CS4450

Computer Networks:
Architecture and Protocols

Lecture 22
More TCP Congestion Control

Rachit Agarwal

Recap: Transmission Control Protocol (TCP)

- Reliable, in-order delivery
    - Ensures byte stream (eventually) arrives intact
    - In the presence of corruption, delays, reordering, loss
- Connection oriented
    - Explicit set-up and tear-down of TCP session
- Full duplex stream of **byte service**
    - **Sends and receives stream of bytes**
- **Flow control**
    - Ensures the sender does not overwhelm the receiver
- **Congestion control**
    - Dynamic adaptation to network path's capacity

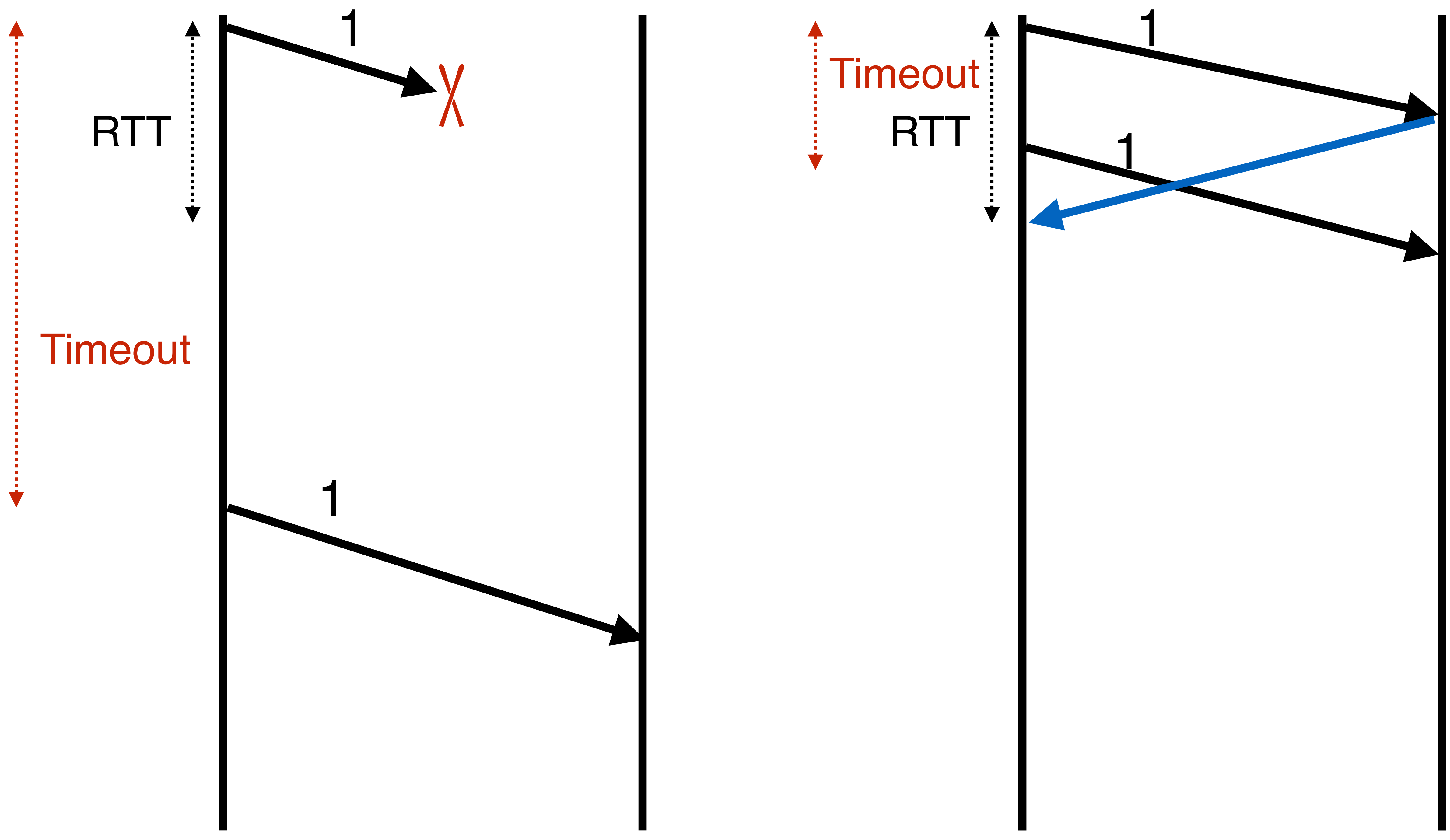
Recap: Basic Components of TCP

- **Segments, Sequence numbers, ACKs**
  - TCP uses byte sequence numbers to identify payloads
  - ACKs referred to sequence numbers
  - Window sizes expressed in terms of # of bytes
- **Retransmissions**
  - Can’t be correct without retransmitting lost/corrupted data
  - TCP retransmits based on timeouts and duplicate ACKs
    - Timeouts based on estimate of RTT
- **Flow Control**
- **Congestion Control**

Recap: Loss with Cumulative ACKs

- Sender sends packets with 100B and seqnos
  - 100, 200, 300, 400, 500, 600, 700, 800, 900
- Assume 5th packet (seqno 500) is lost, but no others
- Stream of ACKs will be
  - 200, 300, 400, 500, 500, 500, 500, 500
- Duplicate ACKs are a sign of an isolated loss
  - The lack of ACK progress means 500 hasn't been delivered
  - Stream of ACKs means some packets are being delivered
- Therefore, could trigger resend upon receiving k duplicate ACKs
  - Large k -> fewer retransmissions, more latency, lower rate (why?)
  - Small k -> more retransmissions, lower latency

Recap: Setting the Timeout Value (RTO)



Timeout too long -> inefficient

Timeout too short -> duplicate packets

Recap: Flow Control (Sliding Window)

