Congestion Control

Reading: Sections 6.1-6.4

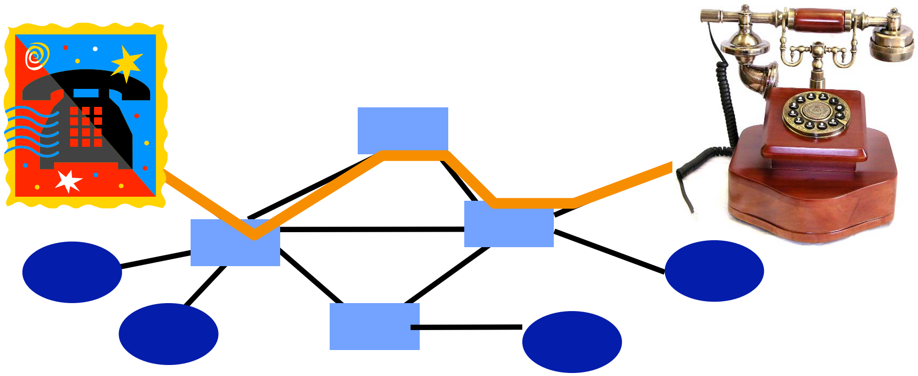
Acknowledgments: Lecture slides are from Computer networks course thought by Jennifer Rexford at Princeton University. When slides are obtained from other sources, a a reference will be noted on the bottom of that slide. A full list of references is provided on the last slide.

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
- Active Queue Management (AQM)
  - Random Early Detection (RED)
  - Explicit Congestion Notification (ECN)

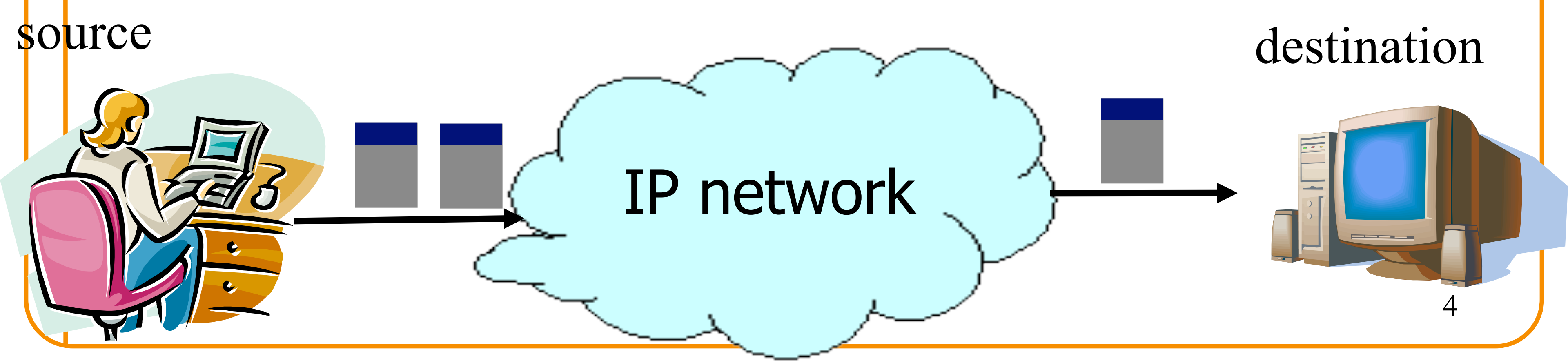
No Problem Under Circuit Switching

- 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



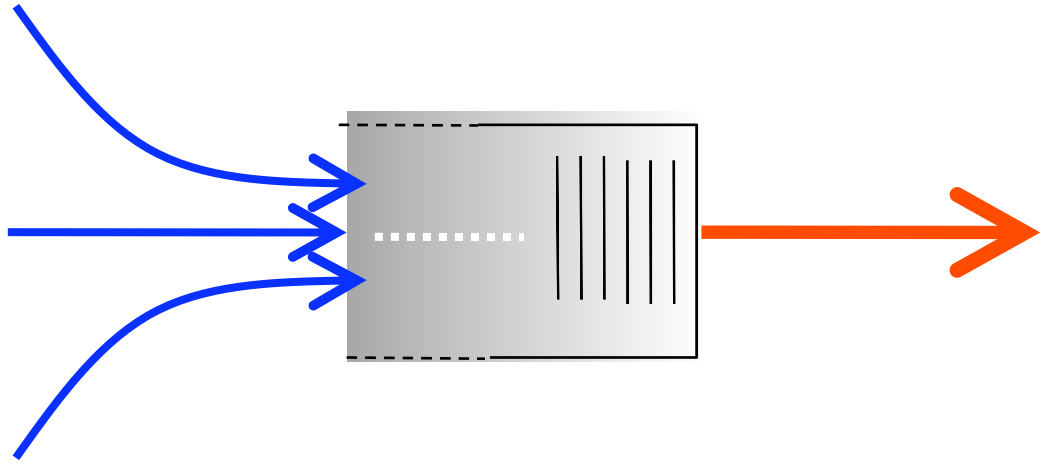
IP Best-Effort Design Philosophy

- Best-effort delivery
  - –Let everybody send
  - –Try to deliver what you can
  - –… and just drop the rest



