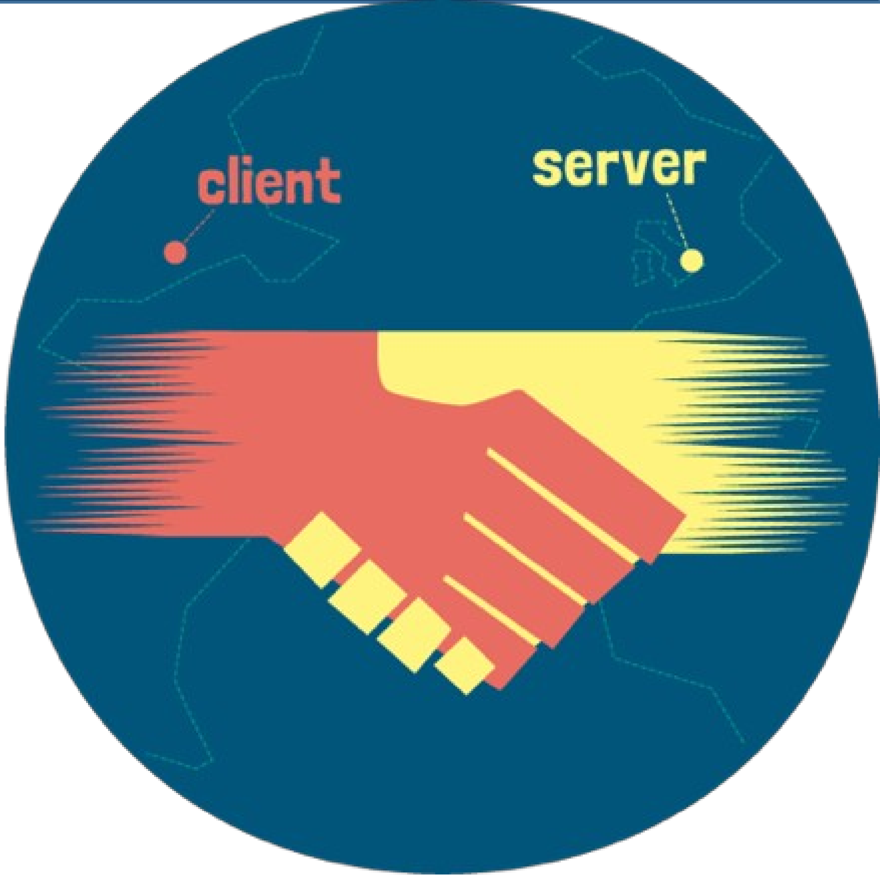
CS 31006: Computer Networks – The Internet Transport Protocols



Department of Computer Science and Engineering

INDIAN INSTITUTE OF TECHNOLOGY KHARAGPUR

Rajat Subhra Chakraborty
rschakraborty@cse.iitkgp.ac.in

Sandip Chakraborty
sandipc@cse.iitkgp.ac.in

Transmission Control Protocol (TCP)

- TCP was specifically designed to provide a reliable, end-to-end byte stream over an unreliable **internetwork**.

- **Internetwork** – different parts may have widely different topologies, bandwidths, delays, packet sizes and other parameters

- **TCP** dynamically adapts to properties of the internetwork and is robust in the face of many kinds of failures.

- RFC 793 (September 1981) – Base protocol
  - RFC 1122 (clarifications and bug fixes), RFC 1323 (High performance), RFC 2018 (SACK), RFC 2581 (Congestion Control), RFC

TCP Service Model

- Uses **Sockets** to define an end-to-end connection (Source IP, Source Port, Source Initial Sequence Number, Destination IP, Destination Port, Destination Initial Sequence Number)

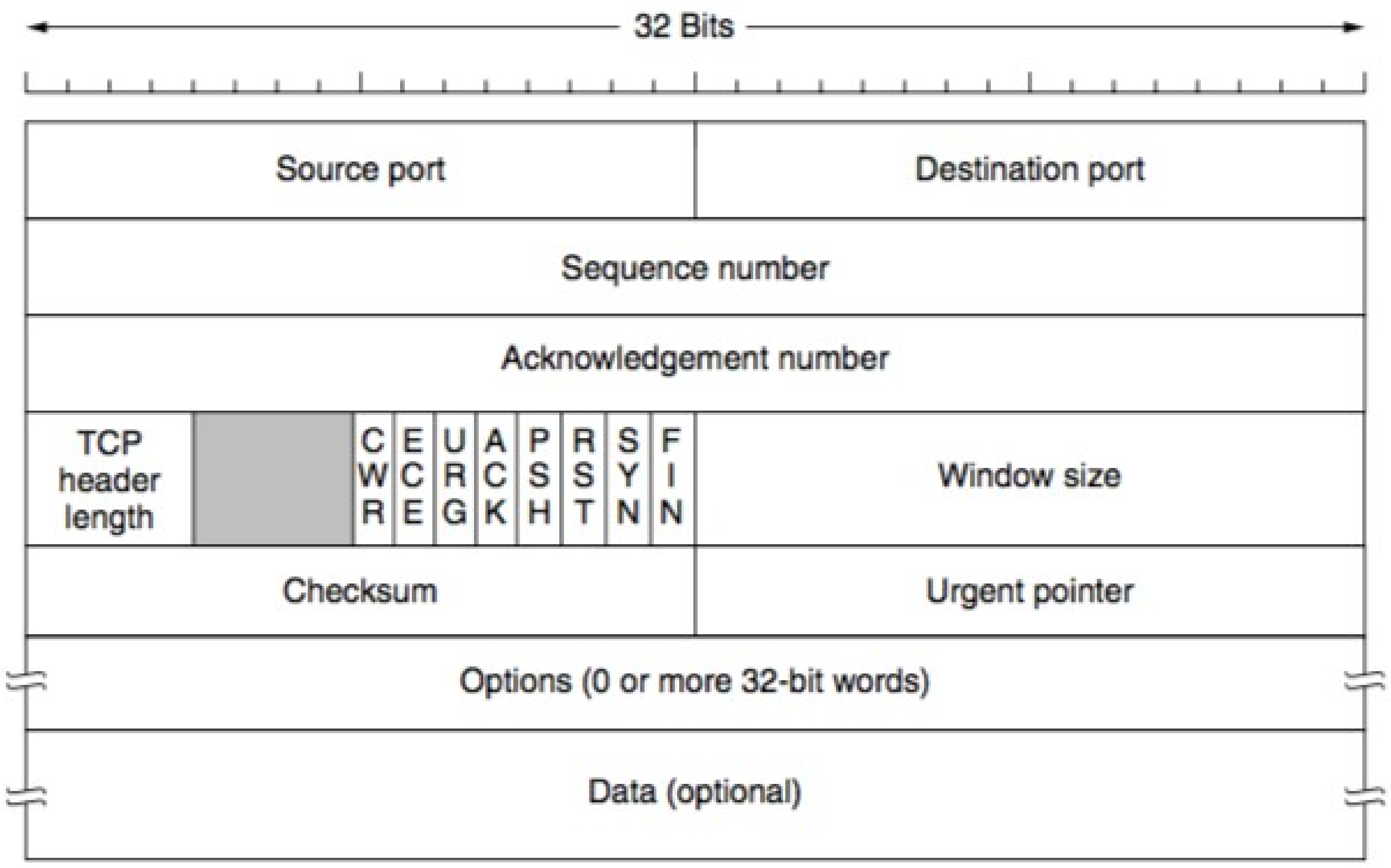
- **Unix Model of Socket Implementation:**
  - A single daemon process, called **Internet Daemon (inetd)** runs all the times at different **well known ports**, and wait for the first incoming connection
  - When a first incoming connection comes, *inetd* forks a new process and starts the corresponding daemon (for example *httpd* at port 80, *ftpd* at port 21 etc.)

- All TCP connections are full-diplex and point-to-point. TCP does not support multicasting or broadcasting.

TCP Service Model

- A TCP connection is a **byte stream**, not a message stream
- Message boundaries are not preserved end-to-end
- Example:
  - The sending process does four 512 byte writes to a TCP stream – for `write()` call to the TCP socket
  - These data may be delivered as – four 512 byte chunks, two 1024 byte chunks, one 2048 byte chunk or some other way
  - There is no way for the receiver to detect the unit(s) in which the data were written by the sending process.

The TCP Protocol – The Header



Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

TCP Sequence Number and Acknowledgement Number

- 32 bits sequence number and acknowledgement number

- Every byte on a TCP connection has its own 32 bit sequence number – a **byte stream** oriented connection

- TCP uses sliding window based flow control – the acknowledgement number contains next expected byte in order, which acknowledges the **cumulative bytes** that has been received by the receiver.
  - ACK number 31245 means that the receiver has correctly received up to 31244 bytes and expecting for byte 31245

TCP Segments

- The sending and receiving TCP entities exchange data in the form of **segments**.
- A TCP segment consists of a fixed 20 byte header (plus an optional part) followed by zero or more data bytes.
- TCP can accumulate data from several `write()` calls into one segment, or split data from one `write()` into multiple segments
- A segment size is restricted by two parameters
  - IP Payload (65515 bytes)
  - Maximum Transmission Unit (MTU) of the link

How a TCP Segment is Created

- `Write()` calls from the applications write data to the TCP sender buffer.
- sender maintains a dynamic window size based on the flow and congestion control algorithm

- Modern implementations of TCP uses **path MTU discovery** to determine the MTU of the end-to-end path (uses ICMP protocol), and sets up the **Maximum Segment Size** (**MSS**) during connection establishment
  - May depend on other parameters (buffer implementation).

- Check the sender window after receiving an ACK. If the window size is less than MSS, construct a single segment;

Challenges in TCP Design

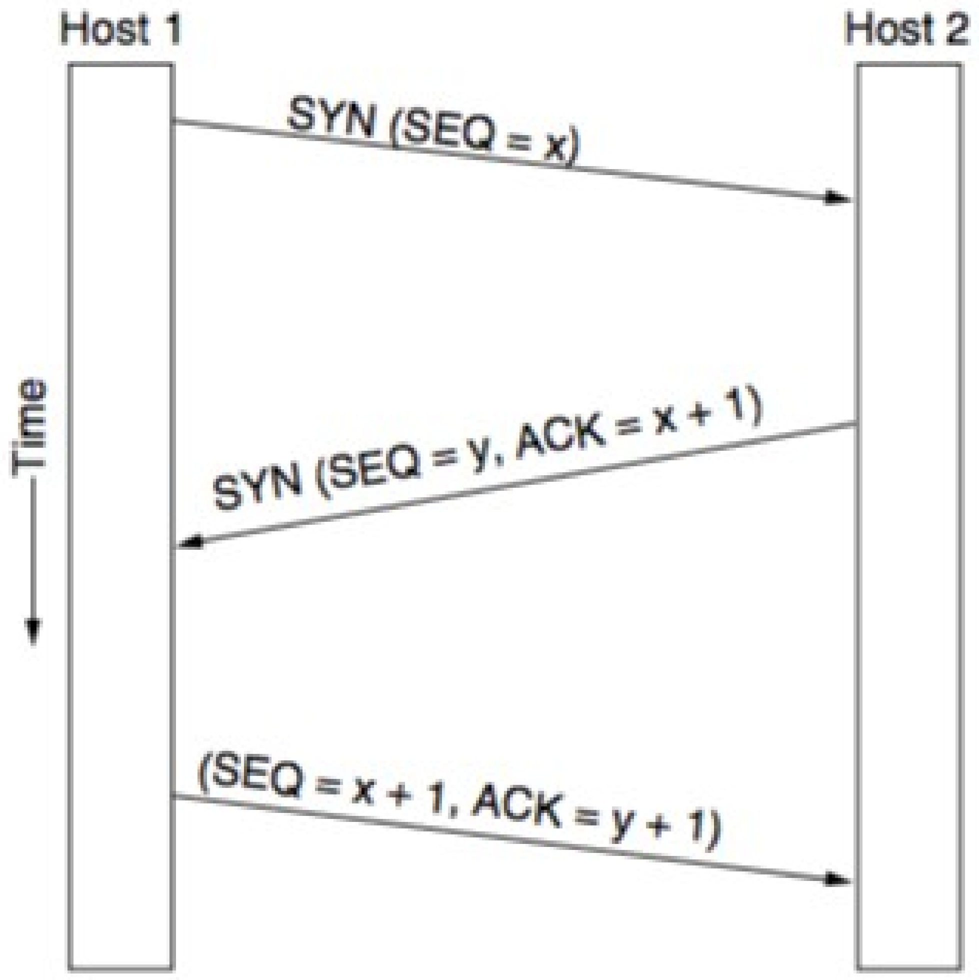
- Segments are constructed dynamically, so retransmissions do not guarantee the retransmission of the same data segment – a retransmission may contain additional data or less data

- Segments may arrive out-of-order. TCP receiver should handle out-of-order segments in a proper way, so that data wastage is minimized.

