The Transport Layer

CS168, Fall 2014
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Material thanks to Ion Stoica, Jennifer Rexford, Nick McKeown, and many other colleagues

Preliminaries

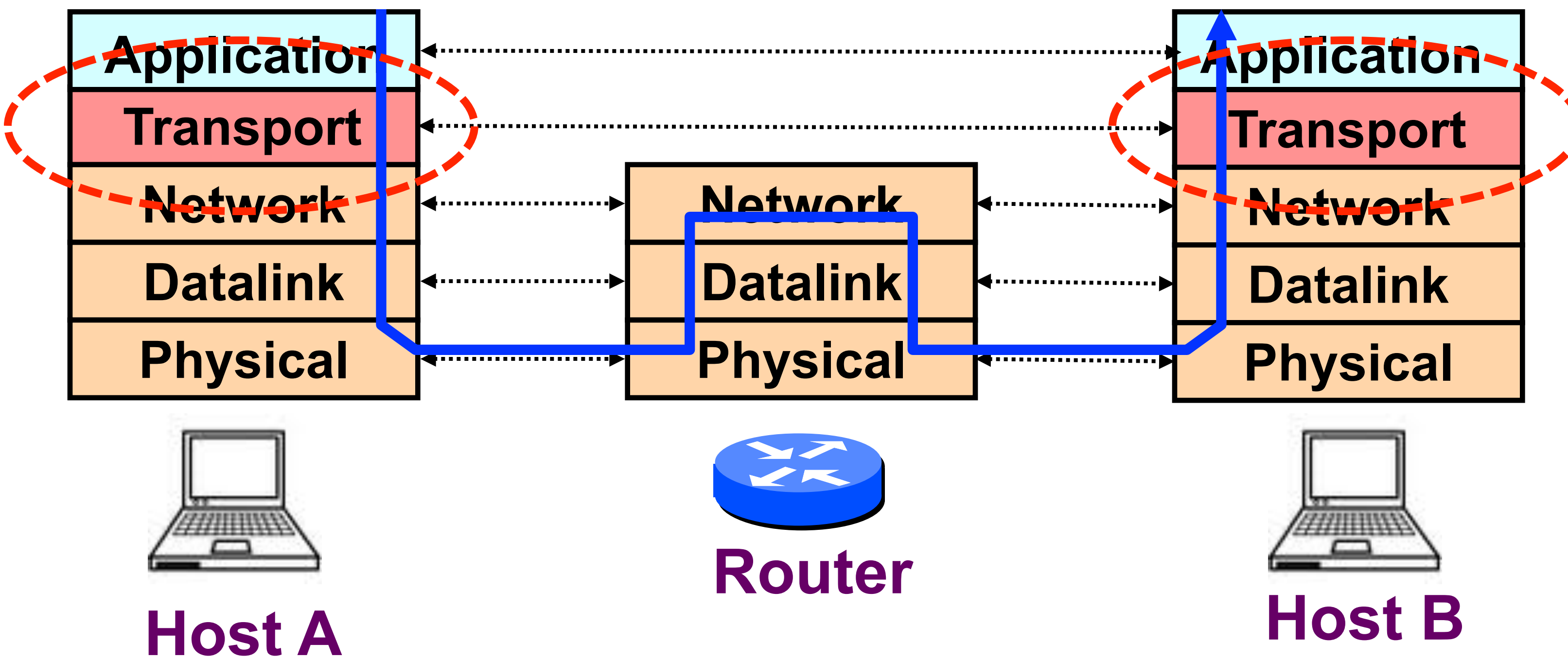
- Sylvia will be back next week
  - You are stuck with me this week
- Please ask questions….
- I will ask a few questions during this lecture
  - Someone should answer….
  - But for the rest of you, I ask questions to give you a chance to think, not because I want an answer…

The Transport Layer

(brief review from last lecture)

From Lecture#3: Transport Layer

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



Why a transport layer?

- Transport layer and application both on host
- Why not just combine the two?
- And what should that code do anyway?

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

Role of the Transport Layer

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

Role of the Transport Layer

- Communication between application 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, MTCP, 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, no-frills 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, no-frills transport protocol
- TCP is the whole-hog protocol
  - offers apps a reliable, in-order, bytestream abstraction
  - with congestion control
  - but no performance guarantees (delay, bw, etc.)

Transport Design Issues

Context: Applications and Sockets

- Socket: software abstraction by which an application process exchanges network messages with the (transport layer in the) operating system
  - socketID = socket(…, socket.TYPE)
  - socketID.sendto(message, …)
  - socketID.recvfrom(…)
  - will cover in detail after midterm

