CS4450

Computer Networks:
Architecture and Protocols

Lecture 23
Transport Layer Wrap Up

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Announcements

- **Finals on 18th and 19th**
    - Cumulative
    - Same format as prelims
- **Prelim almost graded**
    - Grades definitely released 24 hours before S/U deadline (Tuesday)
- **Last lecture next Tuesday**
    - Please plan to attend
    - I will talk about networking trends and grand challenges

Recap

Recap: Four Goals for Reliable Transfer

- **Correctness**
    - As defined
- **“Fairness”**
    - Every flow must get a fair share of network resources
- **Flow Performance**
    - Latency, jitter, etc.
- **Utilization**
    - Would like to maximize bandwidth utilization
    - If network has bandwidth available, flows should be able to use it!

Recap: Complete Correctness Condition

A transport mechanism is "reliable" if and only if

(a) It resends all dropped or corrupted packets

(b) It attempts to make progress

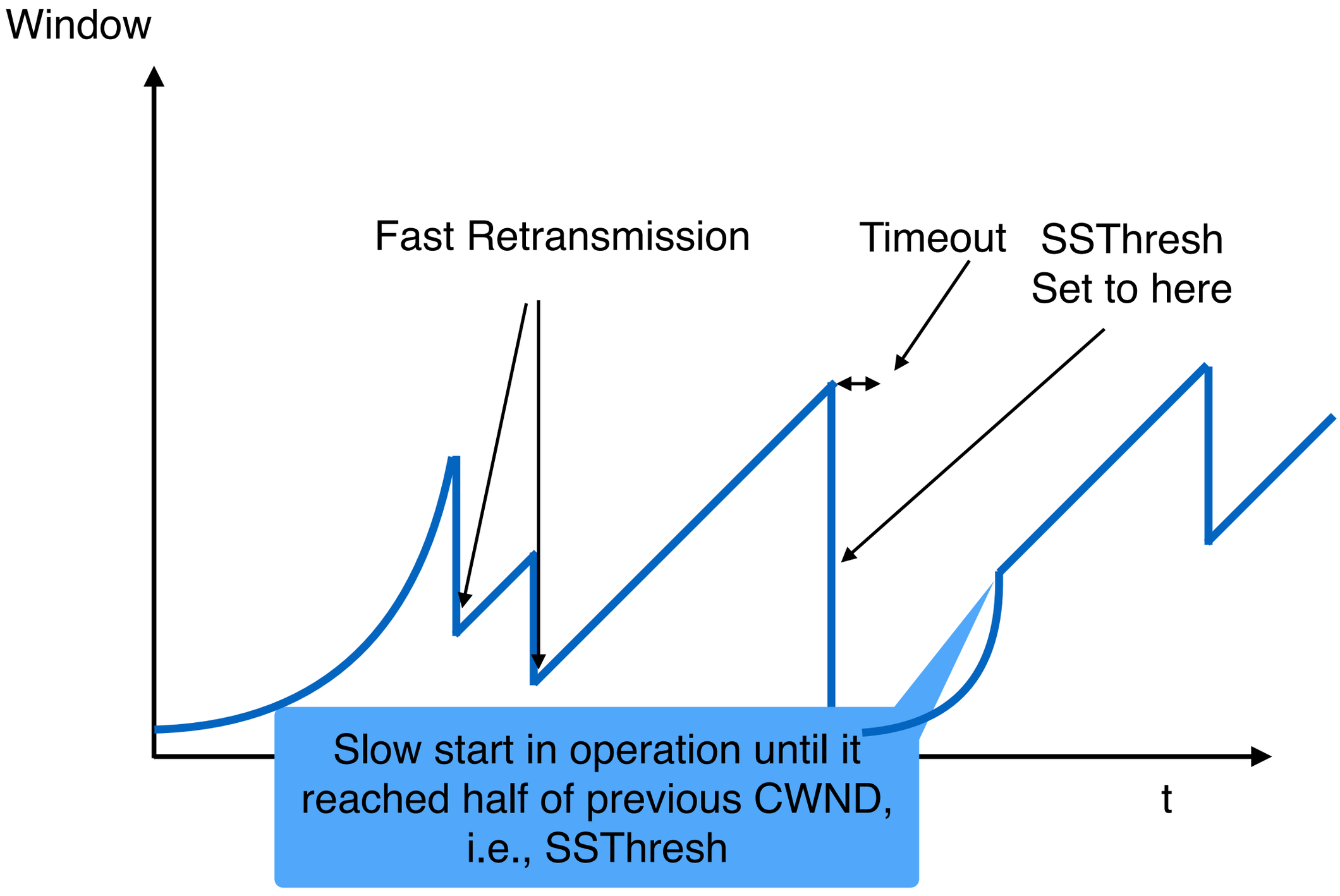
Recap: WHYs behind Transport design

- Started from first principles
    - Correctness condition for reliable transport
- … to understanding **why feedback from receiver is necessary** (sol-v1)
- … to understanding **why timers may be needed** (sol-v2)
- … to understanding **why window-based design may be needed** (sol-v3)
- … to understanding **why cumulative ACKs may be a good idea**
    - Very close to modern TCP

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: Final Time Diagram



One Final Phase: Fast Recovery

- The problem: congestion avoidance too slow in recovering from an isolated loss
- This last feature improves performance, but is not conceptually important

Example

- Consider a TCP connection with:
    - CWND = 10 packets
    - Last ACK was for packet # 101
        - i.e., receiver expecting next packet to have seq no 101
