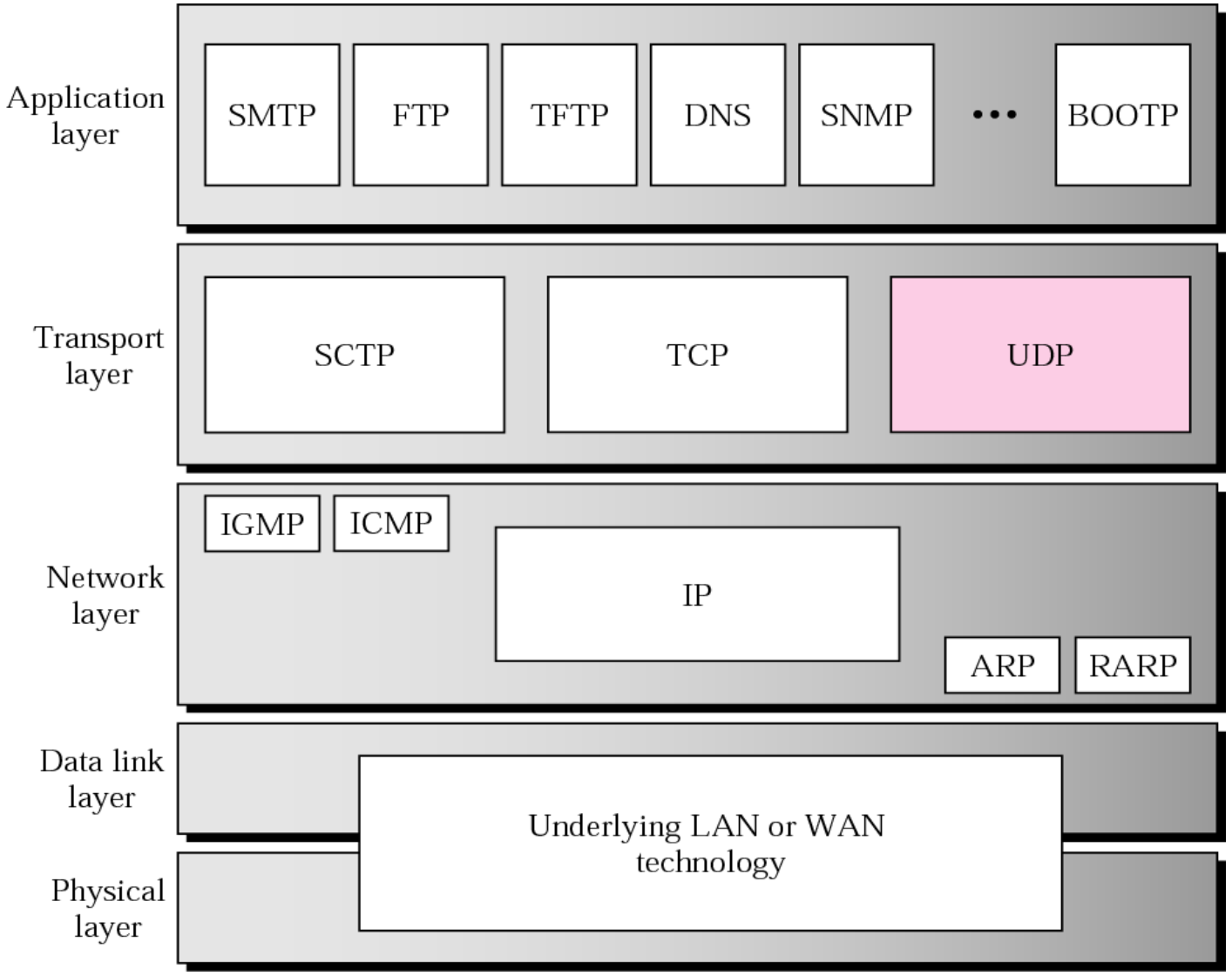
EXAMPLE 15

The example in Figure 12.50 shows what happens if one of the segments in the out-of-order section is also duplicated. In this example, one of the segments (4001:5000) is duplicated. The SACK option announces this duplicate data first and then the out-of-order block. This time, however, the duplicated block is not yet acknowledged by ACK, but because it is part of the out-of-order block (4001:5000 is part of 4001:6000), it is understood by the sender that it defines the duplicate data.