- Advertised Window: W
  - Can send W bytes beyond the next expected byte
- Receiver uses W to prevent sender from overflowing buffer
- Limits number of bytes sender can have in flight

Recap: TCP congestion control: high-level idea

- End hosts adjust sending rate

- Based on implicit feedback from the network
  - Implicit: router drops packets because its buffer overflows, not because it’s trying to send message

- Hosts probe network to test level of congestion
  - Speed up when no congestion (i.e., no packet drops)
  - Slow down when when congestion (i.e., packet drops)

- How to do this efficiently?
  - Extend TCP’s existing window-based protocol…
  - Adapt the window size based in response to congestion

Recap: Not All Losses the Same

- **Duplicate ACKs: isolated loss**
  - Still getting ACKs

- **Timeout: possible disaster**
  - Not enough duplicate ACKs
  - Must have suffered several losses

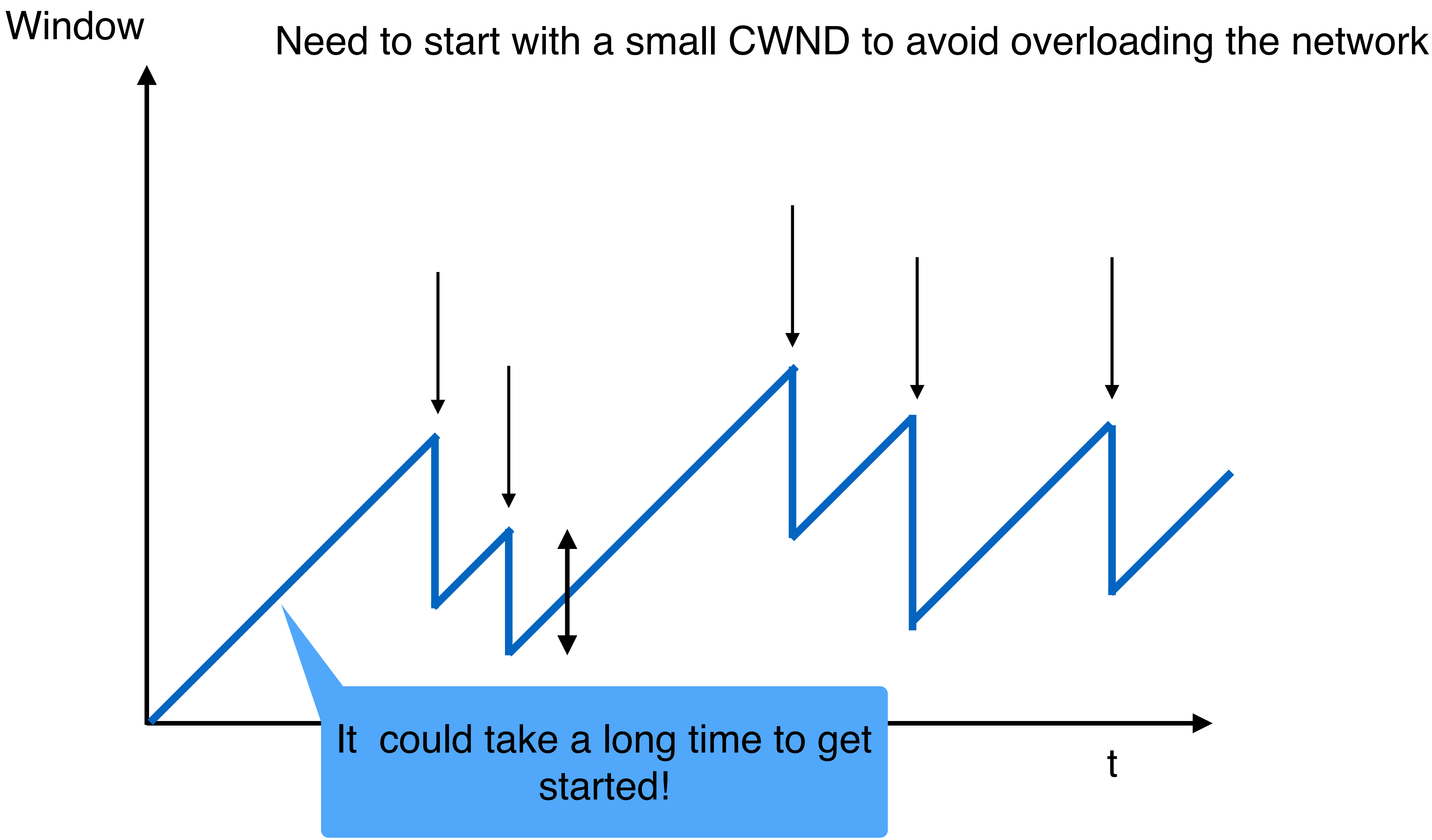
Recap: Additive Increase, Multiplicative Decrease (AIMD)

- Additive increase
    - On success of last window of data, increase by one MSS
    - If W packets in a row have been ACKed, increase W by one
    - i.e., +1/W per ACK
- Multiplicative decrease
    - On loss of packets by DupACKs, divide congestion window by half
    - Special case: when timeout, reduce congestion window to one MSS

Recap: AIMD

- ACK: CWND -> CWND + 1/CWND
  - When CWND is measured in MSS
  - Note: after a full window, CWND increase by 1 MSS
  - Thus, **CWND increases by 1 MSS per RTT**
- 3rd DupACK: CWND -> CWND/2
- Special case of timeout: CWND -> 1 MSS

