

CS 725/825 & IT 725

Lecture 16

Transport Layer

October 23, 2024

Retransmission Timeout

Initialization:

RFC 6298

$$\text{RTO} \leftarrow 1 \text{ sec}$$

After the first measurement:

$$\text{SRTT} \leftarrow R$$

$$\text{RTTVAR} \leftarrow R/2$$

$$\text{RTO} \leftarrow \text{SRTT} + \max(G, K * \text{RTTVAR})$$

After subsequent measurements:

$$\text{RTTVAR} \leftarrow (1 - \text{beta}) * \text{RTTVAR} + \text{beta} * |\text{SRTT} - R'|$$

$$\text{SRTT} \leftarrow (1 - \text{alpha}) * \text{SRTT} + \text{alpha} * R'$$

$$\text{RTO} \leftarrow \text{SRTT} + \max(G, K * \text{RTTVAR})$$

Where:

R - first RTT measurement

R' - subsequent RTT measurement

RTTVAR - RTT variance

SRTT - smoothed RTT estimate

RTO - retransmission timeout

G - clock granularity

Recommended values:

alpha=1/8, beta=1/4, K=4

Exponential Back-off

RTO after a timeout:

Recommended value: $q = 2$

$$\text{RTO} \leftarrow q * \text{RTO}$$

This a **congestion control mechanism** since retransmissions are delayed after packet loss detected. The delay is increasing **exponentially** with more packet losses.

Transmission Window

- ▶ Network provides **no explicit indication of congestion**
- ▶ Source observes **RTT** and **packet loss** and **adjusts transmission rate** according to its estimate of the congestion state of the network
- ▶ Transmission **window size** is **proportional** to the **maximum transmission rate**
- ▶ **Additive Increase Multiplicative Decrease (AIMD)**
 - better safe than sorry

Network Congestion Control

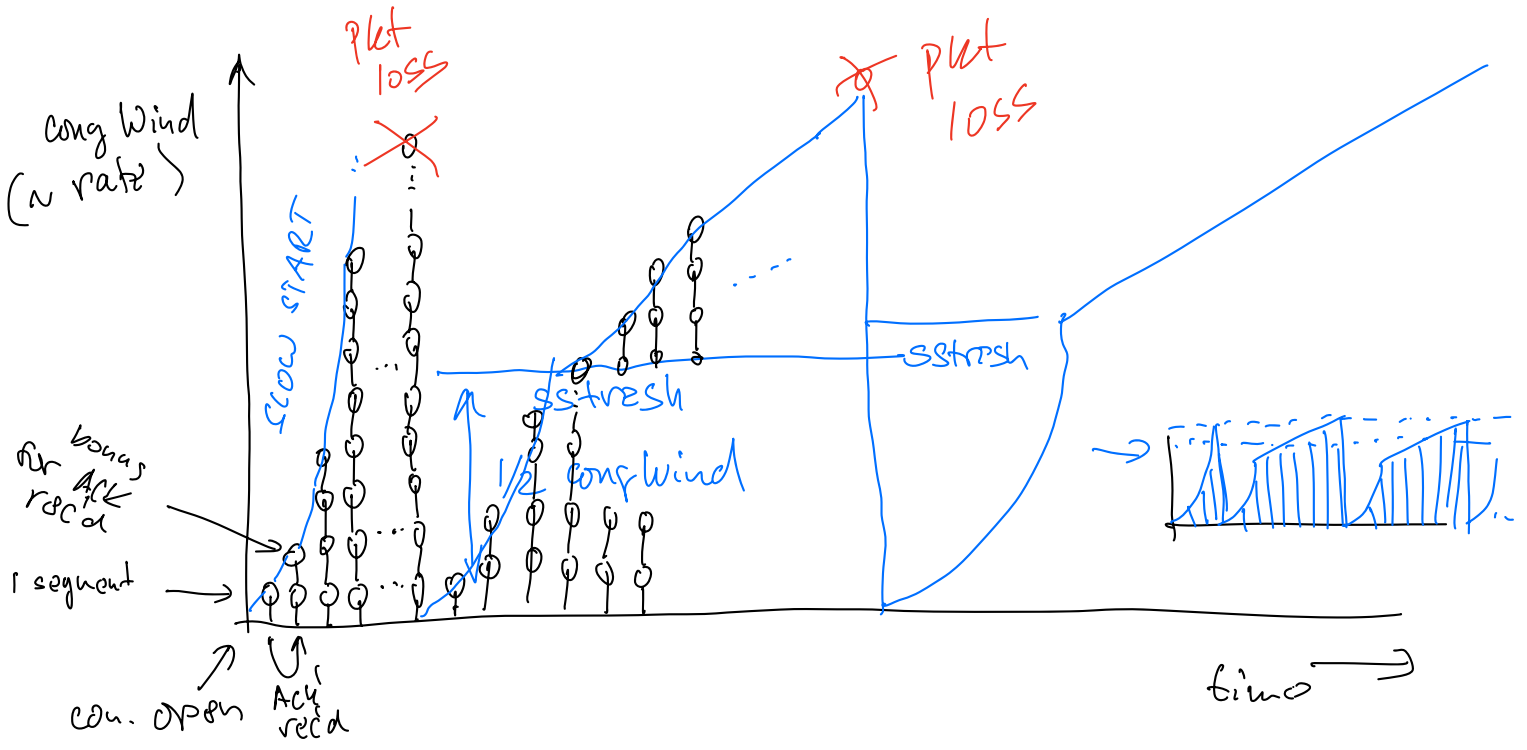
▶ Method:

$TransWind = \min(RecvWind, CongWind)$

$EffectiveWind = TransWind - (LastByteSent - LastByteAckd)$

- ▶ *EffectiveWind* - used in transmission
- ▶ *RecvWind* - from Window Size field
- ▶ *CongWind* - transmitter's estimate of how many unacknowledged packets can be pushed onto the network without causing congestion

