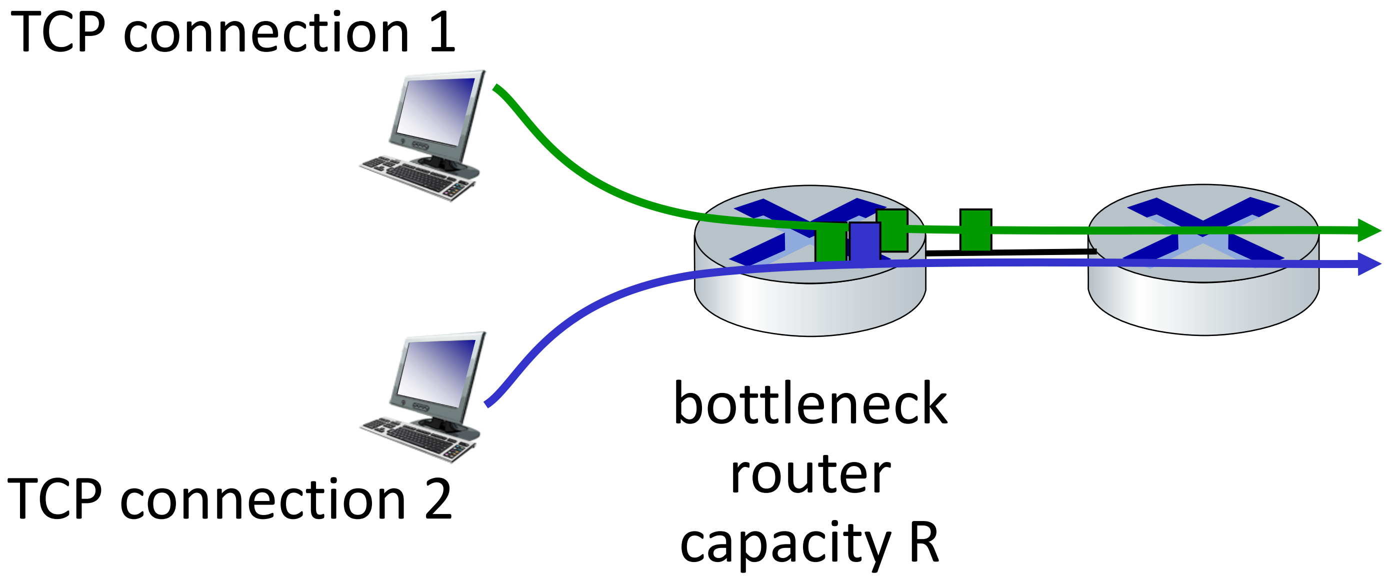
Transport Layer

Anand Baswade

anand@iitbhilai.ac.in

TCP fairness

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



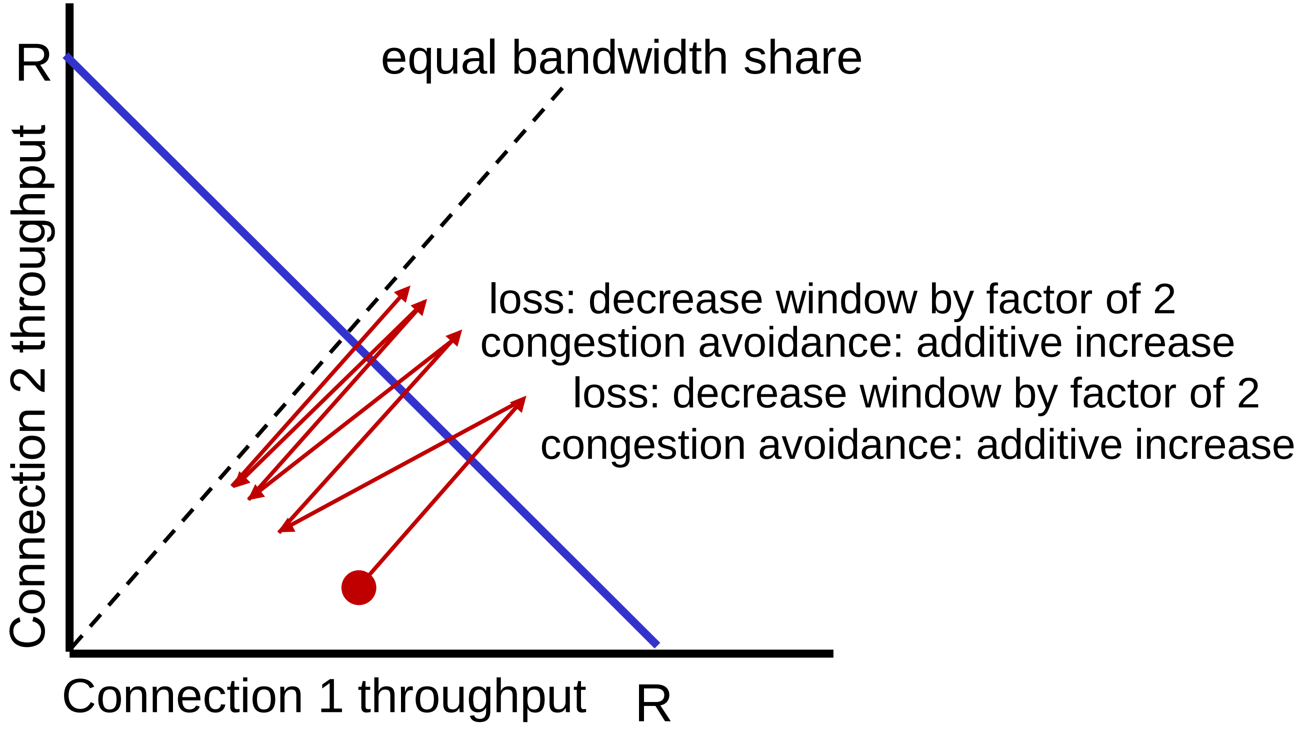
Goal:

1. Fairness: Every user should get a fair share of link
2. Capacity: Network link bandwidth should be well utilized

Q: is TCP Fair?

Example: two competing TCP sessions:

- additive increase – Increases Efficiency
- multiplicative decrease decreases throughput proportionally - Increases Fairness



Is TCP fair?

A: Yes, under idealized assumptions:

