TCP Protocol - Transport Layer

TCP/IP class

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outline

- intro
- sliding window protocol
- tcp concepts
  - header/piggybacking
  - ports and sockets (connection idea)
- open/close state machine
- some protocol mechanics
- performance

intro

- TCP - Transmission Control Protocol
- reliable, connection-oriented stream (point to point) protocol
  - if UDP is like U.S. Mail
  - TCP is like a phone call (cannot broadcast/multicast)
- we need stream delivery because underlying mechanism has flaws
  - out of order due to routing, or loss, or corruption (or dups due to timeout/resend)

intro

- RFC 793 and host requirements 1122
- TCP has own jargon:
  - **socket -** we'll see in a bit
  - **segment**: a TCP packet
  - **MSS: maximum segment size**, max pkt one TCP side can send another, negotiated at connection time
  - **ports**:

TCP properties

- stream orientation. stream of OCTETS (bytes) passed between send/recv
- byte stream is full duplex
  - think of it as two independent streams joined with **piggybacking** mechanism
- piggybacking - one data stream has control info for the other data stream (going the other way)
- unstructured stream
  - tcp doesn't show packet boundaries to applications

TCP properties

- unstructured stream, cont
  - but you can still structure your i/o as “messages” or structures if you want
- virtual circuit connection
  - client connects and server listens/accepts
  - i/o transfers don’t have remote peer address
- tcp provides flow control
  - you don’t have to worry about recv buffering

writing structures down TCP pipe

struct foo { int x; int y; int z; } f;

write (sock, &f, sizeof (struct foo));

