CSCI 6760/4760 Computer Networks: Topology and applications

Chapter 3

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Chapter 3: Transport Layer

our goals:

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

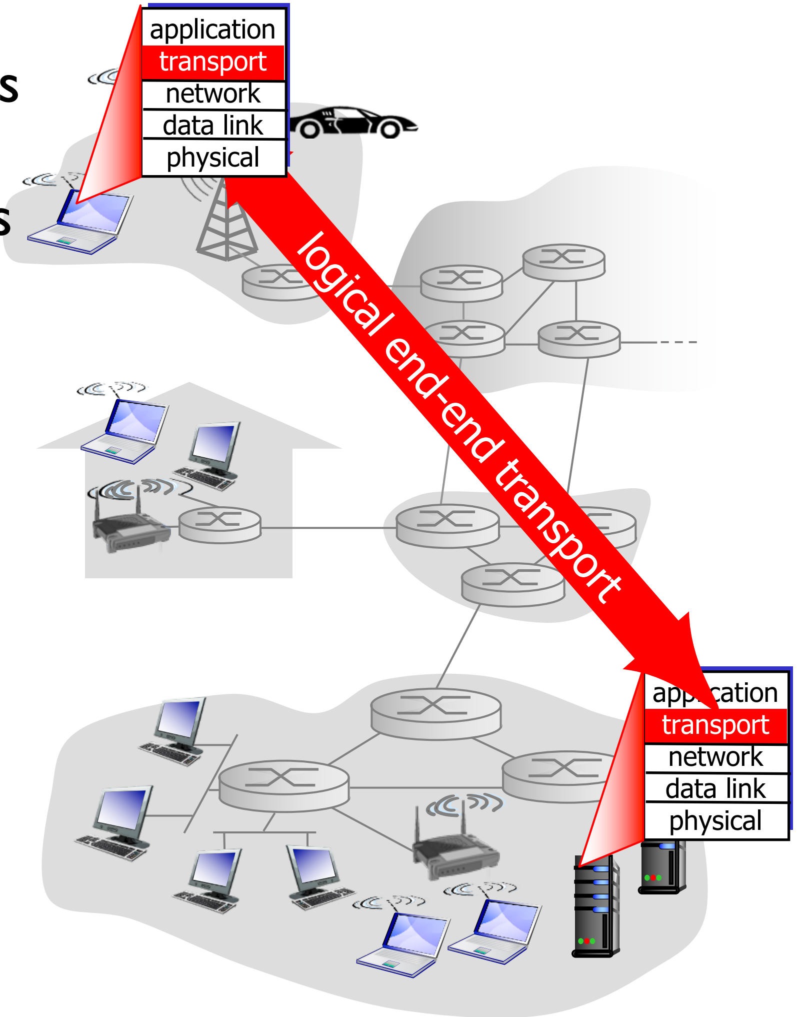
- segment structure
- reliable data transfer
- flow control
- connection management

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
- By *logical communication*, we mean that from an application's perspective, it is as if the hosts running the processes were directly connected;
- in reality, the hosts may be on opposite sides of the planet, 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

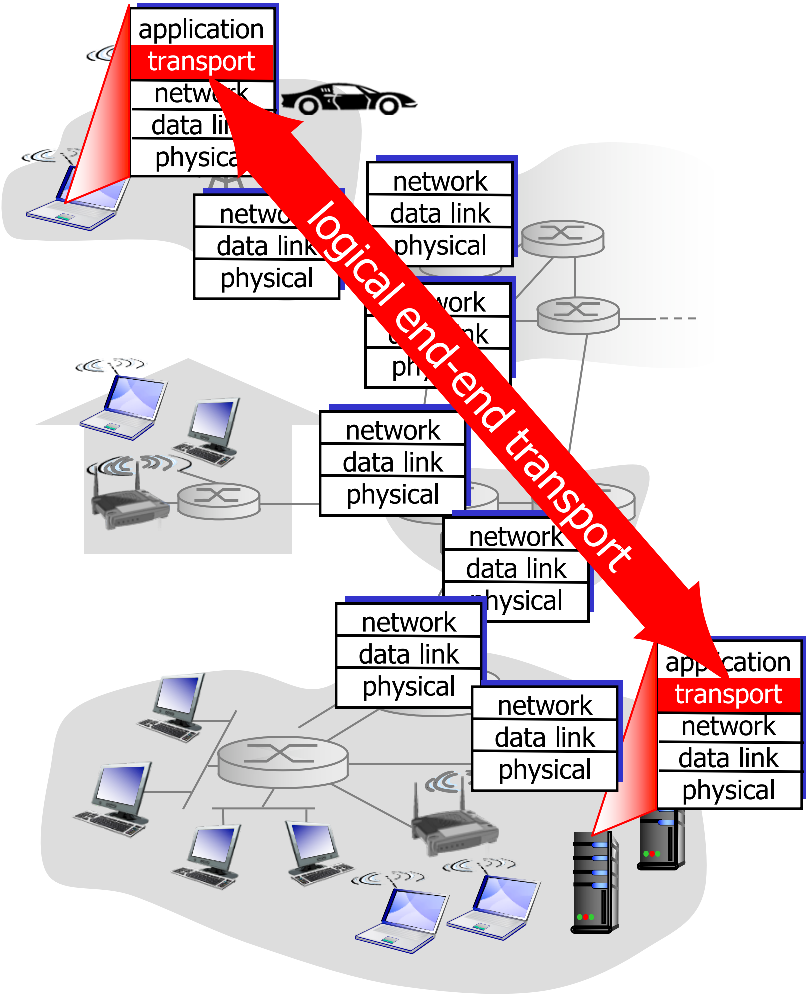
household analogy:

12 kids in Ann's house sending letters to 12 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- 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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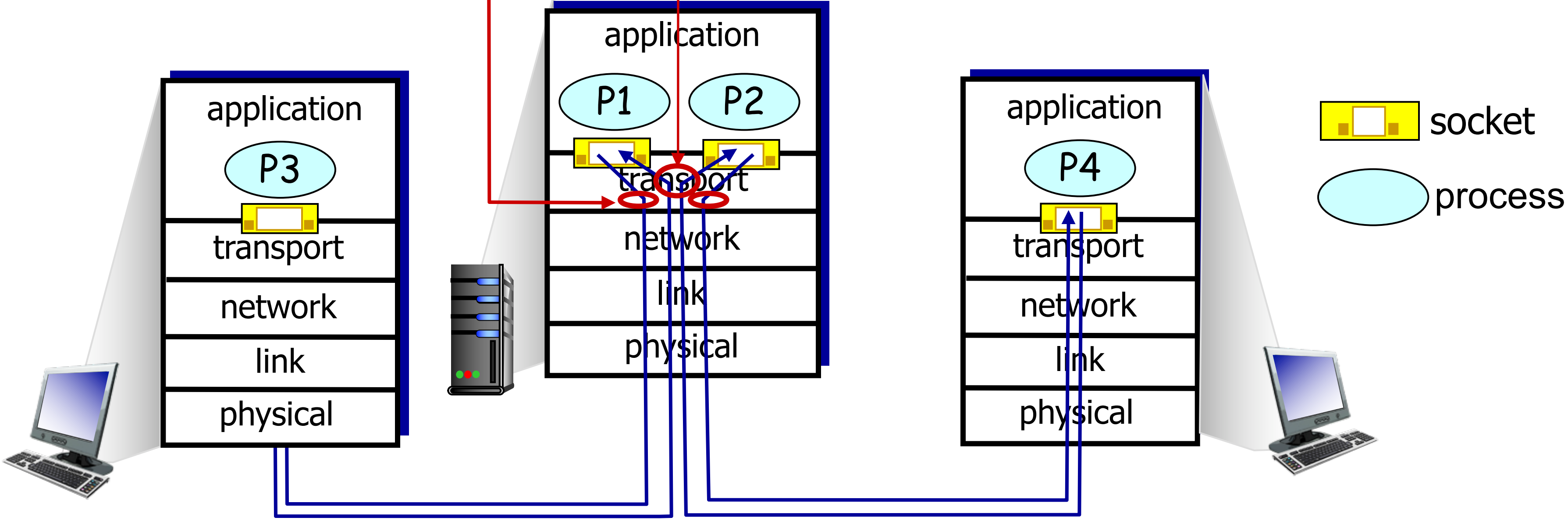
Multiplexing/demultiplexing

multiplexing at sender:

handle data from multiple sockets, add transport header (later used for demultiplexing)

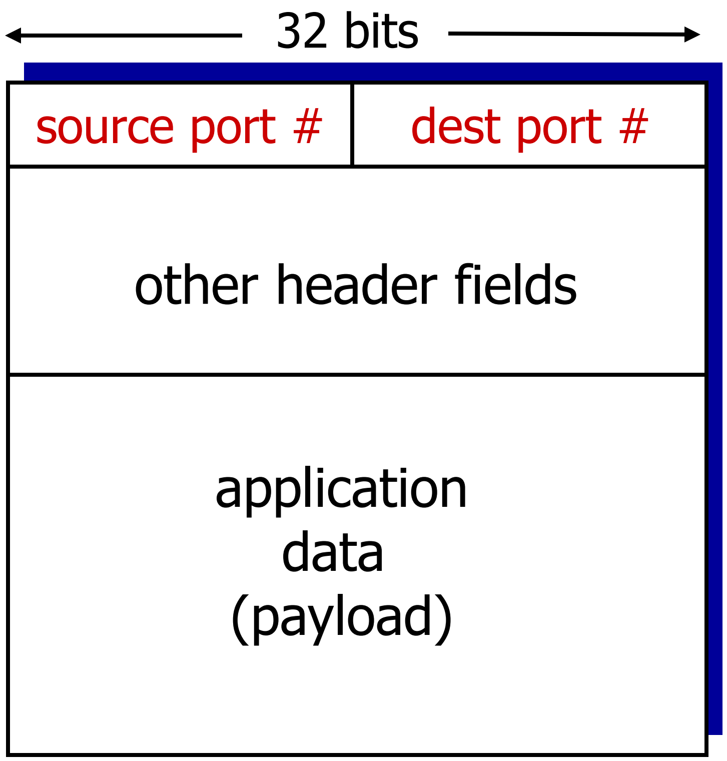
demultiplexing at receiver:

use header info to deliver received segments to correct socket



How demultiplexing works

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



32 bits

source port #

dest port #

other header fields

application data (payload)

TCP/UDP segment format

Connectionless demultiplexing

- created socket has host-local port #:

- when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

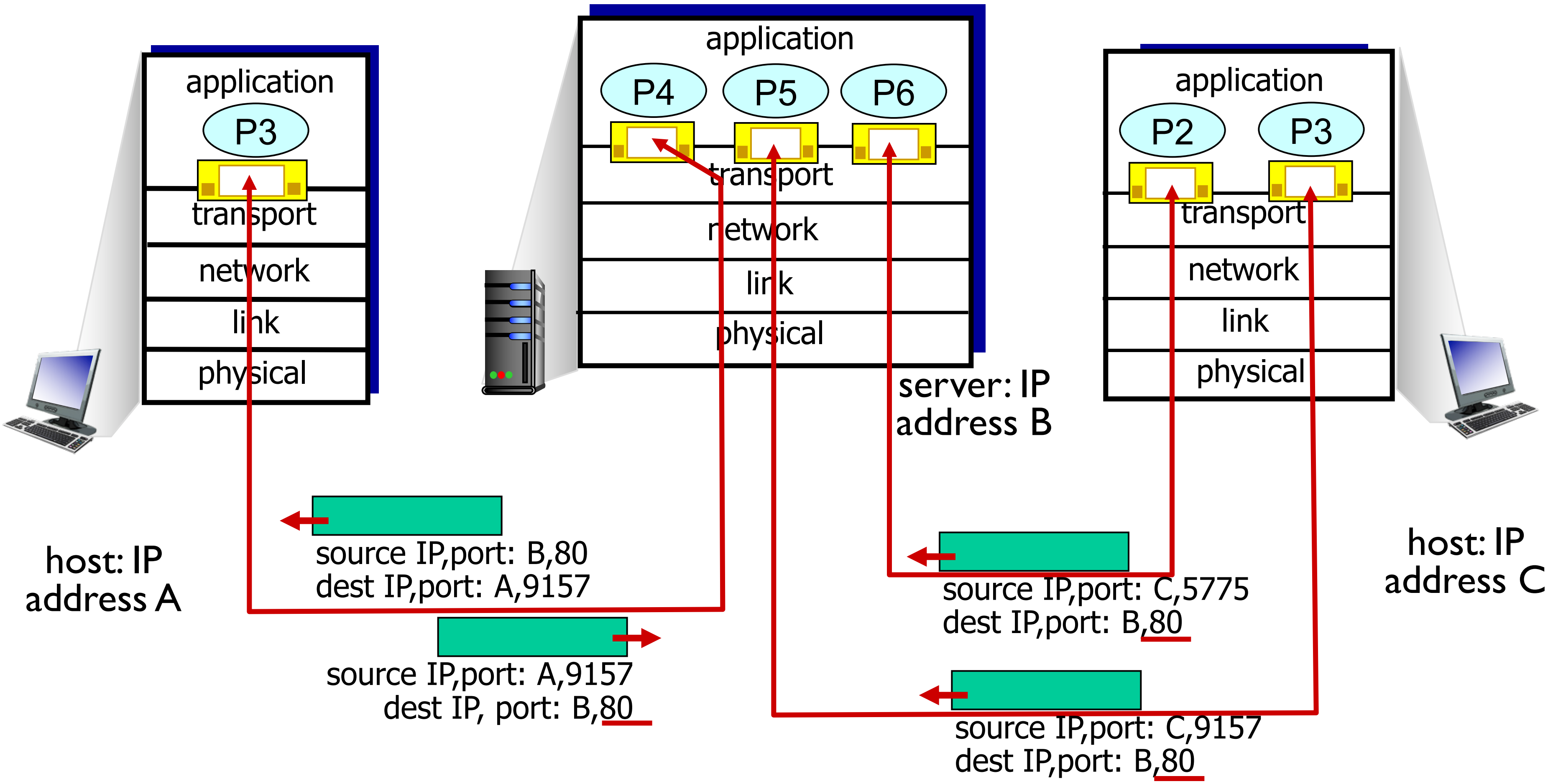
→ IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connection-oriented demux

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

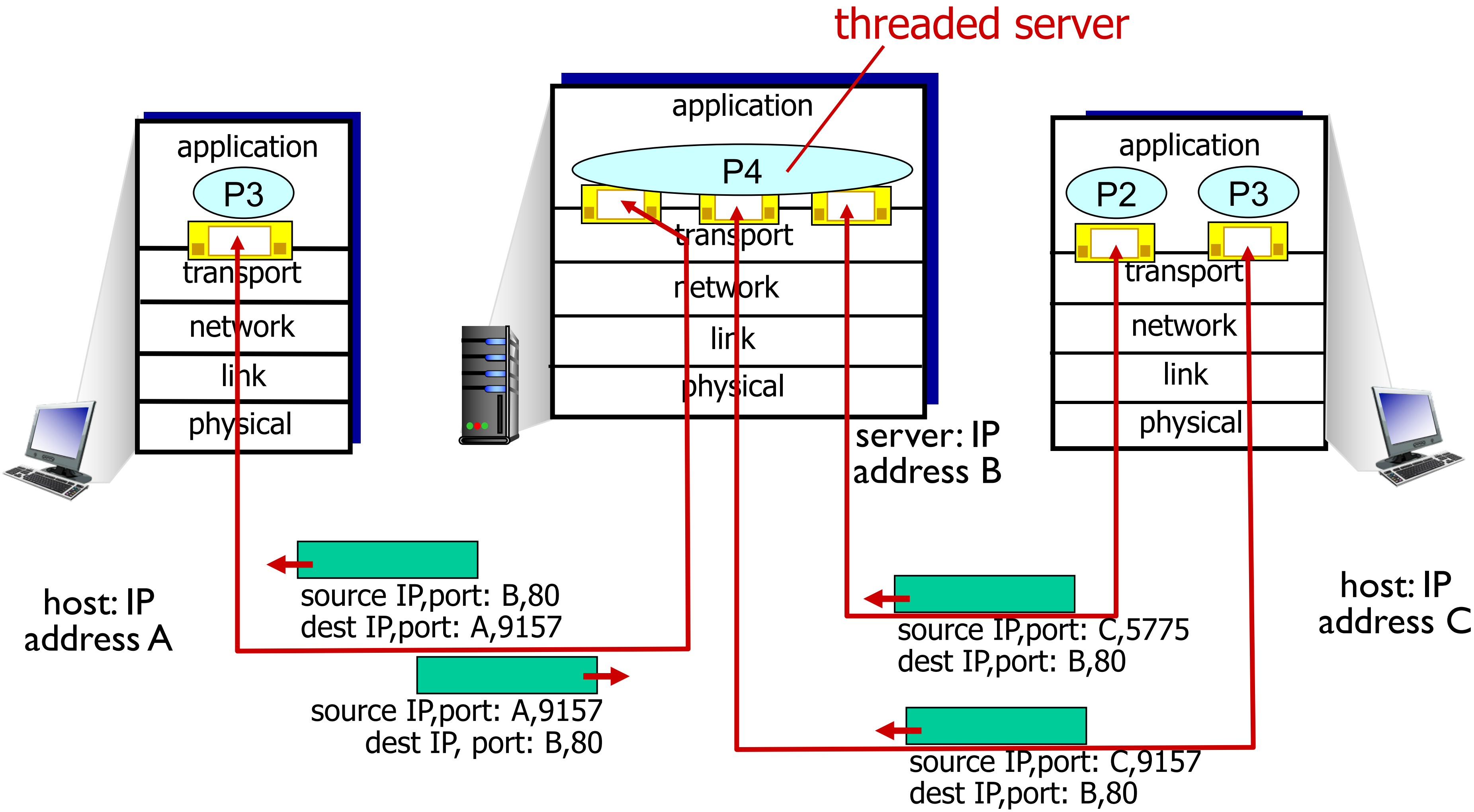
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



three segments, all destined to IP address: B, dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



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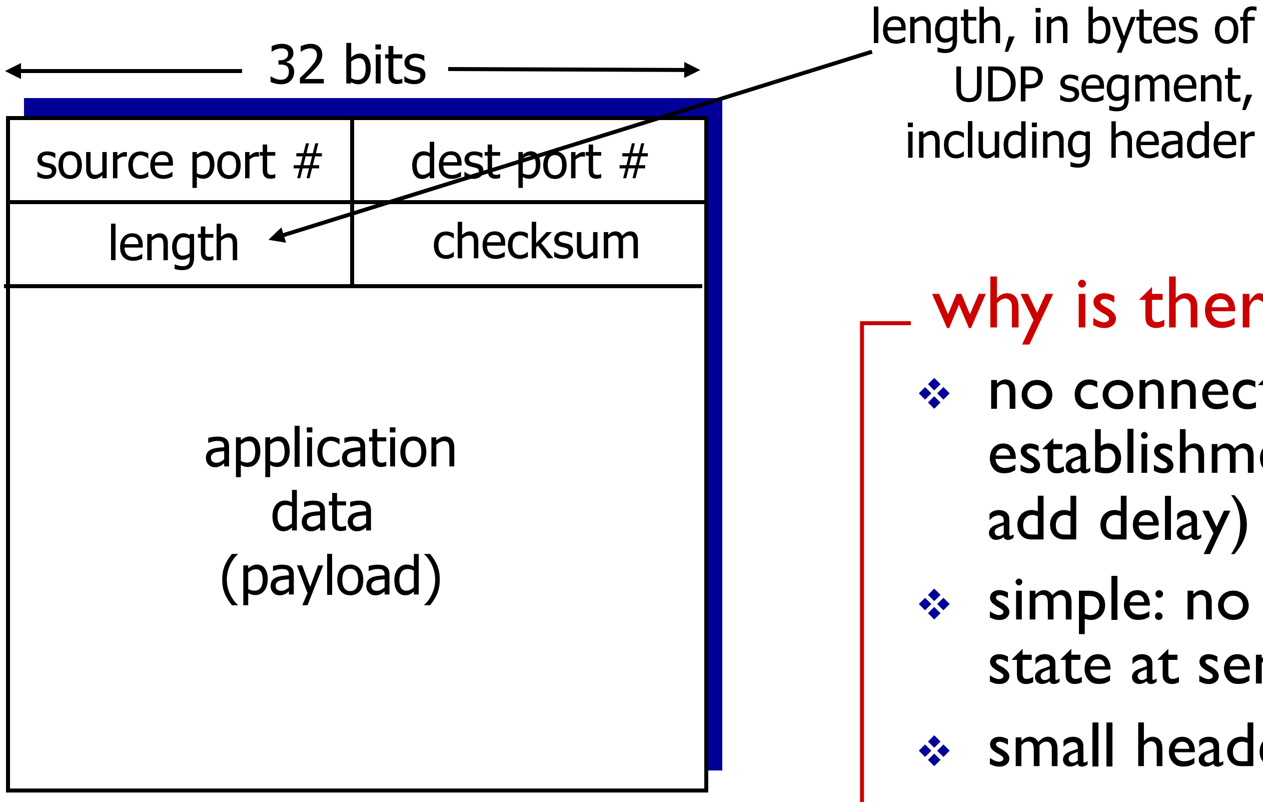
UDP: User Datagram Protocol [RFC 768]

- "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

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

UDP: segment header



UDP segment format

why is there a UDP?

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

why is there a UDP?

- Finer control over what data is sent and when
  - As soon as an application process writes into the socket
  - … UDP will package the data and send the packet
- no delay for connection establishment
- no connection state at sender, receiver
  - No allocation of buffers, parameters, sequence #s, etc.
  - … making it easier to handle many active clients at once
- small header size

Popular Applications That Use UDP

- Multimedia streaming
  - By the time the packet is retransmitted, it's too late
  - E.g., telephone calls, video conferencing, gaming
- Simple query protocols like Domain Name System

UDP checksum

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

sender:

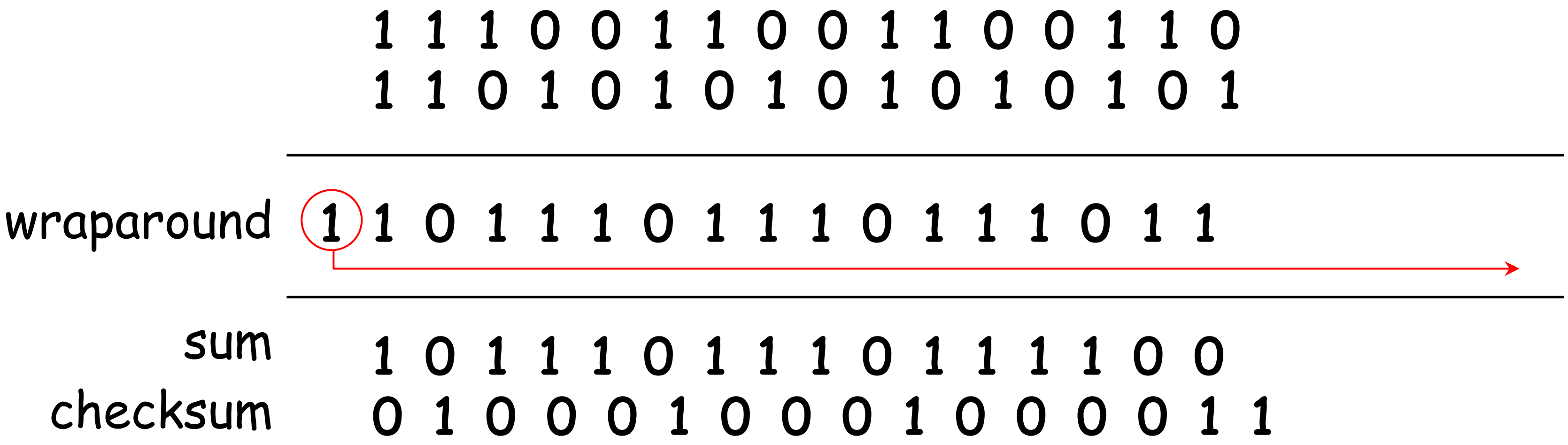
- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one’s complement sum) of segment contents
- sender puts checksum value into UDP checksum field