- same RTT
- fixed number of sessions

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

TCP Options

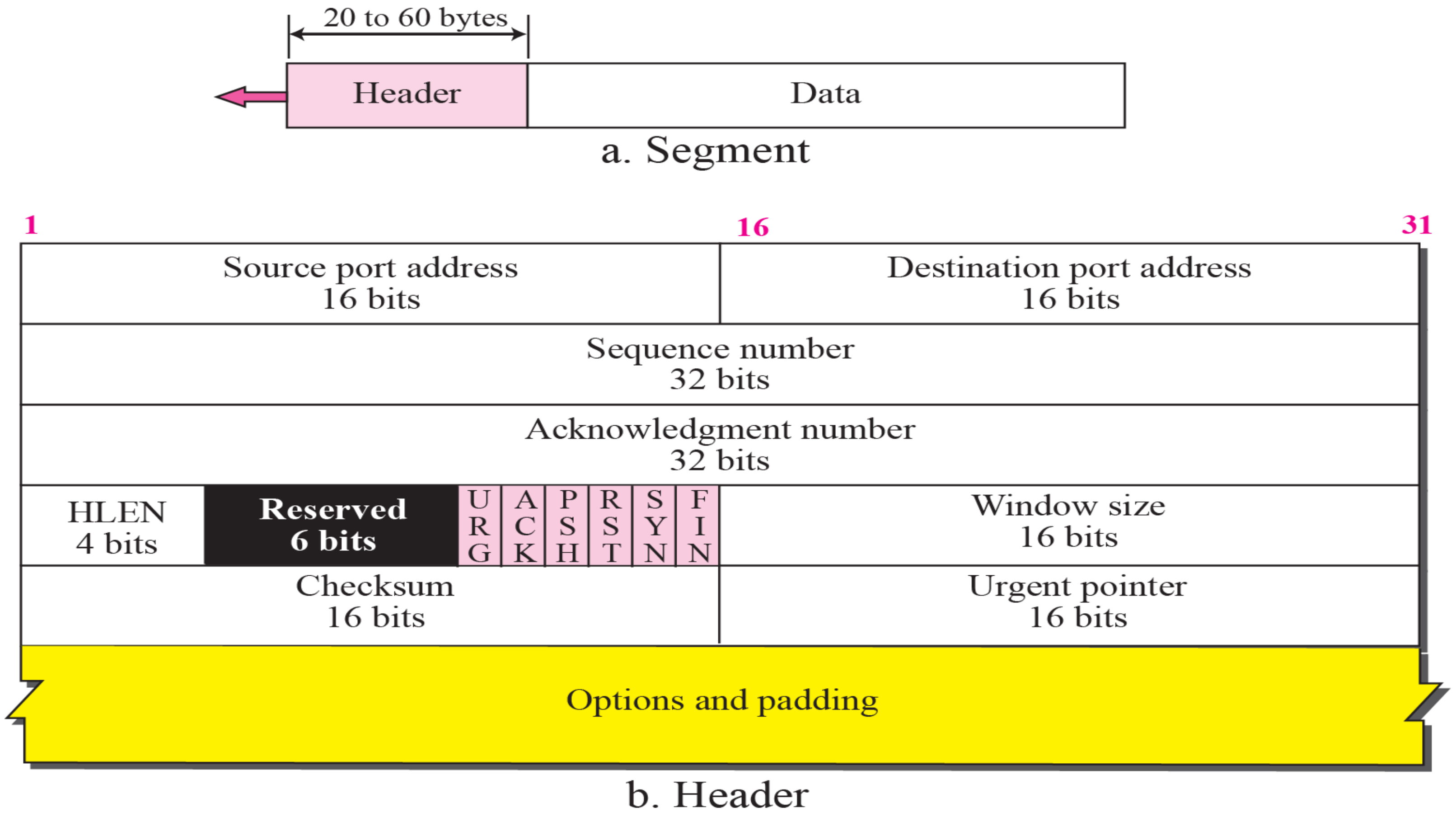
- The TCP header can have up to 40 bytes of optional information. Options convey additional information to the destination.

- We can define two categories of options: 1-byte options and multiple-byte options.

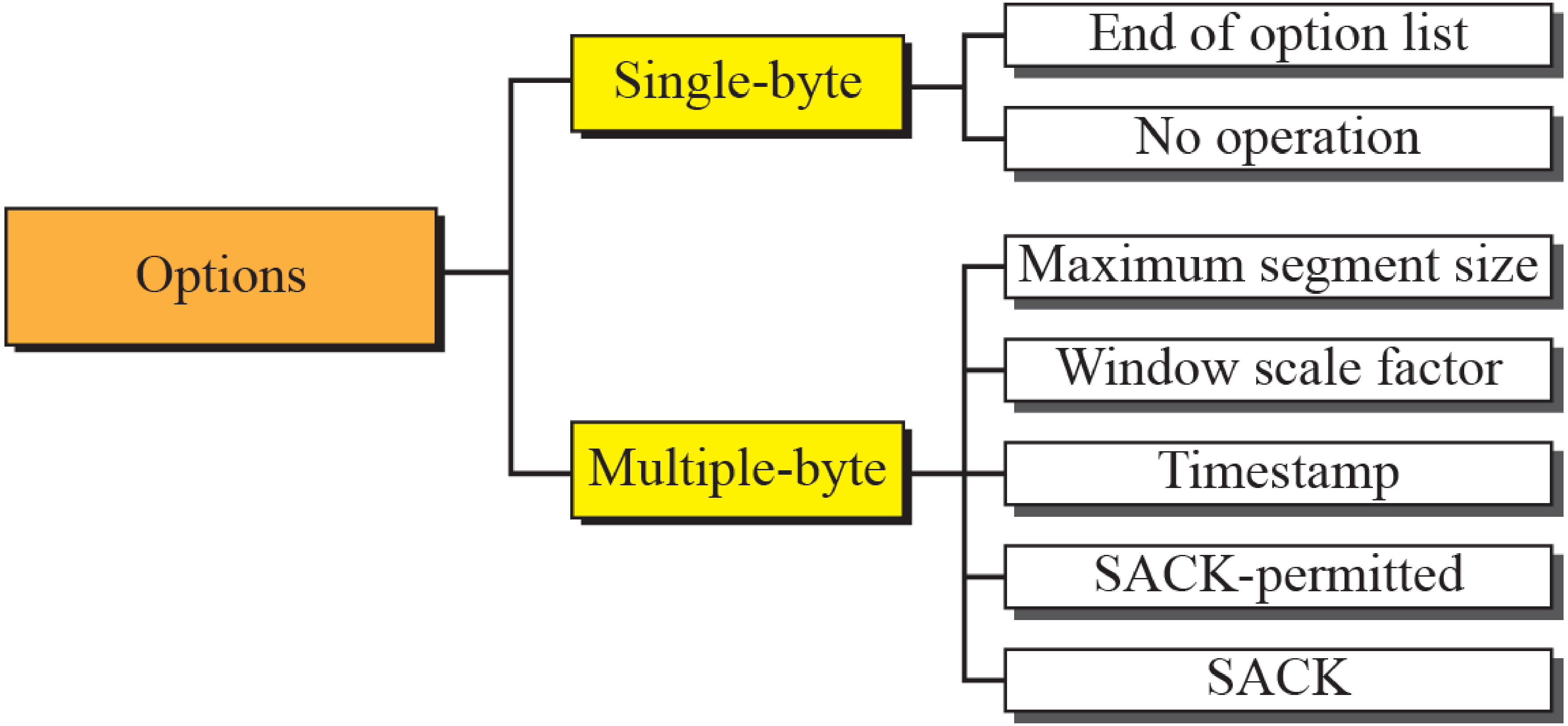
- The first category contains two types of options: end of option list and no operation.

- The second category, in most implementations, contains five types of options: maximum segment size, window scale factor, timestamp, SACK-permitted, and SACK.

