Lesson 05-04:
TCP Congestion Control

CS 326E Elements of Networking

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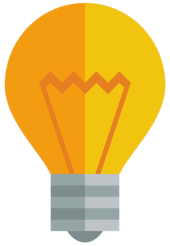
Internet Reference Model



Example Protocols	Layer	Responsible for
FTP, HTTP, SMTP	Application	application specific needs
TCP, UDP	Transport	process to process data transfer
IP	Network	host to host data transfer across different network
Ethernet, WiFi	Link	data transfer between physically adjacent nodes
802.3 PHY	Physical	bit-by-bit or symbol-by-symbol delivery

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Outline

1. Approaches to Congestion Control

Congestion control has 2 approaches

- **First, solely based on sender's detection**
  - Loss-based: Increase sending rate until a loss (timeout) and then cut back
  - Delay-based: Do the same until RTT reaches $RTT_{congested}$

- **Second, network assisted approach**
  - Sender, network core (routers), and the receiver all participates

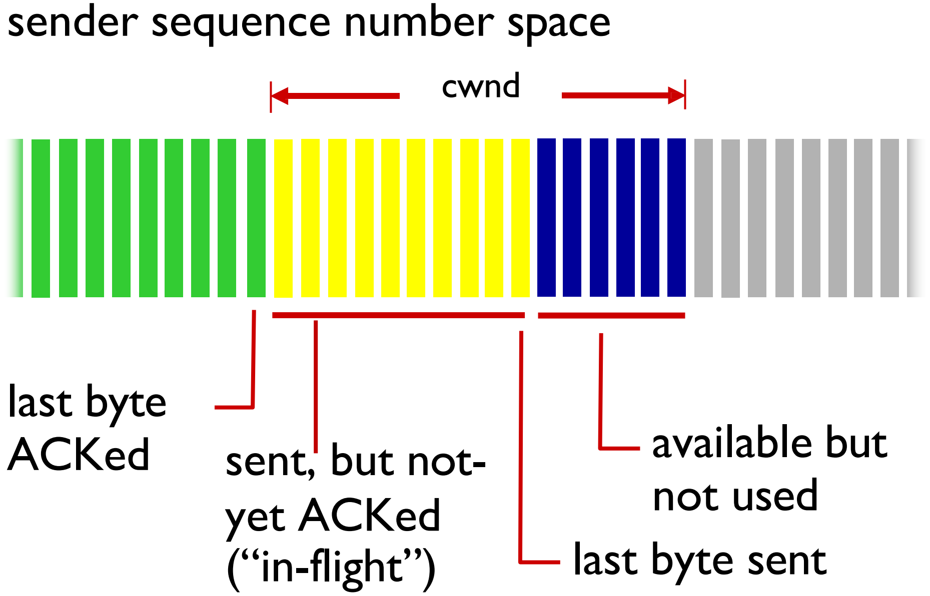
Let's first look at the loss-based approach!

- AIMD
- TCP CUBIC

Outline

1. Approaches to Congestion Control
2. TCP's AIMD

TCP sending rate is limited by congestion window cwnd



$$\text{TCP sending rate} \sim \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

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

True/False? cwnd is a fixed value

How should we adjust cwnd?

cwnd is dynamically adjusted in response to observed congestion

We need to probe what the optimal sending rate is at the moment!

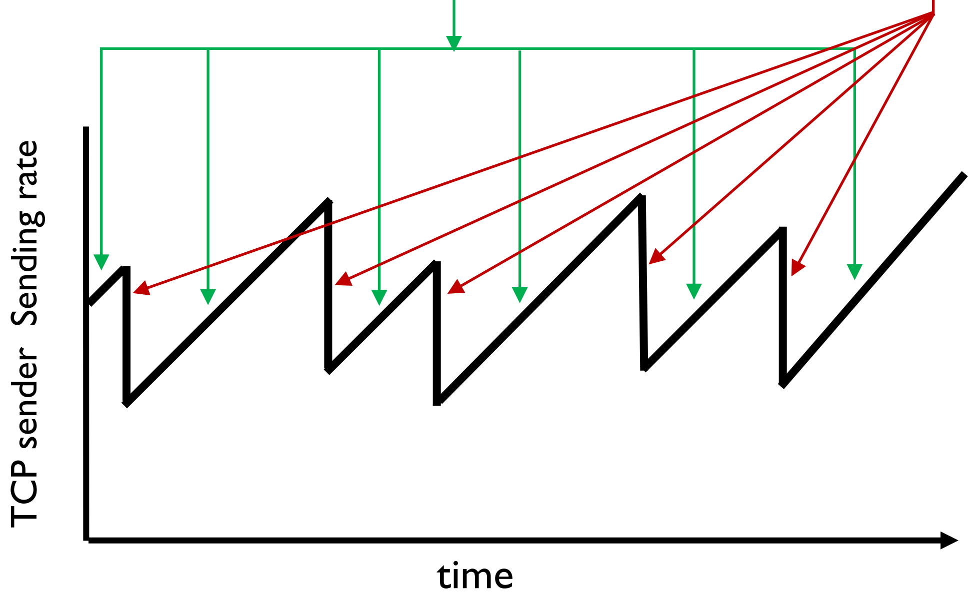
AIMD: sender increases sending rate until packet loss then decrease sending rate on loss

