ELEC / COMP 177 – Fall 2014

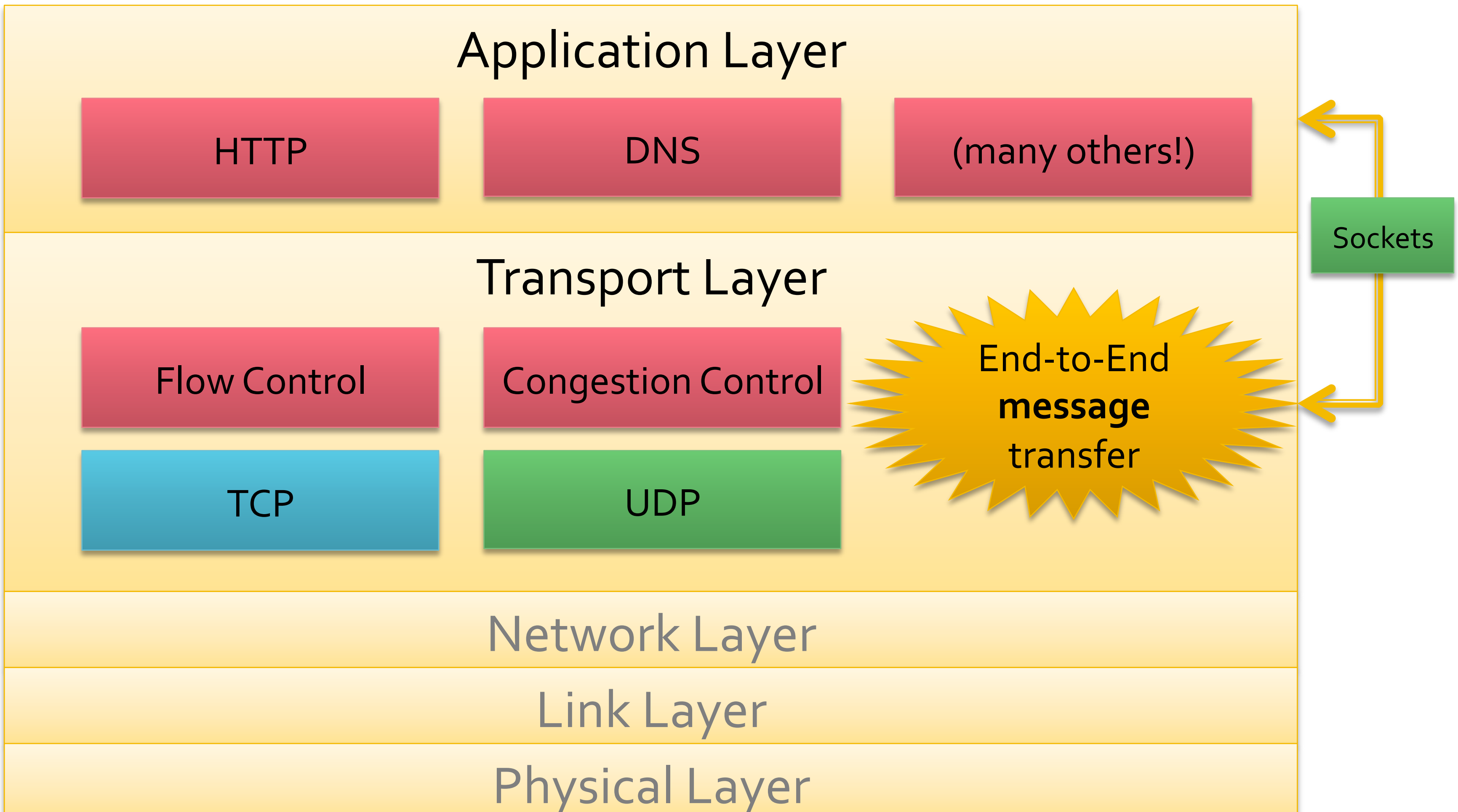
Computer Networking → Transport Layer (TCP & UDP)

Some slides from Kurose and Ross, *Computer Networking*, 5th Edition

Schedule

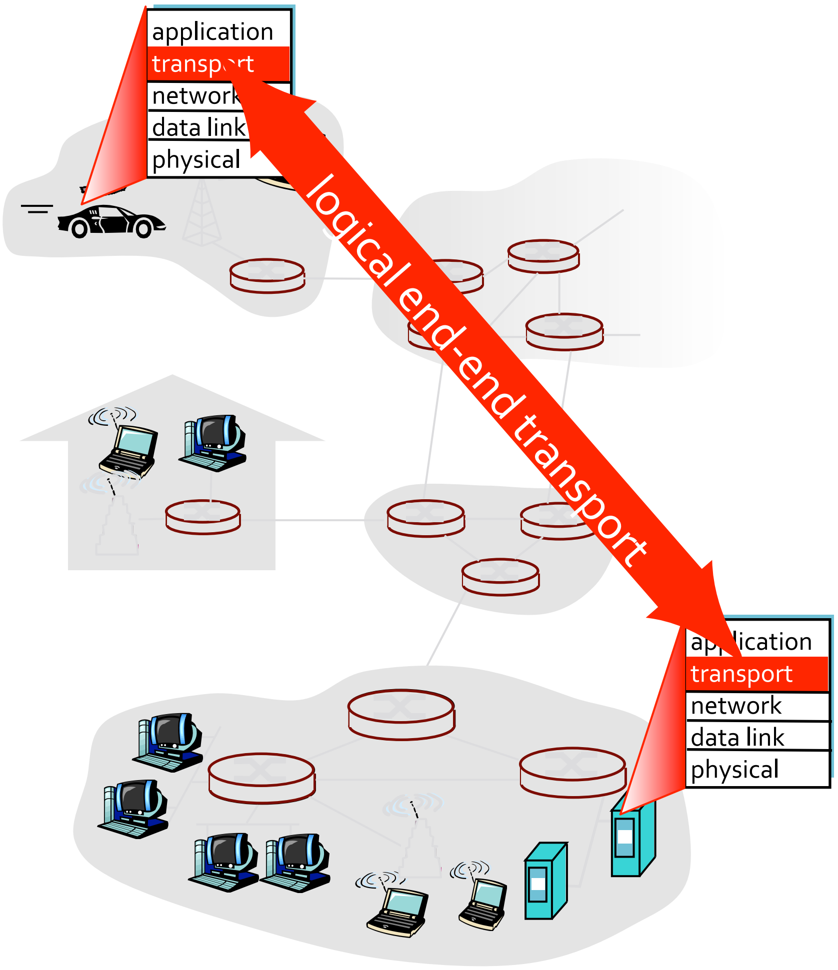
- **Project 3** – Due Thursday October 23rd
- **Quiz 4** – Tuesday October 28th
  - Topics: Transport Layer (TCP, UDP)
- **Presentation 2** – Topic selection due Thursday October 30th

Introducing the Transport Layer



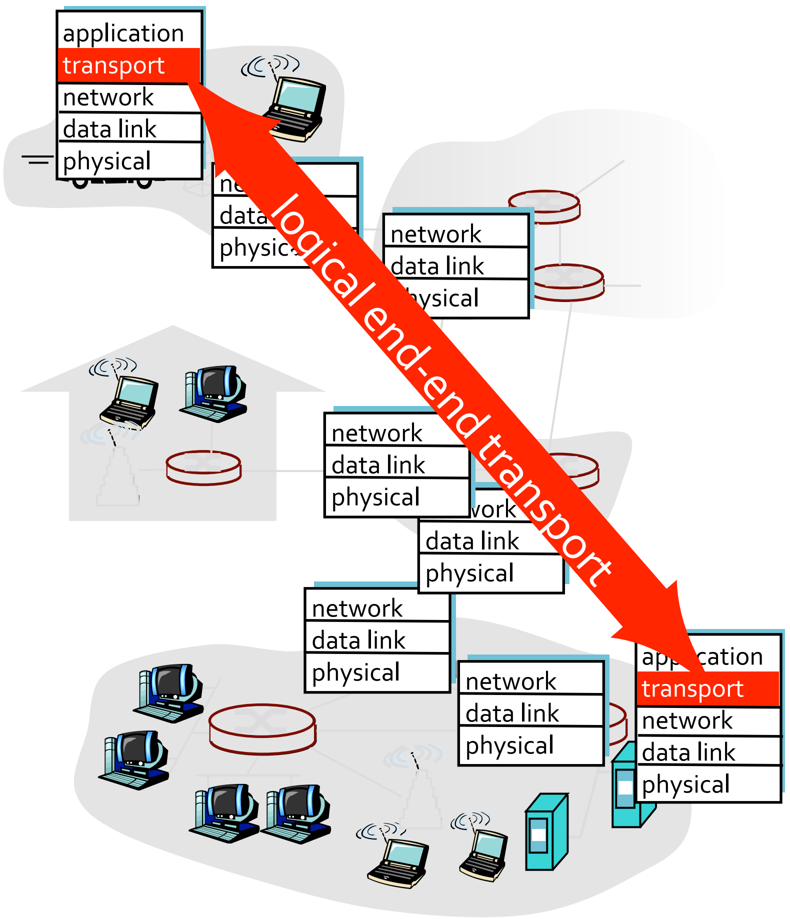
Goal of Transport Layer

- Provide *logical communication* between application processes running on different hosts
- Transport protocols run in end systems
  - Send side: breaks app messages into segments, passes to network layer
  - Receive side: reassembles segments into messages, passes to app layer
- More than one transport protocol available to apps
  - Internet: TCP and UDP



Internet Transport-layer Protocols

- Unreliable, unordered delivery (UDP)
  - No-frills extension of "best-effort" IP
- Reliable, in-order delivery (TCP)
  - Congestion control
  - Flow control
  - Connection setup
- Services not available:
  - Delay guarantees
  - Bandwidth guarantees



UDP – User Datagram Protocol

Connectionless Transport

UDP: User Datagram Protocol [RFC 768]

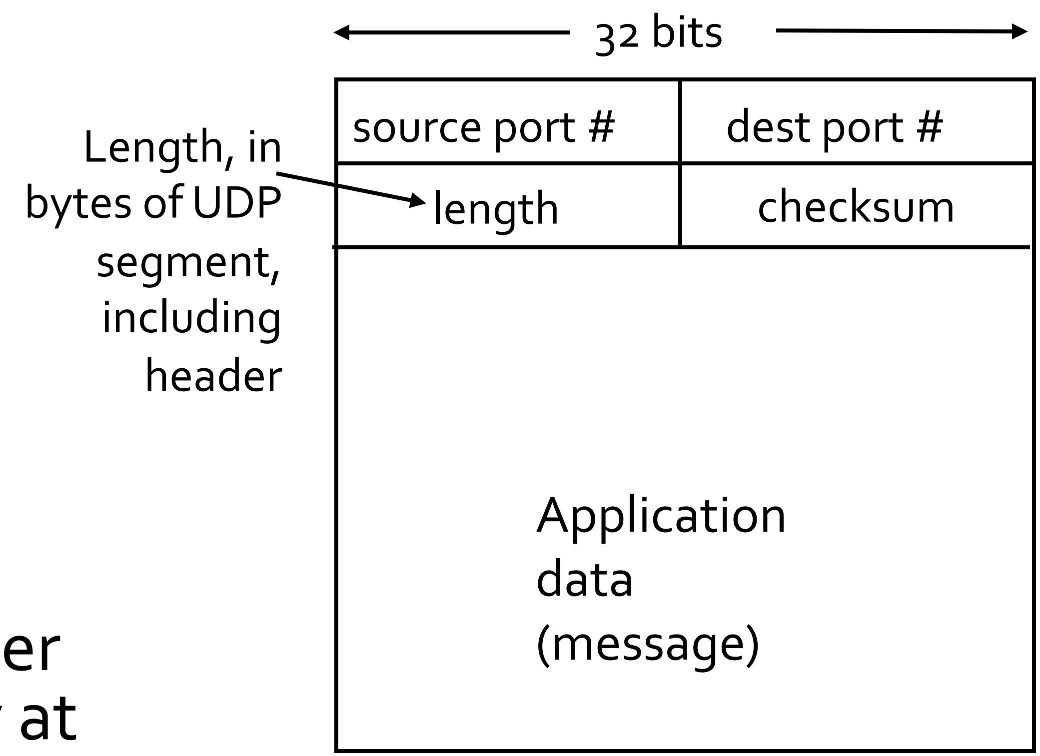
- "No frills, bare bones" Internet transport protocol
- "Best effort" service
- UDP segments may be:
  - Lost
  - Delivered out of order to app
- Connectionless
  - No handshaking between UDP sender, receiver
  - Each UDP segment handled independently of others

Why is there a UDP?

