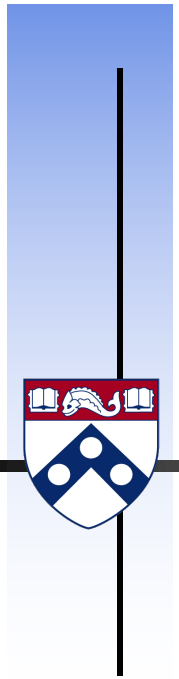


CIS 553: Networked Systems

Flow and congestion control

March 17, 2021

Agenda

- Transmission Control Protocol 
  - Reliable in-order delivery 

  - Connection-oriented
  - Flow control
  - Congestion control

Transmission Control Protocol (TCP)

1. Stream-of-bytes service
   - Sends and receives a stream of bytes
2. Reliable, in-order delivery
   - Detect corruption, loss, and reordering
   - Reliable delivery: acknowledgments and retransmissions
3. Connection-oriented
   - Explicit set-up and tear-down of TCP connection
4. Flow control
   - Prevent overflow of the receiver's buffer space
5. Congestion control
   - Adapt to network congestion for the greater good

Many more optimizations

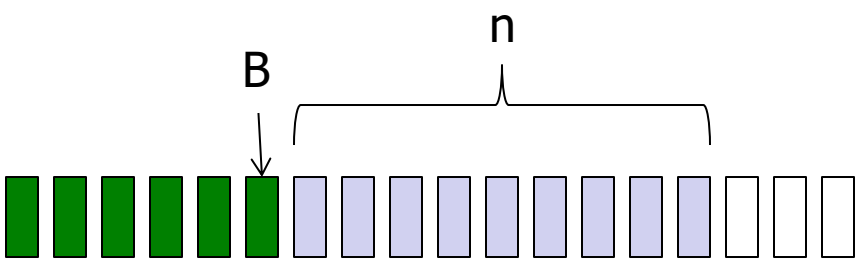
- Checksums (to detect bit errors)
- Acknowledgements (plus retransmissions)
  - Can we avoid using a timeout per packet?
- Sequence numbers (to deal with duplicates)
  - Can we defend against stale packets?
- Timers (to detect loss)
  - How do we set the timer?

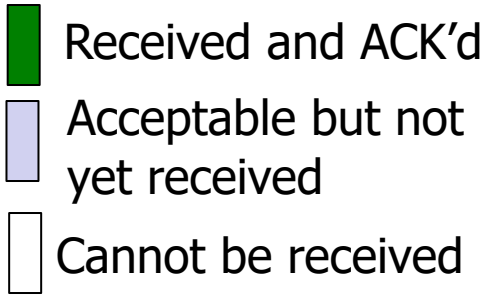
Acknowledgements

- Problem: timeouts are very slow
- Idea: ACKs can carry information beyond just the current packet
  - Cumulative ACKs: ACK carries next in-order sequence number that the receiver expects
  - Selective ACKs: ACK individually acknowledges correctly received packets

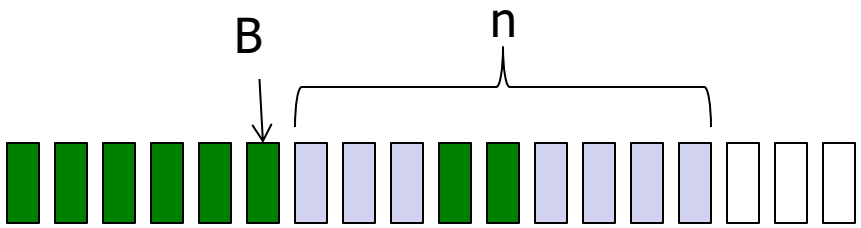
Cumulative acknowledgements (cont'd)

- At receiver





- After receiving B+4, B+5



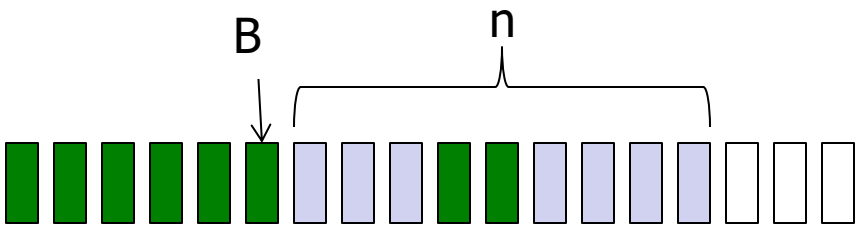
- Receiver sends ACK(B+1)

Taking advantage of cumulative ACKs

- When packet n is lost…
  - … packets n+1, n+2, and so on may get through
- Exploit the ACKs of these packets
  - ACK says receiver is still awaiting nth packet
  - Duplicate ACKs suggest later packets arrived
  - Sender uses "duplicate ACKs" as a hint
- Fast retransmission
  - Retransmit after "triple duplicate ACK"

Selective ACKs

- Send in the ACK the ranges of bytes received
- Example:



ACK(B+1, [B+4, B+6])

