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Computer Networks

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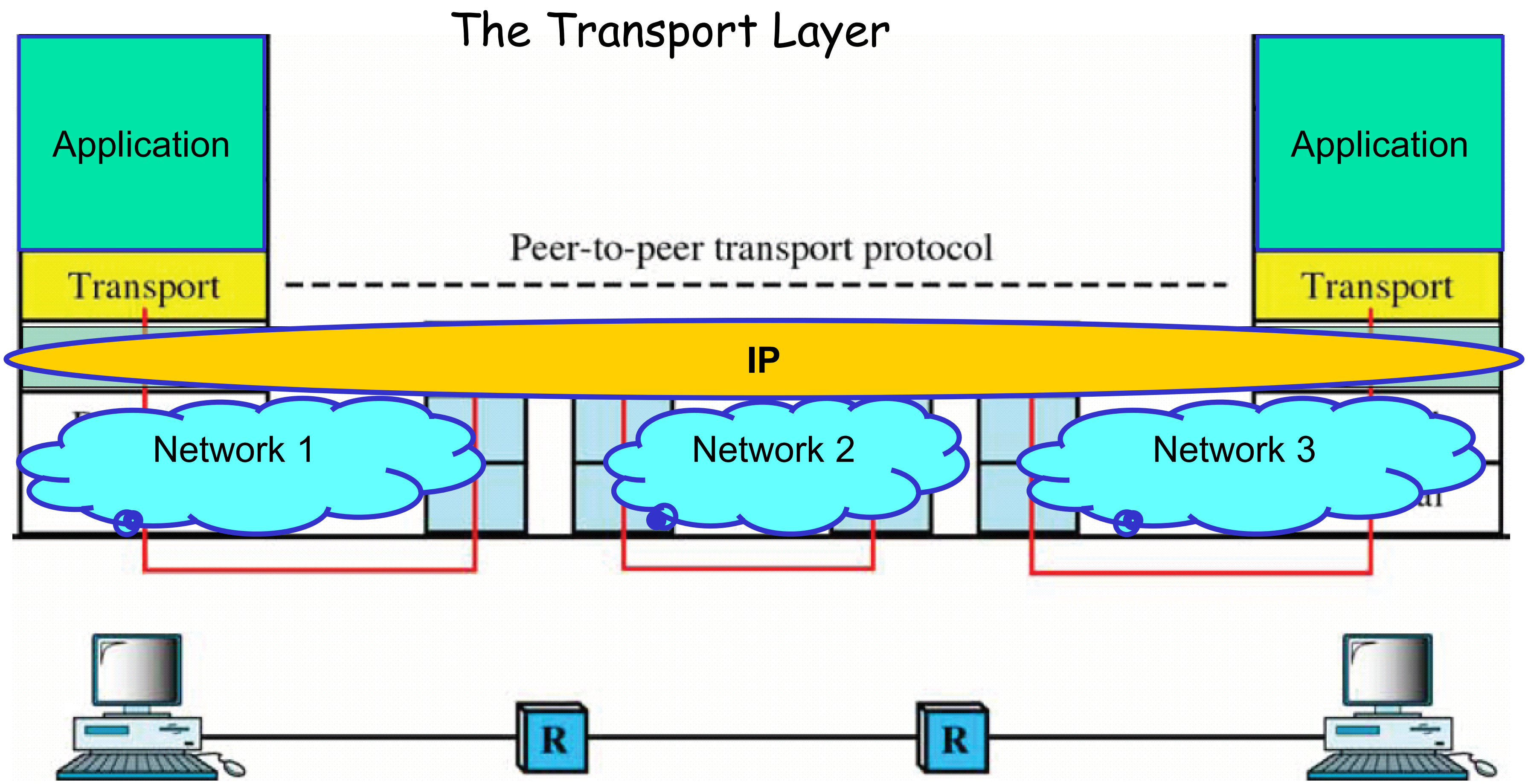
Material with thanks to James F. Kurose, Mosharaf Chowdhury, and other colleagues.

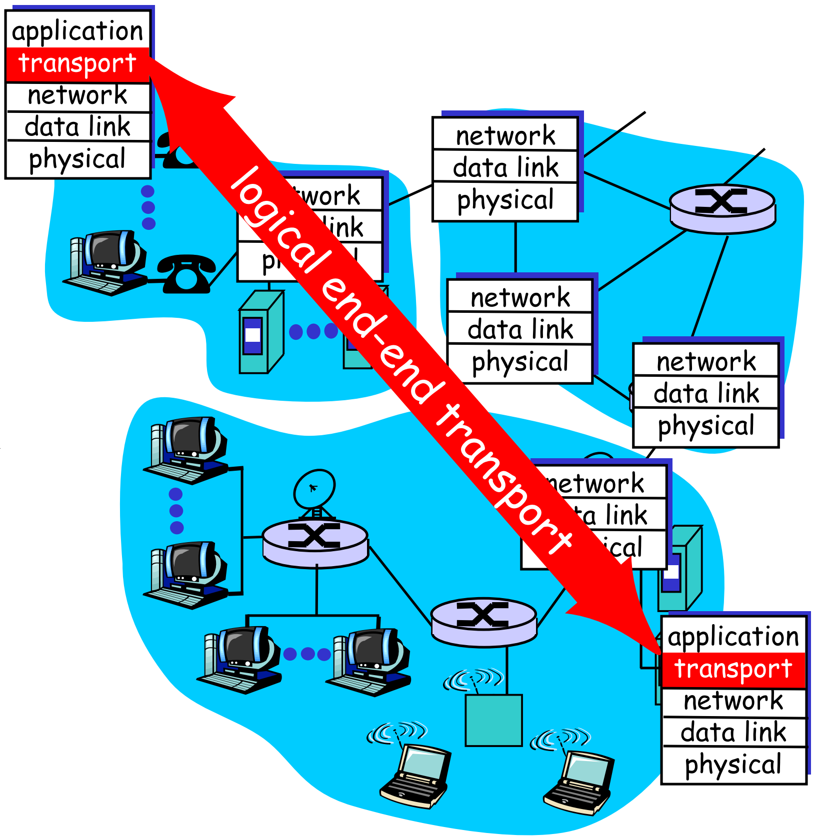
Transport layer basics

Transport Services and Mechanisms



Internet Transport Services

- Provide logical communication between app processes running on different hosts
- Transport protocols run in end systems
  - Send side: breaks app messages into segments, passes to network layer
  - Receive side: reassembles segments into messages, passes to app layer
- More than one transport protocol available to apps
  - Internet: TCP and UDP



Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
  - Need a way to decide which packets go to which applications (multiplexing/demultiplexing)
- IP provides a weak service model (best-effort)
  - Packets can be corrupted, delayed, dropped, reordered, duplicated
  - No guidance on how much traffic to send and when
  - Dealing with this is tedious for application developers

Multiplexing & demultiplexing

- Multiplexing (Mux)
  - Gather and combining data chunks at the source host from different applications and delivering to the network layer
- Demultiplexing (Demux)
  - Delivering correct data to corresponding sockets from multiplexed a stream

Role of the transport layer

- Communication between processes
  - Mux and demux from/to application processes
  - Implemented using *ports*

Role of the transport layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
  - Reliable, in-order data delivery
  - Well-paced data delivery
    - Too fast may overwhelm the network
    - Too slow is not efficient

Role of the transport layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
  - Also SCTP, MPTCP, SST, RDP, DCCP, …

Role of the transport layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist transport protocol
  - Only provides mux/demux capabilities

Role of the transport layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist transport protocol
- TCP offers a reliable, in-order, byte stream abstraction
  - With congestion control, but w/o performance guarantees (delay, b/w, etc.)

Applications and sockets

- Socket: software abstraction for an application process to exchange network messages with the (transport layer in the) operating system

- Transport layer addressing
  - <HostIP, Port>, called a socket

- Two important types of sockets
  - UDP socket: TYPE is SOCK_DGRAM
  - TCP socket: TYPE is SOCK_STREAM

Ports

- 16-bit numbers that help distinguishing apps
  - Packets carry src/dst port no in transport header
  - Well-known (0-1023) and ephemeral ports
- OS stores mapping between sockets and ports
  - Port in packets and sockets in OS
  - For UDP ports (SOCK_DGRAM)
    - OS stores (local port, local IP address) ←→ socket
  - For TCP ports (SOCK_STREAM)
    - OS stores (local port, local IP, remote port, remote IP) ←→ socket

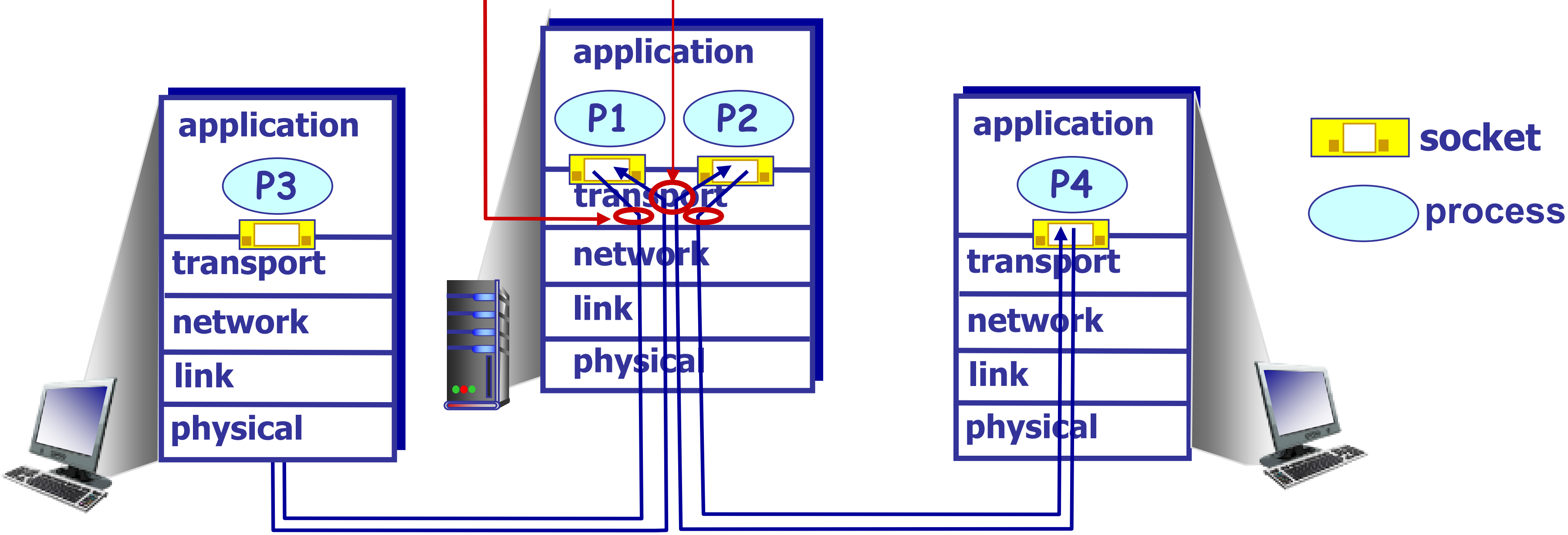
Multiplexing/demultiplexing

multiplexing at sender:

handle data from multiple sockets, add transport header (later used for demultiplexing)