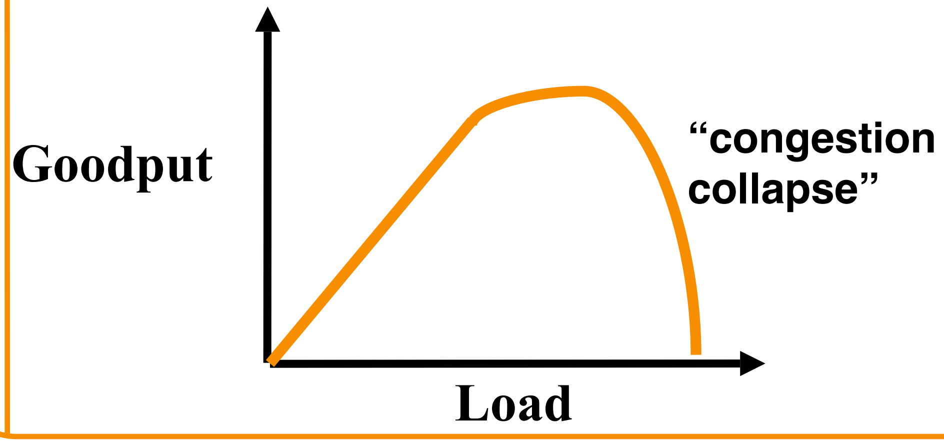
Congestion is Unavoidable

- Two packets arrive at the same time
  - The node can only transmit one
  - … and either buffer or drop the other
- If many packets arrive in short period of time
  - The node cannot keep up with the arriving traffic
  - … and the 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 is this bad?
  - Wasted bandwidth for retransmissions



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

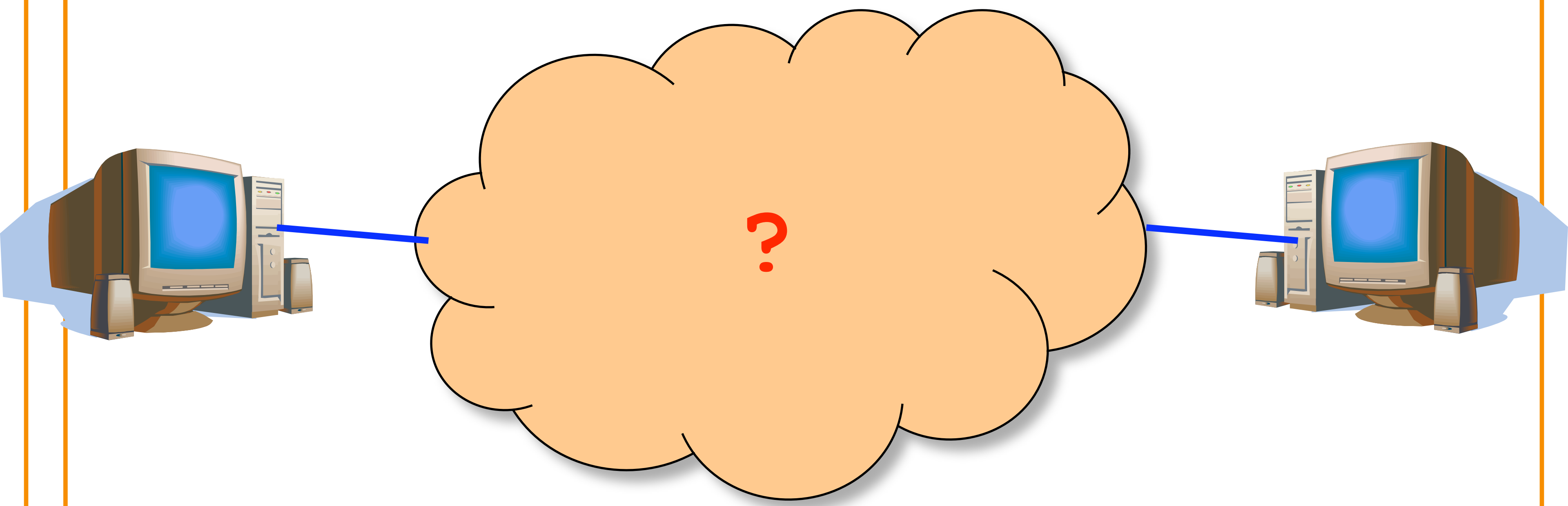
Ways to Deal With Congestion

- 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
- Dynamic adjustment (TCP)
  - Every sender infers the level of congestion
  - And 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?

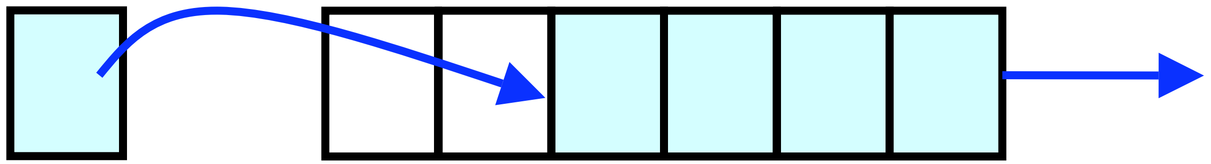
Inferring From Implicit Feedback



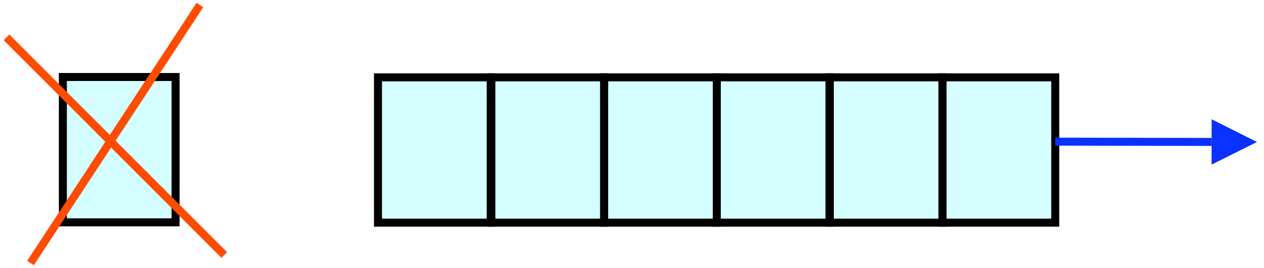
- What does the end host see?
- What can the end host change?

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

- Packet delay
  - Packet experiences high delay
- Packet loss
  - Packet gets dropped along the way
- How does TCP sender learn this?
  - Delay
    - Round-trip time estimate
  - Loss
    - Timeout
    - Duplicate acknowledgments

What Can the End Host Do?

