CSE 461: TCP (part 2)

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Transport: Practice

Protocols

- IP -- Internet protocol
- UDP -- user datagram protocol
- TCP -- transmission control protocol
- RPC -- remote procedure call
- HTTP -- hypertext transfer protocol
- And a bunch more…

How do we connect processes?

IP provides host to host packet delivery

- header has source, destination IP address

For applications to communicate, need to demux packets sent to host to target app

- Web browser (HTTP), Email servers (SMTP), hostname translation (DNS), RealAudio player (RTSP), etc.
- Process id is OS-specific and transient

Ports

Port is a mailbox that processes "rent"

- Uniquely identify communication endpoint as (IP address, protocol, port)

How do we pick port #'s?

- Client needs to know port # to send server a request
- Servers bind to "well-known" port numbers
  - Ex: HTTP 80, SMTP 25, DNS 53, …
  - Ports below 1024 reserved for "well-known" services
- Clients use OS-assigned temporary (ephemeral) ports
  - Above 1024, recycled by OS when client finished

Sockets

- OS abstraction representing communication endpoint
  - Layer on top of TCP, UDP, local pipes
- server (passive open)
  - bind -- socket to specific local port
  - listen -- wait for client to connect
- client (active open)
  - connect -- to specific remote port

User Datagram Protocol (UDP)

Provides application – application delivery

Header has source & dest port #'s

- IP header provides source, dest IP addresses

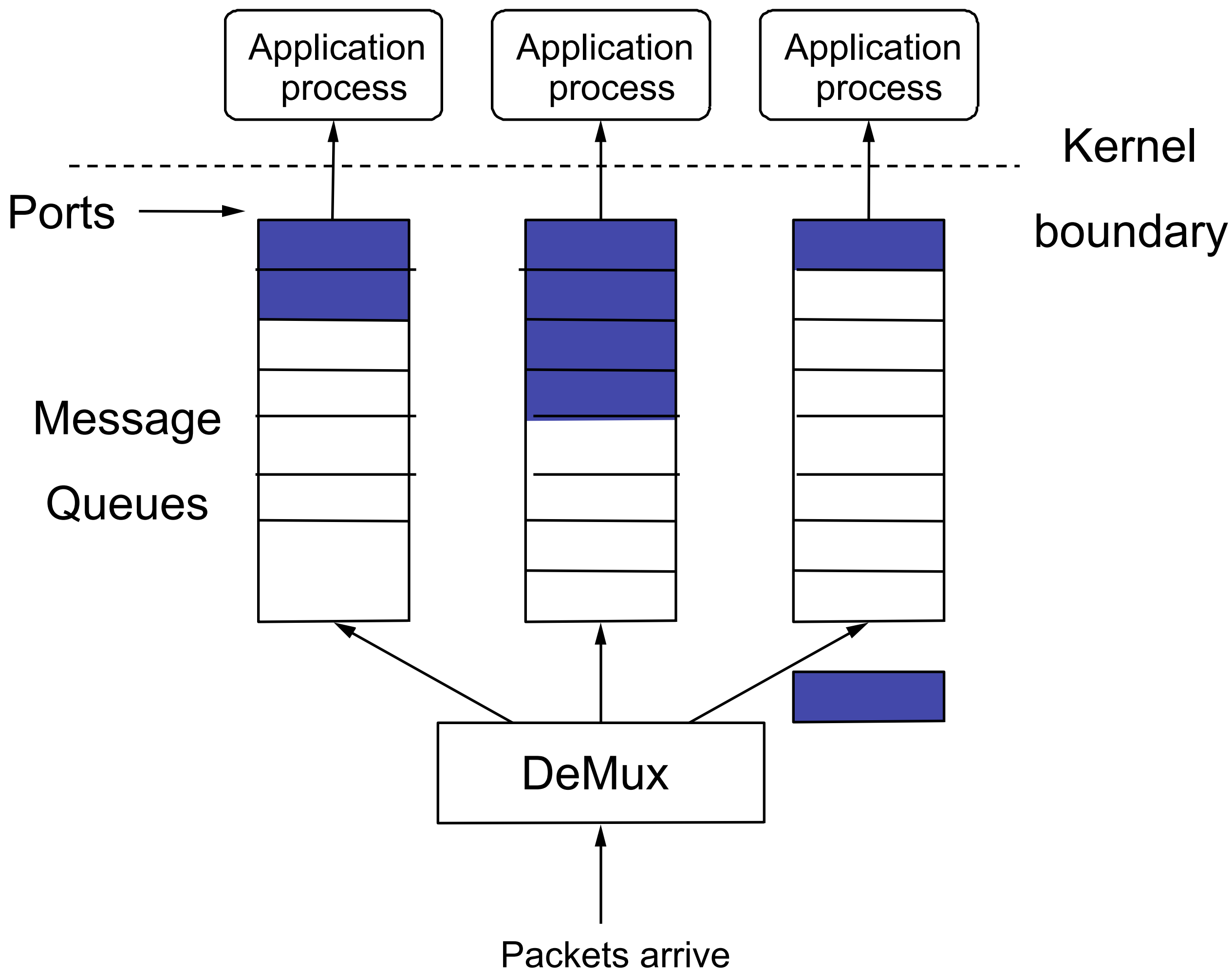
Deliver to destination port on dest machine

Reply returns to source port on source machine

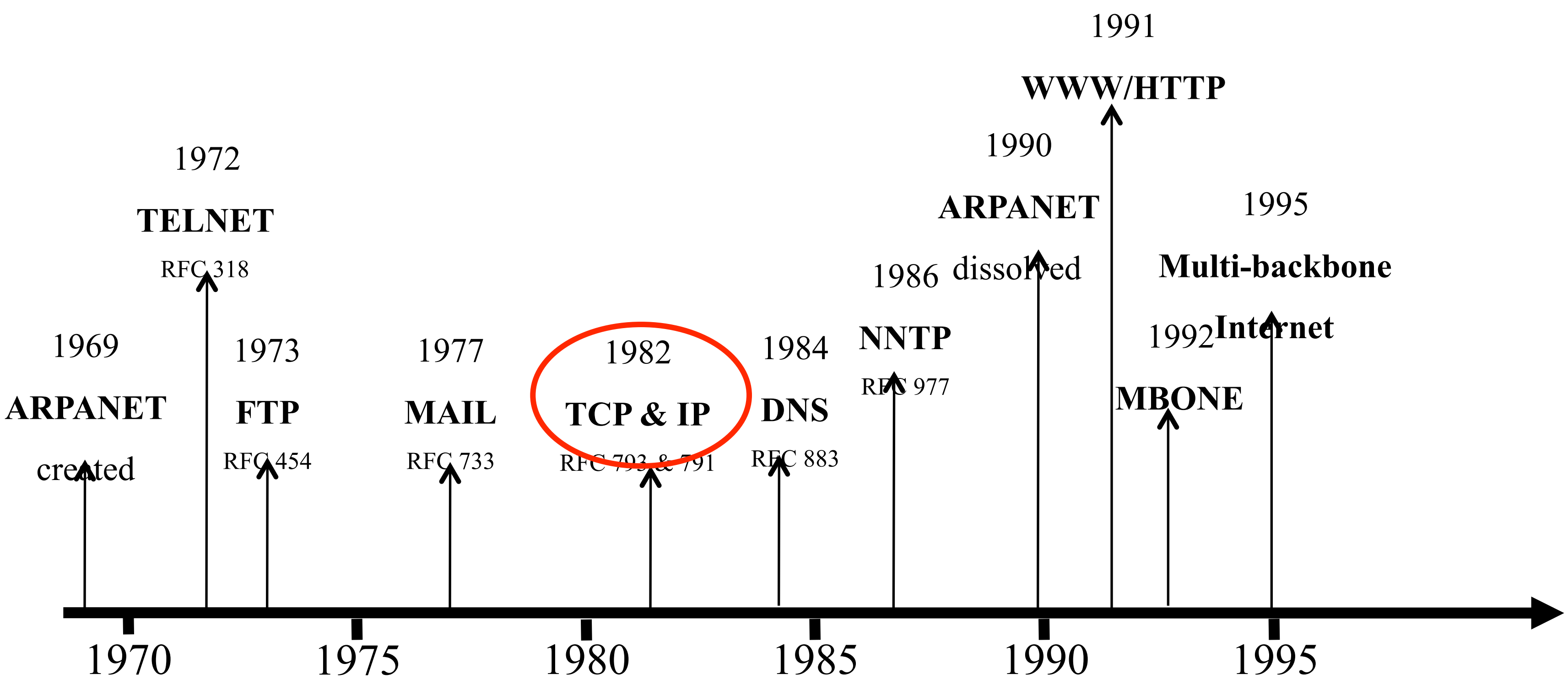
No retransmissions, no sequence #s

=> stateless

UDP Delivery



A brief Internet history...



