# 15-441/641: Computer Networks
# The Transport Layer, Part 2 of 3

15-441/641 Fall 2019

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# Questions to discuss with a friend

- What are some things that make reliable transmission hard?
  - *Think: what went wrong in our reliable transmission race?*
- What is the difference between a “cumulative ACK” and a “basic ACK”?
  - What is one benefit of each?
- How do Selective Repeat and Go-back-N improve upon Stop-and-Wait?
- Can the transport layer guarantee:
  - That all packets will arrive at their destination?
  - That packets will be delivered at a certain throughput?
  - That packets will be delivered with a certain latency?

# Last Time: Reliable Transmission

- When transmitting across the Internet, how can we be sure that every message reaches its destination?
  - Retransmit!
- Three approaches:
  - Stop and Wait
  - Go Back N
  - Selective Repeat

# Stop-and-Wait: Summary

- **Sender:**
  - Transmit packets one by one. Label each with a sequence number. Set timer after transmitting.
  - If receive ACK, send the next packet.
  - If timer goes off, re-send the previous packet.
- **Receiver:**
  - When receive packet, send ACK.
  - If packet is corrupted, just ignore it — sender will eventually re-send.

Can I get some volunteers to act it out?

# Selective Repeat

- **Sender:**
  - Send packets from the window. Set timeout for each packet.
  - On receiving ACKs for the “left side” of the window, slide forward.
    - Send packets that have now entered the window.
  - On timeout, retransmit only the timed out packet
- **Receiver**
  - Keep a buffer of size of the window.
  - On receiving packets, send ACKs for every packet.
  - If packets come in out of order, just store them in the buffer and send ACK anyway.

Can I get some volunteers to act it out?

# Today's Agenda

- #1: Starting/Closing the Connection
  - Headers, mechanics
- #2: Deciding how big to set the window
  - Analysis, algorithms

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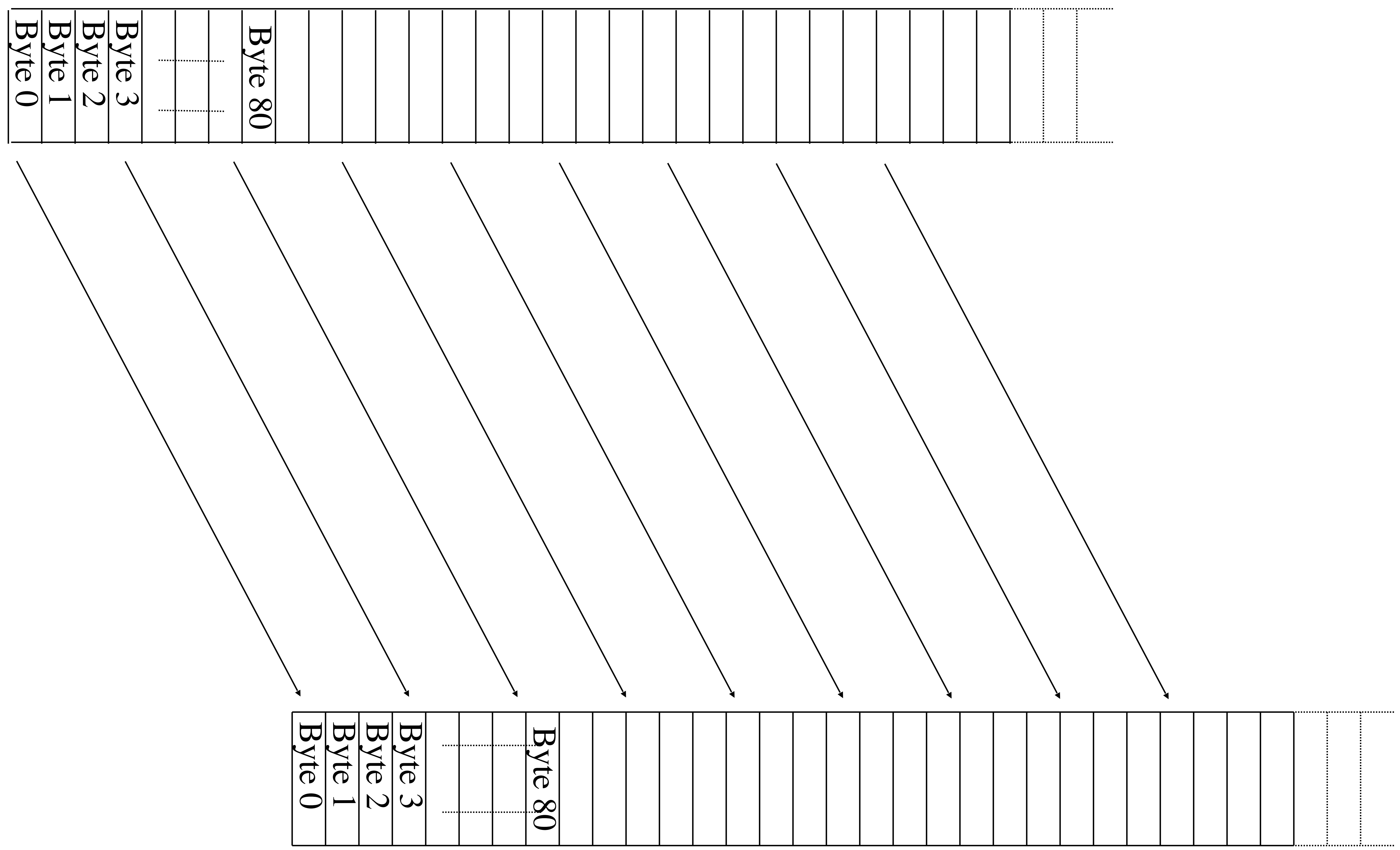
# TCP Header

Used to mux and demux

| | | | |
|---|---|---|---|
| Source port | | | Destination port |
| Sequence number | | | |
| Acknowledgment | | | |
| HdrLen | 0 | Flags | Advertised window |
| Checksum | | | Urgent pointer |
| Options (variable) | | | |
| Data | | | |

# TCP “Stream of Bytes” Service…

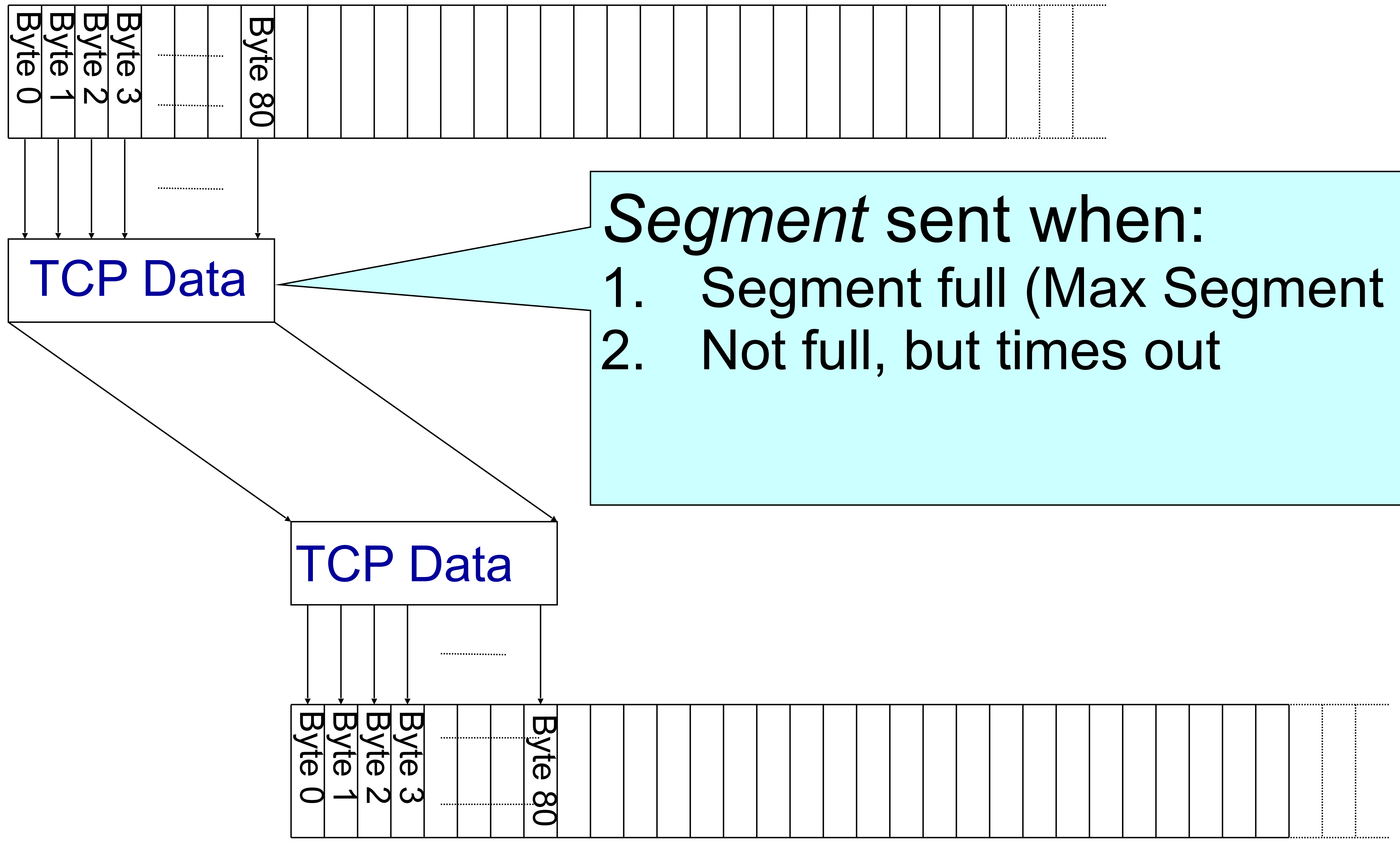
Application @ Host A




Application @ Host B

# … Provided Using TCP "Segments"

Host A

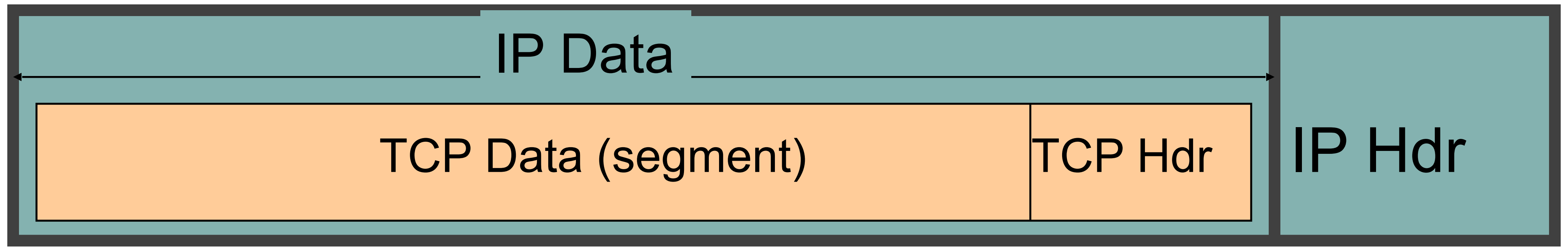



Host B

*Segment* sent when:

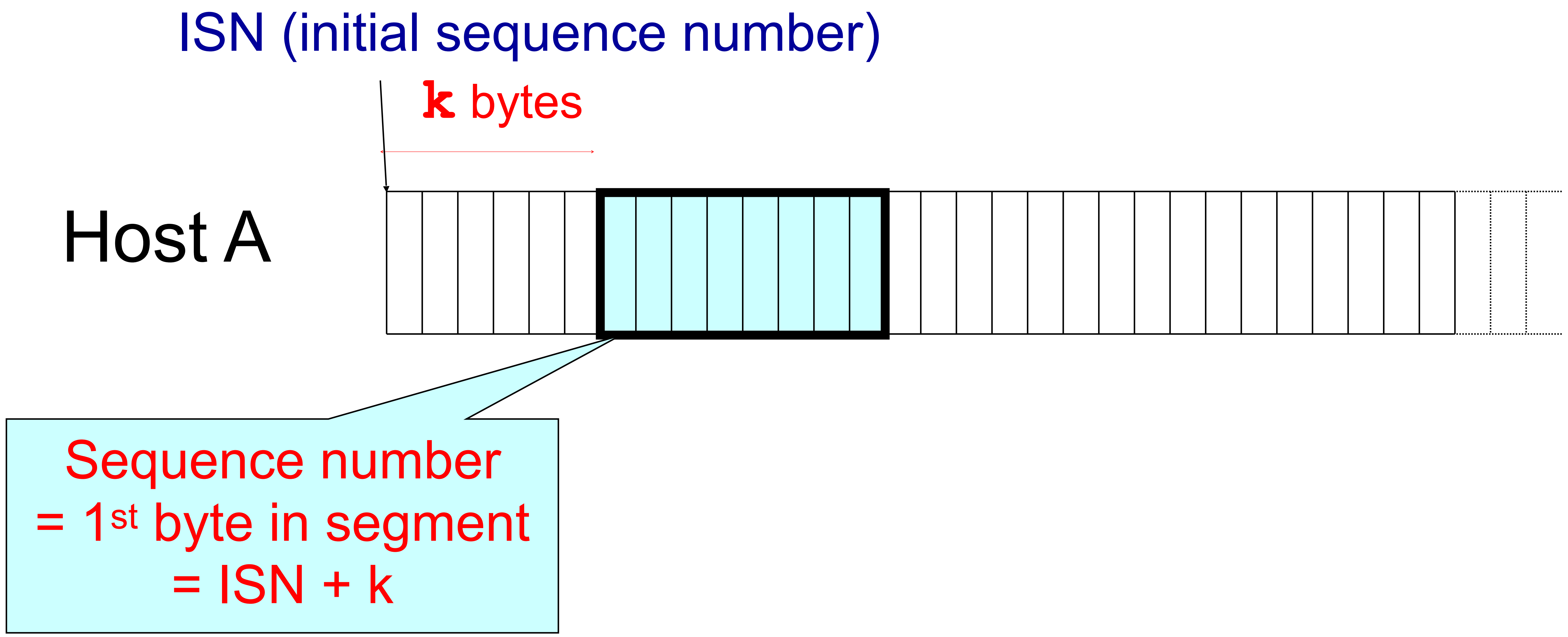
1. Segment full (Max Segment Size),
2. Not full, but times out

# TCP Segment

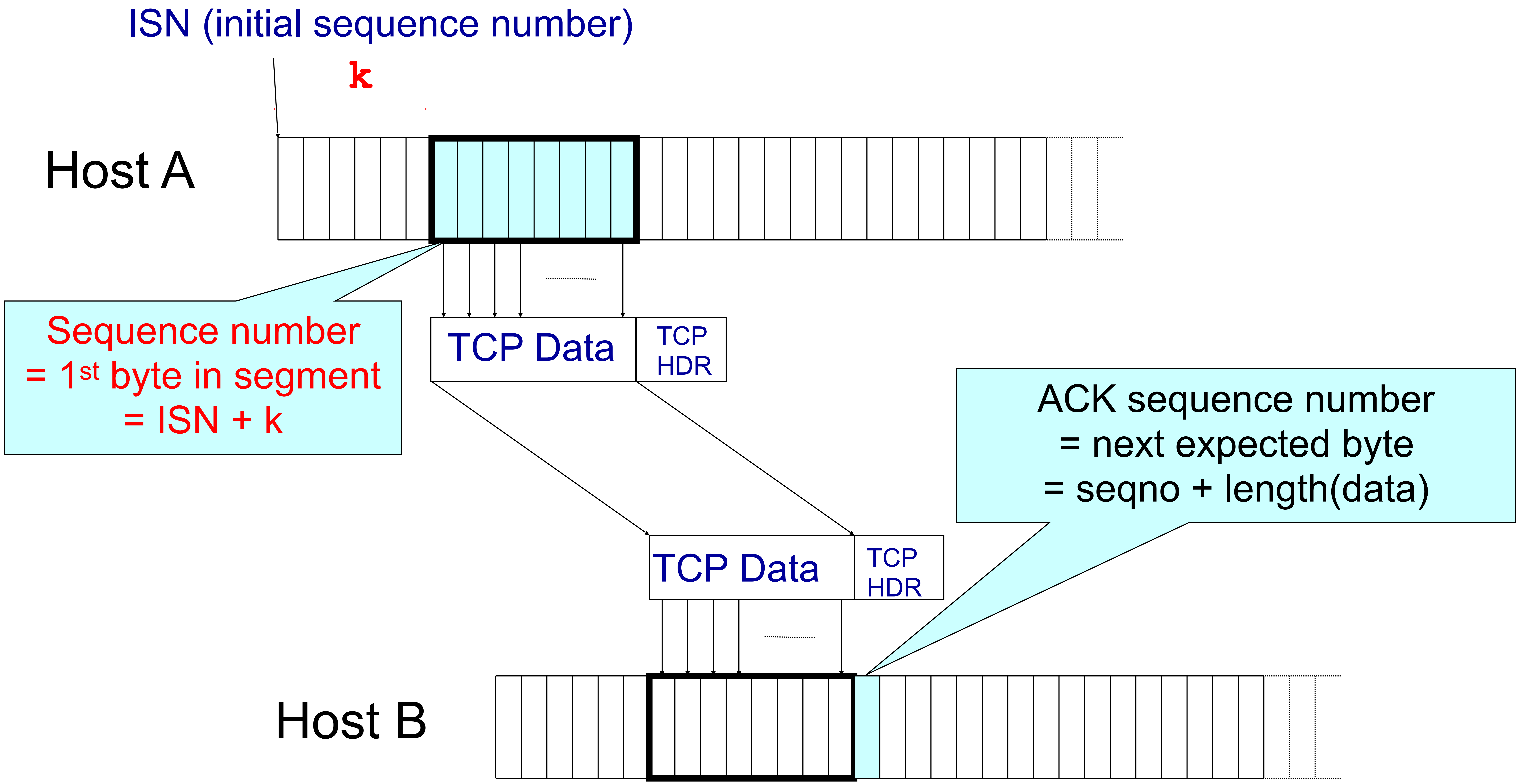



- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long
- TCP **segment**
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU – (IP header) – (TCP header)

# Sequence Numbers

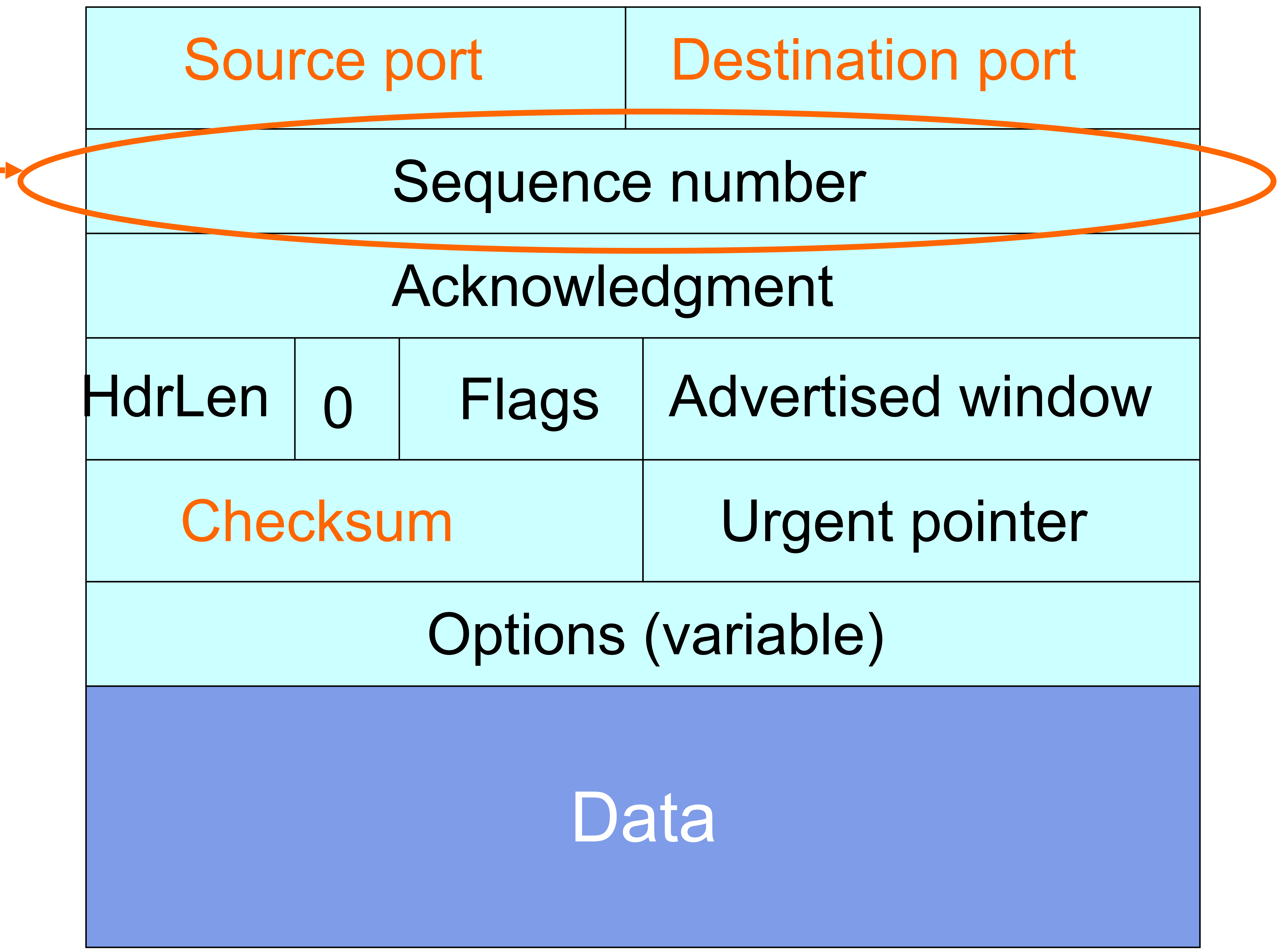


# Sequence Numbers



# TCP Header

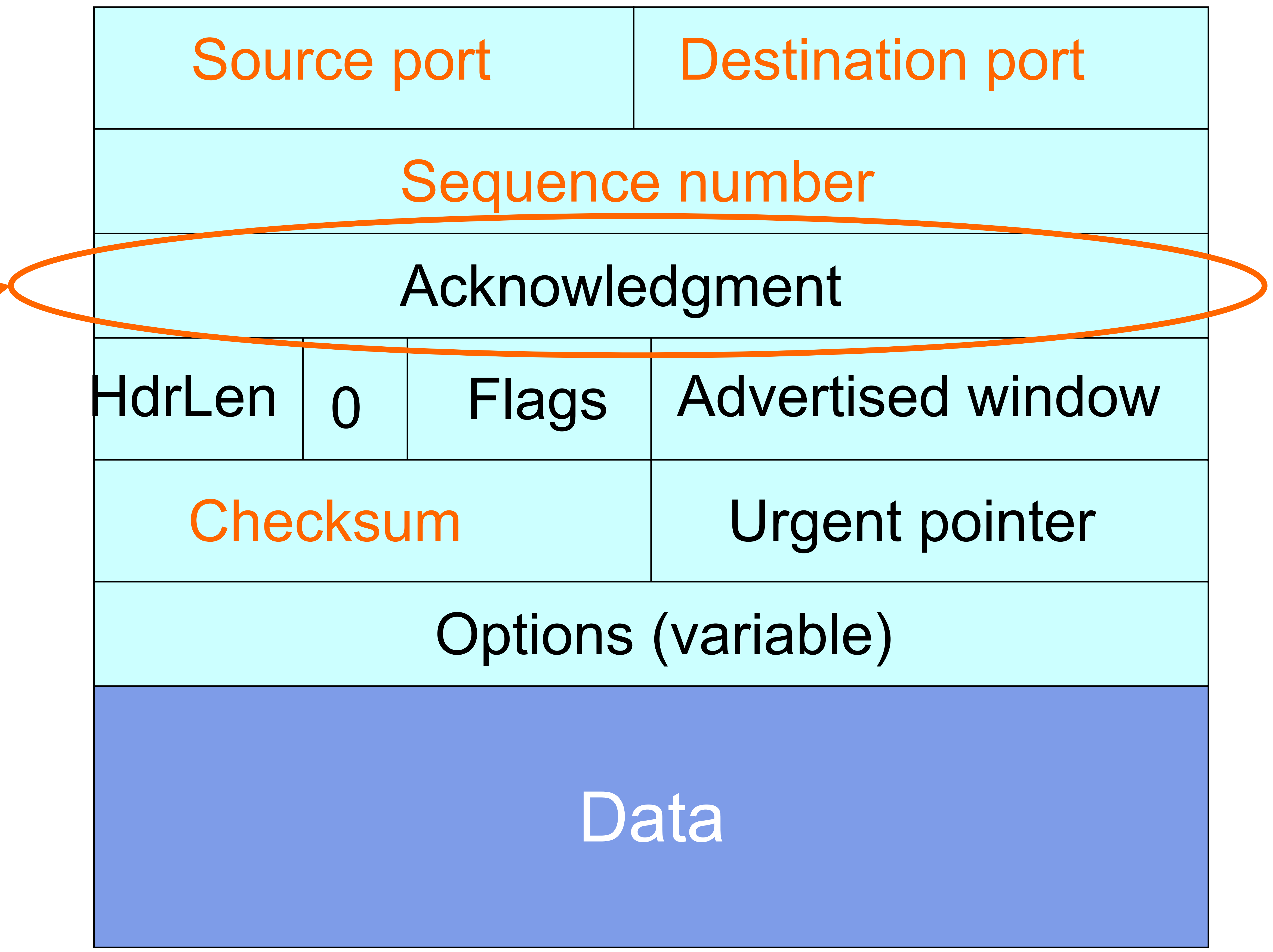
Starting byte offset of data carried in this segment