TCP CONGESTION WINDOW



Congestion Window (original)

- ▶ Components algorithms of TCP network congestion control (RFC 2001):
 - **Slow Start** - initial growth of **CongWind**
 - **Congestion Avoidance** - AIMD-based “search” for optimal rate
 - **Fast Retransmit** - quick recovery from isolated packet losses
 - **Fast Recovery** - undoing congestion control steps under Fast Recovery

Variants of TCP (examples)

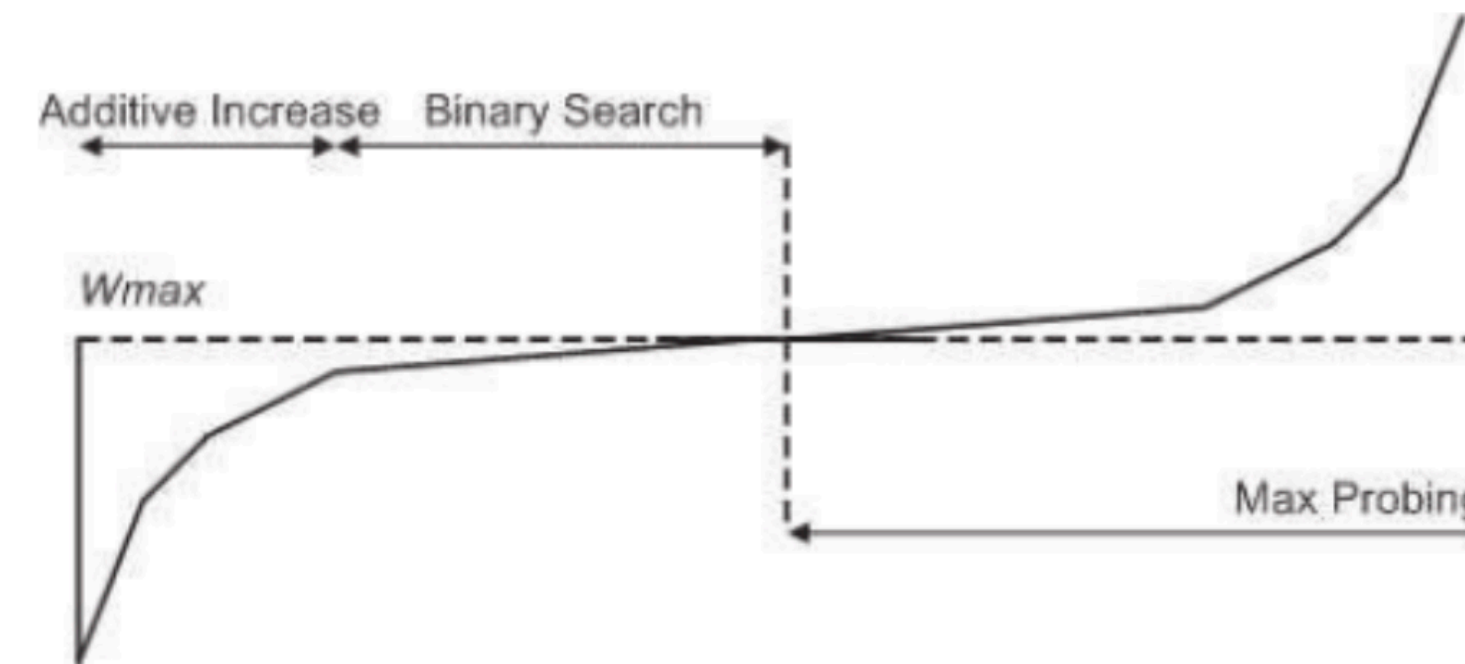
- ▶ Original TCP (RFC1122)
- ▶ TCP Tahoe (adds Fast Retransmit)
- ▶ TCP **Reno** (adds Fast Recovery)
- ▶ TCP Vegas (RTT-based)
- ▶ TCP BIC and **CUBIC** (Linux up to kernel 3.2)
- ▶ Compound TCP (Windows since Vista)
- ▶ TCP Proportional Rate Reduction (PRR) (Linux)
- ▶ TCP Bottleneck Bandwidth and Round-trip propagation time (**BBR**) (RTT-based, developed by Google)

TCP Vegas

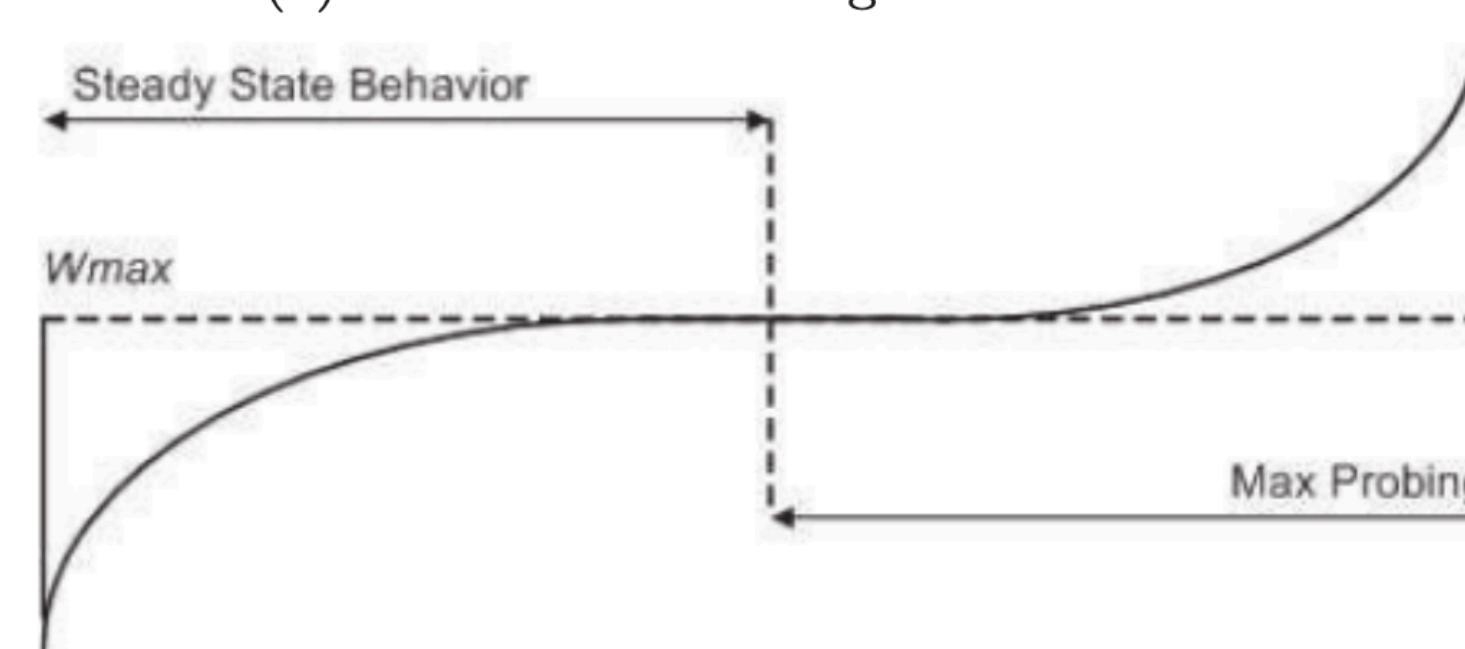
- ▶ RTT observed
- ▶ An increase in RTT indicates congestion
 - reduce transmission rate
- ▶ Steady RTT measurements indicate underutilization
 - slowly increase transmission rate until RTT starts increasing

