



CSC/CPE 138 - Computer Network Fundamentals

Transport Layer

The presentation was adapted from the textbook: *Computer Networking: A Top-Down Approach* 8th edition Jim Kurose, Keith Ross, Pearson, 2020

Redefine the Possible™

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- **TCP congestion control**
- Evolution of transport-layer functionality



TCP congestion control: AIMD



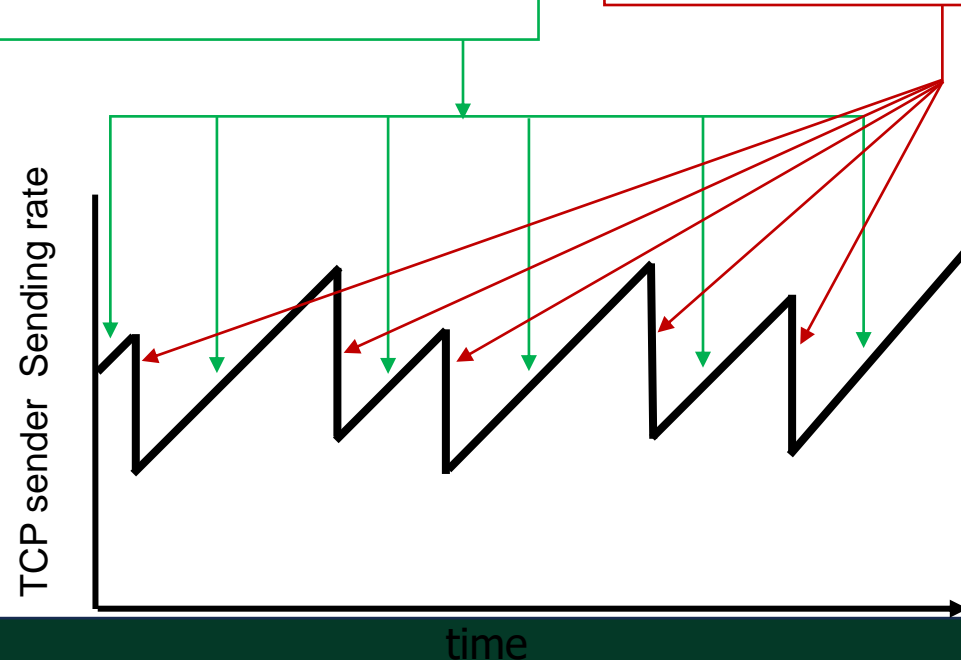
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

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

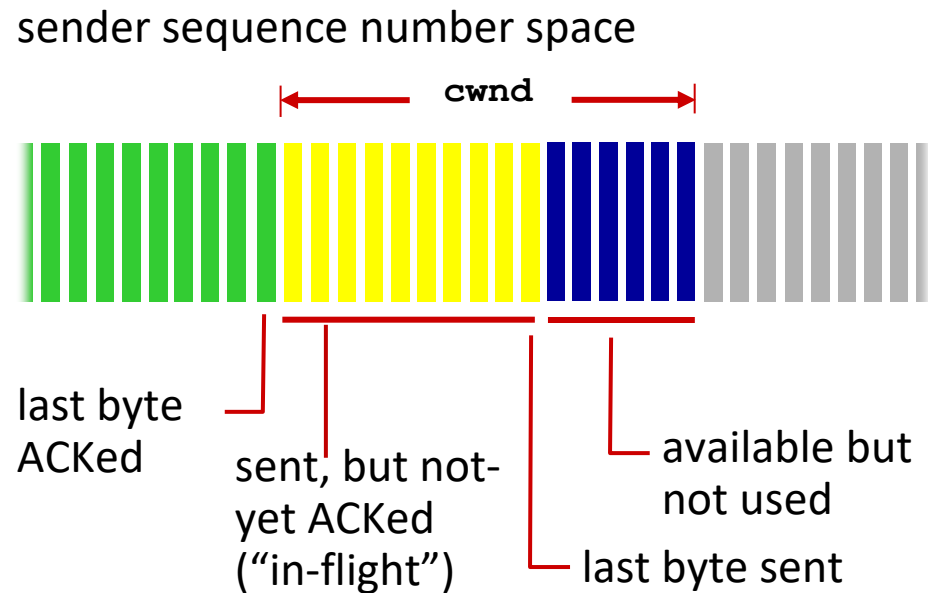
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
 - optimize congested flow rates network wide!
 - have desirable stability properties

