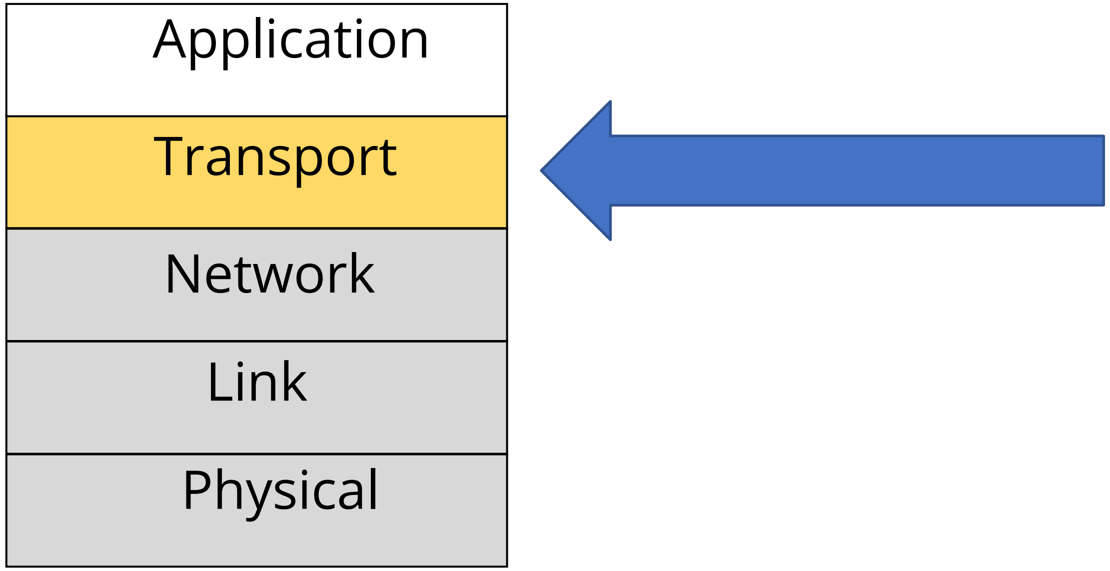
Transport Layer (TCP/UDP)

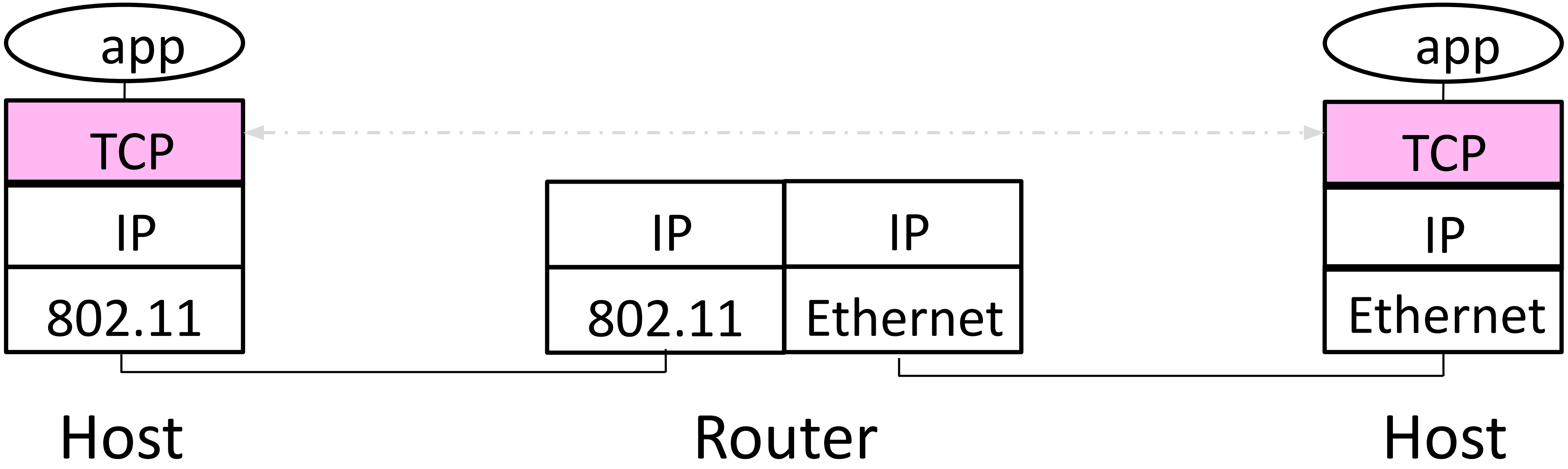
Where we are in the Course

Now: Transport!!!

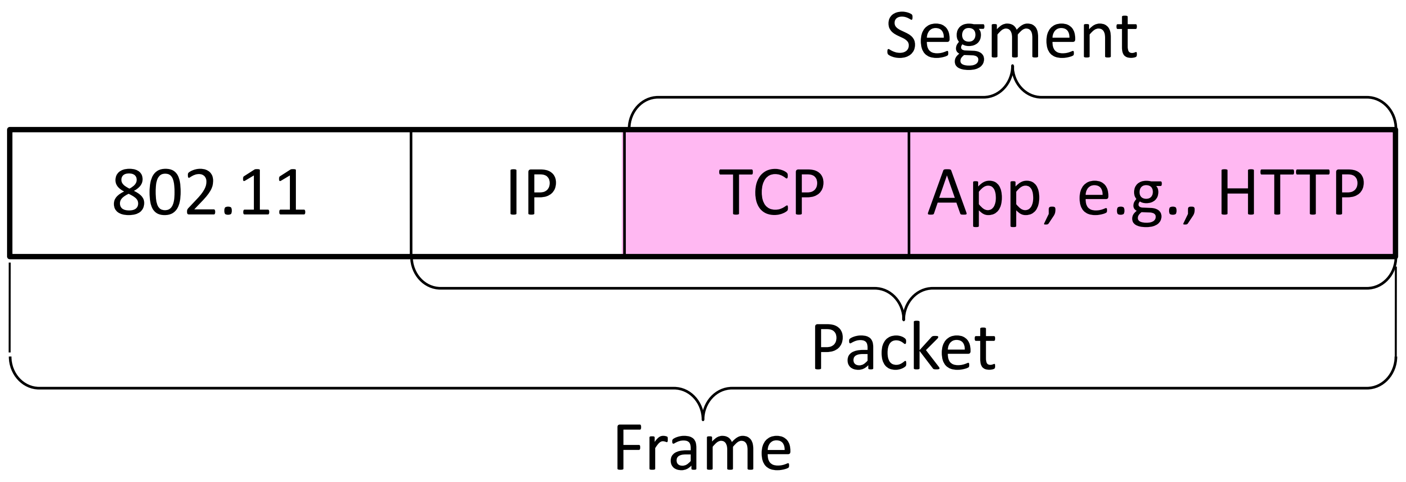
“end-to-end” connectivity across networks of networks



- Transport layer provides end-to-end connectivity across the network



- Segments carry application data across the network
- Segments are carried within packets within frames



Transport Layer Services

- Provide different kinds of data delivery across the network to applications

	Unreliable	**Reliable**
Messages	Datagrams (UDP)	SCTP
Bytestream		Streams (TCP)

Comparison of Internet Transports

- TCP is full-featured, UDP is a glorified packet

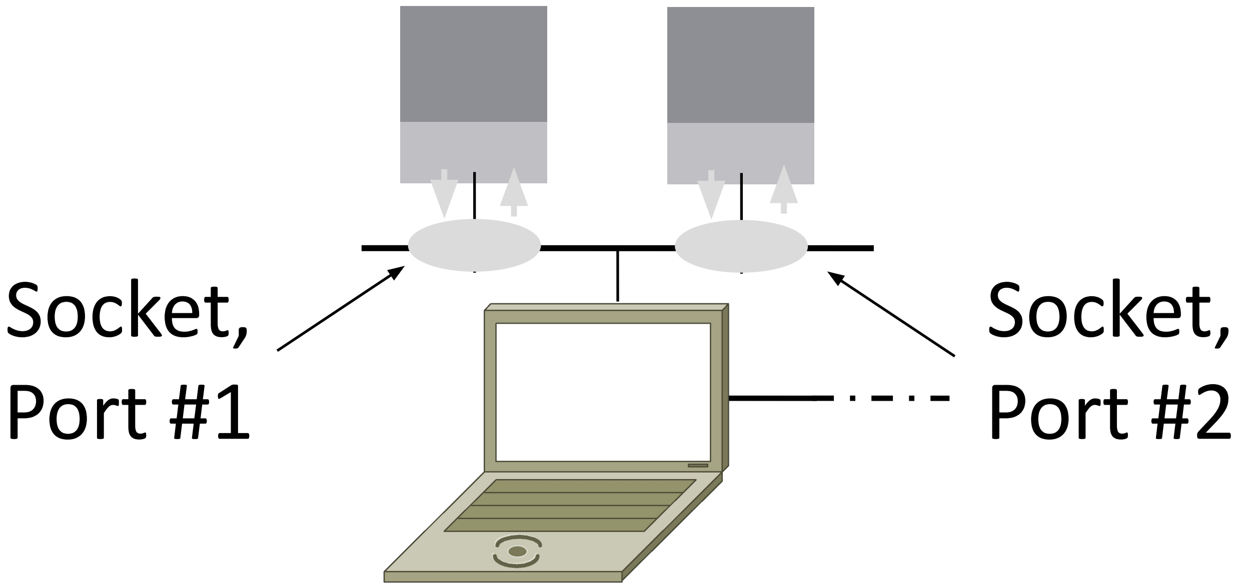
TCP (Streams)	**UDP (Datagrams)**
Connections	Datagrams
Bytes are delivered once, reliably, and in order	Messages may be lost, reordered, duplicated
Arbitrary length content	Limited message size
Flow control matches sender to receiver	Can send regardless of receiver state
Congestion control matches sender to network	Can send regardless of network state

Socket API

- Simple abstraction to use the network
  - The “network” API (really Transport service) used to write all Internet apps
  - Part of all major OSes and languages; originally Berkeley (Unix) ~1983
- Supports both Internet transport services (Streams and Datagrams)

Socket API (2)

- Sockets let apps attach to the local network at different ports



Socket API (3)

- Same API used for Streams and Datagrams

Primitive	Meaning
SOCKET	Create a new communication endpoint
BIND	Associate a local address (port) with a socket
LISTEN	Announce willingness to accept connections
ACCEPT	Passively establish an incoming connection
CONNECT	Actively attempt to establish a connection
SEND(TO)	Send some data over the socket
RECEIVE(FROM)	Receive some data over the socket
CLOSE	Release the socket

Only needed for Streams

To/From for Datagrams

Ports

- Application process is identified by the tuple IP address, transport protocol, and port
  - Ports are 16-bit integers representing local “mailboxes” that a process leases
- Servers often bind to “well-known ports”
  - <1024, require administrative privileges
- Clients often assigned “ephemeral” ports
  - Chosen by OS, used temporarily

Some Well-Known Ports

Port	Protocol	Use
20, 21	FTP	File transfer
22	SSH	Remote login, replacement for Telnet
25	SMTP	Email
80	HTTP	World Wide Web
110	POP-3	Remote email access
143	IMAP	Remote email access
443	HTTPS	Secure Web (HTTP over SSL/TLS)
543	RTSP	Media player control
631	IPP	Printer sharing

Topics

- Service models
  - Socket API and ports
  - Datagrams, Streams
- User Datagram Protocol (UDP)
- Connections (TCP)
- Sliding Window (TCP)
- Flow control (TCP)
- Retransmission timers (TCP)
- Congestion control (TCP)

UDP

User Datagram Protocol (UDP)

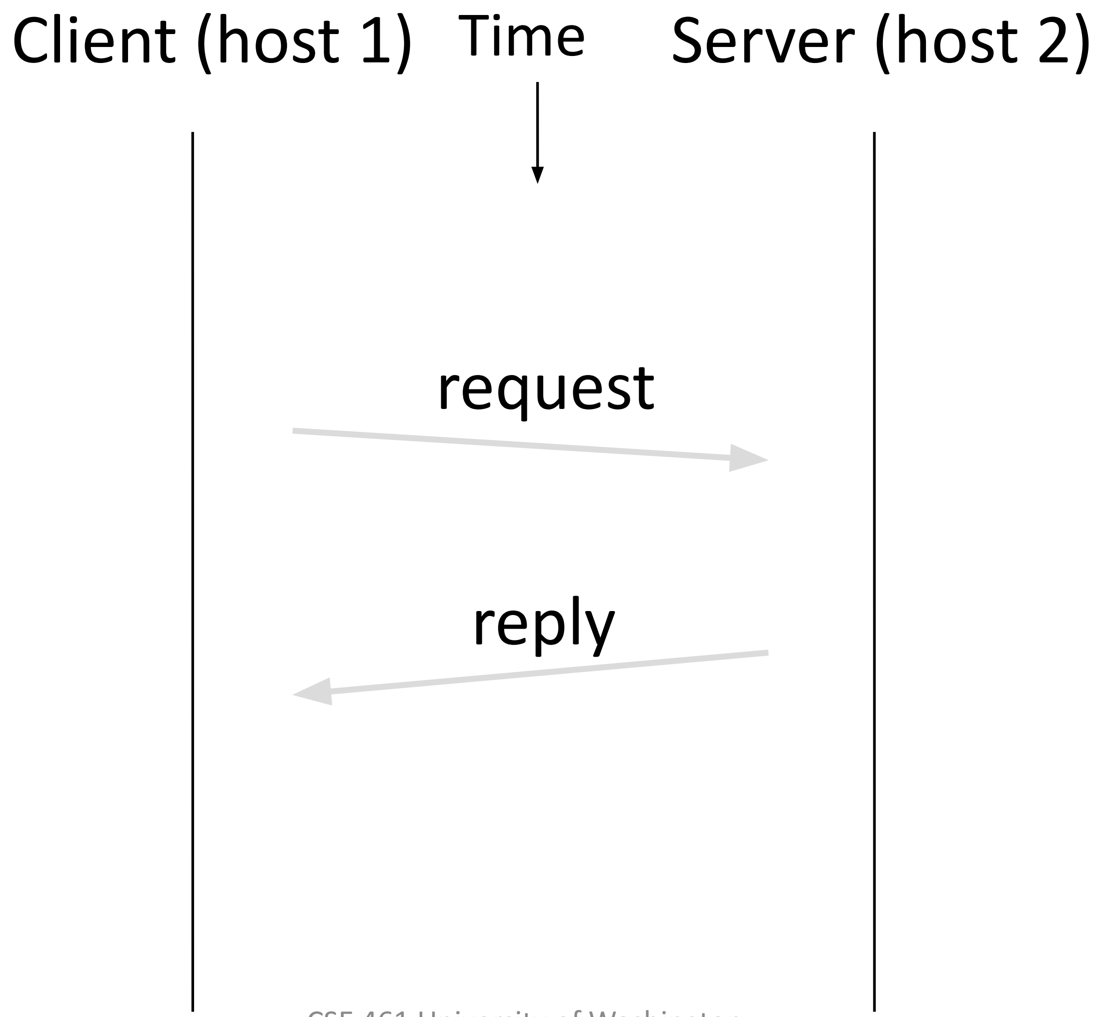
- Used by apps that don’t want reliability or bytestreams
  - Like what?

User Datagram Protocol (UDP)

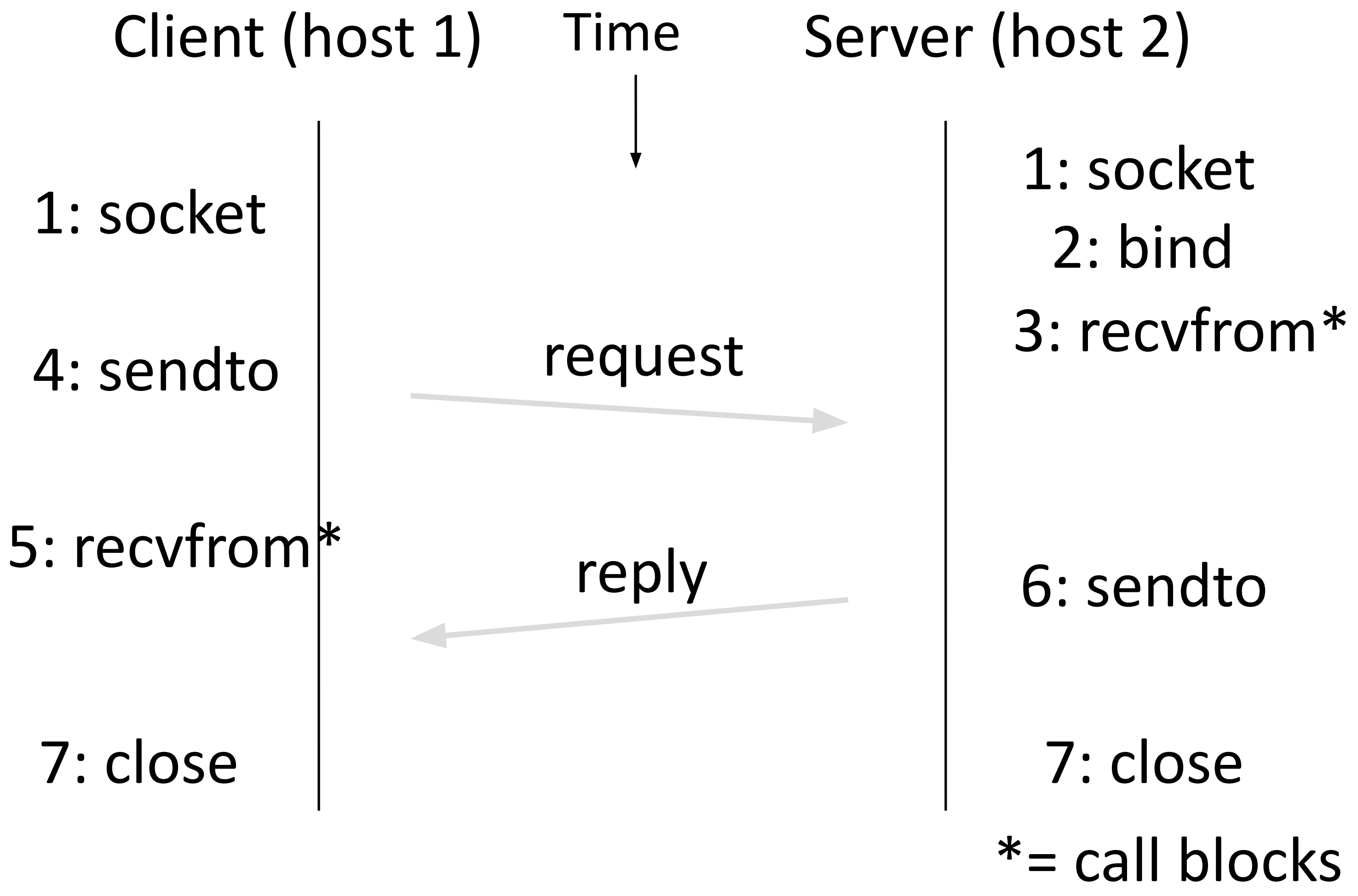
- Used by apps that don’t want reliability or bytestreams
  - Voice-over-IP
  - DNS, RPC
  - DHCP

(If application wants reliability and messages then it has work to do!)

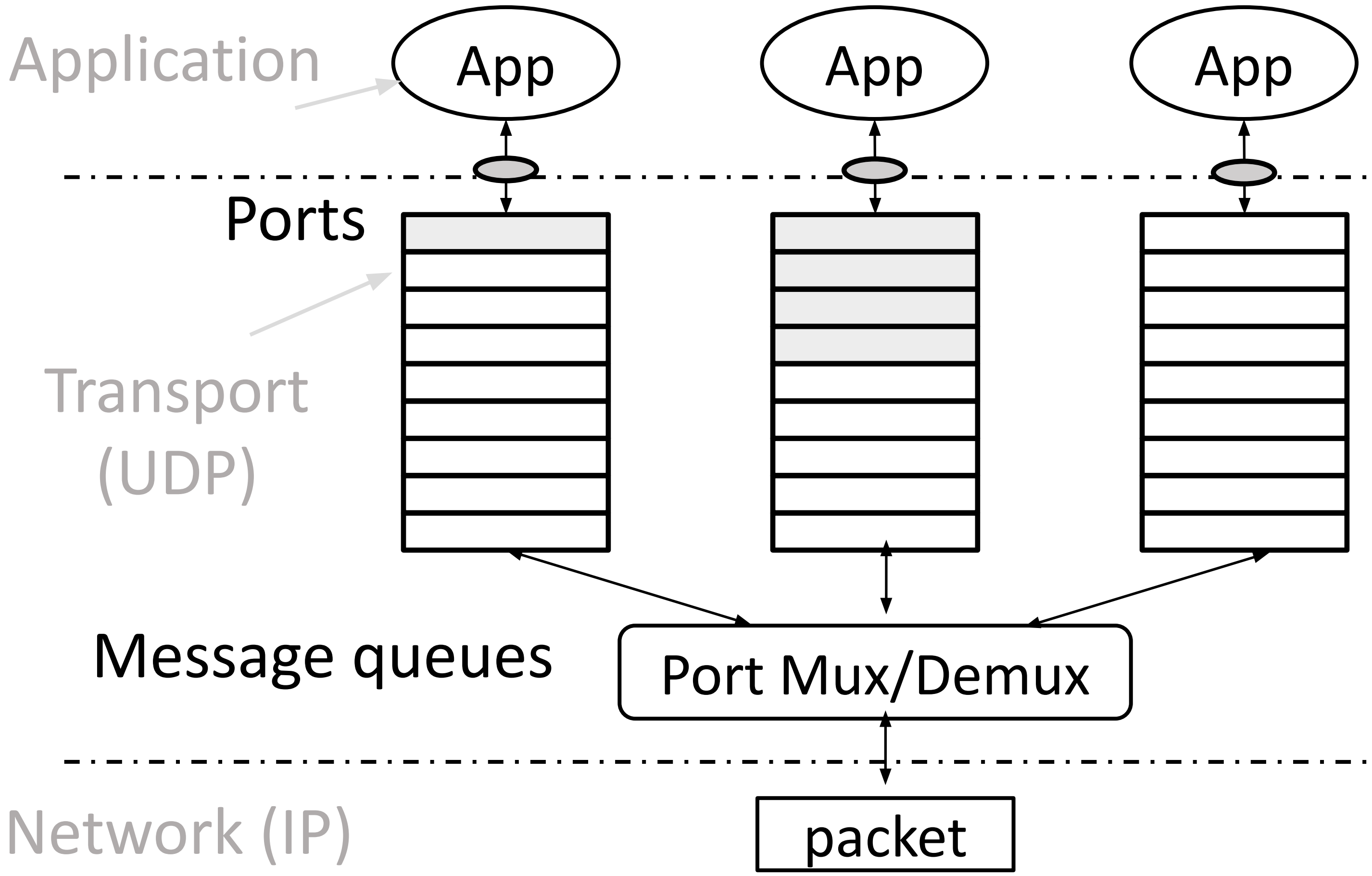
Datagram Sockets



Datagram Sockets (2)



UDP Buffering



UDP Header

- Uses ports to identify sending and receiving application processes
- Datagram length up to 64K
- Checksum (16 bits) for reliability

32 Bits

Source port	Destination port
UDP length	UDP checksum

UDP Header (2)

- Optional checksum covers UDP segment and IP pseudoheader
  - Checks key IP fields (addresses)
  - Value of zero means “no checksum”

32 Bits

Source address		
Destination address		
0 0 0 0 0 0 0 0	Protocol = 17	UDP length

TCP

TCP

- TCP Consists of 3 primary phases:
  1. Connection Establishment (Setup)
  2. Sliding Windows/Flow Control
  3. Connection Release (Teardown)

TCP Signaling

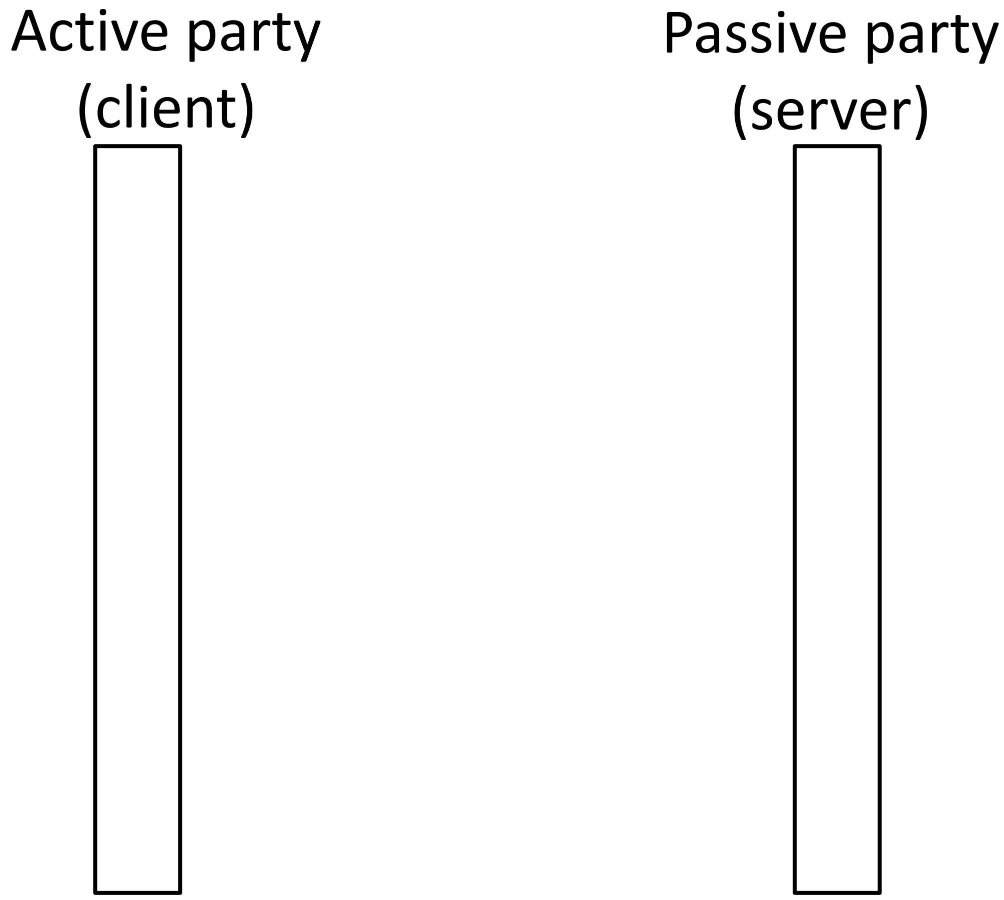
Connection Establishment & Release

Connection Establishment

- Both sender and receiver must be ready before we start the transfer of data
  - Need to agree on a set of parameters
  - e.g., the Maximum Segment Size (MSS)
- This is “signaling”
  - It sets up state at the endpoints
  - Like “dialing” for a telephone call

Three-Way Handshake

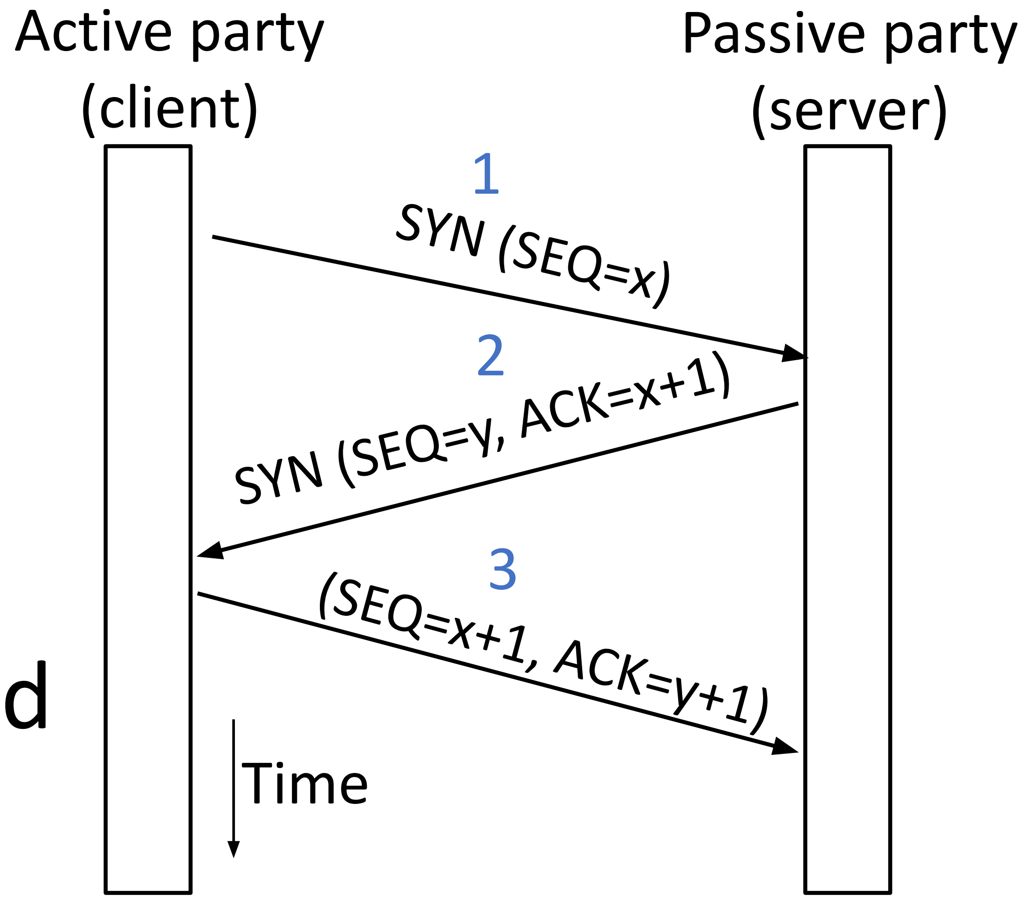
- Used in TCP; opens connection for data in both directions
- Each side probes the other with a fresh Initial Sequence Number (ISN)
  - Sends on a SYNchronize segment
  - Echo on an ACKnowledge segment
- Chosen to be robust even against delayed duplicates



Three-Way Handshake (2)

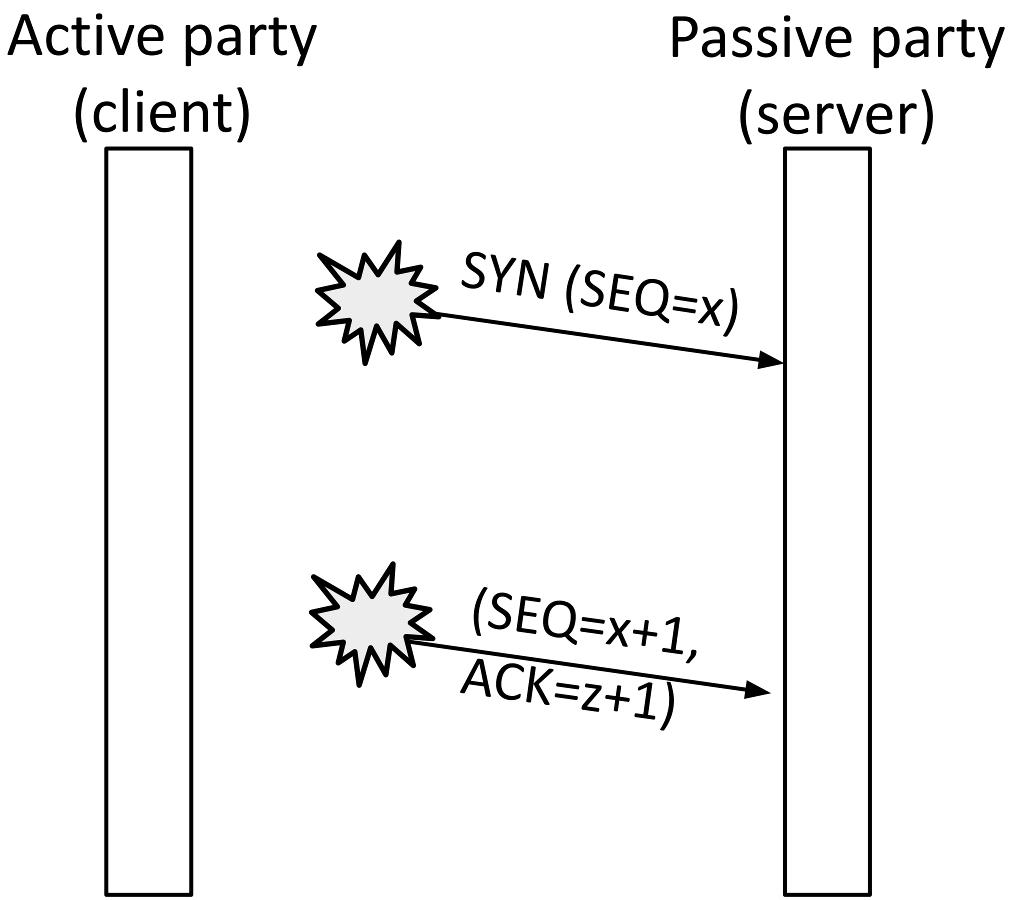
- Three steps:
  - Client sends SYN(x)
  - Server replies with SYN(y)ACK(x+1)
  - Client replies with ACK(y+1)
  - SYNs are retransmitted if lost

- Sequence and ack numbers carried on further segments



Three-Way Handshake (3)

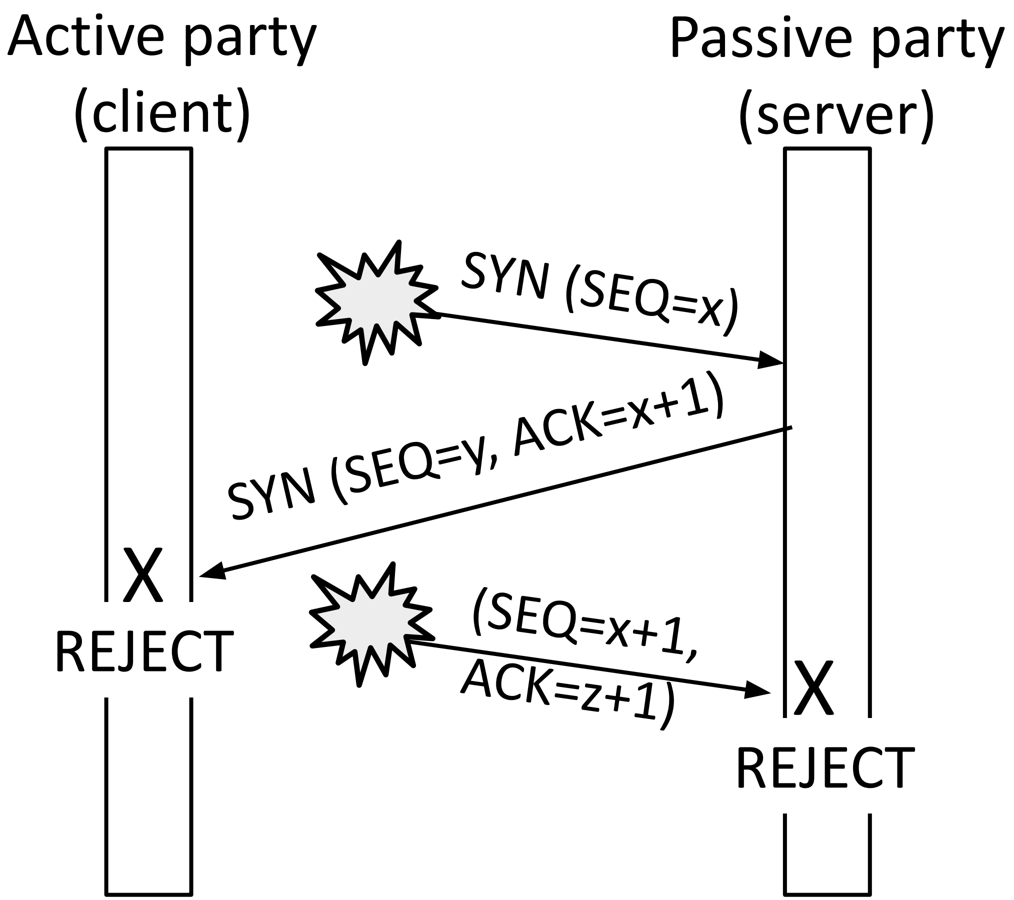
- Suppose delayed, duplicate copies of the SYN and ACK arrive at the server!
  - Improbable, but anyhow …



