The Transport Layer

CPSC 433/533, Spring 2021 Anurag Khandelwal

Administrivia

• HWI grades posted, HW2 out!

- Any issues with HWI grading (e.g., regrade request), email Ramla, cc-me
- Hopefully Project# I is going well due next Tue!
 - No class next Tue though (break day)...

Periodic check-in:

• Are you keeping up with the course fine? If not, reach out! Email, come to OH...

Best-effort global delivery of packets

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- Control Plane: Routing

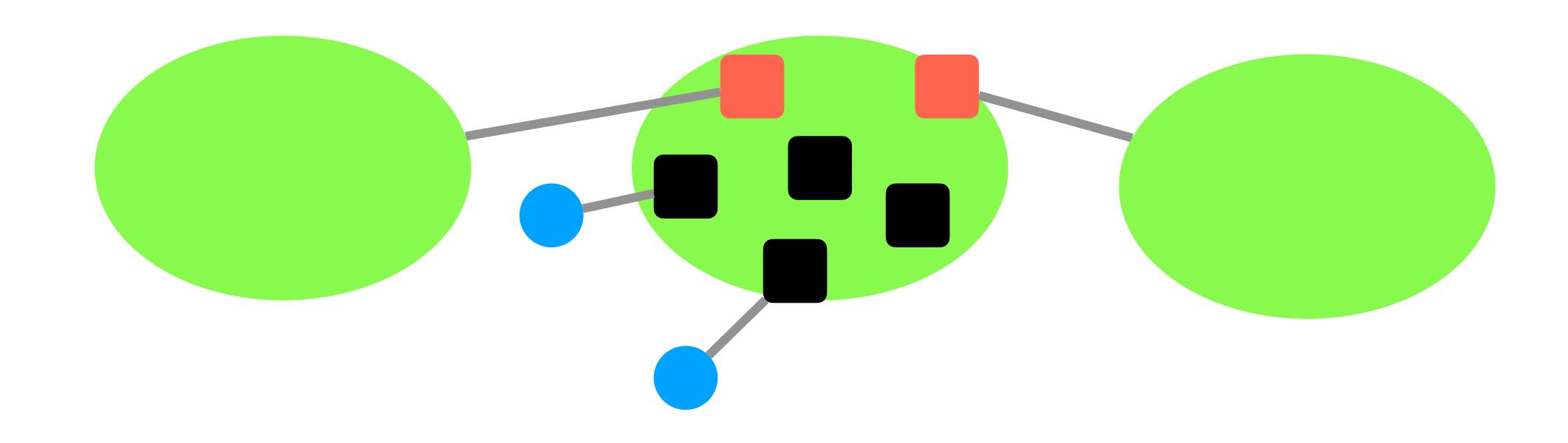
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- Data Plane: Forwarding

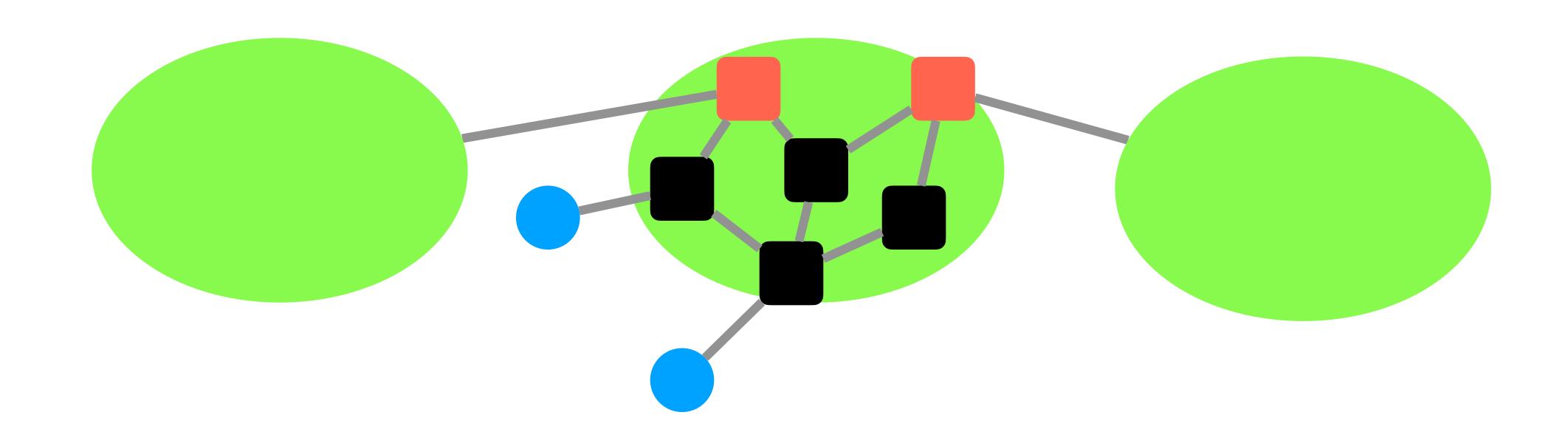
- Best-effort global delivery of packets
- Control Plane: Routing
- Data Plane: Forwarding
- Key enabler of scalability: Addressing

• Hierarchical address structure

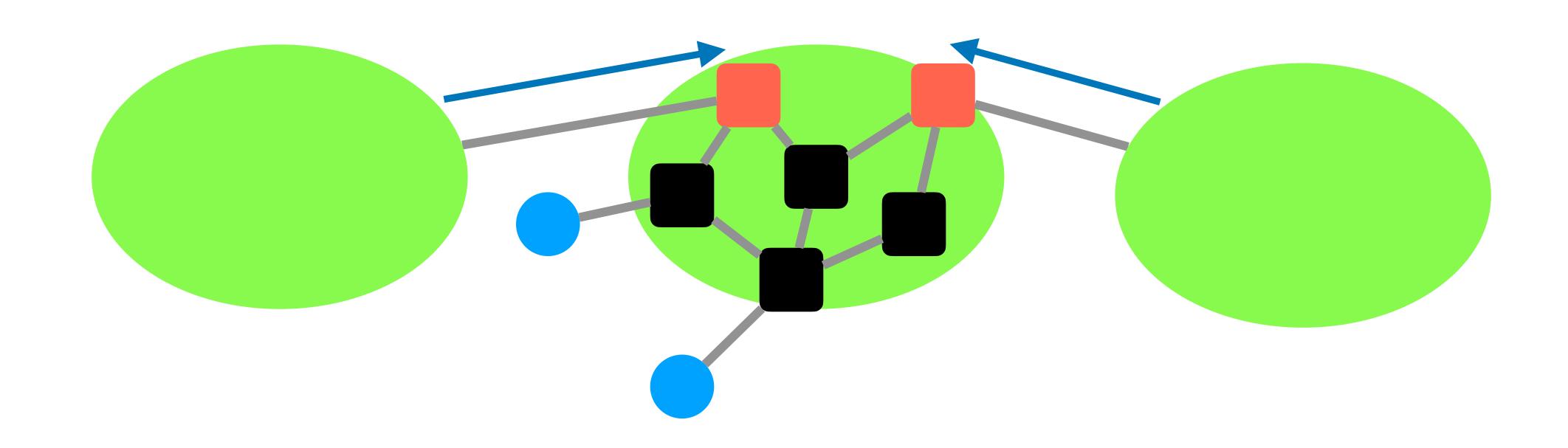
- Hierarchical address structure
- Hierarchical address allocation

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- Hierarchical address allocation
- Hierarchical addresses and routing scalability

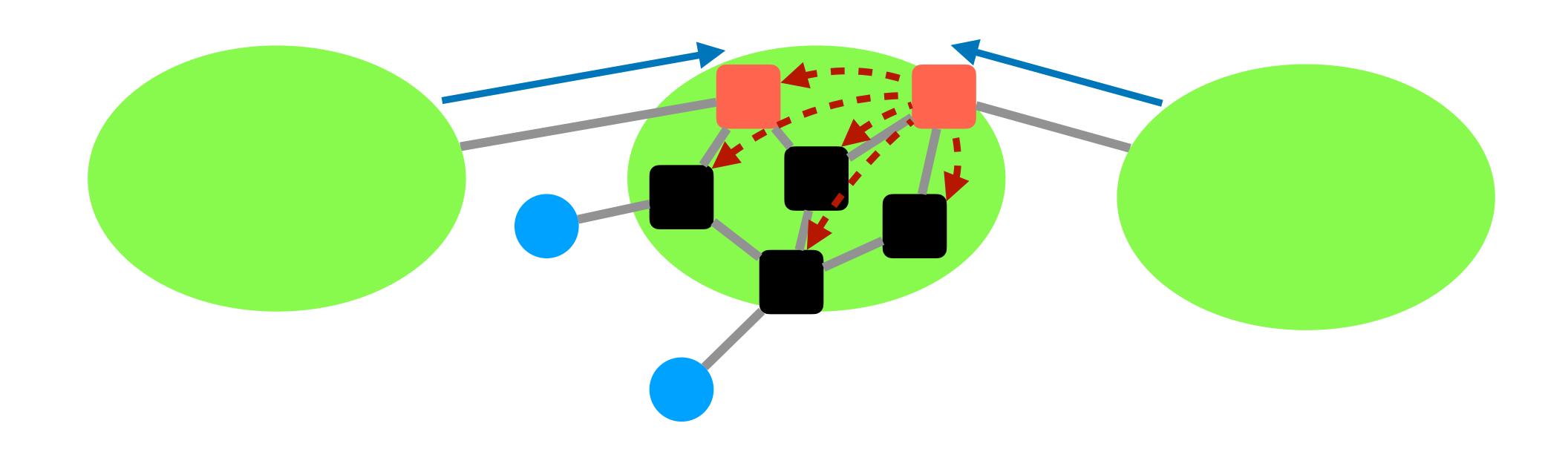




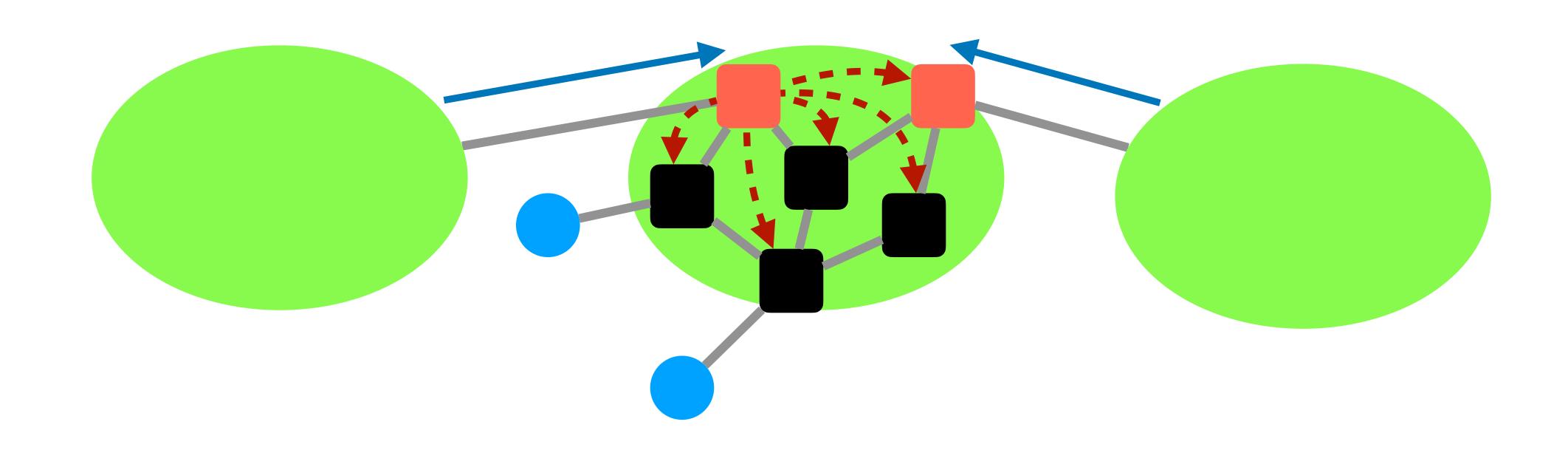
1. Provide internal reachability (IGP)



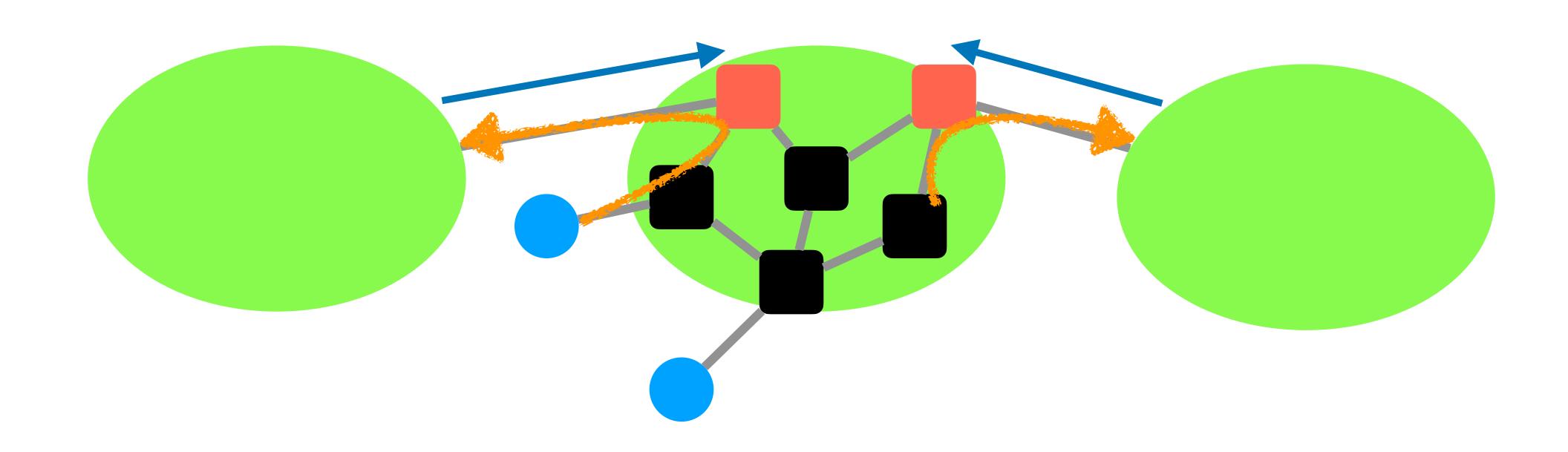
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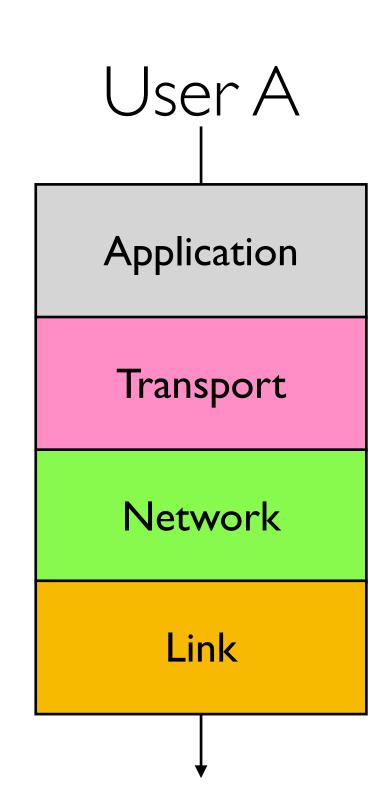
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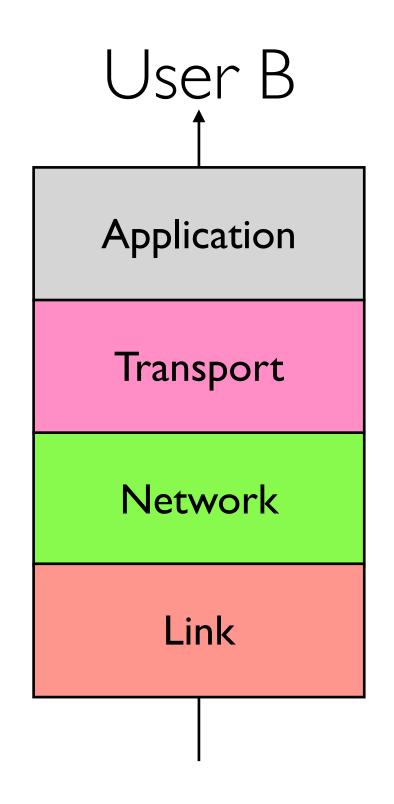


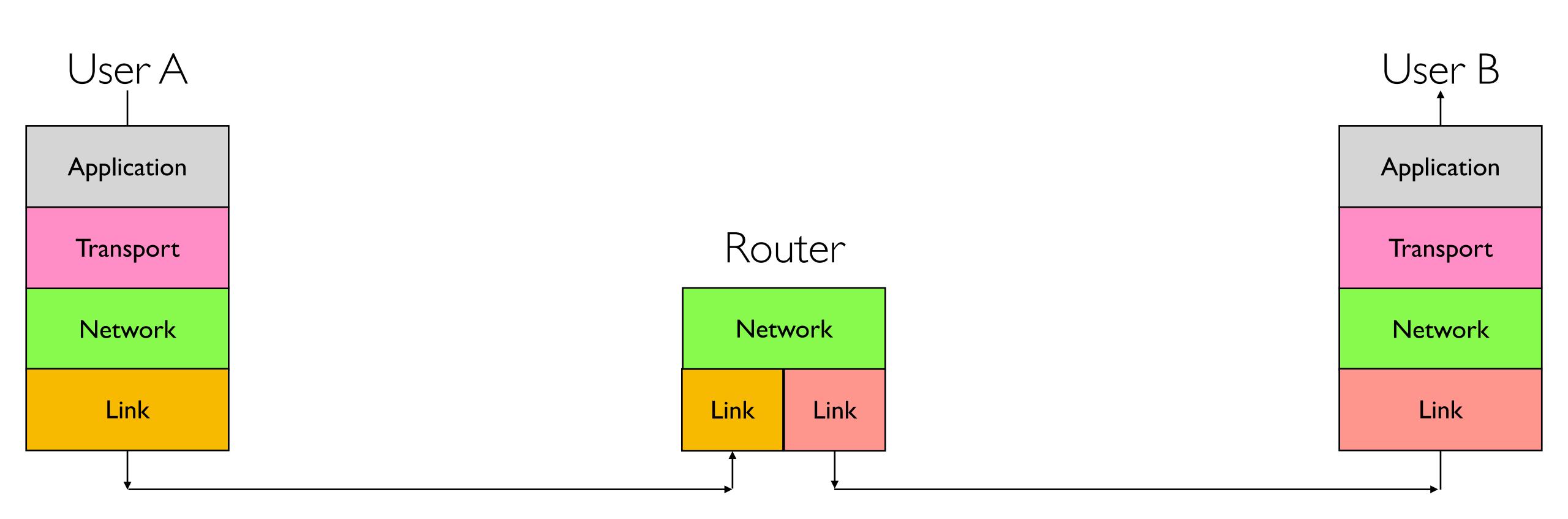
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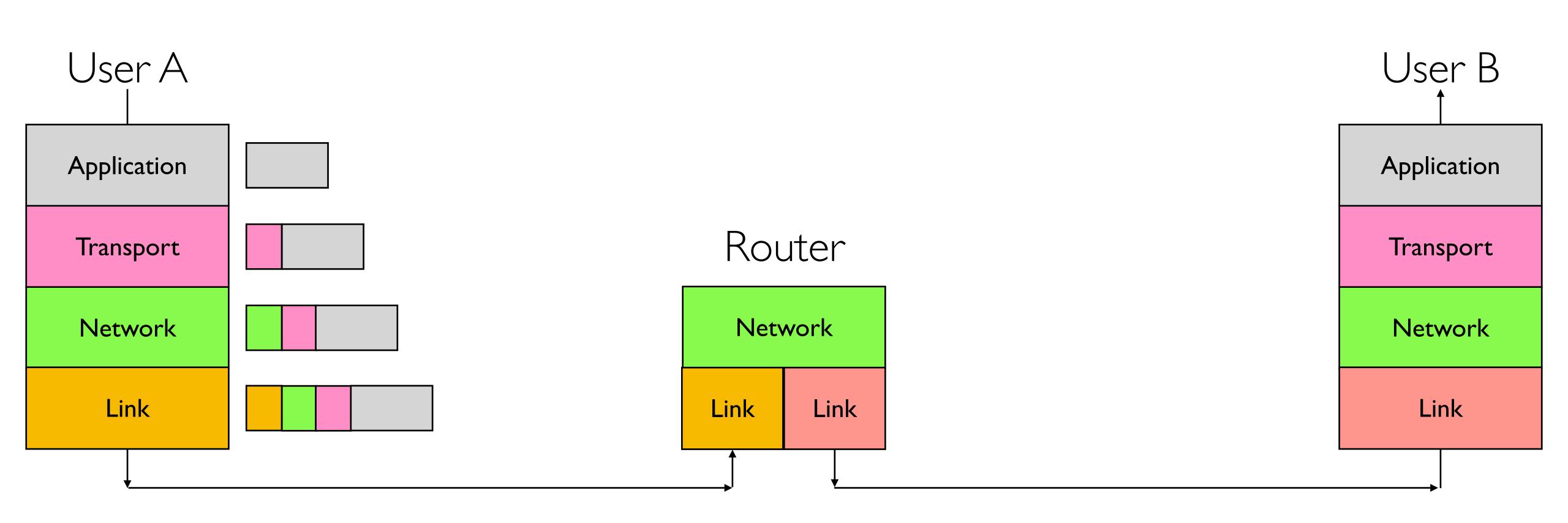


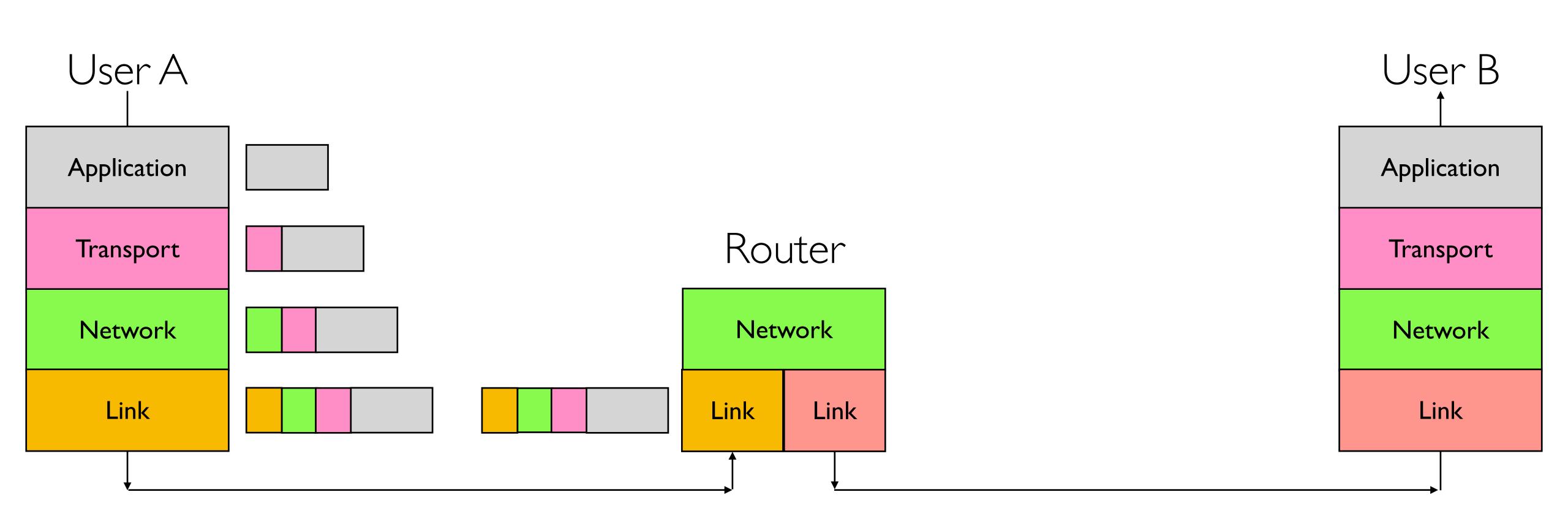
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- 4. Travel shortest path to egress (**IGP**)

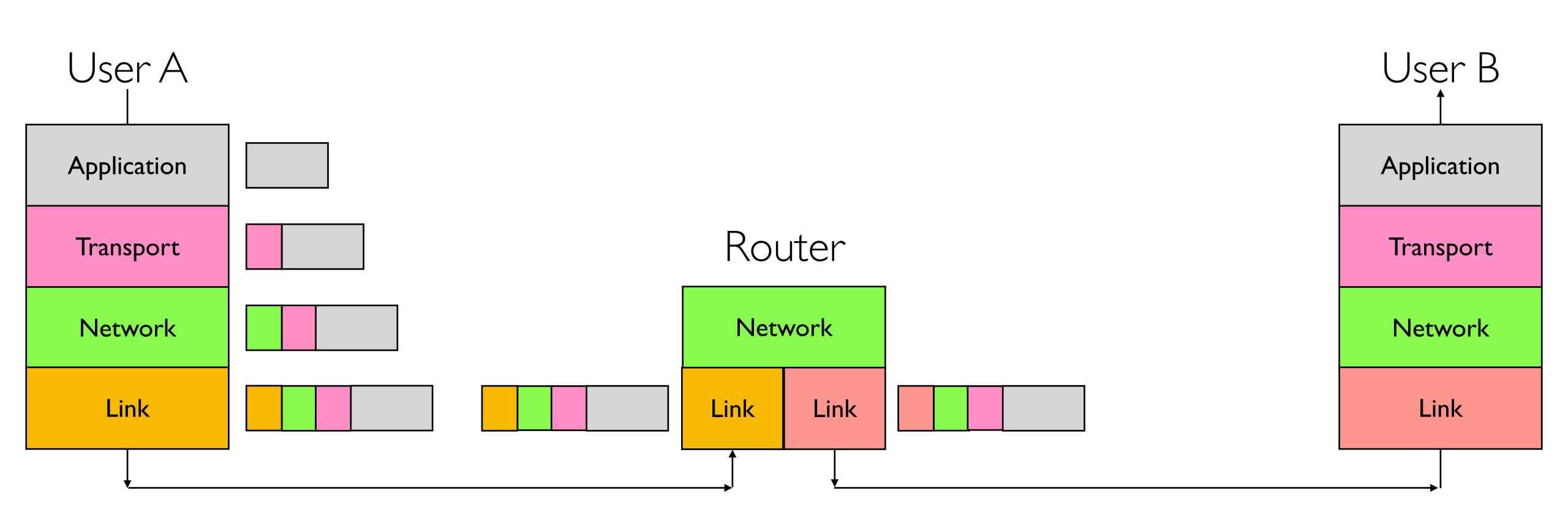


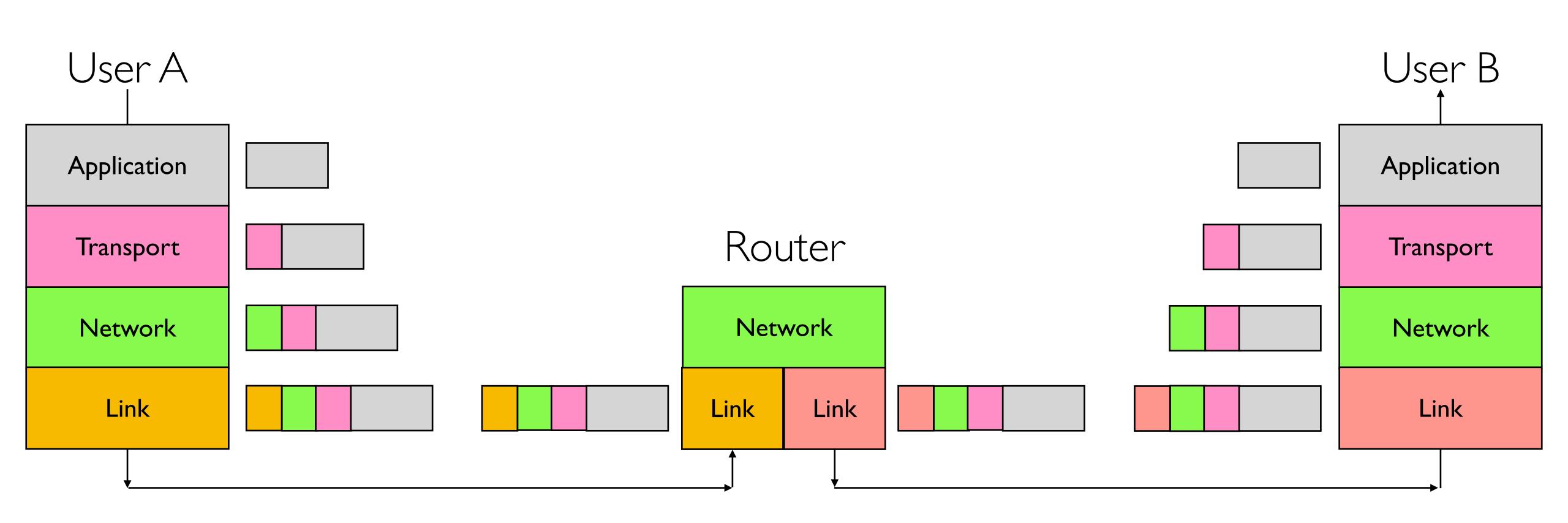


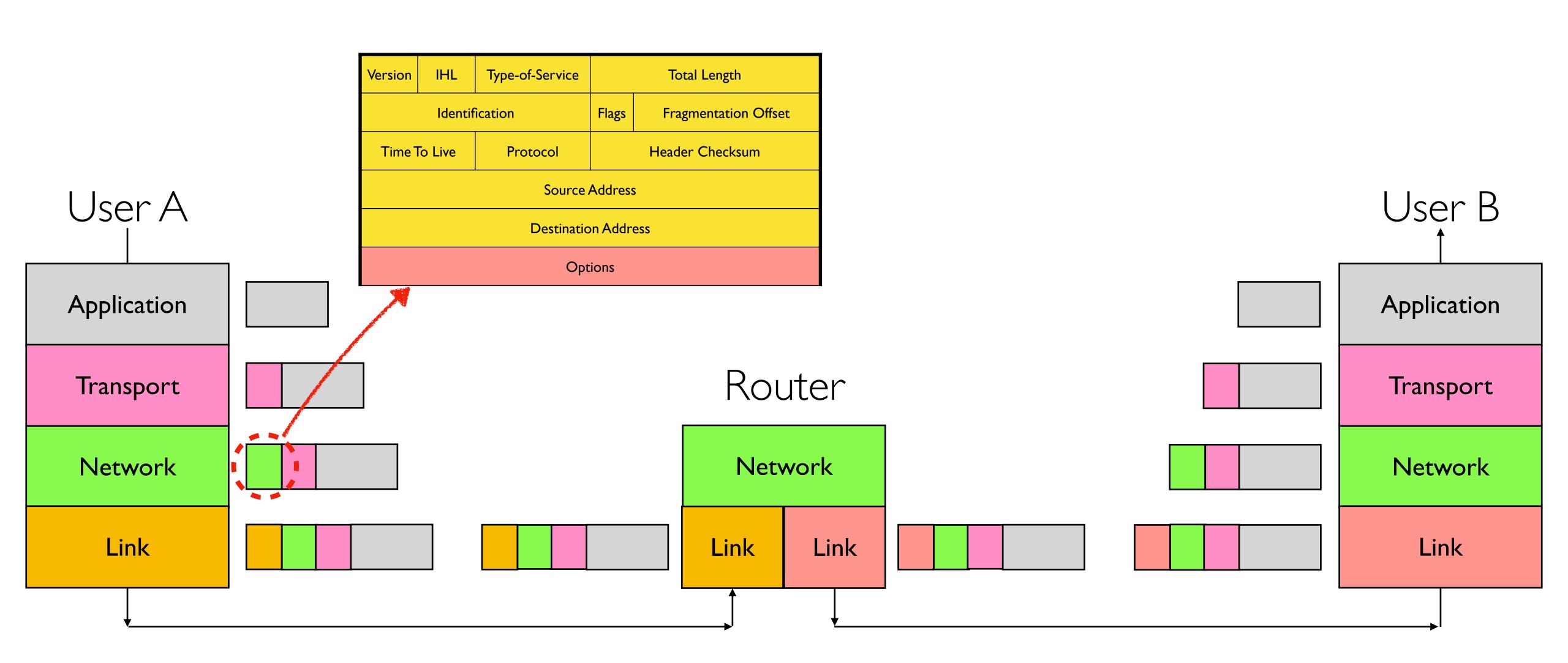


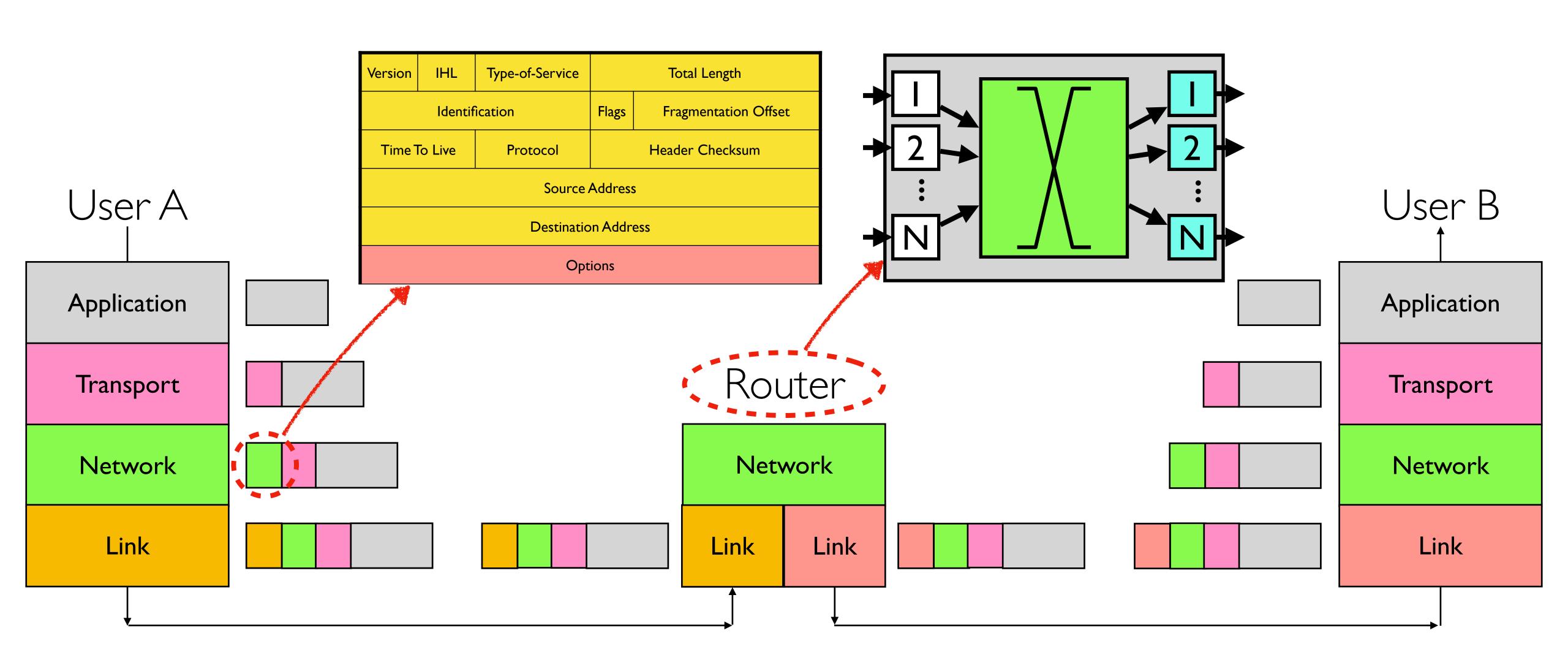




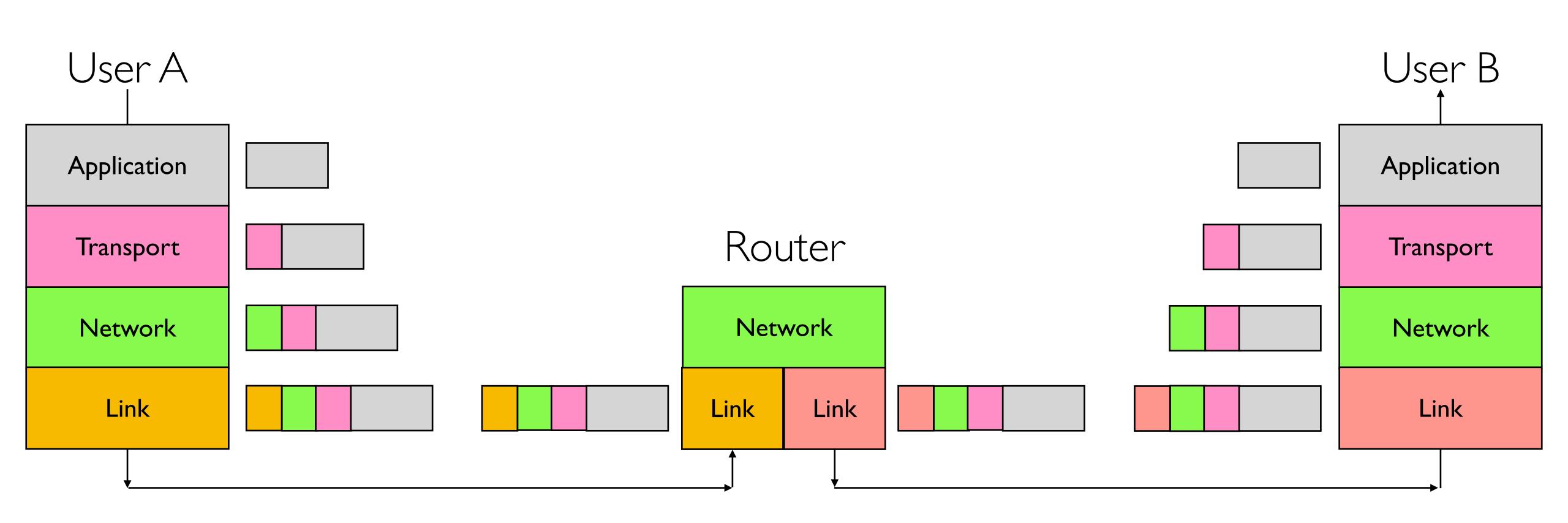






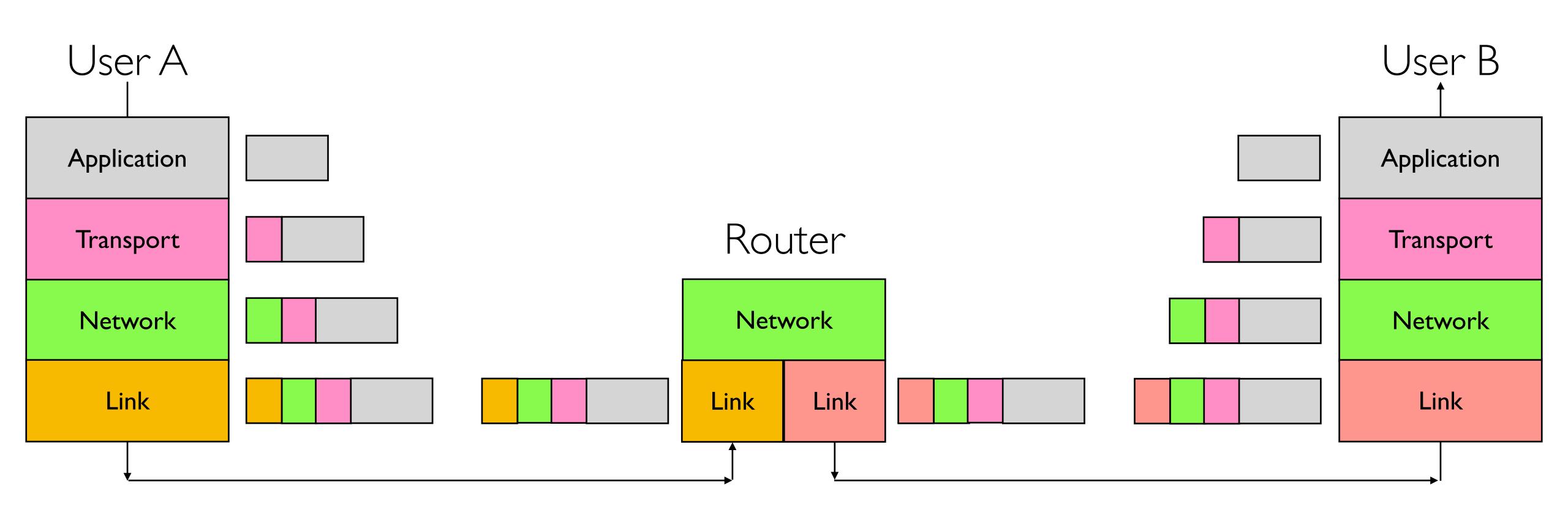


Up Next...