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

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
```

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

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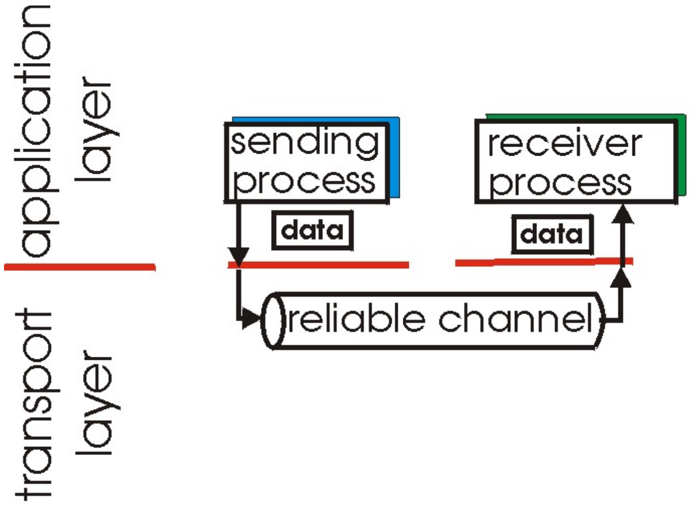
Principles of reliable data transfer

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



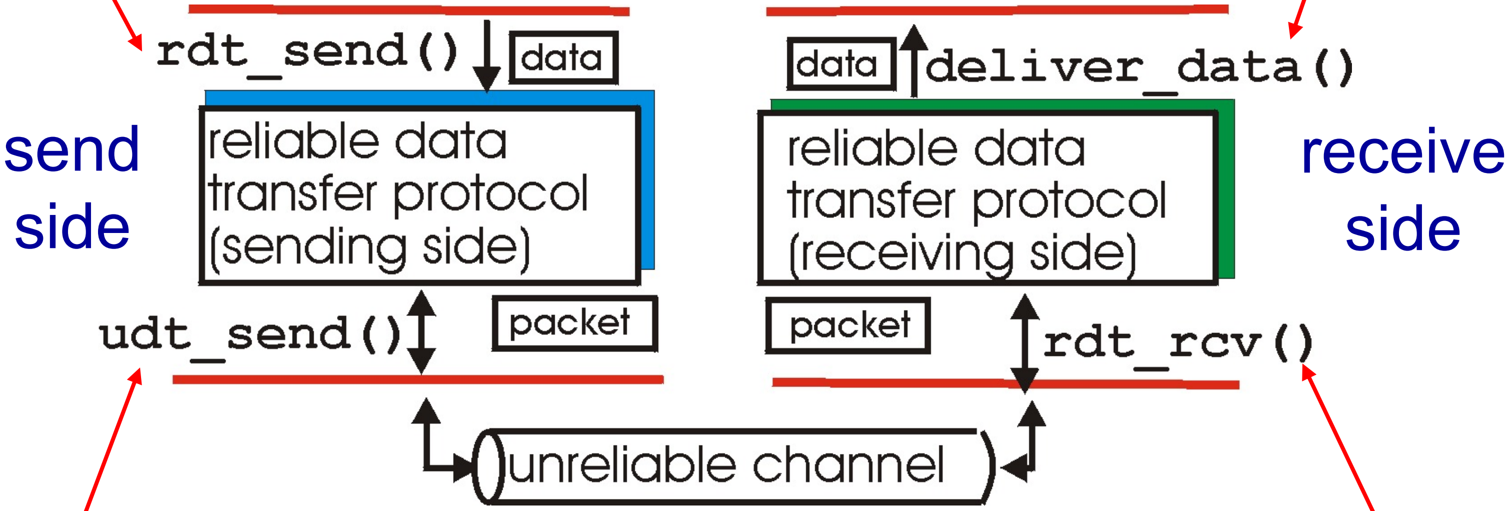
(a) provided service

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

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

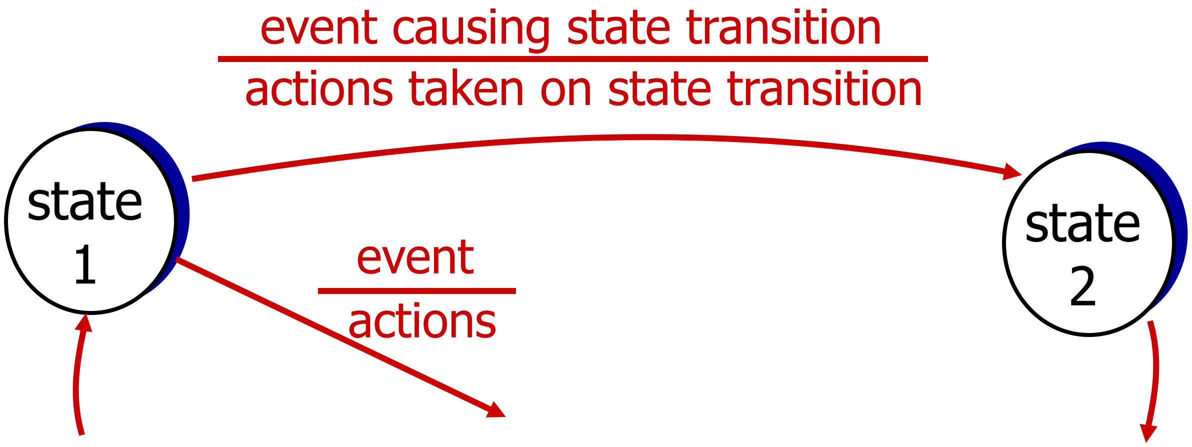
rdt_rcv(): called when packet arrives on rcv-side of channel

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



event causing state transition

actions taken on state transition

state 1

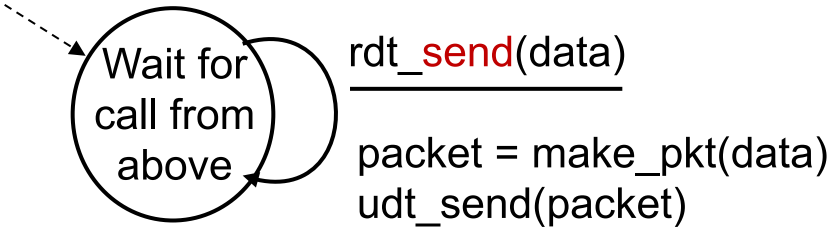
event

actions

state 2

rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors (no flipped from 0 to 1, or vice versa)
  - no loss of packets
  - All the packets are delivered in the order that were sent
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel

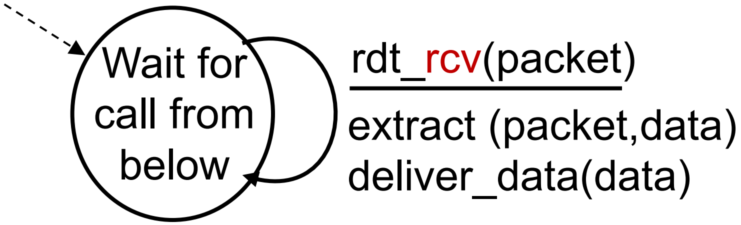


Wait for call from above

rdt_send(data)

packet = make_pkt(data)
udt_send(packet)

sender



Wait for call from below

rdt_rcv(packet)

extract (packet,data)
deliver_data(data)

receiver

rdt2.0: channel with bit errors

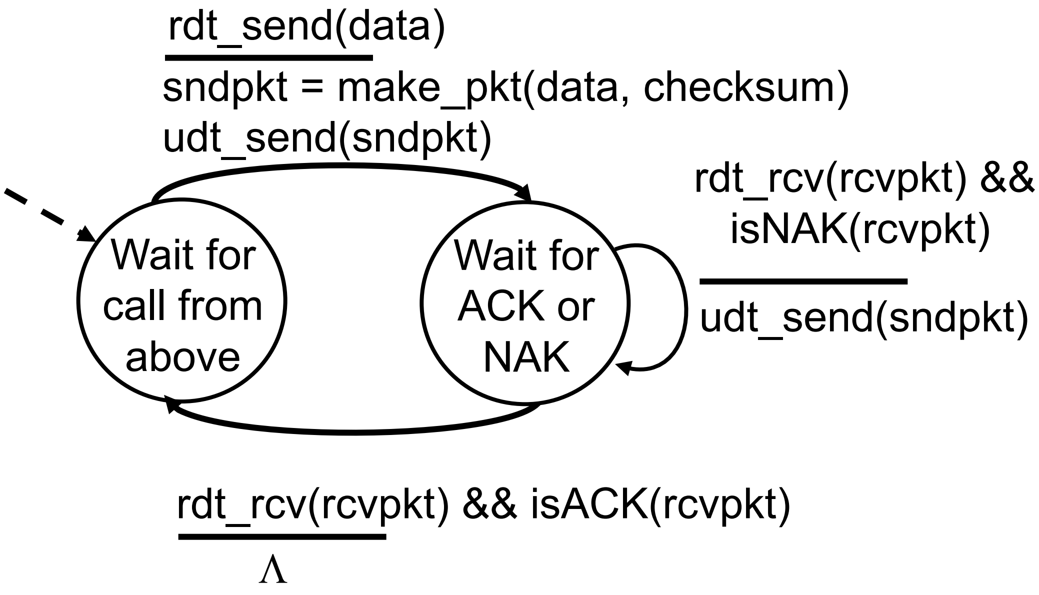
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

How do humans recover from "errors" during conversation?

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
  - feedback: control msgs (ACK,NAK) from receiver to sender

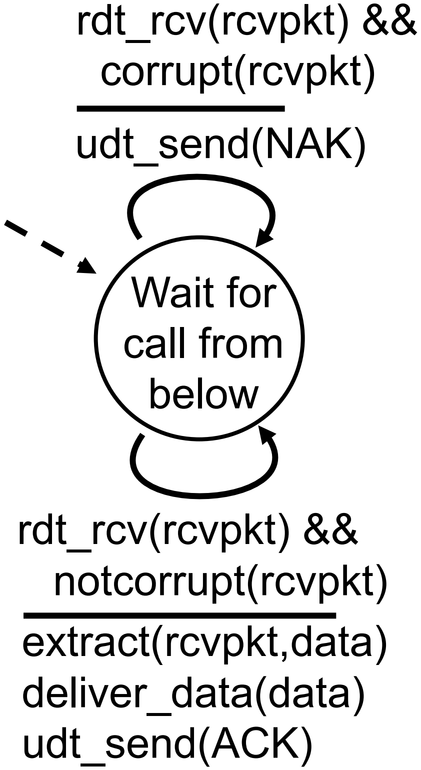
rdt2.0: FSM specification



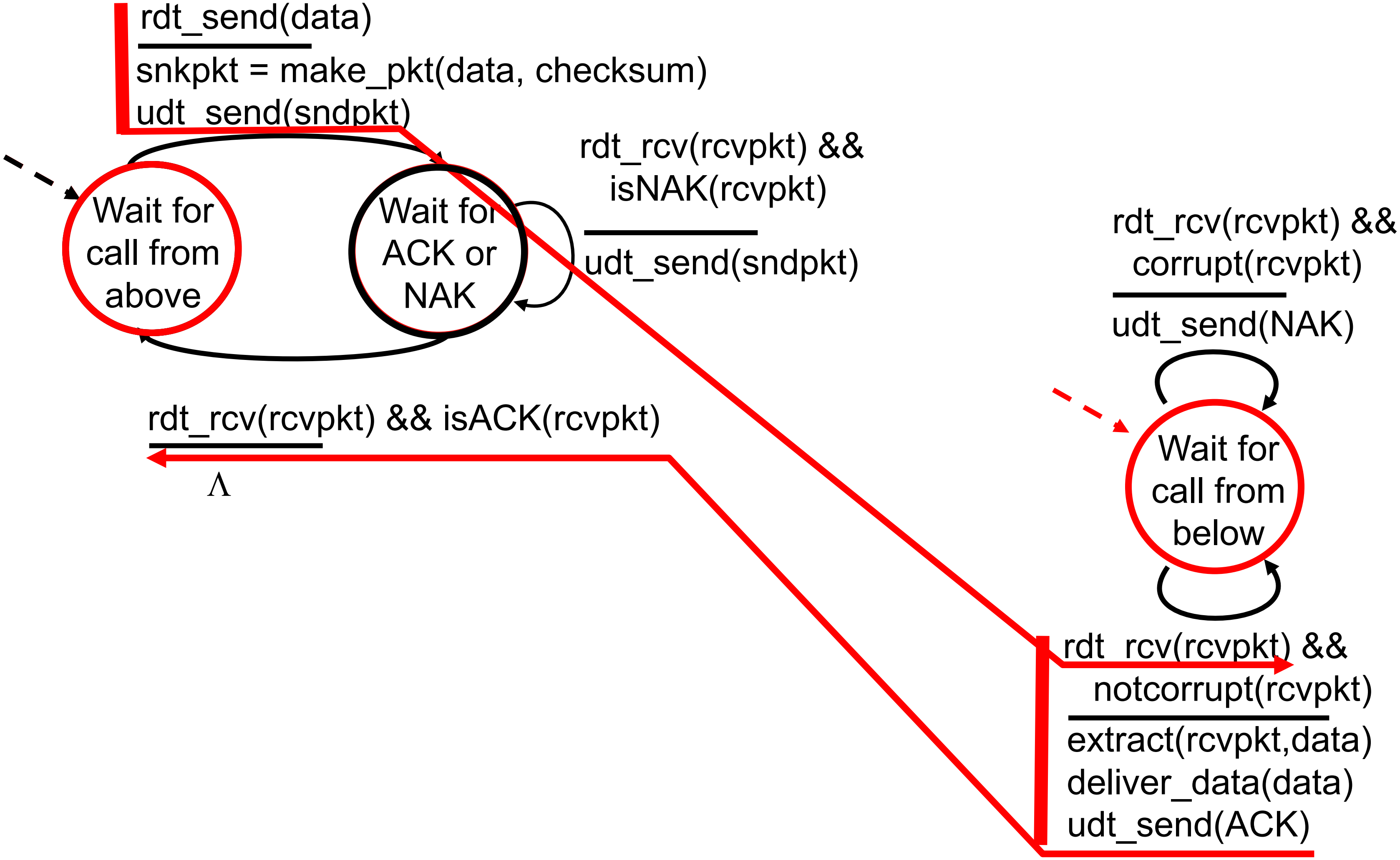
rdt_send(data)
sndpkt = make_pkt(data, checksum)
udt_send(sndpkt)
Wait for call from above
Wait for ACK or NAK
rdt_rcv(rcvpkt) && isNAK(rcvpkt)
udt_send(sndpkt)
rdt_rcv(rcvpkt) && isACK(rcvpkt)
Λ

sender

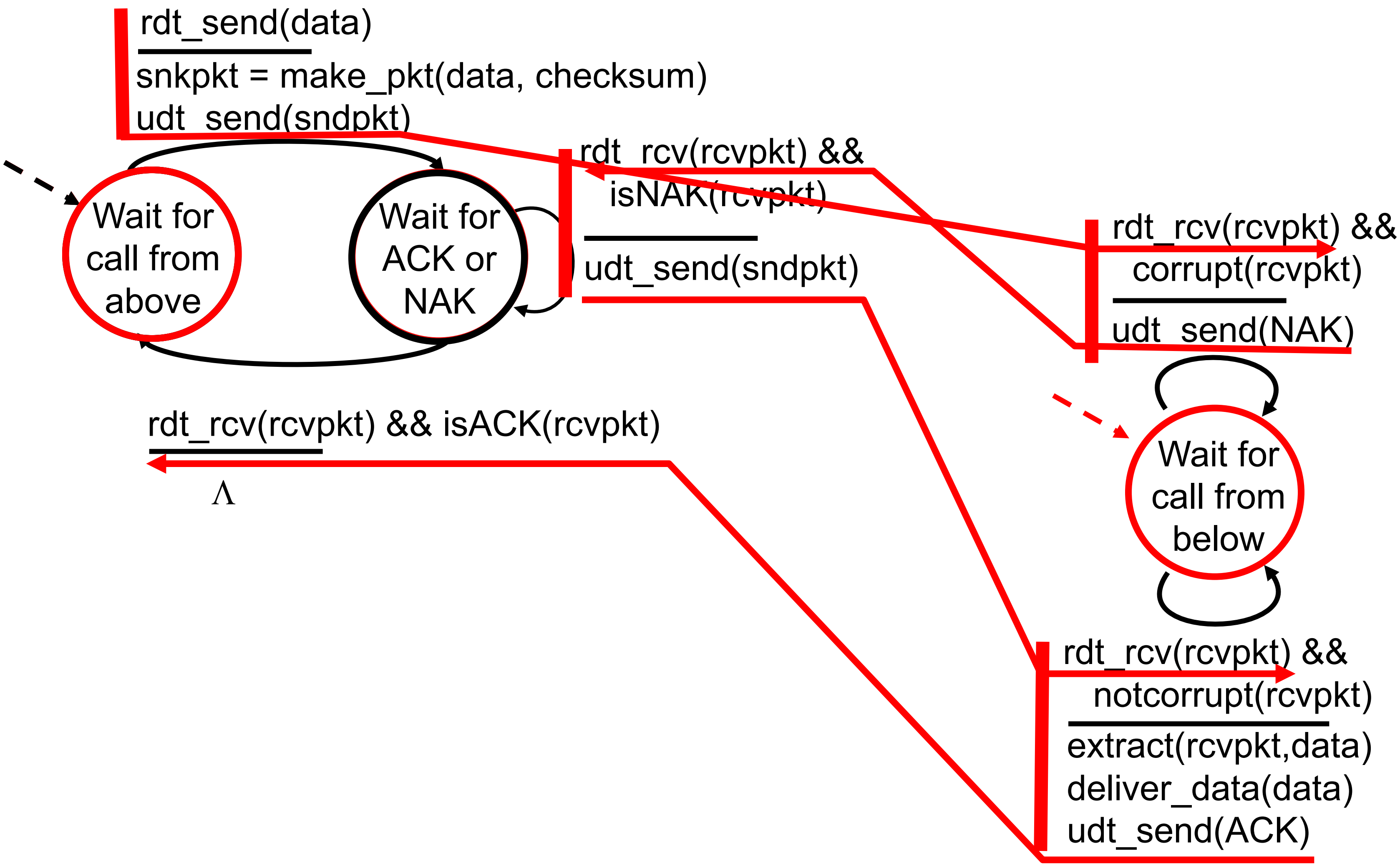
receiver



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) && isNAK(rcvpkt)

udt_send(sndpkt)

rdt_rcv(rcvpkt) && corrupt(rcvpkt)

udt_send(NAK)

rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

Wait for call from below

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

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

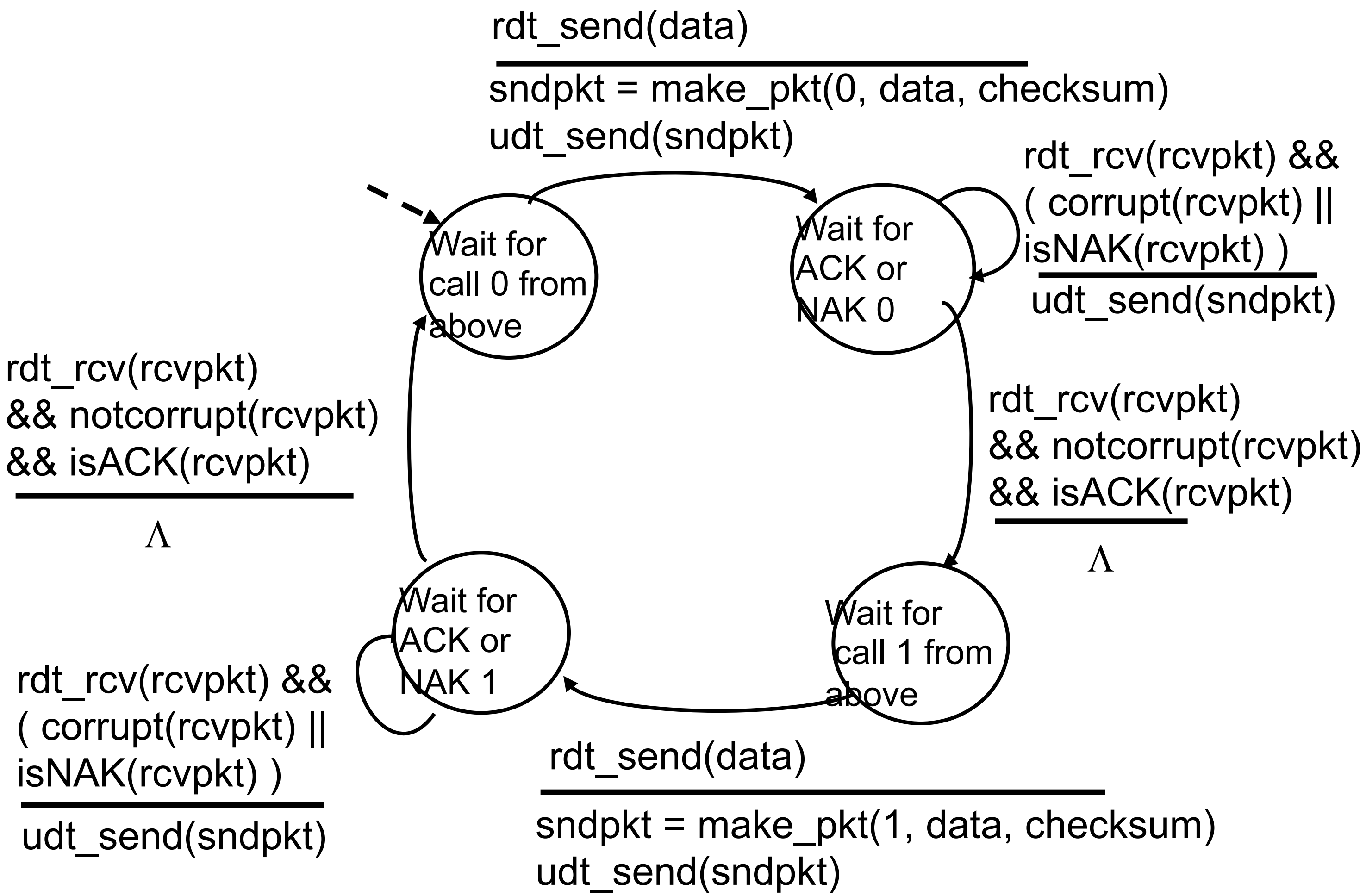
handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- 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



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isNAK(rcvpkt))

udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)

Λ

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)

Λ

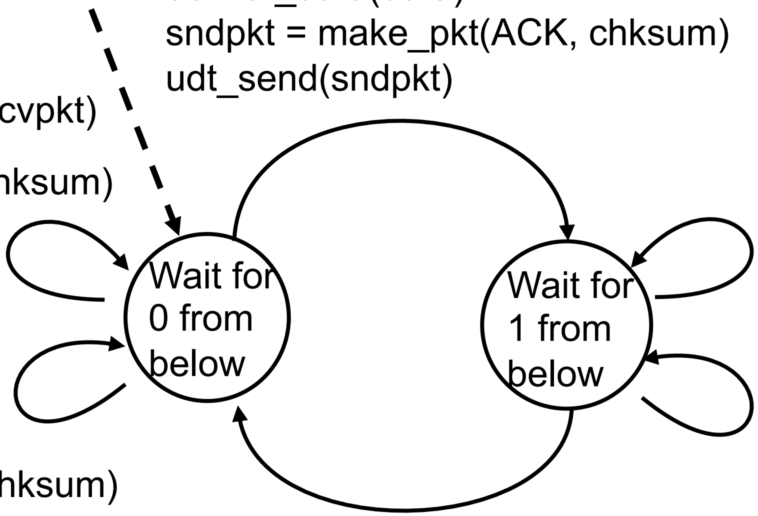
rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isNAK(rcvpkt))

udt_send(sndpkt)

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

rdt2.1: receiver, handles garbled ACK/NAKs



rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)

---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

---

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq1(rcvpkt)

---

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

---

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq0(rcvpkt)

---

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

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 "expected" pkt should have seq # of 0 or 1

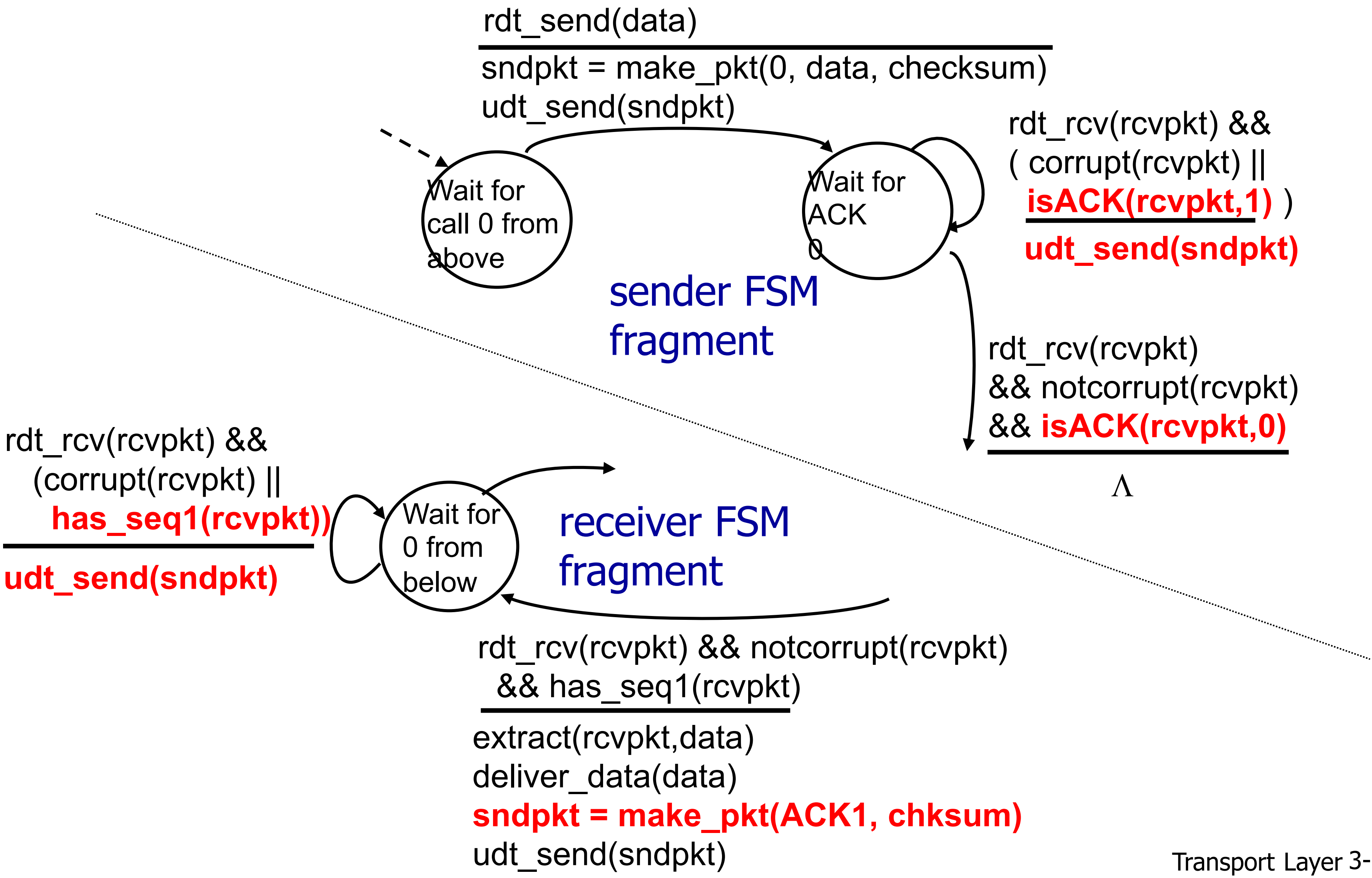
receiver:

- 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



sender FSM fragment

rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& **isACK(rcvpkt,0)**

Λ

receiver FSM fragment

