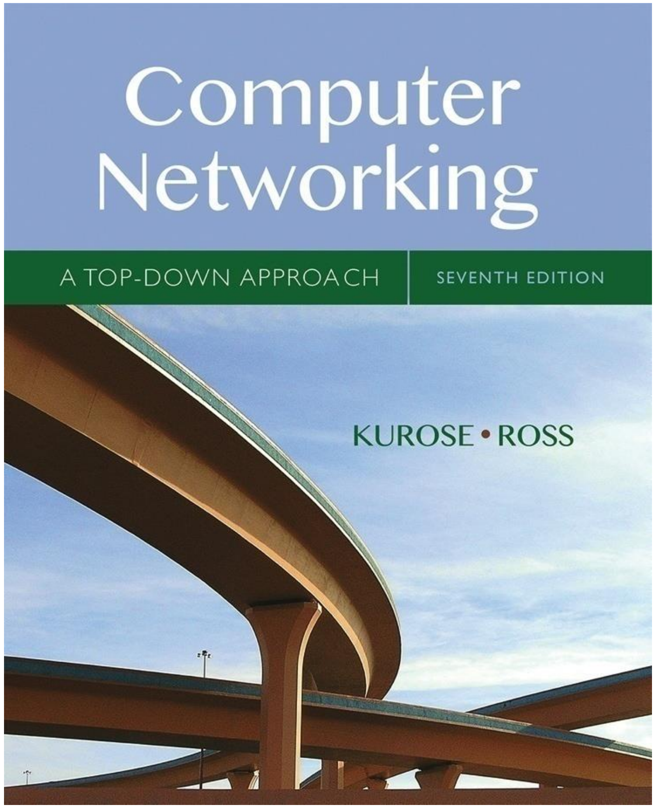
Chapter 3
Transport Layer



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Computer Networking: A Top Down Approach

7th edition
Jim Kurose, Keith Ross
Pearson/Addison Wesley
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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

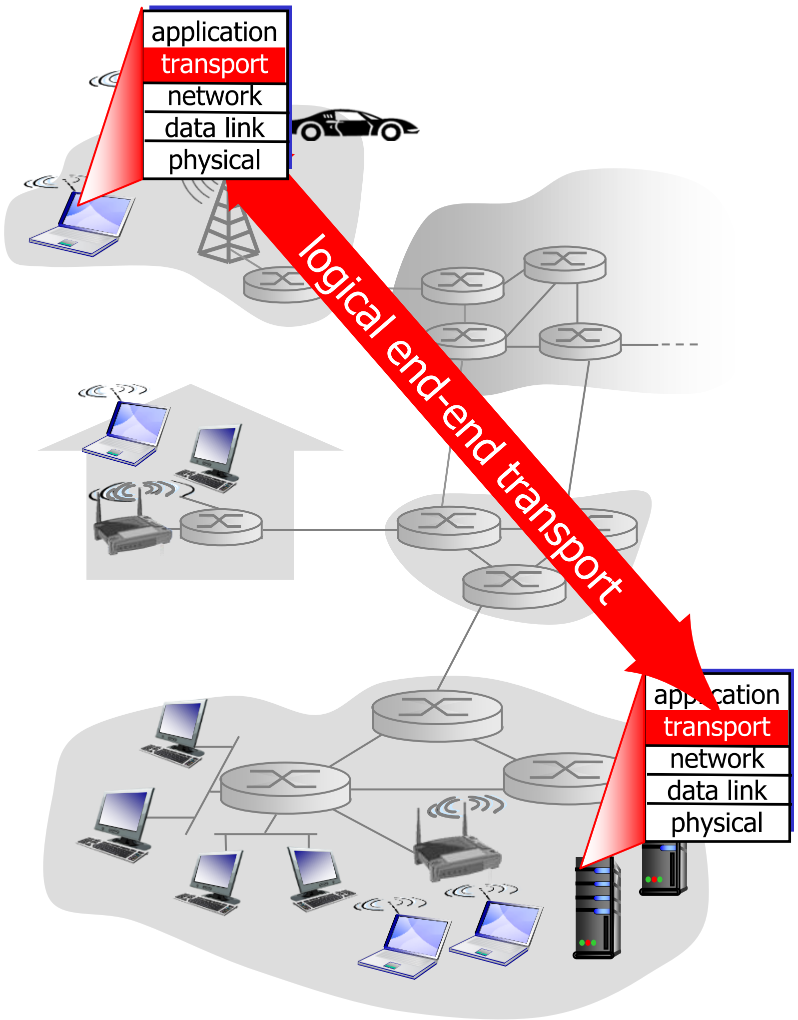
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

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
- 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

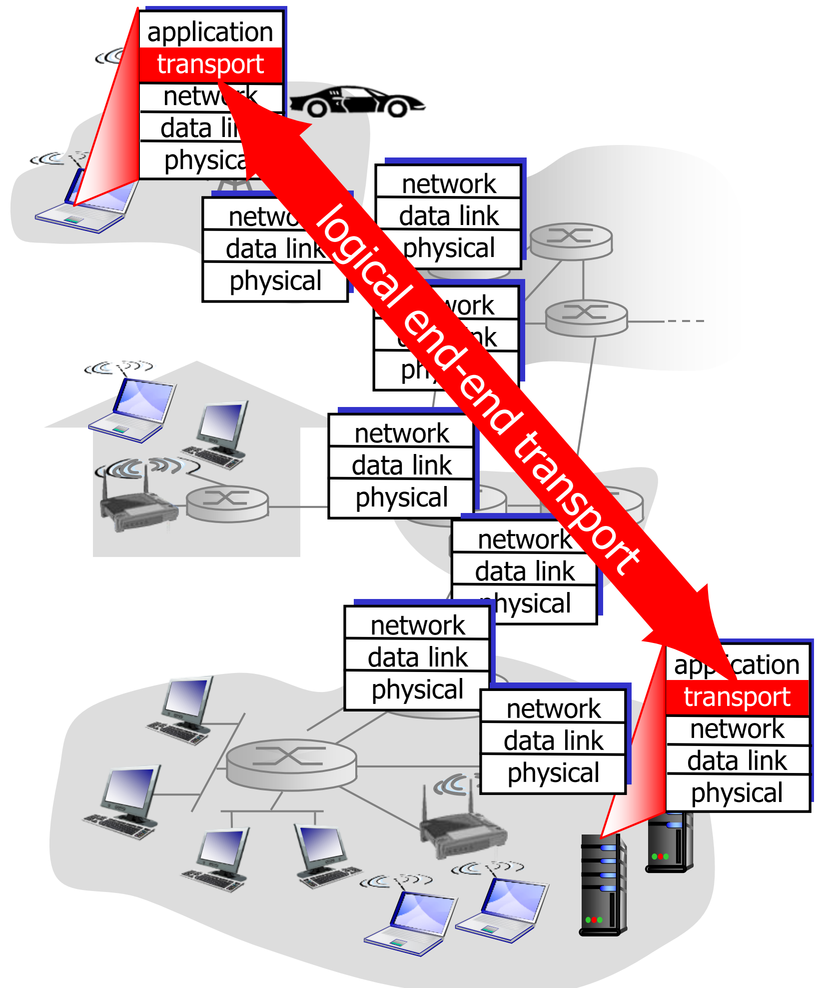
household analogy:

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（拥塞控制）
  - flow control（流控制）
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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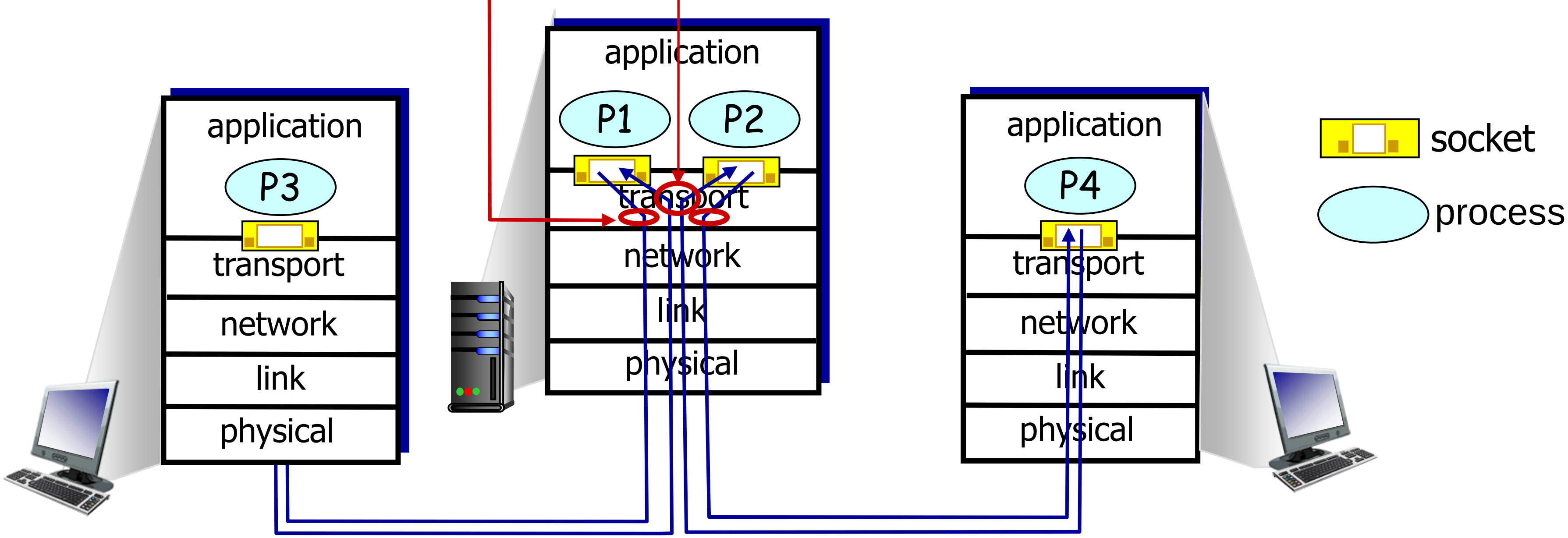
Multiplexing/demultiplexing

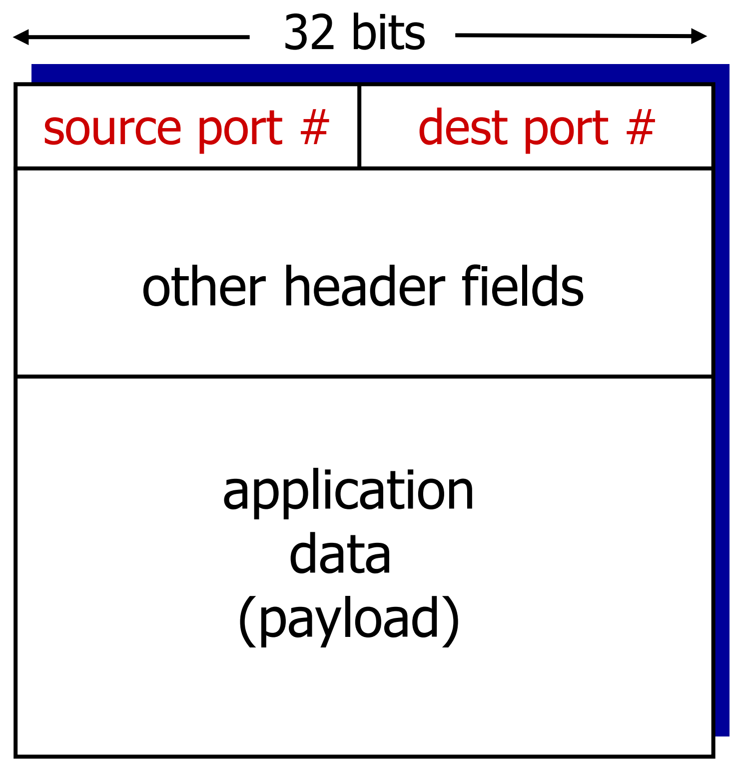
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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



TCP/UDP segment format

Connectionless demultiplexing

- *recall:* created socket has host-local port #:

```
sock.bind('localhost',10000);
```

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

```
clientSocket.sendto(message,(serverName, serverPort))
```

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

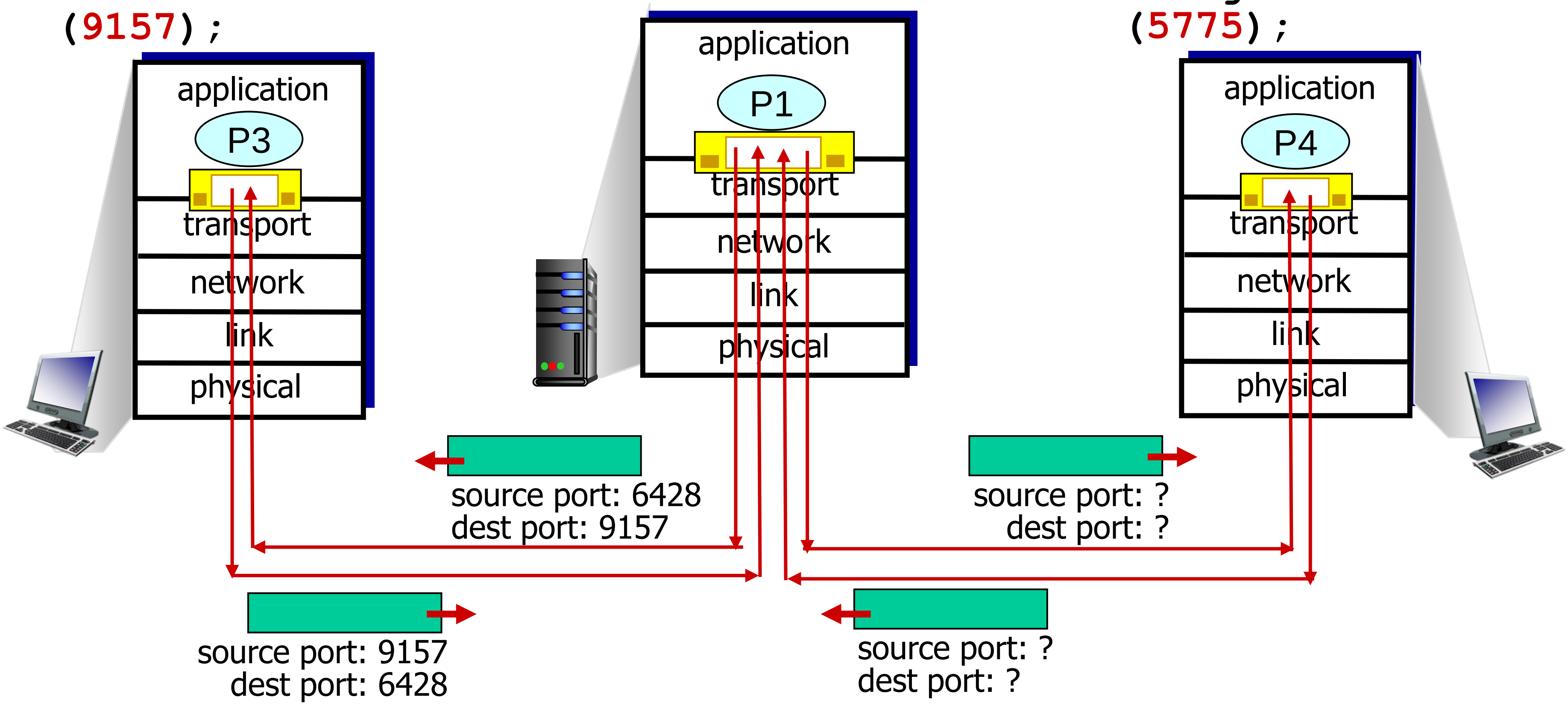
IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connectionless demux: example

```
DatagramSocket
 mySocket2 = new
 DatagramSocket
 (9157);
```

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

```
DatagramSocket
 mySocket1 = new
 DatagramSocket
 (5775);
```



Connection-oriented demux

- Reconsider TCP c/s program
  - Server: welcome socket on port 12000
  - Client: create a socket and send connection establish request (src IP+port)

```
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName,12000))
```

  - Server: create a new socket identified with src IP, src port, dst IP, and dst port

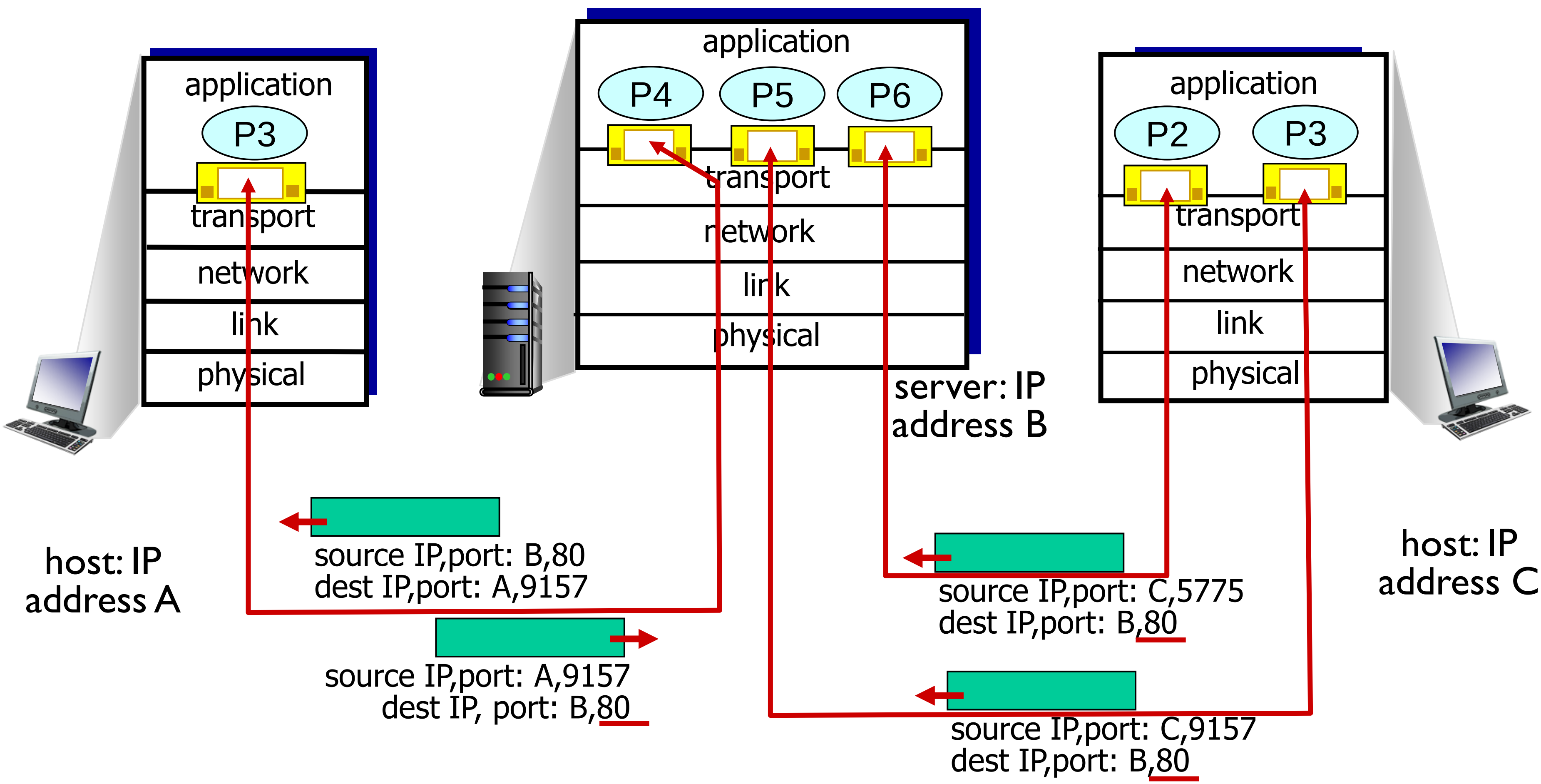
```
serverSocket.bind(('localhost',12000))
serverSocket.listen(1)
connectionSocket, addr = serverSocket.accept()
```

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

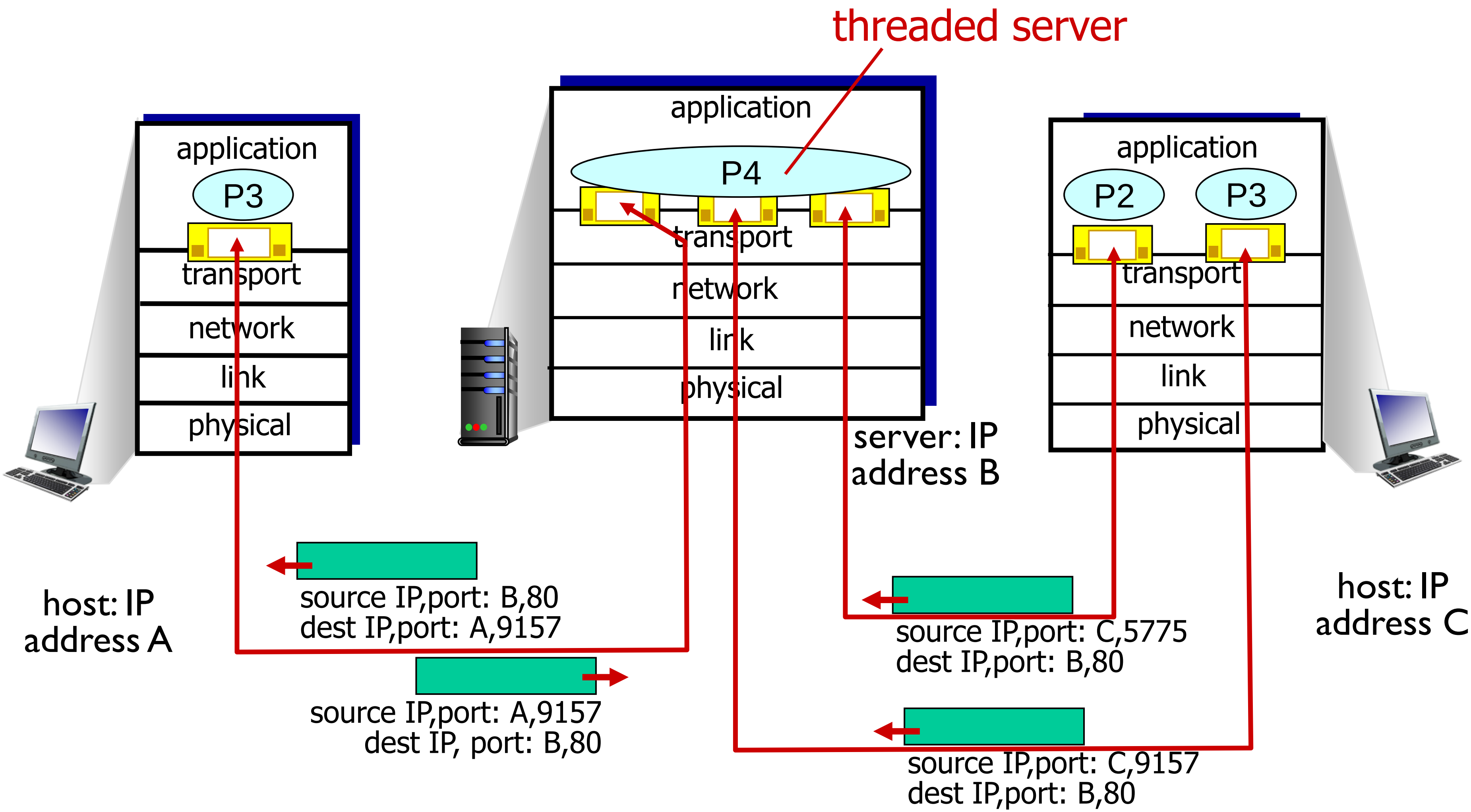
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
  - May have same server port
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each object request

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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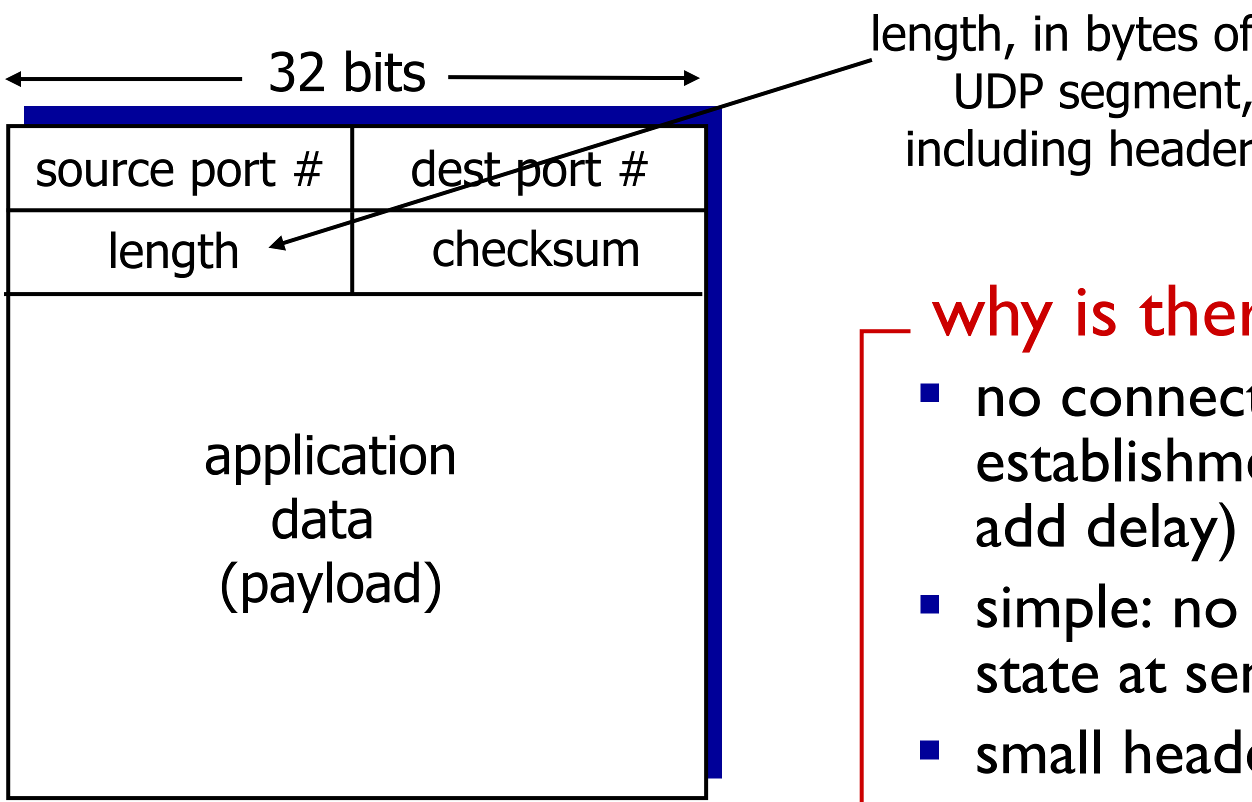
UDP: User Datagram Protocol [RFC 768]

- "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
  - SNMP
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

Application	Application-Layer Protocol	Underlying Transport Protocol
Electronic mail	SMTP	TCP
Remote terminal access	Telnet	TCP
Web	HTTP	TCP
File transfer	FTP	TCP
Remote file server	NFS	Typically UDP
Streaming multimedia	typically proprietary	UDP or TCP
Internet telephony	typically proprietary	UDP or TCP
Network management	SNMP	Typically UDP
Routing protocol	RIP	Typically UDP
Name translation	DNS	Typically UDP

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

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

checksum+sum=全1

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

求和、取补

* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

Pseudo header

- UDP computes a checksum that covers a part of IP header
  - By prepending a pseudo header to actual UDP header
  - Save IP header from error
- Pseudo header contains
  - Source and destination IP
  - Reserved field set as ‘0’s
  - Upper protocol ID = 17 (for UDP)
  - Lens

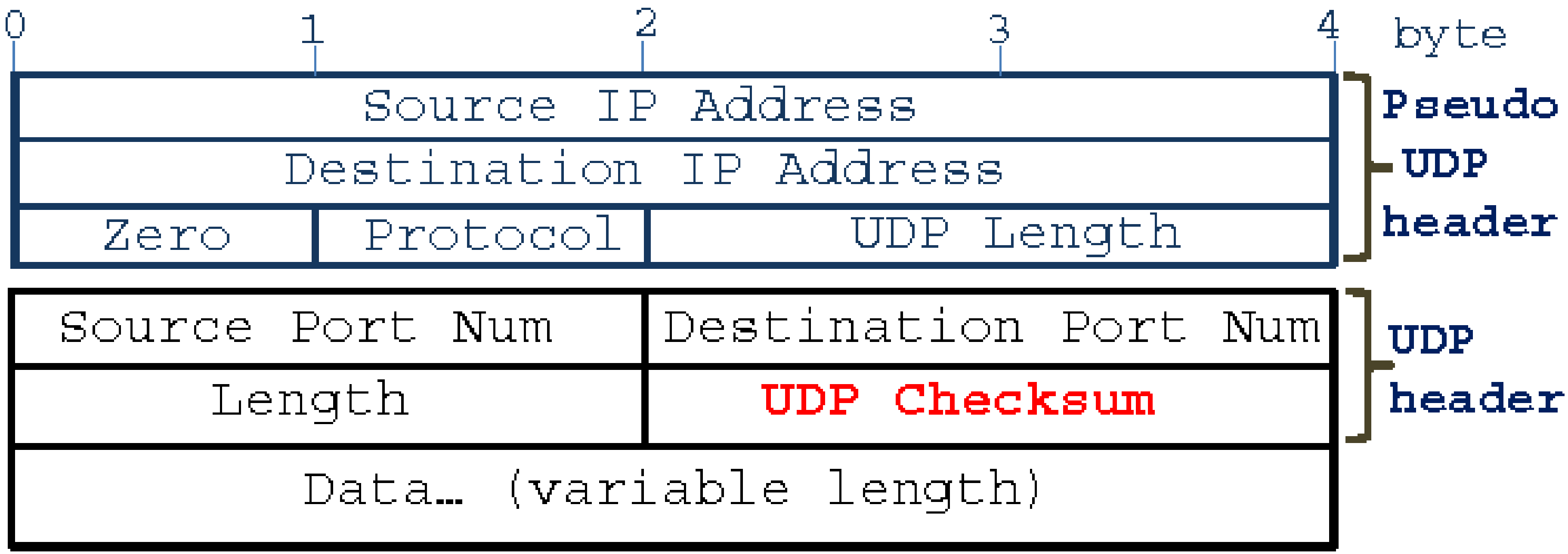


Fig. 1. UDP datagram and pseudo header.