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event



AIMD sawtooth behavior: probing for bandwidth

AIMD’s multiplicative decrease

- AIMD has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties
  - WITHOUT any coordination

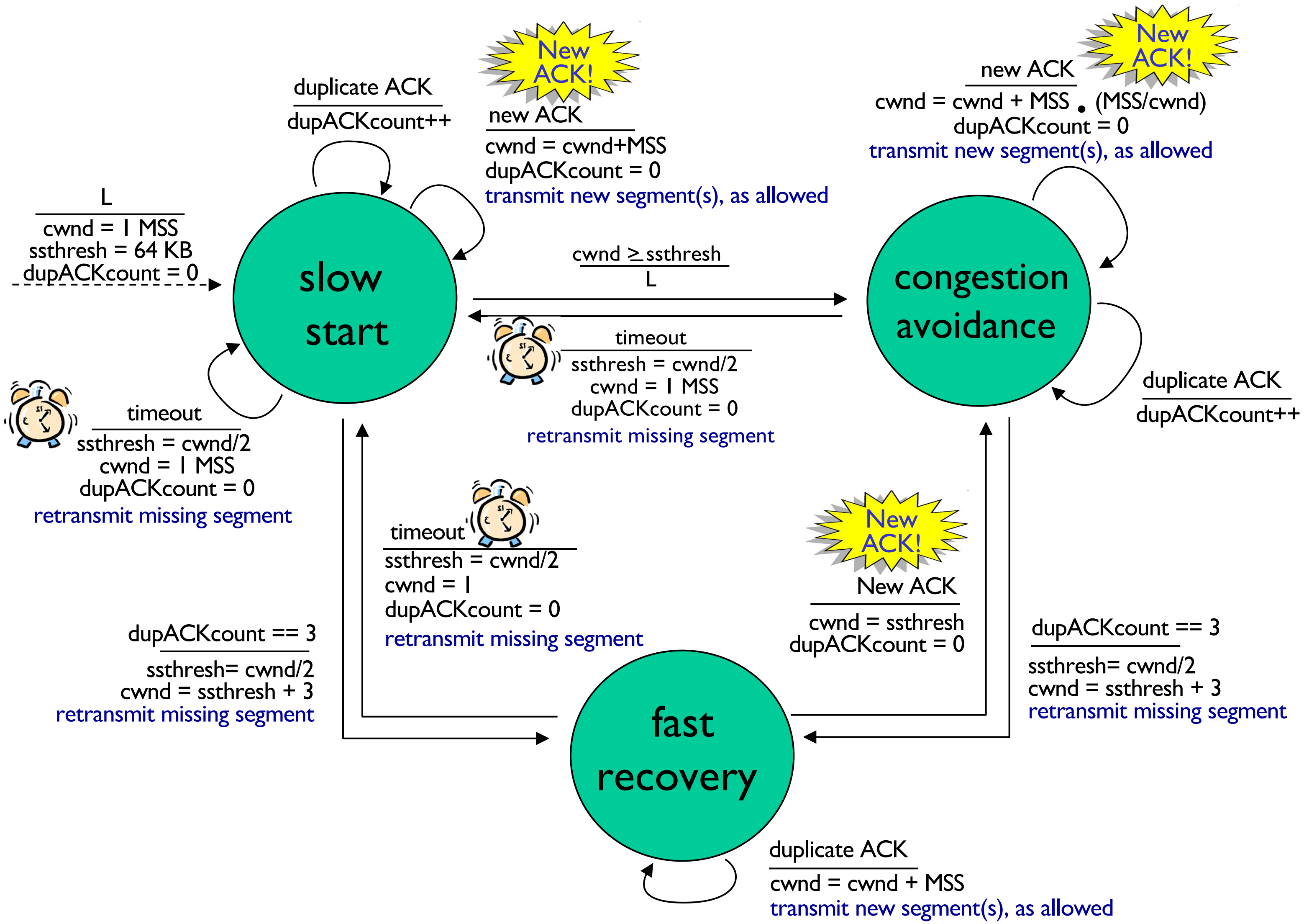
Different versions Reno vs Tahoe

- Reno: Cut to roughly half on loss detected by triple duplicate ACK
- Tahoe: Cut to 1 MSS when loss detected (either t-d-ACK or timeout)

