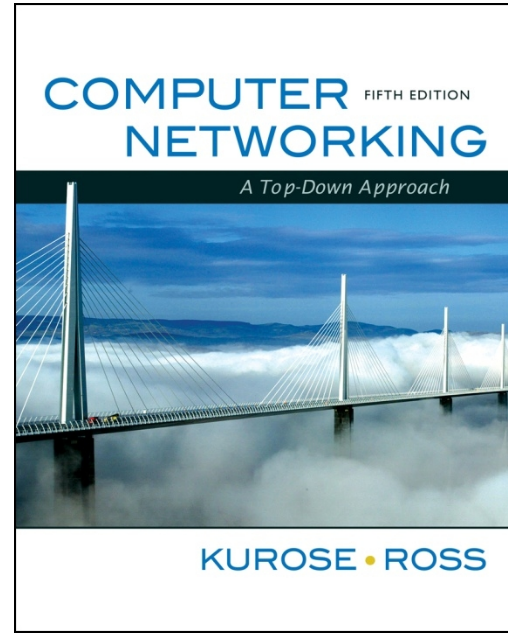
Chapter 3
Transport Layer



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Thanks and enjoy! JFK/KWR

Computer Networking: A Top Down Approach
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

Chapter 3: Transport Layer

Our goals:

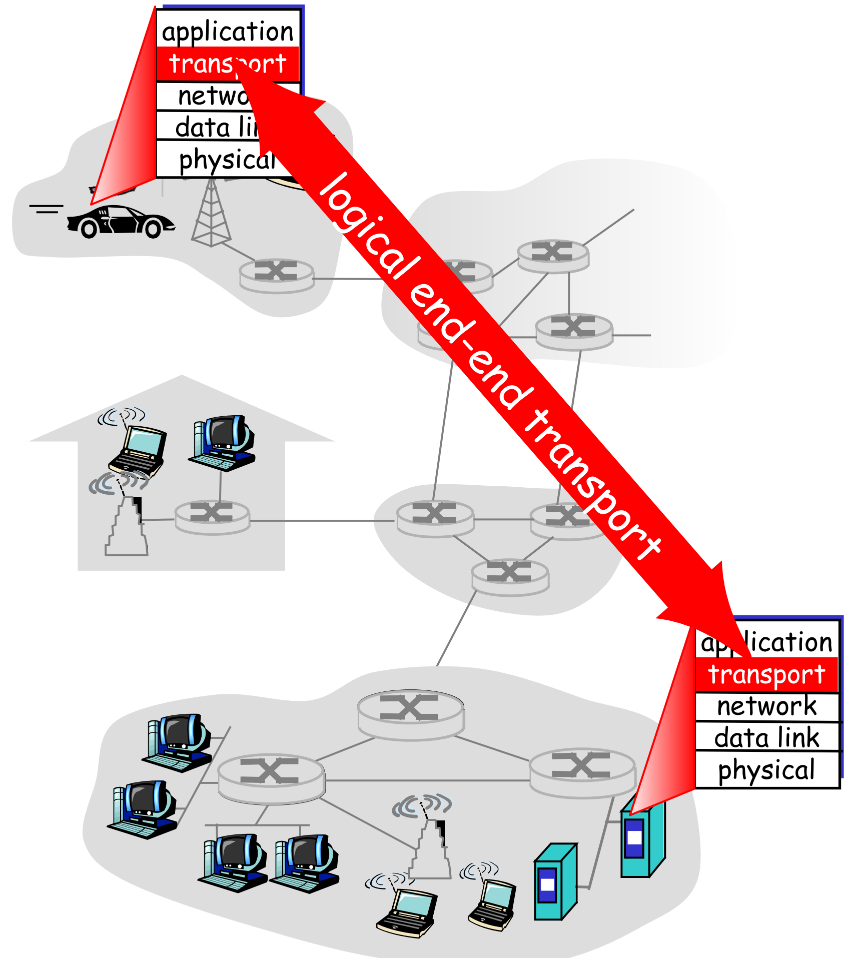
- understand principles behind transport layer services:
  - multiplexing/ demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

Chapter 3 outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

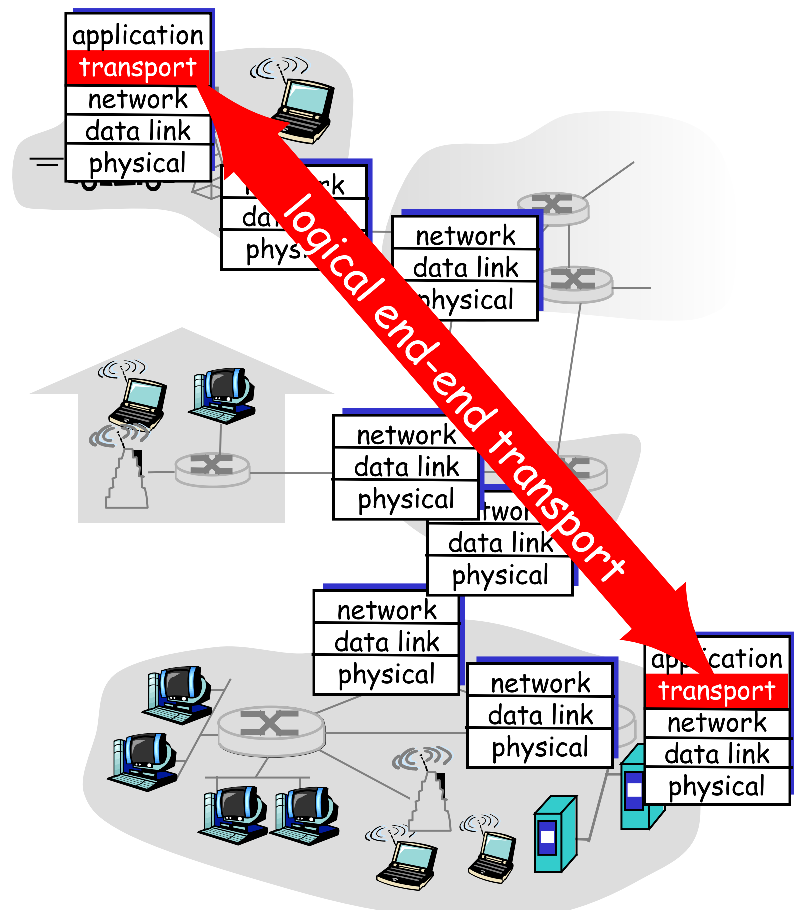
Household analogy:

12 kids sending letters to 12 kids

- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill
- network-layer protocol = postal service

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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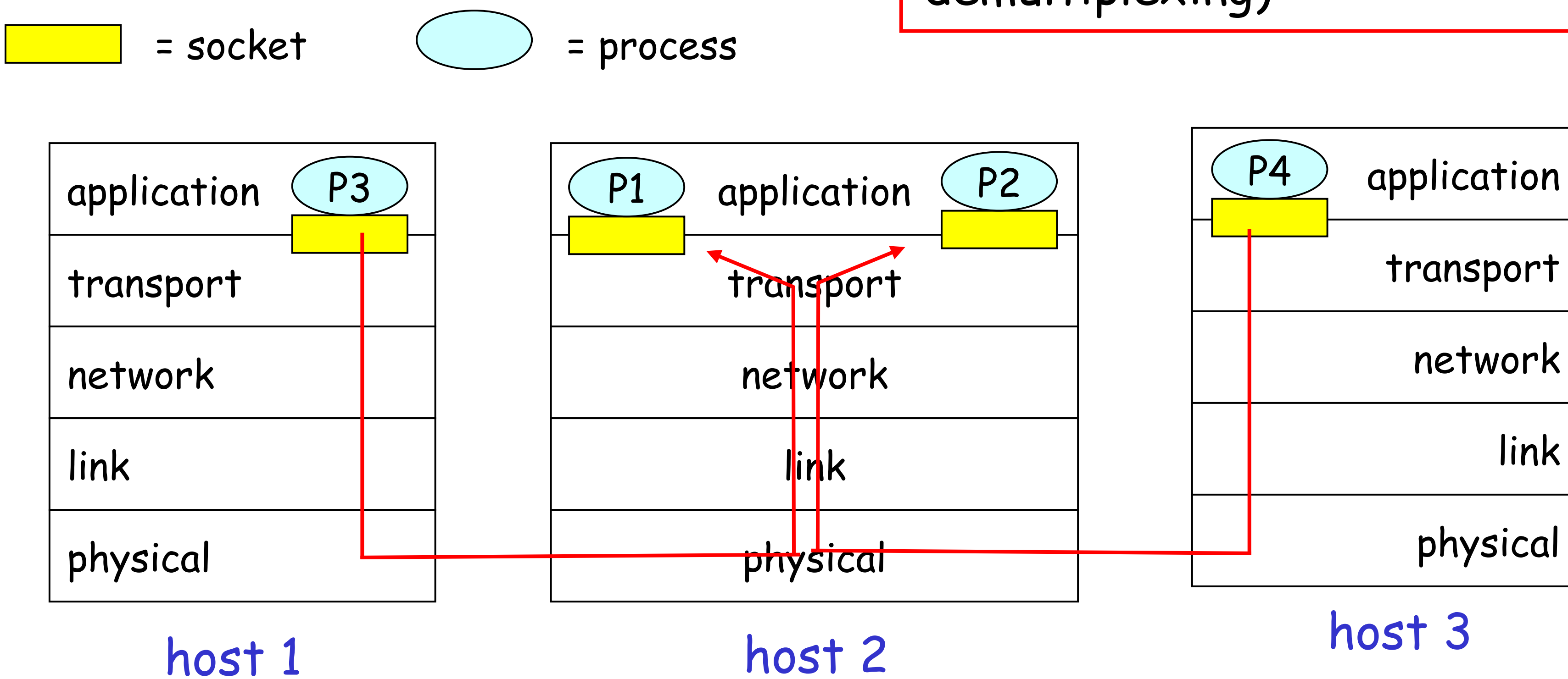
Multiplexing/demultiplexing

Demultiplexing at rcv host:

delivering received segments to correct socket

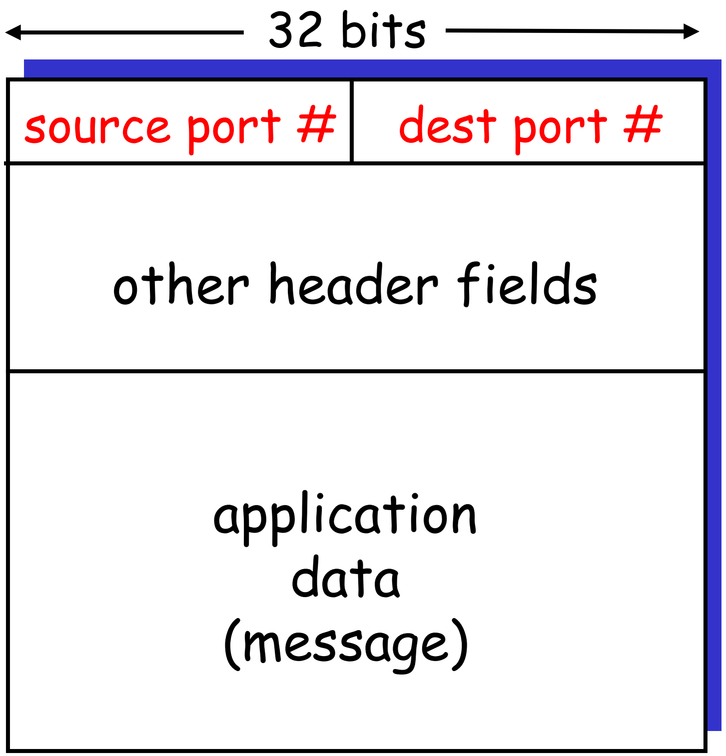
Multiplexing at send host:

gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 transport-layer segment
  - each segment has source, destination port number
- host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
   DatagramSocket(12534);
DatagramSocket mySocket2 = new
   DatagramSocket(12535);
```

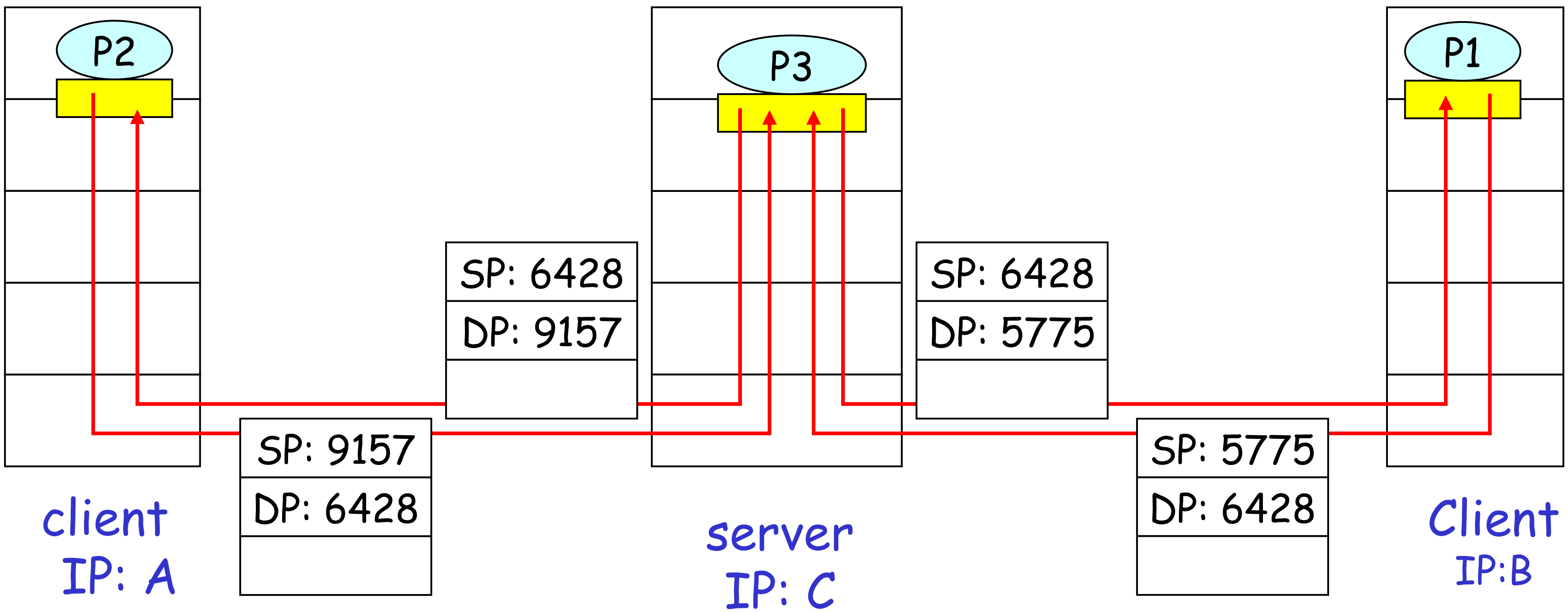
- UDP socket identified by two-tuple:

(dest IP address, dest port number)

- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

Connectionless demux (cont)

