Chapter 3
Transport Layer

Transport Layer

Chapter goals:

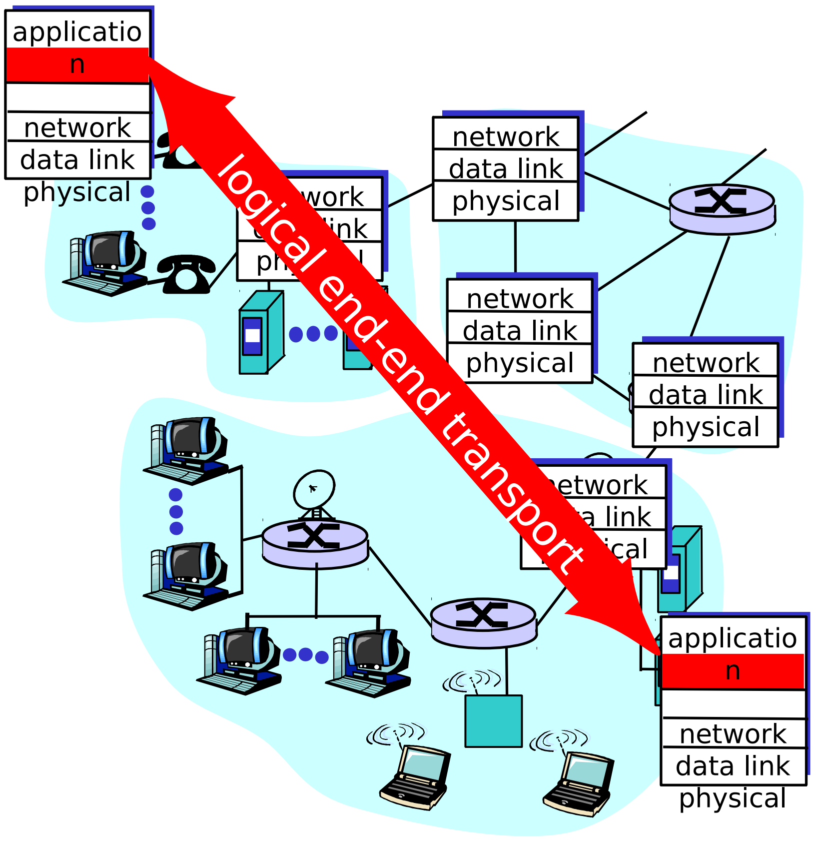
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet

Chapter Overview:

- transport layer services
- multiplexing/demultiplexing
- connectionless transport: UDP
- principles of reliable data transfer
- connection-oriented transport: TCP
  - reliable transfer
  - flow control
  - connection management
- principles of congestion control
- TCP congestion control

Transport services and protocols

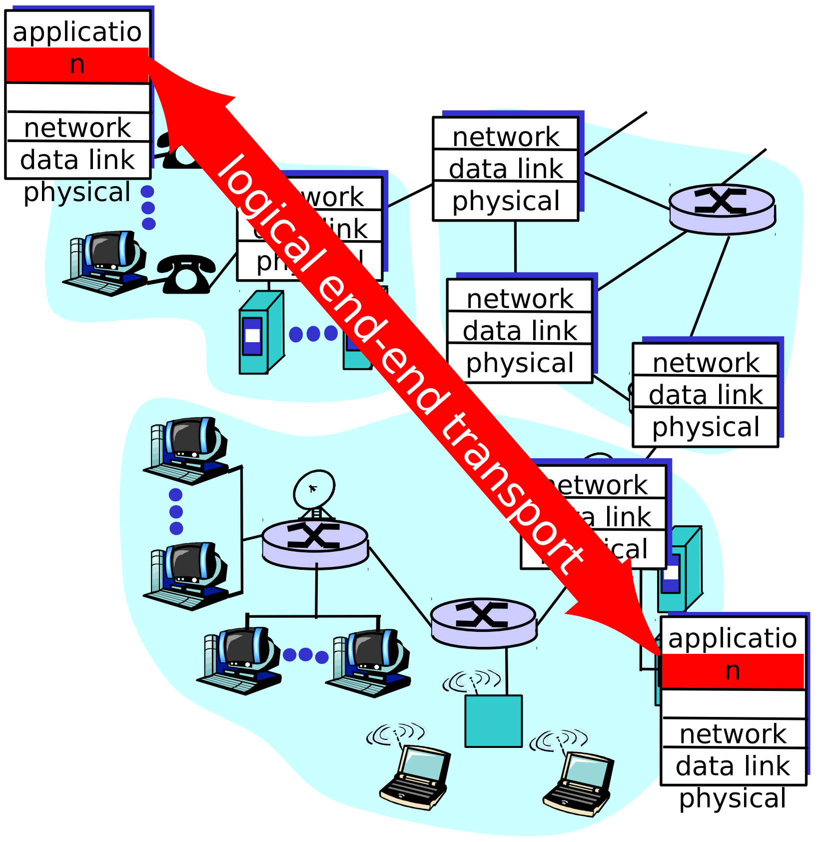
- provide *logical communication* between app' processes running on different hosts
- transport protocols run in end systems
- transport vs network layer services:
- *network layer:* data transfer between end systems
- *transport layer:* data transfer between processes
  - relies on, enhances, network layer services



Transport-layer protocols

Internet transport services:

- reliable, in-order unicast delivery (TCP)
  - congestion
  - flow control
  - connection setup
- unreliable ("best-effort"), unordered unicast or multicast delivery: UDP
- services not available:
  - real-time
  - bandwidth guarantees
  - reliable multicast

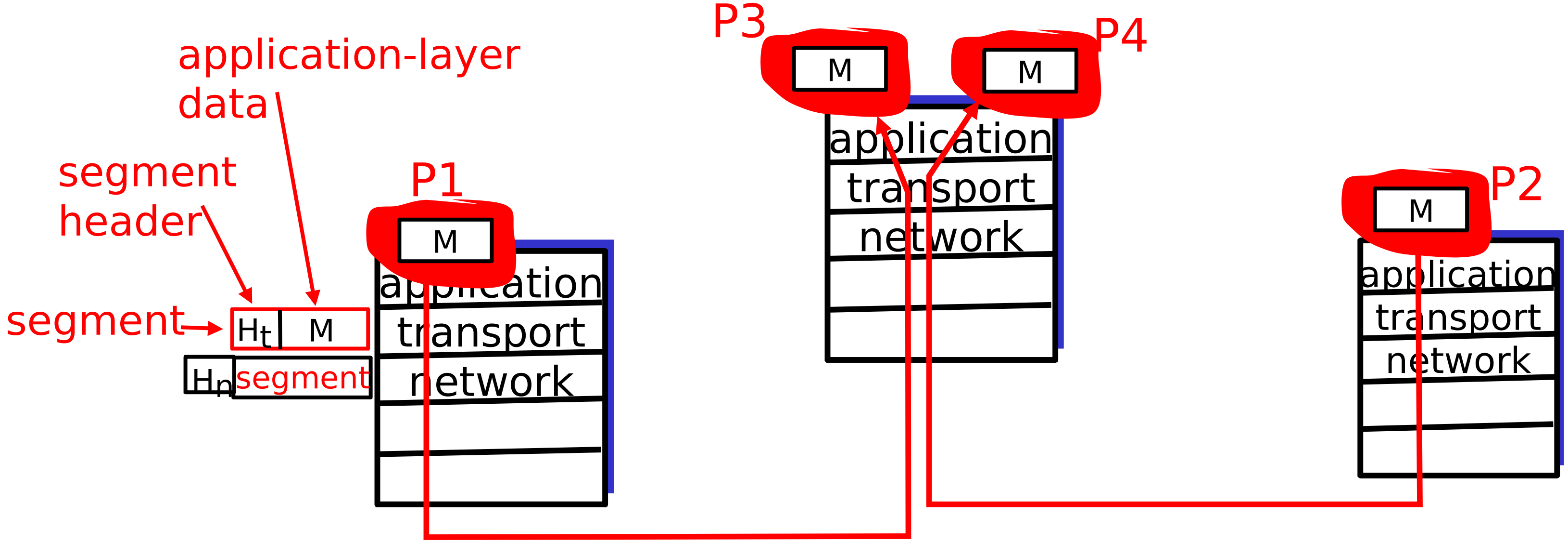


Multiplexing/demultiplexing

Recall: *segment* - unit of data exchanged between transport layer entities

- aka TPDU: transport protocol data unit

Demultiplexing: delivering received segments to correct app layer processes

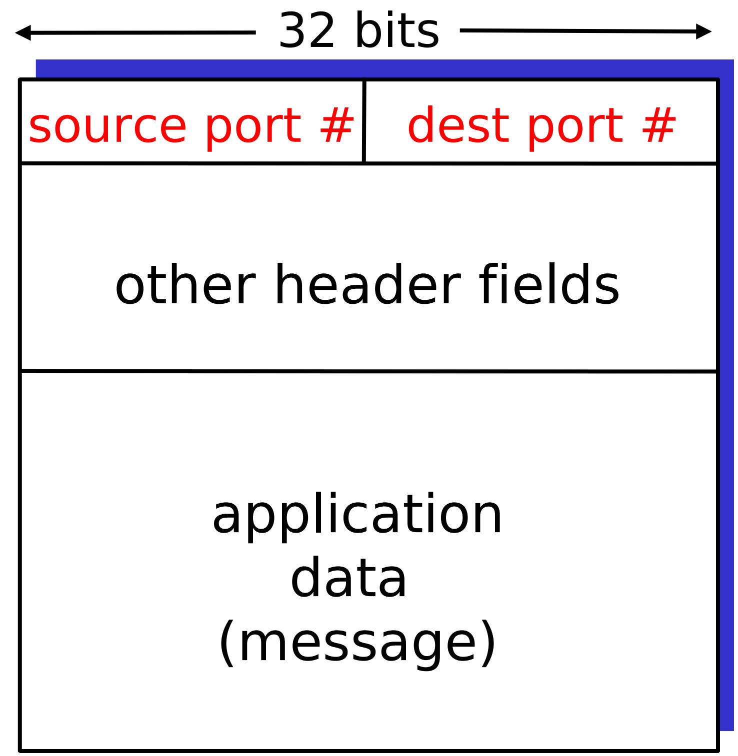


Multiplexing/demultiplexing

Multiplexing:
gathering data from multiple app processes, enveloping data with header (later used for demultiplexing)

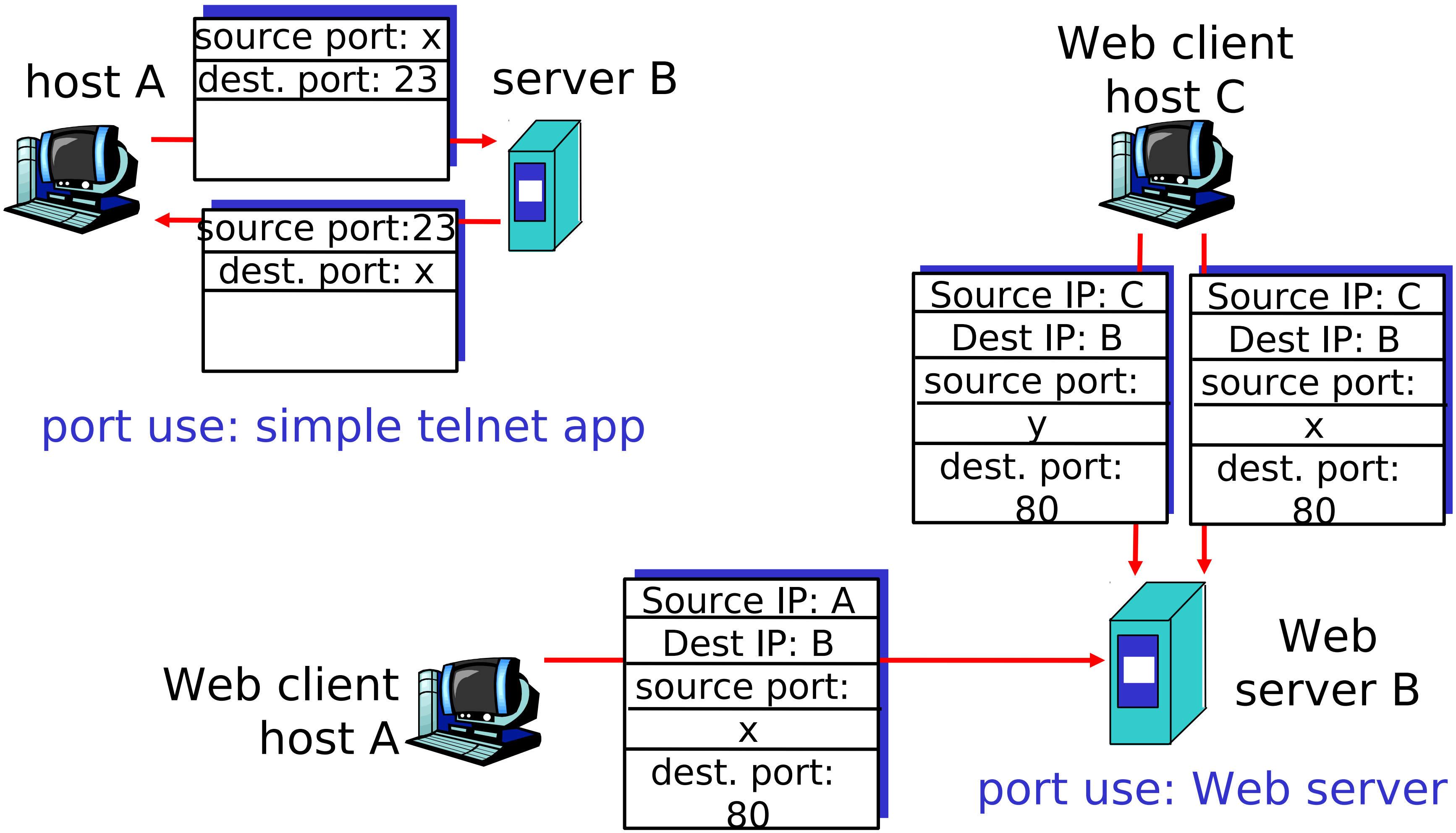
multiplexing/demultiplexing:

- based on sender, receiver port numbers, IP addresses
  - source, dest port #s in each segment
  - recall: well-known port numbers for specific applications



TCP/UDP segment format

Multiplexing/demultiplexing: examples



UDP: User Datagram Protocol [RFC 768]

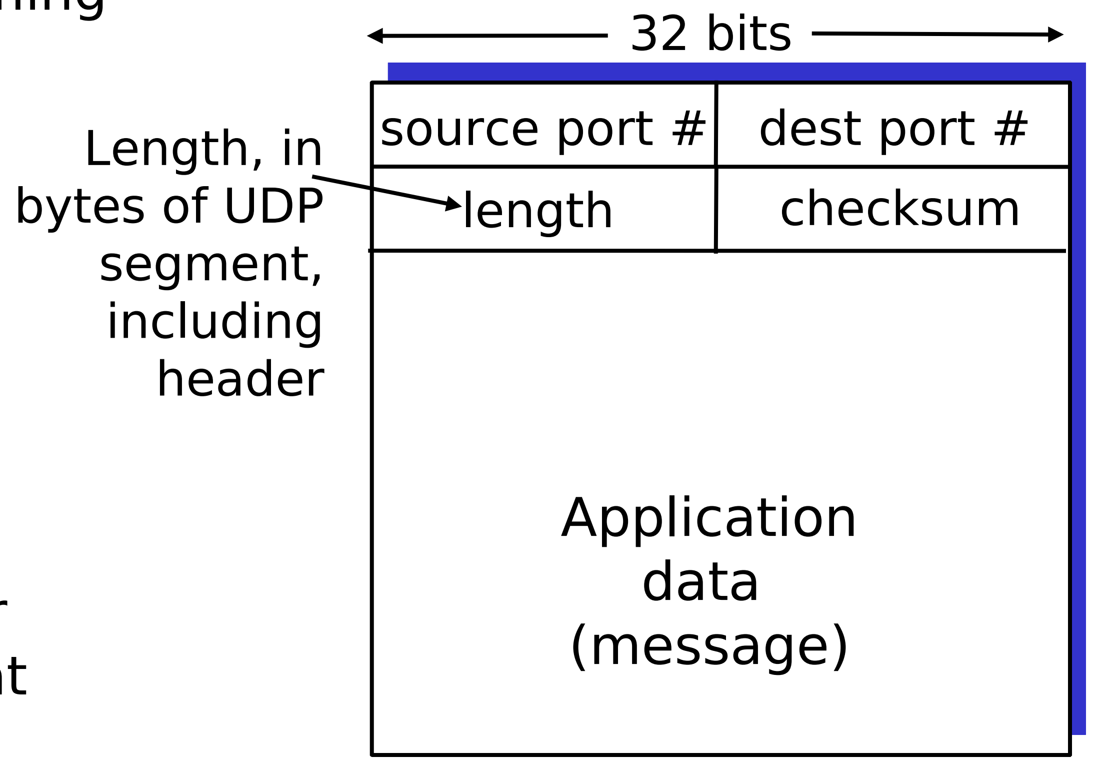
- “no frills,” “bare bones” Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

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 (why?):
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recover!



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:

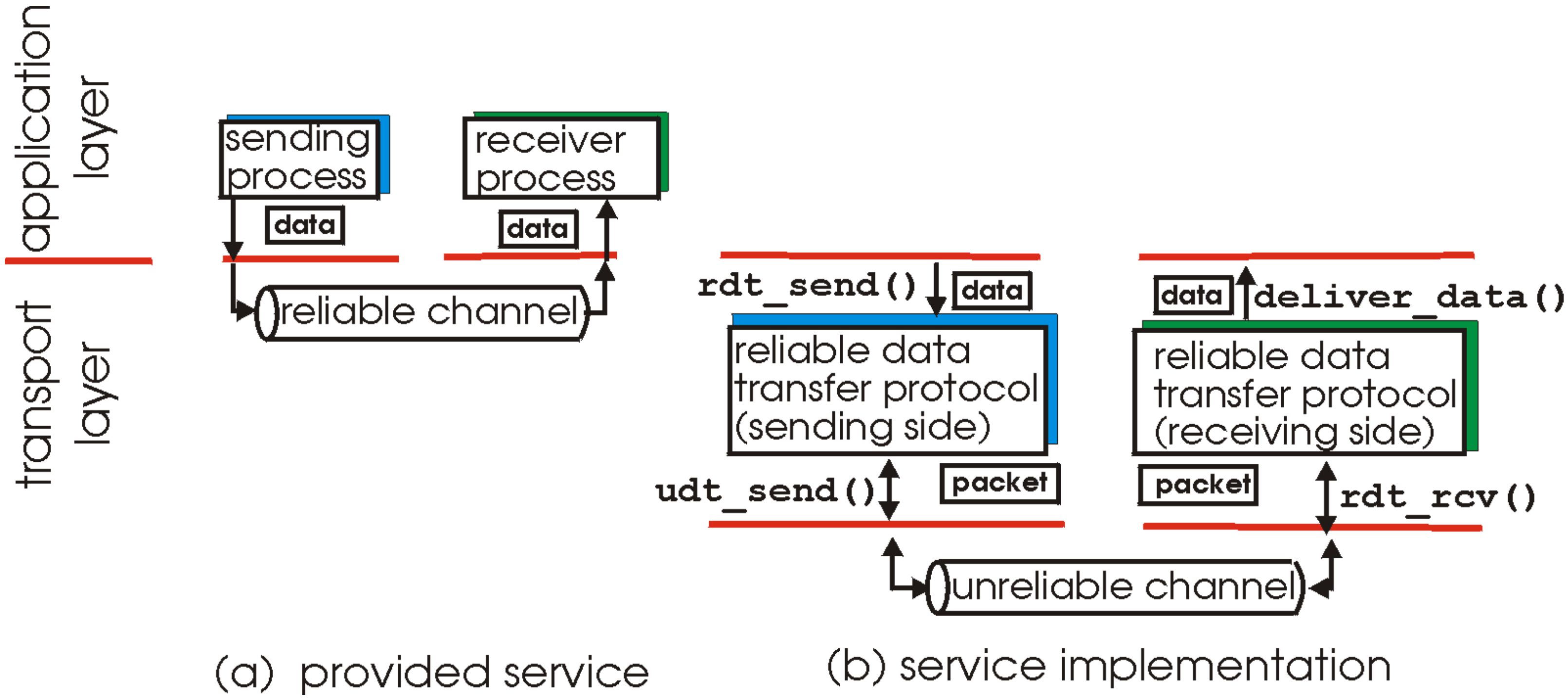
- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonethless?* More later ….

