Congestion Control

Reading: Sections 6.1-6.4

COS 461: Computer Networks
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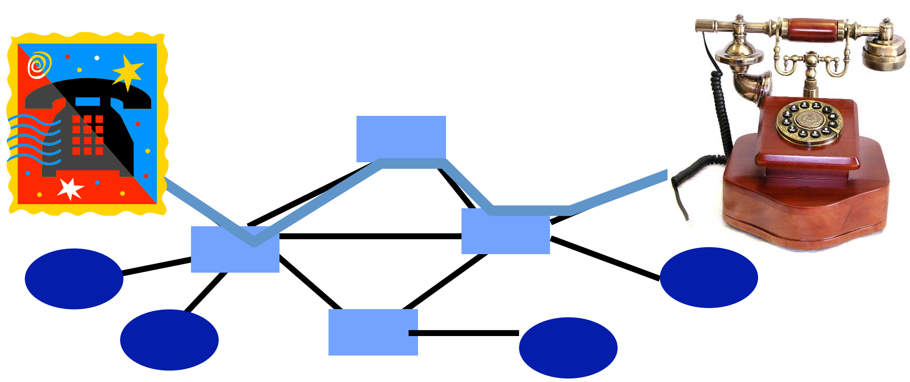
Goals of Today's Lecture

- Congestion in IP networks
  - Unavoidable due to best-effort service model
  - IP philosophy: decentralized control at end hosts
- Congestion control by the TCP senders
  - Infers congestion is occurring (e.g., from packet losses)
  - Slows down to alleviate congestion, for the greater good
- TCP congestion-control algorithm
  - Additive-increase, multiplicative-decrease
  - Slow start and slow-start restart

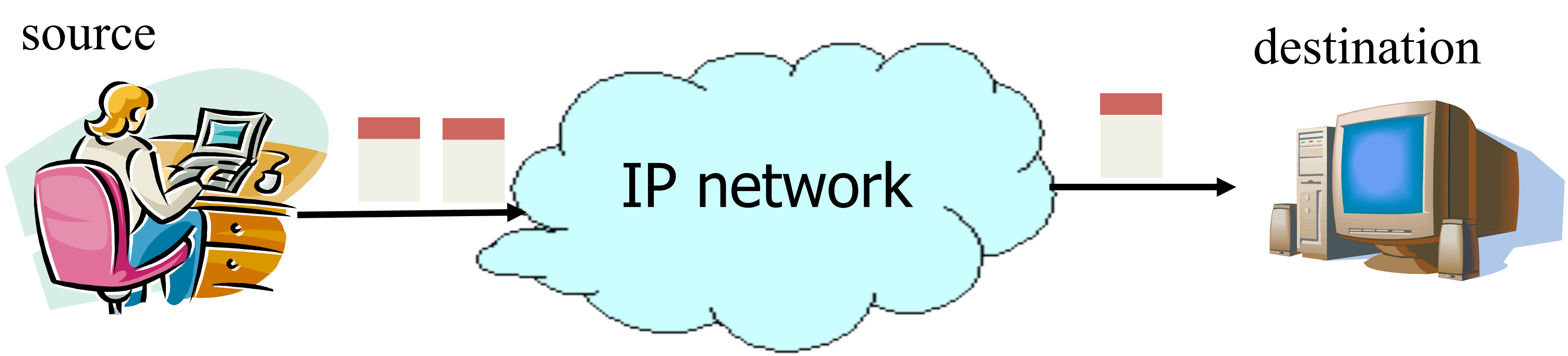
No Problem Under Circuit Switching

- Source establishes connection to destination
  - Nodes reserve resources for the connection
  - Circuit rejected if the resources aren't available
  - Cannot have more than the network can handle