TCP CUBIC

- ▶ An update of TCP BIC (Binary Increase Congestion control)
- ▶ “modifies the linear **window growth function** of existing TCP standards to be a **cubic function** in order to improve the scalability of TCP over fast and long distance networks”



(a) BIC-TCP window growth function.



(b) CUBIC window growth function.

From: Sangtae Ha, Injong Rhee, and Lisong Xu. 2008. CUBIC: a new TCP-friendly high-speed TCP variant. SIGOPS Oper. Syst. Rev. 42, 5 (July 2008), 64–74. DOI:<https://doi.org/10.1145/1400097.1400105>

TCP BBR

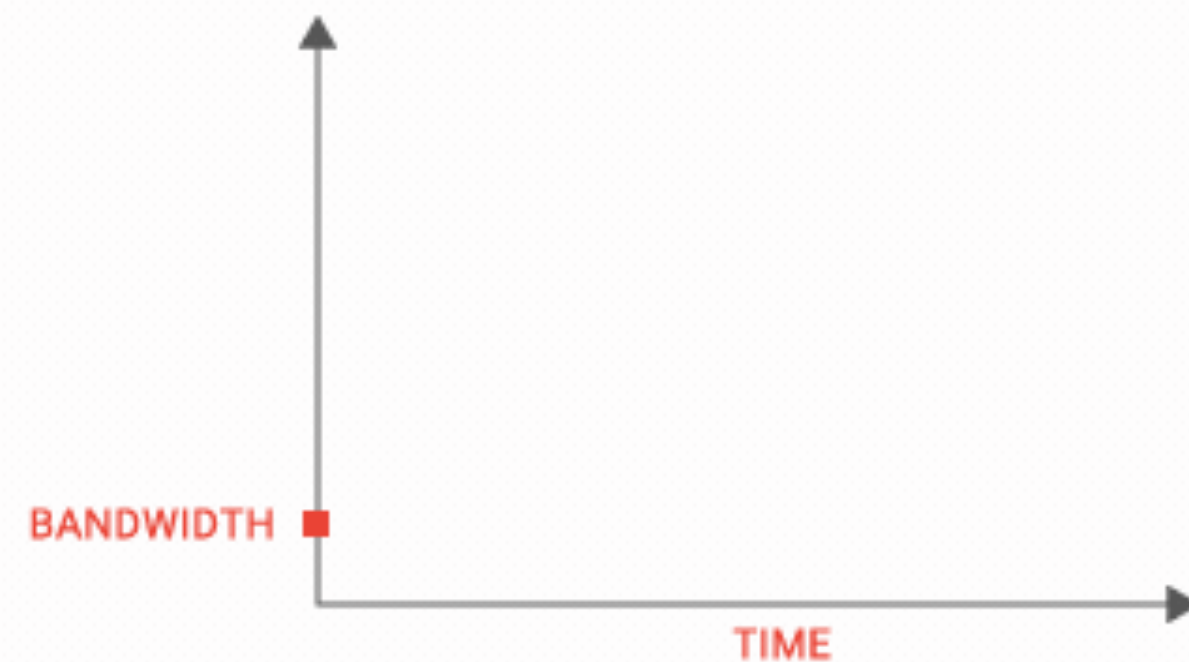
- ▶ Bottleneck Bandwidth and Round-trip propagation time
- ▶ Designed by Google (~2016)
 - with YouTube as the motivating use case
 - available in Linux kernel 4.9+
- ▶ As the protocol name suggests:
 - “BBR congestion control computes the sending rate based on the delivery rate (throughput) estimated from ACKs” (comment in tcp-bbr.c in Linux kernel)

TCP BBR

- ▶ One has to be careful when making claims:

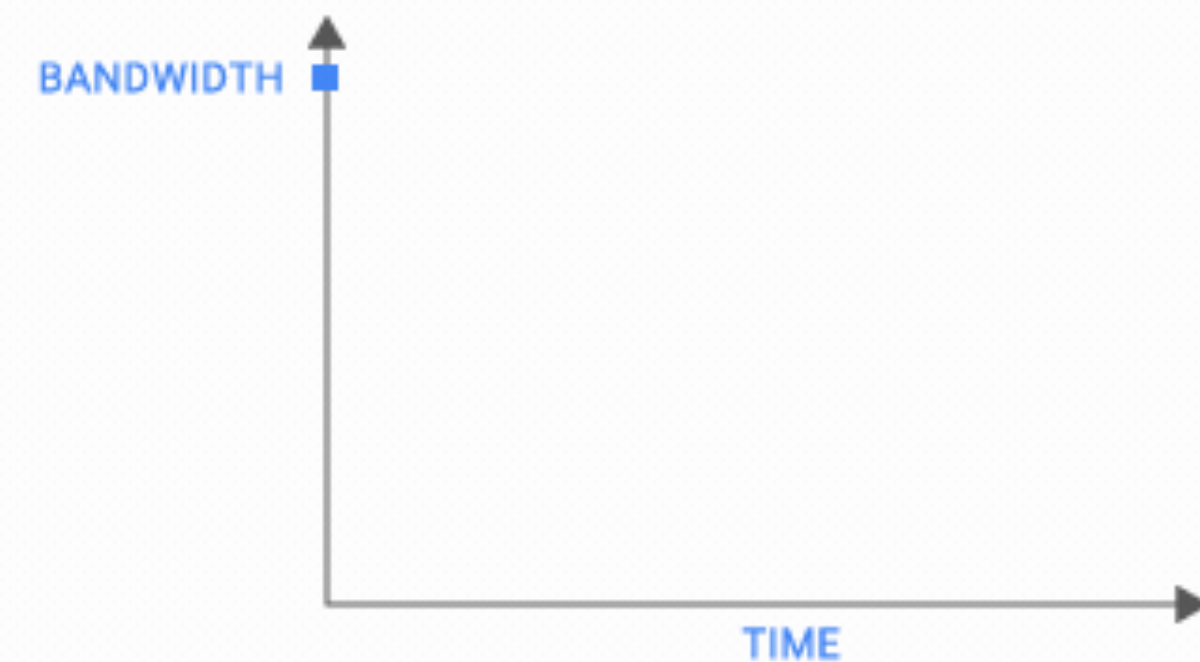
TCP before BBR

Today's Internet is not moving data as well as it should. TCP sends data at lower bandwidth because the 1980s-era algorithm assumes that packet loss means network congestion.



TCP BBR

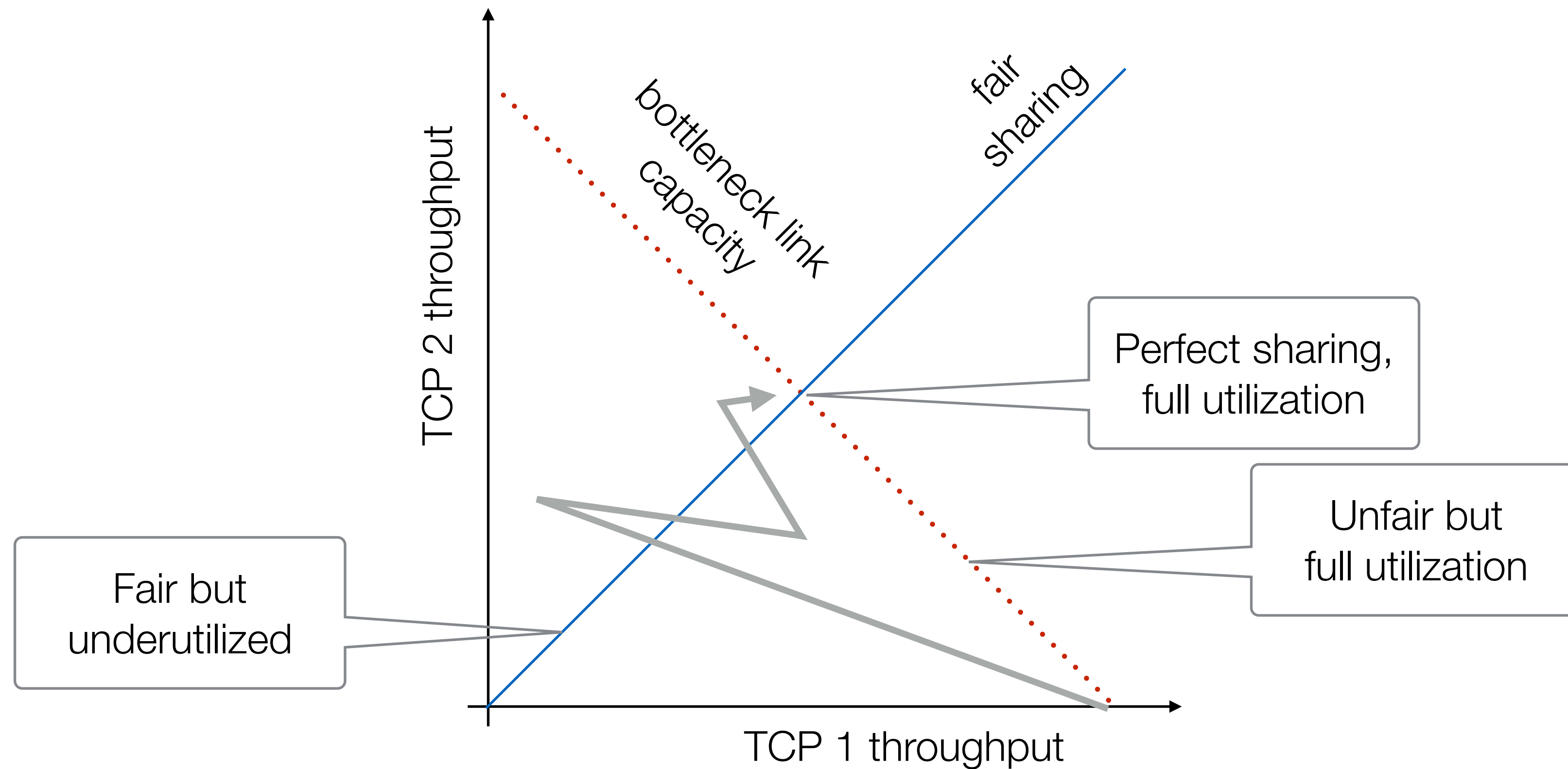
BBR models the network to send as fast as the available bandwidth and is 2700x faster than previous TCPs on a 10Gb, 100ms link with 1% loss. BBR powers google.com, youtube.com, and apps using Google Cloud Platform services.