UDP checksum example:

- Three packets of 16 bits each
  - 0110011001100110
  - 0101010101010101
  - 0000111100001111
- adding the three, calling it 'r':
  - 1100101011001010
- Send the four packets, the original three and 1's complement of 'r' to destination
- The 1's complement of 'r' is:
  - 0011010100110101
- at destination the sum of four packets should be:
  - 1111111111111111
- If the packet is damaged:
  - 1111101111111111 (zeros!!)

Principles of Reliable data transfer

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

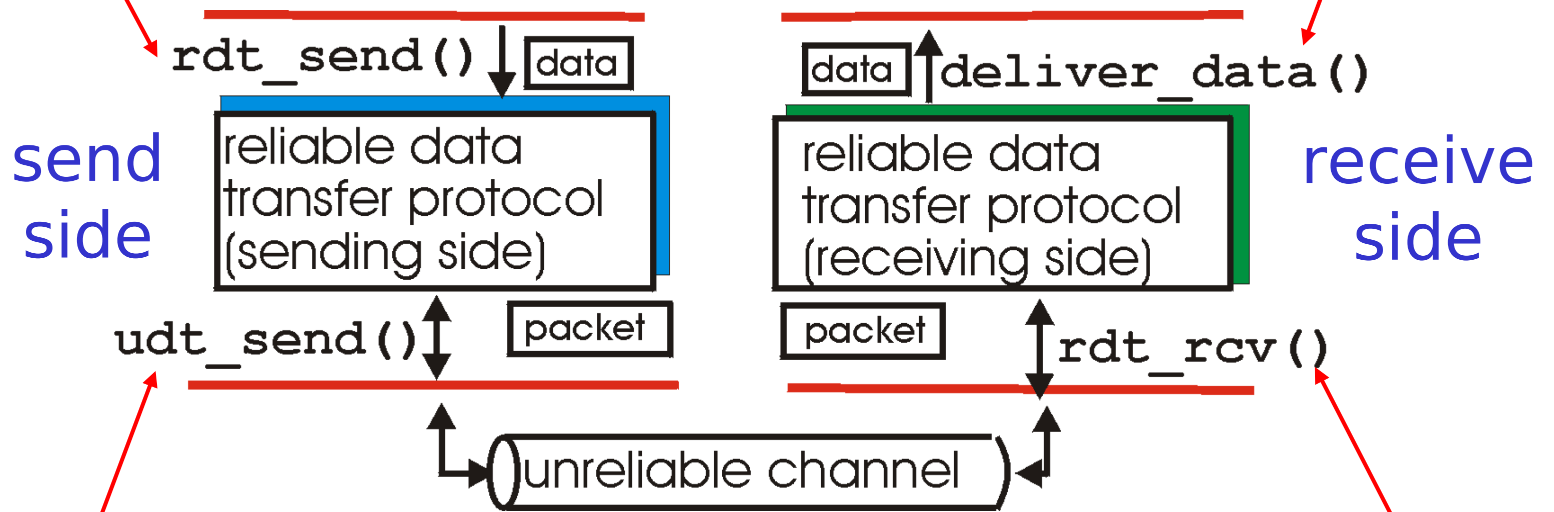


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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

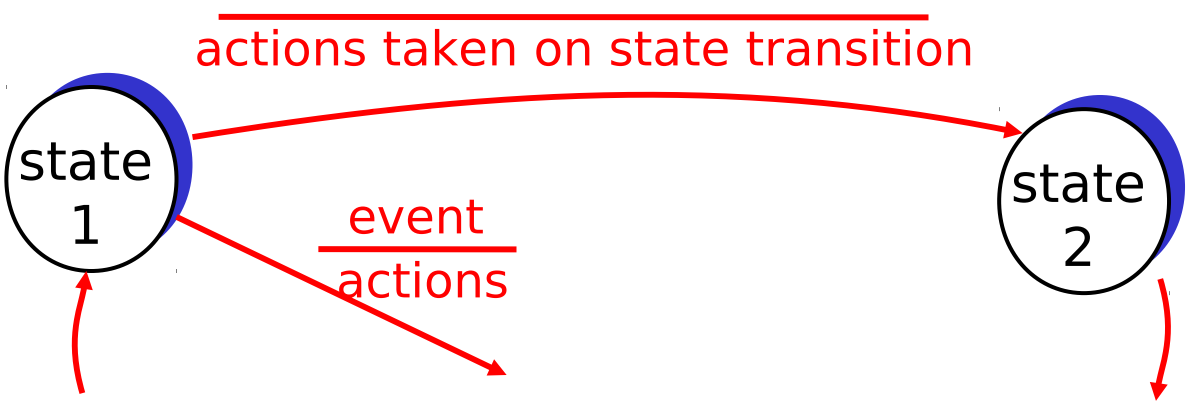
rdt_rcv(): called when packet arrives on rcv-side of channel

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

state: when in this "state" next state uniquely determined by next event



Rdt1.0: reliable transfer over a reliable channel

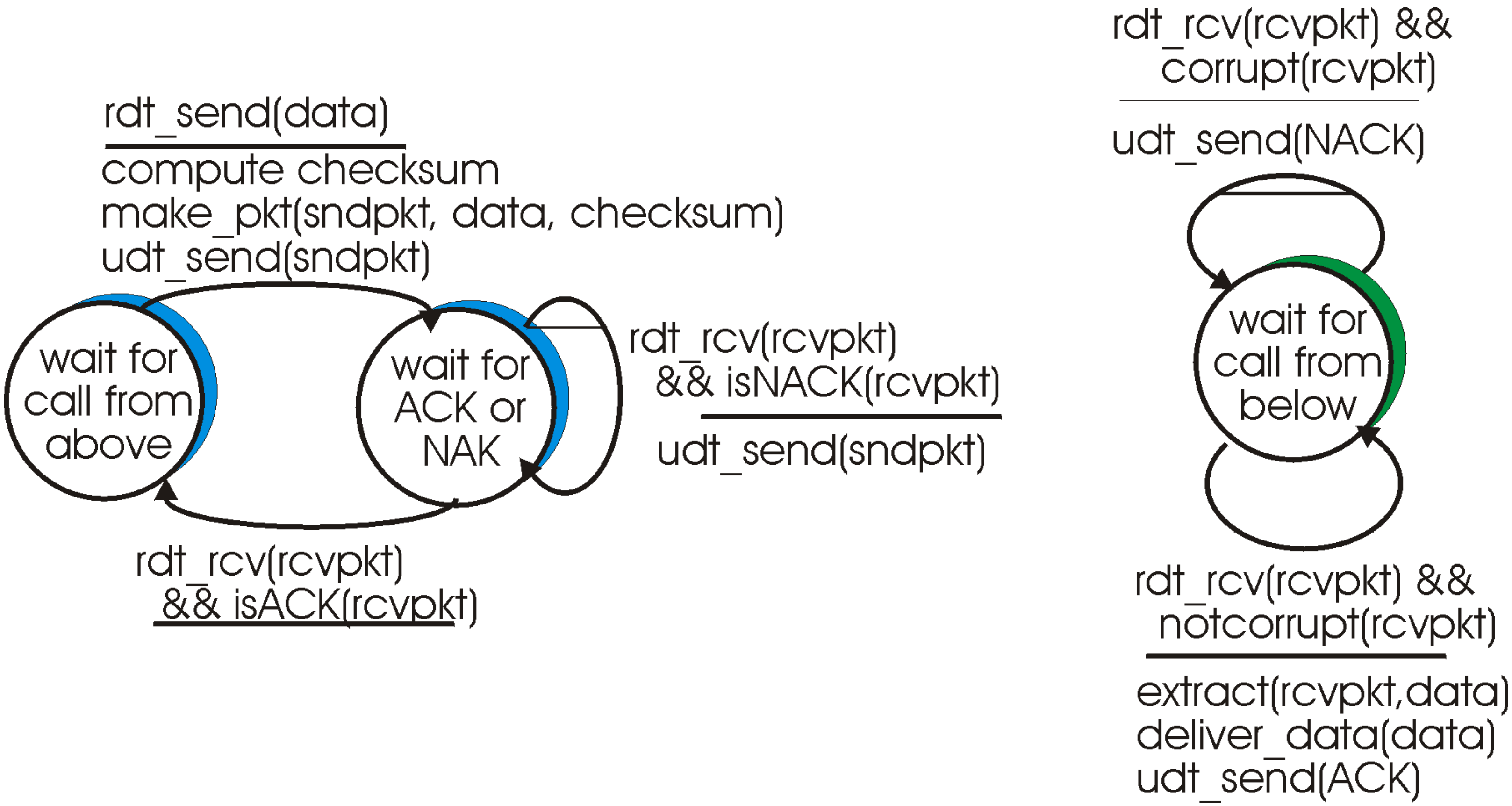
- underlying channel perfectly reliable
  - no bit erros
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel

(a) rdt1.0: sending side (b) rdt1.0: receiving side

Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - recall: UDP 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
  - human scenarios using ACKs, NAKs?
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

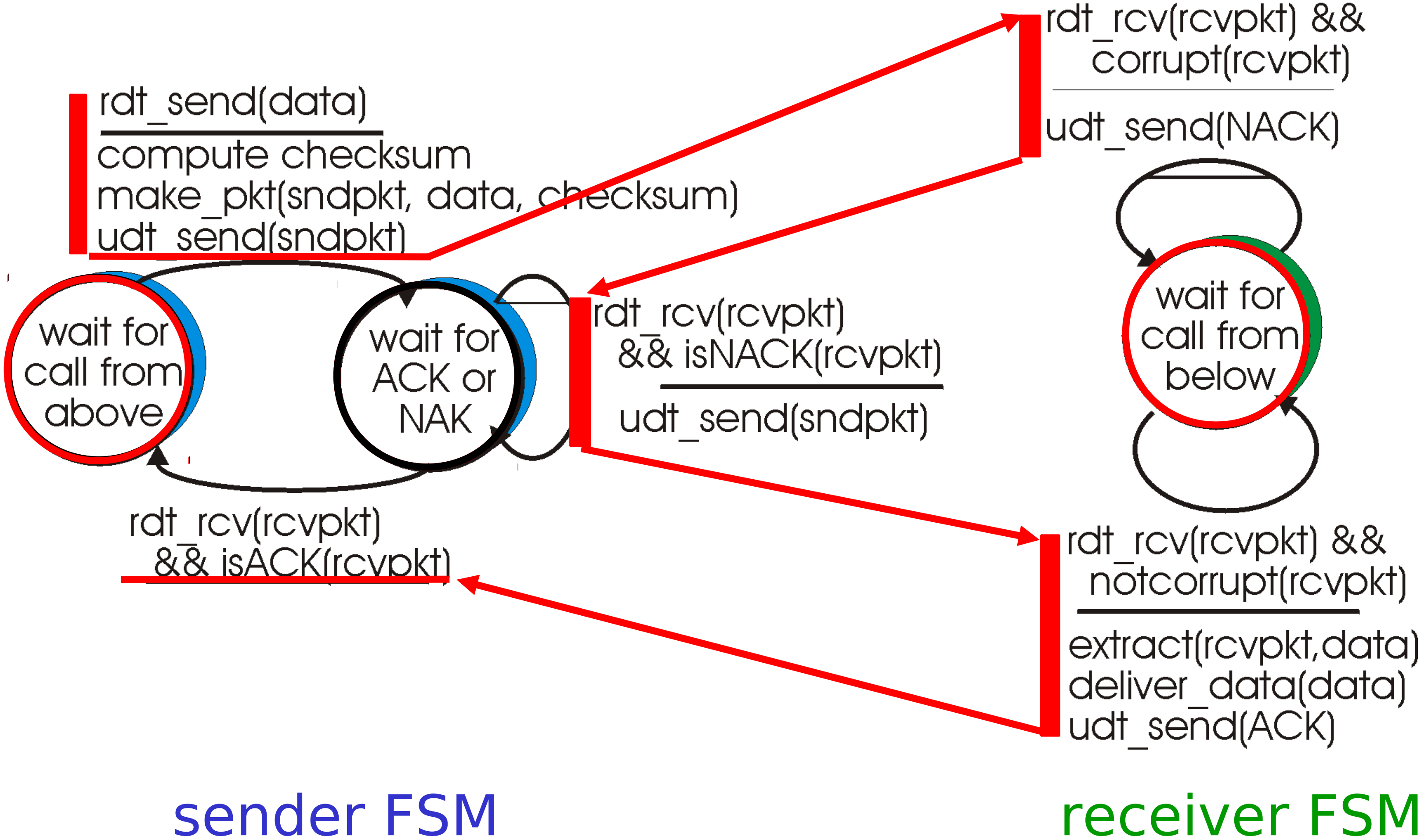
rdt2.0: FSM specification



sender FSM

receiver FSM

rdt2.0: in action (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

What to do?

- sender ACKs/NAKs receiver’s ACK/NAK? What if sender ACK/NAK lost?
- retransmit, but this might cause retransmission of correctly received pkt!

Handling duplicates:

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

stop and wait

Sender sends one packet, then waits for receiver response

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

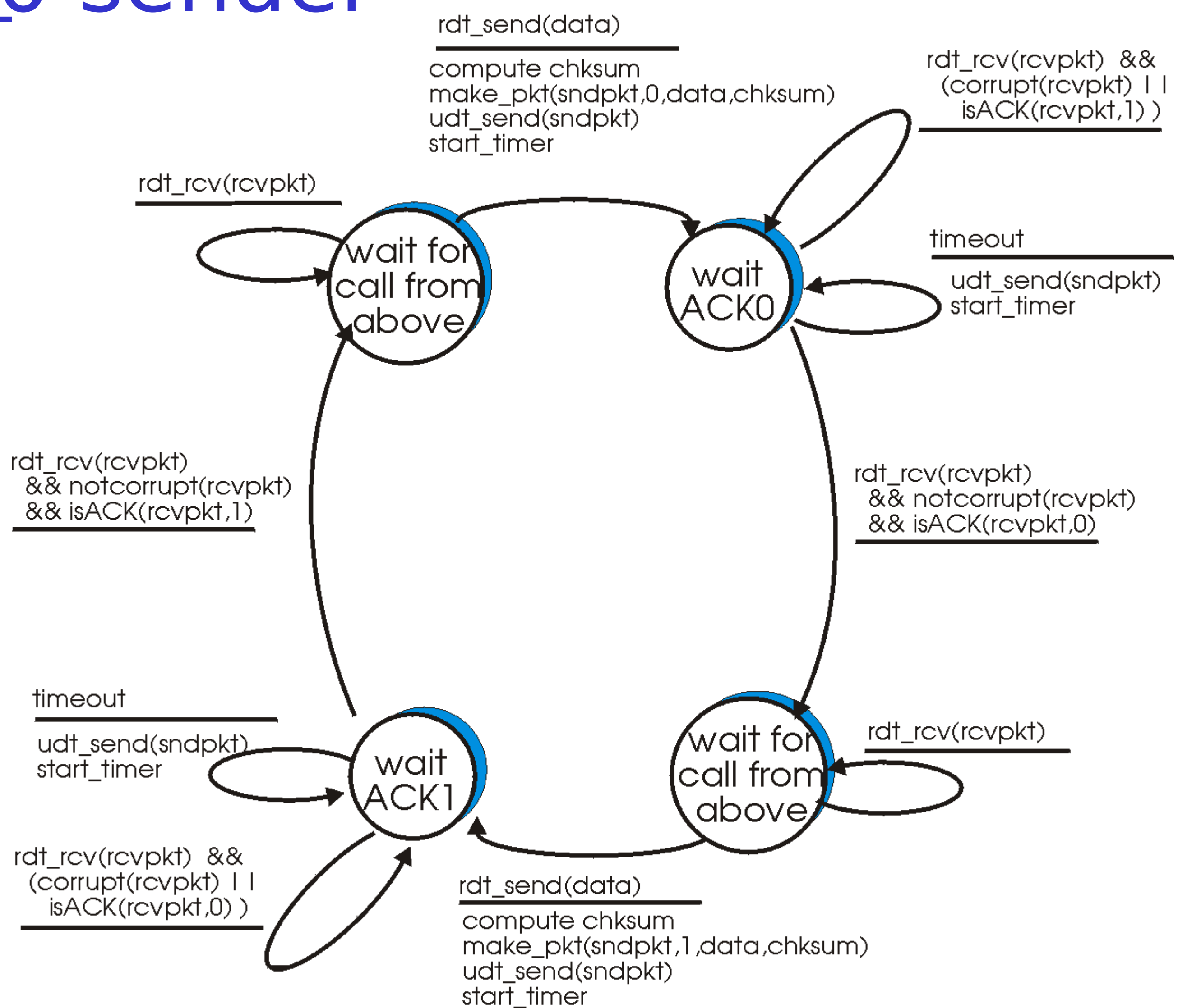
Q: how to deal with loss?

- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

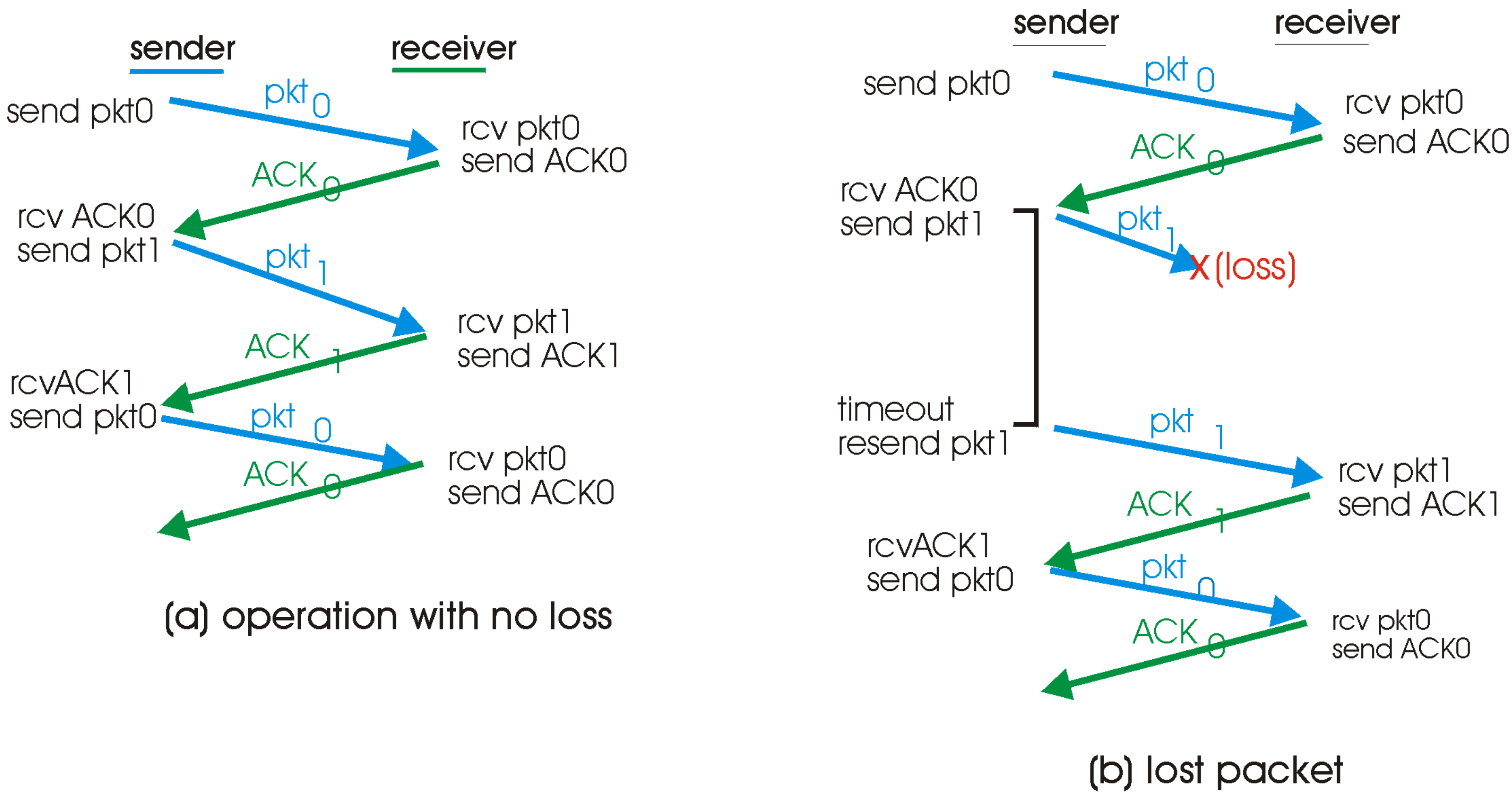
Approach: sender waits "reasonable" amount of time for ACK

- 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

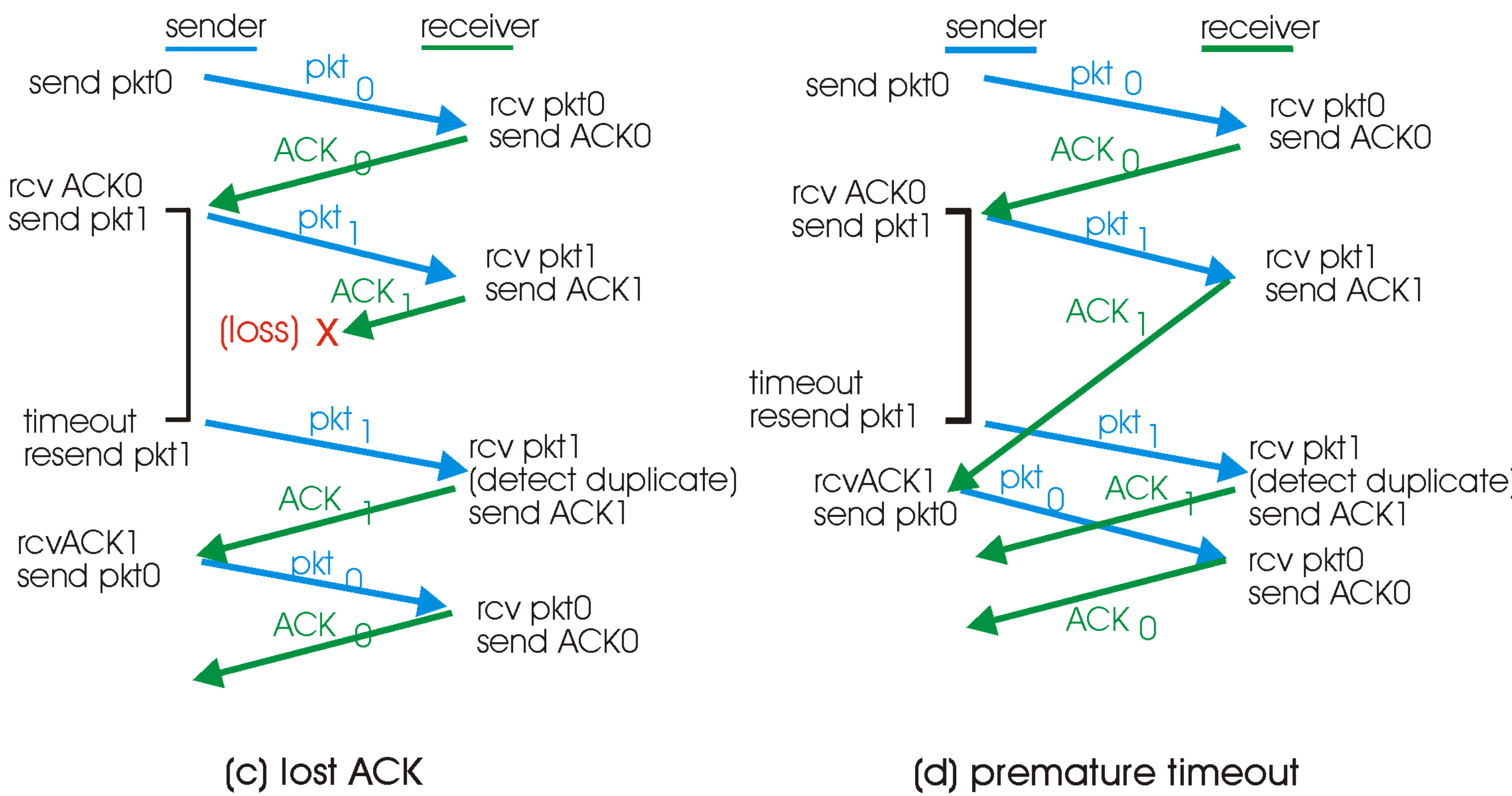
rdt3.0 sender



rdt3.0 (hypothetical protocol)in action



rdt3.0 in action



Performance of rdt3.0

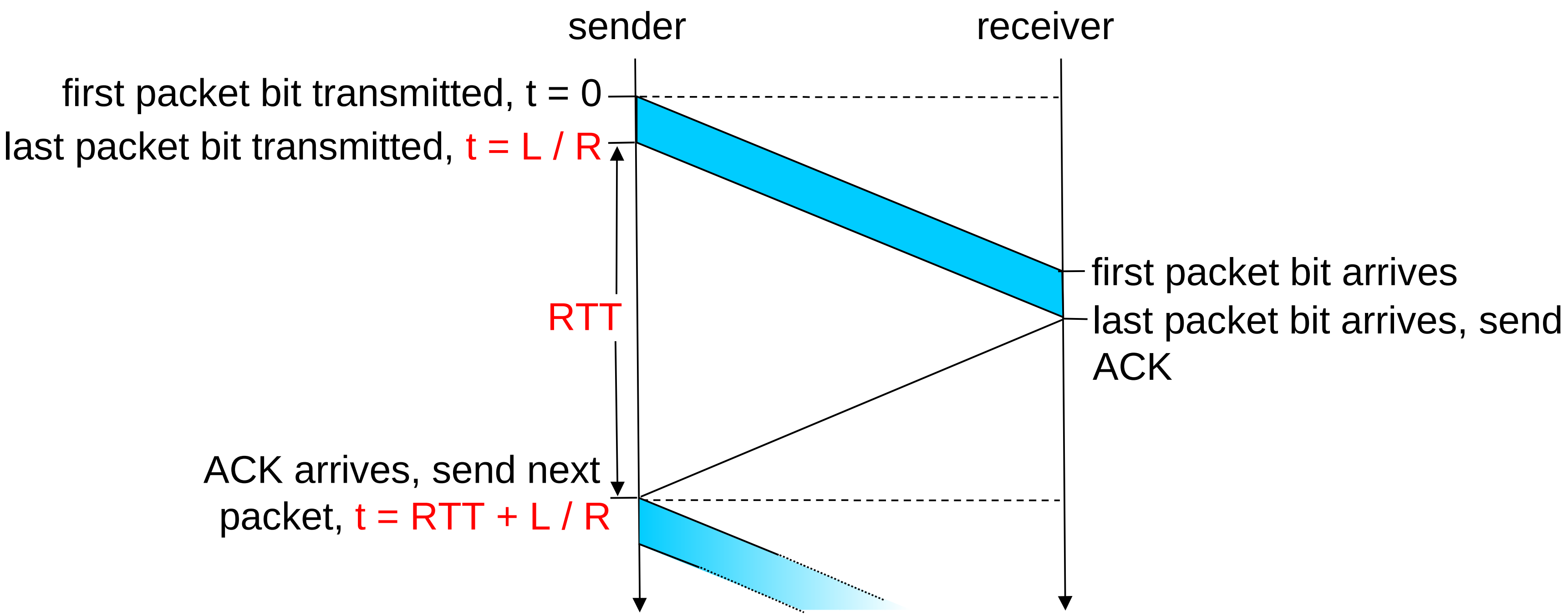
- rdt3.0 works, but performance is not acceptable
- example: 1 Gbps link, 15 ms end to end propagation delay, 1KB packet (1KByte = 8Kbit)
- Utilization of sender (time busy sending)

$$T_{transmit} = \frac{8kb/pkt}{10^{**}9\ b/sec} = 8\ microsec$$

$$\text{Utilization} = U = \frac{\text{fraction of time}}{\text{sender busy sending}} = \frac{0.008\ msec}{30.008\ msec} = 0.00027$$

- 1KB packet every 30 msec -> 267kb/sec throughput over 1 Gbps link
- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

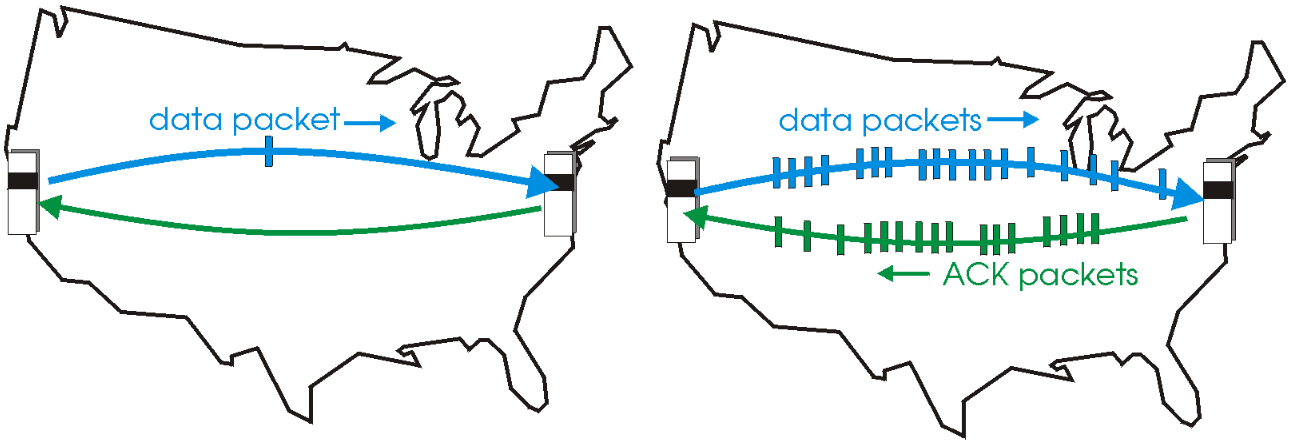


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged packets

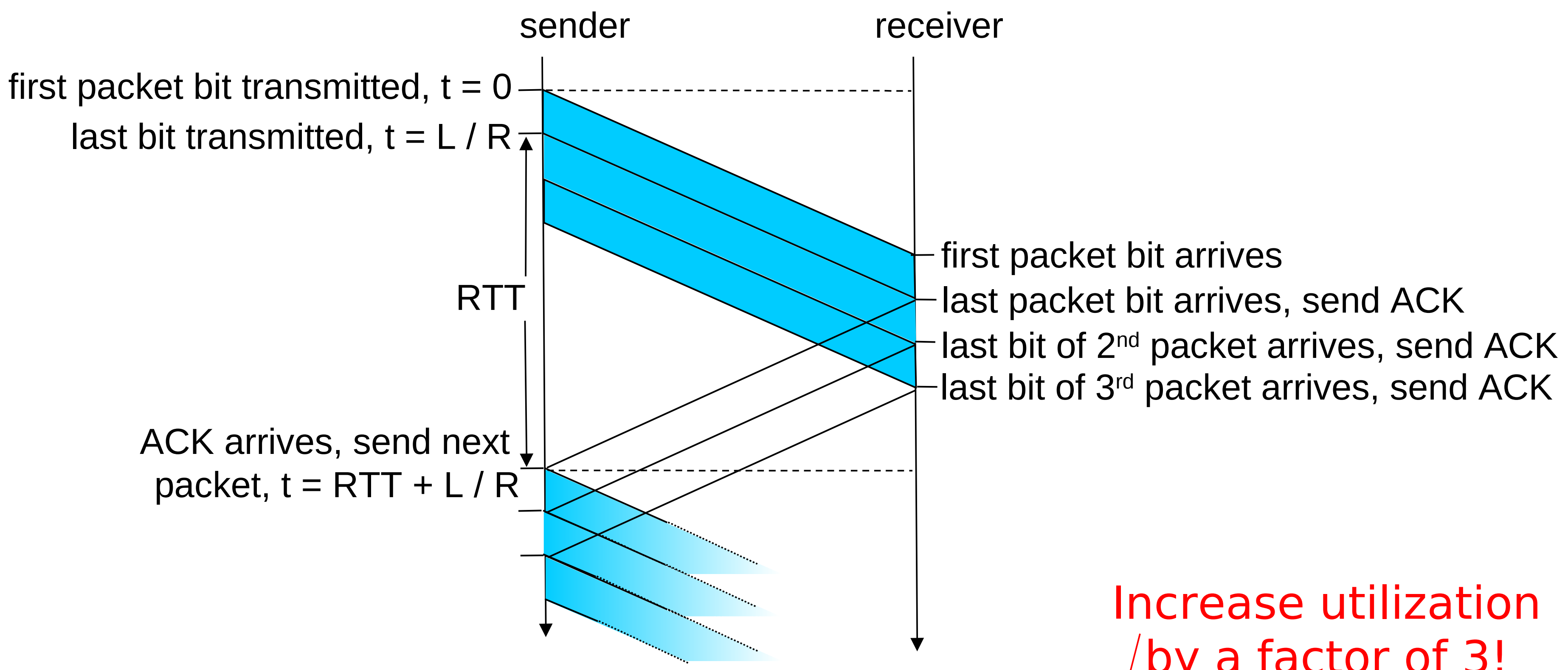
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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



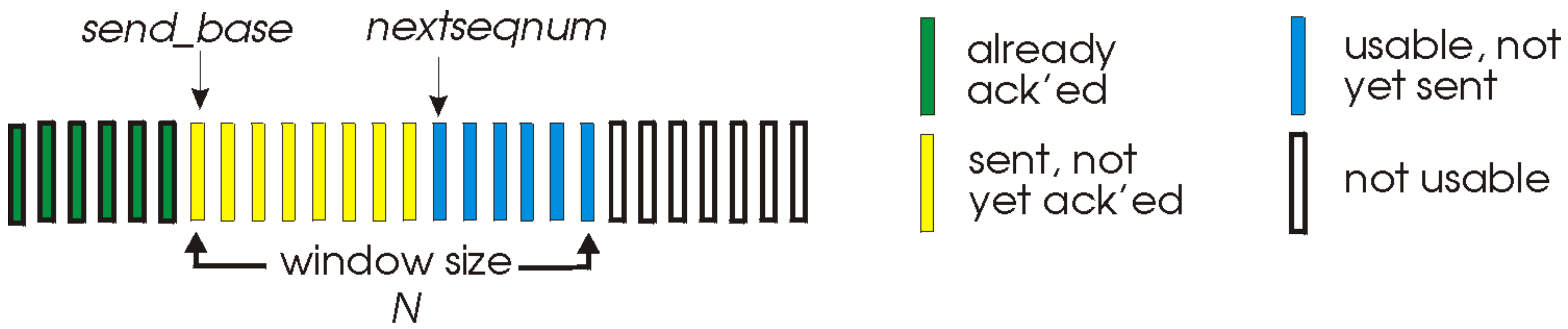
Increase utilization by a factor of 3!

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

Go-Back-N

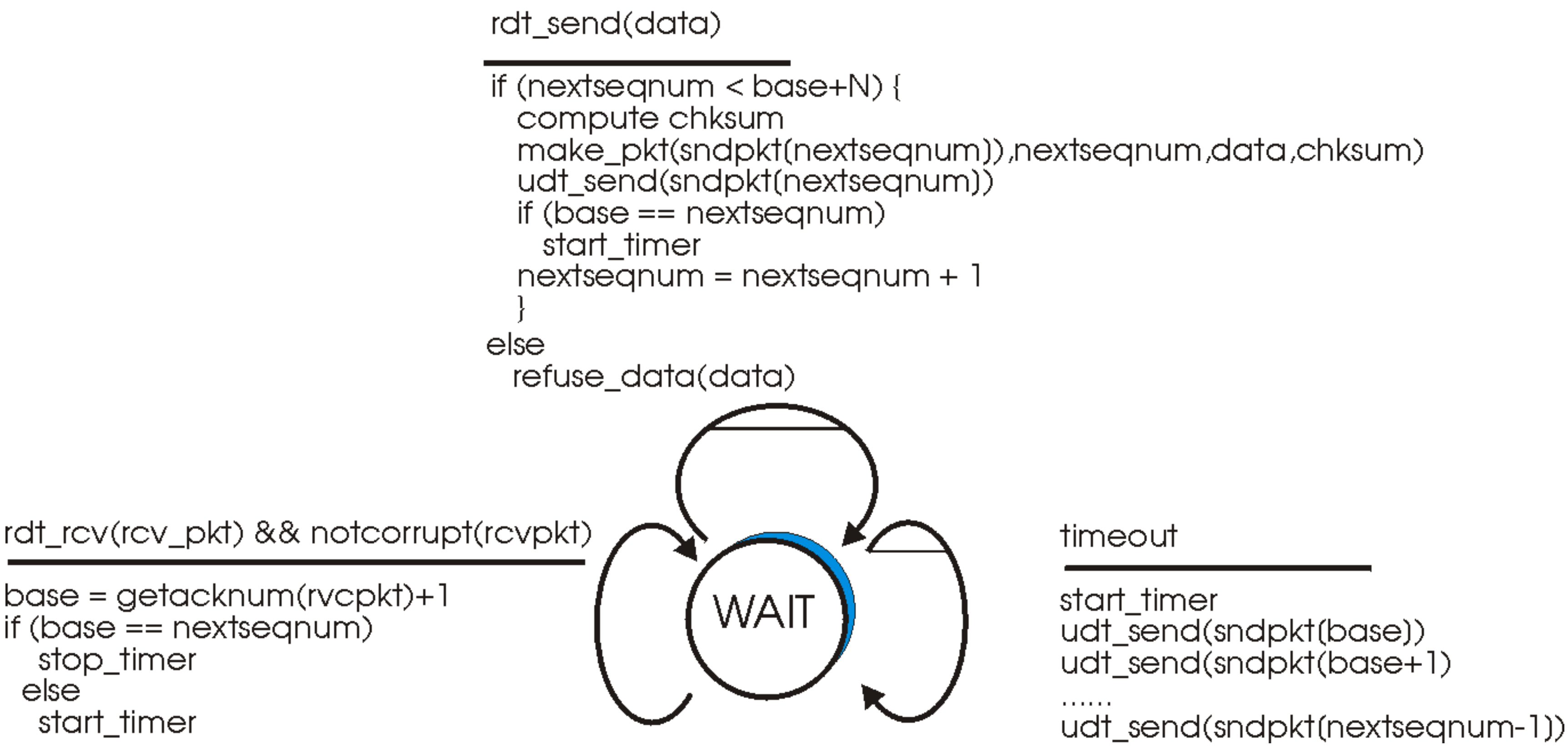
Sender:

- k-bit sequence # in packet header
- “window” of up to N, consecutive unack’ed packets allowed

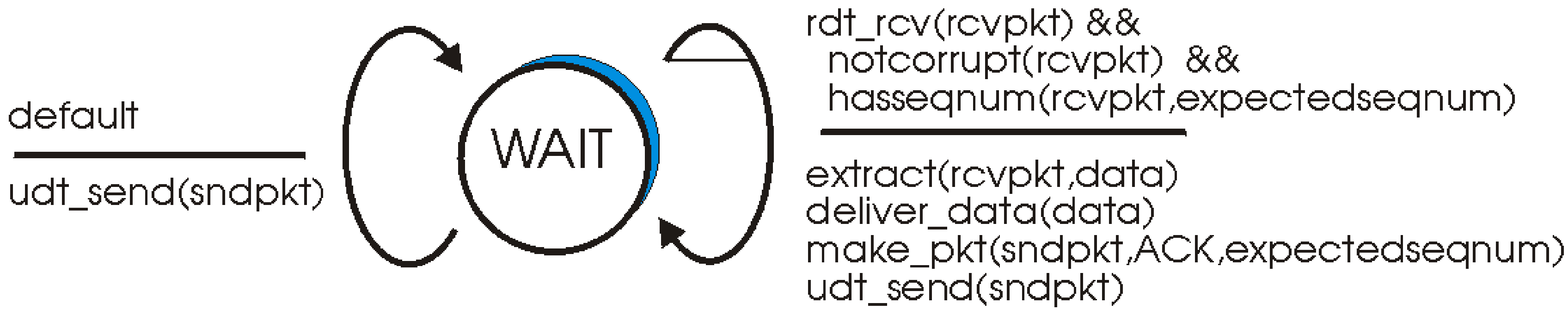


- ACK(n): ACKs all pkts up to, including seq # n - “cumulative ACK”
  - may deceive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



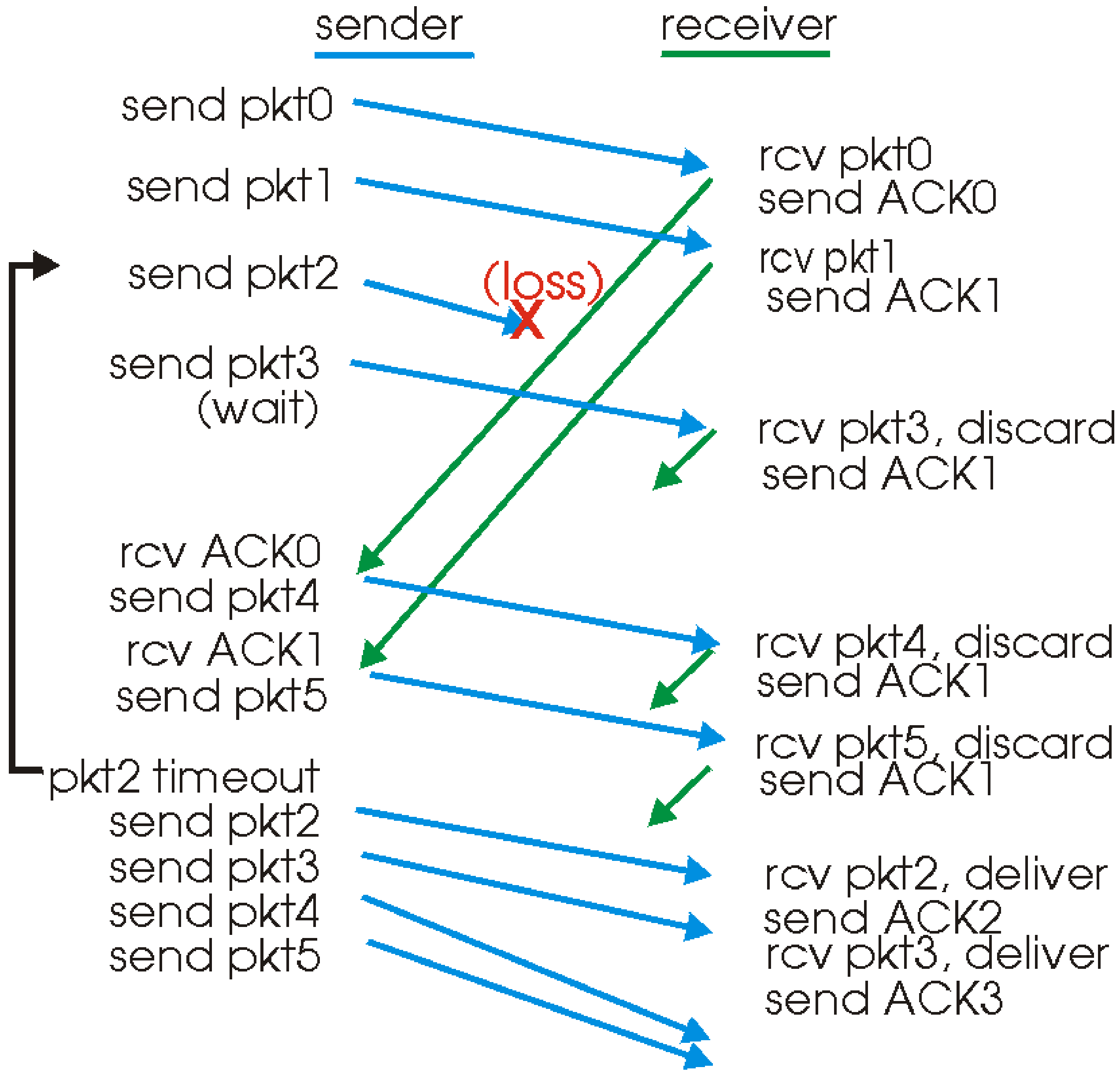
GBN: receiver extended FSM



receiver simple:

- 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!
  - ACK pkt with highest in-order seq #

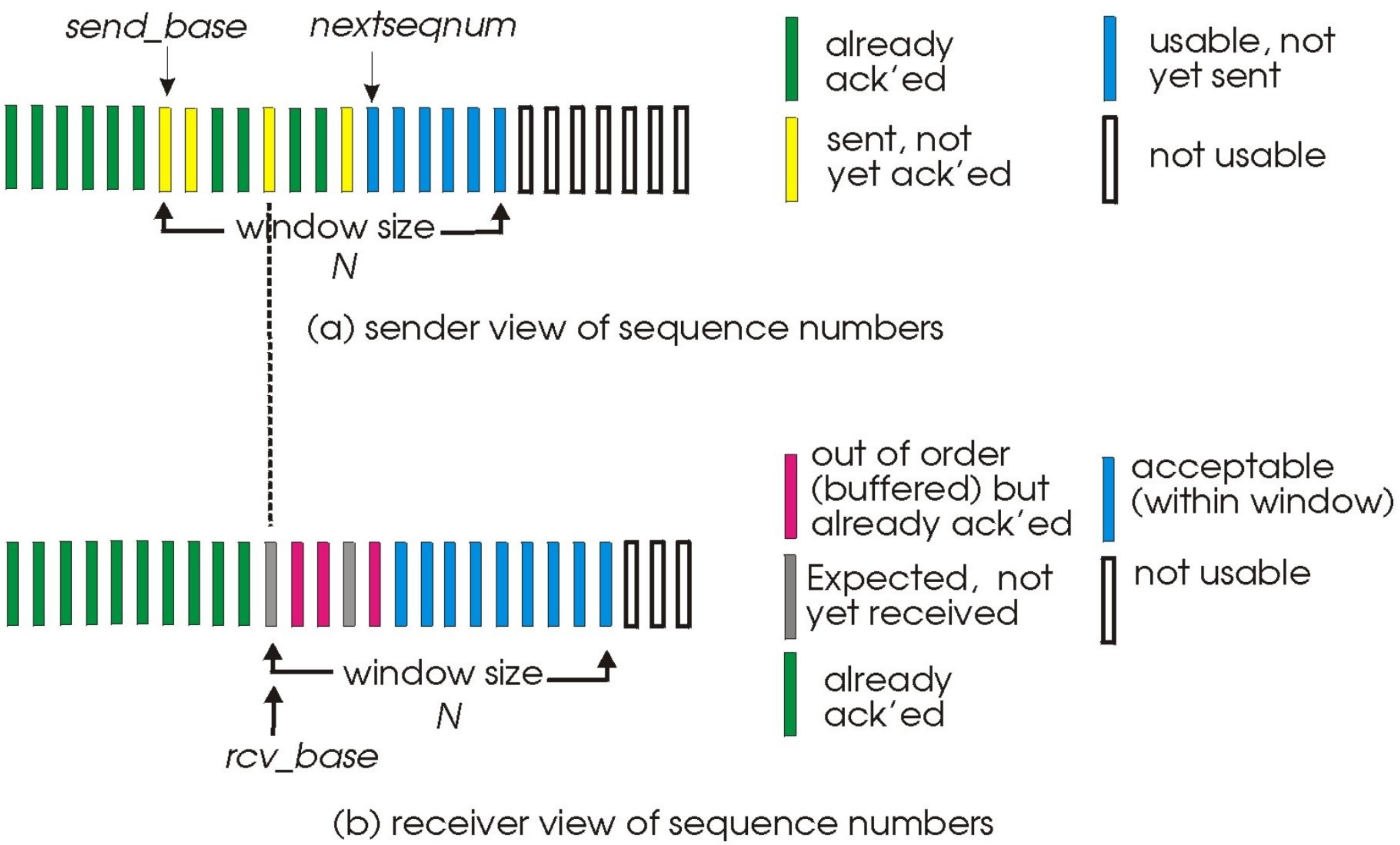
GBN in action



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

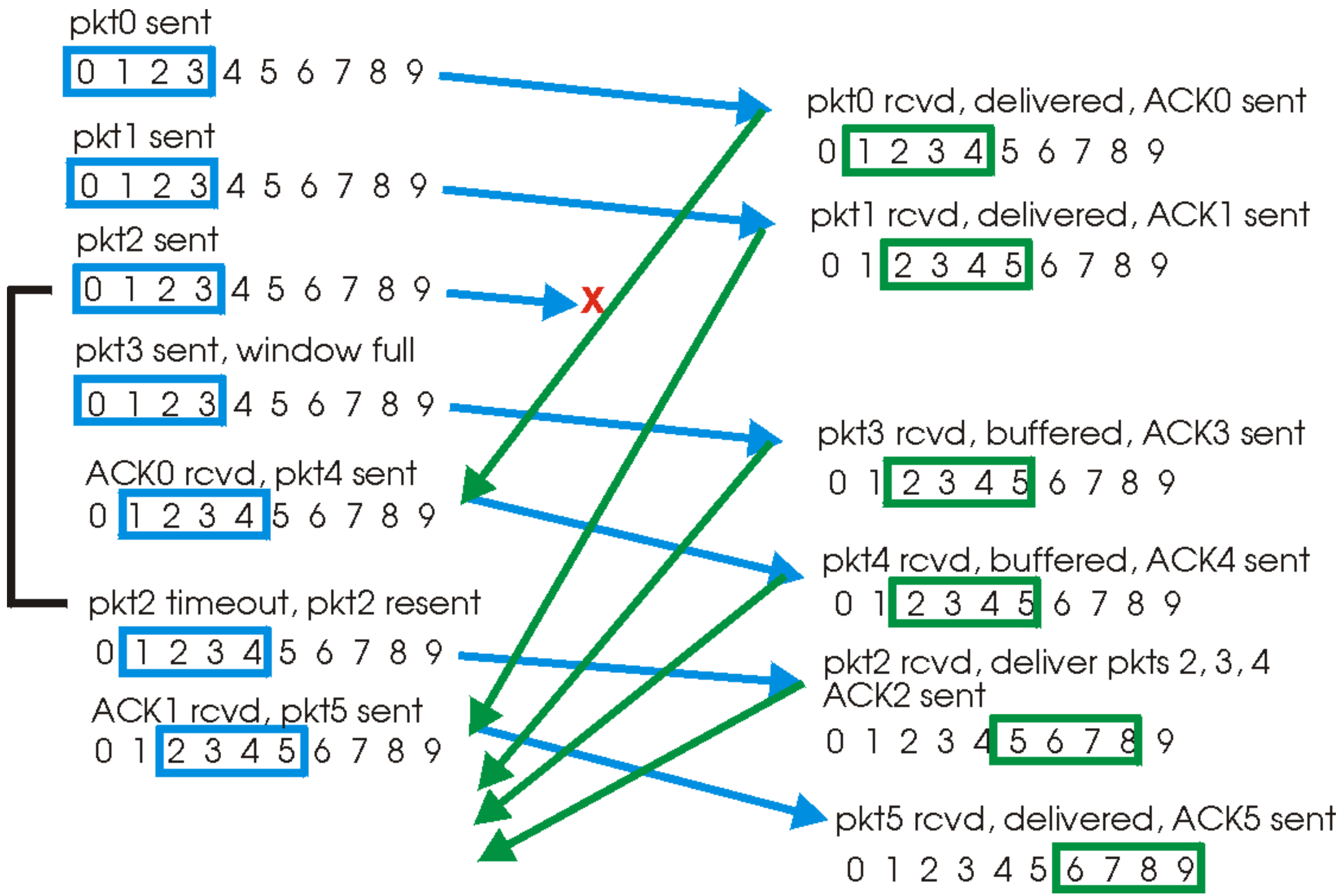
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

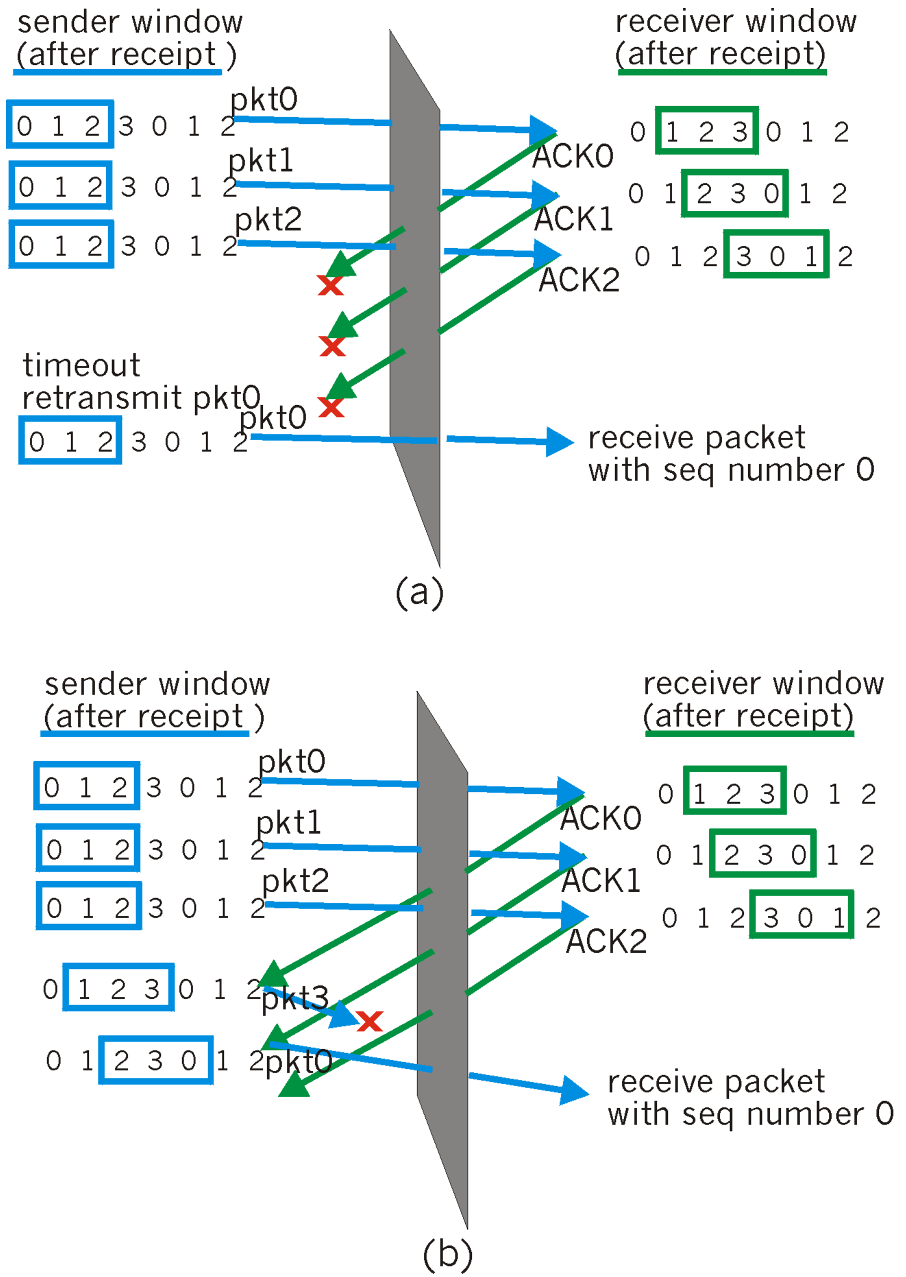
Selective repeat in action



Selective repeat: dilemma

Example:

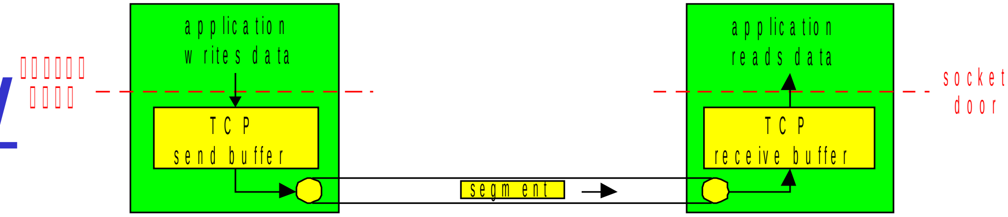
- seq #'s: 0, 1, 2, 3
- window size=3



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

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

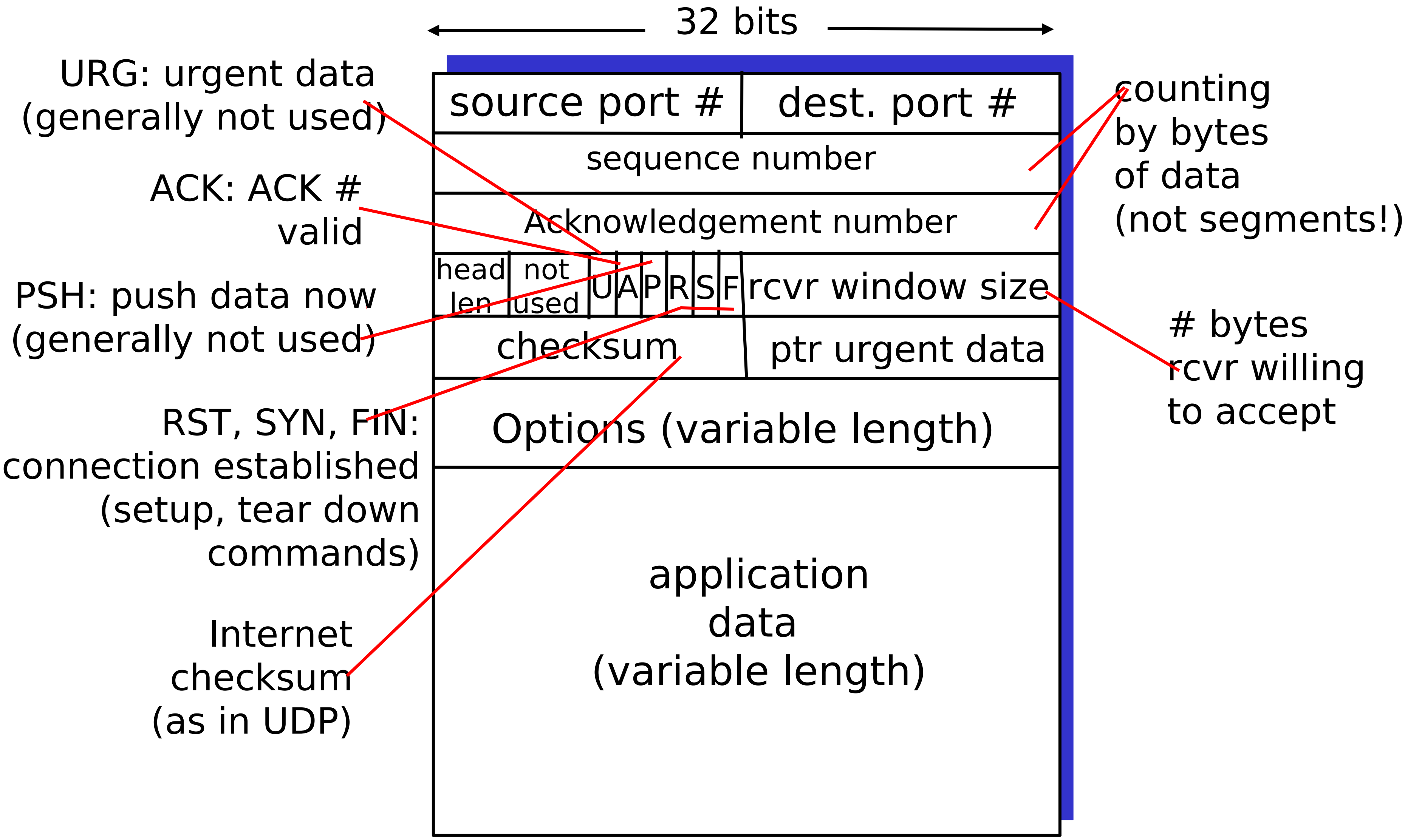
TCP: Overview



- 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 "flood" receiver with data

TCP segment structure



TCP sequence #'s and ACKs

Segment 1 | Segment 2 | ...

