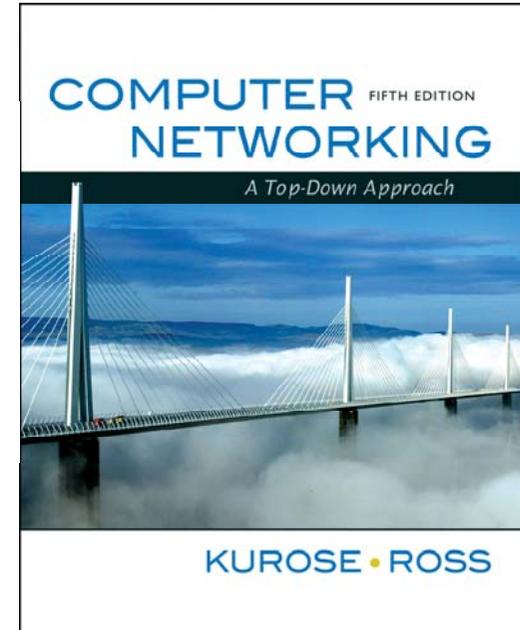


Chapter 3

Transport Layer



A note on the use of these ppt slides:

We're making these slides freely available to all (faculty, students, readers). They're in PowerPoint form so you can add, modify, and delete slides (including this one) and slide content to suit your needs. They obviously represent a *lot* of work on our part. In return for use, we only ask the following:

- If you use these slides (e.g., in a class) in substantially unaltered form, that you mention their source (after all, we'd like people to use our book!)
- If you post any slides in substantially unaltered form on a www site, that you note that they are adapted from (or perhaps identical to) our slides, and note our copyright of this material.

Thanks and enjoy! JFK/KWR

All material copyright 1996-2009
J.F Kurose and K.W. Ross, All Rights Reserved

*Computer Networking:
A Top Down Approach
5th edition.*

*Jim Kurose, Keith Ross
Addison-Wesley, April
2009.*

Chapter 3: Transport Layer

Our goals:

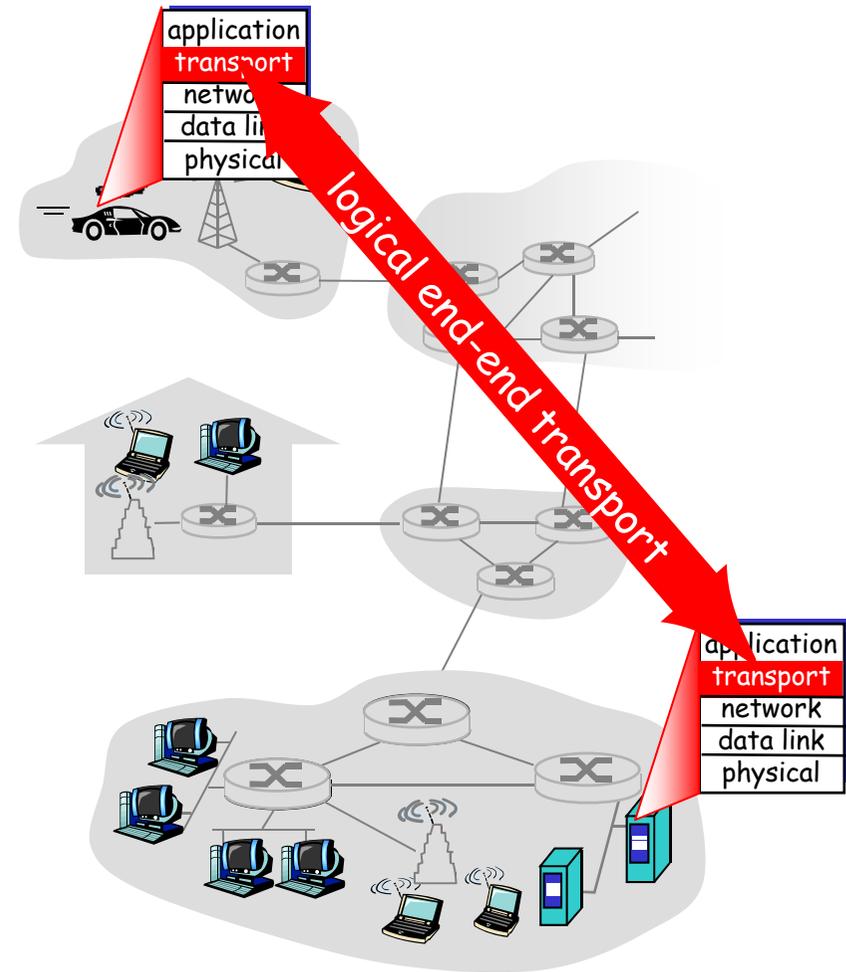
- ❑ understand principles behind transport layer services:
 - multiplexing/demultiplexing
 - reliable data transfer
 - flow control
 - congestion control
- ❑ learn about transport layer protocols in the Internet:
 - UDP: connectionless transport
 - TCP: connection-oriented transport
 - TCP congestion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

Chapter 3 outline

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

Transport services and protocols

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



Transport vs. network layer

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

Household analogy:

12 kids sending letters to 12 kids

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

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

Multiplexing/demultiplexing

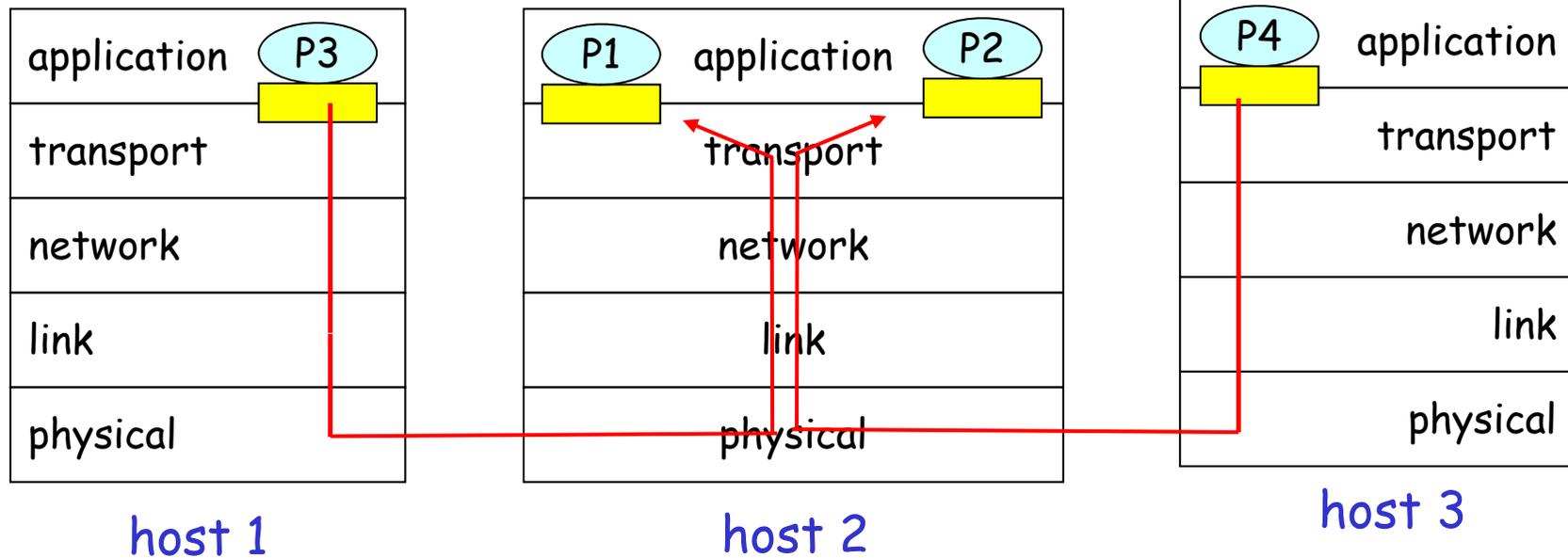
Demultiplexing at rcv host:

delivering received segments to correct socket

Multiplexing at send host:

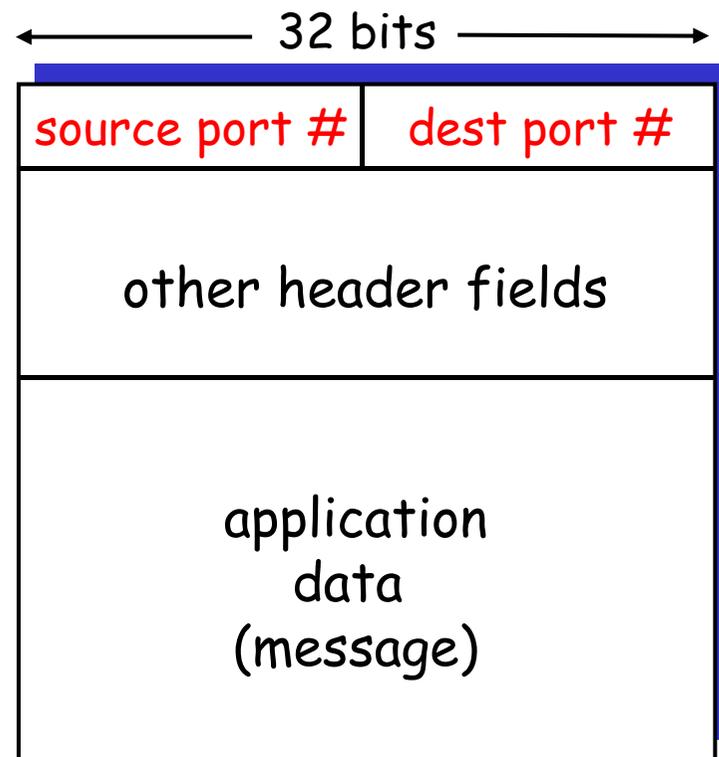
gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

■ = socket ○ = process



How demultiplexing works

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



TCP/UDP segment format

Connectionless demultiplexing

- ❑ Create sockets with port numbers:

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

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

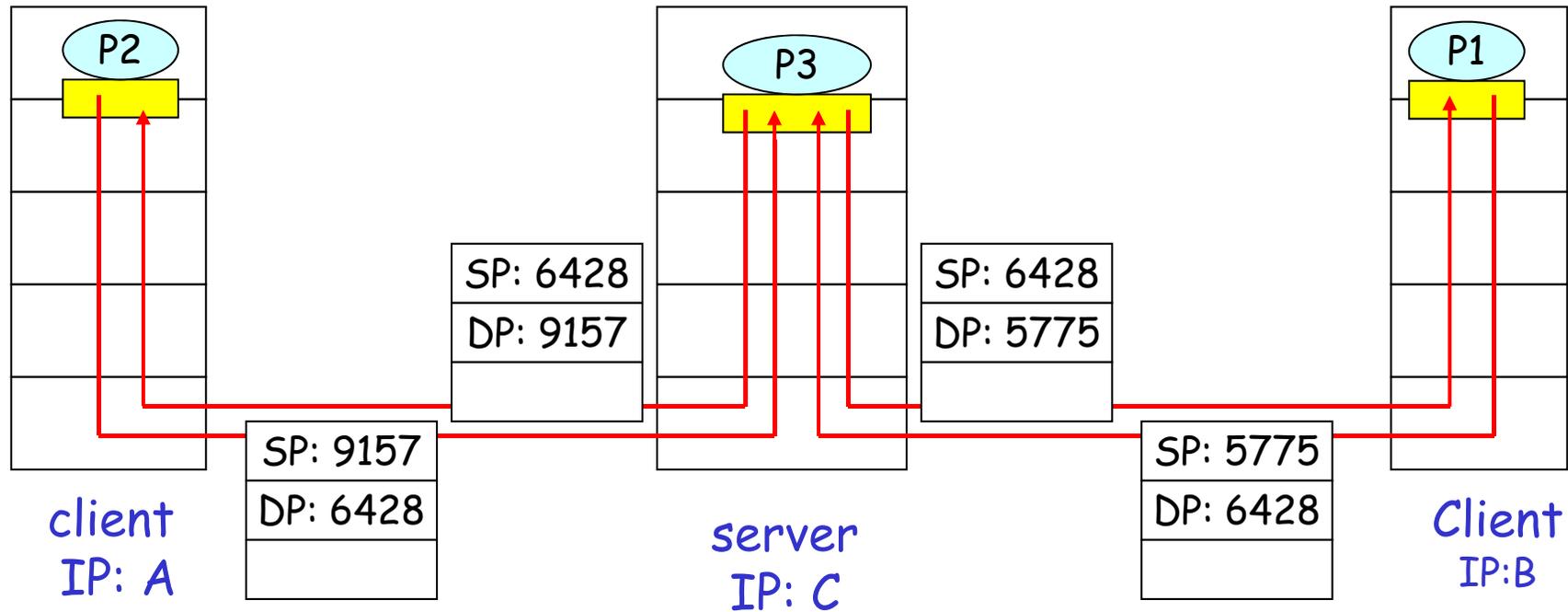
- ❑ UDP socket identified by two-tuple:

(dest IP address, dest port number)

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

Connectionless demux (cont)

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

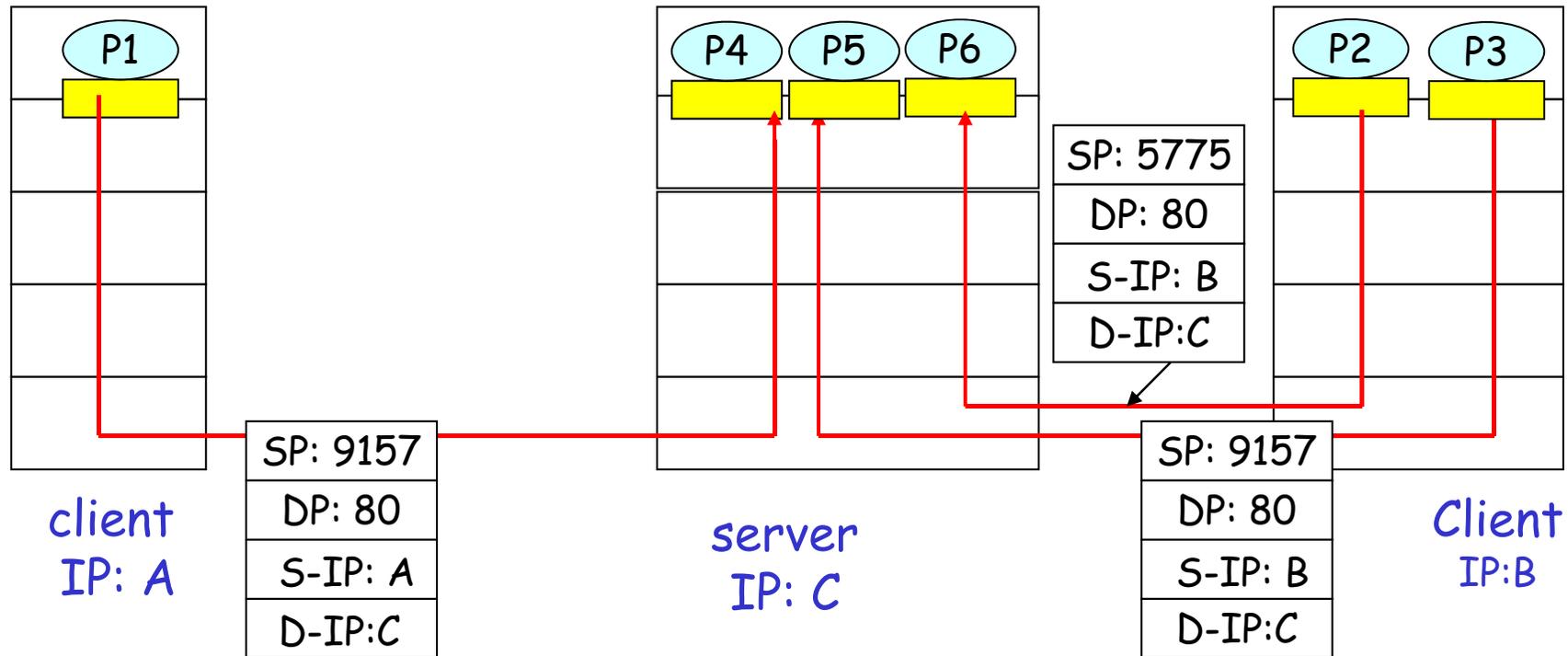


SP provides "return address"

Connection-oriented demux

- ❑ TCP socket identified by 4-tuple:
 - source IP address
 - source port number
 - dest IP address
 - dest port number
- ❑ rcv host uses all four values to direct segment to appropriate socket
- ❑ Server host may support many simultaneous TCP sockets:
 - each socket identified by its own 4-tuple
- ❑ Web servers have different sockets for each connecting client
 - non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



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

UDP: User Datagram Protocol [RFC 768]

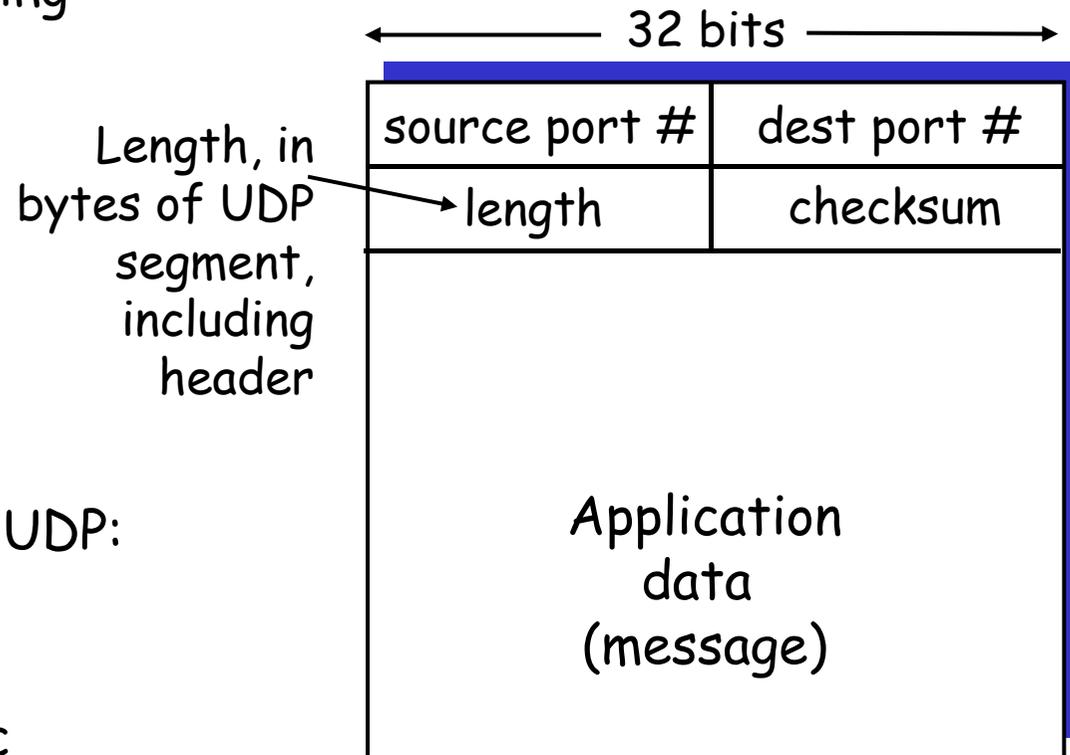
- ❑ “no frills,” “bare bones” Internet transport protocol
- ❑ “best effort” service, UDP segments may be:
 - lost
 - delivered out of order to app
- ❑ *connectionless*:
 - no handshaking between UDP sender, receiver
 - each UDP segment handled independently of others

Why is there a UDP?