# TCP Header

Acknowledgment gives seqno just beyond highest seqno received **in order**
*("What Byte is Next")*

*Remember: CUMULATIVE — this means I have every byte before this sequence number*

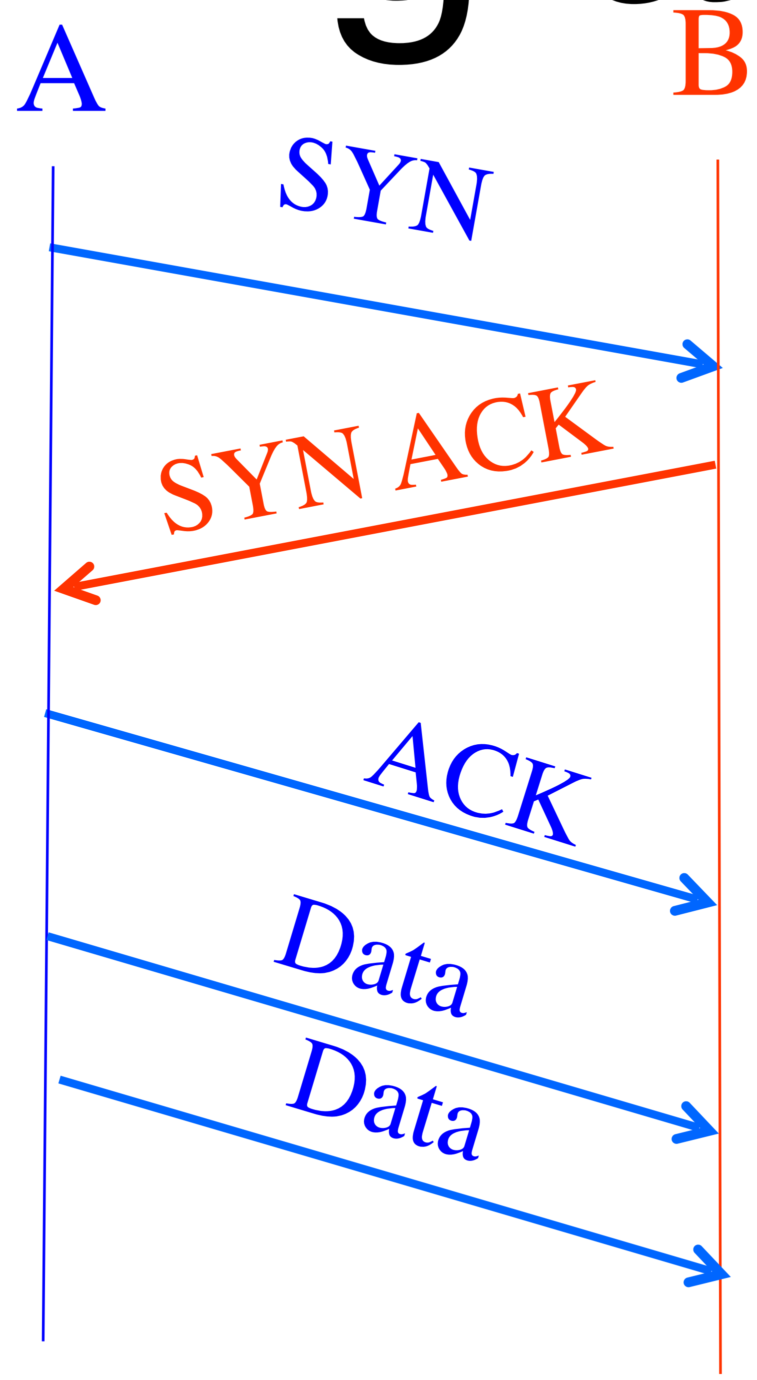


# TCP Connection Establishment and Initial Sequence Numbers

# Initial Sequence Number (ISN)

- Sequence number for the very first byte
- Why not just use ISN = 0?
- Practical issue
  - IP addresses and port #s uniquely identify a connection
  - Eventually, though, these port #s do get used again
  - … small chance an old packet is still in flight
- TCP therefore requires changing ISN
- Hosts exchange ISNs when they establish a connection

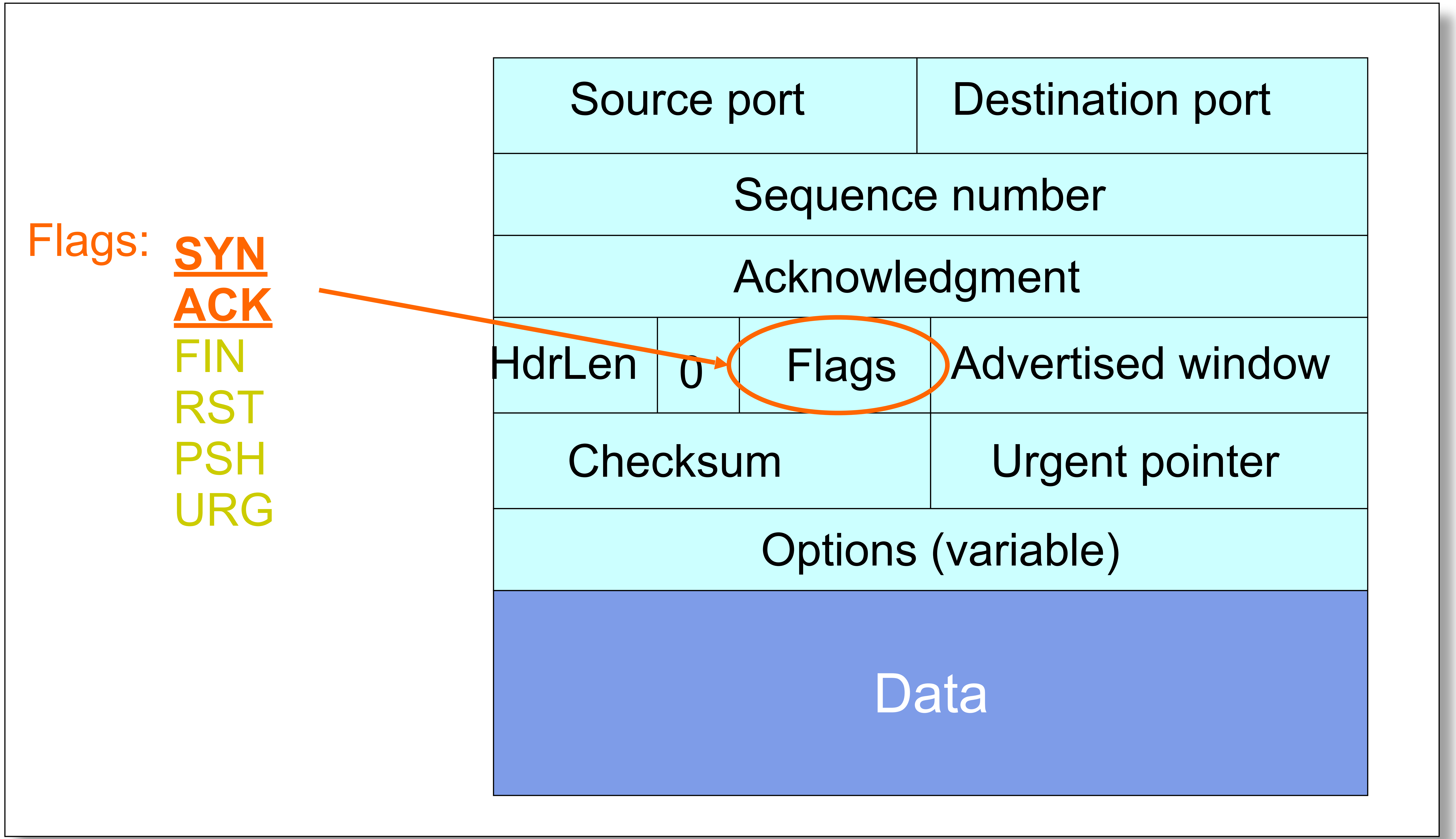
# Establishing a TCP Connection



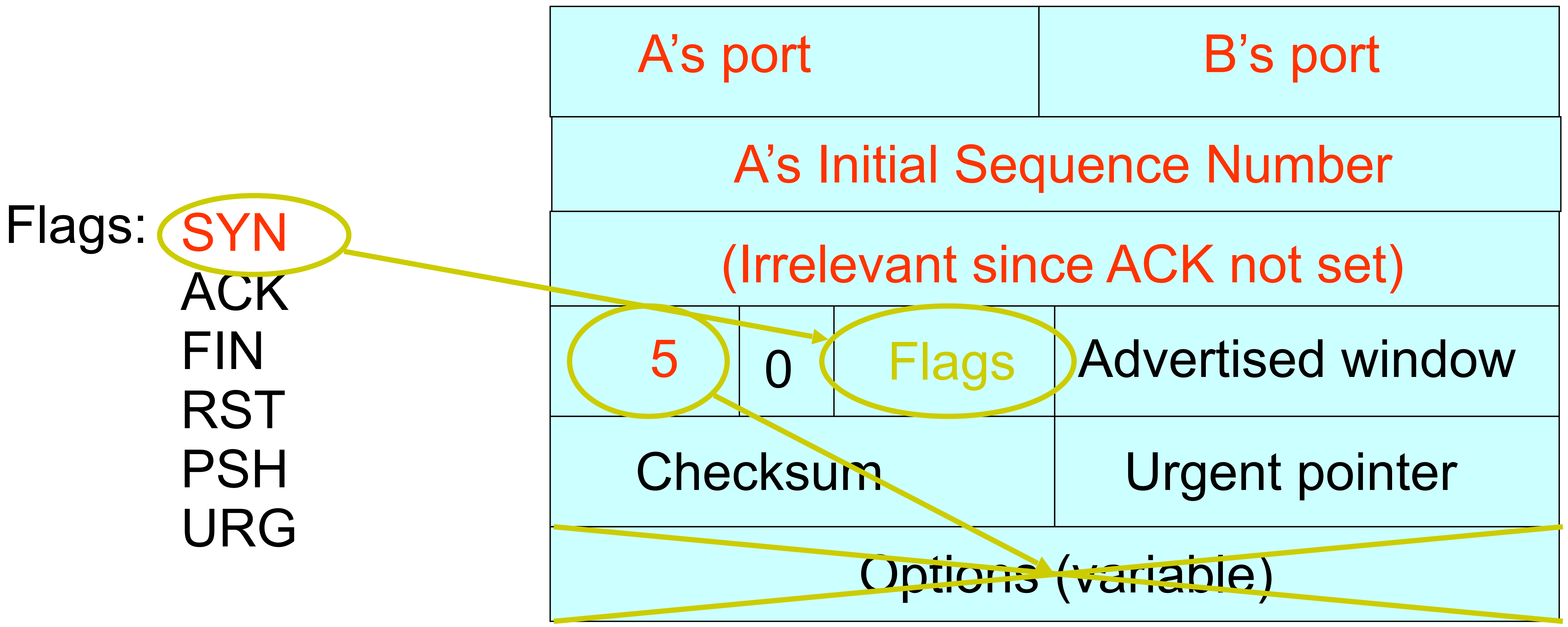

**Each host tells its ISN to the other host.**

- Three-way handshake to establish connection
  - Host A sends a **SYN** (open; "synchronize sequence numbers") to host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

# TCP Header

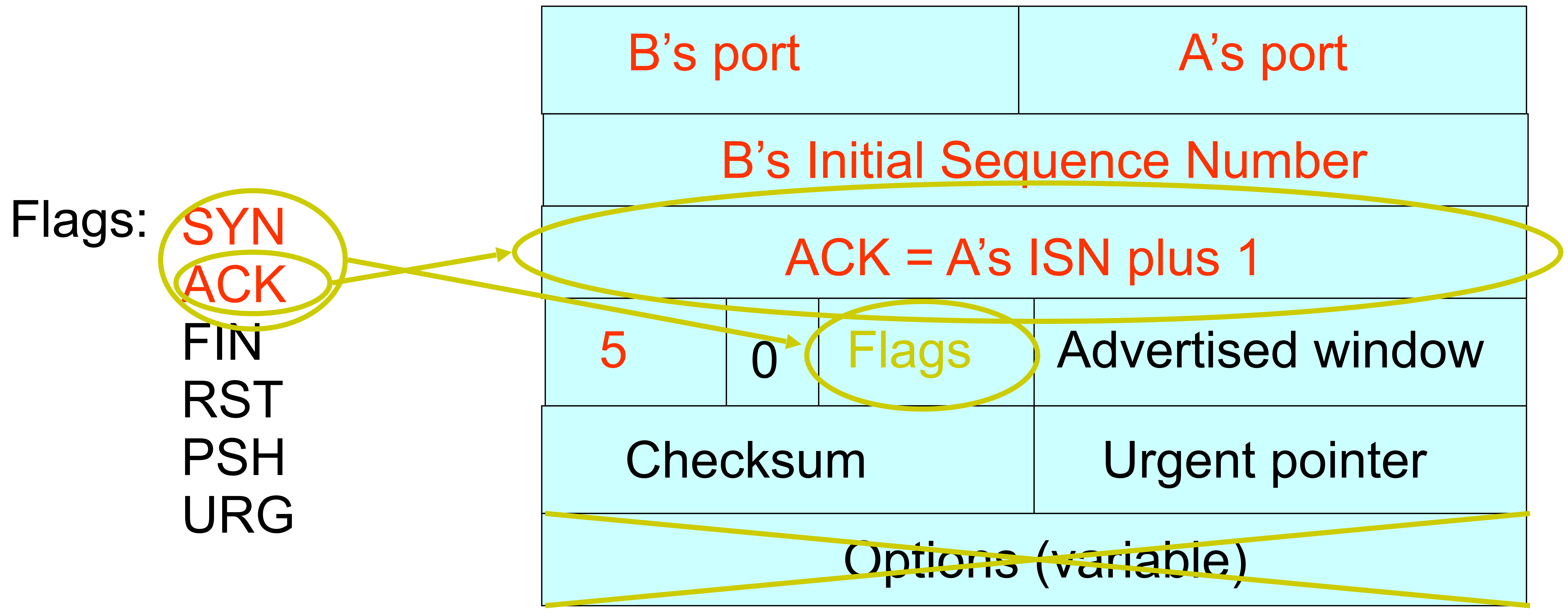

Flags: SYN
ACK
FIN
RST
PSH
URG
Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

# Step 1: A’s Initial SYN Packet




**A tells B it wants to open a connection…**

# Step 2: B’s SYN-ACK Packet




**B tells A it accepts, and is ready to hear the next byte…**

**… upon receiving this packet, A can start sending data**

# Step 3: A's ACK of the SYN-ACK

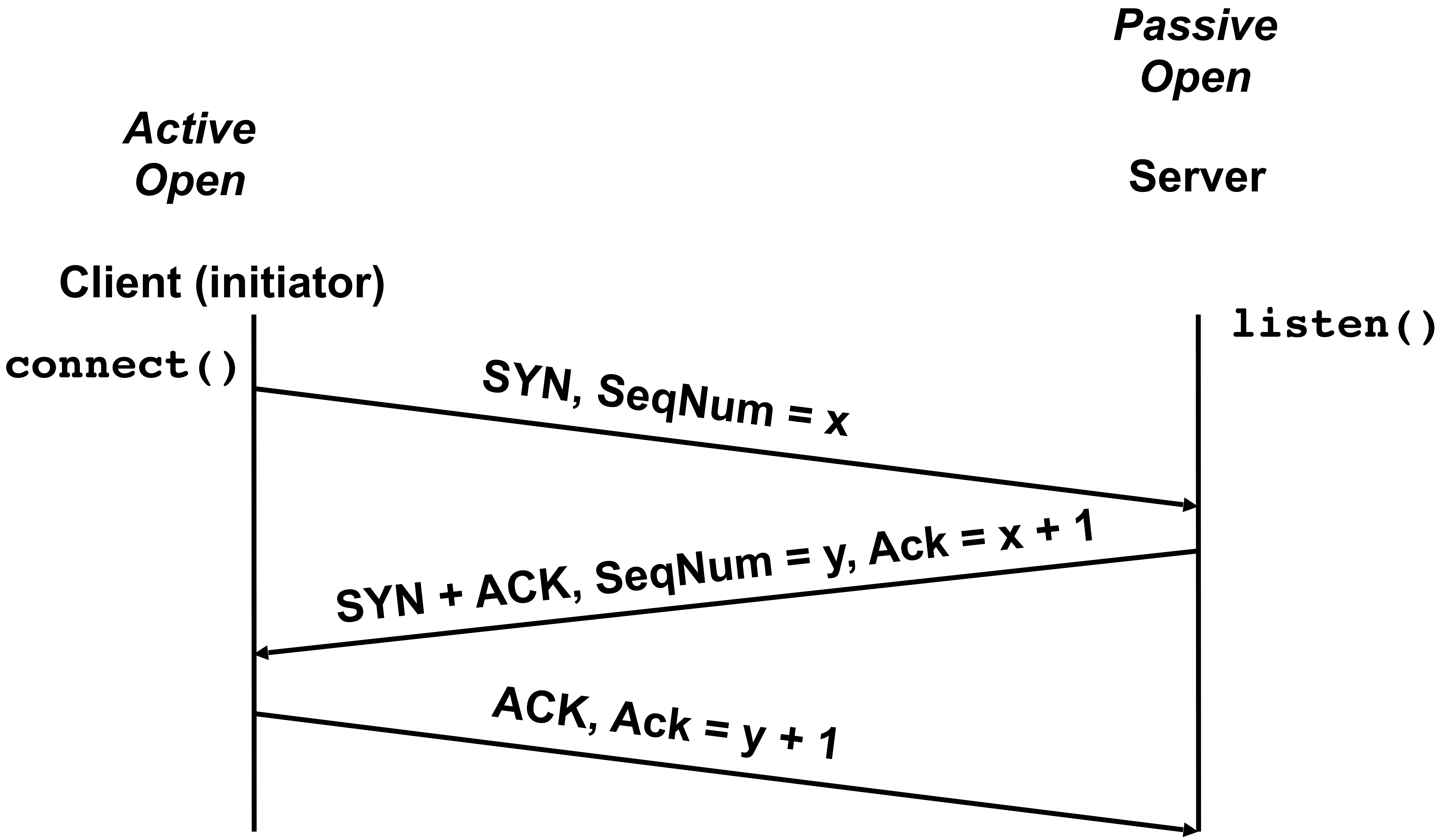
Flags: SYN
ACK
FIN
RST
PSH
URG

| | | | |
|---|---|---|---|
| A's port | | B's port | |
| A's Initial Sequence Number | | | |
| B's ISN plus 1 | | | |
| 20B | 0 | Flags | Advertised window |
| Checksum | | Urgent pointer | |
| ~~Options (variable)~~ | | | |

**A tells B it's likewise okay to start sending**

**… upon receiving this packet, B can start sending data**

# Timing Diagram: 3-Way Handshaking



# What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or:
  - Server discards the packet (e.g., it's too busy)

- Eventually, no SYN-ACK arrives
  - Sender sets a timer and waits for the SYN-ACK
  - … and retransmits the SYN if needed

