TCP recap

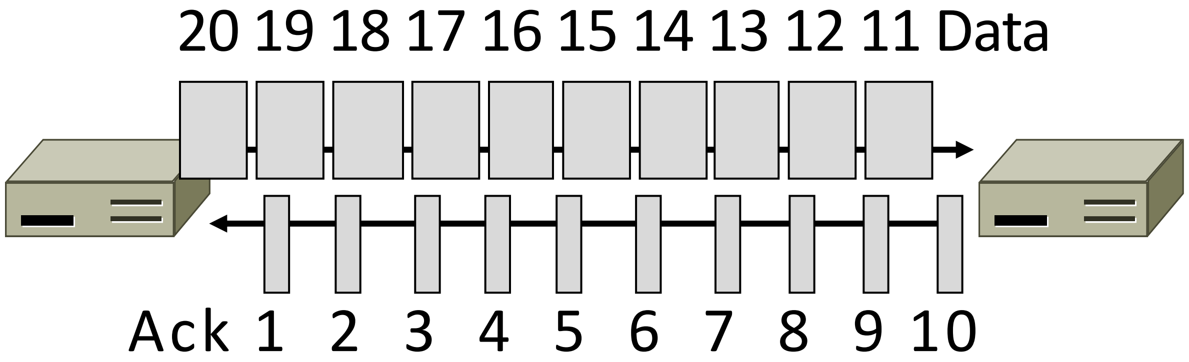
Three phases

1. Connection setup
2. Data transfer
    - Flow control – don’t overwhelm the receiver
        - ARQ – one outstanding packet
        - Go-back-N, selective repeat -- sliding window of W packets
        - **Tuning flow control (ack clocking, RTT estimation)**
    - **Congestion control**
3. Connection release

ACK Clocking

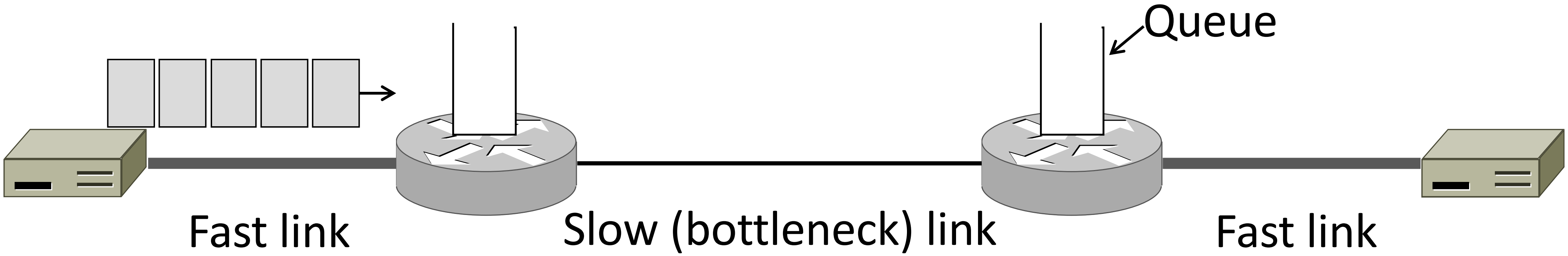
Sliding Window ACK Clock

- Typically, the sender does not know B or D
- Each new ACK advances the sliding window and lets a new segment enter the network
  - ACKs "clock" data segments



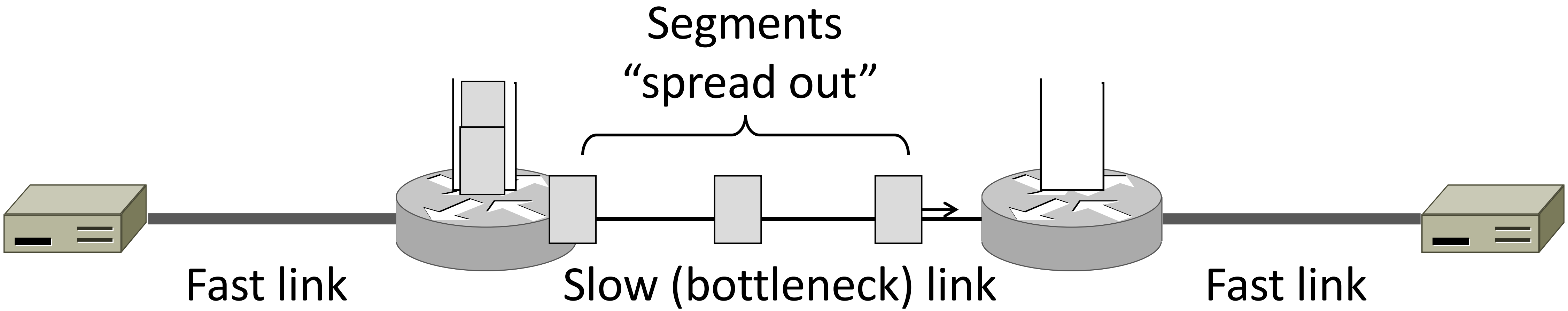
Benefit of ACK Clocking

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



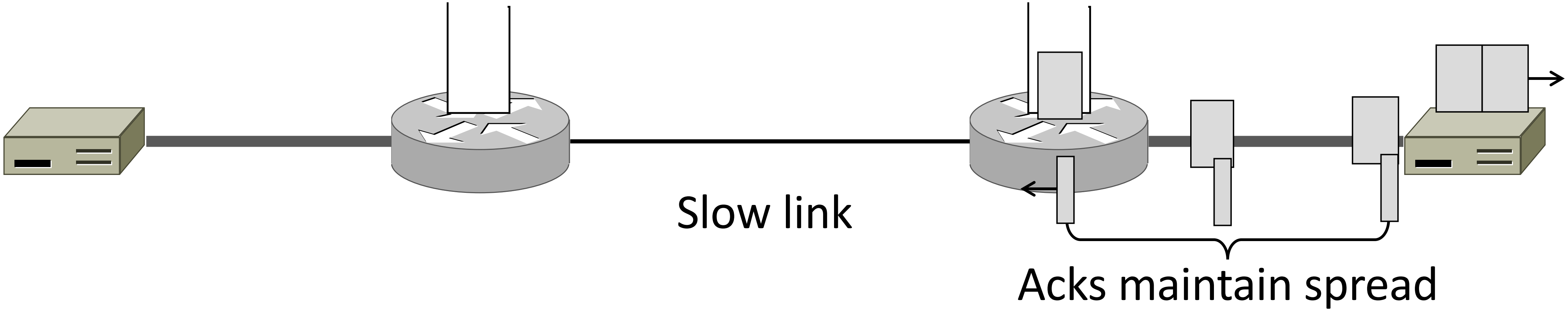
Benefit of ACK Clocking (2)

- Segments are buffered and spread out on slow link



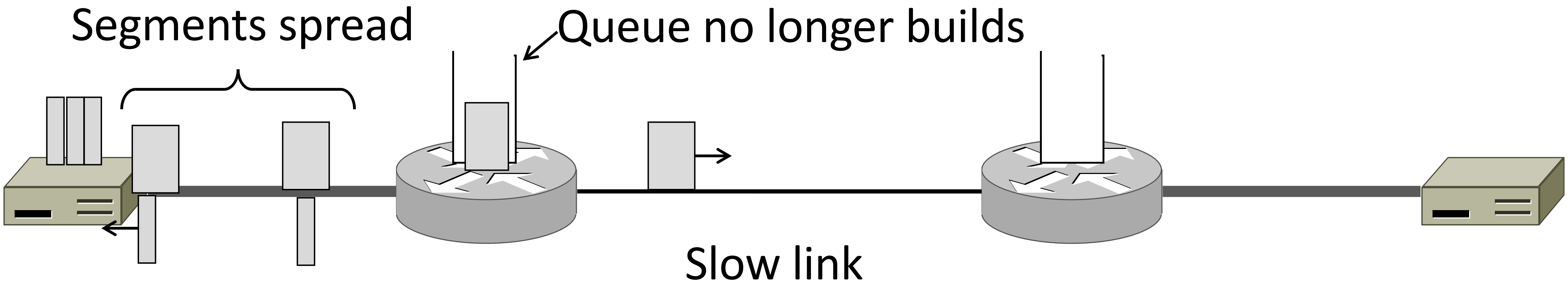
Benefit of ACK Clocking (3)

- 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 (4)

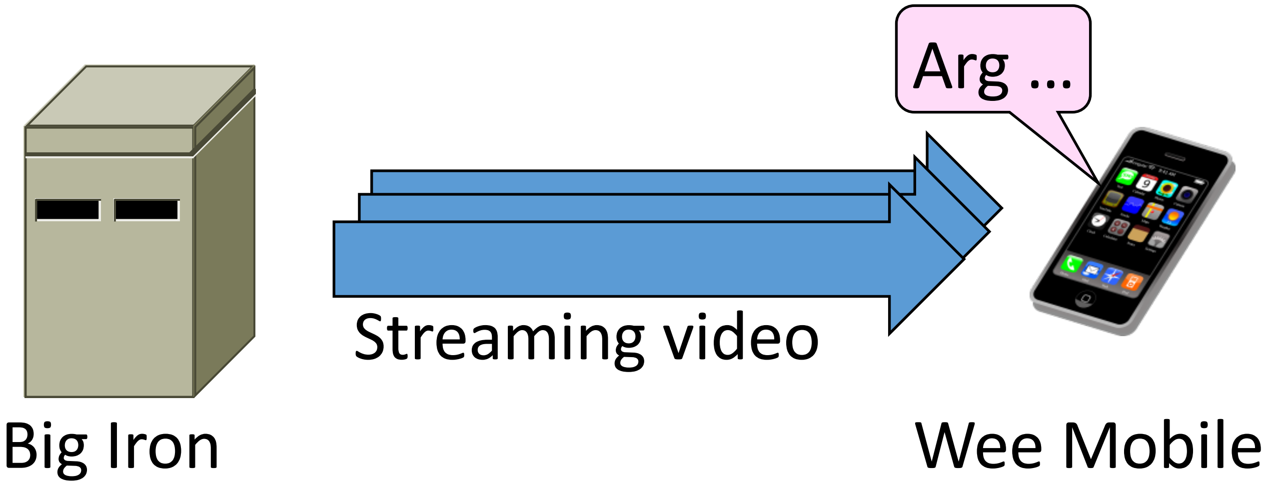
- Helps run with low levels of loss and delay!
- The network smooths out the burst of data segments
- ACK clock transfers this smooth timing back to sender
- Subsequent data segments are not sent in bursts so do not queue up in the network