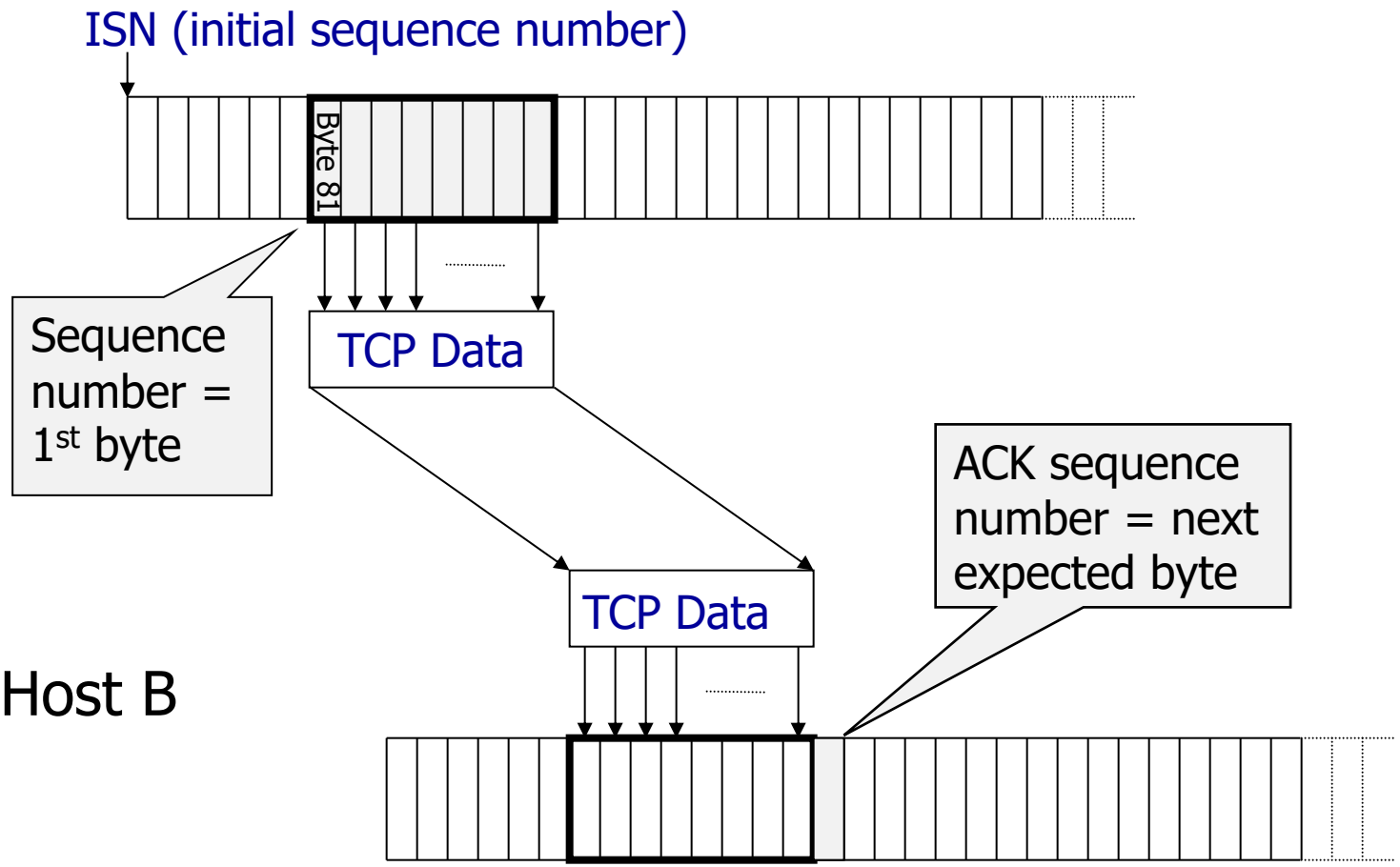
- Selective ACKs offer more precise information but require more complicated book-keeping

Many more optimizations

- Checksums (to detect bit errors)
- Acknowledgements (plus retransmissions)
  - Can we avoid using a timeout per packet?
- Sequence numbers (to deal with duplicates)
  - Can we defend against stale packets?
- Timers (to detect loss)
  - How do we set the timer?

Recap: Sequence Numbers

Host A



Initial Sequence Number (ISN)

- Problem: Can't always use ISN of 0
  - Why?
- Reuse of port numbers
  - Port numbers must (eventually) get used again
  - … and an old packet may still be in flight
  - … and associated with the new connection
- Idea: TCP must change the ISN over time
  - Set from a 32-bit clock that ticks every 4 microsec
  - … which wraps around once every 4.55 hours!

Many more optimizations

- Checksums (to detect bit errors)
- Acknowledgements (plus retransmissions)
  - Can we avoid using a timeout per packet?
- Sequence numbers (to deal with duplicates)
  - Can we defend against stale packets?
- Timers (to detect loss)
  - How do we set the timer?

Retransmission timeout

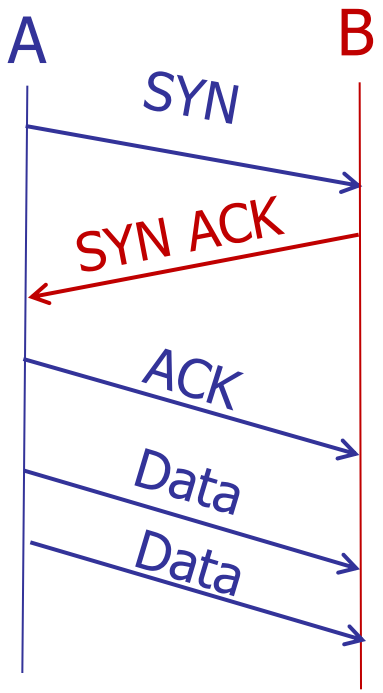
- Problem: How to set timeout for when to retransmit the first packet in the window?
  - Too long: connection has low throughput
  - Too short: retransmit packet that was just delayed
- Solution: make timeout proportional to RTT
  - RTO = EstimatedRTT + 4 x StdDevRTT

Agenda

- Transmission Control Protocol 
  - Reliable in-order delivery 
  - Connection-oriented 

  - Flow control
  - Congestion control

Establishing a TCP Connection



Each host tells its ISN to the other host

- Three-way handshake to establish connection
  - Host A sends a **SYN** (open) to the host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header

Flags: SYN
FIN
RST
PSH
URG
ACK

Source port			Destination port
Sequence number			
Acknowledgment			
HdrLen	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			
Data			

Step 1: A's Initial SYN Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

A's port			B's port
A's Initial Sequence Number			
Acknowledgment			
20	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			