UDP checksum

- Many link-layer protocol provide error-checking, then why checksum in UDP?
  - no guarantee that all the links between source and destination provide error checking;
  - error may happen when segment stored in a router's memory
- End-end principle in system design

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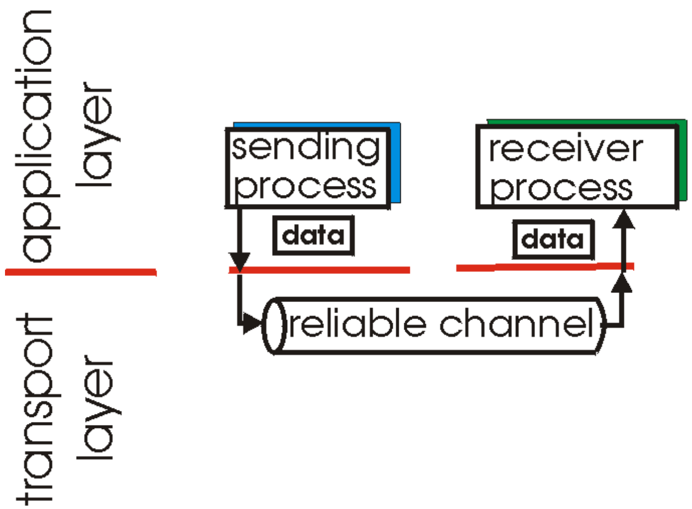
Principles of reliable data transfer

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

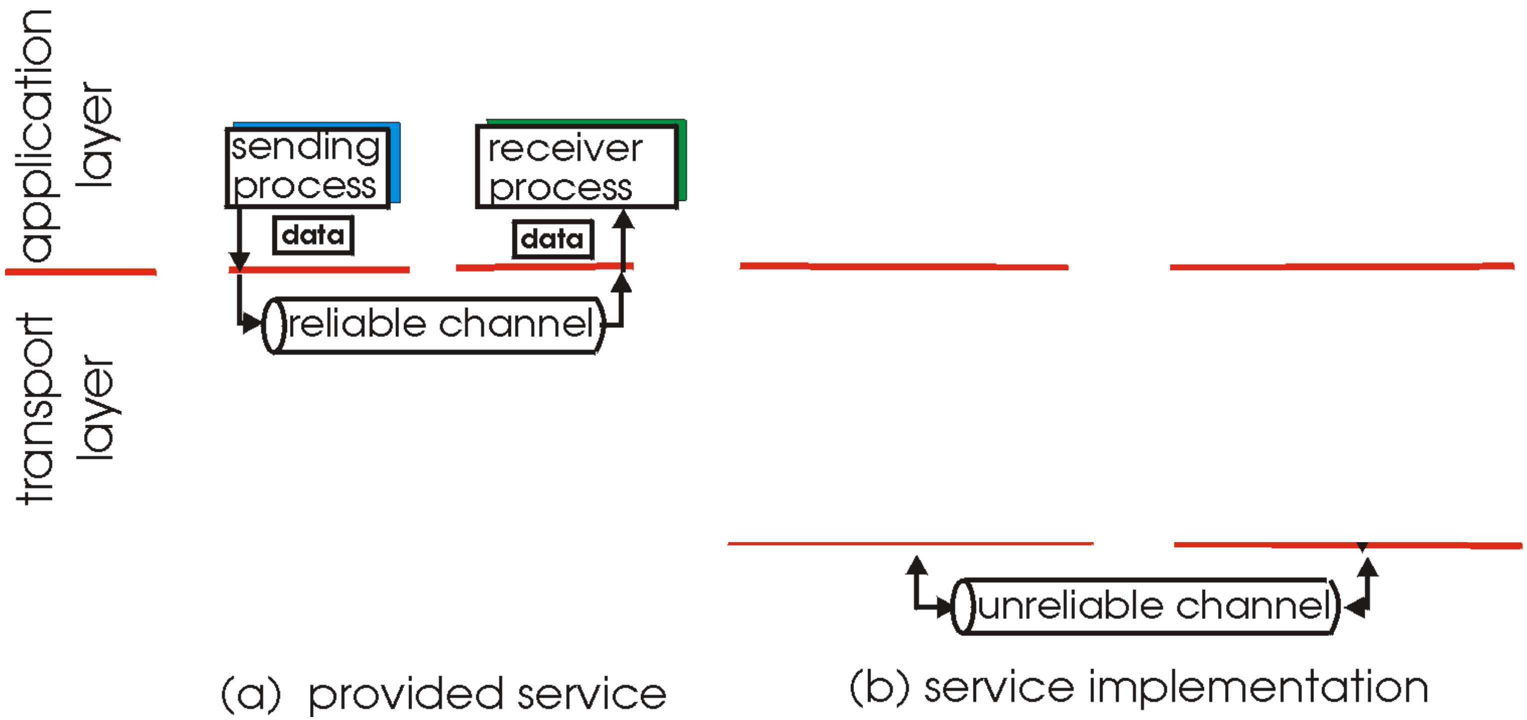


(a) provided service

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

Principles of reliable data transfer

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



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Principles of reliable data transfer

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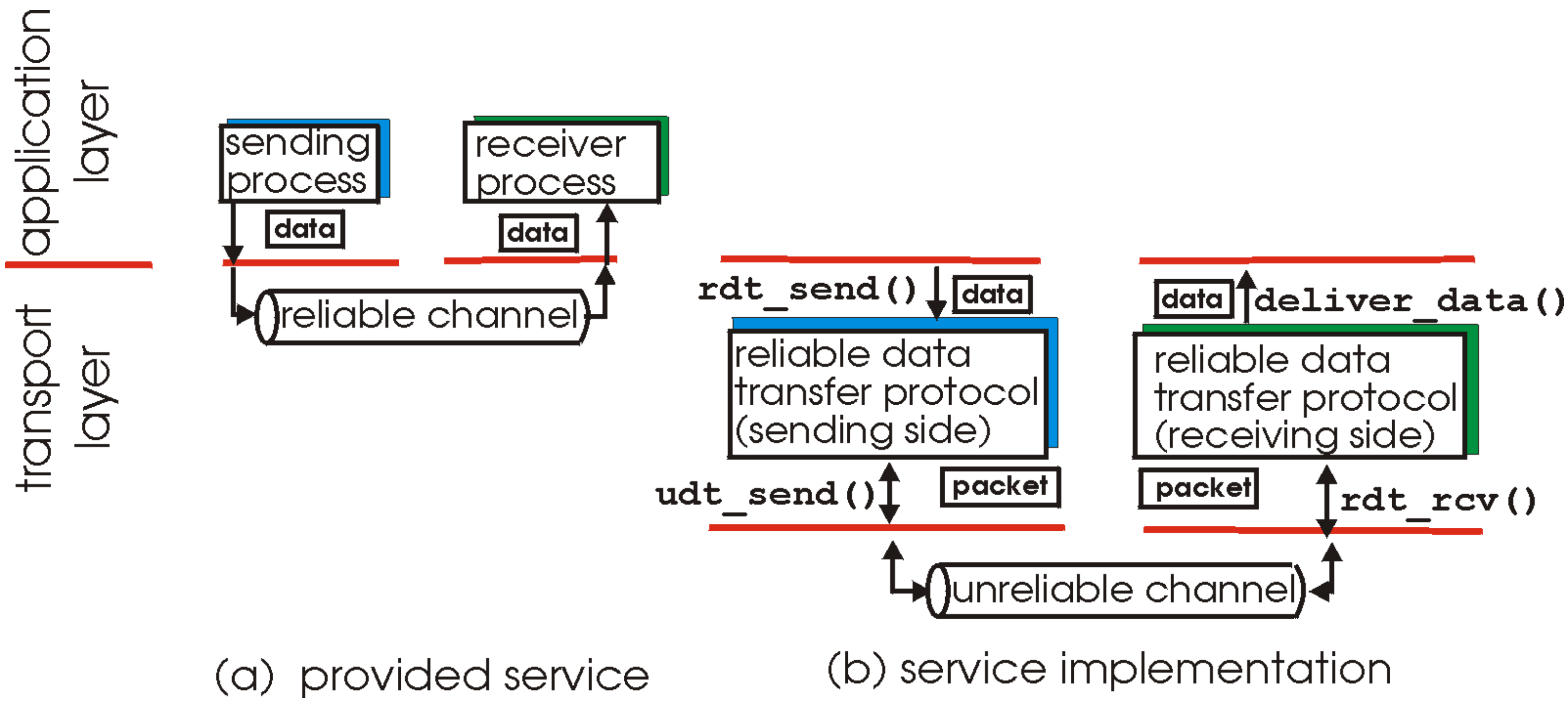
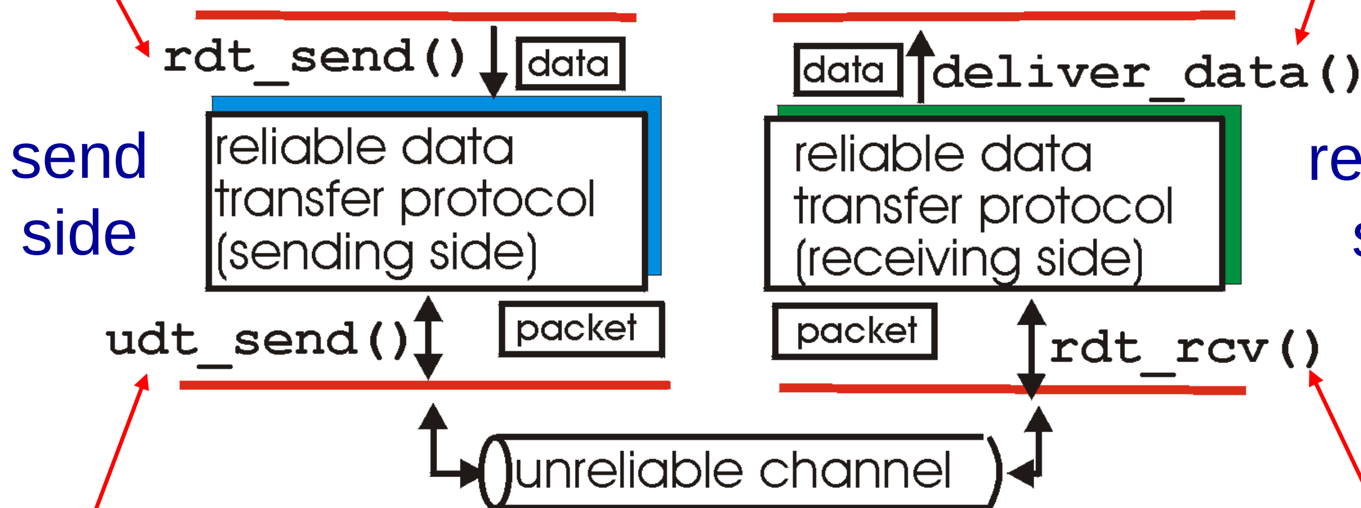


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

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

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



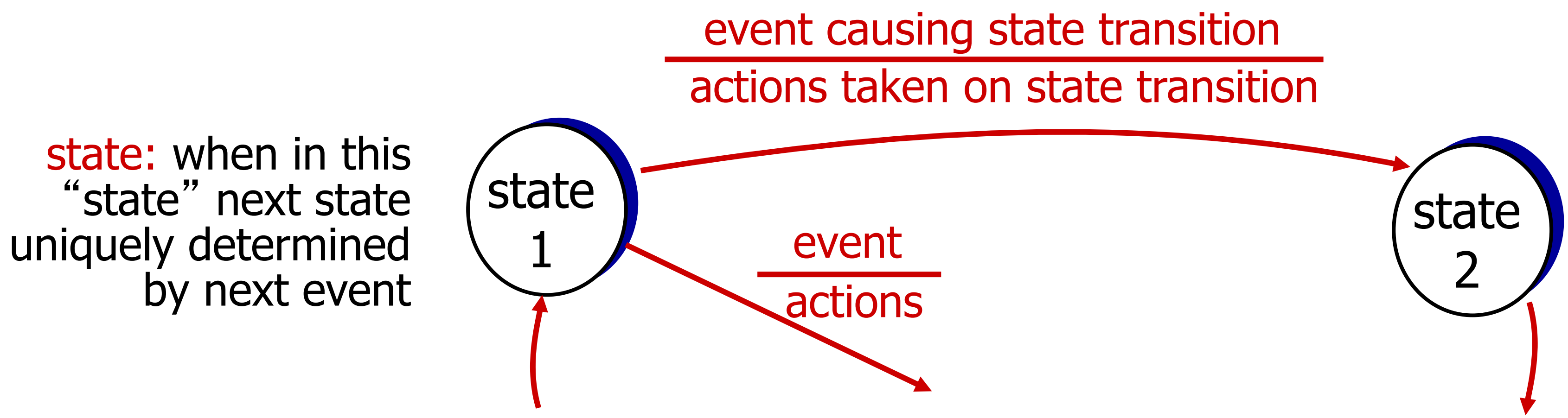
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

Reliable data transfer: getting started

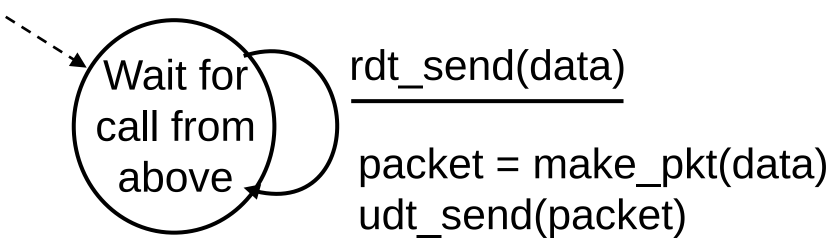
we'll:

- incrementally develop sender, receiver sides of <u>r</u>eliable <u>d</u>ata <u>t</u>ransfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

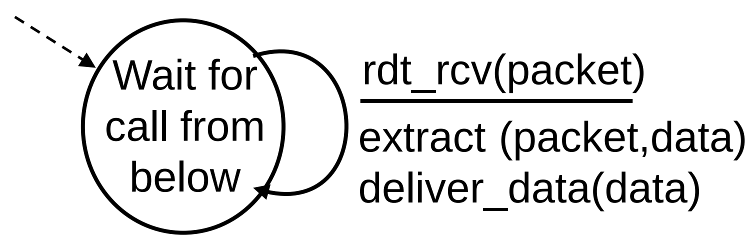


rdt1.0: reliable transfer over a reliable channel

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



sender



receiver

rdt2.0: channel with bit errors

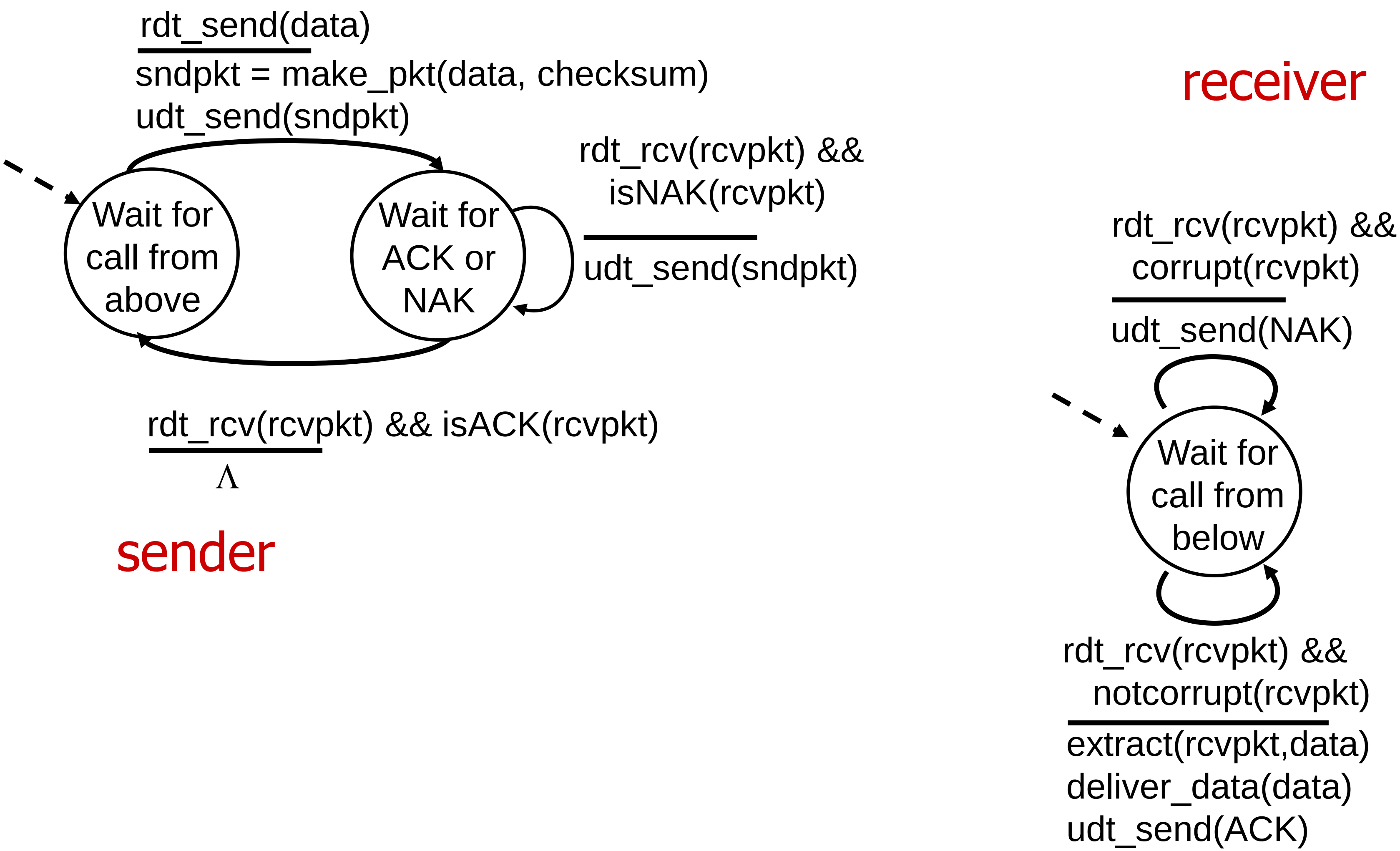
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

How do humans recover from "errors" during conversation?

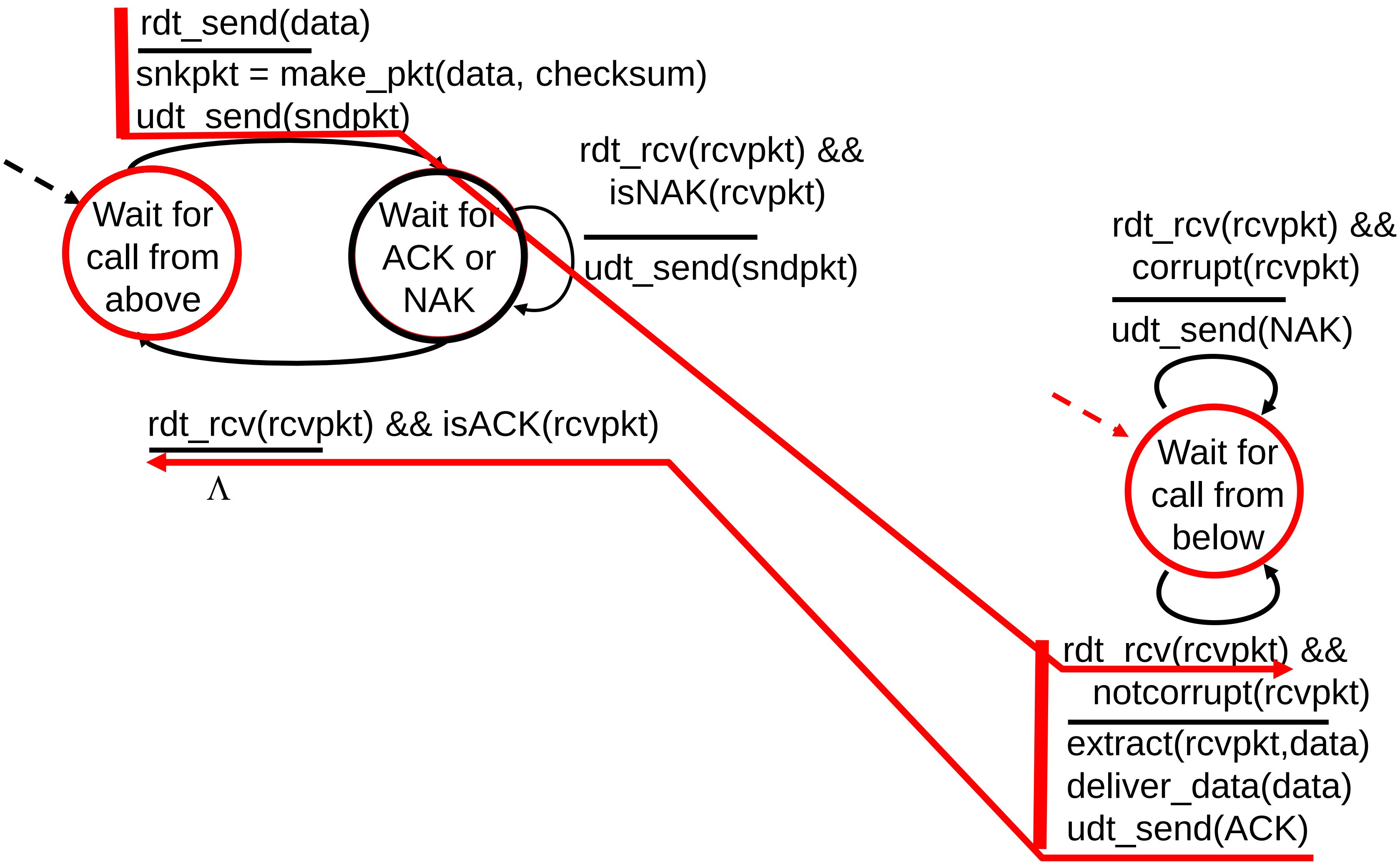
rdt2.0: channel with bit errors

- 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
  - reliable data transfer protocols based on such retransmission are known as ARQ (Automatic Repeat reQuest) protocols.

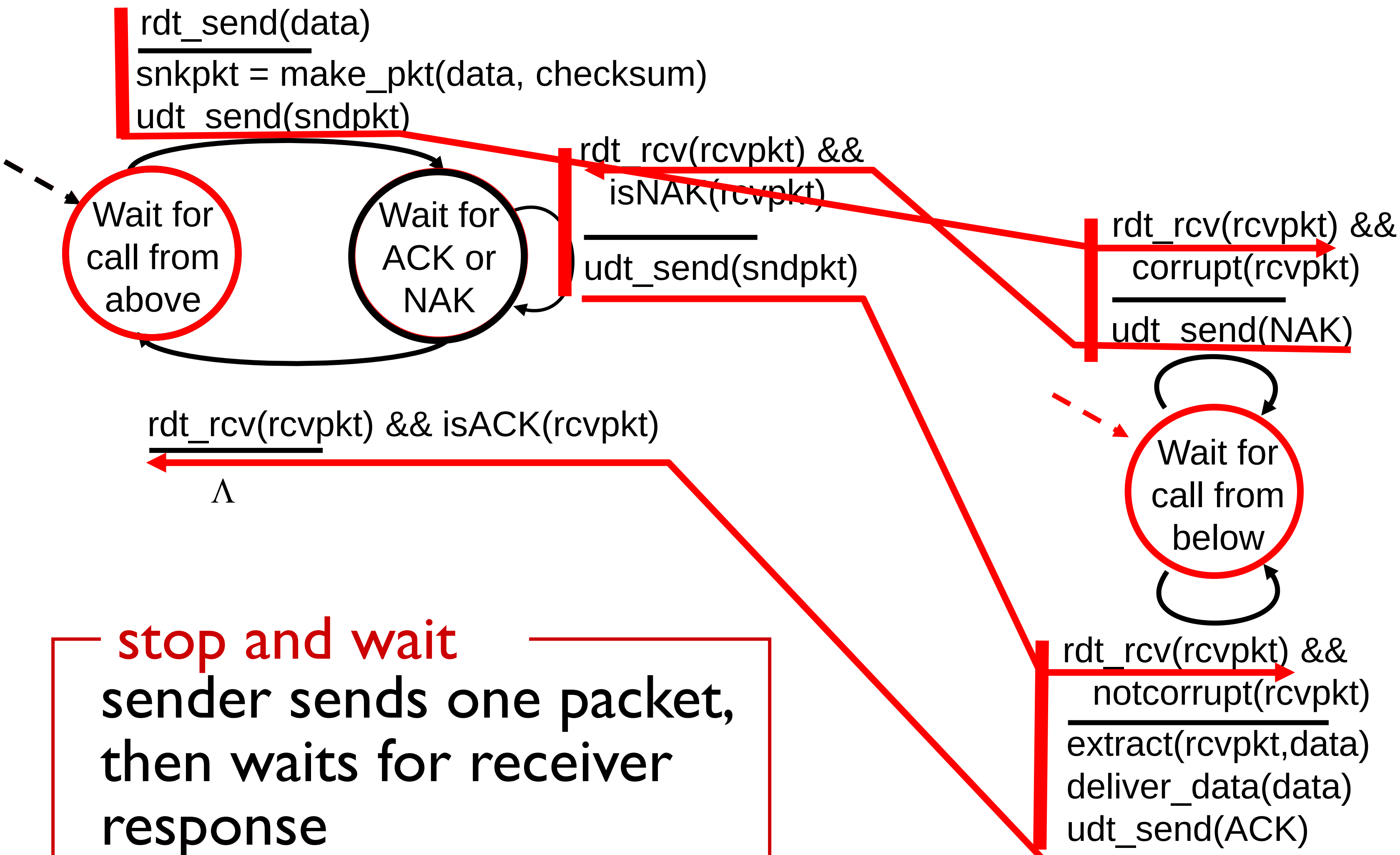
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



stop and wait

sender sends one packet, then waits for receiver response

rdt2.0 has a fatal flaw!