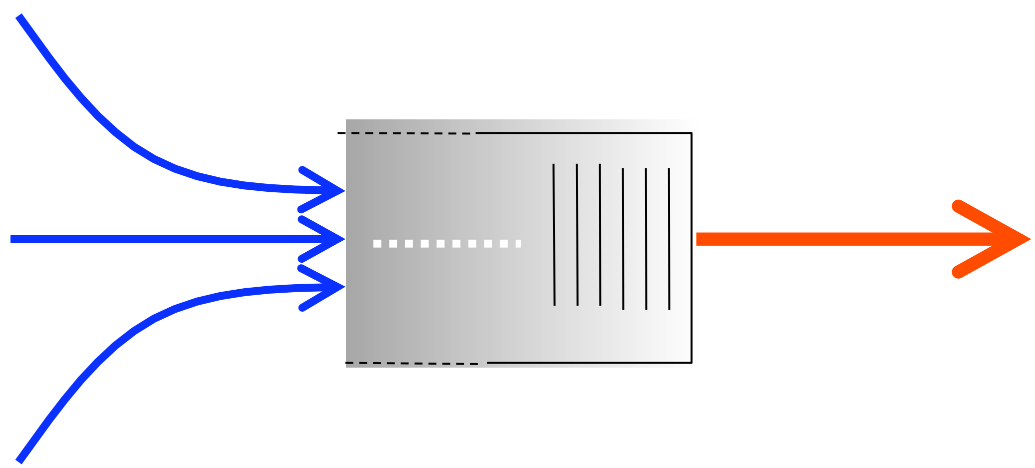
IP Best-Effort Design Philosophy

- Best-effort delivery
  - Let everybody send
  - Network tries to deliver what it can
  - … and just drop the rest



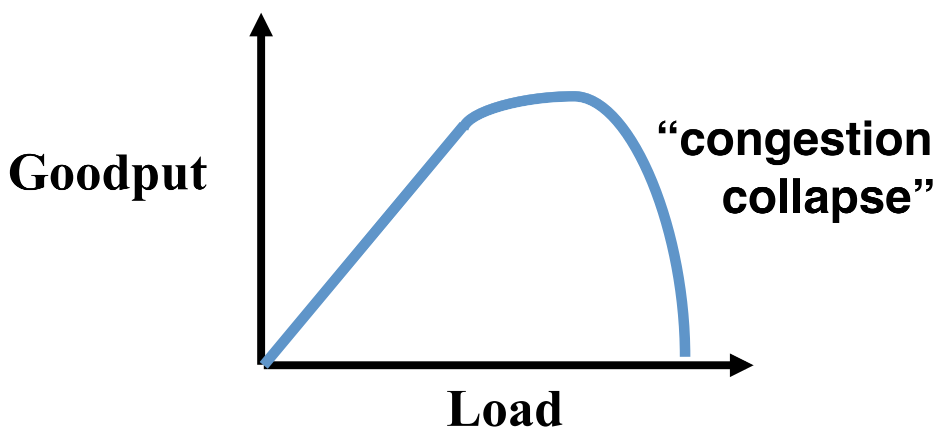
Congestion is Unavoidable

- Two packets arrive at same time
  - Router can only transmit one: must buffer or drop other
- If many packets arrive in short period of time
  - Router cannot keep up with the arriving traffic
  - Buffer may eventually overflow



The Problem of Congestion

- What is congestion? Load is higher than capacity
- What do IP routers do? Drop the excess packets
- Why bad? Wasted bandwidth for retransmissions