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
has_seq1(rcvpkt))

udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK1, chksum)
udt_send(sndpkt)

rdt3.0: channels with errors *and* loss

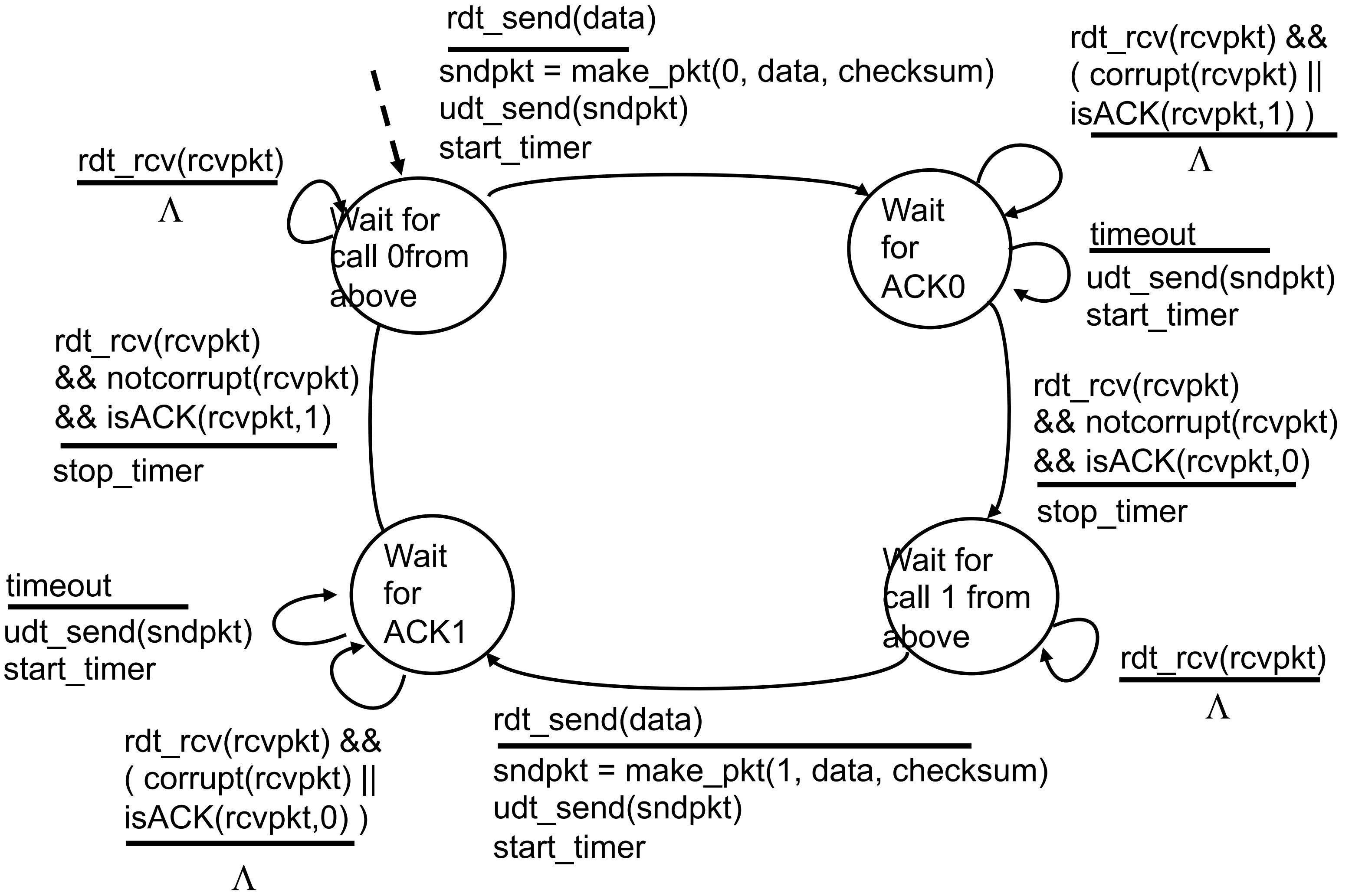
new assumption: underlying channel can also lose packets (data, ACKs)

- 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 seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt3.0 sender



rdt_send(data)
___
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
___
Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))
___
Λ

timeout
___
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)
___
stop_timer

rdt_rcv(rcvpkt)
___
Λ

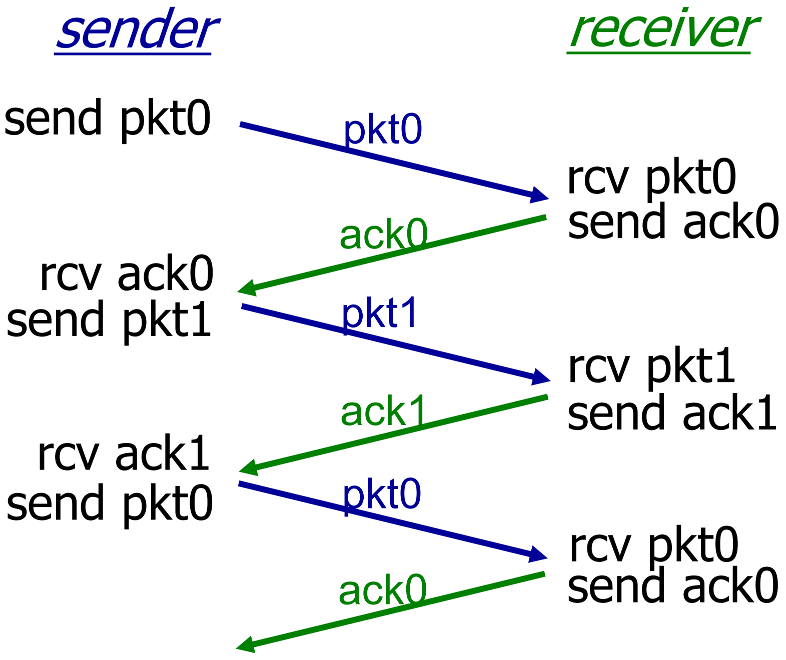
rdt_send(data)
___
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

timeout
___
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,0))
___
Λ

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)
___
stop_timer

rdt3.0 in action



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

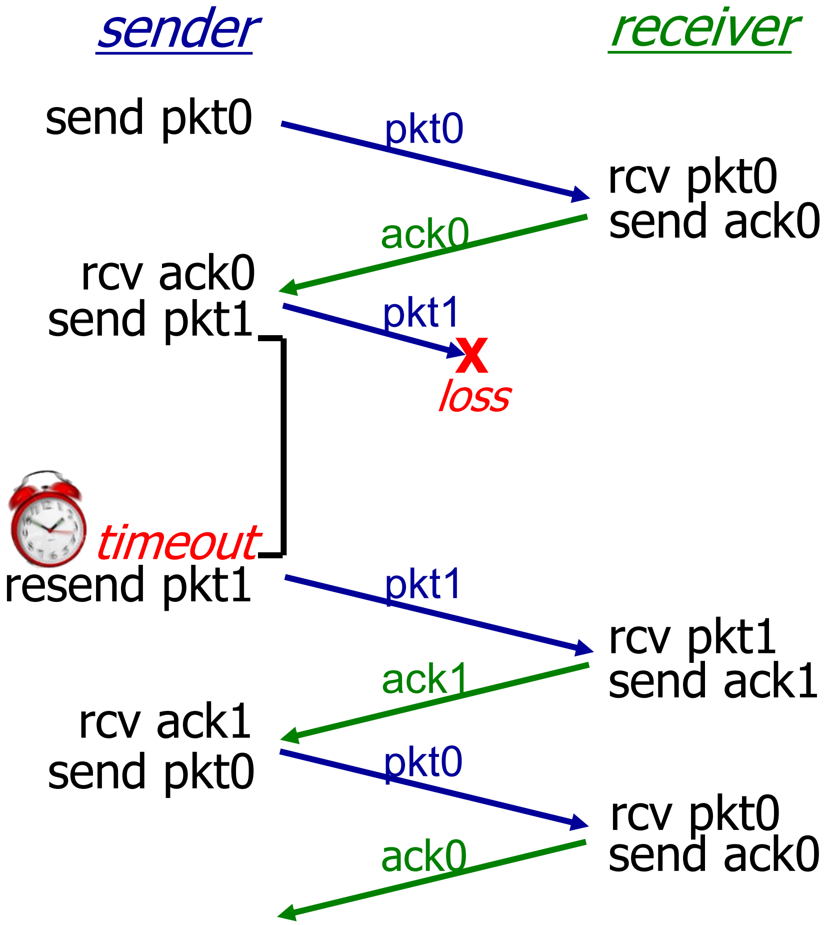
rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(a) no loss



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

loss

timeout
resend pkt1

pkt1

rcv pkt1
send ack1

ack1

rcv ack1
send pkt0

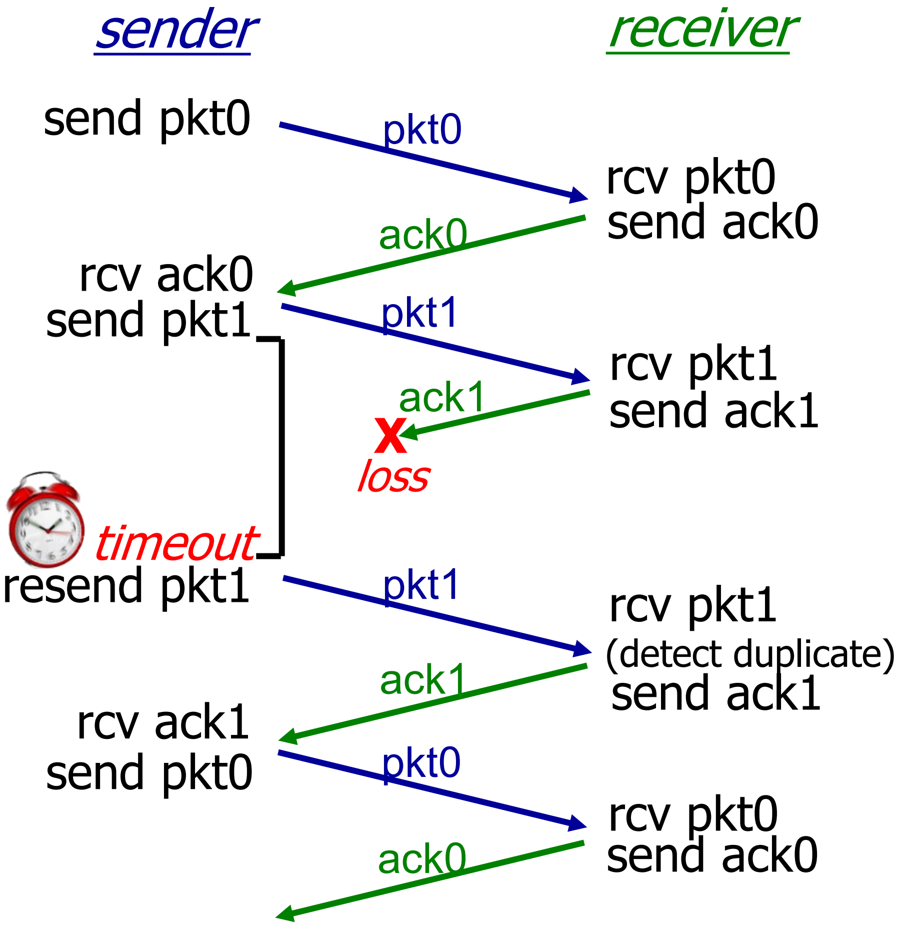
pkt0

rcv pkt0
send ack0

ack0

(b) packet loss

rdt3.0 in action



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

X
loss

timeout
resend pkt1

pkt1

rcv pkt1
(detect duplicate)
send ack1

ack1

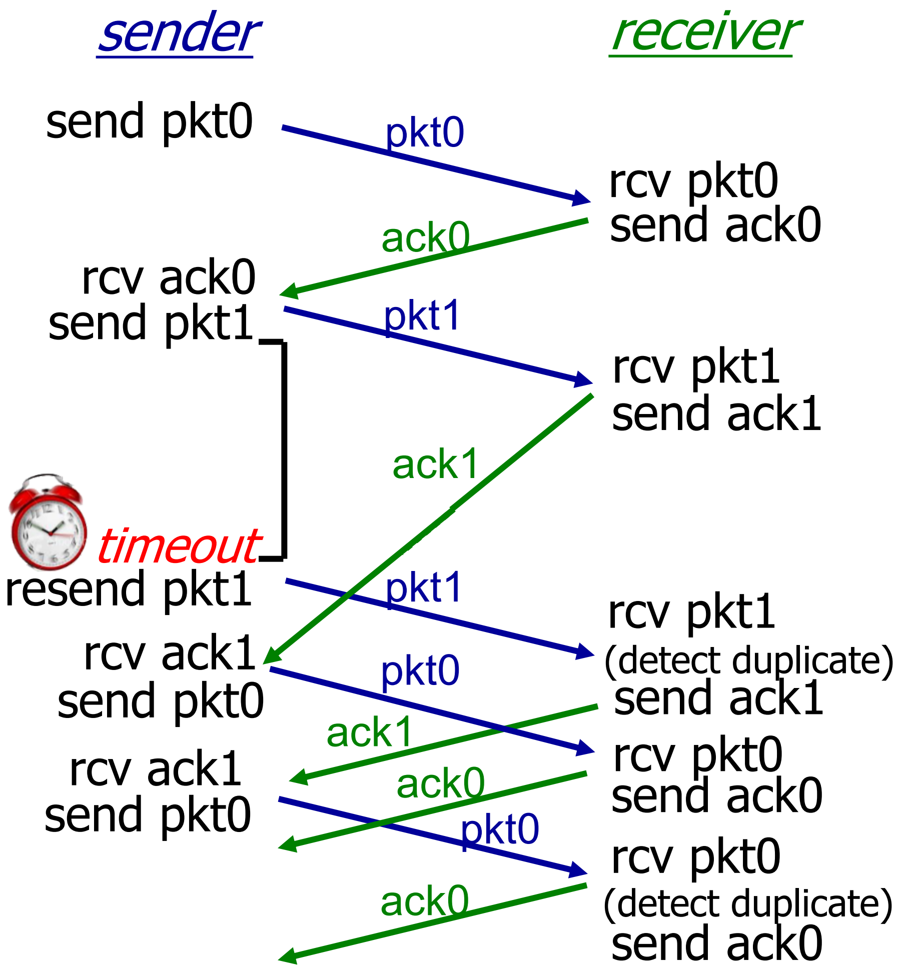
rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(c) ACK loss



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

timeout
resend pkt1

pkt1

rcv ack1
send pkt0

pkt0

rcv pkt1
(detect duplicate)
send ack1

ack1

rcv pkt0
send ack0

rcv ack1
send pkt0

ack0

pkt0

rcv pkt0
(detect duplicate)
send ack0

ack0

(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance is not good
- e.g.: 1 Gbps link, 15 ms prop. Delay (RTT=30 ms), 8000 bit packet:

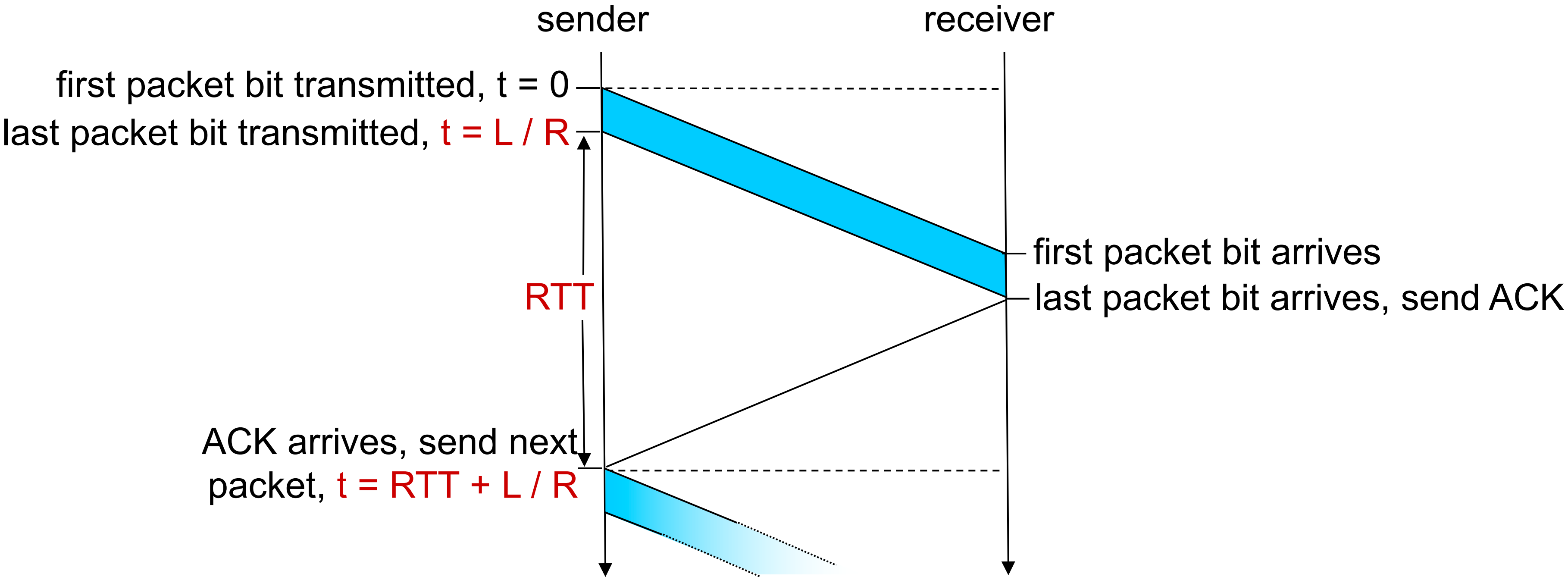
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

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

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

  - if RTT=30 msec, 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

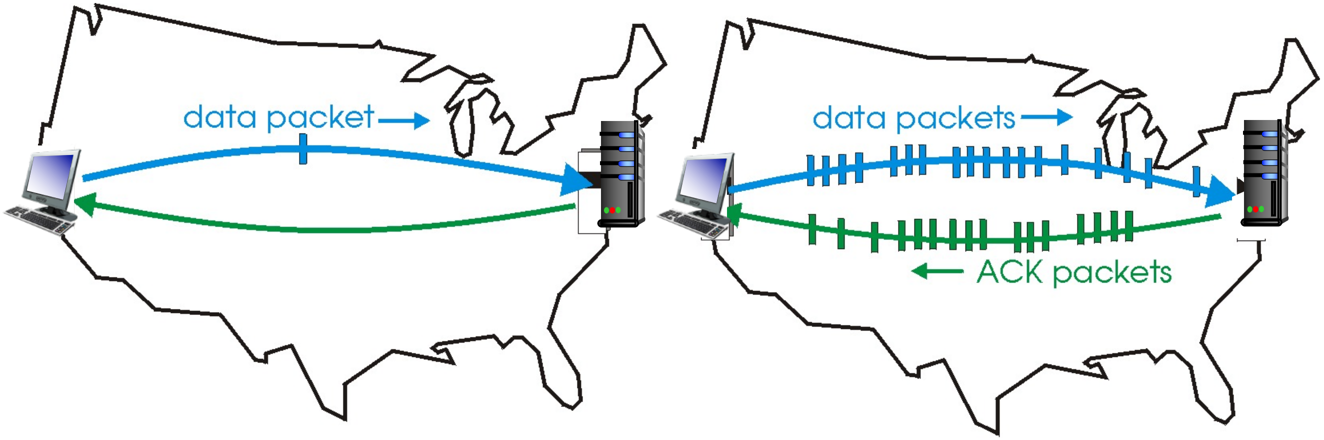

sender
receiver
first packet bit transmitted, t = 0
last packet bit transmitted, t = L / R
RTT
first packet bit arrives
last packet bit arrives, send ACK
ACK arrives, send next
packet, t = RTT + L / R

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

Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

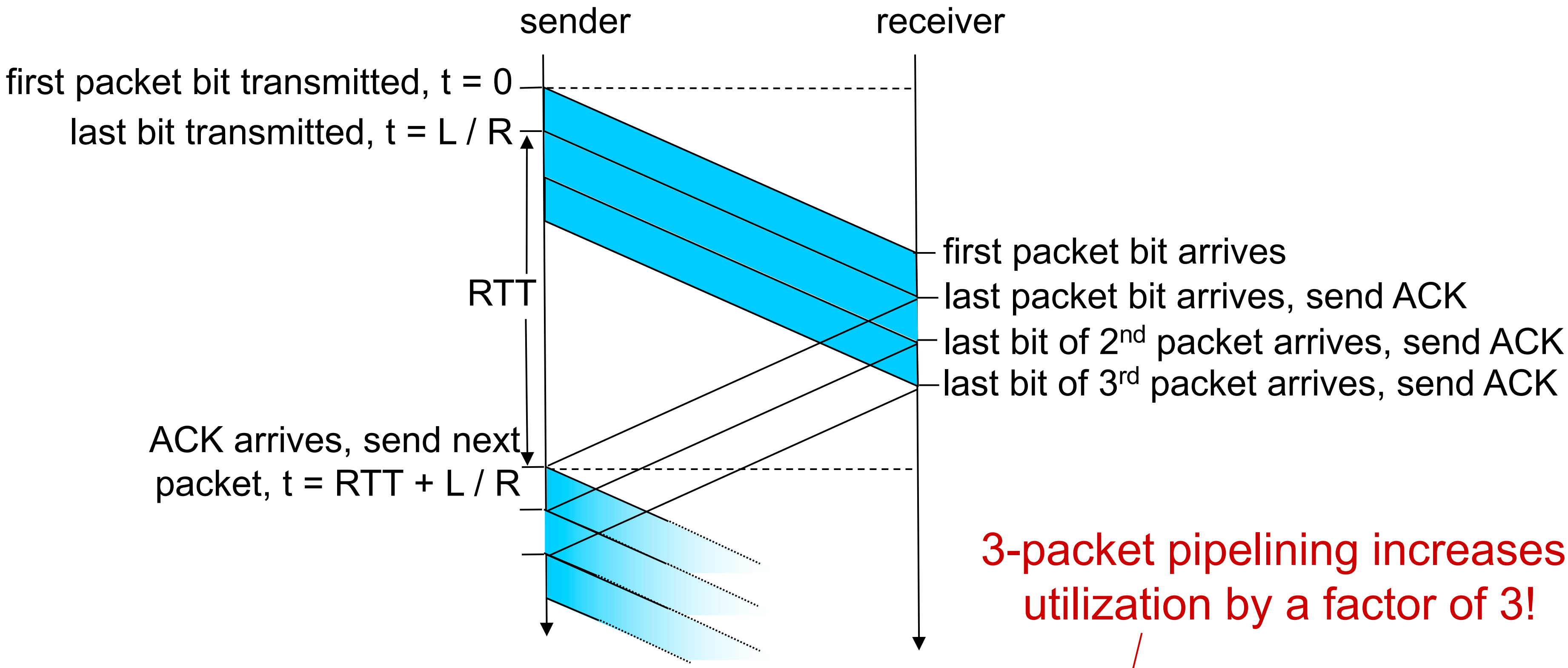
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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


sender
receiver
first packet bit transmitted, t = 0
last bit transmitted, t = L / R
RTT
ACK arrives, send next packet, t = RTT + L / R
first packet bit arrives
last packet bit arrives, send ACK
last bit of 2nd packet arrives, send ACK
last bit of 3rd packet arrives, send ACK

3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

Pipelined protocols: overview

Go-back-N:

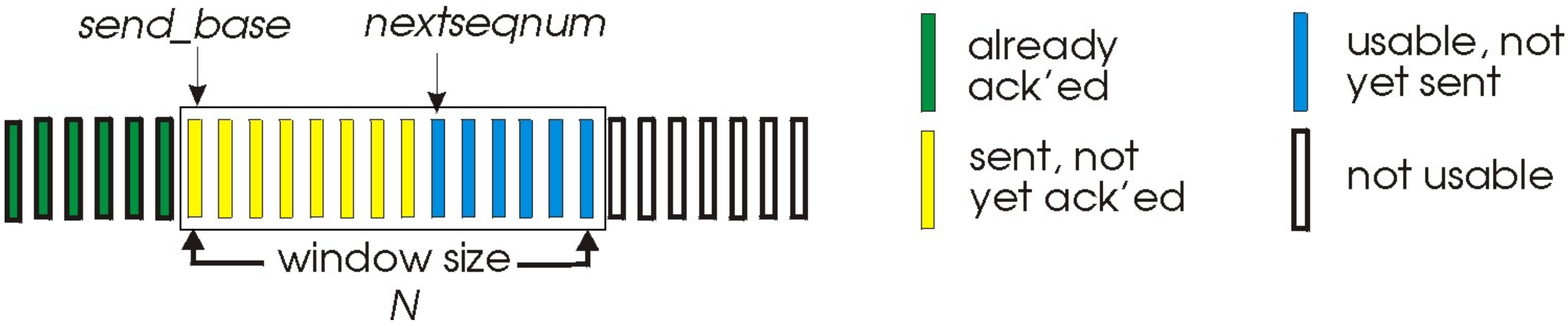
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

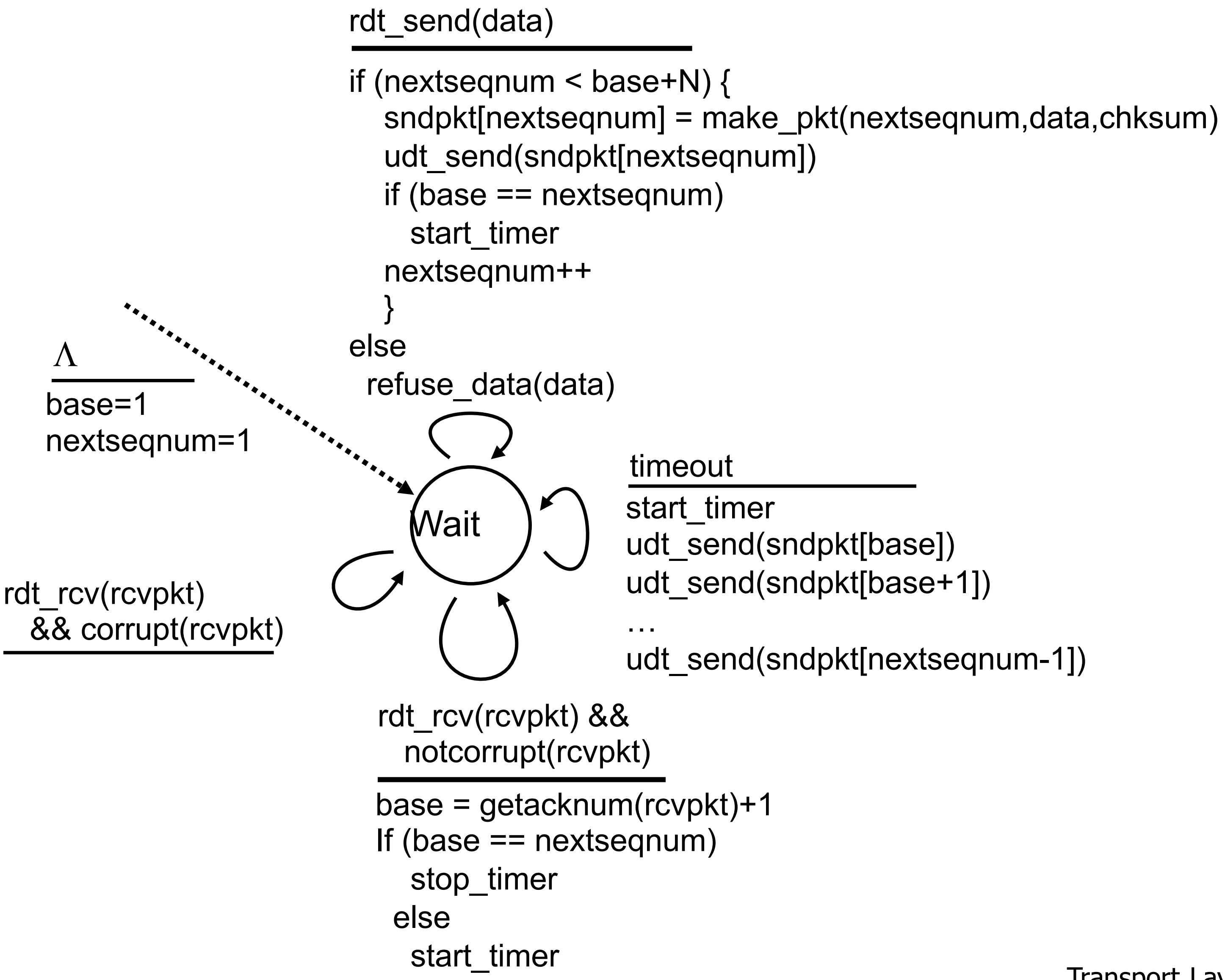
Go-Back-N: sender

- k-bit seq # in pkt header
- “window” of up to N, consecutive unack’ed pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - *“cumulative ACK”*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet 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)
```