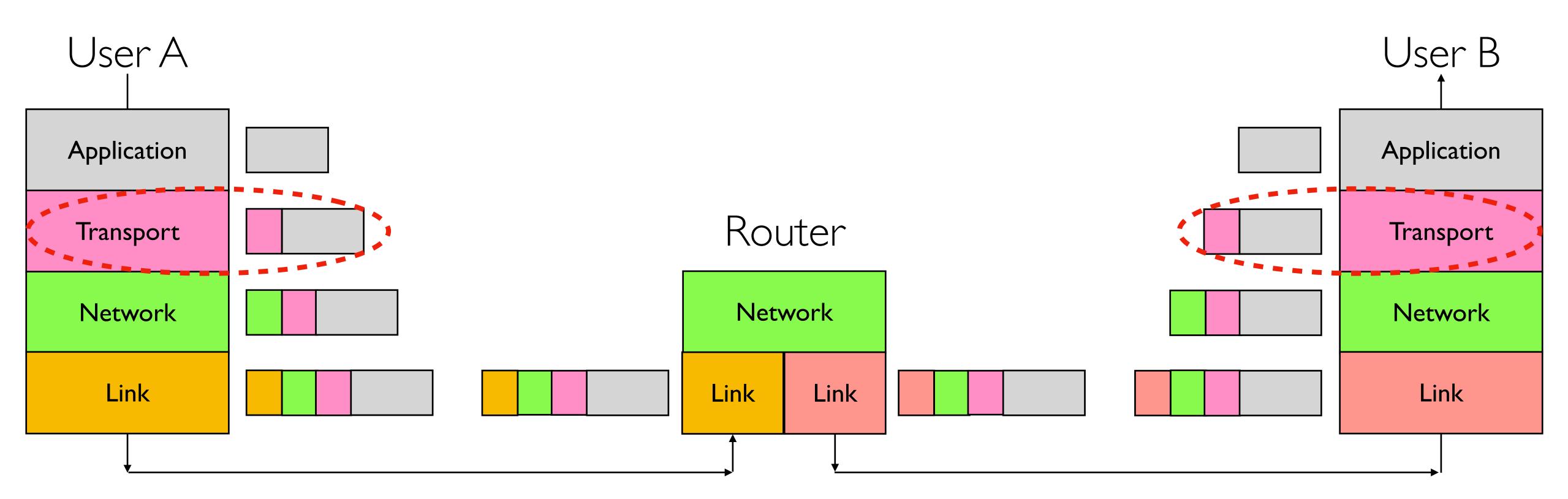
Up Next...

• So far: Network Layer, Best-effort global delivery of packets



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- Next: Transport Layer, Reliable (or unreliable) delivery of data
 - Layer at end-hosts, between the application and network layers



Transport layer and application both on host

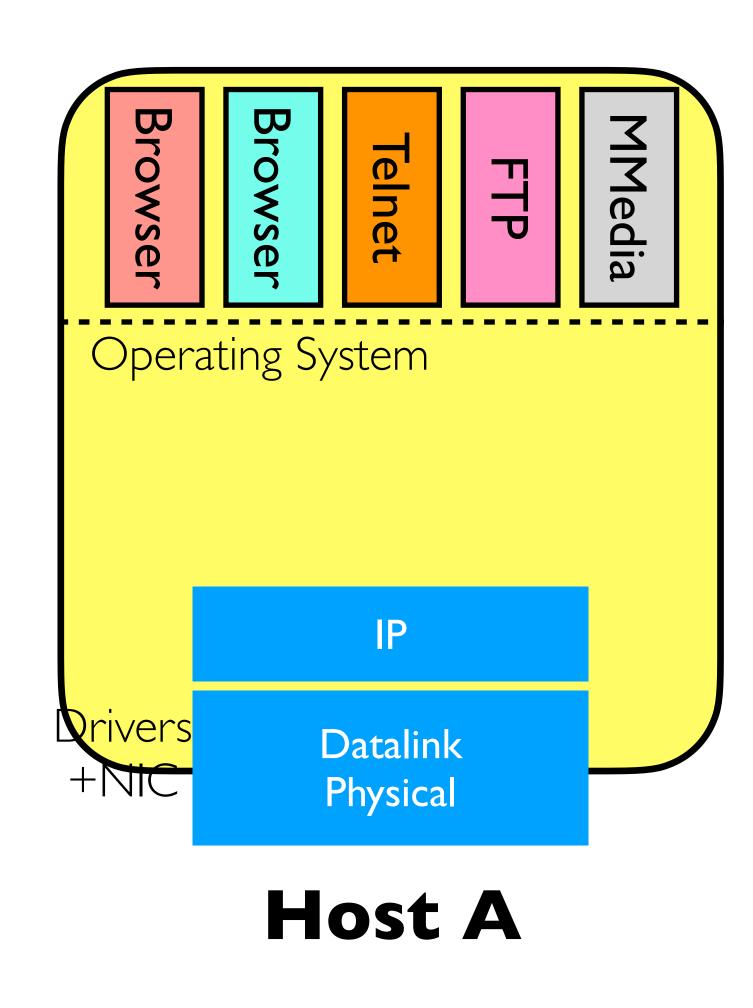
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- Why not combine the two?

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- Why not combine the two?
- And what should that code do anyway?

1. De-multiplex packets between many applications

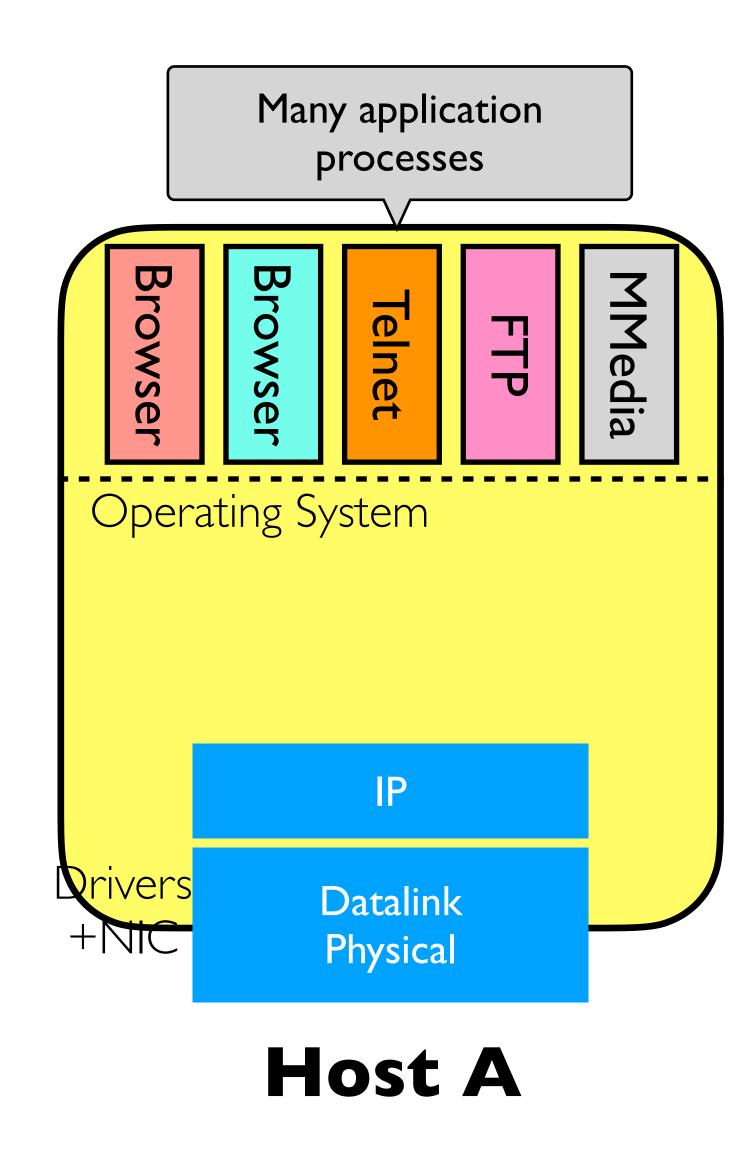
- 1. De-multiplex packets between many applications
- 2. Additional (optional) services on top of IP

- 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)



Application Transport Network Data Link **Physical**

Host B



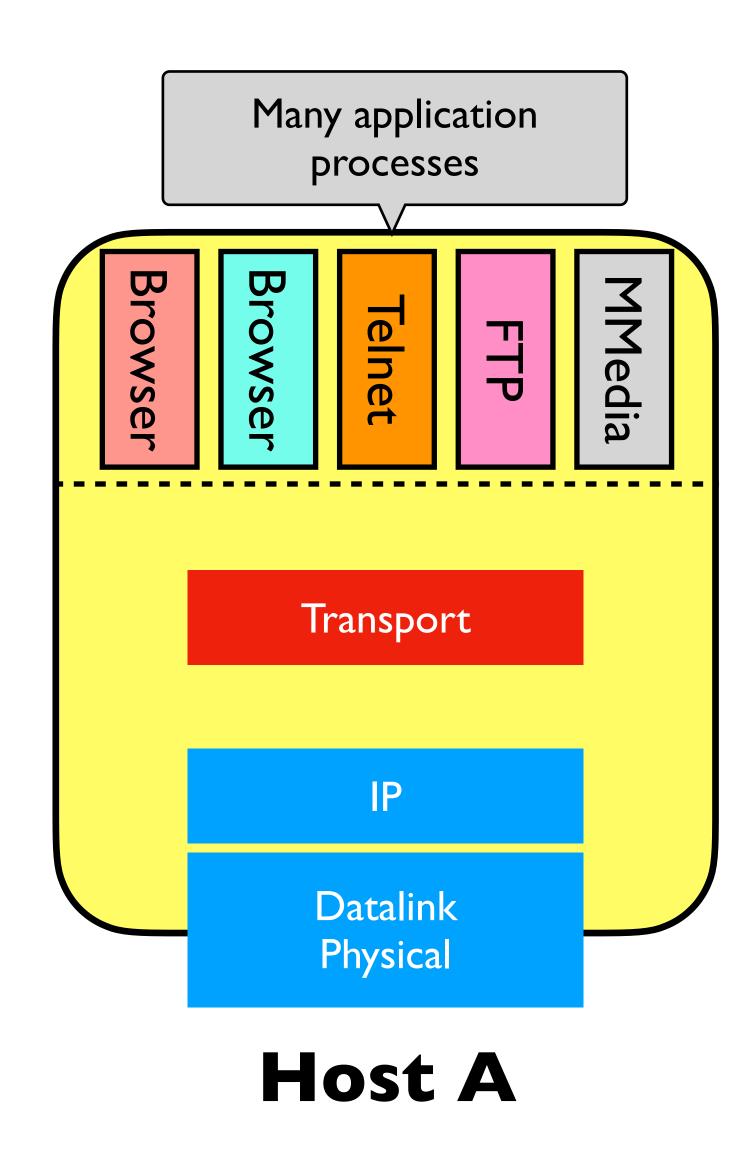
Application

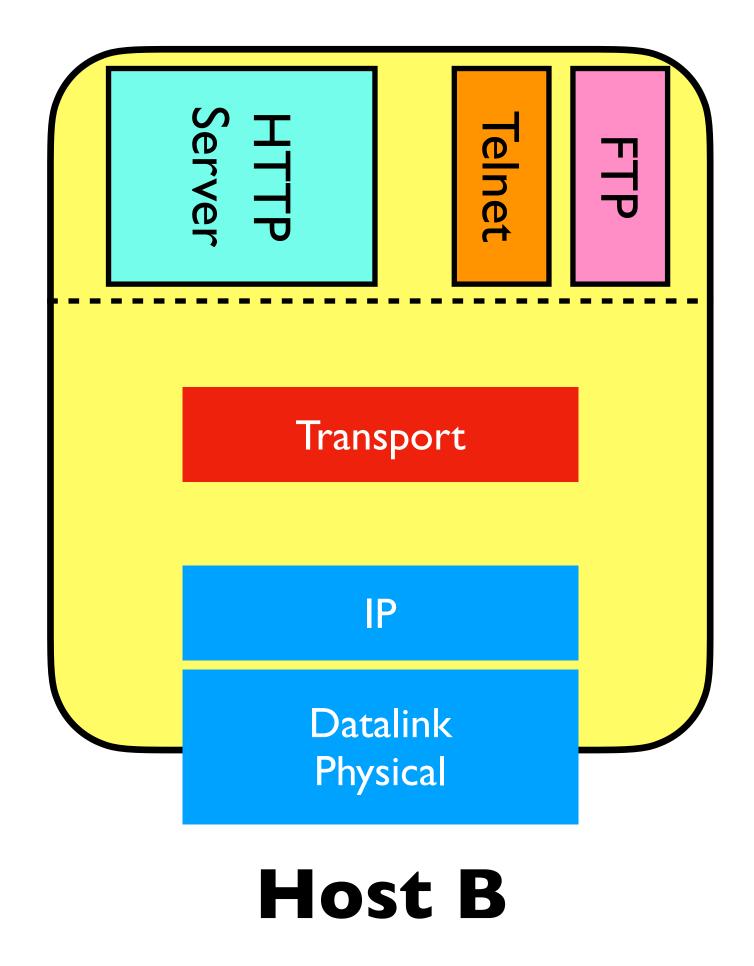
Transport

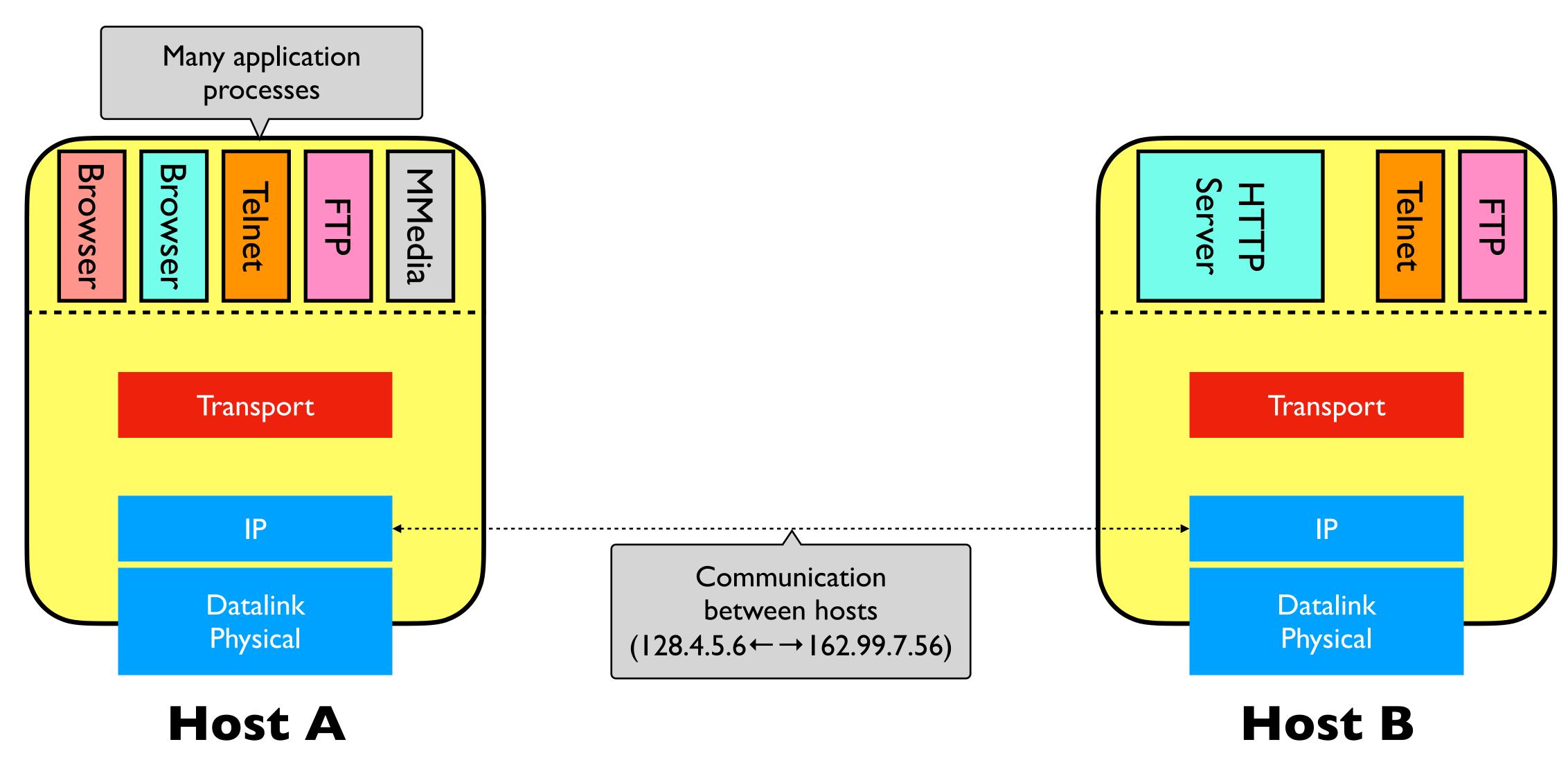
Network

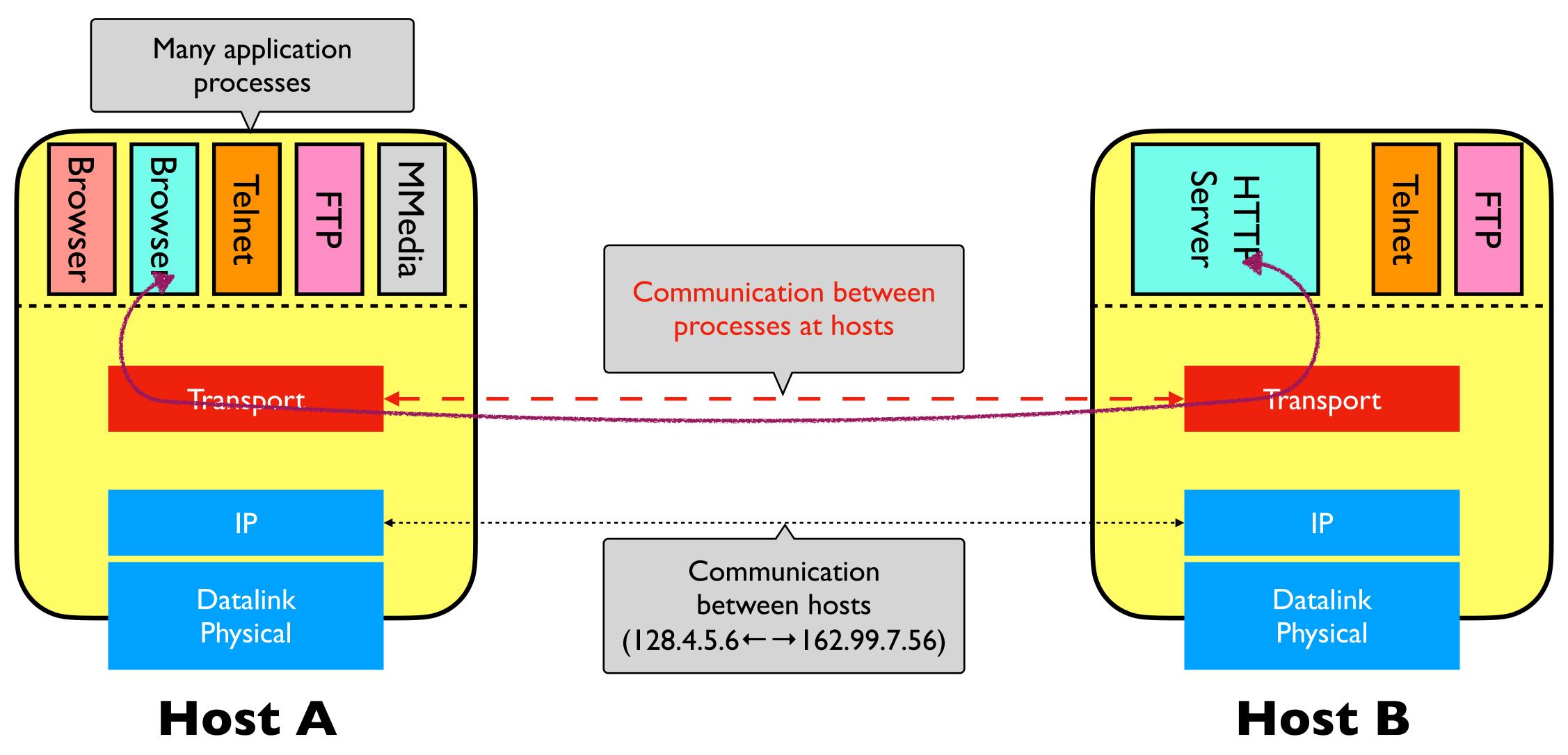
Data Link

Physical









Why a Transport Layer: Improved Service Model

Why a Transport Layer: Improved Service Model

• 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

Communication between application processes

- Multiplex and demultiplex from/to application processes
- Implemented using **ports** (not the same as router ports!)

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

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- TCP and UDP are the common transport protocols
 - Also SCTP, MTCP, SST, RDP, DCCP, ...

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- TCP and UDP are the common transport protocols
- UDP is a minimalist, no-frills transport protocol
 - Only provides multiplex/demultiplex capabilities

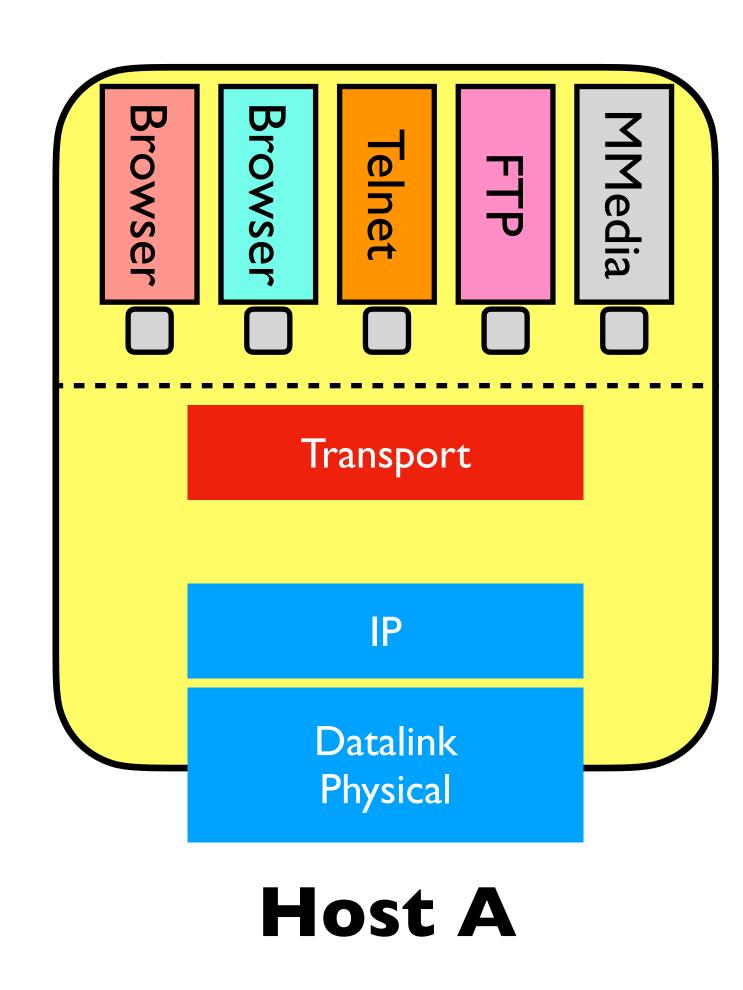
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- Provide common end-to-end services for application layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist, no-frills transport protocol
- TCP is the whole-hog protocol
 - Offers applications a reliable, in-order, byte stream abstraction
 - With congestion control
 - But no performance guarantees (delay, bw, etc.)

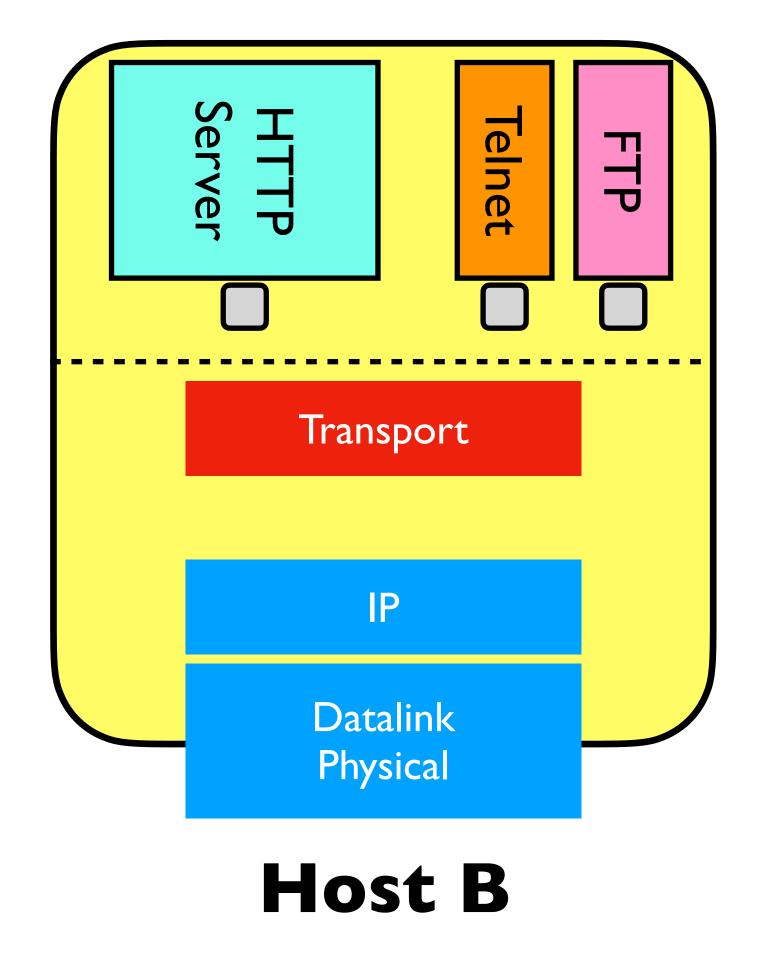
Why a transport layer?

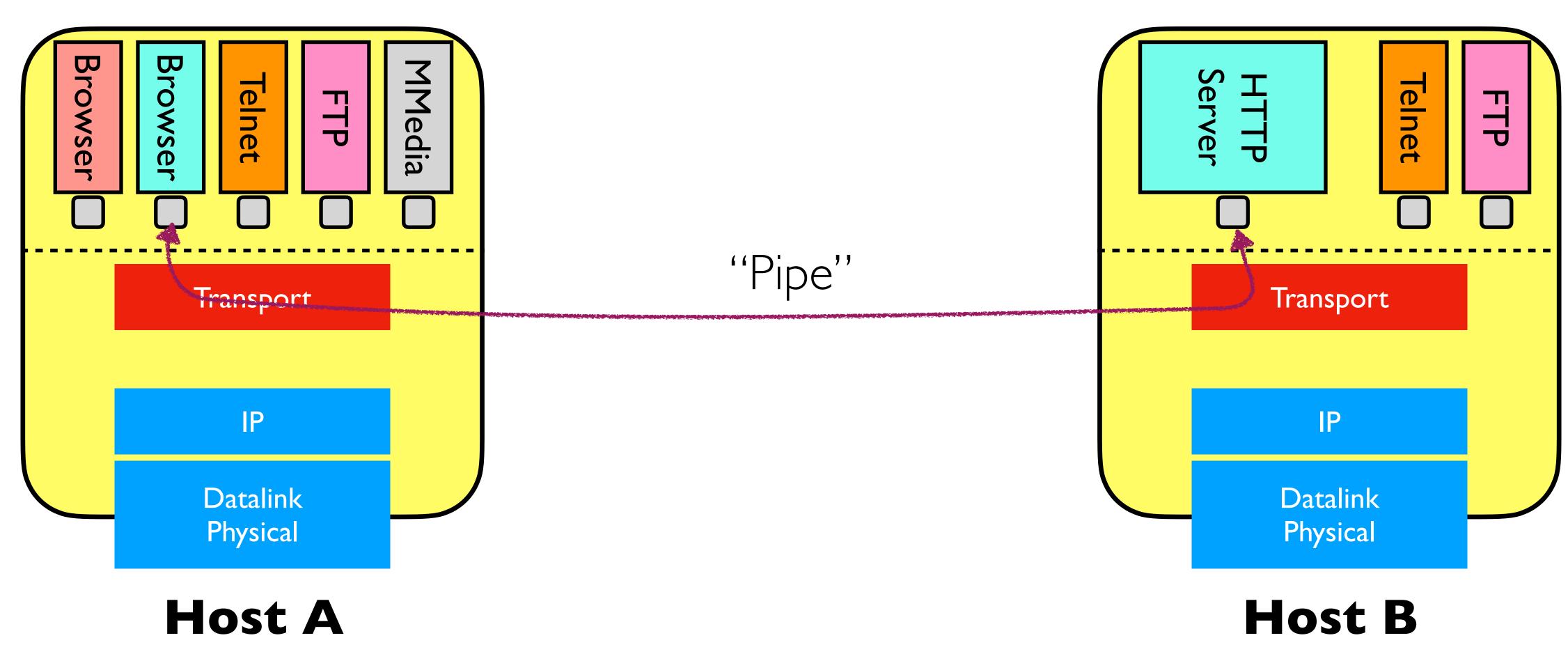
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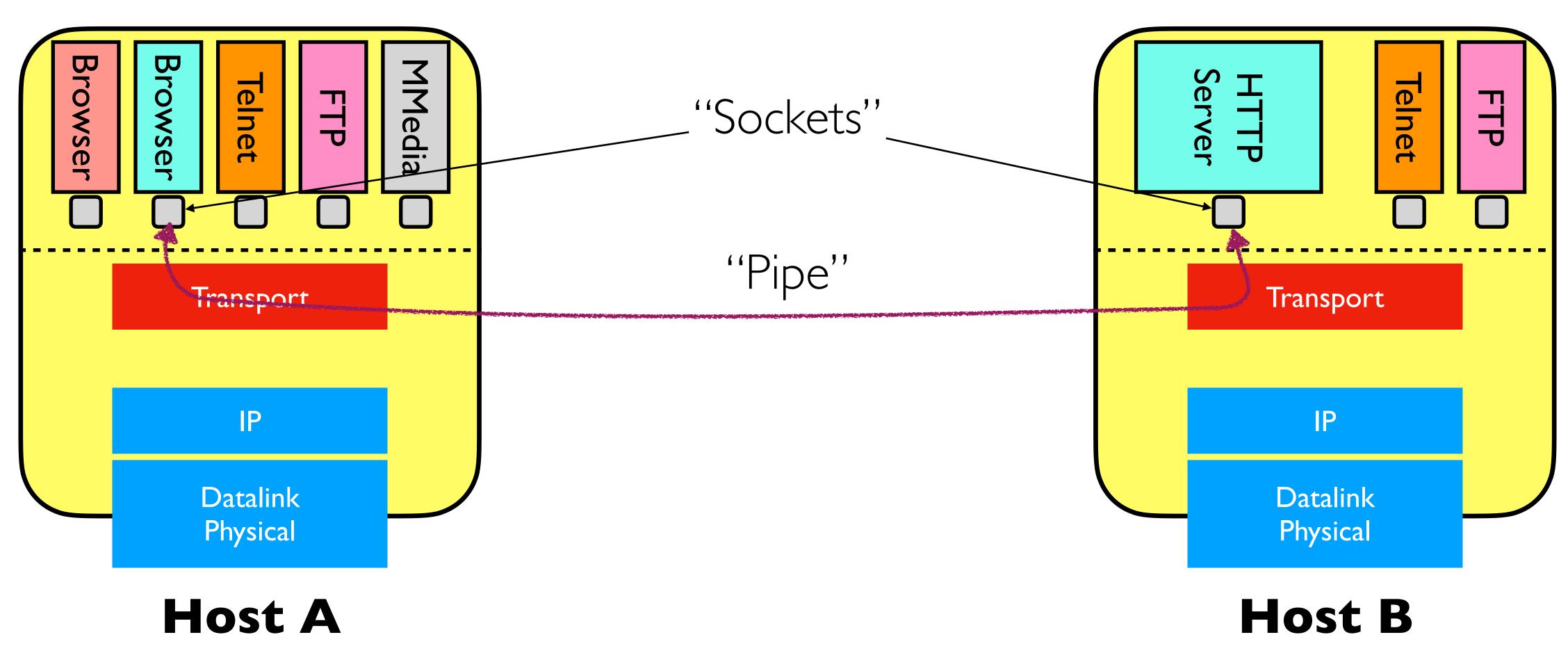
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 - Messages are limited to a single packet ("datagrams")
 - Pipe is "leaky": applications need to deal with leaks (lost/corrupted datagrams)
- Reliable byte stream: Transmission Control Protocol (TCP)
 - Bytes inserted into pipe by sender
 - They emerge, in order, at the receiver (to the application)

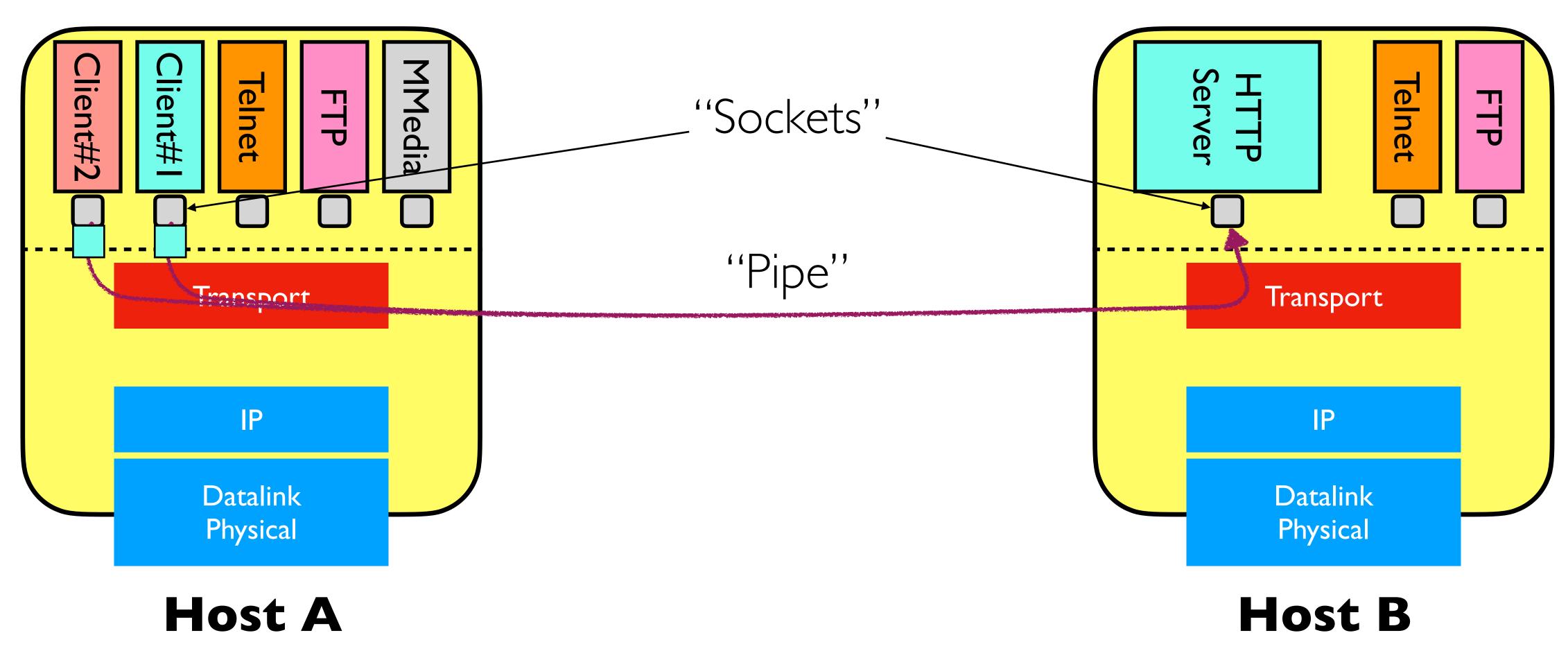
• Sources send packets, destinations receive packets

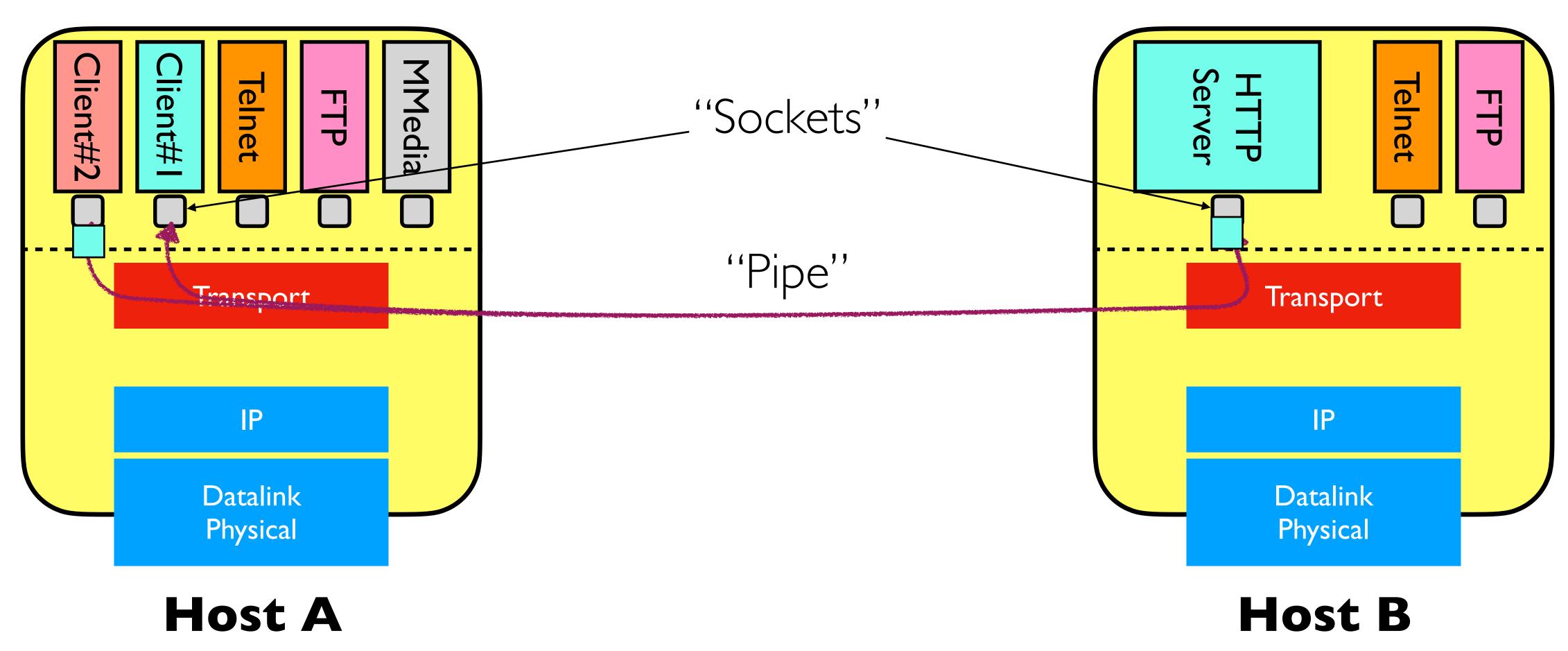
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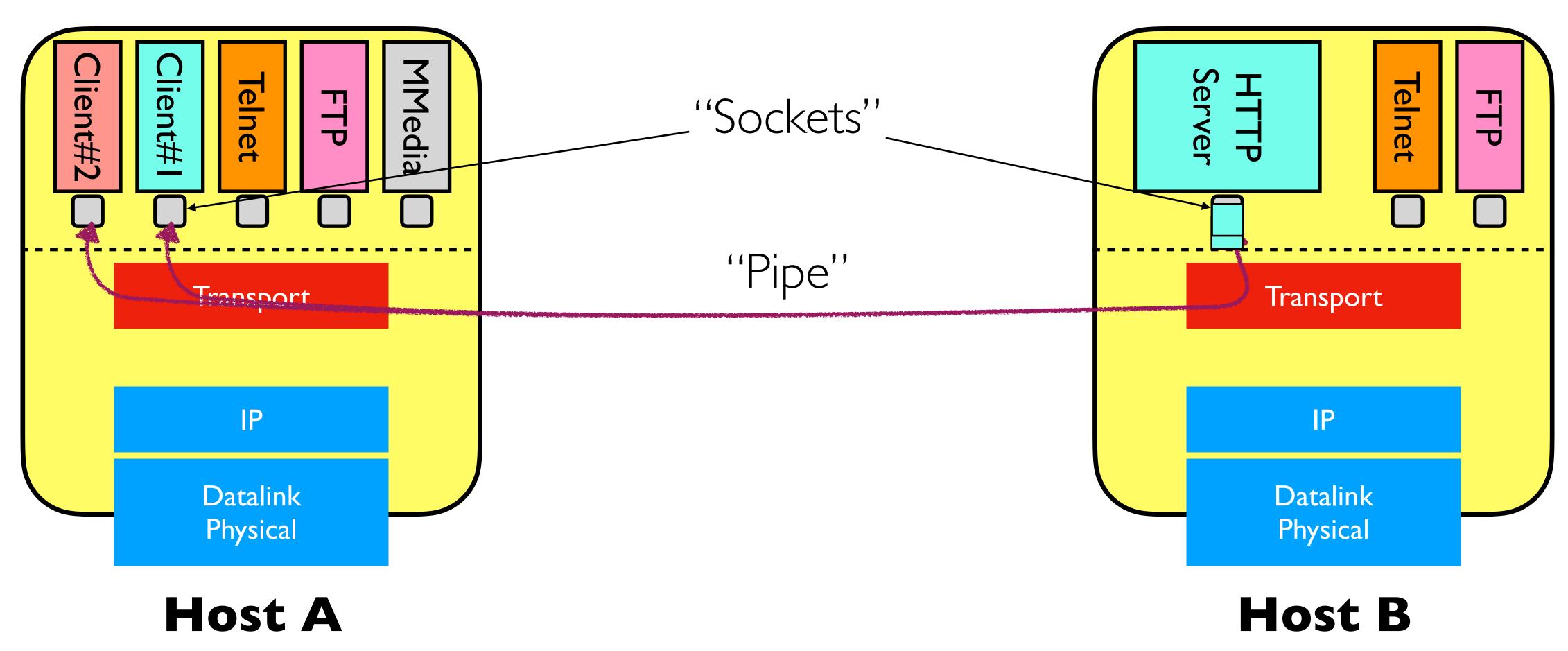
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- Nothing else!
- Minimal extension to IP





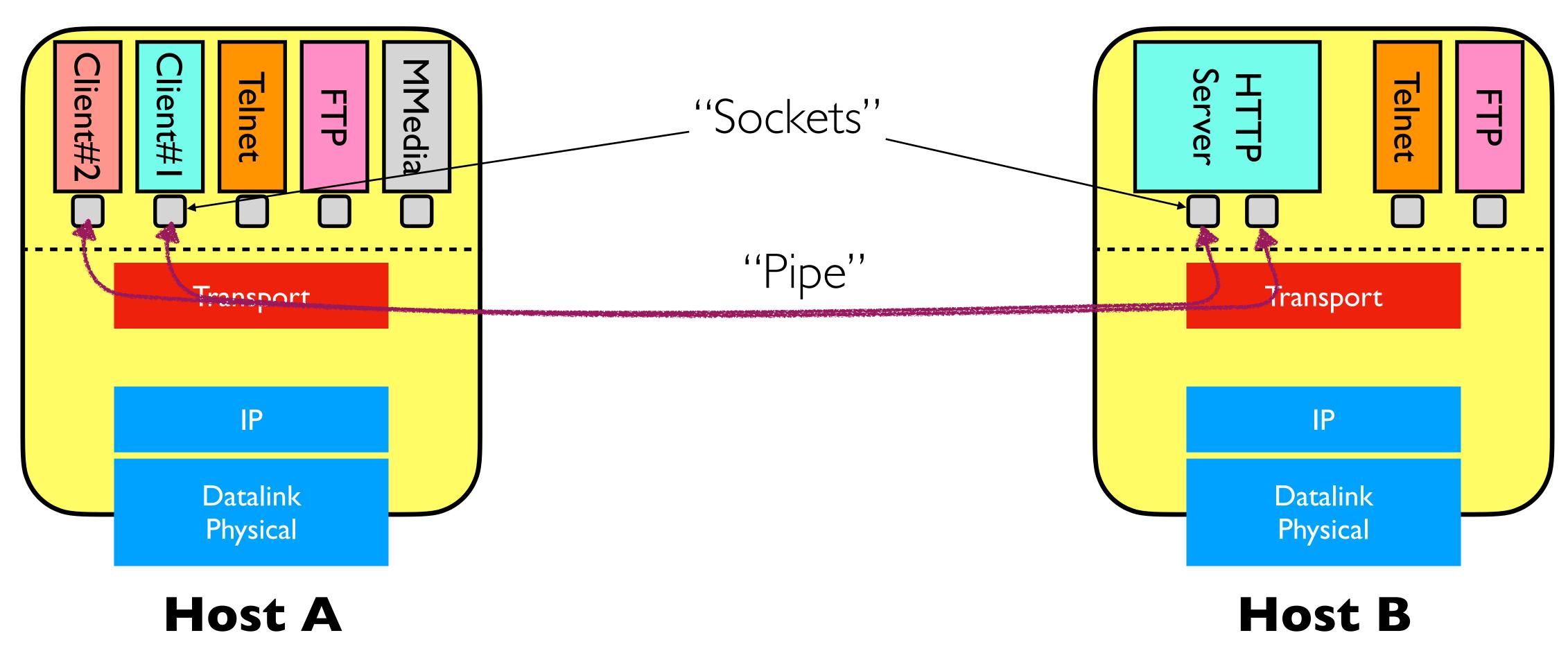


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- Every source-destination pair has a dedicated stream oriented "connection" (or "session")

Abstraction: Pipes & Sockets



Sockets

Sockets

- **Socket:** software abstraction by which an application process exchanges network messages with the (transport layer of the) operating system
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 - socketID.sendto(message, ...)
 - socketID.recvfrom(...)
 - Will cover in detail later in the course

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

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- Q: Why the difference?

