Chapter 3
Transport Layer

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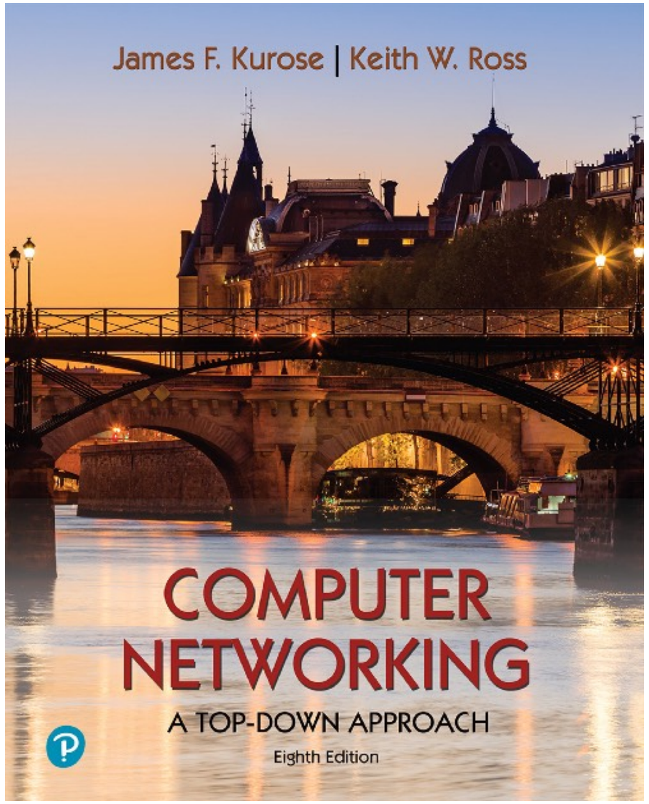


Computer Networking: A Top-Down Approach
8th edition
Jim Kurose, Keith Ross
Pearson, 2020

Day 14: Evolution of TCP and Transport

CSEE 4119
Computer Networks
Ethan Katz-Bassett

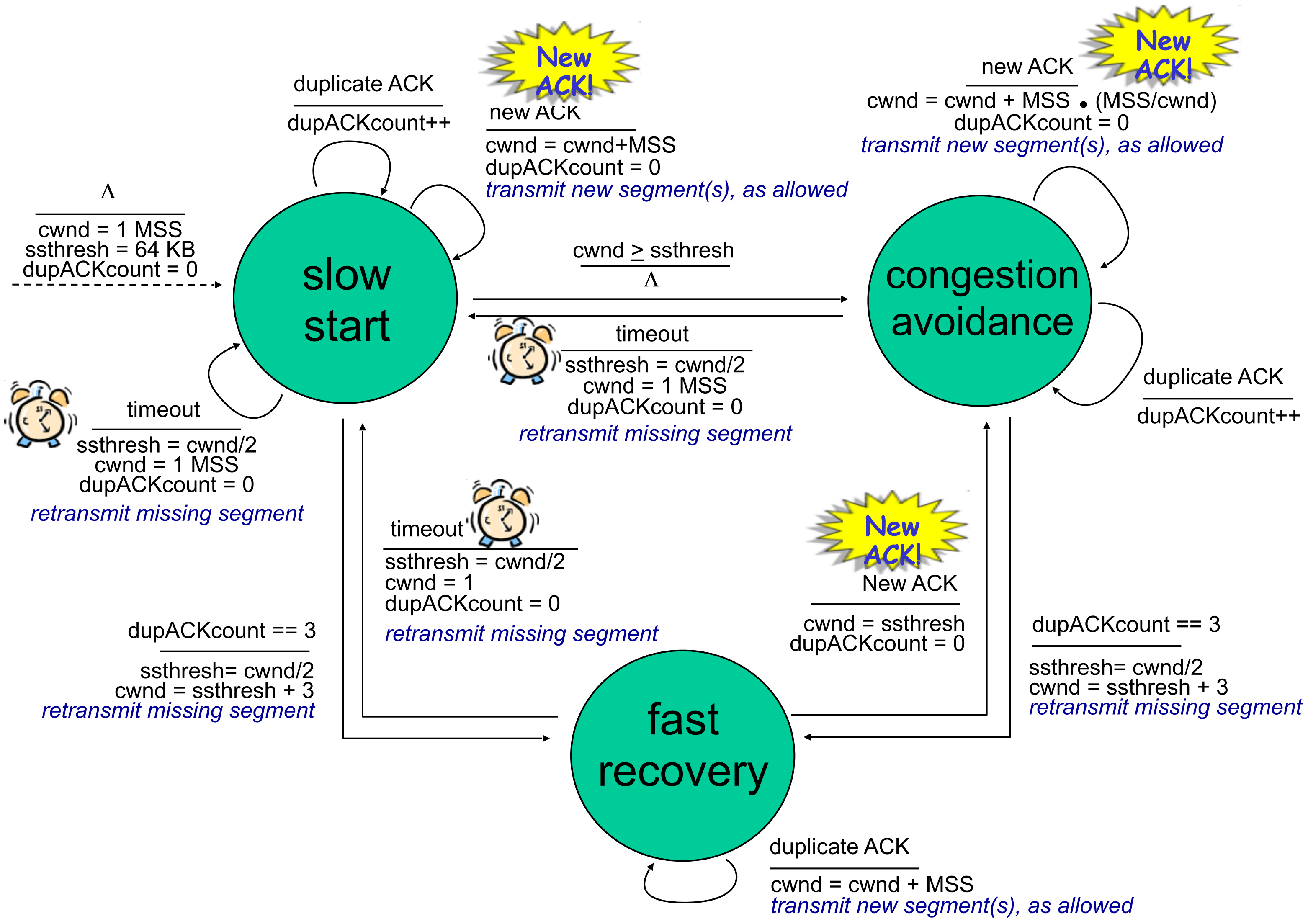
COLUMBIA UNIVERSITY
IN THE CITY OF NEW YORK

Oct 18 admin

- Masks ***strongly recommended***, over nose and mouth
- Attendance & participation ***not*** required
- If you are not feeling well or were exposed to COVID, please stay home
- Videos of lectures available

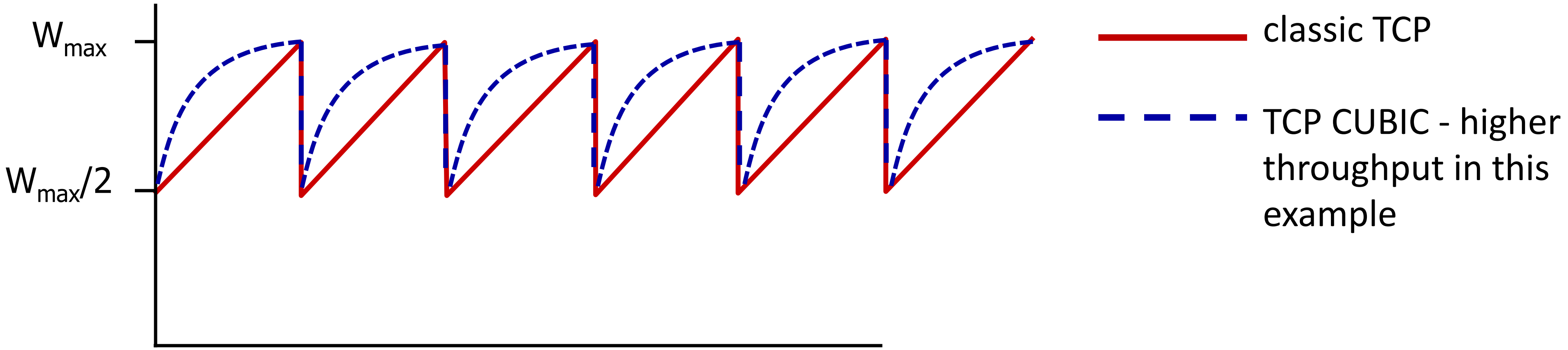
- HW2 due yesterday
- 1 extra slip day granted (5 total)
- HW3 assigned, due October 31
    - ***If submitted LATE, SCORE=0, NO exceptions***
- Project 1 Final Stage released this week
- Next week: guest lectures, Henning on Tuesday, Tom on Thursday
    - I'll be at the Internet Measurement Conference