TCP Uses ACK Clocking

- 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

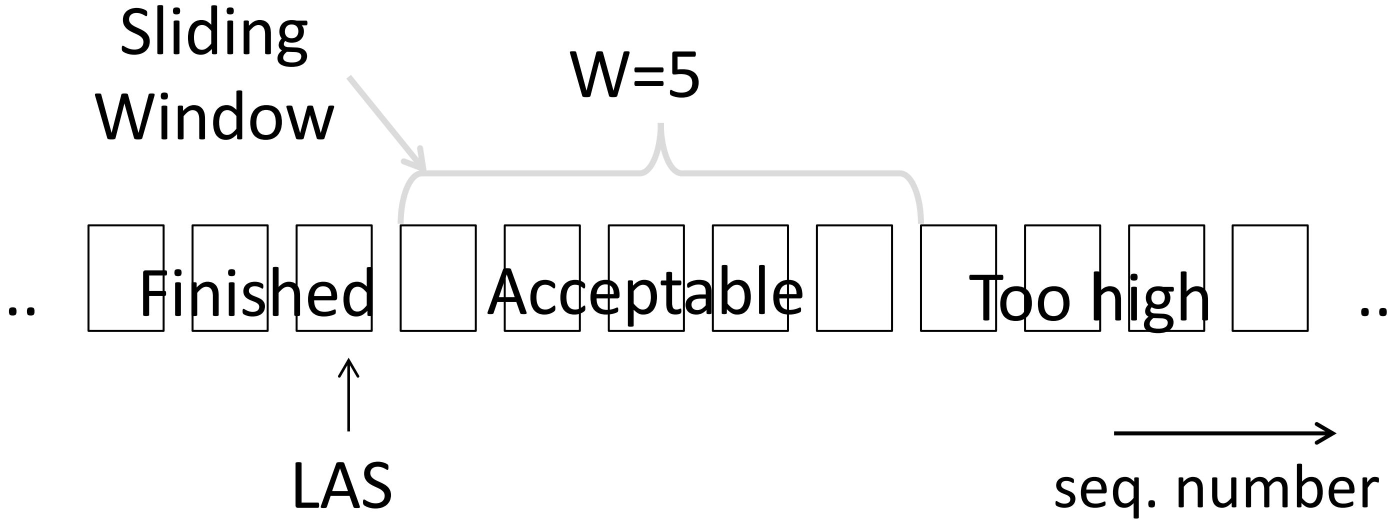
Problem

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



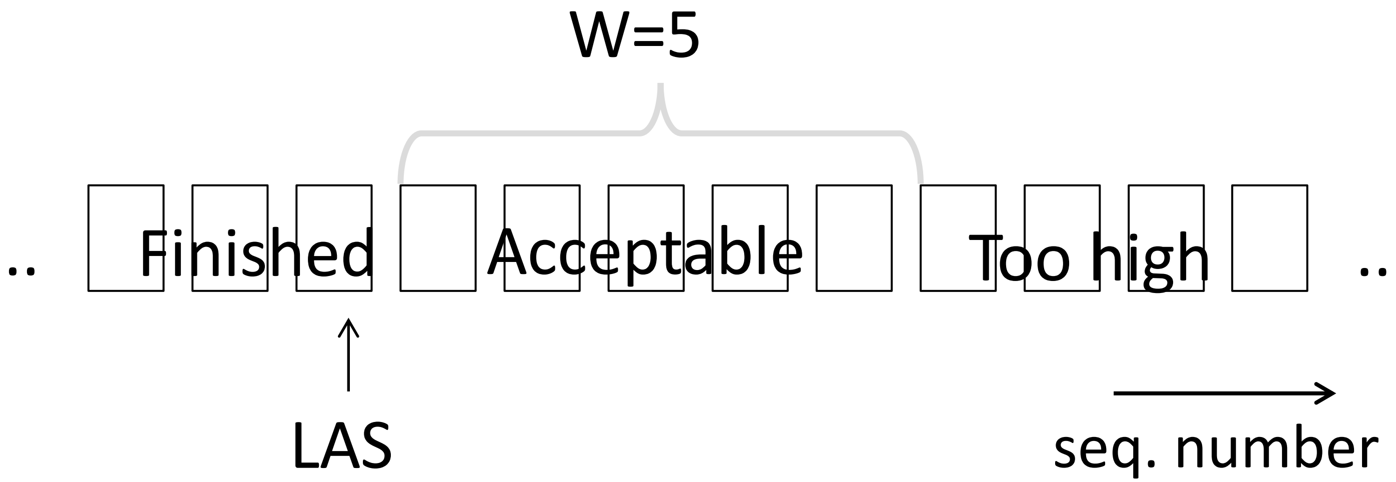
Receiver Sliding Window

- Consider receiver with W buffers
  - LAS=LAST ACK SENT
  - app pulls in-order data from buffer with recv() call



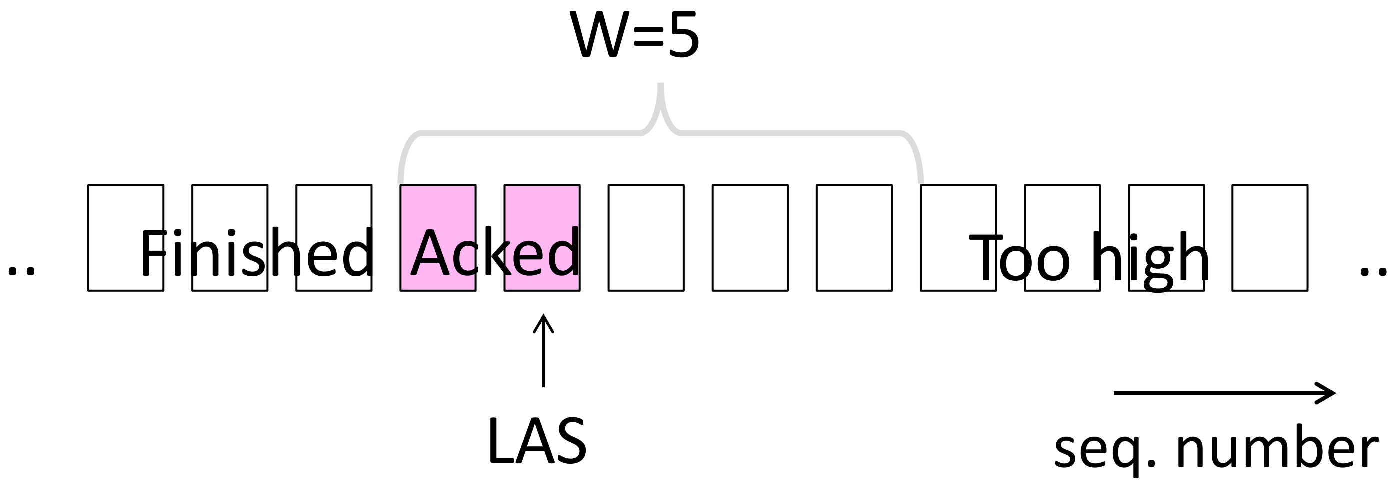
Receiver Sliding Window (2)

- Suppose the next two segments arrive but app does not call recv()



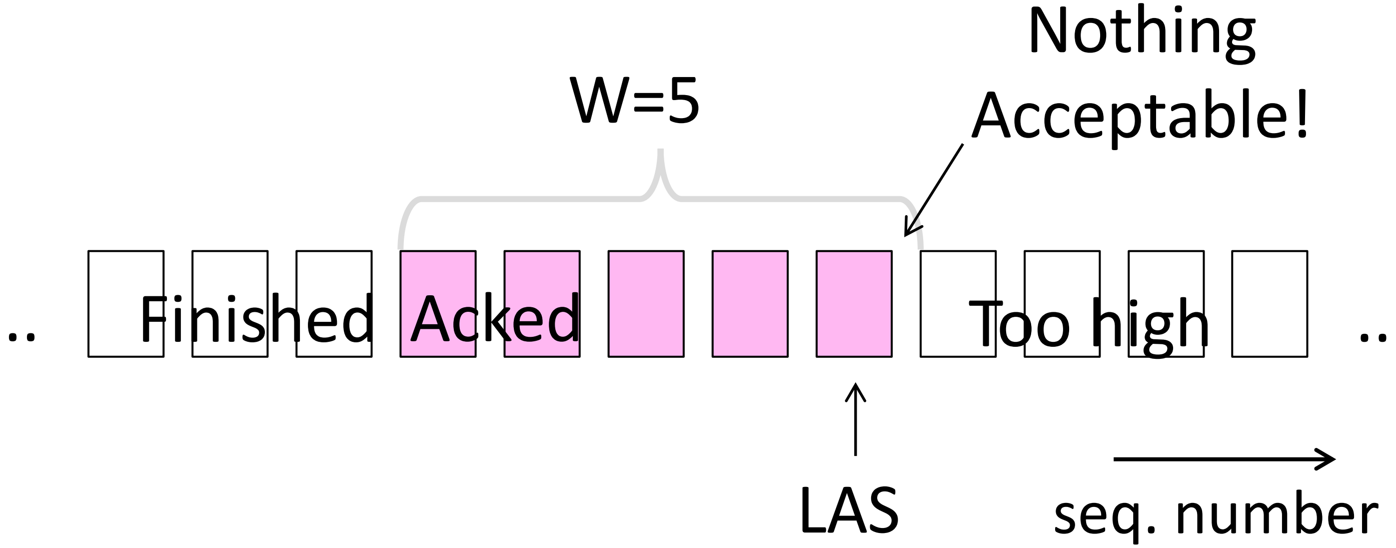
Receiver Sliding Window (3)

- Suppose the next two segments arrive but app does not call recv()
  - LAS rises, but we can't slide window!



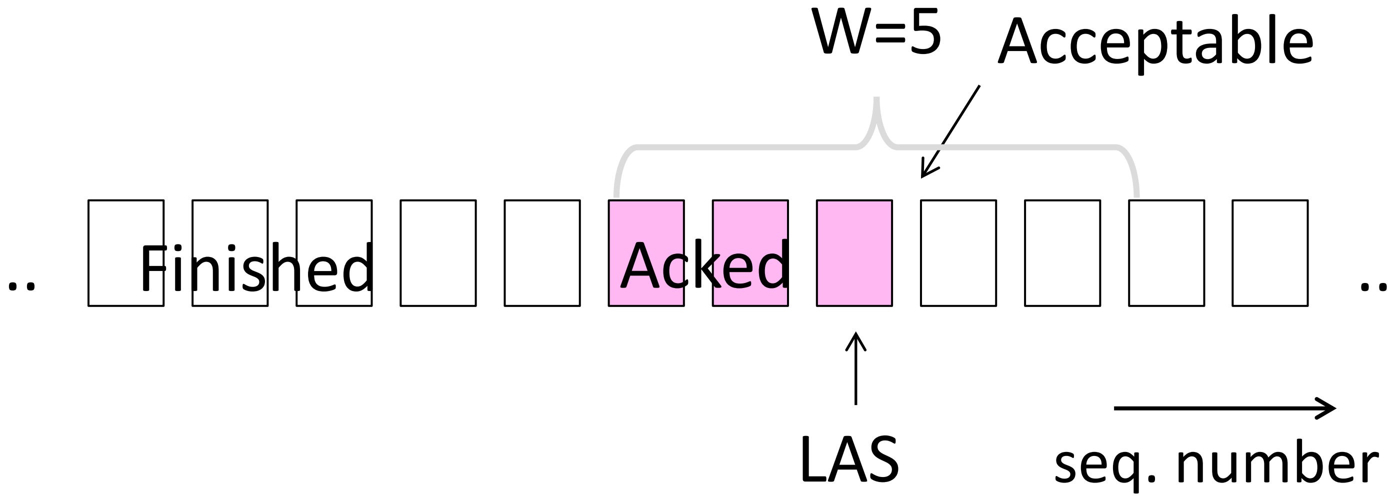
Receiver Sliding Window (4)

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



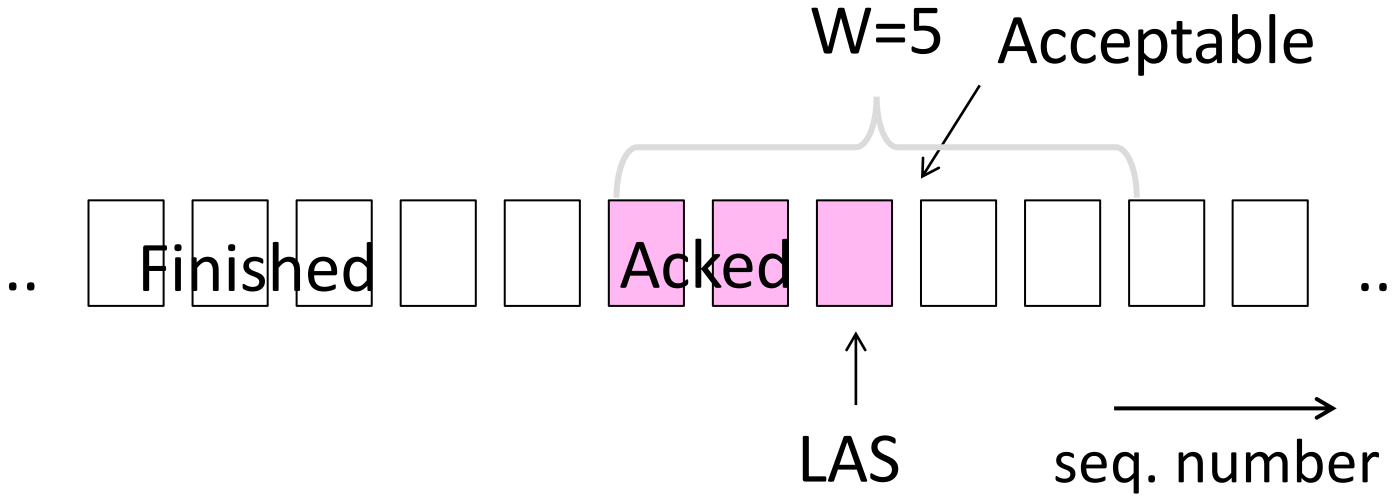
Receiver Sliding Window (5)

- App recv() takes two segments
  - Window slides (phew)



