2110472 Computer Network

Transport Layer & Multimedia Networking

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Course Information

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Course website:
http://www.cp.eng.chula.ac.th/~kultida/classes.html

Lecture schedule: Friday 13:00-16:00

Course materials: Lecture slides
Selected textbooks

Chapter 3
Transport Layer

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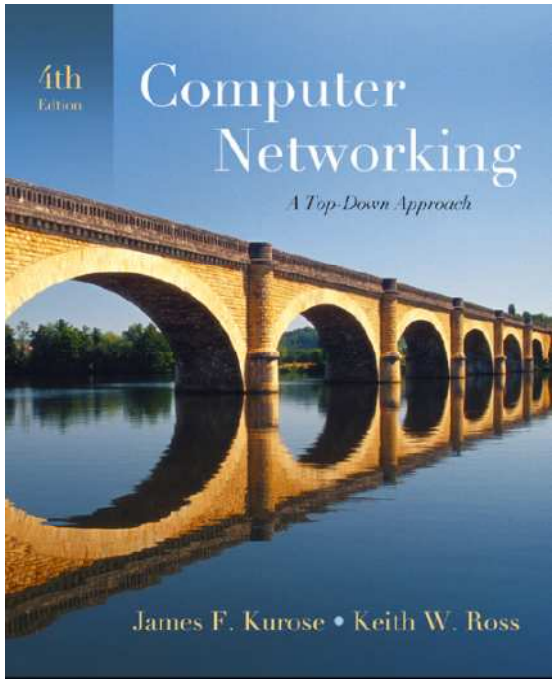


Computer Networking: A Top Down Approach 4th edition.
Jim Kurose, Keith Ross
Addison-Wesley, July 2007.

A note on the use of these ppt slides:

Chapter 3: Transport Layer

Our goals:

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

Chapter 3 outline

3.1 Transport-layer services

3.2 Multiplexing and demultiplexing

3.3 Connectionless transport: UDP

3.4 Principles of reliable data transfer

3.5 Connection-oriented transport: TCP

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

3.6 Principles of congestion control

3.7 TCP congestion control

Transport services and protocols

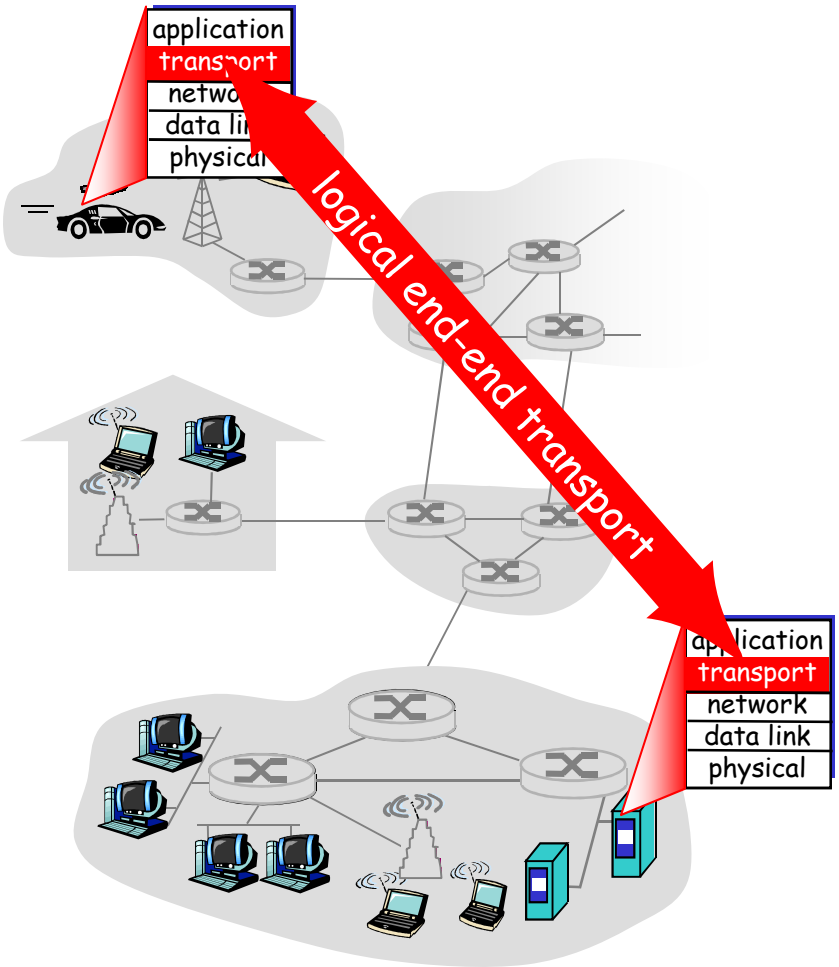
provide *logical communication* between app processes running on different hosts

transport protocols run in end systems

- send side: breaks app messages into segments, passes to network layer
- rcv side: reassembles segments into messages, passes to app layer

more than one transport protocol available to apps

- Internet: TCP and UDP



Transport vs. network layer

network layer: logical communication between hosts

transport layer: logical communication between processes

- relies on, enhances, network layer services

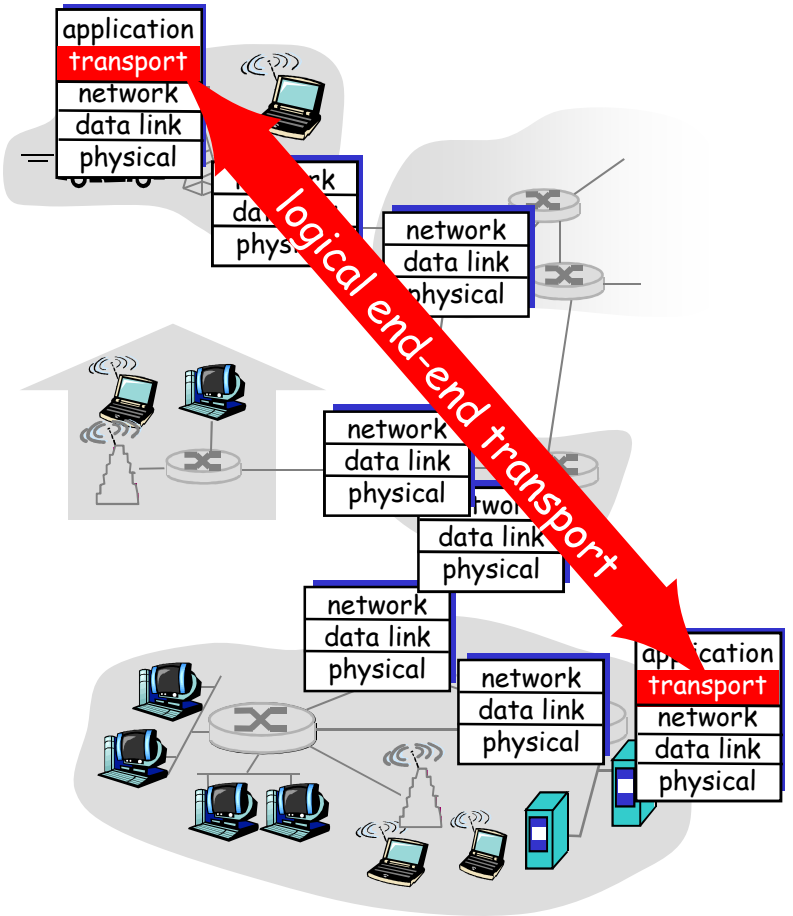
Household analogy:

12 kids sending letters to 12 kids

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

Internet transport-layer protocols

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



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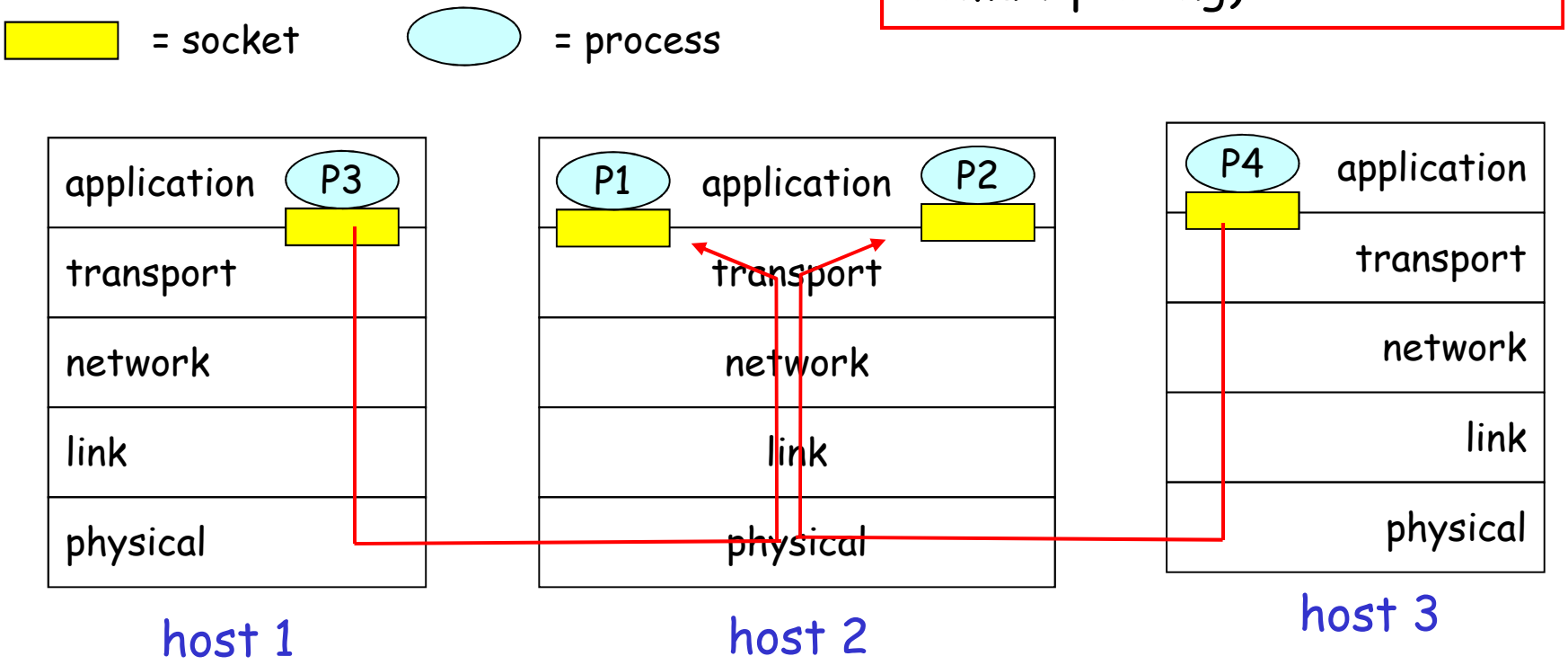
Multiplexing/demultiplexing

Demultiplexing at rcv host:

delivering received segments to correct socket

Multiplexing at send host:

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

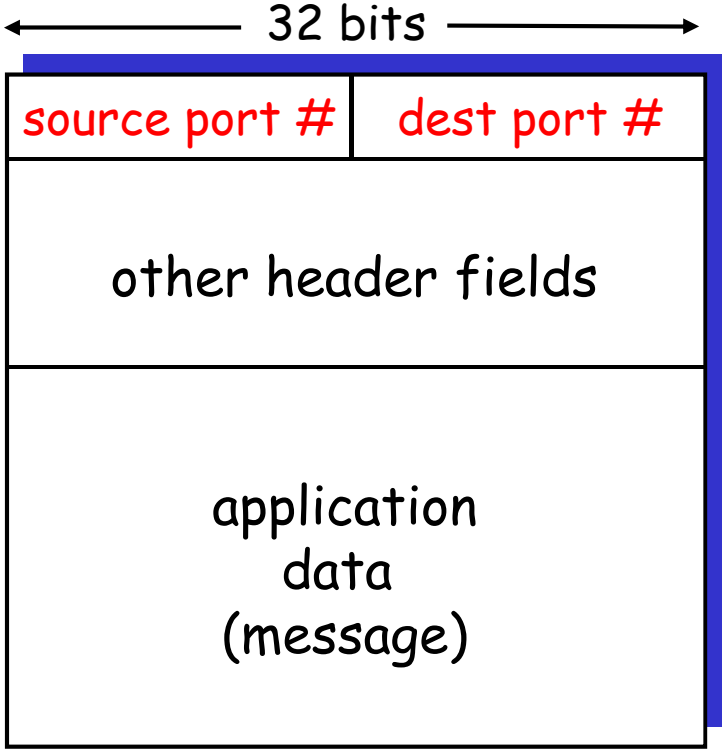


How demultiplexing works

host receives IP datagrams

- each datagram has source IP address, destination IP address
- each datagram carries 1 transport-layer segment
- each segment has source, destination port number

host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

Create sockets with port numbers:

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

UDP socket identified by two-tuple:

(dest IP address, dest port number)

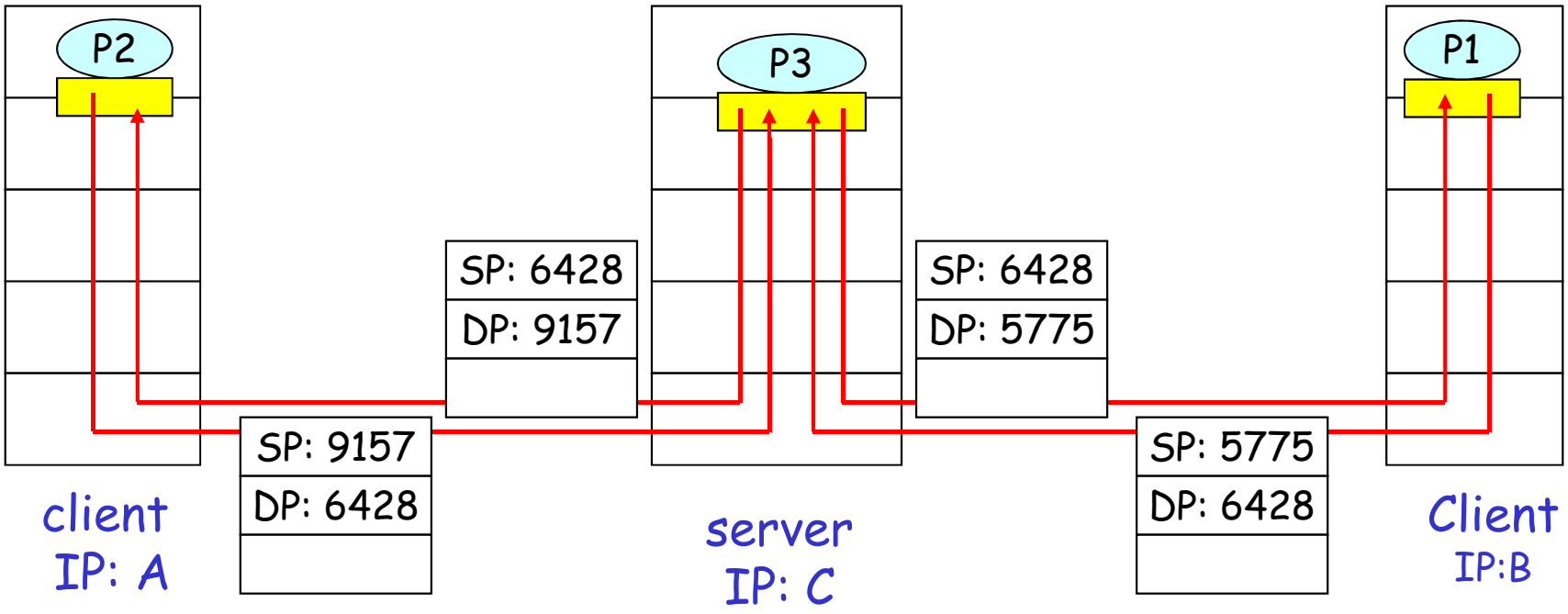
When host receives UDP segment:

- checks destination port number in segment
- directs UDP segment to socket with that port number

IP datagrams with different source IP addresses and/or source port numbers directed to same socket

Connectionless demux (cont)

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



SP provides "return address"

Connection-oriented demux

TCP socket identified by 4-tuple:

- source IP address
- source port number
- dest IP address
- dest port number

recv host uses all four values to direct segment to appropriate socket

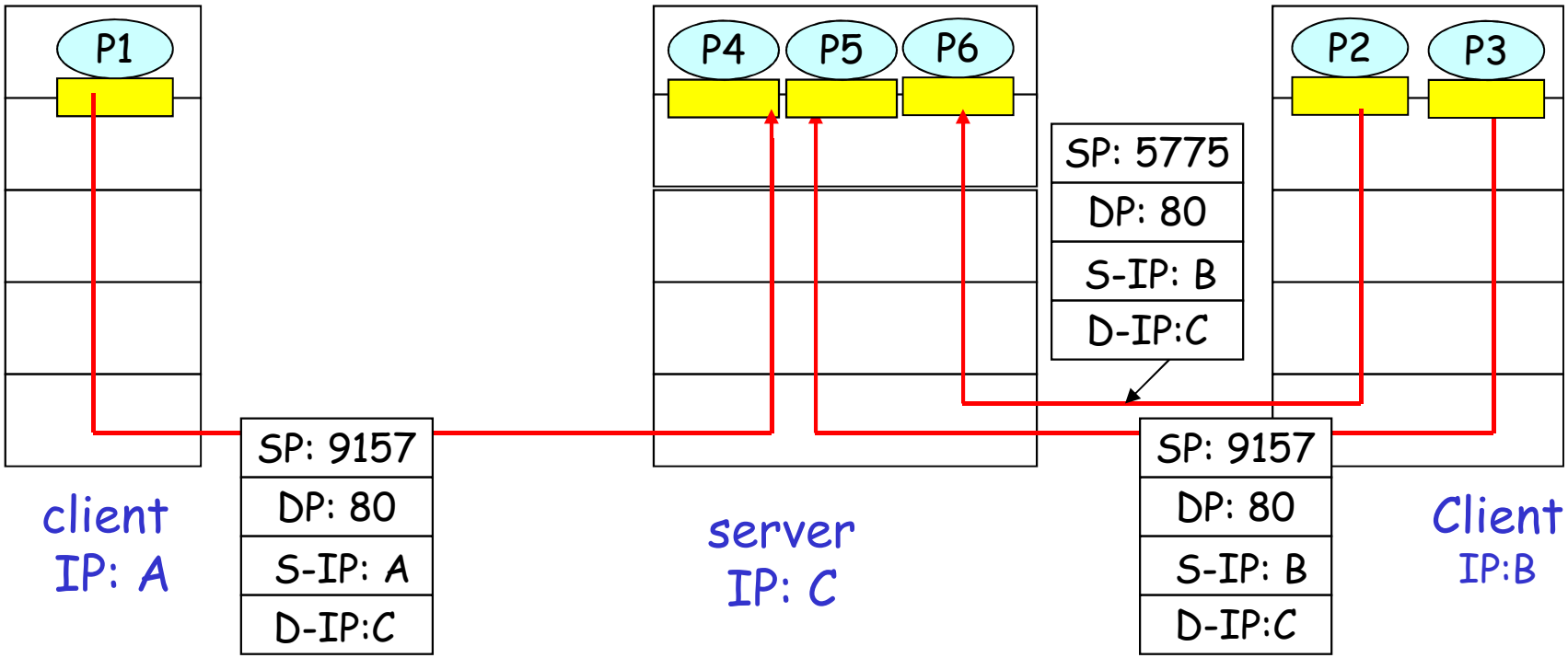
Server host may support many simultaneous TCP sockets:

- each socket identified by its own 4-tuple

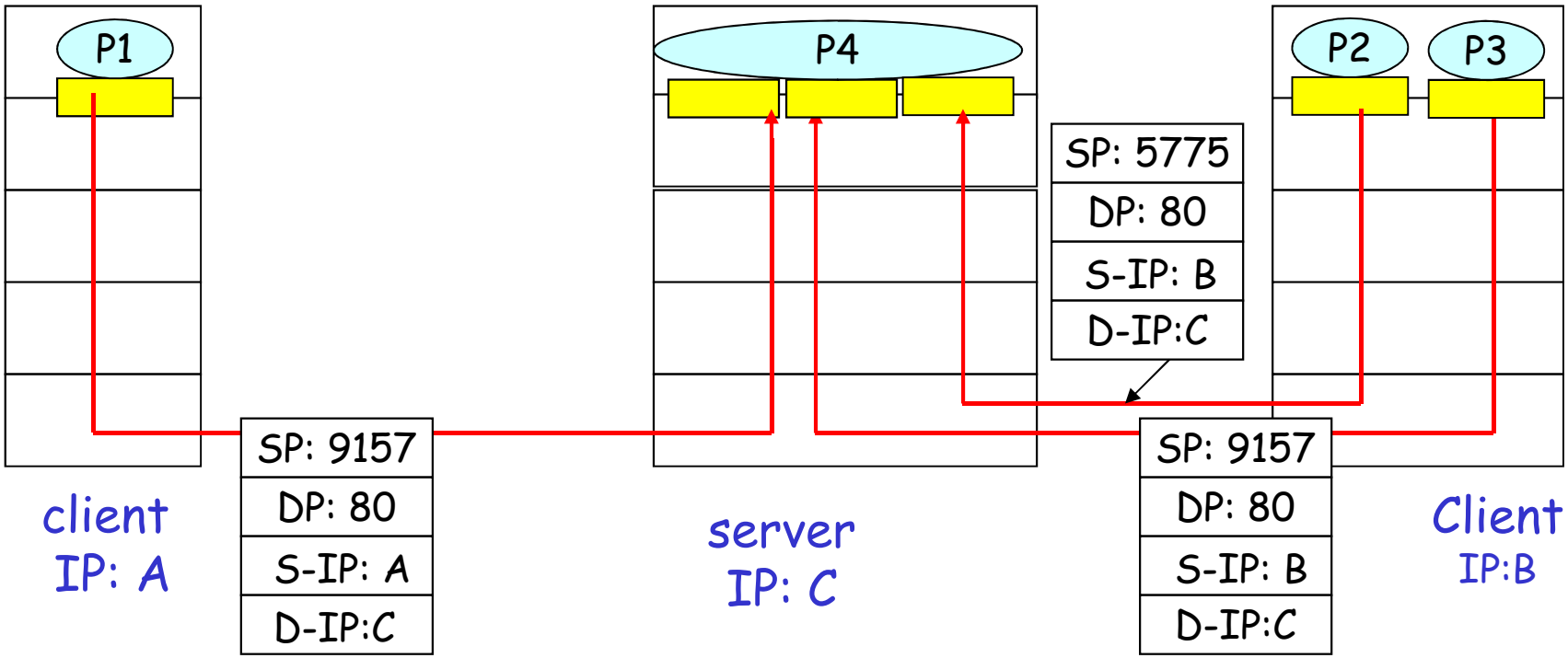
Web servers have different sockets for each connecting client

- non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



Connection-oriented demux: Threaded Web Server



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UDP: User Datagram Protocol [RFC 768]

"no frills," "bare bones" Internet transport protocol

"best effort" service, UDP segments may be:

- lost
- delivered out of order to app

connectionless:

- no handshaking between UDP sender, receiver
- each UDP segment handled independently of others

Why is there a UDP?