Three-Way Handshake (4)

- Suppose delayed, duplicate copies of the SYN and ACK arrive at the server!
  - Improbable, but anyhow …

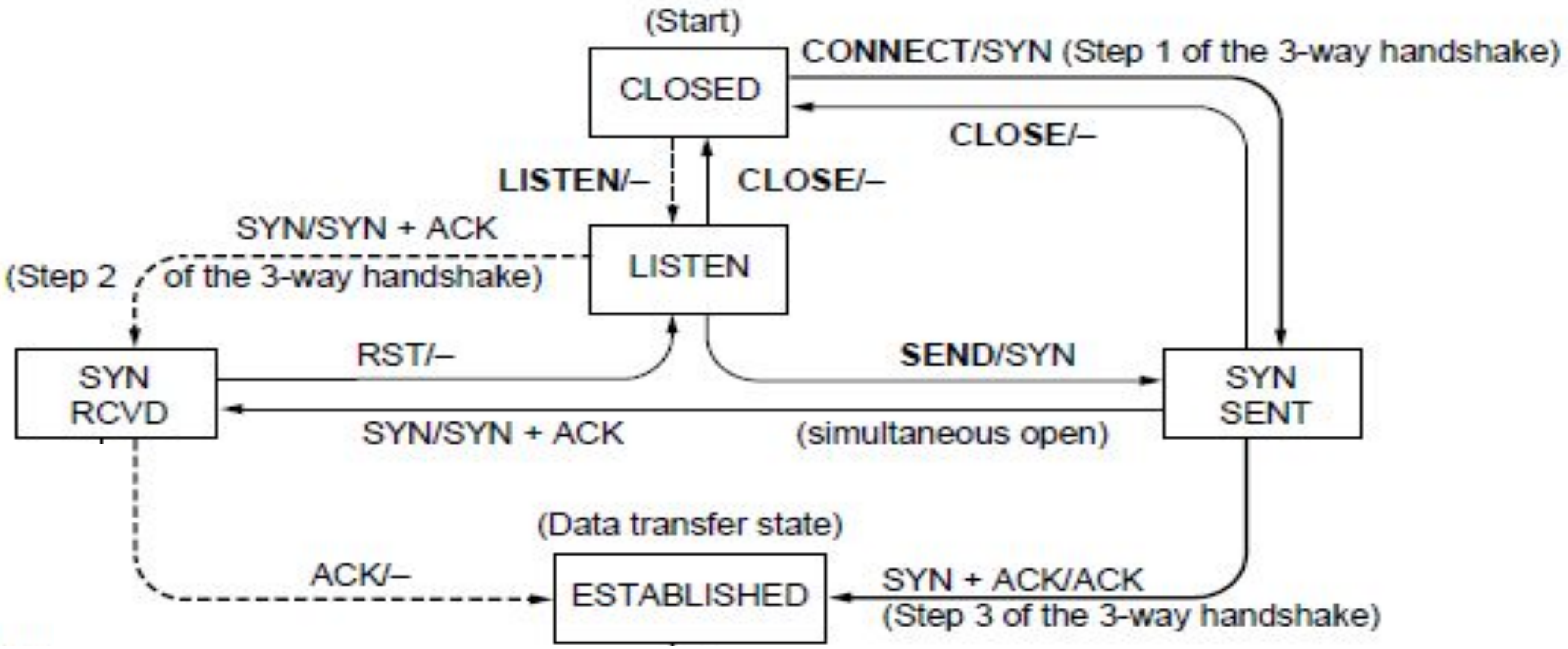
- Connection will be cleanly rejected on both sides



TCP Connection State Machine

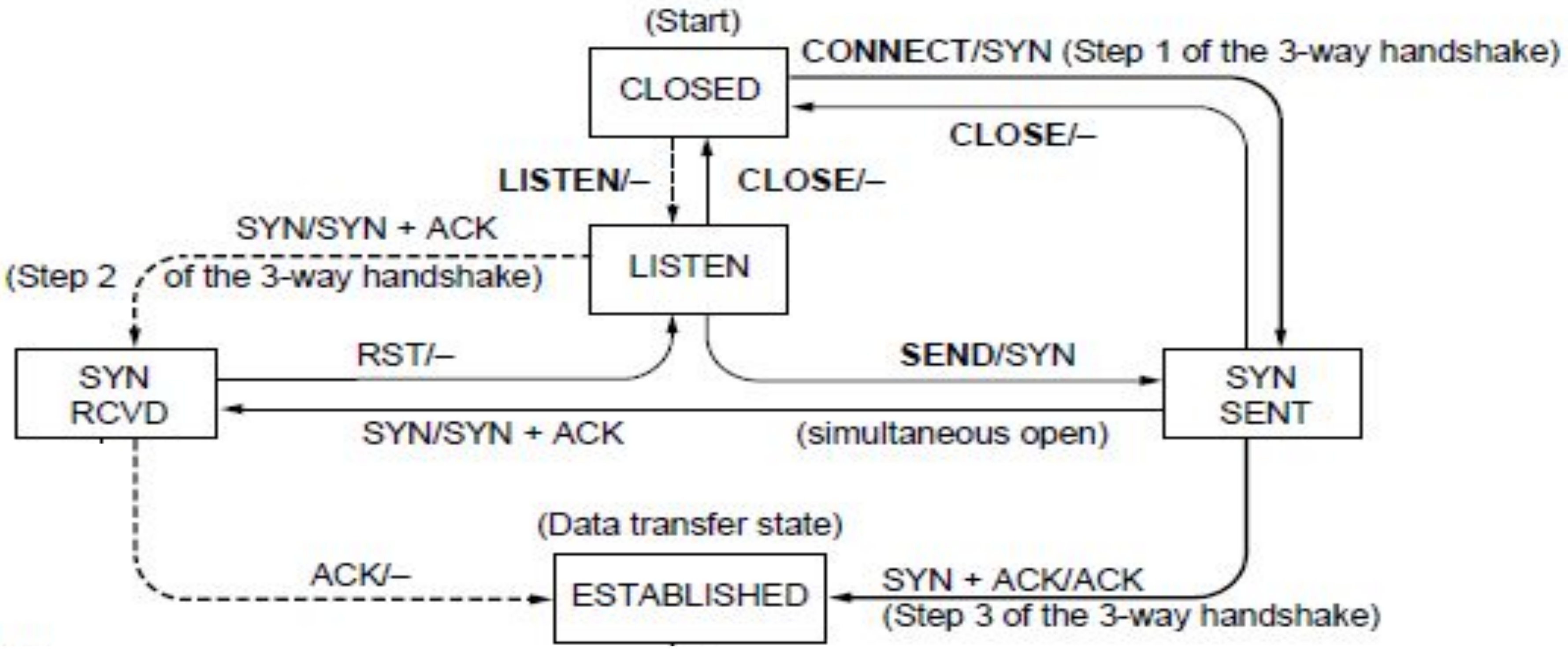
- Captures the states ([]) and transitions (->)
  - A/B means event A triggers the transition, with action B

Both parties run instances of this state machine



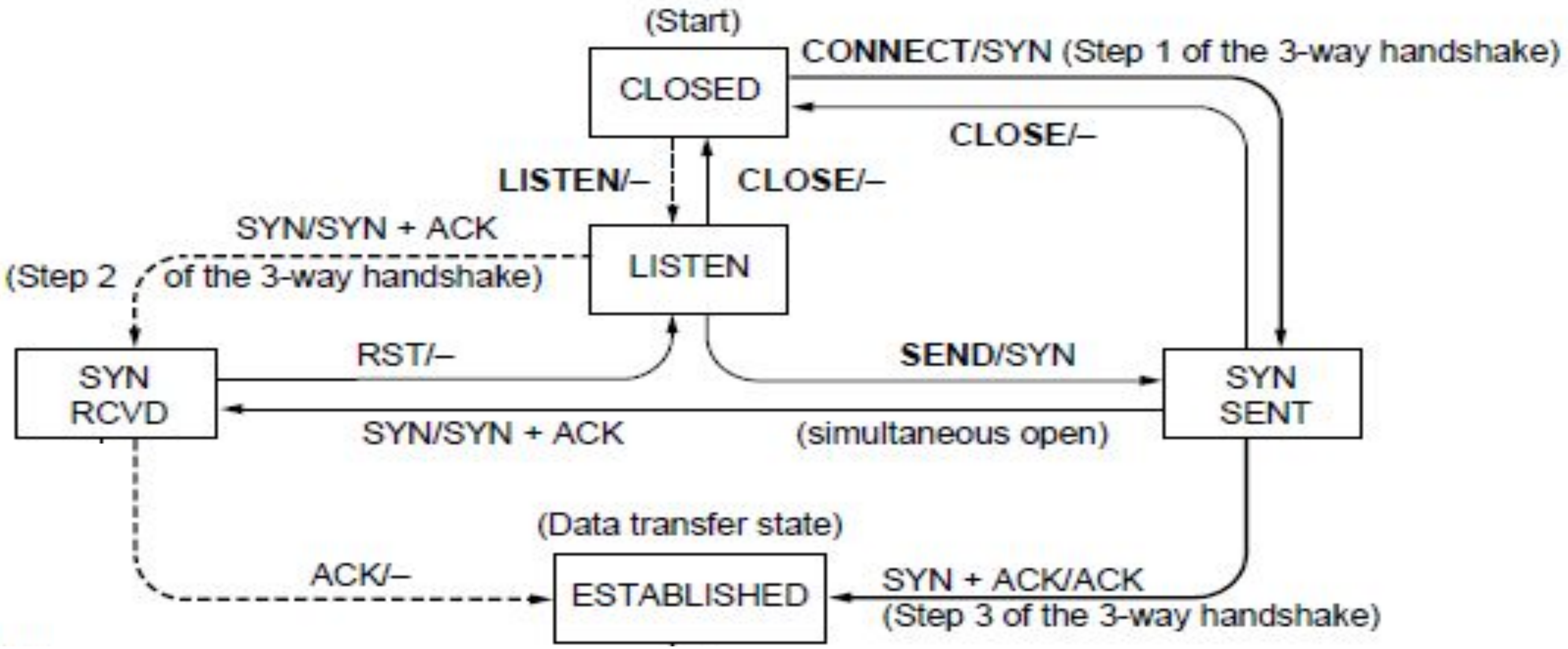
TCP Connections (2)

- Follow the path of the client:



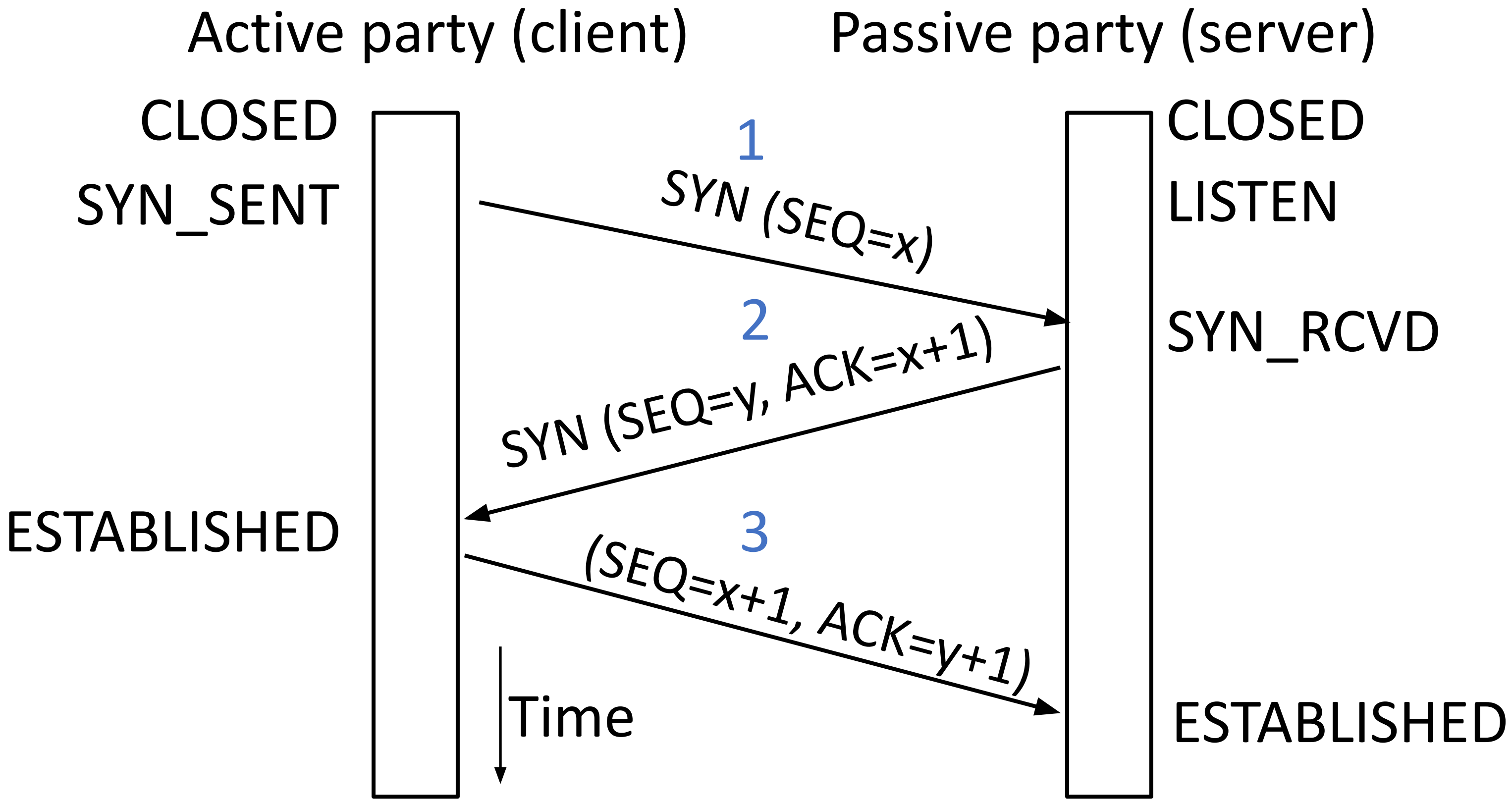
TCP Connections (3)

- And the path of the server:



TCP Connections (4)

- Again, with states …



TCP Connections (5)

- Finite state machines are a useful tool to specify and check the handling of all cases that may occur
  - This feels like classic distributed systems :)

- TCP allows for simultaneous open
  - i.e., both sides open instead of the client-server pattern
  - Try at home to confirm it works

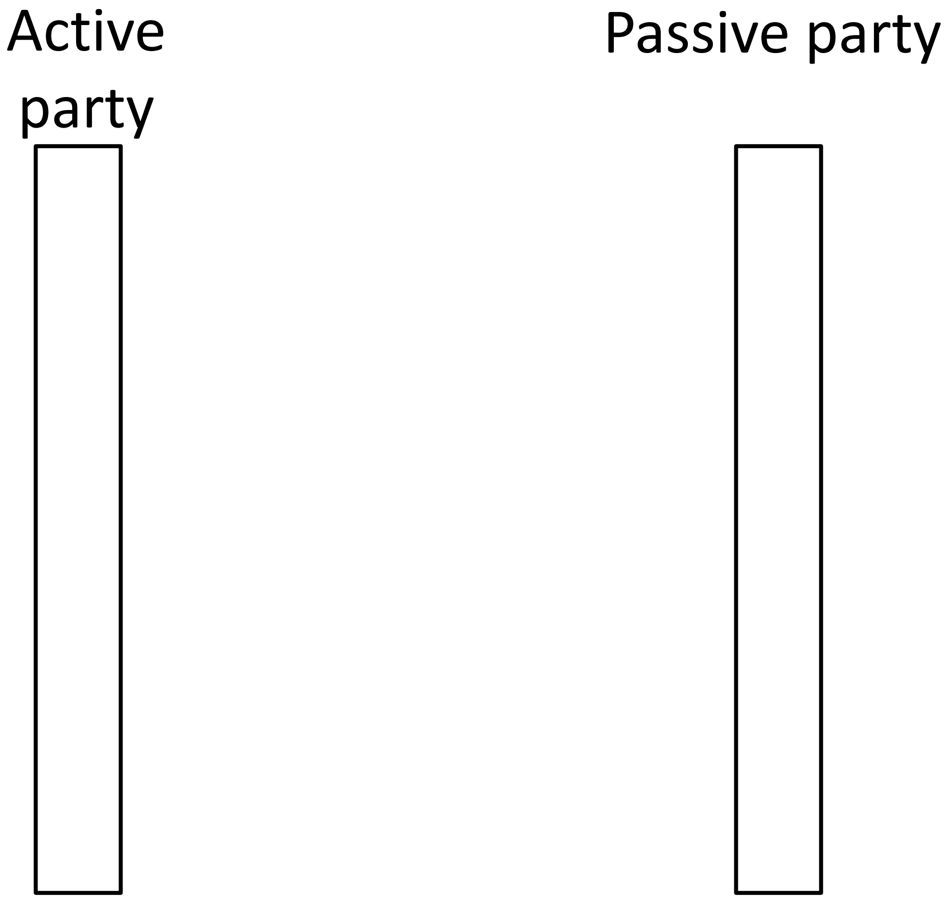
Connection Release

- Orderly release by both parties when done
  - Delivers all pending data and “hangs up”
  - Cleans up state in sender and receiver
- Key problem is to provide reliability while releasing
  - TCP uses a “symmetric” close in which both sides shutdown independently

TCP Connection Release

- Two steps:
  - Active sends FIN(x), passive ACKs
  - Passive sends FIN(y), active ACKs
  - FINs are retransmitted if lost

- Each FIN/ACK closes one direction of data transfer



TCP Connection Release (2)

- Two steps:
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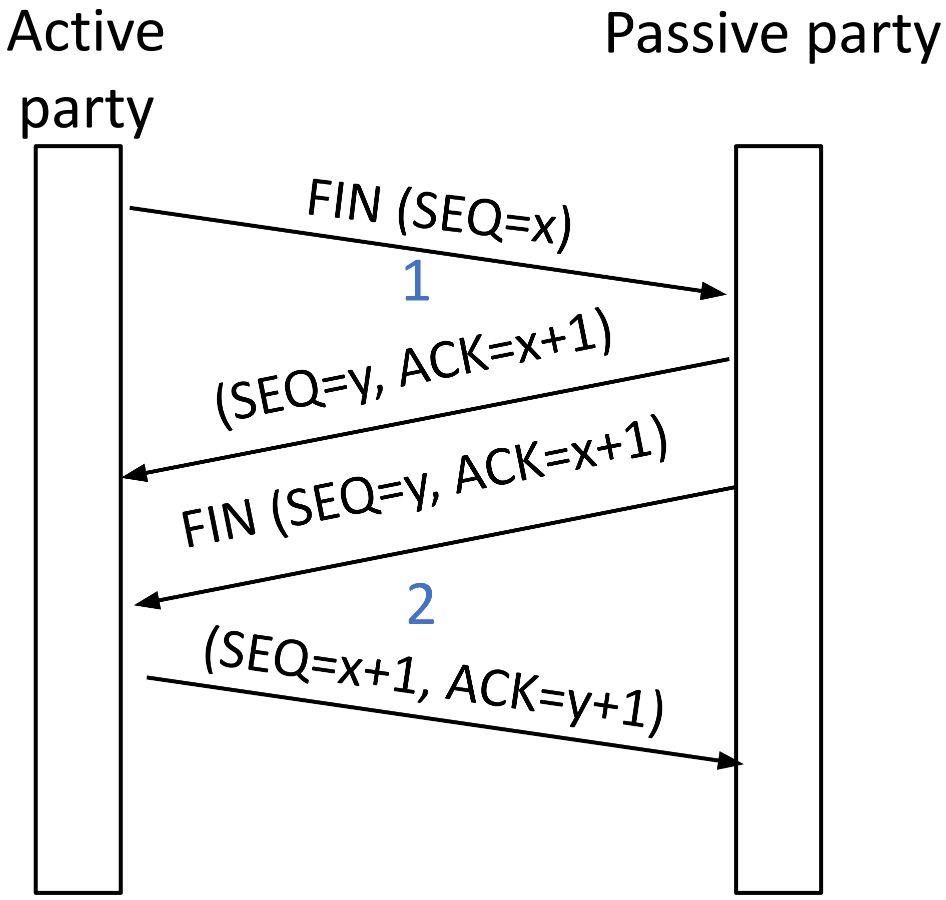

TCP Connection State Machine

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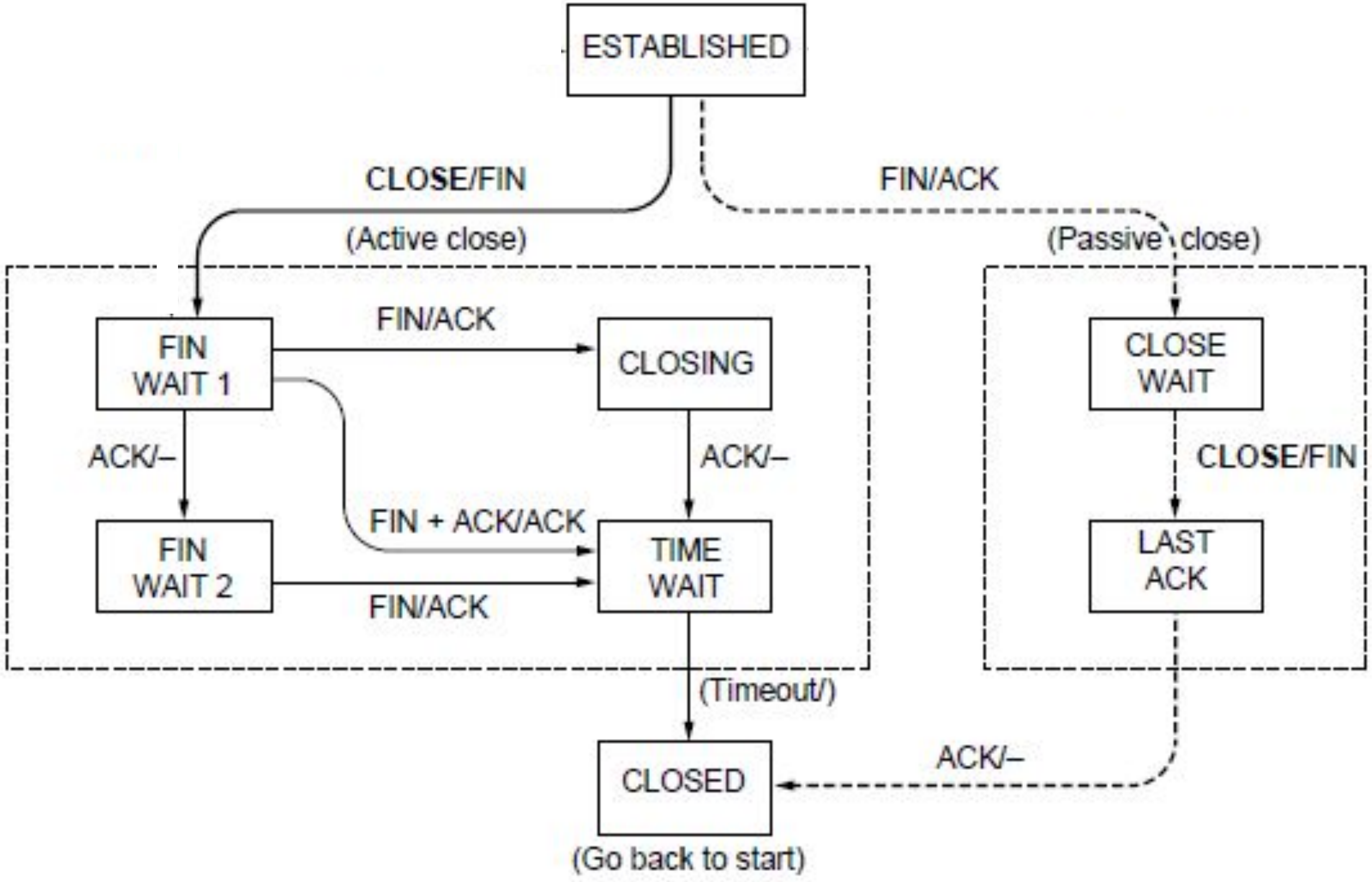
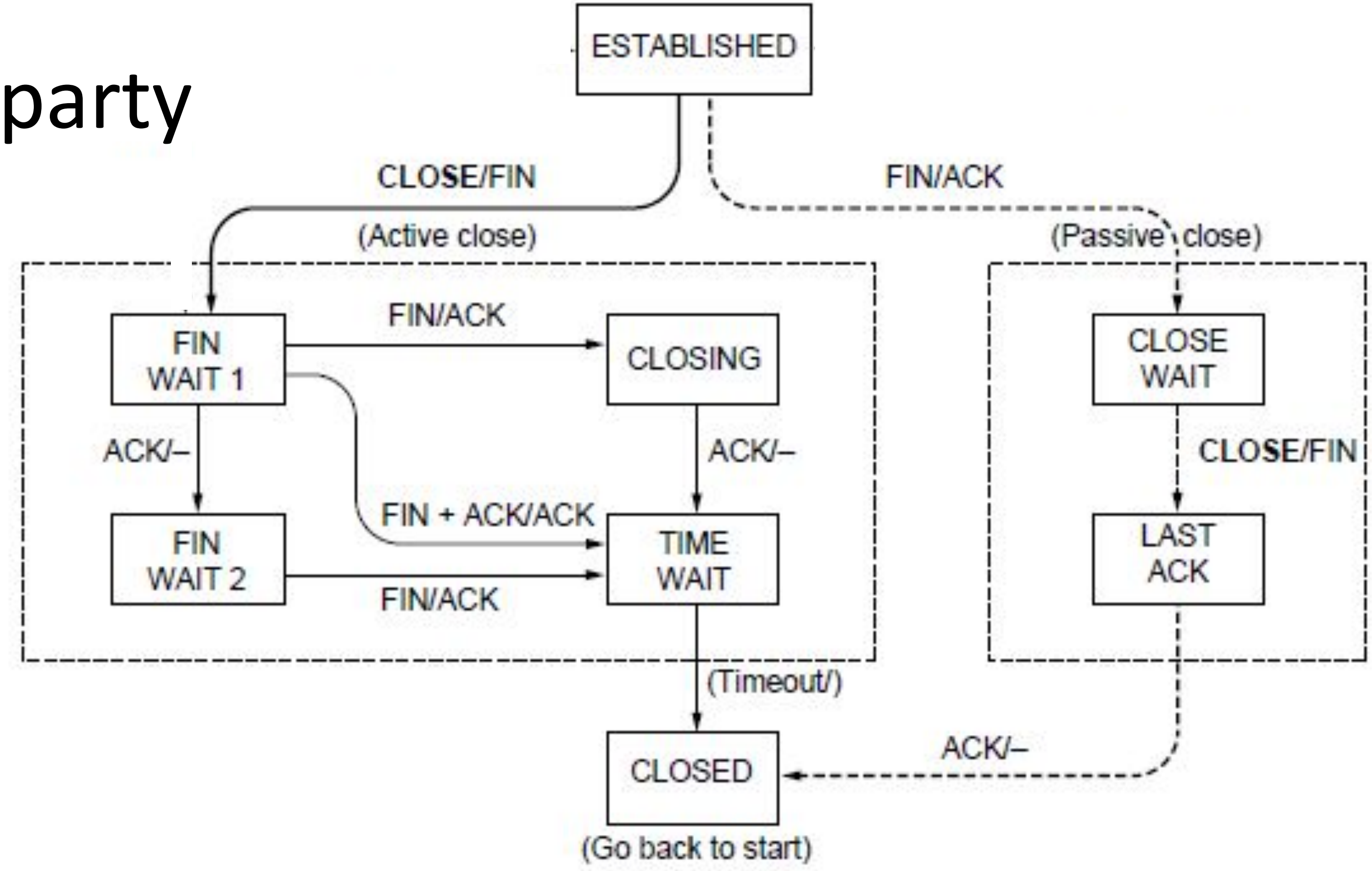


Both parties run instances of this state machine

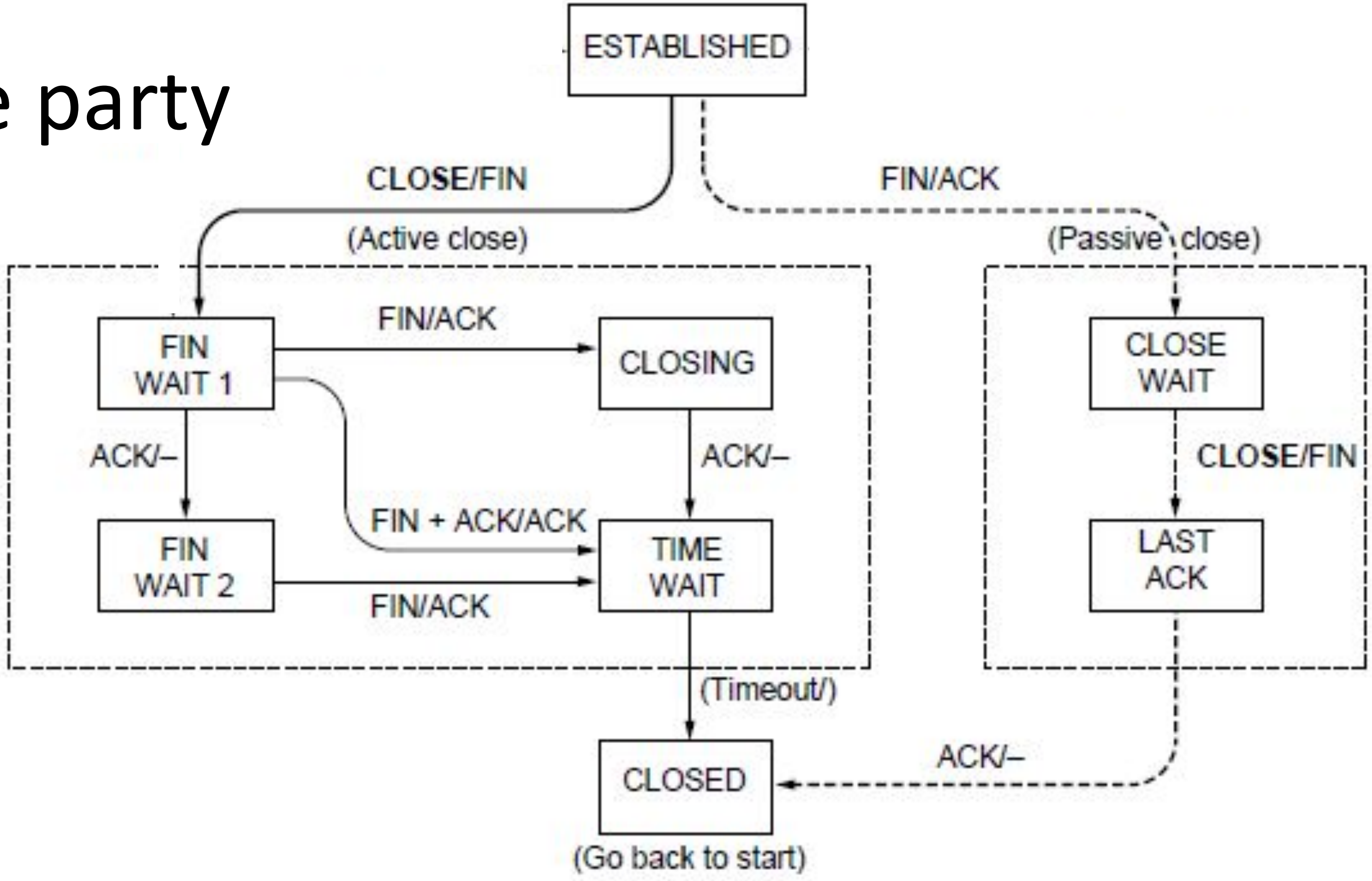
TCP Release

- Follow the active party



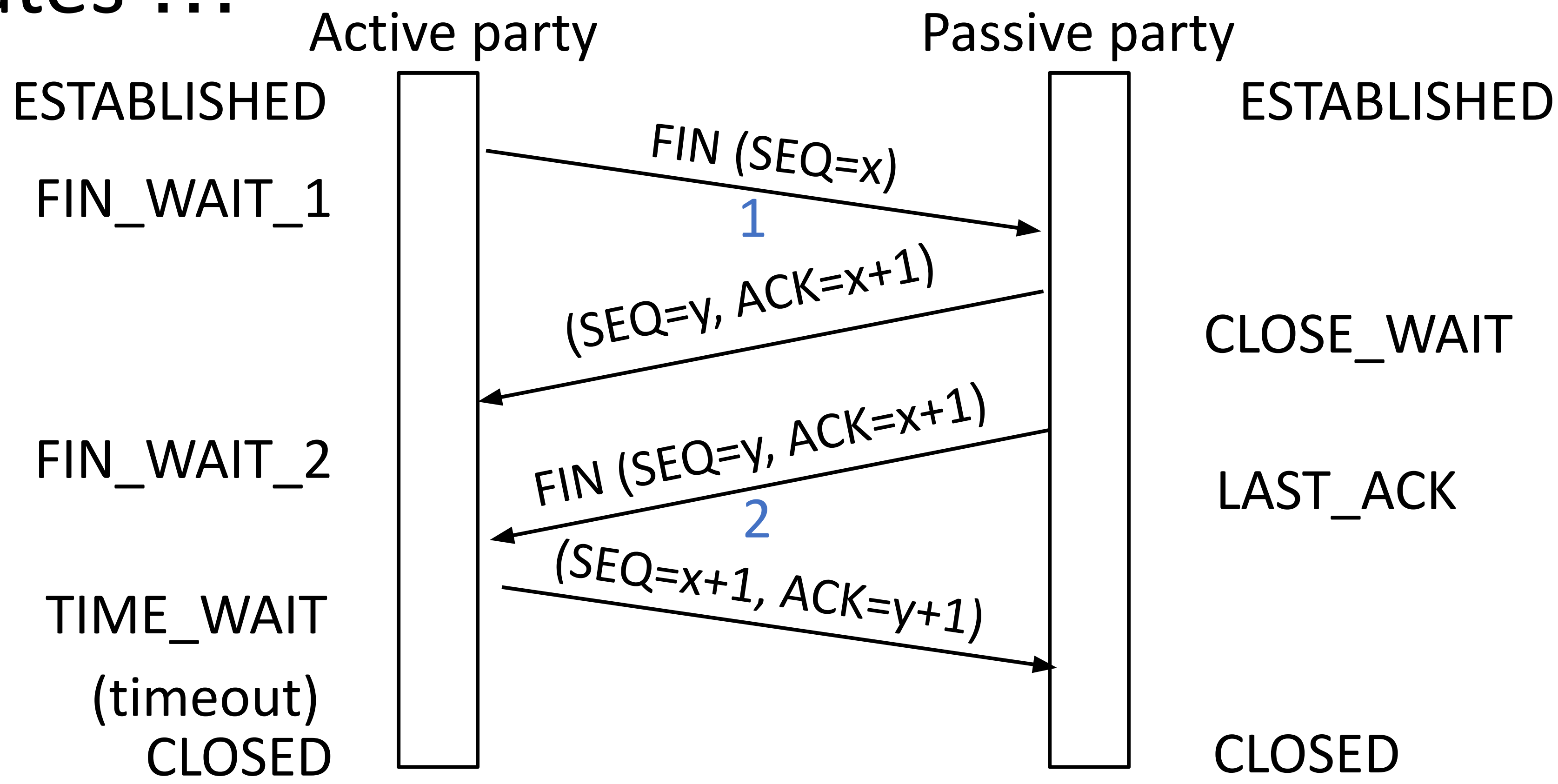
TCP Release (2)

- Follow the passive party



TCP Release (3)

- Again, with states …



TIME_WAIT State

- Wait a long time after sending all segments and before completing the close
  - Two times the maximum segment lifetime of 60 seconds
- Why?
  - ACK might have been lost, in which case FIN will be resent for an orderly close
  - Could otherwise interfere with a subsequent connection

Chaotic optimization in the real world…

- The TCP close procedure we just described is “clean”, but…
  - Also requires more messages
  - Means the connection state needs to be maintained by the server during the close…
    - What if your server is serving 10k, 100k, 1m clients???
    - What if the data transferred is small and you know at the application layer that the client received it all…
- TCP *also has an unorganized close*… the “reset”
  - Intended for use if the connection becomes corrupted
  - Can be sent by either party, *no guarantee all data transmitted or received*
  - Handled as an error/exception, but often sent by real-world endpoints instead of a full close!