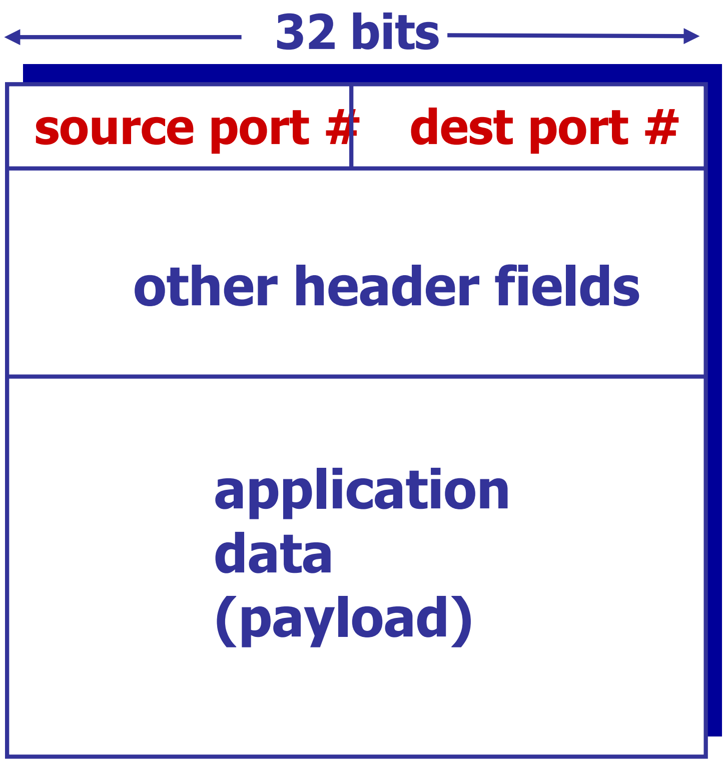
demultiplexing at receiver:

use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- *recall:* created socket has host-local port #:

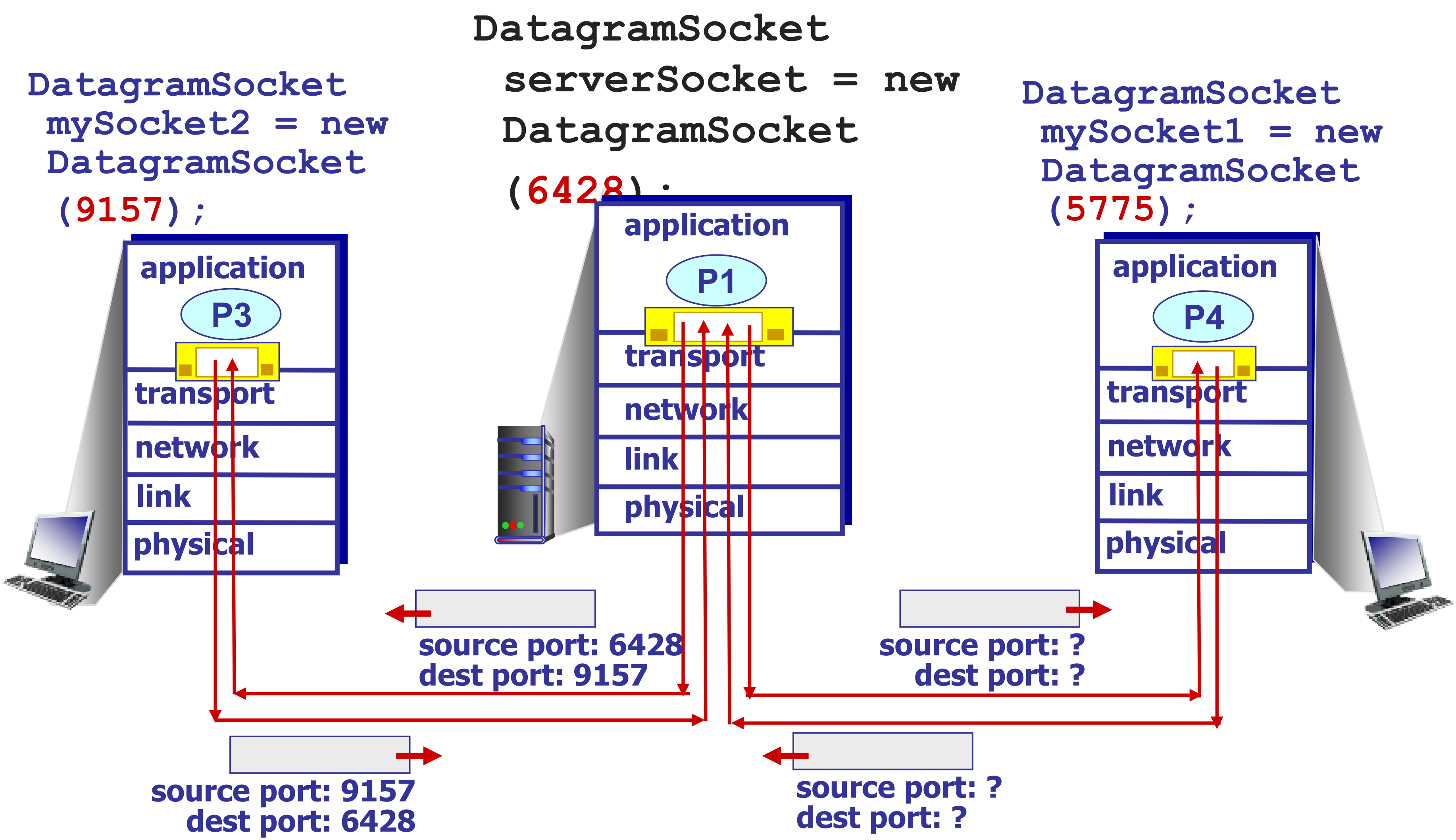
```
DatagramSocket mySocket1
= new DatagramSocket(12534);
```

- ***recall:*** **when creating datagram to send into UDP socket, must specify**
  - **destination IP address**
  - **destination port #**

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connectionless demux: example

