CS 43: Computer Networks

Transport Layer & Reliable Data Transfer

October 24, 2019

SWARTHMORE COLLEGE

Announcements

- Final Exam on Dec 17th 2 – 5 PM
- Lab 4 Deadline extended to Thursday
- Lab 5 Partnerships: extended to 5 labs with the same lab partner.
- Discussion questions with answers after class.

Reading Quiz

Transport Layer

Today

- Unreliable, unordered service: UDP
- Principles of reliability
- Class of protocols: Automatic Repeat Requests

Moving down a layer!

Application Layer

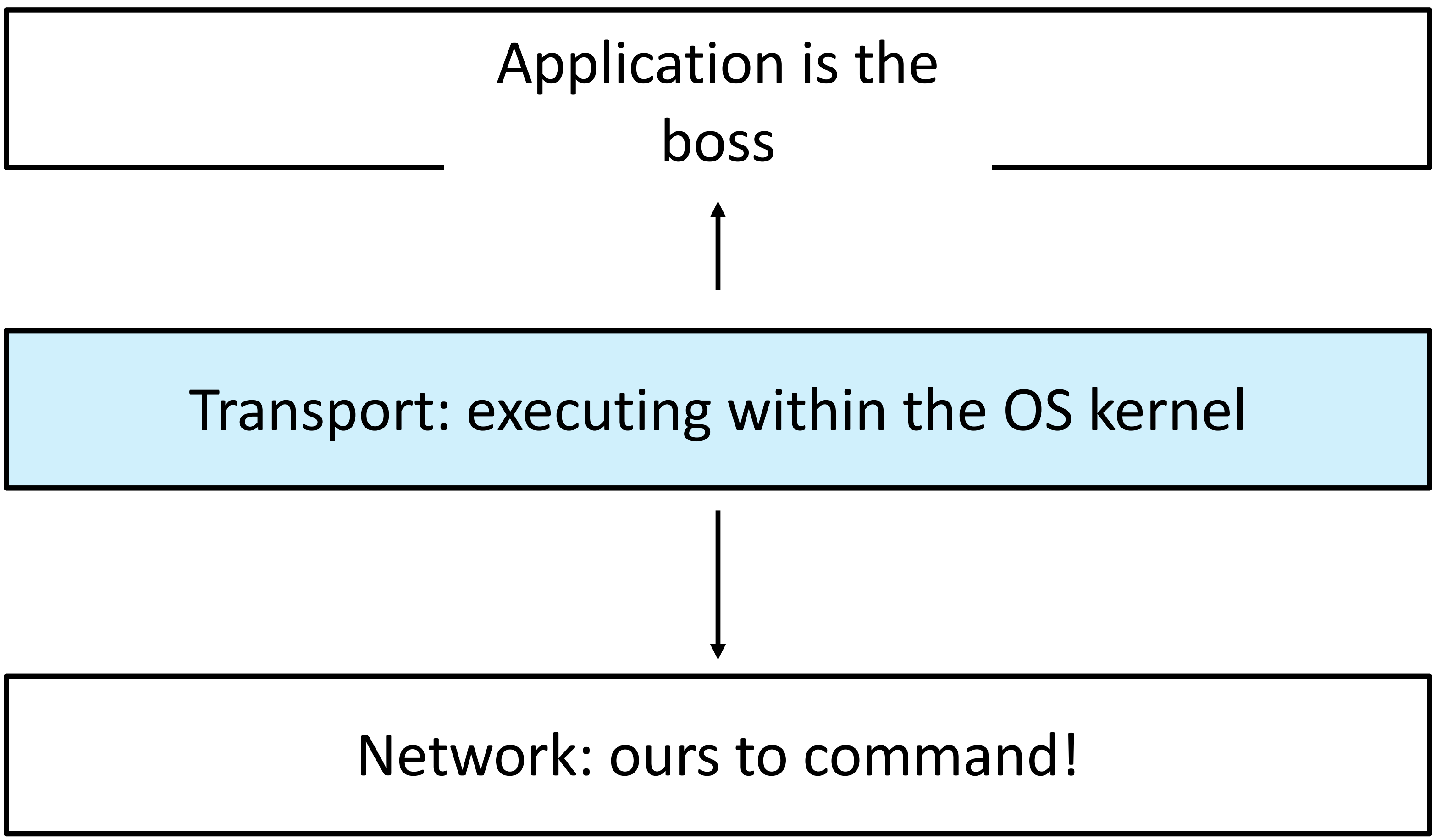
Transport: end-to-end connections, reliability

Network: routing

Link (data-link): framing, error detection

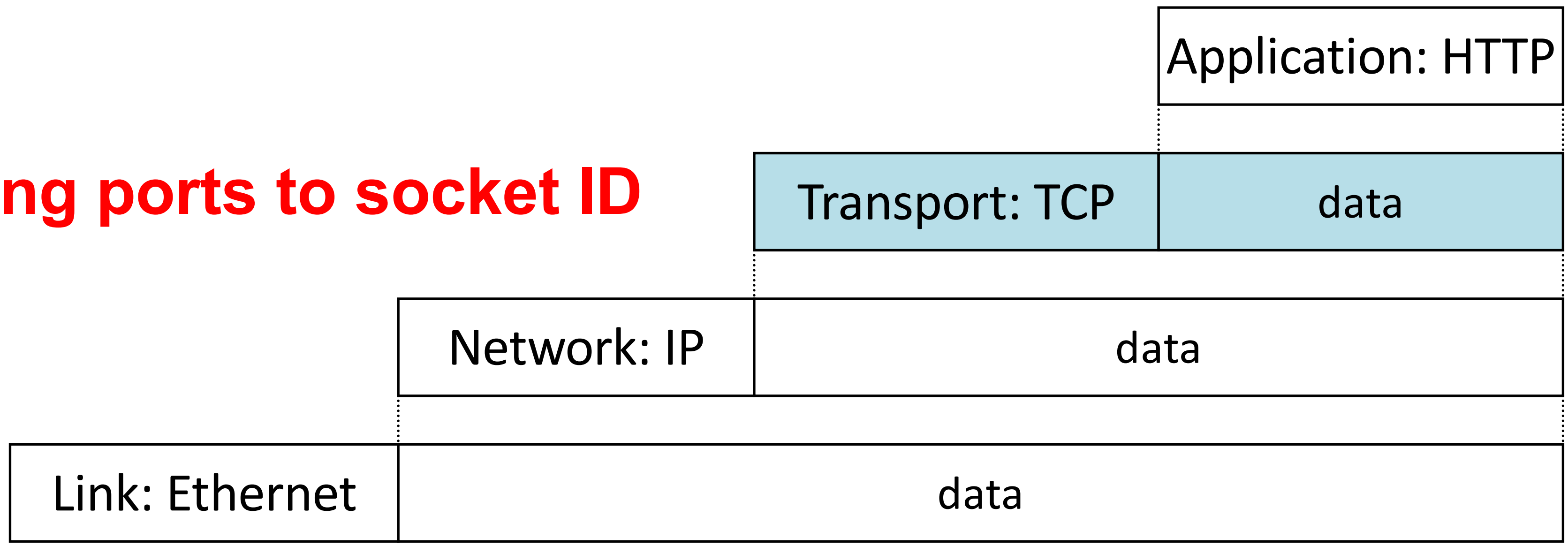
Physical: 1's and 0's/bits across a medium
(copper, the air, fiber)