A tells B it wants to open a connection…

Step 2: B's SYN-ACK Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

B's port			A's port
B's Initial Sequence Number			
A's ISN plus 1			
20	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			

B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

Step 3: A's ACK of the SYN-ACK

Flags: SYN
FIN
RST
PSH
URG
ACK

A's port			B's port
Sequence number			
B's ISN plus 1			
20	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			

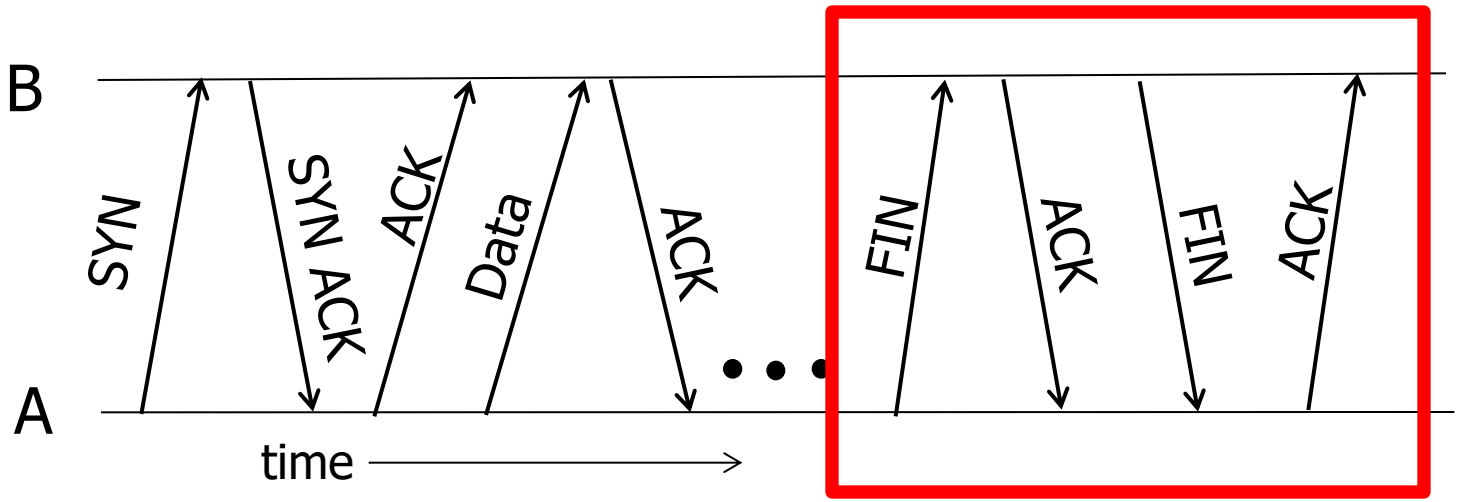
A tells B it is okay to start sending

… upon receiving this packet, B can start sending data

SYN Loss

- Upon sending SYN, sender sets a timer
  - If SYN lost, timer expires before SYN-ACK received
  - Sender retransmits SYN
- How should the TCP sender set the timer?
  - No idea how far away the receiver is
  - Some TCPs use default of 3 or 6 seconds
- Usually handled out-of-band…
  - User gets impatient and hits reload
  - … Users aborts connection, initiates new socket
  - Essentially, forces a fast send of a new SYN!

Tearing Down the Connection



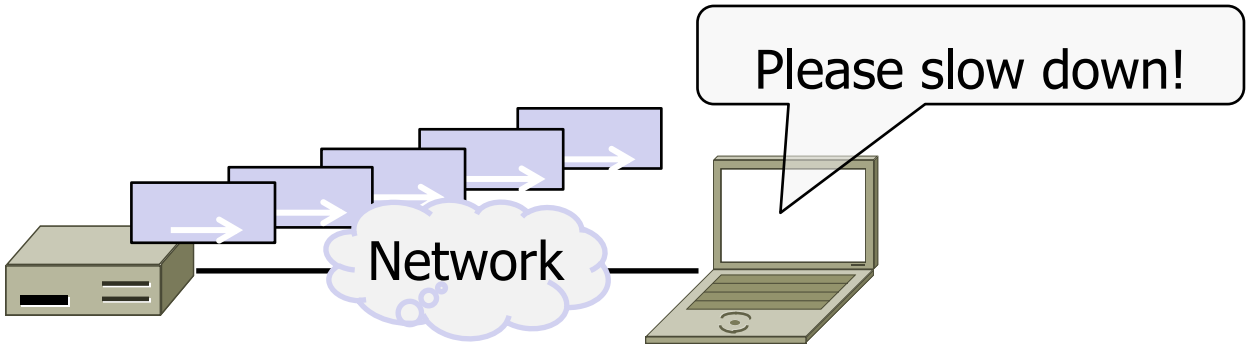
- Closing (each end of) the connection
  - Finish (FIN) to close and receive remaining bytes
  - And other host sends a FIN ACK to acknowledge
  - Reset (RST) to close and not receive remaining bytes

