Electrical Engineering E6761
Computer Communication Networks
Lecture 3
Transport Layer Services:
reliability, connection setup, flow control

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Tues 4:10-6:40, Mudd 1127
Course URL:
http://www.cs.columbia.edu/~danr/EE6761

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Today

- PA#2 – due 10/3
- HW#0 – solutions on-line (see week2 on materials pg)
- HW#1 – changes, questions?
- Java for PA#2: Yes – all Java-related questions to Vasillis, please
- Transport Layer
  - e2e argument
  - multiplexing / demultiplexing
  - reliability
  - connection setup / teardown
  - flow control
  - Example protocol: TCP
  - congestion control… next time

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Policy Refresh…

- Collaboration is O.K. on
  - Homework
  - Programming Assignments
  - Project
- How much help should you get?
  - so that next time, you could do similar types of problems on your own
- How much help should you give?
  - enough to get the person moving again
- Who should you help?
  - anybody who asks you

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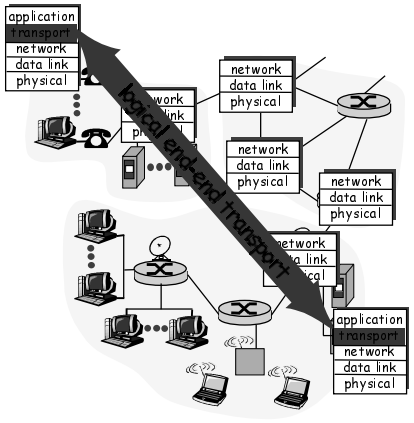
Transport services and protocols

- provide *logical communication* between apps' processes running on different hosts
- transport protocols
  - run in end systems
  - transfer info between **processes**
- *runs on top of the network layer, which :*
  - transfers info between network components
  - can delay, reorder or drop packets



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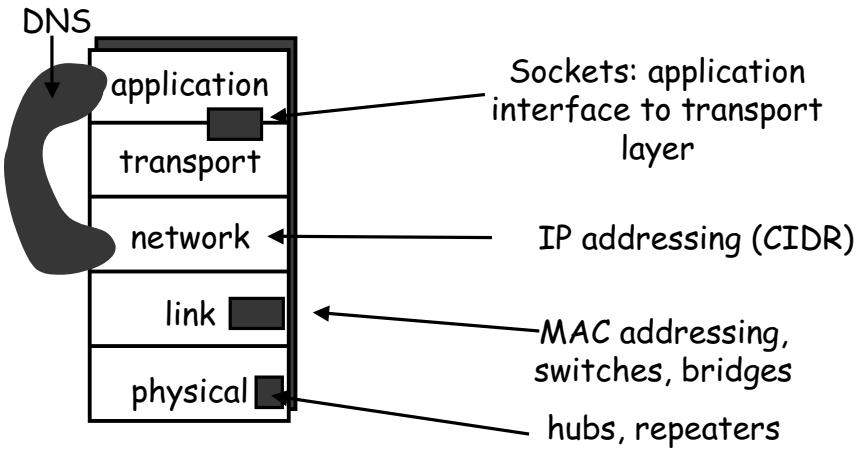
What we've seen so far (layered perspective)…



Today: part1 of transport layer details

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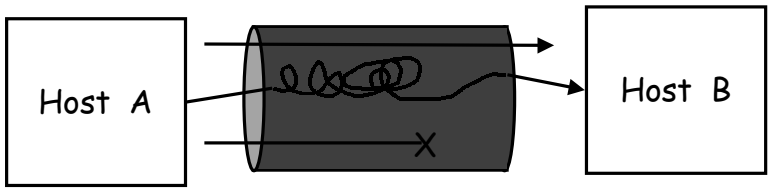
Transport Layer "view" of the network

- A "pipe" connects every pair of hosts
- packets sent into the pipe might
  - come out the other end quickly
  - come out the other end eventually
  - disappear



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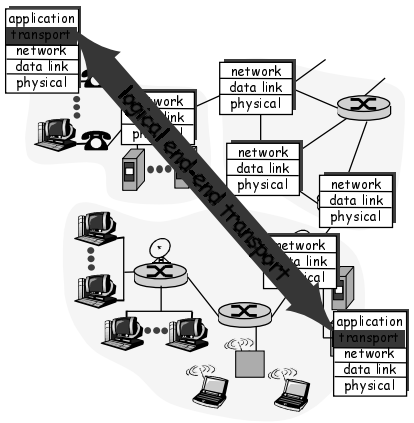
Transport-layer protocols

Internet transport services:

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



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e2e argument [see Saltzer, Reed, Clark article]

- Philosophy behind Intenet design: Move complex operations to the edges of the network
  - Why? Not all apps may require complex ops, e.g.,
    - reliability (audio, video)
    - security
  - Also, some functionality difficult to implement in network core
    - duplicates suppression
    - FIFO ordering
  - ops often repeated at edge anyways as safety check
- Implications of e2e argument to the Internet:

  most complex ops should be performed toward the top of the protocol stack

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e2e argument pros and cons

- Pros:
  - reduces network complexity – eases deployment, network recovery
  - reduces redundant checks since app often provides checks anyways
  - bugs harder to fix
- Cons:
  - network less efficient (e.g., hop-to-hop reliability would reduce b/w reqmts and delivery delays)
  - more responsibility lies with the application
    - longer development cycle, frequent bugs

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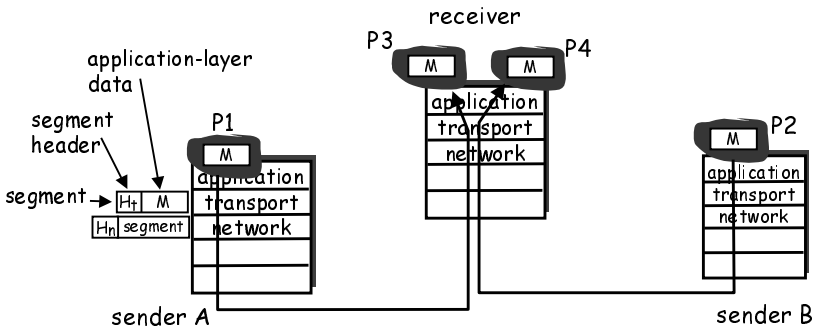
Multiplexing/demultiplexing