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

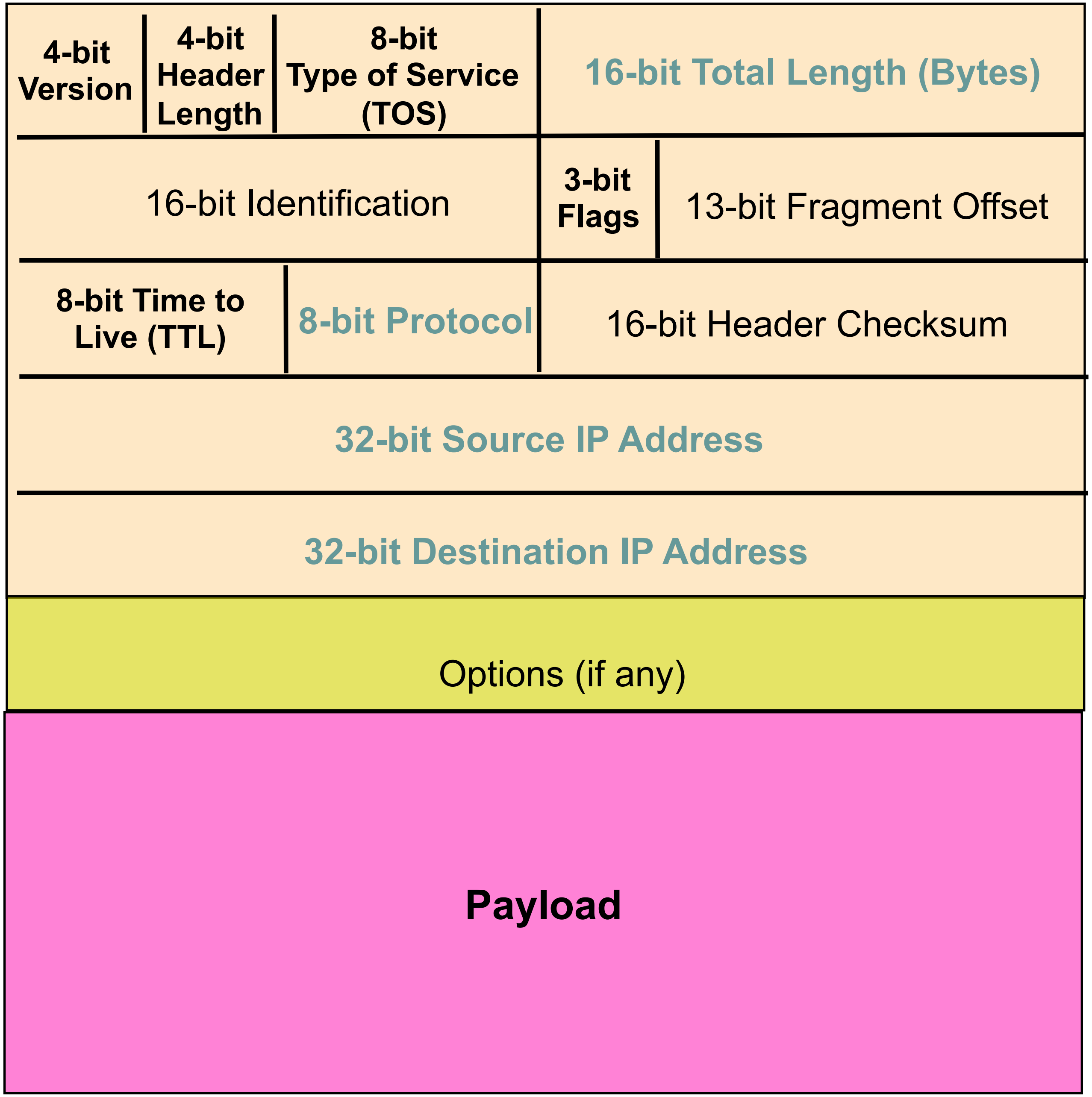
Ports

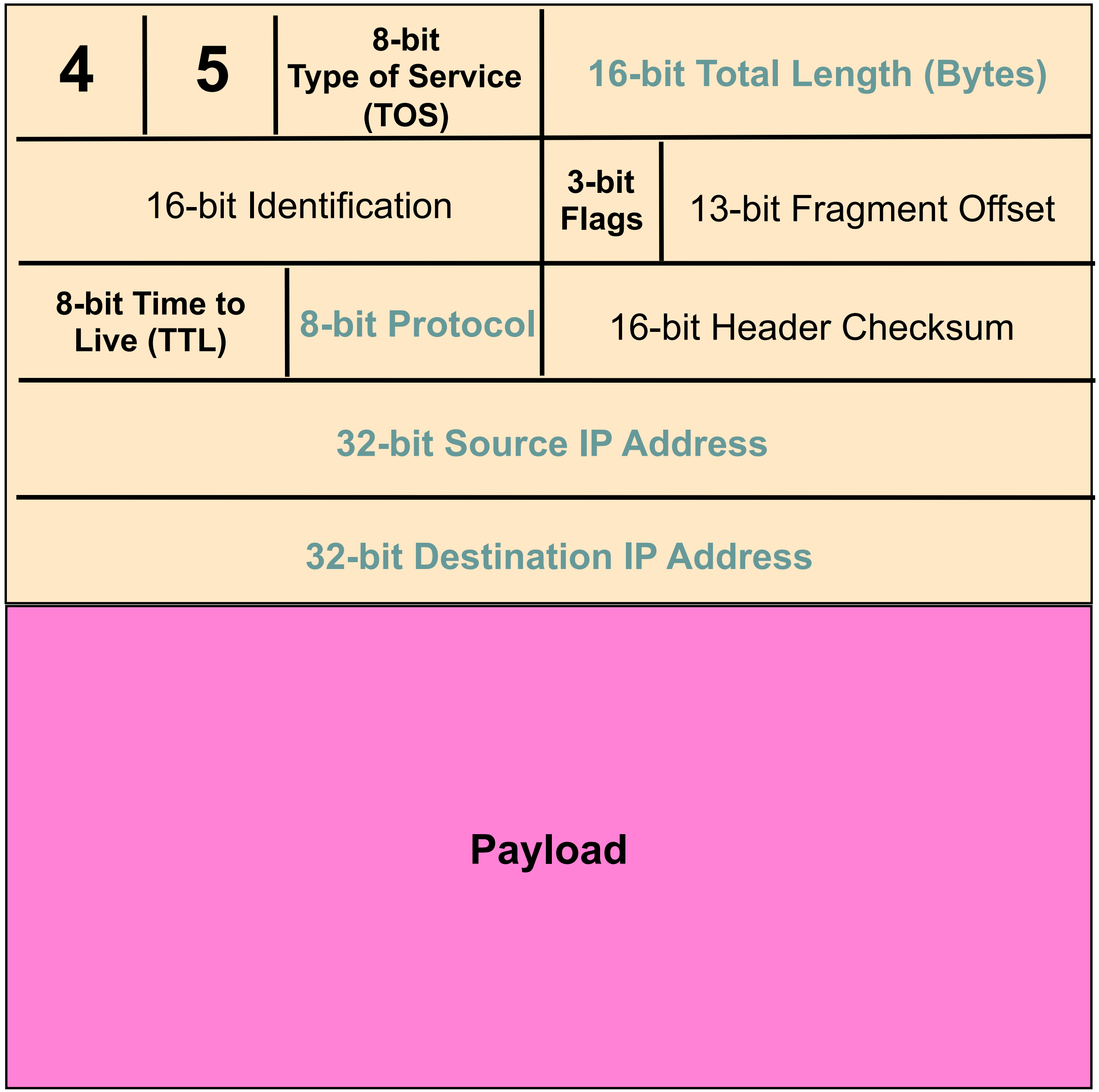
- Problem: deciding which app (socket) gets which packets
- Solution: ***port*** as a transport layer identifier (16 bits)
  - packet carries source/destination port numbers in transport header
- OS stores mapping between sockets and ports
  - **Port: in packets**
  - **Socket: in OS**
- For UDP ports (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

Two Questions

- Why the difference?
  - 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

- Why do you need to include local IP?


4-bit Version
4-bit Header Length
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
8-bit Protocol
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Options (if any)
Payload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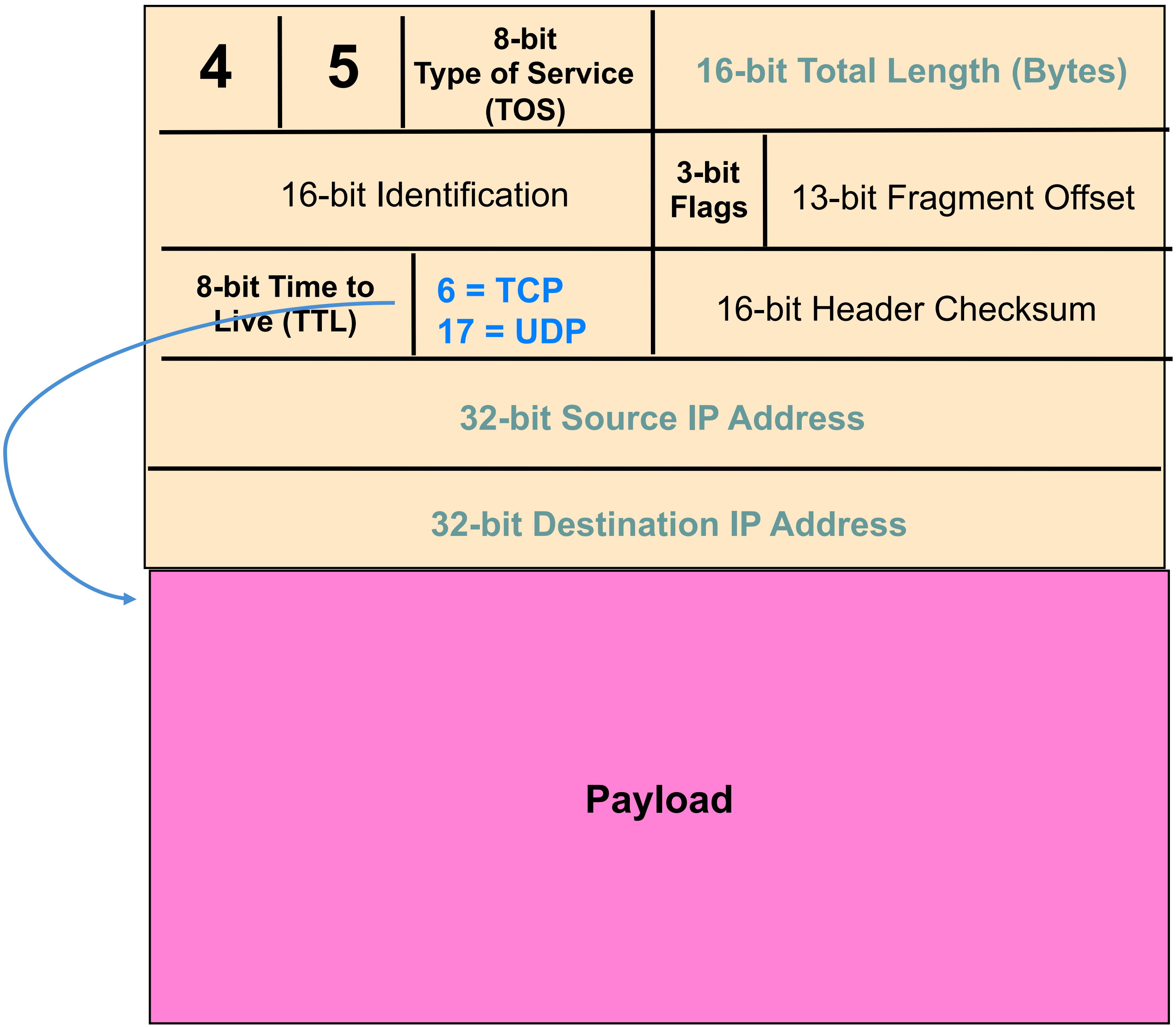
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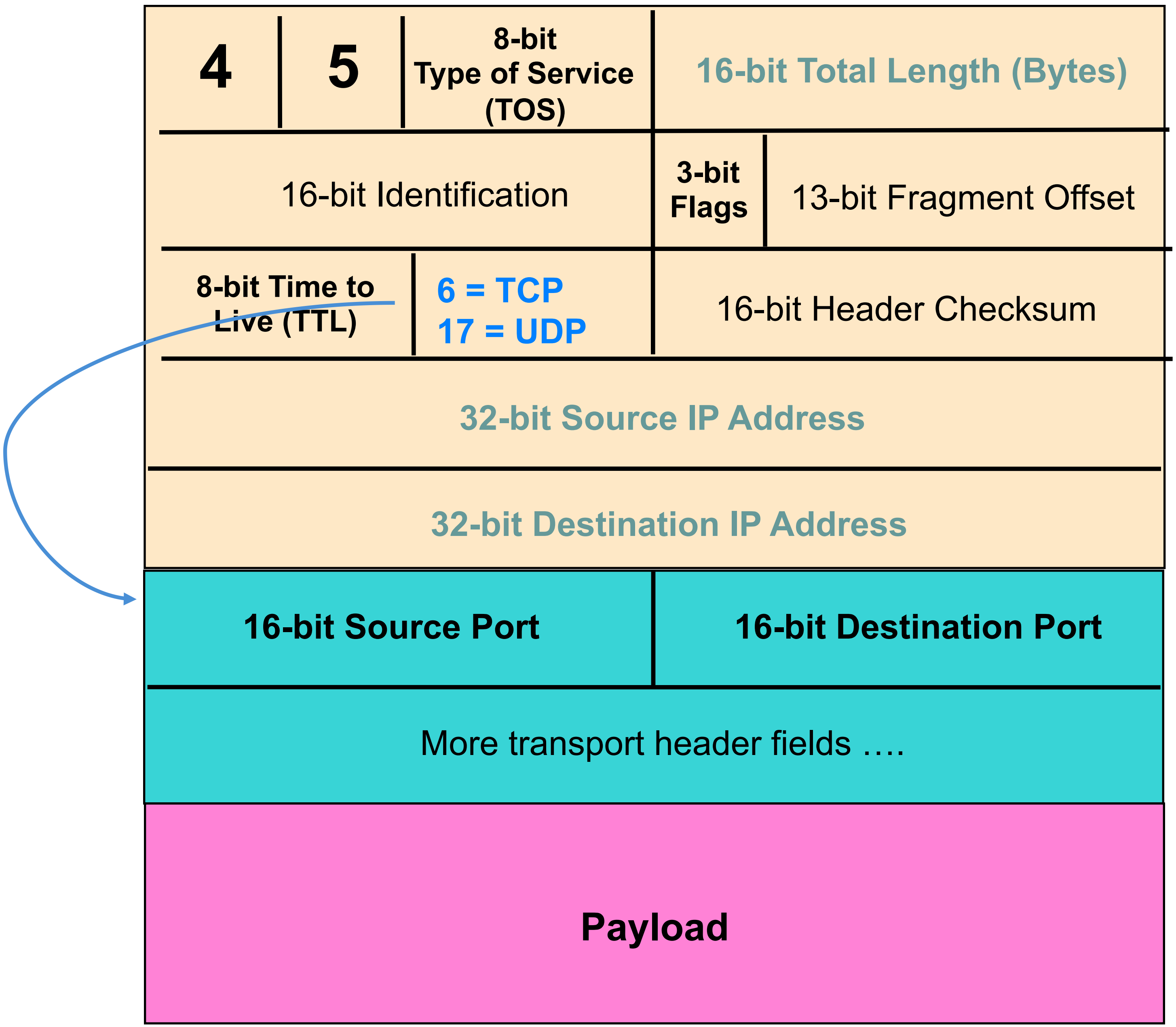
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Payload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6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
16-bit Source Port
16-bit Destination Port
More transport header fields ….
Payload