Increase in load that results in a *decrease* in useful work done

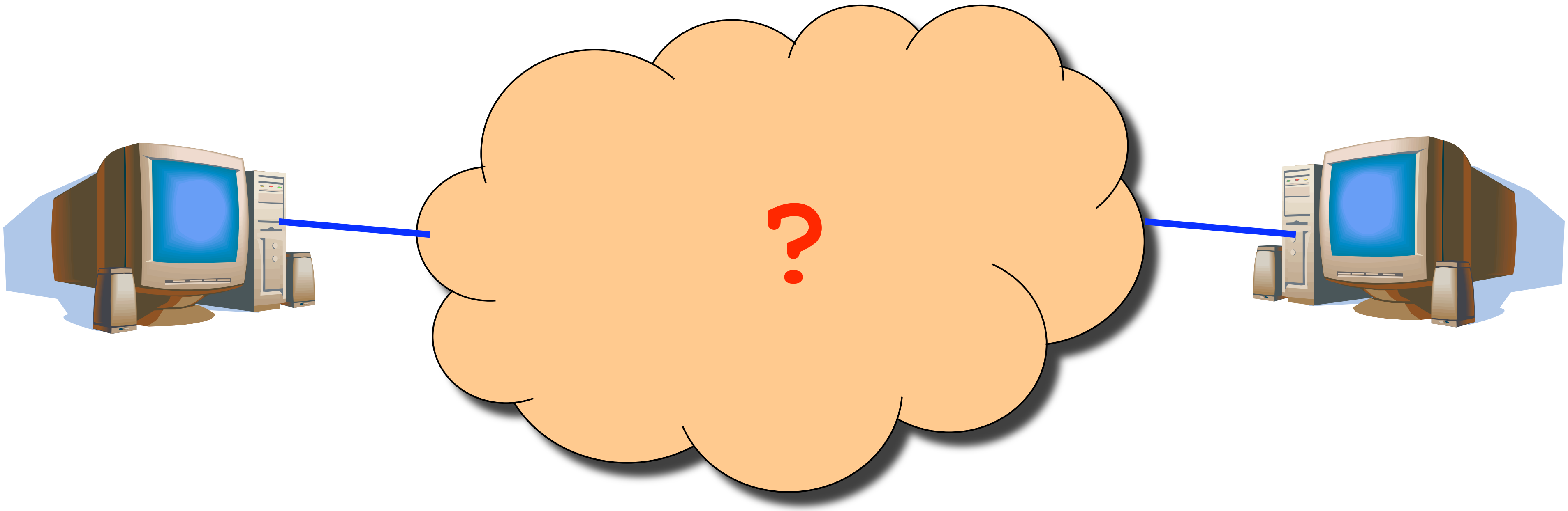
Ways to Deal With Congestion

- Ignore the problem
  - Many dropped (and retransmitted) packets
  - Can cause congestion collapse
- Reservations, like in circuit switching
  - Pre-arrange bandwidth allocations
  - Requires negotiation before sending packets
- Pricing
  - Don’t drop packets for the high-bidders
  - Requires a payment model, and low-bidders still dropped
- Dynamic adjustment (TCP)
  - Every sender infers the level of congestion
  - Each adapts its sending rate “for the greater good”

Many Important Questions

- How does the sender know there is congestion?
  - Explicit feedback from the network?
  - Inference based on network performance?
- How should the sender adapt?
  - Explicit sending rate computed by the network?
  - End host coordinates with other hosts?
  - End host thinks globally but acts locally?
- What is the performance objective?
  - Maximizing goodput, even if some users suffer more?
  - Fairness? (Whatever *that* means!)
- How fast should new TCP senders send?

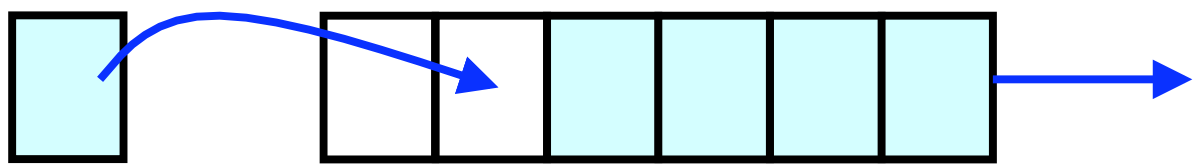
Inferring From Implicit Feedback



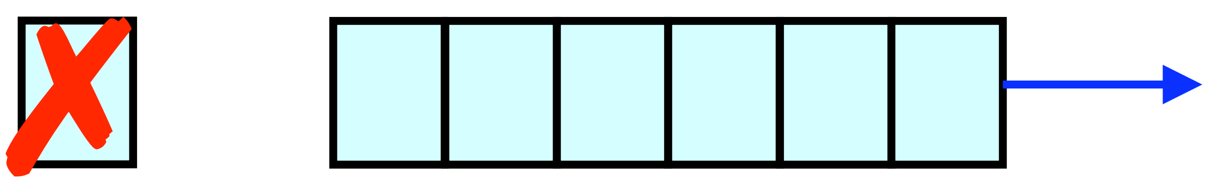
- What does the end host see?
- What can the end host change?

Where Congestion Happens: Links

- Simple resource allocation: FIFO queue & drop-tail
- Access to the bandwidth: first-in first-out queue
  - Packets transmitted in the order they arrive



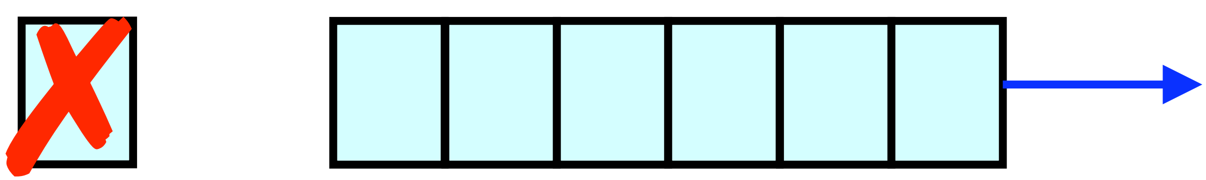
- Access to the buffer space: drop-tail queuing
  - If the queue is full, drop the incoming packet



How it Looks to the End Host

- Delay: Packet experiences high delay
- Loss: Packet gets dropped along path

- How does TCP sender learn this?
  - Delay: Round-trip time estimate
  - Loss: Timeout and/or duplicate acknowledgments



What Can the End Host Do?