- Need *something* to provide port numbers (specific source/destination application)
- No connection establishment (adds delay)
- Simple: no connection state at sender / receiver
- Small segment header
- No congestion control
  - UDP can blast away as fast as desired

UDP

- 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

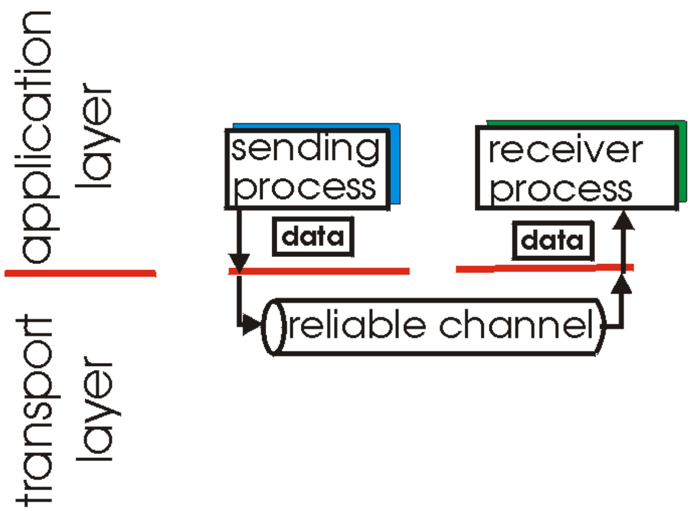
- **Sender**
  - Treat segment contents as sequence of 16-bit integers
  - Checksum: addition (1's complement sum) of segment contents
  - Sender puts checksum value into UDP checksum field
- **Receiver**
  - Compute checksum of received segment
  - Check if computed checksum equals checksum field value:
    - NO - error detected
    - YES - no error detected. But maybe errors nonetheless?

Reliable Data Transfer

Stepping through the design of TCP

Principles of Reliable data transfer

- Reliability is important in application, transport, and link layers

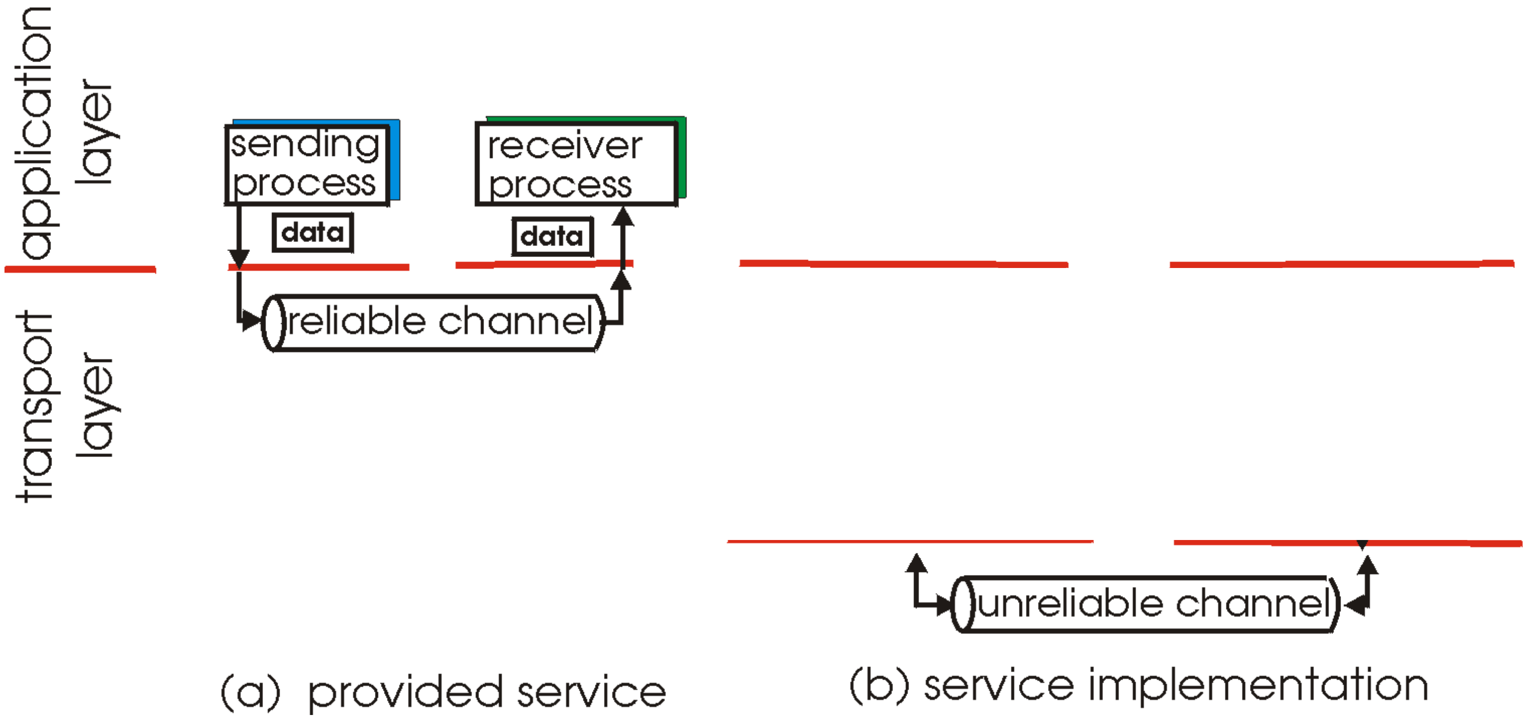


(a) provided service

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

Principles of Reliable data transfer

- Reliability is important in application, transport, and link layers



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Principles of Reliable data transfer

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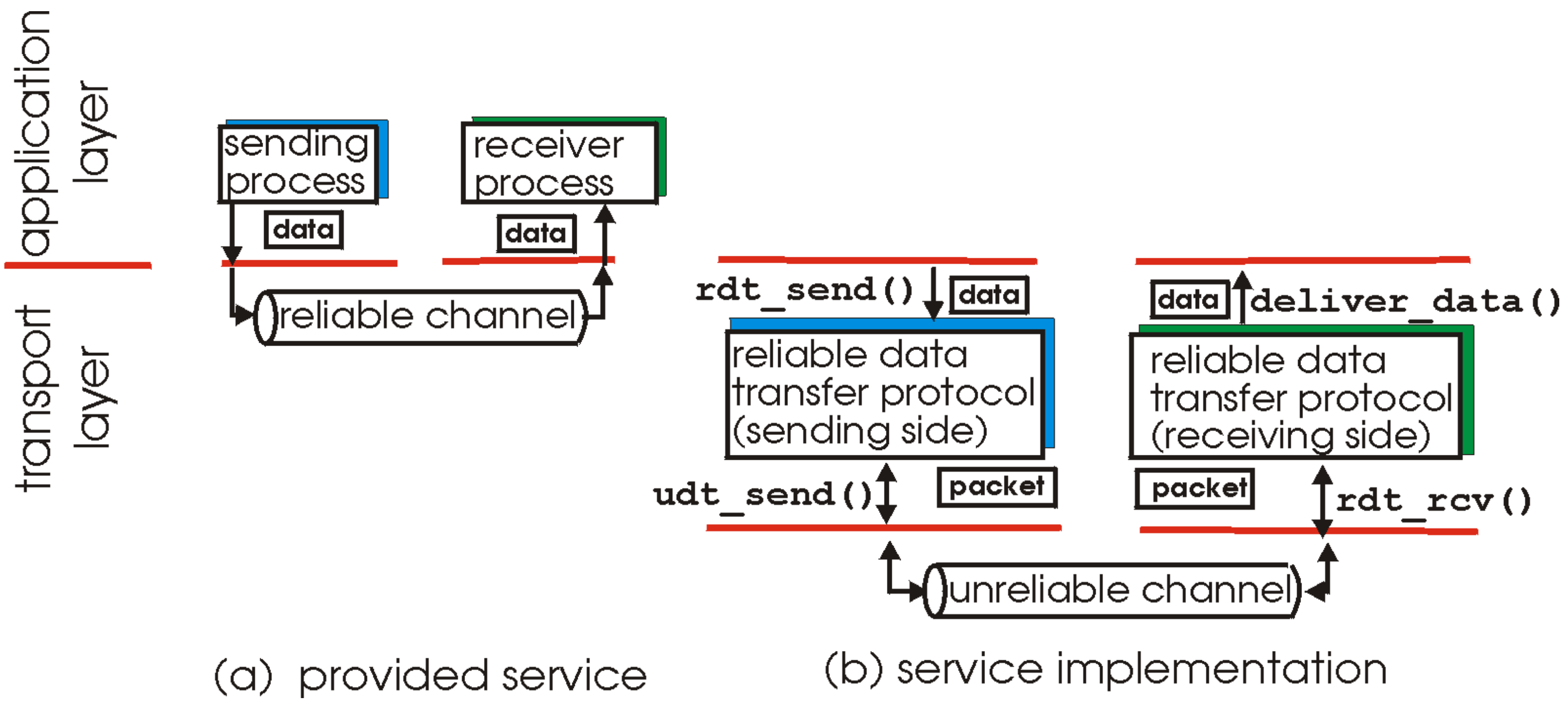
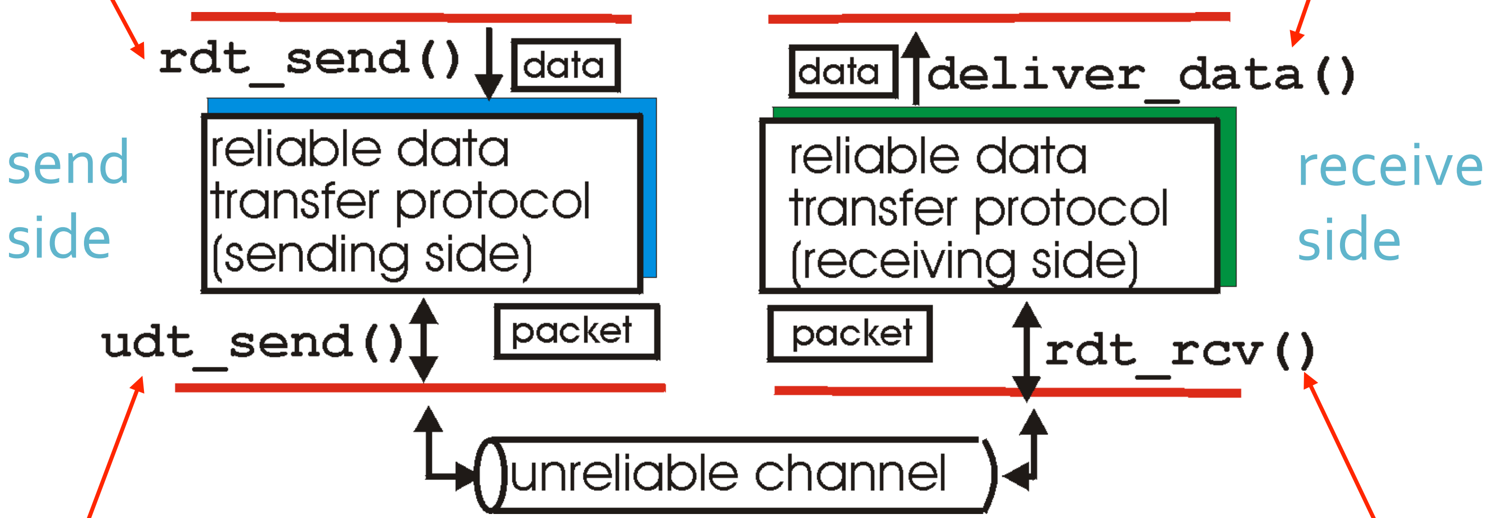


- Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

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

deliver_data() : called by **rdt** to deliver data to app.

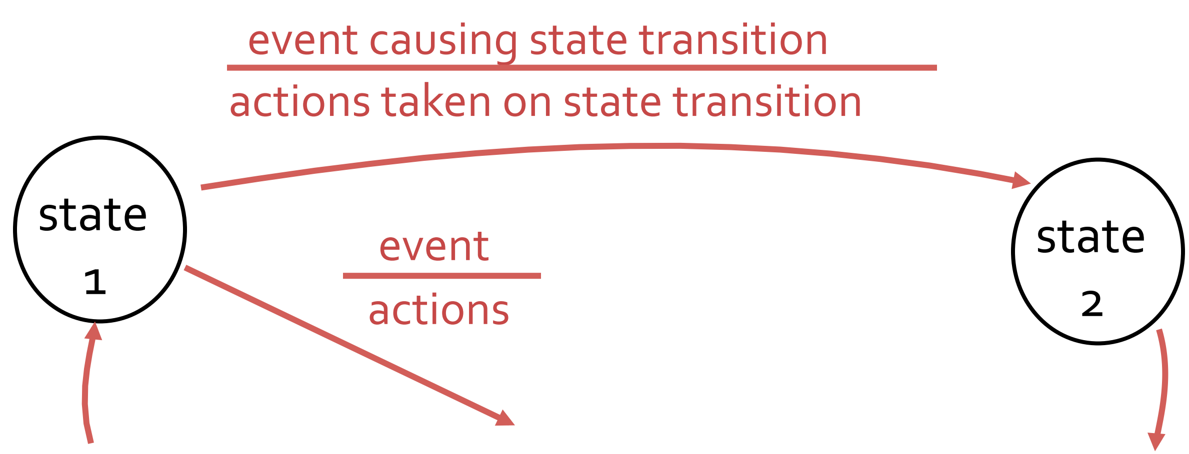


udt_send() : called by rdt, to transfer packet over unreliable channel to receiver

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

Intro to Reliable Data Transfer

- The plan: **Incrementally** develop sender / receiver sides of **reliable data transfer protocol (rdt), a fictional protocol**
  - **TCP is similar to RDT but too complex to describe all at once**
- Consider only unidirectional data transfer
  - but control info will flow on both directions!
- Use **finite state machines (FSM)** to specify sender, receiver

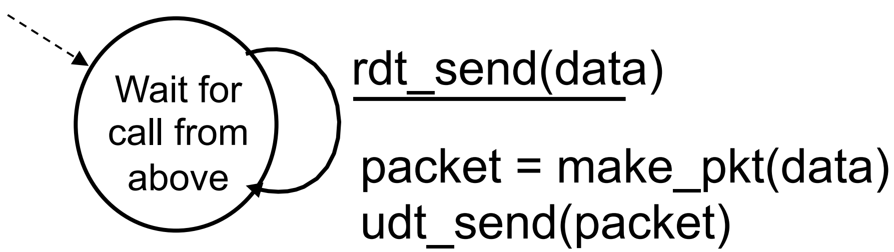


State: When in this "state", next state uniquely determined by next event

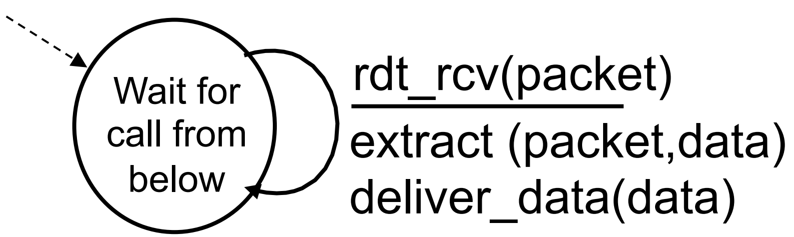
Λ (uppercase Lambda = empty set)

rdt1.0: Reliable Transfer Over a Reliable Channel

- Underlying channel perfectly reliable
  - No bit errors
  - No loss of packets
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying channel
  - Receiver reads data from underlying channel