TCP congestion control: details



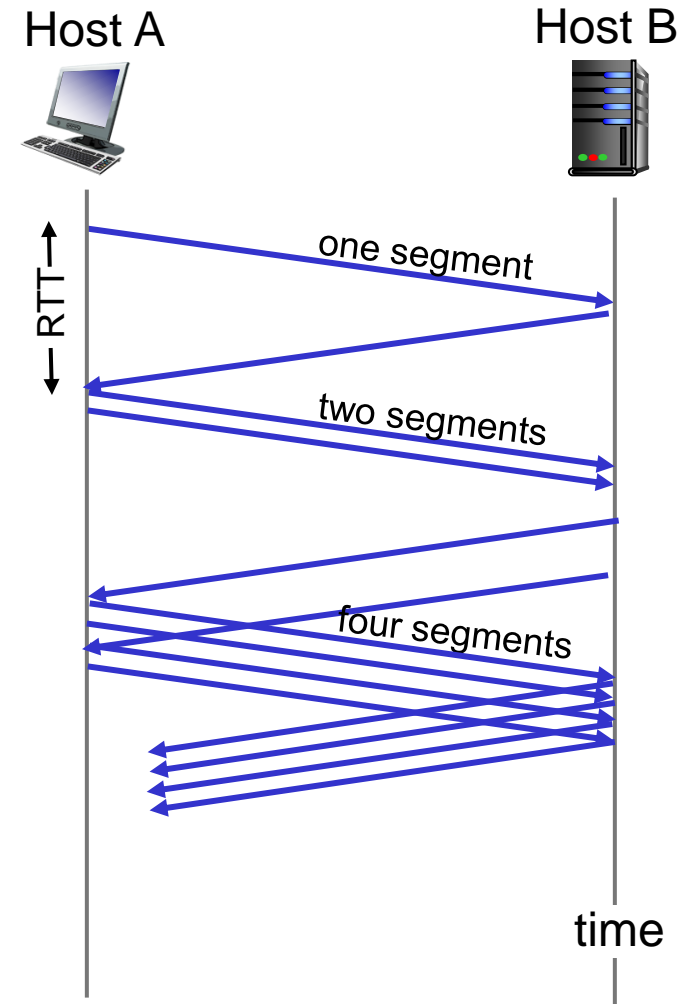
TCP sending behavior:

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

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

- TCP sender limits transmission: $\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

- 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
- *summary*: initial rate is slow, but ramps up exponentially fast



TCP: from slow start to congestion avoidance

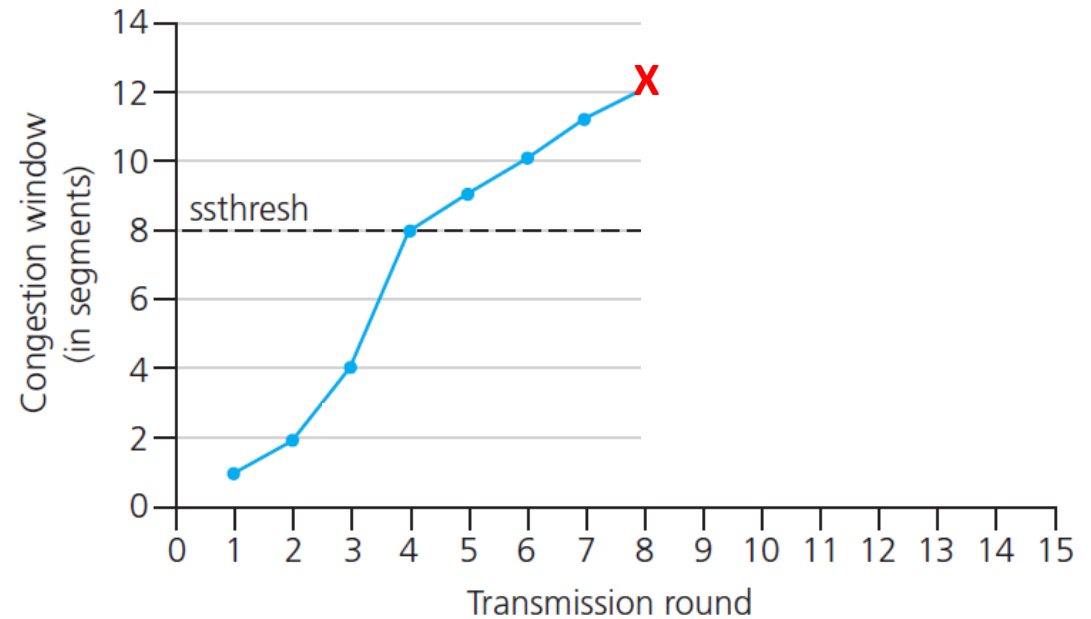


Q: when should the exponential increase switch to linear?