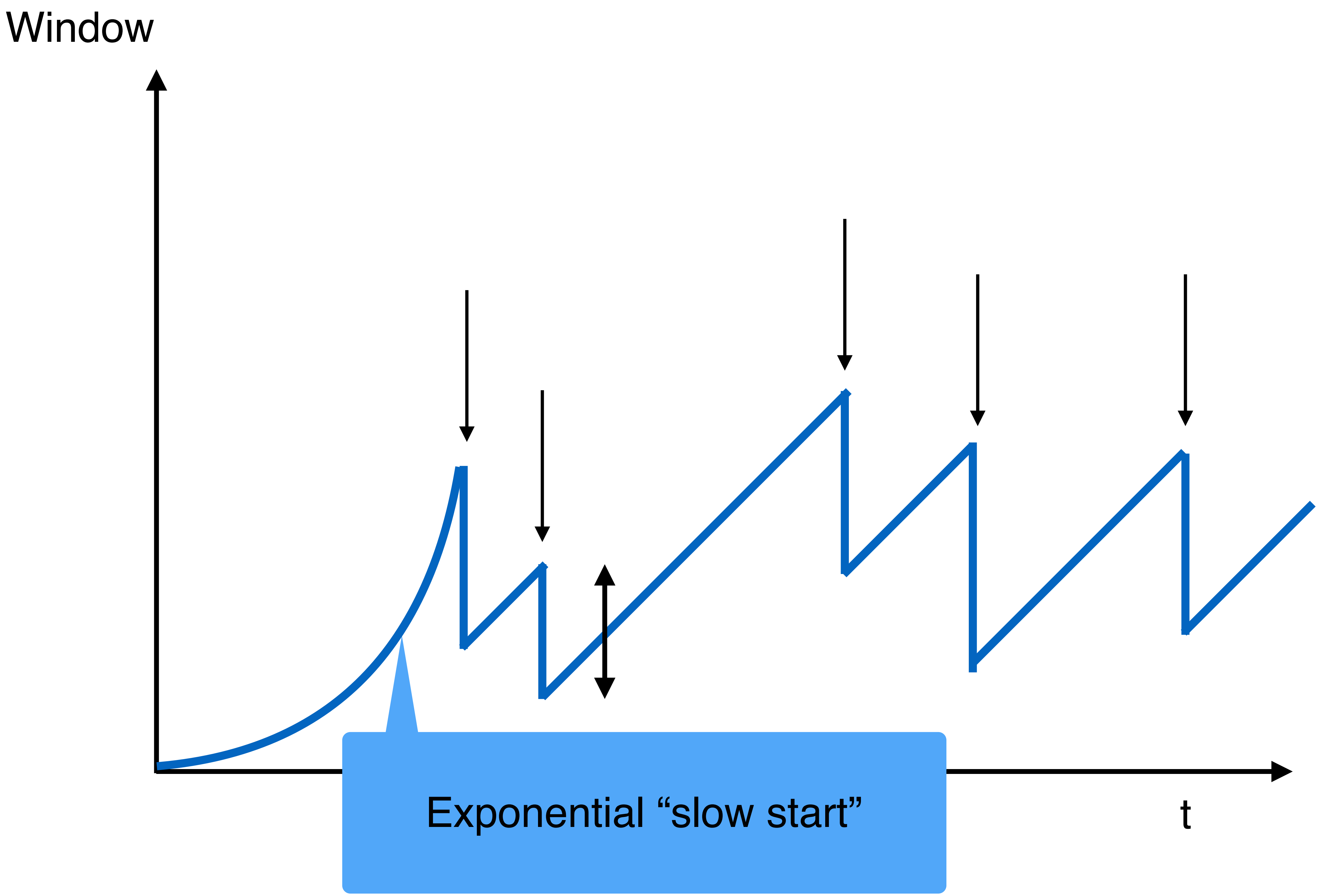
Recap: AIMD Starts Too Slowly



Recap: “Slow Start” Phase

- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS/RTT
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (**actually “fast start”**)
  - Sender starts at a slow rate (hence the name)
  - … but increases exponentially until first loss

Recap: Slow Start and the TCP Sawtooth (no timeouts)