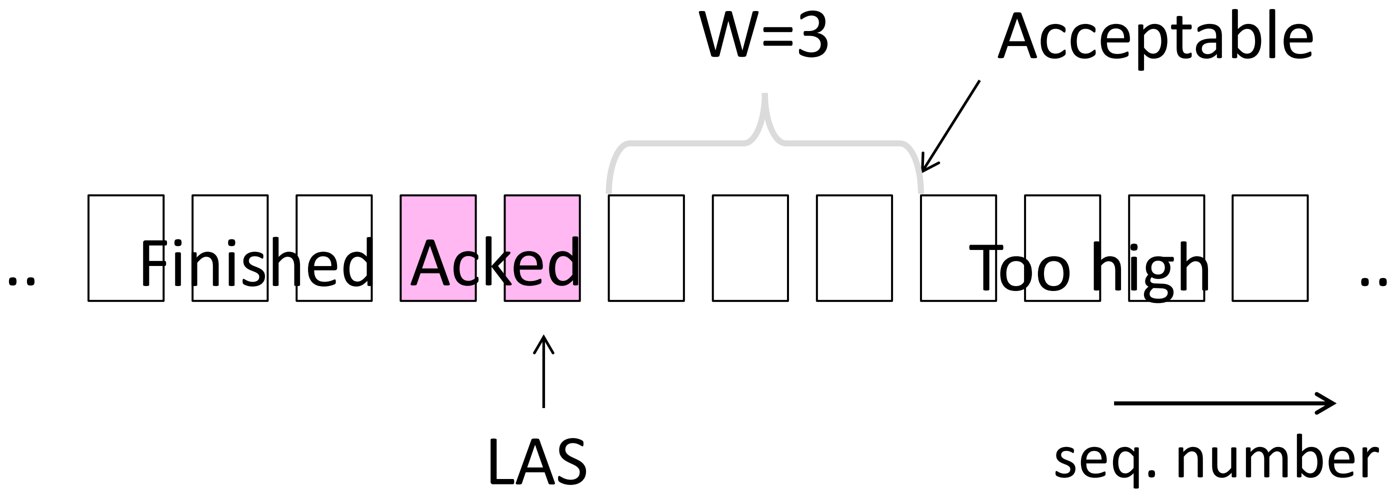
Flow Control

- Avoid loss at receiver by telling sender the available buffer space
    - WIN=#Acceptable, not W (from LAS)



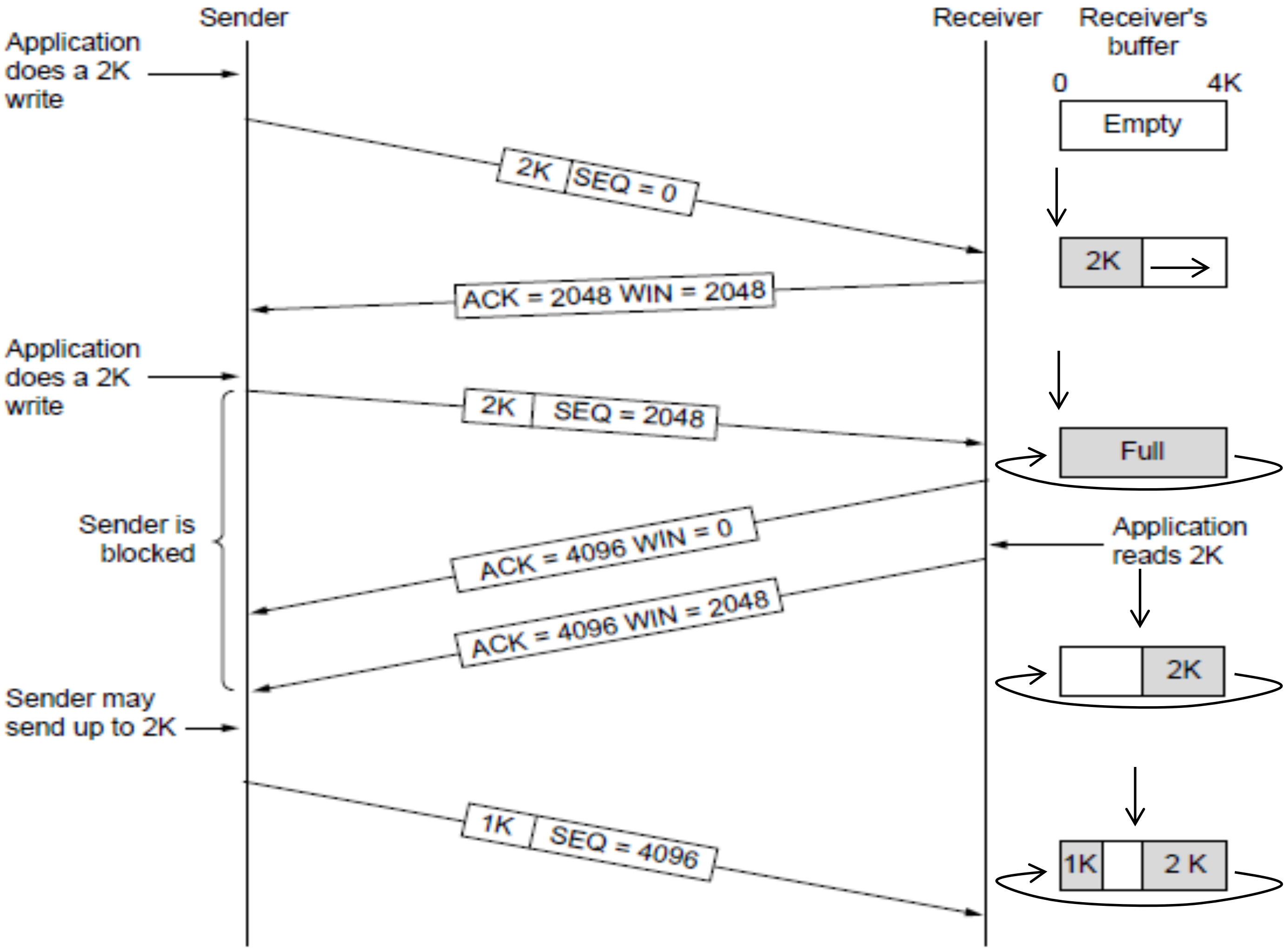
Flow Control (2)

- Sender uses lower of the sliding window and flow control window (WIN) as the effective window size



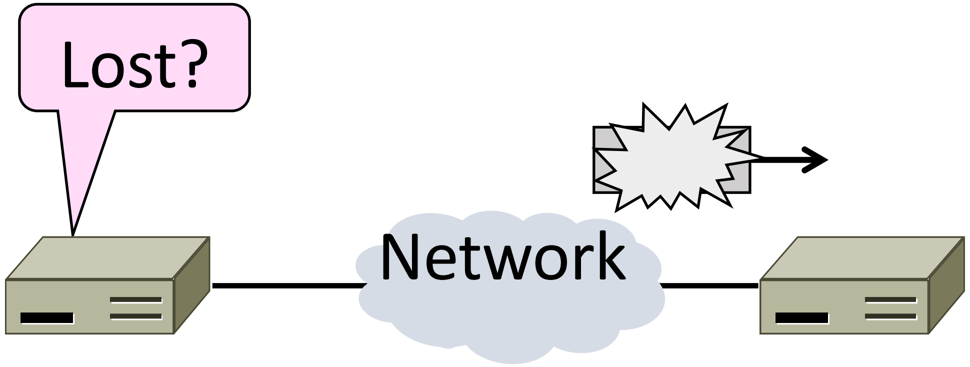
Flow Control (3)

- TCP-style example
  - SEQ/ACK sliding window
  - Flow control with WIN
  - SEQ + length < ACK+WIN
  - 4KB buffer at receiver
  - Circular buffer of bytes



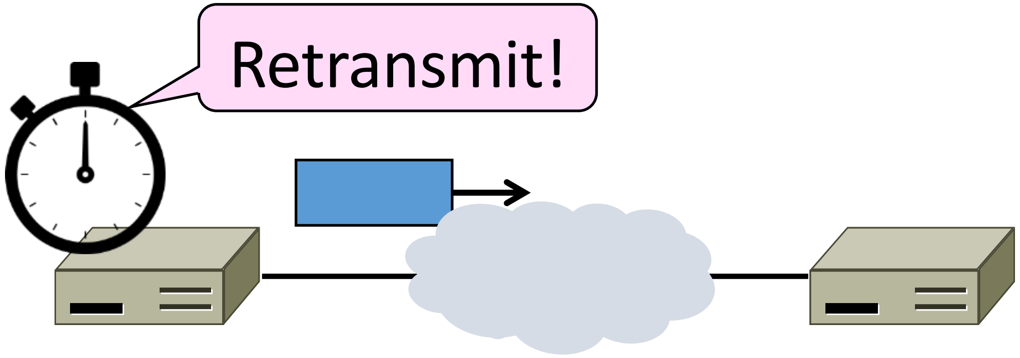
Topic

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



Retransmissions

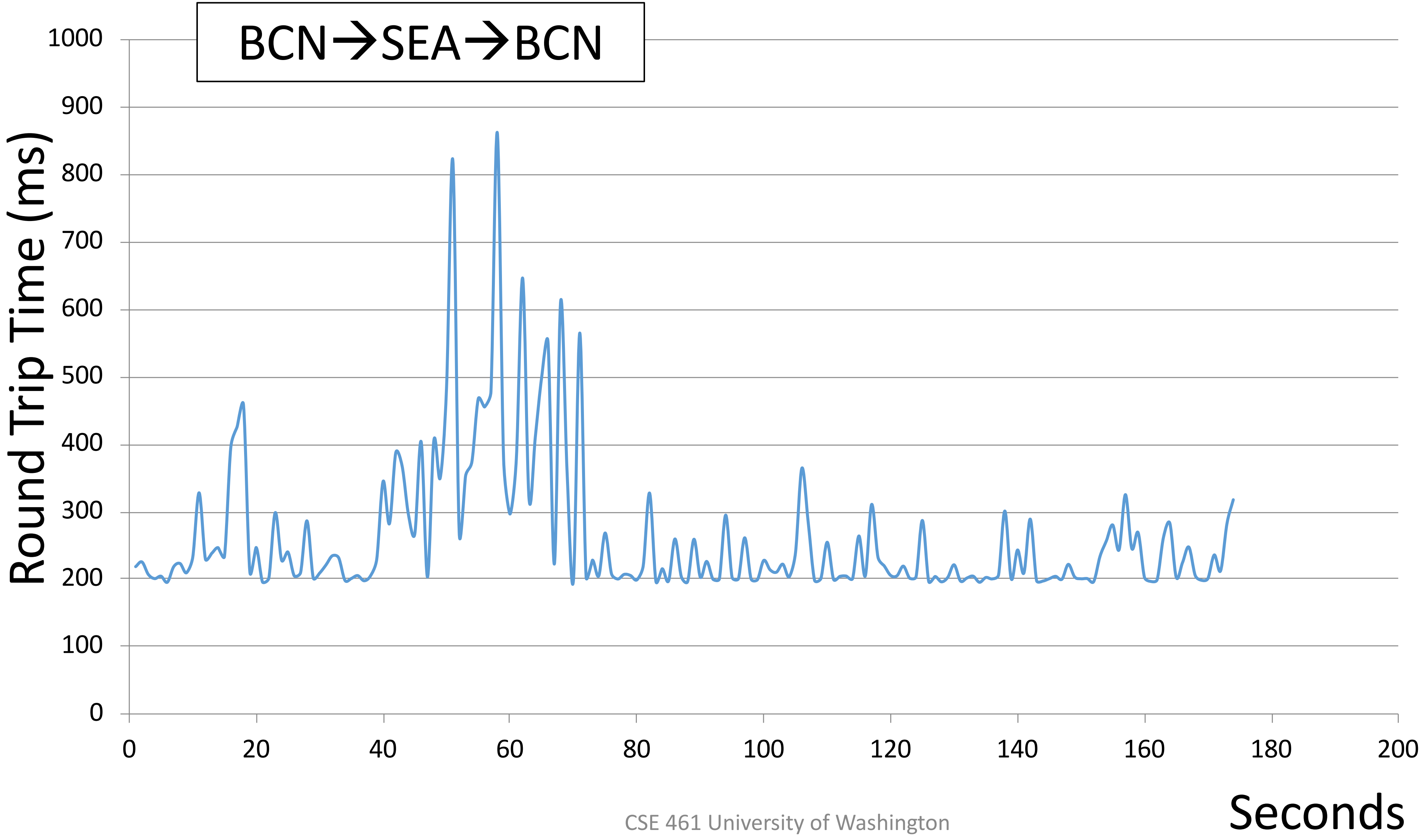
- With sliding window, detecting loss with <u>timeout</u>
  - Set timer when a segment is sent
  - Cancel timer when ack is received
  - If timer fires, <u>retransmit</u> data as lost