4-bit Version Length	8-bit Type-of-Service	I 6-bit Total Length (Bytes)		
I6-bit Identification			I3-bit Fragmentation 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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32-bit Source IP Address					
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I6-bit Source Port			I6-bit Destination Port		
More transport header fields					
Payload					

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 - e.g., ssh:22, http:80
 - Helps client know server's port
 - Services can listen on well-known ports
- Ephemeral ports (most of 1024-65535): given to clients

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Host receives IP packets

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Host receives IP packets

- Each IP header has source and destination IP address
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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

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Rest of Lecture

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- Reliable Transport
- Next Lecture: Details of TCP

In a perfect world, reliable transport is easy

@Sender Send packets

ReceiverWait for packets

- In a perfect world, reliable transport is easy
- All the bad things "best-effort" can permit:
 - 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?)

Mechanisms for coping with bad things:

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 - Forward error correction: a way to mask errors without retransmission

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

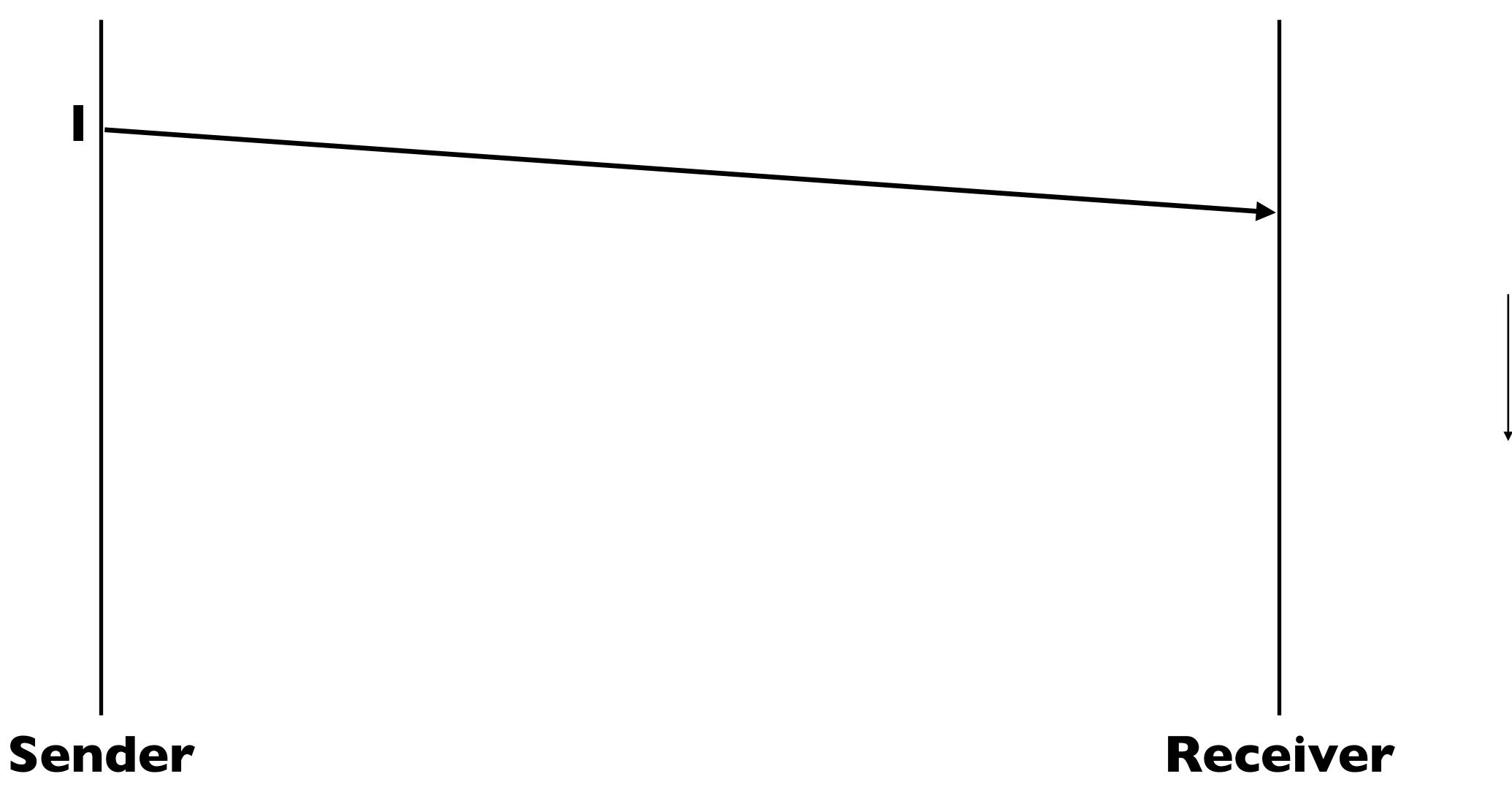
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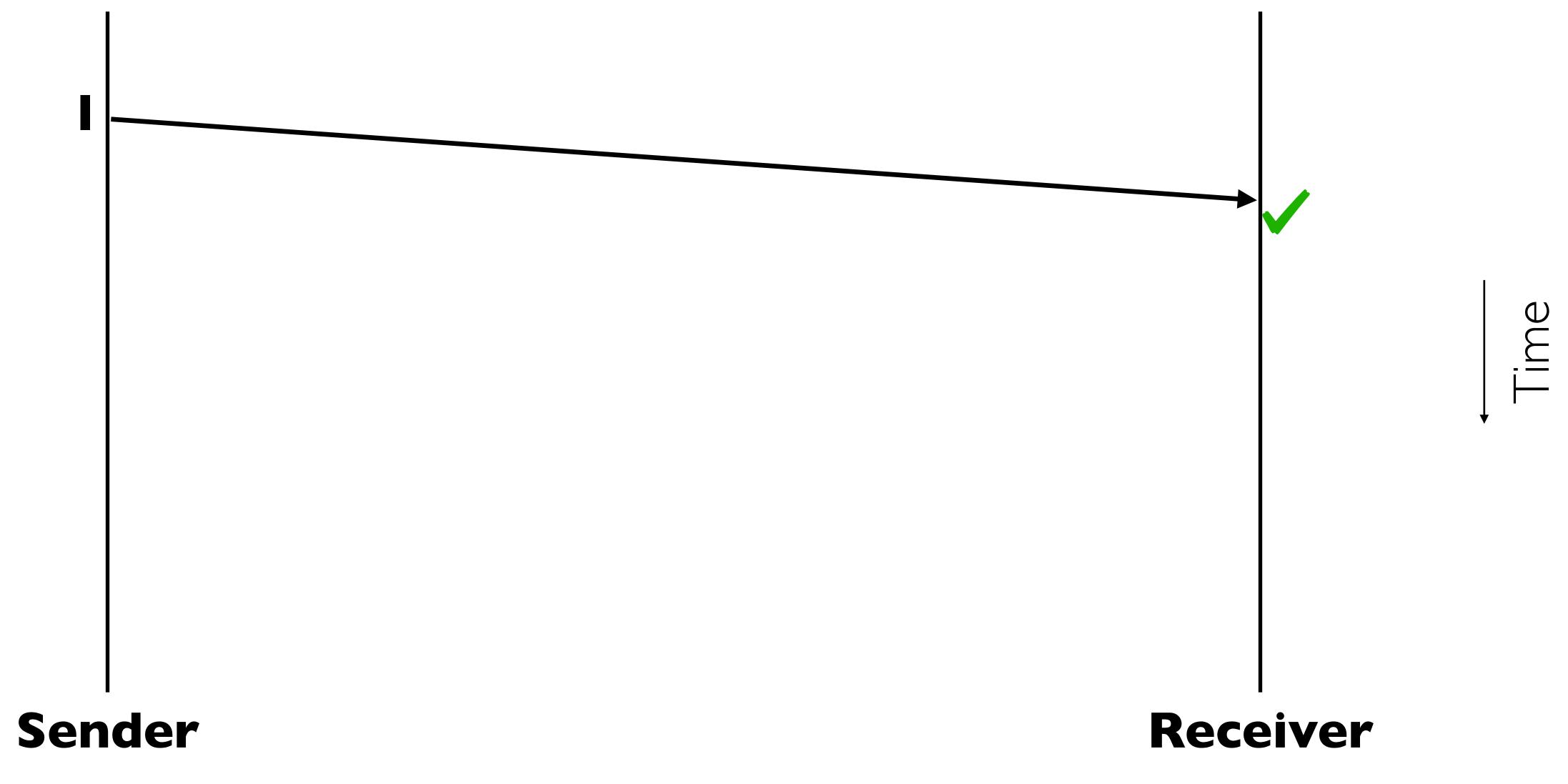
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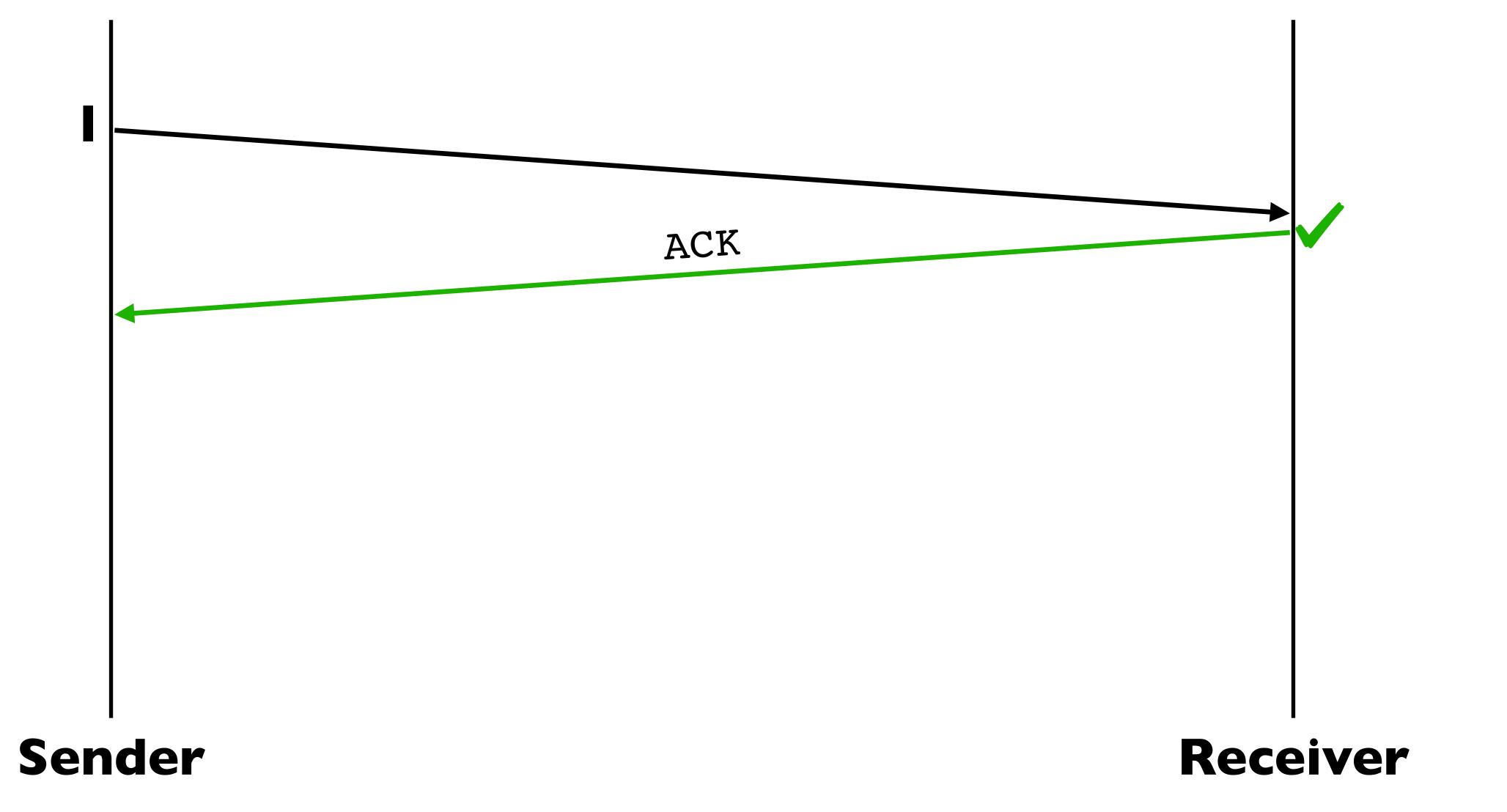
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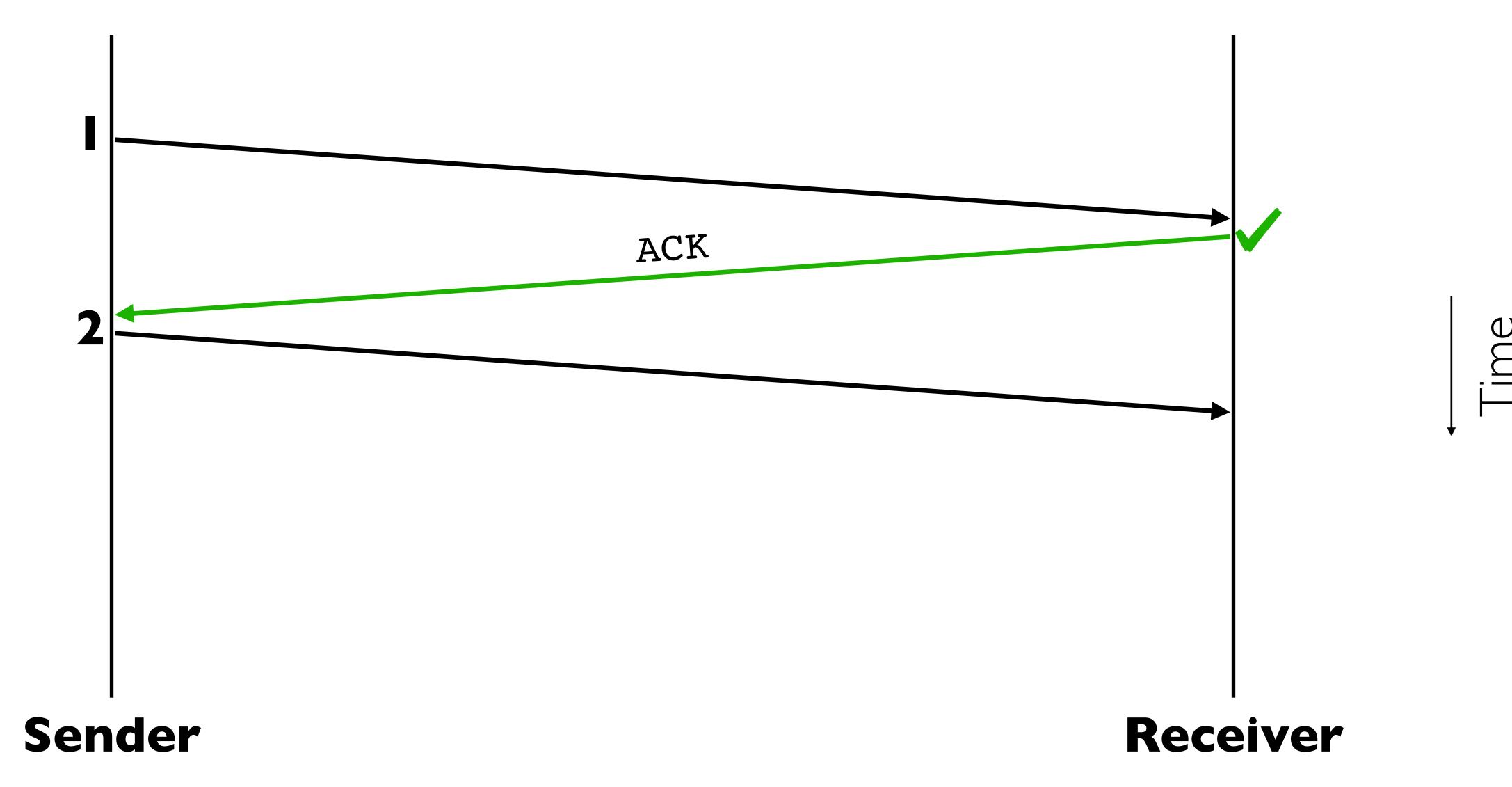
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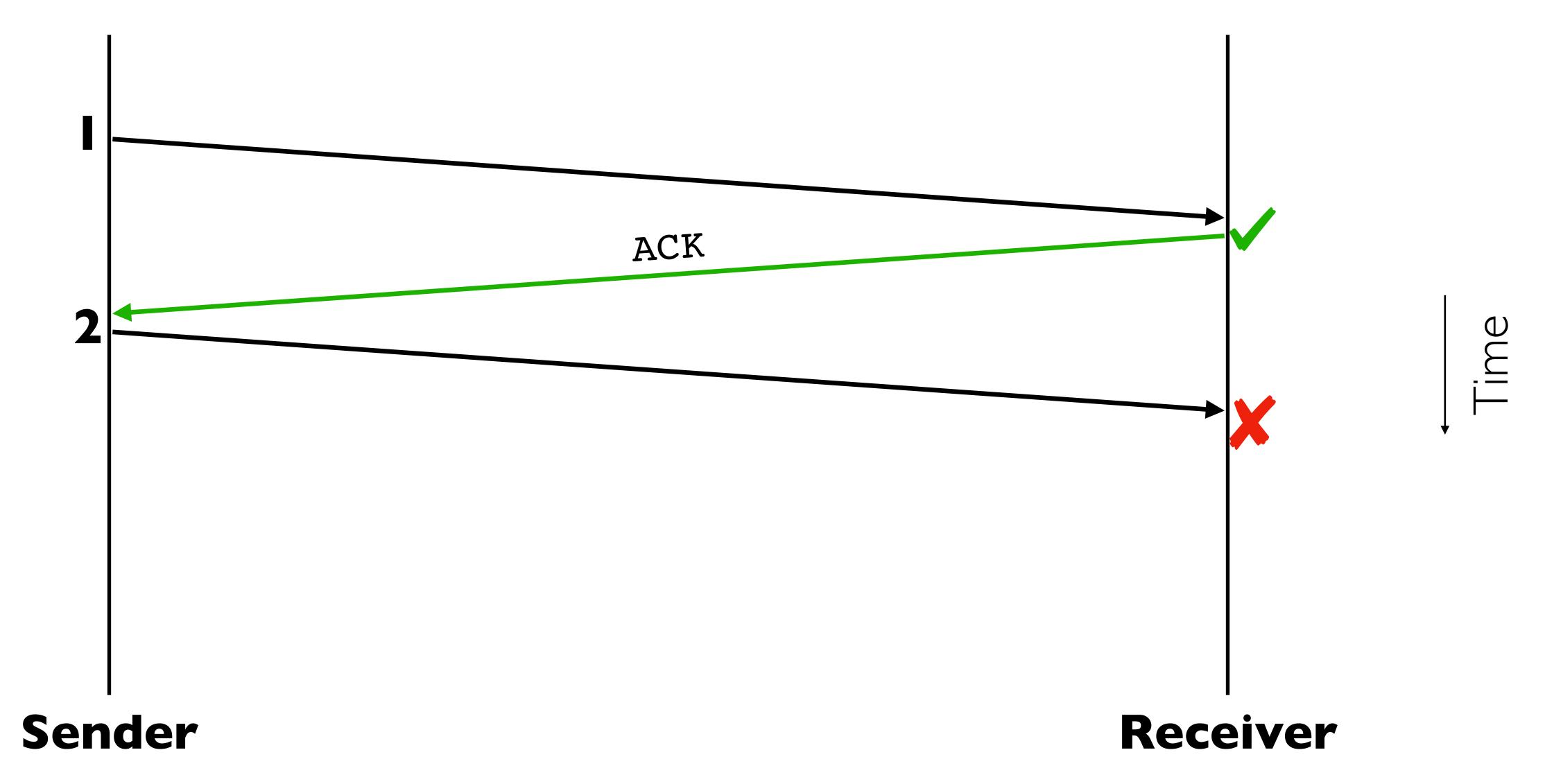