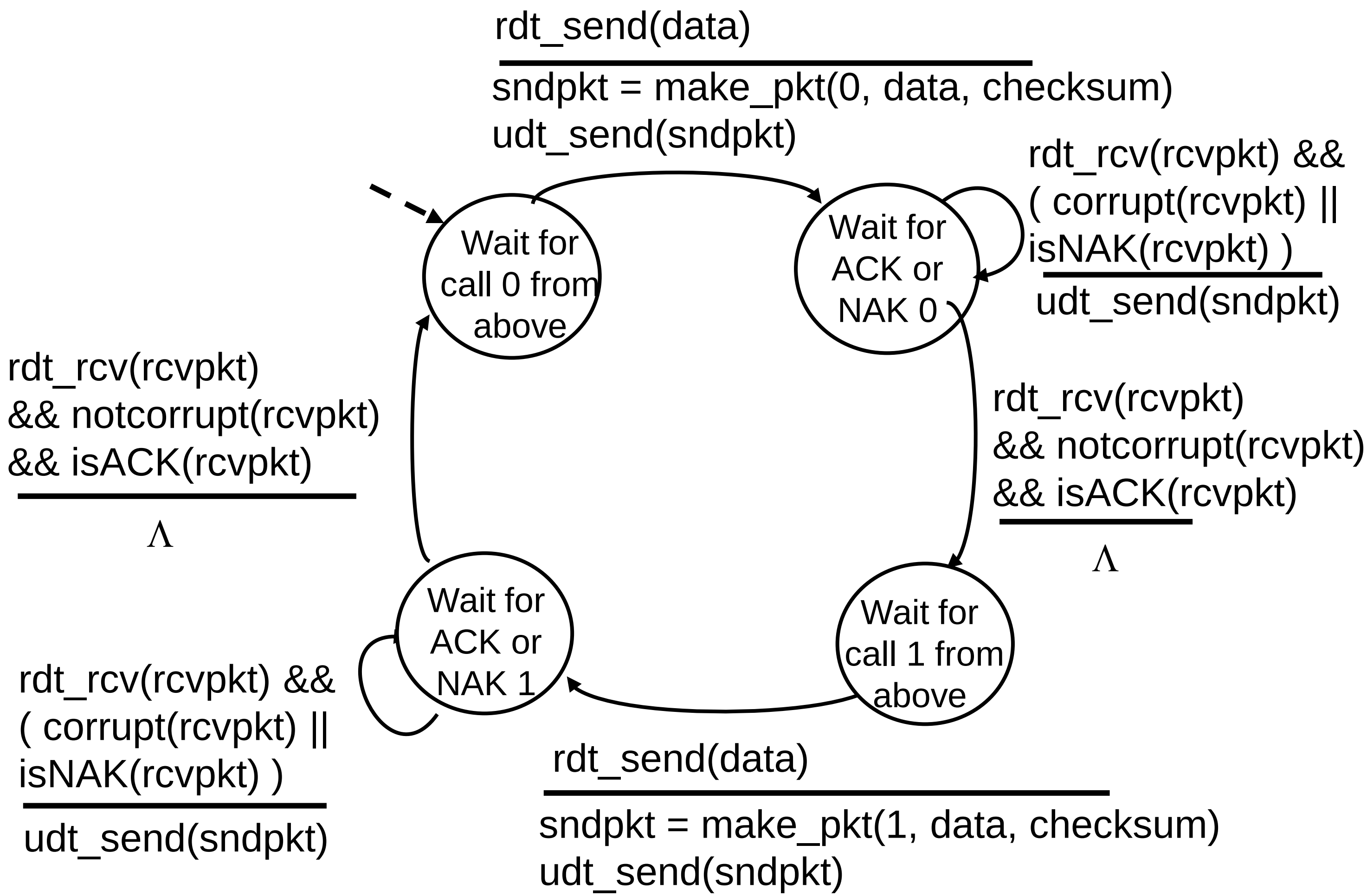
what happens if ACK/NAK corrupted?

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

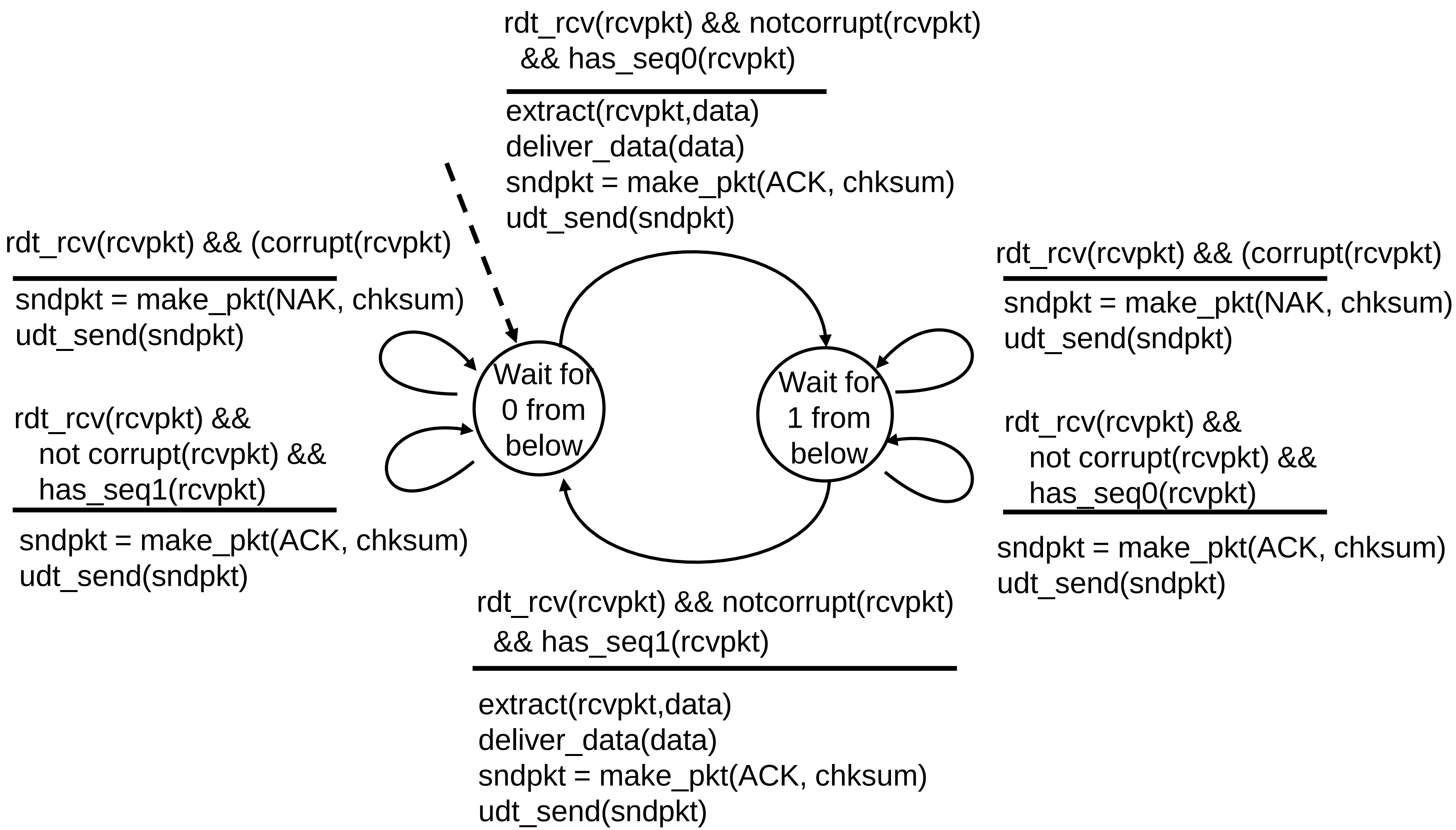
handling duplicates:

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

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

sender:

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

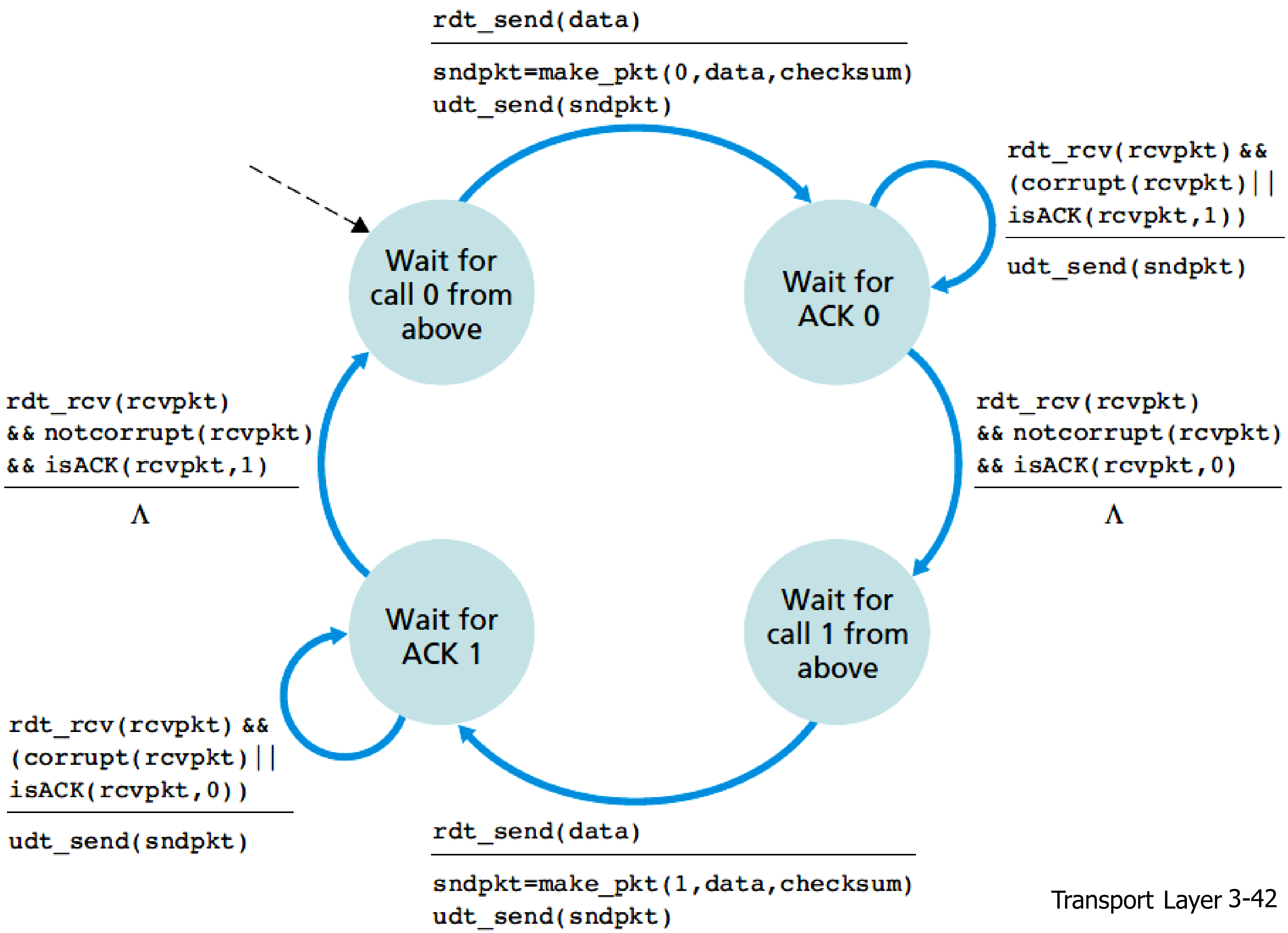
receiver:

- 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

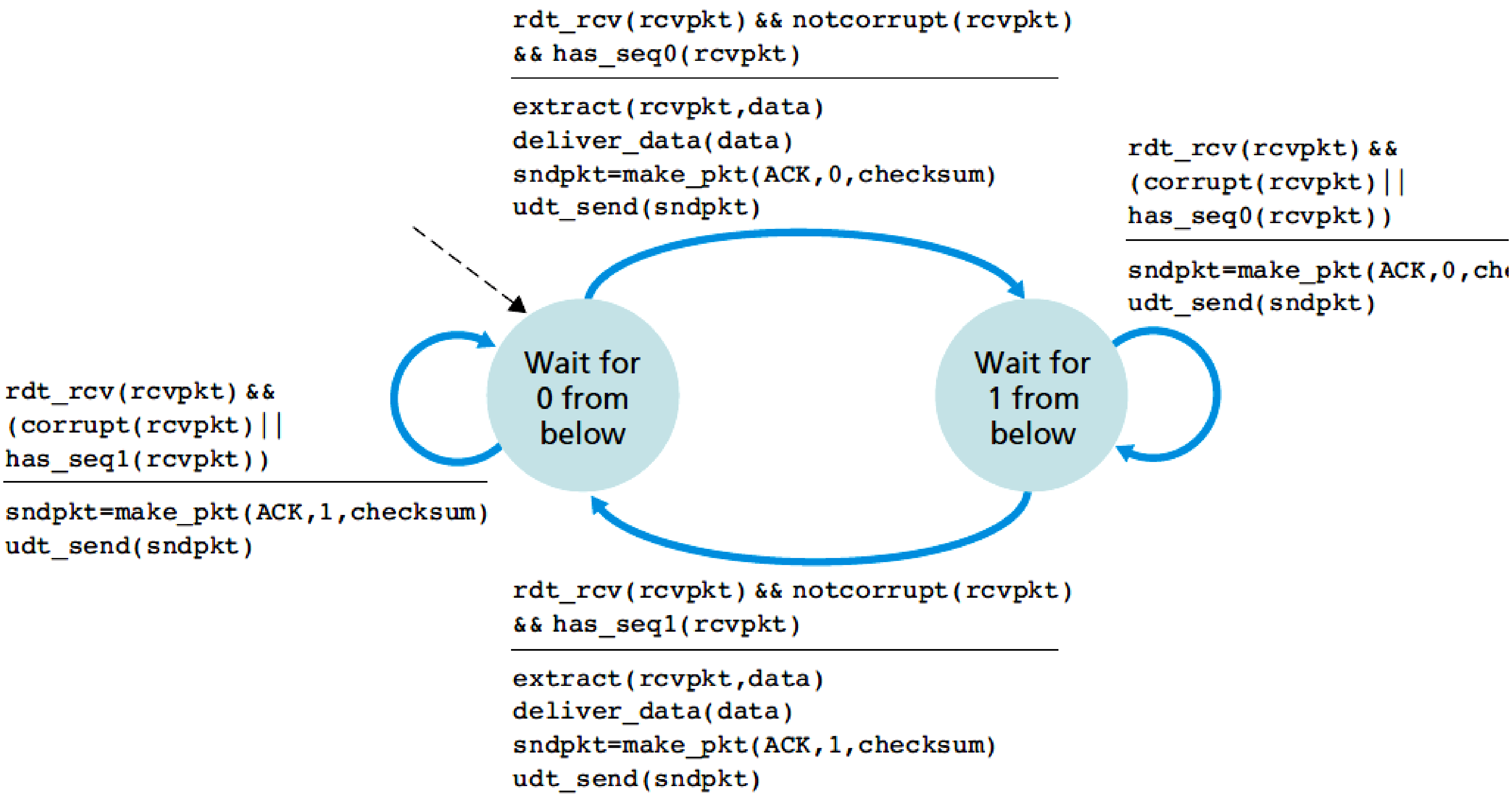
rdt2.2: a NAK-free protocol

- 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



rdt2.2: receiver



rdt3.0: channels with errors *and* loss

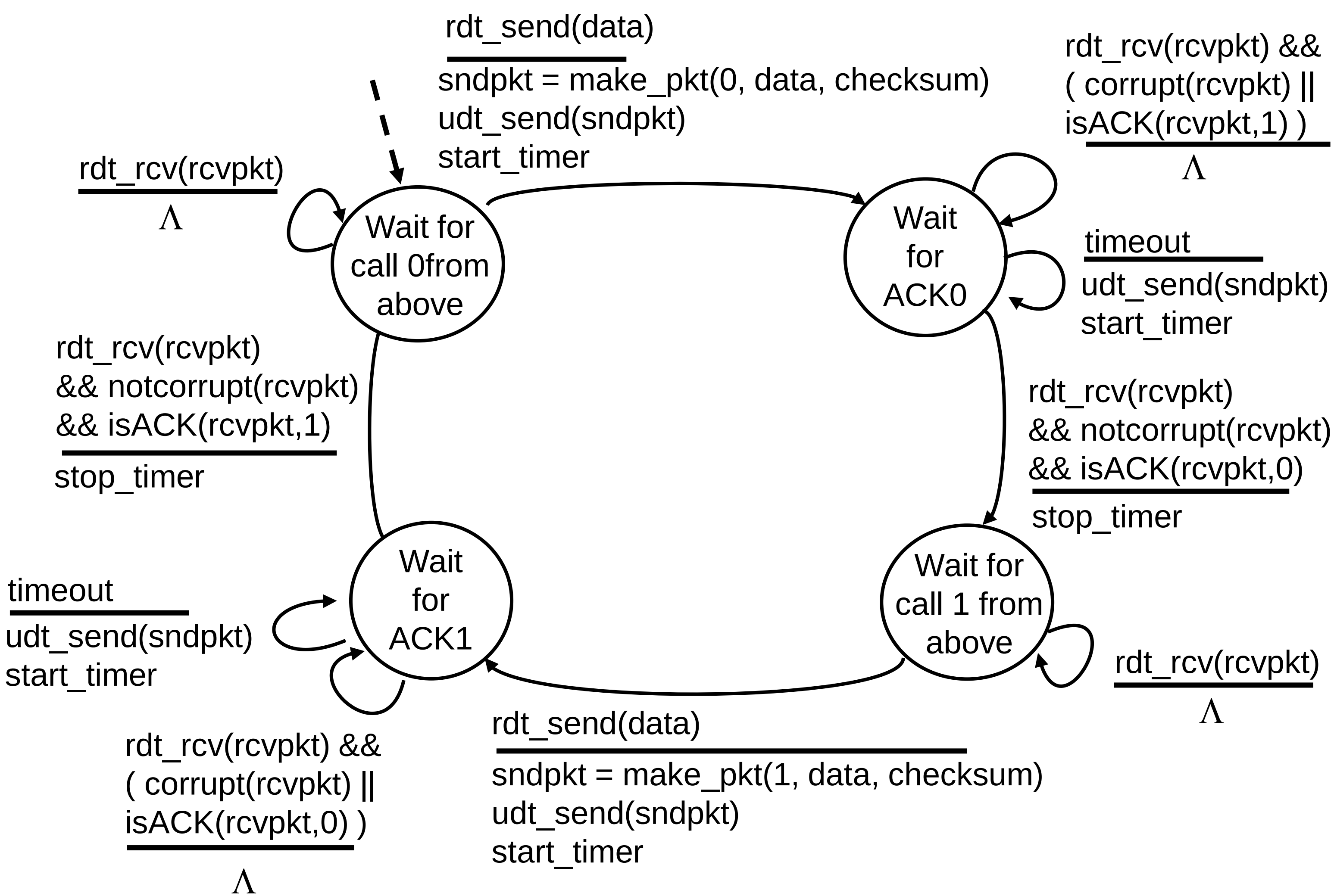
new assumption: underlying channel can also lose packets (data, ACKs)

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

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

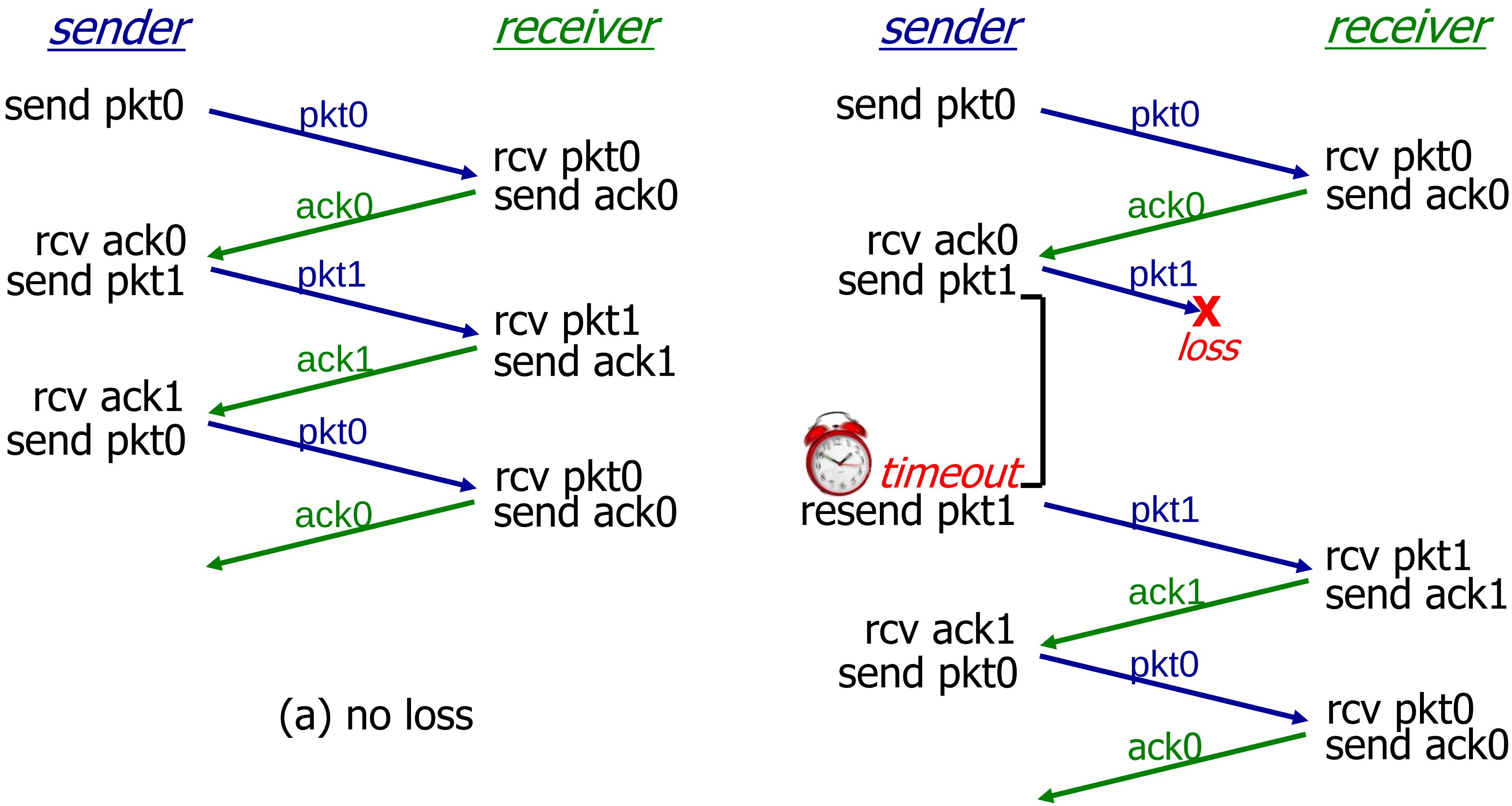
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt3.0 sender



Receiver's FSM is same as rdt2.2

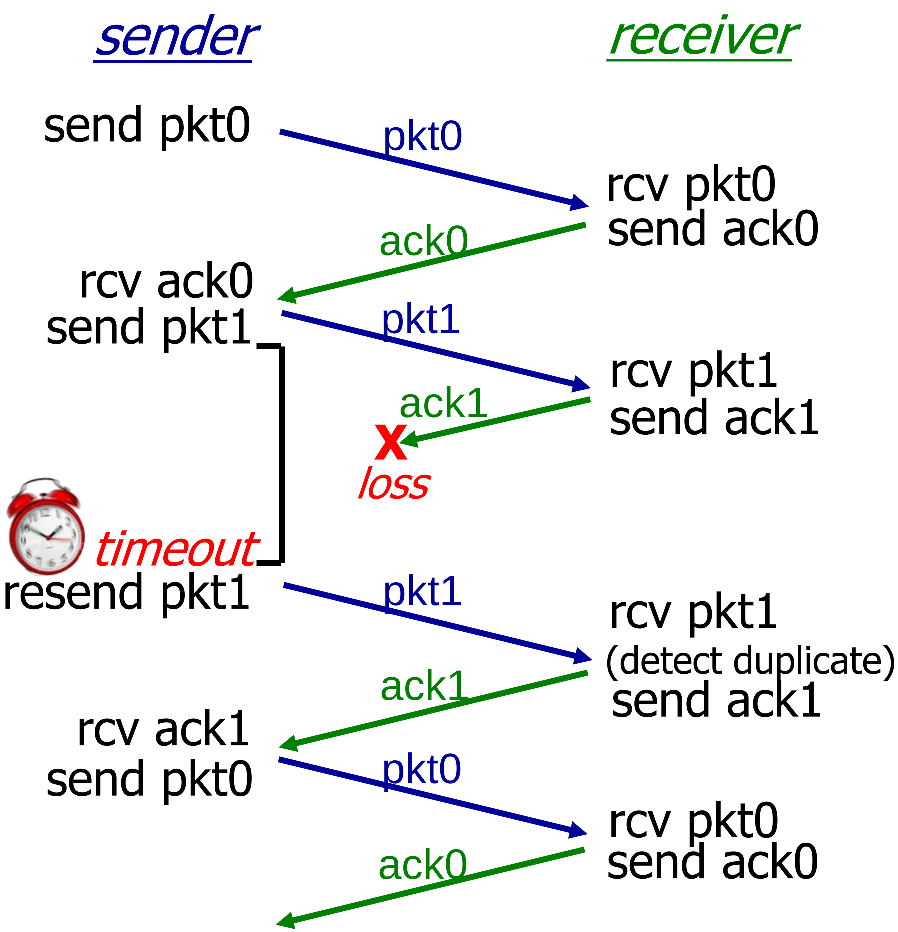
rdt3.0 in action



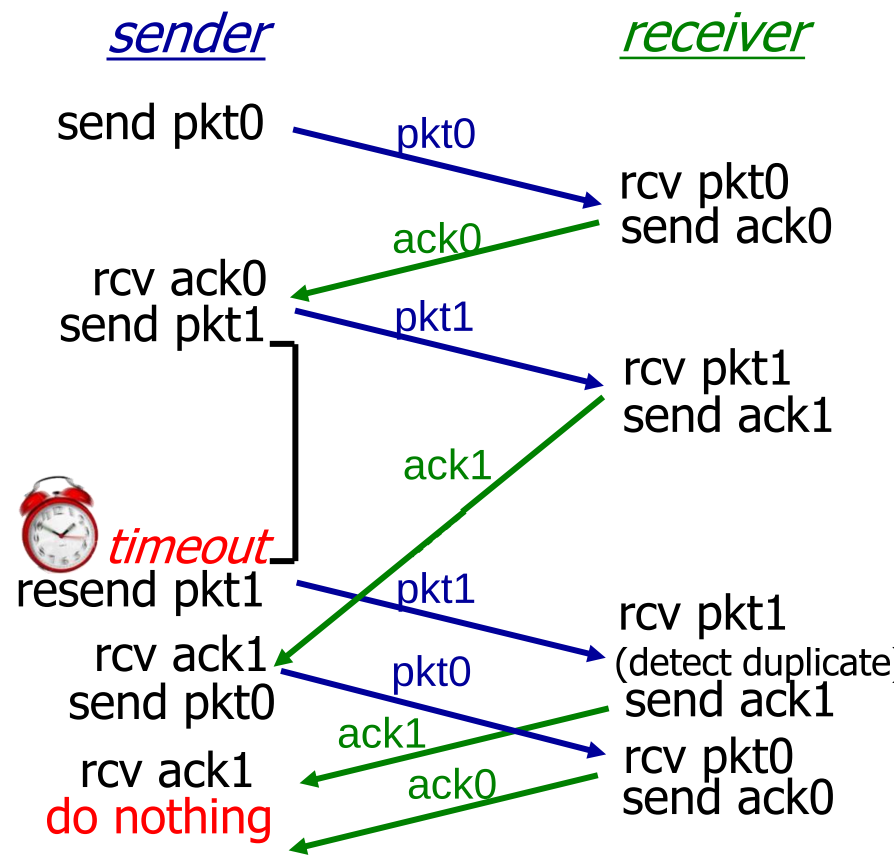
(a) no loss

(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

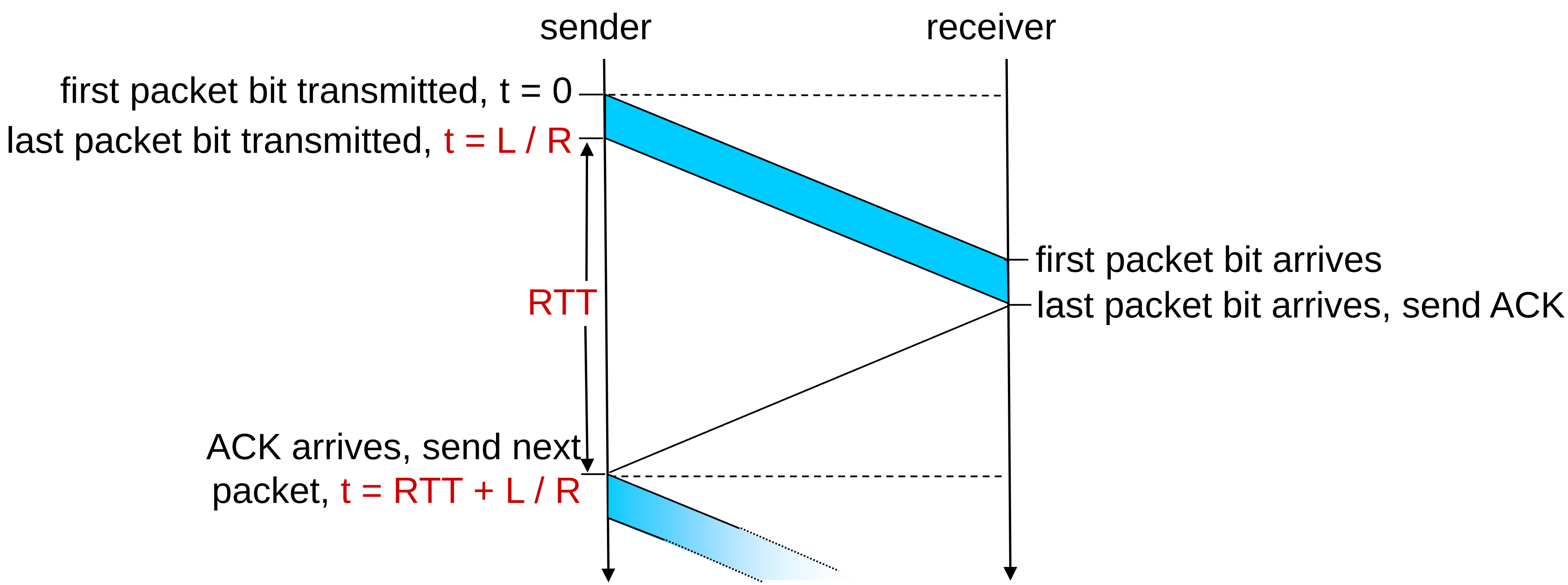
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