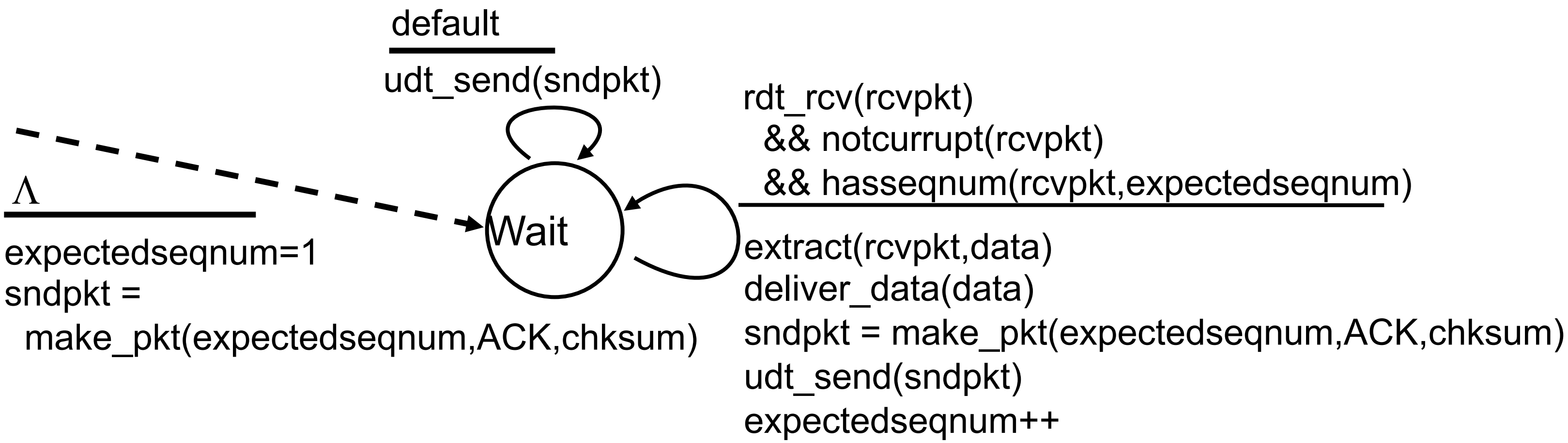
```
Λ
________________
base=1
nextseqnum=1
```

```
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

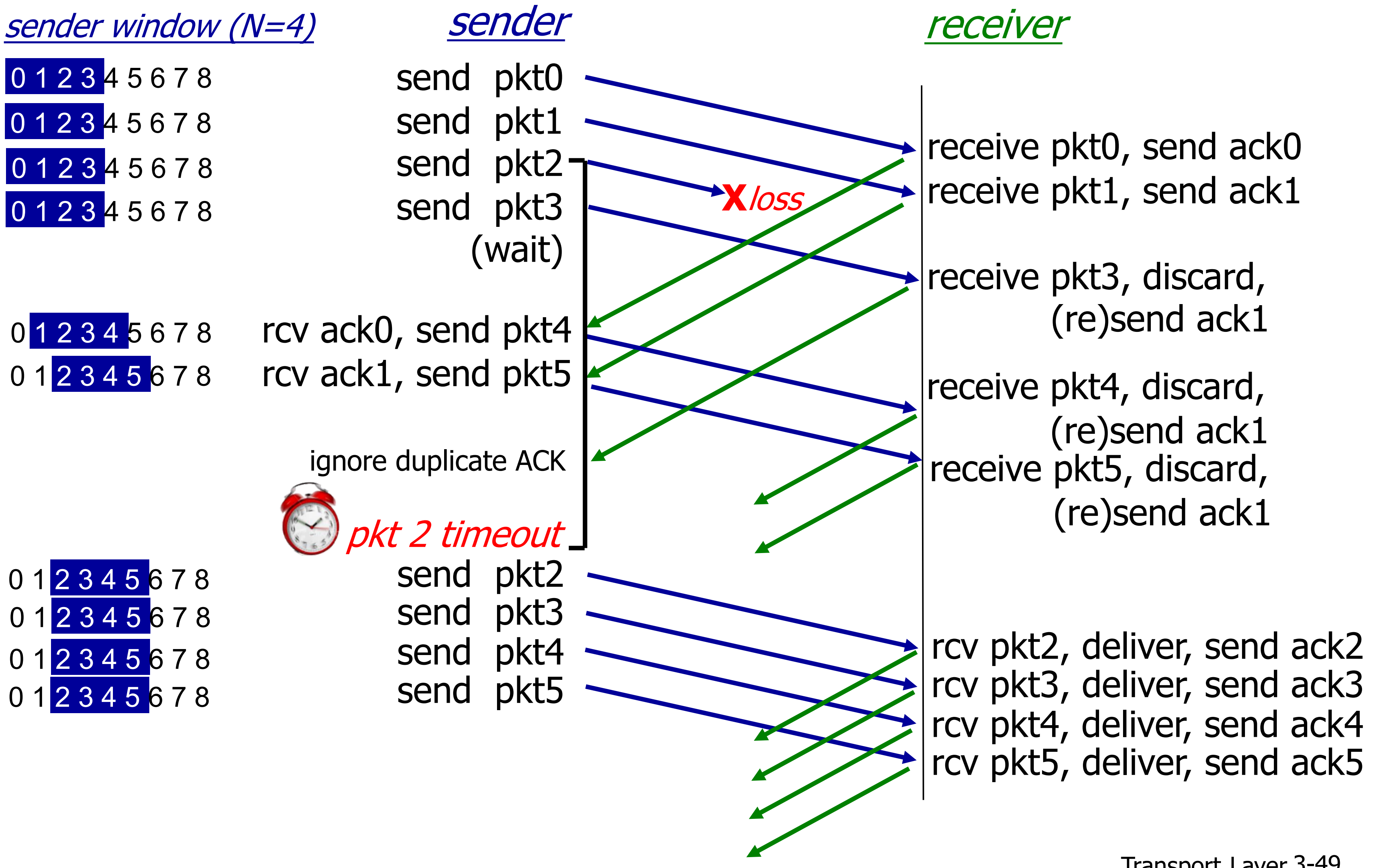


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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



sender window (N=4)

sender

receiver

0 1 2 3 4 5 6 7 8 send pkt0

0 1 2 3 4 5 6 7 8 send pkt1

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

(wait)

receive pkt0, send ack0

receive pkt1, send ack1

X *loss*

receive pkt3, discard,
(re)send ack1

0 1 2 3 4 5 6 7 8 rcv ack0, send pkt4

0 1 2 3 4 5 6 7 8 rcv ack1, send pkt5

receive pkt4, discard,
(re)send ack1

ignore duplicate ACK

receive pkt5, discard,
(re)send ack1

pkt 2 timeout

0 1 2 3 4 5 6 7 8 send pkt2

0 1 2 3 4 5 6 7 8 send pkt3

0 1 2 3 4 5 6 7 8 send pkt4

0 1 2 3 4 5 6 7 8 send pkt5

rcv pkt2, deliver, send ack2

rcv pkt3, deliver, send ack3

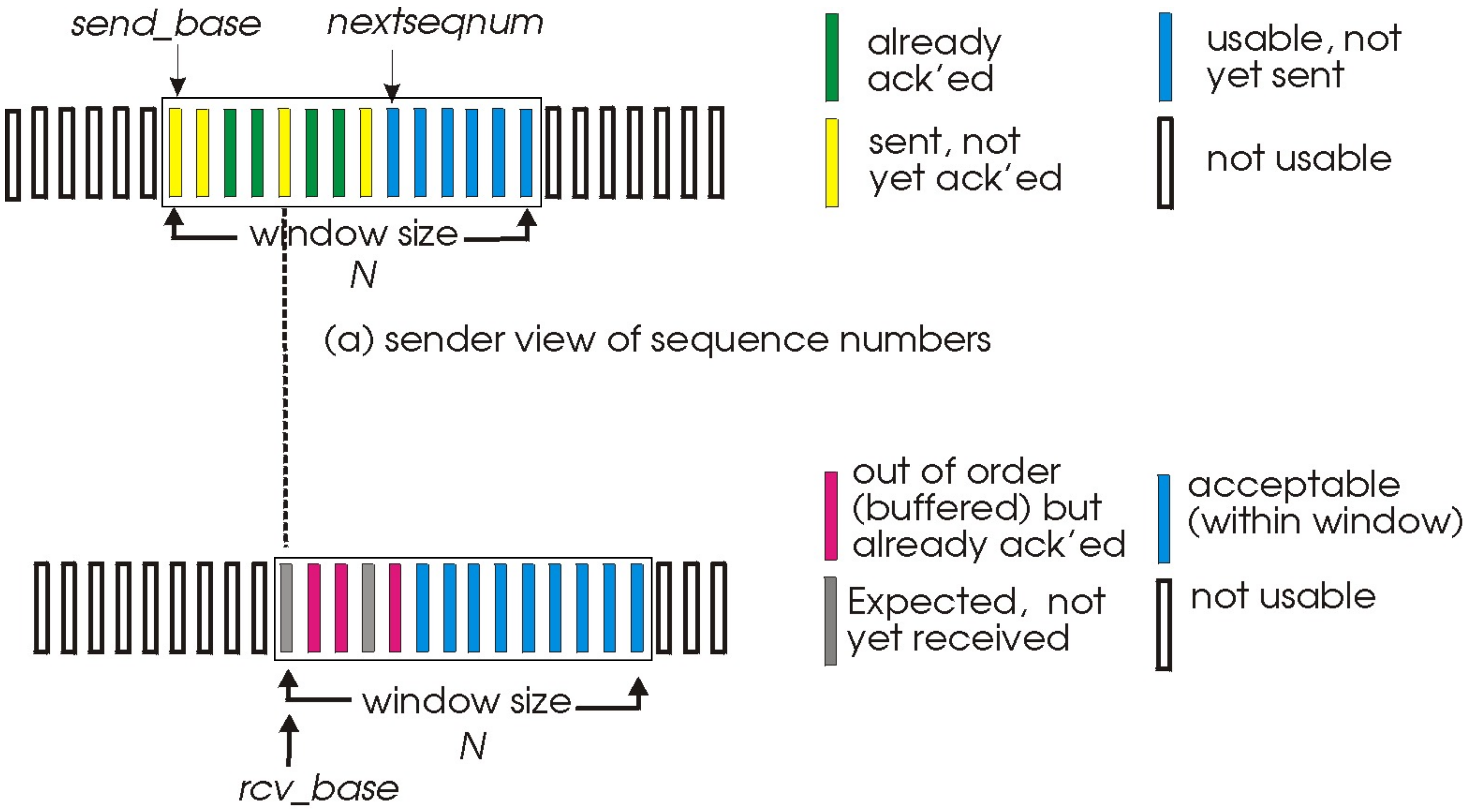
rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

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
  - limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers

(b) receiver view of sequence numbers

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

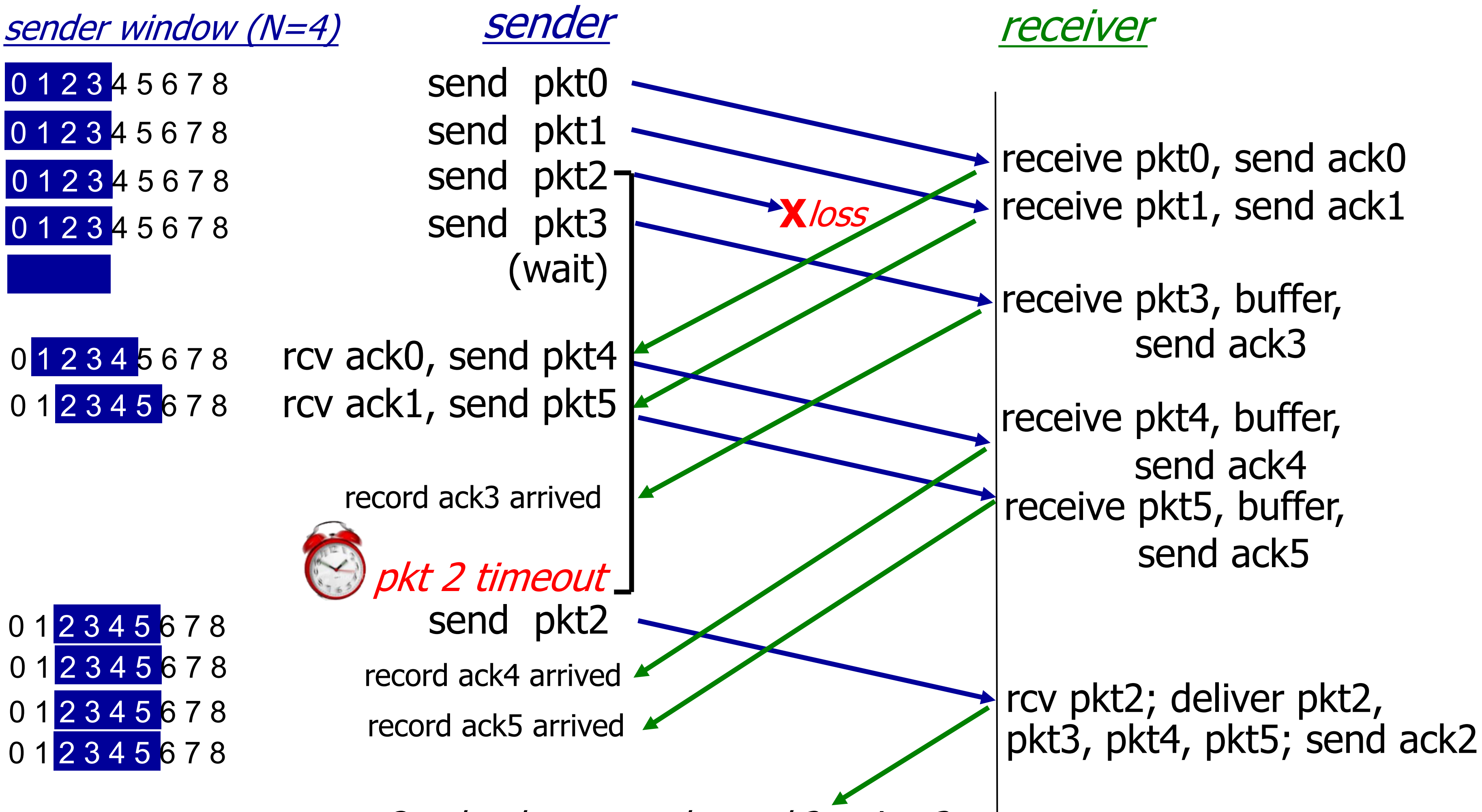
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n) (this pkt has previously acked)

otherwise:

- ignore

Selective repeat in action



sender window (N=4)

sender

receiver

sender window	sender	receiver
0 1 2 3 4 5 6 7 8	send pkt0	receive pkt0, send ack0
0 1 2 3 4 5 6 7 8	send pkt1	receive pkt1, send ack1
0 1 2 3 4 5 6 7 8	send pkt2	X loss
0 1 2 3 4 5 6 7 8	send pkt3 (wait)	receive pkt3, buffer, send ack3
0 1 2 3 4 5 6 7 8	rcv ack0, send pkt4	receive pkt4, buffer, send ack4
0 1 2 3 4 5 6 7 8	rcv ack1, send pkt5	receive pkt5, buffer, send ack5
	record ack3 arrived	
	pkt 2 timeout	
0 1 2 3 4 5 6 7 8	send pkt2	rcv pkt2; deliver pkt2, pkt3, pkt4, pkt5; send ack2
0 1 2 3 4 5 6 7 8	record ack4 arrived	
0 1 2 3 4 5 6 7 8	record ack5 arrived	
0 1 2 3 4 5 6 7 8		

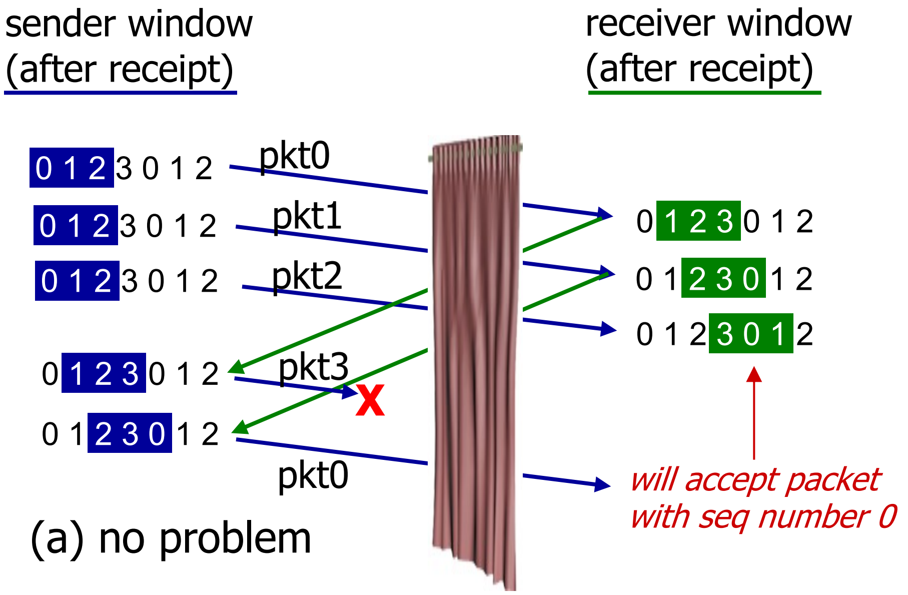
Q: what happens when ack2 arrives?

Selective repeat: dilemma

example:

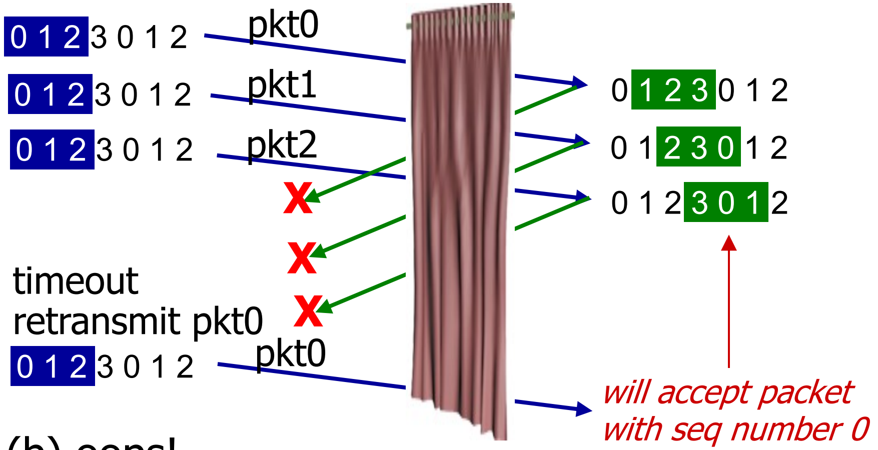
- seq #' s: 0, 1, 2, 3
- window size=3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)

Q: what relationship between seq # size and window size to avoid problem in (b)?



(a) no problem

receiver can't see sender side.
receiver behavior identical in both cases!
something's (very) wrong!



(b) oops!

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- flow control
- connection management

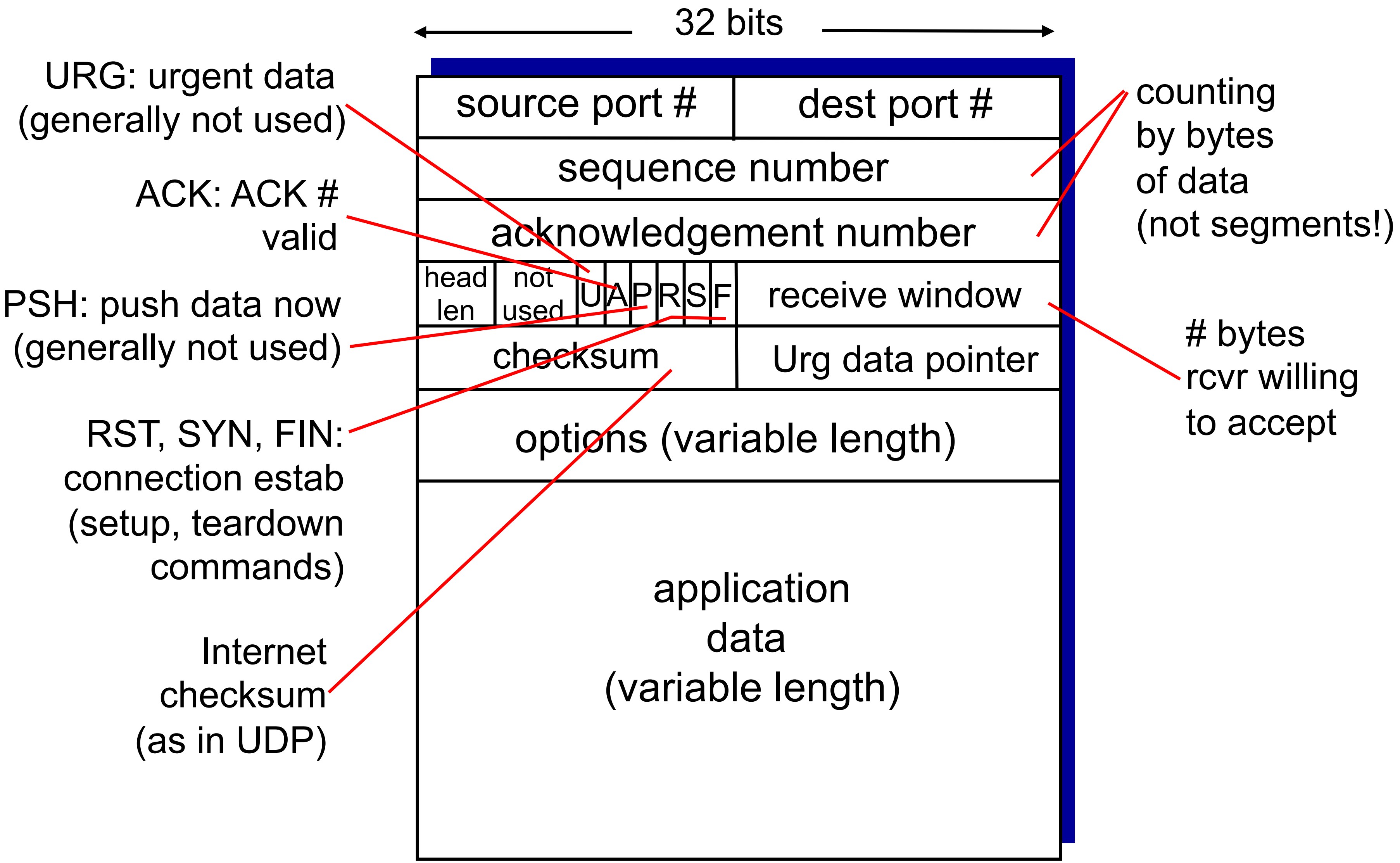
3.6 principles of congestion control

3.7 TCP congestion control

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

TCP segment structure



32 bits

source port # | dest port #

sequence number

acknowledgement number

head len | not used | U | A | P | R | S | F | receive window

checksum | Urg data pointer

options (variable length)

application data (variable length)

URG: urgent data (generally not used)

ACK: ACK # valid

PSH: push data now (generally not used)

RST, SYN, FIN: connection estab (setup, teardown commands)

Internet checksum (as in UDP)

counting by bytes of data (not segments!)

# bytes rcvr willing to accept

TCP seq. numbers

- Suppose that the data stream consists of a file consisting of 500,000 bytes, that the MSS is 1,000 bytes, and that the first byte of the data stream is numbered 0.
- TCP constructs 500 segments out of the data stream. The first segment gets assigned sequence number 0, the second segment gets 1,000 and so on.

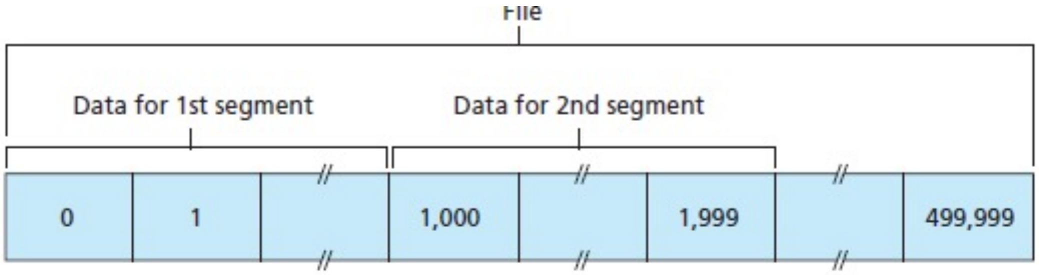


Figure 3.30 ♦ Dividing file data into TCP segments

TCP seq. numbers, ACKs

sequence numbers:

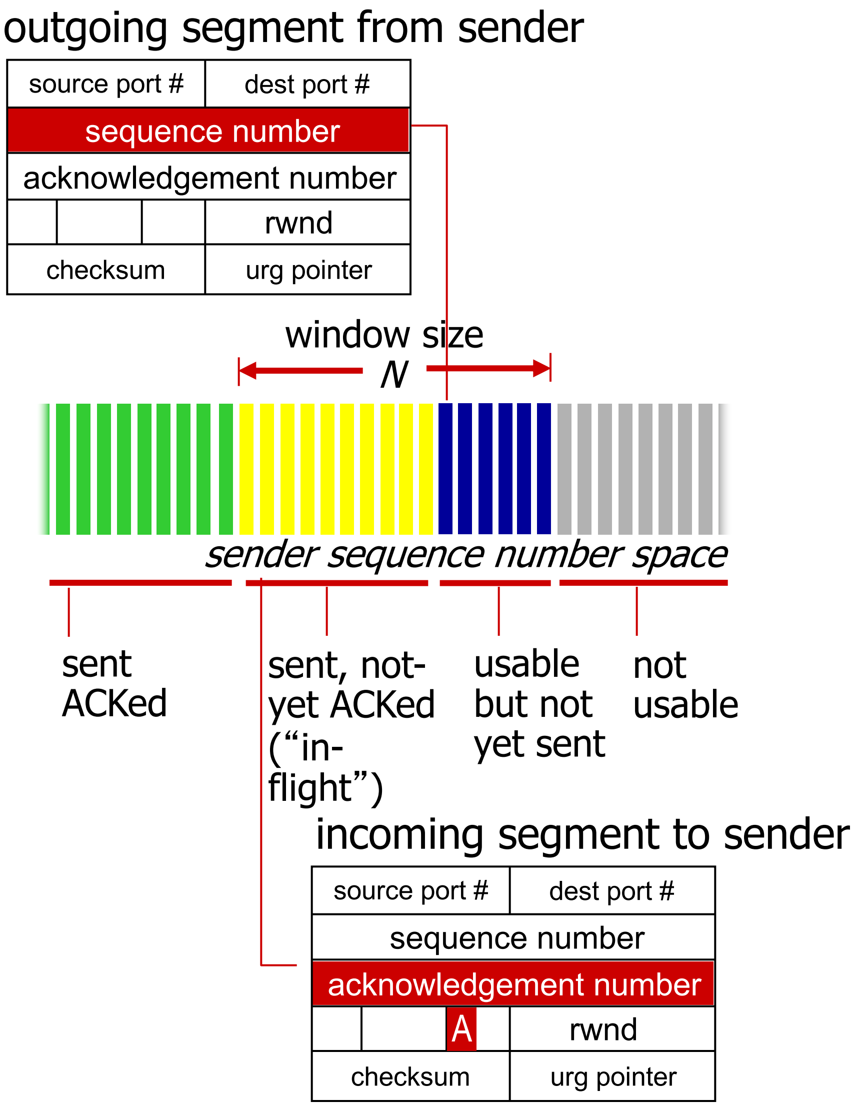
- byte stream "number" of first byte in segment's data

acknowledgements:

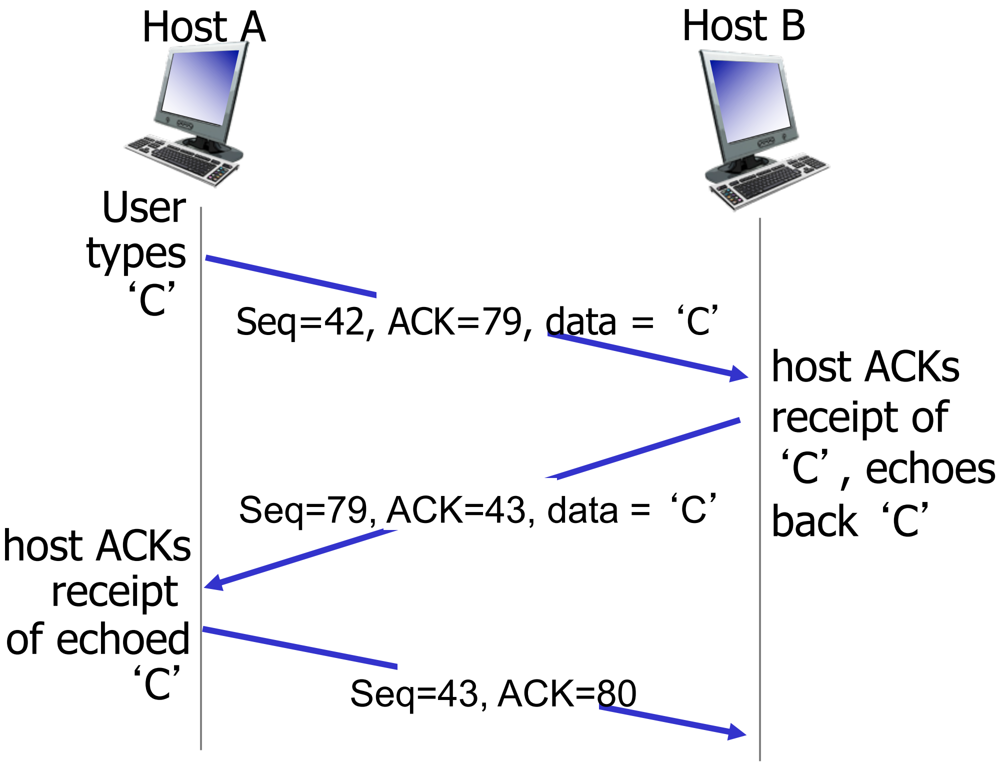
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP seq. numbers, ACKs


Host A
Host B
User types 'C'
Seq=42, ACK=79, data = 'C'
host ACKs receipt of 'C', echoes back 'C'
Seq=79, ACK=43, data = 'C'
host ACKs receipt of echoed 'C'
Seq=43, ACK=80

simple telnet scenario

TCP round trip time, 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

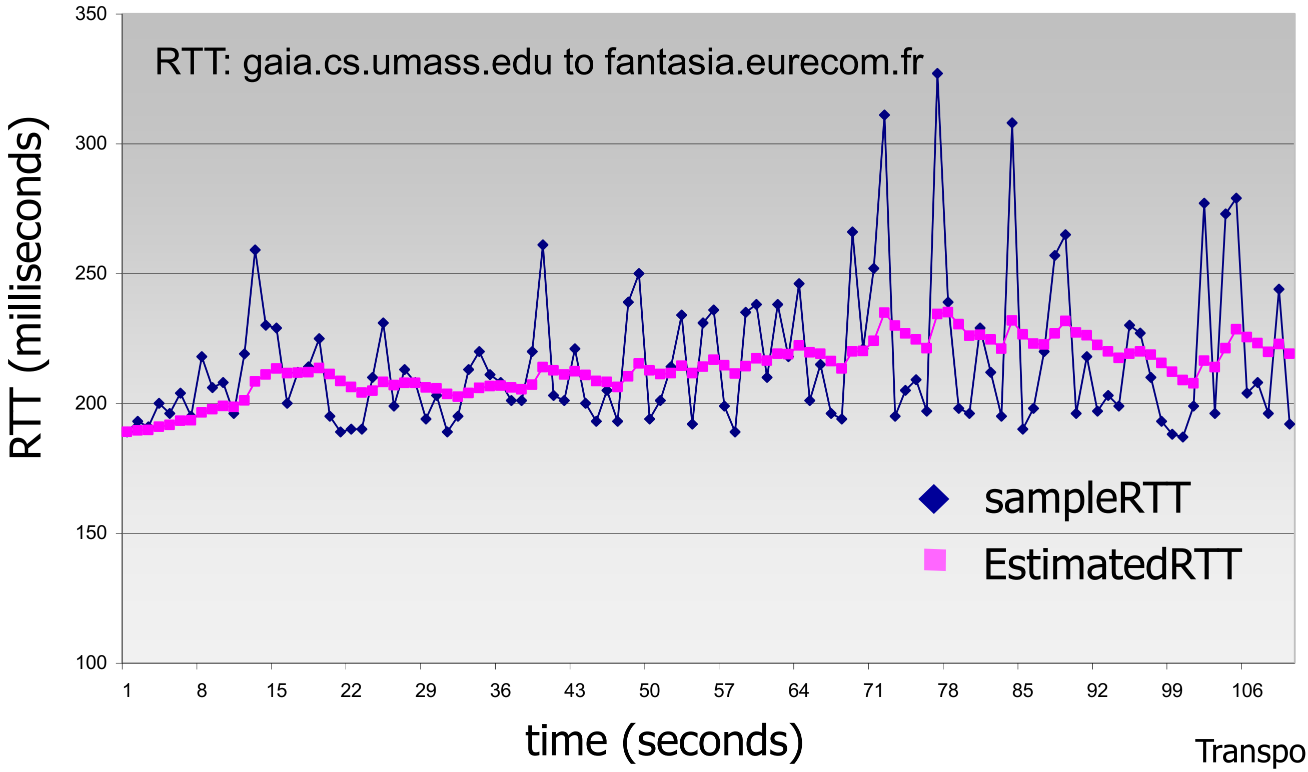
Q: how to estimate RTT?

- **`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, timeout

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

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** **`->`** larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

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

             (typically, β = 0.25)
```