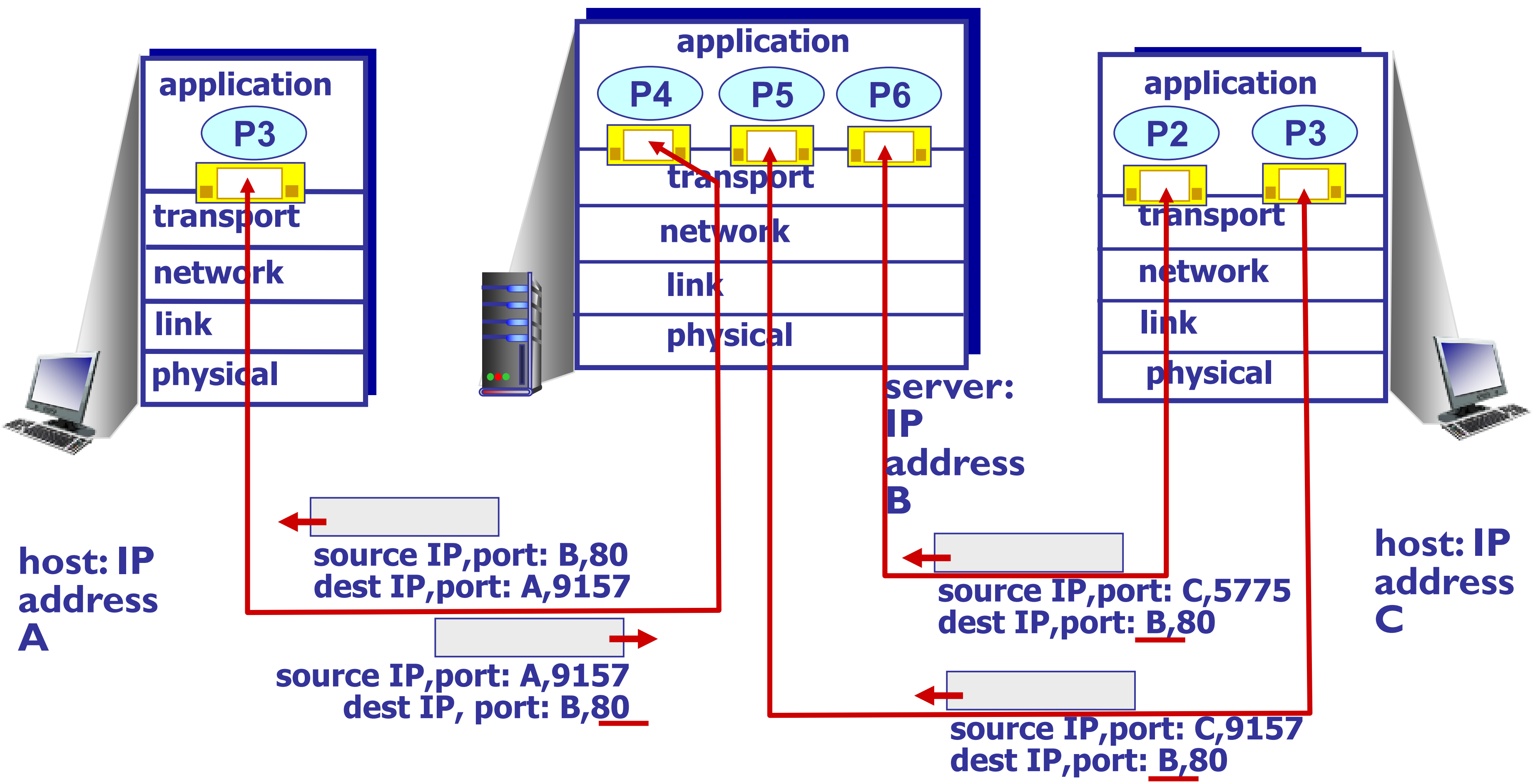


Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket

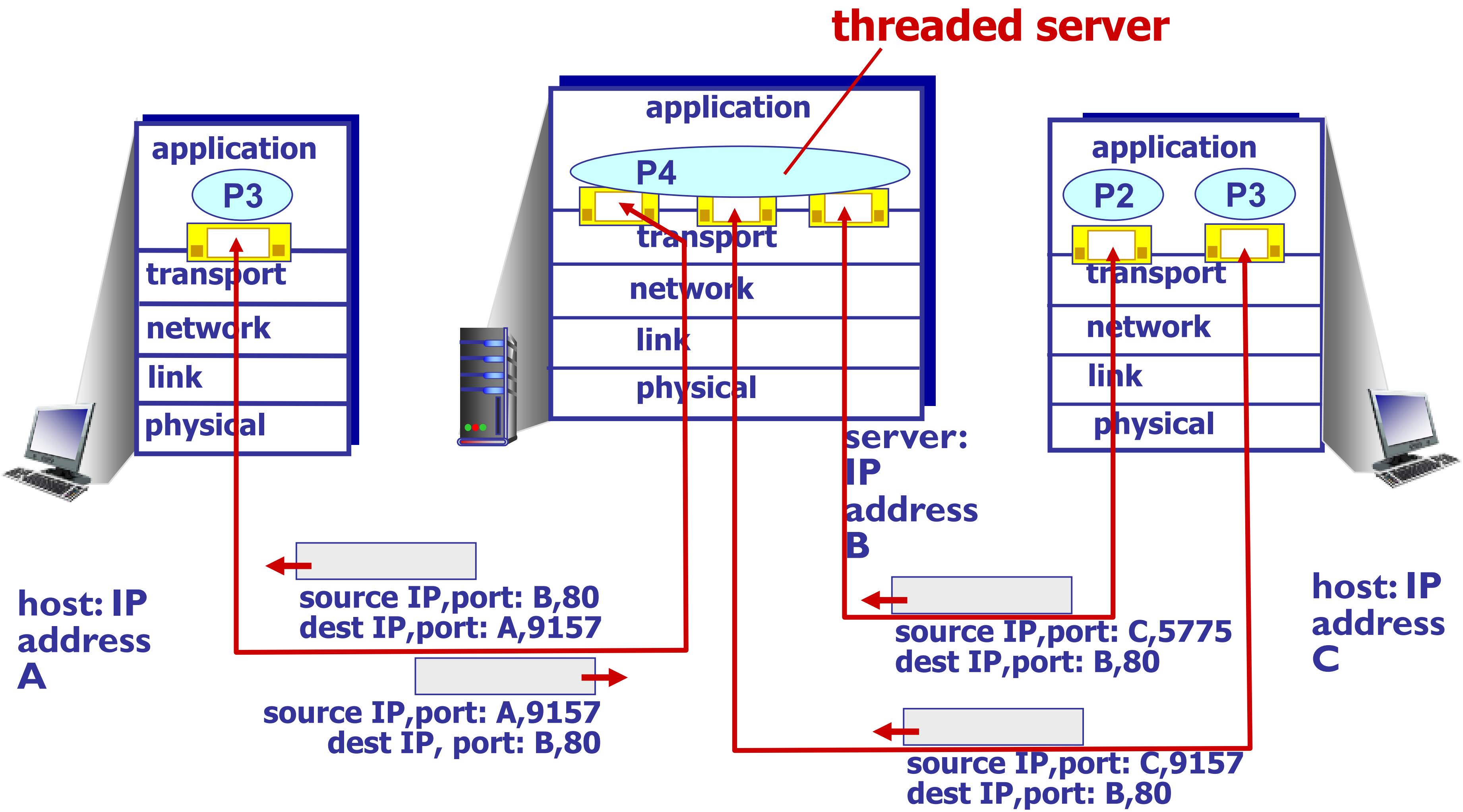
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



three segments, all destined to IP address: B, dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



Design of reliable transport

Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
  - Need a way to decide which packets go to which applications (mux/demux)
- IP provides a weak service model (best-effort)
  - Packets can be corrupted, delayed, dropped, reordered, duplicated
  - No guidance on how much traffic to send and when
  - Dealing with this is tedious for application developers

Reliable transport

- In a perfect world, reliable transport is easy

@Sender

- Send packets

@Receiver

- Wait for packets

Reliable transport

- In a perfect world, reliable transport is easy
- All the bad things best-effort can do
  - A packet is corrupted (bit errors)
  - A packet is lost (*why?*)
  - A packet is delayed (*why?*)
  - Packets are reordered (*why?*)
  - A packet is duplicated (*why?*)

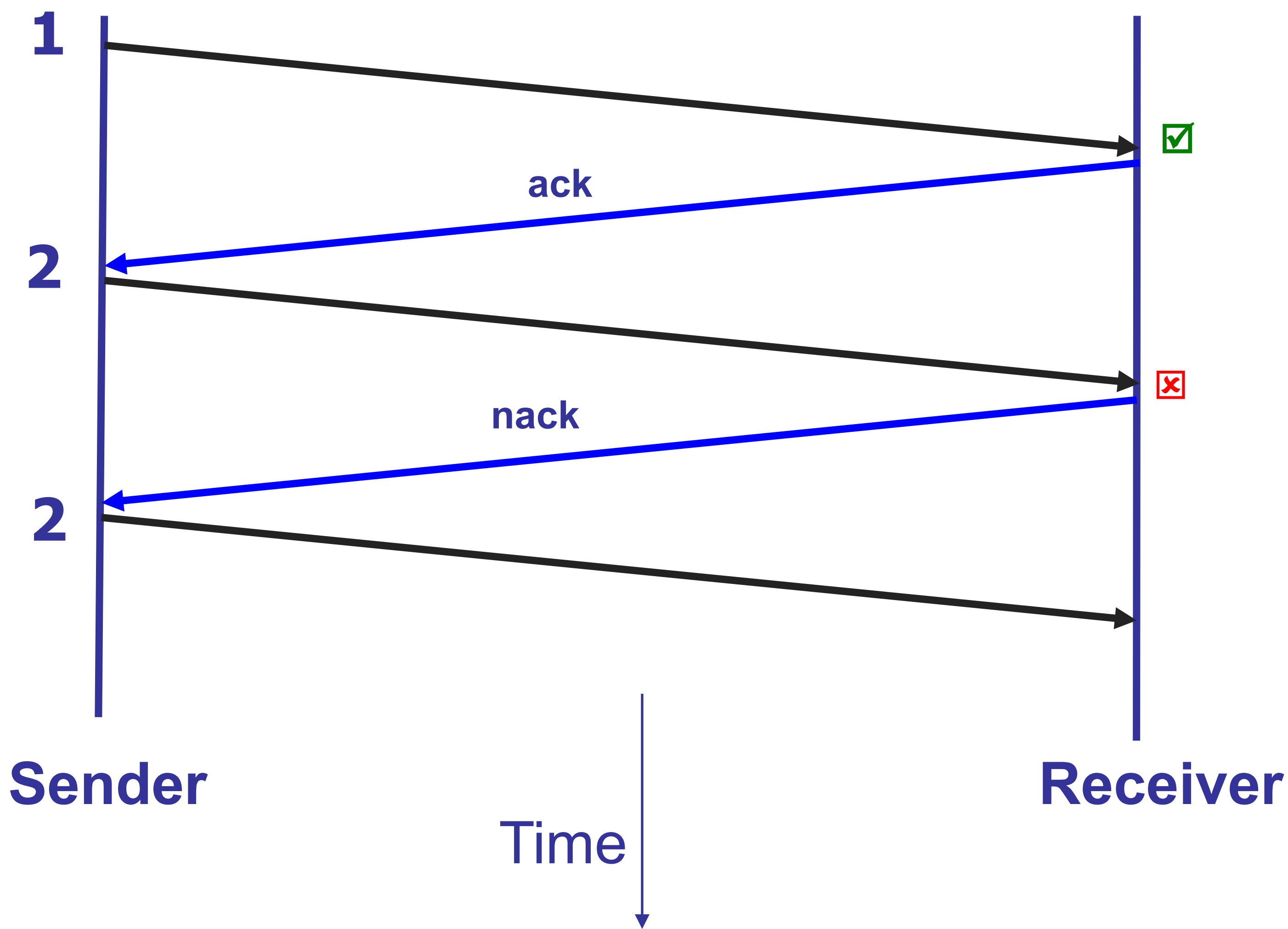
Reliable transport

- Mechanisms for coping with bad events
  - Checksums: to detect corruption
  - ACKs: receiver tells sender that it received packet
  - NACK: receiver tells sender it did not receive packet
  - Sequence numbers: a way to identify packets
  - Retransmissions: sender resends packets
  - Timeouts: a way of deciding when to resend packets
  - *Forward error correction: a way to mask errors without retransmission*
  - *Network encoding: an efficient way to repair errors*

Dealing with packet corruption

❖ *the* question: how to recover from errors:

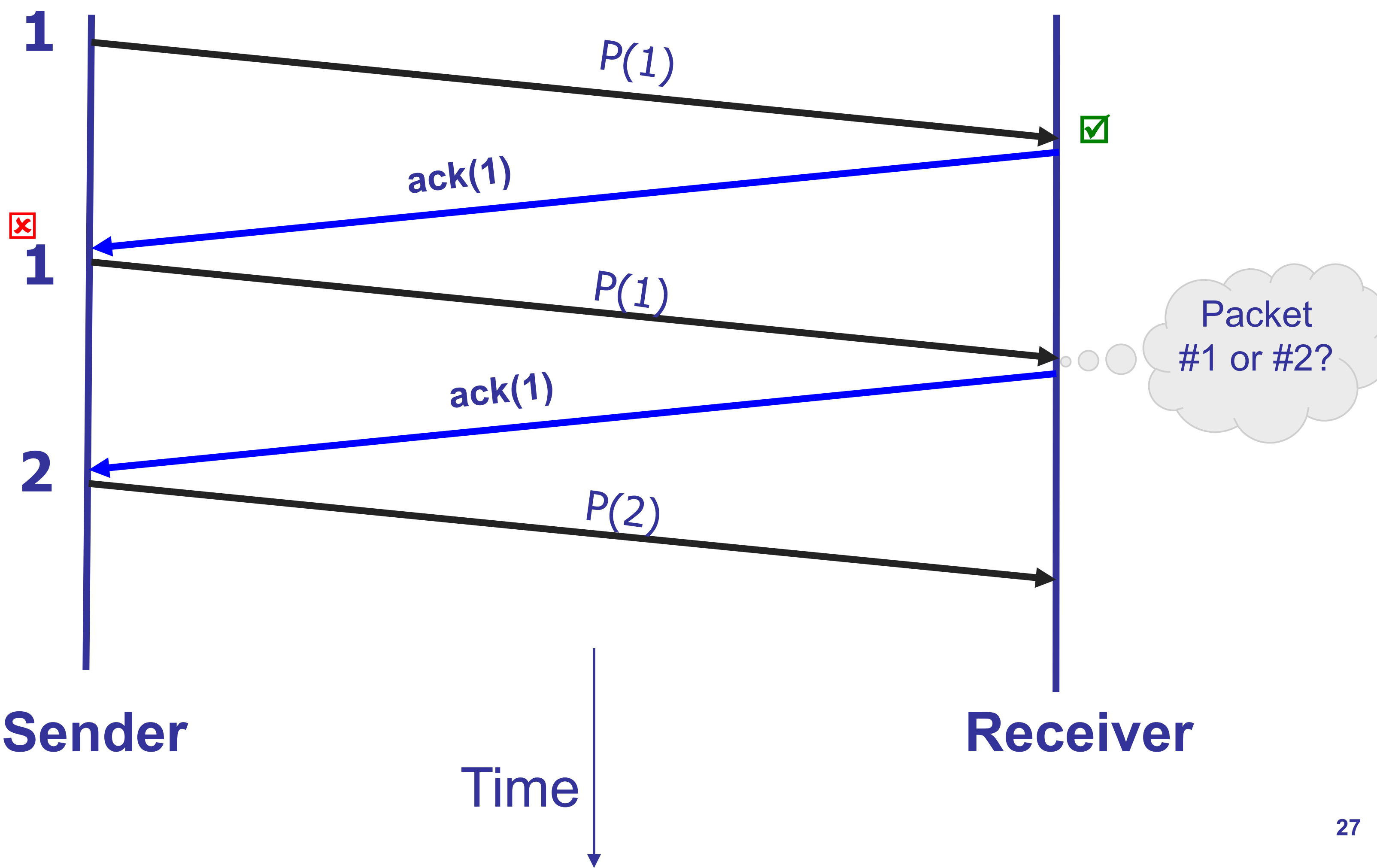
- *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
- *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
- sender retransmits pkt on receipt of NAK



Dealing with packet corruption

What if the ACK/NACK is corrupted?

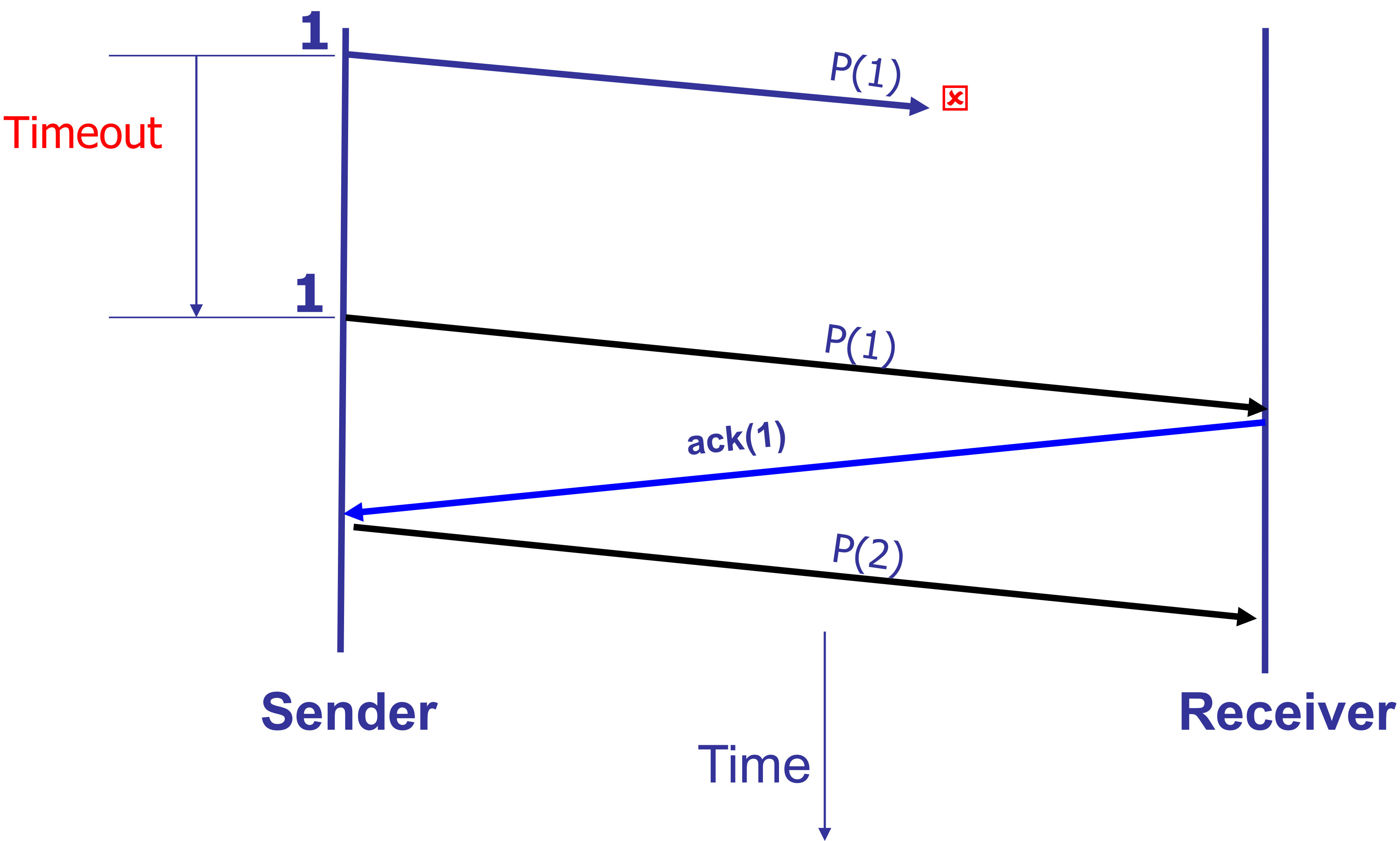
Data and ACK packets carry <u>sequence numbers</u>