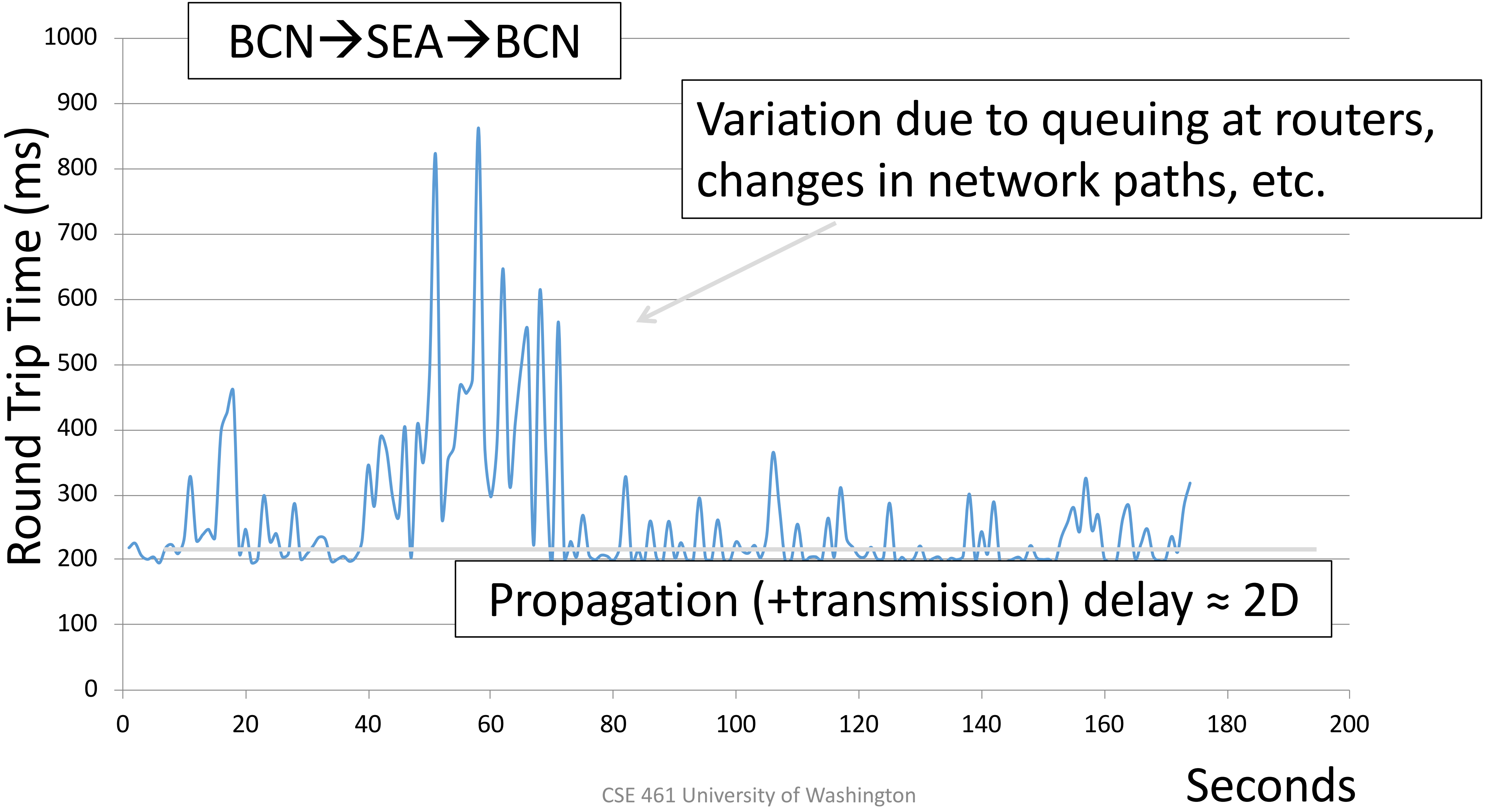
Timeout Problem

- Timeout should be "just right"
  - Too long → inefficient network capacity use
  - Too short → spurious resends waste network capacity
- 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

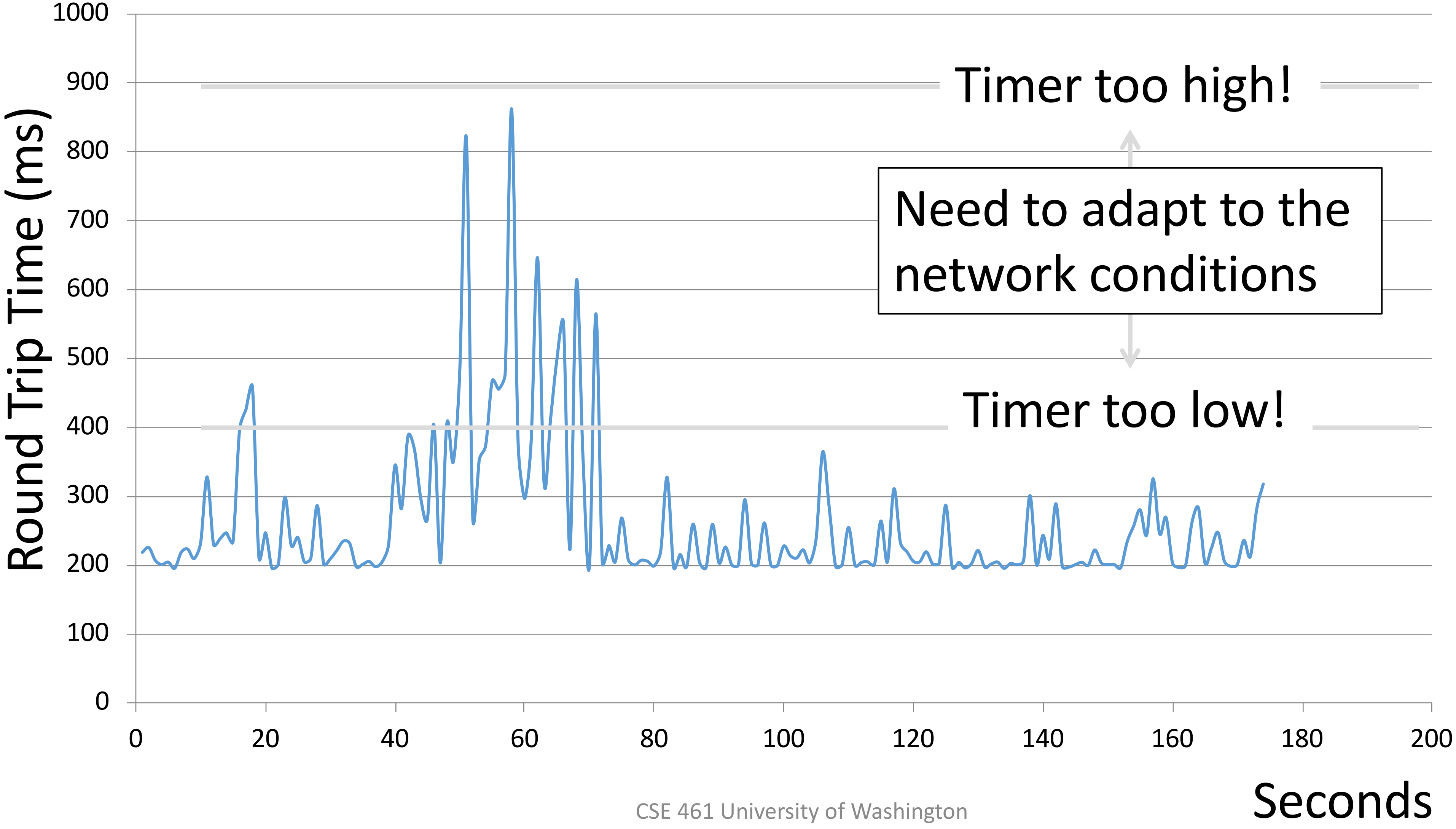
Example of RTTs



Example of RTTs (2)



Example of RTTs (3)



Need to adapt to the network conditions

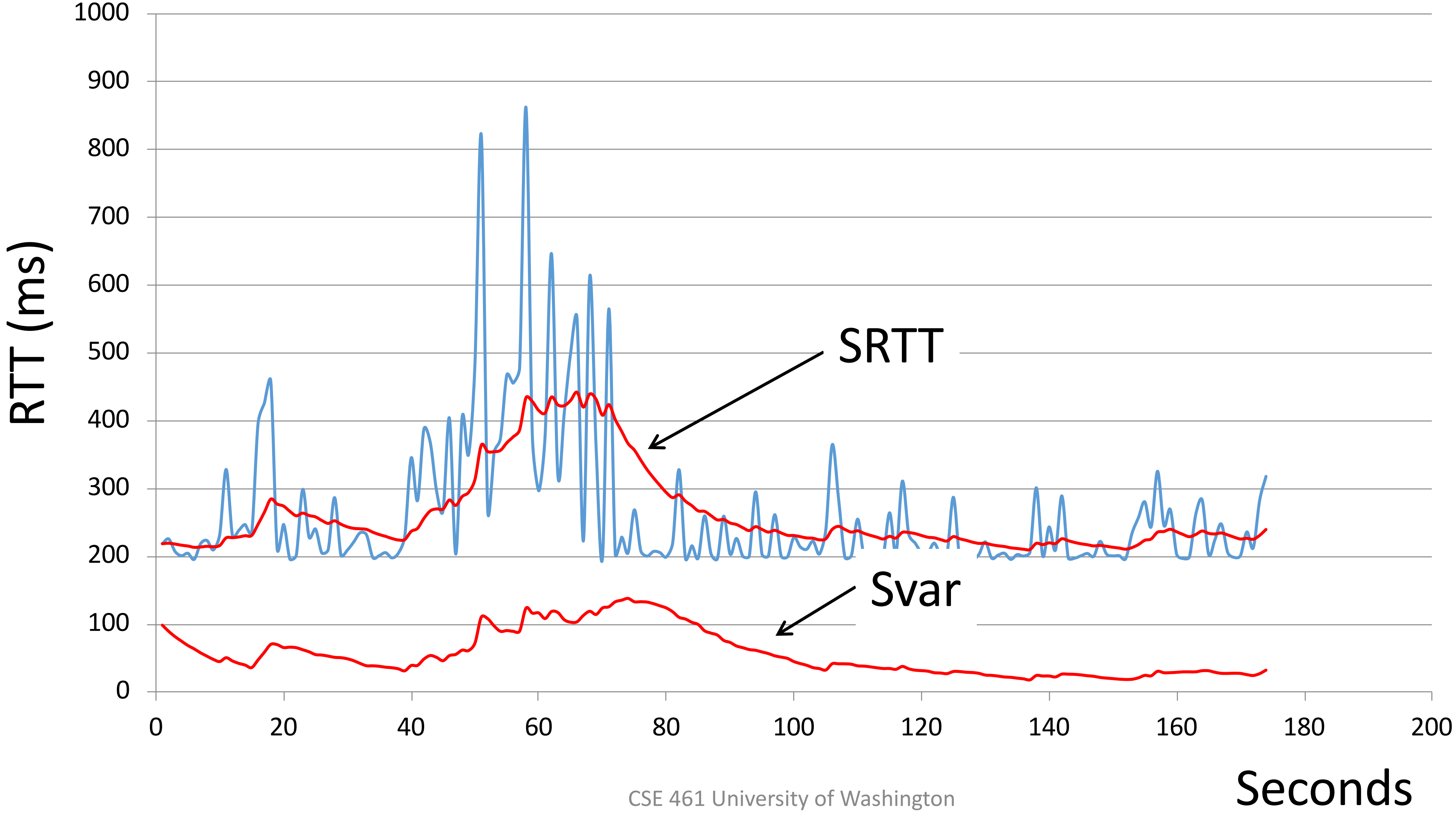
Adaptive Timeout

- Smoothed estimates of the RTT (1) and variance in RTT (2)
  - Update estimates with a moving average

