CSC 358 – Principles of Computer Networks

Handout # 9: Congestion Control

Joe Lim
Department of Mathematical and Computational Sciences
joe.lim@utoronto.ca

Announcements

- PS2 Due: Fri, Mar 16 @11:59PM
- PA2: Due: Fri, Mar 30 @11:59PM
  - 6% : Working PA1
  - 12%: Working PA2
- Need to work on PA1?
  - Use PA1-Test on MarkUs
- PA2 Autotester running on MarkUs
- Finals: Mon, Apr 9, 2018 from 9-12 in IB120
  - Please consult the official timetable website https://student.utm.utoronto.ca/examschedule/finalexams.php

Extra Office Hours

- After semester end and final exams
- Is there a need?
- If so, when is a good time?

Today's Lecture

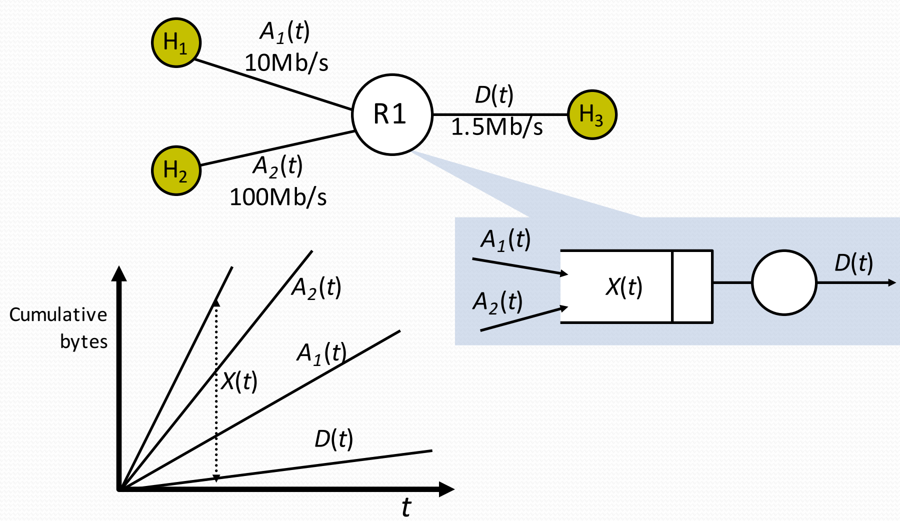
- Principles of congestion control
  - Learning that congestion is occurring
  - Adapting to alleviate the congestion
- TCP congestion control
  - Additive-increase, multiplicative-decrease
  - Slow start and slow-start restart
- Related TCP mechanisms
  - Nagle's algorithm and delayed acknowledgments
- Active Queue Management (AQM)
  - Random Early Detection (RED)
  - Explicit Congestion Notification (ECN)

Congestion Control

congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

What is Congestion?



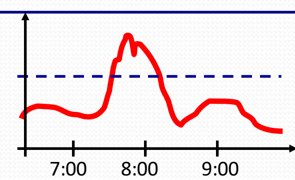
Flow Control vs. Congestion Control

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

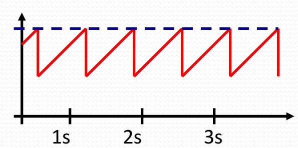
- Different concepts, but similar mechanisms
  - TCP flow control: receiver window
  - TCP congestion control: congestion window
  - TCP window: min{congestion window, receiver window}

Time Scales of Congestion

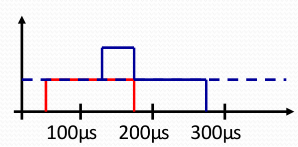
Too many users using a link during a peak hour



TCP flows filling up all available bandwidth

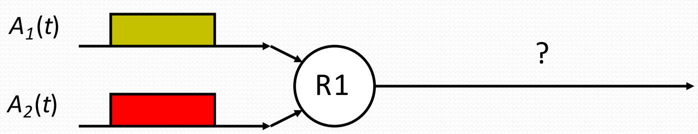


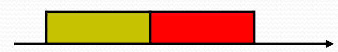
Two packets colliding at a router – also referred to as contention



