EECS 489

Computer Networks

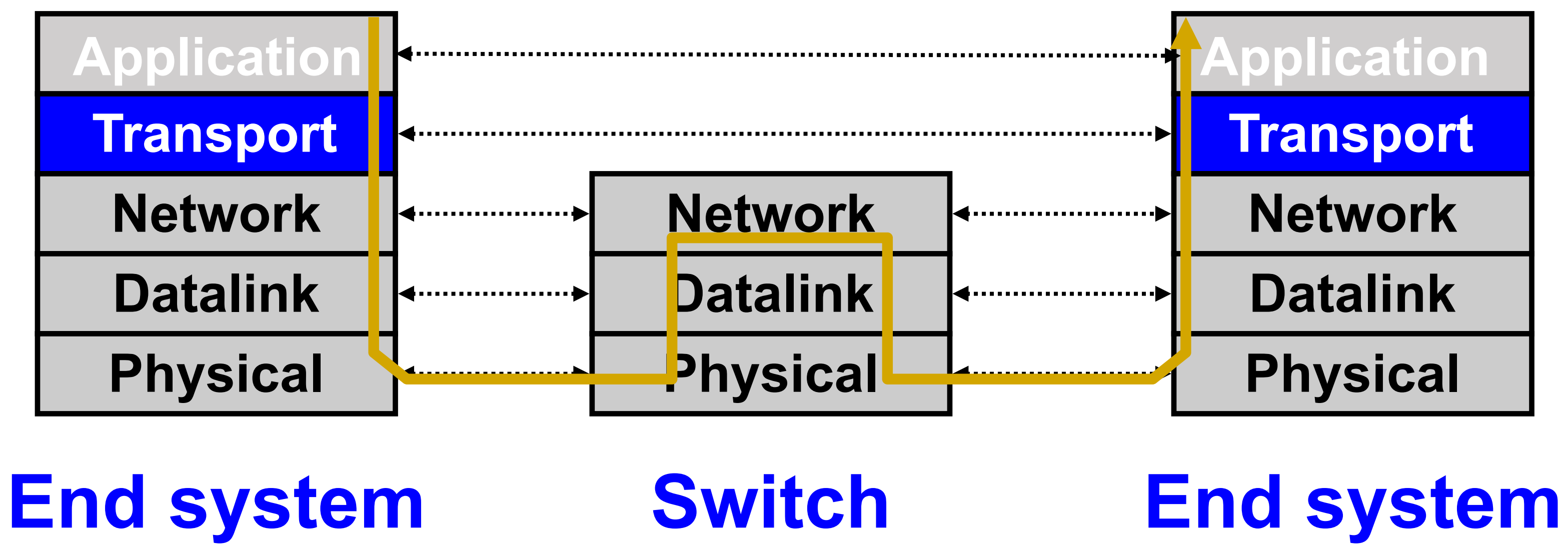
Transport Layer Basics

Agenda

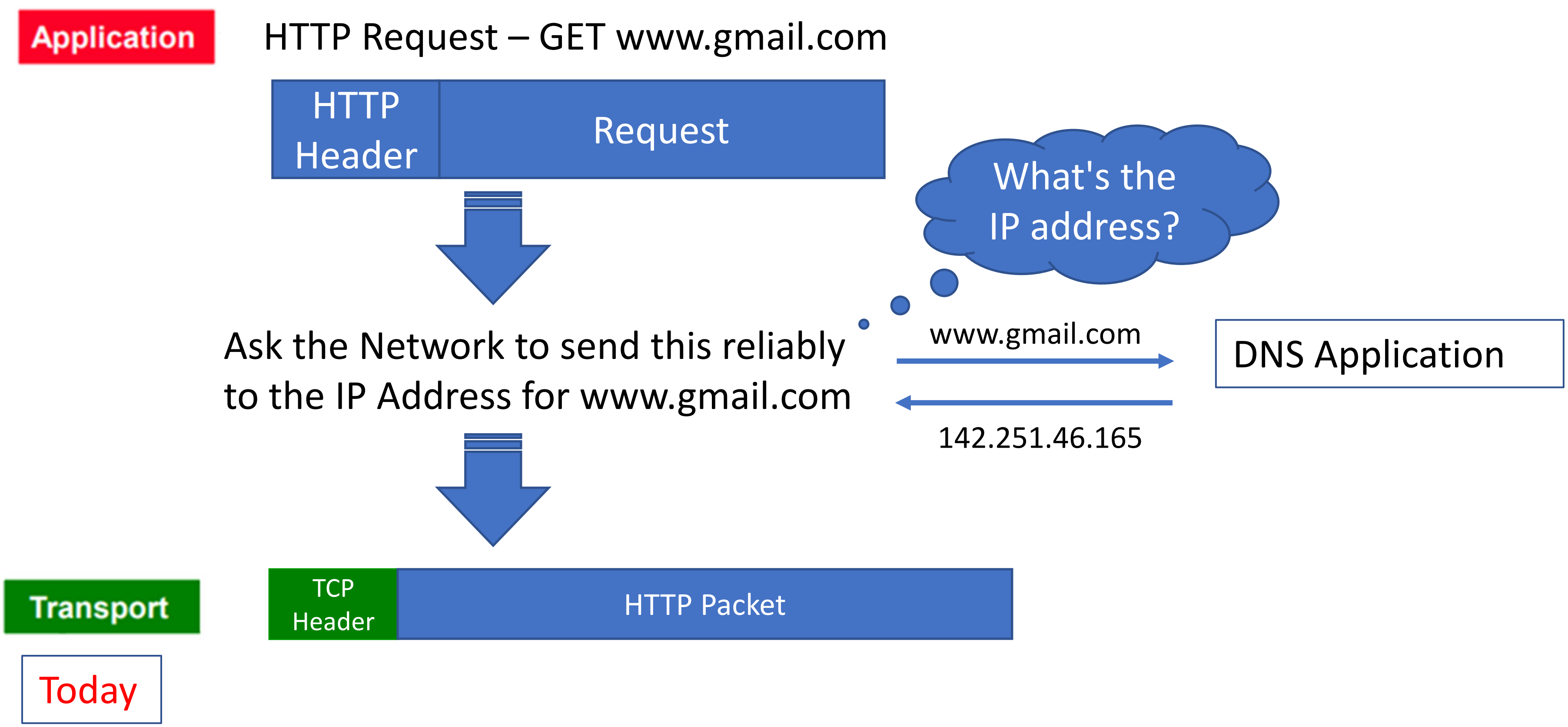
- Transport layer basics
- UDP
- Designing a reliable transport protocol