Window Size field in the TCP Segment Header

- Flow control in TCP is handled using a variable sized sliding window.
- The *window size* field tells how many bytes the receiver can receive based on the current free size at its buffer space.
- **What is meant by window size 0?**
- TCP Acknowledgement – combination of acknowledgement number and window size

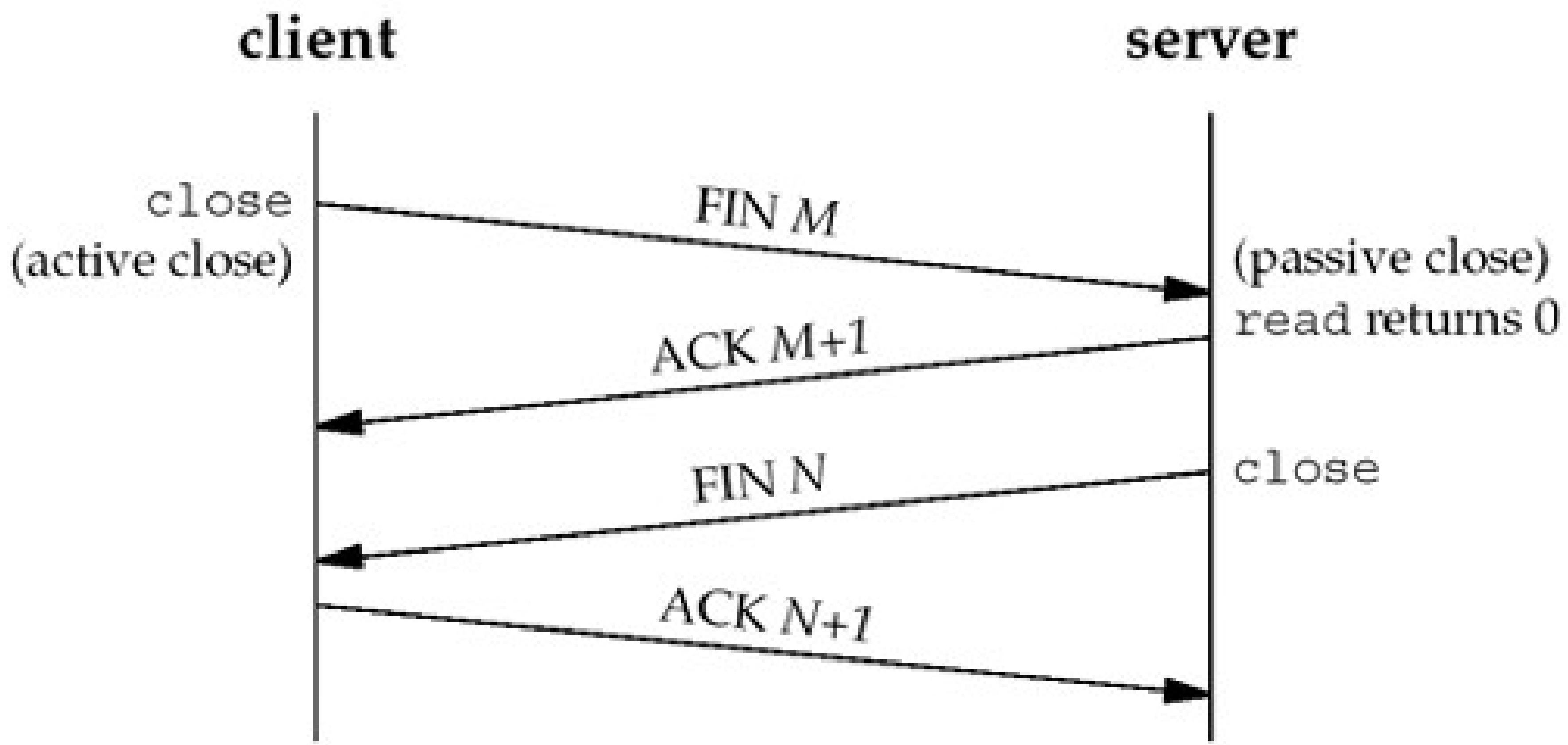
TCP Connection Establishment



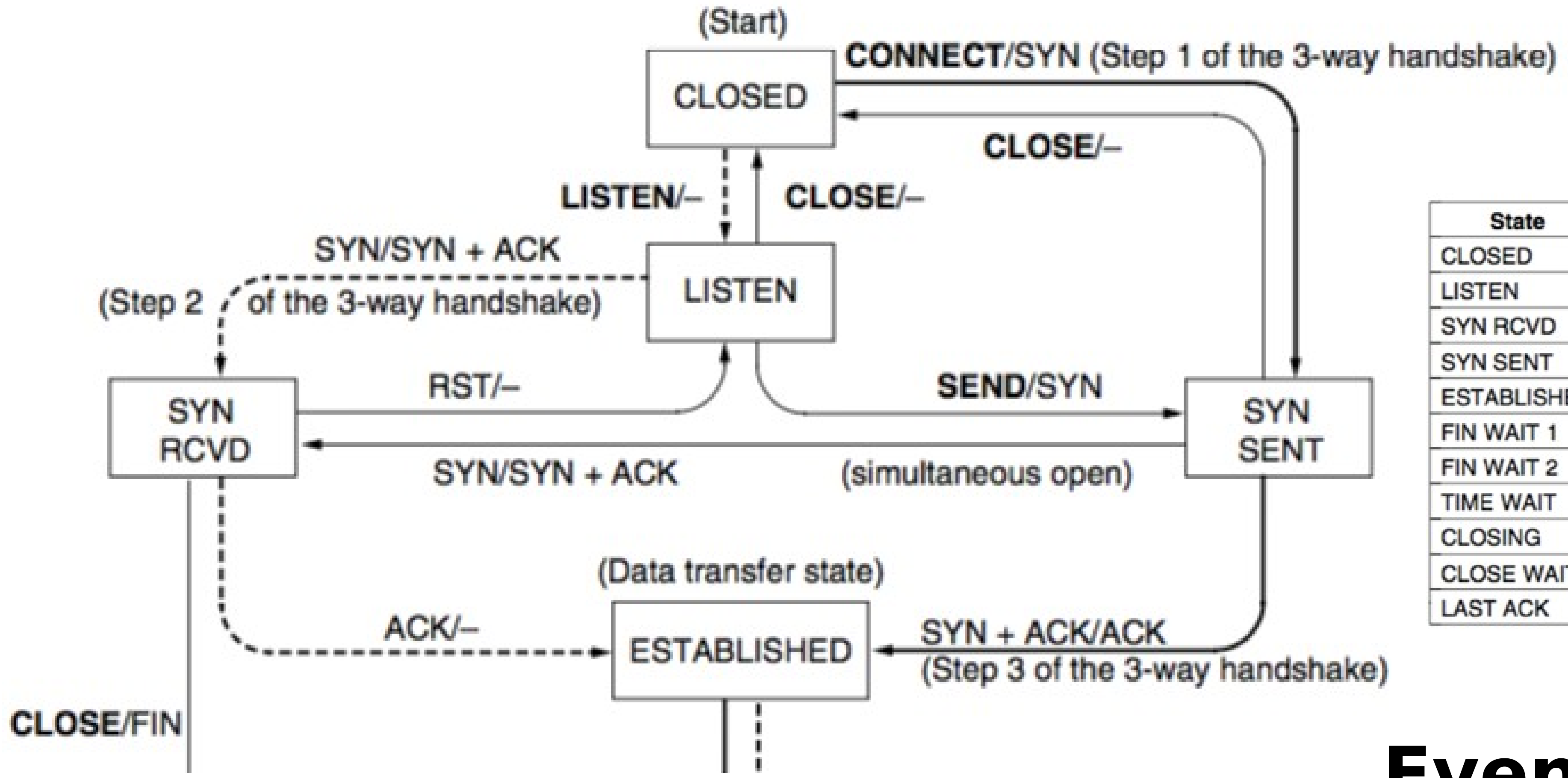
- **How to choose the initial sequence number?**
  - Protect delayed duplicates, do now generate the initial sequence number for every connection from 0
  - Original implementation of TCP used a clock based approach, the clock ticked every 4 microseconds, the value of the clock cycles from 0 to 2^{32}-1. The value of the clock gives the initial sequence number

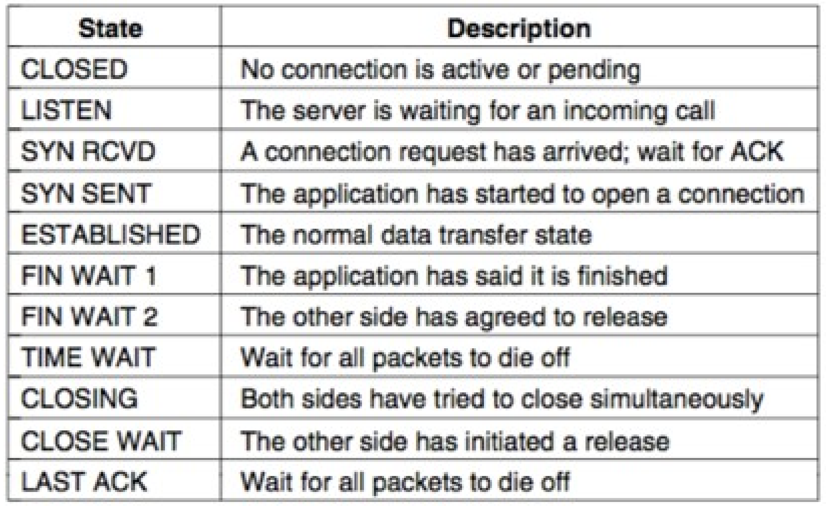
- **TCP SYN flood attack**
  - Solution: Use cryptographic function to generate sequence numbers

TCP Connection Release



TCP State Transition Diagram – Connection Modeling



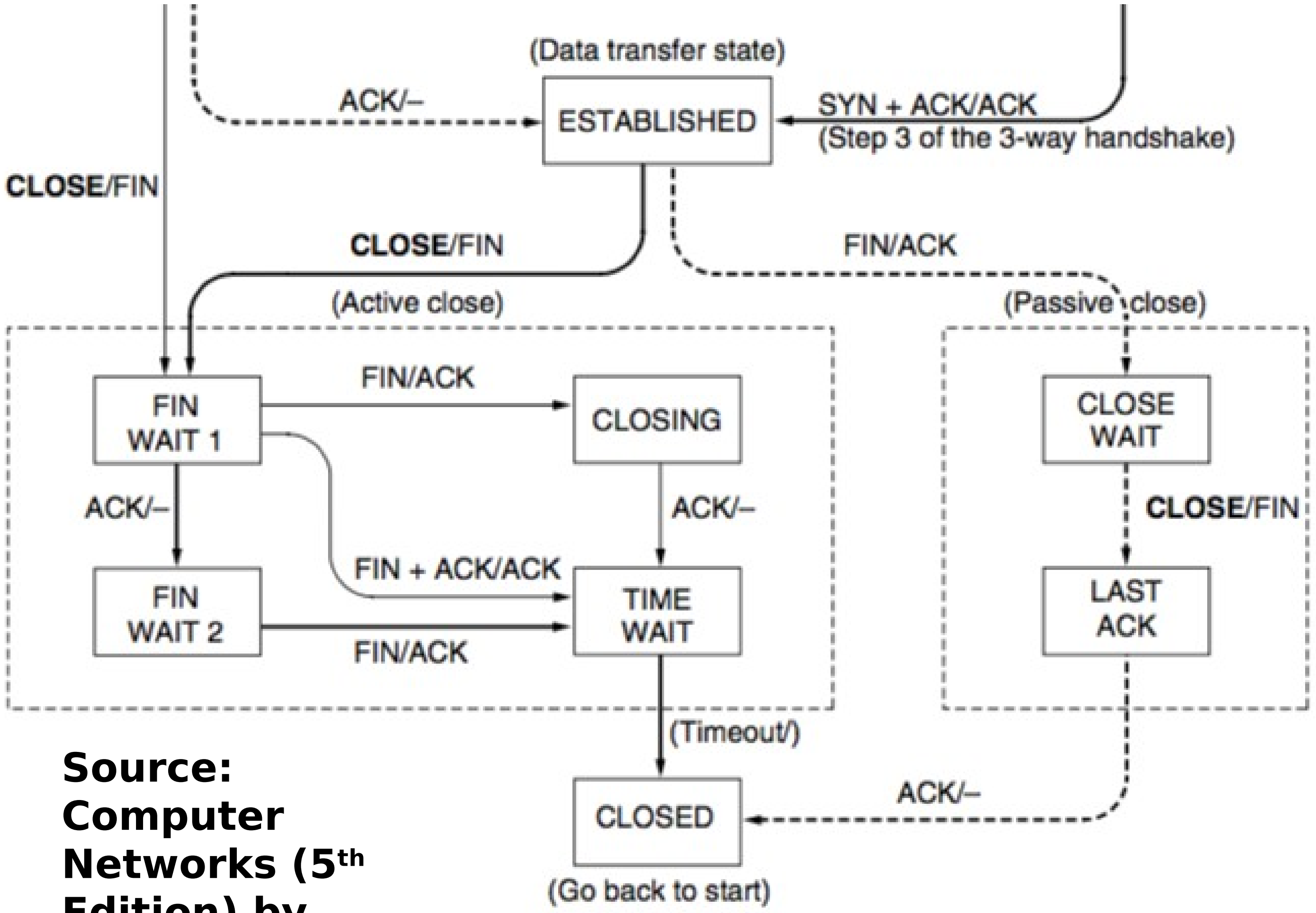


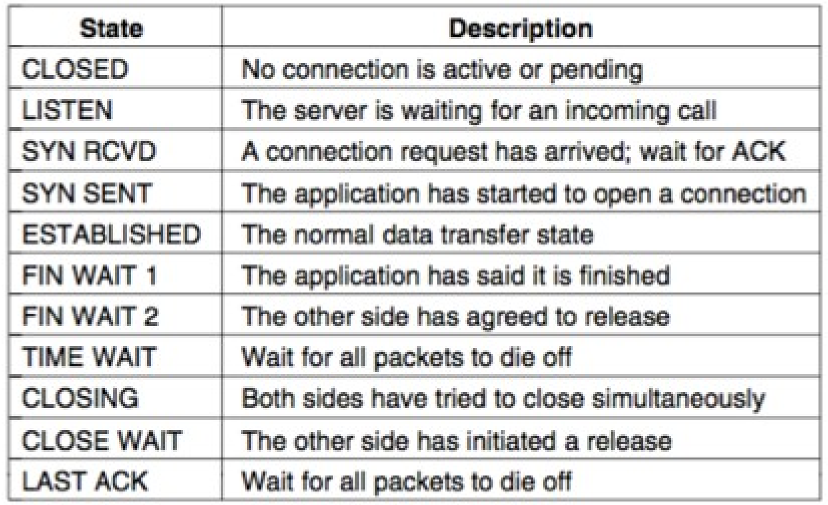
State	Description
CLOSED	No connection is active or pending
LISTEN	The server is waiting for an incoming call
SYN RCVD	A connection request has arrived; wait for ACK
SYN SENT	The application has started to open a connection
ESTABLISHED	The normal data transfer state
FIN WAIT 1	The application has said it is finished
FIN WAIT 2	The other side has agreed to release
TIME WAIT	Wait for all packets to die off
CLOSING	Both sides have tried to close simultaneously
CLOSE WAIT	The other side has initiated a release
LAST ACK	Wait for all packets to die off