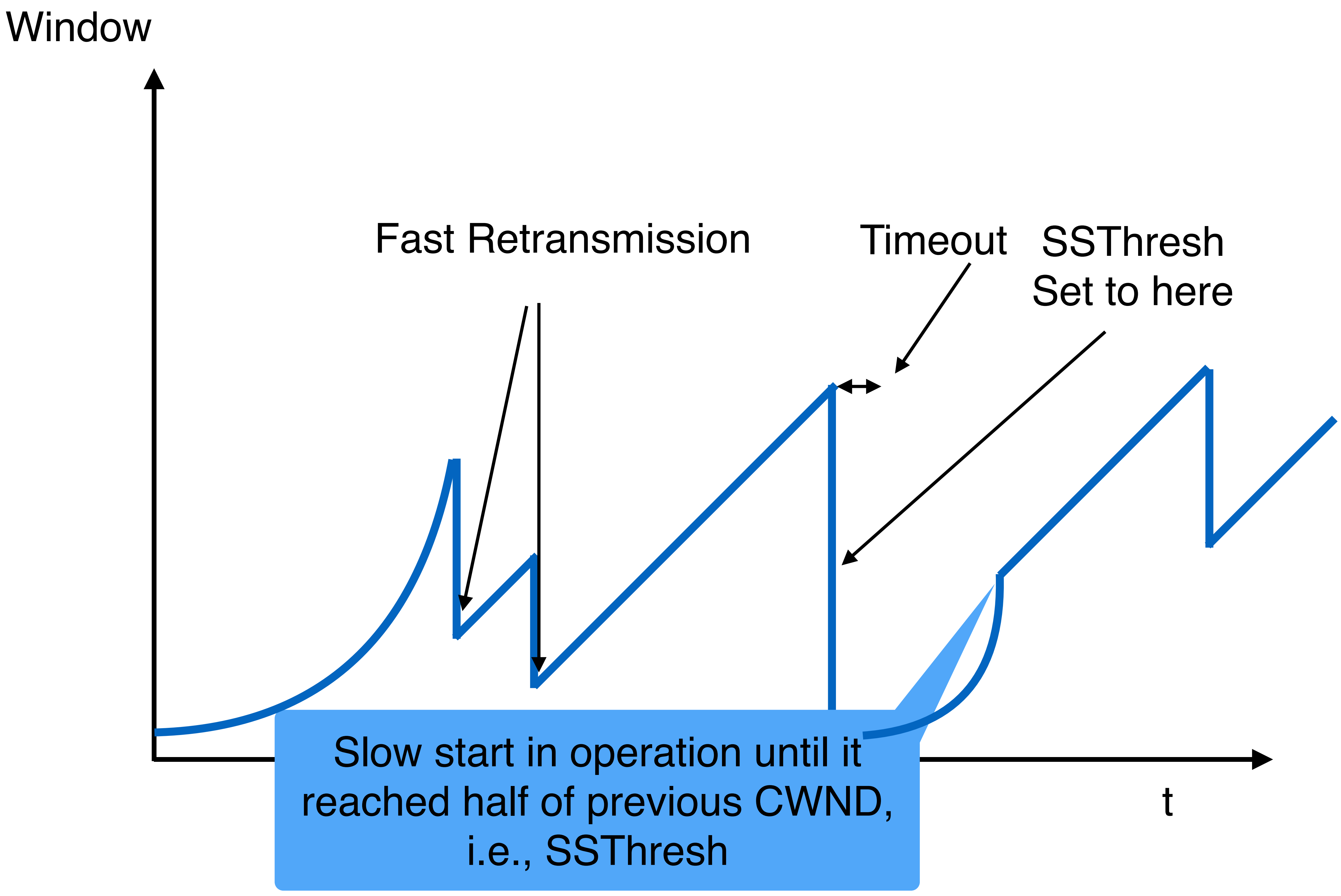


Timeouts

Loss Detected by Timeout

- Sender starts a timer that runs for RTO seconds
- **Restart timer whenever ACK for new data arrives**
- If timer expires
    - Set SSHTHRESH <- CWND/2 (“Slow Start Threshold”)
    - Set CWND <- 1 (MSS)
    - Retransmit **first** lost packet
    - Execute Slow Start until CWND > SSTHRESH
    - After which switch to Additive Increase

TCP Time Diagram (with timeouts)



Summary of Increase

- "Slow start": increase CWND by 1 (MSS) for each ACK
  - A factor of 2 per RTT
- Leave slow-start regime when either:
  - CWND > SSTHRESH
  - Packet drop detected by dupacks
- Enter AIMD regime
  - Increase by 1 (MSS) for each window's worth of ACKed data

Summary of Decrease

- Cut CWND half on loss detected by dupacks
  - **Fast retransmit to avoid overreacting**
- Cut CWND all the way to 1 (MSS) on **timeout**
  - Set ssthresh to CWND/2
- Never drop CWND below 1 (MSS)
  - Our correctness condition: always try to make progress

TCP Congestion Control Details

Implementation

- State at sender
    - CWND (initialized to a small constant)
    - ssthresh (initialized to a large constant)
    - dupACKcount
    - Timer, as before
- Events at sender
    - ACK (new data)
    - dupACK (duplicate ACK for old data)
    - Timeout
- What about receiver? Just send ACKs upon arrival
    - Assuming RWND > CWND

Event: ACK (new data)

- If in slow start
    - CWND += 1

CWND packets per RTT
Hence after one RTT with
no drops:
CWND = 2 x CWND

Event: ACK (new data)

- If CWND <= ssthresh
  - CWND += 1
- Else
  - CWND = CWND + 1/CWND

Slow Start Phase

Congestion Avoidance Phase

(additive increase)

CWND packets per RTT
Hence after one RTT with no drops:
CWND = CWND + 1

Event: Timeout

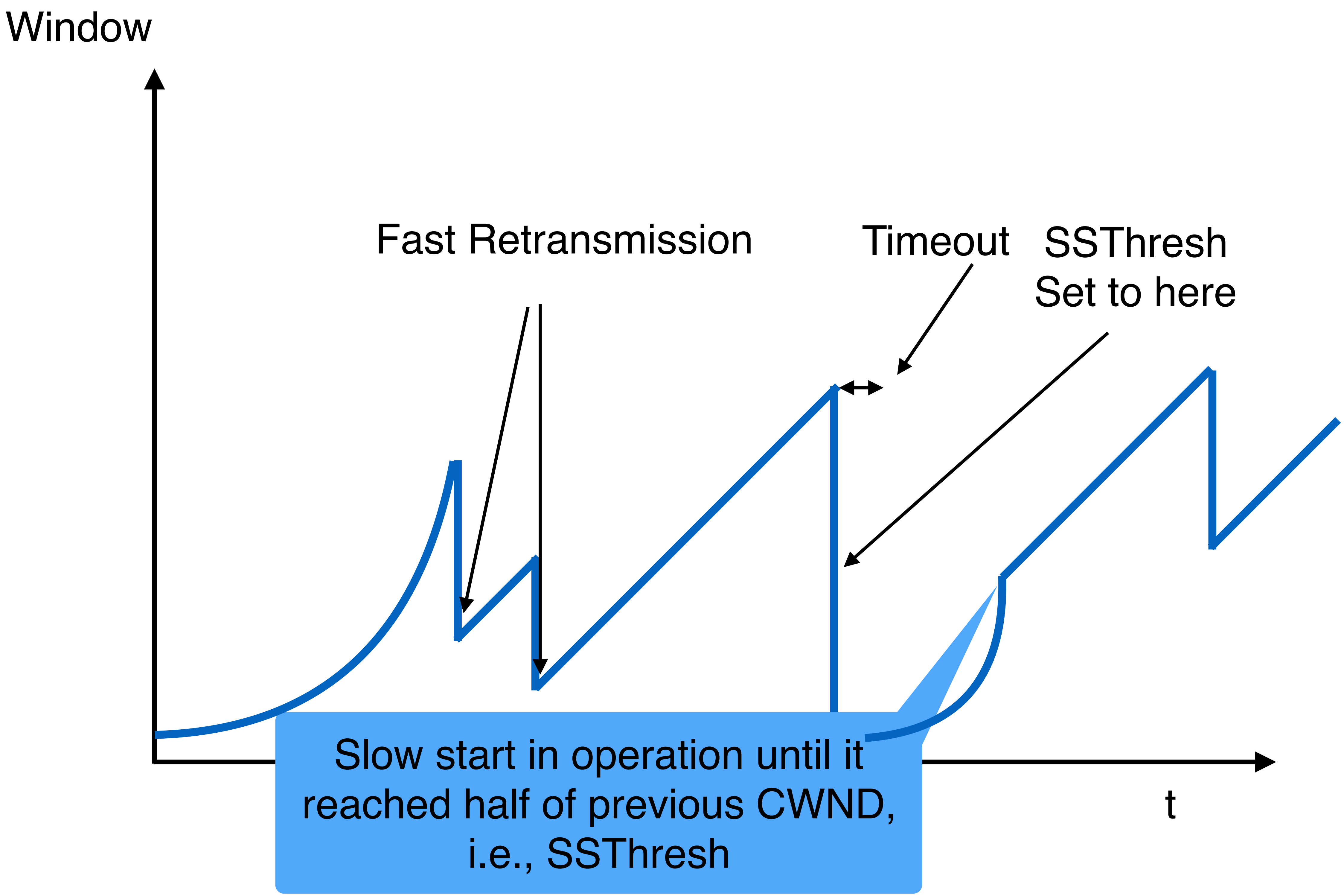
- On Timeout
    - ssthresh <- CWND/2
    - CWND <- 1

