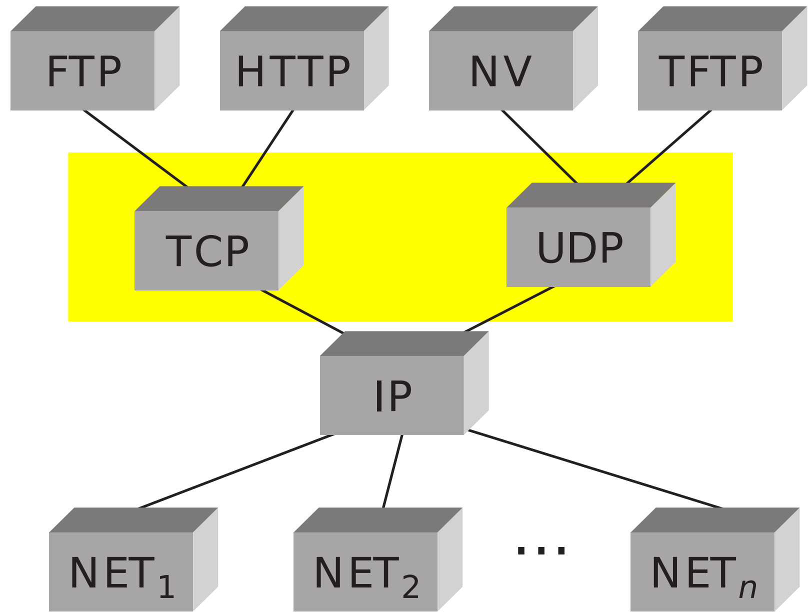
Overview

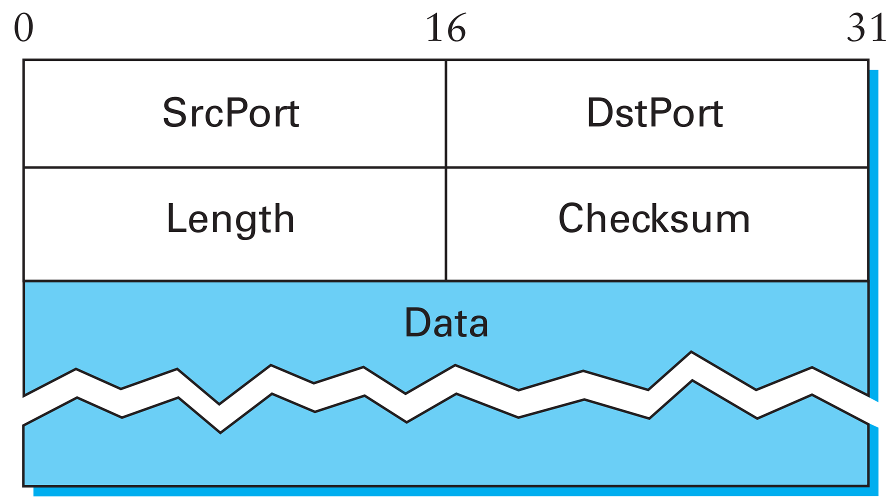
- User datagram protocol (UDP)
- Packet checksums
- Reliability: sliding window
- TCP connection setup
- TCP windows, retransmissions, and acknowledgments

Transport Protocol Review



- **Transport protocols sit on top of the network layer (IP)**
- **Can provide:**
  - Application-level multiplexing ("ports")
  - Error detection, reliability, etc.

UDP – *user datagram protocol*



- **Unreliable and unordered datagram service**
- **Adds multiplexing, checksum on whole packet**
- **No flow control, reliability, or order guarantees**
- **Endpoints identified by ports**
- **Checksum aids in error detection**

Error detection

- **Transmission errors definitely happen**
  - Cosmic rays, radio interference, etc.
  - If error probability is 2^{-30}, that's 1 error per 128 MB!
- **Some link-layer protocols provide error detection**
  - But UDP/IP must work over many link layers
  - Not all links on a path may have error detection
  - Moreover, recall end-to-end argument!
    Need end-to-end check
- **UDP detects errors with a *checksum***
  - Compute small checksum value, like a hash of the packet
  - If packet corrupted in transit, checksum likely to be wrong
  - Similar checksum on IP header, but doesn't cover payload

Checksum algorithms

- **Good checksum algorithms**
  - Should detect errors that are likely to happen
    (E.g., should detect any single bit error)
  - Should be efficient to compute
- **IP, UDP, and TCP use 1s complement sum:**
  - Set checksum field to 0
  - Sum all 16-bit words in pkt
  - Add any carry bits back in
    (So `0x8000` + `0x8000` = `0x0001`)
  - Flip bits (`sum = ~sum;`) to get checksum (`0xf0f0` → `0x0f0f`),
    Unless sum is `0xffff`, then checksum just `0xffff`
  - To check: Sum whole packet (including sum), should get `0xffff`

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

UDP pseudo-header

```
0                   1                   2                   3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
```

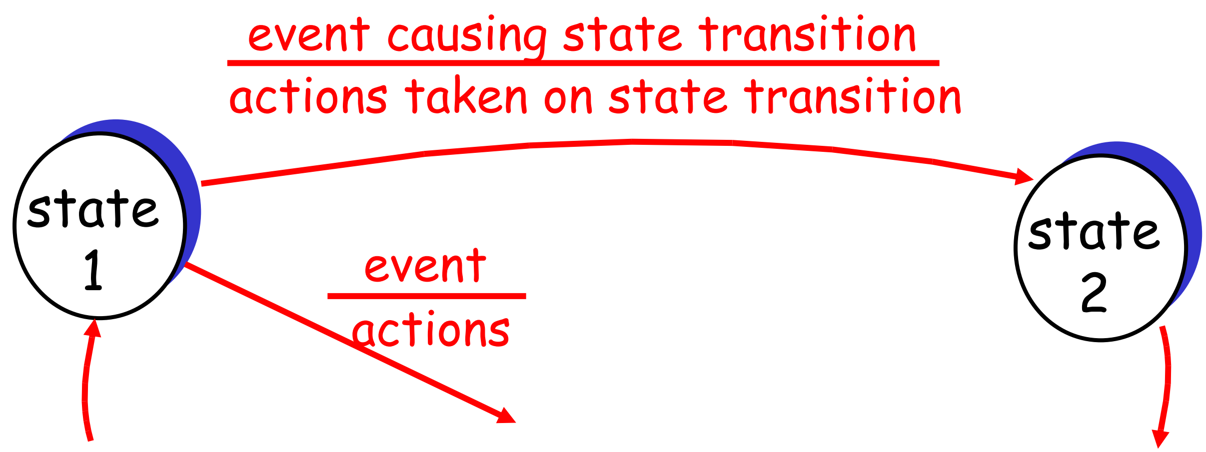
Source IP address		
Destination IP address		
Zero	Protocol (=17)	UDP length
Source Port		Destination Port
UDP Payload		

- **Checksum actually includes "pseudo-header"**
  - Not transmitted, just pre-pended to compute checksum
  - Ensures UDP checksum includes IP addresses
- **Trick question: Is UDP a layer on top of IP?**

How good is UDP/IP checksum?