Recap: TCP (Reno) congestion control



TCP CUBIC

- 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*

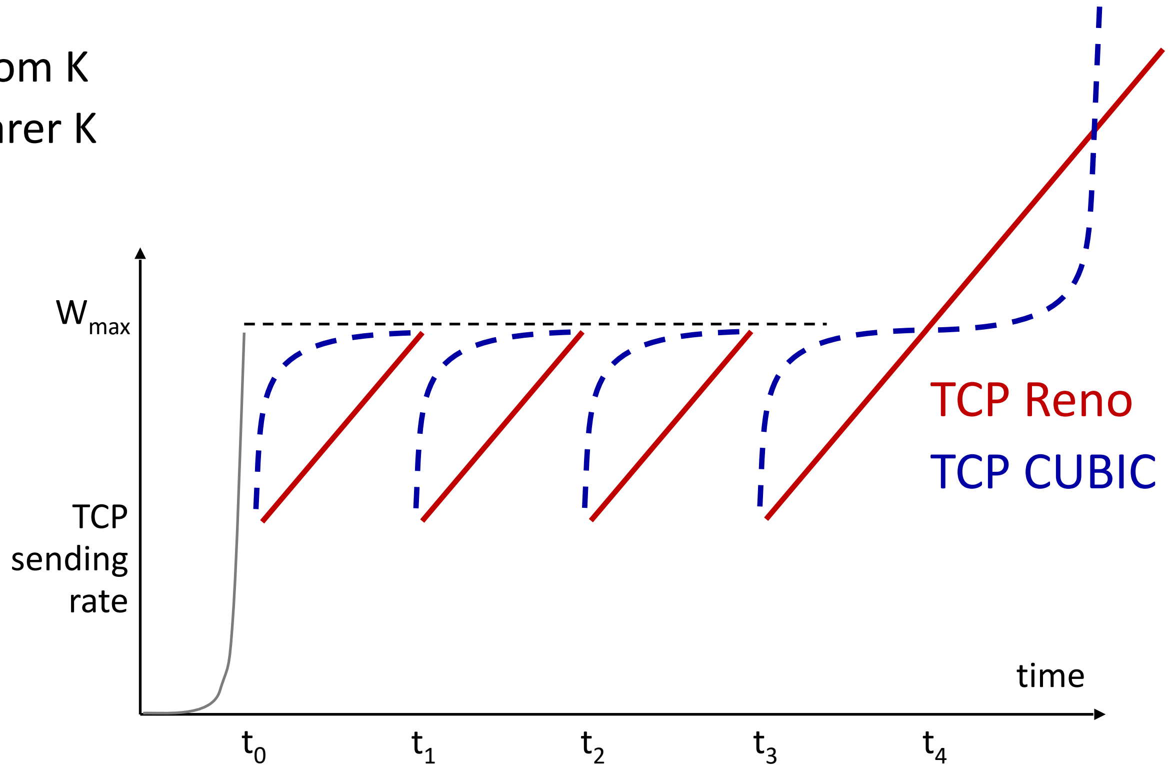


TCP CUBIC

- K: point in time when TCP window size will reach W_{max}
  - K is configurabletunable

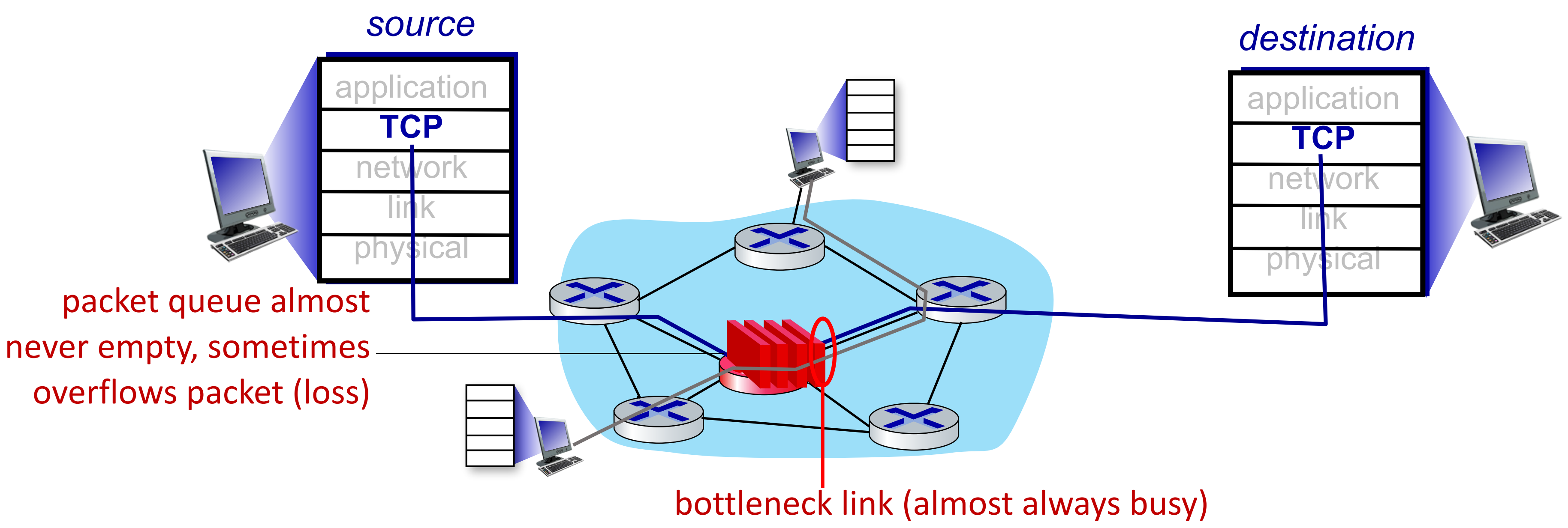
- increase W as a function of the *cube* of the distance between current time and K
  - larger increases when further away from K
  - smaller increases (cautious) when nearer K

- TCP CUBIC default in Linux
  - As of 2020, most popular TCP for popular Web servers
  - Today:
    - Netflix uses Reno
    - Google, YouTube, Akamai, Amazon, DropBox use BBR (next! And on HW!)