Source: Computer Networks (5th Edition) by

Event/Action
Dashed: Server
Solid: Client

TCP State Transition Diagram – Connection Modeling



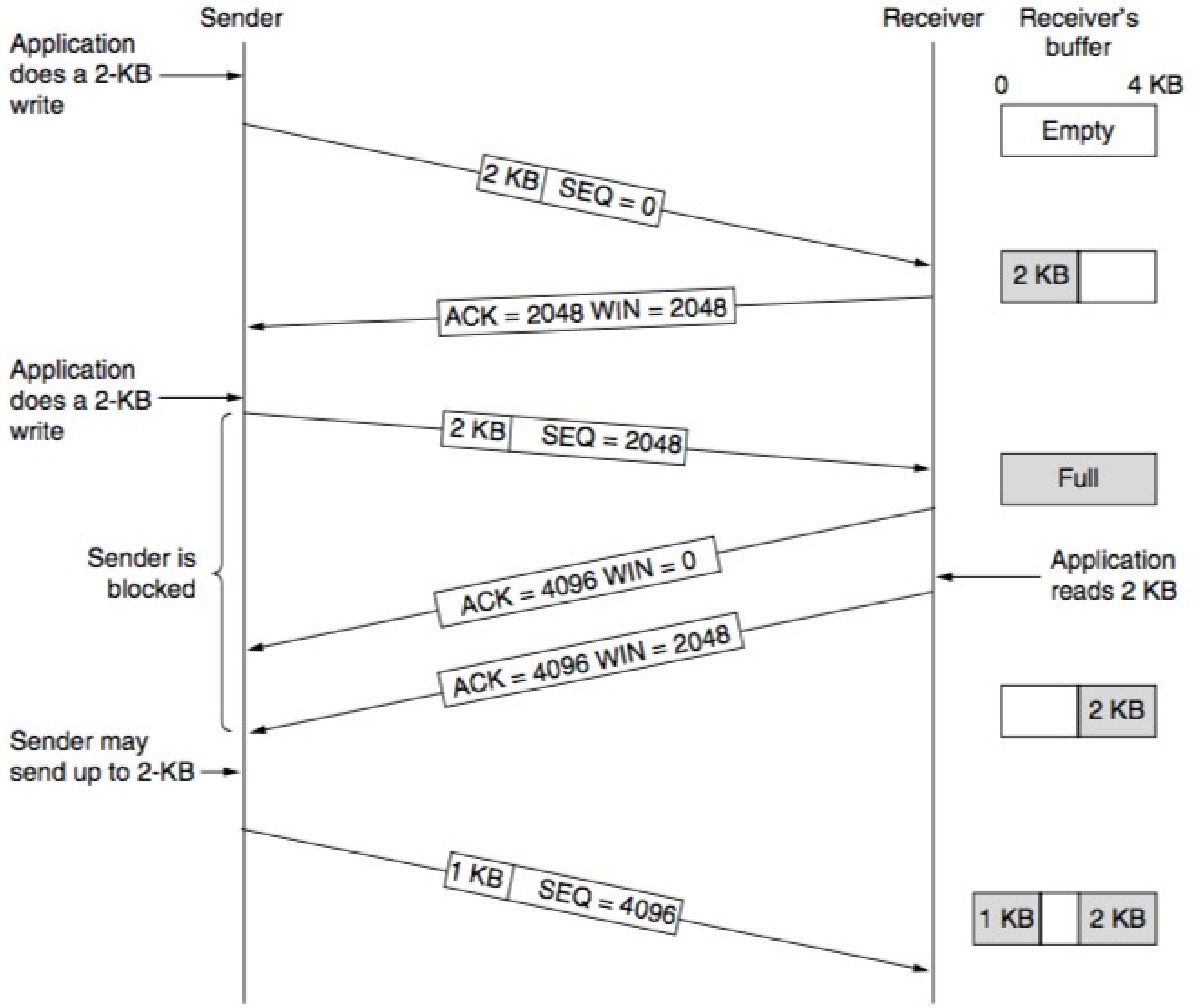


State	Description
CLOSED	No connection is active or pending
LISTEN	The server is waiting for an incoming call
SYN RCVD	A connection request has arrived; wait for ACK
SYN SENT	The application has started to open a connection
ESTABLISHED	The normal data transfer state
FIN WAIT 1	The application has said it is finished
FIN WAIT 2	The other side has agreed to release
TIME WAIT	Wait for all packets to die off
CLOSING	Both sides have tried to close simultaneously
CLOSE WAIT	The other side has initiated a release
LAST ACK	Wait for all packets to die off

Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

Event/Action
Dashed: Server
Solid: Client

TCP Sliding Window



Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

Delayed Acknowledgements

- Consider a telnet connection, that reacts on every keystroke.
- In the worst case, whenever a character arrives at the sending TCP entity, TCP creates a 21 byte TCP segment, 20 bytes of header and 1 byte of data. For this segment, another ACK and window update is sent when the application reads that 1 byte. This results in a huge wastage of bandwidth.
- **Delayed acknowledgements:** Delay acknowledgement and window updates for up to 500 msec in the hope of receiving few more data packets within that interval.
- **However, the sender can still send multiple short data**

Nagle’s Algorithm

- When data come into the sender in small pieces, just send the first piece and buffer all the rest until the first piece is acknowledged.

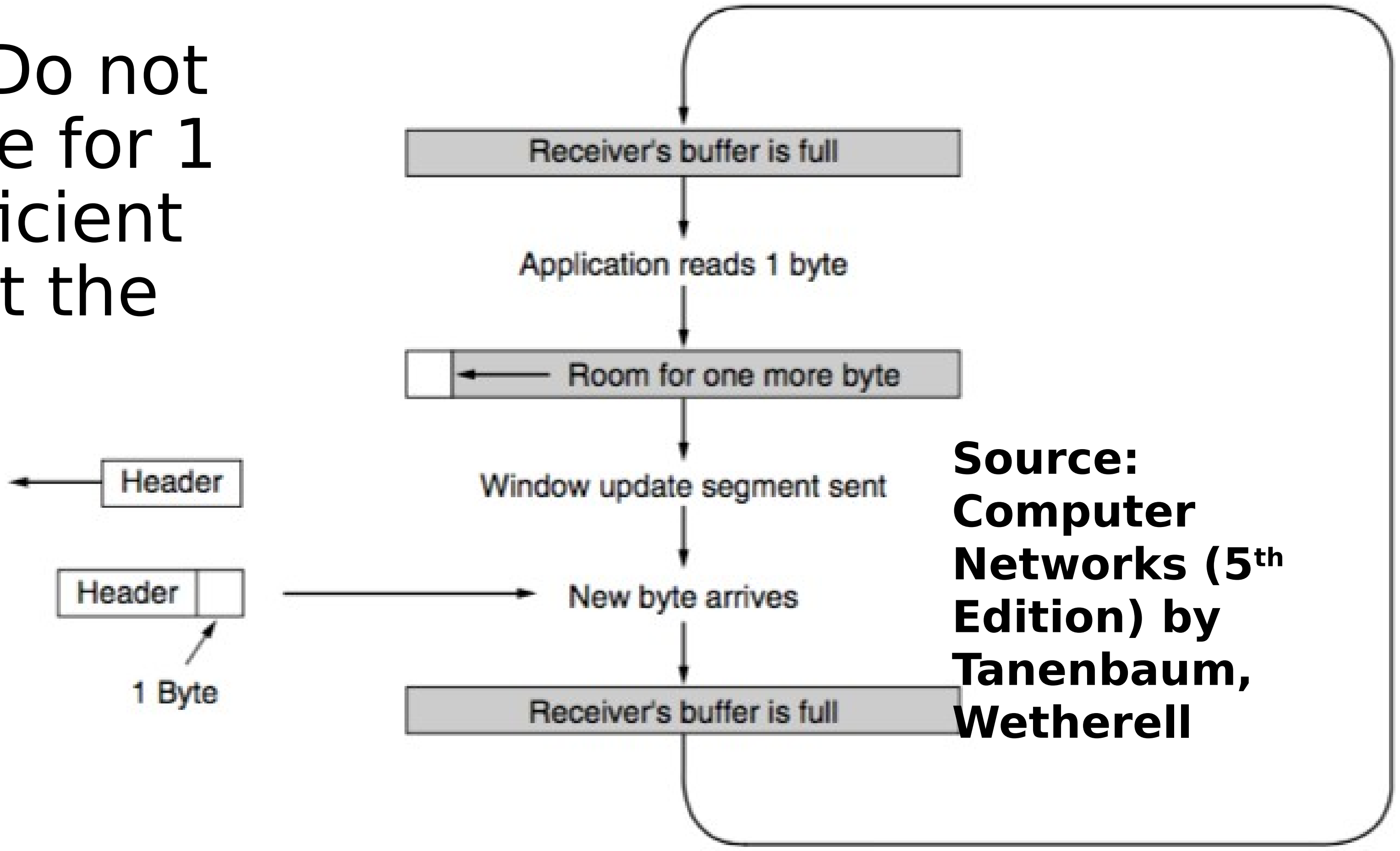
- Then send all buffered data in one TCP segment and start buffering again until the next segment is acknowledged.
  - **Only one short packet can be outstanding at any time.**

- **Do we want Nagle’s Algorithm all the time?**

- **Nagle’s Algorithm and Delayed Acknowledgement**
  - Receiver waits for data and sender waits for acknowledgement – results in starvation

Silly Window Syndrome

- Data are passed to the sending TCP entity in large blocks, but an interactive application on the receiver side reads data only 1 byte at a time.
- **Clark's solution:** Do not send window update for 1 byte. Wait until sufficient space is available at the receiver buffer.



Source: Computer Networks (5th Edition) by Tanenbaum, Wetherell

Handling Short Segments – Sender and Receiver Together

- Nagle’s algorithm and Clark’s solution to silly window syndrome are **complementary**
- **Nagle’s algorithm:** Solve the problem caused by the sending application delivering data to TCP a byte at a time
- **Clark’s solution:** Receiving application fetching the data up from TCP a byte at a time
- Exception: The PSH flag is used to inform the sender to create a segment immediately without waiting for more data

Handling Out of Order in TCP

- TCP buffers out of order segments and forward a duplicate acknowledgement to the sender.
- **Acknowledgement in TCP – Cumulative acknowledgement**
- Receiver has received bytes 0, 1, 2, _, 4, 5, 6, 7
  - TCP sends a cumulative acknowledgement with ACK number 3, acknowledging everything up to byte 2
  - Once 4 is received, a duplicate ACK with ACK number 3 (next expected byte) is forwarded – **triggers congestion control**
  - After timeout, sender retransmits byte 3
  - Once byte 3 is received, it can send another cumulative ACK with ACK number 8 (next ecpected byte)

TCP Timer Management

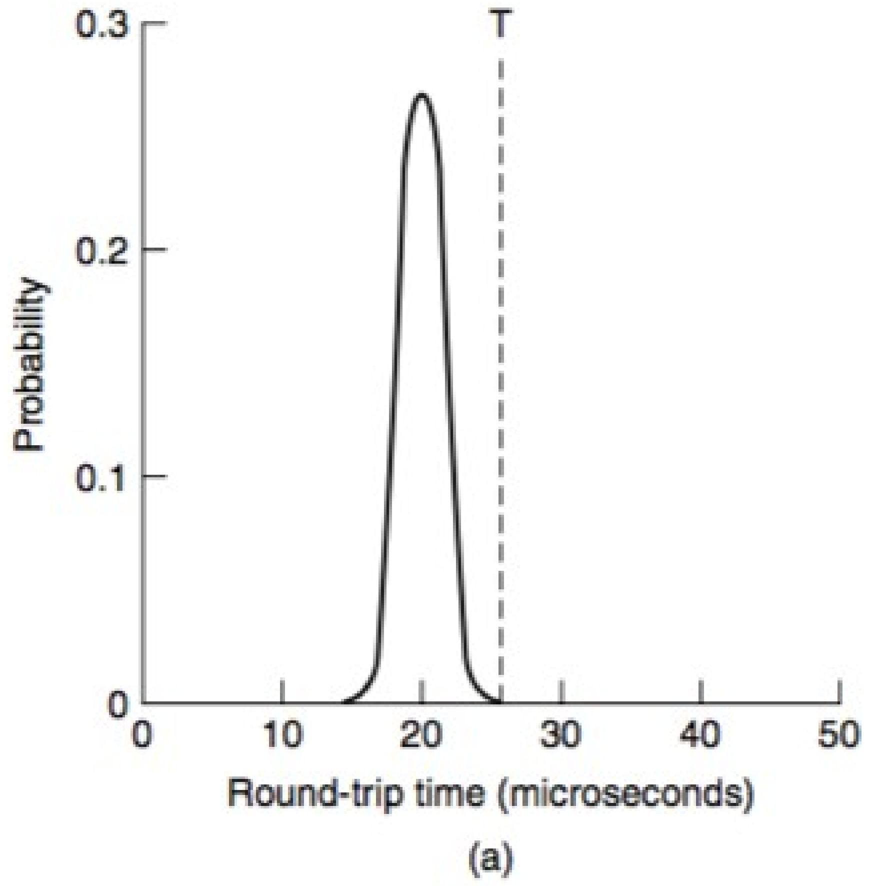
- **TCP Retransmission Timeout (RTO)**: When a segment is sent, a retransmission timer is started
  - **If the segment is acknowledged before the timer expires, the timer is stopped**
  - **If the timer expires before the acknowledgement comes, the segment is retransmitted**

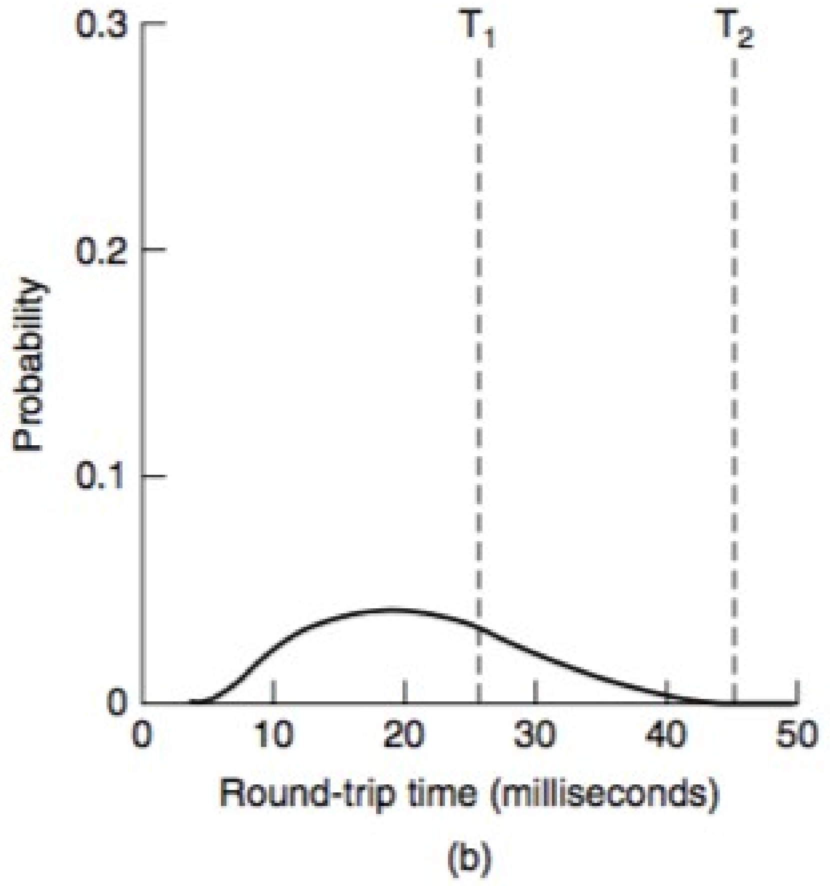
- **What can be an ideal value of RTO ?**

- **Possible solution**: Estimate RTT, and RTO is some positive multiples of RTT

- **RTT estimation is difficult for transport layer – why?**

RTT at Data Link Layer vs RTT at Transport Layer





Data Link Layer **Transport Layer**

RTT Estimation at the Transport Layer

- Use a dynamic algorithm that constantly adapts the timeout interval, based on continuous measurements of network performance.