read(sock,
&f,
sizeof (struct
foo));
* not exactly

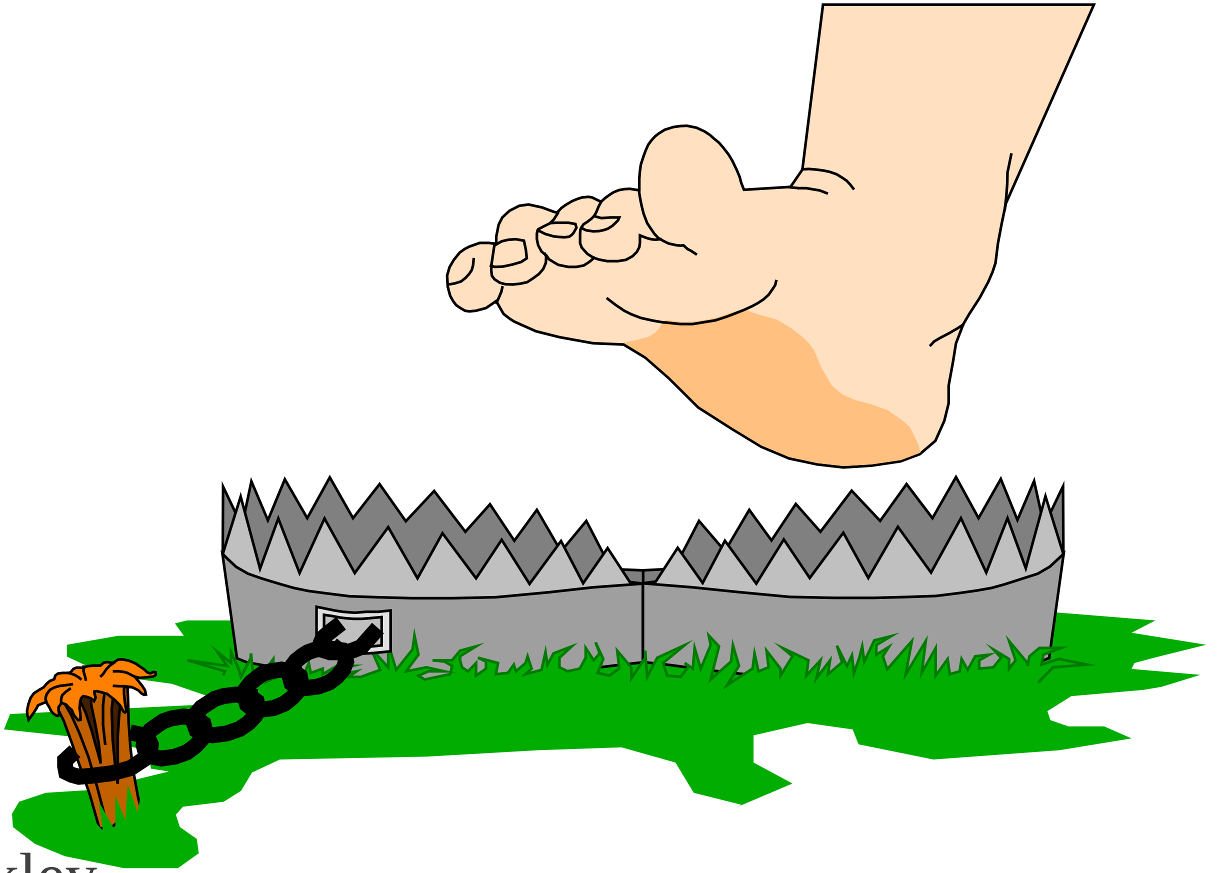
TCP properties

- efficiency - not easy to duplicate over WAN environment
- congestion detection end to end
  - backs off if it thinks net is congested
- **most TCP/IP error handling is in TCP**
  - end to end
- complex protocol
  - can treat telnet (interactive) and ftp (bulk transfer) differently + acks/timers, etc

TCP buffering and prog POV

- can’t predict very well as programmer how tcp will buffer data
  - write 2 512 byte packets, might read 1 1k pkt
  - write 1 1024 byte packet, might read 2 512
  - type in 3 characters, tcp might send them as 1 3 byte packet
- It’s a data stream
- writes are atomic, unless you use non-blocking I/O. reads are not

reinventing tcp...

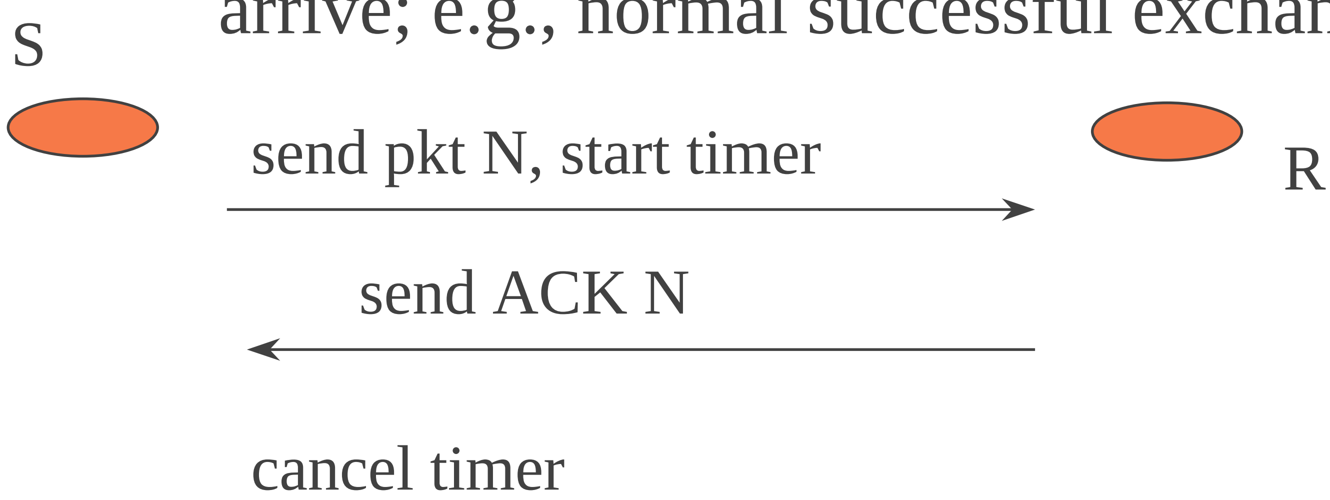


TCP - Complex protocol

- Doug Comer states that TCP is a protocol and not any particular specification or chunk of software. True, but
- Binkley states: “Don’t try this at home, borrow one from the Internet”
  - 4.4 BSD, KA9Q, linux
- TCP is not easy to debug or test in terms of interoperability (or replace with a reliable UDP app)