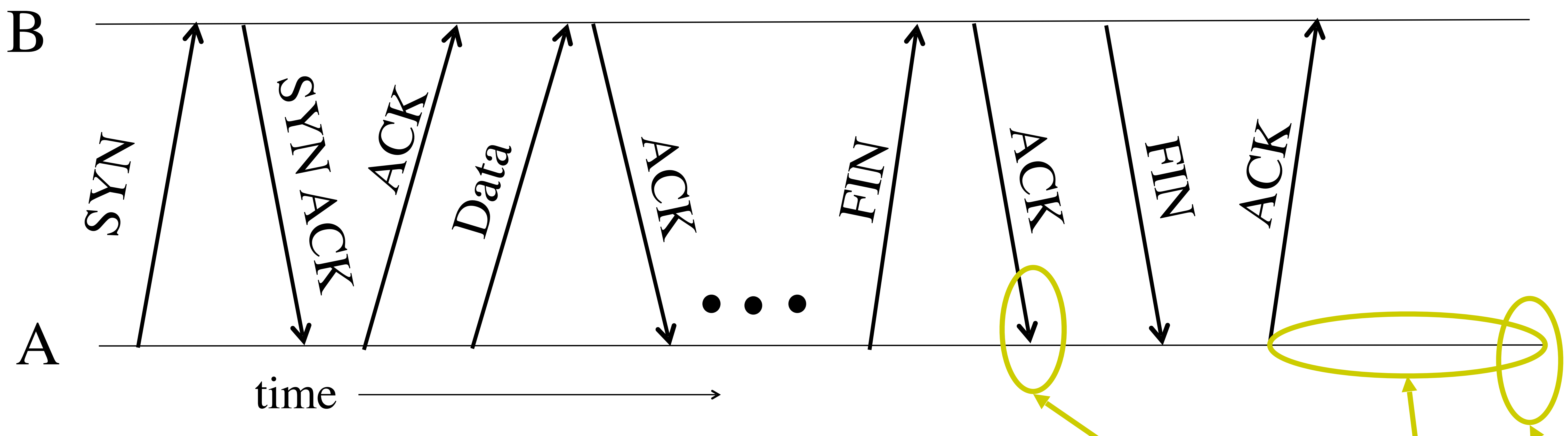
- How should the TCP sender set the timer?
  - Sender has no idea how far away the receiver is
  - Hard to guess a reasonable length of time to wait
  - **SHOULD** (RFCs 1122 & 2988) use default of 3 seconds
    - Some implementations instead use 6 seconds

# SYN Loss and Web Downloads

- User clicks on a hypertext link
  - Browser creates a socket and does a “connect”
  - The “connect” triggers the OS to transmit a SYN
- If the SYN is lost…
  - 3-6 seconds of delay: can be very long
  - User may become impatient
  - … and click the hyperlink again, or click “reload”
- User triggers an “abort” of the “connect”
  - Browser creates a new socket and another “connect”
  - Essentially, forces a faster send of a new SYN packet!
  - Sometimes very effective, and the page comes quickly

# Tearing Down the Connection

# Normal Termination, One Side At A Time




- Finish (**FIN**) to close and receive remaining bytes
  - **FIN** occupies one byte in the sequence space
- Other host acks the byte to confirm
- Closes A's side of the connection, but not B's
  - Until B likewise sends a **FIN**
  - Which A then acks

Connection now **closed**

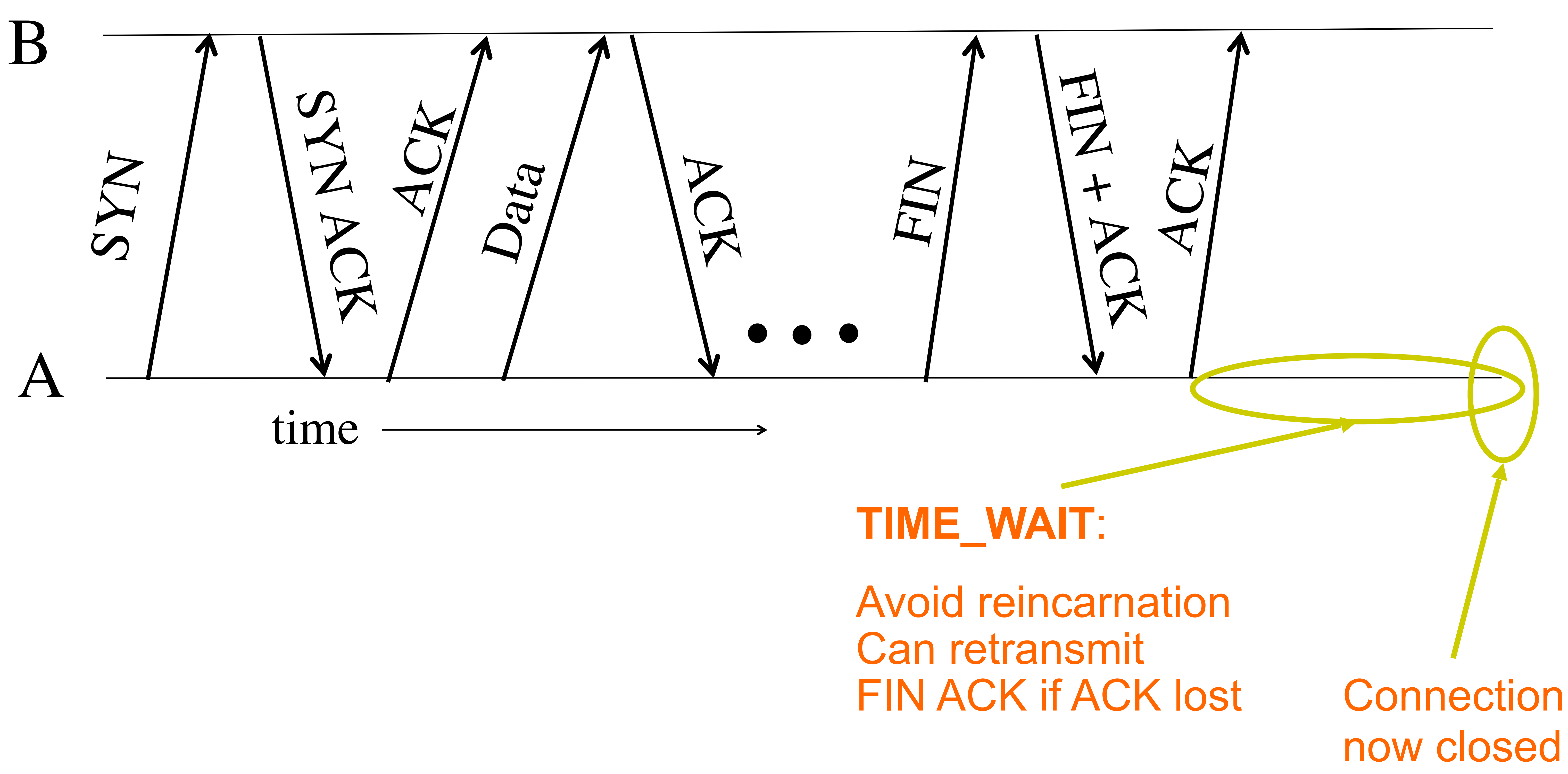
Connection now **half-closed**

**TIME_WAIT**:

Avoid reincarnation

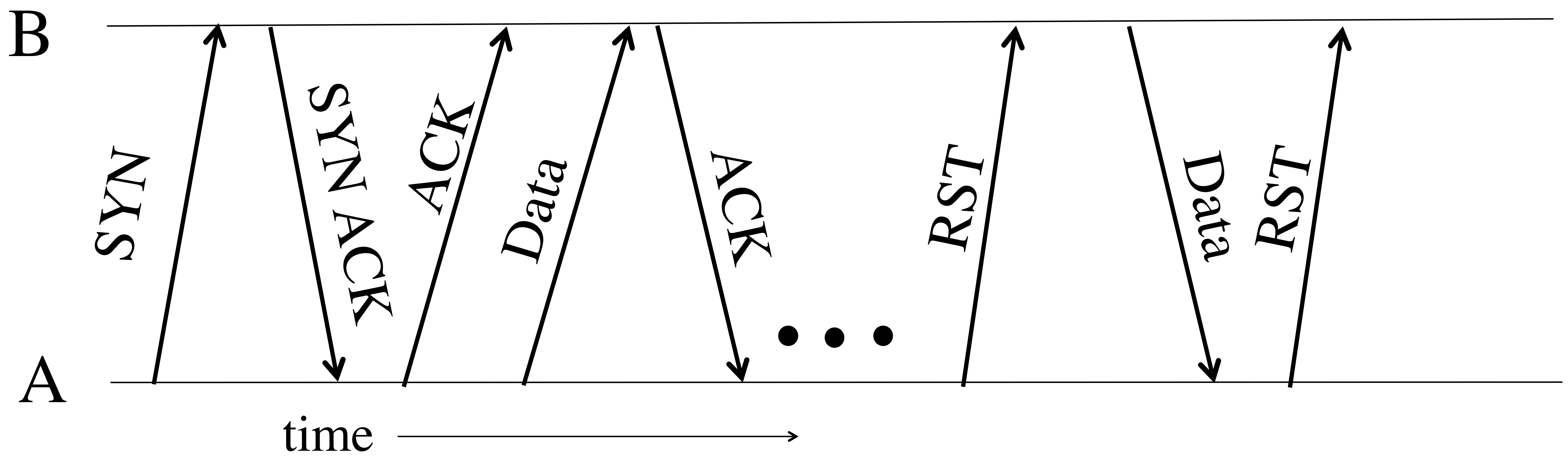
B will retransmit FIN if ACK is lost

# Normal Termination, Both Together



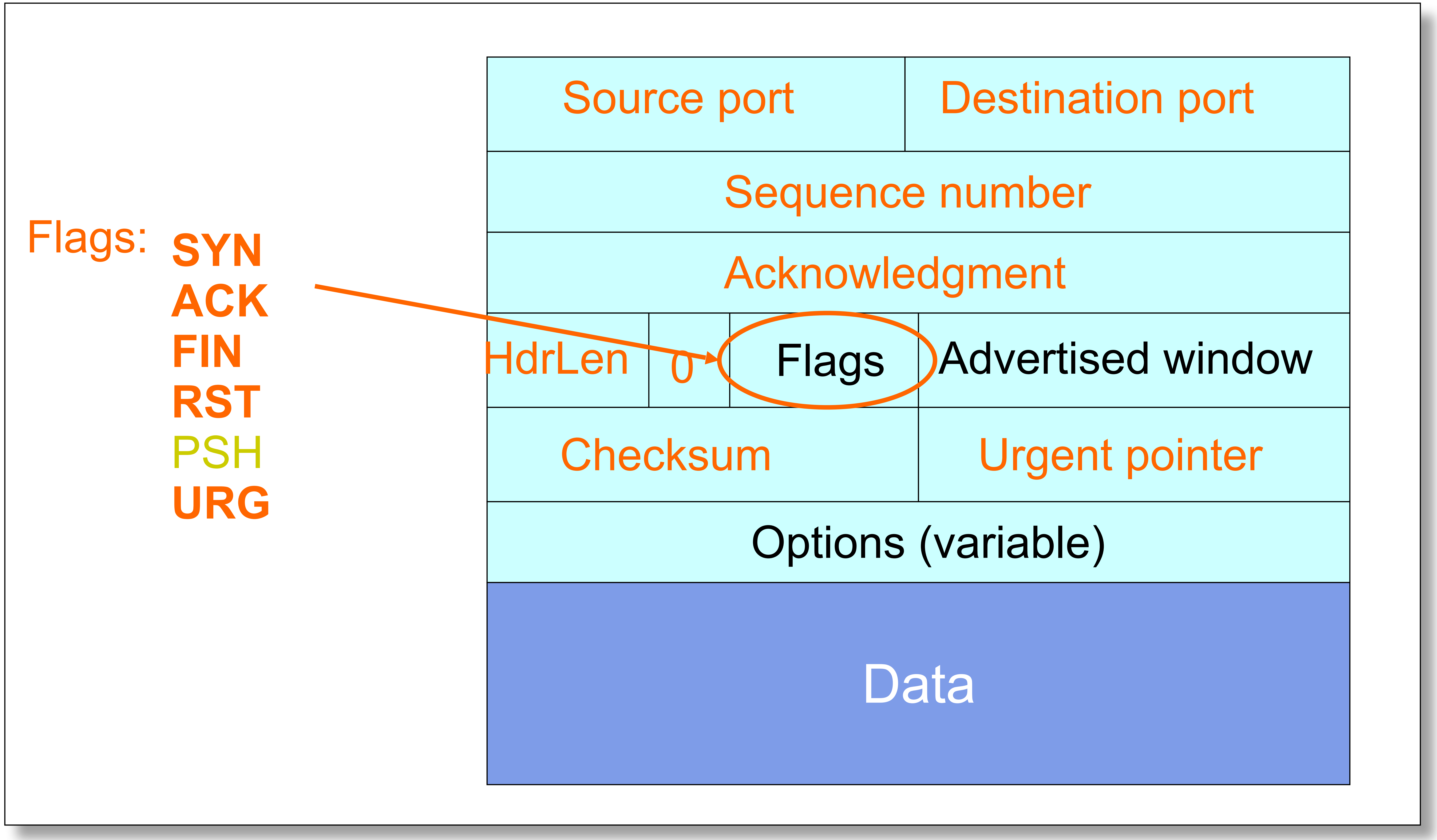

- Same as before, but B sets **FIN** with their ack of A's **FIN**

# Abrupt Termination




- A sends a RESET (**RST**) to B
  - E.g., because application process on A crashed
- That's it
  - B does not ack the **RST**
  - Thus, **RST** is not delivered reliably
  - And: any data in flight is lost
  - But: if B sends anything more, will elicit another **RST**

# TCP Header


Flags:
SYN
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Source port
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HdrLen
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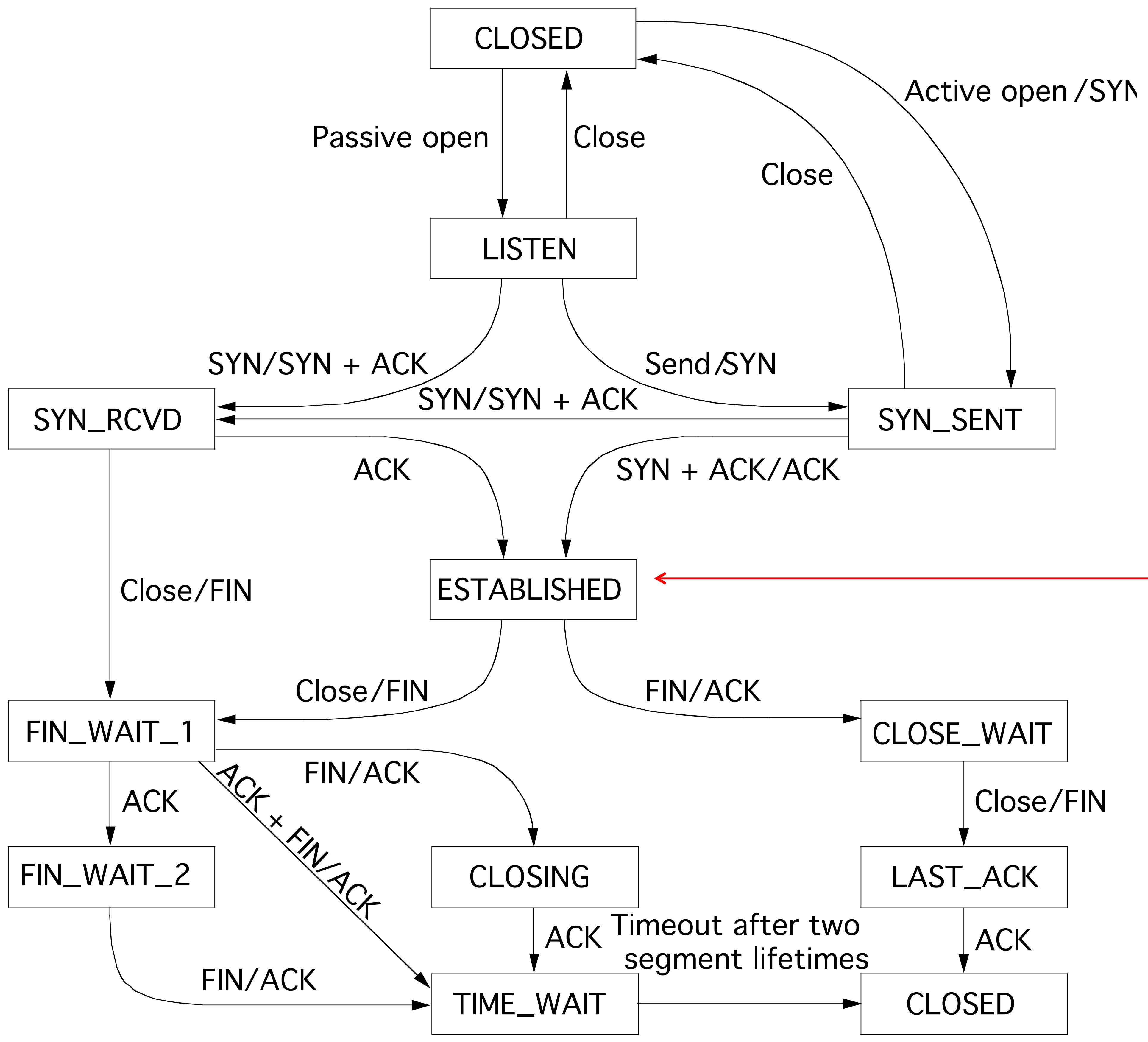
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

# TCP State Transitions




Data, ACK exchanges are in here

# After all that work…

- ESTABLISHED is the part where we transmit data!
- In checkpoint 1 of P2, you will have a basic Stop-And-Wait sender given to you, but you will need to enable the handshake and session termination.

# Today’s Agenda

- #1: Starting/Closing the Connection
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- **#2: Deciding how big to set the window**
  - Analysis, algorithms

# Sliding Windows

- A sender's "window" contains a set of packets that have been transmitted but not yet acked.
- Sliding windows improve the efficiency of a transport protocol.
- Two questions we need to answer to use windows:
    - (1) How do we handle loss with a windowed approach?
    - (2) How big should we make the window?

# Last Time

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# Today

- A sender's "window" contains a set of packets that have been transmitted but not yet acked.
- Sliding windows improve the efficiency of a transport protocol.
- Two questions we need to answer to use windows:
  - (1) How do we handle loss with a windowed approach?
  - (2) How big should we make the window?

Why not send as fast as we can?

# Problem #1: Flow Control

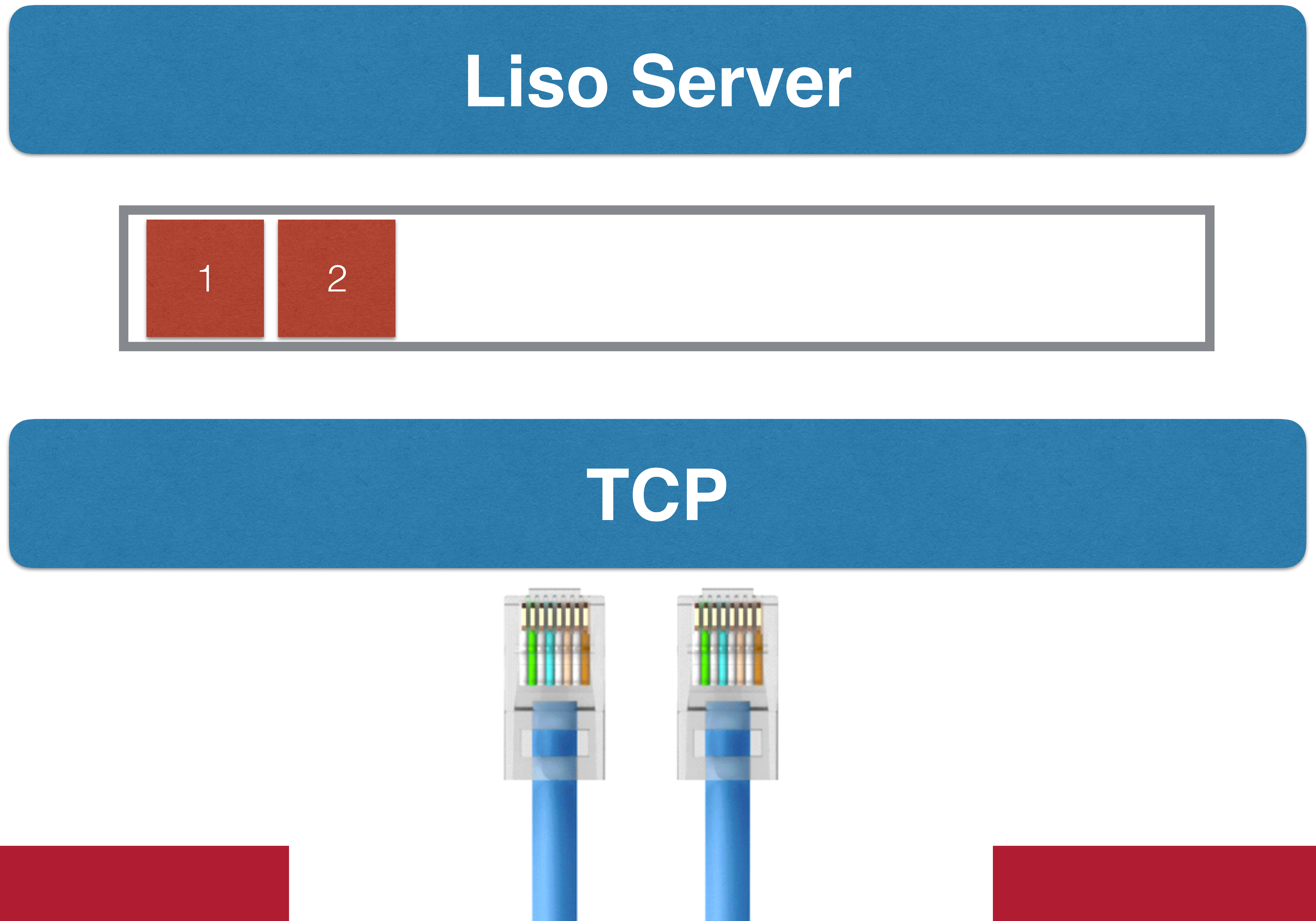
Yet another demo…

I need two volunteers, one of whom is confident reading out loud in English!

Flow Control: Don't overload the receiver.

Bonus candy: who wrote the essay in the packets? What is the essay named?