- **Jacobson's algorithm (1988) - used in TCP**
  - For each connection, TCP maintains a variable, ***SRTT (smoothed Round Trip Time)*** – best current estimate of the round trip time to the destination
  - When a segment is sent, a timer is started (both to see how long the acknowledgement takes and also to trigger a retransmission if it takes too long)
  - If the ACK gets back – measure the time (say, *R*)
  - Update SRTT as follows

    (Exponentially Weighted Moving Average – EWMA)
  - α is a smoothing factor that determines how quickly the old values

Problem with EWMA

- Even given a good value of SRTT, choosing a suitable RTO is nontrivial.

- Initial implementation of TCP used RTO = 2SRTT

- Experience showed that a constant value was too inflexible, because it failed to response when the **variance went up (RTT fluctuation is high) – happens normally at high load**

- **Consider variance of RTT during RTO estimation.**

RTO Estimation

- Update RTT variation (*RTTVAR*) as follows.

- Typically β = ¾

- RTO is estimated as follows,

- **Why 4 ?**
  - **Somehow arbitrary**
  - **Jacobson's paper is full of clever tricks – use integer addition, subtraction and shift – computation is lightweight**

Karn’s Algorithm

- How will you get the RTT estimation, when a segment is lost and retransmitted again?

- **Karn’s algorithm:**
  - Do not update estimates on any segments that has been retransmitted
  - The timeout is doubled each successive retransmission until the segments gets through the first time

Other TCP Timers

- **Persistent TCP Timer:** Avoid deadlock when receiver buffer is announced as zero
  - **After the timer goes off, sender forwards a probe packet to the receiver to get the updated window size**

- **Keepalive Timer:** Close the connection when a connection has been idle for a long duration

- **TCP TIME_WAIT:** Wait before closing a connection – twice the packet lifetime

TCP Congestion Control

- Based on implementation of AIMD using a window and with packet loss as the binary signal

- TCP maintains a **Congestion Window (CWnd) –** number of bytes the sender may have in the network at any time

- **Sending Rate = Congestion Window / RTT**

- **Sender Window (SWnd) = Min (CWnd, RWnd)**

- RWnd – Receiver advertised window size

1986 Congestion Collapse

- In 1986, the growing popularity of Internet led to the first occurrence of congestion collapse – a prolonged period during which goodput dropped precipitously (more than a factor of 100)

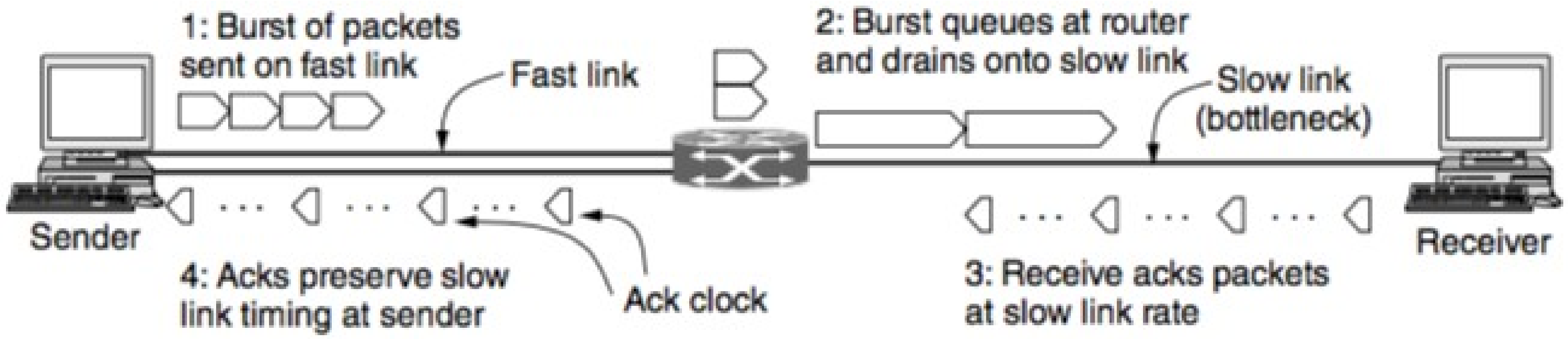
- Early TCP Congestion Control algorithm – Effort by Van Jancobson (1988)

- **Challenged for Jacobson** – Implement congestion control without making much change in the protocol (made it instantly deployable)

- **Packet loss is a suitable signal for congestion – use**

Adjust CWnd based on AIMD

- **One of the most interesting ideas – use ACK for clocking**


1: Burst of packets sent on fast link
Fast link
2: Burst queues at router and drains onto slow link
Slow link (bottleneck)
Sender
Receiver
4: Acks preserve slow link timing at sender
Ack clock
3: Receive acks packets at slow link rate

- **ACK returns to the sender at about the rate that packets can be sent over the slowest link in the path.**

- **Trigger CWnd adjustment based on the rate at which ACK are received.**

Increase Rate Exponentially at the Beginning – The Slow Start

- AIMD rule will take a very long time to reach a good operating point on fast networks if the CWnd is started from a small size.

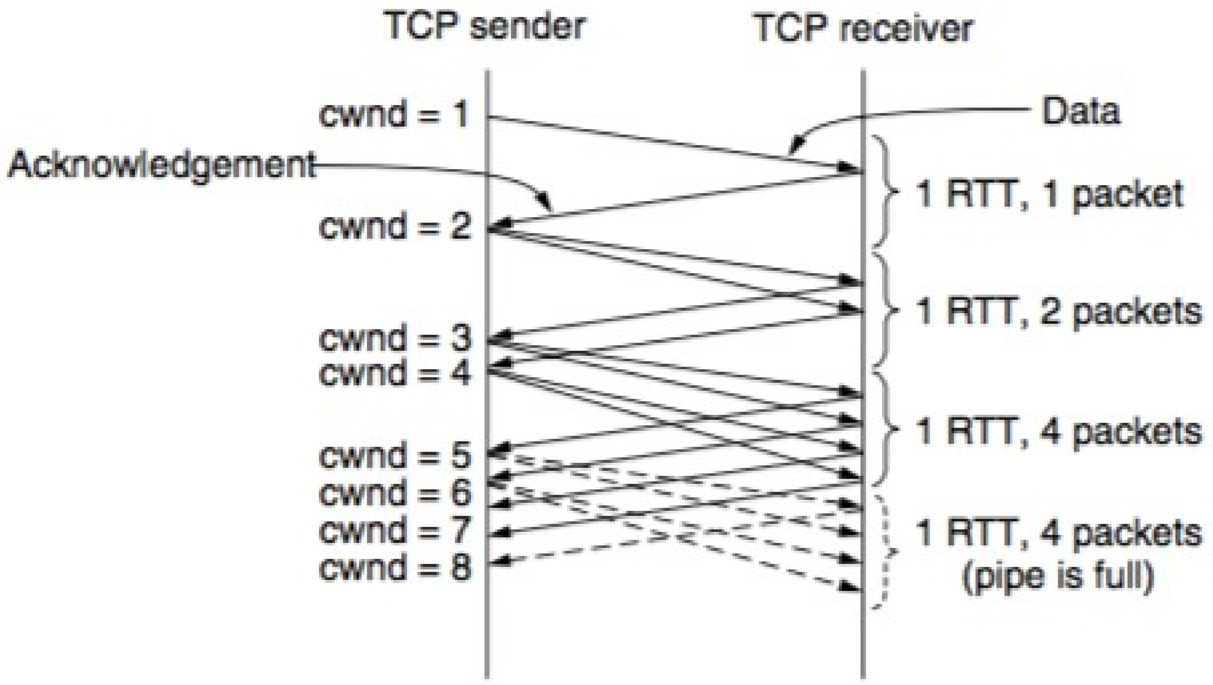
- A 10 Mbps link with 100 ms RTT
  - Appropriate CWnd = BDP = 1 Mbit
  - 1250 byte packets -> 100 packets to reach BDP
  - CWnd starts at 1 packet, and increased 1 packet at every RTT
  - 100 RTTs are required 10 sec before the connection reaches to a moderate rate

- **Slow Start - Exponential increase of rate to avoid slow convergence**
  - **Rate is not slow at all !** ☹

TCP Slow Start

- Every ACK segment allows two more segments to be sent

- For each segment that is acknowledged before the retransmission timer goes off, the sender adds one segment’s worth of bytes to th



Slow Start Threshold

- Slow start causes exponential growth, eventually it will send too many packets into the network too quickly.

- To keep slow start under control, the sender keeps a threshold for the connection called the **slow start threshold (ssthresh).**

- Initially ssthresh is set to BDP (or arbitrarily high), the maximum that a flow can push to the network.

- Whenever a packet loss is detected by a RTO, the ssthresh is set to be half of the congestion window

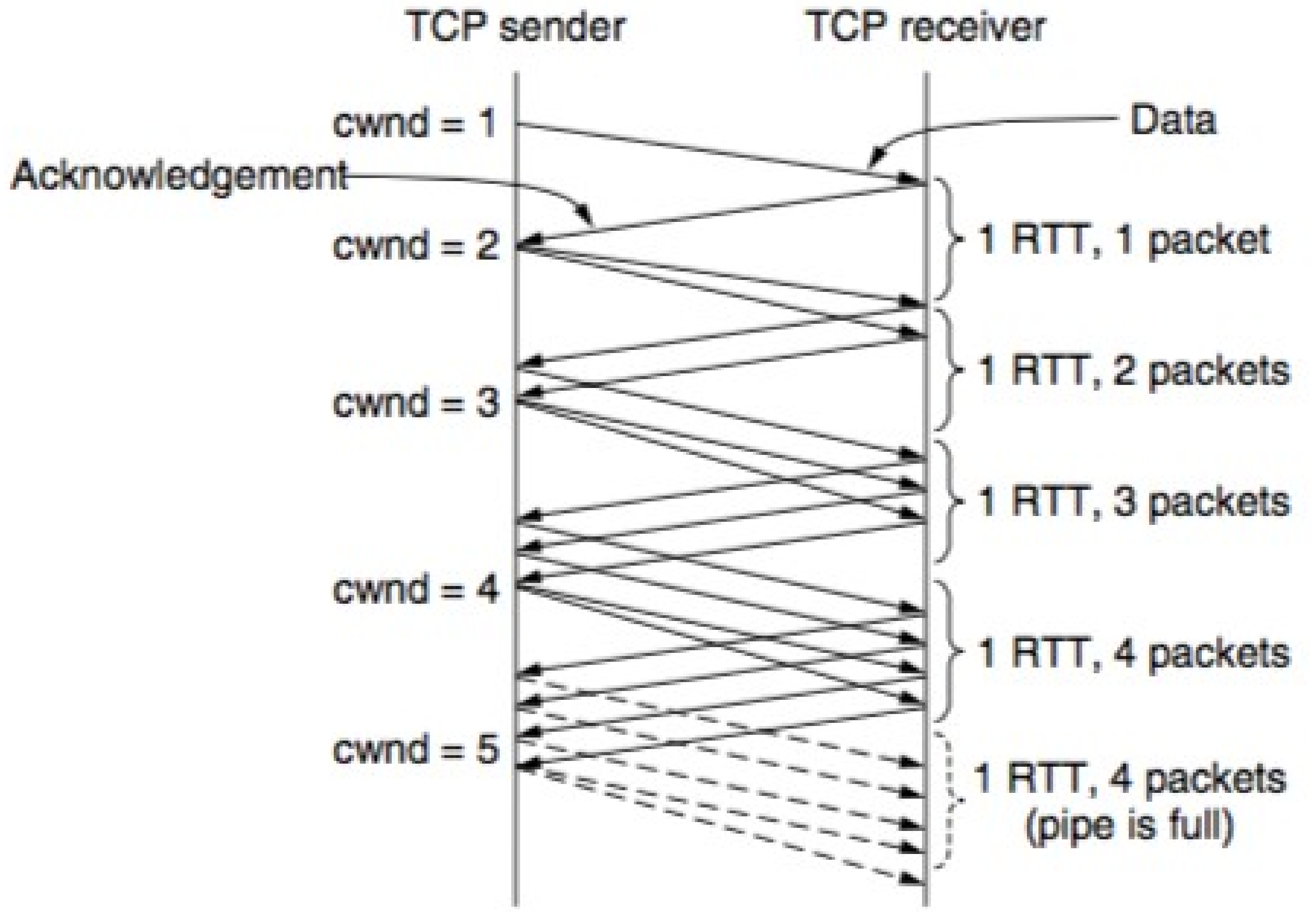
Additive Increase (Congestion Avoidance)

- Whenever ssthresh is crossed, TCP switches from slow start to additive increase.

- Usually implemented with an partial increase for every segment that is acknowledged, rather than an increase of one segment per RTT.