```
DatagramSocket serverSocket = new DatagramSocket(6428);
```



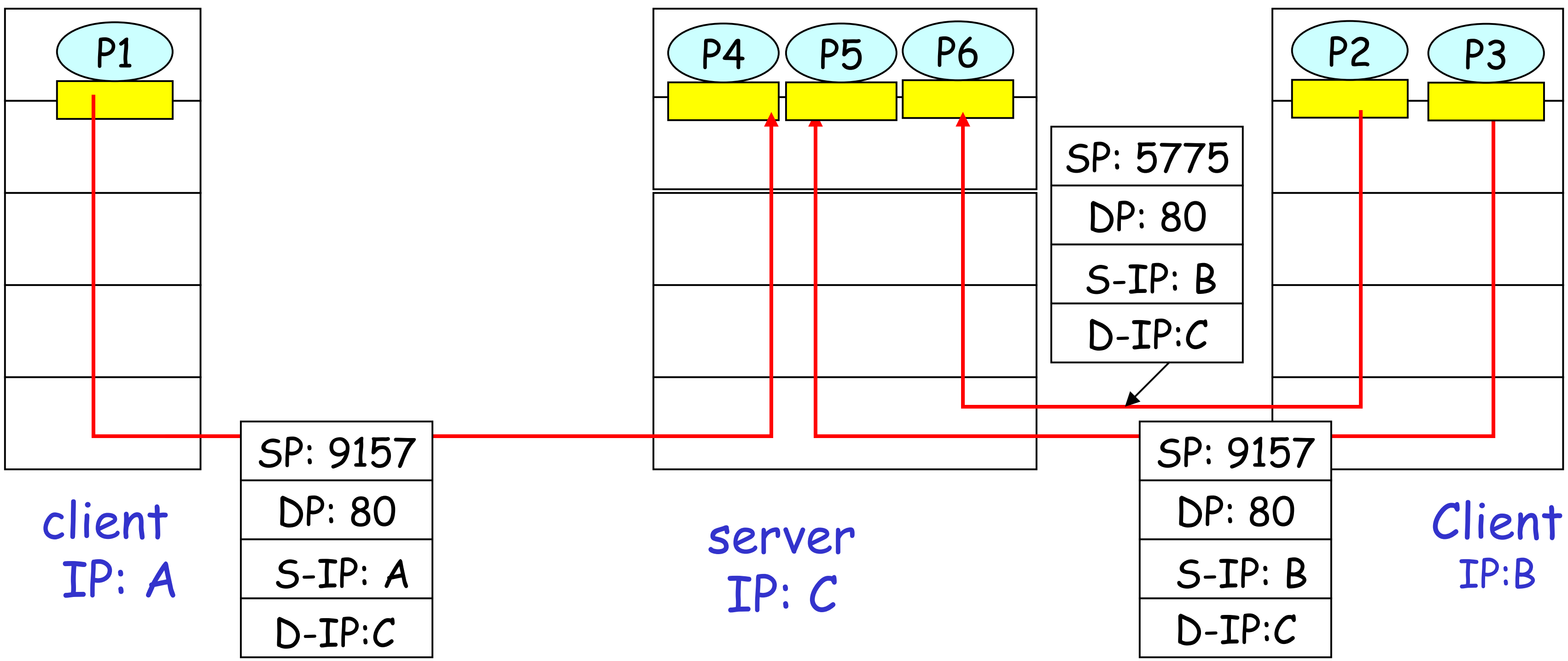
SP provides "return address"

Connection-oriented demux

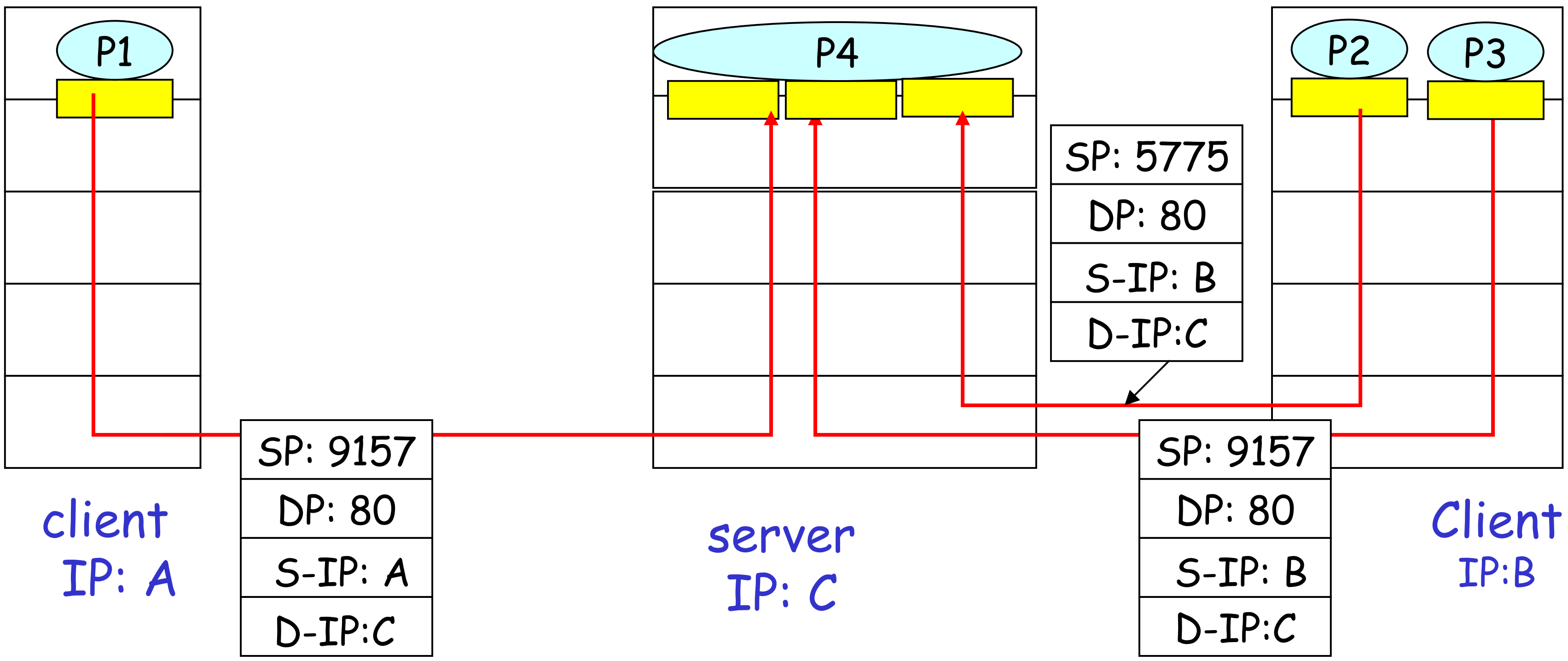
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- receiving host uses all four values to direct segment to appropriate socket

- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



Connection-oriented demux: Threaded Web Server



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UDP: User Datagram Protocol [RFC 768]

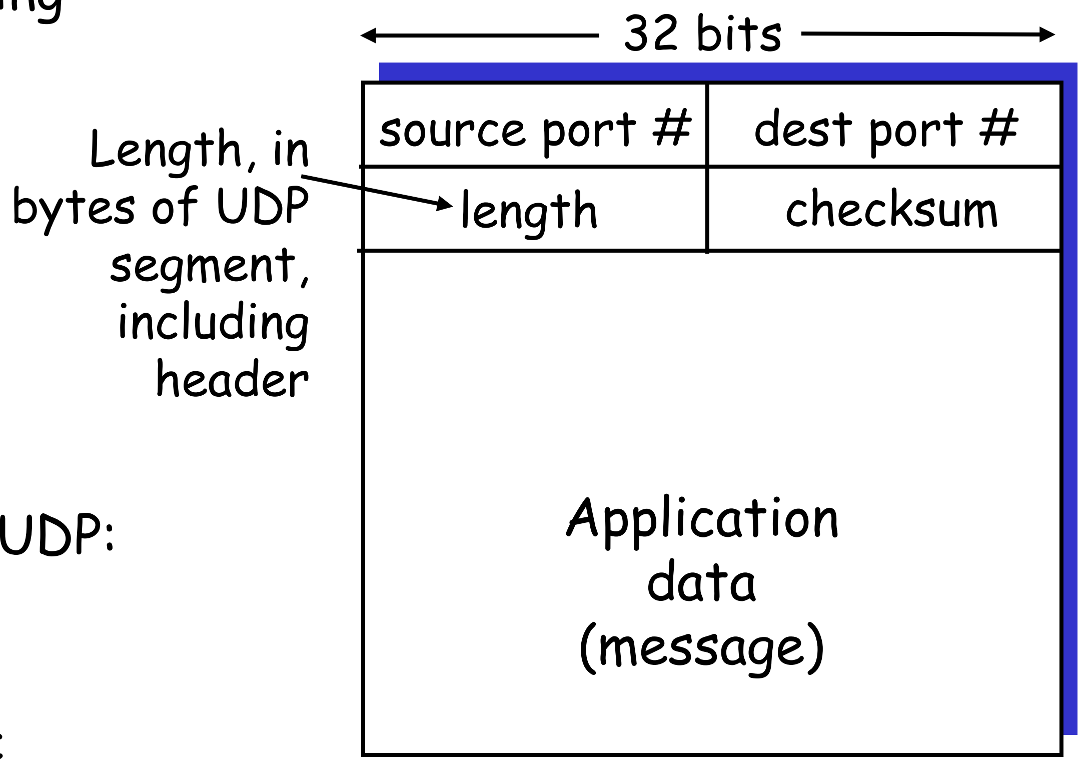
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired

UDP: more

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



UDP segment format

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet Checksum Example

- Note
  - When adding numbers, a carryout from the most significant bit needs to be added to the result
- Example: add two 16-bit integers

		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

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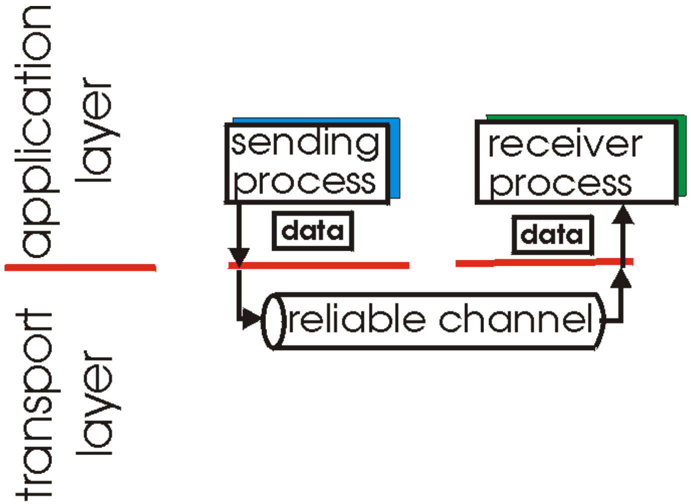
Principles of Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!



(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of Reliable data transfer

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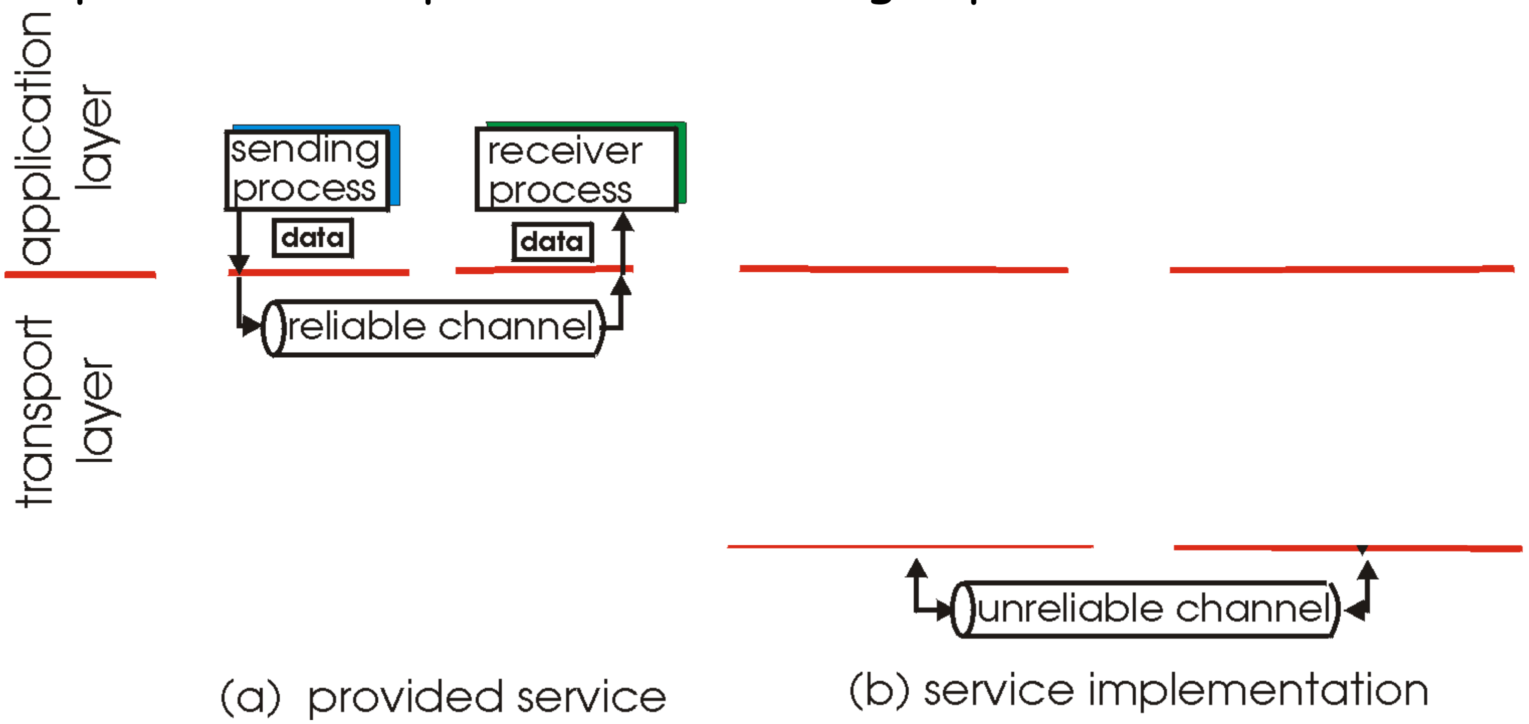

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Principles of Reliable data transfer

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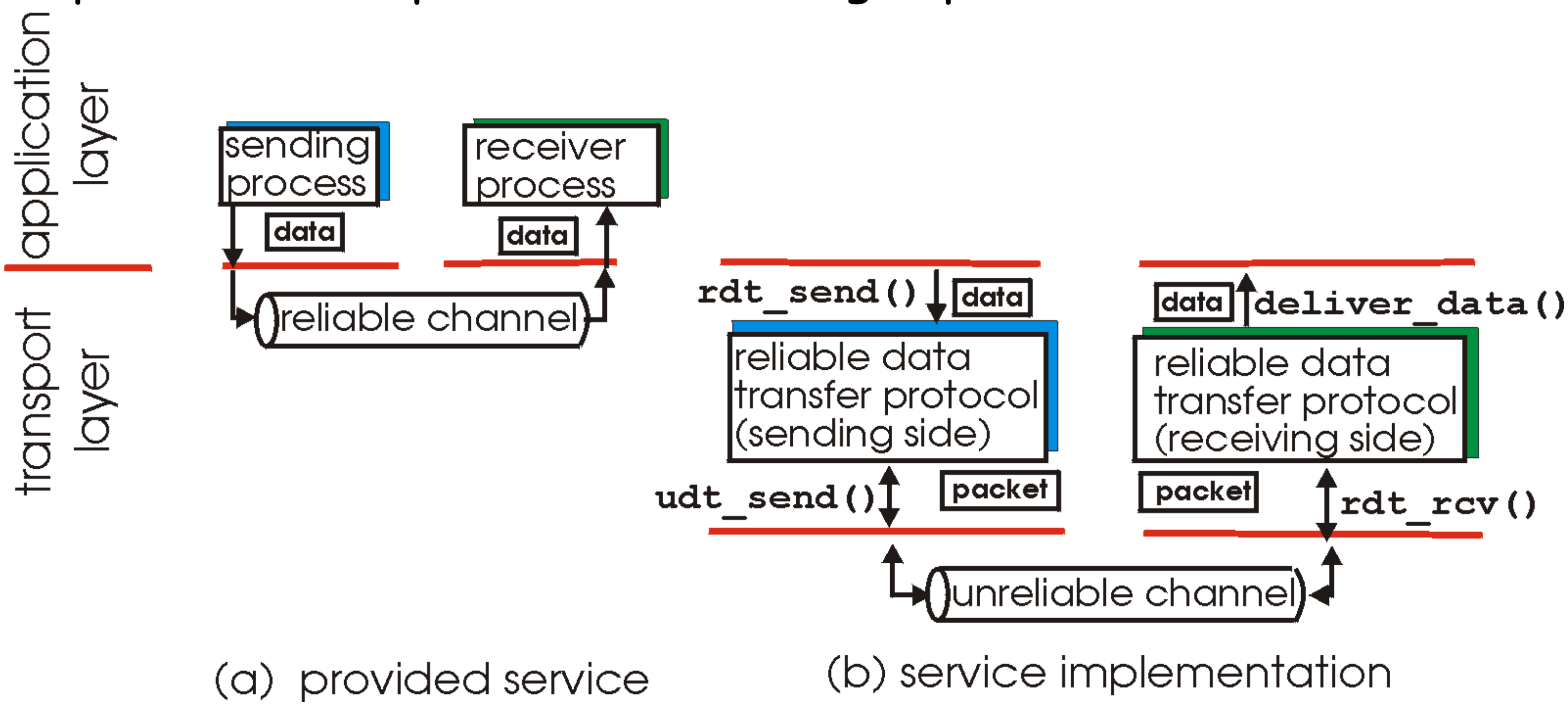
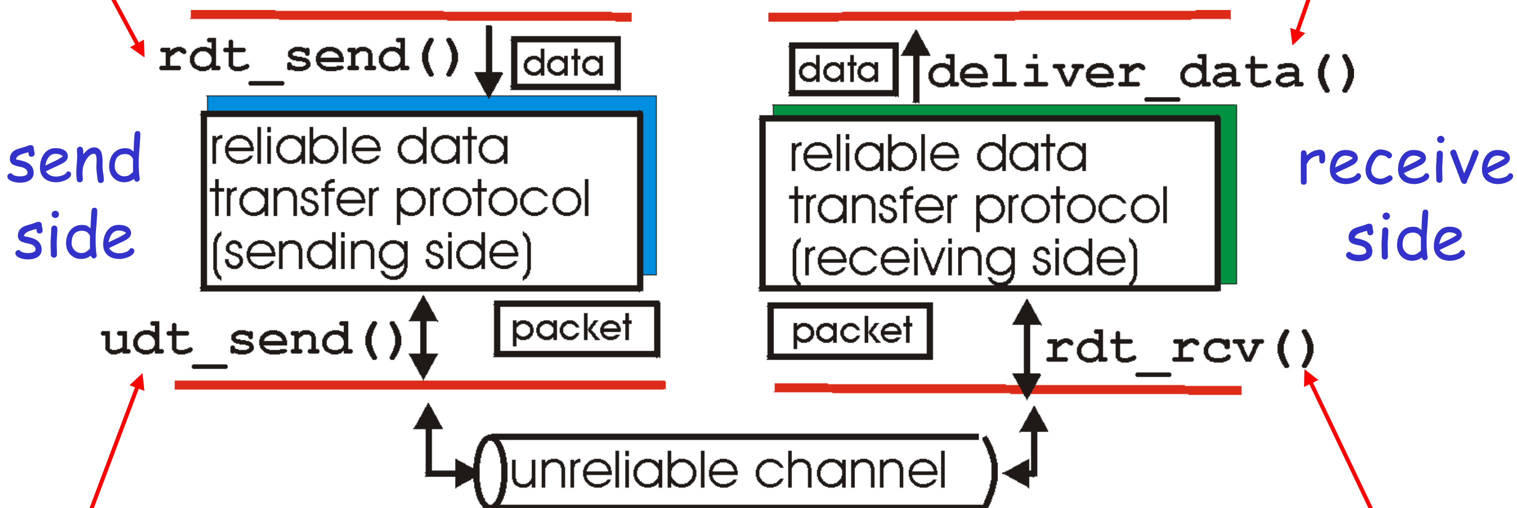


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by **rdt** to deliver data to upper



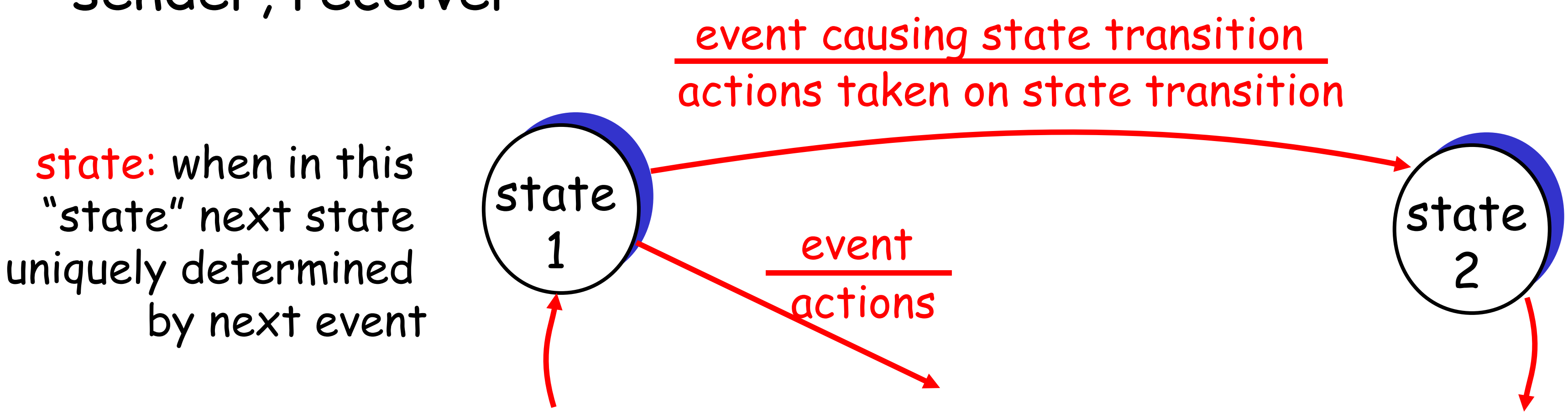
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

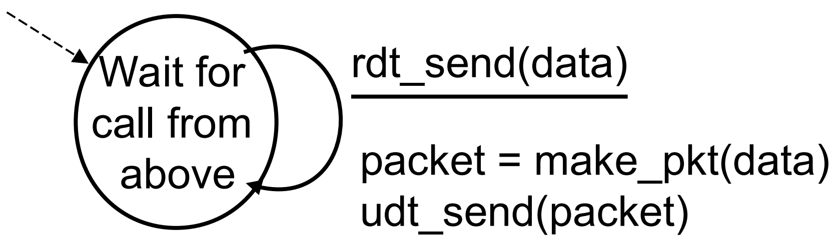
We'll:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

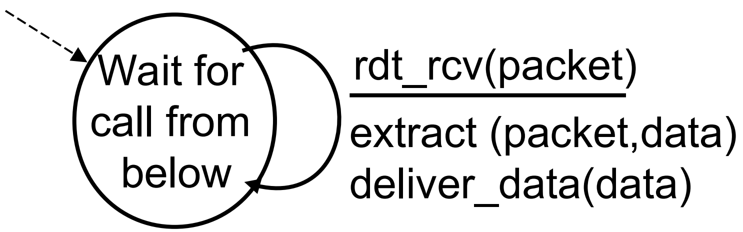


Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



sender

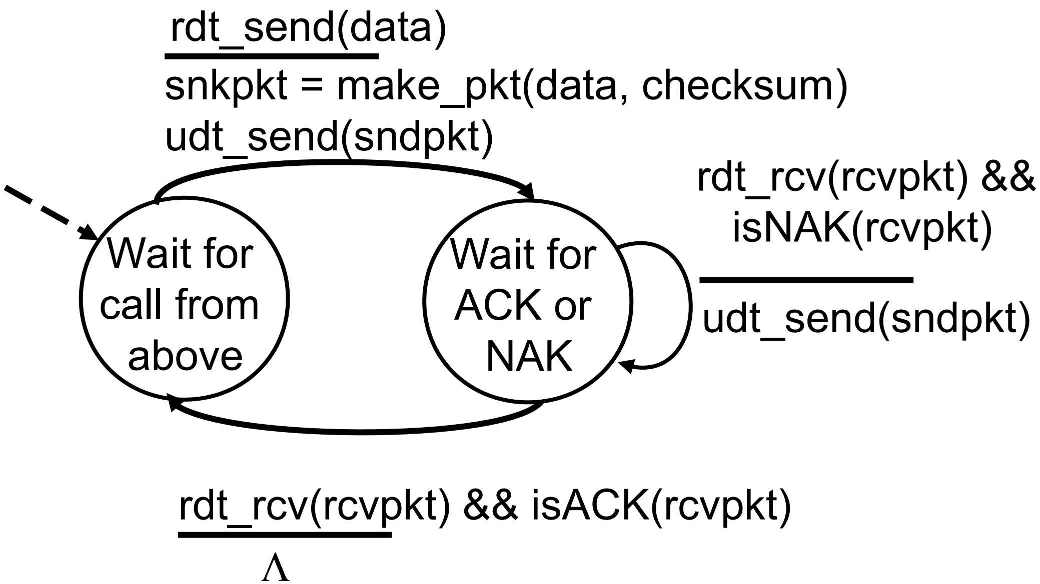


receiver

Rdt2.0: channel with bit errors

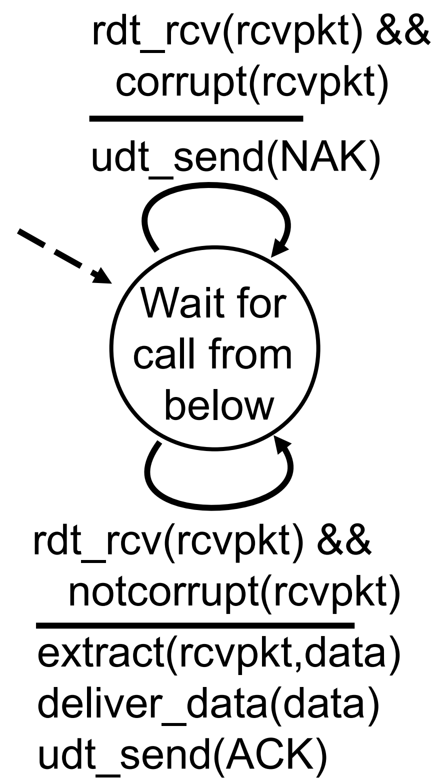
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

rdt2.0: FSM specification

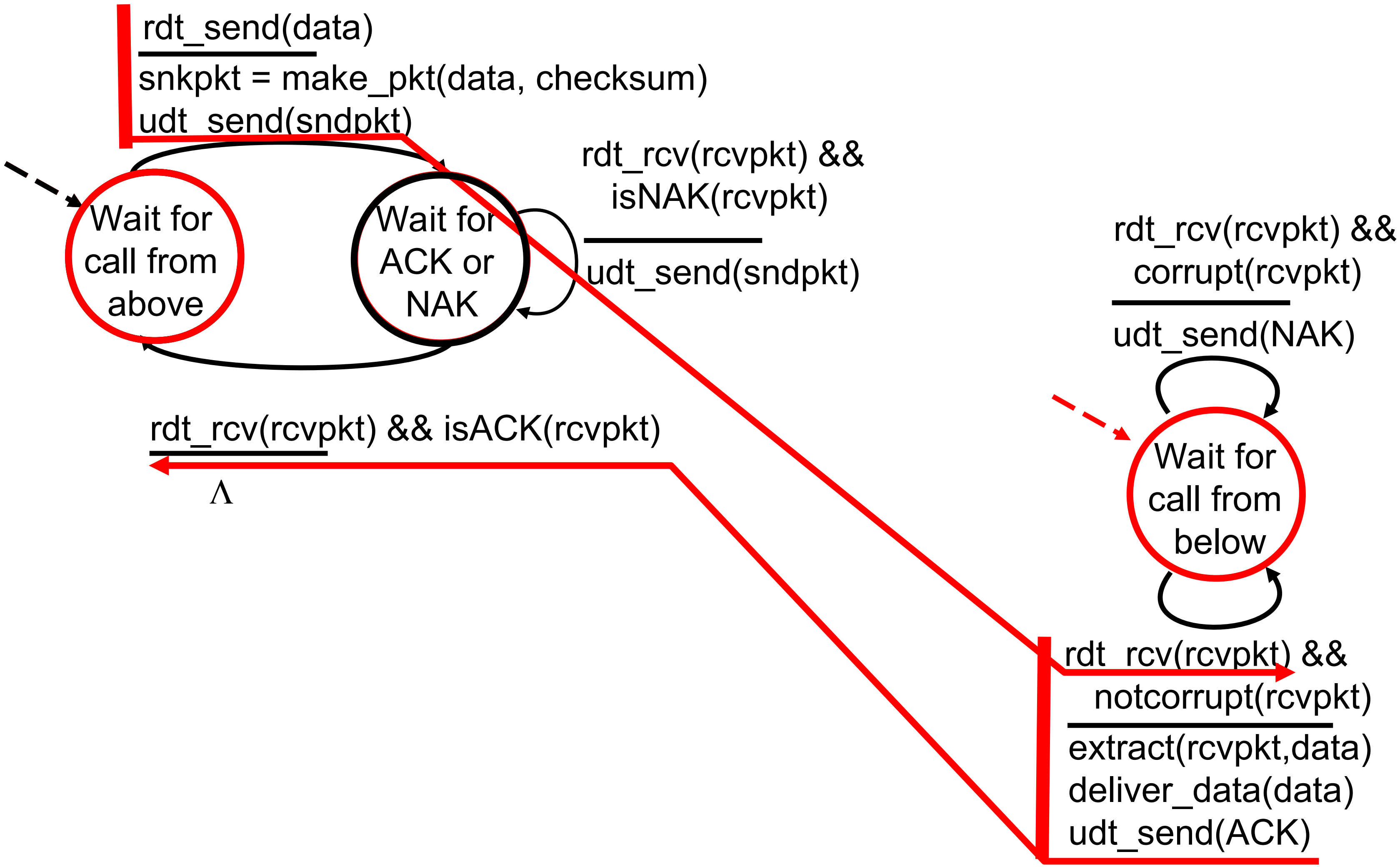


sender

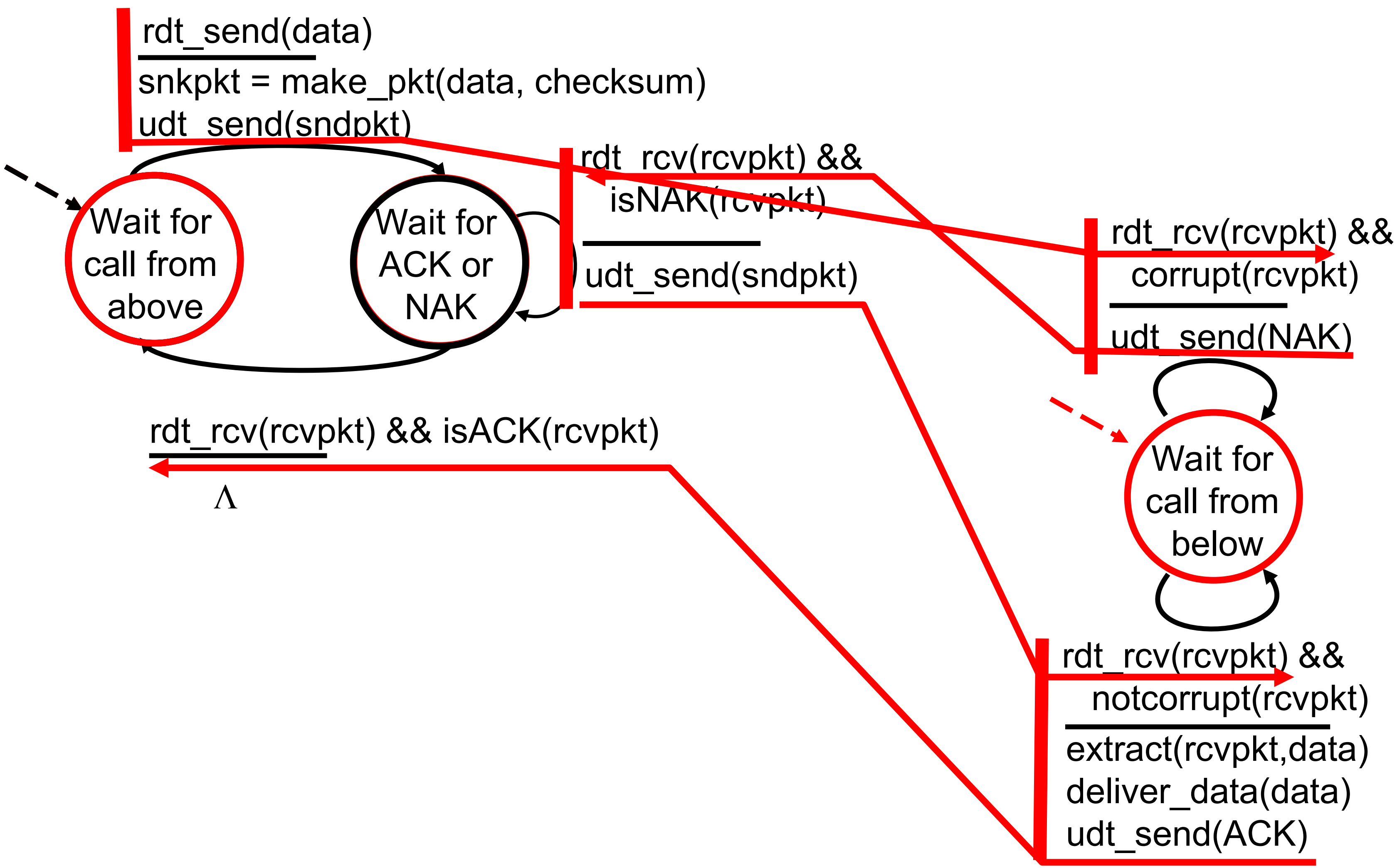
receiver



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

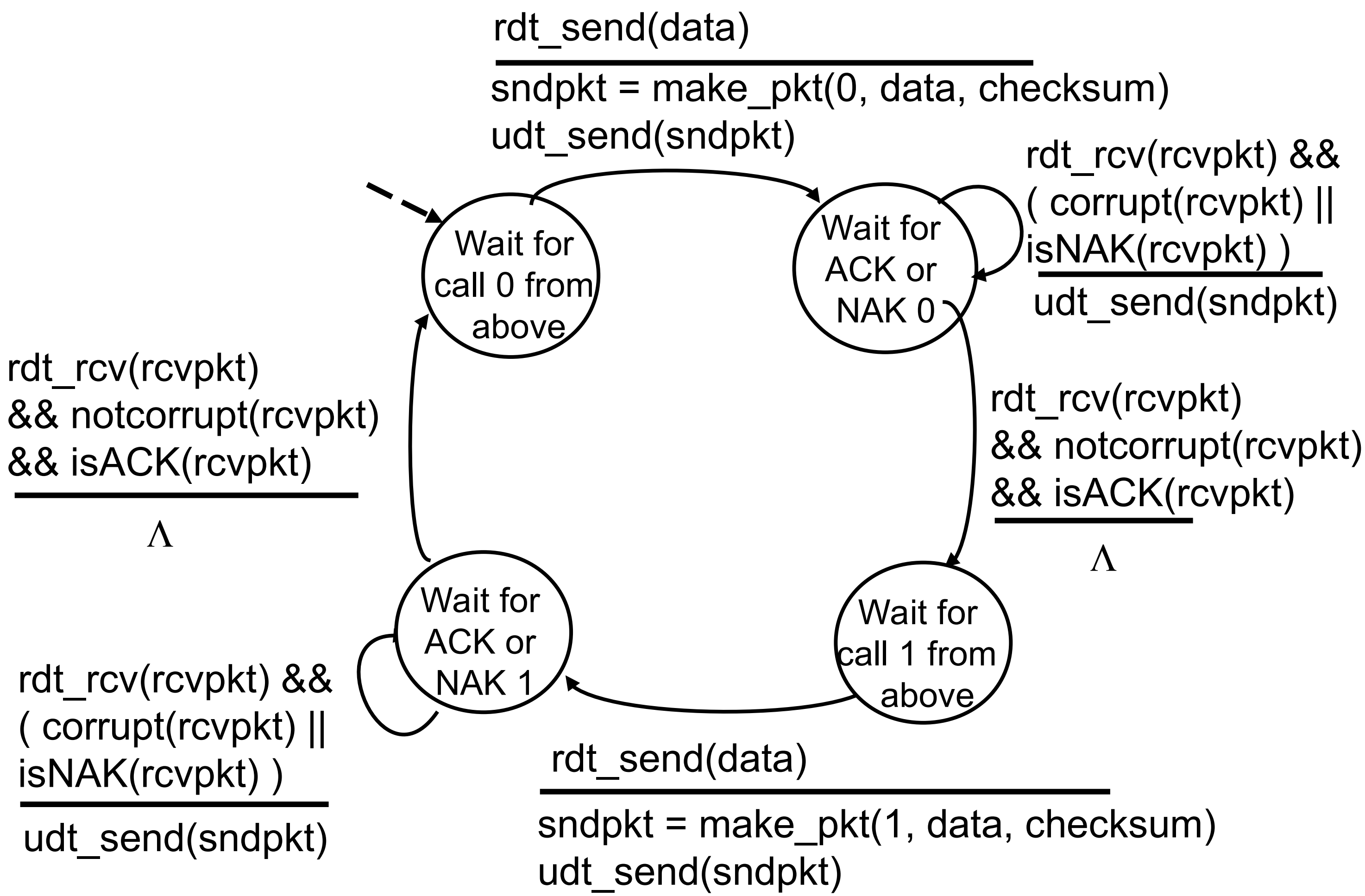
Handling duplicates:

- sender retransmits current pkt if ACK/NAK garbled
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

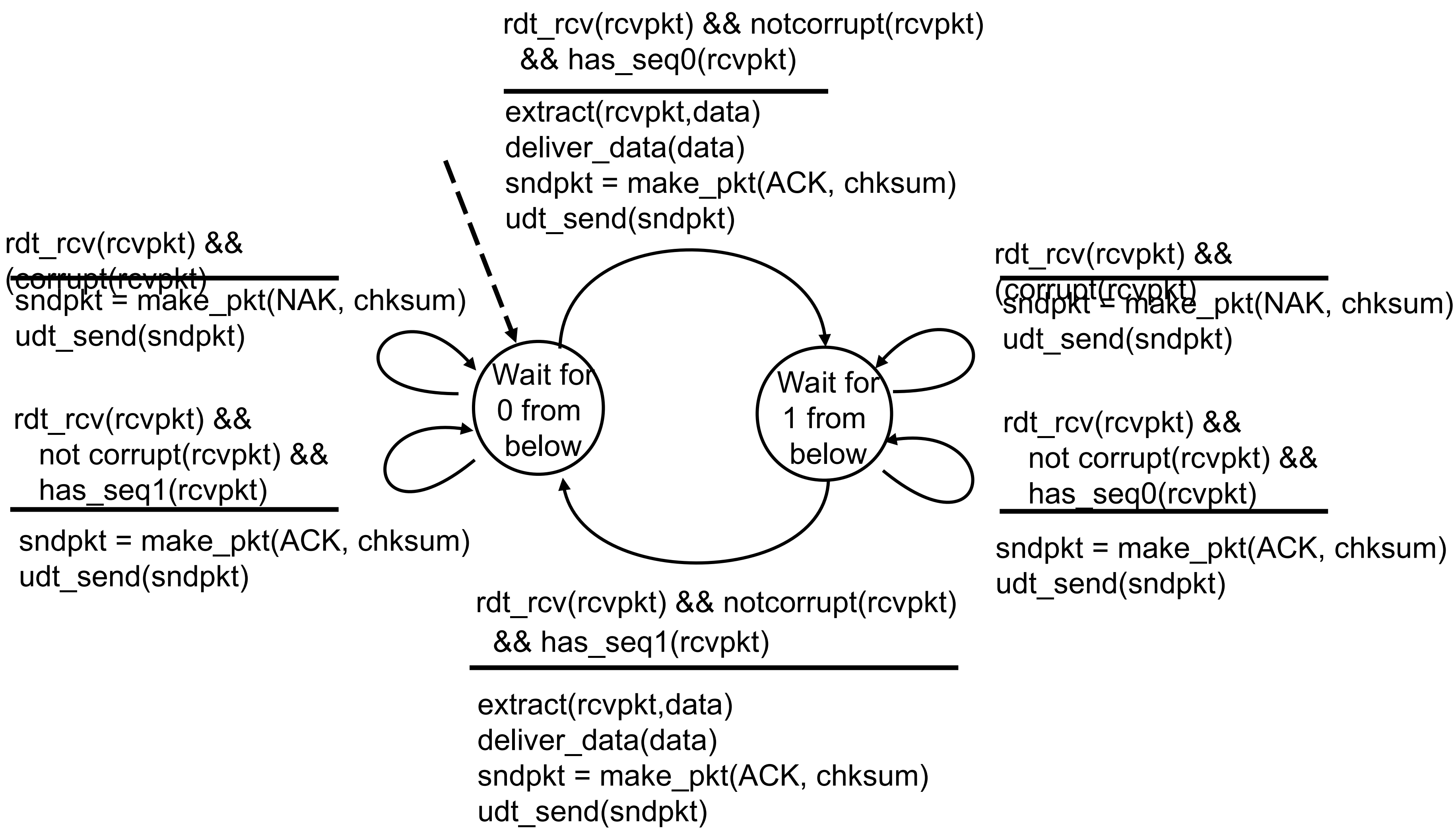
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

Sender:

- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

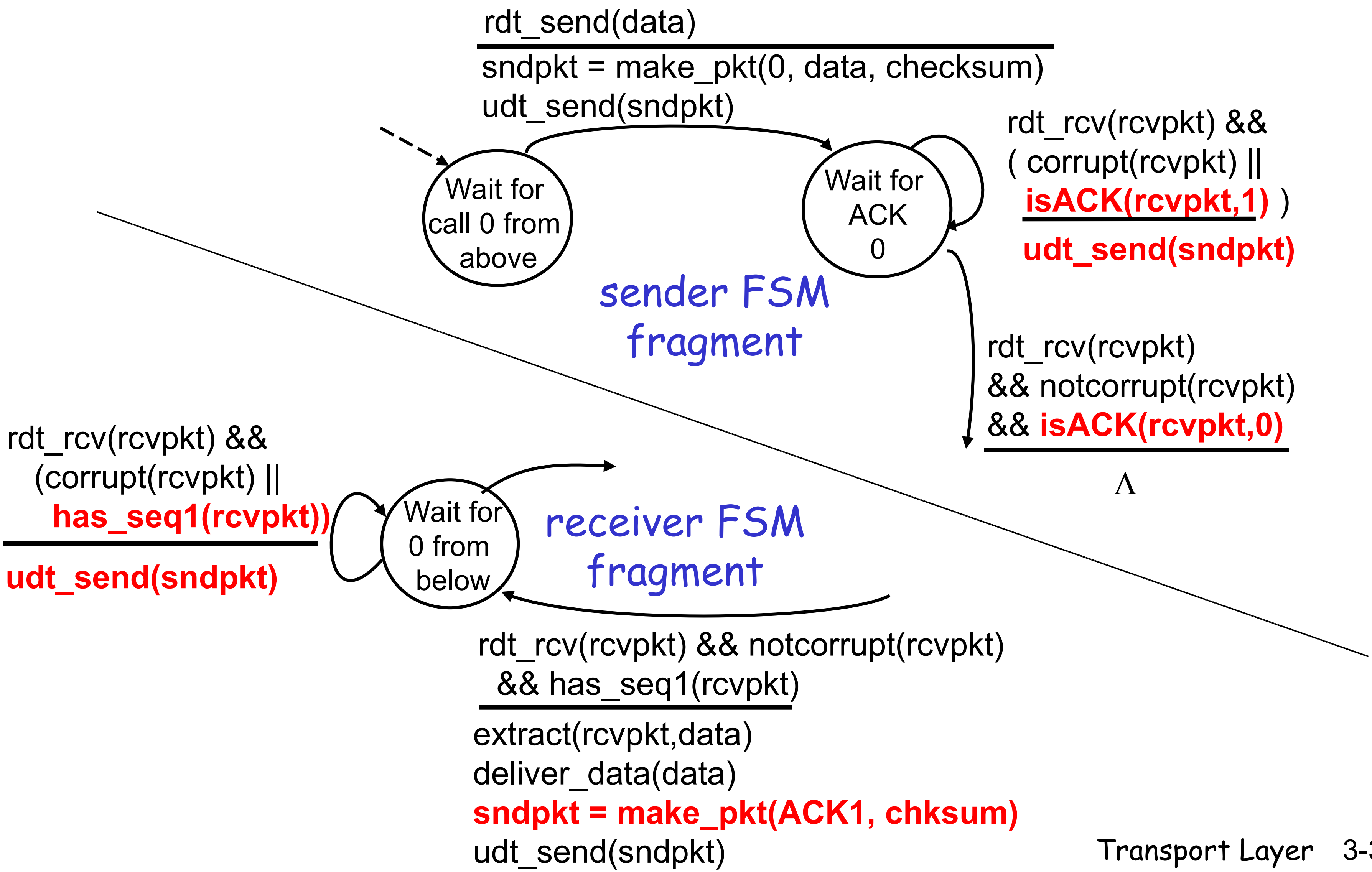
Receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

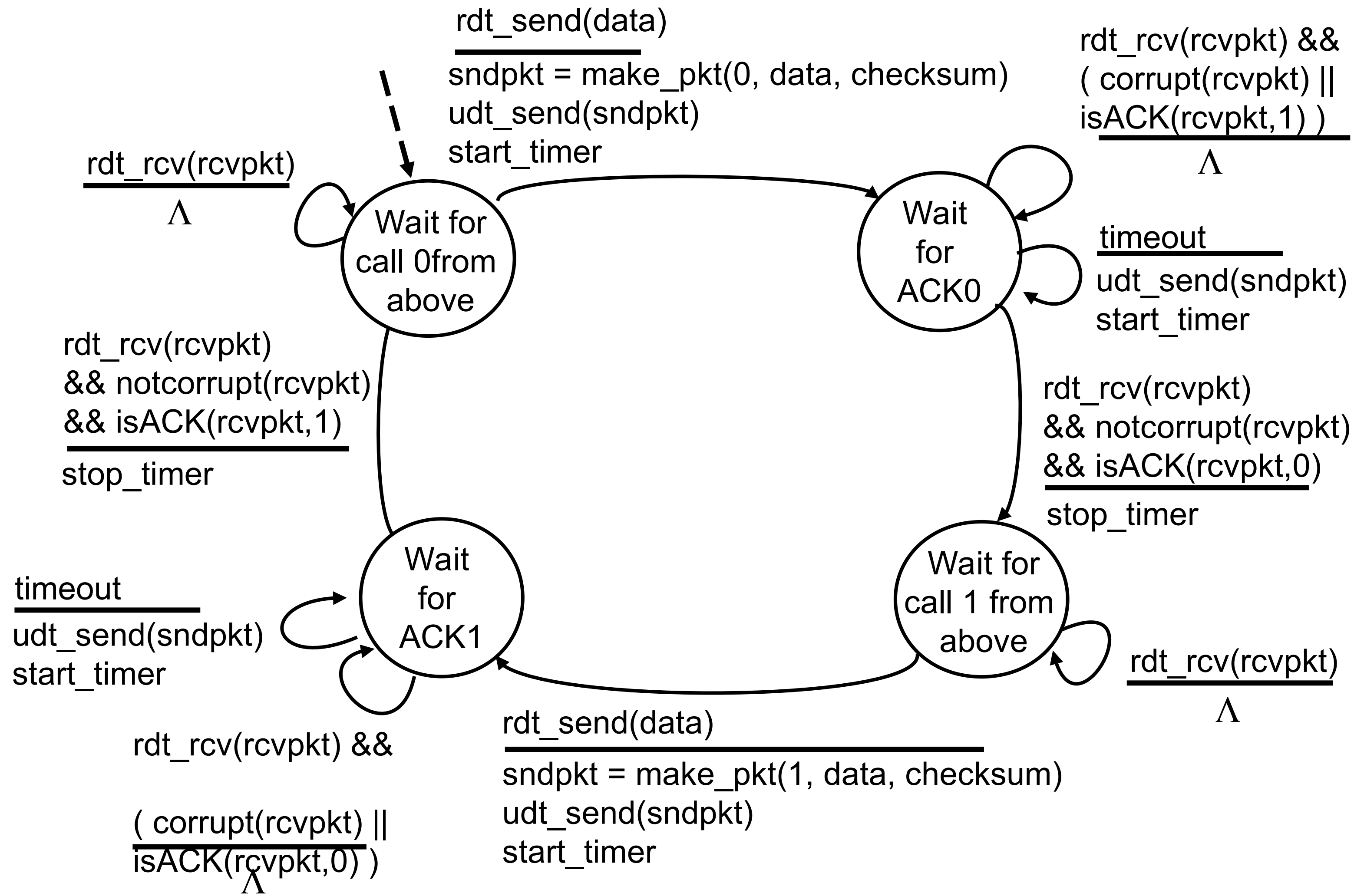
New assumption: underlying channel can also lose packets (data or ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

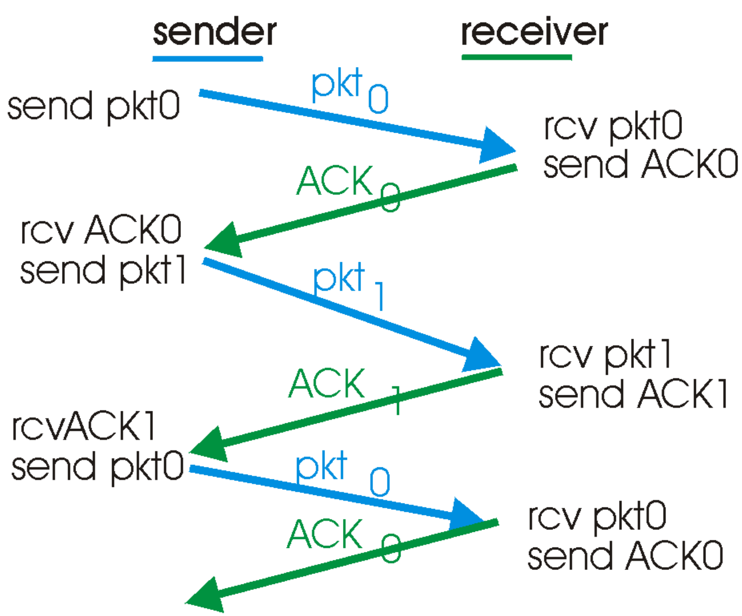
Approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

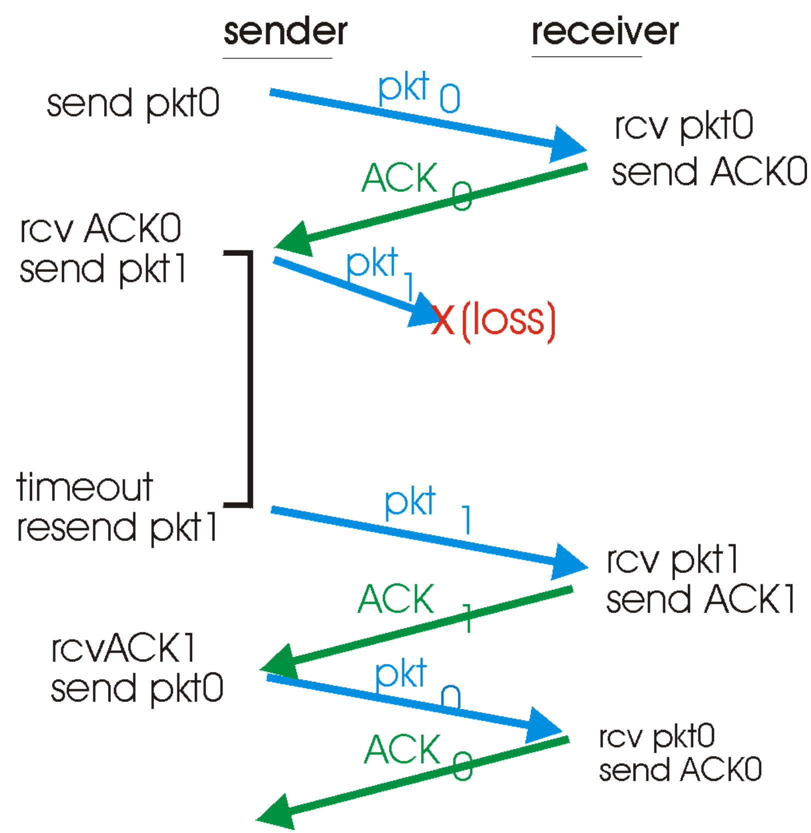
rdt3.0 sender



rdt3.0 in action

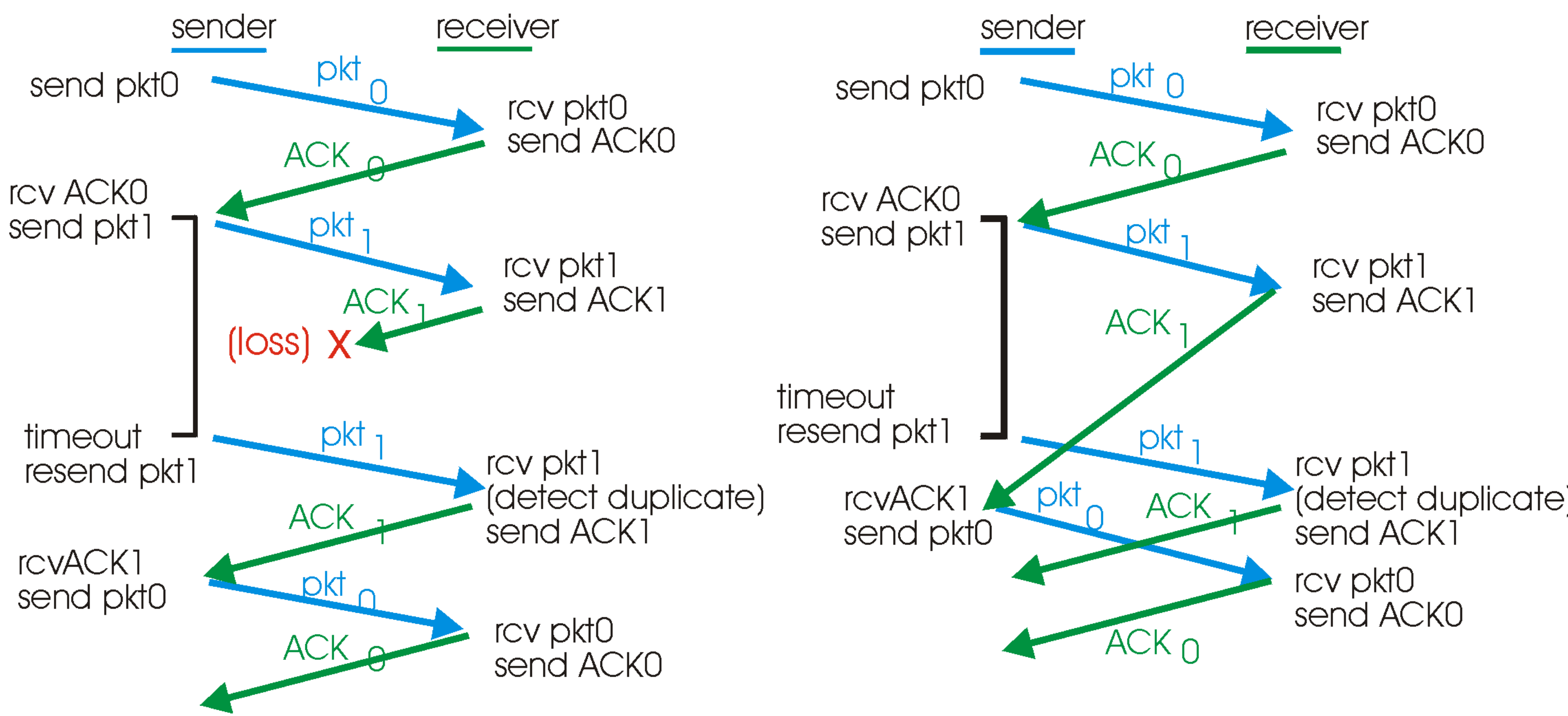


(a) operation with no loss



(b) lost packet

rdt3.0 in action



(c) lost ACK (d) premature timeout

Performance of rdt3.0

- rdt3.0 works, but performance stinks
- ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

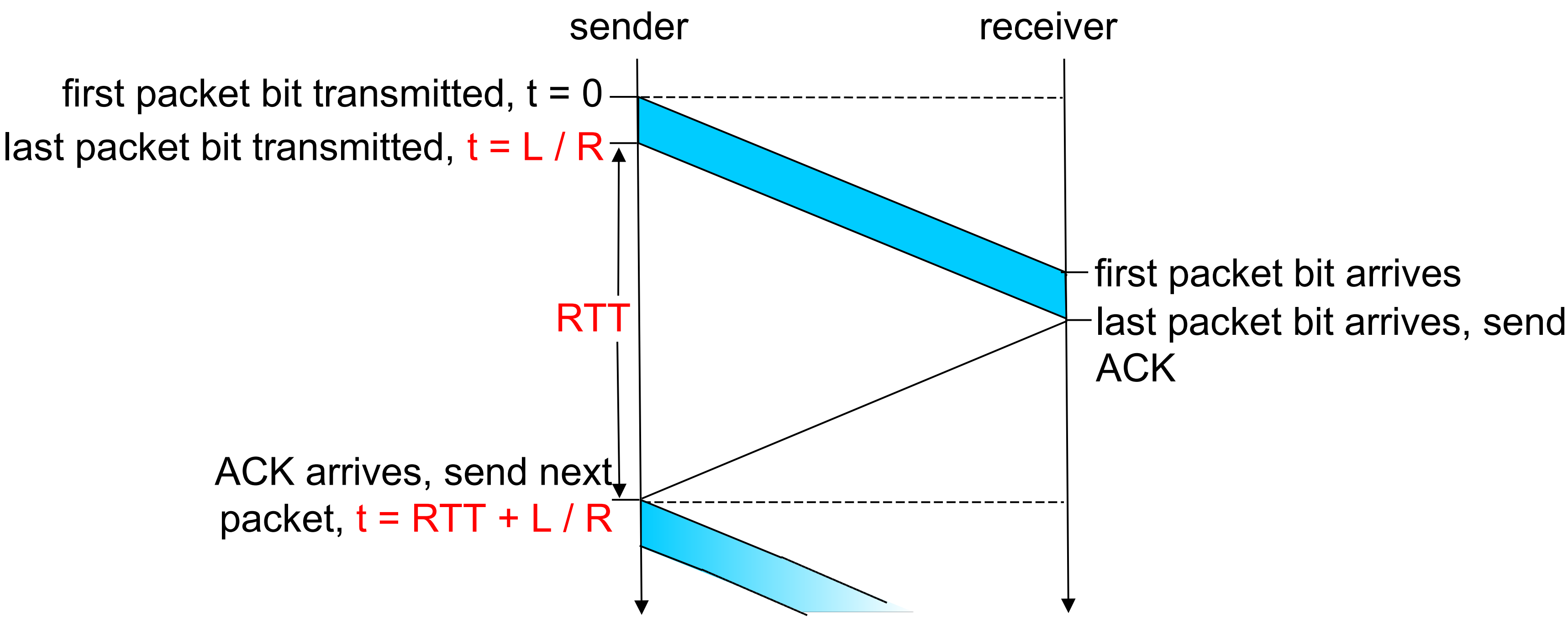
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{ bps}} = 8\text{ microseconds}$$

  - U_{sender}: utilization – fraction of time sender busy sending

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

  - 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
  - network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

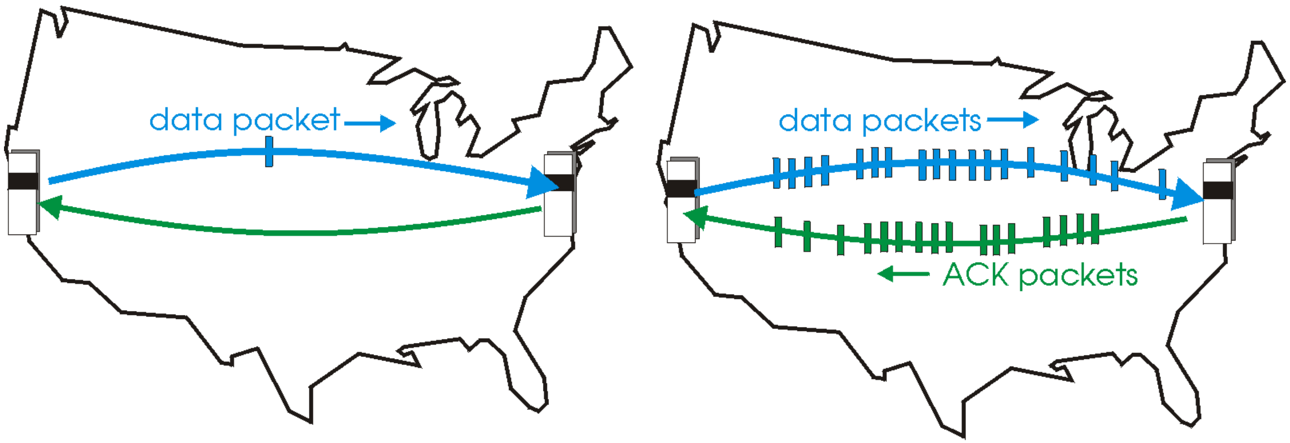


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

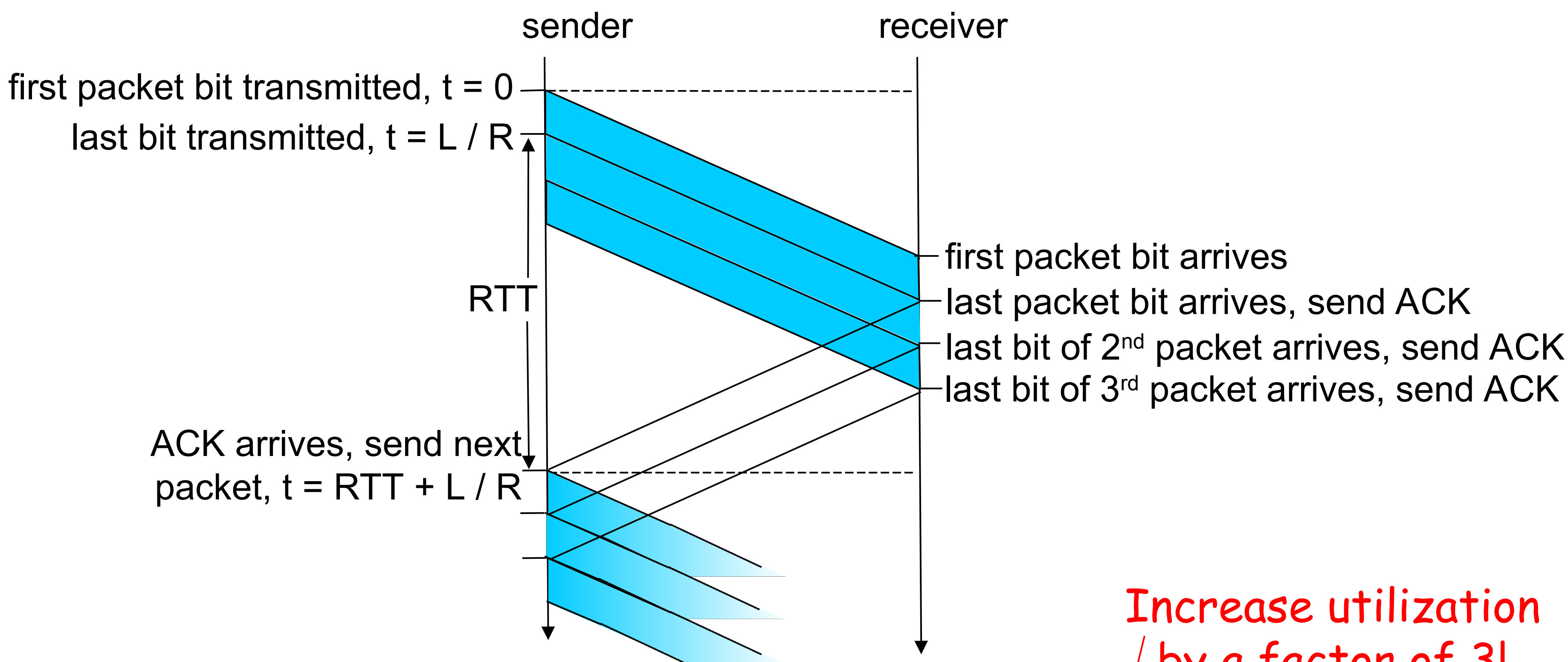


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

- Two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Pipelining Protocols

Go-back-N: overview

- *sender:* up to N unACKed pkts in pipeline
- *receiver:* only sends cumulative ACKs
  - doesn't ACK pkt if there's a gap
- *sender:* has timer for oldest unACKed pkt
  - if timer expires: retransmit all unACKed packets

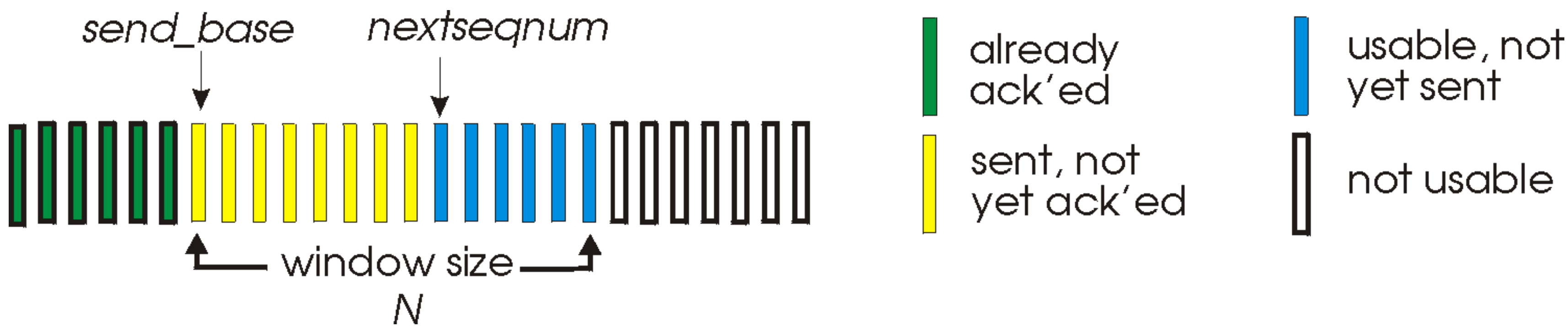
Selective Repeat: overview

- *sender:* up to N unACKed packets in pipeline
- *receiver:* ACKs individual pkts
- *sender:* maintains timer for each unACKed pkt
  - if timer expires: retransmit only unACKed packet

Go-Back-N

Sender:

- k-bit seq # in pkt header
- "window" of up to N, consecutive unACKed pkts allowed

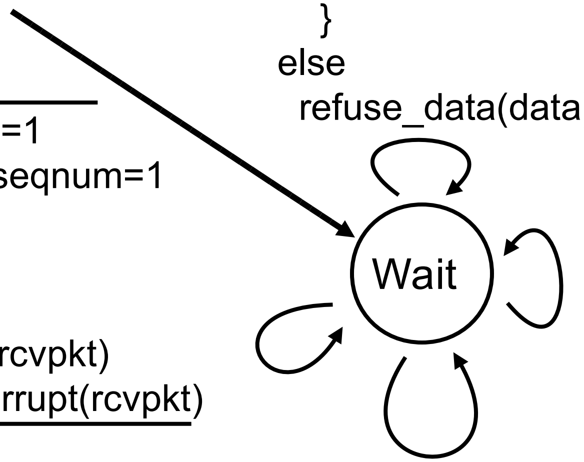


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM

```
rdt_send(data)
____________________________
if (nextseqnum < base+N) {
    sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
       start_timer
    nextseqnum++
    }
else
  refuse_data(data)
```

```
Λ
____________
base=1
nextseqnum=1
```

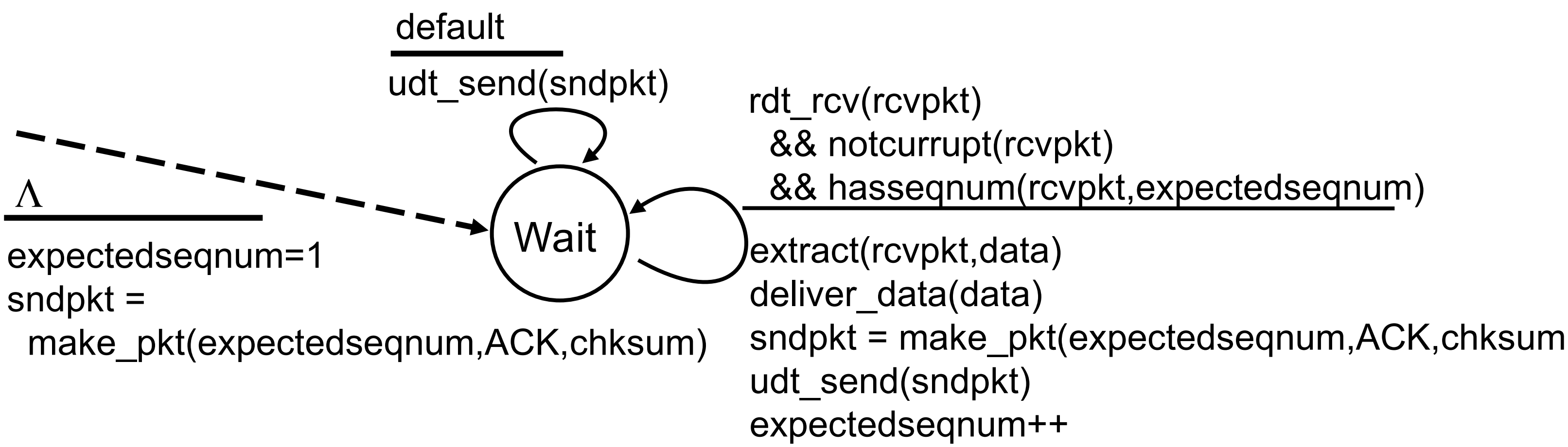



```
timeout
________________________
start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
…
udt_send(sndpkt[nextseqnum-1])
```