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

UDP: more

often used for streaming multimedia apps

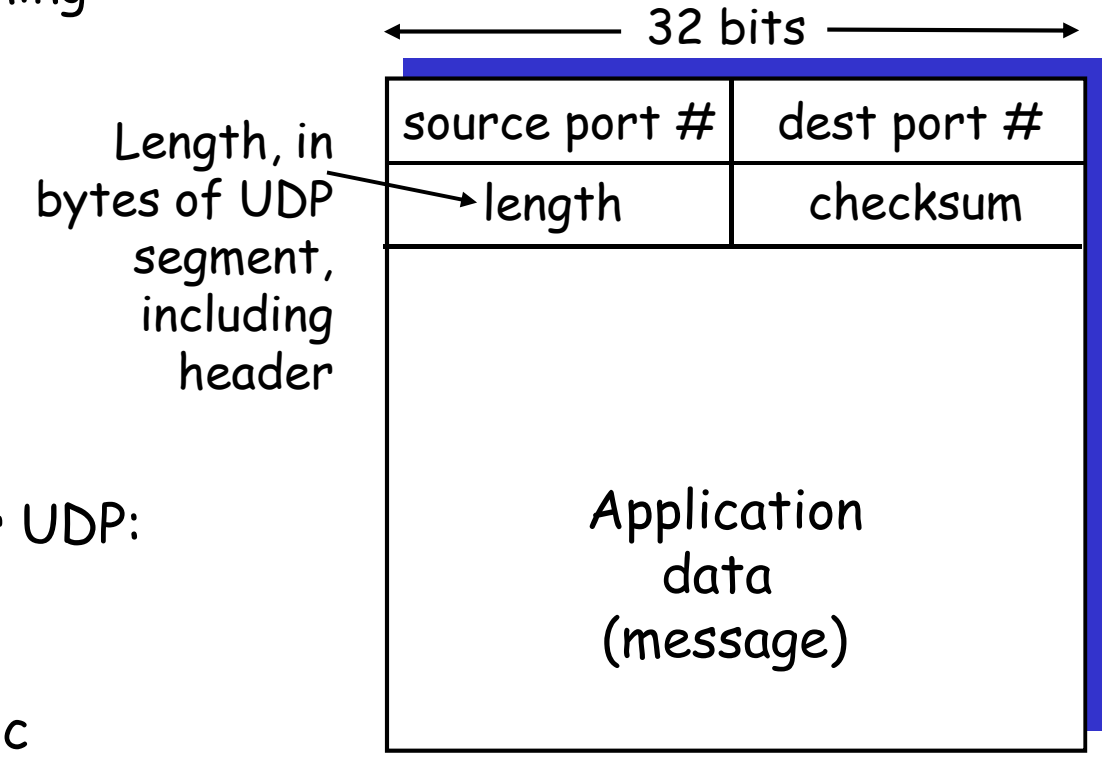
- loss tolerant
- rate sensitive

other UDP uses

- DNS
- SNMP

reliable transfer over UDP: add reliability at application layer

- application-specific error recovery!



UDP segment format

UDP checksum

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

Sender:

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

Receiver:

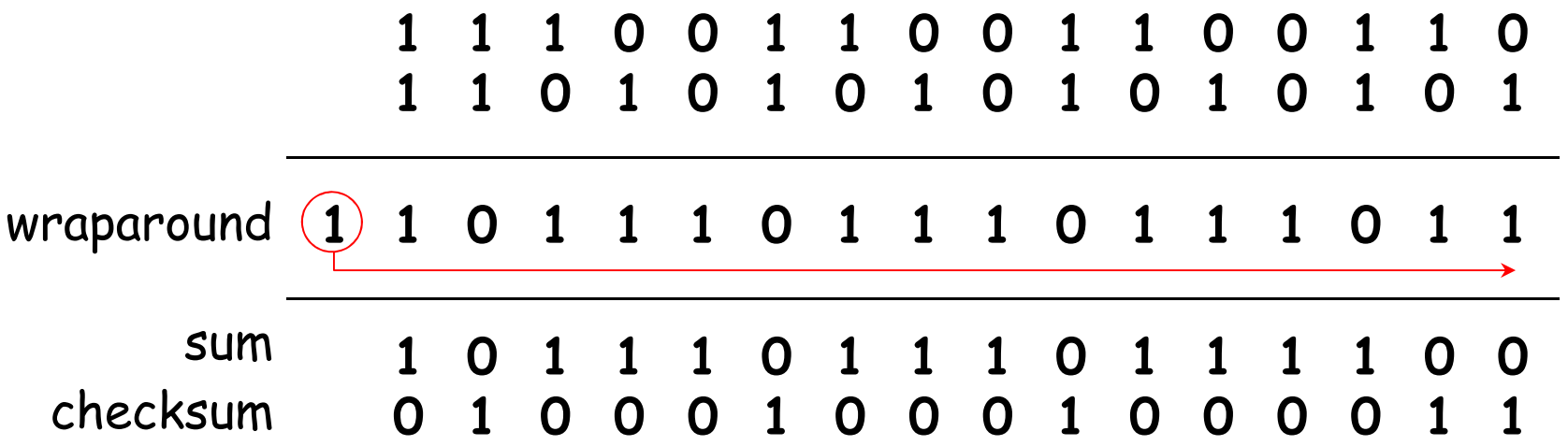
- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later
  - ....

Internet Checksum Example

Note

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

Example: add two 16-bit integers



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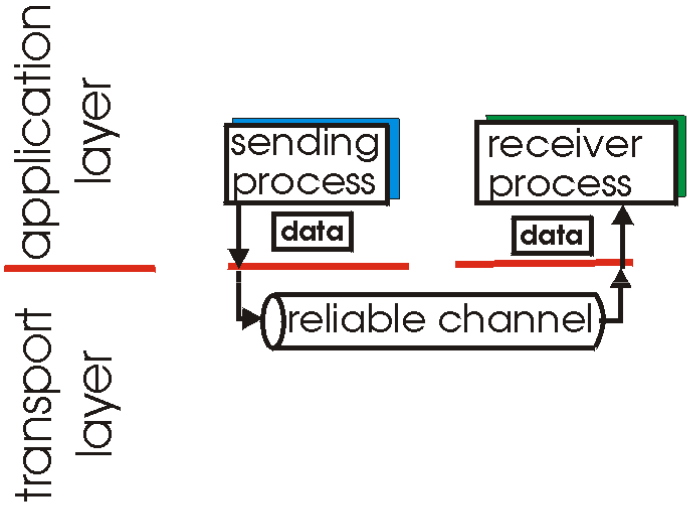
Principles of Reliable data transfer

important in app., transport, link layers

top-10 list of important networking topics!



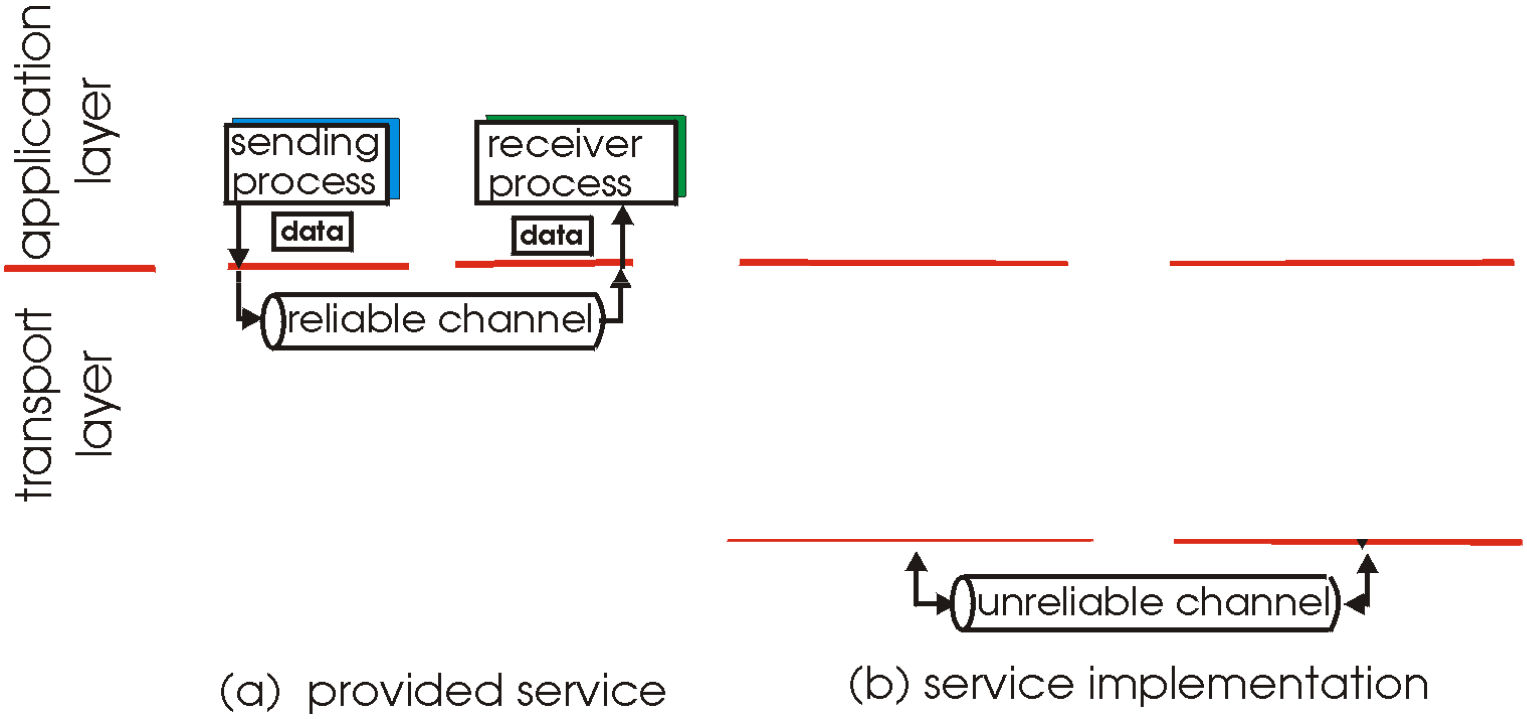
(a) provided service

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

Principles of Reliable data transfer

important in app., transport, link layers

top-10 list of important networking topics!



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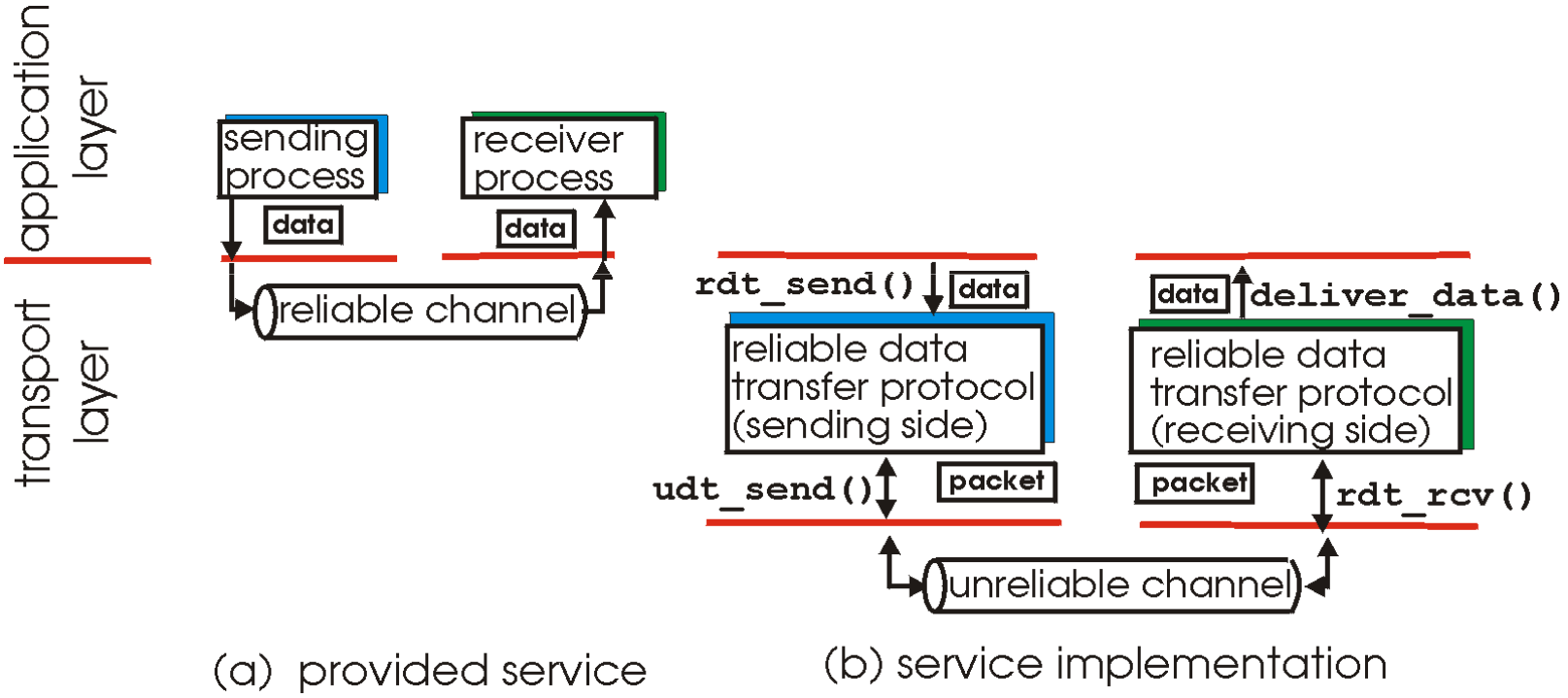
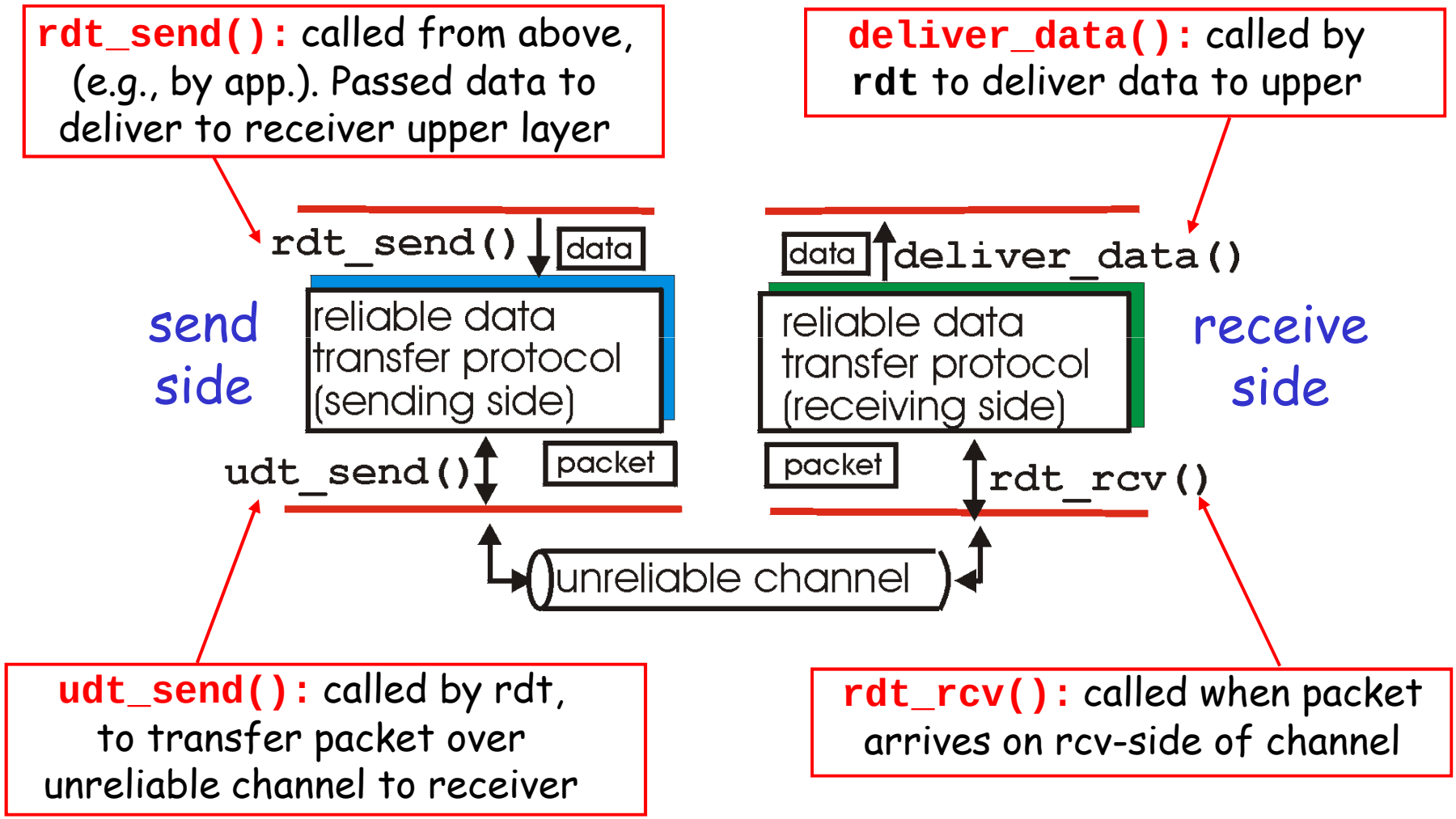


characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

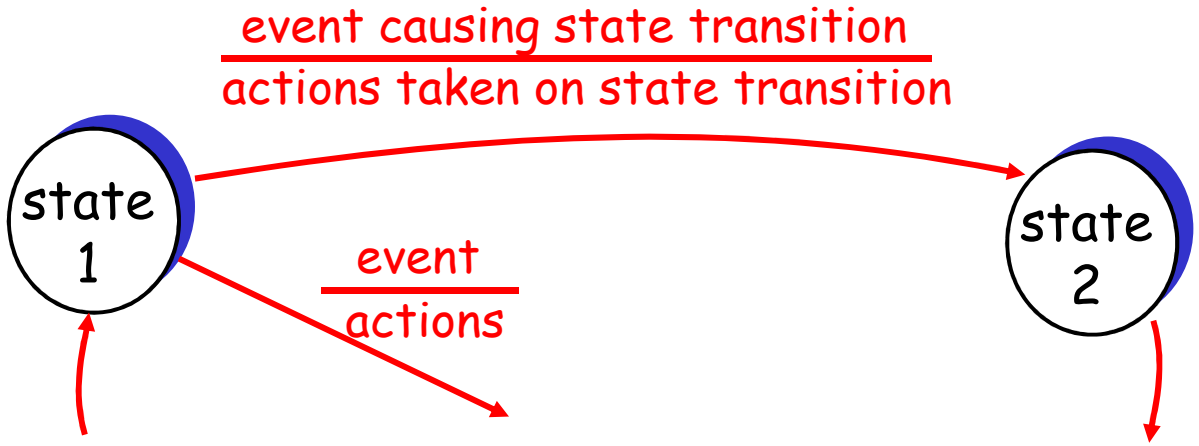


Reliable data transfer: getting started

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

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



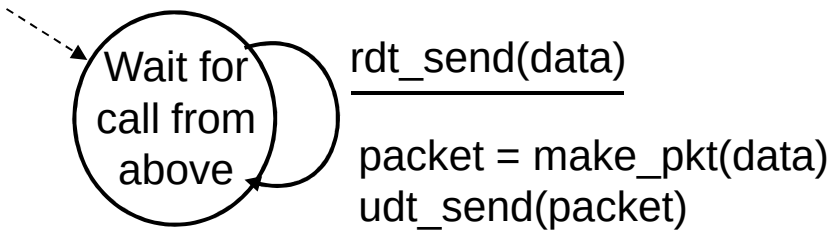
Rdt1.0: reliable transfer over a reliable channel

underlying channel perfectly reliable

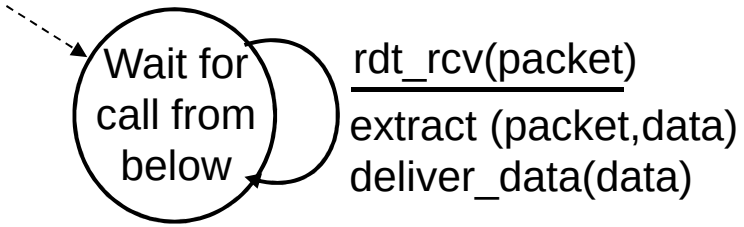
- no bit errors
- no loss of packets

separate FSMs for sender, receiver:

- sender sends data into underlying channel
- receiver read data from underlying channel



sender



receiver

Rdt2.0: channel with bit errors

underlying channel may flip bits in packet

- checksum to detect bit errors

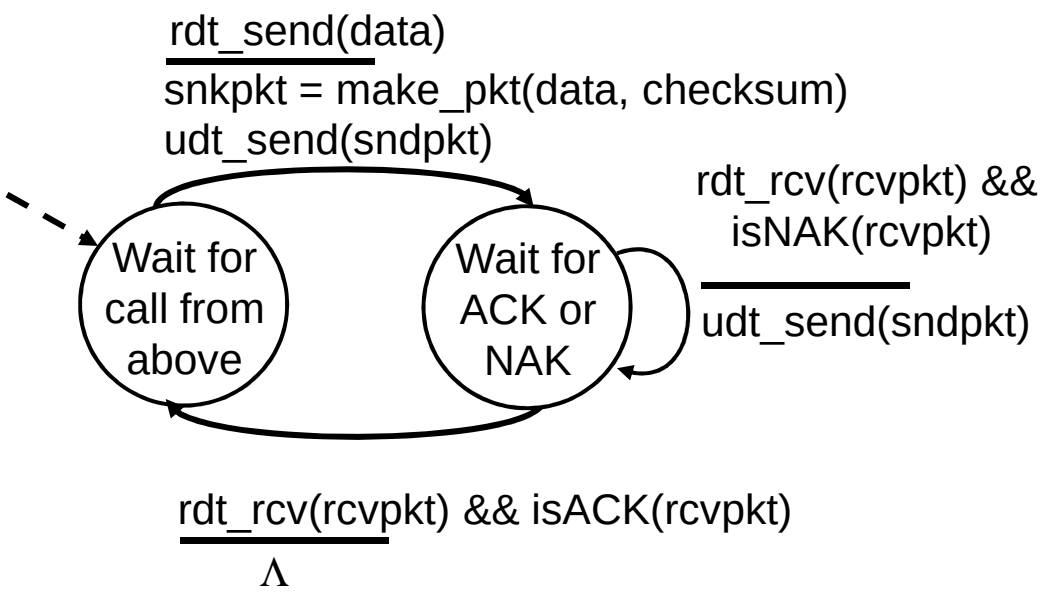
the question: how to recover from errors:

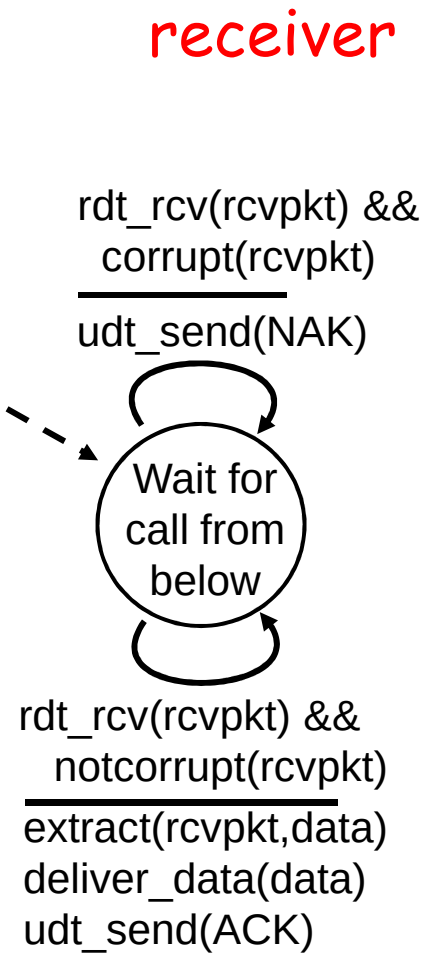
- *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
- *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
- sender retransmits pkt on receipt of NAK

new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):

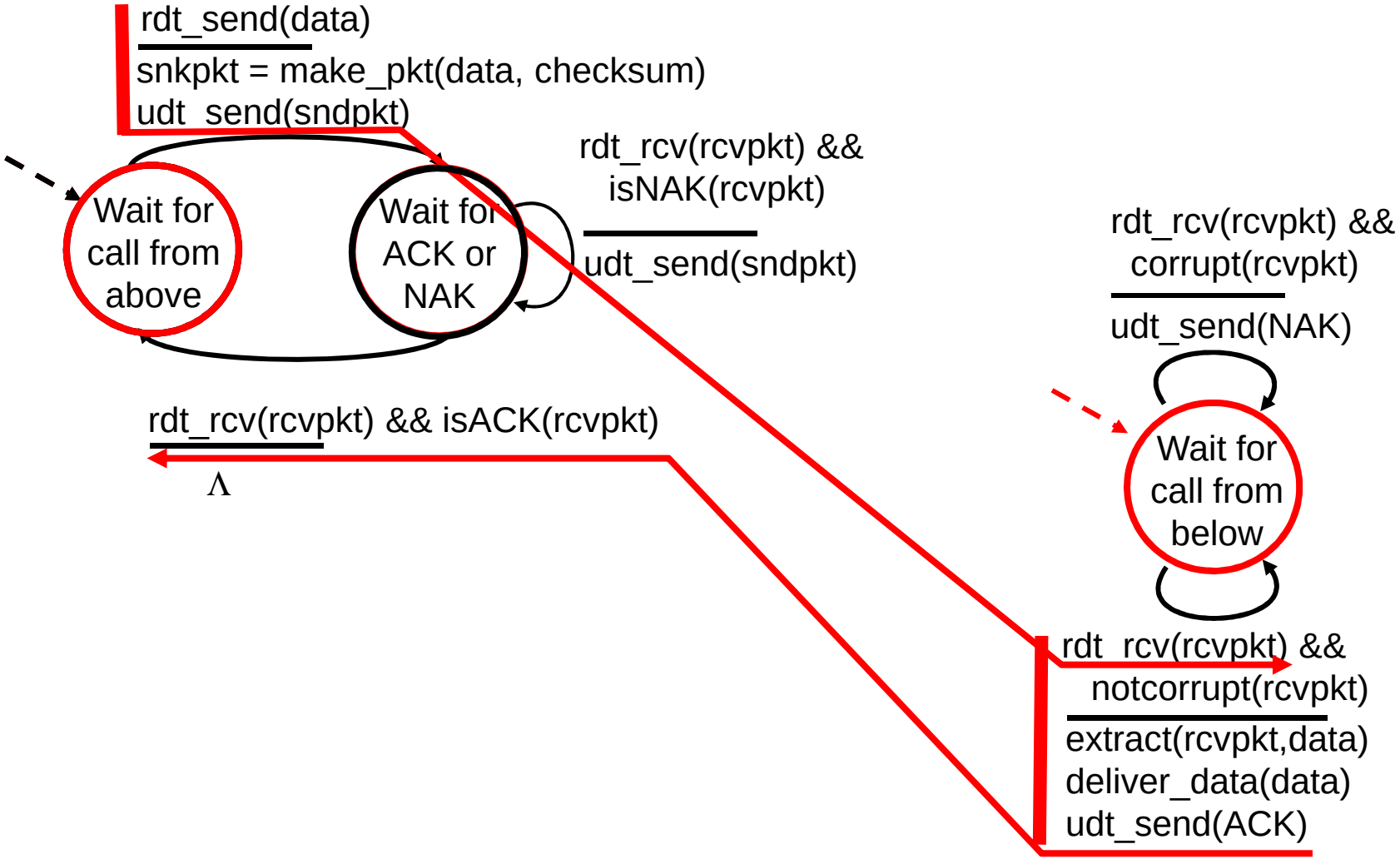
- error detection
- receiver feedback: control msgs (ACK,NAK) rcvr->sender

rdt2.0: FSM specification

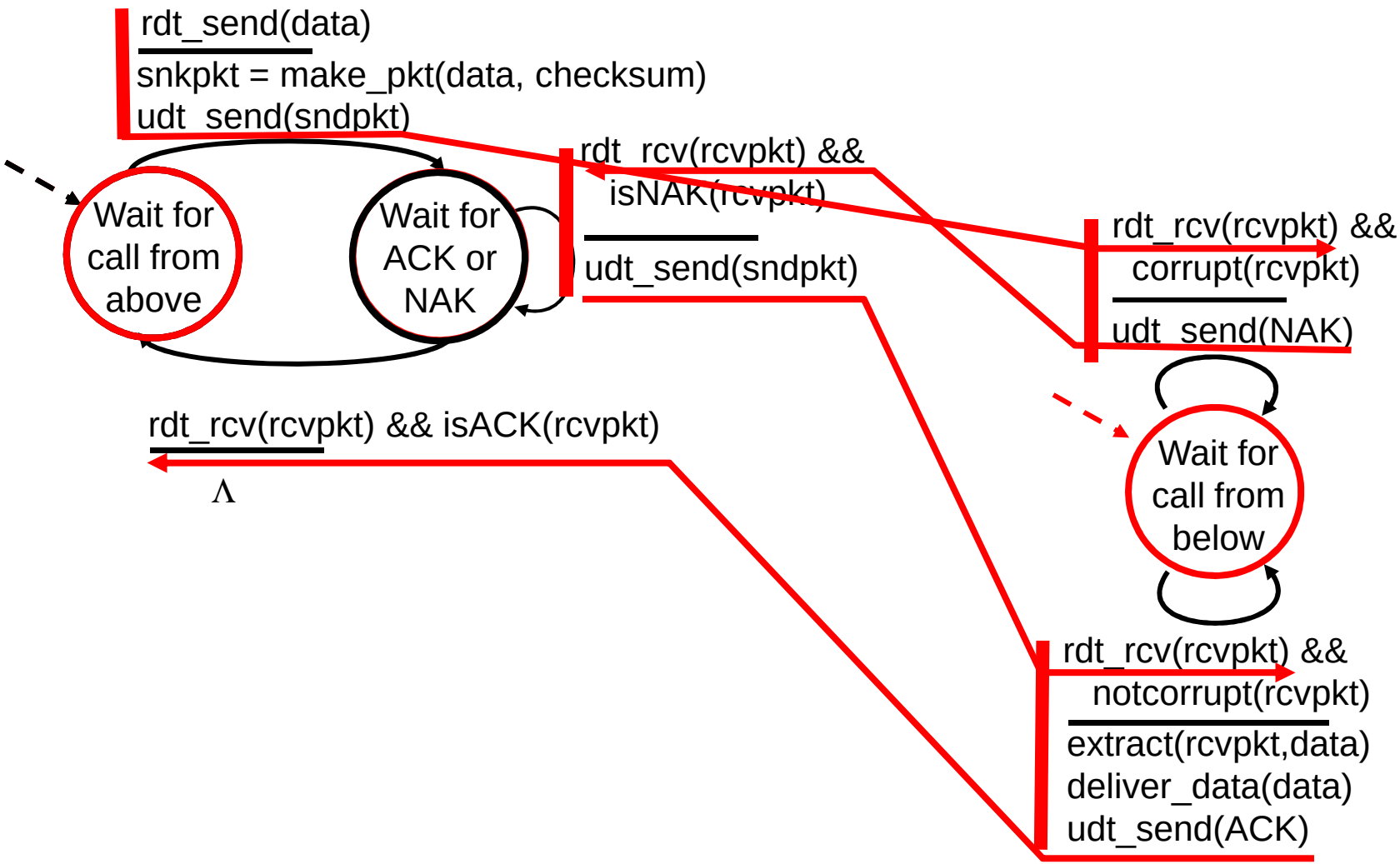




rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

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

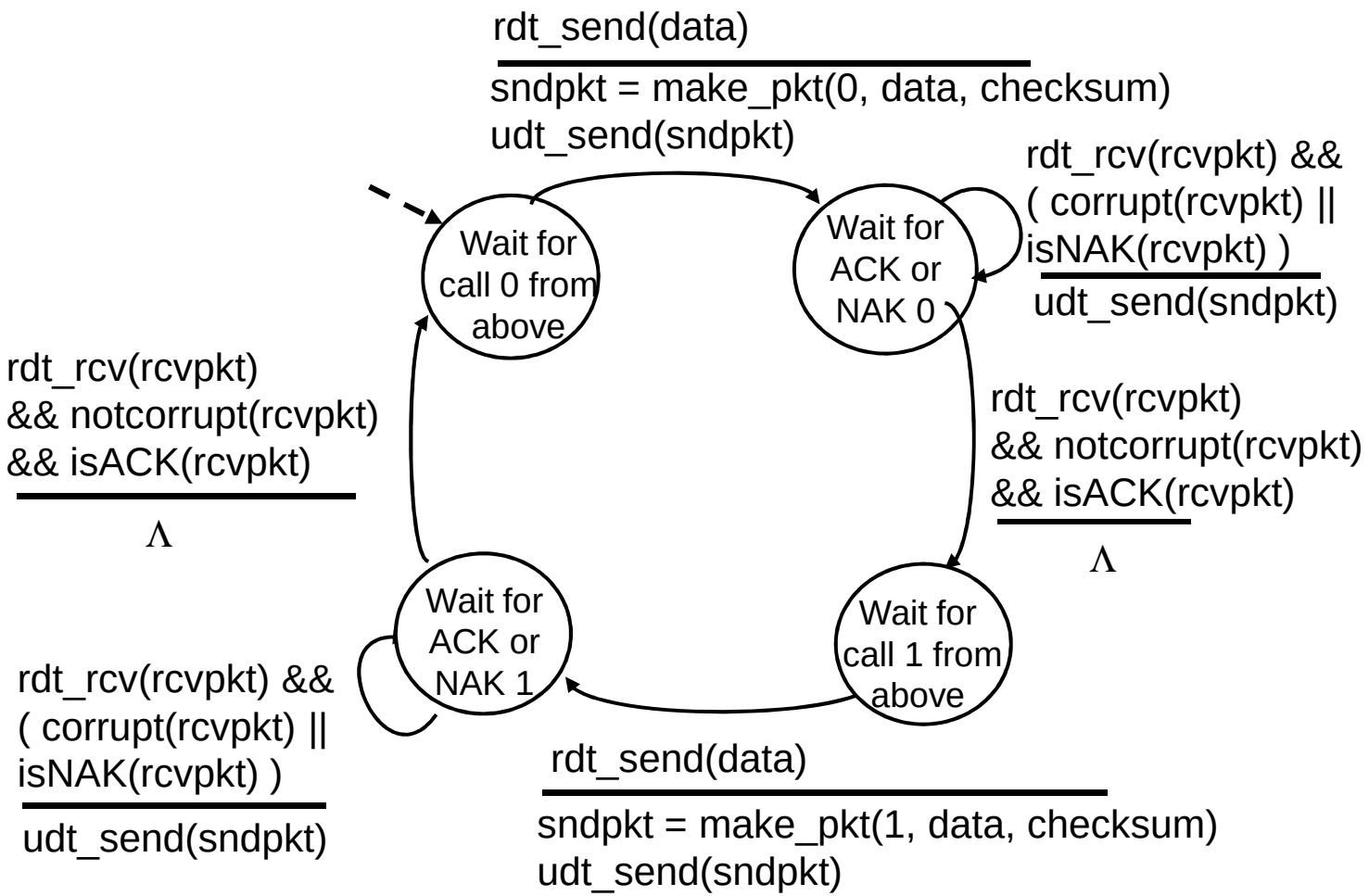
Handling duplicates:

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

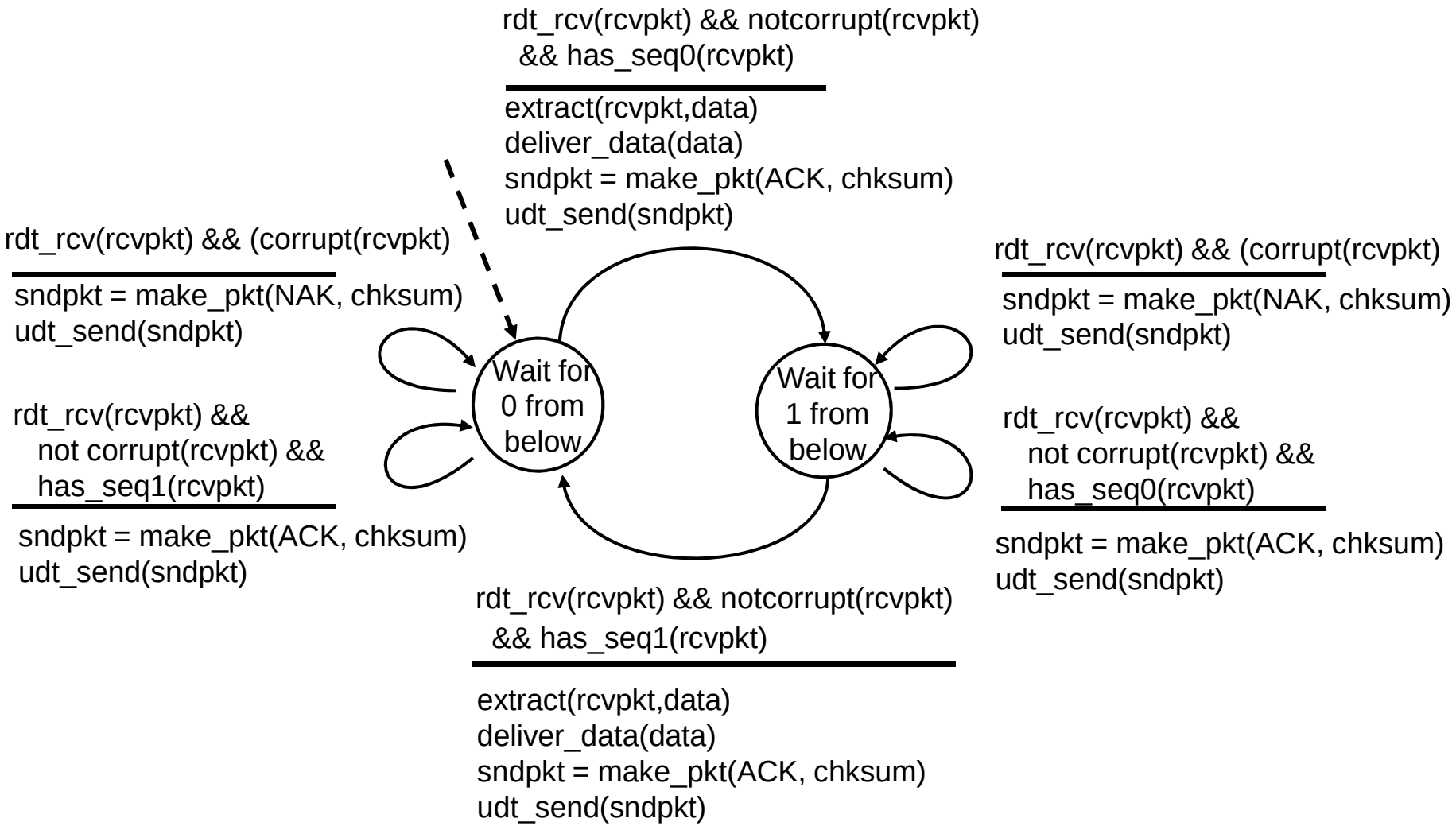
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

Sender:

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

Receiver:

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

rdt2.2: a NAK-free protocol

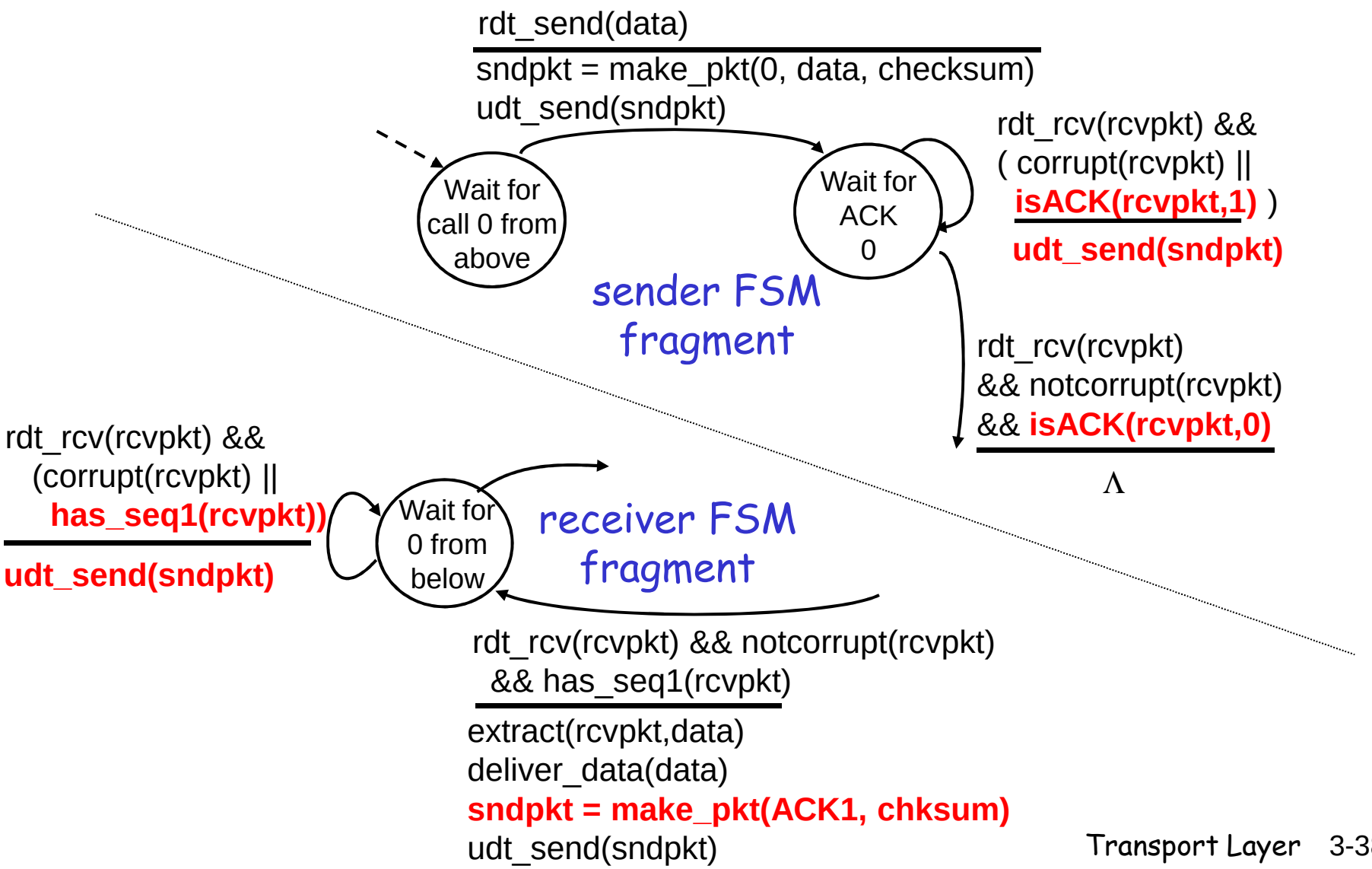
same functionality as rdt2.1, using ACKs only

instead of NAK, receiver sends ACK for last pkt received OK

- receiver must *explicitly* include seq # of pkt being ACKed

duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

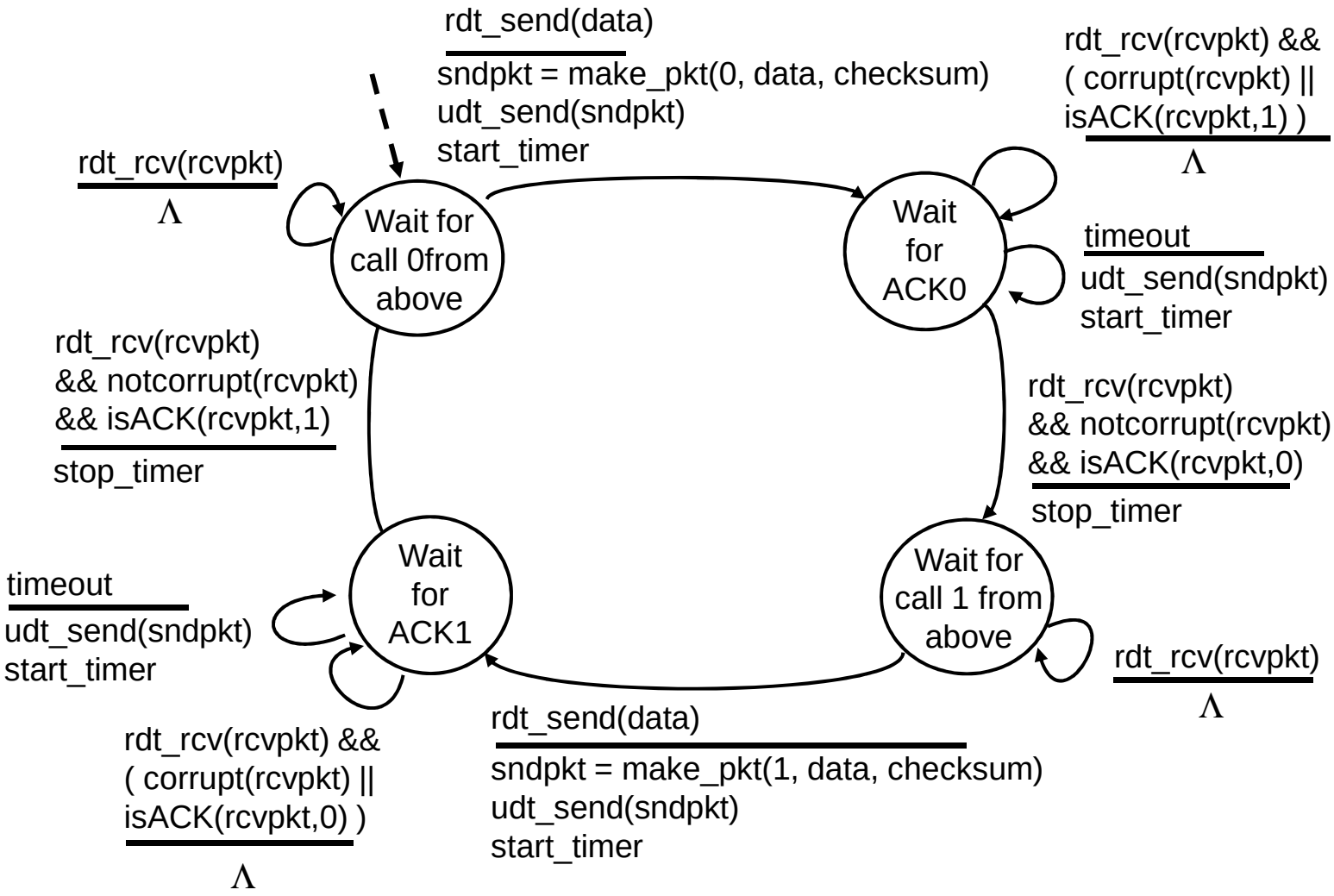
New assumption:

- underlying channel can also lose packets (data or ACKs)
  - checksum, seq. #, ACKs, retransmissions will be of help, but not enough

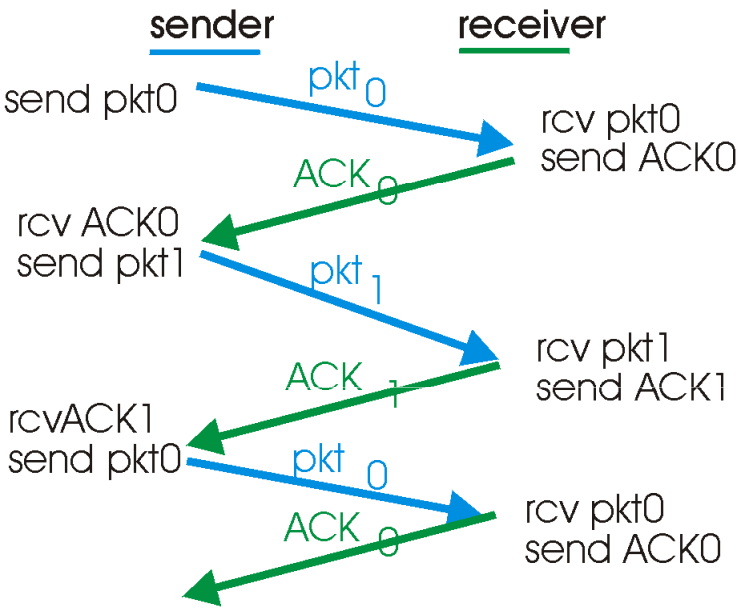
Approach: sender waits "reasonable" amount of time for ACK

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

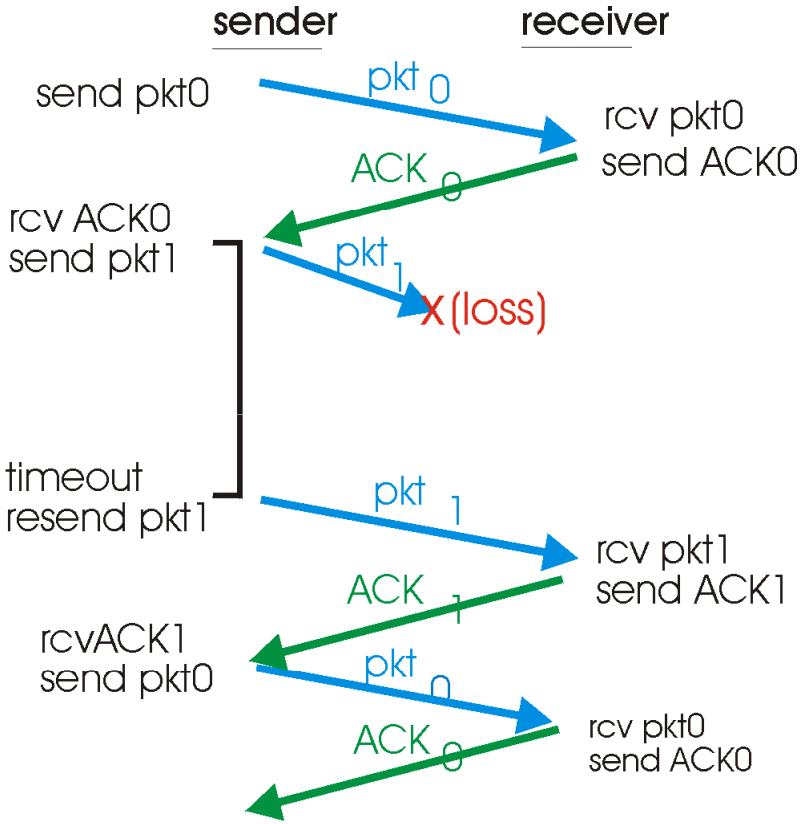
rdt3.0 sender



rdt3.0 in action

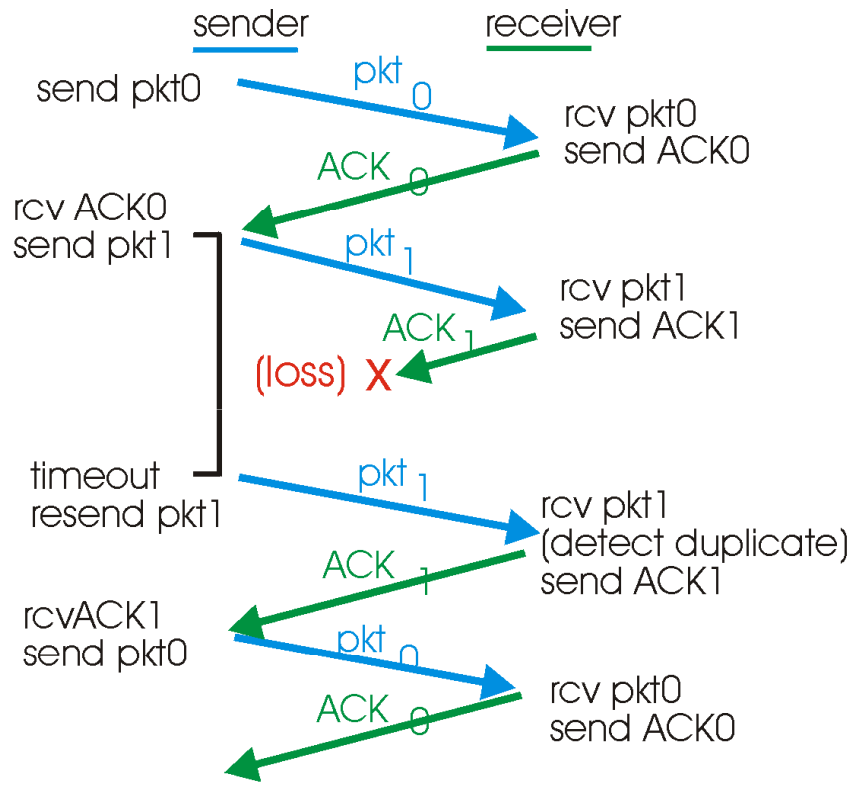


(a) operation with no loss

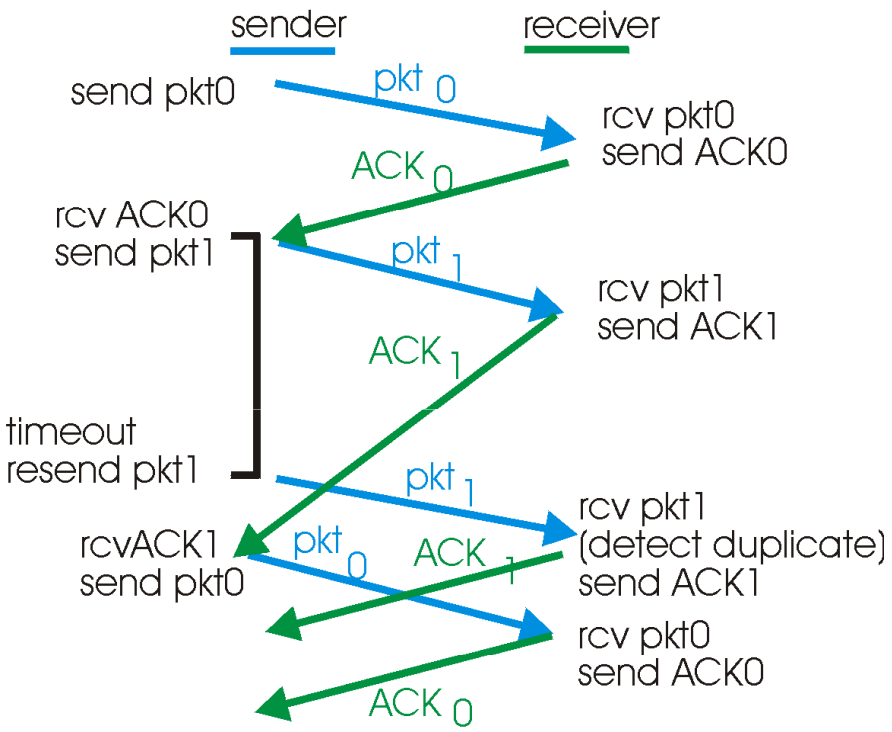


(b) lost packet

rdt3.0 in action



(c) lost ACK



(d) premature timeout

Performance of rdt3.0

rdt3.0 works, but performance stinks

ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

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

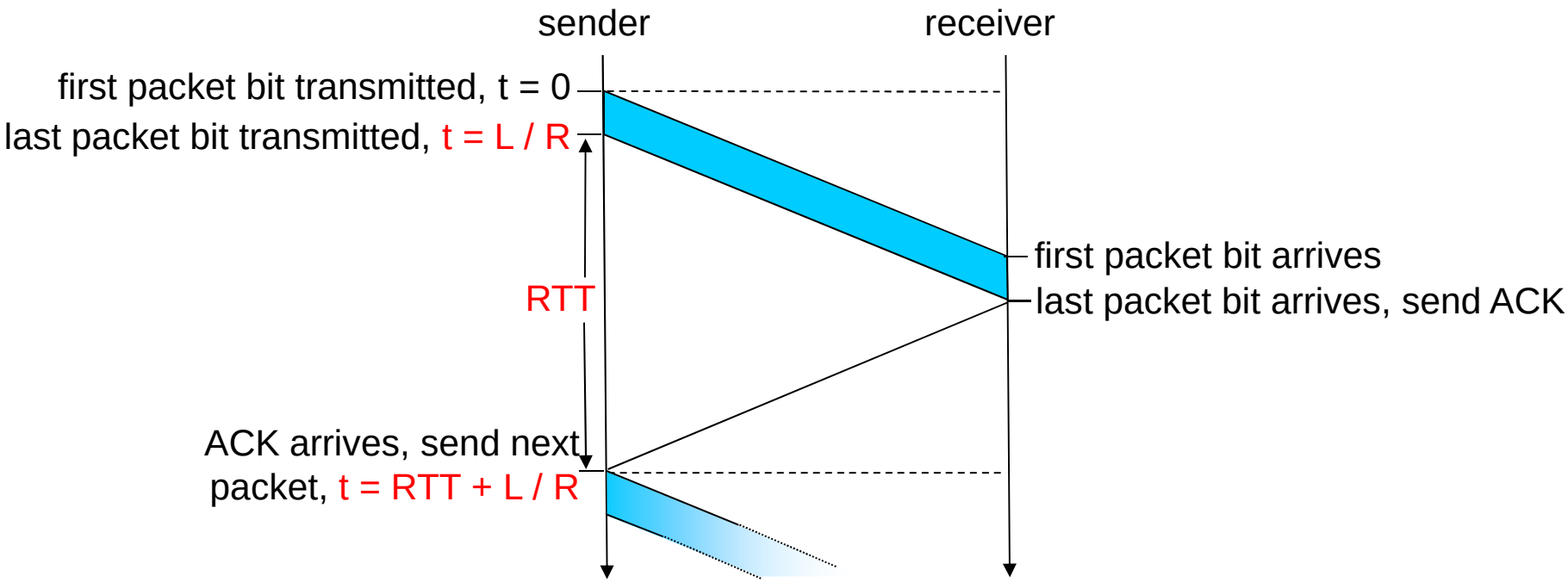
U_{sender}: utilization – fraction of time sender busy sending

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

1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link

network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

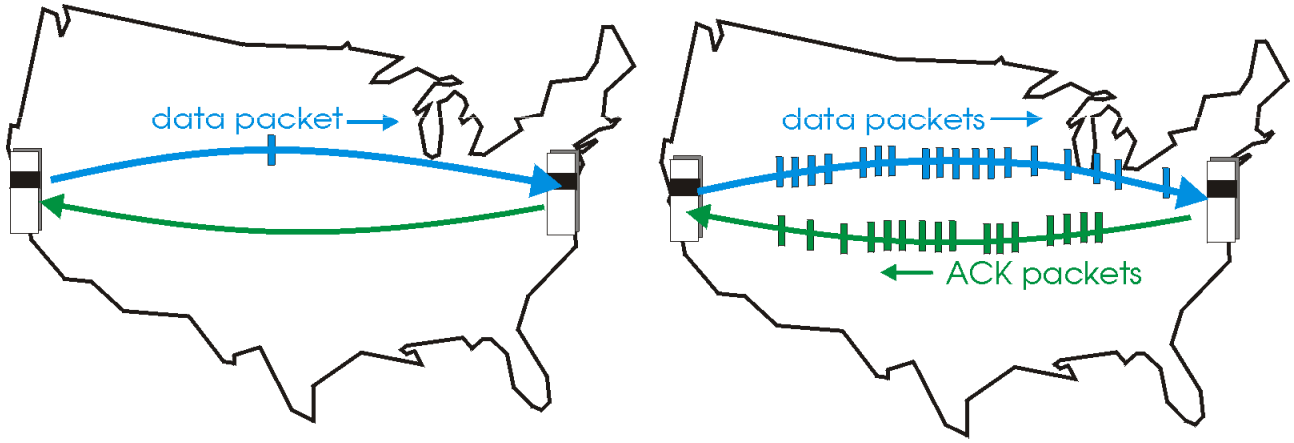


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

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

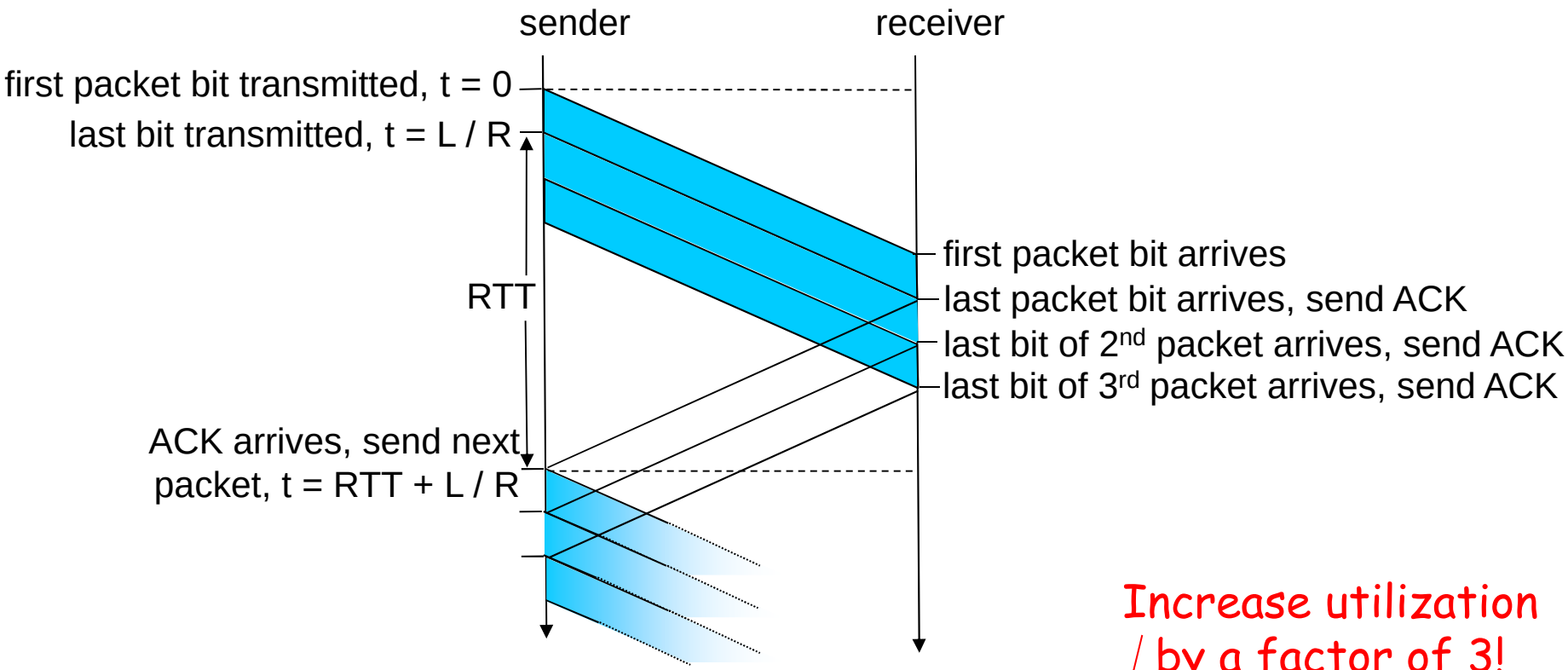


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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

Pipelining: increased utilization



Increase utilization by a factor of 3!