- 10 packets [101, 102, 103, …, 110] are in flight
    - Packet 101 is dropped
    - What ACKs do they generate?
    - And how does the sender respond?

Timeline

- ACK 101(due to 102) CWND = 10 dupACK #1 (no xmit)
- ACK 101(due to 103) CWND = 10 dupACK #2 (no xmit)
- ACK 101(due to 104) CWND = 10 dupACK #3 (no xmit)
- RETRANSMIT 101 ssthresh = 5 CWND = 5
- ACK 101 (due to 105) CWND=5 (no xmit)
- ACK 101 (due to 106) CWND=5 (no xmit)
- ACK 101 (due to 107) CWND=5 (no xmit)
- ACK 101 (due to 108) CWND=5 (no xmit)
- ACK 101 (due to 109) CWND=5 (no xmit)
- ACK 101 (due to 110) CWND=5 (no xmit)
- ACK 111 (due to 101)<- only now can we transmit new packets
- Plus no packets in flight so no ACKs for another RTT

Note that you do not restart dupACKcounter on same packet!

Solution: Fast Recovery

- Idea: Grant the sender temporary “credit” for each dupACK so as to keep packets in flight (each ACK due to arriving pkt)
- If dupACKcount = 3
    - ssthresh = CWND / 2
    - CWND = ssthresh + 3
- While in fast recovery
    - CWND = CWND + 1 for each additional duplicate ACK
- Exit fast recovery after receiving **new** ACK
    - Set CWND = ssthresh (which had been set to CWND/2 after loss)

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- RETRANSMIT 101 ssthresh = 5 CWND = 8 (5 + 3)
- ACK 101 (due to 105) CWND=9 (no xmit)
- ACK 101 (due to 106) CWND=10 (no xmit)
- ACK 101 (due to 107) CWND=11 (xmit 111)
- ACK 101 (due to 108) CWND=12 (xmit 112)
- ACK 101 (due to 109) CWND=13 (xmit 113)
- ACK 101 (due to 110) CWND=14 (xmit 114)
- ACK 111 (due to 101) CWND = 5 (xmit 115) <- exiting fast recovery
- Packets 111-114 already in flight (and now sending 115)
- ACK 112 (due to 111) CWND = 5 + 1/5 <- back to congestion avoidance

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 $\text{Sum}(a_i) = C$ 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.