Outline

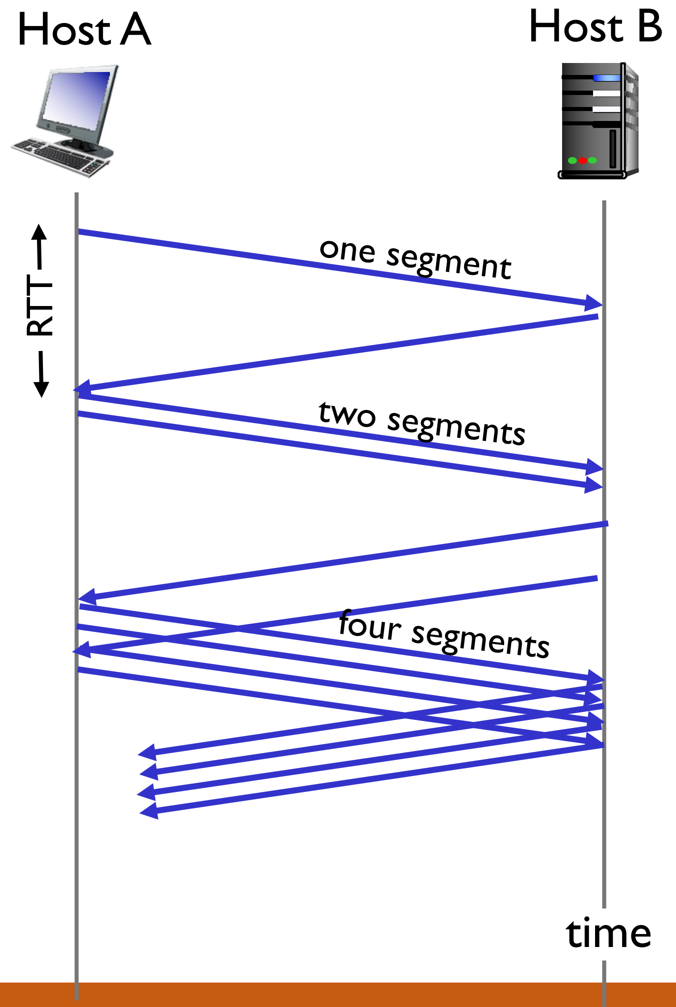
1. Approaches to Congestion Control
2. AIMD
3. 3 States in TCP Congestion Control

TCP CC has 3 states implementing AIMD



TCP slow start is not that slow

- when connection begins, increase rate exponentially until first loss event:
  - initially cwnd = 1 MSS
  - double cwnd every RTT
  - done by incrementing cwnd for every ACK received



Initial rate is slow but ramps up exponentially fast!

Two states that increases cwnd

- Slow Start does exponential increase (initial ramp up)
- Congestion Avoidance does linear increase
- When should we switch from exponential to linear increase?

When it reaches half of last max cwnd value just before the loss

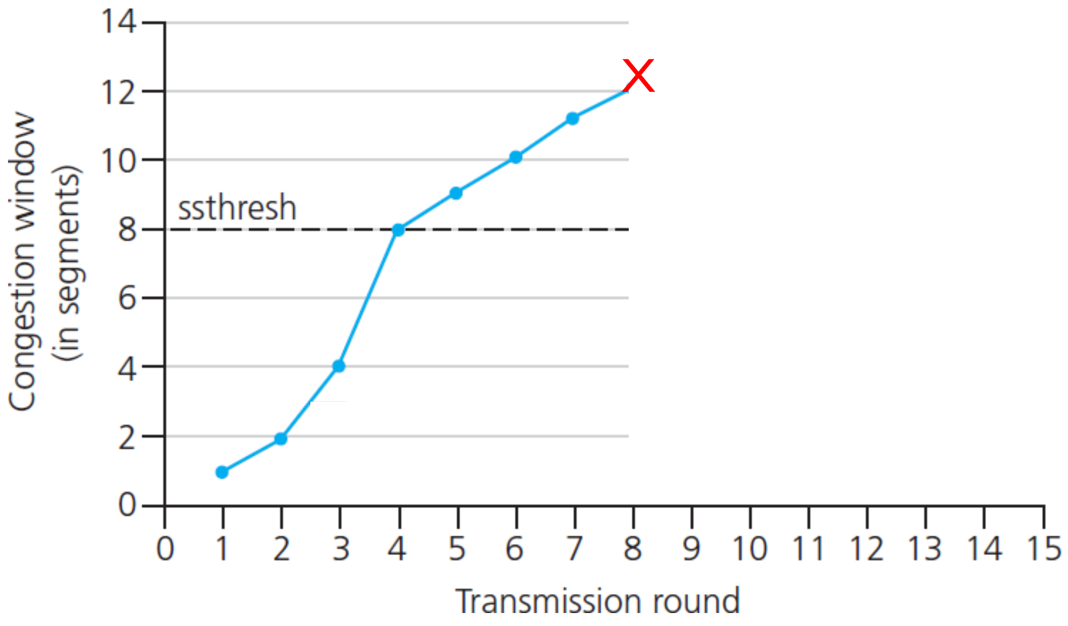
SSthresh stores half of last max cwnd value

Q: when should the exponential increase switch to linear?