From: <https://cloud.google.com/blog/products/networking/tcp-bbr-congestion-control-comes-to-gcp-your-internet-just-got-faster> (interestingly, the link no longer works, a copy of the article is still available at <https://www.googblogs.com/tcp-bbr-congestion-control-comes-to-gcp-your-internet-just-got-faster/>)

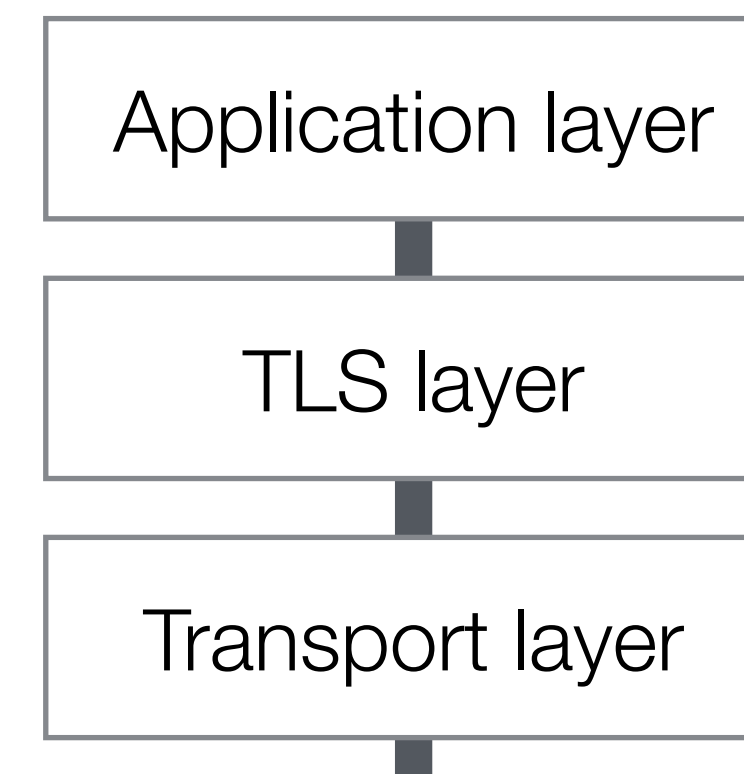
TCP Fairness

- ▶ **Example:** two TCP connections competing with each other on a bottleneck link:



Transport Layer Security

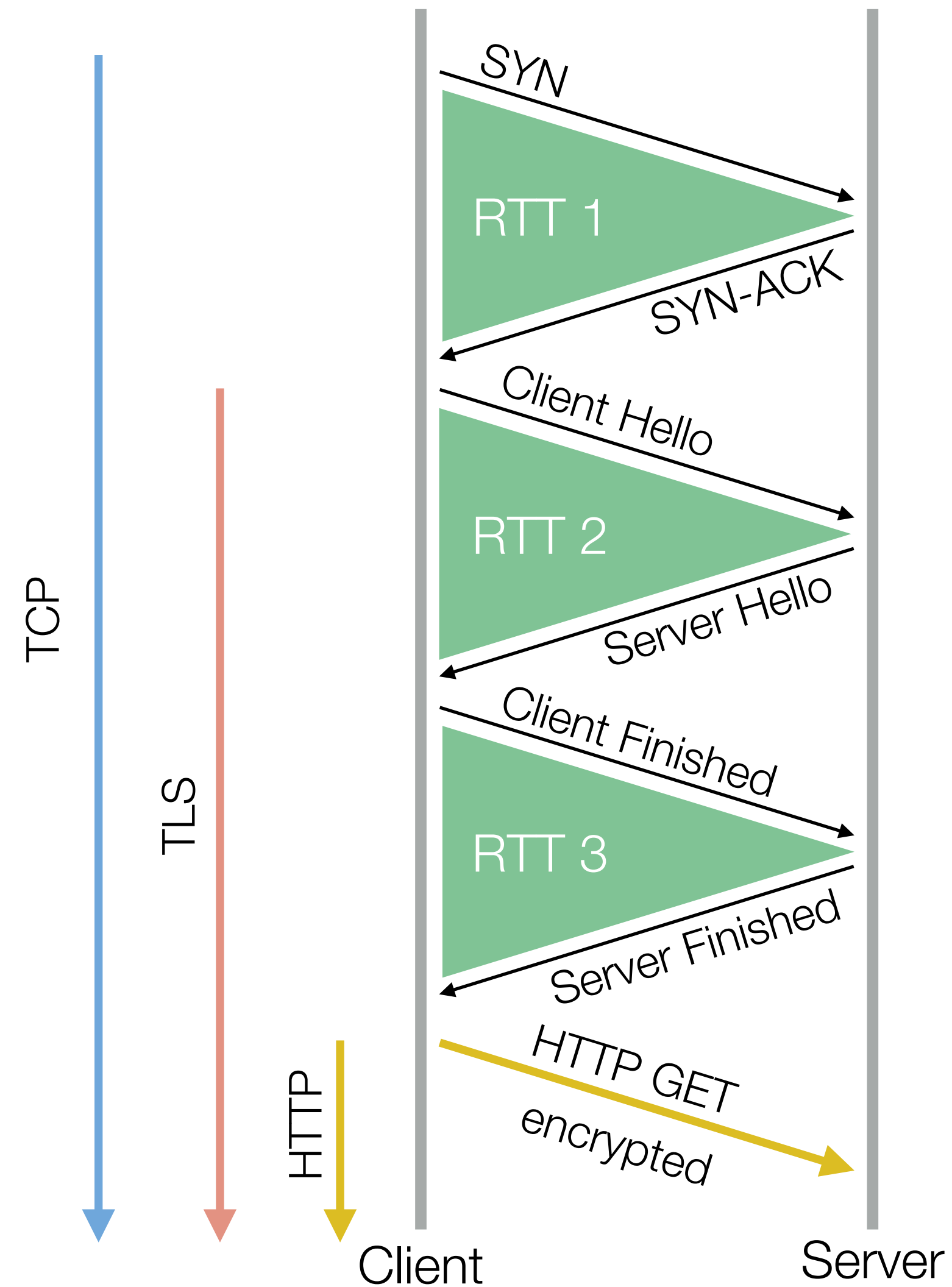
- ▶ **Transport Layer Security (TLS)** - cryptographic protocols that to provide privacy (encryption) and data integrity protection
- ▶ ... earlier versions known as SSL (Secure Socket Layer) is now deprecated but the term is widely used as a synonym for TLS
- ▶ Most used version TLS 1.2 (2008)
- ▶ Current version: TLS 1.3