Dealing with Congestion

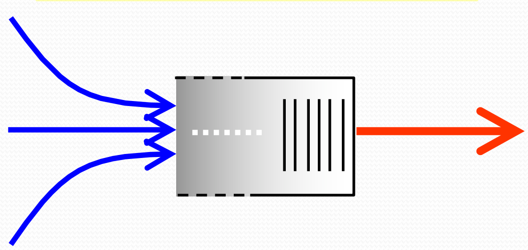
Example: two flows arriving at a router



Strategy	
Drop one of the flows	
Buffer one flow until the other has departed, then send it	
Re-Schedule one of the two flows for a later time	
Ask both flows to reduce their rates	

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 a short period of time
  - The node cannot keep up with the arriving traffic
  - … and the buffer may eventually overflow



Arguably Congestion is Good!

- We use packet switching because it makes efficient use of the links. Therefore, buffers in the routers are frequently occupied.
- If buffers are always empty, delay is low, but our usage of the network is low.
- If buffers are always occupied, delay is high, but we are using the network more efficiently.
- So how much congestion is too much?

Congestion Collapse

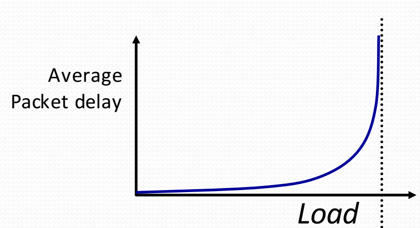
- **Definition**: Increase in network load results in a decrease of useful work done
- Many possible causes
  - Spurious retransmissions of packets still in flight
    - Classical congestion collapse
    - Solution: better timers and TCP congestion control
  - Undelivered packets
    - Packets consume resources and are dropped elsewhere in network
    - Solution: congestion control for ALL traffic

What Do We Want, Really?

- High throughput
  - Throughput: measured performance of a system
  - E.g., number of bits/second of data that get through
- Low delay
  - Delay: time required to deliver a packet or message
  - E.g., number of msec to deliver a packet
- These two metrics are sometimes at odds
  - E.g., suppose you drive a link as hard as possible
  - … then, throughput will be high, but delay will be, too

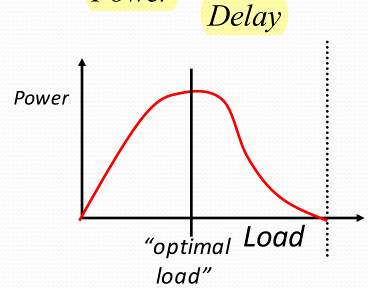
Load, Delay, and Power

Typical behavior of queuing systems with random arrivals:



A simple metric of how well the network is performing:

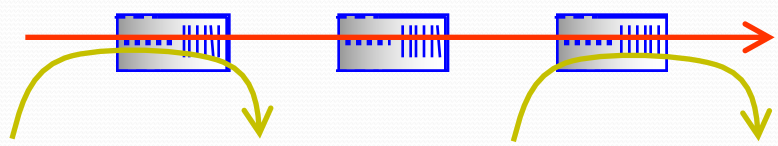
$$Power = \frac{Load}{Delay}$$



Goal: maximize power

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?
  - But, what if the flows traverse different paths?