- Upon detecting congestion
  - Decrease the sending rate (e.g., divide in half)
  - End host does its part to alleviate the congestion
- But, what if conditions change?
  - Suppose there is more bandwidth available
  - Would be a shame to stay at a low sending rate
- Upon *not* detecting congestion
  - Increase the sending rate, a little at a time
  - And see if the packets are successfully delivered

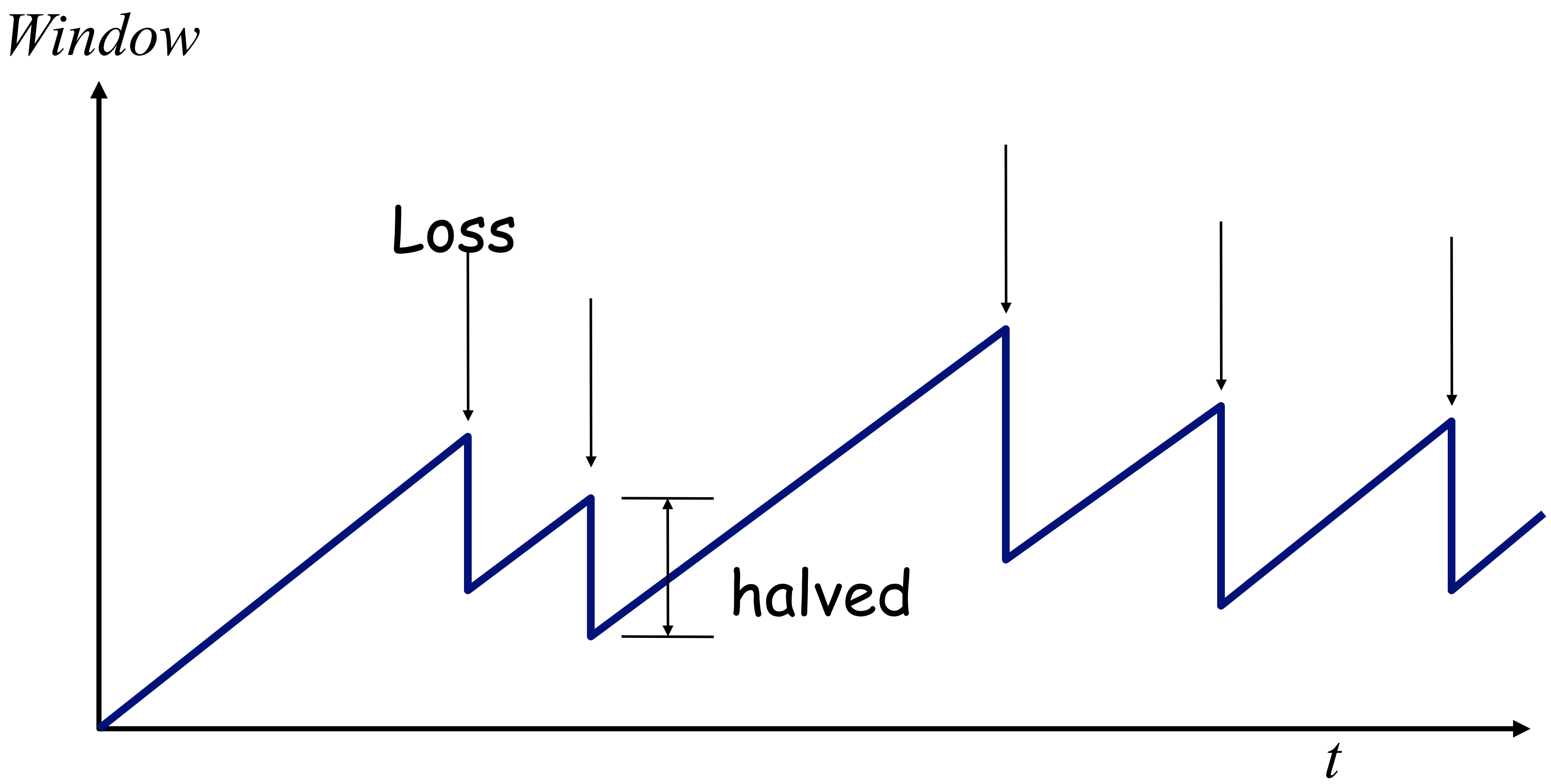
TCP Congestion Window

- Each TCP sender maintains a congestion window
  - Maximum number of bytes to have in transit
  - I.e., number of bytes still awaiting acknowledgments
- Adapting the congestion window
  - Decrease upon losing a packet: backing off
  - Increase upon success: optimistically exploring
  - Always struggling to find the right transfer rate
- Both good and bad
  - Pro: avoids having explicit feedback from network
  - Con: under-shooting and over-shooting the rate

Additive Increase, Multiplicative Decrease

- How much to increase and decrease?
  - Increase linearly, decrease multiplicatively
  - A necessary condition for stability of TCP
  - Consequences of over-sized window are much worse than having an under-sized window
    - Over-sized window: packets dropped and retransmitted
    - Under-sized window: somewhat lower throughput
- Multiplicative decrease
  - On loss of packet, divide congestion window in half
- Additive increase
  - On success for last window of data, increase linearly

Leads to the TCP "Sawtooth"