A legit TCP joke

A: Hello

B: Howdy, and I've heard everything up to "Hello".

A: I have heard everything up to "Howdy"

A: Would you like to hear a TCP joke?

B: Yes, and I have heard everything up to "joke?"

A: Excellent. I have heard everything up to "Yes".

B: I have heard everything up to "Excellent."

A: Buffer bloat and congestion control.

A. I have finished.

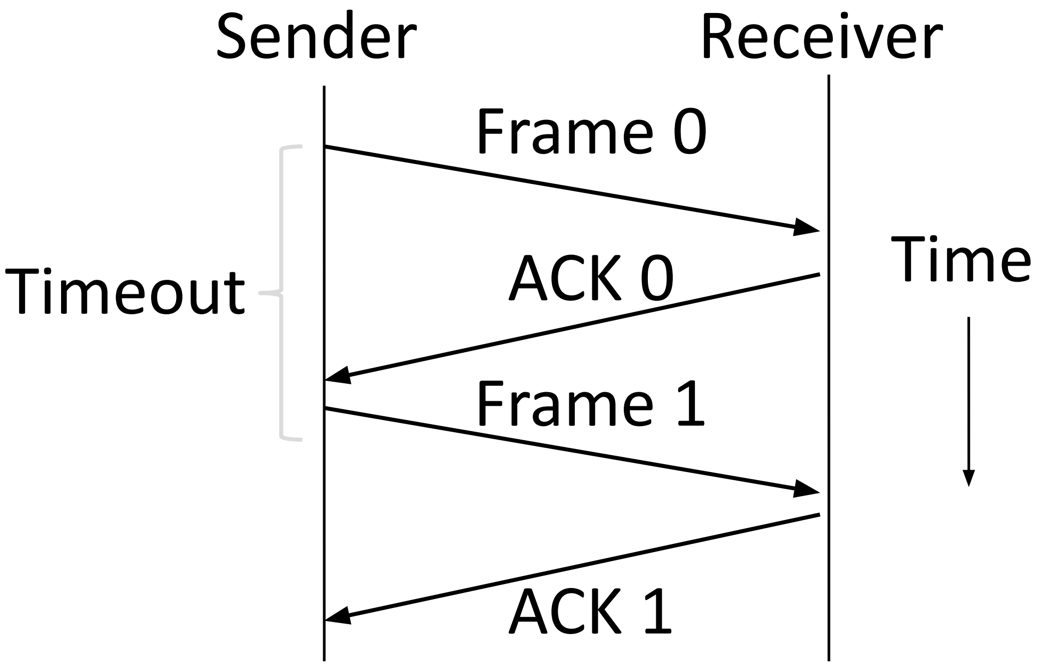
B. Haha, and I am finished. I have heard everything up to that you are finished.

A: I have heard everything up to that you are finished.

Flow Control

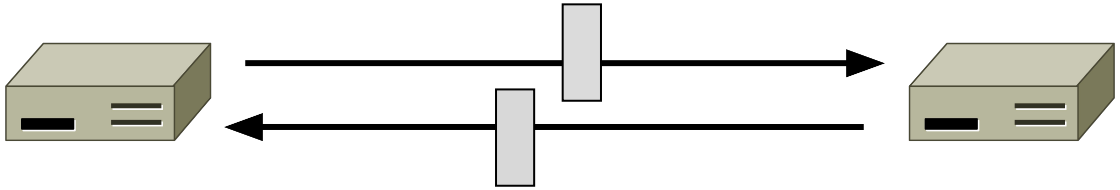
Recall

- ARQ with one message at a time is Stop-and-Wait (normal case below)



Limitation of Stop-and-Wait

- It allows only a single message to be outstanding from the sender:
  - Fine for LAN (only one frame fits in network anyhow)
  - Not efficient for network paths with BD >> 1 packet



Limitation of Stop-and-Wait Example

For this example, ignore transmission delay & time to process the message

- Example: R=1 Mbps, D = 50 ms, 10kb packets
  - Simple RTT (Round Trip Time) ~ 2D = 100 ms
  - How many packets/sec?

~1 packet / RTT -> 10 packets / second -> 10kb/packet * 10 packets/second -> 100kbps
10% link efficiency!

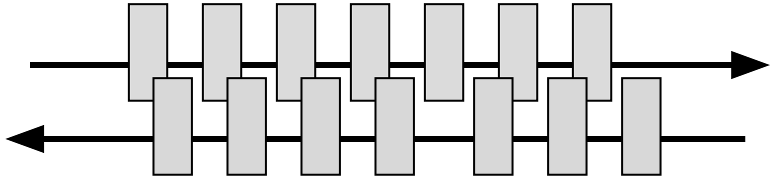
- What if R=10 Mbps?

1 packet / RTT -> 10 packets / second -> 10kb/packet * 10 packets/second -> 100kbps
1% link efficiency!

In practice even worse… since real world implementation cannot ignore the transmission + processing delay!

Sliding Window

- Generalization of stop-and-wait
  - Allows W packets to be outstanding
  - Can send W packets per RTT (=2D)



- Pipelining improves performance
- Need W=2BD to fill network path

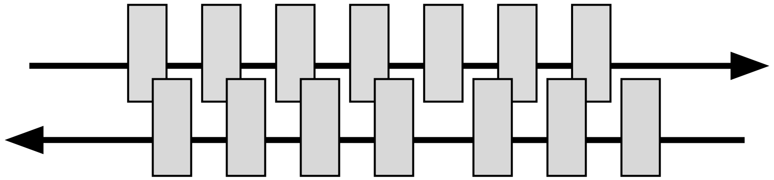
Sliding Window (2)

- What W will use the network capacity?
  - Remember: W = 2BD
  - Assume 10kb packets
- Ex: R=1 Mbps, D = 50 ms

- Ex: What if R=10 Mbps?

Sliding Window (3)

- Ex: R=1 Mbps, D = 50 ms
  - 2BD = 10^6 b/sec x 100 10^{-3} sec = 100 kbit
  - W = 2BD = 10 packets of 1250 bytes



- Ex: What if R=10 Mbps?
  - 2BD = 1000 kbit
  - W = 2BD = 100 packets of 1250 bytes

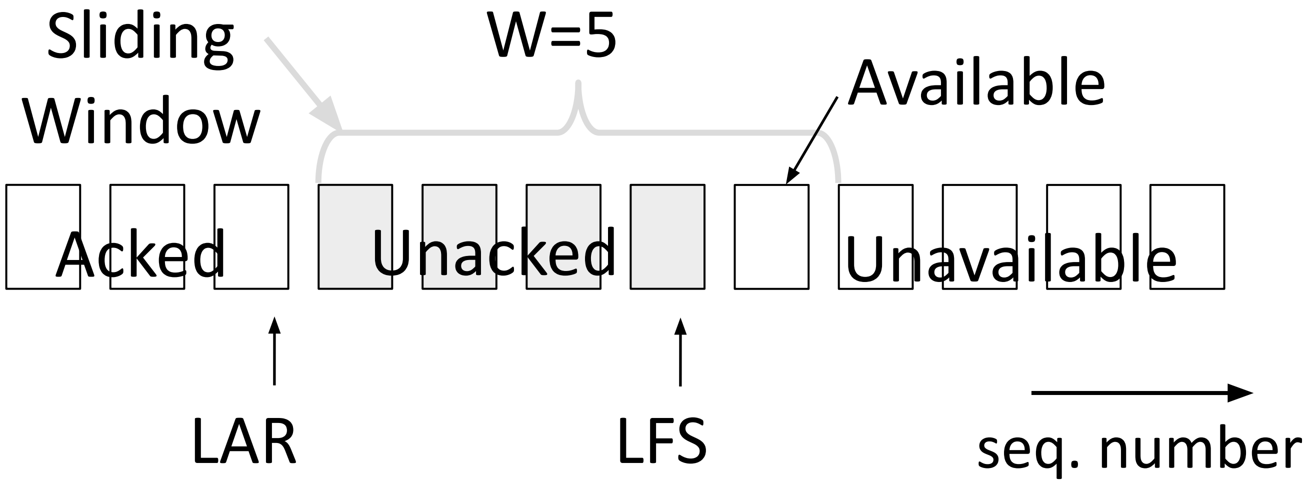
Sliding Window Protocol

- Many variations, depending on how buffers, acknowledgements, and retransmissions are handled

- <u>Go-Back-N</u>
  - Simplest version, can be inefficient
- <u>Selective Repeat</u>
  - More complex, better performance

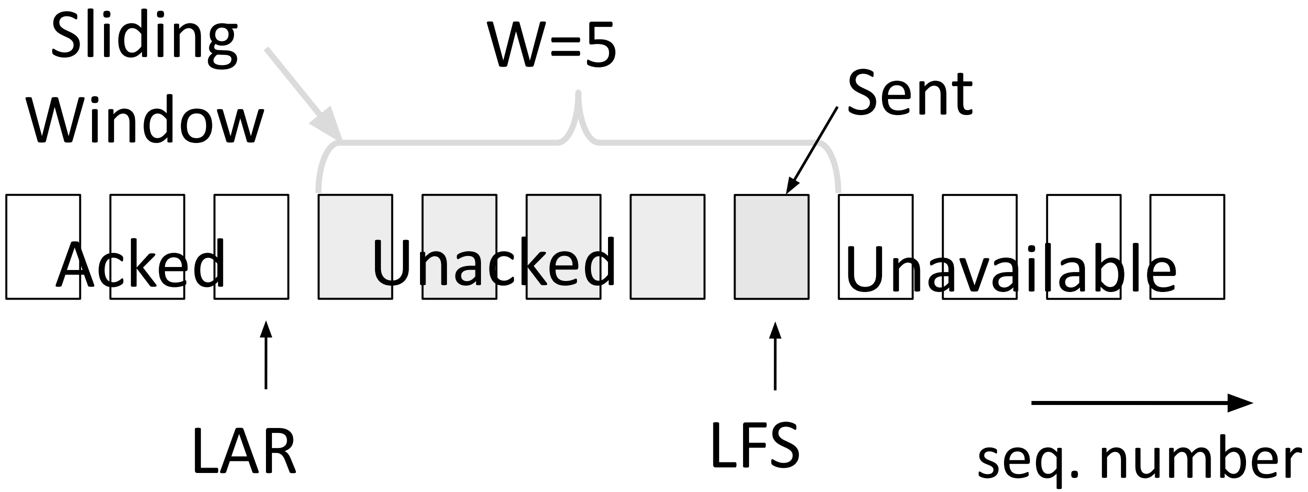
Sliding Window – Sender

- Sender buffers up to W segments until they are acknowledged
  - LFS=LAST FRAME SENT, LAR=LAST ACK REC'D
  - Sends while LFS – LAR < W



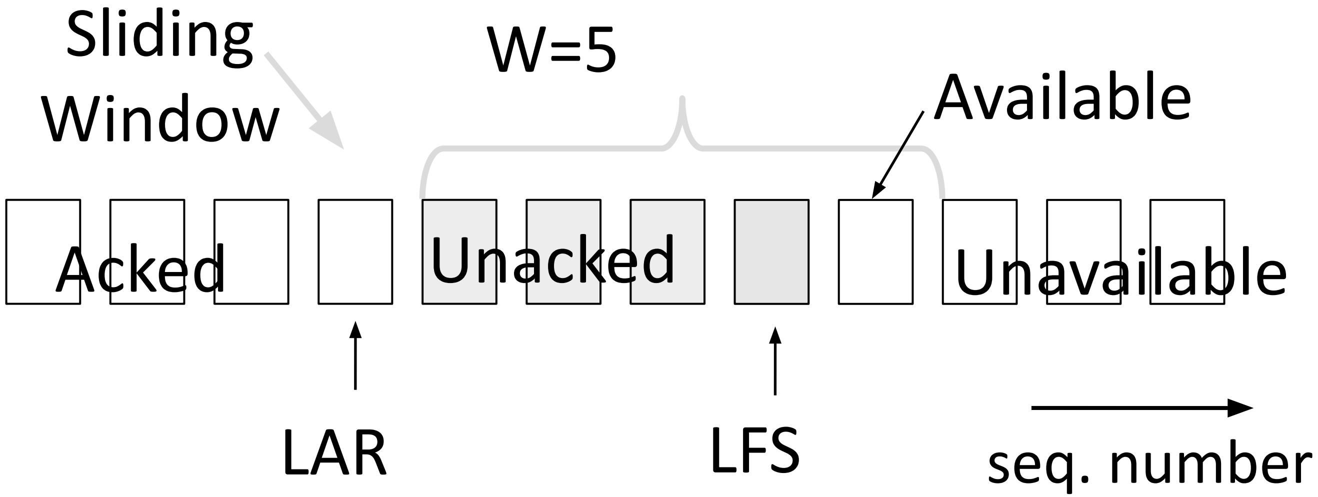
Sliding Window – Sender (2)

- Transport accepts another segment of data from the Application ...
  - Transport sends it (as LFS–LAR < 5)



Sliding Window – Sender (3)

- Next higher ACK arrives from peer…
  - Window advances, buffer is freed
  - LFS–LAR < 5 (can send one more)



Sliding Window – Go-Back-N

- Receiver keeps only a single packet buffer for the next segment
  - State variable, LAS = LAST ACK SENT
- On receive:
  - If seq. number is LAS+1, accept and pass it to app, update LAS, send ACK
  - Otherwise discard (as out of order)

Sliding Window – Selective Repeat

- Receiver passes data to app in order, and buffers out-of-order segments to reduce retransmissions

- ACK conveys highest in-order segment
  - Plus hints about out-of-order segments in modern implementations
- TCP uses a selective repeat design; we'll see the details later

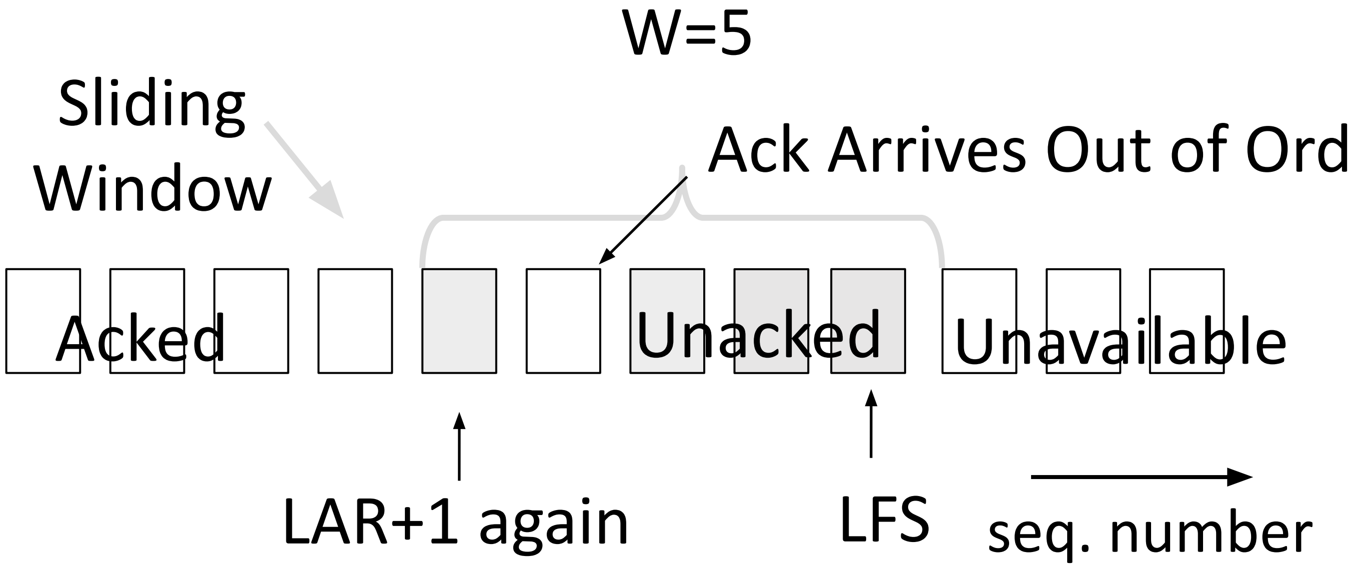
Sliding Window – Selective Repeat (2)

- Buffers W segments, keeps state variable LAS = LAST ACK SENT

- On receive:
  - Buffer segments that arrive in [LAS+1, LAS+W]
  - Send app in-order segments from LAS+1, and update LAS
  - Send ACK for LAS regardless

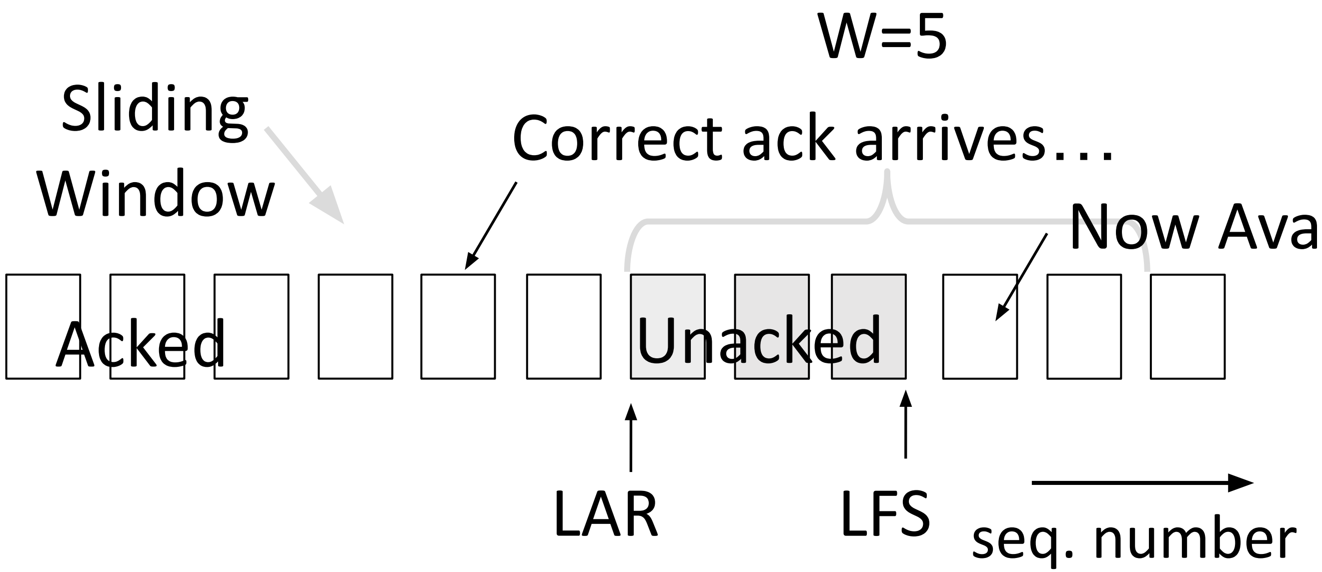
Sliding Window – Selective Retransmission (3)

- Keep normal sliding window
- If receive something out of order
  - Send last unacked packet again!



Sliding Window – Selective Retransmission (4)

- Keep normal sliding window
- If correct ACK arrives, move window and LAR, send more messages



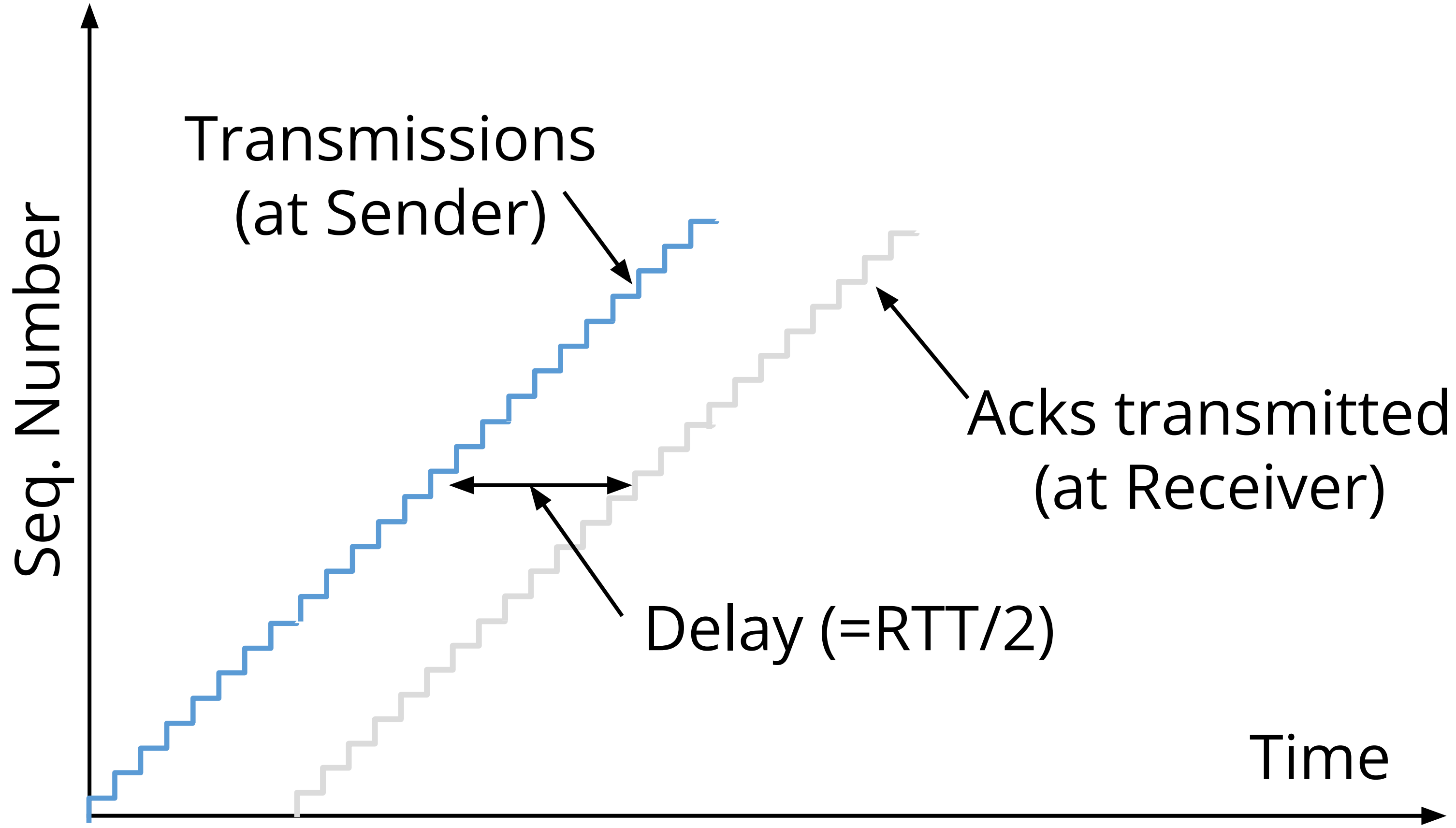
Sliding Window – Retransmissions

- Go-Back-N uses a single timer to detect losses
    - On timeout, resends buffered packets starting at LAR+1
- Selective Repeat uses a timer per unacked segment to detect losses
    - On timeout for segment, resend it
    - Hope to resend fewer segments

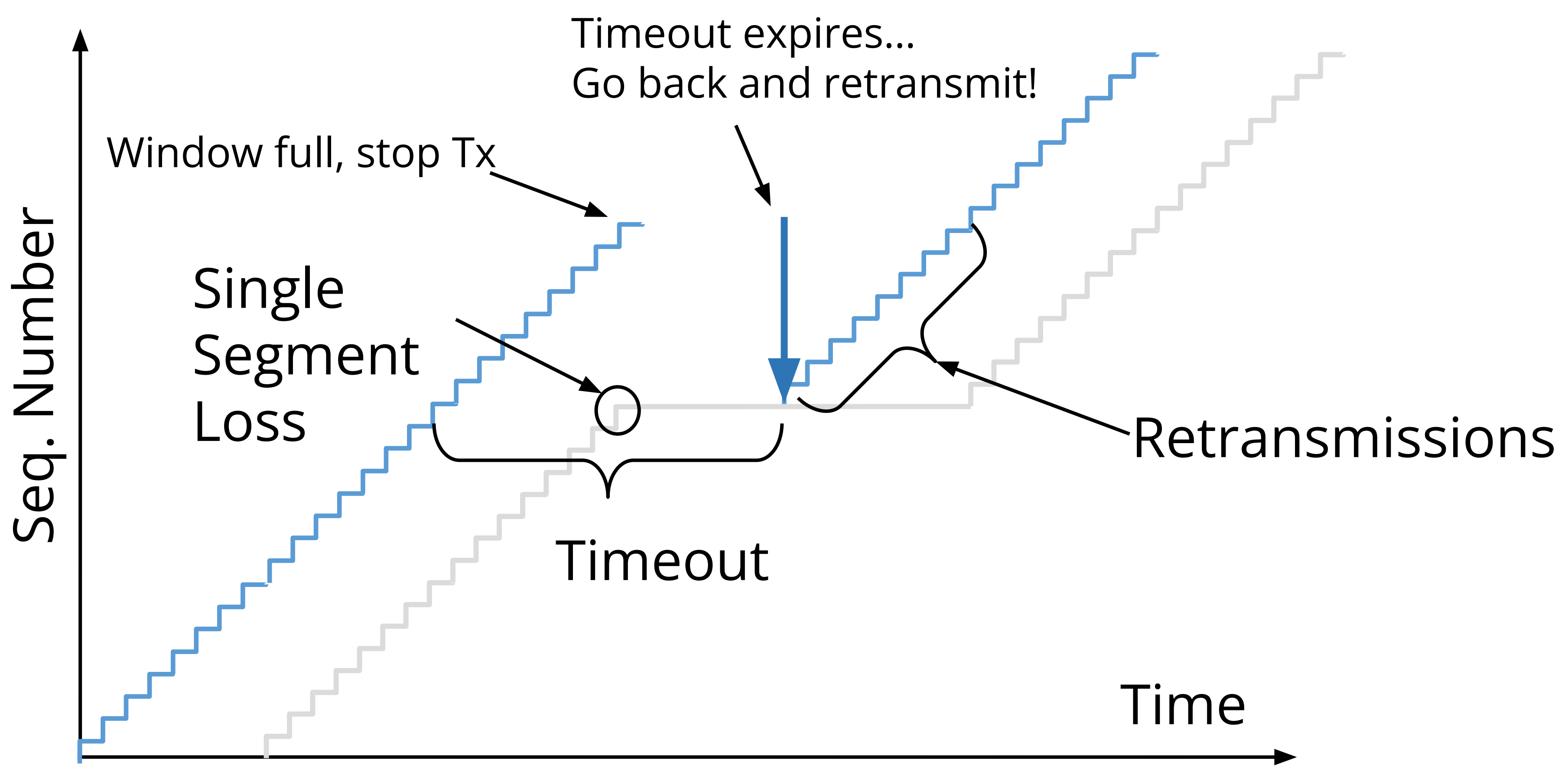
Sequence Numbers

- Need more than 0/1 for Stop-and-Wait …
  - But how many?
- For Selective Repeat, need W numbers for packets, plus W for acks of earlier packets
  - 2W seq. numbers
  - Fewer for Go-Back-N (W+1)
- Typically implement seq. number with an N-bit counter that wraps around at $2^N—1$
  - E.g., N=8: …, 253, 254, 255, 0, 1, 2, 3, …

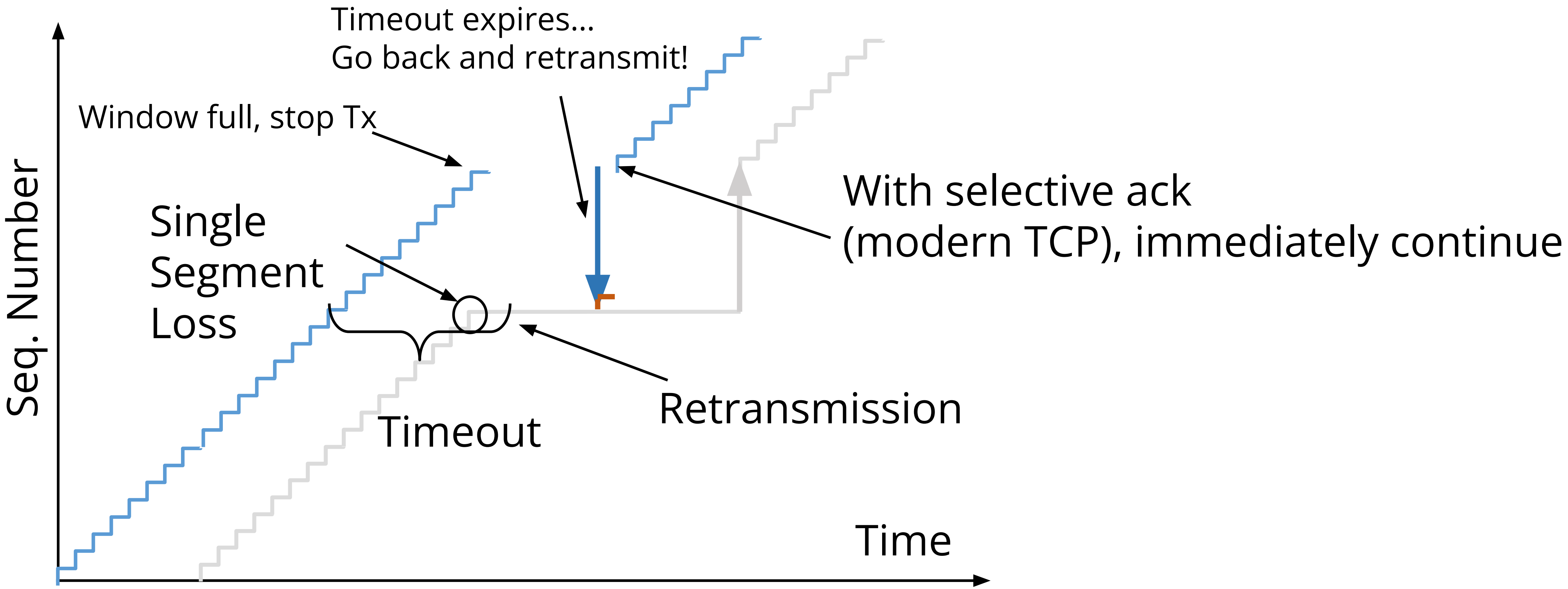
Sequence vs. Time Plot



Sequence vs. Time Plot: Go-Back-N



Sequence vs. Time Plot: Selective Retransmit

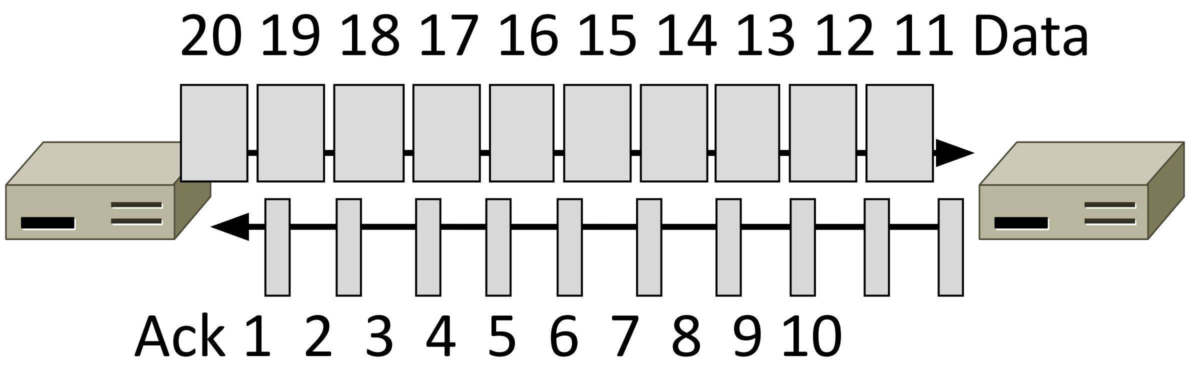


ACK Clocking

So now we can control the flow! But how do we make it smooth?