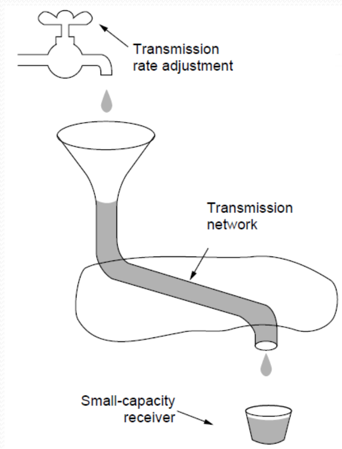


Resource Allocation vs. Congestion Control

- Resource allocation
  - How nodes meet competing demands for resources
  - E.g., link bandwidth and buffer space
  - When to say no, and to whom
- Congestion control
  - How nodes prevent or respond to overload conditions
  - E.g., persuade hosts to stop sending, or slow down
  - Typically has notions of fairness (i.e., sharing the pain)

Regulating the Sending Rate

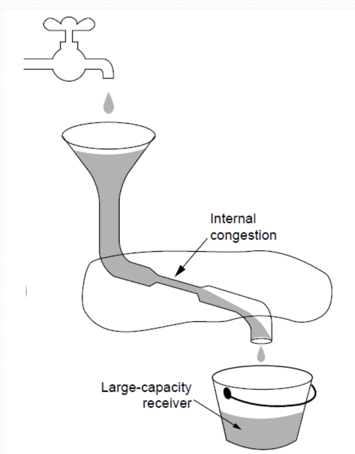
- Sender may need to slow down for different reasons:
  - Flow control, when the receiver is not fast enough [right]
  - Congestion, when the network is not fast enough [over]



A fast network feeding a low-capacity receiver → flow control is needed

Regulating the Sending Rate

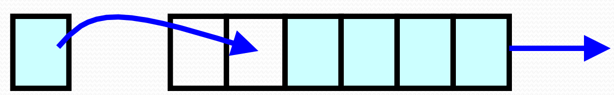
- Our focus is dealing with this problem – congestion



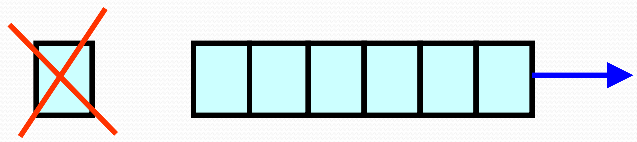
A slow network feeding a high-capacity receiver → congestion control is needed

Simple Resource Allocation

- Simplest approach: FIFO queue and drop-tail
- Link bandwidth: first-in first-out queue
  - Packets transmitted in the order they arrive



- Buffer space: drop-tail queuing
  - If the queue is full, drop the incoming packet



Simple Congestion Detection

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

Options for Congestion Control

- Implemented by host versus network
- Reservation-based, versus feedback-based
- Window-based versus rate-based.

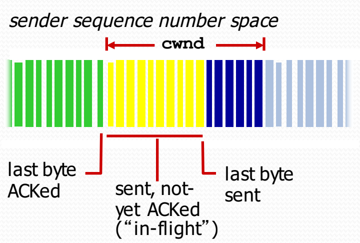
TCP Congestion Control

- TCP implements host-based, feedback-based, window-based congestion control.
- TCP sources attempts to determine how much capacity is available
- TCP sends packets, then reacts to observable events (loss).

Idea of TCP Congestion Control

- Each source determines the available capacity
  - … so it knows how many packets to have in transit
- Congestion window
  - Maximum # of unacknowledged bytes to have in transit
  - The congestion-control equivalent of receiver window
  - MaxWindow = min{congestion window, receiver window}
  - Send at the rate of the slowest component: receiver or network
- Adapting the congestion window
  - Decrease upon losing a packet: backing off
  - Increase upon success: optimistically exploring

TCP Congestion Control: Details



- sender limits transmission:

$$\text{LastByteSent-LastByteAcked} \leq \text{cwnd}$$

- **cwnd** is dynamic, function of perceived network congestion

TCP sending rate:

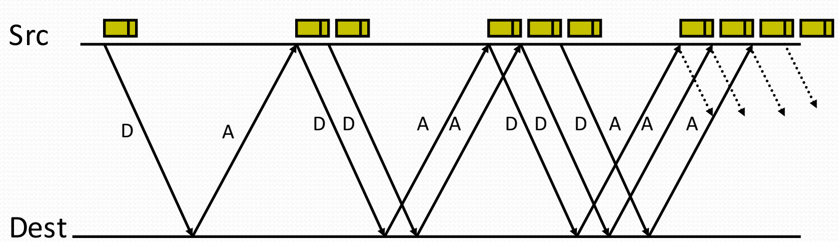
- *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

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

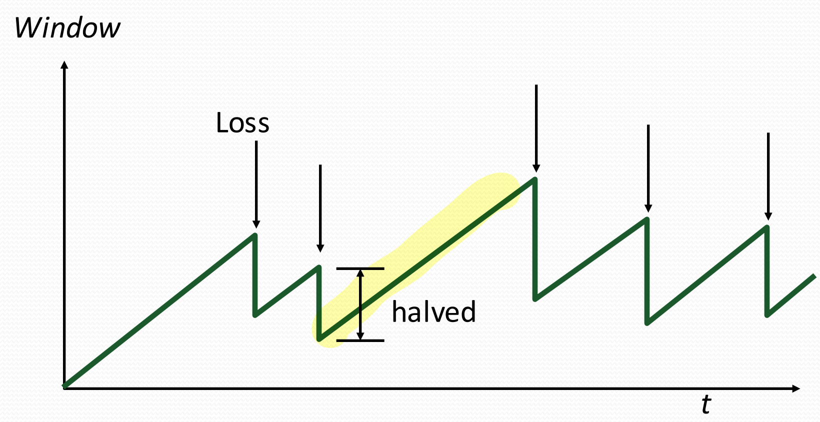
Additive Increase



Actually, TCP uses bytes, not segments to count:
When ACK is received:

$$cwnd += MSS\left(\frac{MSS}{cwnd}\right)$$

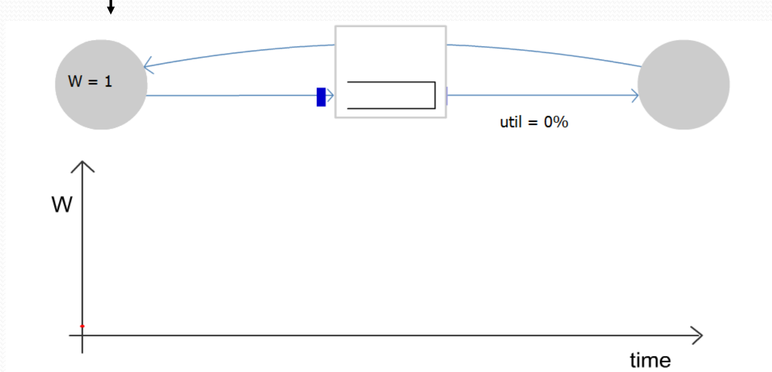
Leads to the TCP "Sawtooth"



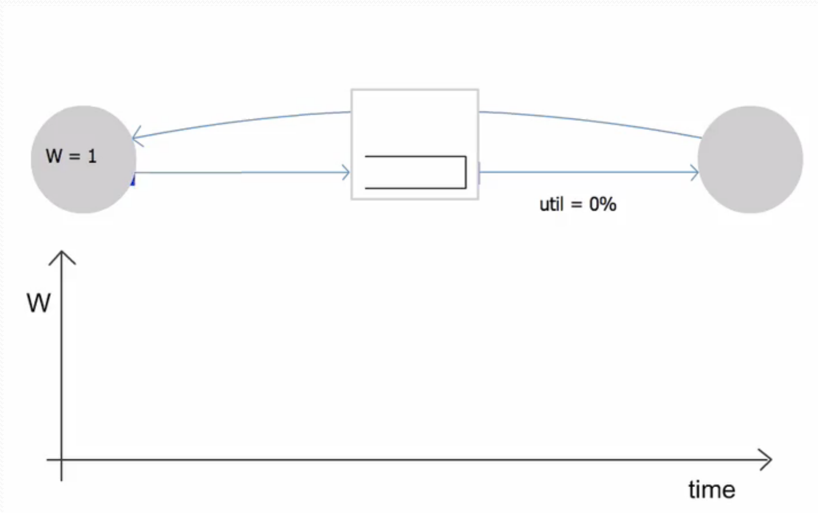
Congestion Window Evolution

Only W packets may be outstanding

Rule for adjusting W
- If an ACK is received: W ← W+1/W
- If a packet is lost: W ← W/2