How hosts handle more than one session simultaneously

segment - unit of data exchanged between transport layer entities

Demultiplexing: delivering received segments to correct app layer processes



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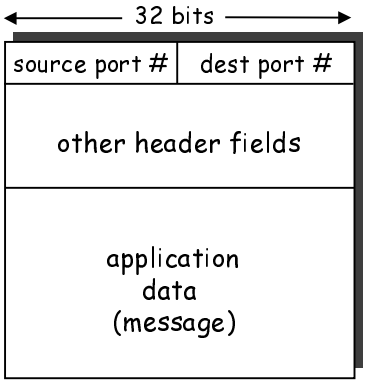
Multiplexing/demultiplexing

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

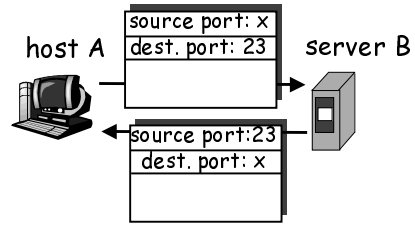
multiplexing/demultiplexing in the Internet:

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

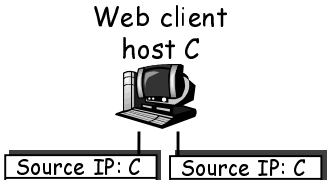


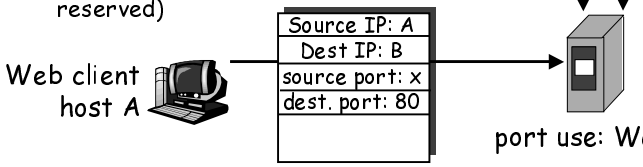
TCP/UDP segment format

Multiplexing/demultiplexing: examples



port use: simple telnet app
(Note how port 23 at server must be shared whereas port x at host can be reserved)





port use: Web server

Q: how does the server know which packets go with which process?

UDP: User Datagram Protocol [RFC 768]

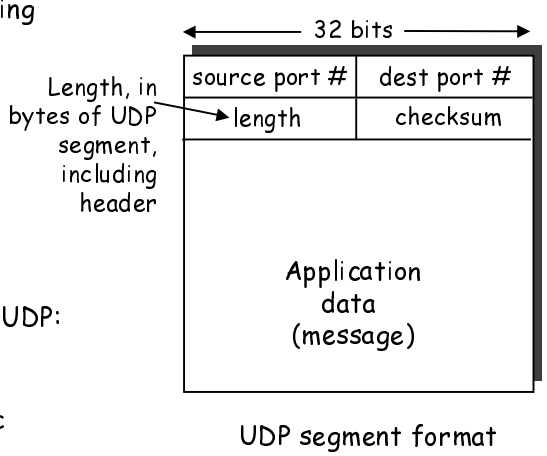
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

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

UDP: more

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses (why?):
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error 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 nonethless?* More later ….

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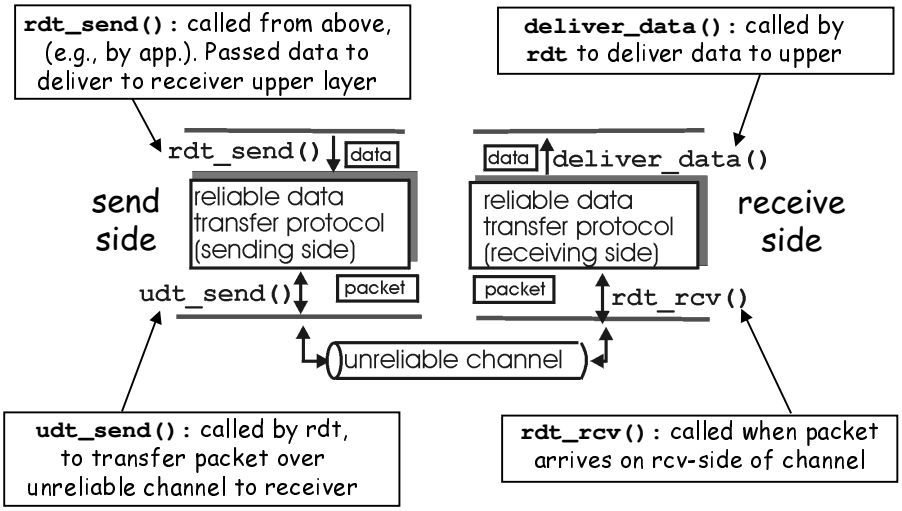
Reliable data transfer: getting started



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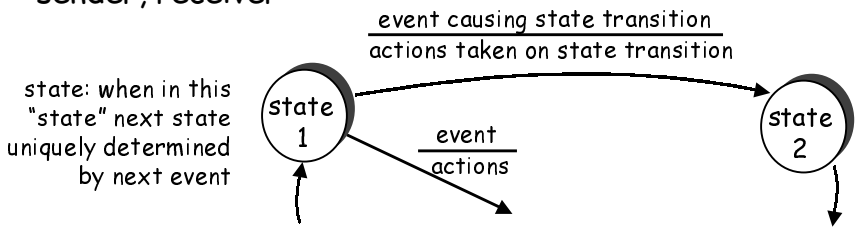
Reliable data transfer: getting started

We'll:

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



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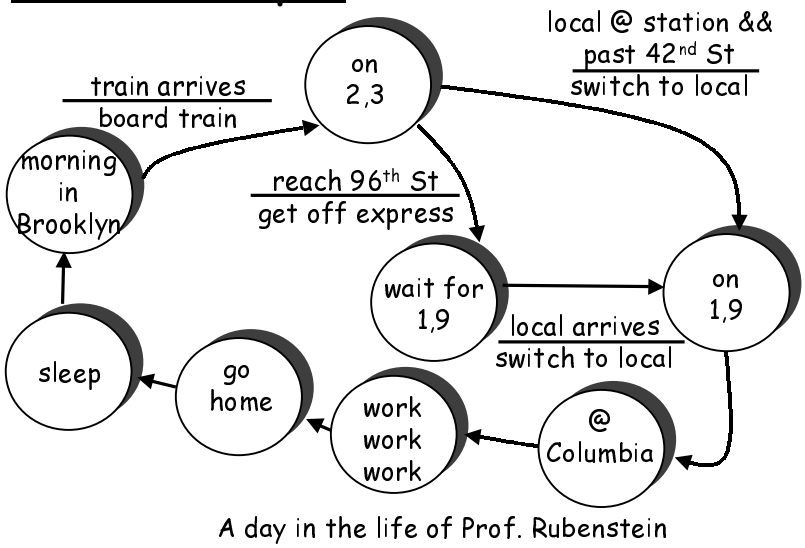
FSM example



A day in the life of Prof. Rubenstein

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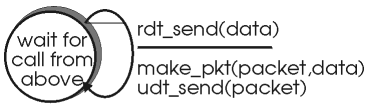
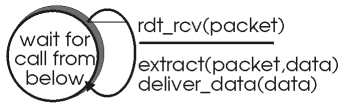
Rdt1.0: reliable transfer over a reliable channel

- 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



(a) rdt1.0: sending side

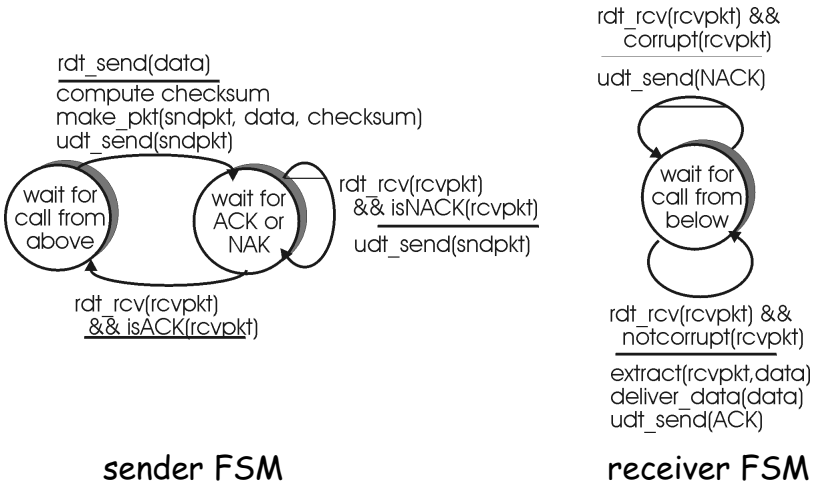


(b) rdt1.0: receiving side

Rdt2.0: channel with bit errors

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

rdt2.0: FSM specification



sender FSM

receiver FSM

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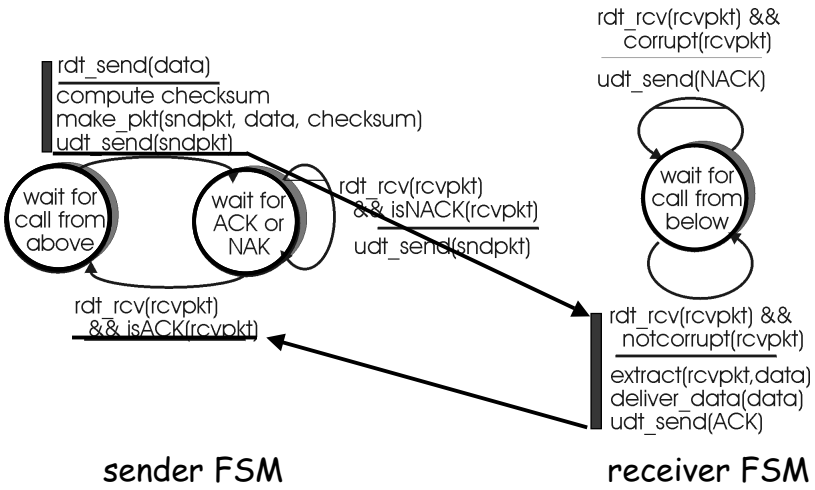
rdt2.0: in action (no errors)



sender FSM

receiver FSM

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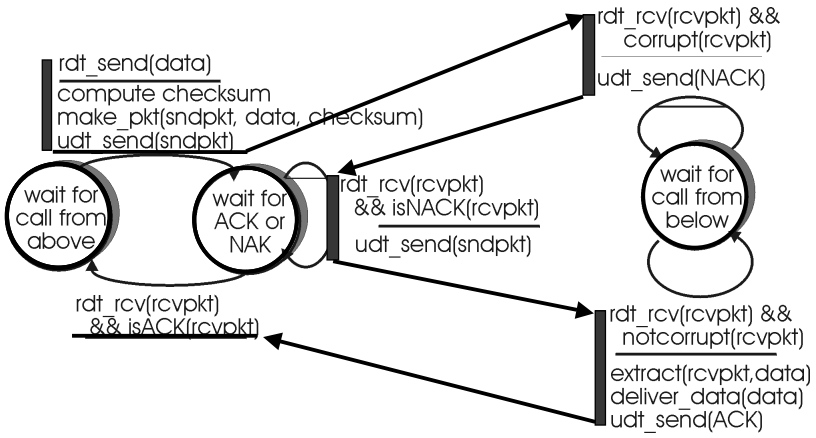
rdt2.0: in action (error scenario)



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

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

What to do?

