Transport layer

Our goals:

- Understand principles behind transport layer services:
  - Multiplexing/demulti-plexing
  - 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

1

Transport layer: Outline

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - Segment structure
  - Reliable data transfer
  - Flow control
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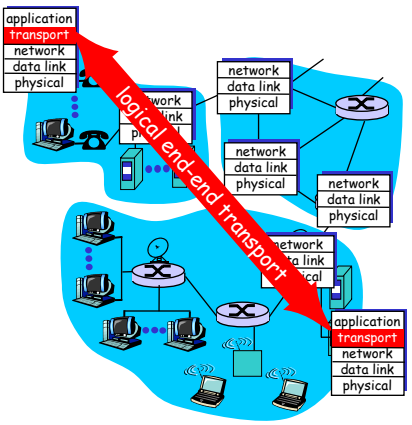
2

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



3

Transport vs. network layer

- *Network layer:* logical communication between hosts
- *Transport layer:* logical communication between processes
  - Relies on, enhances, network layer services

Household analogy:

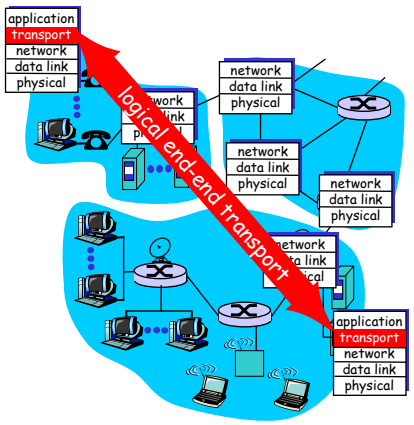
12 kids sending letters to 12 kids

- Processes = kids
- App messages = letters in envelopes
- Hosts = houses
- Transport protocol = Ann and Bill
- Network-layer protocol = postal service

4

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



5

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6

Multiplexing/demultiplexing

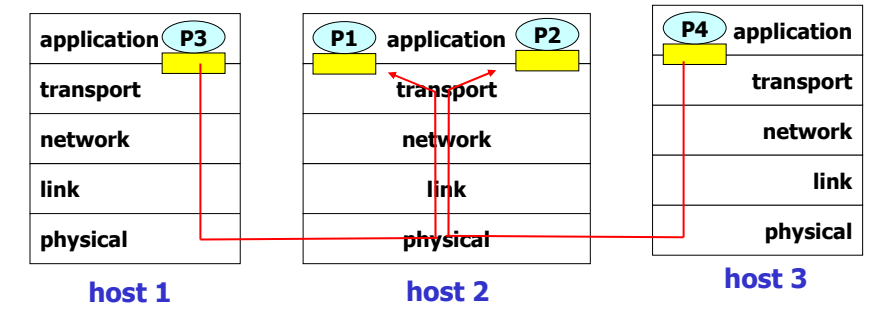
Demultiplexing at rcv host:

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



7

Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
   DatagramSocket(9111);
DatagramSocket mySocket2 = new
   DatagramSocket(9222);
```

- UDP socket identified by two-tuple:

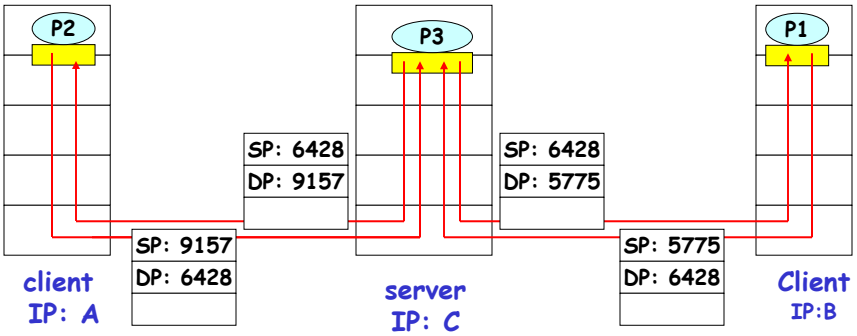
(dest IP address, dest port number)

- 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

8

Connectionless demux (cont.)

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



SP provides "return address"

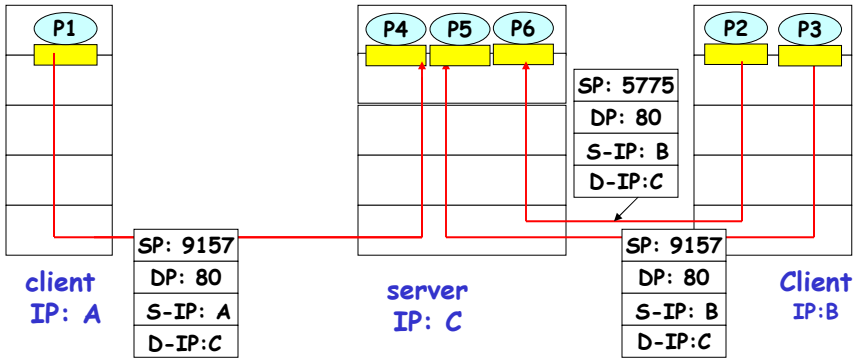
9

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

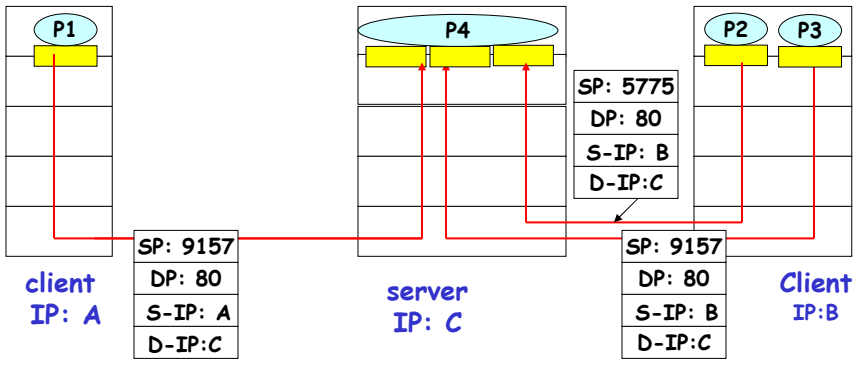
10

Connection-oriented demux (cont.)



11

Connection-oriented demux: Threaded Web Server



12

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13

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

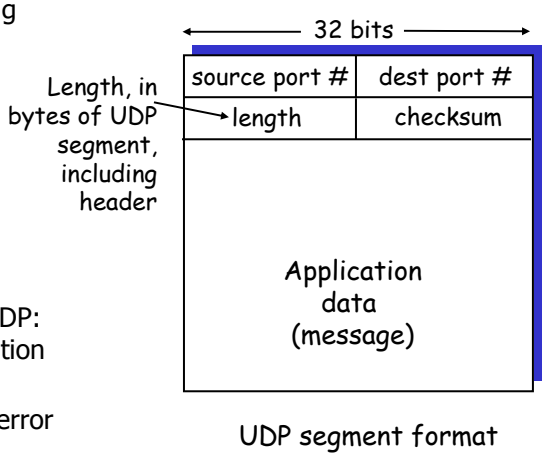
