Wrapping up the IP header & Reliability Concepts

Spring 2024
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Designing IP: two remaining topics

- IPv4 \rightarrow IPv6
- Security implications of the IP header

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 - Addresses four times as big
- Took the opportunity to do some "spring cleaning"
 - Got rid of all fields that were not absolutely necessary
- Result is an elegant, if unambitious, protocol

What "clean up" would you do?

4-bit Header Length	8-bit Type of Service	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					

Expanded addresses

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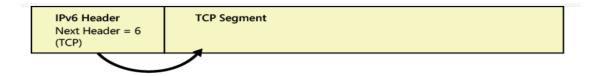
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- New options mechanism → "next header"

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 - Example: "follow this route" (source routing)

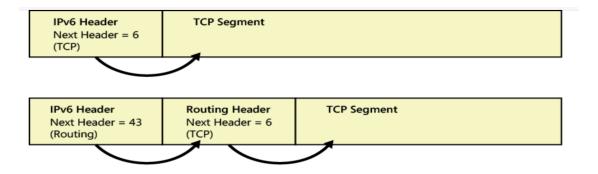
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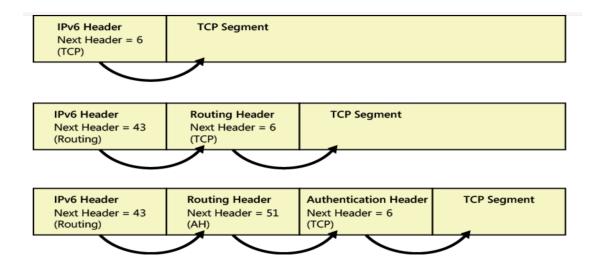
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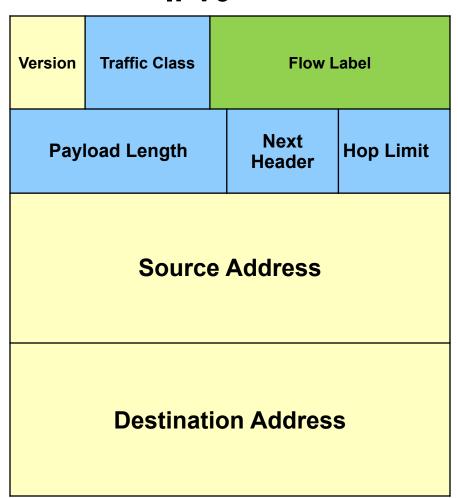
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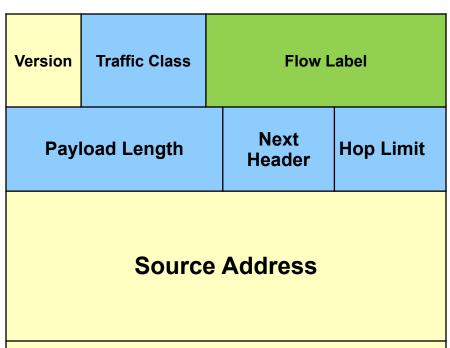
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- Eliminated header length
- Added Flow Label
 - Explicit mechanism to denote related streams of packets



IPv4

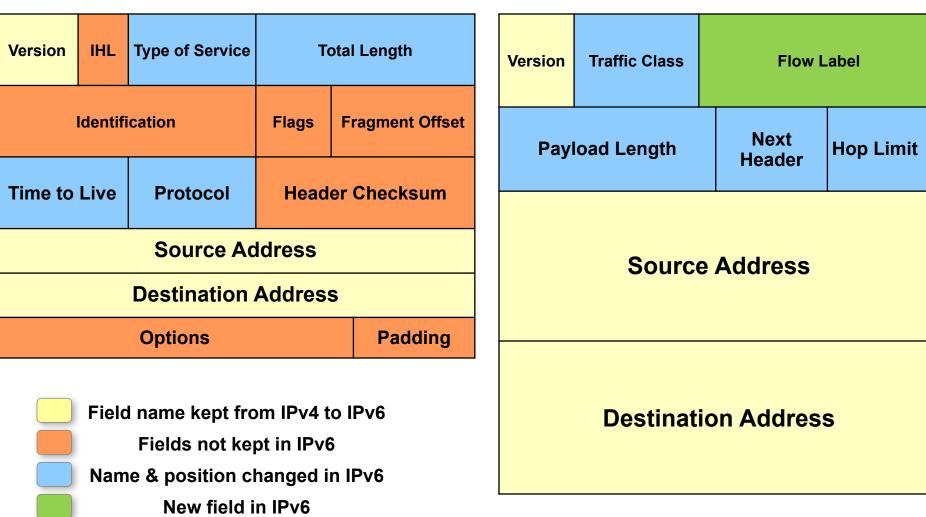
Version	IHL	Type of Service	Total Length		
Identification		Flags	Fragment Offset		
Time to	Live	Protocol	Header Checksum		
Source Address					
Destination Address					
Options				Padding	

IPv6



Destination Address

IPv4 IPv6



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 - general next-header approach
 - general flow label for packet

Quick Security Analysis of IP Header

Focus on Sender Attacks

- Vulnerabilities a sender can exploit
- Note: not a comprehensive view of potential attacks!
 - For example, we'll ignore attackers other than the sender

IP Packet Structure

4-bit Header Length		8-bit Type of Service	16-bit Total Length (Bytes)	
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Options (if any)				
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IP Address Integrity

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- Source address should be the sending host
 - But who's checking?
 - You could send packets with any source you want

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- Attack the destination
 - Send excessive packets, overload network path to destination
 - But: victim can identify/filter you by the source address
 - Hence, evade detection by putting different source addresses in the packets you send ("spoofing")

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- Or: as a way to bother the spoofed host
 - Spoofed host is wrongly blamed
 - Spoofed host may receive return traffic from the receiver(s)

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- What if the network charges for TOS traffic ...
 - ... and attacker spoofs the victim's source address?
- Today, mostly set/used by operators, not end-hosts

Security Implications of Fragmentation?

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- Send packets larger than MTU

 make routers do extra work
 - Can lead to resource exhaustion

More Security Implications

• IP options

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 - Processing IP options often processed in router's control plane
 (i.e., slow path) → attacker can try to overload routers

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- IP options
 - Processing IP options often processed in router's control plane (i.e., slow path) → attacker can try to overload routers
- Routers often ignore options / drop packets with options

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Allows discovery of topology (a la traceroute)

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- Allows discovery of topology (a la traceroute)
- Some routers do not respond with a TTL exceeded error message

Other Security Implications?

- No apparent problems with protocol field (8 bits)
 - It's just a de-muxing handle
 - If set incorrectly, next layer will find packet ill-formed
- Bad IP checksum field (16 bits) will cause packet to be discarded by the network
 - Not an effective attack...

Recap: IP header design

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More nuanced than it first seems!

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More nuanced than it first seems!

- Must juggle multiple goals
 - Efficient implementation
 - Security
 - Future needs

Questions?

Next topic: Reliable Transport

- Today: focus on concepts and mechanisms
- Next lecture: the design of TCP

Reliable Delivery Is Necessary

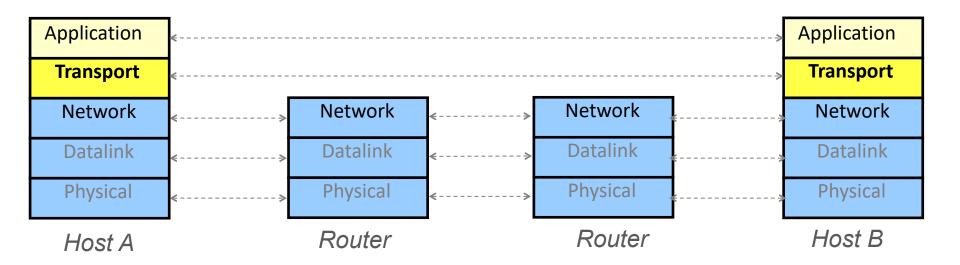
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 - E.g., file transfer

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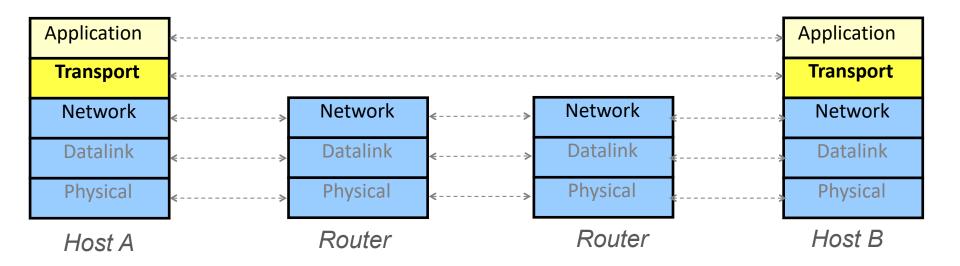
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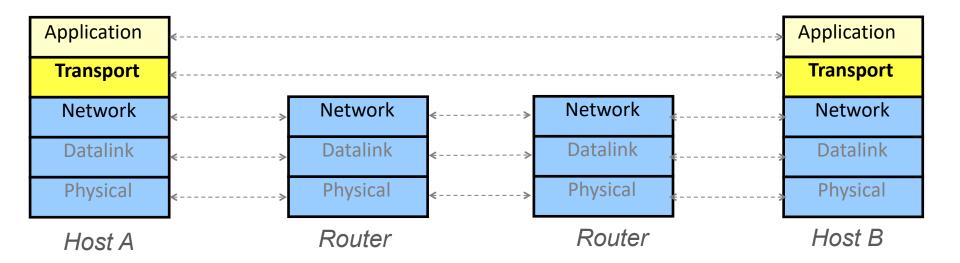
- Many app semantics involve reliable delivery
 - E.g., file transfer
- Challenge: building a reliable service on top of unreliable packet delivery
- Bridging the gap between
 - the abstractions application designers want
 - the abstractions networks can easily support



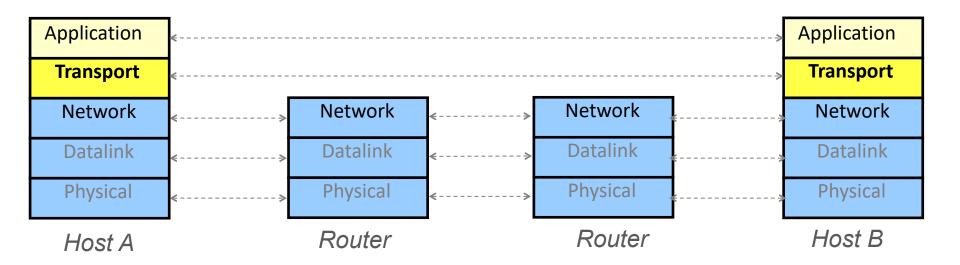
• At network layer: best-effort delivery



- At network layer: best-effort delivery
- At transport layer: at-least-once delivery



- At network layer: best-effort delivery
- At transport layer: *at-least-once* delivery
- At the app layer: exactly-once delivery



Goals For Reliable Transfer

(at the Transport Layer)

Goals For Reliable Transfer (at the Transport Layer)

Correctness

 The destination receives every packet, uncorrupted, at least once

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Minimize time until data is transferred

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Efficiency

- Would like to minimize use of bandwidth
- i.e., avoid sending packets unnecessarily

Note!

- A reliability protocol (at the transport layer) can "give up", but must announce this to application
 - E.g., if the network is partitioned
- But it can never falsely claim to have delivered a packet