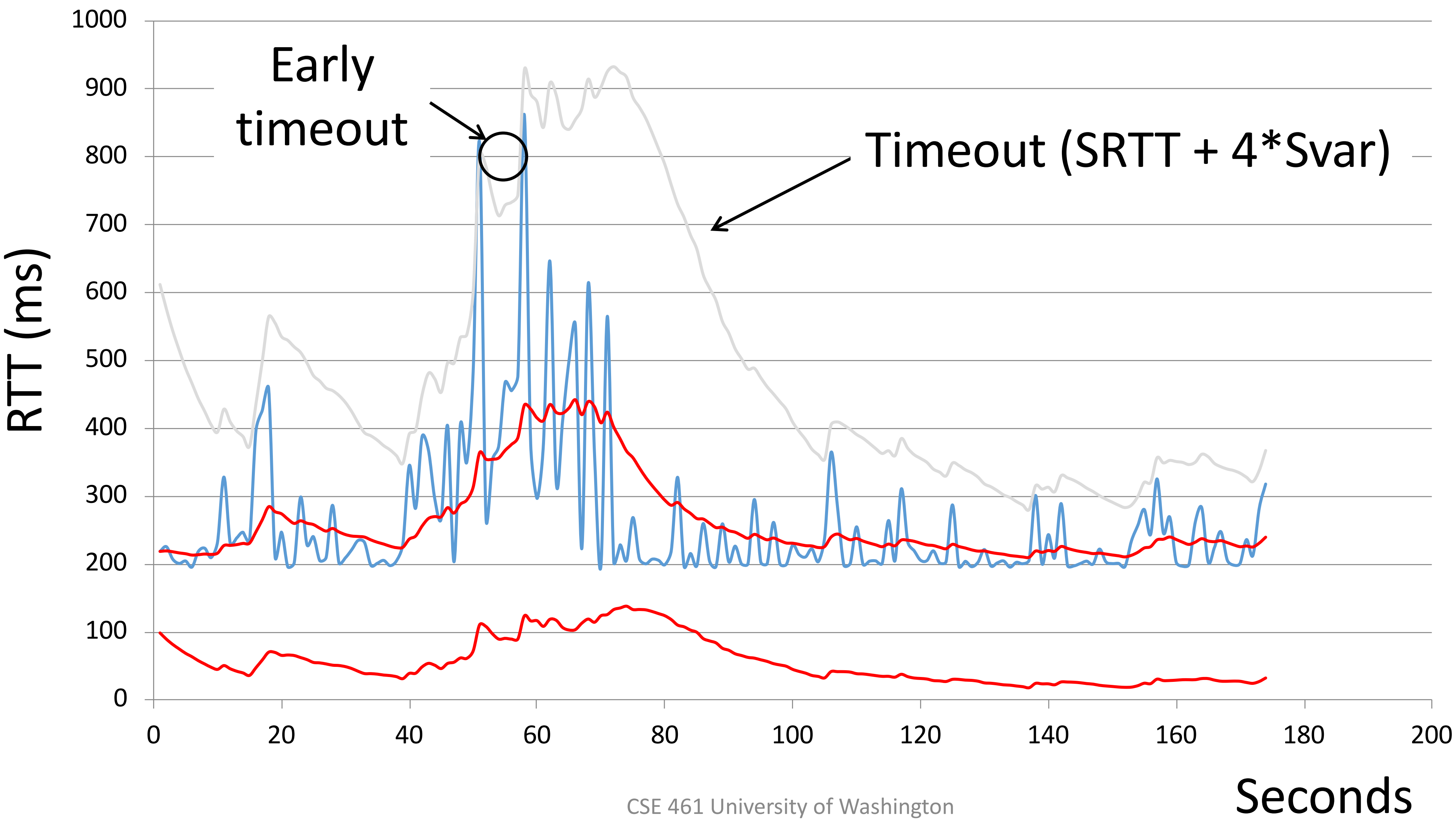
1. $SRTT_{N+1} = 0.9*SRTT_N + 0.1*RTT_{N+1}$
2. $Svar_{N+1} = 0.9*Svar_N + 0.1*|RTT_{N+1} - SRTT_{N+1}|$

- 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 (2)

- 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

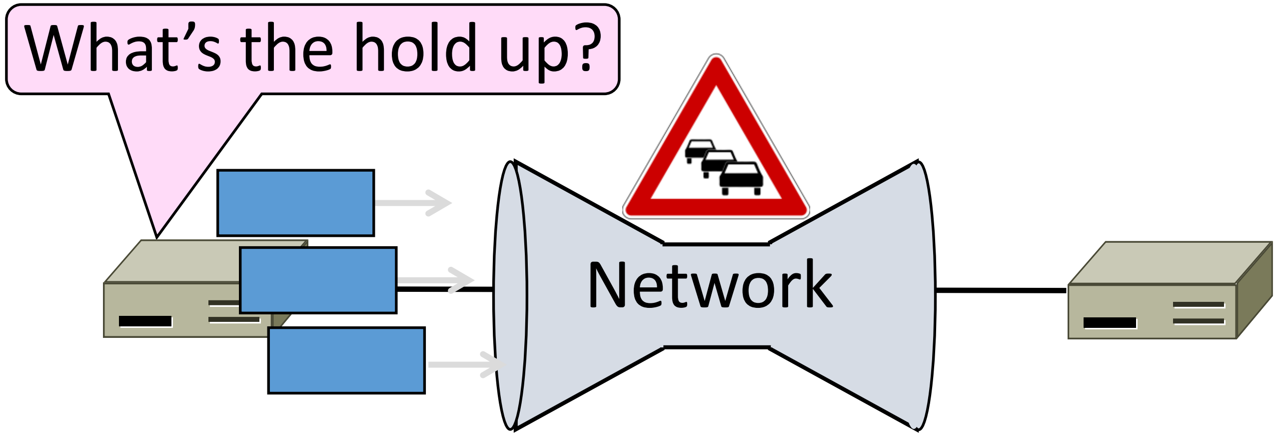
TCP to date:

- We can set up and tear connections
  - Connection establishment and release handshakes
- 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

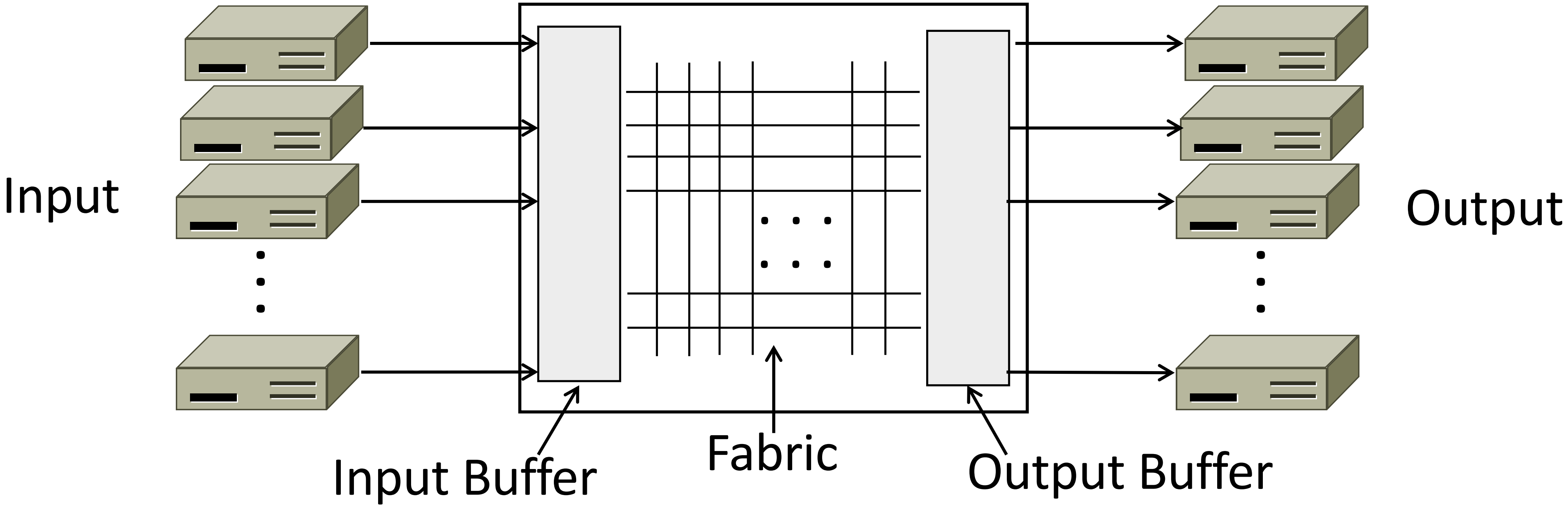


Congestion Collapse in the 1980s

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

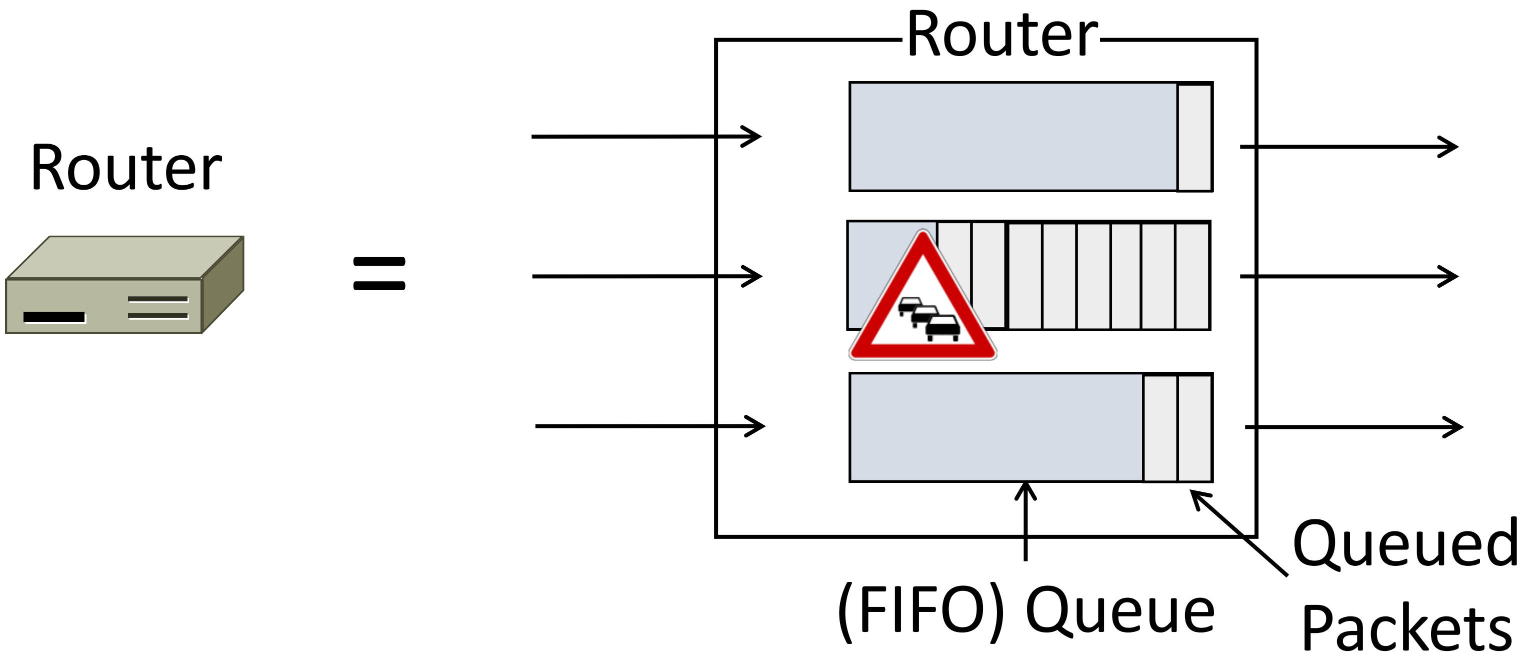
Nature of Congestion

- 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

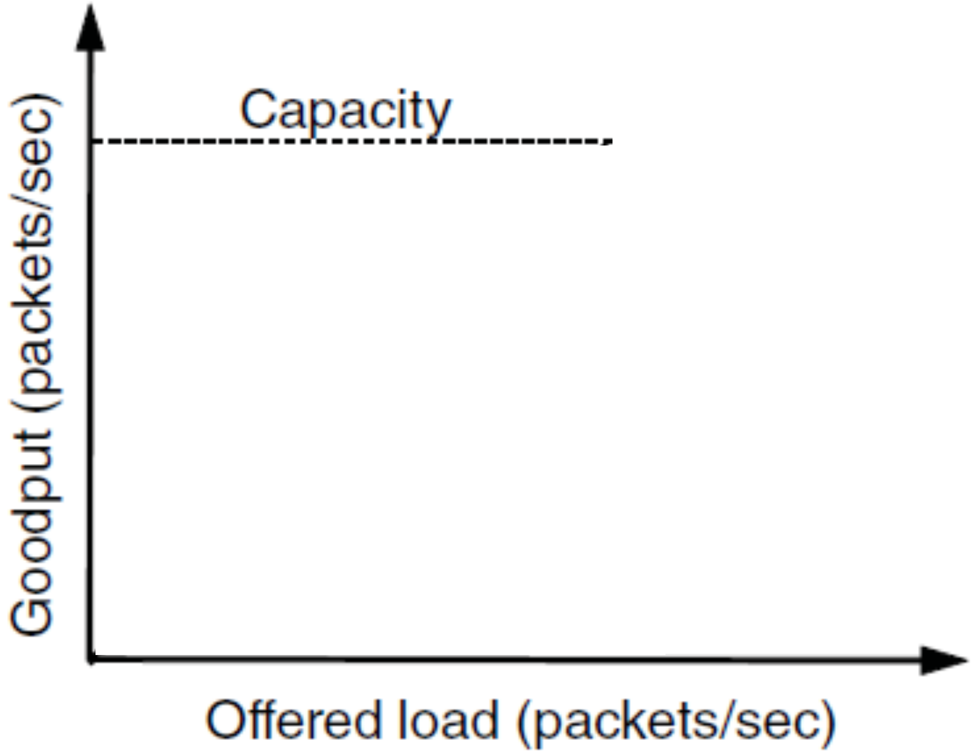


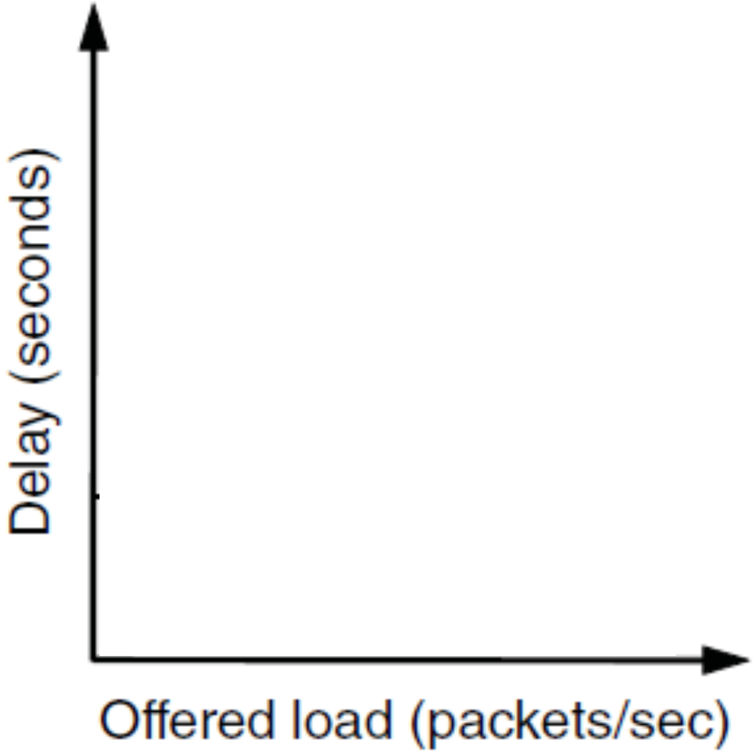
Nature of Congestion (3)