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

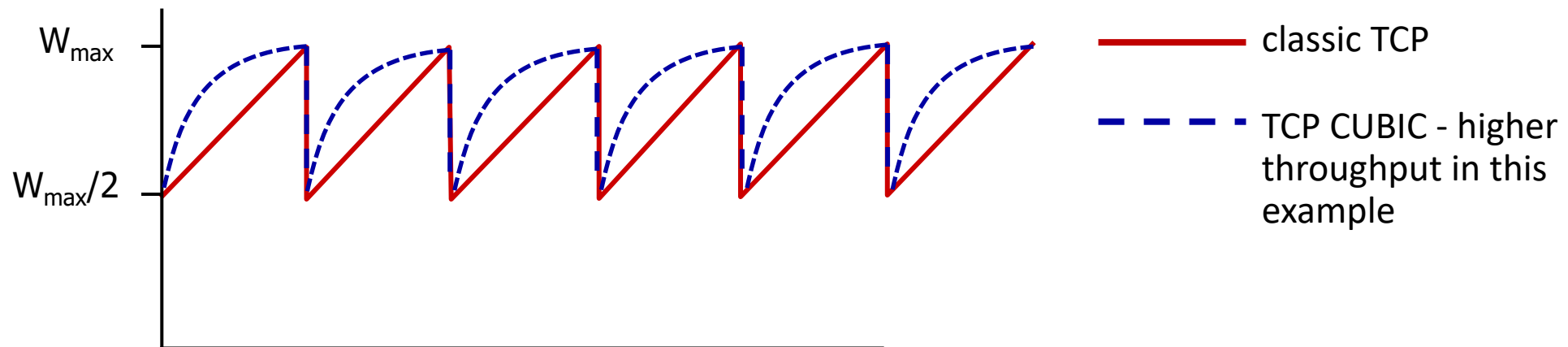
Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event