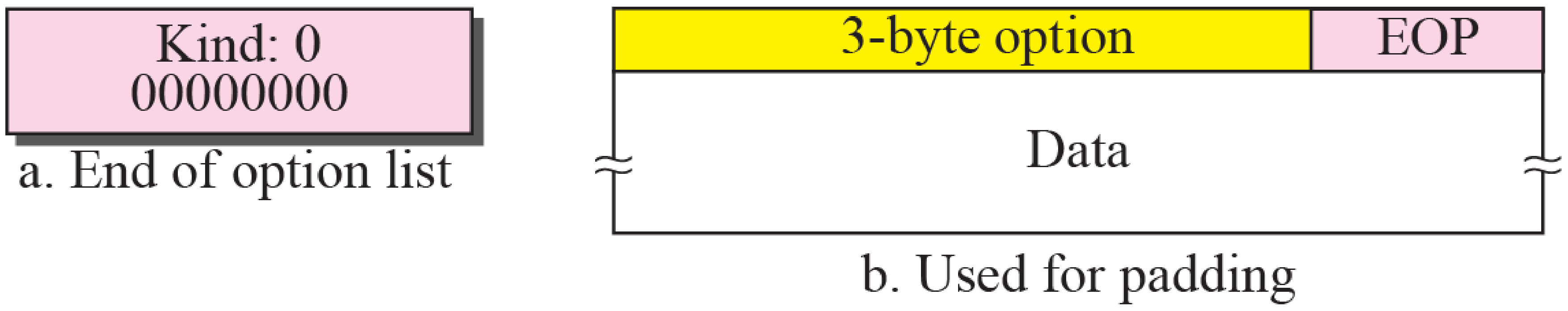
TCP segment with options format



Options


Options
Single-byte
Multiple-byte
End of option list
No operation
Maximum segment size
Window scale factor
Timestamp
SACK-permitted
SACK

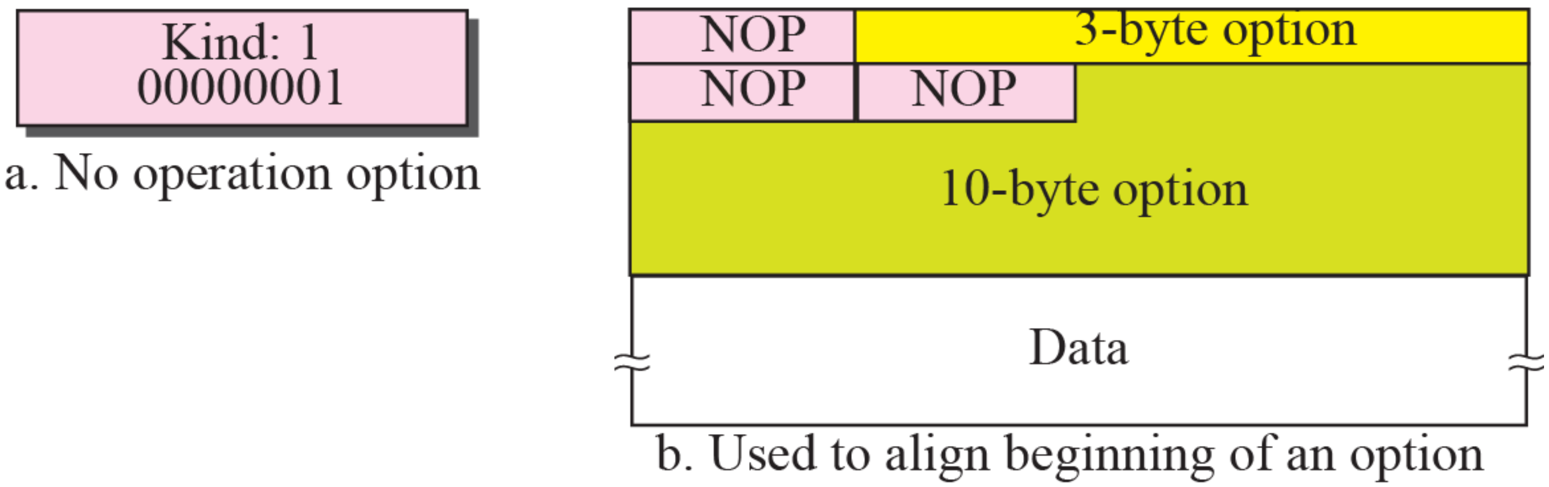
End-of-option option



EOP can be used only once.

No-operation option

- It is used to separate the different options used within the TCP Option field.



NOP can be used more than once.

Minimum-segment-size option

1) The MSS (Maximum Segment Size) is defined as the largest block of data that a sender using TCP will send to the receiver.
2) When a connection is initiated a SYN segment is sent, in the process of sending a SYN segment, the sender has the option of announcing its MSS.
3) If a sender doesn't use the options field to declare the MSS then TCP assumes a default of 536bytes (minus the 20 byte TCP header).

Kind: 2 00000010	Length: 4 00000100	Maximum segment size
1 byte	1 byte	2 bytes

The value of MSS is determined during connection establishment and does not change during the connection.

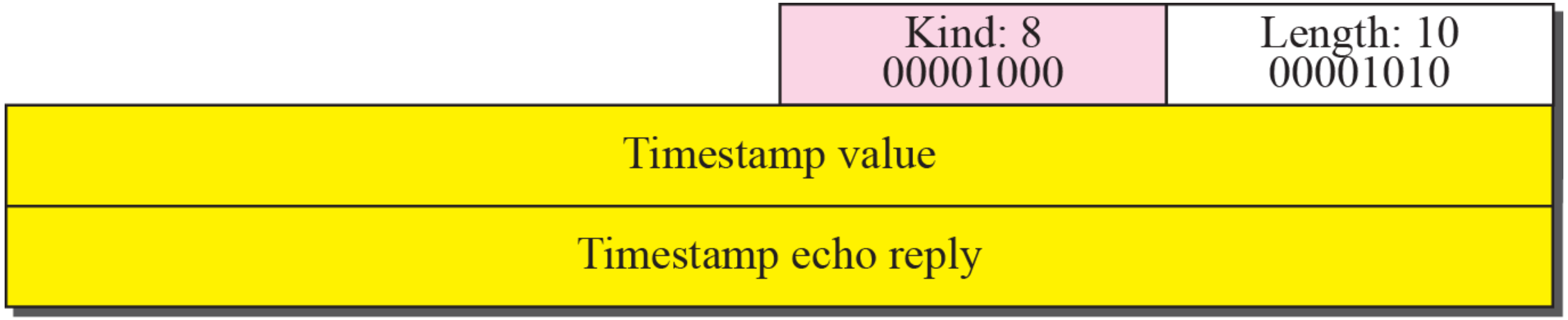
Window-scale-factor option

1. For lines with high bandwidth, high delay or high bandwidth-delay product, 64KB is not enough to keep the sender busy all the time.
2. In RFC 1323, the use of the options field is proposed permitting the use of a window scale factor.
3. This scale factor permits shifting the window size up to 14 bits to the left therefore permitting window sizes of up to $2^{(16+14)} = 2^{30}$ bytes.

Example window size =65535, and scale factor = 2 (2^ 2 = 4 multiply by 4 to window size to calculate the actual window size)