TCP: This is your life...

1974

TCP described by

Vint Cerf and *Bob Kahn*

In IEEE Trans Comm

1975

Three-way handshake

Raymond Tomlinson

In SIGCOMM 75

1982

TCP & IP

RFC 793 & 791

1983

BSD Unix 4.2

supports TCP/IP

1984

Nagel's algorithm

to reduce overhead

of small packets;

predicts congestion

collapse

1986

Congestion collapse

observed

1987

Karn's algorithm

to better estimate

round-trip time

1988

Van Jacobson's algorithms

congestion avoidance and congestion control

(*most* implemented in **4.3BSD Tahoe**)

1990

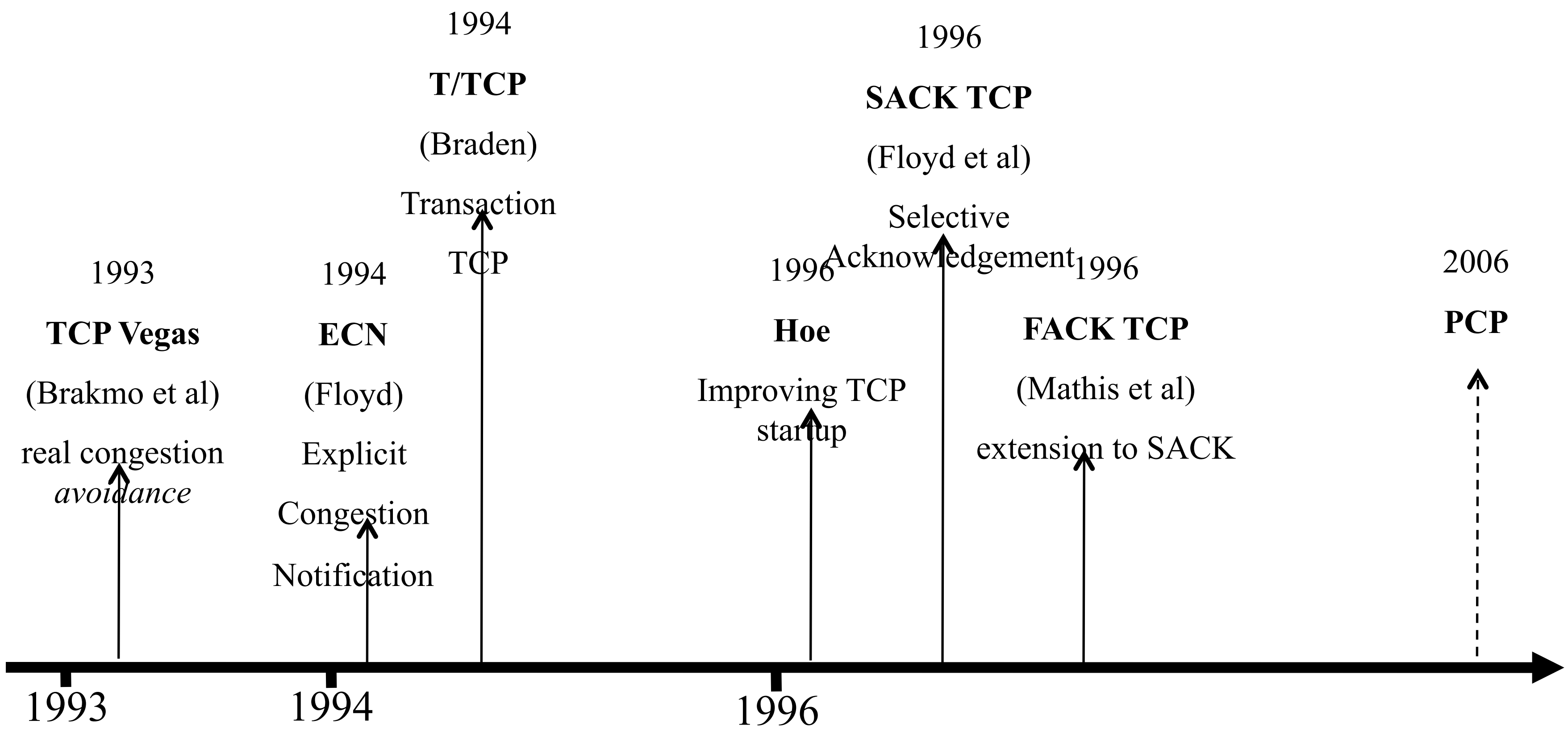
4.3BSD Reno

fast retransmit

delayed ACK's

1975 1980 1985 1990

TCP: After 1990



Transmission Control Protocol (TCP)

Reliable bi-directional byte stream
- No message boundaries
- Ports as application endpoints

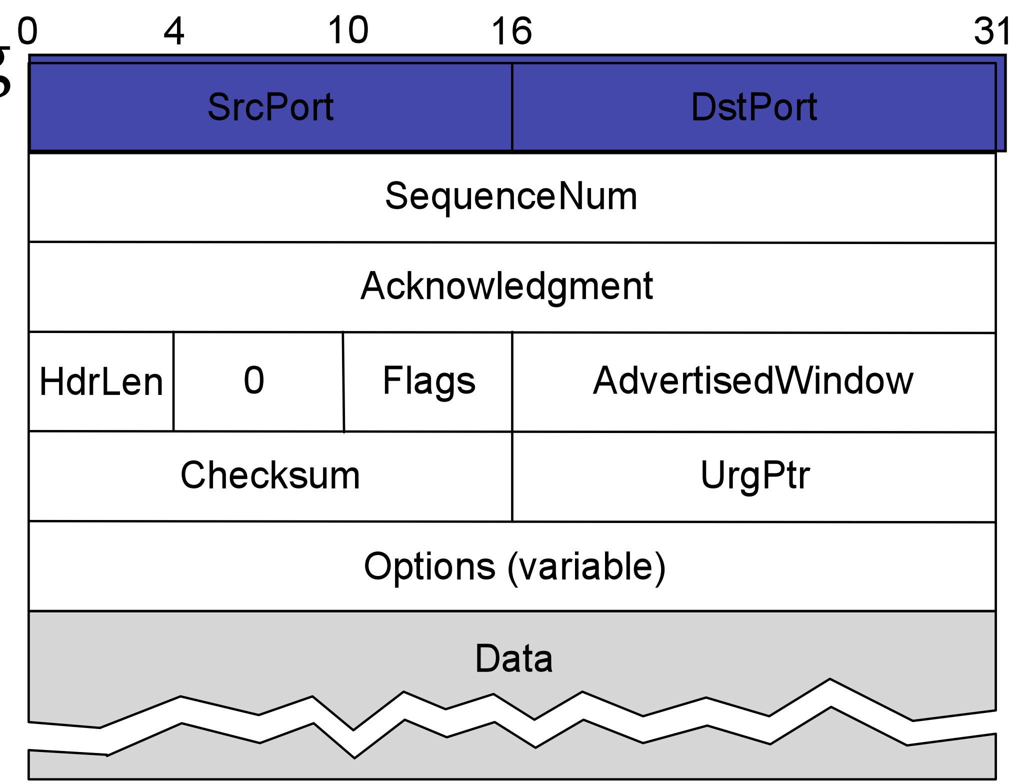
Sliding window, go back N/SACK, RTT est, …
- Highly tuned congestion control algorithm