Recap: Multiplexing and Demultiplexing

- Host receives IP packets
  - Each IP header has source and destination IP address
  - Each Transport Layer header has source and destination port number

- Host uses IP addresses and port numbers to direct the message to appropriate socket
  - UDP maps local destination port and address to socket
  - TCP maps address pair and port pair to socket

Rest of Lecture

- More on ports
- UDP
- Reliable Transport
- Next lecture: Details of TCP

More on Ports

- Separate 16-bit port address space for UDP and TCP
- “Well known” ports (0-1023): everyone agrees which services run on these ports
  - e.g., ssh:22, http:80
  - helps client know server’s port
  - Services can listen on well-known port
- Ephemeral ports (most 1024-65535): given to clients

UDP: User Datagram Protocol

- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery

- UDP described in RFC 768 – (1980!)
  - Destination IP address and port to support demultiplexing
  - Optional error checking on the packet contents
    - (checksum field = 0 means "don't verify checksum")

SRC port	DST port
checksum	length
DATA	

Question

- Why do UDP packets carry the sender's port?

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

Reliable Transport

- In a perfect world, reliable transport is easy

@Sender
- send packets

@Receiver
- wait for packets

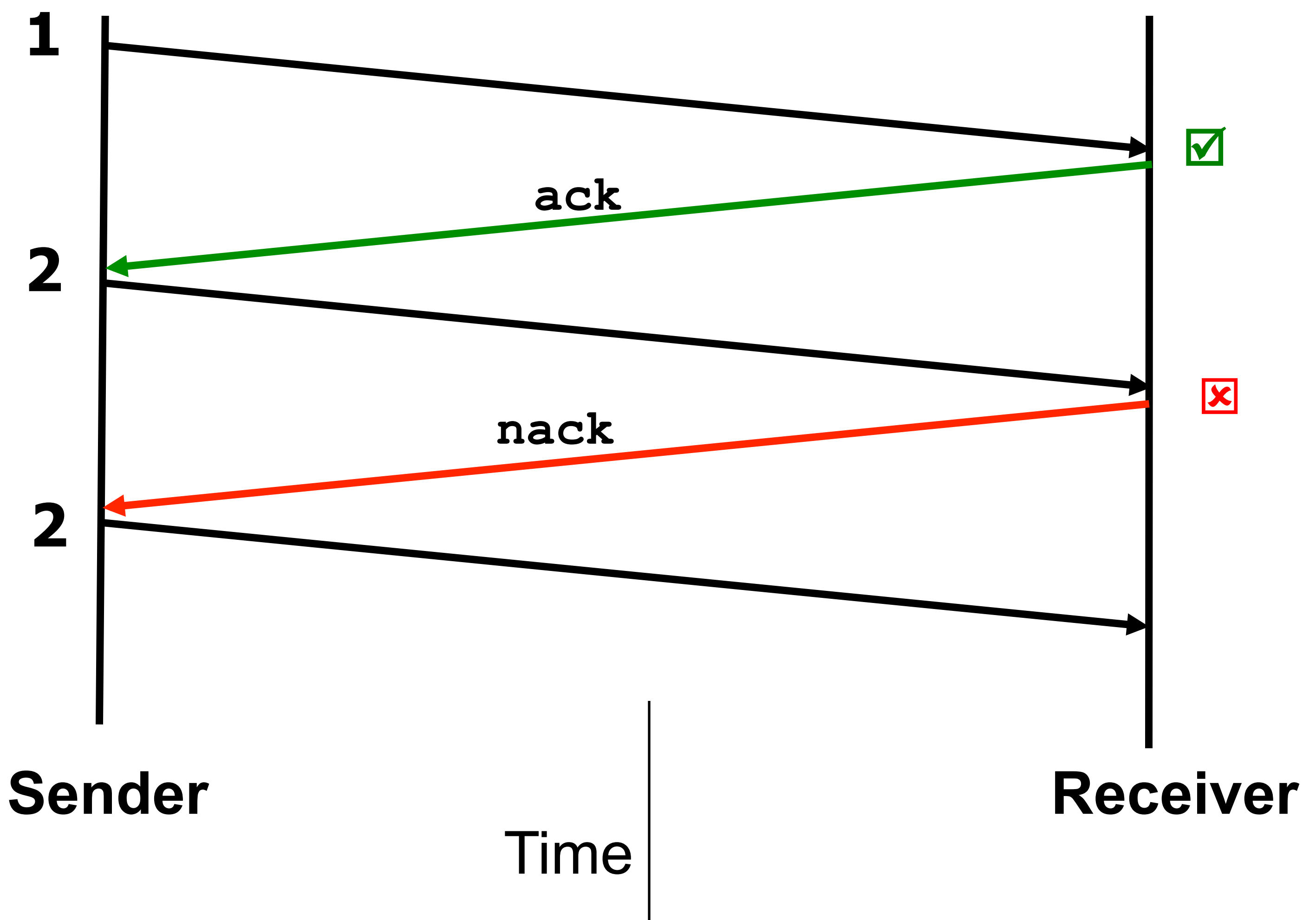
Reliable Transport

- 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?*)