Agenda

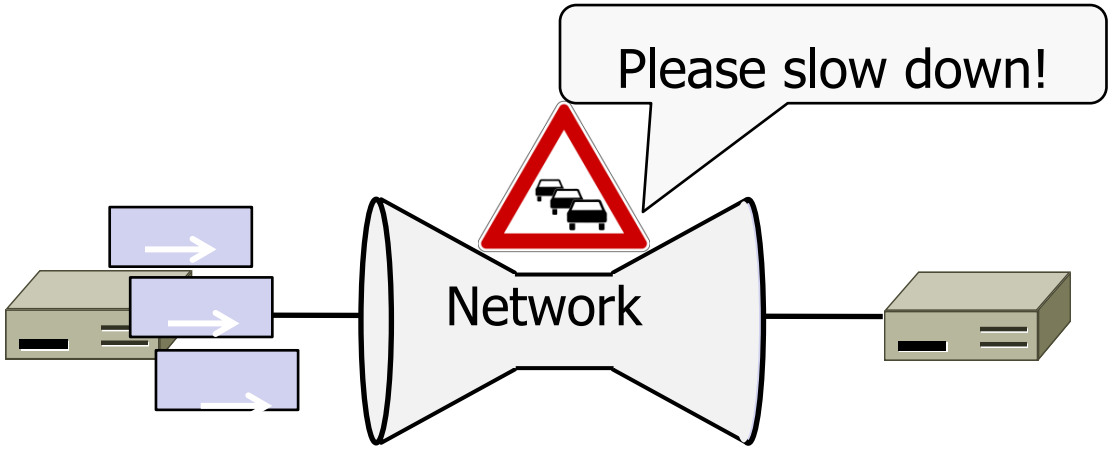
- Transmission Control Protocol 
  - Reliable in-order delivery 
  - Connection-oriented 
  - Flow control 

