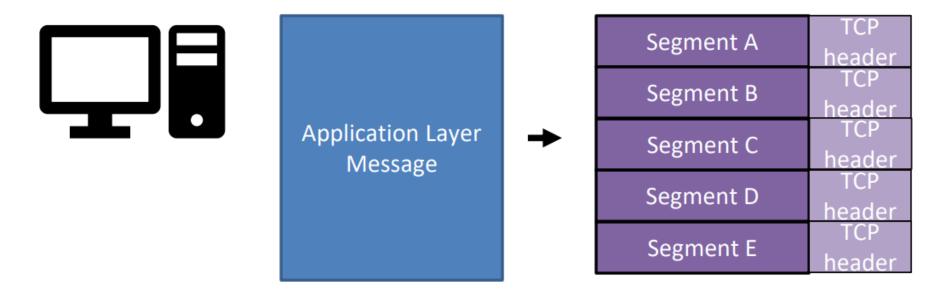
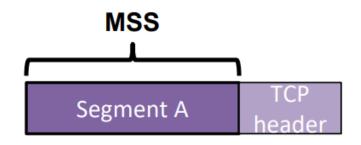
CSCI 466: Networks

TCP Flow Control, Timeout, Congestion Control

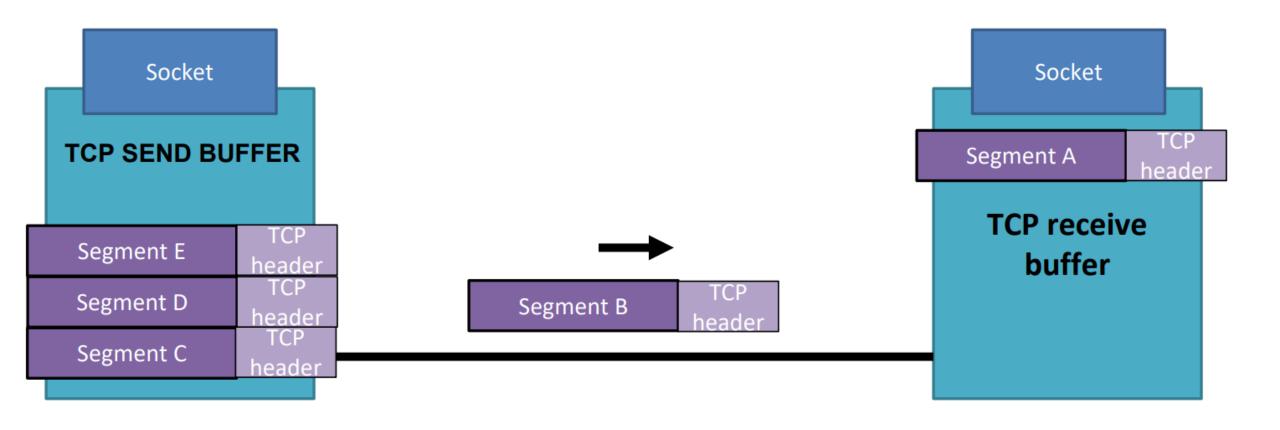
Reese Pearsall Fall 2023



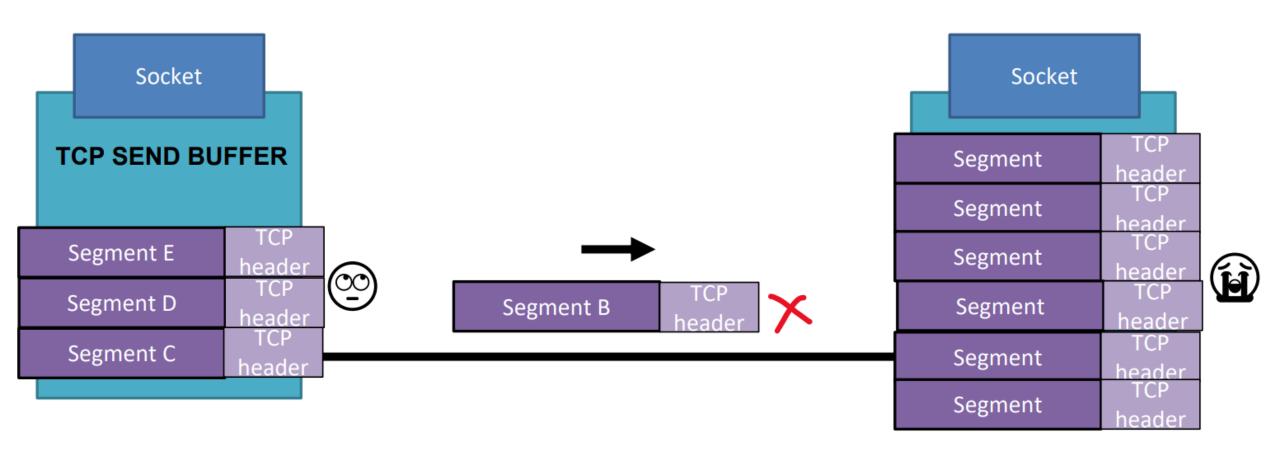
Application layer messages are split into smaller chunks called **segments**