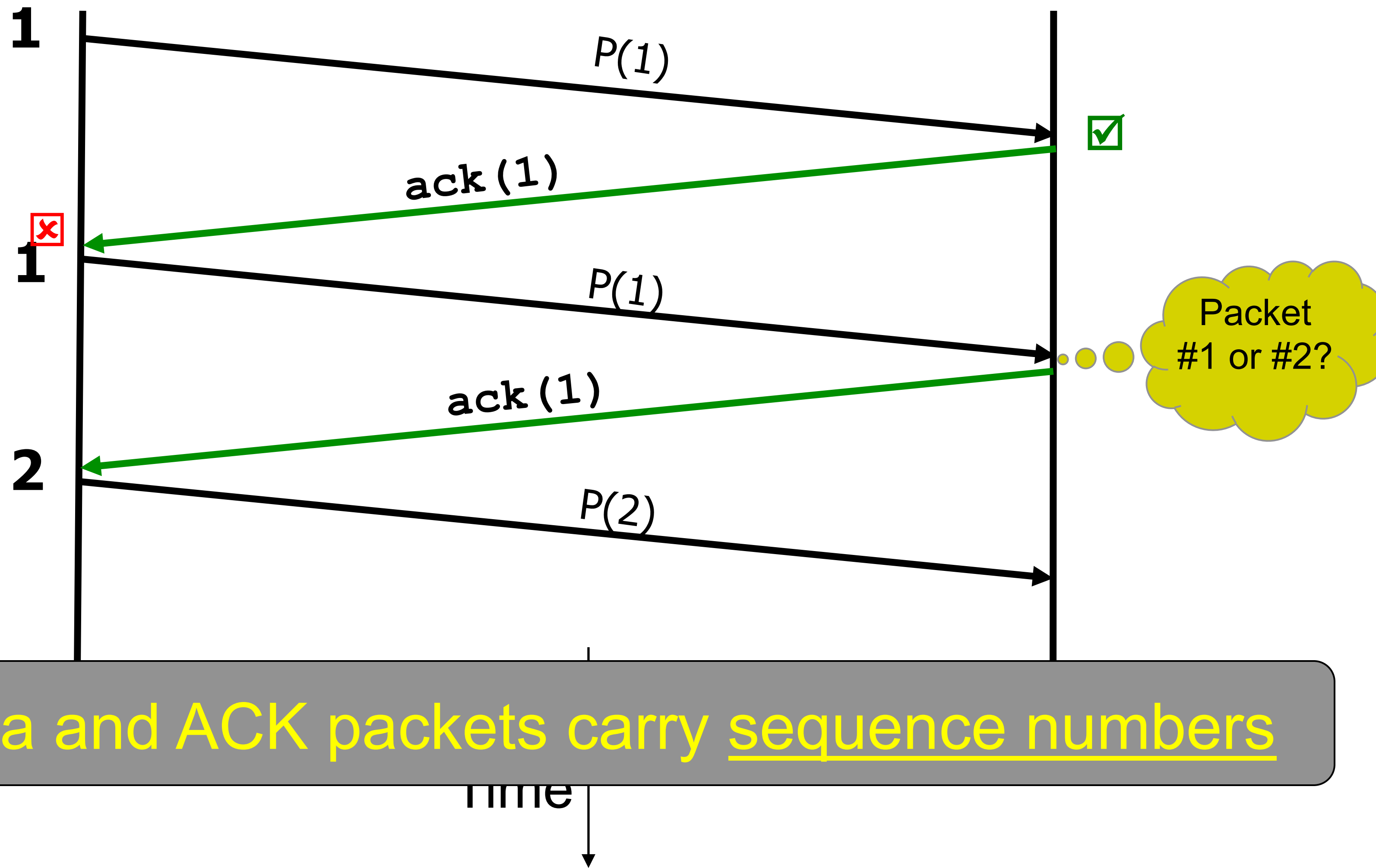
Reliable Transport

- 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 a packet
  - *Forward error correction: a way to mask errors without retransmission*
  - *Network encoding: an efficient way to repair errors*
  - ….