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

UDP: more

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



UDP segment format

UDP checksum

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

Sender:

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

Receiver:

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

Internet Checksum Example

- Note

- When adding numbers, a carryout from the most significant bit needs to be added to the result

- Example: add two 16-bit integers

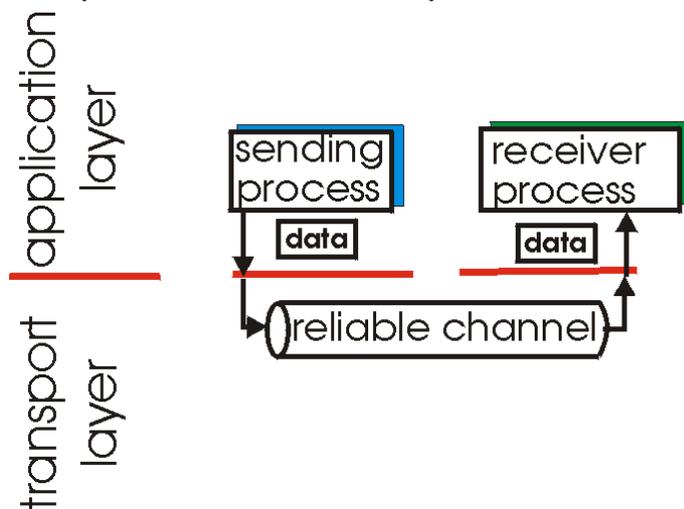
		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
		<hr/>															
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
		<hr/>															
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

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 Reliable data transfer

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

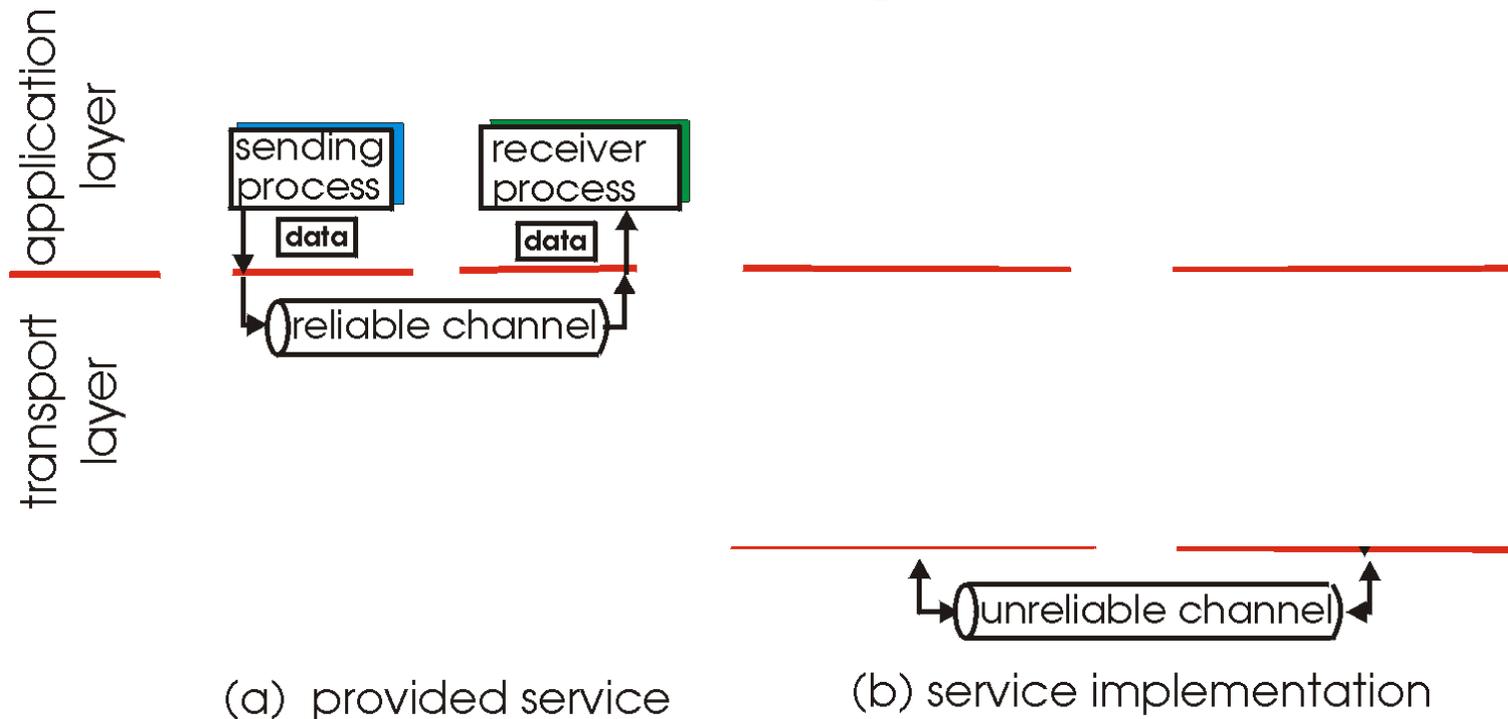


(a) provided service

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

Principles of Reliable data transfer

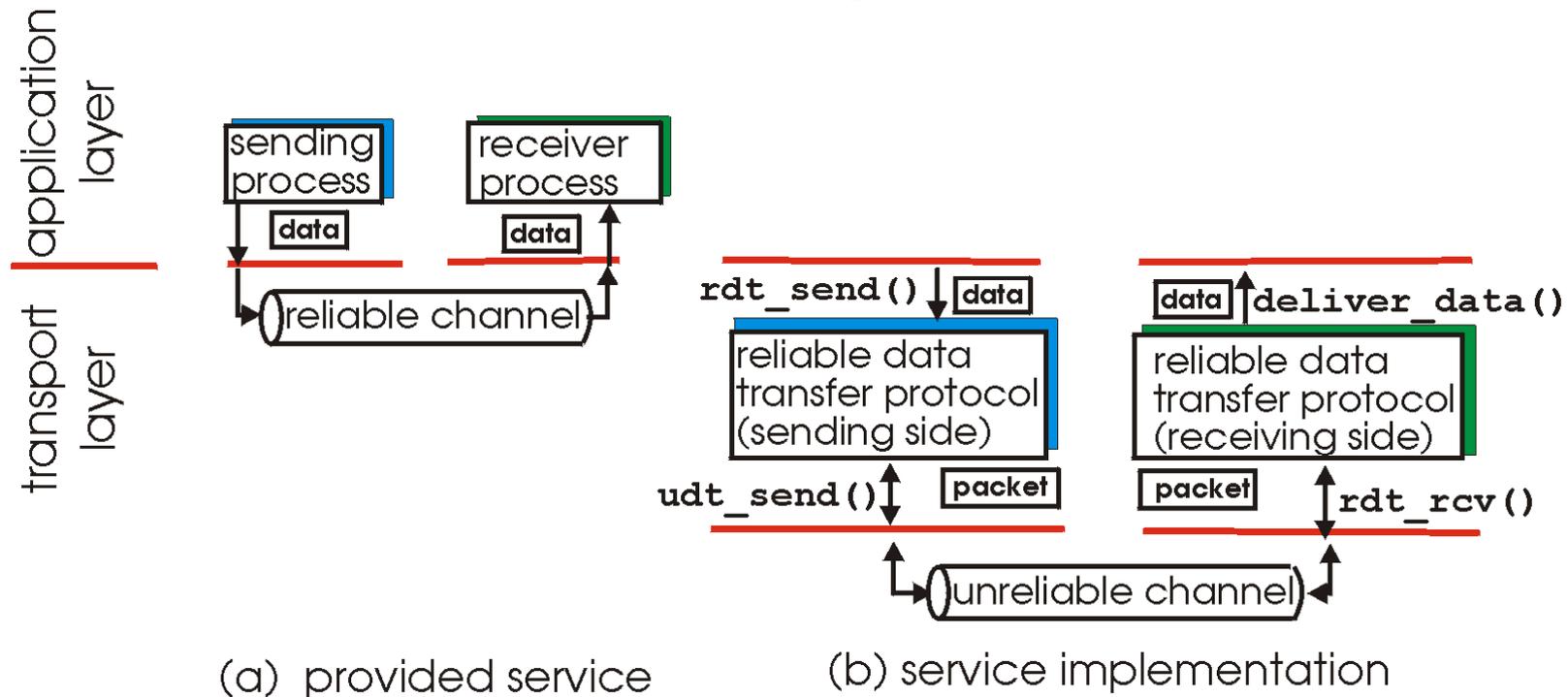
- important in app., transport, link layers
- top-10 list of important networking topics!



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

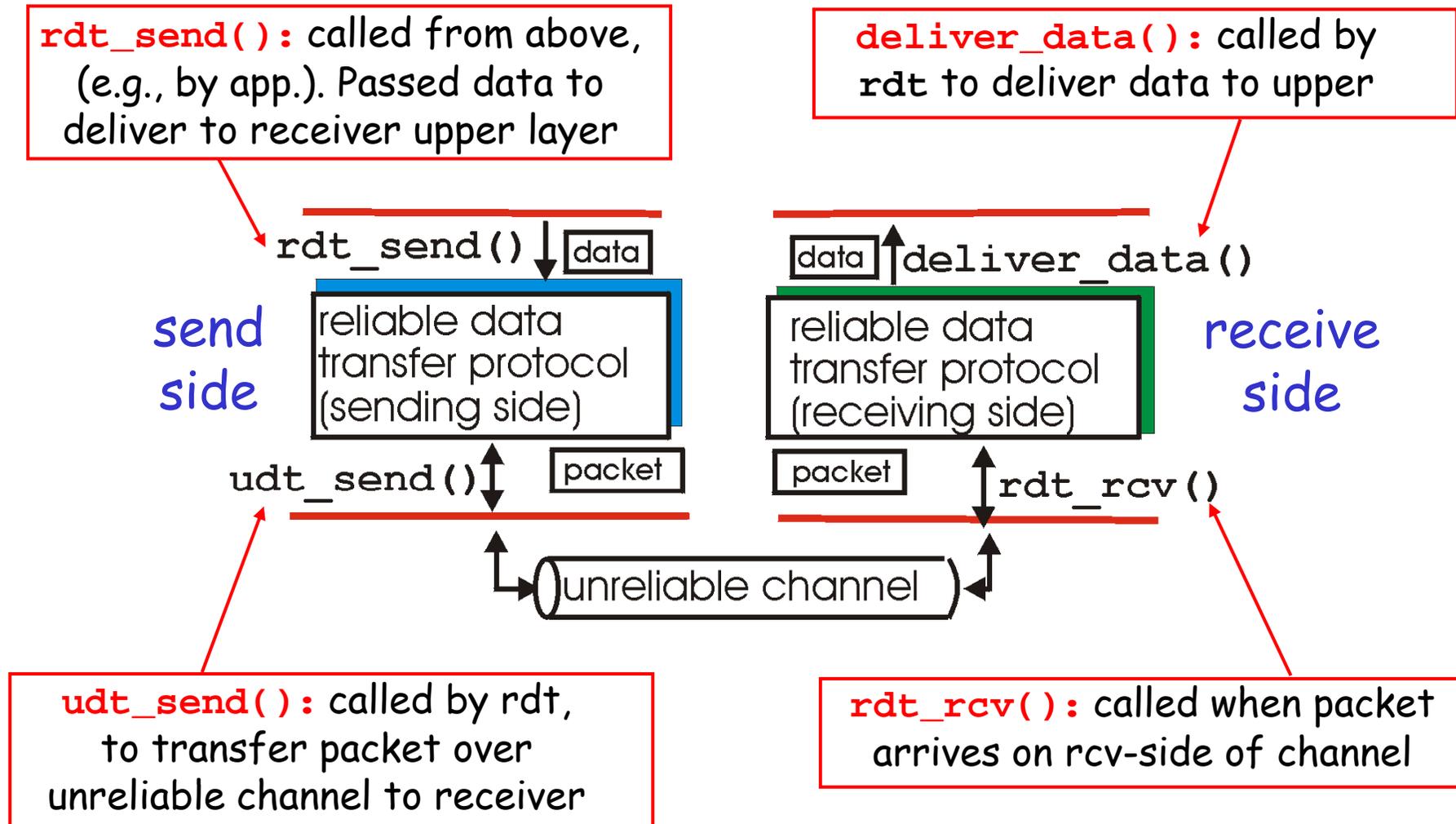
Principles of Reliable data transfer

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



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

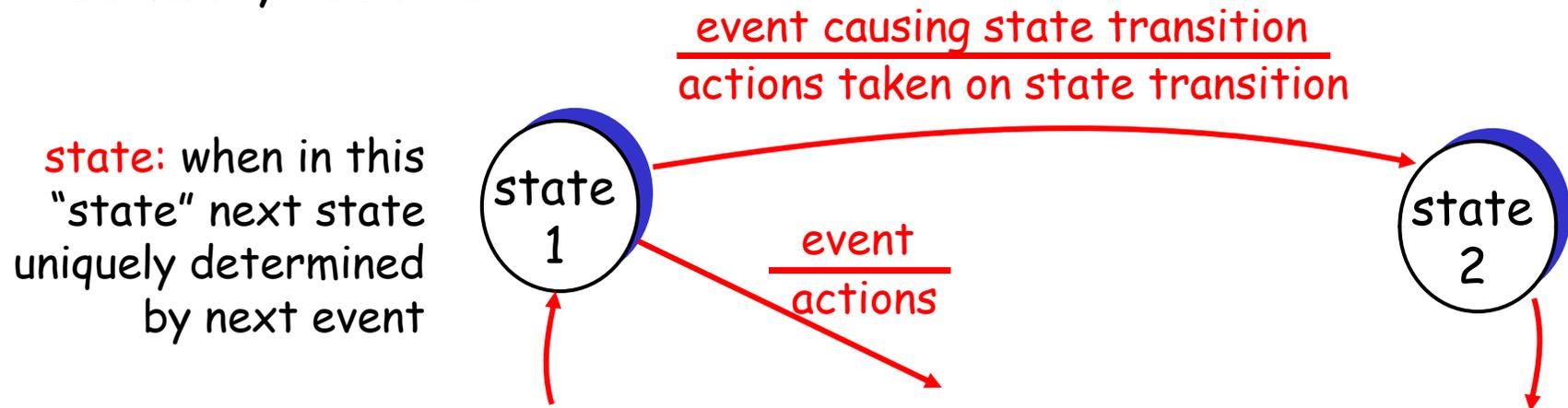
Reliable data transfer: getting started



Reliable data transfer: getting started

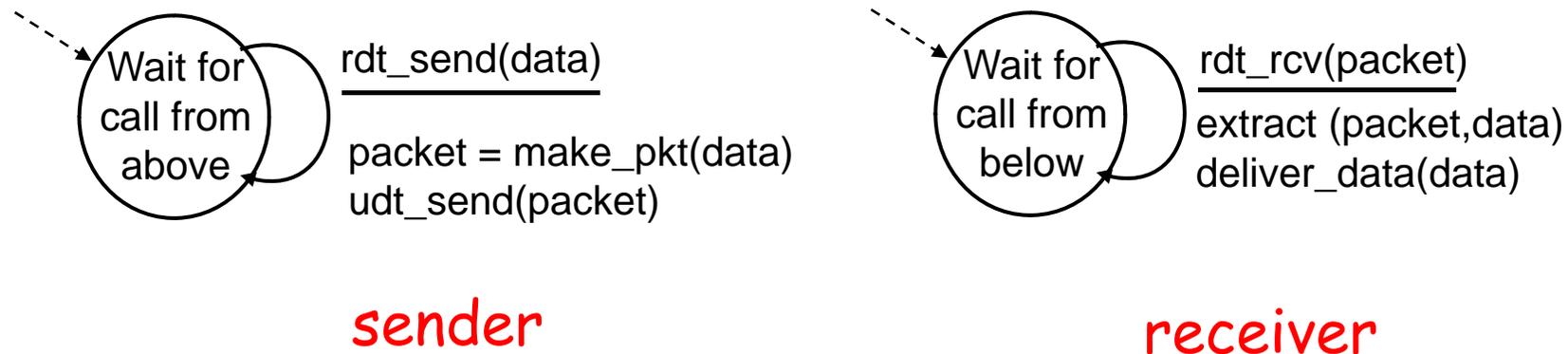
We'll:

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



Rdt1.0: reliable transfer over a reliable channel

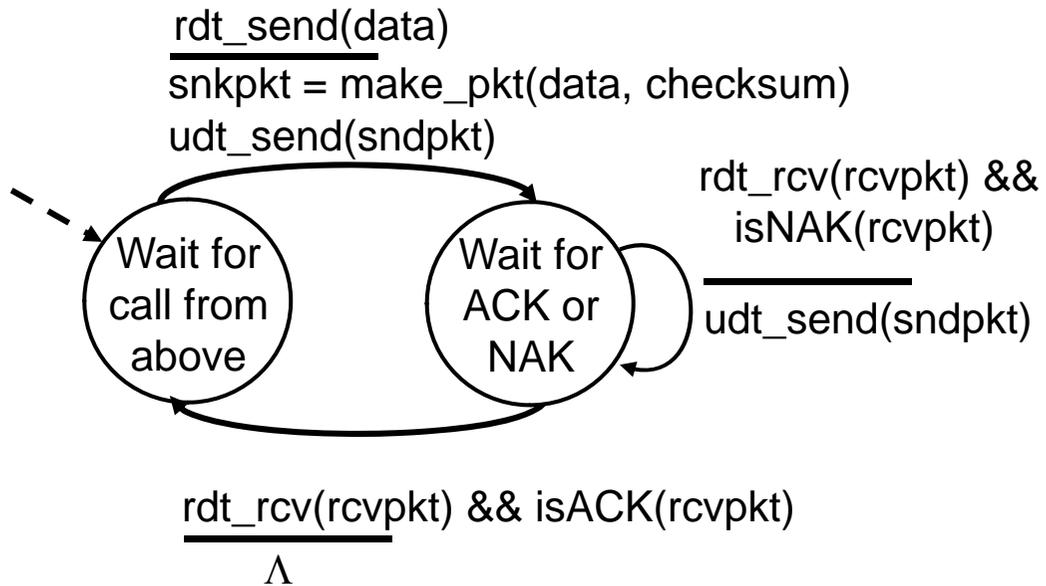
- underlying channel perfectly reliable
 - no bit errors
 - no loss of packets
- separate FSMs for sender, receiver:
 - sender sends data into underlying channel
 - receiver read data from underlying channel



Rdt2.0: channel with bit errors

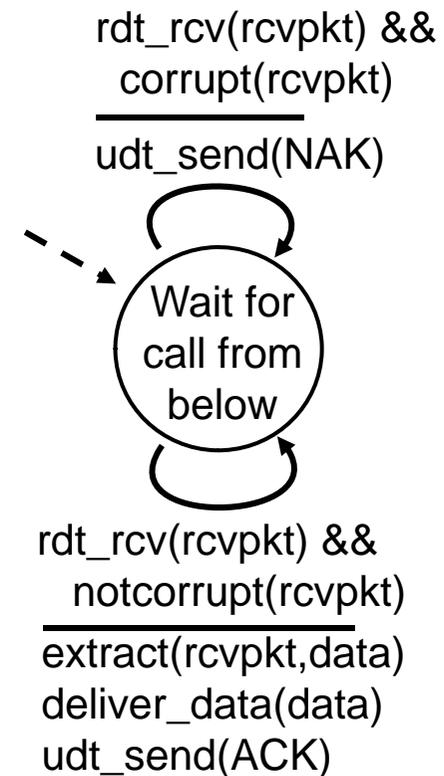
- underlying channel may flip bits in packet
 - checksum to detect bit errors
- *the question*: how to recover from errors:
 - *acknowledgements (ACKs)*: receiver explicitly tells sender that pkt received OK
 - *negative acknowledgements (NAKs)*: receiver explicitly tells sender that pkt had errors
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