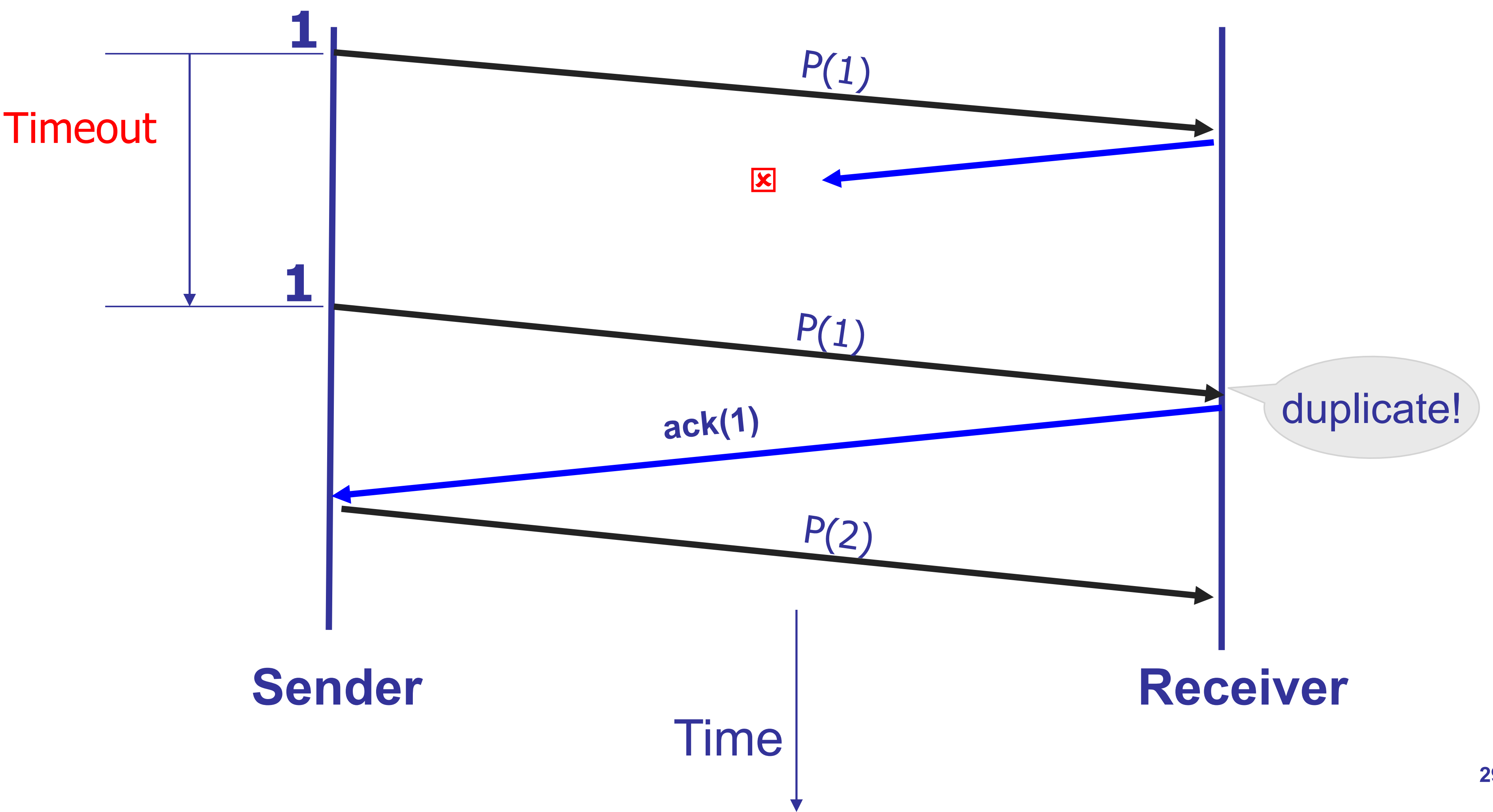
Dealing with packet loss

Timer-driven loss detection

Set timer when packet is sent; retransmit on timeout

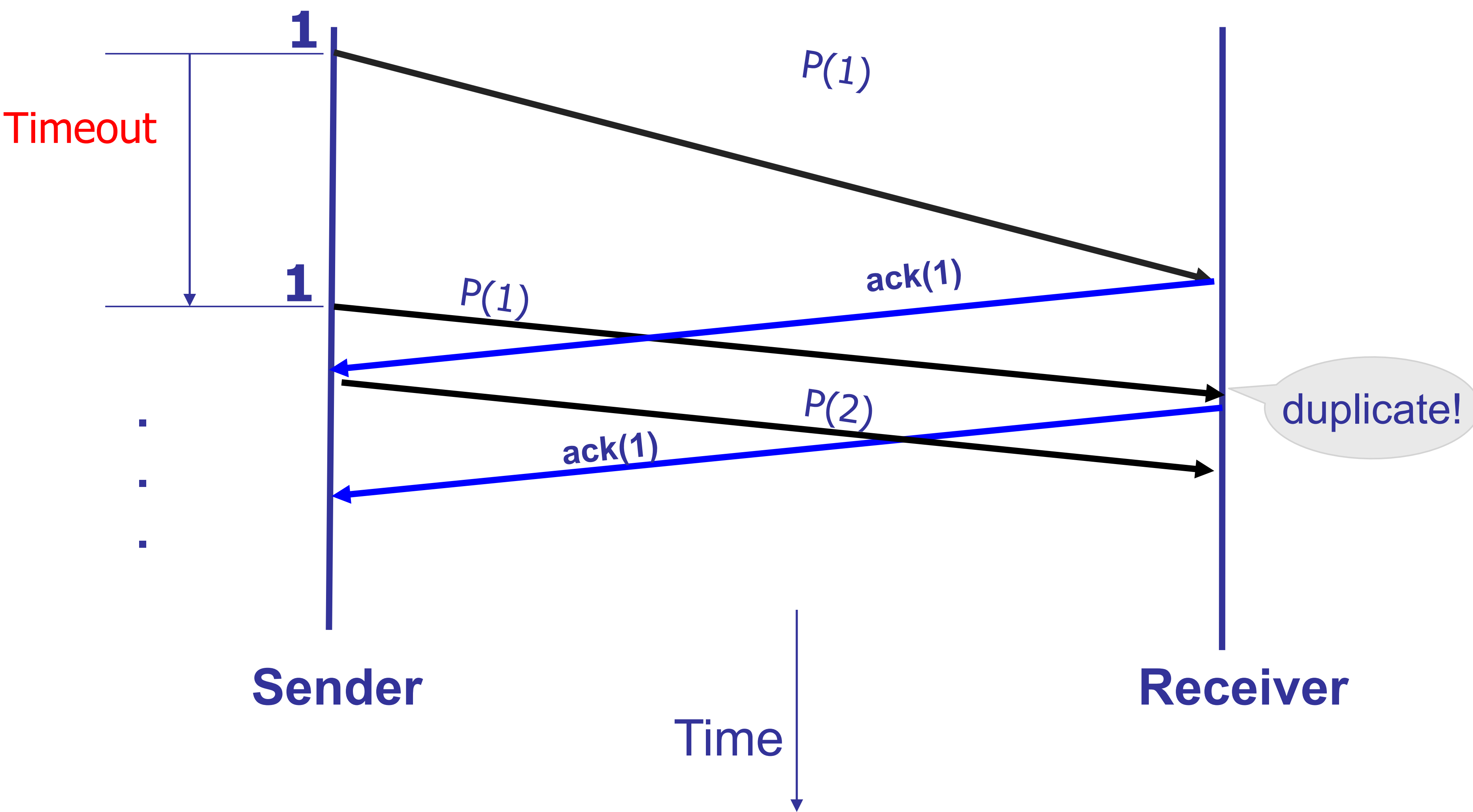


Dealing with packet loss (of ack)



Dealing with packet loss

Timer-driven retransmission can lead to <u>duplicates</u>



Components of a solution

- Checksums (to detect bit errors)
- Timers (to detect loss)
- Acknowledgements (positive or negative)
- Sequence numbers (to deal with duplicates)

Designing a reliable transport protocol？

A Solution: “Stop and Wait”

@Sender

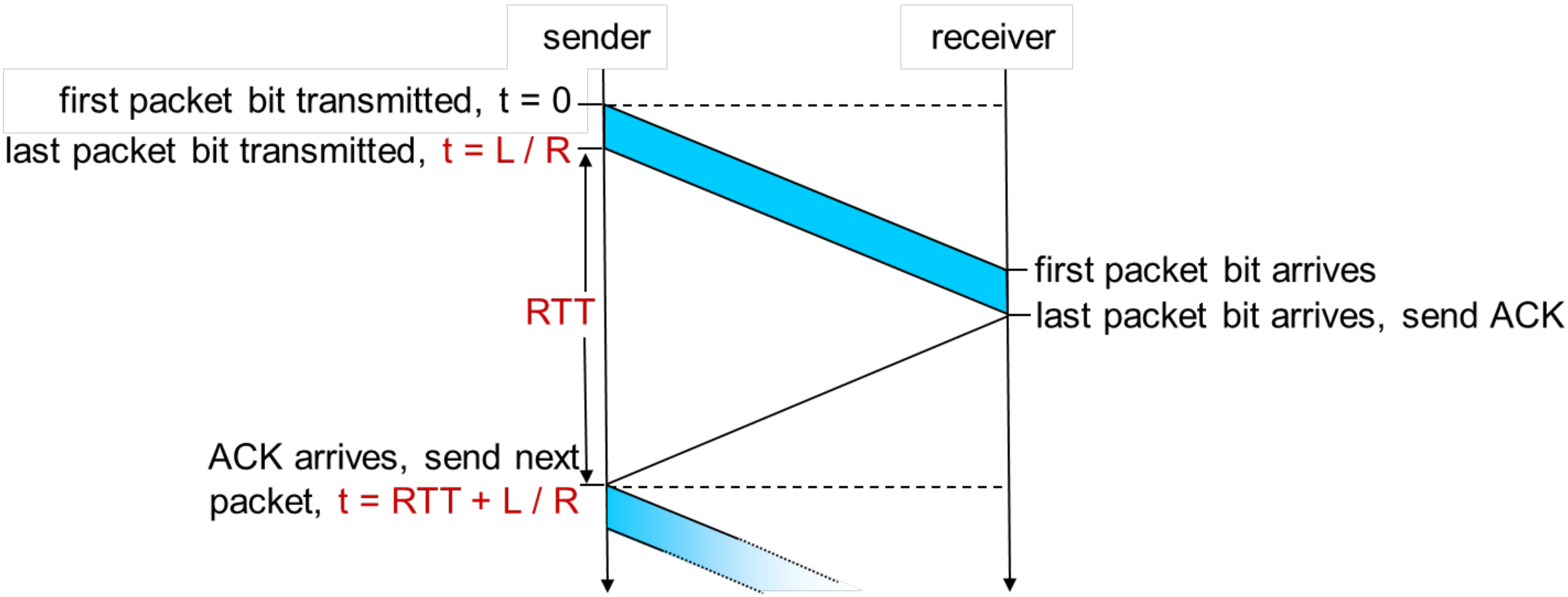
- Send packet(I); (re)set timer; wait for ack
- If (ACK)
  - I++; repeat
- If (NACK or TIMEOUT)
  - repeat