Transport Layer perspective

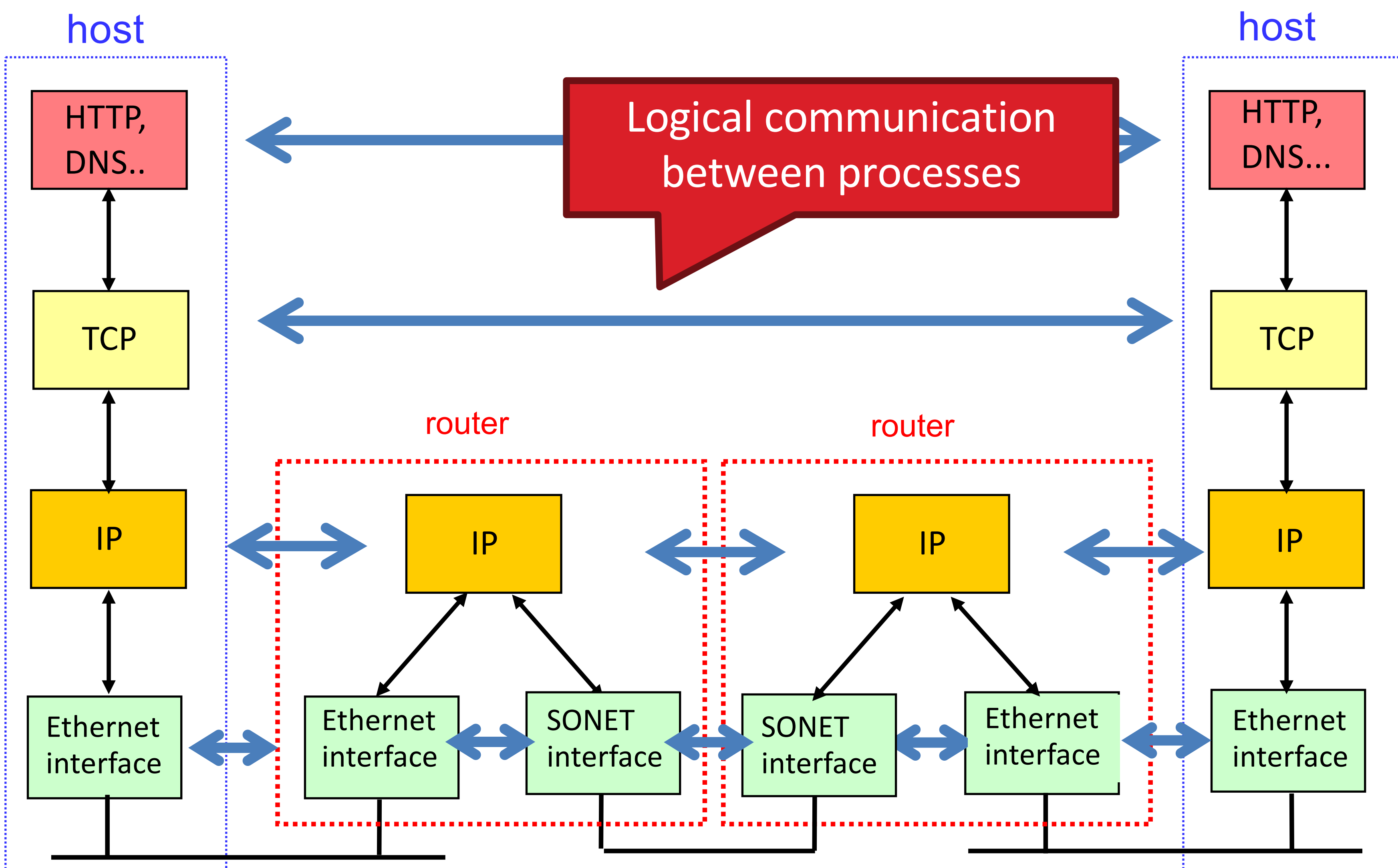


Transport Layer Header

Assigning ports to socket ID

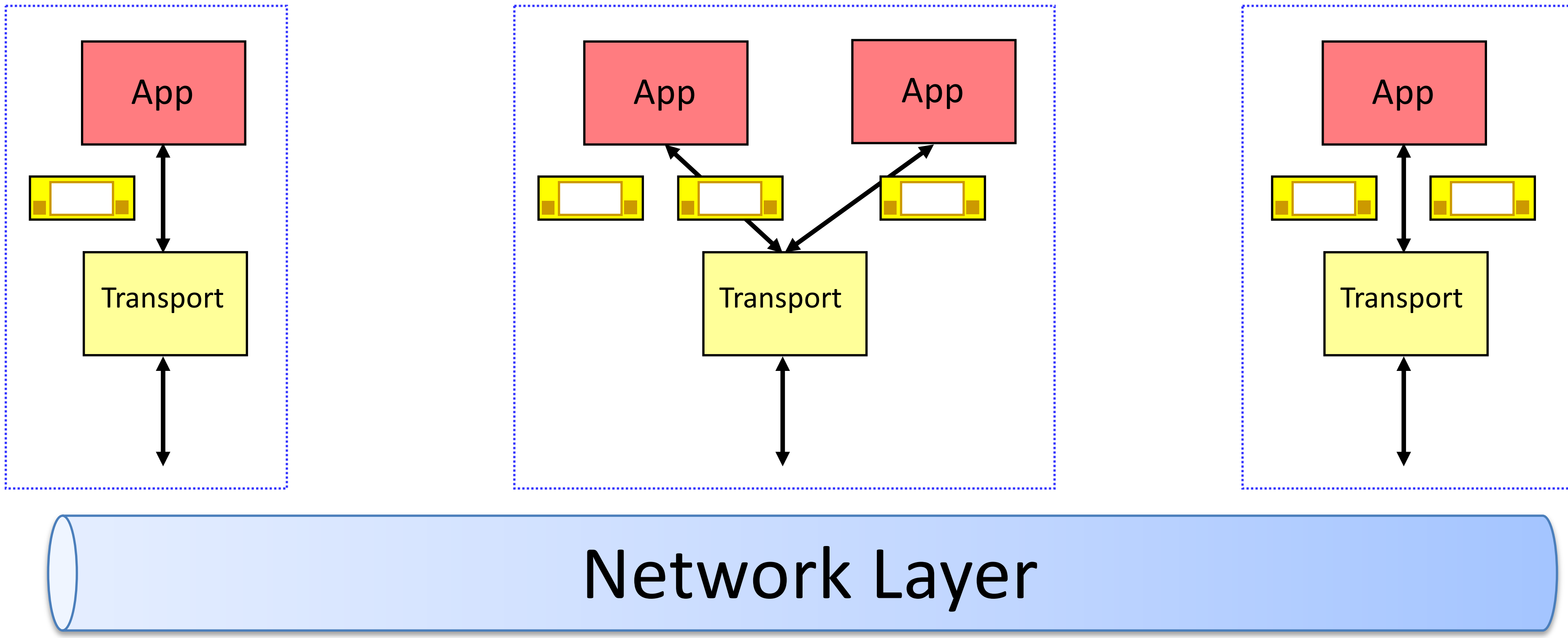


Transport Layer: Runs on end systems



Last Class: Multiplexing/De-Multiplexing

(Simultaneous transmission of two or more signals/messages over a single channel.)



- The network is a shared resource.
  – It does NOT care about your applications, sockets, etc.

- Senders mark segments, in header, with identifier (port)

How many of these services might we provide at the transport layer? Which?

- Reliable transfers
- Error detection
- Error correction
- Bandwidth guarantees
- Latency guarantees
- Encryption
- Message ordering
- Link sharing fairness

A. 4 or fewer
B. 5
C. 6
D. 7
E. All 8

How many of these services might we provide at the transport layer? Which?

- Reliable transfers (T)
- Error detection (U, T)
- Error correction (T)
- Bandwidth guarantees
- Latency guarantees
- Encryption
- Message ordering (T)
- Link sharing fairness (T)

Critical question: Can it be done at the end host?

A. 4 or fewer
B. 5
C. 6
D. 7
E. All 8

TCP sounds great! UDP…meh. Why do we need it?

A. It has good performance characteristics.

B. Sometimes all we need is error detection.

C. We still need to distinguish between sockets.

D. It basically just fills a gap in our layering model.

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UDP – User Datagram Protocol

- Unreliable, unordered service
- Adds:
  - **end points identified by ports**
  - multiplexing
  - checksum (error detection)

UDP: User Datagram Protocol [RFC 768]

- “No frills,” “Bare bones” Internet transport protocol
  - RFC 768 (1980)
  - Length of the document?

UDP: User Datagram Protocol [RFC 768]

¯_(ツ)_/¯

- "Best effort" service,
- UDP segments may be:
  - Lost
  - Delivered out of order (same as underlying network layer)

How many of the following steps does UDP implement? (which ones?)

1. exchange an initiate handshake (connection setup)
2. break up packet into segments at the source and number them
3. place segments in order at the destination
4. error-checking with checksum

How many of the following steps does UDP implement? (which ones?)

1. exchange an initiate handshake (connection setup)
2. break up packet into segments at the source and number them
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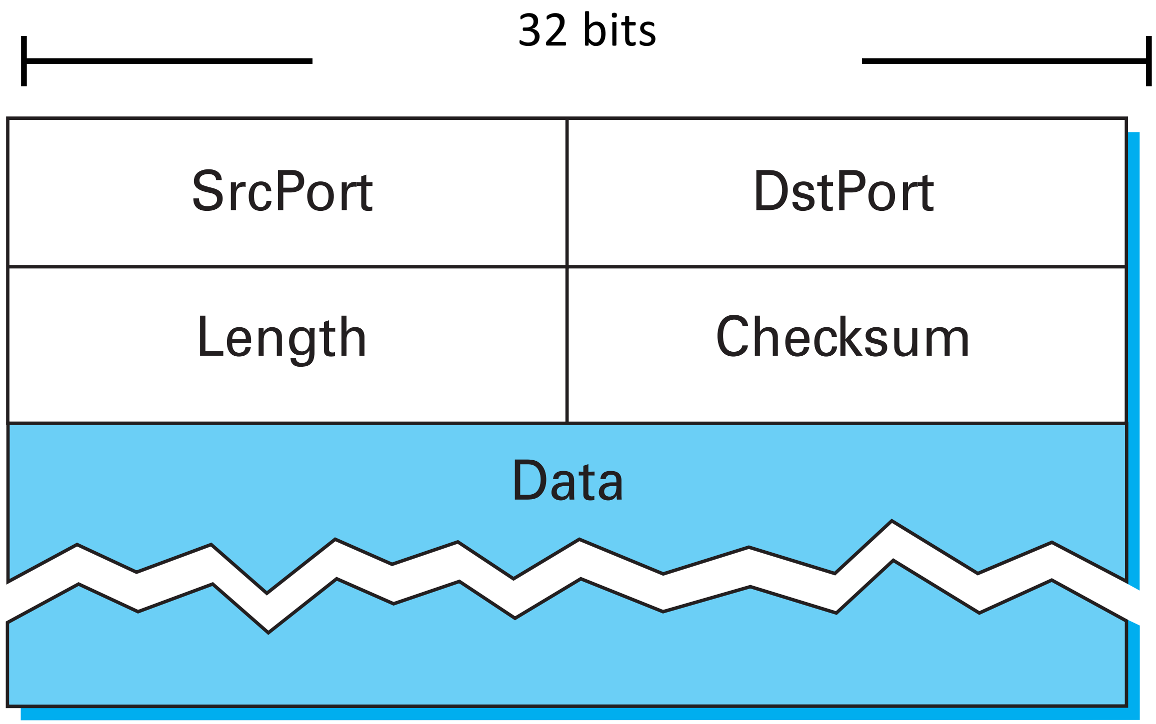
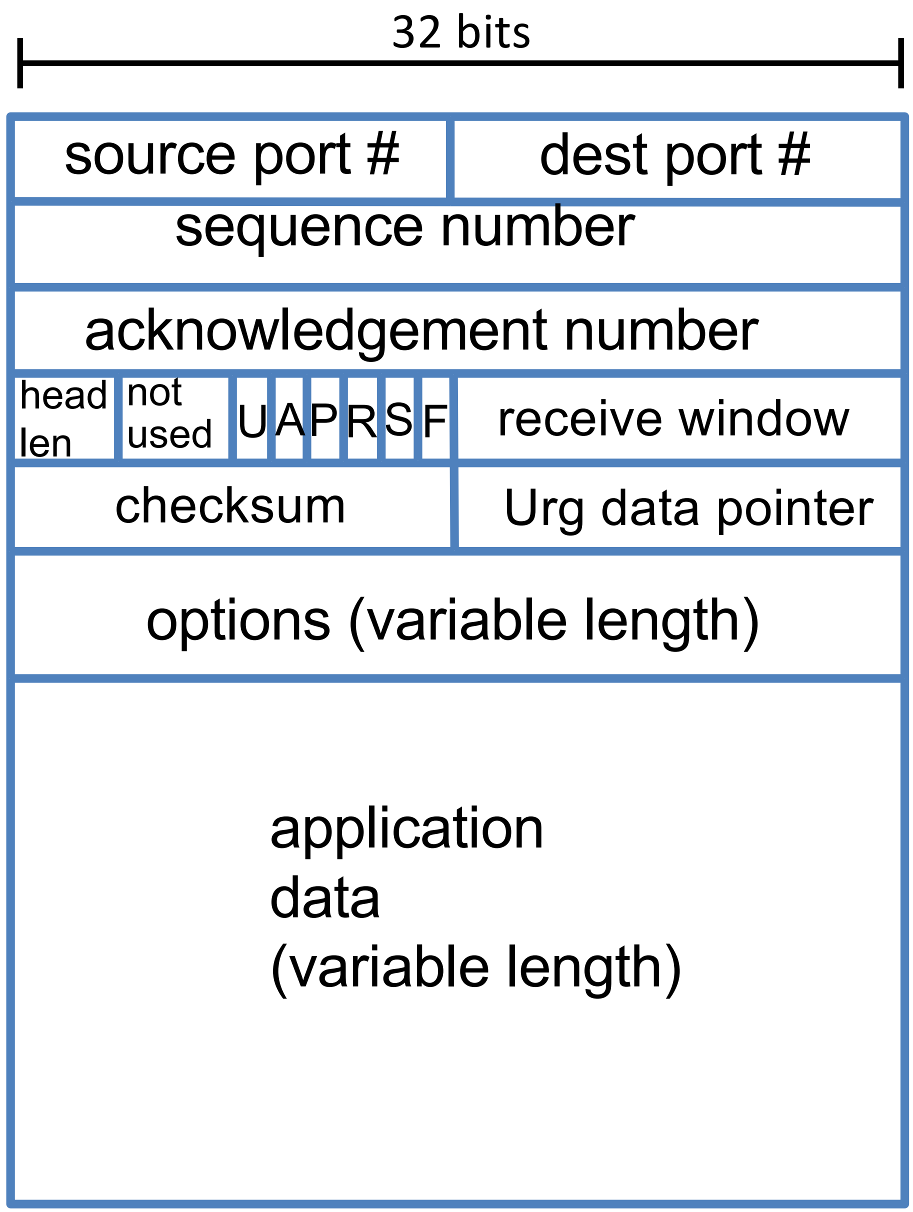
UDP Segment