TLS connection latency

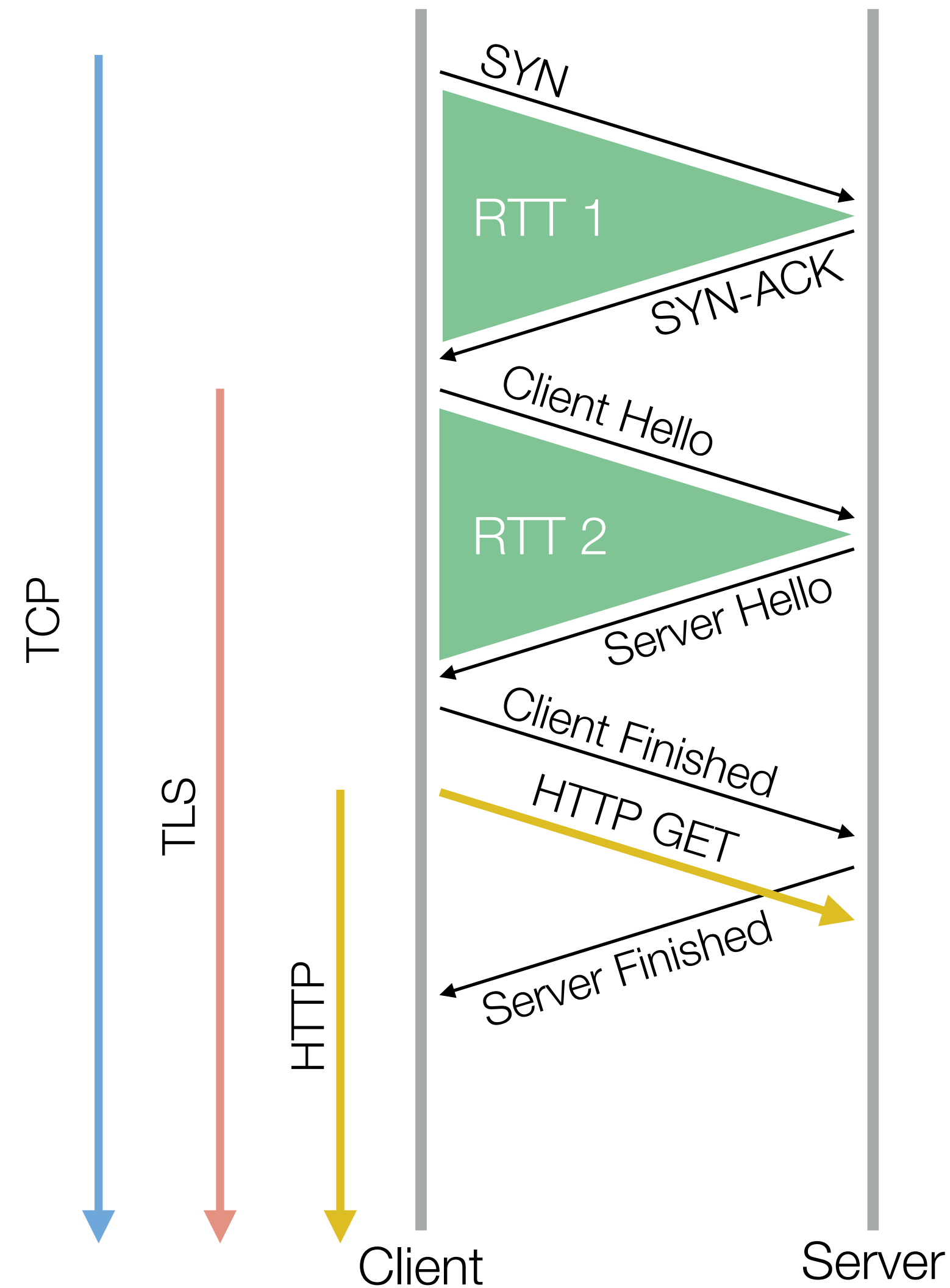
► TLS 1.2

- 3 RTTs required to establish a secure connection



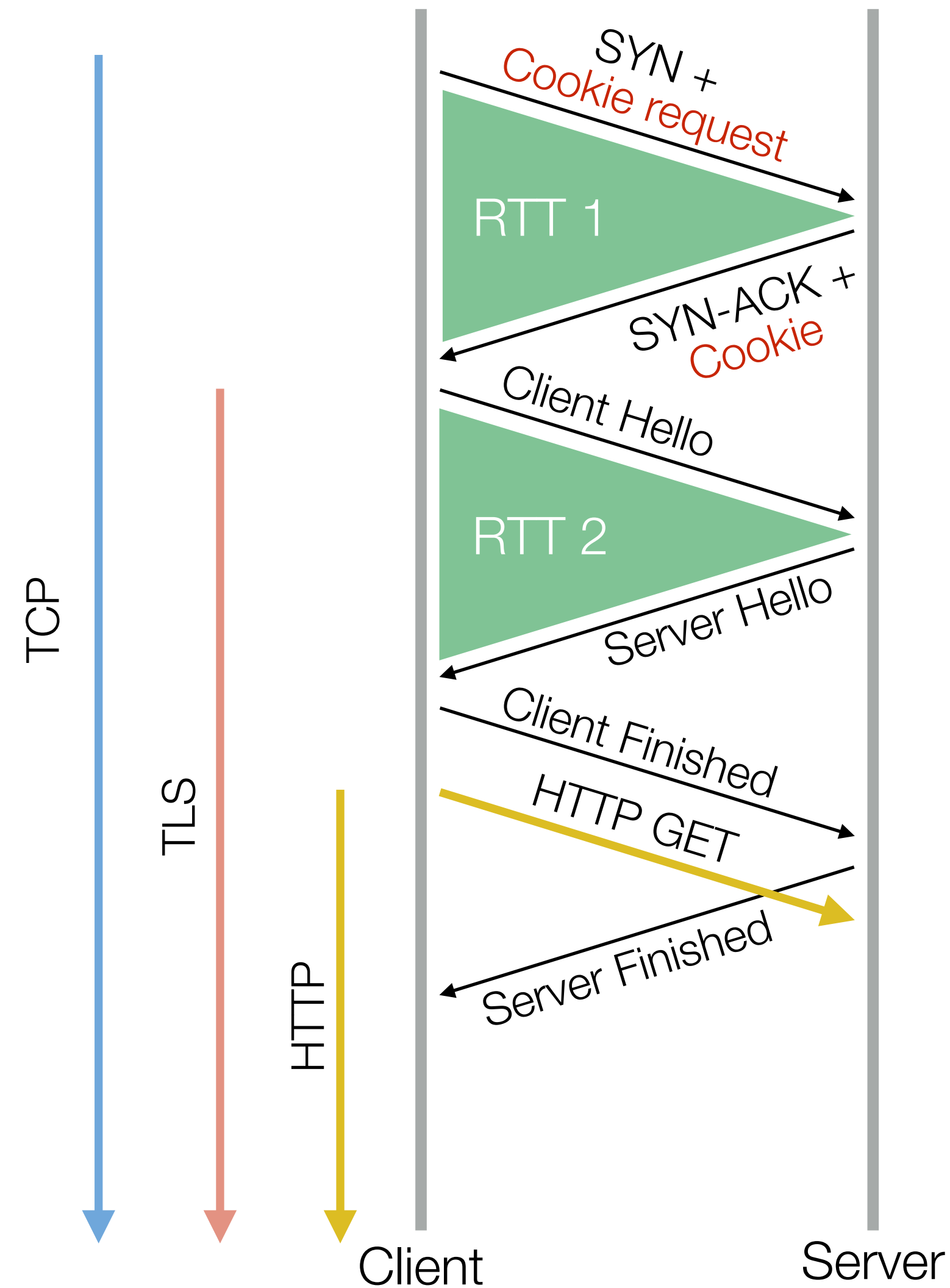
TLS connection latency

- ▶ **TLS False Start** option
 - 2 RTTs required to establish a secure connection