sender

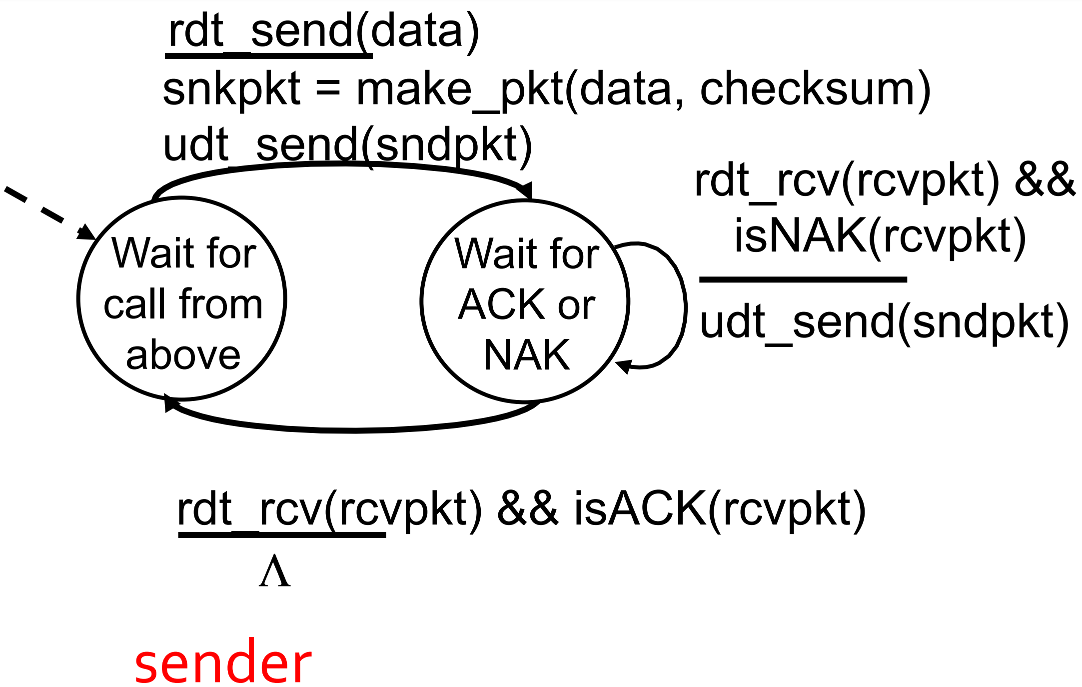


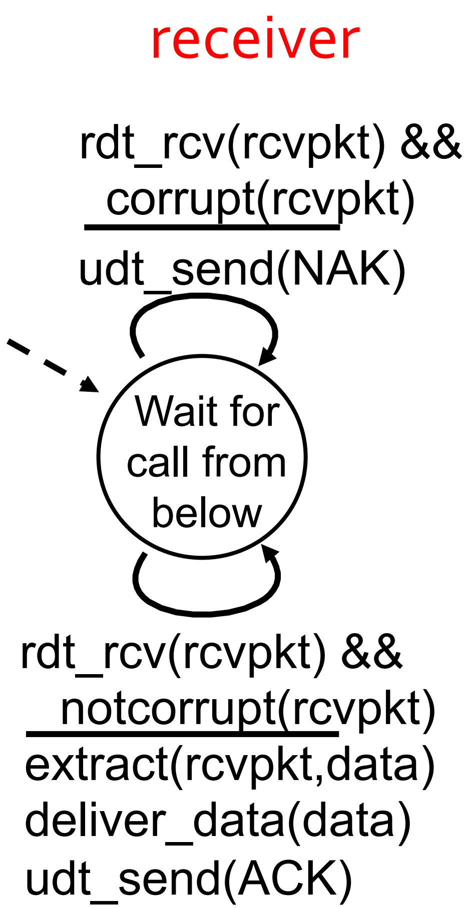
receiver

rdt2.0: Channel with Bit Errors

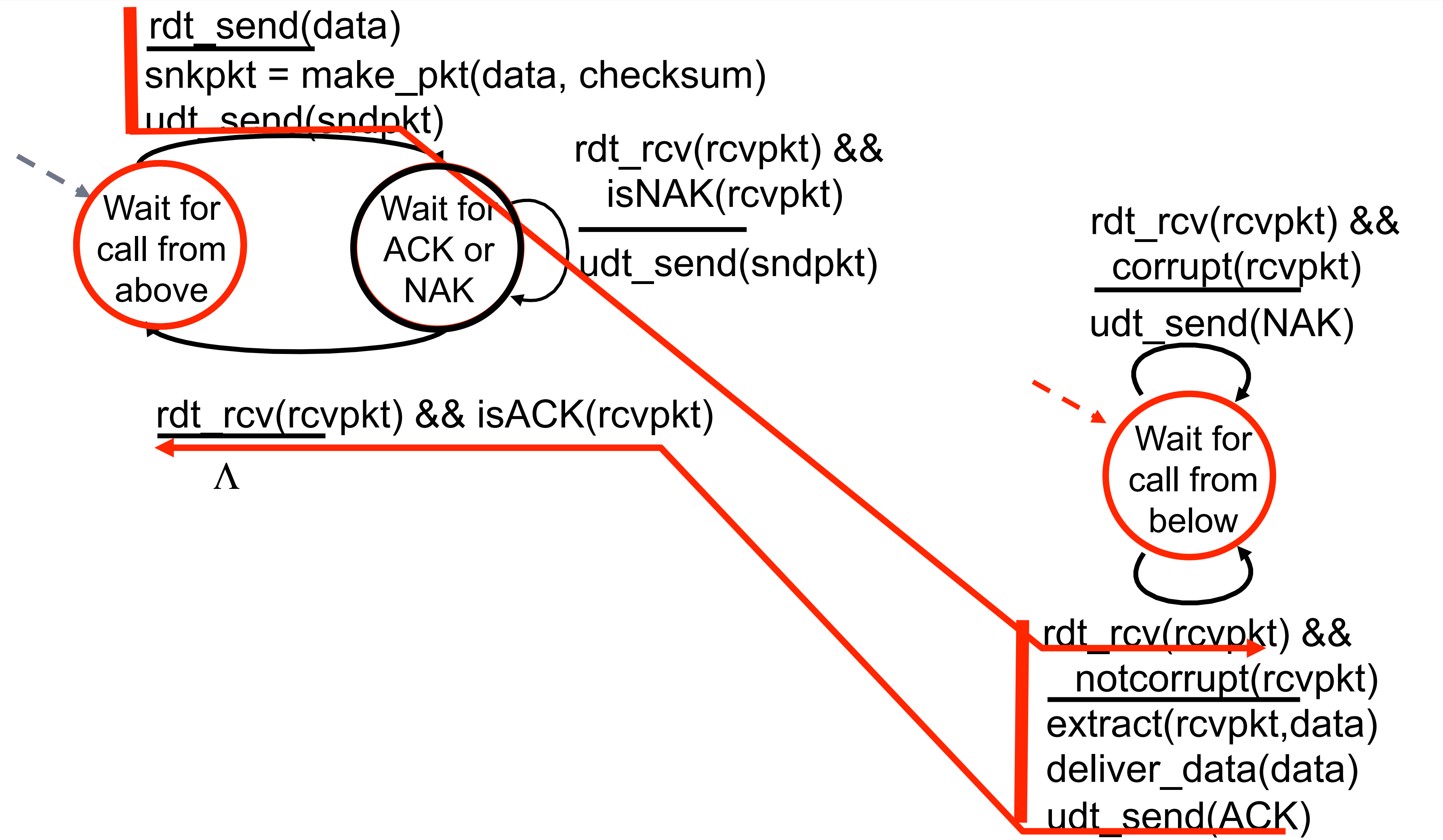
- Underlying channel may **flip bits** in packet
  - Checksum to detect bit errors
- But, how do we recover from errors?
  - **Acknowledgements (ACKs)**: receiver explicitly tells sender that packet received OK
  - **Negative acknowledgements (NAKs):** receiver explicitly tells sender that packet had errors
  - Sender retransmits packet on receipt of NAK
- New mechanisms in `rdt2.0` (beyond `rdt1.0`):
  - Error detection
  - Receiver feedback
    - Control msgs (ACK,NAK) go from receiver to sender

rdt2.0: FSM specification

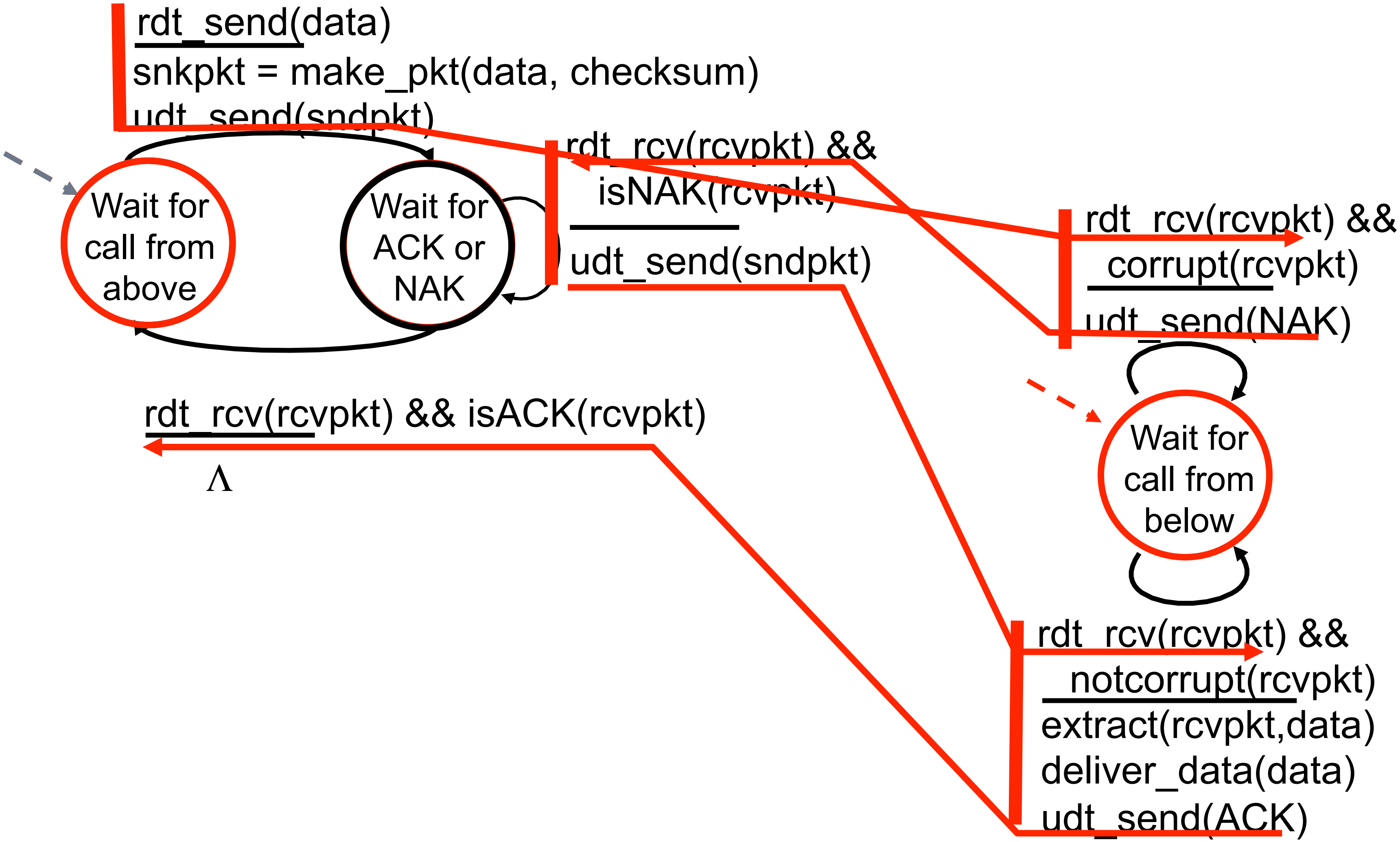




rdt2.0: Operation with No Errors



rdt2.0: Error Scenario



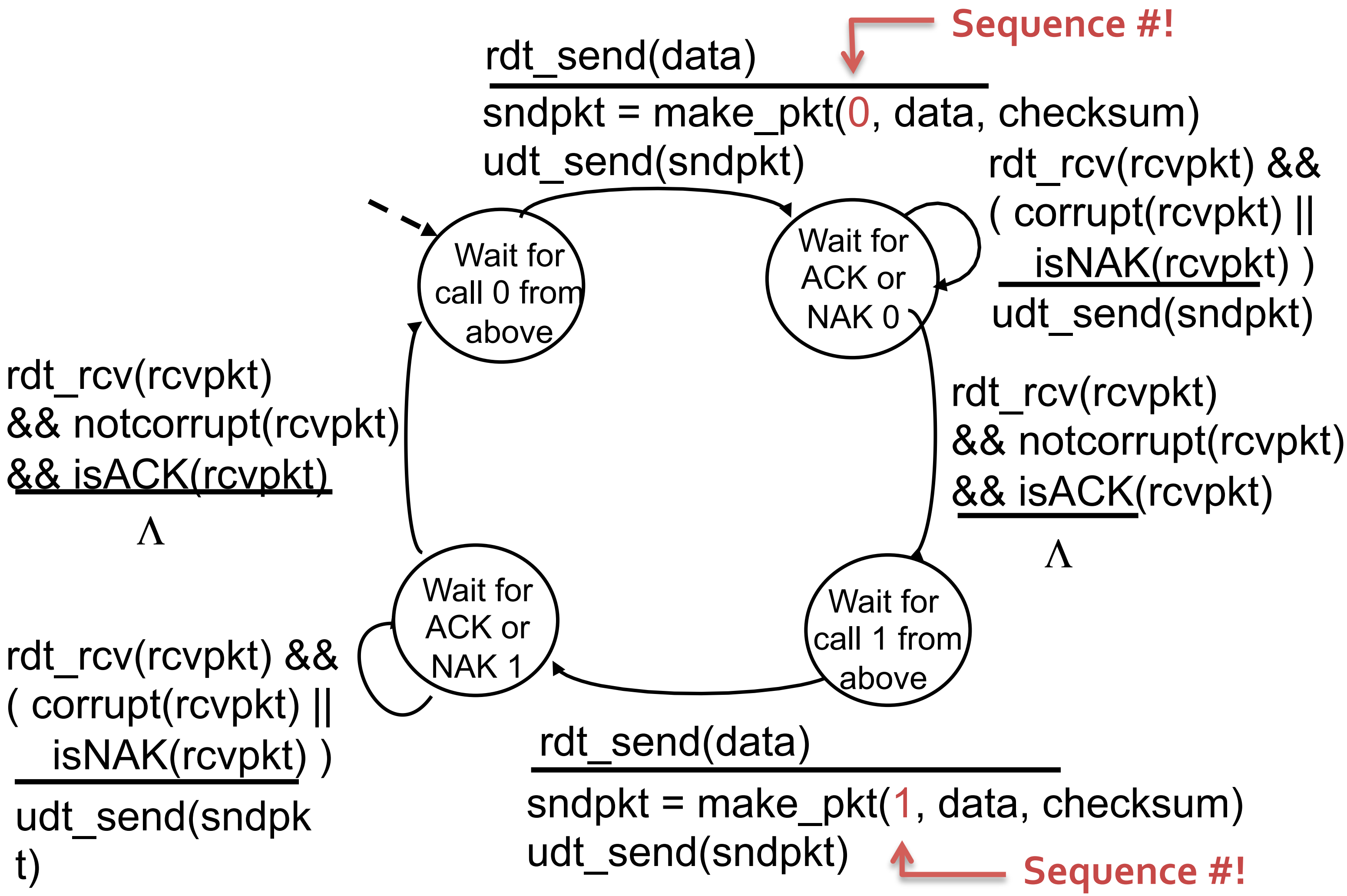
rdt2.0 has a Fatal Flaw!