Transport layer

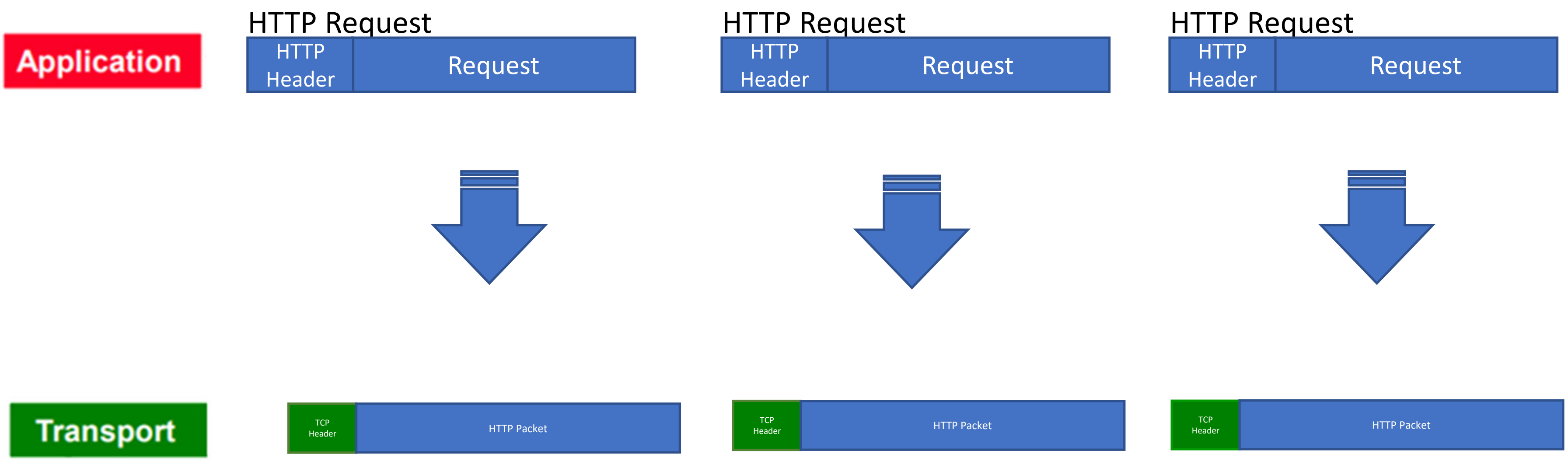
- Layer at end hosts, between the application and network layer



Packet Generation



Packet Generation



Why a transport layer?

- IP addresses capture hosts, but end-to-end communication happens between applications
  - 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 from different applications and delivering to the network layer
- Demultiplexing (Demux)
  - Delivering correct data to corresponding sockets from a multiplexed 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.)

QUIC transport protocol

- The QUIC transport protocol (RFC 9000)
- Built on top of UDP
- QUIC packets are encrypted individually
- Faster connection setup by reusing the negotiated parameters from a previous connection
- Many other benefits: extensibility, reduced sensitivity to packet loss, reduced HoL, etc.

Applications and sockets

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

UDP: User Datagram Protocol

- Lightweight communication between processes
  - Avoid overhead and delays of order & reliability
- UDP described in RFC 768 – (1980!)
  - Destination IP address and port to support demultiplexing

SRC port	DST port
Length	Checksum
DATA	

UDP (cont’d)

- Optional error checking on the packet contents
  - (checksum field = 0 means “don’t verify checksum”)
- Source port is also optional
  - Useful to respond back to the sender in some cases