- $+$ **Very fast to compute in software**
  - Same implementation works on big & little endian CPUs
- $-$ **16 bits is not very long (misses $1/2^{16}$ errors)**
- $+$ **Checksum does catch any 1-bit error**
- $-$ **But not any two-bit error**
  - E.g., increment one word ending 0, decrement one ending 1
- $-$ **Checksum also optional on UDP**
  - All 0s means no checksum calculated
  - If checksum word gets wiped to 0 as part of error, bad news
- **Good thing most link layers have stronger checksums**
- **Next problem: If you discard bad packets, how to ensure reliable delivery? (E.g., stop & wait lab 1)**

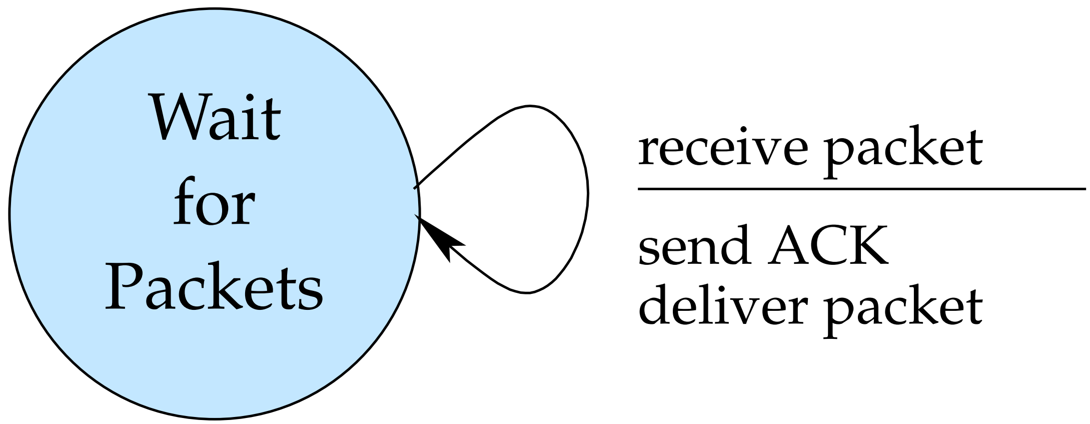
Finite State Machines



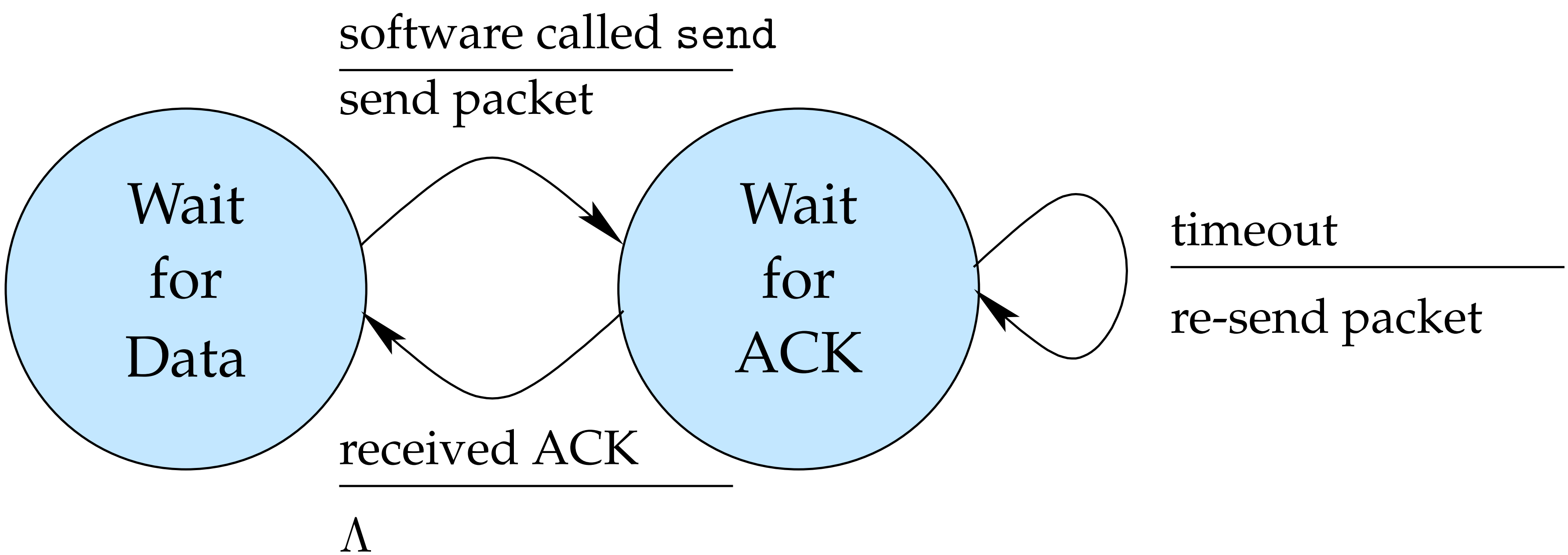
- **Represent protocols using state machines**
  - Sender and receiver each have a state machine
  - Start in some initial state
  - Events cause each side to select a state transition
- **Transition specifies action taken**
  - Specified as events/actions
  - E.g., software calls `send`/put packet on network
  - E.g., packet arrives/send acknowledgment