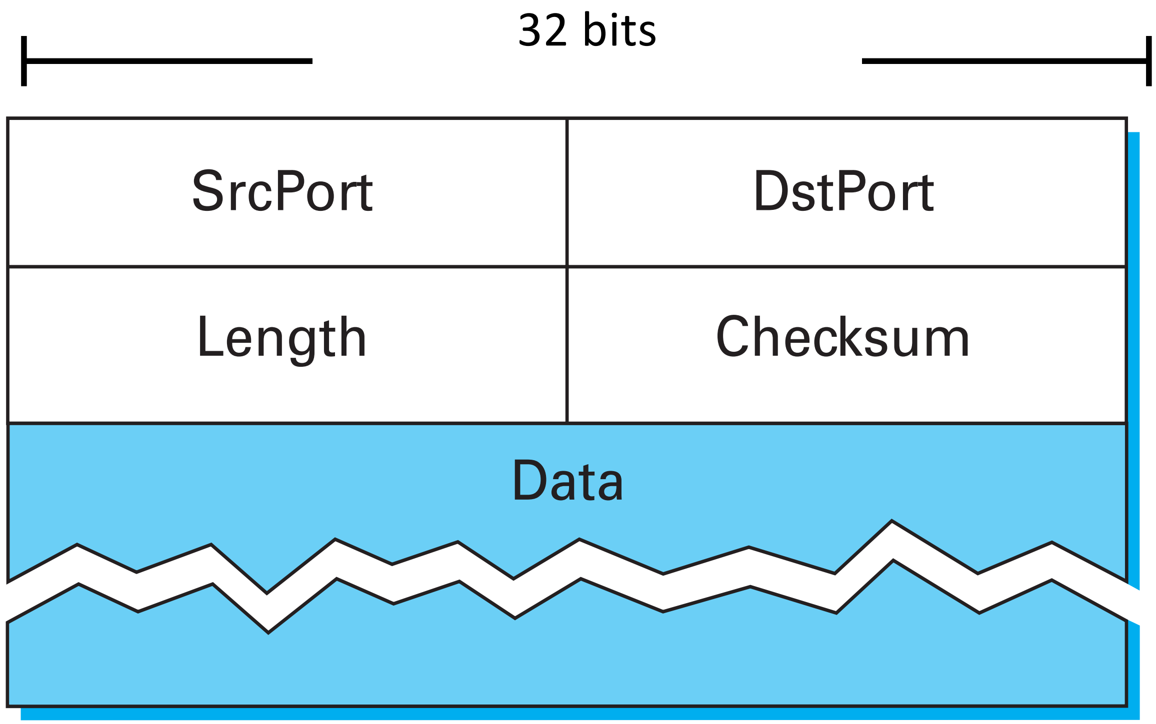


TCP Segment!



32 bits
source port #
dest port #
sequence number
acknowledgement number
head len
not used
U
A
P
R
S
F
receive window
checksum
Urg data pointer
options (variable length)
application data (variable length)

UDP Segment



UDP Checksum

- Goal: Detect transmission errors (e.g. flipped bits)
  - Router memory errors
  - Driver bugs
  - Electromagnetic interference

UDP Checksum at the Sender

- Treat the entire segment as 16-bit integer values
- Add them all together (sum)
- Put the 1's complement in the checksum header field

One's Compliment

- In bitwise compliment, all of the bits in a binary number are flipped.
- So 1111000011110000 -> 0000111100001111

Checksum example

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

Receiver

- Add all the received data together as 16-bit integers
- Add that to the checksum
- If result is not 1111 1111 1111 1111, there are errors!

If our checksum addition yields all ones, are we guaranteed to be error-free?

A. Yes

B. No

If our checksum addition yields all ones, are we guaranteed to be error-free?

A. Yes

B. No: we could have two bit flips that cancel each other out.

UDP Applications

- Latency sensitive
  - Quick request/response (DNS)
  - Network management (SNMP, TFTP)
  - Voice/video chat
- Error correction unnecessary (periodic msgs)
- Communicating with lots of others

What if you want something more reliable than UDP, but faster/not as full featured as TCP?

A. Sorry, you're out of luck.

B. Write your own transport protocol.

C. Add in the features you want at the application layer.

What if you want something more reliable than UDP, but faster/not as full featured as TCP?

Write your own transport protocol:

i. more control: OS controls scheduling, prioritization

ii. reusability: all application layer protocols can use it – great if you own a datacenter (e.g. Google) and you can run your own environment.

What if you want something more reliable than UDP, but faster/not as full featured as TCP?

Add in the features you want at the application layer.

i. easier than modifying the OS.
ii. write into your application custom features on top of UDP that are specific to your application
iii. don’t overburden the transport layer

Irrespective of which layer we choose, adopting something new is always hard on the Internet!

TCP: send() Blocking

- Recall: With TCP, send() blocks if buffer full.

UDP sendto() blocking

With TCP, send() blocks if buffer full.

- Does UDP need to block? Should it?

A. Yes, if buffers are full, it should.
B. It doesn't need to, but it might be useful.
C. No, it does not need to and shouldn't do so.

UDP sendto() blocking

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Summary

Transport Layer:

- Provides a logical communication between processes/ applications
- packets are called segments at the transport layer
- Transport layer protocol: responsible for adding port numbers (mux/demux segments)

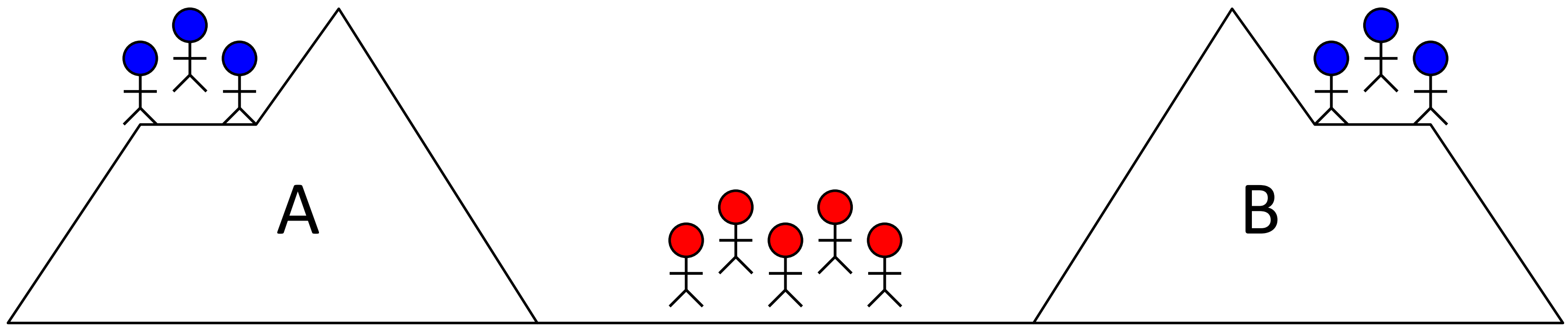
UDP:

- No “frills” protocol
- No state maintained about the packet
- Checksum (1’s complement) over IP + UDP + payload.
  - can only correct for 1 bit errors.
- adds port numbers over unreliable network (best effort)
- applications:
  - latency sensitive applications: real-time audio, video
  - communicating with a lot of end-hosts (like DNS)
- UDP Sockets:
  - do not need to be implemented as blocking system calls for correctness since the only guarantee UDP makes is best-effort delivery.
  - however send/recv can be implemented as blocking system calls depending on the application

Today

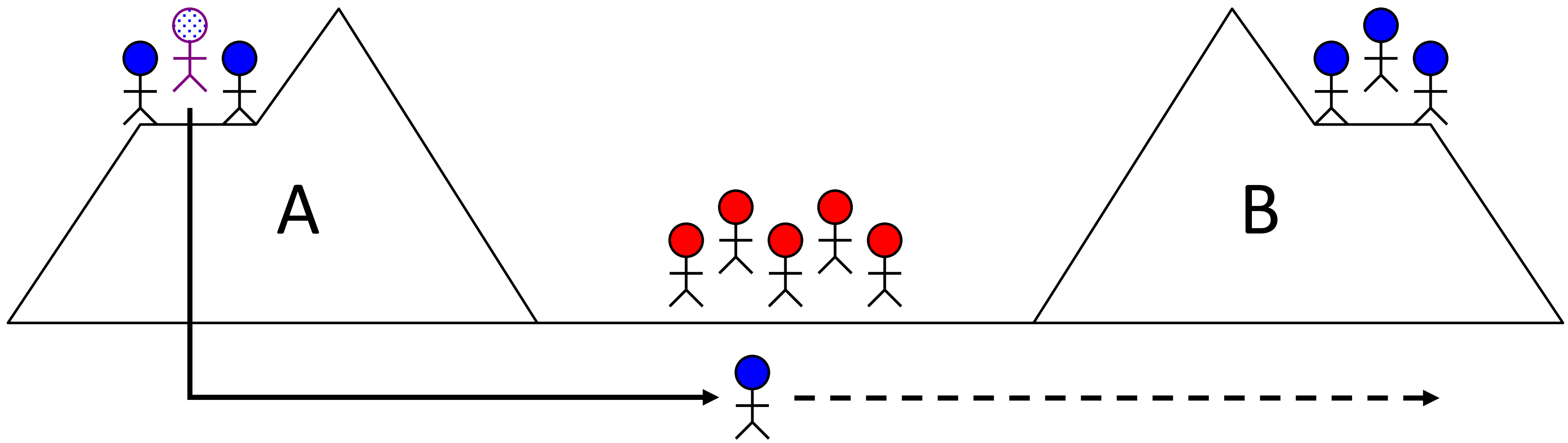
- Principles of reliability
  - The Two Generals Problem
- Automatic Repeat Requests
  - Stop and Wait
  - Timeouts and Losses
  - Pipelined Transmission