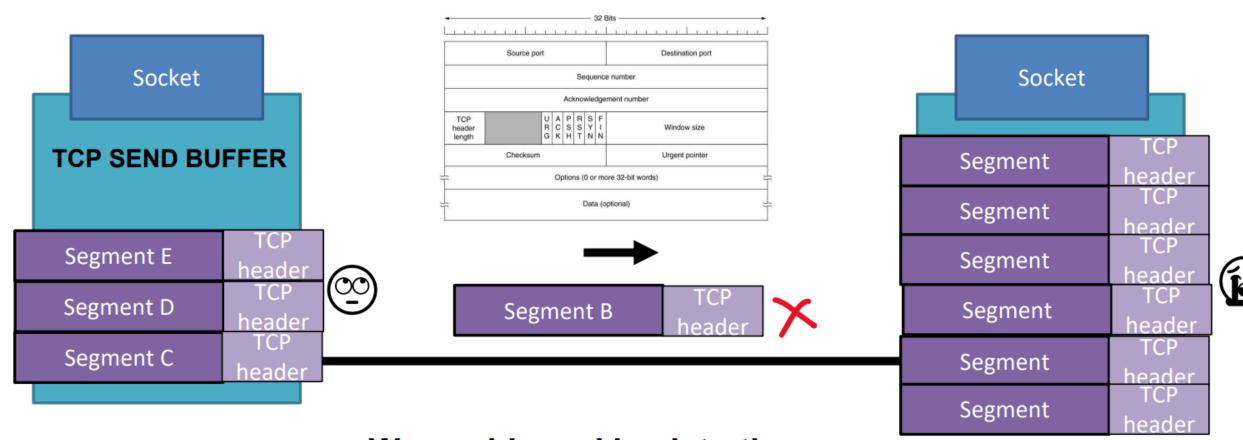
The size of these segments is determined by the maximum segment size (MSS)



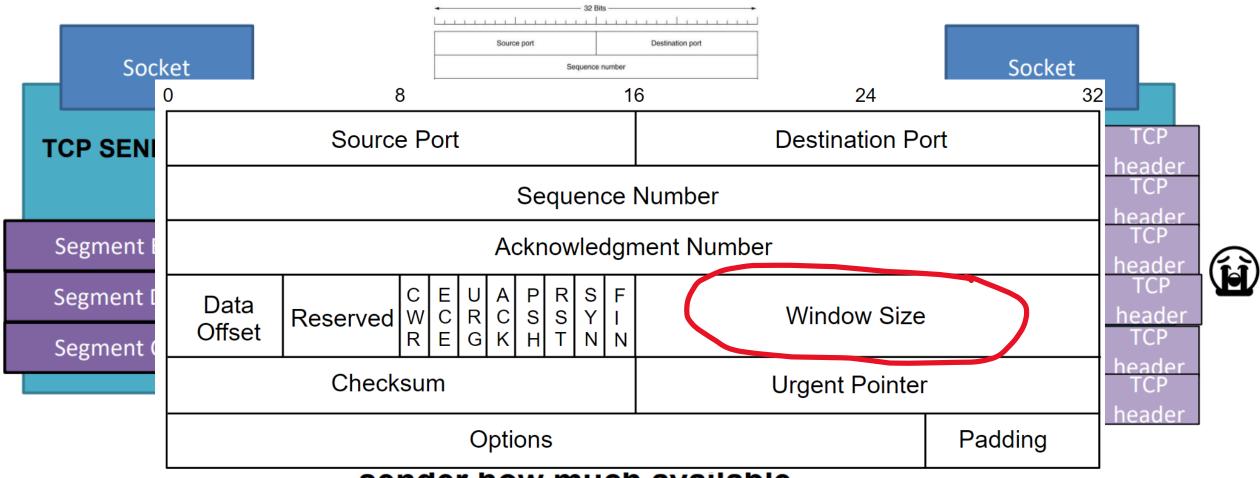
Applications read streams of data from a TCP buffer



How could we prevent something like this from happening?