# Receive Buffer



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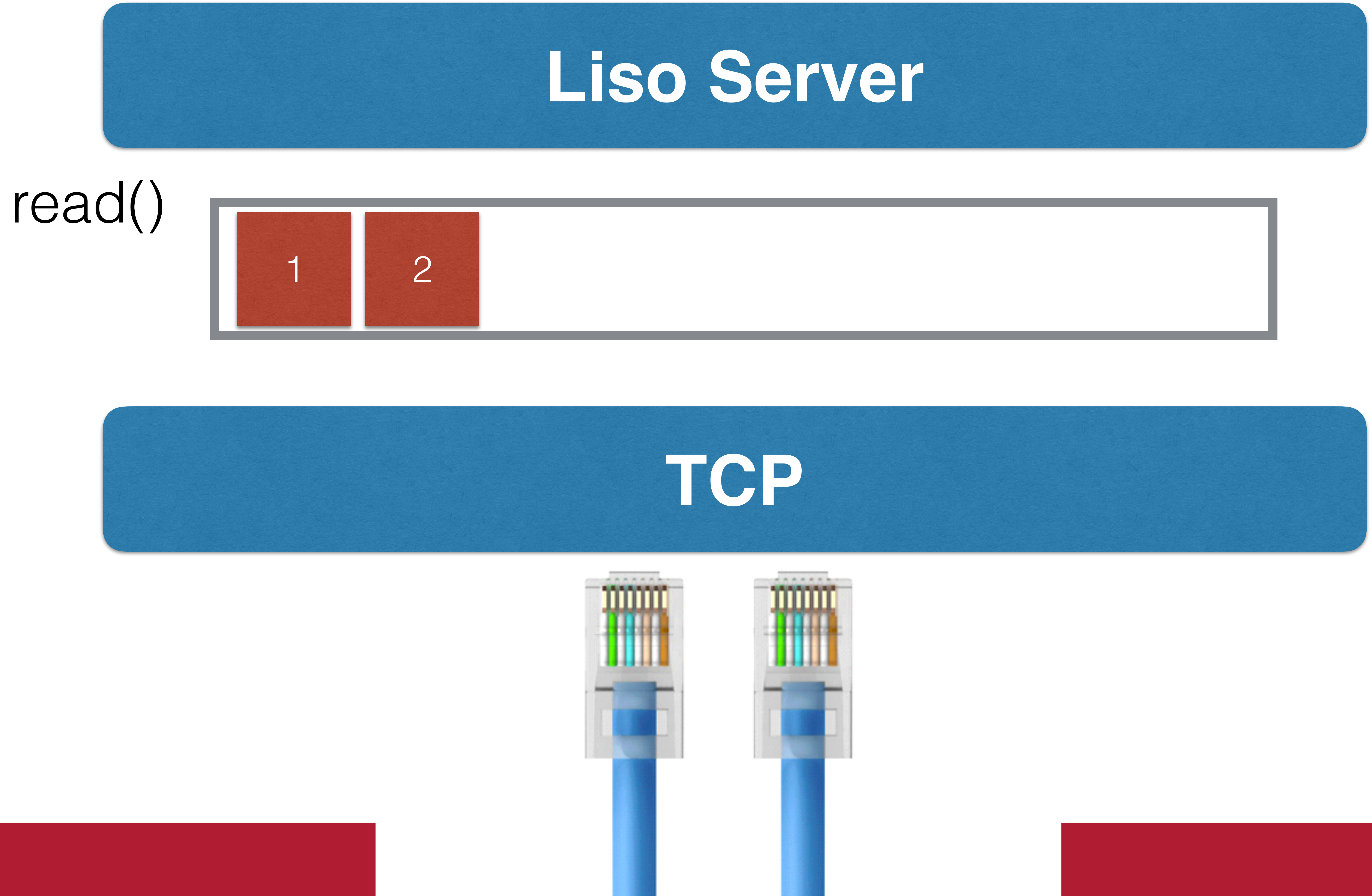

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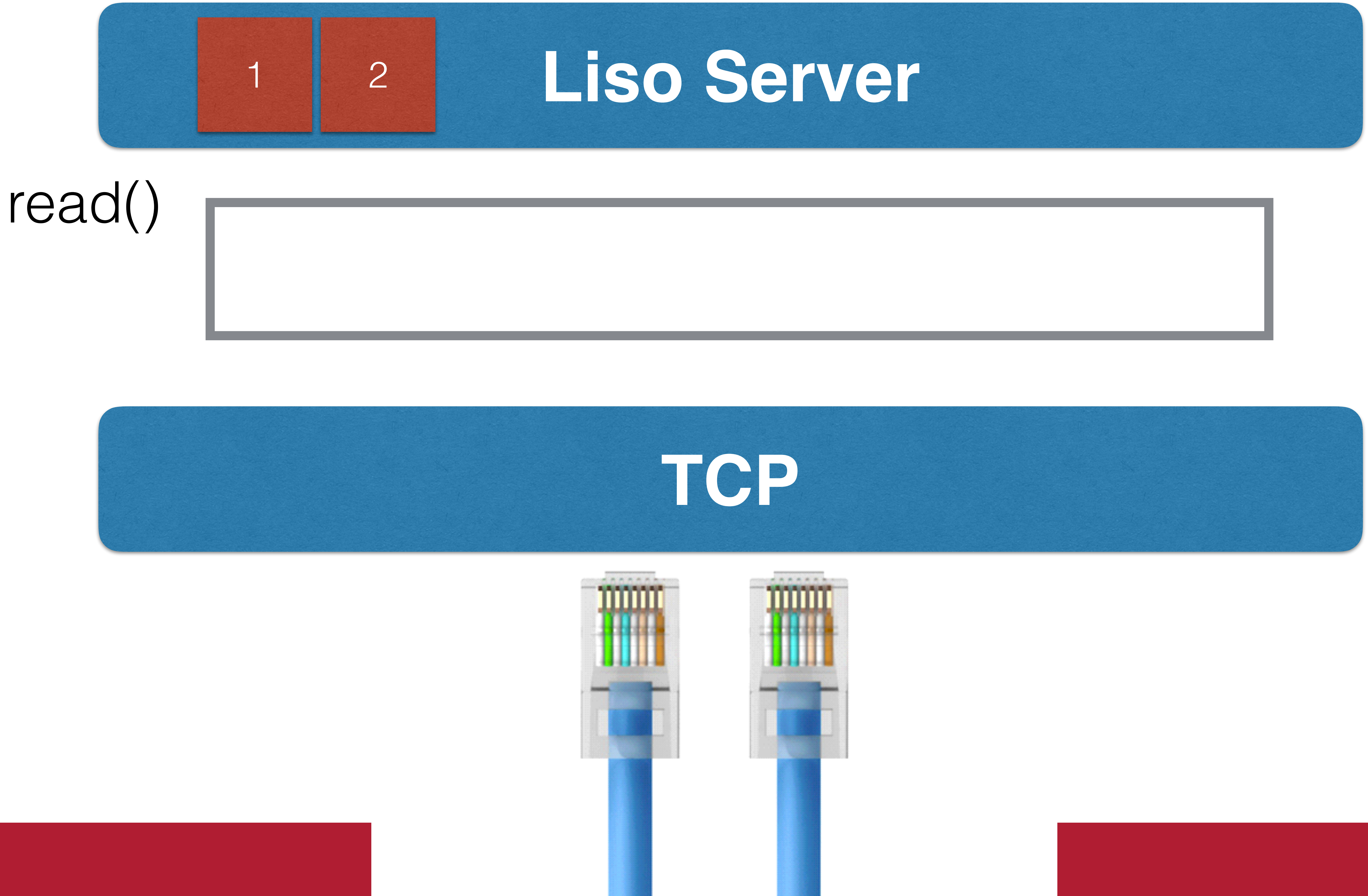

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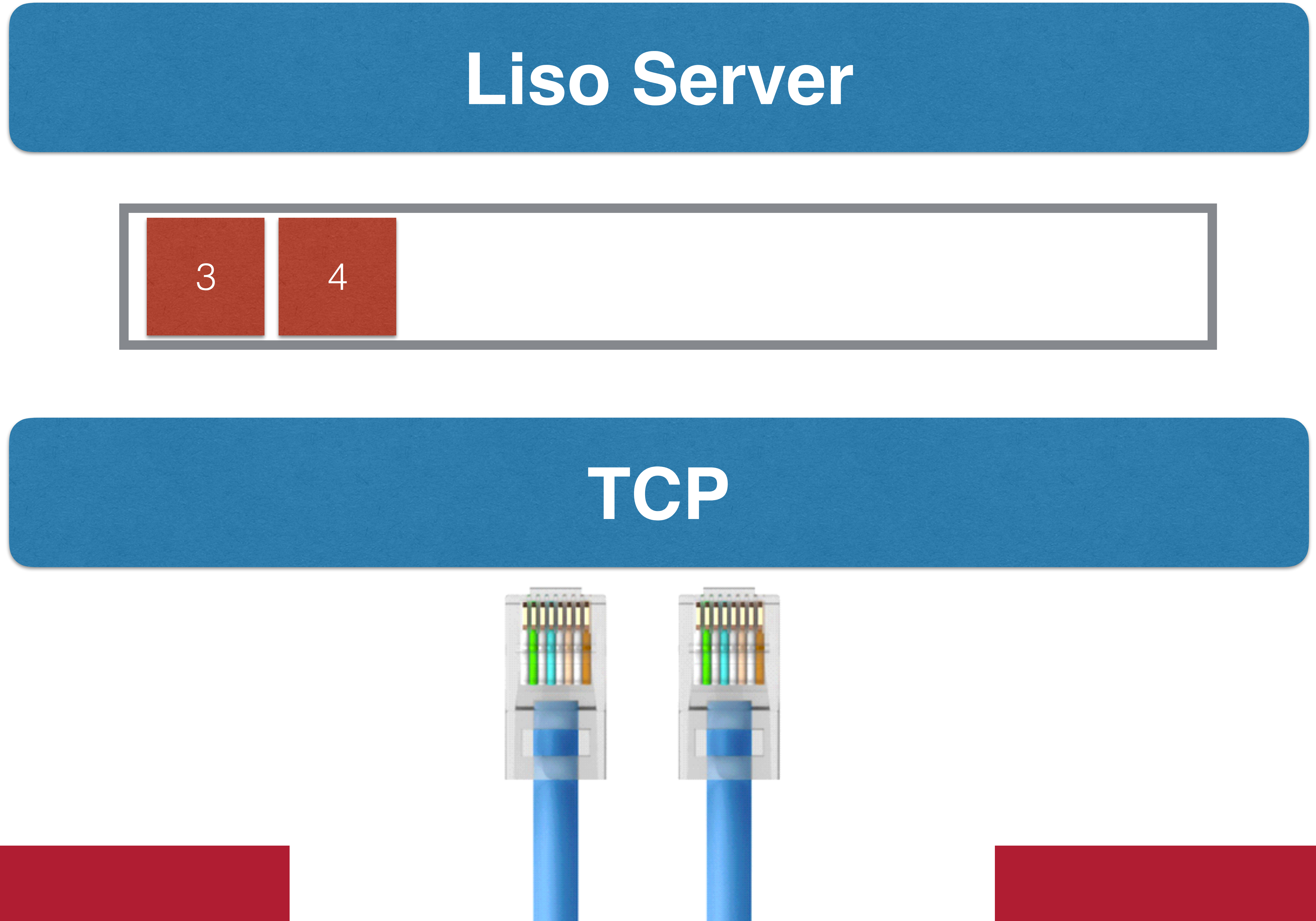

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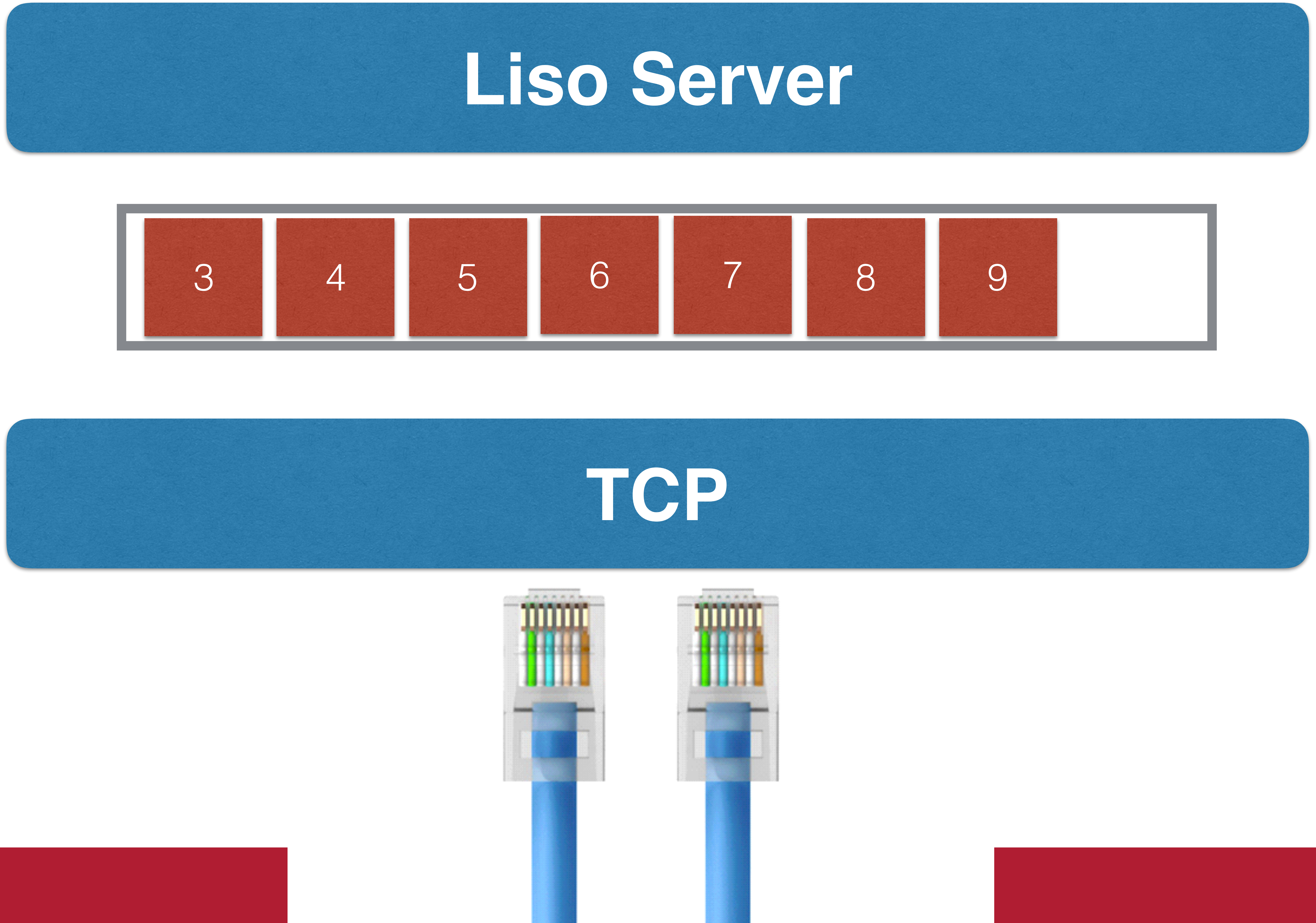

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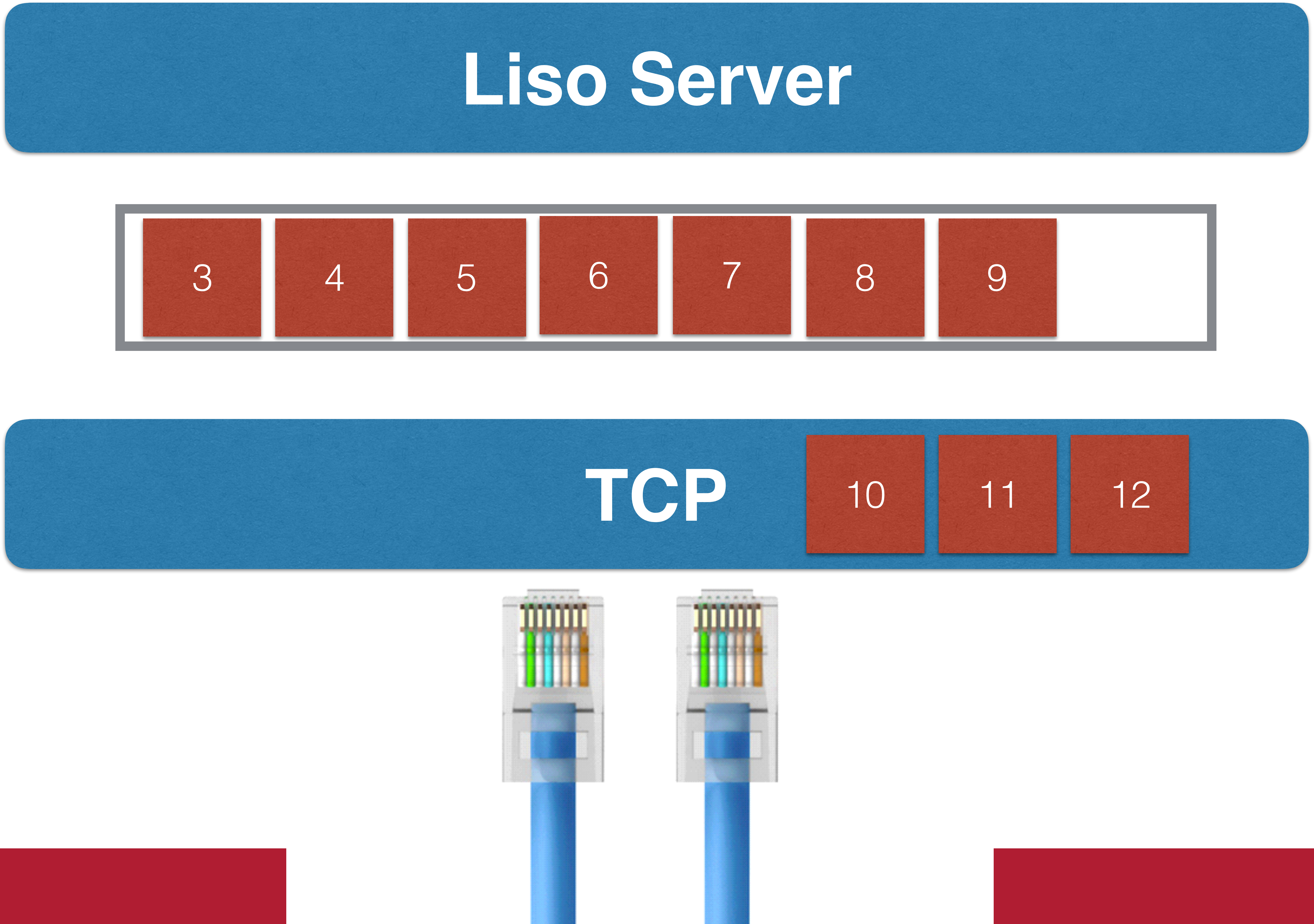

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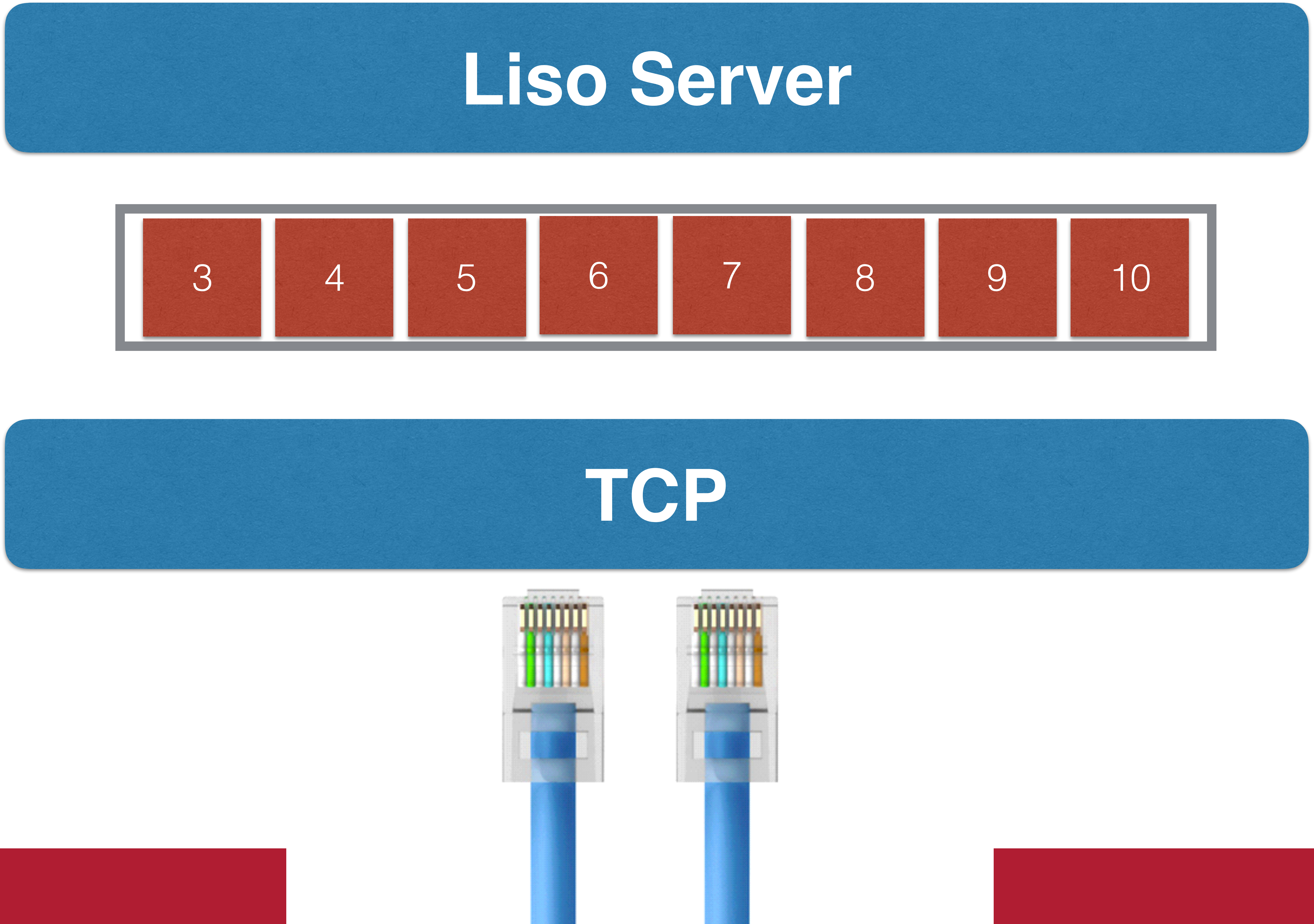


11 and 12 just get dropped :(

# Solution: Advertised Window

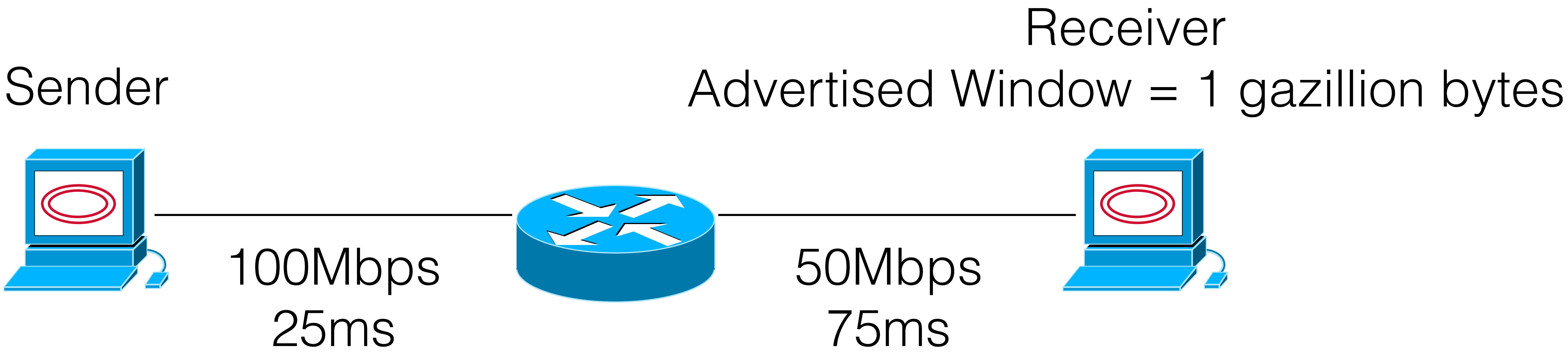
- Receiver uses an "Advertised Window" (W) to prevent sender from overflowing its window
    - Receiver indicates value of W in ACKs
    - Sender limits number of bytes it can have in flight <= W
- If I only have 10KB left in my buffer, tell the receiver in my next ACK!

