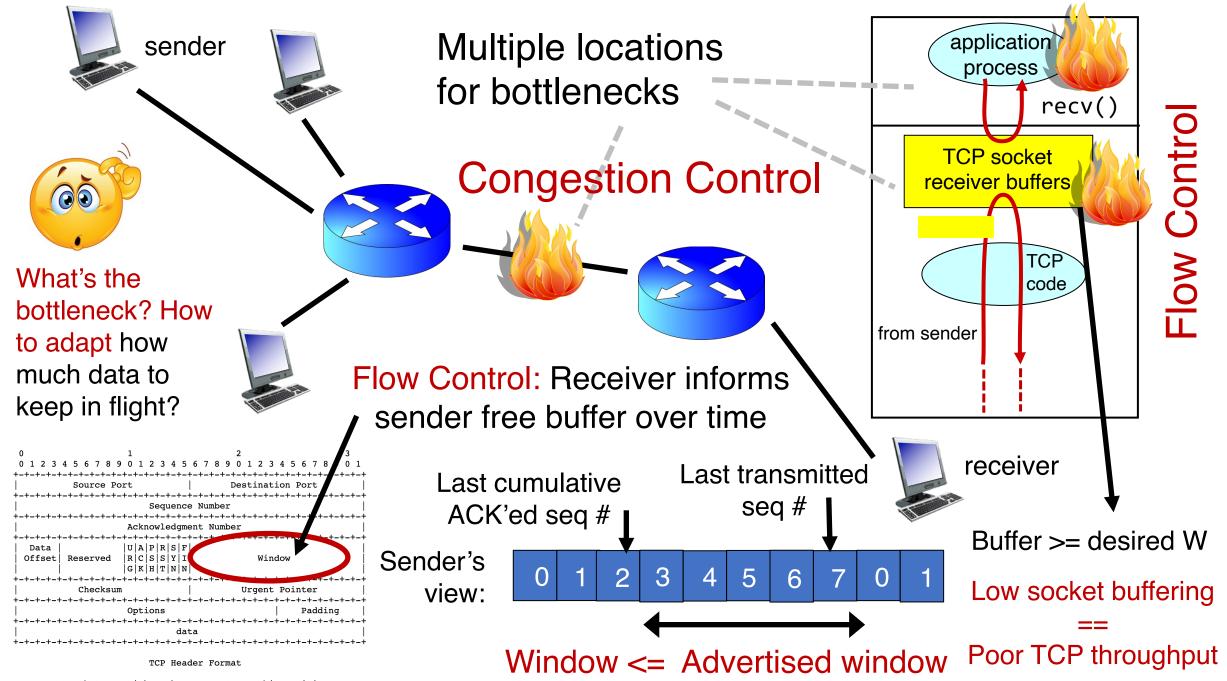
CS 352 Congestion Control (Part 1)

Lecture 16

http://www.cs.rutgers.edu/~sn624/352-F22

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Note that one tick mark represents one bit position.

Congestion control

The approach that the Internet takes is to use a distributed algorithm to converge to an efficient and fair outcome.

Each endpoint acts by itself. No central vantage point or control. Use whatever bottleneck capacity available, even with a single TCP connection.

Share bottleneck capacity equitably



Sense and React

Signals and Knobs in Congestion Control

Signals

- Packets being ACK'ed
- Packets being dropped (e.g. RTO fires)
- Packets being delayed (RTT)
- Rate of incoming ACKs

Implicit feedback signals measured directly at sender. (There are also explicit signals that the network might provide.)

Knobs

- What can you change to "probe" the available bottleneck capacity?
- Suppose receiver buffer is unbounded:
- Increase window/sending rate: e.g., add x or multiply by a factor of x
- Decrease window/sending rate: e.g., subtract x or reduce by a factor of x

Sense and react, sure...but how?