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

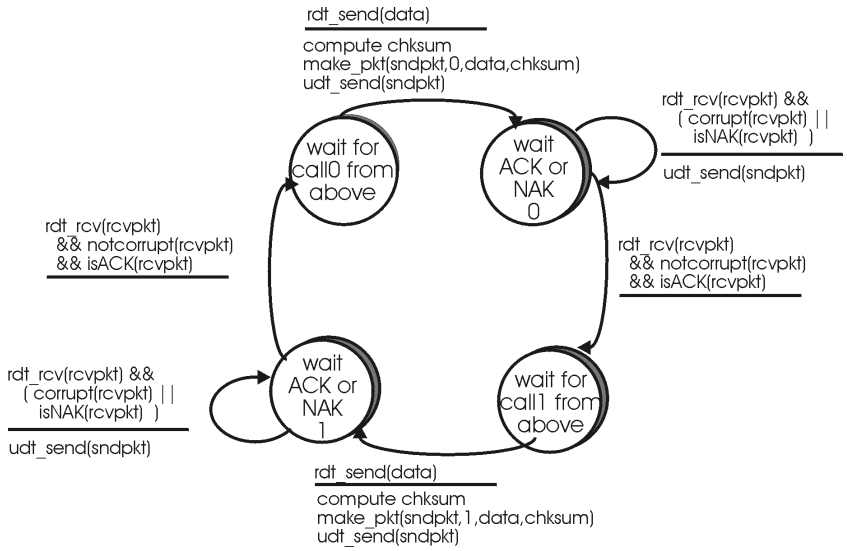
Handling duplicates:

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

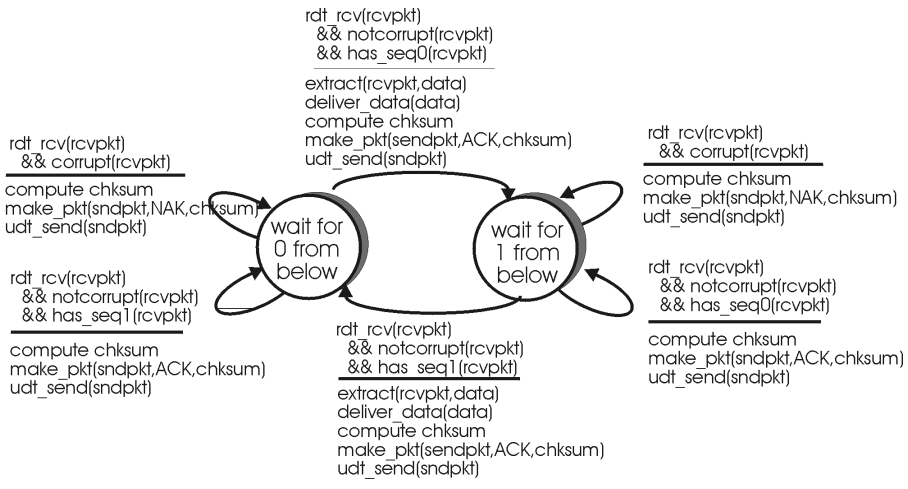
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



Note: sender & rcvr must agree on initial seqno

rdt2.1: discussion

Sender:

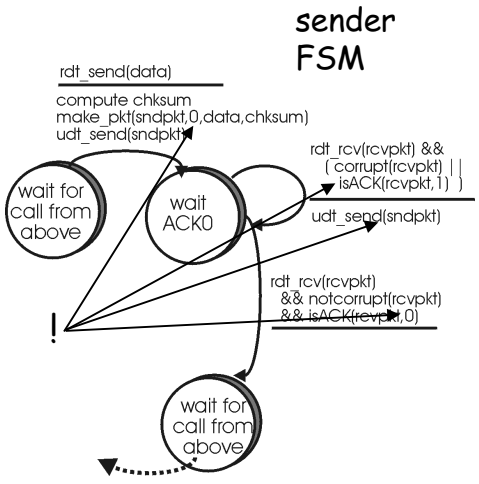
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

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*



rdt3.0: channels with errors *and* loss

New assumption: underlying channel can also lose packets (data or ACKs)

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

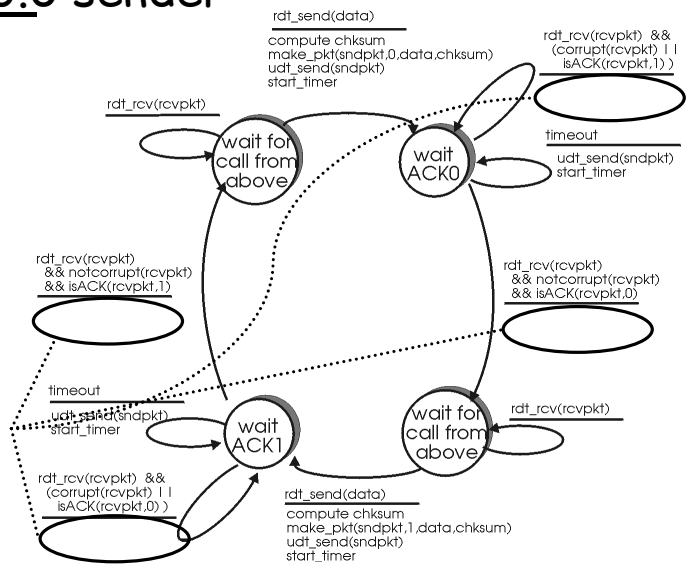
Q: how to deal with loss?

- one possibility: sender waits until certain data or ACK definitely lost, then retransmits
- drawbacks?