```
rdt_rcv(rcvpkt)
   && corrupt(rcvpkt)
______________________
```

```
rdt_rcv(rcvpkt) &&
   notcorrupt(rcvpkt)
________________________
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer
  else
    start_timer
```

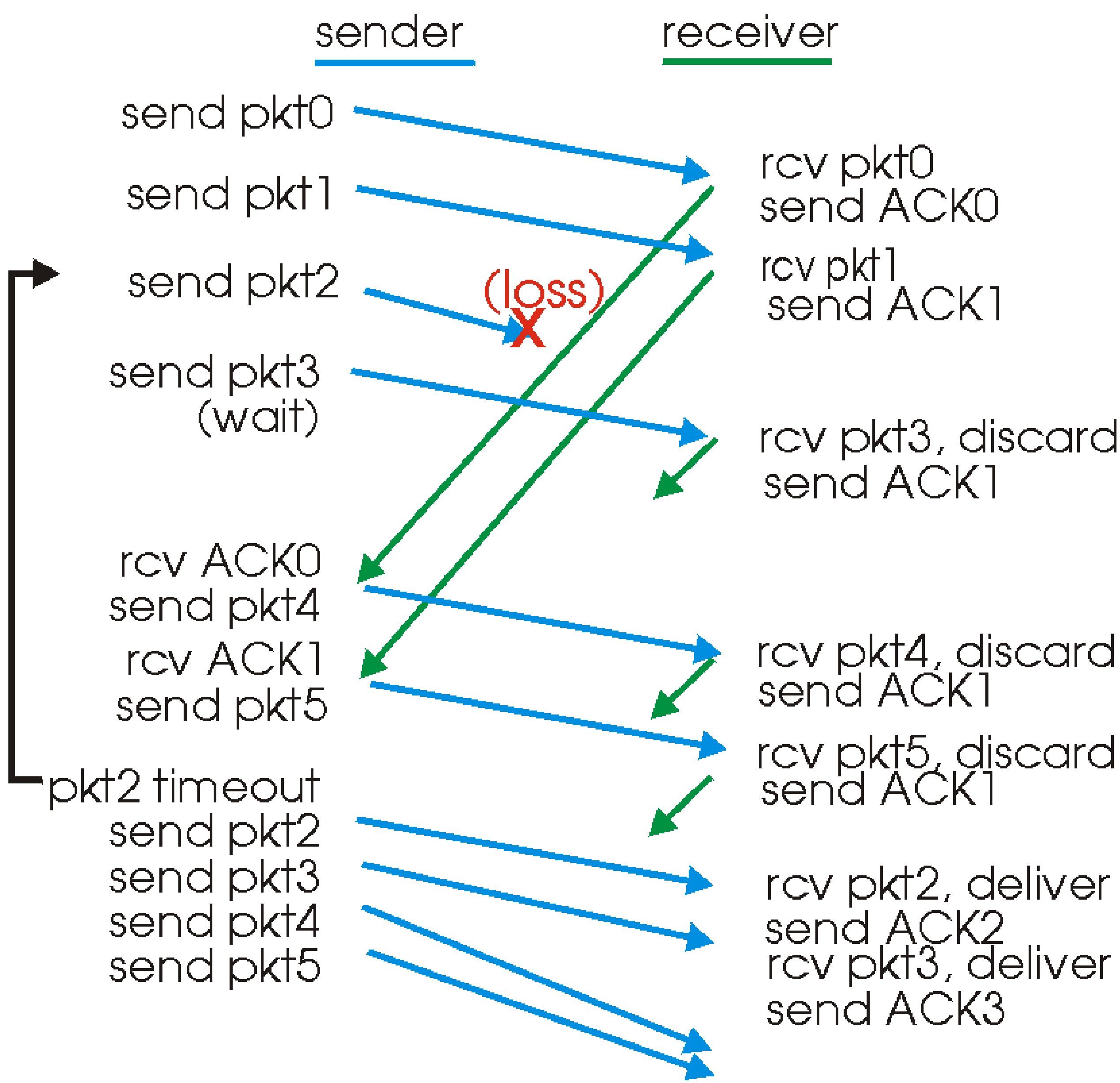
GBN: receiver extended FSM



ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #
- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK pkt with highest in-order seq #

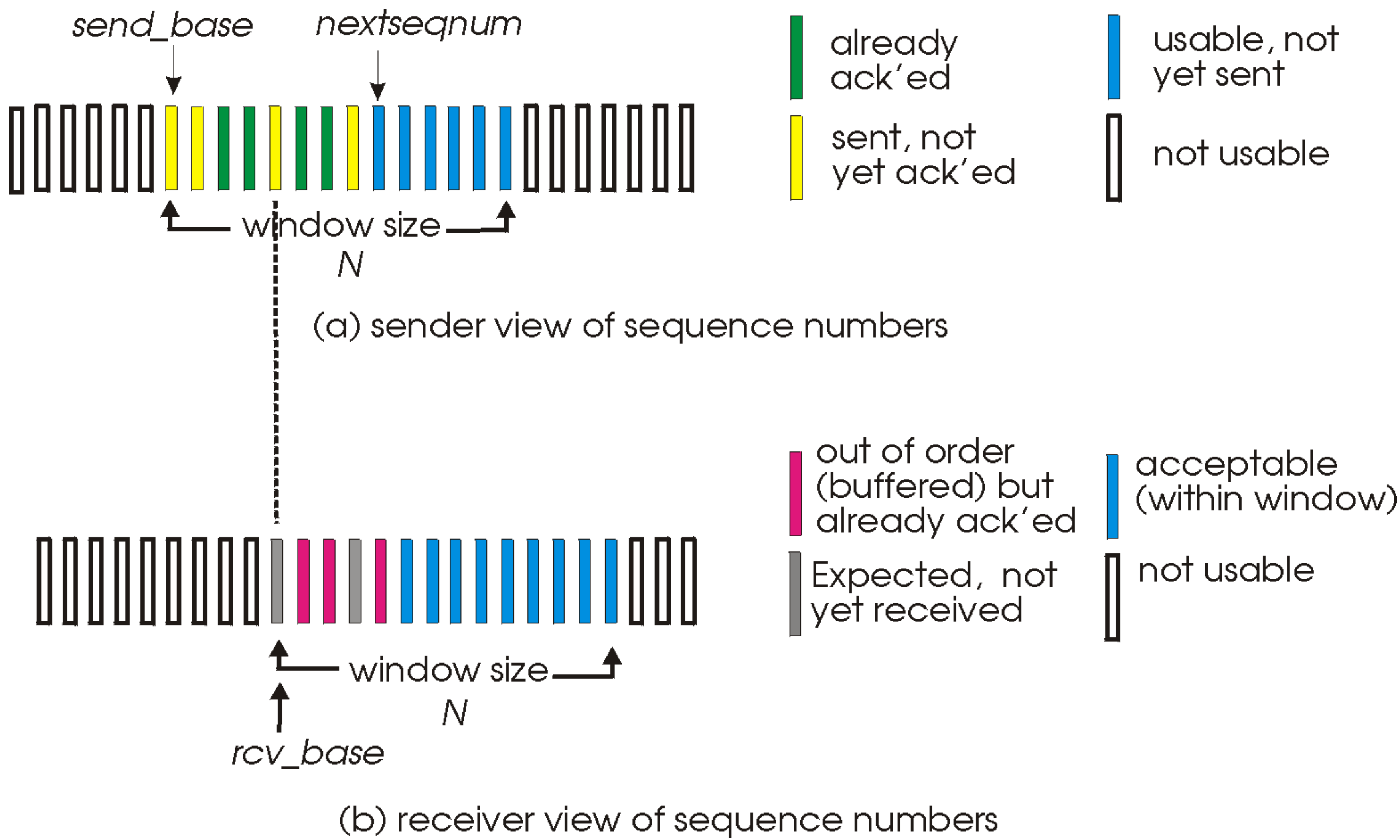
GBN in action



Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



Selective repeat

sender

data from above :

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

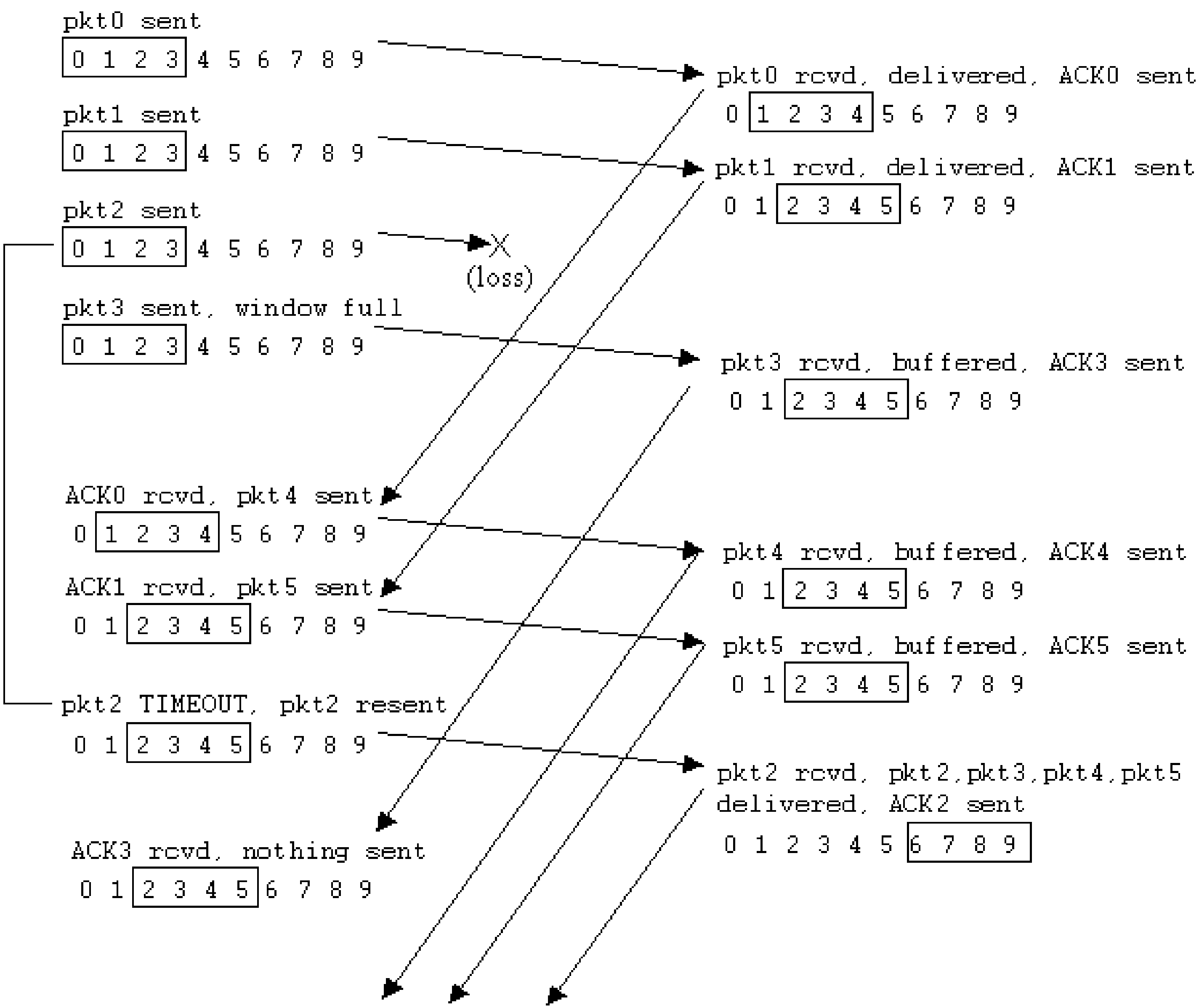
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

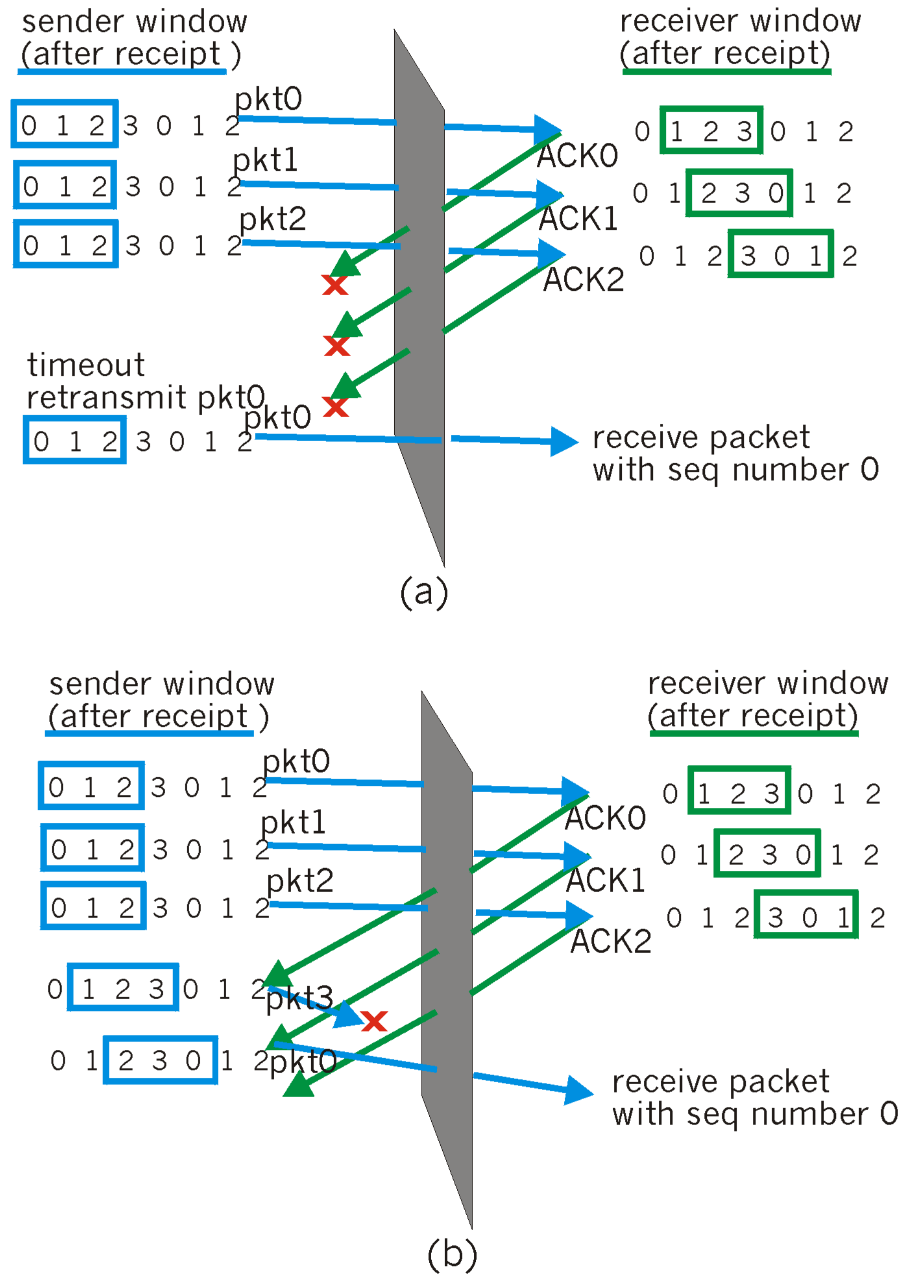
Selective repeat in action



Selective repeat: dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size=3



- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?

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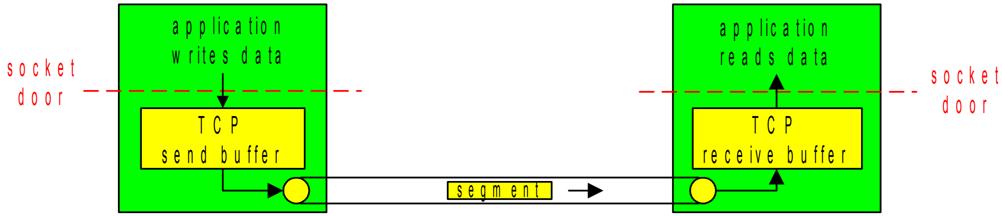
TCP: Overview

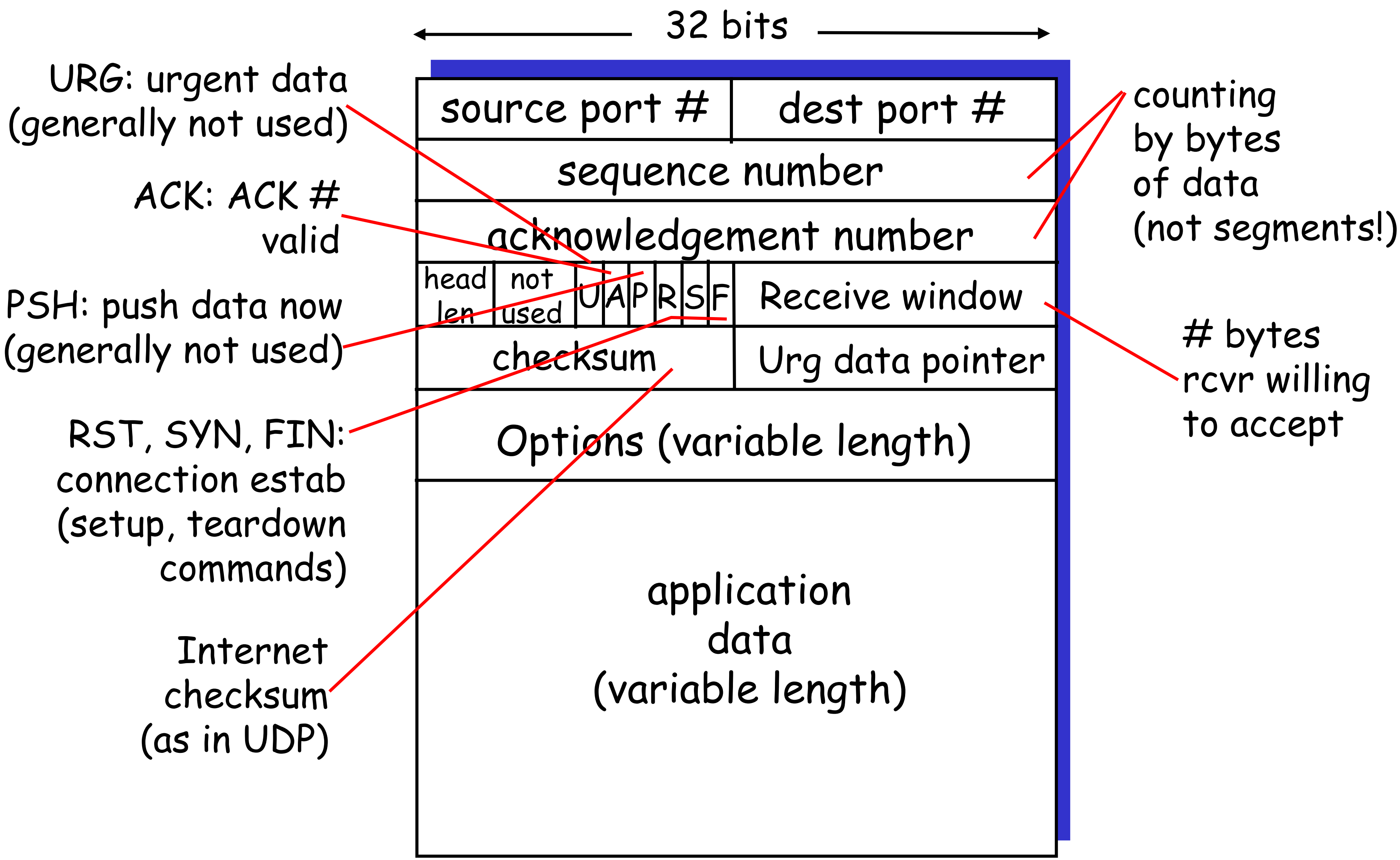
RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*



- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

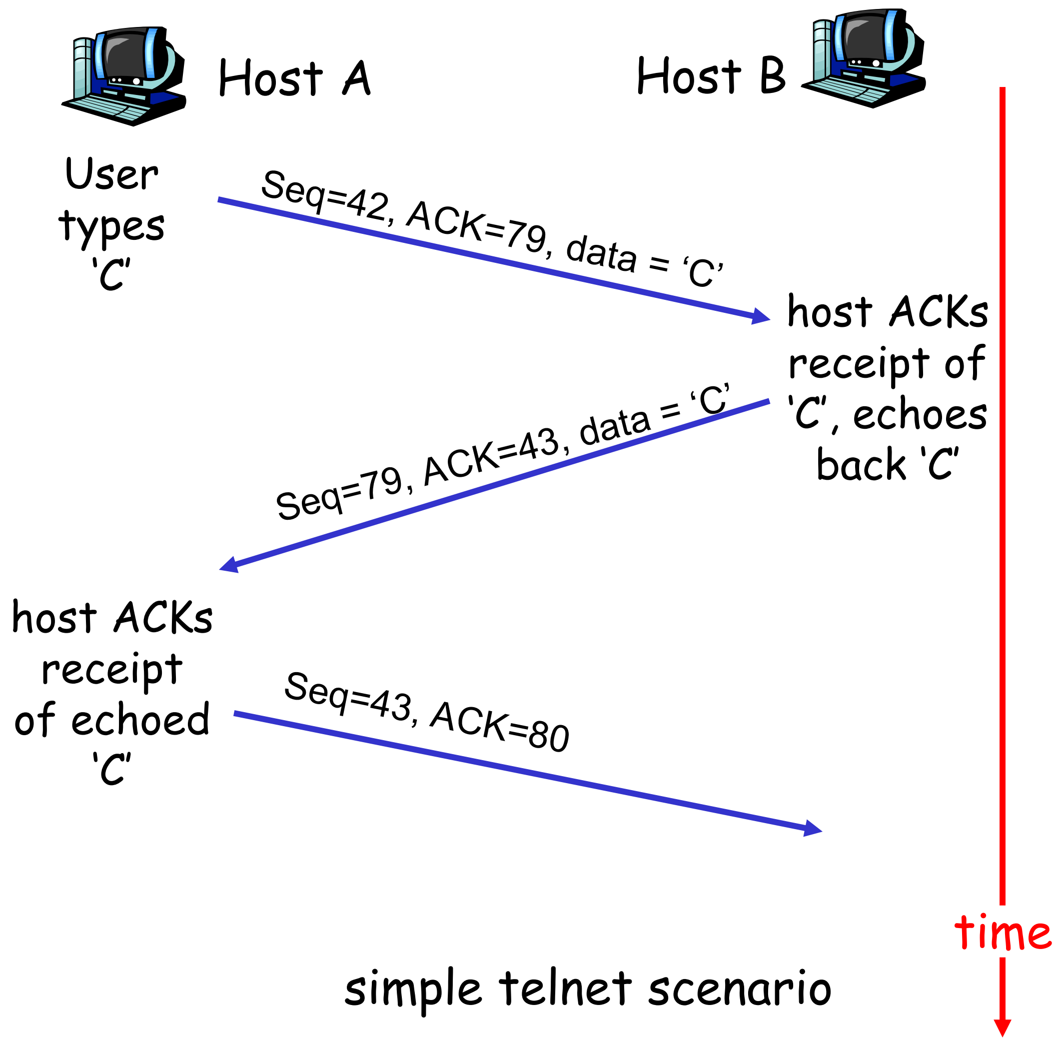
- byte stream "number" of first byte in segment's data

ACKs:

- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementer



simple telnet scenario

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- too short: premature timeout
  - unnecessary retransmissions
- too long: slow reaction to segment loss

Q: how to estimate RTT?

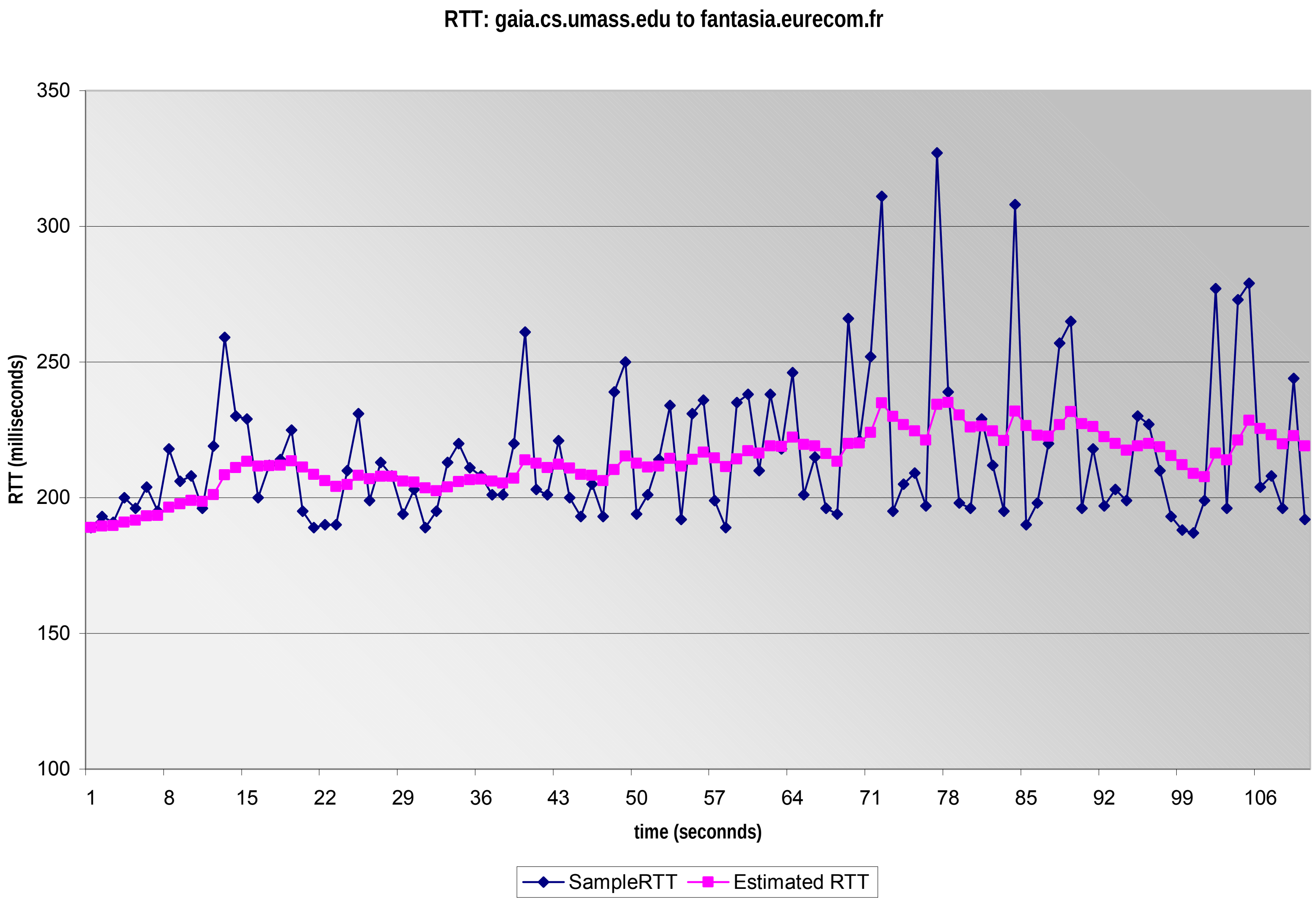
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current `SampleRTT`

TCP Round Trip Time and Timeout

$$\texttt{EstimatedRTT} = (1-\alpha)*\texttt{EstimatedRTT} + \alpha*\texttt{SampleRTT}$$

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

- **`EstimtedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

DevRTT = (1-β)*DevRTT +
β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)

Then set timeout interval:

TimeoutInterval = EstimatedRTT + 4*DevRTT

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TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- pipelined segments
- cumulative ACKs
- TCP uses single retransmission timer
- retransmissions are triggered by:
  - timeout events
  - duplicate ACKs
- initially consider simplified TCP sender:
  - ignore duplicate ACKs
  - ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running (think of timer as for oldest unACKed segment)
- expiration interval: `TimeOutInterval`

timeout:

- retransmit segment that caused timeout
- restart timer

ACK rcvd:

- if acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are outstanding segments

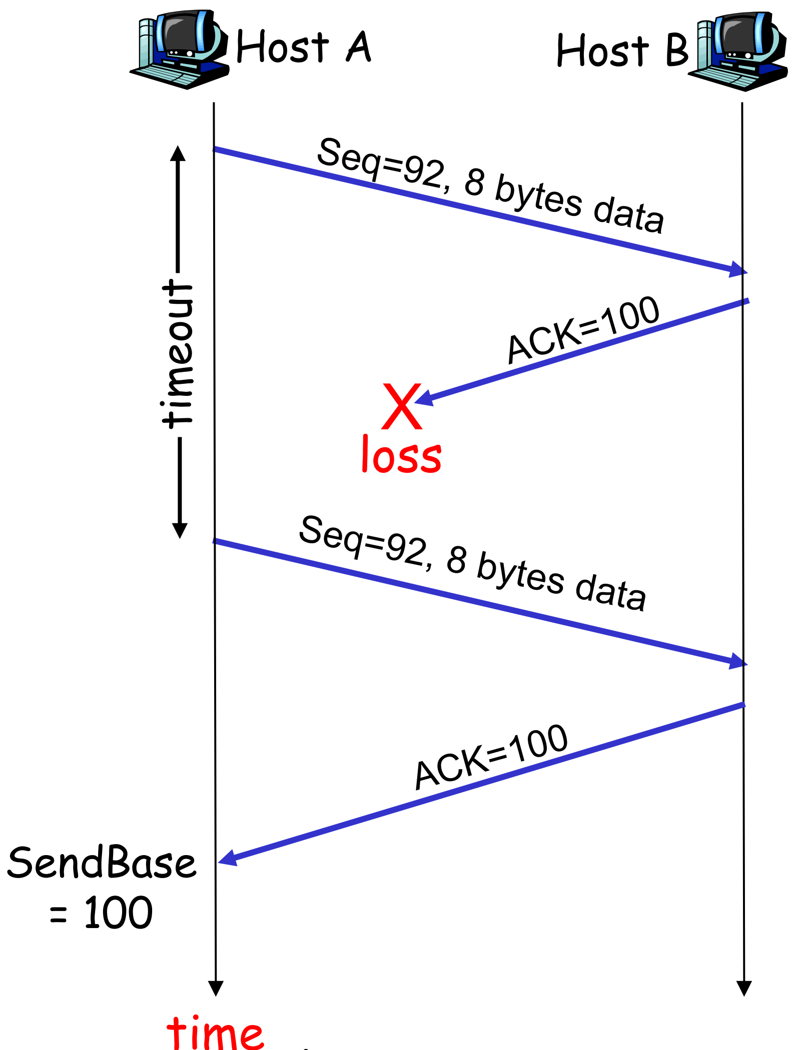
TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

 loop (forever) {
   switch(event)

   event: data received from application above
         create TCP segment with sequence number NextSeqNum
         if (timer currently not running)
              start timer
         pass segment to IP
         NextSeqNum = NextSeqNum + length(data)

    event: timer timeout
         retransmit not-yet-acknowledged segment with
                smallest sequence number
         start timer

    event: ACK received, with ACK field value of y
         if (y > SendBase) {
              SendBase = y
             if (there are currently not-yet-acknowledged segments)
                    start timer
             }

  }  /* end of loop forever */
```

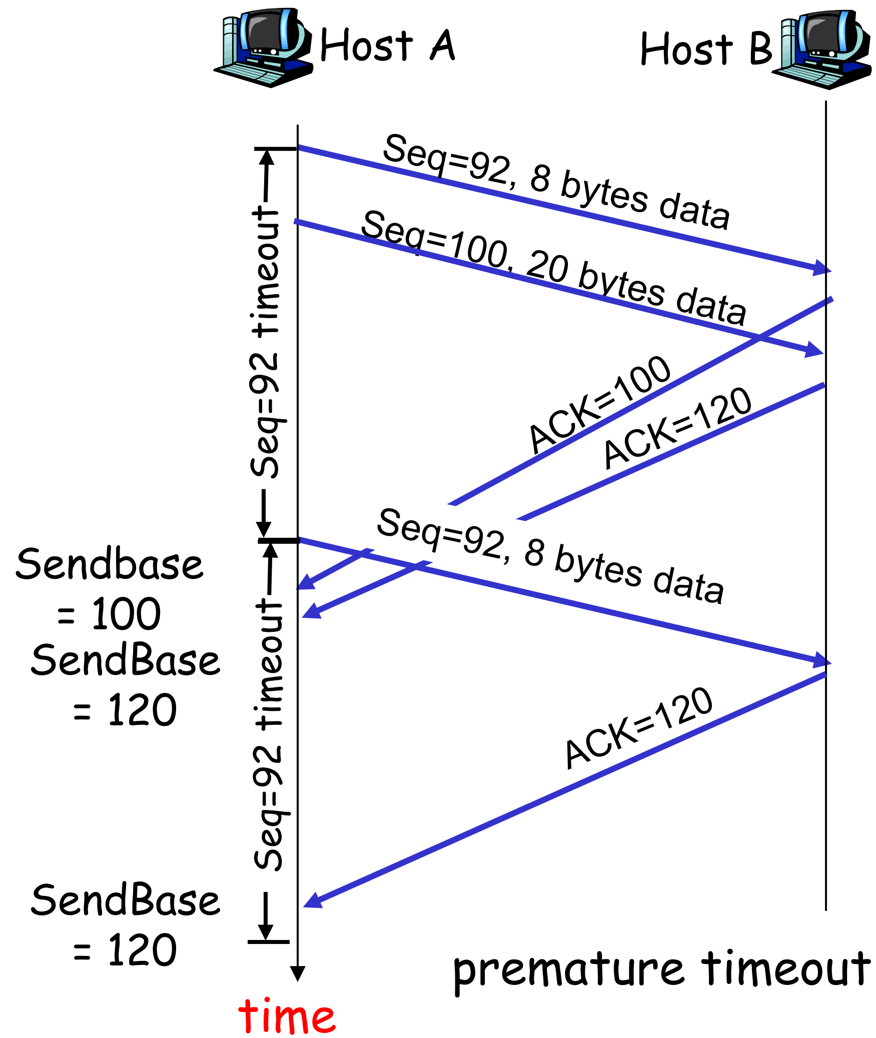
Comment:
- SendBase-1: last cumulatively ACKed byte

Example:
- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is ACKed

TCP: retransmission scenarios

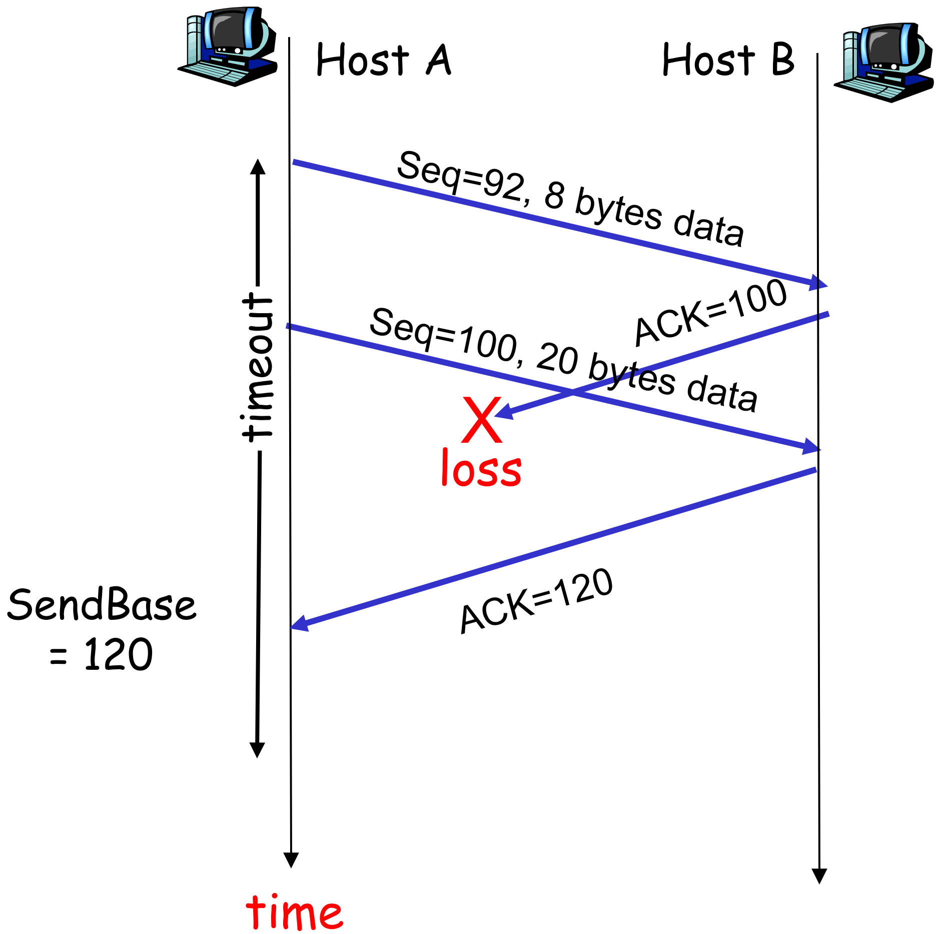


lost ACK scenario



premature timeout

TCP retransmission scenarios (more)



Cumulative ACK scenario

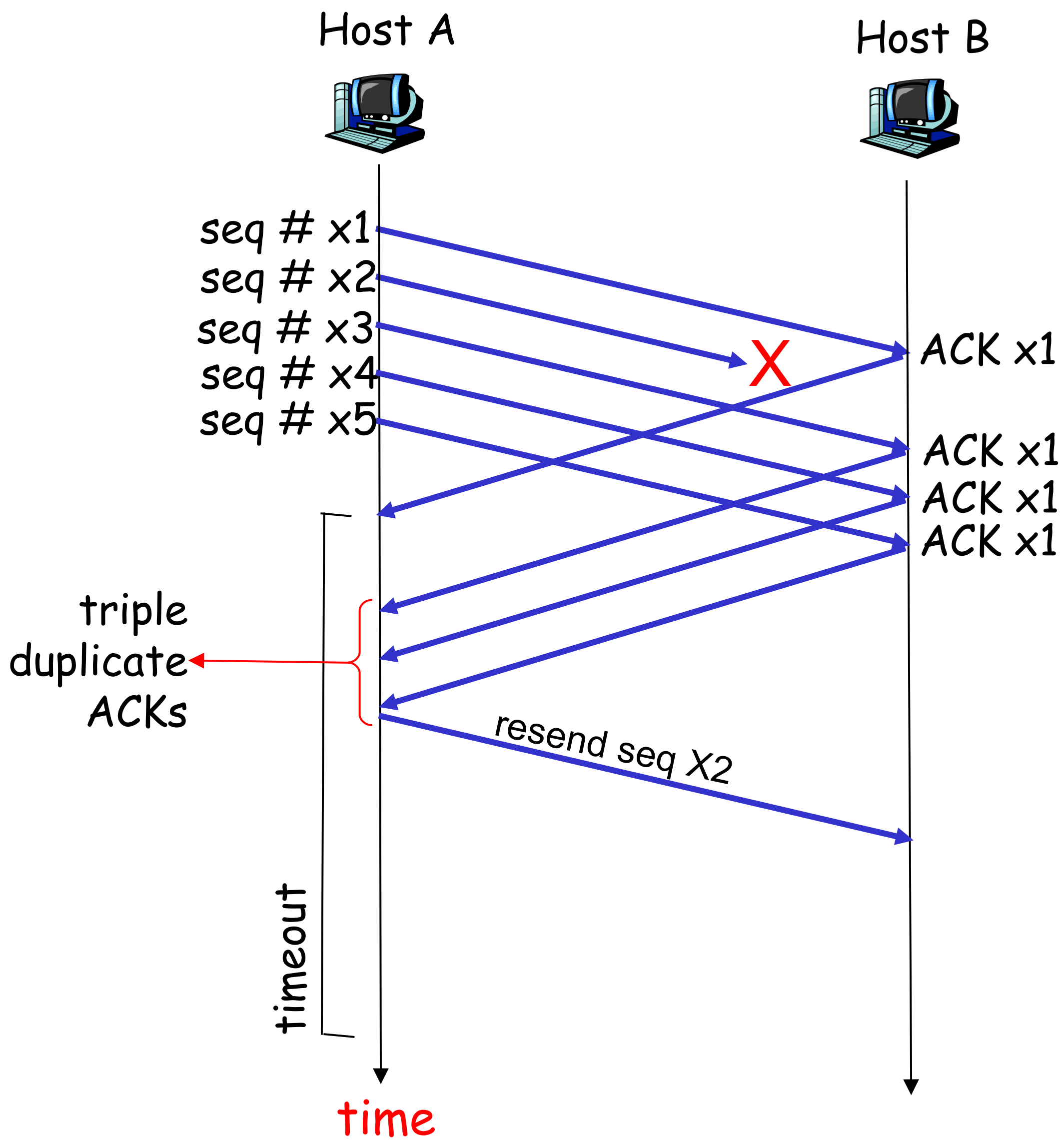
TCP ACK generation [RFC 1122, RFC 2581]