Why is there a UDP?

- 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

14

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 recover!



UDP segment format

UDP checksum

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

Sender:

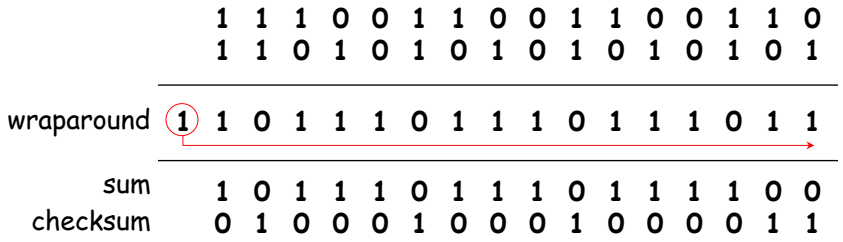
- Treat segment contents as sequence of 16-bit integers
- Checksum: addition (1's complement sum) of segment contents
- Sender puts checksum value into UDP checksum field

Receiver:

- Compute checksum of received segment
- Check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonethleess?* 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



17

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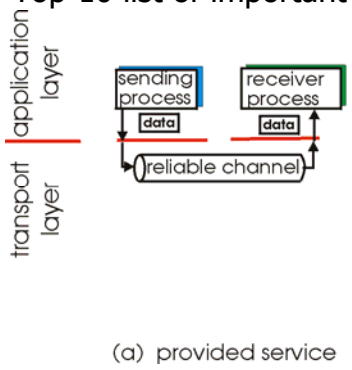
18

Principles of reliable data transfer

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



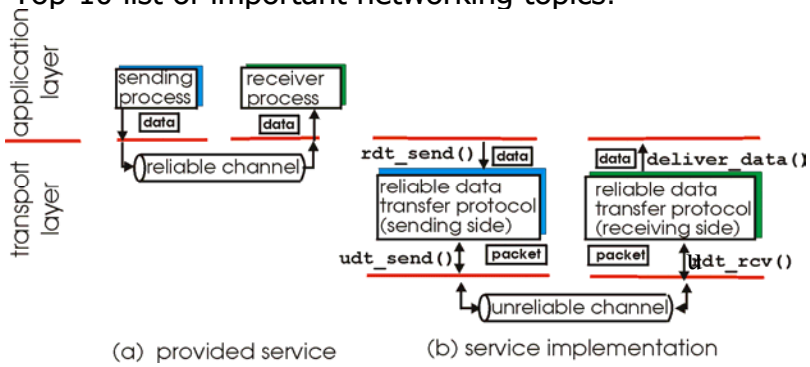