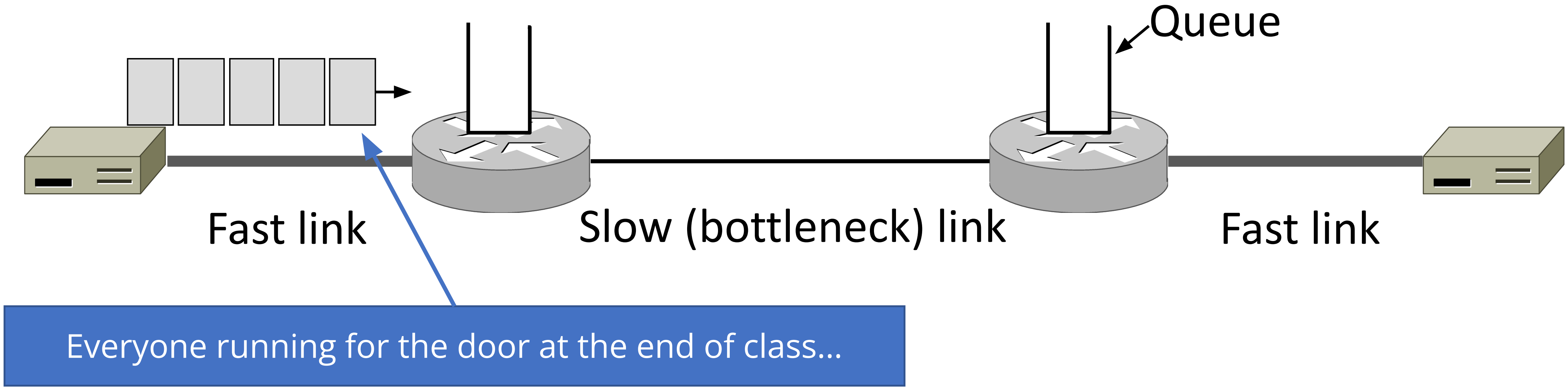
Sliding Window ACK Clock

- Each in-order ACK advances the sliding window and lets a new segment enter the network
  - ACKS "clock" data segments



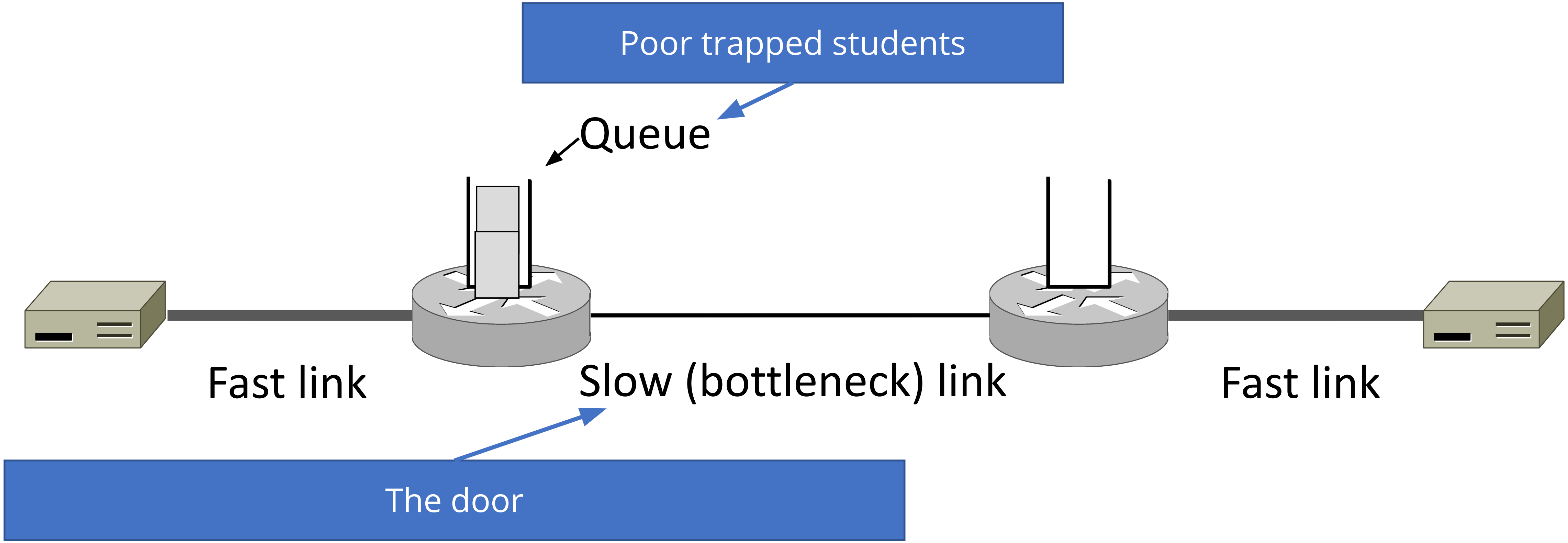
Benefit of ACK Clocking

- Consider what happens when sender injects a burst of segments into the network



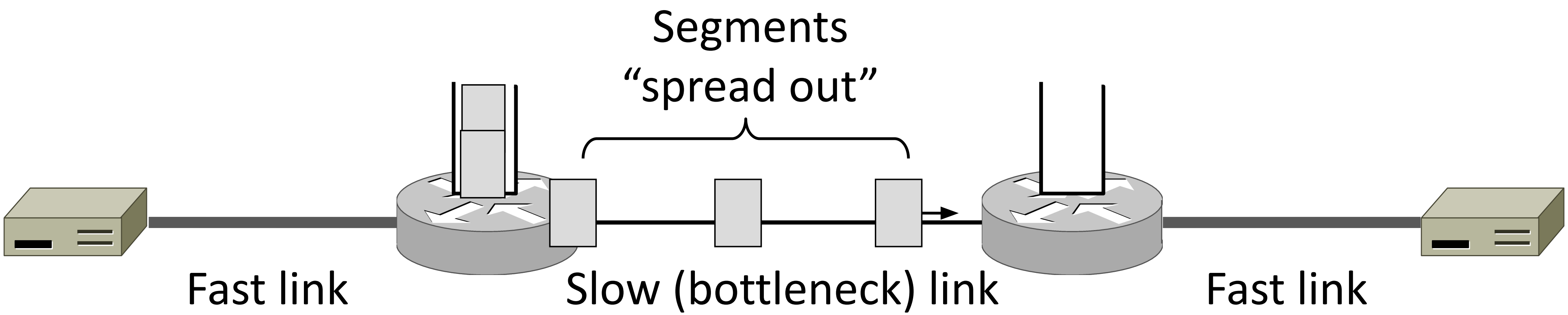
Benefit of ACK Clocking

- Intermediate routers overloaded with traffic (bad)



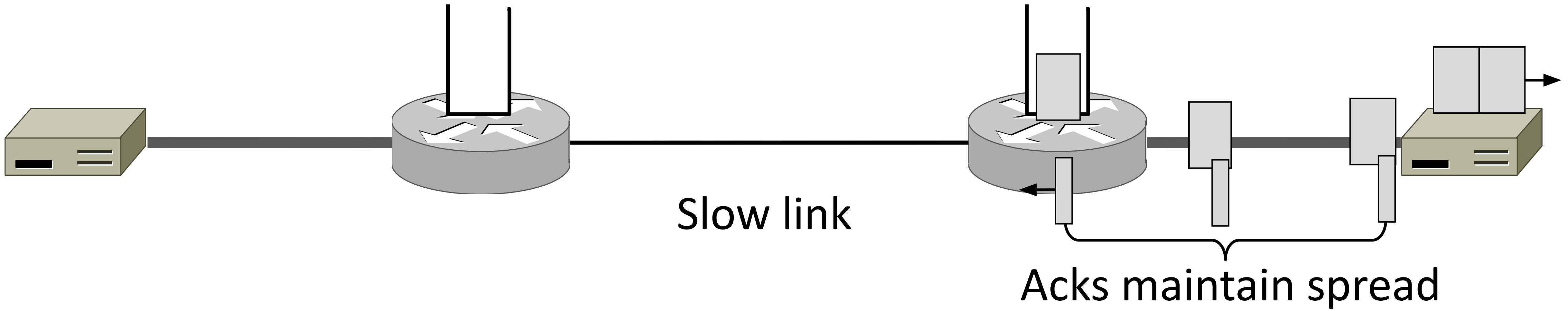
Benefit of ACK Clocking (2)

- Fortunately, segments are buffered and spread out on slow link



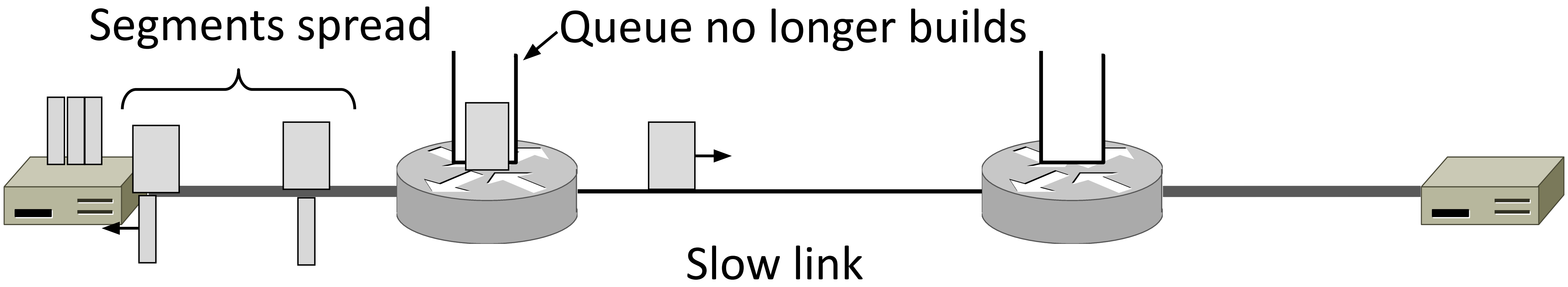
Benefit of ACK Clocking (3)

- ACKS maintain the spread back to the original sender



Benefit of ACK Clocking (4)

- Sender clocks new segments with the spread
  - Now sending at the bottleneck link without queuing!

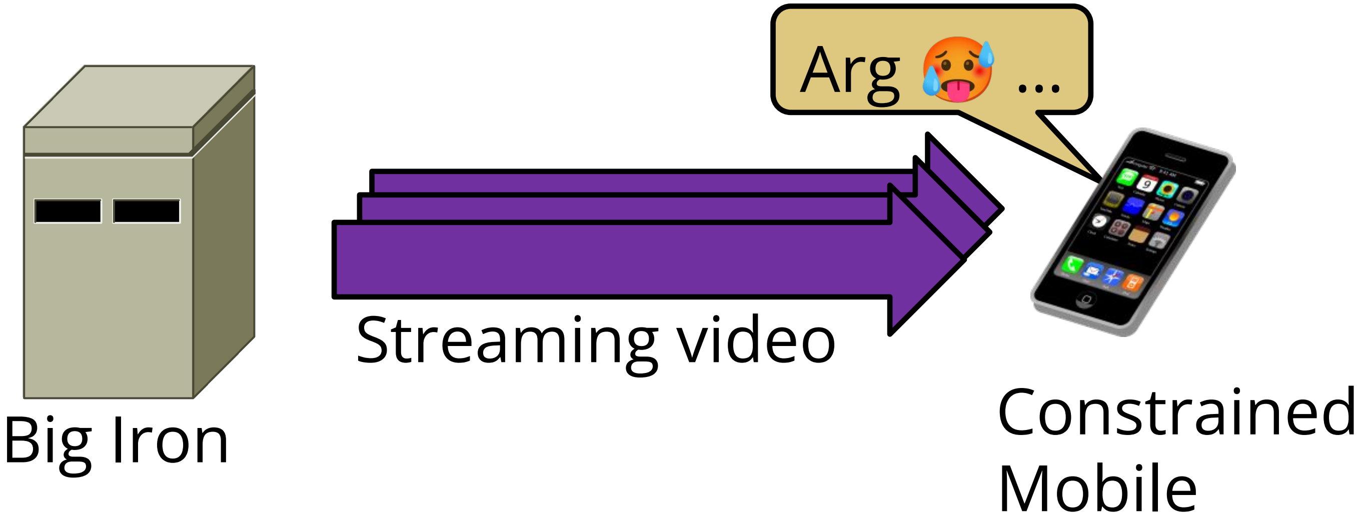


Benefit of ACK Clocking Summary

- Helps run with low levels of loss and delay!
- The network smooths out the burst of data segments
- ACK clock transfers this smooth timing back to sender
- Subsequent data segments are not sent in bursts so do not queue up in the network

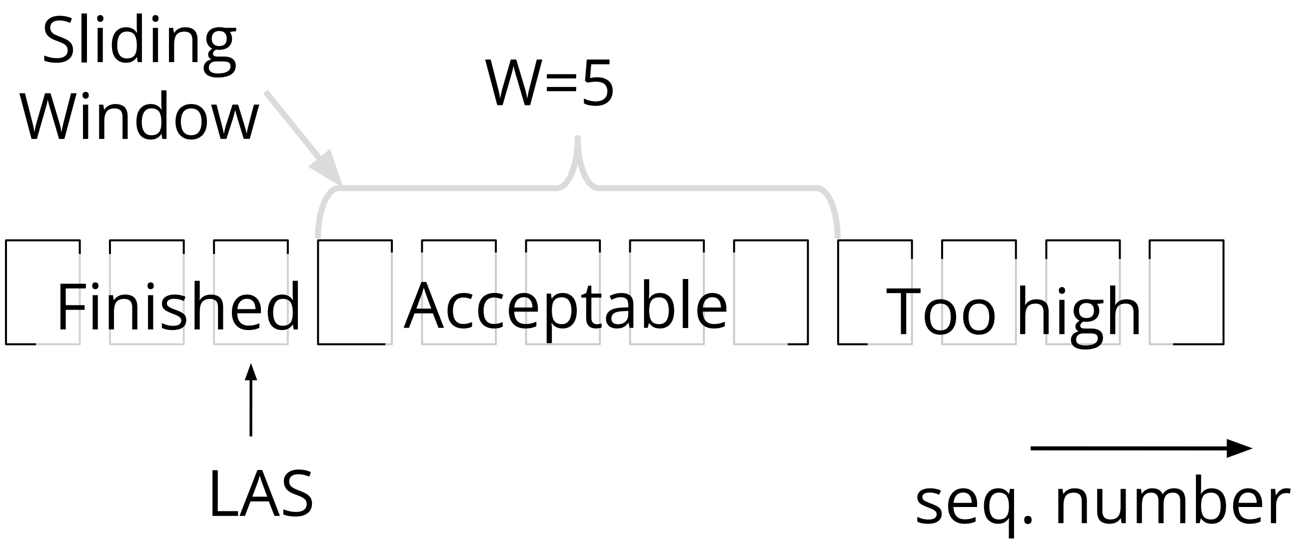
A Related Problem…

- Sliding window has sliding window to keep network busy
  - What if the receiver is overloaded?



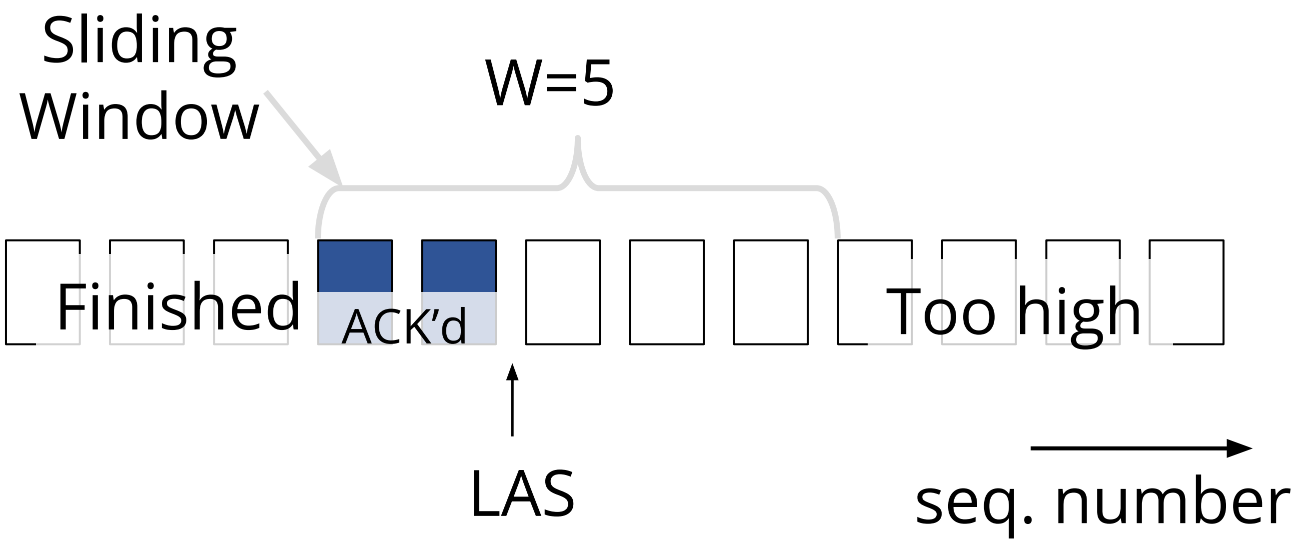
Sliding Window – Receiver Constraints

- Consider receiver with `W` buffers
- **`LAS`**`=last ack sent`
  - App pulls in-order data from buffer with recv() call



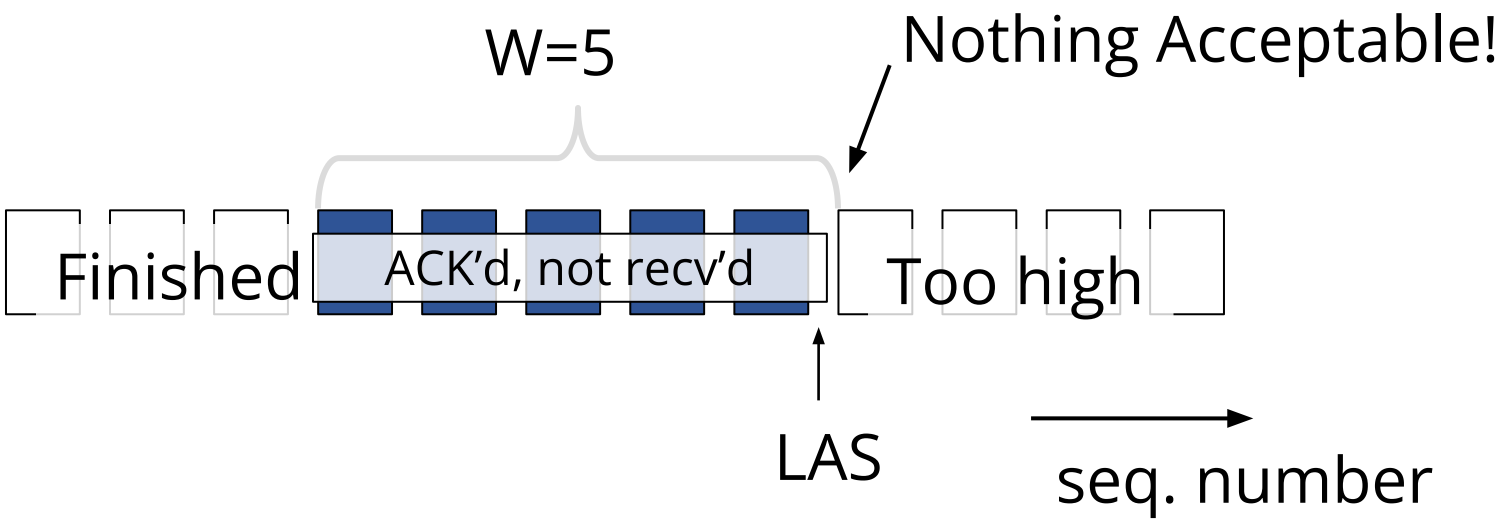
Sliding Window – Receiver Constraints (2)

- Suppose the next two segments arrive…
  - but app does not call recv()
  - **LAS** rises, *but we can't slide window*! No buffer space!



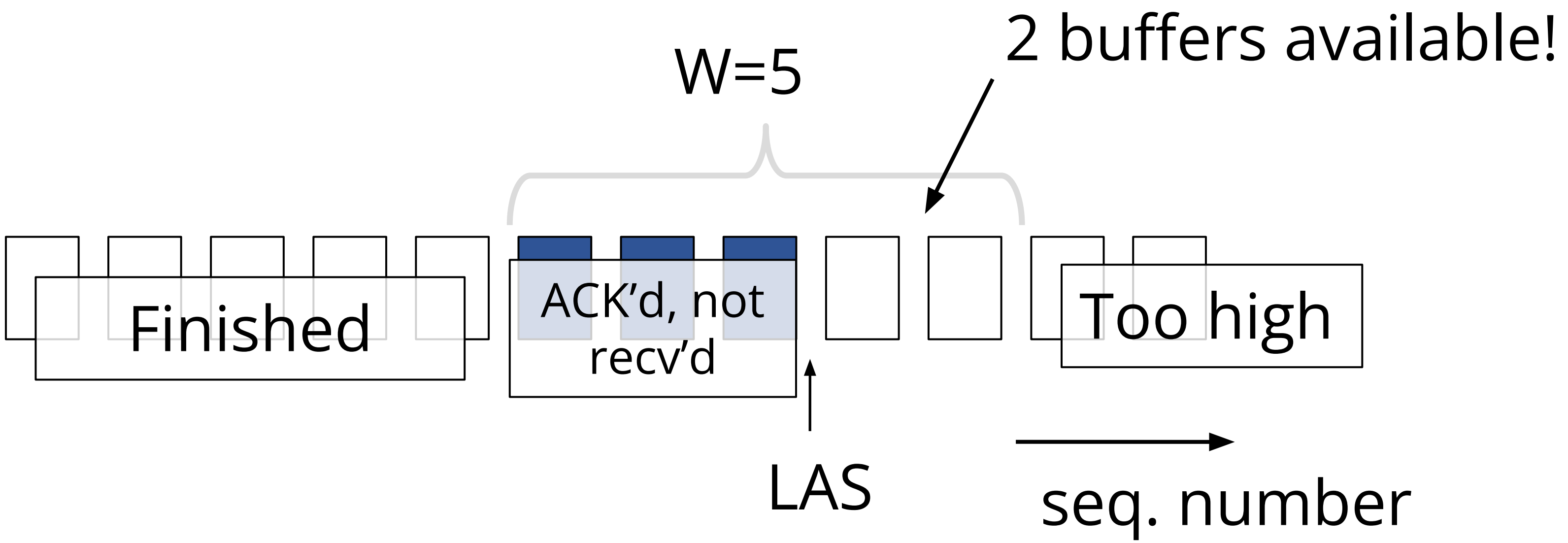
Sliding Window – Receiver Constraints (3)

- Further segments arrive (in order)… fill entire buffer
  - Must drop segments until app recvs!



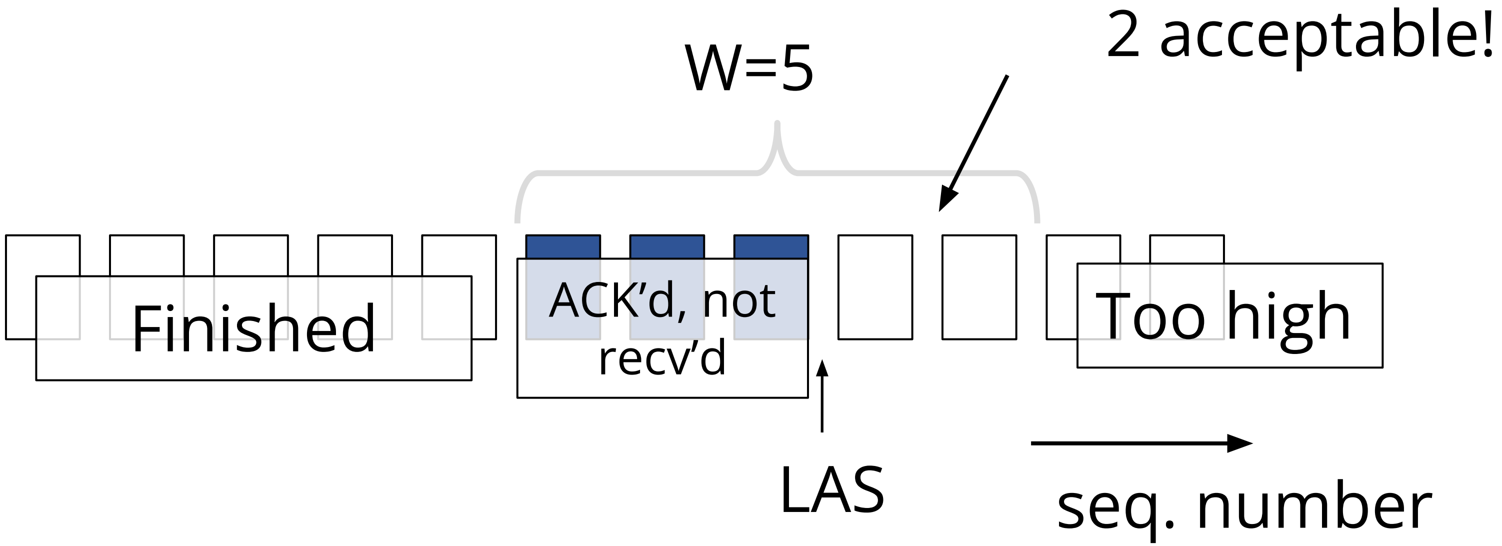
Sliding Window – Receiver Constraints (4)

- App recv() takes two segments
  - Buffer window slides (phew)



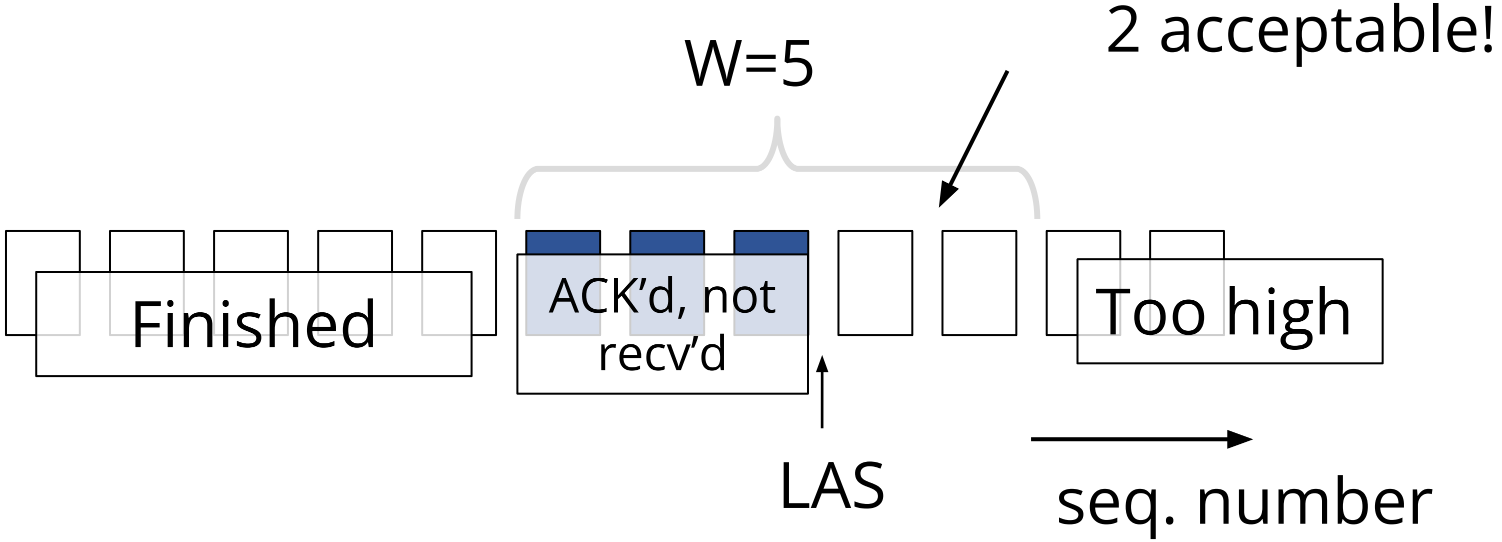
Flow Control

- Avoid loss at receiver by telling sender the available buffer space
  - WIN=#Acceptable, not just `W + LAS`



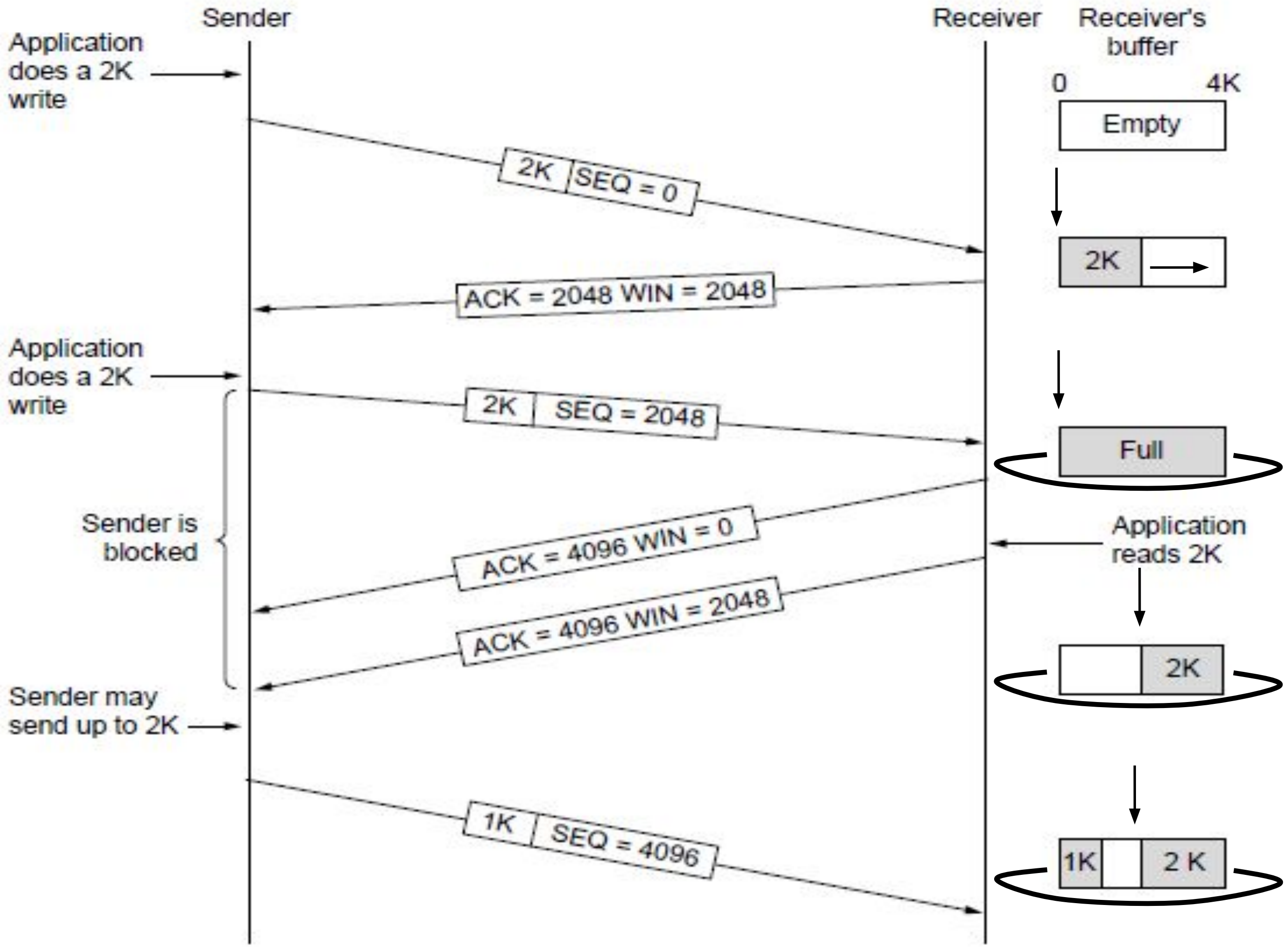
Flow Control (2)

- Acks include receiver's `WIN`
- Sender uses lesser value as the effective window size
  - min({sliding window `W + LAS`}, {flow control window (`WIN`)})



Flow Control (3)

- TCP-style example
  - seq/ack sliding window
  - Flow control with `WIN`
  - `seq + length < ack + win`
  - 4KB buffer at receiver
    - Implemented as a circular buffer of bytes



TCP Uses ACK Clocking + Flow Control

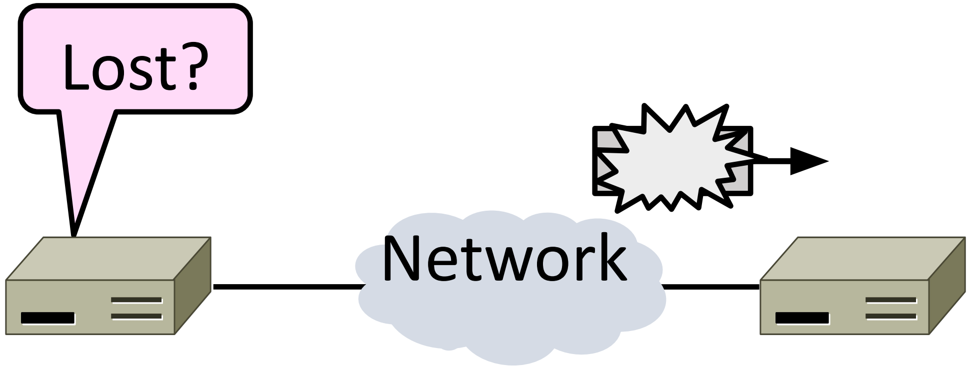
- TCP uses a sliding window because of the value of ACK clocking
- Sliding window controls how many segments are inside the network
- TCP only sends small bursts of segments to let the network keep the traffic smooth
- Some implementation details
  - (see “Silly Window Syndrome” in reading)
  - **“Nagle’s Algorithm” has implications for high-performance work!**

Optimizing the Timeout

How do we actually know the value to use??? 🤨😵😩

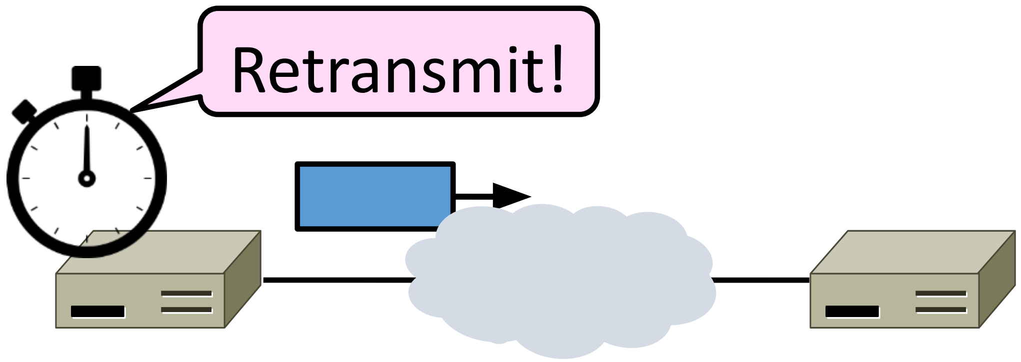
Topic

- How to set the timeout for sending a retransmission
  - Adapting to the network path



Retransmissions

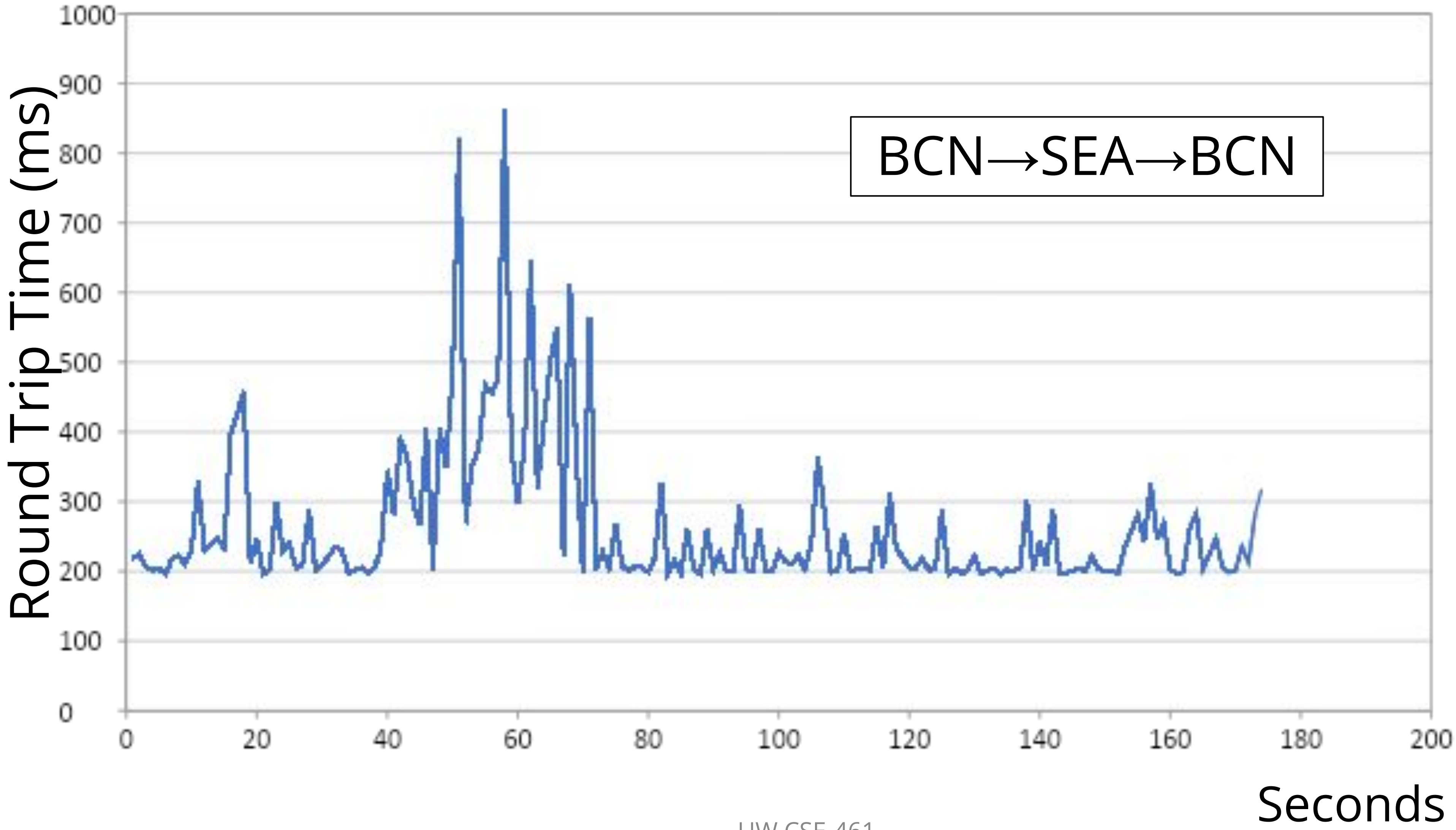
- With sliding window, detecting loss with <u>timeout</u>
  - Set timer when a segment is sent
  - Cancel timer when ack is received
  - If timer fires, <u>retransmit</u> data as lost