- Upon detecting congestion (well, packet loss)
  - Decrease the sending rate
  - End host does its part to alleviate the congestion
- But, what if conditions change?
  - If bandwidth becomes available, unfortunate if host remains sending at low rate
- Upon *not* detecting congestion
  - Increase sending rate, a little at a time
  - And see if packets are successfully delivered

TCP Congestion Window

- Each TCP sender maintains a congestion window
  - Max number of bytes to have in transit (not yet ACK'd)

- Adapting the congestion window
  - Decrease upon losing a packet: backing off
  - Increase upon success: optimistically exploring
  - Always struggling to find right transfer rate

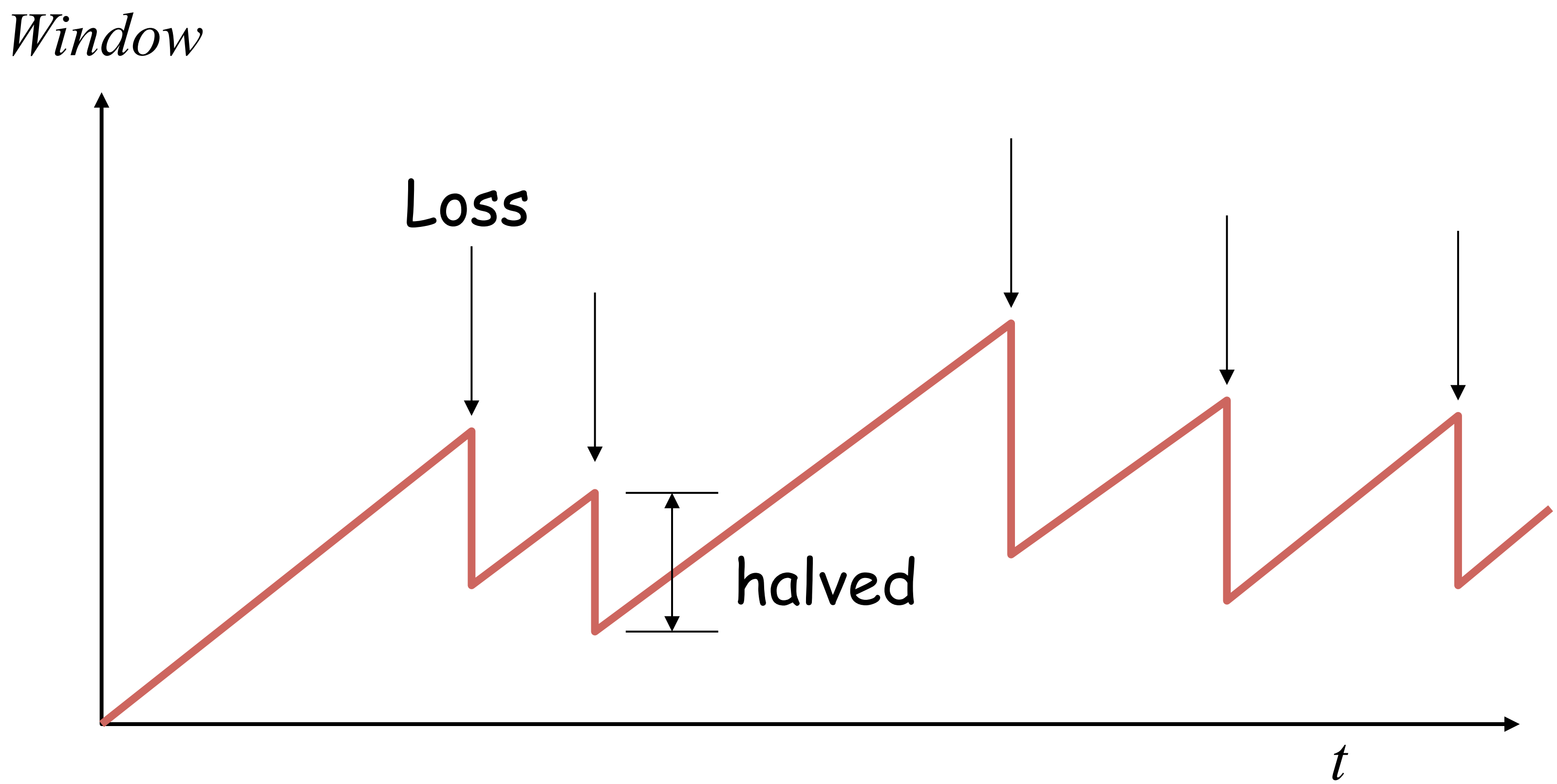
- Tradeoff
  - Pro: avoids needing explicit network feedback
  - Con: continually under- and over-shoots "right" rate

Additive Increase, Multiplicative Decrease (AIMD)

- How much to adapt?
  - Additive increase: On success of last window of data, increase window by 1 Max Segment Size (MSS)
  - Multiplicative decrease: On loss of packet, divide congestion window in half

- Much quicker to slow than speed up!
  - Over-sized windows (causing loss) are much worse than under-sized windows (causing lower thruput)
  - AIMD: A necessary condition for stability of TCP