Congestion Window Evolution



Practical Details

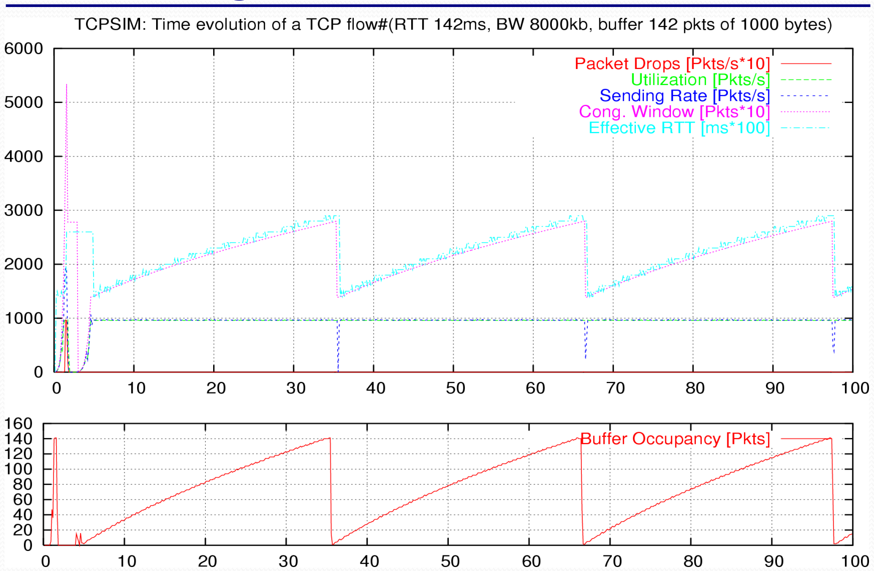
- Congestion window
  - Represented in bytes, not in packets (Why?)
  - Packets have MSS (Maximum Segment Size) bytes
- Increasing the congestion window
  - Increase by MSS on success for last window of data
  - In practice, increase a fraction of MSS per received ACK
    - # packets per window: CWND / MSS
    - Increment per ACK: MSS * (MSS / CWND)
- Decreasing the congestion window
  - Never drop congestion window below 1 MSS

TCP Sending Rate

- What is the sending rate of TCP?
- Acknowledgement for sent packet is received after one RTT
- Amount of data sent until ACK is received is the current window size W
- Therefore sending rate is R = W/RTT

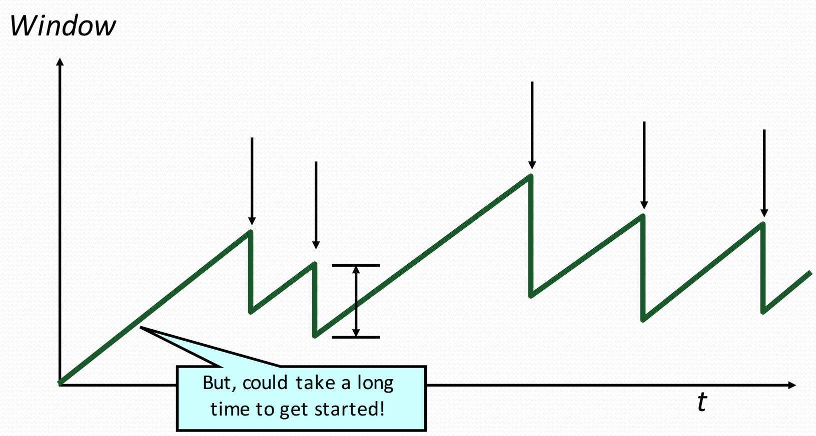
- Is the TCP sending rate saw tooth shaped as well?