A: when cwnd gets to 1/2 of its value before timeout.

Implementation:

- variable Ssthresh (slow start threshold)
- on loss event, ssthresh is set to 1/2 of cwnd just before loss event



If cwnd < ssthresh, we are in slow start

If cwnd ≥ ssthresh, we are in congestion avoidance

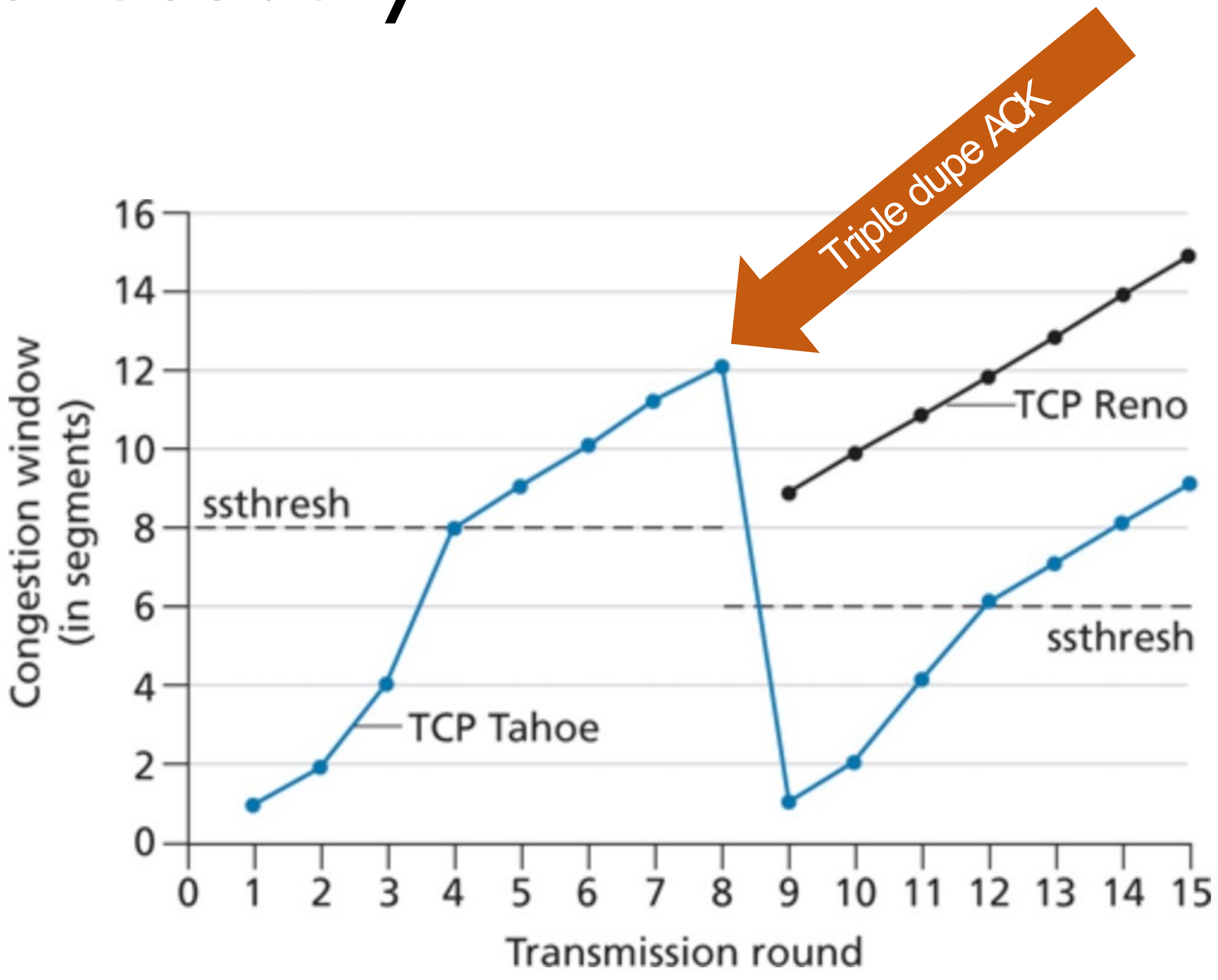
Tahoe vs Reno's fast recovery

Tahoe (no fast recovery)