The Two Generals Problem



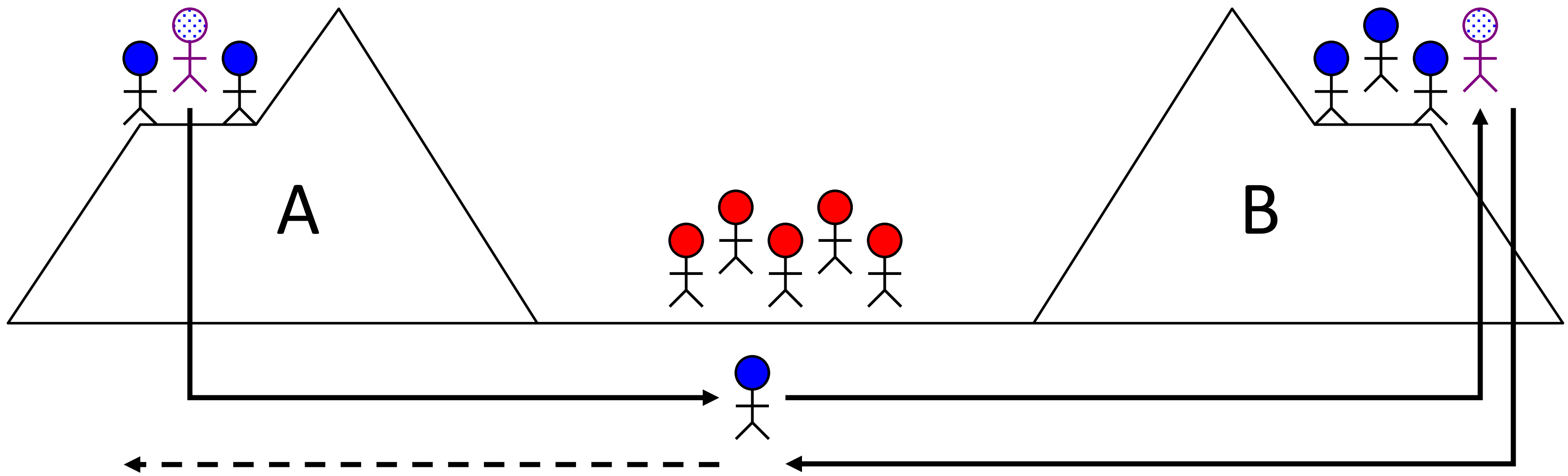
- Two army divisions (blue) surround enemy (red)
  - Each division led by a general
  - Both must agree when to simultaneously attack
  - If either side attacks alone, defeat
- Generals can only communicate via messengers
  - Messengers may get captured (unreliable channel)

The Two Generals Problem



- How to coordinate?
  - Send messenger: “Attack at dawn”
  - What if messenger doesn’t make it?

The Two Generals Problem

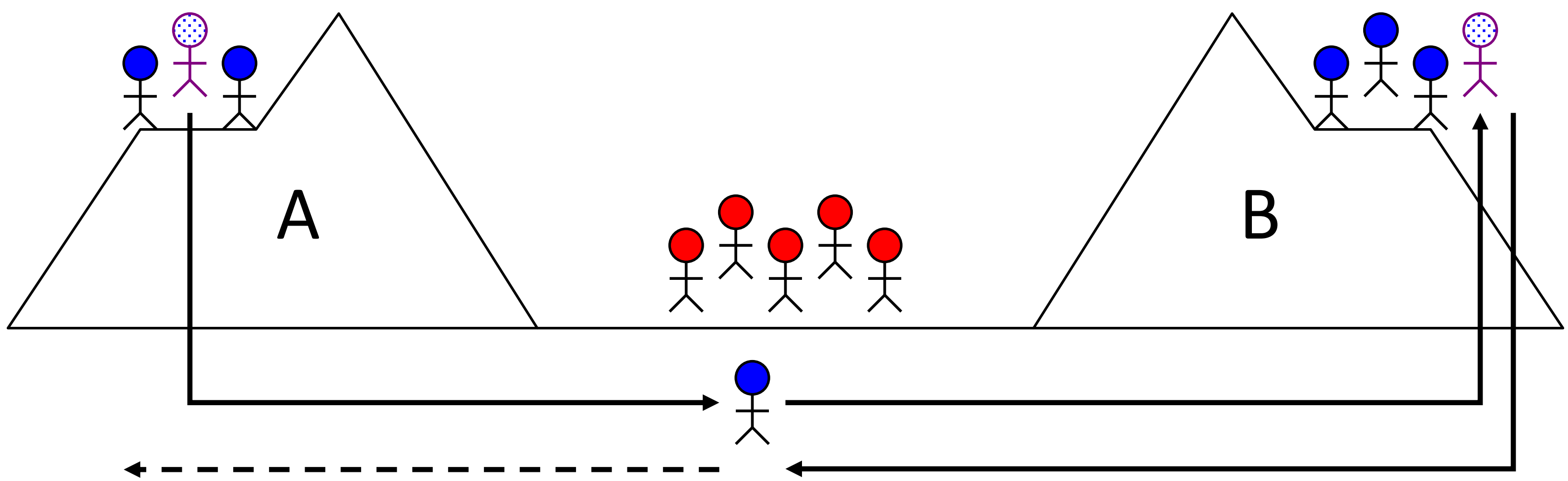


- How to be sure messenger made it?
  - Send acknowledgment: “I delivered message”

In the “two generals problem”, can the two armies reliably coordinate their attack? (using what we just discussed)

- A. Yes (explain how)
- B. No (explain why not)

The Two Generals Problem



- Result
  - Can't create perfect channel out of faulty one
  - <u>Can only increase probability of success</u>

Give up? No way!

As humans, we like to face difficult problems.

- We can’t control oceans, but we can build canals
- We can’t fly, but we’ve landed on the moon
- We just need engineering!

What can possibly go wrong....

Engineering

- Concerns
  - Message corruption
  - Message duplication
  - Message loss
  - Message reordering
  - Performance
- Our toolbox
  - Checksums
  - Timeouts
  - Acks & Nacks
  - Sequence numbering
  - Pipelining

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We use these to build Automatic Repeat Request (ARQ) protocols.

(We'll briefly talk about alternatives at the end.)

Automatic Repeat Request (ARQ)

- Intuitively, ARQ protocols act like you would when using a cell phone with bad reception.
  - Receiver: Message garbled? Ask to repeat.
  - Sender: Didn’t hear a response? Speak again.
- Refer to book for building state machines.
  - We’ll look at TCP’s states soon

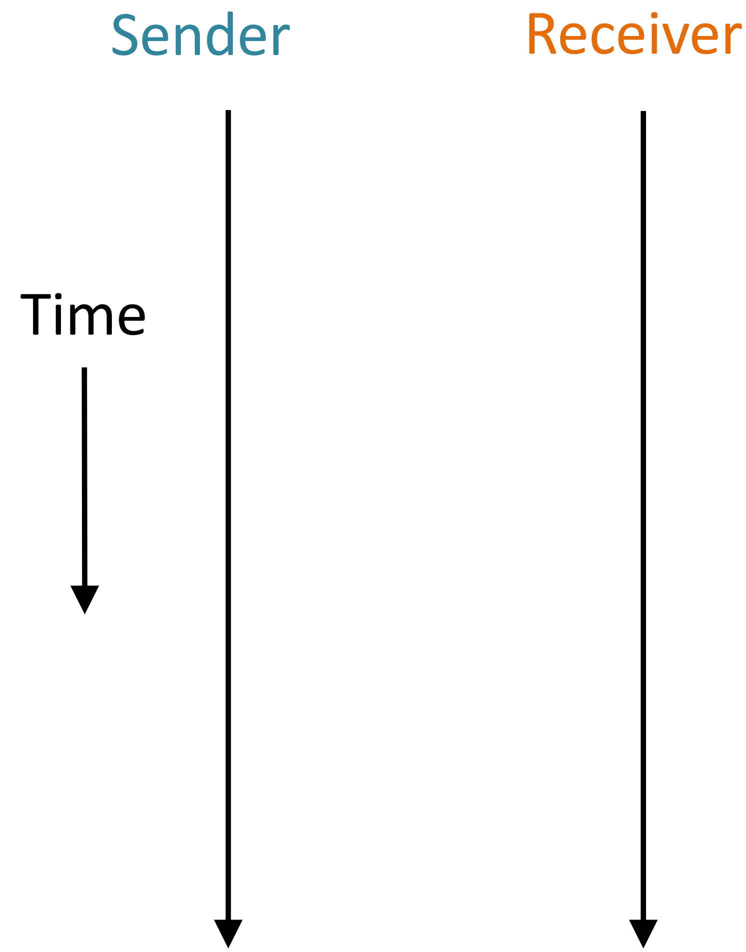
ARQ Broad Classifications

1. Stop-and-wait

Stop and Wait

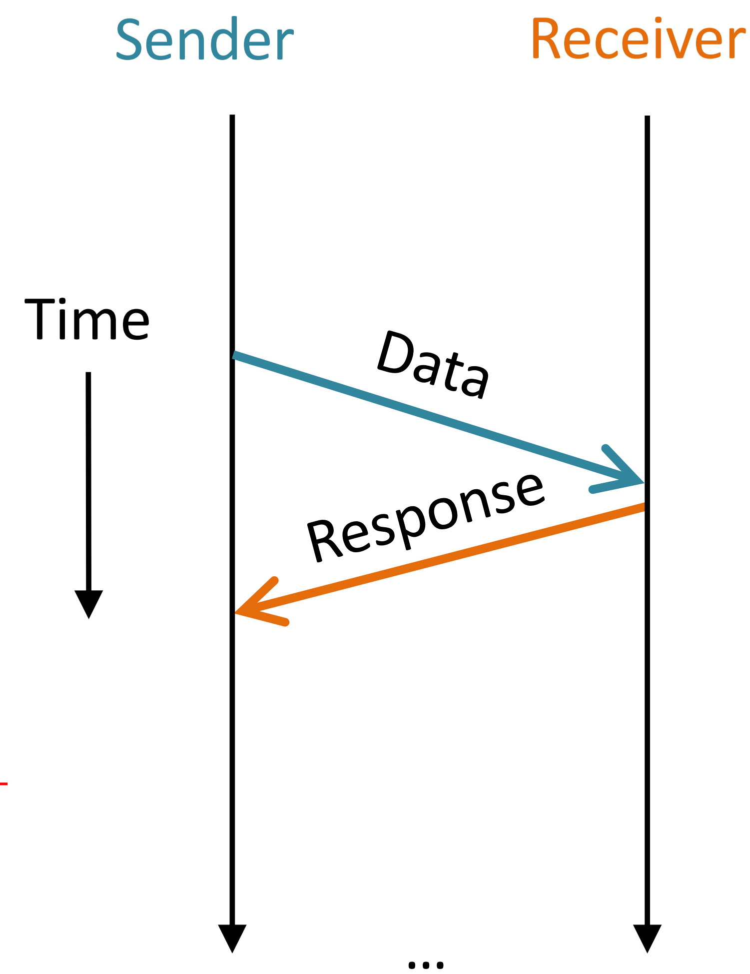
We have:

- a sender
- a receiver
- time: represented by downwards arrow



Stop and Wait

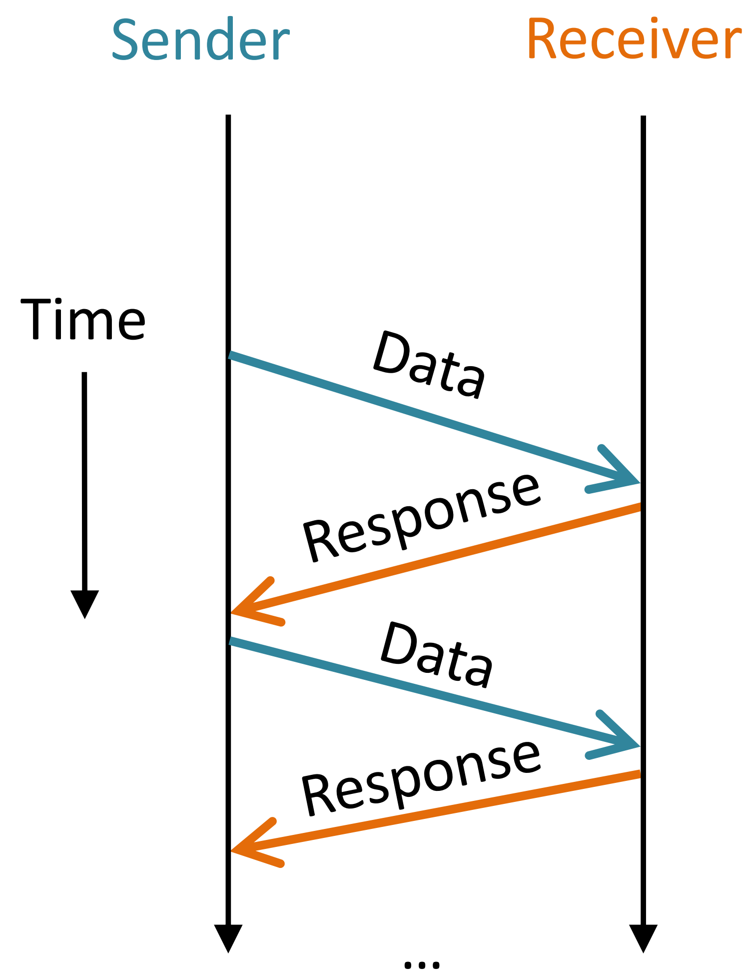
Sender sends data and waits till they get the response message from the receiver.



Buffer data, and don't send till response received

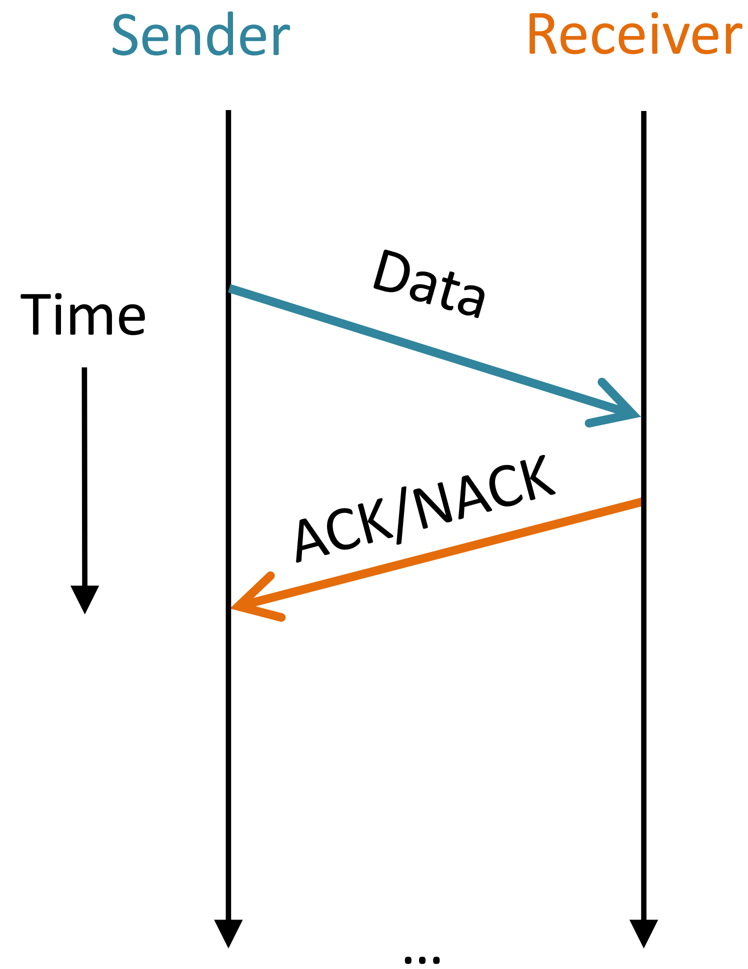
Stop and Wait

- Up next: concrete problems and mechanisms to solve them.
- These mechanisms will build upon each other
- Questions?



Corruption?

- Error detection mechanism: checksum
  - Data good – receiver sends back ACK
  - Data corrupt – receiver sends back NACK

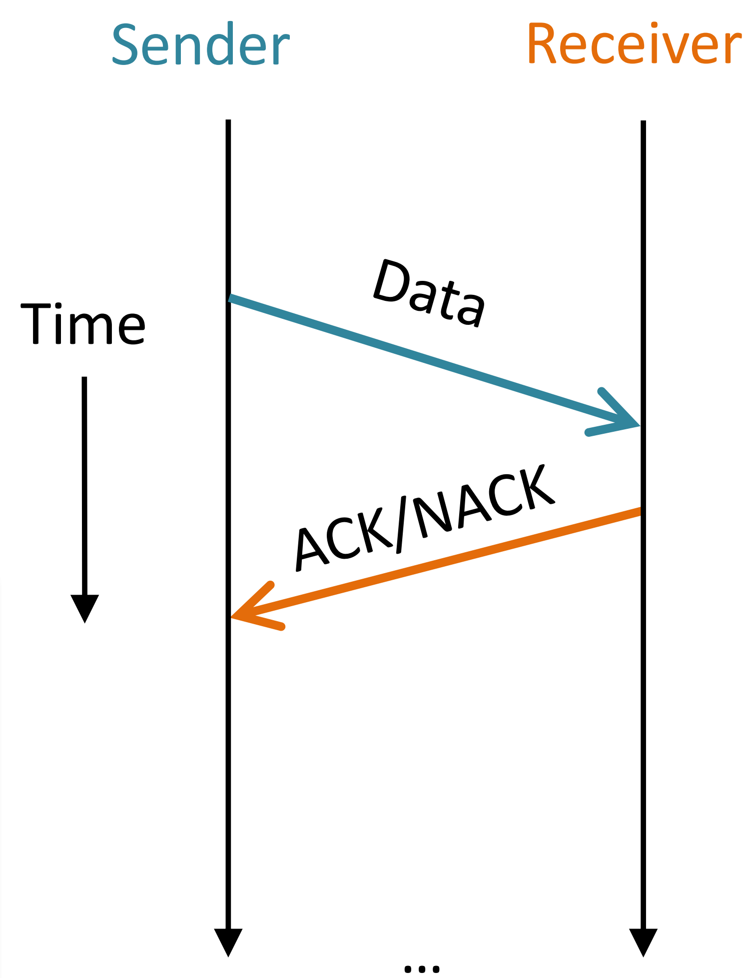


Could we do this with just ACKs or just NACKs?

Error detection mechanism: checksum

- Data good – receiver sends back ACK
- Data corrupt – receiver sends back NACK

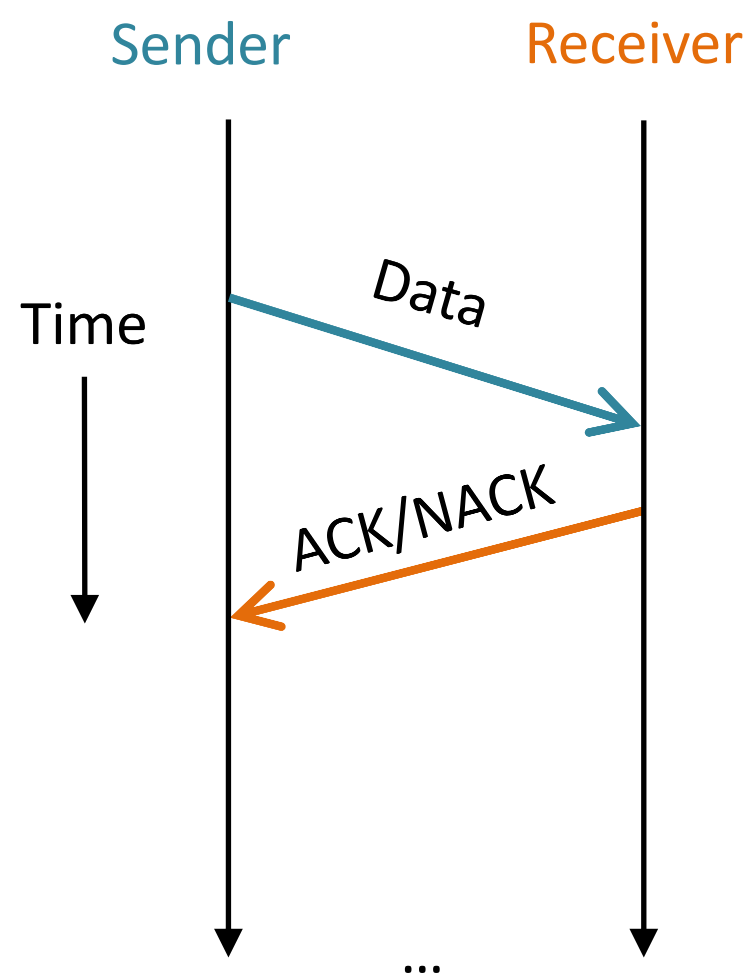
A. No, we need them both.
B. Yes, we could do without one of them, but we'd need some other mechanism.
C. Yes, we could get by without one of them.



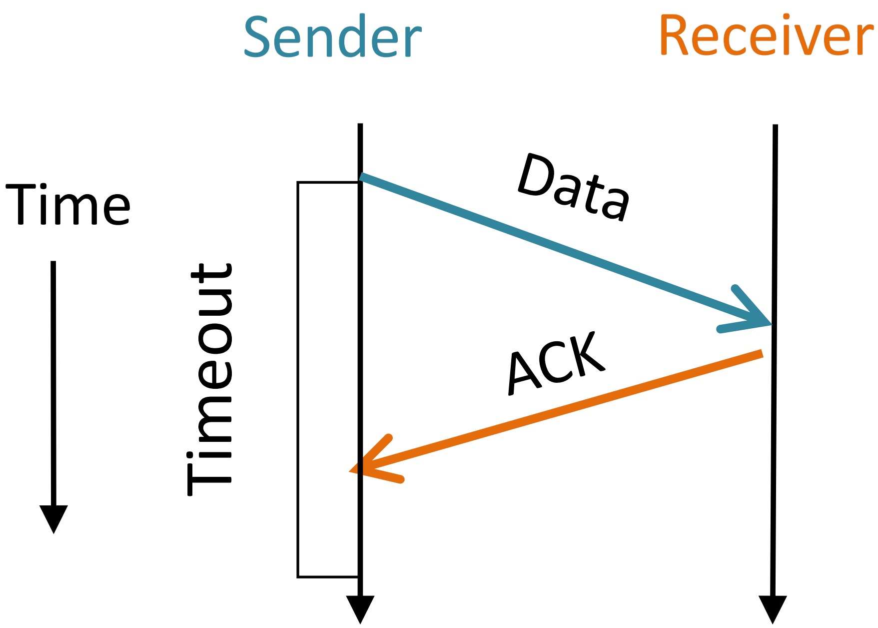
Could we do this with just ACKs or just NACKs?