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

Pipelining Protocols

Go-back-N: 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

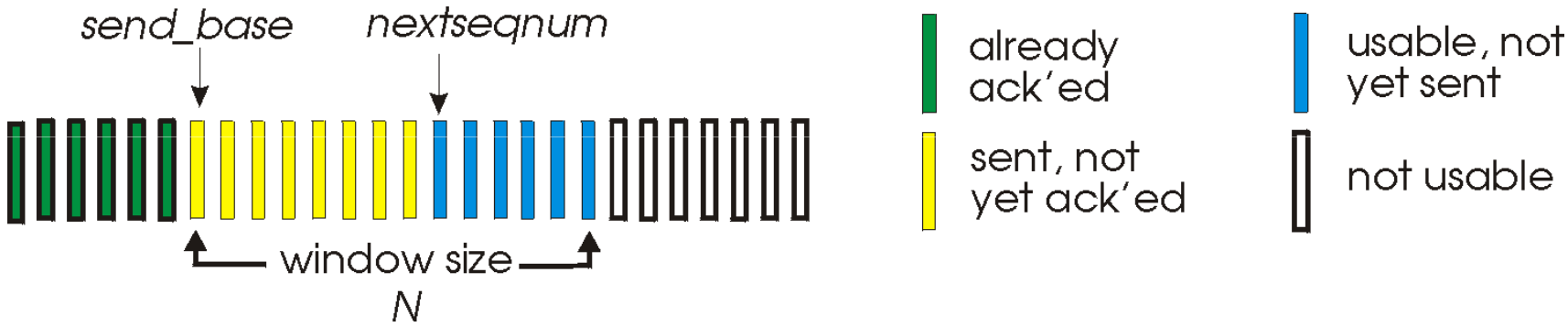
- Sender can have up to N unacked packets in pipeline
- Rcvr acks individual packets
- Sender maintains timer for each unacked packet
  - When timer expires, retransmit only unack packet

Go-Back-N

Sender:

k-bit seq # in pkt header

"window" of up to N, consecutive unack'ed pkts allowed



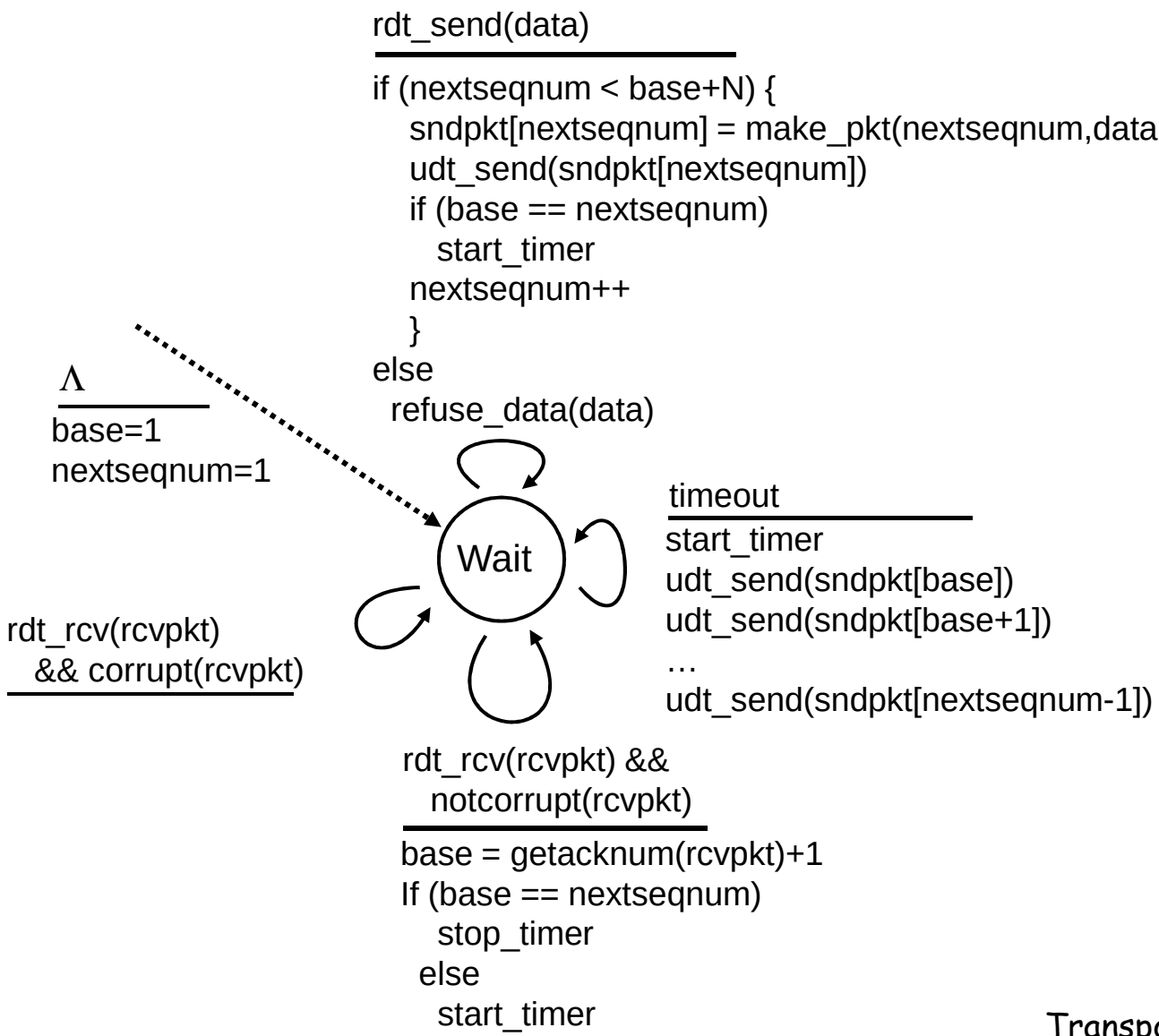
ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"

may receive duplicate ACKs (see receiver)

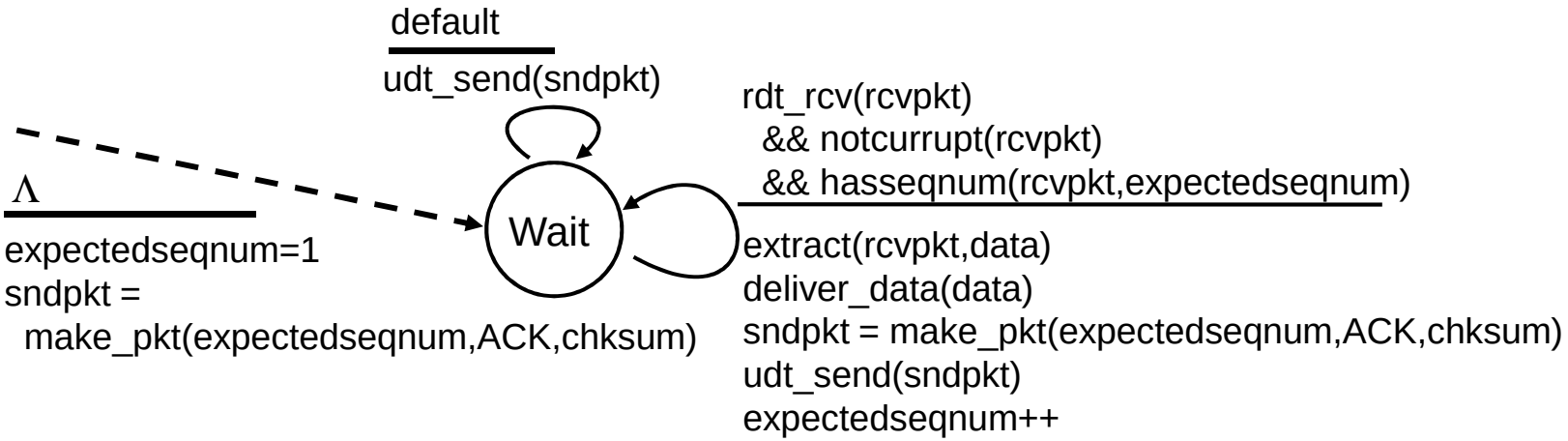
timer for each in-flight pkt

timeout(n): retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM



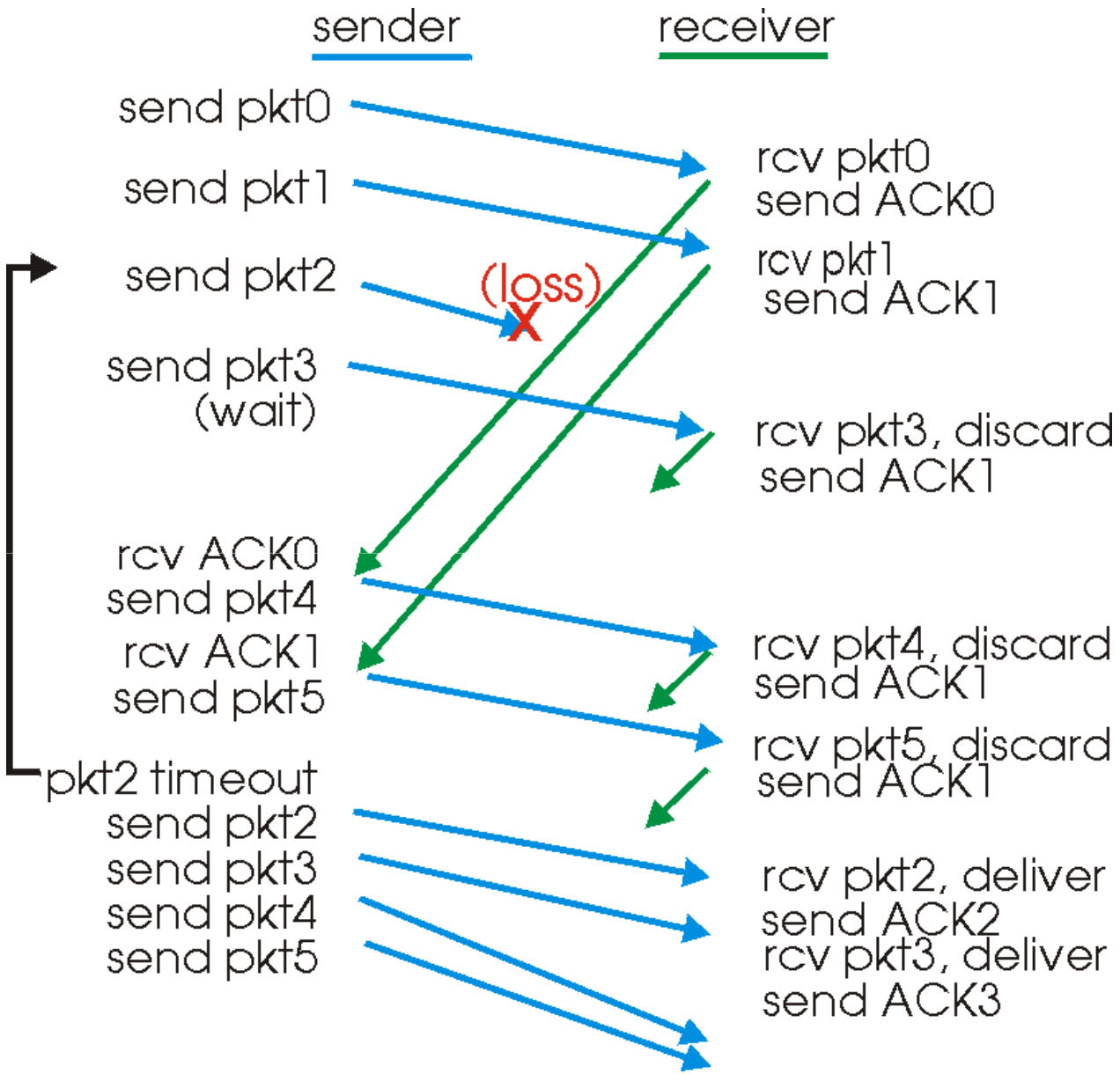
ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

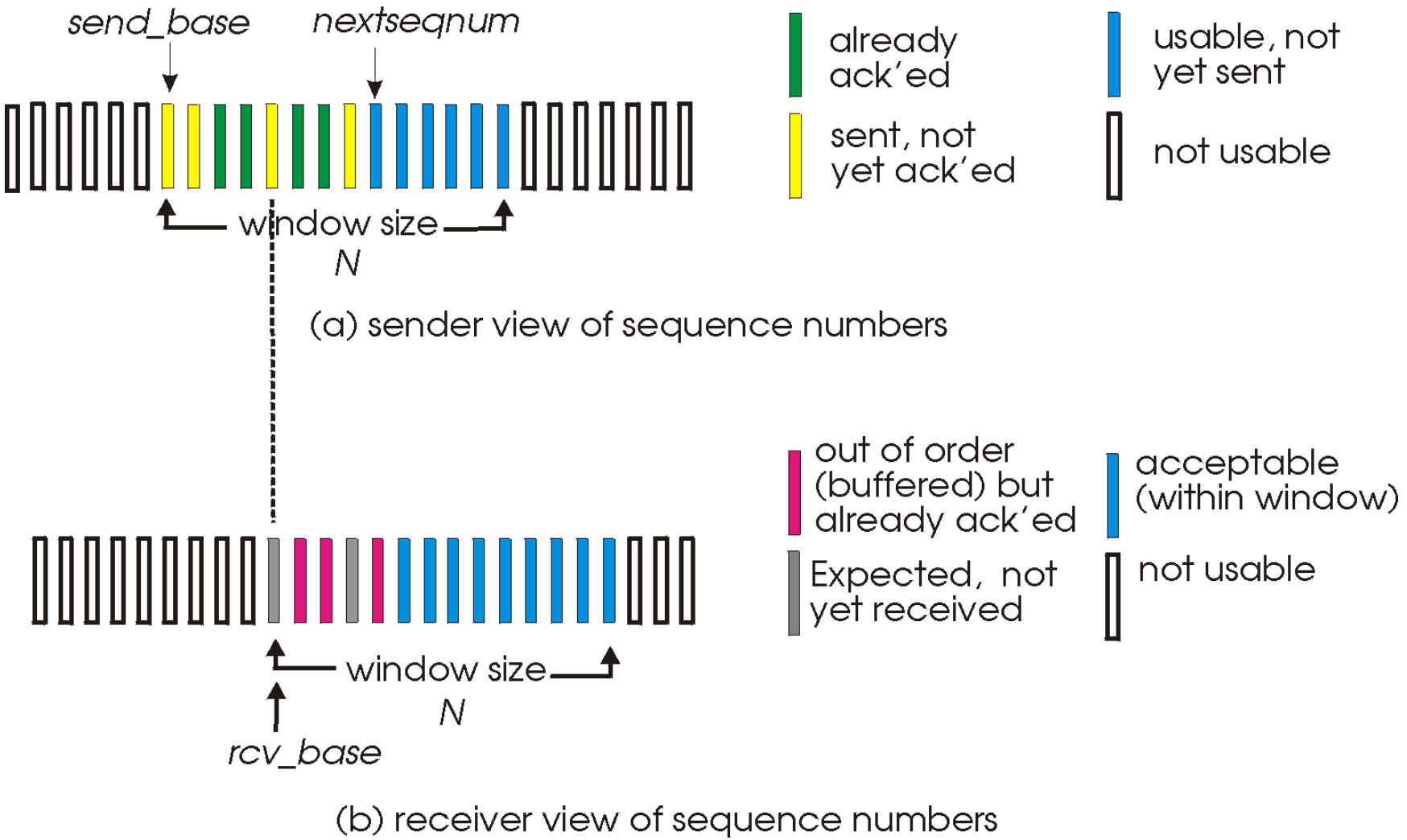
GBN in action



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



Selective repeat

sender

data from above :

if next available seq # in window, send pkt

timeout(n):

resend pkt n, restart timer

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

mark pkt n as received

if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

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

send ACK(n)

out-of-order: buffer

in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

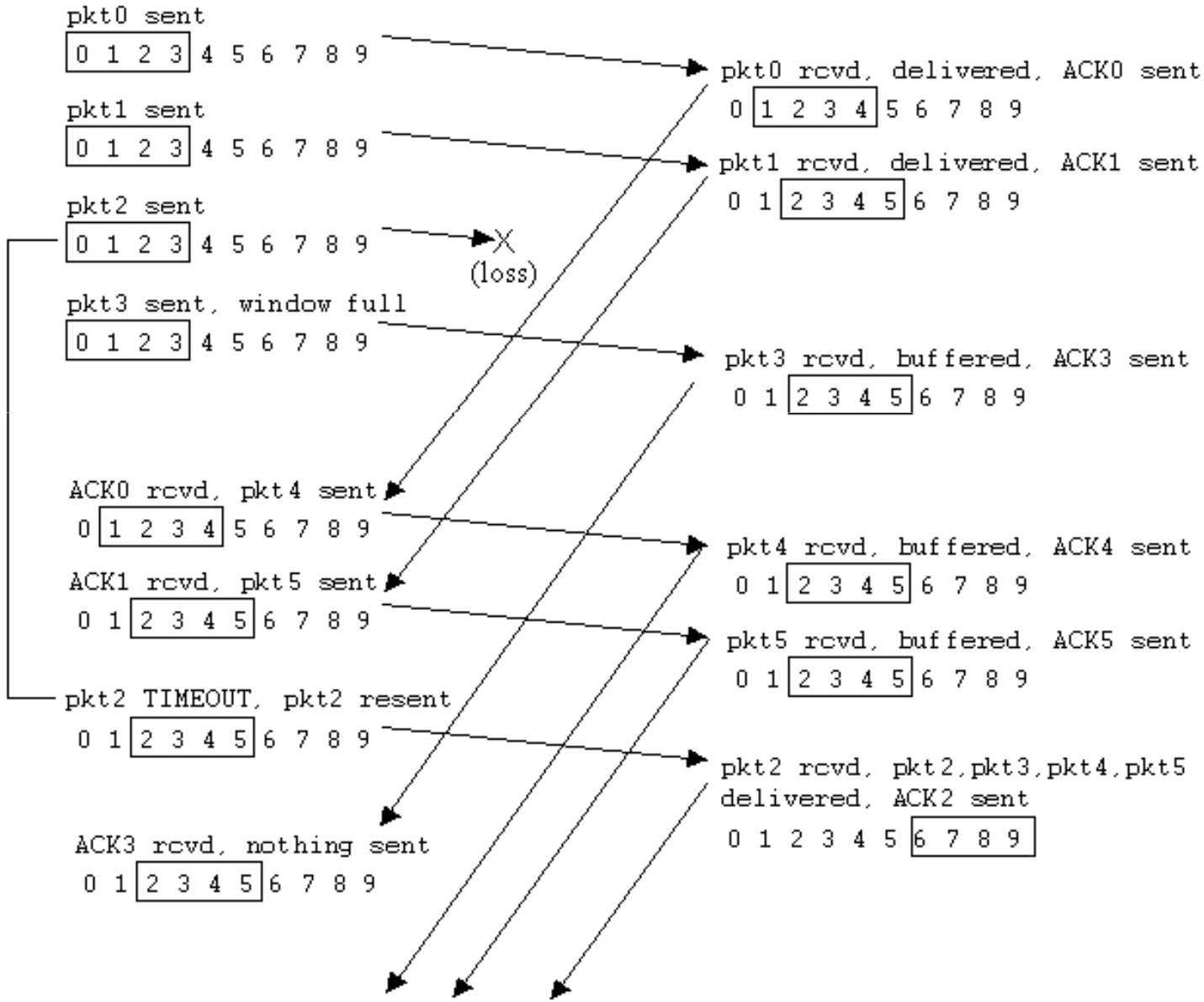
pkt n in [rcvbase-N,rcvbase-1]

ACK(n)

otherwise:

ignore

Selective repeat in action



Selective repeat: dilemma

Example:

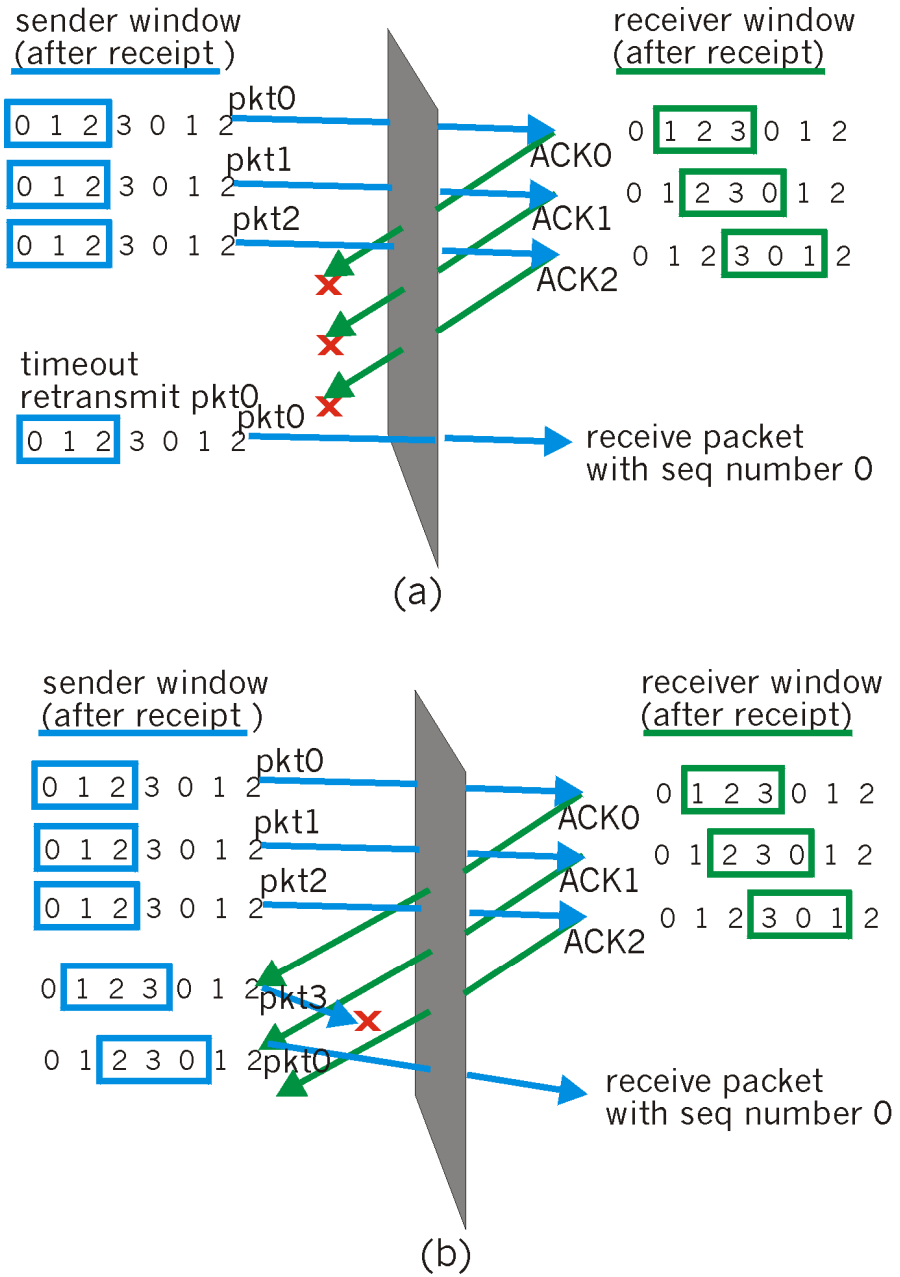
seq #'s: 0, 1, 2, 3

window size=3

receiver sees no difference in two scenarios!