Flow control
- prevent sender from overrunning receiver buffers

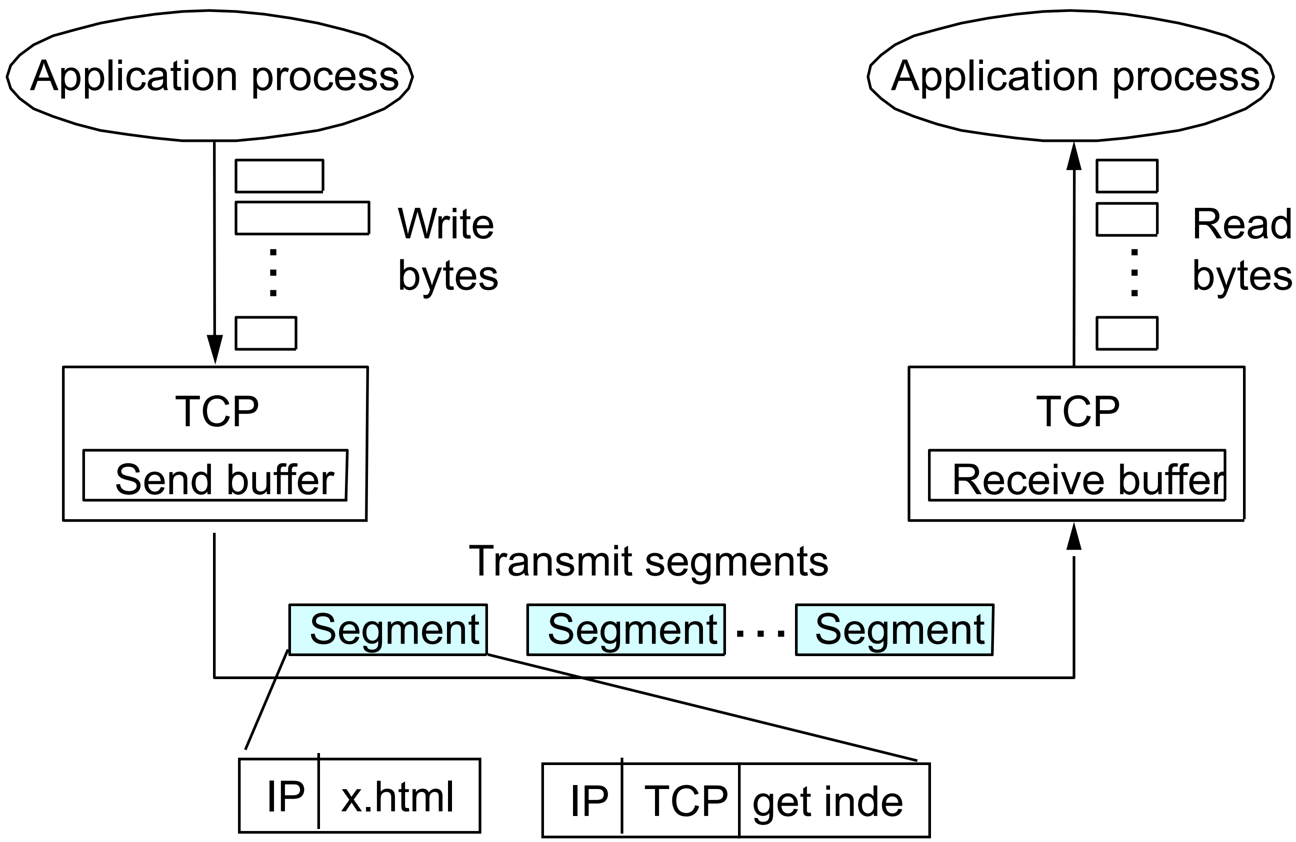
Connection setup
- negotiate buffer sizes and initial seq #s
- Needs to work between all types of computers (supercomputer -> 8086)

TCP Packet Header

- Source, destination ports
- Sequence # (bytes being sent)
- Ack # (next byte expected)
- Receive window size
- Checksum
- Flags: SYN, FIN, RST



TCP Delivery



TCP Sliding Window

Per-byte, not per-packet (why?)

- send packet says “here are bytes j-k”
- ack says “received up to byte k”

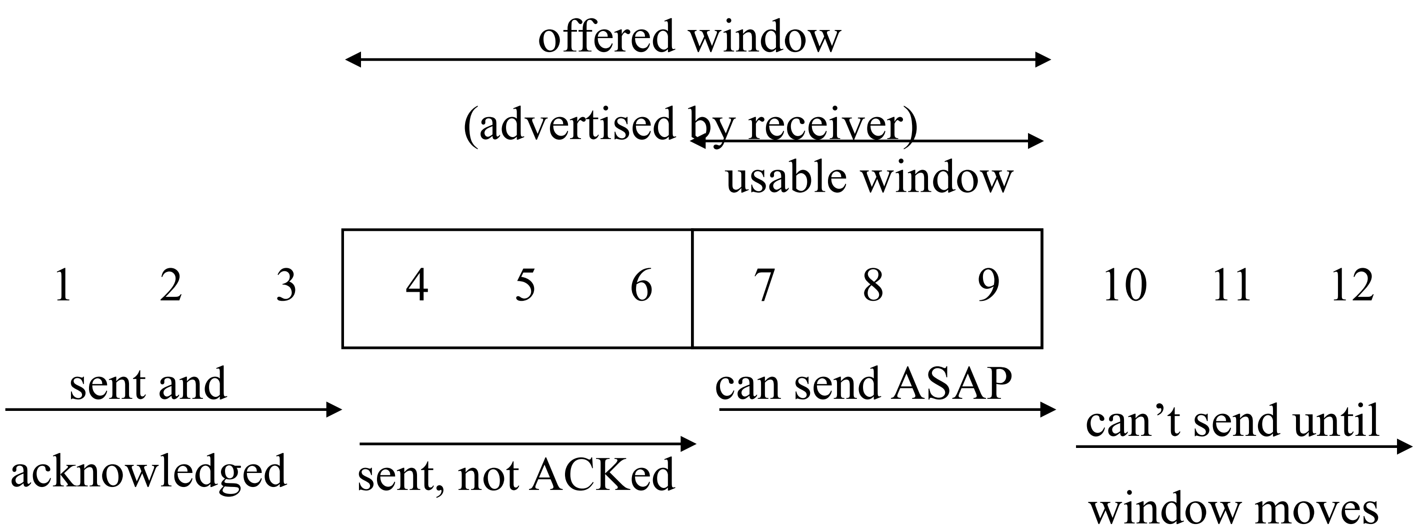
Send buffer >= send window

- can buffer writes in kernel before sending
- writer blocks if try to write past send buffer

Receive buffer >= receive window

- buffer acked data in kernel, wait for reads
- reader blocks if try to read past acked data

Visualizing the window



Left side of window advances when data is acknowledged.

Right side controlled by size of window advertisement

Flow Control

What if sender process is faster than receiver process?

- Data builds up in receive window
- if data is acked, sender will send more!
- If data is not acked, sender will retransmit!

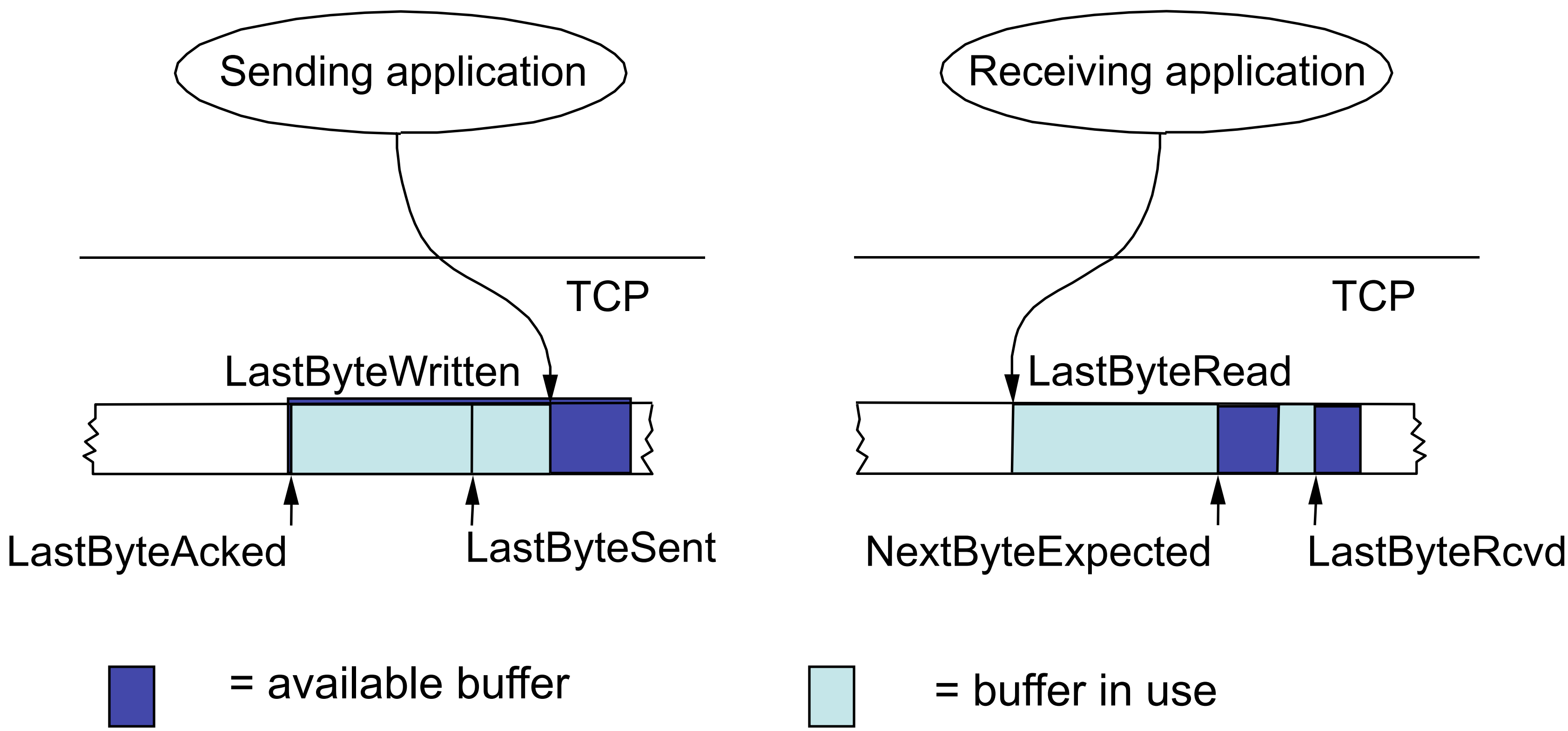
Sender must transmit data no faster than it can be consumed by the receiver