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

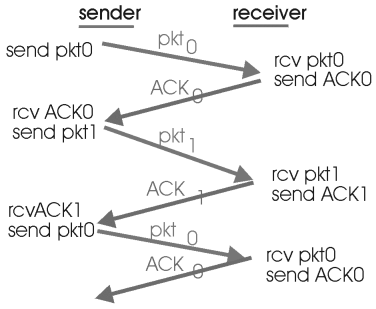
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt3.0 sender

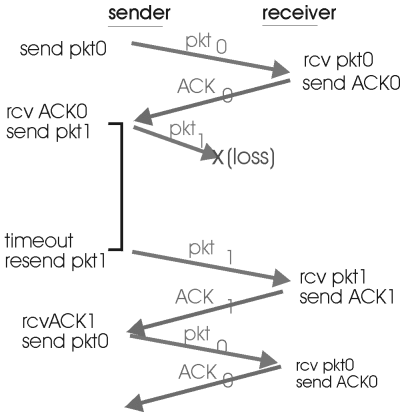


why not retransmit pkts at these events?

rdt3.0 in action



(a) operation with no loss



(b) lost packet

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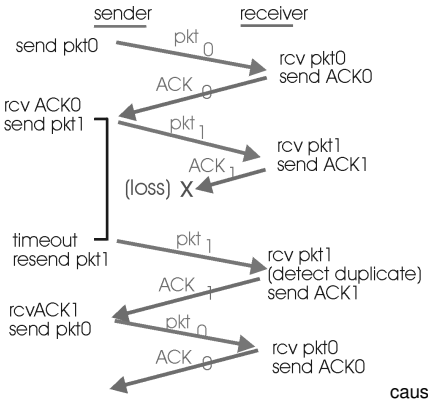
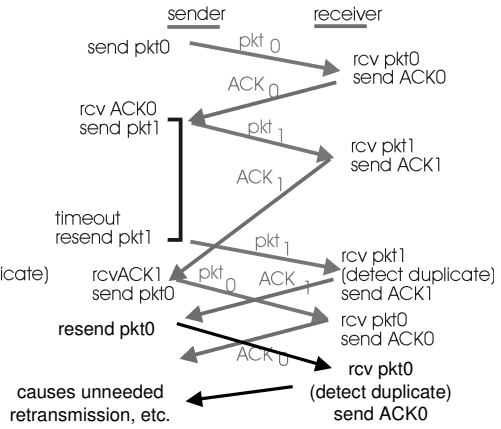
rdt3.0 in action



(c) lost ACK



(d) premature timeout

Transmitting on those "other" events (2 slides ago)
would have caused cascade of redundant retransmissions

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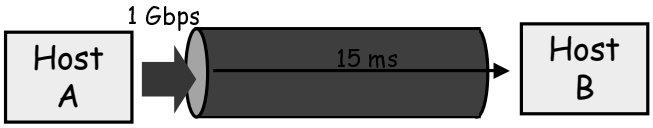
Performance of rdt3.0

- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:



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

$$Utilization = U = \frac{\text{fraction of time sender busy sending}}{} = \frac{8\ microsec}{30.016\ msec} = 0.00015$$

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

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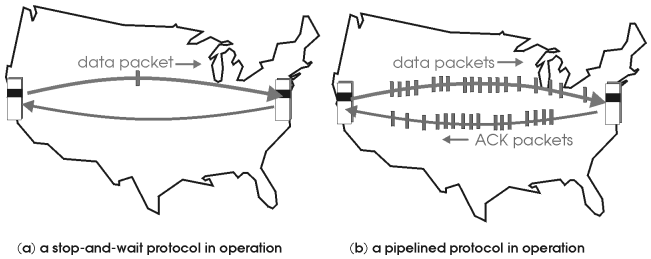
Pipelined protocols

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

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



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

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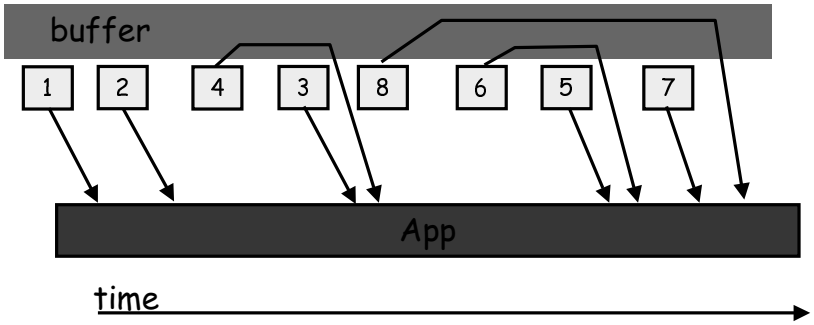
In-order buffering



- Transport layer maintains a per-session buffer
  - pkts possibly placed in buffer out of order (e.g., due to network loss)
  - pkts are sent up to app (and then removed from buffer) in order

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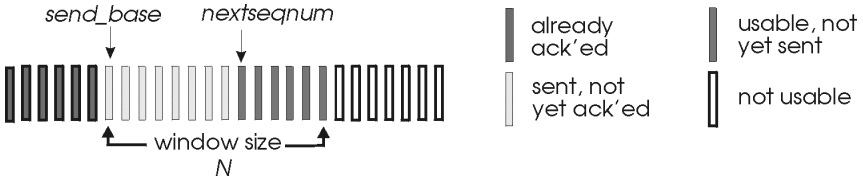
Go-Back-N

Sender:

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

Rcvr:

- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may deceive duplicate ACKs (see receiver)



More Sender:

- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

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GBN: receiver extended FSM

sender

- have N pkts "in transit"
- roll window past largest ACK
- on timeout of lowest seqno packet in window, retransmit current window (and reset timers)

receiver simple:

- ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember **expectedseqnum**
- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - ACK pkt with highest in-order seq #