Stop and wait FSMs

- **Receiver FSM:**

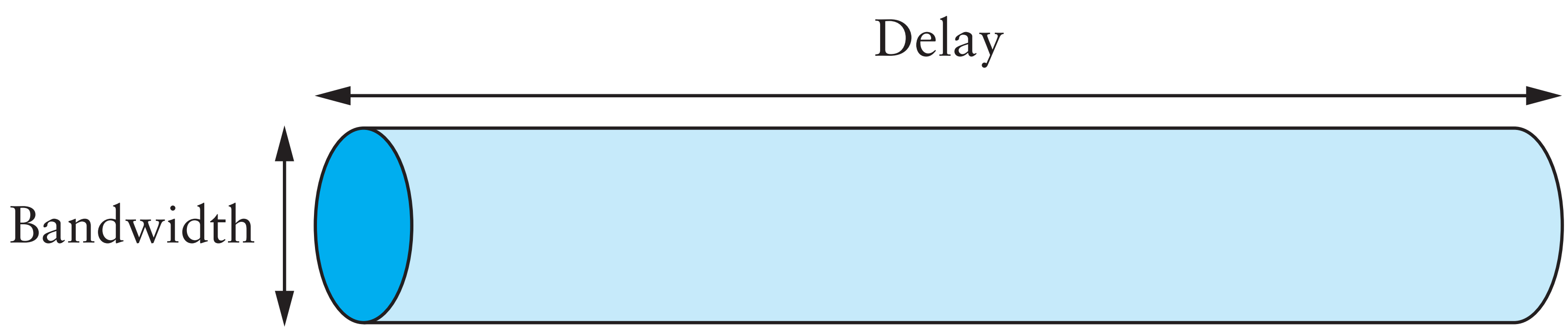


- **Sender FSM:**

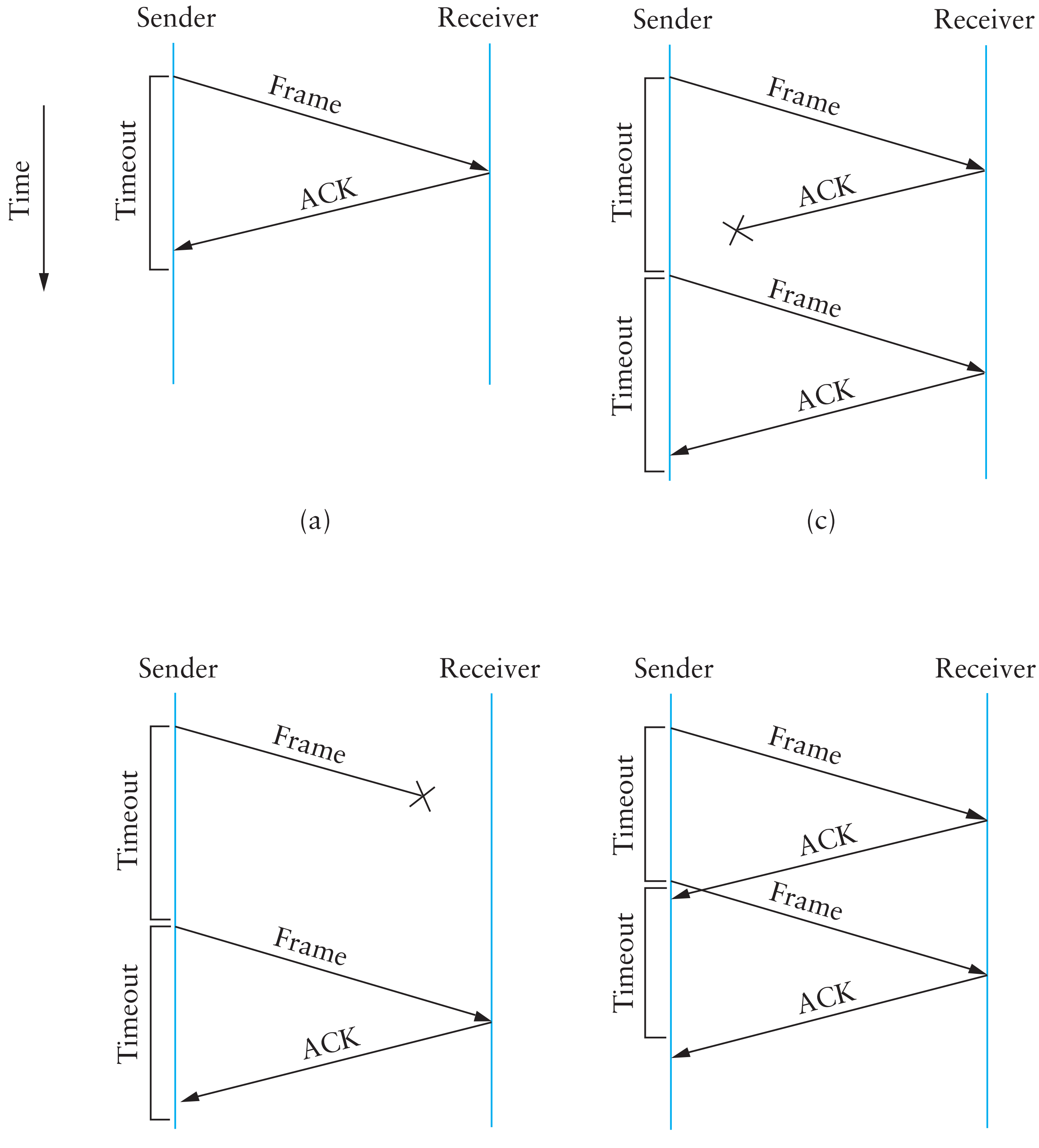


Problems with Stop and Wait

- **Might duplicate packet… how?**
- **Can't keep pipe full**



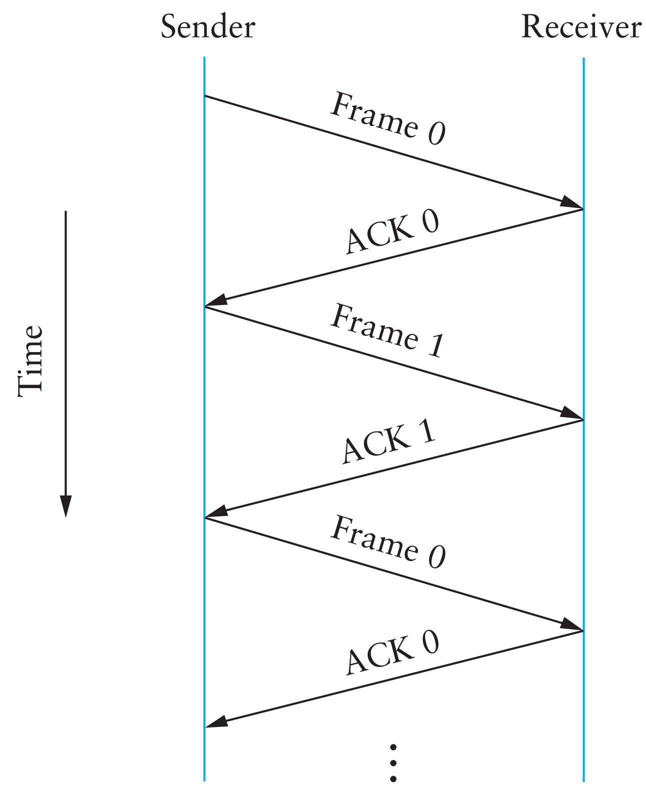
  - For full utilization want # bytes in flight $\geq$ bandwidth$\times$delay
    (But don't want to overload the network, either)