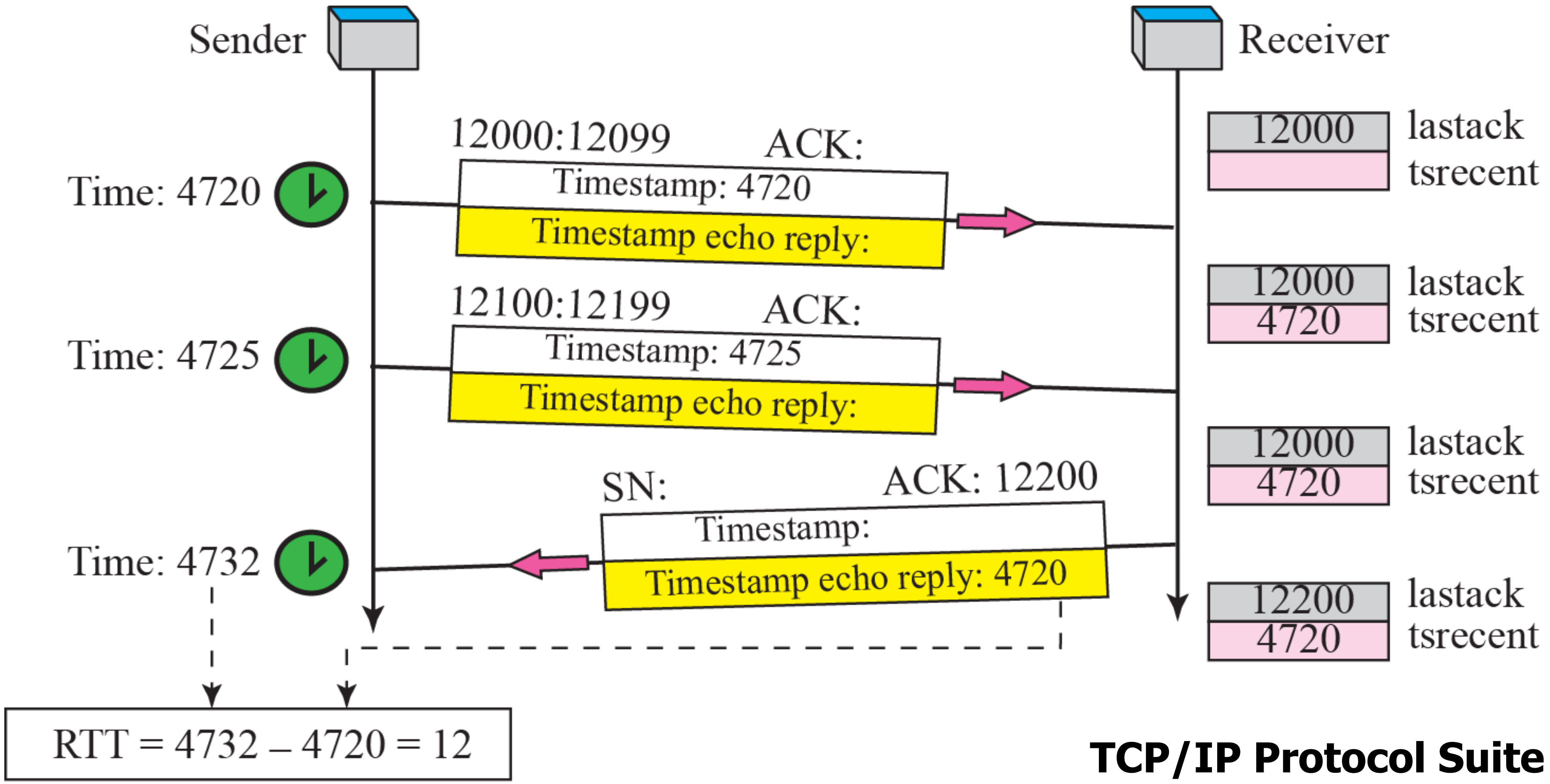
Kind: 3 00000011	Length: 3 00000011	Scale factor
1 byte	1 byte	1 byte

The value of the window scale factor can be determined only during connection establishment; it does not change during the connection.

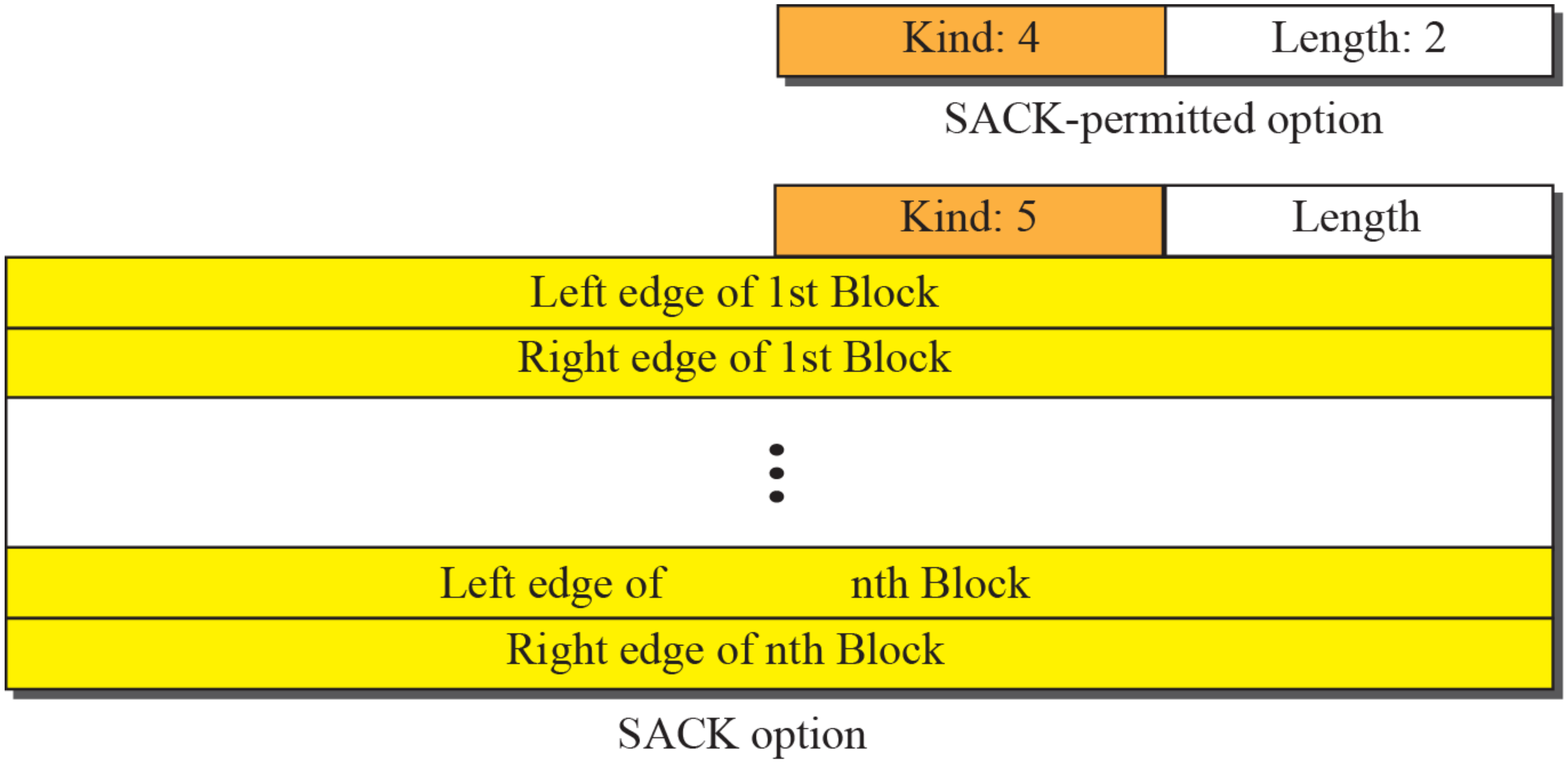


One application of the timestamp option is the calculation of round-trip time (RTT).