0 1 2 3 4 5	1000 1001 1002	

Sequence. Numbers (#'s):

- byte stream 'number' of first byte in segment's data
- Not necessarily starts from 0, use random initial number R
  - Segment 1: 0 + R
  - Segment 2: 1000 + R etc...

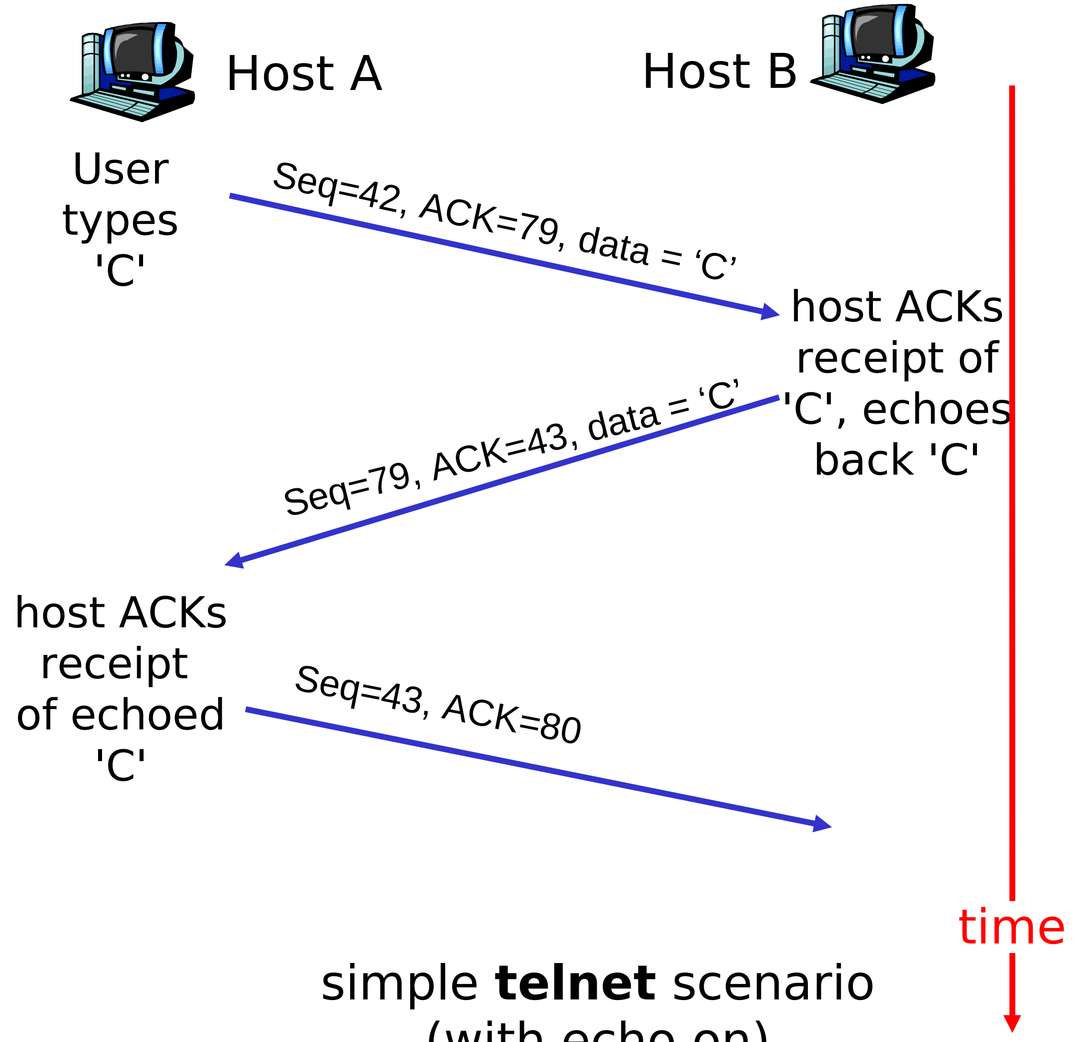
ACKs (acknowledgment):

- seq # of next byte expected from other side (last byte +1)
- cumulative ACK
- If received segment 1, waits for segment 2
- Ack=1000 + R (received up to 999^{th} byte)

TCP sequence #'s and ACKs

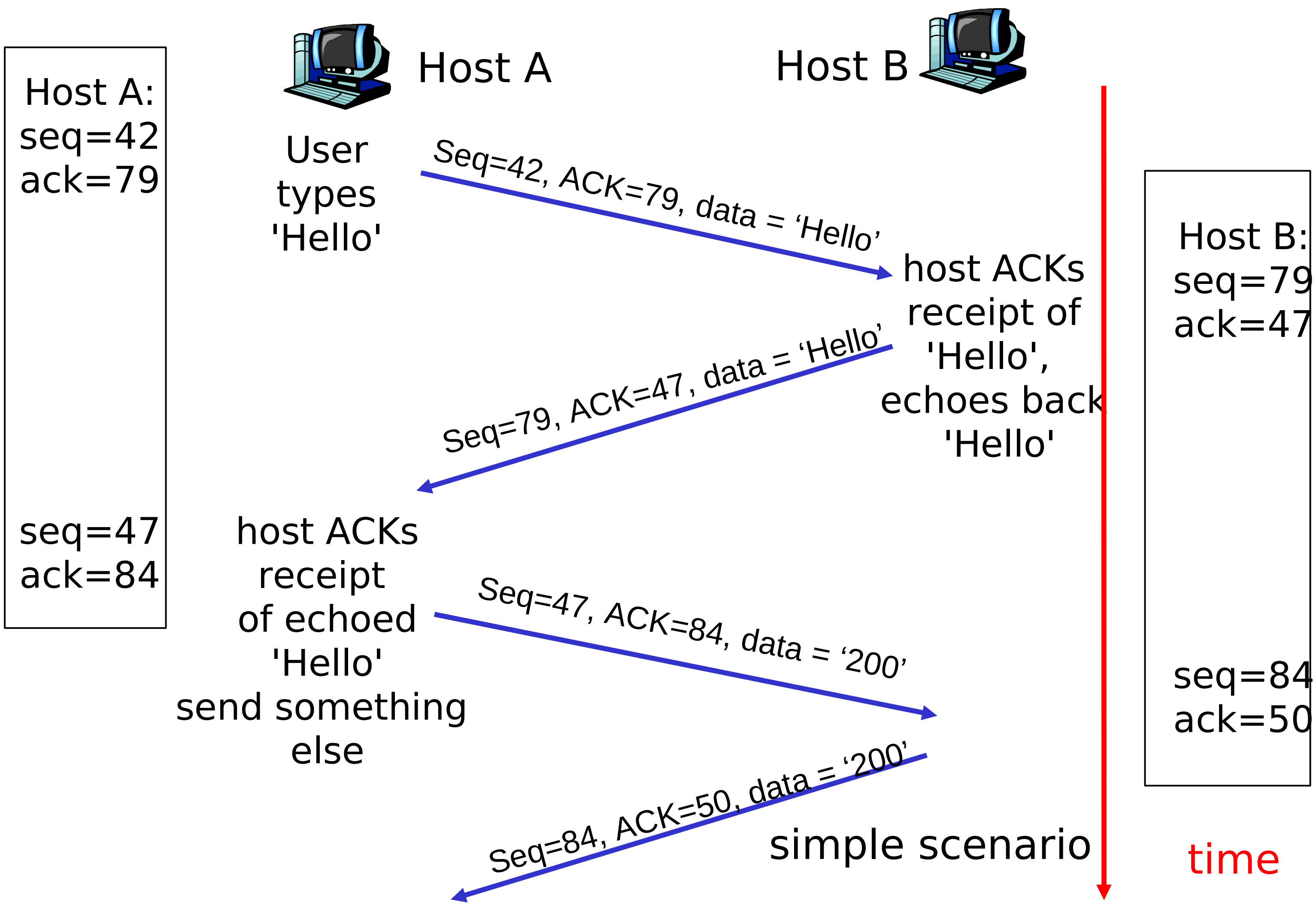
Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - decide when implementing

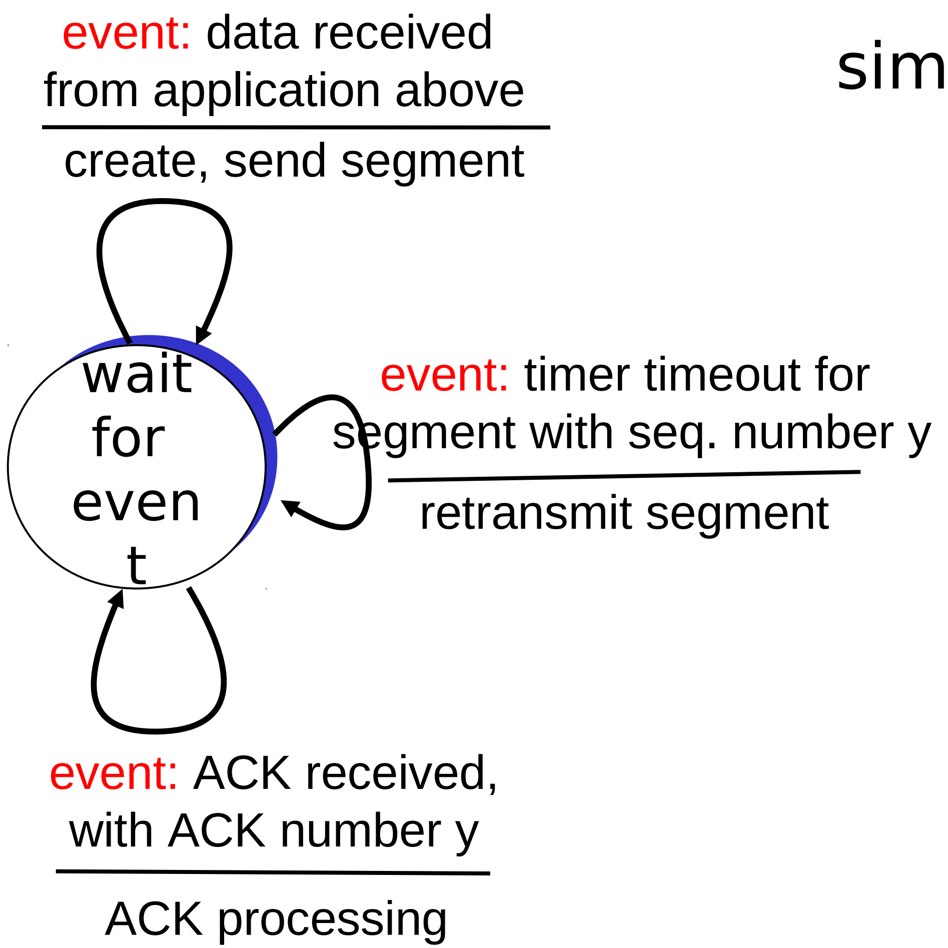


simple **telnet** scenario (with echo on)

Yet another example



TCP: reliable data transfer



simplified sender, assuming

- one way data transfer
- no flow, congestion control

TCP: reliable data transfer

Simplified TCP sender

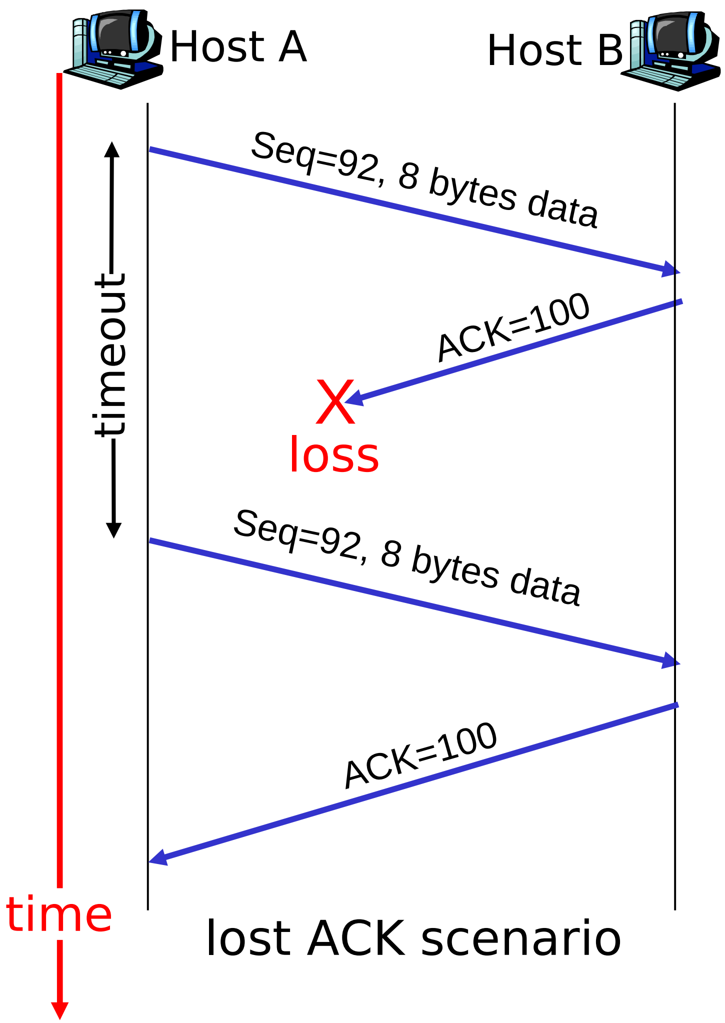
```
sendbase = initial_sequence number
nextseqnum = initial_sequence number

loop (forever) {
  switch(event)
  event: data received from application above
       create TCP segment with sequence number nextseqnum
       start timer for segment nextseqnum
       pass segment to IP
       nextseqnum = nextseqnum + length(data)
  event: timer timeout for segment with sequence number y
       retransmit segment with sequence number y
       compute new timeout interval for segment y
       restart timer for sequence number y
  event: ACK received, with ACK field value of y
       if (y > sendbase) { /* cumulative ACK of all data up to y */
           cancel all timers for segments with sequence numbers < y
           sendbase = y
           }
       else { /* a duplicate ACK for already ACKed segment */
           increment number of duplicate ACKs received for y
           if (number of duplicate ACKS received for y == 3) {
               /* TCP fast retransmit */
               resend segment with sequence number y
               restart timer for segment y
           }
  } /* end of loop forever */
```

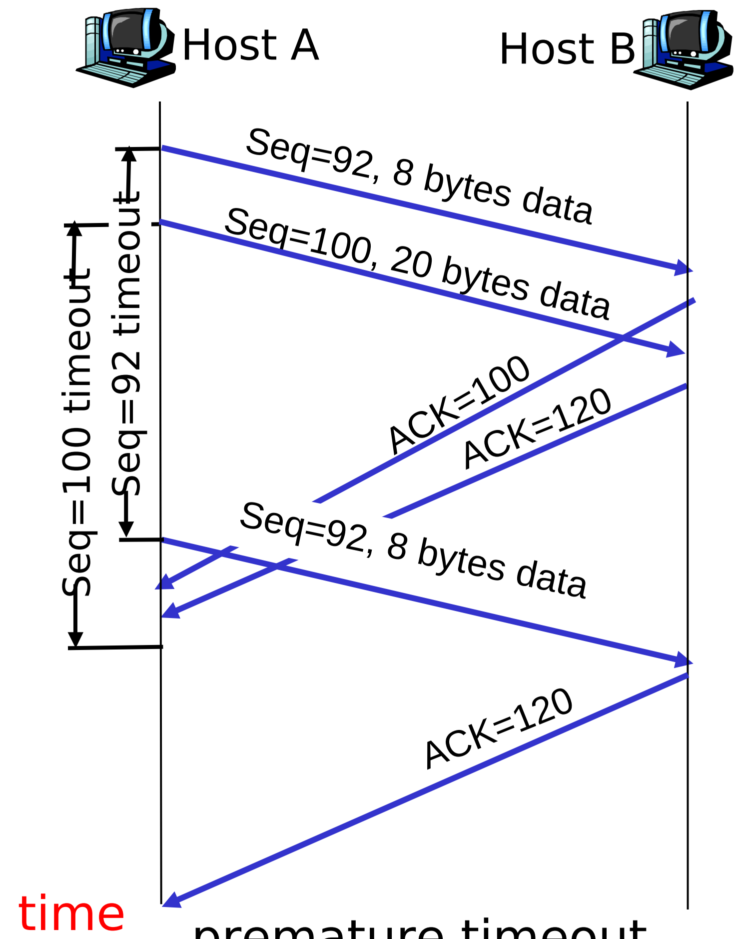
TCP ACK generation [RFC 1122, RFC 2581]

Event	TCP Receiver action
in-order segment arrival, no gaps, everything else already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
in-order segment arrival, no gaps, one delayed ACK pending	immediately send single cumulative ACK
out-of-order segment arrival higher-than-expect seq. # gap detected	send duplicate ACK, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate ACK if segment starts at lower end of gap

TCP: retransmission scenarios

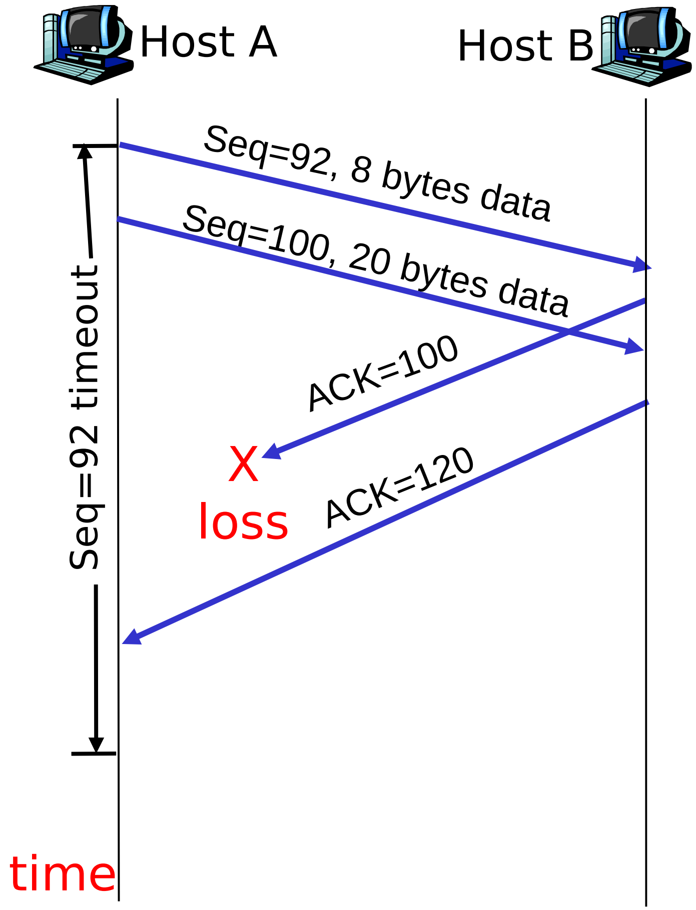


lost ACK scenario



premature timeout,
cumulative ACKs

TCP: retransmission scenarios



cumulative ACKs,
avoids retransmission of the first segment

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.