  - Congestion control

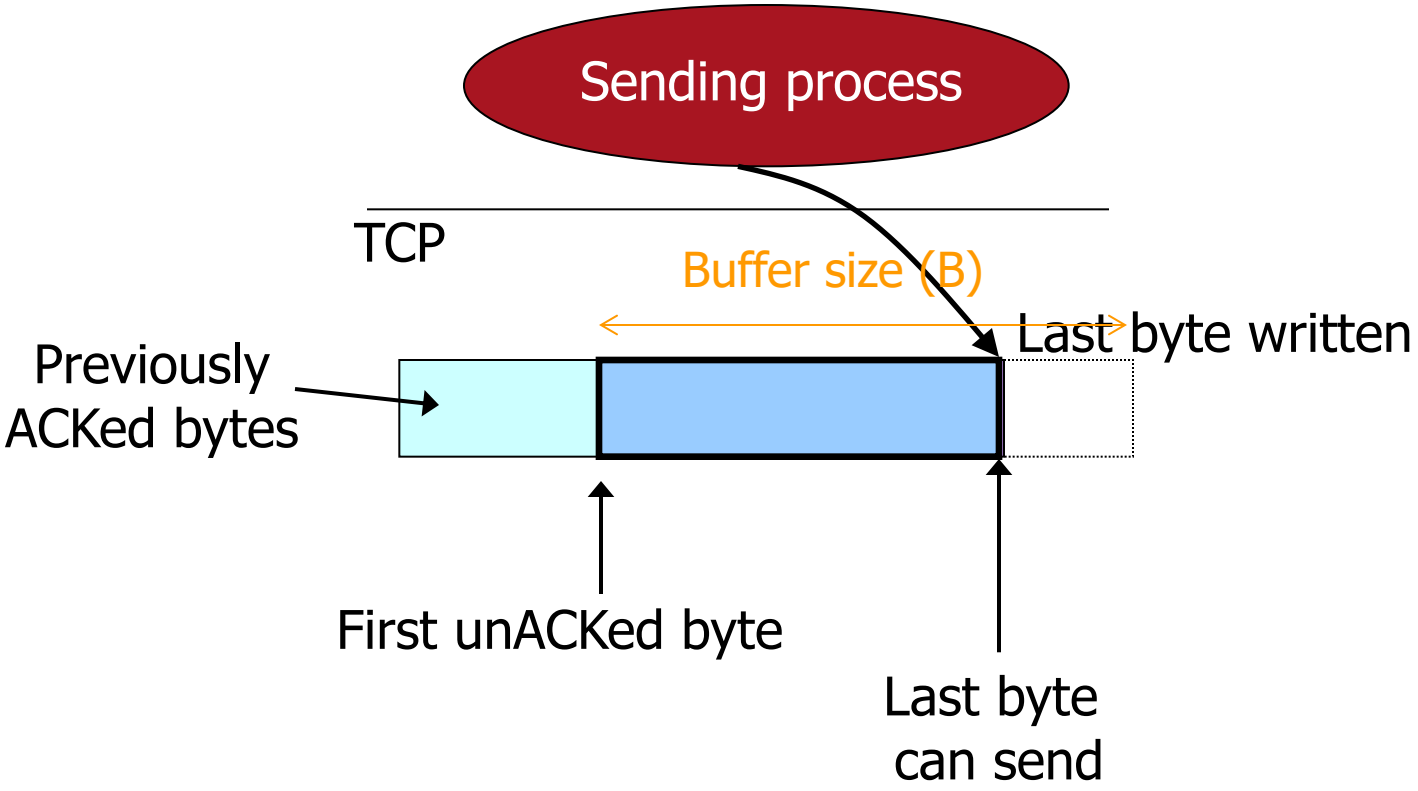
Flow control



Congestion control

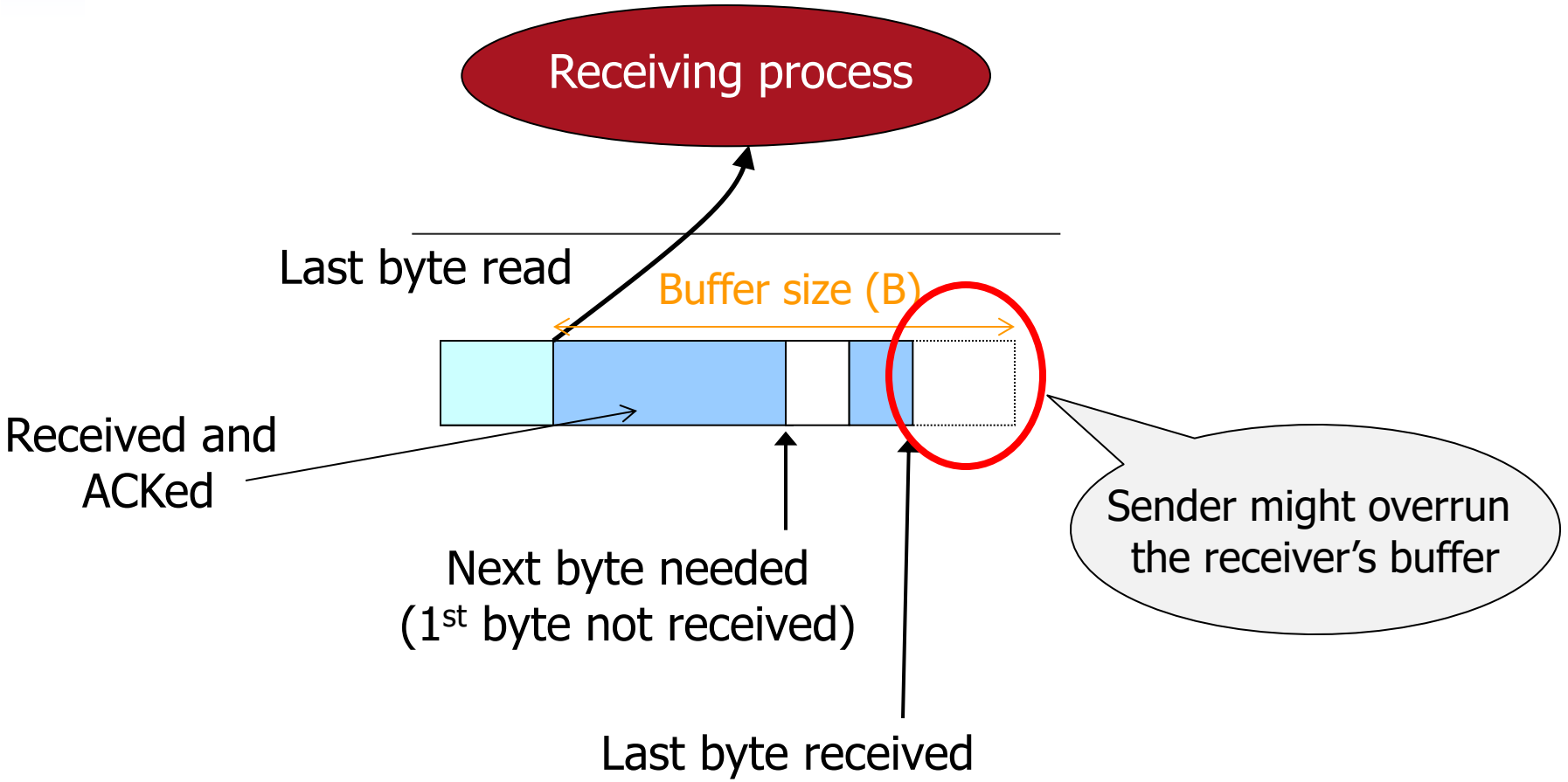


Sliding window at sender



How do we prevent the sender from overflowing the send buffer?