Dealing with Packet Corruption

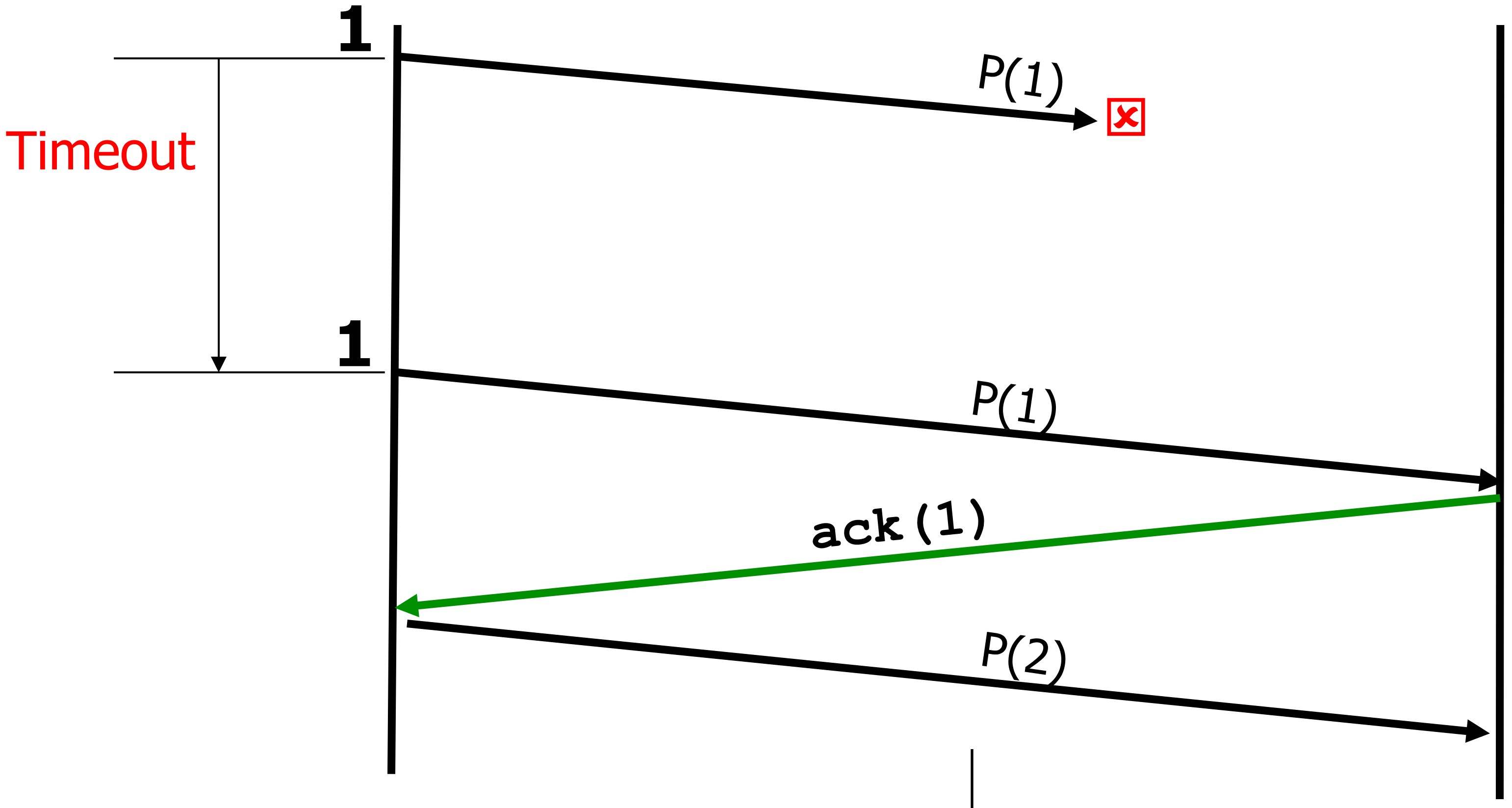


Dealing with Packet Corruption



Data and ACK packets carry sequence numbers

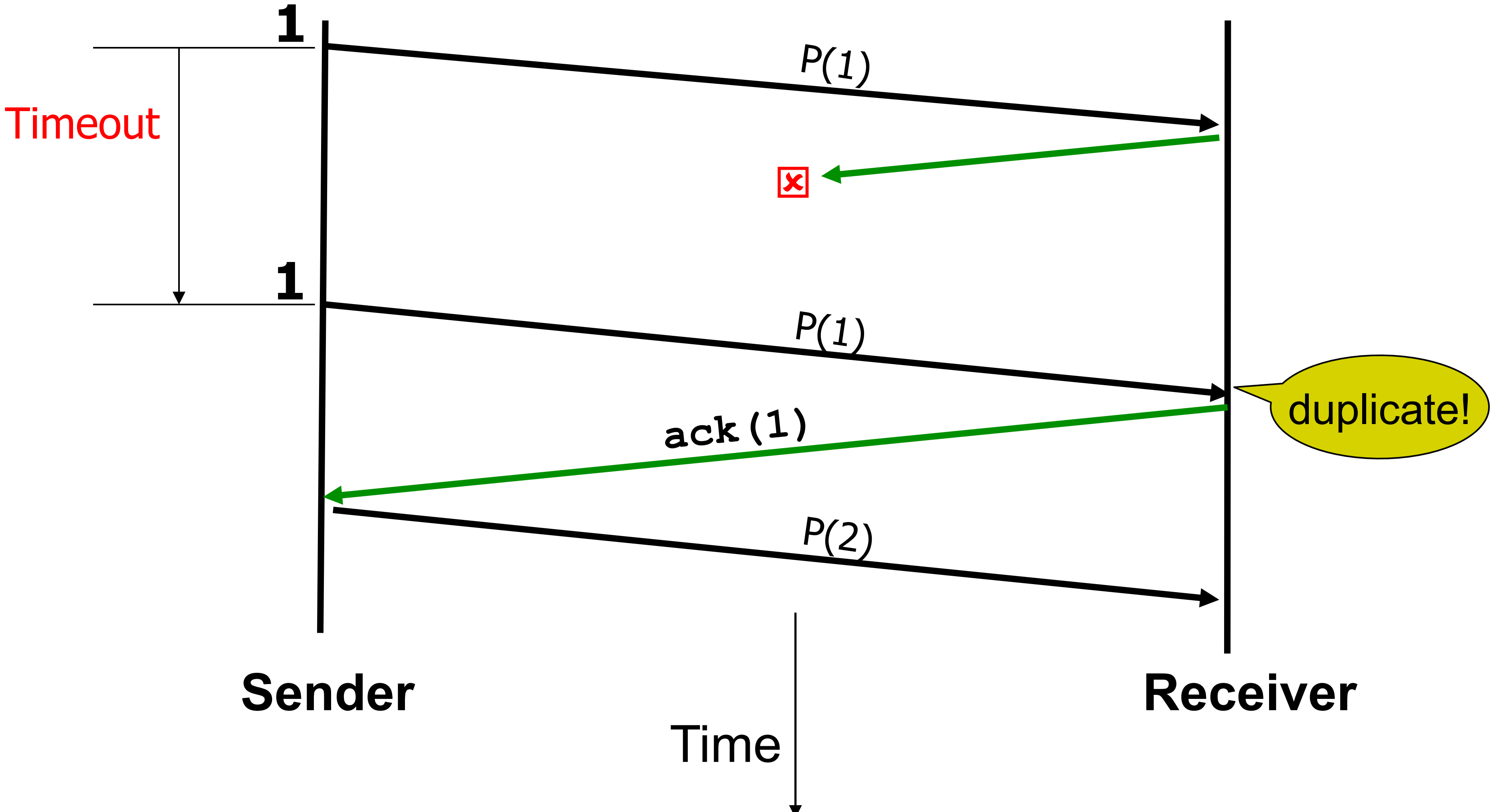
Dealing with Packet Loss



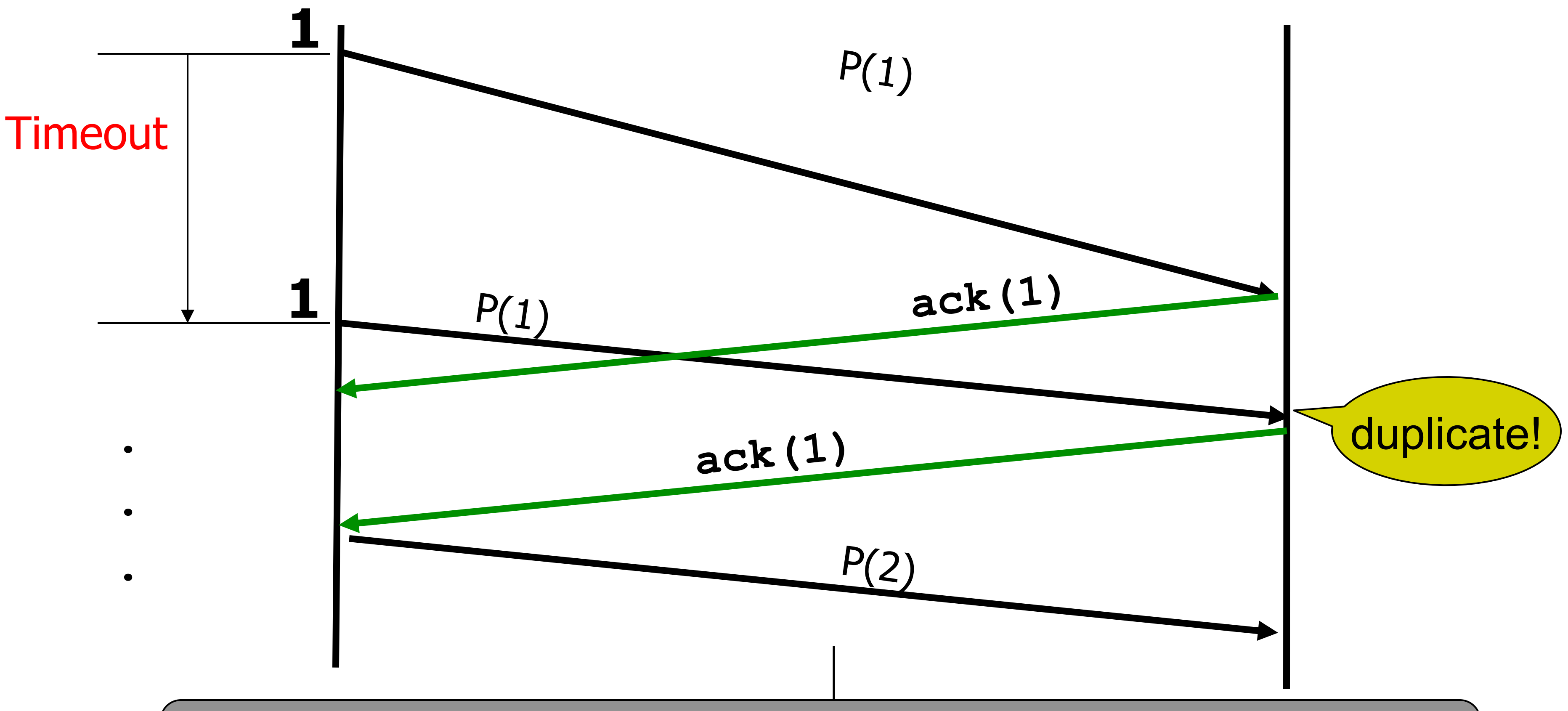
<u>Timer-driven loss detection</u>

Set timer when packet is sent; retransmit on timeout

Dealing with Packet Loss (of ack)



Dealing with Packet Loss



Timer-driven retx. can lead to duplicates

Components of a solution (so far)

- checksums (to detect bit errors)
- timers (to detect loss)
- acknowledgements (positive or negative)
- sequence numbers (to deal with duplicates)

- But we haven’t put them together into a coherent design…

Designing Reliable Transport

A Solution: “Stop and Wait”

@Sender

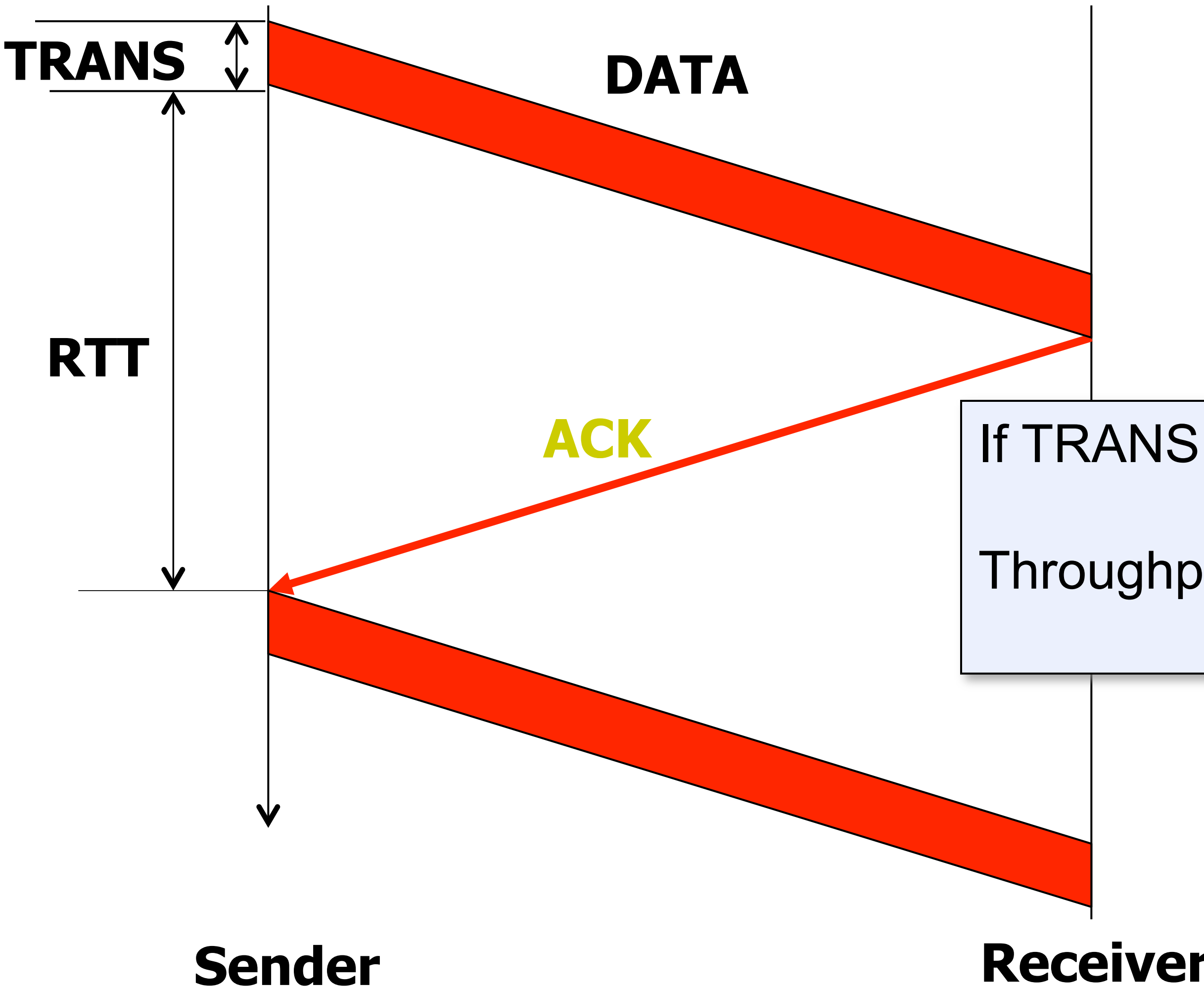
- send packet(I); (re)set timer; wait for ack
- If (ACK)
  - I++; repeat
- If (NACK or TIMEOUT)
  - repeat