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

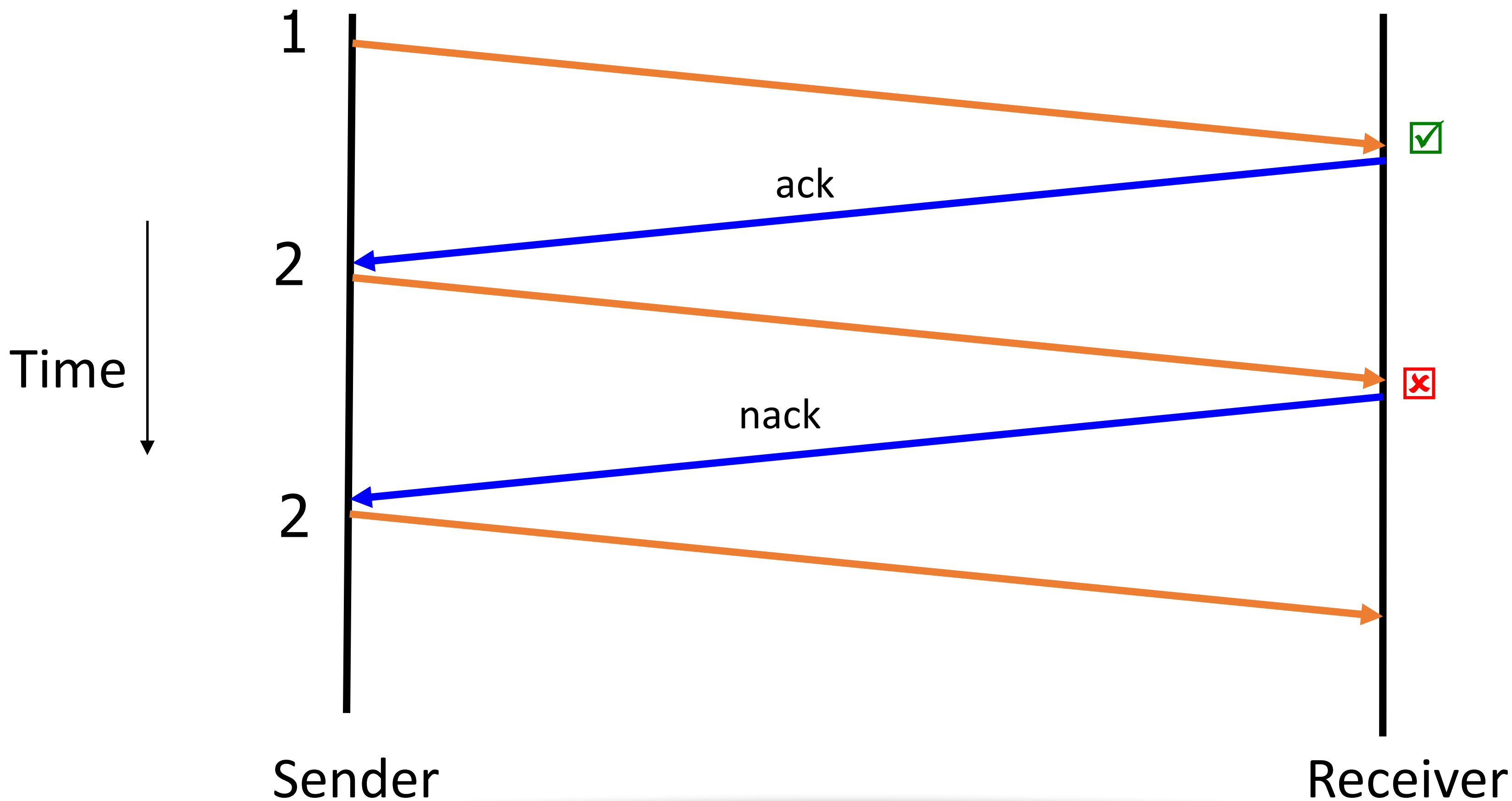
@Receiver

- Wait for packets

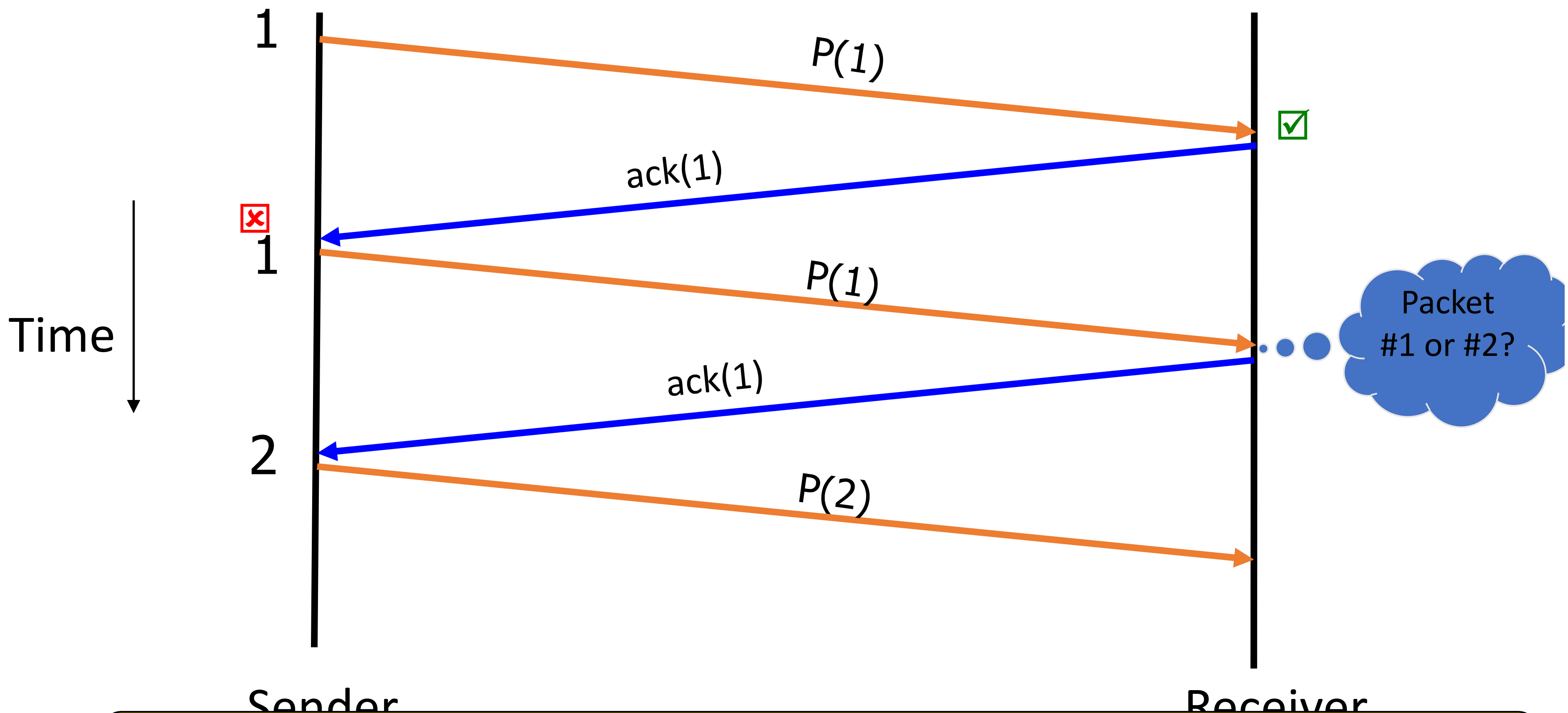
Reliable transport

- 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?*)

