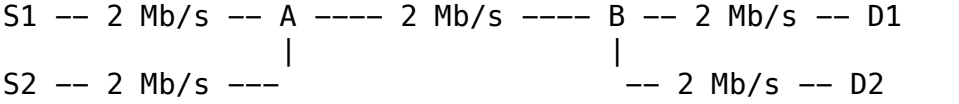
6.02 - Network: Reliable Transport and Congestion Control
Lecture #10
Katrina LaCurts, lacurts@mit.edu

0. Introduction
  - Last week: how to route scalably in the face of policy and economy
  - This week: how to transport scalably in the face of diverse application demands

1. TCP
  - Goals: provide reliable transport, prevent congestion
  - Broader questions: how do we do this scalably, and how do we share the network efficiently and fairly?
  - Today: TCP Congestion Control
    - In particular, a version of TCP known as "New Reno"
  - Next lecture: An alternative approach to "resource management" on the Internet

2. Reliable Transport via sliding-window protocol
  - Goal: receiving application gets a complete, in-order bytestream from the sender. One copy of every packet, in order.
  - Why do we need it? Network is unreliable. Packets get dropped, can arrive out-of-order.
  - Basics:
    - Every data packet gets a sequence number (1, 2, 3, ...)
    - Sender has W outstanding packets at any given time. W = window size
    - When receive gets a packet, it sends an ACK back. ACKs are cumulative: An ACK for X indicates "I have received all packets up to and including X."
    - If sender doesn't receive an ACK indicating that packet X has been received, after some amount of time it will "timeout" and retransmit X.
      - Maybe X was lost, its ACK was lost, or its ACK is delayed
      - The timeout = proportional to (but a bit larger than) the RTT of the path between sender and receiver
    - At receiver: keep buffer to avoid delivering out-of-order packets, keep track of last-packet-delivered to avoid delivering duplicates.

3. Main motivation
  - What's the "right" value for W?
  - In particular, what if there are multiple senders?
    - Ex:



    - What should happen? Debatable. Reasonable alternative:

```
S1 -- 1 Mb/s -- A ---- 2 Mb/s ---- B -- 1 Mb/s -- D1
                  |                 |
S2 -- 1 Mb/s ---                     -- 1 Mb/s -- D2
```

   - How do S1 and S2 figure this out? What happens if S3 arrives? Or if S1 starts sending less? Etc.

5. Congestion Control: controlling the source rate to achieve high performance
   - Goals: Efficiency and fairness
     - Minimize drops, minimize delay, maximize utilization
     - Share bandwidth fairly among all connections that are using it
   - FOR NOW: assume all senders have infinite offered load. Fairness = splitting bandwidth equally amongst them.
   - But no senders knows how many other senders there are, and that number can change over time.
   - We'll use window-based congestion control. Switches are dumb (can only drop packets); senders are smart

6. AIMD
   - Need a signal for congestion in the network, so senders can react to it.
   - Our signal: packet drops
   - Every RTT:
     - If there is no loss, W = W+1
     - If there is loss, W = W/2
   - This is "Additive Increase Multiplicative Decrease" (AIMD)
   - Senders constantly readjust => adapt to a changing number of senders, or changing offered loads
   - Window size exhibits sawtooth behavior (see slides)
   - Why AIMD?
     - It's "safe": senders are conservative about increasing, but scale back dramatically in the face of congestion
     - Efficient and fair

7. Finite Offered Load
   - Remove the assumption that everyone has infinite offered load
   - Suppose S1 and S2 have offered load of 1Mb/s, S3 has offered load of .5Mb/s, and they all share a bottleneck with capacity 2Mb/s
   - What happens?
     - In theory: S3 stops increase once it's sending .5Mb/s. S1 and S2 continue increasing until they reach .75Mb/s
   - Is this fair?
     - In some sense. It achieves a type of fairness known as "max-min fairness". But there are other definitions (e.g., "proportional fairness")
   - What happens in practice?
     - We might get max-min fairness, or one of the senders might experience a much longer RTT and so not increase its window at

the same rate.
- So: TCP's congestion control utilizes the network reasonably well, but it's hard to measure fairness, or claim that fairness is achieved under skewed workloads, varying RTTs, etc.

8. Additional Mechanisms
- Slow Start
  - At the beginning of the connection, exponential increase the window (double it every RTT until you see loss)
  - Decreases the time it takes for the initial window to "ramp up"
  - (See slide for diagram)
- Fast Retransmit/Fast Recovery
  - When a sender receives an ACK with sequence number X, and then three duplicates of that packet, it immediately retransmits packet X+1 (remember: ACKs are cumulative)
    Ex: Send   1 2 3 4 5 6
        Receive 1 2   2 2 2
        Sender receives 4 ACKs total with sequence number "2"; infers that packet 3 is lost, immediately retransmits
  - On fast-retransmit, window decrease is as before: W = W/2
  - In fact, when a packet is lost due to timeout, TCP behaves differently: W = 1, then do slow-start until the last good window, and then start additive increase.
  - (See slide for diagram)
  - Reasoning: if there is a retransmission due to timeout, then there is significant loss in the network, and senders should back *way* off.

9. Reflection
- TCP has been a massive success, requires no changes to the Internet's infrastructure, is something endpoints can opt-in to, allows the network to be shared among tons of different users, all with different -- and changing -- types of traffic, in a distributed manner.
- BUT: TCP doesn't react to congestion until it's already happening. Is there something better we could do?