Event: dupACK

- dupACKcount++
- If dupACKcount = 3 /* fast retransmit */
    - ssthresh <- CWND/2
    - CWND <- CWND/2

Remains in congestion avoidance after fast retransmission

Time Diagram



Slow-start restart: Go back to CWND of 1 MSS, but take advantage of knowing the previous value of CWND.

TCP Flavors

- TCP Tahoe
  - CWND = 1 on triple dupACK
- TCP Reno
  - CWND = 1 on timeout
  - CWND = CWND/2 on triple dupACK
- TCP-newReno
  - TCP-Reno + improved fast recovery
- TCP-SACK
  - Incorporates selective acknowledgements

Our default assumption

Any Questions?

TCP and fairness guarantees

Consider A Simple Model

- Flows **ask** for an amount of bandwidth r_i
    - In reality, this request is implicit (the amount they send)
- The link gives them an amount a_i
    - Again, this is implicit (by how much is forwarded)
    - $a_i <= r_i$
- There is some total capacity C
    - Sum $a_i <= C$

Fairness

- When all flows want the same rate, fair is easy
  - Fair share = C/N
  - C = capacity of link
  - N = number of flows
- Note:
  - This is fair share per link. This is not a global fair share
- When not all flows have the same demand?
  - What happens here?

Example 1

- Requests: r_i Allocations: a_i
- $C = 20$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- Solution
  - $a_1 = 6$, $a_2 = 5$, $a_3 = 4$
- When bandwidth is plentiful, everyone gets their request
- This is the easy case

Example 2

- Requests: r_i Allocations: a_i
- $C = 12$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- One solution
  - $a_1 = 4$, $a_2 = 4$, $a_3 = 4$
  - Everyone gets the same
- Why not proportional to their demands?
  - $a_i = (12/15)\ r_i$
- Asking for more gets you more!
  - Not incentive compatible (i.e., cheating works!)
  - You can’t have that and invite innovation!

Example 3

- Requests: r_i Allocations: a_i
- $C = 14$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- $a_3 = 4$ (can't give more than a flow wants)
- Remaining bandwidth is 10, with demands 6 and 5
  - From previous example, if both want more than their share, they both get half
  - $a_1 = a_2 = 5$

Max-Min Fairness

- Given a set of bandwidth demands r_i and total bandwidth C, max-min bandwidth allocations are $a_i = \min(f, r_i)$
    - Where f is the unique value such that $\mathrm{Sum}(a_i) = C$ or set f to be infinite if no such value exists
- **This is what round-robin service gives**
    - If all packets are MTU
- Property:
    - If you don't get full demand, no one gets more than you

Computing Max-Min Fairness

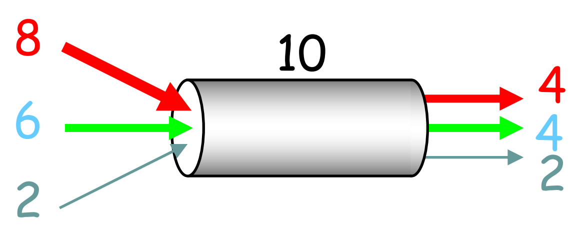
- Assume demands are in increasing order…
- If $C/N <= r_1$, then $a_i = C/N$ for all i
- Else, a_1 = r1, set $C = C - a_1$ and $N = N-1$
- Repeat
- Intuition: all flows requesting less than fair share get their request. Remaining flows divide equally

Example

- Assume link speed C is 10Mbps
- Have three flows:
  - Flow 1 is sending at a rate 8 Mbps
  - Flow 2 is sending at a rate 6 Mbps
  - Flow 3 is sending at a rate 2 Mbps
- How much bandwidth should each get?
  - According to max-min fairness?
- Work this out, talk to your neighbors

Example

- Requests: r_i Allocations: a_i
- Requests: $r_1 = 8$, $r_2 = 6$, $r_3 = 2$
- $C = 10$, $N = 3$, $C/N = 3.33$
  - Can serve all for r_3
  - Remove r_3 from the accounting: $C = C - r_3 = 8$, $N = 2$
- $C/2 = 4$
  - Can't service all for r_1 or r_2
  - So hold them to the remaining fair share: $f = 4$