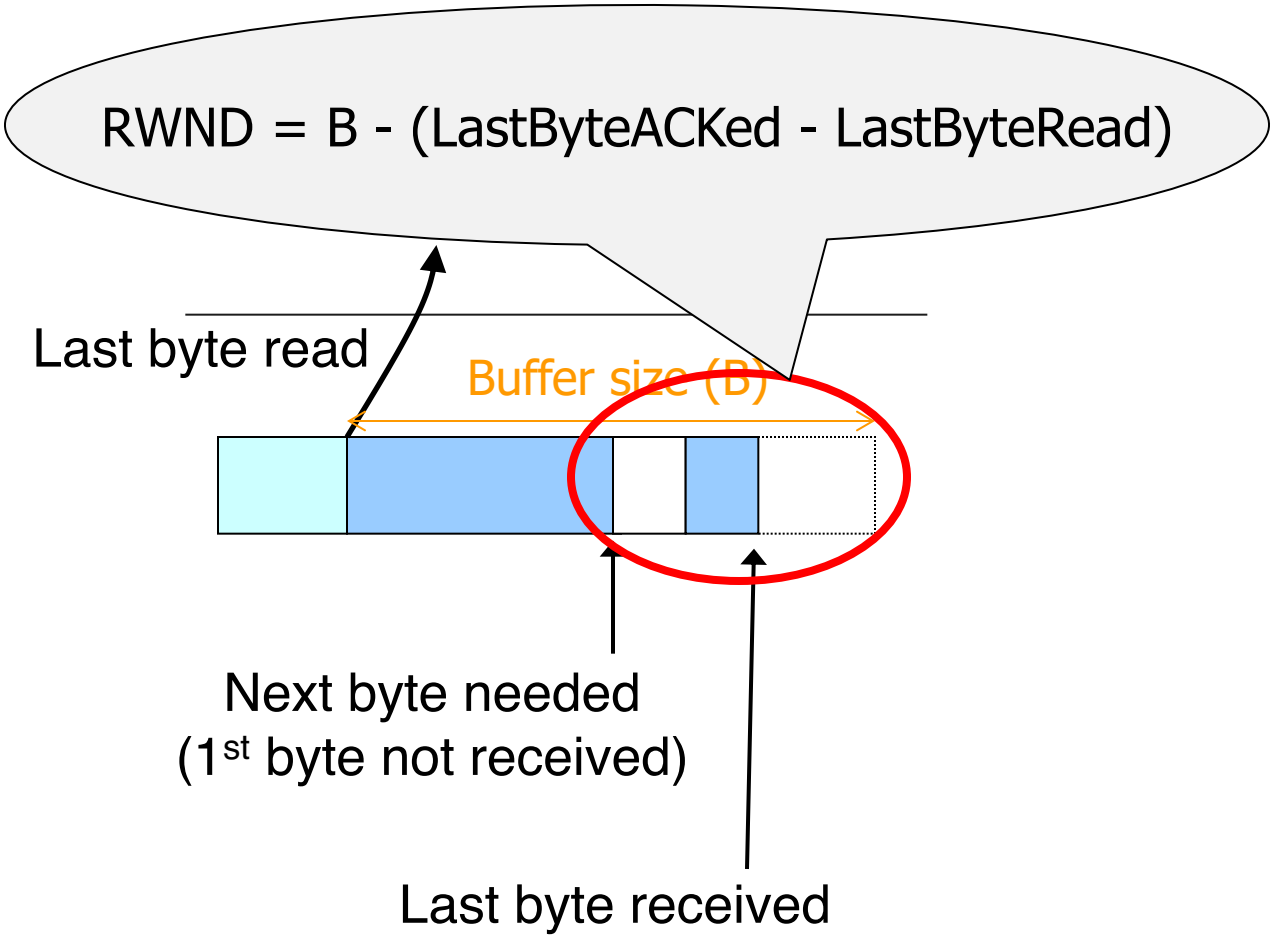
Sliding window at receiver



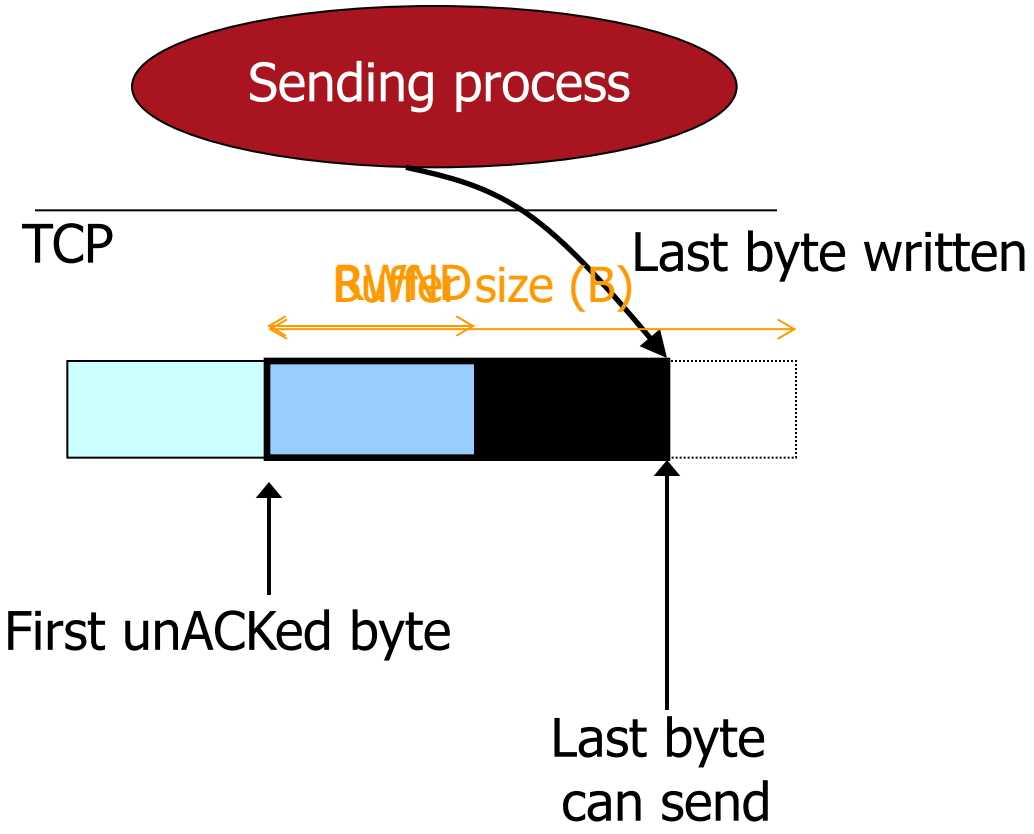
Solution: Flow Control

- Receiver uses an "Advertised Window" (RWND) to prevent sender from overflowing its window
  - Receiver indicates value of RWND in ACKs
  - Sender ensures that the total number of bytes in flight <= RWND

Sliding window at receiver



Sliding window at sender



Sliding window with flow control

- Sender: window advances when new data ACK'd
- Receiver: window advances as receiving process consumes data
- Receiver advertises to the sender where the receiver window currently ends ("righthand edge")
  - Sender agrees not to exceed this amount

- UDP does not have flow control
  - Data can be lost due to buffer overflow

Advertised window limits rate

- Sender can send no faster than RWND/RTT bytes/sec
- Receiver only advertises more space when it has consumed old arriving data
- What happens when RWND=0?
  - Sender keeps probing with one data bytes

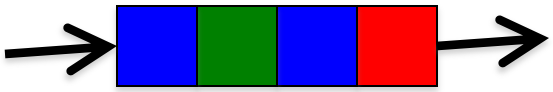
- In original TCP design, that was the sole protocol mechanism controlling sender's rate…

Agenda

- Transmission Control Protocol ✓
  - Reliable in-order delivery ✓
  - Connection-oriented ✓
  - Flow control ✓
  - Congestion control NEXT