- 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

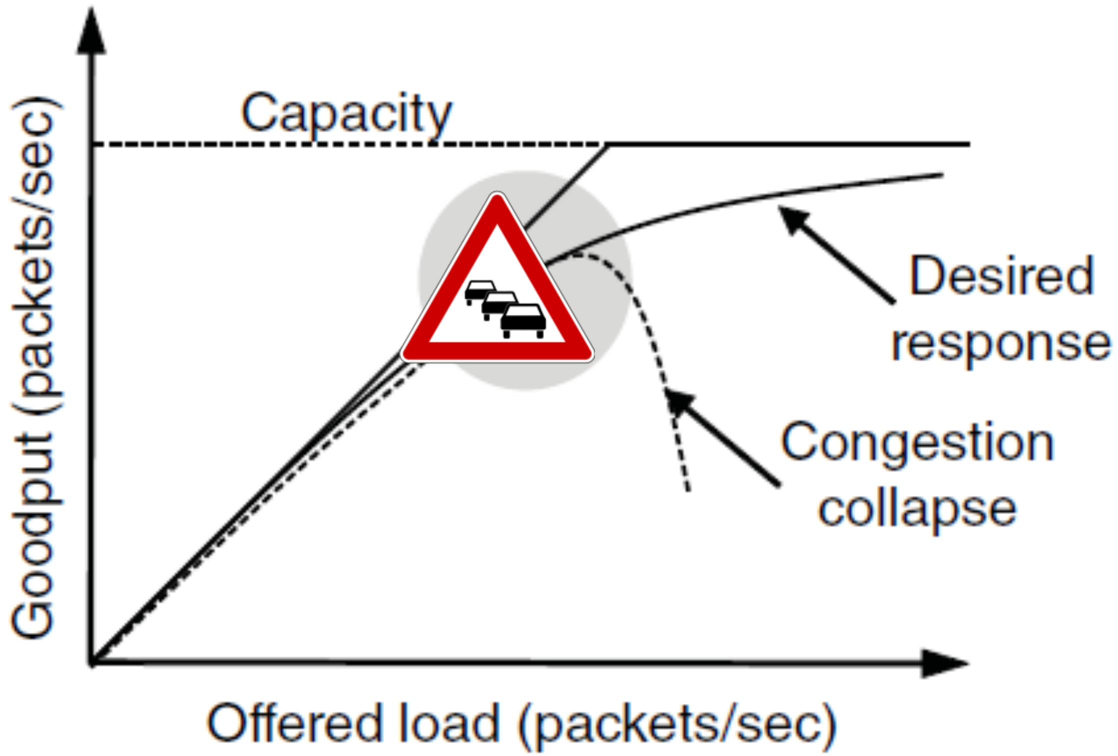
- What happens to performance as we increase load?

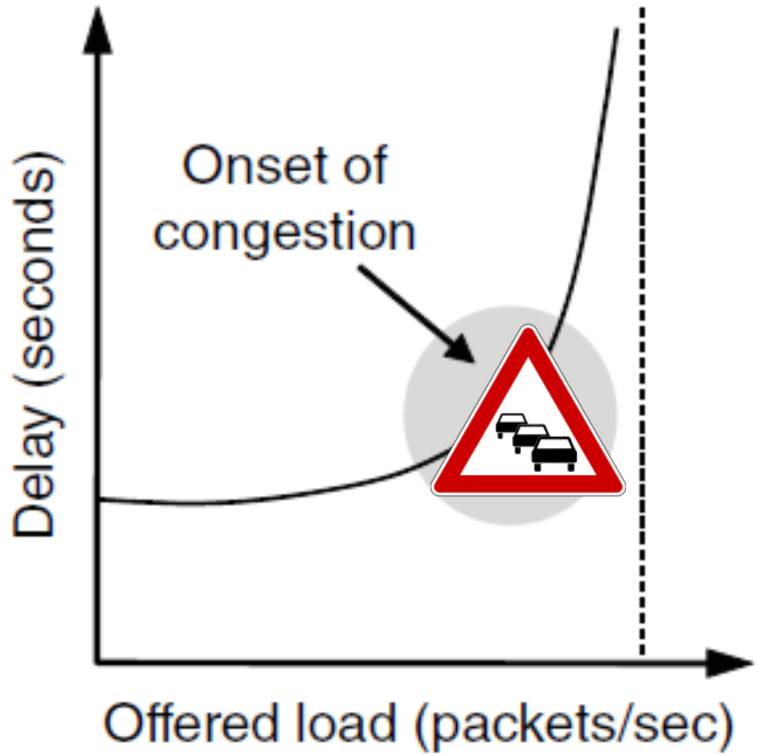




Effects of Congestion (2)

- What happens to performance as we increase load?





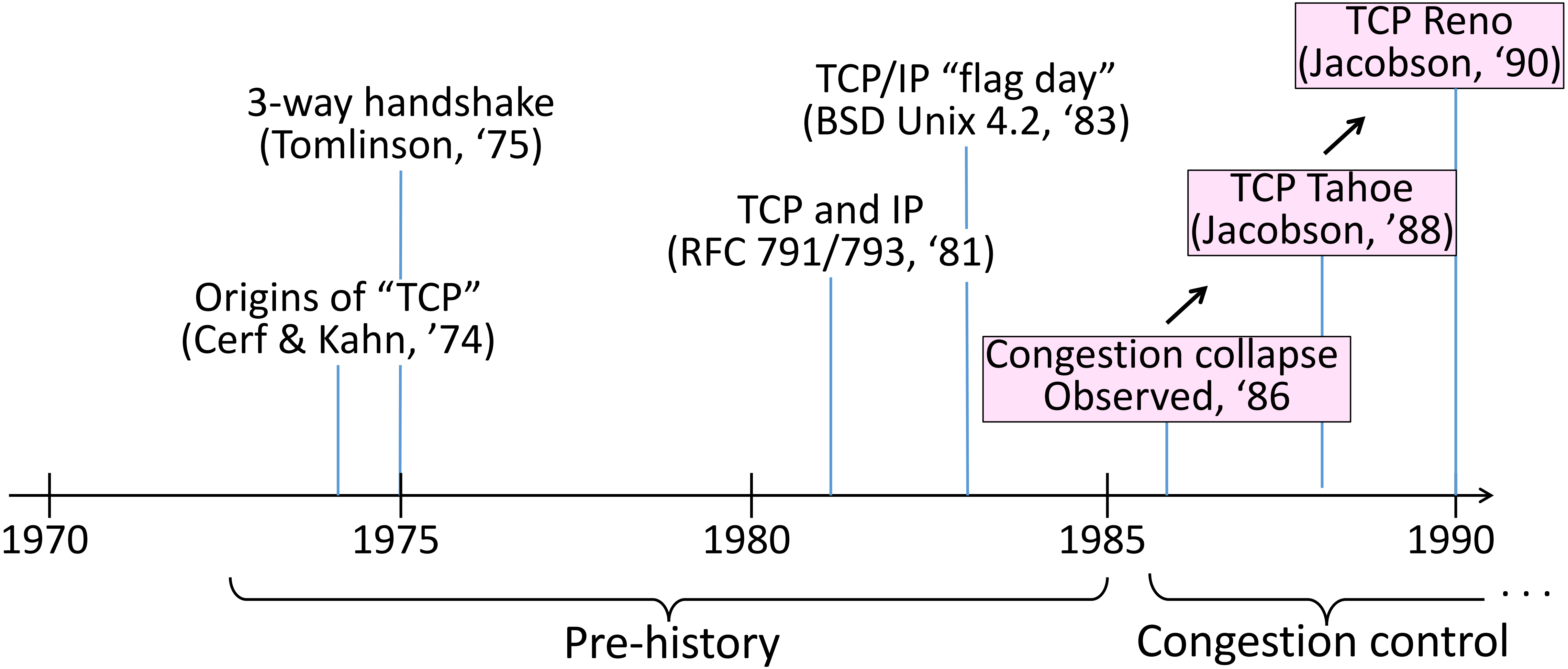
Effects of Congestion (3)

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

TCP Tahoe/Reno

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

TCP Timeline



TCP Reno
(Jacobson, ‘90)

TCP/IP “flag day”
(BSD Unix 4.2, ‘83)

3-way handshake
(Tomlinson, ‘75)

TCP Tahoe
(Jacobson, ’88)

TCP and IP
(RFC 791/793, ‘81)

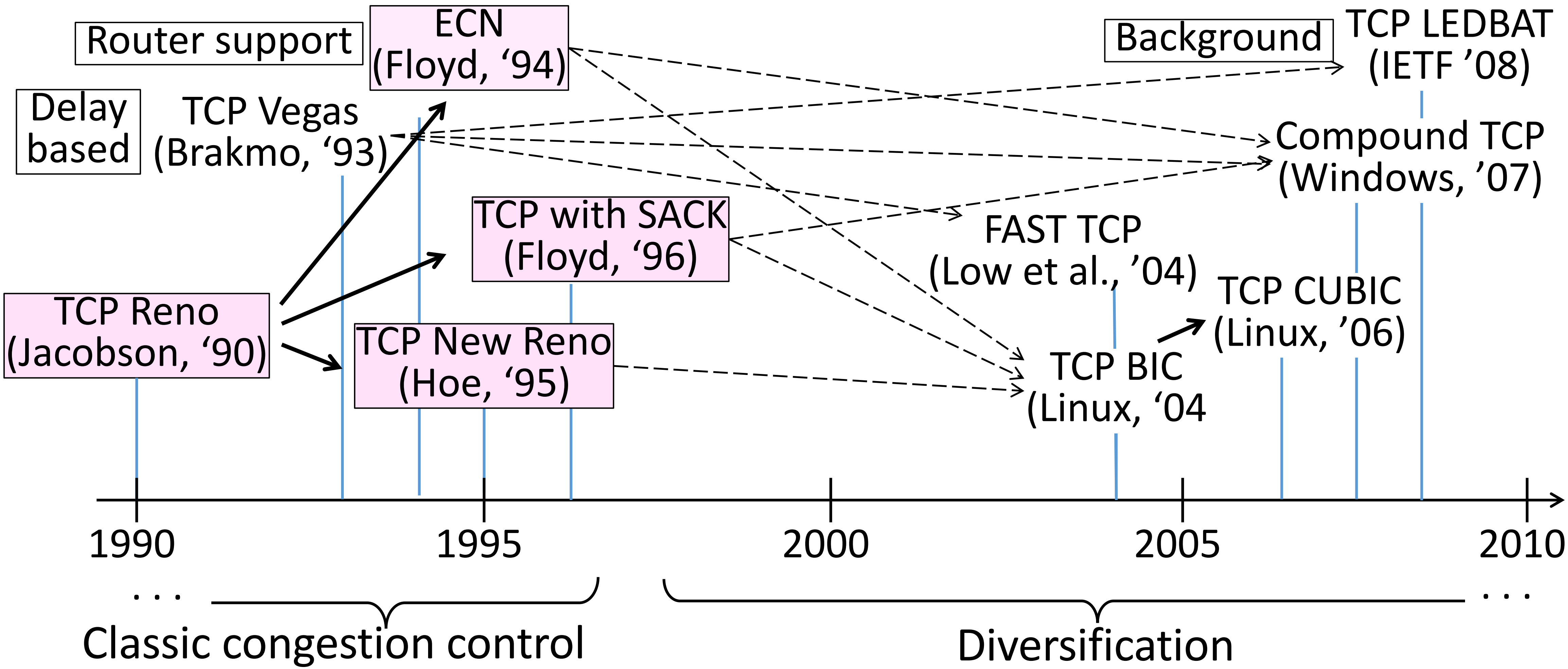
Origins of “TCP”
(Cerf & Kahn, ’74)