Practical Details

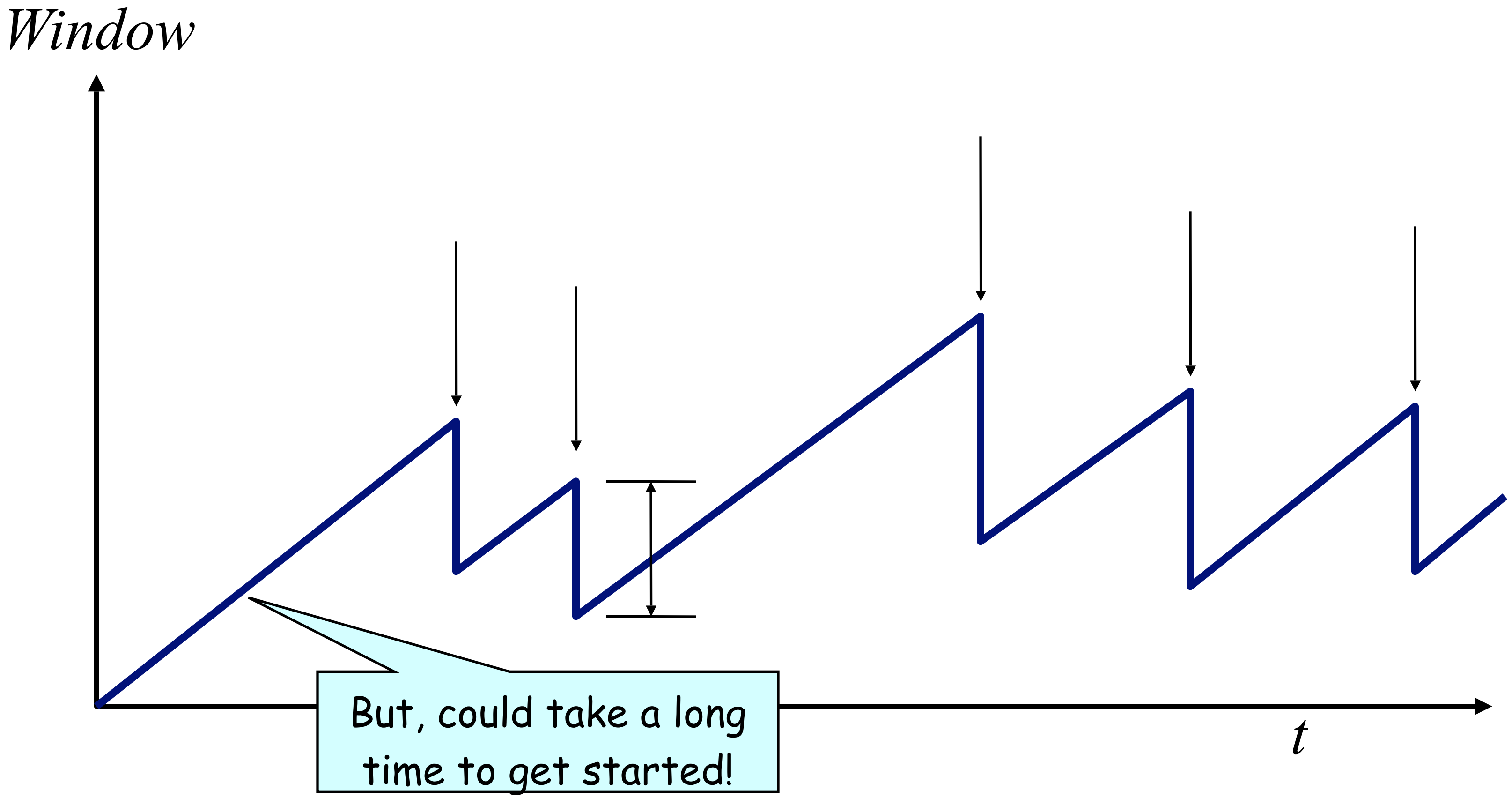
- Congestion window
  - Represented in bytes, not in packets (Why?)
  - Packets have MSS (Maximum Segment Size) bytes
- Increasing the congestion window
  - Increase by MSS on success for last window of data
- Decreasing the congestion window
  - Never drop congestion window below 1 MSS

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
  - TCP window: min{congestion window, receiver window}

How Should a New Flow Start

Need to start with a small CWND to avoid overloading the network.