(a) provided service

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

19

Principles of reliable data transfer

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

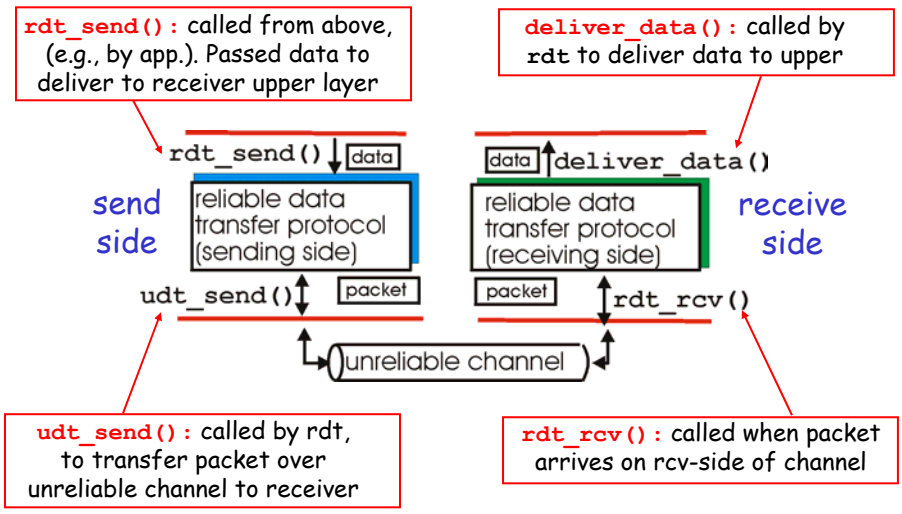


(a) provided service (b) service implementation

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

20

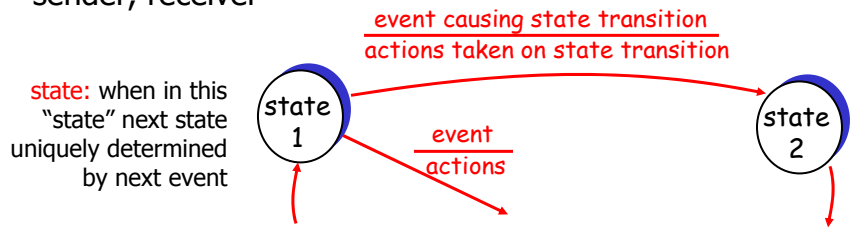
Reliable data transfer: Getting started



Reliable data transfer: Getting started

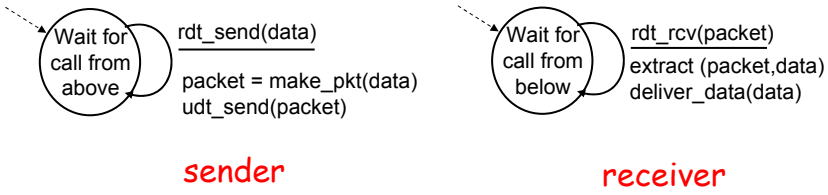
We'll:

- 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 erros
  - No loss of packets
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying channel
  - Receiver read data from underlying channel



23

Rdt2.0: Channel with bit errors

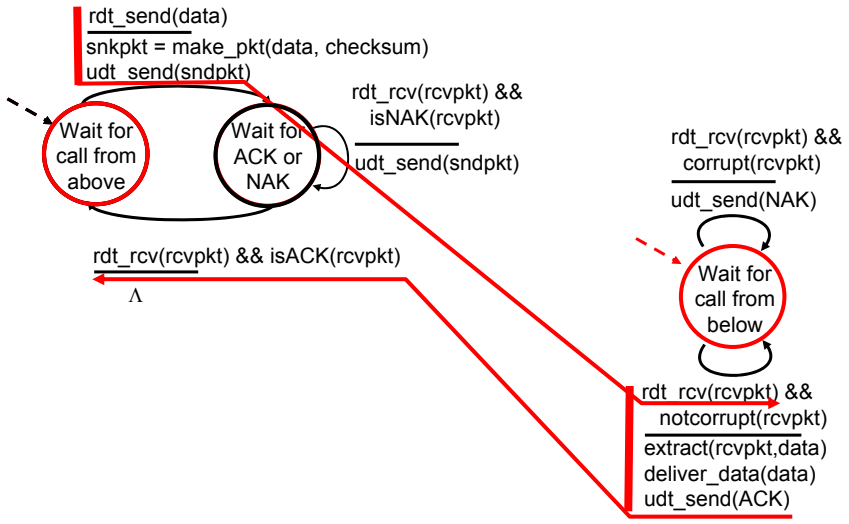
- Underlying channel may flip bits in packet
  - Recall: UDP checksum to detect bit errors