Dealing with packet corruption

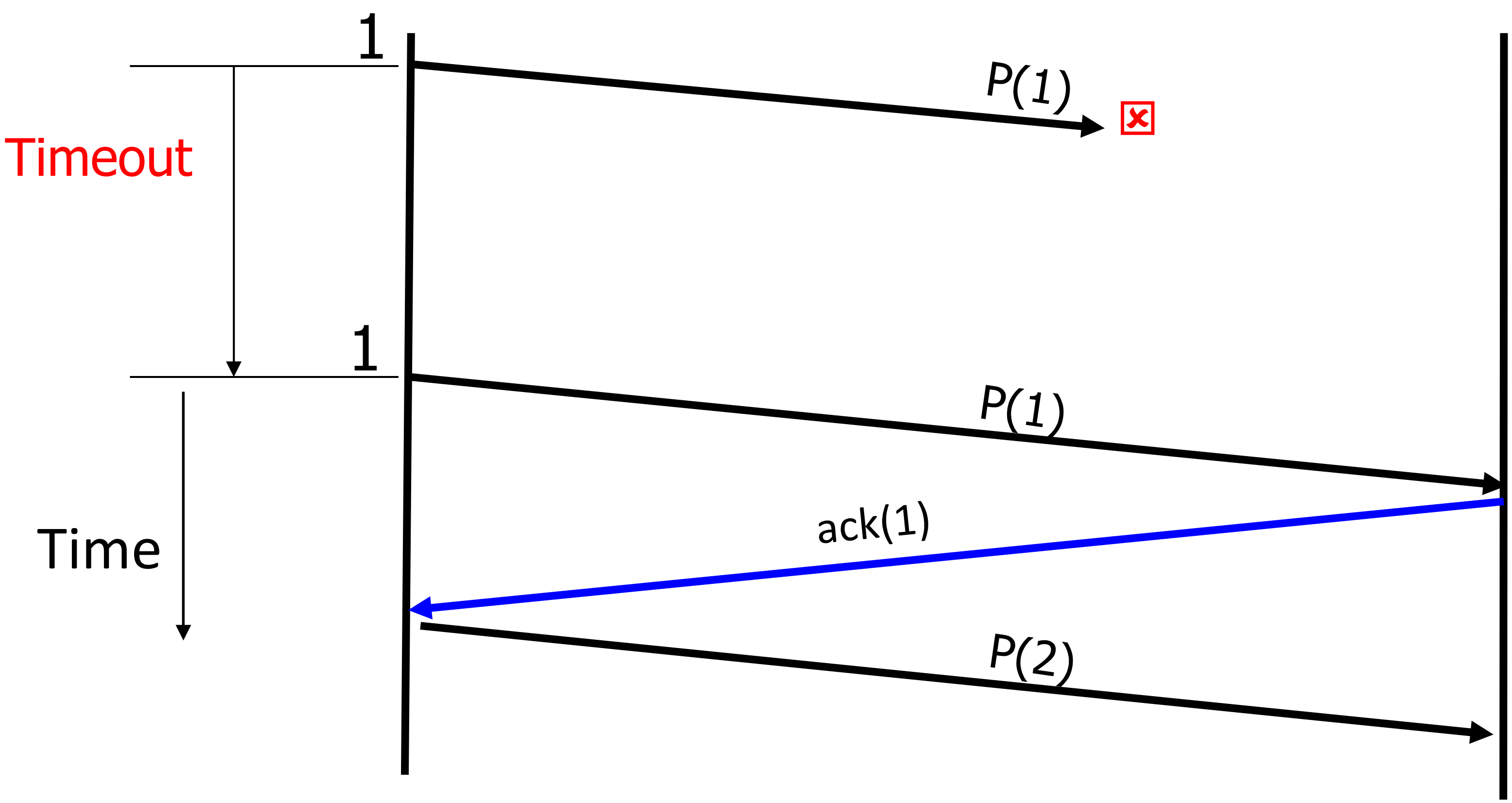


Dealing with packet corruption



Data and ACK packets carry sequence numbers

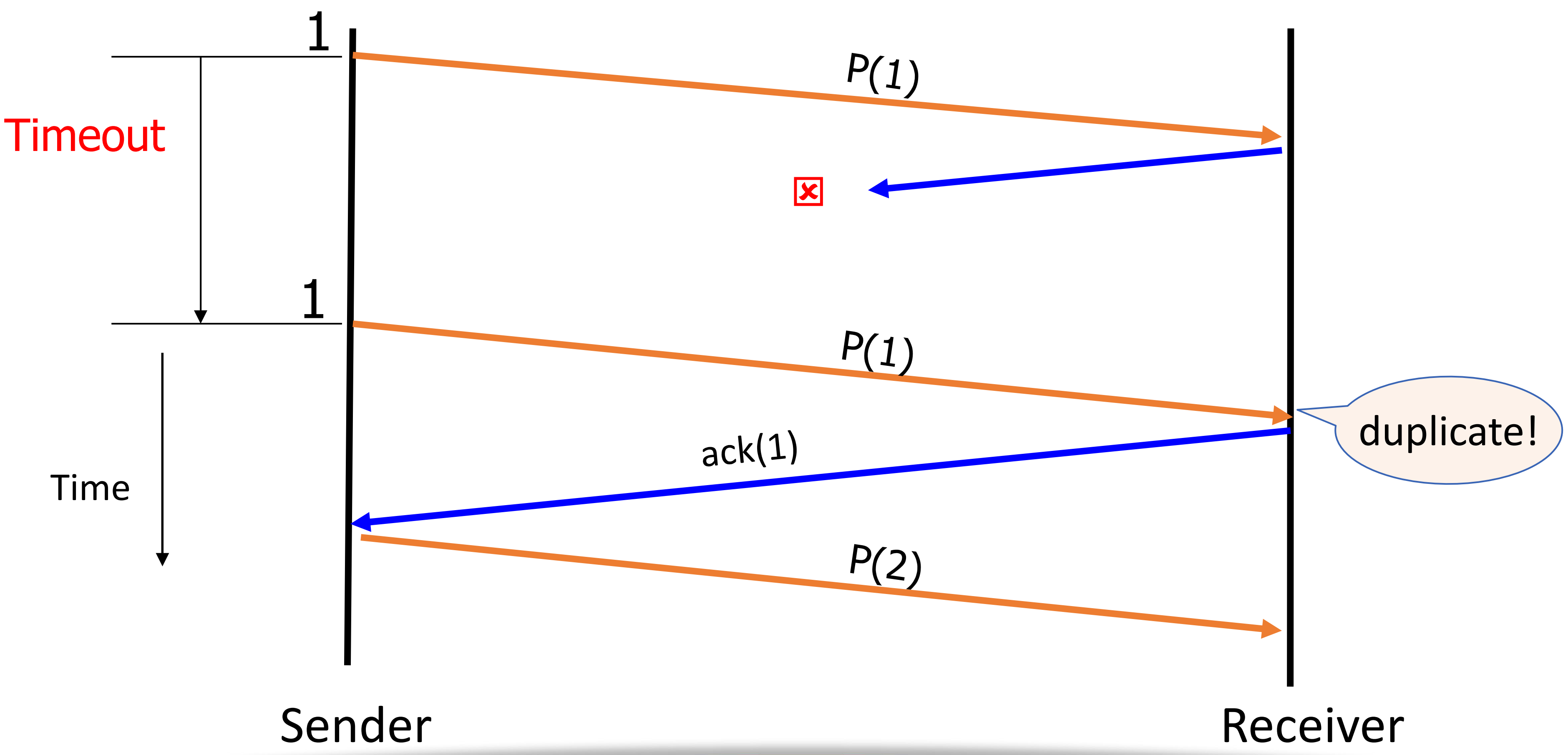
Dealing with packet loss



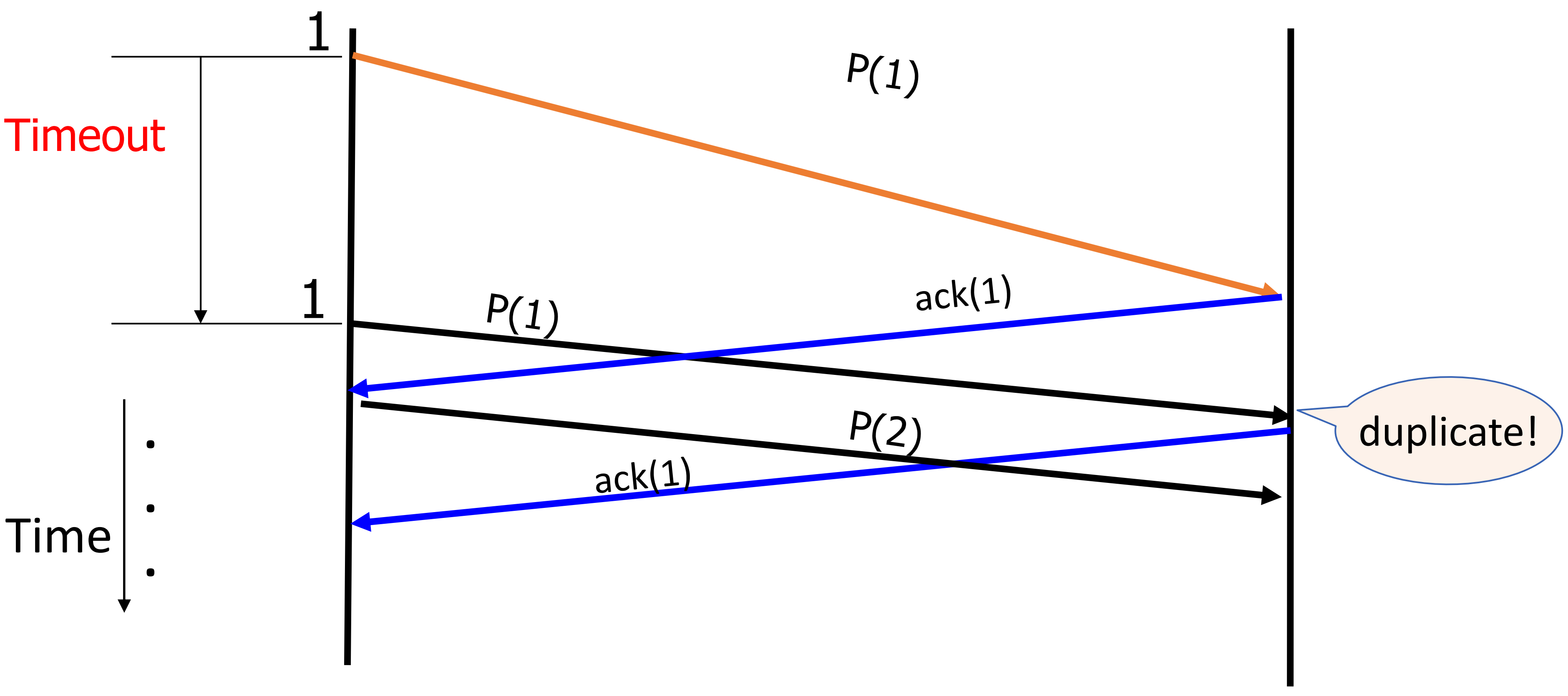
Timer-driven loss detection

Set timer when packet is sent; retransmit on timeout

Dealing with packet loss (of ack)