Leads to the TCP “Sawtooth”



Receiver Window vs. Congestion Window

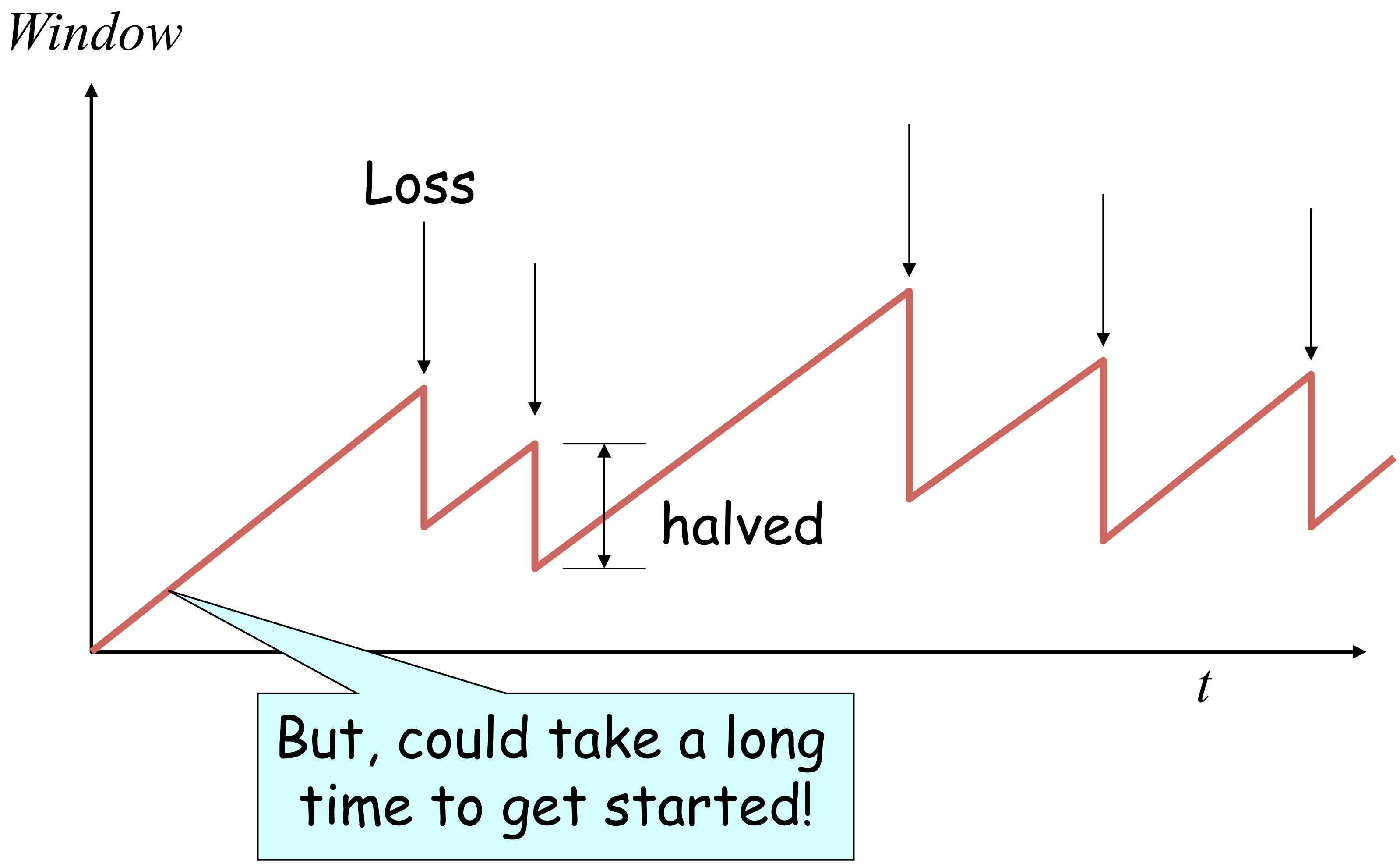
- Flow control
  - Keep a *fast sender* from overwhelming *a slow receiver*
- Congestion control
  - Keep a *set of senders* from overloading the *network*

- Different concepts, but similar mechanisms
  - TCP flow control: receiver window
  - TCP congestion control: congestion window
  - Sender TCP window =

    min { congestion window, receiver window }

How Should a New Flow Start?