Congestion

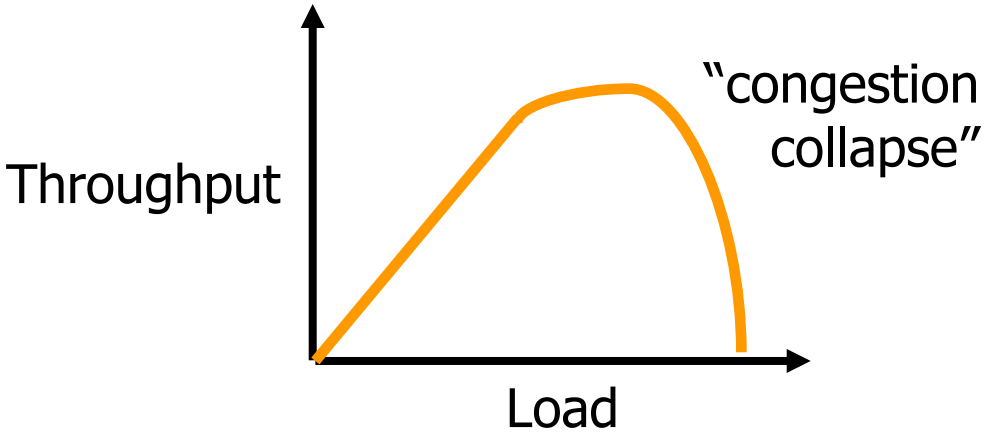
- Best-effort network does not "block" calls
  - So, they can easily become overloaded
  - Congestion == "Load higher than capacity"
- Examples of congestion
  - Link layer: Ethernet frame collisions
  - Network layer: full IP packet buffers
- Excess packets are simply dropped
  - And the sender can simply retransmit



queue

Congestion Collapse

- Easily leads to congestion collapse
  - Senders retransmit the lost packets
  - Leading to even greater load
  - … and even more packet loss



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

Van Jacobson's solution

- Reuse the sliding window
  - Controls number of packets in flight
- Sending rate ~Window/RTT
- Vary window size to control sending rate

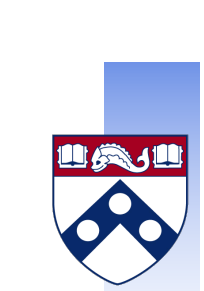
Windows to keep in mind

- Congestion Window: CWND
  - Bytes that can be sent without overflowing routers
  - Computed by sender using congestion control algo.
- Flow control window: RWND
  - Bytes that can be sent without overflowing receiver
  - Determined by the receiver and reported to the sender

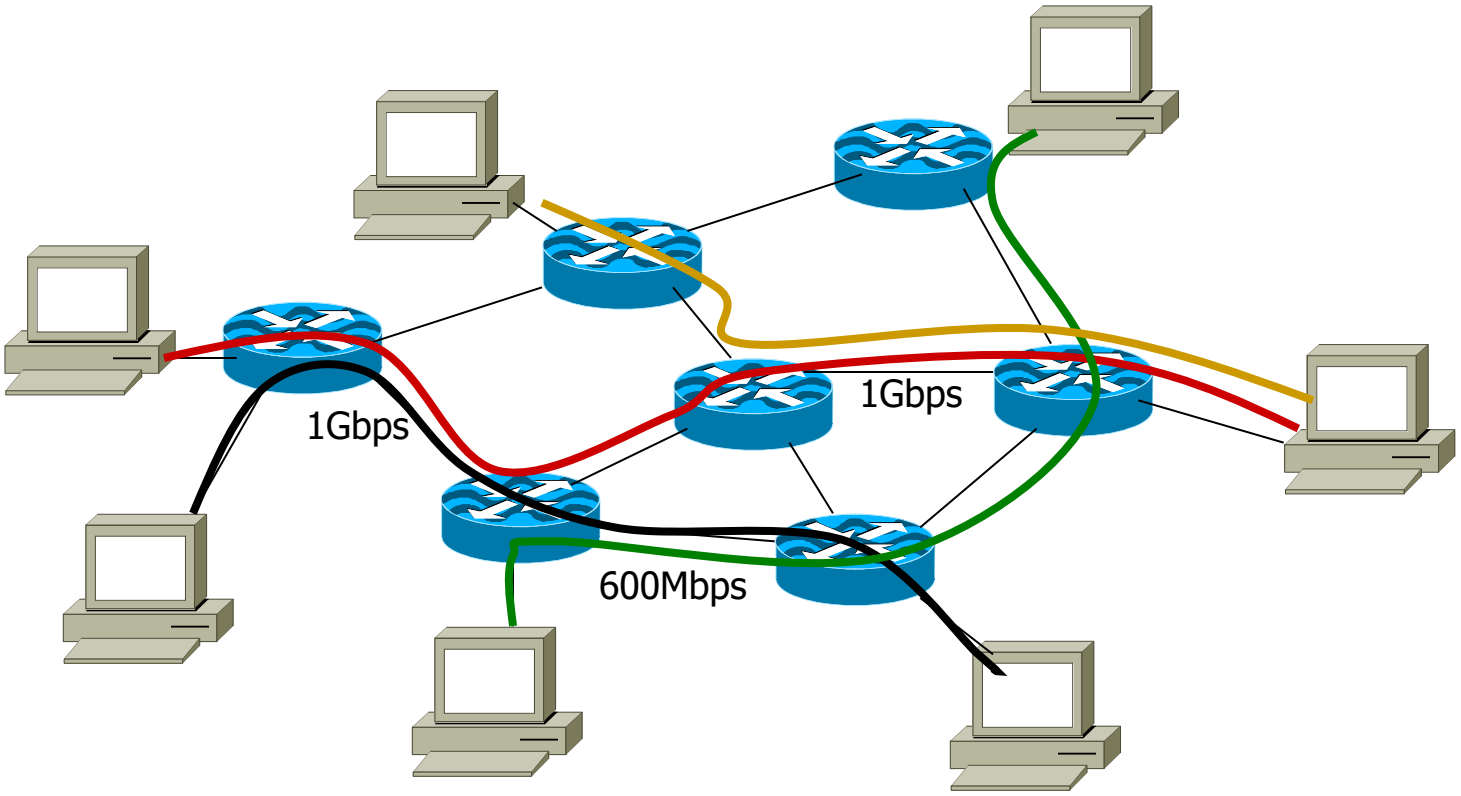
- Sender-side window = min {CWND, RWND}
  - Assume for remainder of lecture that RWND >> CWND

Note

- This lecture talks about CWND in units of MSS
  - MSS (Maximum Segment Size): the amount of payload data in a TCP packet
  - This is only for the simplicity of presentation
- Real implementations maintain CWND in bytes



A complex problem!