- Receiver might be a slow machine
- App might consume data slowly

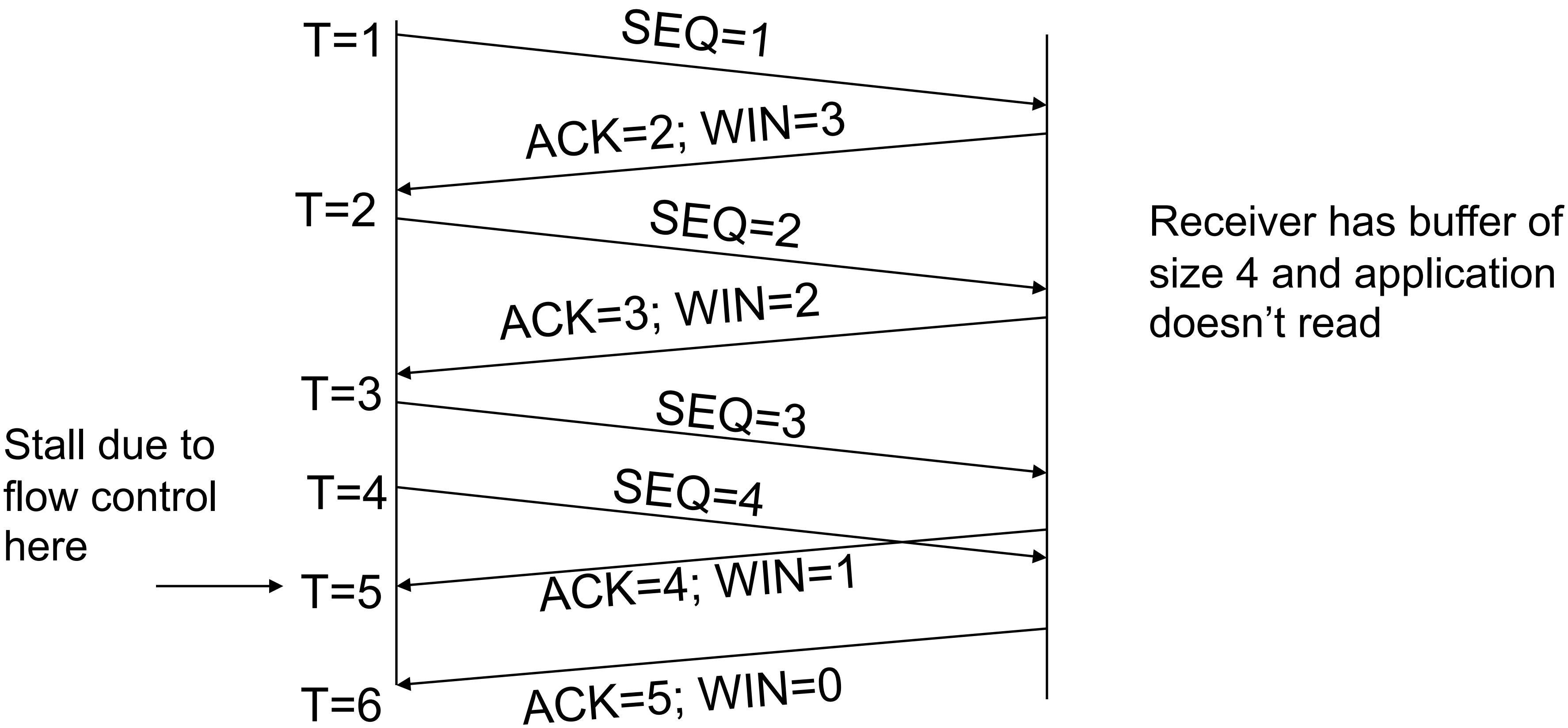
Sender sliding window <= free receiver buffer

- Advertised window = # of free bytes; if zero, stop

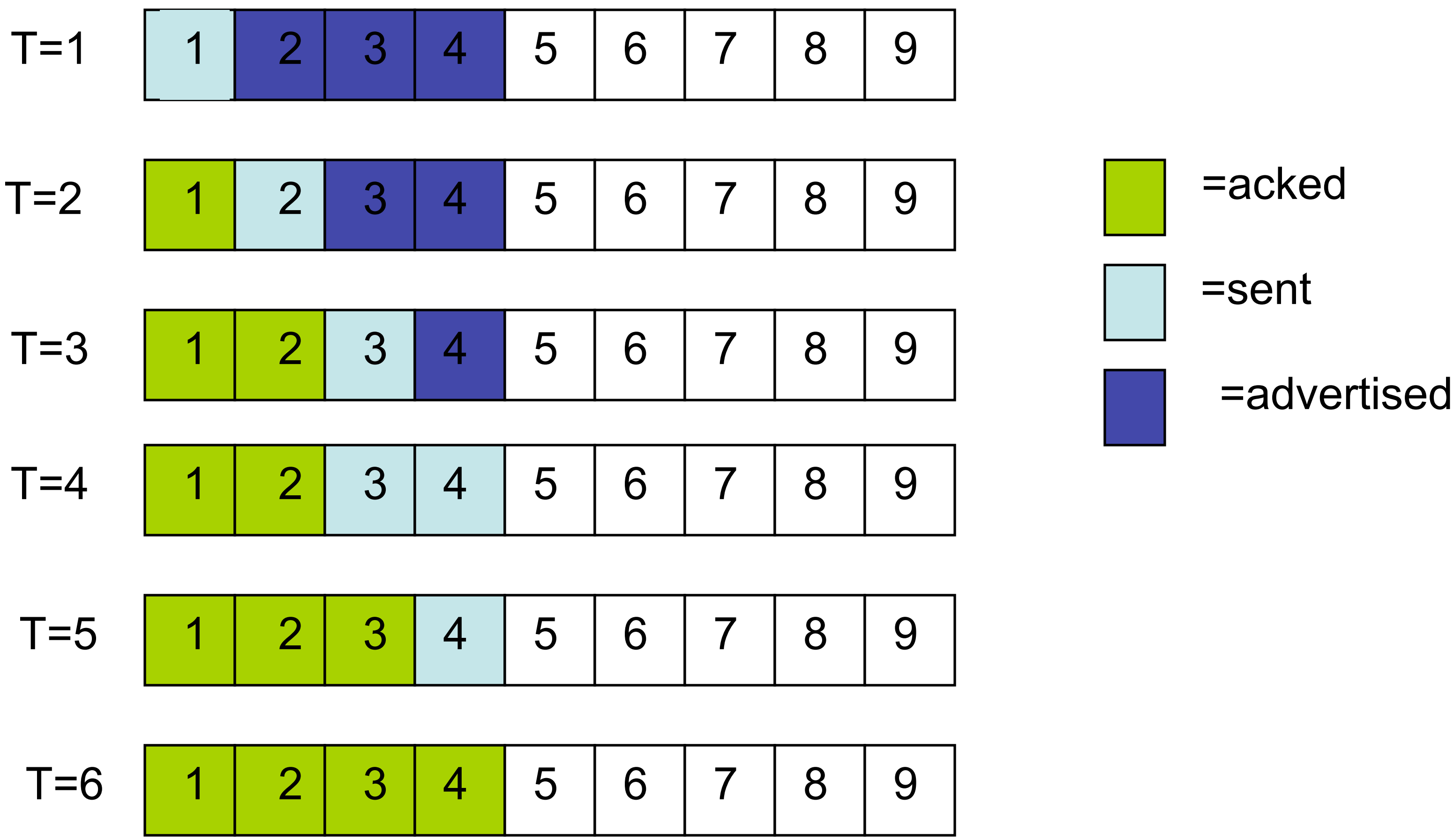
Sender and Receiver Buffering



Example – Exchange of Packets



Example – Buffer at Sender



How does sender know when to resume sending?