- Where do you want to be?
 - The steady state
- How do you get there?
 - Congestion control algorithms
- Sense accurately
- React proportionately

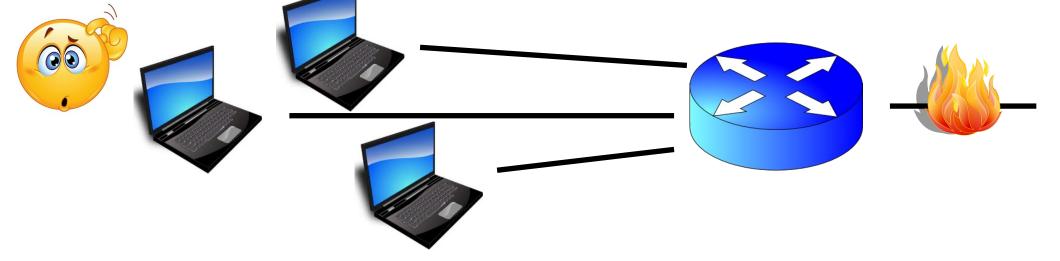


The Steady State

Efficiency of a single TCP conversation

What does efficiency look like?

 Suppose we want to achieve an efficient outcome for one TCP conversation by observing network signals from the endpoint

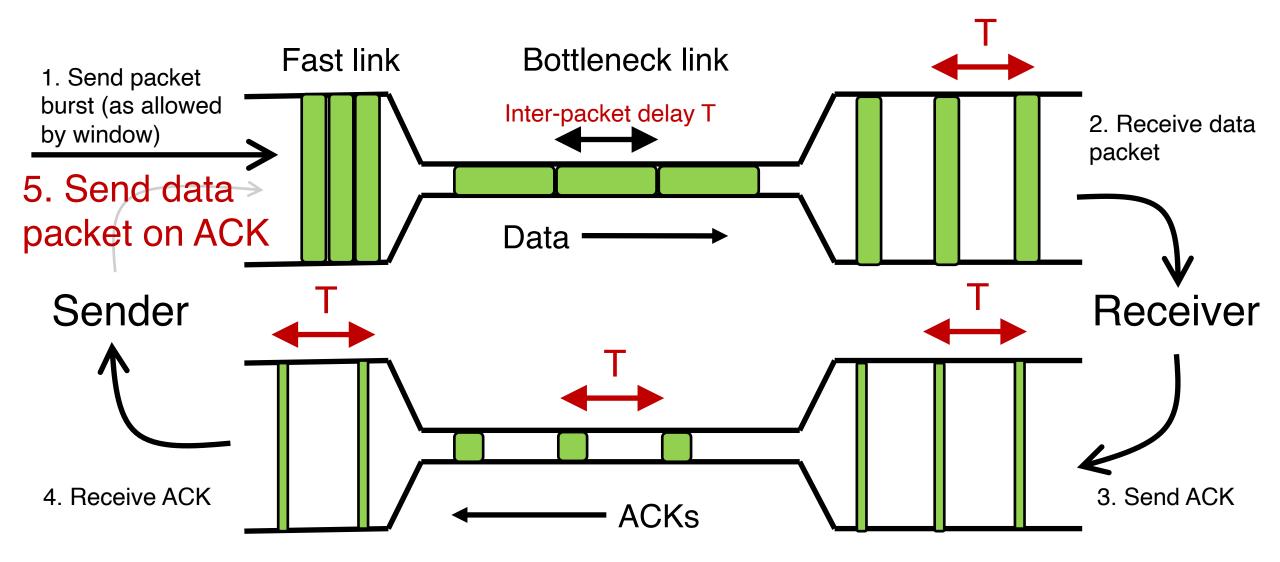


- Q: How should the endpoint behave at steady state?
- Challenge: bottleneck link is remotely located

Steady state: Ideal goal

- High sending rate: Use the full capacity of the bottleneck link
- Low delay: Minimize the overall delay of packets to get to the receiver
 - Overall delay = propagation + queueing + transmission
 - Assume propagation and transmission components fixed
- "Low delay" reduces to low queueing delay
- i.e., don't push so much data into the network that packets have to wait in queues
- Key question: When to send the next packet?

When to send the next packet?