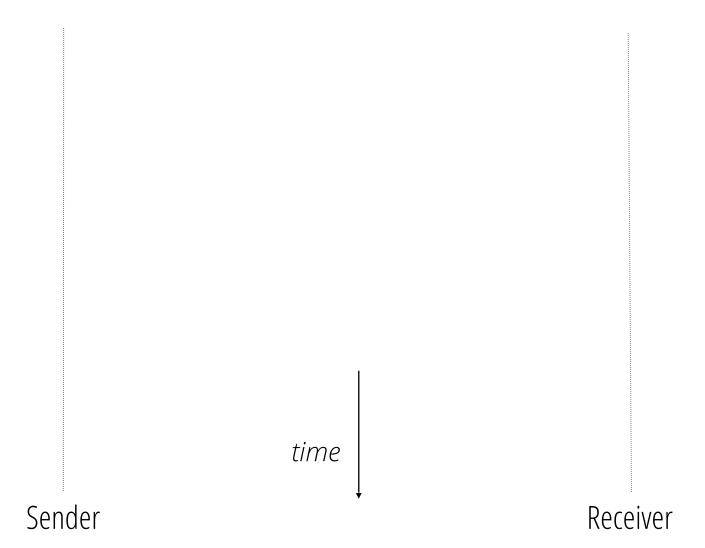
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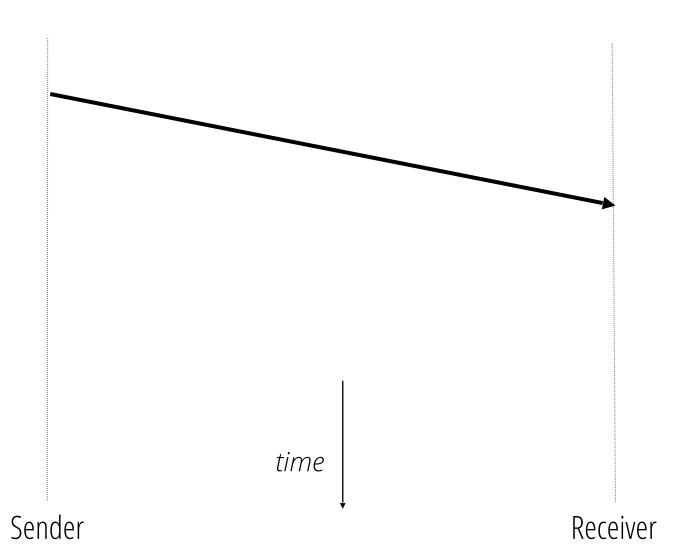
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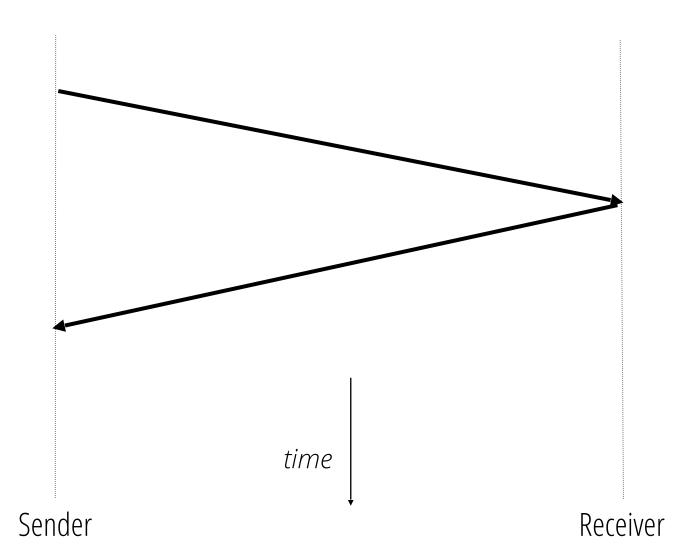
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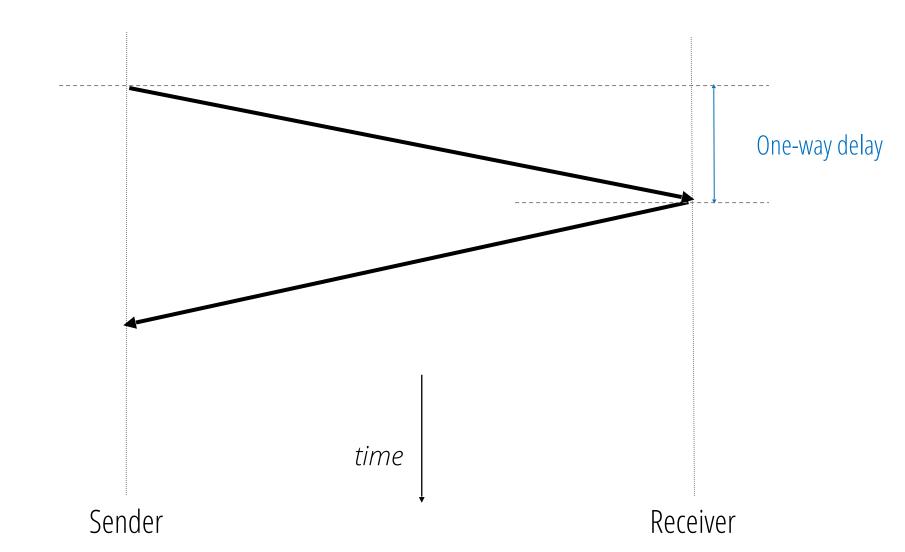
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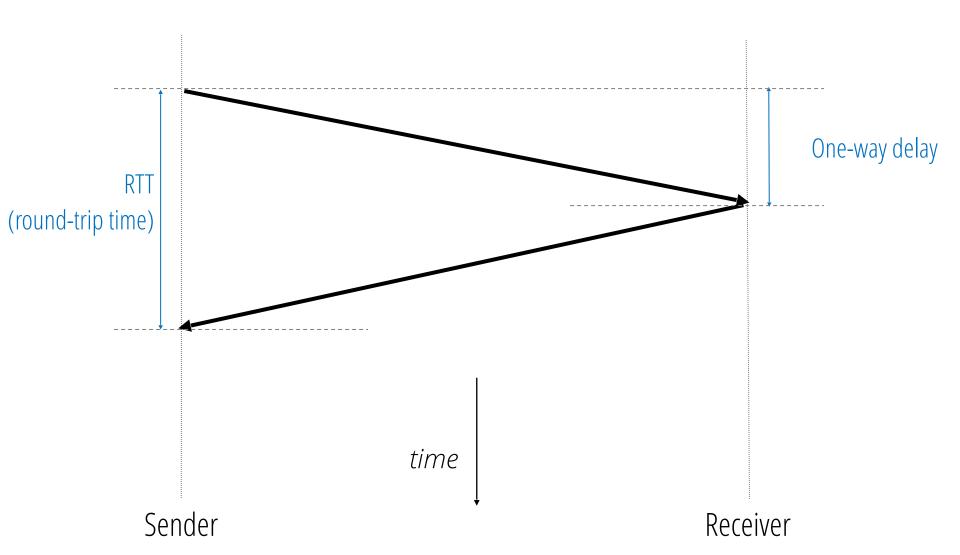
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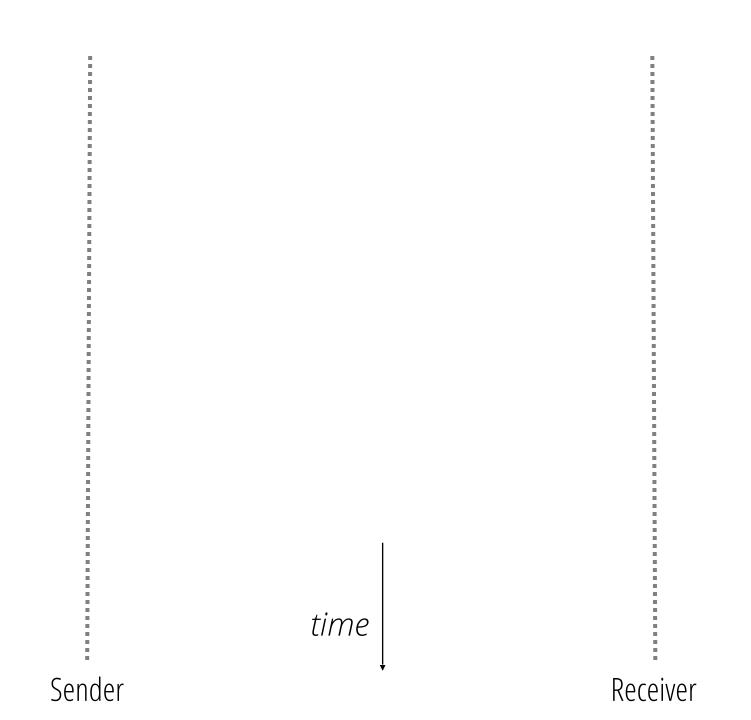
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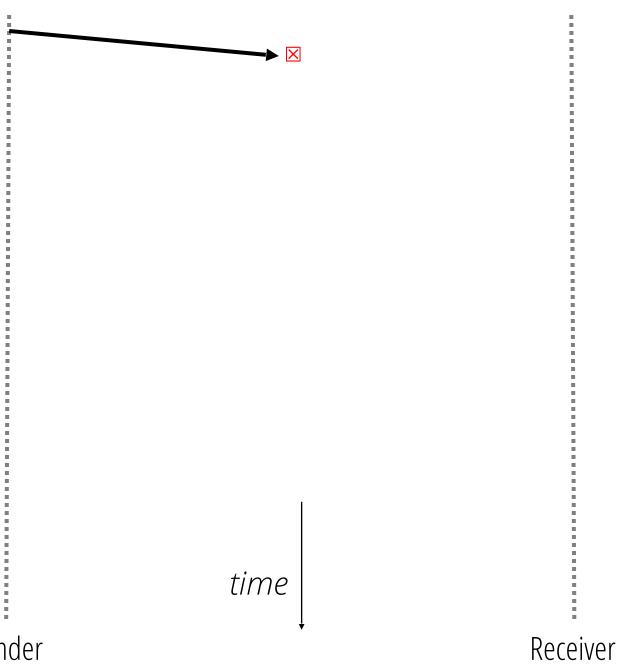
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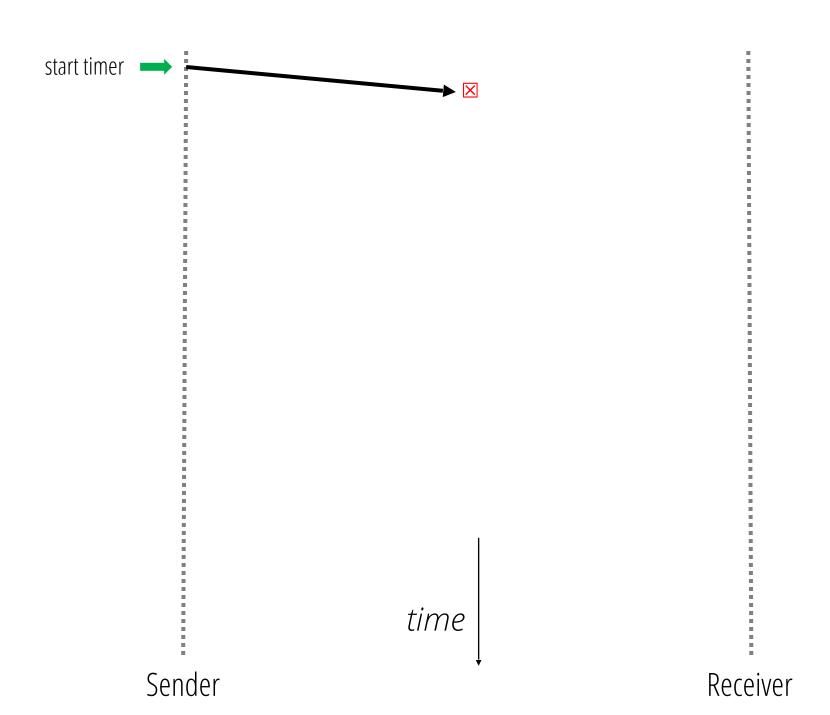
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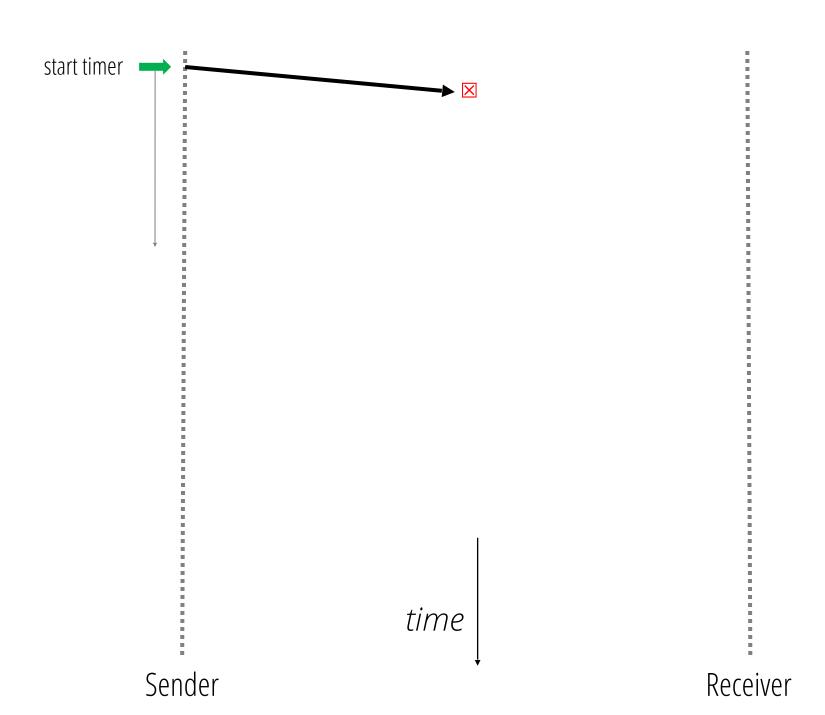
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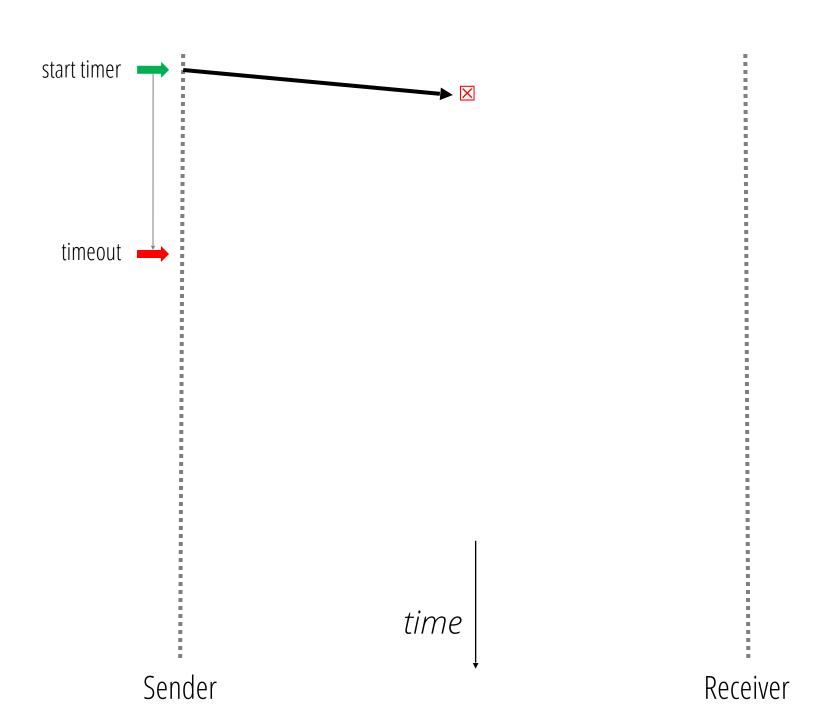


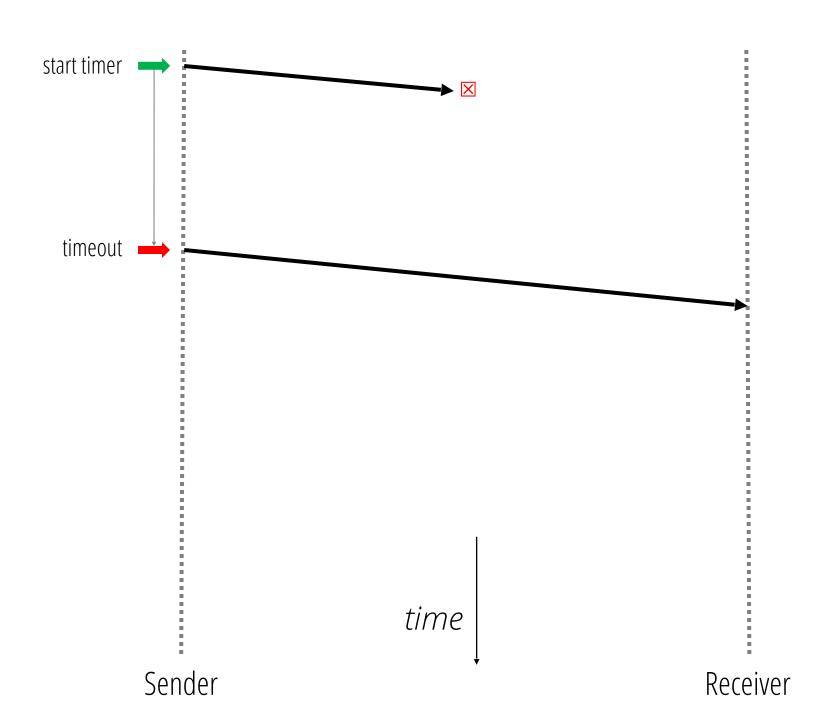


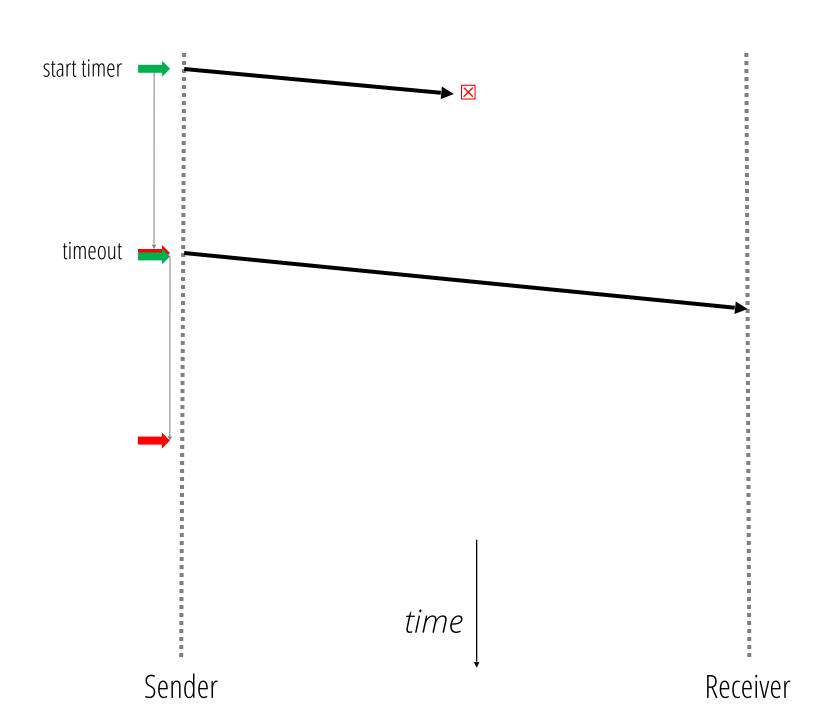
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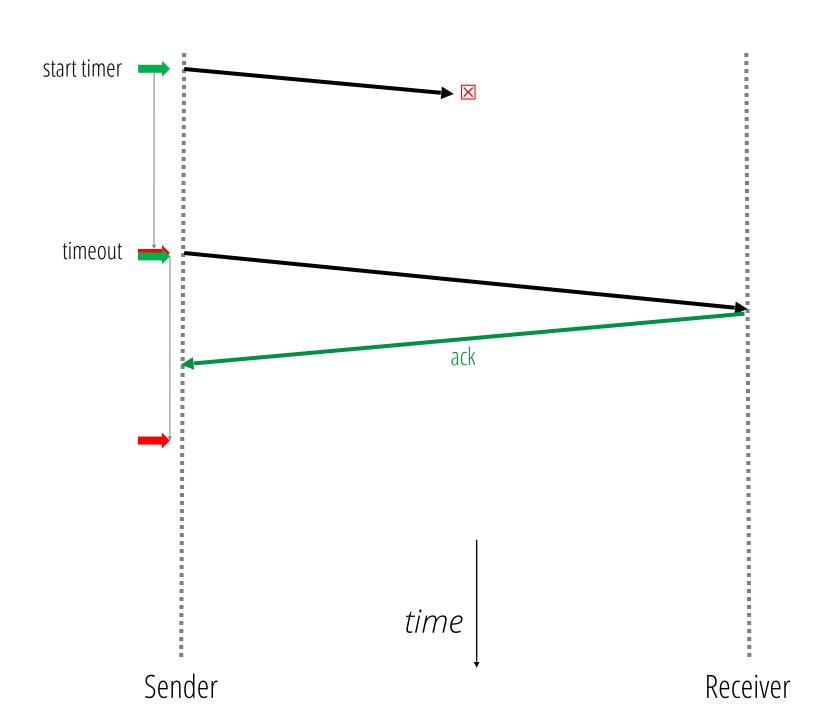