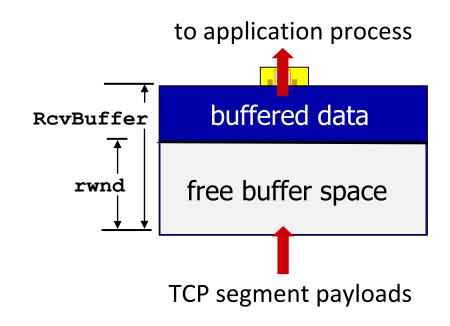


We could send back to the sender how much available space we have in our buffer!



sender how much available space we have in our buffer!

- TCP receiver "advertises" free buffer space in rwnd field in TCP header
 - RcvBuffer size set via socket options (typical default is 4096 bytes)
 - many operating systems auto-adjust
 RcvBuffer
- sender limits amount of unACKed ("in-flight") data to received rwnd
- guarantees receive buffer will not overflow



TCP receiver-side buffering

https://media.pearsoncmg.com/aw/ecs_kurose_compnetwork_7/c w/content/interactiveanimations/flow-control/index.html

TCP Timer

What is a good way to determine when to timeout? (aka the length of timer)

- 1. Too short: premature timeout, unnecessary retransmissions
- 2. Too long: slow reaction to segment loss

The TCP timeout value should around the same time it take to

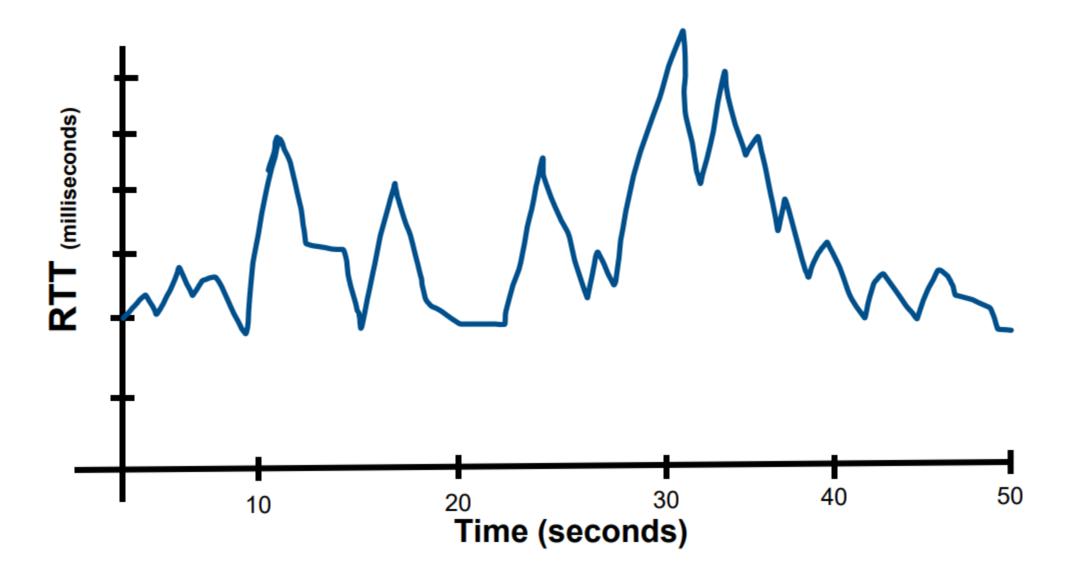
TCP Timer

What is a good way to determine when to timeout? (aka the length of timer)

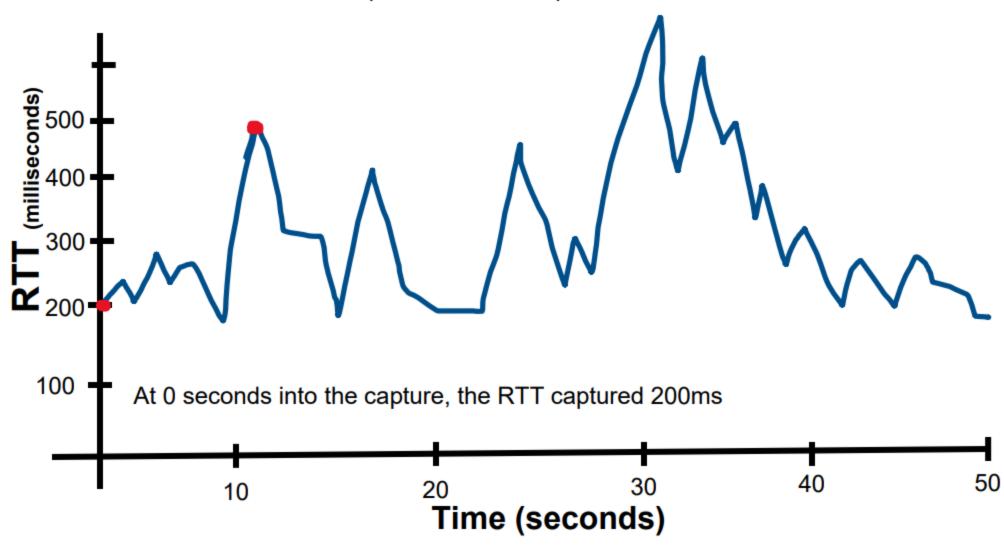
- 1. Too short: premature timeout, unnecessary retransmissions
- 2. Too long: slow reaction to segment loss