- What happens if **ACK/NAK is corrupted**?
  - Sender doesn't know what happened at receiver!
- Can't just retransmit
  - Receiver might get duplicate data

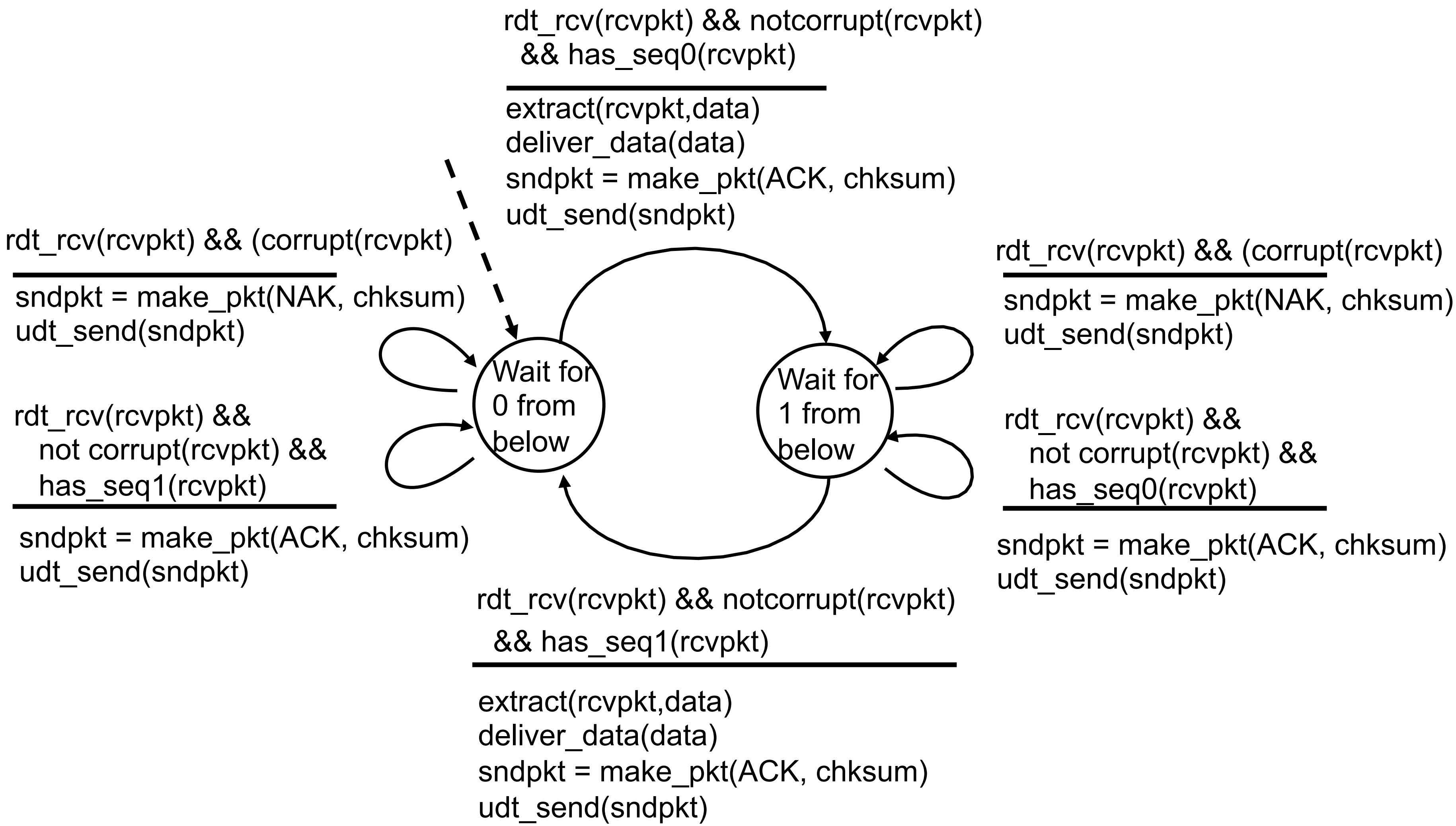
- Handling duplicates:
  - Sender retransmits current packet if ACK/NAK garbled
  - Sender adds **sequence number** to each packet
  - Receiver discards (doesn't deliver) duplicate packet

- **Stop and wait design**
  - Sender sends 1 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" packet has sequence number of 0 or 1

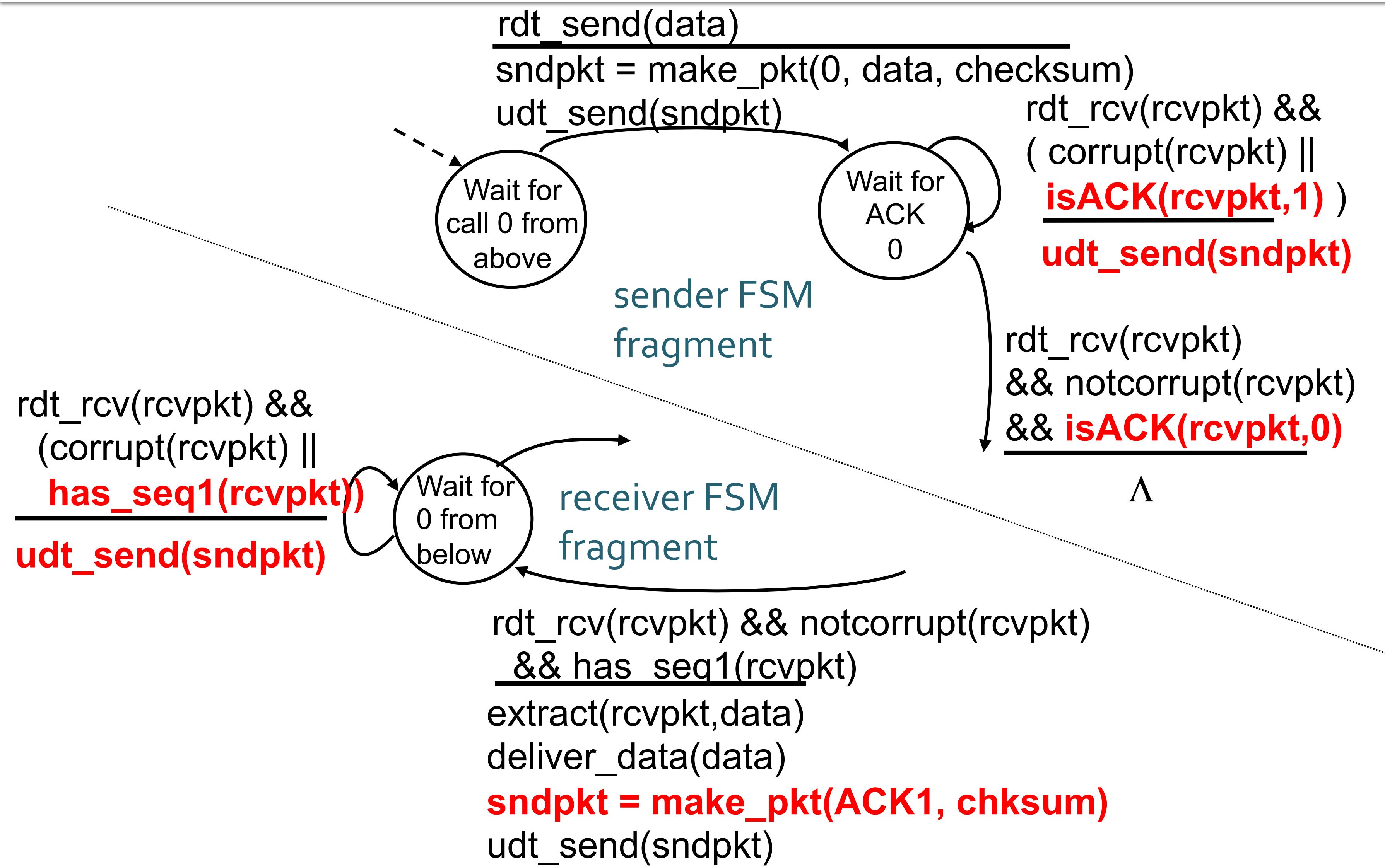
Receiver:

- Must check if received packet is duplicate
  - State indicates whether 0 or 1 is expected packet sequence number
- Receiver can *not* know if its last ACK/NAK received OK at sender
  - *Packet corruption can affect ACK/NAK packets…*

rdt2.2: a NAK-free protocol

- Same *functionality* as rdt2.1
- No NAKs!
  - Receiver instead sends ACK for last packet received OK
  - Receiver must explicitly include seq # of packet being ACKed
- Duplicate ACK at sender results in same action as NAK
  - Retransmit current packet