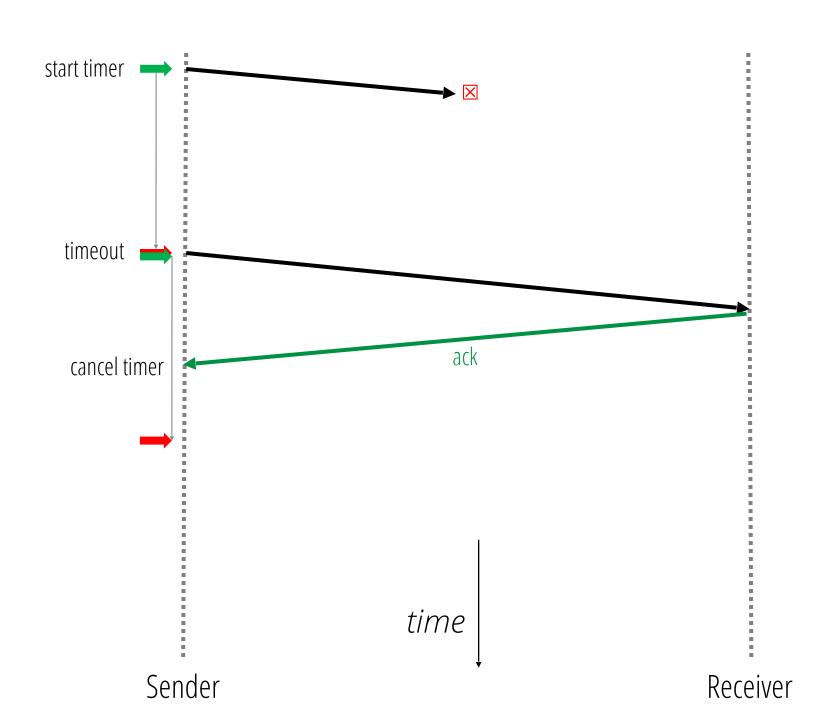


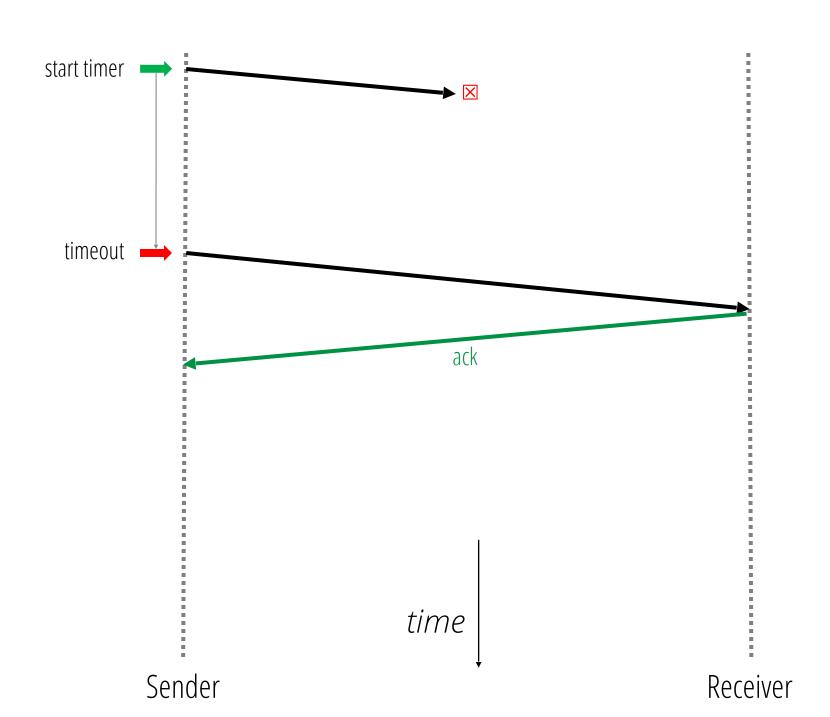


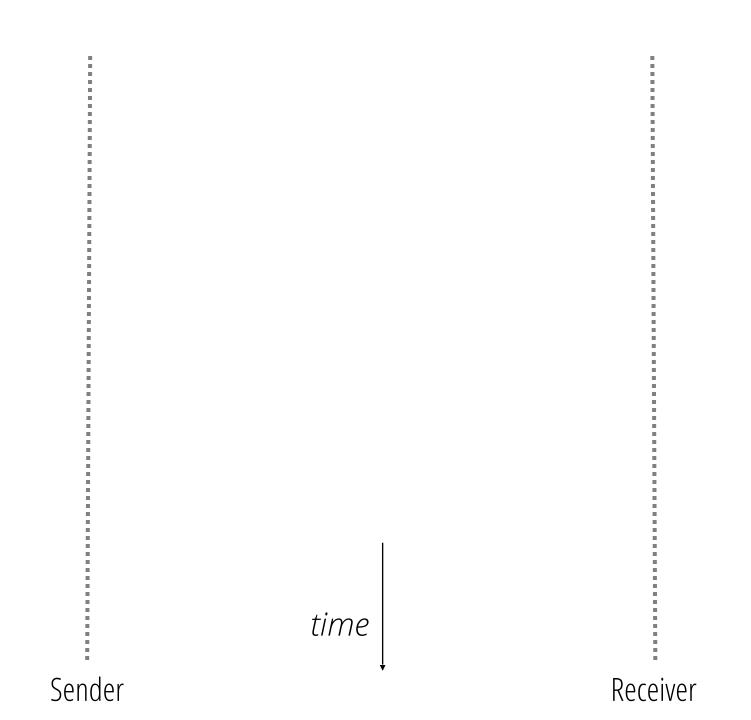


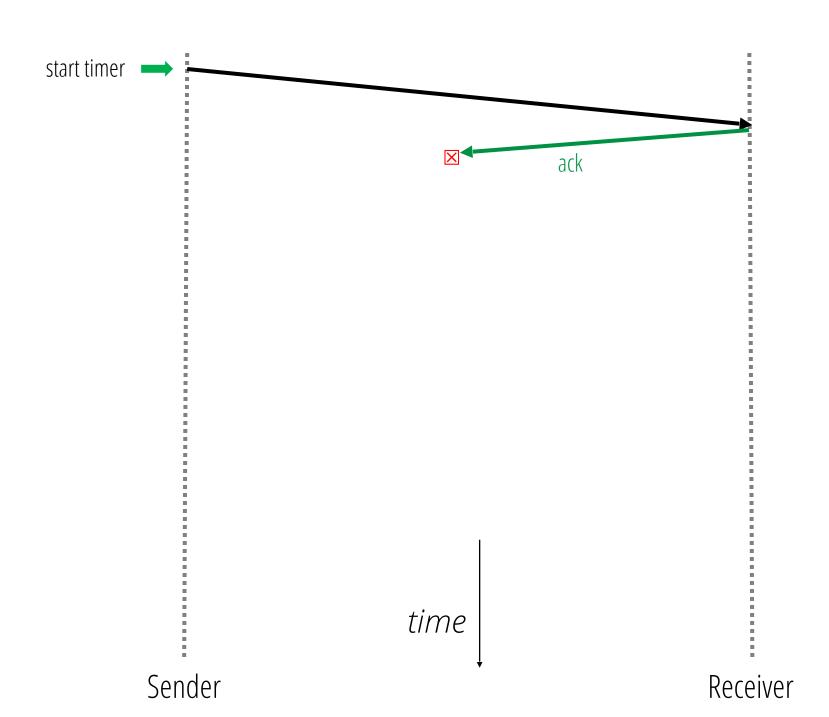


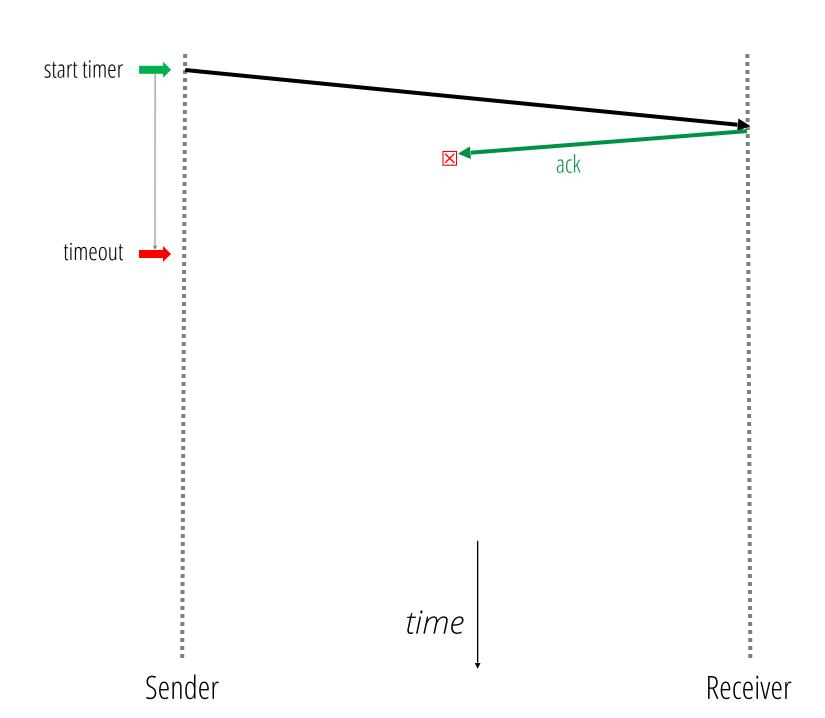


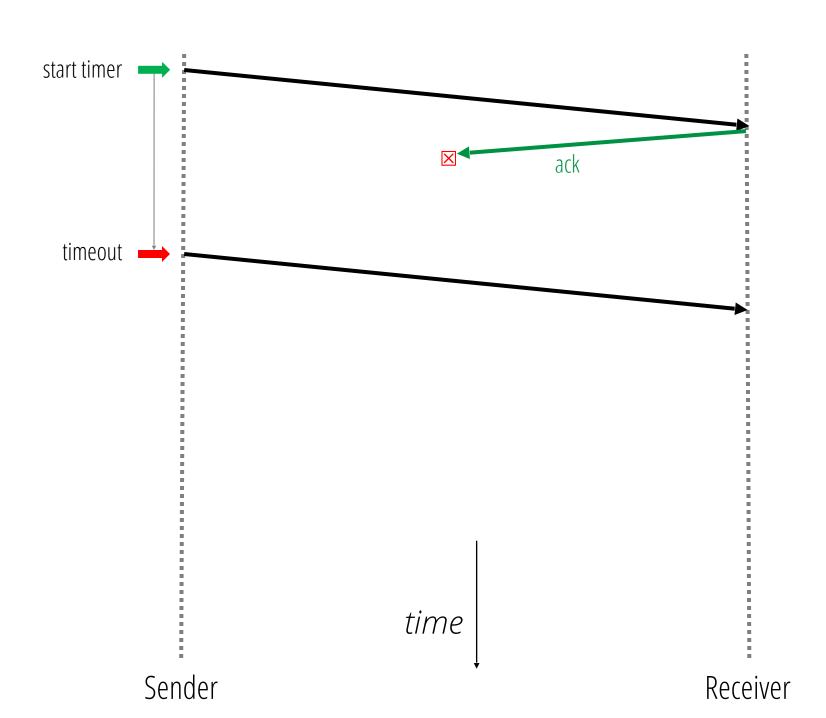


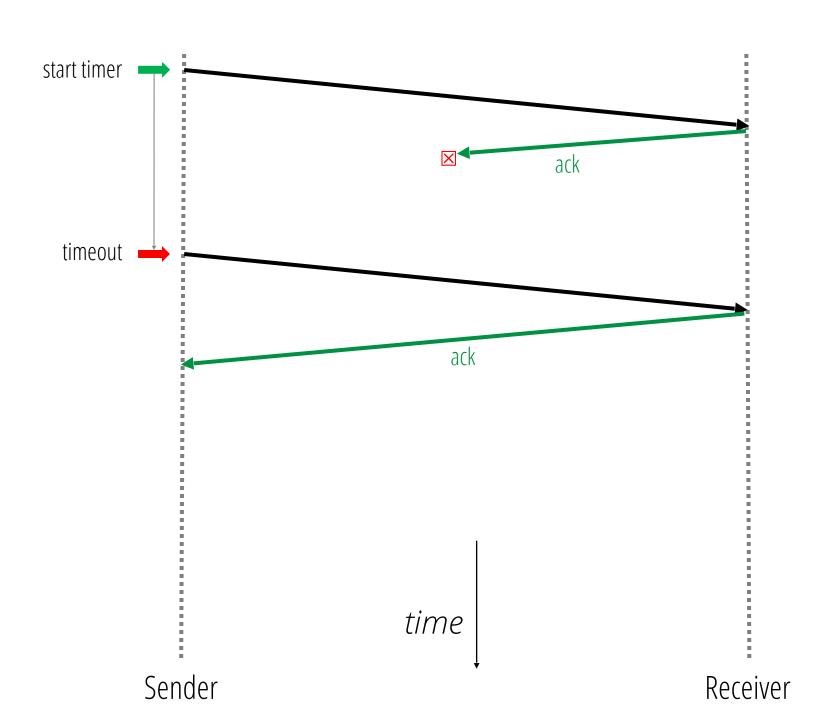


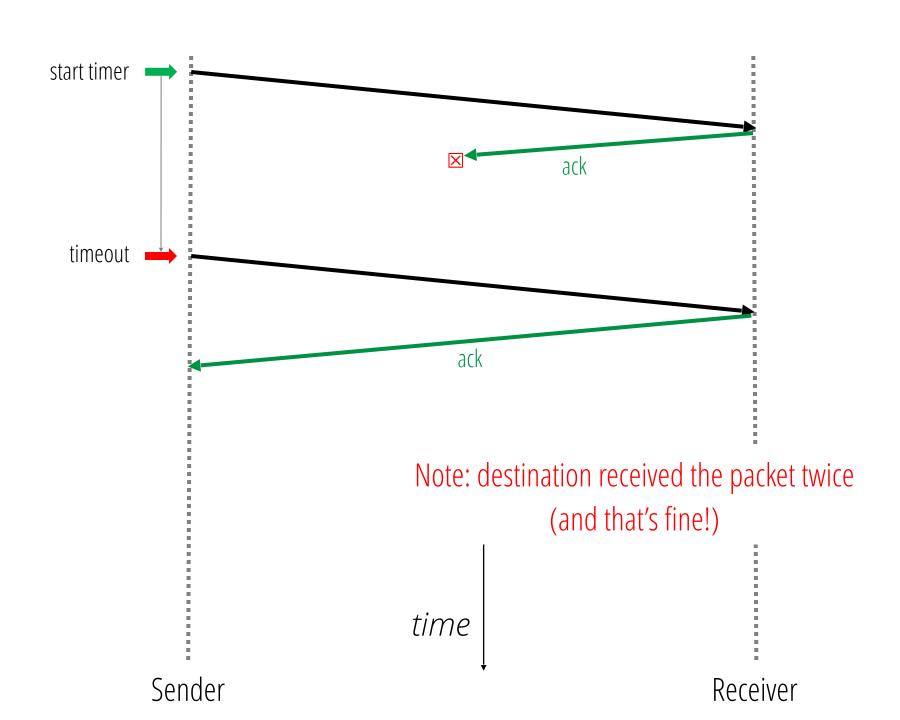












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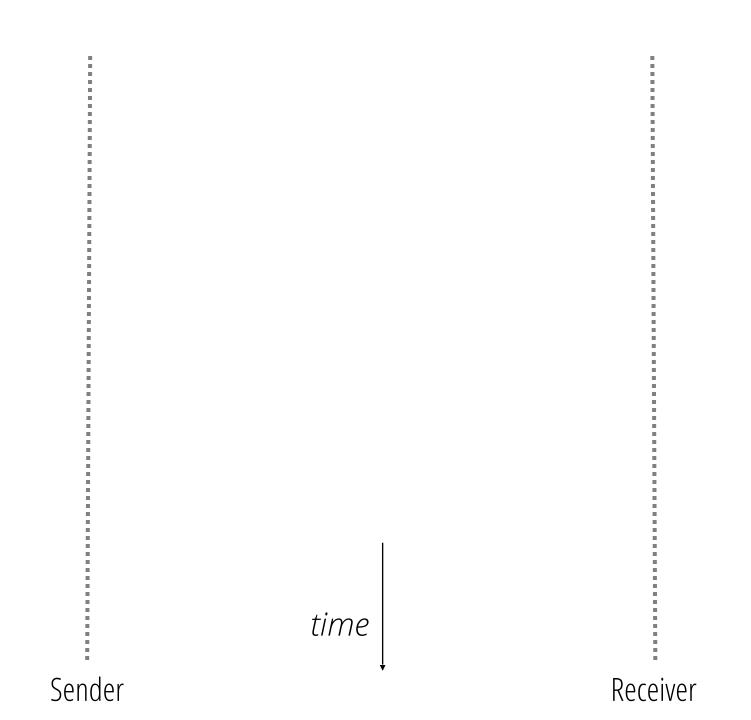
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- Hence, often used as last resort