  - 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

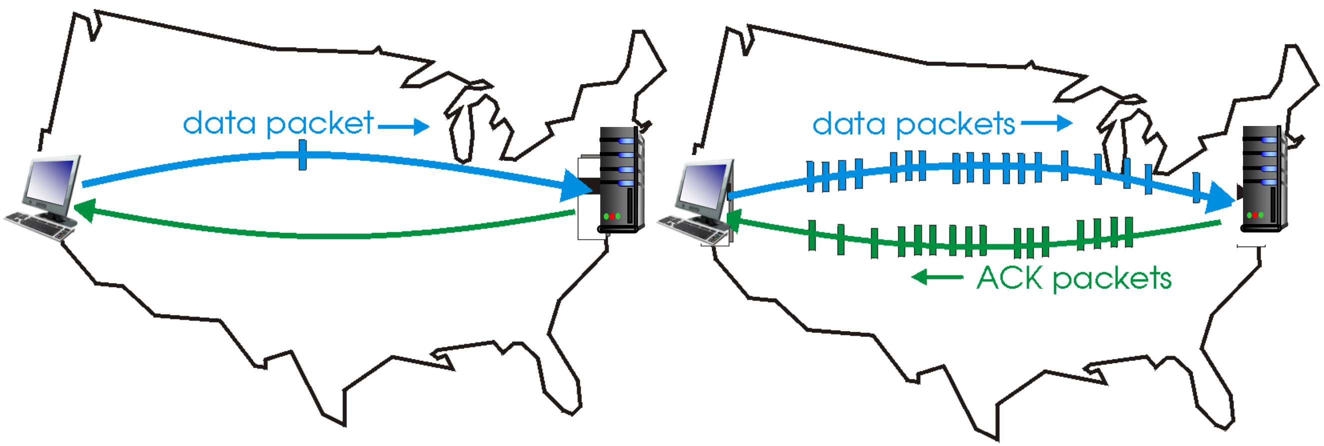


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

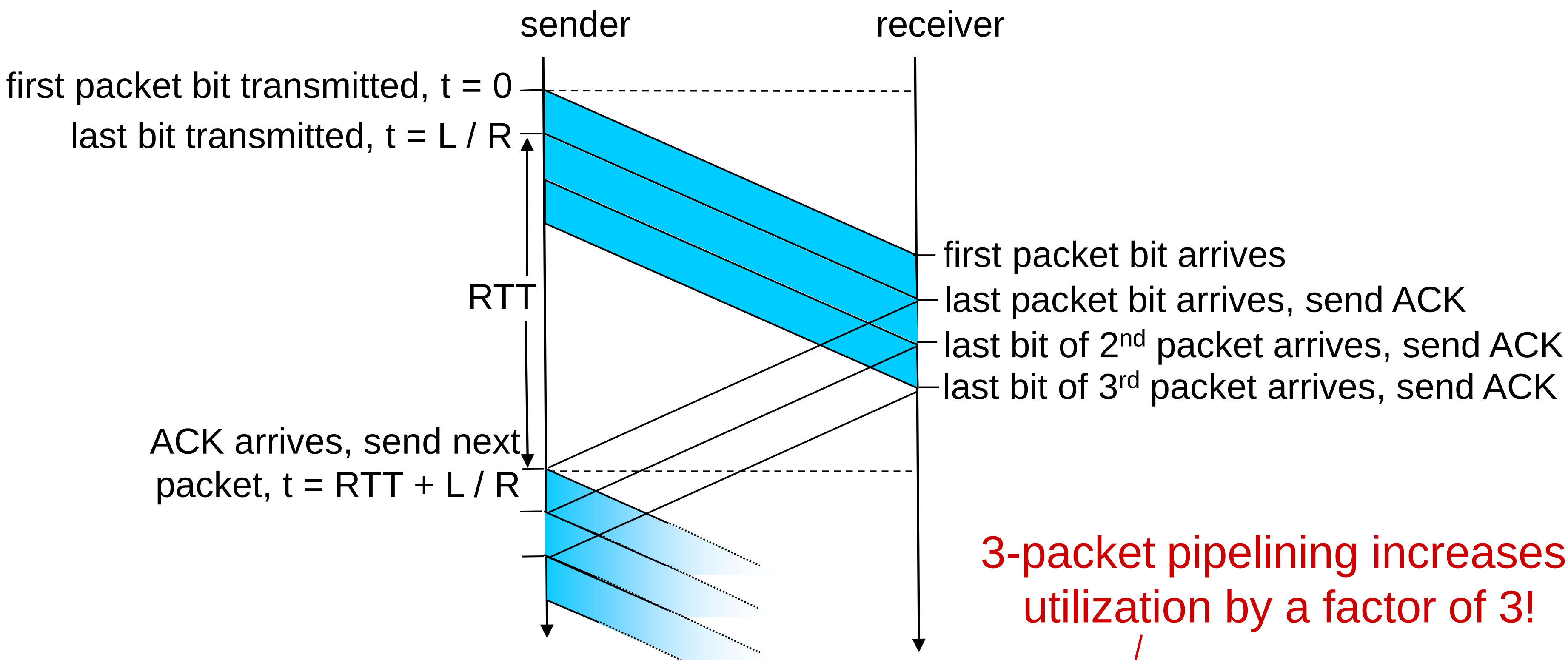
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

- two generic forms of pipelined protocols: *go-Back-N（回退N）, selective repeat（选择性重传）*

Pipelining: increased utilization



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:

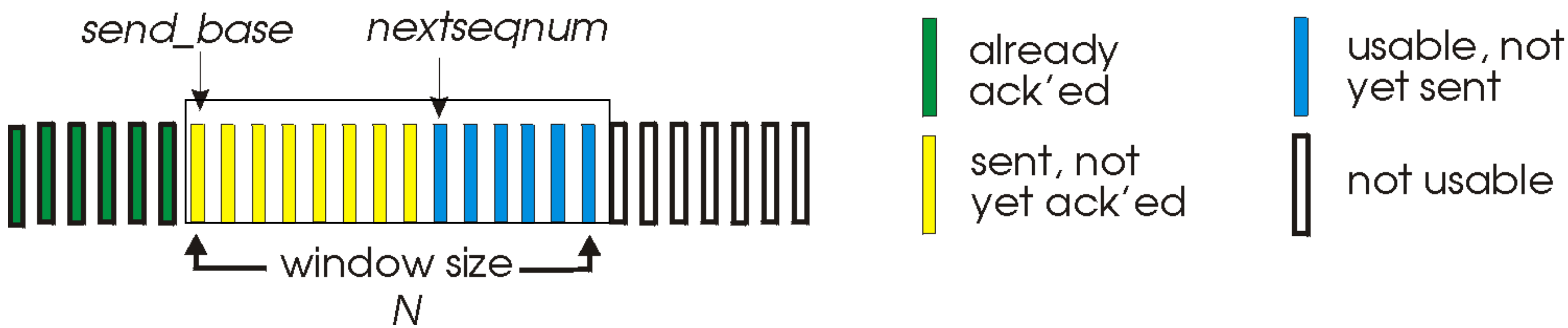
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack（累积确认）*
  - doesn’t ack packet if there’s a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack’ed packets in pipeline
- rcvr sends *individual ack（逐个确认）* 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



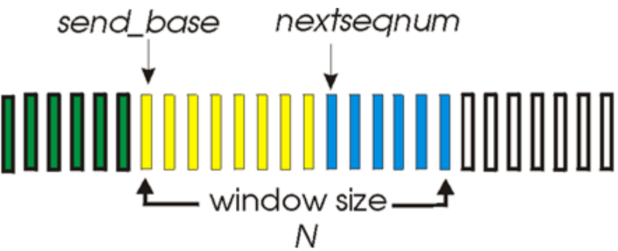
[0, *send_base*-1]: packets transmitted and acked
[*send_base*, *nextseqnum*-1]: packets transmitted but not acked, in-flight packets
[*nextseqnum*, *send_base*+*N*-1]: packets can be sent
[*send_base*+*N*,]: packets can not be sent

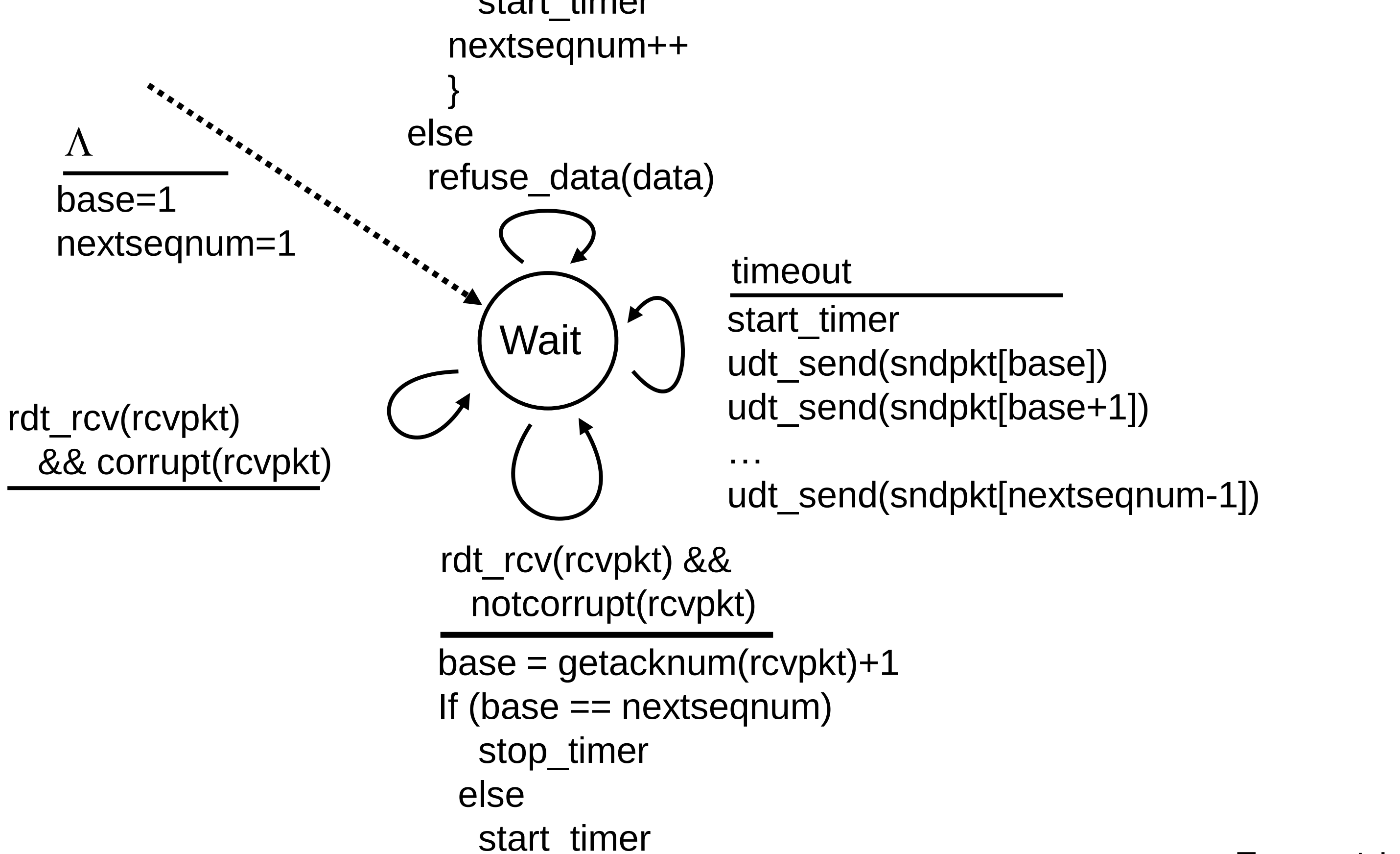
- N: window size
- GBN is called a sliding-window protocol

Go-Back-N: sender

- 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

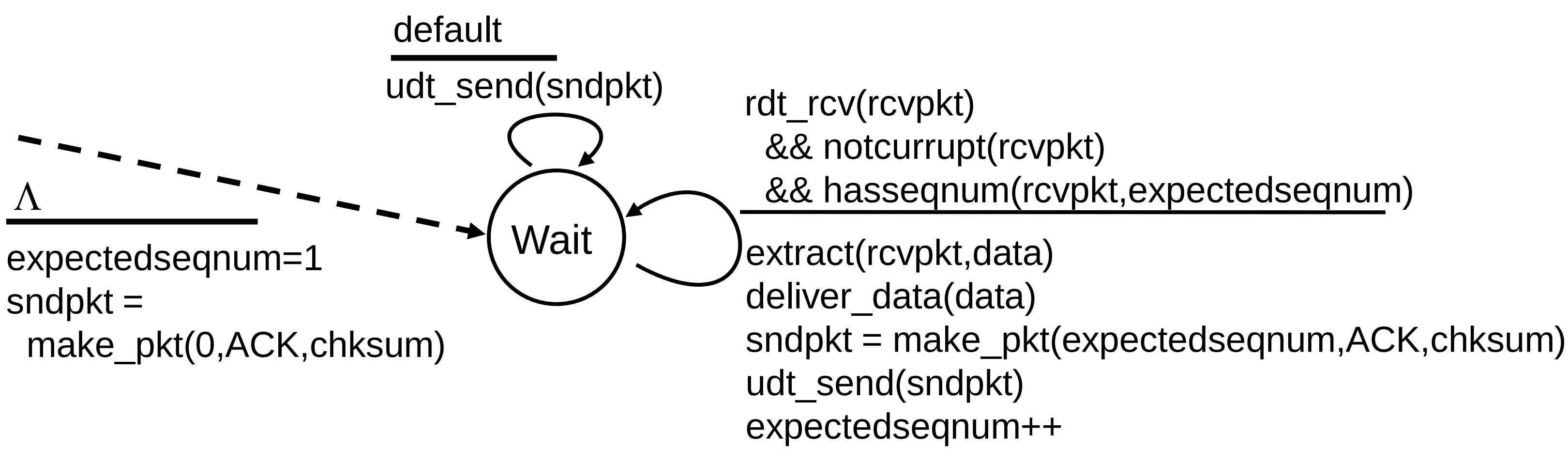




GBN: sender extended FSM

- When rdt_send() is called from above, check if the window is full
  - Not full: send packet, update variables
    - If first packet in window, start timer
  - Full: return data to the uppper layer
- Receive ACK with seq. *n*:
  - All the packets up to *n* (including *n*) is acked,
  - Stop timer if all sent-out packets are acked
  - Restart timer if some packets are acked.
- Timeout:
  - Resend all the packets in [*base*, *nextseqnum*-1]

GBN: receiver extended FSM

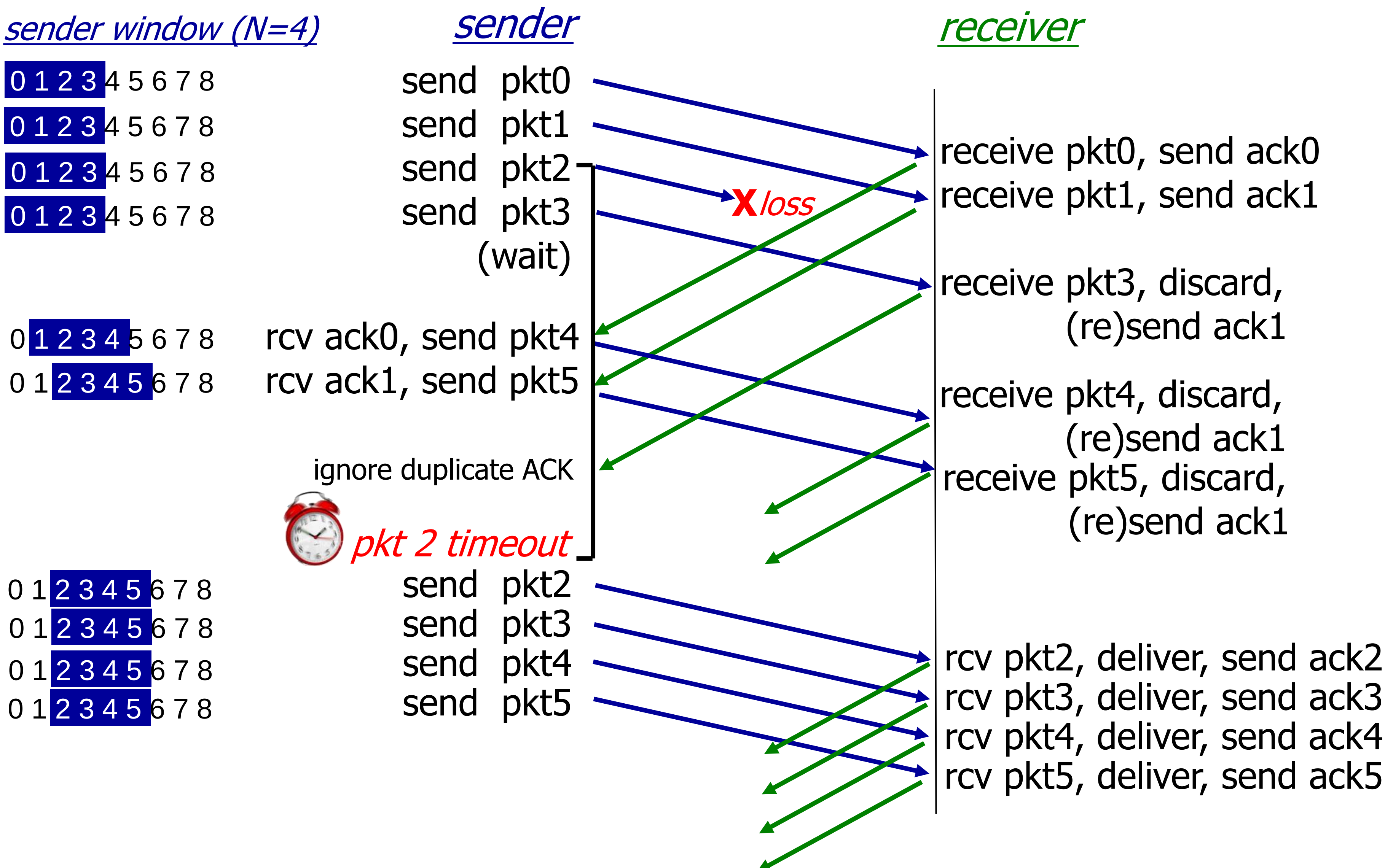


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **expectedseqnum**

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

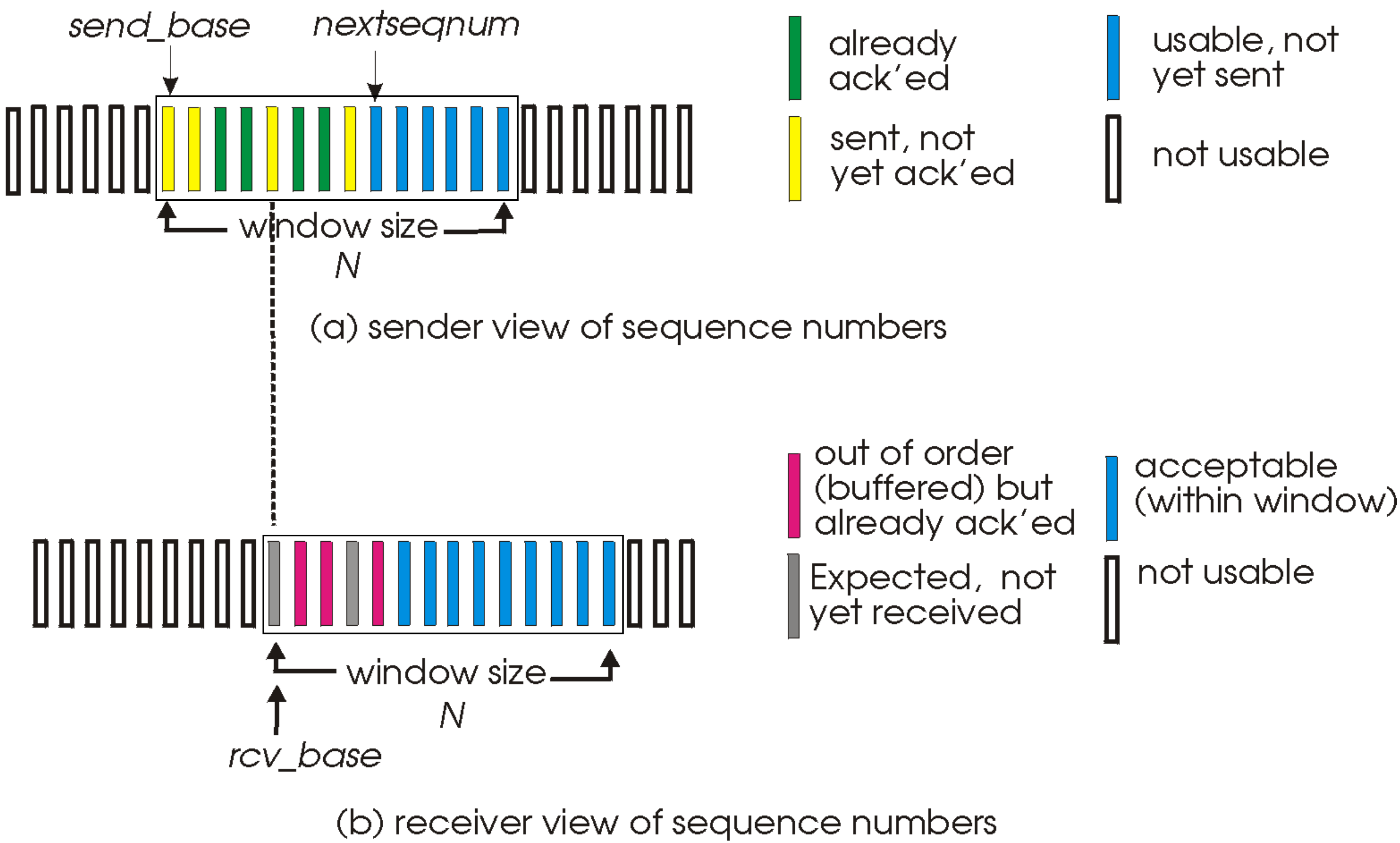
GBN in action



Selective repeat

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

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 is 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]

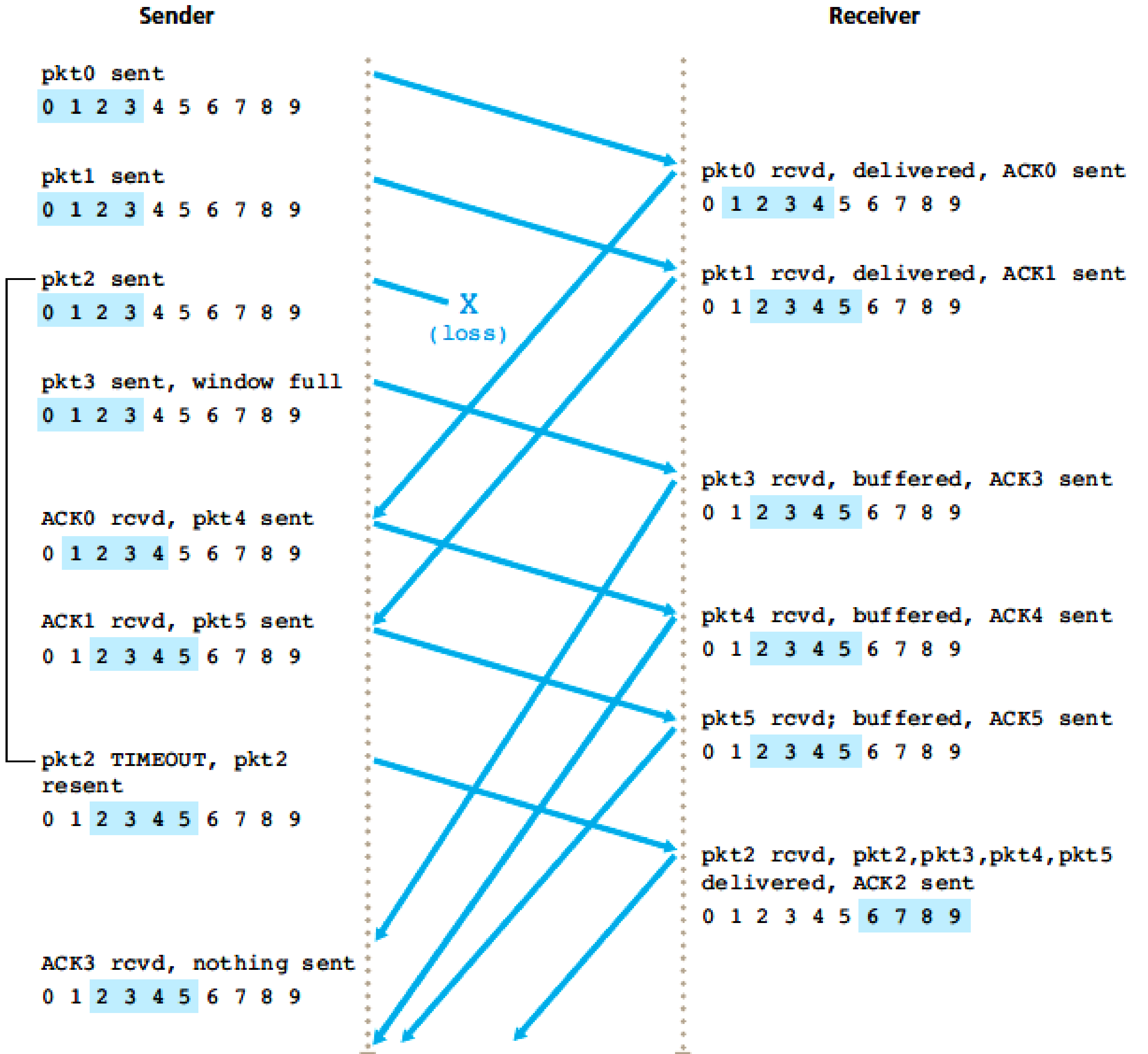
- ACK(n)

otherwise:

- ignore

Selecti

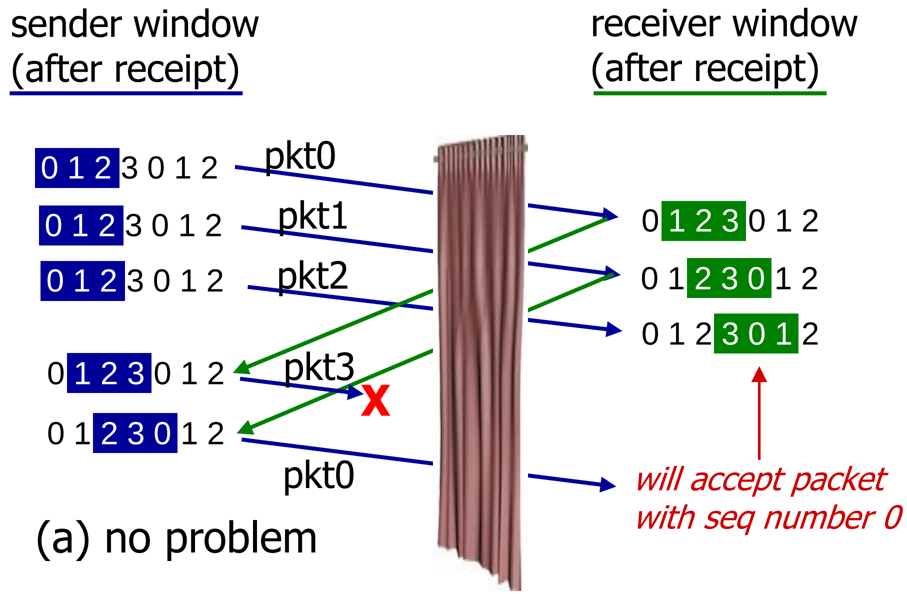
Q: what happens when ack2 arrives?