- A common approximation is to increase Cwnd for additive increase as follows:

Additive Increase – Packet Wise Approximation



Triggering an Congestion

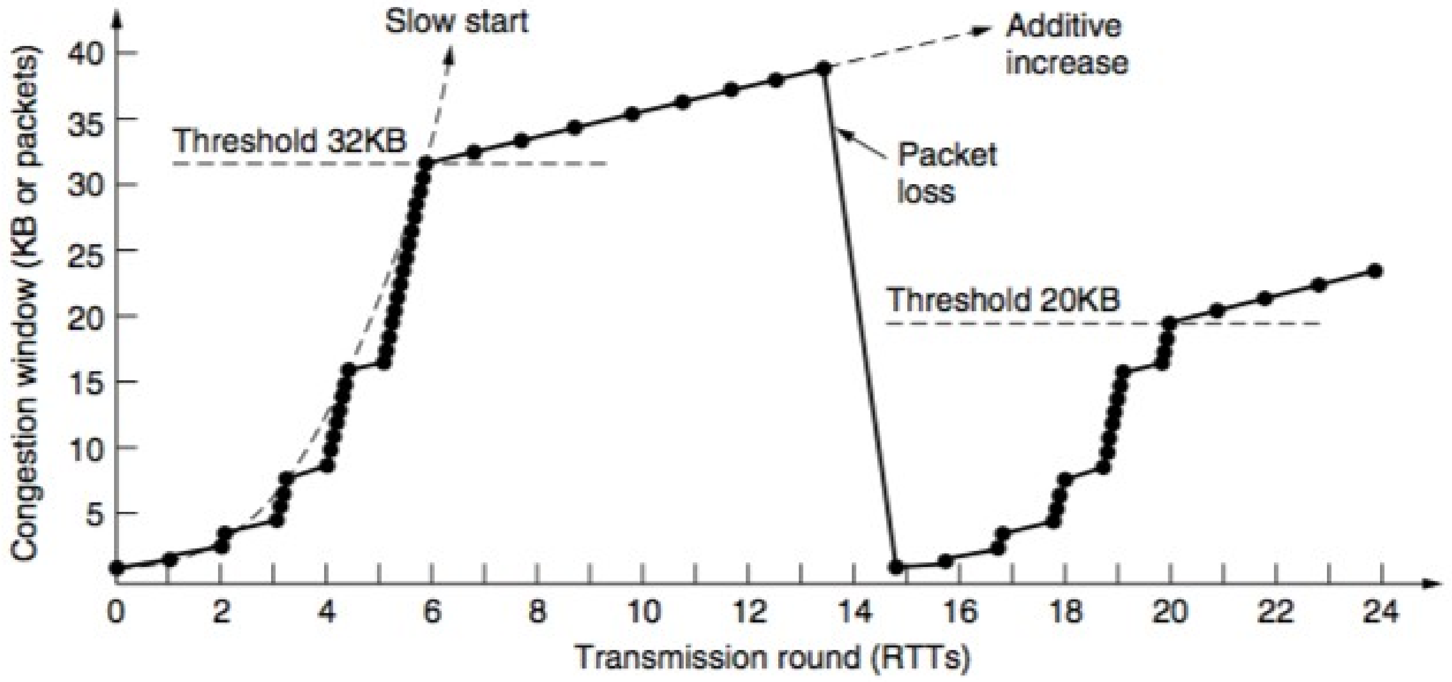
- Two ways to trigger a congestion notification in TCP – (1) RTO, (2) Duplicate ACK

- **RTO**: A sure indication of congestion, however time consuming

- **Duplicate ACK:** Receiver sends a duplicate ACK when it receives out of order segment
  - **A loose way of indicating congestion**
  - **TCP arbitrarily assumes that THREE duplicate ACKs (DUPACKs) imply that a packet has been lost – triggers congestion control mechanism**
  - The identity of the lost packet can be inferred – **the very next packet in sequence**

Fast Retransmission – TCP Tohoe

- Use THREE DUPACK as the sign of congestion
- Once 3 DUPACKs have been received,
  - Retransmit the lost packet (**fast retransmission)** – takes one RTT
  - Set ssthresh as half of the current CWnd
  - Set CWnd to 1 MS



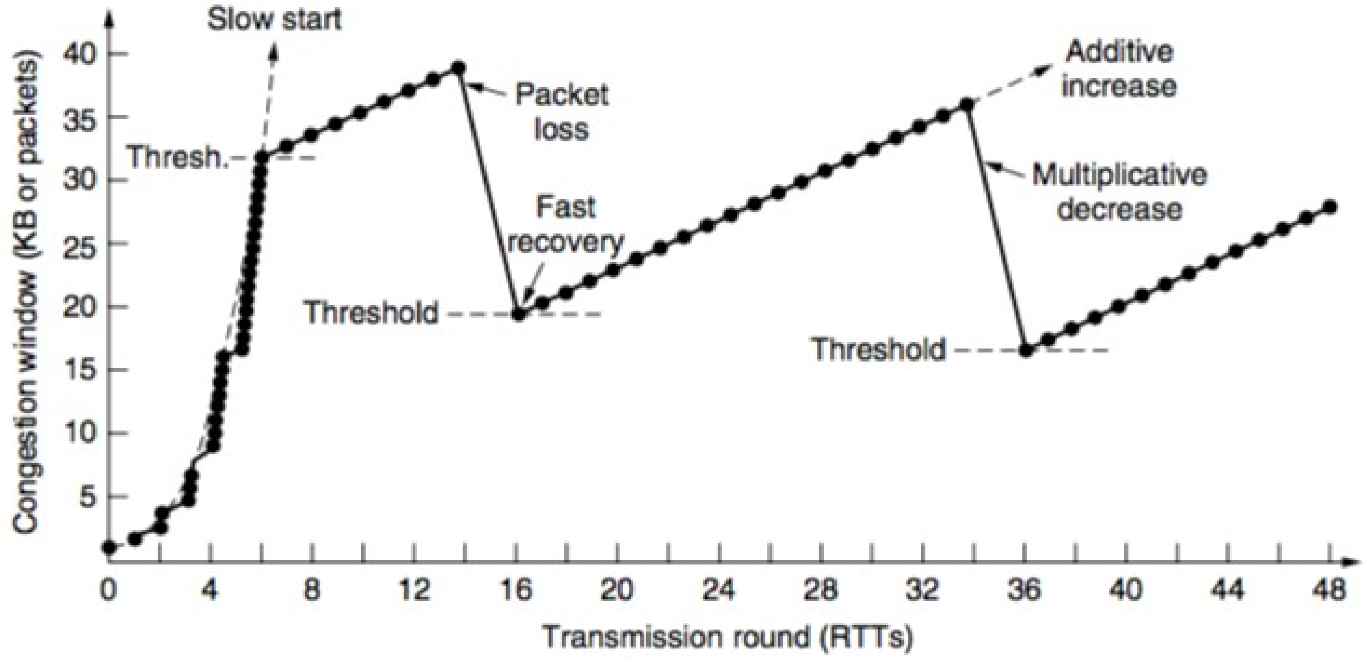
Fast Recovery – TCP Reno

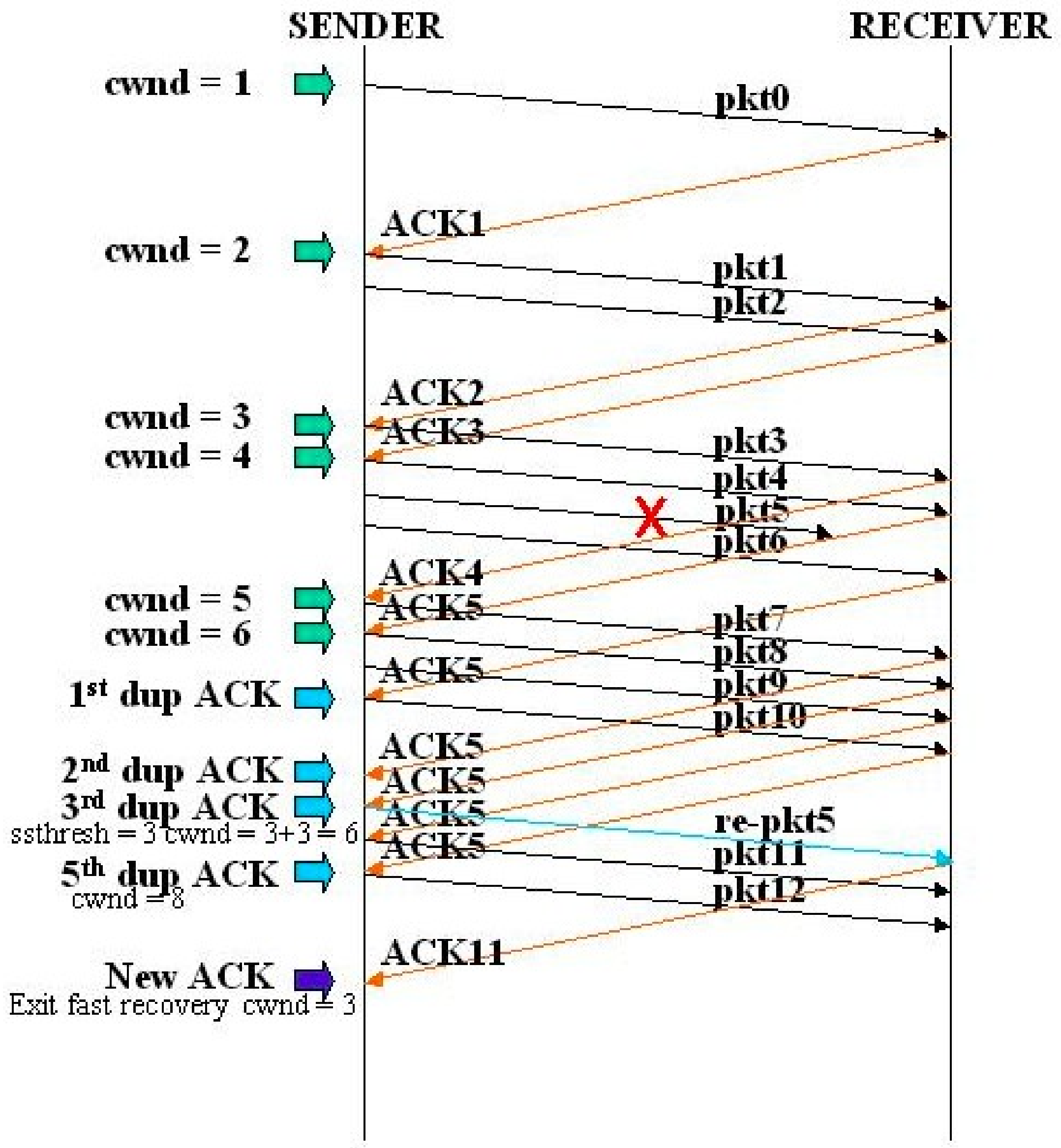
- **Once a congestion is detected through 3 DUPACKs, do TCP really need to set CWnd = 1 MSS ?**
- DUPACK means that **some segments are still flowing in the network** – a signal for temporary congestion, but not a prolonged one
- Immediately transmit the lost segment (**fast retransmit)**, then transmit additional segments based on the DUPACKs received **(fast recovery)**

Fast Recovery – TCP Reno

- **Fast recovery:**
  1. set ssthresh to one-half of the current congestion window. Retransmit the missing segment.
  2. set cwnd = ssthresh + 3.
  3. Each time another duplicate ACK arrives, set cwnd = cwnd + 1. Then, send a new data segment if allowed by the value of cwnd.
  4. Once receive a new ACK (an ACK which acknowledges all intermediate segments sent between the lost packet and the receipt of the first duplicate ACK), exit fast recovery. This causes setting cwnd to ssthresh (the ssthresh in step 1). Then, continue with linear increasing due to congestion avoidance algorithm.

Fast Recovery – TCP Reno

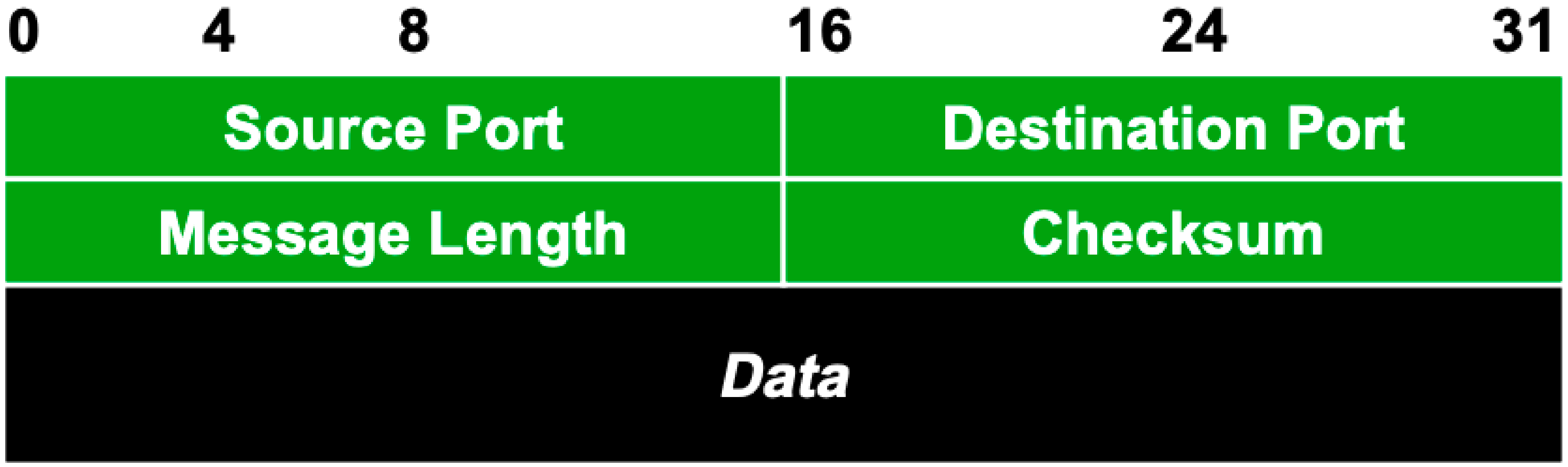




User Datagram Protocol (UDP)

- Basically a wrapper on top of IP
- Provides unreliable datagram service
  - Packets may be lost or delivered out of order
  - Users exchange datagrams (not streams)
  - Connectionless
  - Not buffered -- UDP accepts data and transmits immediately (no buffering before transmission)
  - Full duplex -- concurrent transfers can take place in both directions

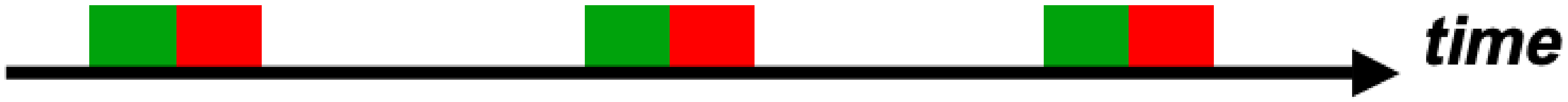
UDP Datagram Format



UDP versus TCP

- Choice of UDP versus TCP is based on
  - Functionality
  - Performance

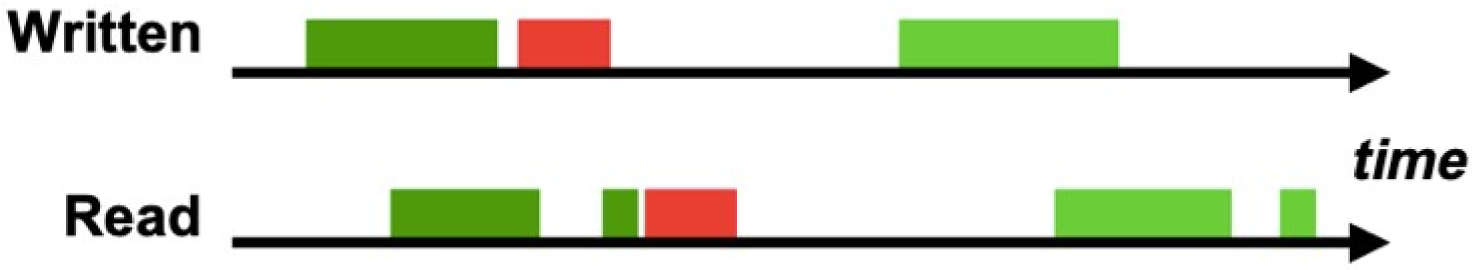
- **Performance**
  - TCP’s window-based flow control scheme leads to bursty bulk transfers (not rate based)
  - TCP’s “slow start” algorithm can reduce throughput
  - TCP has extra overhead per segment
  - UDP can send small, inefficient datagrams (constant bit-rate traffic)