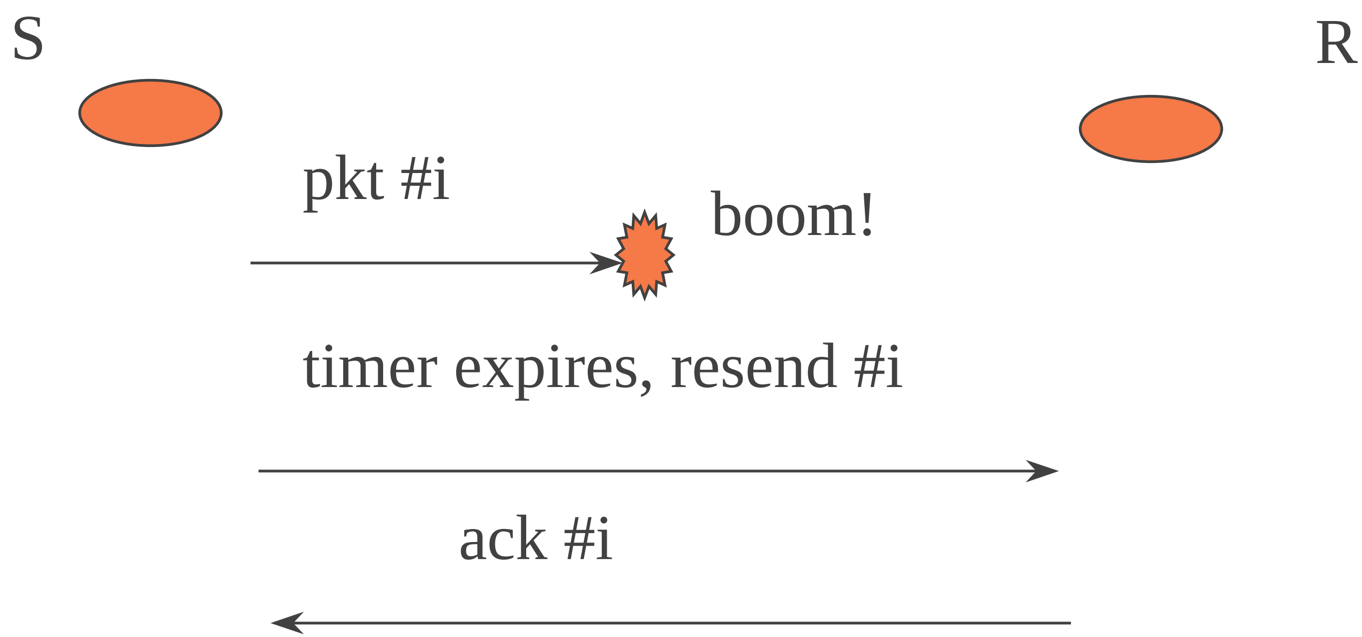
sliding window protocol basics

- positive acknowledge with retransmission
  - recv must send ACK with seq #
  - sender must timeout and resend if ACK fails to arrive; e.g., normal successful exchange:



sliding window

- and when it doesn't work



because we resend: we can duplicate pkts and ACKS!

sliding window protocol

- simple pos. ack. with retransmission is called “ping-pong” protocol, not efficient
- send && wait for ACK, therefore data only flows one way at a time, not both ways, **thus we cut efficiency of channel in half**
- we want to send many packets (asap) and get back ACKS (possibly combined into 1 cumulative ACK)

sliding window

- sliding window is more complex form of pos. ack. with retransmission
- we still retransmit and we still want ACKS

packets: 1 2 3 | 4 5 6 7 | 8 9 10 11 12 13 14

done | window | not sent

acked unacked

1-3 sent and ACKED, 4-7 in window and sent but not ACKED, if ACK arrives, sender slides window up

sliding window, cont.

- goal in previous slide: cumulative ACK e.g., [ACK up to #7]
- tcp uses bytes not packets for sequencing
- recv-side controls sliding window and views that as available buffering, can stop sending by telling it window size is 0 in ACK, thus **flow control**
- with window size == 1, we get simple positive ack with retransmission

TCP - encapsulation

e.2=14	20	20	1460
ethernet	ip header	tcp header	data (but maybe not)