Selective repeat: dilemma

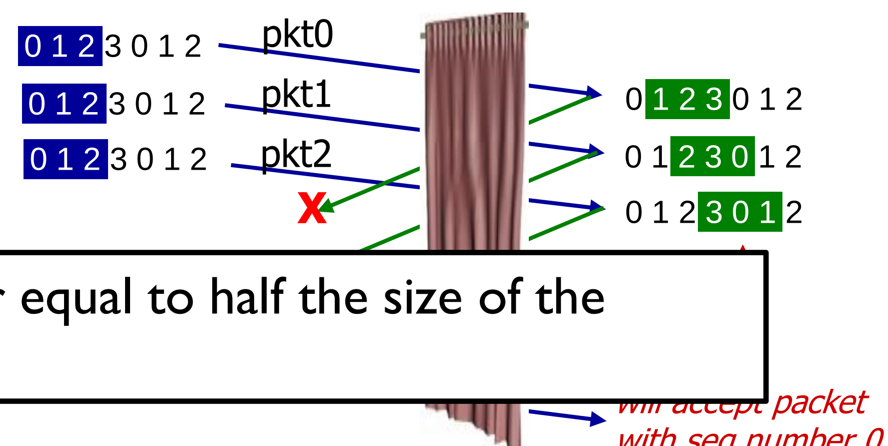
example:

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



(a) no problem

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



(b) oops!

Q: what relationship between seq # size

window size must be less than or equal to half the size of the sequence number space

Comparing GBN and SR

GBN

- Pros
  - Sender maintains one timer
  - Receiver does not need to buffer out-of-order packets
  - Easy to implement
- Cons
  - Waste bandwidth, causing congestions

SR

- Pros
  - Do not waste bandwith
- Cons
  - Sender maintains timer for each unacked packet
  - Reciever buffers out-of-order packets
  - Complex

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

TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
  - multicasting is not possible
- 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
  - First determine the maximum transmission unit (MTU) of the link layer, then set MSS by deducing the TCP/IP header length (typically 40 bytes)
  - Ethernet/PPP have MSS of 1500 bytes, which means MSS=1460 bytes

TCP: Overview

- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange

```
clientSocket.connect((serverName,serverPort))
```

  - Client sends a special segment, sender returns a special segment; and client responds with a third segment
  - Three-way handshake
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure

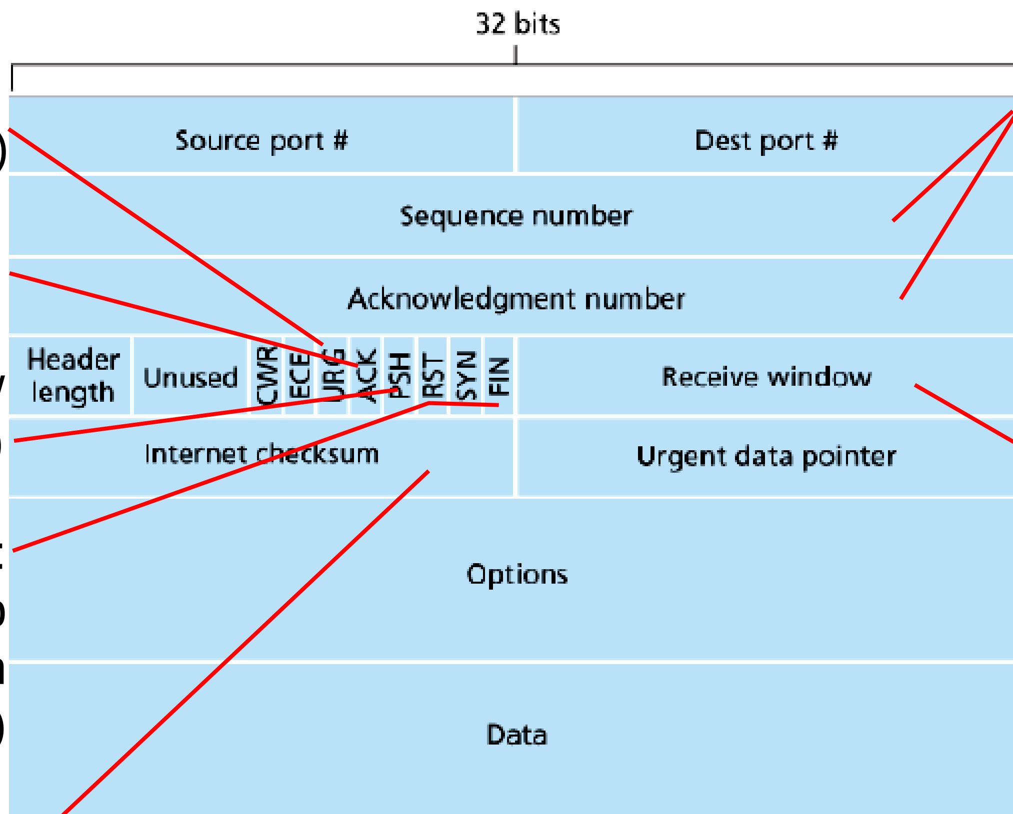
URG: urgent data
(generally not used)

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, ACKs

sequence numbers:

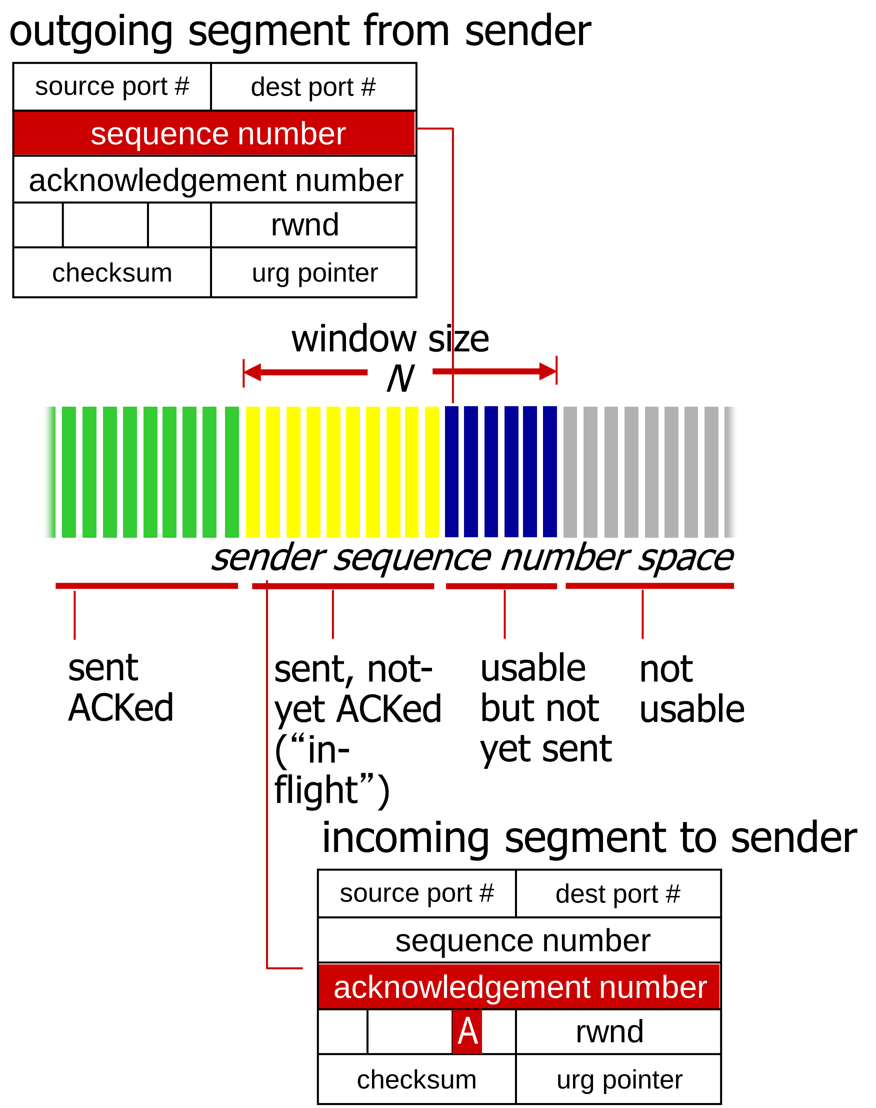
- byte stream "number" of first byte in segment's data

acknowledgements:

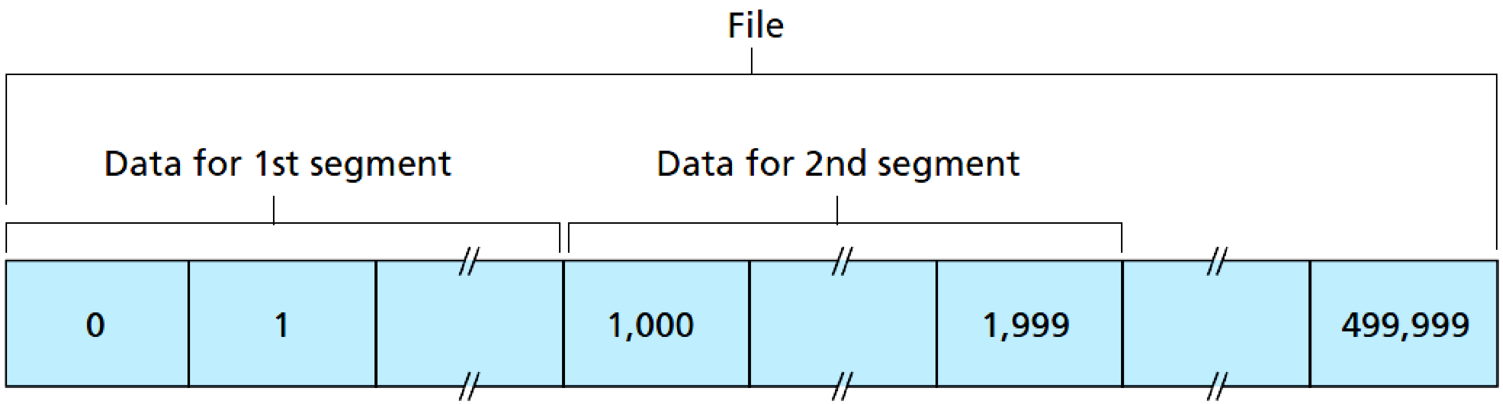
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

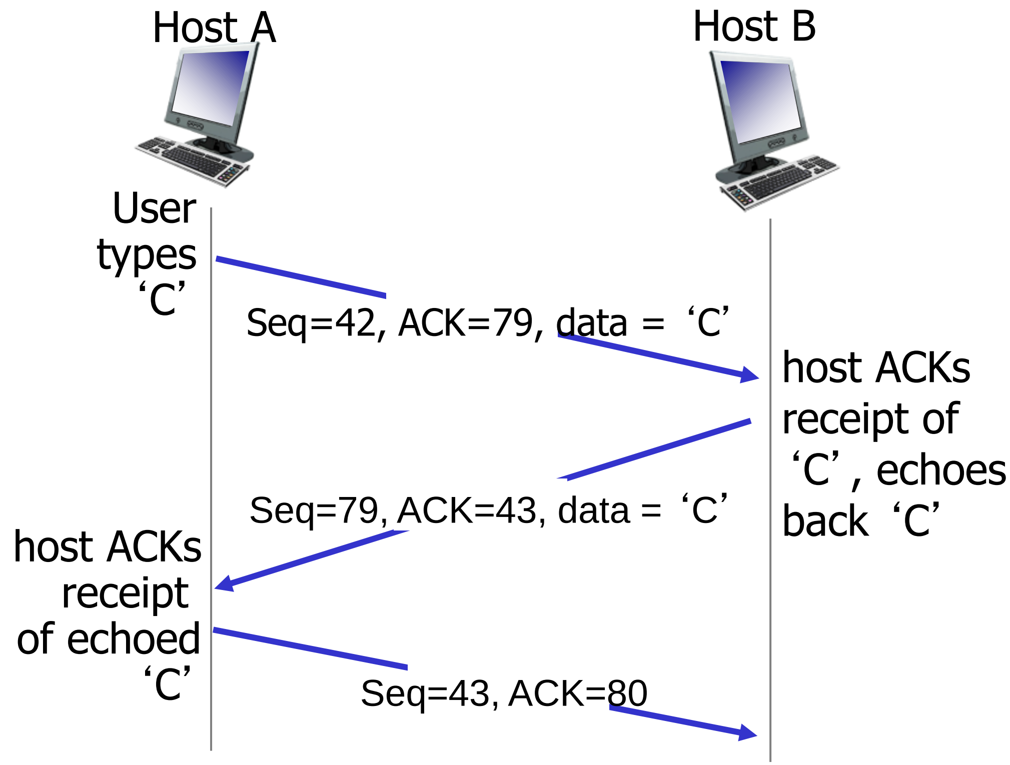


TCP seq. numbers, ACKs



- Example 1: A file of 500,000 bytes, MSS=1000 bytes
  - Segment 1, seq#=0
  - Segment 2, seg#=1000
  - Segment 3, seg#=2000
  - …
- Example 2: A receives a segment from B containing 0~535 bytes, and another segment containing 900~1000 bytes, but not the bytes 536~899.
  - A's segment to B ack#=536
  - Accumulative acknowledgement

TCP seq. numbers, ACKs



simple telnet scenario

the acknowledgment for the first client-to-server data is said to be piggybacked on the server-to-client data segment.

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

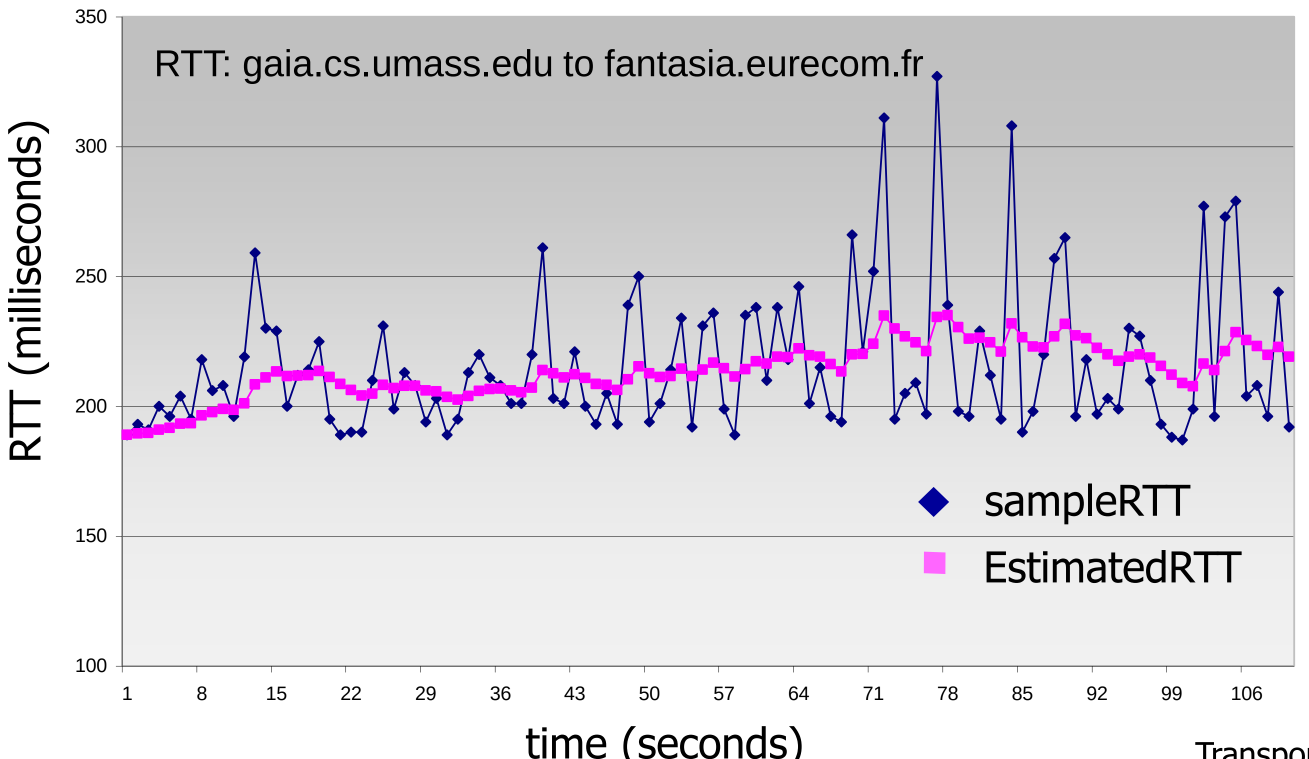
Q: how to estimate RTT?

- **`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（指数滑动平均）
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- Measure the variability of the RTT

$$\texttt{DevRTT} = (1 - \beta) \cdot \texttt{DevRTT} + \beta \cdot |\texttt{SampleRTT} - \texttt{EstimatedRTT}|$$

- EWMA of the difference between SampleRTT and EstimatedRTT
- Recommend value for β is 0.25

TCP round trip time, timeout

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

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

estimated RTT   "safety margin"

- Initial TimeoutInterval=1 sec.

Chapter 3 outline

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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:

data rcvd from app:

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

timeout:

- retransmit segment that caused timeout
- restart timer

只重传1个 segment

ack rcvd:

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

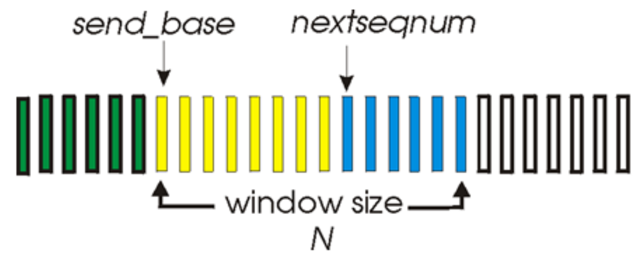
TCP sender events:

```
NextSeqNum=InitialSeqNumber
SendBase=InitialSeqNumber

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)
            break;

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

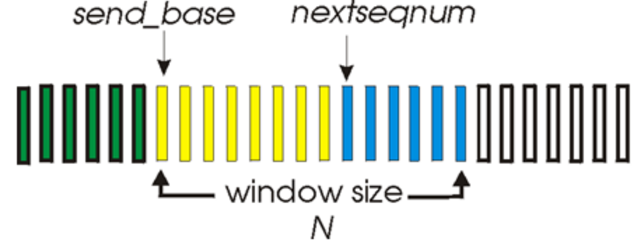


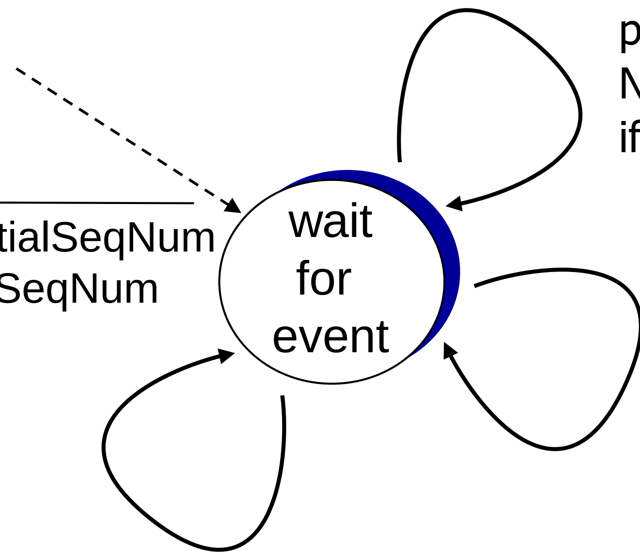
TCP sender events:

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

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

TCP sender (simplified)





Λ

NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

data received from application above

create segment, seq. #: NextSeqNum
pass segment to IP (i.e., “send”)
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
    start timer

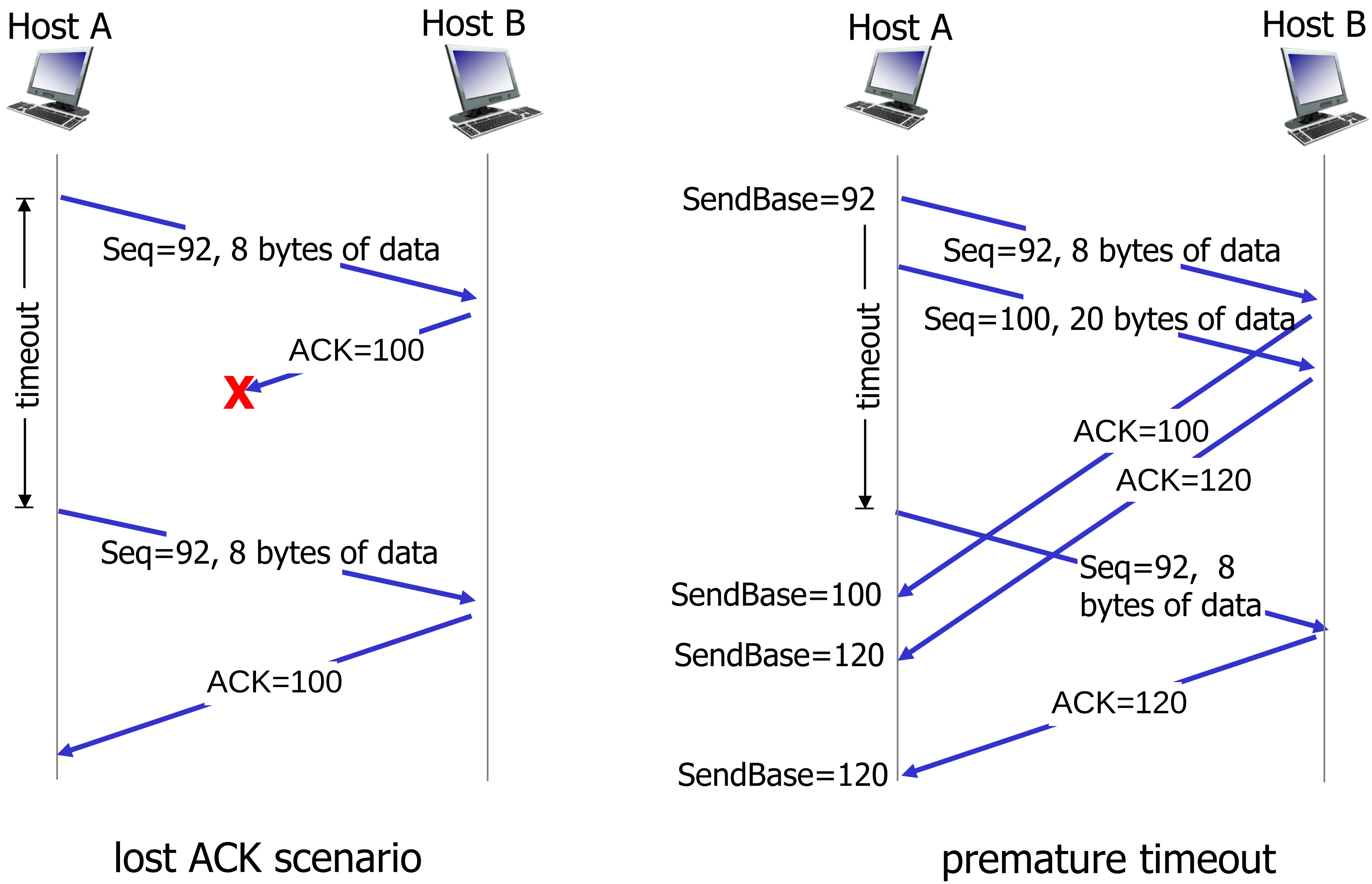
timeout

retransmit not-yet-acked segment
    with smallest seq. #
start timer

ACK received, with ACK field value y

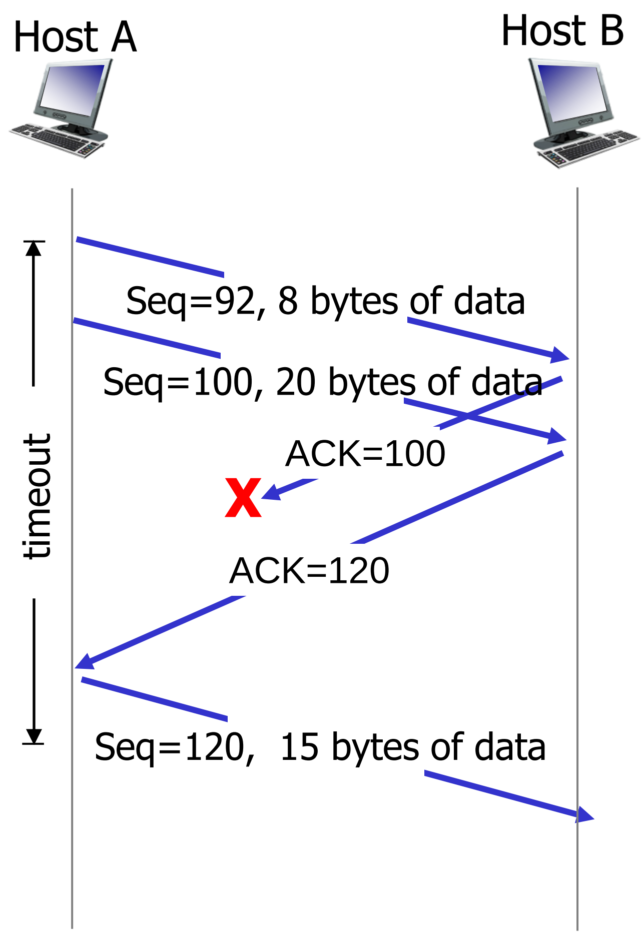
```
if (y > SendBase) {
   SendBase = y
   /* SendBase–1: last cumulatively ACKed byte */
   if (there are currently not-yet-acked segments)
       start timer
     else stop timer
 }
```

TCP: retransmission scenarios



The second segment will not be retransmitted until another timeout

TCP: retransmission scenarios



cumulative ACK：ACK=120表示到119的所有segment都收到了

Doubling timeout

- whenever the timeout event occurs, TCP retransmits the not-yet acknowledged segment with the smallest sequence number, as described above.
  - But each time TCP retransmits, it sets the next timeout interval to twice the previous value.
  - A limited form of congestion control
- whenever the timer is started after either of the two other events (that is, data received from application above, and ACK received), the TimeoutInterval is derived from the most recent values of **EstimatedRTT** and **DevRTT**.

TCP ACK generation [RFC 1122, RFC 2581]

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment waiting for ACK transmission.	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, given that segment starts at lower end of gap

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.

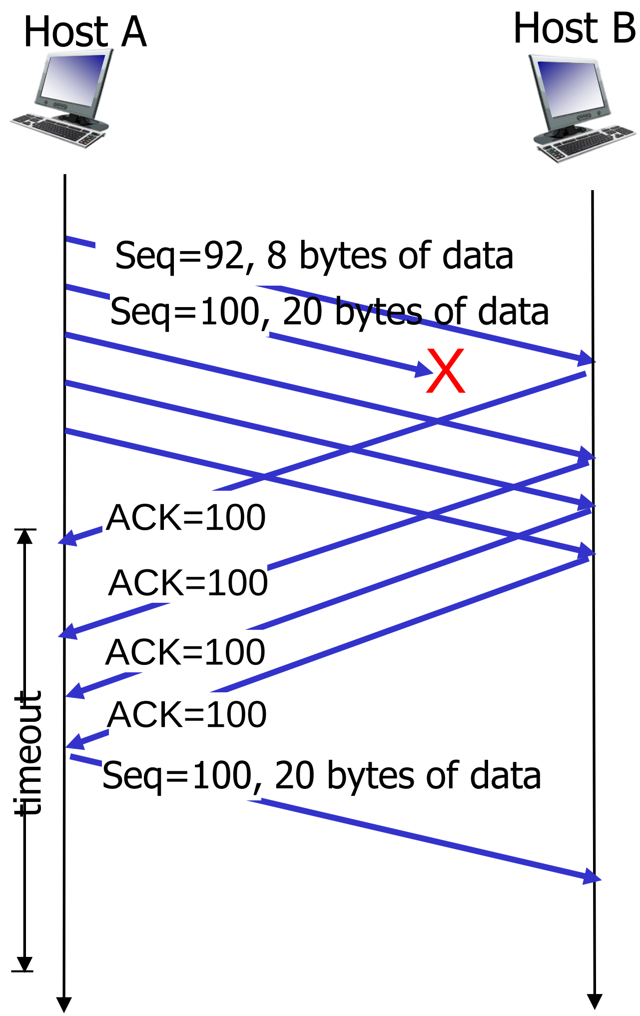
TCP fast retransmit

if sender receives 3 ACKs for same data

("triple duplicate ACKs"), resend unacked segment with smallest seq #

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

TCP fast retransmit



The first ACK is not duplicated ACK

fast retransmit after sender receipt of triple duplicate ACK

Discussion

- Maintains the smallest seq # of transmitted but not acked byte (SendBase) and the sequence of the next byte to send (NextSeqNum); accumulative acknowledgement
  - GBN-like
- Buffer out-of-order segments, only need to retransmit lost data
  - SR-like

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

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- segment structure
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- connection management

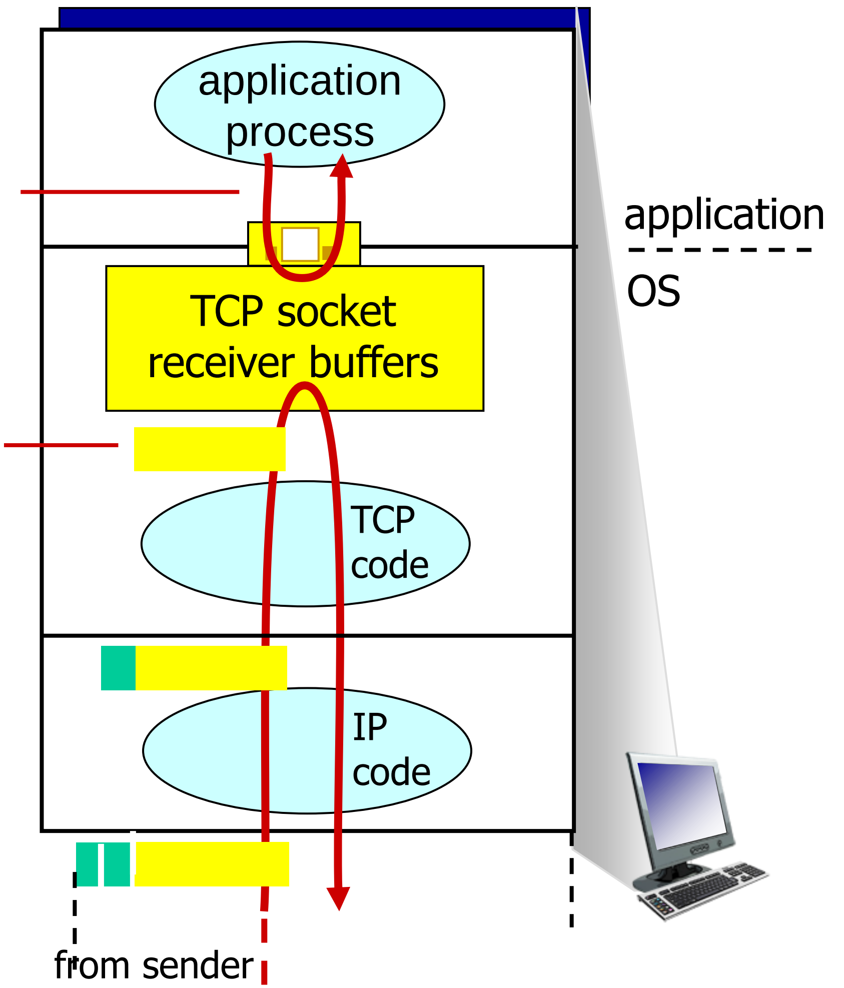
3.6 principles of congestion control

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)



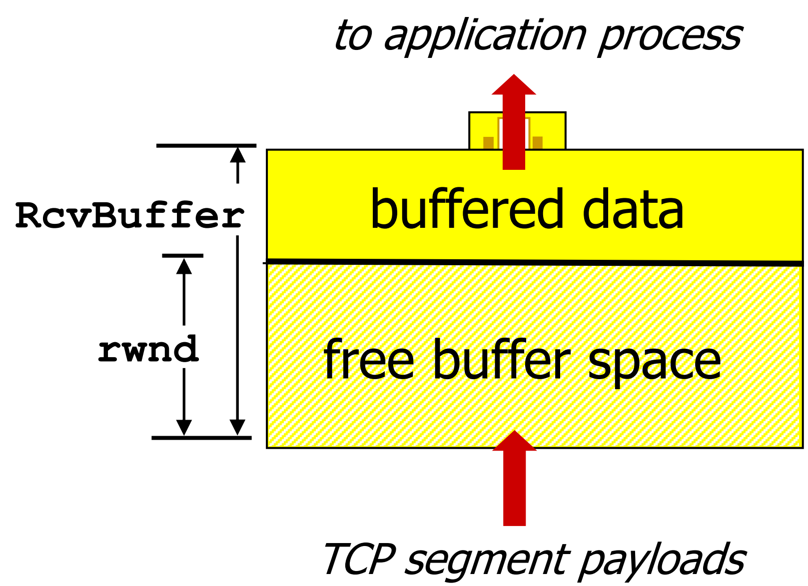
receiver protocol stack

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

- Receiver maintains
  - LastByteRead: last byte read by the application process
  - LastByteRcvd: last byte arrived in the receiver buffer

rwnd = RcvBuffer - [LastByteRead - LastByteRcvd]

- Sender maintains
  - LastByteSent
  - LastByteAcked
  - Ensure than

LastByteSent – LastByteAcked ≤ rwnd

TCP flow control

- One problem
  - Receiver announce rwnd=0 to the sender
  - Sender stops to send
  - Receiver has no data to acknowledge, sender has no way to know the receiver's new rwnd
- TCP specification requires sender to continue to send segments with one data byte when receiver's receive window is zero.

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
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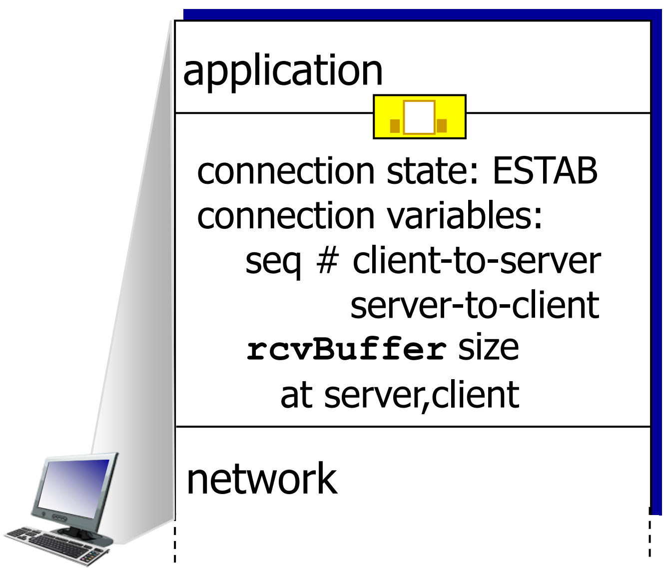
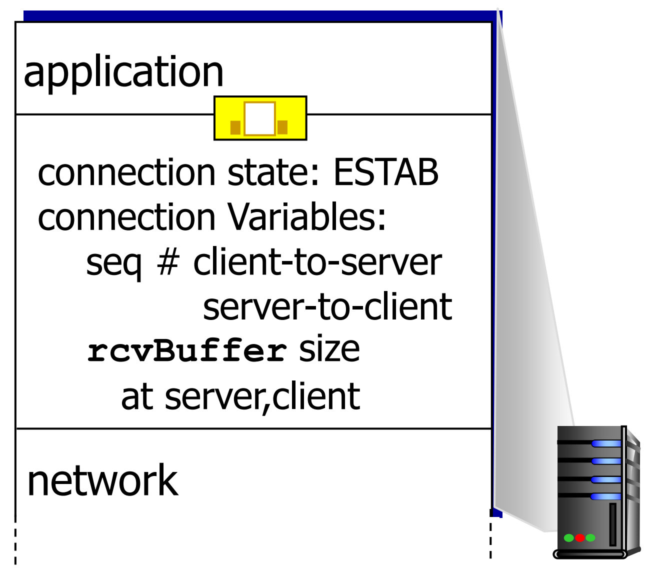
Connection Management

before exchanging data, sender/receiver "handshake":

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

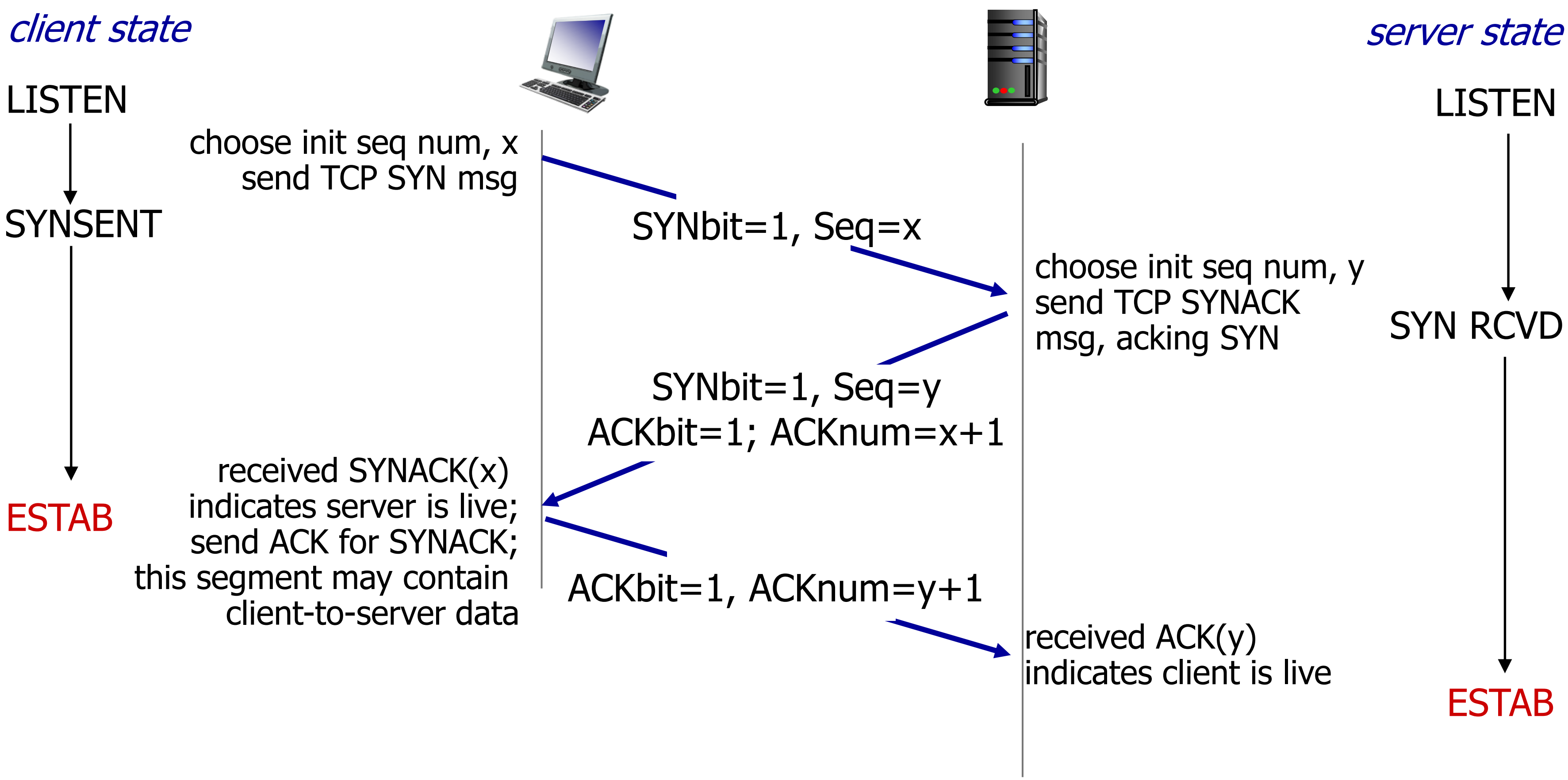