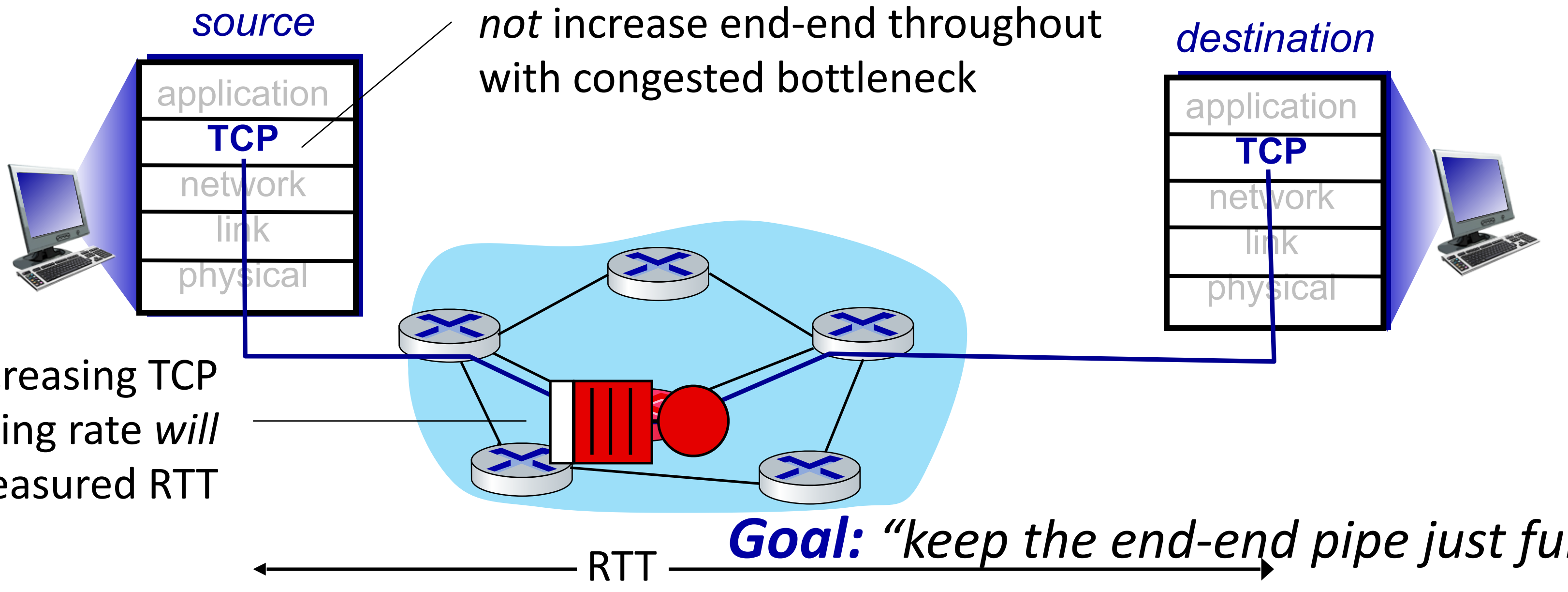
TCP and the congested "bottleneck link"

- 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



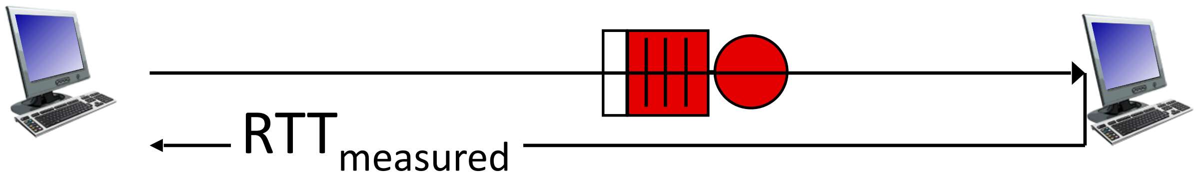
insight: increasing TCP sending rate will *not* increase end-end throughout with congested bottleneck

insight: increasing TCP sending rate *will* increase measured RTT

Goal: *"keep the end-end pipe just full, but not fuller"*

Delay-based TCP congestion control

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



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

Delay-based approach:

- RTT_{min} - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window `cwnd` is cwnd/RTT_{min}

if measured throughput "very close" to uncongested throughput

increase `cwnd` linearly /* since path not congested */

else if measured throughput "far below" uncongested throughout

decrease `cwnd` linearly /* since path is congested */

Delay-based TCP congestion control

- congestion control without inducing/forcing loss
- maximizing throughout (“keeping the just pipe full… ”) while keeping delay low (“…but not fuller”)
- What’s the problem if you just use delay…and others use Reno (loss)?
- a number of deployed TCPs take a delay-based approach
  - BBR developed and deployed by Google uses a modified delay-based approach

BBR: a CC written from scratch at Google

The story of BBR goes back to 2013...

- Many Google services complained about TCP performance...
  - Internal B4 backbone TCP throughput often < 10Mbps
  - Youtube.com: sometimes terrible video quality, with RTT > 10 secs
  - Google.com: poor latency in developing regions
- Services started to “work around” TCP
  - Use parallel connections, tweak TCP knobs, add more buffer to the network, ...

What was the root cause of these problems?

- TCP congestion control at Google in 2013 was CUBIC (Linux default)
  - CUBIC is a loss-based congestion control algorithm
  - Packet loss was the sole signal

The problem: loss-based congestion control

- Loss-based congestion control Reno (NetFlix/FreeBSD), CUBIC (Linux/Apple/Windows)
  - Keeps sending faster until it sees a loss

- But packet loss **alone** is **not** a good proxy for congestion

- If loss comes **before** sustained congestion, loss-based CC gets low throughput
  - 10Gbps over 100ms RTT needs < 0.0000029% (2.9e-8) packet loss (infeasible)
  - 1% loss (feasible) over 100ms RTT gets < 3Mbps

- If loss comes **after** congestion, loss-based CC bloats buffers, suffers high delays

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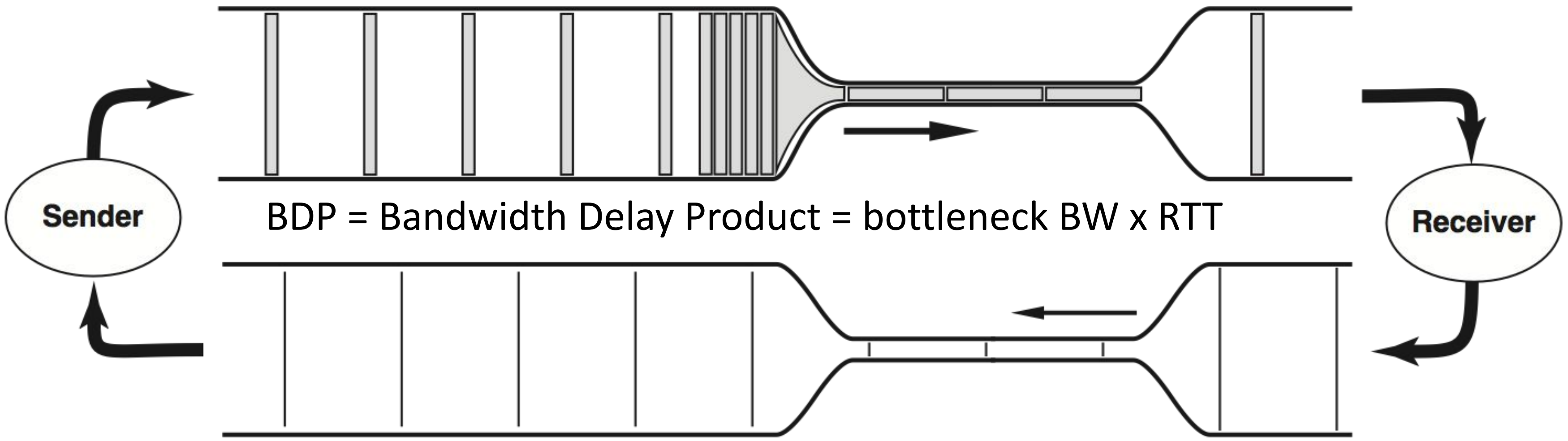
Network congestion and bottlenecks: a model



