

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



Evolving transport-layer functionality

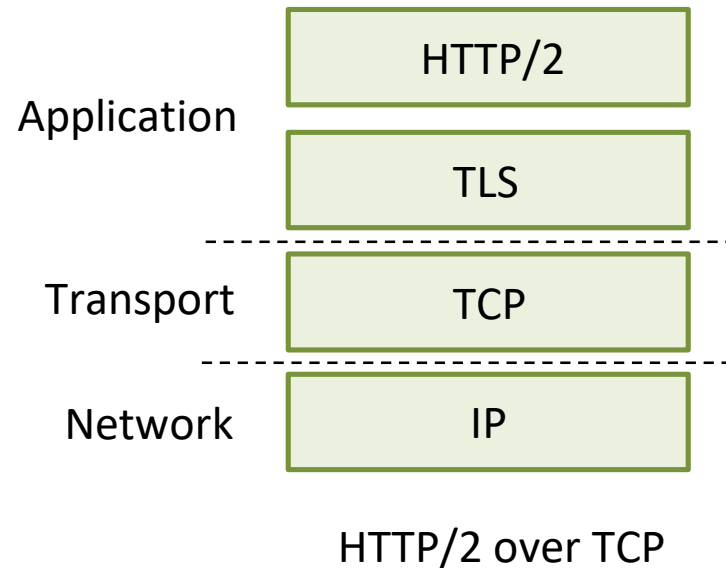
- TCP, UDP: principal transport protocols for 40 years
- different “flavors” of TCP developed, for specific scenarios:

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)

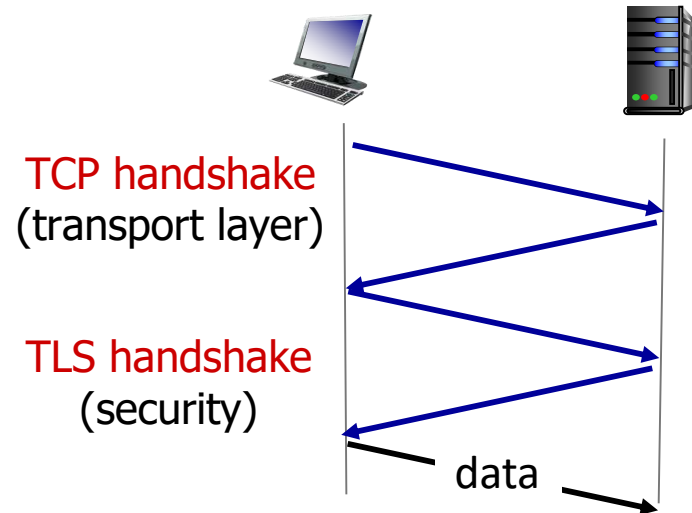


QUIC: Quick UDP Internet Connections

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

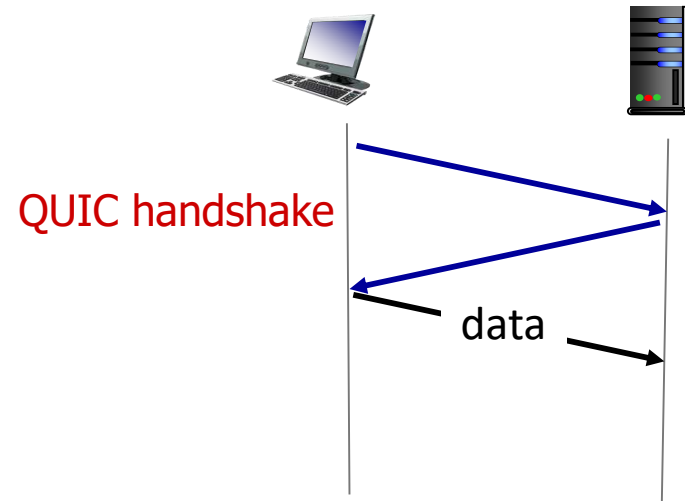
- **error and congestion control:** “Readers familiar with TCP’s loss detection and congestion control will find algorithms here that parallel well-known TCP ones.” [from QUIC specification]
- **connection establishment:** reliability, congestion control, authentication, encryption, state established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