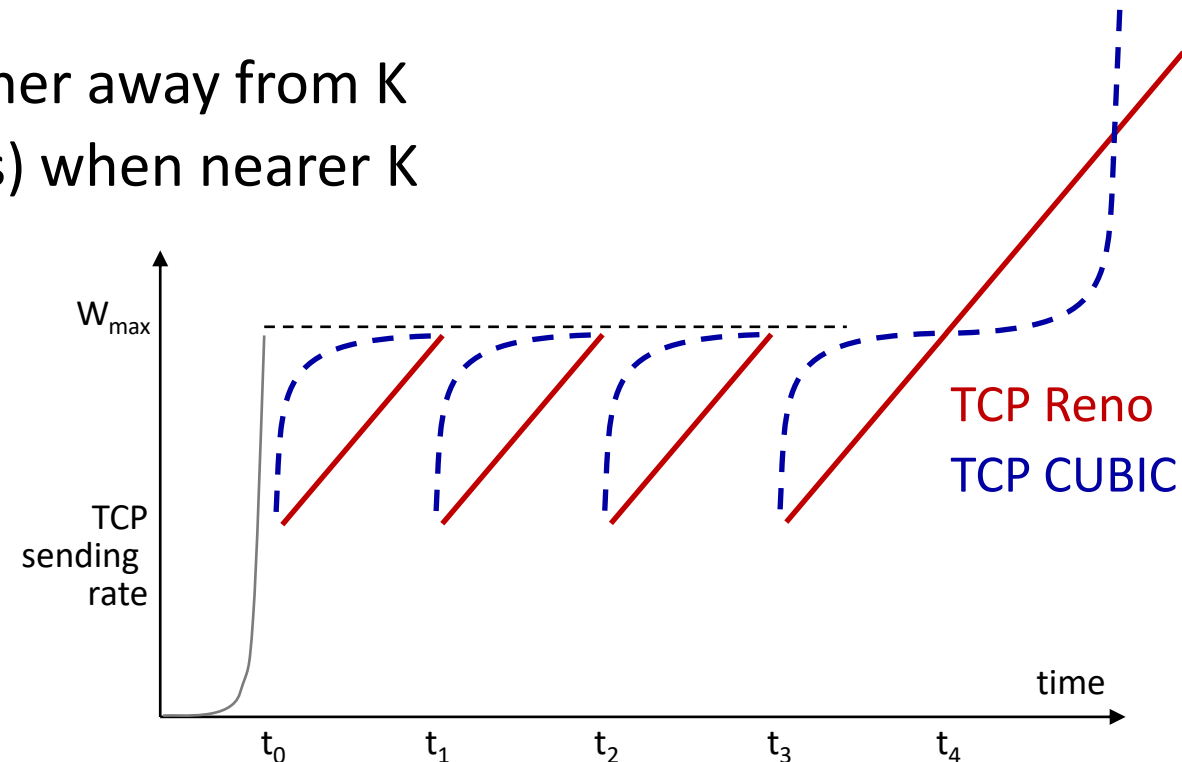


* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

- Is there a better way than AIMD to “probe” for usable bandwidth?
- 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*



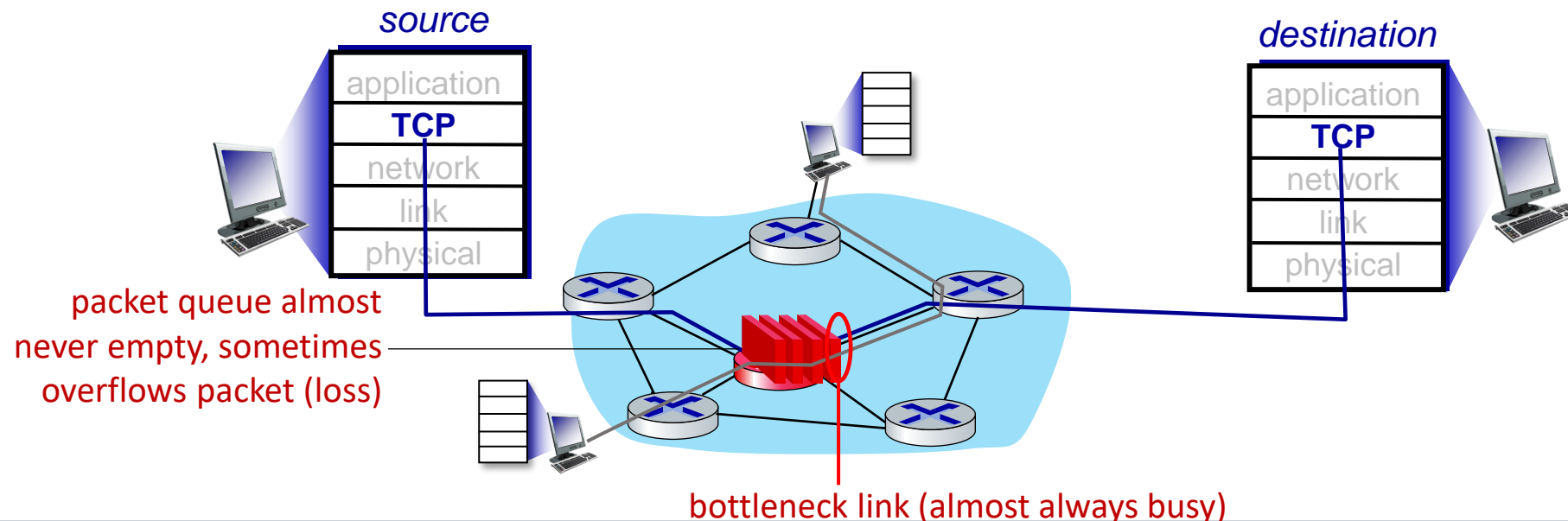
- K: point in time when TCP window size will reach W_{\max}
 - K itself is tuneable
- increase W as a function of the *cube* of the distance between current time and K
 - larger increases when further away from K
 - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular TCP for popular Web servers



TCP and the congested “bottleneck link”