@Receiver

- wait for packet
- if packet is OK, send ACK
- else, send NACK
- repeat

- We have a correct reliable transport protocol!
- Probably the world’s most inefficient one (*why?*)

Stop & Wait is Inefficient



If TRANS << RTT then

Throughput ~ DATA/RTT

Orders of Magnitude

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

Three Design Decisions

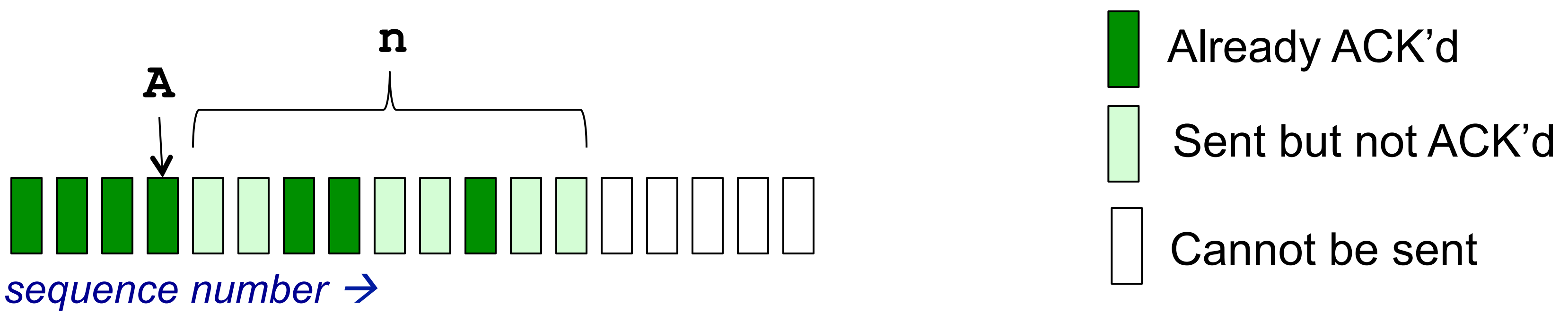
- Which packets can sender send?
  - Sliding window
- How does receiver ack packets?
  - Cumulative
  - Selective
- Which packets does sender resend?
  - GBN
  - Selective repeat

Sliding Window

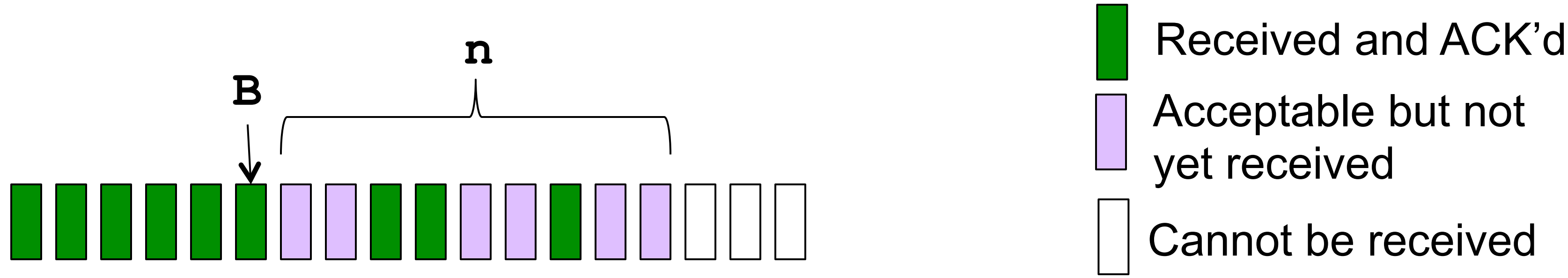
- window = set of adjacent sequence numbers
  - The size of the set is the window size; assume window size is *n*
- General idea: send up to *n* packets at a time
  - Sender can send packets in its window
  - Receiver can accept packets in its window
  - Window of acceptable packets "slides" on successful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

- Compare to Stop and Wait: Data/RTT

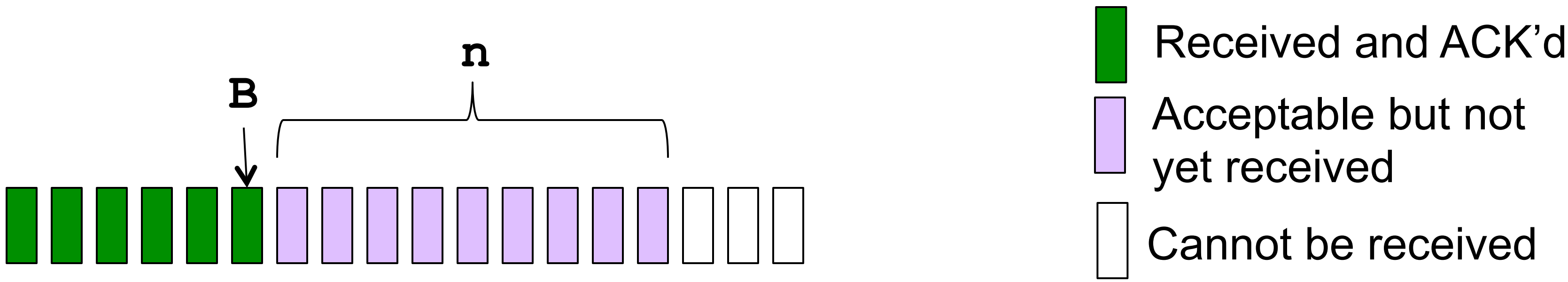
- Two questions:
  - What happens when n gets too large?
  - How do we choose n?

Acknowledgements w/ Sliding Window

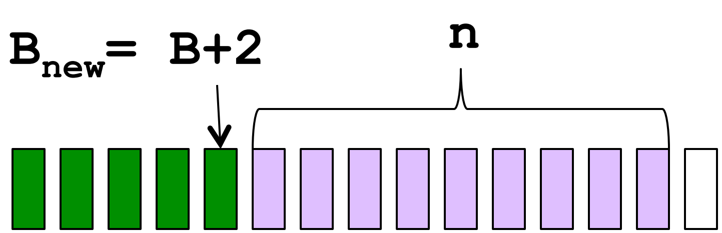
- Two common options
  - cumulative ACKs: ACK carries next in-order sequence number that the receiver expects

Cumulative Acknowledgements (1)