- *The* question: how to recover from errors:
  - *Acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *Negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - Sender retransmits pkt on receipt of NAK
  - Human scenarios using ACKs, NAKs?

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

- Error detection
- Receiver feedback: control msgs (ACK,NAK) rcvr->sender

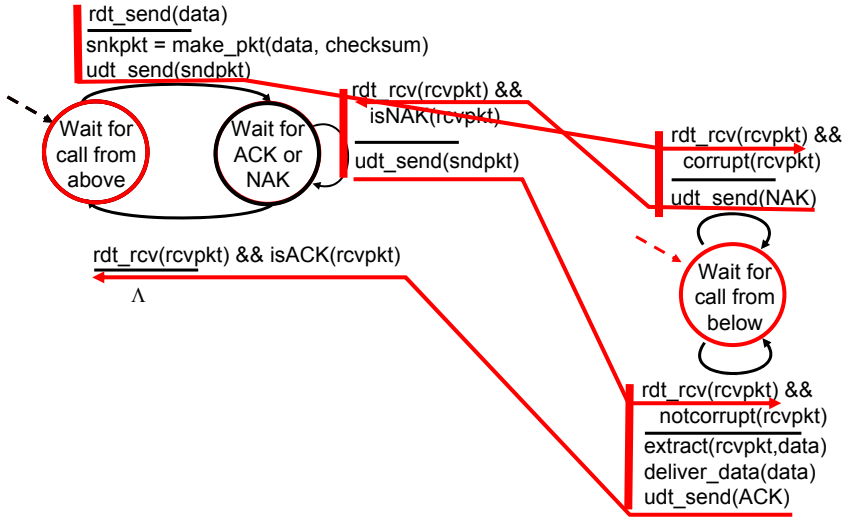
24

Rdt2.0: Operation with no errors



25

Rdt2.0: Error scenario



26

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!

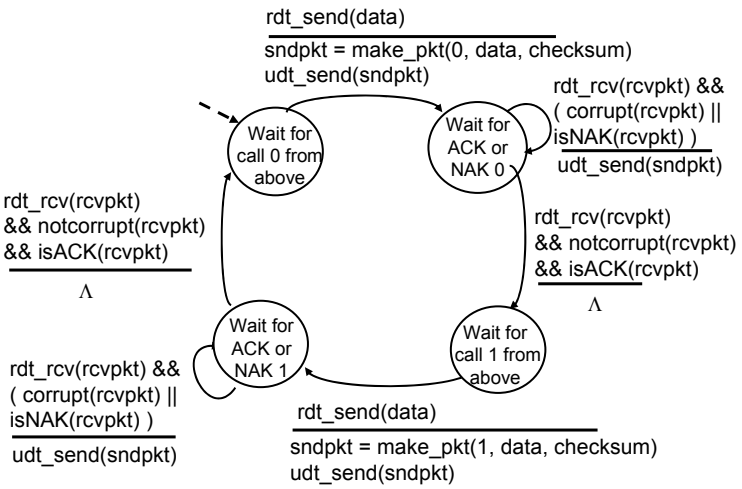
Handling duplicates:

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

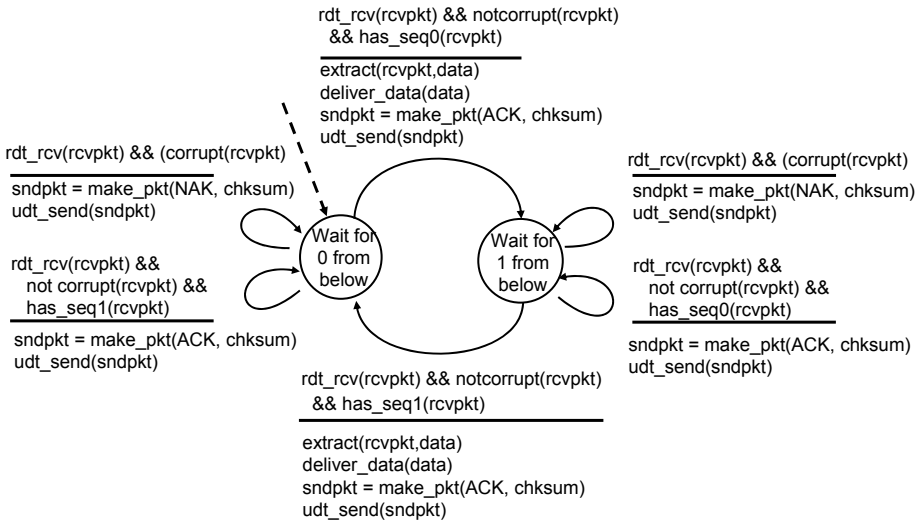
stop and wait

Sender sends one packet, then waits for receiver response

Rdt2.1: Sender with garbled ACK/NAKs



Rdt2.1: Receiver with garbled ACK/NAKs



29

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