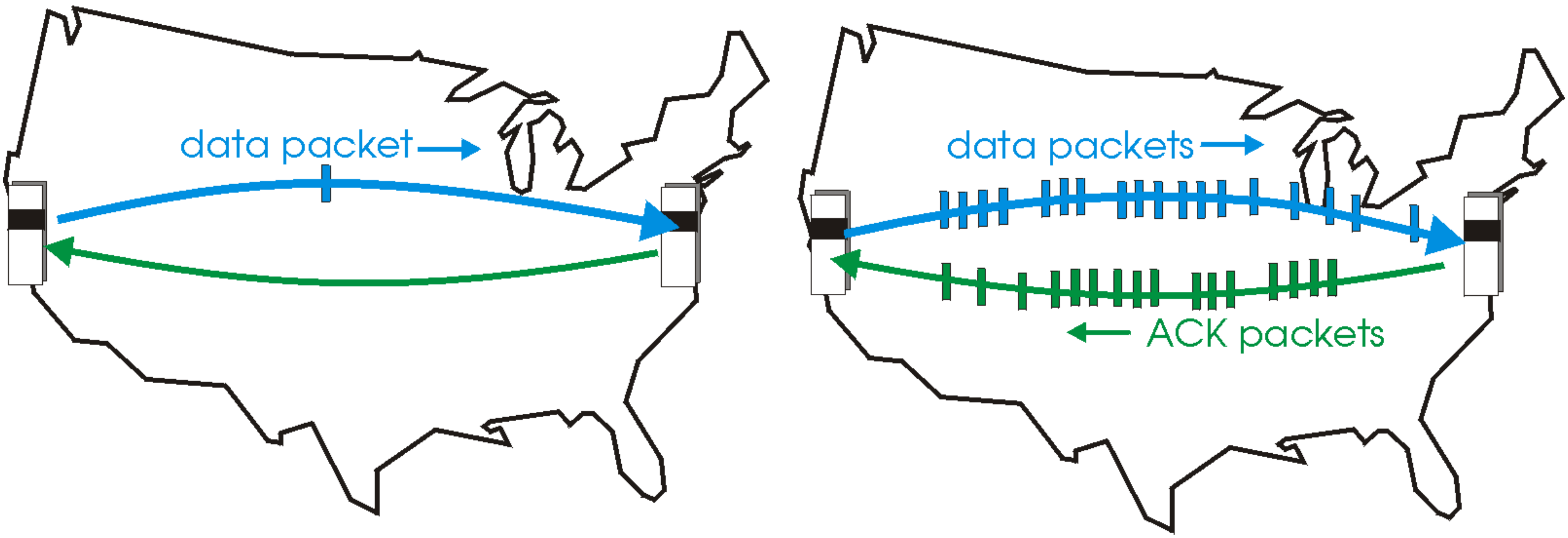
Sender
Receiver
Time
Timeout
Frame
ACK
(a)
Sender
Receiver
Timeout
Frame
ACK
Timeout
Frame
ACK
(c)
Sender
Receiver
Timeout
Frame
Timeout
Frame
ACK
(b)
Sender
Receiver
Timeout
Frame
ACK
Timeout
Frame
ACK
(d)

Duplicates

- **Solve problem with 1-bit counter**
  - Place in both Frame and ACK
  - Receiver knows if duplicate of last frame
  - Sender won’t interpret duplicate old ACK as for new packet
- **This still requires some simplifying assumptions**
  - Network itself might duplicates packets
  - Packet might be heavily delayed and reordered
  - Assume these don’t happen for now
  - But usually prefer weaker assumption: *Maximum Segment Lifetime* (MSL)



Effect of RTT on performance



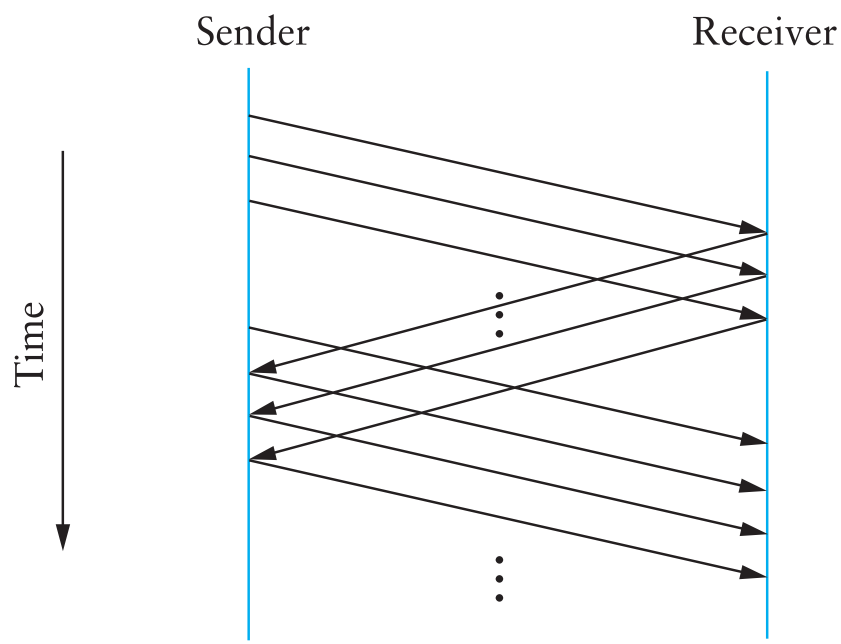
(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

- **Stop & wait goodput depends on Round-Trip Time (RTT)**
  - Capped by packet size/RTT regardless of underlying link b/w
- **Need *pipelineing* for goodput to approach link throughput**

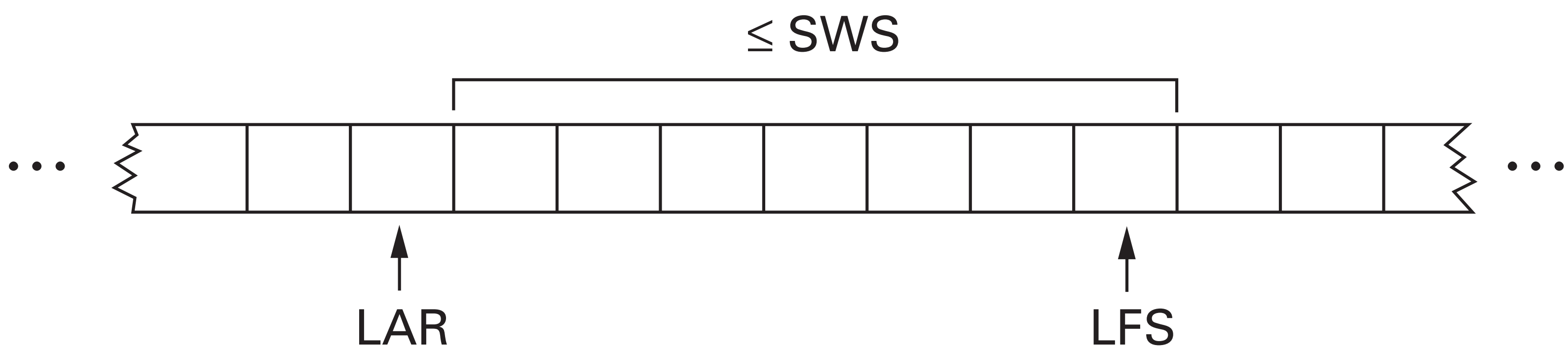
Sliding window protocol

- **Addresses problem of keeping the pipe full**
  - Generalize previous protocol with $>$ 1-bit counter
  - Allow multiple outstanding (unACKed) frames
  - Upper bound on unACKed frames, called window