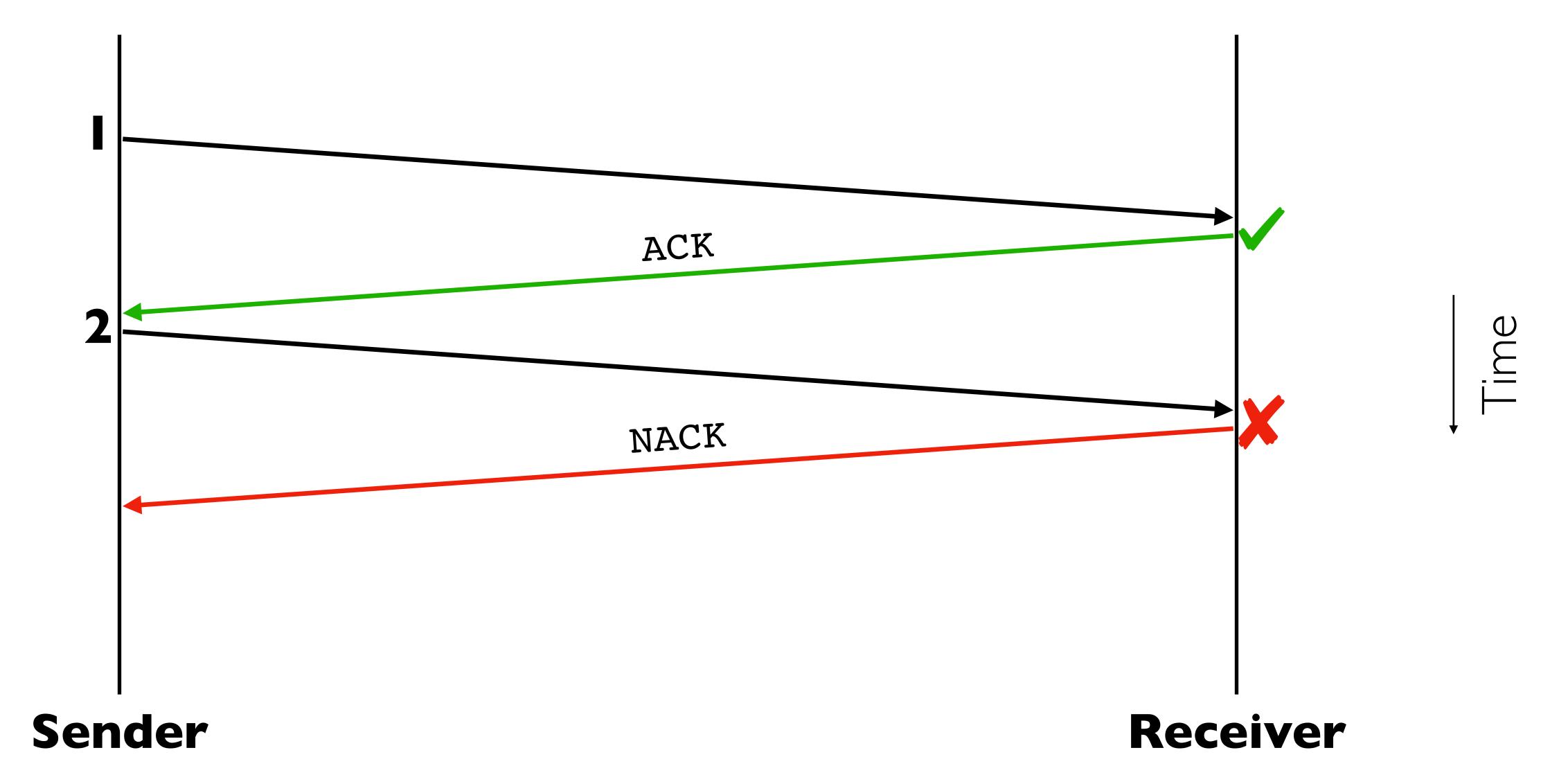


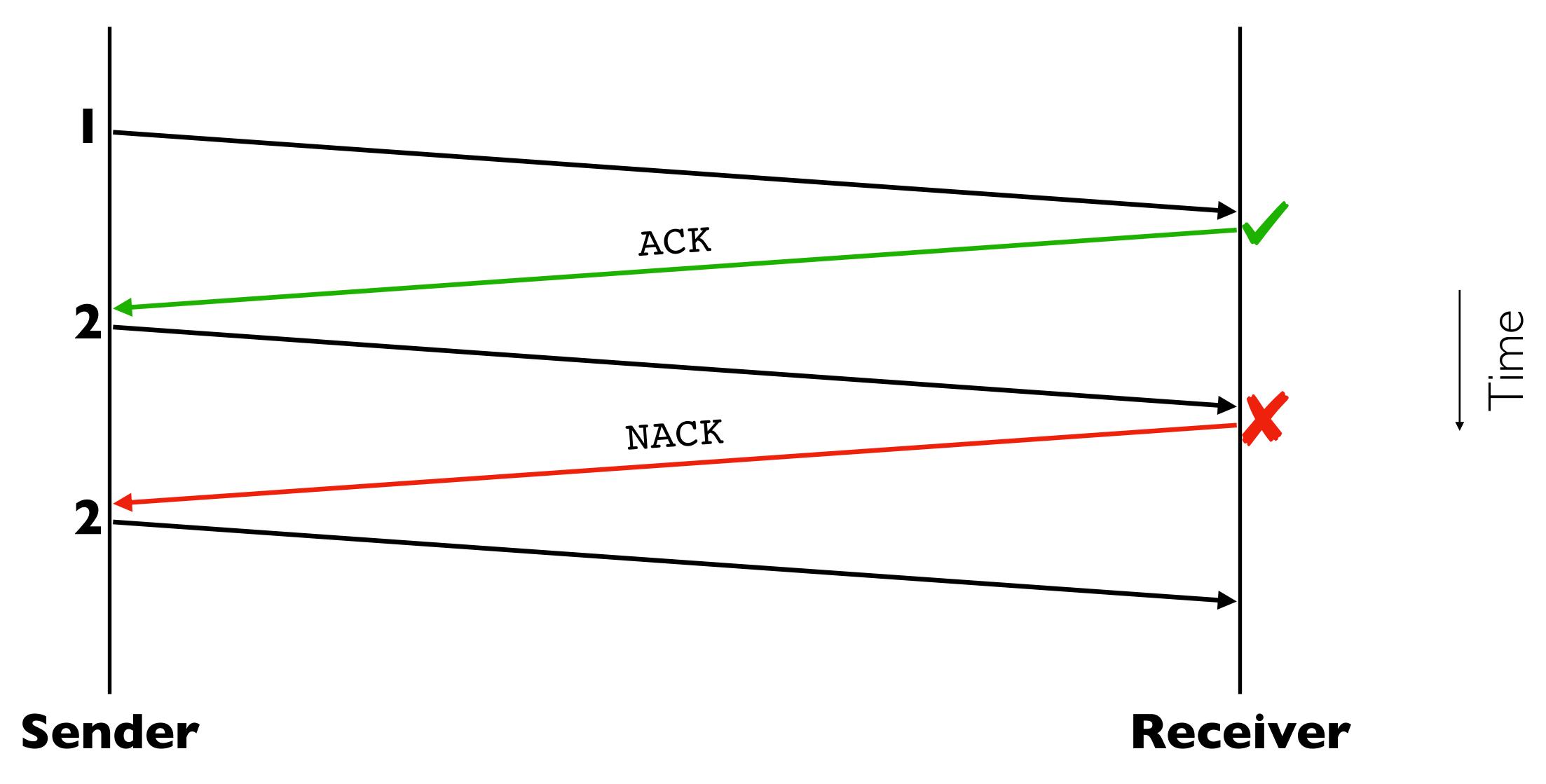


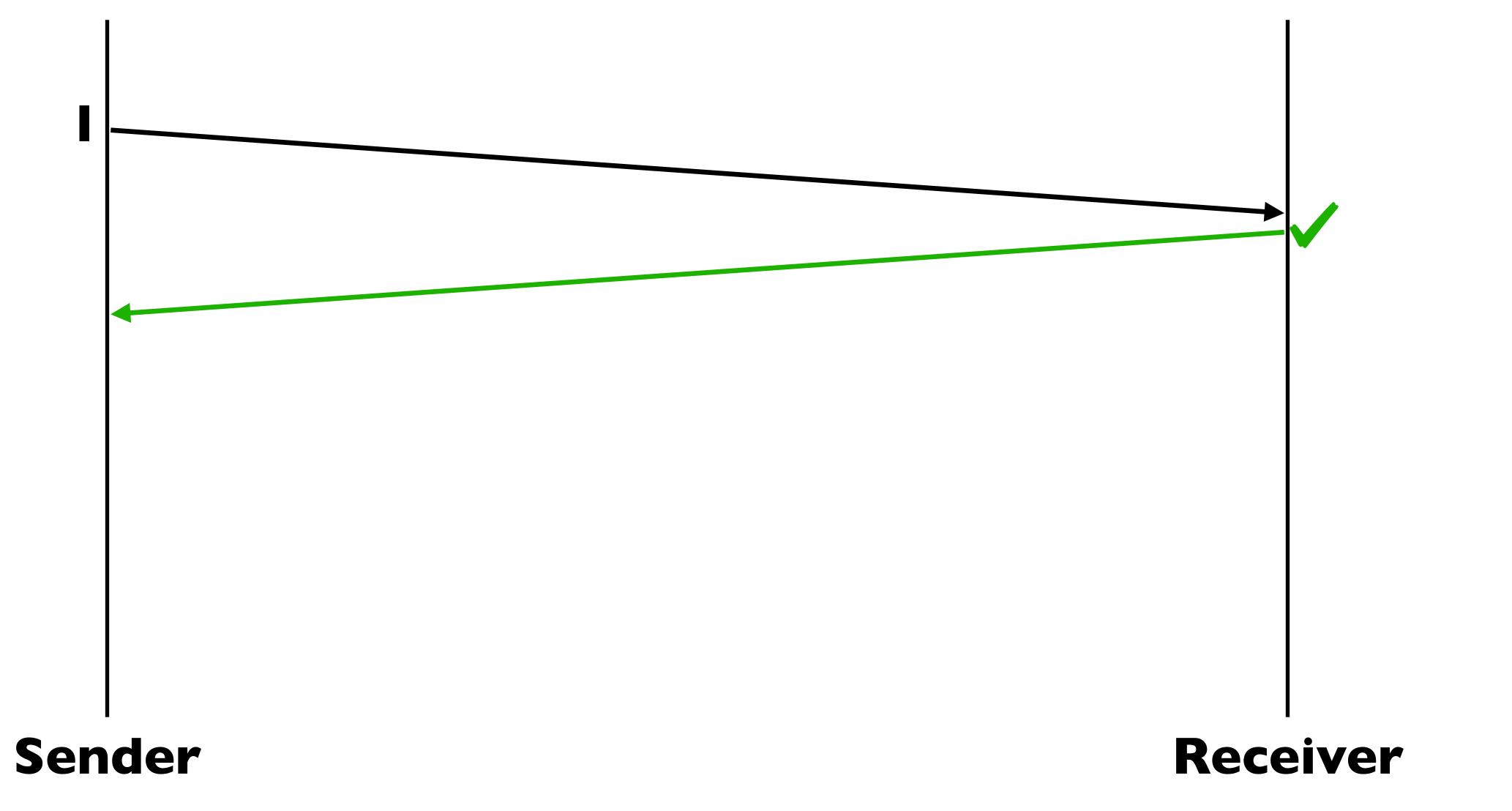


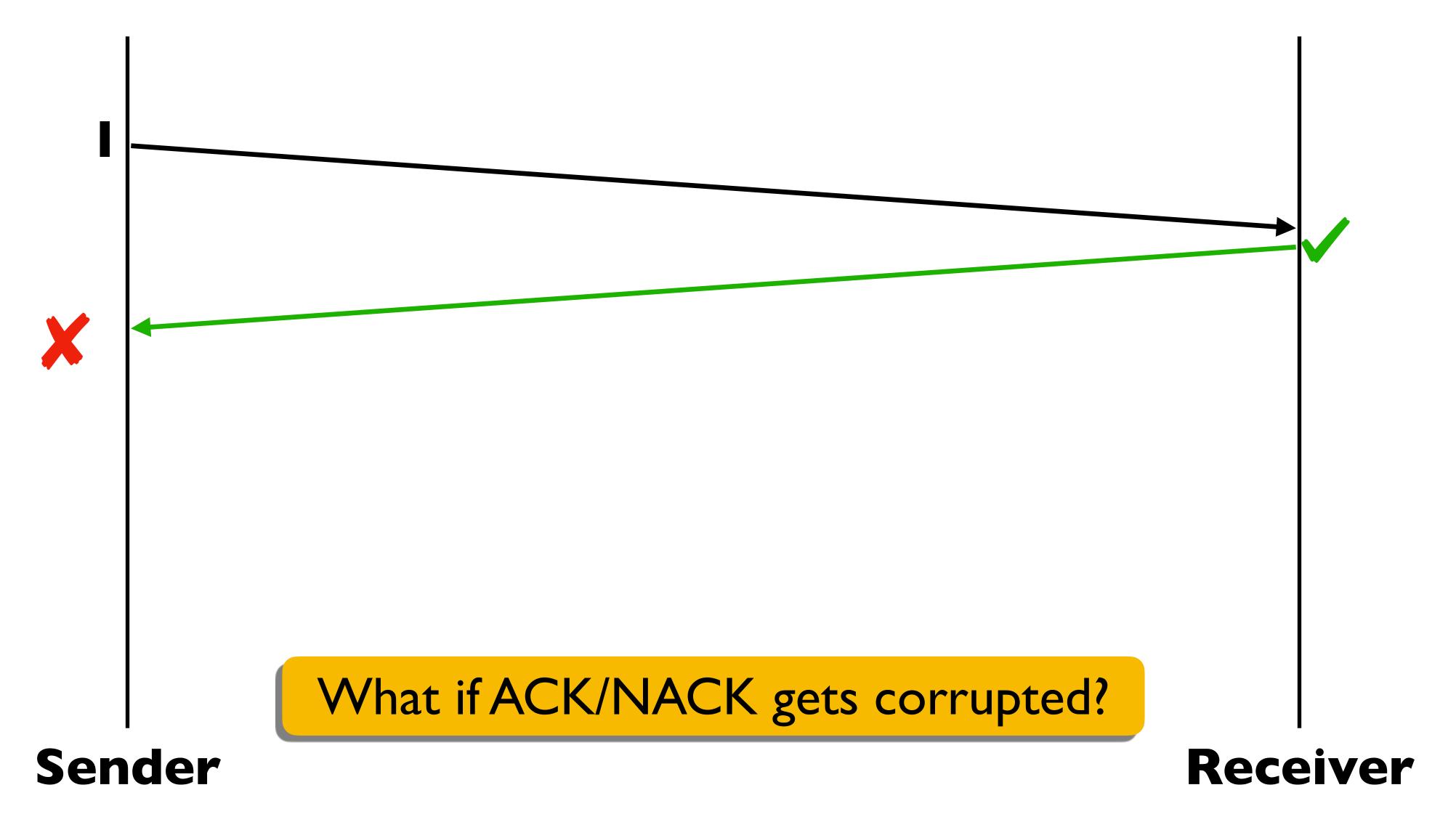


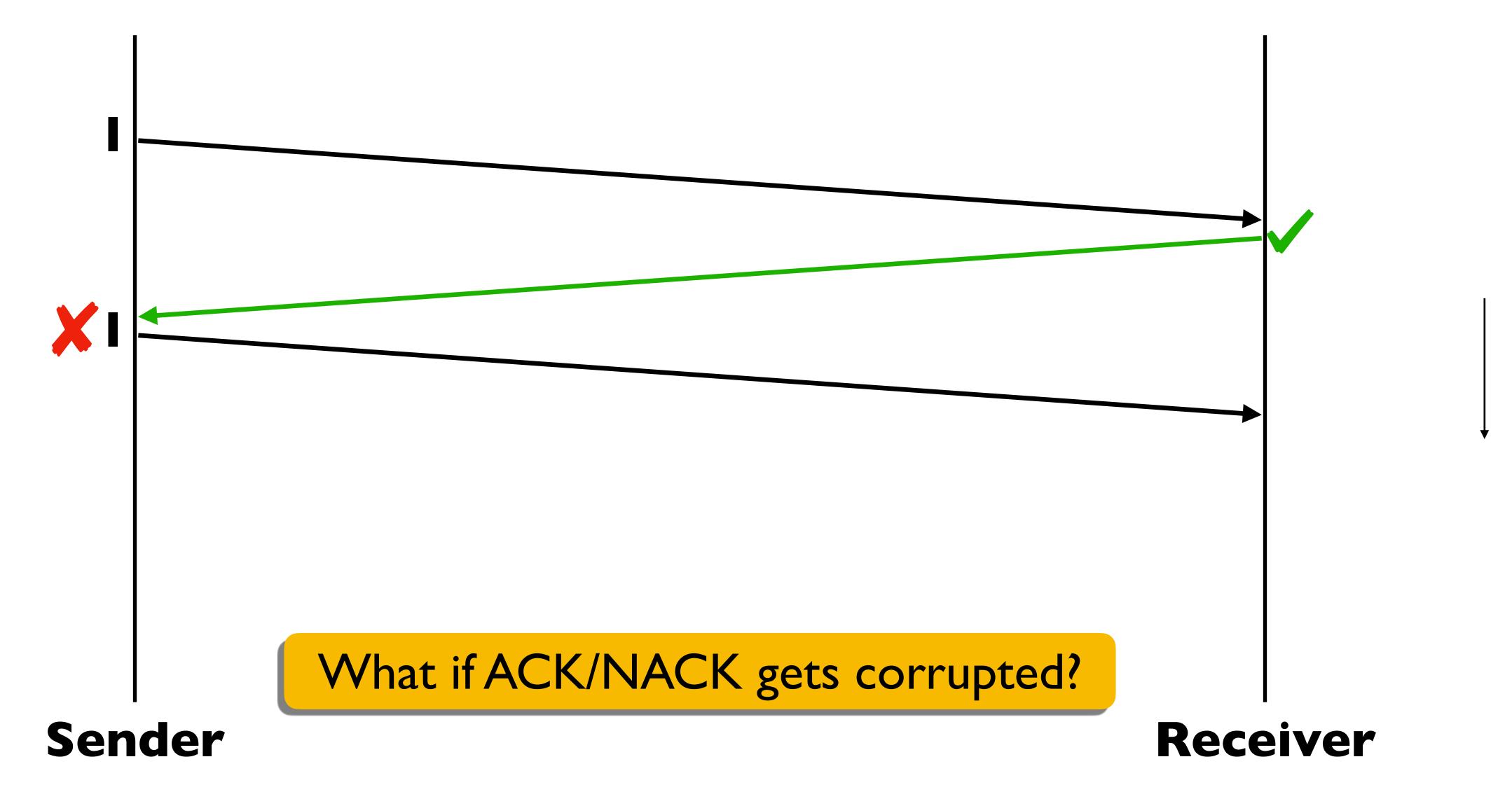


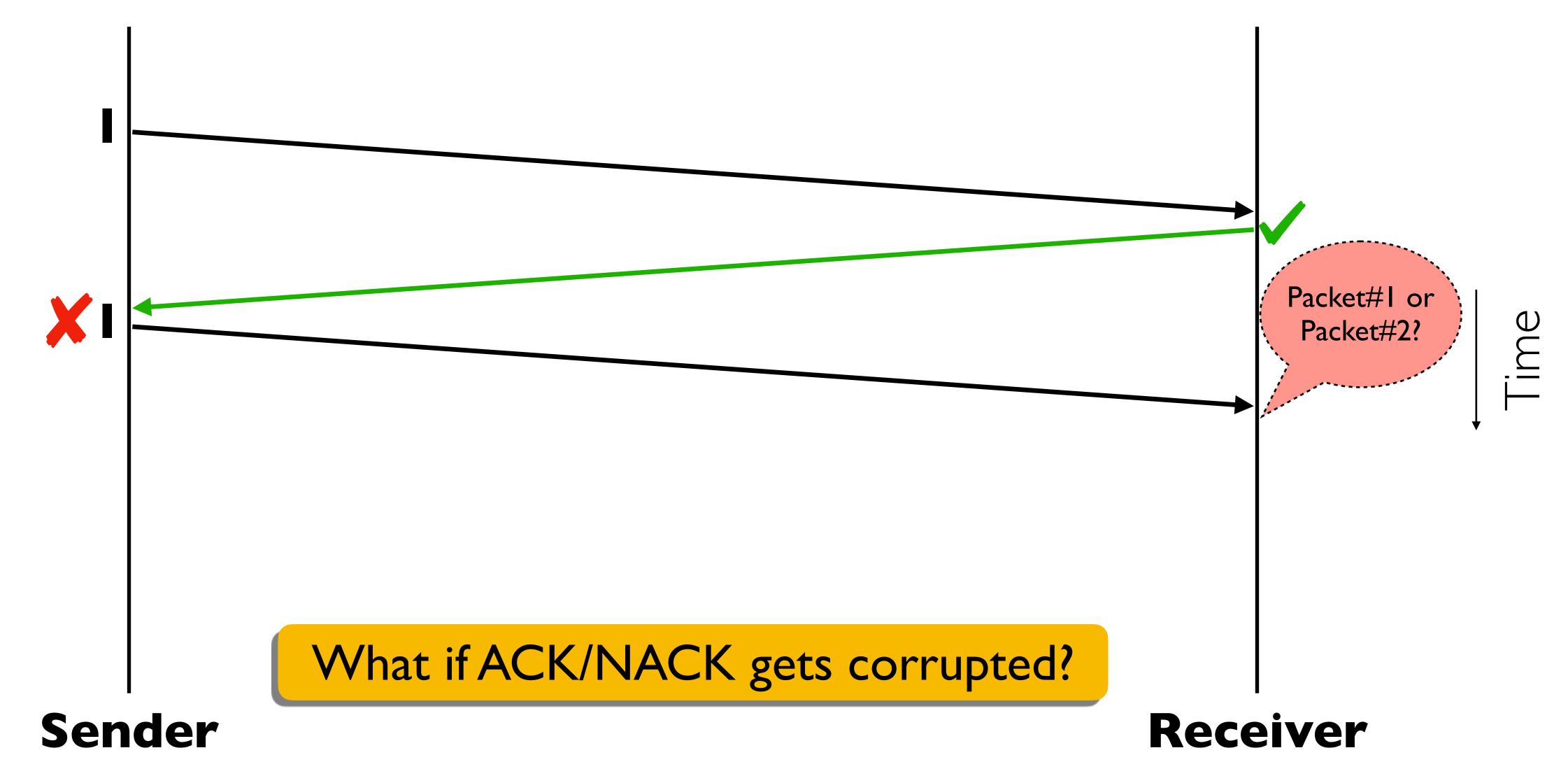


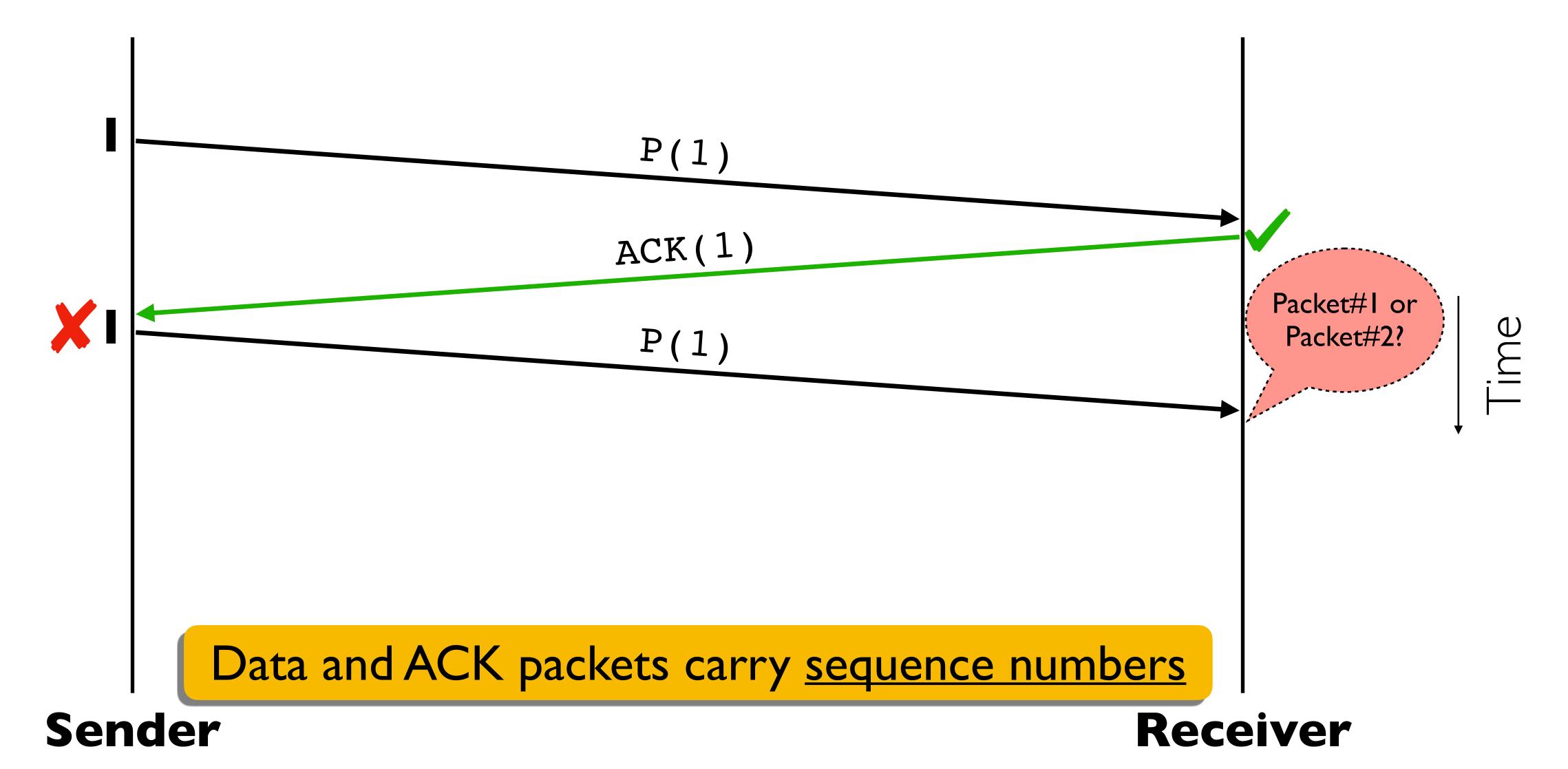


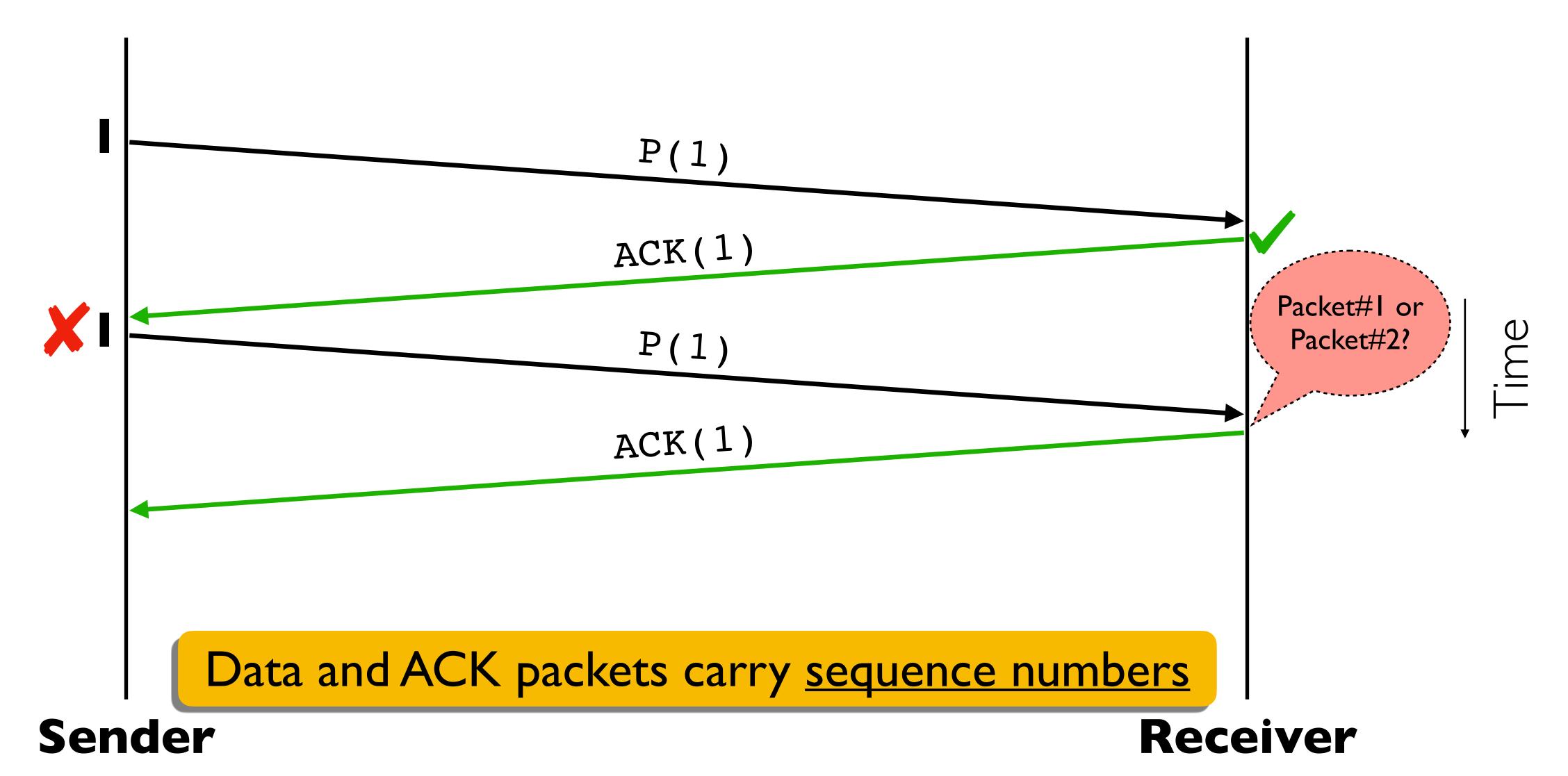


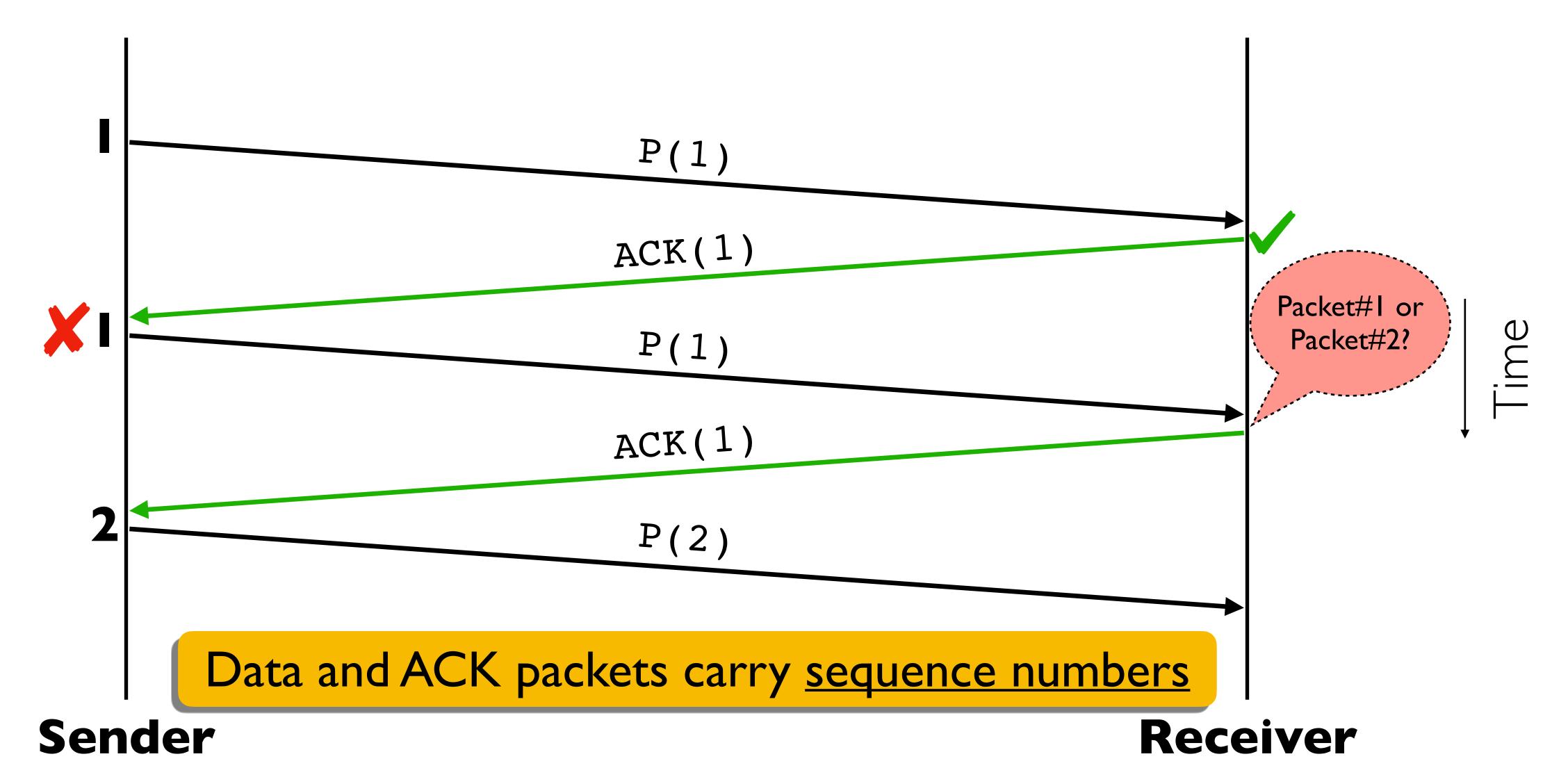


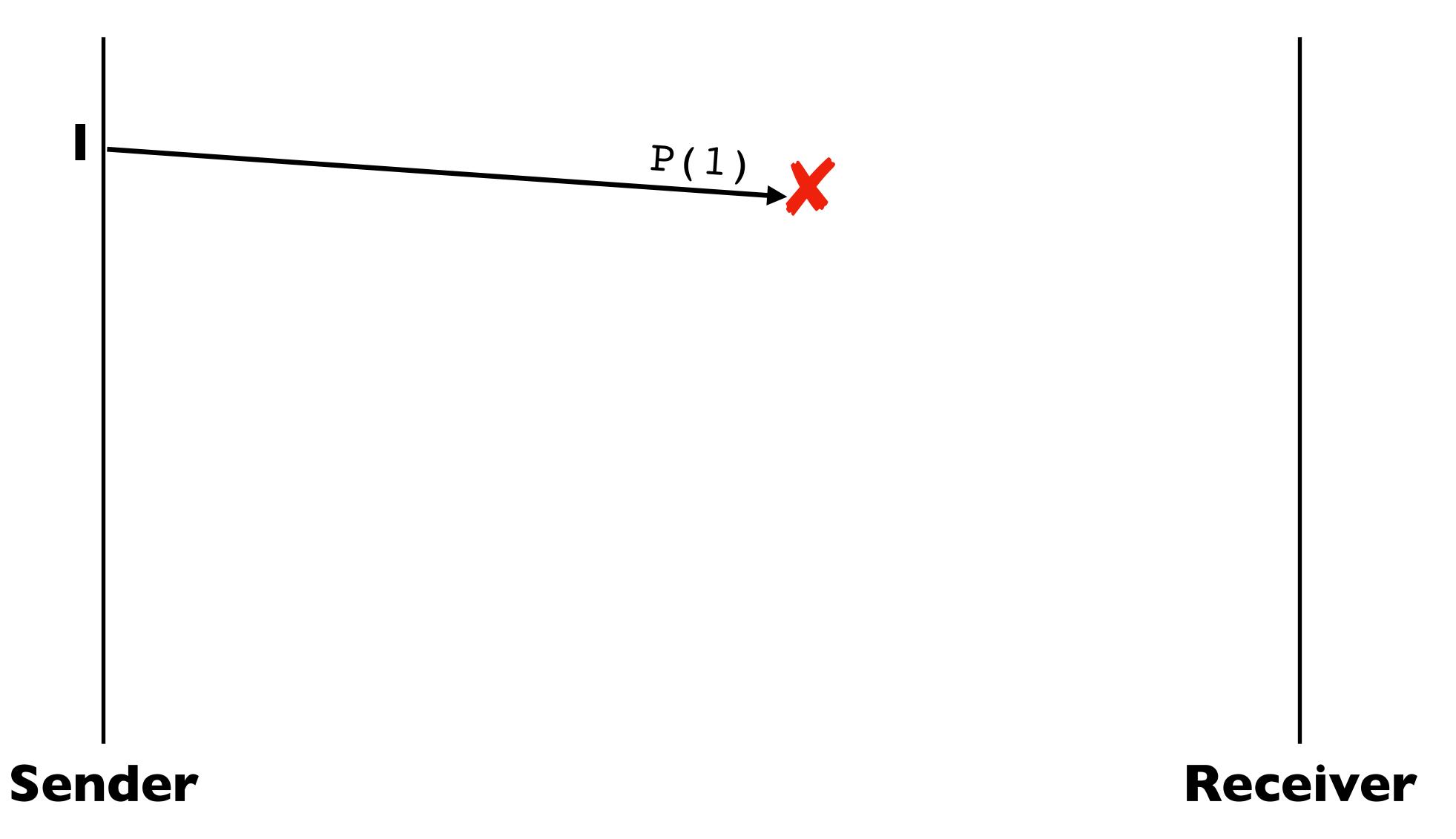


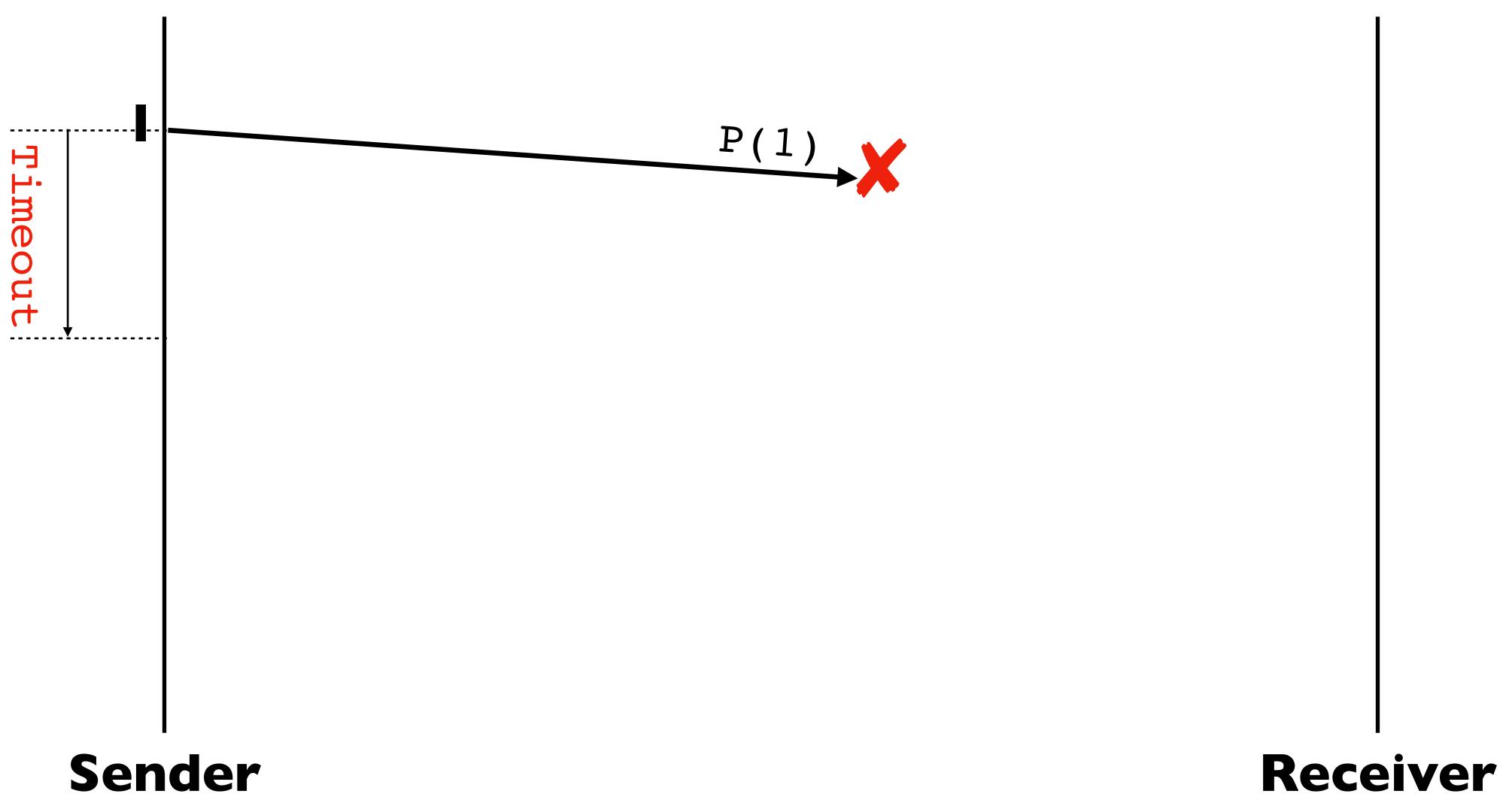


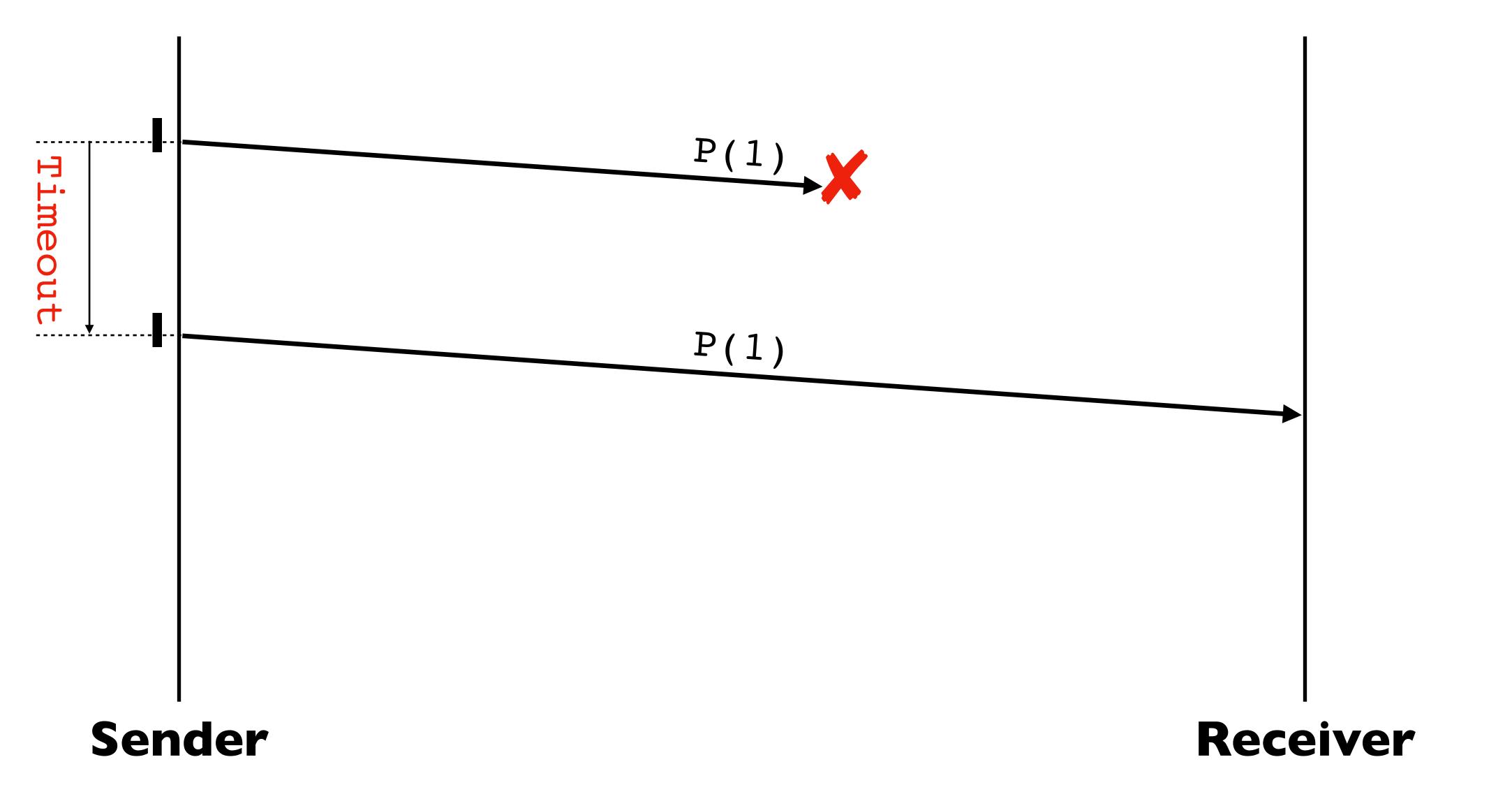


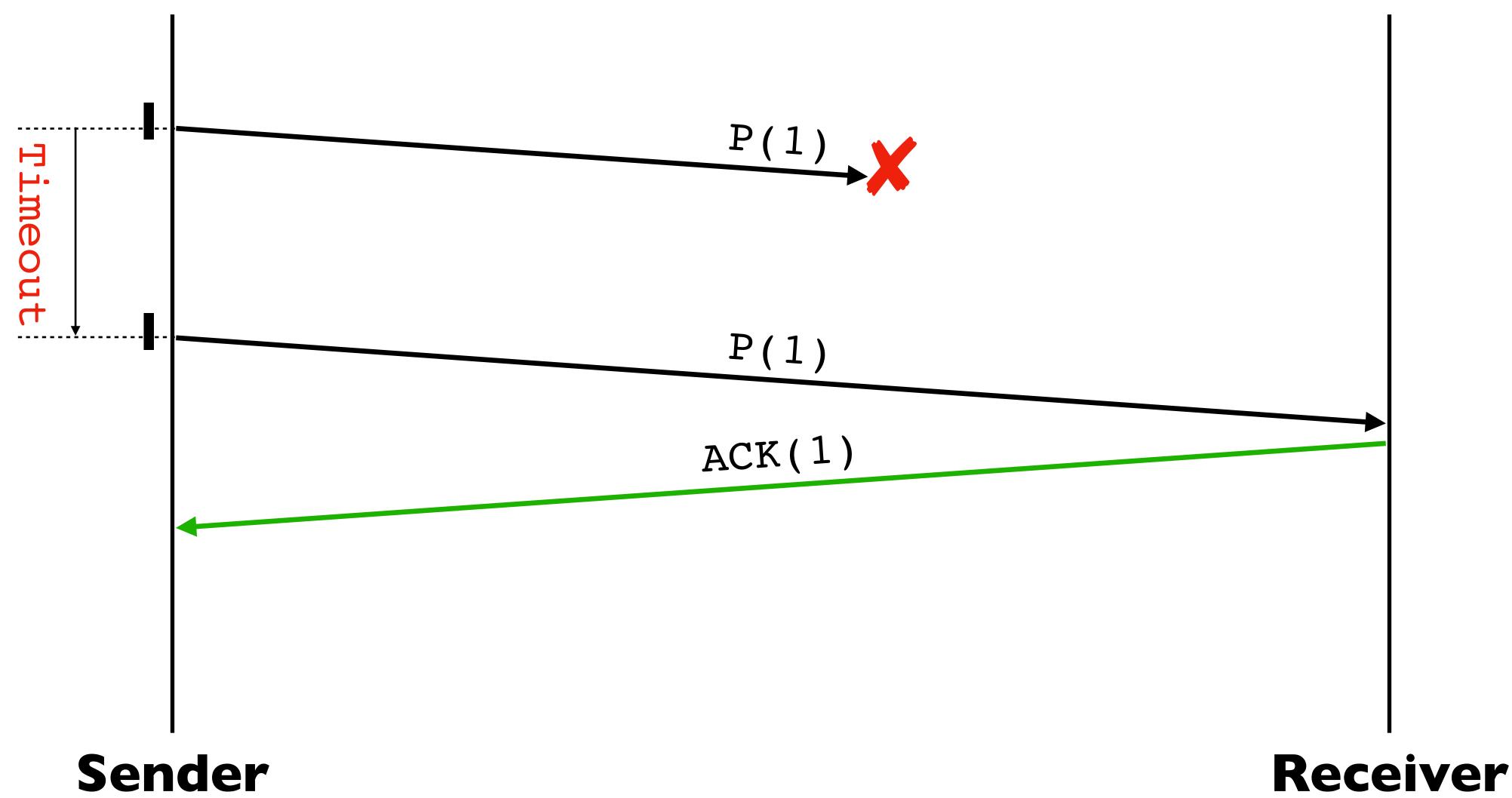


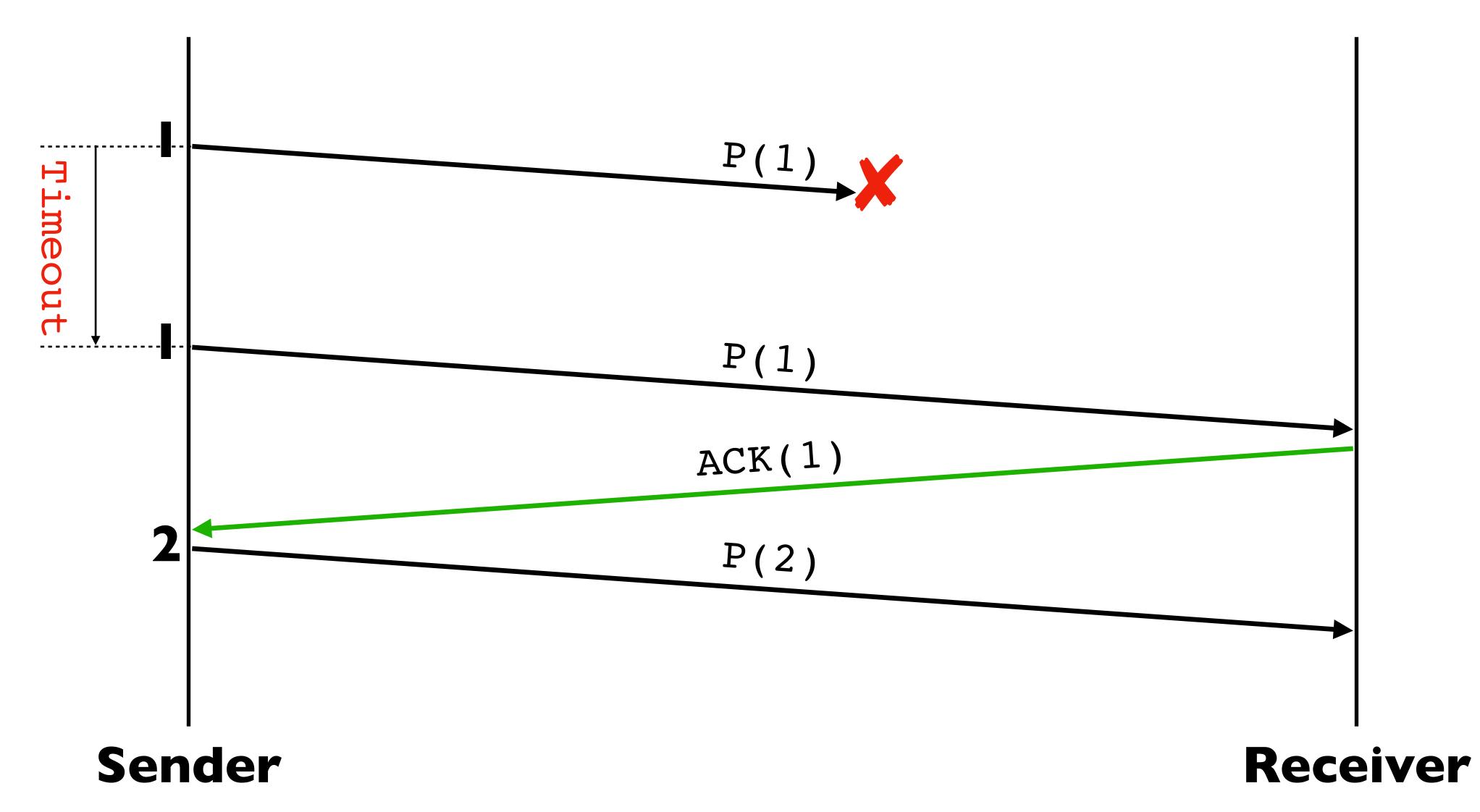


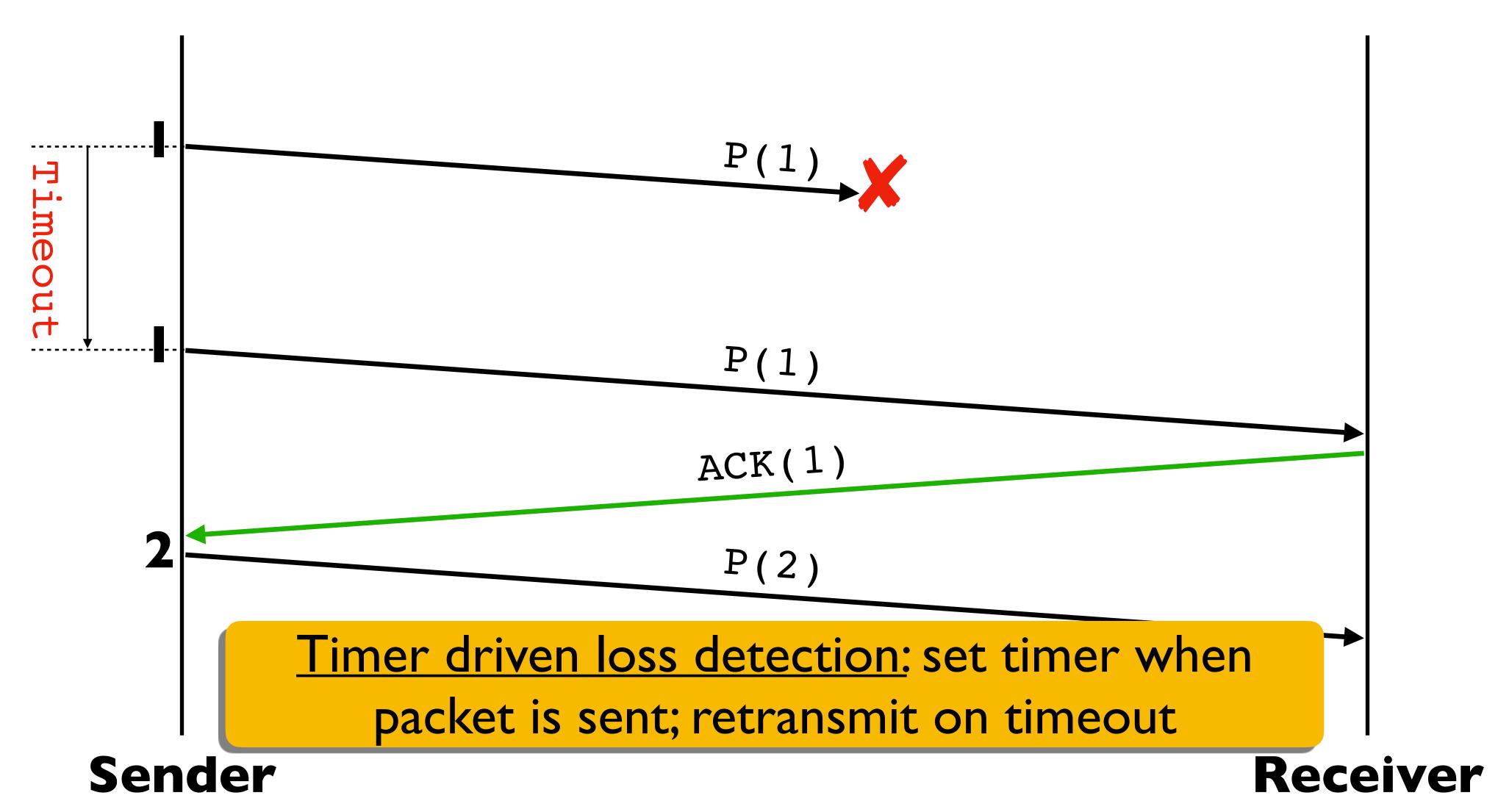


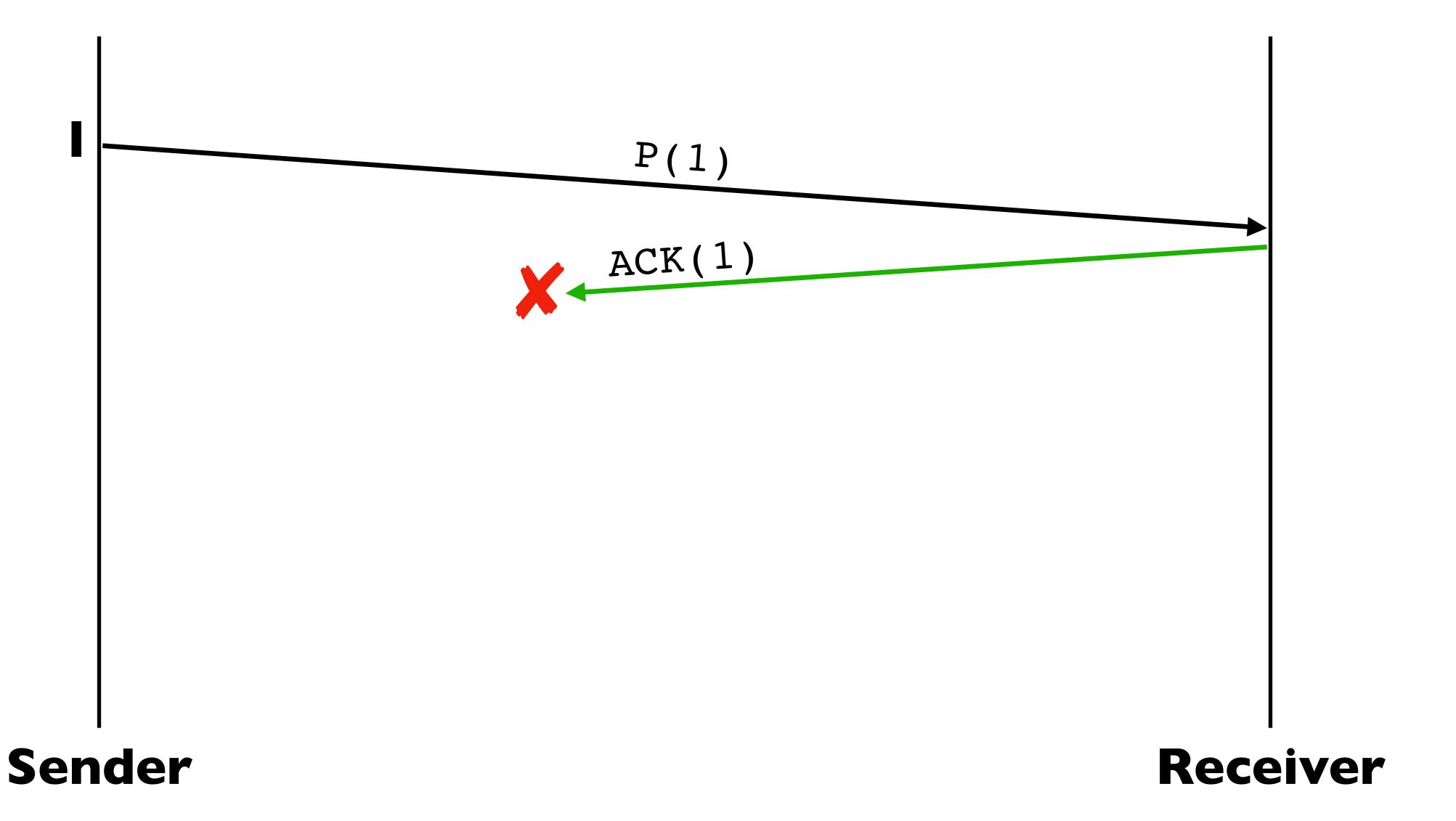


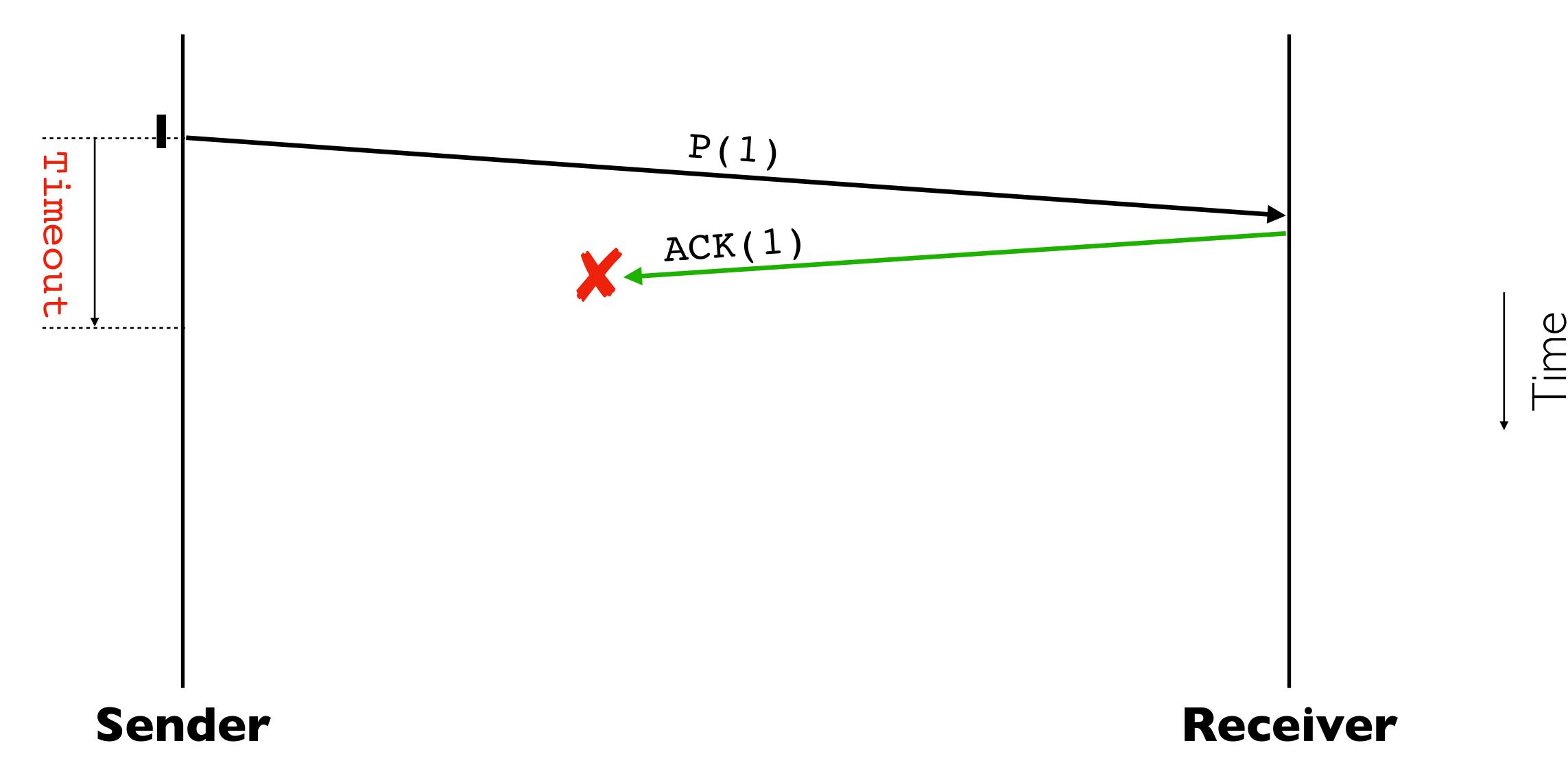


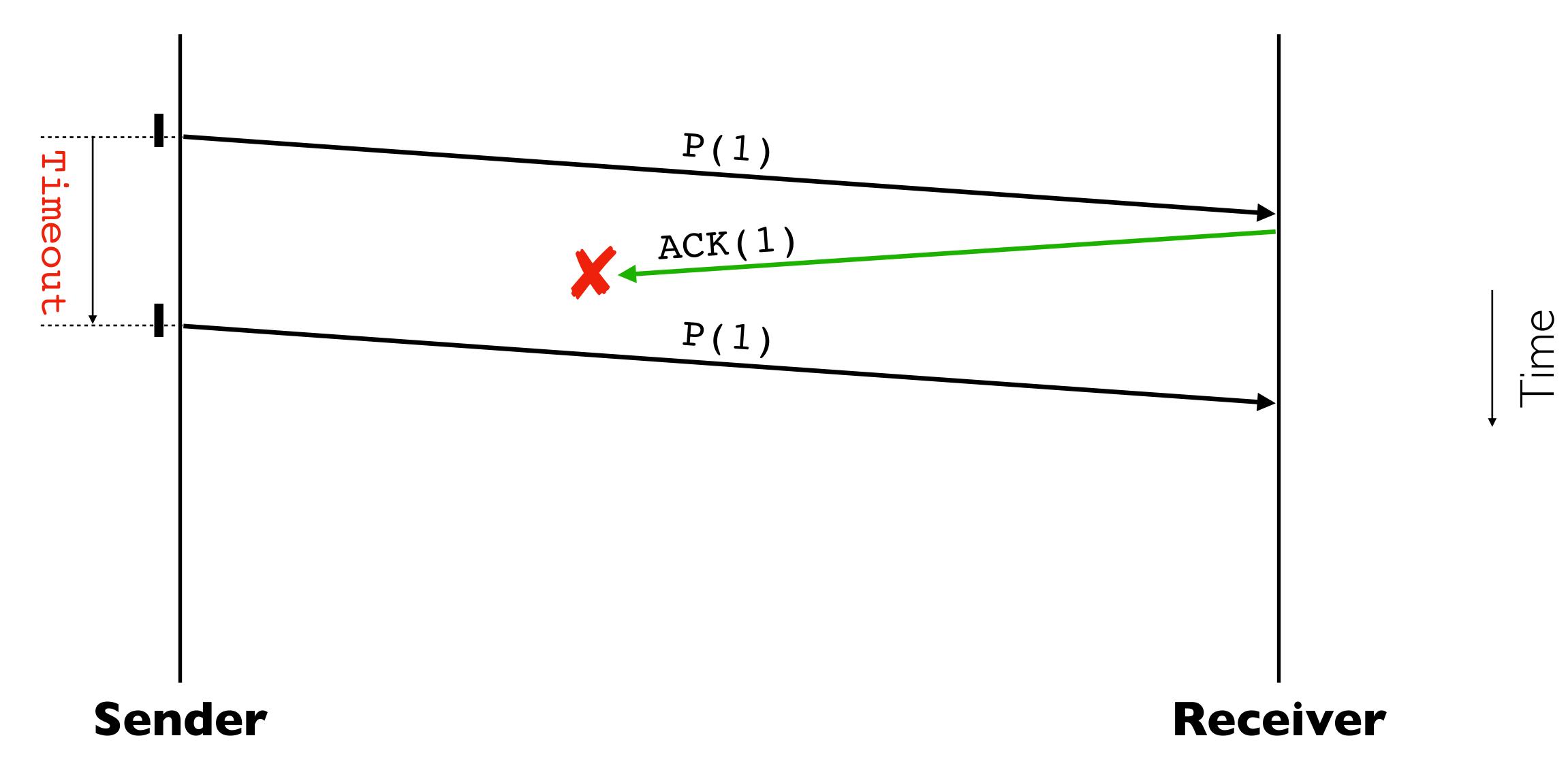


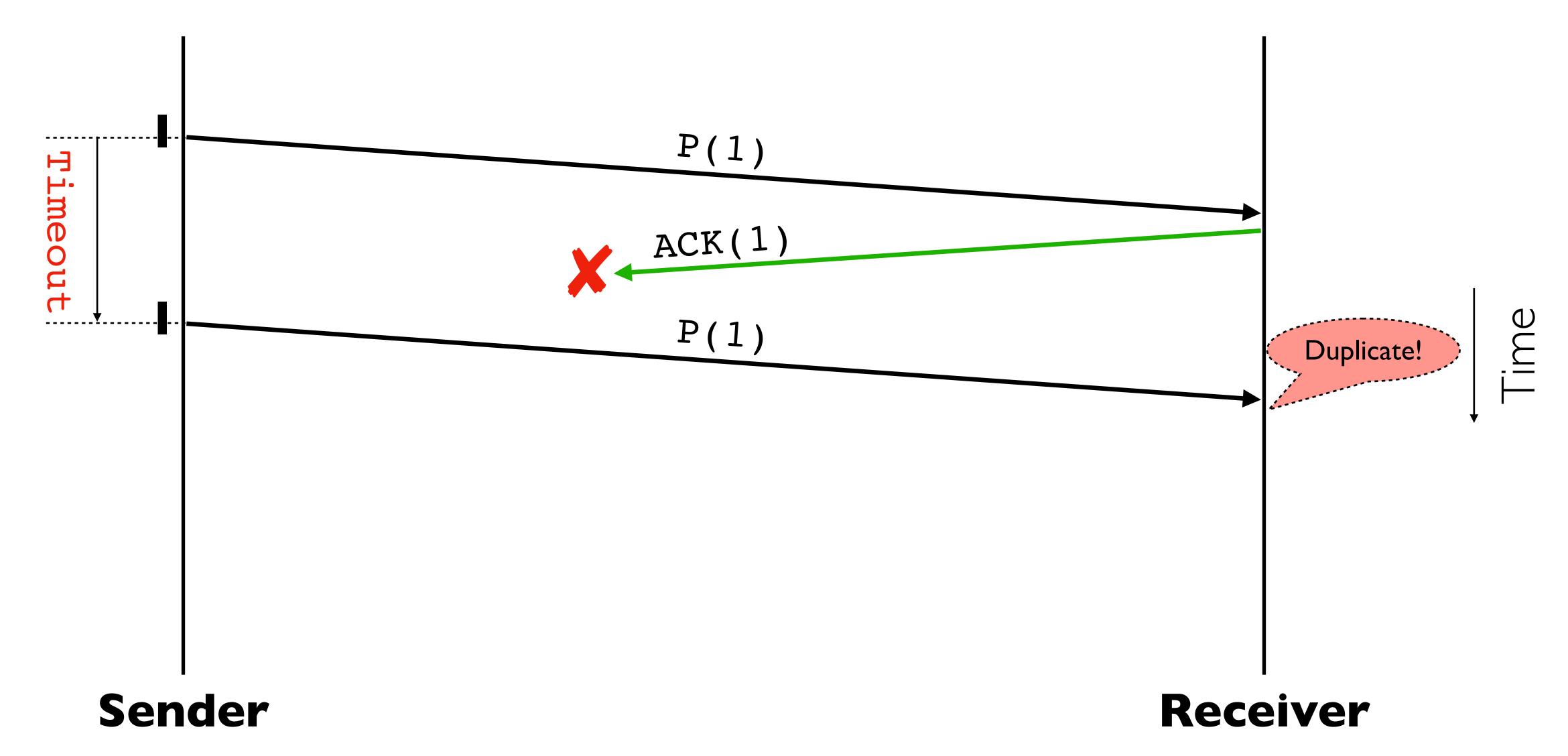


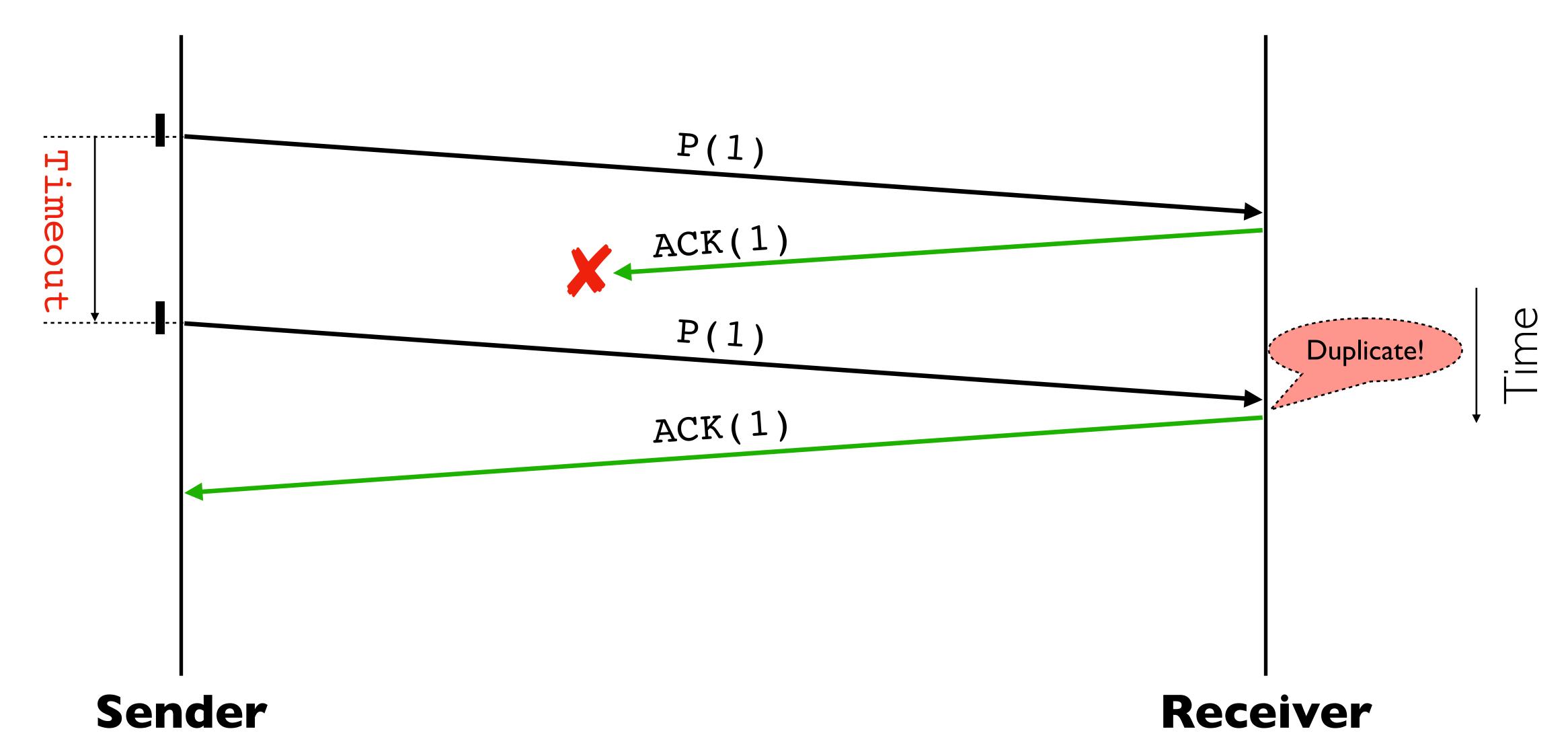