TCP header may have options, but default size is 20 bytes

TCP header

<table>
<tr><td colspan="8">0 15</td><td>16 31</td></tr>
<tr><td colspan="6">source port: 16 bits</td><td colspan="3">dest. port: 16 bits</td></tr>
<tr><td colspan="9">sequence number: 32 bits</td></tr>
<tr><td colspan="9">acknowledge number: 32 bits</td></tr>
<tr><td>hlen:4</td><td>resv:6</td><td>U</td><td>A</td><td>P</td><td>R</td><td>S</td><td>F</td><td>window size: 16</td></tr>
<tr><td colspan="6">TCP checksum:16</td><td colspan="3">urgent pointer: 16</td></tr>
<tr><td colspan="9">TCP options (if any)</td></tr>
</table>

Header - explained

- header sent in every TCP packet, may just be control message (SYN/FIN/ACK) with no data
- view TCP as 2 sender/recv data streams with control information sent back the other way (piggybacking)

header slides

- source port: 16 bits, the TCP source port
- destination port: 16 bits, note ports in 1st 8 bytes
- sequence number: 1st data octet in this segment (from send to recv): 32 bit space
- ack: if ACK flag set, next expected sequence number (piggybacking; i.e., we are talking about the flow the other way)

header, cont.

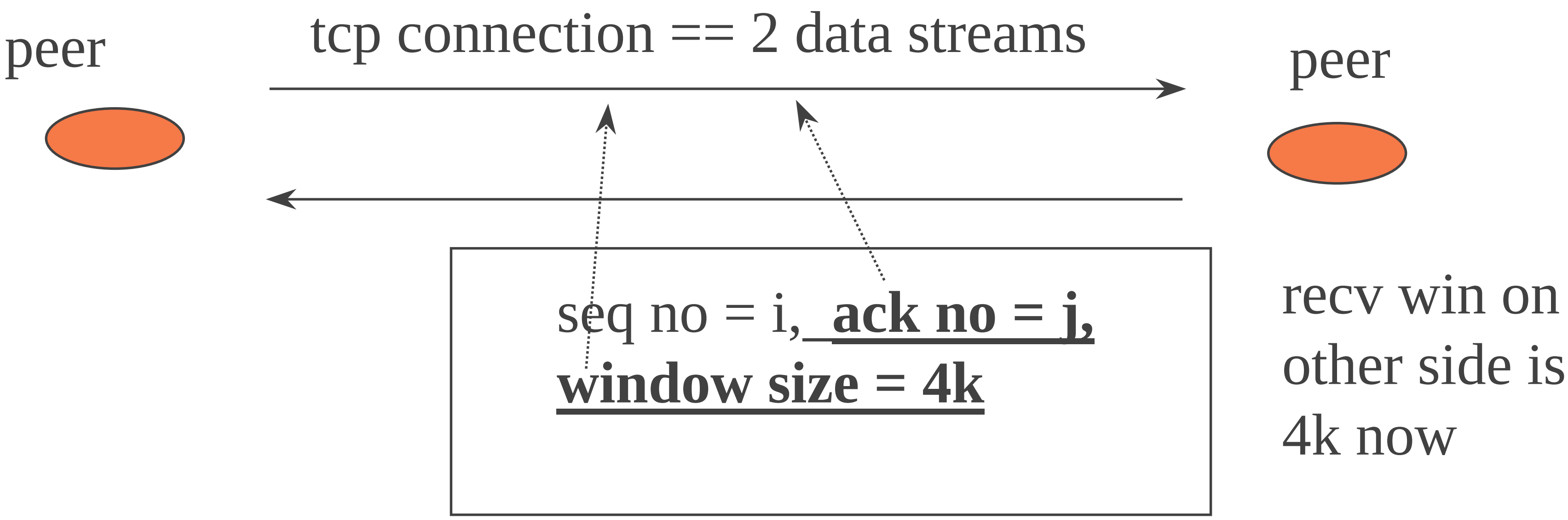
- hlen: # of 32 bit words in header
- reserved: not used
- flags
  - URG: - urgent pointer field significant
  - ACK:- ack field significant (this pkt is an ACK!)
  - PSH: - push function (mostly ignored)
  - RST: - reset (give up on) the connection (error)
  - SYN: - initial synchronization packet (start connect)
  - FIN: - final hangup packet (end connect)

header, cont.