TCP Sending Rate and Buffers



Getting Started

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

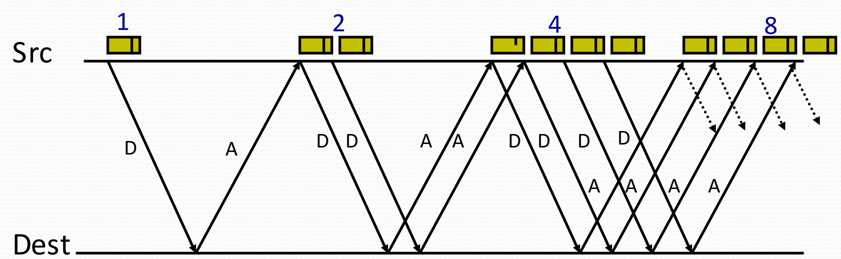


"Slow Start" Phase

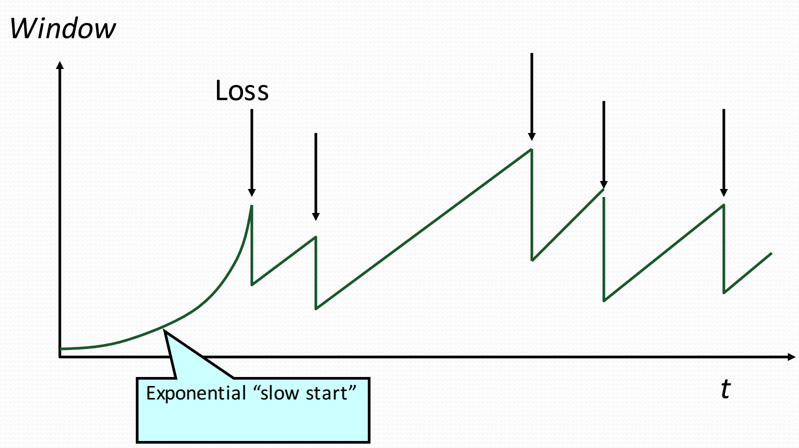
- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS/RTT
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (really "fast start")
  - Sender starts at 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
=
Increase CWND by 1 for each ACK received



Slow Start and the TCP Sawtooth

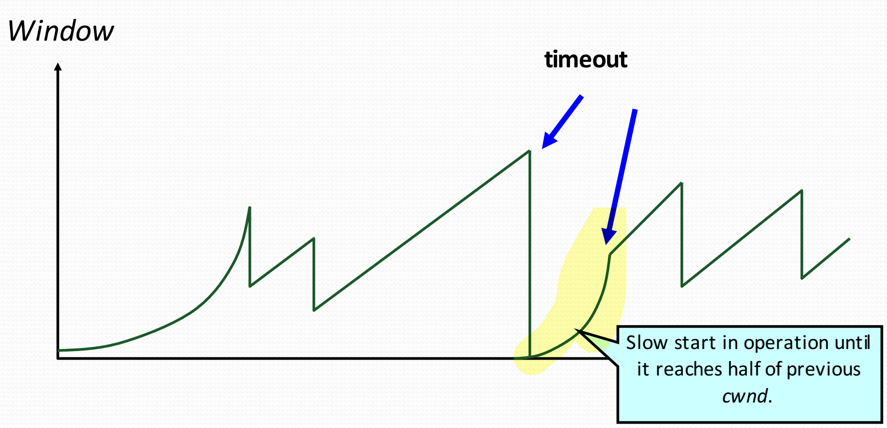


Why is it called slow-start? Because TCP originally had no congestion control mechanism. The source would just start by sending a whole window's worth of data.