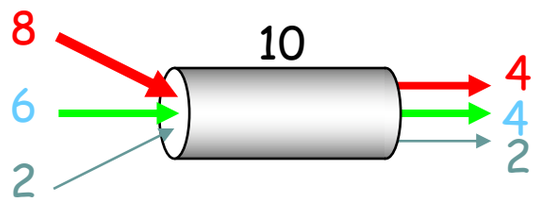
Remaining flows divide equally

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

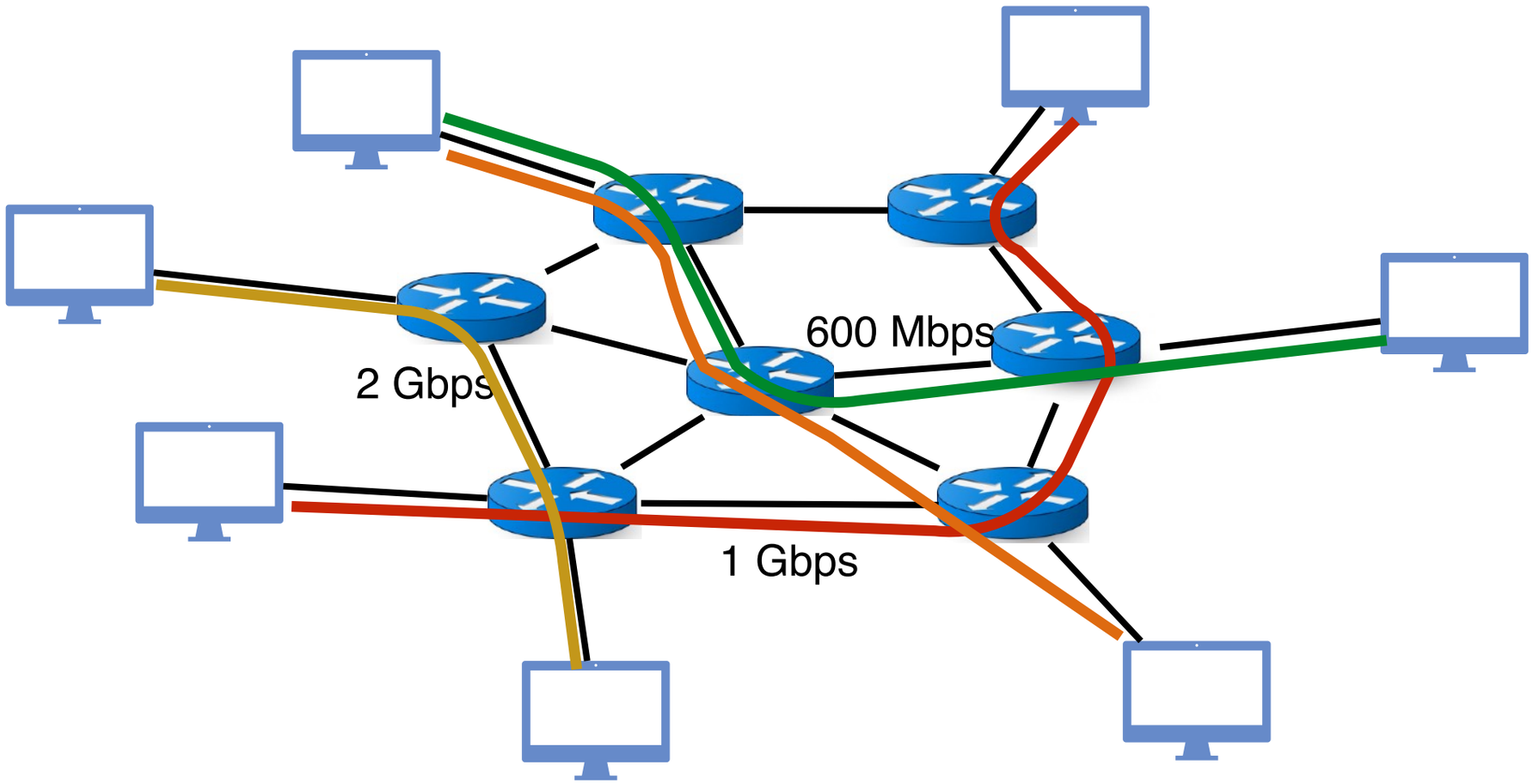


$f = 4$:
min(8, 4) = 4
min(6, 4) = 4
min(2, 4) = 2

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**
- 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