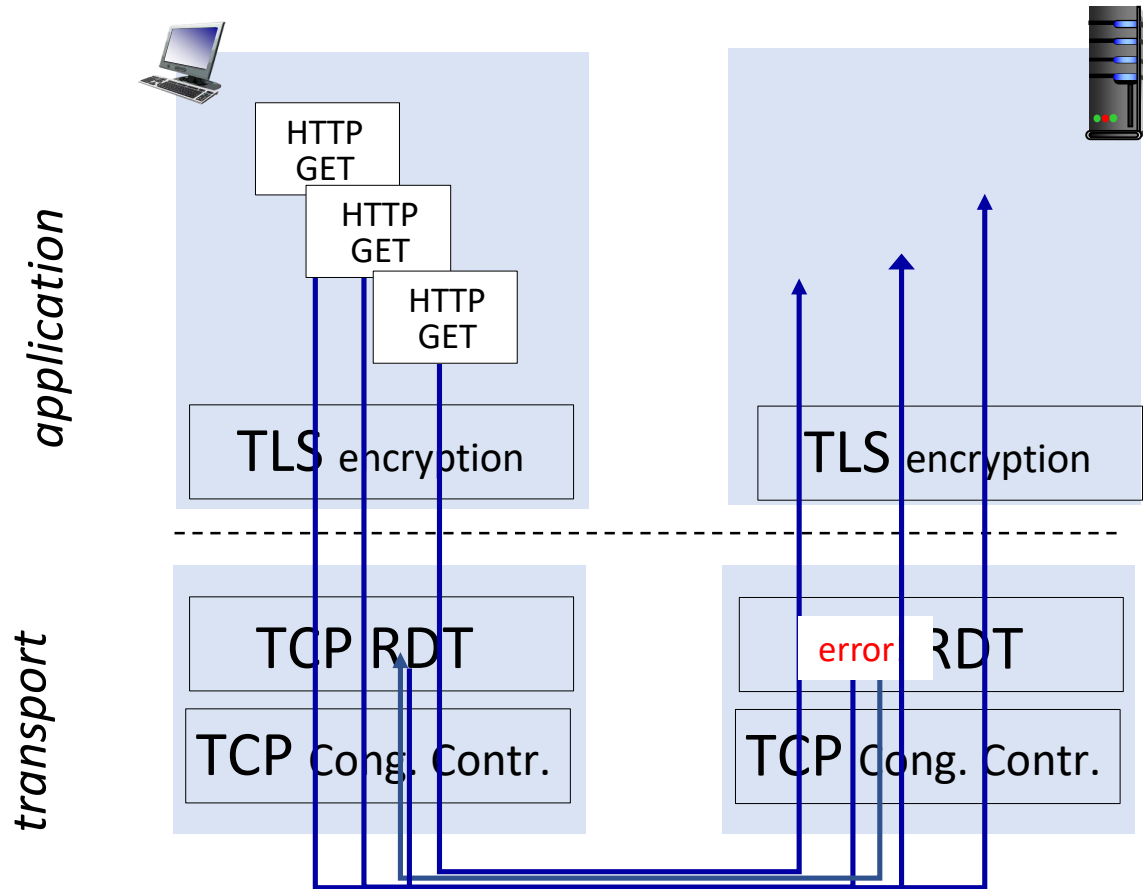
- 2 serial handshakes



QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

Chapter 3: summary

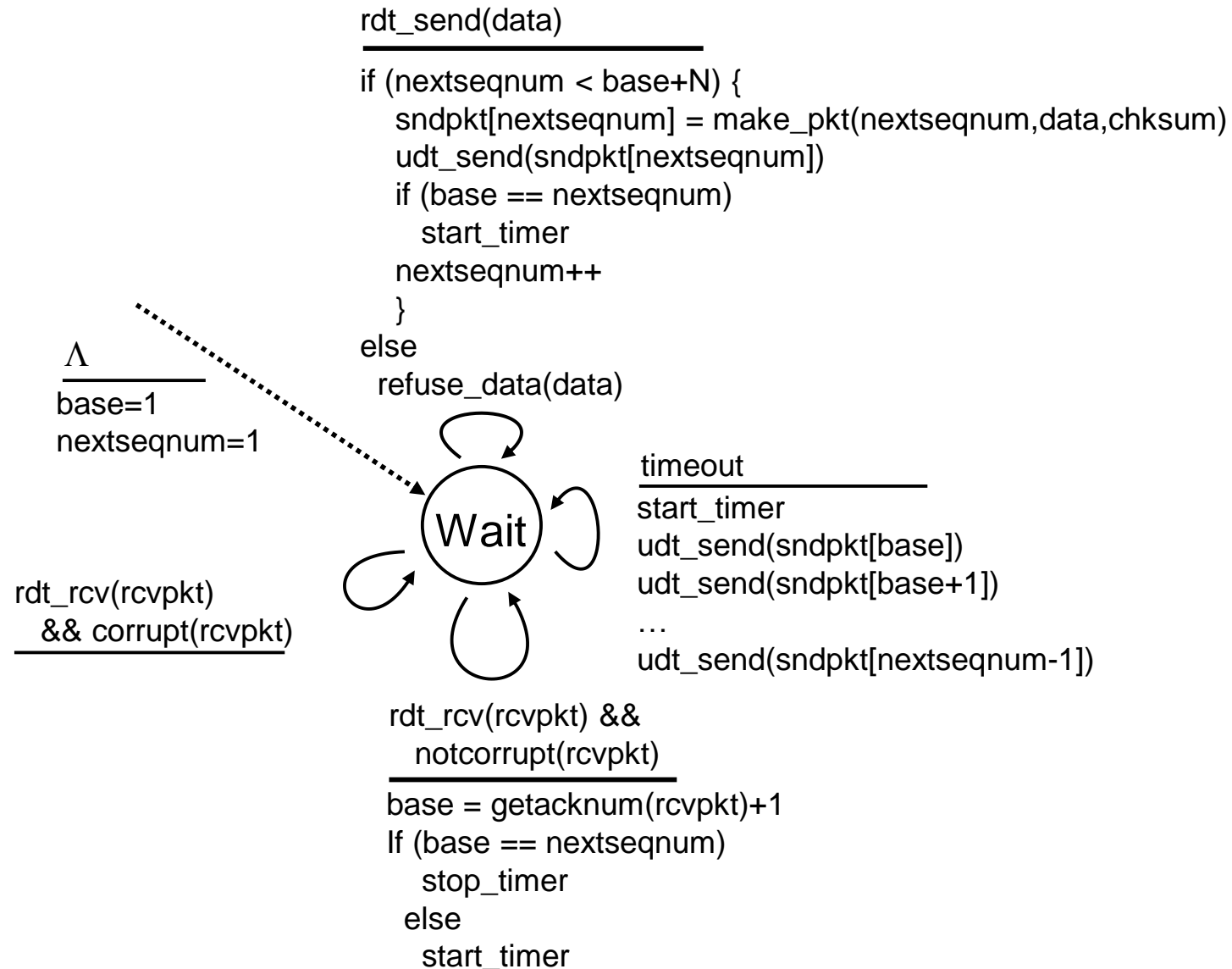
- principles behind transport layer services:
 - multiplexing, demultiplexing
 - reliable data transfer
 - flow control
 - congestion control
- instantiation, implementation in the Internet
 - UDP
 - TCP

Up next:

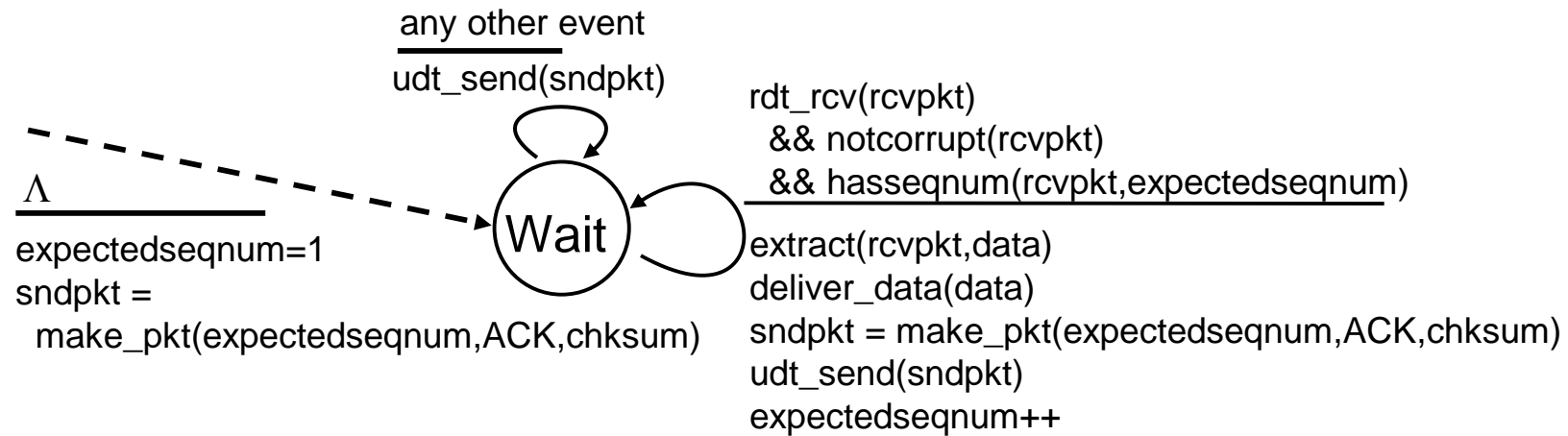
- leaving the network “edge” (application, transport layers)
- into the network “core”
- two network-layer chapters:
 - data plane
 - control plane