Dealing with delay



Timer-driven retransmission can lead to duplicates

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

A Solution: “Stop and Wait”

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

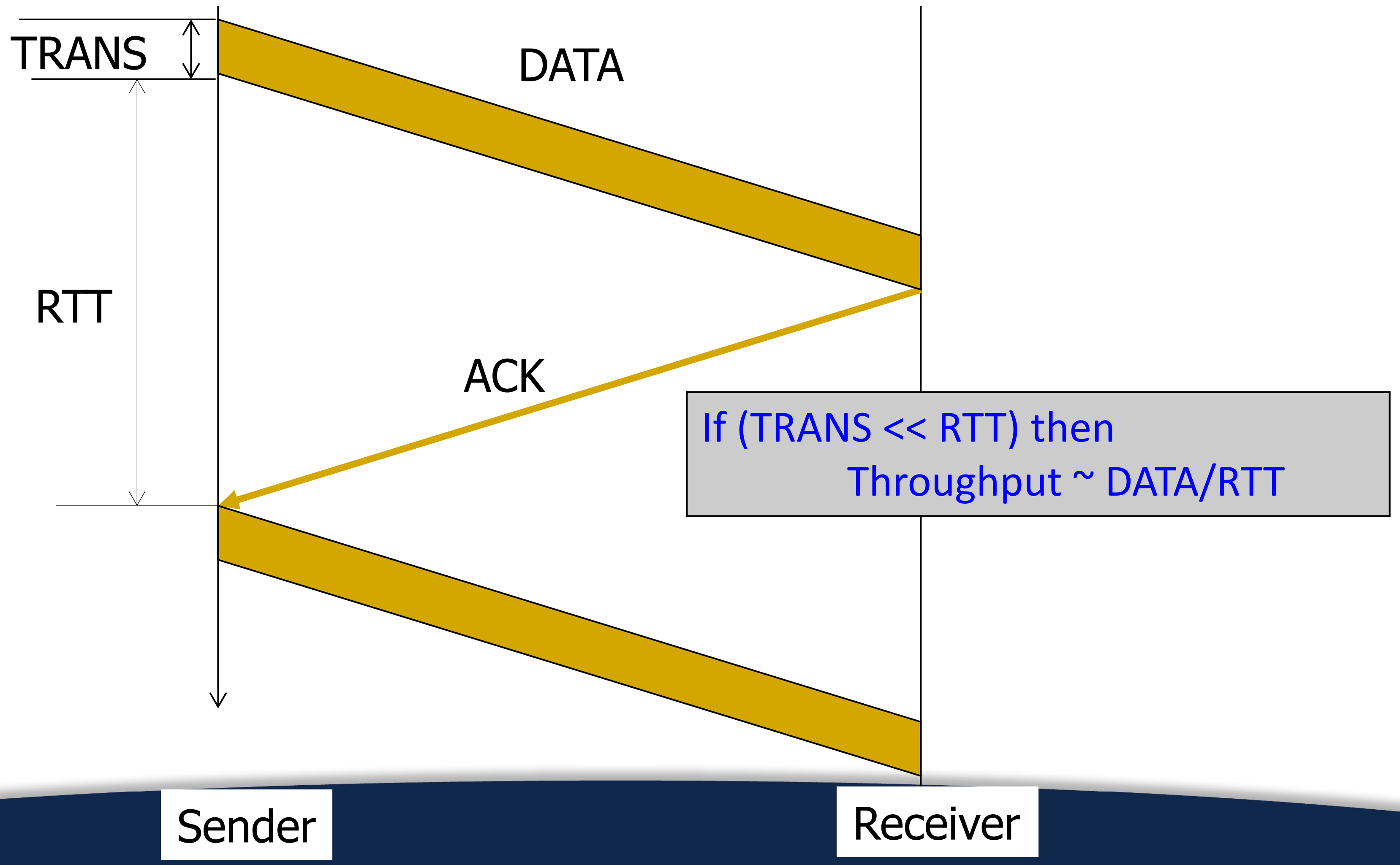
@Sender

- 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

Stop & Wait is inefficient



Orders of magnitude

- Transmission time for 10Gbps link:
  - ~ microsecond for 1500 byte packet
- RTT:
  - 1,000 kilometers ~ O(10) milliseconds

Three design decisions

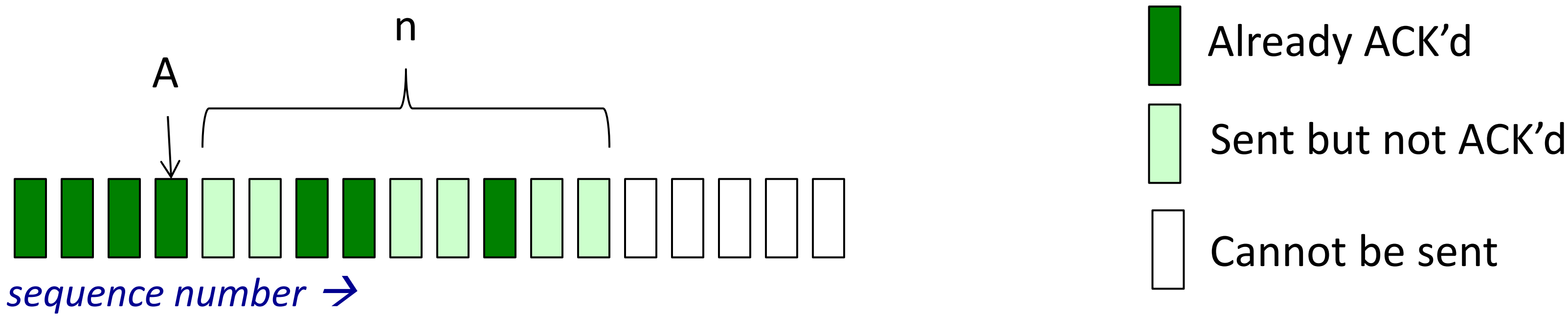
- Which packets can sender send?
- How does receiver ack packets?
- Which packets does sender resend?