- At receiver



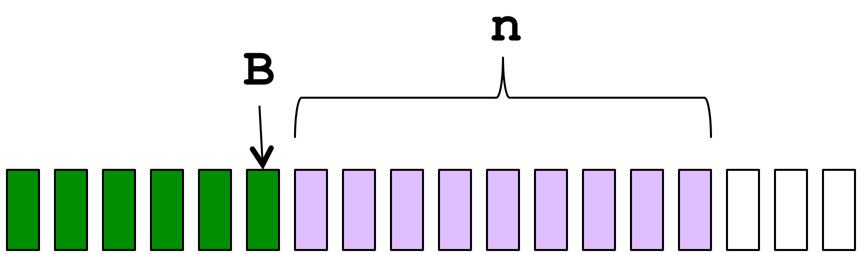
- After receiving B+1, B+2

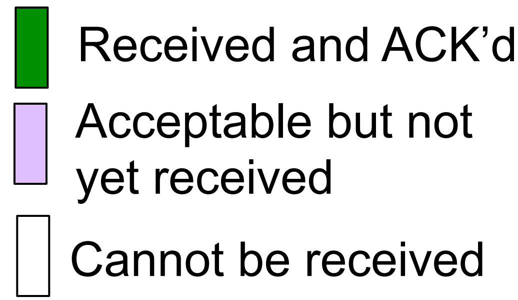


- Receiver sends ACK(B+3) = ACK(B_{new}+1)

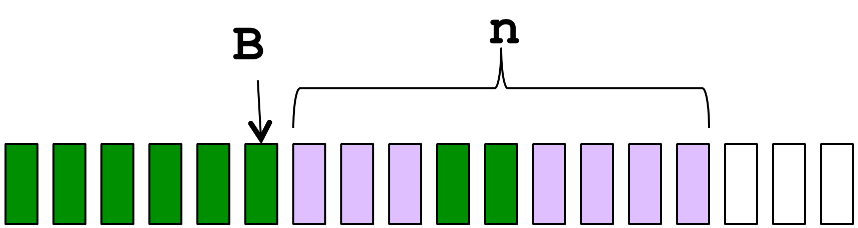
Cumulative Acknowledgements (2)

- 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

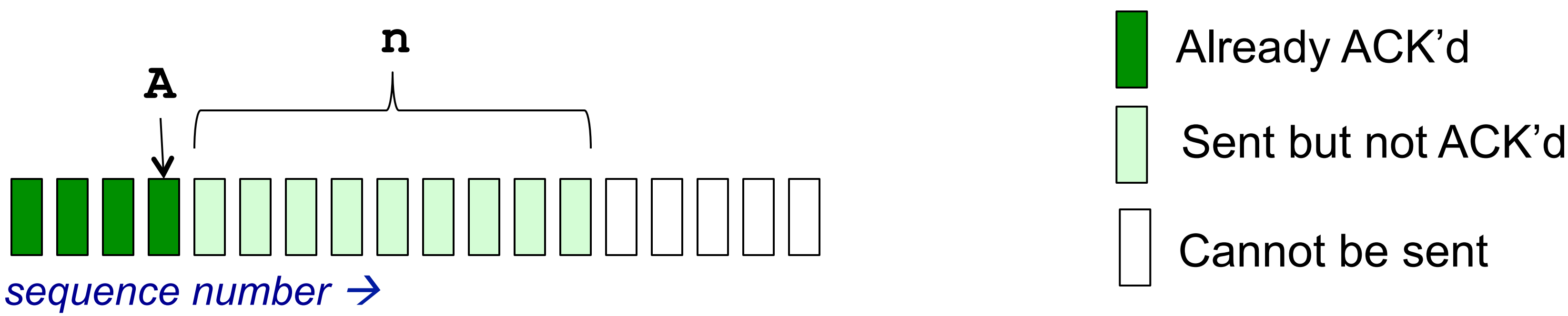
- Resending packets: two canonical approaches
  - Go-Back-N
  - Selective Repeat
- Many variants that differ in implementation details

Go-Back-N (GBN)

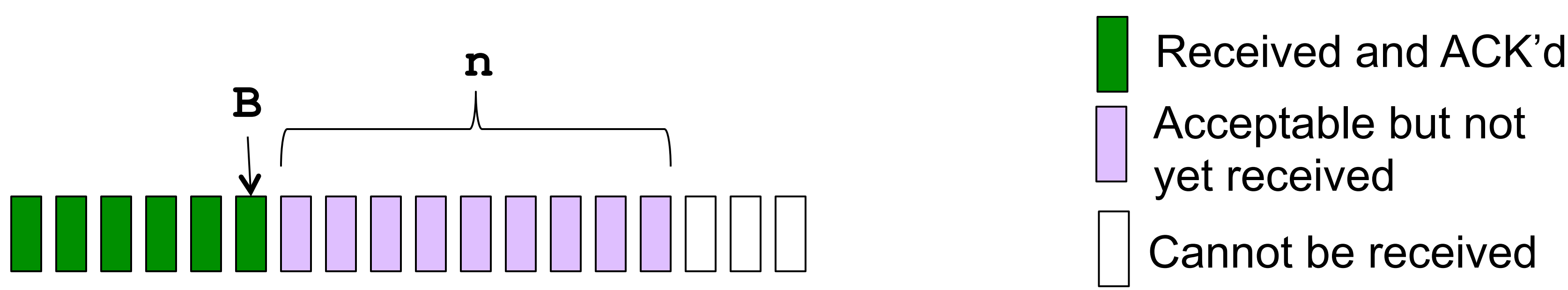
- Sender transmits up to *n* unacknowledged packets
- Receiver only accepts packets in order
  - discards out-of-order packets (i.e., packets other than *B+1*)
- Receiver uses cumulative acknowledgements
  - i.e., sequence# in ACK = next expected in-order sequence#
- Sender sets timer for 1st outstanding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

Sliding Window with GBN

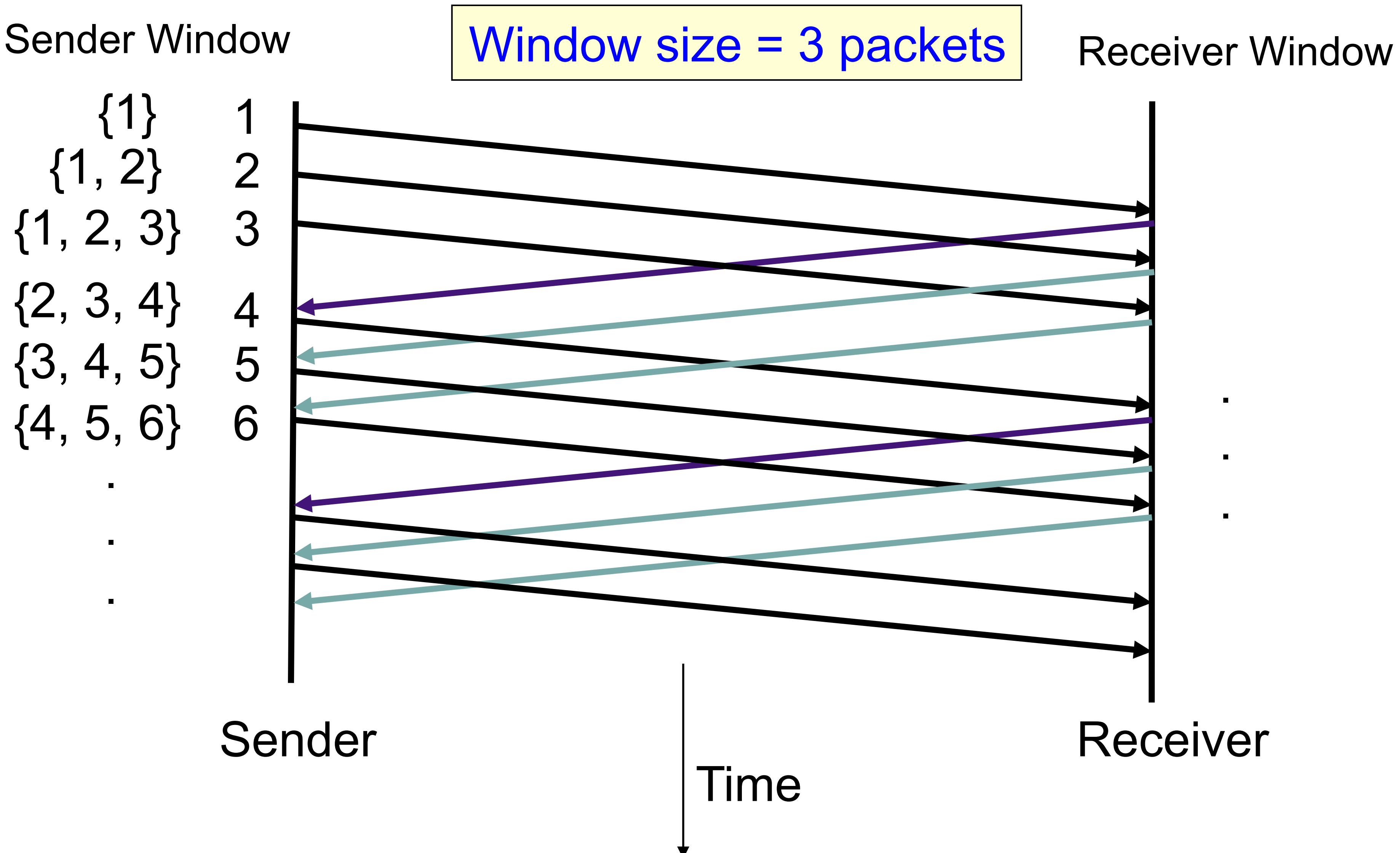
- Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