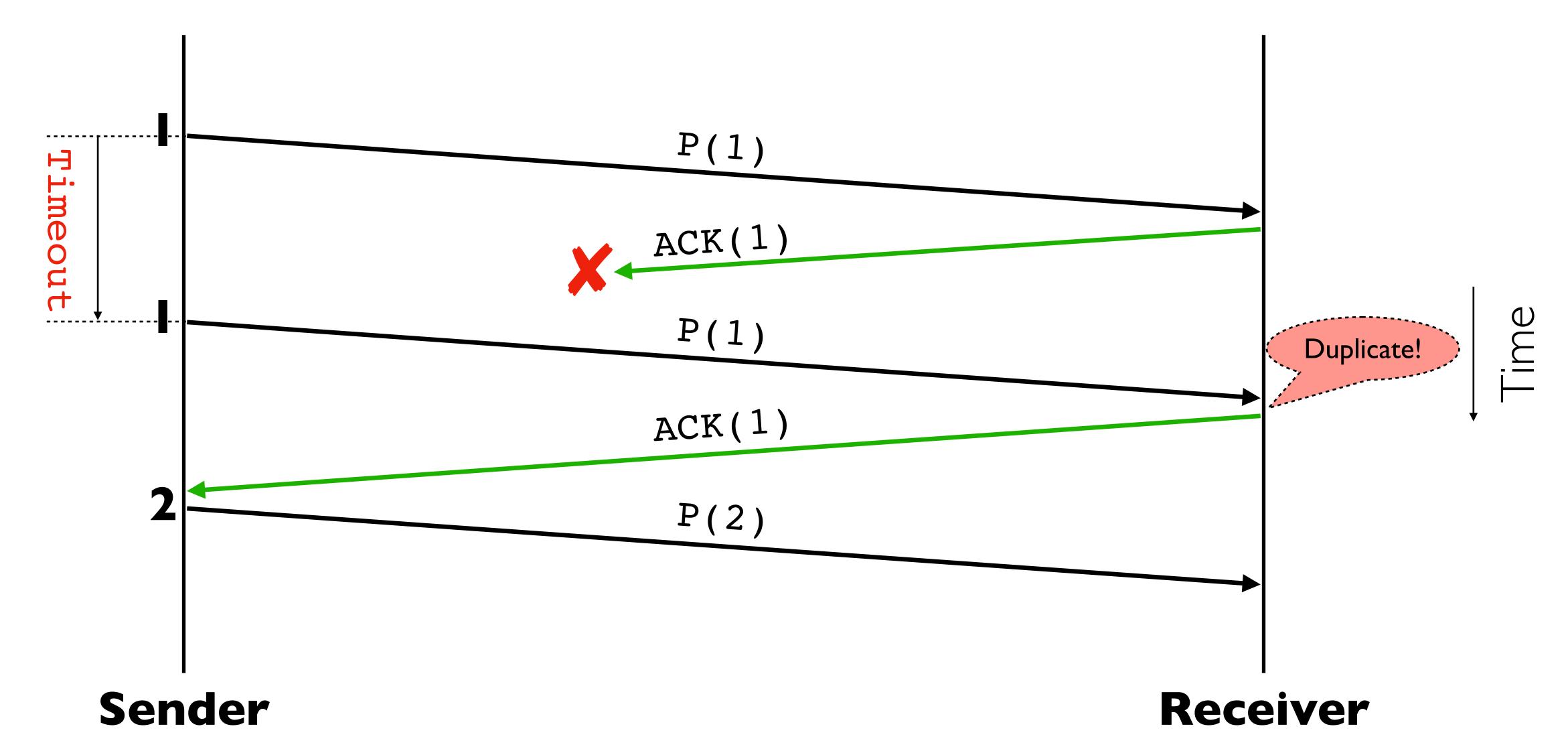


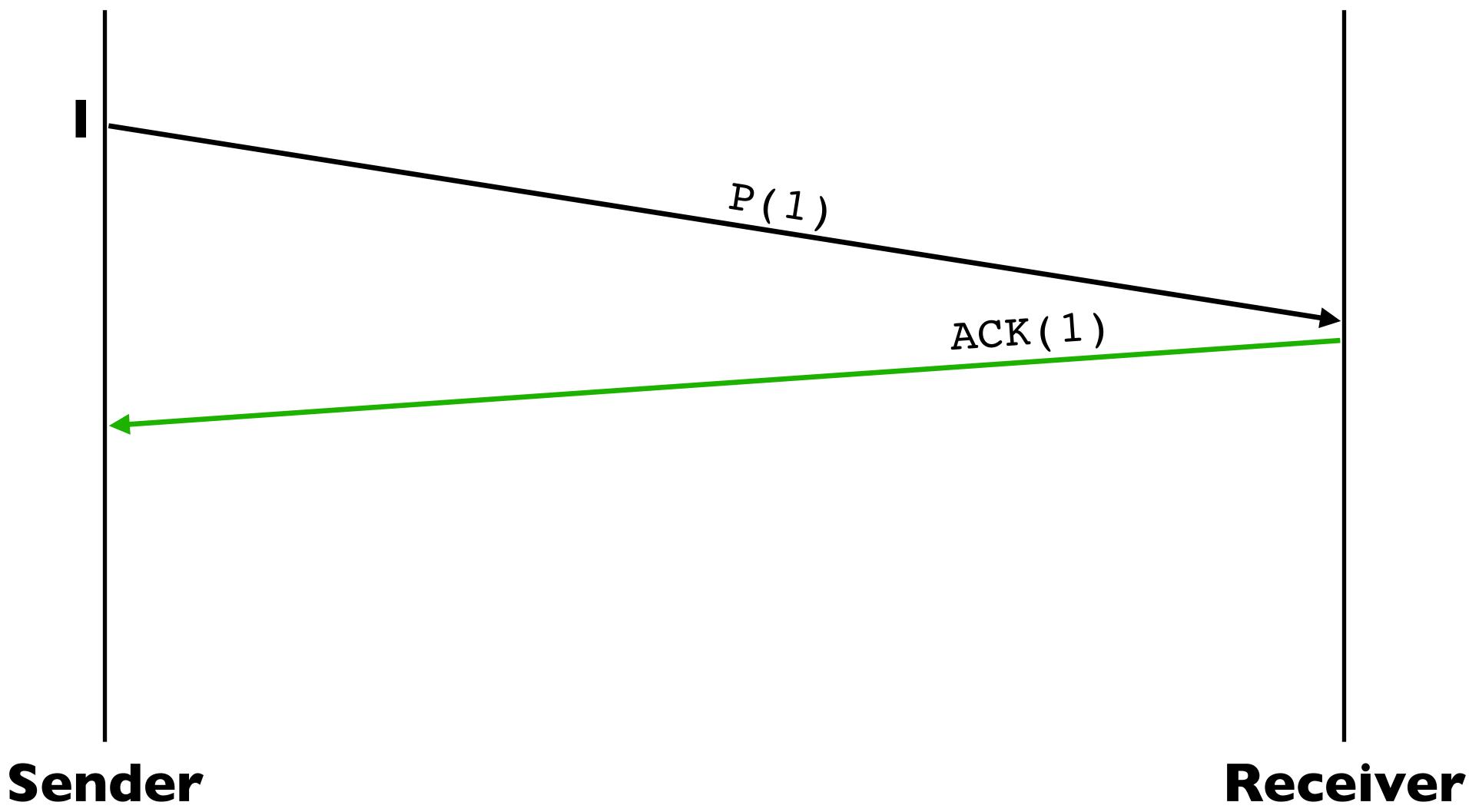


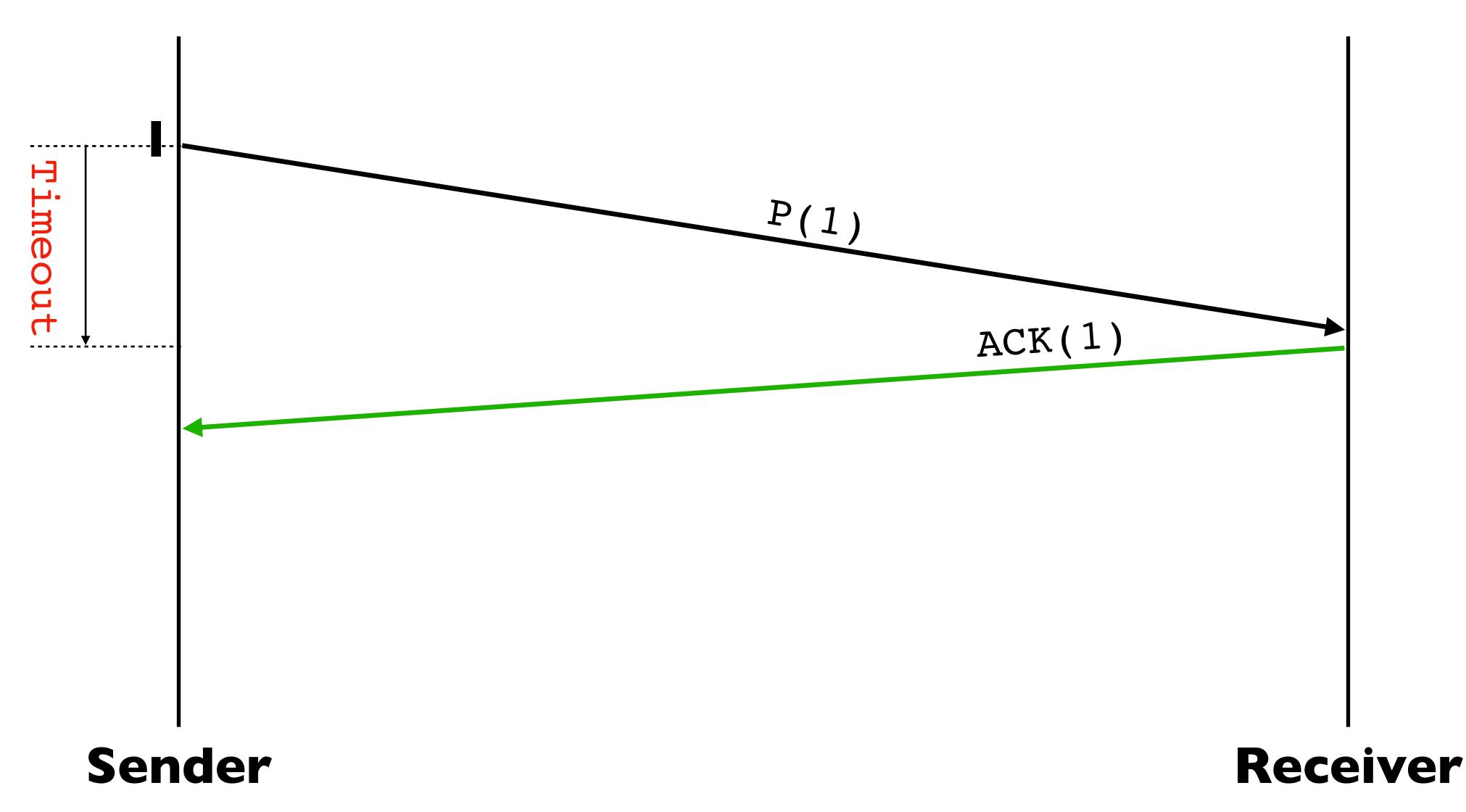


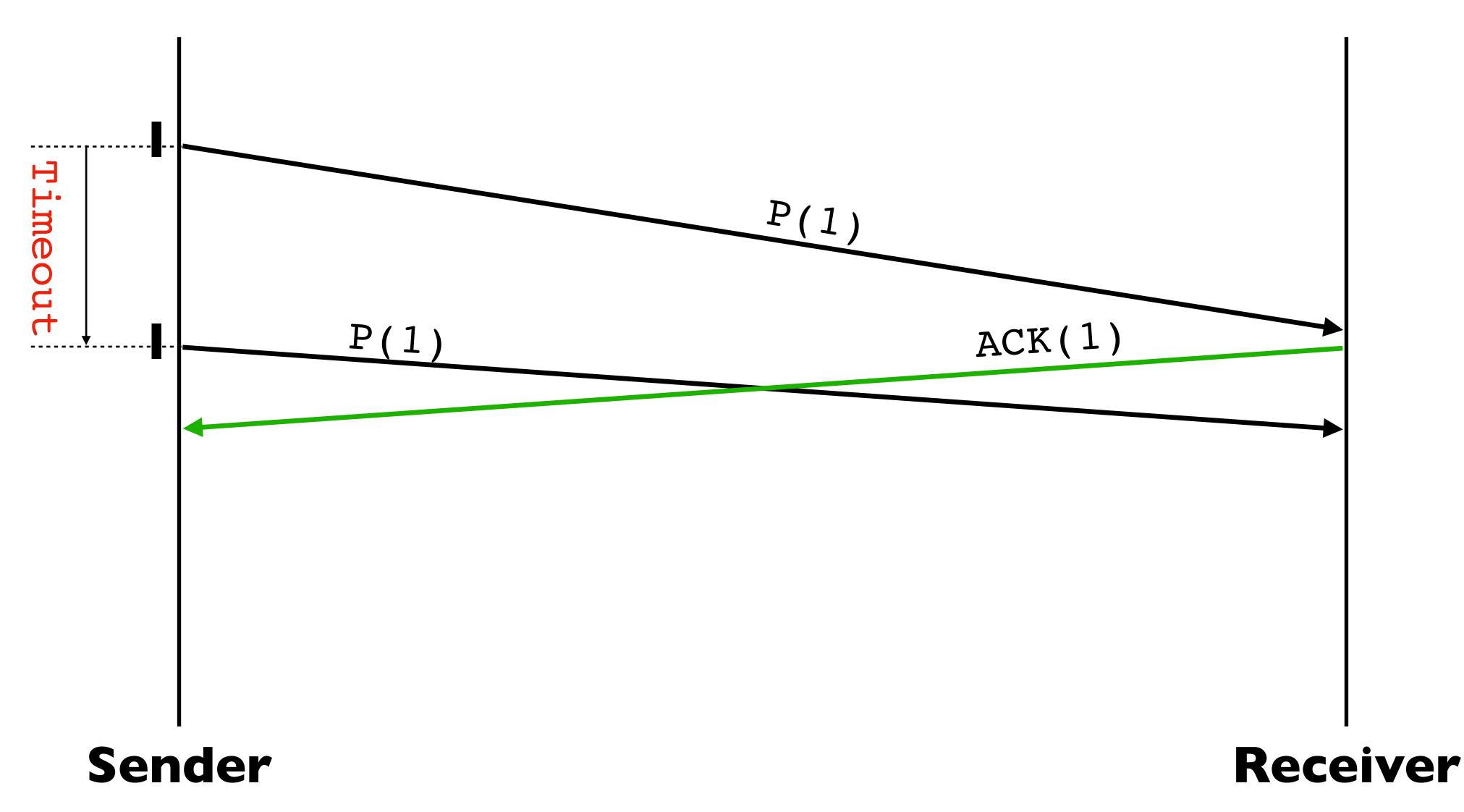


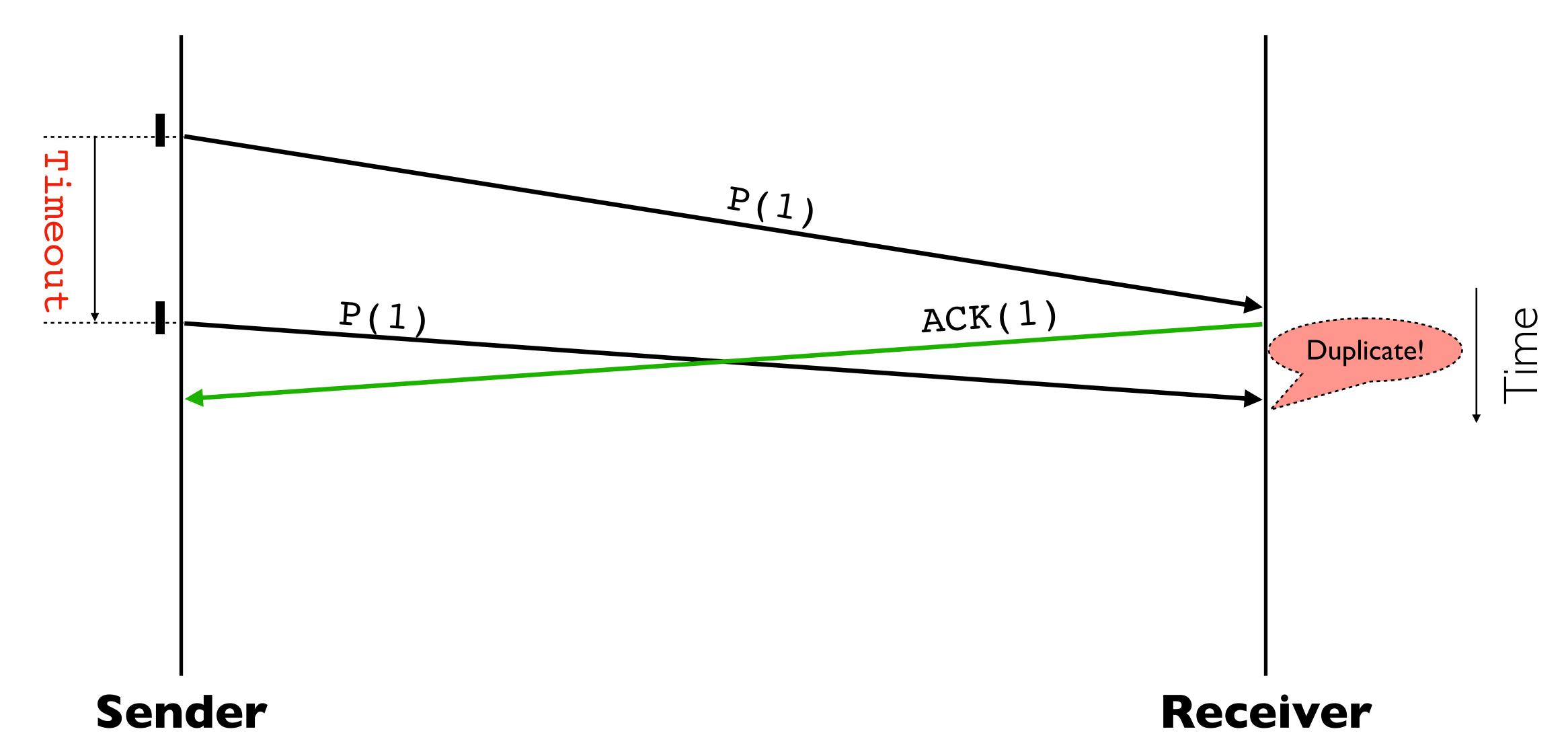


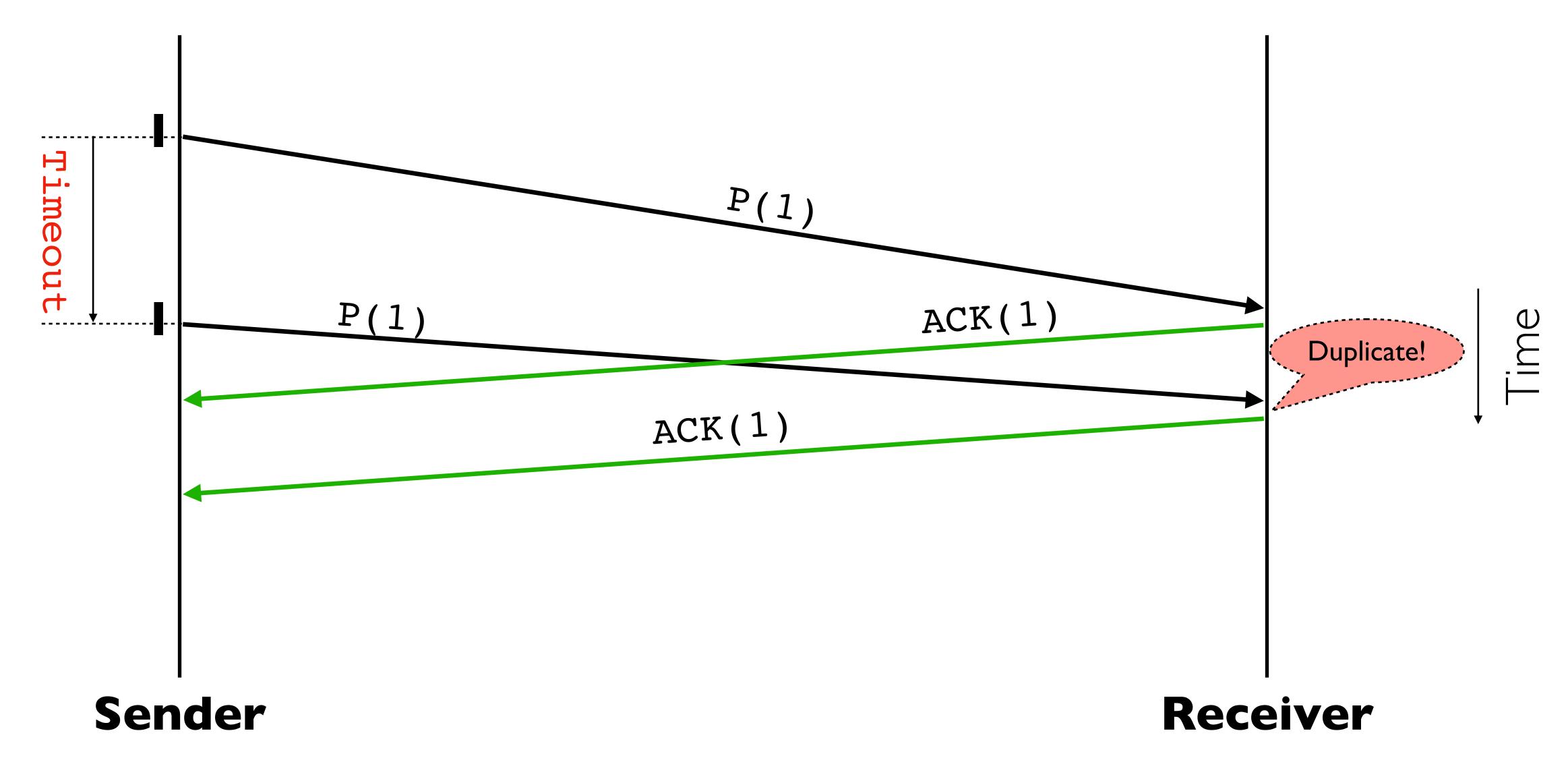


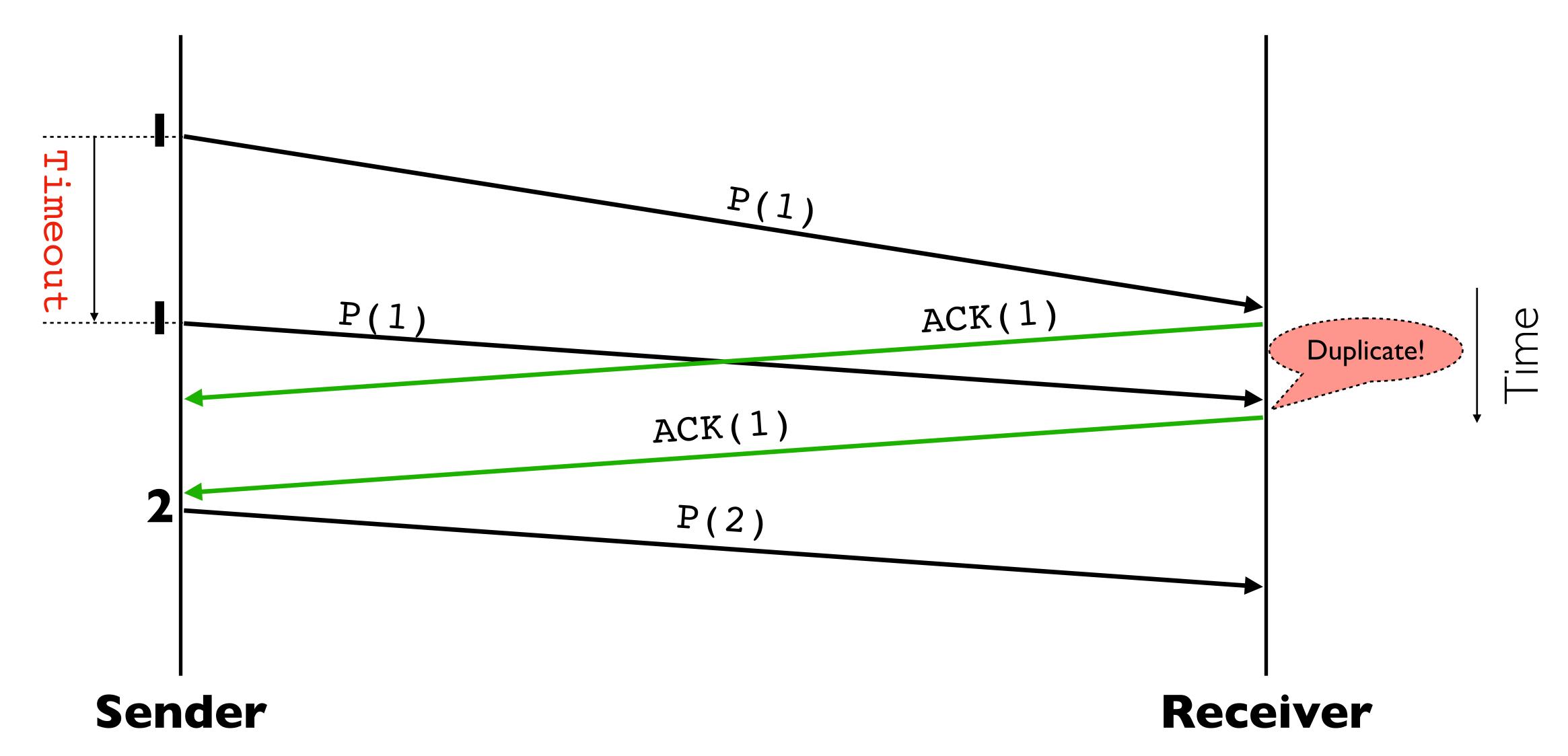




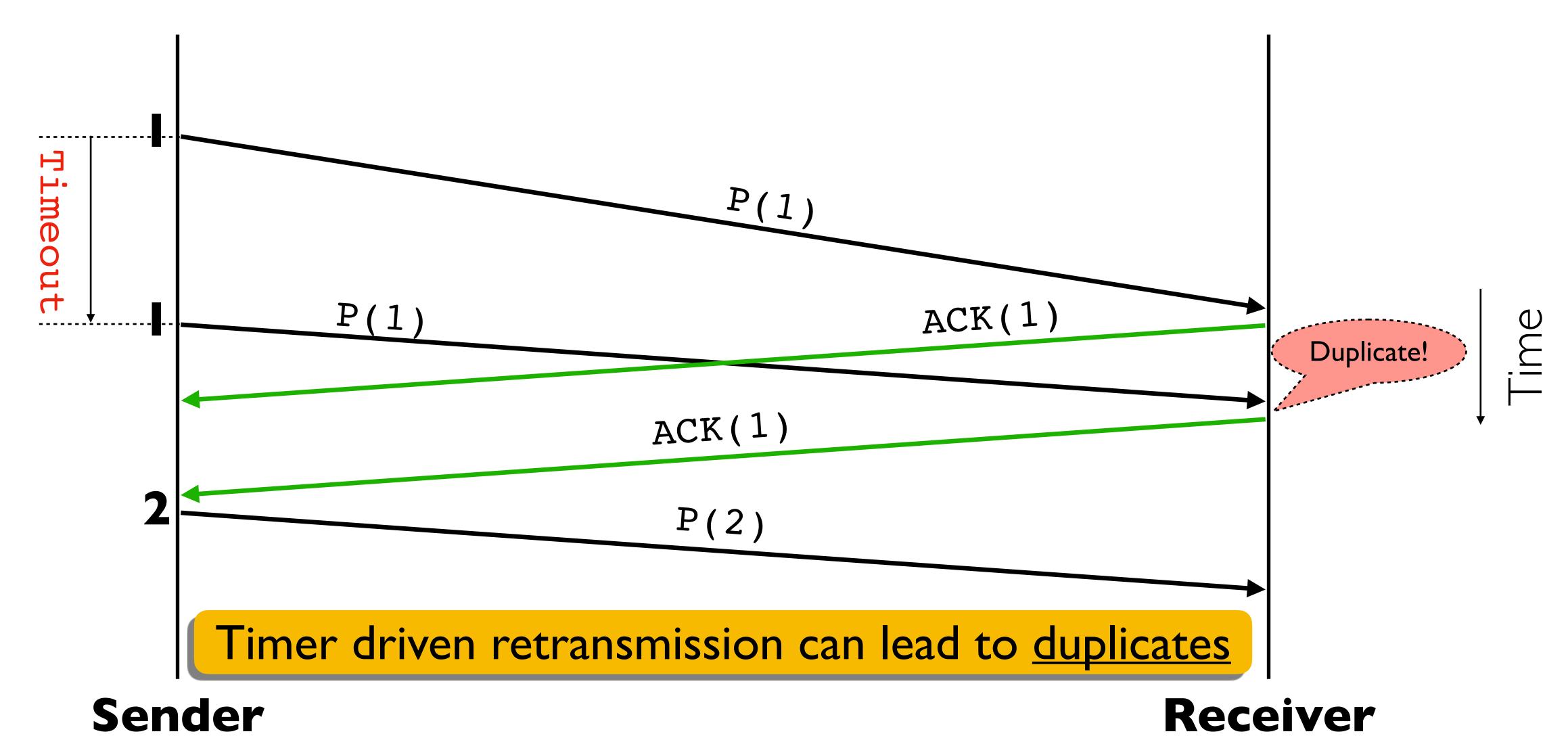








Dealing with Packet Delays



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- Timeouts: a way of deciding when to resend a packet
- But we have not put 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(i)
- If (packet is OK)
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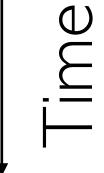
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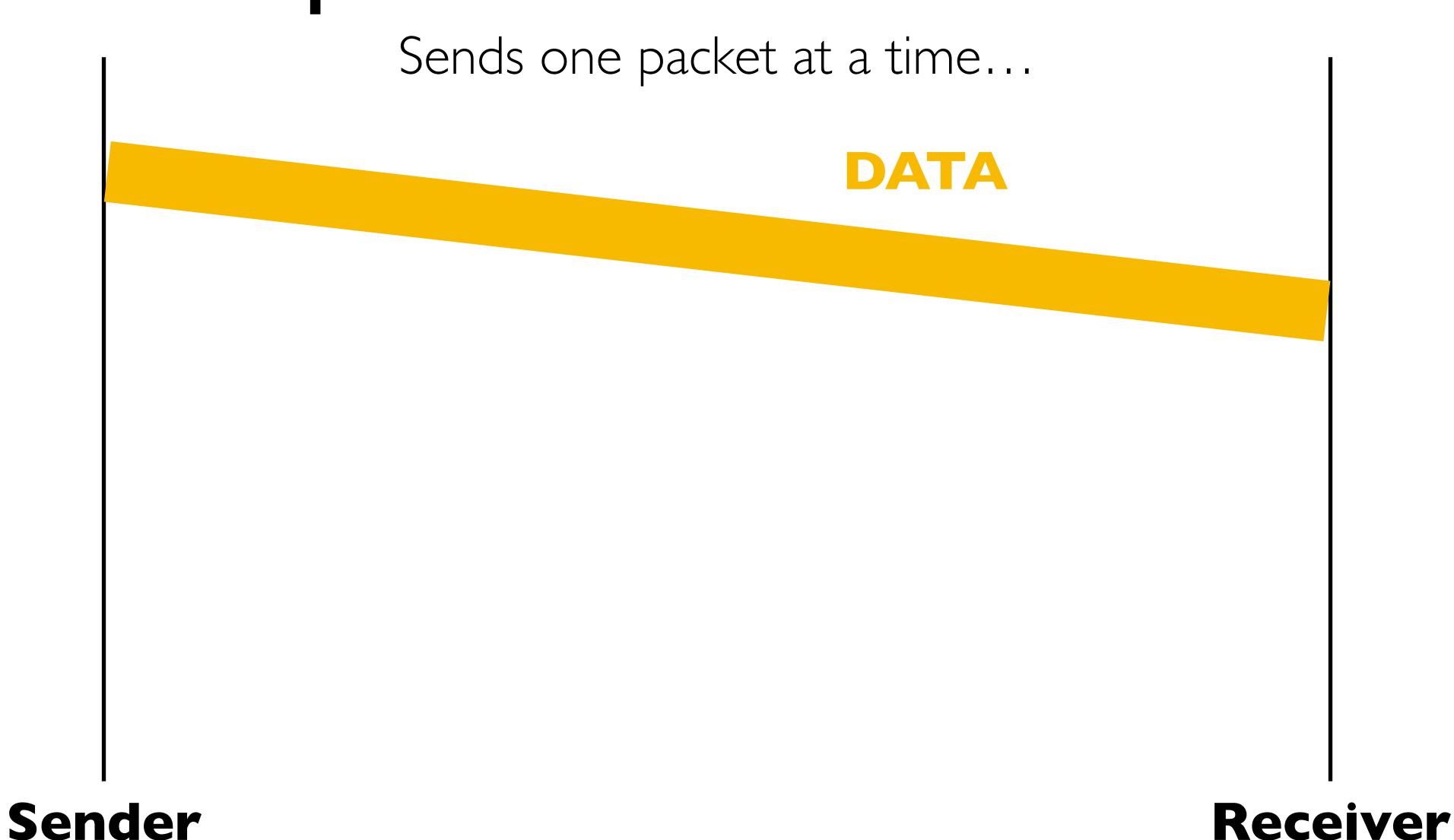
- We have a correct reliable protocol!
- Probably the world's most inefficient one... why?