- window: window size, begins with ACK field that recv-side will accept (piggyback)
- checksum: 16 bits, pseudo-header, tcp header, and data
- urgent pointer: offset from sequence number, points to data following urgent data, URG flag must be set
- options - e.g., Max Segment Size (MSS)

TCP piggybacking (header)

- data may be sent 2-ways, sender to recv (1-way data flow) may contain piggybacked state (ack/window fields) for other data channel. This info is feedback on other channel



returned pkt either ACK or may have data

ports and sockets

- TCP clients and servers have a TCP port in the TCP port space 0 (not used)..64k-1
- unlike UDP, TCP uses connection as fundamental abstraction, not port
- when we connect, we end up with:
  peer (client): 18.25.0.36,1069; 128.10.2.3,25
  peer (server): 128.10.2.3,25; 18.25.0.36,1069
- each side has 4-tuple (**socket**) which is used to id incoming packets (demux to app)

ports and sockets

- TCP server architecture thus

  connected_fd = accept(listen_fd, ...);
- server may spinoff “slave” thread on connected fd to take care of application-level protocol (some sequence of read/write calls)
- all server processes bound to WK port have same server-side port #; e.g., http/80

TCP open/close

- TCP distinguishes **passive** and **active** open
- servers usually do passive open, means they LISTEN
- clients usually do active open, means they connect
- reach ESTABLISHED state after 3-way handshake

open/close, 3 way handshake

sender **recv**

1. send SYN seq=x

recv SYN,
2. send SYN seq=y,
ACK x+1

3. recv SYN/ACK,
send ACK y+1

---

both sides can SYN at the same time and it will work

results include established connection, initial sequence numbers exchanged, ACKS ack next expected byte (cumulative)

closing a tcp connection

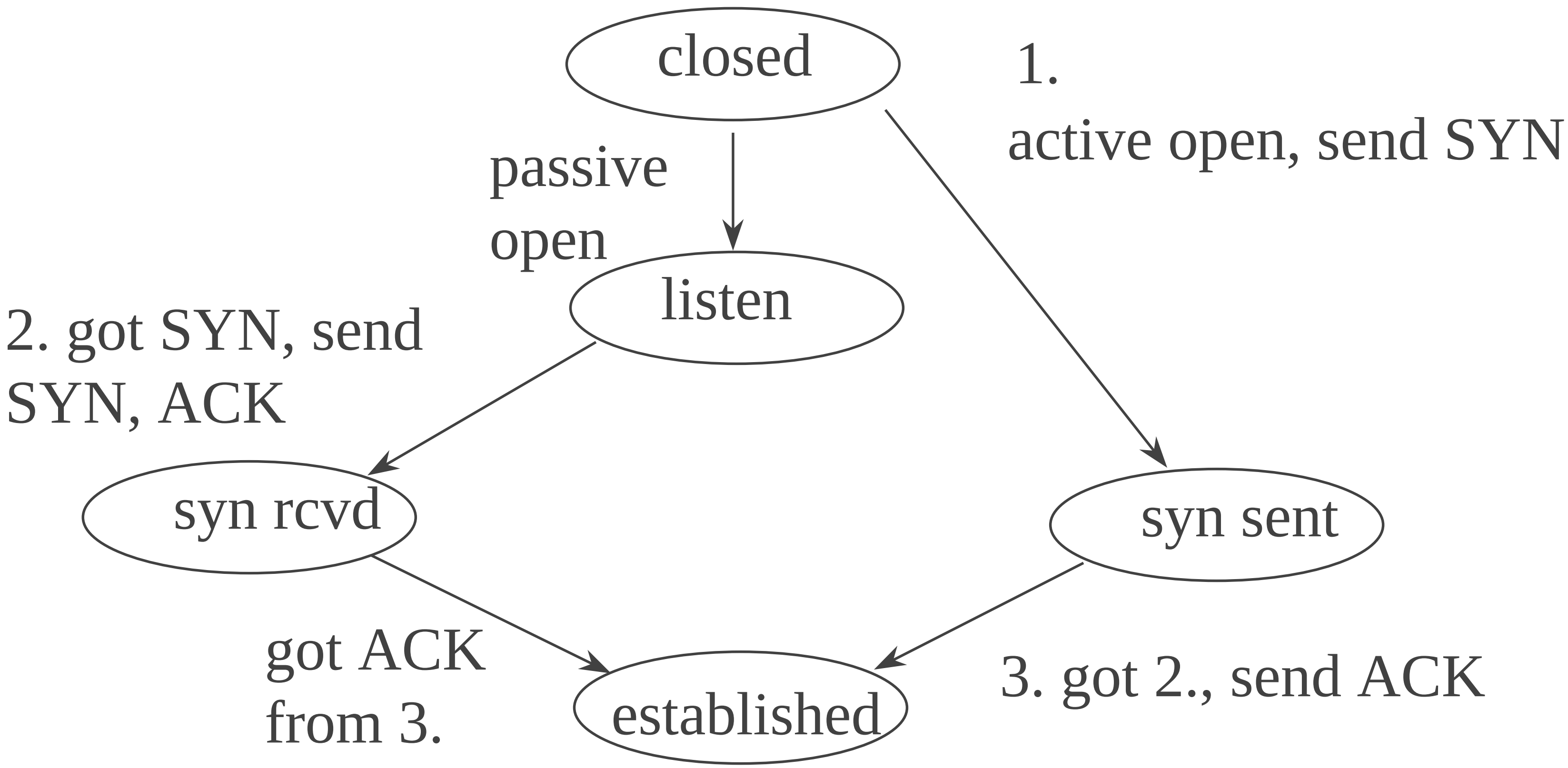
- connections are full duplex and it is possible to shutdown(2) one side at a time
- close(2) closes everything and the UNIX version doesn’t quite jibe with TCP - UNIX close is async and doesn’t wait for handshake
- really just 2 2-way handshakes (send FIN, recv replies with ACK per channel)
- interesting problem: how do you make sure last ACK got there (can’t ACK it...)