Example 15


out-of-order segments
0001:1000
4001:5000
5001:6000
4001:5000
Accumulative ACK
Block 1
Duplicate
1001
ACK
5
18
4001
5000
4001
6000
SACK
Duplicate block

Position of UDP in the TCP/IP protocol suite

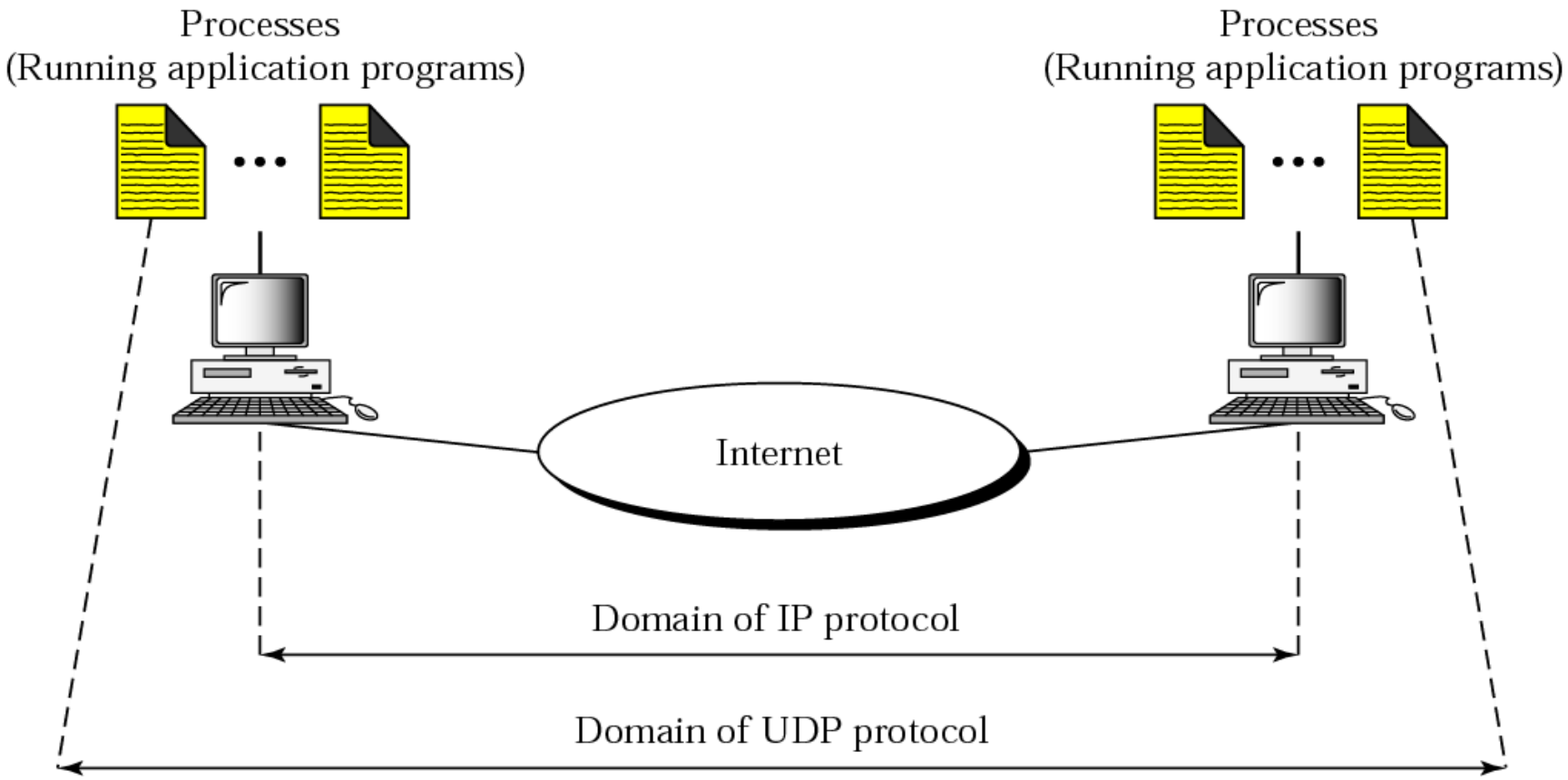


PROCESS-TO-PROCESS COMMUNICATIOn

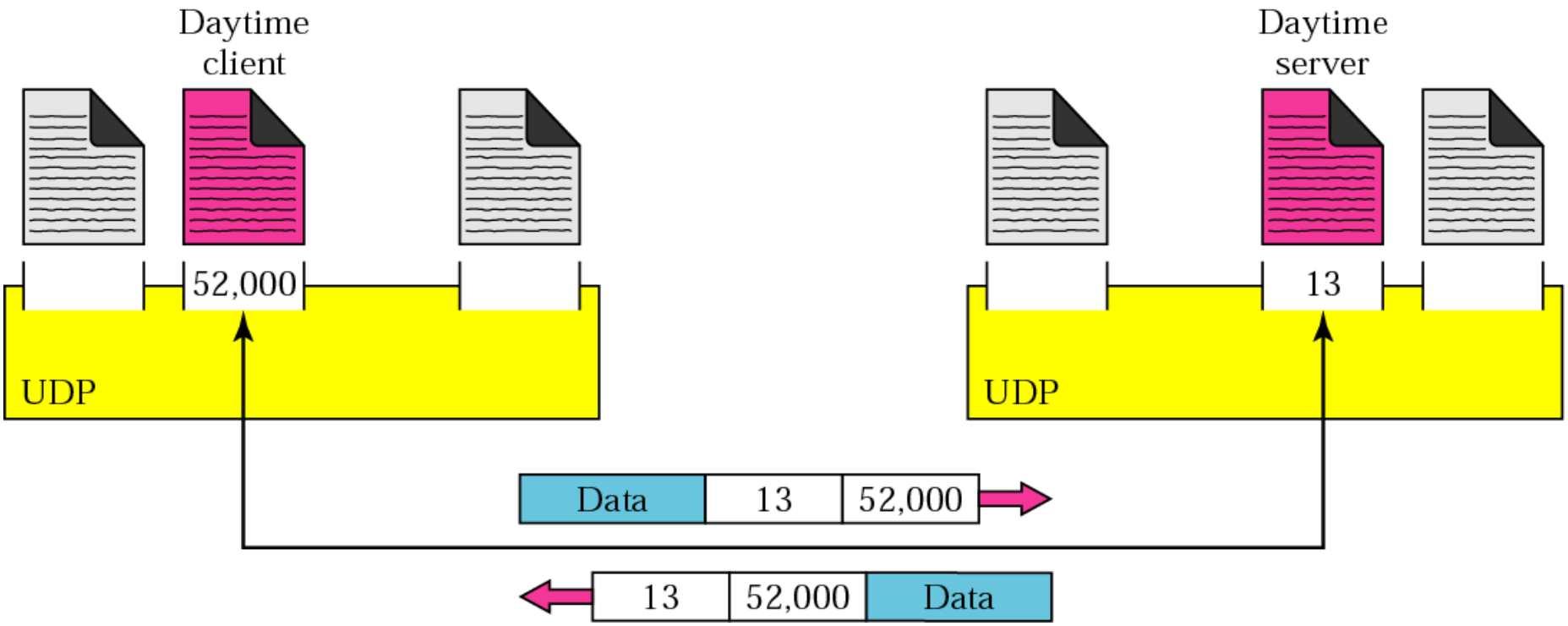
Before we examine UDP, we must first understand host-to-host communication and process-to-process communication and the difference between them.