- With only ACK, we could get by with a timeout.
- With only NACK, we couldn't advance (no good).

A. No, we need them both.
B. Yes, we could do without one of them, but we'd need some other mechanism.
C. Yes, we could get by without one of them.

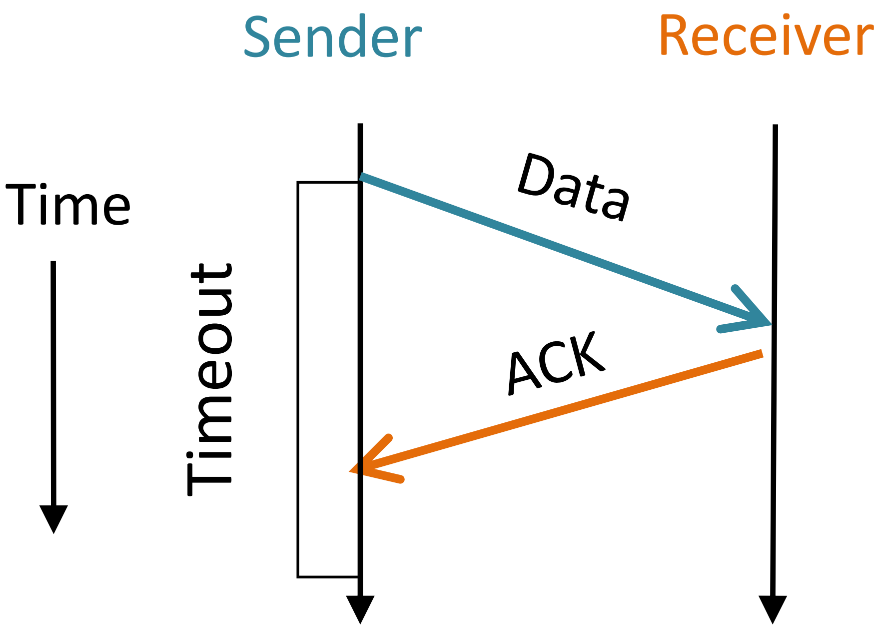


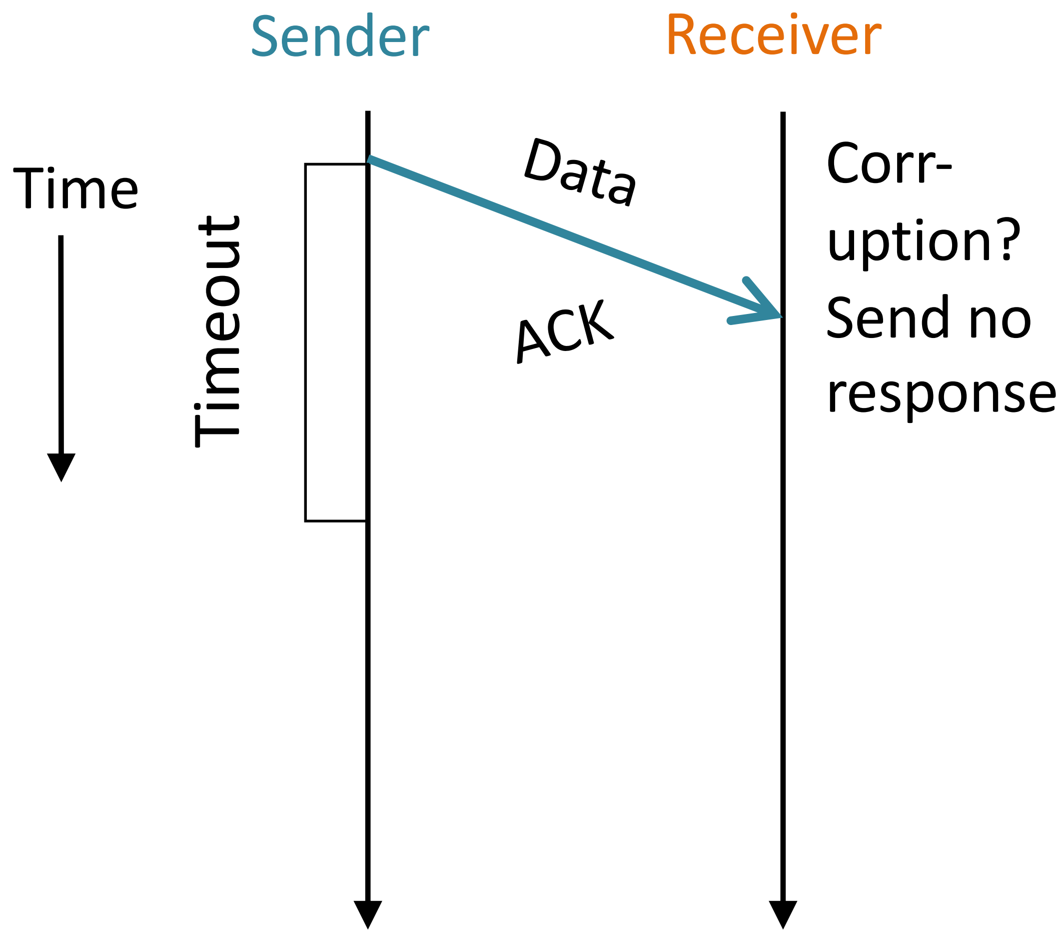
Timeouts and Losses



- Sender starts a clock. If no response, retry.

Timeouts and Losses





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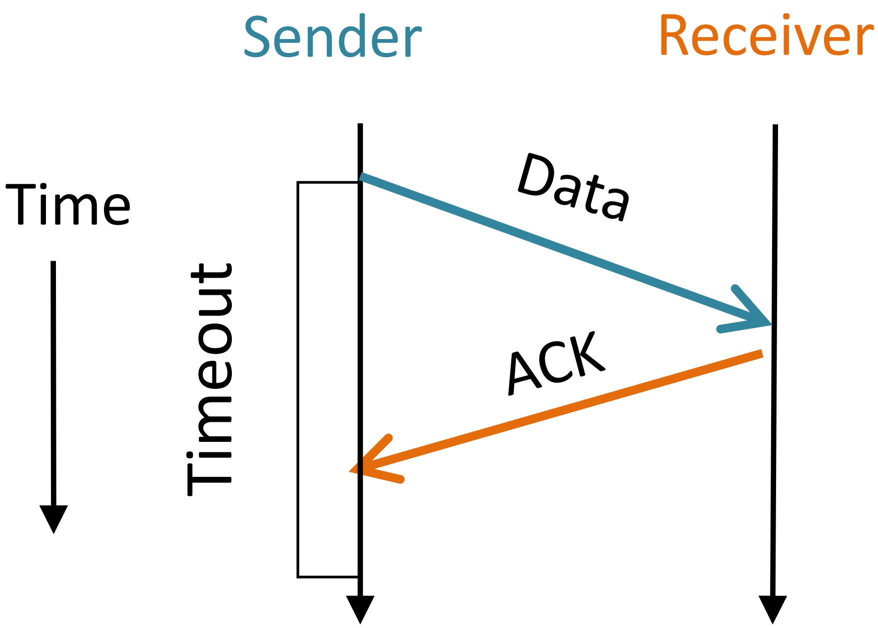
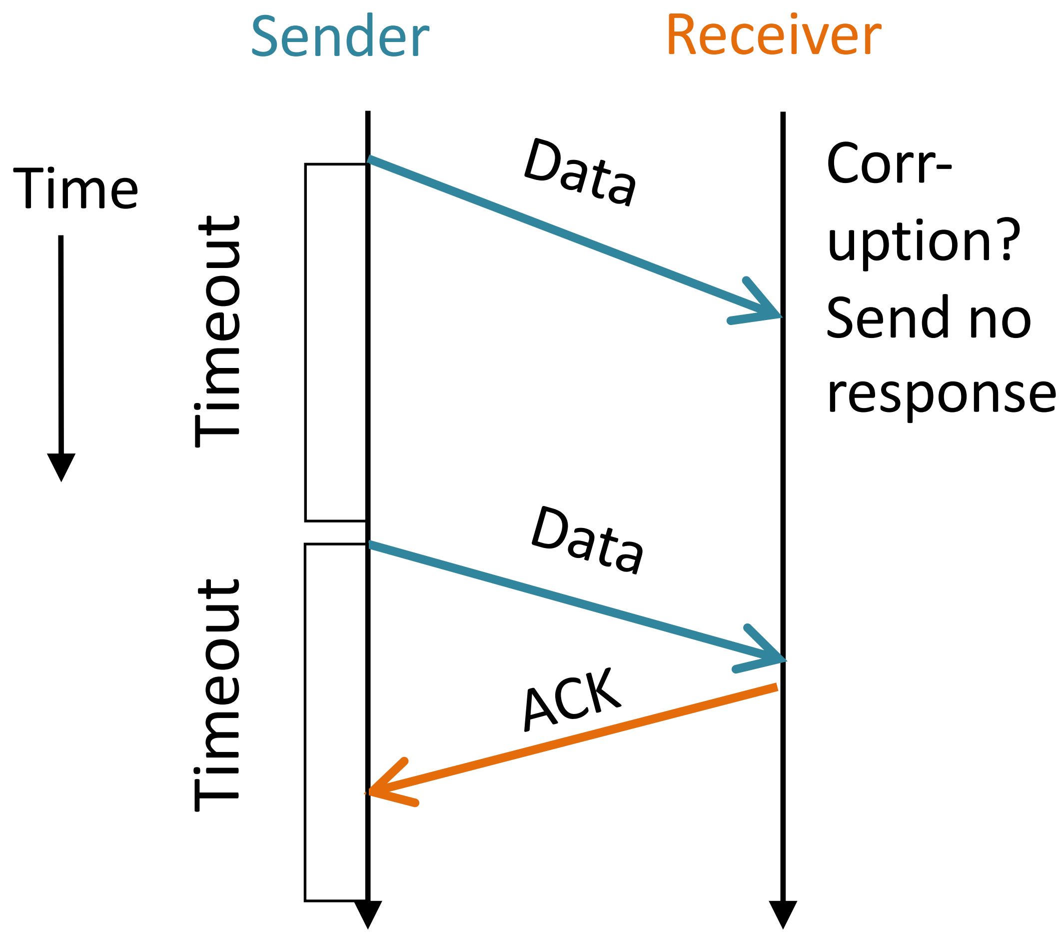
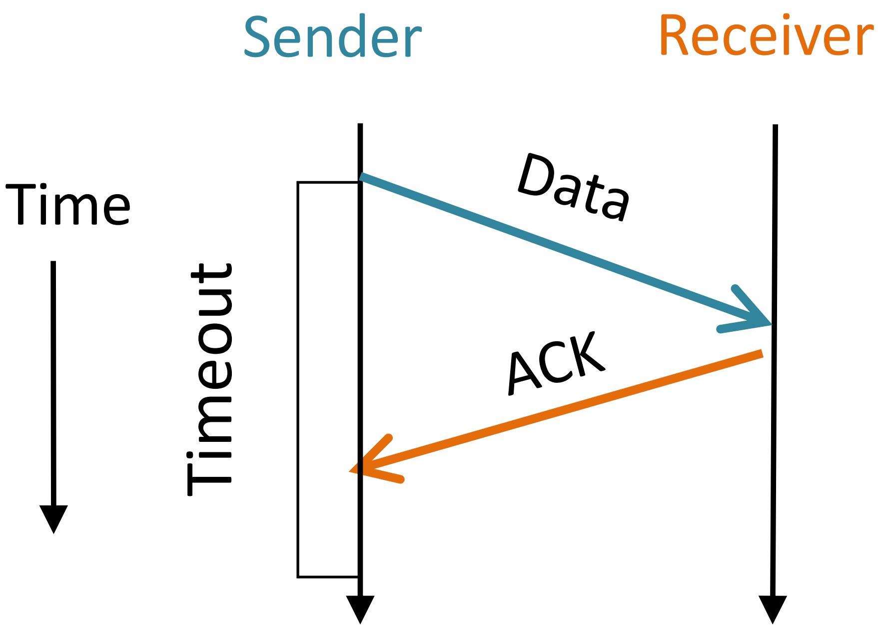
Timeouts and Losses