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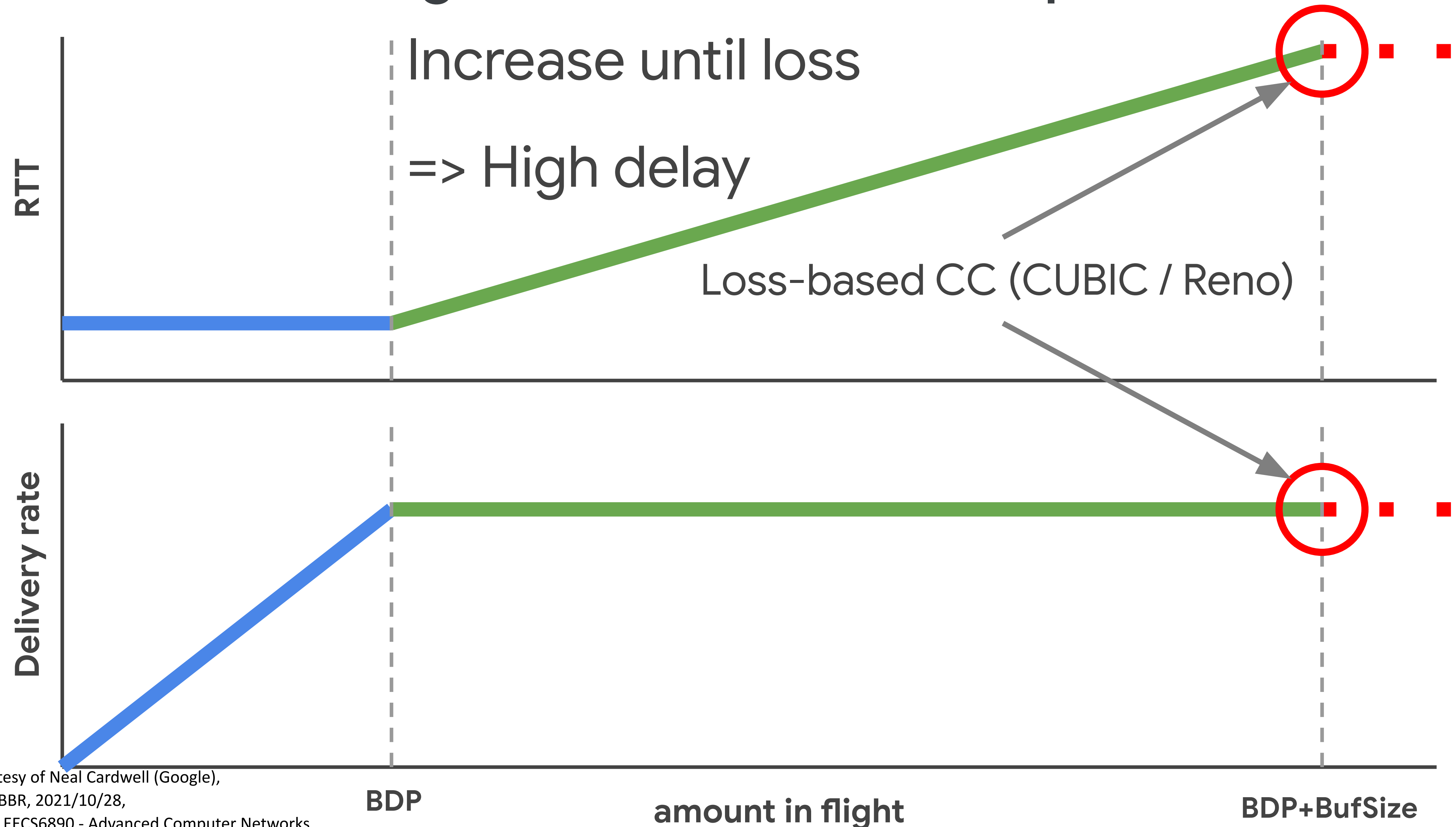
Loss-based congestion control in deep buffers



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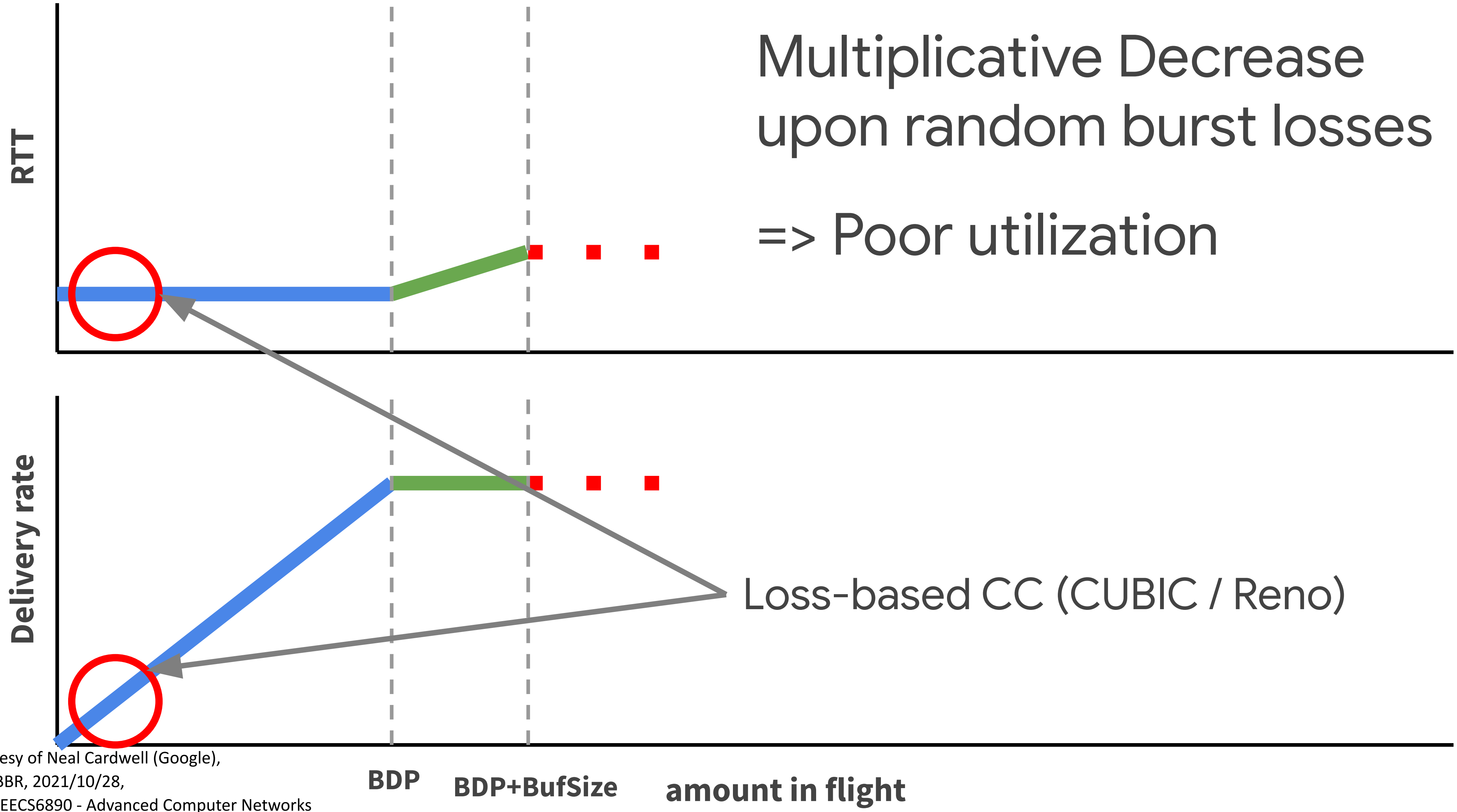
Loss-based congestion control in shallow buffers