- *Port Numbers*
- *Socket Addresses*

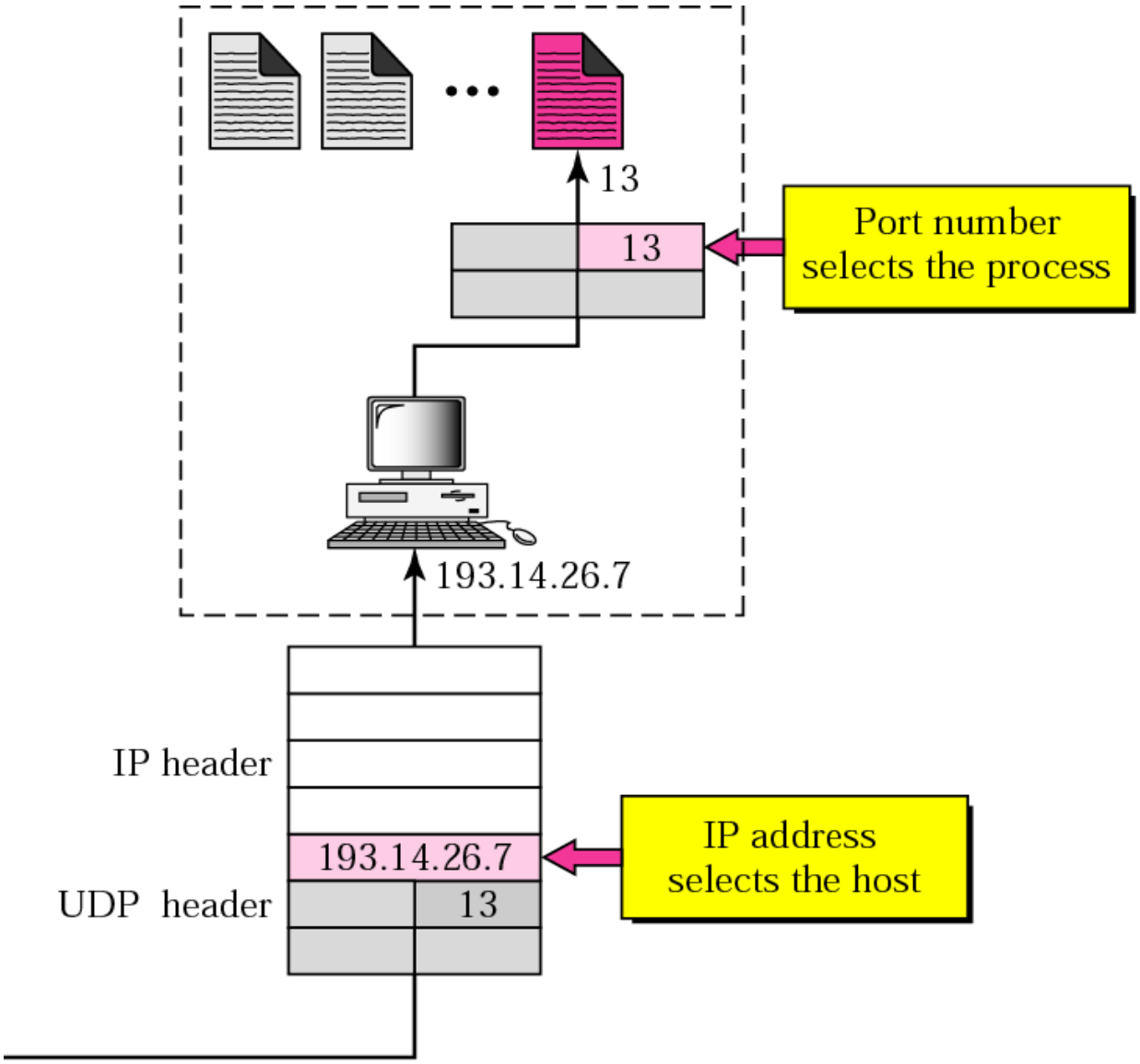
UDP versus IP



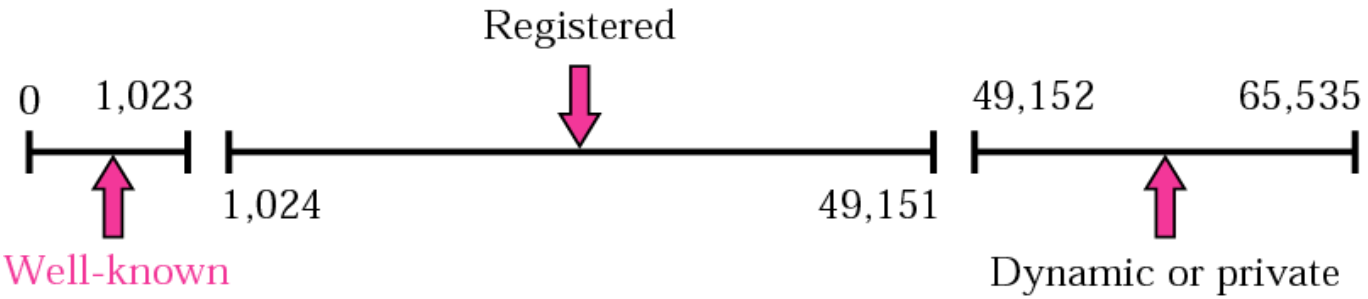
Port numbers



IP addresses versus port numbers



ICANN ranges



The well-known port numbers are less than 1024.

Table 11.1 *Well-known ports used with UDP*

Port	*Protocol*	*Description*
7	Echo	Echoes a received datagram back to the sender
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
53	Nameserver	Domain Name Service
67	Bootps	Server port to download bootstrap information
68	Bootpc	Client port to download bootstrap information
69	TFTP	Trivial File Transfer Protocol
111	RPC	Remote Procedure Call
123	NTP	Network Time Protocol
161	SNMP	Simple Network Management Protocol
162	SNMP	Simple Network Management Protocol (trap)

EXAMPLE 1

In UNIX, the well-known ports are stored in a file called /etc/services. Each line in this file gives the name of the server and the well-known port number. We can use the grep utility to extract the line corresponding to the desired application. The following shows the port for TFTP. Note TFTP can use port 69 on either UDP or TCP.