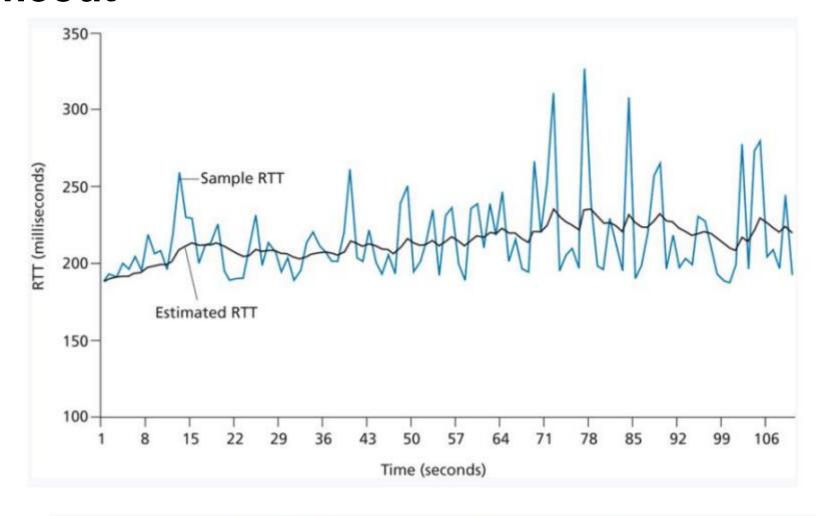
The TCP timeout value should around the same time it take to receive an acknowledgement on a sent packet (on average)

Let's consider setting it to be a dynamic value!



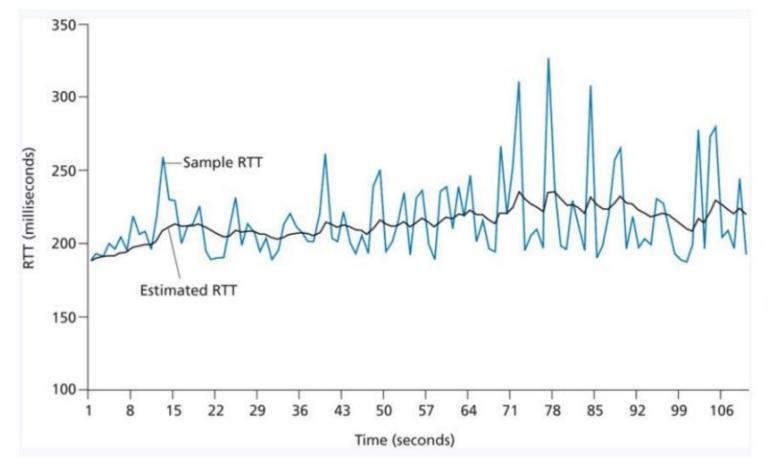
11 seconds into the capture, the RTT captured 500ms





EstimatedRTT =
$$(1 - \alpha) \cdot \text{EstimatedRTT} + \alpha \cdot \text{SampleRTT}$$

a=0.125



In addition, we also want some kind of safety margin

DevRTT =
$$(1-\beta)$$
 *DevRTT + β * | SampleRTT-EstimatedRTT | (typically, β = 0.25)