Sliding window

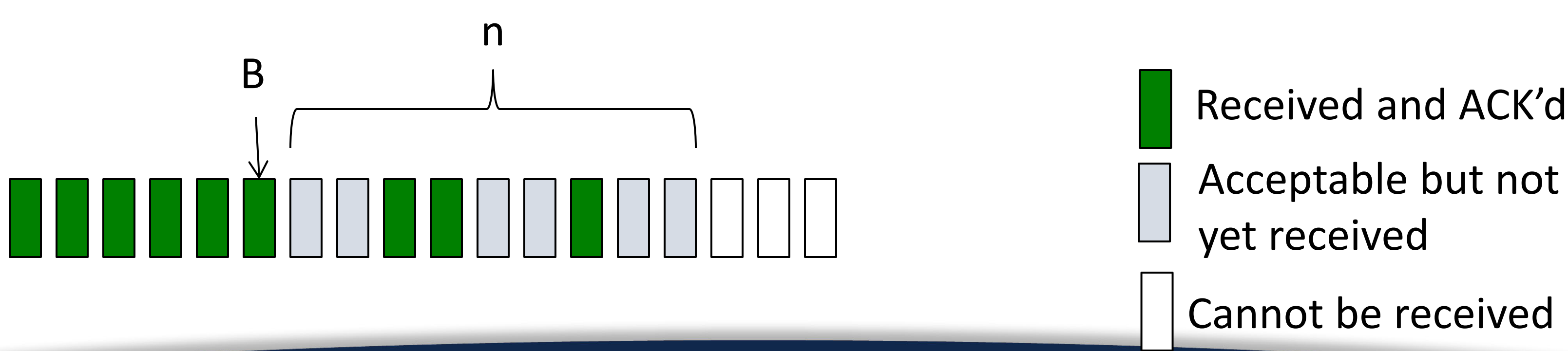
- Window = set of adjacent sequence numbers
  - The size of the set is the window size; assume window size is n
- General idea: send up to n packets at a time
  - Sender can send packets in its window
  - Receiver can accept packets in its window
  - Window of acceptable packets "slides" on successful 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

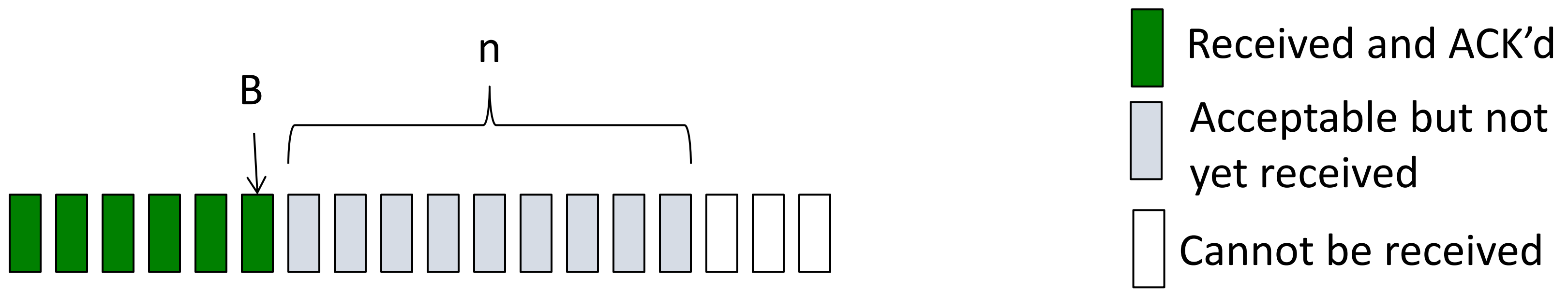
- What happens when n gets too large?

Acknowledgements w/ sliding window

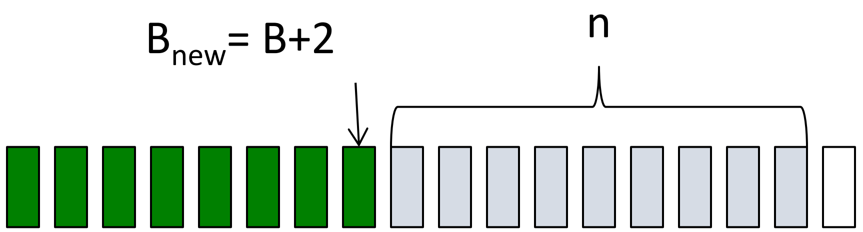
- Two common options
  - Cumulative ACKs: ACK carries next in-order sequence number that the receiver expects

Cumulative acknowledgements

- At receiver



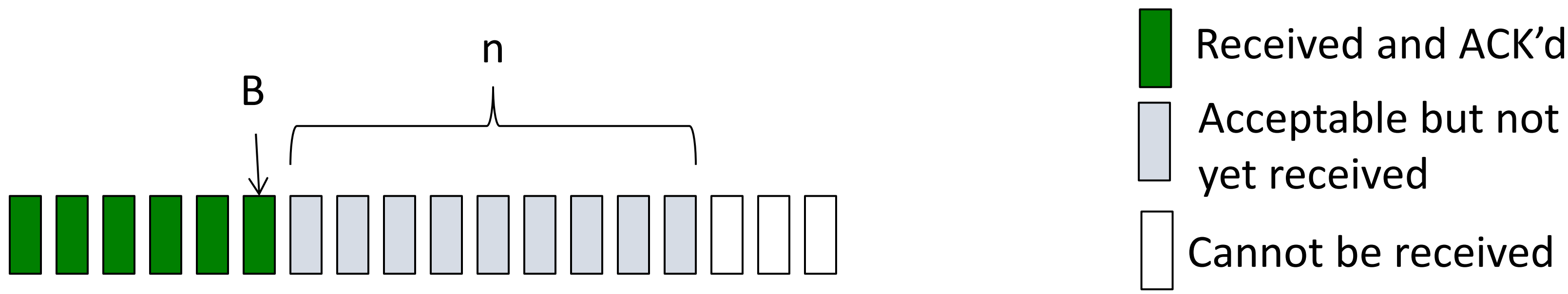
- After receiving B+1, B+2



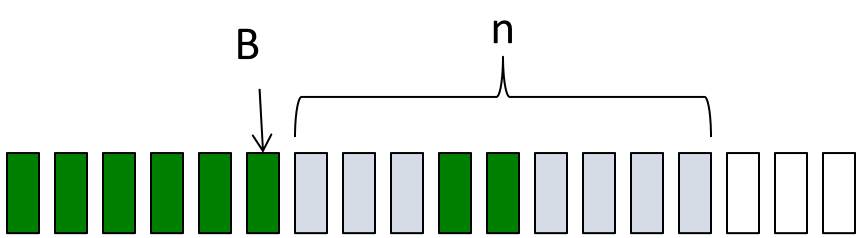
- Receiver sends ACK(B+3) = ACK(B_{new}+1)

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

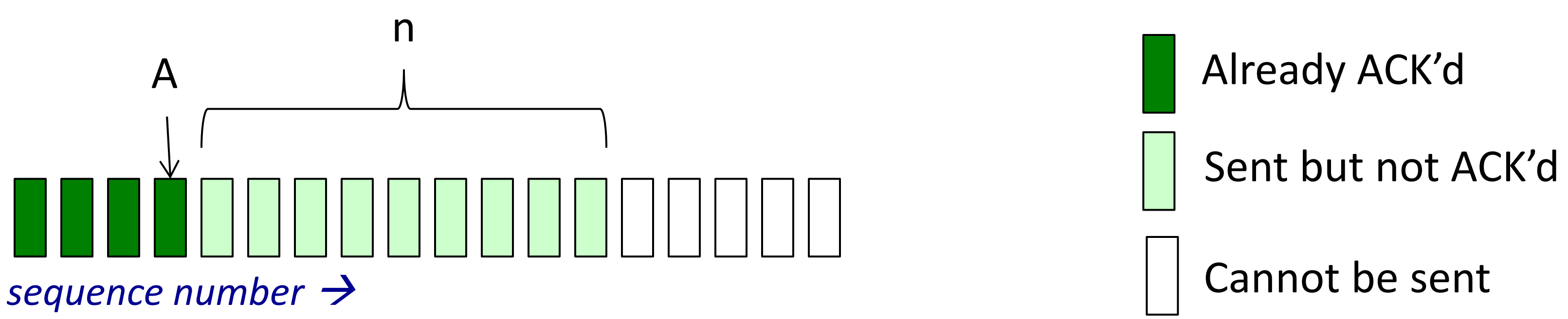
- Resending packets: two canonical approaches
  - Go-Back-N
  - Selective Repeat
- Many variants that differ in implementation details