If receive window = 0, sender stops

- no data => no acks => no window updates

Sender periodically pings receiver with one byte packet

- receiver acks with current window size

Why not have receiver ping sender?

Should sender be greedy (I)?

Should sender transmit as soon as any space opens in receive window?

- Silly window syndrome
  - receive window opens a few bytes
  - sender transmits little packet
  - receive window closes

Solution (Clark, 1982): sender doesn't resume sending until window is half open

Should sender be greedy (II)?

App writes a few bytes; send a packet?

- Don't want to send a packet for every keystroke
- If buffered writes >= max segment size
- if app says "push" (ex: telnet, on carriage return)
- after timeout (ex: 0.5 sec)

Nagle's algorithm

- Never send two partial segments; wait for first to be acked, before sending next
- Self-adaptive: can send lots of tinygrams if network is being responsive

But (!) poor interaction with delayed acks (later)

TCP Connection Management

Setup

- assymetric 3-way handshake

Transfer

- sliding window; data and acks in both directions

Teardown

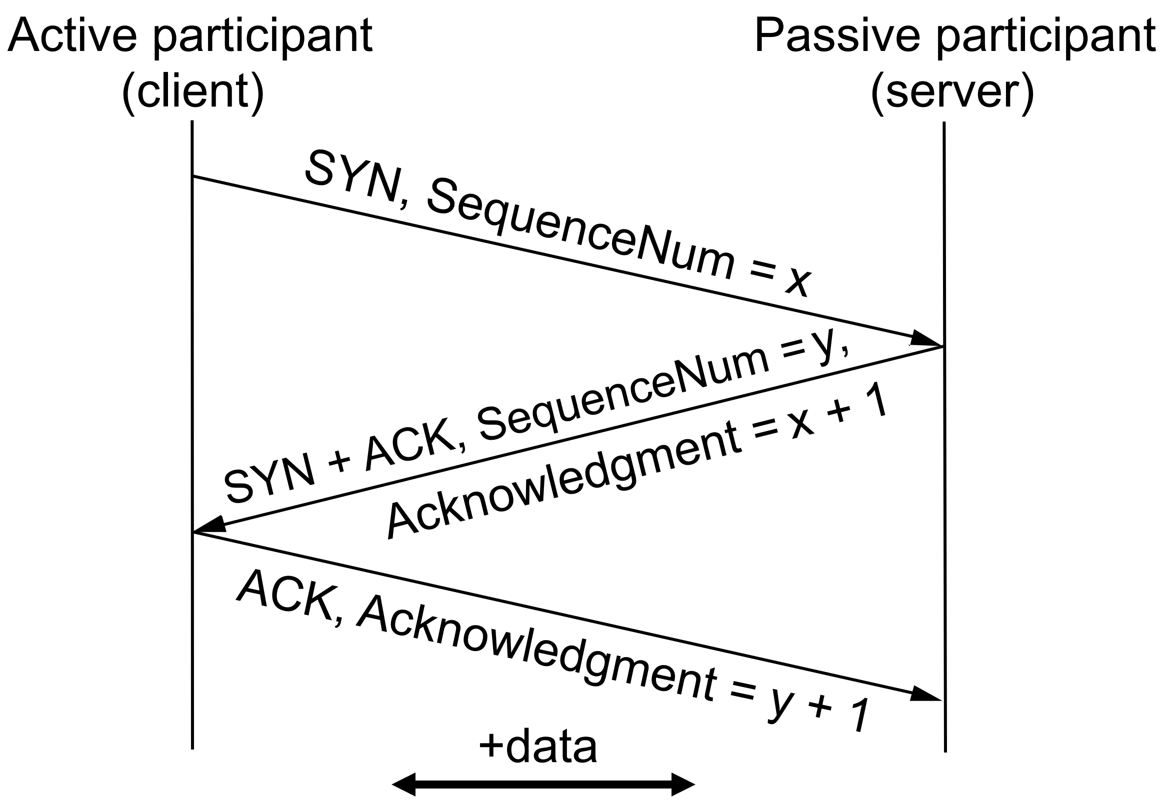
- symmetric 2-way handshake

Client-server model

- initiator (client) contacts server
- listener (server) responds, provides service

Three-Way Handshake

Opens both directions for transfer



Do we need 3-way handshake?

Allows both sides to

- allocate state for buffer size, state variables, ...
- calculate estimated RTT, estimated MTU, etc.

Helps prevent

- Duplicates across incarnations
- Intentional hijacking
  - random nonces => weak form of authentication

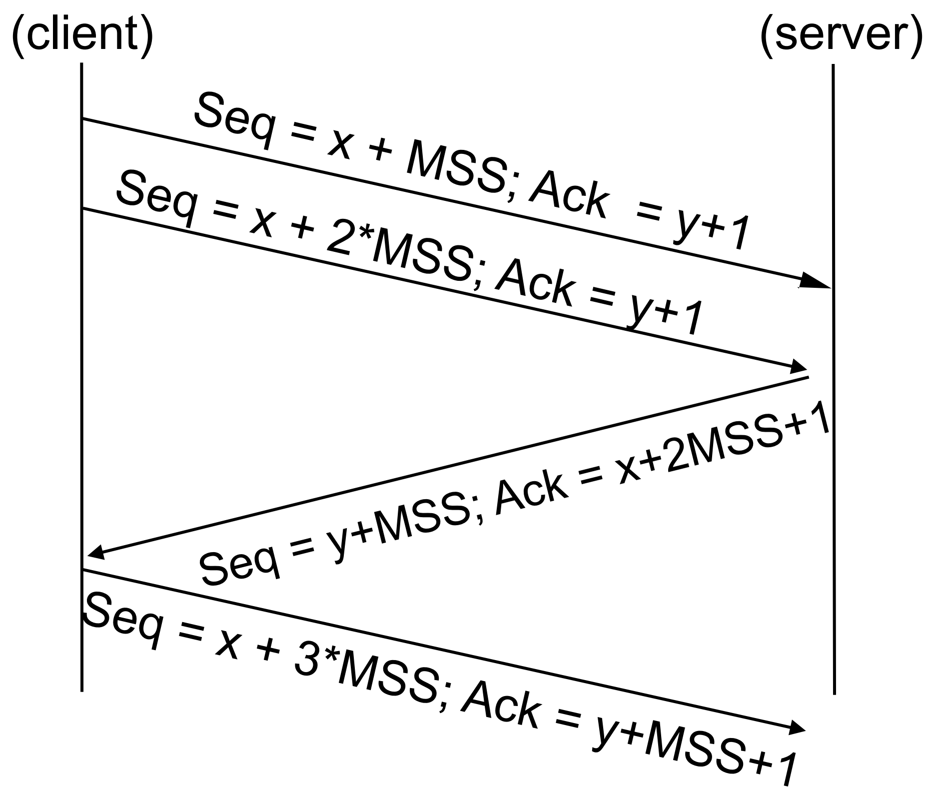
Short-circuit?