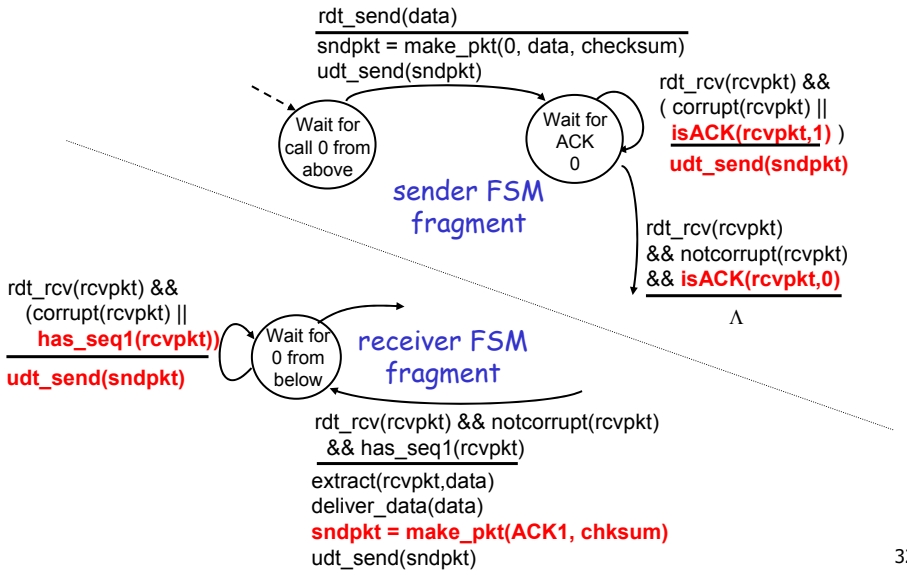
30

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*

31

Rdt2.2: Sender, receiver fragments



32

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

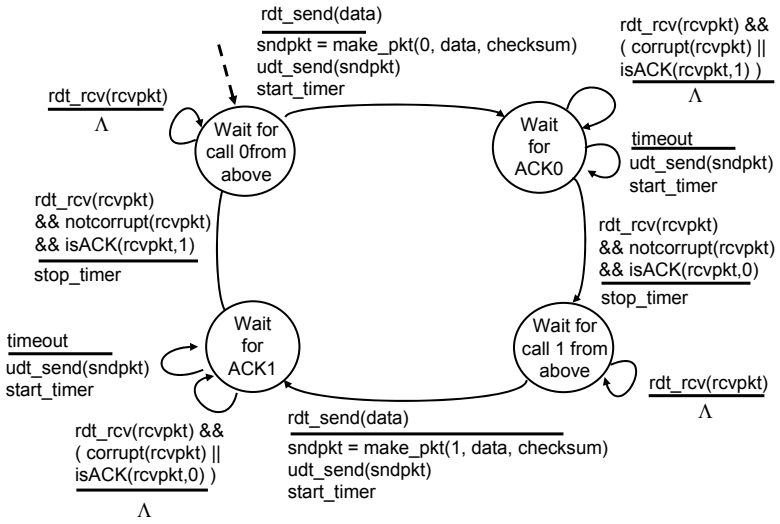
Q: How to deal with loss?

- Sender waits until certain data or ACK lost, then retransmits

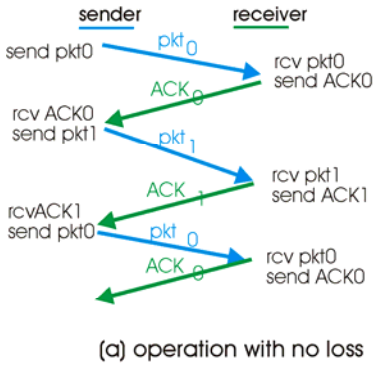
Approach: sender waits "reasonable" amount of time for ACK

- 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

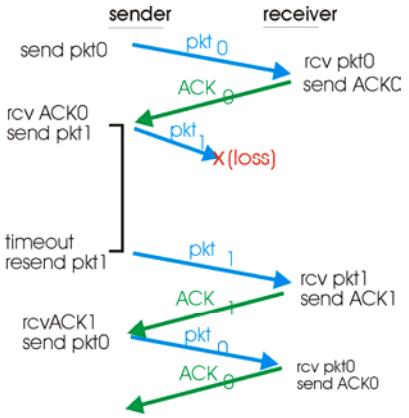
Rdt3.0 sender



Rdt3.0 in action



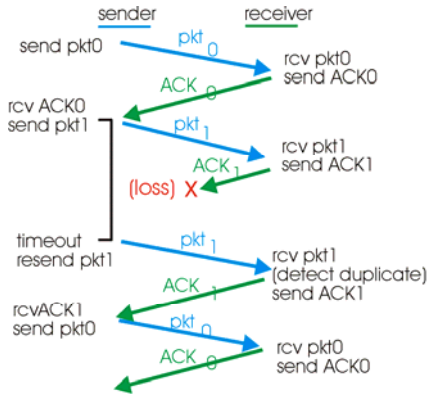
(a) operation with no loss



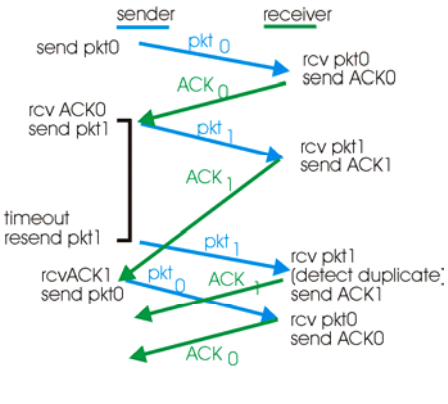
(b) lost packet

35

Rdt3.0 in action (cont.)



(c) lost ACK



(d) premature timeout

36

Performance of rdt3.0

- Rdt3.0 works, but performance stinks