Event at Receiver	TCP Receiver action
Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
Arrival of in-order segment with expected seq #. One other segment has ACK pending	Immediately send single cumulative ACK, ACKing both in-order segments
Arrival of out-of-order segment higher-than-expect seq. # . Gap detected	Immediately send *duplicate ACK*, indicating seq. # of next expected byte
Arrival of segment that partially or completely fills gap	Immediate send ACK, provided that segment starts at lower end of gap

Fast Retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs for that segment

- If sender receives 3 ACKs for same data, it assumes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires


Host A
Host B
seq # x1
seq # x2
seq # x3
seq # x4
seq # x5
X
ACK x1
ACK x1
ACK x1
ACK x1
triple
duplicate
ACKs
resend seq X2
timeout
time

Fast retransmit algorithm:

```
event: ACK received, with ACK field value of y
          if (y > SendBase) {
              SendBase = y
              if (there are currently not-yet-acknowledged segments)
                  start timer
             }
          else {
               increment count of dup ACKs received for y
               if (count of dup ACKs received for y = 3) {
                   resend segment with sequence number y
               }
```

a duplicate ACK for already ACKed segment

fast retransmit

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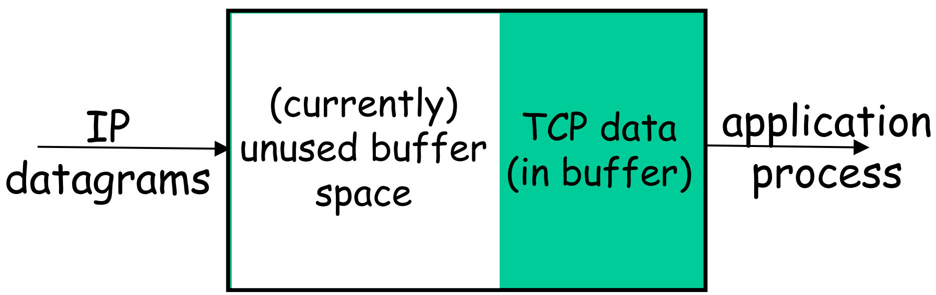
TCP Flow Control

- receive side of TCP connection has a receive buffer:



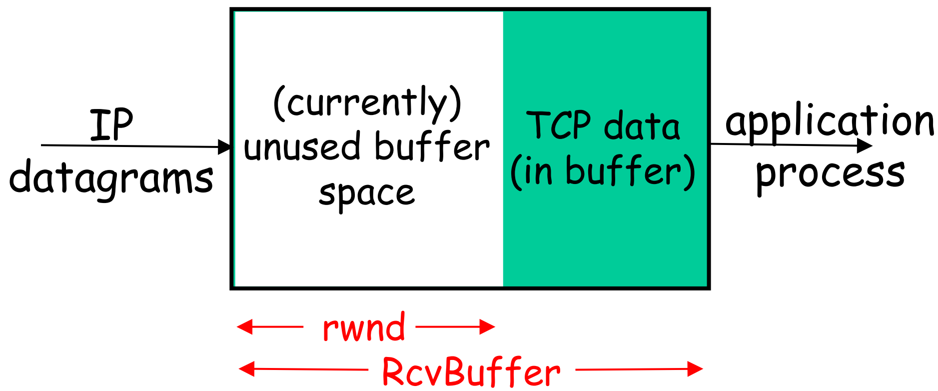
- app process may be slow at reading from buffer

flow control

sender won't overflow receiver's buffer by transmitting too much,
too fast

- *speed-matching service:* matching send rate to receiving application's drain rate

TCP Flow control: how it works



(suppose TCP receiver discards out-of-order segments)

- unused buffer space:

= **rwnd**

= **RcvBuffer-[LastByteRcvd - LastByteRead]**

- receiver: advertises unused buffer space by including `rwnd` value in segment header
- sender: limits # of unACKed bytes to `rwnd`
  - guarantees receiver's buffer doesn't overflow

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TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. **RcvWindow**)
- *client:* connection initiator

```
Socket clientSocket = new
 Socket("hostname","port

 number");
```

- *server:* contacted by client

```
Socket connectionSocket =
 welcomeSocket.accept();
```

Three way handshake:

Step 1: client host sends TCP SYN segment to server
- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment
- server allocates buffers
- specifies server initial seq. #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

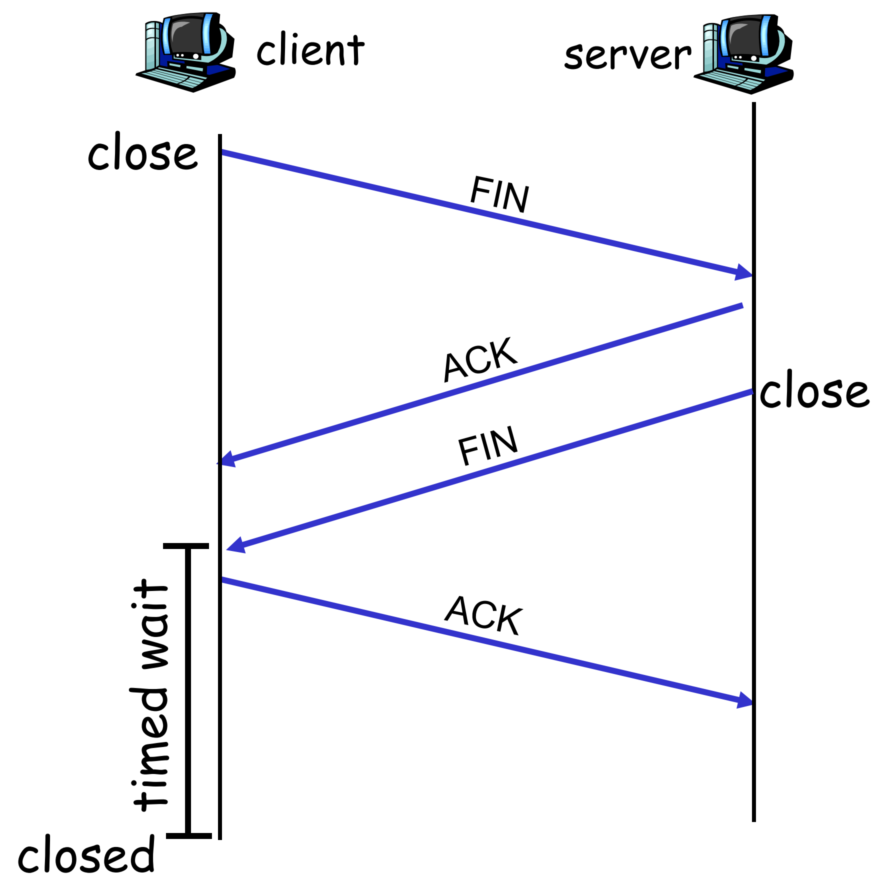
TCP Connection Management (cont.)

Closing a connection:

client closes socket:
```
clientSocket.close();
```

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.



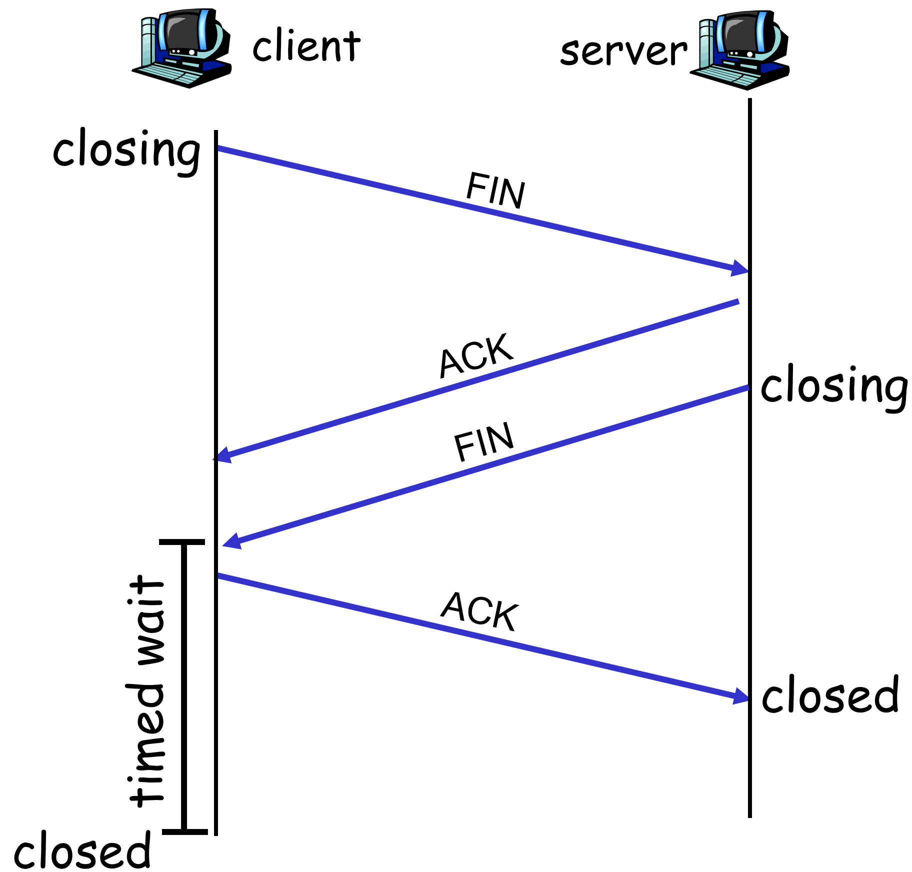
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

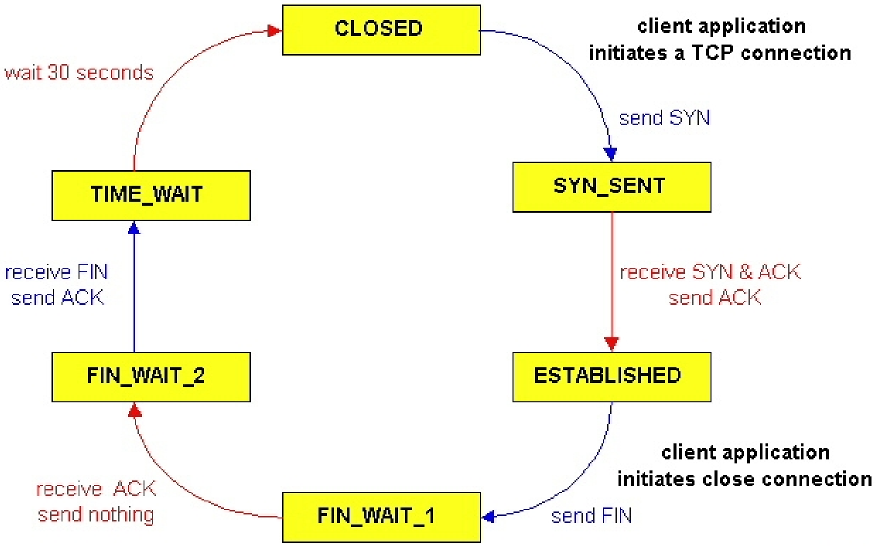
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

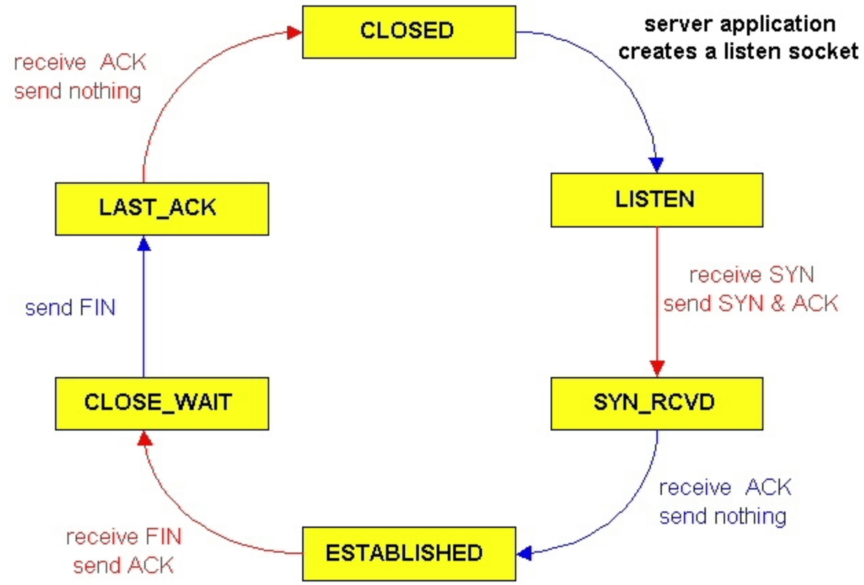


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



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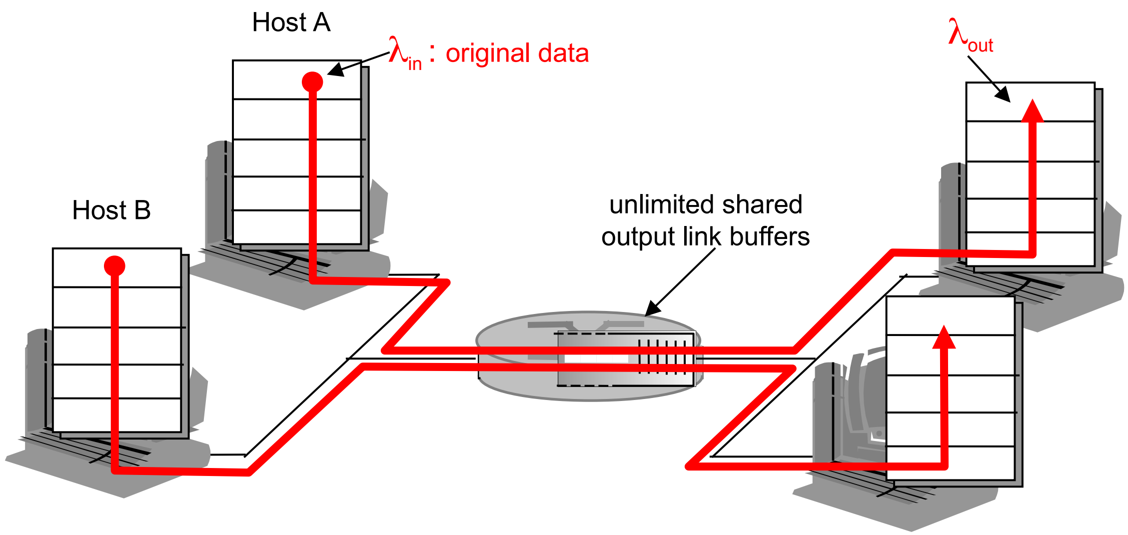
Principles of Congestion Control

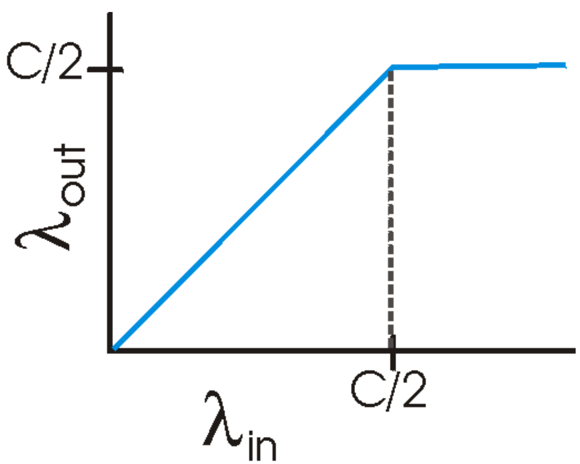
Congestion:

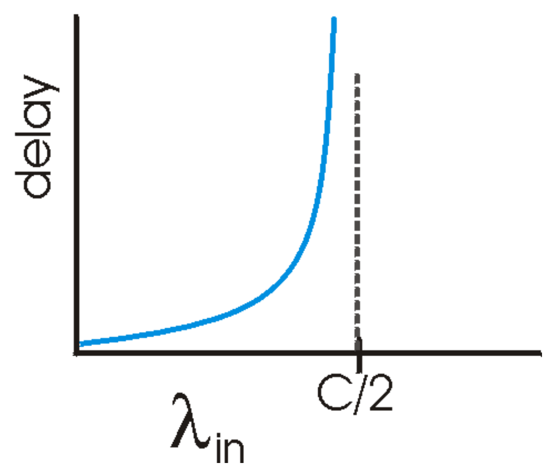
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



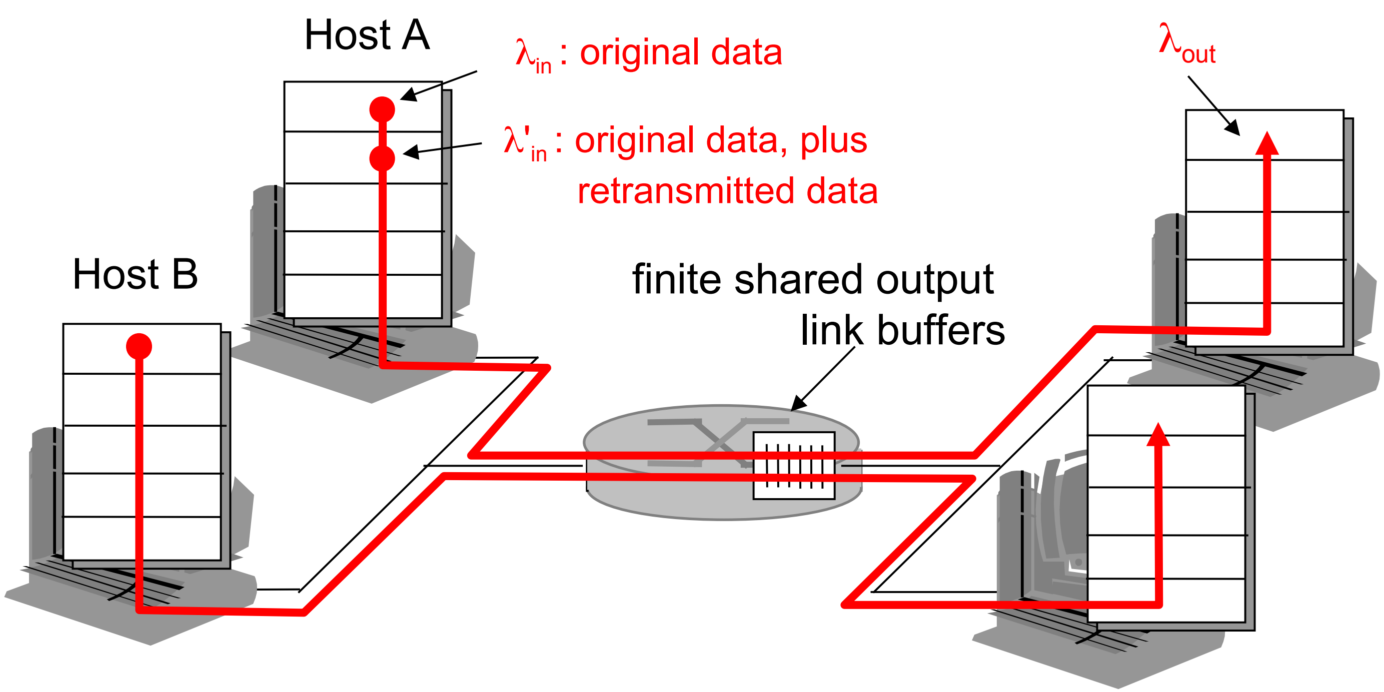




- large delays when congested
- maximum achievable throughput

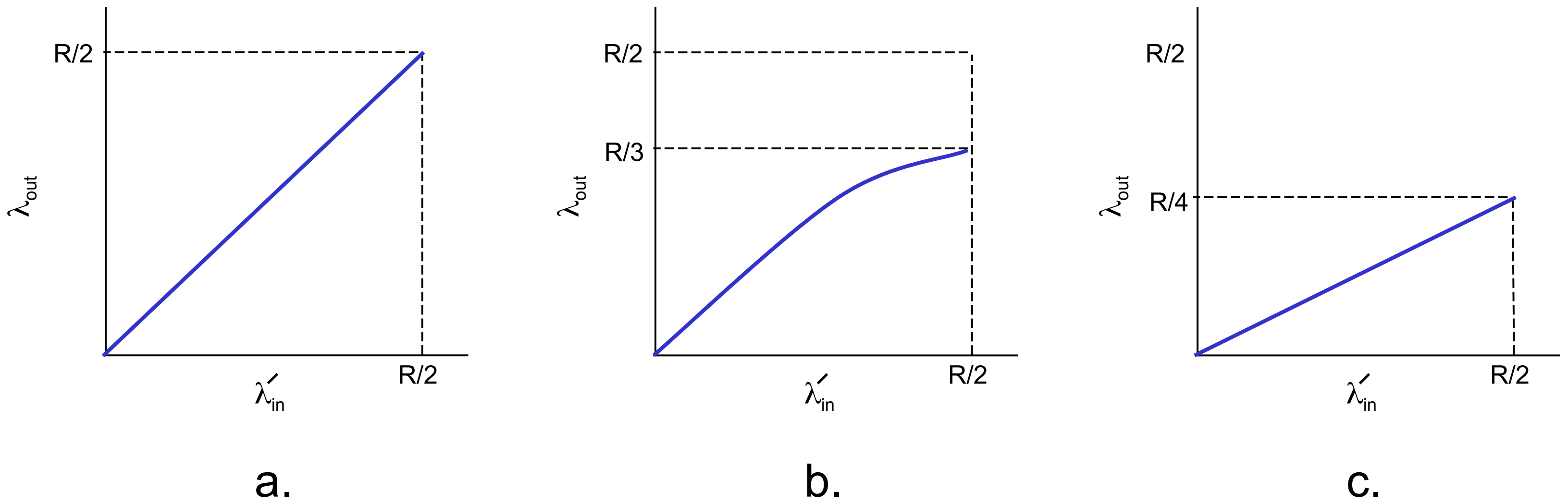
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



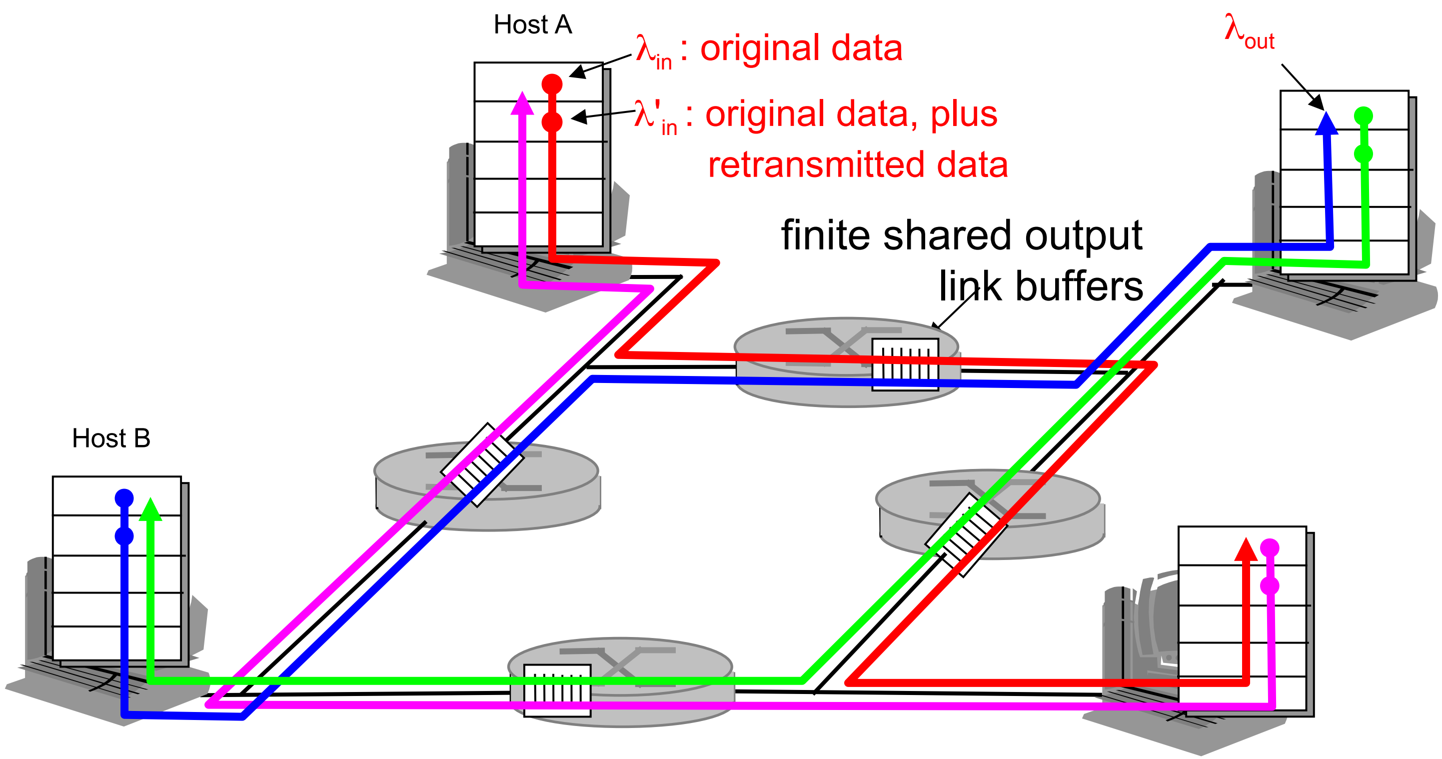
"costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

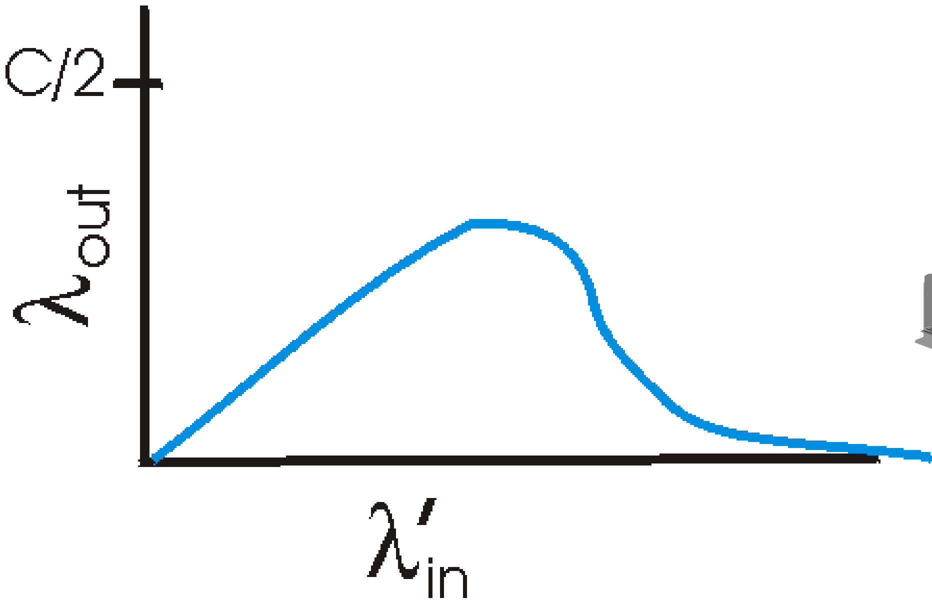
Causes/costs of congestion: scenario 3

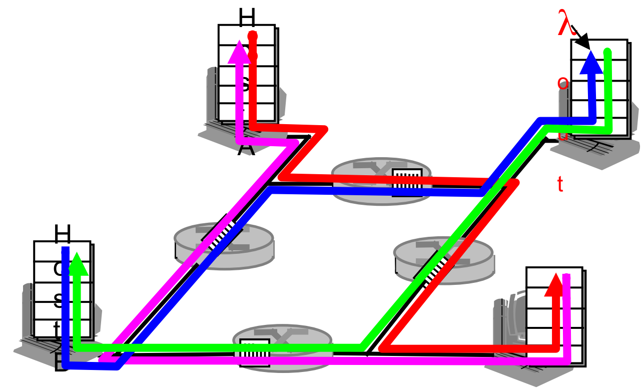
- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3





another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!

Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

Case study: ATM ABR congestion control

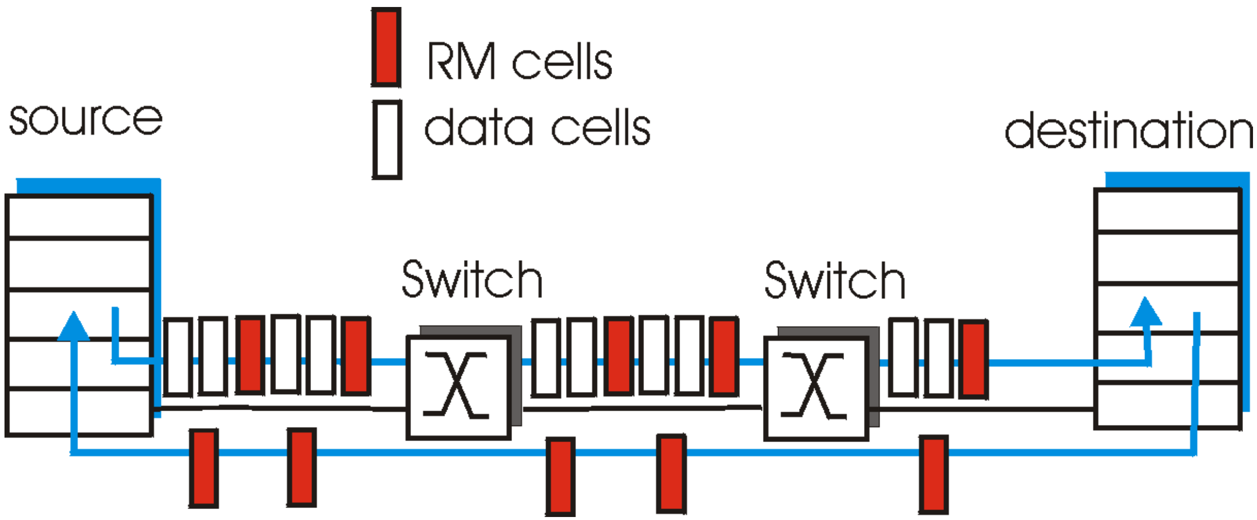
ABR: available bit rate:

- "elastic service"
- if sender's path "underloaded":
  - sender should use available bandwidth
- if sender's path congested:
  - sender throttled to minimum guaranteed rate

RM (resource management) cells:

- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - NI bit: no increase in rate (mild congestion)
  - CI bit: congestion indication
- RM cells returned to sender by receiver, with bits intact

Case study: ATM ABR congestion control



- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - sender' send rate thus maximum supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell

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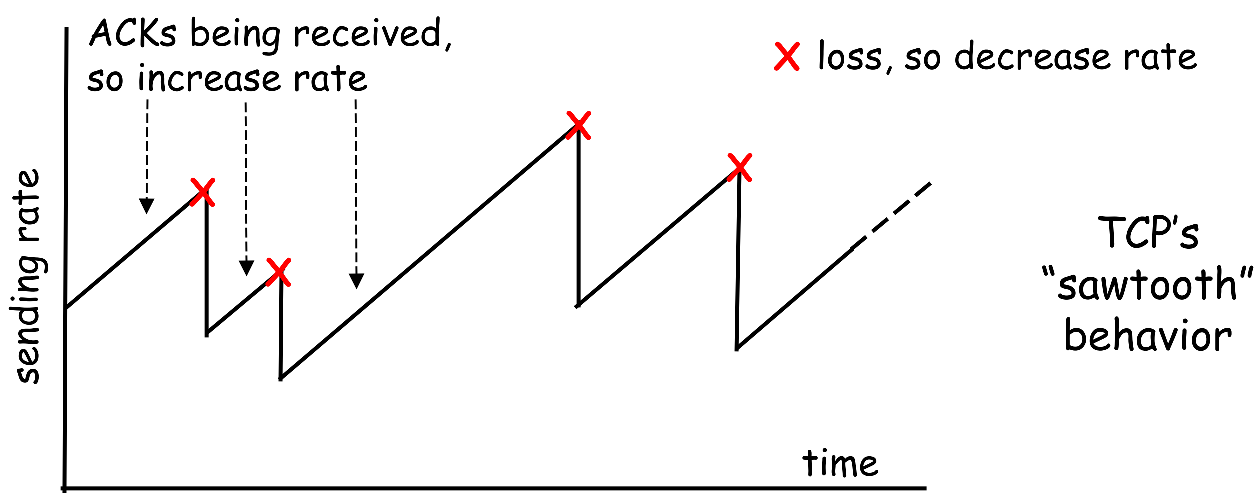
TCP congestion control:

- *goal:* TCP sender should transmit as fast as possible, but without congesting network
  - Q: how to find rate *just* below congestion level
- decentralized: each TCP sender sets its own rate, based on *implicit* feedback:
  - *ACK:* segment received (a good thing!), network not congested, so increase sending rate
  - *lost segment:* assume loss due to congested network, so decrease sending rate

TCP congestion control: bandwidth probing

- "probing for bandwidth": increase transmission rate on receipt of ACK, until eventually loss occurs, then decrease transmission rate
  - continue to increase on ACK, decrease on loss (since available bandwidth is changing, depending on other connections in network)



- Q: how fast to increase/decrease?
  - details to follow

TCP Congestion Control: details

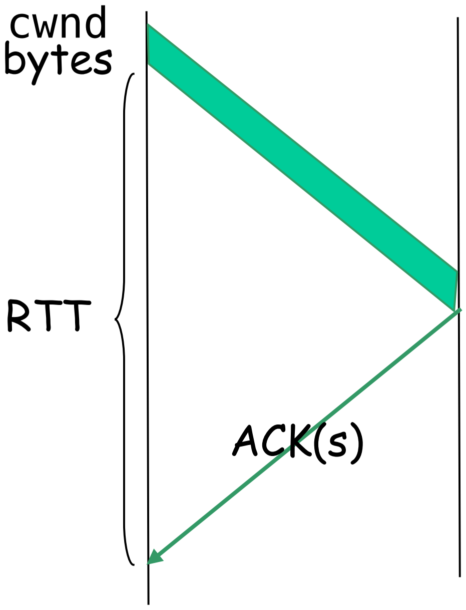
- sender limits rate by limiting number of unACKed bytes "in pipeline":

  LastByteSent-LastByteAcked ≤ cwnd

  - **cwnd:** differs from **rwnd** (how, why?)
  - sender limited by **min(cwnd,rwnd)**
- roughly,

$$\boxed{\text{rate} = \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}}$$

- **cwnd** is dynamic, function of perceived network congestion



TCP Congestion Control: more details

segment loss event: reducing `cwnd`

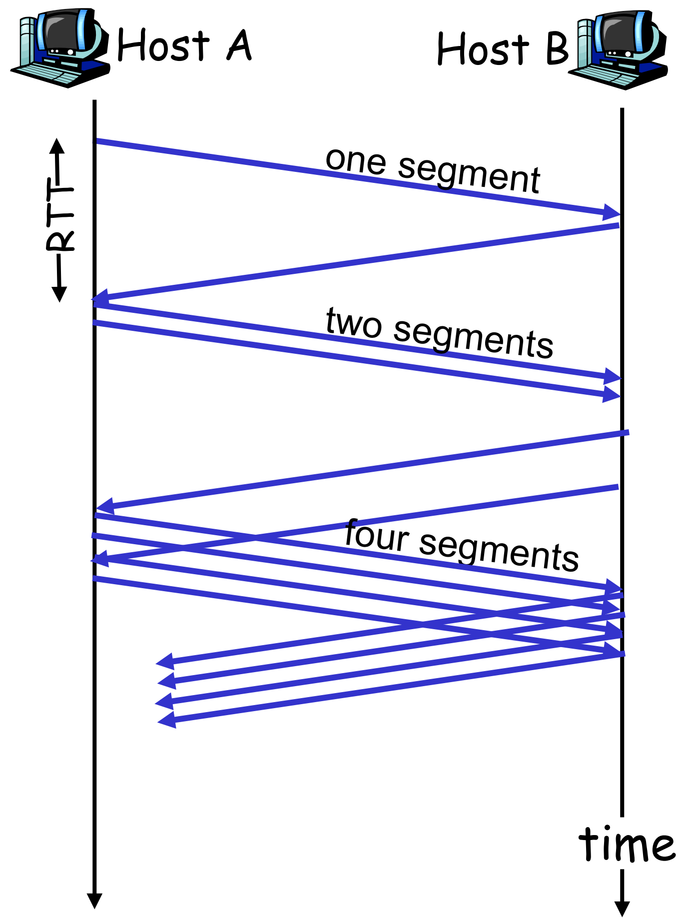
- timeout: no response from receiver
  - cut `cwnd` to 1
- 3 duplicate ACKs: at least some segments getting through (recall fast retransmit)
  - cut **cwnd** in half, less aggressively than on timeout

ACK received: increase `cwnd`

- slowstart phase:
  - increase exponentially fast (despite name) at connection start, or following timeout
- congestion avoidance:
  - increase linearly

TCP Slow Start

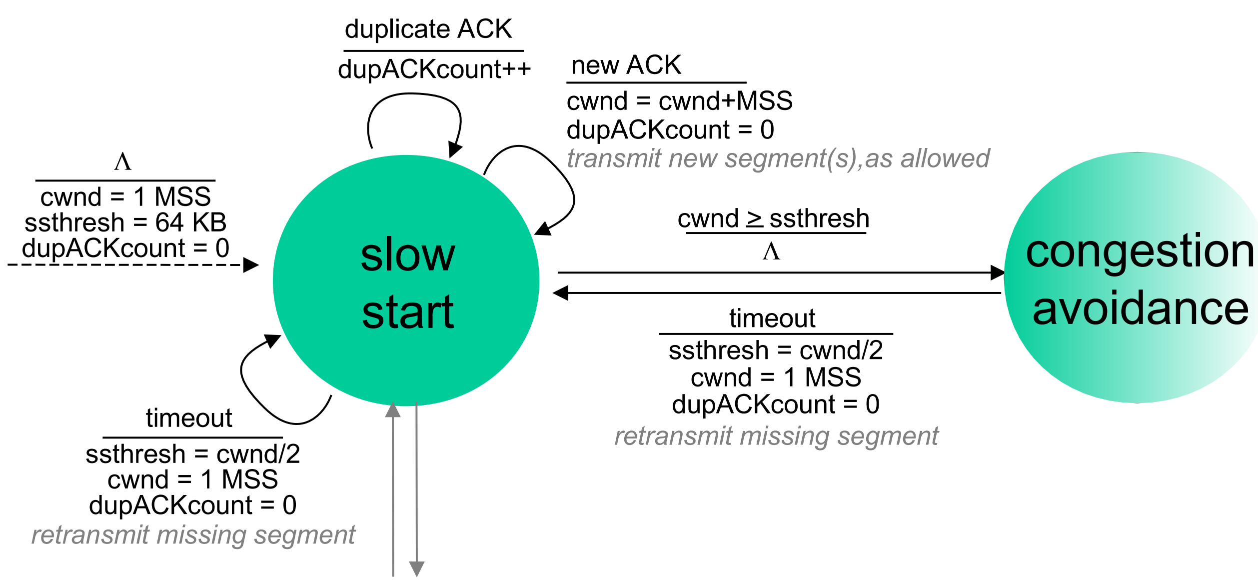
- when connection begins, **cwnd** = 1 MSS
  - example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- increase rate exponentially until first loss event or when threshold reached
  - double **cwnd** every RTT
  - done by incrementing **cwnd** by 1 for every ACK received



Transitioning into/out of slowstart

ssthresh: cwnd threshold maintained by TCP

- on loss event: set **ssthresh** to **cwnd/2**
  - remember (half of) TCP rate when congestion last occurred
- when **cwnd >= ssthresh**: transition from slowstart to congestion avoidance phase



TCP: congestion avoidance

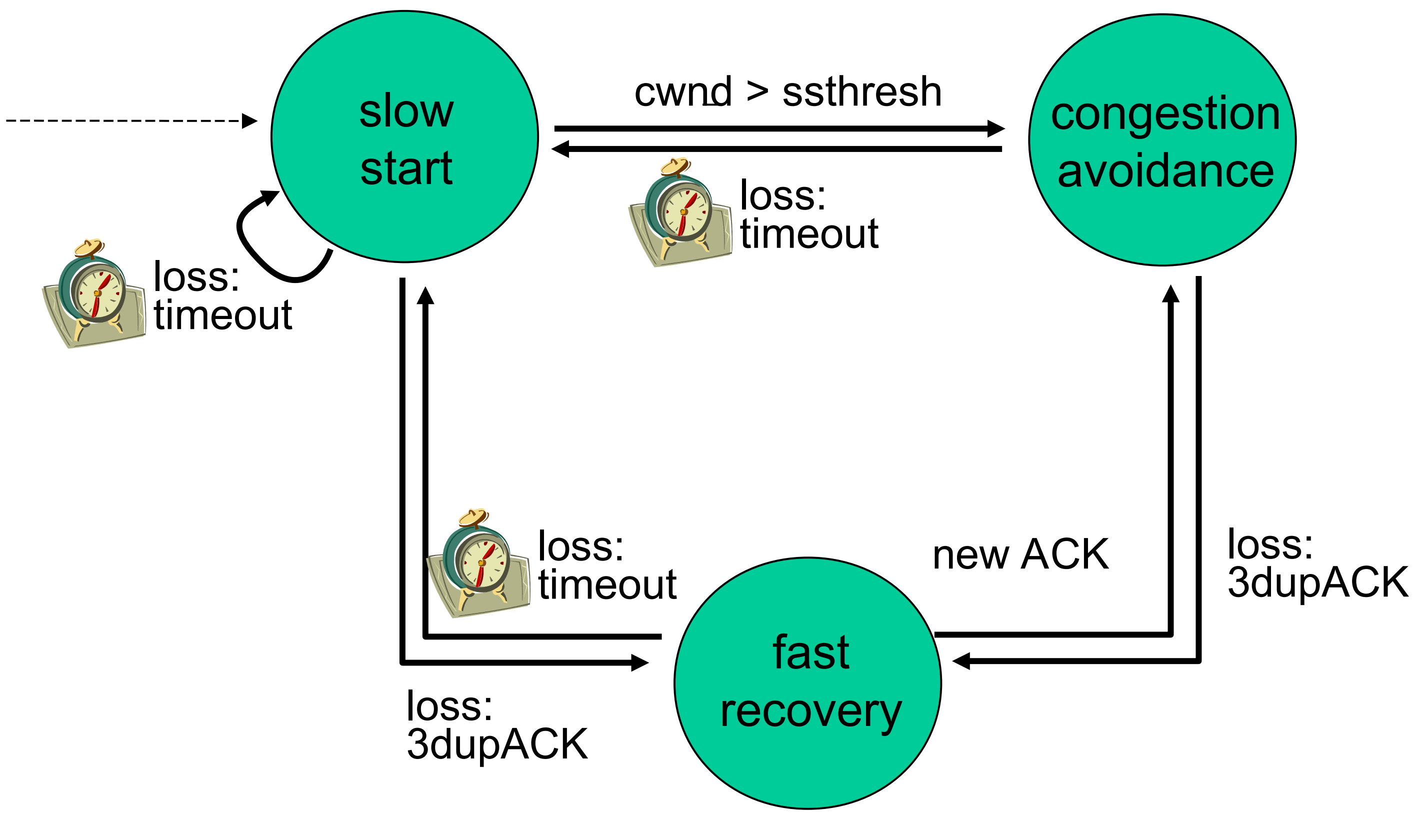
- when `cwnd > ssthresh` grow `cwnd` linearly
  - increase `cwnd` by 1 MSS per RTT
  - approach possible congestion slower than in slowstart
  - implementation: `cwnd = cwnd + MSS/cwnd` for each ACK received

AIMD

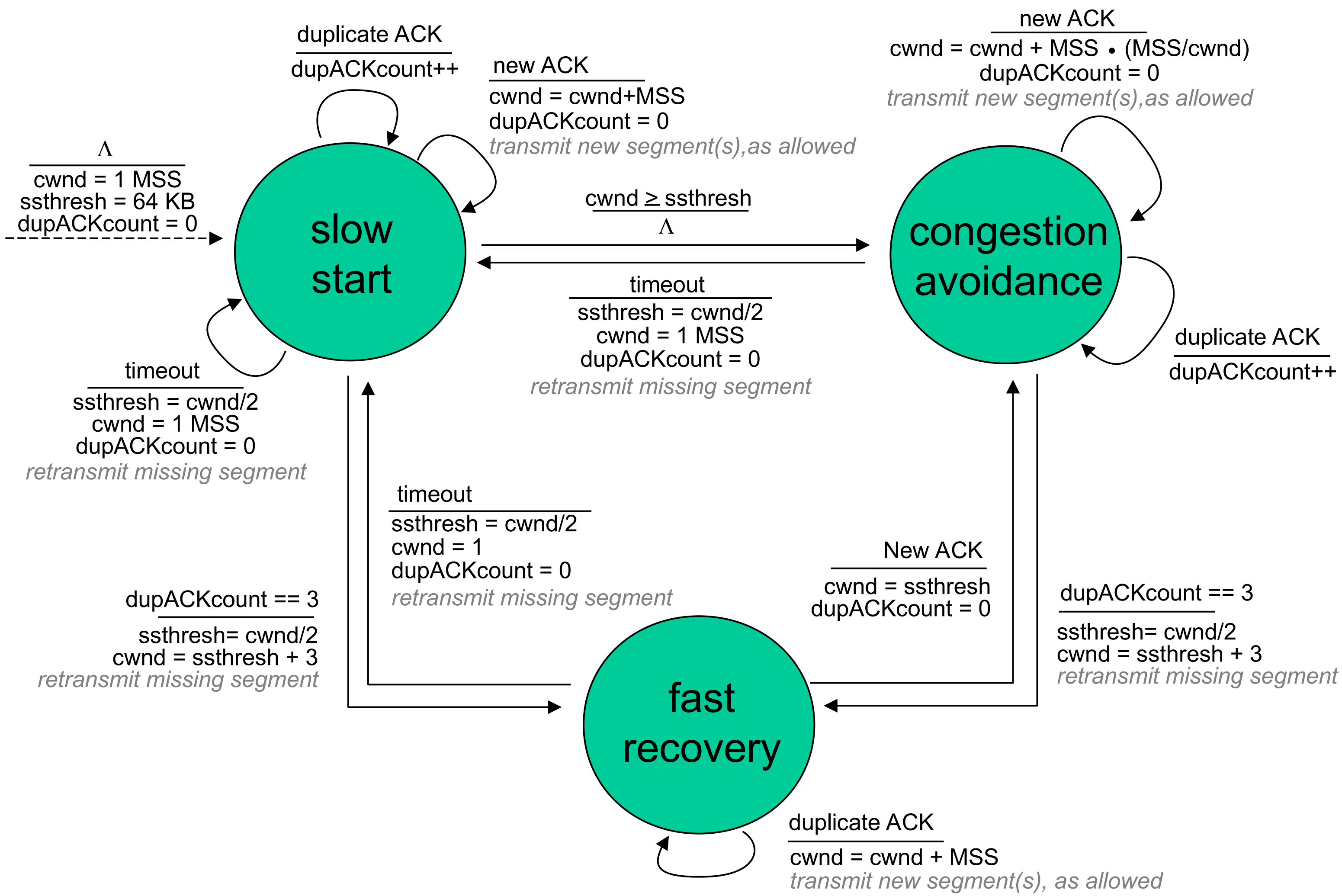
- ACKs: increase `cwnd` by 1 MSS per RTT: additive increase
- loss: cut `cwnd` in half (non-timeout-detected loss): multiplicative decrease

AIMD: Additive Increase Multiplicative Decrease

TCP congestion control FSM: overview

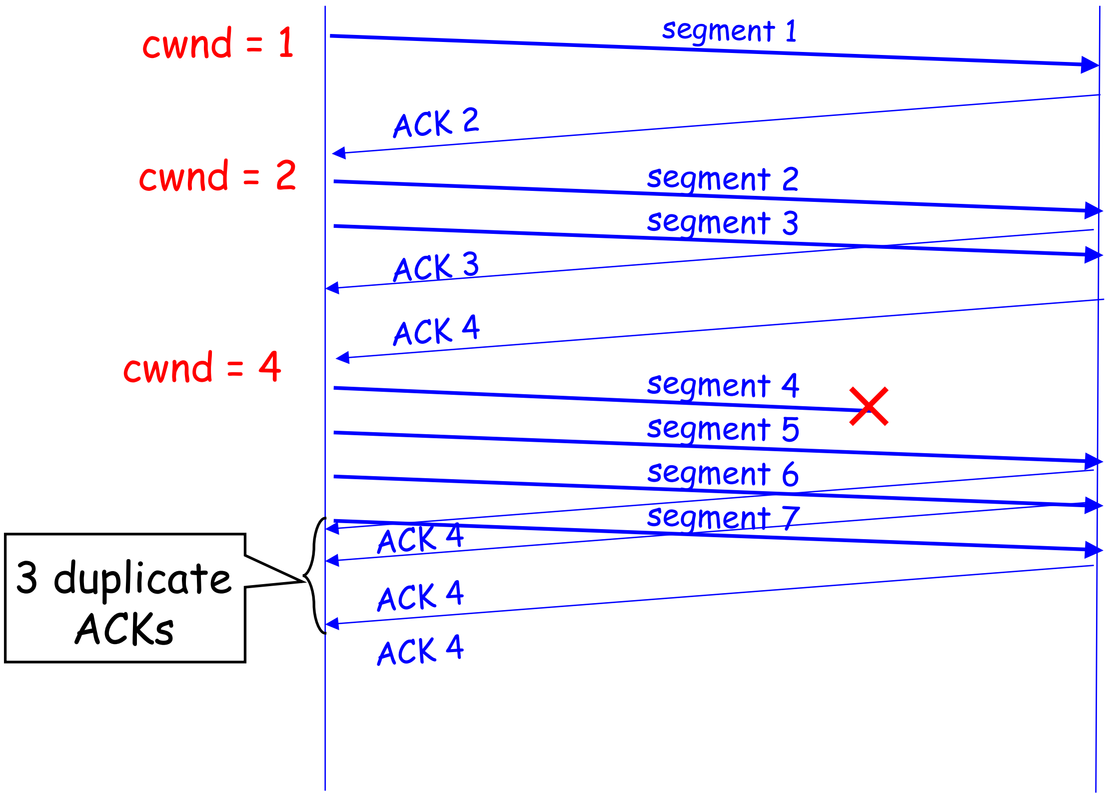


TCP congestion control FSM: details



Fast Retransmits

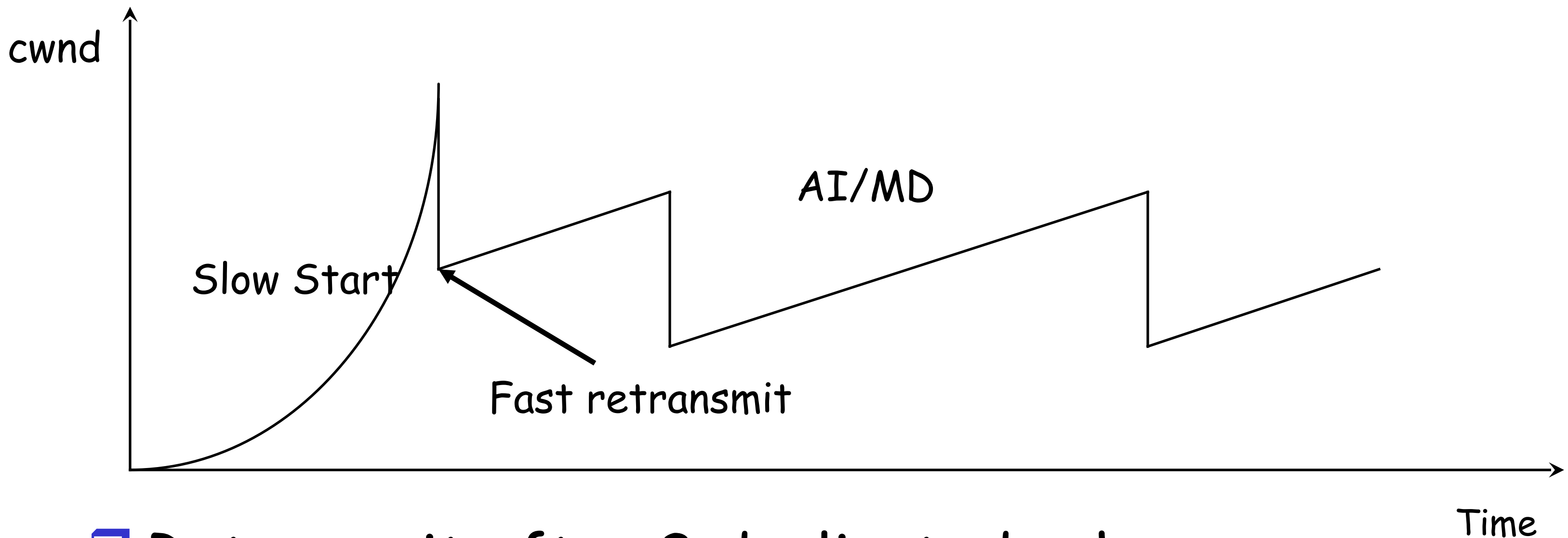
- Resend a segment after 3 duplicate ACKs
  - Duplicate ACK means that an out-of sequence segment was received
- Notes:
  - ACKs are for next expected packet
  - Packet reordering can cause duplicate ACKs
  - Window may be too small to get enough duplicate ACKs



Fast Recovery: After a Fast Retransmit

- *ssthresh* = *cwnd* / 2
- *cwnd* = *ssthresh*
  - Instead of setting *cwnd* to 1, cut cwnd in half (multiplicative decrease)
- For each dup ack arrival
  - *dupack*++
  - Indicates packet left network, so we may be able to send more
  - MaxWindow = min(*cwnd* + *dupack*, AdvWin)
- Receive ack for new data (beyond initial dup ack)
  - *dupack* = 0
  - Exit fast recovery
- But when RTO expires still do *cwnd* = 1

Fast Retransmit and Fast Recovery



- Retransmit after 3 duplicated acks
  - Prevent expensive timeouts
- Reduce slow starts
- At steady state, *cwnd* oscillates around the optimal window size

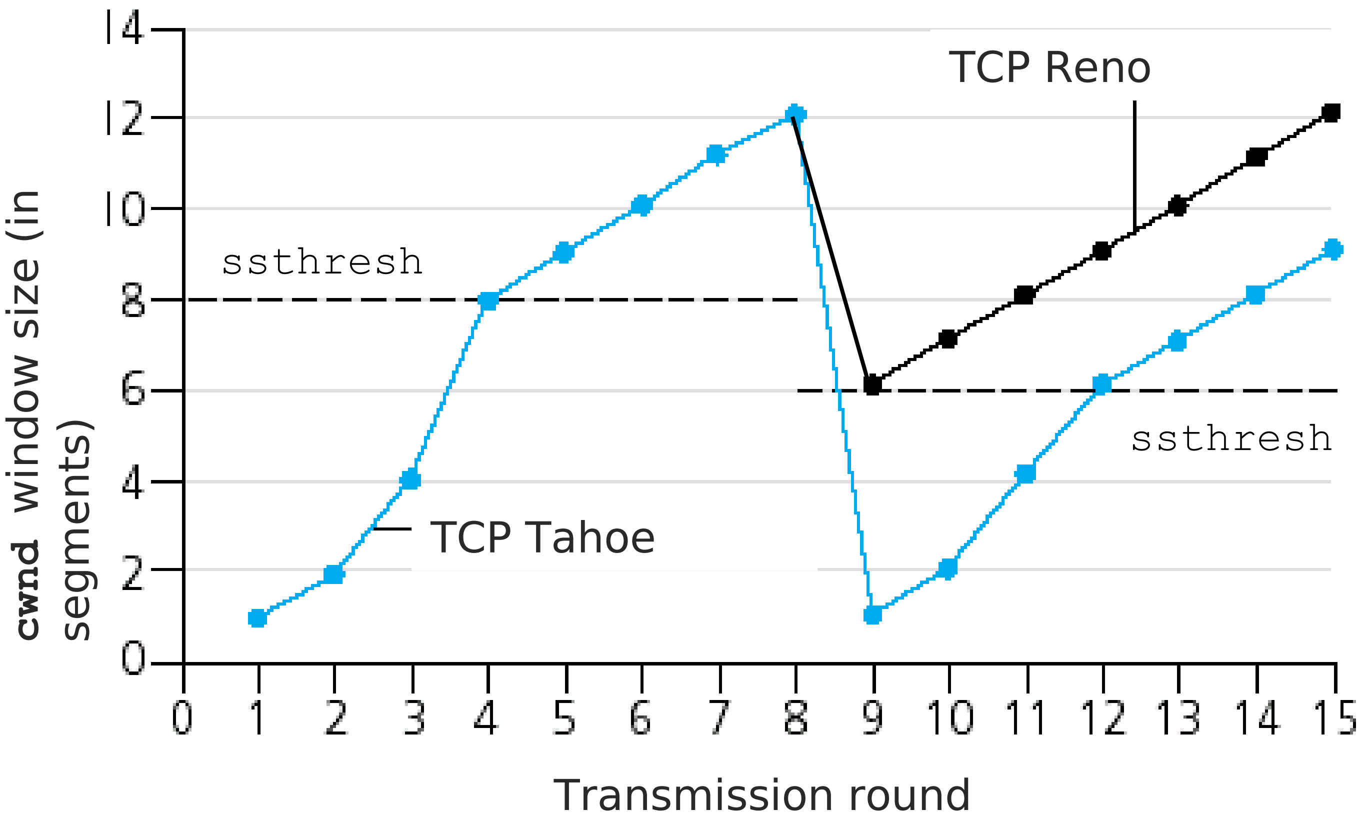
TCP Congestion Control Summary

- Measure available bandwidth
  - Slow start: fast, hard on network
  - AIMD: slow, gentle on network
- Detecting congestion
  - Timeout based on RTT
    - Robust, causes low throughput
  - Fast Retransmit: avoids timeouts when few packets lost
    - Can be fooled, maintains high throughput
- Recovering from loss
  - Fast recovery: don't set cwnd=1 with fast retransmits

TCP Flavors

- TCP-Tahoe
  - cwnd =1 whenever drop is detected
- TCP-Reno
  - cwnd =1 on timeout
  - cwnd = cwnd/2 on dupack
- TCP-newReno
  - TCP-Reno + improved fast recovery
- TCP-SACK

Popular "flavors" of TCP

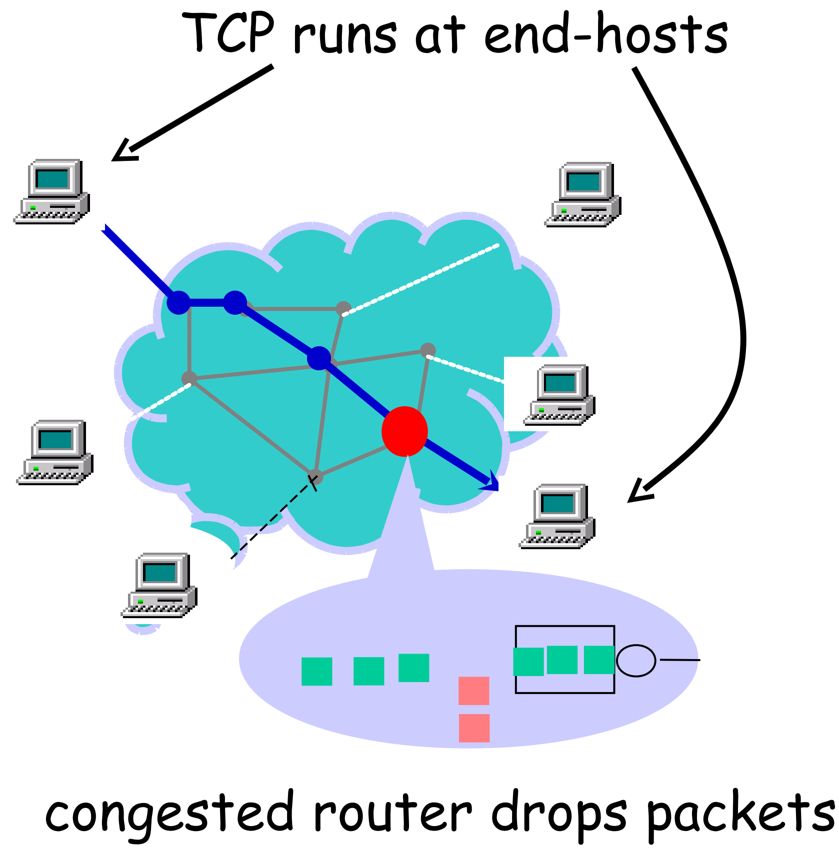


TCP-SACK

- SACK = Selective Acknowledgements
- ACK packets identify exactly which packets have arrived
- Makes recovery from multiple losses much easier

TCP behavior

- congestion control:
  - decrease sending rate when loss detected, increase when no loss
- routers
  - discard, mark packets when congestion occurs
- interaction between end systems (TCP) and routers?
  - want to understand (quantify) this interaction