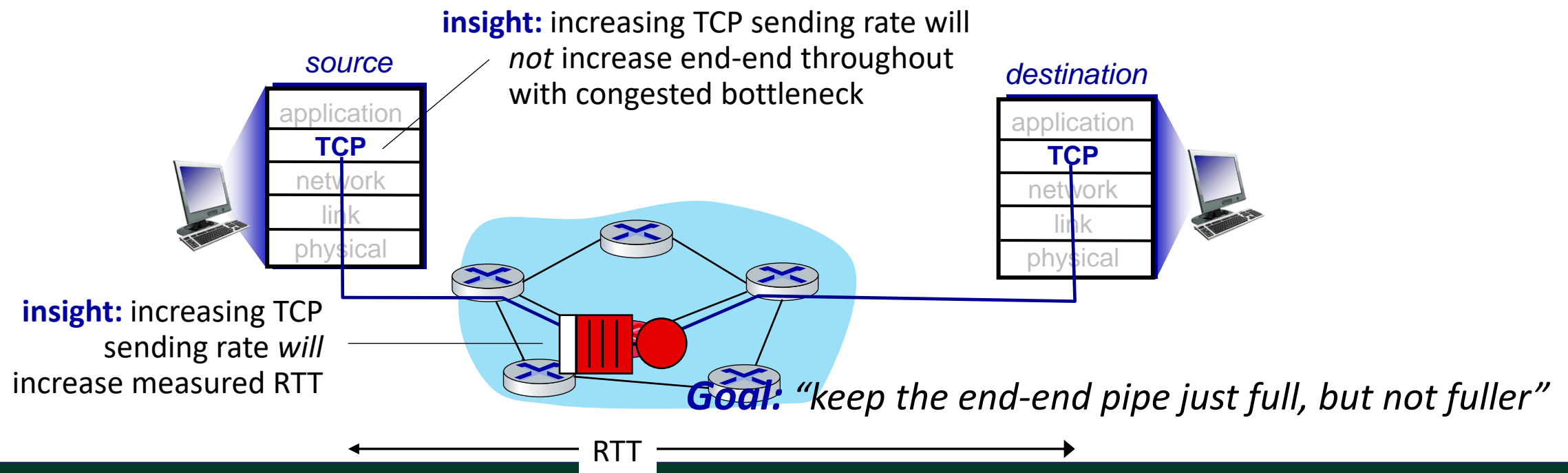
- TCP (classic, CUBIC) increase TCP's sending rate until packet loss occurs at some router's output: the *bottleneck link*



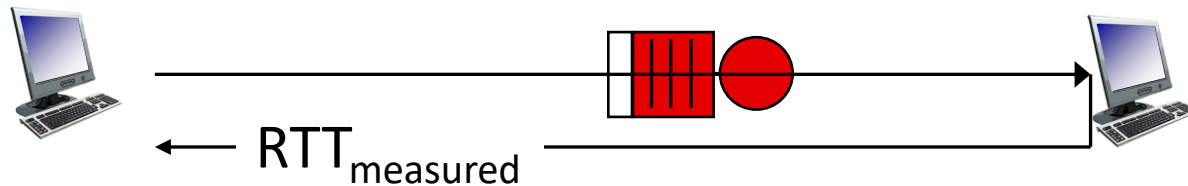
TCP and the congested “bottleneck link”



- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link



Keeping sender-to-receiver pipe “just full enough, but no fuller”: keep bottleneck link busy transmitting, but avoid high delays/buffering



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

Delay-based approach:

- RTT_{\min} - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window cwnd is $\text{cwnd}/\text{RTT}_{\min}$

if measured throughput “very close” to uncongested throughput
 increase cwnd linearly /* since path not congested */
else if measured throughput “far below” uncongested throughput
 decrease cwnd linearly /* since path is congested */