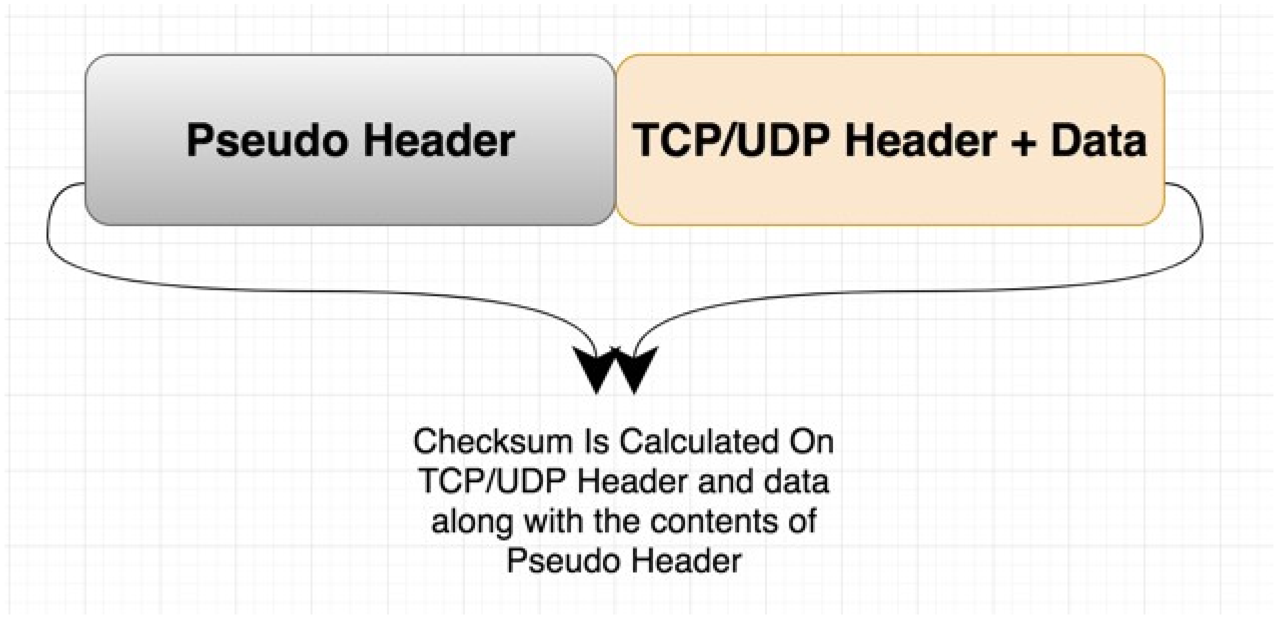
UDP versus TCP

- **Reliability**
  - TCP provides reliable, in-order transfers
  - UDP provides unreliable service -- application must accept or deal with (a) packet loss due to overflows and errors, (b) out-of-order datagrams

- **Application complexity**
  - Application-level framing can be difficult using TCP because of Nagle algorithm
  - Nagle algorithm controls when TCP segments are sent to use IP datagrams efficier
  - But, data may be r ıits that how it was se :P)



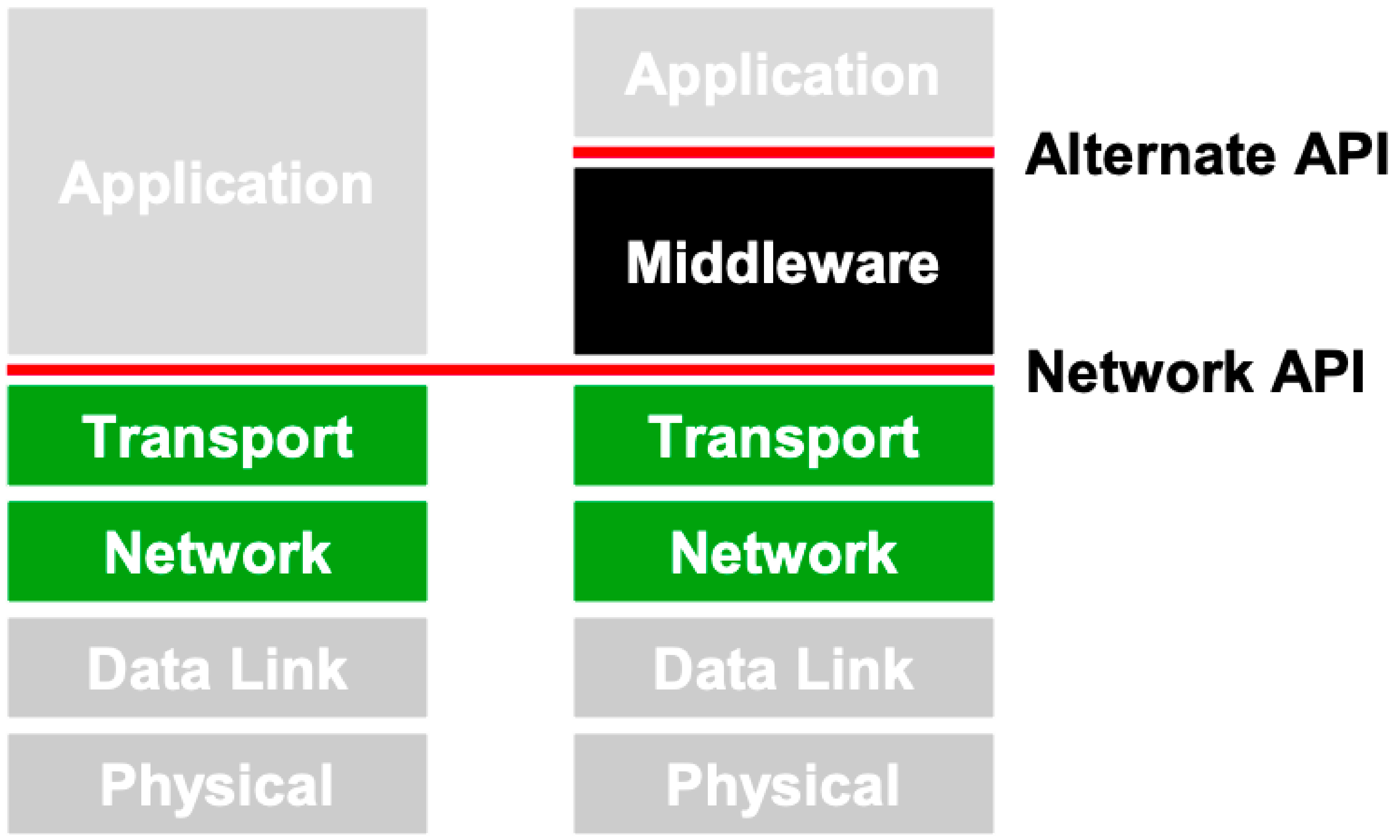
TCP and UDP Checksum Calculation


Pseudo Header
TCP/UDP Header + Data
Checksum Is Calculated On TCP/UDP Header and data along with the contents of Pseudo Header

- **Pseudo Header contains**
  - **Source IP**
  - **Destination IP**
  - **Protocol Header (TCP, UDP or ICMP)**
  - **Segment or datagram length**
  - **Reserved 8 bit**

- An additional layer of verification that the packet has reached to the intended destination (IP also has its own checksum)

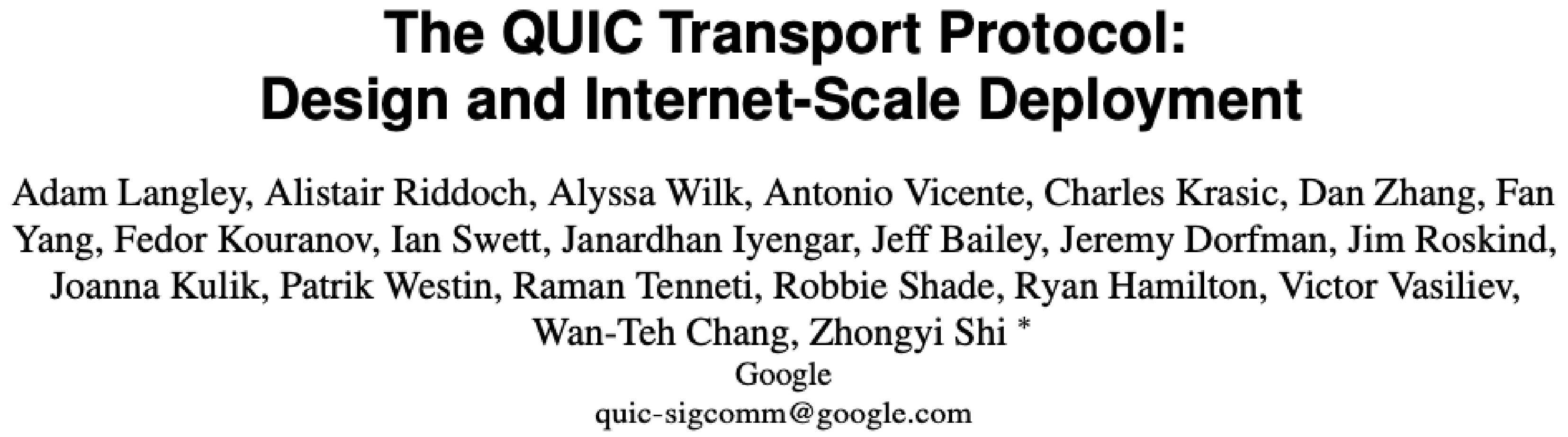
Middleware Model



Quick UDP Internet Connections (QUIC)

- Experimental protocol -- deployed at google starting in 2014
  - Between Google services and Chrome
  - Improved page load latency, video rebuffer rate
  - ~35% of Google's egress traffic (~7% of Internet traffic)
  - Akamai deployed in 2016

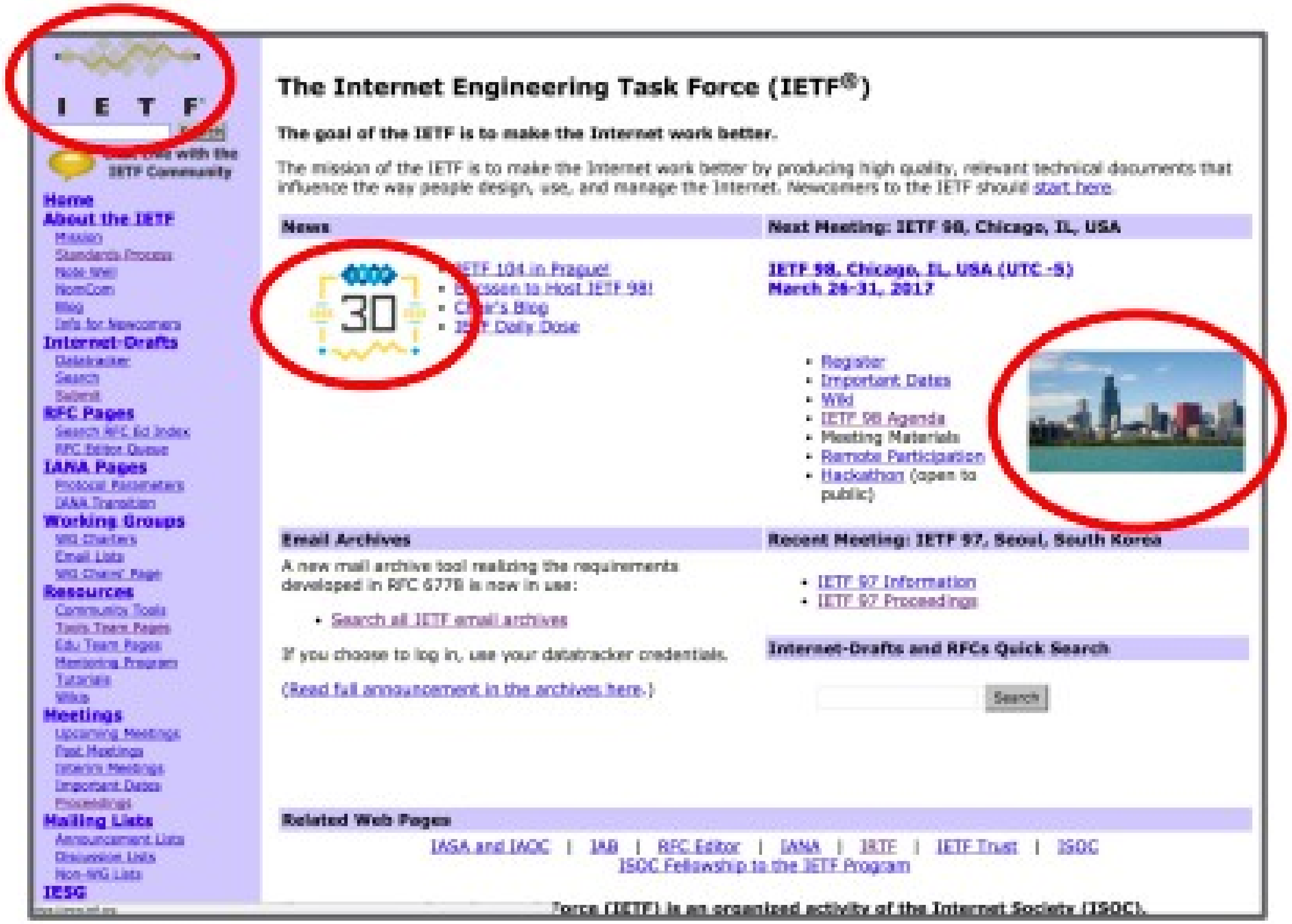
- A transport service (middleware) on top of UDP



The QUIC Transport Protocol: Design and Internet-Scale Deployment

Adam Langley, Alistair Riddoch, Alyssa Wilk, Antonio Vicente, Charles Krasic, Dan Zhang, Fan Yang, Fedor Kouranov, Ian Swett, Janardhan Iyengar, Jeff Bailey, Jeremy Dorfman, Jim Roskind, Joanna Kulik, Patrik Westin, Raman Tenneti, Robbie Shade, Ryan Hamilton, Victor Vasiliev, Wan-Teh Chang, Zhongyi Shi *
Google
quic-sigcomm@google.com

How Does a Web Page Look Like?



I E T F

Chat live with the IETF Community

Home
About the IETF
Mission
Standards Process
Note Well
NomCom
Blog
Info for Newcomers
Internet-Drafts
Datatracker
Search
Submit
RFC Pages
Search RFC Ed Index
RFC Editor Queue
IANA Pages
Protocol Parameters
IANA Transition
Working Groups
WG Charters
Email Lists
WG Chairs' Page
Resources
Community Tools
Tools Team Pages
Edu Team Pages
Mentoring Program
Tutorials
Wikis
Meetings
Upcoming Meetings
Past Meetings
Interim Meetings
Important Dates
Proceedings
Mailing Lists
Announcement Lists
Discussion Lists
Non-WG Lists
IESG

The Internet Engineering Task Force (IETF®)

The goal of the IETF is to make the Internet work better.