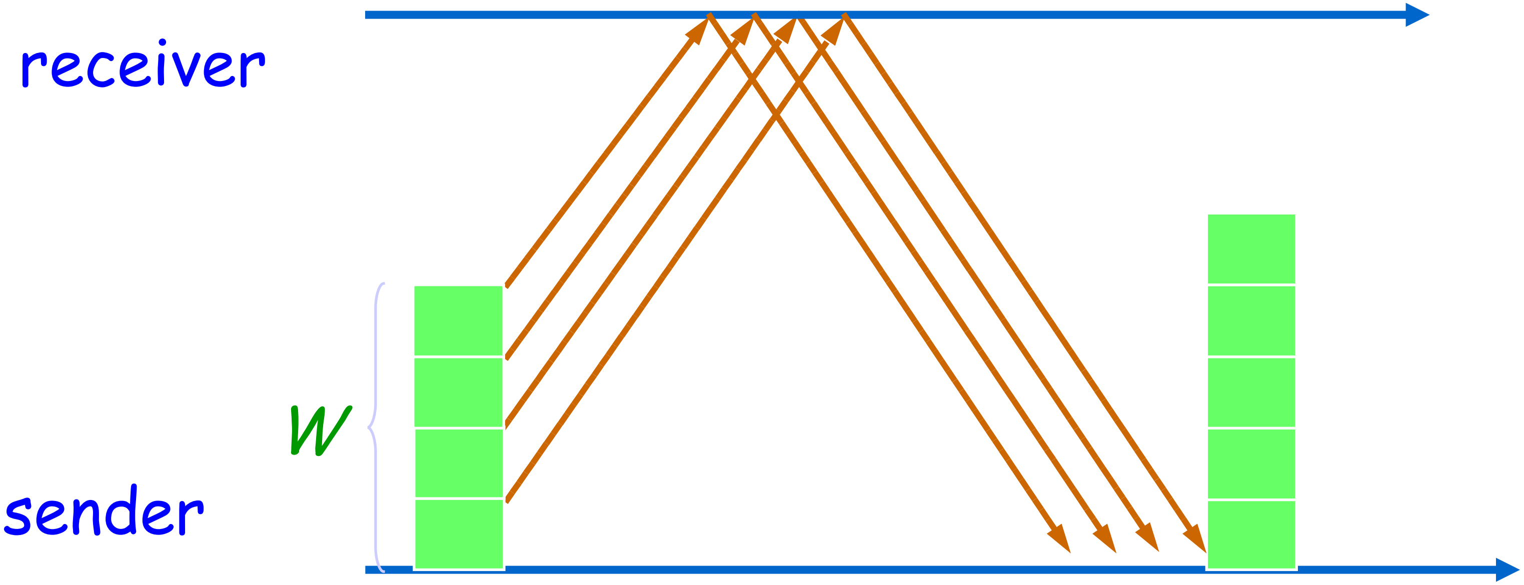
Generic TCP behavior

- window algorithm (window W)
  - up to W packets in network
  - return of ACK allows sender to send another packet
  - cumulative ACKS
- increase window by one per RTT

$$W \leftarrow W + 1/W \text{ per ACK}$$

$$\Rightarrow W \leftarrow W + 1 \text{ per RTT}$$

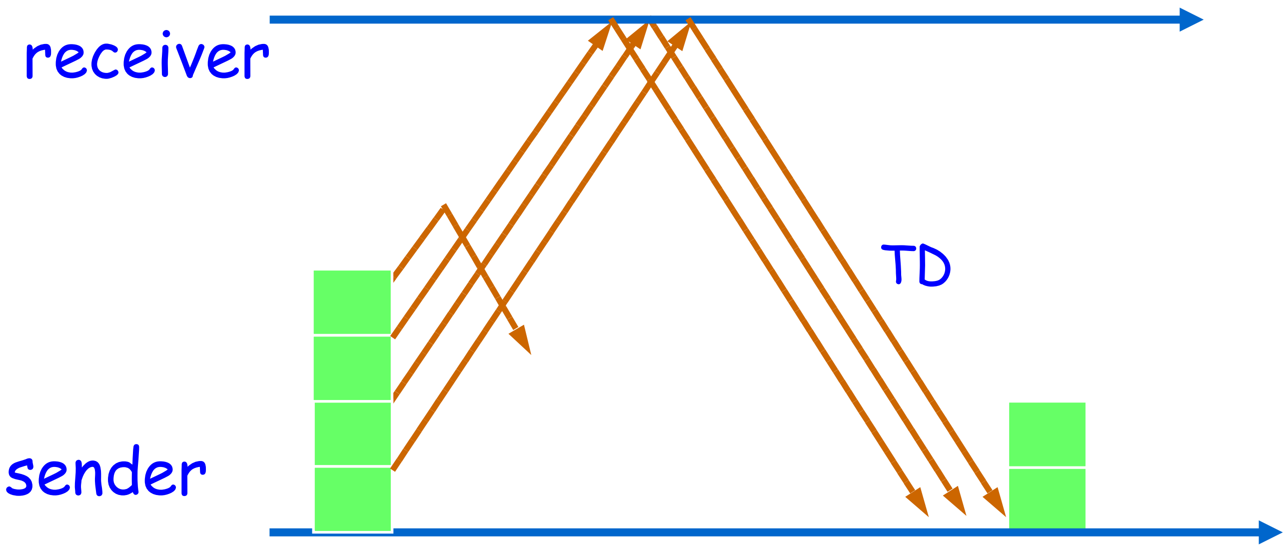
- seeks available network bandwidth


receiver
W
sender

Generic TCP behavior

- window algorithm (window W)
- increase window by one per RTT

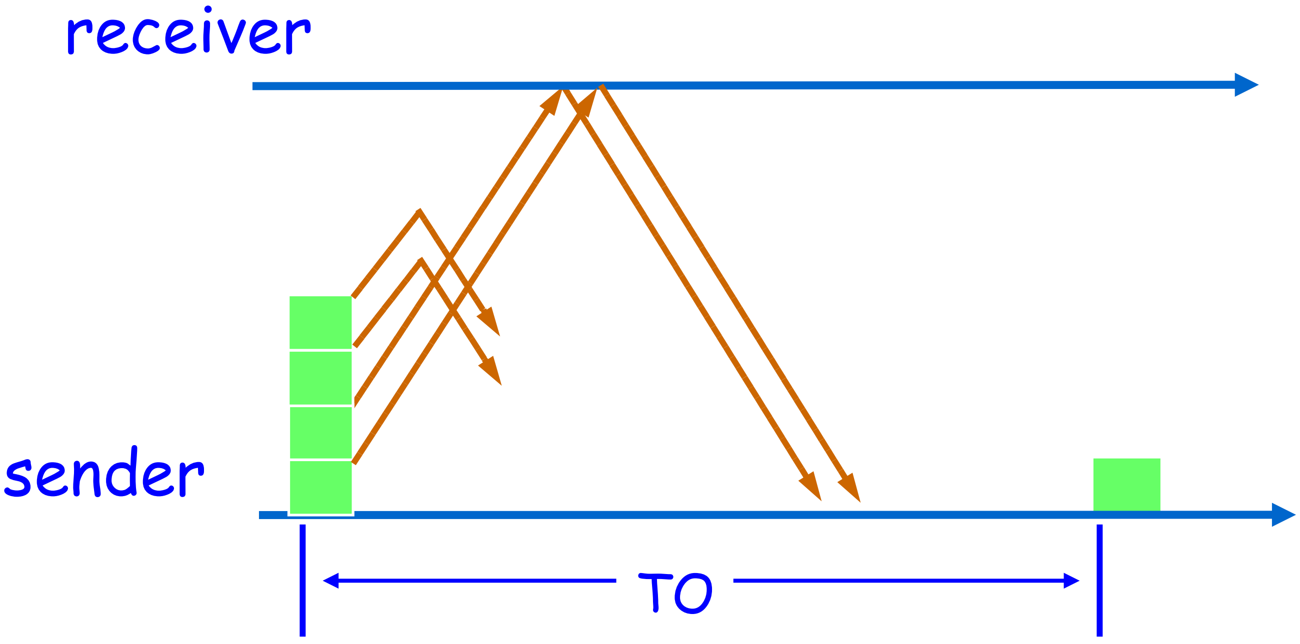
  W <- $W + 1/W$ *per ACK*
- loss indication of congestion
- decrease window by half on detection of loss, (triple duplicate ACKs), W <- $W/2$


receiver
TD
sender

Generic TCP Behavior

- window algorithm (window W)
- increase window by one per RTT

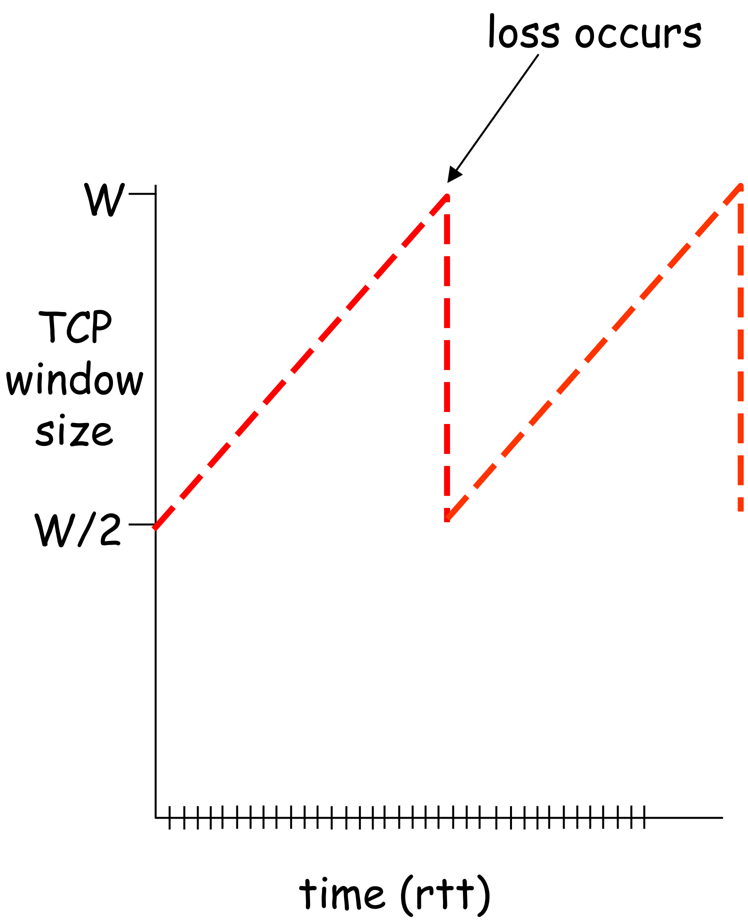
  W <- W +1/ W *per* ACK
- halve window on detection of loss, W <- W/2
- timeouts due to lack of ACKs -> window reduced to one, W <- 1


receiver
sender
TO

Generic TCP Behavior

- window algorithm (window W)
- increase window by one per RTT (or one over window per ACK, W <- W+1/W)
- halve window on detection of loss, W <- W/2
- timeouts due to lack of ACKs, W <- 1
- successive timeout intervals grow exponentially long up to six times

TCP throughput/loss relationship

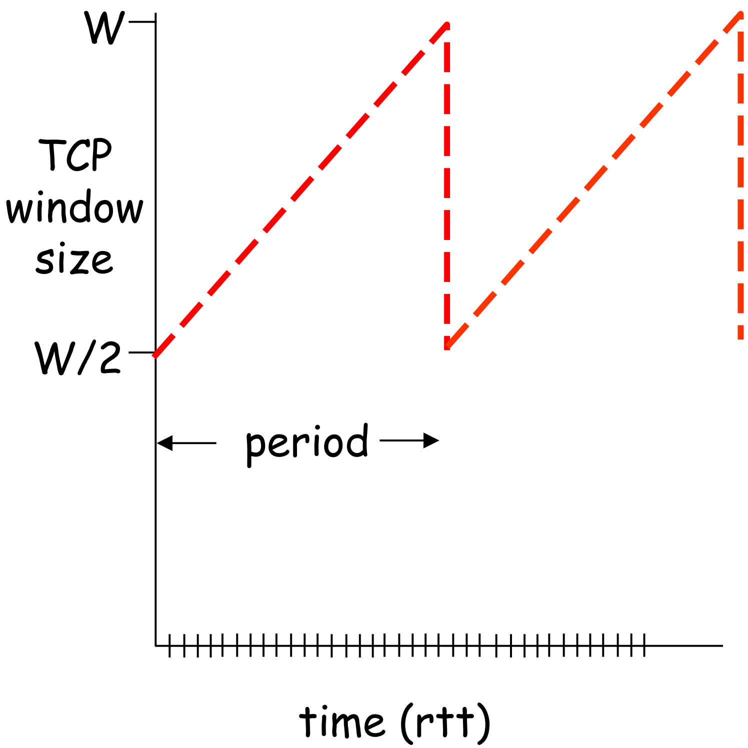


Idealized model:

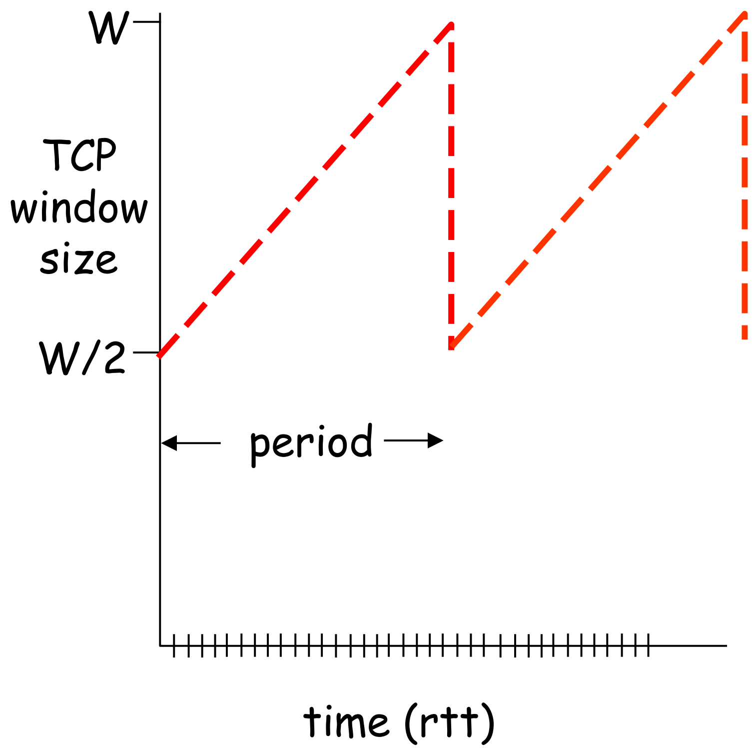
- W is maximum supportable window size (then loss occurs)
- TCP window starts at W/2 grows to W, then halves, then grows to W, then halves…
- one window worth of packets each RTT
- *to find:* throughput as function of loss, RTT

TCP throughput/loss relationship

packets sent per “period” =



TCP throughput/loss relationship



\# packets sent per "period" =

$$\frac{W}{2}+\left(\frac{W}{2}+1\right)+\ldots+W=\sum_{n=0}^{W/2}\left(\frac{W}{2}+n\right)$$

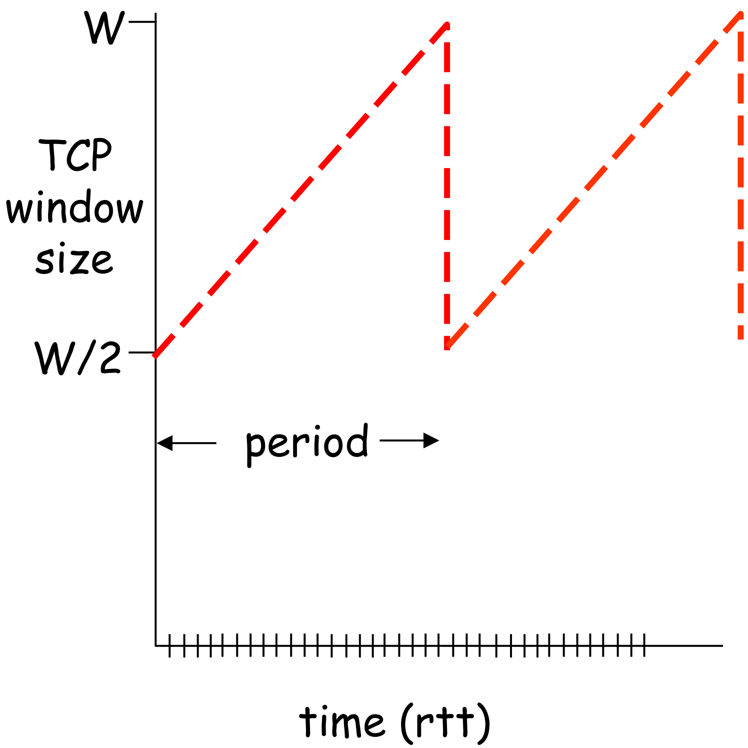
$$=\left(\frac{W}{2}+1\right)\frac{W}{2}+\sum_{n=0}^{W/2}n$$

$$=\left(\frac{W}{2}+1\right)\frac{W}{2}+\frac{W/2(W/2+1)}{2}$$

$$=\frac{3}{8}W^2+\frac{3}{4}W$$

$$\approx\frac{3}{8}W^2$$

TCP throughput/loss relationship



packets sent per "period" $\approx \frac{3}{8}W^2$

1 packet lost per "period" implies:

$p_{loss} \approx \frac{8}{3W^2}$ or: $W = \sqrt{\frac{8}{3p_{loss}}}$

$$B = \text{avg.\_thruput} = \frac{3}{4}W \frac{\text{packets}}{\text{rtt}}$$

$$\boxed{B = \text{avg.\_thruput} = \frac{1.22}{\sqrt{p_{loss}}} \frac{\text{packets}}{\text{rtt}}}$$

B throughput formula can be extended to model timeouts and slow start (PFTK)

TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires window size W = 83,333 in-flight segments
- throughput in terms of loss rate:

$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

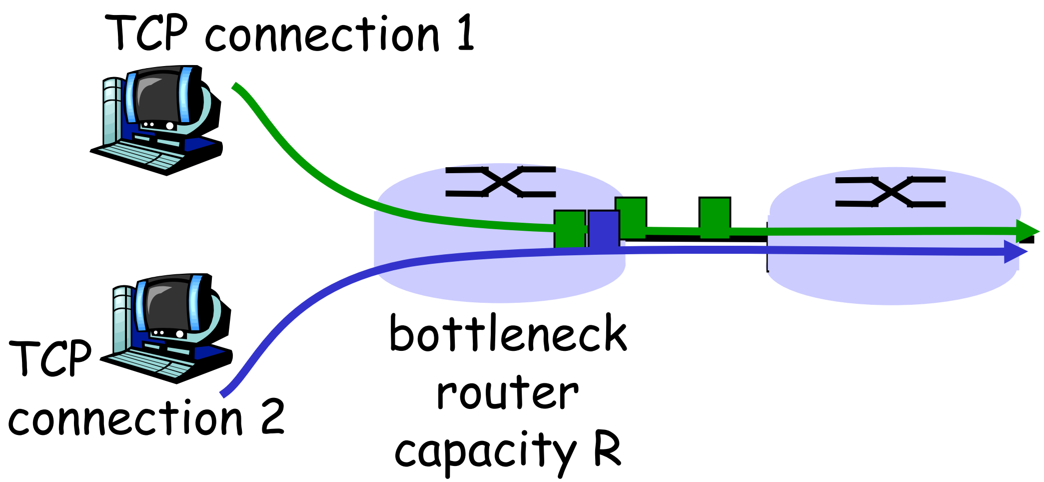
- → $L = 2 \cdot 10^{-10}$ *Wow*
- new versions of TCP for high-speed

Summary: TCP Congestion Control

- when **cwnd < ssthresh**, sender in slow-start phase, window grows exponentially.
- when **cwnd >= ssthresh**, sender is in congestion-avoidance phase, window grows linearly.
- when triple duplicate ACK occurs, **ssthresh** set to **cwnd/2**, **cwnd** set to ~ **ssthresh**
- when timeout occurs, **ssthresh** set to **cwnd/2**, **cwnd** set to 1 MSS.

TCP Fairness

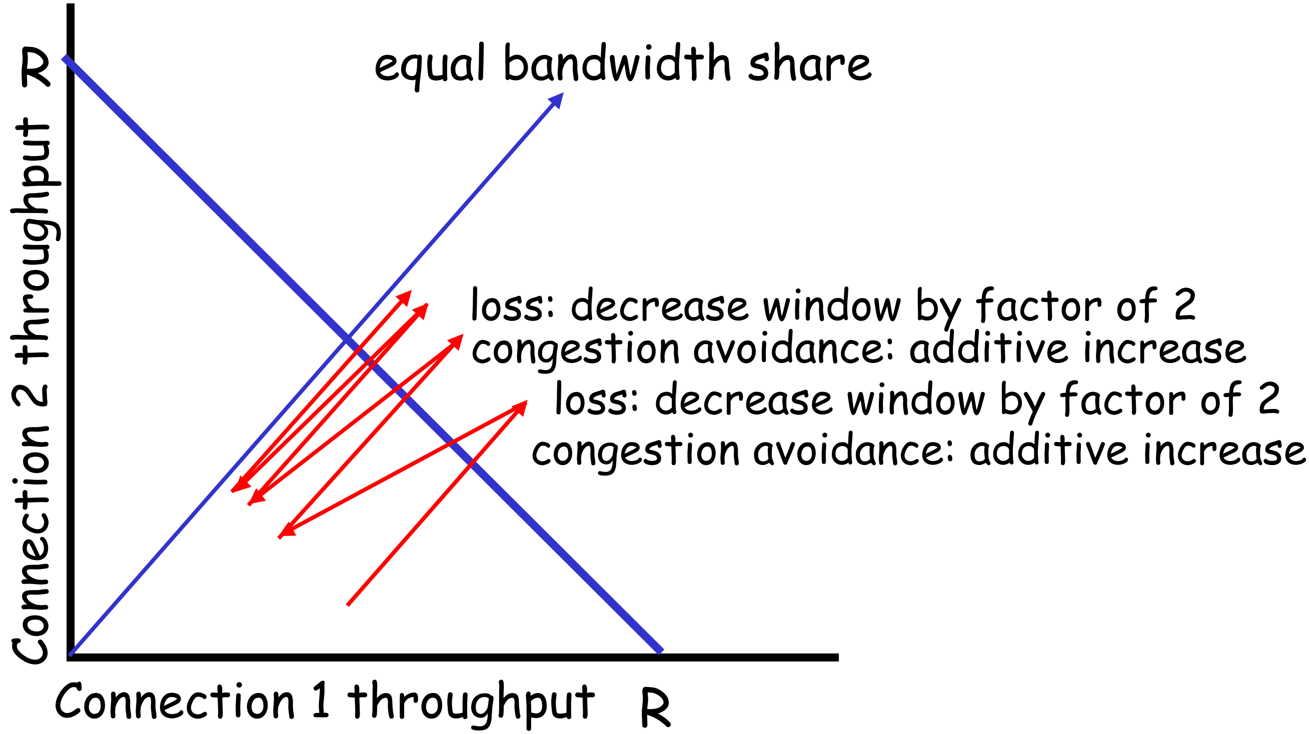
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



Fairness (more)

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- web browsers do this
- example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !

Recap : TCP/ Reno

- **Two cases**
  - 3 duplicate ACKs (network capable of delivering some packets)
  - Timeout (more alarming)

- **Two phases**

  1. Slow start (SS) - MI

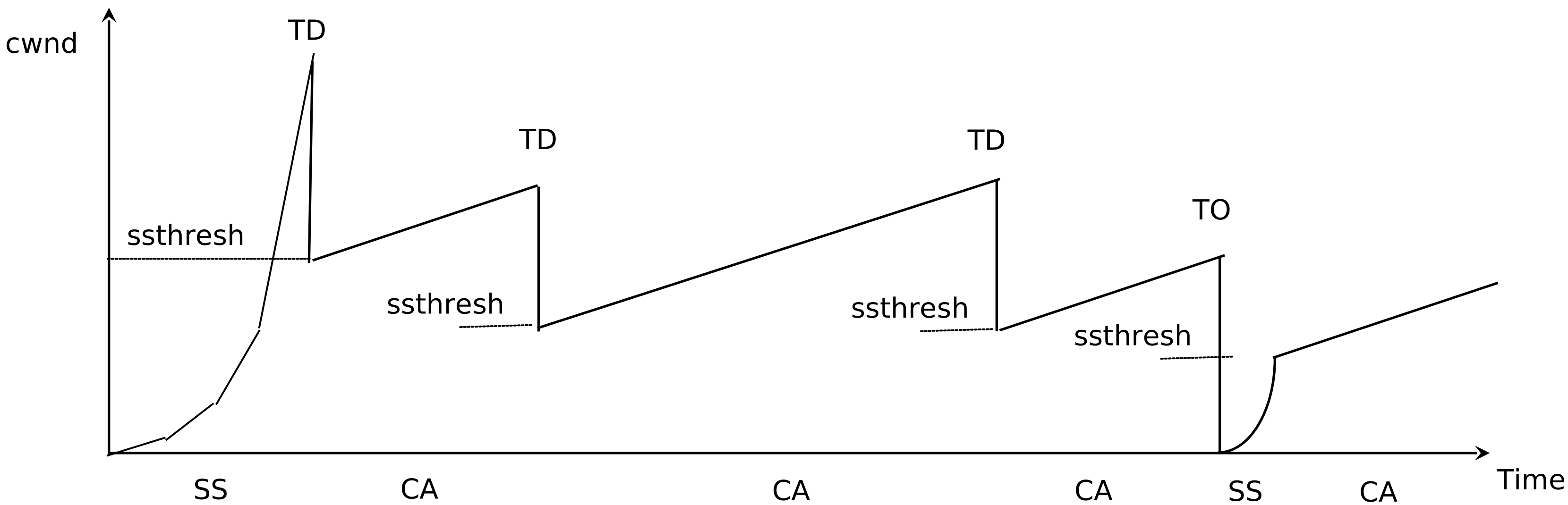
     2*cwnd per RTT till congestion

  2. Congestion avoidance(CA) – AIMD

     cwnd increase by 1 per RTT

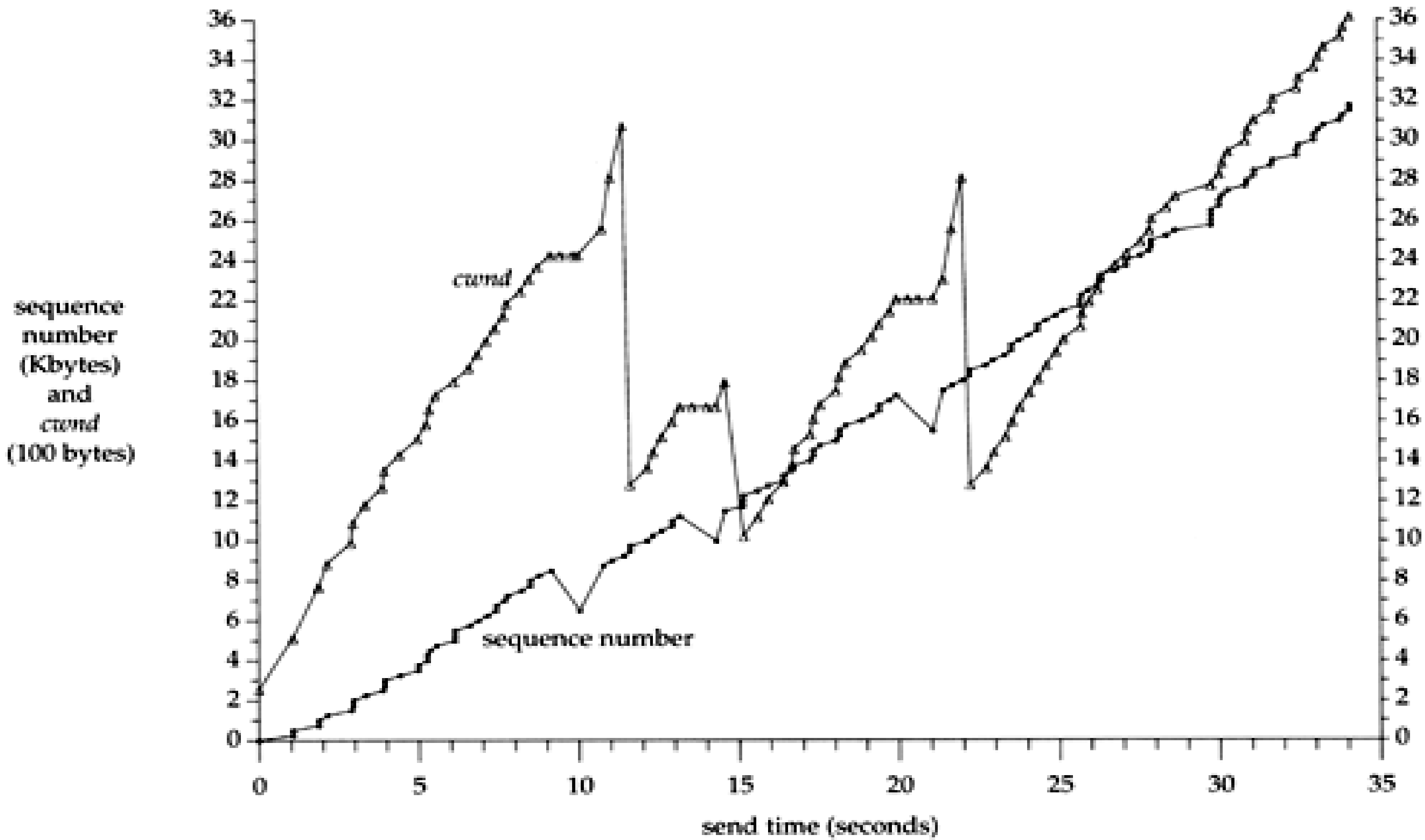
     - **3 duplicate ACKs ➔ cwnd = cwnd/2**
     - Timeout ➔ cwnd =1
     - In timeout ➔ timeout = 2*timeout

Recap : TCP/ Reno



SS – Slow Start
CA – Congestion Avoidance
TD – Three Duplicate ACKs
TO - Timeout

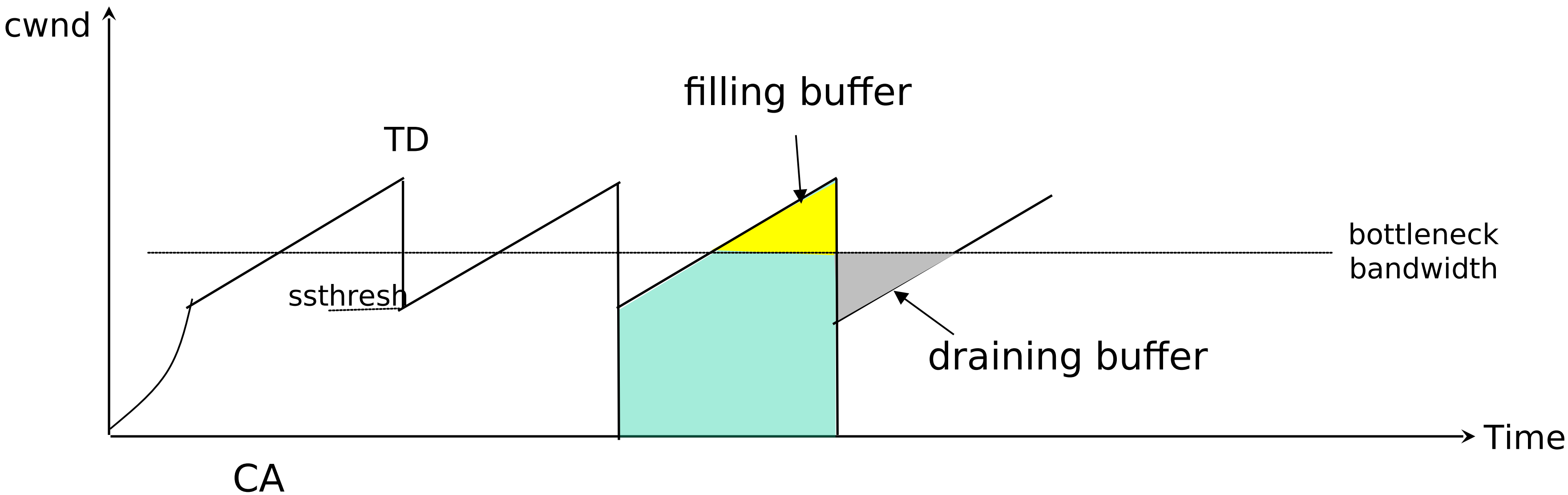
Recap : TCP/ Reno

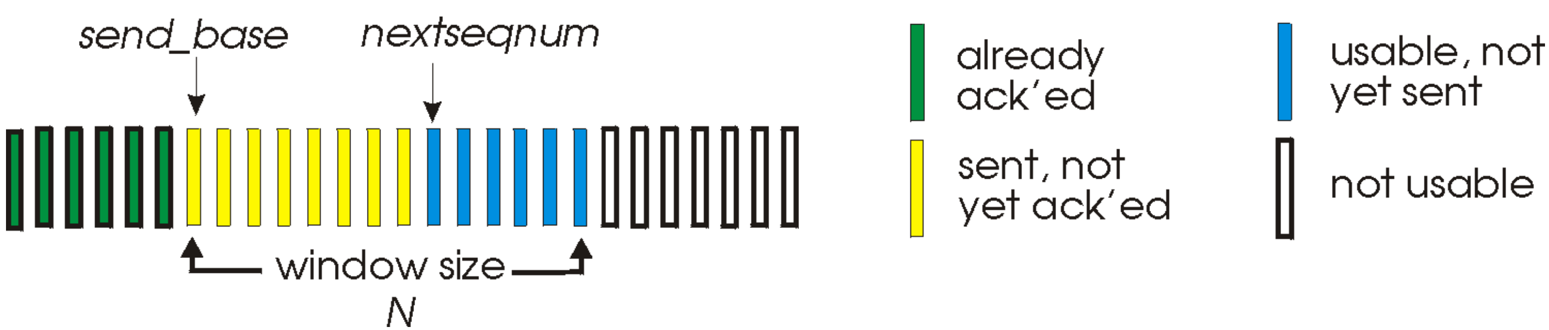


When cwnd is cut to half, why does sending rate not get cut?