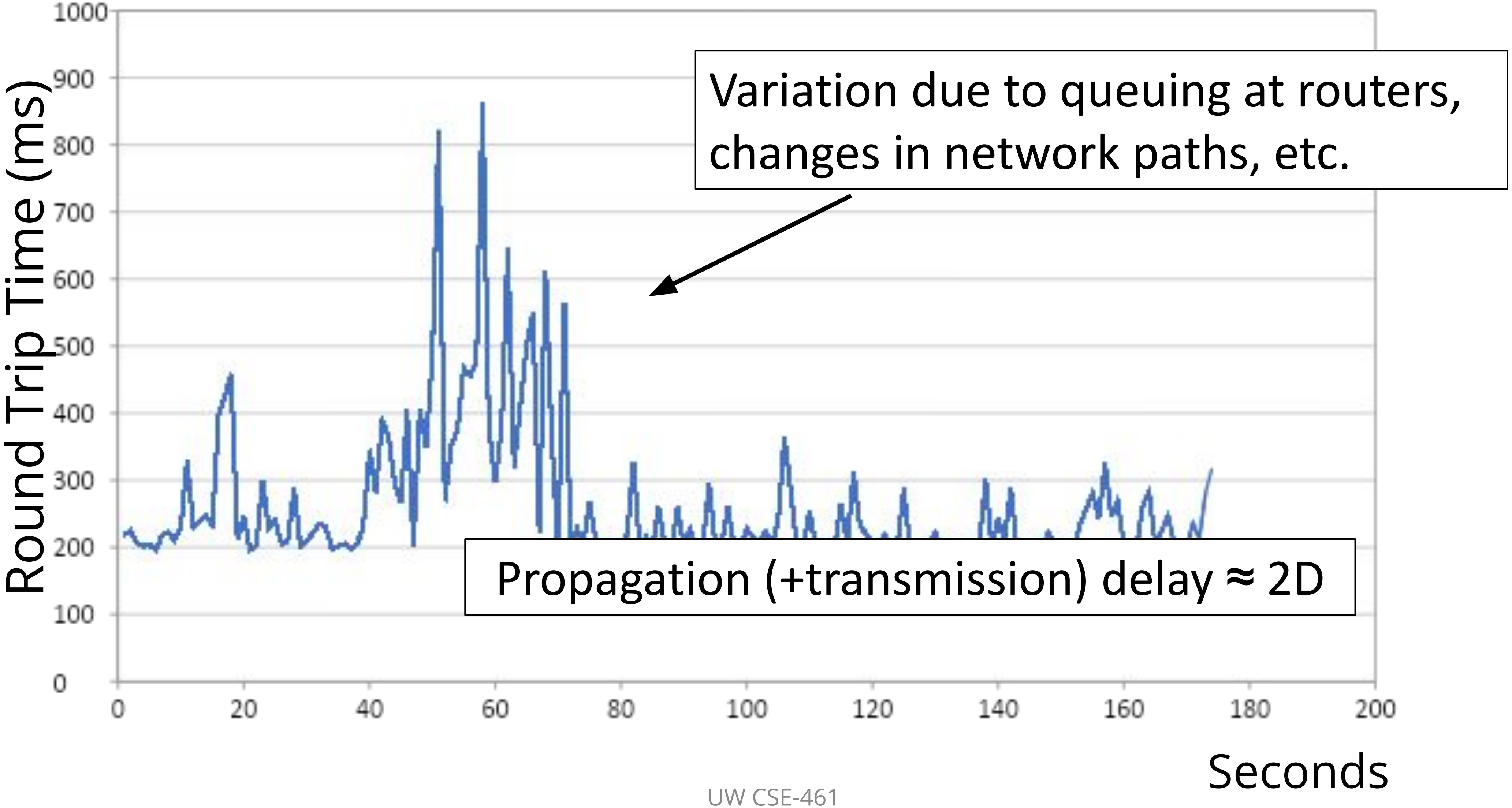
The Timeout Problem

- Timeout should be "just right"
  - Too long wastes network capacity
  - Too short leads to spurious resends
  - But what is "just right"?
- Easy to set on a LAN (Link)
  - Short, fixed, predictable RTT
- Hard on the Internet (Transport)
  - Wide range, variable RTT

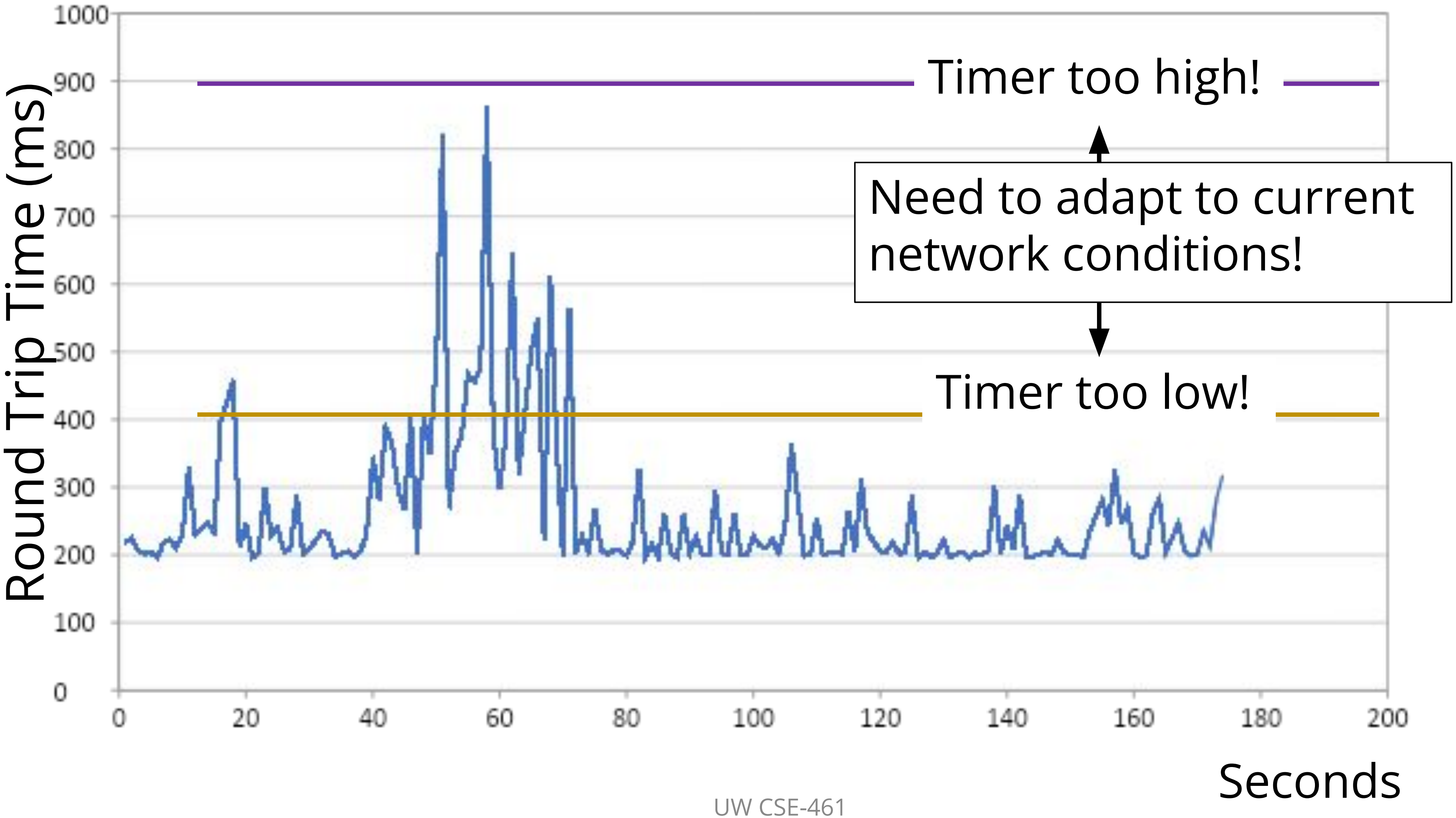
Example of RTTs



Example of RTTs (2)



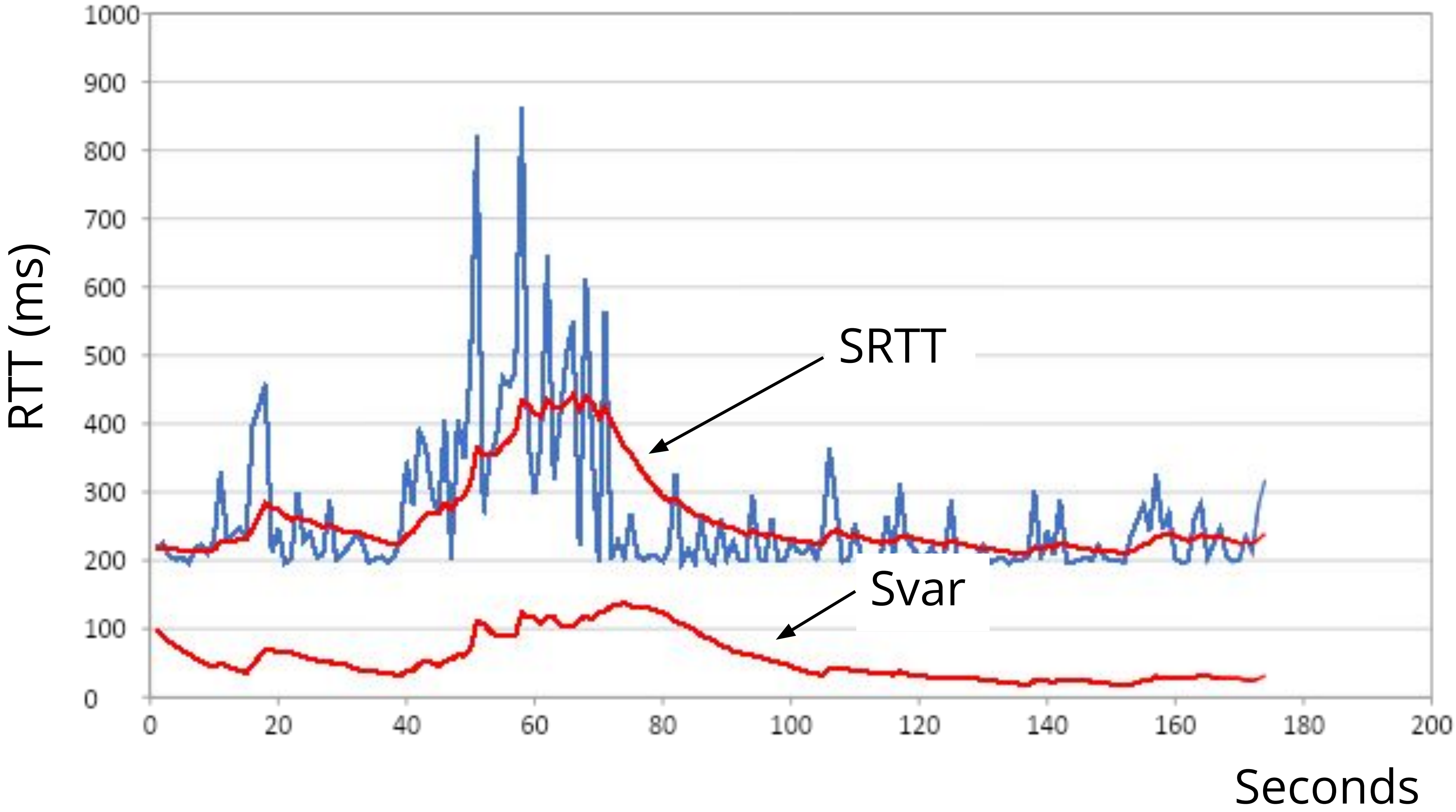
Example of RTTs (3)



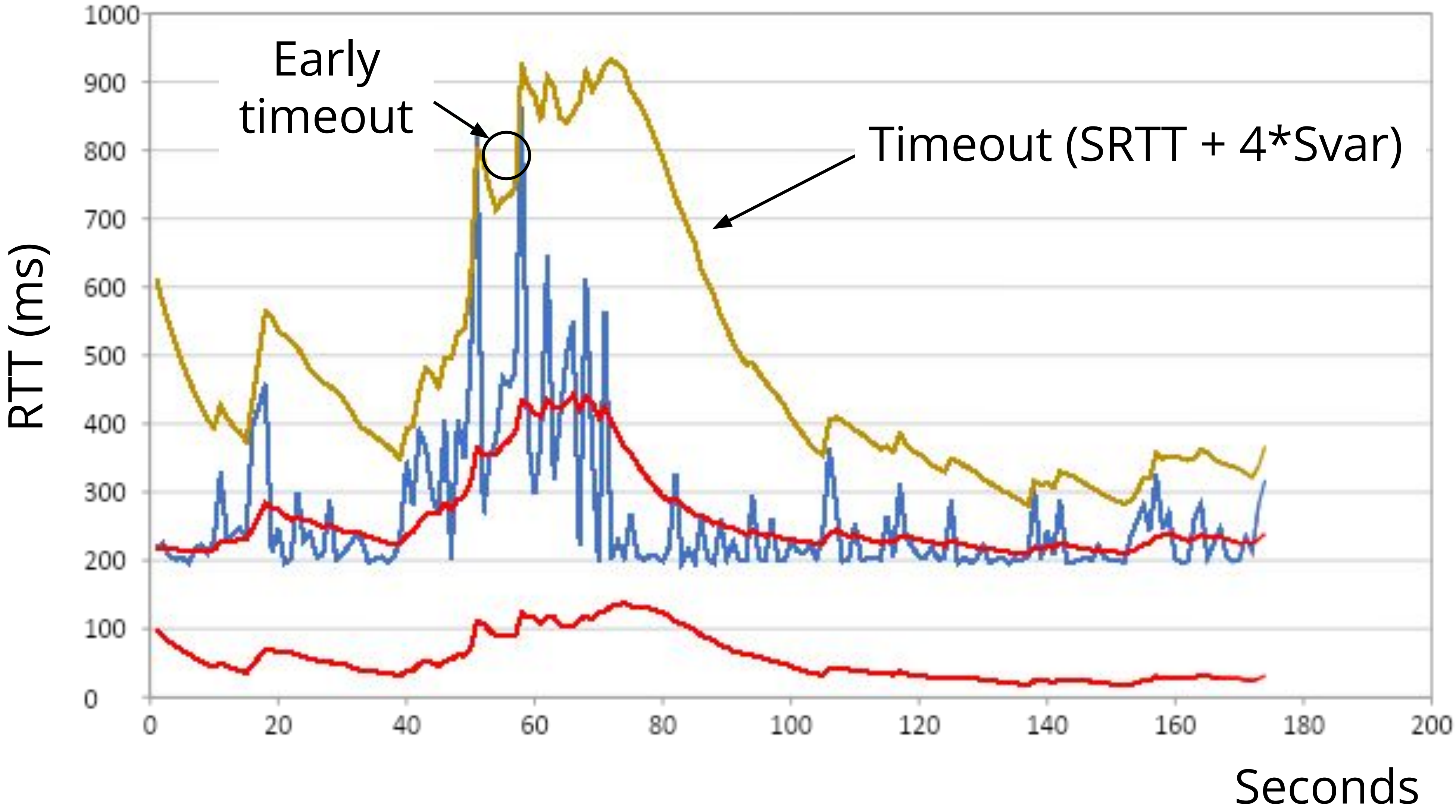
Simple Adaptive Timeout

- Smoothed estimates of the RTT (1) *and* variance in RTT (2)
  - Update estimates with an exponentially weighted moving average
    - $SRTT_{N+1} = 0.9*SRTT_N + 0.1*RTT_{N+1}$
    - $Svar_{N+1} = 0.9*Svar_N + 0.1* |RTT_{N+1} – SRTT_{N+1}|$
- Set timeout to a multiple of estimates
  - To estimate the upper RTT in practice
    - TCP $Timeout_N = SRTT_N + 4*Svar_N$

Example of Adaptive Timeout



Example of Adaptive Timeout (2)



Adaptive Timeout Summary

- Simple to compute, does a good job of tracking actual RTT
  - Little “headroom” to lower
  - Yet very few early timeouts

- Turns out to be important for good performance and robustness

Congestion 🤧

🚐🚗🚙🚚🚗🚗

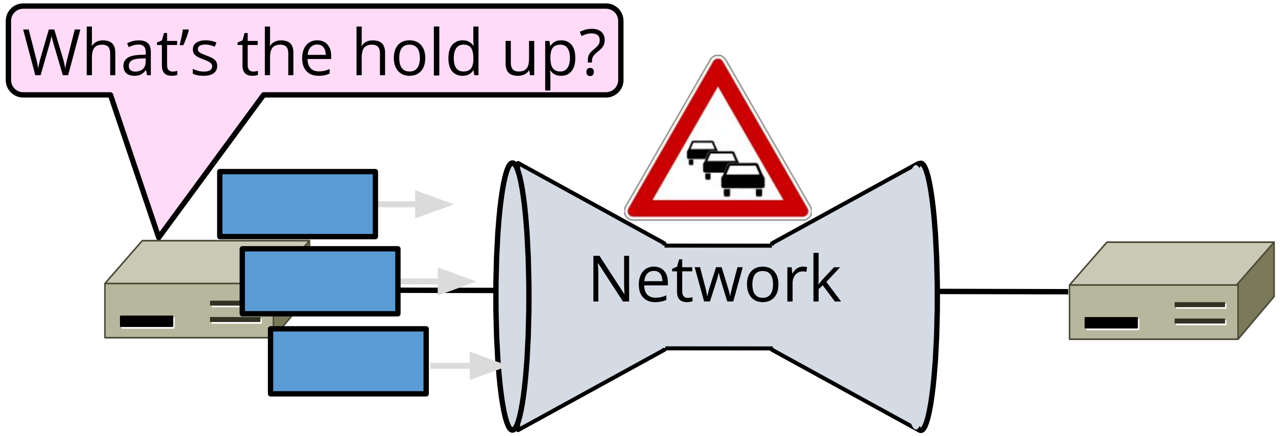
TCP to date:

- We can set up a connection
  - (connection establishment)
- Tear down a connection
  - (connection release)
- Keep the sending and receiving buffers from overflowing
  - (flow control)

What’s missing?

Network Congestion

- A “traffic jam” in the network
  - Later we will learn how to control it

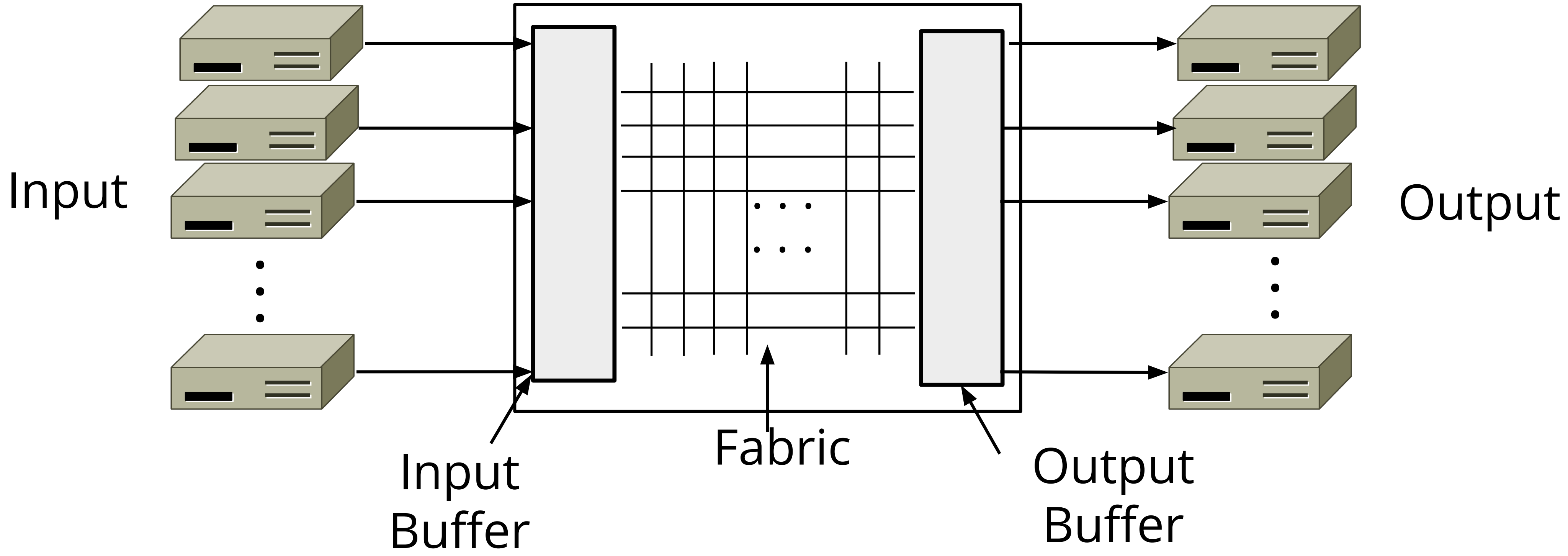


Congestion Collapse in the 1980s

- Early TCP used fixed size window (e.g., 8 packets)
  - Initially fine for reliability
- But something happened as the ARPANET grew
  - Links stayed busy but transfer rates fell by orders of magnitude!

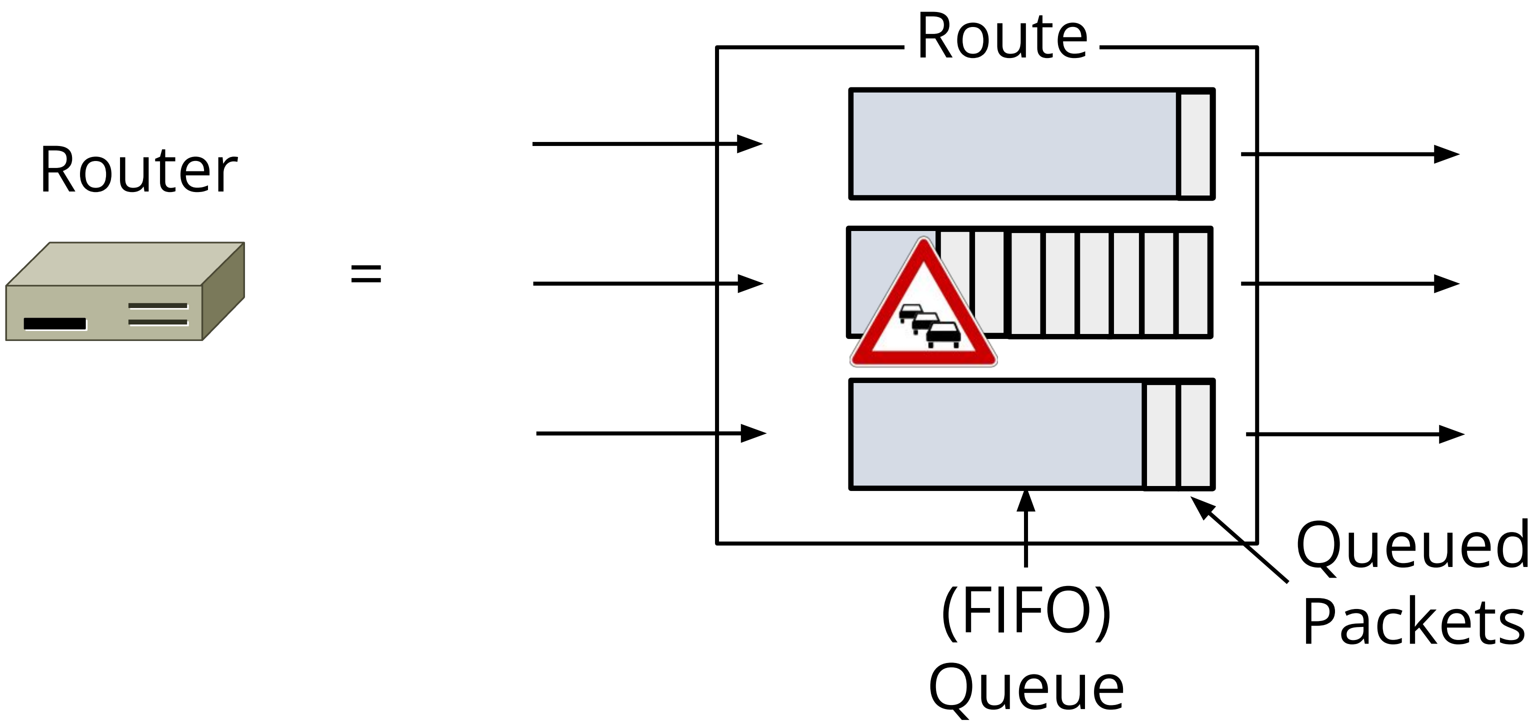
Nature of Congestion

- Routers/switches have internal buffering



Nature of Congestion (2)

- Simplified view of per port output queues
  - Typically FIFO (First In First Out), discard when full

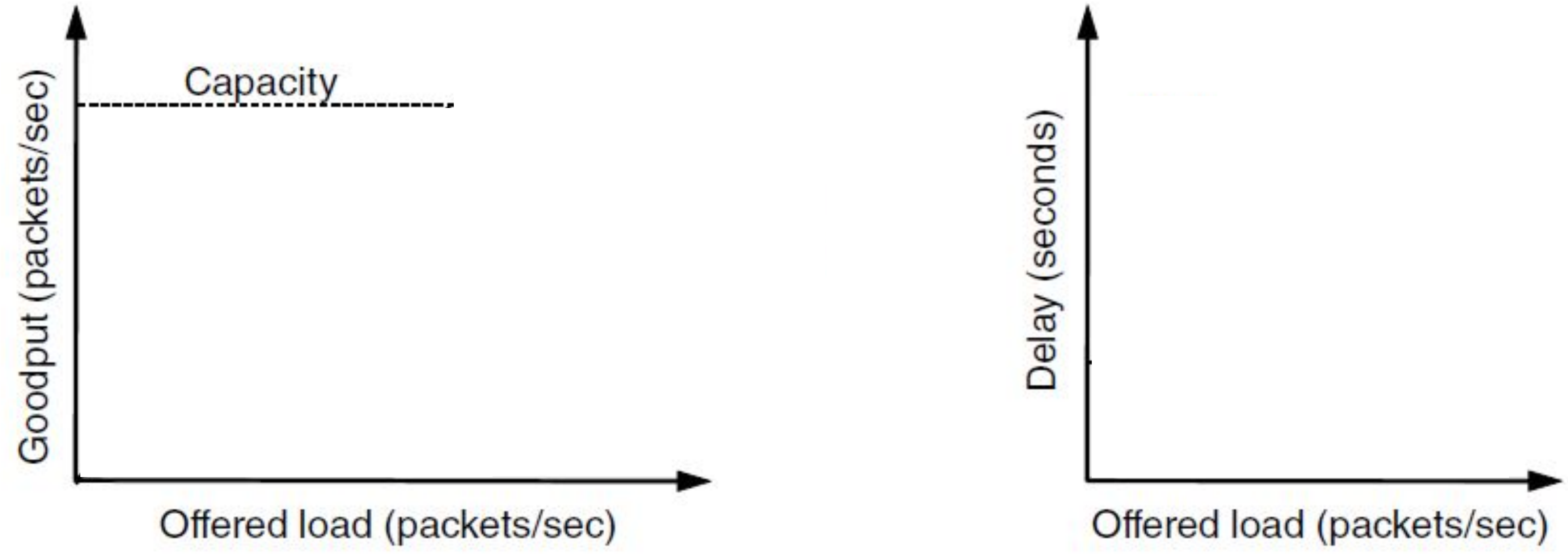


Nature of Congestion (3)

- Queues help by absorbing bursts when input > output rate
- But if input > output rate persistently, *queue will overflow*
  - This is congestion
- Congestion is a function of the traffic patterns
  - Can occur even if every link has the same capacity!

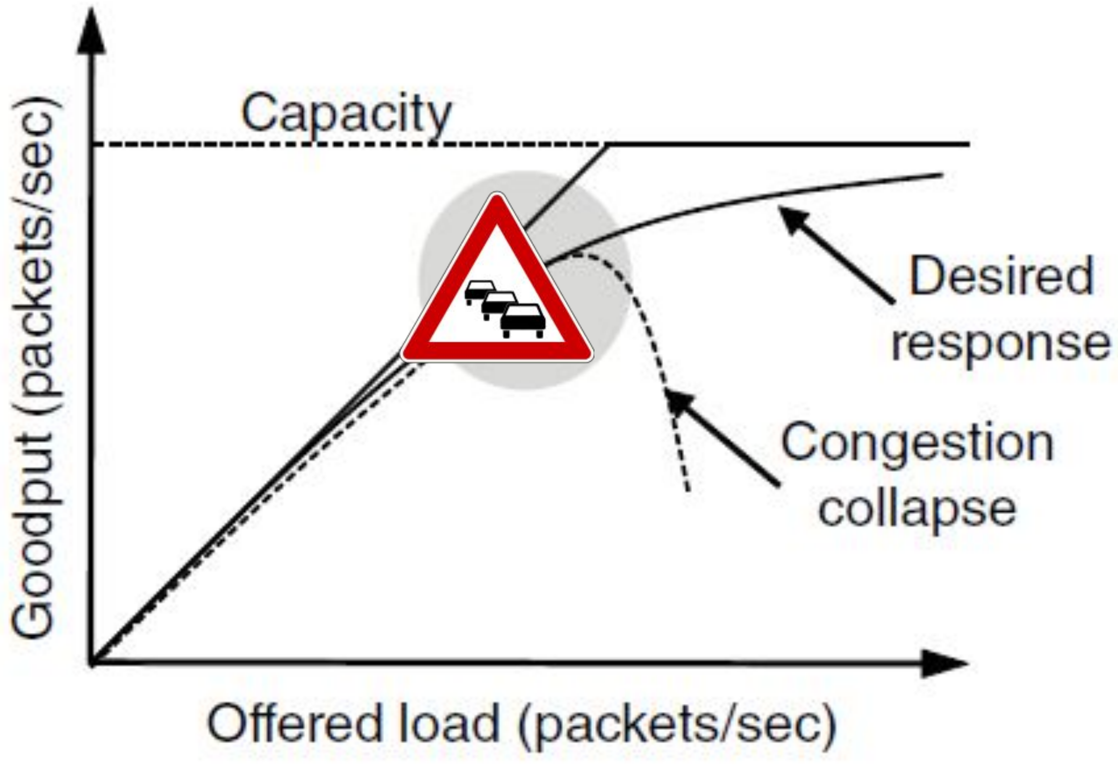
Effects of Congestion

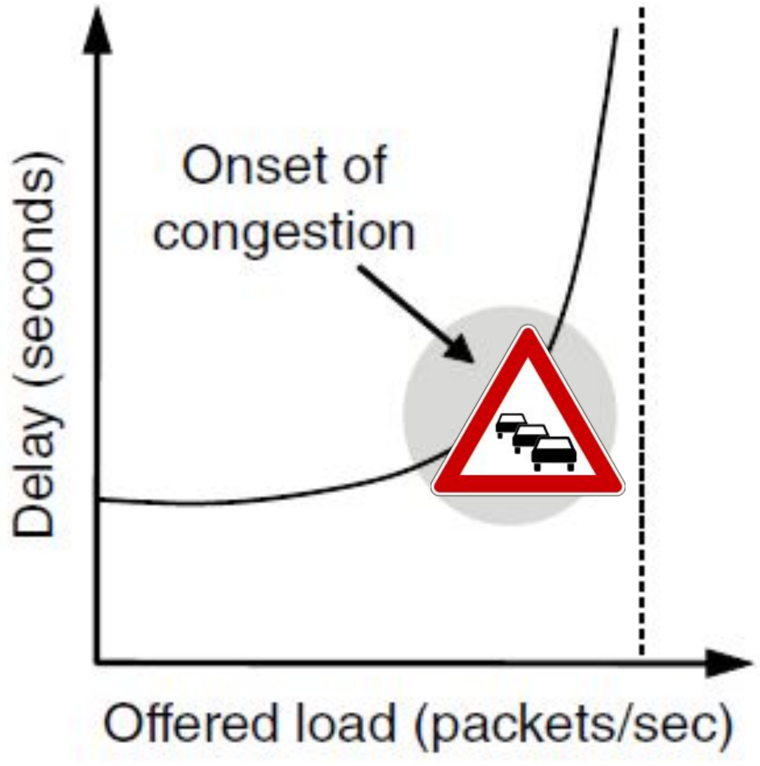
- What happens to performance as we increase load?



Effects of Congestion (2)

- What happens to performance as we increase load?