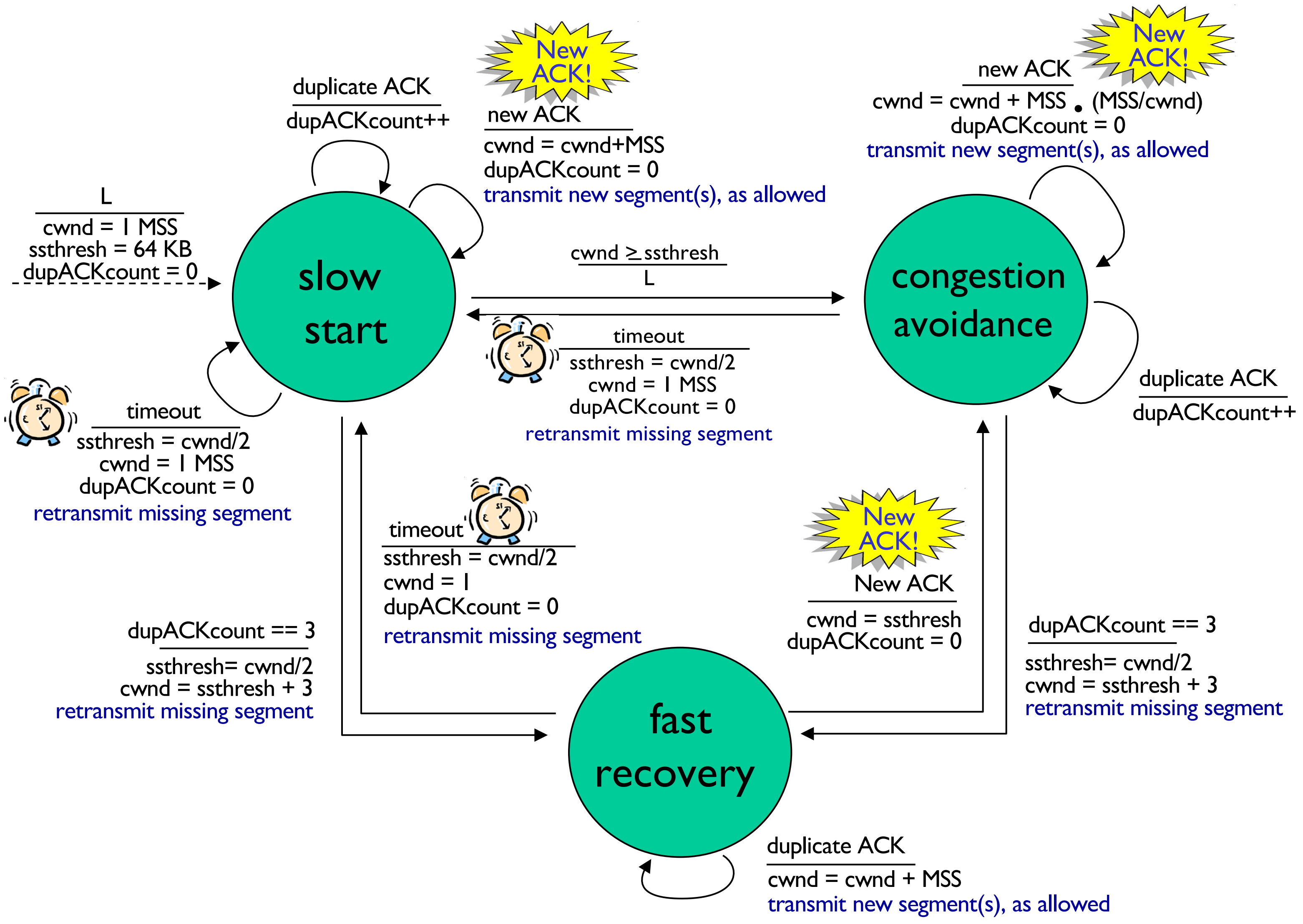
- ssthresh = cwnd/2
- cwnd = 1 MSS

Reno

- ssthresh = cwnd/2
- cwnd = ssthresh + 3MSS



Summary: TCP congestion control



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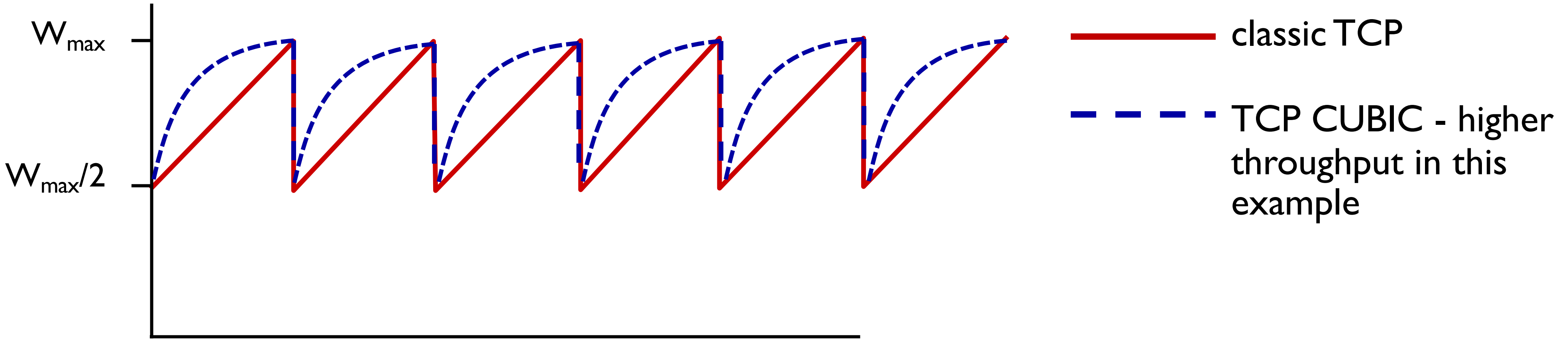
4. TCP CUBIC

Is there a better way
to "probe" available bandwidth?