Start slow (a small CWND) to avoid overloading network

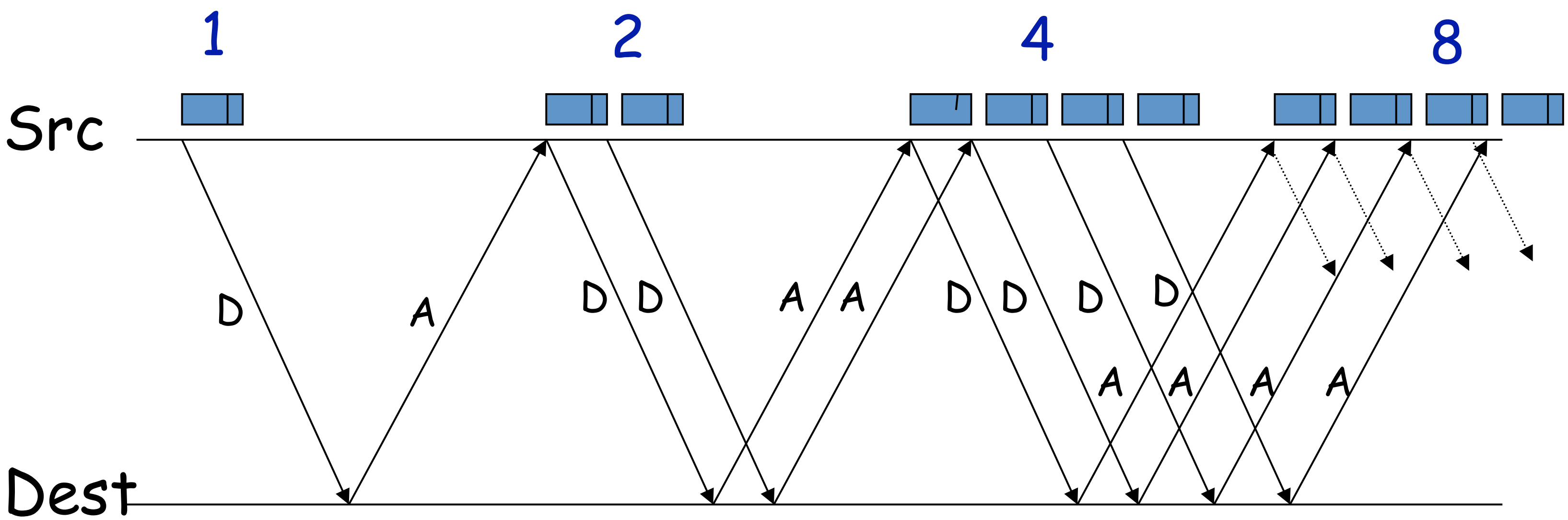


"Slow Start" Phase

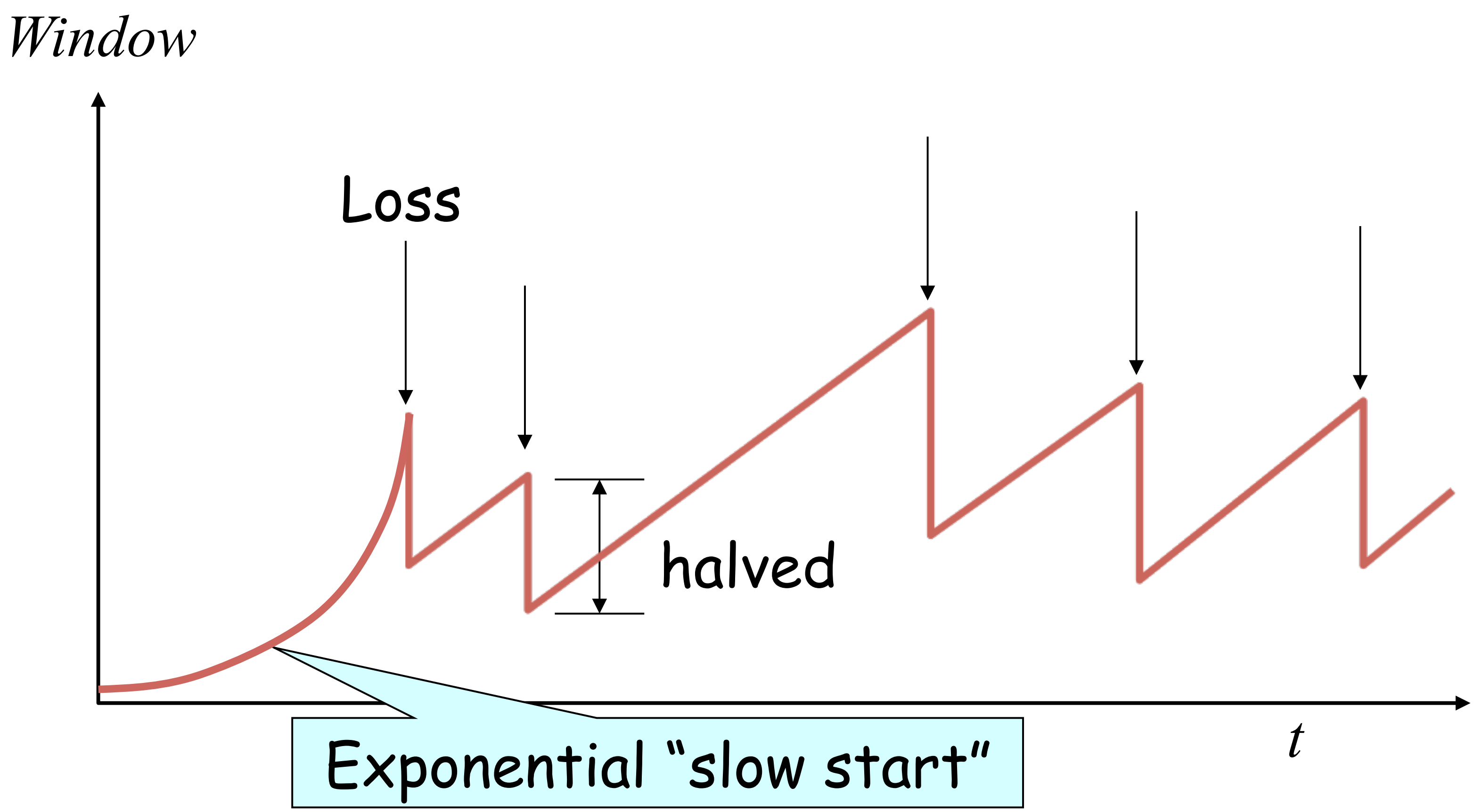
- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS / RTT
- Could be pretty wasteful
  - Might be much less than actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (really "fast start")
  - Sender starts at a slow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