If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:

fast retransmit: resend segment before timer expires

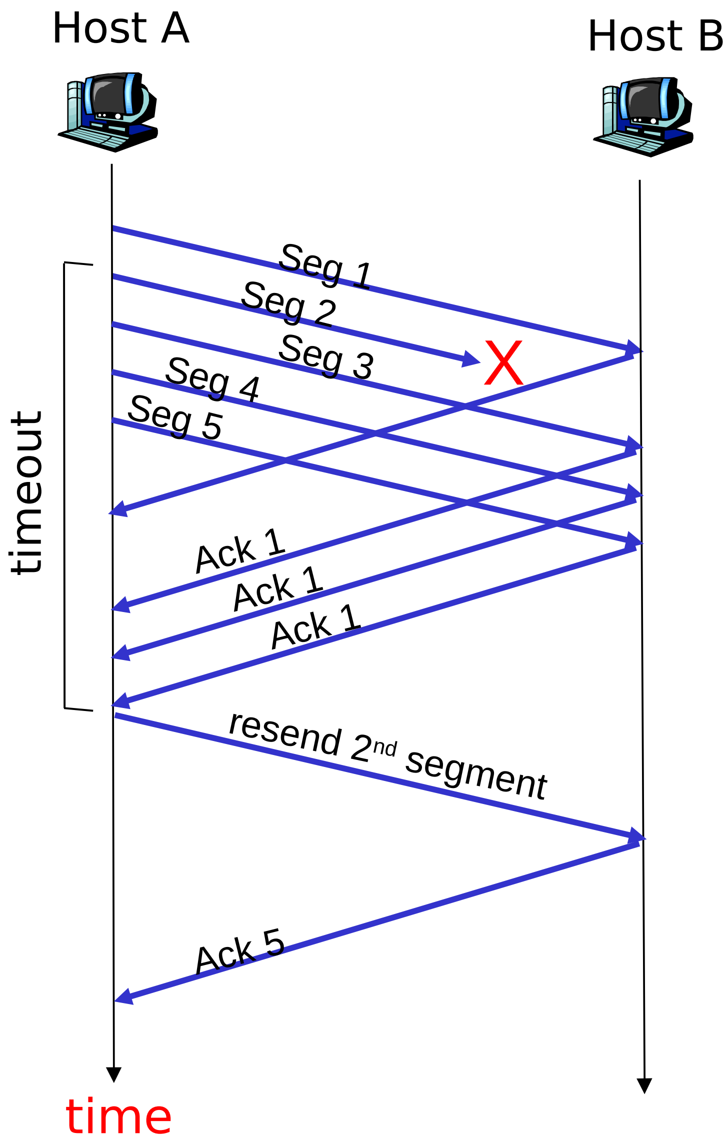


Figure 3.37 Resending a segment after triple duplicate ACK

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

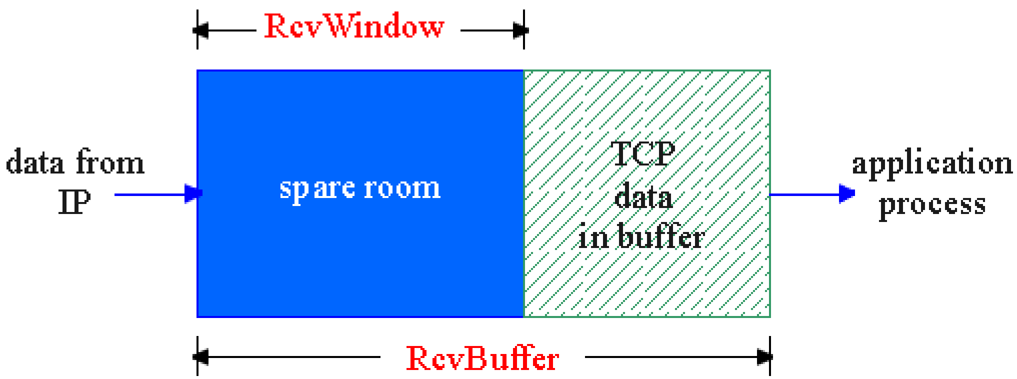
fast retransmit

TCP Flow Control

flow control: sender won't overrun receiver's buffers by transmitting too much, too fast

`RcvBuffer` = size or TCP Receive Buffer

`RcvWindow` = amount of spare room in Buffer



receiver buffering

receiver: explicitly informs sender of (dynamically changing) amount of free buffer space

- **`RcvWindow` field** in TCP segment

sender: keeps the amount of transmitted, unACKed data less than most recently received **`RcvWindow`**

question: What happens when Rcv buffer is full?

Sender keeps sending 1 byte...

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

- longer than RTT *
  - note: RTT will vary
- too short: premature timeout
  - unnecessary retransmissions
- too long: slow reaction to segment loss

* RTT = round trip time

Q: how to estimate RTT?

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

TCP Round Trip Time and Timeout

```
EstimatedRTT = (1-x)*EstimatedRTT + x*SampleRTT
```

- Exponential weighted moving average
- influence of given sample decreases exponentially fast
- typical value of x: 0.125

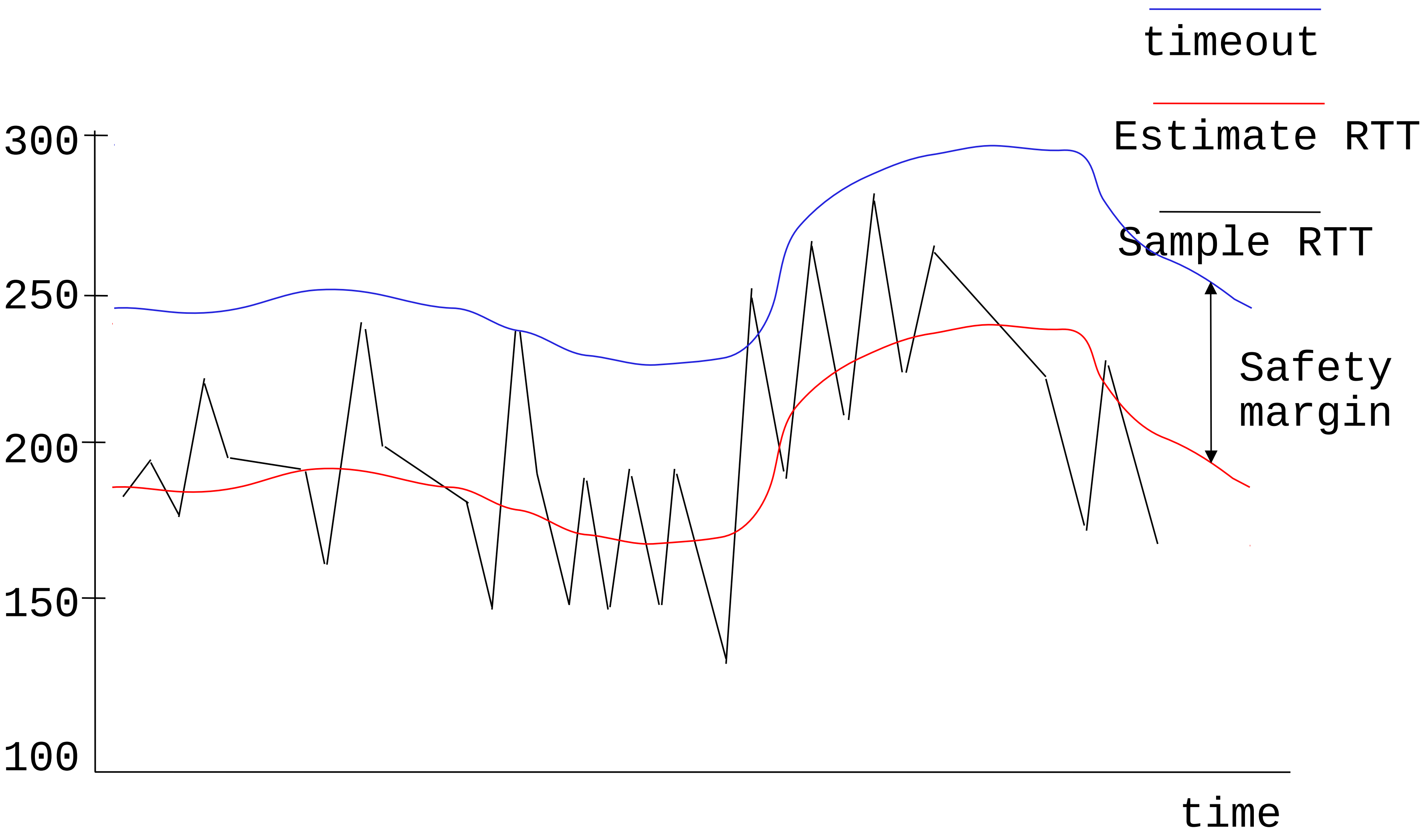
Setting the timeout

- **`EstimtedRTT`** plus "safety margin"
- large variation in **`EstimatedRTT ->`** larger safety margin

```
Timeout = EstimatedRTT + 4*Deviation

Deviation = (1-x)*Deviation +
            x*|SampleRTT-EstimatedRTT|
```

TCP Round Trip Time and Timeout



TCP Connection Management

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

- initialize TCP variables:
  - sequence numbers
  - buffers, flow control info (e.g. **`RcvWindow`**)
- *client:* connection initiator

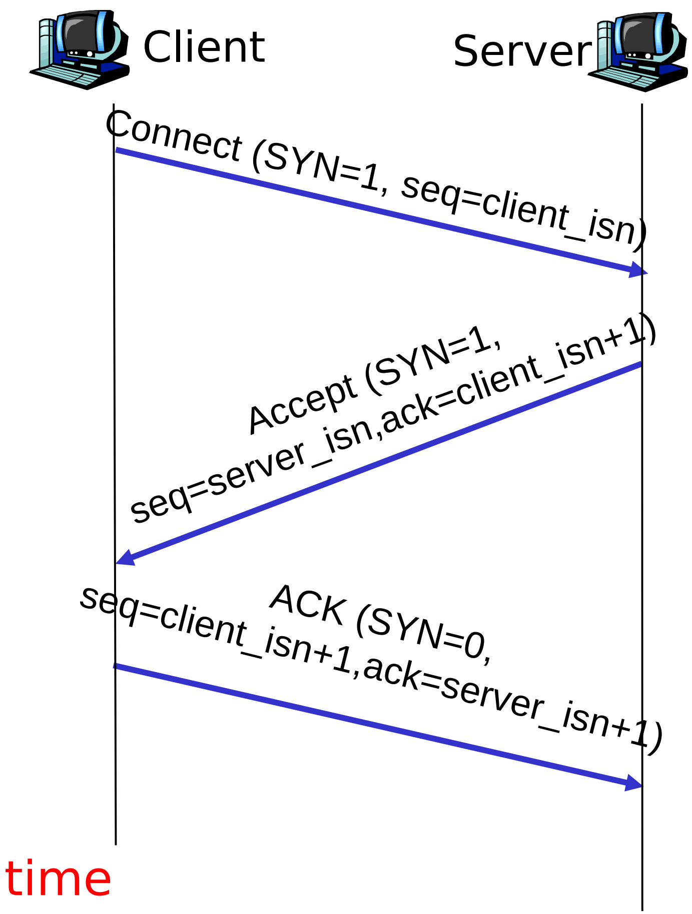
```
Socket clientSocket = new   Socket("hostname","port number");
```

```
connect;
```

- *server:* contacted by client

```
Socket accept();
```

TCP Connection Management



Three way handshake:

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

- specifies initial seq number (isn)

Step 2: server end system receives SYN, replies with SYNACK control segment

- ACKs received SYN
- allocates buffers
- specifies server-> receiver initial seq. number

Step 3: client ACK the connection:

server_isn +1 and SYN=0

Connection established!

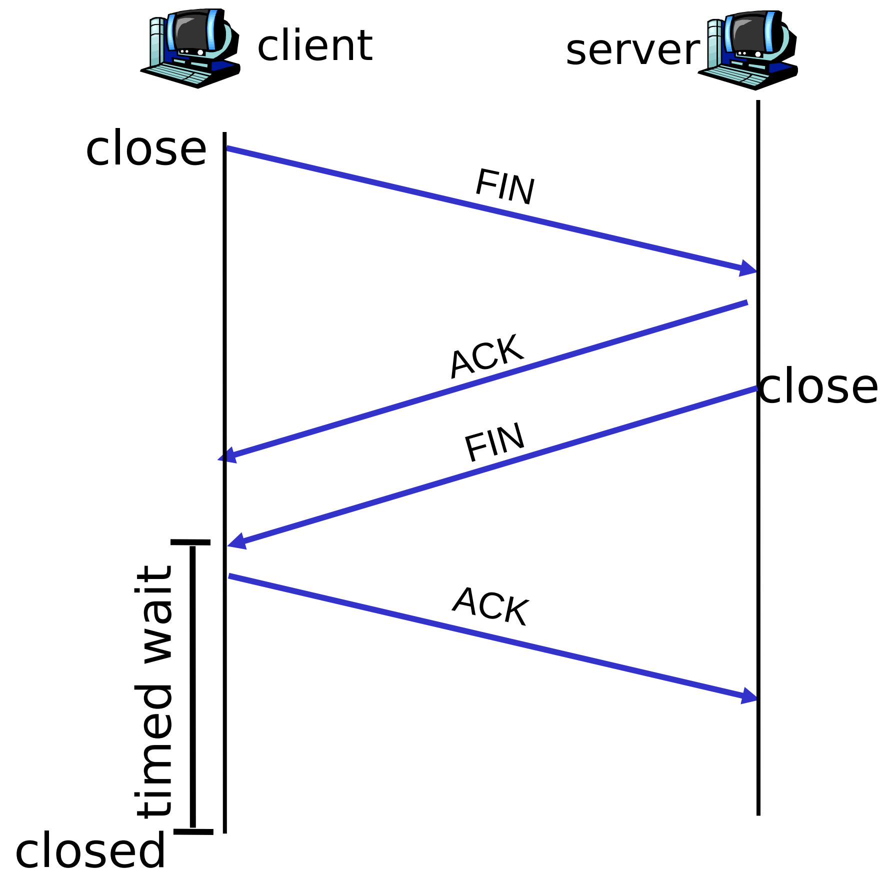
TCP Connection Management (cont.)

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.



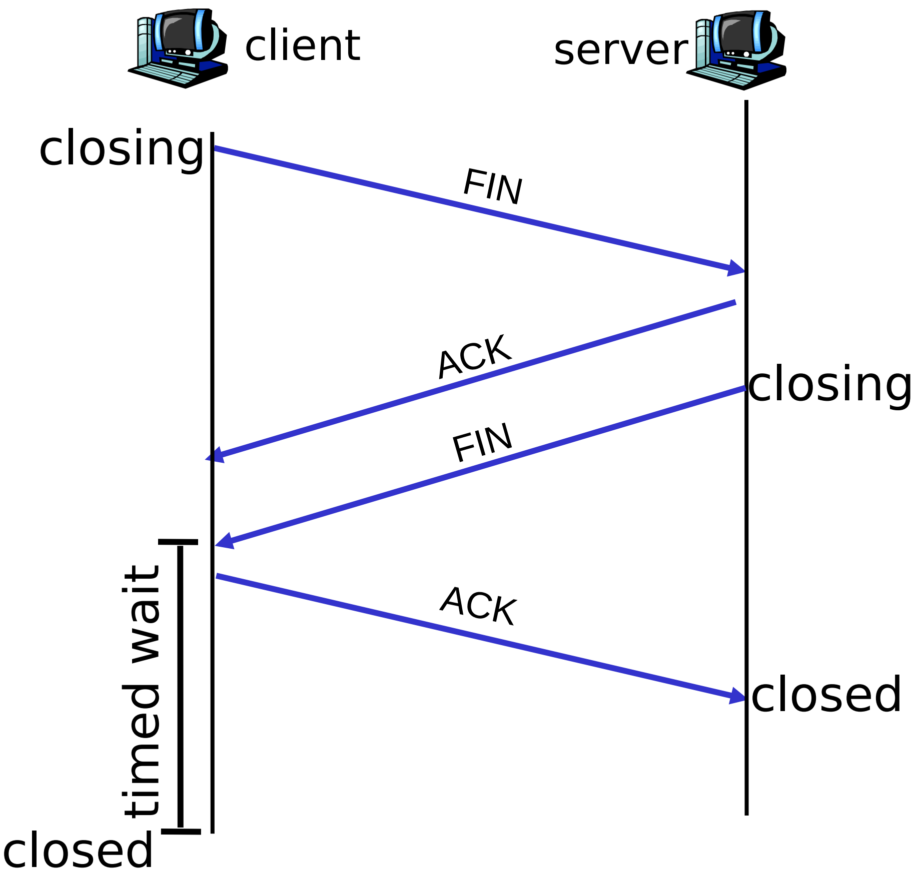
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

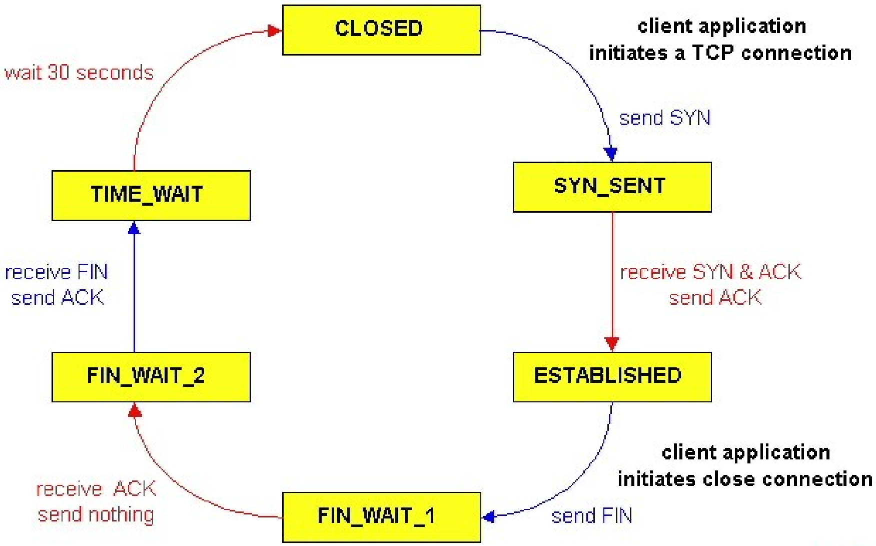
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