Rationale

- When the sender receives an ACK, that's a signal that the previous packet has left the bottleneck link (and the rest of the network)
- Hence, it must be safe to send another packet without congesting the bottleneck link
- Such transmissions are said to follow packet conservation
- ACK clocking: "Clock" of ACKs governs packet transmissions

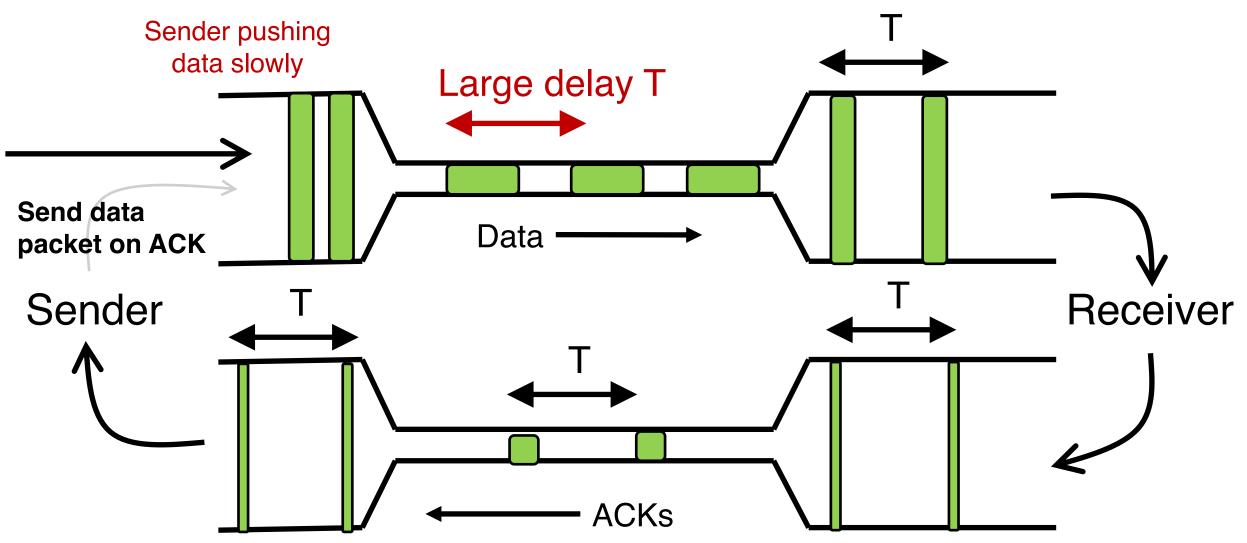
ACK clocking: analogy

- How to avoid crowding a grocery store?
- Strategy: Send the next waiting customer exactly when a customer exits the store



• However, this strategy alone can lead to inefficient use of resources...

ACK clocking alone can be inefficient



ACK clocking alone can be inefficient Sender pushing data slowly Large delay T Send data Data packet on ACK Sender Receiver

The sending rate should be high enough to keep the "pipe" full Analogy: a grocery store with only 1 customer in entire store If the store isn't "full", you're using store space inefficiently

Steady State of Congestion Control

- Send at the highest rate possible (to keep the pipe full)
- while being ACK-clocked (to avoid congesting the pipe)
- So, how to get to steady state?

Finding the Right Congestion Window

Let's play a game

- Suppose I'm thinking of a positive integer. You need to guess the number I have in mind.
- Each time you guess, I will tell you whether your number is smaller or larger than (or the same as) the one I'm thinking of
- Note that my number can be very large
- How would you go about guessing the number?