@Receiver

- Wait for packet
- If packet is OK, send ACK
- Else, send NACK
- Repeat

- A correct reliable transport protocol, but an extremely inefficient one

Stop & Wait is inefficient



L: packet size
R: bandwidth of the link
RTT = 2*PropDelay: roundtrip time

If (L/R<< RTT) then
Throughput ~ DATA/RTT

Orders of magnitude

- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

  - **if RTT=30 msec,**

- **U $_{sender}$: *utilization* – fraction of time sender busy sending**

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

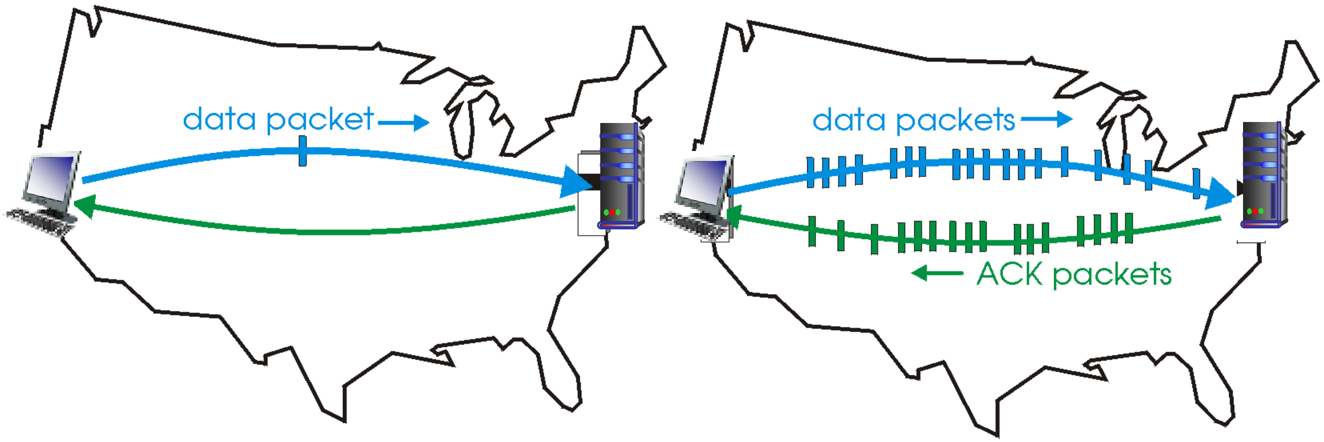
  - **33kB/sec thruput over 1 Gbps link!**

- **network protocol limits use of physical resources!**

Pipelined protocols

pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

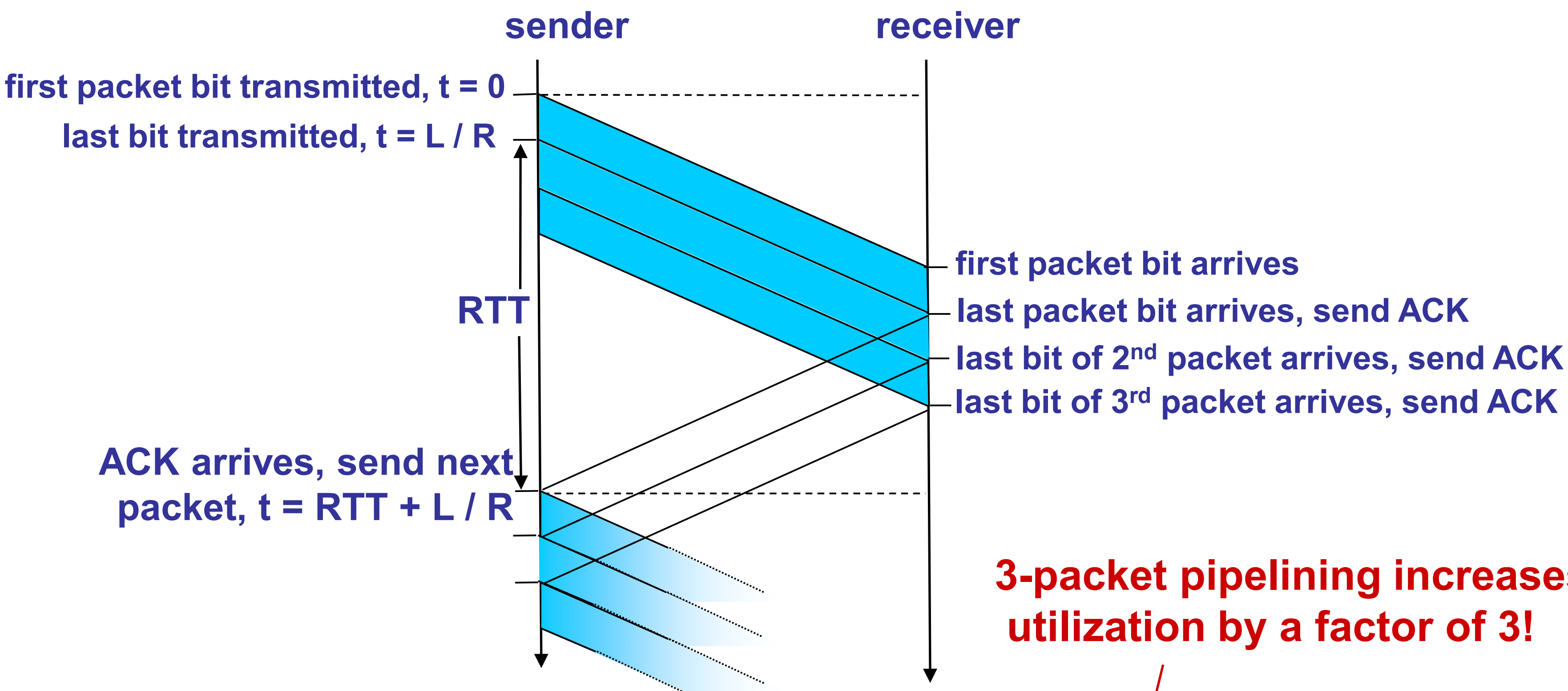
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

Pipelining: increased utilization



3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

Three design decisions

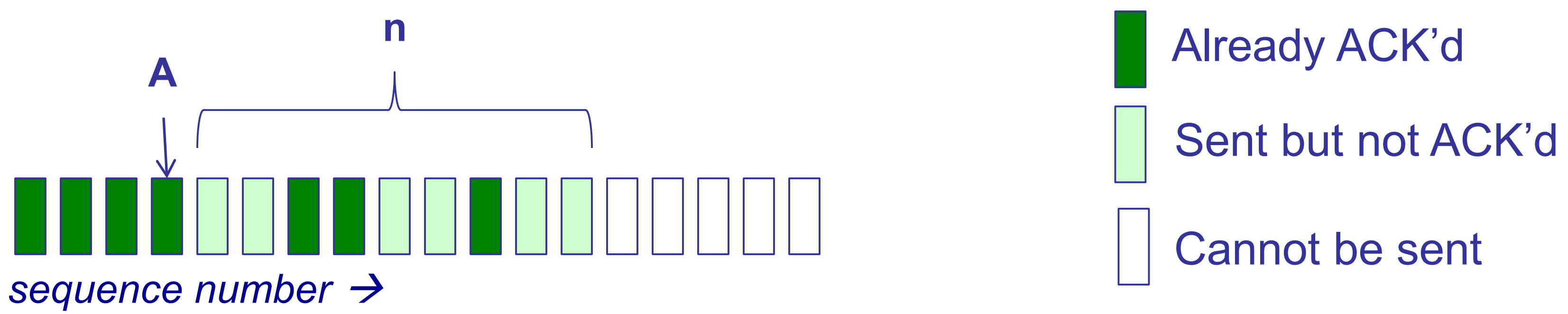
- Which packets can sender send?
  - Sliding window
- How does receiver ack packets?
  - Cumulative
  - Selective
- Which packets does sender resend?
  - Go-Back N (GBN)
  - Selective Repeat (SR)

Sliding window

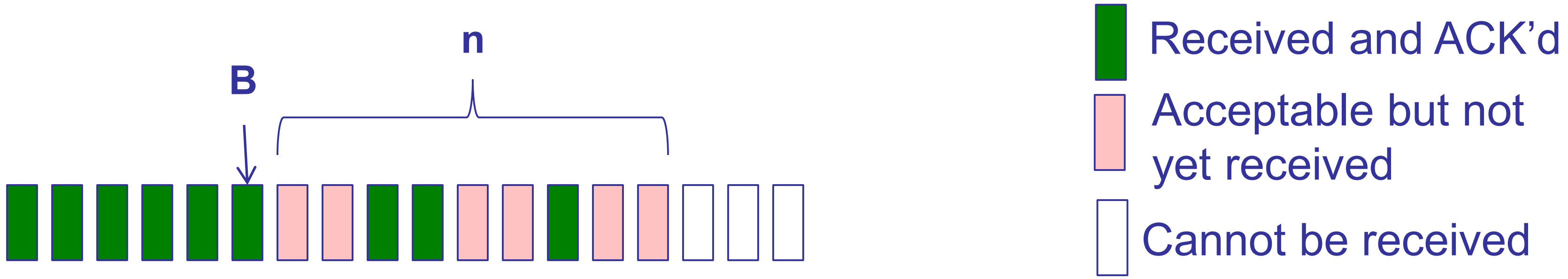
- Window = set of adjacent sequence numbers
  - The size of the set is the window size; assume window size is n
- General idea: send up to n packets at a time
  - Sender can send packets in its window
  - Receiver can accept packets in its window
  - Window of acceptable packets "slides" on successful reception/acknowledgement
  - Window contains all packets that might still be in transit
- Sliding window often called "packets in flight"

Sliding window

- Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



Throughput of sliding window

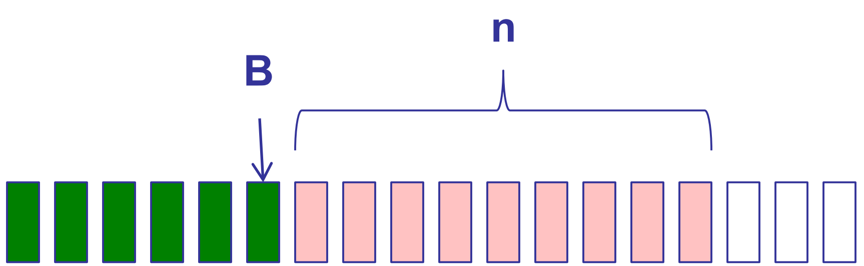
- If window size is n, then throughput is roughly
  - MIN(n*DATA/RTT, Link Bandwidth)
- Compare to Stop and Wait: Data/RTT
- What happens when n gets too large?