rdt2.0: FSM specification

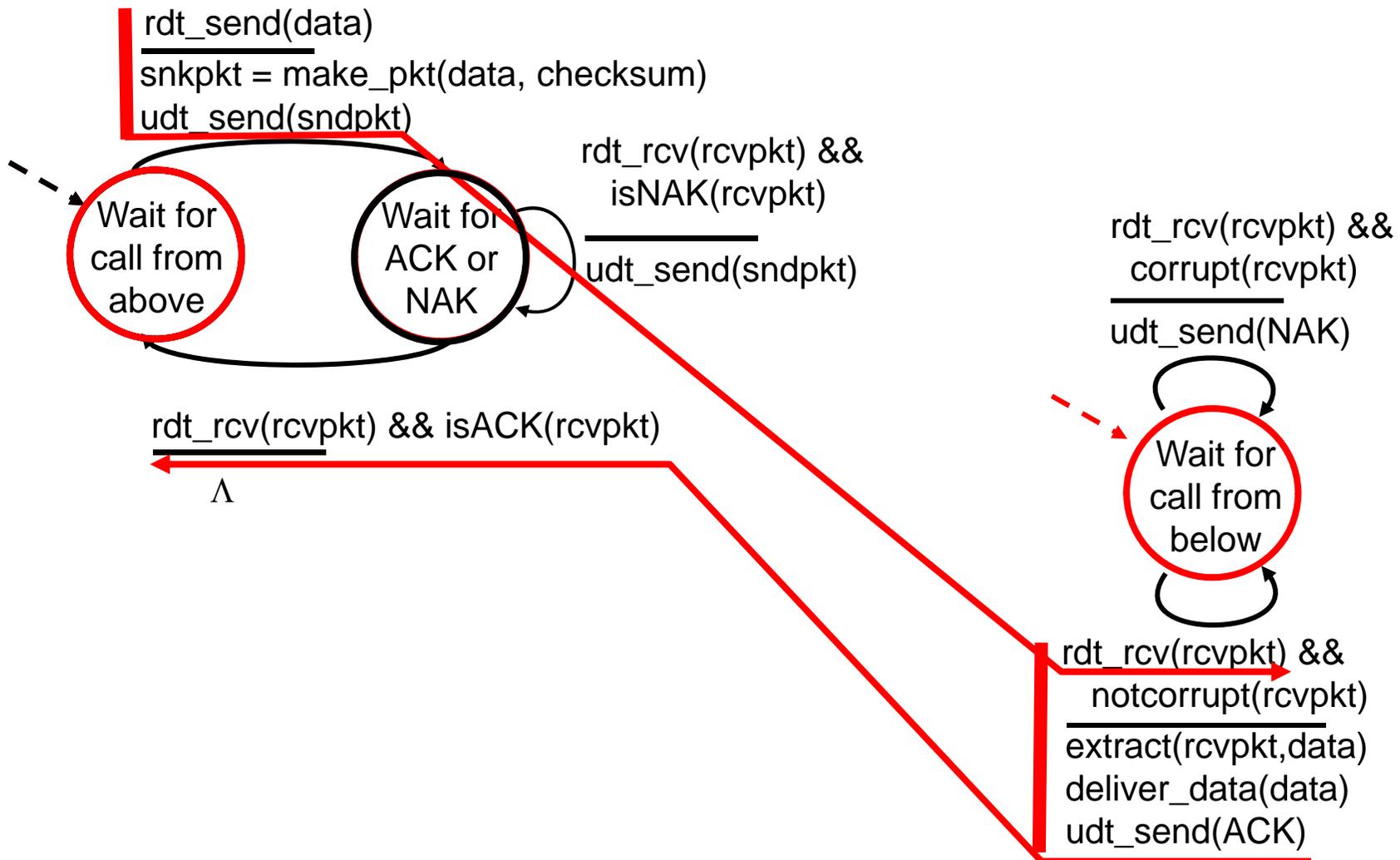


sender

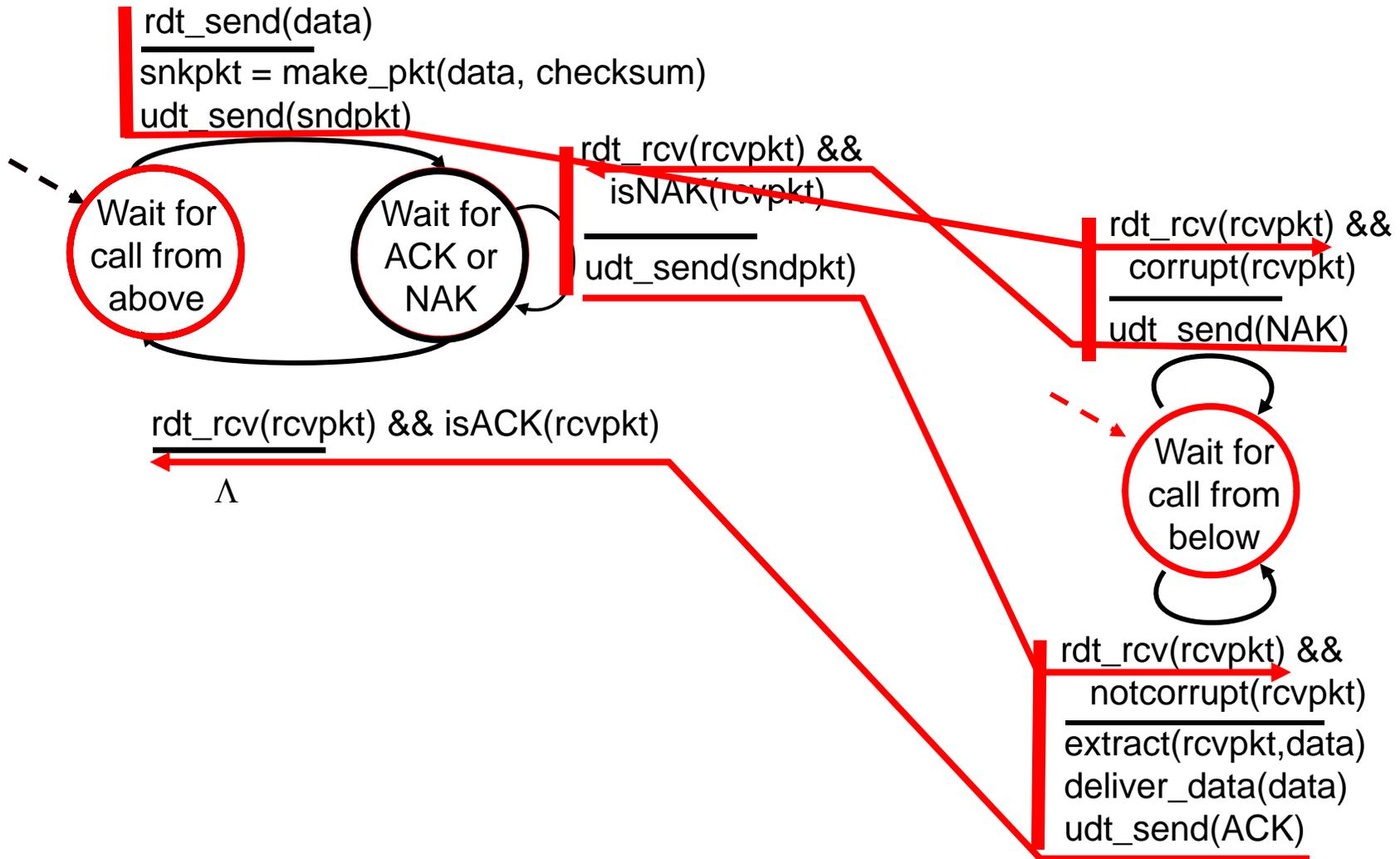
receiver



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

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

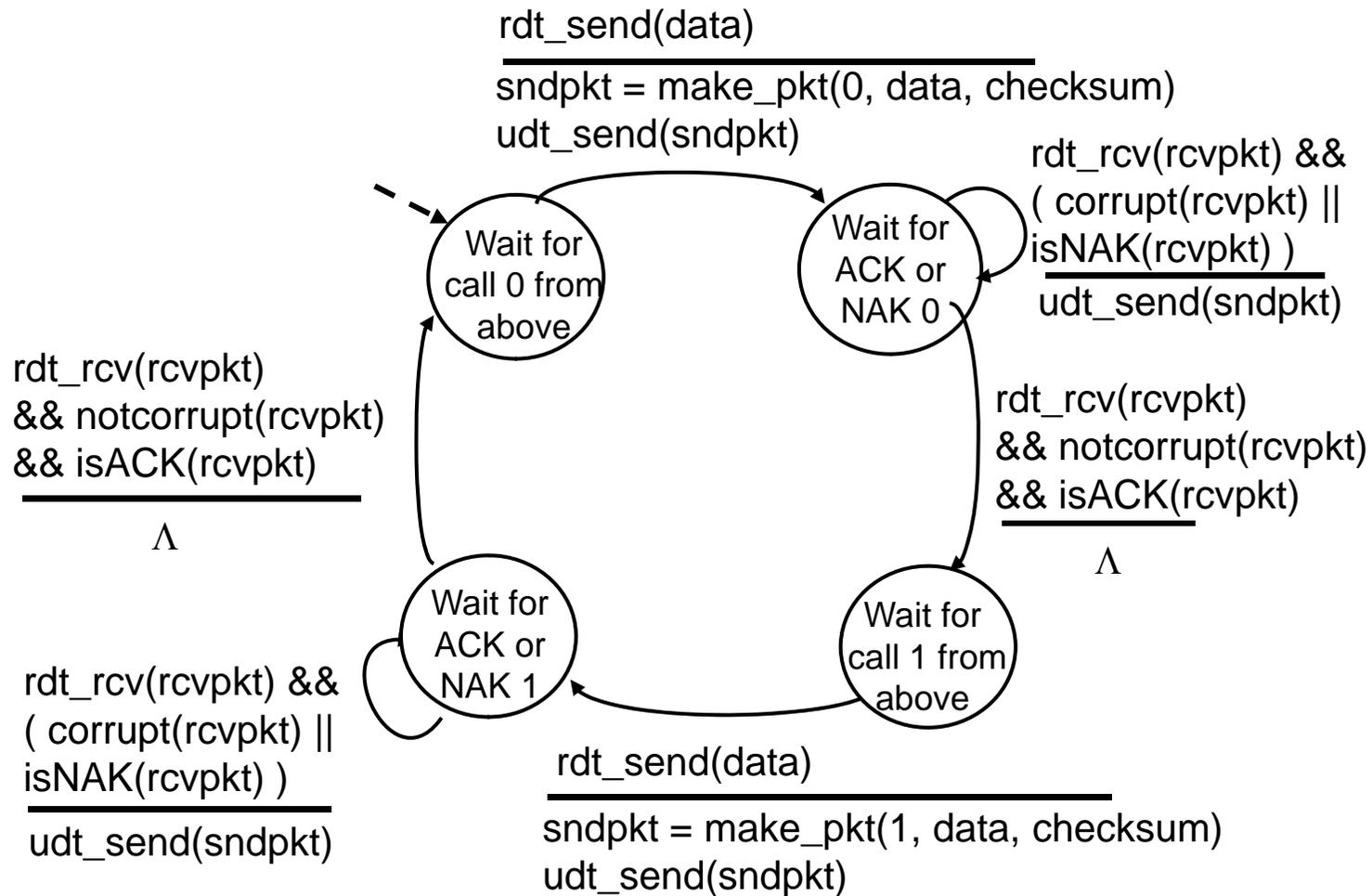
Handling duplicates:

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

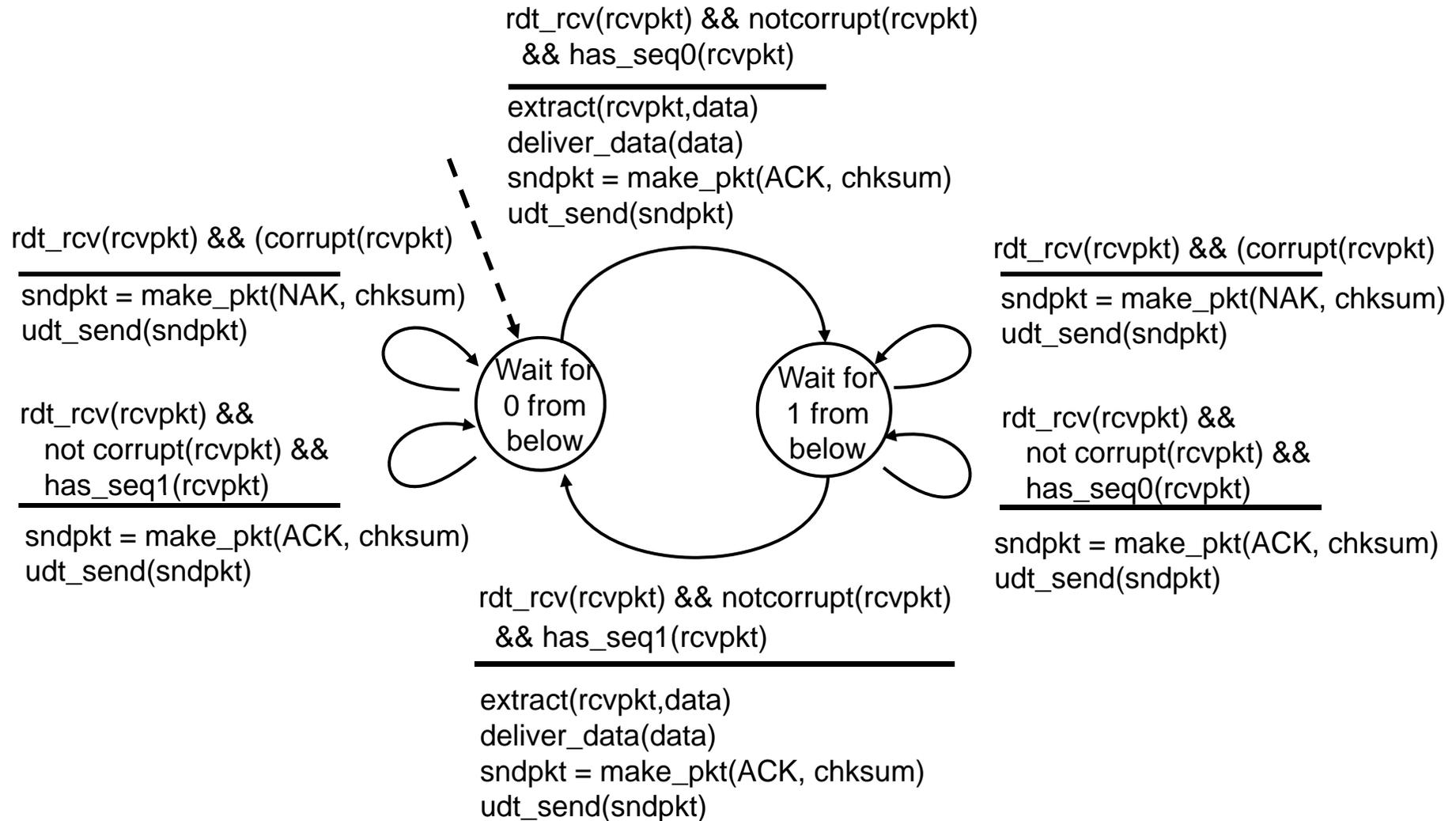
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

Sender:

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

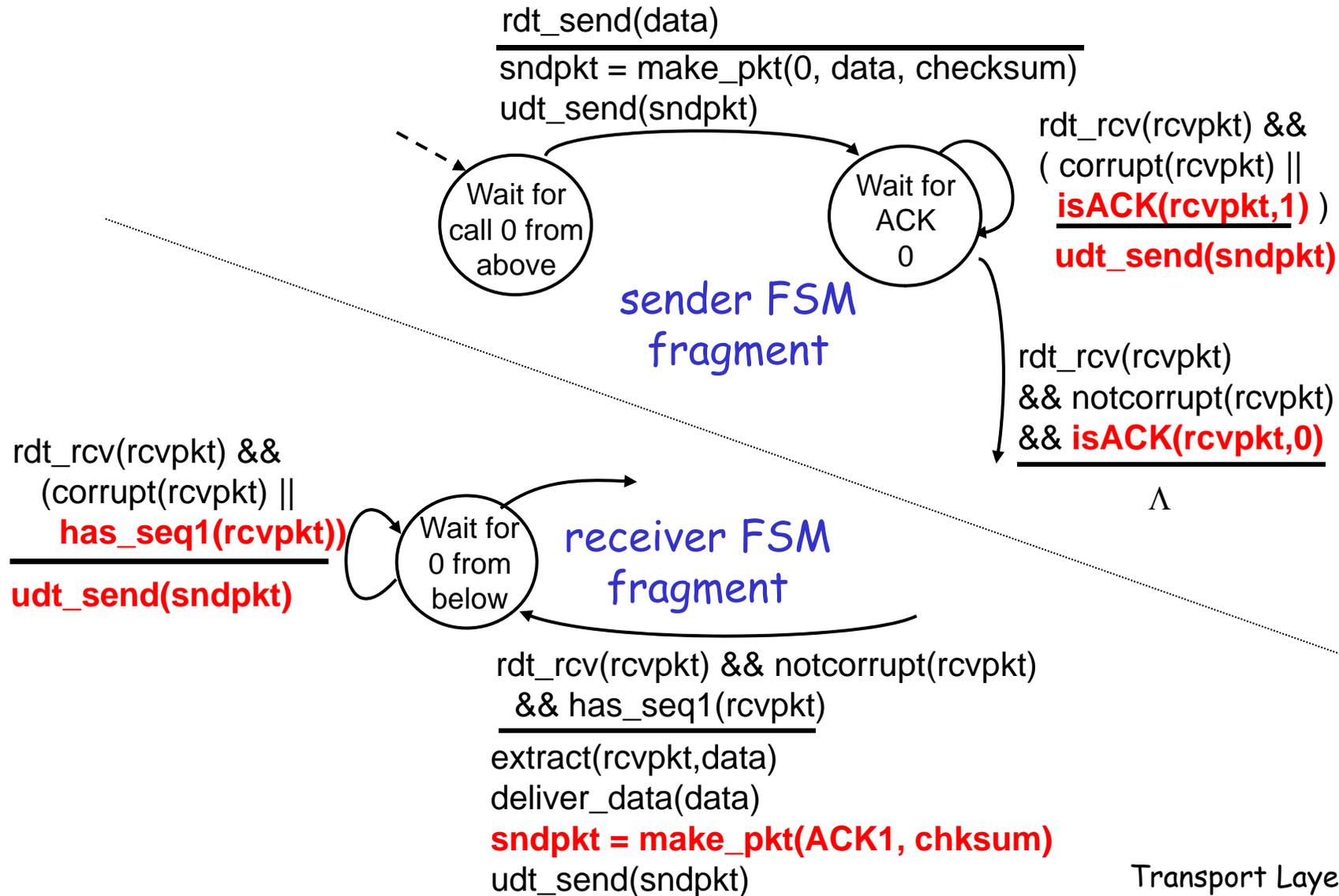
Receiver:

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

rdt2.2: a NAK-free protocol

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

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors and loss

New assumption:

underlying channel can also lose packets (data or ACKs)

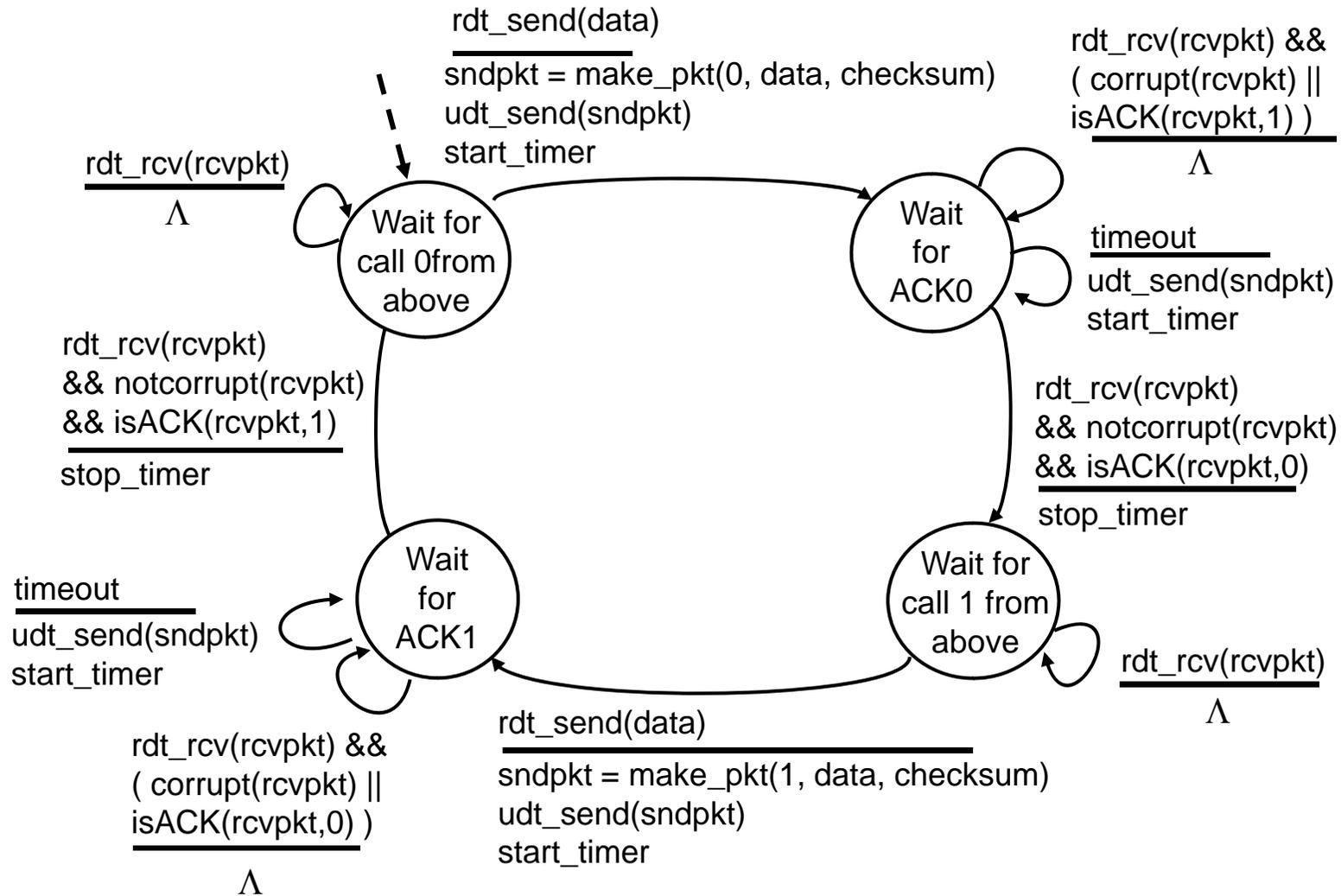
- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