SW sender

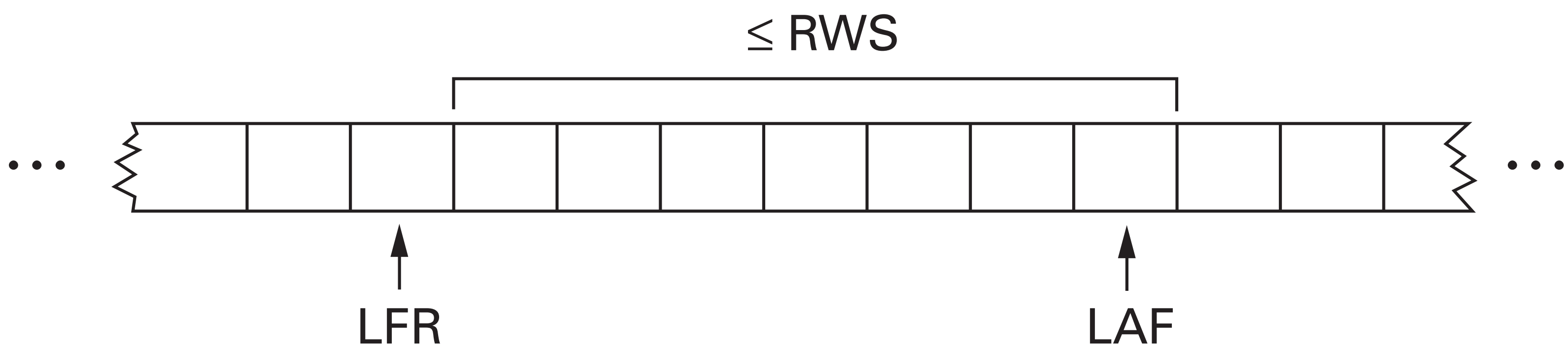
- **Assign sequence number to each frame (SeqNum)**
- **Maintain three state variables:**
  - Send Window Size (SWS)
  - Last Acknowledgment Received (LAR)
  - Last Frame Sent (LFS)



- **Maintain invariant: LFS $-$ LAR $\leq$ SWS**
- **Advance LAR when ACK arrives**
- **Buffer up to SWS frames**

SW receiver

- **Maintain three state variables**
  - Receive Window Size (RWS)
  - Largest Acceptable Frame (LAF)
  - Last Frame Received (LFR)



- **Maintain invariant: LAF $-$ LFR $\leq$ RWS**

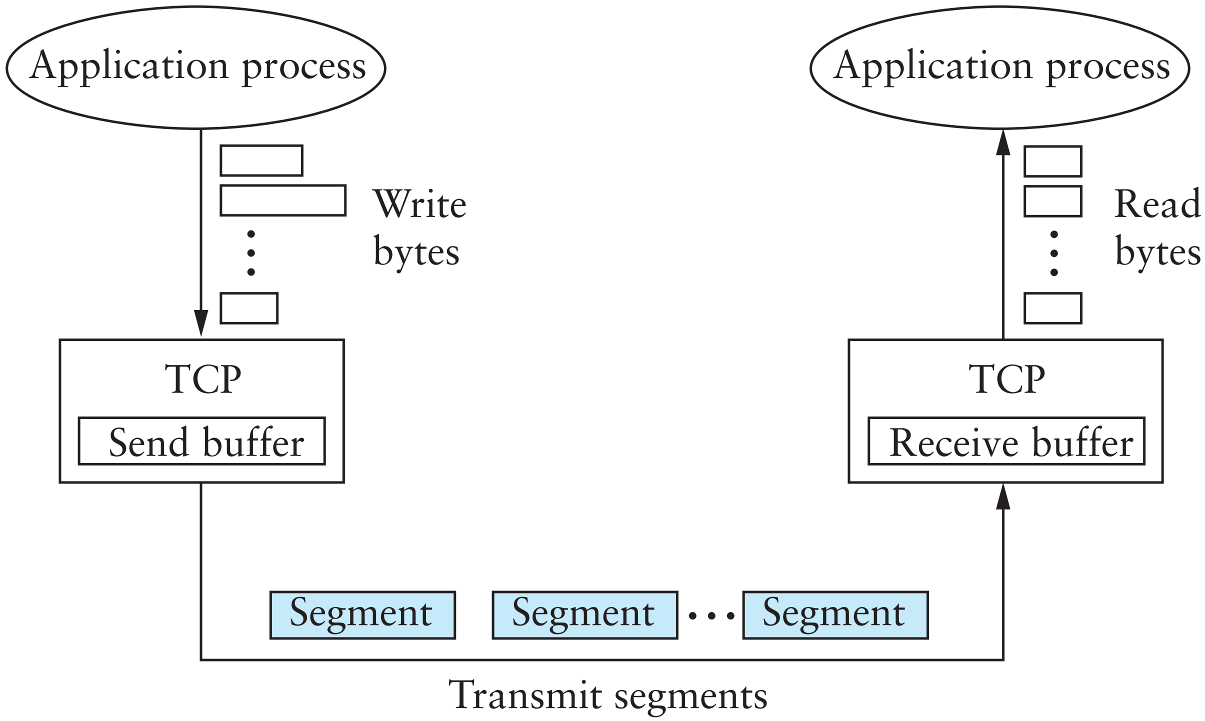
SW receiver, continued

- **When frame # SeqNum arrives:**
  - if LFR $<$ SeqNum $\leq$ LFA accept
  - if SeqNum $\leq$ LFR or SeqNum $>$ LFA discarded
- **Send *cumulative* ACKs**
  - I.e., ACK n means received all packets w. SeqNo $\leq n$
  - E.g., if received packets $1, 2, 3, 5$, must ACK 3
- **Or can alternatively use TCP-style ACKs, which specify first pkt. *not* received**
  - E.g., if received packets $1, 2, 3, 5$, must ACK 4, not 3
  - Note Labs 1 and 2 use TCP-style cumulative ACKs

Sequence number space

- **How big should RWS be?**
  - At least 1. No bigger than SWS
    (Don't accept a packet the sender shouldn't have sent)
- **How many distinct sequence numbers needed?**
- **If RWS=1, need at least SWS+1**
  - This protocoal is often called "Go-Back-N"
- **If RWS=SWS, need at least 2SWS**
  - Otherwise, bad news if ACKs are lost
  - Sender may retransmit a window that was already received
  - Receiver will think retransmissions are from next window
- **Generally need at least RWS+SWS**
  - RWS packets in unknown state (ACK may/may not be lost)
  - SWS packets in flight must not overflow sequence space