# How big should we make the window?

- Window should be:
  - Less than or equal to the advertised window so that we do not overload the receiver.
    - This is called Flow Control.

Alright, so let's set the window to W?

# What will happen here?



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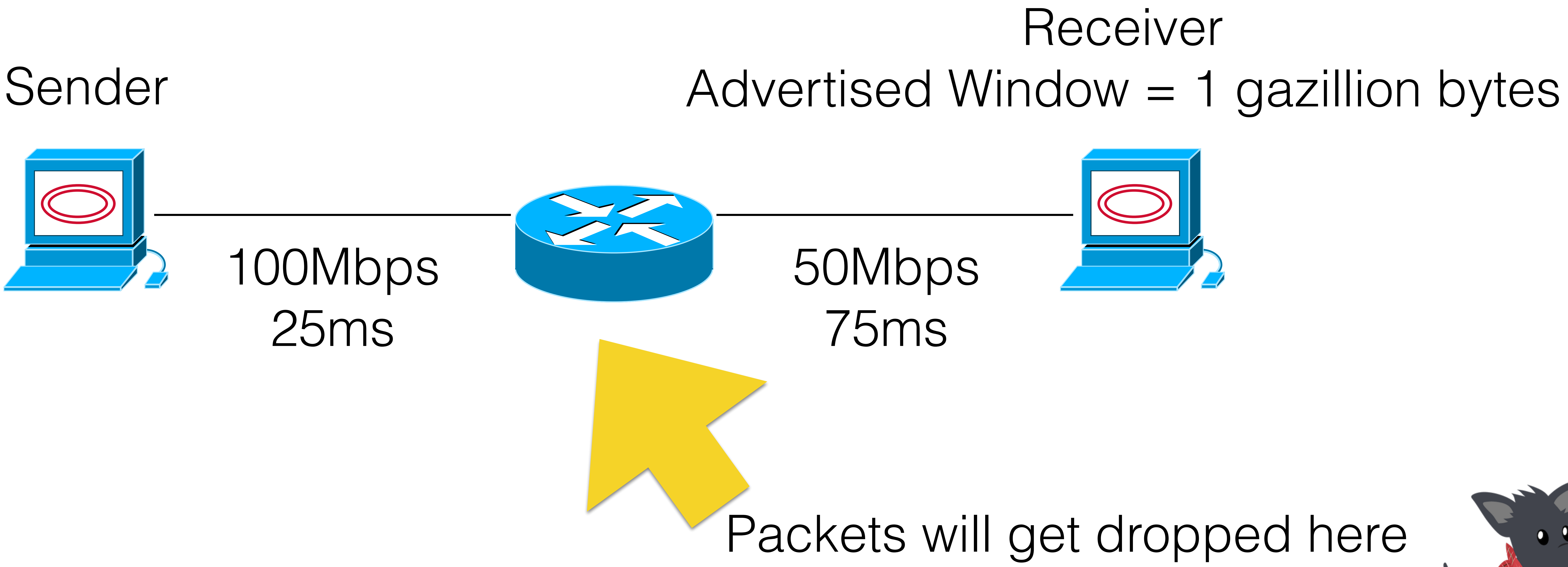

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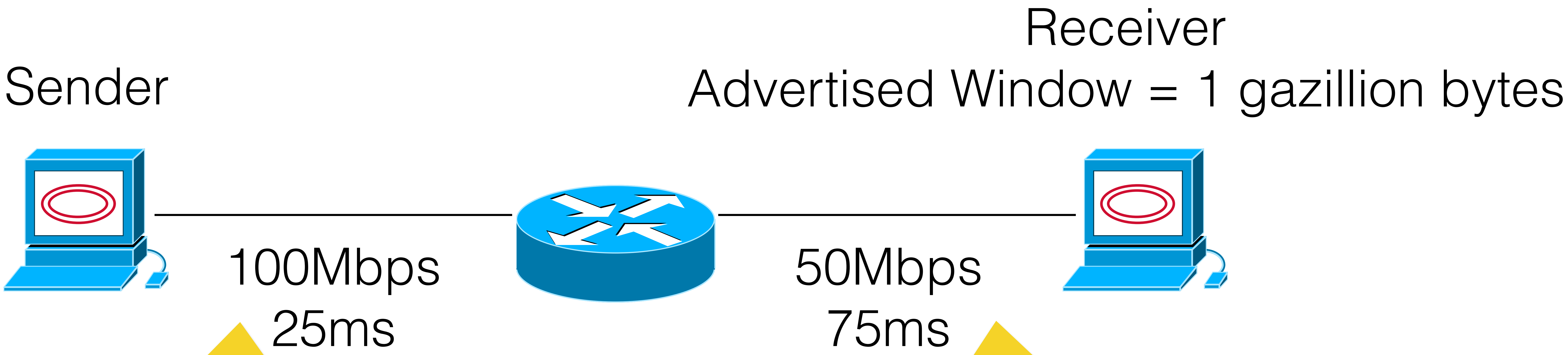
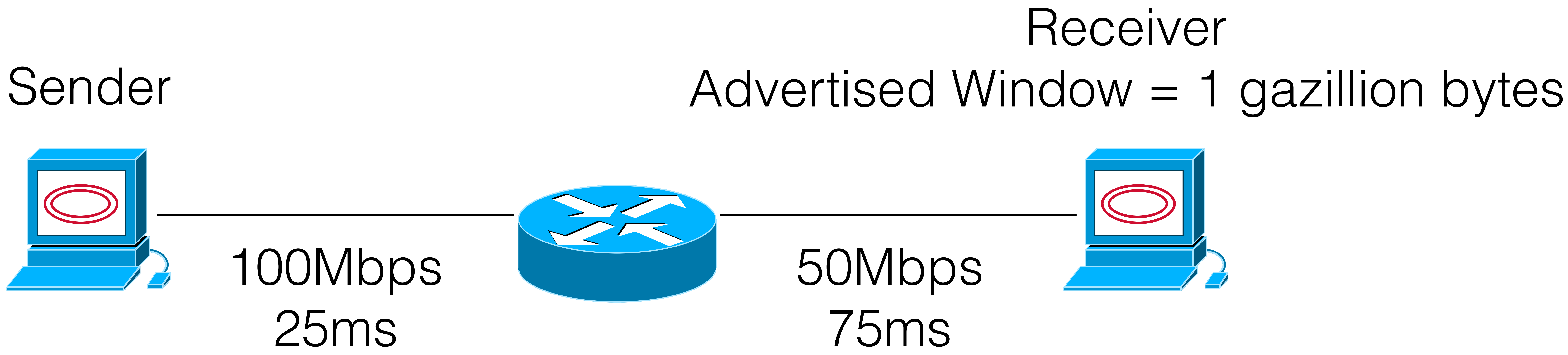



Arrival rate is faster than departure rate

How big should we set the window to be?

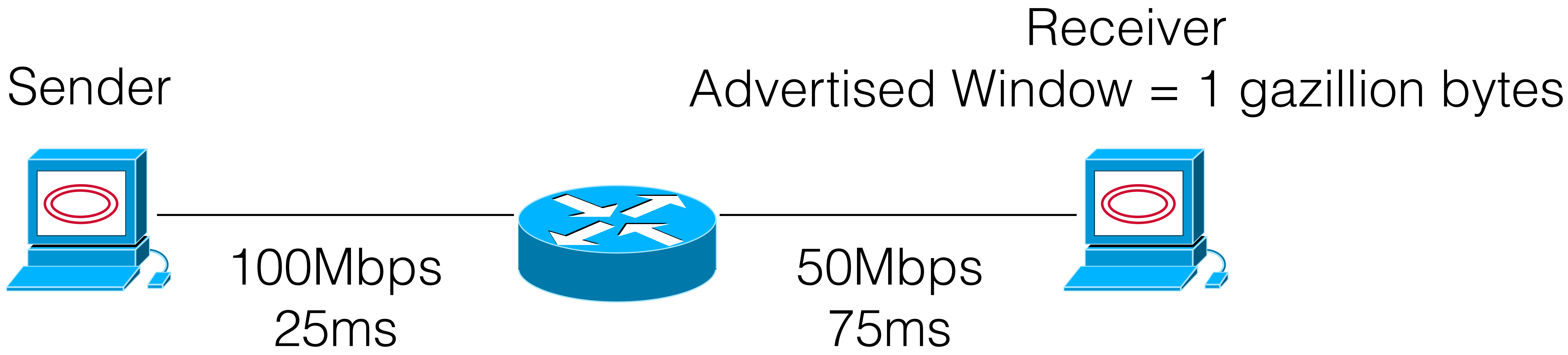
# “I just want to send at 50Mbps — how does that translate into a window size?”



Remind me: what is the definition of a Window?

Recall: Window is the number of bytes I may have transmitted but not yet received an ACK for.

# How long will it take for me to receive an ACK back for the first packet?



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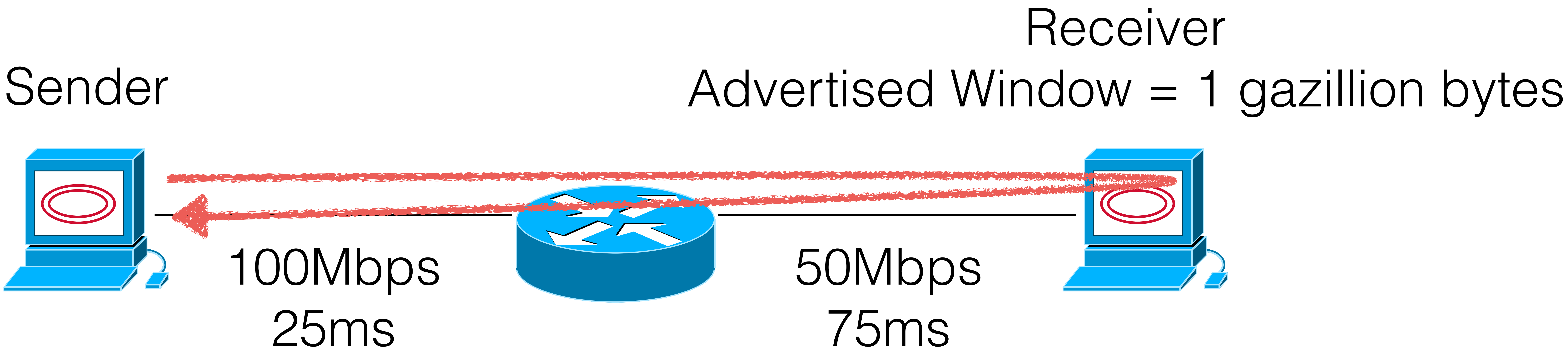



One round-trip-time (RTT) = 200 milliseconds

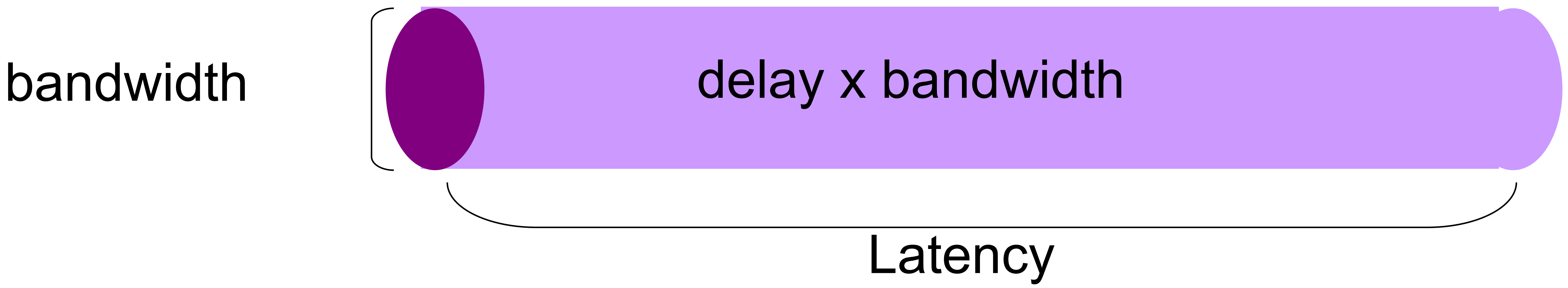
How much data will I send, at 50Mbps, in 200ms?

50Mbps * 200ms = 1.25 MB

We call this the

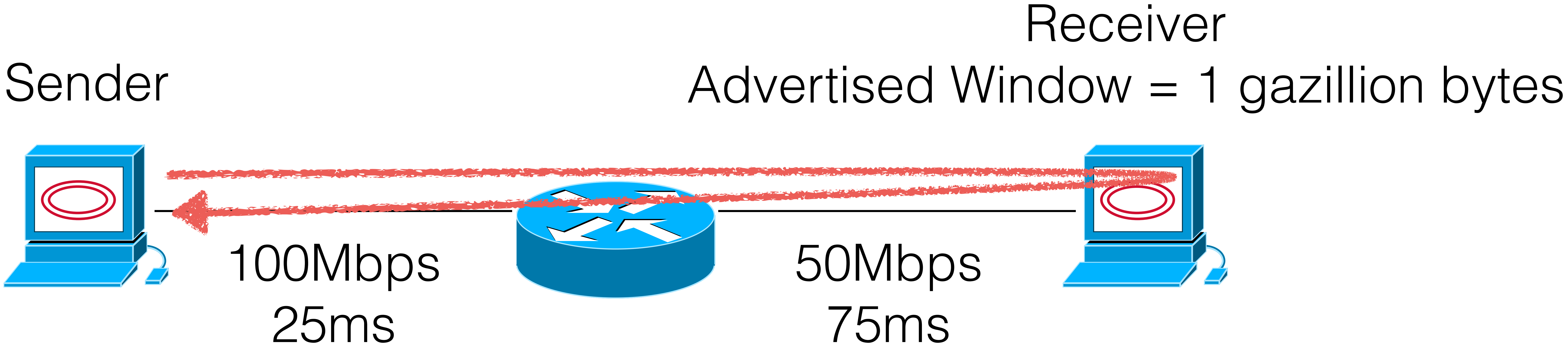
*bandwidth-delay product.*

# Pipe Model



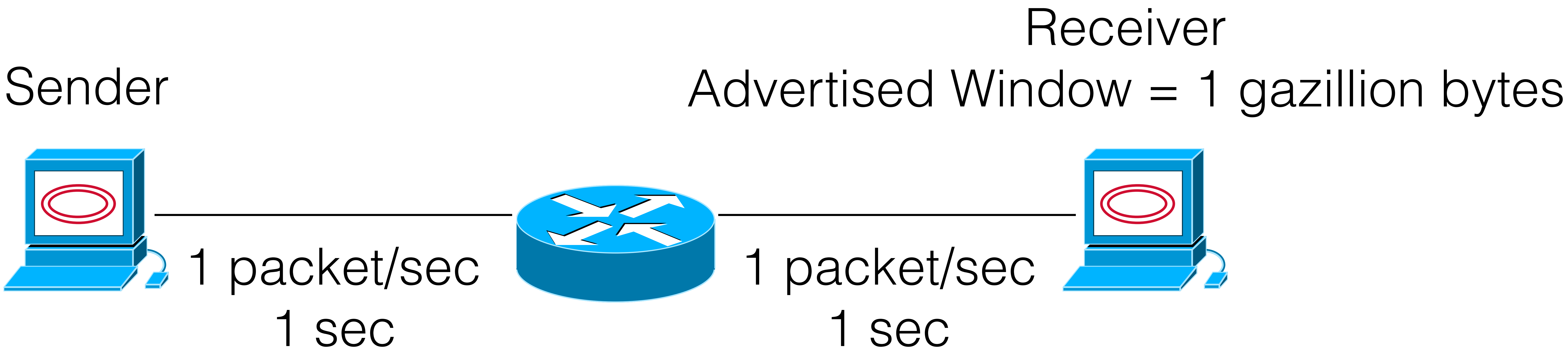

- Bandwidth-Delay Product (BDP): “volume” of the link
    - amount of data that can be “in flight” at any time
    - propagation delay × bits/time = total bits in link

# When we set our window to the BDP, we get into a very convenient loop called “ACK Clocking”




One round-trip-time (RTT) = 200 milliseconds

# I receive new ACKs back at *just* the right rate so that I can keep transmitting at 1 packet/sec.



# How big should we make the window?

- Window should be:
  - Less than or equal to the advertised window so that we do not overload the receiver.
    - This is called Flow Control.
  - Less than or equal to the bandwidth-delay product so that we do not overload the network.
    - This is called Congestion Control.
- (That's it).

# What are we missing?

How do we actually figure out the BDP?!?!

# Today's Agenda

- #1: Starting/Closing the Connection
  - Headers, mechanics
- #2: Deciding how big to set the window: **Equal to BDP**
  - Analysis, algorithms
  - **How do we compute the BDP?**

# Problem Constraints

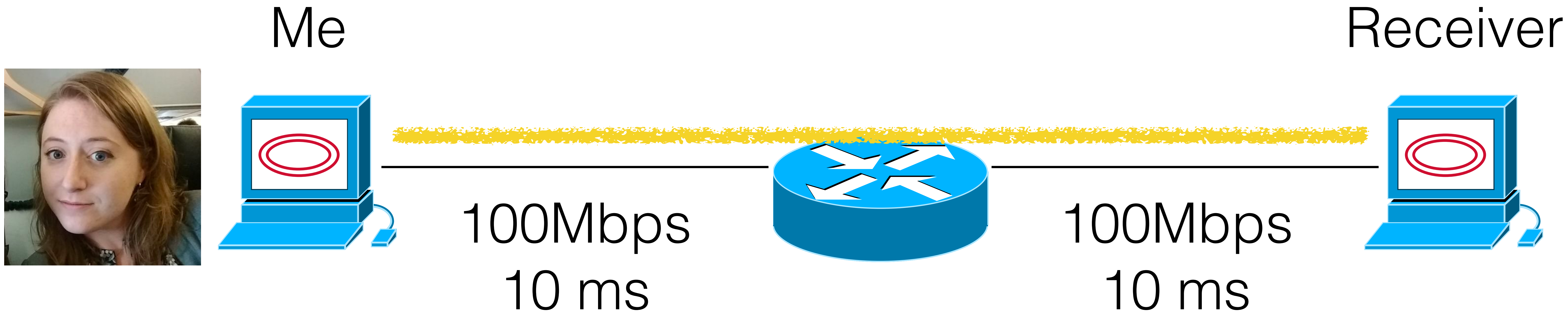
- The network does not tell us the bandwidth or the round trip time.
    - *Implication: Need to infer appropriate window size from the transmitted packets.*

Let’s make it harder…

# Problem Constraints

- The network does not tell us the bandwidth or the round trip time.
- My share of bandwidth is dependent on the other users on the network.

# My window size: 100Mbps x 10ms

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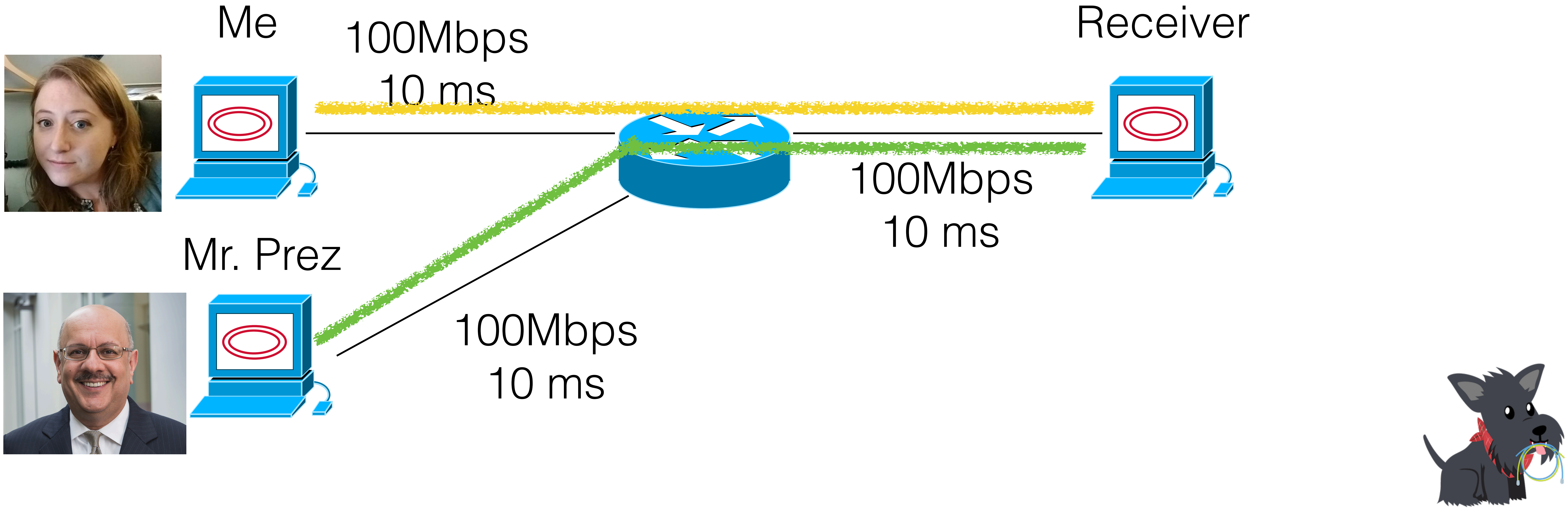