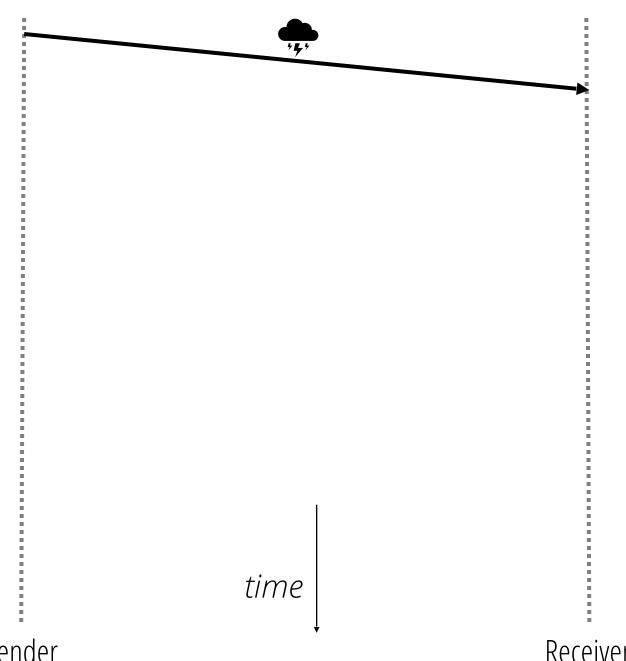
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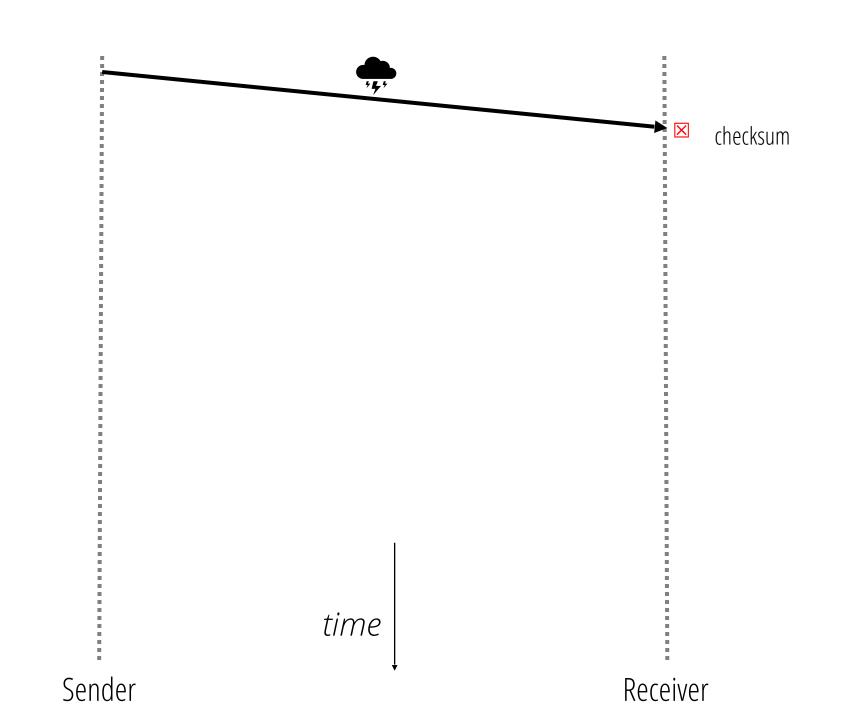


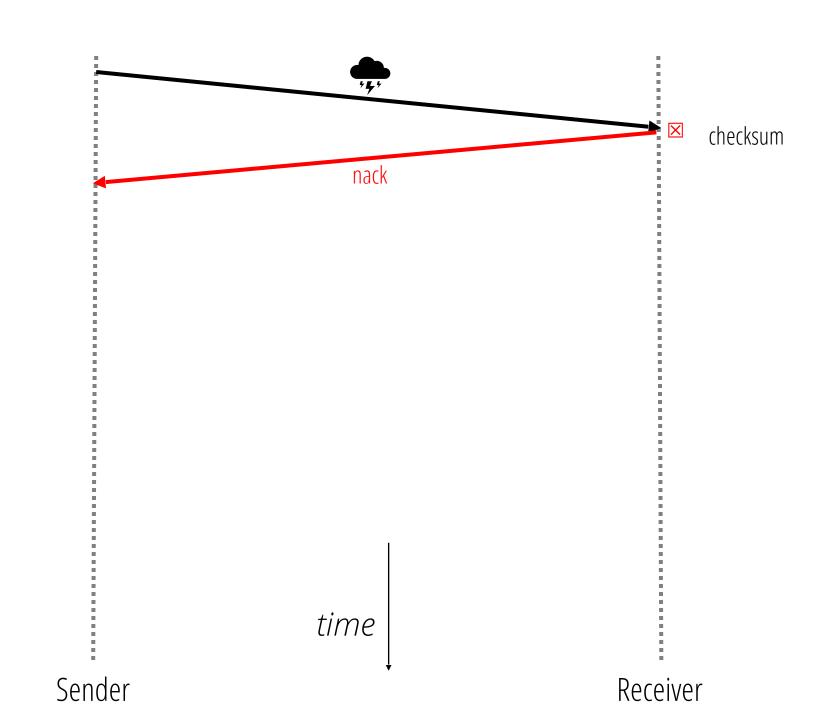
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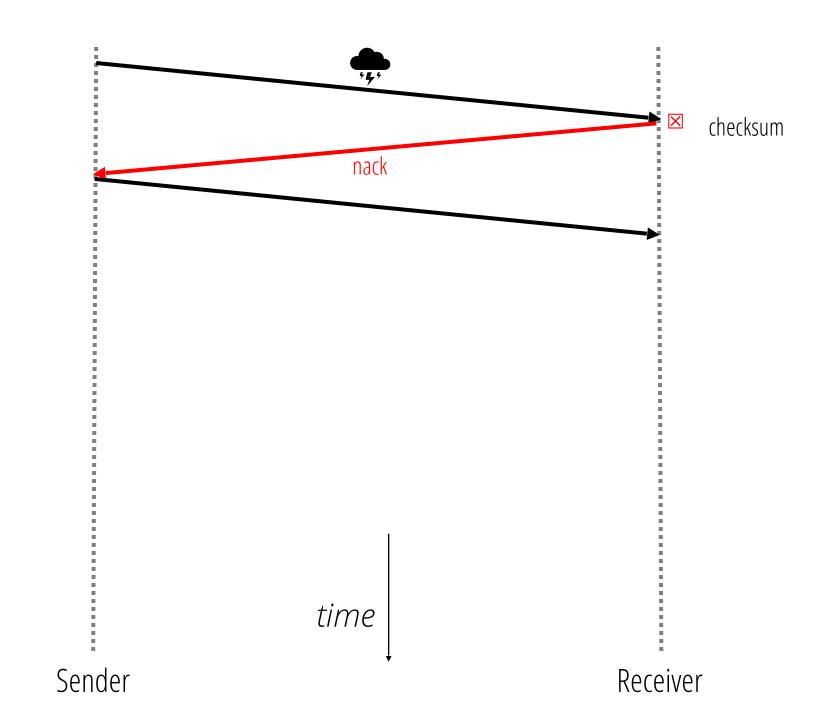


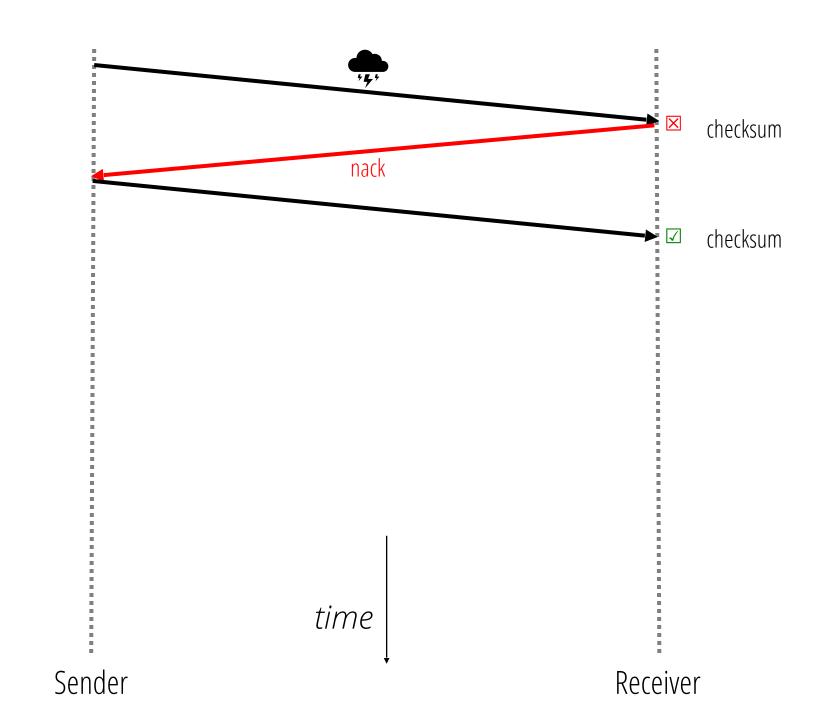


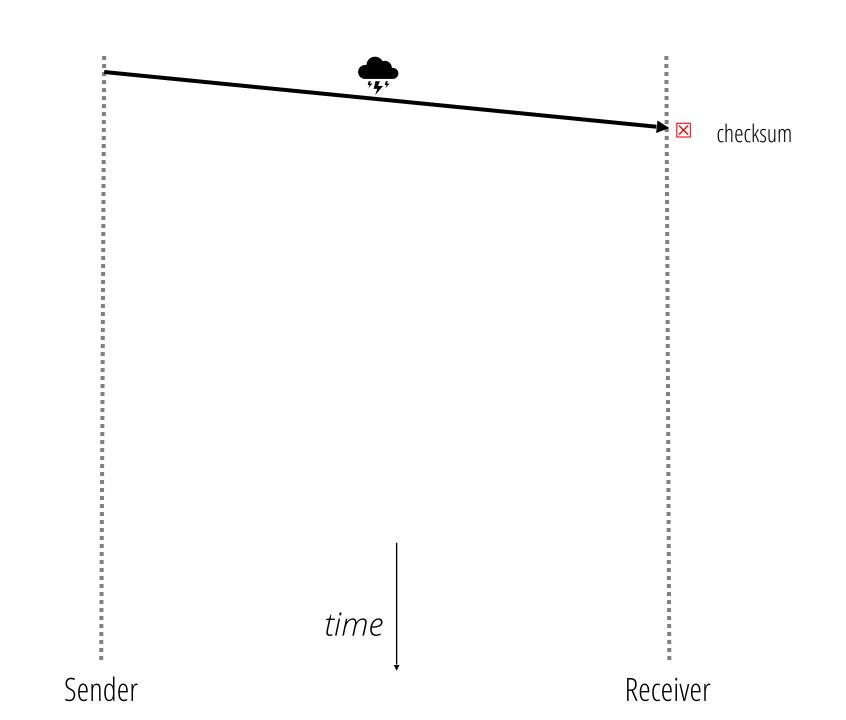
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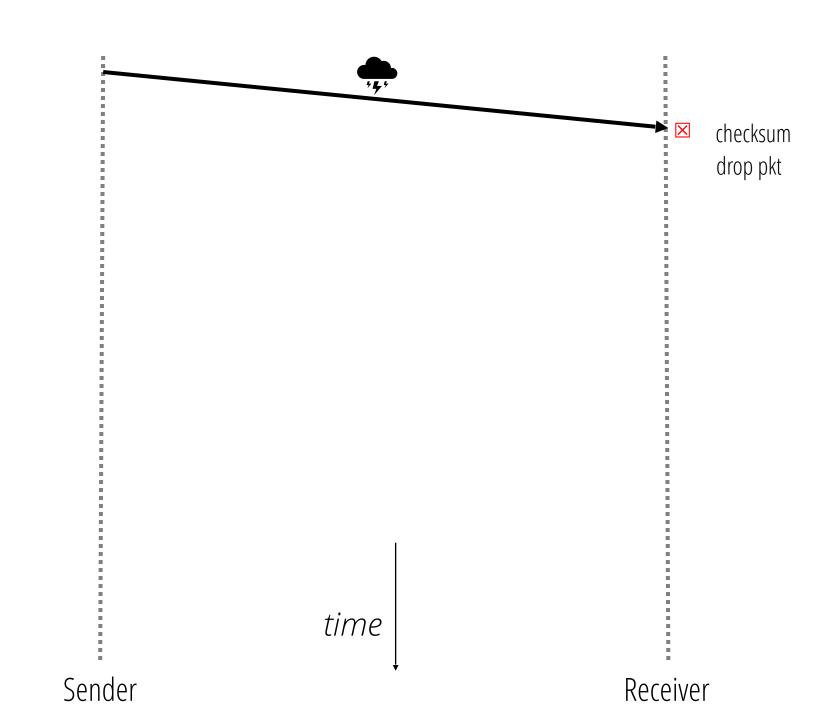