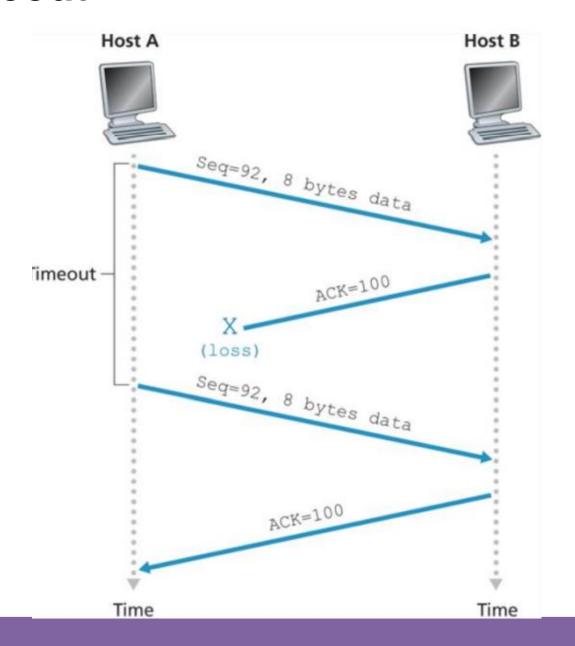
TimeoutInterval =

EstimatedRTT + 4*DevRTT

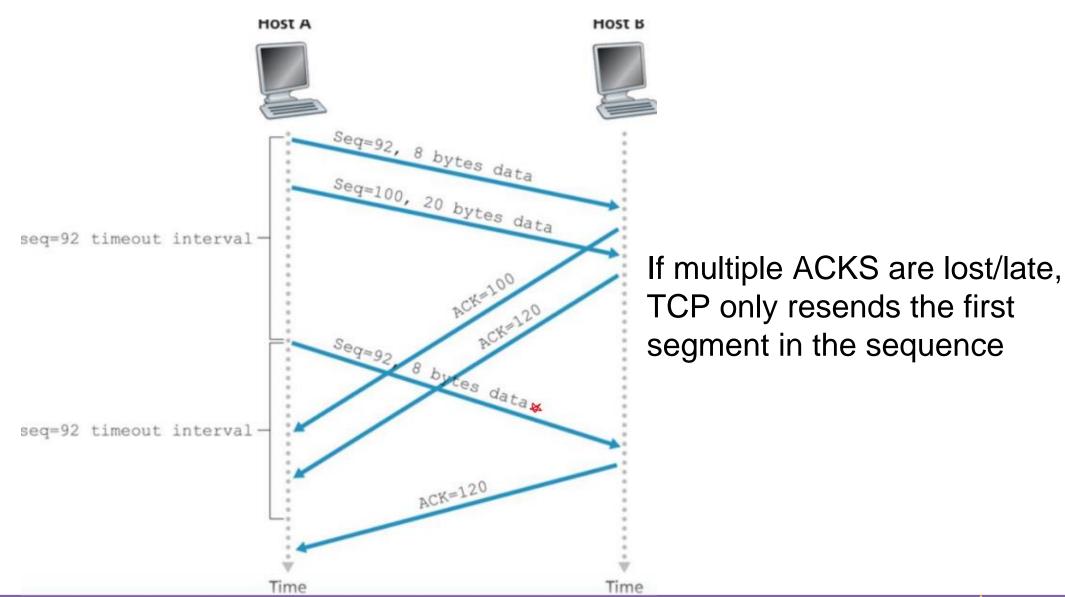
(safety margin)

EstimatedRTT =
$$(1 - \alpha) \cdot \text{EstimatedRTT} + \alpha \cdot \text{SampleRTT}$$

a=0.125



TCP retransmits on ACK loss

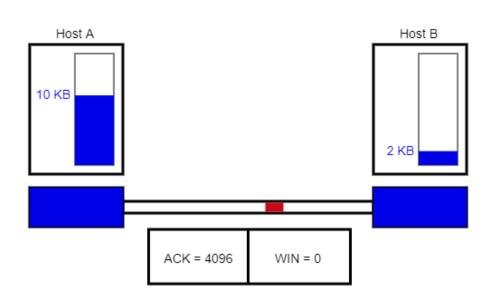


Event	TCP Receiver Action
Arrival of in-order segment with expected sequence number. All data up to expected sequence number already acknowledged.	Delayed ACK. Wait up to 500 msec for arrival of another in-order segment. If next in-order segment does not arrive in this interval, send an ACK.
Arrival of in-order segment with expected sequence number. One other in-order segment waiting for ACK transmission.	Immediately send single cumulative ACK, ACKing both in-order segments.
Arrival of out-of-order segment with higher-than-expected sequence number. Gap detected.	Immediately send duplicate ACK, indicating sequence number of next expected byte (which is the lower end of the gap).
Arrival of segment that partially or completely fills in gap in received data.	Immediately send ACK, provided that segment starts at the lower end of gap.