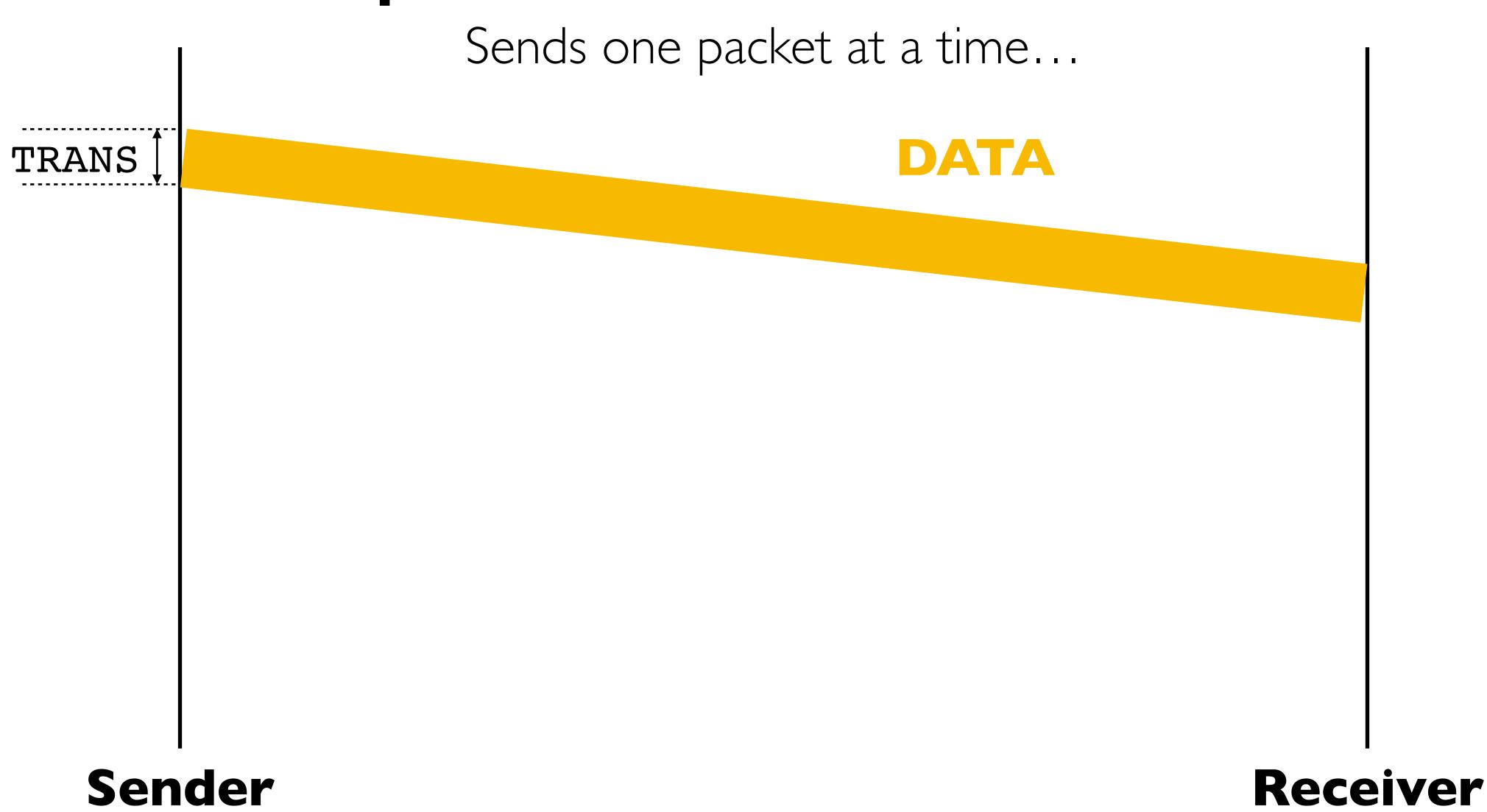


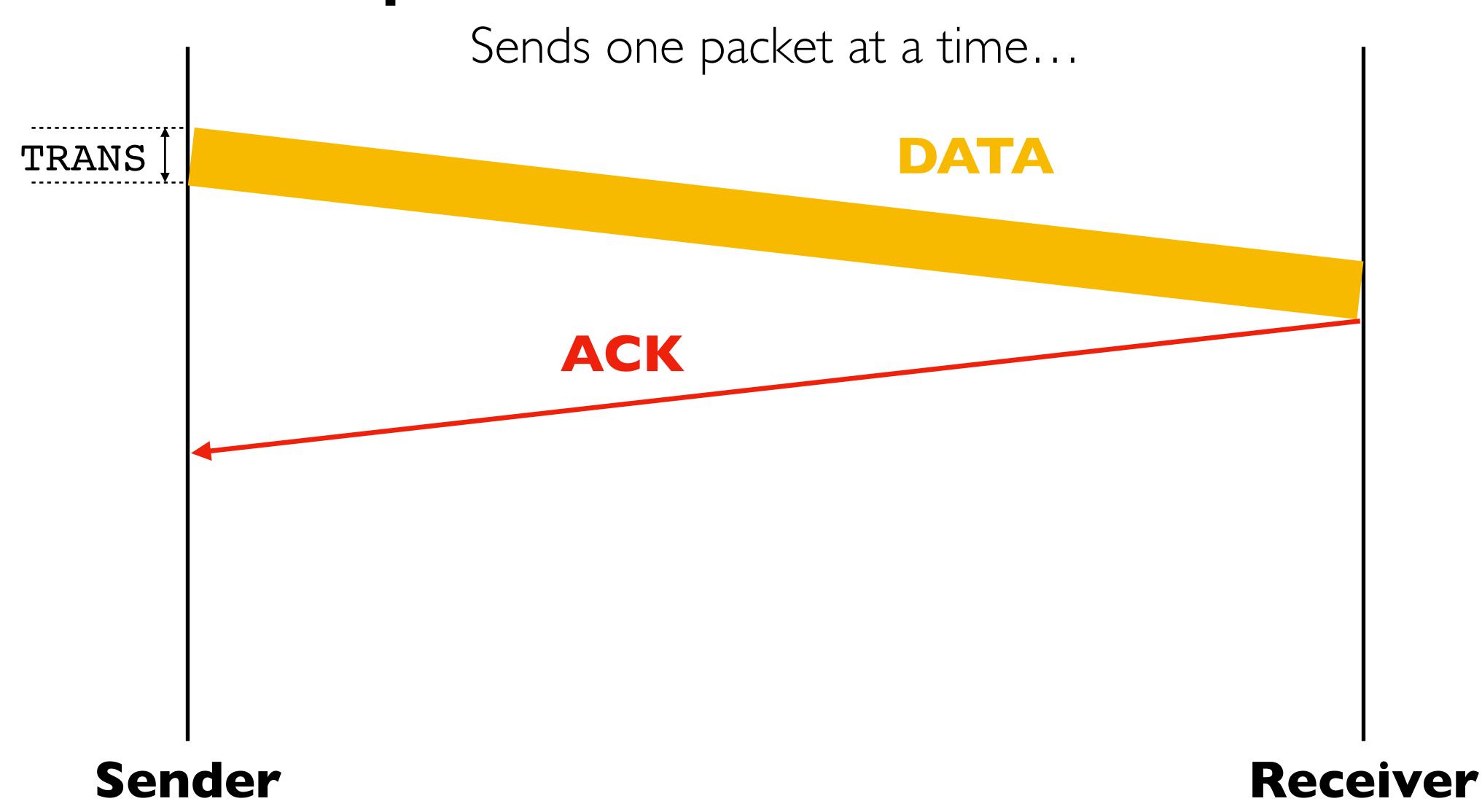
Sends one packet at a time...

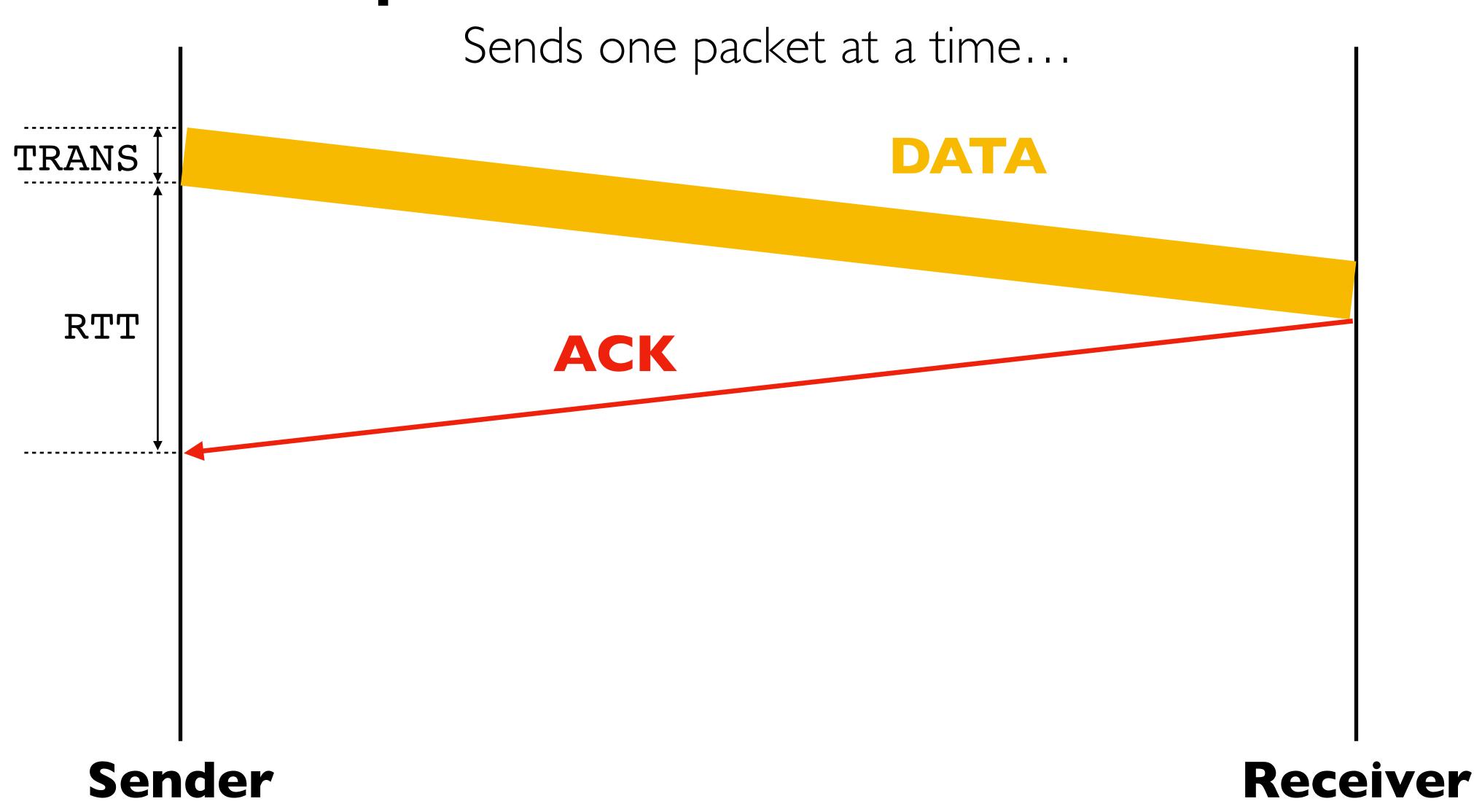


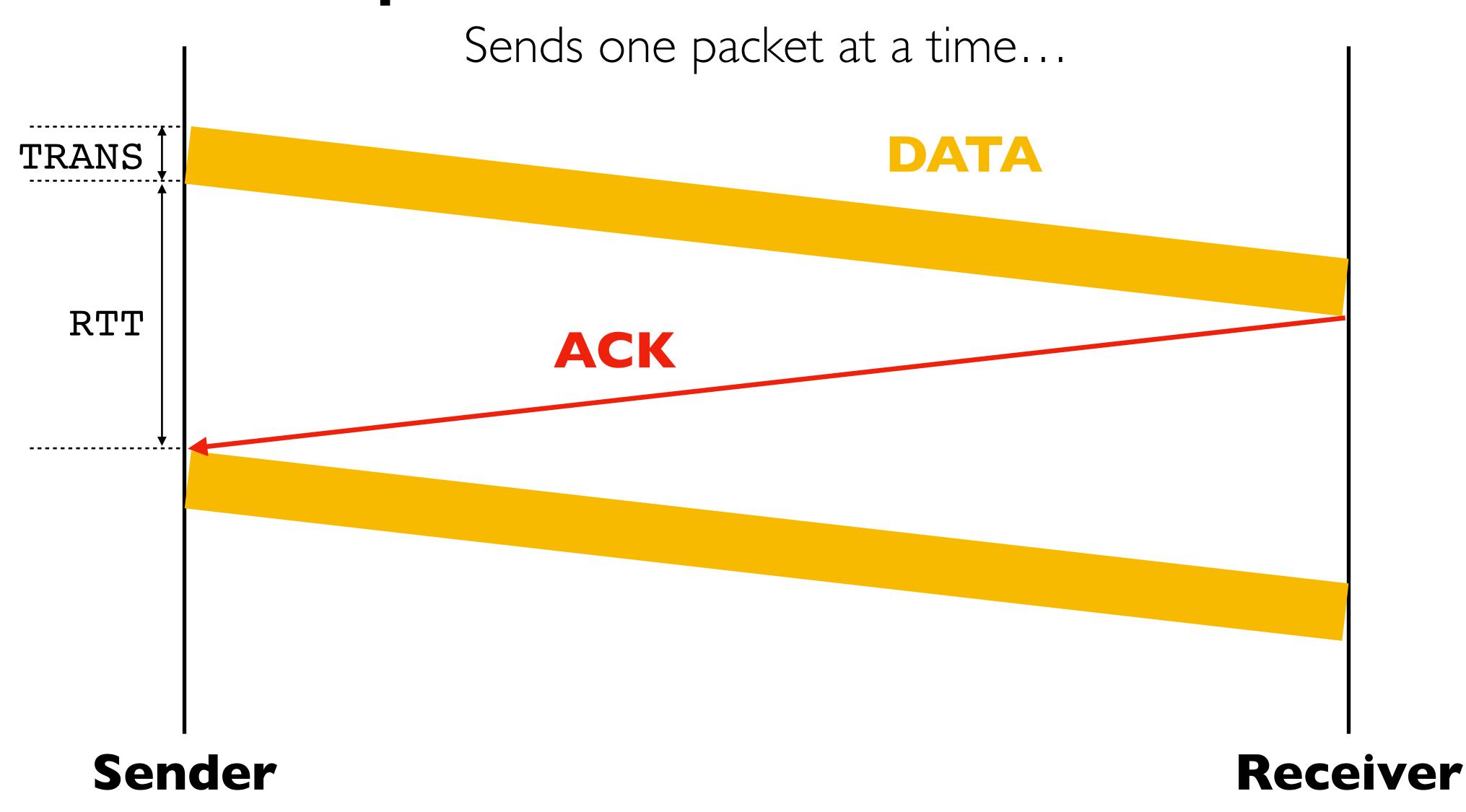
Sender Receiver

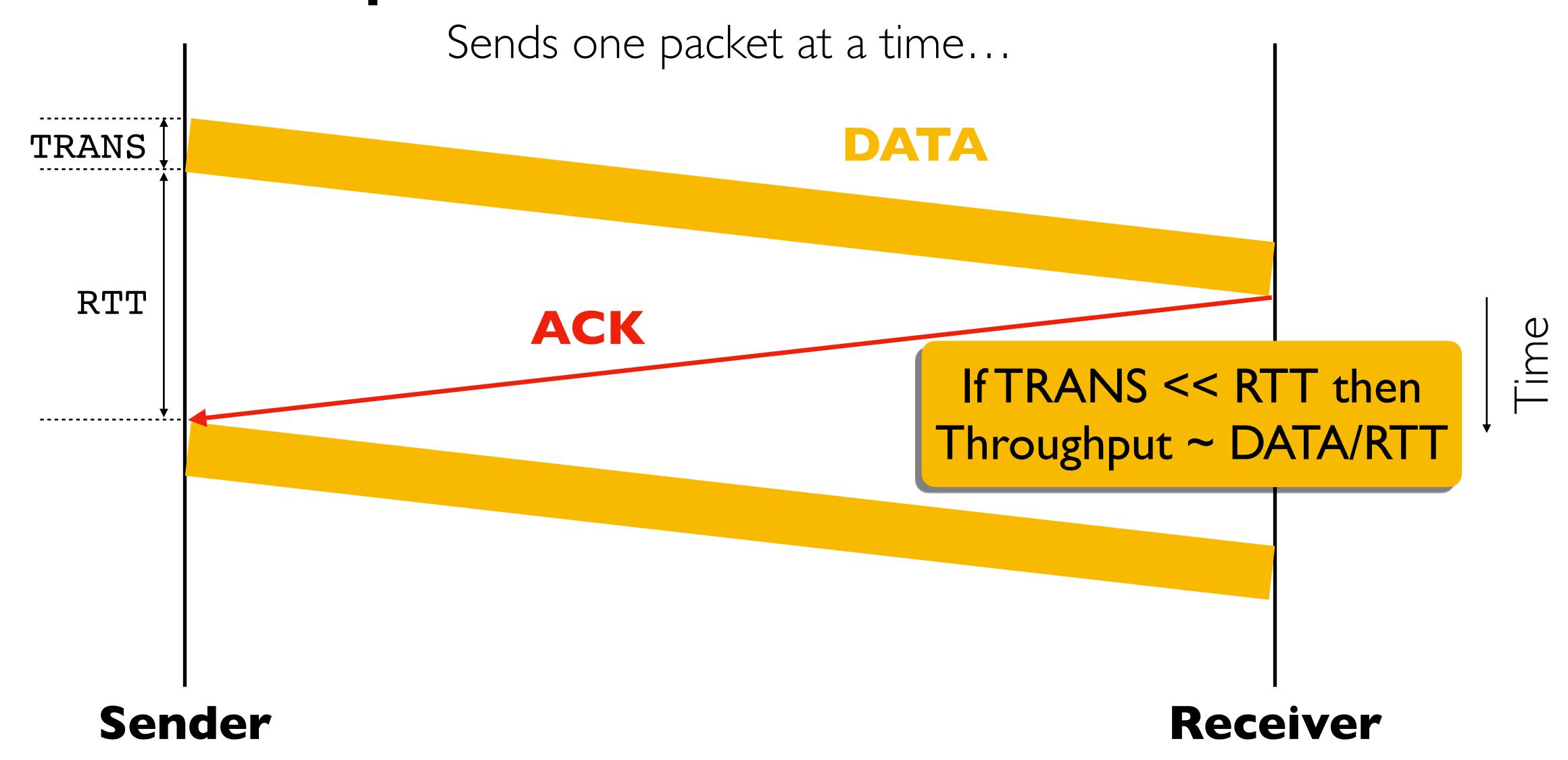












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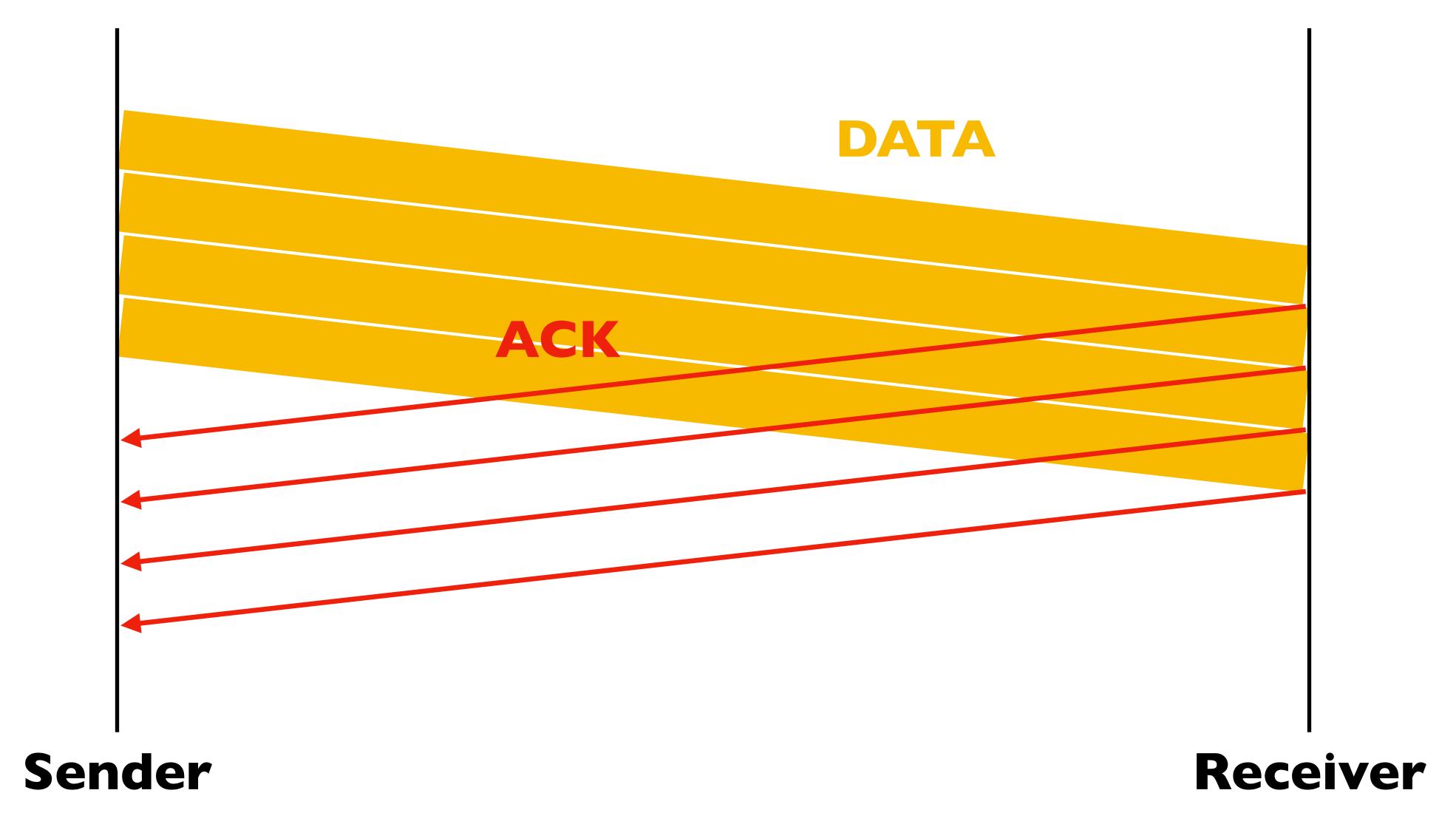
Throughput:

• 1500 byte / 10 ms = 1.5 Mbps!

How to make it more efficient?



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• Which packets can sender send?

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 - Sliding window

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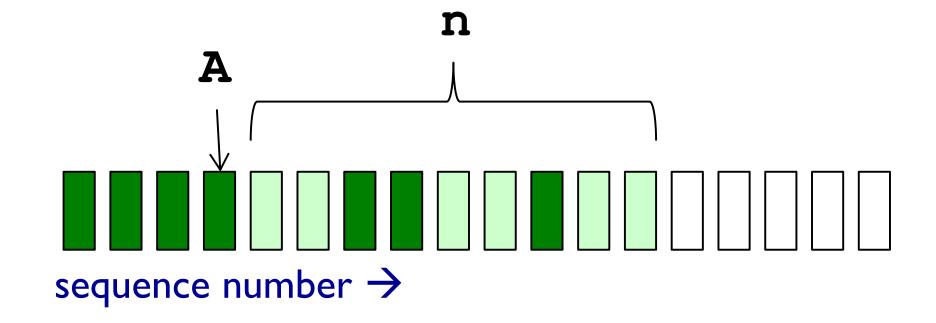
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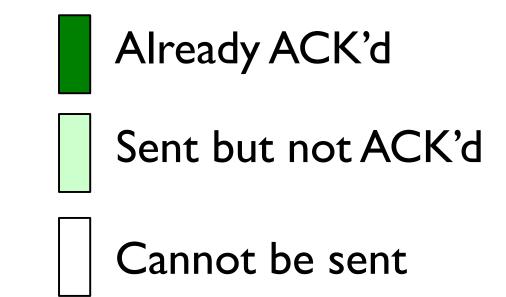
• General Idea: send up to n packets at a time

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- Window of acceptable packets slides on successful reception/acknowledgement

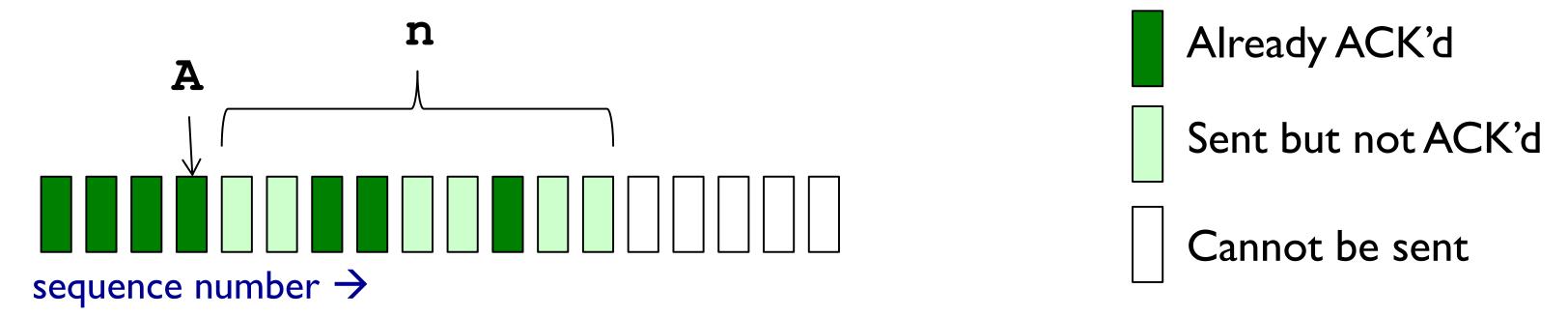
- Window = set of adjacent sequence numbers
 - The size of the set is the window size; assume window size is n
- General Idea: send up to n packets at a time
 - Sender can send packets in its window
 - Receiver can accept packets in its window
 - Window of acceptable packets slides on successful reception/acknowledgement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

• Let A be the last ACK'd packet of sender without gap; then window of sender = $\{A+1,A+2,...,A+n\}$





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• Let B be the **last received packet without gap** by receiver; then window of receiver = $\{B+1, ..., B+n\}$



• If window size is n, then throughput is roughly:

MIN[n * DATA/RTT, Link bandwidth]

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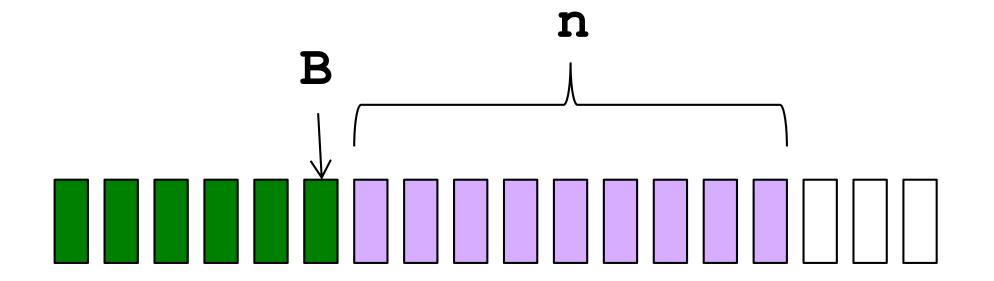
MIN[DATA/RTT, Link bandwidth]

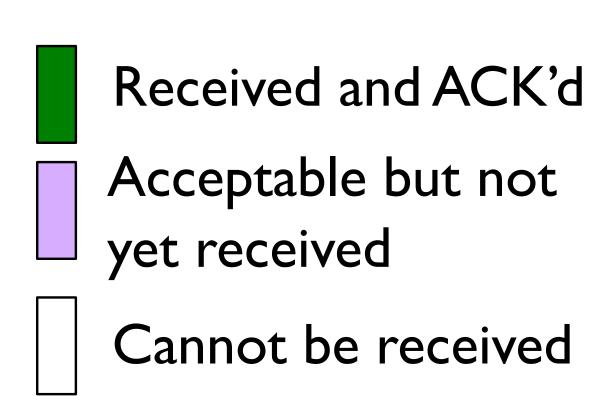
- Two questions:
 - What happens when n gets too large?
 - How do we choose n?

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

Cumulative Acknowledgements (I)

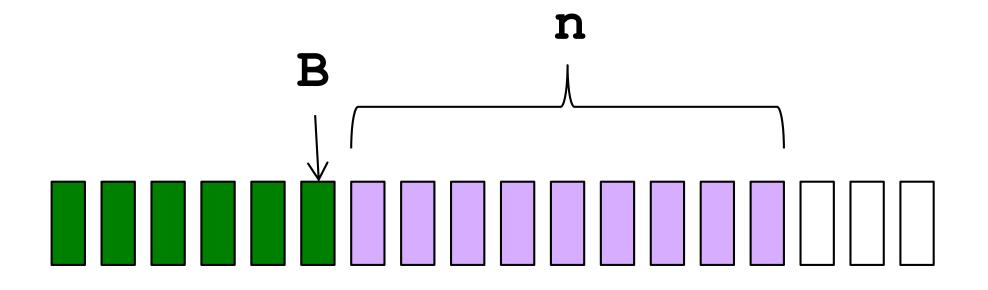
At receiver



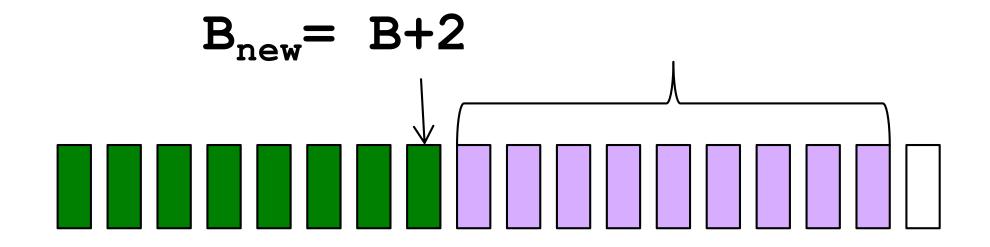


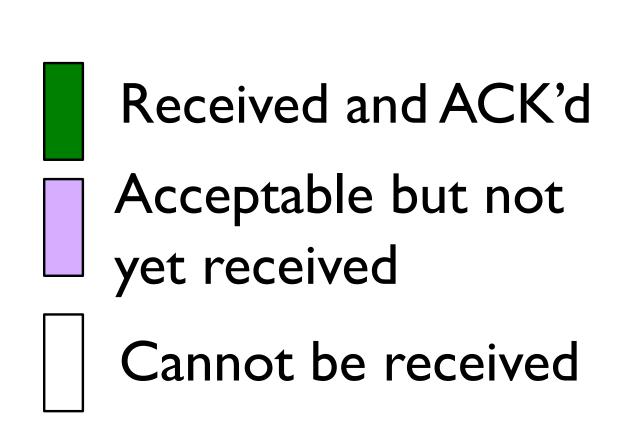
Cumulative Acknowledgements (I)

At receiver



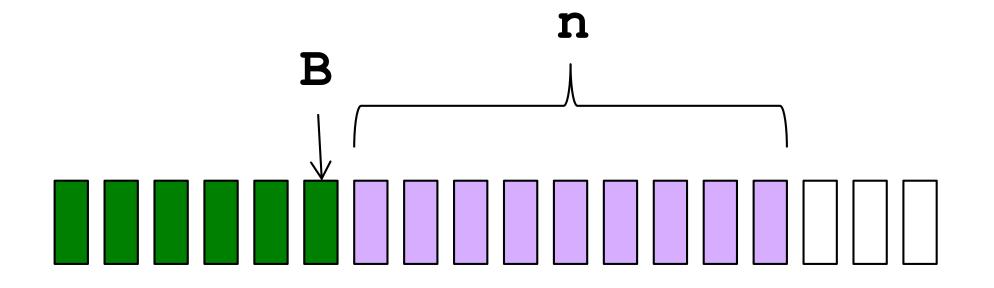
• After receiving B+1, B+2



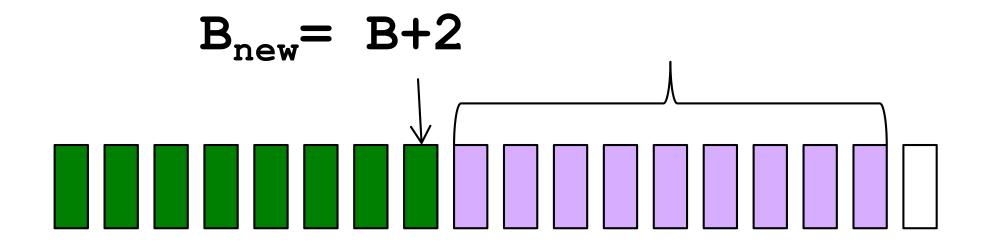


Cumulative Acknowledgements (I)

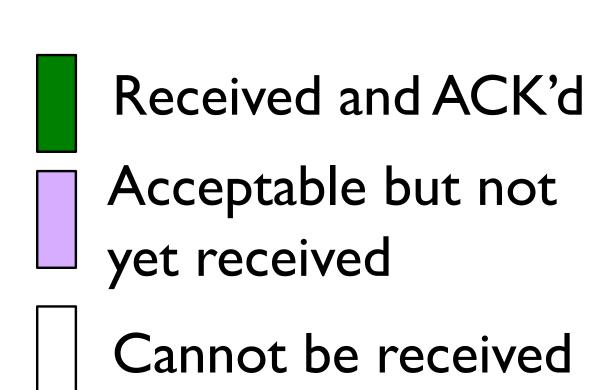
At receiver



• After receiving B+I, B+2

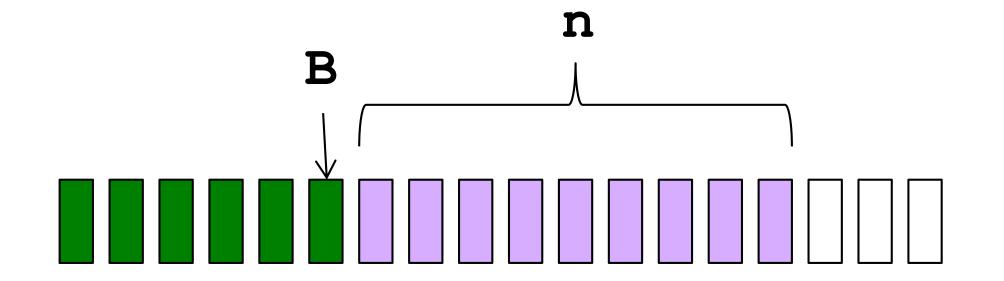


• Receiver sends ACK(B+3) = ACK(B_{new}+1)



Cumulative Acknowledgements (2)