Slow Start in Action

Double CWND per round-trip time



Slow Start and the TCP Sawtooth



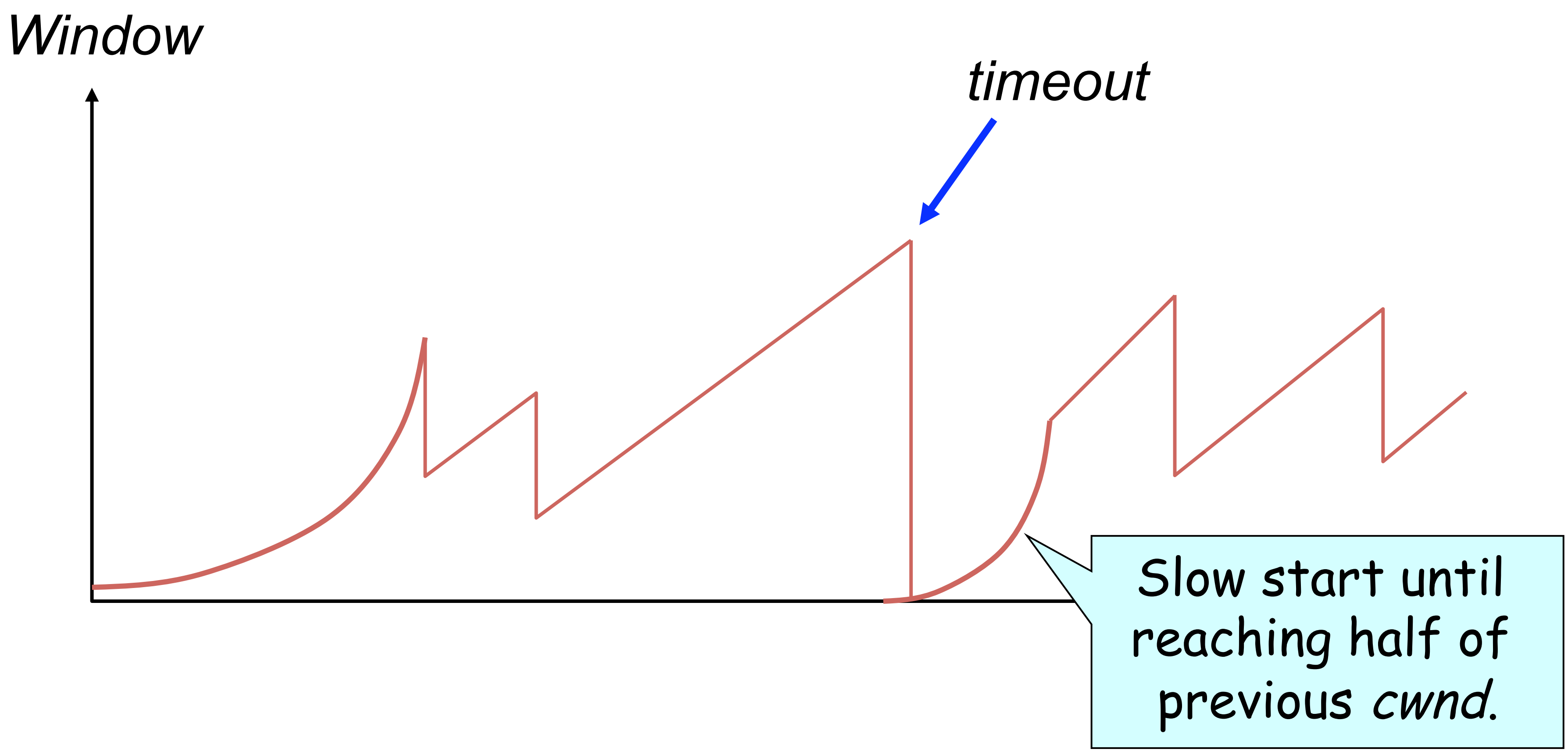
- So-called because TCP originally had no congestion control
  - Source would start by sending an entire receiver window
  - Led to congestion collapse!