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

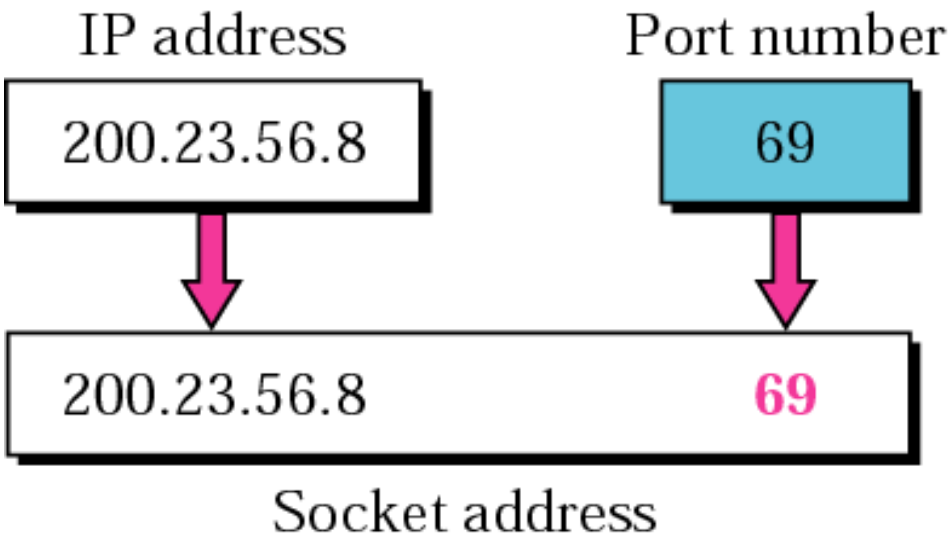
See Next Slide

EXAMPLE 1 (CONTINUED)

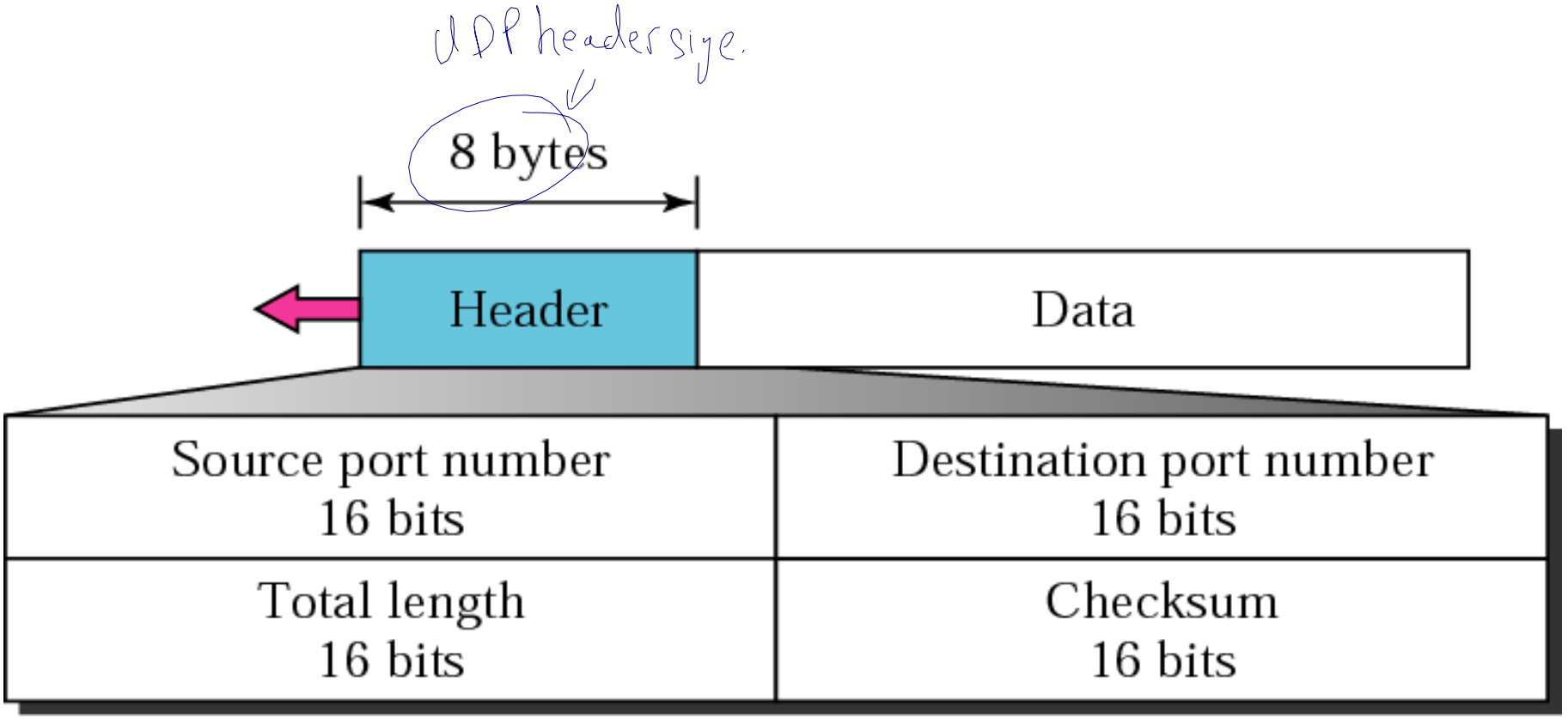
SNMP uses two port numbers (161 and 162), each for a different purpose, as we will see in Chapter 21.

```
$ grep snmp /etc/services
snmp        161/tcp        #Simple Net Mgmt Proto
snmp        161/udp        #Simple Net Mgmt Proto
snmptrap    162/udp        #Traps for SNMP
```

Socket address



User datagram format



UDP packets are called user datagrams and have a fixed-size header of 8 bytes.