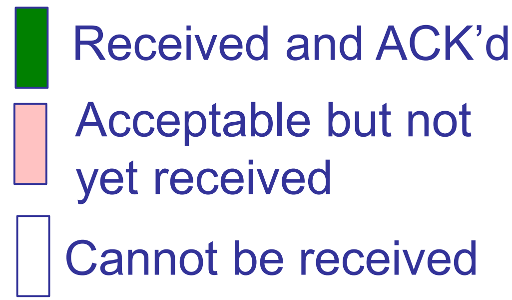
Acknowledgements w/ sliding window

- Two common options
  - Cumulative ACKs: ACK carries next in-order sequence number that the receiver expects

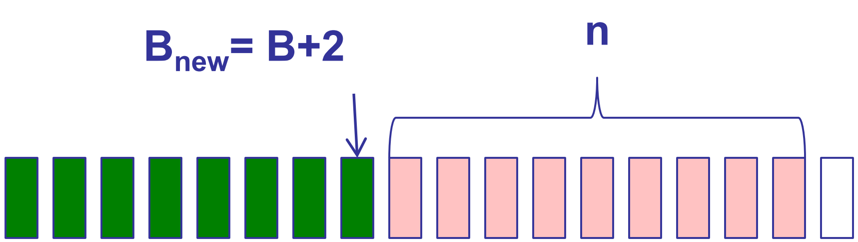
Cumulative acknowledgements

- At receiver





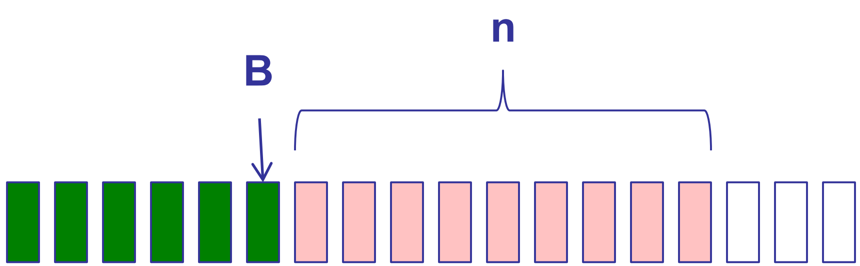
- After receiving B+1, B+2

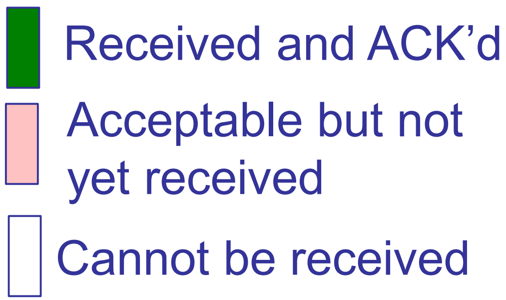


- Receiver sends ACK(B+3) = ACK(B_{new}+1)

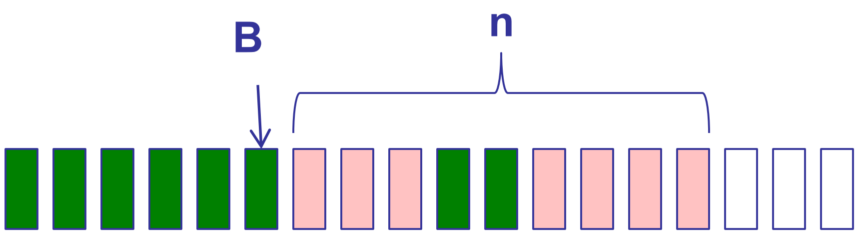
Cumulative acknowledgements (cont'd)

- At receiver





- After receiving B+4, B+5



- Receiver sends ACK(B+1)

Acknowledgements w/ sliding window

- Two common options
  - Cumulative ACKs: ACK carries next in-order sequence number the receiver expects
  - Selective ACKs: ACK individually acknowledges correctly received packets

- Selective ACKs offer more precise information but require more complicated book-keeping

Sliding window protocols

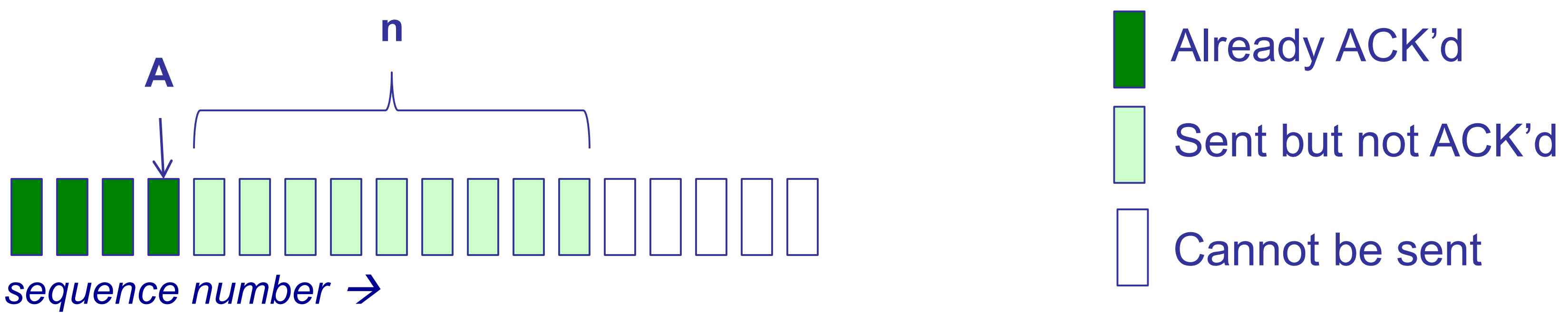
- Resending packets: two canonical approaches
  - Go-Back-N
  - Selective Repeat
- Many variants that differ in implementation details

Go-Back-N (GBN)

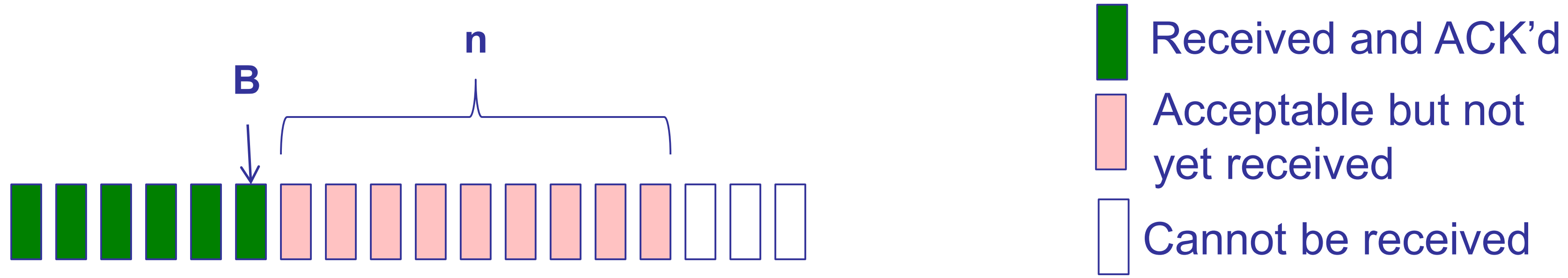
- Sender transmits up to n unacknowledged packets
- Receiver only accepts packets in order
  - Discards out-of-order packets (i.e., packets other than B+1)
- Receiver uses cumulative acknowledgements
  - i.e., sequence# in ACK = next expected in-order sequence#
- Sender sets timer for 1st outstanding ack (A+1)
- If timeout, retransmit A+1, … , A+n

Sliding window with GBN

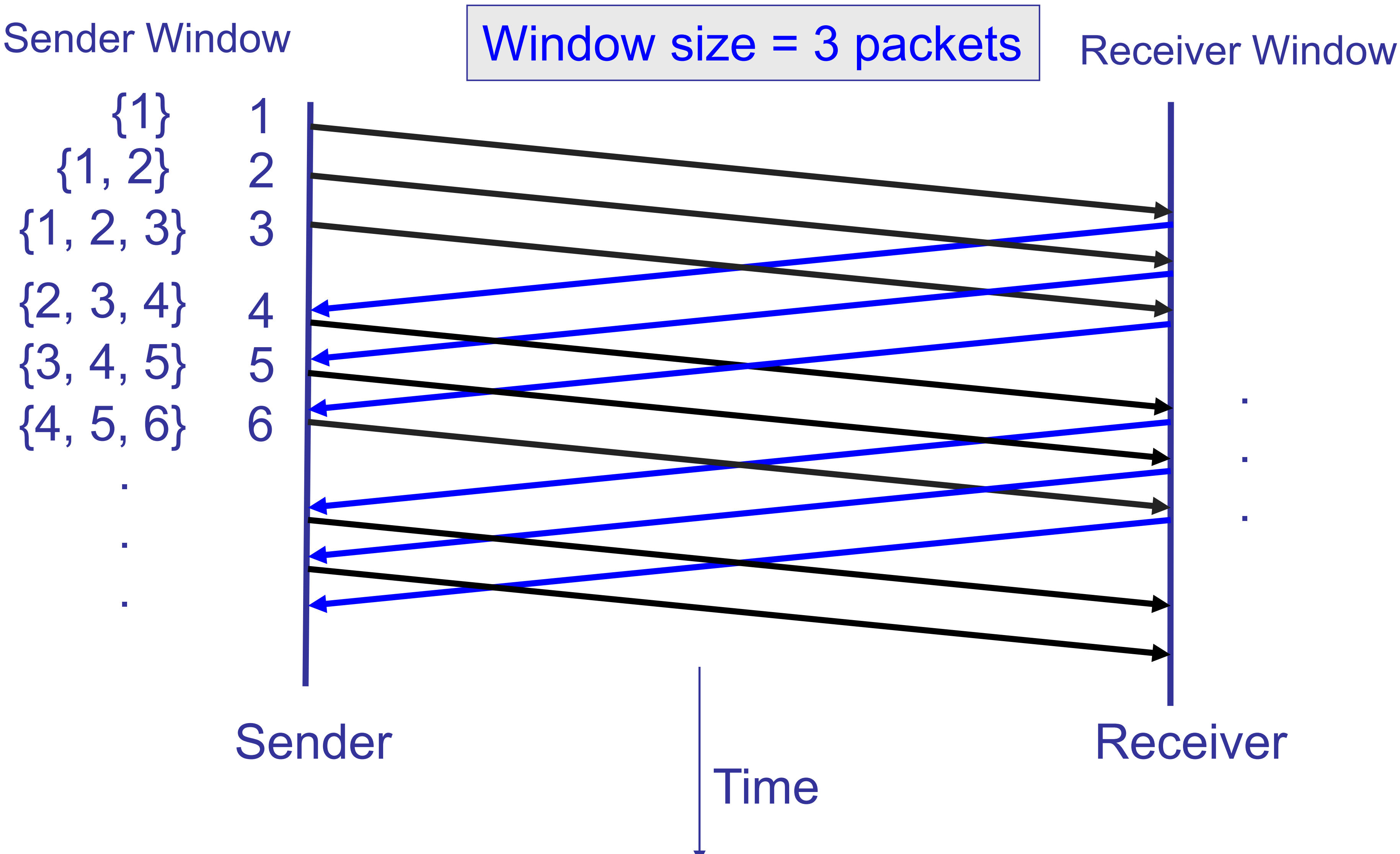
- Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