Multiplicative Decrease upon random burst losses

=> Poor utilization

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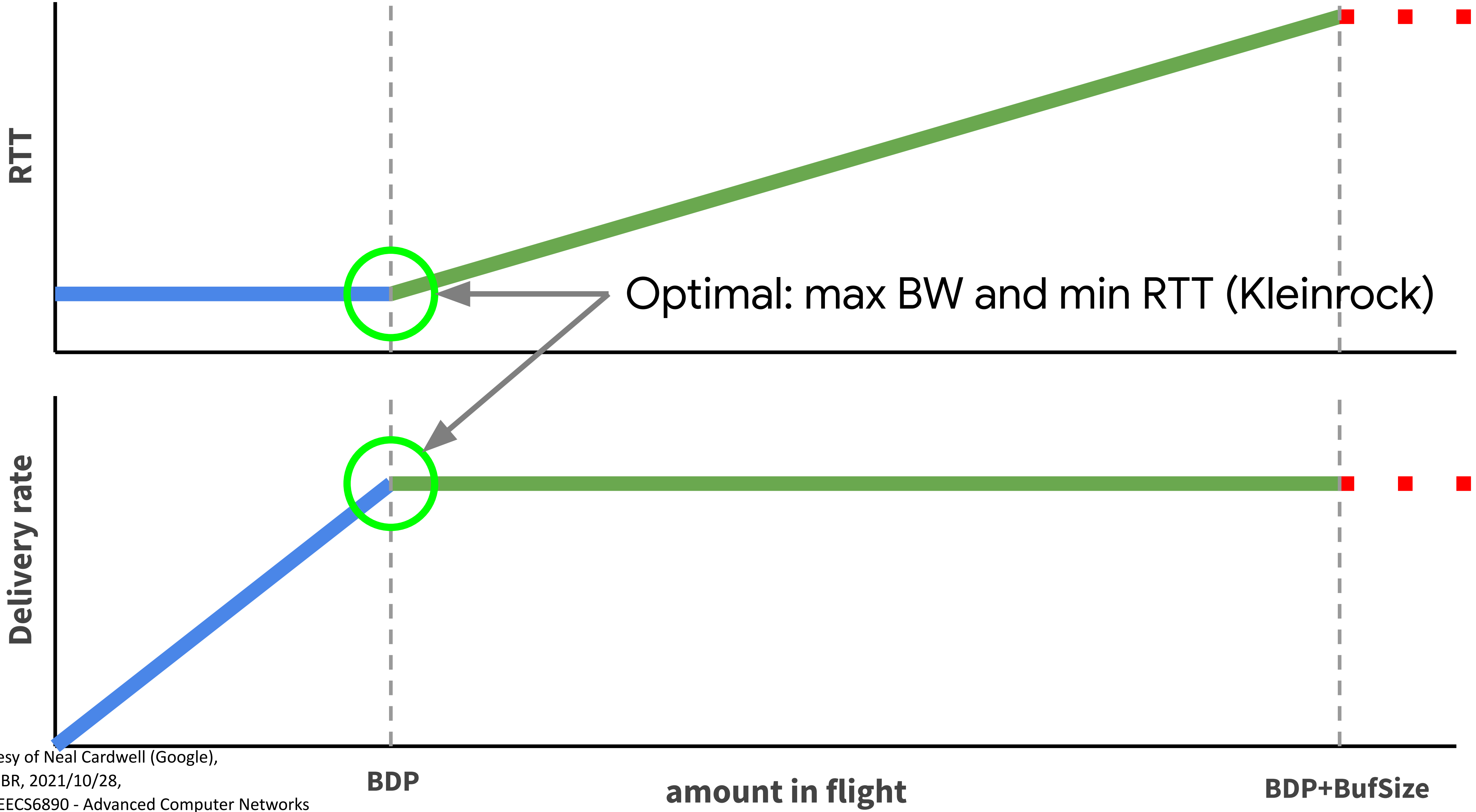
What is the optimal operating point?



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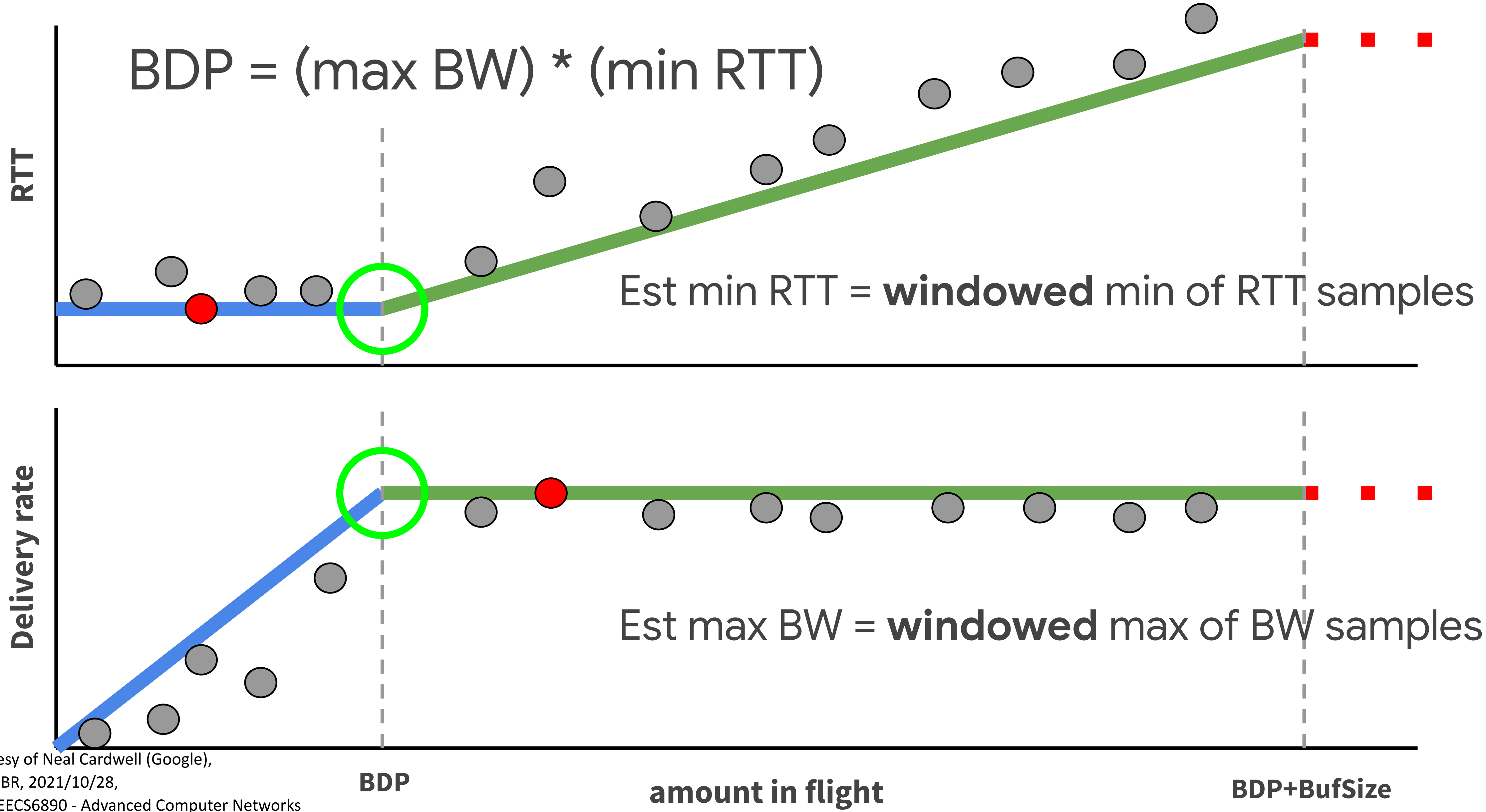
Estimating optimal point (max BW, min RTT)