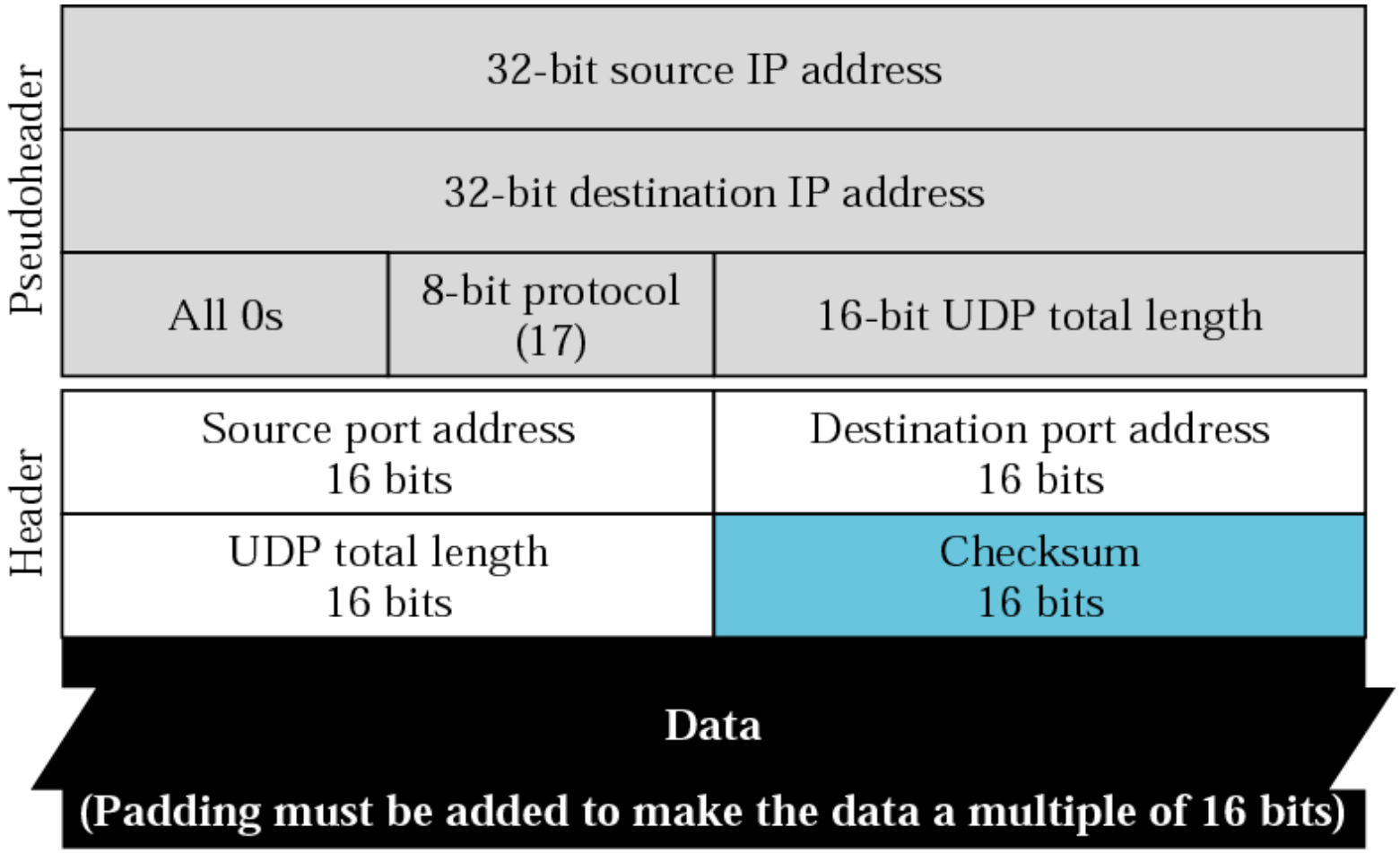
UDP length =
IP length − IP header's length