Additional Chapter 3 slides

Go-Back-N: sender extended FSM



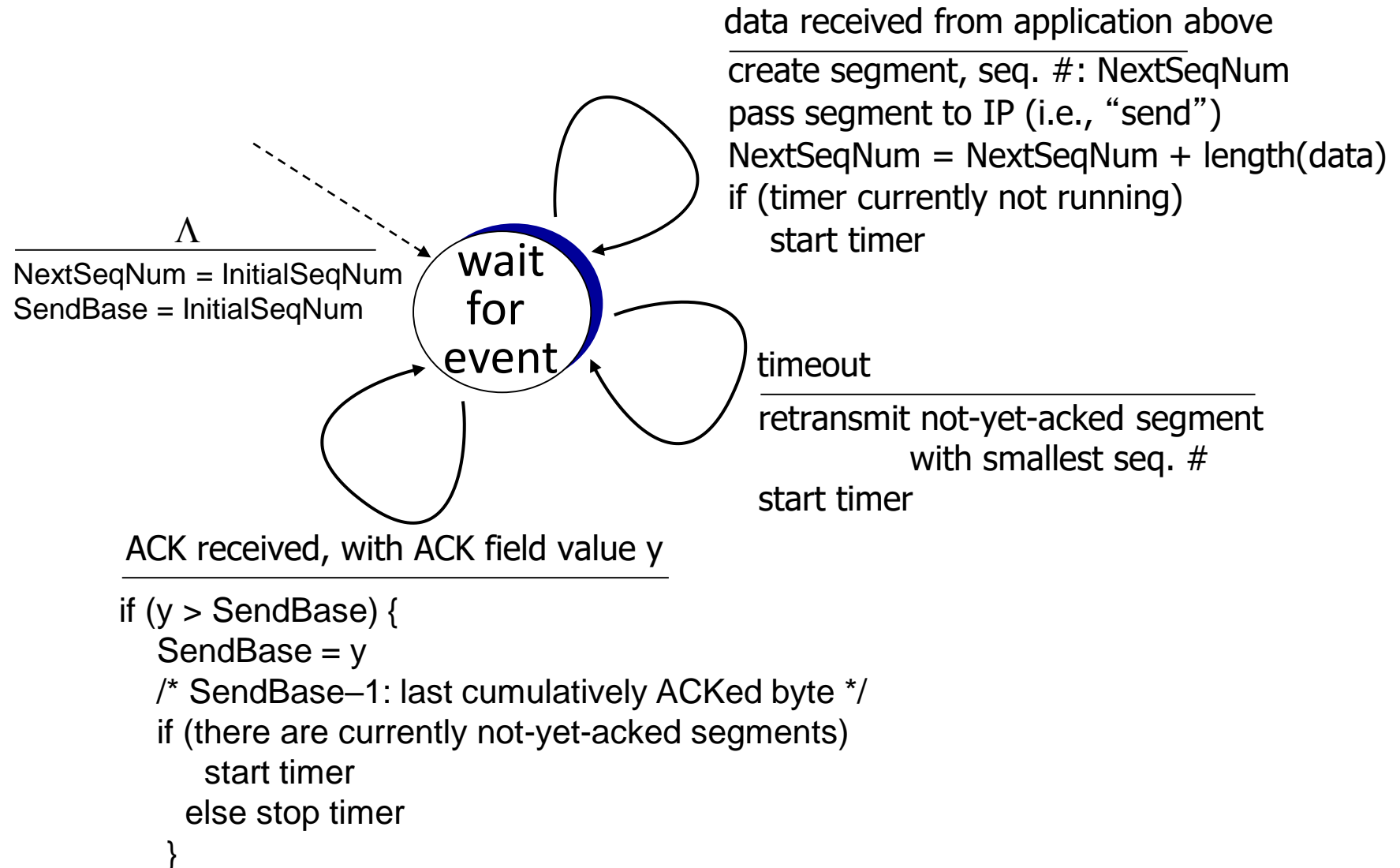
Go-Back-N: receiver extended FSM