```
sock.connect("servername",
  port);
```




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

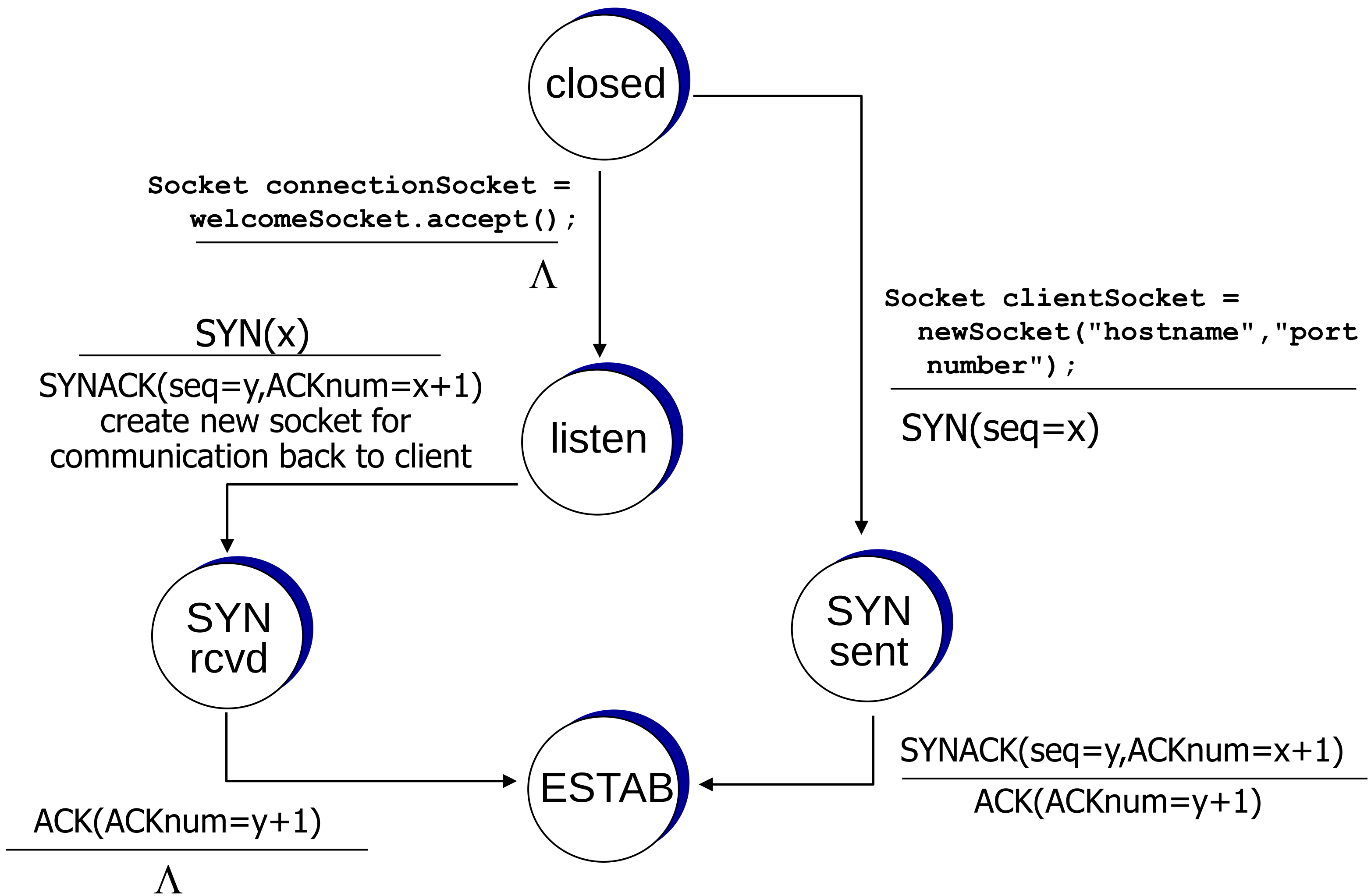
TCP 3-way handshake



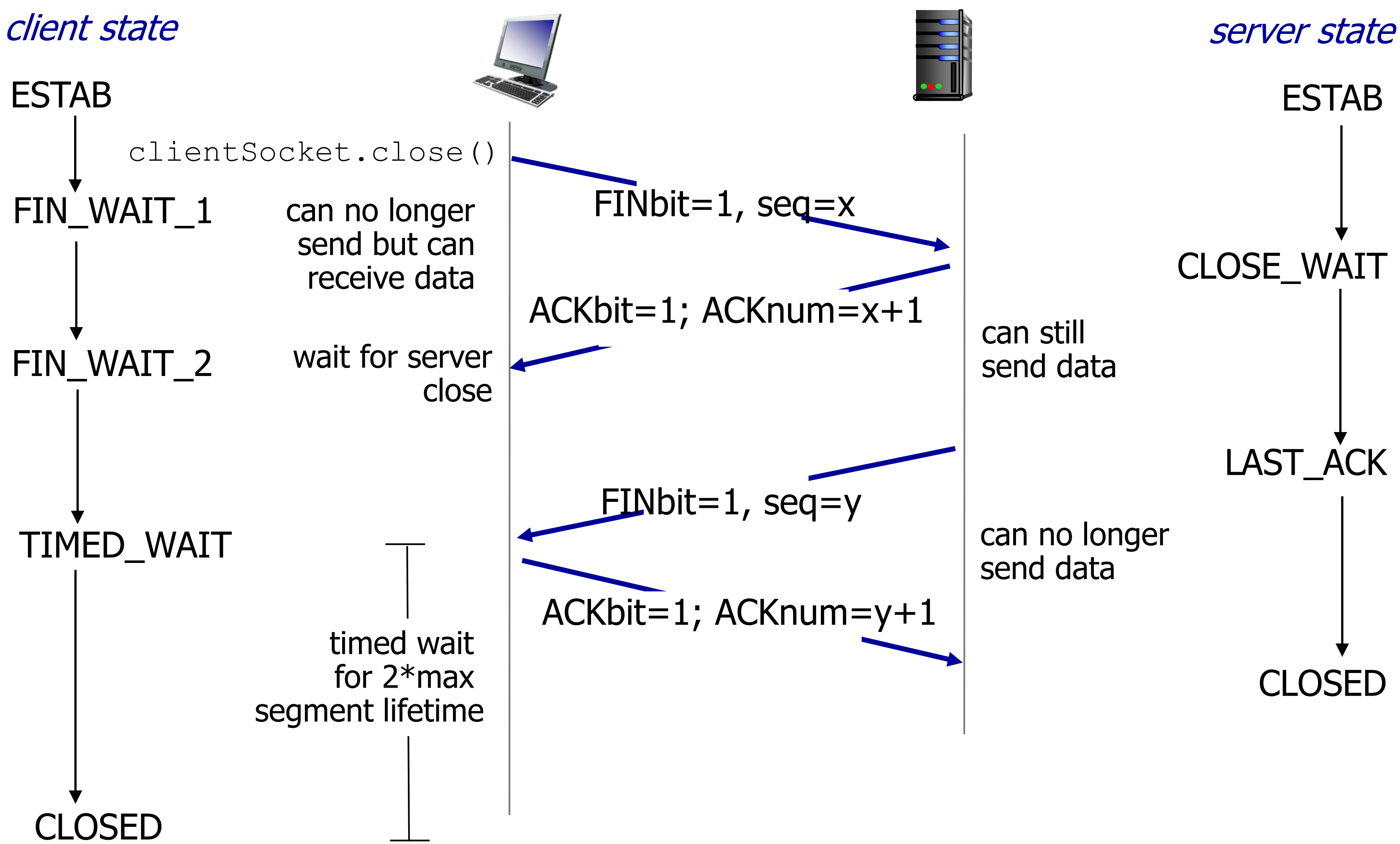
TCP 3-way handshake

- Client: send initial TCP SYN segment
  - SYNbit=1
  - Choose an initial sequence number x
- Server:
  - Receive TCP SYN segment, allocate buffer and variables
  - Send SYNACK segment (SYNbit=1, ACKbit=1), choose an initial sequence number y, acknowledge sequence number x+1
- Client:
  - Receive SYNACK segment
  - Send ACK, acknowledge sequence number y+1

TCP 3-way handshake: FSM



TCP: closing a connection



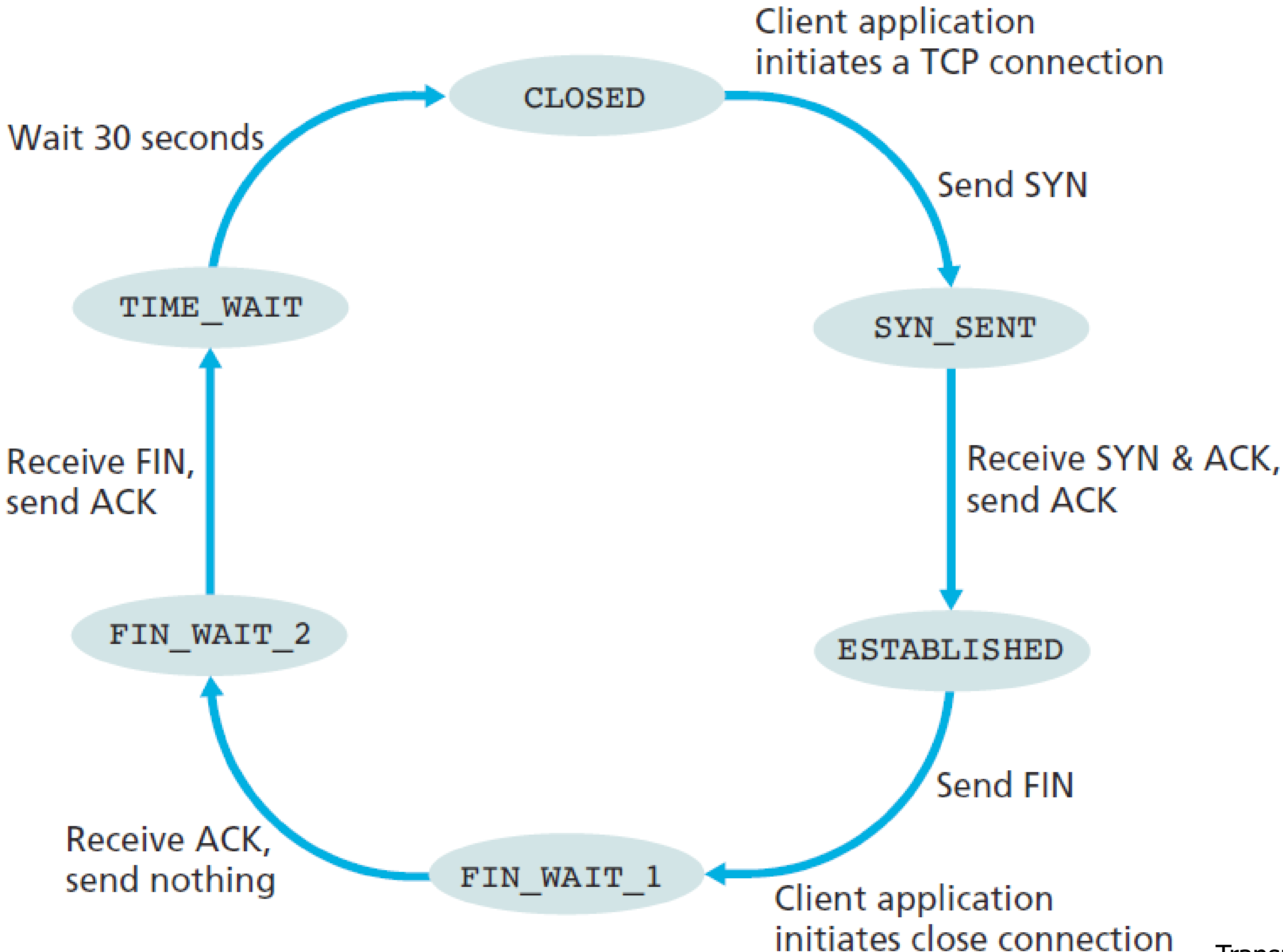
TCP: closing a connection

- Client:
  - Send segment with FINbit=1, enters into FIN_WAIT_1 state
  - Wait and receive acknowledgement from server, enters into FIN_WAIT_2 state
  - Wait and receive segment with FINbit=1, acknowledges the segment, enters into TIME_WAIT state
  - Stay in the TIME_WAIT state for a while, retransmit acknowledgement if lost, then enters into CLOSED state
- Server
  - Receive segment with FINbit=1 from the client, acknowledges it, enters into CLOSE_WAIT state
  - Send segment with FINbit=1, enters into LAST_ACK state
  - Receive acknowledgement from the client, enters into CLOSED state.

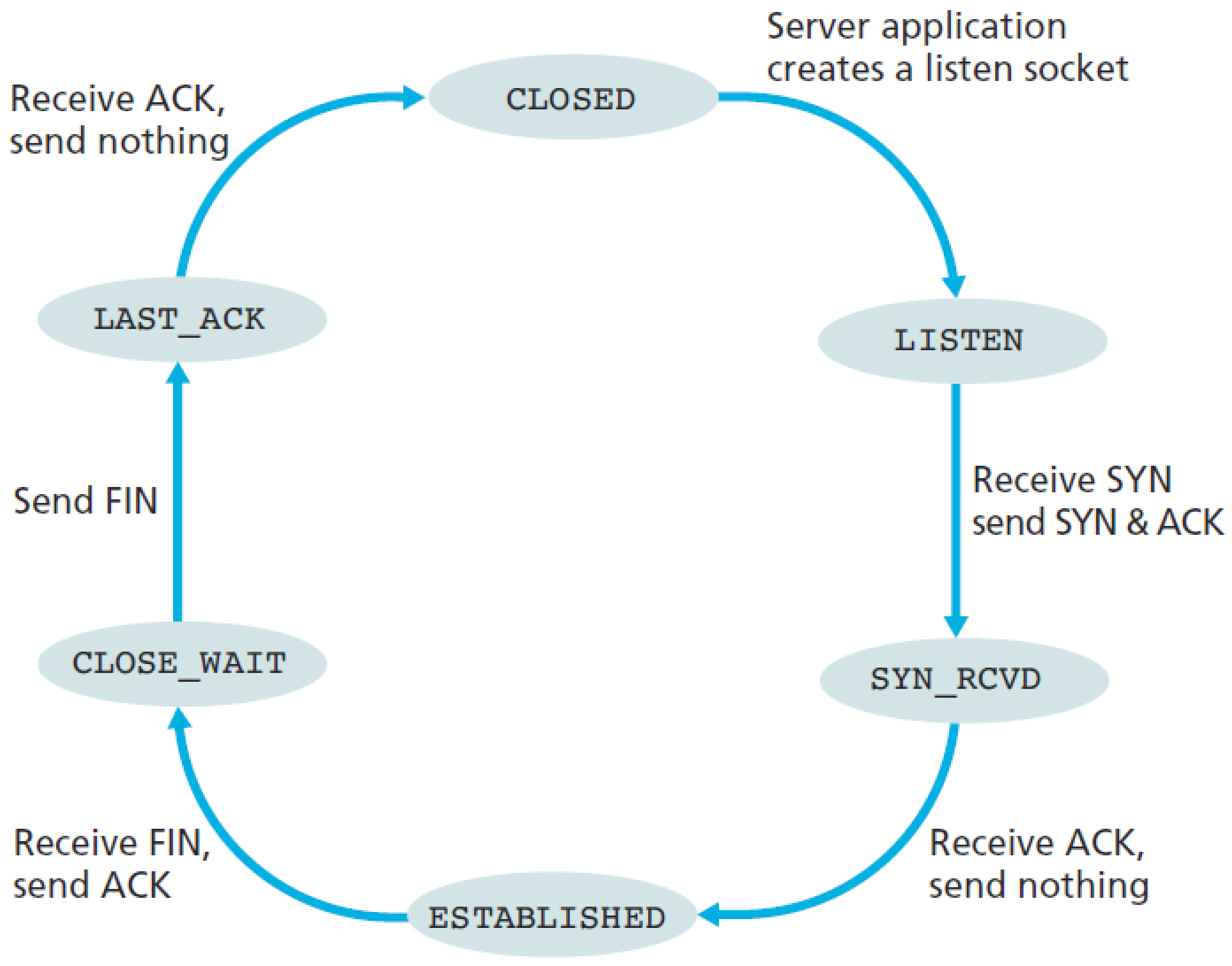
TCP: closing a connection

- 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

Client TCP FSM



Server TCP FSM



Reject connection

- A server receives a TCP SYN packet with destination port, but the host is not accepting connections on that port.
- Send a special reset segment to the source. This TCP segment has the RSTbit=1.

- nmap: port scan tool

Chapter 3 outline

3.1 transport-layer services

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3.4 principles of reliable data transfer

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3.6 principles of congestion control

3.7 TCP congestion control

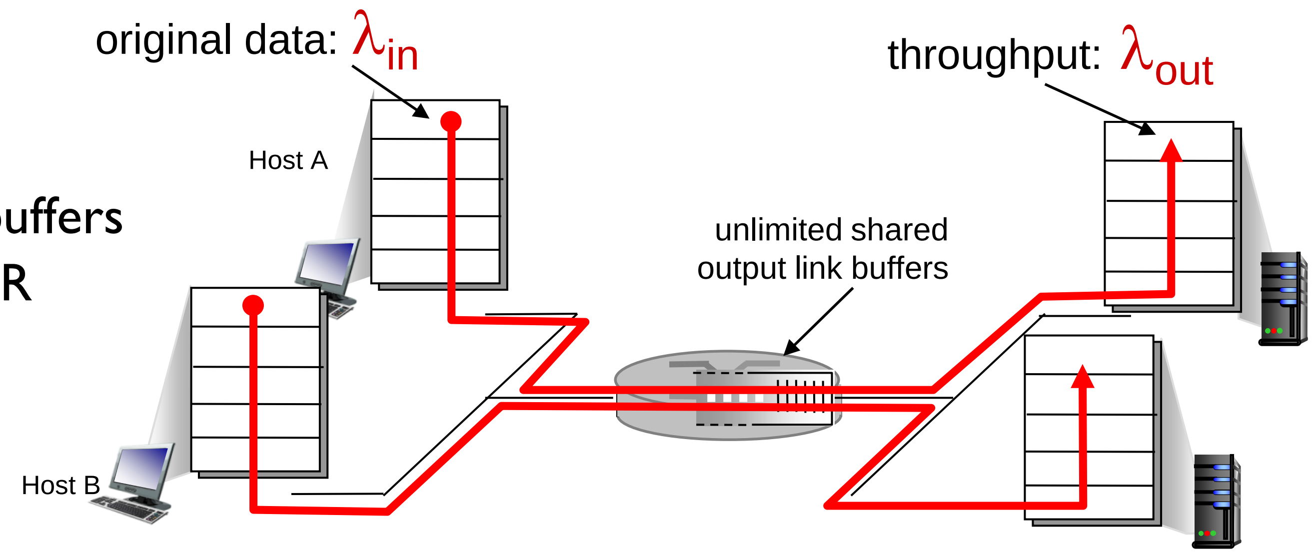
Principles of congestion control

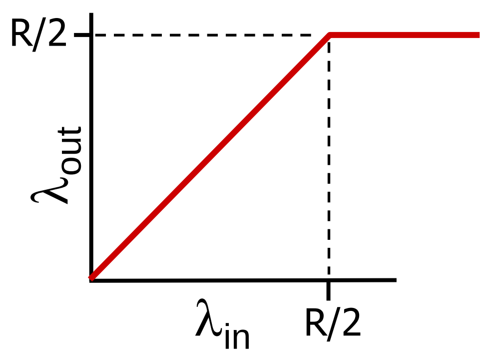
congestion:

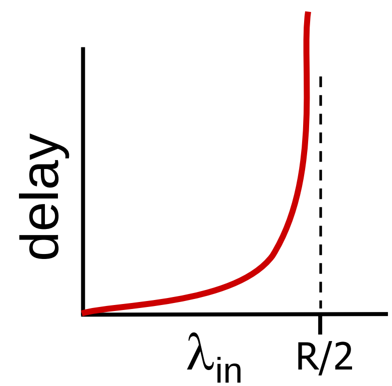
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

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





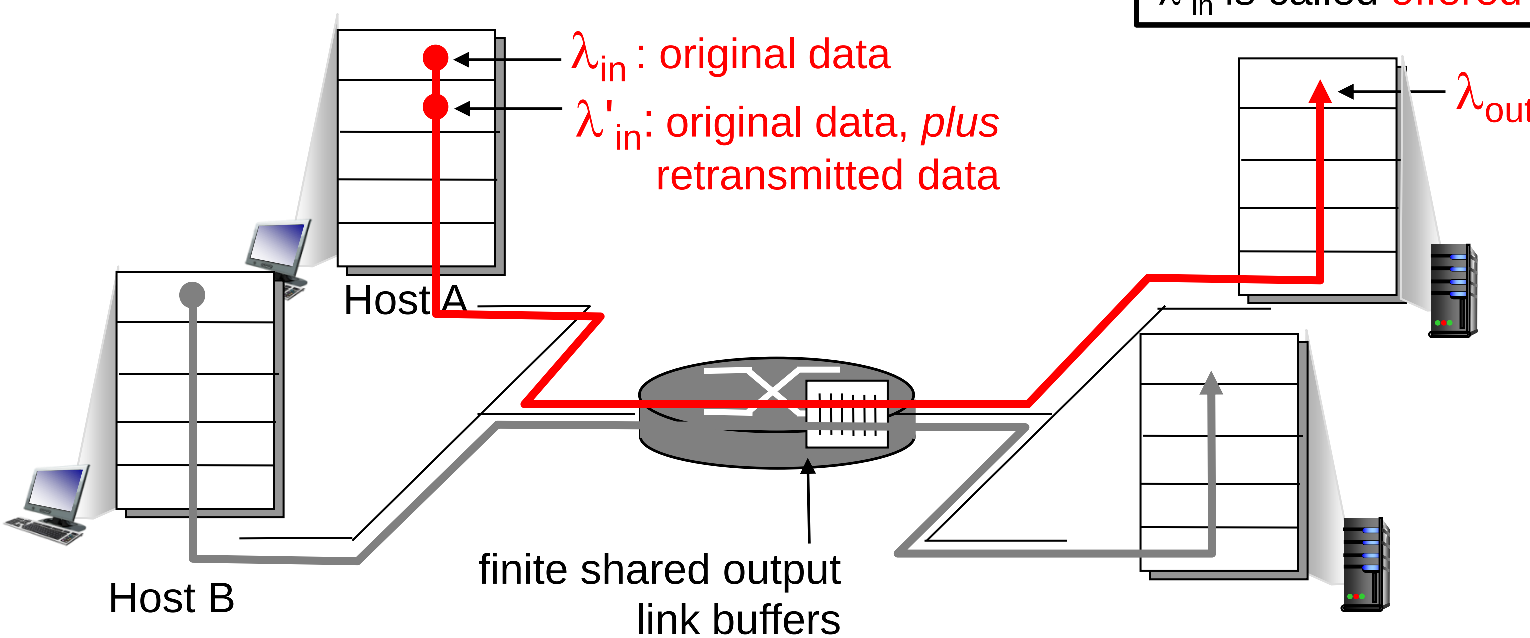


- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

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* : $\lambda'_{in} \geq \lambda_{in}$

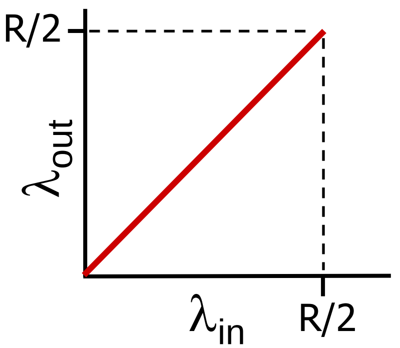
λ'_{in} is called offered load

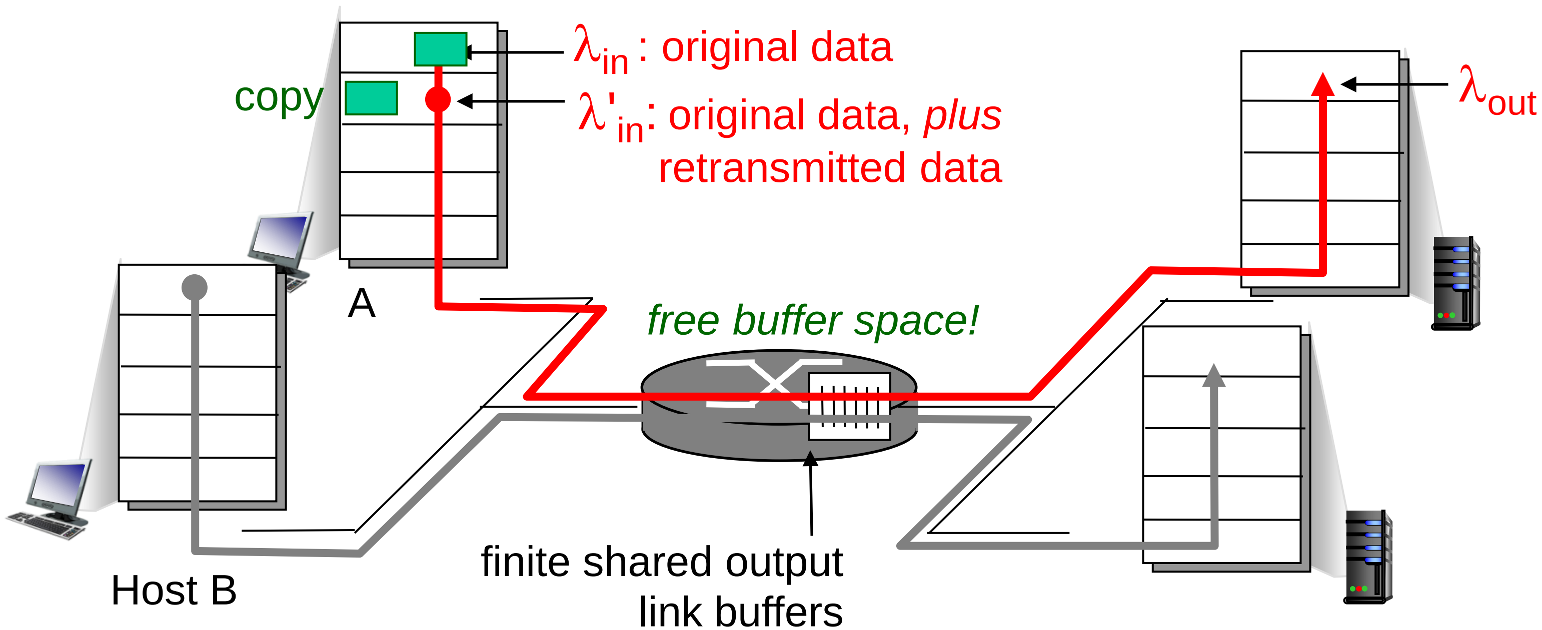


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available



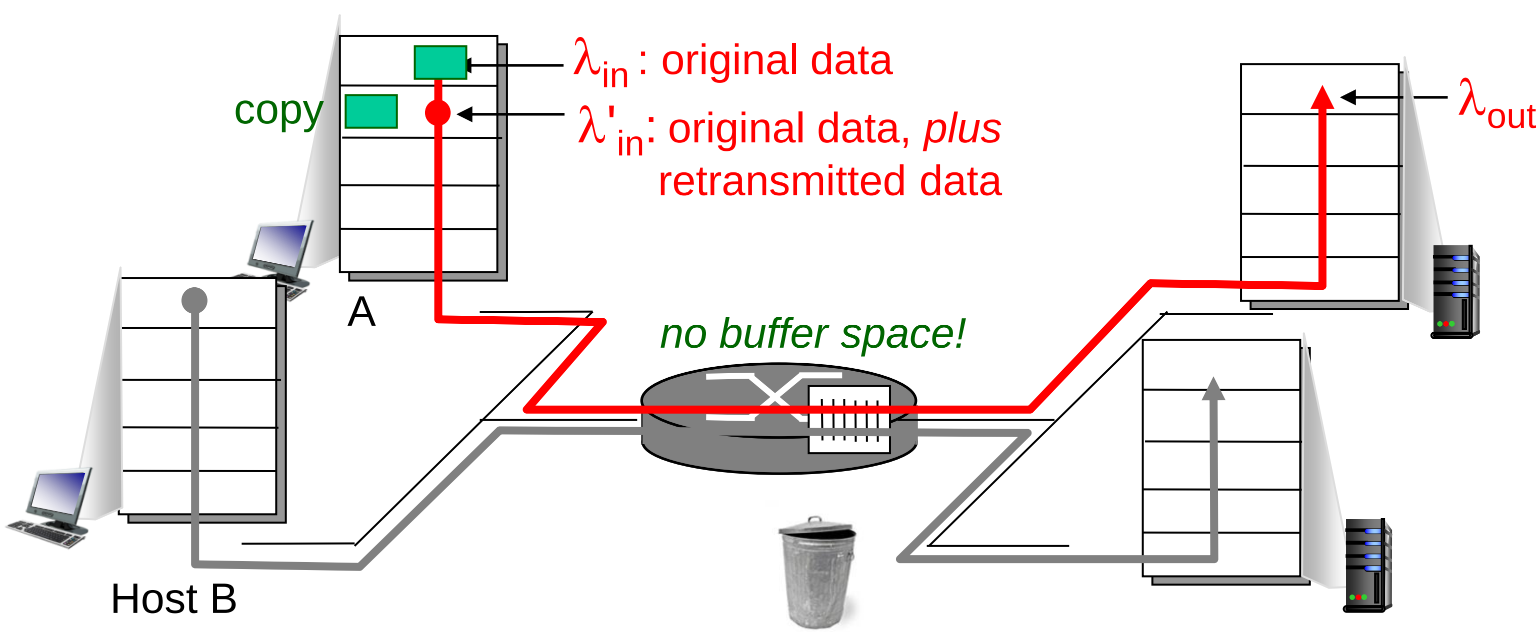


Causes/costs of congestion: scenario 2

Idealization: known loss

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

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



Causes/costs of congestion: scenario 2

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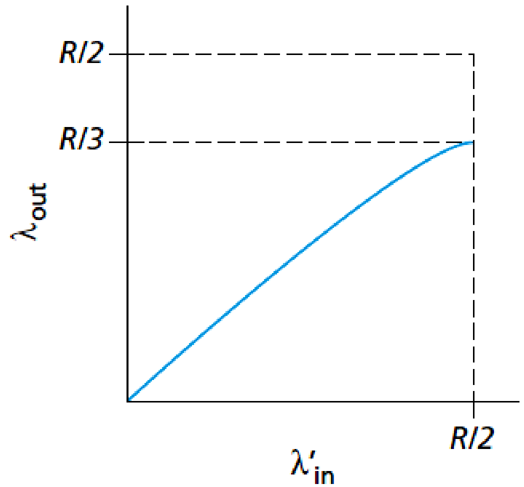
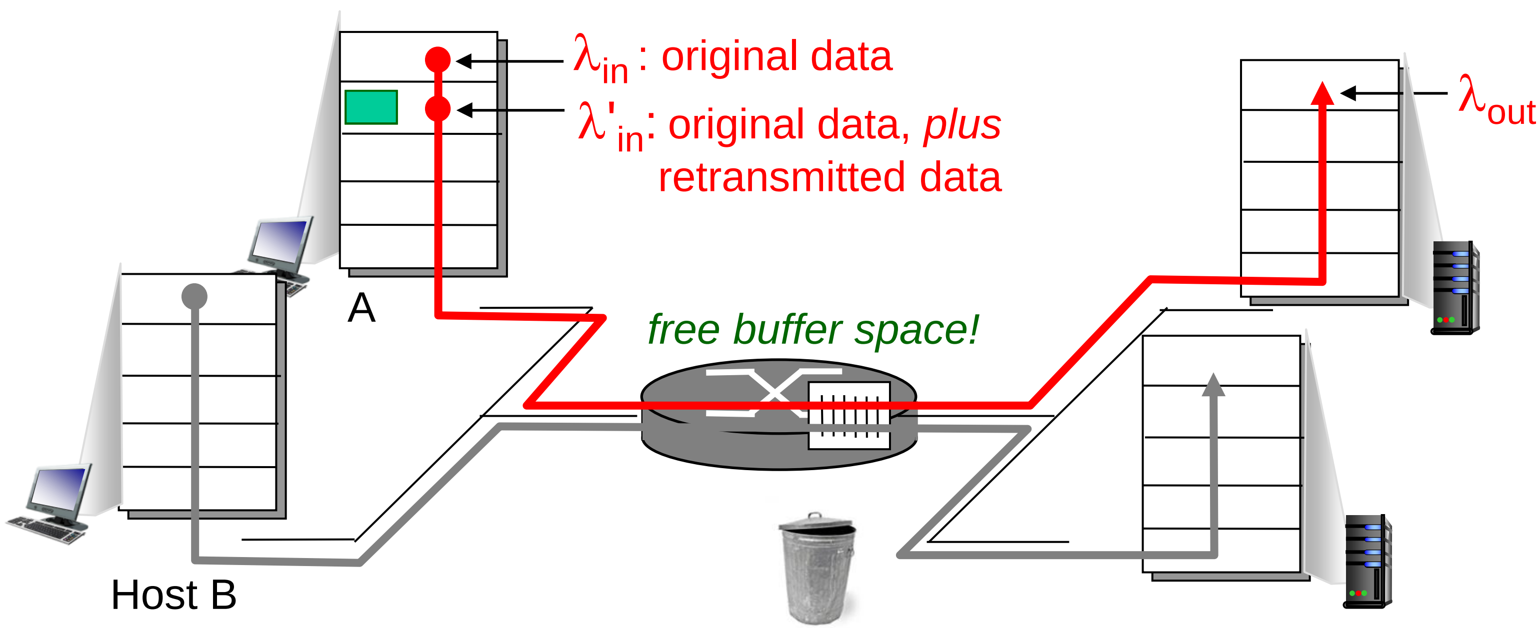


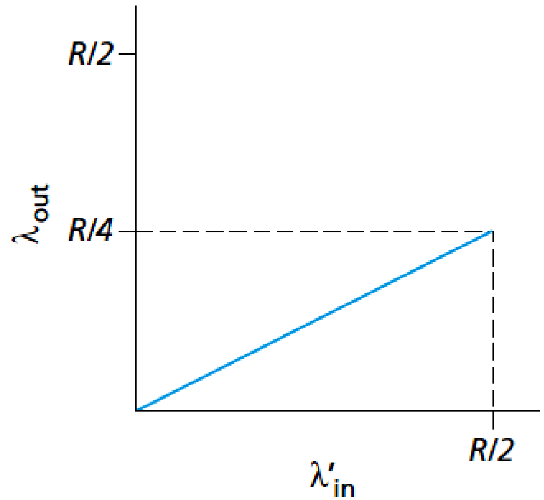
out of the 0.5R units of data transmitted, 0.333R bytes/sec (on average) are original data and 0.166R bytes/sec (on average) are retransmitted data.



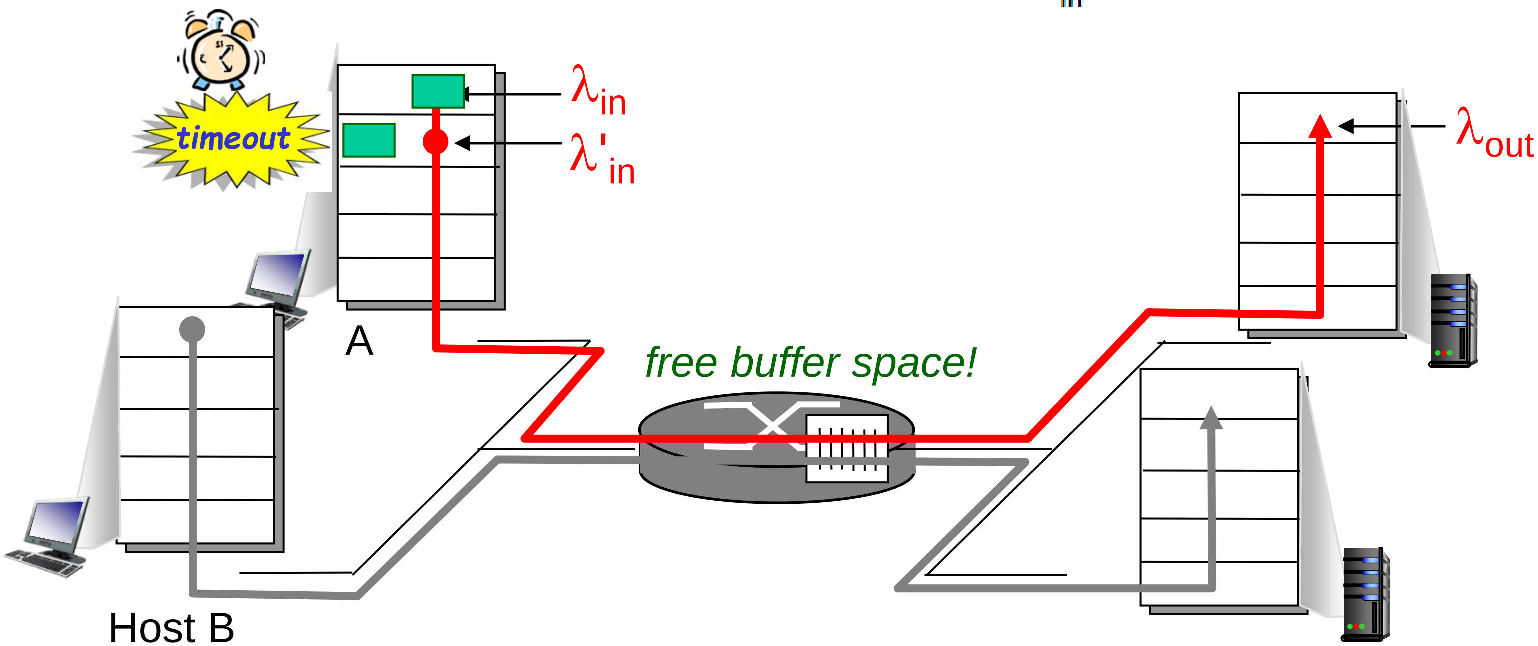
Causes/costs of congestion: scenario 2

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



the work done by the router in forwarding the retransmitted copy of the original packet was wasted.



Causes/costs of congestion: scenario 2

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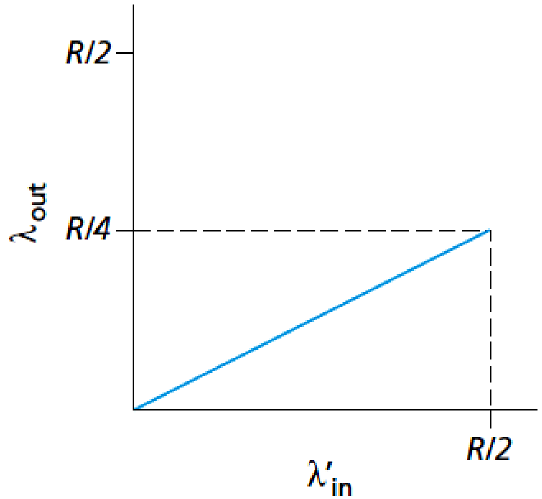


"costs" of congestion:

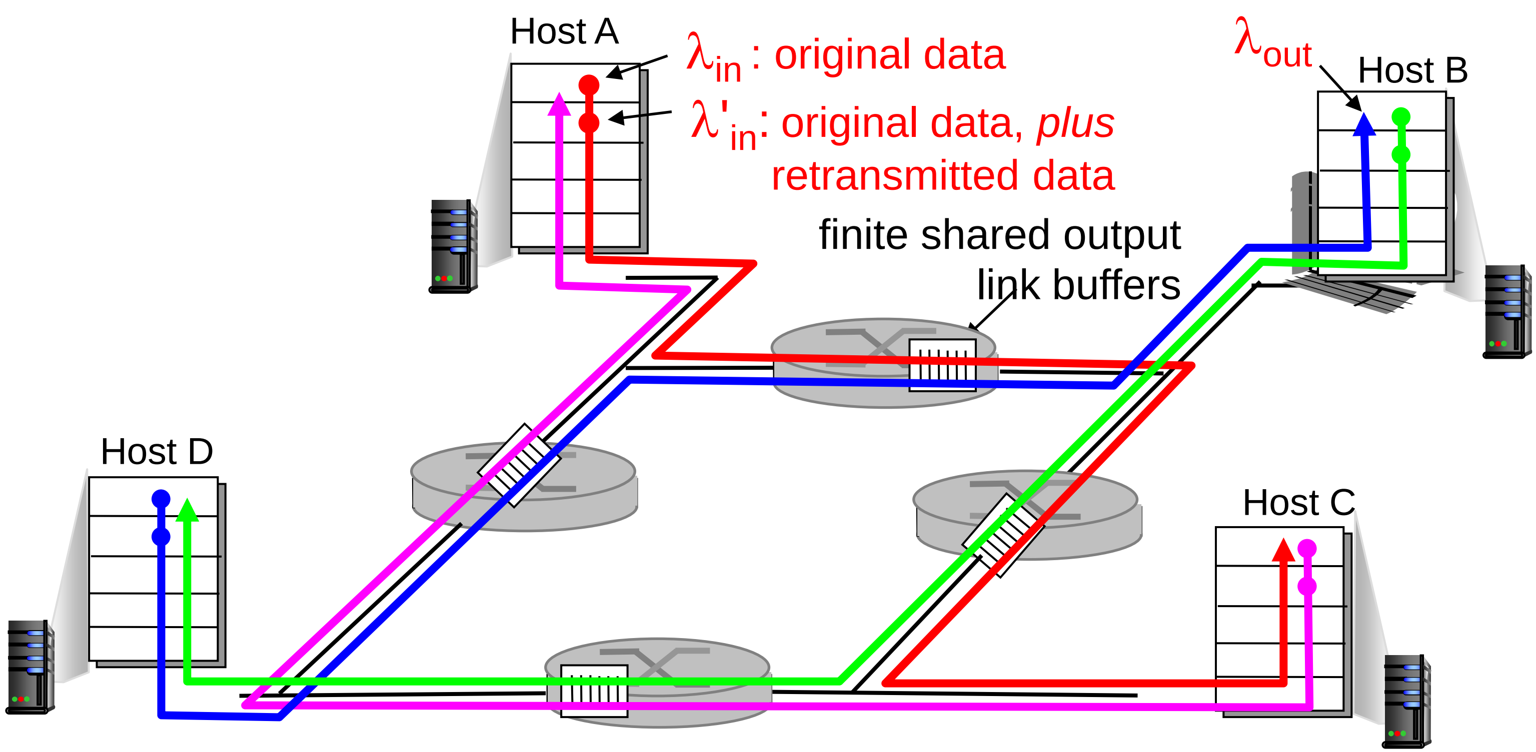
- delay
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

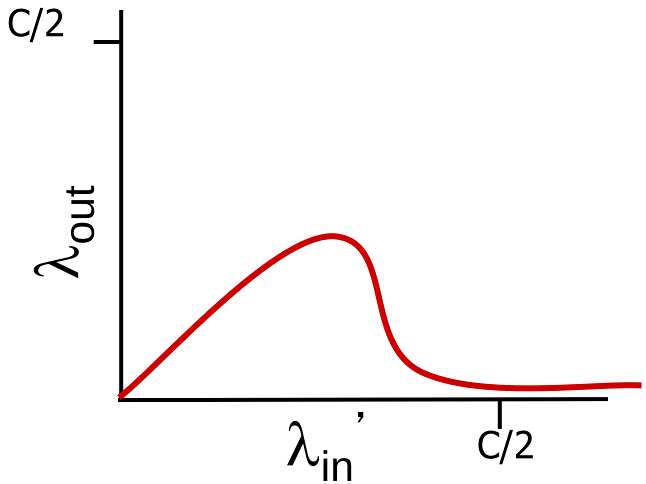
- four senders
- multihop paths
- timeout/retransmit

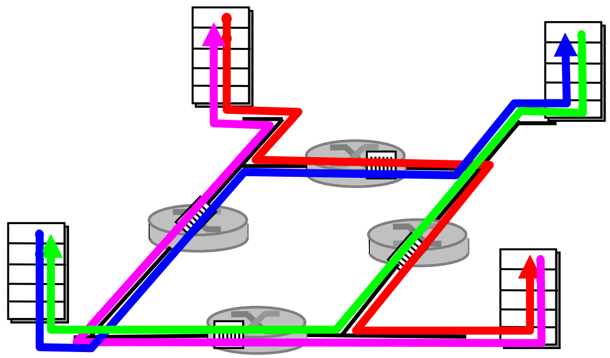
<u>Q:</u> what happens as λ_{in} and λ_{in}' increase ?

<u>A:</u> as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow$ 0



Causes/costs of congestion: scenario 3





Blue traffic arrives at router with rate of R (router-router capacity)
Red traffic arrives at router with extremely large rate λ_{in}'

another "cost" of congestion:

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

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

Congestion Window

- TCP stack maintains variables
  - LastByteSent
  - LastByteAcked
  - rwnd
- Sender's congestion control mechanism tracks additional variable
  - Congestion window, cwnd
  - Ensure that

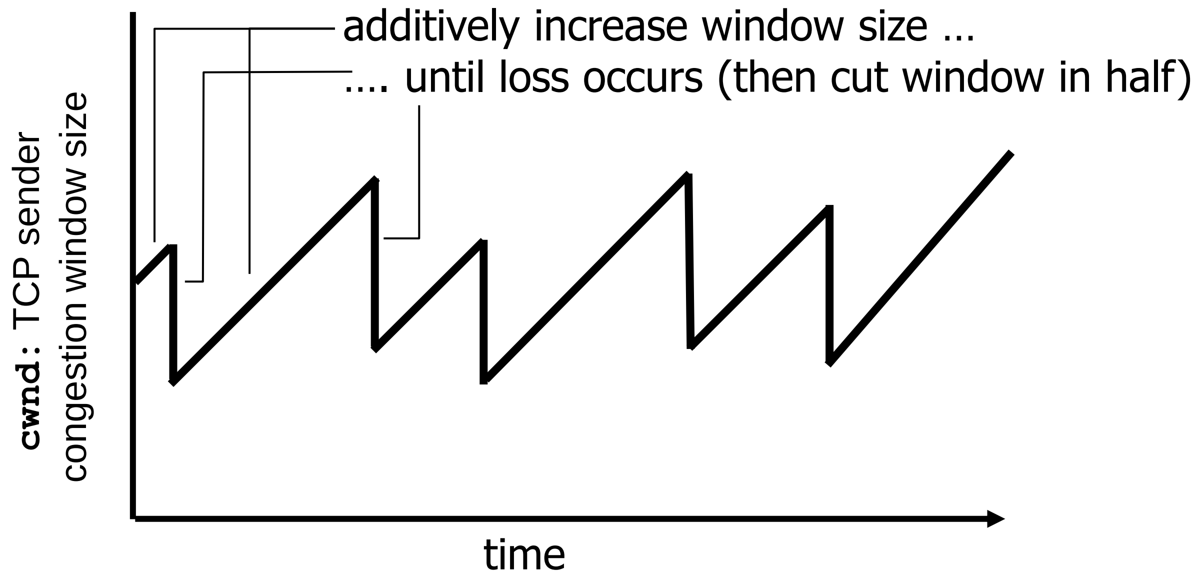