TCP CUBIC: more aggressive initially but more cautious later with higher probability of loss

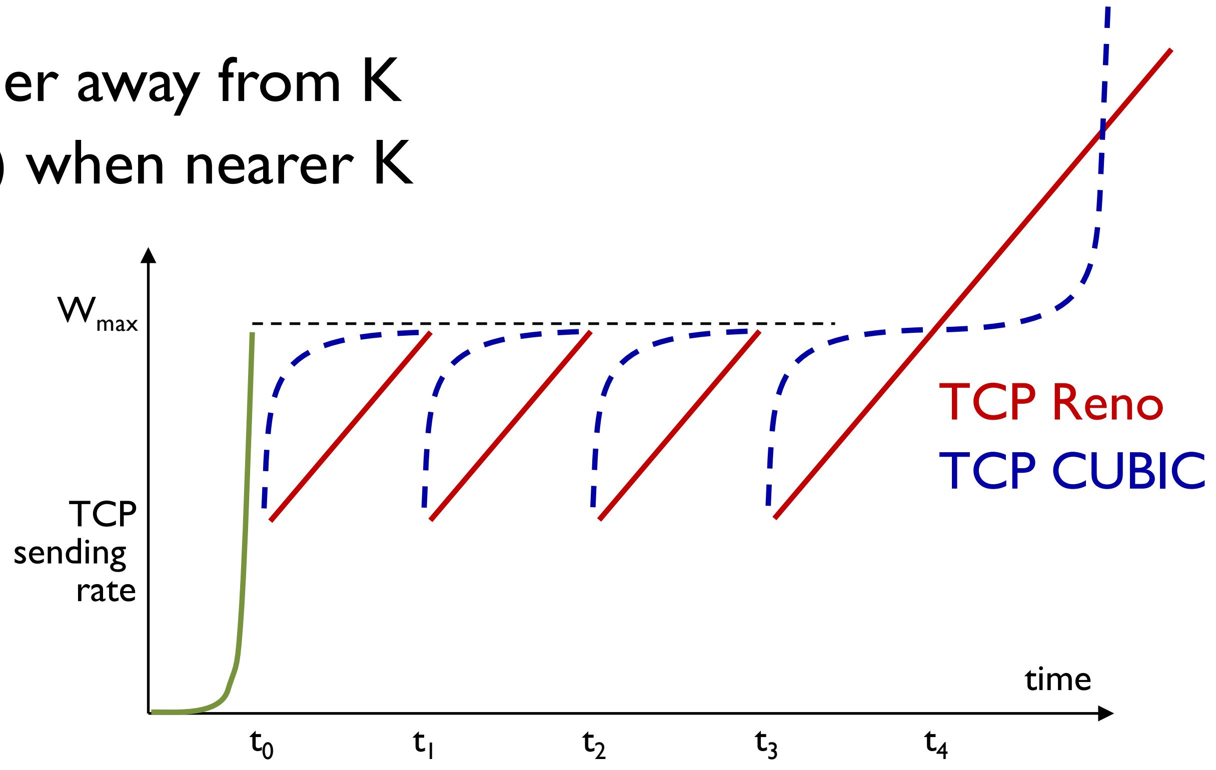
- Insight/intuition:
  - W_{max}: sending rate at which congestion loss was detected
  - congestion state of bottleneck link probably (?) hasn't changed much
  - after cutting rate/window in half on loss, initially ramp to to W_{max} faster, but then approach W_{max} more slowly



TCP CUBIC has higher throughput than Reno

- K: point in time when TCP window size will reach W_{max}
  - K itself is tunable
- increase W as a function of the cube of the distance between current time and K
  - larger increases when further away from K
  - smaller increases (cautious) when nearer K

CUBIC is default in Linux, widely used among popular Web servers



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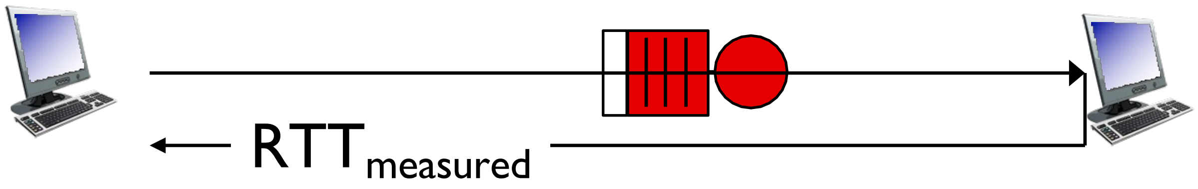
5. Delay-based CC