Approach: sender waits

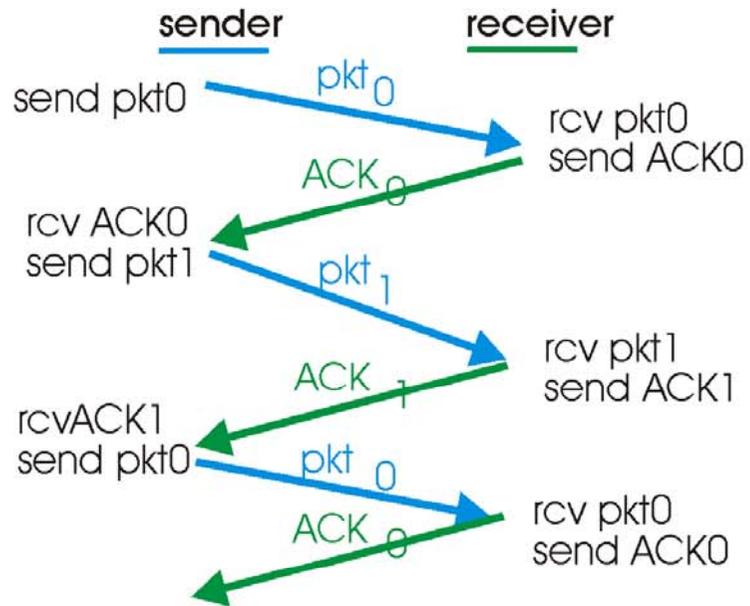
“reasonable” amount of time for ACK

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

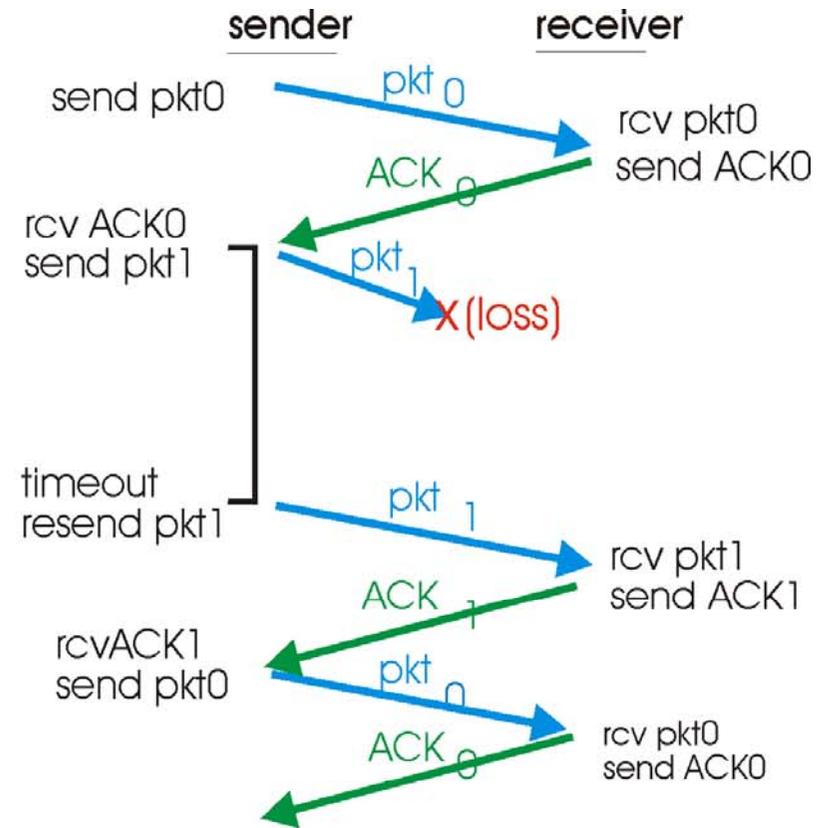
rdt3.0 sender



rdt3.0 in action

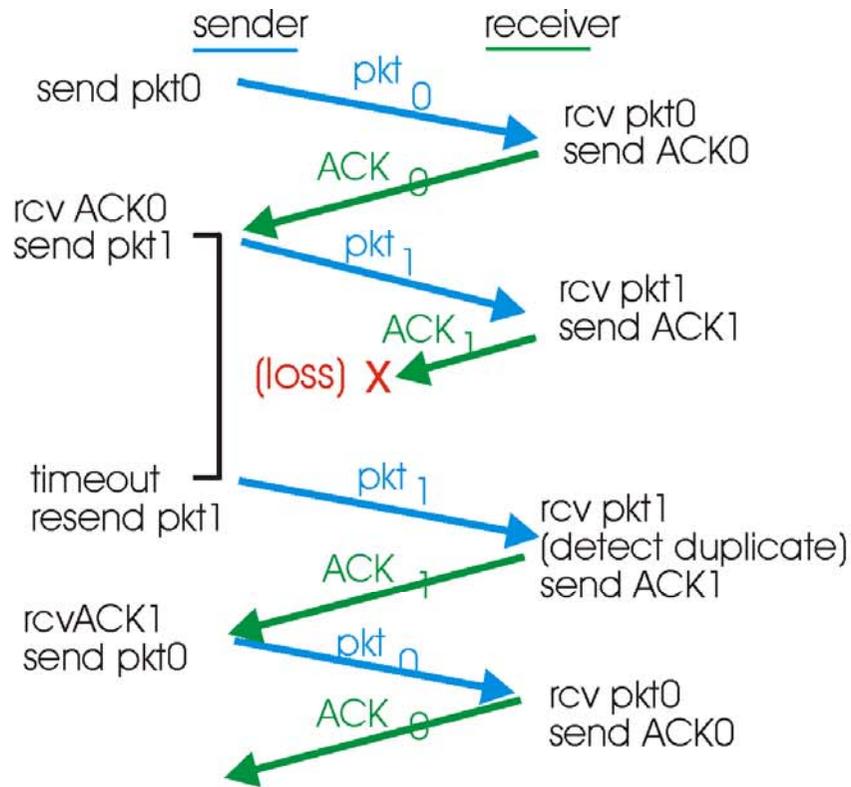


(a) operation with no loss

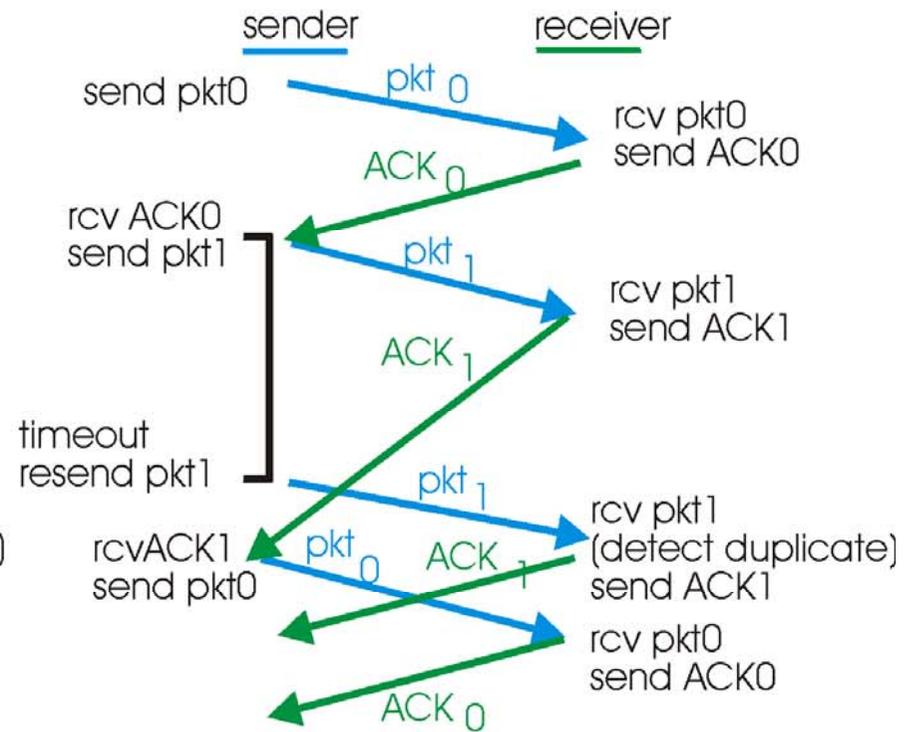


(b) lost packet

rdt3.0 in action



(c) lost ACK



(d) premature timeout

Performance of rdt3.0

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

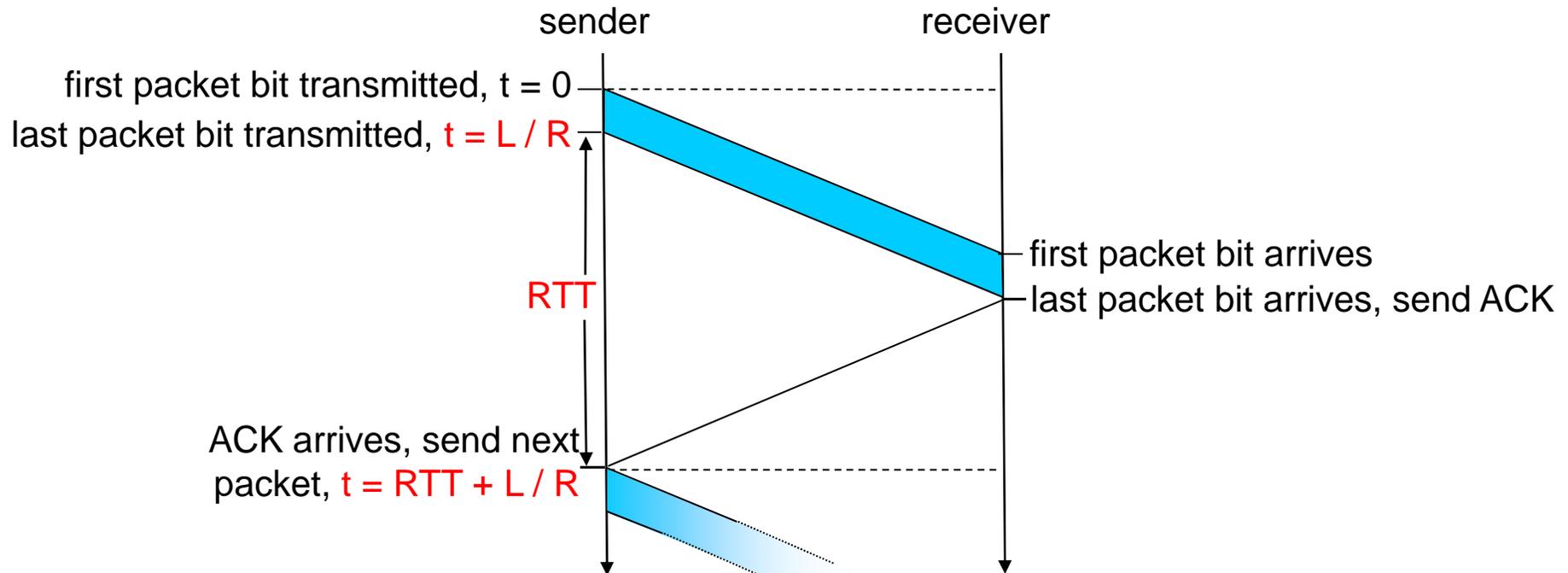
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\text{microseconds}$$

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

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

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

rdt3.0: stop-and-wait operation

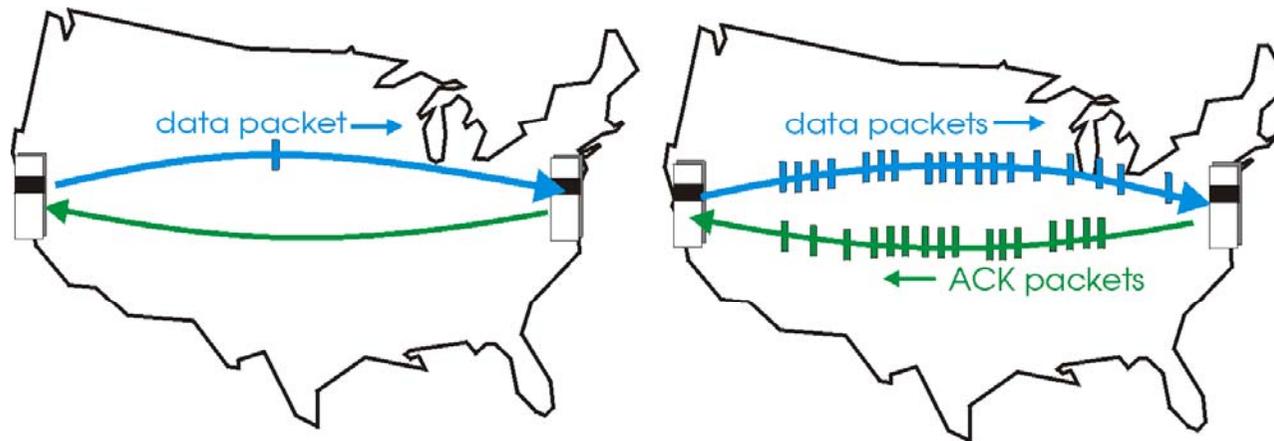


$$U_{\text{sender}} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

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

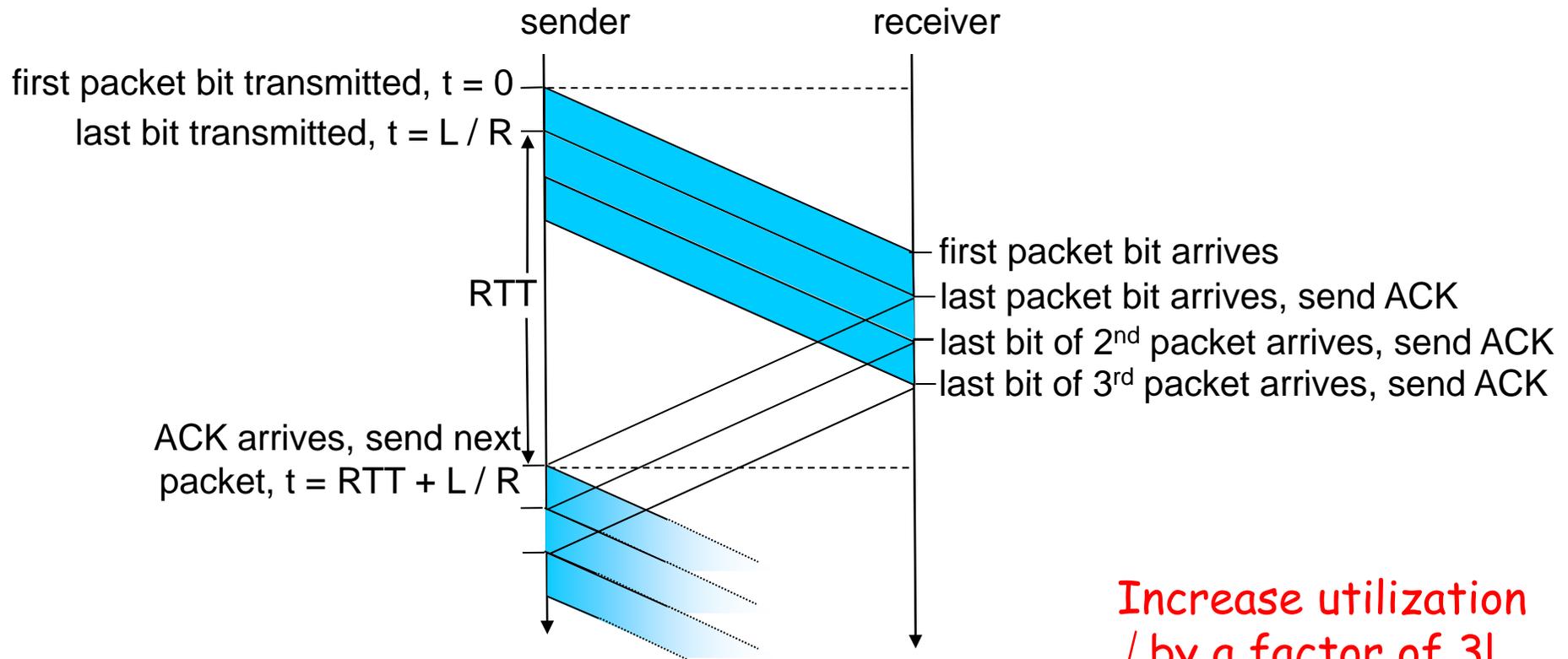


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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

Pipelining: increased utilization



Increase utilization
by a factor of 3!

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

Pipelining Protocols

Go-back-N: big picture:

- ❑ Sender can have up to N unacked packets in pipeline
- ❑ Rcvr only sends cumulative acks
 - Doesn't ack packet if there's a gap
- ❑ Sender has timer for oldest unacked packet
 - If timer expires, retransmit all unacked packets

Selective Repeat: big pic

- ❑ Sender can have up to N unacked packets in pipeline
- ❑ Rcvr acks individual packets
- ❑ Sender maintains timer for each unacked packet
 - When timer expires, retransmit only unack packet

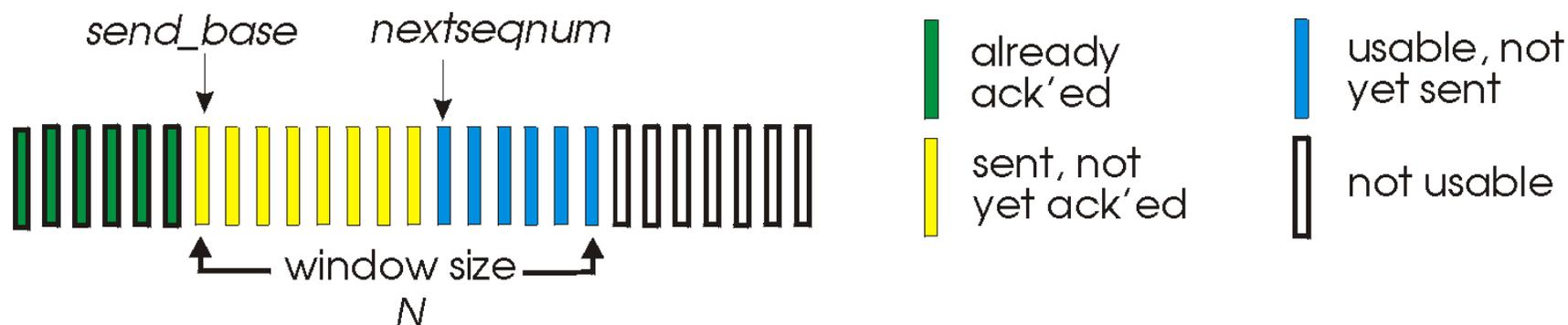
Selective repeat: big picture

- ❑ Sender can have up to N unacked packets in pipeline
- ❑ Rcvr acks individual packets
- ❑ Sender maintains timer for each unacked packet
 - When timer expires, retransmit only unack packet

Go-Back-N

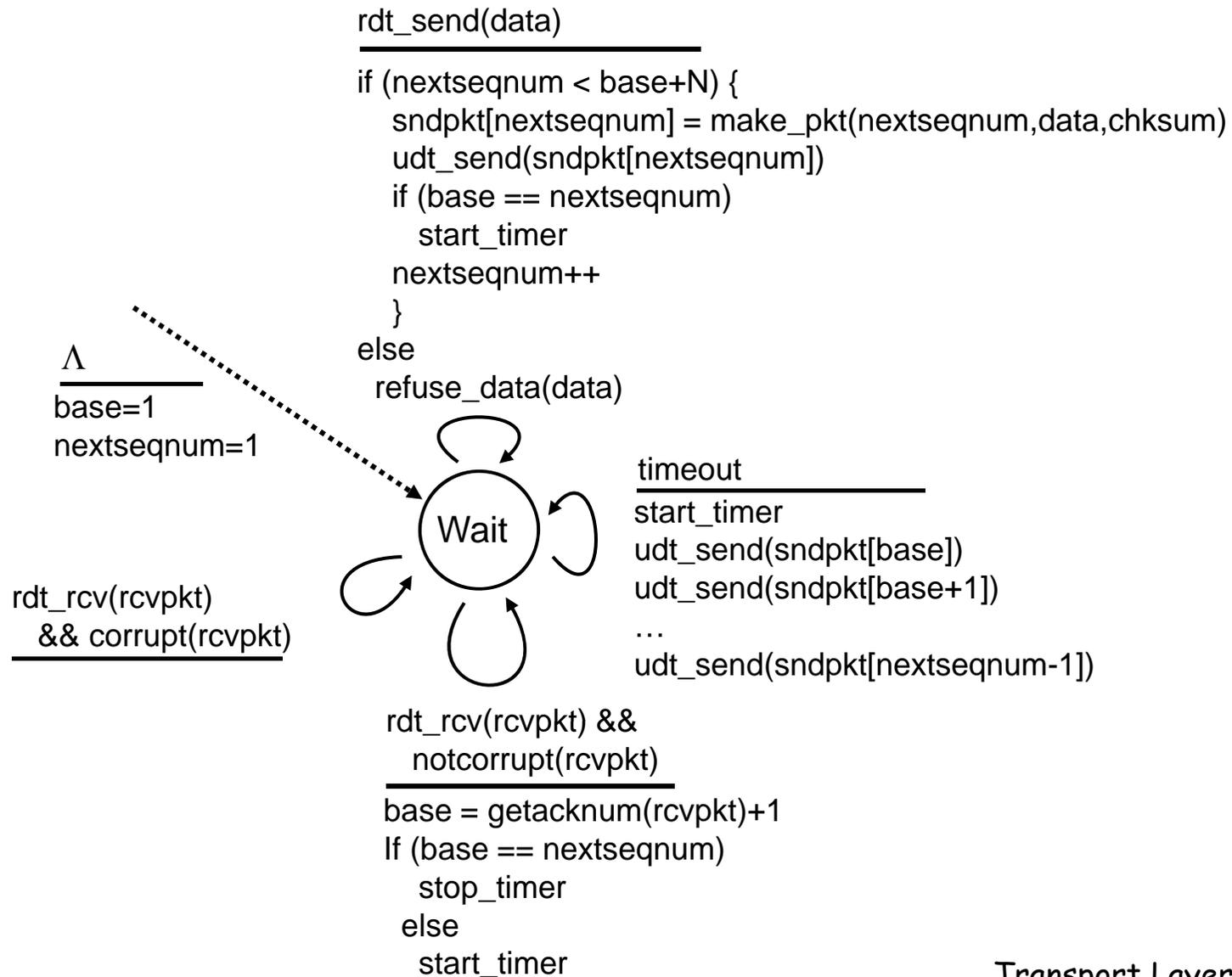
Sender:

- ❑ k-bit seq # in pkt header
- ❑ "window" of up to N, consecutive unack'ed pkts allowed