Go-Back-N (GBN)

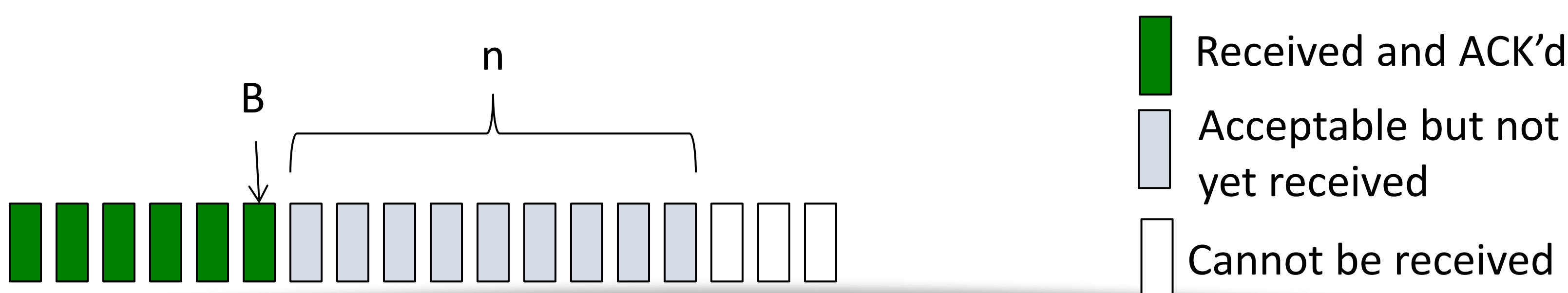
- Sender transmits up to n unacknowledged packets
- Receiver only accepts packets in order
  - Discards out-of-order packets (i.e., packets other than B+1)
- Receiver uses cumulative acknowledgements
  - i.e., sequence# in ACK = next expected in-order sequence#
- Sender sets timer for 1st outstanding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

Sliding window with GBN

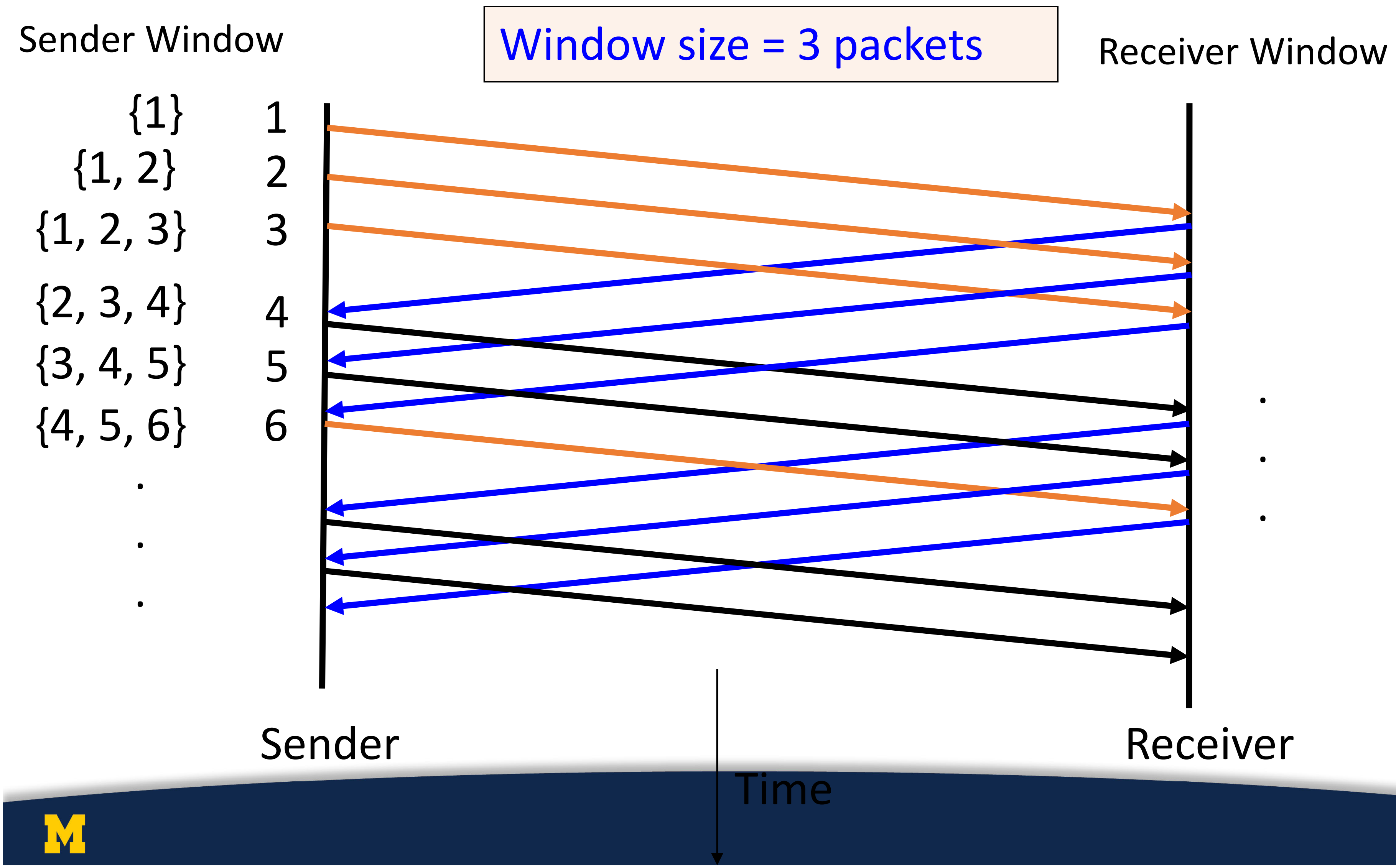
- Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