incorrectly passes duplicate data as new in (a)

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



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TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

point-to-point:

one sender, one receiver

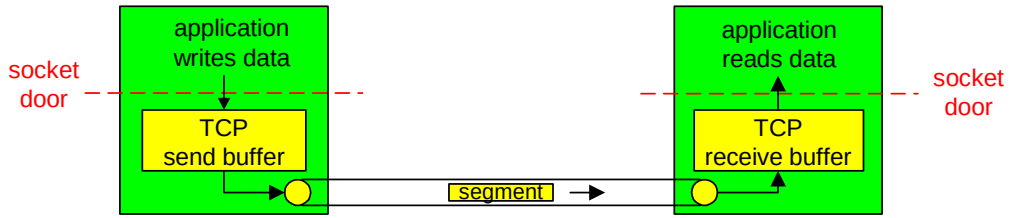
reliable, in-order *byte steam:*

no "message boundaries"

pipelined:

TCP congestion and flow control set window size

send & receive buffers



full duplex data:

bi-directional data flow in same connection

MSS: maximum segment size

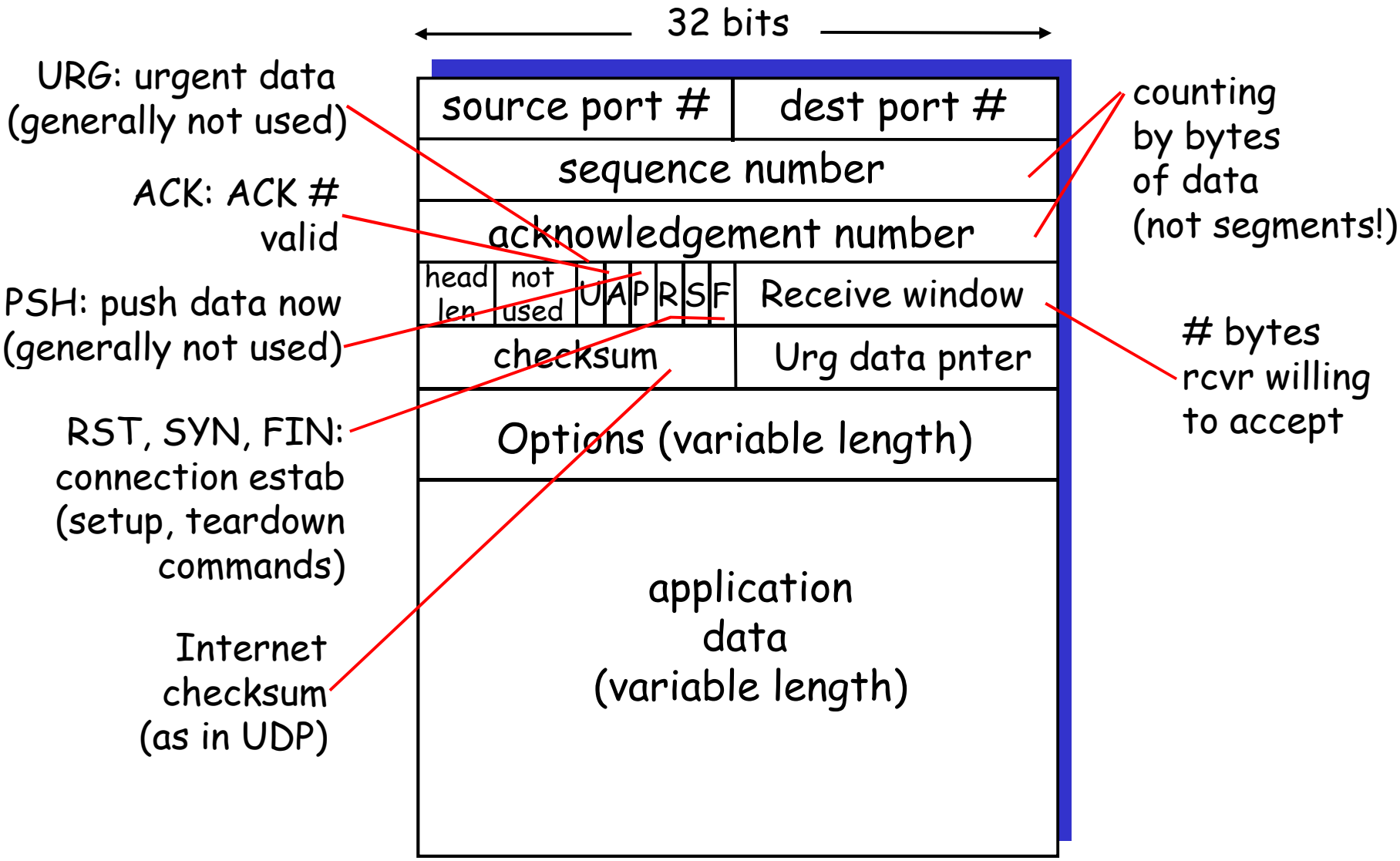
connection-oriented:

handshaking (exchange of control msgs) init's sender, receiver state before data exchange

flow controlled:

sender will not overwhelm receiver

TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

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

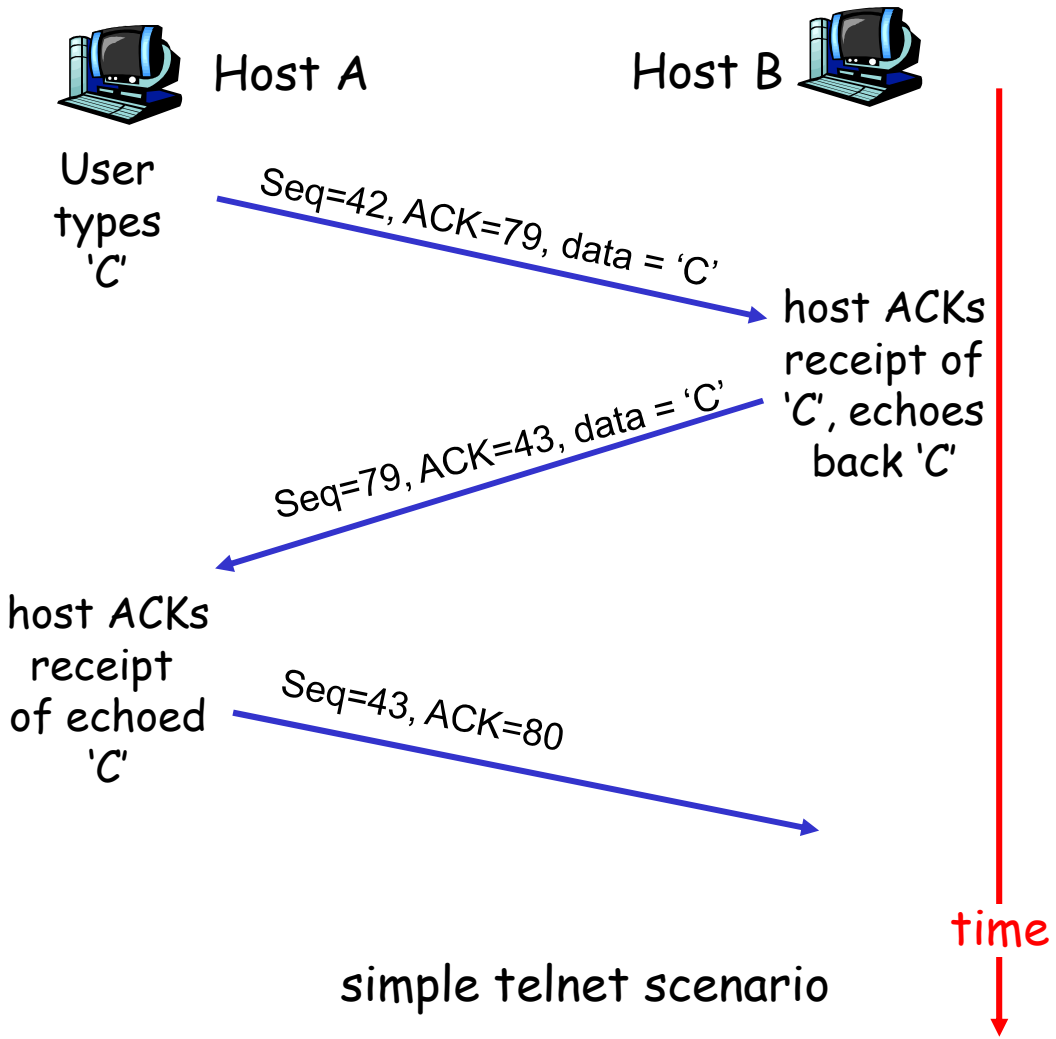
ACKs:

seq # of next byte expected from other side

cumulative ACK

Q: how receiver handles out-of-order segments

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



simple telnet scenario

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

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

Q: how to estimate RTT?

SampleRTT: measured time from segment transmission until ACK receipt

- ignore retransmissions

SampleRTT will vary, want estimated RTT "smoother"

- average several recent measurements, not just current **SampleRTT**

TCP Round Trip Time and Timeout

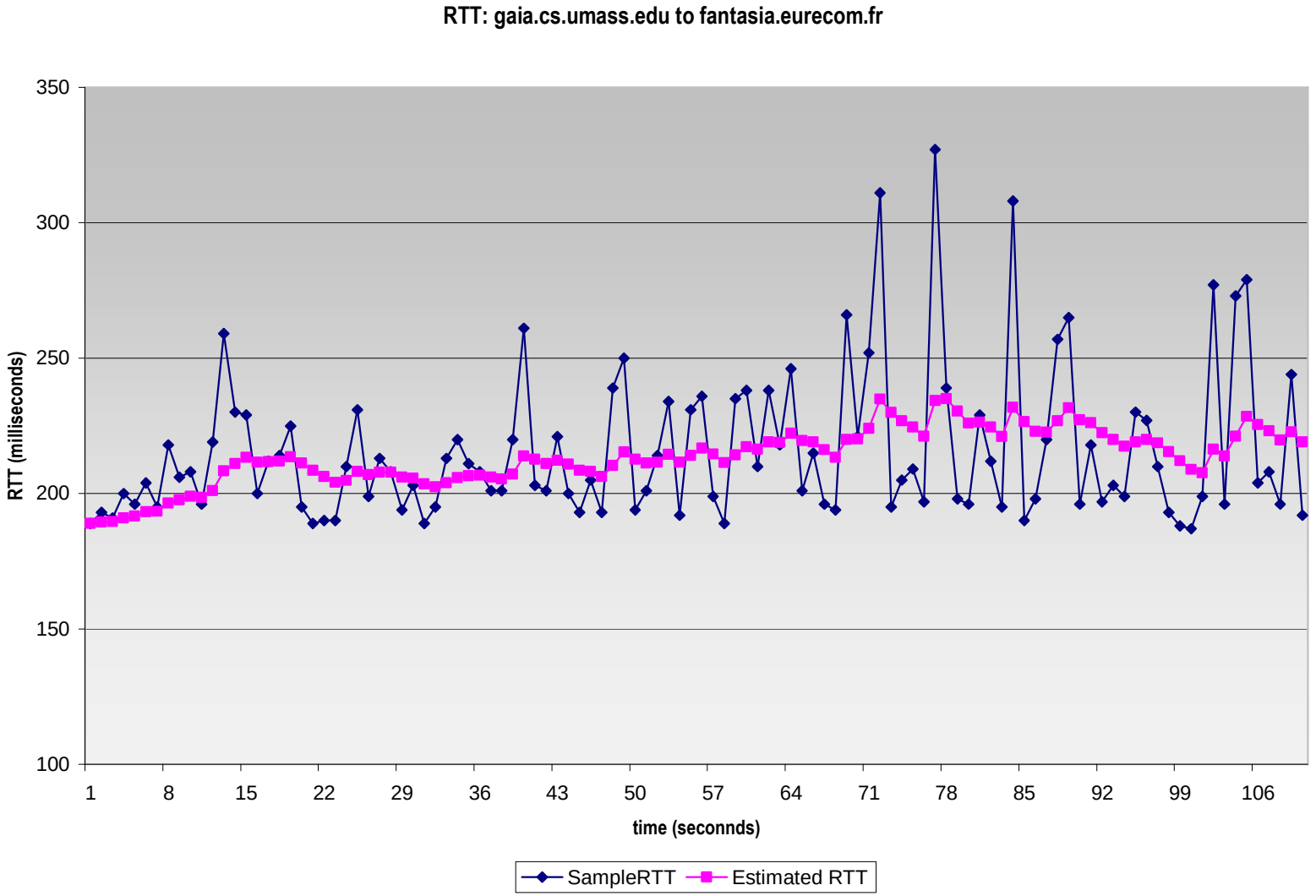
EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

Exponential weighted moving average

influence of past sample decreases exponentially fast

typical value: α = 0.125

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

EstimtedRTT plus "safety margin"

large variation in **EstimatedRTT** -> larger safety margin

first estimate of how much SampleRTT deviates from EstimatedRTT:

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

(typically, β = 0.25)

Then set timeout interval:

TimeoutInterval = EstimatedRTT + 4*DevRTT

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TCP reliable data transfer

TCP creates rdt service on top of IP's unreliable service

Pipelined segments

Cumulative acks

TCP uses single retransmission timer

Retransmissions are triggered by:

- timeout events
- duplicate acks

Initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

data rcvd from app:

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

timeout:

- retransmit segment that caused timeout
- restart timer

Ack rcvd:

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

```
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:

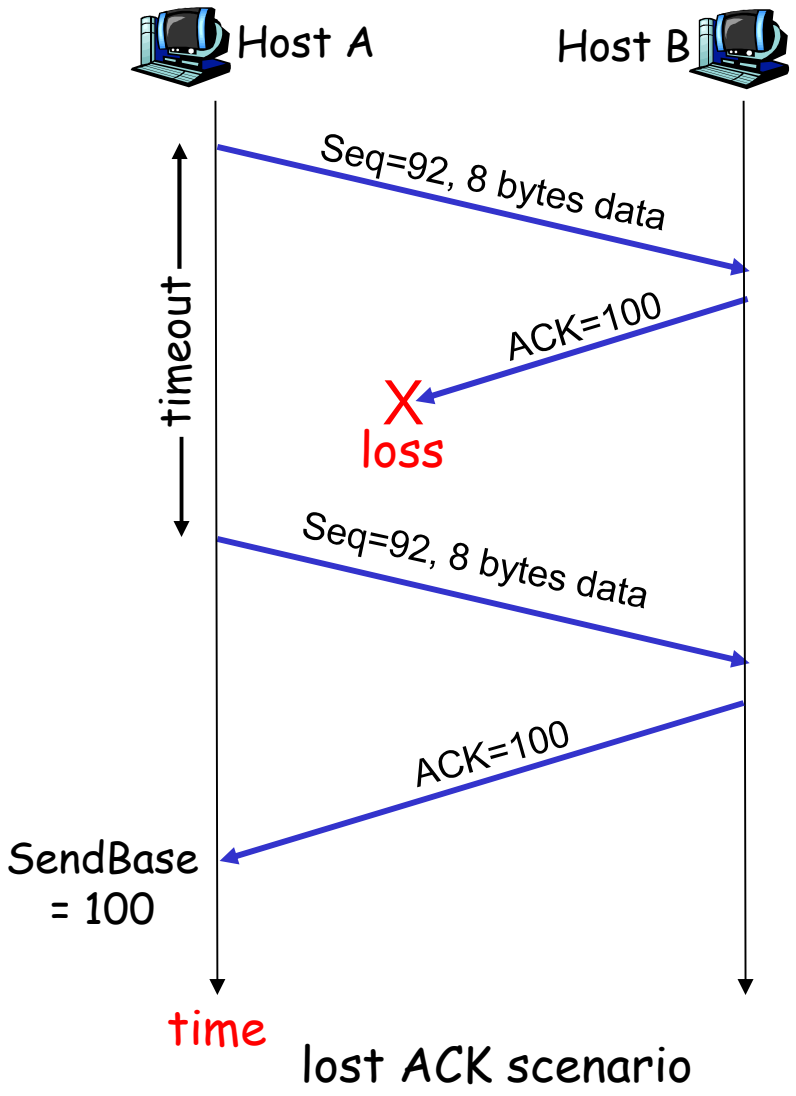
- SendBase-1: last cumulatively ack'ed byte

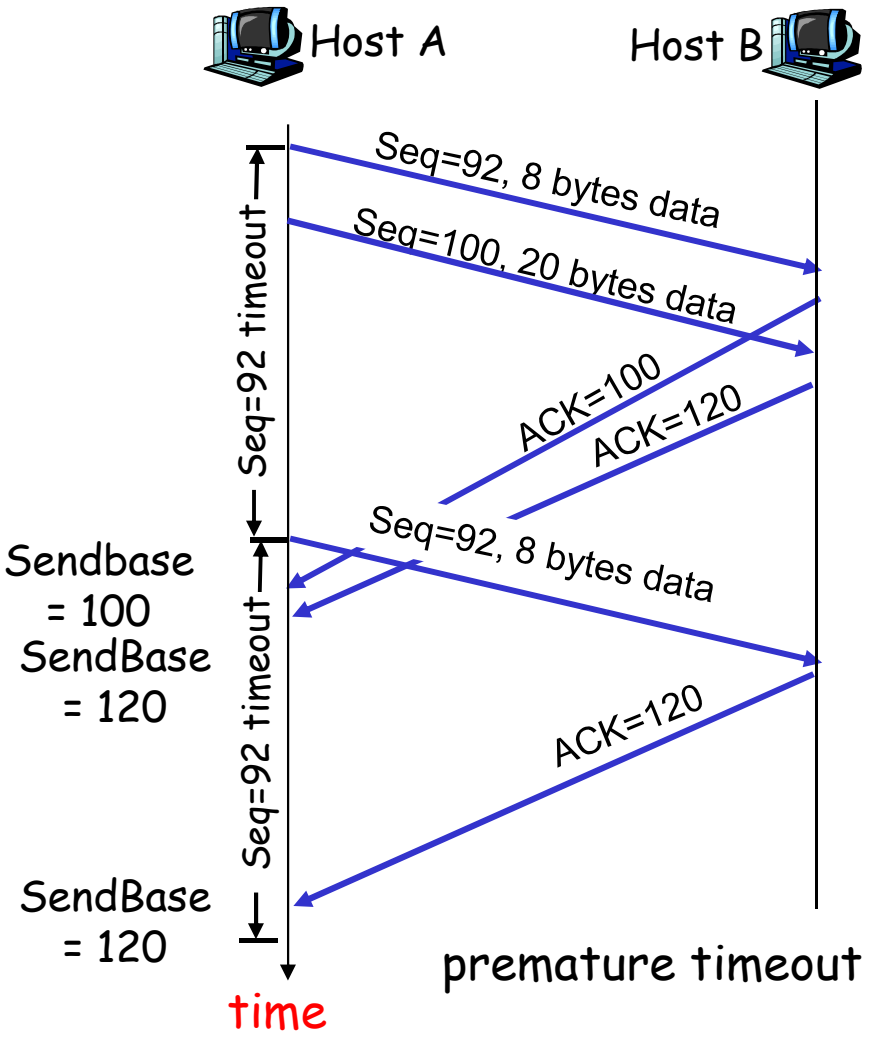
Example:

- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ;

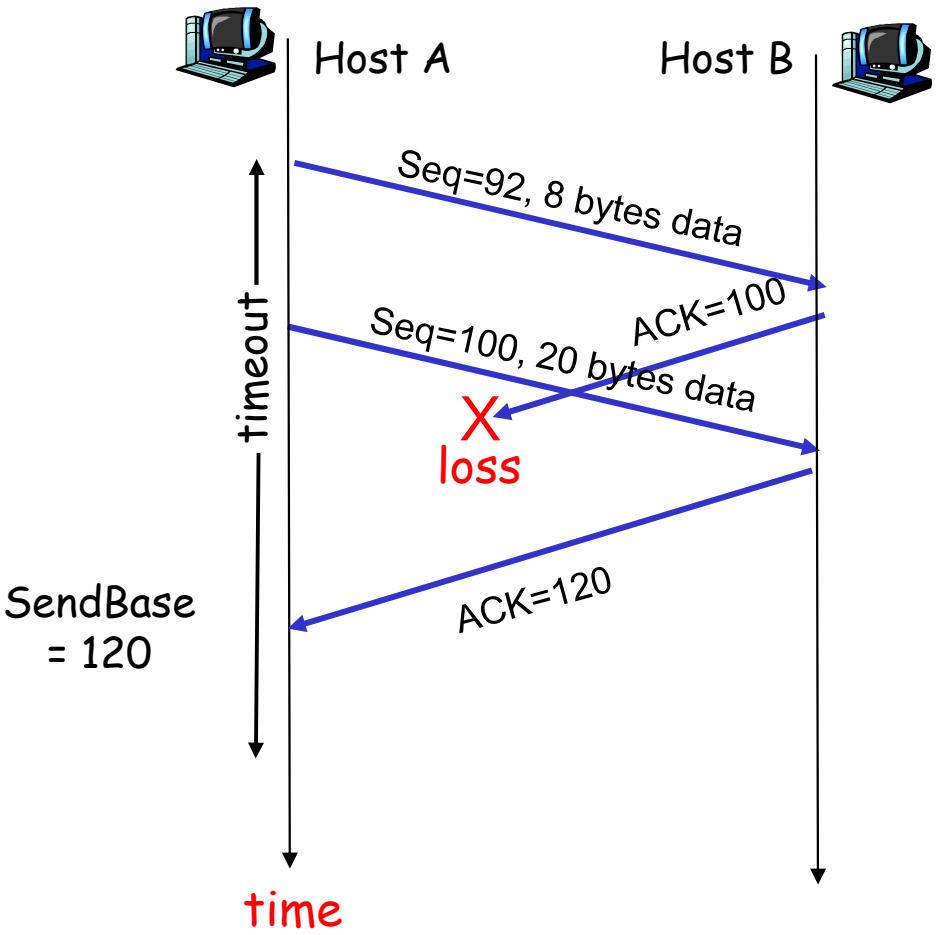
y > SendBase, so that new data is acked

TCP: retransmission scenarios





TCP retransmission scenarios (more)



Cumulative ACK scenario

TCP ACK generation [RFC 1122, RFC 2581]

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

Fast Retransmit

Time-out period often relatively long:

- long delay before resending lost packet

Detect lost segments via duplicate ACKs.

- Sender often sends many segments back-to-back
- If segment is lost, there will likely be many duplicate ACKs.

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

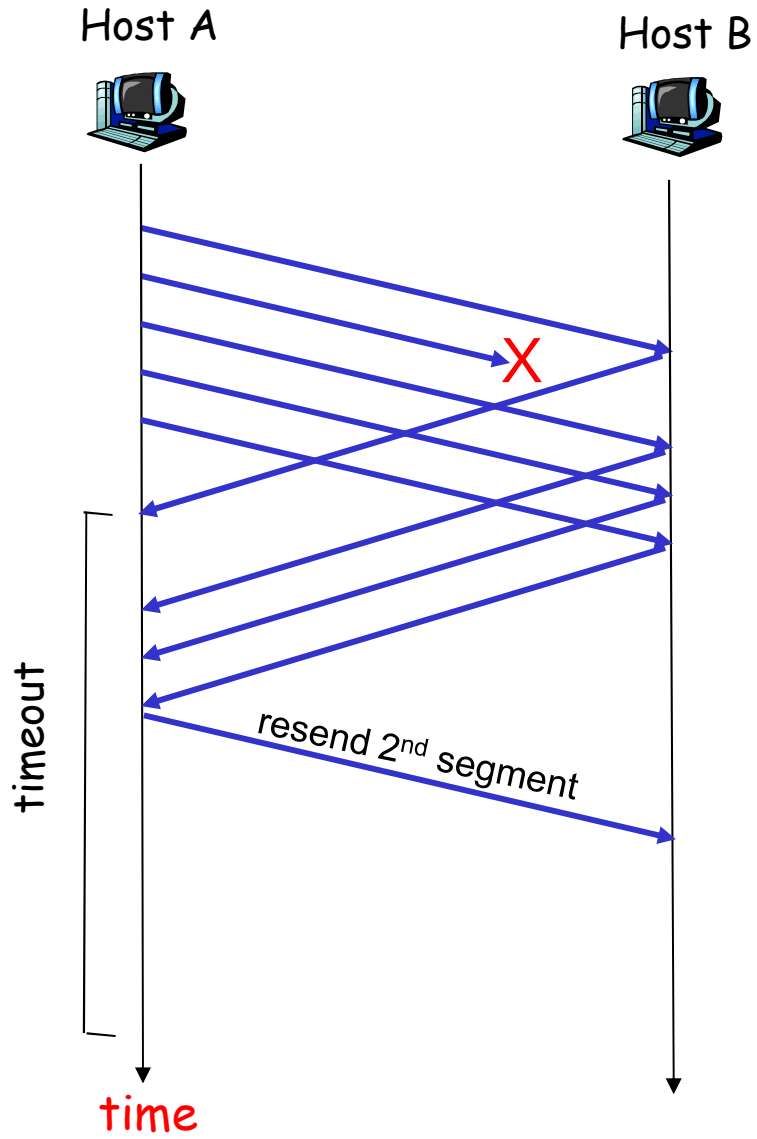


Figure 3.37 Resending a segment after triple duplicate ACK

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

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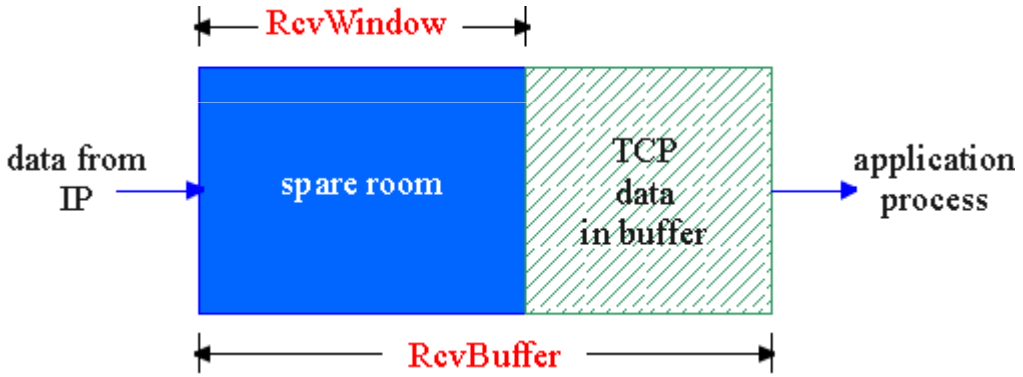
TCP Flow Control

receive side of TCP connection has a receive buffer:



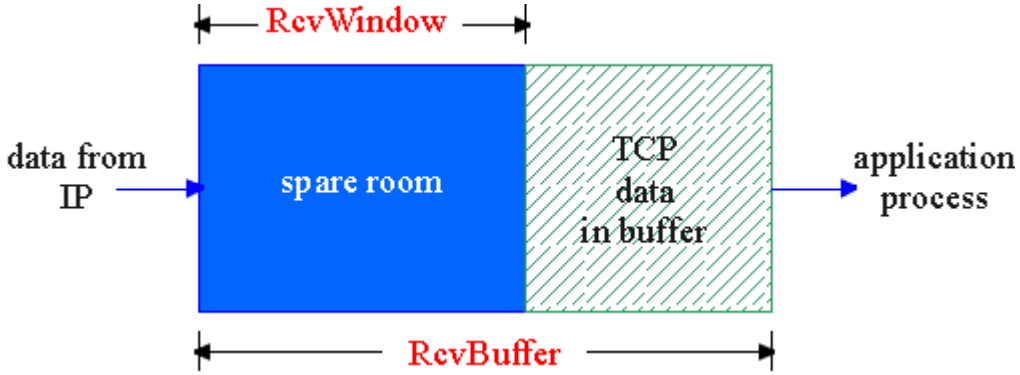
app process may be slow at reading from buffer

flow control

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

speed-matching service: matching 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

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TCP Connection Management

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

initialize TCP variables:

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

client: connection initiator

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

server: contacted by client

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

Three way handshake:

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

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

- server allocates buffers
- specifies server initial seq. #

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

TCP Connection Management (cont.)

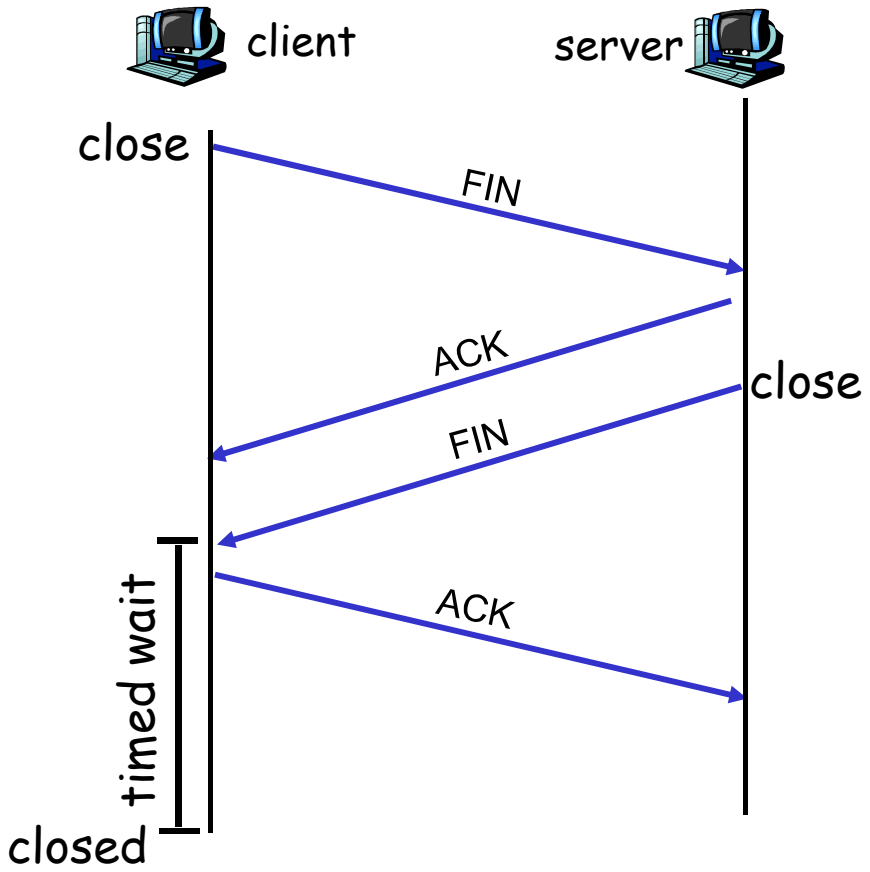
Closing a connection:

client closes socket:

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

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

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



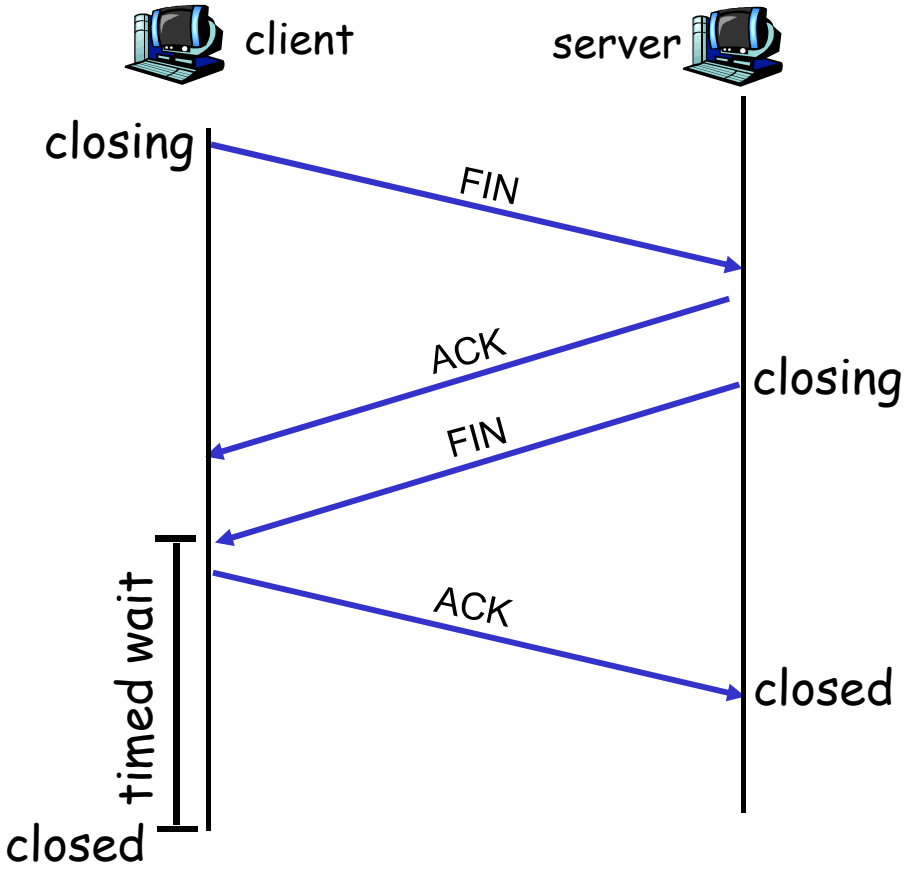
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

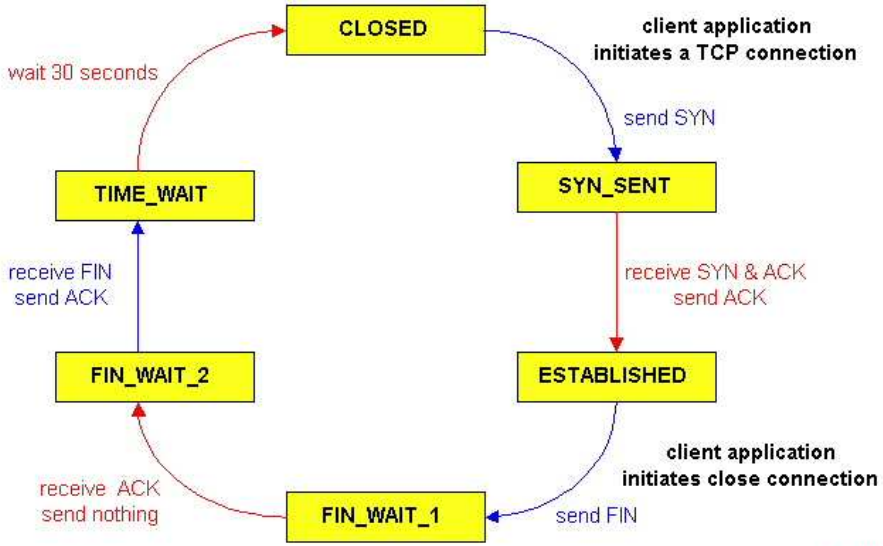
Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

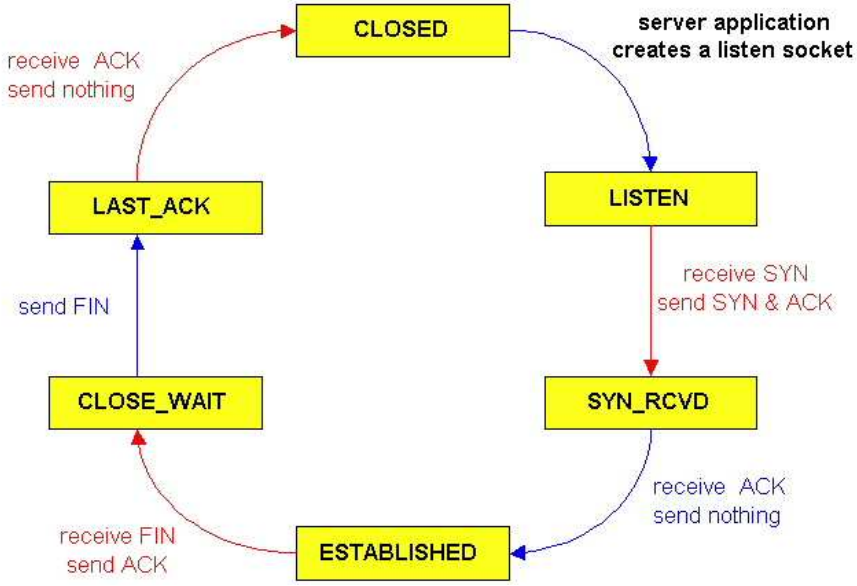


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



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Principles of Congestion Control

Congestion:

informally: "too many sources sending too much data too fast for *network* to handle"

different from flow control!

manifestations:

- lost packets (buffer overflow at routers)
- long delays (queueing in router buffers)

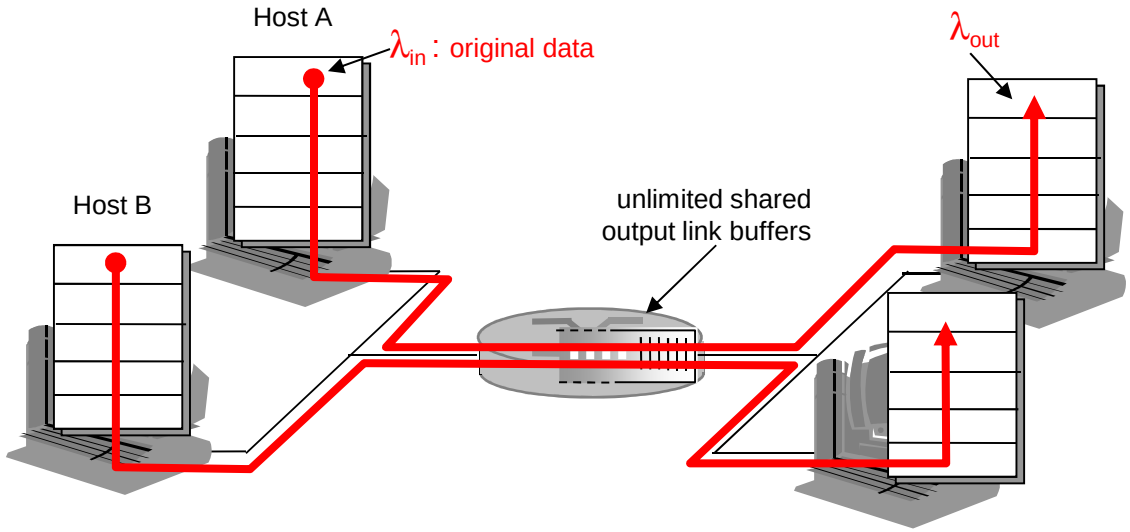
a top-10 problem!

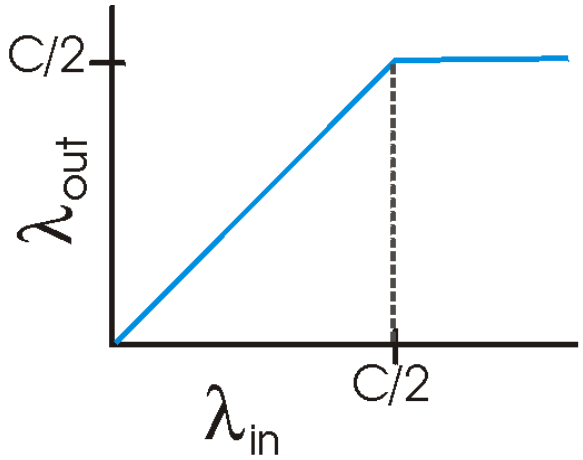