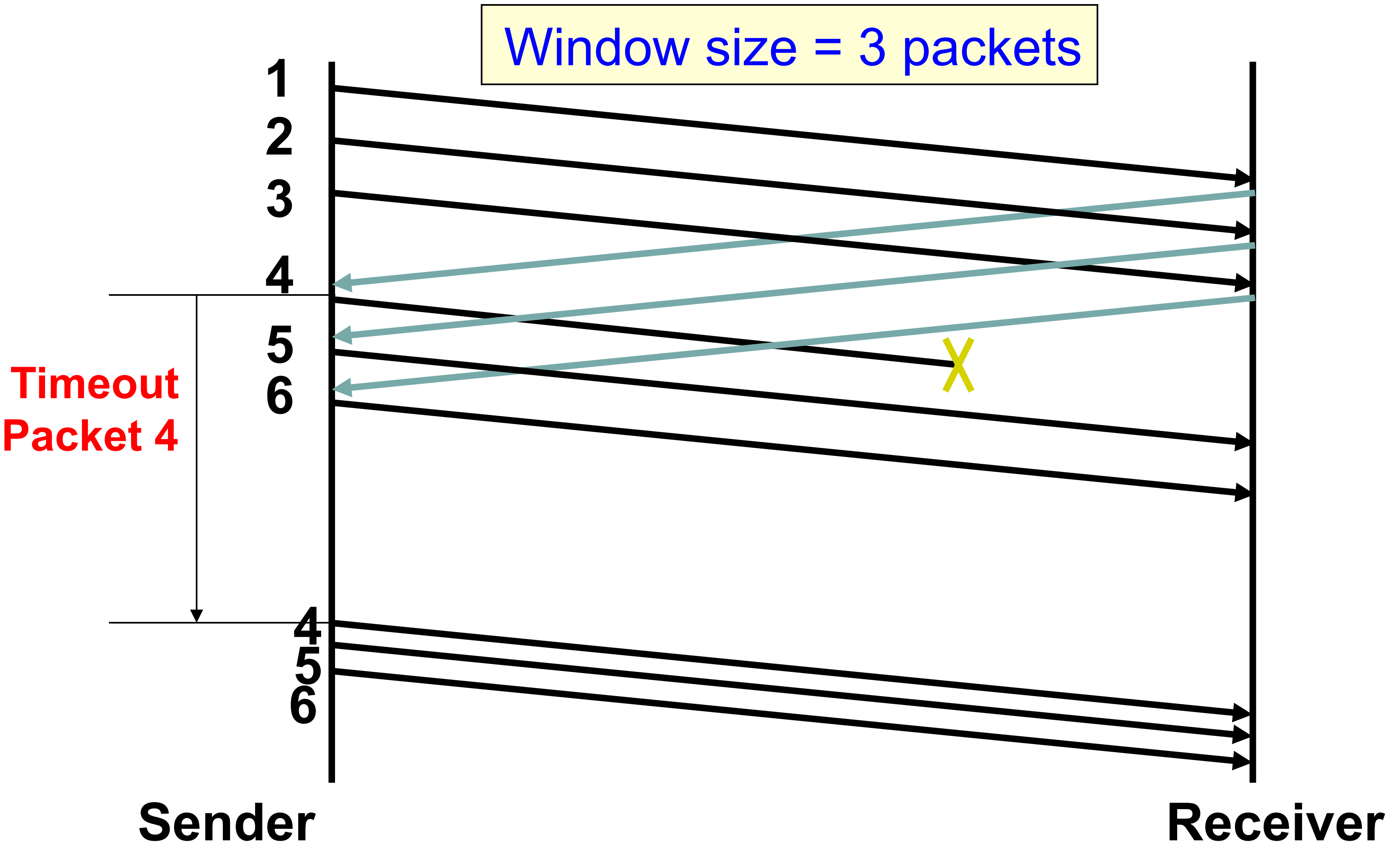
- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



GBN Example w/o Errors



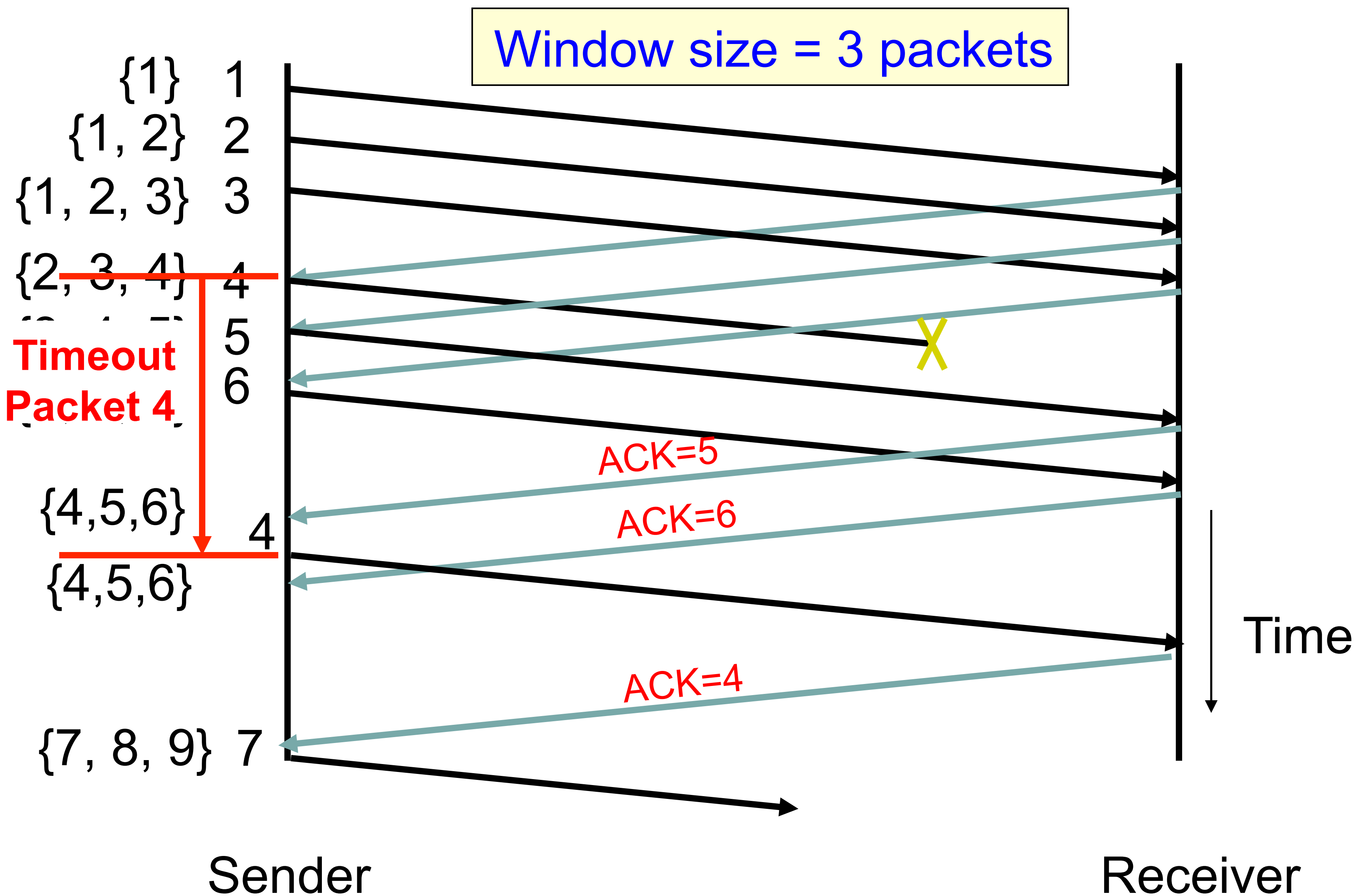
GBN Example with Errors



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 would SR be better?

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)

Recap: 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

What does TCP do?

Most of our previous tricks + a few differences

- Sequence numbers are byte offsets
- Sender and receiver maintain a sliding window
- Receiver sends cumulative acknowledgements (like GBN)
- Sender maintains a single retx. timer
- Receivers do not drop out-of-sequence packets (like SR)
- Introduces fast retransmit : optimization that uses duplicate ACKs to trigger early retx (next time)
- Introduces timeout estimation algorithms (next time)

Next Time

- TCP
  - Reliability
  - Congestion control