Specifics about when/how/why to send ACKs are described in TCP Congestion Control's **RFC** (request for comments)

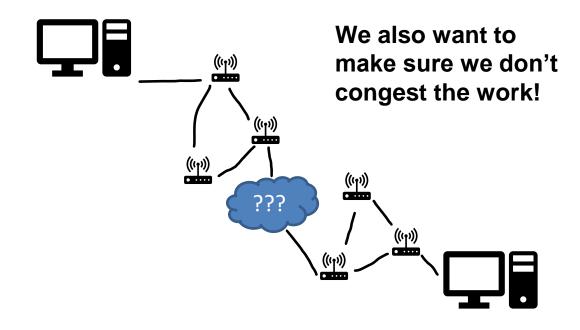
9293

5681



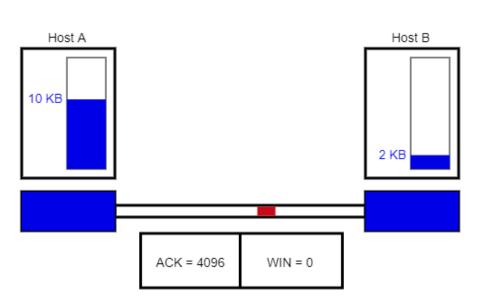
TCP sends back amount of available buffer space in the receiver

This helps make sure we don't overwhelm the receiver

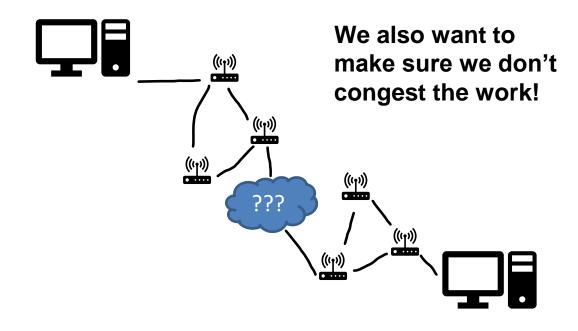


Issues:

- If the network is congested, we want to slow down our sending rate
- If the network is not congested, we should try to send more stuff



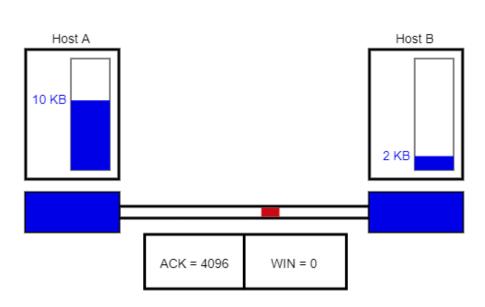
TCP sends back amount of available buffer space in the receiver This helps make sure we don't overwhelm the receiver



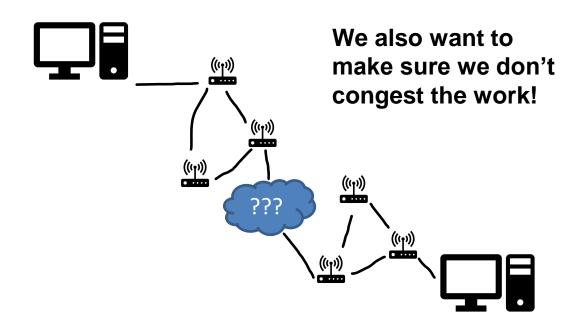
Issues:

- If the network is congested, we want to slow down our sending rate
- If the network is not congested, we should try to send more stuff

From the sender perspective, how could we measure how congested the network is?



TCP sends back amount of available buffer space in the receiver This helps make sure we don't overwhelm the receiver



Issues:

- If the network is congested, we want to slow down our sending rate
- If the network is not congested, we should try to send more stuff

Some ways we could measure how congested the network is

- -See how many dropped packets we are getting
- -Amount of duplicate ACKs received
- -Amount of UnAcked packets

TCP sender also has a **congestion window (cwnd)**, which controls the amount of unAck'd that can be sent out

TCP is **self-clocking**

(It uses acknowledgements to trigger, or clock, its increase in congestion window size)

The amount of unacknowledged data at a sender may not exceed the *minimum* of the congestion window and receiving window

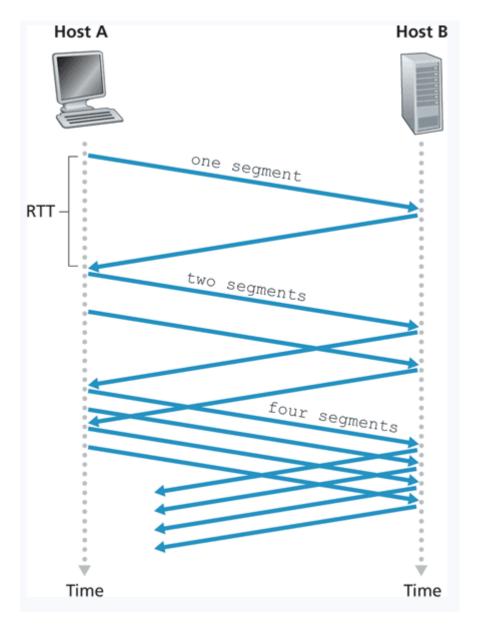
We also want to make sure we don't congest the work!