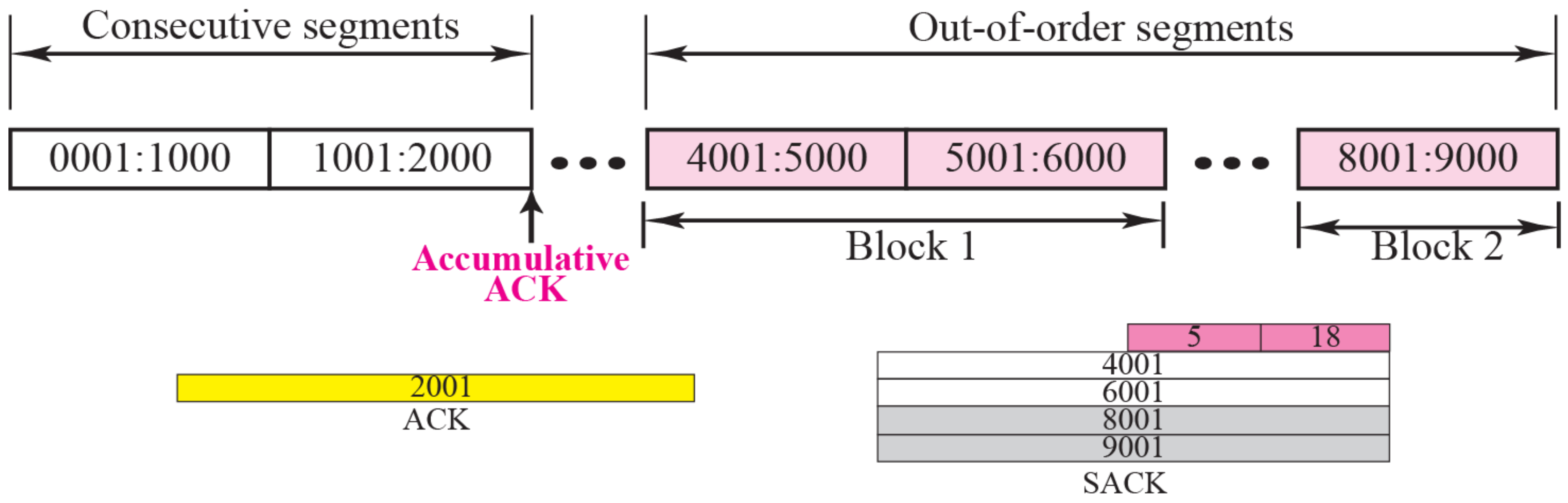
Figure shows an example that calculates the round-trip time for one end. Everything must be flipped if we want to calculate the RTT for the other end.



Selective ACK (*SACK)*

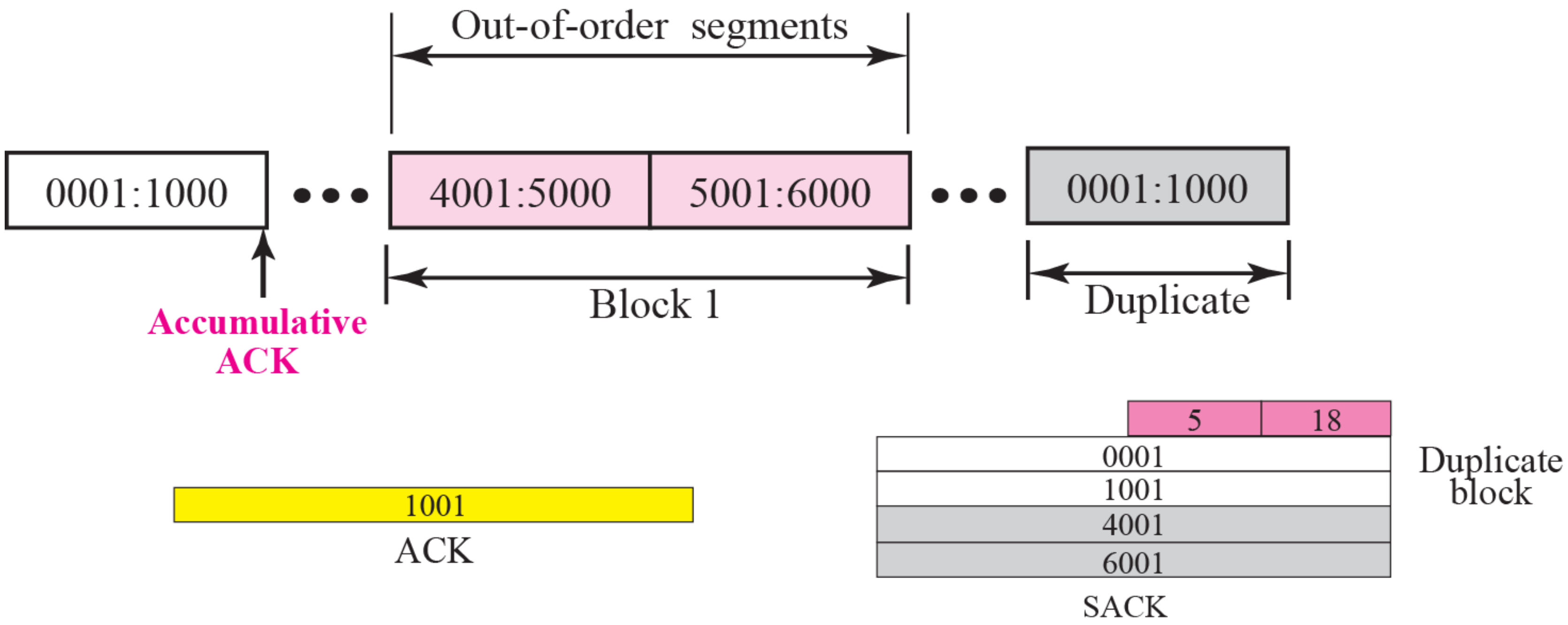


Let us see how the SACK option is used to list out-of-order blocks. In Figure an end has received five segments of data.



TCP/IP Protocol Suite

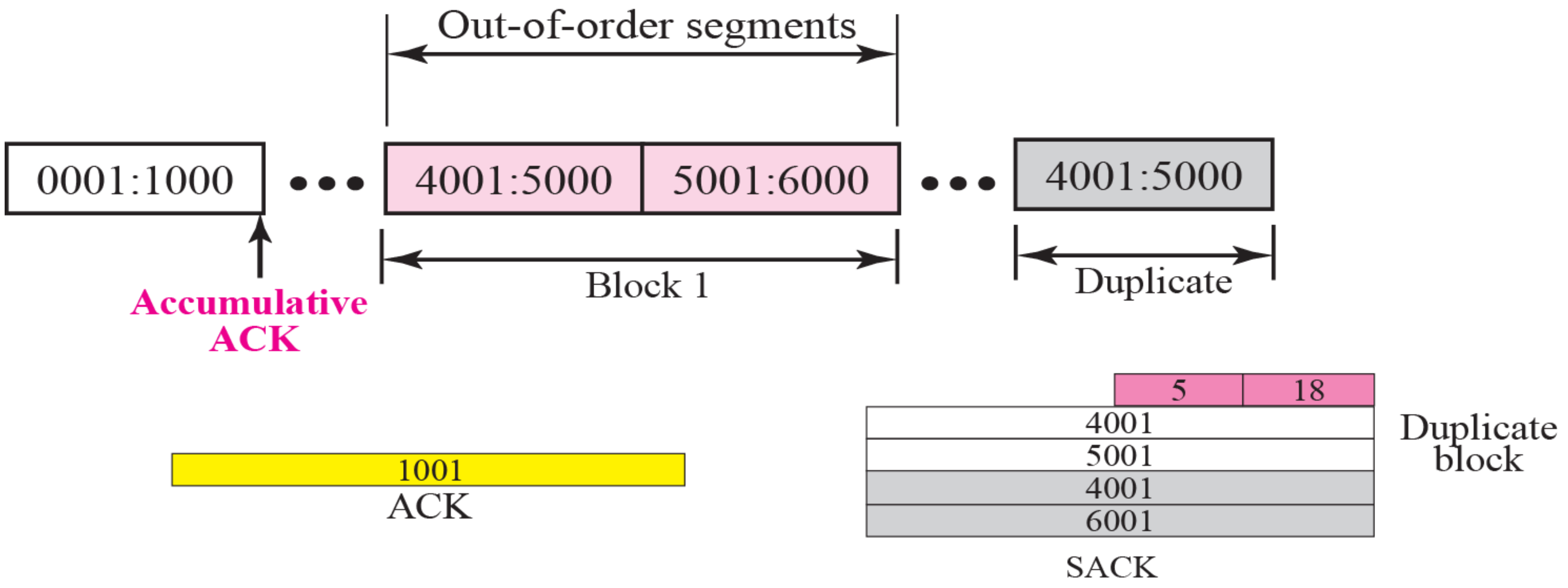
Figure shows how a duplicate segment can be detected with a combination of ACK and SACK.