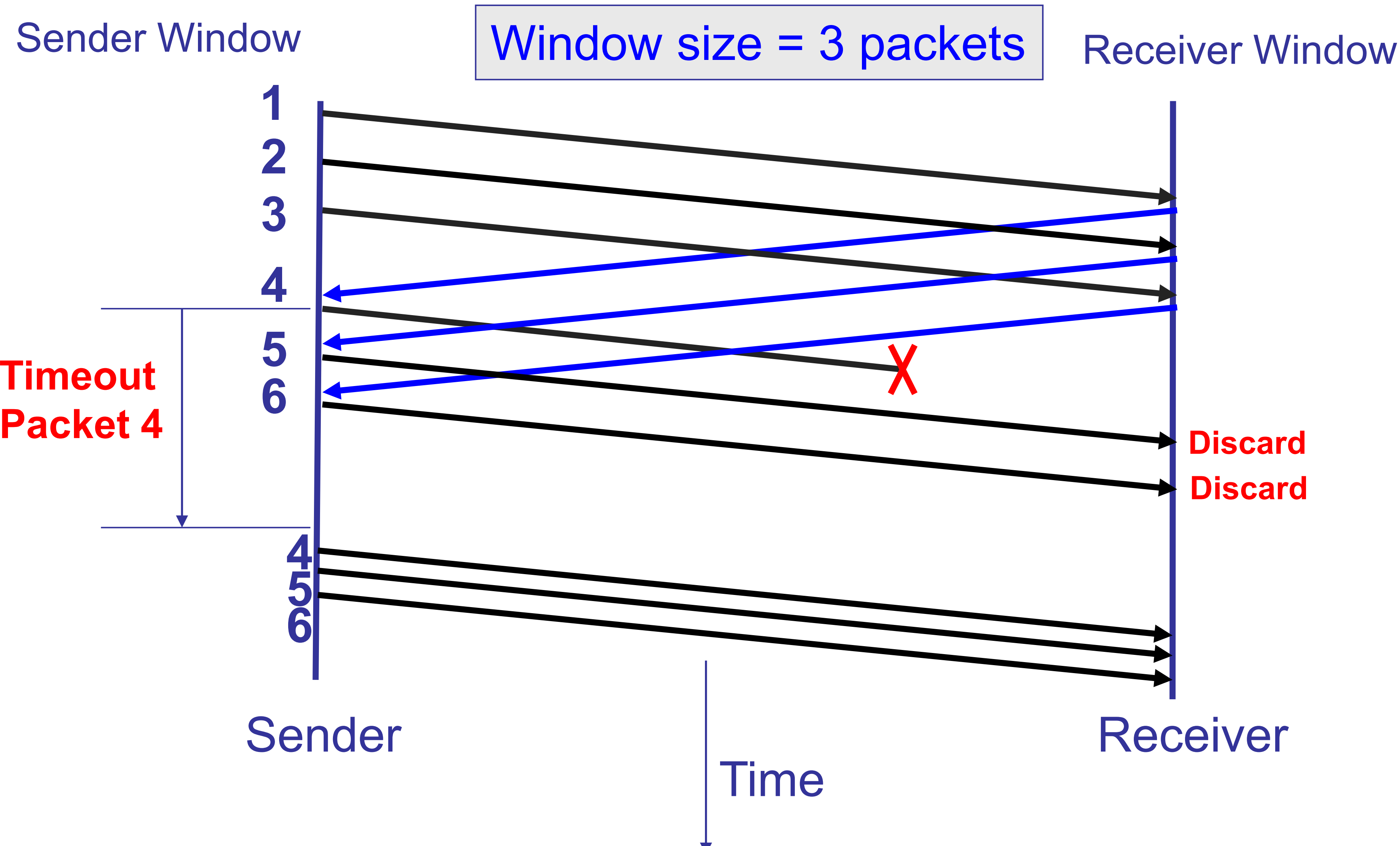
- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



GBN example w/o errors



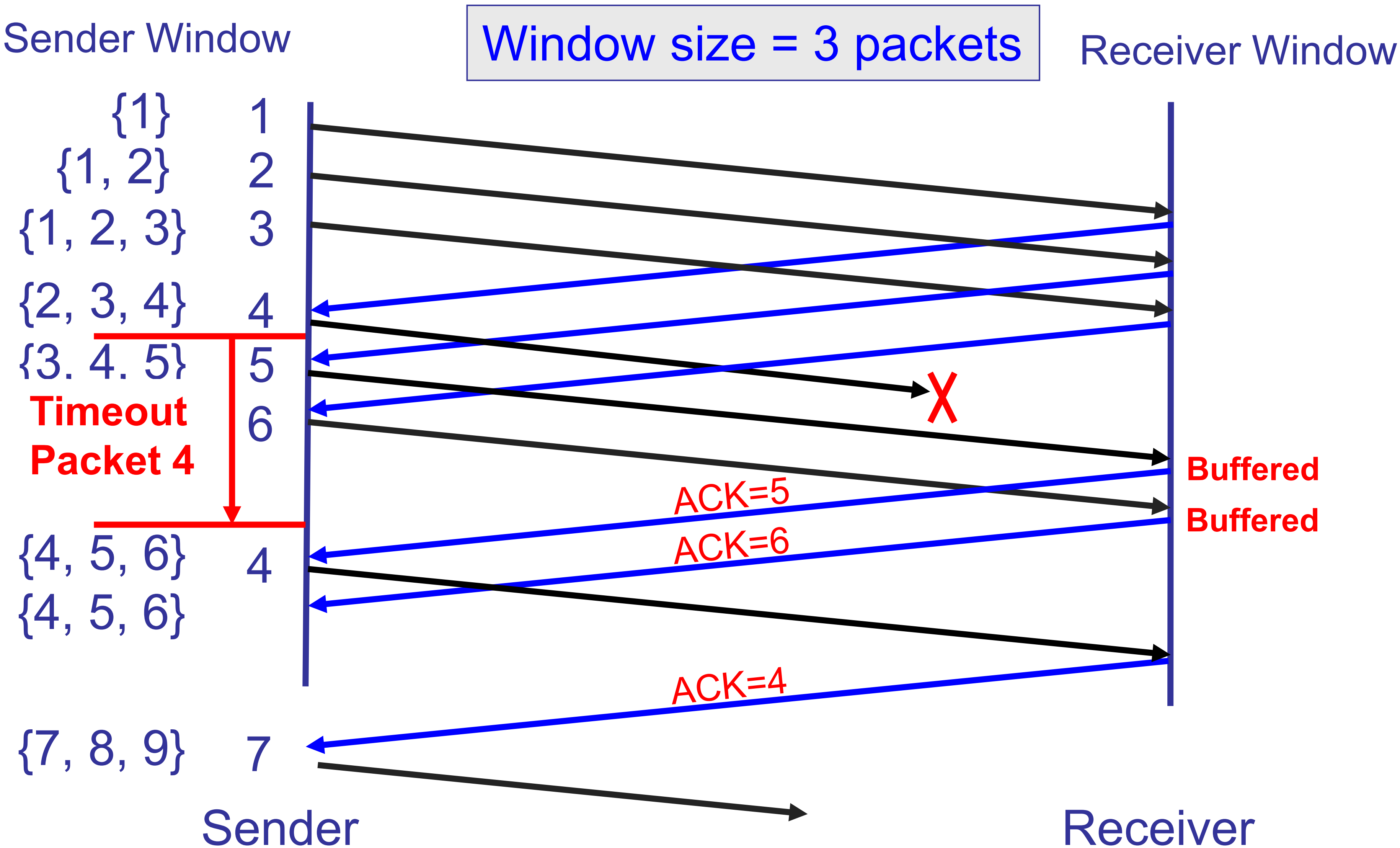
GBN example with errors



Selective Repeat (SR)

- Sender: transmit up to n unacknowledged packets
- Assume packet k is lost, k+1 is not
  - Receiver: indicates packet k+1 correctly received
  - Sender: retransmit only packet k on timeout
- Efficient in retransmissions but complex book-keeping
  - Need a timer per packet

SR example with errors



GBN vs. Selective Repeat

- When would GBN be better?
  - When error rate is low; wastes bandwidth otherwise
- When would SR be better?
  - When error rate is high; otherwise, too complex

Observations

- With sliding windows, it is possible to fully utilize a link, provided the window size is large enough.
- Sender has to buffer all unacknowledged packets, because they may require retransmission
- Receiver may be able to accept out-of-order packets, but only up to its buffer limits
- Implementation complexity depends on protocol details (GBN vs. SR)

Components of a solution

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - Cumulative
  - Selective
- Sequence numbers (duplicates, windows)
- Sliding windows (for efficiency)
- Reliability protocols use the above to decide when and what to retransmit or acknowledge

Summary

- Transport Layer Service
  - Addressing
  - Multiplexing
- Design of reliable transport
  - Dealing with packet corruption
  - Flow control
    - Stop-and-wait
    - Sliding Window

Homework

- 第3章:R5, R8, P1, P19