- Persistent connections in HTTP (keep connection open)
- Transactional TCP (save seq #, reuse on reopen)
- But congestion control effects dominate

TCP Transfer

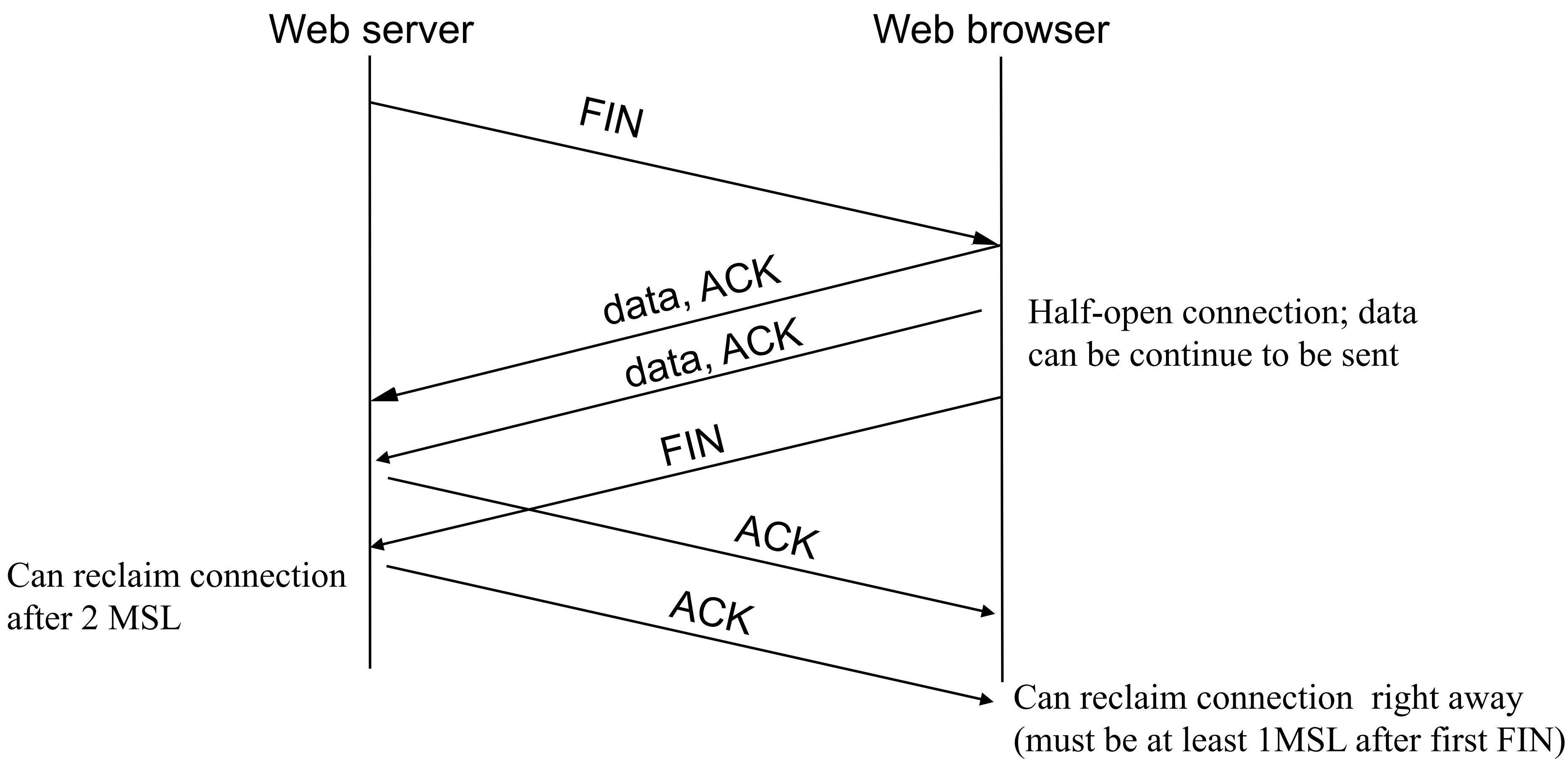
Connection is bi-directional

- acks can carry response data

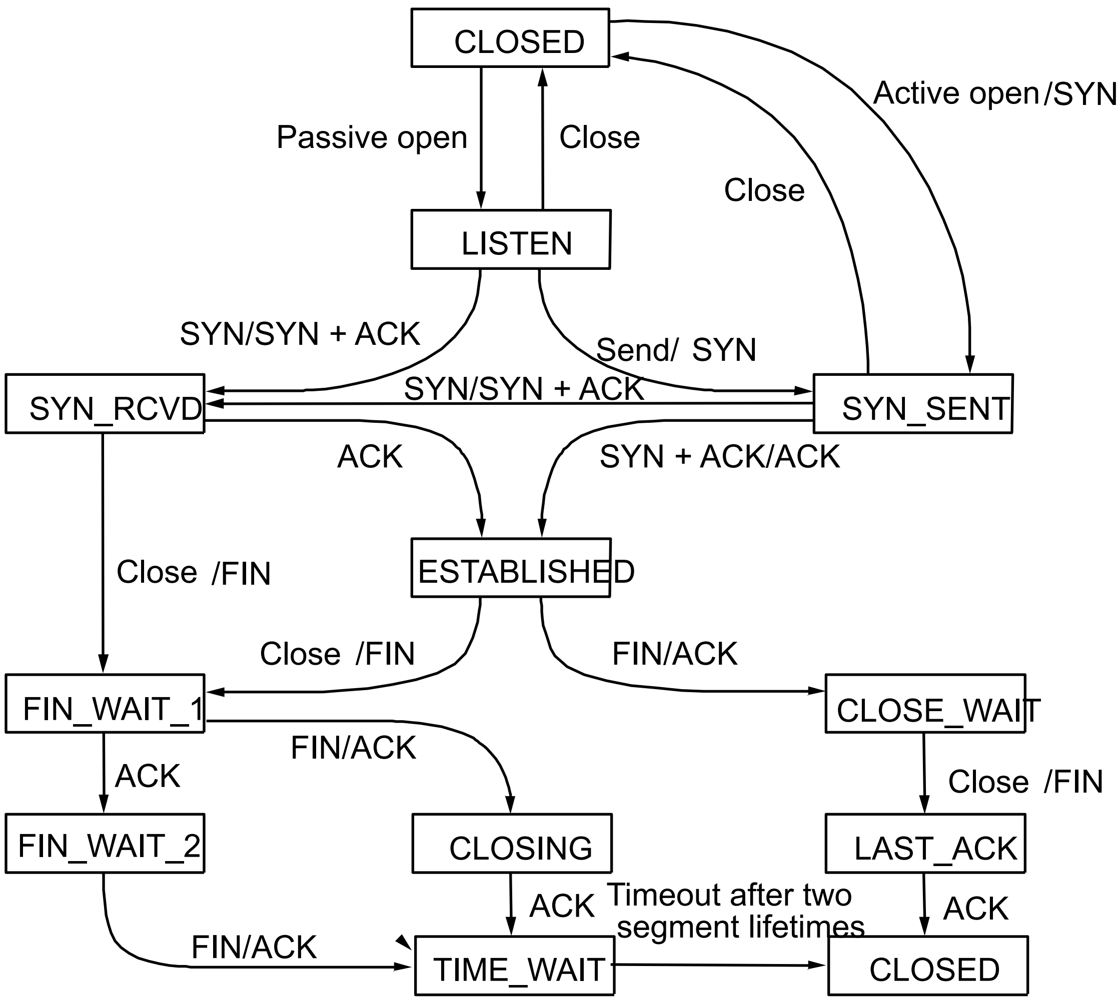


TCP Connection Teardown

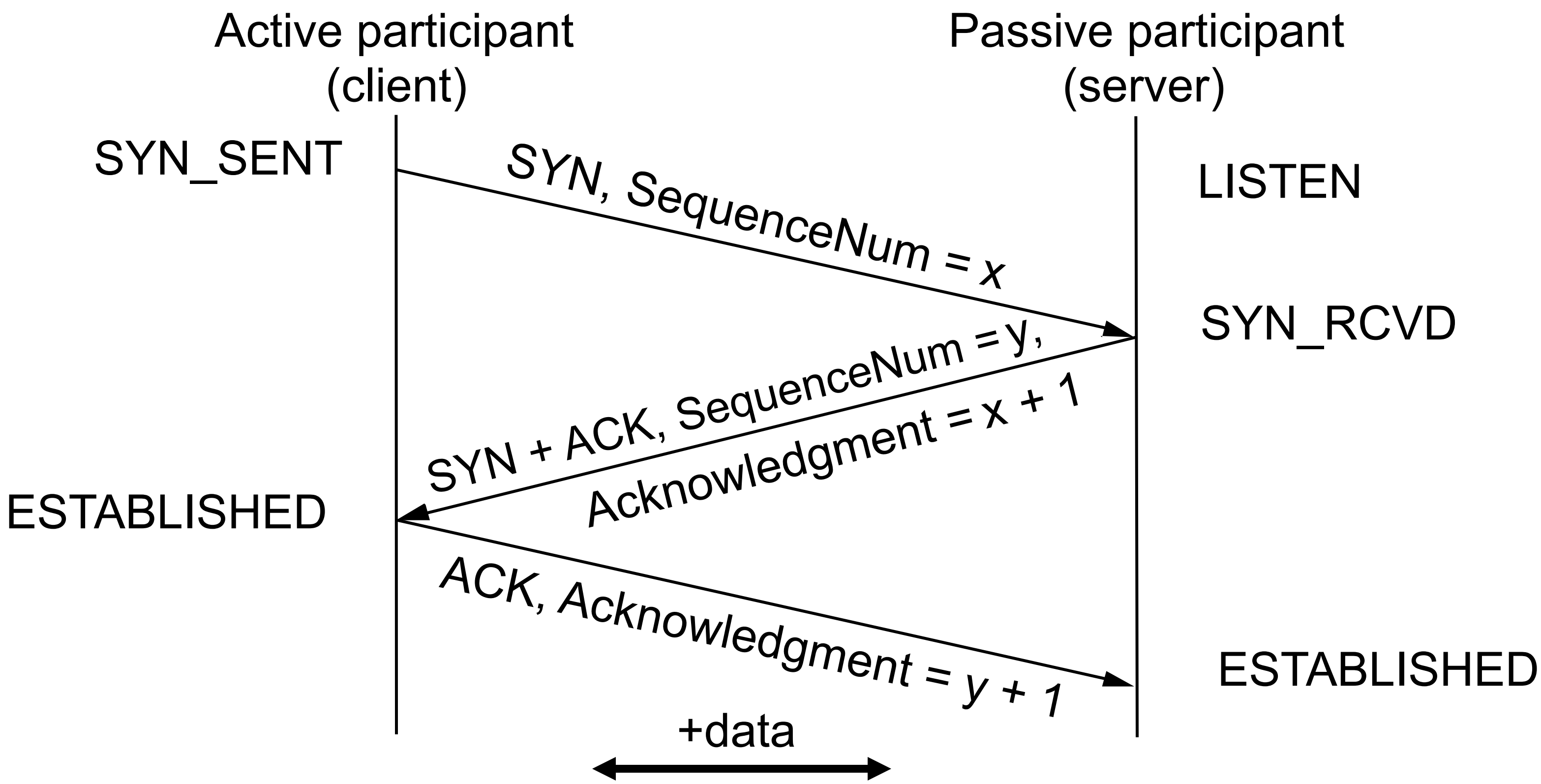
Symmetric: either side can close connection (or RST!)