Delay-based TCP CC monitors throughput

Keeping the pipe "just full enough, but no fuller"



$$\text{Throughput}_{measured} = \frac{\text{\# bytes sent in last RTT interval}}{\text{RTT}_{measured}}$$

- RTT_{min} - minimum observed RTT
- uncongested throughput - $\text{cwnd}/\text{RTT}_{min}$

if $\text{Throughput}_{measured}$ "very close" to $\text{cwnd}/\text{RTT}_{min}$ //not congested
    increase cwnd linearly
else if $\text{Throughput}_{measured}$ "far below" $\text{cwnd}/\text{RTT}_{min}$ //congested
    decrease cwnd linearly

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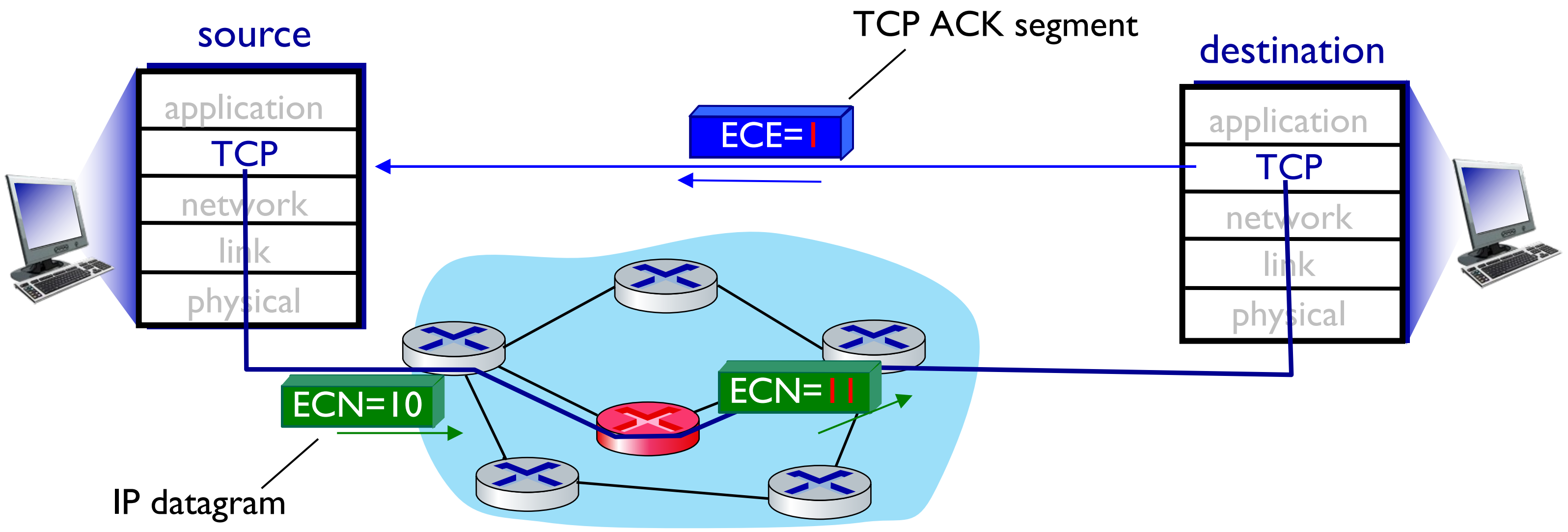
6. Network assisted CC

Network-assisted approach: Explicit congestion notification (ECN)

- two bits in IP header (ToS field) marked by network router to indicate congestion
  - policy to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit (ECN-Echo) on ACK segment to notify sender of congestion



ECN approach involves both IP and TCP

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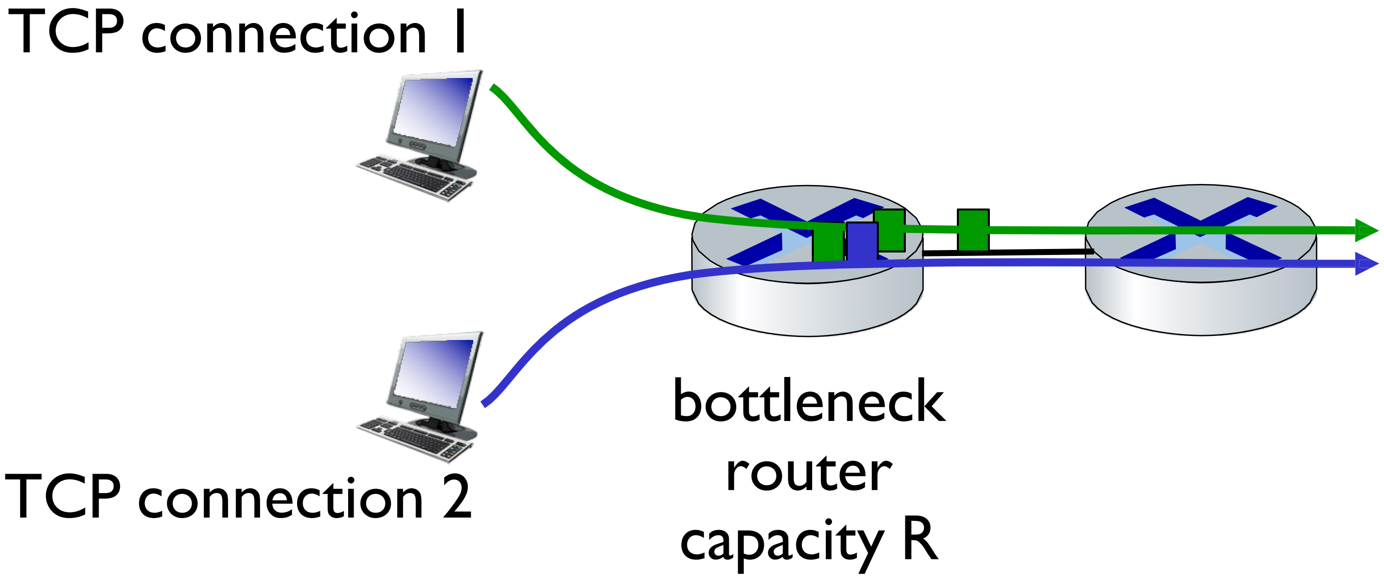
7. TCP fairness