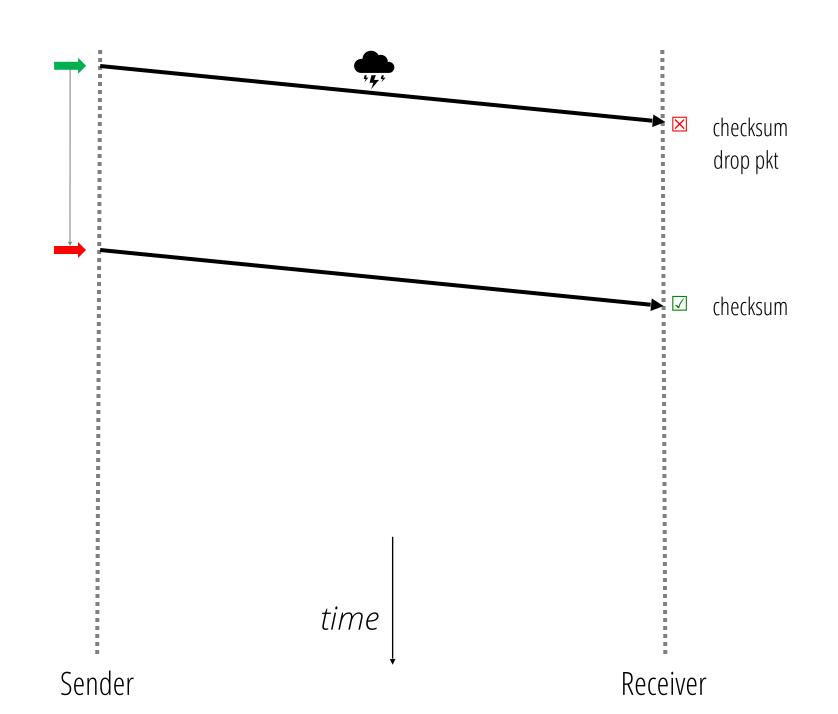












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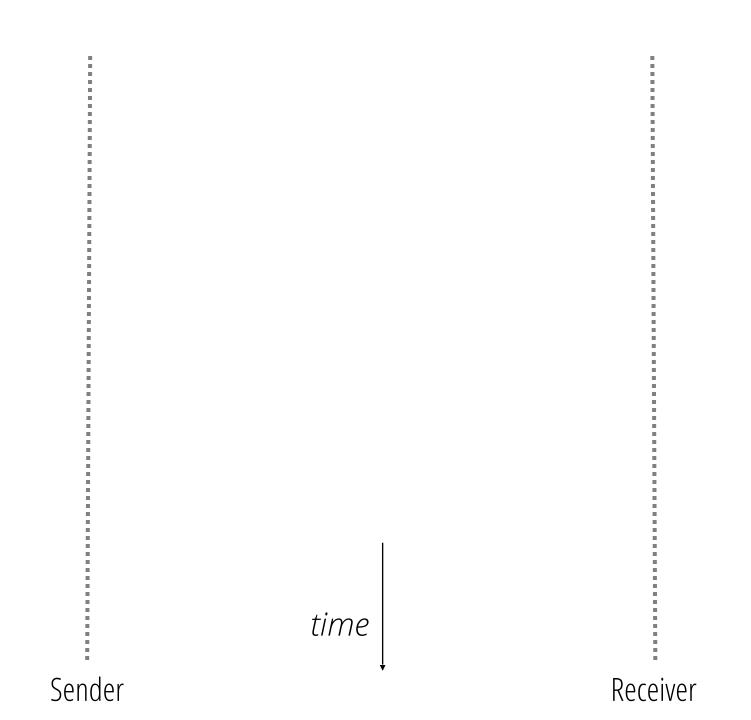
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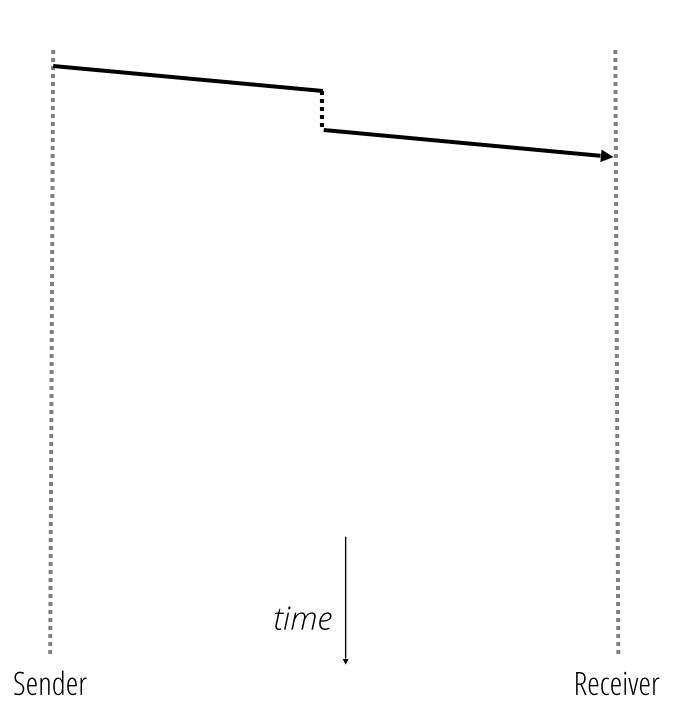


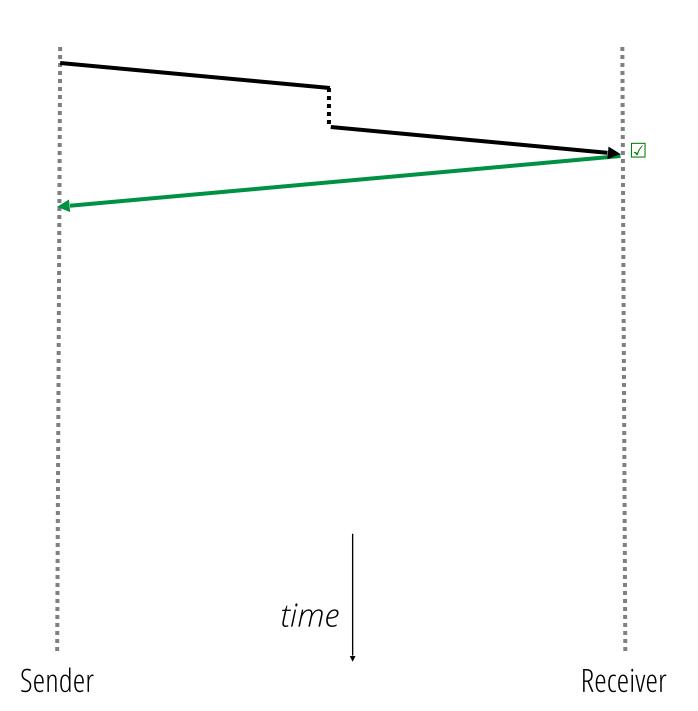
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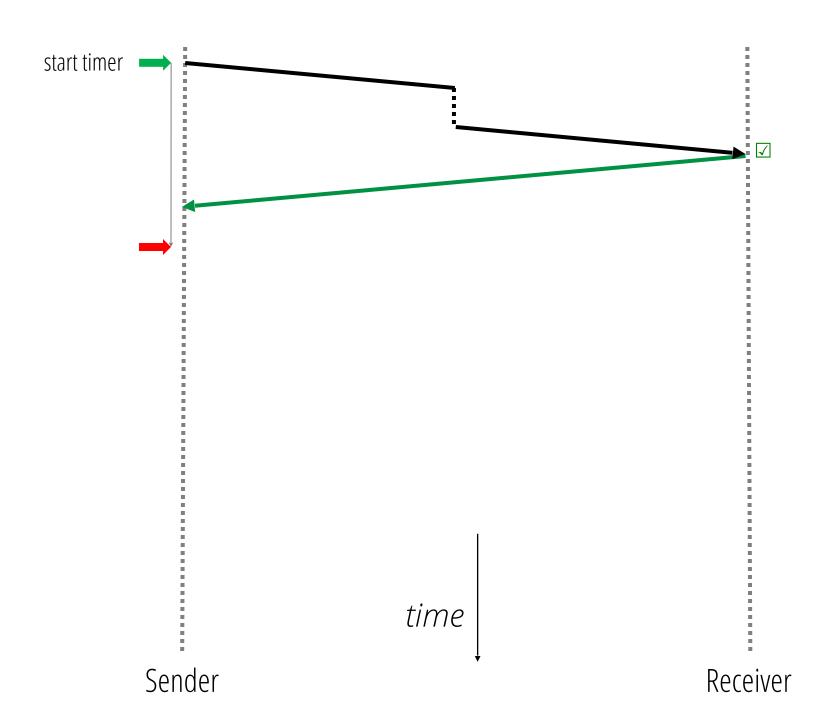


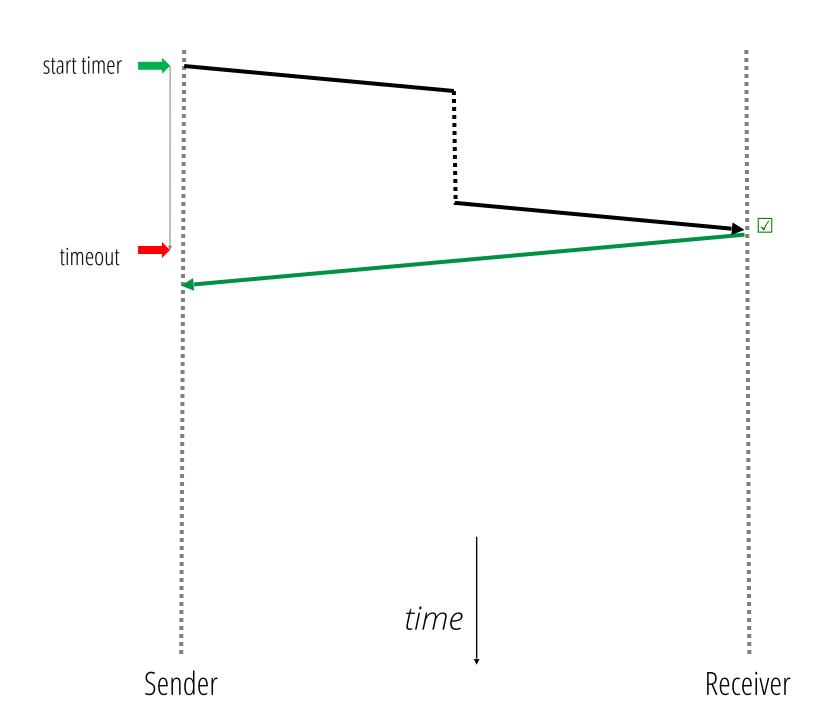
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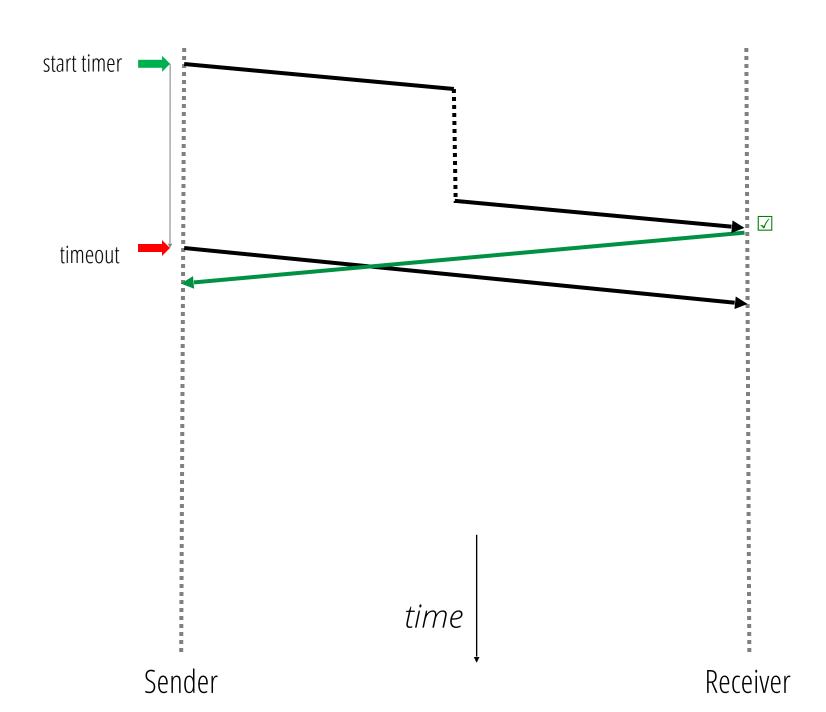