close

peer1		peer2
send FIN seq=x	$\longrightarrow$	
		recv FIN
	$\longleftarrow$	send ACK x+1
recv ACK		
	$\longleftarrow$	send FIN seq=y
recv FIN		
send ACK y+1	$\longrightarrow$	
		recv ACK

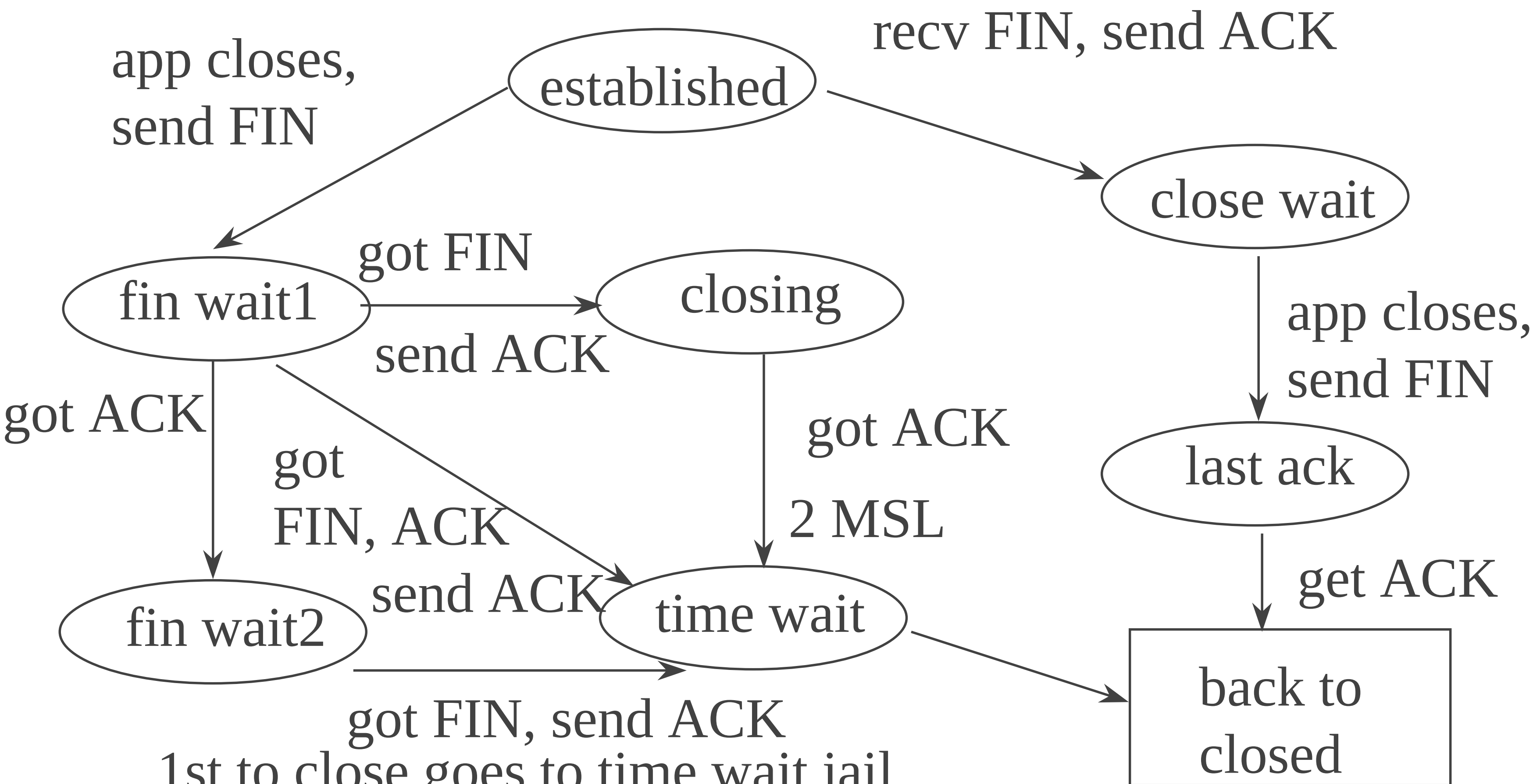
1st to send FIN is last to send ACK,
no way to know if that ACK arrived