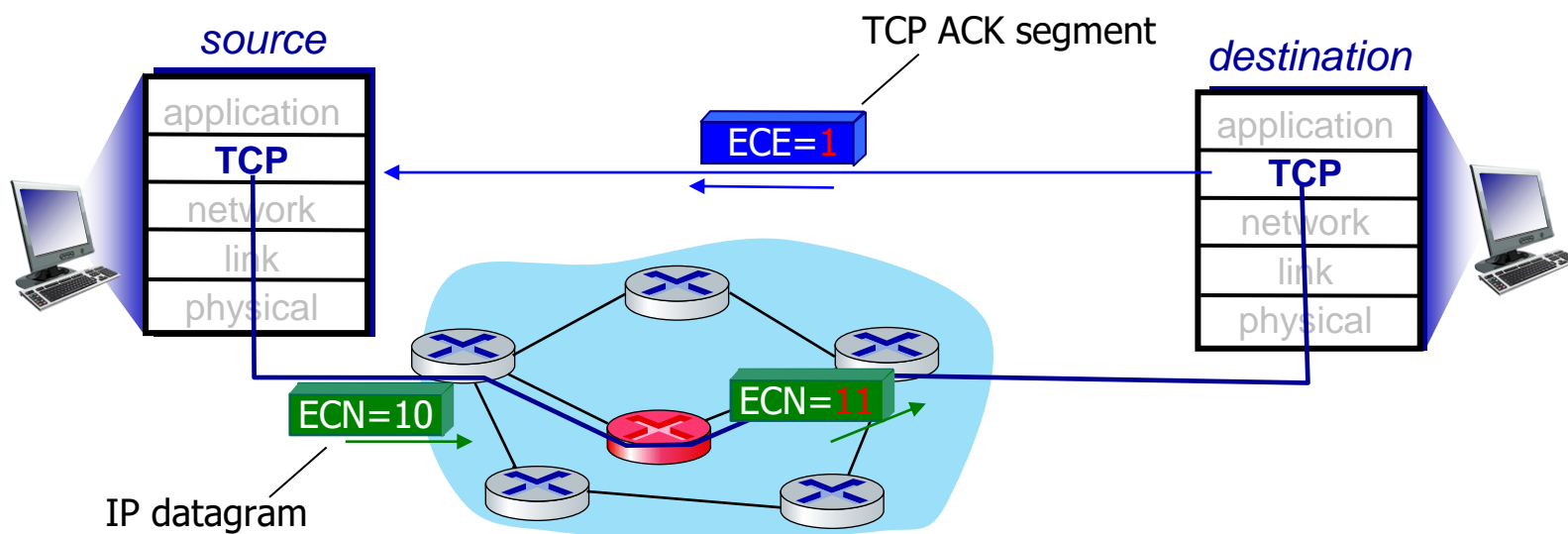
- congestion control without inducing/forcing loss
- maximizing throughput (“keeping the just pipe full...”) while keeping delay low (“...but not fuller”)
- a number of deployed TCPs take a delay-based approach
 - BBR deployed on Google’s (internal) backbone network

Explicit congestion notification (ECN)