Congestion control is a resource allocation problem involving many flows, many links, and complicated global dynamics

Two Basic Questions

- How does the end host detect congestion?

Detecting congestion

- Packet delays
  - Tricky: noisy signal (delay often varies considerably)
- Routers tell end hosts when they're congested
- Packet loss
  - Fail-safe signal that TCP already has to detect
  - Complications:
    - Non-congestive loss (e.g., checksum errors)
    - Different types of loss

Different types of packet loss

- Duplicate ACKs: isolated loss
  - Still getting ACKs
  - How to tell difference between loss and reordering?
    - Triple duplicate acks
- Timeout: much more serious
  - Must have suffered several losses
- Will adjust rate differently for each case
  - Assume duplicate ACKs for now

Two Basic Questions

- How does the end host detect congestion?
- How fast should the end host send?
  - How can it discover available bandwidth?
  - How should it adjust to changes in available bandwidth?
  - How should it share with other flows?

Responding to Congestion

- Upon detecting congestion
  - Decrease the sending rate
- But, what if conditions change?
  - If more bandwidth becomes available,
  - … unfortunate to keep sending at a low rate
- Upon *not* detecting congestion
  - Increase sending rate, a little at a time
  - See if packets get through

TCP Rate adjustment

- Basic structure
  - Upon receipt of ACK (of new data): increase rate
  - Upon detection of loss: decrease rate
- How we increase/decrease the rate depends on the phase of congestion control we're in:
  - Discovering available bottleneck bandwidth vs.
  - Adjusting to bandwidth variations

Bandwidth discovery with "Slow Start"

- Goal: estimate available bandwidth
  - Start slow (for safety)
  - Ramp up quickly (for efficiency)
- Consider
  - RTT = 100ms, MSS = 1000bytes
  - Window size to fill 1Mbps of BW = 12.5 packets
  - Window size to fill 1Gbps = 12,500 packets
  - Either is possible!

Slow Start phase

- Sender starts at a slow rate, but increases exponentially until first loss
- Start with a small congestion window
  - Initially, CWND = 1
  - So, initial sending rate is MSS/RTT
- Double the CWND for each RTT with no loss

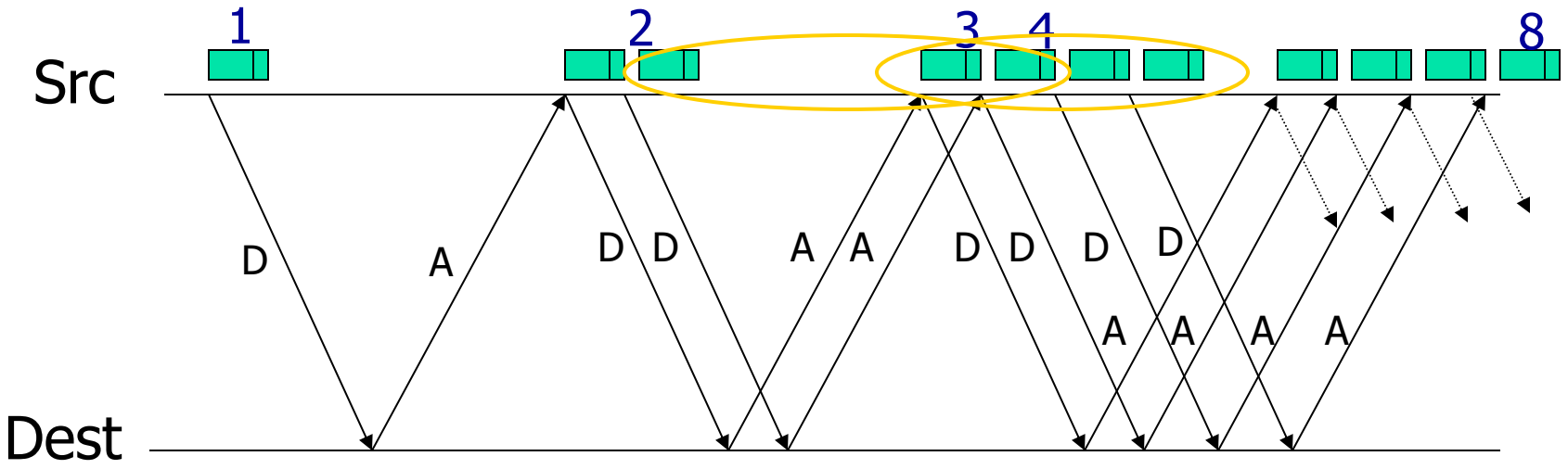
Slow Start in action

- For each RTT: double CWND
  - i.e., for each ACK, CWND += 1

Linear increase per ACK(CWND+1) ➔ exponential increase per RTT (2*CWND)

Slow Start in action

- For each RTT: double CWND
  - i.e., for each ACK, CWND += 1



When does Slow Start stop?

- Slow Start gives an estimate of available bandwidth
  - At some point, there will be loss
- Introduce a "slow start threshold" (ssthresh)
  - Initialized to a large value
- If CWND > ssthresh, stop Slow Start

Adjusting to varying bandwidth

- CWND > ssthresh
  - Stop rapid growth and focus on maintenance
- Now, want to track variations in this available bandwidth, oscillating around its current value
  - Repeated probing (rate increase) and backoff (decrease)

AIMD

- Additive increase
  - For each ACK, CWND = CWND+ 1/CWND
  - CWND is increased by one only if all segments in a CWND have been acknowledged
- Multiplicative decrease
  - On packet loss, CWND = CWND/2

Leads to the TCP "Sawtooth"