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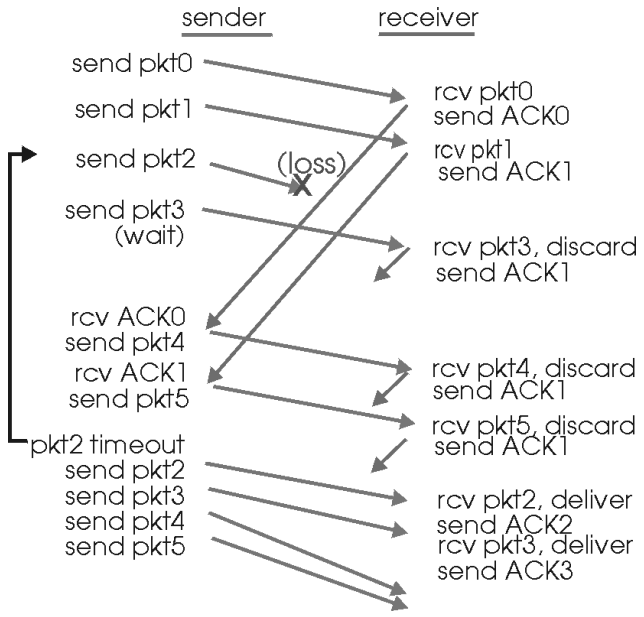
GBN in action

Here, N=3



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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 maintains timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq #s of sent, unACKed pkts

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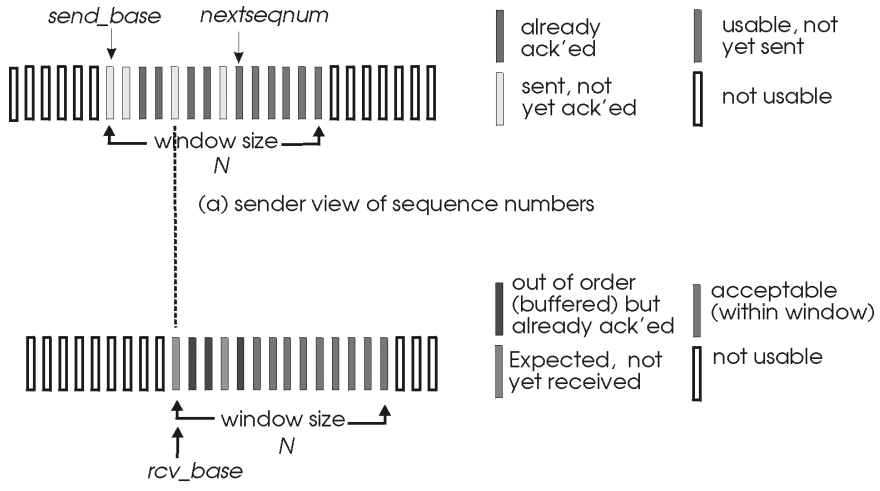
Selective repeat: sender, receiver windows



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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

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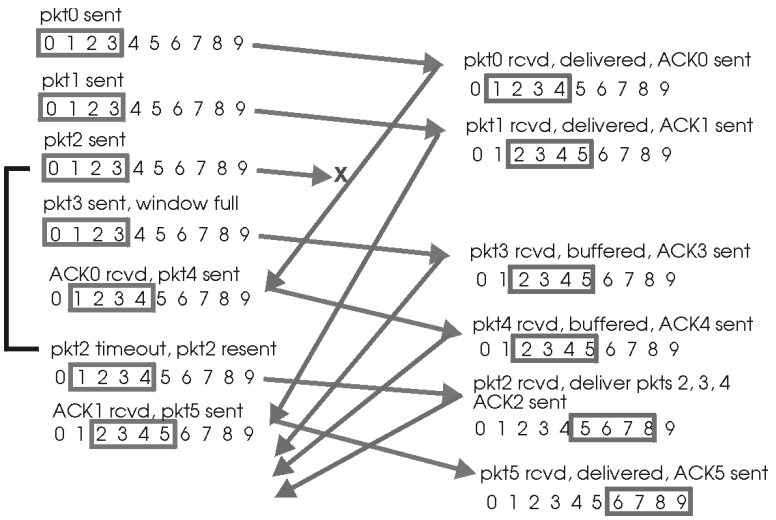
Selective repeat in action



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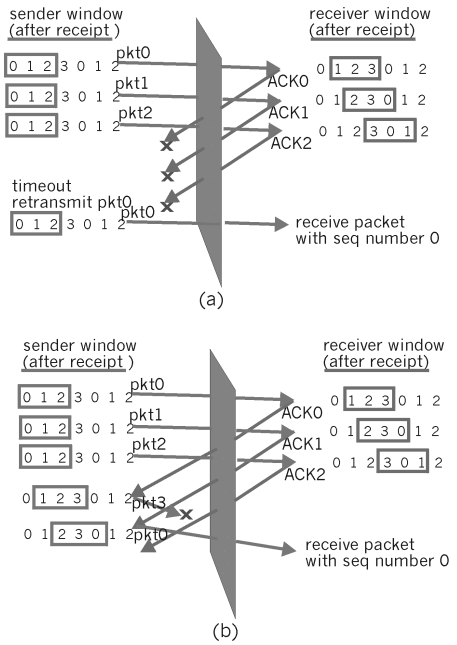
Selective repeat: dilemma

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?



Go-back-N vs. Selective Repeat

- Q: How do bandwidth requirements compare?
- Let's do a simple analytical comparison
- Model:
  - any packet transmission lost with probability, p
  - ACKs never lost
  - selective repeat:
    - sender knows exactly what rcvr needs
  - Go-back-N
    - each round, sender transmits block of N pkts
    - rcvr informs sender of 1st lost pkt
    - sender sends N pkts starting at 1st point of loss
    - rcvr dumps any pkts in window after a loss

Selective Repeat Analysis

- Each pkt can be "examined" in isolation
- T_{SR} = # of transmissions of a pkt
- $P(T_{SR} > i) = p^i$
- $E[T_{SR}] = P(T_{SR}=1) + 2\,P(T_{SR}=2) + 3P(T_{SR}=3) + \ldots$
  $= P(T_{SR} > 0) + P(T_{SR} > 1) + P(T_{SR} > 2) + P(T_{SR} > 3) + \ldots$
  $= 1 / (1-p)$
- e.g., p = .2, then $E[T_{SR}] = 5$