- Example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8kb/pkt}{10**9\ b/sec} = 8 \text{ microsec}$$

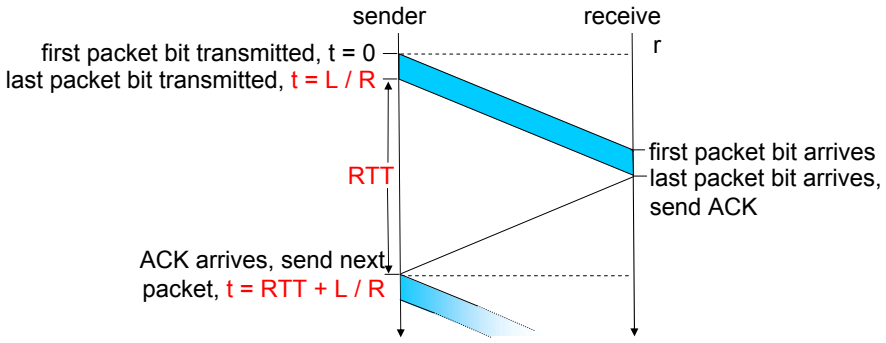
  - U_{sender}: utilization – fraction of time sender busy sending

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

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

37

Rdt3.0: Stop-and-wait operation



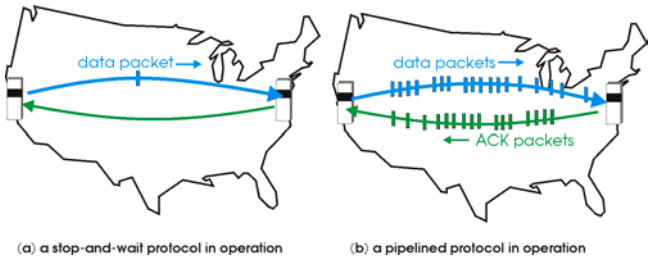
$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

38

Pipelined protocols

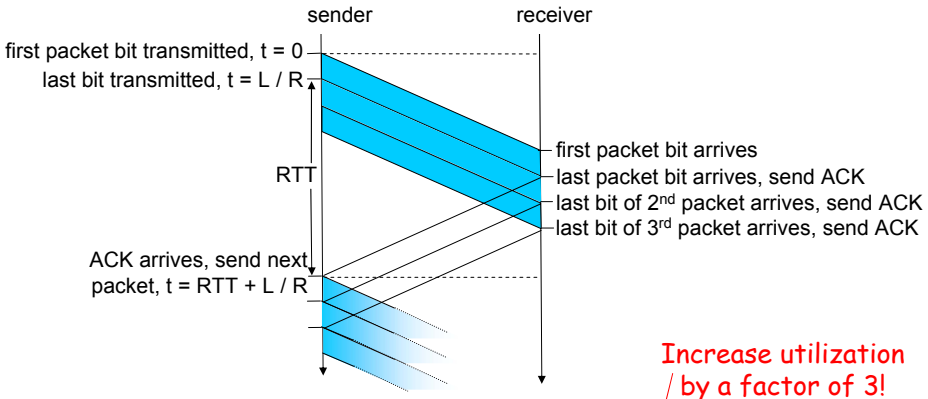
Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- Range of sequence numbers must be increased
- Buffering at sender and/or receiver



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

Pipelining: Increased utilization



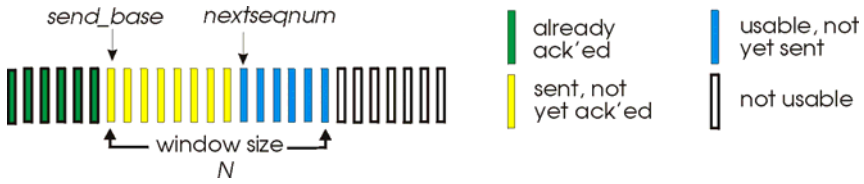
Increase utilization by a factor of 3!

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

Go-Back-N