Congestion collapse
Observed, ‘86

Pre-history

Congestion control

TCP Timeline (2)



Router support

ECN
(Floyd, ‘94)

Delay based

TCP Vegas
(Brakmo, ‘93)

TCP with SACK
(Floyd, ‘96)

TCP Reno
(Jacobson, ‘90)

TCP New Reno
(Hoe, ‘95)

Background

TCP LEDBAT
(IETF ’08)

Compound TCP
(Windows, ’07)

FAST TCP
(Low et al., ’04)

TCP CUBIC
(Linux, ’06)

TCP BIC
(Linux, ‘04

Classic congestion control

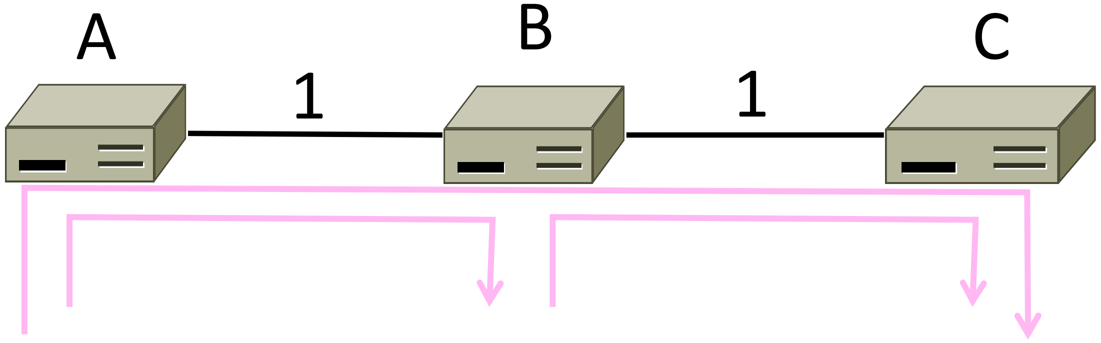
Diversification

Bandwidth Allocation

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

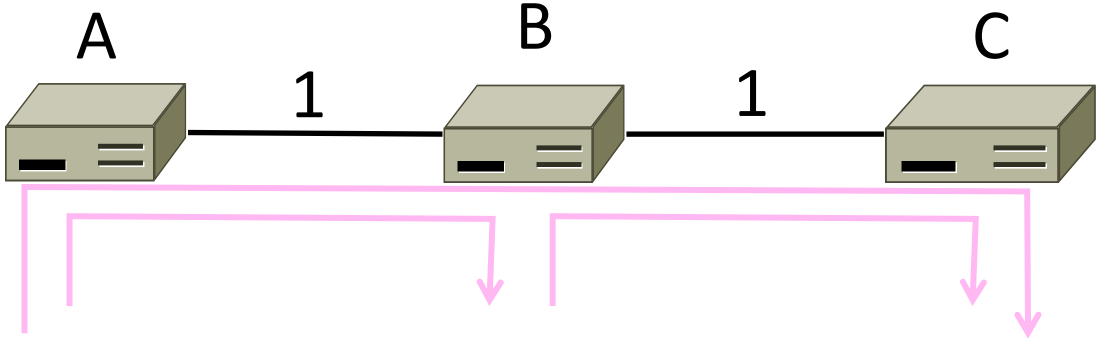
Efficiency vs. Fairness

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



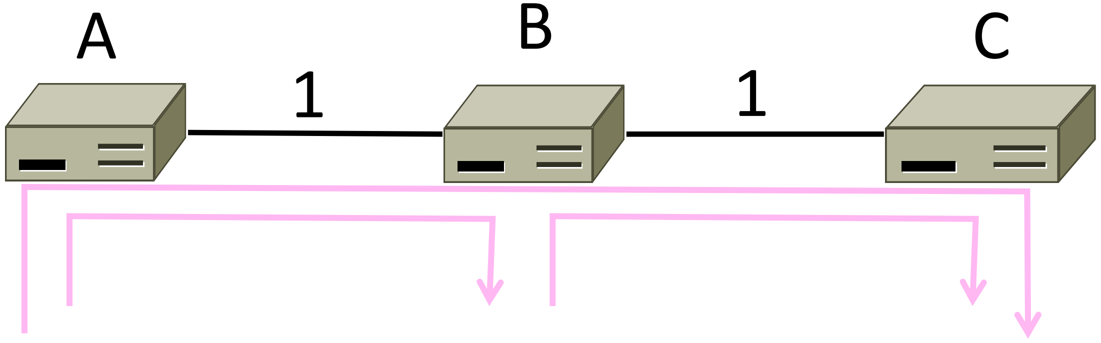
Efficiency vs. Fairness (2)

- 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!



Fairness

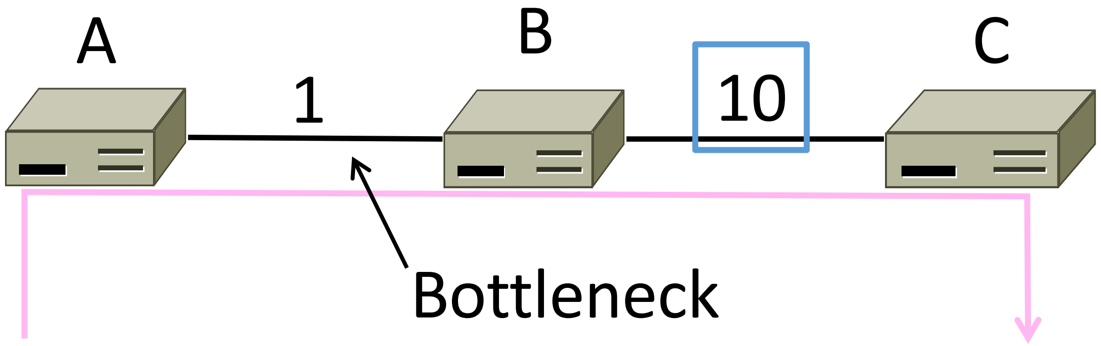
- What's a "fair" bandwidth allocation?
  - The max-min fair allocation

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 starvation
    - A node that cannot use any bandwidth
  - “Equal per flow” is good enough

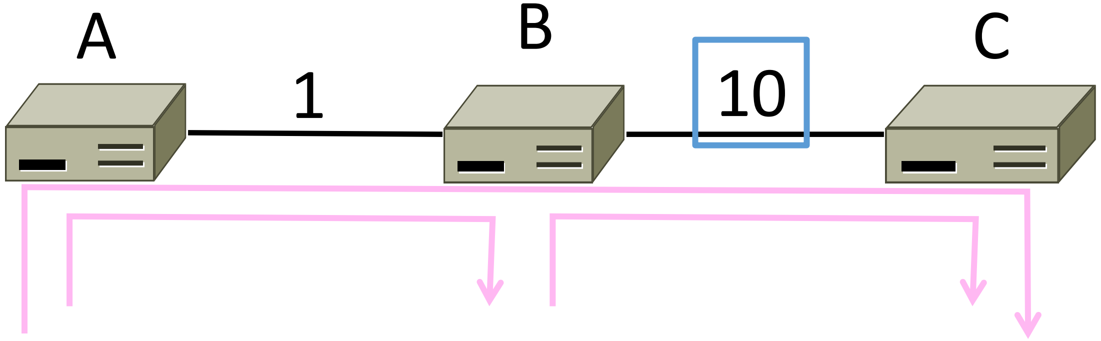
Generalizing "Equal per Flow"

- Bottleneck 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

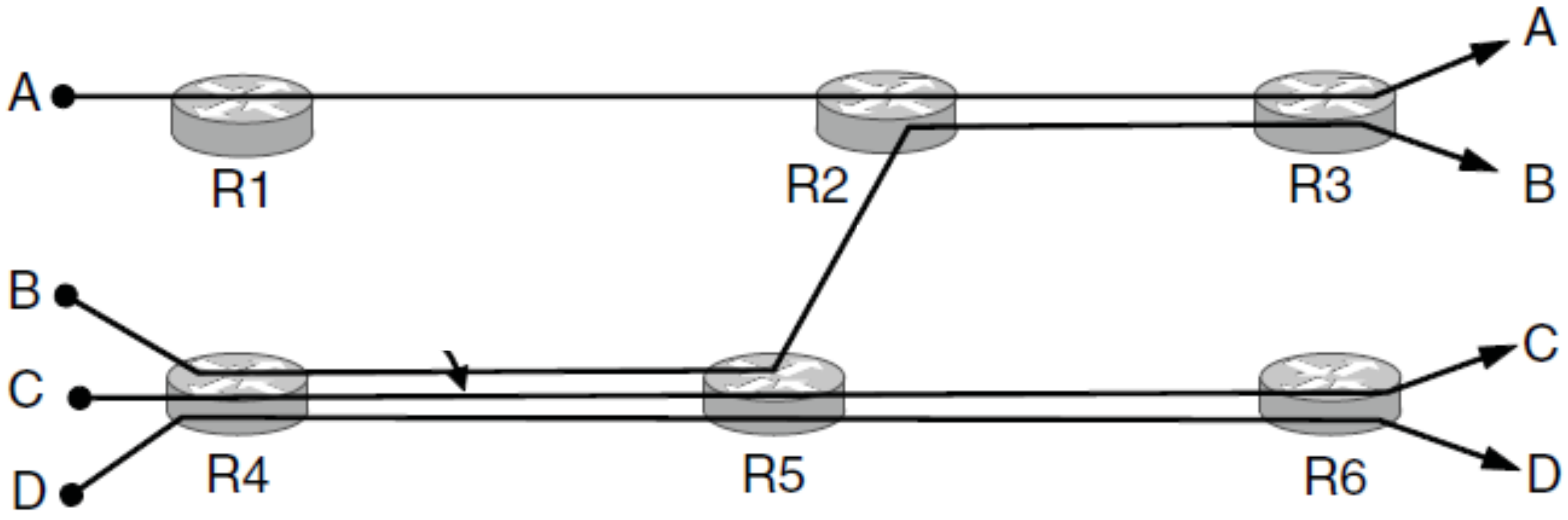
- Intuitively, flows bottlenecked on a link get an equal share of that link
- <u>Max-min fair allocation</u> is one that:
  - Increasing the rate of one flow will decrease the rate of a smaller flow
  - This "maximizes the minimum" flow