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT

"safety margin"

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- connection management

3.6 principles of congestion control

3.7 TCP congestion control

TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

let's initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

Three events in sender:

1. 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`

2. timeout:

- retransmit segment that caused timeout
- restart timer

3. ack rcvd:

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

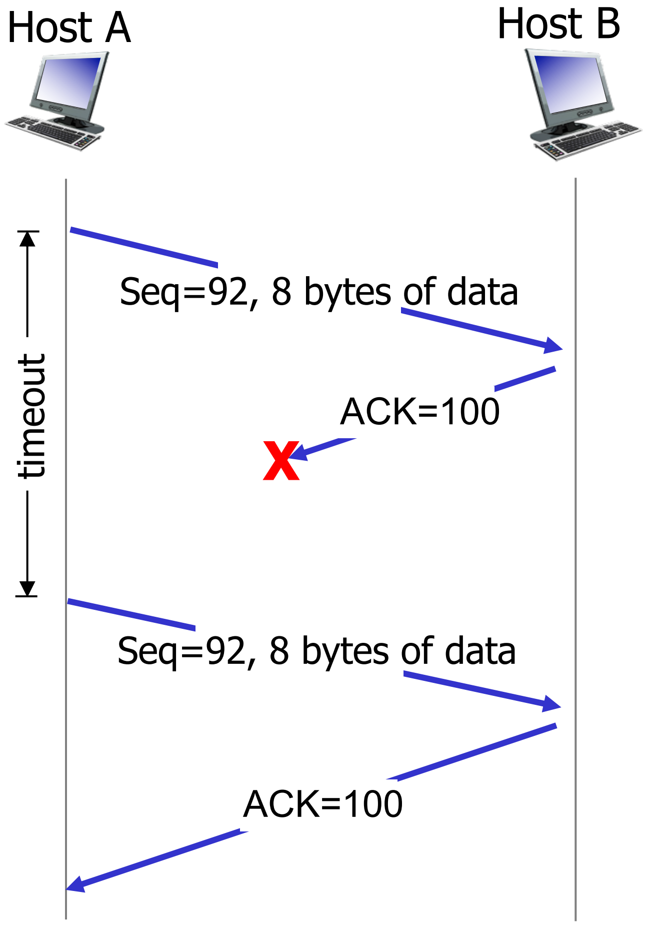
TCP sender (simplified)

```
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 */
```

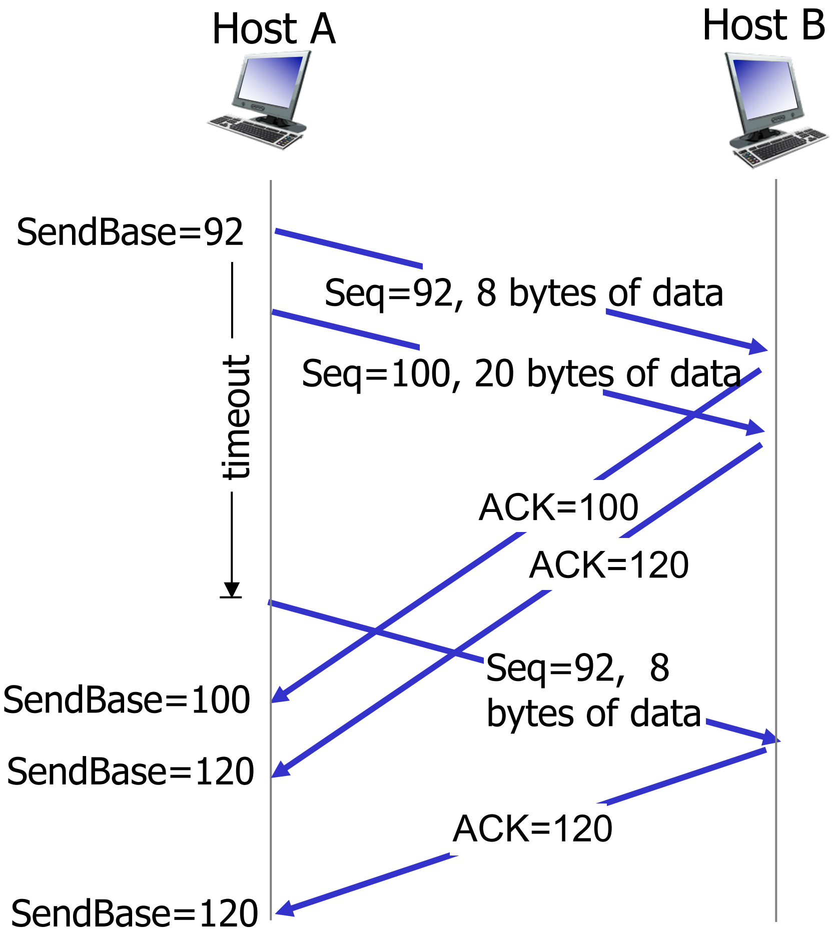
Comment:
- SendBase-1: last cumulatively ack'ed byte

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

TCP: retransmission scenarios

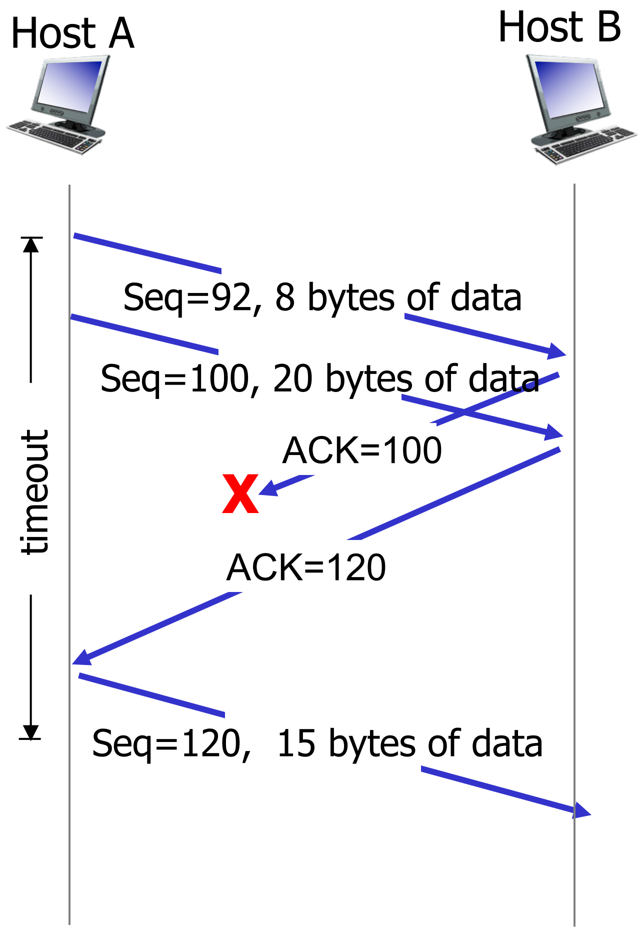


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK

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**(prevents creating extra traffic)**
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments **(do not let sender wait too long)**
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte **(enables fast retransmit)**
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

Doubling the Timeout Interval

- Each time TCP retransmits, it sets the next timeout interval to twice the previous value, rather than deriving it from the last EstimatedRTT and DevRTT
- If Time out Interval is .75 sec when the timer first expires. TCP will then retransmit this segment and set the new expiration time to 1.5 sec.
- If the timer expires again 1.5 sec later, TCP will again retransmit this segment, now setting the expiration time to 3.0 sec. Thus the intervals grow exponentially after each retransmission

TCP 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.

TCP fast retransmit

if sender receives 3 duplicate ACKs for same data

("triple duplicate ACKs"),

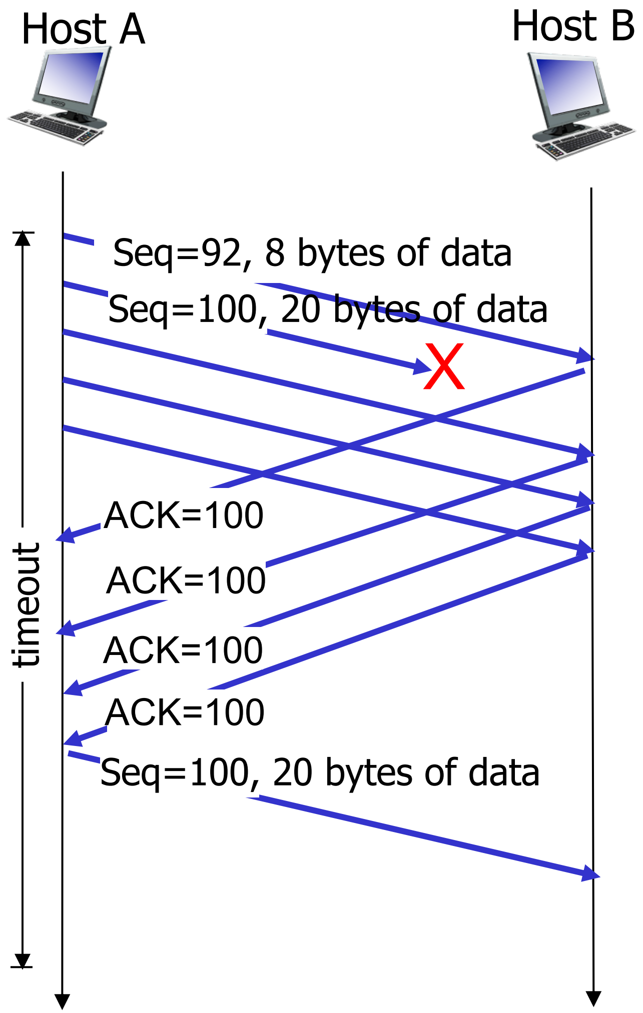
resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

Is TCP a GBN or SR?