LastByteSent - LastByteAcked ≤ min{cwnd, rwnd}

TCP Algorithm to prevent network congestion

- Slow Start
- Congestion Avoidance
- Fast recovery

Start sending slow, but exponentially grows up to a *threshold*

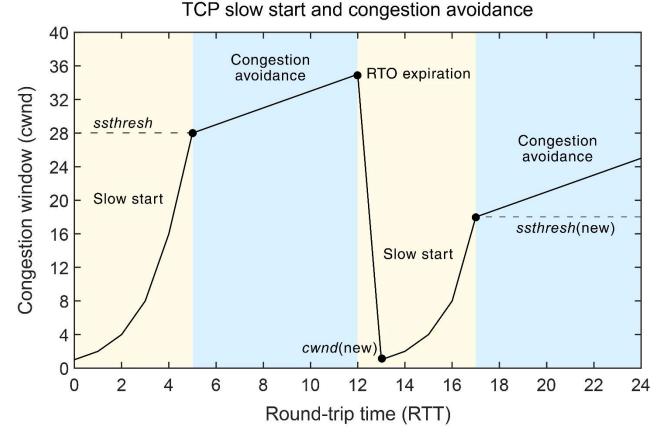


TCP Algorithm to prevent network congestion

- Slow Start
- Congestion Avoidance
- Fast recovery

Linearly increase congestion window for each ACK received

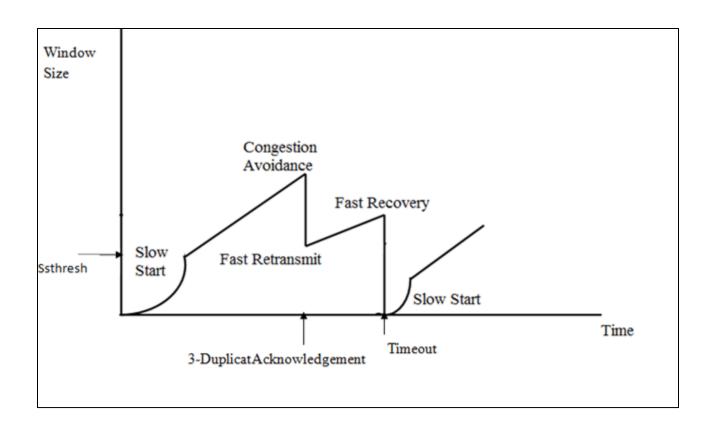
When a loss event occurs, significantly decrease congestion window and slow down transmission rate, and enter **fast recovery**