UDP Checksum

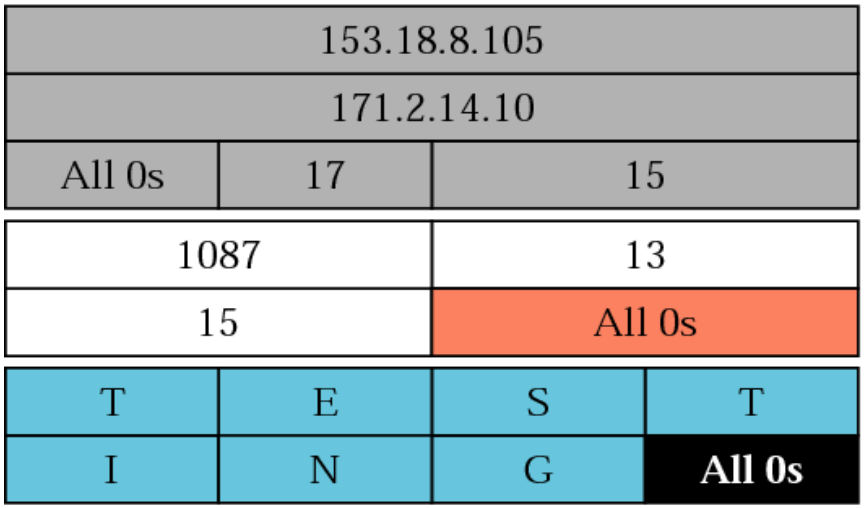
UDP checksum calculation is different from the one for IP and ICMP. Here the checksum includes three sections: a pseudoheader, the UDP header, and the data coming from the application layer.

- Checksum Calculation at Sender
- Checksum Calculation at Receiver
- Optional Use of the Checksum