- TCP looks a lot like GBN but TCP will buffer correctly received but out-of-order segments
- Some version, has selective ack
- So it is hybrid GBN and SR

TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

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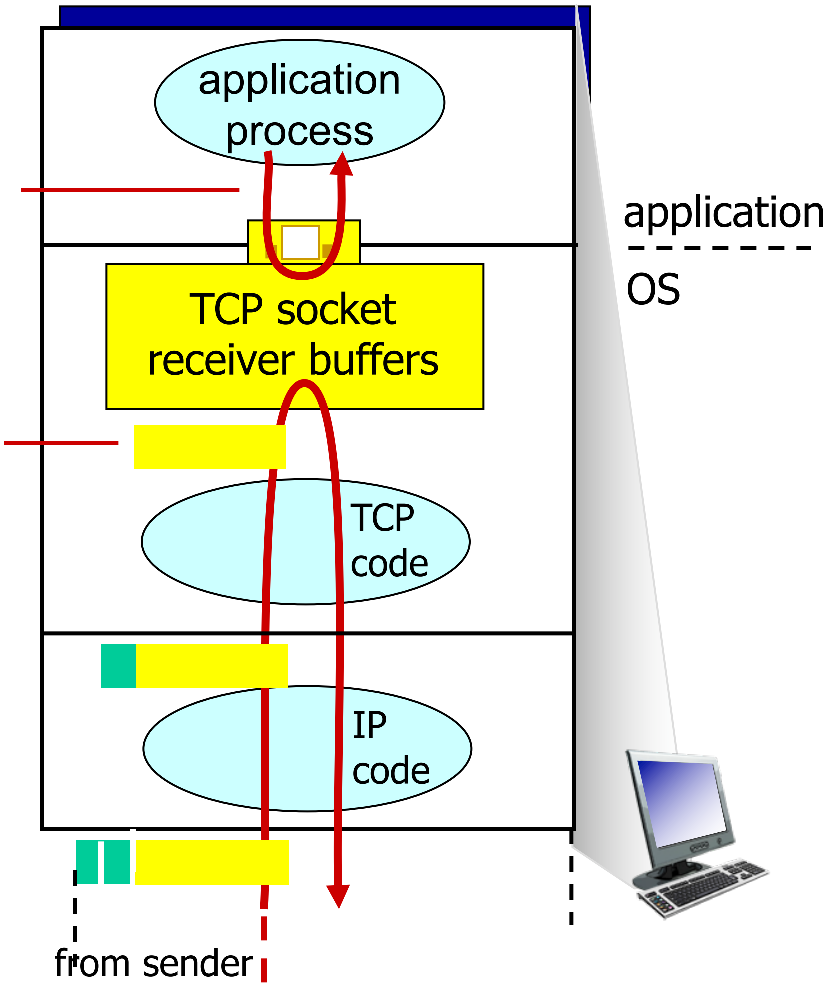
3.7 TCP congestion control

TCP flow control

application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)

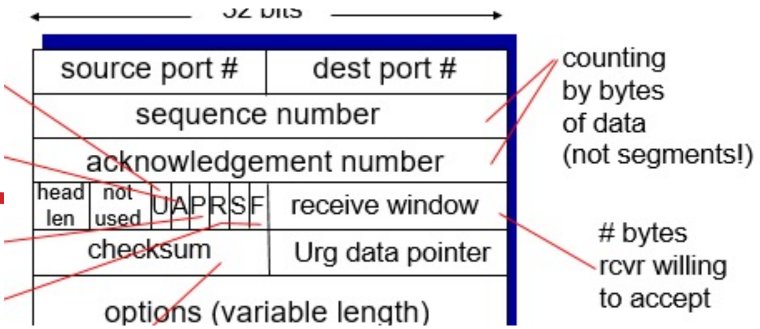


receiver protocol stack

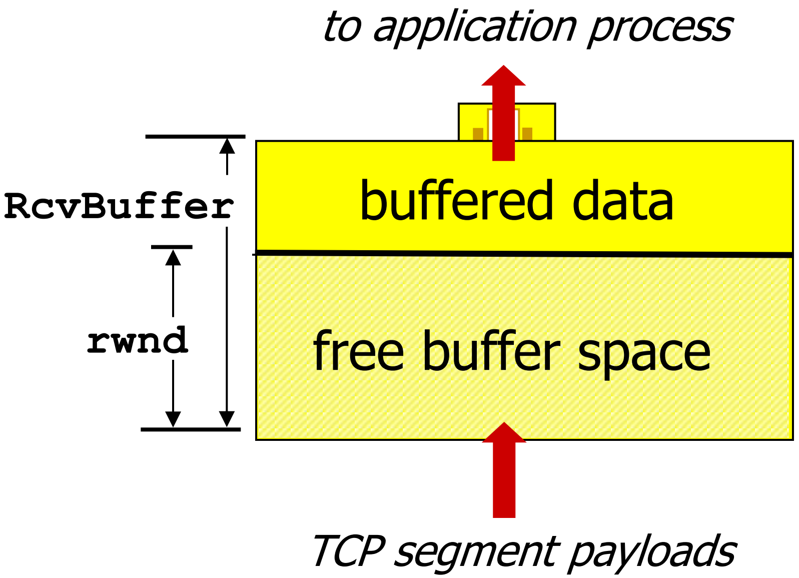
flow control

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

TCP flow control



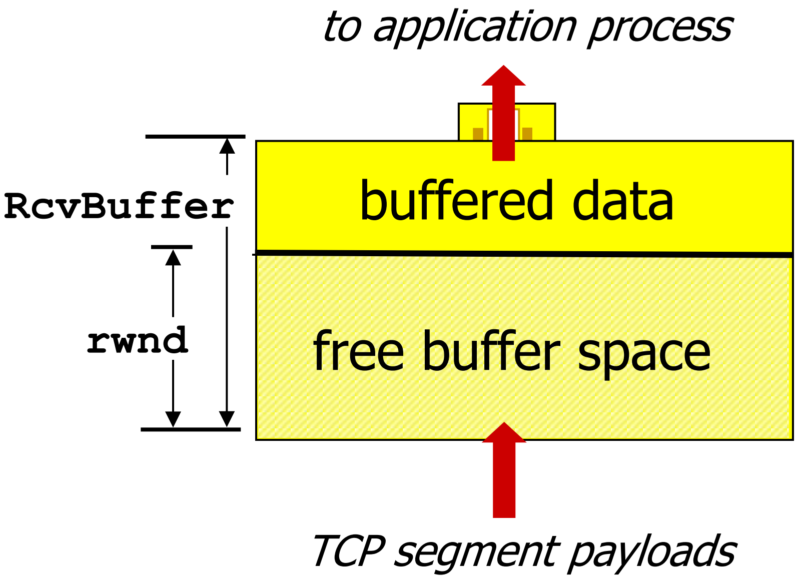
- receiver "advertises" free buffer space by including **`rwnd`** value in TCP header of receiver-to-sender segments
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow



receiver-side buffering

TCP flow control

- Because TCPis not permitted to overflow the allocated buffer, we must have
- LastByteRcvd – LastByteRead less than or equal to RcvBuffer
- The receive window, denoted rwnd is set to the amount of spare room in the buffer:
- rwnd = RcvBuffer – [LastByteRcvd – LastByteRead]



receiver-side buffering

TCP flow control

- Host A is never informed that some space has opened up in Host B’s receive buffer
- Host A is blocked and can transmit no more data!
- To solve this problem, the TCP specification requires Host A to continue to send segments with one data byte when B’s receive window is zero.
- These segments will be acknowledged by the receiver.
- Eventually the buffer will begin to empty and the acknowledgments will contain a nonzero rwnd value.

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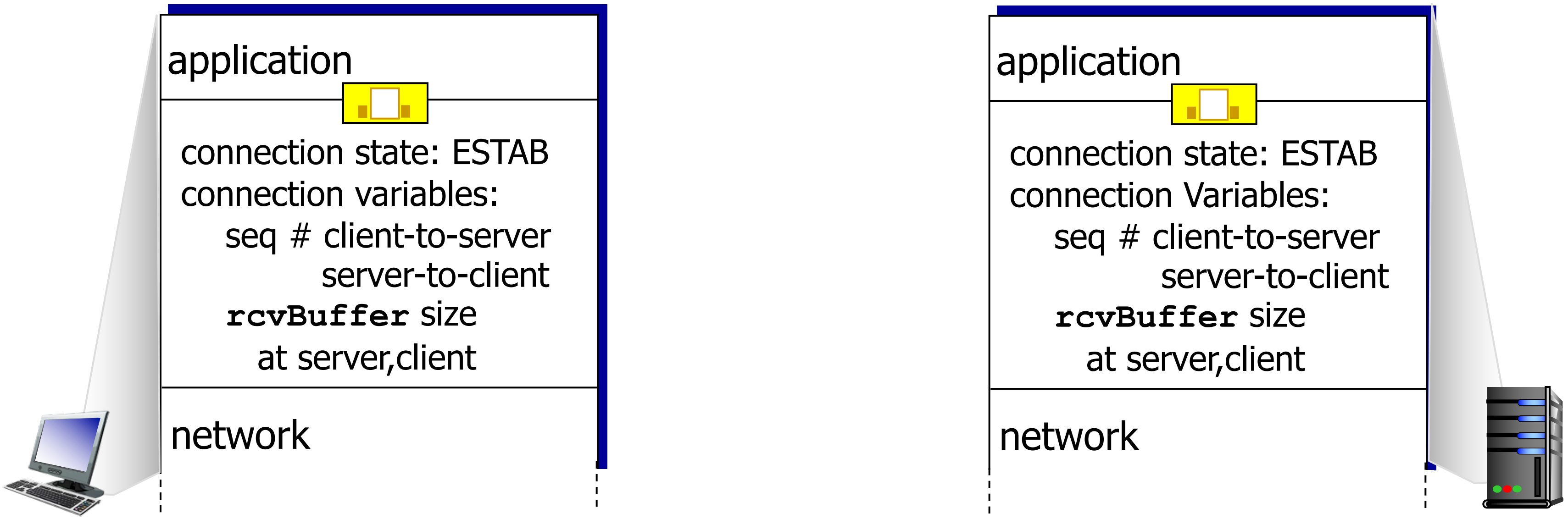
3.7 TCP congestion control

Connection Management

before exchanging data, sender/receiver "handshake":

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters



application

connection state: ESTAB
connection variables:
seq # client-to-server
server-to-client
rcvBuffer size
at server,client

network

application

connection state: ESTAB
connection Variables:
seq # client-to-server
server-to-client
rcvBuffer size
at server,client