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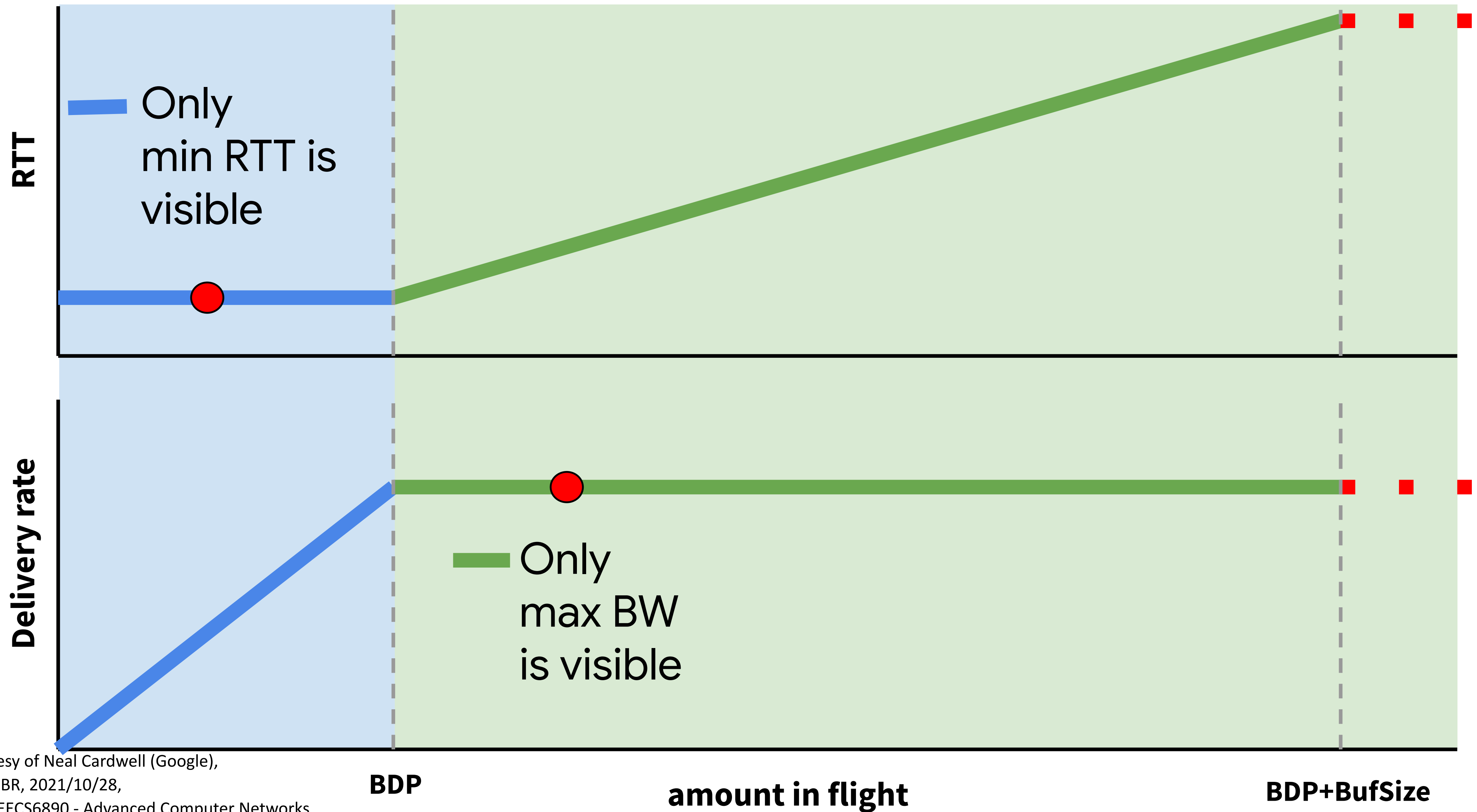
To see max BW, min RTT: probe both sides of BDP



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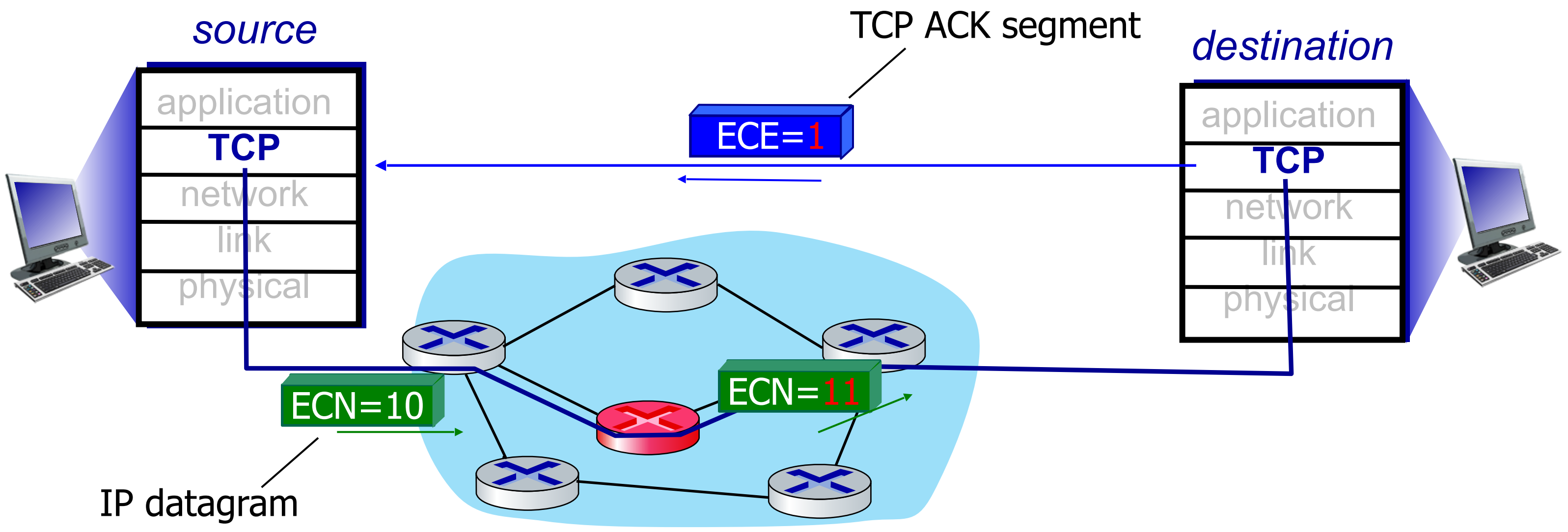
Explicit congestion notification (ECN)