Recap : TCP/ Reno

There is a filling and draining of buffers for each TCP flow.





TCP/ Reno Analysis

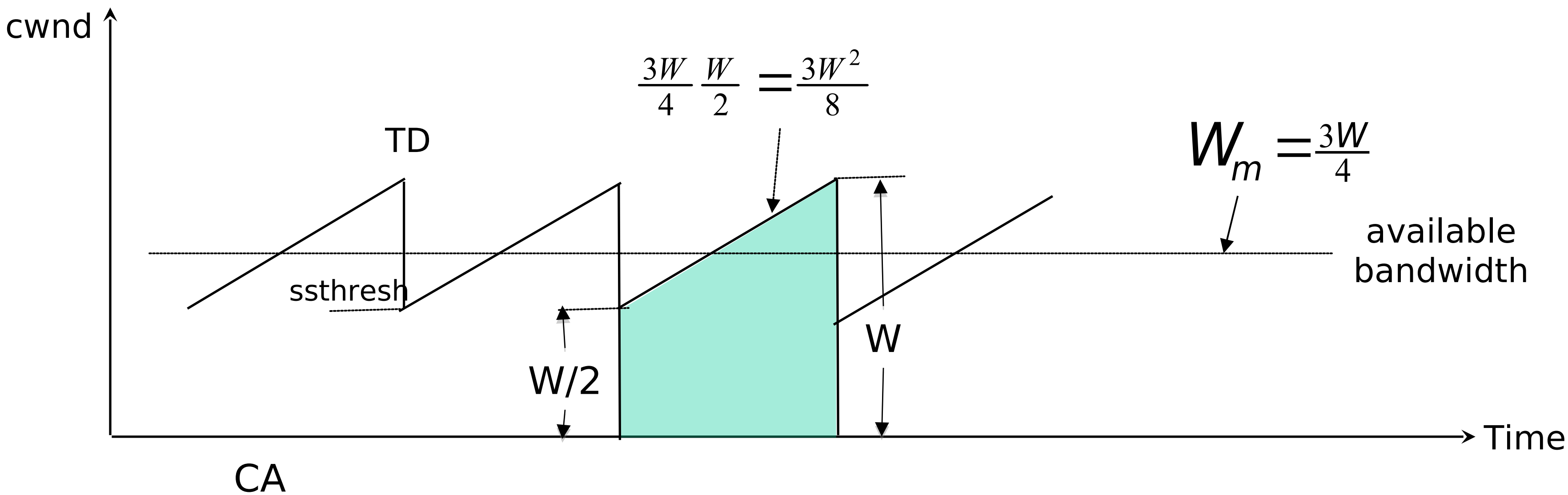
TCP/ Reno Throughput Analysis

- **Understand throughput in terms of**
  - RTT
  - Packet loss rate (p)
  - Packet size (S)

- **Throughput calculations**
  - Assume congestion avoidance and no timeouts occur
  - Mean window size W_m segments, round trip time RTT & pack size S
  - Throughput ≈ $\frac{W_m * S}{RTT}$ bytes/sec

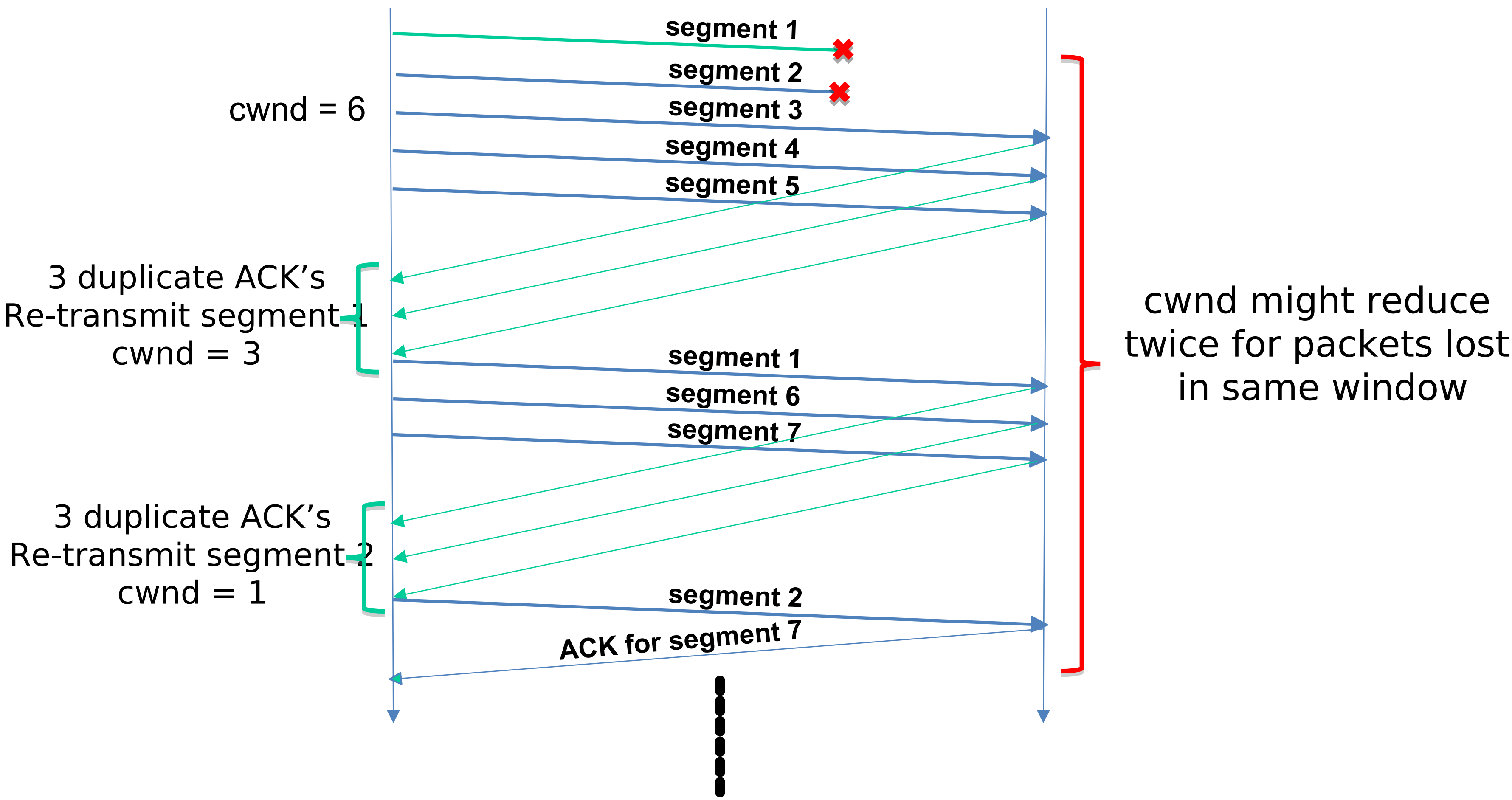
Deterministic Analysis

- Consider congestion avoidance
  - Assume one packet is lost per cycle
  - Total packets sent per cycle $= \frac{1}{2}*(W + W/2) * W/2 = 3W^2/8$
  - Packet loss (p) $= 1/(3W^2/8) = 8/(3W^2) \rightarrow W = \frac{\sqrt{8/3}}{\sqrt{p}} = \frac{1.6}{\sqrt{p}}$
- Throughput $= \frac{S*W_m}{RTT} = \frac{S}{RTT}\left(\frac{3}{4}\frac{1.6}{\sqrt{p}}\right) = \frac{1.2S}{RTT\sqrt{p}}$



TCP/ Reno Drawbacks

- Multiple packets lost simultaneously cannot be accounted f(



TCP/ Reno Drawbacks

- RTT unfairness
  - Flows with different RTT's grow their congestion windows differently
  - Users with shorter RTT ramp up faster!
  - On long distance links, RTT is high and cwnd takes longer to increase leading to underutilization of link.

- Synchronized losses
  - Simultaneous packet loss events for multiple competing flows.

New Protocol Necessary!!

Desired Characteristics in TCP

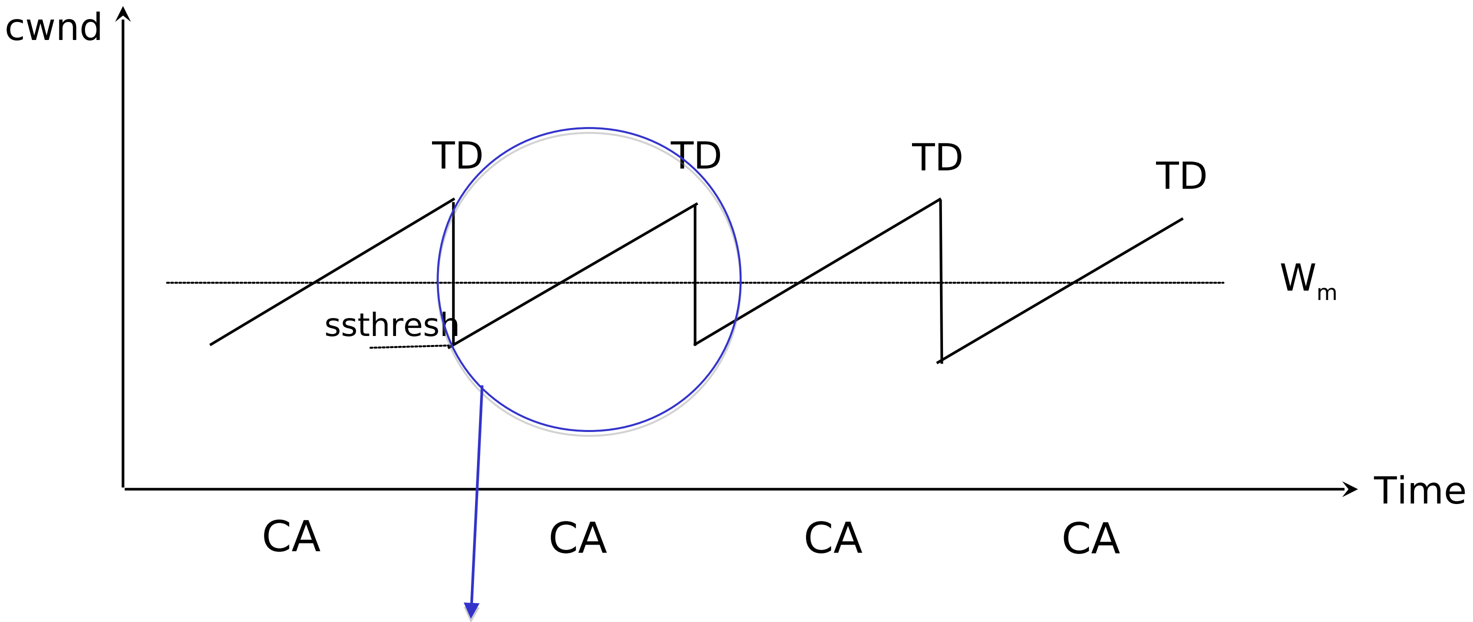
- Adaptive schemes that grow the congestion window dep
network conditions
  - Scalable
  - RTT Fairness
  - Faster convergence to better utilize full bandwidth

TCP BIC

http://www.land.ufrj.br/~classes/coppe-redes-2007/projeto/BIC-TCP-infocom-04.pdf

Growth functions

- Consider TCP/Reno growth function



Grows linearly throughout

TCP BIC

Binary Increase Congestion Control (BIC) algorithm

PHASE 1

- cwnd < low_wind, follows TCP
  - ACK received : cwnd = cwnd + 1
  - Loss event: cwnd = cwnd/2

PHASE 2

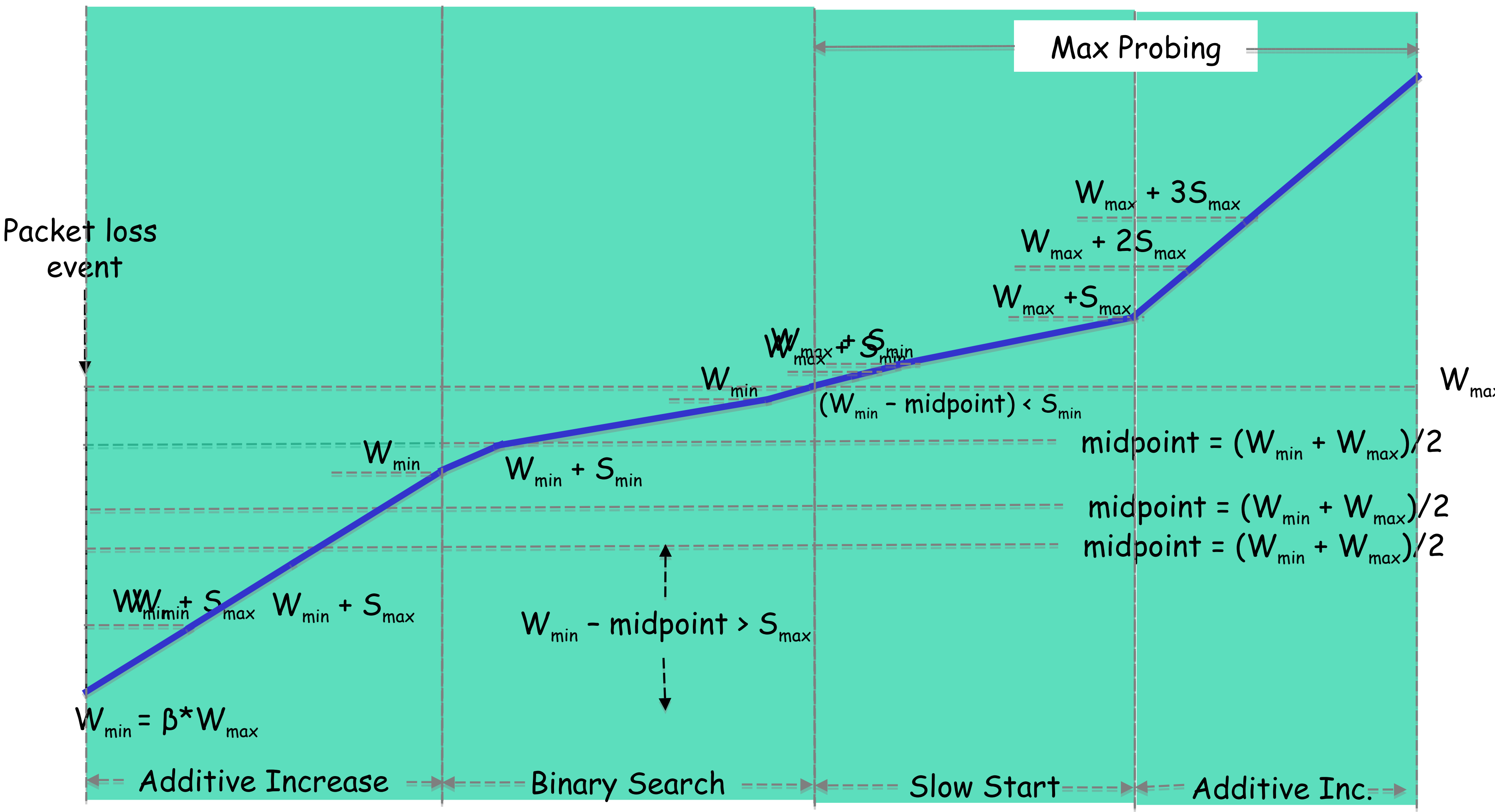
- cwnd > low_wind, follows BIC

BIC Algorithm

- Some preliminaries
  - βmultiplicative decrease factor
  - W_{max} = cwnd size before the reduction
  - $W_{min} = \beta * W_{max}$ – just after reduction
  - midpoint = (Wmax + Wmin)/2

BIC performs binary search between W_{max} and W_{min} looking for the midpoint.

BIC Algorithm



BIC Algorithm

```
while (cwnd != W_max){
If ((W_min - midpoint) > S_max)
      cwnd = cwnd + S_max          → Additive Increase
   else
      If ((W_min – midpoint) < S_min)
            cwnd = W_max
      else
            cwnd = midpoint        → Binary Search
   If (no packet loss)
      W_min = cwnd
   else
      W_min = β*cwnd
      W_max = cwnd
   midpoint = (W_max + W_min)/2
   }
```

BIC Algorithm

while ($cwnd >= W_{max}$){

If ($cwnd < W_{max} + S_{max}$)

$cwnd = cwnd + S_{min}$ → Slow Start

else

$cwnd = cwnd + S_{max}$ → Additive Increase

If (packet loss)

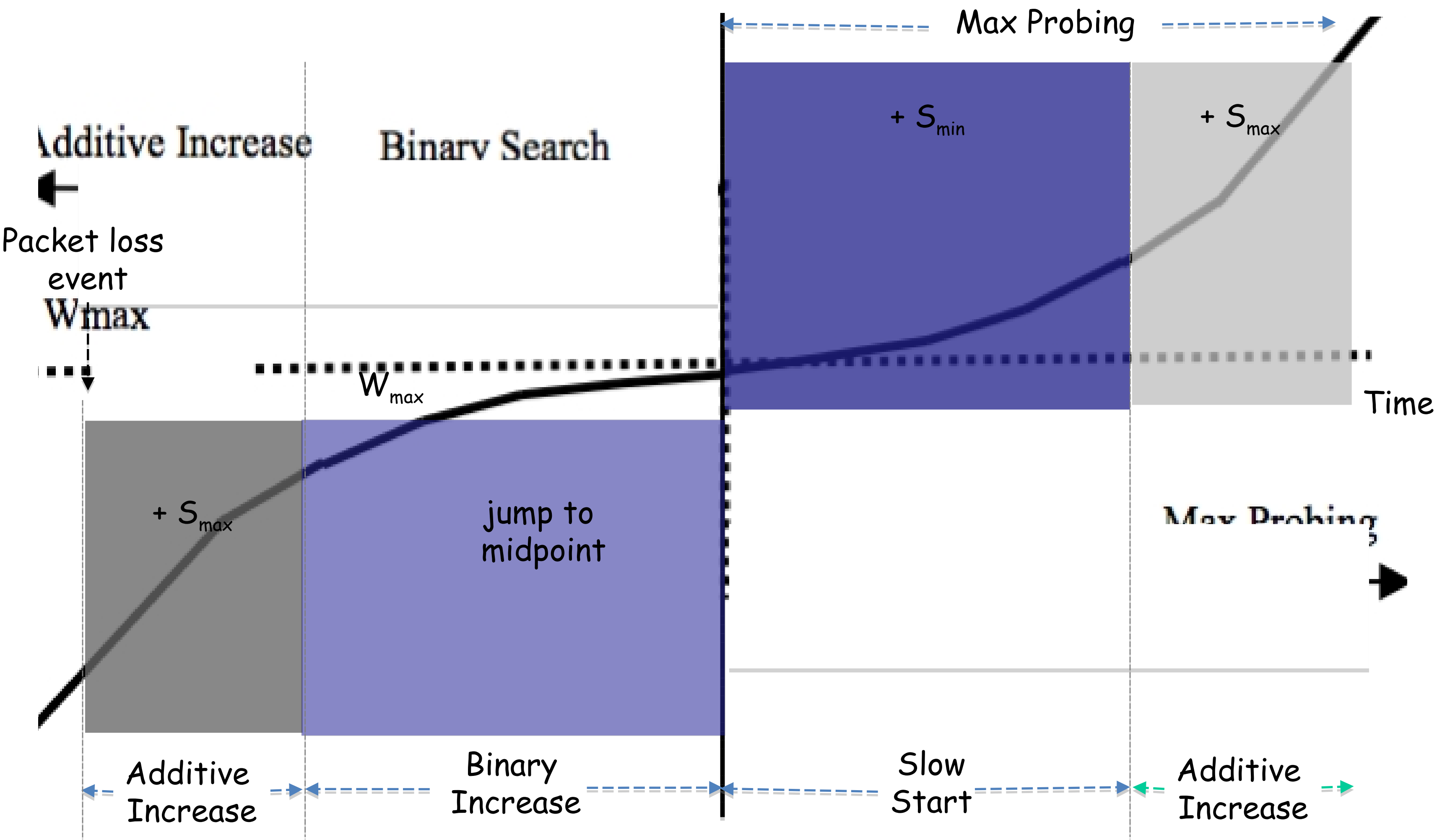
$W_{min} = \beta * cwnd$

$W_{max} = cwnd$

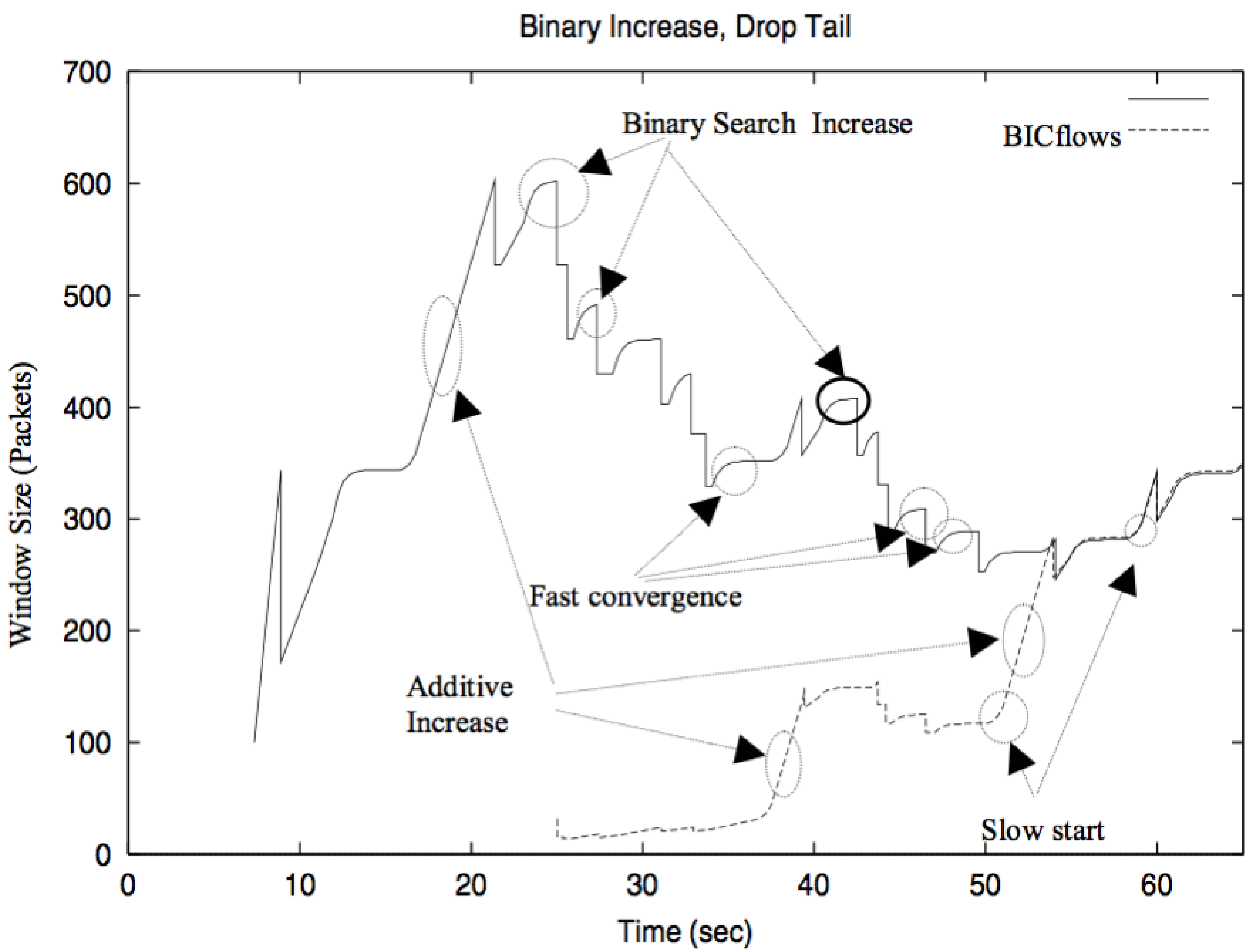
}

Max Probing

TCP BIC - Summary



TCP BIC in Action



TCP BIC Advantages

- Scalability: quickly scales to fair BW share
- Fairness and convergence: Achieves better fairness and faster convergence
- Slow Growth around W_{max} ensures that unnecessary timeouts do not occur.

TCP BIC Drawbacks

- cwnd growth is aggressive for TCP with short RTT or low speed
  - Short RTT makes cwnd ramp up soon
- Still dependent on RTT
  - Proportional to inverse square of the RTT like TCP/ Reno
- Complex window growth function
  - Difficult for analysis and actual implementation

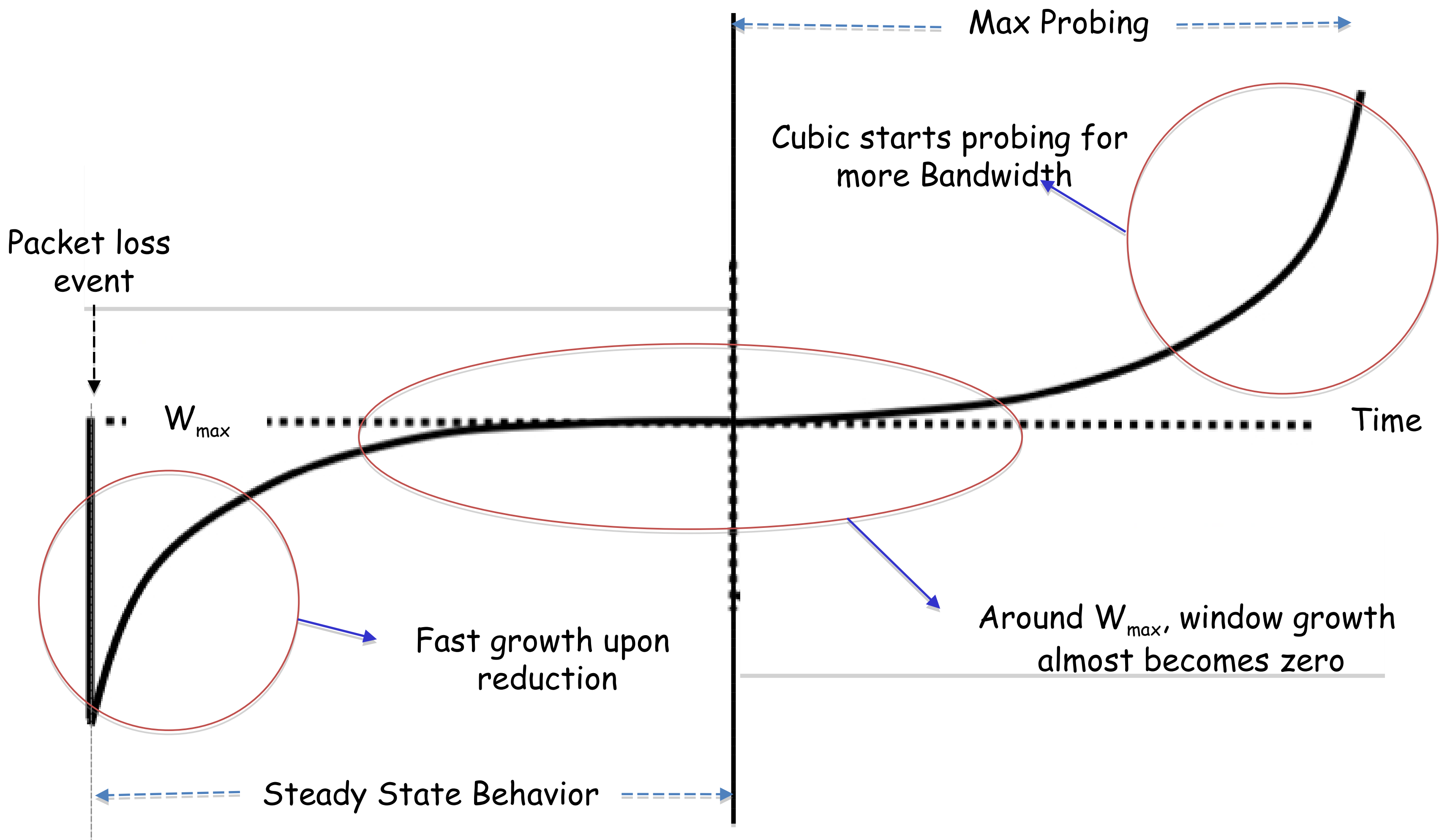
TCP Cubic

http://www4.ncsu.edu/~rhee/export/bitcp/cubic-paper.pdf

TCP Cubic

- cwnd = $C(t - K)^3 + W_{max}$
  - W_{max} = cwnd before last reduction
  - βmultiplicative decrease factor
  - C scaling factor
  - $$K = \sqrt[3]{W\beta / C}$$
  - t is the time elapsed since last window reduction

TCP CUBIC


Max Probing
Cubic starts probing for more Bandwidth
Packet loss event
W_{max}
Time
Fast growth upon reduction
Around W_{max}, window growth almost becomes zero
Steady State Behavior

TCP Cubic Advantages

- Good RTT fairness
  - Growth dominated by t, competing flows have same t after synchronized packet loss

- Real-time dependent
  - Similar to BIC but linear increases are time dependent
  - Does not depend on ACK's like TCP/ Reno

- Scalability
  - Cubic increases window to W_{max} (or its vicinity) quickly and keeps it there longer

TCP Cubic Drawbacks

- Slow Convergence
  - Flows with higher cwnd are more aggressive initially
  - Prolonged unfairness between flows

- Bandwidth Delay Products
  - Linear increase artefacts

Chapter 3: Summary

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

Next:

- leaving the network "edge" (application, transport layers)
- into the network "core"