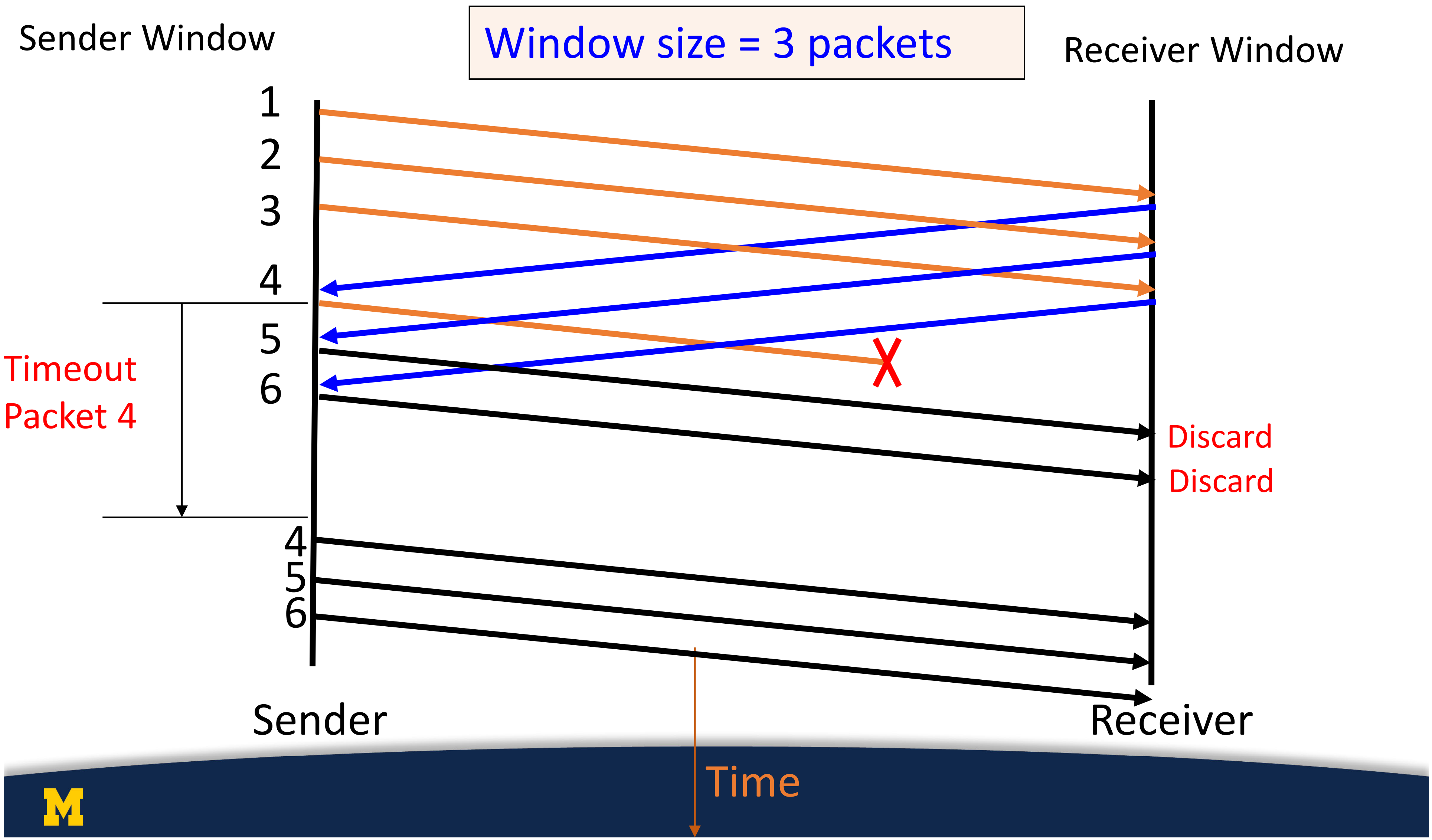
- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



GBN example w/o errors



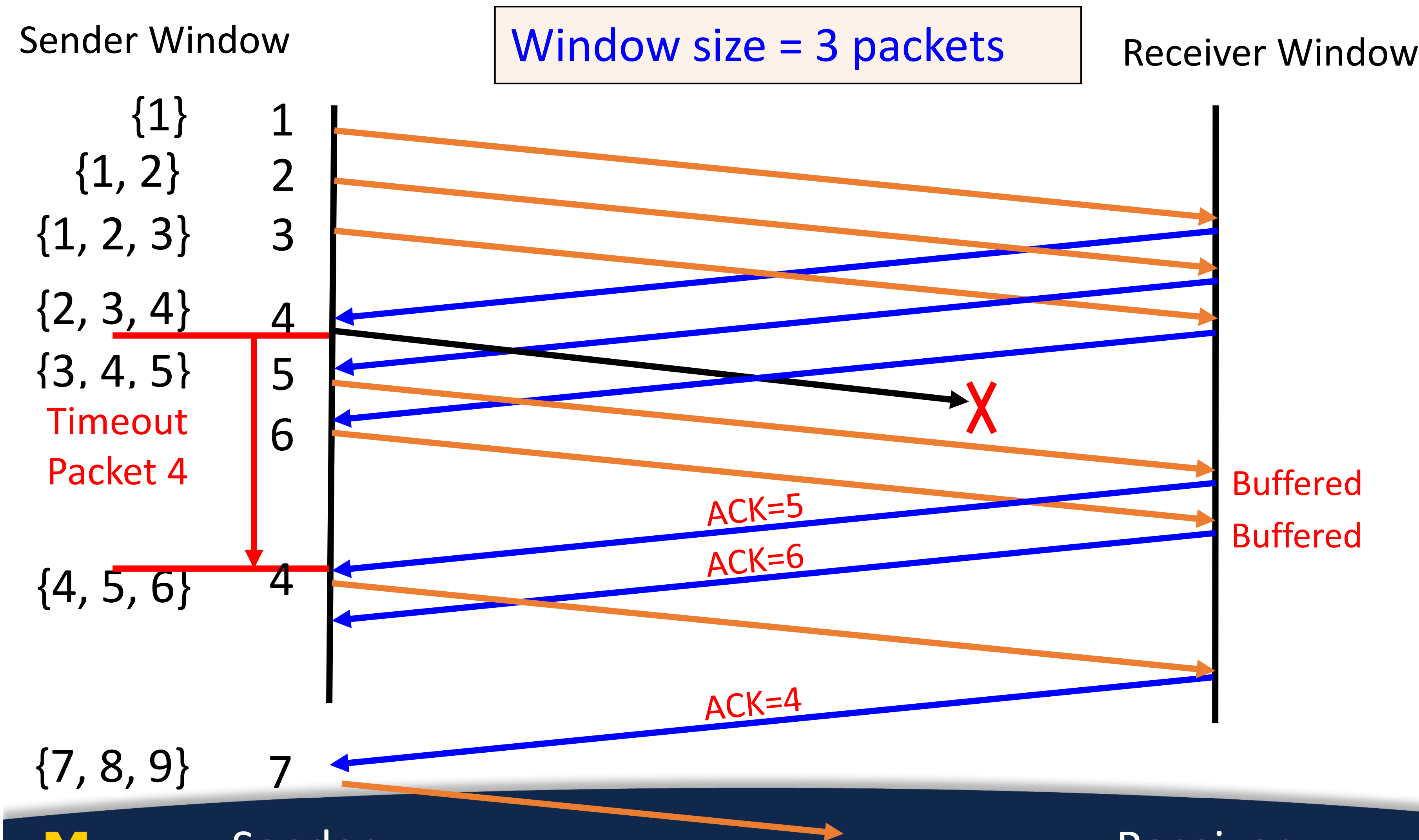
GBN example with errors



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

- For a large-enough window, it is possible to fully utilize a link with sliding windows
- 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 allows applications to communicate with each other
- Provides unreliable and reliable mechanisms
- Possible to build reliable transport over unreliable medium

- Next lecture
  - TCP

Bonus Quiz

- https://forms.gle/kqze7Eoi2Lva6KpJ7