state machine - simplified open



i/o, i/o, it’s off to work we go ...

state machine - close



1st to close goes to time wait jail,

the FINE for the FIRST FIN is ...

- both apps could close first and send FIN, hence left side is more complex, but state machine supports async close
- s/he who closes first, gets stuck in TIME_WAIT state since we aren't getting an ACK back for our ACK sent, must wait 2 MSL (max segment length) time, 1 or 2 minutes typically

Protocol Mechanisms (some)

- Stevens, p. 227
  "There is no single correct way for two TCPs to exchange a given amount of data"
- window size - flow control
- delayed ack
- nagle algorithm
- adaptive retransmission + backoff
- congestion control

TCP variable window size

- flow control occurs because the receive side controls the window size
- if window size == 0, the sender cannot send data
- sender will send window probe (1 byte of data) to see if window is open (ack might be lost). Separate timer for this function called persistance timer
- this is end to end flow control, doesn’t include routers

delayed ACK

- try to not send ACK immediately and hope that data will show up so that ACK can piggyback (free ride, not extra packet)
- delay typically 200 milliseconds
- this is recv-side timer, not send ACK timer
- with telnet might see

  send 1 char

  echo char and ACK it

  delayed ACK for echo

nagle algorithm

- traditional telnet over wan can add to congestion because we have 40 bytes of header for 1 echoed byte of data
- rfc 896 - nagle algorithm
- tcp connection can only have one unacked outstanding small segment. Can't send more until you get an ACK, sender may collect more data
- only affects sender who sends small data amounts
- TCP_NODELAY socket option turns this off

nagle, cont.

- algorithm is said to be “self-clocking”, you can go as fast as round trip latency will allow since you wait for return ACK
- hope is that sender slows down to give data opportunity “to pile up” before it is sent
- X-windows would not want nagle algorithm since it would send small data chunks (mouse clicks) and want those sent as real-time as possible

timeout and retransmission

- can’t use fixed time for send ACK timer
- if too long, response not good if timeout occurs,
- if too short, can’t know apriori how long to wait (and congestion might change the time)
- TCP uses **adaptive retransmission timer,**
- see text for details, uses fixed-point arithmetic

simple timer backoff

- if no acks at all are received tcp will use a modified form of exponential backoff
- Stevens (p.299) gives 1,3,6,12,48, 64 on one implementation, retries at a minute until 9 minutes then a reset
- will this work for the Mars Mission?
- if packets start showing up backoff is removed

congestion control in TCP

- routers may drop packets as space is not pre-allocated by definition - congestion
- routers don't have effective mechanism to indicate congestion (ICMP source quench is not it...) to sender
- assumption: packet loss due to damage is small, therefore TCP assumes it means congestion since ACKS do not come back

congestion control

- TCP uses **slow start** and **multiplicative decrease** to deal with congestion
- Van Jacobson 1988 outlined these ideas
- slow-start roughly: whenever starting traffic or recovering from congestion, start congestion window at the size of a single segment and increase it (up to a point) as ACKs show up

congestion avoidance

- multiplicative decrease - upon loss of a segment, reduce the congestion window by half down to a minimum of 1. For those segments that remain in the send window, backoff the retransmission timer exponentially.