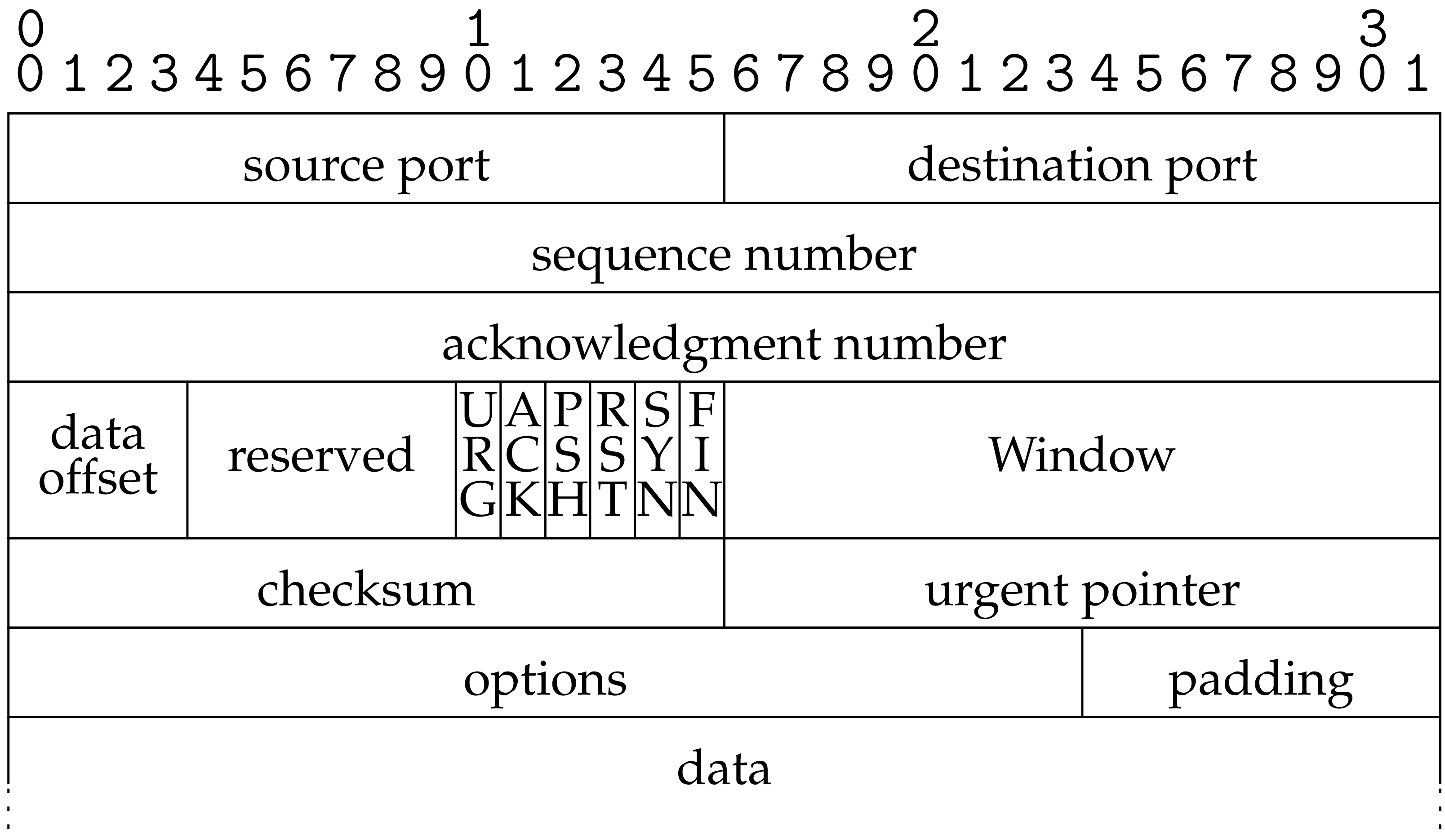
High-level view of TCP



- **Full duplex, connection-oriented byte stream**
- **Flow control**
  - If one end stops reading, writes at other eventuall block/fail
- **Congestion control**
  - Keeps sender from overrunning network [more next lecture]

2-minute stretch

TCP segment


0 1 2 3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
source port
destination port
sequence number
acknowledgment number
data offset
reserved
URG
ACK
PSH
RST
SYN
FIN
Window
checksum
urgent pointer
options
padding
data

TCP fields

- **Ports**
- **Seq no. – segment position in byte stream**
  - Unlike Lab 1, sequence #s corresponds to *bytes*, not packets
- **Ack no. – seq no. sender expects to receive next**
- **Data offset – # of 4-byte header & option words**
- **Window – willing to receive**
  - Lets receiver limit SWS (possibly to 0) for flow control (more in a few slides)
- **Checksum**
- **Urgent pointer**

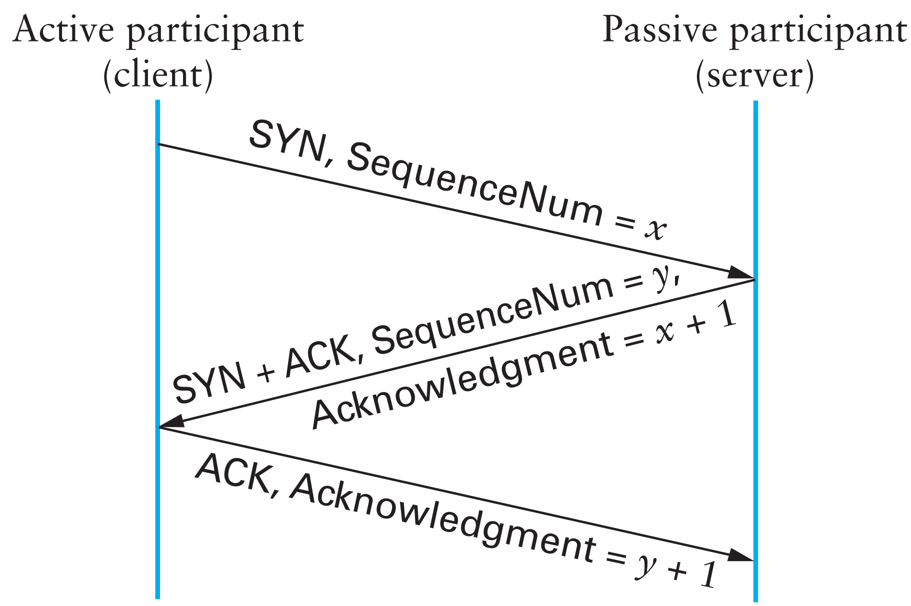
TCP Flags

- URG – urgent data present
- ACK – ack no. valid (all but first segment)
- PSH – push data up to application immediately
- RST – reset connection
- SYN – “synchronize” establishes connection
- FIN – close connection

A TCP Connection (no data)

```
orchard.48150 > essex.discard:
        S 1871560457:1871560457(0) win 16384
essex.discard > orchard.48150:
        S 3249357518:3249357518(0) ack 1871560458 win 17376
orchard.48150 > essex.discard: . ack 1 win 17376
orchard.48150 > essex.discard: F 1:1(0) ack 1 win 17376
essex.discard > orchard.48150: . ack 2 win 17376
essex.discard > orchard.48150: F 1:1(0) ack 2 win 17376
orchard.48150 > essex.discard: . ack 2 win 17375
```

Connection establishment



- **Need SYN packet in each direction**
  - Typically second SYN also acknowledges first
  - Supports "simultaneous open," seldom used in practice
- **If no program listening: server sends RST**
- **If server backlog exceeded: ignore SYN**
- **If no SYN-ACK received: retry, timeout**