The mission of the IETF is to make the Internet work better by producing high quality, relevant technical documents that influence the way people design, use, and manage the Internet. Newcomers to the IETF should start here.

News

- IETF 104 in Prague!
- Ericsson to Host IETF 98!
- Chair's Blog
- IETF Daily Dose

Next Meeting: IETF 98, Chicago, IL, USA

IETF 98, Chicago, IL, USA (UTC -5)
March 26-31, 2017

- Register
- Important Dates
- Wiki
- IETF 98 Agenda
- Meeting Materials
- Remote Participation
- Hackathon (open to public)

Email Archives

A new mail archive tool realizing the requirements developed in RFC 6778 is now in use:

- Search all IETF email archives

If you choose to log in, use your datatracker credentials.

(Read full announcement in the archives here.)

Recent Meeting: IETF 97, Seoul, South Korea

- IETF 97 Information
- IETF 97 Proceedings

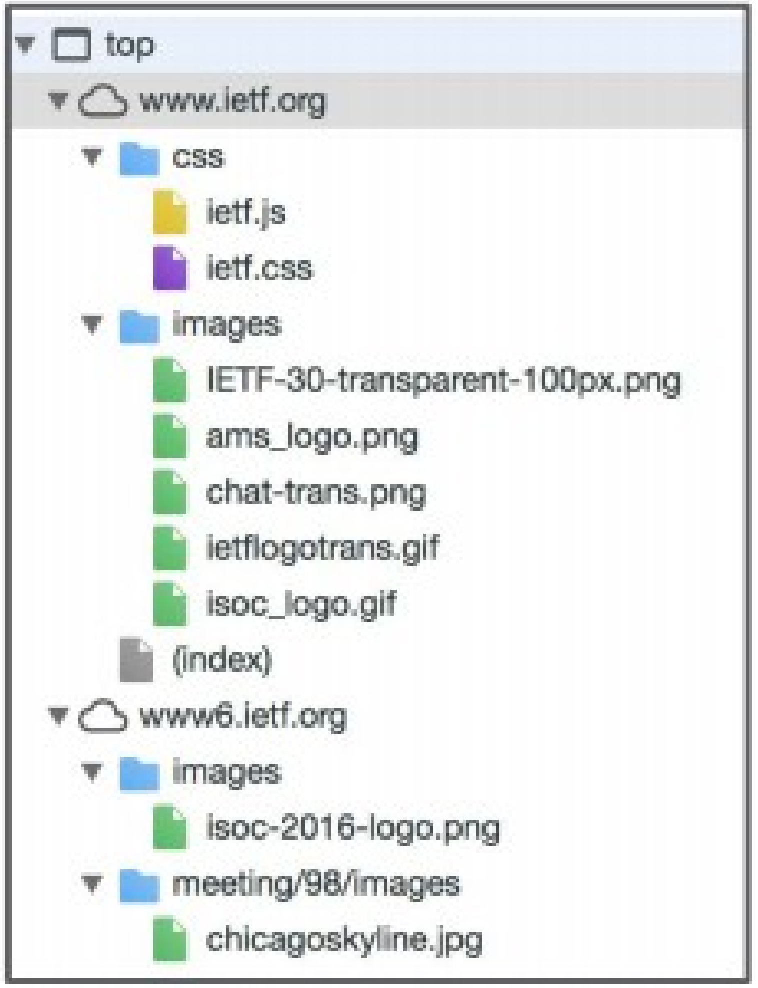
Internet-Drafts and RFCs Quick Search

Search

Related Web Pages

IASA and IAOC | IAB | RFC Editor | IANA | IRTF | IETF Trust | ISOC
ISOC Fellowship to the IETF Program

Force (IETF) is an organized activity of the Internet Society (ISOC).



```
▼ top
  ▼ www.ietf.org
    ▼ css
        ietf.js
        ietf.css
    ▼ images
        IETF-30-transparent-100px.png
        ams_logo.png
        chat-trans.png
        ietflogotrans.gif
        isoc_logo.gif
      (index)
  ▼ www6.ietf.org
    ▼ images
        isoc-2016-logo.png
    ▼ meeting/98/images
        chicagoskyline.jpg
```

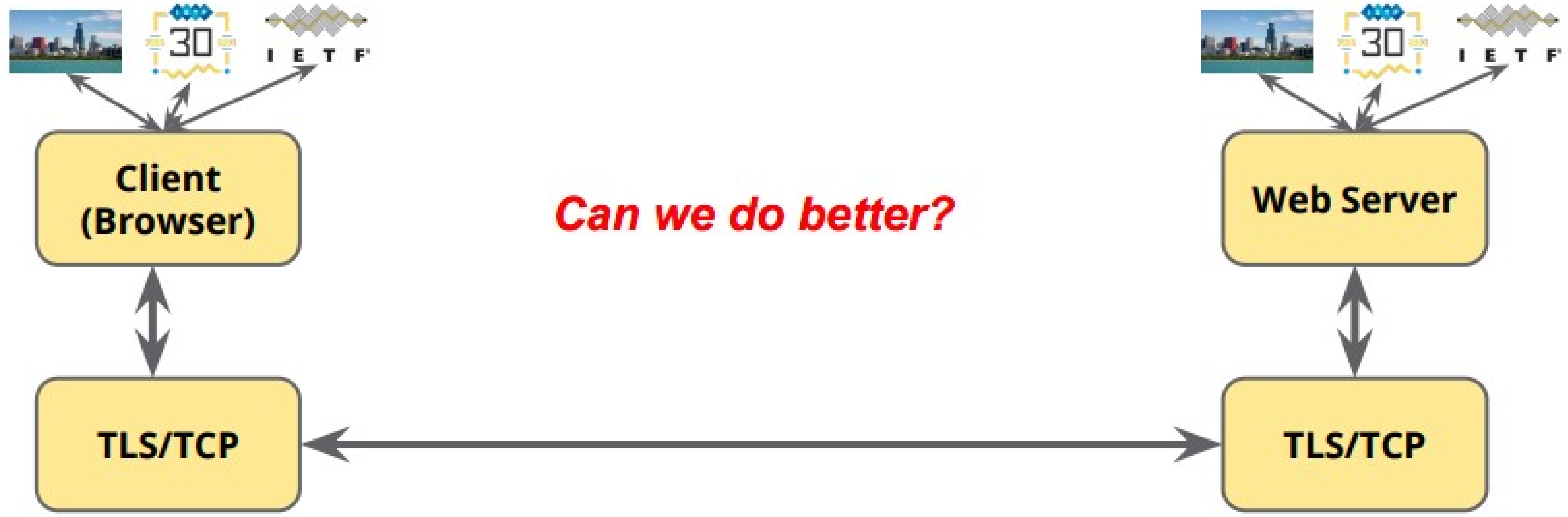
HTTP/1

- Connection setup
  - 1 round-trip to set up a TCP connection
  - 2 round-trips to set up a TLS 1.2 connection

- After the connection setup, HTTP requests/responses flow over the connection
  - Persistent HTTP -- all the embedded objects are transferred over the TCP/TLS connection

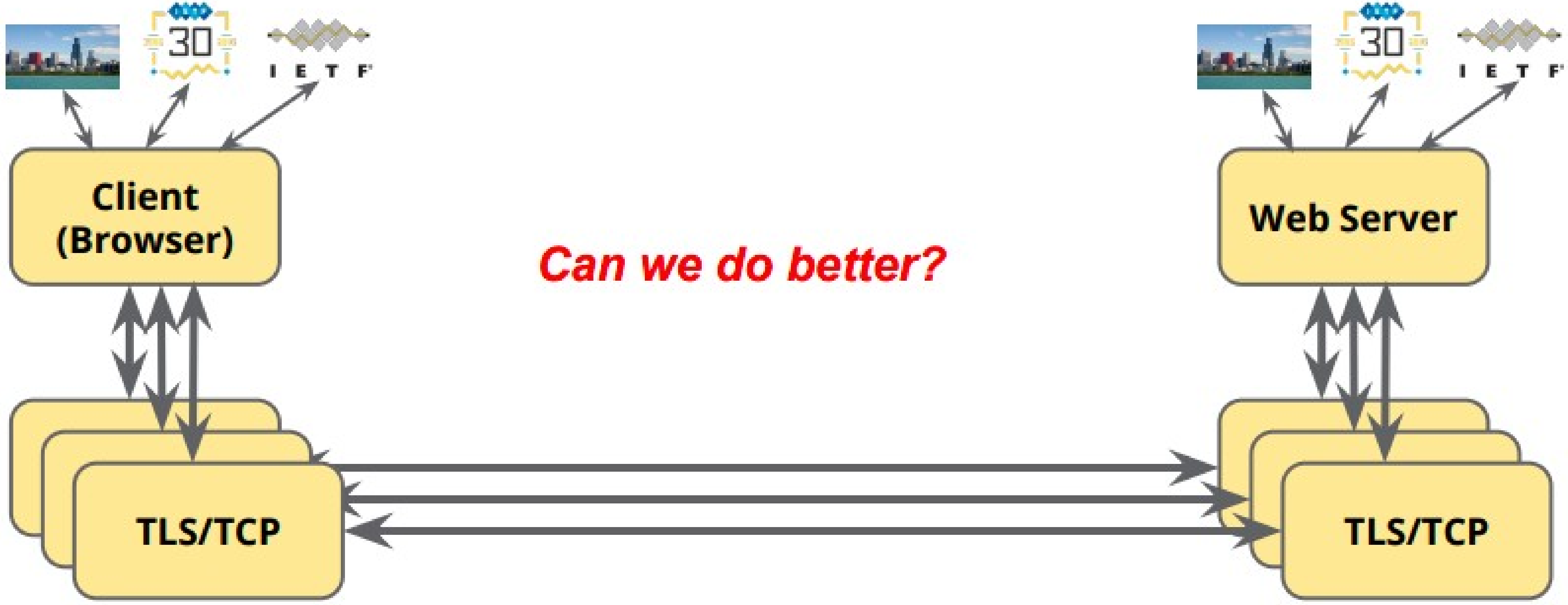
HTTP/1 - Head of Line (HoL) Blocking

- TCP congestion control blocks the sender side when there is a segment loss
  - The receiver keeps on sending the duplicate acknowledgements
  - The sender-side window is blocked until the lost packet is recovered, and a fresh ACK is received



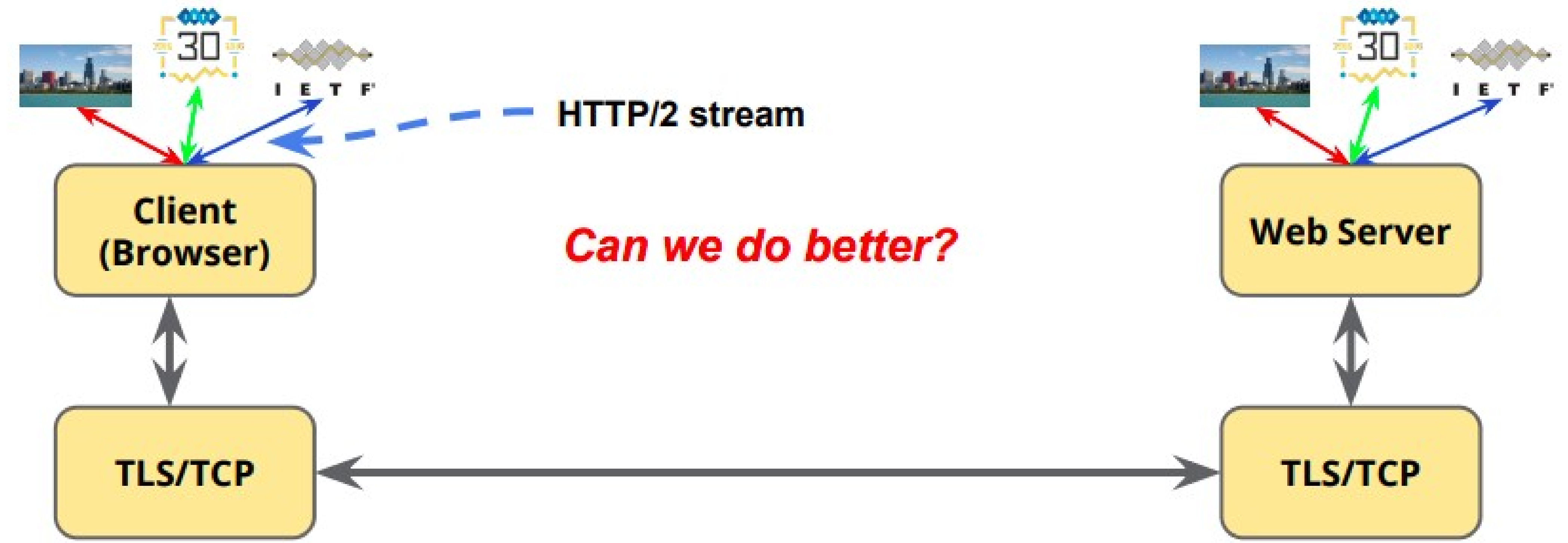
HTTP/1.1 Parallel Connections

- Use parallel connections for each HTTP request-response pair



HTTP/1.1 - Can we do Better?

- Use parallel streams -- multiplex the HTTP request-response streams over a single TCP connection (SPDY -- the precursor of HTTP/2)
  - Use congestion control for each streams
  - How this is different from HTTP/1.1 (parallel connections)?