TCP/IP Protocol Suite

Figure shows what happens if one of the segments in the out-of-order section is also duplicated. In this example, one of the segments (4001:5000) is duplicated.

The SACK option announces this duplicate data first and then the out-of-order block.



A New Transport Protocol

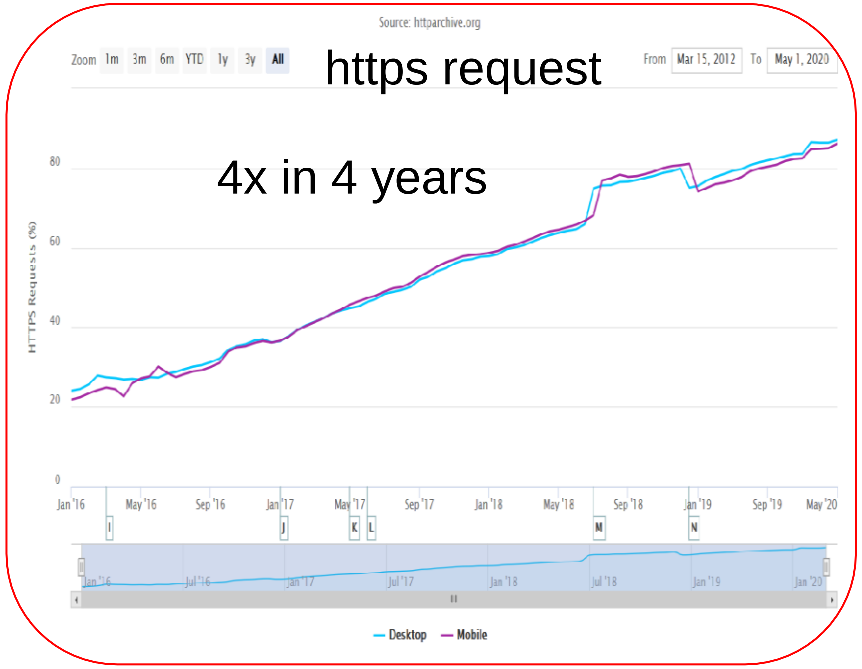
QUIC: Quick UDP Internet Connections

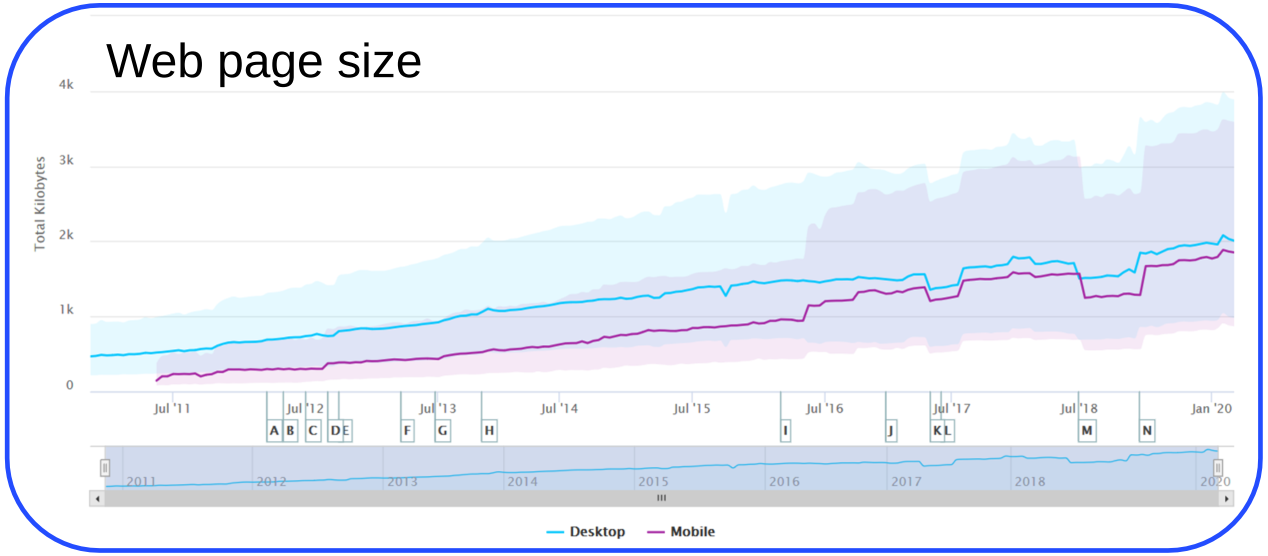
HTTP/3: HTTP over QUIC is next Generation

On 6 June 2022, IETF published HTTP/3 as a Proposed Standard in RFC 9114

Introduction: Change

- **Increasing scale of .... everything**
  - Flow size changes
  - Flow count increases (e.g., web pages)
  - Flow diversity increase (e.g., web pages)
    - Multiple connections





HTTP Network Stack

HTTP
TLS 1.2+
TCP
IP

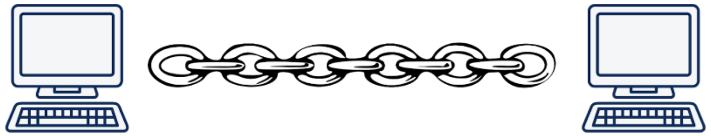
HTTP / 1.1

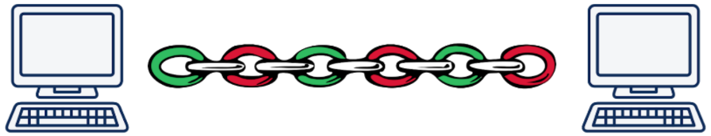
- January 1997
- **Many parallel TCP connection (6 connections per host name)**
- **HTTP head of line blocking**

HTTP / 2

- May 2015
- **Using Single connection per host**
- **Many parallel streams**
- **TCP head of line blocking**

TLS – Transport Layer Security
TCP – Transport Control Protocol
IP – Internet Protocol