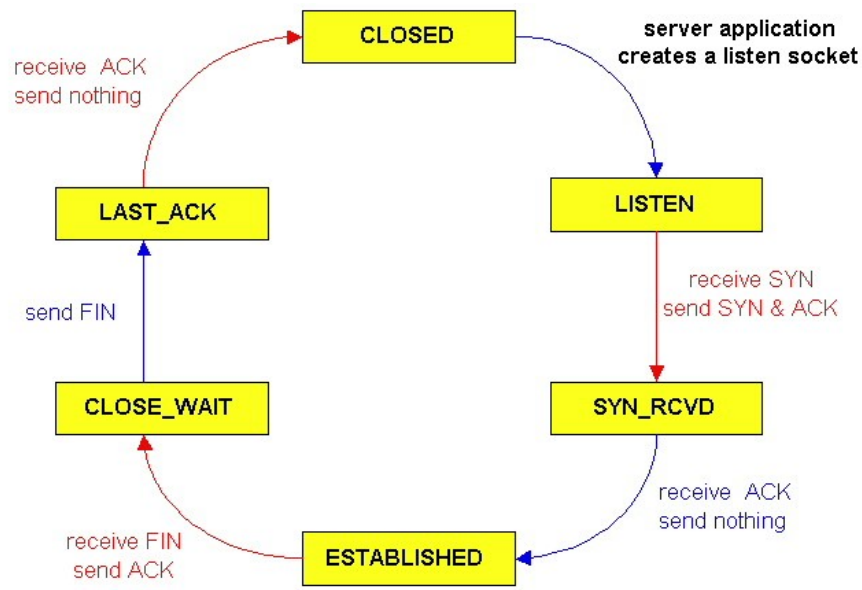


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



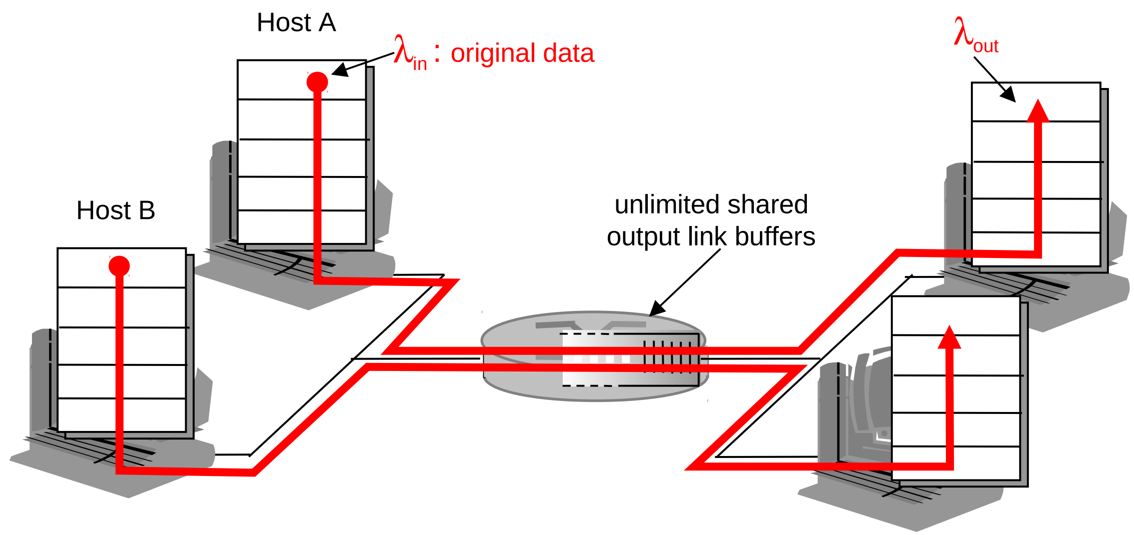
Principles of Congestion Control

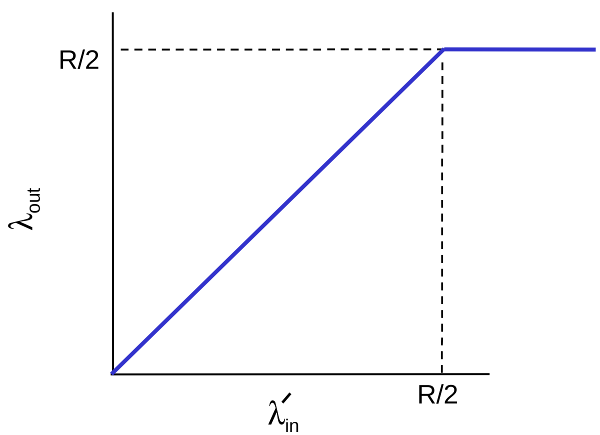
Congestion:

- 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 (queuing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

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





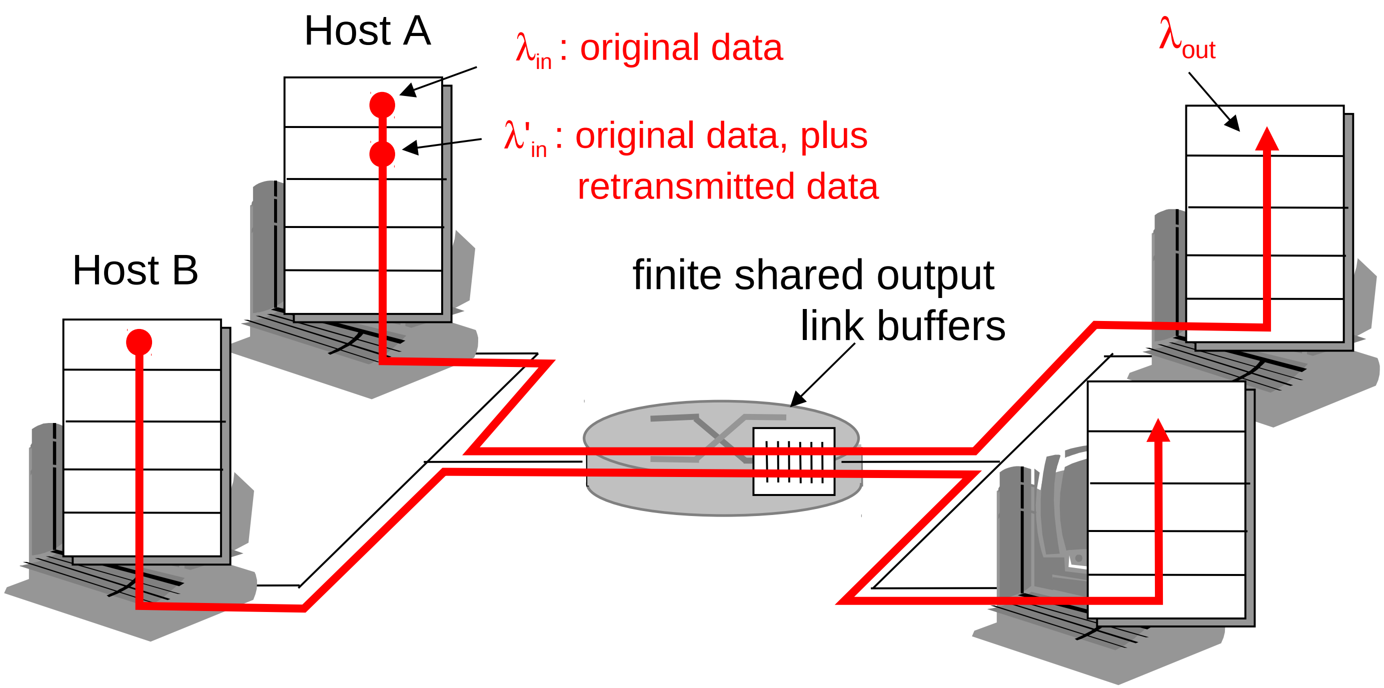
a.

- large delays when congested
- maximum achievable throughput

Causes/costs of congestion: scenario 2

one router, *finite* buffers

sender retransmission of lost packet

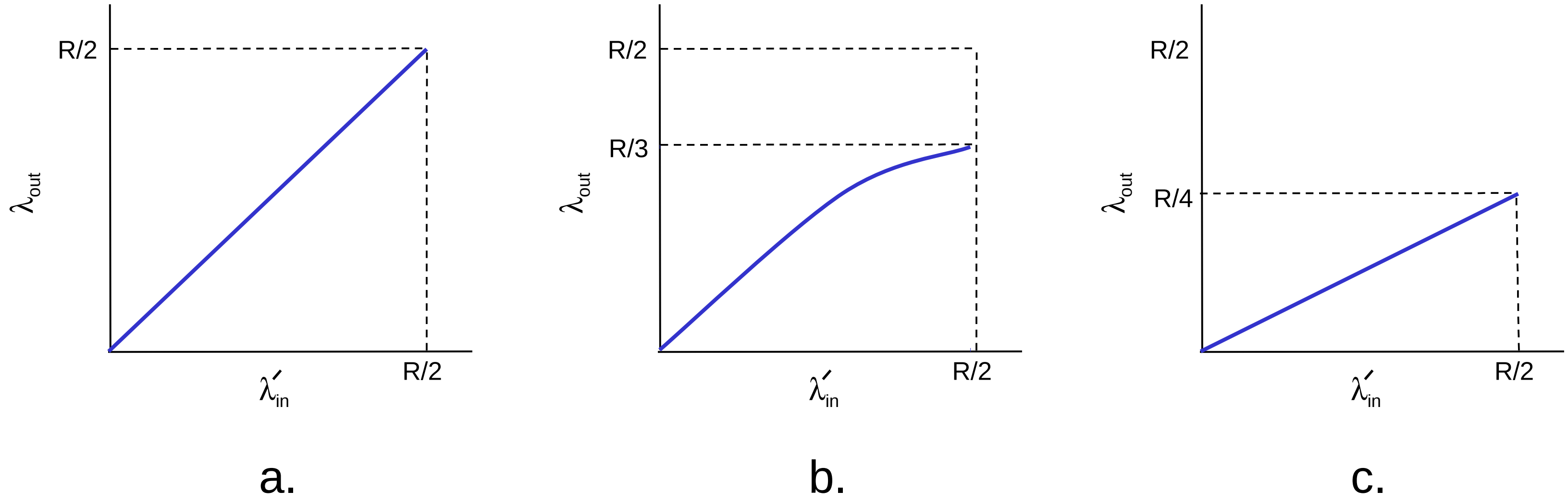


Causes/costs of congestion: scenario 2

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}



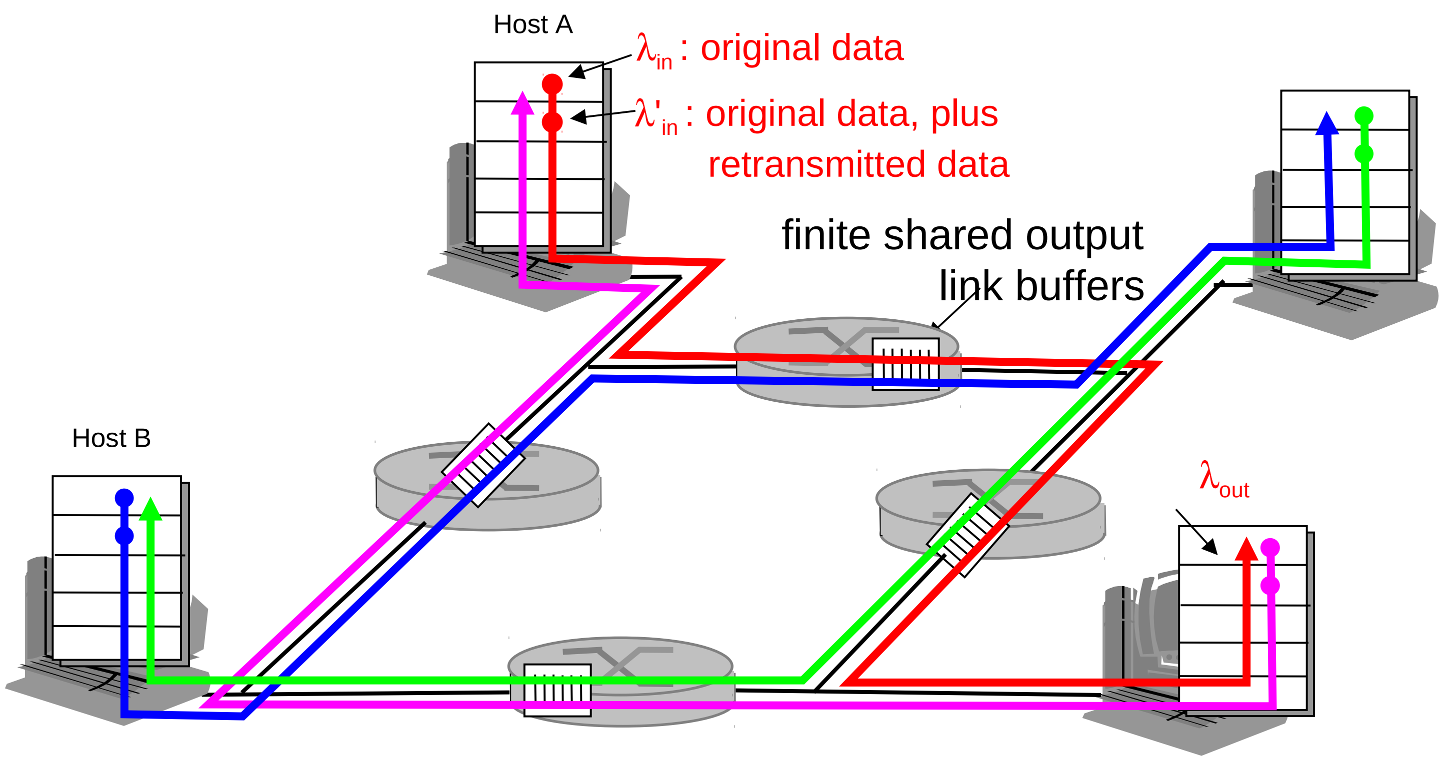
"costs" of congestion:

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

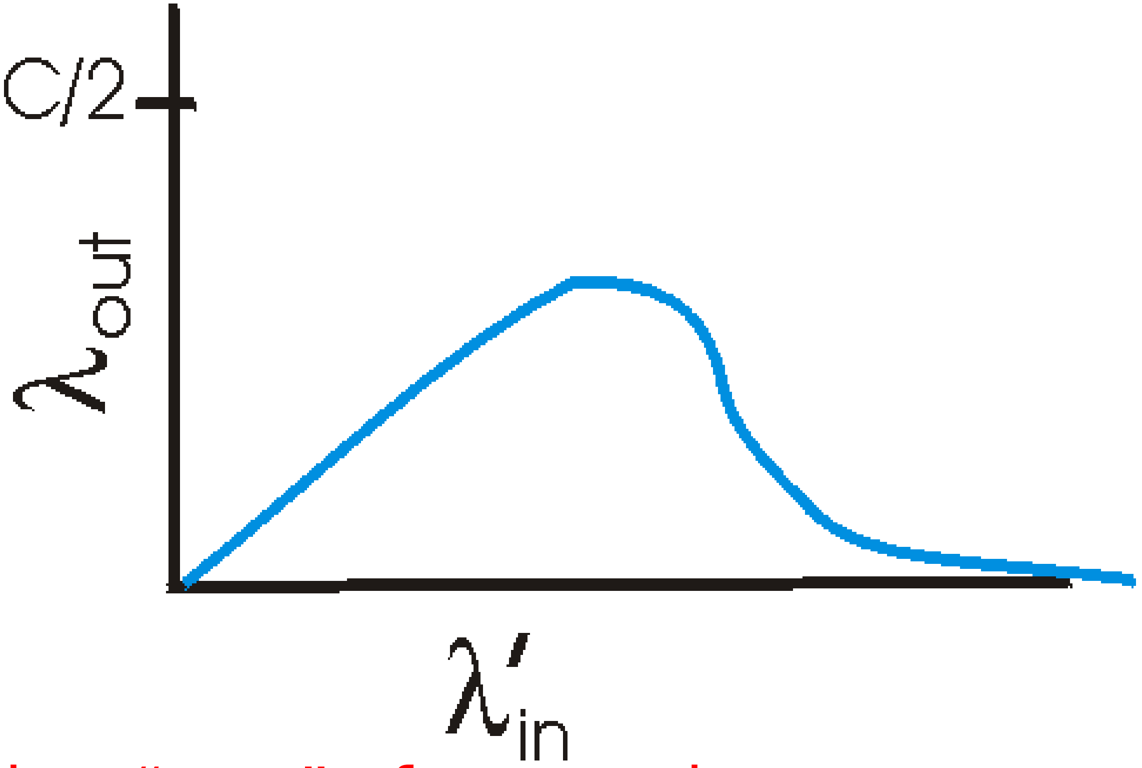
Causes/costs of congestion: scenario 3

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:

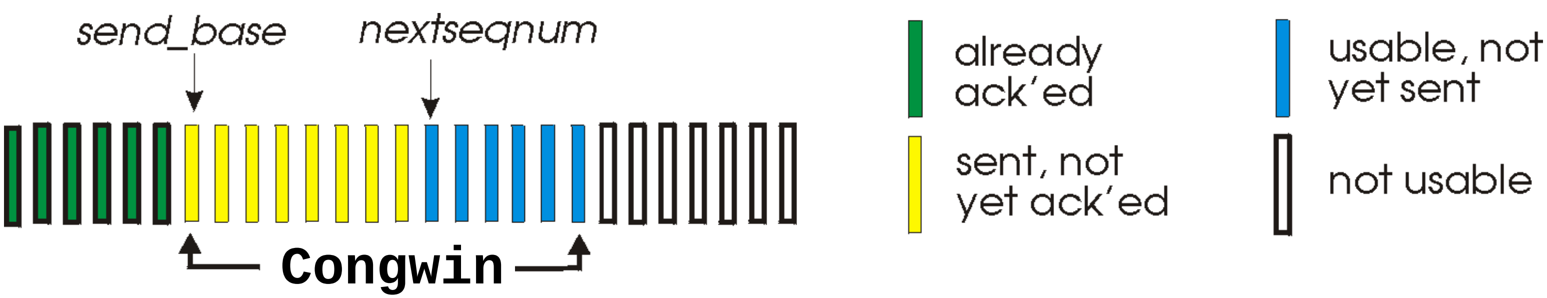
- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

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

TCP Congestion Control

- end-end control (no network assistance)
- transmission rate limited by congestion window size, **`Congwin`**, over segments:



- w segments, each with MSS bytes sent in one RTT:

$$\boxed{\text{throughput} = \frac{w * MSS}{RTT} \text{ Bytes/sec}}$$

TCP congestion control:

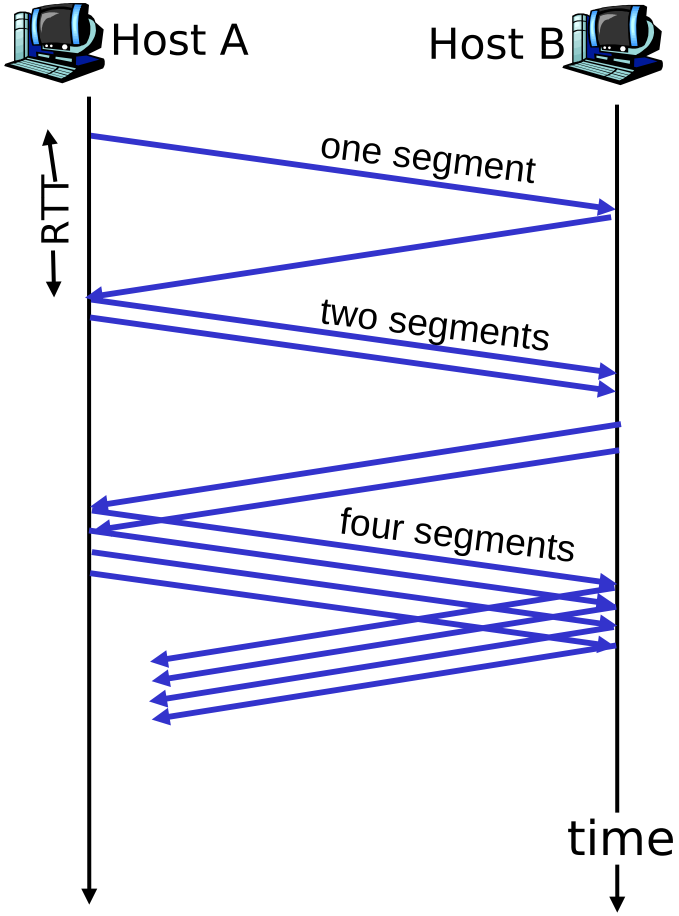
- "probing" for usable bandwidth:
  - ideally: transmit as fast as possible (**`Congwin`** as large as possible) without loss
  - *increase* **`Congwin`** until loss (congestion)
  - loss: *decrease* **`Congwin`**, then begin probing (increasing) again
- Two "phases"
  - slow start
  - congestion avoidance
- important variables:
  - **`Congwin`**
  - **`threshold:`** defines threshold between two slow start phase, congestion control phase

TCP Slowstart

Slowstart algorithm

initialize: Congwin = 1
for (each transmission completed)
Congwin=Congwin*2
until (loss event OR
CongWin > threshold)

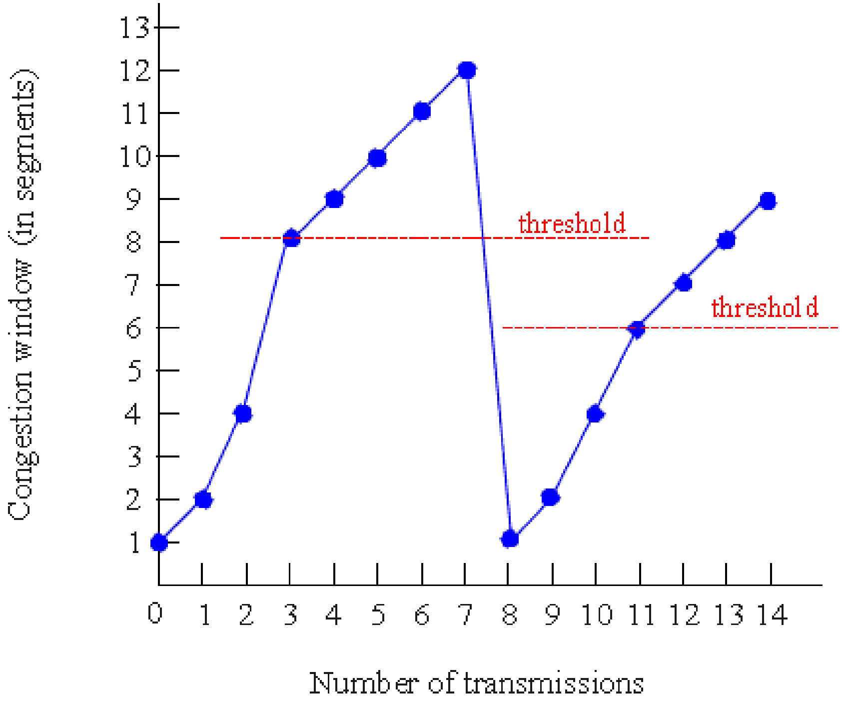
- exponential increase (per RTT) in window size (not so slow!)
- loss event: timeout and/or or three duplicate ACKs



TCP Congestion Avoidance

Congestion avoidance

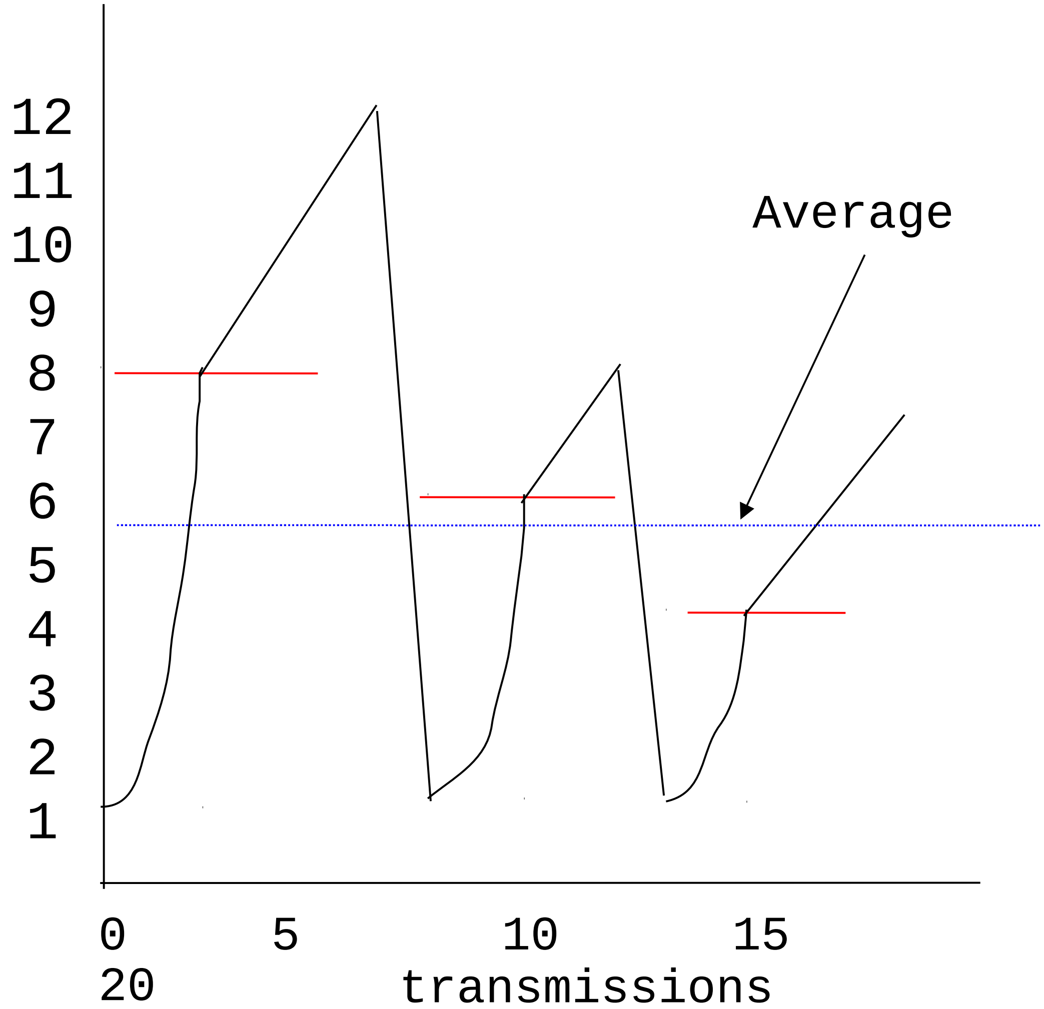
```
/* slowstart is over       */
/* Congwin > threshold */
Until (loss event) {
  every w segments ACKed:
      Congwin++
  }
threshold = Congwin/2
Congwin = 1
perform slowstart[1]
```



AIMD

TCP congestion avoidance:

- AIMD: *additive increase, multiplicative decrease*
  - increase window by 1 per RTT
  - decrease threashold by factor of 2 on loss event



Refinement: inferring loss

After 3 dup ACKs:

`CongWin` is cut in half

window then grows linearly

<u>But</u> after timeout event:

`CongWin` instead set to 1 MSS;

window then grows exponentially

to a threshold, then grows linearly

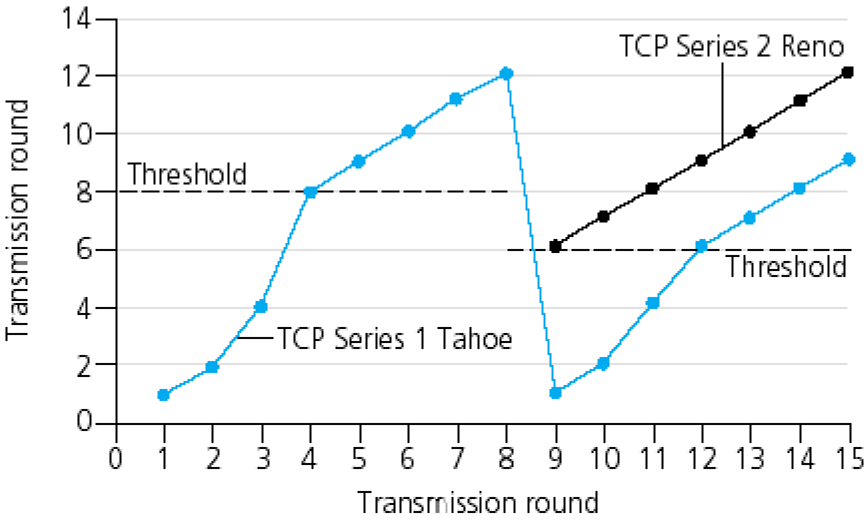
Philosophy:

- 3 dup ACKs indicates network capable of delivering some segments
- timeout indicates a "more alarming" congestion scenario

Refinement

Q: When should the exponential increase switch to linear?

A: When **`CongWin`** gets to 1/2 of its value before timeout.



Implementation:

Variable Threshold

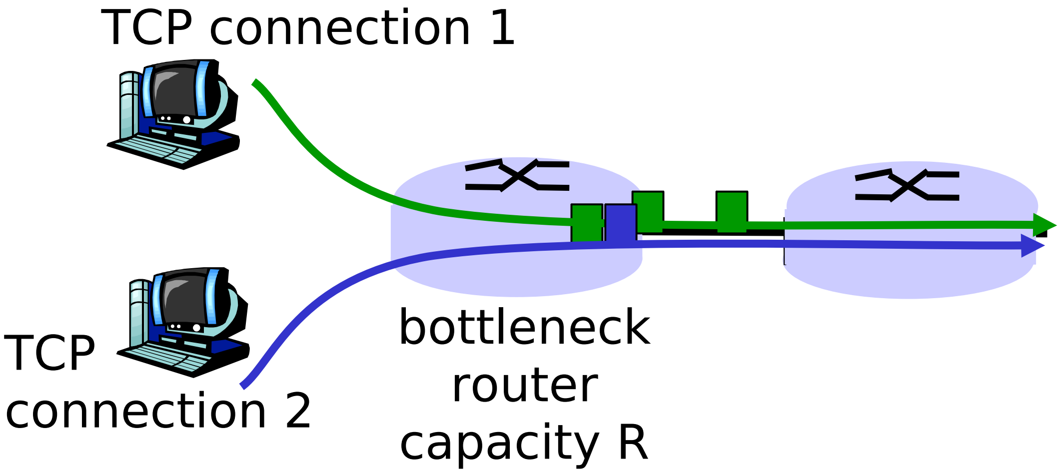
At loss event, Threshold is set to 1/2 of CongWin just before loss event

TCP sender congestion control

State	Event	TCP Sender Action	Commentary
Slow Start (SS)	ACK receipt for previously unacked data	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to “Congestion Avoidance”	Resulting in a doubling of CongWin every RTT
Congestion Avoidance (CA)	ACK receipt for previously unacked data	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
SS or CA	Loss event detected by triple duplicate ACK	Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance”	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
SS or CA	Timeout	Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start”	Enter slow start
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP Fairness

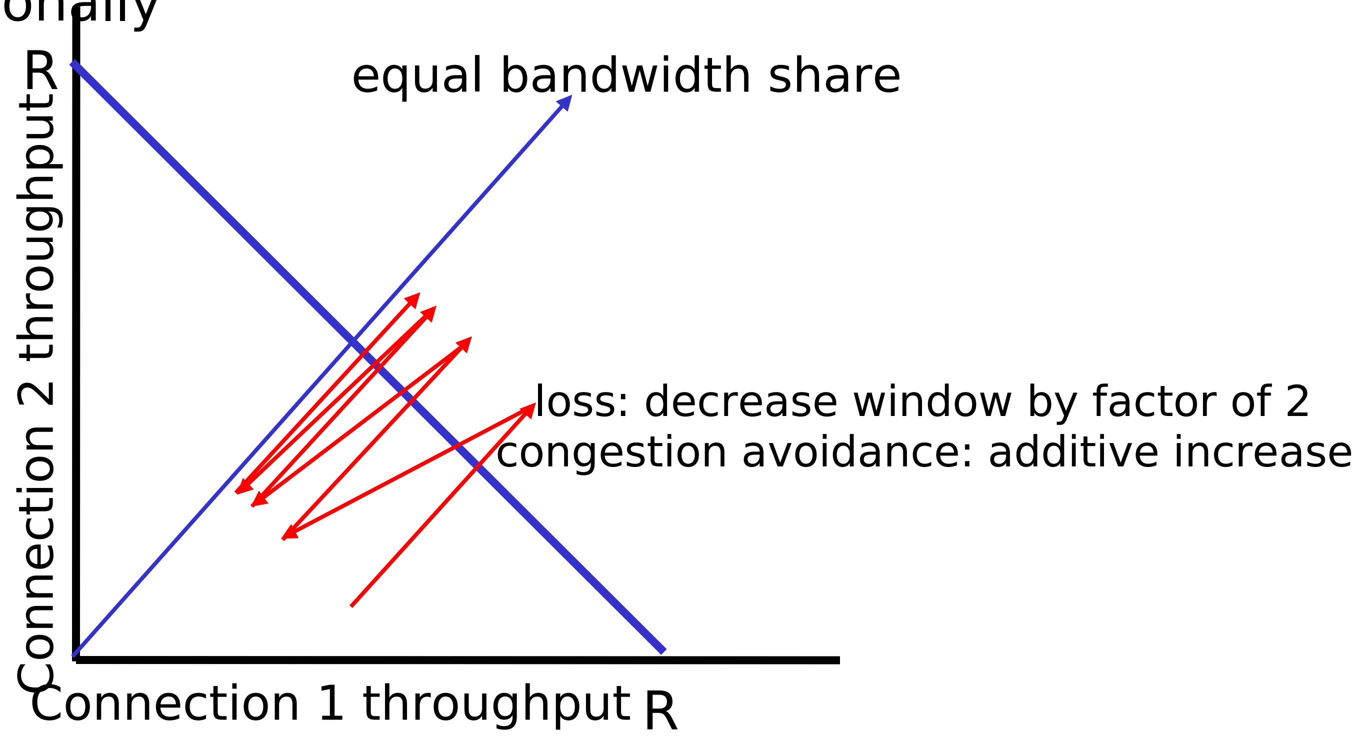
Fairness goal: if N TCP sessions share same bottleneck link, each should get 1/N of link capacity



Why is TCP fair?

Two competing sessions:

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



TCP latency modeling

Q: How long does it take to receive an object from a Web server after sending a request?

- TCP connection establishment
- data transfer delay

Notation, assumptions:

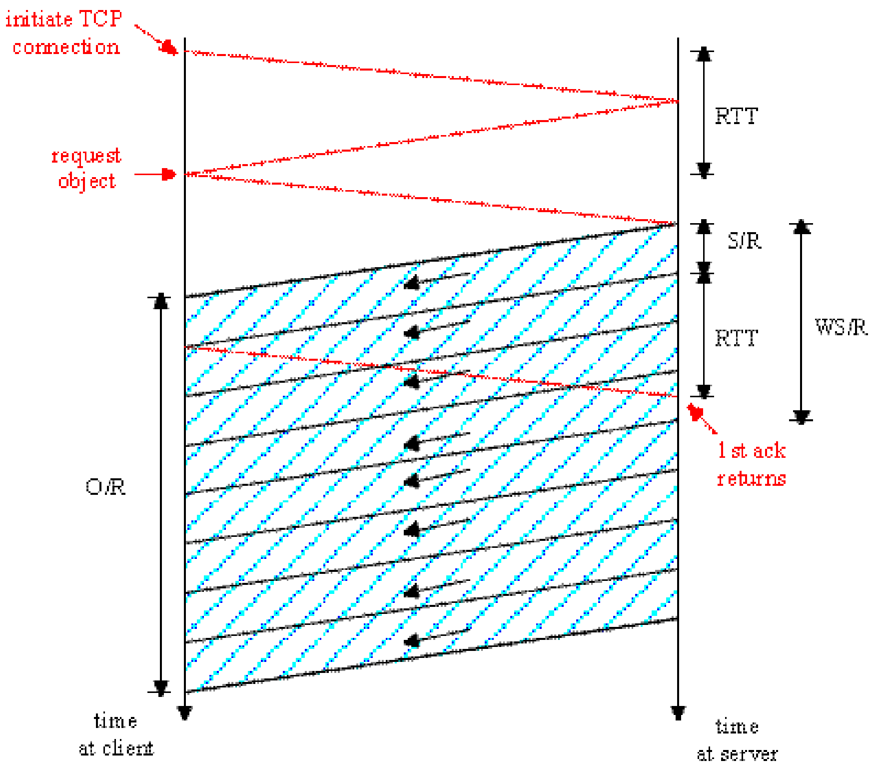
- Assume one link between client and server of rate R
- Assume: fixed congestion window, W segments
- S: MSS (bits)
- O: object size (bits)
- no retransmissions (no loss, no corruption)

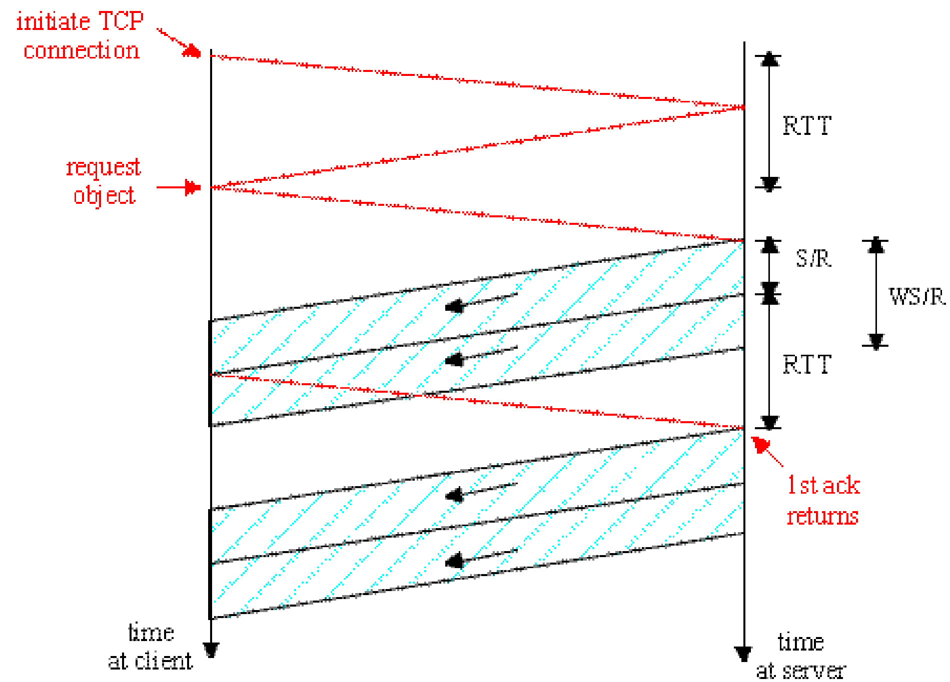
Two cases to consider:

- WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent
- WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

TCP latency Modeling

K:= O/WS





Case 1: latency = 2RTT + O/R

Case 2: latency = 2RTT + O/R + (K-1)[S/R + RTT - WS/R]