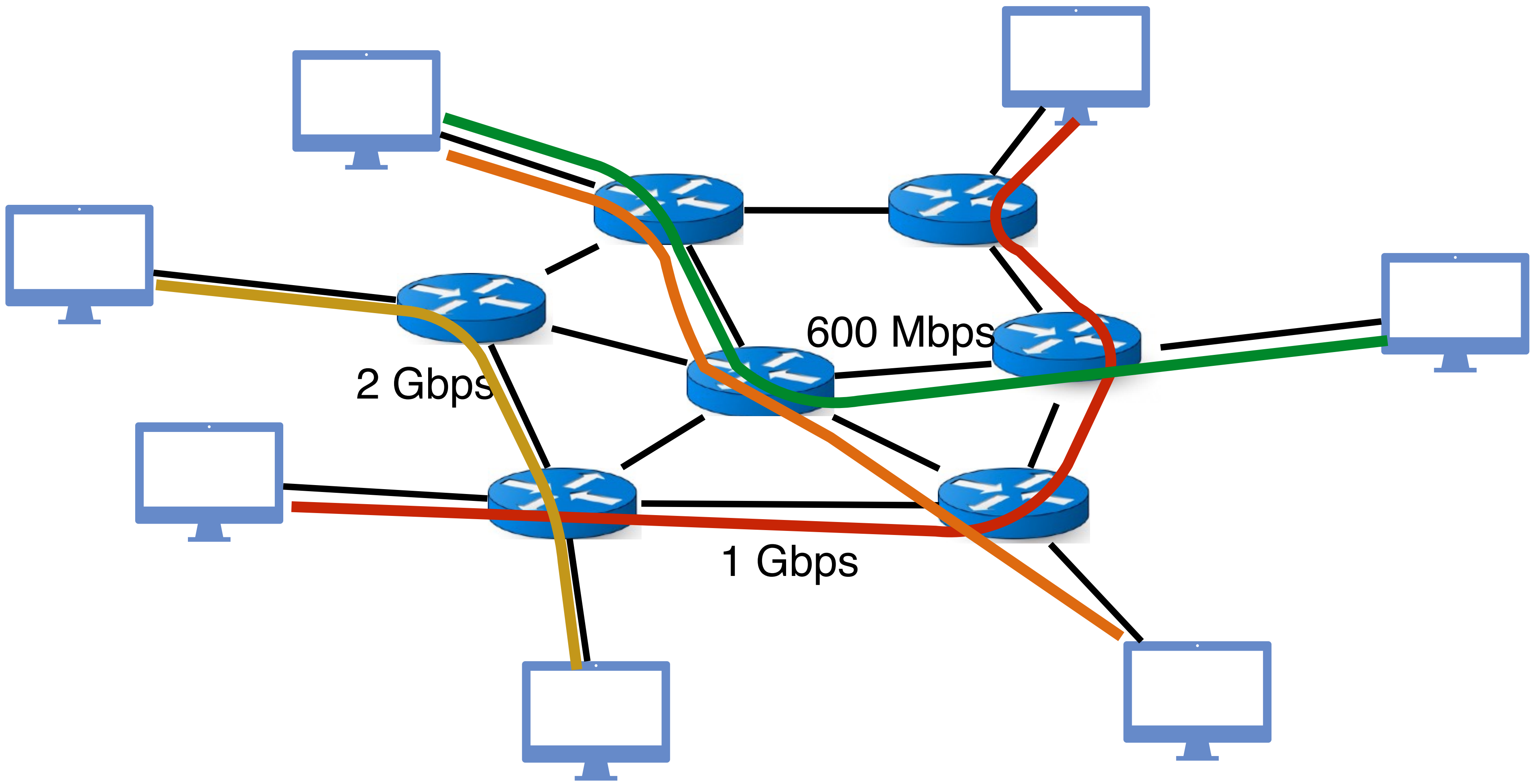


$f = 4$:
min(8, 4) = 4
min(6, 4) = 4
min(2, 4) = 2

Max-Min Fairness

- Max-min fairness the natural per-link fairness
- Only one that is
  - Symmetric
  - Incentive compatible (asking for more doesn't help)

Reality of Congestion Control



Congestion control is a resource allocation problem involving many flows, many links and complicated global dynamics

Classical result:

In a stable state
(no dynamics; all flows are infinitely long; no failures; etc.)

TCP guarantees max-min fairness

Any Questions?

The Many Failings of TCP Congestion Control

1. Fills up queues (large queueing delays)
2. Every segment not ACKed is a loss (non-congestion related losses)
3. Produces irregular saw-tooth behavior
4. Biased against long RTTs (unfair)
5. Not designed for short flows
6. Easy to cheat

(1) TCP Fills Up Queues

- TCP only slows down when queues fill up
    - High queueing delays
- Means that it is not optimized for latency
    - What is it optimized for then?
        - **Answer: Fairness (discussion in next few slides)**
- And many packets are dropped when buffer fills
- Alternative 1: Use small buffers
    - Is this a good idea?
    - Answer: No, bursty traffic will lead to reduced utilization
- Alternative: **Random Early Drop (RED)**
    - Drop packets on purpose **before** queue is full
    - A very clever idea

Random Early Drop (or Detection)

- Measure average queue size A with exponential weighting
  - Average: Allows for short bursts of packets without over-reacting
- Drop probability is a function of A
  - No drops if A is very small
  - Low drop rate for moderate A's
  - Drop everything if A is too big
- Drop probability applied to incoming packets
- Intuition: link is fully utilized well before buffer is full

Advantages of RED

- Keeps queues smaller, while allowing bursts
  - Just using small buffers in routers can’t do the latter
- Reduces synchronization between flows
  - Not all flows are dropping packets at once
  - Increases/decreases are more gentle
- Problem
  - Turns out that RED does not guarantee fairness

(2) Non-Congestion-Related Losses?