Effects of Congestion (3)

- As offered load rises, congestion occurs as queues begin to fill:
  - Delay and loss rise sharply with more load
  - Throughput falls below load (due to loss)
  - Goodput may fall below throughput (due to spurious retransmissions)
- None of the above is good!
  - Want network performance just before congestion

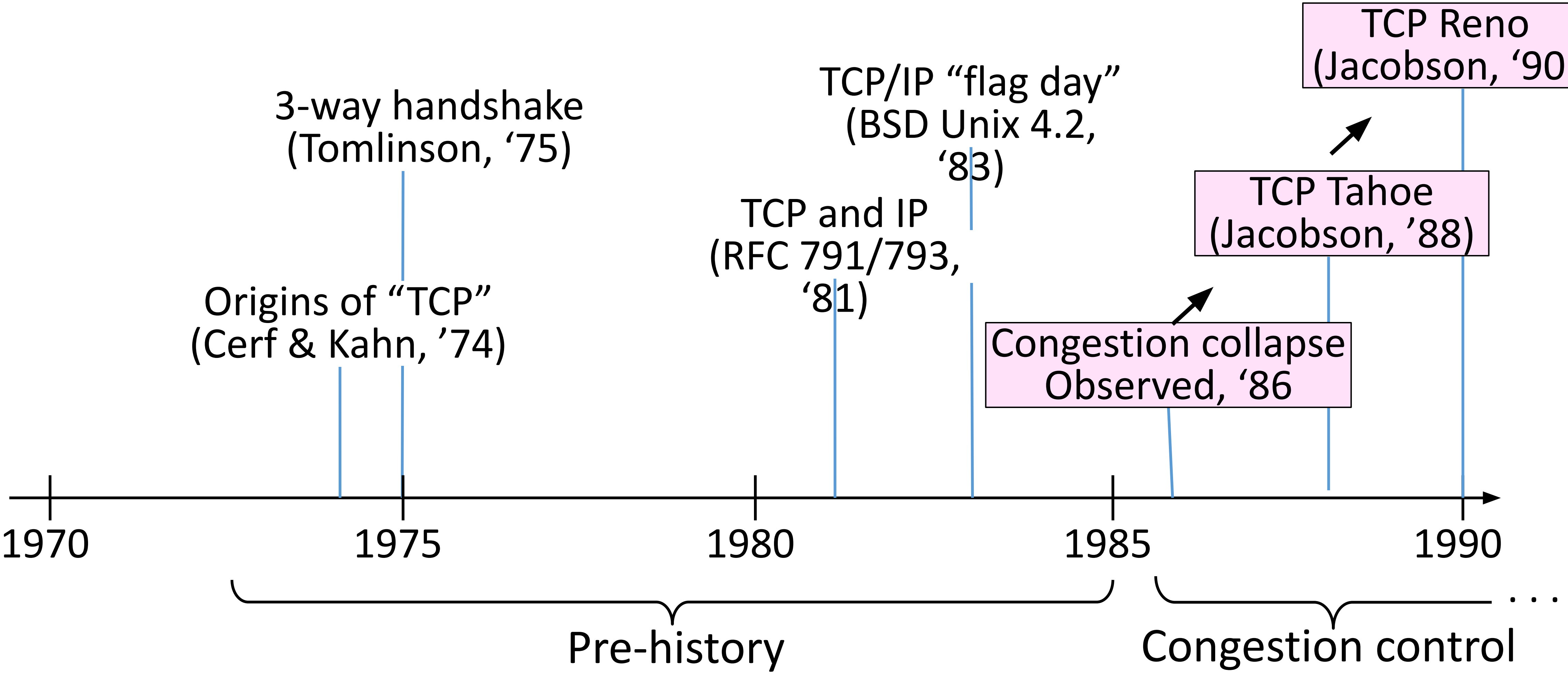
Van Jacobson (1950—)

- Widely credited with saving the Internet from congestion collapse in the late 80s
  - Introduced congestion control principles
  - Practical solutions (TCP Tahoe/Reno)
- Much other pioneering work:
  - Tools like traceroute, tcpdump, pathchar
  - IP header compression, multicast tools

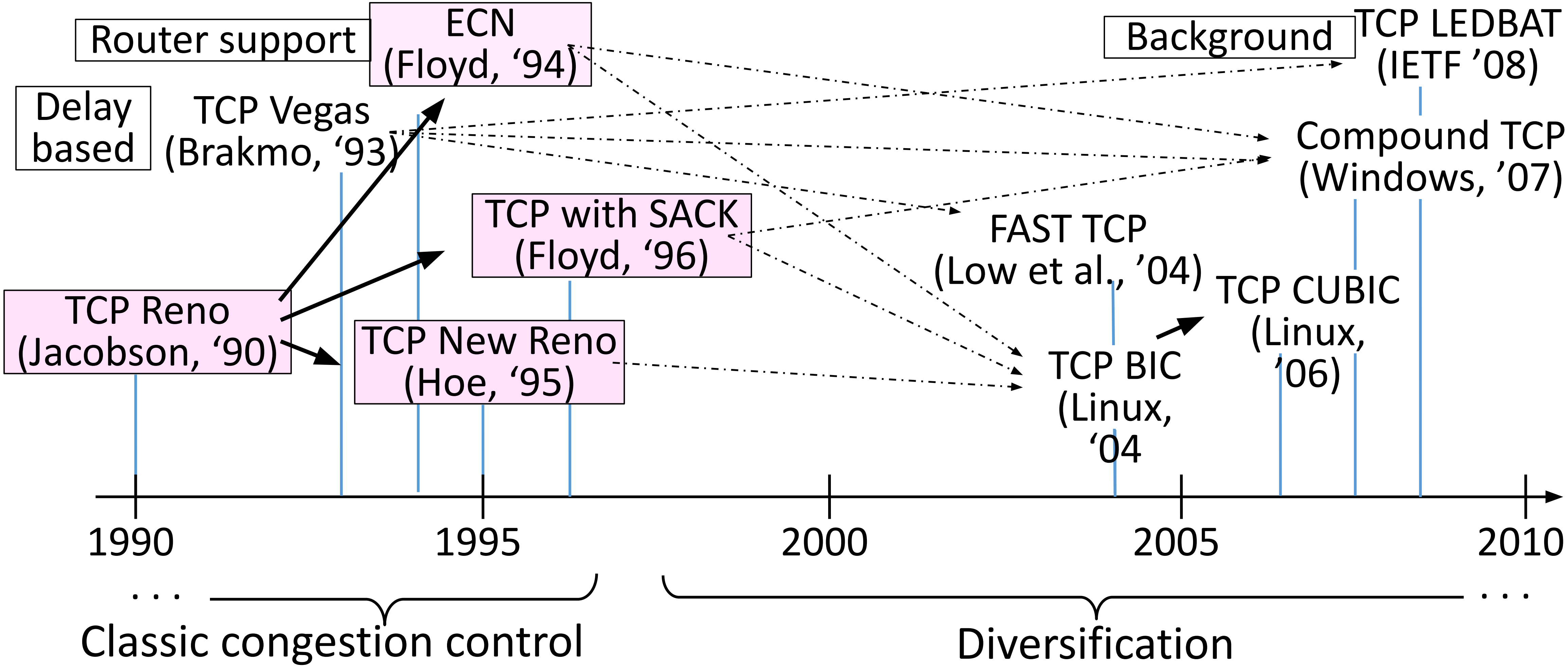
TCP Tahoe/Reno

- TCP extensions and features we will study:
  - AIMD
  - Fair Queuing
  - Slow-start
  - Fast Retransmission
  - Fast Recovery

TCP Timeline



TCP Timeline (2)



Bandwidth Allocation

- Important task for network is to allocate its capacity to senders
  - Good allocation is both efficient and fair
- <u>Efficient</u> means most capacity is used but there is no congestion
- <u>Fair</u> means every sender gets a reasonable share the network

Bandwidth Allocation (2)

- Key observation:
  - In an effective solution, Transport and Network layers must work together
- Network layer witnesses congestion
  - Only it can provide direct feedback
- Transport layer causes congestion
  - Only it can reduce offered load

Bandwidth Allocation (3)

- Why is it hard? (Just split equally!)
  - Number of senders and their offered load changes
  - Senders may lack capacity in different parts of network
  - Network is distributed; no single party has an overall picture of its state

Bandwidth Allocation (4)

- Solution context:
  - Senders adapt concurrently based on their own view of the network
  - Design this adaption so the network usage as a whole is efficient and fair
  - Adaption is continuous since offered loads continue to change over time

Fair Allocations

Fair Allocation

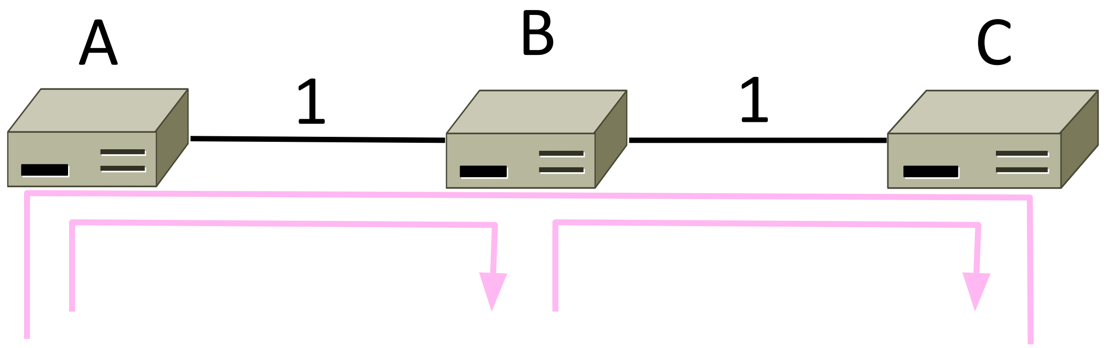
- What's a "fair" bandwidth allocation?
  - The max-min fair allocation is one that's often cited
    - There are *huge* assumptions baked into this definition of fairness, which we'll talk about the last week!

Recall

- We want a good bandwidth allocation to be both fair and efficient
  - Now we learn what fair means
- Caveat: in practice, efficiency is more important than fairness

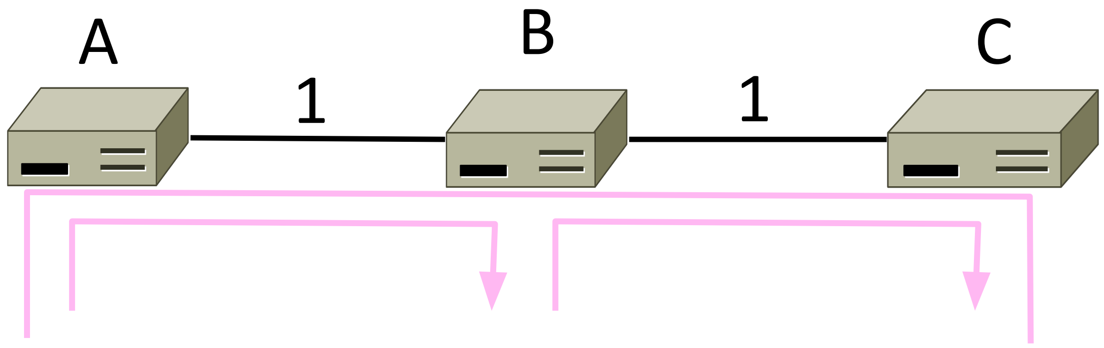
Efficiency vs. Fairness

- Cannot always have both!
  - Example network with traffic:
    - A→B, B→C and A→ C
  - How much traffic can we carry?



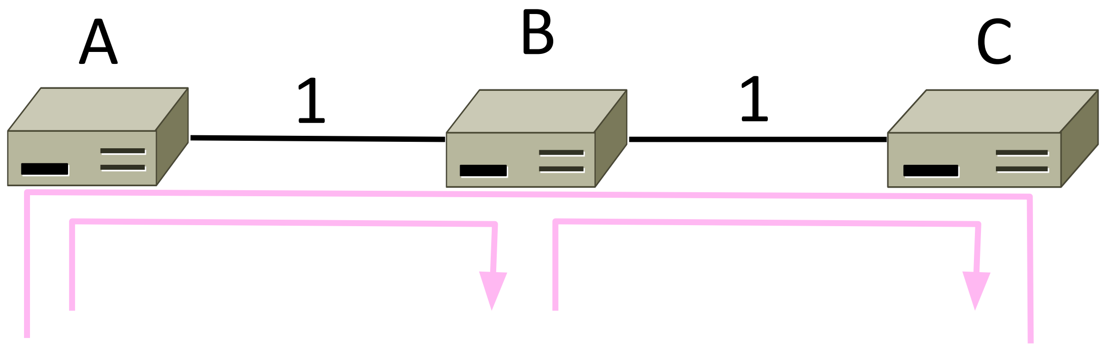
Efficiency vs. Fairness (2)

- If we care about fairness:
  - Give equal bandwidth to each flow
  - A→B: ½ unit, B→C: ½, and A→C, ½
  - Total traffic carried is 1 ½ units



Efficiency vs. Fairness (3)

- If we care about efficiency:
  - Maximize total traffic in network
  - A→B: 1 unit, B→C: 1, and A→C, 0
  - Total traffic rises to 2 units!

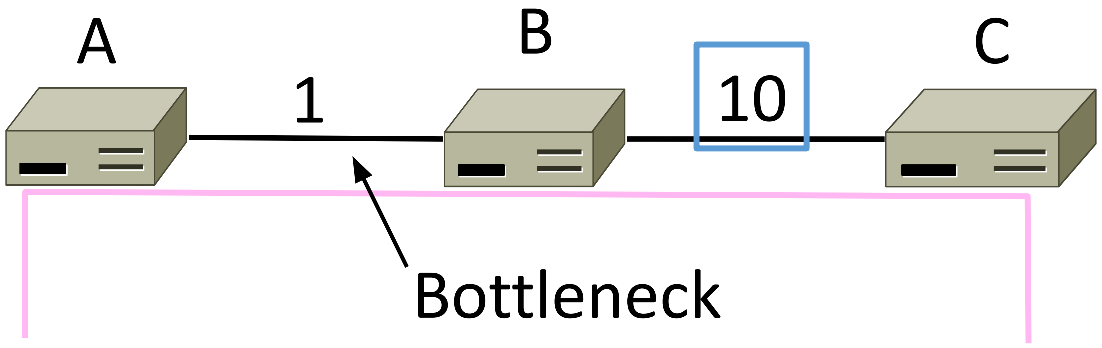


The Slippery Notion of Fairness

- Why is “equal per flow” fair anyway?
  - A→C uses more network resources than A→B or B→C
  - Host A sends two flows, B sends one
- Not productive to seek exact fairness
  - More important to avoid <u>starvation</u>
    - A node that cannot use any bandwidth
  - “Equal per flow” is good enough

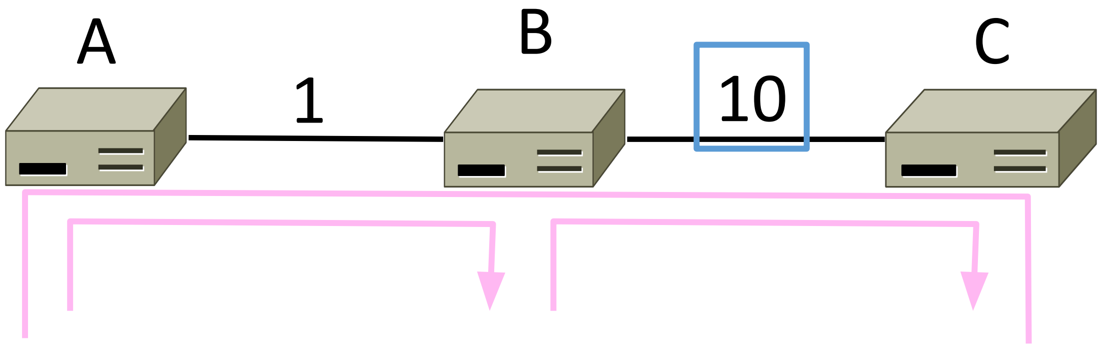
Generalizing "Equal per Flow"

- <u>Bottleneck</u> for a flow of traffic is the link that limits its bandwidth
  - Where congestion occurs for the flow
  - For A→C, link A–B is the bottleneck



Generalizing "Equal per Flow" (2)

- Flows may have different bottlenecks
  - For A→C, link A–B is the bottleneck
  - For B→C, link B–C is the bottleneck
  - Can no longer divide links equally …



Max-Min Fairness

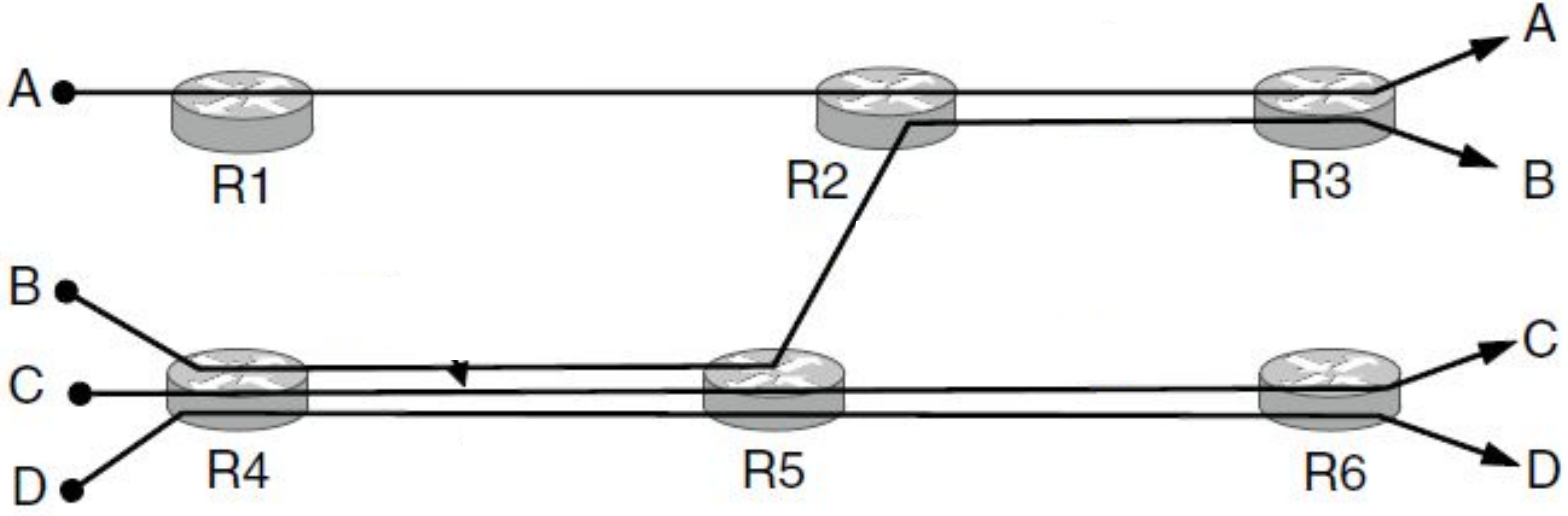
- Intuitively, flows bottlenecked on a link get an equal share of that link
- <u>Max-min fair allocation</u> is one that:
  - Increasing the rate of one flow will decrease the rate of a smaller flow
  - This “maximizes the minimum” flow

Max-Min Fairness (2)

- To find it given a network, imagine "pouring water into the network"
  1. Start with all flows at rate 0
  2. Increase the flows until there is a new bottleneck in the network
  3. Hold fixed the rate of the flows that are bottlenecked
  4. Go to step 2 for any remaining flows

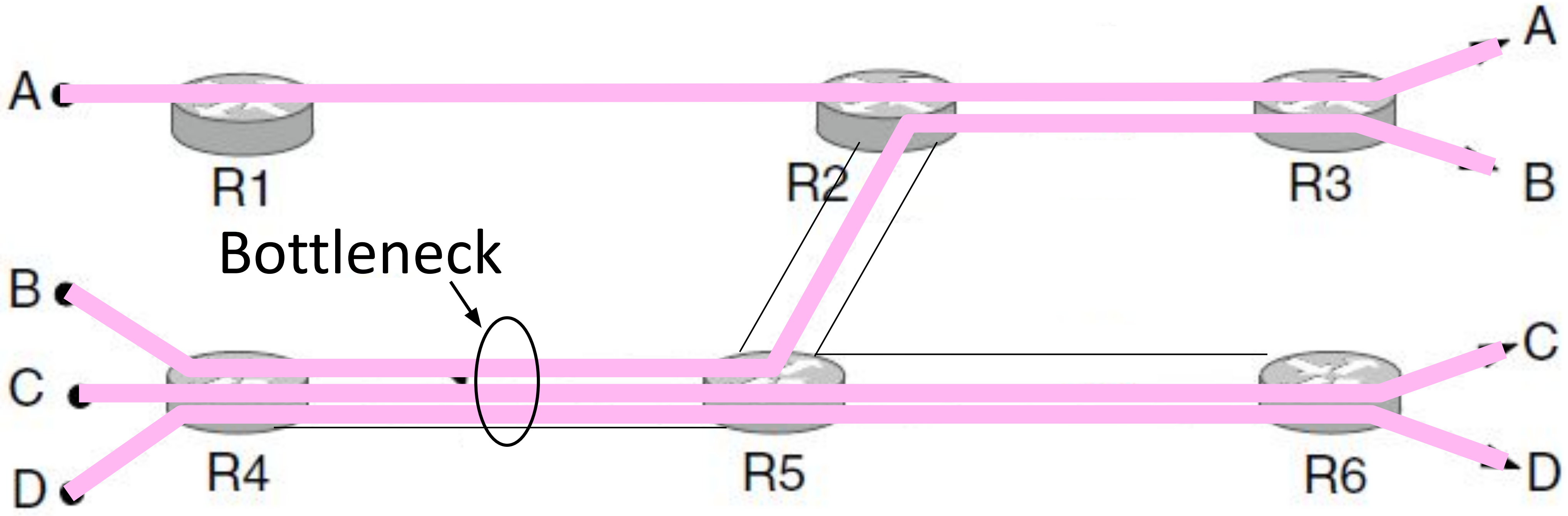
Max-Min Example

- Example: network with 4 flows, link bandwidth = 1
  - What is the max-min fair allocation?



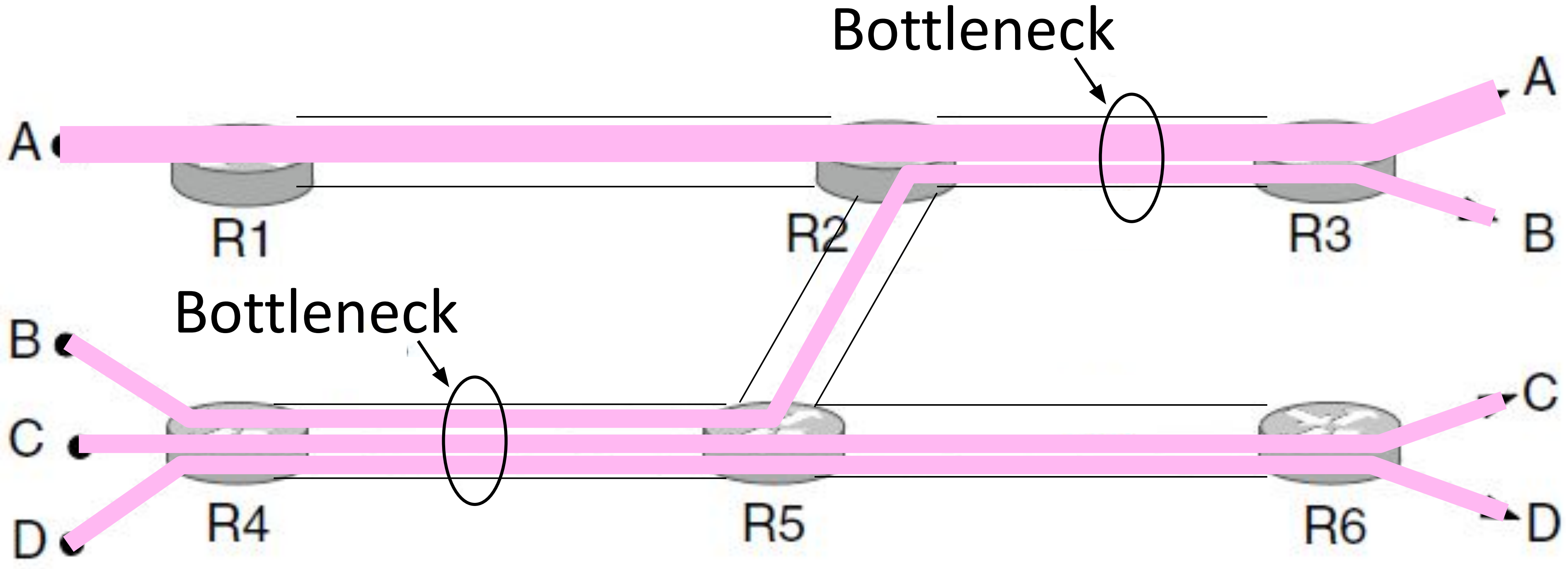
Max-Min Example (2)

- When rate=1/3, flows B, C, and D bottleneck R4—R5
  - Fix B, C, and D, continue to increase A



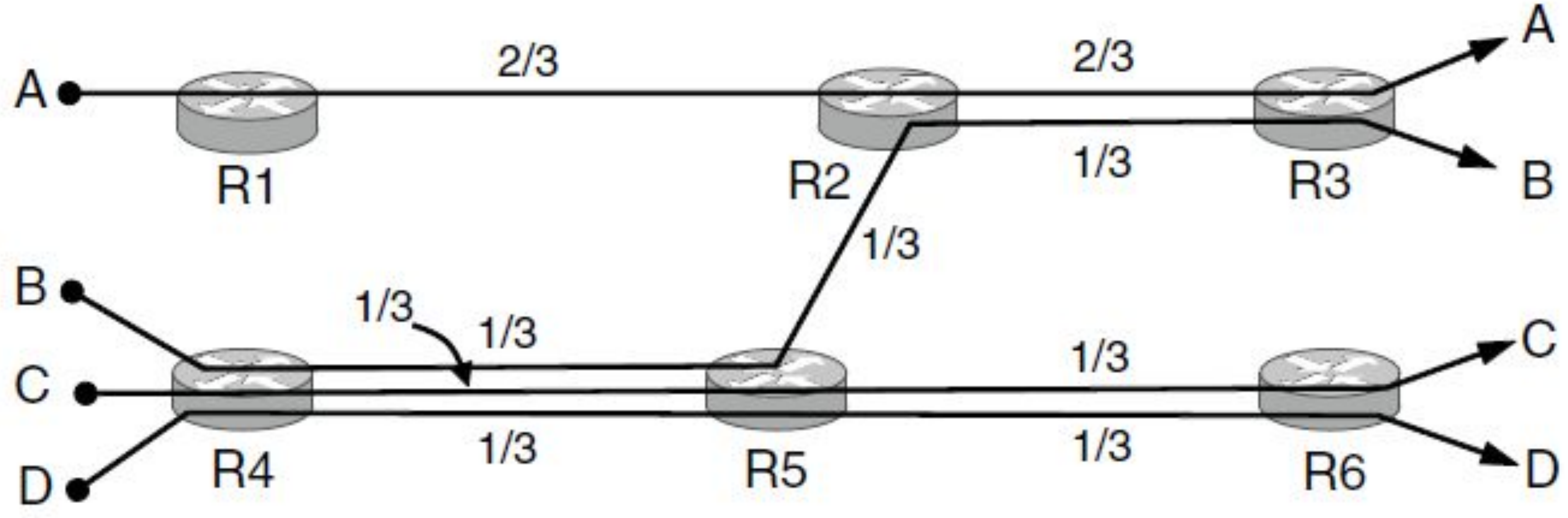
Max-Min Example (3)

- When rate=2/3, flow A bottlenecks R2—R3. Done.



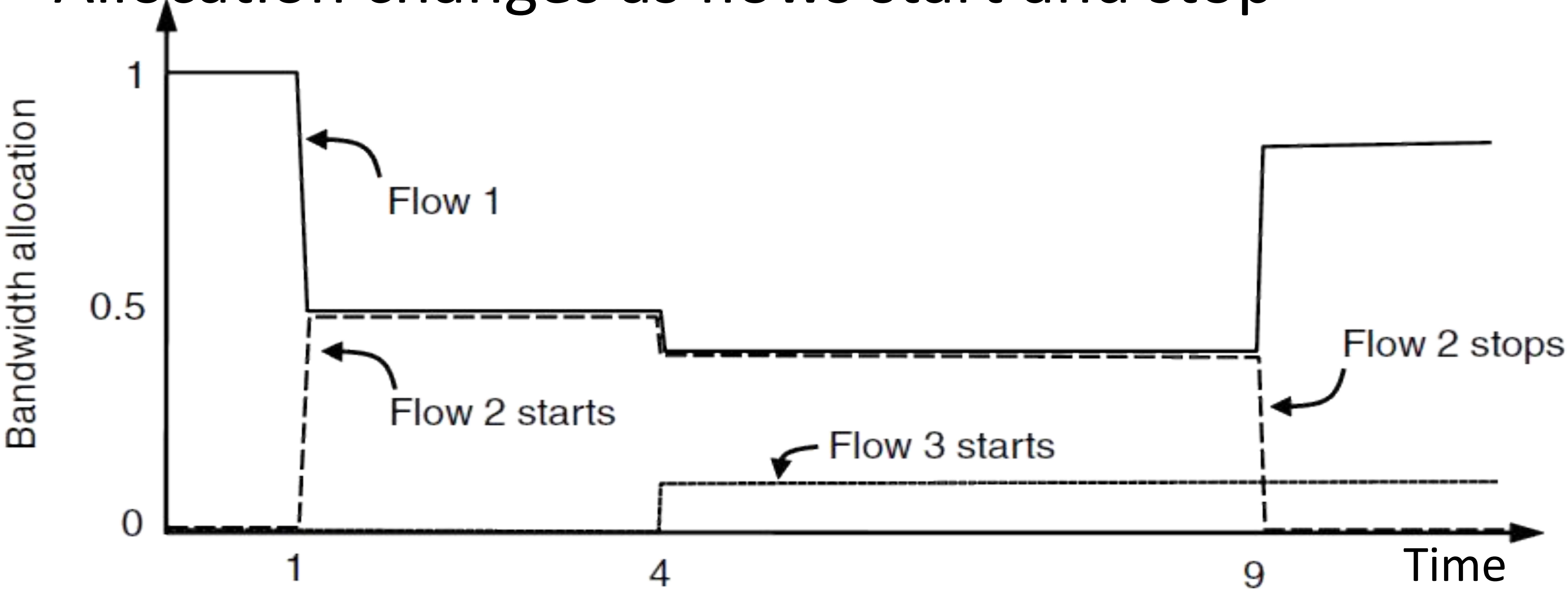
Max-Min Example (4)

- End with A=2/3, B, C, D=1/3, and R2/R3, R4/R5 full
  - Other links have extra capacity that can't be used

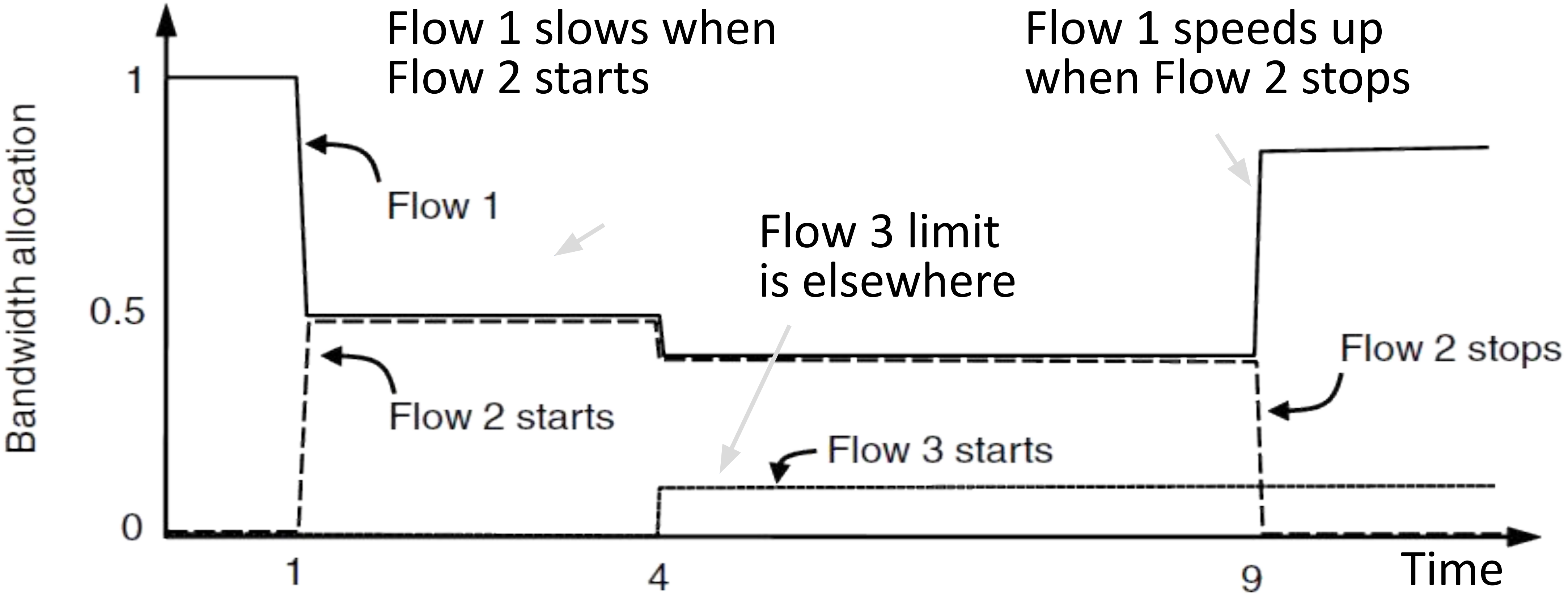


Adapting over Time

- Allocation changes as flows start and stop



Adapting over Time (2)



Bandwidth Allocation

Recall

- Want to allocate capacity to senders
  - Network layer provides feedback
  - Transport layer adjusts offered load
  - A good allocation is efficient and fair
- How should we perform the allocation?
  - Several different possibilities …

Bandwidth Allocation Models

- Open loop versus closed loop
  - Open: reserve bandwidth before use
  - Closed: use feedback to adjust rates
- Host versus Network support
  - Who is sets/enforces allocations?
- Window versus Rate based
  - How is allocation expressed?