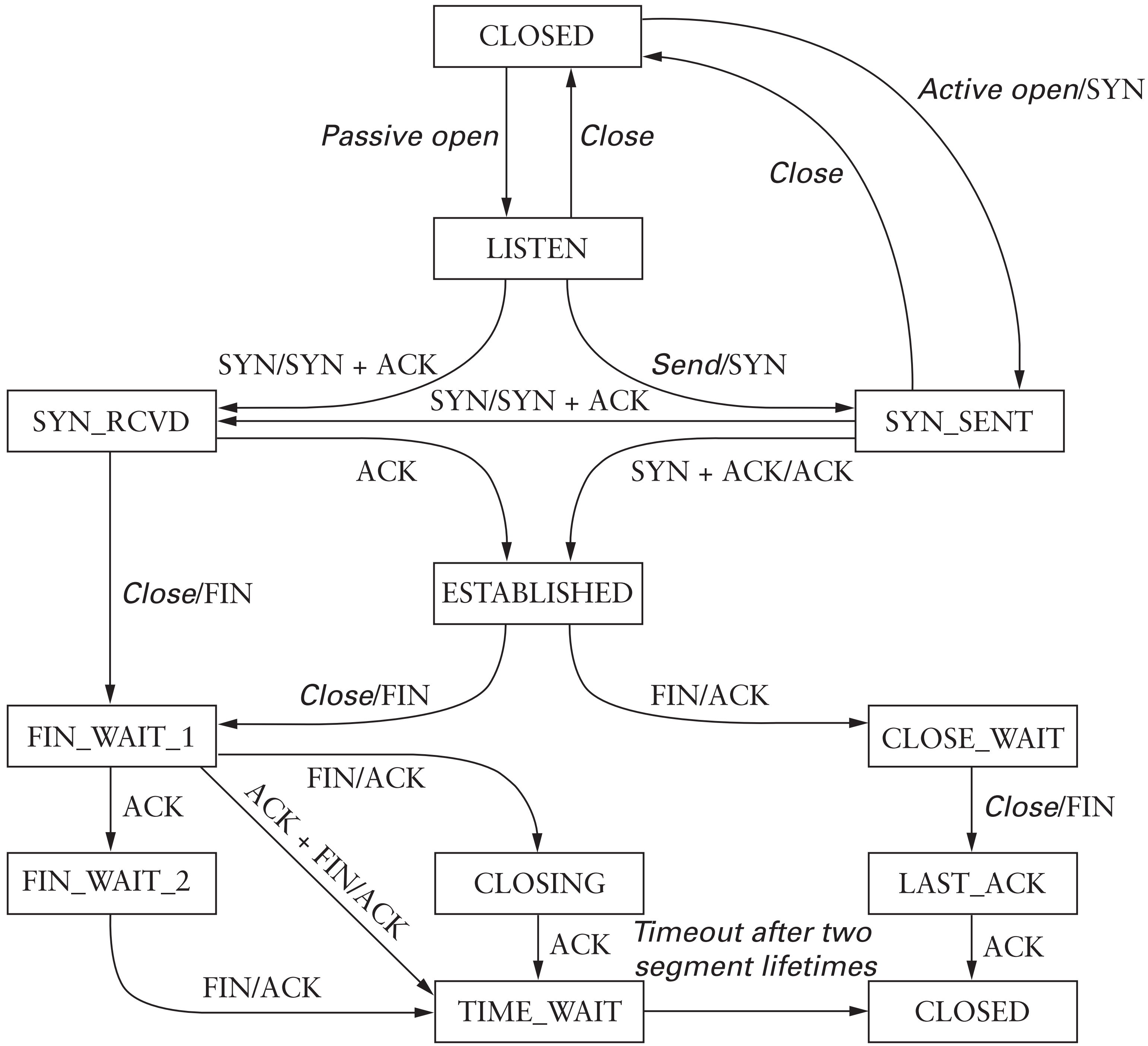
Connection termination

- **FIN bit says no more data to send**
  - Caused by close or shutdown on sending end
  - Both sides must send FIN to close a connection
- **Typical close:**
  - $A \rightarrow B$: FIN, seq S_A, ack S_B
  - $B \rightarrow A$: ack $S_A + 1$
  - $B \rightarrow A$: FIN, seq S_B, ack $S_A + 1$
  - $A \rightarrow B$: ack $S_B + 1$
- **Can also have simultaneous close**
- **After last message, can A and B forget about closed socket?**

TIME_WAIT

- **Problems with closed socket**
  - What if final ack is lost in the network?
  - What if the same port pair is immediately reused for a new connection? (Old packets might still be floating around.)
- **Solution: “active” closer goes into TIME_WAIT**
  - Active close is sending FIN before receiving one
  - After receiving ACK and FIN, keep socket around for 2MSL (twice the “maximum segment lifetime”)
- **Can pose problems with servers**
  - OS has too many sockets in TIME_WAIT, slows things down
    Hack: Can send RST and delete socket, set SO_LINGER socket option to time 0 (useful for benchmark programs)
  - OS won’t let you re-start server because port still in use
    SO_REUSEADDR option lets you re-bind used port number

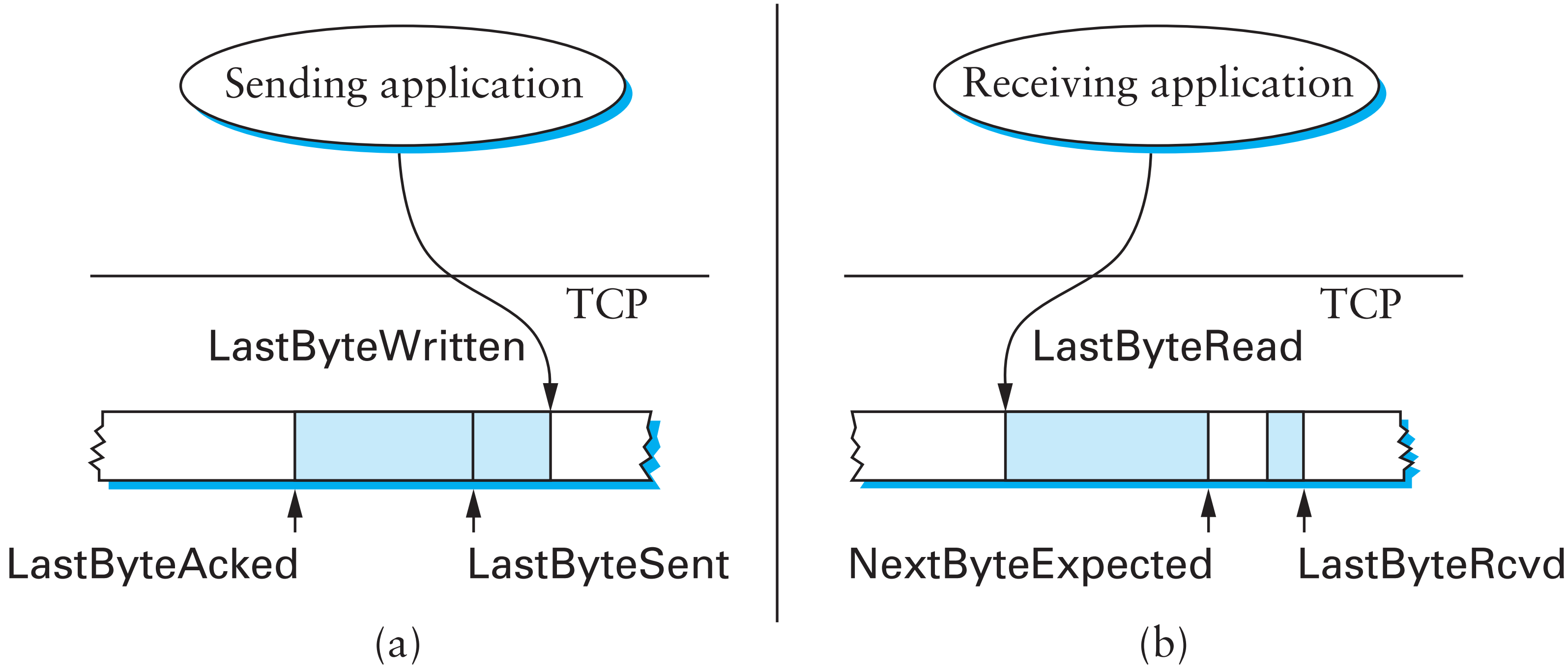
State summary [RFC 793]