- For instance, RED drops packets intentionally
    - TCP would think the network is congested
- Can use **Explicit Congestion Notification (ECN)**
- Bit in IP packet header (actually two)
    - TCP receiver returns this bit in ACK
- When RED router would drop, it sets bit instead
    - Congestion semantics of bit exactly like that of drop
- Advantages
    - Doesn't confuse corruption with congestion

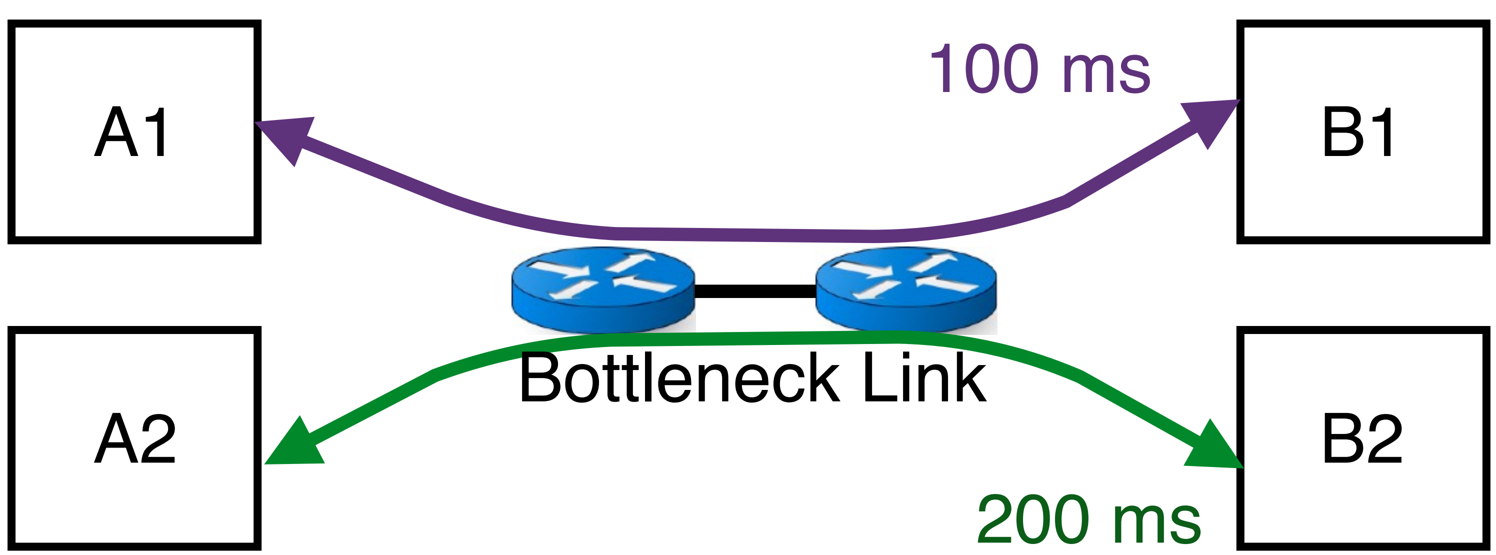
(3) Sawtooth Behavior Uneven

- TCP throughput is “choppy"
  - Repeated swings between W/2 to W
- Some apps would prefer sending at a steady rate
  - E.g., streaming apps
- A solution: “Equation-based congestion control”
  - Ditch TCP’s increase/decrease rules and just follow the equation:
  - **[Matthew Mathis, 1997] TCP Throughput = MSS/RTT sqrt(3/2p)**
    - **Where p is drop rate**
  - Measure drop percentage p and set rate accordingly
- Following the TCP equation ensures we’re TCP friendly
  - I.e., use no more than TCP does in similar setting

Any Questions?

(4) Bias Against Long RTTs

- Flows get throughput inversely proportional to RTT
- **TCP unfair in the face of heterogeneous RTTs!**
- [Matthew Mathis, 1997] TCP Throughput = MSS/RTT sqrt(3/2p)
    - Where p is drop rate
- Flows with long RTT will achieve lower throughput



Possible Solutions

- Make additive constant proportional to RTT
- But people don’t really care about this…

(5) How Short Flows Fare?

- Internet traffic:
  - Elephant and mice flows
  - Elephant flows carry most bytes (>95%), but are very few (<5%)
  - Mice flows carry very few bytes, but most flows are mice
    - 50% of flows have < 1500B to send (1 MTU);
    - 80% of flows have < 100KB to send
- Problem with TCP?
  - Mice flows do not have enough packets for duplicate ACKs!!
  - Drop ~=~ Timeout (unnecessary high latency)
  - These are precisely the flows for which latency matters!!!
- Another problem:
  - Starting with small window size leads to high latency

Possible Solutions?