Two Kinds of Loss in TCP

- Timeout
  - Packet n is lost and detected via a timeout
    - When? n is last packet in window, or all packets in flight lost
  - After timeout, blasting entire CWND would cause another burst
  - Better to start over with a low CWND

- Triple duplicate ACK
  - Packet n is lost, but packets n+1, n+2, etc. arrive
    - How detected? Multiple ACKs that receiver waiting for n
    - When? Later packets after n received
  - After triple duplicate ACK, sender quickly resends packet n
  - Do a multiplicative decrease and keep going

Repeating Slow Start After Timeout



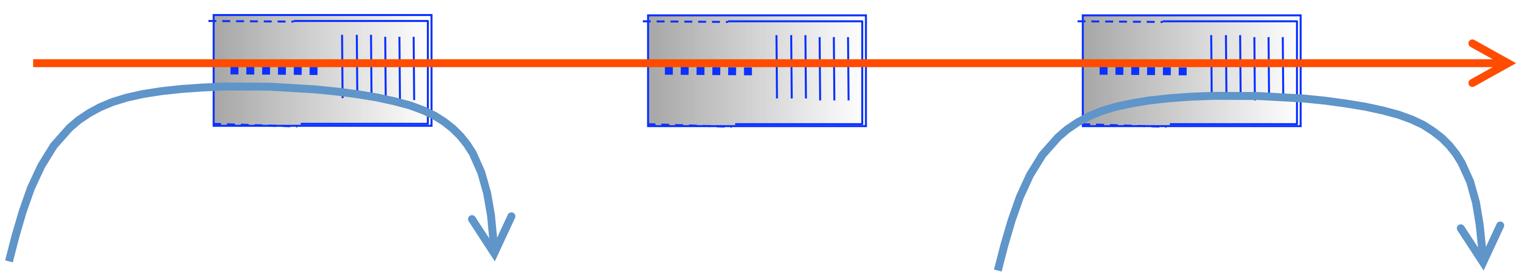
Slow-start restart: Go back to CWND of 1, but take advantage of knowing the previous value of CWND.

Repeating Slow Start After Idle Period

- Suppose a TCP connection goes idle for a while
- Eventually, the network conditions change
  - Maybe many more flows are traversing the link
- Dangerous to start transmitting at the old rate
  - Previously-idle TCP sender might blast network
  - … causing excessive congestion and packet loss
- So, some TCP implementations repeat slow start
  - Slow-start restart after an idle period

TCP Achieves Some Notion of Fairness

- Effective utilization is not only goal
  - We also want to be *fair* to various flows
  - … but what does *that* mean?
- Simple definition: equal shares of the bandwidth
  - N flows that each get 1/N of the bandwidth?
  - But, what if flows traverse different paths?
  - Result: bandwidth shared in proportion to RTT



What About Cheating?

- Some folks are more fair than others
  - Running multiple TCP connections in parallel (BitTorrent)
  - Modifying the TCP implementation in the OS
    - Some cloud services start TCP at > 1 MSS
  - Use the User Datagram Protocol
- What is the impact
  - Good guys slow down to make room for you
  - You get an unfair share of the bandwidth
- Possible solutions?
  - Routers detect cheating and drop excess packets?
  - Per user/customer failness?
  - Peer pressure?

Conclusions

- Congestion is inevitable
  - Internet does not reserve resources in advance
  - TCP actively tries to push the envelope
- Congestion can be handled
  - Additive increase, multiplicative decrease
  - Slow start and slow-start restart
- Active Queue Management can help
  - Random Early Detection (RED)
  - Explicit Congestion Notification (ECN)
- Fundamental tensions
  - Feedback from the network?
  - Enforcement of "TCP friendly" behavior?