Two Kinds of Loss in TCP

- 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
- 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

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

Other TCP Mechanisms

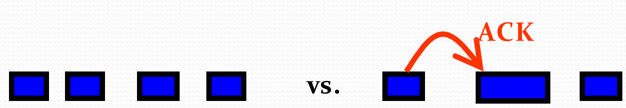
- Nagle's Algorithm and Delayed ACK

Motivation for Nagle's Algorithm

- Interactive applications
  - Telnet, ssh 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

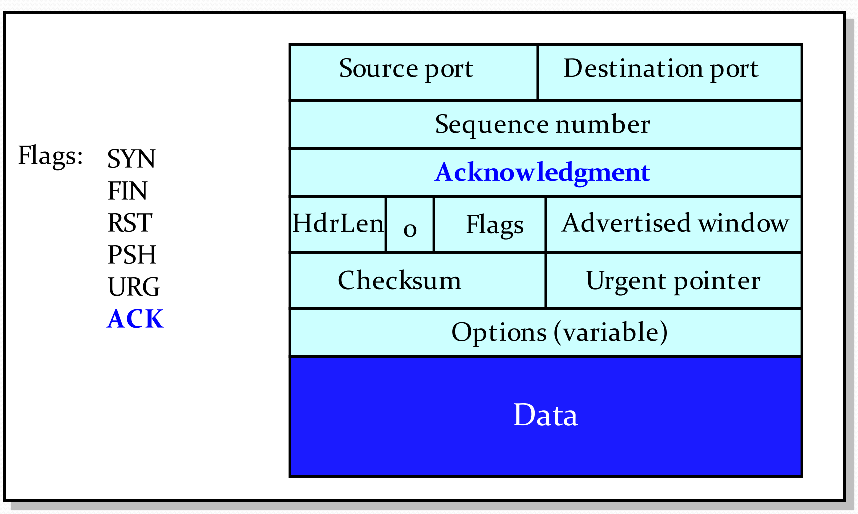


- Influence on performance
  - Interactive applications: enables batching of bytes
  - Bulk transfer: transmits in MSS-sized packets anyway

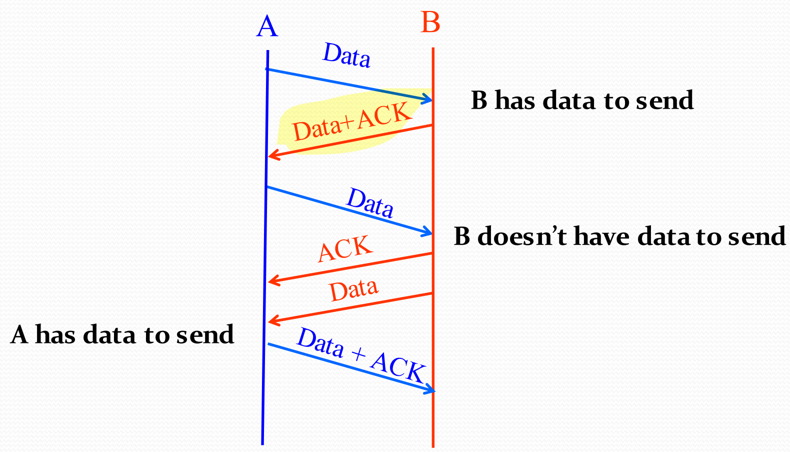
Motivation for Delayed ACK

- 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

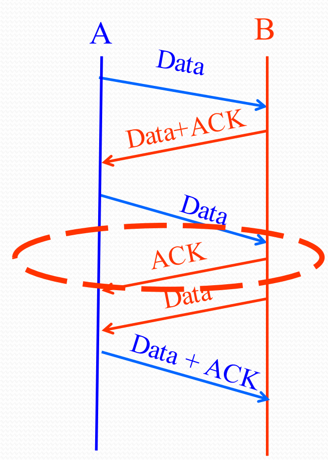


Example of Piggybacking



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)