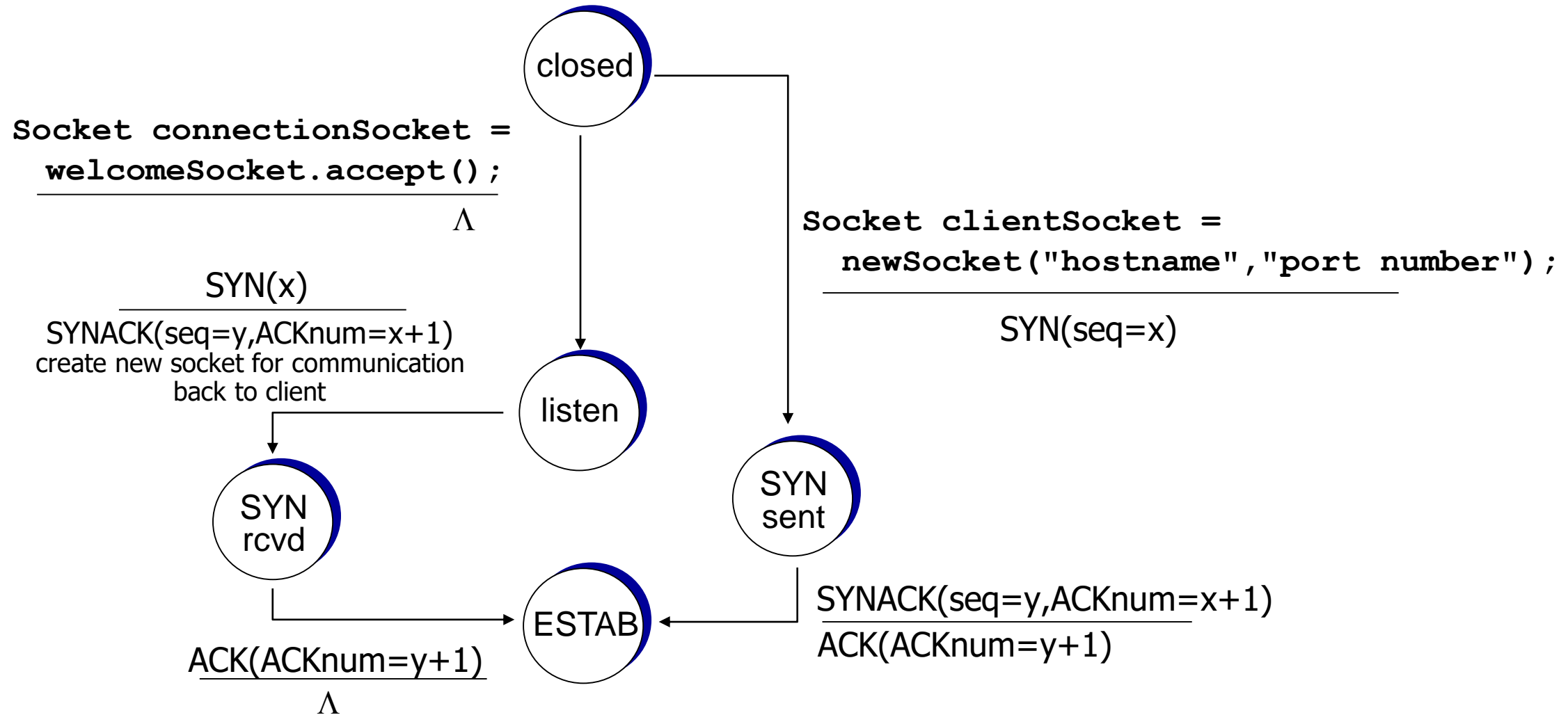
ACK-only: always send ACK for correctly-received packet with highest *in-order* seq #