TCP is a closed loop, host-driven, and window-based

Bandwidth Allocation Models (2)

- We'll look at closed-loop, host-driven, and window-based too
- Network layer returns <u>feedback</u> on current allocation to senders
  - For TCP signal is "a packet dropped"
- Transport layer adjusts sender's behavior via window in response
  - How senders adapt is a <u>control law</u>

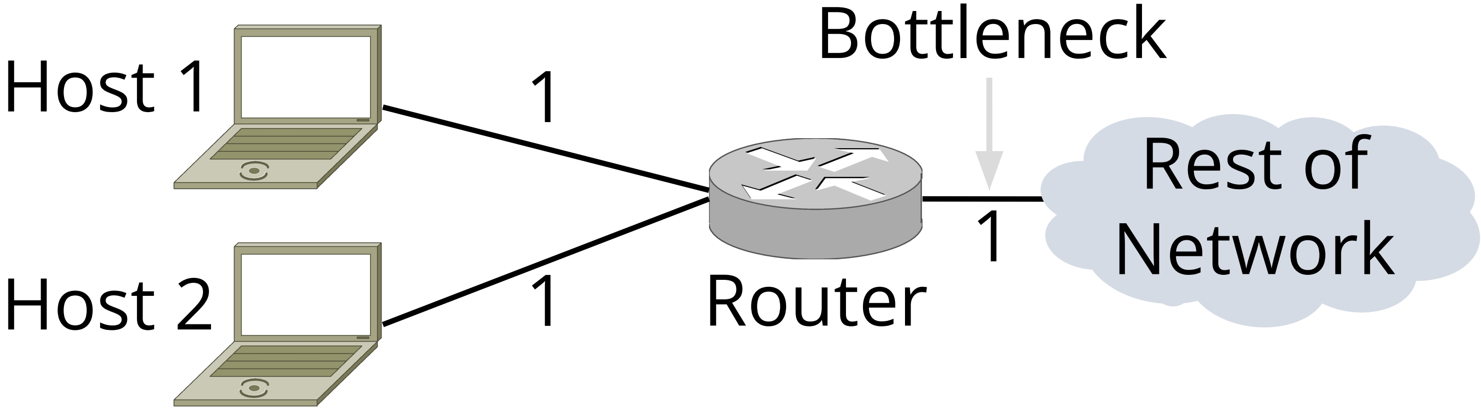
Additive Increase Multiplicative Decrease

- AIMD is a control law hosts can use to reach a good allocation
  - Hosts additively increase rate while network not congested
  - Hosts multiplicatively decrease rate when congested
  - Used by TCP

- Let’s explore the AIMD game …

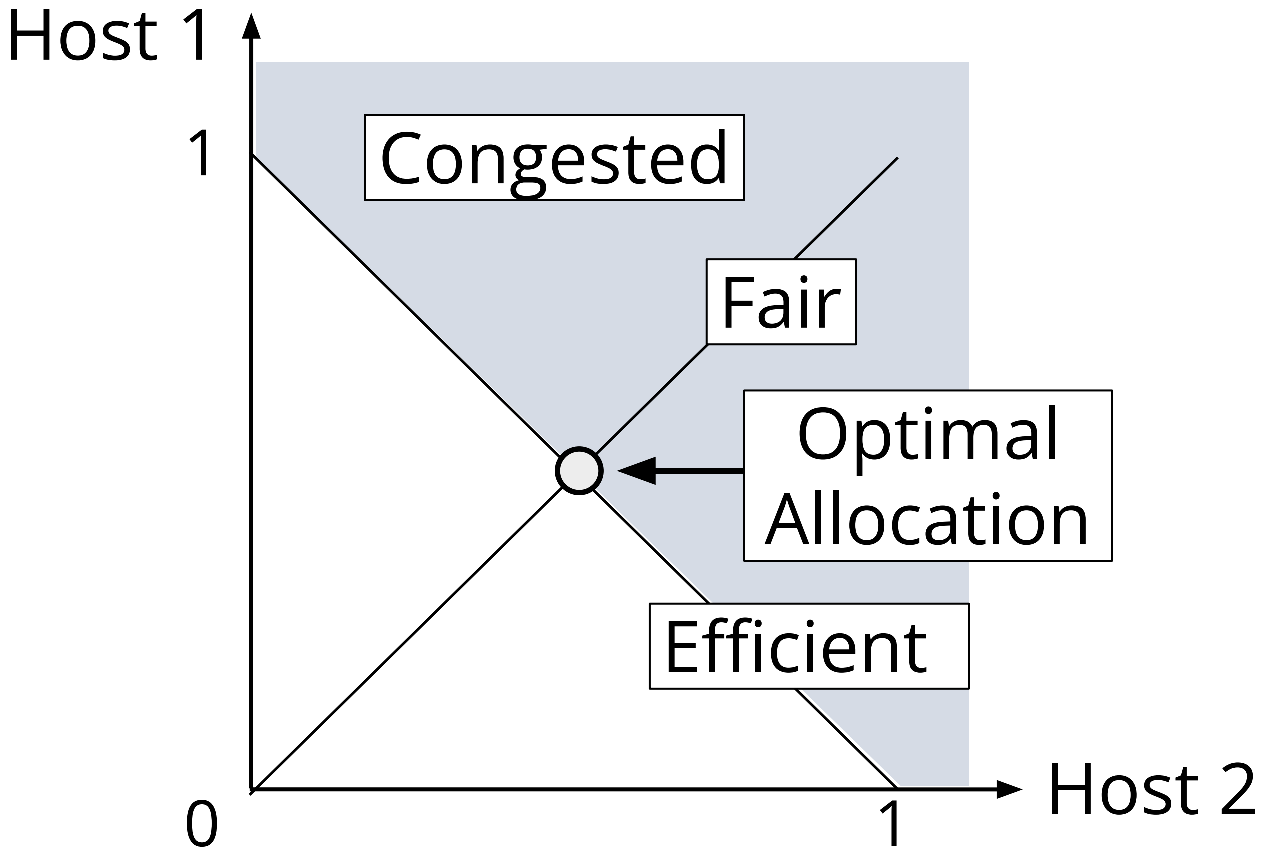
AIMD Game

- Hosts 1 and 2 share a bottleneck
  - But do not talk to each other directly
- Router provides binary feedback
  - Tells hosts if network is congested



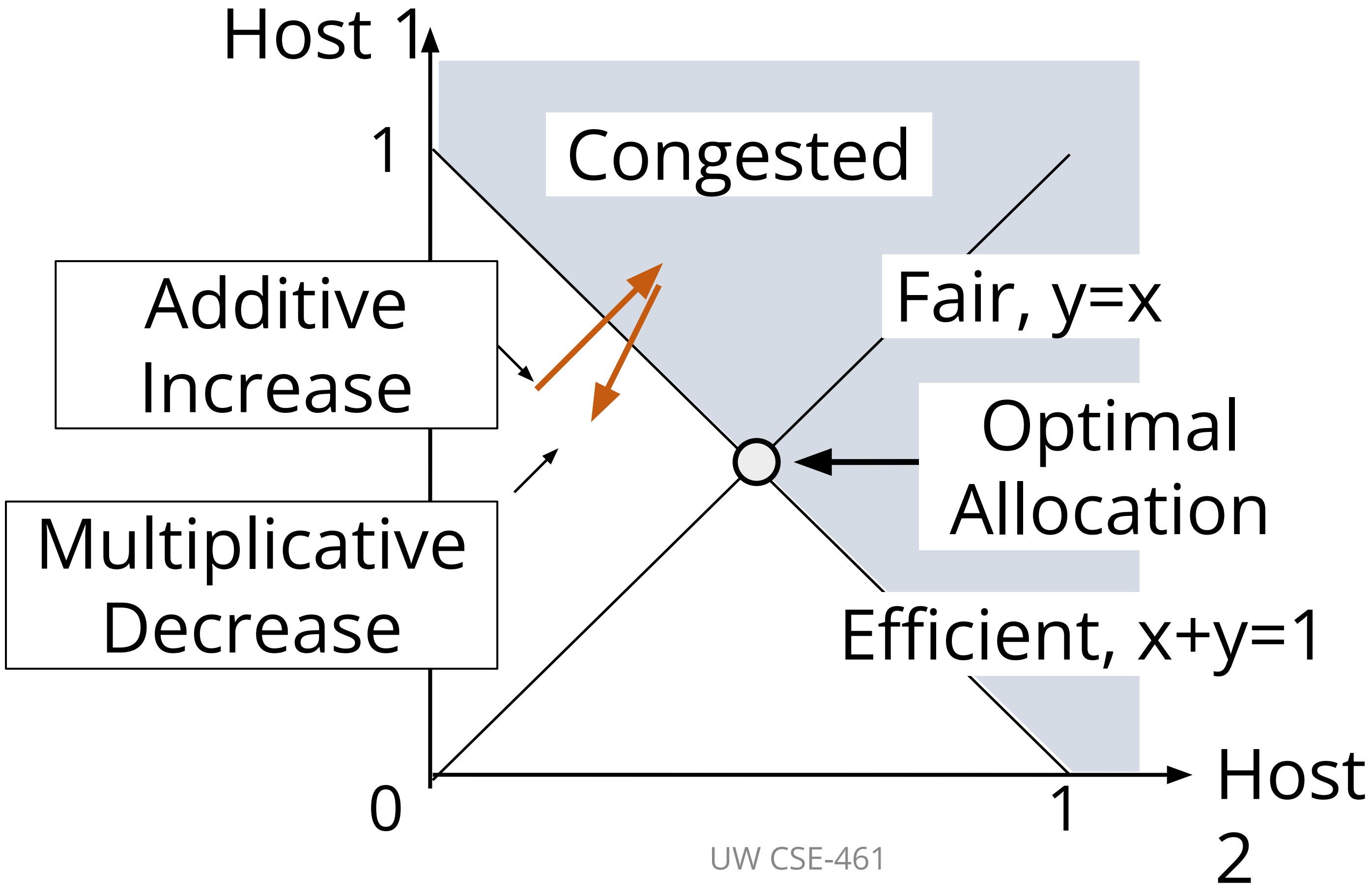
AIMD Game (2)

- Each point is a possible allocation



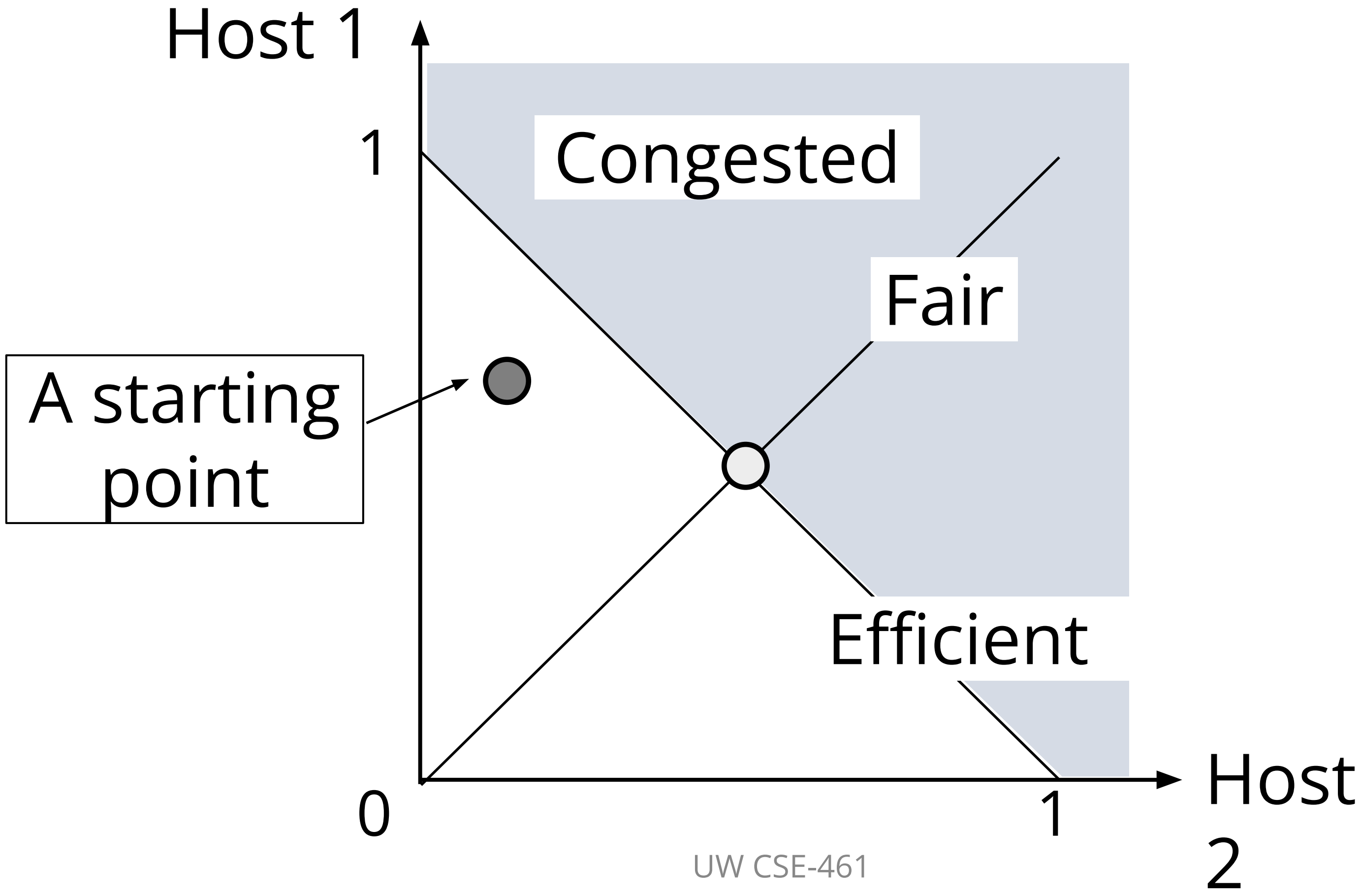
AIMD Game (3)

- AI and MD move the allocation



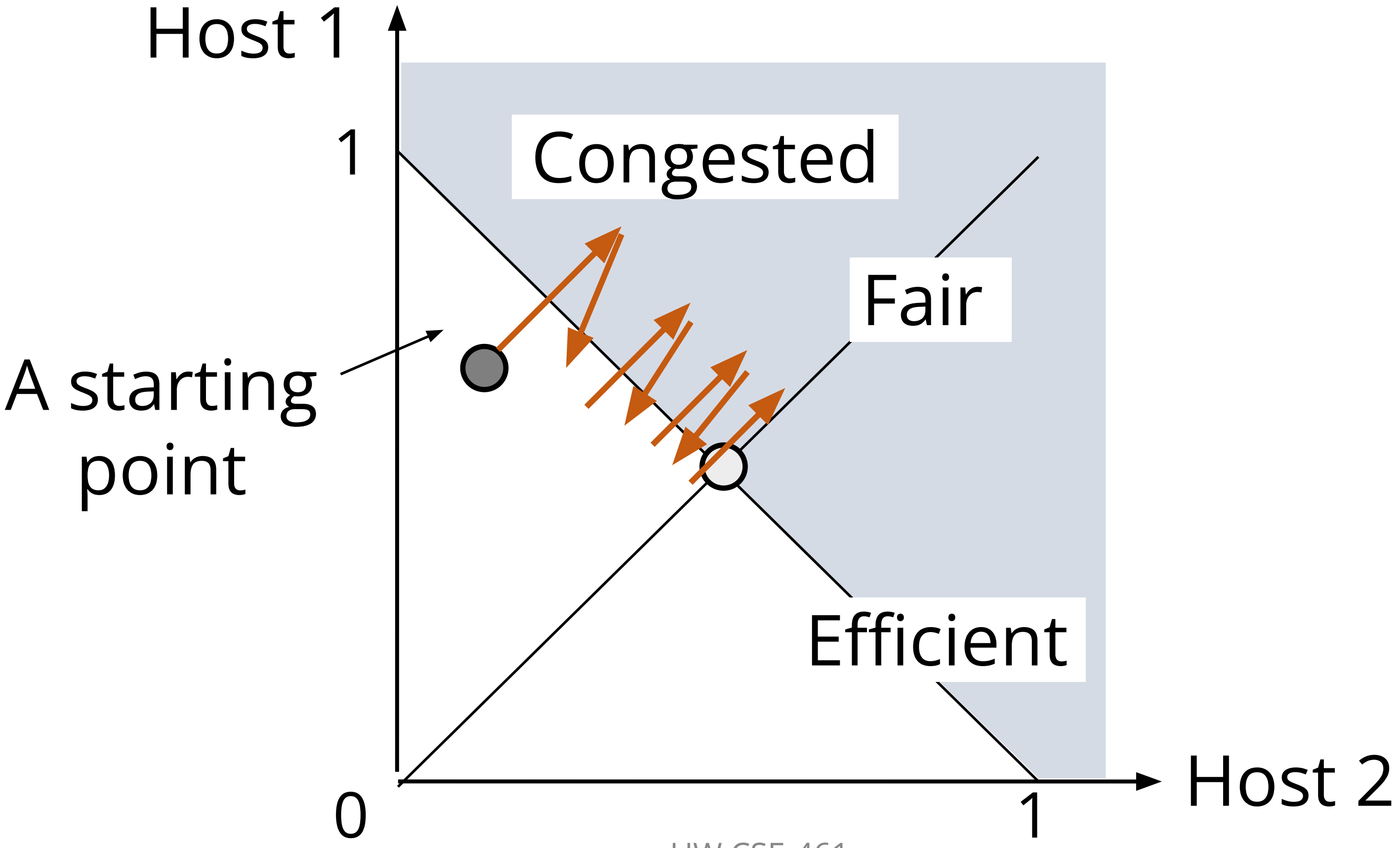
AIMD Game (4)

- Play the game!



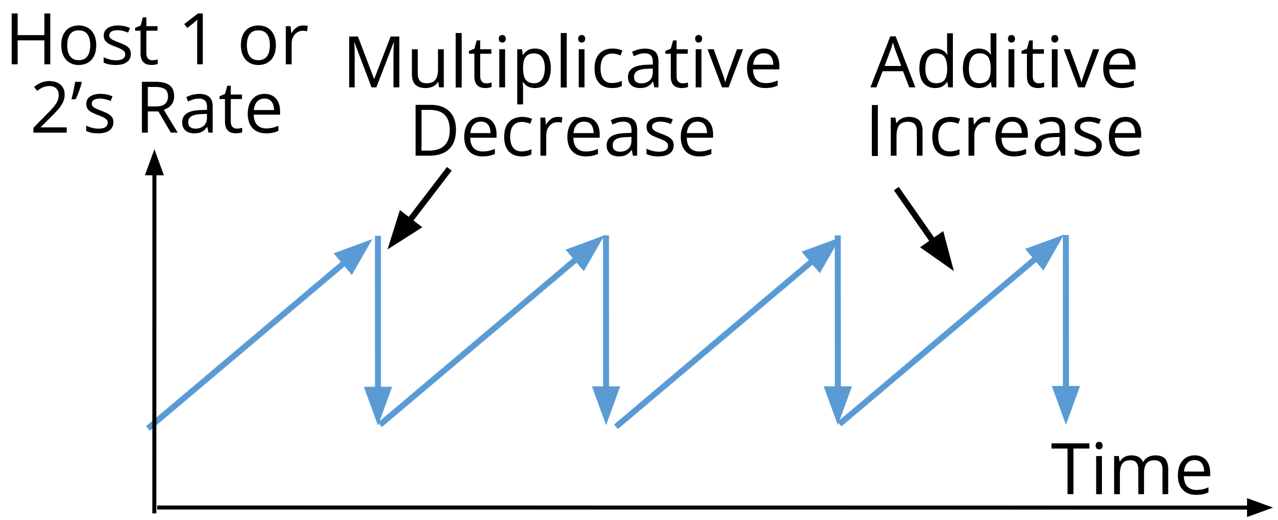
AIMD Game (5)

- Always converge to good allocation!



AIMD Sawtooth

- Produces a “sawtooth” pattern over time for rate of each host
  - This is the “TCP sawtooth”



AIMD Properties

- Converges to an allocation that is efficient and fair when hosts run it
  - Holds for more general topologies
- Other increase/decrease control laws do not!
  - (Try MIAD, MIMD, MIAD)
- Requires only binary feedback from the network

Feedback Signals

- Several possible signals, with different pros/cons

Activity:

1. In a group of 3, come up with two different possible feedback signals (possibly recall from the reading)
2. And what are their pros/cons

Feedback Signals

- Several possible signals, with different pros/cons

Signal	Example Protocol	Pros / Cons
Packet loss	TCP NewReno Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	TCP BBR (Youtube)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification DCTCP (Datacenters)	Hear about congestion early Require router support

Feedback Signals

- Several possible signals, with different pros/cons
  - We'll look at classic TCP that uses packet loss as a signal

Signal	**Example Protocol**	**Pros / Cons**
Packet loss	TCP NewReno Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	TCP BBR (Youtube)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification DCTCP (Datacenters)	Hear about congestion early Require router support

Slow Start
(TCP Additive Increase)

Practical AIMD

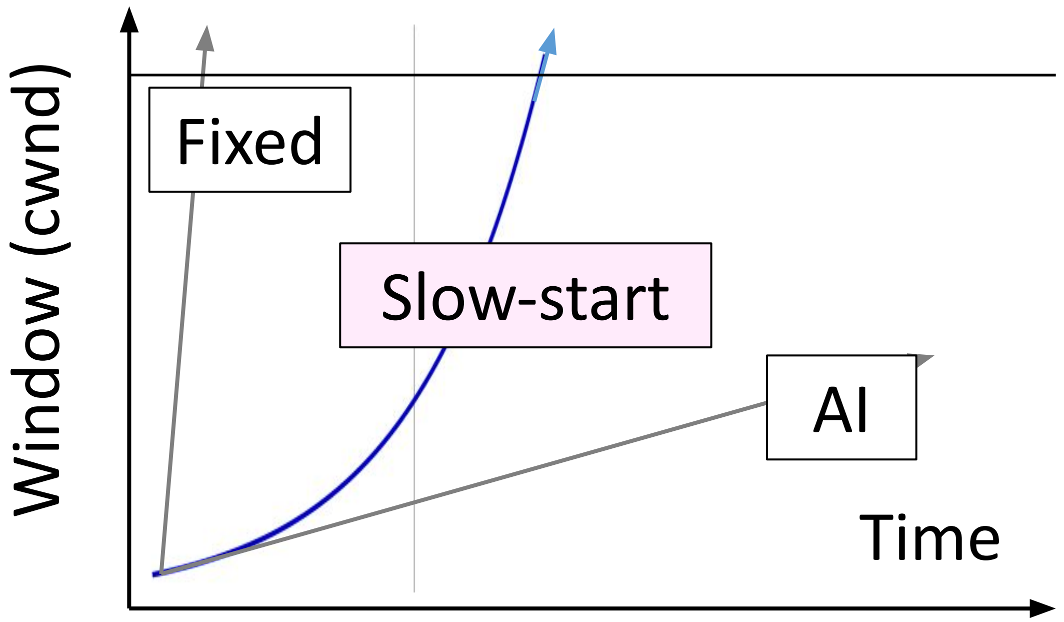
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a <u>congestion window</u> or <u>cwnd</u> to set its rate (≈cwnd/RTT)
- Sender uses loss as network congestion signal
- Need TCP to work across a very large range of rates and RTTs

TCP Startup Problem

- We want to quickly near the right rate, $cwnd_{IDEAL}$, but it varies greatly
  - Fixed sliding window doesn't adapt and is rough on the network (loss!)
  - Additive Increase with small bursts adapts cwnd gently to the network, but might take a long time to become efficient

Slow-Start Solution

- Start by doubling cwnd every RTT
  - Exponential growth (1, 2, 4, 8, 16, …)
  - Start slow, but quickly reach large values

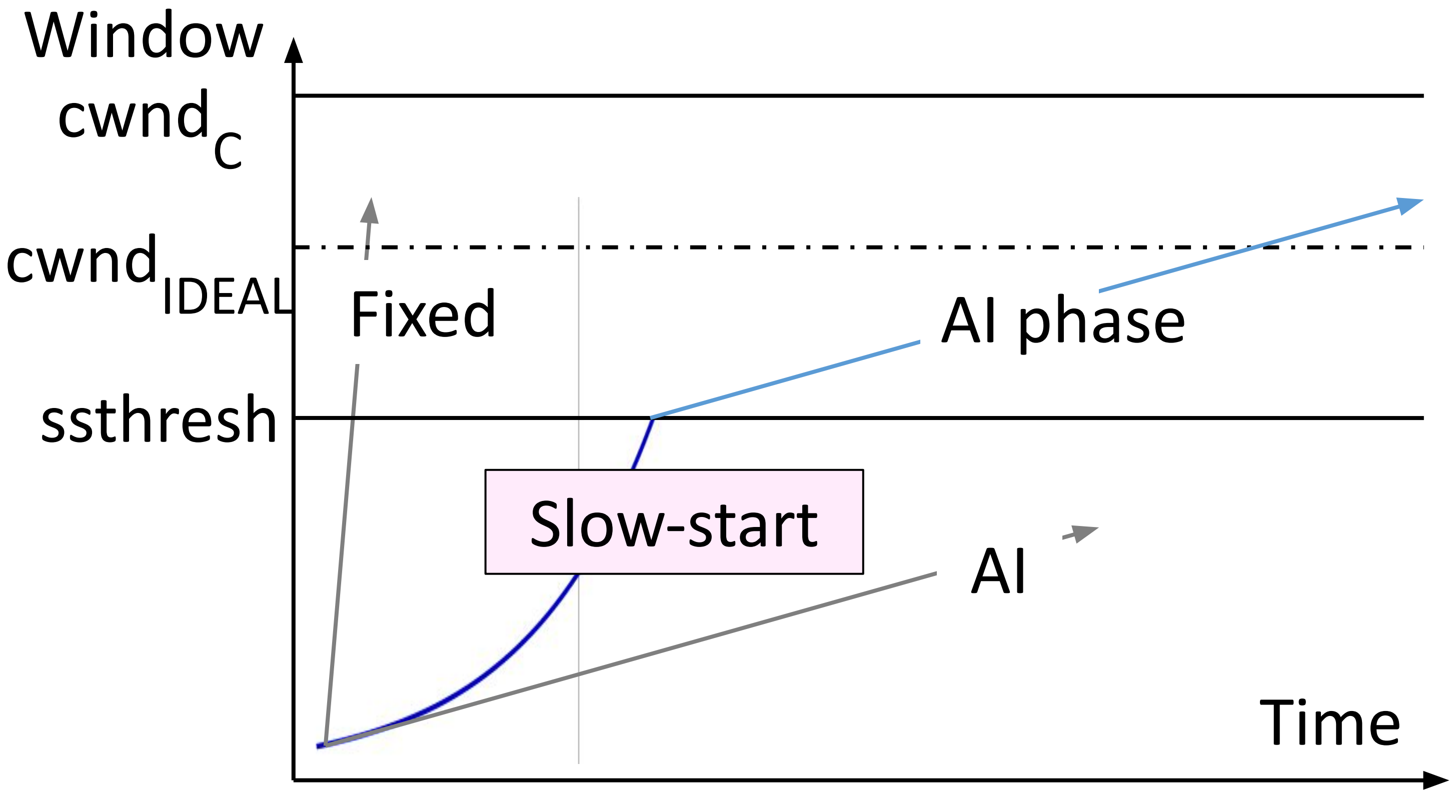


Slow-Start Solution (2)

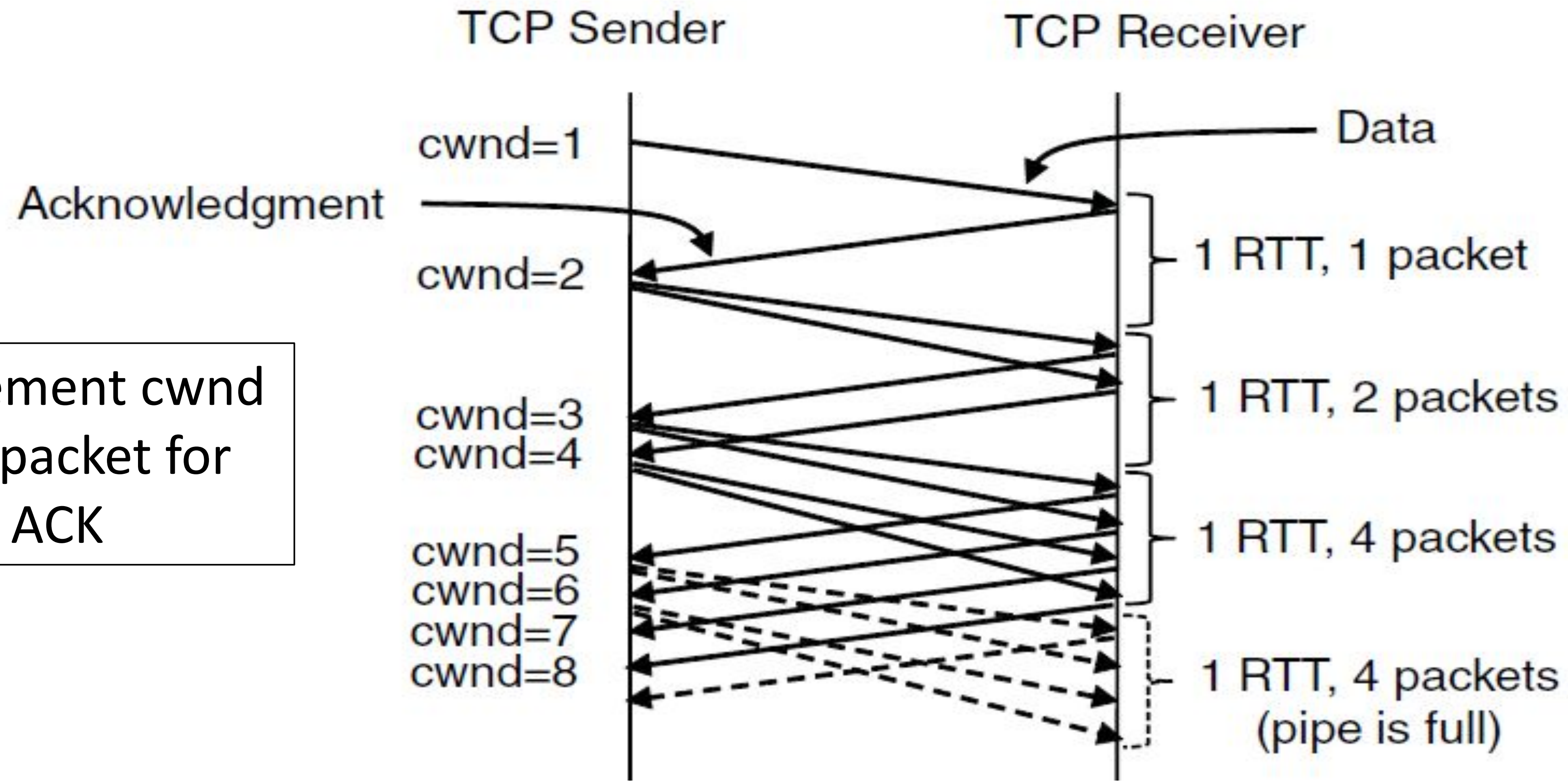
- Eventually packet loss will occur when the network is congested
  - Loss timeout tells us cwnd is too large
  - Next time, switch to AI beforehand
  - Slowly adapt cwnd near right value
- In terms of cwnd:
  - Expect loss for $cwnd_C \approx 2BD$+queue
  - Use ssthresh = $cwnd_C/2$ to switch to AI

Slow-Start Solution (3)

- Combined behavior, after first time
  - Most time spend near right value

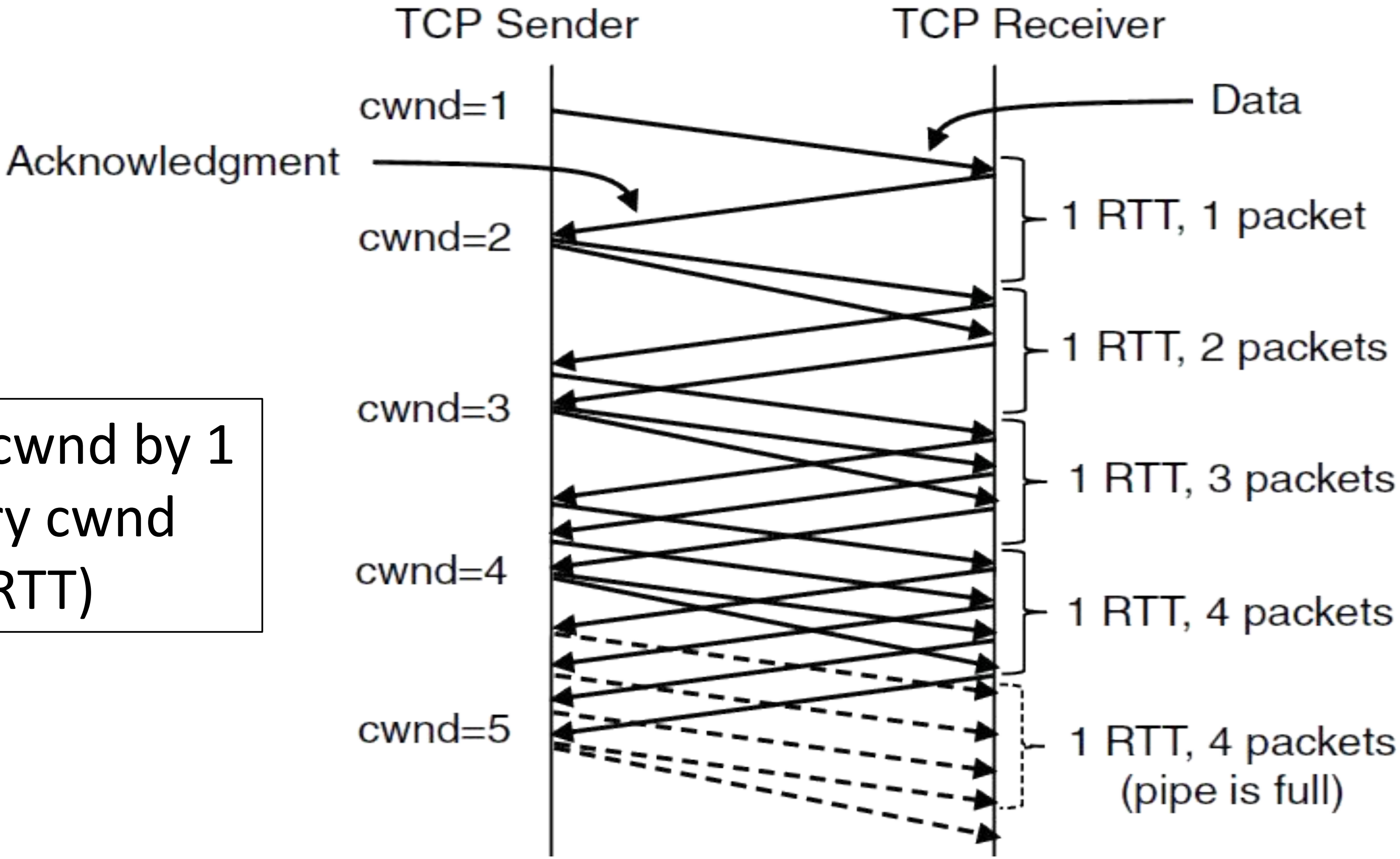


Slow-Start (Doubling) Timeline



Increment cwnd by 1 packet for each ACK

Additive Increase Timeline



Increment cwnd by 1 packet every cwnd ACKs (or 1 RTT)

TCP Tahoe (Implementation)

- Initial slow-start (doubling) phase
  - Start with cwnd = 1 (or small value)
  - cwnd += 1 packet per ack
- Later Additive Increase phase
  - cwnd += 1/cwnd packets per ack
  - Roughly adds 1 packet per RTT
- Switching threshold (initially infinity)
  - Switch to AI when cwnd > ssthresh
  - Set ssthresh = cwnd/2 after loss
  - Begin with slow-start after timeout

Timeout Misfortunes

- Why do a slow-start after timeout?
  - Instead of MD cwnd (for AIMD)
- Timeouts are sufficiently long that the ack clock will have run down 😩
  - Slow-start ramps up the ack clock
- *We need to detect loss before a timeout to get to full AIMD*
  - TCP Tahoe doesn’t

Fast Recovery

(Enabling TCP Multiplicative Decrease)

Practical AIMD (2)

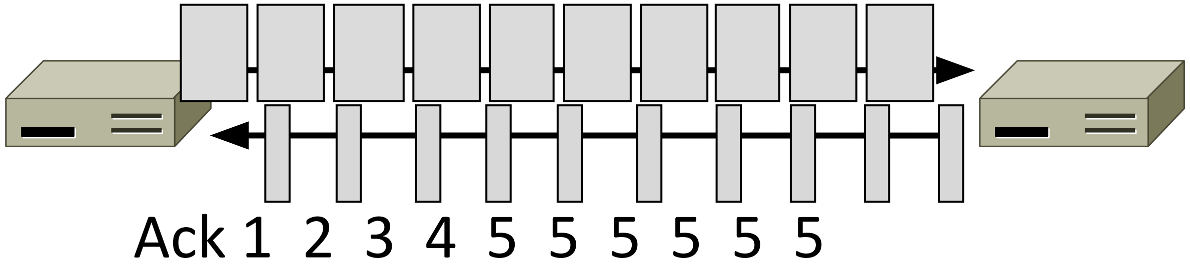
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a <u>congestion window</u> or <u>cwnd</u> to set its rate (≈cwnd/RTT)
- Sender uses slow-start to ramp up the ACK clock, followed by Additive Increase
- But after a timeout, sender slow-starts again with cwnd=1 (as if no ACK clock)

Inferring Loss from ACKs

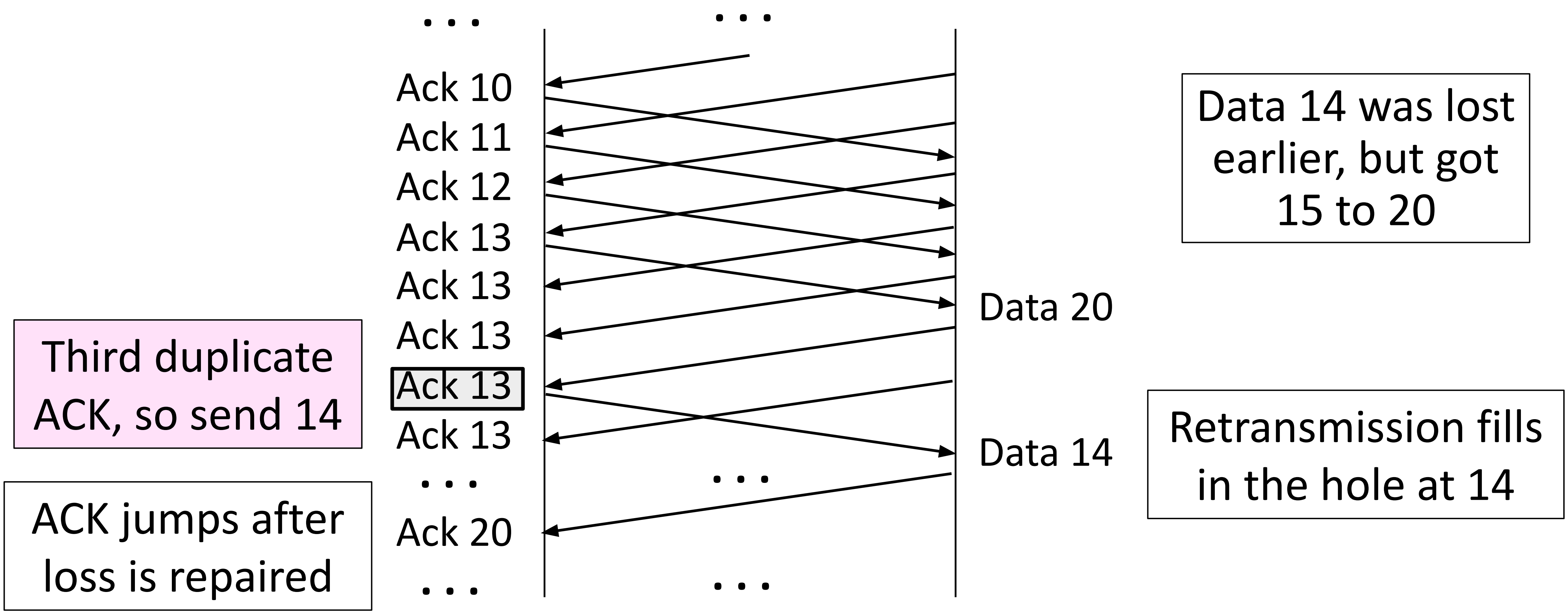
- TCP uses a cumulative ACK
  - Carries highest in-order seq. number
  - Normally a steady advance
- Duplicate ACKs give us hints about what data hasn't arrived
  - Tell us some new data did arrive, but it was not next segment
  - Thus the next segment may be lost