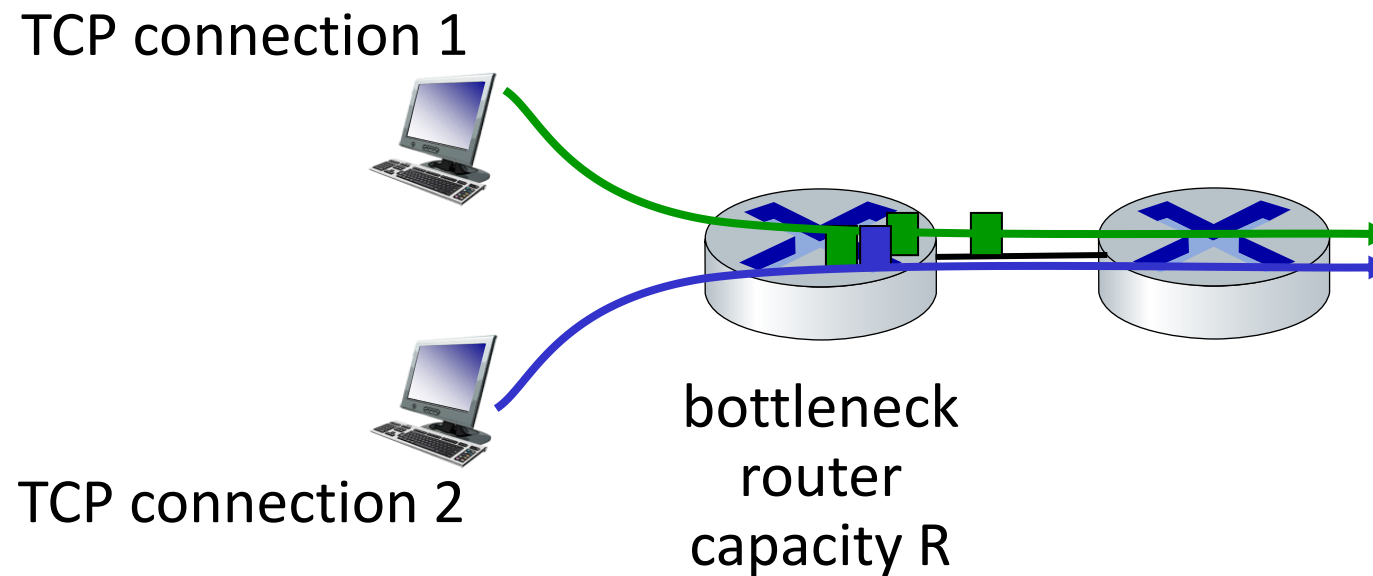


TCP deployments often implement *network-assisted* congestion control:

- 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 on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)

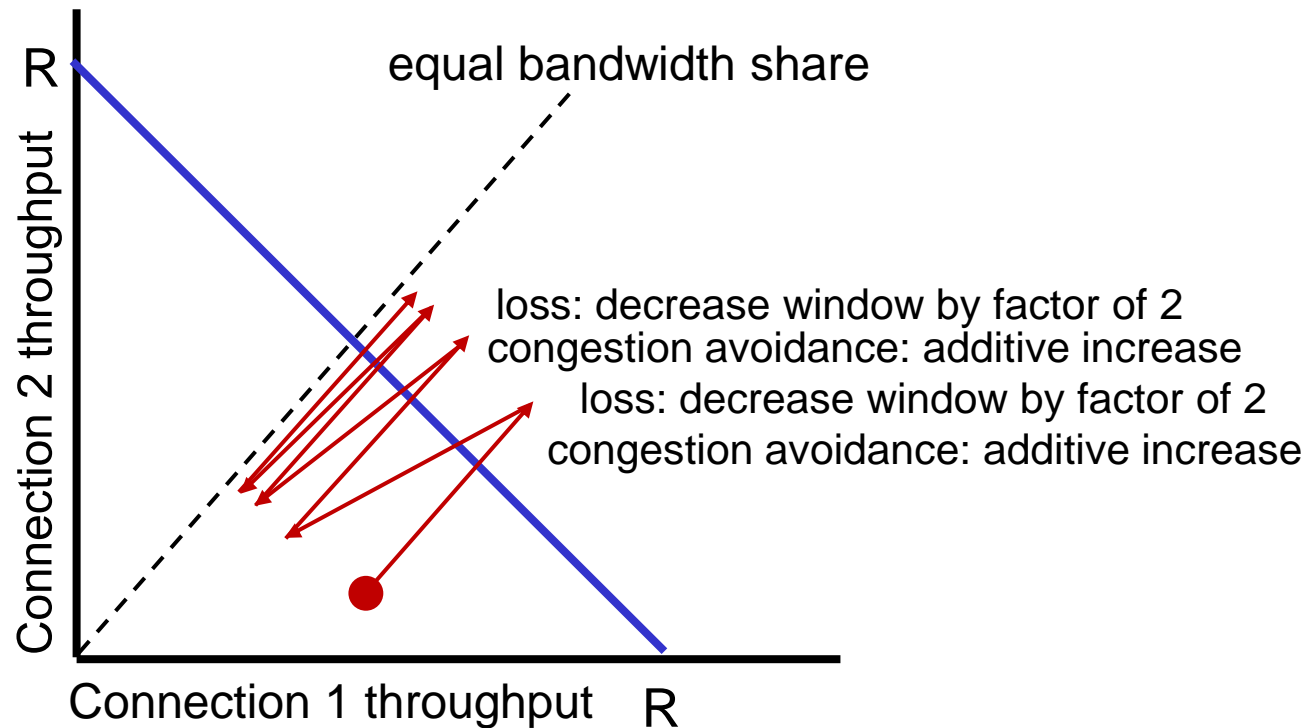


Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



Example: two competing TCP sessions:

- additive increase gives slope of 1, as throughput 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 and UDP

- multimedia apps often do not use TCP
 - do not want rate throttled by congestion control
- instead use UDP:
 - send audio/video at constant rate, tolerate packet loss
- there is no “Internet police” policing use of congestion control

Fairness, parallel TCP connections

- application can open *multiple* parallel connections between two hosts
- web browsers do this , e.g., 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 $R/2$