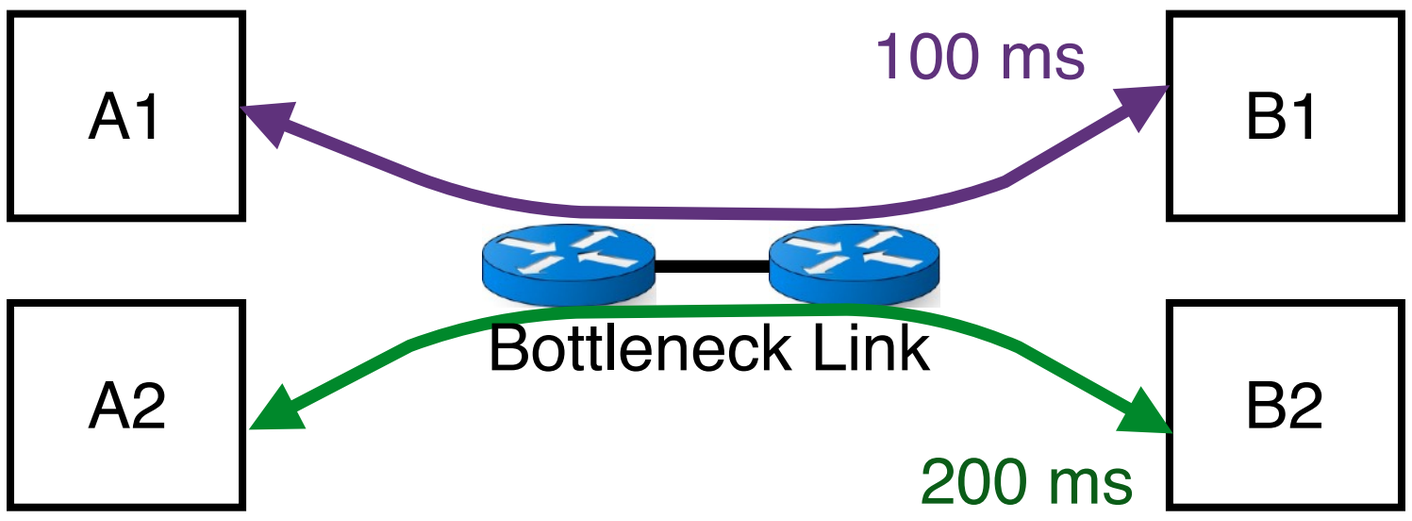
(3) Sawtooth Behavior Uneven

- 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?

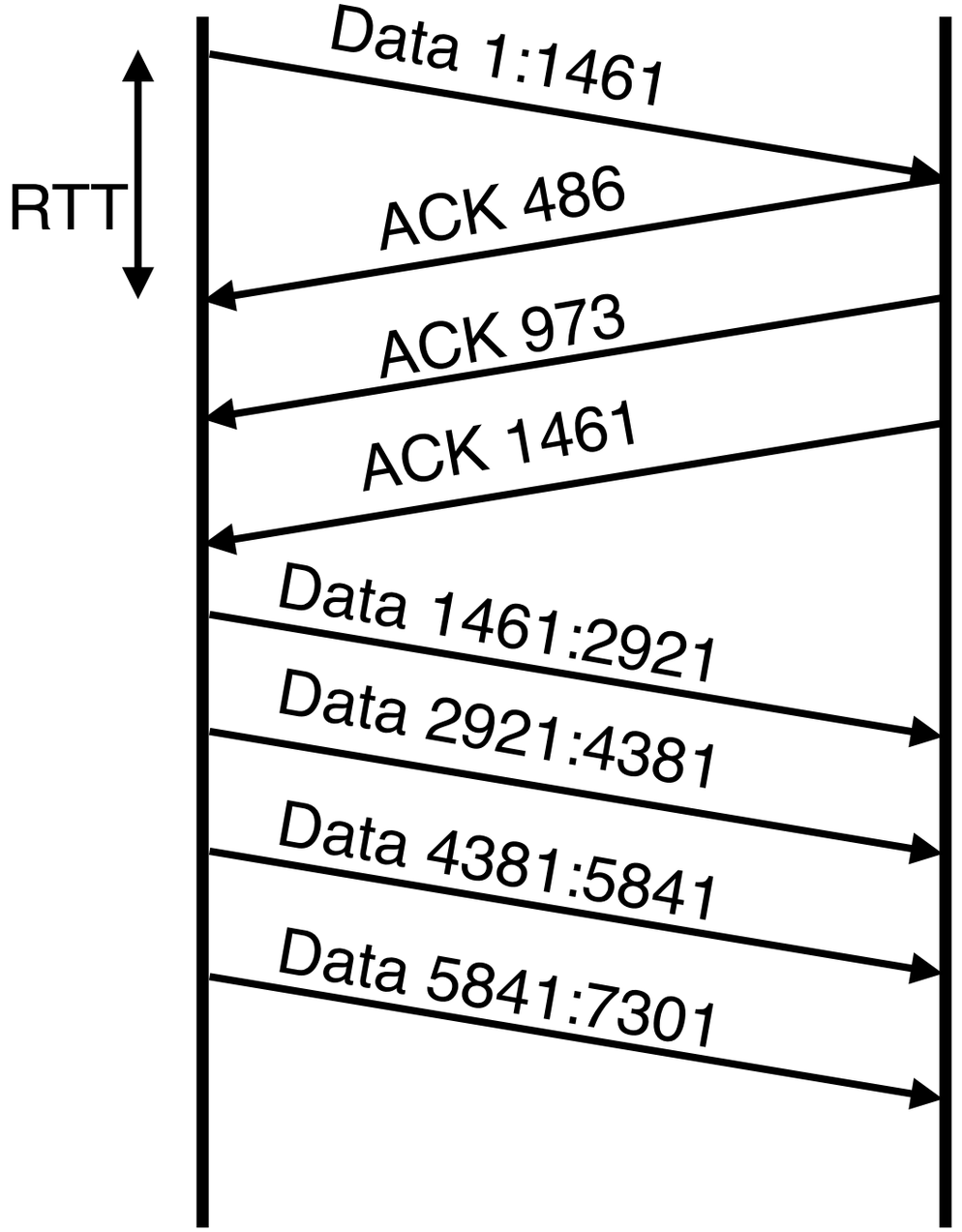
- Larger initial window?
    - Google proposed moving from ~4KB to ~15KB
    - Covers ~90% of HTTP Web
    - Decreases delay by 5%
- Many recent research papers on the timeout problem
    - Require network support