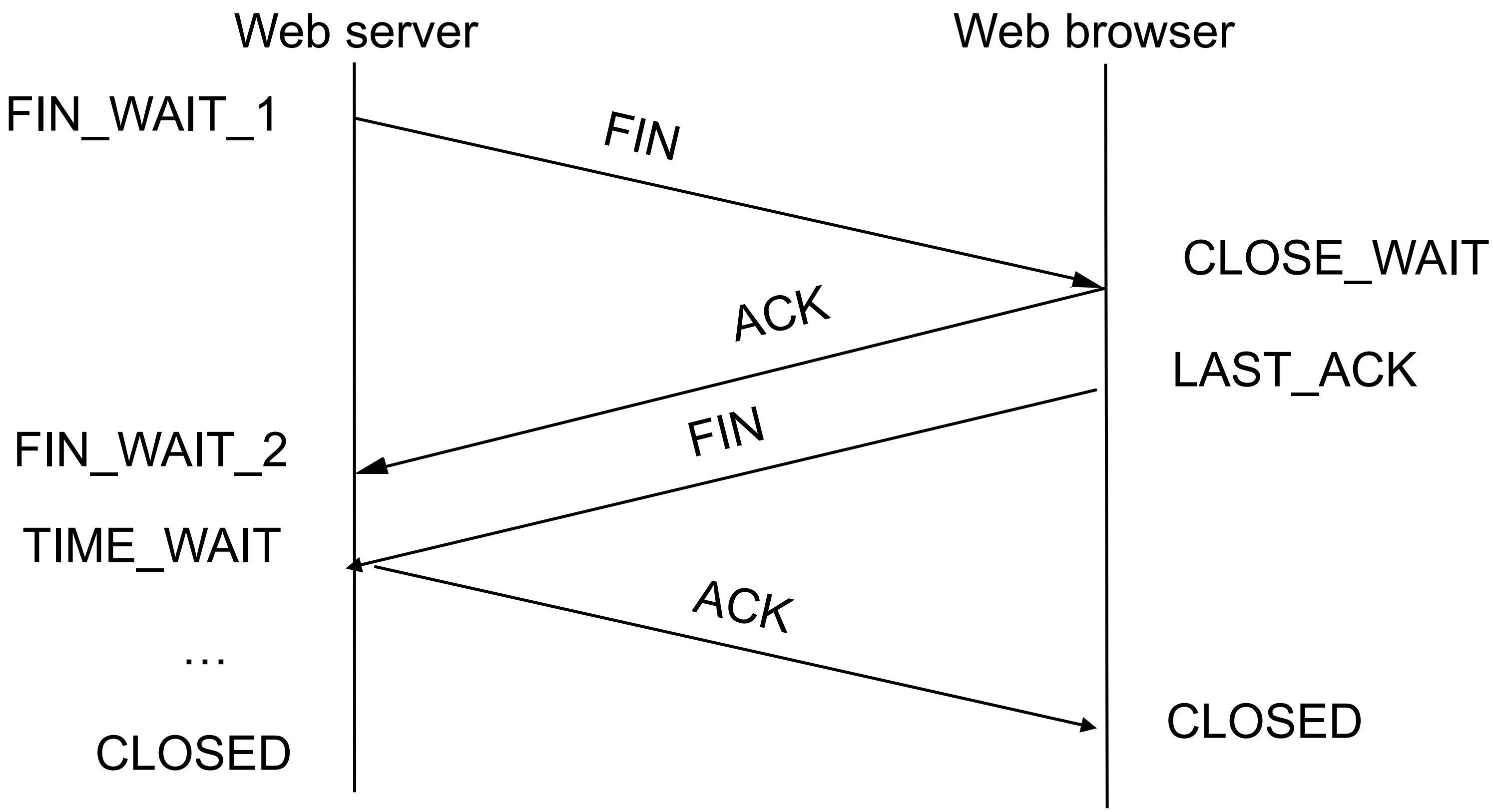
TCP State Transitions



TCP Connection Setup, with States



TCP Connection Teardown



The TIME_WAIT State

We wait 2MSL (two times the maximum segment lifetime of 60 seconds) before completing the close

Why?

ACK might have been lost and so FIN will be resent

Could interfere with a subsequent connection

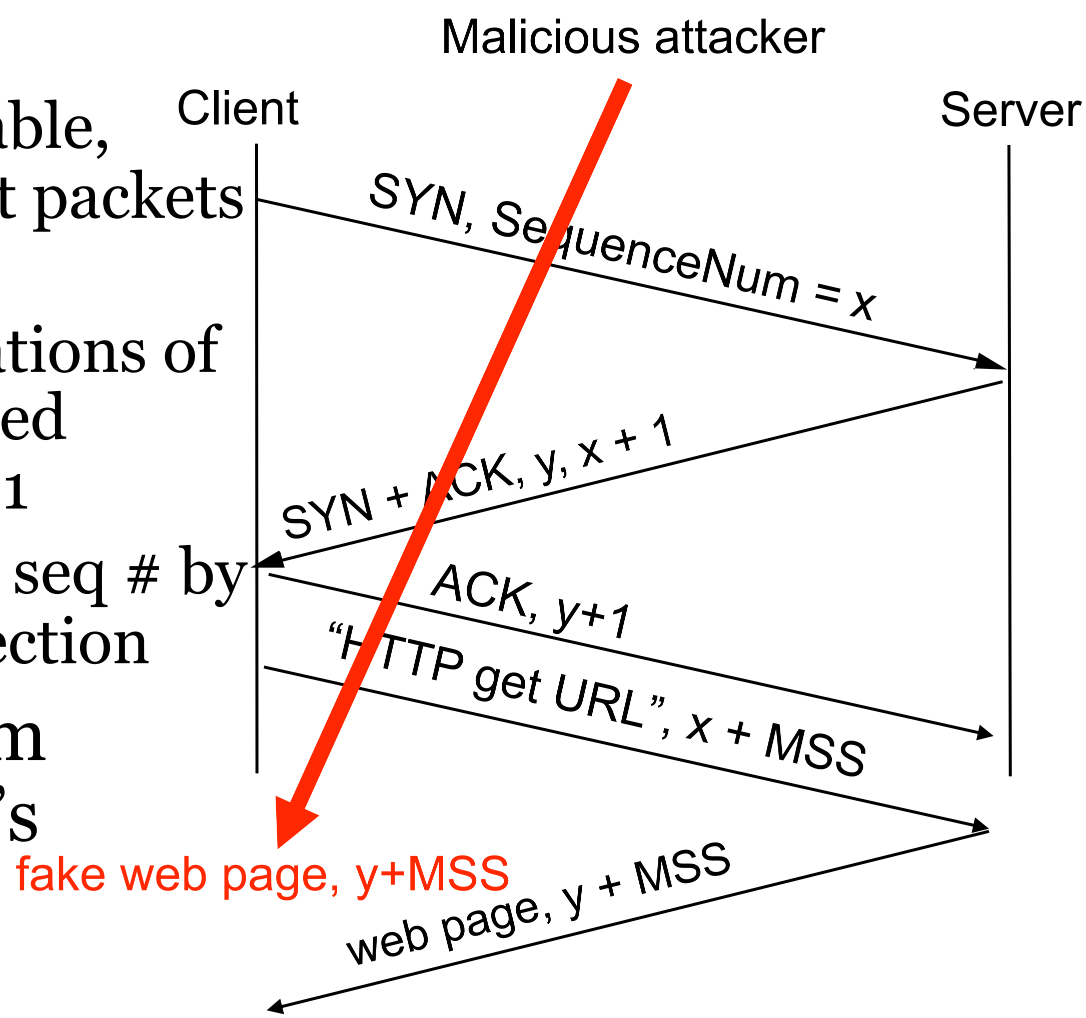
TCP Handshake in an Uncooperative Internet