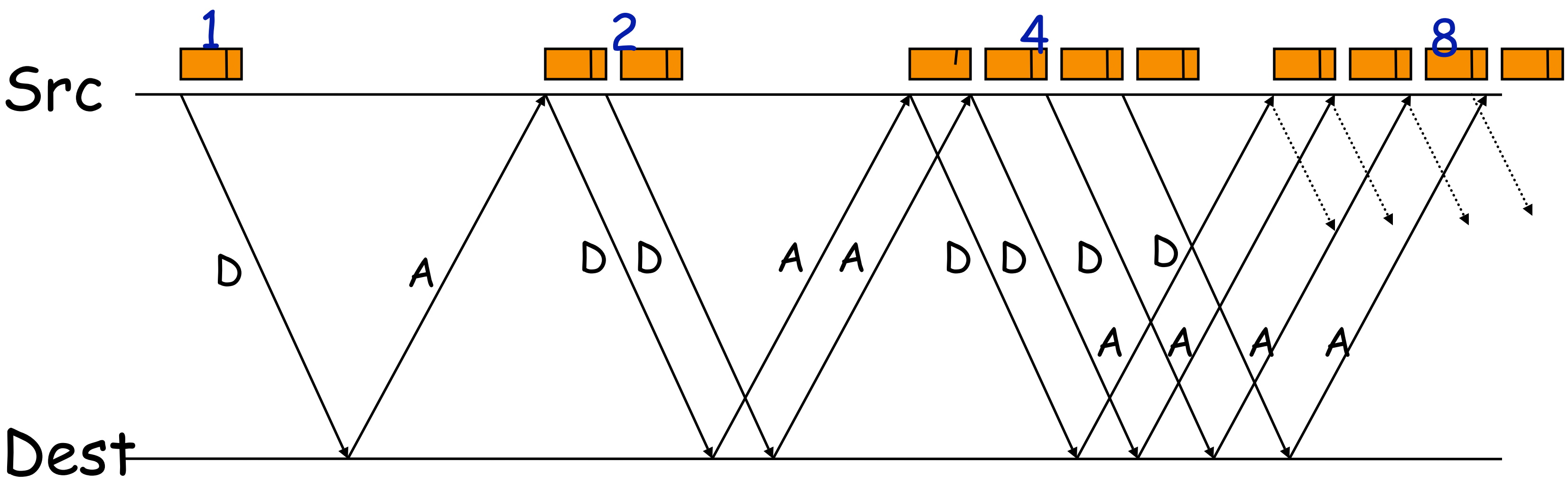


"Slow Start" Phase

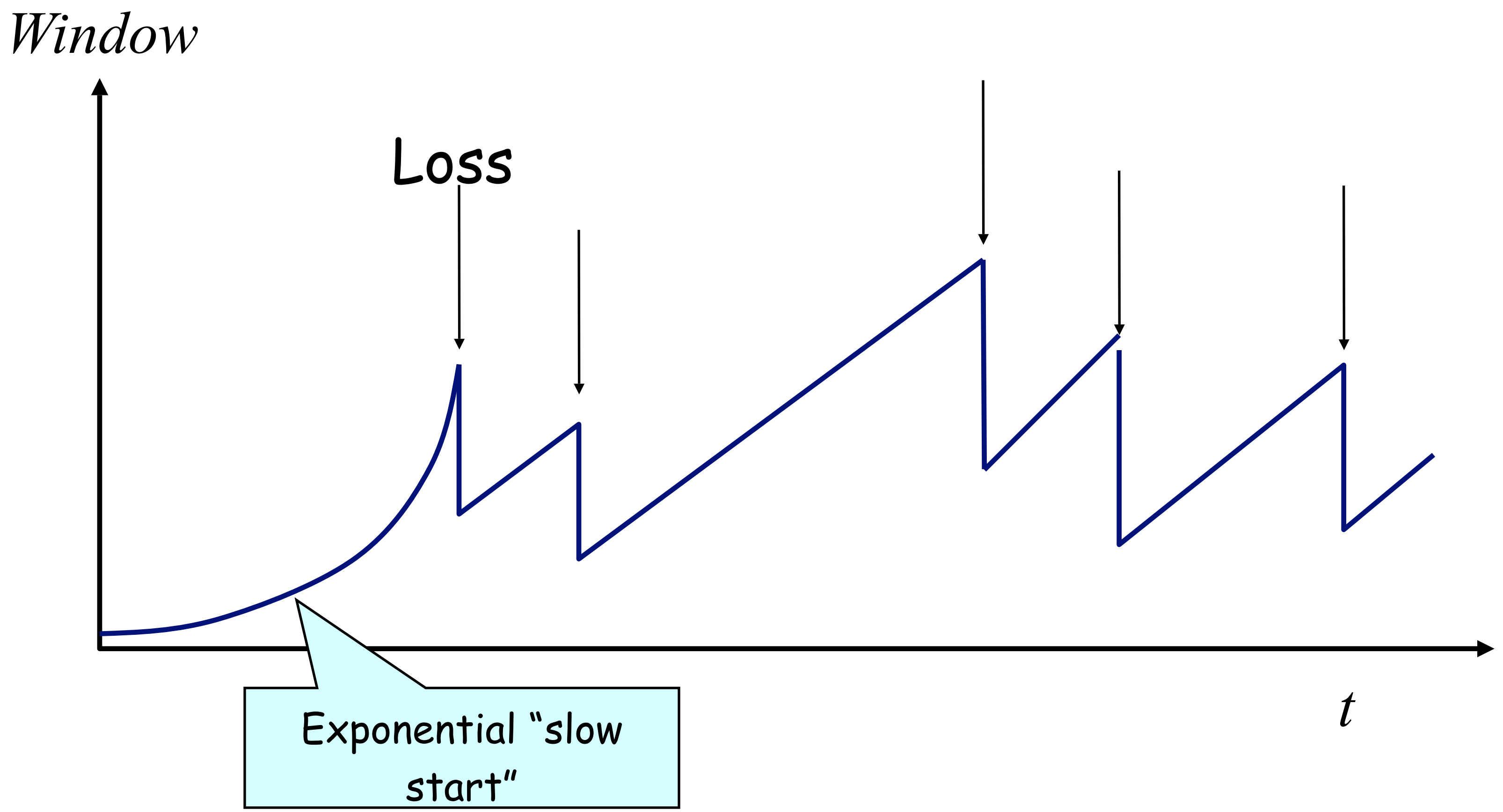
- Start with a small congestion window
  - Initially, CWND is 1 Max Segment Size (MSS)
  - So, initial sending rate is MSS/RTT
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (really "fast start")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