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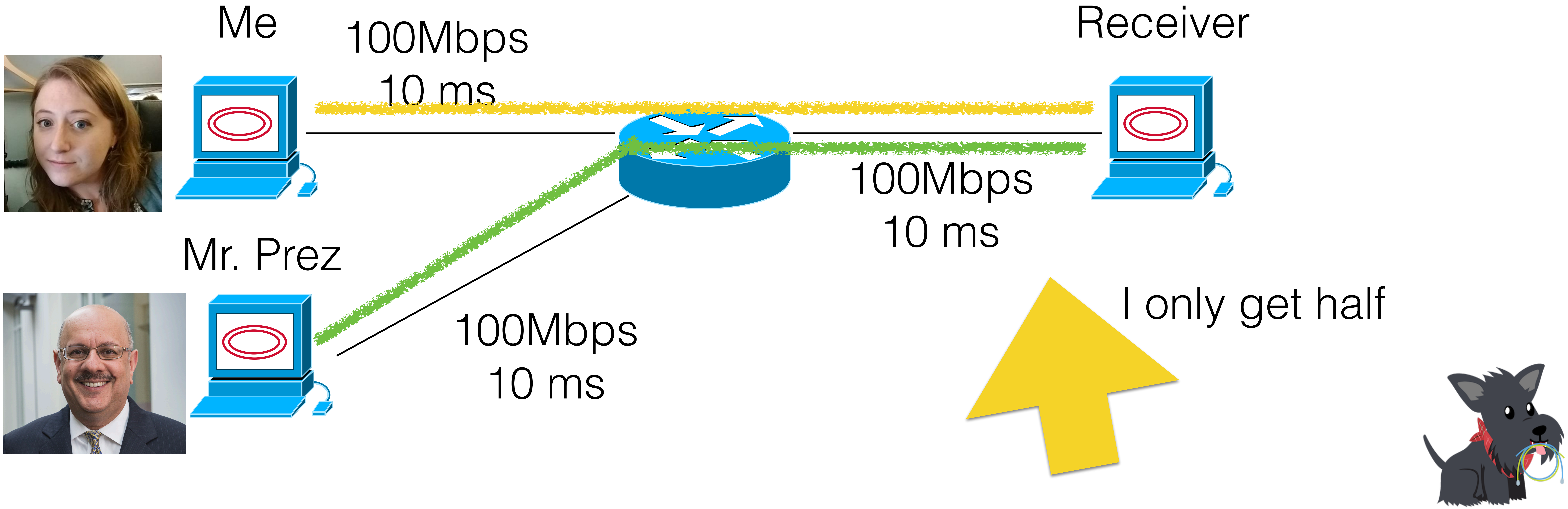
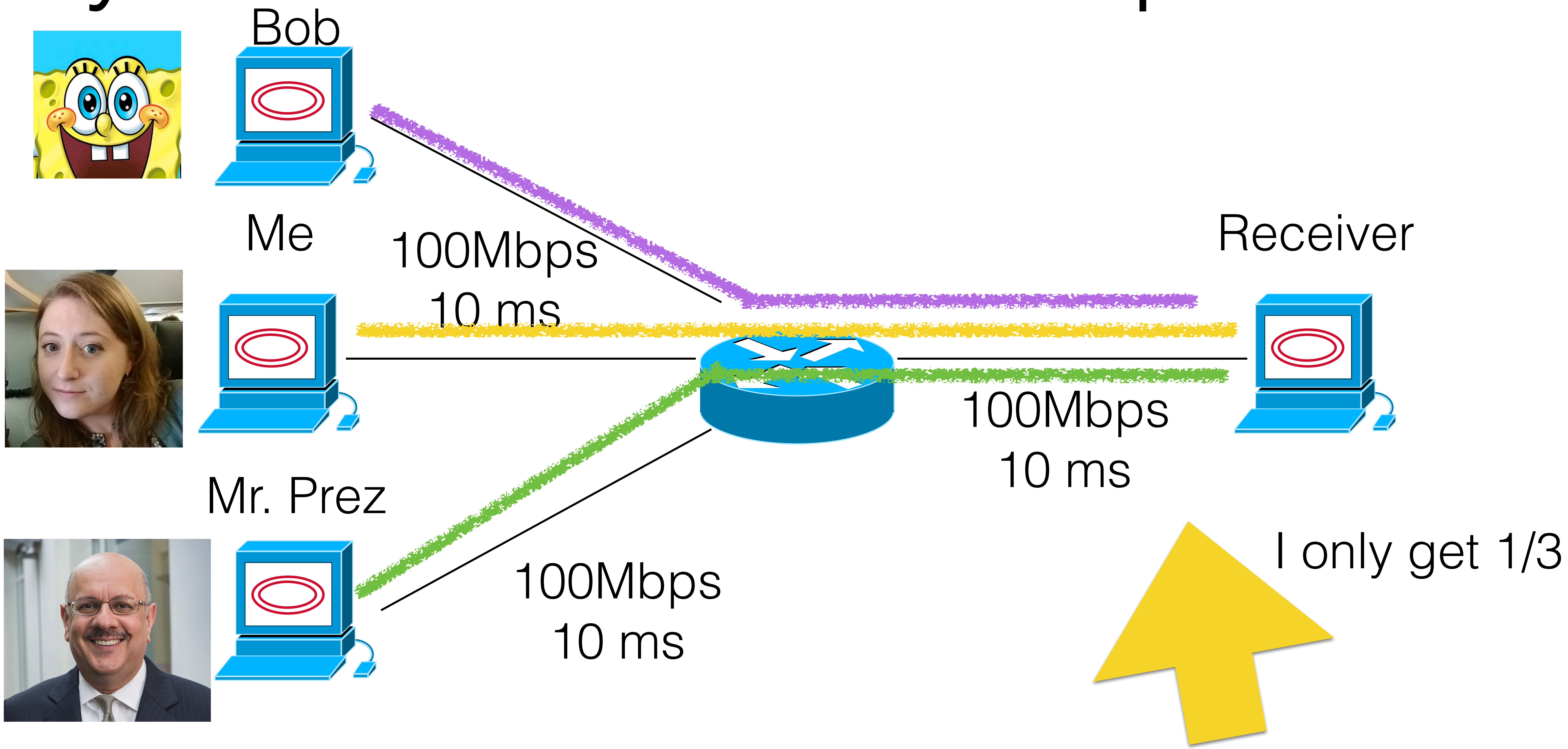


# My window size: 33Mbps x 10ms



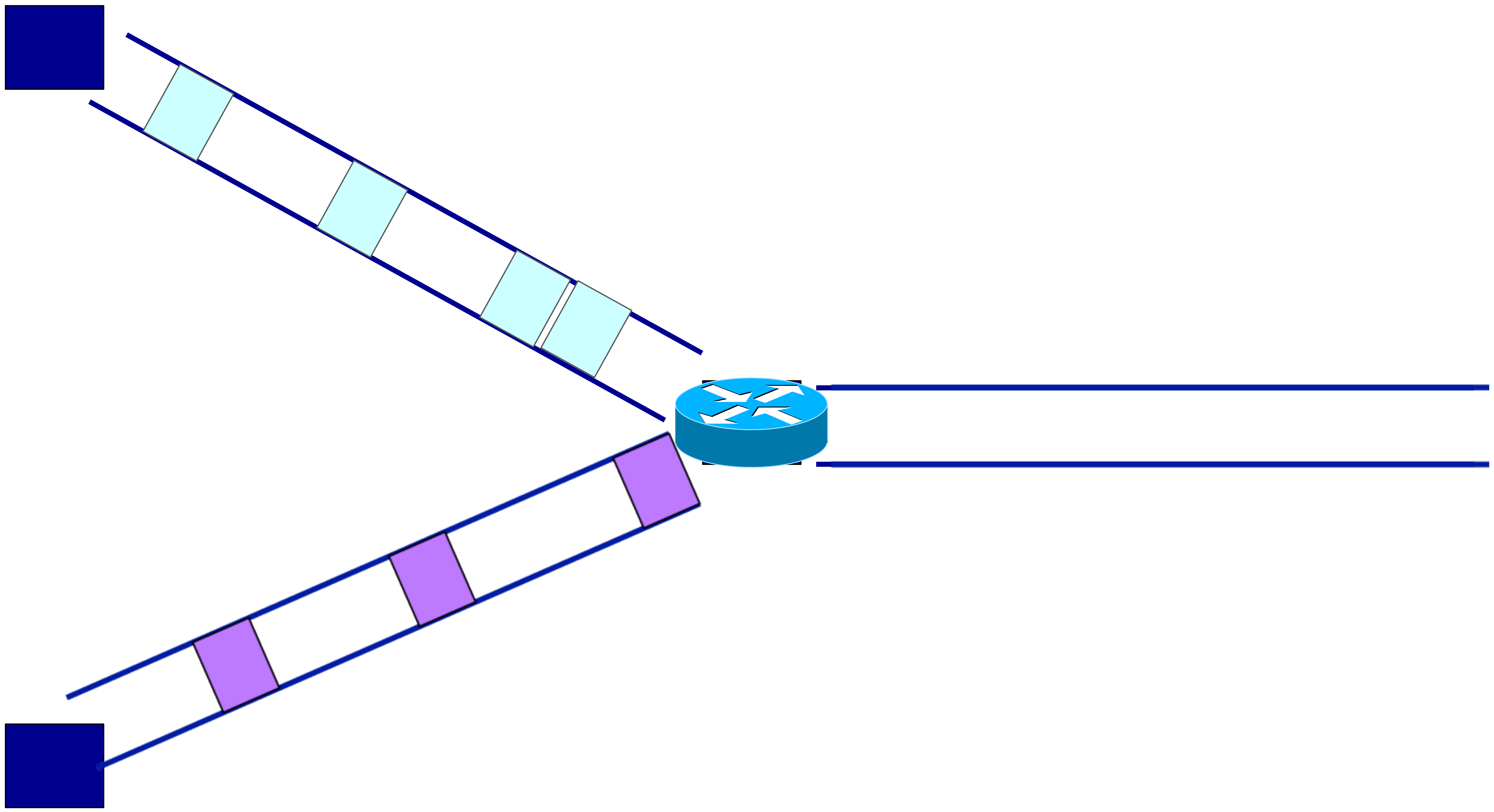
# Problem Constraints

- The network does not tell us the bandwidth or the round trip time.
- My share of bandwidth is dependent on the other users on the network.
  - *Implication: my window size will change as other users start or stop sending.*

# Problem Constraints

- The network does not tell us the bandwidth or the round trip time.
- My share of bandwidth is dependent on the other users on the network.
- Excess packets may not be dropped, but instead stalled in a bottleneck queue.

# All routers have queues to avoid packet drops.



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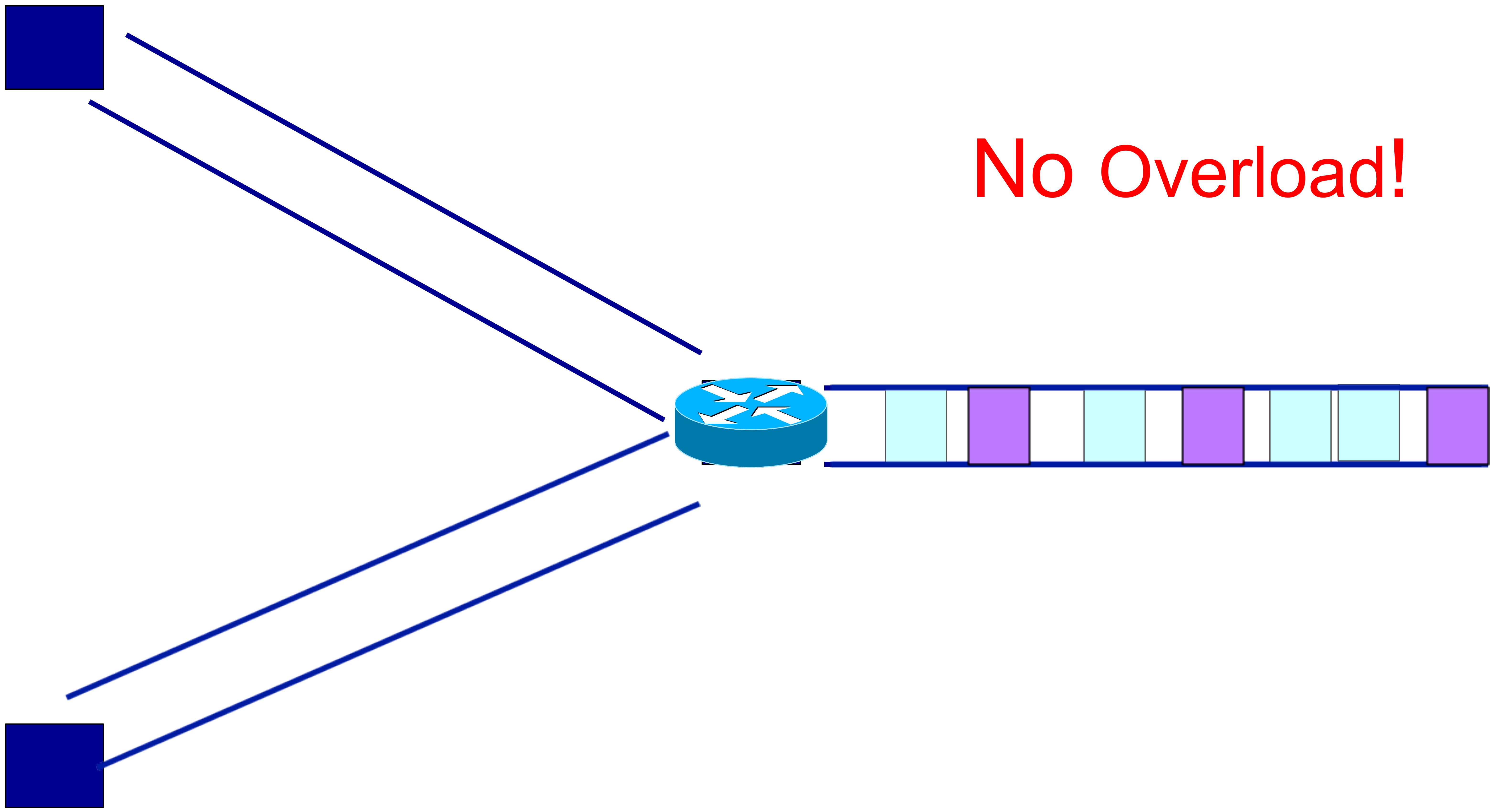
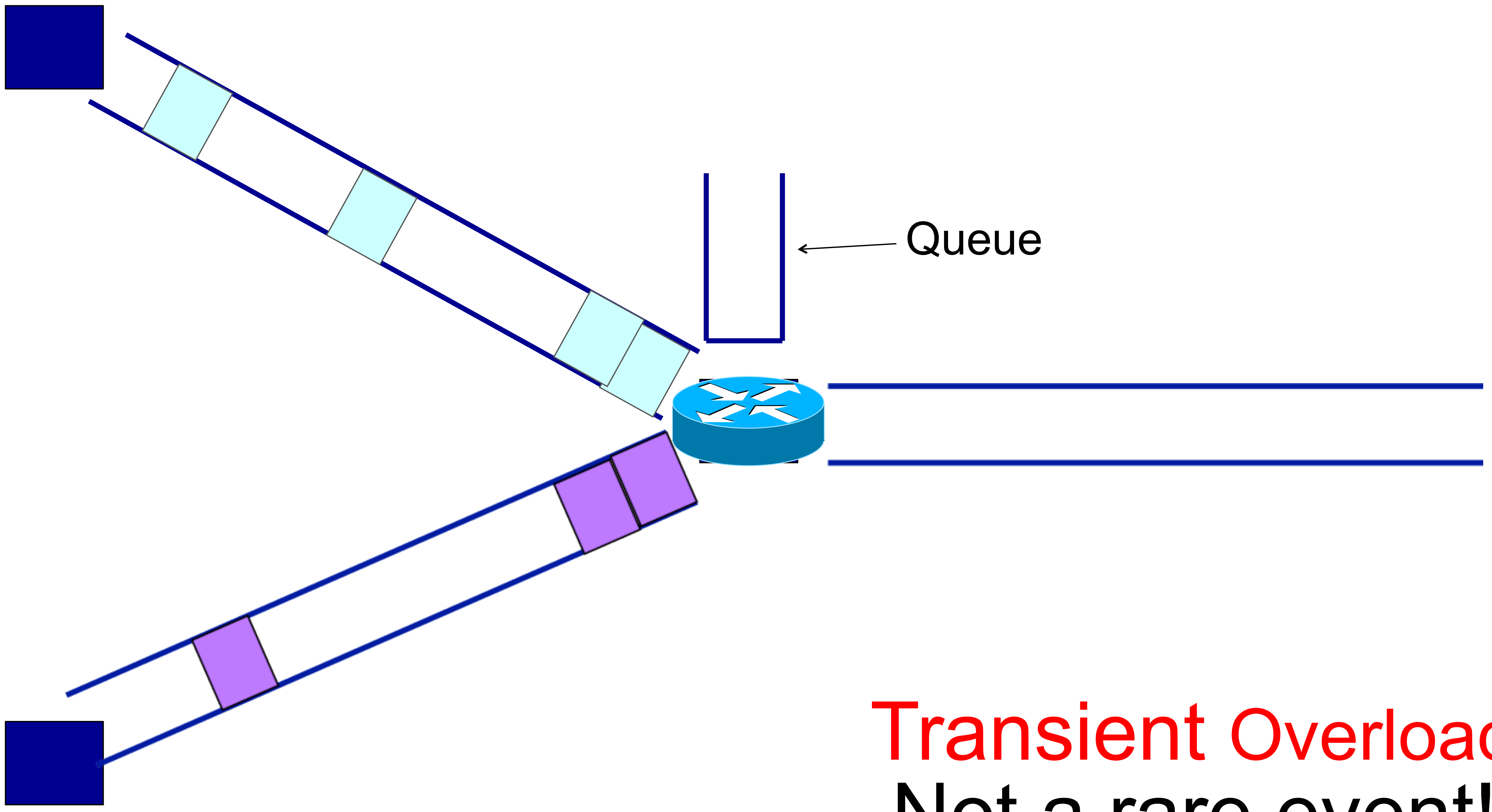


# Statistical multiplexing: pipe view




Transient Overload

Not a rare event!

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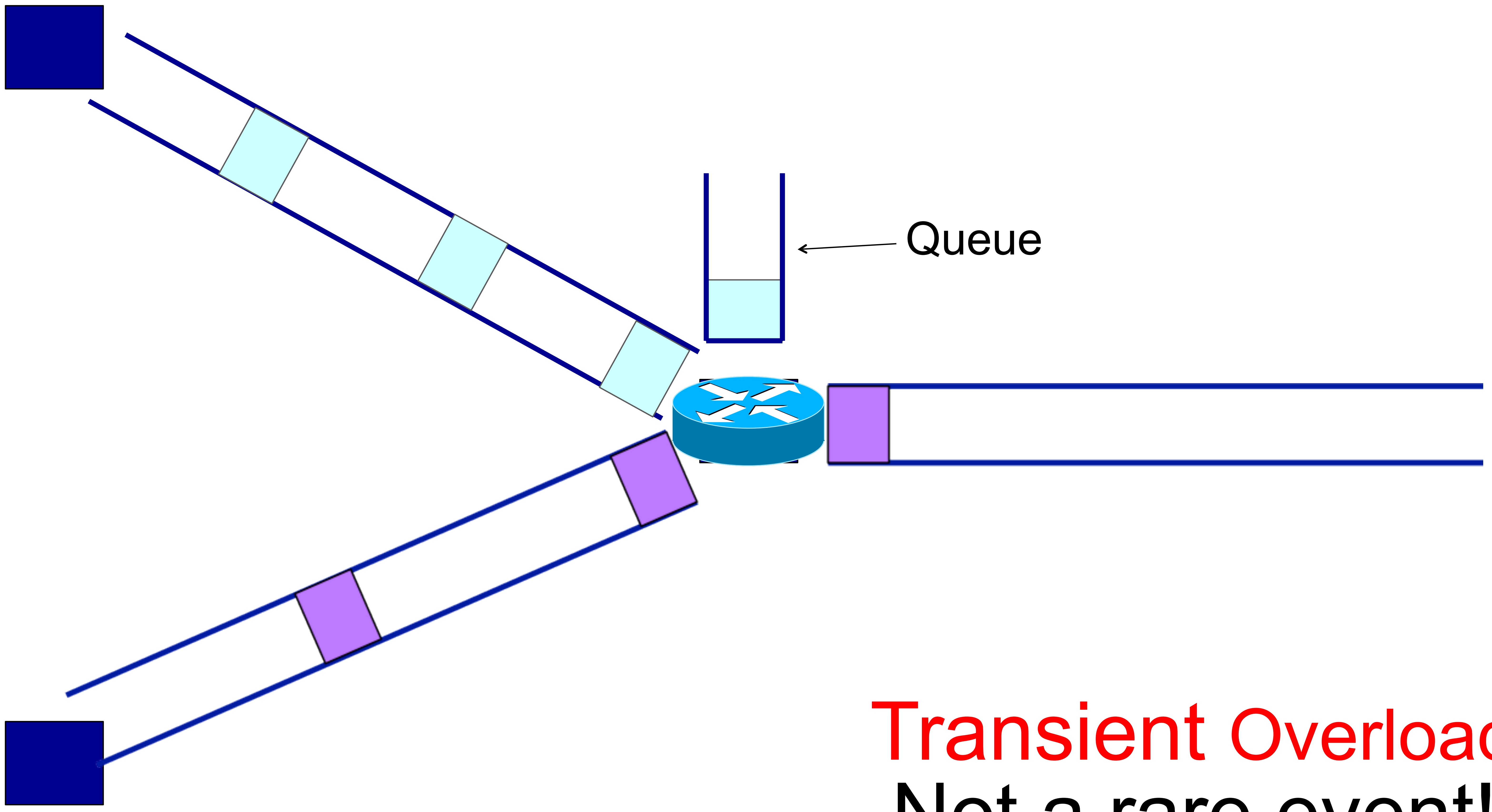


Transient Overload
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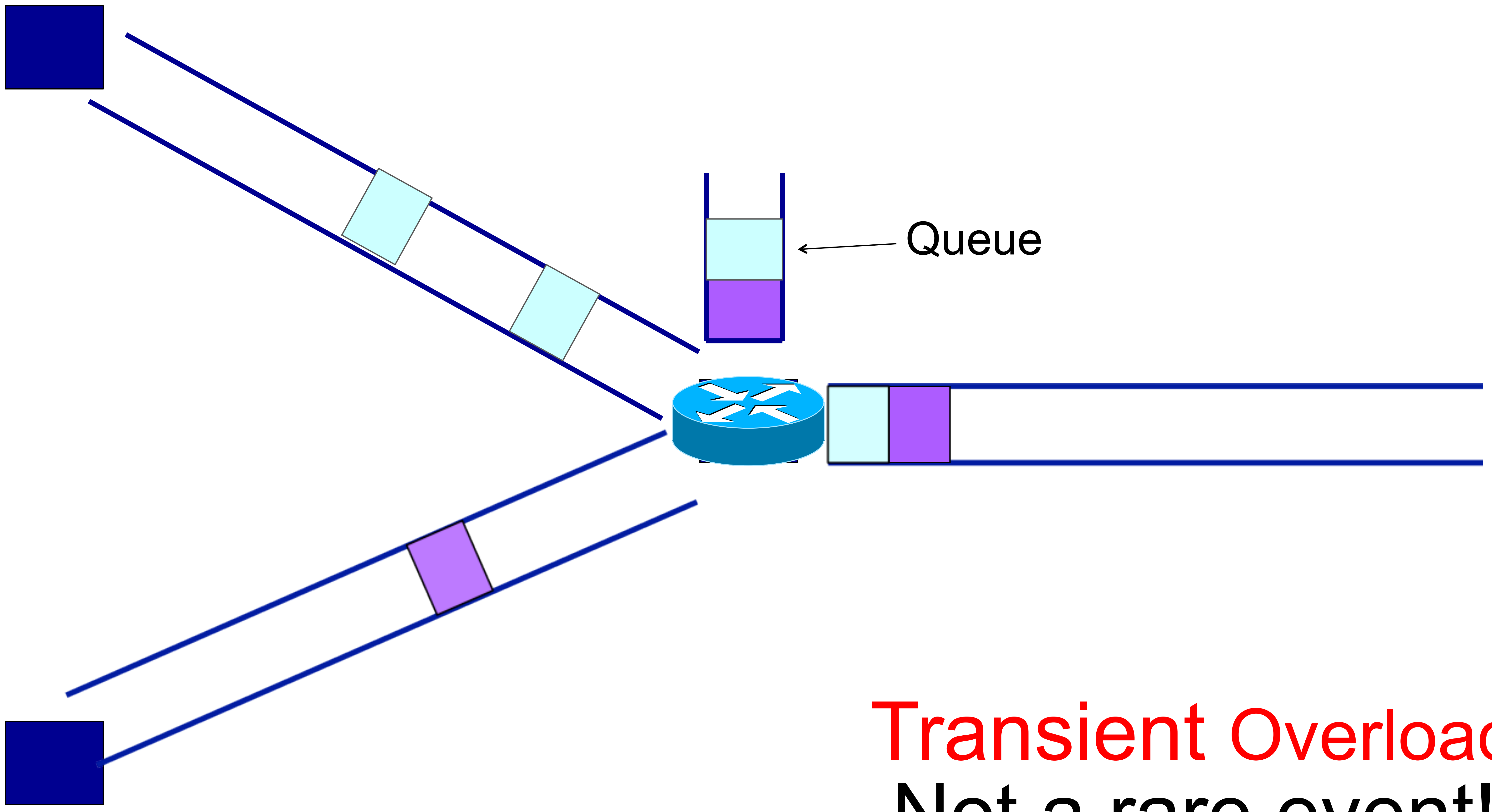


Transient Overload
Not a rare event!