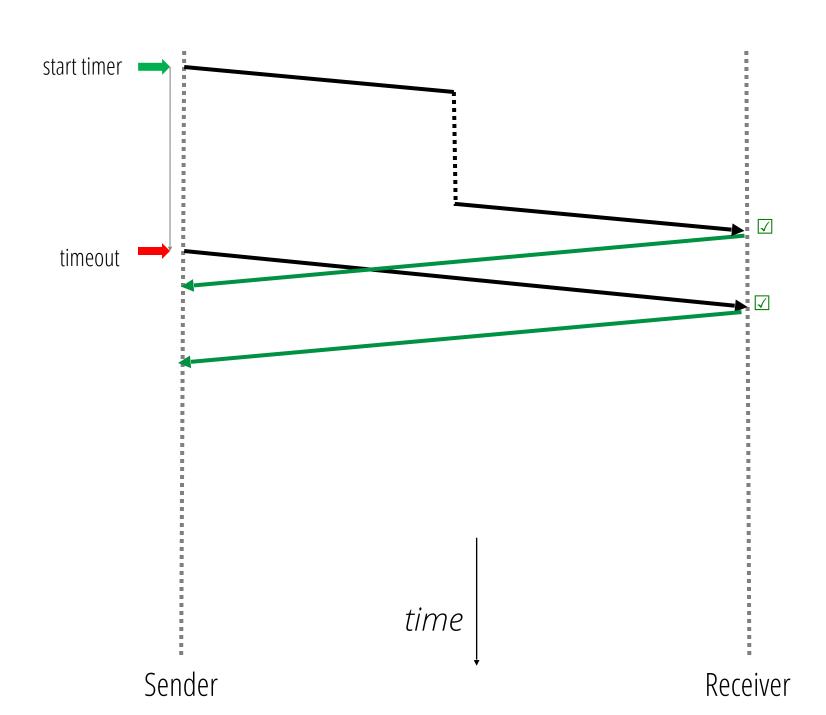


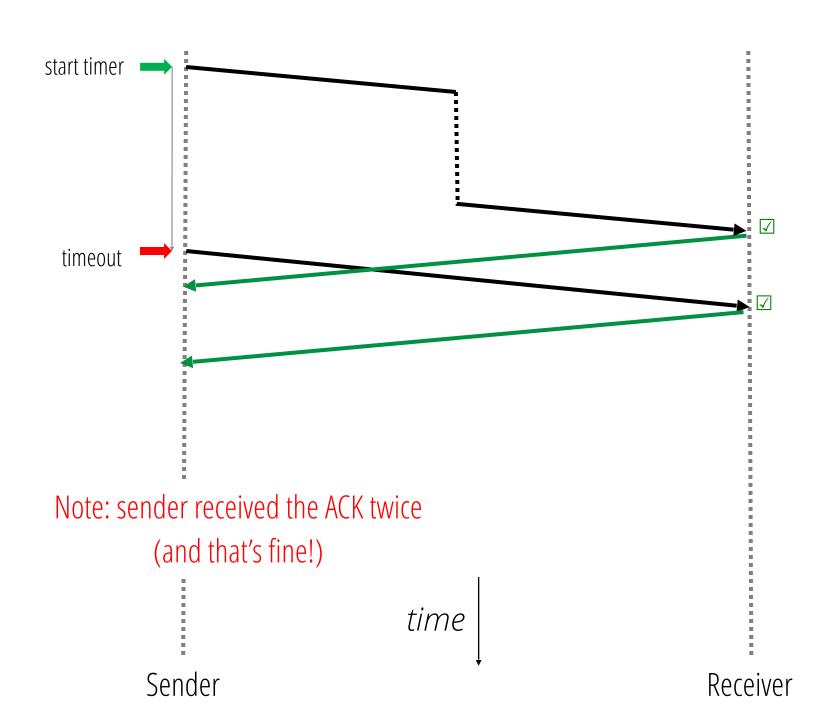


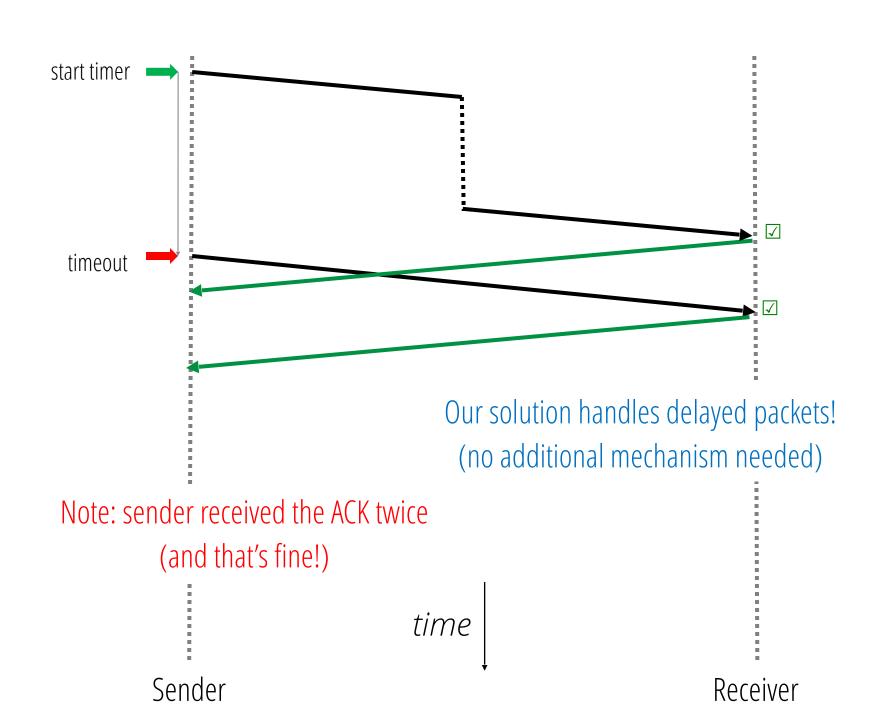












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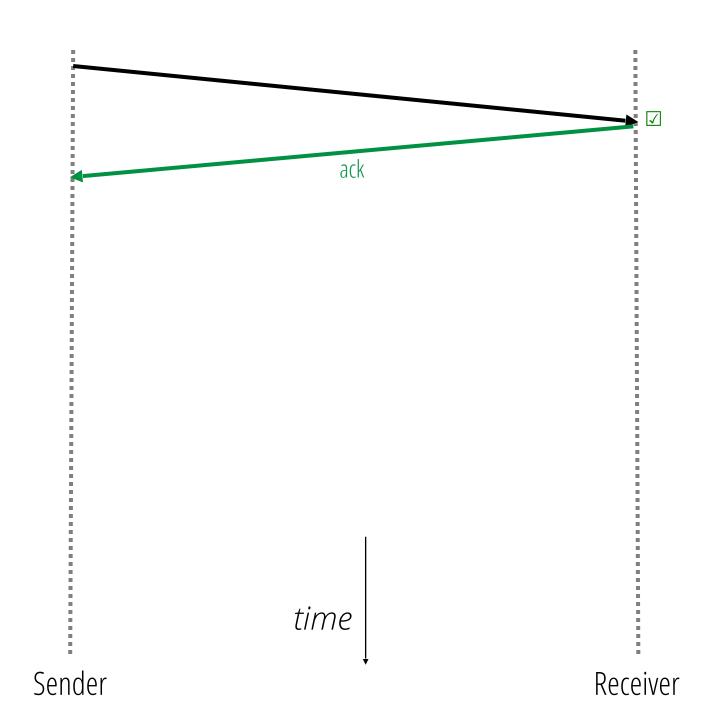
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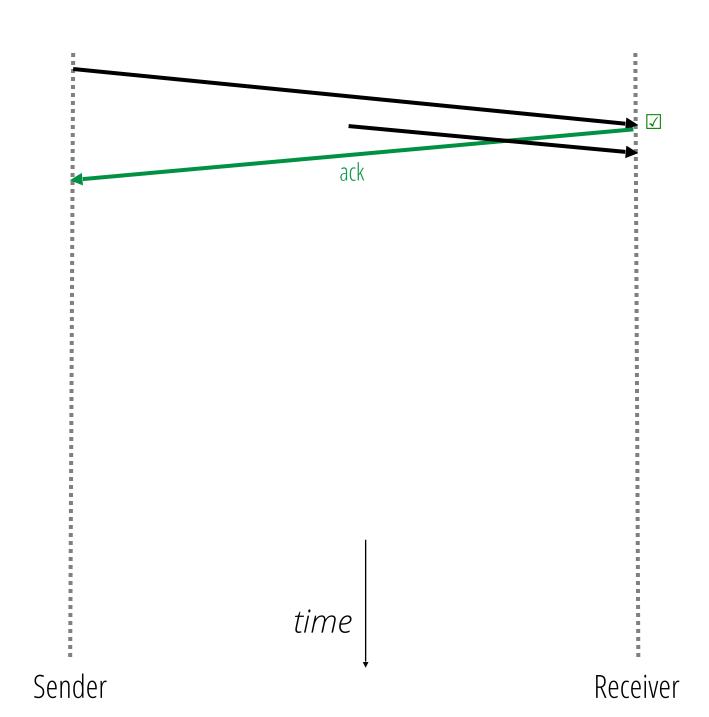


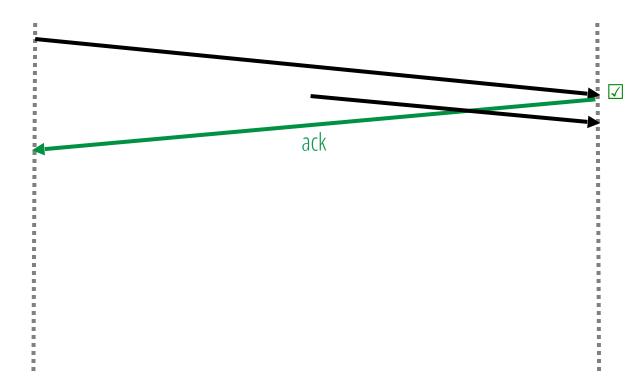
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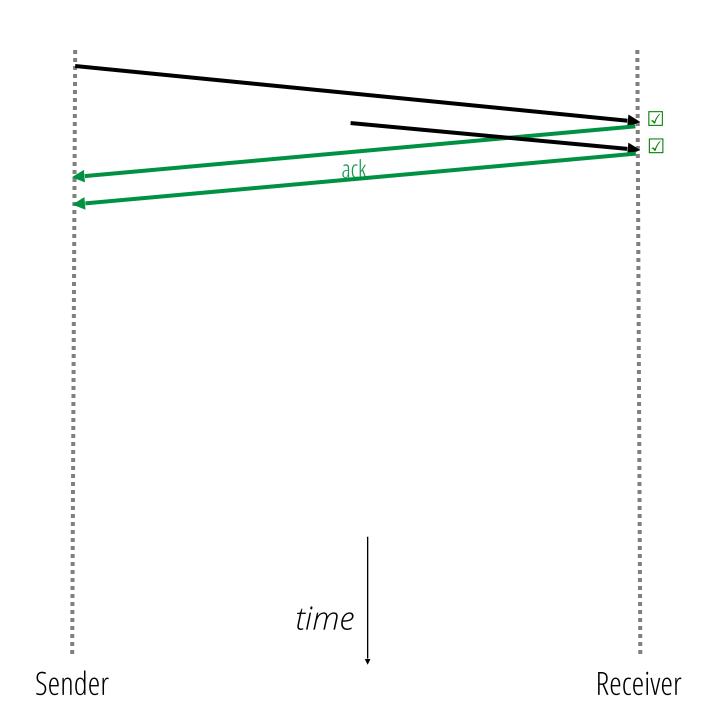


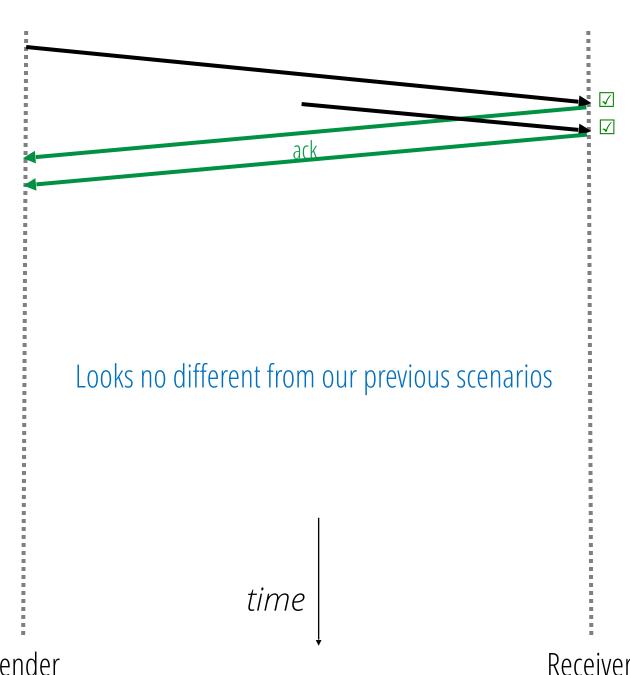
Why would the network even duplicate a packet?
Usually, because of link-level reliability gone wrong (very rare)

time

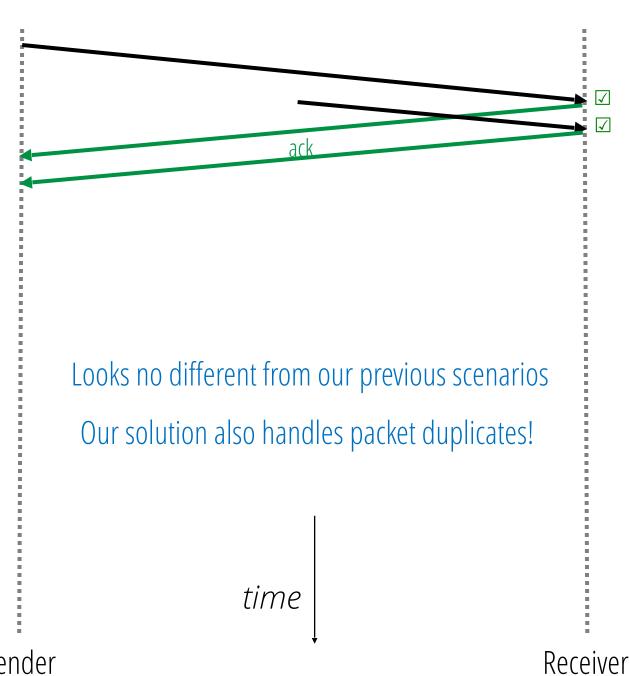
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Sender Receiver



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Packets can be reordered

Have solved the single packet case!

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- Sender:
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 - And reset timer

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Sender:

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Receiver

When receiver gets packet, sends ACK

What have we learnt?

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- Building blocks for a solution
 - Checksums: to detect corruption
 - Feedback from receiver: positive/negative (ack/nack)
 - Retransmissions: sender resends packets
 - **Timeouts**: when to resend a packet

What have we learnt?

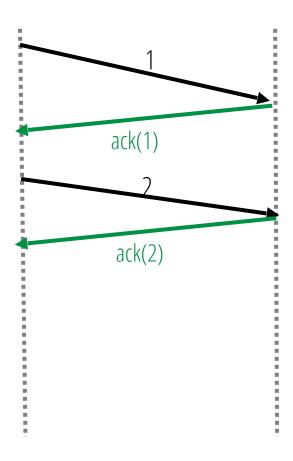
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 - Feedback from receiver: positive/negative (ack/nack)
 - Retransmissions: sender resends packets
 - Timeouts: when to resend a packet
- Semantics of a solution: "at least once"
 - Receiver can receive the same packet more than once
 - Sender can see the same ack/nack more than once

Questions?

Next: reliably send multiple packets

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Will need +1 design component: sequence numbers!



Data packets carry sequence numbers; and ACKs indicate what sequence numbers have been received

Next: reliably send multiple packets

- Will need +1 design component: sequence numbers!
- We now have all the necessary building blocks!

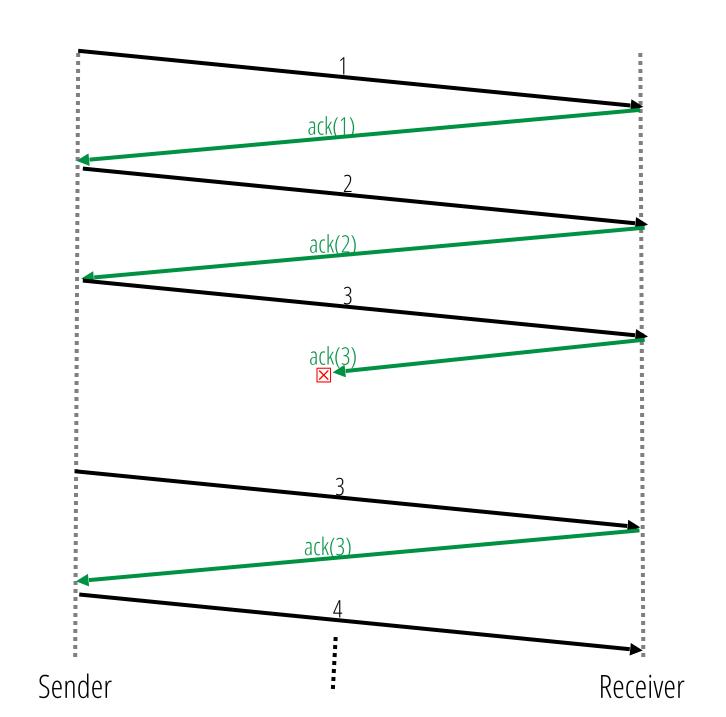
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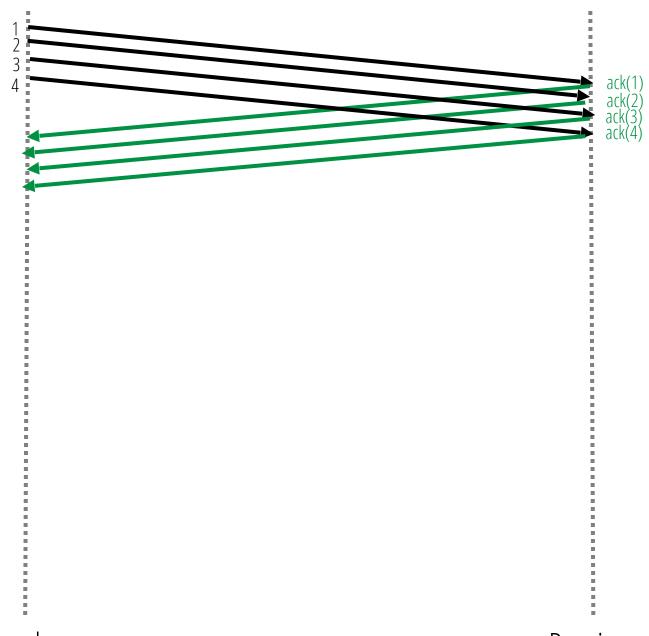
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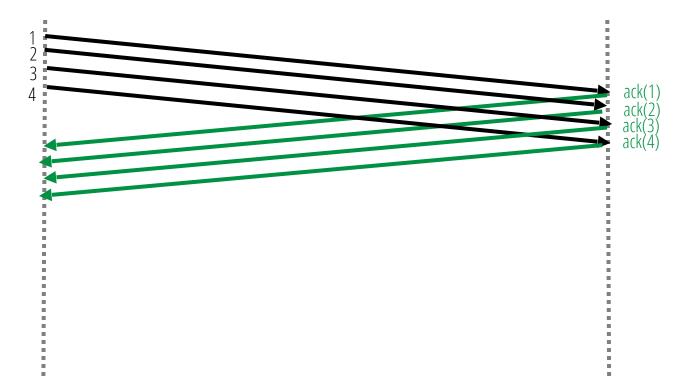
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 - Wait for packet i to be acknowledged before sending i+1
- We have a correct reliable delivery protocol!