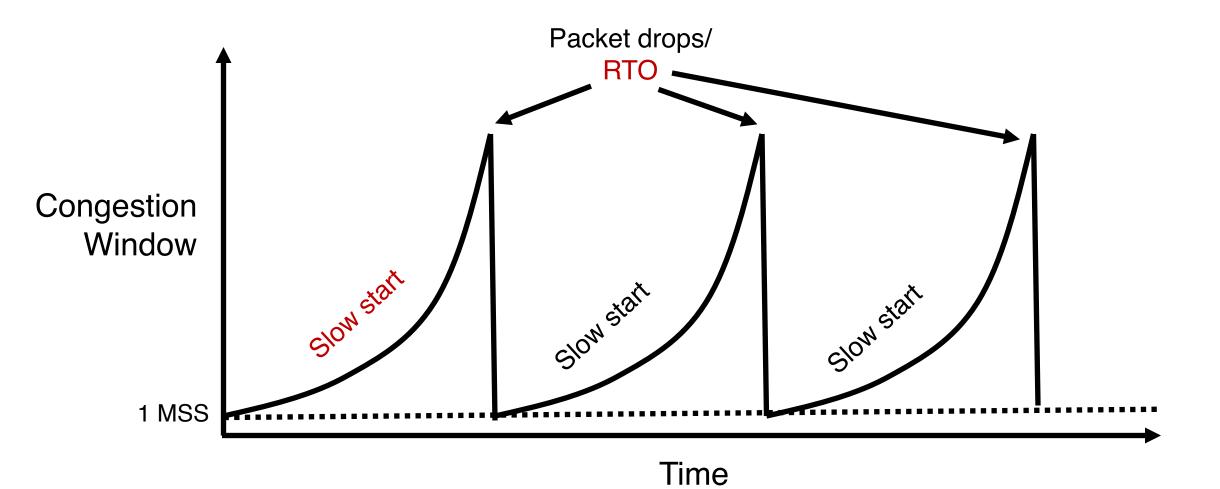
Finding the right congestion window

- TCP congestion control algorithms solve a similar problem!
- There is an unknown bottleneck link rate that the sender must match
- If sender sends more than the bottleneck link rate:
 - packet loss, delays, etc.
- If sender sends less than the bottleneck link rate:
 - all packets get through; successful ACKs

Quickly finding a rate: TCP slow start

Payload Host B Host A Initially cwnd = 1 MSS MSS is "maximum segment size" MSS one segment • Upon receiving an ACK of each MSS, increase the cwnd by 1 MSS RTT two segments Effectively, double cwnd every RTT four segments Initial rate is slow but ramps up exponentially fast • On loss (RTO), restart from cwnd := 1 time MSS

Behavior of slow start



Slow start has problems

- Congestion window increases too rapidly
 - Example: suppose the "right" window size cwnd is 17
 - cwnd would go from 16 to 32 and then dropping down to 1
 - Result: massive packet drops
- Congestion window decreases too rapidly
 - Suppose the right cwnd is 31, and there is a loss when cwnd is 32
 - Slow start will resume all the way back from cwnd 1
 - Result: unnecessarily low throughput
- Instead, perform finer adjustments of cwnd based on signals

Use slow start mainly at the beginning

• You might accelerate your car a lot when you start, but you want to make only small adjustments after.

• Want a smooth ride, not a jerky one!

- Slow start is a good algorithm to get close to the bottleneck link rate when there is little info available about the bottleneck, e.g., starting of a connection
- Once close enough to the bottleneck link rate, use a different set of strategies to perform smaller adjustments to cwnd
 - Called TCP congestion avoidance

TCP Congestion Avoidance

Two congestion control algorithms

TCP New Reno

• The most studied, classic "textbook" TCP algorithm

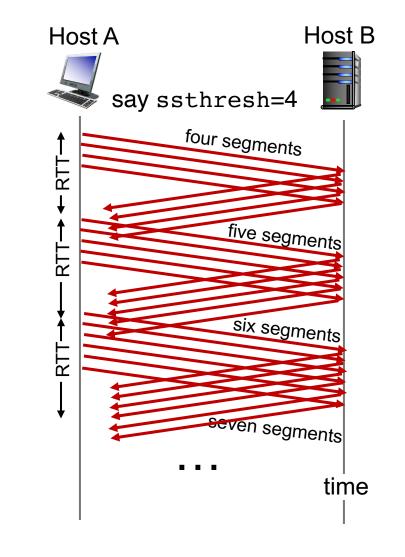
TCP BBR

- Recent algorithm developed & deployed by Google
- The primary knob is congestion
 The primary knob is sending rate window
- The primary signal is packet loss (RTO)
- Adjustment using additive increase