- may generate duplicate ACKs
 - need only remember **expectedseqnum**
- out-of-order packet:
- discard (don't buffer): *no receiver buffering!*
 - re-ACK pkt with highest in-order seq #

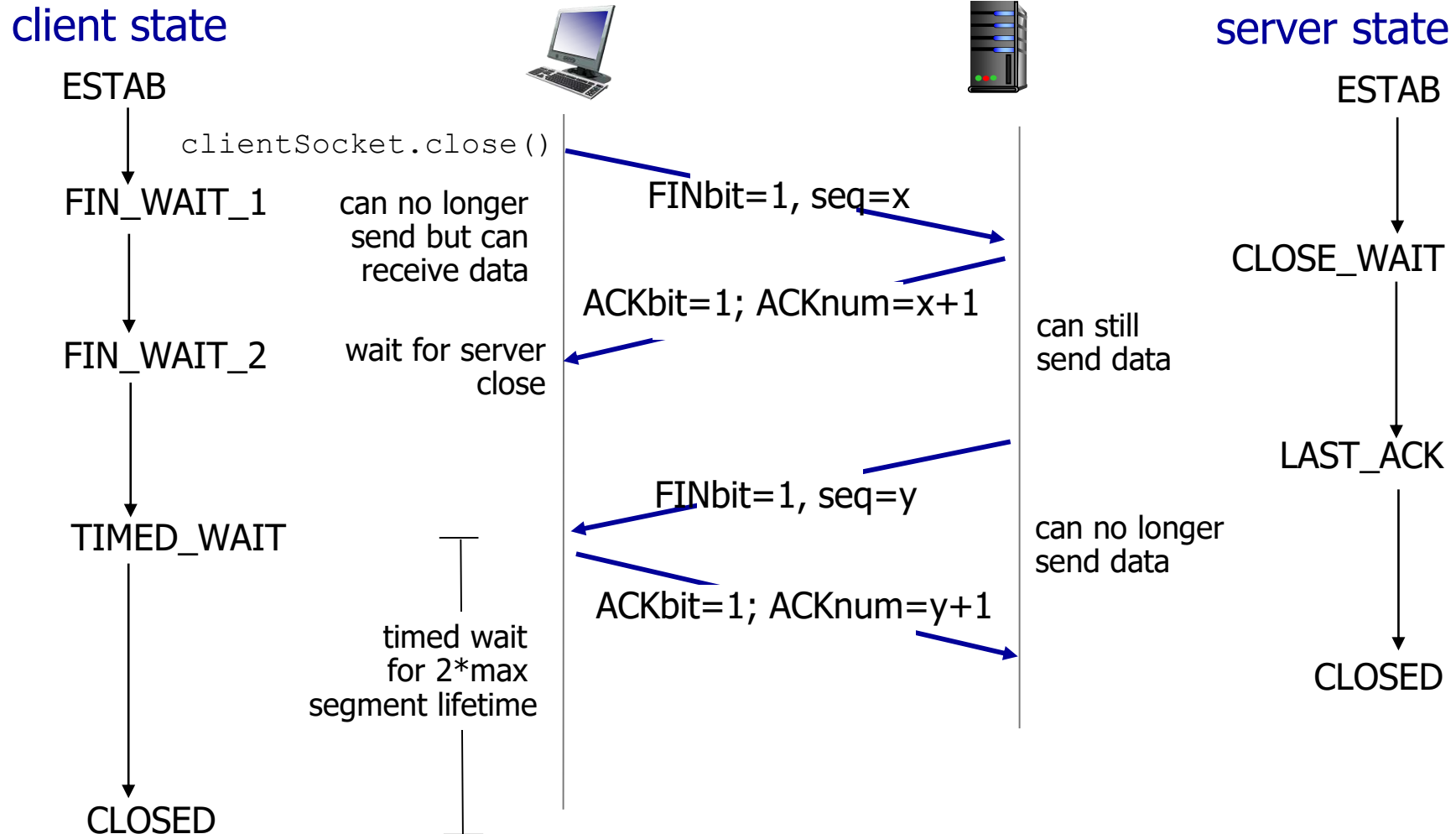
TCP sender (simplified)