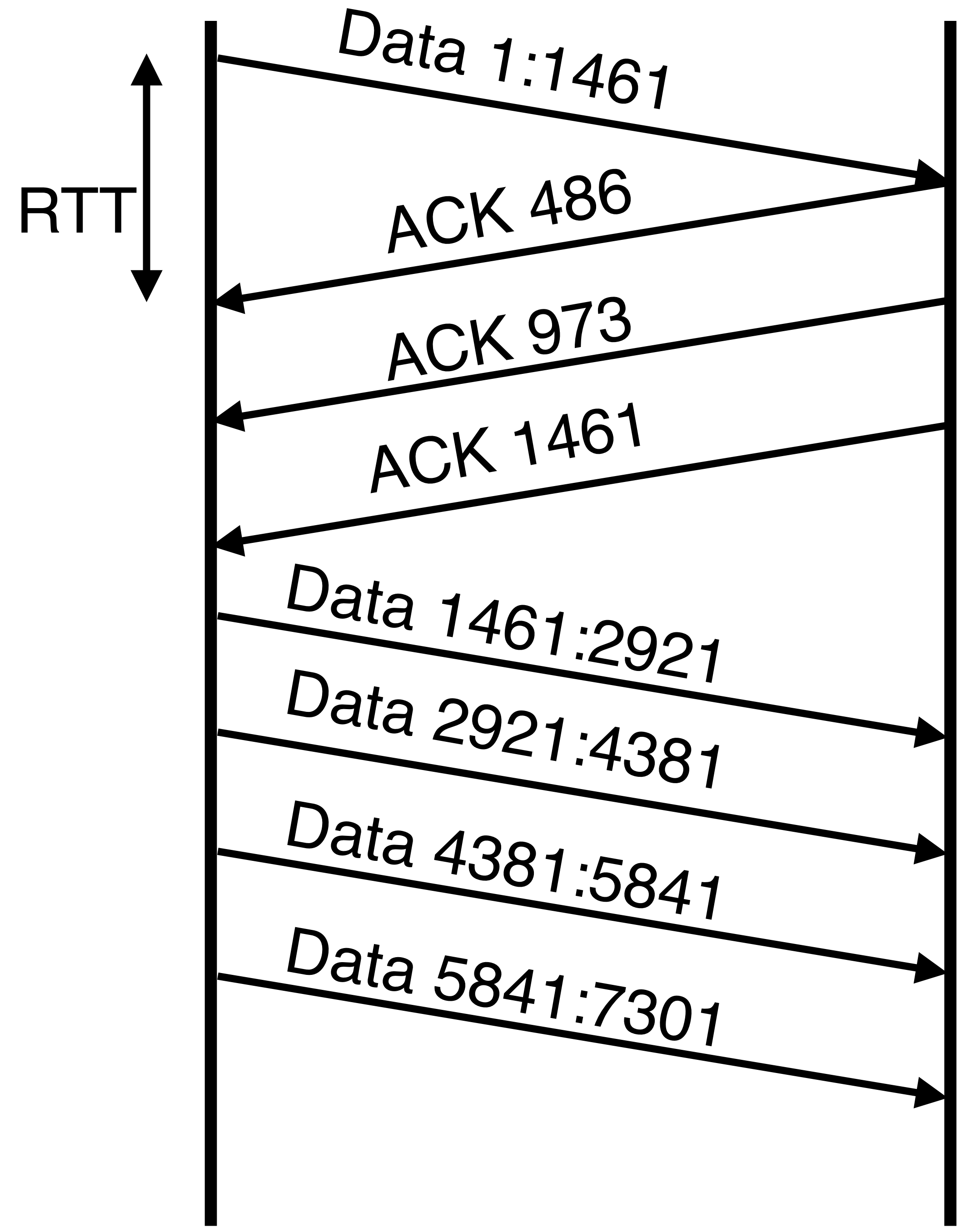
- Larger initial window?
  - Google proposed moving from ~4KB to ~15KB
  - Covers ~90% of HTTP Web
  - Decreases delay by 5%
- Many recent research papers on the timeout problem
  - Require network support

(6) Cheating

- TCP was designed assuming a cooperative world
- No attempt was made to prevent cheating
- Many ways to cheat, will present three

Cheating #1: ACK-splitting (receiver)

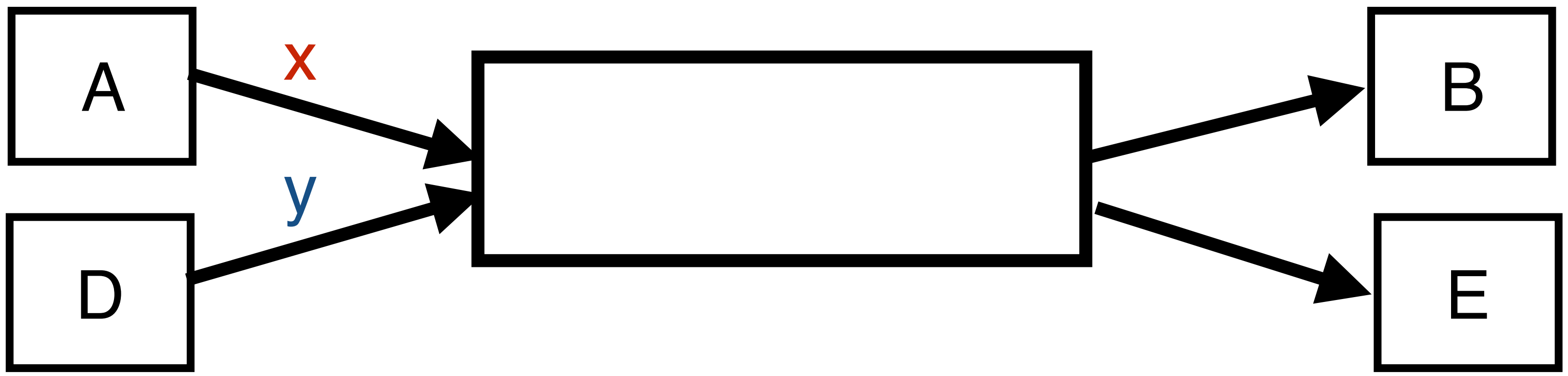
- TCP Rule: grow window by one MSS for each valid ACK received
- Send **M** (distinct) ACKs for one MSS
- Growth factor proportional to **M**



Cheating #2: Increasing CWND Faster (source)

- TCP Rule: increase window by one MSS for each valid ACK received
- Increase window by **M** per ACK
- Growth factor proportional to **M**

Cheating #3: Open Many Connections (source/receiver)



- Assume
  - A start 10 connections to B
  - D starts 1 connection to E
  - Each connection gets about the same throughput
- Then A gets 10 times more throughput than D

Cheating

- Either sender or receiver can independently cheat!
- **Why hasn't Internet suffered congestion collapse yet?**
  - Individuals don't hack TCP (not worth it)
  - Companies need to avoid TCP wars
- How can we prevent cheating
  - Verify TCP implementations
  - Controlling end points is hopeless
- Nobody cares, really

Any Questions?

How Do You Solve These Problems?

- Bias against long RTTs
- Slow to ramp up (for short-flows)
- Cheating
- Need for uniformity