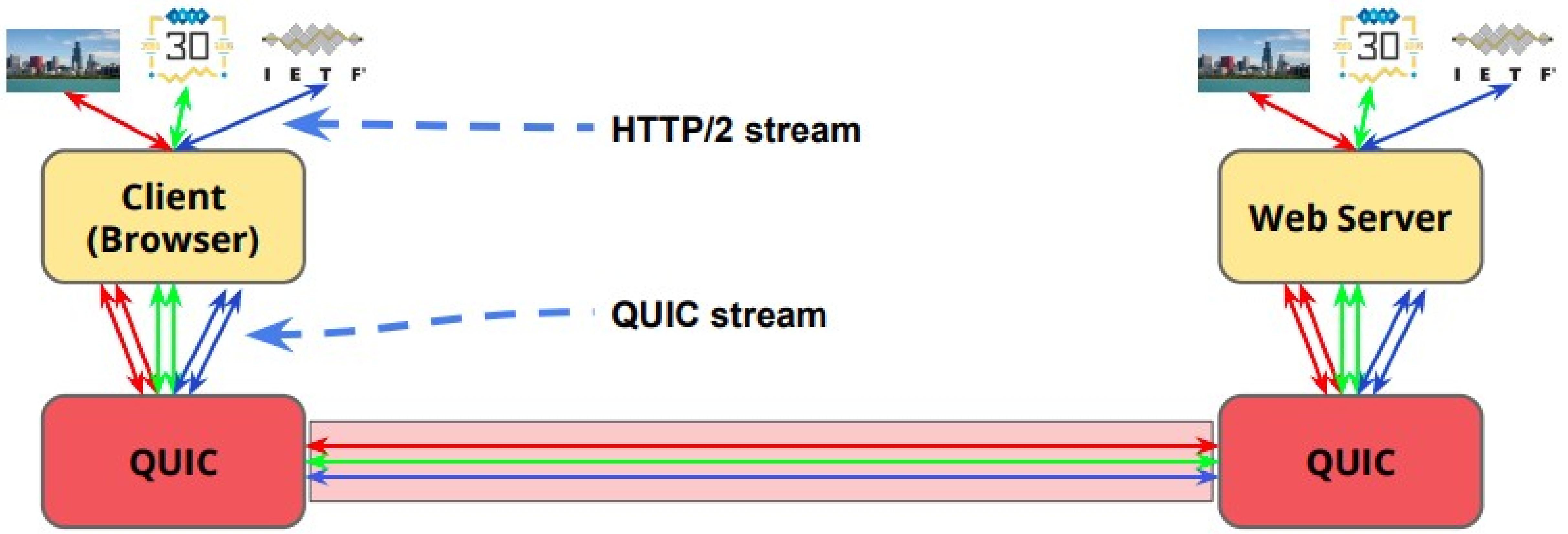


Problems with TCP

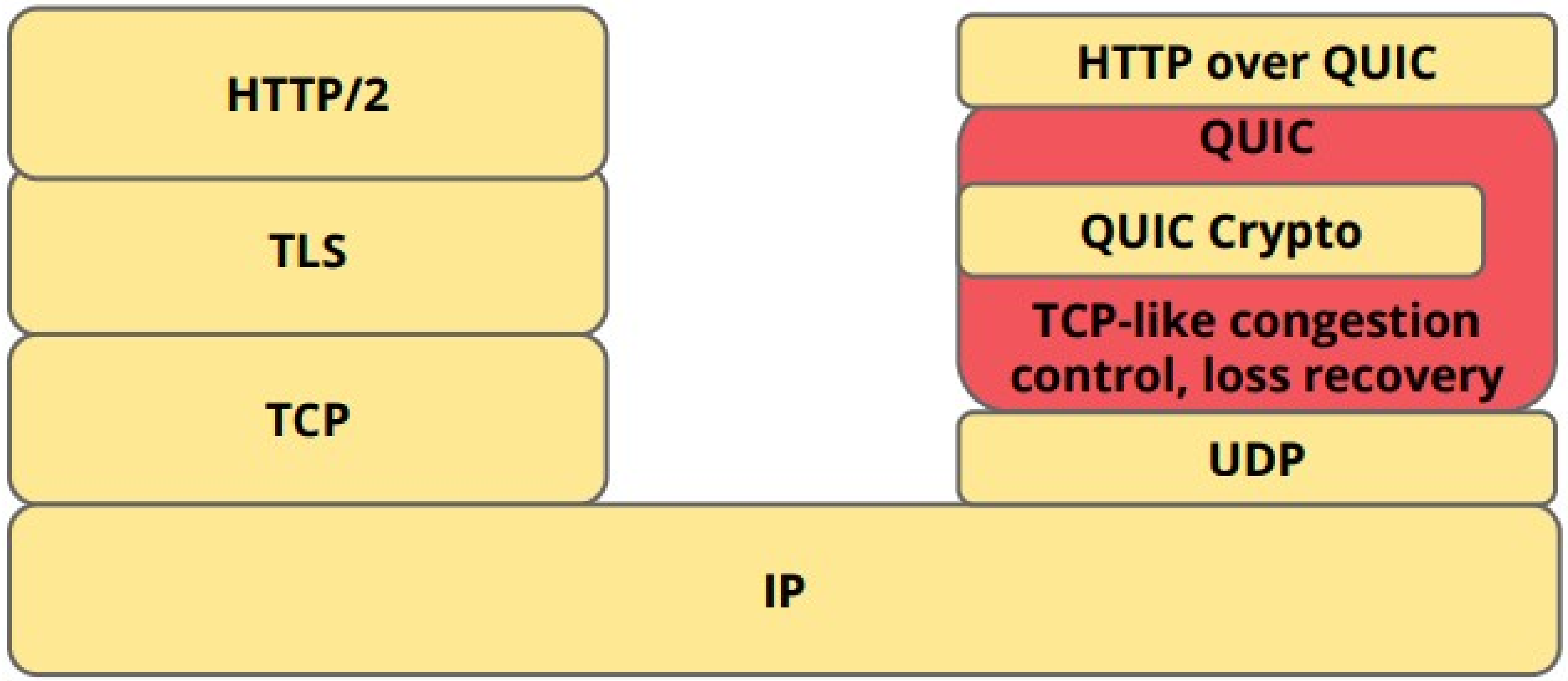
- **Implementation Entrenchment**
  - TCP is implemented in OS kernel -- needs kernel modifications for every updates -- less control for the application
  - Application-specific tuning is difficult -- need to push changes in the TCP stack -- requires OS upgrade

- **Handshake Delay**
  - 3 round-trips are required for establishing a TCP/TLS connection
  - Most transfers in the Internet are short transfers -- 3 RTT handshaking is an overhead

QUIC Streams



QUIC Protocol Stack

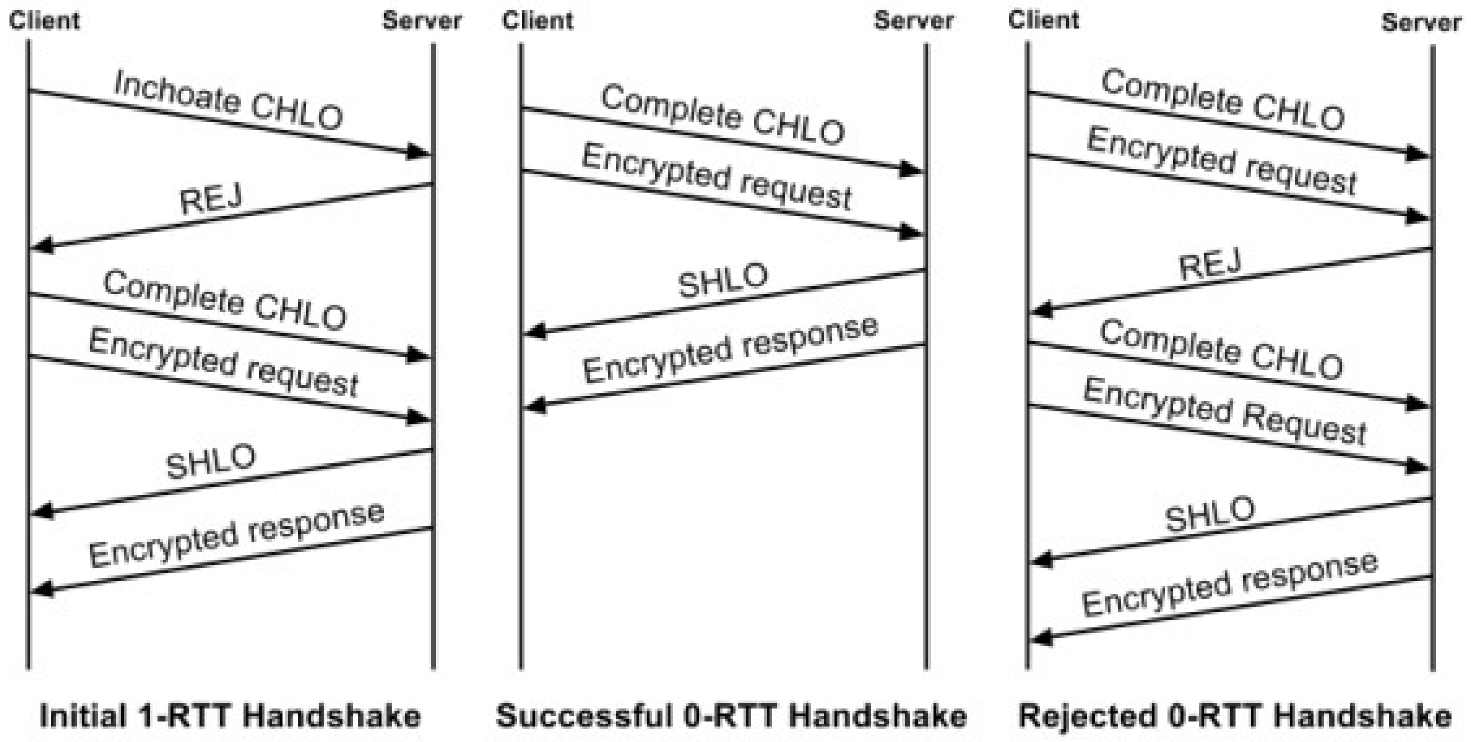


QUIC Handshaking -- Connection Establishment

- QUIC client caches information about the origin
  - On subsequent connections to the same origin, the client can establish an encrypted connection with no additional round trips
  - 0-RTT connection to a known server

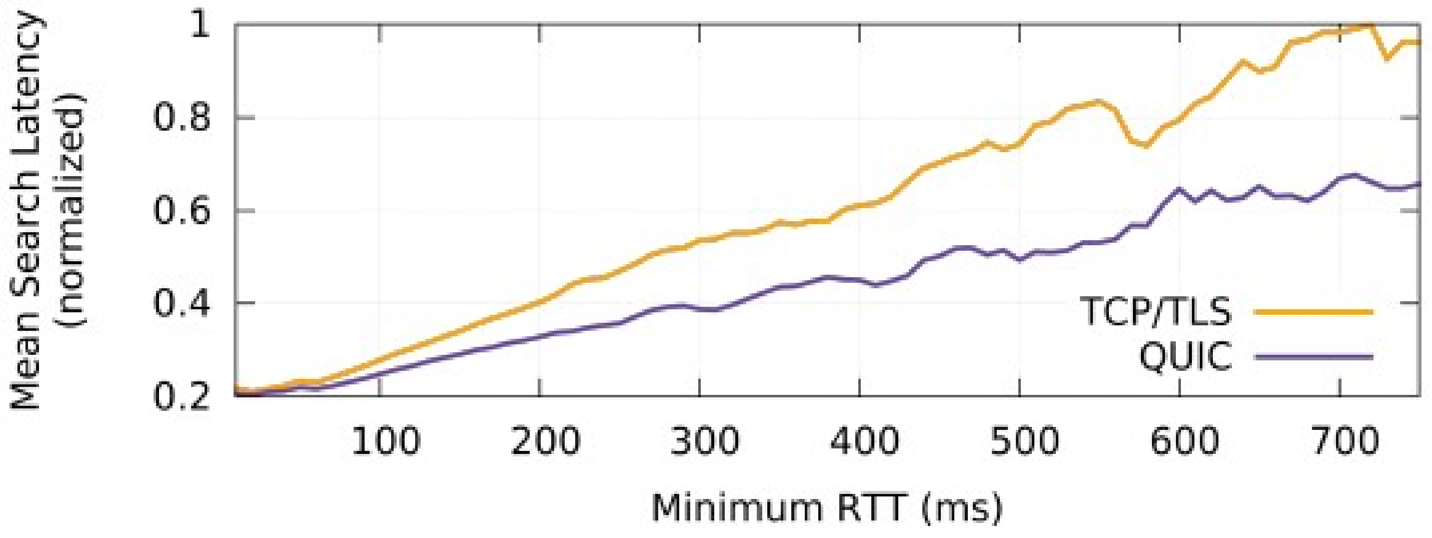
- Connection establishment to an unknown server
  - 1-RTT if crypto keys are not new -- QUIC embeds the key exchange protocol within the transport protocol itself -- no separate handshaking like TCP and TLS
  - 2-RTTs if QUIC version negotiation needed

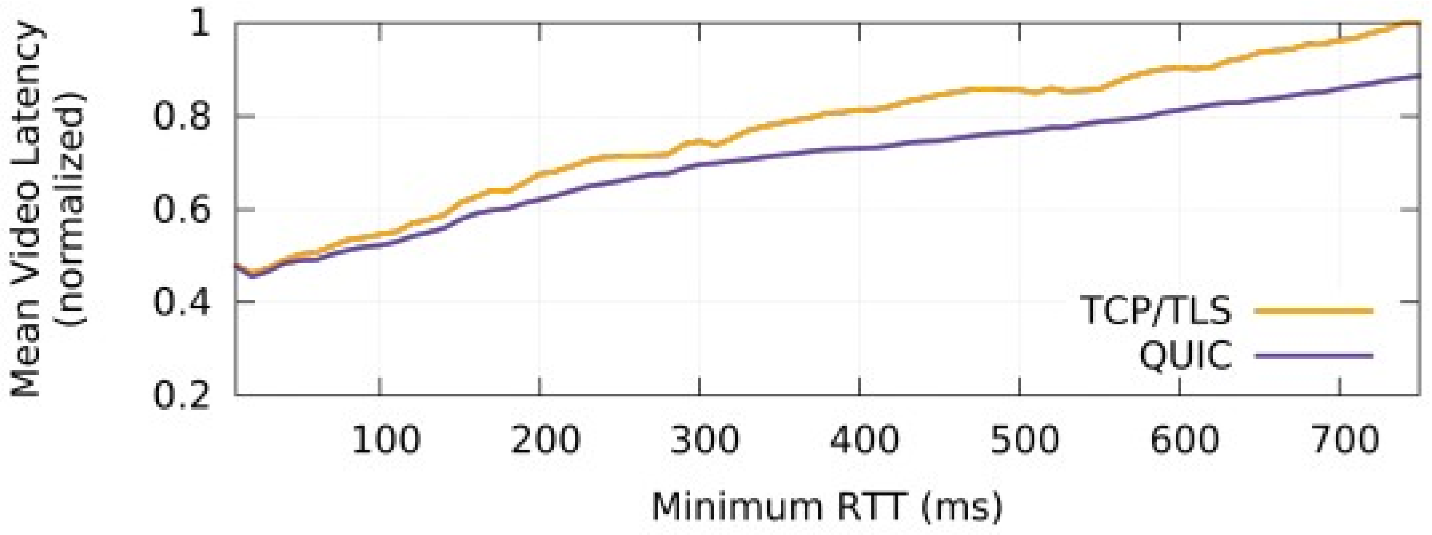
QUIC Connection Establishment



- The REJ message contains
  - a server *config* that contains server's long-term Diffie-Hellman public value
  - a certificate chain authenticating the server
  - a signature of the server config
  - a source-address token: An authenticated encryption block that contains client's IP address and a time-stamp by the server

QUIC Performance





This is a broad discussion of the Transport Layer … Next we'll move to the discussion of Network layer and the IP Protocol