Pseudoheader for checksum calculation



Checksum calculation of a simple UDP user datagram

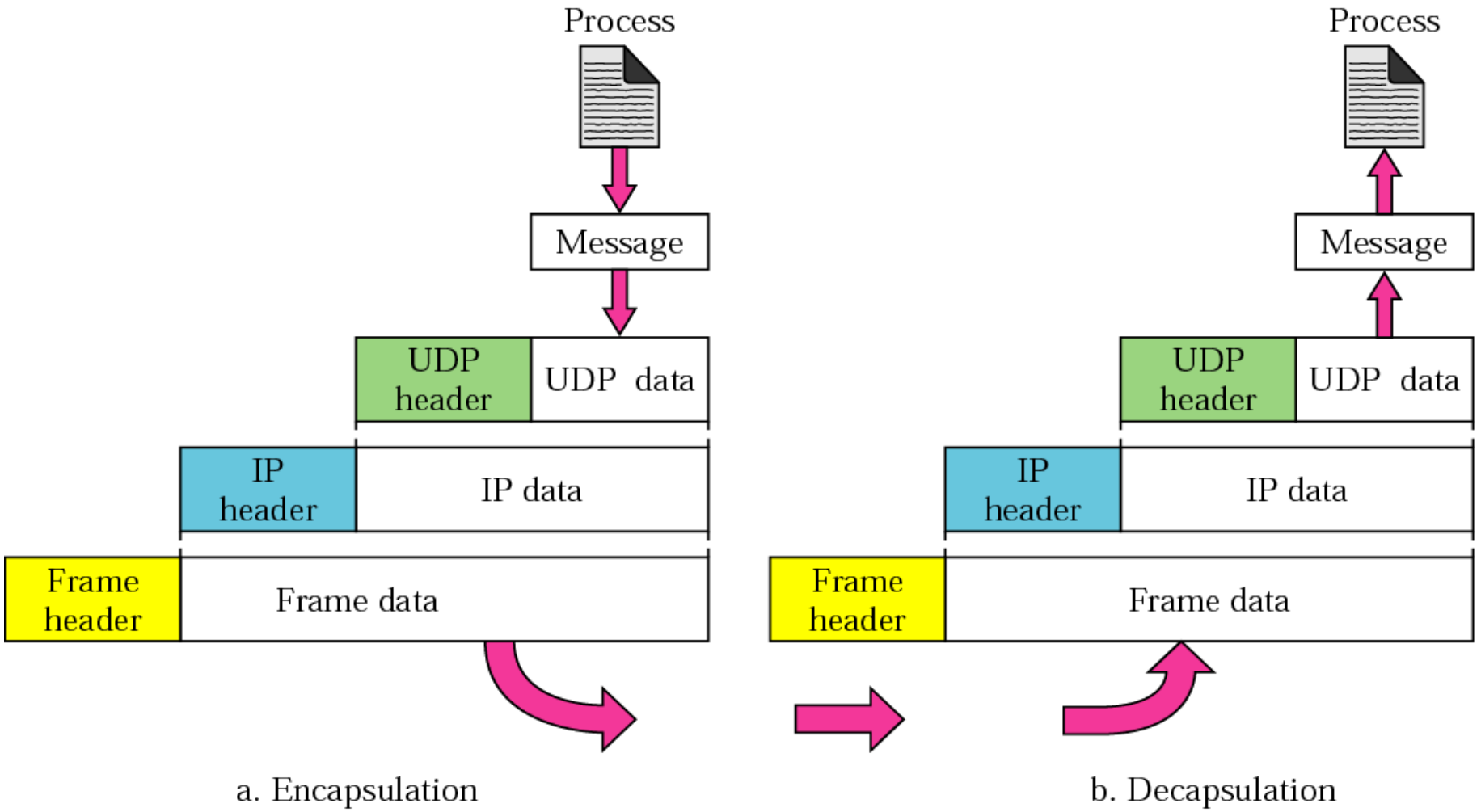


10011001	00010010	153.18
00001000	01101001	8.105
10101011	00000010	171.2
00001110	00001010	14.10
00000000	00010001	0 and 17
00000000	00001111	15
00000100	00111111	1087
00000000	00001101	13
00000000	00001111	15
00000000	00000000	0 (checksum)
01010100	01000101	T and E