At receiver



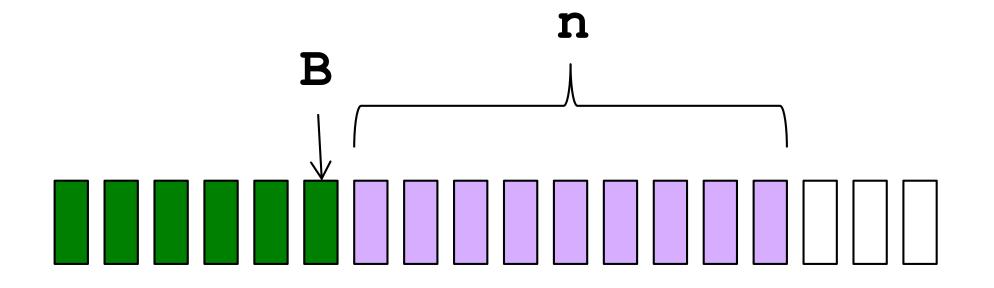
Received and ACK'd

Acceptable but not yet received

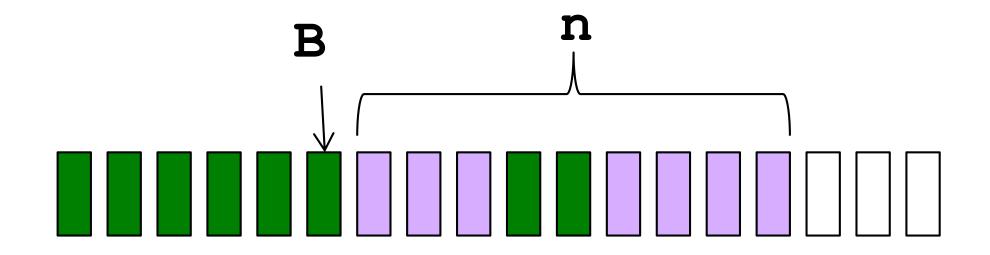
Cannot be received

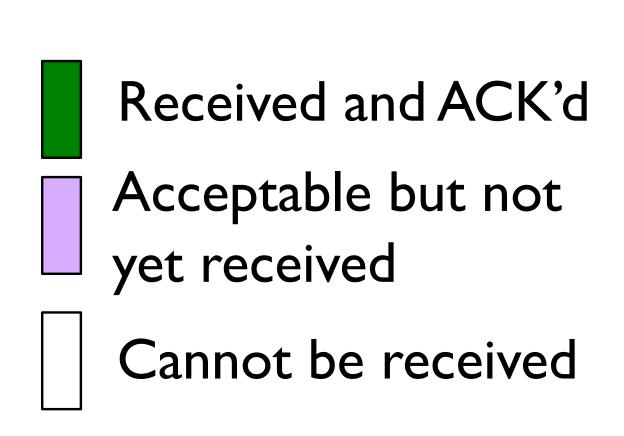
Cumulative Acknowledgements (2)

At receiver



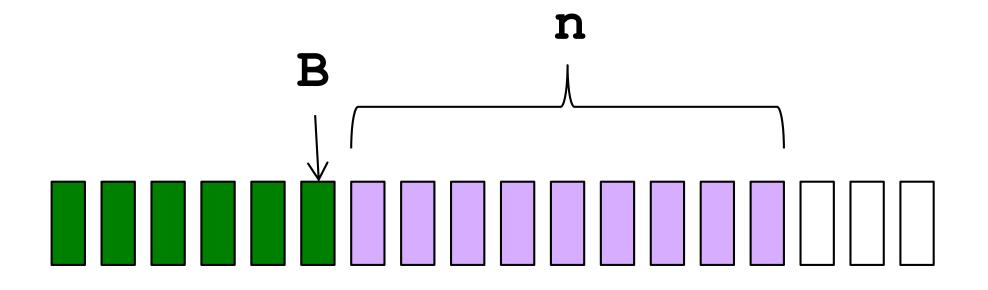
• After receiving B+4, B+5



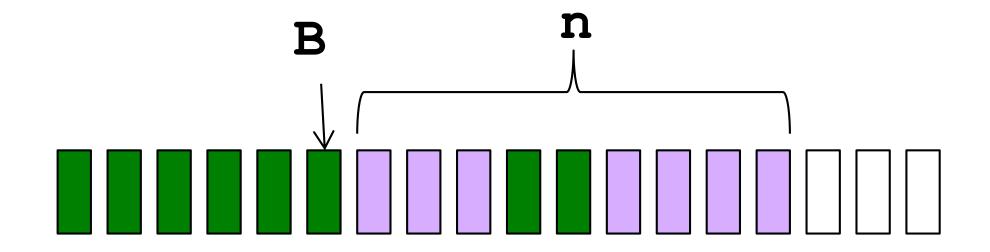


Cumulative Acknowledgements (2)

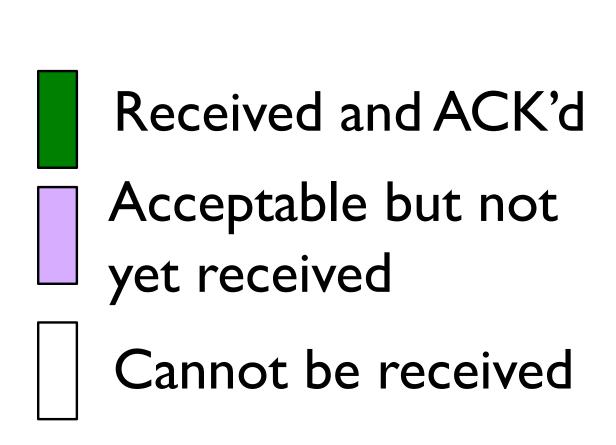
At receiver



• After receiving B+4, B+5



• Receiver sends ACK(B+I)



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- Two common options
 - Cumulative ACKs: ACK carries next in-order sequence number that 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 (GBN)
 - Selective Repeat
- Many variants that differ in implementation details

• Sender transmits up to *n* unacknowledged packets

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 - Discards out-of-order packets (i.e., packets other than B+I)

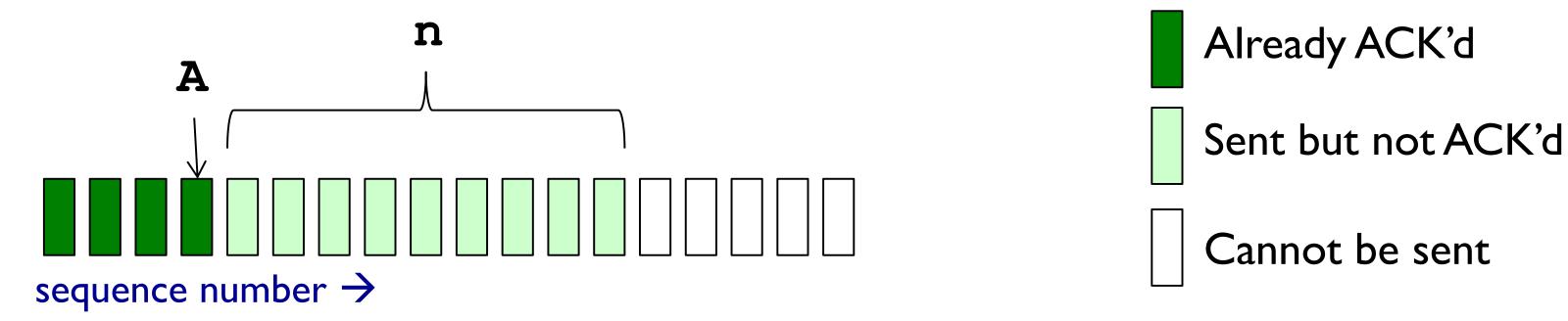
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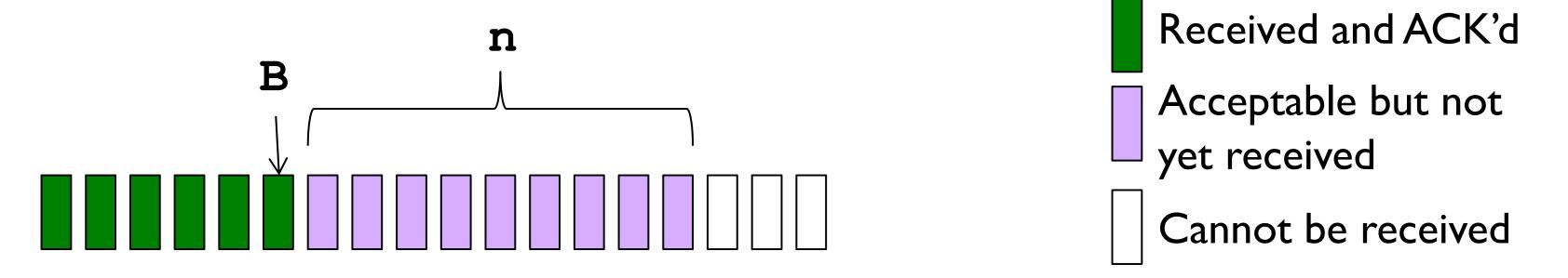
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- Sender sets timer for 1st outstanding ACK(A+1)
- If timeout, retransmit A+I, ..., A+n