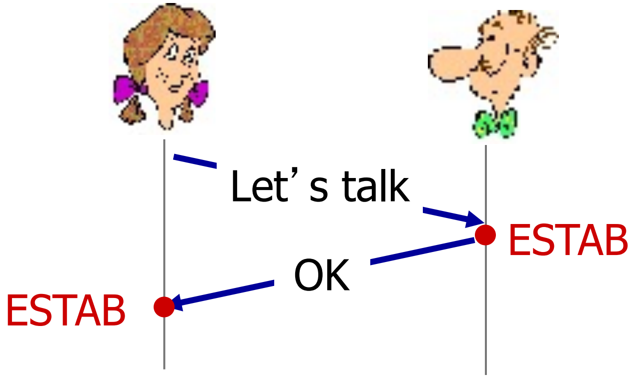
network

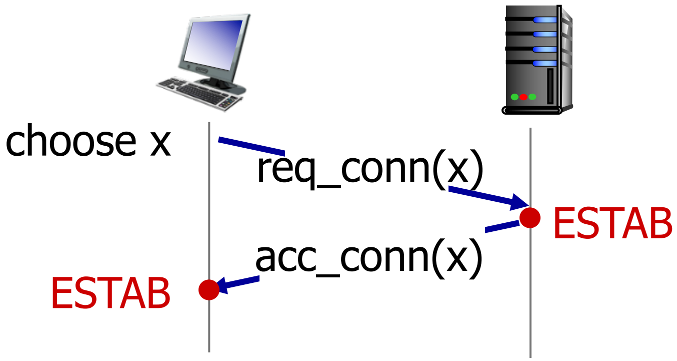
```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

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

Agreeing to establish a connection

2-way handshake:





Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can’t “see” other side

Three way handshake:

Step 1: client host sends TCP SYN segment to server.

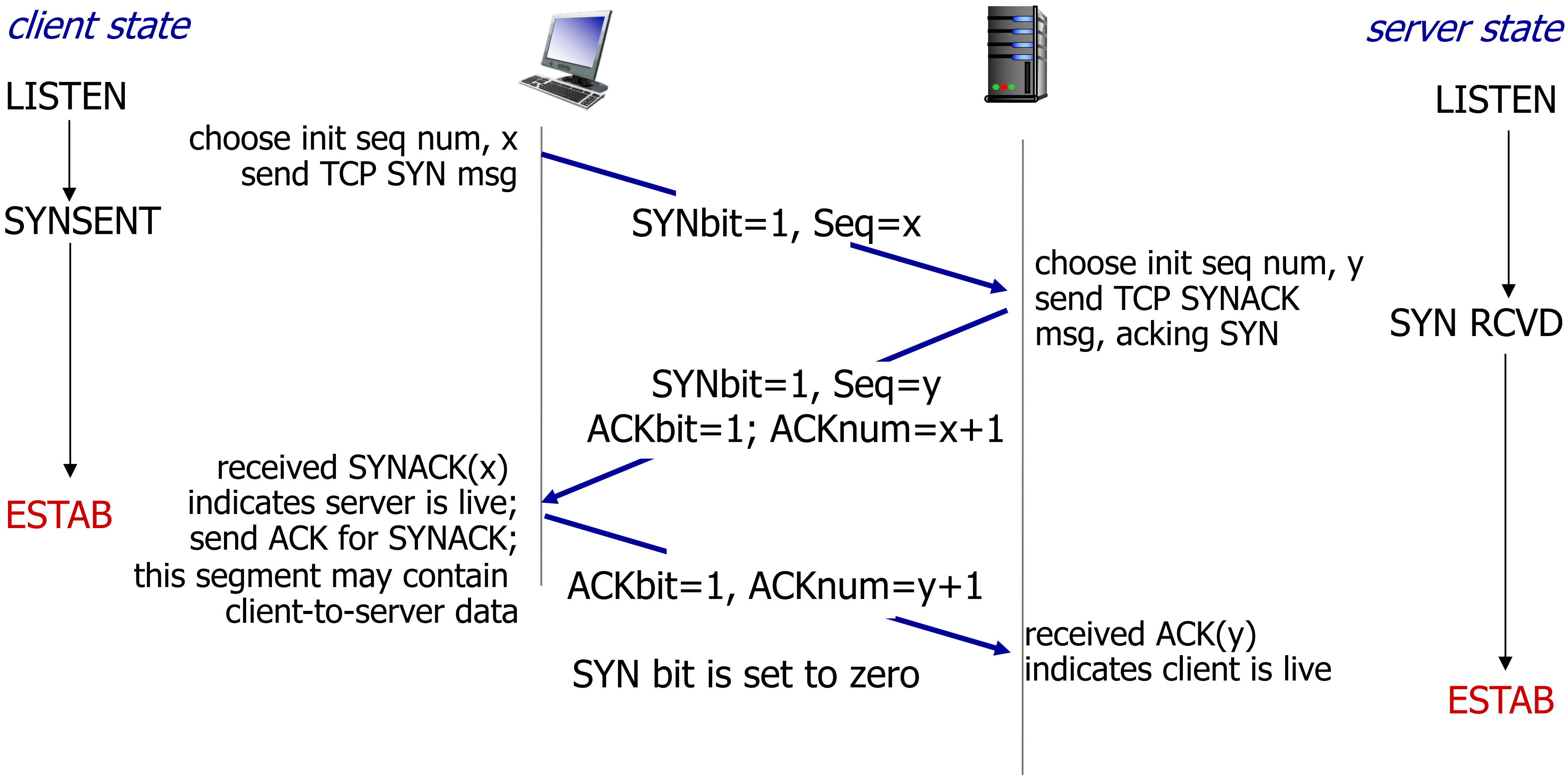
- SYN bit is set to 1
- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment

- SYN bit is set to 1
- server allocates buffers
- specifies server initial seq. #
- no data

Step 3: client receives SYNACK, replies with ACK segment, which may contain data. SYN bit is set to 0. Client also allocate buffers

TCP 3-way handshake



client state

LISTEN

SYNSENT

ESTAB

server state

LISTEN

SYN RCVD

ESTAB

choose init seq num, x
send TCP SYN msg

SYNbit=1, Seq=x

choose init seq num, y
send TCP SYNACK
msg, acking SYN

SYNbit=1, Seq=y
ACKbit=1; ACKnum=x+1

received SYNACK(x)
indicates server is live;
send ACK for SYNACK;
this segment may contain
client-to-server data

ACKbit=1, ACKnum=y+1

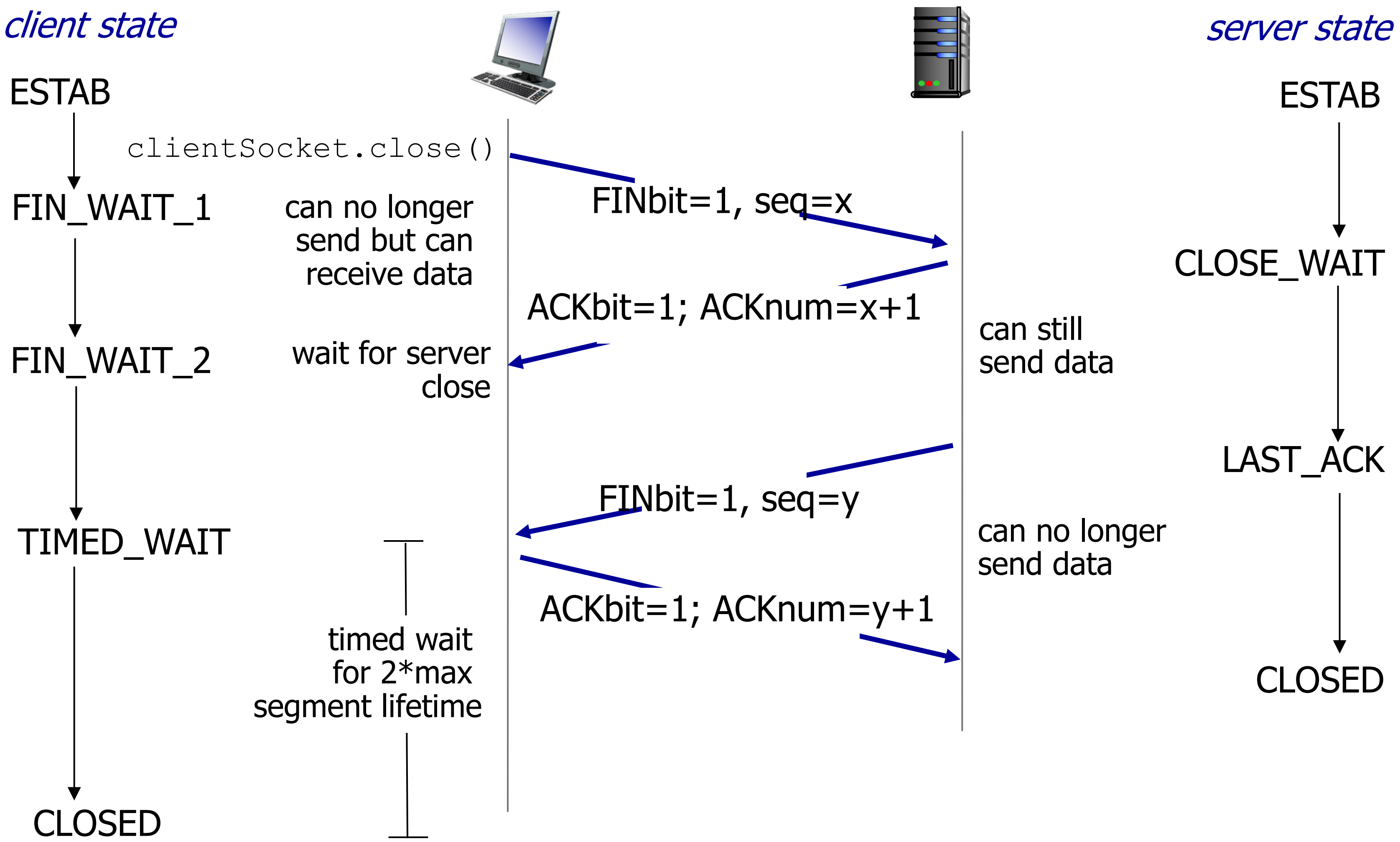
SYN bit is set to zero

received ACK(y)
indicates client is live

TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



client state

server state

ESTAB

`clientSocket.close()`

FIN_WAIT_1

can no longer send but can receive data

FINbit=1, seq=x

ESTAB

CLOSE_WAIT

ACKbit=1; ACKnum=x+1

FIN_WAIT_2

wait for server close

can still send data

LAST_ACK

FINbit=1, seq=y

TIMED_WAIT

can no longer send data

ACKbit=1; ACKnum=y+1

timed wait for 2*max segment lifetime

CLOSED

CLOSED

RST flag

- Suppose a host receives a TCP SYN packet with destination port 80 but host does not accept connection on port 80.
- Host will send a reset segment to client (set RST to 1)
- By this, host says “ I don’t have a socket for that segment”

Sending SYN could have three possible outcomes

- SYNACK is received
- RST is received
- Nothing received which means that SYN segment was blocked by firewall and never reached the target host

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

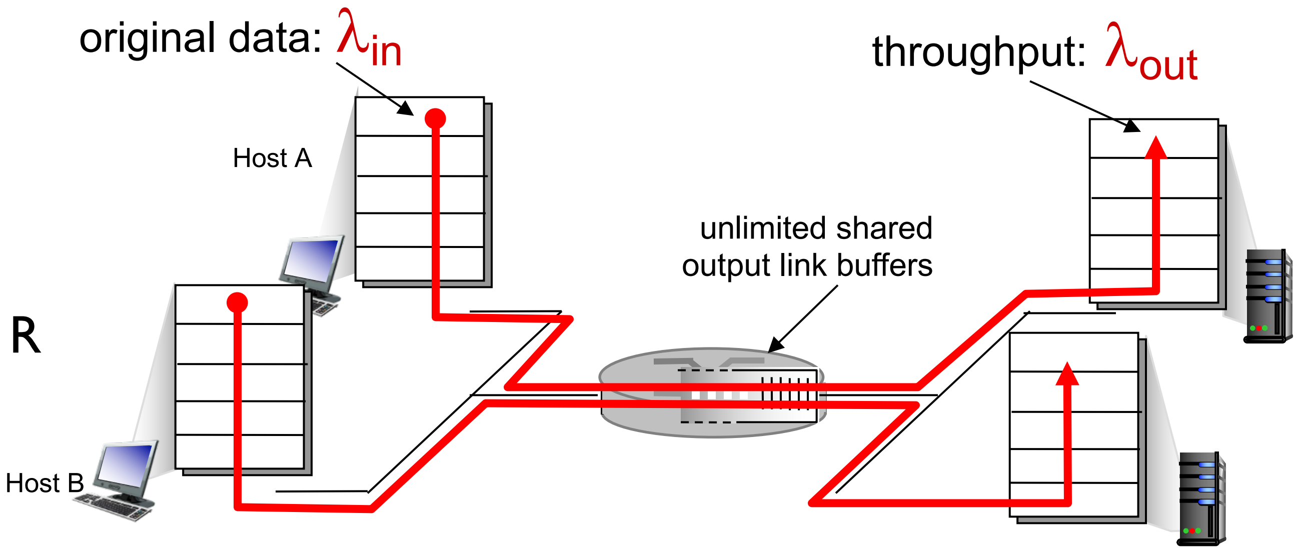
Principles of congestion control

congestion:

- What is congestion?
- –Load is higher than capacity
- What do IP routers do? –Drop the excess packets
- 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!

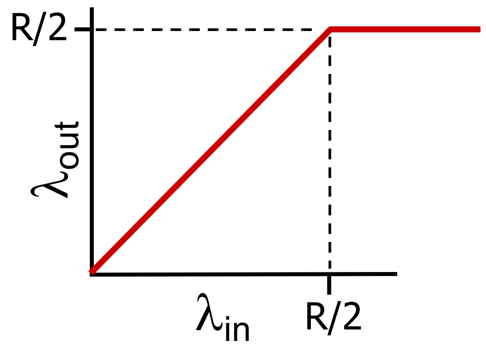
Causes/costs of congestion: scenario 1

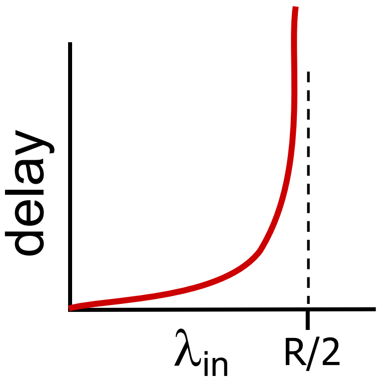
- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



original data: λ_{in}

throughput: λ_{out}

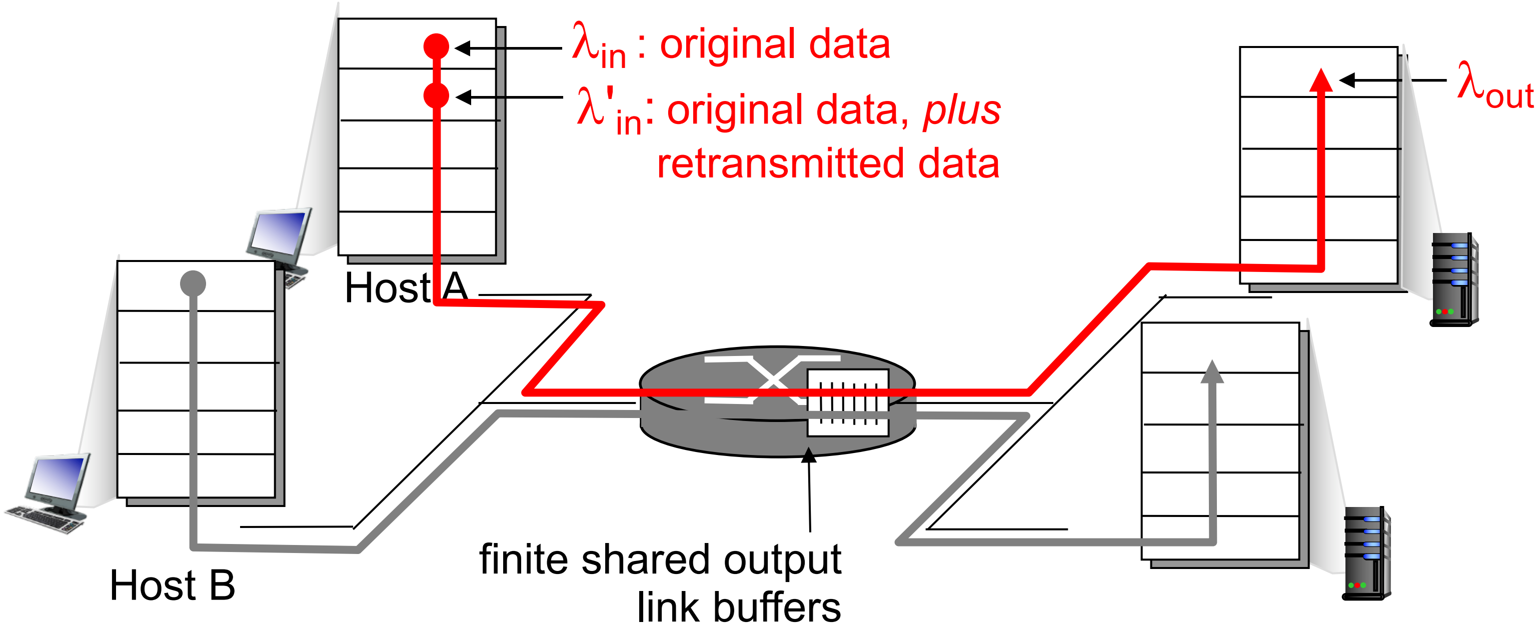




- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

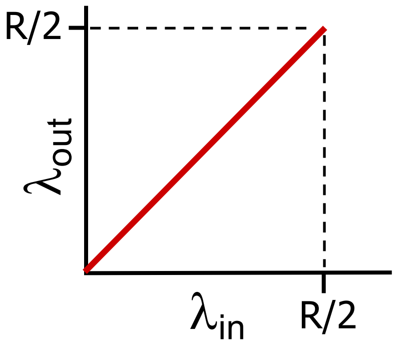


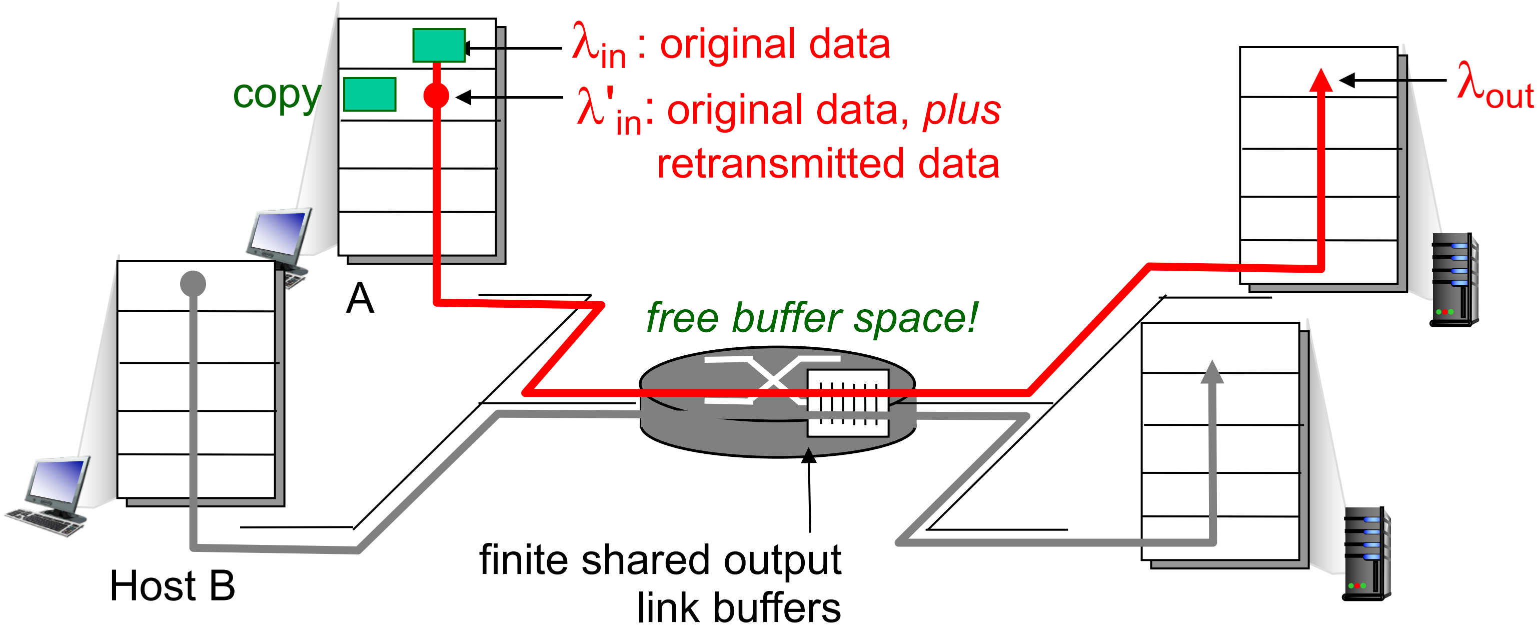
λ_{in} : original data
λ'_{in}: original data, *plus* retransmitted data
λ_{out}
Host A
Host B
finite shared output link buffers

Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

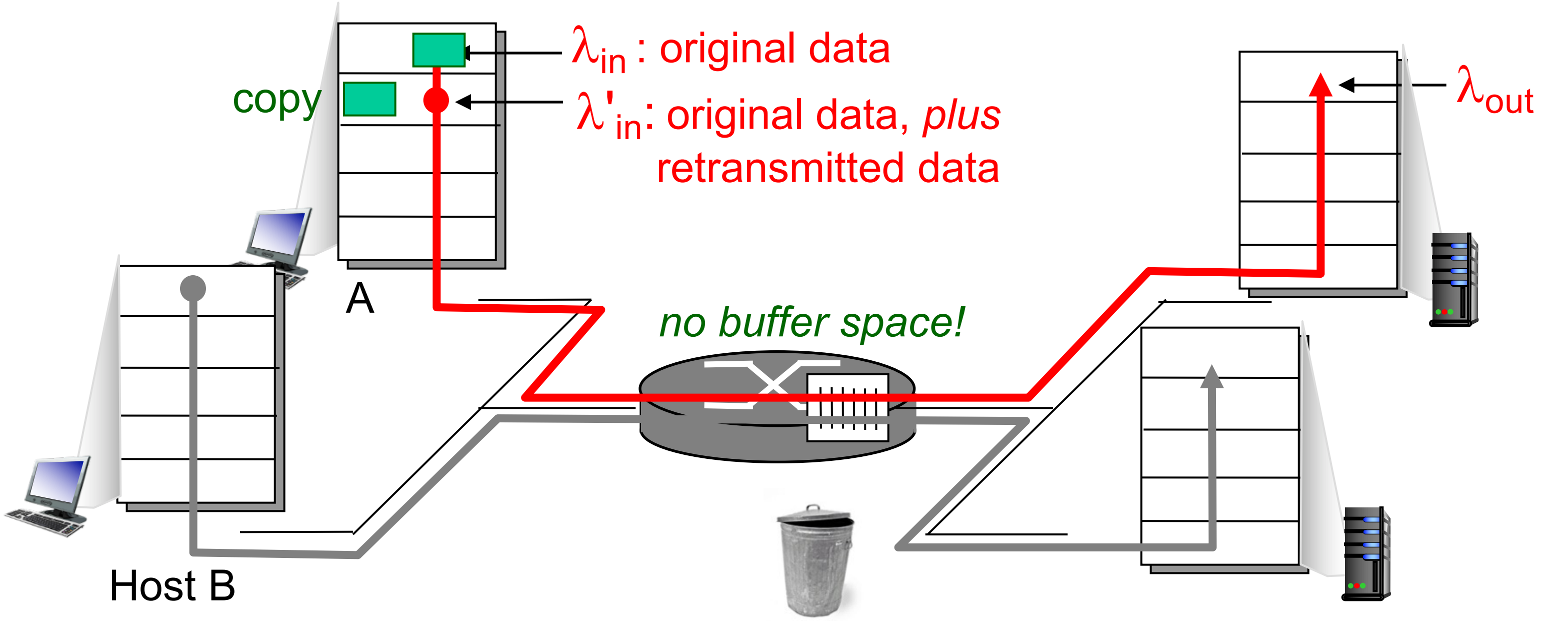



copy
λ_{in} : original data
λ'_{in}: original data, *plus* retransmitted data
λ_{out}
A
free buffer space!
Host B
finite shared output link buffers

Causes/costs of congestion: scenario 2

Idealization: known loss packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



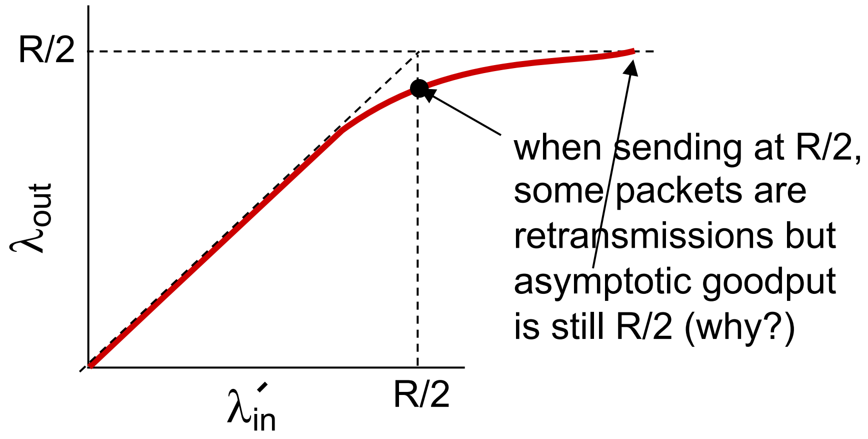
λ_{in} : original data
copy
λ'_{in}: original data, *plus* retransmitted data
λ_{out}
A
no buffer space!
Host B

Causes/costs of congestion: scenario 2

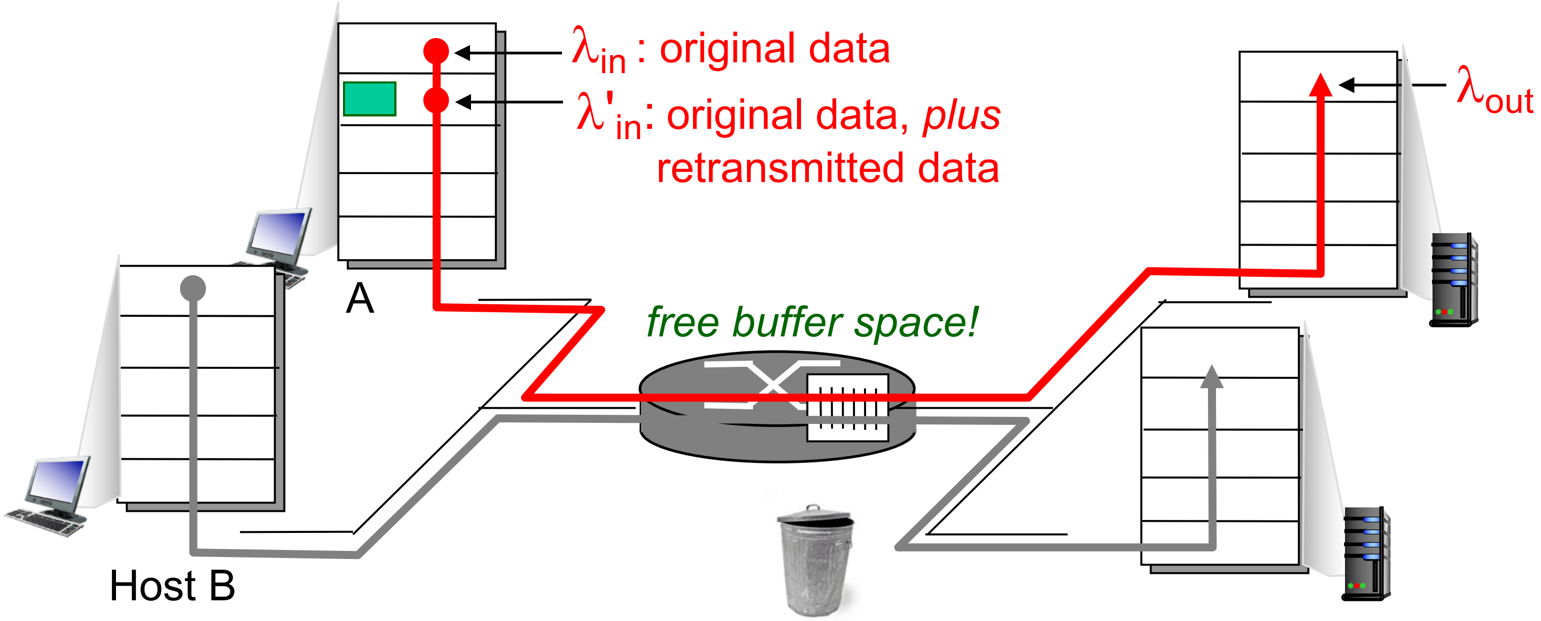
Idealization: known loss

packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



when sending at R/2, some packets are retransmissions but asymptotic goodput is still R/2 (why?)



λ_{in} : original data

λ'_{in}: original data, *plus* retransmitted data

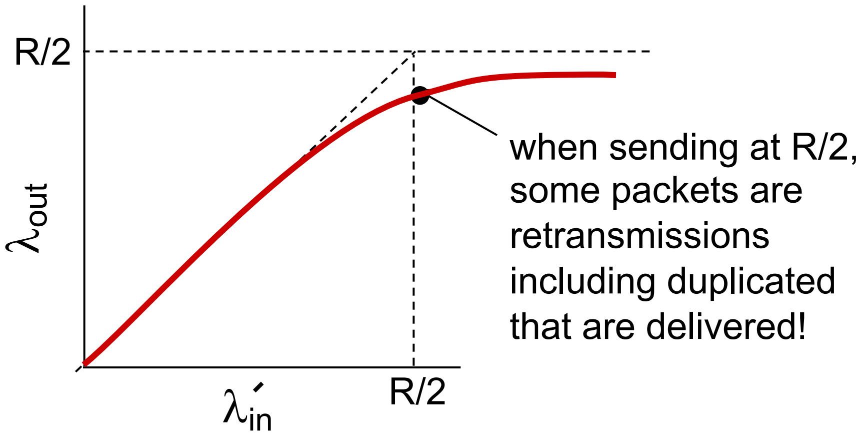
λ_{out}

free buffer space!

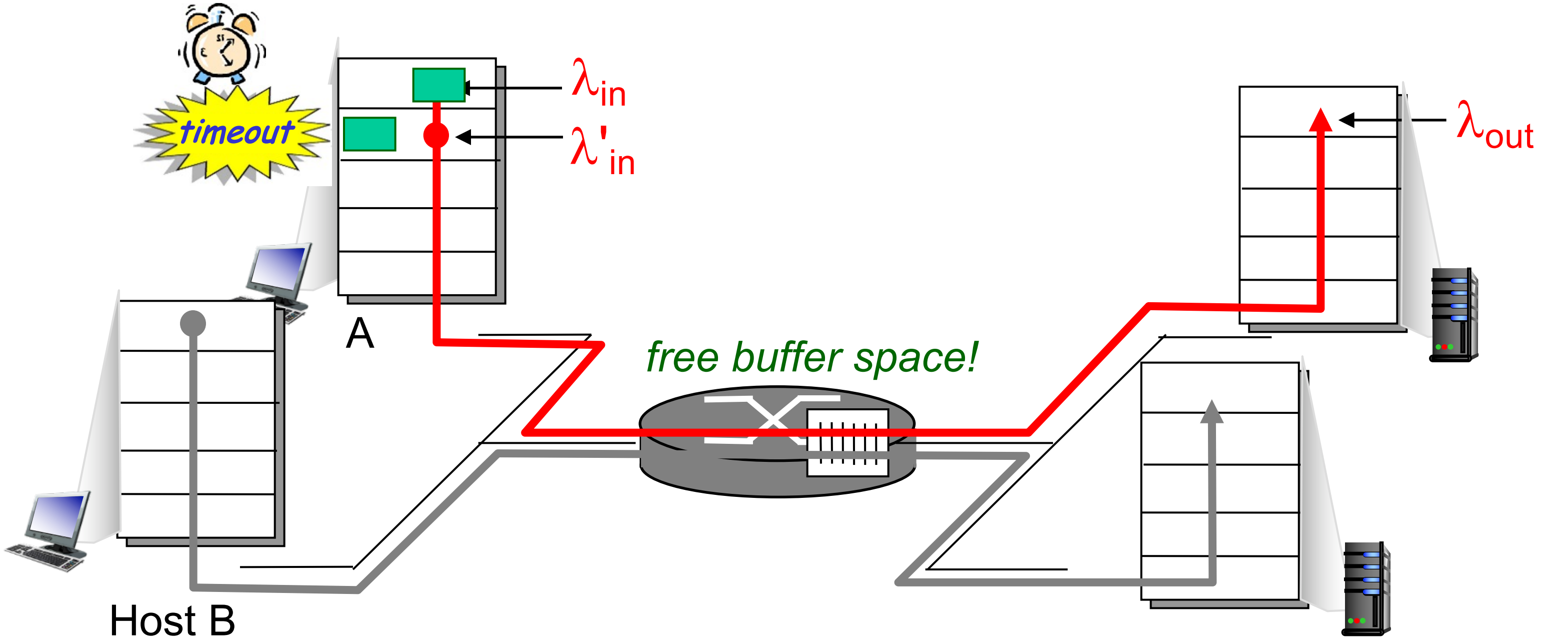
Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



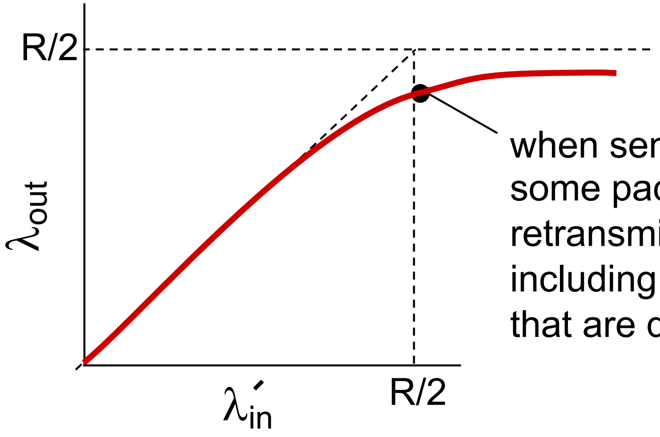
when sending at R/2, some packets are retransmissions including duplicated that are delivered!



Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



when sending at R/2, some packets are retransmissions including duplicated that are delivered!

"costs" of congestion:

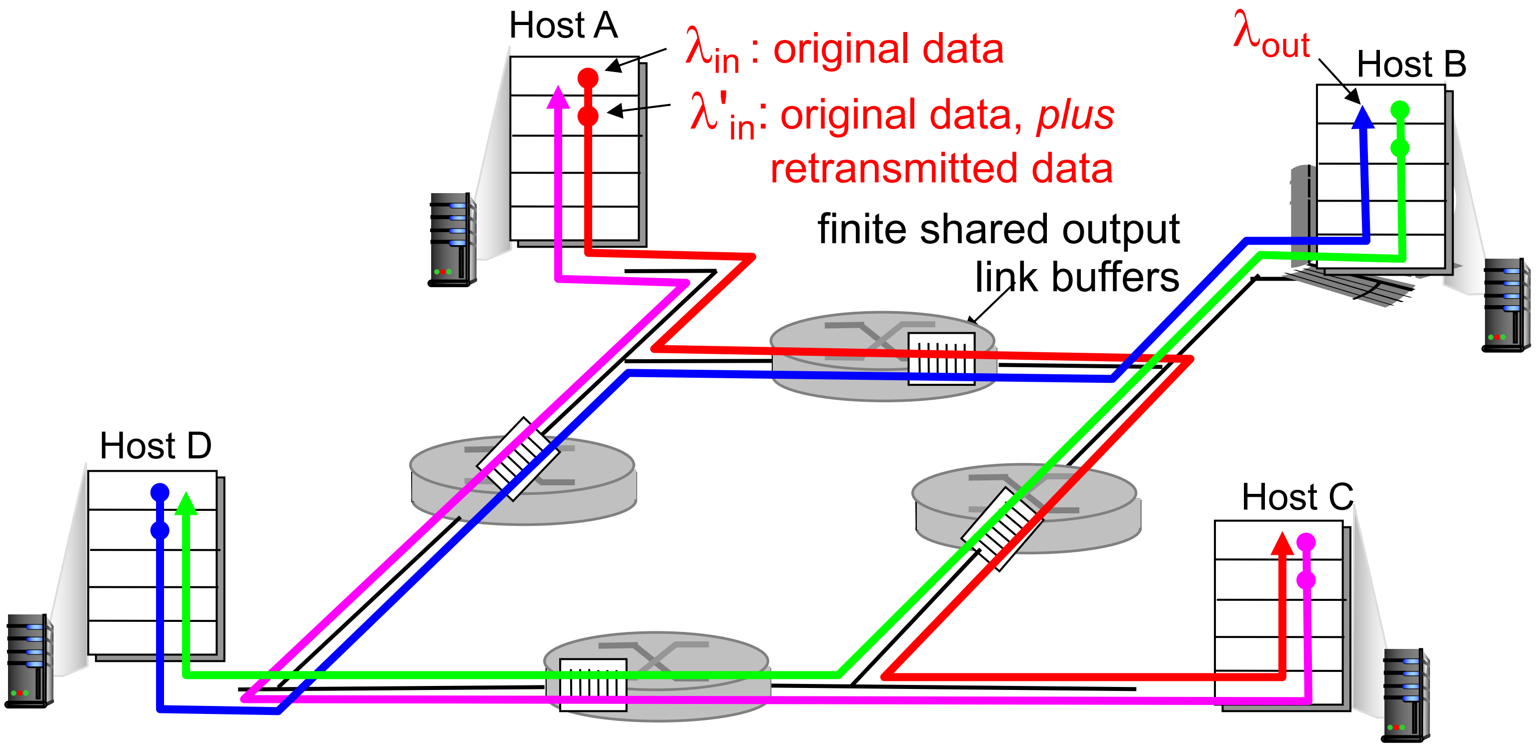
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

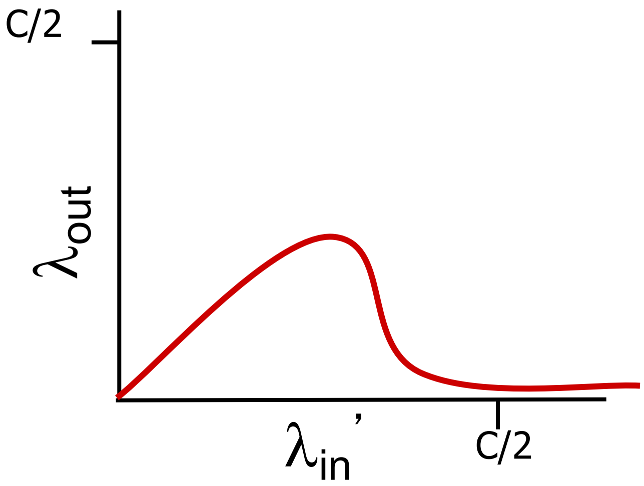
Q: what happens as λ_{in} and λ_{in}' increase ?

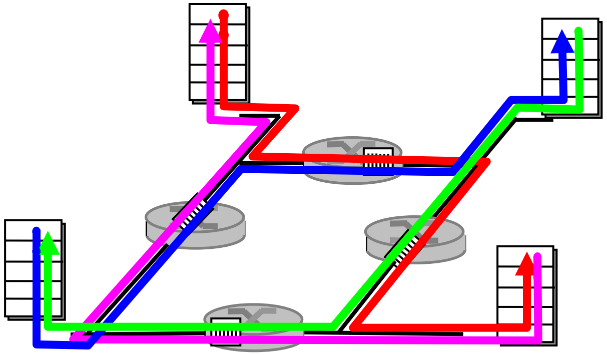
A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 0



Host A
λ_{in} : original data
λ'_{in}: original data, *plus* retransmitted data
finite shared output link buffers
λ_{out}
Host B
Host D
Host C

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 layer to transport layer
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:

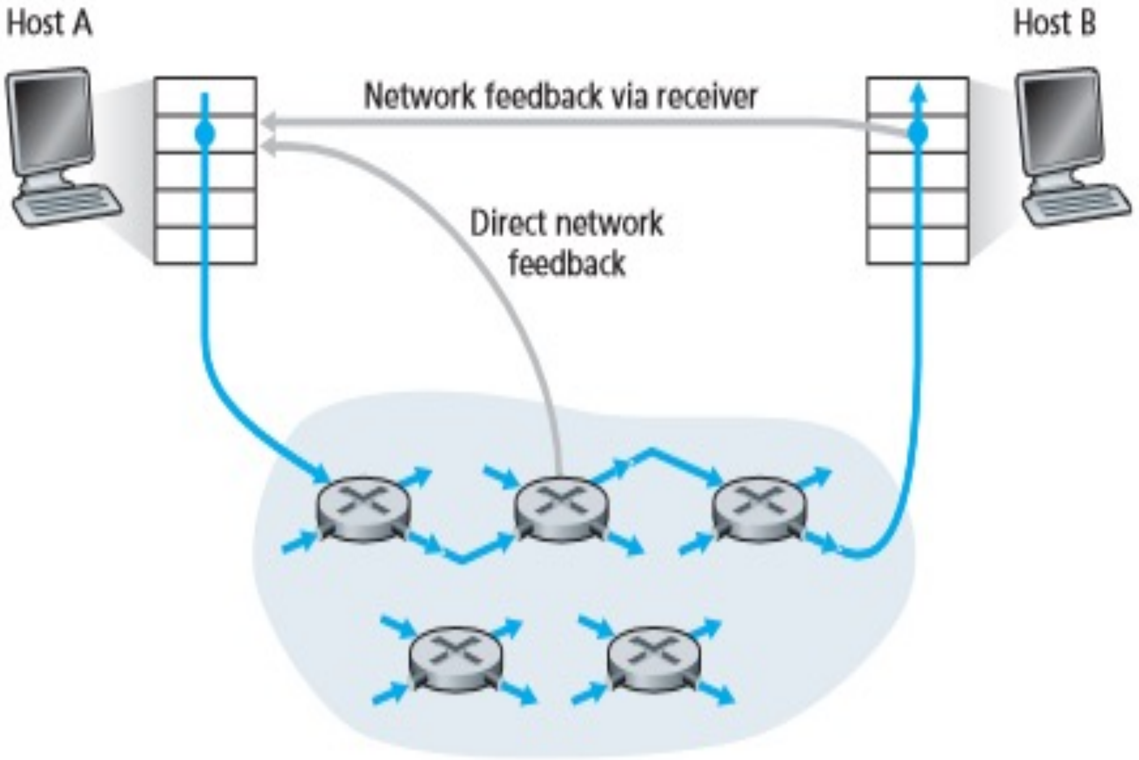
- routers provide feedback to end systems
  - Feedback could be a single bit indicating congestion Examples: TCP/IP ECN, ATM ABR)
  - explicit rate for sender to send at

Two ways of feedback from the network to the sender

1. Direct feedback: sent from a network router to the sender, essentially saying, “I’m congested!”.
2. Router marks/updates a field in a packet flowing from sender to receiver to indicate congestion.

- Upon receipt of a marked packet, the receiver notifies the sender of the congestion indication.
- Note: this form of notification takes at least a full round-trip time.

Two ways of feedback from the network to the sender



ATM (**Asynchronous Transfer Mode**

- ATM takes virtual circuit (VC) approach toward packet switching. This means that each switch on the source-to-destination path will maintain state about VC
- Cell used instead of Packet
- Switch used instead of Router
- Resource-management (RM) cells are used to send congestion-related information among the hosts and switches

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 limited to minimum guaranteed rate

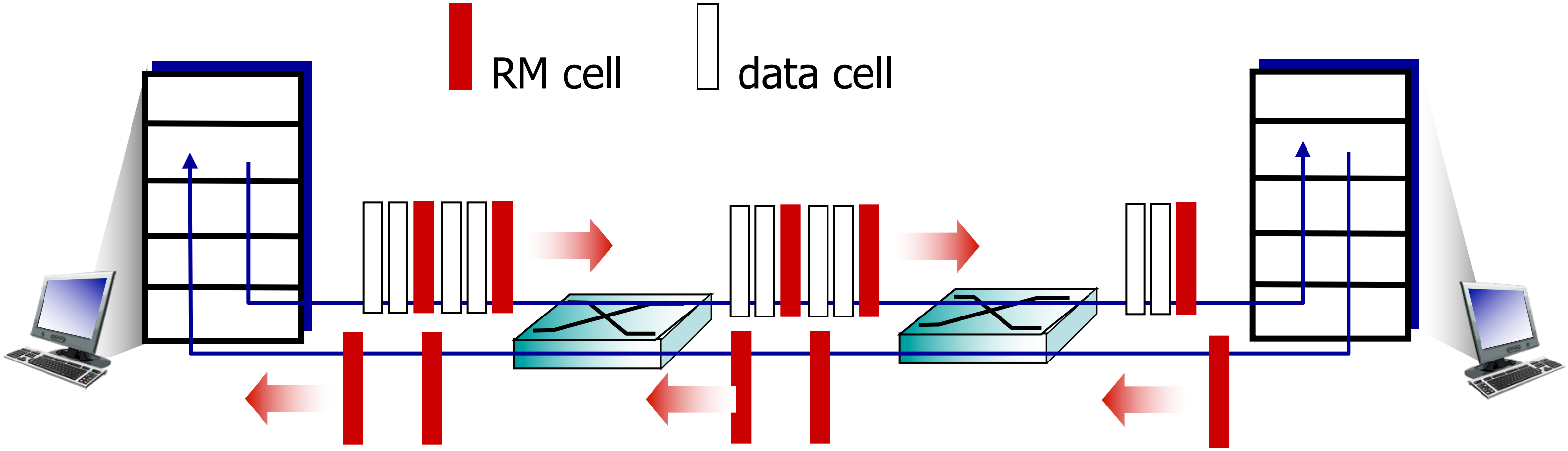
RM (resource management) cells:

- sent by sender, between data cells
- RM cells can be used to provide both direct network feedback and network feedback via the receiver
- bits in RM cell set by switches ("*network-assisted*")
- RM cells returned to sender by receiver, with bits intact

ABR provides three mechanisms for signaling congestion

1. EFCI bit. Each data cell contains an explicit forward congestion indication (EFCI) bit. A congested network switch can set EFCI bit in a data cell to 1. The destination host checks the EFCI bit. When an RM cell arrives at the destination, then the destination sets the congestion indication bit (the CI bit) of the RM cell to 1 and sends the RM cell back to the sender
2. CI and NI bits: RM cells have a congestion indication (CI) bit (severe congestion) and a no increase (NI) bit (mild congestion) in RM cells
3. ER setting. Each RM cell also contains a 2-byte explicit rate (ER) field. A congested switch lower the value contained in the ER field in a passing RM cell

Case study: ATM ABR congestion control



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

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3.7 TCP congestion control

TCP congestion control

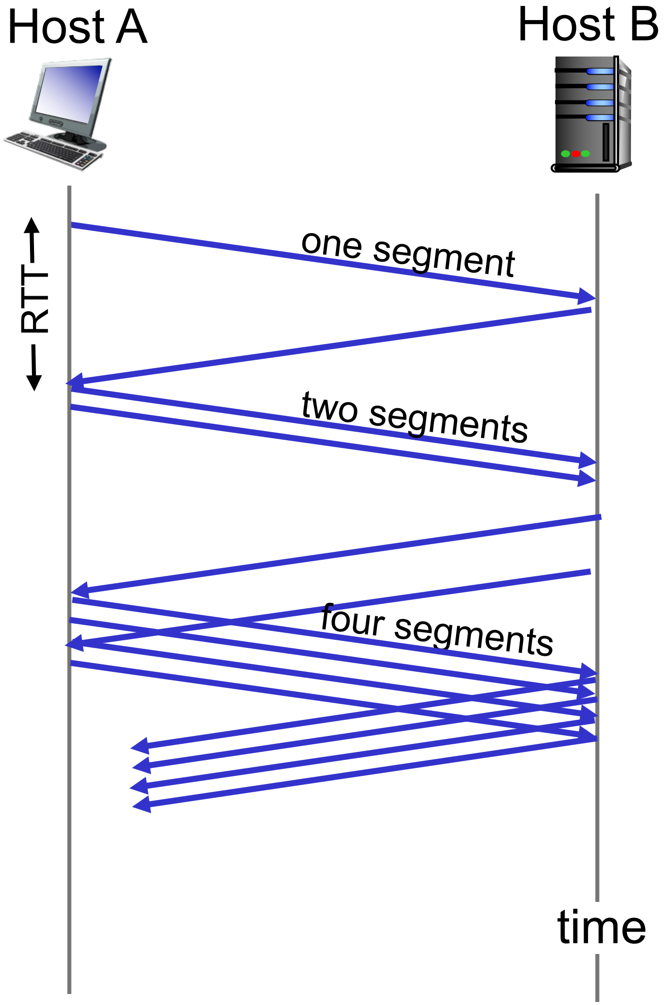
- sender keeps track of an, congestion window (cwnd) the amount of unacknowledged data at a sender may not exceed the minimum of cwnd and rwnd, that is:

$$\texttt{LastByteSent} - \texttt{LastByteAcked} \leq \texttt{min\{cwnd, rwnd\}}$$

- Assuming TCP receive buffer is very large, then the sender's send rate is roughly cwnd/RTT bytes/sec
- TCP detects congestion if "lost event". Lost is detected using timeout and Duplicate ACK.

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast



Slow start after loss event

- if there is a loss event (i.e., congestion) indicated by a timeout, the TCP sender sets the value of cwnd to 1 and sets ssthresh to cwnd/2
- when the value of cwnd equals ssthresh, slow start ends and TCP transitions into congestion avoidance mode
- slow start can end if three duplicate ACKs are detected, in which case TCP performs a fast retransmit and enters fast recovery state.

Congestion Avoidance

- Rather than doubling cwnd every RTT, TCP increases cwnd by just a single MSS every RTT
- TCP's congestion-avoidance algorithm behaves the same as in the case of slow start when a timeout occurs.
- But for the case of loss detection using a triple duplicate ACK, TCP halves the value of cwnd and enter fast-recovery

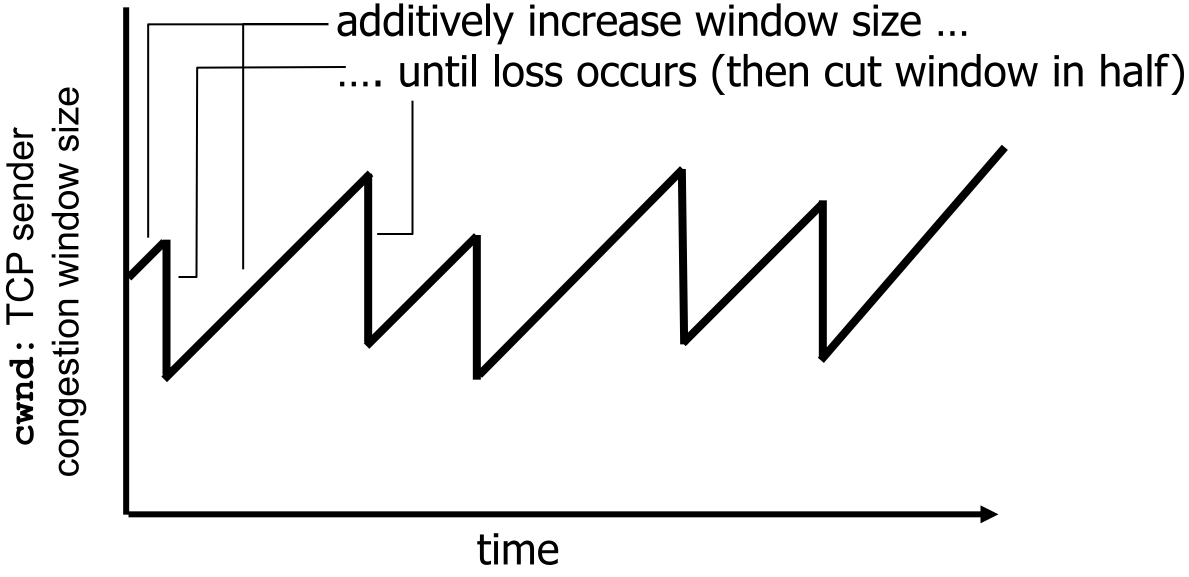
Fast Recovery

- Set SSTHRESH=CWND/2
- Set CWND=SSTHRESH+3 MSS
- Every subsequent duplicate ack: CWND=CWND+1MSS
- When a new ACK arrives (not a duplicate ack), TCP enters the congestion-avoidance state
- If a timeout event occurs, fast recovery transitions to the slow-start state

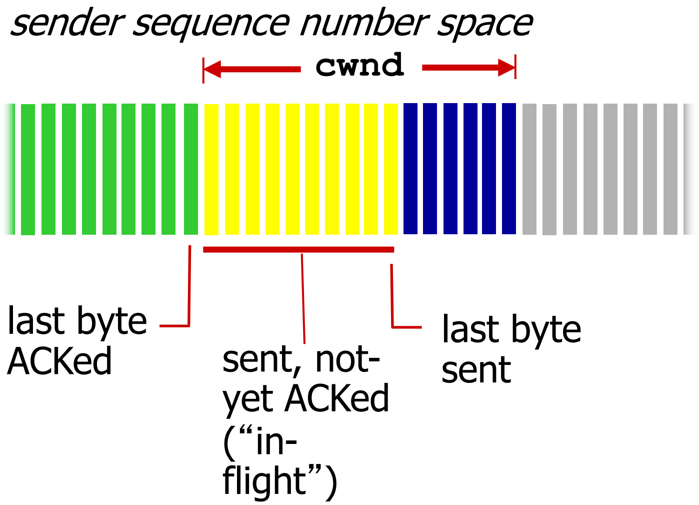
TCP congestion control: additive increase multiplicative decrease

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details


sender sequence number space
cwnd
last byte ACKed
sent, not-yet ACKed ("in-flight")
last byte sent

- sender limits transmission:

```
  LastByteSent-
    LastByteAcked ≤  cwnd
```

- cwnd is dynamic, function of perceived network congestion

TCP sending rate:

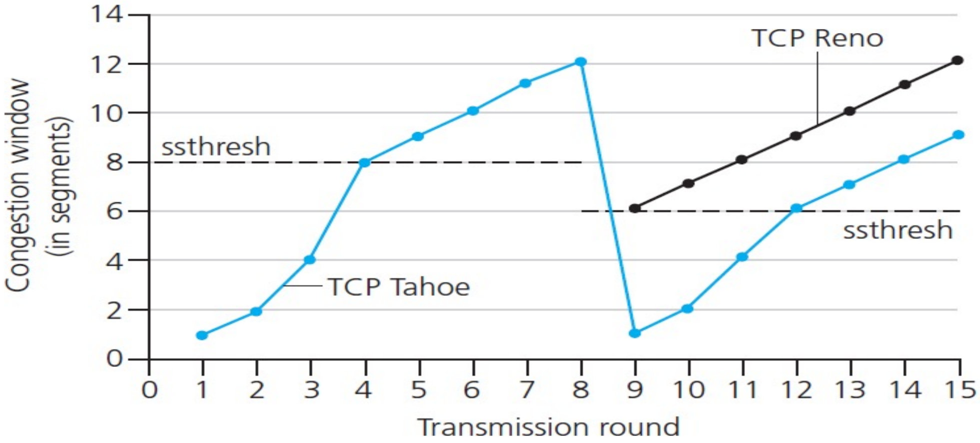
- *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

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

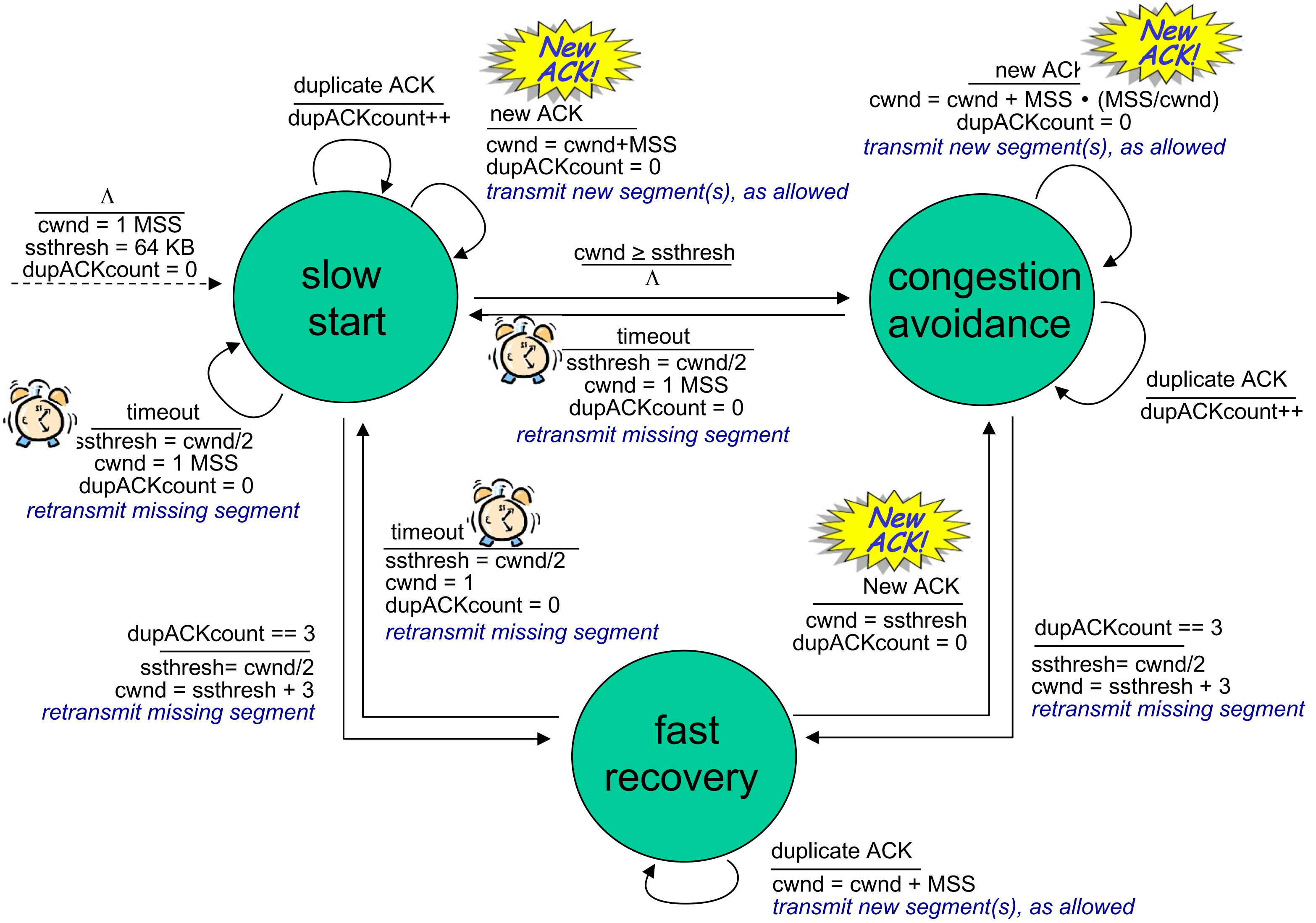
TCP Taho and TCP Reno

TCP Tahoe, unconditionally cut its congestion window to 1 MSS and entered the slow-start phase after either a timeout or triple-duplicate-ACK.

TCP Reno, incorporated fast recovery



Summary: TCP Congestion Control



Λ
cwnd = 1 MSS
ssthresh = 64 KB
dupACKcount = 0

duplicate ACK
dupACKcount++

New ACK!

new ACK
cwnd = cwnd+MSS
dupACKcount = 0
transmit new segment(s), as allowed

slow start

cwnd ≥ ssthresh
Λ

congestion avoidance

New ACK!

new ACK
cwnd = cwnd + MSS • (MSS/cwnd)
dupACKcount = 0
transmit new segment(s), as allowed

timeout
ssthresh = cwnd/2
cwnd = 1 MSS
dupACKcount = 0
retransmit missing segment

duplicate ACK
dupACKcount++

timeout
ssthresh = cwnd/2
cwnd = 1 MSS
dupACKcount = 0
retransmit missing segment

timeout
ssthresh = cwnd/2
cwnd = 1
dupACKcount = 0
retransmit missing segment

New ACK!

New ACK
cwnd = ssthresh
dupACKcount = 0

dupACKcount == 3
ssthresh= cwnd/2
cwnd = ssthresh + 3
retransmit missing segment

dupACKcount == 3
ssthresh= cwnd/2
cwnd = ssthresh + 3
retransmit missing segment

fast recovery

duplicate ACK
cwnd = cwnd + MSS
transmit new segment(s), as allowed

Summary: TCP Congestion Control

- When **`CongWin`** is below **`Threshold`**, sender in slow-start phase, window grows exponentially.
- When **`CongWin`** is above **`Threshold`**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** set to **`Threshold`**.
- When timeout occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** is set to 1 MSS.

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

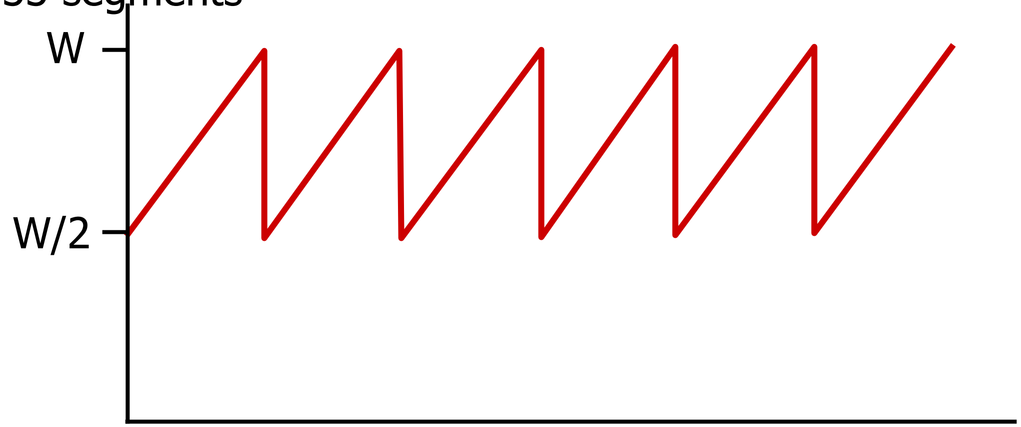
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP throughput

- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes)
  - avg. window size (# in-flight bytes) is ¾ W (Average of W/2 and W)
  - avg. thruput is 3/4W per RTT

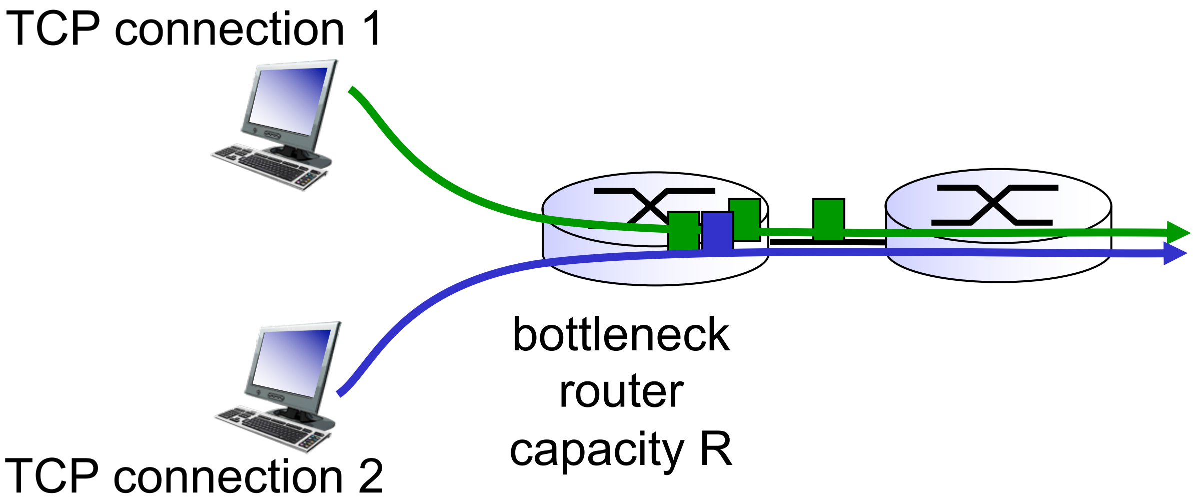
$$\text{avg TCP thruput} = \frac{3}{4}\,\frac{W}{RTT}\ \text{bytes/sec}$$

example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput requires W = 83,333 in-flight segments (Throughput is 1.25 GB, 1.25 GB /1500 =83,333 segments



TCP Fairness

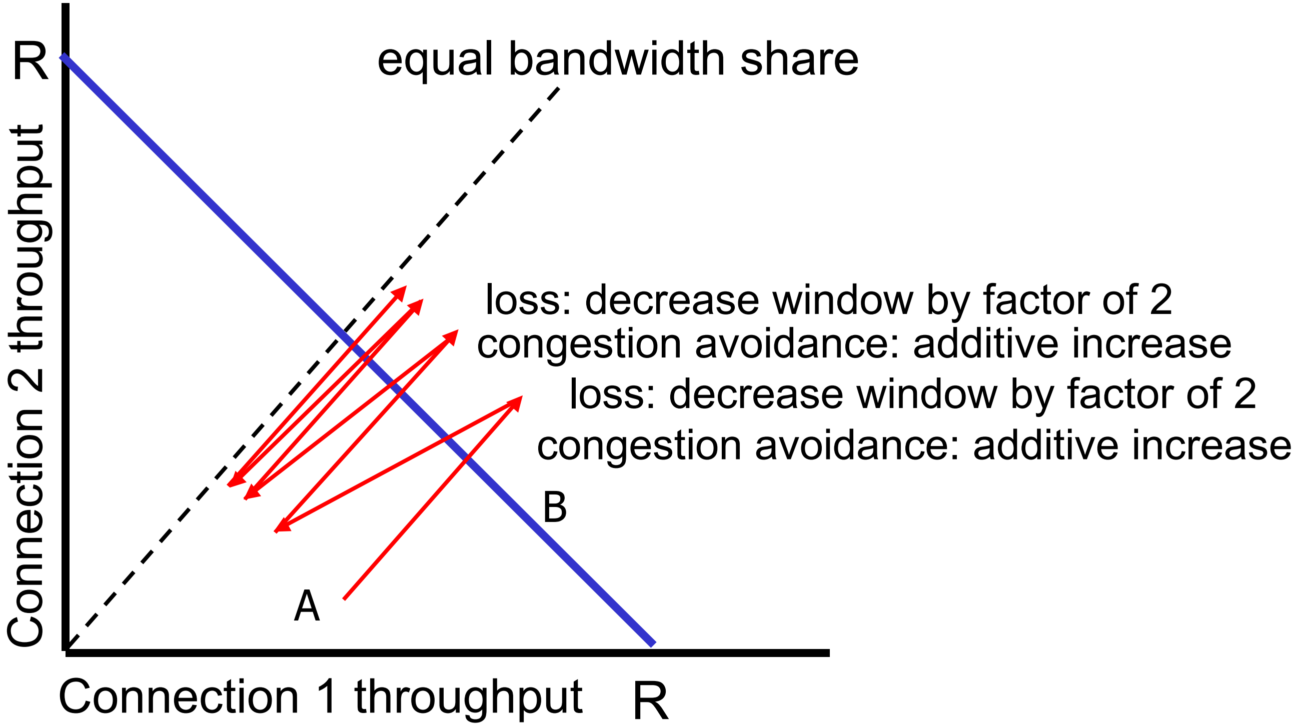
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

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


R
equal bandwidth share
loss: decrease window by factor of 2
congestion avoidance: additive increase
loss: decrease window by factor of 2
congestion avoidance: additive increase
B
A
Connection 2 throughput
Connection 1 throughput
R

- Suppose that the TCP window sizes are at point *A*. Because the amount of link bandwidth jointly consumed by the two connections is less than *R*, both connections will increase their window by 1 MSS.
- Throughput of the two connections proceeds along a 45-degree line
- Both connections decrease their
- windows by a factor of two (Point C).

Fairness (more)

Fairness and UDP

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

Fairness, parallel TCP connections

- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets more than R/2

Chapter 3: summary

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

next:

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