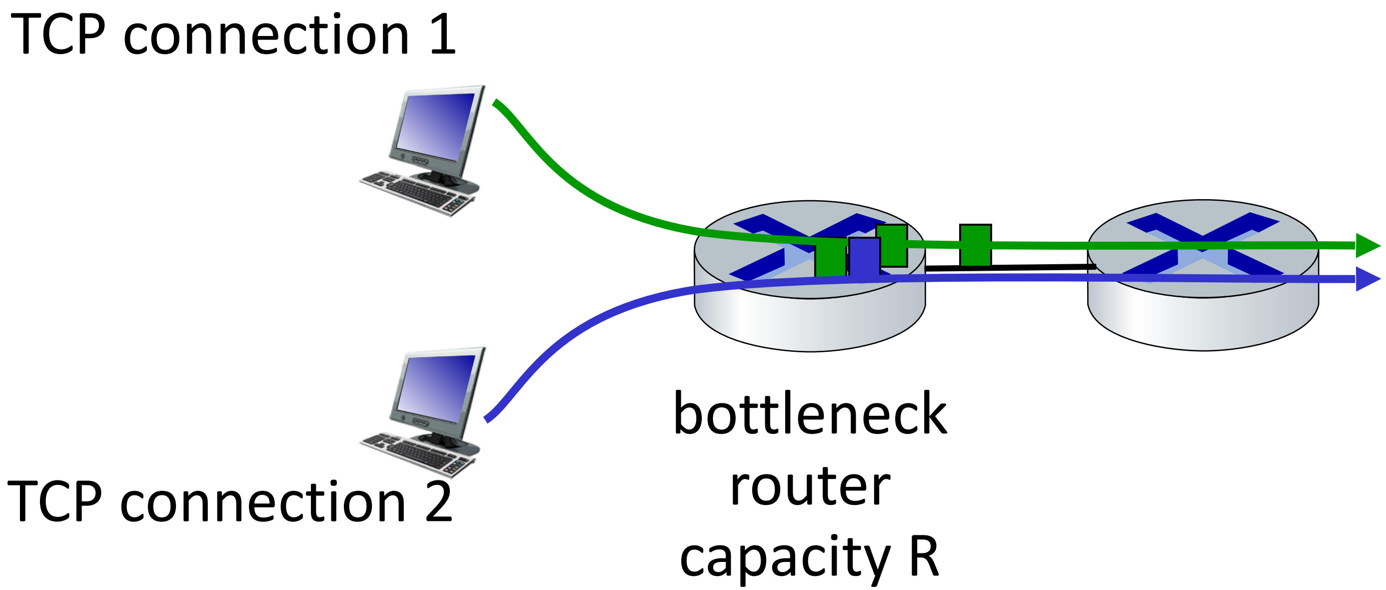
TCP deployments often implement *network-assisted* congestion control:

- 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)



TCP fairness

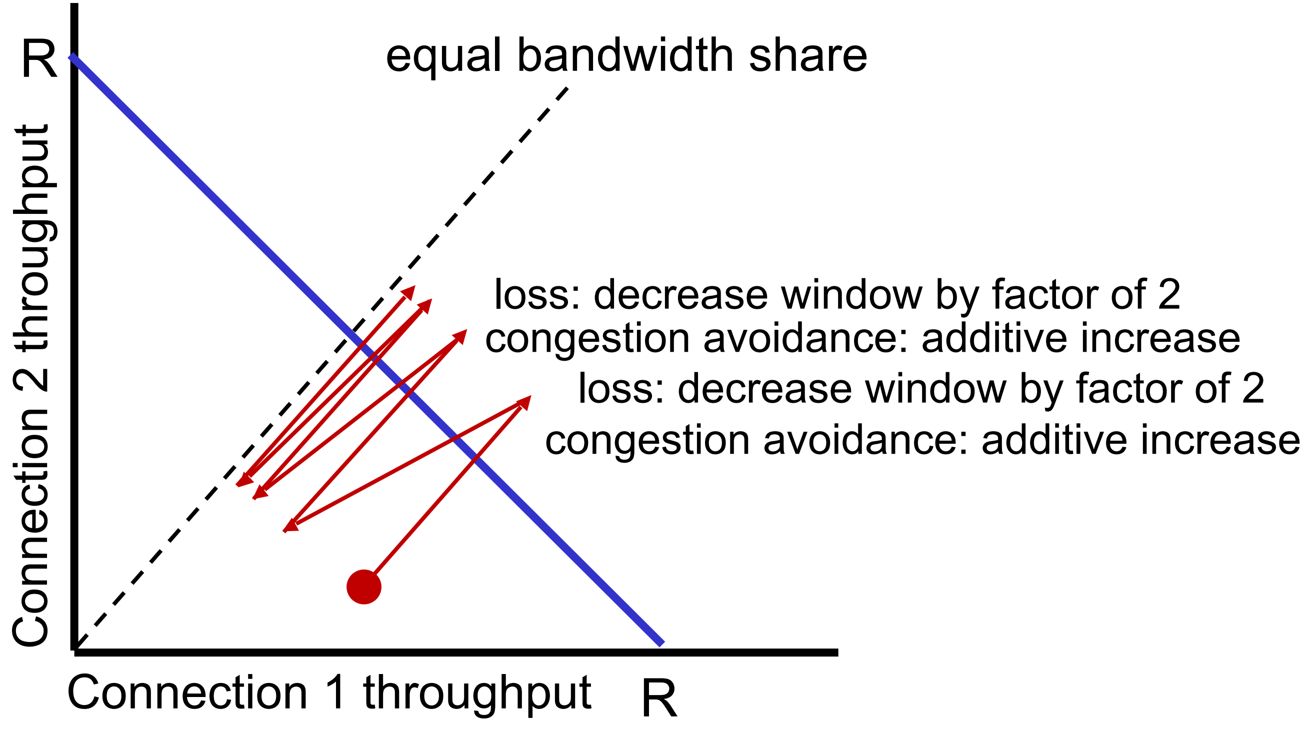
Fairness goal: if *K* TCP sessions share same bottleneck link of bandwidth *R*, each should have average rate of *R/K*



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 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

Transport layer: roadmap

- 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 over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of $L = 2\cdot 10^{-10}$ *– a very small loss rate!*

- versions of TCP for long, high-speed scenarios

Evolving transport-layer functionality

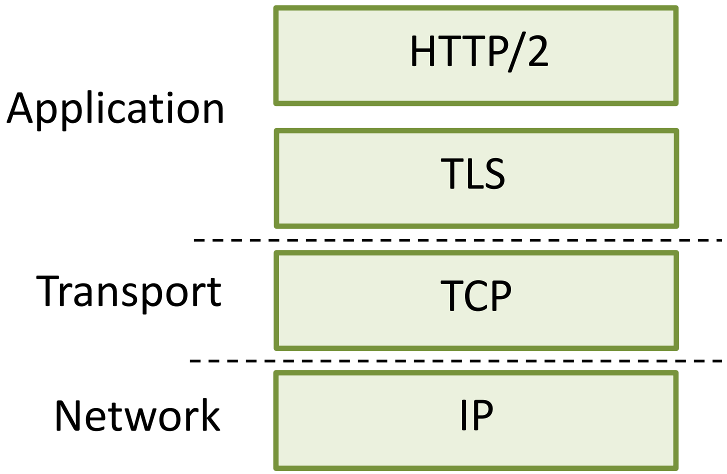
- TCP, UDP: principal transport protocols for 40 years
- different “flavors” of TCP developed, for specific scenarios:

Scenario	Challenges
Long, fat pipes (large data transfers)	Many packets “in flight”; loss shuts down pipeline
Wireless networks	Loss due to noisy wireless links, mobility; TCP treat this as congestion loss
Long-delay links	Extremely long RTTs
Data center networks	Latency sensitive
Background traffic flows	Low priority, “background” TCP flows

- moving transport–layer functions to application layer, on top of UDP
  - HTTP/3: QUIC

QUIC: Quick UDP Internet Connections

- application-layer protocol, on top of UDP
  - increase performance of HTTP
  - deployed on many Google servers, apps (Chrome, mobile YouTube app)



HTTP/2 over TCP

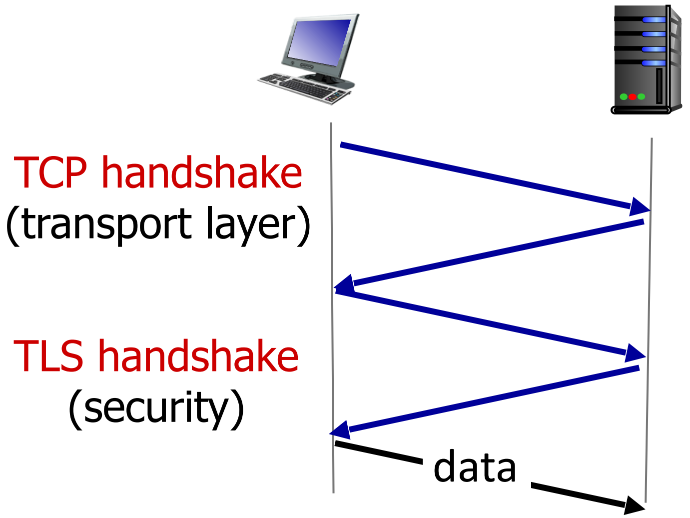
QUIC: Quick UDP Internet Connections

adopts approaches we've studied in this chapter for connection establishment, error control, congestion control

- **error and congestion control:** "Readers familiar with TCP's loss detection and congestion control will find algorithms here that parallel well-known TCP ones." [from QUIC specification]
- **connection establishment:** reliability, congestion control, authentication, encryption, state established in one RTT

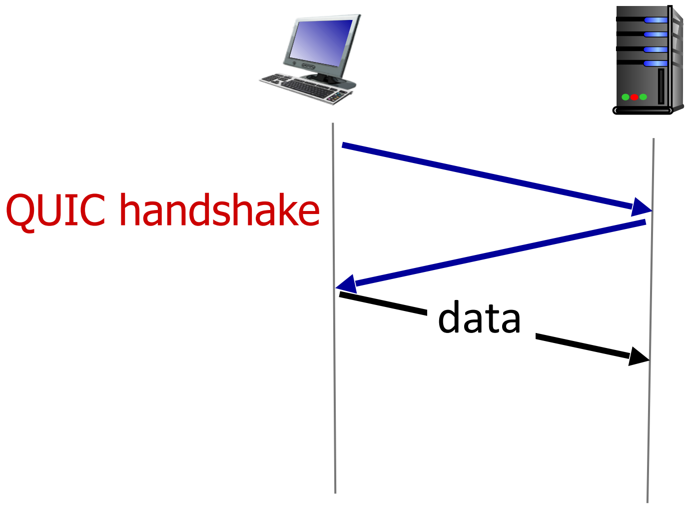
- multiple application-level "streams" multiplexed over single QUIC connection
  - separate reliable data transfer, security
  - common congestion control

QUIC: Connection establishment



TCP (reliability, congestion control state)
+ TLS (authentication, crypto state)

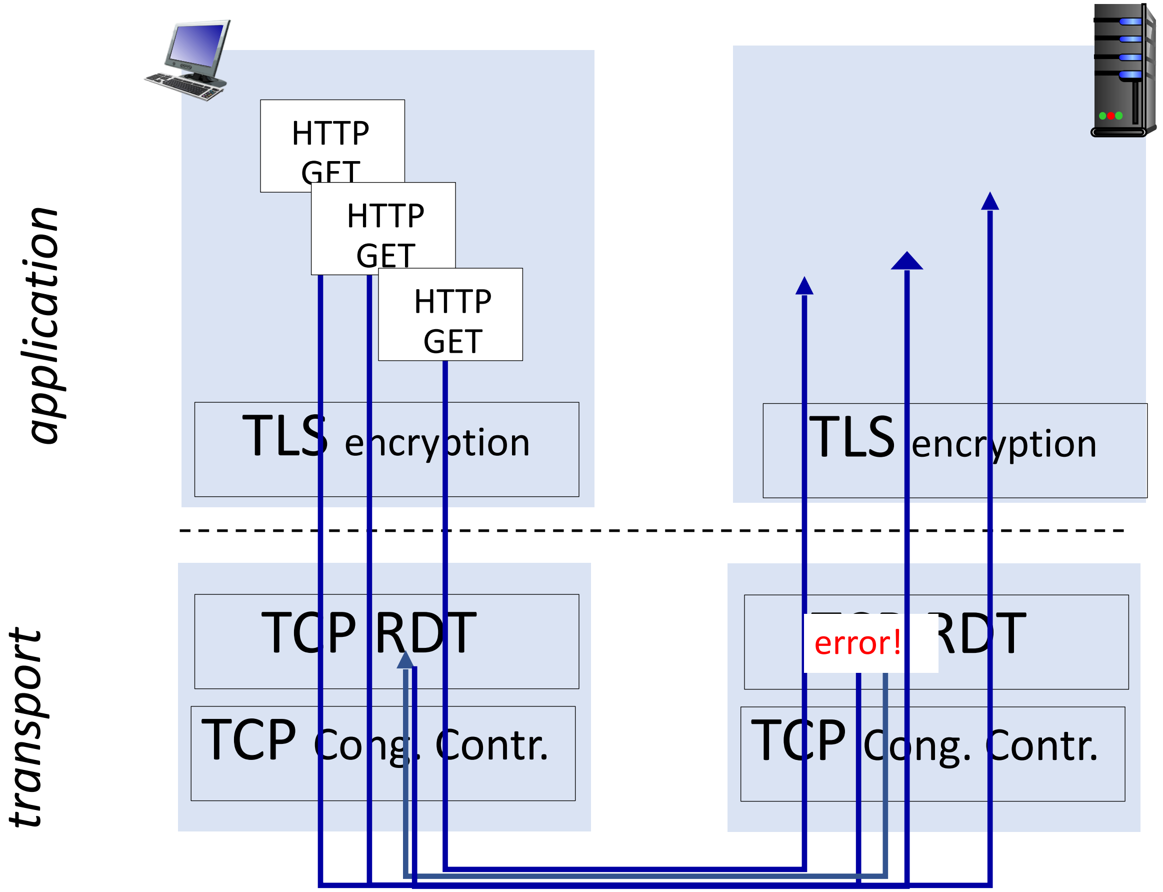
- 2 serial handshakes



QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

Chapter 3: summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation, implementation in the Internet
  - UDP
  - TCP

Up next:

- leaving the network "edge" (application, transport layers)
- into the network "core"
- two network-layer chapters:
  - data plane
  - control plane

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