- The primary signal is rate of incoming ACKs
- Adjustment using gain cycling and filters

TCP New Reno: Additive Increase

- Remember the recent past to find a good estimate of link rate
- The last good cwnd without packet drop is a good indicator
 - TCP New Reno calls this the slow start threshold (ssthresh)
- Increase cwnd by 1 MSS every RTT after cwnd hits ssthresh
 - Effect: increase window additively per RTT

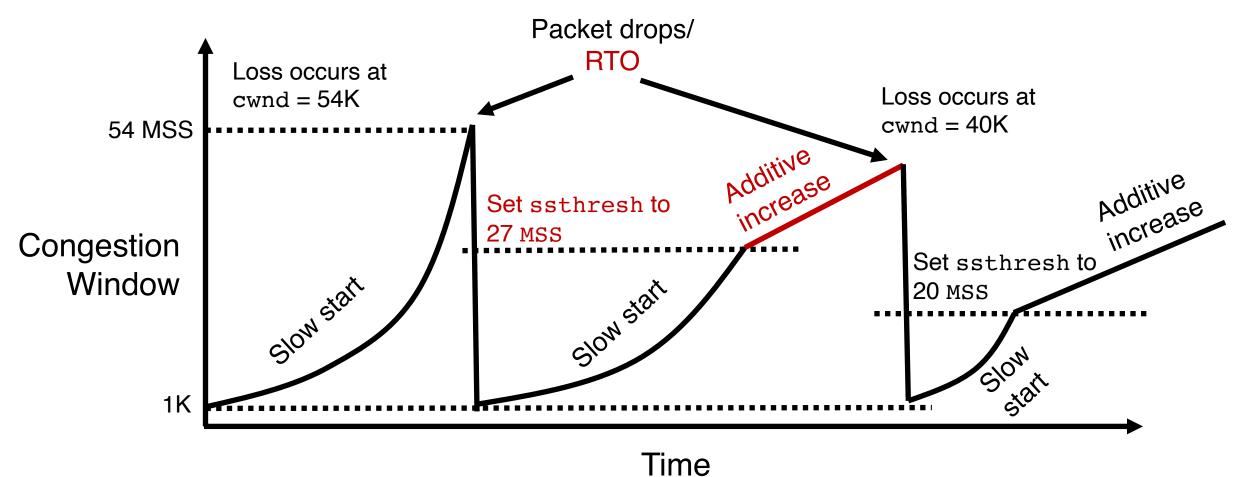


TCP New Reno: Additive increase

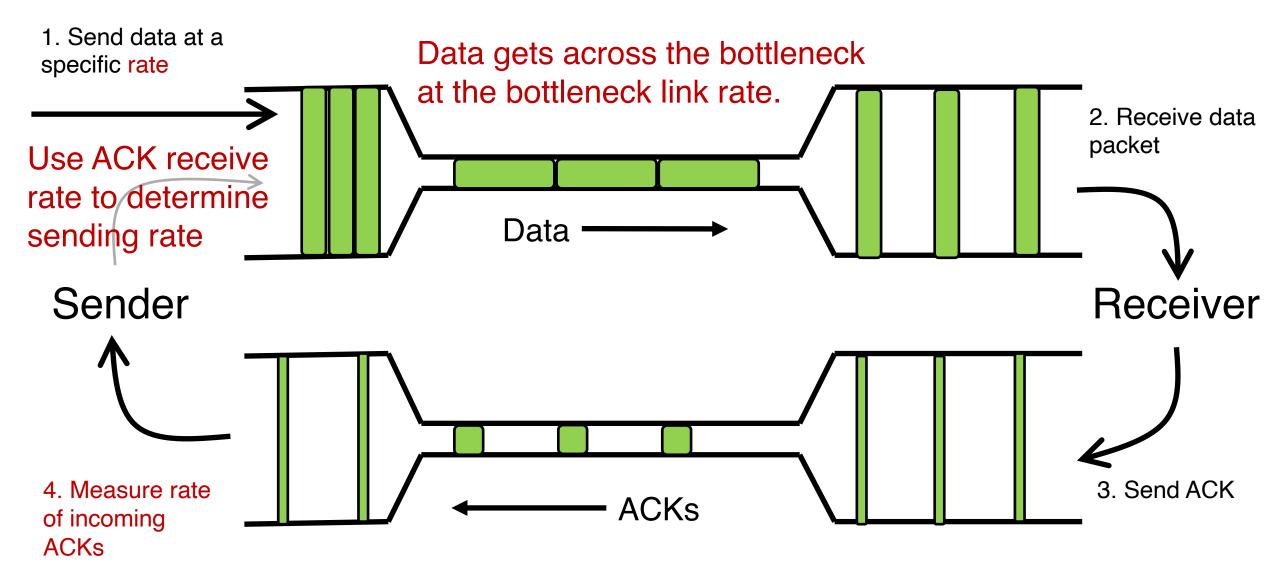
- Start with ssthresh = 64K bytes (TCP default)
- Do slow start until ssthresh
- Once the threshold is passed, do additive increase
 - Add one MSS to cwnd for each cwnd worth data ACK'ed
 - For each MSS ACK'ed, cwnd = cwnd + (MSS * MSS) / cwnd
- Upon a TCP timeout (RTO),
 - Set cwnd = 1 MSS
 - Set ssthresh = max(2 * MSS, 0.5 * cwnd)
 - i.e., the next linear increase will start at half the current cwnd