TCP 3-way handshake FSM

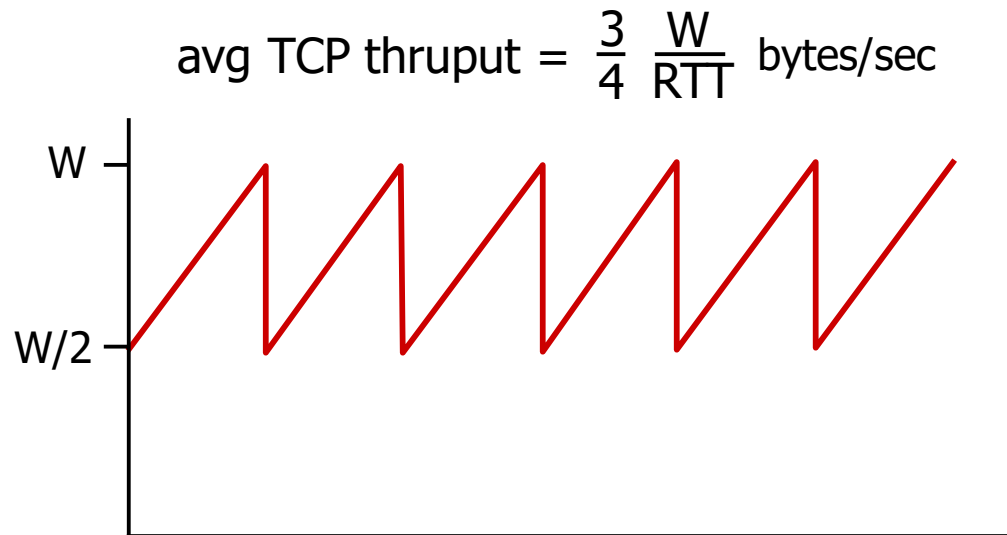


Closing a TCP connection



TCP throughput

- avg. TCP thruput as function of window size, RTT?
 - ignore slow start, assume there is always data to send
- W : window size (measured in bytes) where loss occurs
 - avg. window size (# in-flight bytes) is $\frac{3}{4} W$
 - avg. thruput is $3/4W$ per RTT



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