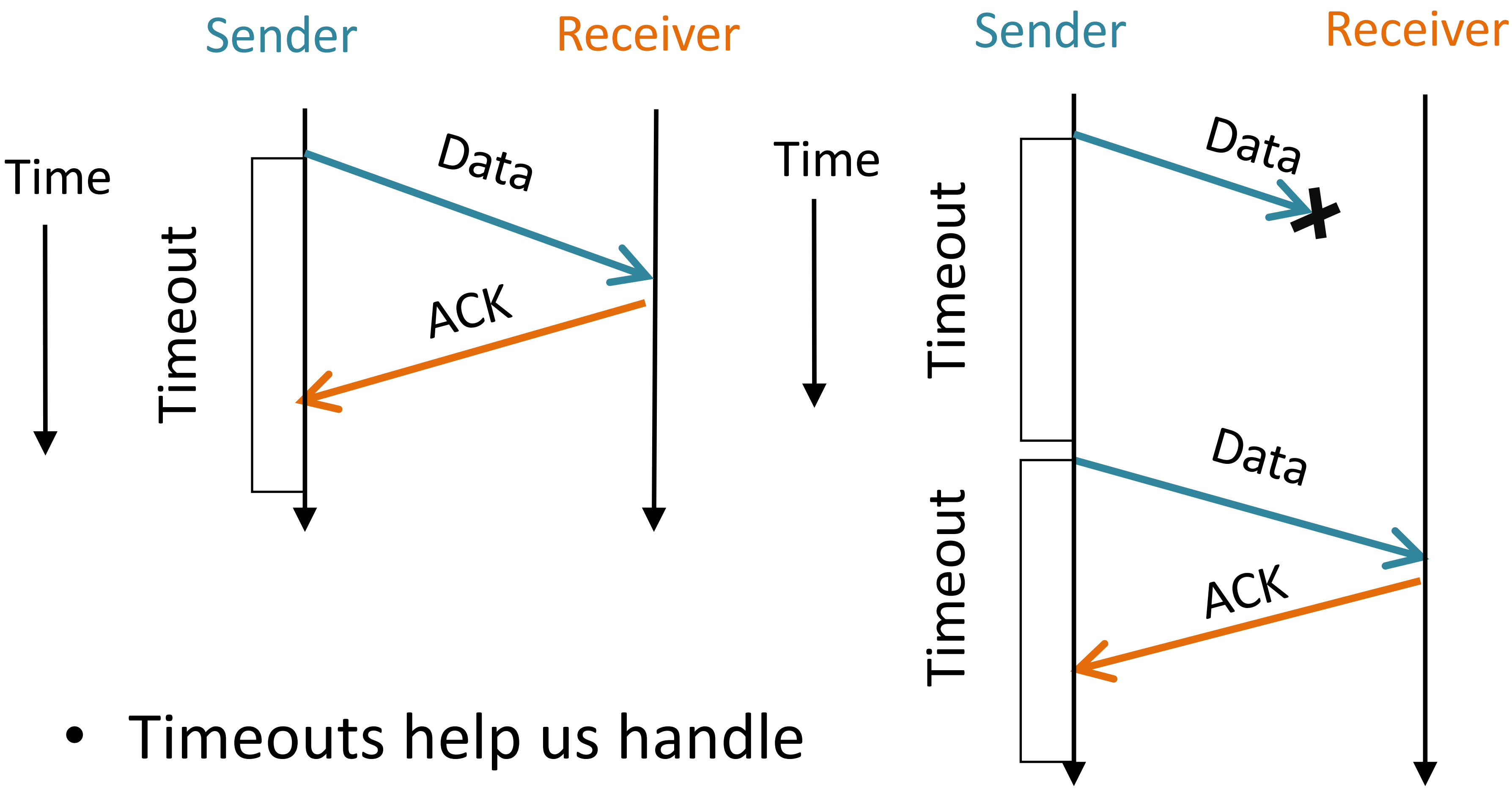
- Sender starts a clock. If no response, retry.
- Probably not a great idea for handling corruption, but it works.

Timeouts and Losses



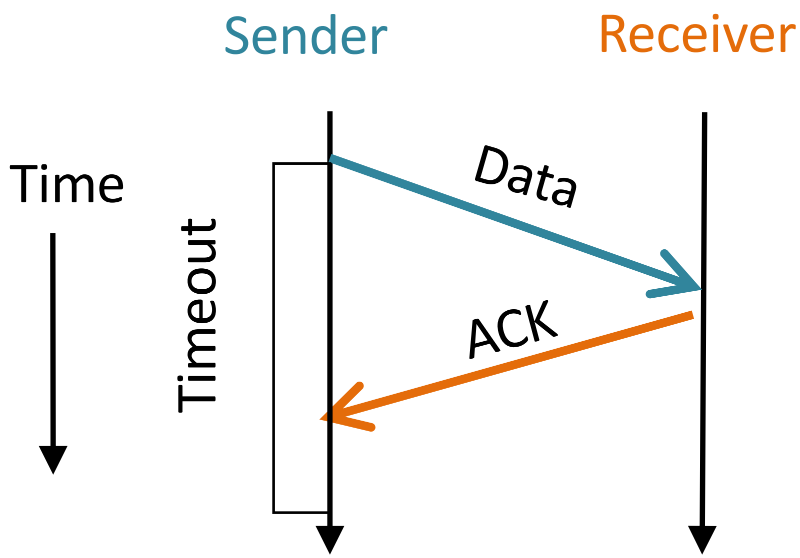
- Timeouts help us handle message losses too!

Timeouts and Losses



- Timeouts help us handle message losses too!

Adding timeouts might create new problems for us to worry about. How many? Examples?



A. No new problems (why not?)
B. One new problem (which is..)
C. Two new problems (which are..)
D. More than two new problems (which are..)

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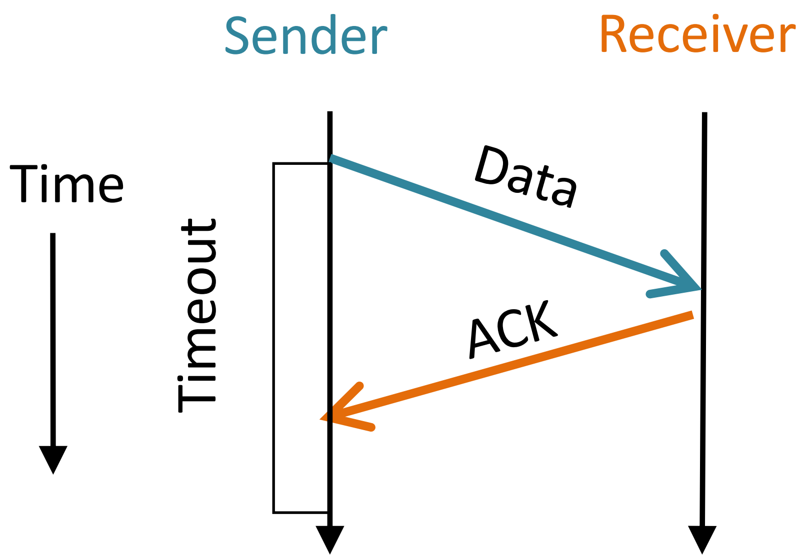