Behavior of Additive Increase

Say MSS = 1 KByte Default ssthresh = 64KB = 64 MSS



TCP BBR: finding the bottleneck link rate



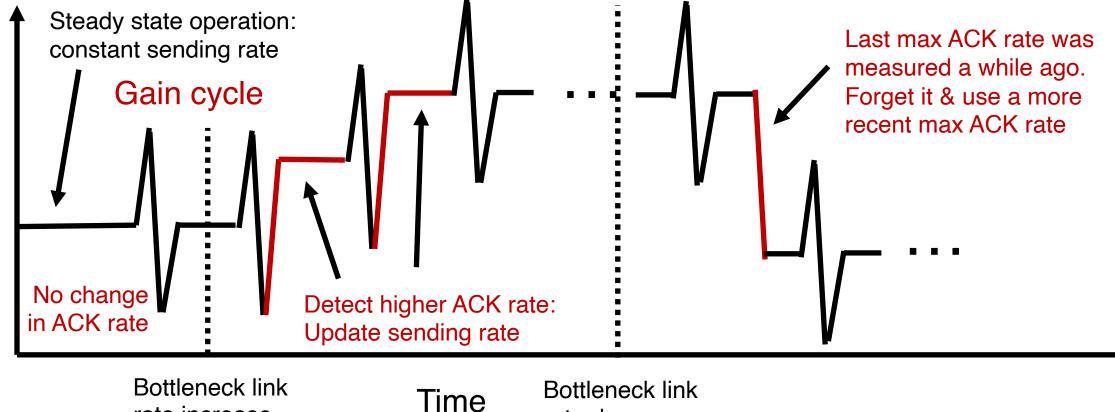
TCP BBR: finding the bottleneck link rate

Assuming that the link rate of the bottleneck

- == the rate of data getting across the bottleneck link
- == the rate of data getting to the receiver
- == the rate at which ACKs are generated by the receiver
- == the rate at which ACKs reach the sender
- Measuring ACK rate provides an estimate of bottleneck link rate
- BBR: Send at the maximum ACK rate measured in the recent past
 - Update max with new bottleneck rate estimates, i.e., larger ACK rate
 - Forget estimates last measured a long time ago
 - Incorporated into a rate filter

TCP BBR: Adjustments by gain cycling

• BBR periodically increases its sending rate by a gain factor to see if the link rate has increased (e.g., due to a path change)



rate increase

Sending rate

Summary: Getting to Steady State

- Want to get to highest sending rate that doesn't congest the bottleneck link
- Slow start: Exponential increase towards a reasonable estimate of link rate
- Congestion avoidance: milder adjustments to get close to correct link rate estimate.
- TCP New Reno: additive increase
- TCP BBR: gain cycling and filters