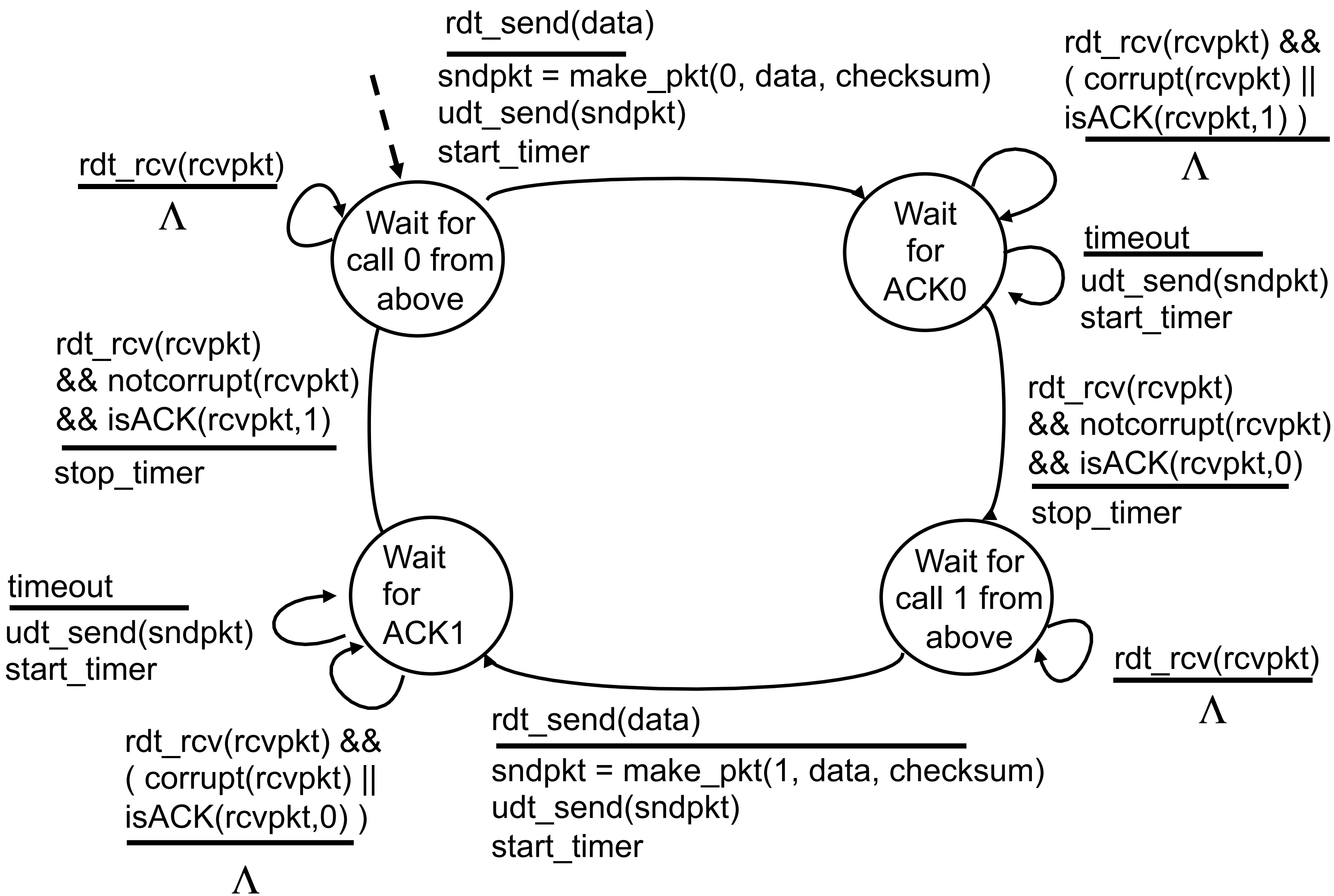
rdt2.2: Partial Sender and Receiver



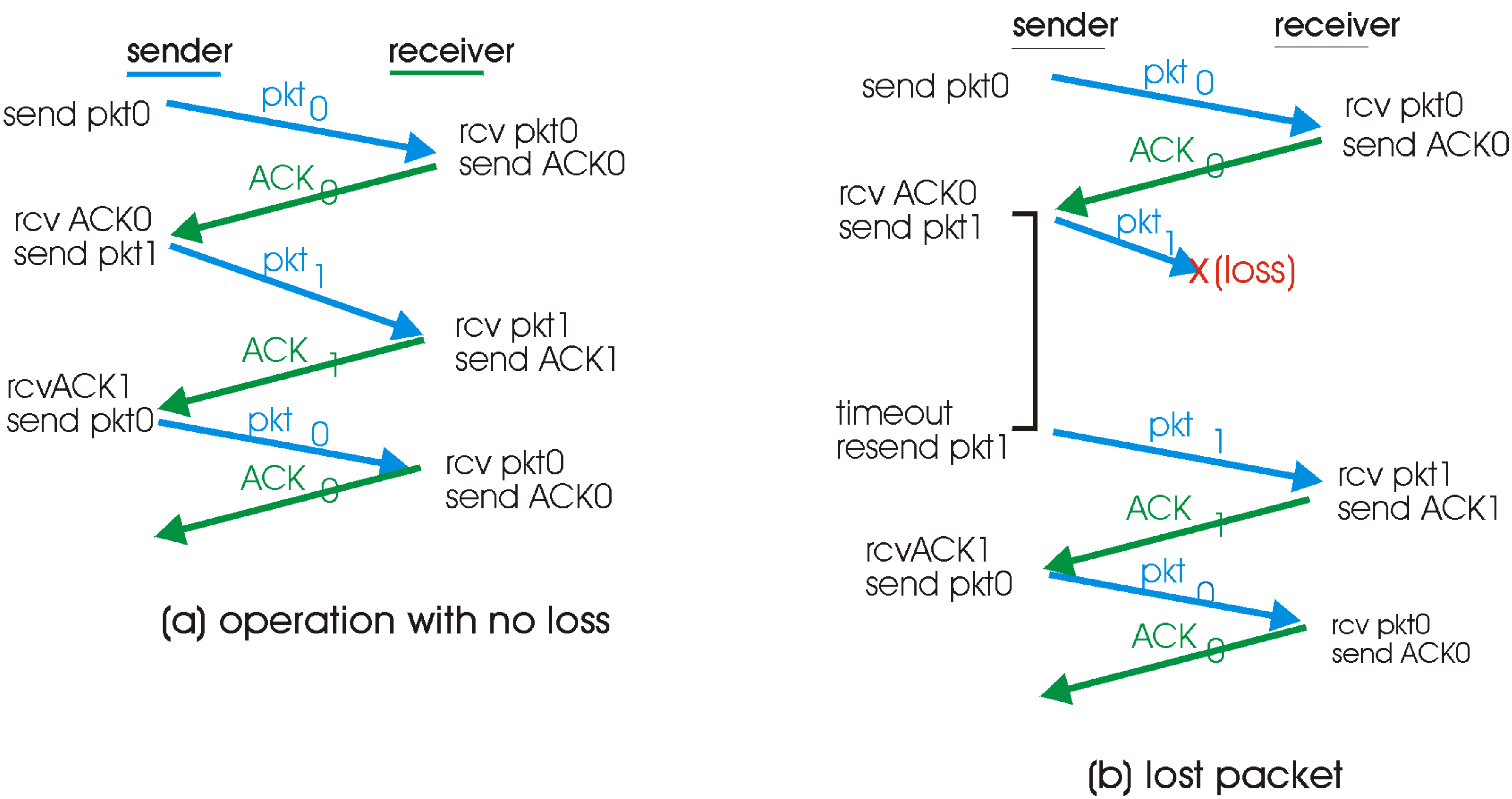
rdt3.0: Channels with Errors and Loss

- New assumption
  - Underlying channel can also **lose packets** (data **<u>or</u>** ACKs)
  - Checksum, seq. #, ACKs, and retransmissions will help but are not sufficient
- New approach
  - Sender waits "reasonable" amount of time for ACK
  - Retransmits if no ACK received in this time
  - If pkt (or ACK) is just delayed but not lost:
    - Retransmission will be duplicate, but seq. #'s solves this problem
    - Receiver must specify seq # of pkt being ACKed
  - Requires countdown timer

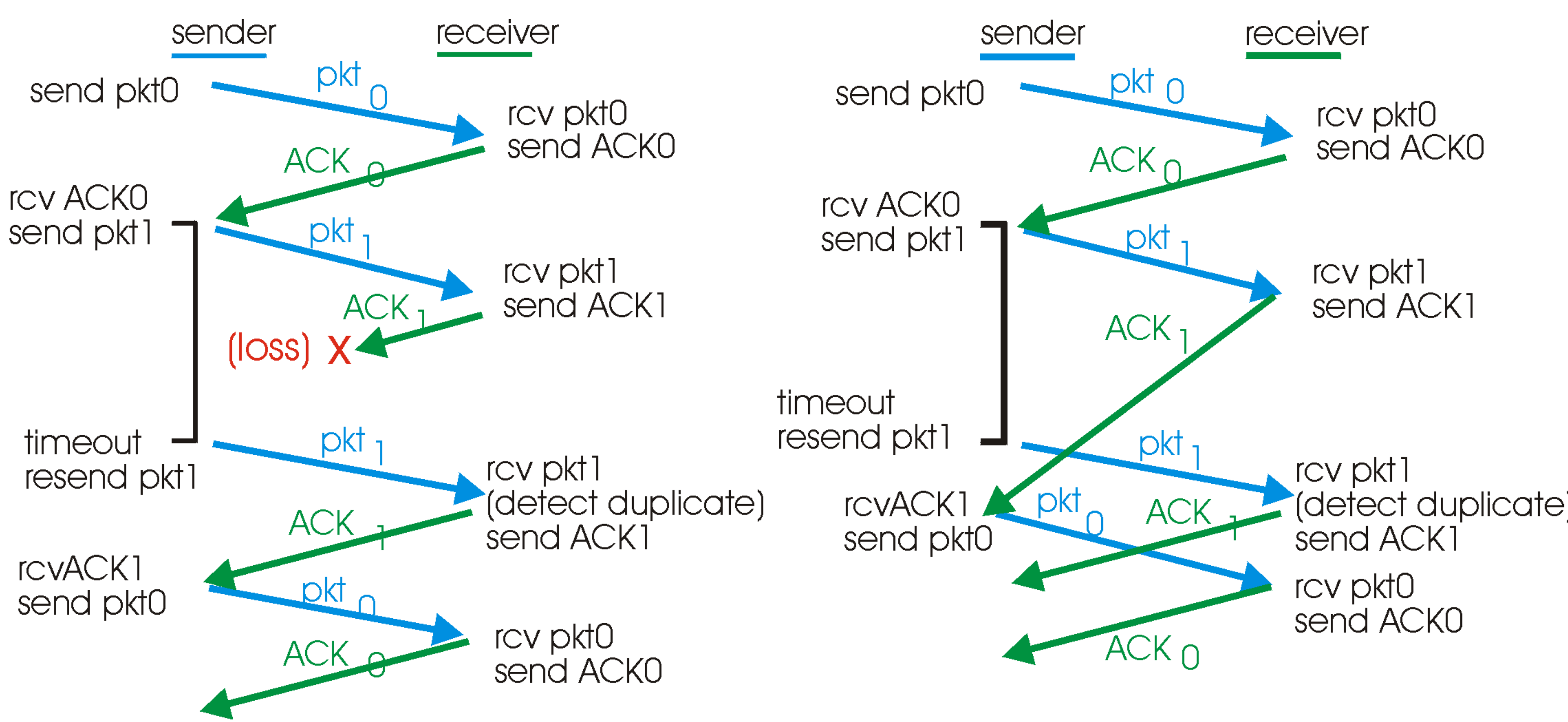
rdt3.0 Sender



rdt3.0 in Action



rdt3.0 in Action



(c) lost ACK

(d) premature timeout

Performance of rdt3.0

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

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

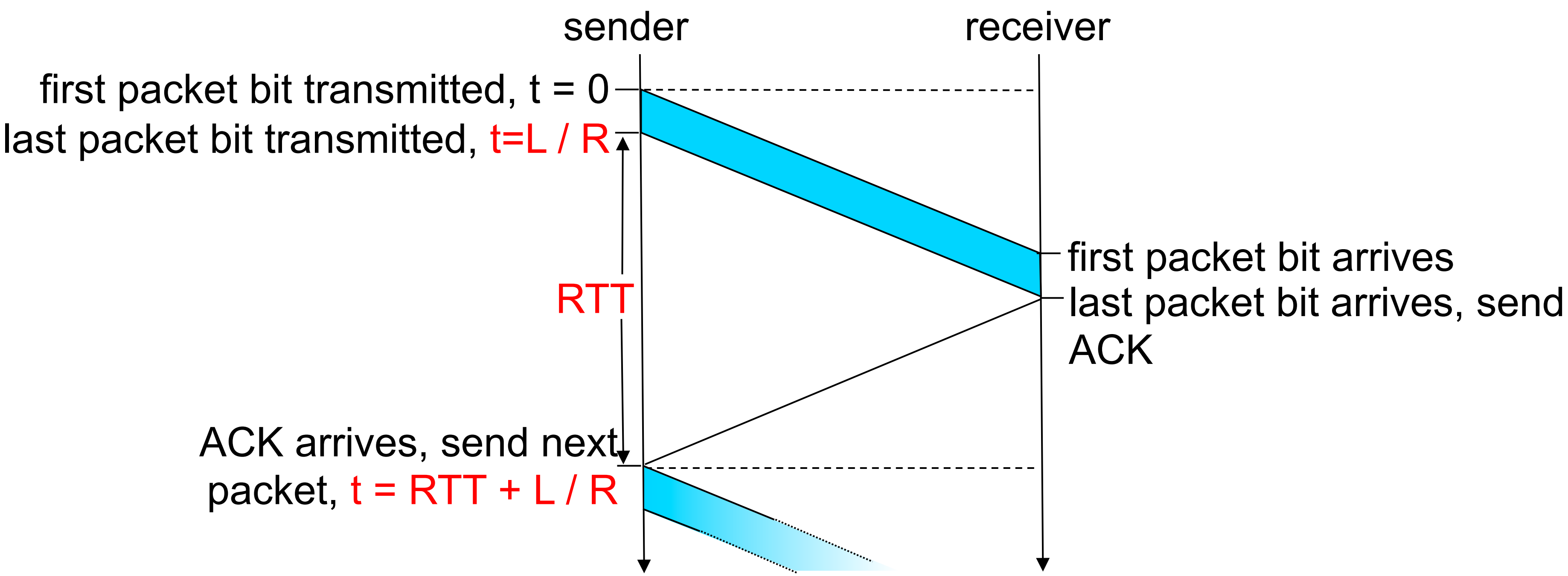
How long it takes to push packet out onto wire