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Go-Back-N analysis

- S_N = # pkts arriving prior to loss in window of N
- $P(S_N > i) = (1-p)^{i+1}$, $0 \le i < N$, $= 0$ for $i \ge N$
- $E[S_N] = P(S_N > 0) + P(S_N > 1) + \ldots + P(S_N > N-1)$
  $= (1 - p - (1-p)^{N+1}) / p$
- Let $S_{N,j}$ = # of pkts accepted in the j^{th} transmission
- $E[T_{GBN}]$ = avg. # of transmissions of pkt

$$= \frac{\sum_{j=1}^{m} N}{\sum_{j=1}^{m} (S_{N,j})} \qquad = \frac{(\sum_{j=1}^{m} N) / m}{(\sum_{j=1}^{m} S_{N,j}) / m}$$

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Go-back-N analysis (cont'd)

- as $m \to \infty$ $= N / E[S_N]$

$$E[T_{GBN}] = \frac{Np}{1 - p - (1-p)^{N+1}}$$

- How does $E[T_{SR}]$ compare with $E[T_{GBN}]$

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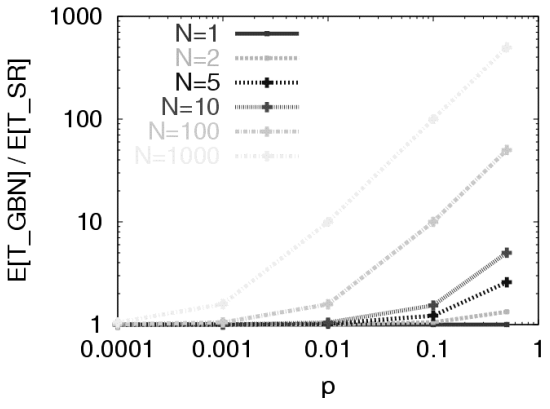
Go-Back-N vs. Selective Repeat

- Using our analysis, for various N, p: how much more efficient is Selective Repeat vs. Go-Back-N?



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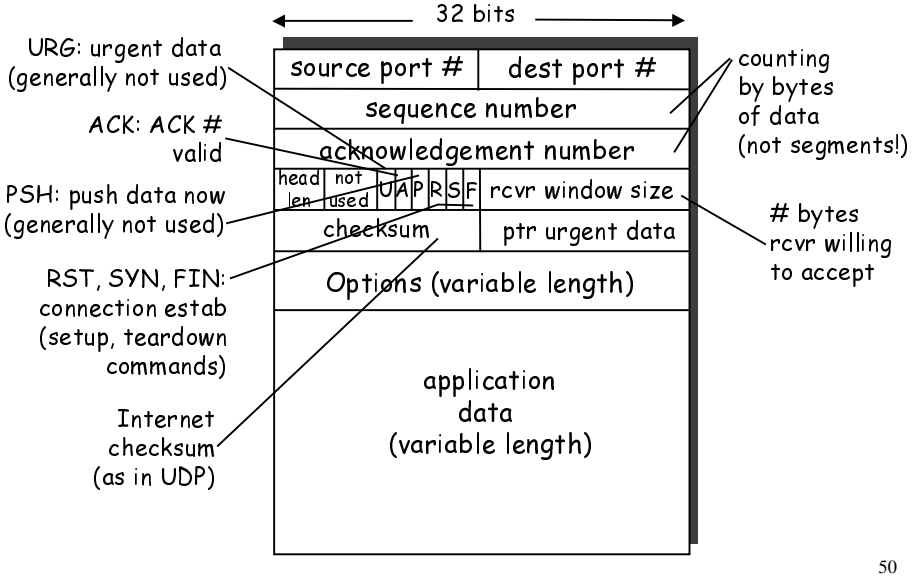
TCP: Overview

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

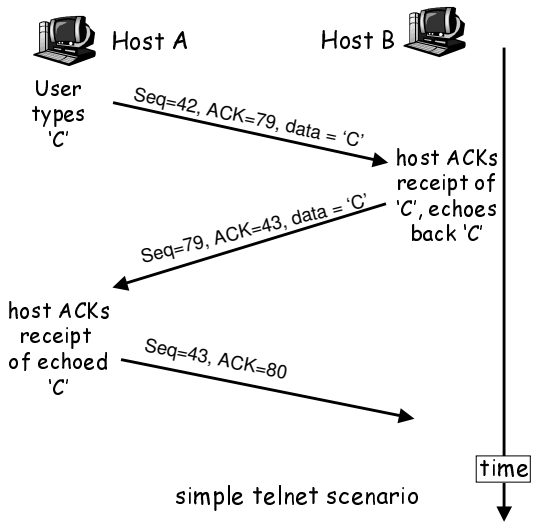


TCP seq. #'s and ACKs

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

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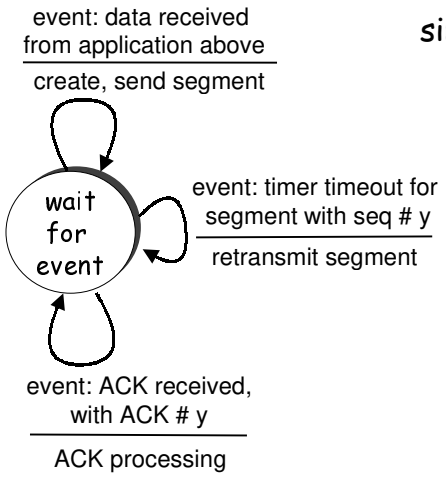
TCP: reliable data transfer



simplified sender, assuming
- one way data transfer
- no flow, congestion control