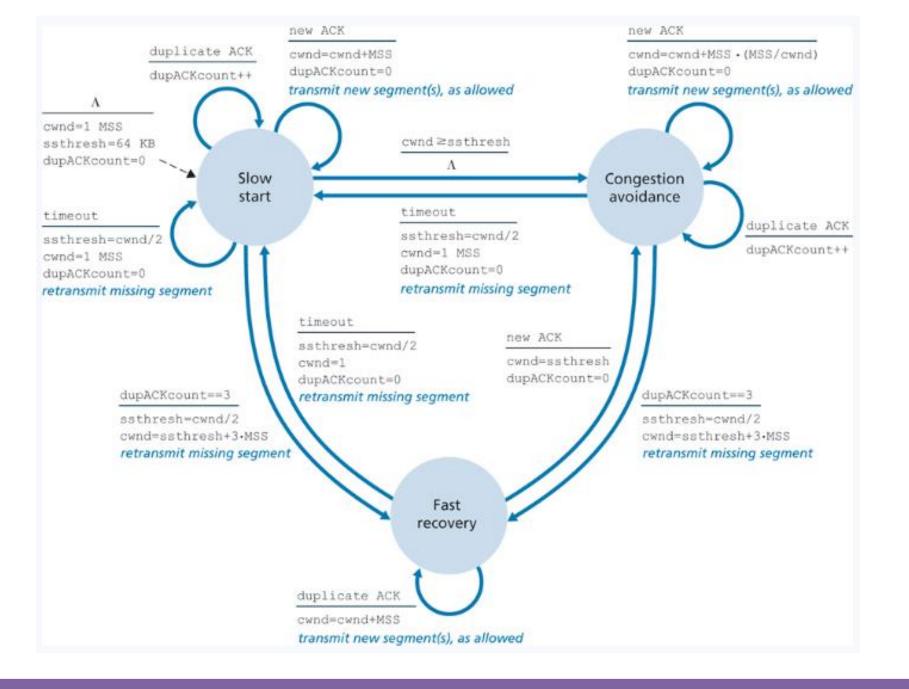


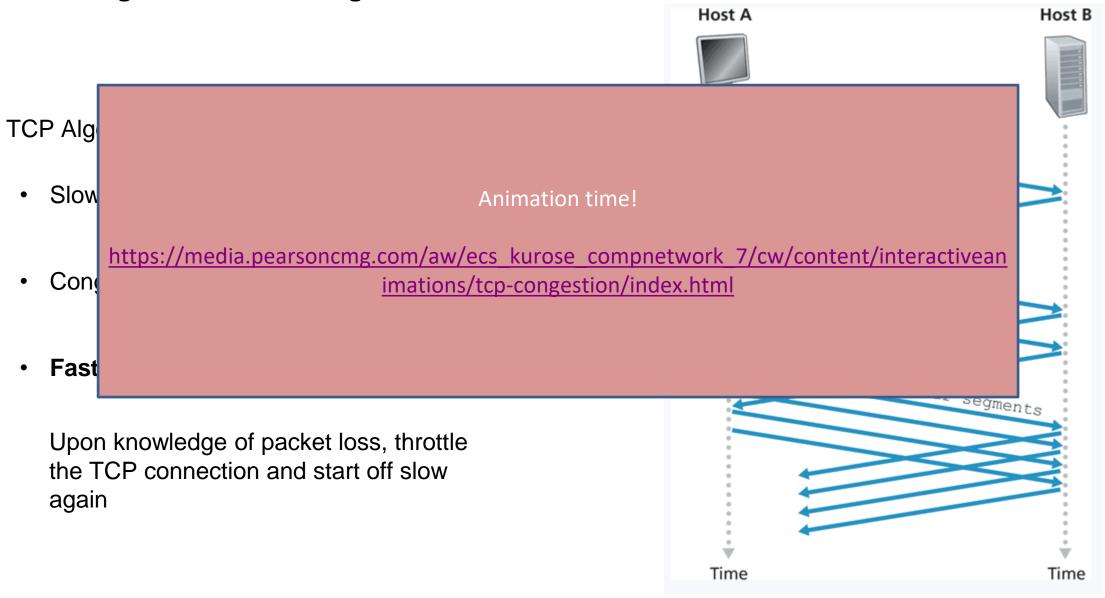
TCP Algorithm to prevent network congestion

- Slow Start
- Congestion Avoidance
- Fast recovery

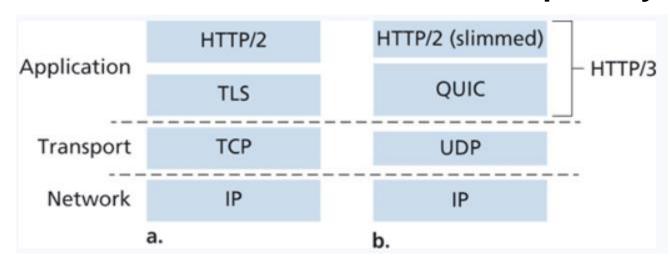
Upon knowledge of packet loss, throttle the TCP connection and start off slow again



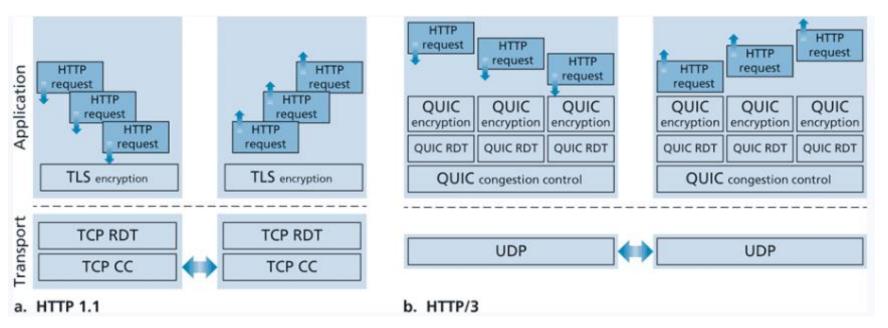




Current transport layer implementation



Transport layer protocols and congestion control is still a heavily researched area!



FIN