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 = $\{B+1, ..., B+n\}$

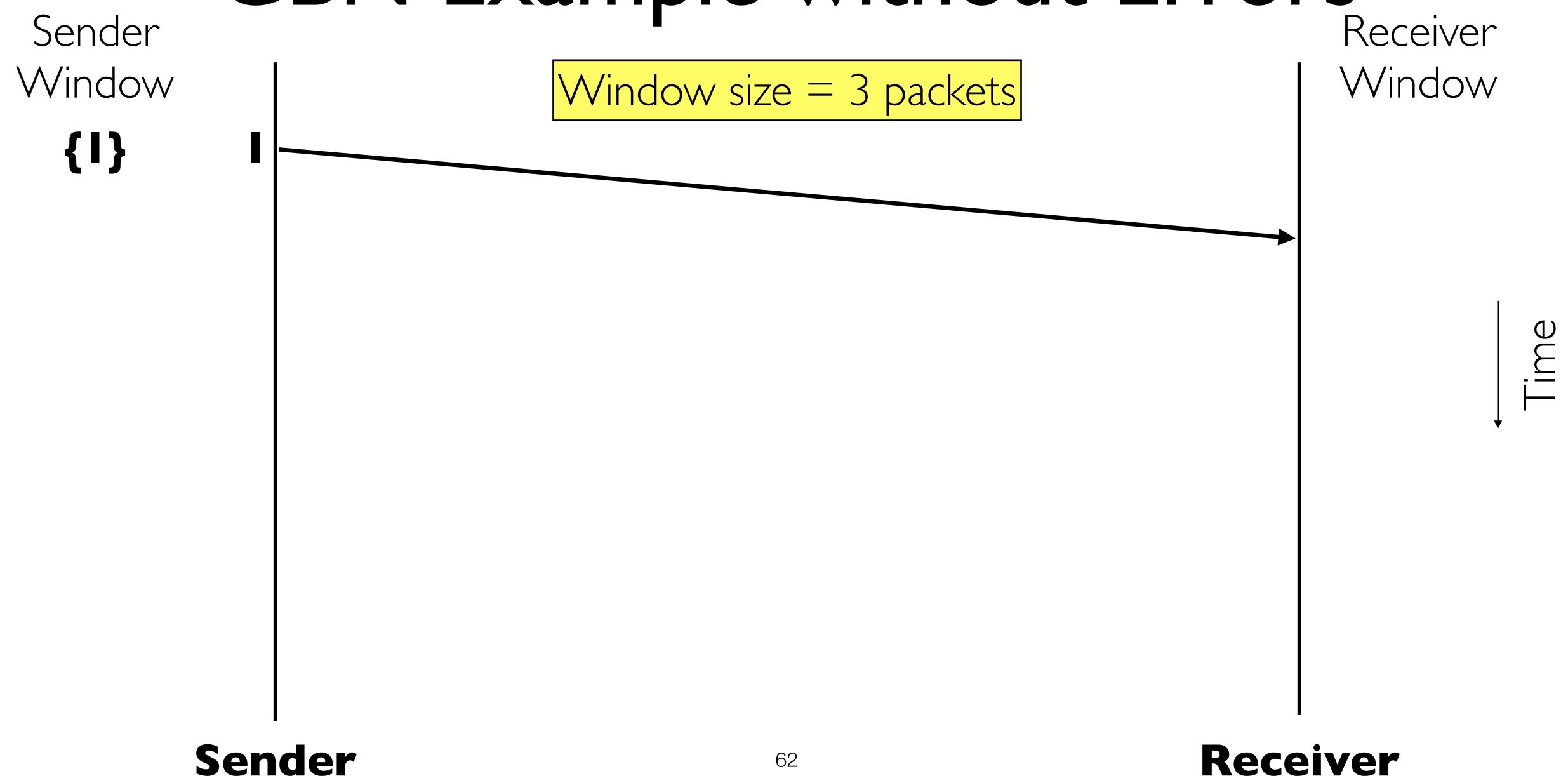


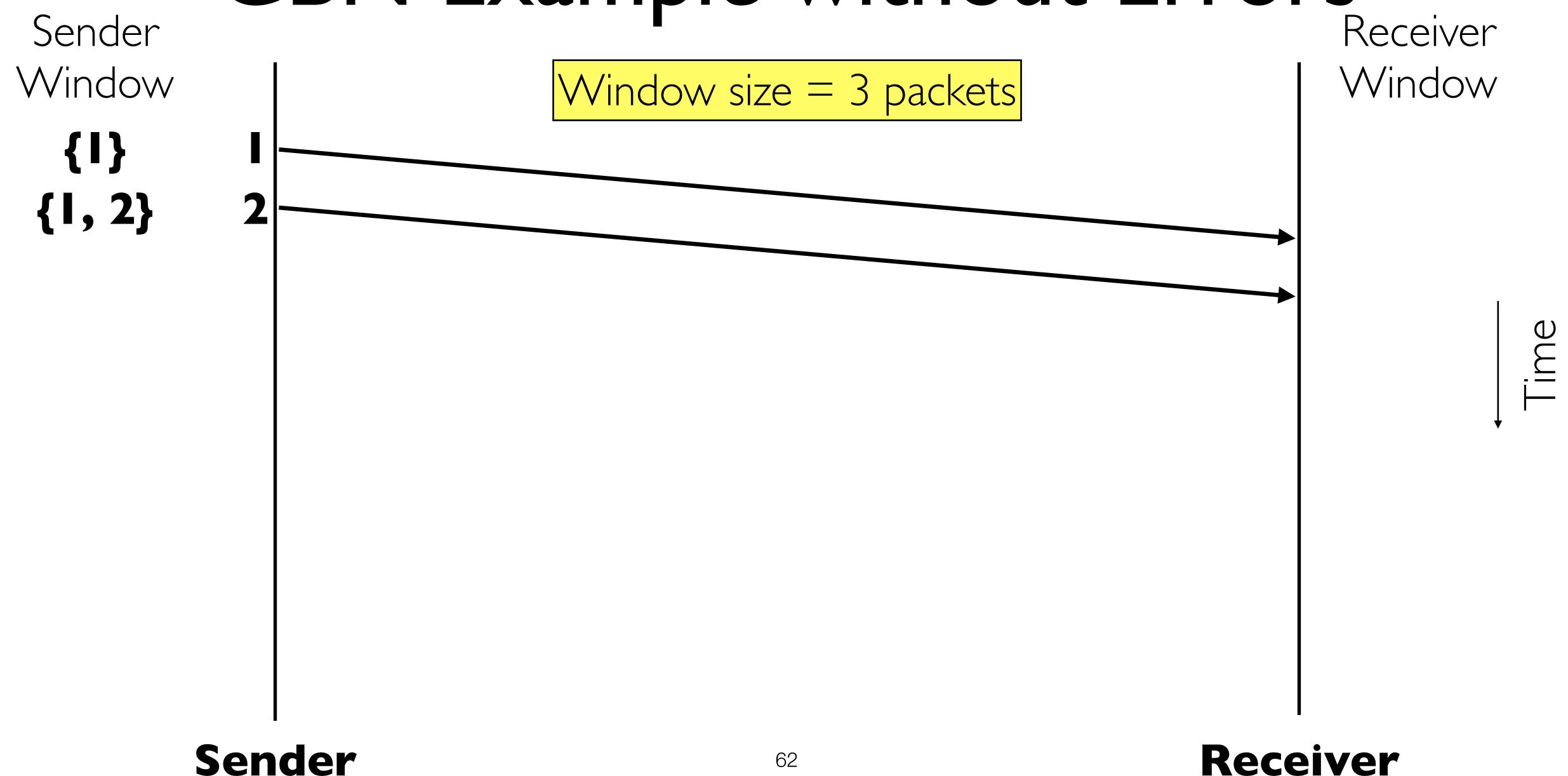
Sender Window

Window size = 3 packets

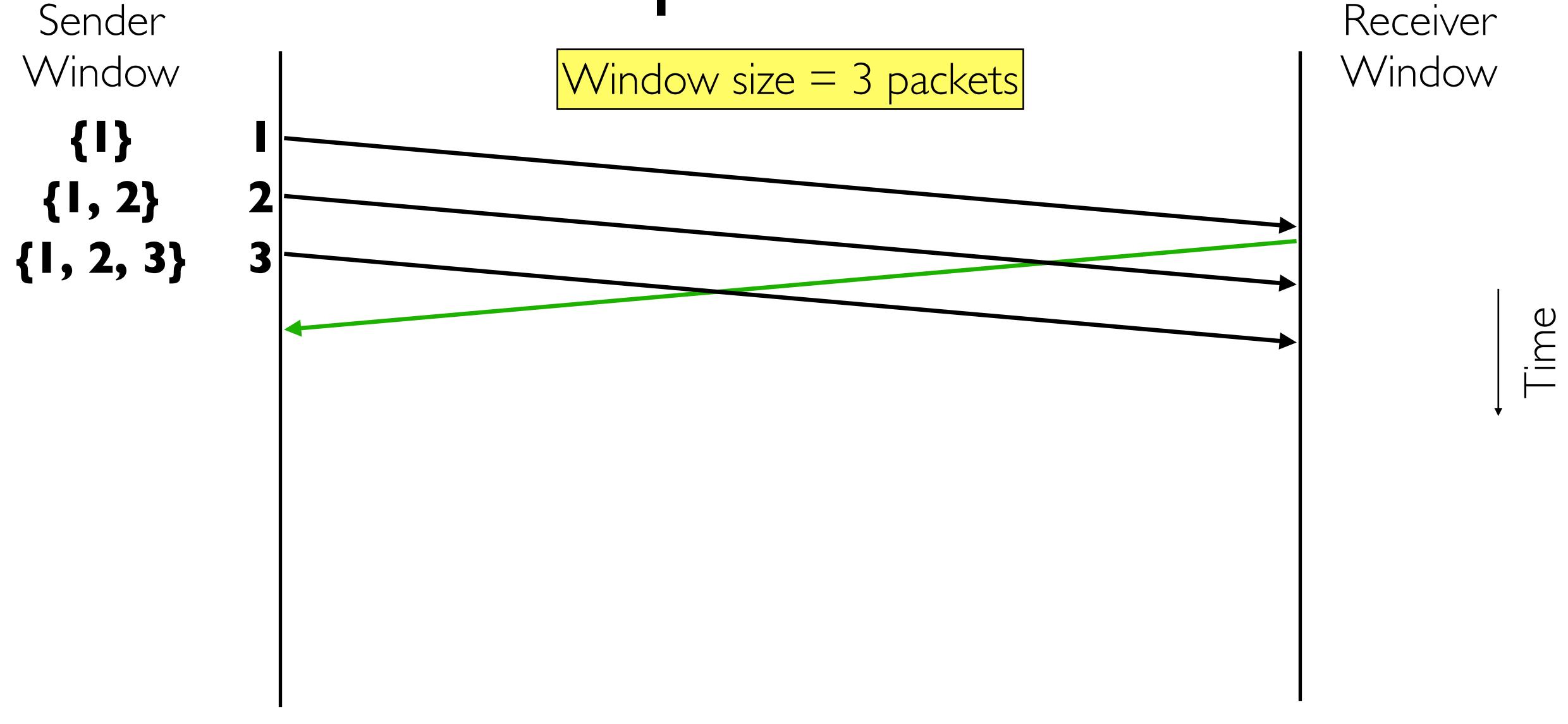
Receiver Window

Time





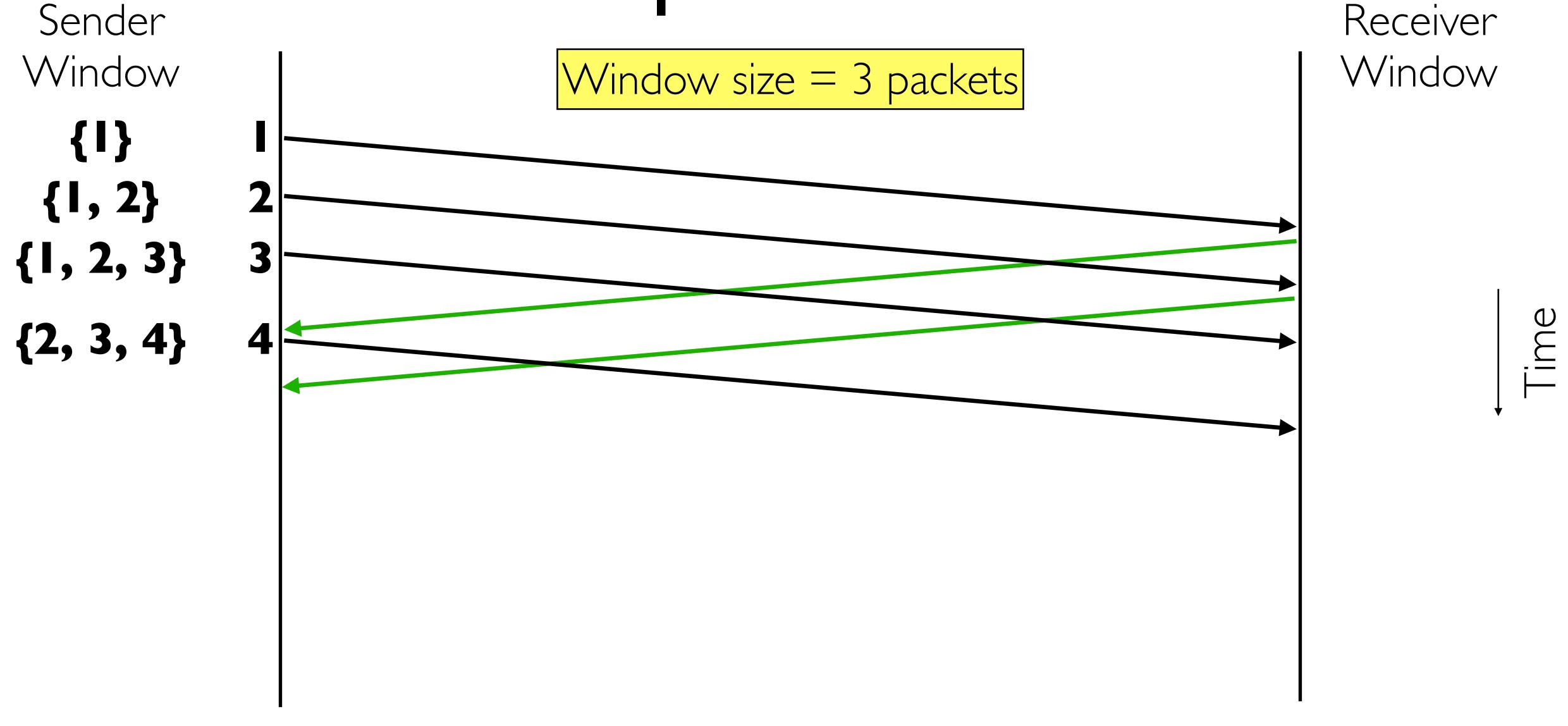
62



62

Receiver

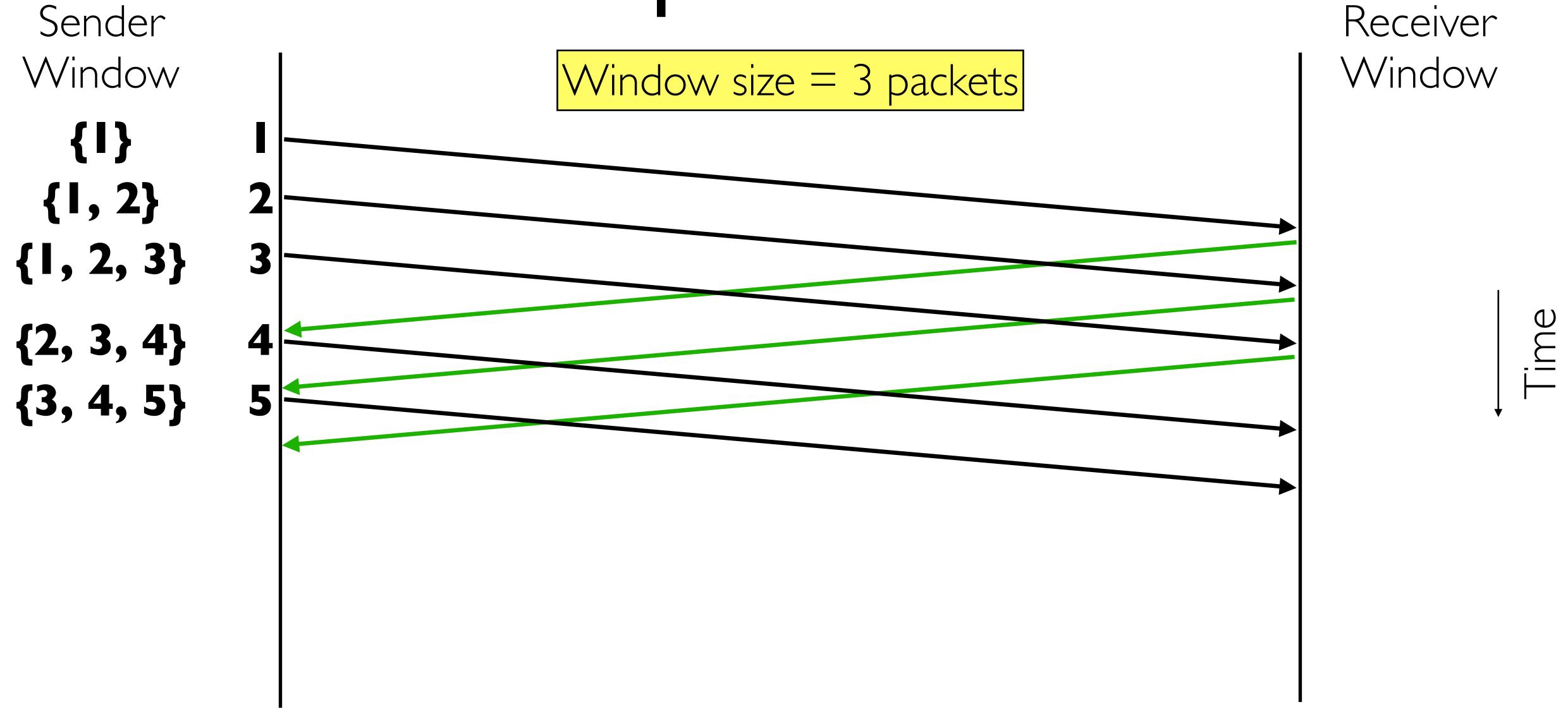
Sender



62

Receiver

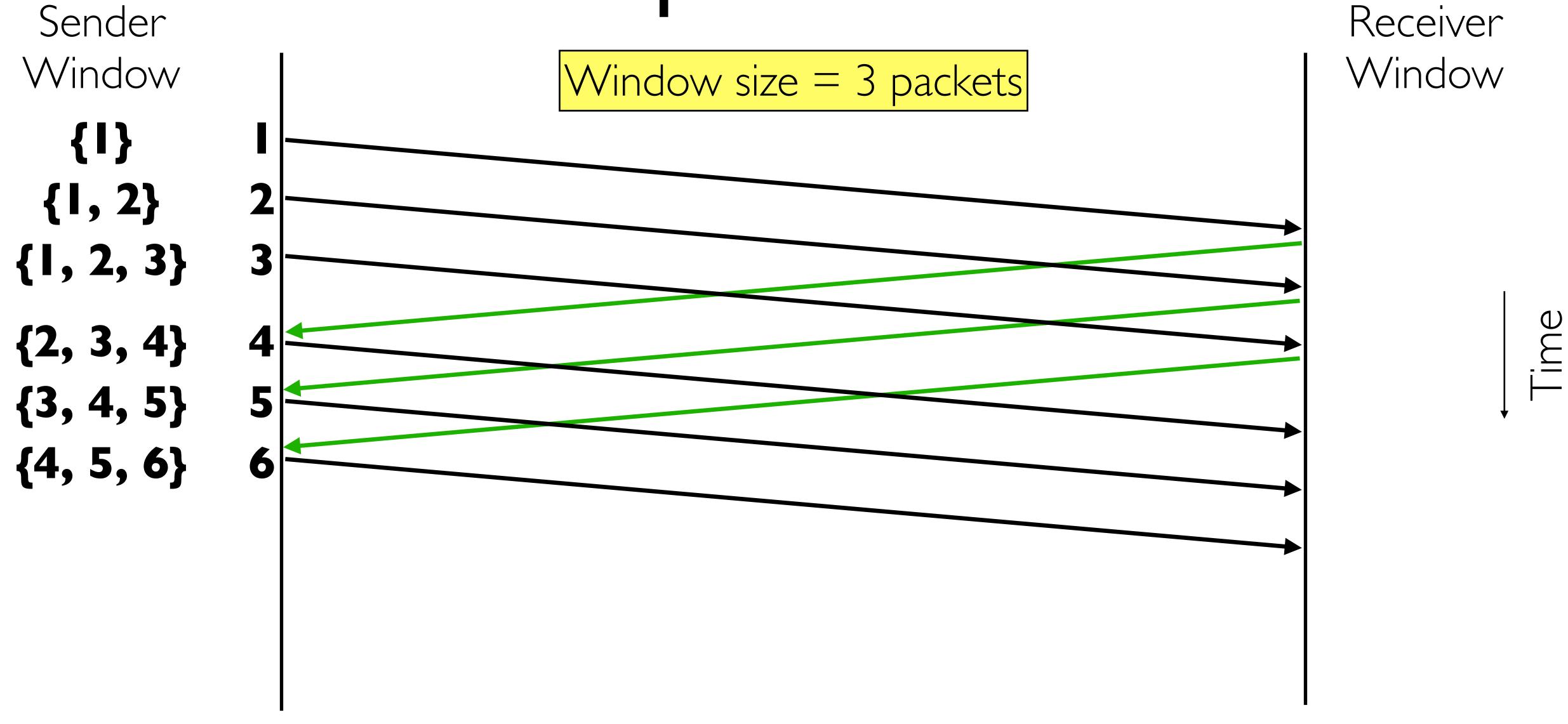
Sender



62

Receiver

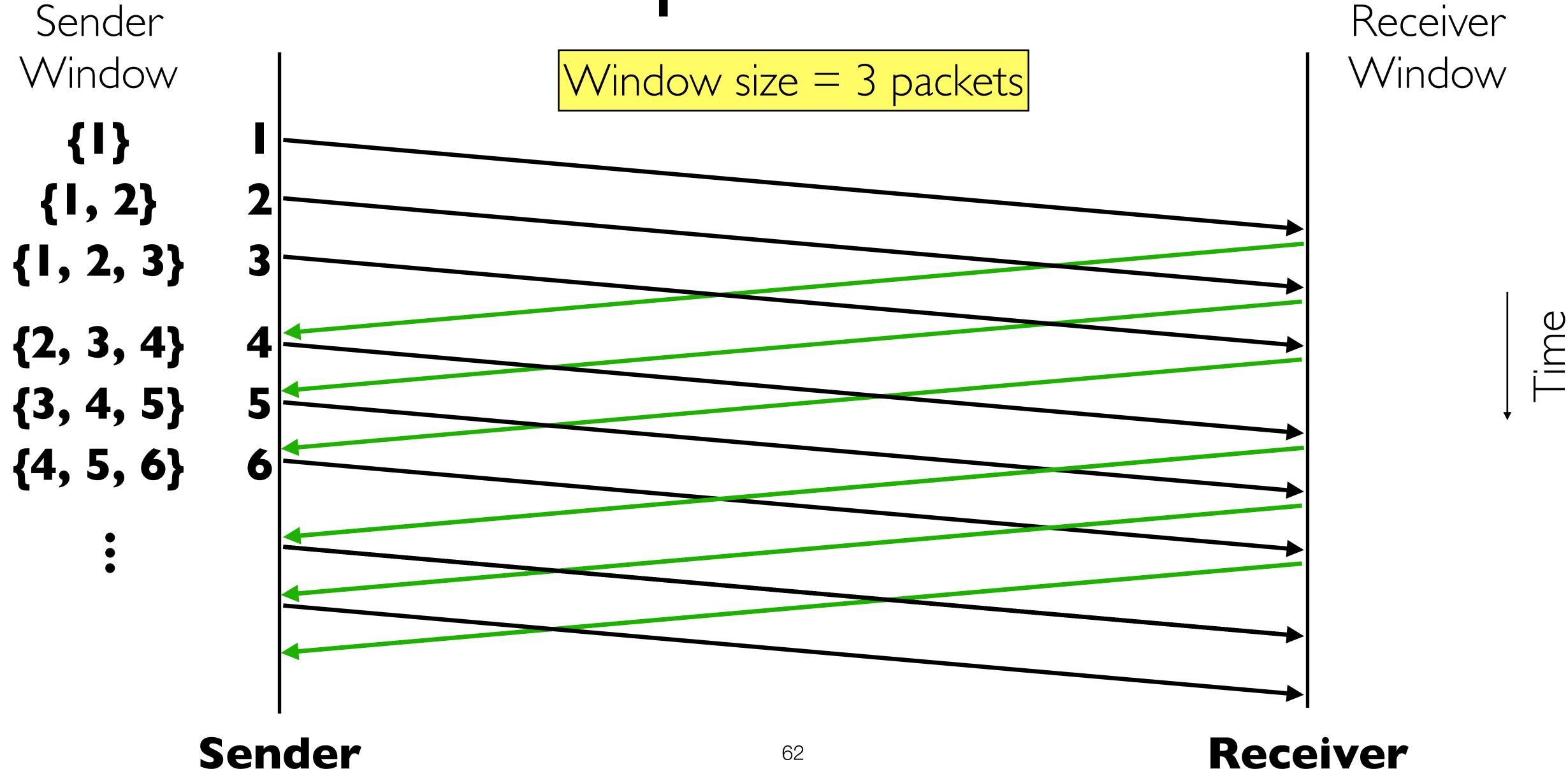
Sender

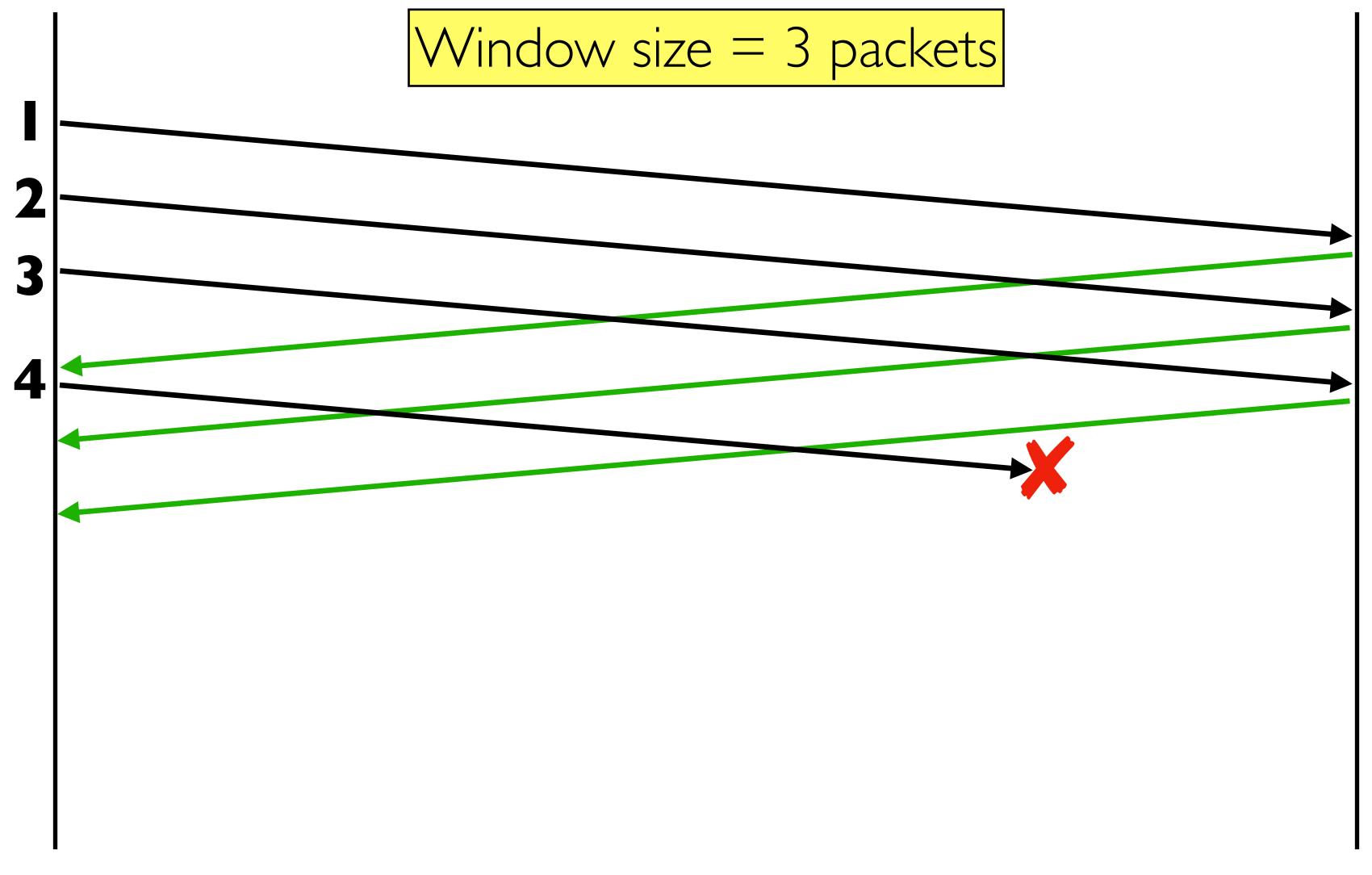


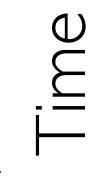
62

Receiver

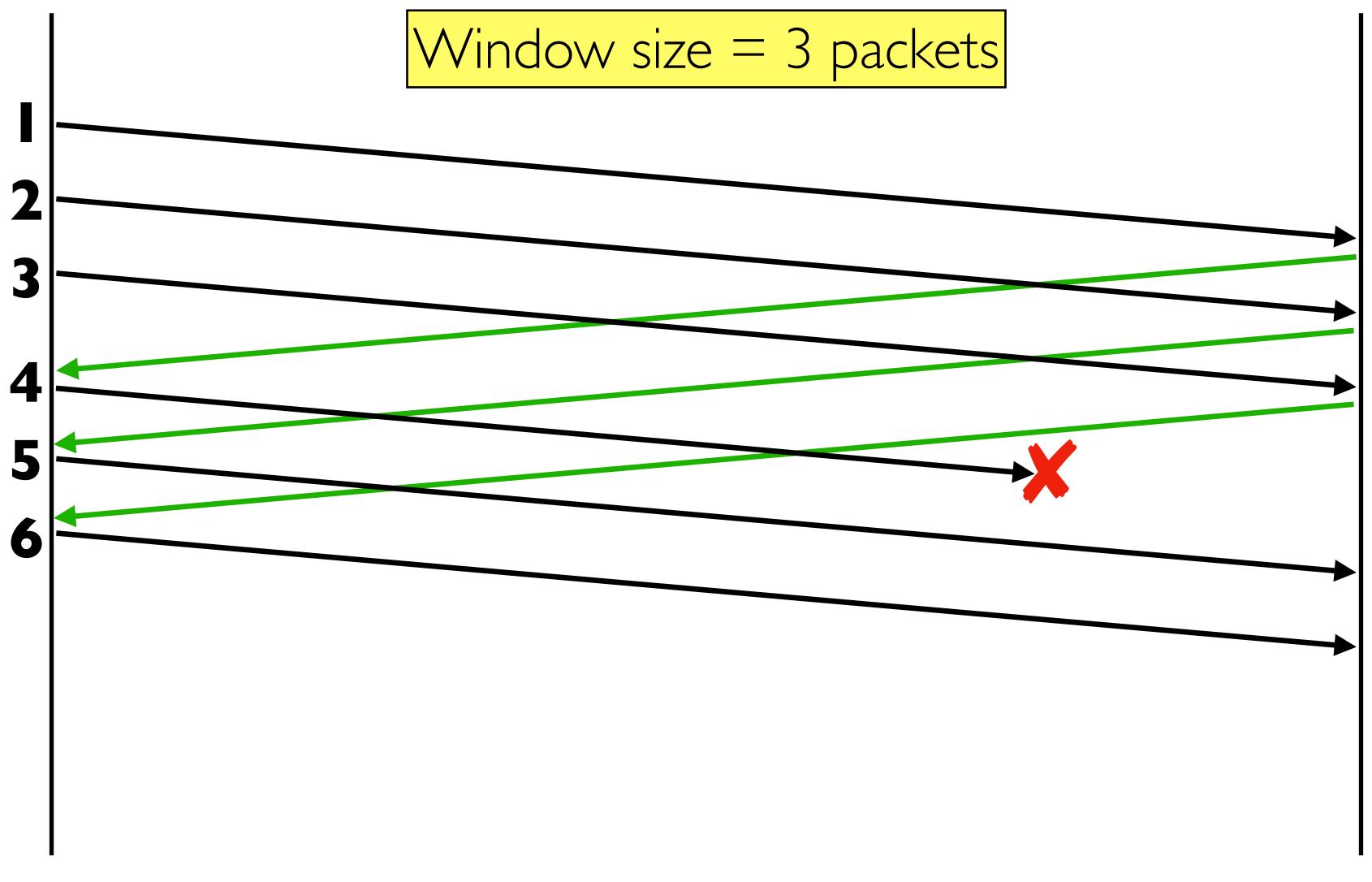
Sender

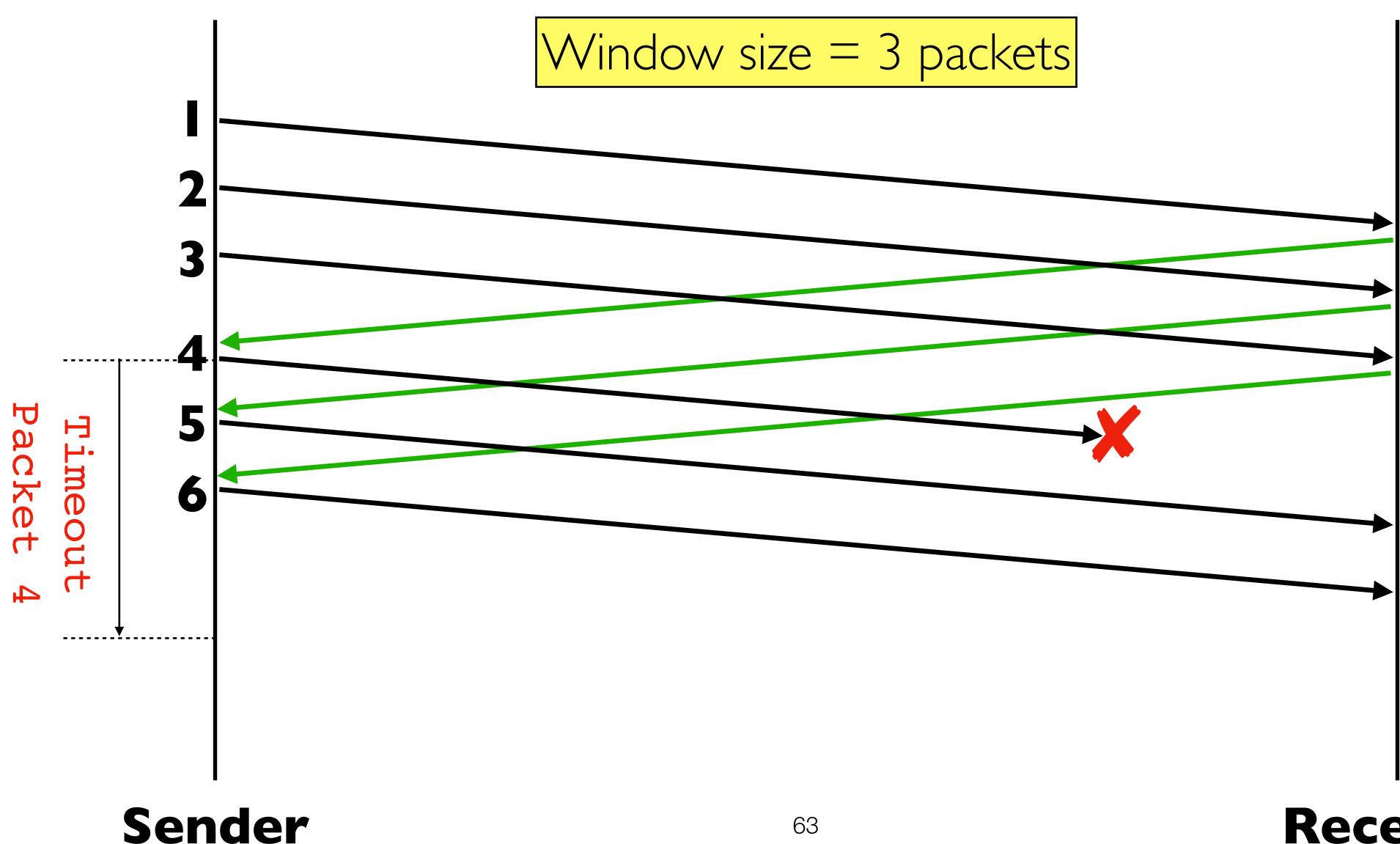


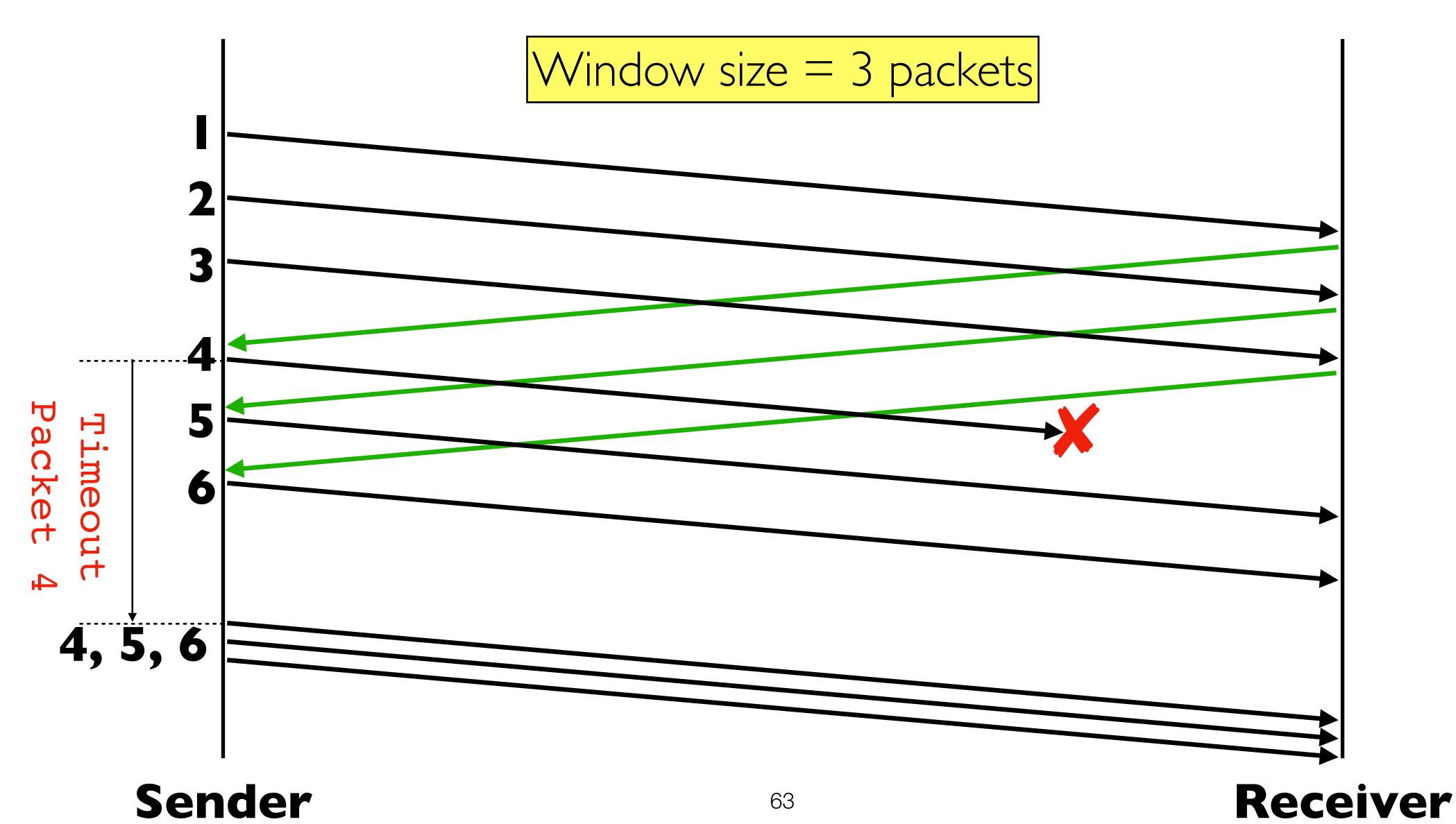


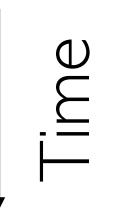


Sender Receiver









• Sender transmits up to *n* unacknowledged packets

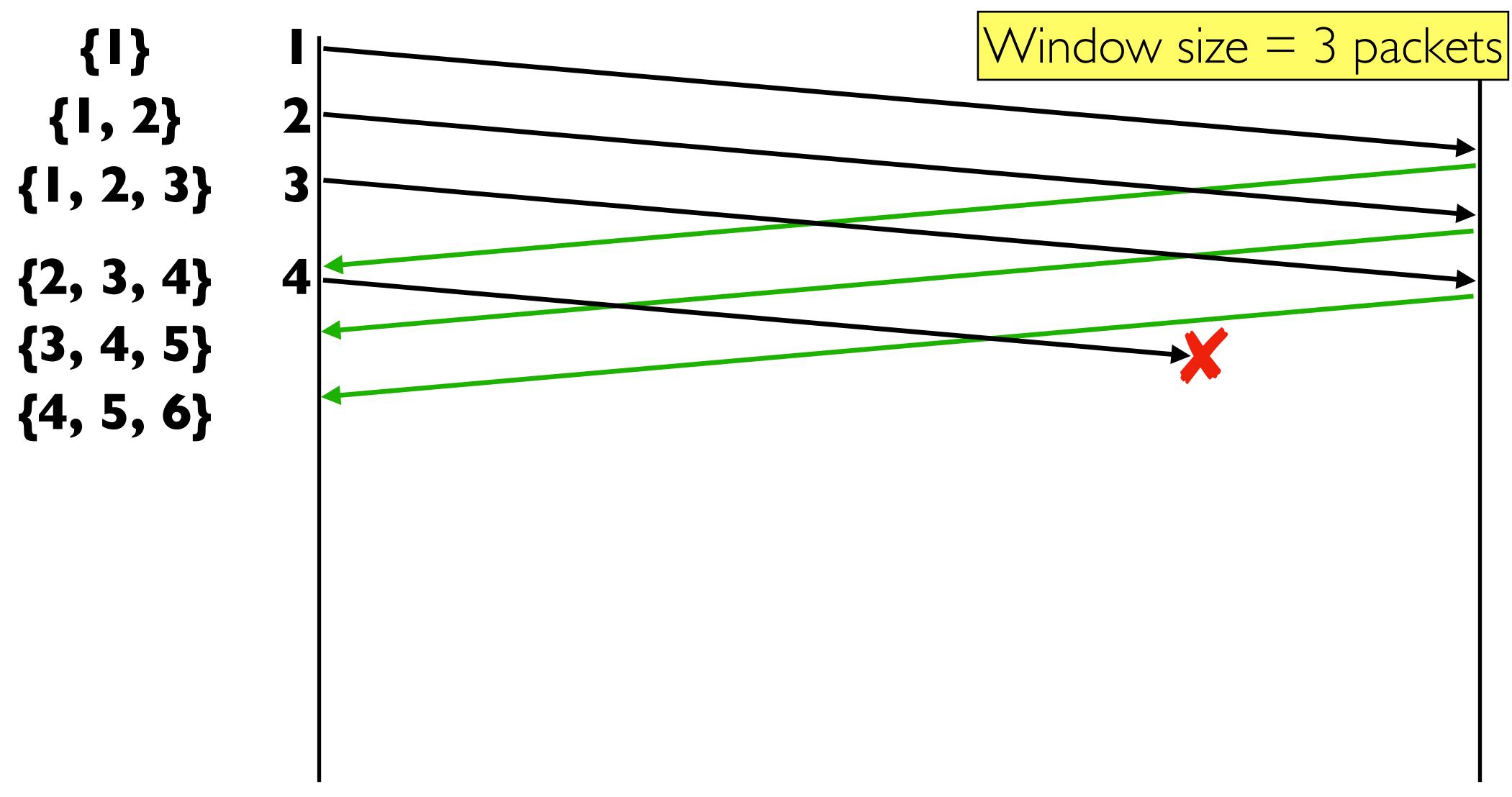
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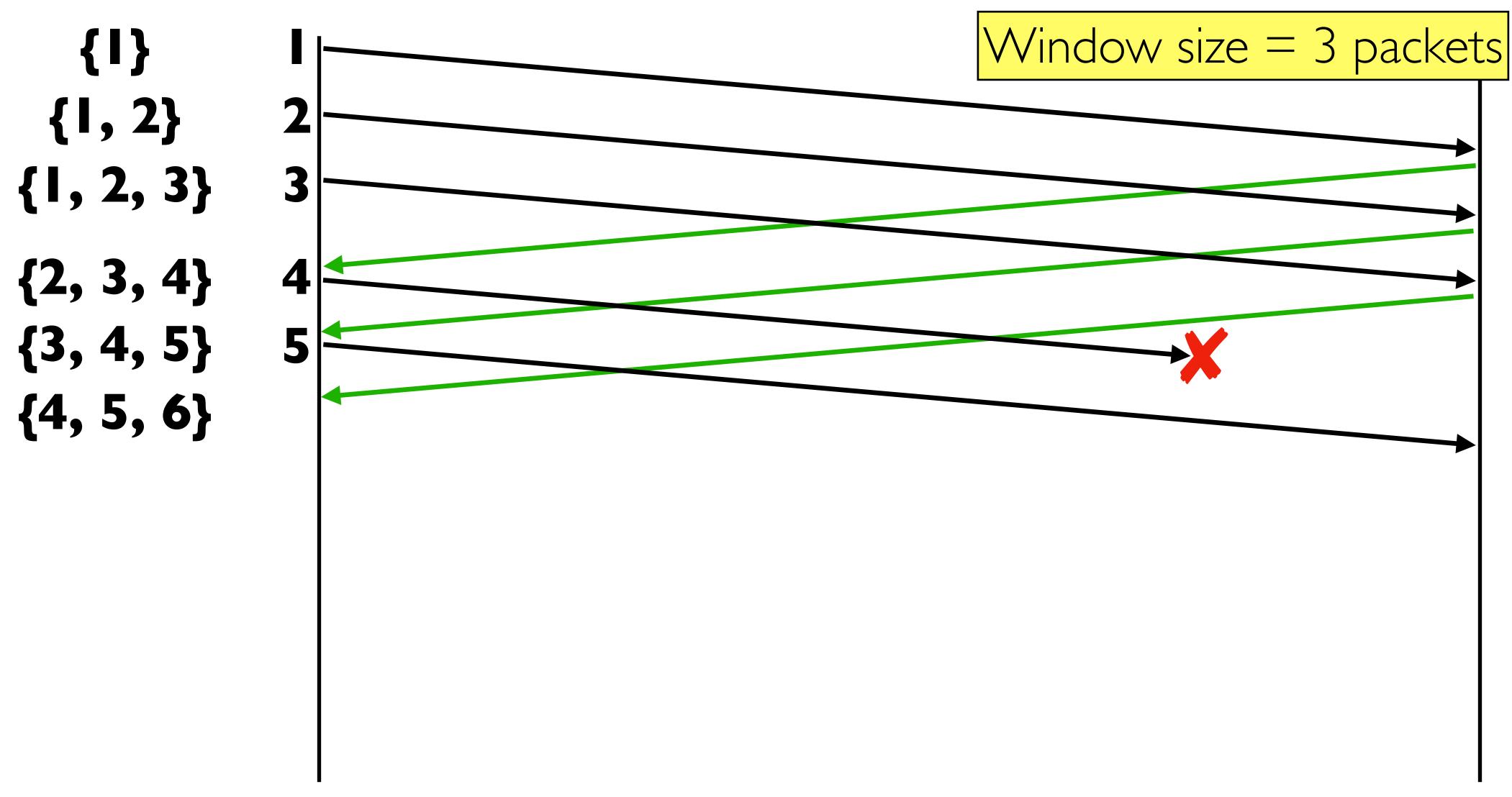
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- Receiver indicates packet k+1 correctly received
- Sender retransmits only packet k on timeout
- Efficient in retransmissions but complex bookkeeping
 - Need a timer per packet!

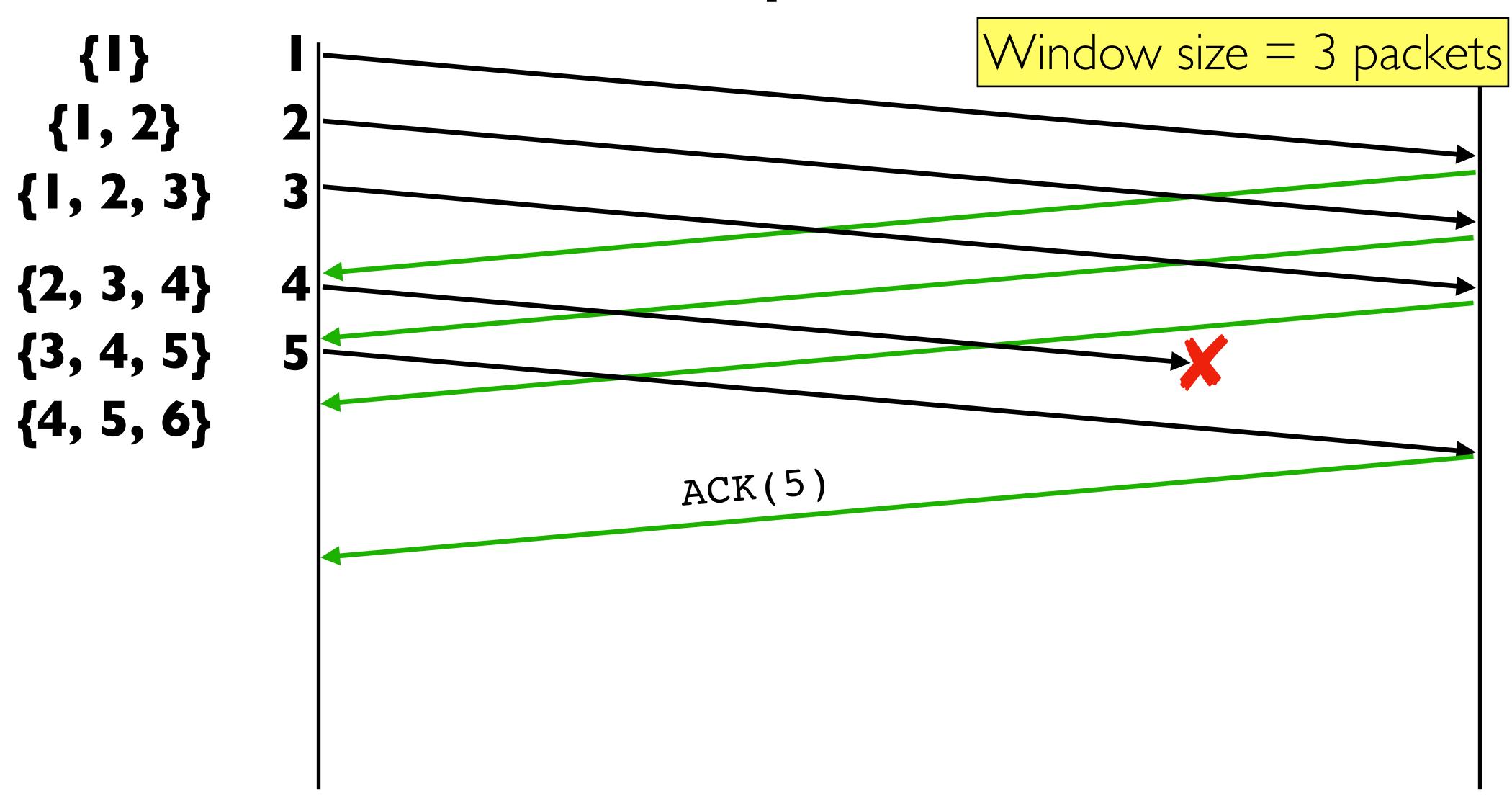
SR Example with Errors



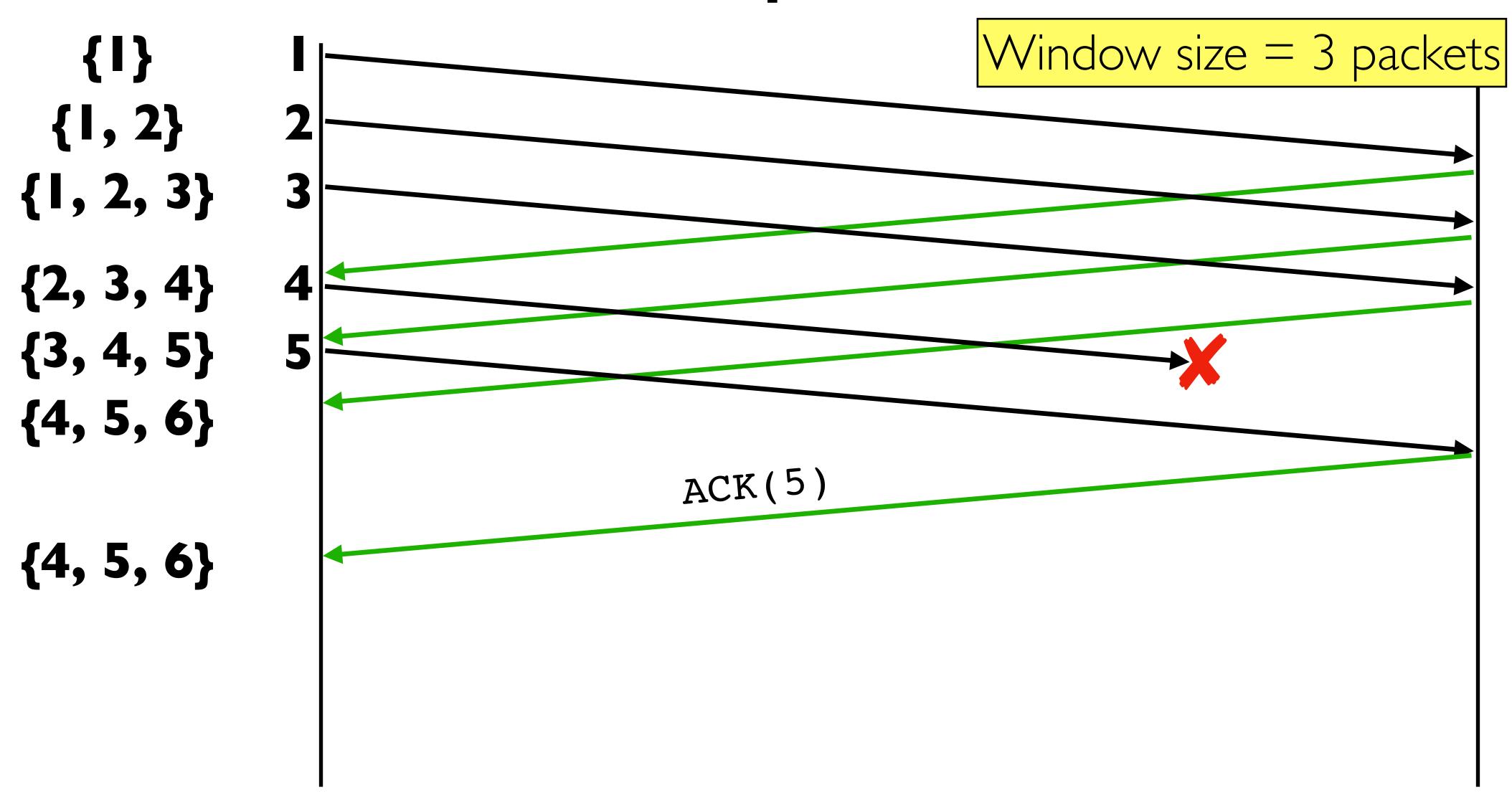
SR Example with Errors



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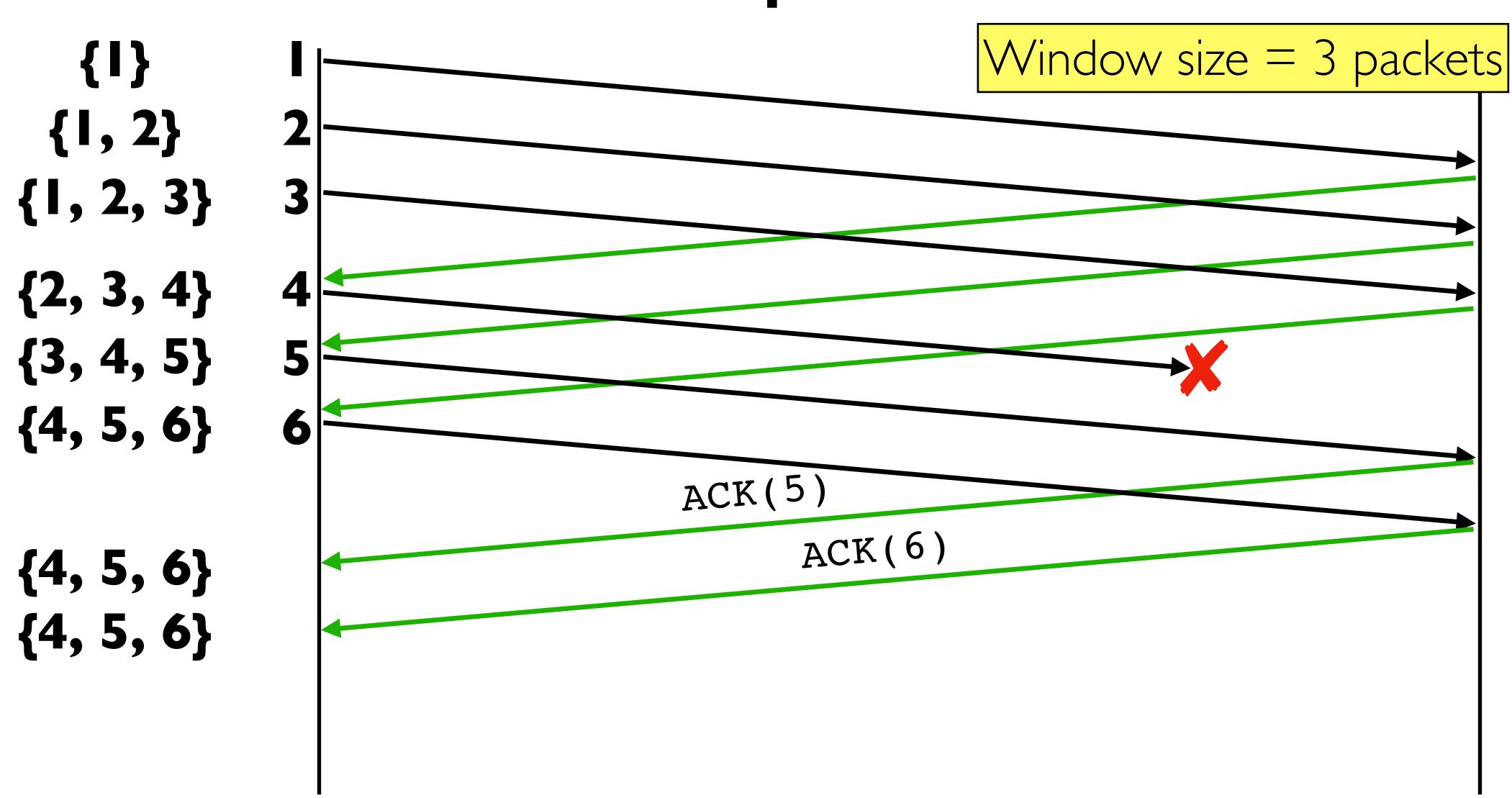
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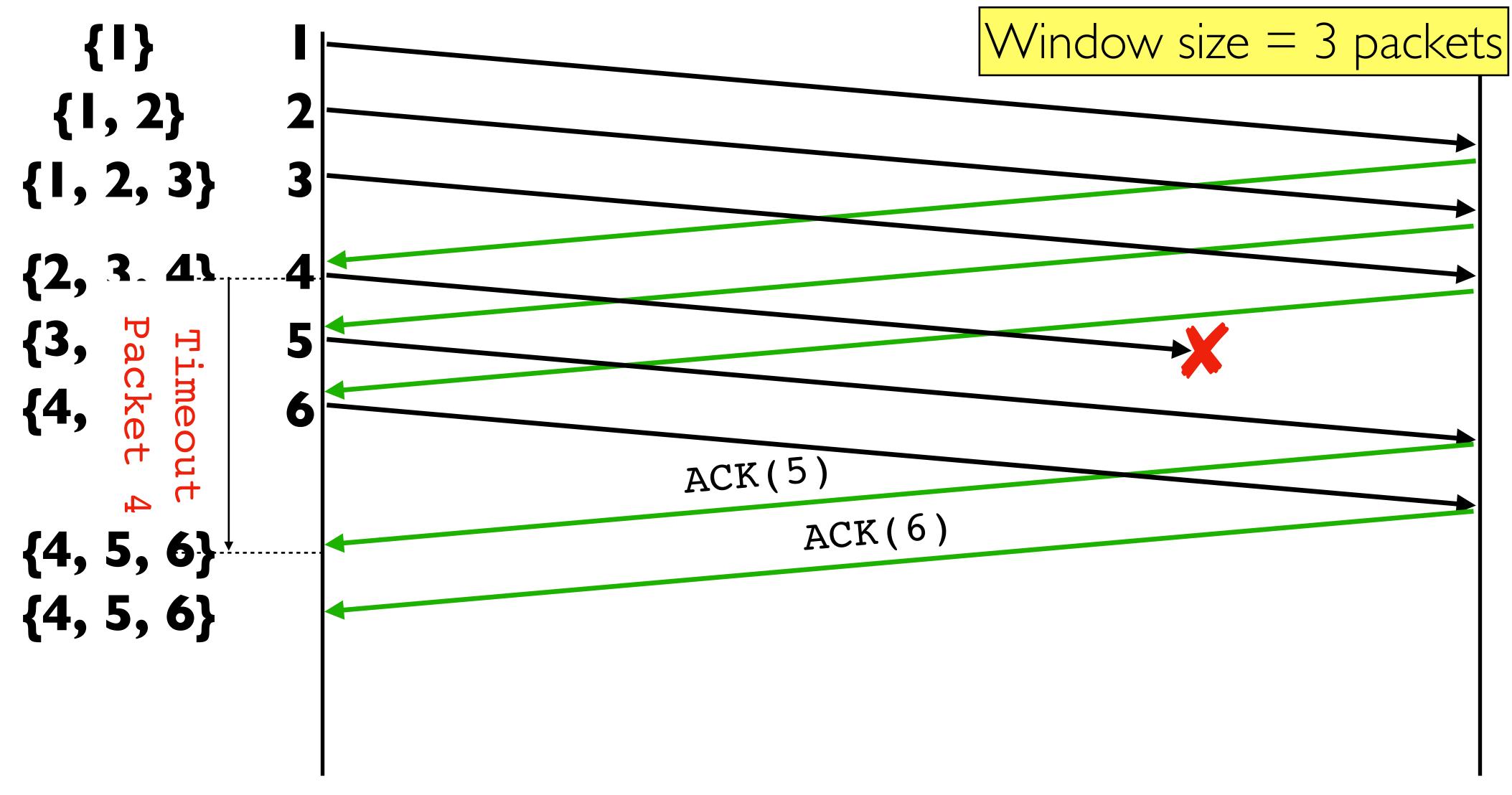
Time

Sender ⁶⁵ Receiver

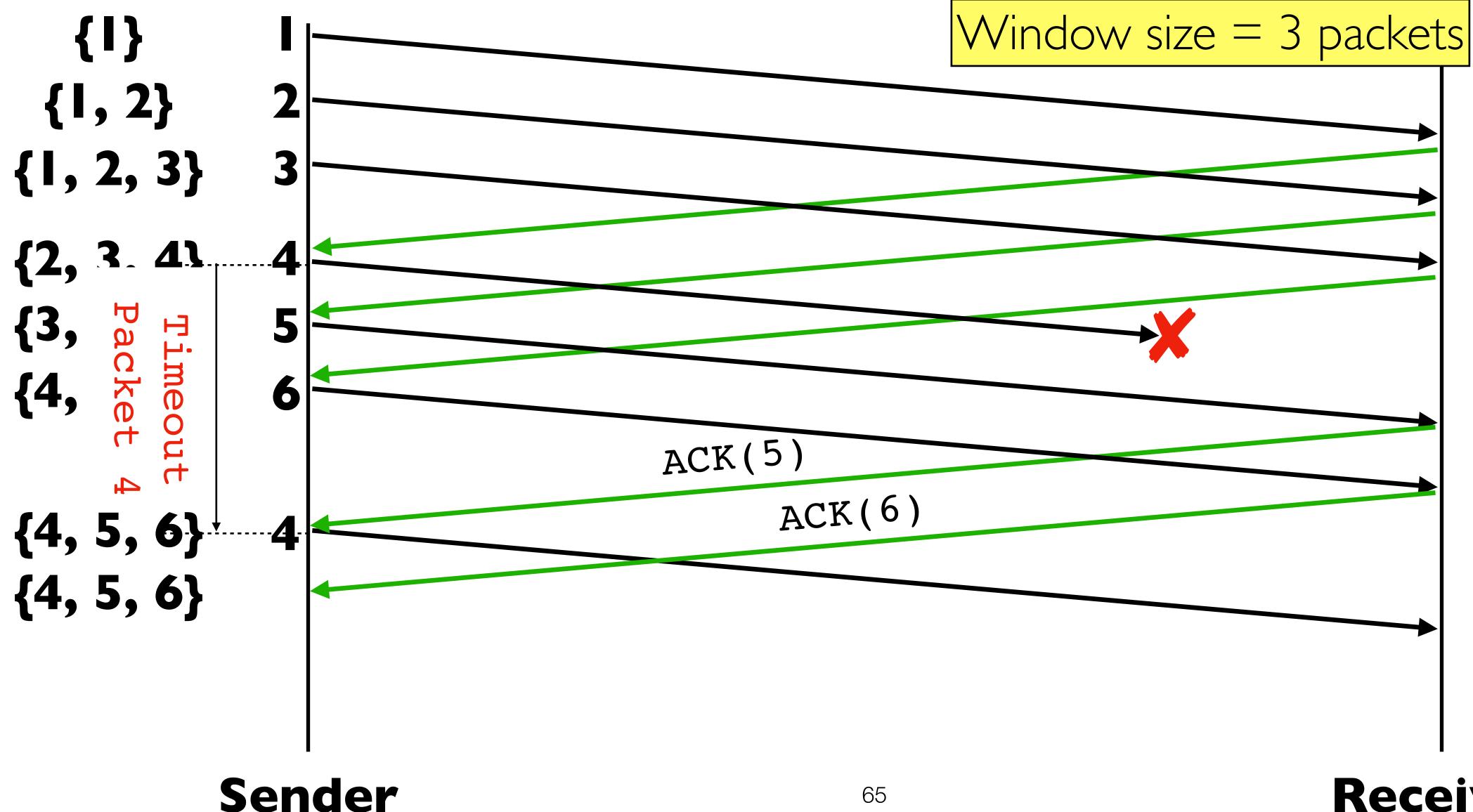
SR Example with Errors



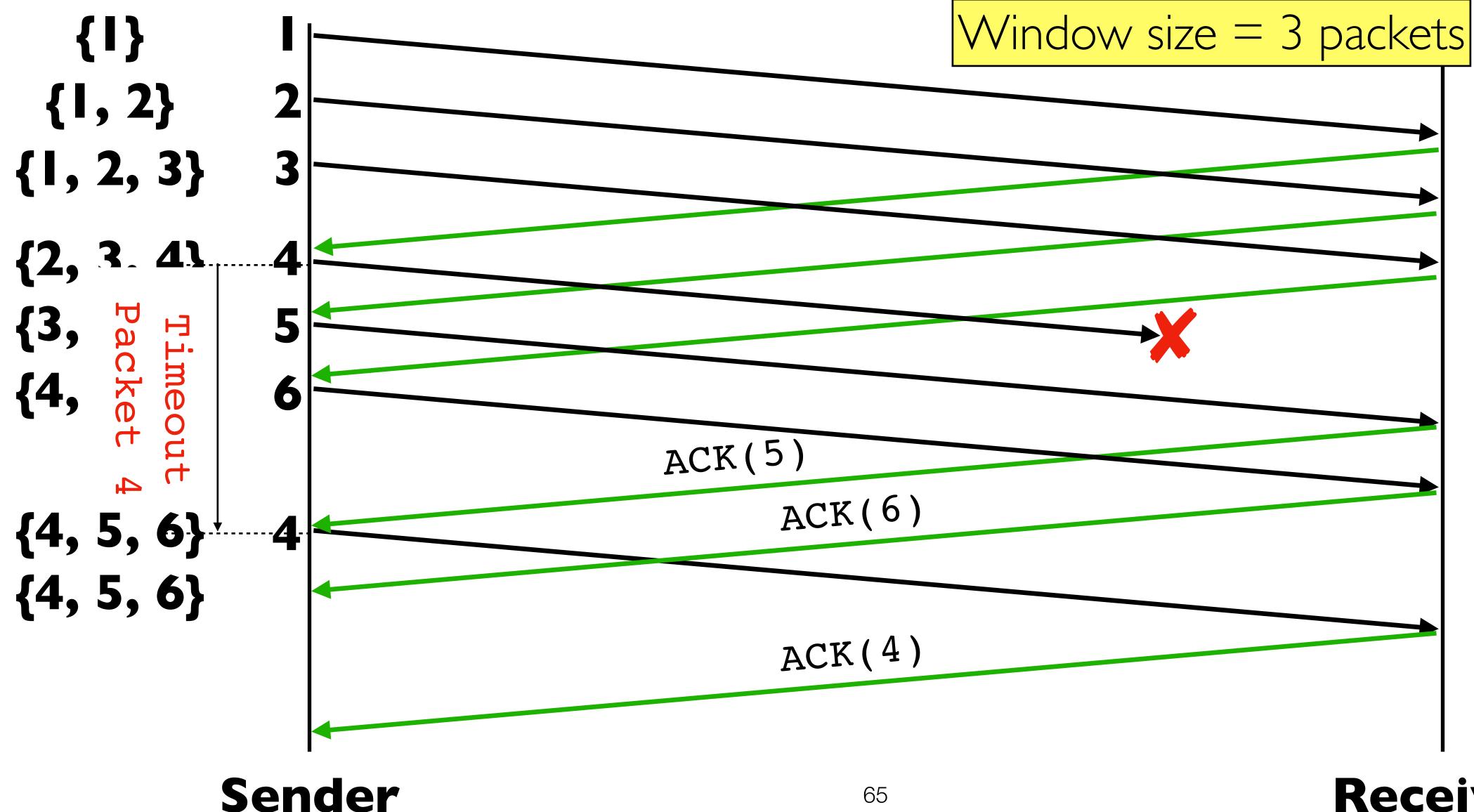
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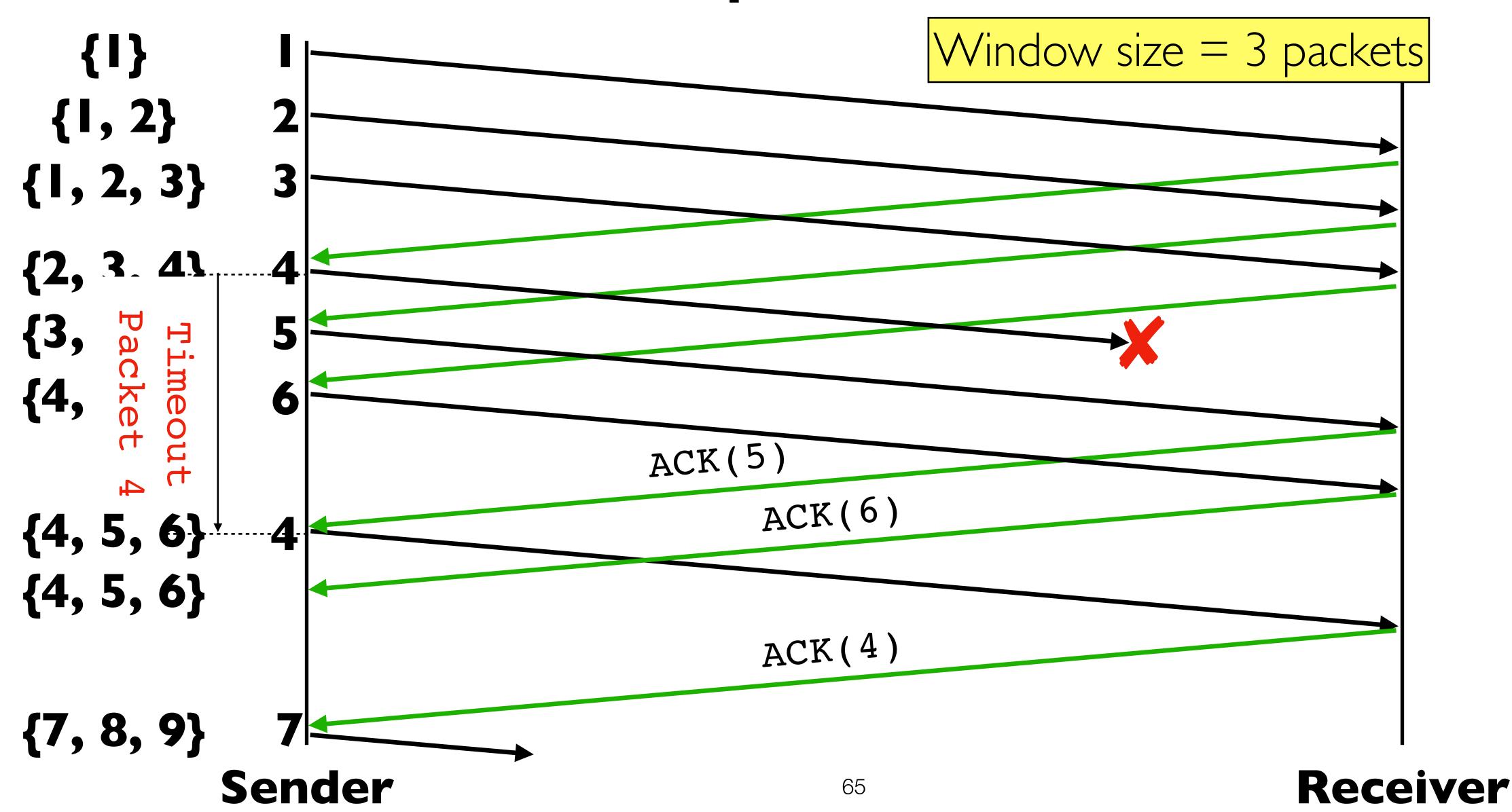


SR Example with Errors



Receiver

SR Example with Errors



GBN vs Selective Repeat

- When would GBN be better?
- When would SR be better?

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- Sender has to buffer all unacknowledged packet, 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
- ACKs
 - Cumulative
 - Selective
- Sequence numbers: tracking duplicates, windows
- Sliding Windows: for efficiency

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- Checksums: for error detection
- Timers: for loss detection
- ACKs
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- Sequence numbers: tracking duplicates, windows
- Sliding Windows: for efficiency
- Reliability protocols use the above to decide when and what to retransmit or acknowledge.

- Most of our previous tricks + a few differences
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- Introduces timeout estimation algorithms (next time)

Next Time

• TCP

- Reliability
- Congestion control