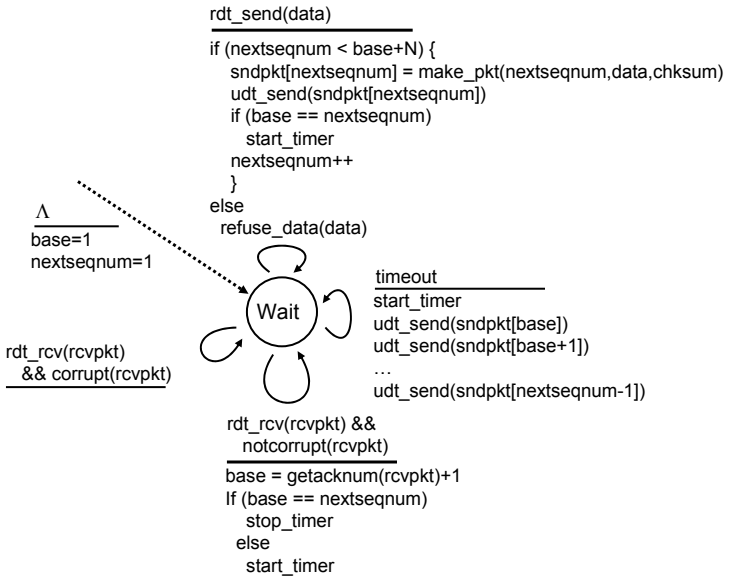
Sender:

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

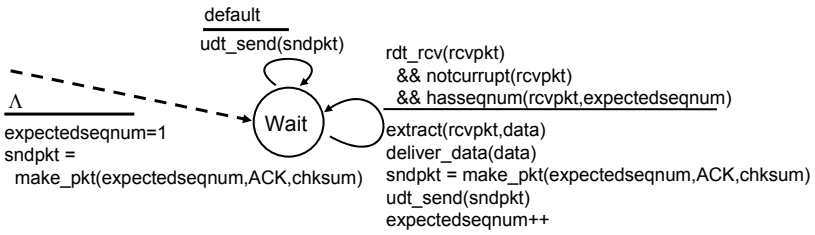


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - May deceive duplicate ACKs (see receiver)
- Timer for each in-flight pkt
- *Timeout(n):* Retransmit pkt n and all higher seq # pkts in window

GBN: Sender extended FSM



GBN: Receiver extended FSM



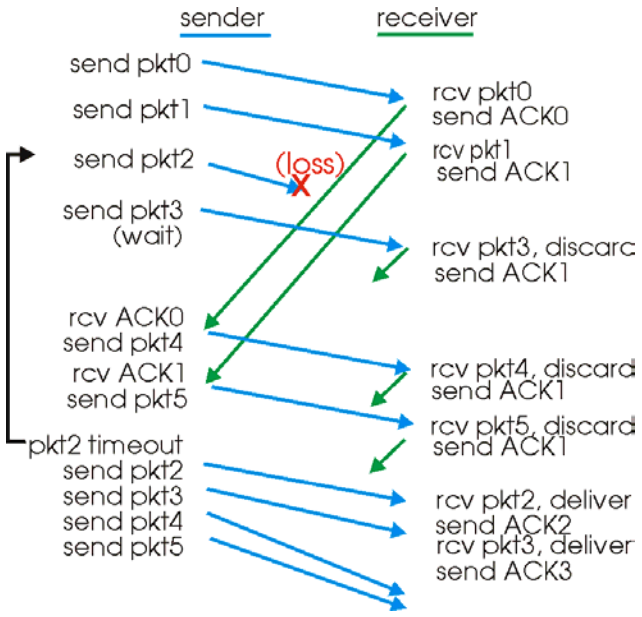
ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

43

GBN in action



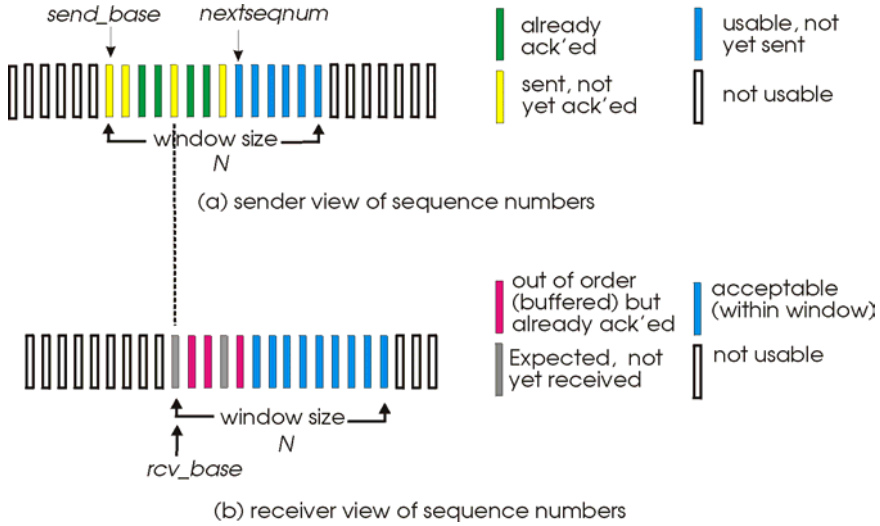
44

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

45

Selective repeat: Sender, receiver windows



46

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]

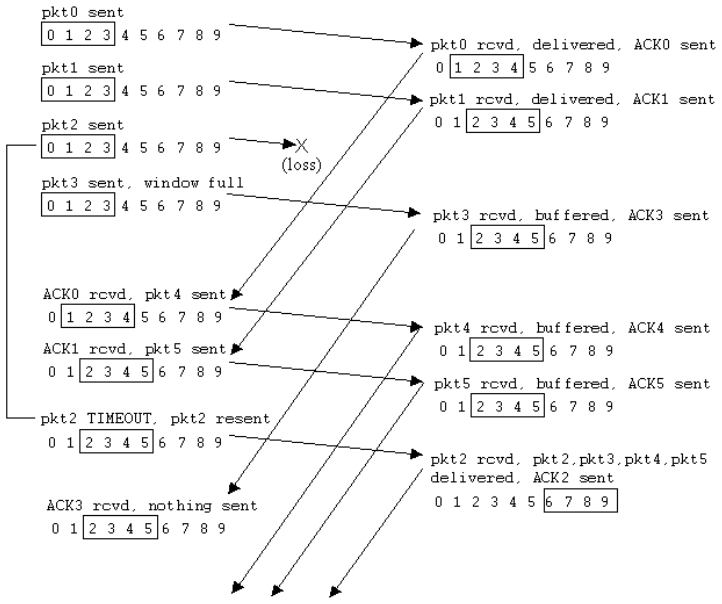
- ACK(n)

Otherwise:

- Ignore

47

Selective repeat in action



48

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?