Fast Retransmit

- Treat three duplicate ACKs as a loss
  - Retransmit next expected segment
  - Some repetition allows for reordering, but still detects loss quickly



Fast Retransmit (2)



Fast Retransmit (3)

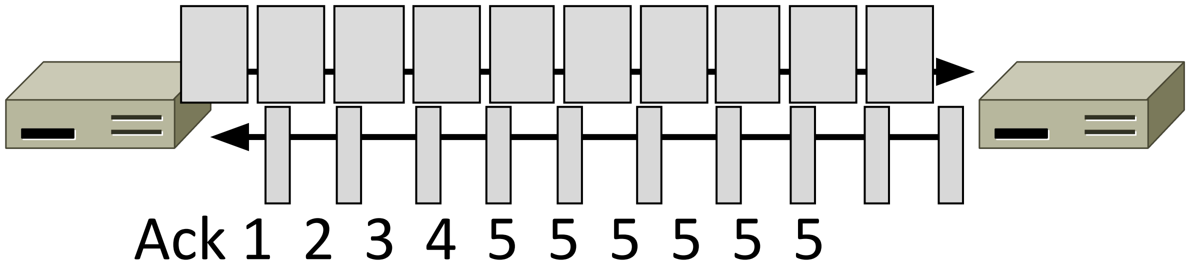
- It can repair single segment loss quickly, typically before a timeout
- However, we have quiet time at the sender/receiver while waiting for the ACK to jump
- And we still need to MD cwnd …

Inferring Non-Loss from ACKs

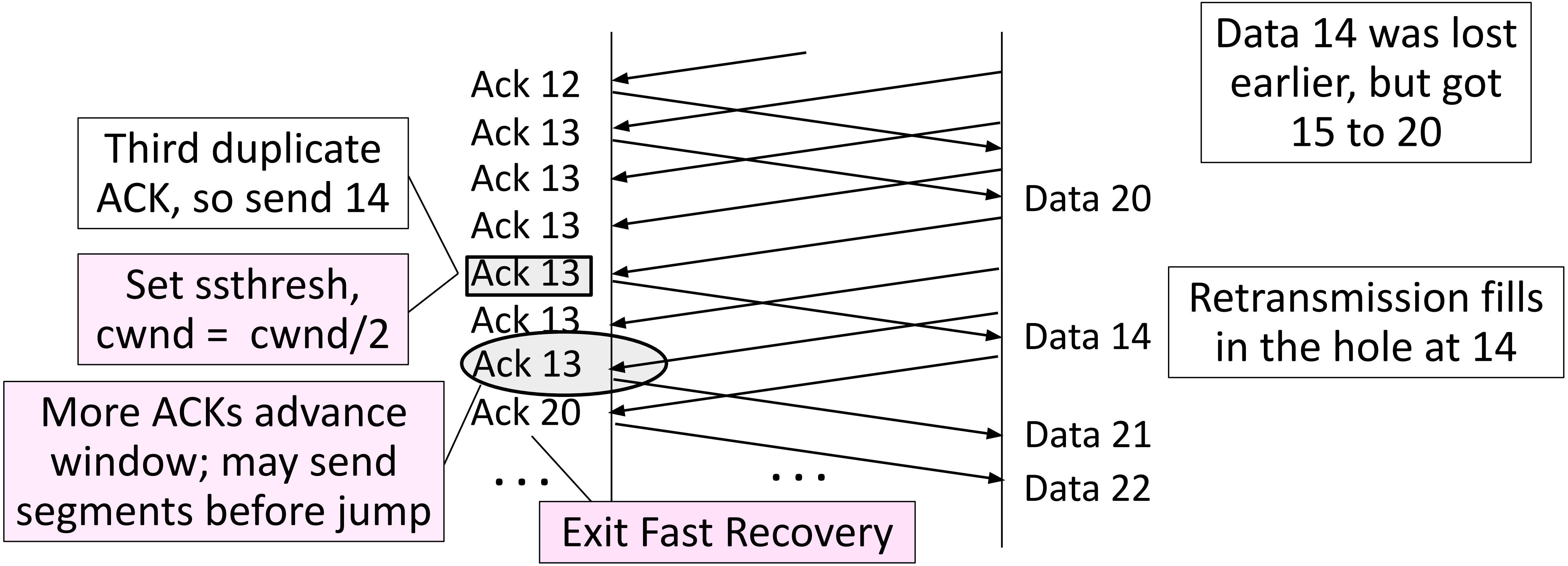
- Duplicate ACKs also give us hints about what data has arrived
  - Each new duplicate ACK means that some new segment has arrived
  - It will be the segments after the loss
  - Thus advancing the sliding window will not increase the number of segments stored in the network

Fast Recovery

- First fast retransmit, and MD cwnd
- Then pretend further duplicate ACKs are the expected ACKs
  - Lets new segments be sent for ACKs
  - Reconcile views when the ACK jumps



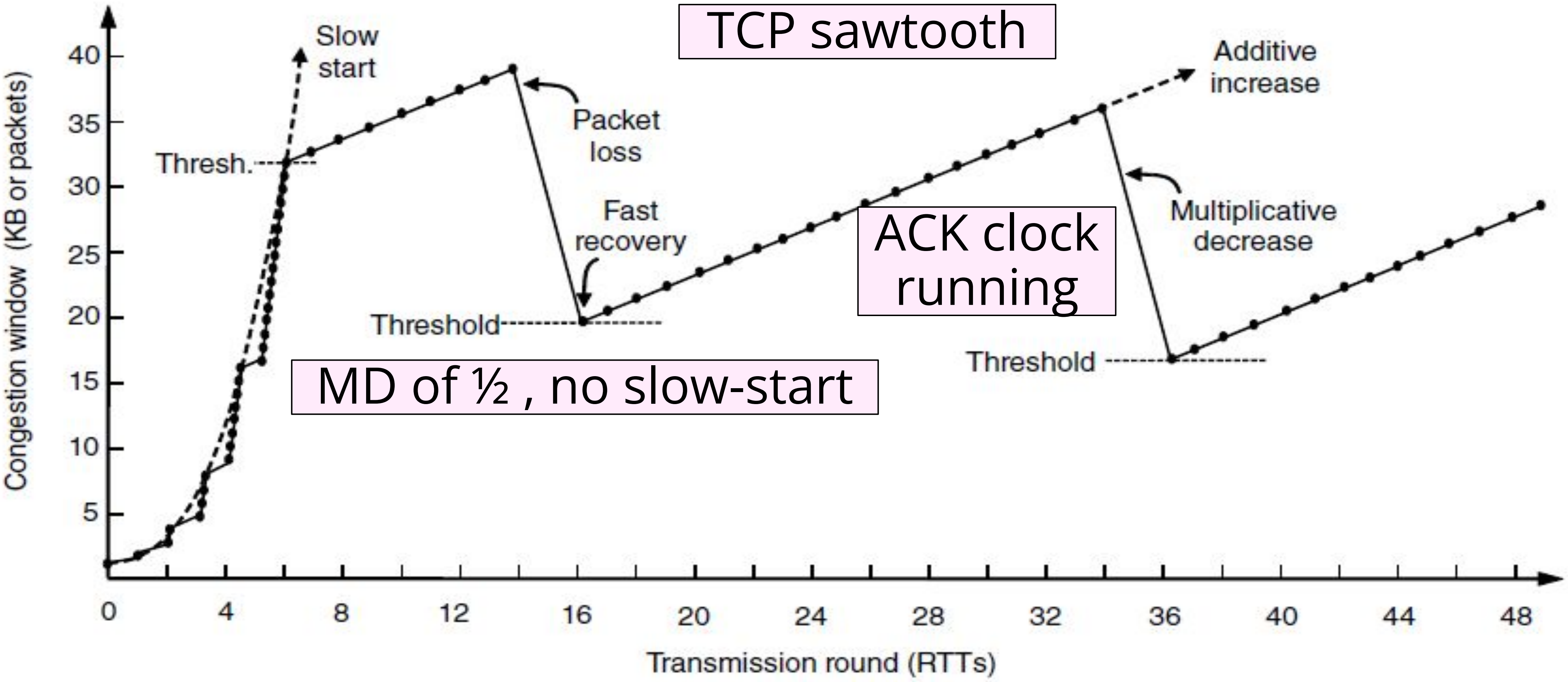
Fast Recovery (2)



Fast Recovery (3)

- With fast retransmit, it repairs a single segment loss quickly and keeps the ACK clock running
- This allows us to realize AIMD
  - No timeouts or slow-start after loss, just continue with a smaller cwnd
- TCP Reno combines slow-start, fast retransmit and fast recovery
  - Multiplicative Decrease is ½

TCP Reno



TCP sawtooth
ACK clock running
MD of ½ , no slow-start
Slow start
Additive increase
Packet loss
Fast recovery
Multiplicative decrease
Thresh.
Threshold
Threshold
Congestion window (KB or packets)
Transmission round (RTTs)
0
4
8
12
16
20
24
28
32
36
40
44
48
5
10
15
20
25
30
35
40

TCP Reno, NewReno, and SACK

- Reno can repair one loss per RTT
  - Multiple losses cause a timeout
- NewReno further refines ACK heuristics
  - Repairs multiple losses without timeout
- Selective ACK (SACK) is a better idea
  - Receiver sends ACK ranges so sender can retransmit without guesswork
  - Requires header extension, widely used in practice

TCP CUBIC

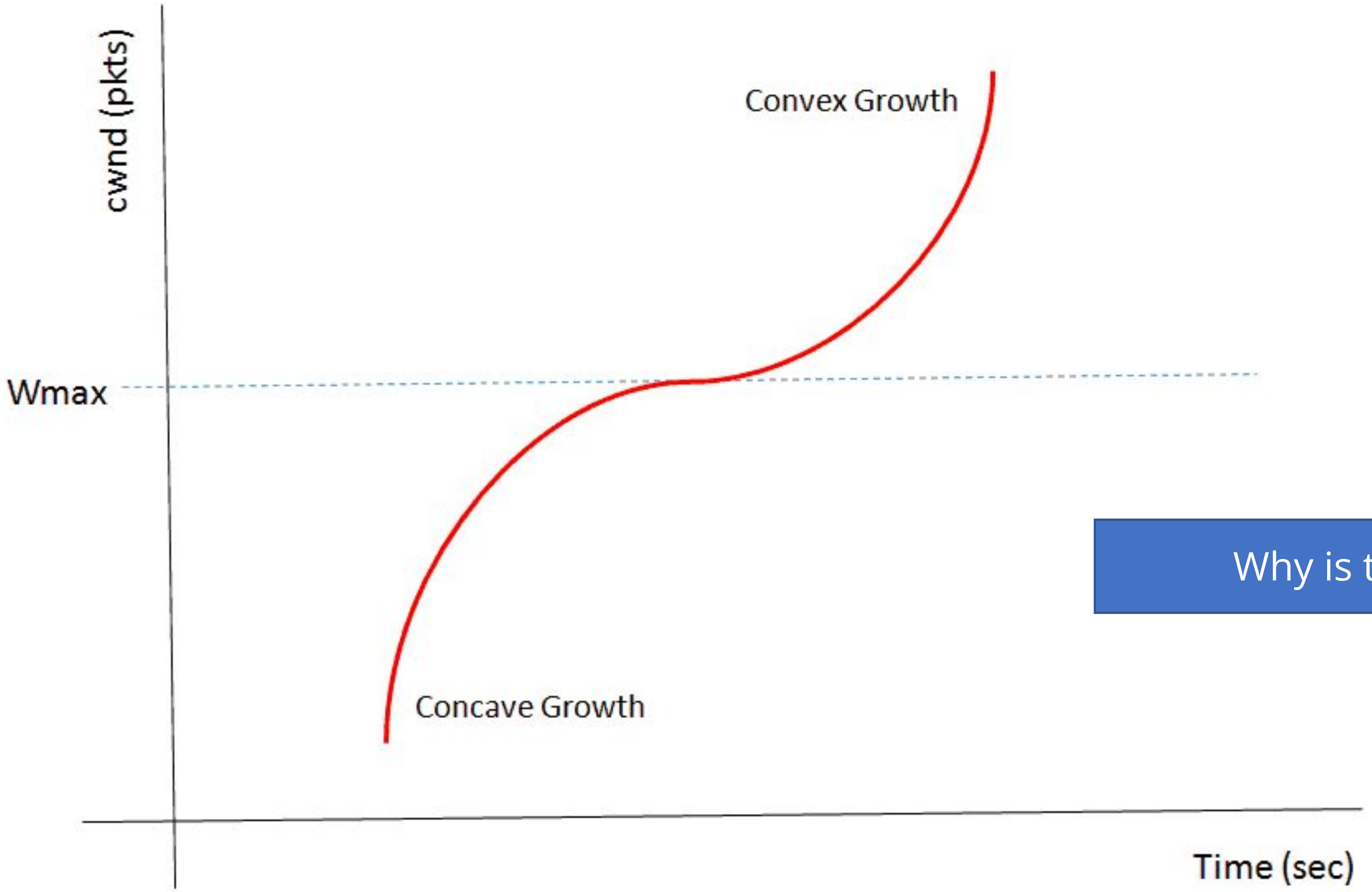
- Current standard TCP Stack
  - Linux (>= 2.6.19)
  - Windows (>= 10.1709)
  - MacOS (>= Yosemite)
- Internet grows to have more long-distance, high bandwidth connections
- Seeks to resolve two key problems with “standard” TCP:
  - Flows with lower RTT’s “grow” faster than those with higher RTTs
  - Flows grow too “slowly” (linearly) after congestion

TCP CUBIC

1. At the time of experiencing congestion event the window size for that instant will be recorded as W_{max} or the maximum window size.
2. The W_{max} value will be set as the inflection point of the cubic function that will govern the growth of the congestion window.
3. The transmission will then be restarted with a smaller window value (20%) and, if no congestion is experienced, this value will increase according to the concave portion of the cubic function (**not depending on received ACKs for cadence**).
4. As the window approaches W_{max} the increments will slow down.
5. Once the tipping point has been reached, i.e. W_{max}, the value of the window will continue to increase discreetly.
6. Finally, if the network is still not experiencing any congestion, the window size will continue to increase according to the convex portion of the function.

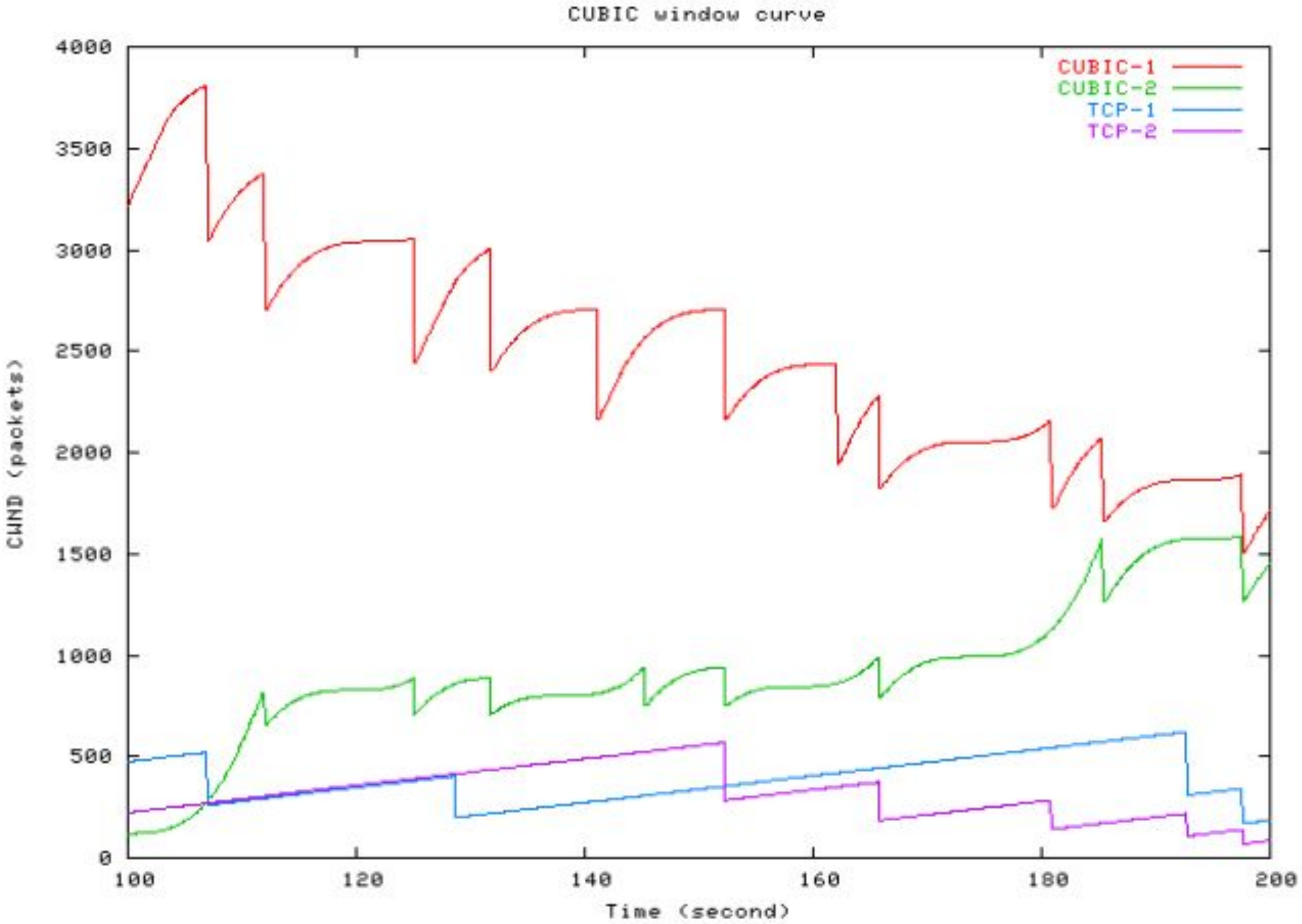
TCP CUBIC

Still keeps all the same tricks from NewReno about fast recovery and fast retransmit w/ SACK,
Replaces additive increase with a cubic function



Why is this helpful?

TCP CUBIC vs Everyone



The next generation? TCP BBR

- Bottleneck Bandwidth and Round-trip propagation time
- Developed at Google in 2016 primarily for YouTube traffic
- Attempting to solve “bufflerbloat” problem
- “Model-based” (Vegas) rather than “Loss-based” (CUBIC)
    - Measure RTT, latency, bottleneck bandwidth
    - Use this to predict window size

Bufferbloat

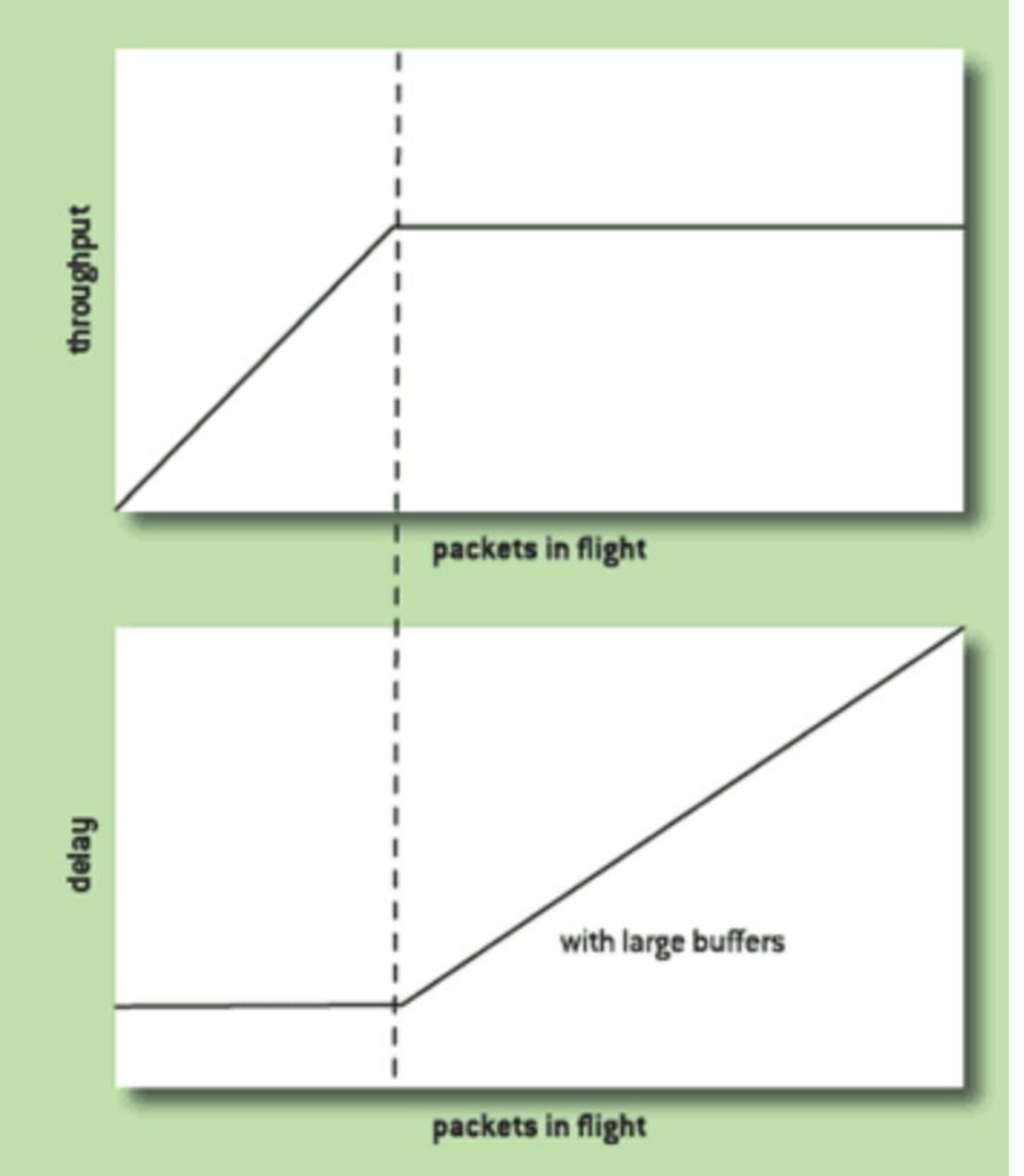
- Larger queues are better than smaller queues right?

When might this not be the case?

You already know the answer if you went to section!)

Bufferbloat

- Given TCP loss semantics…
- Performance can decrease as buffer size is increased
- Consider a mostly full buffer:
  - New packets arrive and have to wait
  - Then are transmitted to next mostly full buffer
  - *No drops* but performance (in terms of latency!) degrades

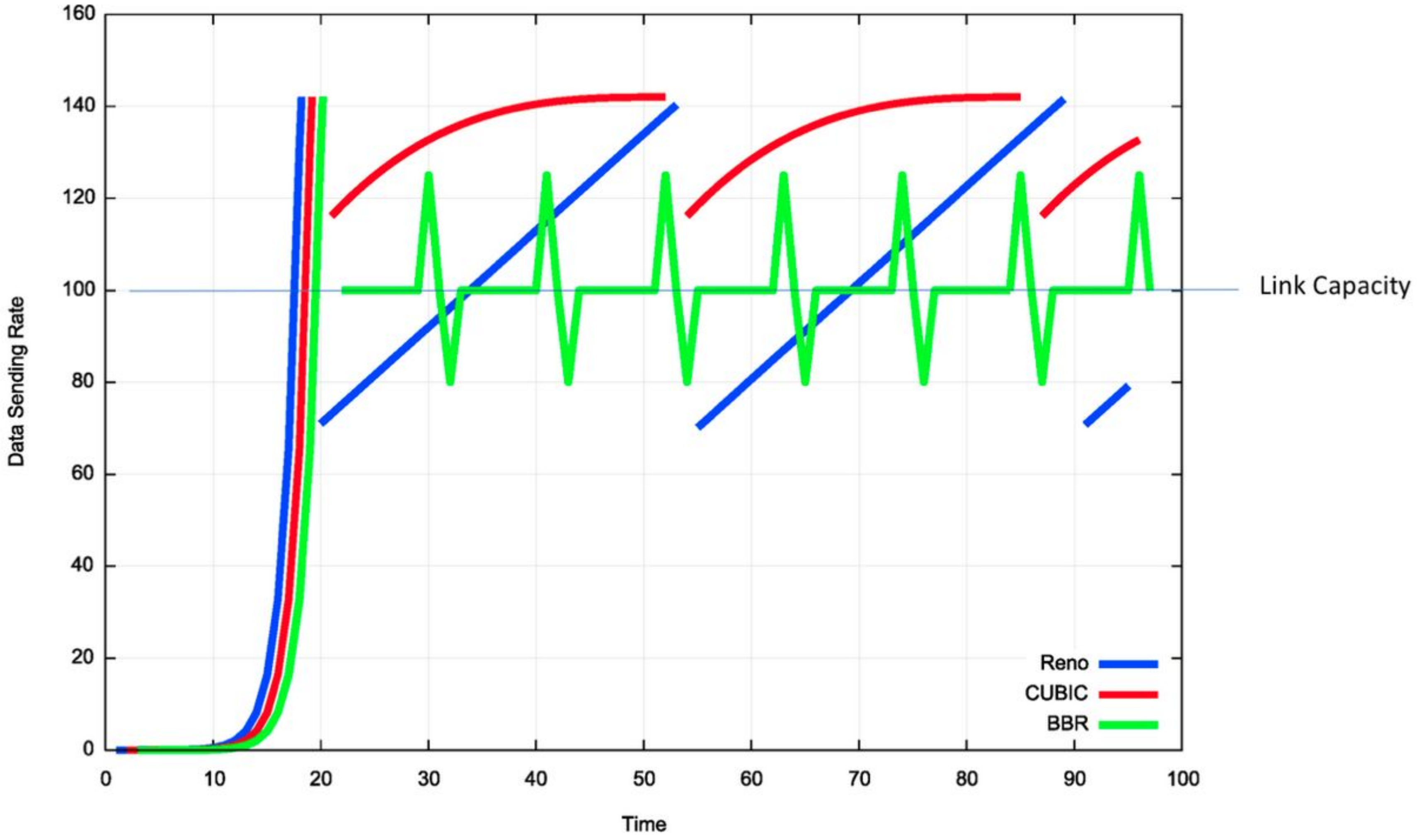


TCP BBR

BBR Has 4 Distinct Phases:

1. Startup: Basically identical to Cubic. Exponentially grow until RTTs start to increase (instead of dropped packet). Set cwnd.
2. Drain: Startup filled a queue. Temporarily reduce sending rate (known as “pacing gain”)
3. Probe Bandwidth: Increase sending rate to see if there’s more capacity. If not, drain again.
4. Probe RTT: Reduce rate dramatically (4 packets) to measure RTT. Use this as our baseline for above.

TCP BBR vs Everyone



Network-Side Congestion Control

Congestion Avoidance vs. Control

- Classic TCP drives the network into congestion and then recovers
  - Needs to see loss to slow down
- Would be better to use the network but avoid congestion altogether!
  - Reduces loss and delay
- But how can we do this?

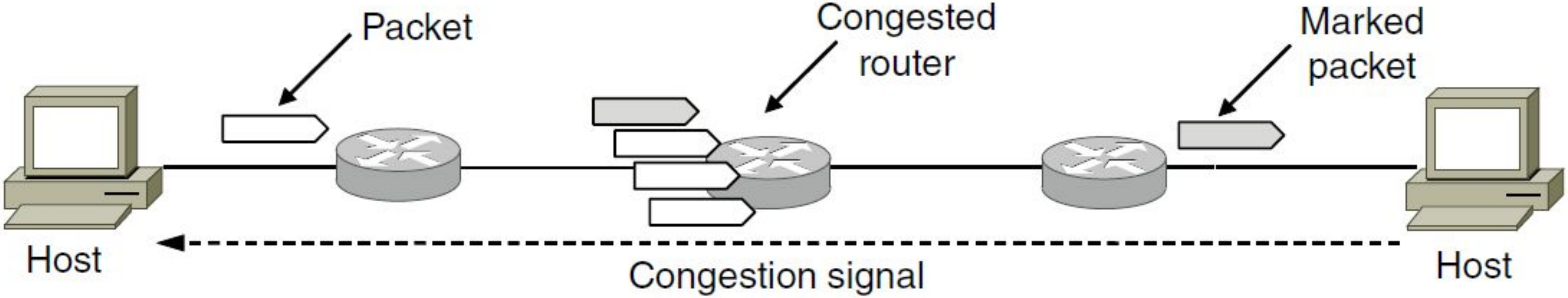
Feedback Signals

Delay and router signals can let us avoid congestion

Signal	Example Protocol	Pros / Cons
Packet loss	Classic TCP Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	TCP BBR (Youtube)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification	Hear about congestion early Require router support

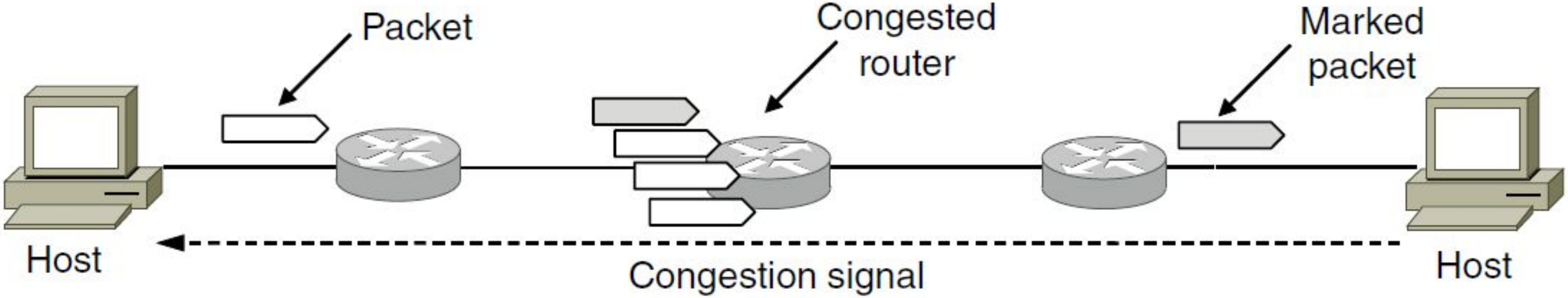
ECN (Explicit Congestion Notification)

- Router detects the onset of congestion via its queue
  - When congested, it <u>marks</u> affected packets (IP header)



ECN (2)

- Marked packets arrive at receiver; treated as loss
  - TCP receiver reliably informs TCP sender of the congestion



ECN (3)

In hindsight, ECN is a much better approach than using loss…

Loss-based signaling causes active harm to the flow in the process of notifying about congestion : (

- Advantages:
  - Routers deliver clear signal to hosts
  - Congestion is detected early, (no loss🎉🤩💯)
  - No extra packets need to be sent
- Disadvantages:
  - Routers and hosts must be upgraded
  - More work at router
    - With IPv4 even have to recompute that pesky checksum : (

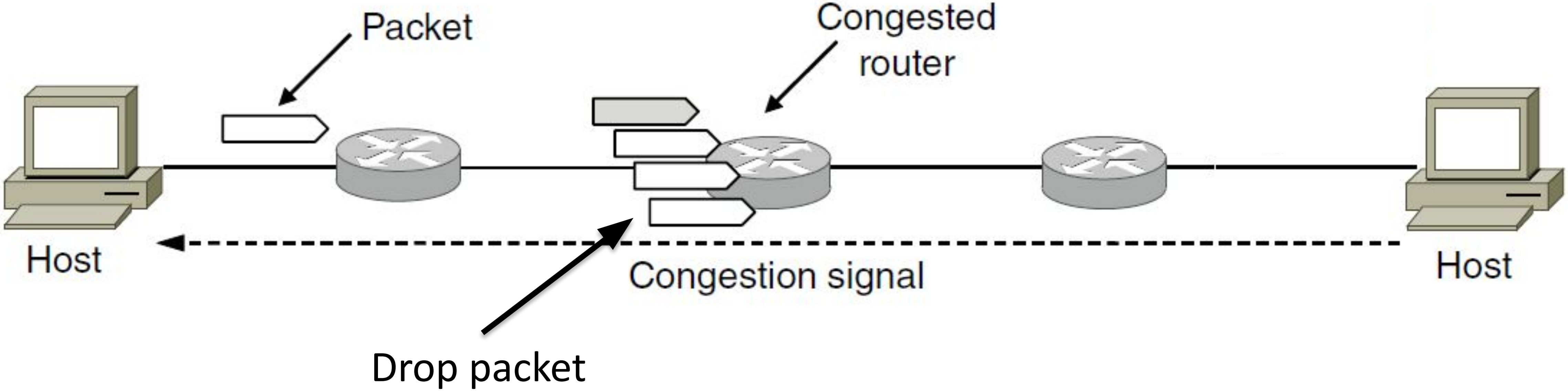
An interim solution, better than nothing until ECN is widespread

Random Early Detection (RED)

- Jacobson (again!) and Floyd
- Alternative idea: instead of marking packets, drop
  - We know they're using TCP, make use of that fact
- Signals congestion to sender
  - But without adding headers or doing packet inspection
- Drop at random, depending on queue size
  - If queue empty, accept packet always
  - If queue full, always drop
  - As queue approaches full, increase likelihood of packet drop
    - Example: 1 queue slot left, 10 packets expected, 90% chance of drop

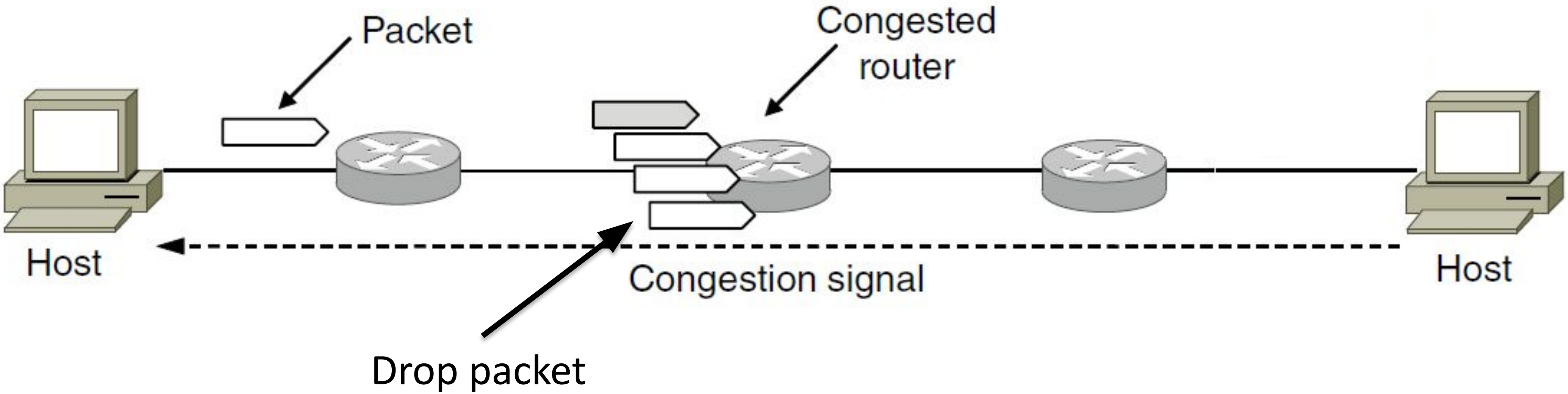
RED (Random Early Detection)

- Router detects the onset of congestion via its queue
  - Prior to congestion, drop a packet to signal
  - Lightweight: no per-flow state, no header modification



RED (Random Early Detection)

- Sender enters MD (multiplicative decrease), slows packet flow
  - We shed load, everyone is happy



Final thoughts on congestion control

- End-host approaches by their nature are *cooperative,* and can be abused by malicious hosts
- What would you do if you were an internet service provider?

In practice drives lots of complexity in real-world access networks!

In cellular the network sets per-user rate limits + sharing priorities

Fiber + Cable networks add user ratelimiting too, inline hardware to detect and police “nonresponsive” flows that don’t cooperate