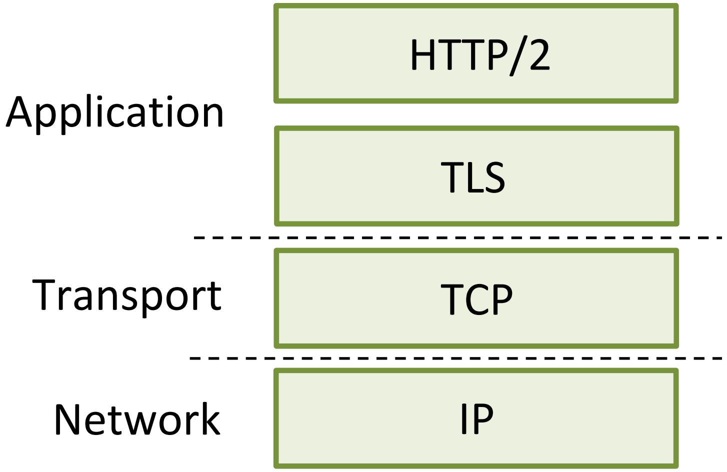




https://http3-explained.haxx.se/en/why-quic

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)

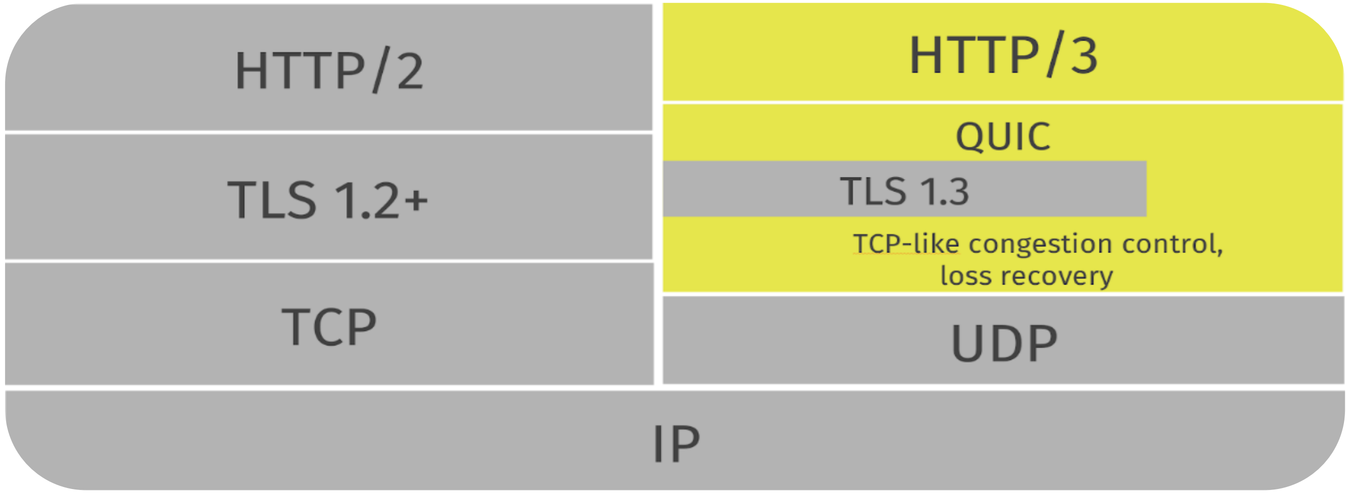


HTTP/2 over TCP

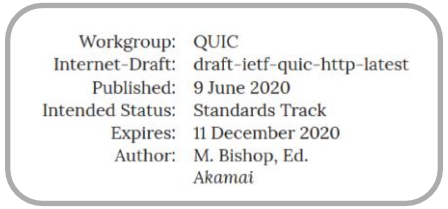
HTTP Over QUIC Network Stack

HTTP over TCP

HTTP over QUIC

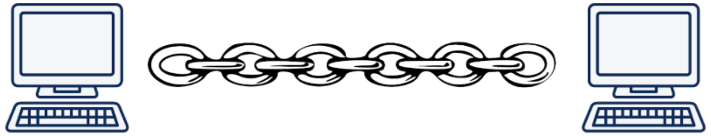


HTTP / 3

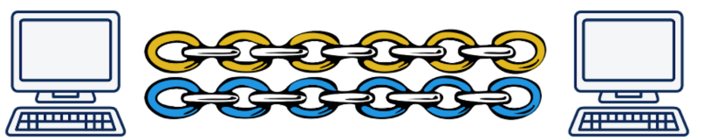


Workgroup:	QUIC
Internet-Draft:	draft-ietf-quic-http-latest
Published:	9 June 2020
Intended Status:	Standards Track
Expires:	11 December 2020
Author:	M. Bishop, Ed.
	Akamai

- **No - TCP head of line blocking**
  - **streams are independent to each other**
- **Faster handshake**
  - **Earlier data**
- **More encryption, always**
- **Over UDP (*Connection less, No resend, No flow control*)**



- Steam impact not connection impact



https://http3-explained.haxx.se/en/why-quic

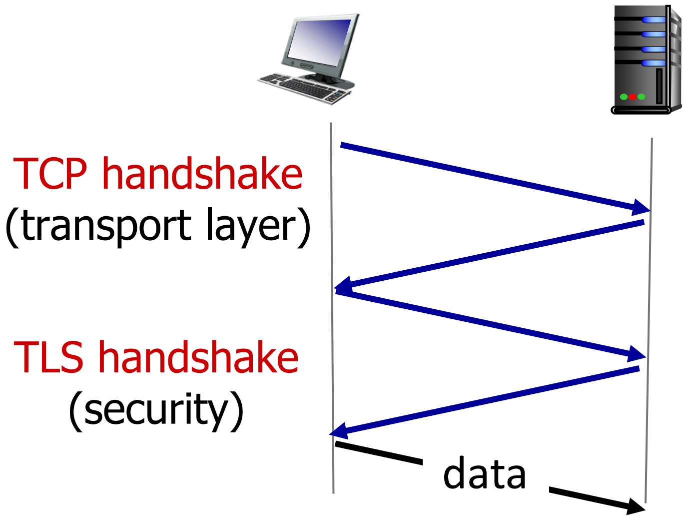
QUIC: Quick UDP Internet Connections

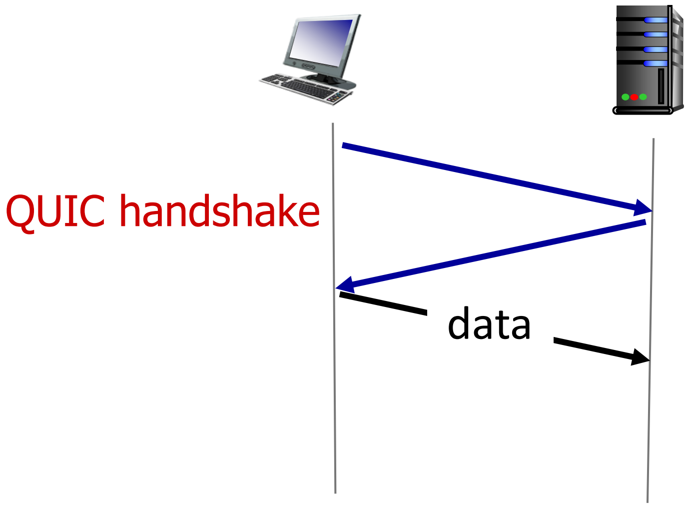
adopts approaches we've studied in this chapter for connection establishment, error control, congestion control