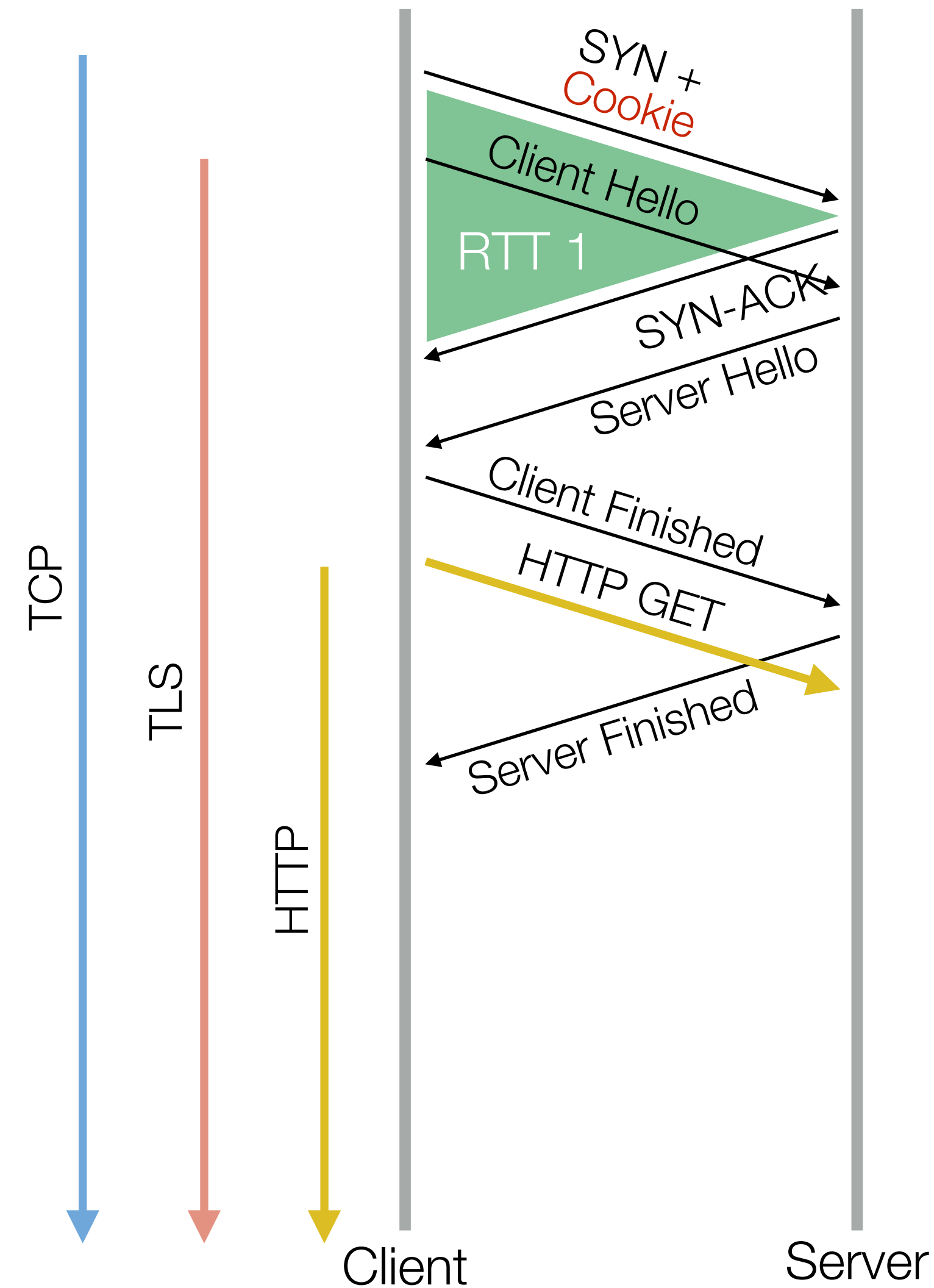
TLS connection latency

- ▶ **TLS Fast Open** option
 - when client connects for the first time, 2 RTTs are still required to establish a secure connection
 - server provides **Fast Open Cookie** to be used to speed-up subsequent connections



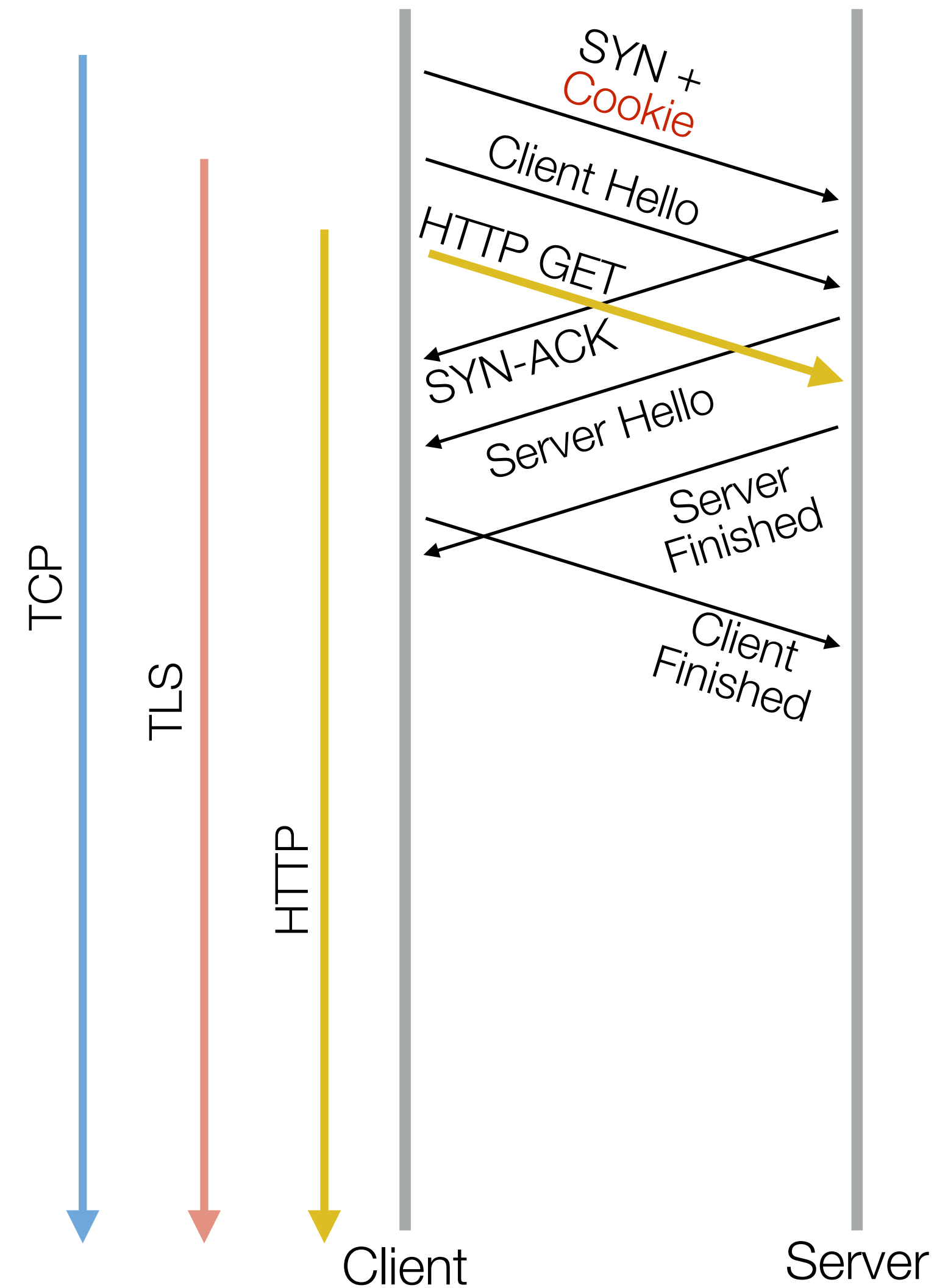
TLS connection latency

- ▶ **TLS Fast Open** option
 - for subsequent connections, only one RTTs required to establish a secure connection
 - client sends previously received **Fast Open Cookie**



TLS connection latency

- ▶ 0-RTT with **TLS 1.3**
 - for subsequent connections (using **Fast Open Cookie**), HTTP command is set before the TLS connection is fully established
 - However, the initial data sent to the server is susceptible (e.g., replay attack)



UDP

- ▶ **User Datagram Protocol** (RFC 768)
 - A wrapper protocol for IP to add port numbers
 - 8 bytes

Source Port	Destination Port
Length	Checksum