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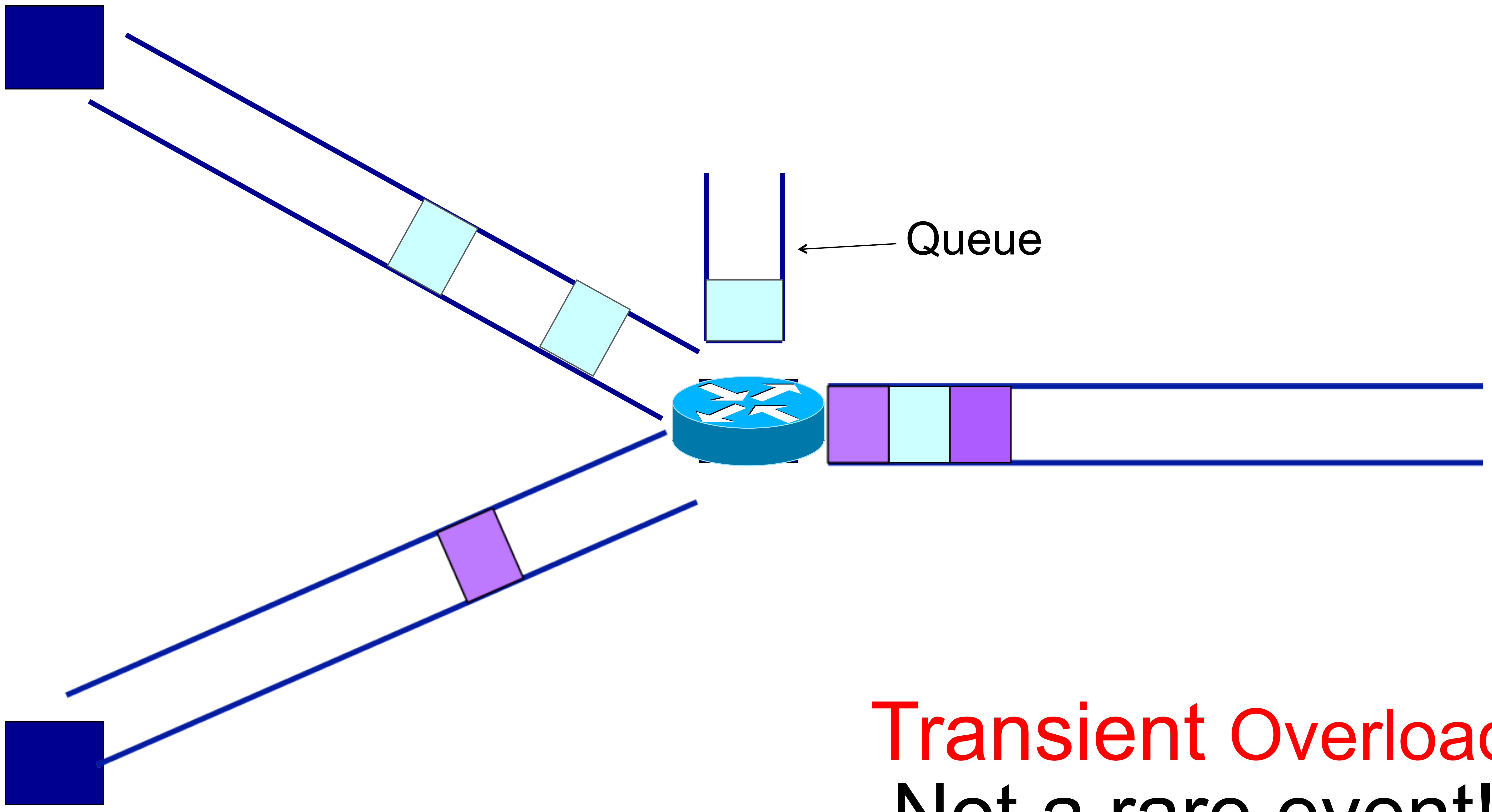


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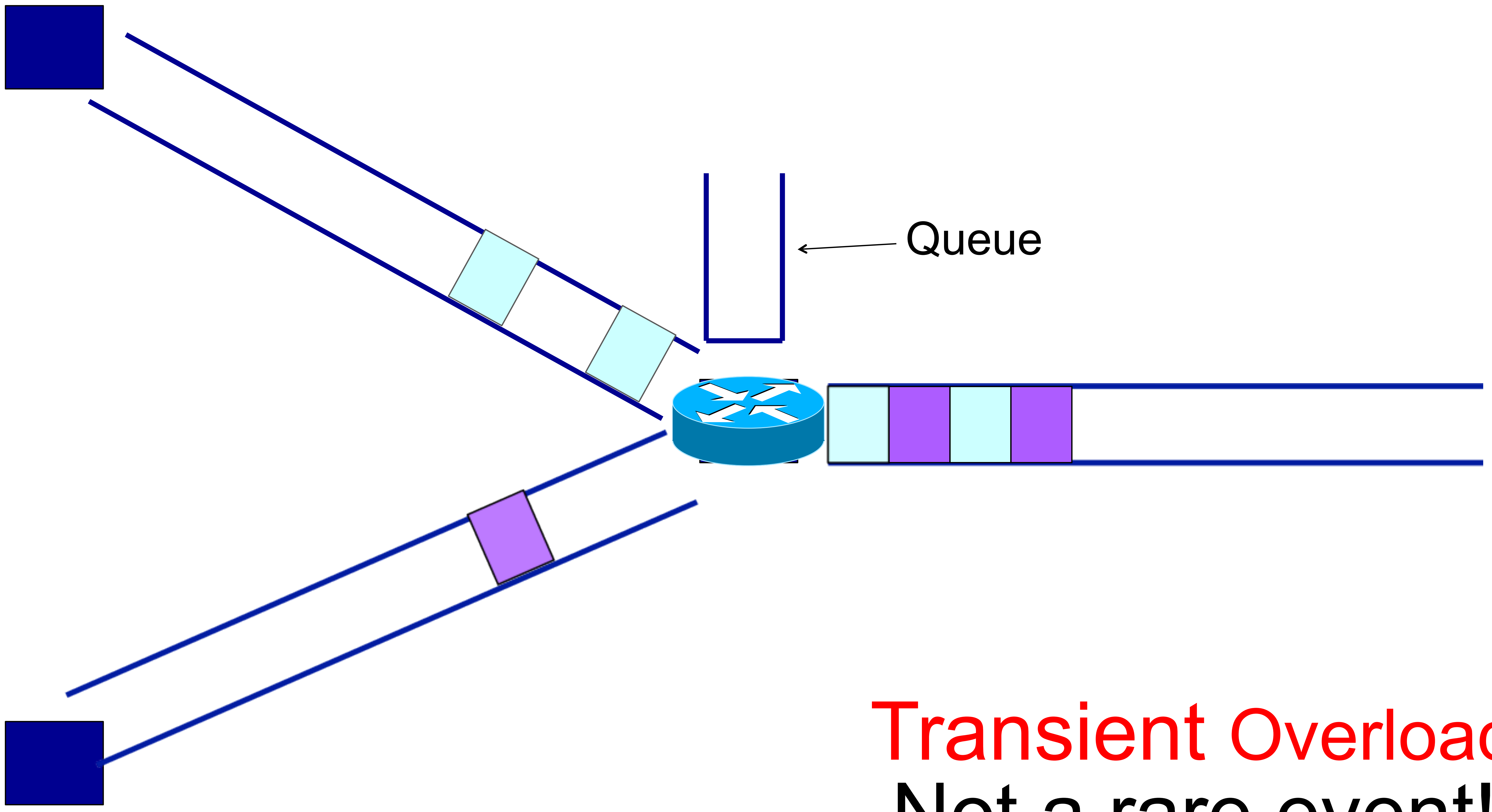


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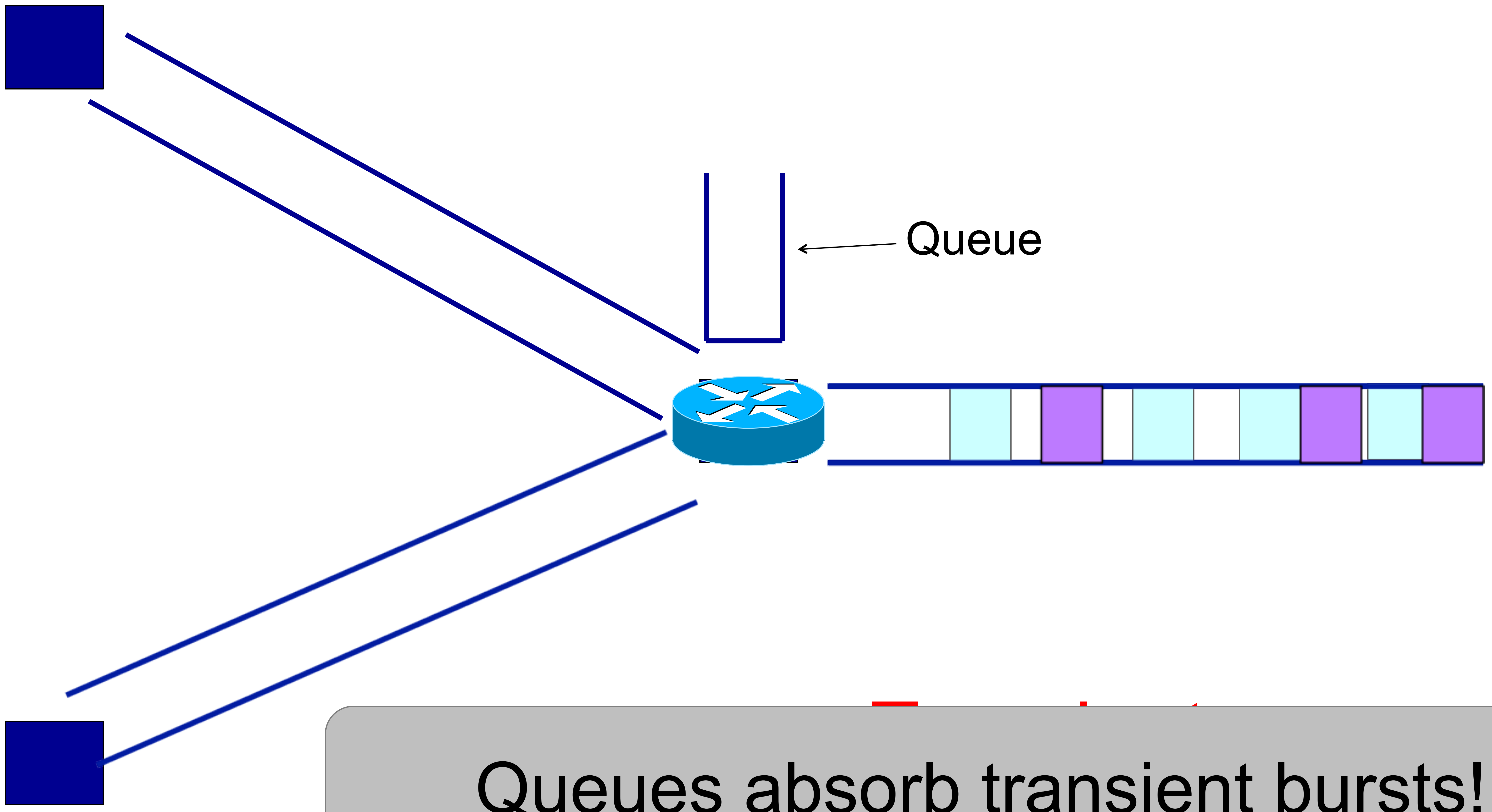
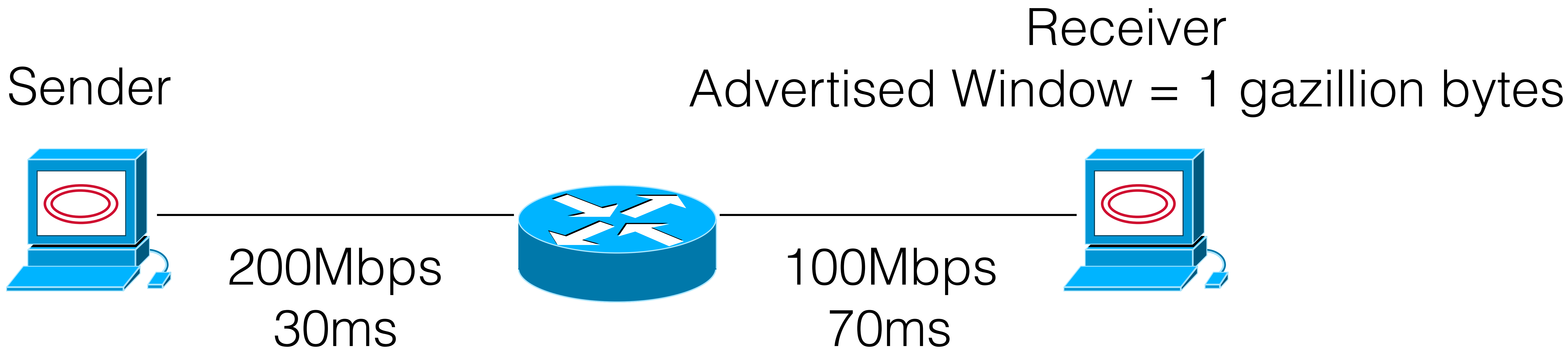
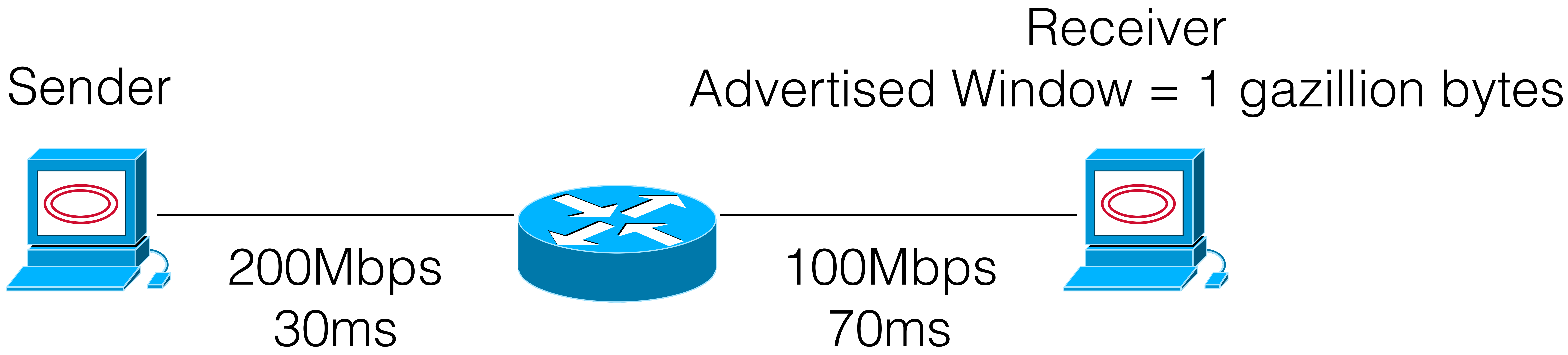



Queues absorb transient bursts!

# BDP: 100Mbps * 200ms = 2.5MB

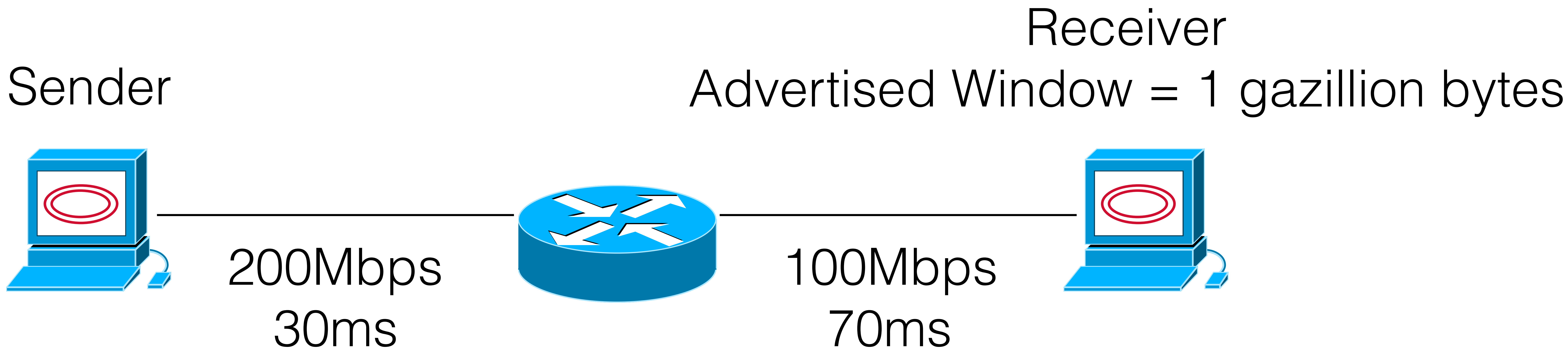


# BDP: 100Mbps * 200ms = 2.5MB




If I have 1000B payloads, my window will be 2500 packets.

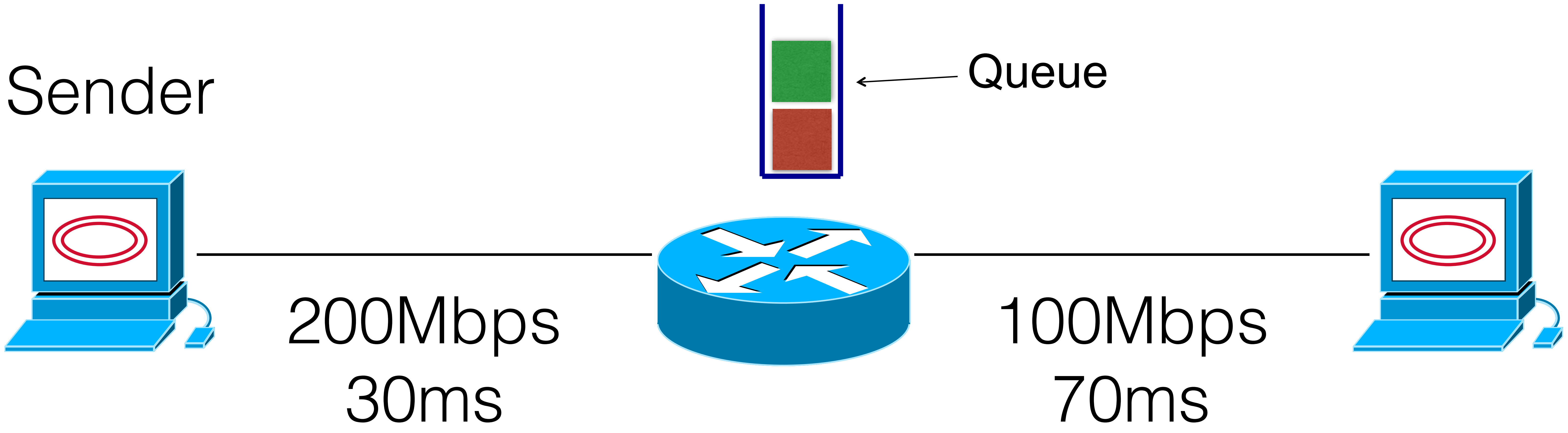
# BDP: 100Mbps * 200ms = 2.5MB




Will packets get dropped if I set my window to, say, 2.6MB or 2600 packets?

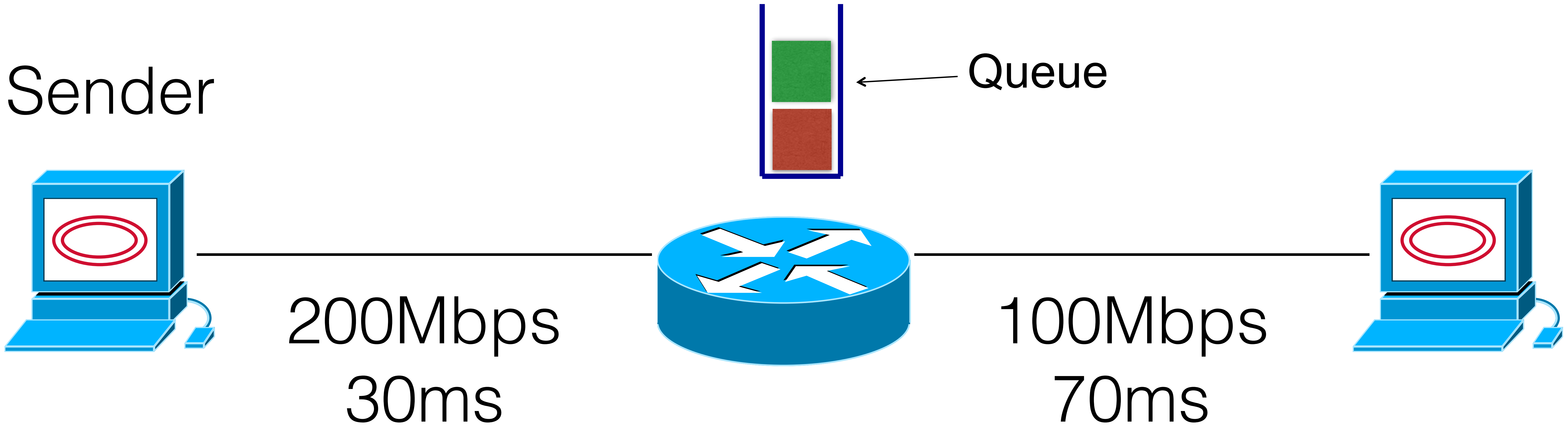
What do you think?