- **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





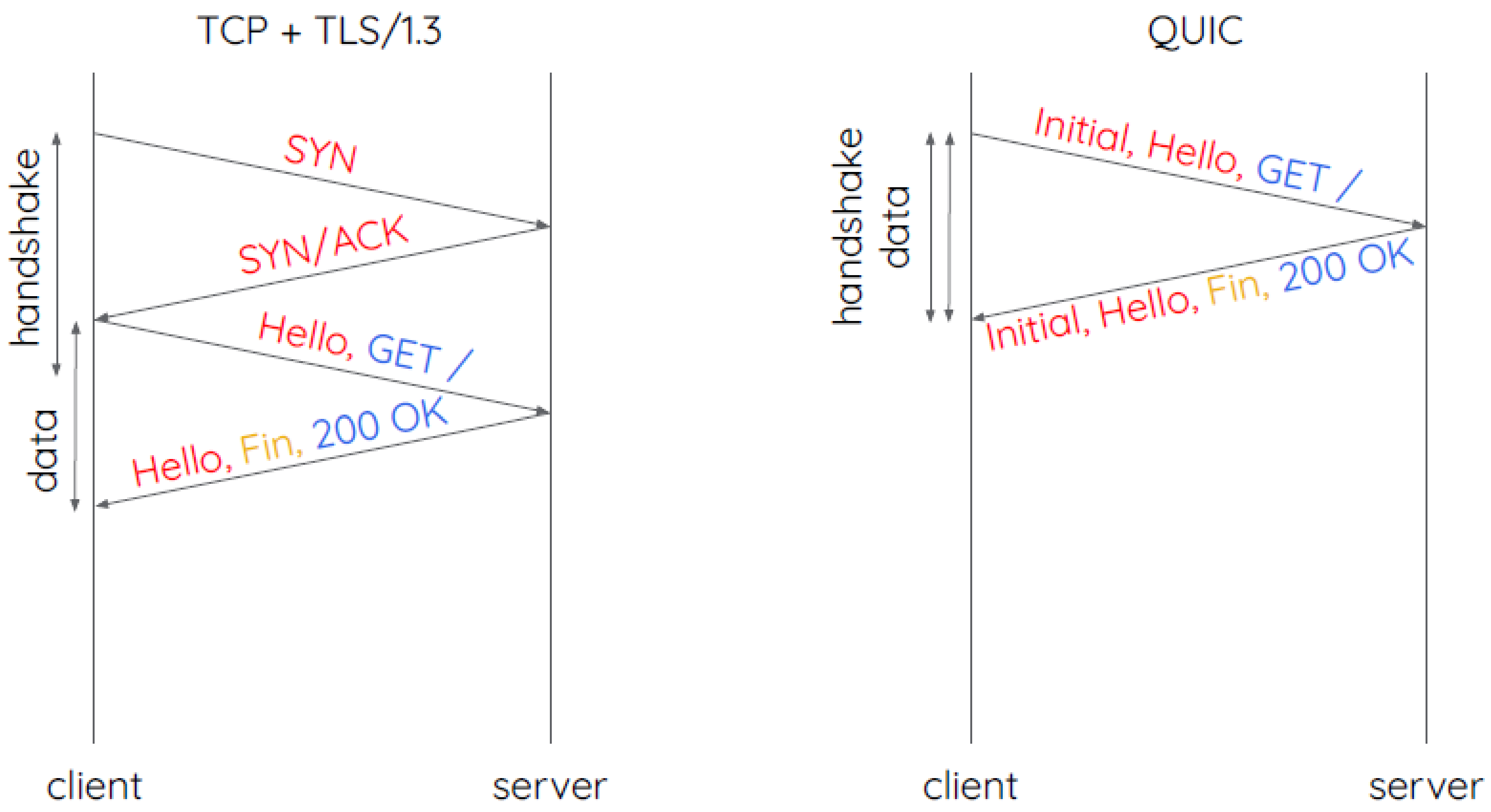
TCP (reliability, congestion control state) + TLS (authentication, crypto state)

- 2 serial handshakes

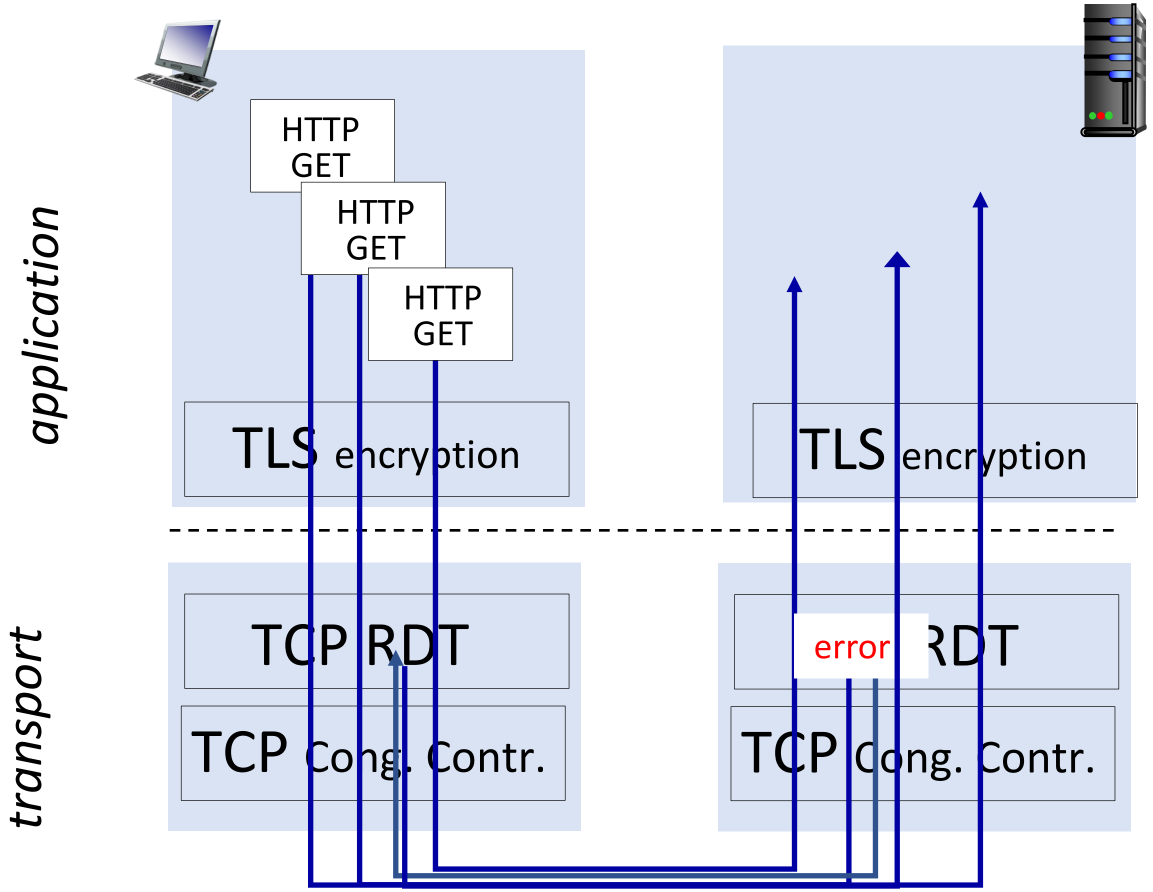
QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

Subsequent Connection to the same server



QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

QUIC Status

- In May 2021, the IETF standardized QUIC in RFC 9000.

Implementations:

Apple, Facebook, Fastly, Firefox, F5, Google, Microsoft ...

Server deployments have been going on for a while

Akamai, Cloudflare, Facebook, Fastly, Google ...

Clients are at different stages of deployment

Chrome, Firefox, Edge, Safari

iOS, MacOS

Source: SIGCOMM 2020, QUIC Tutorial, https://conferences.sigcomm.org/sigcomm/2020/tutorial-quic.html