Slow Start in Action

Double CWND per round-trip time



Slow Start and the TCP Sawtooth

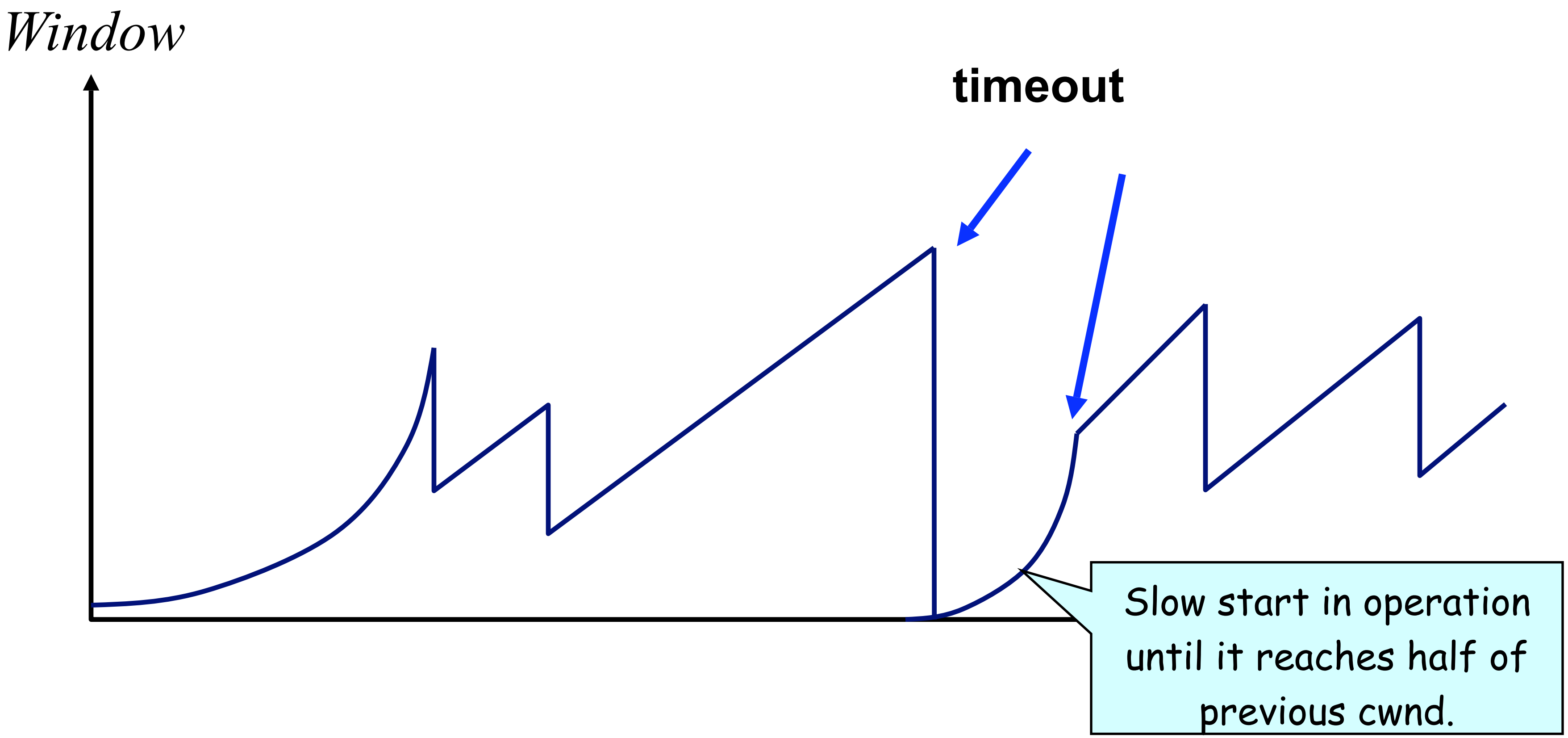


Why is it called slow-start? Because TCP originally had no congestion control mechanism. The source would just start by sending a whole receiver window's worth of data.

Two Kinds of Loss in TCP

- Timeout
  - Packet n is lost and detected via a timeout
  - E.g., because all packets in flight were lost
  - After the timeout, blasting away for the entire CWND
  - … would trigger a very large burst in traffic
  - So, better to start over with a low CWND
- Triple duplicate ACK
  - Packet n is lost, but packets n+1, n+2, etc. arrive
  - Receiver sends duplicate acknowledgments
  - … and the sender retransmits packet n quickly
  - 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
  - E.g., Telnet session where you don't type for an hour
- Eventually, the network conditions change
  - Maybe many more flows are traversing the link
  - E.g., maybe everybody has come back from lunch!
- Dangerous to start transmitting at the old rate
  - Previously-idle TCP sender might blast the 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 the only goal
  - We also want to be *fair* to the various flows
  - … but what the heck does *that* mean?
- Simple definition: equal shares of the bandwidth
  - N flows that each get 1/N of the bandwidth?

What About Cheating?

- Some folks are more fair than others
  - Running multiple TCP connections in parallel
  - Modifying the TCP implementation in the OS
  - 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?
  - Peer pressure?
  - ???

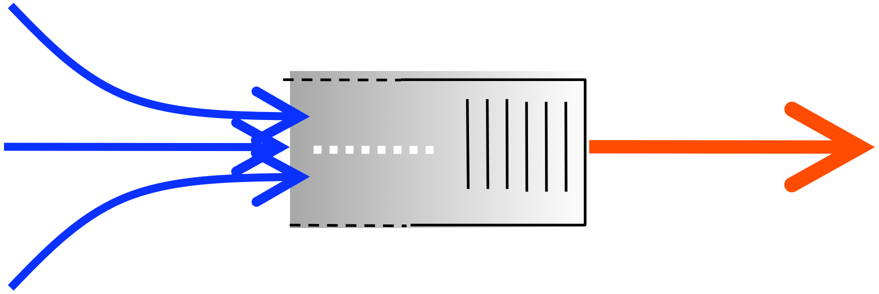
Queuing Mechanisms

Random Early Detection (RED)

Explicit Congestion Notification (ECN)

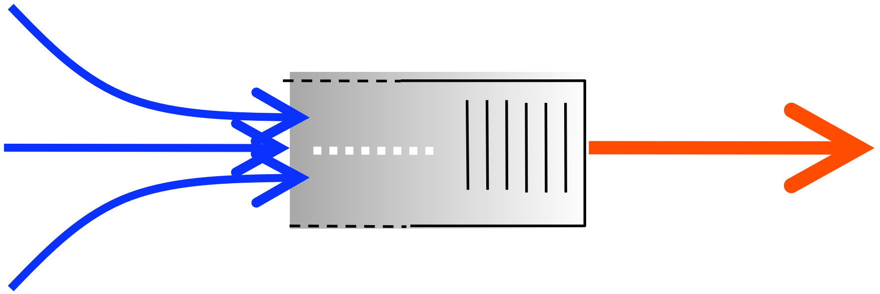
Bursty Loss From Drop-Tail Queuing

- TCP depends on packet loss
  - Packet loss is the indication of congestion
  - In fact, TCP *drives* the network into packet loss
  - … by continuing to increase the sending rate
- Drop-tail queuing leads to *bursty* loss
  - When a link becomes congested…
  - … many arriving packets encounter a full queue
  - And, as a result, many flows divide sending rate in half
  - … and, many individual flows lose multiple packets