TCP Hijacking

- if seq # is predictable, attacker can insert packets into TCP stream
- many implementations of TCP simply bumped previous seq # by 1
- attacker can learn seq # by setting up a connection

Solution: use random initial sequence #'s

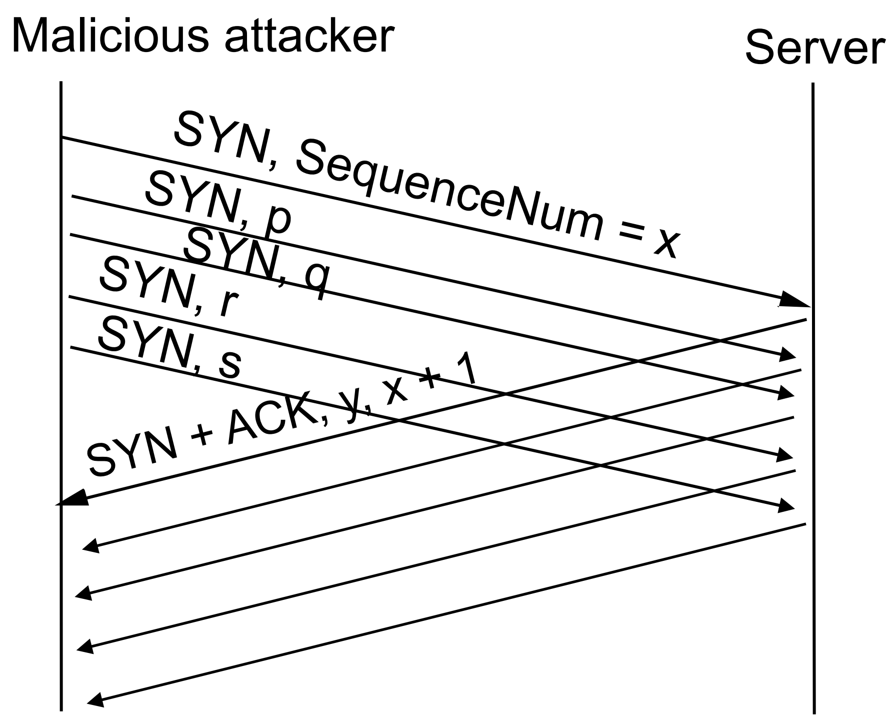
- weak form of authentication



TCP Handshake in an Uncooperative Internet

TCP SYN flood

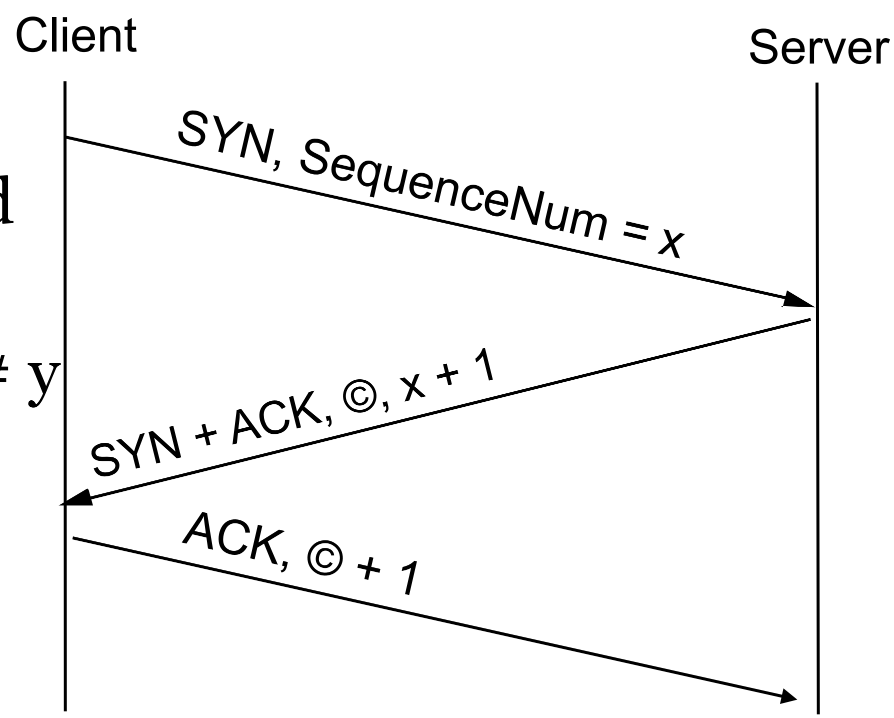
- server maintains state for every open connection
- if attacker spoofs source addresses, can cause server to open lots of connections
- eventually, server runs out of memory



TCP SYN cookies

Solution: SYN cookies

- Server keeps no state in response to SYN; instead makes client store state
- Server picks return seq # y = © that encrypts x
- Gets © +1 from sender; unpacks to yield x



Can data arrive before ACK?

How can TCP choose segment size?

- Pick LAN MTU as segment size?
  - LAN MTU can be larger than WAN MTU
  - E.g., Gigabit Ethernet jumbo frames
- Pick smallest MTU across all networks in Internet?
  - Most traffic is local!
    - Local file server, web proxy, DNS cache, ...
  - Increases packet processing overhead
- Discover MTU to each destination? (IP DF bit)
- Guess?

Layering Revisited

IP layer "transparent" packet delivery

- Implementation decisions affect higher layers (and vice versa)
  - Fragmentation => reassembly overhead
    - path MTU discovery
  - Packet loss => congestion or lossy link?
    - link layer retransmission
  - Reordering => packet loss or multipath?
    - router hardware tries to keep packets in order
  - FIFO vs. active queue management

IP Packet Header Limitations

Fixed size fields in IPv4 packet header

- source/destination address (32 bits)
  - limits to ~ 4B unique public addresses; about 600M allocated
  - NATs map multiple hosts to single public address
- IP ID field (16 bits)
  - limits to 65K fragmented packets at once => 100MB in flight?
  - in practice, fewer than 1% of all packets fragment
- Type of service (8 bits)
  - unused until recently; used to express priorities
- TTL (8 bits)
  - limits max Internet path length to 255; typical max is 30
- Length (16 bits)
  - Much larger than most link layer MTU's

TCP Packet Header Limitations

Fixed size fields in TCP packet header

- seq #/ack # -- 32 bits (can't wrap within MSL)
  - T1 ~ 6.4 hours; OC-192 ~ 3.5 seconds
- source/destination port # -- 16 bits
  - limits # of connections between two machines (NATs)
  - ok to give each machine multiple IP addresses
- header length
  - limits # of options
- receive window size -- 16 bits (64KB)
  - rate = window size / delay
  - Ex: 100ms delay => rate ~ 5Mb/sec
  - RFC 1323: receive window scaling
  - Defaults still a performance problem

HTTP on TCP

How do we reduce the # of messages?

Delayed ack: wait for 200ms for reply or another pkt arrival

TCP RST from web server