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

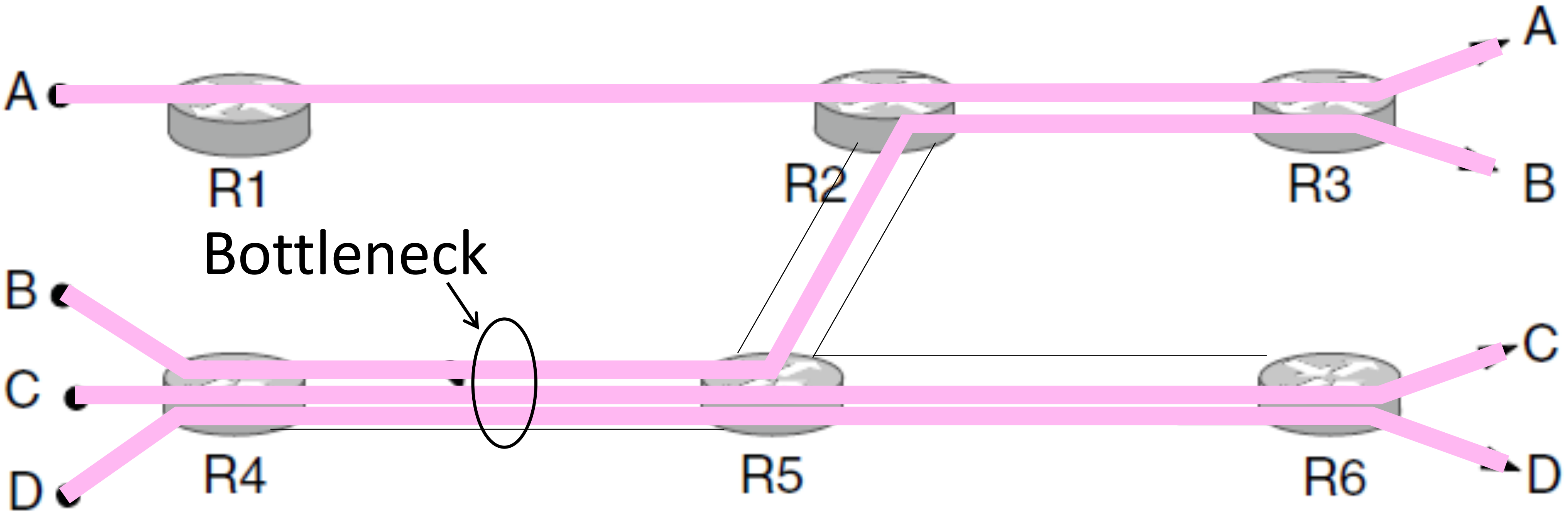
Max-Min Example

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



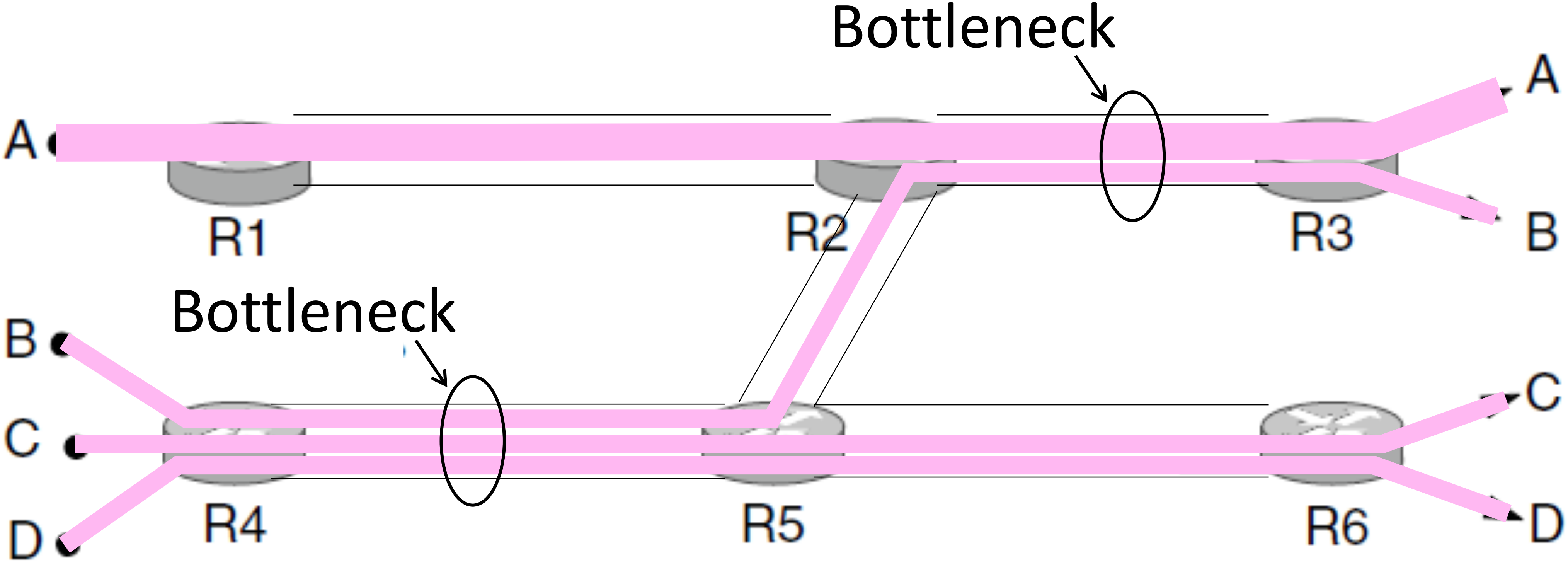
Max-Min Example (2)

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



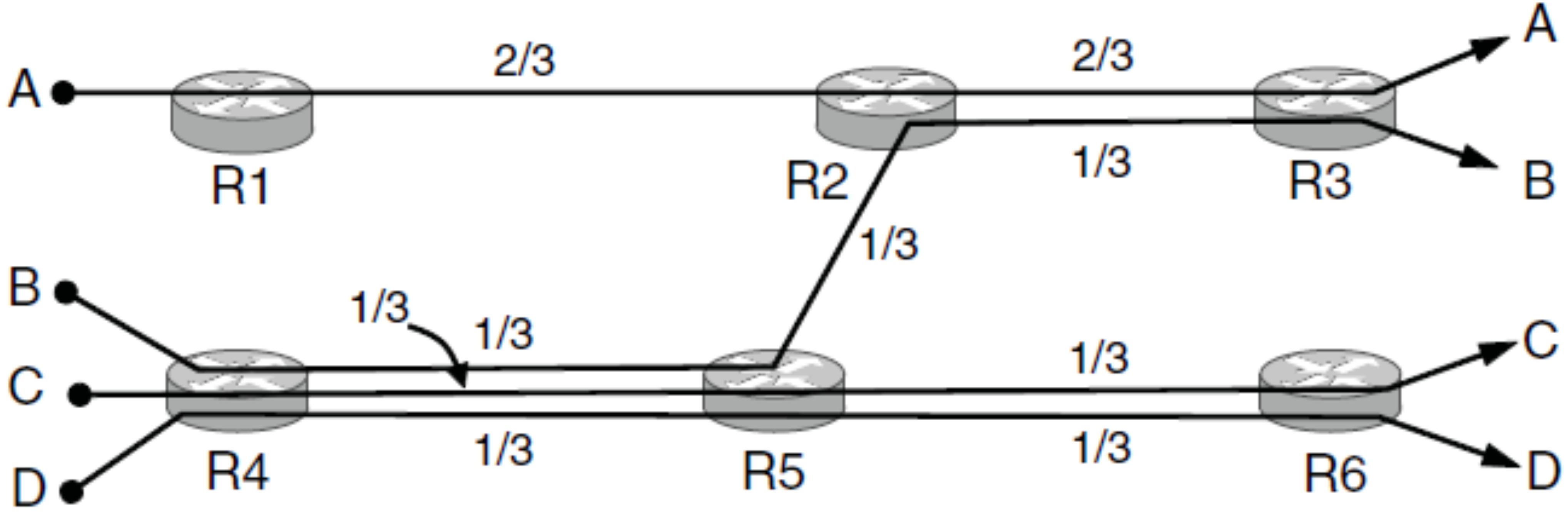
Max-Min Example (3)

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



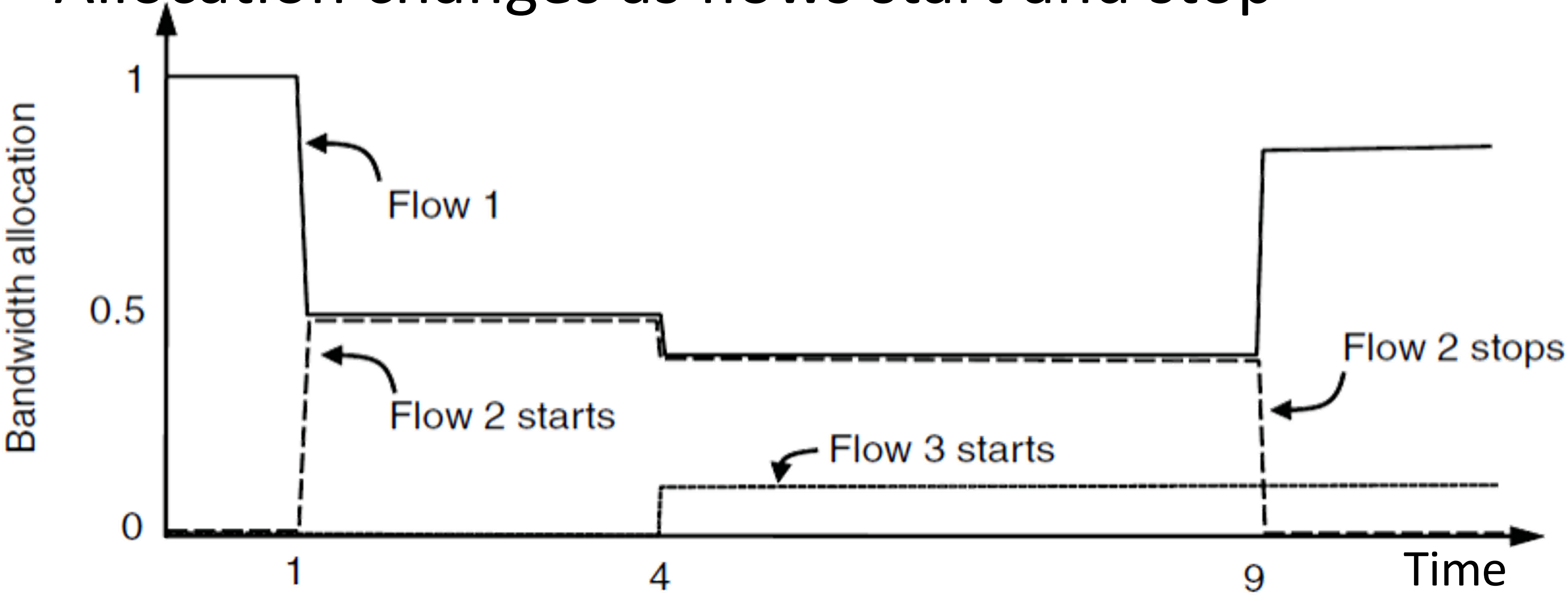
Max-Min Example (4)

- End with A=2/3, B, C, D=1/3, and R2—R3, R4—R5 full

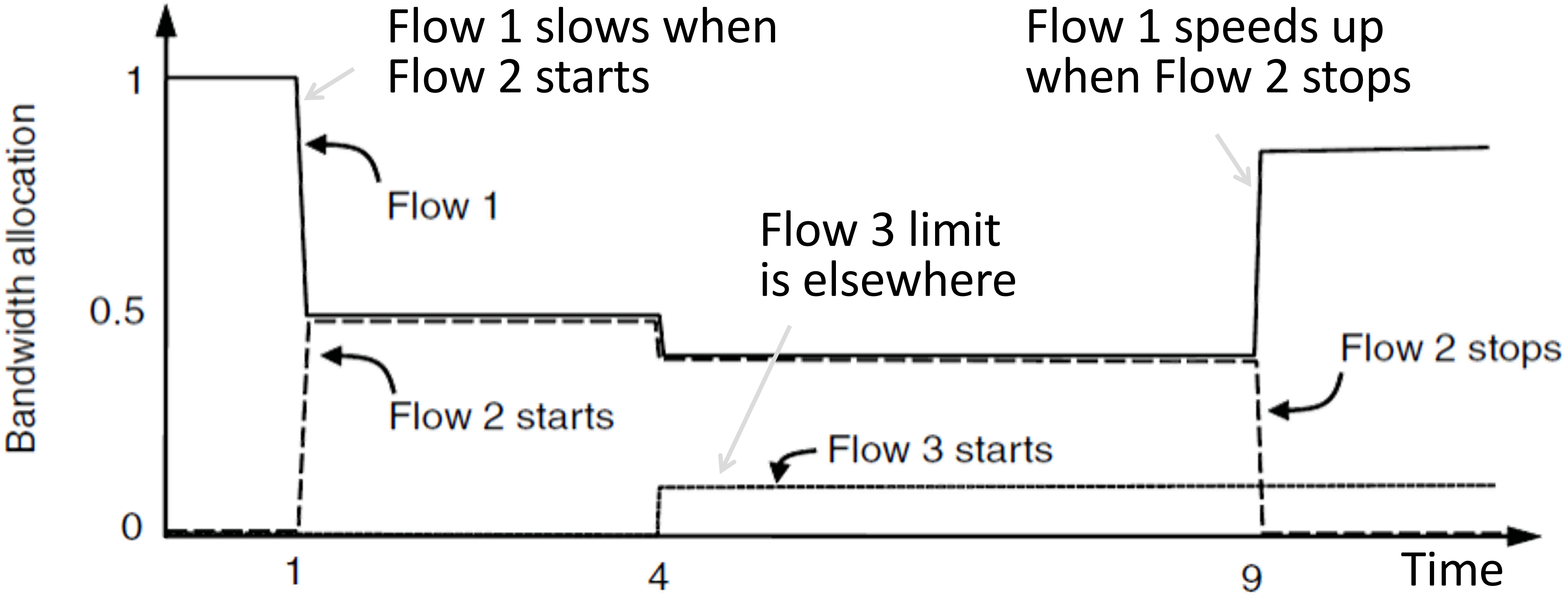


Adapting over Time

- Allocation changes as flows start and stop



Adapting over Time (2)



Flow 1 slows when Flow 2 starts

Flow 1 speeds up when Flow 2 stops

Flow 3 limit is elsewhere