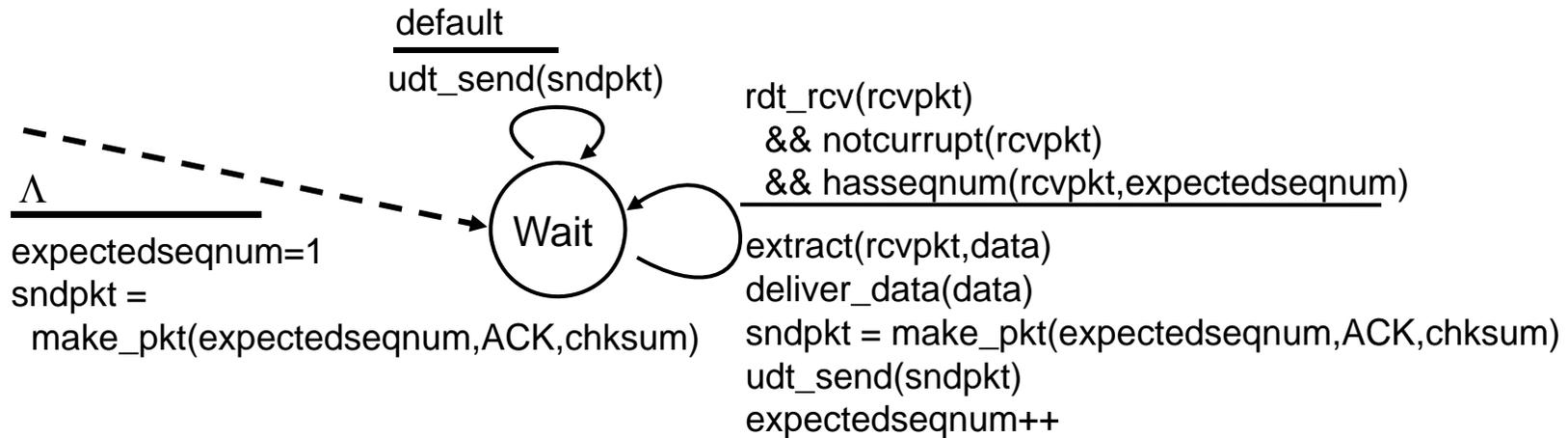


- ❑ $ACK(n)$: ACKs all pkts up to, including seq # n - "cumulative ACK"
 - may receive duplicate ACKs (see receiver)
- ❑ timer for each in-flight pkt
- ❑ $timeout(n)$: retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM



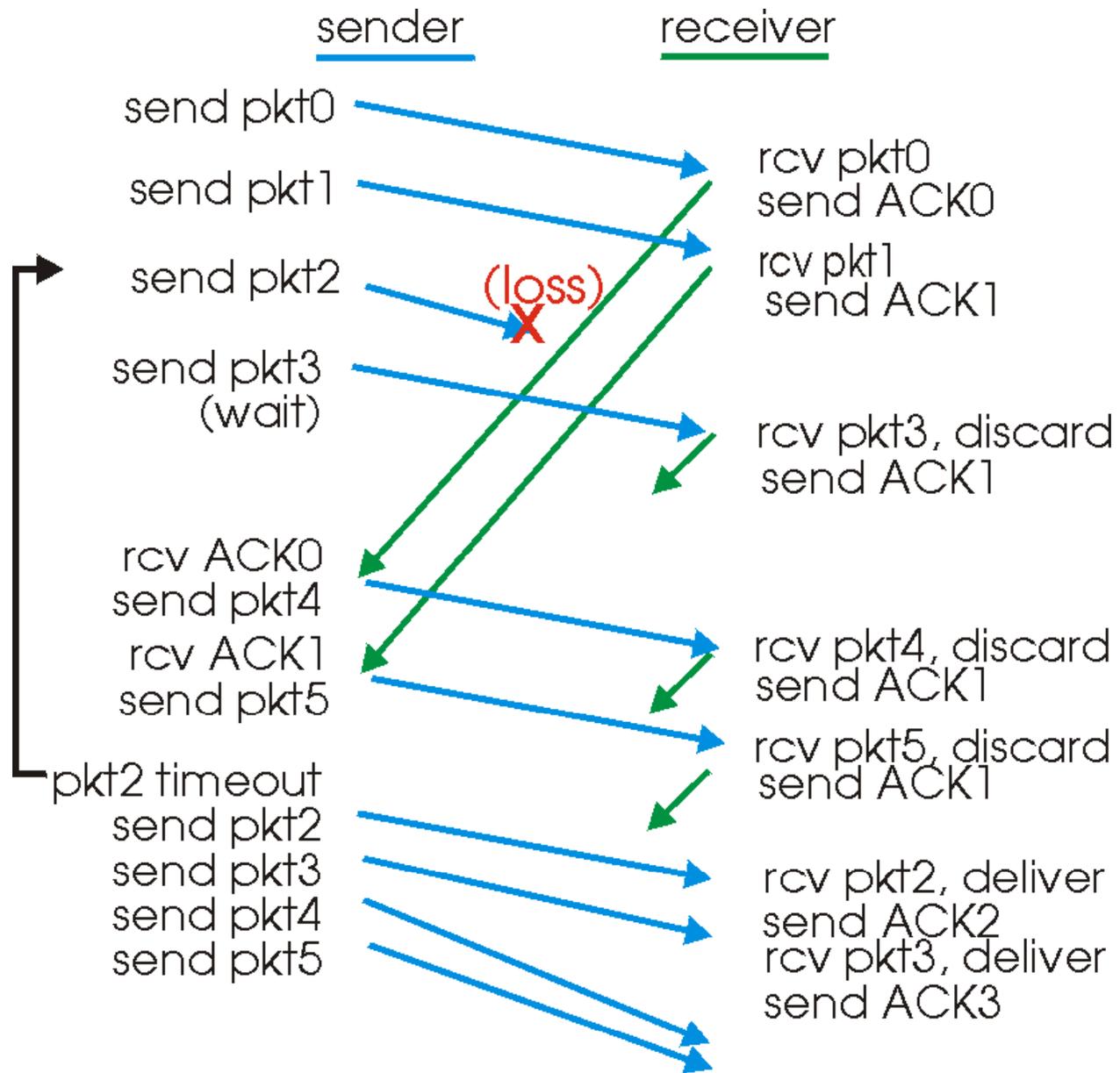
ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

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

□ out-of-order pkt:

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

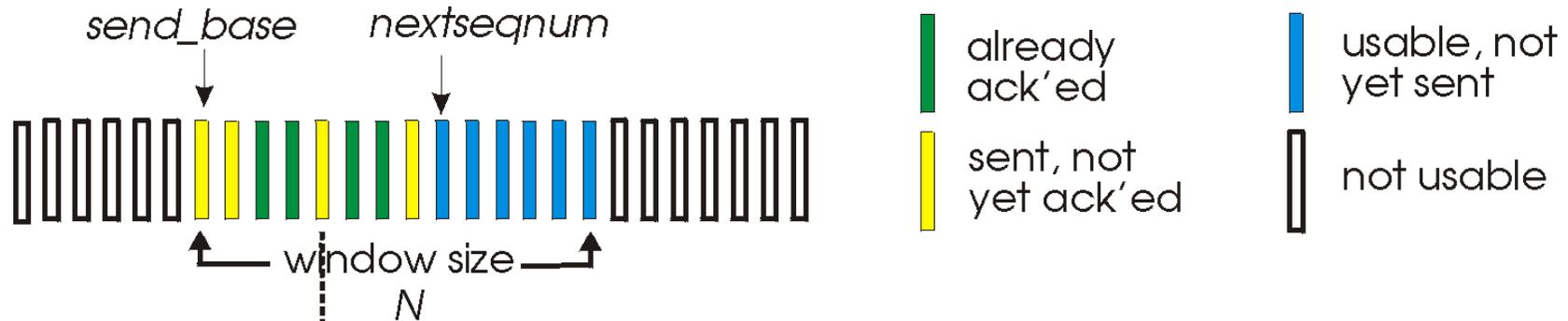
GBN in action



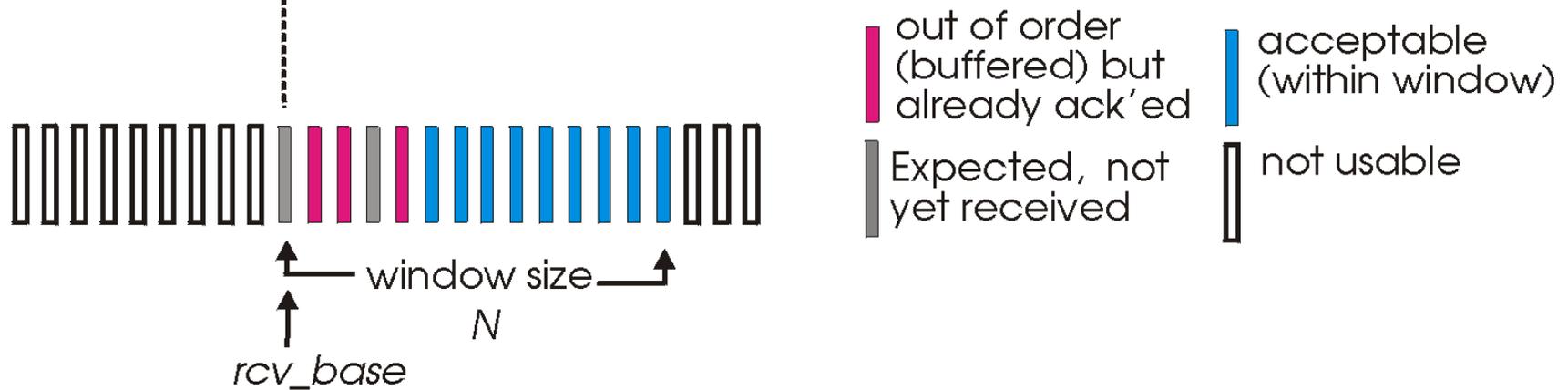
Selective Repeat

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

Selective repeat: sender, receiver windows



(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

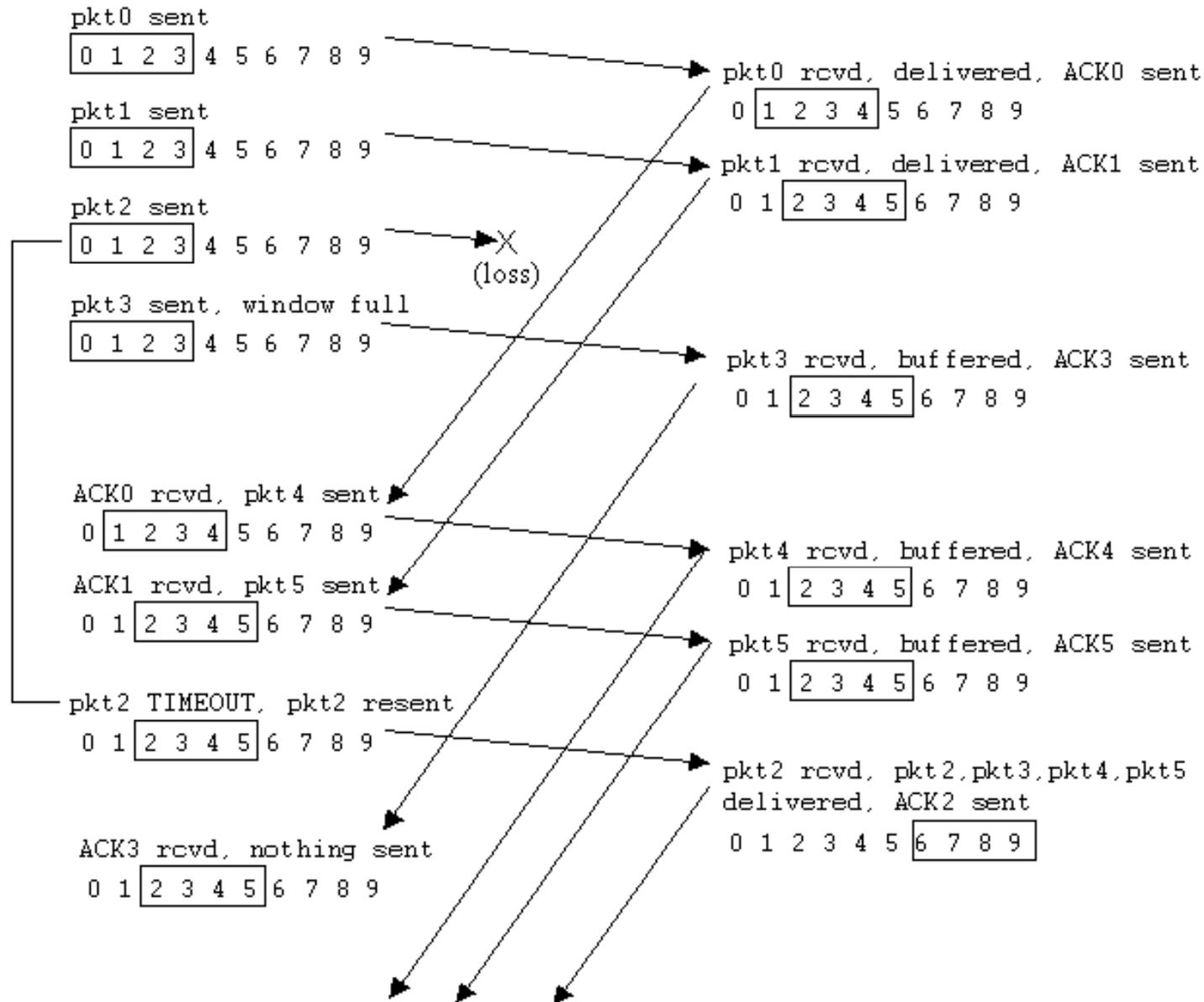
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



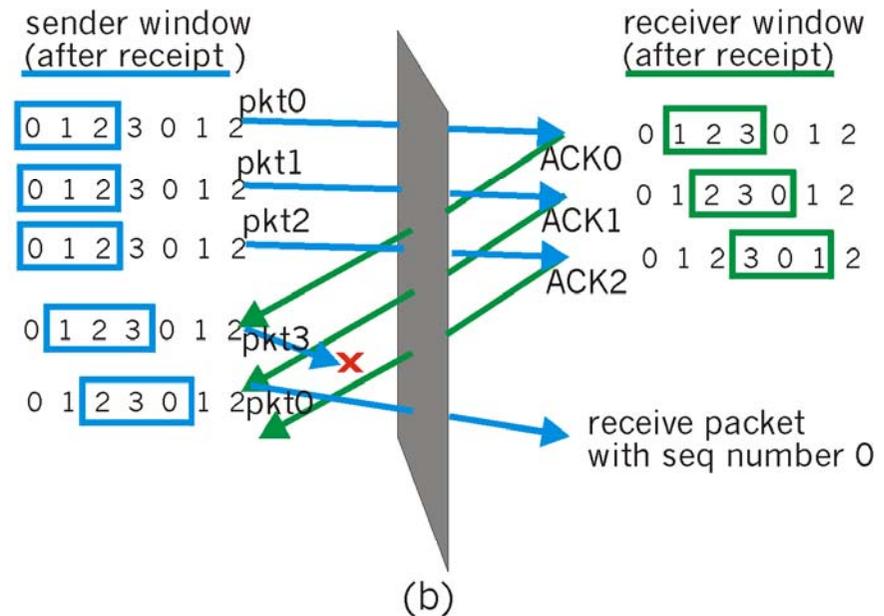
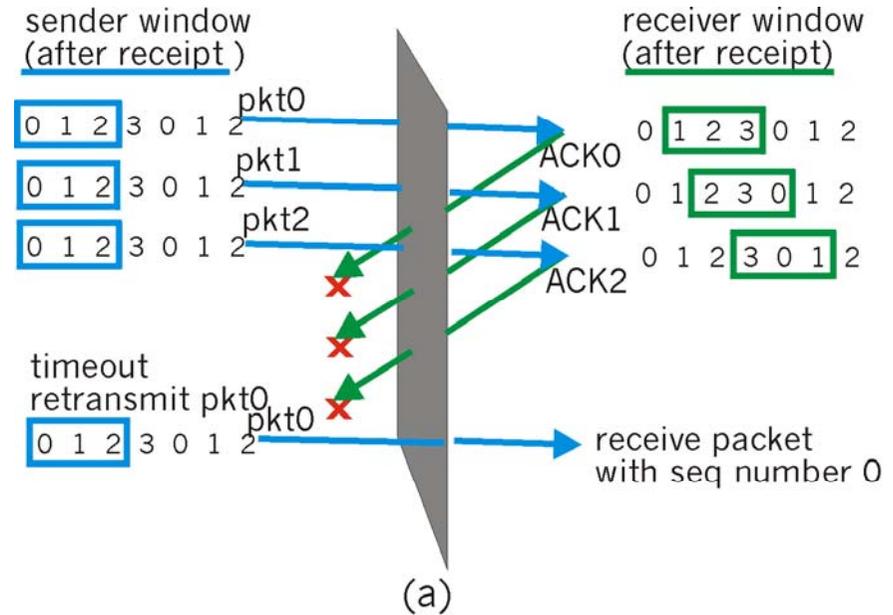
Selective repeat: dilemma

Example:

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

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

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



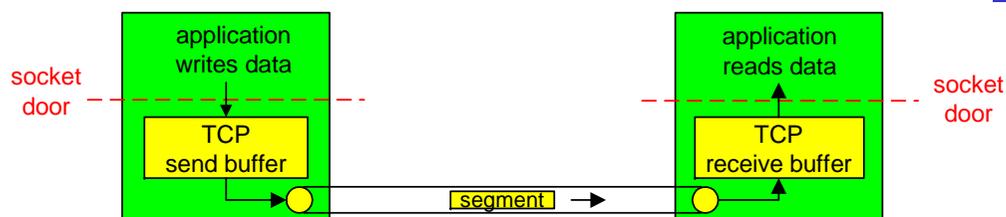
Chapter 3 outline

- ❑ 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

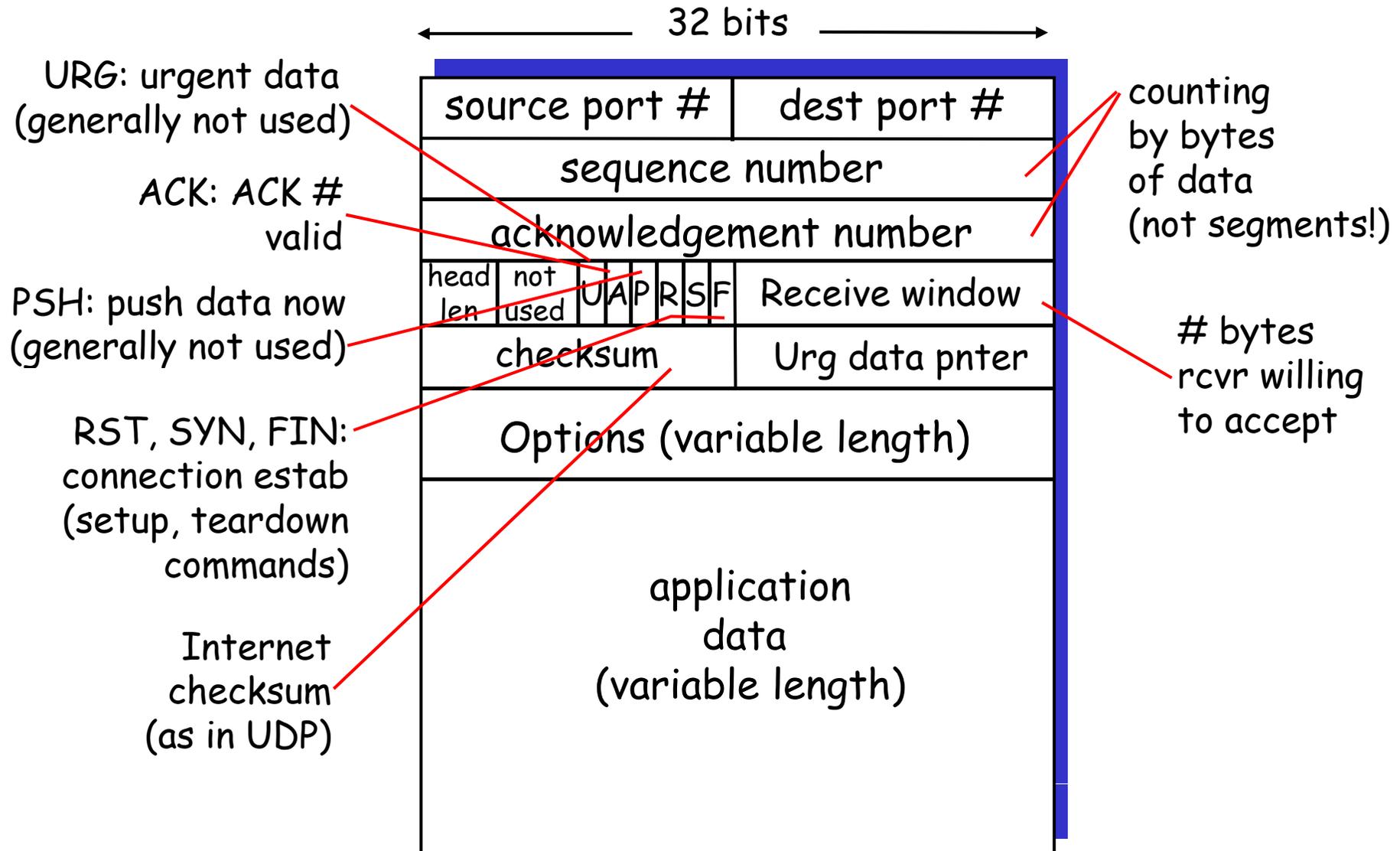
TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- ❑ **point-to-point:**
 - one sender, one receiver
 - ❑ **reliable, in-order *byte stream*:**
 - no "message boundaries"
 - ❑ **pipelined:**
 - TCP congestion and flow control set window size
 - ❑ ***send & receive buffers***
- ❑ **full duplex data:**
 - bi-directional data flow in same connection
 - MSS: maximum segment size
 - ❑ **connection-oriented:**
 - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
 - ❑ **flow controlled:**
 - sender will not overwhelm receiver



TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

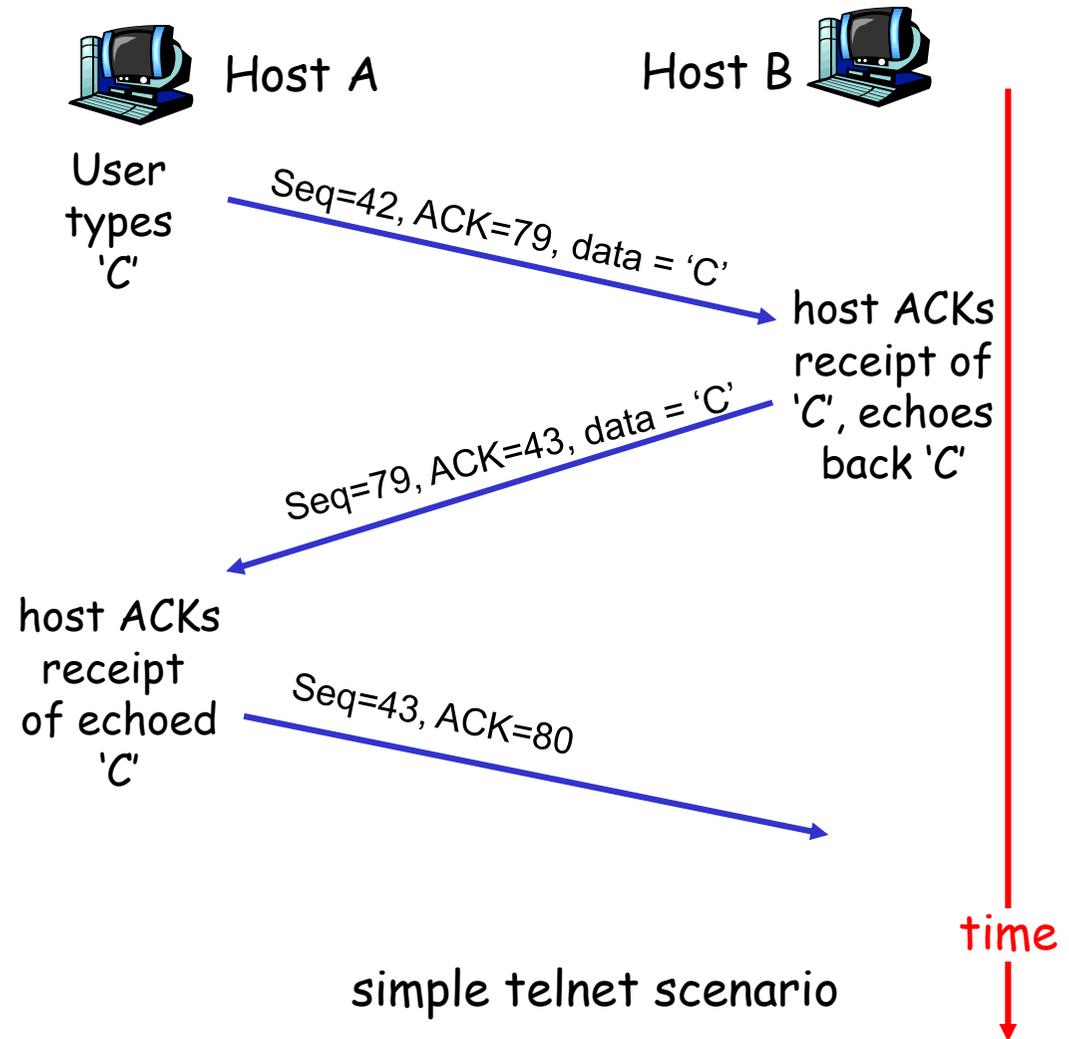
- byte stream
"number" of first byte in segment's data

ACKs:

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

Q: how receiver handles out-of-order segments

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



TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

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

Q: how to estimate RTT?

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

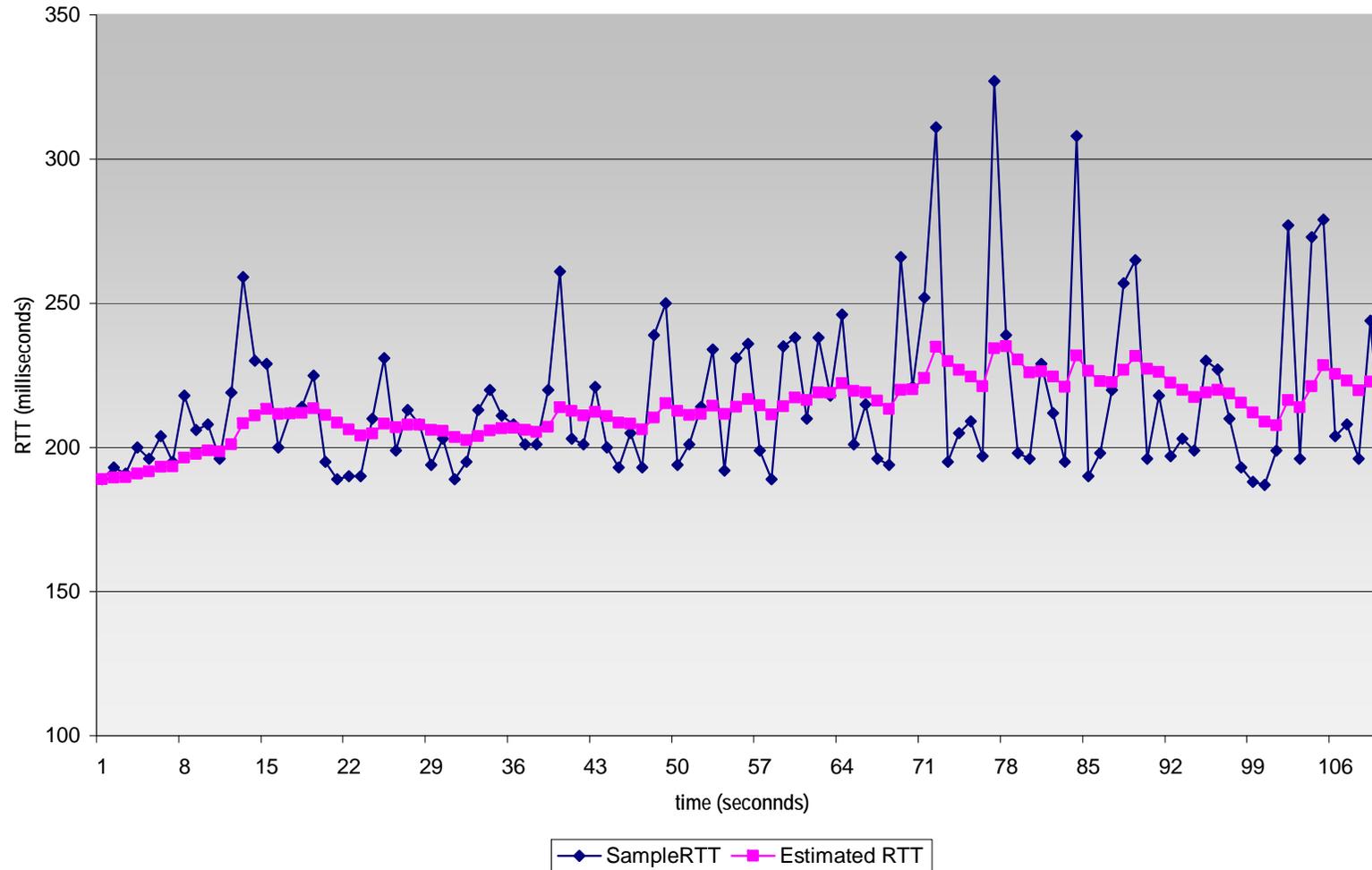
TCP Round Trip Time and Timeout

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

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

Example RTT estimation:

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr



TCP Round Trip Time and Timeout

Setting the timeout

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

$$\text{DevRTT} = (1-\beta) * \text{DevRTT} + \beta * |\text{SampleRTT} - \text{EstimatedRTT}|$$

(typically, $\beta = 0.25$)

Then set timeout interval:

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$

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 reliable data transfer

- ❑ TCP creates rdt service on top of IP's unreliable service
- ❑ Pipelined segments
- ❑ Cumulative acks
- ❑ TCP uses single retransmission timer
- ❑ Retransmissions are triggered by:
 - timeout events
 - duplicate acks
- ❑ Initially consider simplified TCP sender:
 - ignore duplicate acks
 - ignore flow control, congestion control

TCP sender events:

data rcvd from app:

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

timeout:

- ❑ retransmit segment that caused timeout
- ❑ restart timer

Ack rcvd:

- ❑ If acknowledges previously unacked segments
 - update what is known to be acked
 - start timer if there are outstanding segments

```
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 */
```

TCP sender (simplified)

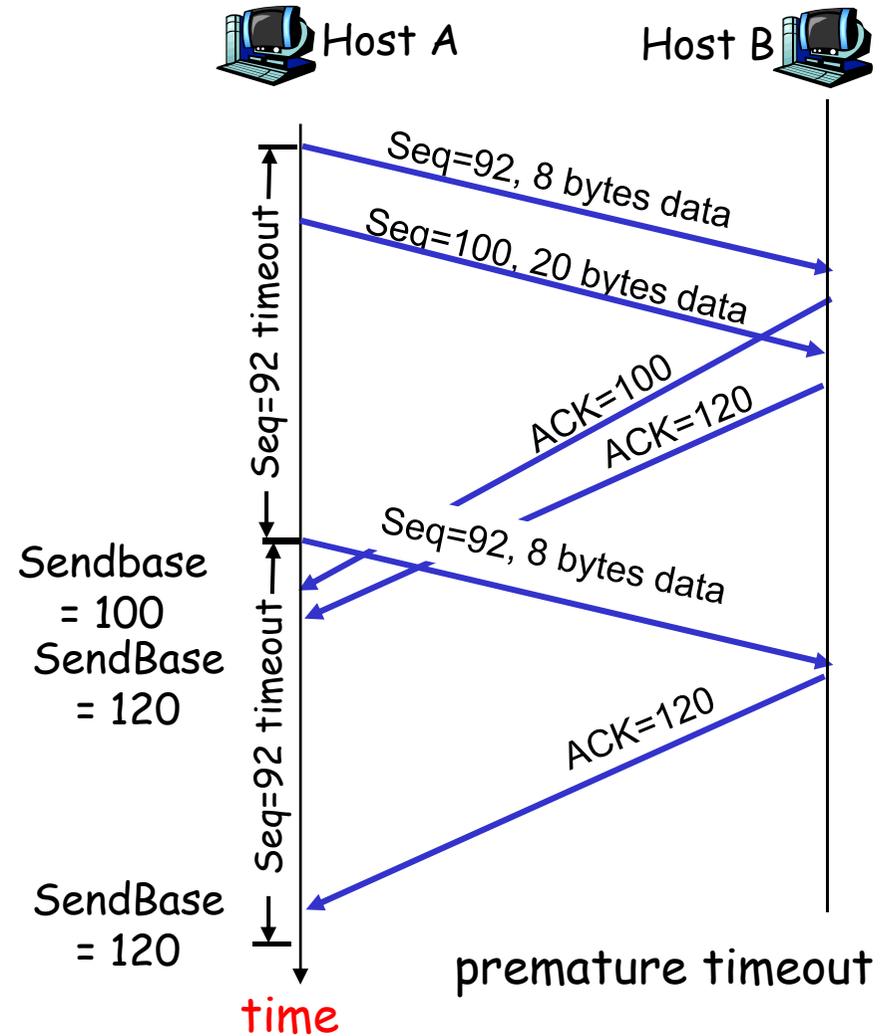
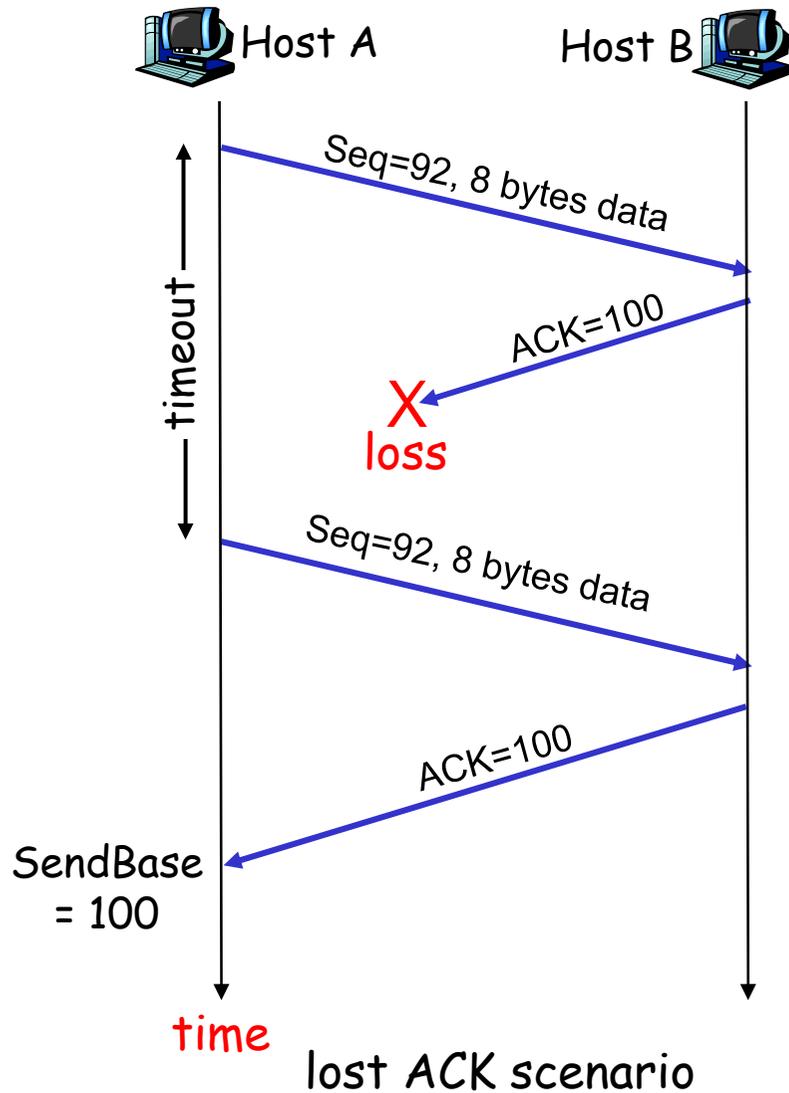
Comment:

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

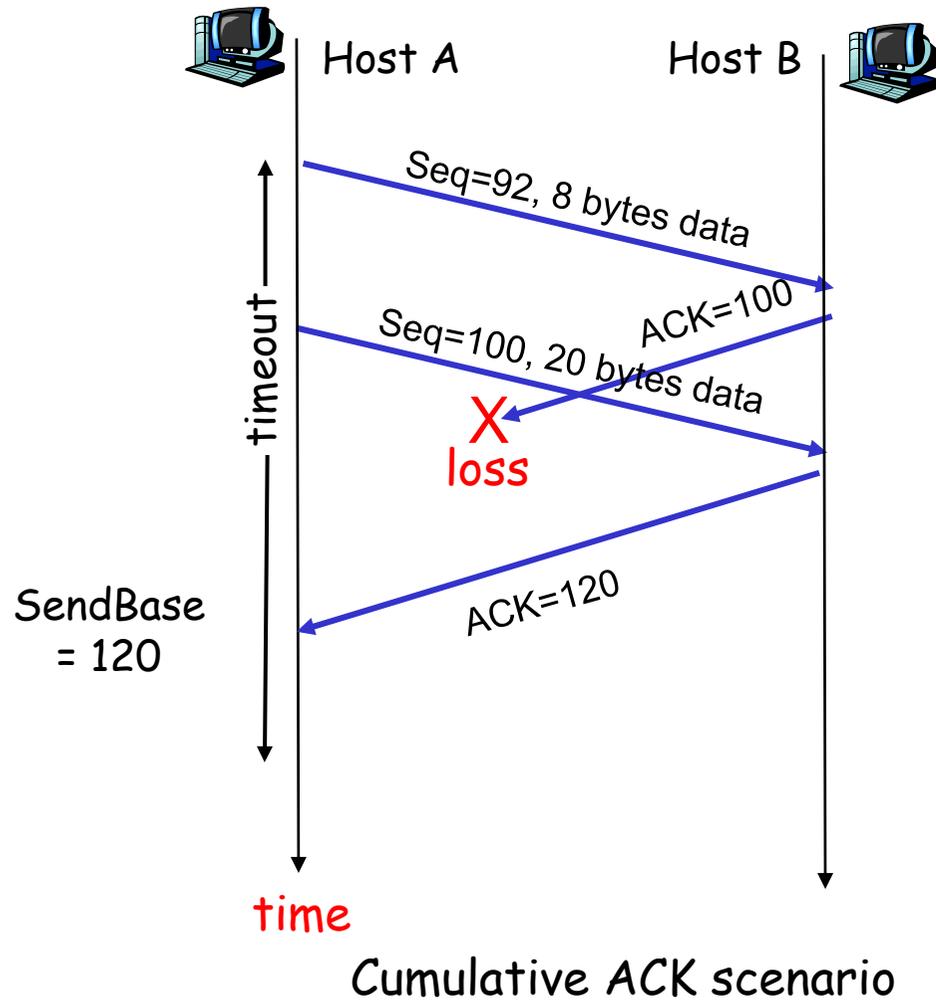
Example:

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

TCP: retransmission scenarios



TCP retransmission scenarios (more)



TCP ACK generation [RFC 1122, RFC 2581]

Event at Receiver

TCP Receiver action

Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed

Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK

Arrival of in-order segment with expected seq #. One other segment has ACK pending

Immediately send single cumulative ACK, ACKing both in-order segments

Arrival of out-of-order segment higher-than-expected seq. # . Gap detected

Immediately send *duplicate ACK*, indicating seq. # of next expected byte

Arrival of segment that partially or completely fills gap

Immediate send ACK, provided that segment starts at lower end of gap

Fast Retransmit

- Time-out period often relatively long:
 - long delay before resending lost packet
- Detect lost segments via duplicate ACKs.
 - Sender often sends many segments back-to-back
 - If segment is lost, there will likely be many duplicate ACKs.
- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
 - fast retransmit: resend segment before timer expires

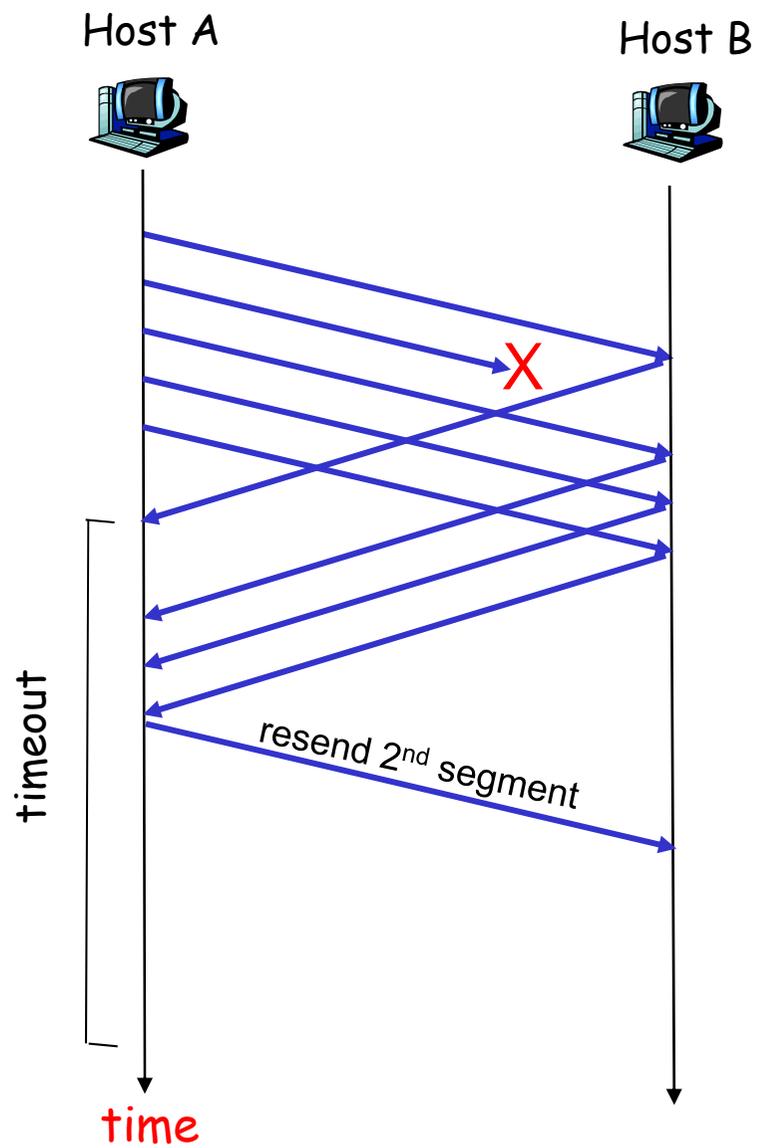


Figure 3.37 Resending a segment after triple duplicate ACK

Fast retransmit algorithm:

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

a duplicate ACK for
already ACKed segment

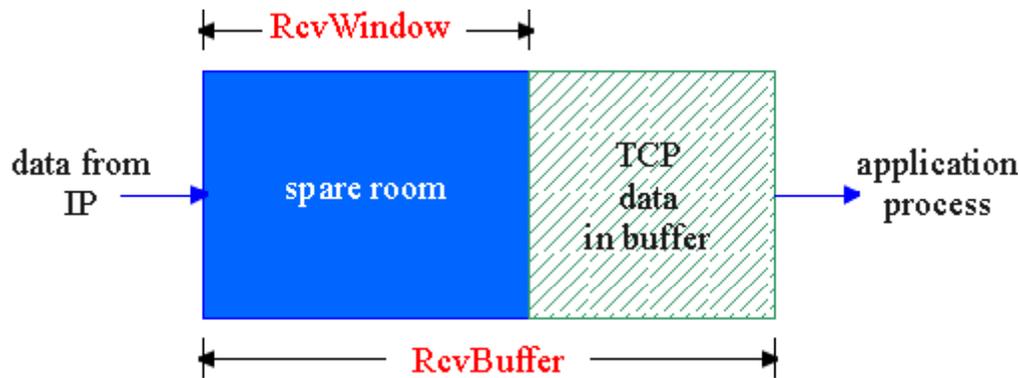
fast retransmit

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 Flow Control

- receive side of TCP connection has a receive buffer:



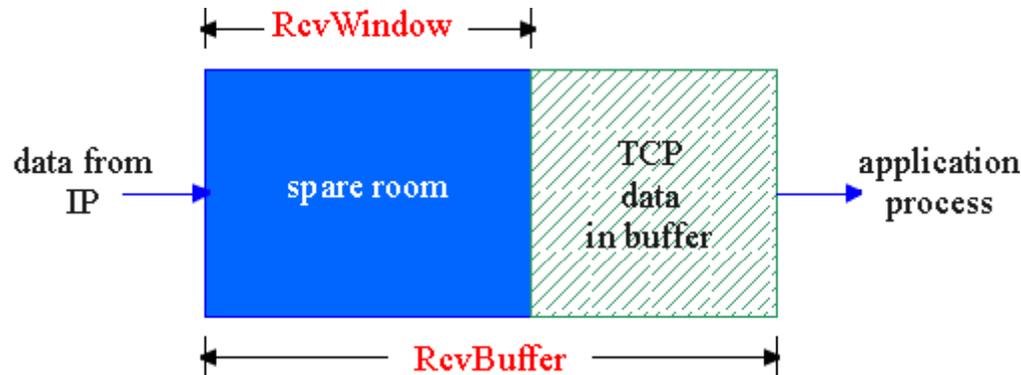
- app process may be slow at reading from buffer

flow control

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

- speed-matching service: matching the send rate to the receiving app's drain rate

TCP Flow control: how it works



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

□ spare room in buffer

= RcvWindow

= RcvBuffer - [LastByteRcvd - LastByteRead]

- Rcvr advertises spare room by including value of RcvWindow in segments
- Sender limits unACKed data to RcvWindow
 - guarantees receive buffer doesn't overflow

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 Connection Management

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

□ initialize TCP variables:

- seq. #s
- buffers, flow control info (e.g. RcvWindow)

□ *client*: connection initiator

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

□ *server*: contacted by client

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

Three way handshake:

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

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

- server allocates buffers
- specifies server initial seq. #

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

TCP Connection Management (cont.)

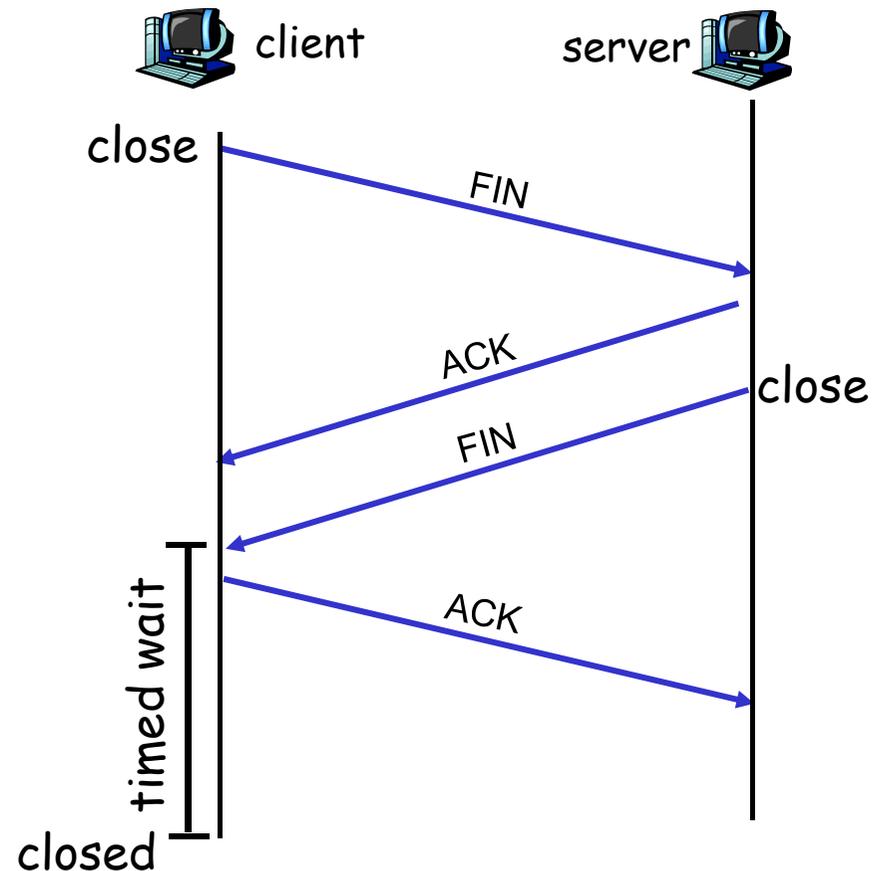
Closing a connection:

client closes socket:

```
clientSocket.close();
```

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

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



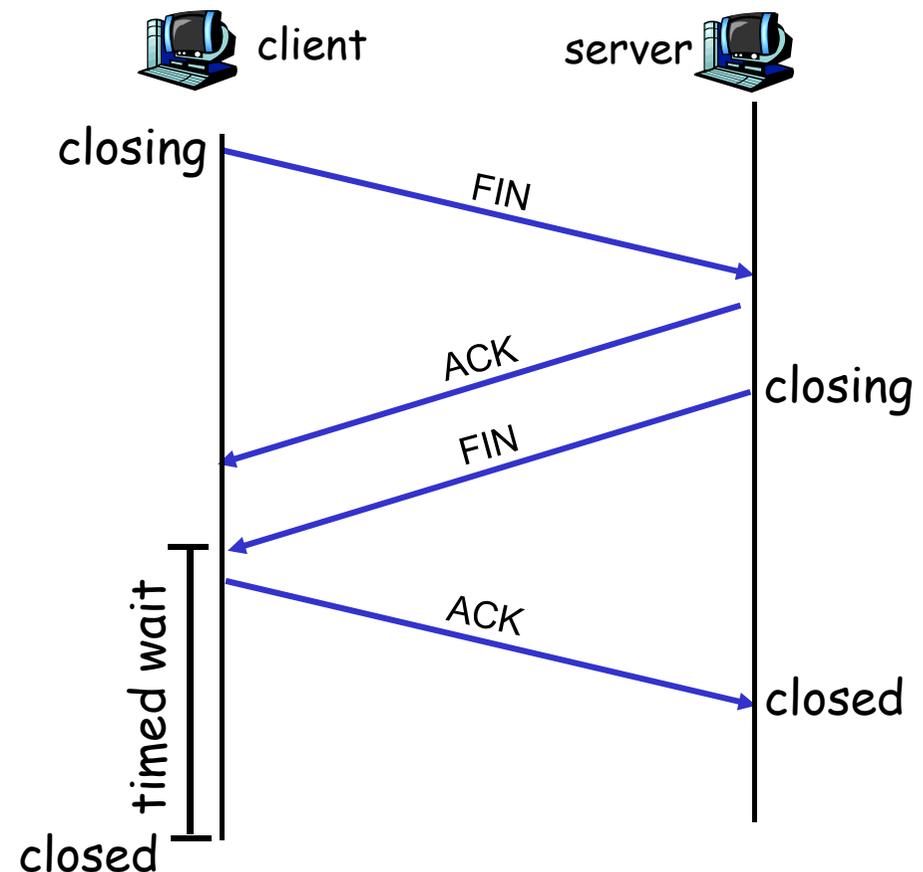
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

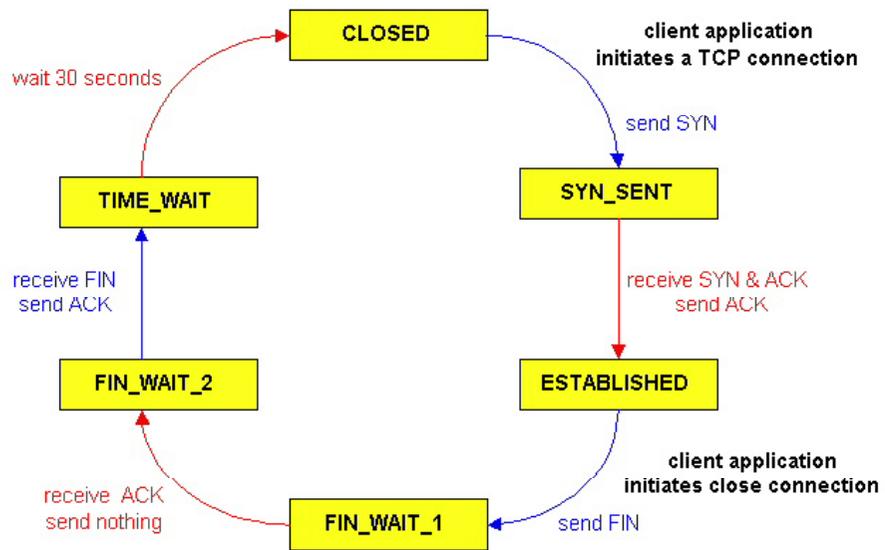
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

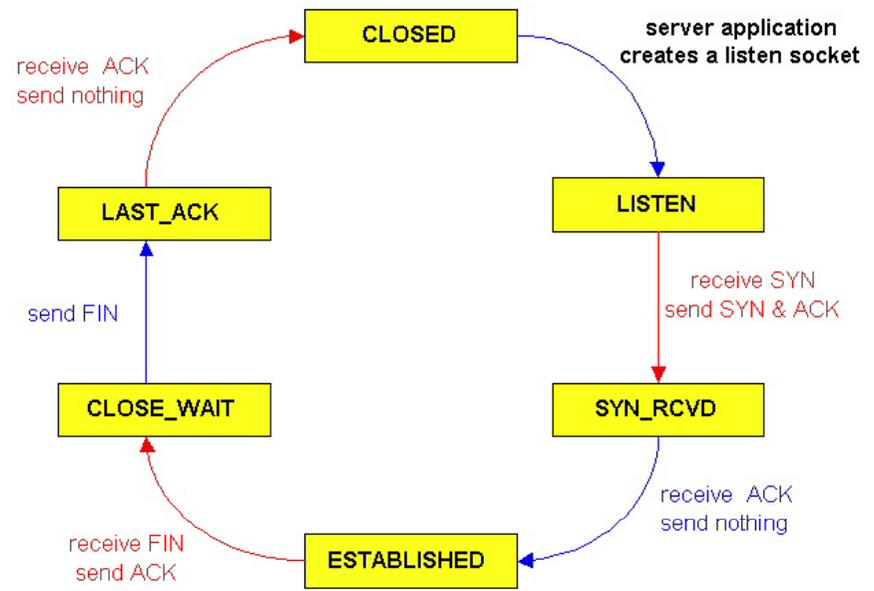


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



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

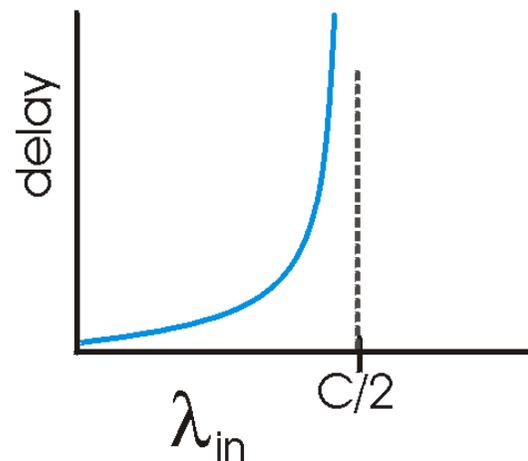
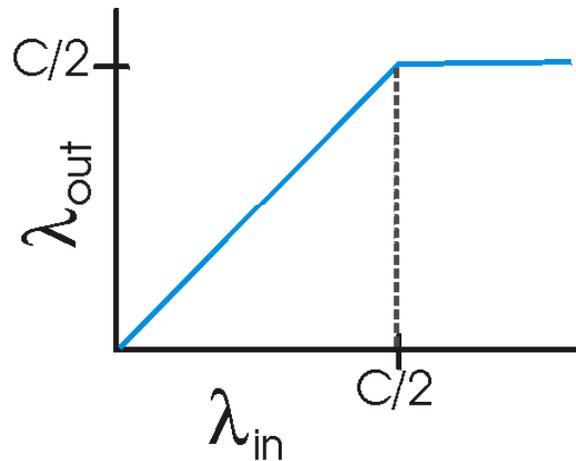
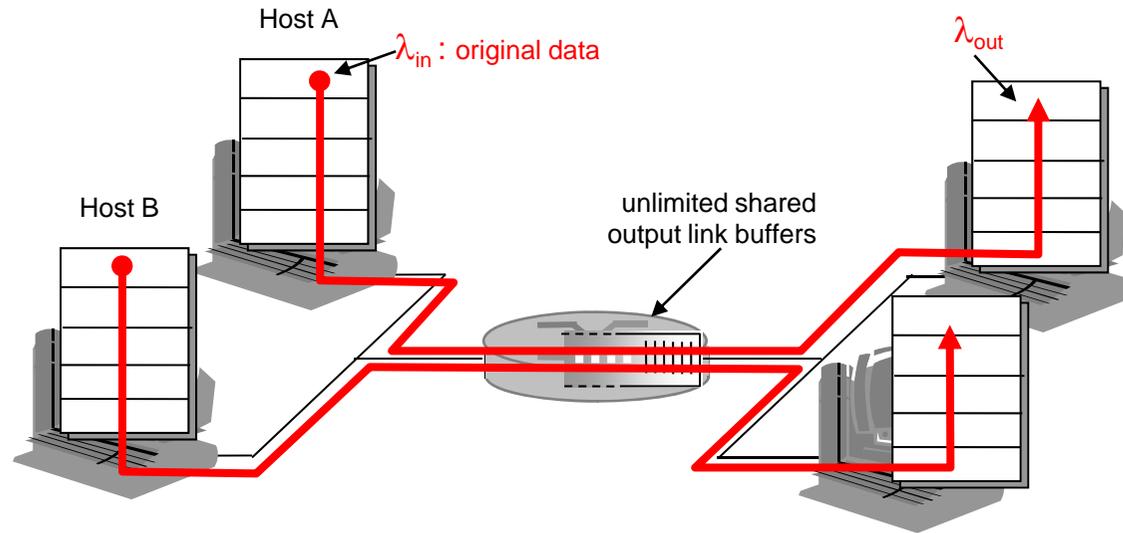
Principles of Congestion Control

Congestion:

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

Causes/costs of congestion: scenario 1

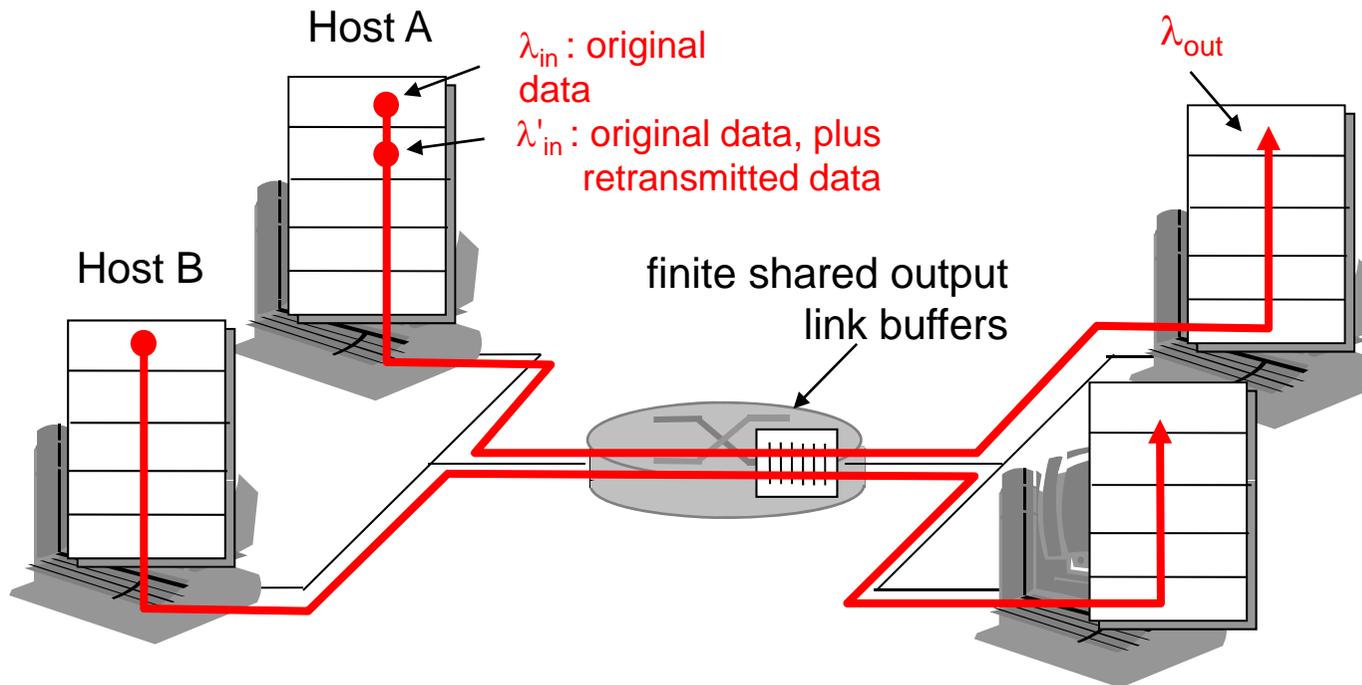
- two senders, two receivers
- one router, infinite buffers
- no retransmission



- large delays when congested
- maximum achievable throughput

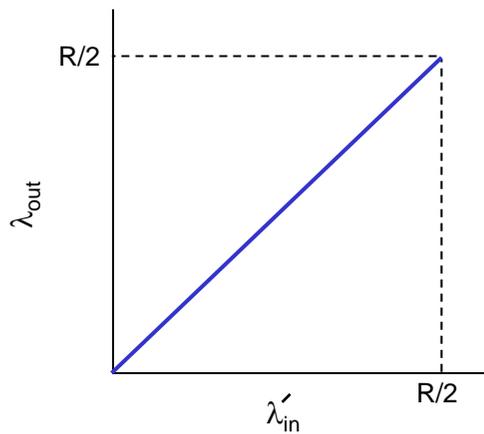
Causes/costs of congestion: scenario 2

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

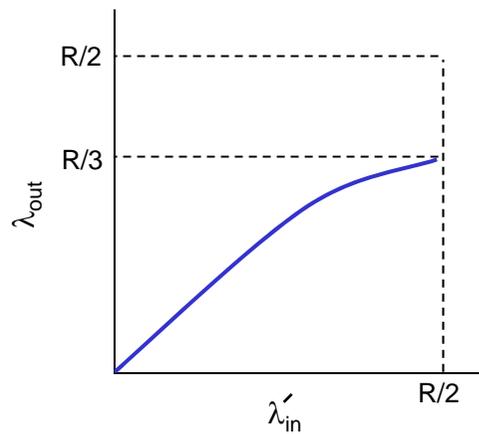


Causes/costs of congestion: scenario 2

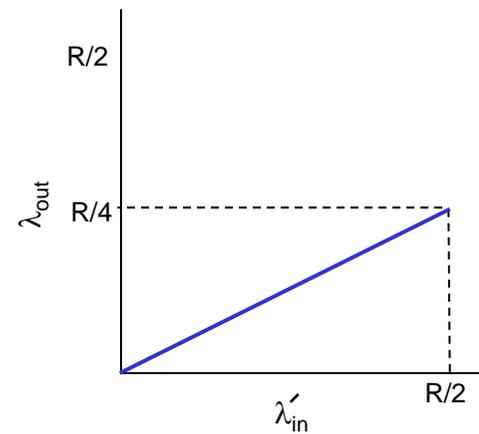
- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



a.



b.



c.