Causes/costs of congestion: scenario 1

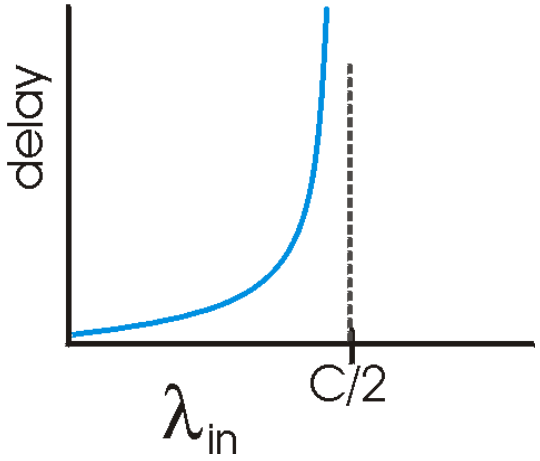
two senders, two receivers

one router, infinite buffers

no retransmission







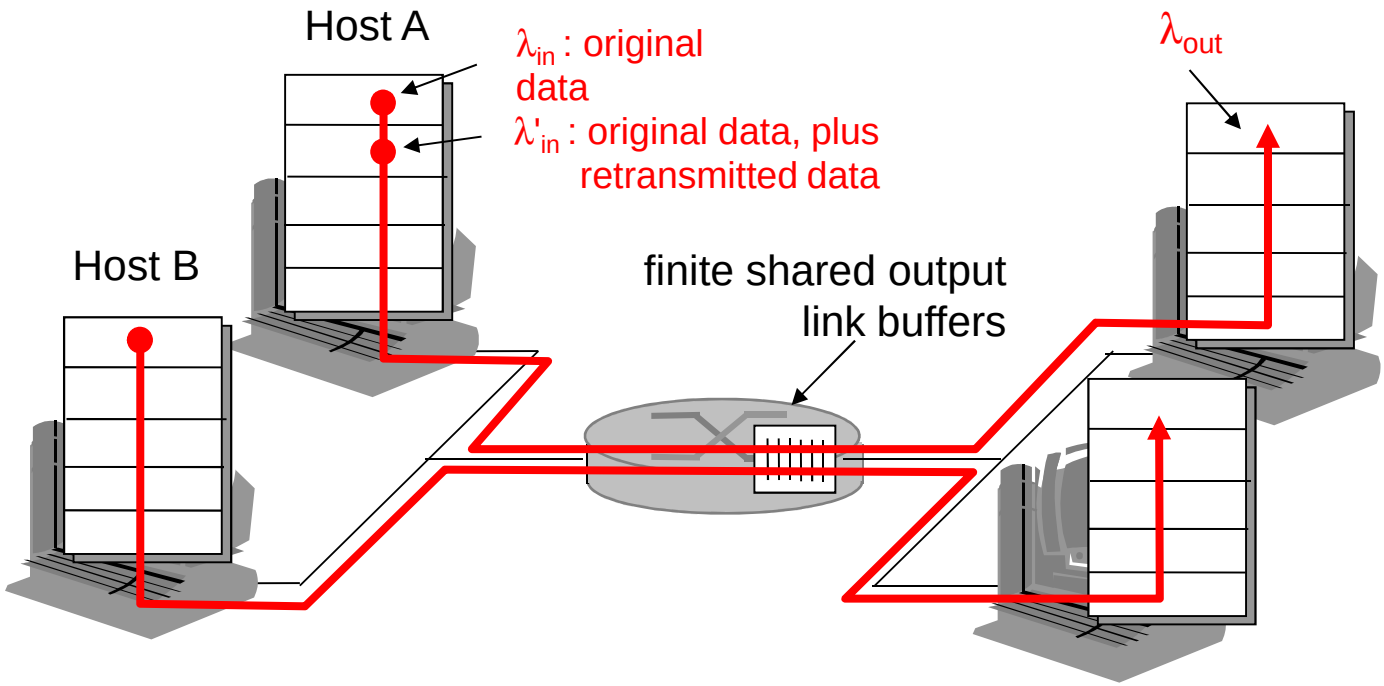
large delays when congested

maximum achievable throughput

Causes/costs of congestion: scenario 2

one router, *finite* buffers

sender retransmission of lost packet

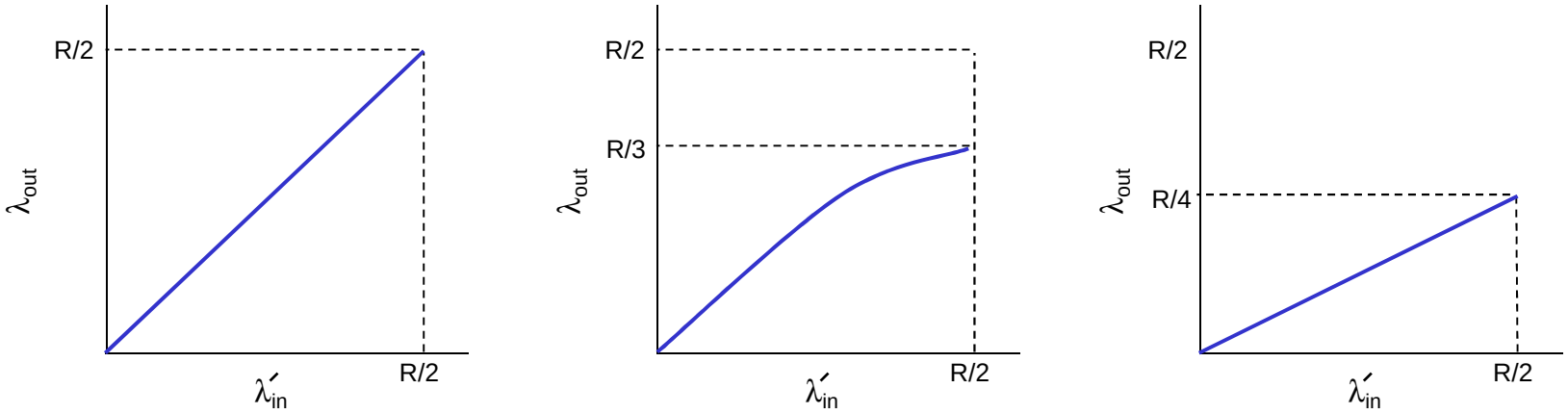


Causes/costs of congestion: scenario 2

always: $\lambda_{in} = \lambda_{out}$ (goodput)

"perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$

retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



"costs" of congestion:

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

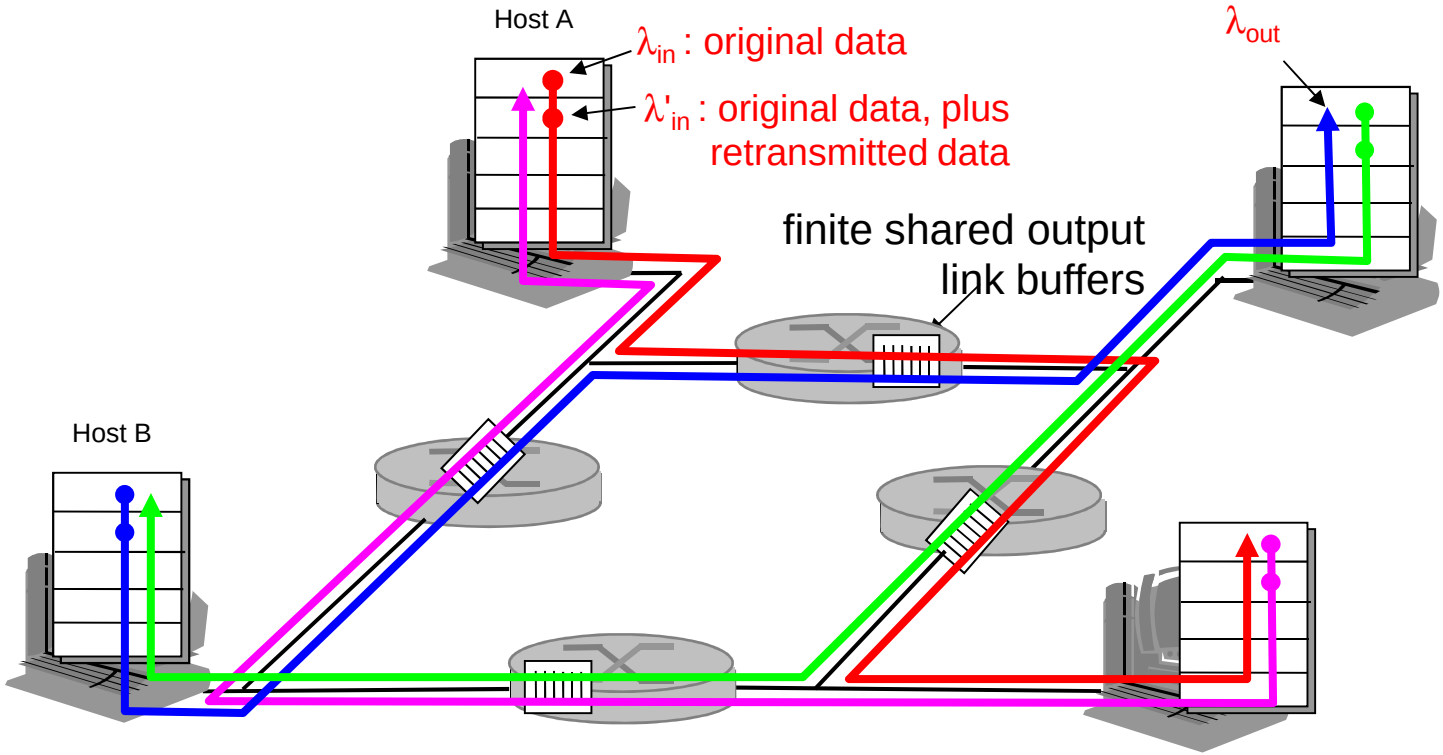
Causes/costs of congestion: scenario 3

four senders

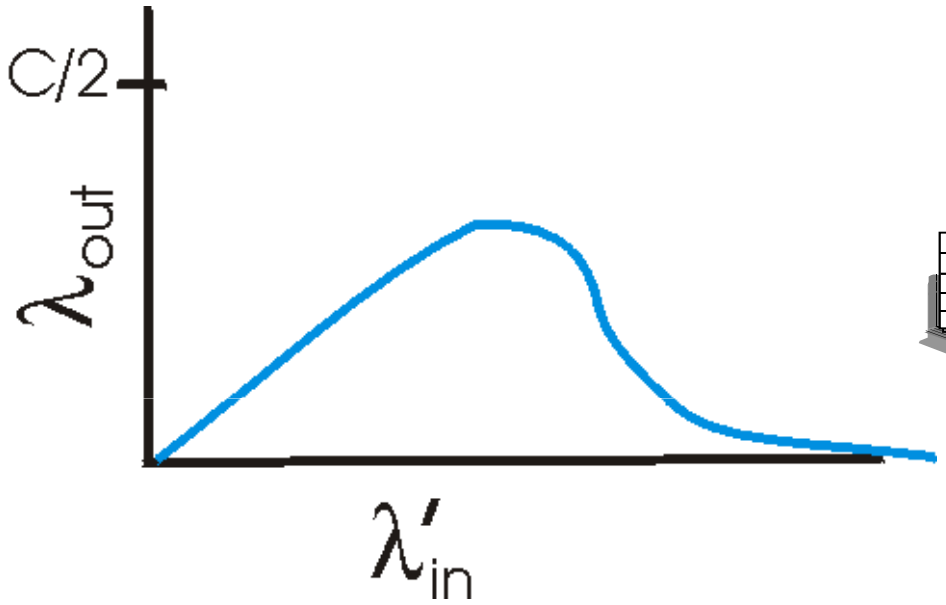
multihop paths

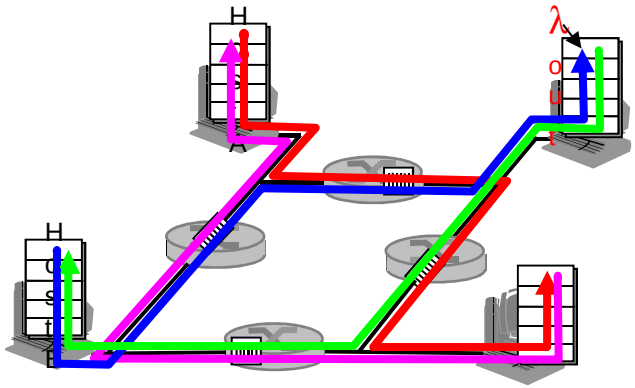
timeout/retransmit

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



Causes/costs of congestion: scenario 3





Another "cost" of congestion:

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

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

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

Network-assisted congestion control:

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

Case study: ATM ABR congestion control

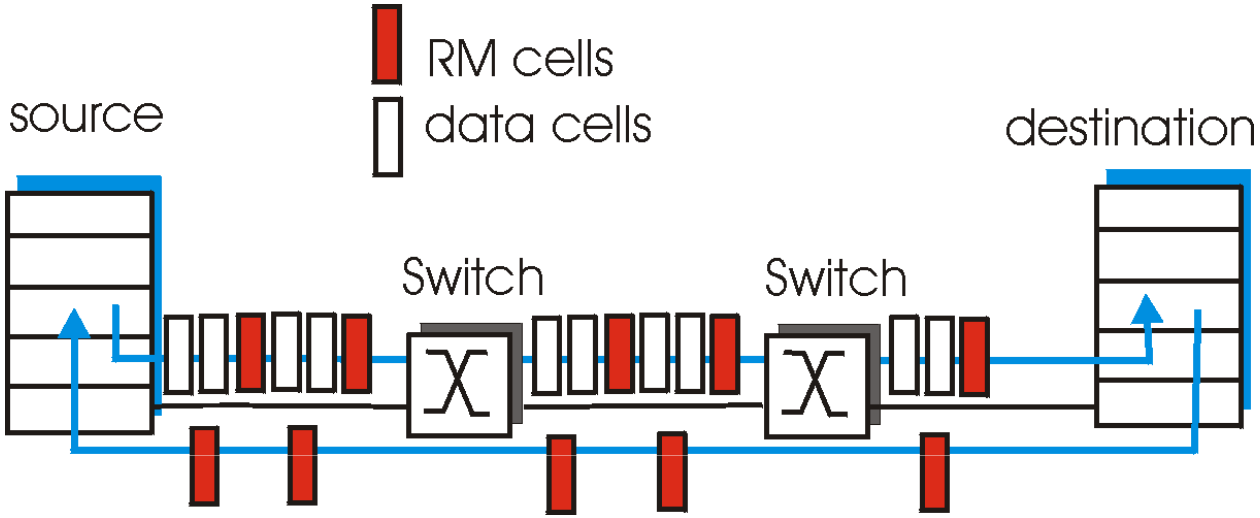
ABR: available bit rate:

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

RM (resource management) cells:

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

Case study: ATM ABR congestion control



two-byte ER (explicit rate) field in RM cell

- congested switch may lower ER value in cell
- sender' send rate thus maximum supportable rate on path

EFCI bit in data cells: set to 1 in congested switch

- if data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell

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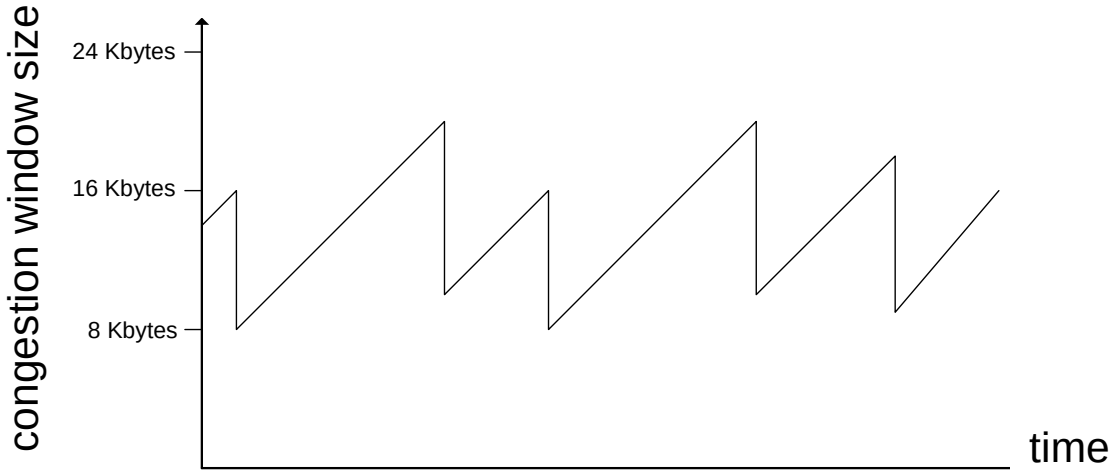
TCP congestion control: additive increase, multiplicative decrease

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:

LastByteSent-LastByteAcked

$\leq$ **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 >> MSS/RTT

- desirable to quickly ramp up to respectable rate

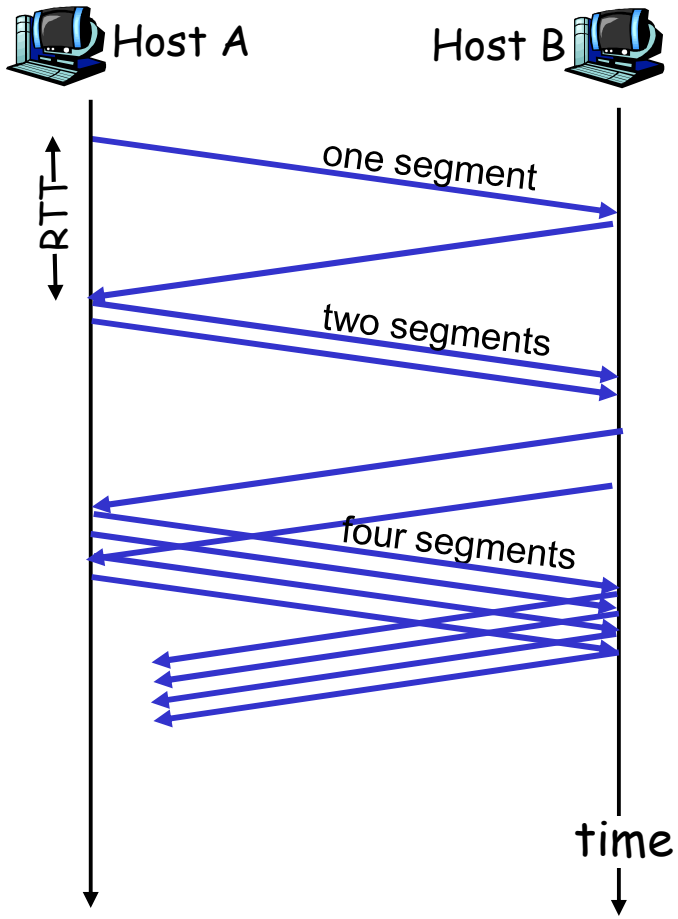
When connection begins, increase rate exponentially fast until first loss event

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

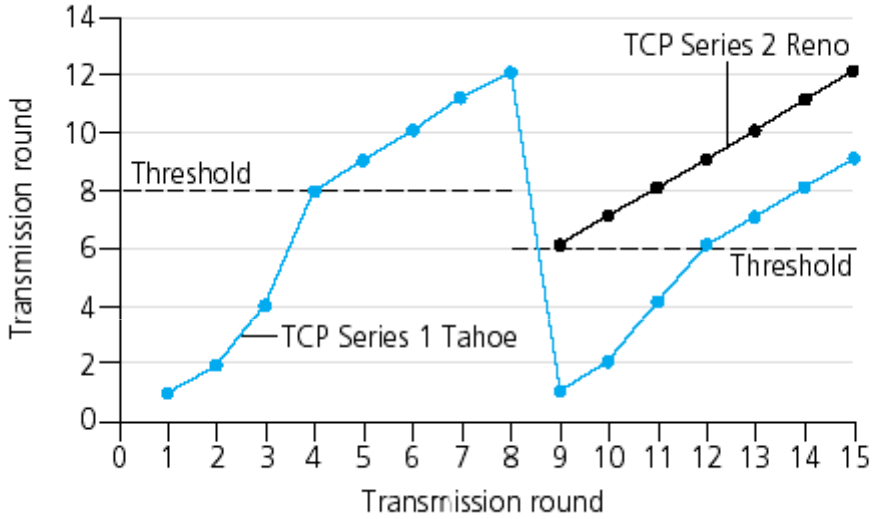
Philosophy:

- 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 throughout 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 throughout: .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:

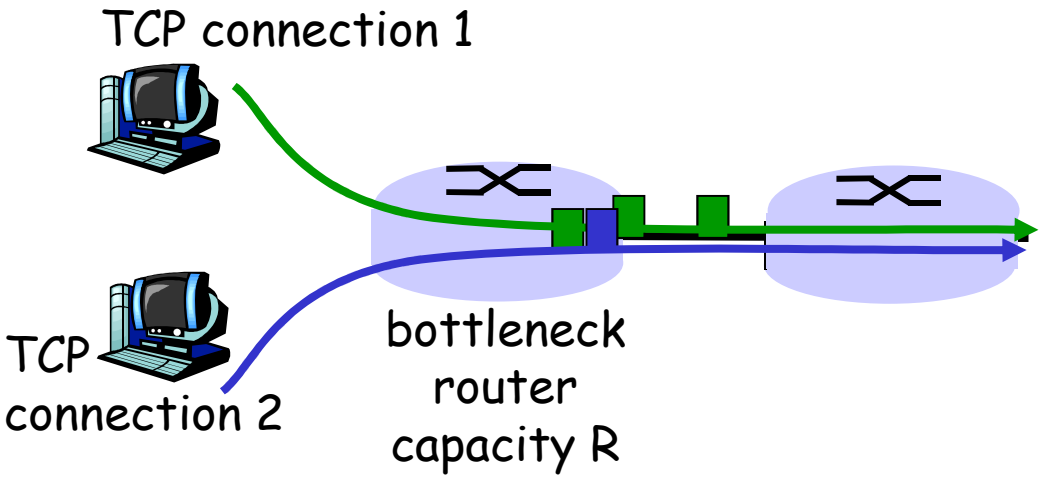
$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

➜ 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

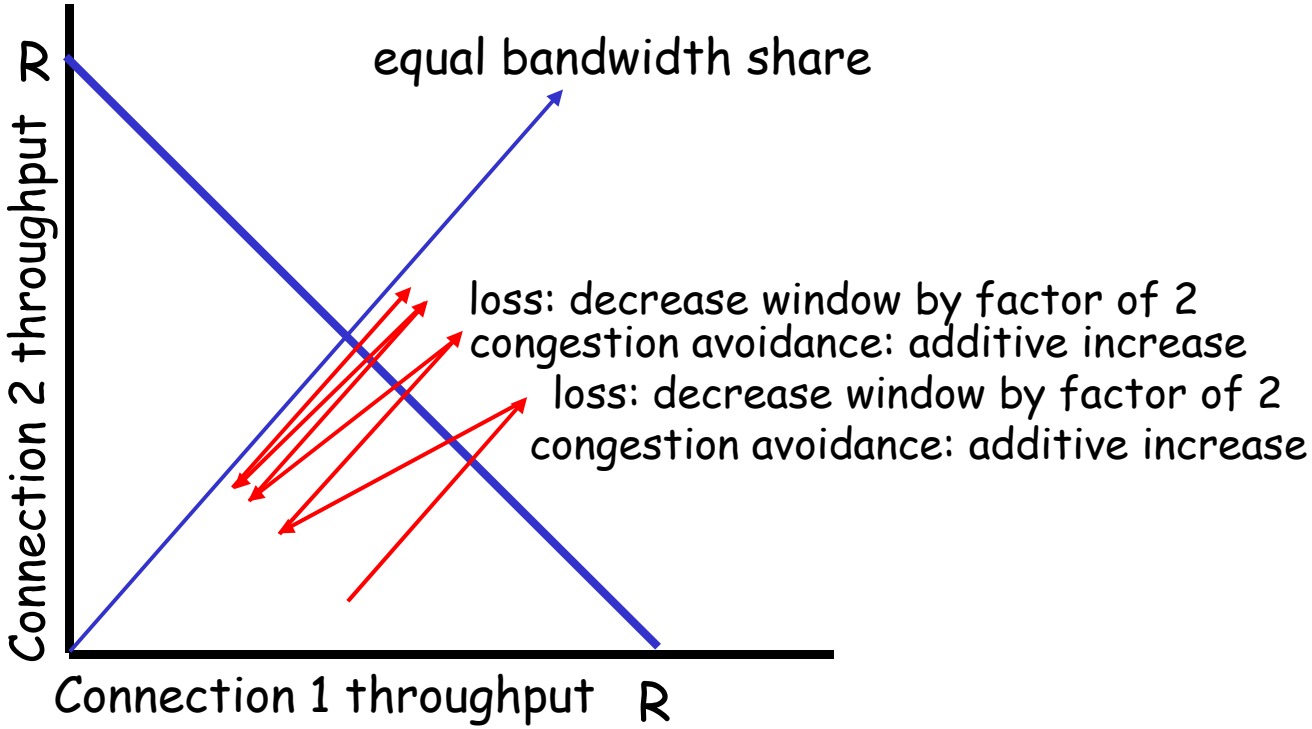


Why is TCP fair?

Two competing sessions:

Additive increase gives slope of 1, as throughout increases

multiplicative decrease decreases throughput proportionally



Fairness (more)

Fairness and UDP

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