TCP fairness

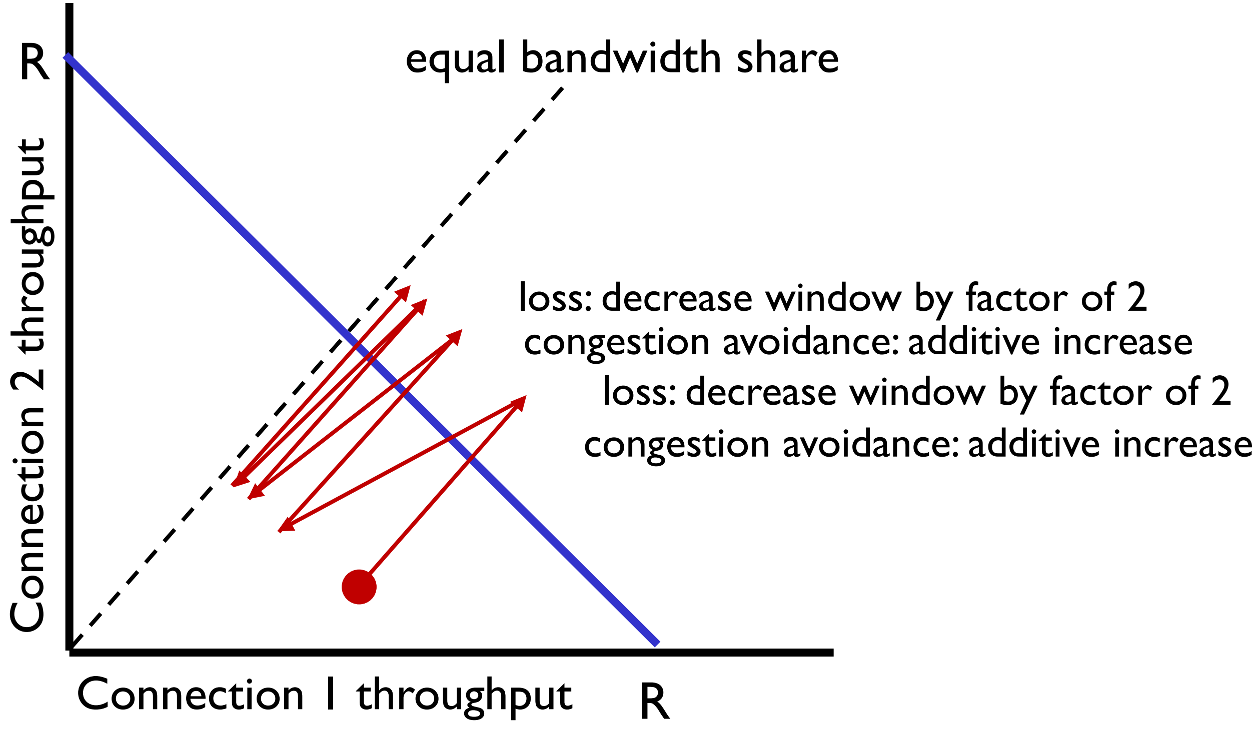
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Q: is TCP Fair?

Example: two competing TCP sessions:

- additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



Is TCP fair?

A: Yes, under idealized assumptions:

- same RTT
- fixed number of sessions only in congestion avoidance

Fairness: must all network apps be "fair"?

Fairness and UDP

- multimedia apps opts out TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss

Fairness, parallel TCP connections

- application can open multiple parallel connections btw 2 hosts (web browsers, etc.)
- Link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2

There is NO "Internet Police" policing use of bandwidth

Acknowledgements

Slides are adopted from Kurose' Computer Networking Slides