(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)

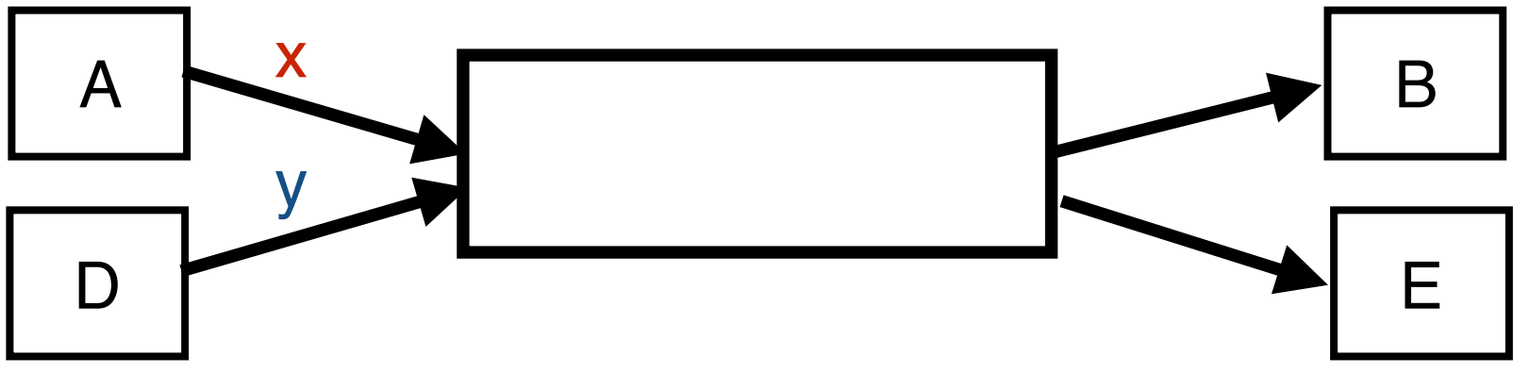
- TCP Rule: grow window by one MSS for each valid ACK received
- Send **M** (distinct) ACKs for one MSS
- Growth factor proportional to **M**



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

End of Transport Layer

Now YOU know as much as I know :-)