Sending data

- **Bulk data sent in MSS-sized segments**
  - Chosen to avoid fragmentation (e.g., 1460 on ethernet LAN)
  - Write of 8K might use 6 segments—PSH set on last one
  - PSH avoids unnecessary context switches on receiver
- **Sender's OS can delay sends to get full segments**
  - Nagle algorithm: Only one unacknowledged short segment
  - TCP_NODELAY option avoids this behavior
- **Segments may arrive out of order**
  - Sequence number used to reassemble in order
- **Window achieves flow control**
  - Receiver sets max window w. `SO_RCVBUF`
  - If window 0 and sender's buffer full, write will block or return EAGAIN

Sliding window revisited



- **Used to guarantee reliable & in-order delivery**
- **New: Used for *flow control***
  - Instead of fixed window size, receiver sends AdvertisedWindow in window field of TCP header
- **Next lecture: used for *congestion control***
  - $\text{SWS} = \min(\text{AdvertisedWindow}, \text{CongestionWindow})$

A TCP connection (3 byte echo)

```
orchard.38497 > essex.echo:
        S 1968414760:1968414760(0) win 16384
essex.echo > orchard.38497:
        S 3349542637:3349542637(0) ack 1968414761 win 17376
orchard.38497 > essex.echo: . ack 1 win 17376
orchard.38497 > essex.echo: P 1:4(3) ack 1 win 17376
essex.echo > orchard.38497: . ack 4 win 17376
essex.echo > orchard.38497: P 1:4(3) ack 4 win 17376
orchard.38497 > essex.echo: . ack 4 win 17376
orchard.38497 > essex.echo: F 4:4(0) ack 4 win 17376
essex.echo > orchard.38497: . ack 5 win 17376
essex.echo > orchard.38497: F 4:4(0) ack 5 win 17376
orchard.38497 > essex.echo: . ack 5 win 17375
```

Path MTU discovery

- **Problem: How does TCP know what MSS to use?**
  - On local network, obvious, but for more distant machines?
- **Solution: Exploit ICMP—another protocol on IP**
  - ICMP for control messages, not intended for buik data
  - IP supports **DF** (don't fragment) bit in IP header
  - Set DF to get ICMP can't fragment when segment too big
- **Can do binary search on packet sizes**
  - But better: Base algorithm on most common MTUs
  - Common algorithm may underestimate slightly (better than overestimating and loosing packet)
  - See RFC1191 for details
- **Is TCP a layer on top of IP?**

Delayed ACKs

- **Goal: Piggy-back ACKs on data**
  - Echo server just echoes, why send separate ack first?
  - Can delay ACKs for 200 msec in case application sends data
  - If more data received, immediately ACK second segment
  - Note: Never delay duplicate ACKs (if segment out of order)
- **Warning: Can interact *very* badly with Nagle**
  - "My login has 200 msec delays"
  - Set TCP_NODELAY

Retransmission

- **TCP dynamically estimates round trip time**
- **If segment goes unacknowledged, must retransmit**
- **Use exponential backoff (in case loss from congestion)**
- **After $\sim$10 minutes, give up and reset connection**
- **Problem: Don't necessarily want to halt everything for one lost packet**
  - Next lecture will explain fast retransmit optimization

Other details

- **Persist timer**
  - Sender can block because of 0-sized receive window
  - Receiver may open window, but ACK message lost
  - Sender keeps probing (sending one byte beyond window)
- **Keep-alives [RFC 1122]**
  - Detect dead connection even when no data to send
  - E.g., remote login server, and client rebooted
  - Solution: Send "illegal" segments with no data and already acknowledged sequence number (`SND.NXT-1`)
  - Or can include one byte of garbage data
  - Remote side will RST (if rebooted), or timeout (if crashed)

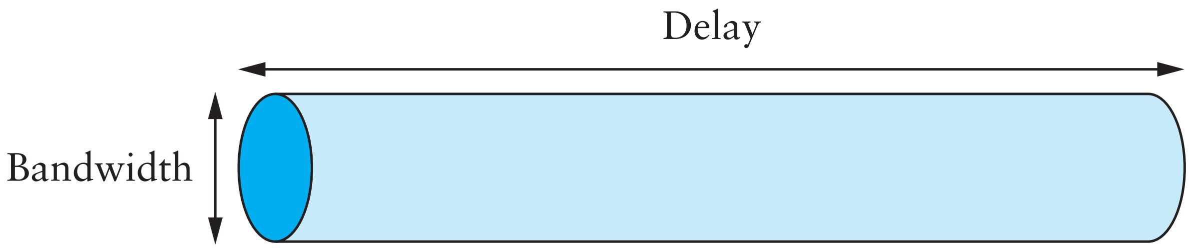
32-bit seqno wrap around

Bandwidth	Time Until Wrap Around
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
Ethernet (100 Mbps)	6 minutes
STS-3 (155 Mbps)	4 minutes
STS-12 (622 Mbps)	55 seconds
STS-24 (1.2 Gbps)	28 seconds

Keeping the pipe full w. 100 msec delay

Bandwidth	Delay $\times$ Bandwidth Product
T1 (1.5 Mbps)	18KB
Ethernet (10 Mbps)	122KB
T3 (45 Mbps)	549KB
Ethernet (100 Mbps)	1.2MB
STS-3 (155 Mbps)	1.8MB
STS-12 (622 Mbps)	7.4MB
STS-24 (1.2 Gbps)	14.8MB

How to fill high bw×delay pipe? [RFC 1323]



- **Extensions implemented as header options**
- **Window scale option for 16-bit window field**
  - Multiplies window by fixed power of 2 in each direction
  - Otherwise, could only fill pipe with 64 KB
- **Extend sequence space with 32-bit timestamp**
  - Protection Against Wrapped Sequence #s (PAWS)
- **Also include most recently received timestamp**
  - Allows much more accurate RTT estimation

Summary

- User datagram protocol (UDP)
- Packet checksums
- Reliability: sliding window
- TCP connection setup
- TCP sliding windows, retransmissions, and acknowledgments
- Using windows for flow control

Limitations of Flow Control

- *Link* **may be the bottleneck**
- **Sending too fast will cause heavy packet loss**
- **Many retransmissions, lost acks, poor performance**
- **Flow control provides** *correctness*
- **Need more for performance: congestion control (Next lecture…)**