- Probably the world's most inefficient one
 - Max throughput ~ one packet per RTT





Sender Receiver



Idea: have multiple packets "in flight" (send additional packets while waiting for ACKs to come in)

Sender Receiver

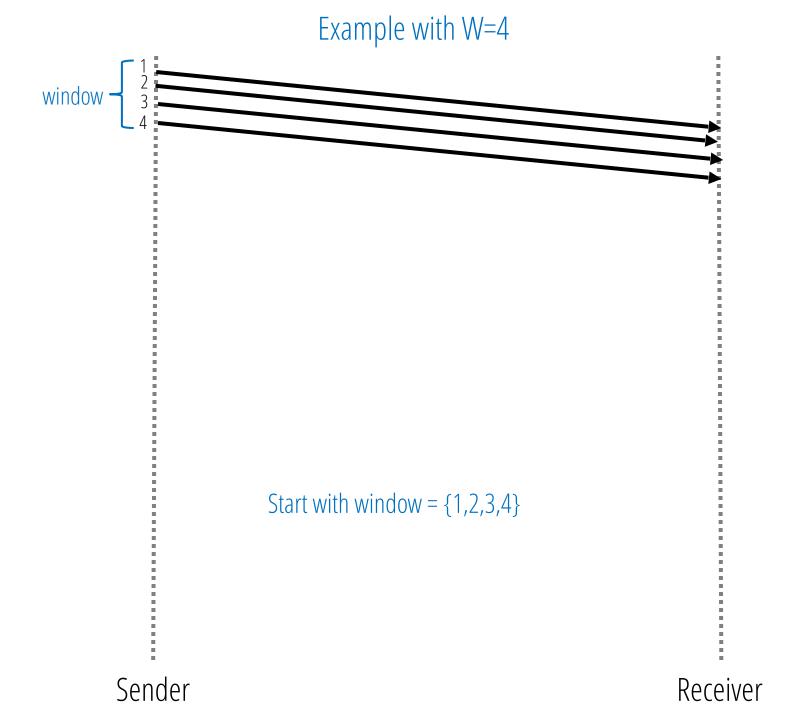
Window-based Algorithms

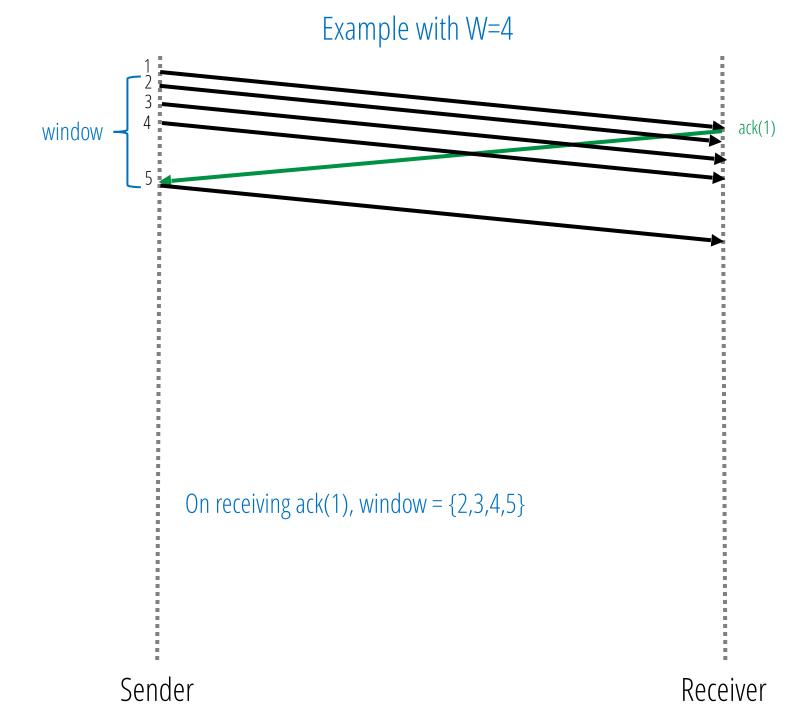
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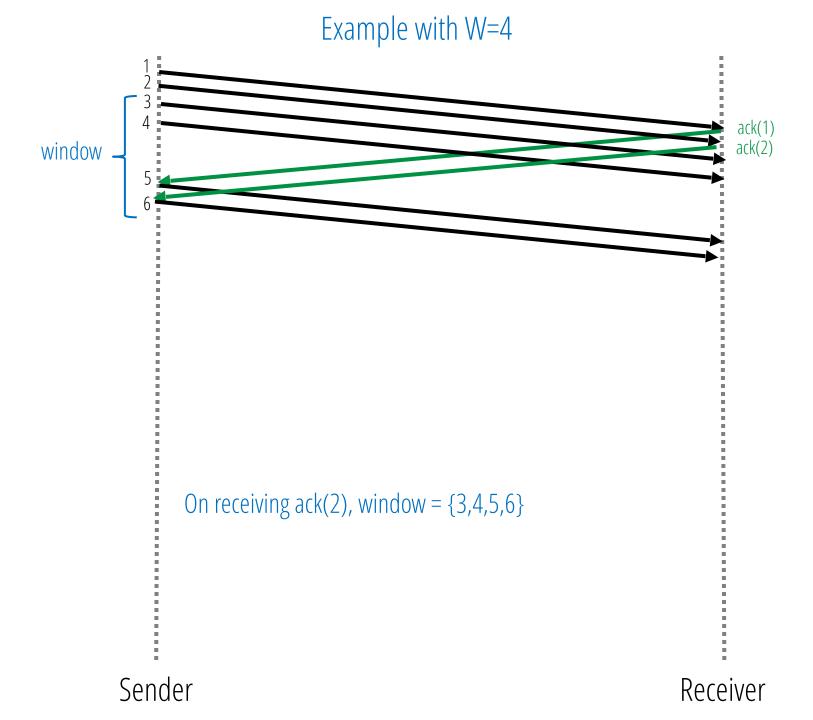
- Basic idea: allow W packets "in flight" at any time
 - W is the size of the window

Window-based Algorithms

- Basic idea: allow W packets "in flight" at any time
 - W is the size of the window
- Hence, a simple algorithm (at sender)
 - Send W packets
 - When one gets ACK'ed, send the next packet in line







Reliably sending many packets

- Will need +1 design component: sequence numbers!
- We now have all the necessary building blocks

Reliably sending many packets

- Will need +1 design component: sequence numbers!
- We now have all the necessary building blocks

- Plus one more, for efficiency (performance)
 - Window

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Pick window size W to balance three goals

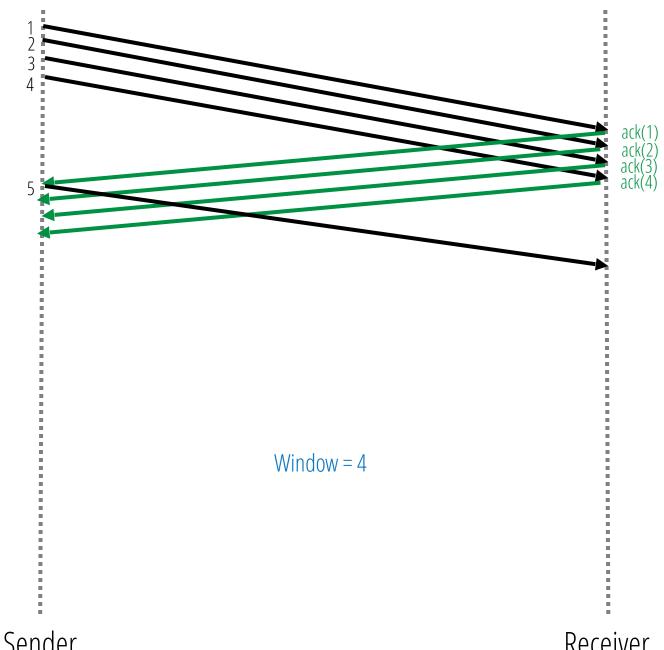
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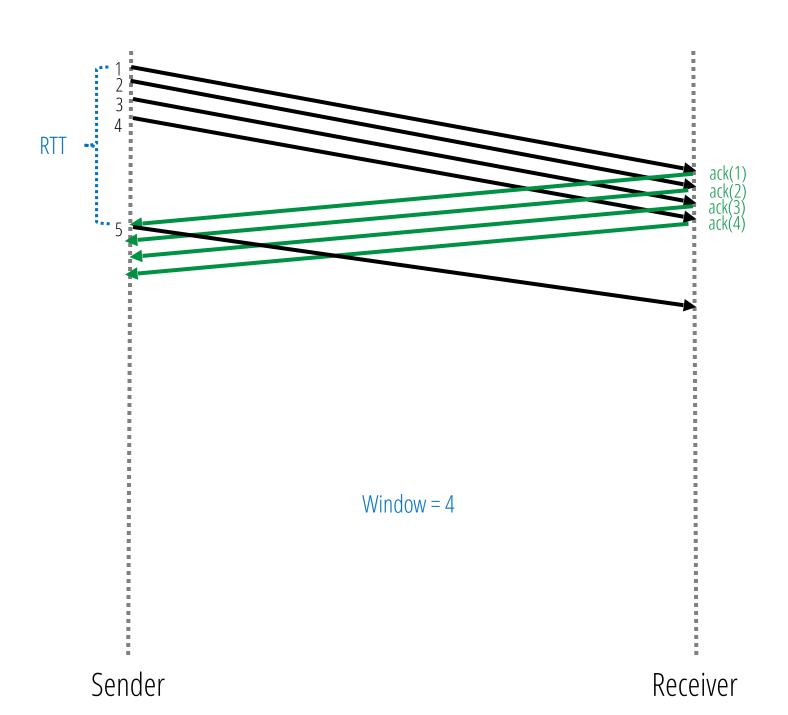
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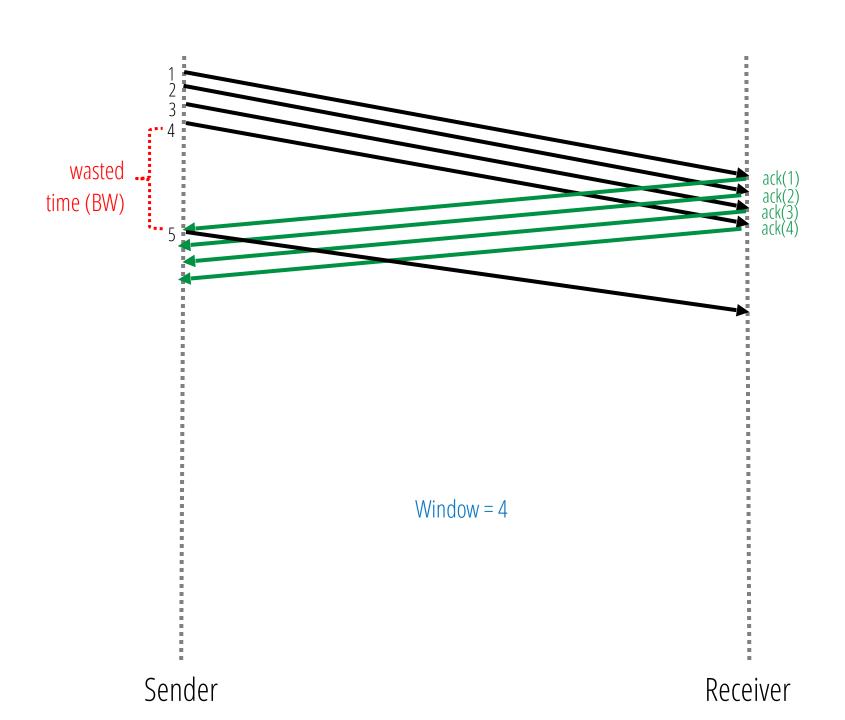
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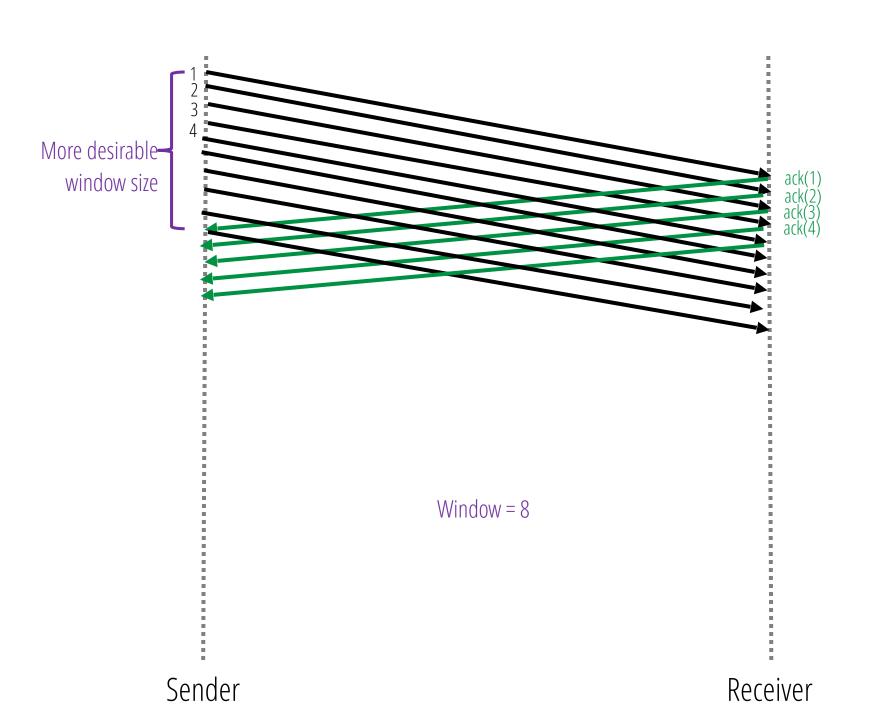
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 - W should allow sender to transmit for entire RTT
 - From sending first packet until receive first ACK



Sender Receiver





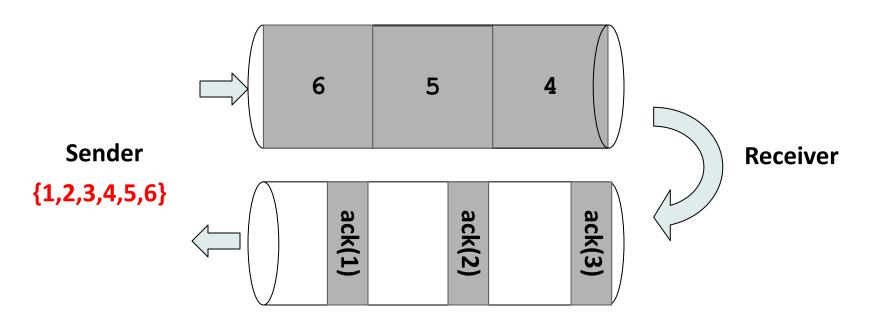


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- Hence, condition: W x Packet-Size ~ RTT x B
 - E.g., for a path with RTT=1 second and bottleneck B = 8 Mbits/second, if packet size = 100 Bytes, we want a window size W = 10,000 packets

Setting W to be one RTT of packets

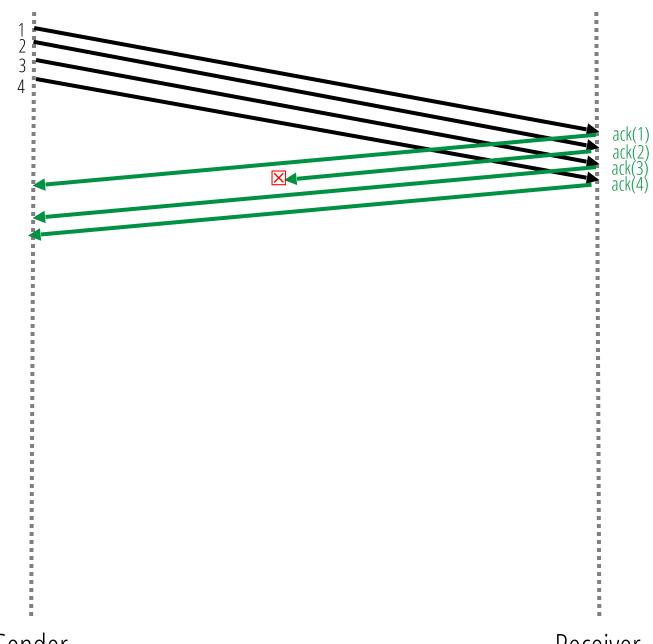


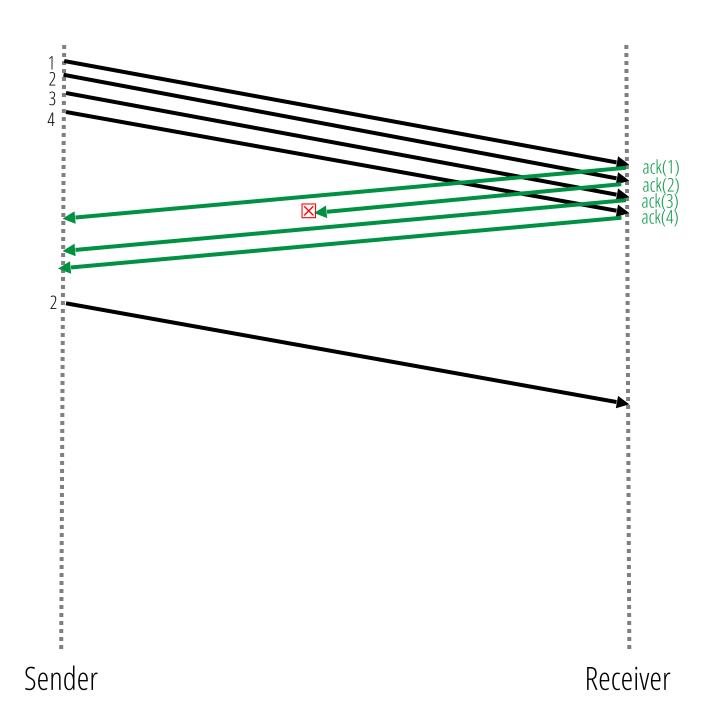
New Design Considerations

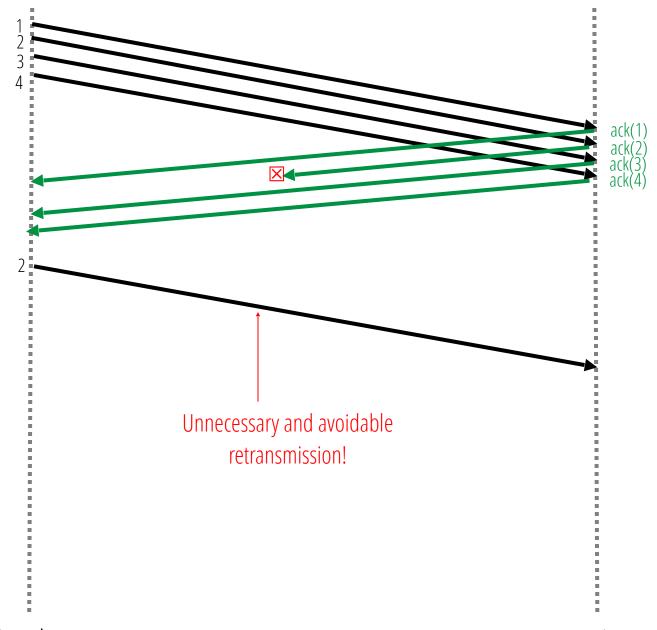
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ACKs: design options