$$\texttt{LastByteSent} - \texttt{LastByteAcked} \leq \texttt{min\{cwnd, rwnd\}}$$

- Sender's data rate: *cwnd/RTT* (bytes/sec)

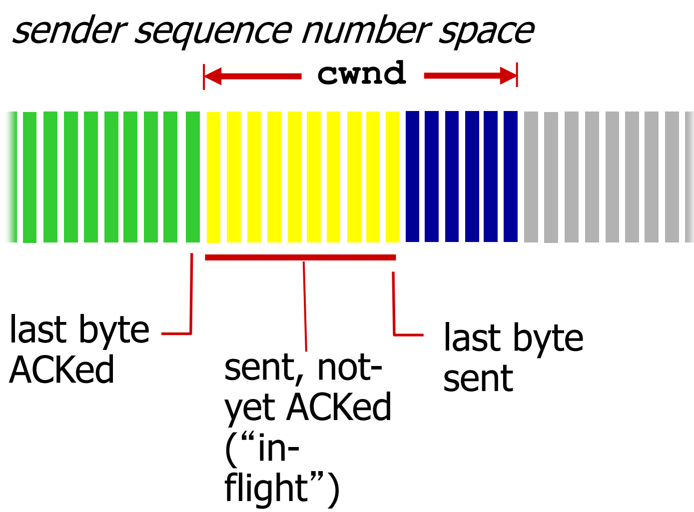
TCP congestion control: additive increase multiplicative decrease

- *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 limits transmission:

$$\texttt{LastByteSent-LastByteAcked} \leq \texttt{cwnd}$$

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

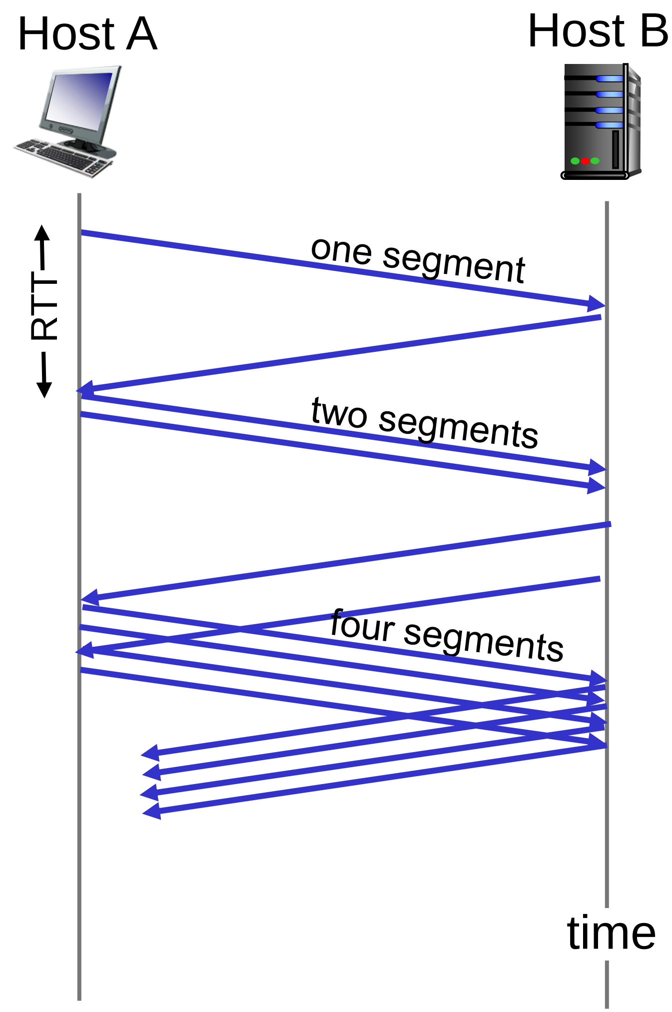
TCP sending rate:

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

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

TCP 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 is not slow!

TCP Slow Start

- When to end the exponential growth?
  1. if there is a loss event (i.e., congestion) indicated by a timeout,
     - the TCP sender sets the value of cwnd to 1 and begins the slow start process anew; It also sets the value of a second state variable, *ssthresh* to cwnd/2 (Tahoe and Reno)
  2. The second way in which slow start may end cwnd equals ssthresh, enter into congestion avoidance state
  3. The final way in which slow start can end is if three duplicate ACKs are detected, in which case TCP performs a fast retransmit and enters the fast recovery state.
     - Reno only

TCP: detecting, reacting to loss

- loss indicated by timeout (Tahoe and Reno):
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then goes to congestion avoidance state
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - Goes to the fast recovery state
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

Congestion Avoidance

- Increases the value of cwnd by just a single MSS every RTT
  - TCP sender increases cwnd by MSS bytes (MSS/cwnd) whenever a new ack arrives.
- When to end?
  - When a timeout occurs: The value of cwnd is set to 1 MSS, and the value of ssthresh is updated to half the value of cwnd when the loss event occurred.
  - Triple duplicate ACK event:
    - halves the value of cwnd (adding in 3 MSS to account for the triple duplicate ACKs received) and records the value of ssthresh to be half the value of cwnd when the triple duplicate ACKs were received. (Reno only)
    - Same as time out (Tahoe only)

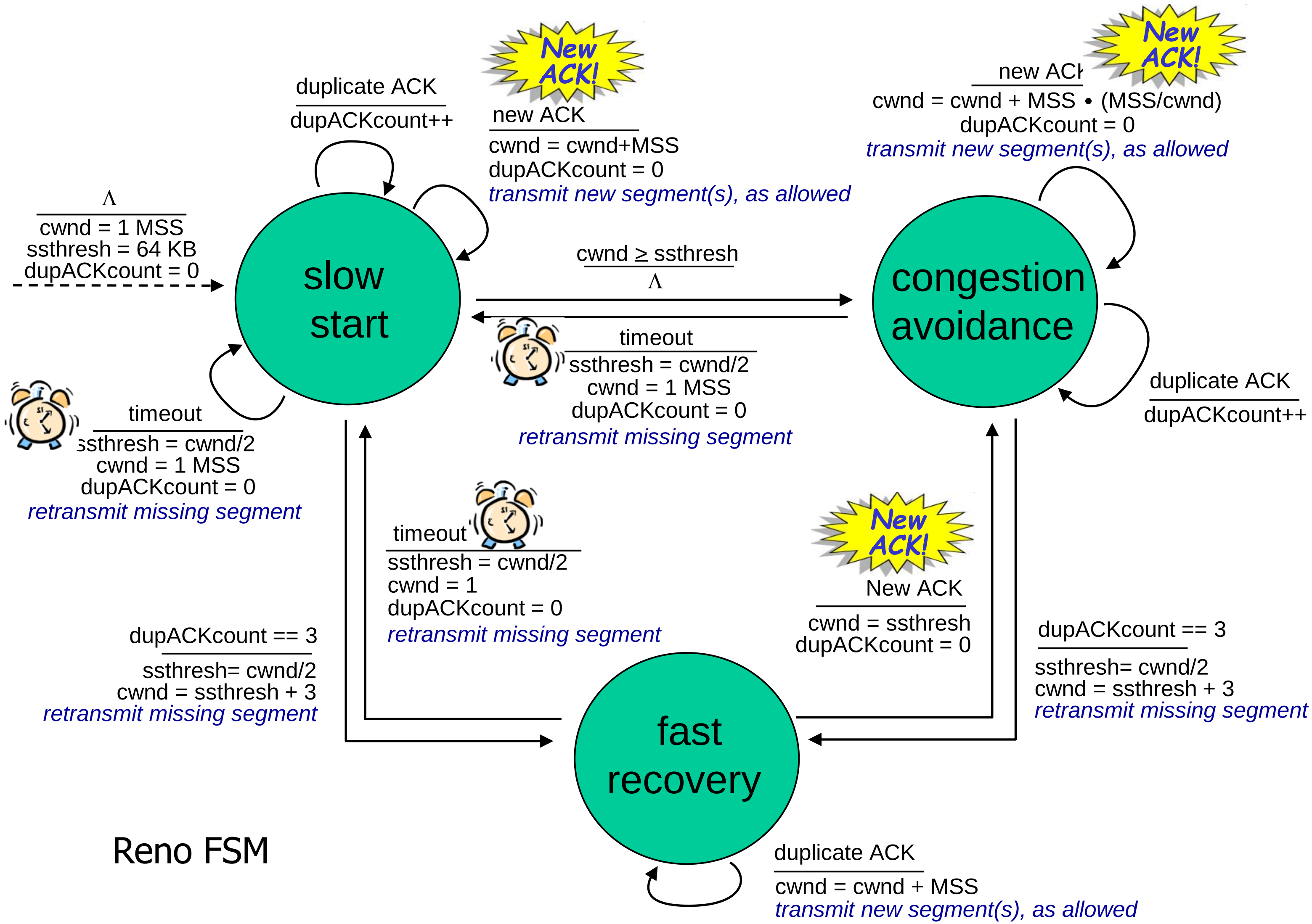
Fast Recovery

- In fast recovery, the value of cwnd is increased by 1 MSS for every duplicate ACK received for the missing segment that caused TCP to enter the fast-recovery state.
- Eventually, when an ACK arrives for the missing segment, TCP enters the congestion-avoidance state after deflating cwnd.
  - cwnd=ssthresh

Fast Recovery

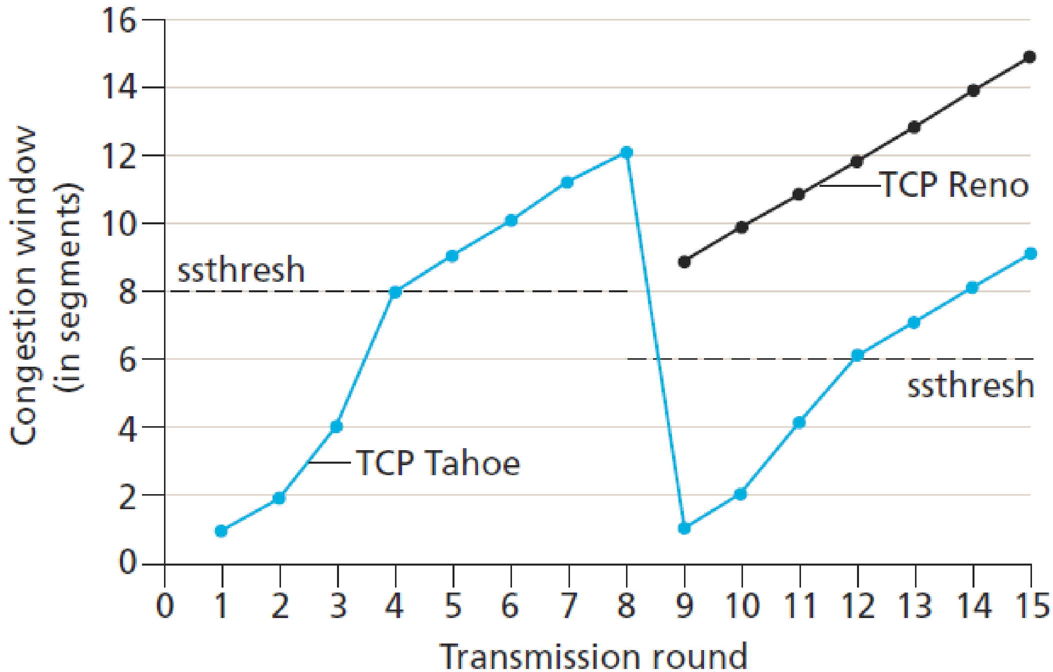
- If a timeout event occurs, fast recovery transitions to the slow-start state after performing the same actions as in slow start and congestion avoidance:
  - The value of cwnd is set to 1 MSS,
  - The value of ssthresh is set to half the value of cwnd when the loss event occurred.
- Reno has fast-recovery state, but Tahoe doesn't
- Tahoe always set cwnd = 1MSS and halves ssthresh when timeout or 3 every duplicate ACKs

Summary: TCP Congestion Control



Reno FSM

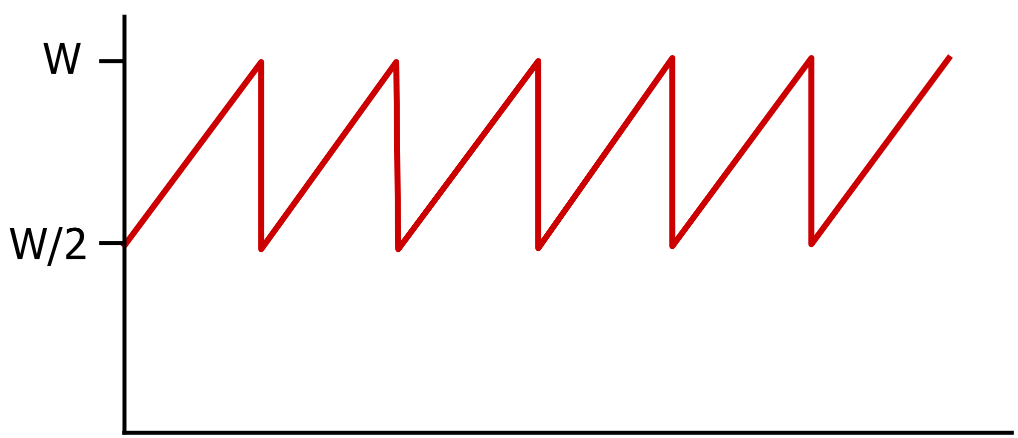
- Initially, ssthresh=8 MSS
- First 4 rounds, same for Tahoe and Reno
- Lost detected at round 8 (3 dup Ack)
    - Set ssthresh=6 MSS (both Tahoe and Reno)
    - Tahoe: cwnd drop to 1 MSS
    - Reno: cwnd drop to 6 +3=9 MSS



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) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

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



TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

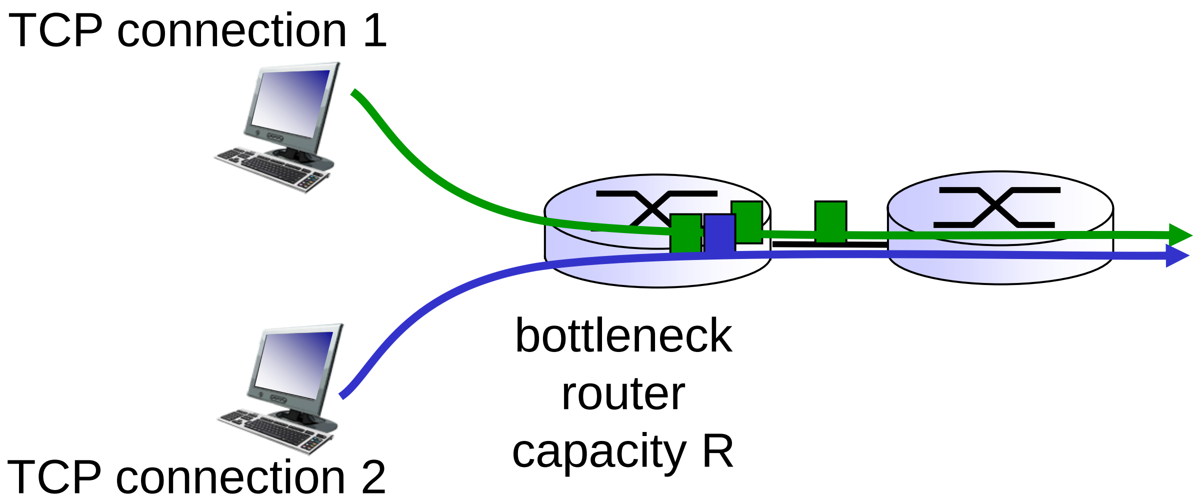
$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- Lead to research of new versions of TCP for high-speed

TCP Fairness

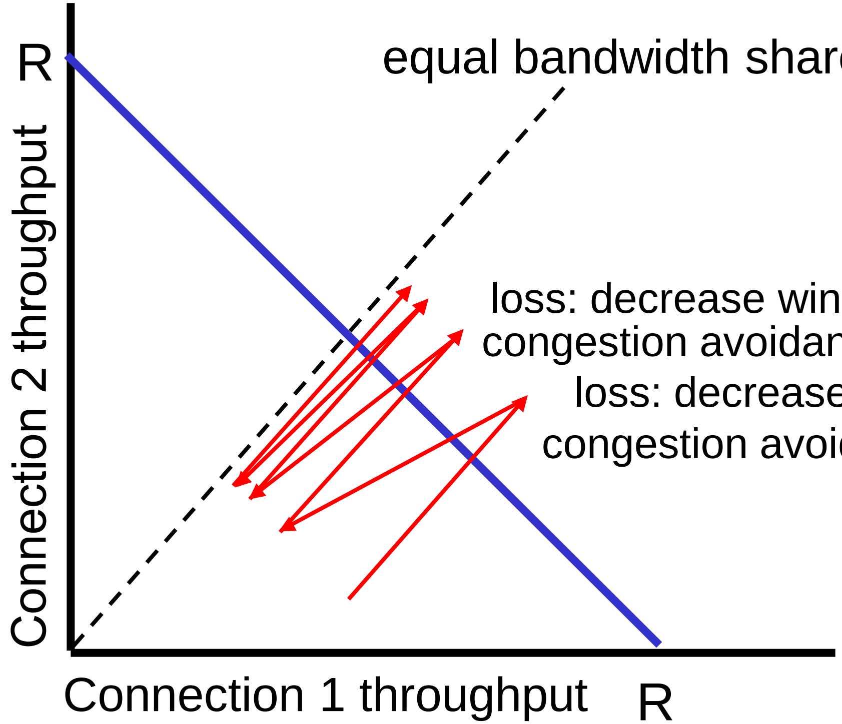
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

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



Fairness (more)

Fairness and UDP

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

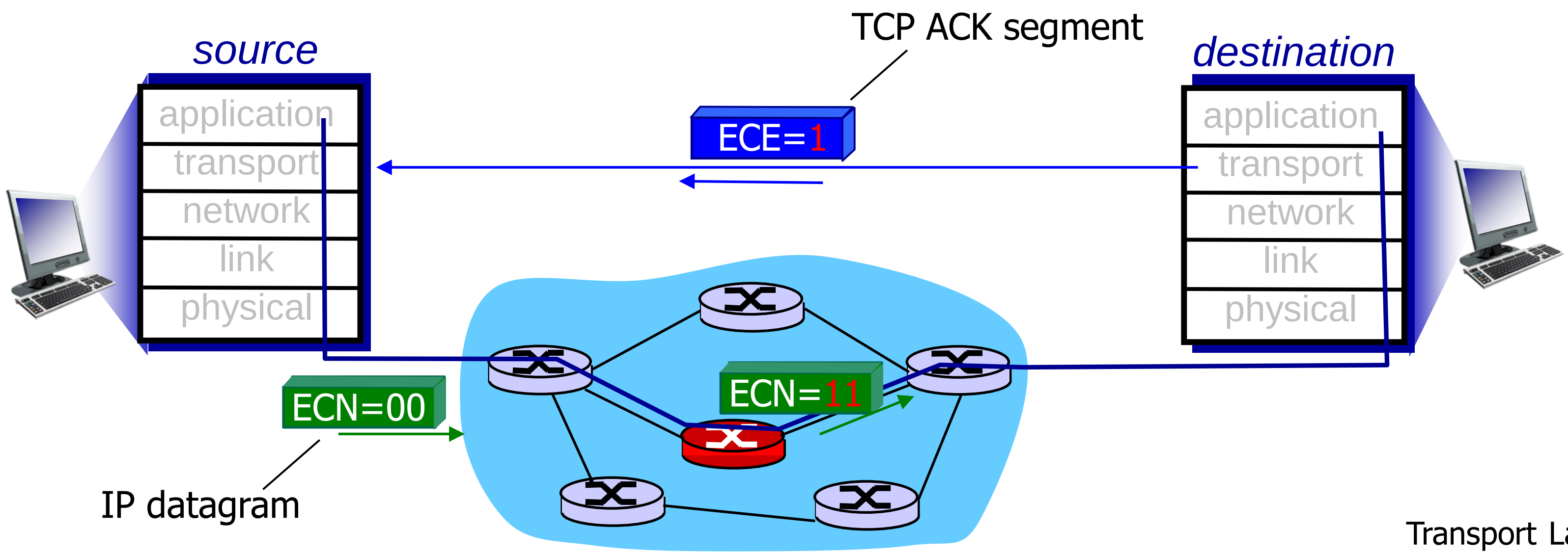
Fairness, parallel TCP connections

- 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 R/2

Explicit Congestion Notification (ECN)

network-assisted congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
- congestion indication carried to receiving host
- receiver (seeing congestion indication in IP datagram)) sets ECE bit on receiver-to-sender ACK segment to notify sender of congestion
- Sender halves cwnd, sets CWR bit on the next segment



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"
- two network layer chapters:
  - data plane
  - control plane