routers and congestion - RED

- routers might use an obvious queue-drop mechanism
  - too many buffers; drop packets at end of queue call this a“**tail-drop**” policy
  - on heavily multiplexed router many TCP connections may lose a packet and be forced into slow-start
- routers may use **Random Early Detection** (or RED) - basically randomly discard packets in Queue at a certain saturation point
  - thus avoid tail-drop policy

TCP efficiency/performance

- mid 80's say with VAX on ethernet, performance was poor, now can find good approximation of 1 gigabit on faster end hosts, 90% of 100BASE common
- assumptions (from Stevens):
  - send two packets into 2 pkt window
  - get one ack
  - 2 hosts on ethernet
  - max data possible

performance, cont.

- equation:

$$\frac{\text{real data bytes}}{\text{header (overhead)}} \qquad \frac{\text{bits/sec}}{\text{8 bits}} \Rightarrow \frac{\text{bytes}}{\text{sec}}$$

assume ethernet:

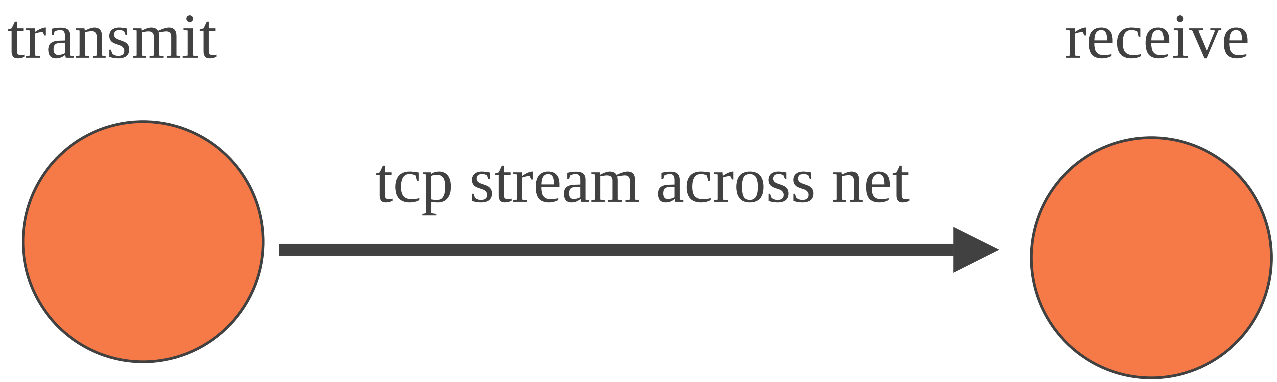
$$\frac{2 * 1460}{2 * 1538 + 84} * \frac{10000000}{8} = 1,155,063$$

$$\frac{1155063}{1250000} = 92.4\%$$

ttcp - used to measure tcp thruput

- common on UNIX hosts
- disks not part of measurement
- normalize 1st say with measurement over localhost
- then between hosts on net
- may use different window sizes,
  - 8k/16k/32k/64k ...

ttcp test - test ttcp



ttcp -t -s ...

ttcp -r -s

note: this is memory to memory, no disks involved

some observations

- commodity P3 cpus with gigabit Ethernet card can easily do 500mbits with TTCP
- 781 mbits between two Crays over 800 Mbs hippi channel
- 907 over Cray loopback
- can’t go faster than slowest link
- can’t go faster than memory bandwidth
- can’t go faster than window size/rtt

constraints are finally

- window-size (new window-size and PAWS options are significant here)
- speed of light

study questions

- assume you have a TCP connection (telnet to site Y) and you reboot a router in between, what should happen?
- if you have a TCP connection and you reboot one of the end-end systems, what should happen?
- what would be the problems with having TCP support multicast addresses?

study questions

- it is widely assumed (if not cherished) that TCP would make a poor mechanism for transfer of audio/video steady-stream data? Why is that? and can you make a case for a contrarian point of view?
- you telnet to mars... what should you think about in terms of tcp timers?
  - hint: it might take about 10 minutes one-way to Mars for light (depends on mars/earth orbits)