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TCP: reliable data transfer

Simplified TCP sender

Similar to GBN, but some slight differences...

```
00  sendbase = initial_sequence number
01  nextseqnum = initial_sequence number
02
03   loop (forever) {
04     switch(event)
05     event: data received from application above
06          create TCP segment with sequence number nextseqnum
07          start timer for segment nextseqnum
08          pass segment to IP
09          nextseqnum = nextseqnum + length(data)
10      event: timer timeout for segment with sequence number y
11          retransmit segment with sequence number y
12          compute new timeout interval for segment y
13          restart timer for sequence number y
14      event: ACK received, with ACK field value of y
15          if (y > sendbase) { /* cumulative ACK of all data up to y */
16              cancel all timers for segments with sequence numbers < y
17               sendbase = y
18               }
19          else { /* a duplicate ACK for already ACKed segment */
20               increment number of duplicate ACKs received for y
21               if (number of duplicate ACKS received for y == 3) {
22                  /* TCP fast retransmit */
23                 resend segment with sequence number y
24                 restart timer for segment y
25               }
26     }  /* end of loop forever */
```

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TCP ACK generation [RFC 1122, RFC 2581]

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

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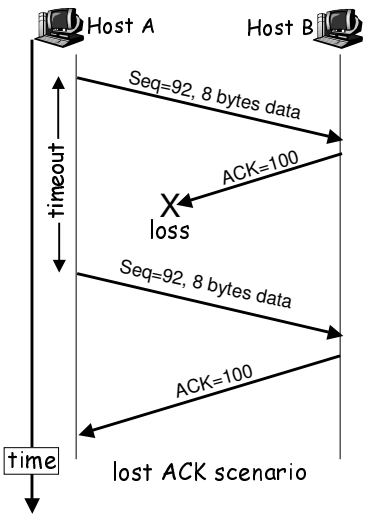
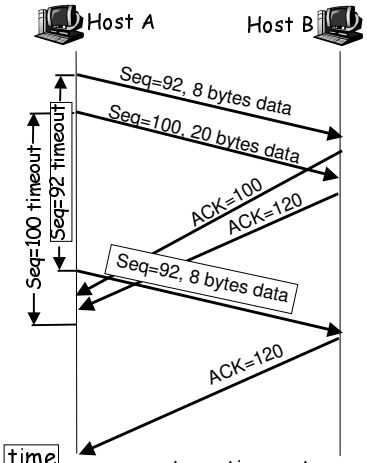
TCP: retransmission scenarios



lost ACK scenario



premature timeout, cumulative ACKs

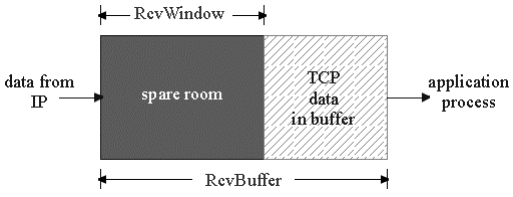
TCP Flow Control

flow control

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

RcvBuffer = size or TCP Receive Buffer

RcvWindow = amount of spare room in Buffer



receiver buffering

- receiver: explicitly informs sender of (dynamically changing) amount of free buffer space
  - **RcvWindow field** in TCP segment
- sender: keeps the amount of transmitted, unACKed data less than most recently received **RcvWindow**

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

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

Q: how to estimate RTT?

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

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TCP Round Trip Time and Timeout

EstimatedRTT = (1−x)*EstimatedRTT + x*SampleRTT

- Exponential weighted moving average
- influence of given sample decreases exponentially fast
- typical value of x: 0.1

Setting the timeout

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

Timeout = EstimatedRTT + 4*Deviation

Deviation = (1−x)*Deviation +
x*|SampleRTT−EstimatedRTT|

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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 end system sends TCP SYN control segment to server
- specifies initial seq #

Step 2: server end system receives SYN, replies with SYNACK control segment
- ACKs received SYN
- allocates buffers
- specifies server-> receiver initial seq. #

Initial seqnos in both directions chosen randomly. Why?

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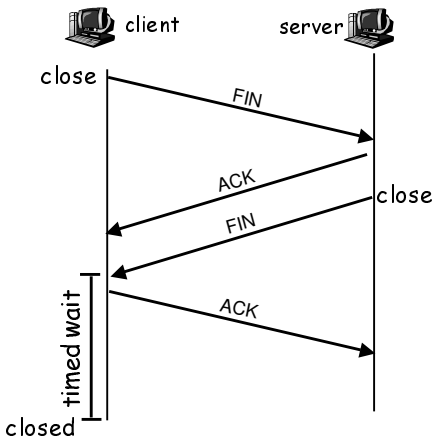
TCP Connection Management (cont.)

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.



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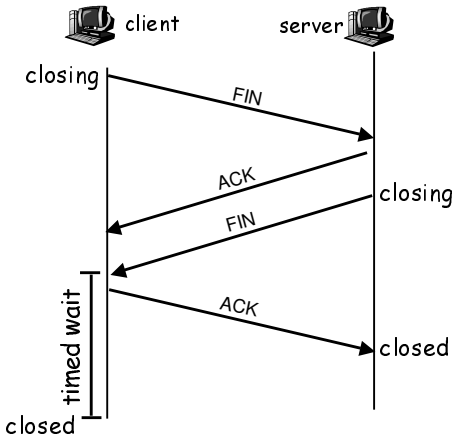
TCP Connection Management (cont.)

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.



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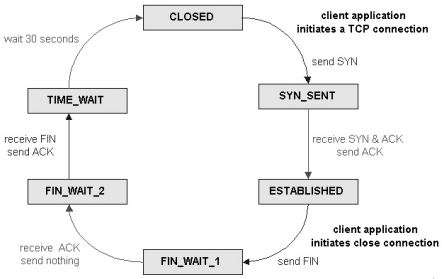
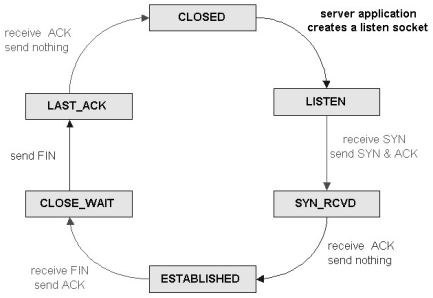
TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



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