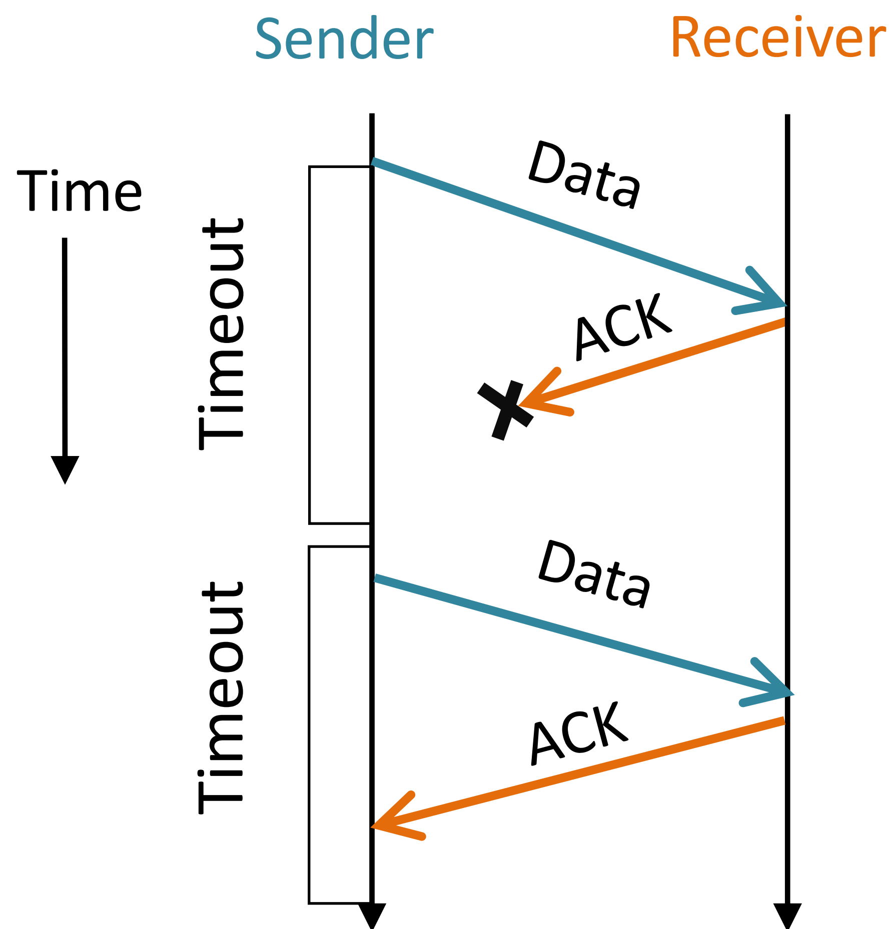

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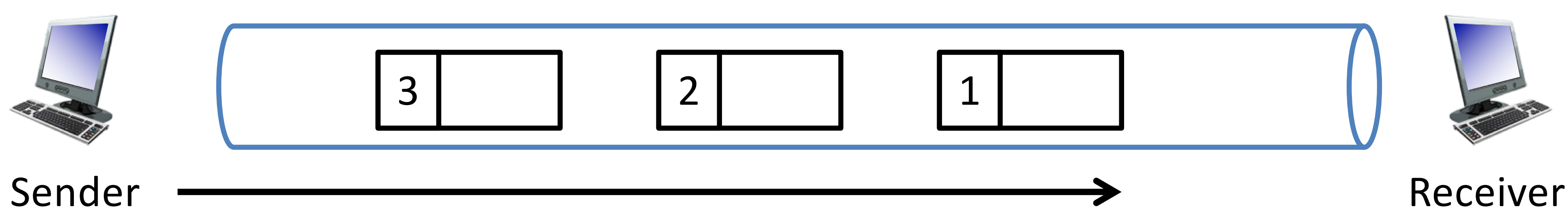
Sequence Numbering

Sender

- Add a monotonically increasing label to each msg

Receiver

- Ignore messages with numbers we've seen before
- When pipelining (a few slides from now)
  - Detect gaps in the sequence (e.g., 1,2,4,5)



What is our link utilization with a stop-and-wait protocol?

System parameters:

Link rate: 8 Mbps (one megabyte per second)

RTT: 100 milliseconds

Segment size: 1024 bytes

A. < 0.1 %
B. ≈ 0.1 %
C. ≈ 1 %
D. 1-10 %
E. > 10 %

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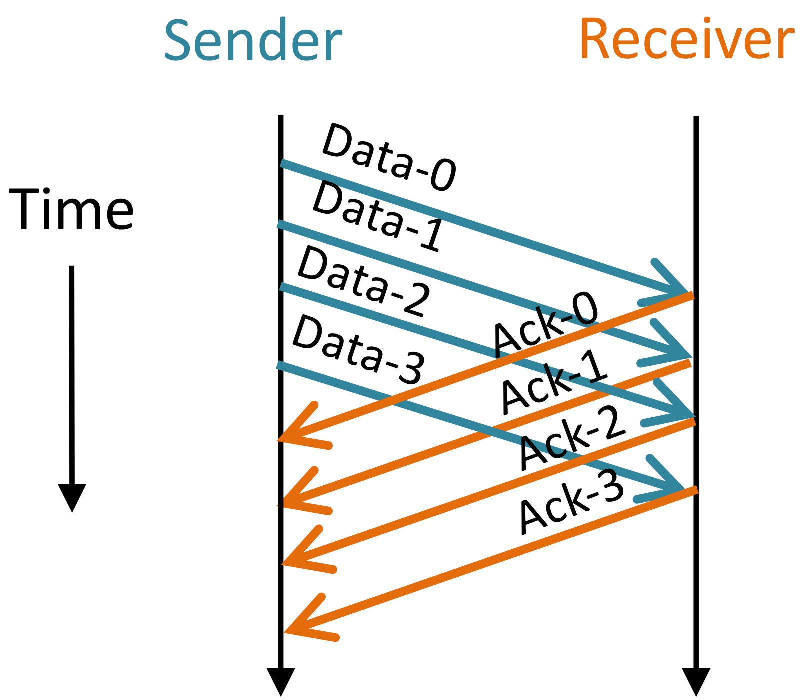
Big Problem: Performance is determined by RTT, not channel capacity!

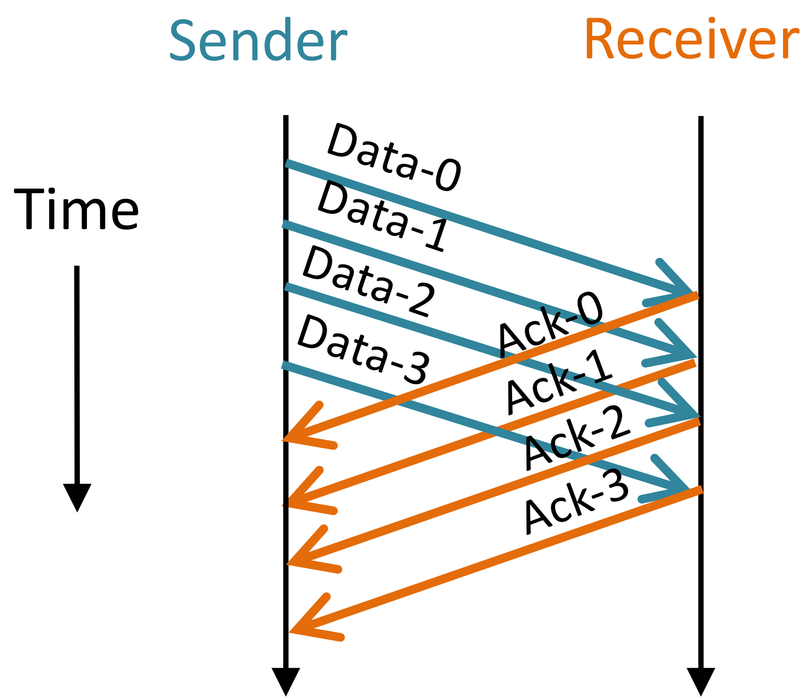
Pipelined Transmission

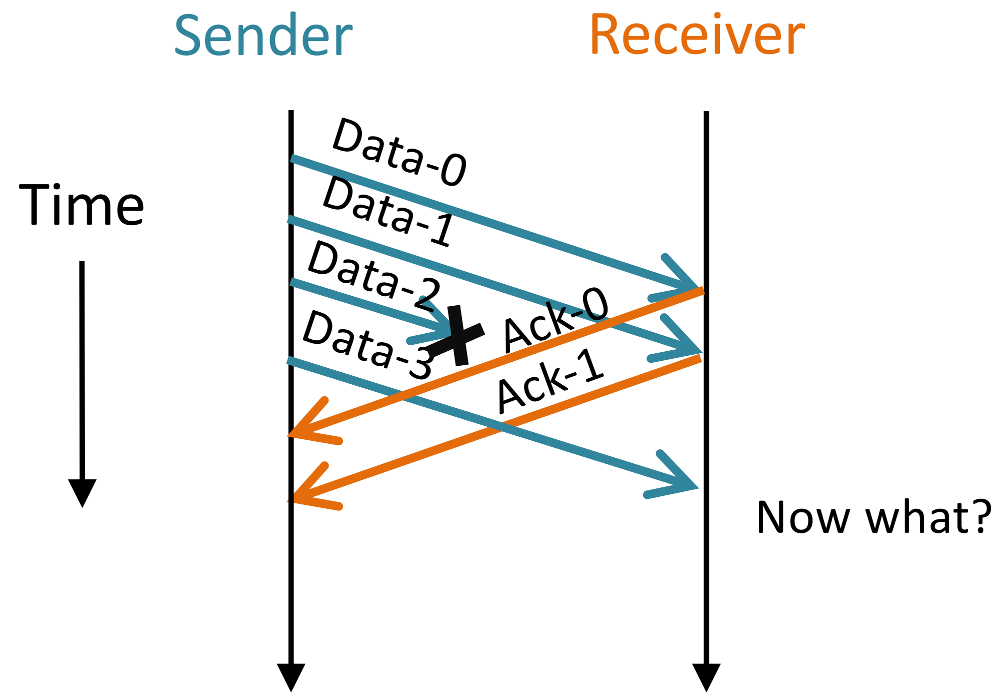


Keep multiple segments "in flight"

- Allows sender to make efficient use of the link
- Sequence numbers ensure receiver can distinguish segments
- We'll talk about "how many" next time (windowing).

Pipelined Transmission





Keep multiple segments "in flight"

- Allows sender to make efficient use of the link
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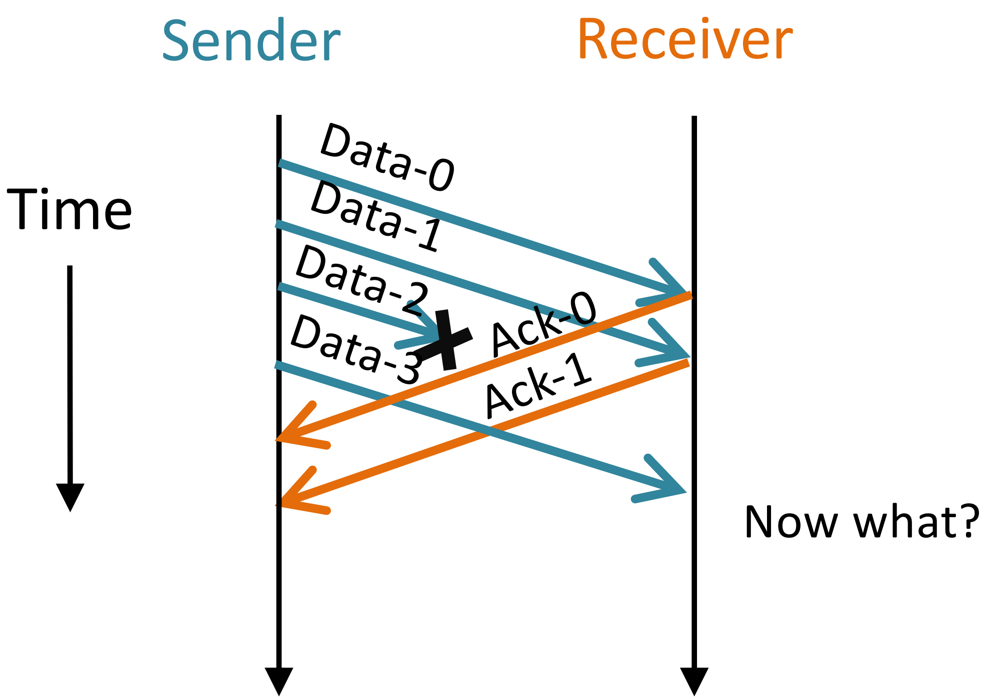
What should the sender do here?



What information does the sender need to make that decision?

What is required by either party to keep track?

A. Start sending all data again from 0.
B. Start sending all data again from 2.
C. Resend just 2, then continue with 4 afterwards.