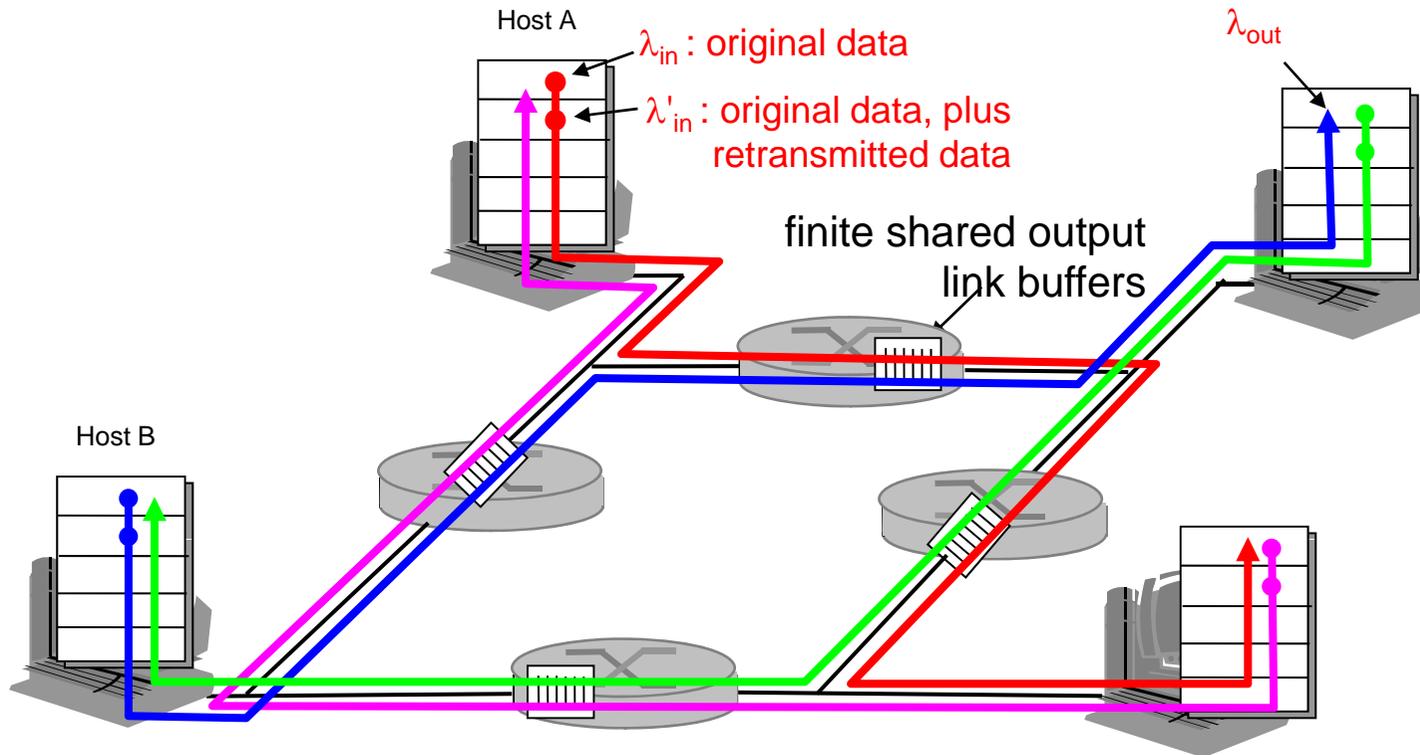
"costs" of congestion:

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

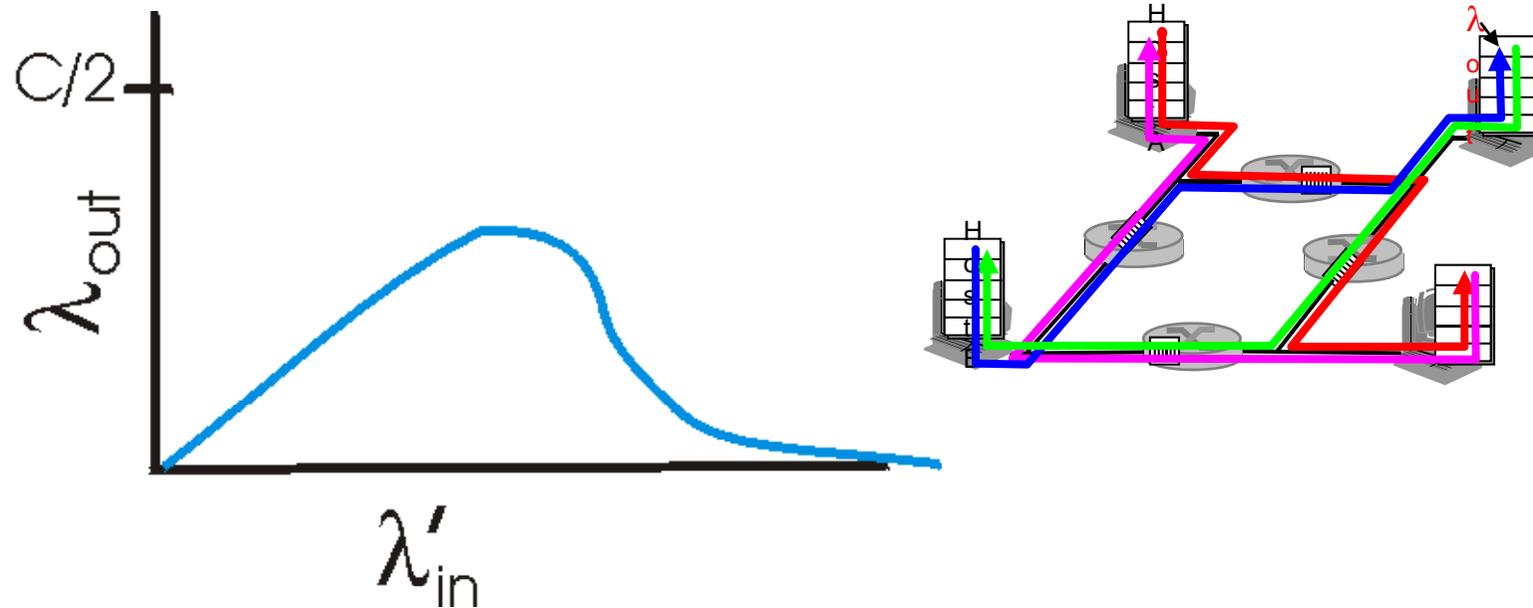
Causes/costs of congestion: scenario 3

- ❑ four senders
- ❑ multihop paths
- ❑ timeout/retransmit

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



Causes/costs of congestion: scenario 3



Another "cost" of congestion:

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

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

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

Network-assisted congestion control:

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

Case study: ATM ABR congestion control

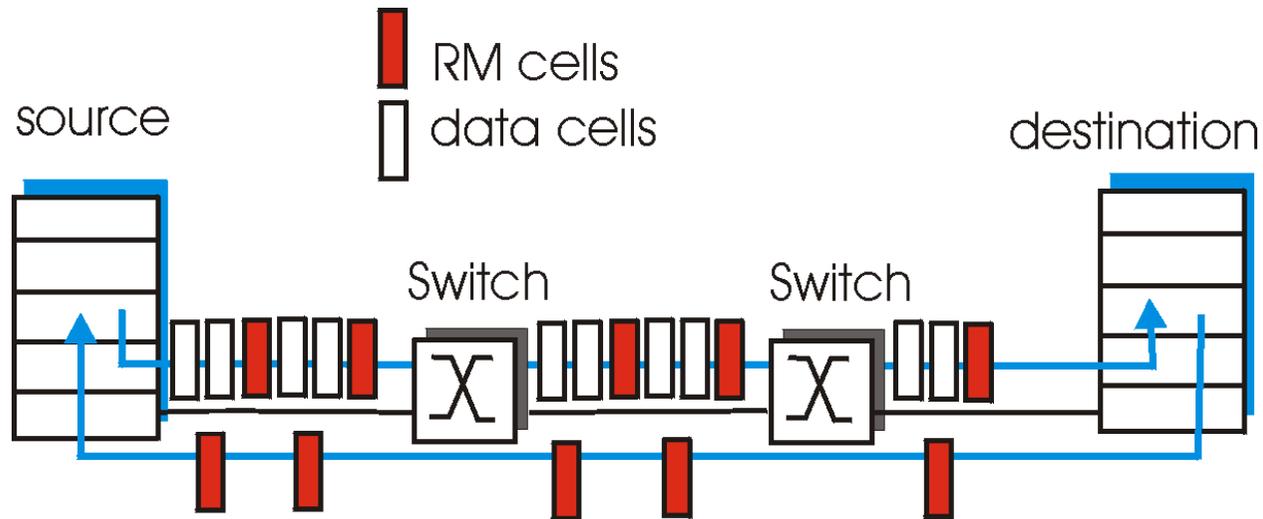
ABR: available bit rate:

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

RM (resource management) cells:

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

Case study: ATM ABR congestion control



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

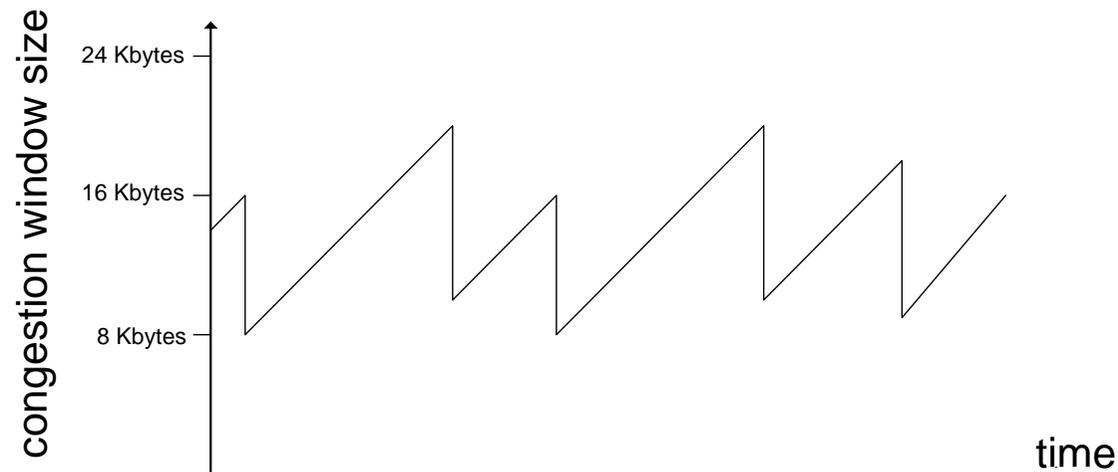
Chapter 3 outline

- ❑ 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 congestion control: additive increase, multiplicative decrease

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

Saw tooth behavior: probing for bandwidth



TCP Congestion Control: details

- sender limits transmission:

$$\text{LastByteSent} - \text{LastByteAcked} \leq \text{CongWin}$$

- Roughly,

$$\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}$$

- CongWin is dynamic, function of perceived network congestion

How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (CongWin) after loss event

three mechanisms:

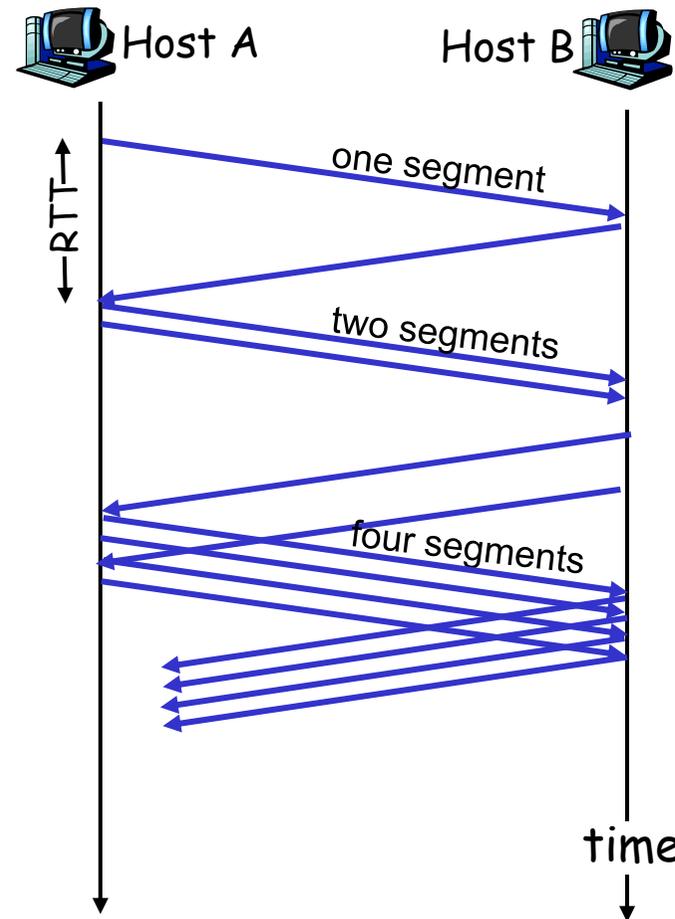
- AIMD
- slow start
- conservative after timeout events

TCP Slow Start

- ❑ When connection begins, CongWin = 1 MSS
 - Example: MSS = 500 bytes & RTT = 200 msec
 - initial rate = 20 kbps
- ❑ available bandwidth may be \gg MSS/RTT
 - desirable to quickly ramp up to respectable rate
- ❑ When connection begins, increase rate exponentially fast until first loss event

TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
 - double CongWin every RTT
 - done by incrementing CongWin for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



Refinement: inferring loss

- ❑ After 3 dup ACKs:
 - CongWin is cut in half
 - window then grows linearly
- ❑ But after timeout event:
 - CongWin instead set to 1 MSS;
 - window then grows exponentially
 - to a threshold, then grows linearly

Philosophy:

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

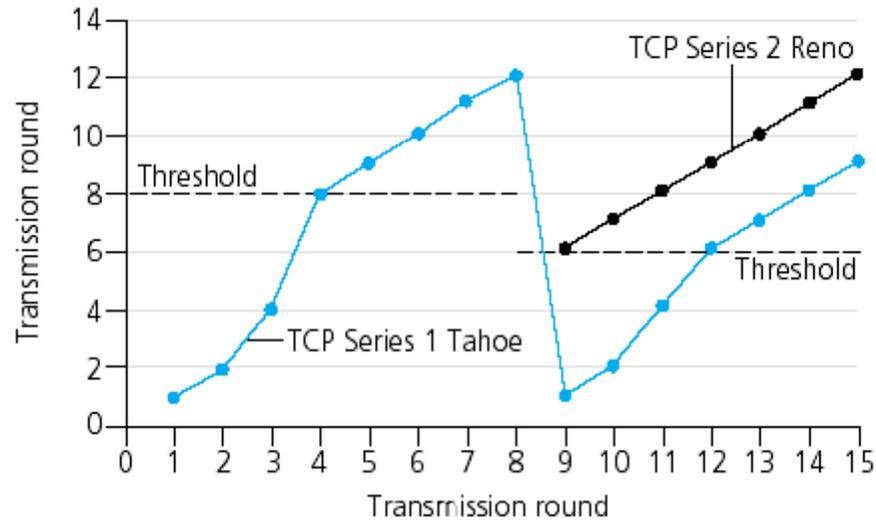
Refinement

Q: When should the exponential increase switch to linear?

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

Implementation:

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



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

- ❑ What's the average throughput of TCP as a function of window size and RTT?
 - Ignore slow start
- ❑ Let W be the window size when loss occurs.
- ❑ When window is W , throughput is W/RTT
- ❑ Just after loss, window drops to $W/2$, throughput to $W/2RTT$.
- ❑ Average throughput: $.75 W/RTT$

TCP Futures: TCP over "long, fat pipes"

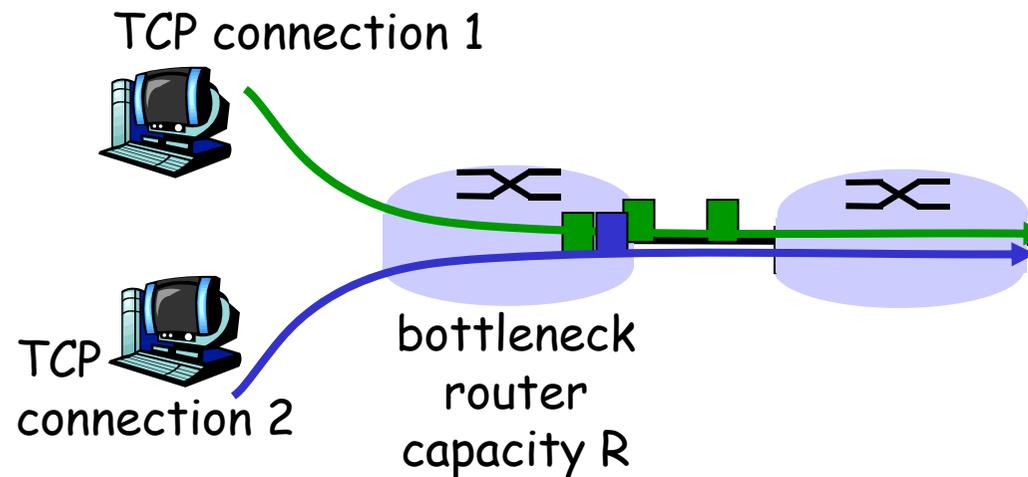
- ❑ Example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- ❑ Requires window size $W = 83,333$ in-flight segments
- ❑ Throughput in terms of loss rate:

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

- ❑ $\rightarrow L = 2 \cdot 10^{-10}$ *Wow*
- ❑ New versions of TCP for high-speed

TCP Fairness

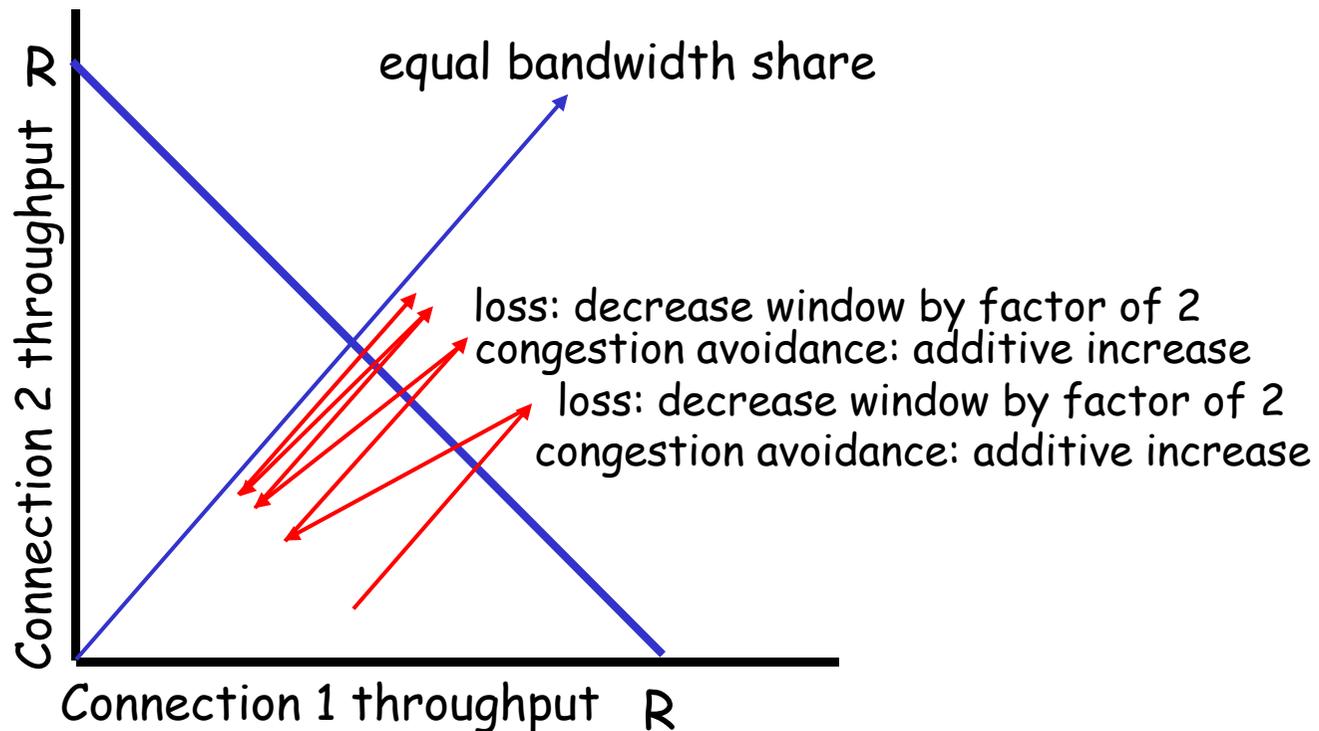
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



Why is TCP fair?

Two competing sessions:

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



Fairness (more)

Fairness and UDP

- ❑ Multimedia apps often do not use TCP
 - do not want rate throttled by congestion control
- ❑ Instead use UDP:
 - pump audio/video at constant rate, tolerate packet loss
- ❑ Research area: TCP friendly

Fairness and parallel TCP connections

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

Chapter 3: Summary

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

Next:

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