- Individual packet ACKs (our design so far)
 - On receiving packet i, send ack(i)







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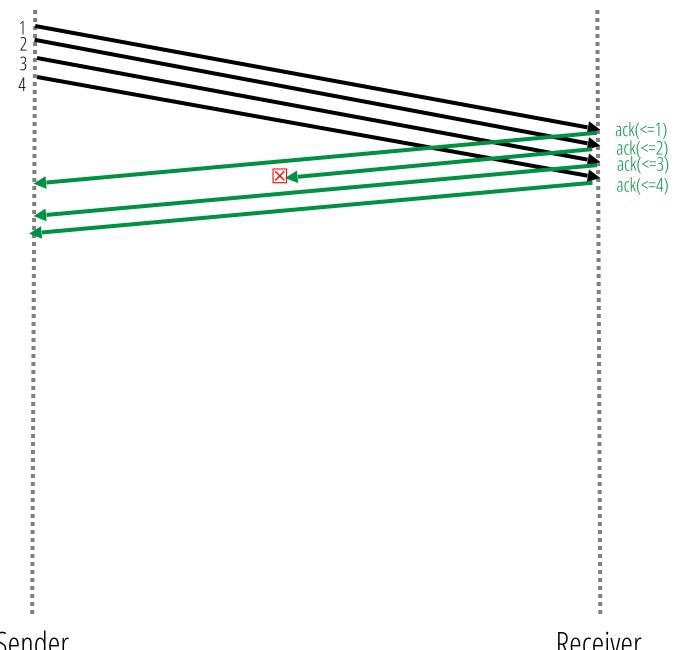
Receiver

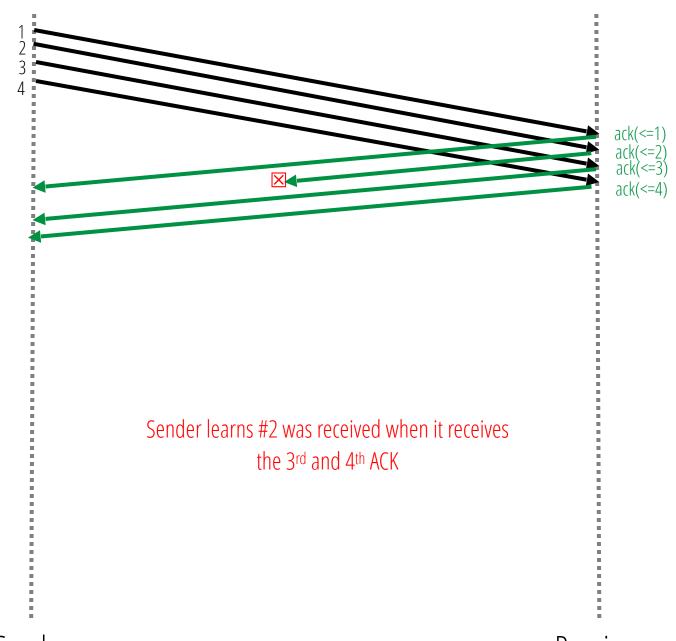
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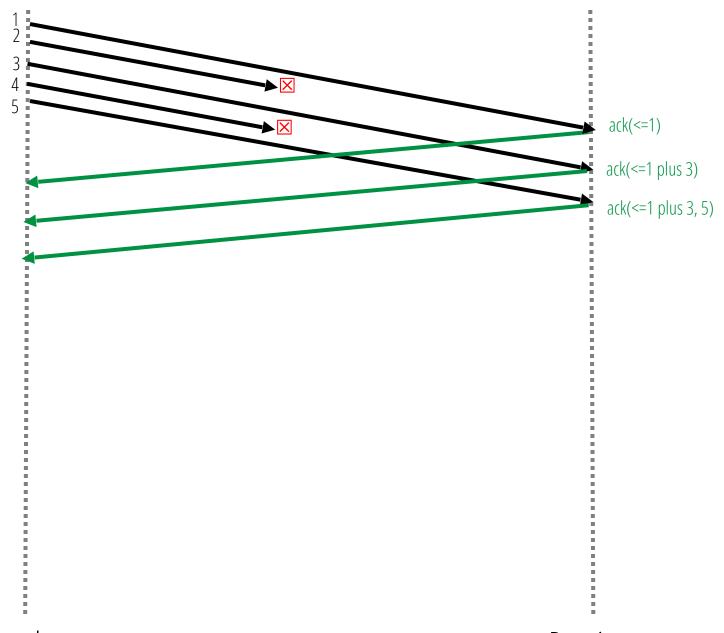
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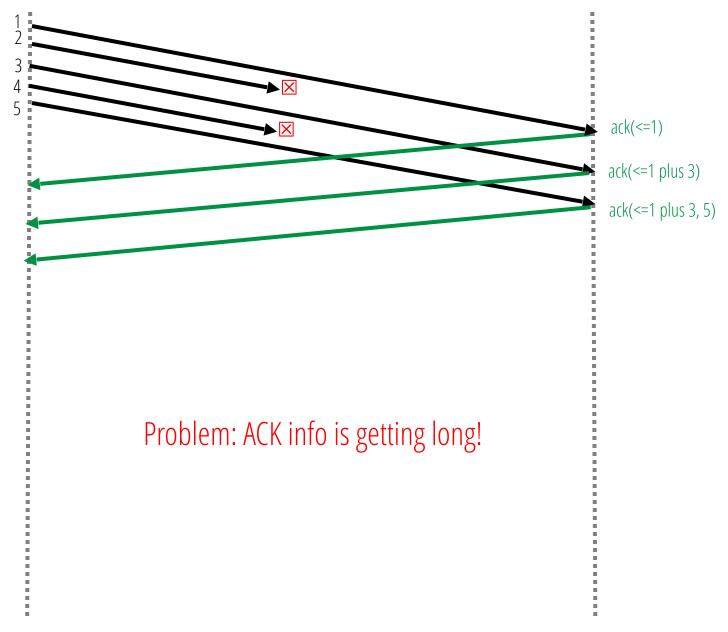
Full Information ACKs

 Give highest cumulative ACK plus any additional packets received ("everything up to #12 and #14, #15")









Sender

Receiver

ACKs: design options

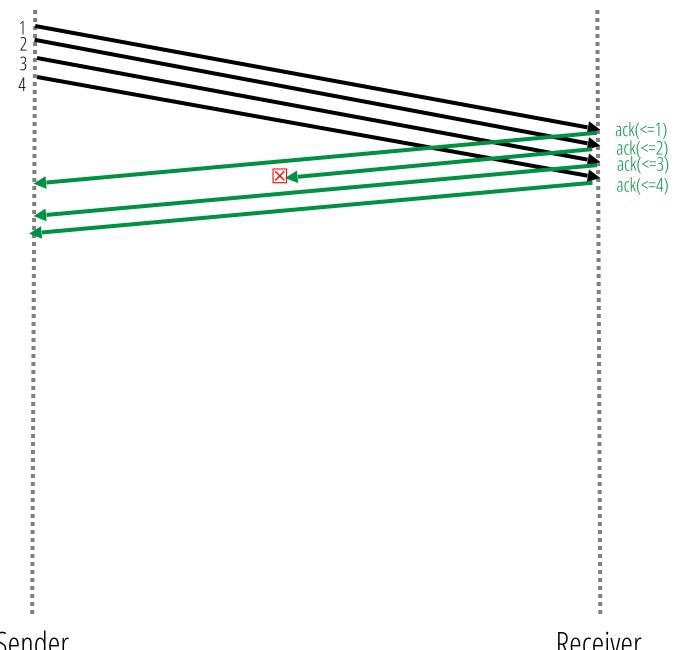
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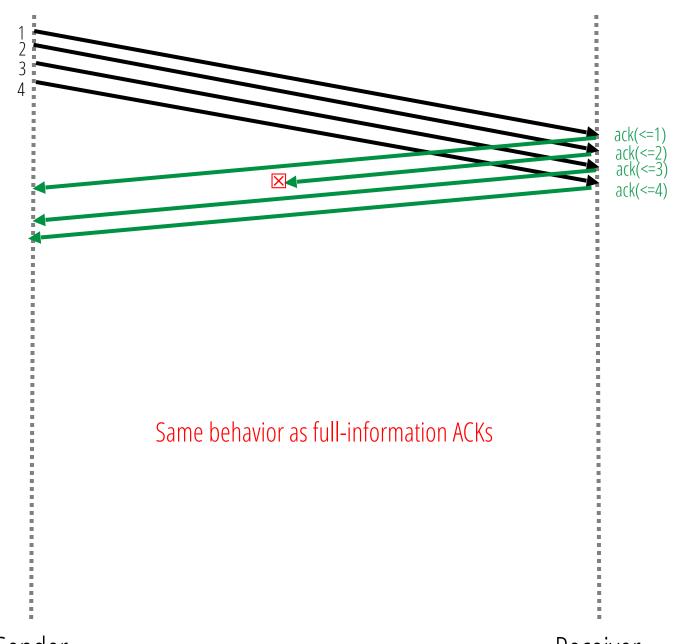
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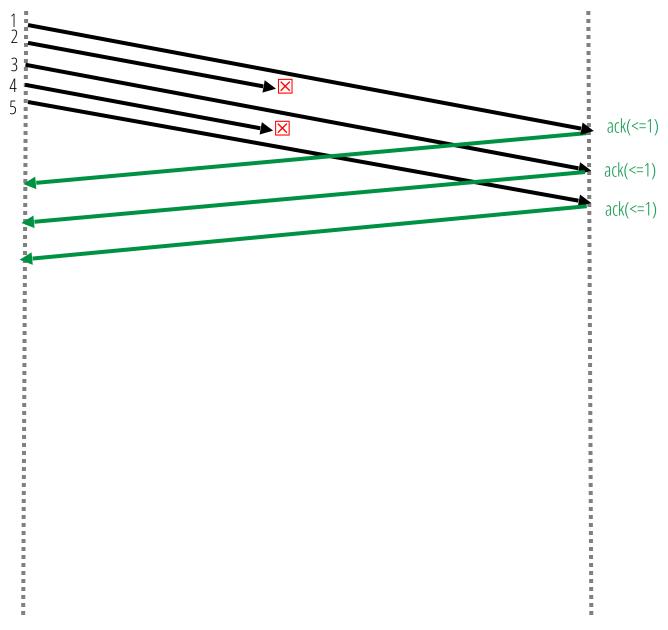
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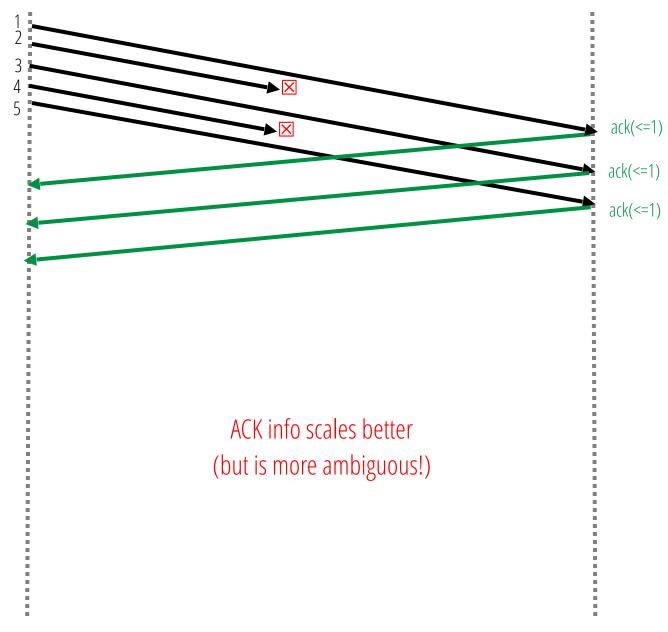
Cumulative ACKs

ACK the highest sequence number for which all previous packets have been received









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- Pro: compact; simple
- Con: loss of ACK packet *always* requires a retransmission

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Cumulative

- Pro: compact; more resilient to ACK loss (vs. individual ACKs)
- Con: Incomplete info on which data packets arrived

New Design Considerations

- Window size
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Detecting Loss

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• If packet times out, assume it is lost...

How else can you detect loss?

Detecting Loss

If packet times out, assume it is lost...

- How else can you detect loss?
- When ACKs for k "subsequent packets" arrive
 - E.g., only packet 5 is lost, will receive ACKs for 6, 7, ...
 - E.g., if k=3, retransmit 5 after we receive ACKs for 6, 7, 8
 - Details look a little different for each ACK option (next slides)

• Assume packet 5 is lost, but no others

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 - 2
 - 3
 - 4
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← Packet 5 lost! (Received k=3 dupACKs)

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 - But can be ambiguous with cumulative ACKs and multiple losses

- Consider a sender with a window size = 6 & k=3
 - Packets 1,2 have been ACKed
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 - ACK 7 arrives (3rd ACK for subsequent packet)

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 - ACK 9 arrives → send 13, and so on...

Response with full-info ACKs

Similar behavior as with Individual ACKs

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- (for packet 8,9,10) ACK 2 → unclear what packet to resend!

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- Unfortunately, TCP uses cumulative ACKs...

Taking Stock...

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 - Checksums
 - ACK/NACKs
 - Timeouts
 - Retransmissions
 - Sequence numbers
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- And discussed tradeoffs in how to apply them
 - Individual vs. Full vs. Cumulative ACKs
 - Timeout vs. ACK-driven loss detection

From design options to design

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- Can put together a variety of reliability protocols from our building blocks!
 - We saw one already: Stop-and-Wait
 - Another possibility: "Go-Back-N" (in section)
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 - We saw one already: Stop-and-Wait
 - Another possibility: "Go-Back-N" (in section)
 - TCP implements yet another (next lecture)
- More important that you know how to design and evaluate a reliability protocol, than that you memorize the details of any one implementation!

Preview: what does TCP do?

- Uses most of our building blocks w/ a few diffs.
 - Checksums
 - ACKs (no explicit NACKs)
 - "Sliding" Windows

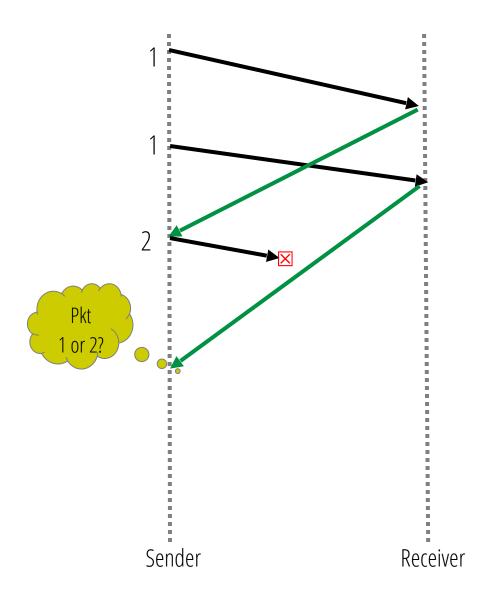
 - Cumulative ACKs (and counting dupACKs)
 - Option for a form of full-information ACKs (SACK)
 - Timers (w/ timer estimation algorithm)

- Sender encodes the data to be resilient to loss
 - Basic idea: add some redundancy to data / packet stream
 - E.g., take k packets, encode as n (>k) packets
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- Vast literature on coding schemes
 - E.g., fountain codes, raptor codes, ...
- Historically not used very much but that could change...

Questions?



Backup#2: We need sequence numbers with stop-and-wait