01010011	01010100	S and T
01001001	01001110	I and N
01000111	00000000	G and 0 (padding)
10010110	**11101011**	Sum
01101001	**00010100**	Checksum

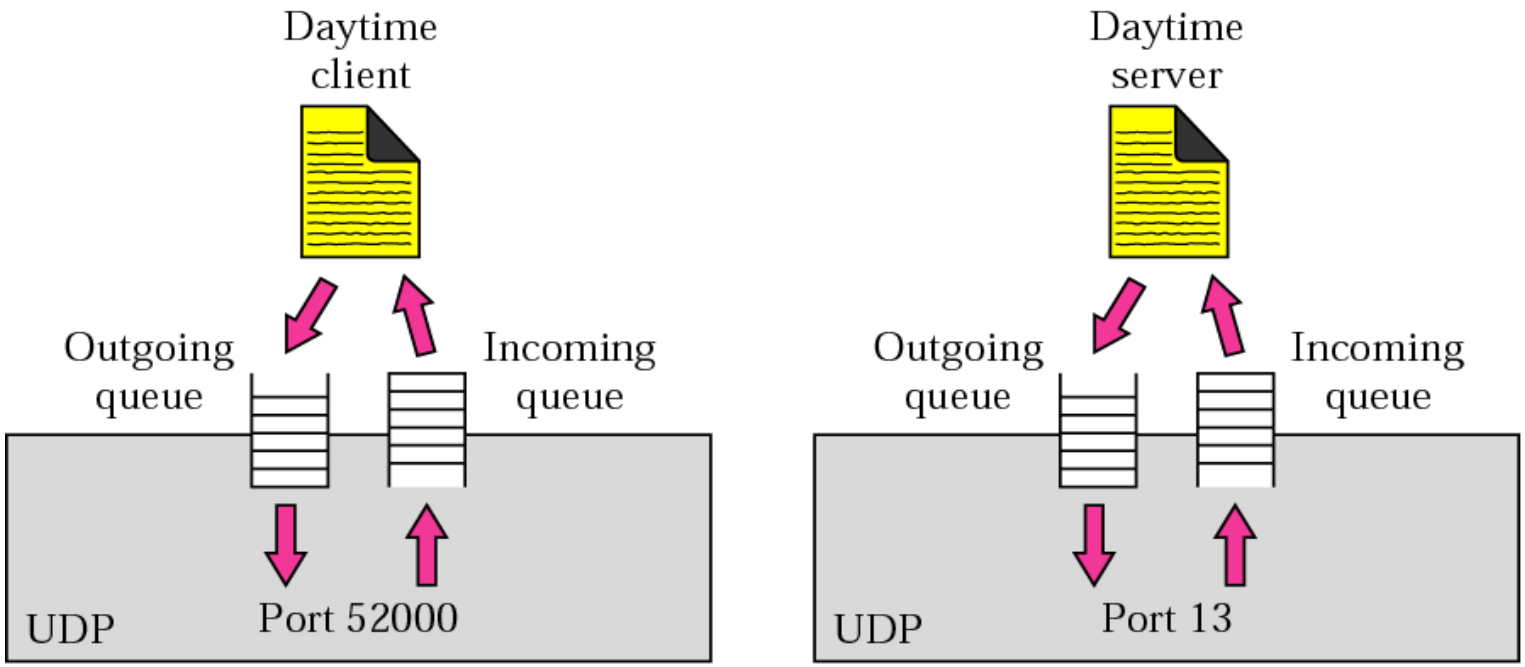
UDP OPERATION

- Connectionless Services
- Flow and Error Control
- Encapsulation and Decapsulation
- Queuing
- Multiplexing and Demultiplexing

Encapsulation and decapsulation



Queues in UDP



Multiplexing and demultiplexing