# BDP: 100Mbps * 200ms = 2.5MB




If the queue can hold 100 more packets, none will be dropped!

# BDP: 100Mbps * 200ms = 2.5MB




If the queue cannot “absorb” the extra packets, they will be dropped.

# Problem Constraints

- The network does not tell us the bandwidth or the round trip time.
- My share of bandwidth is dependent on the other users on the network.
- Excess packets may not be dropped, but instead stalled in a bottleneck queue.
    - *Implication: It’s okay to “overshoot” the window size, a little bit, and you still won’t suffer packet loss.*

**Congestion Control Algorithm:** An algorithm to determine the appropriate window size, given the prior constraints.

# There are *many* congestion control algorithms.

- TCP Reno and NewReno (the OG originals)
- Cubic (Linux, OSX)
- BBR (Google)
- LEDBAT (BitTorrent)
- Compound (Windows)
- FastTCP (Akamai)
- DCTCP (Microsoft Datacenters)
- TIMELY (Google Datacenters)
- Other weird stuff (ask Ranysha on Thursday)

# Some History: TCP in the 1980s

- Sending rate only limited by flow control
  - Packet drops → senders (repeatedly!) retransmit a full window's worth of packets
- Led to "congestion collapse" starting Oct. 1986
  - Throughput on the NSF network dropped from 32Kbits/s to 40bits/sec
- "Fixed" by Van Jacobson's development of TCP's congestion control (CC) algorithms

# Van Jacobsen

Internet Hall of Fame
Kobayashi Award
SIGCOMM Lifetime Achievement Award

- Inventor of TCP Congestion Control
- “TCP Tahoe”
- More recently, one of the co-inventors of Google’s BBR
- Author of many networking tools (traceroute, tcpdump)

LITERALLY SAVED THE INTERNET FROM COLLAPSE

# Jacobson's Approach

- Extend TCP's existing window-based protocol but adapt the window size in response to congestion
  - required no upgrades to routers or applications!
  - patch of a few lines of code to TCP implementations

- A pragmatic and effective solution
  - but many other approaches exist

- Extensively improved upon
  - topic now sees less activity in ISP contexts
  - but is making a comeback in datacenter environments

The default TCP everyone teaches is TCP Reno, so that is what we will teach in this class.

* Even though Reno isn't what Jacobsen invented.

** Even though our research at CMU suggests that it's almost extinct — no one (except Netflix) uses it anymore

# TCP Reno: General Blueprint

- If a packet is lost, slow down! The packet is a signal that you are sending *too fast.*
- If you have been sending for a while and no packets are lost, speed up! No loss is a signal that you are probably are sending less than the link capacity.

# How much should we slow down? Speed up?

- AIAD: Additive Increase, Additive Decrease
  - Every RTT, I increase my window by one. Every time I have a loss, I decrease my window by one.
- MIAD: Multiplicative Increase, Additive Decrease
  - Every RTT, I increase my window by 2x. Every time I have a loss, I decrease my window by one.
- AIMD: Additive Increase, Multiplicative Decrease
  - Every RTT, I increase my window by 1. Every time I have a loss, I decrease my window by 2x.
- MIMD: Additive Increase, Multiplicative Decrease
  - Every RTT, I increase my window by 2x. Every time I have a loss, I decrease my window by 2x.

# Let's Try It

- Turn to a partner. One of you will be "the network", the other will be "the sender."

- Network:
    - Choose a random number between 1 and 30. This is your BDP.
    - Every time your partner guesses, tell them "drop" if they overshoot, or "no drop" if they undershoot.
    - On a piece of paper, keep track of how many times your partner guessed, and keep track of how many packets are "lost"
        - If my partner guesses 40, and my secret number is 28, we "lost" 12 packets and transmitted 28.

- Sender:
    - Choose an algorithm (AIMD, MIMD, MIAD, or AIAD) and an *initial window size* — a random number from 1-30 that is your first window size.
    - Tell your partner "I transmit $windowsize packets"
        - Your partner will tell you whether there were dropped packets or no dropped packets.
    - Adjust your window according to the algorithm and then make another guess.

# Who thinks they had a good algorithm/initial window size?

- What algorithm did you choose?
    - Why is it a good algorithm?
- What initial window size did you choose?
    - Why is it a good initial window size?

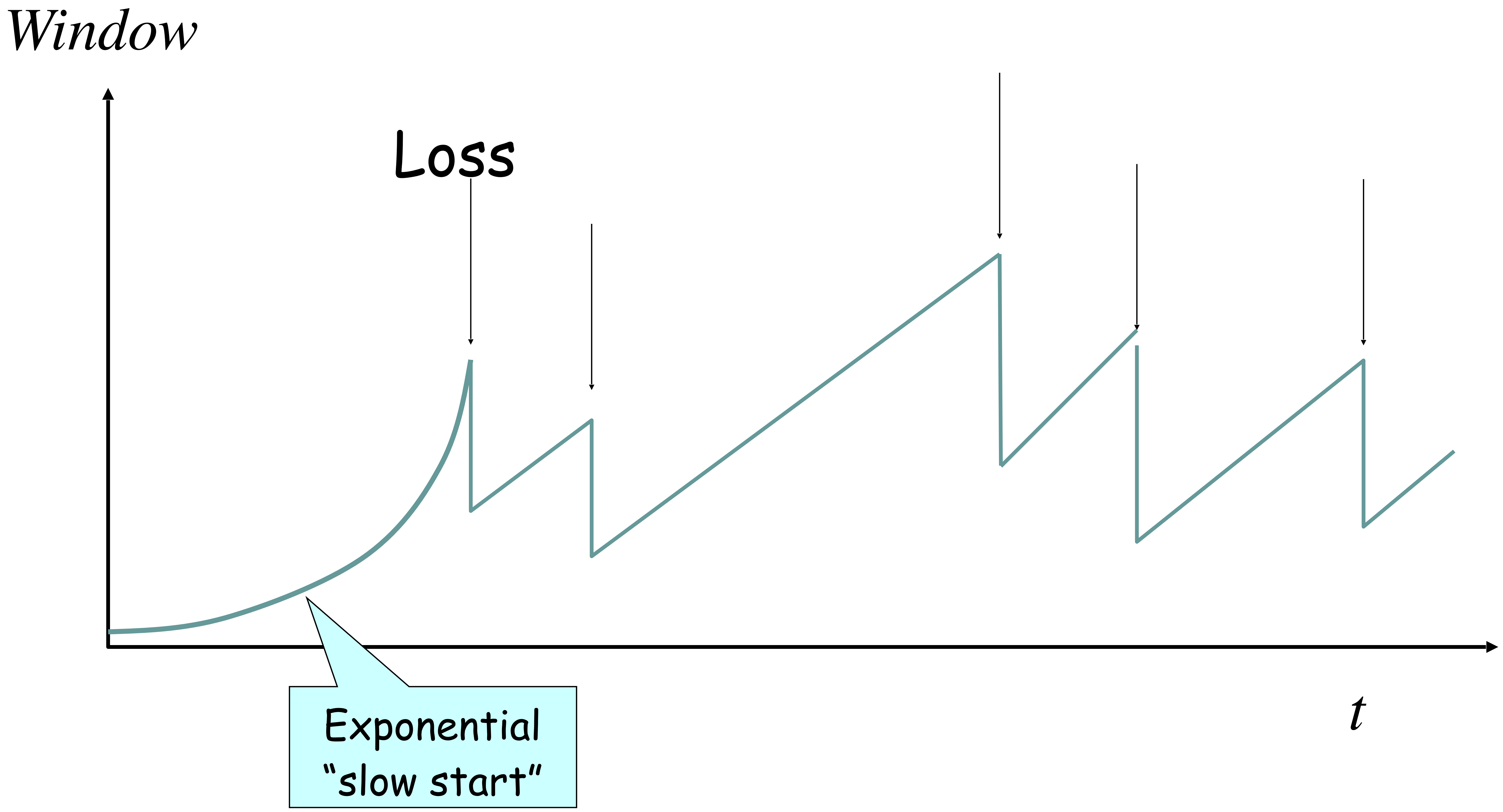
# Challenges

- If you overshoot, lots of packets can be lost — for you and anyone else sharing the link!
    - Wastes network resources
    - Slows down transmission overall (have to wait for timers to go off)
    - Wastes CPU time (complicates book-keeping at sender and receiver)
- If you undershoot your transmission is slower than it could be…. :(

# TCP Reno

- Uses Multiplicative Increase at startup to find the “right” sending rate quickly. Initial window size is set to 4.
    - For historical reasons this is called “slow start” — senders used to just pick an insane high initial window size and this was “slower” than that.
- Under normal operation, uses Additive Increase/Multiplicative Decrease (AIMD) to adjust the sending rate over time.

# Leads to the TCP “Sawtooth”



# Slow-Start vs. AIMD

- When does a sender stop Slow-Start and start Additive Increase?
- Introduce a "slow start threshold" (ssthresh)
  - Initialized to a large value
- When window = ssthresh, sender switches from slow-start to AIMD-style increase
  - Or if a drop happens.

# Why AIMD?

- Key idea:
  - Be cautious in consuming new resources
    - So we don't cause another congestion collapse!
  - Be aggressive in slowing down at packet drops.
    - So we don't cause another congestion collapse!
- Other nice properties: AIMD is guaranteed to converge to a *fair share* between two senders sharing the same link with the same RTT.
  - More on this later.

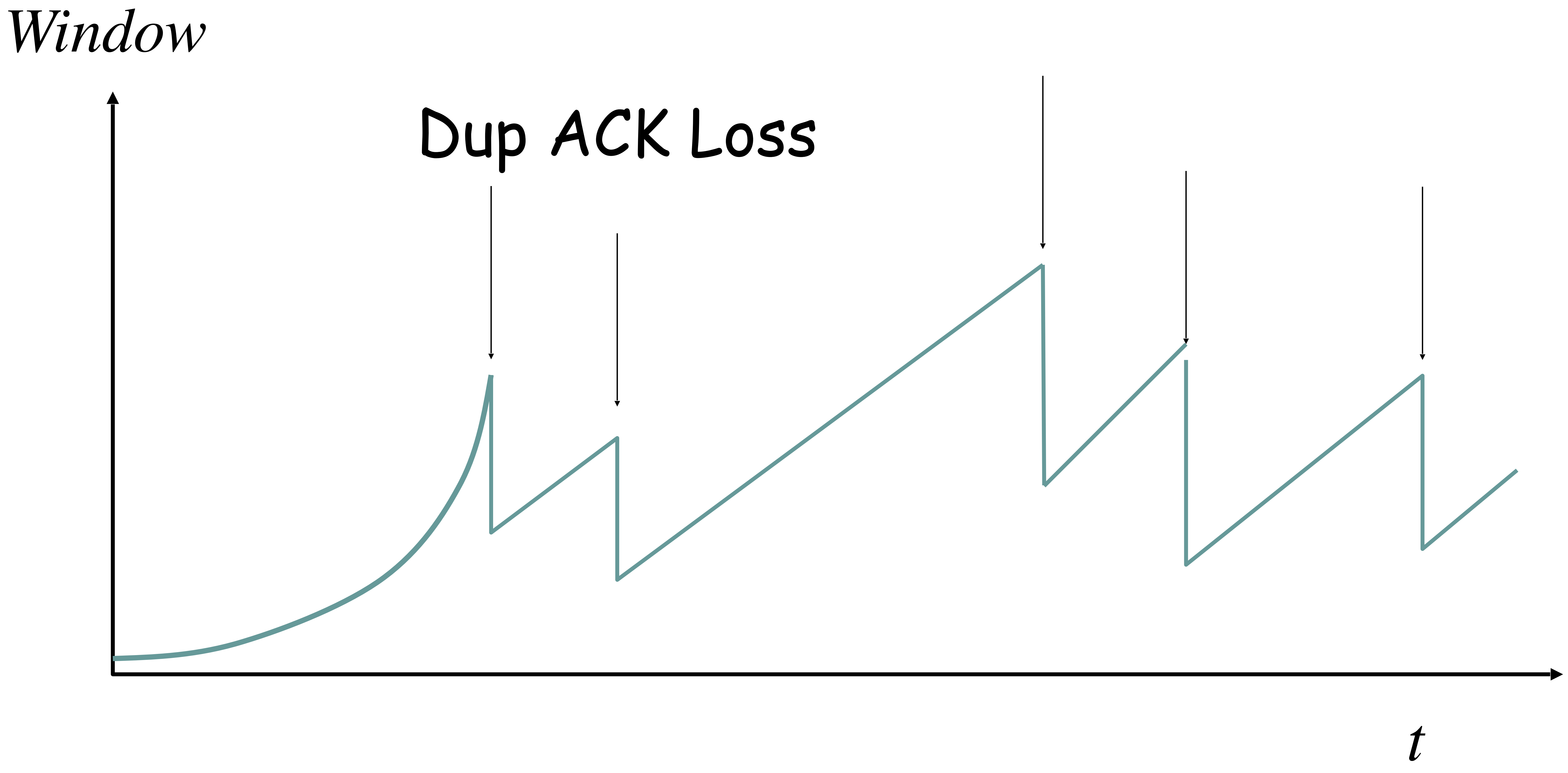
# AIMD Mechanics in Reno

- “CWND” is the measured “congestion window”
  - Sending window is min(CWND, Advertised Window)
- Reno follows three key stages to determine CWND:
  - (1) Slow start, where it uses multiplicative increase
  - (2) Congestion avoidance, where it uses additive increase
  - (3) Fast recovery, where it “recovers” from “easy” packet losses.
    - *What do you mean, Easy Packet Losses?*

# Duplicate ACKs

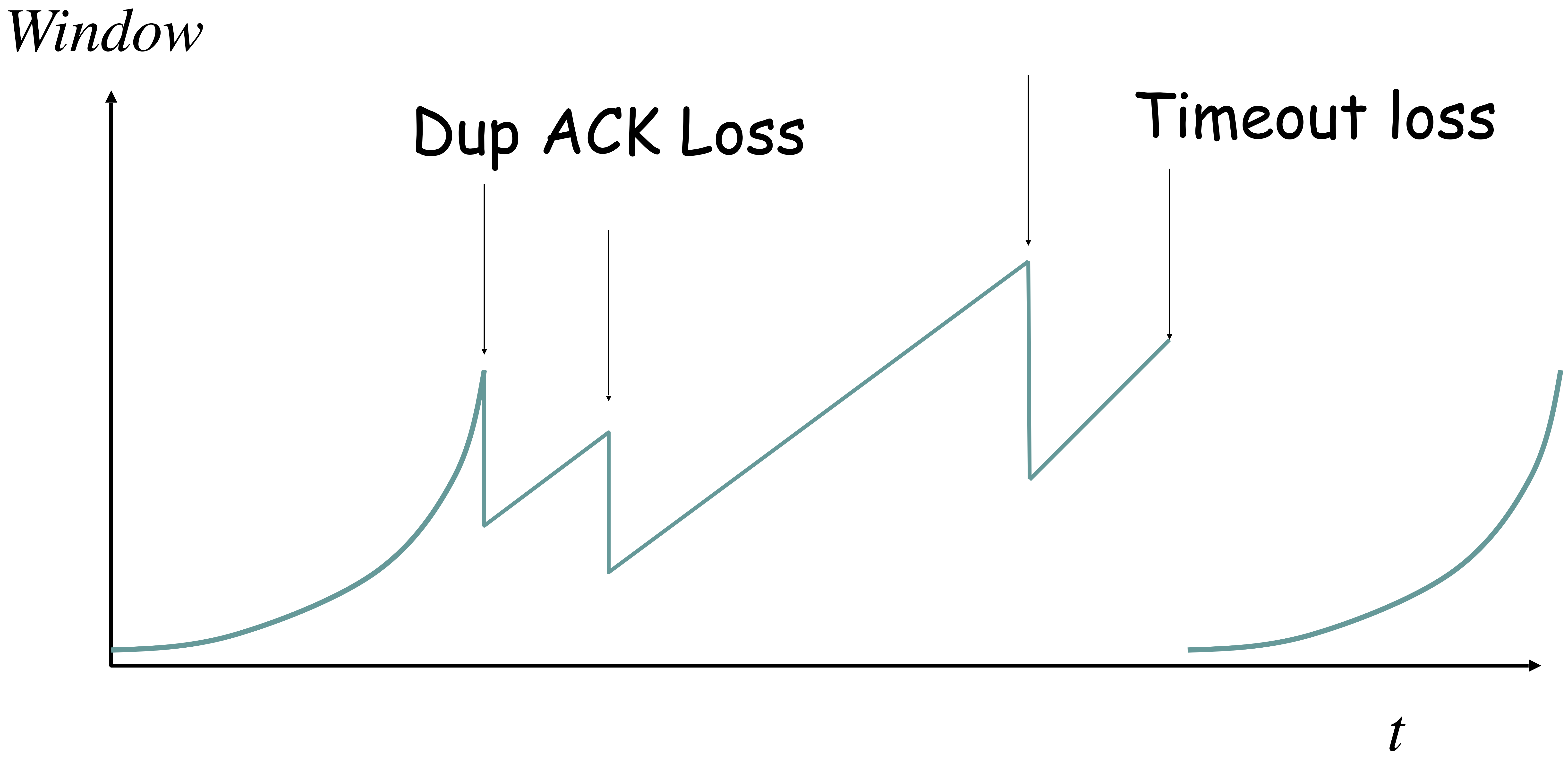
- I can pre-emptively figure out that loss has happened without a timer going off.
- How?
  - Say I receive packets with MSS 1000, sequence numbers 1000, 2000, 4000, 5000, 6000….
  - I know I missed 3000!
- Recall that TCP uses cumulative ACKs — I ACK the next byte such that I have the data for all bytes lower than that.
  - If I see the same “dup” ACK three times, I determine there is a loss.

# Leads to the TCP "Sawtooth"

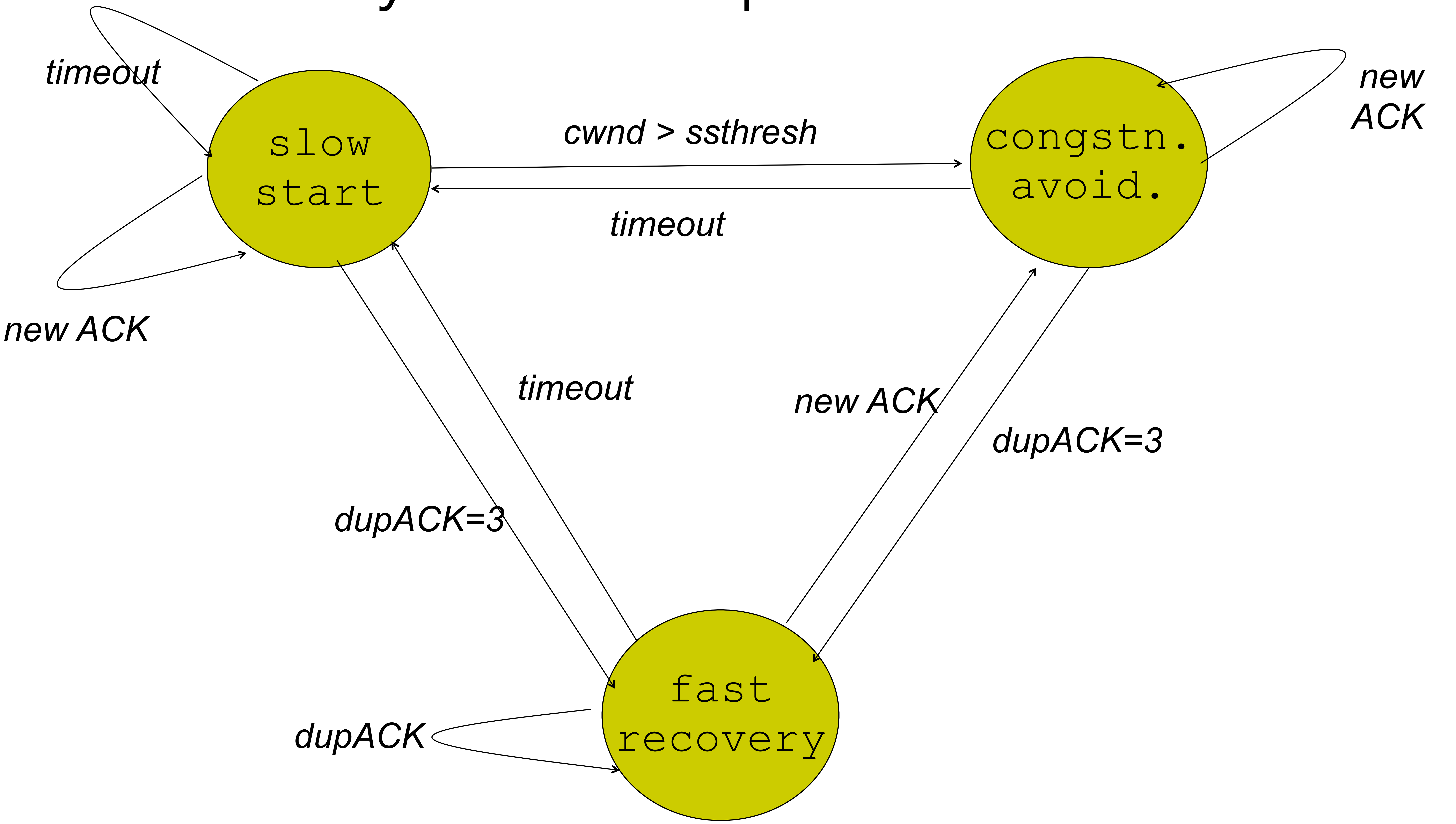


# Assumption: Timeout Losses are Worse

- Timeout *can mean* (but not always) that lots of packets were lost and I have severely overshot.
- So I should react more severely to a timeout.
- Instead of halving my window, I will go all the way back to slow start and start over again!


Window
Dup ACK Loss
Timeout loss
t

# Print this out and tape it above your bed. This is what you will implement for P2 CP2!



# Summary

- All TCP connections use the same handshake, initial sequence number exchange, etc.
- But determining the right window size is *hard* because the network does not tell us directly how much capacity is available to us!
  - There are lots of algorithms to measure “CWND”
  - Reno is the classic algorithm, and it uses AIMD.

# On Tuesday

- Visiting speaker: Dr. T-Y Huang from Netflix
  - She works on making video streaming algorithms
  - Related to our TCP questions: If I can send you a video at 25Mbps, 15Mbps, 10Mbps, or 5Mbps, what rate should I chose?
  - How should I send the video so that if packets are dropped, your video doesn't have glitches?
  - Watch Piazza this weekend: I will make a post inviting the first ten responders to have (free) lunch with Dr. Huang.

# Next Time with Me…

- Why AIMD converges to fairness
- Calculating TCP throughput with loss
- Problems with TCP Reno
- New TCPs: Cubic, BBR
- Is the Internet fair?