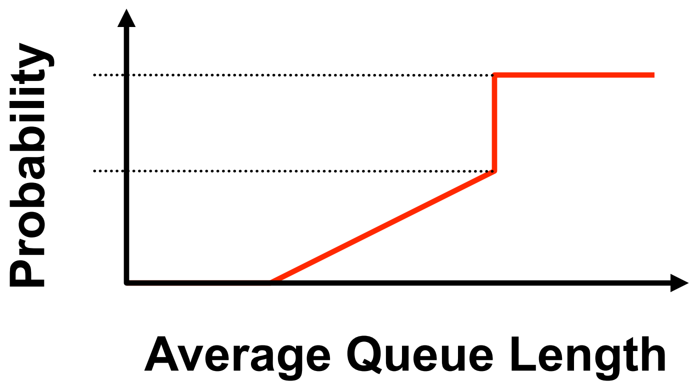
Slow Feedback from Drop Tail

- Feedback comes when buffer is completely full
  - … even though the buffer has been filling for a while
- Plus, the filling buffer is increasing RTT
  - … and the variance in the RTT
- Might be better to give early feedback
  - Get one or two connections to slow down, not all of them
  - Get these connections to slow down before it is too late



Random Early Detection (RED)

- Basic idea of RED
  - Router notices that the queue is getting backlogged
  - … and randomly drops packets to signal congestion
- Packet drop probability
  - Drop probability increases as queue length increases
  - If buffer is below some level, don't drop anything
  - … otherwise, set drop probability as function of queue



Properties of RED

- Drops packets before queue is full
  - In the hope of reducing the rates of some flows
- Drops packet in proportion to each flow's rate
  - High-rate flows have more packets
  - … and, hence, a higher chance of being selected
- Drops are spaced out in time
  - Which should help desynchronize the TCP senders
- Tolerant of burstiness in the traffic
  - By basing the decisions on *average* queue length

Problems With RED

- Hard to get the tunable parameters just right
  - How early to start dropping packets?
  - What slope for the increase in drop probability?
  - What time scale for averaging the queue length?
- Sometimes RED helps but sometimes not
  - If the parameters aren't set right, RED doesn't help
  - And it is hard to know how to set the parameters
- RED is implemented in practice
  - But, often not used due to the challenges of tuning right

Explicit Congestion Notification

- Early dropping of packets
  - Good: gives early feedback
  - Bad: has to drop the packet to give the feedback
- Explicit Congestion Notification
  - Router marks the packet with an ECN bit
  - … and sending host interprets as a sign of congestion
- Surmounting the challenges
  - Must be supported by the end hosts and the routers
  - Requires two bits in the IP header (one for the ECN mark, and one to indicate the ECN capability)
  - Solution: borrow two of the Type-Of-Service bits in the IPv4 packet header

Other TCP Mechanisms

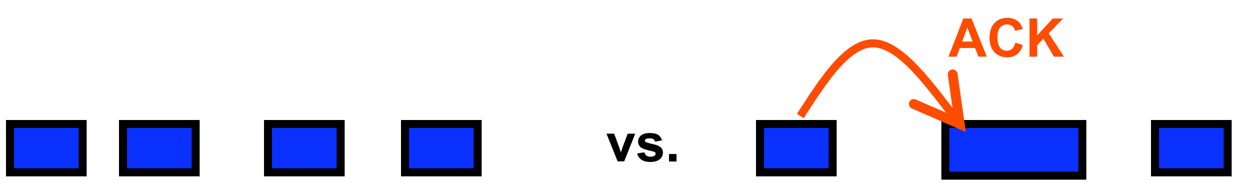
Nagle's Algorithm and Delayed ACK

Motivation for Nagle's Algorithm

- Interactive applications
  - Telnet and rlogin
  - Generate many small packets (e.g., keystrokes)
- Small packets are wasteful
  - Mostly header (e.g., 40 bytes of header, 1 of data)
- Appealing to reduce the number of packets
  - Could force every packet to have some minimum size
  - … but, what if the person doesn't type more characters?
- Need to balance competing trade-offs
  - Send larger packets
  - … but don't introduce much delay by waiting

Nagle's Algorithm

- Wait if the amount of data is small
  - Smaller than Maximum Segment Size (MSS)
- And some other packet is already in flight
  - I.e., still awaiting the ACKs for previous packets
- That is, send at most one small packet per RTT
  - … by waiting until all outstanding ACKs have arrived



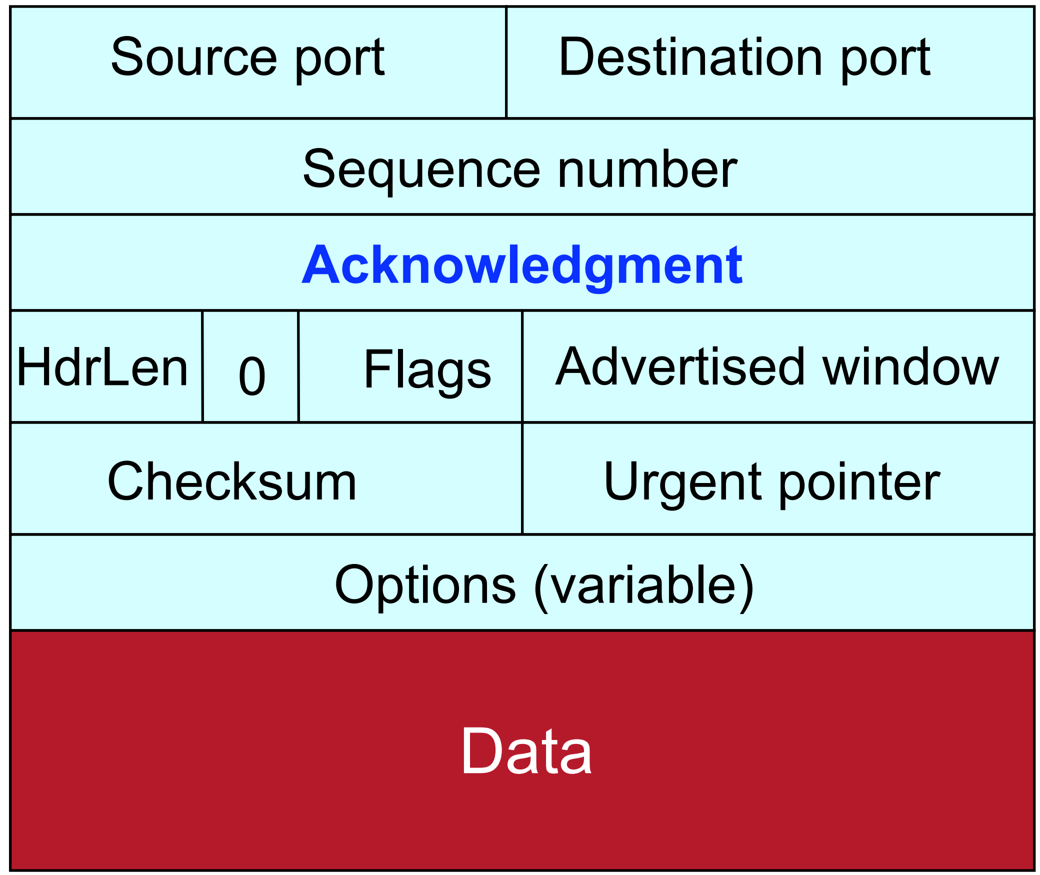
- Influence on performance
  - Interactive applications: enables batching of bytes
  - Bulk transfer: transmits in MSS-sized packets anyway

Motivation for Delayed ACK

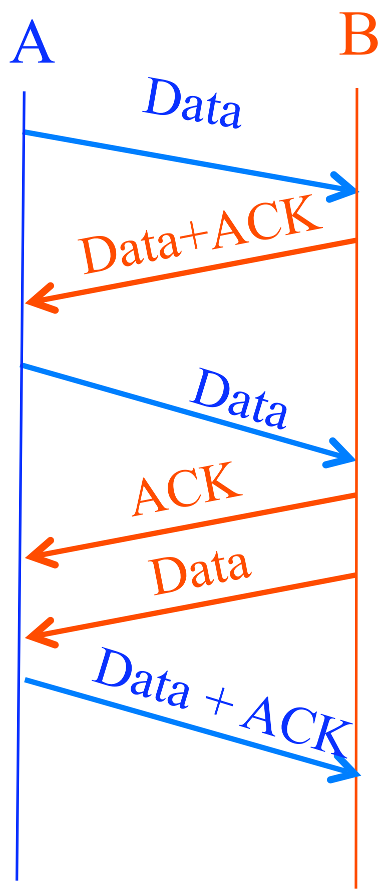
- TCP traffic is often bidirectional
  - Data traveling in both directions
  - ACKs traveling in both directions
- ACK packets have high overhead
  - 40 bytes for the IP header and TCP header
  - … and zero data traffic
- Piggybacking is appealing
  - Host B can send an ACK to host A
  - … as part of a data packet from B to A

TCP Header Allows Piggybacking

Flags: SYN
FIN
RST
PSH
URG
ACK



Example of Piggybacking



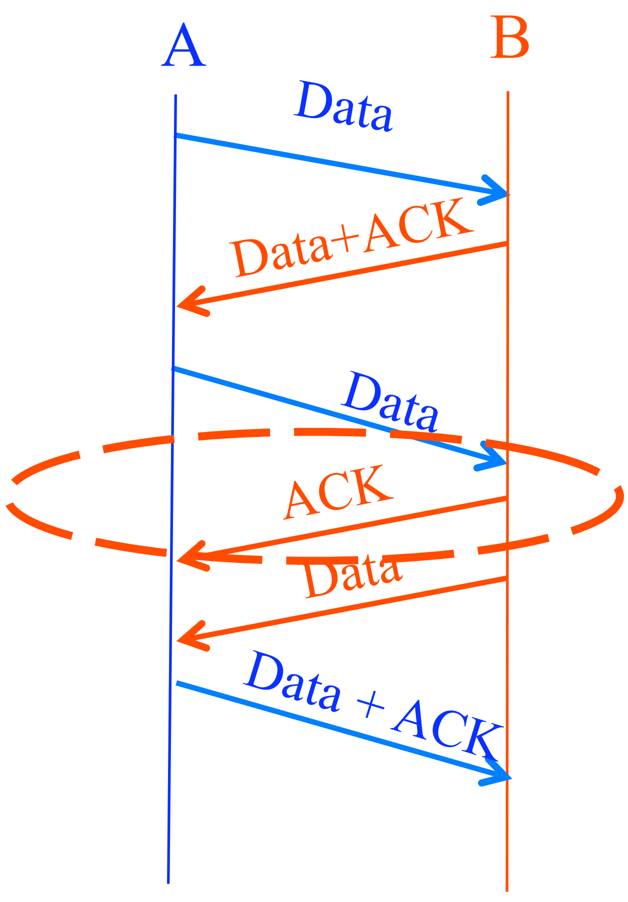
B has data to send

B doesn't have data to send

A has data to send

Increasing Likelihood of Piggybacking

- Increase piggybacking
  - TCP allows the receiver to *wait* to send the ACK
  - … in the hope that the host will have data to send
- Example: rlogin or telnet
  - Host A types characters at a UNIX prompt
  - Host B receives the character and executes a command
  - … and then data are generated
  - Would be nice if B could send the ACK with the new data



Delayed ACK

- Delay sending an ACK
  - Upon receiving a packet, the host B sets a timer
    - Typically, 200 msec or 500 msec
  - If B's application generates data, go ahead and send
    - And piggyback the ACK bit
  - If the timer expires, send a (non-piggybacked) ACK
- Limiting the wait
  - Timer of 200 msec or 500 msec
  - ACK every other full-sized packet

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?