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

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

- 1KB packet every 30 msec
  - **33kB/sec throughput over 1 Gbps link**
  - **Network protocol limits use of physical resources!**

rdt3.0: Stop-and-Wait Operation

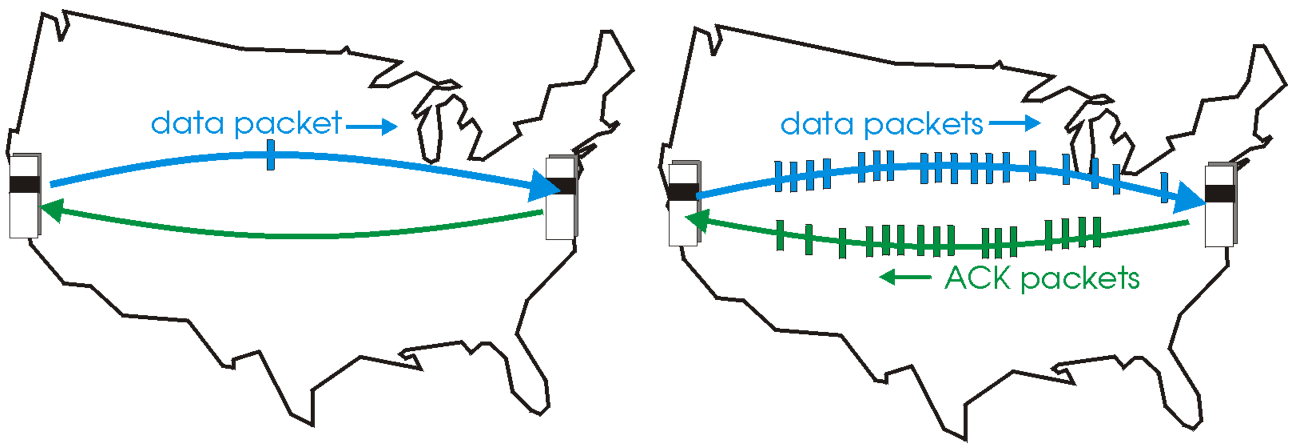


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

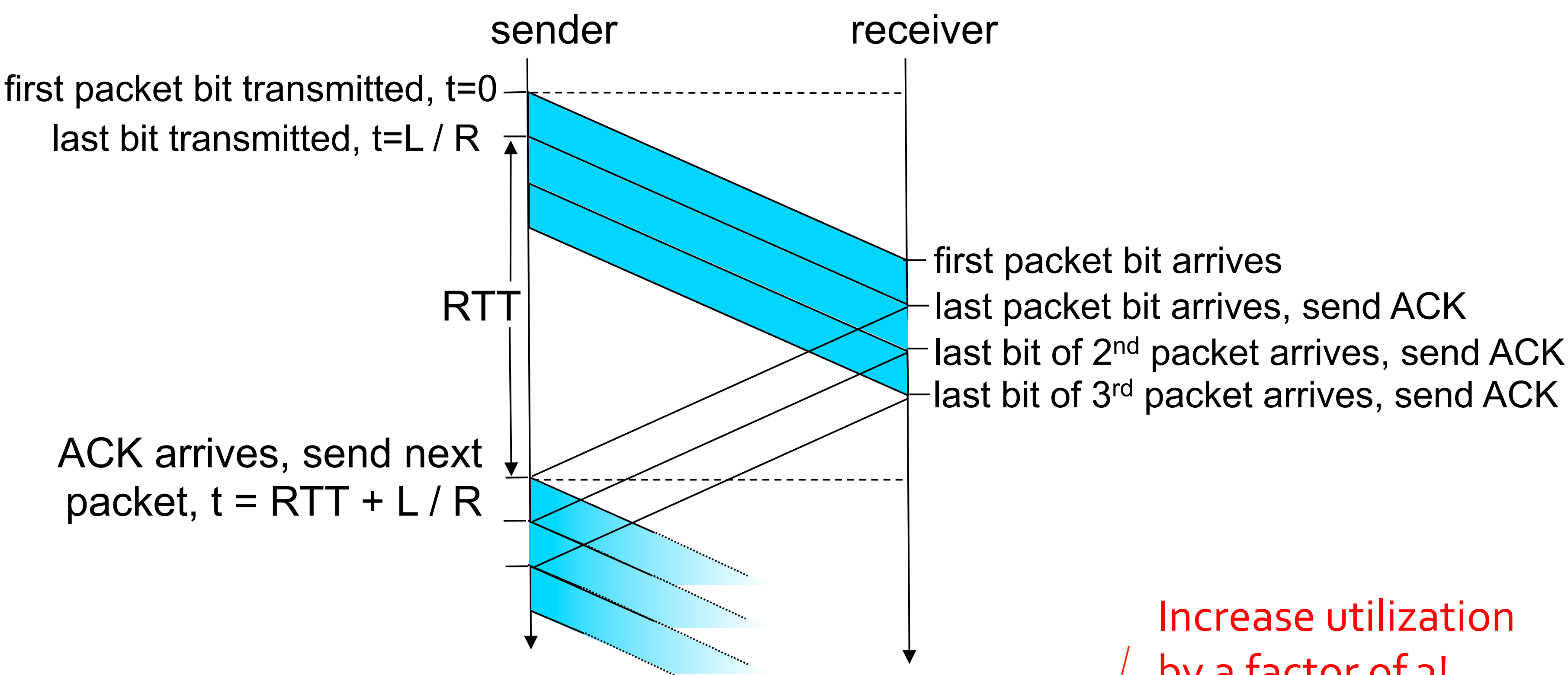
- Range of sequence numbers must be increased
- Buffering at sender and/or receiver



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

(b) a pipelined protocol in operation

Pipelining: Increased Utilization



Increase utilization by a factor of 3!

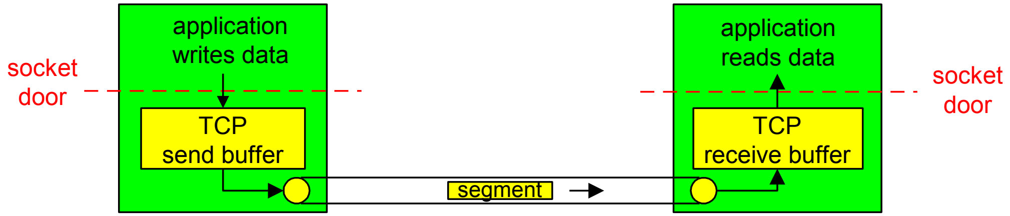
$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

TCP – Transmission Control Protocol

TCP: Overview

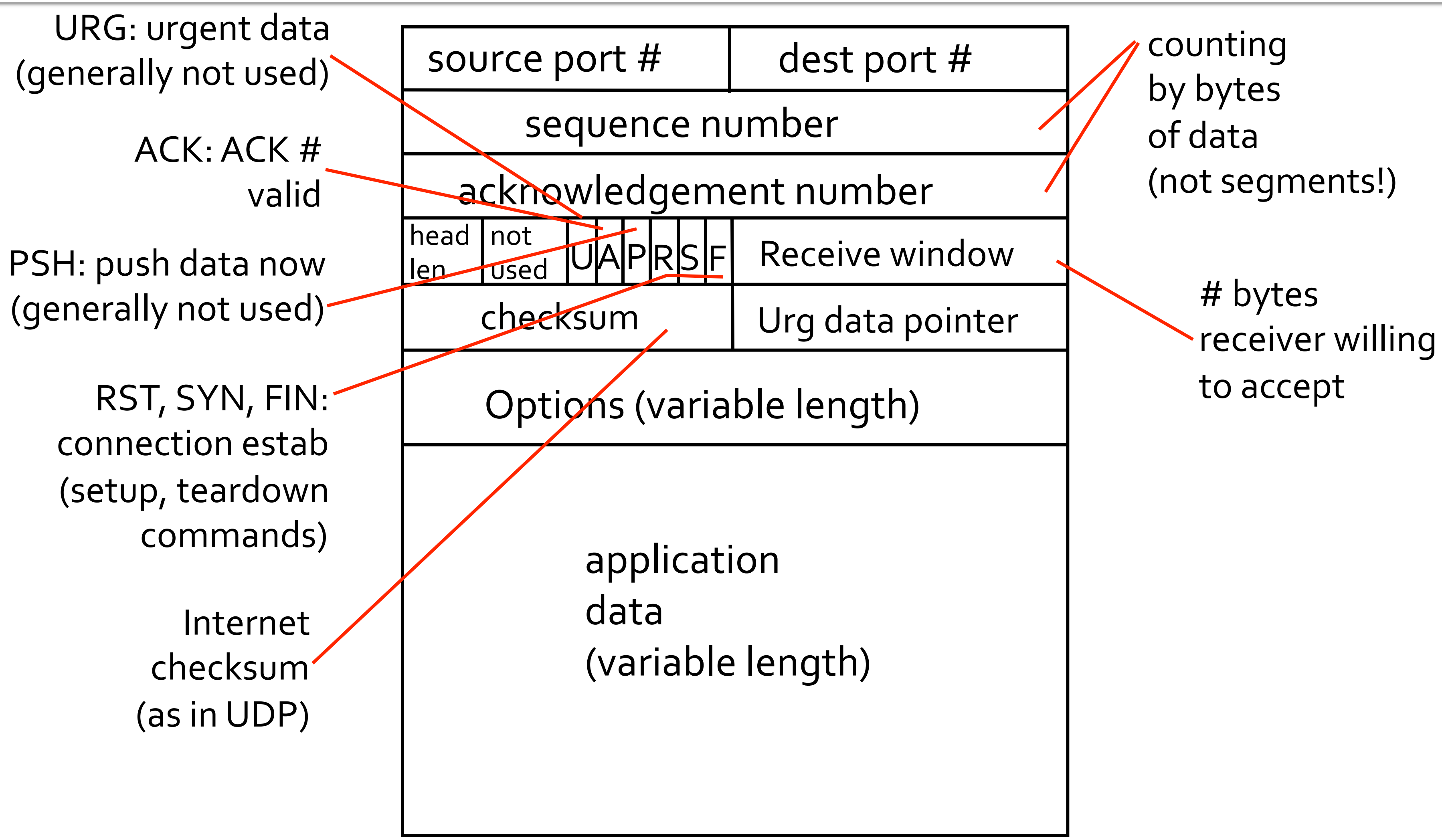
RFCs: 793, 1122, 1323, 2018, 2581

- 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) initializes sender, receiver state before data exchange
- Flow controlled:
  - Sender will not overwhelm receiver

TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

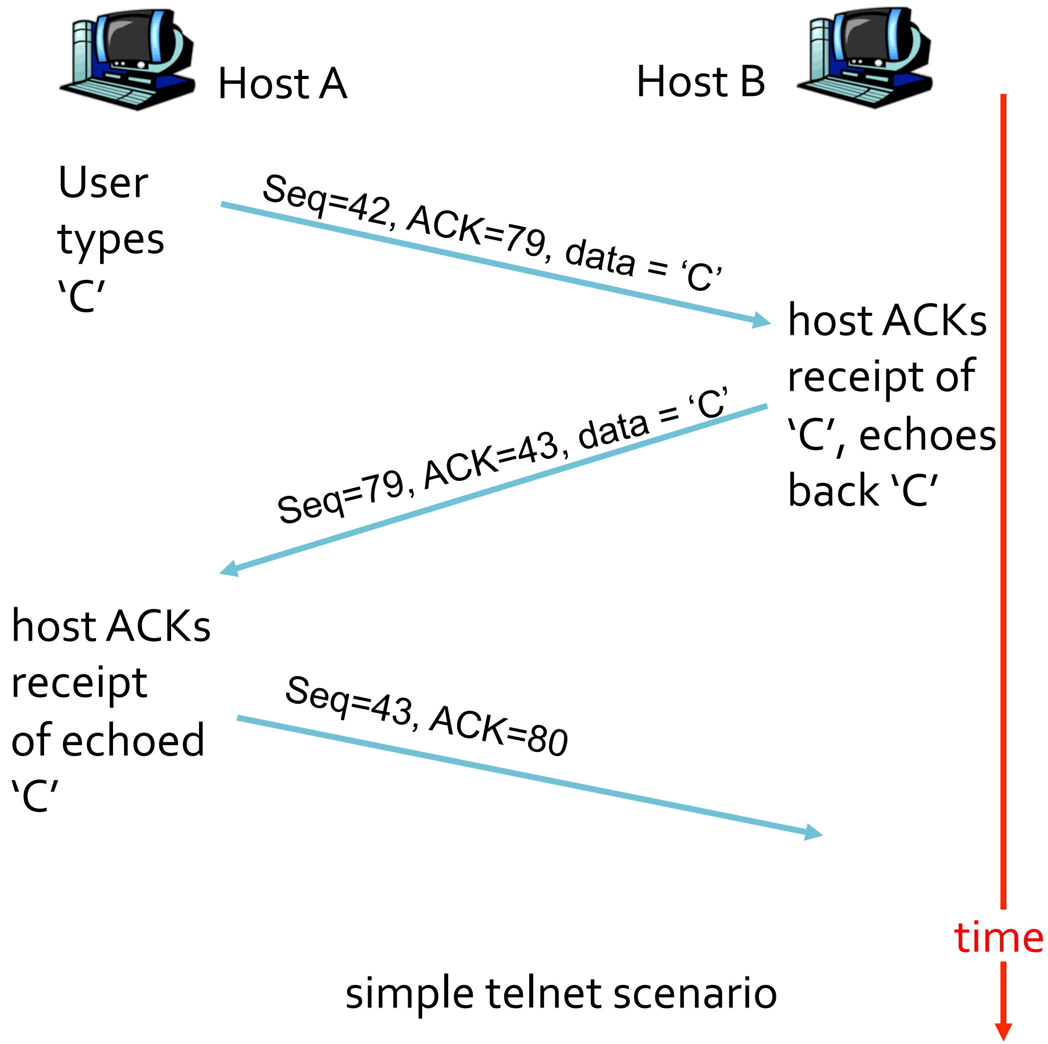
- Byte stream "number" of first byte in segment's data

ACKs:

- Seq # of next byte expected from other side
- Cumulative ACK

How does receiver handle out-of-order segments?

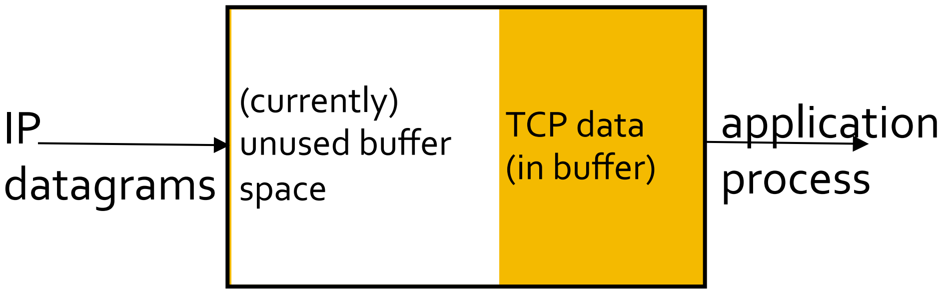
- TCP spec doesn't say, - up to implementer



simple telnet scenario

TCP Flow Control

- Receive side of TCP connection has a receive buffer:



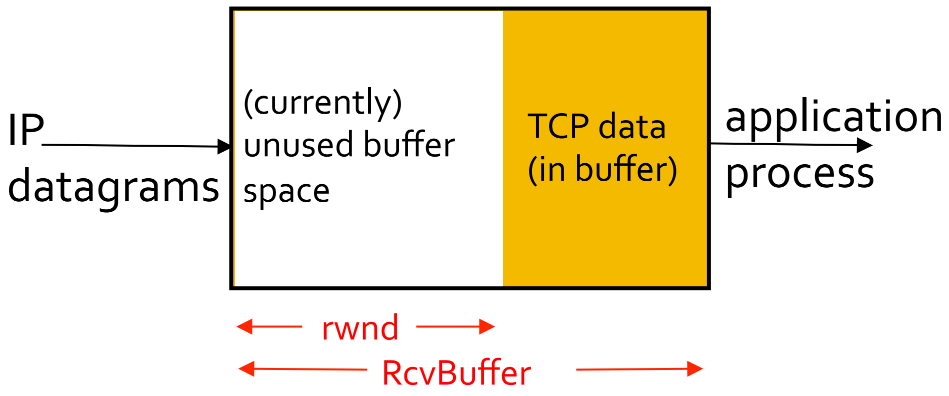
- Application process may be slow at reading from buffer
  - What if buffer fills up?

Flow Control:

Prevents **sender** from **overflowing receiver's buffer** by transmitting too much, too fast

Speed matching service: matching send rate to receiving application's drain rate

TCP Flow Control: How it Works



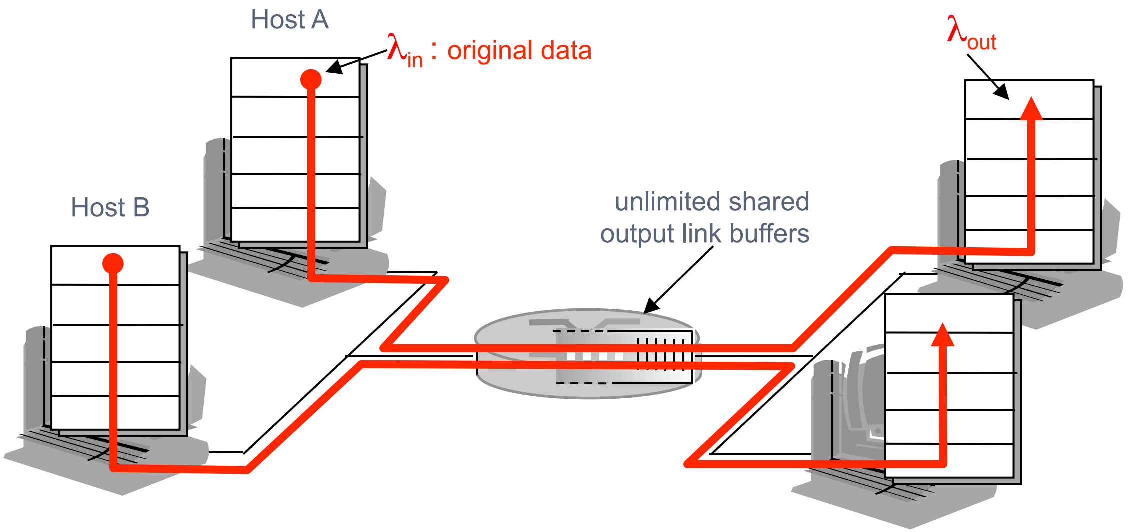
- Suppose TCP receiver discards out-of-order segments…
- Unused buffer space = rwnd
  = RcvBuffer-[LastByteRcvd - LastByteRead]
- Receiver notifies sender of unused buffer space
  - Segment header includes the `rwnd` value
- Sender limits # of unACKed bytes to `rwnd`
  - Guarantees receiver's buffer doesn't overflow

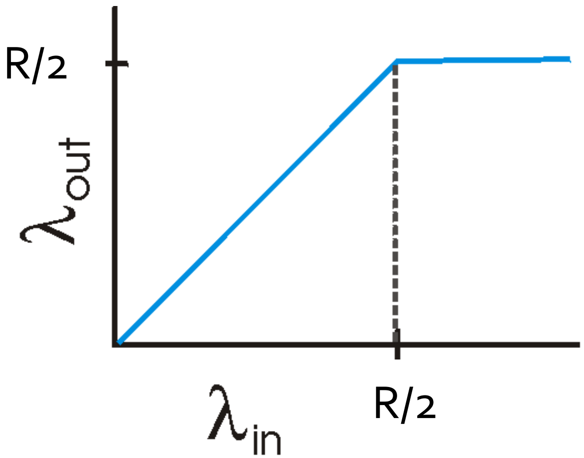
Principles of Congestion Control

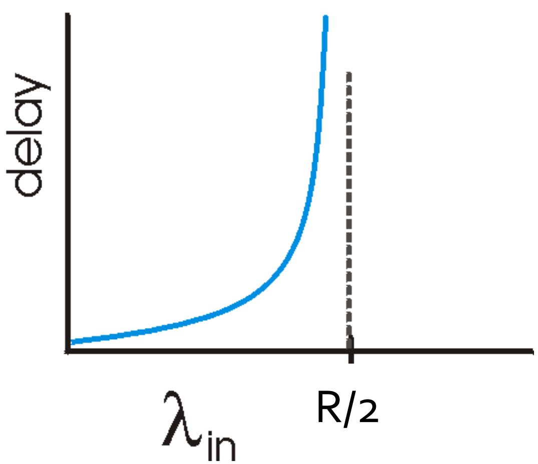
- What is congestion?
  - Informally: "too many sources sending too much data too fast for **network** to handle"
- Different from flow control!
- Manifestations
  - Lost packets (buffer overflow at routers)
  - Long delays (queueing in router buffers)

Causes/costs of Congestion: Scenario 1

- Two senders, two receivers
- One router, **infinite** buffers
- No retransmission
- Link BW of R



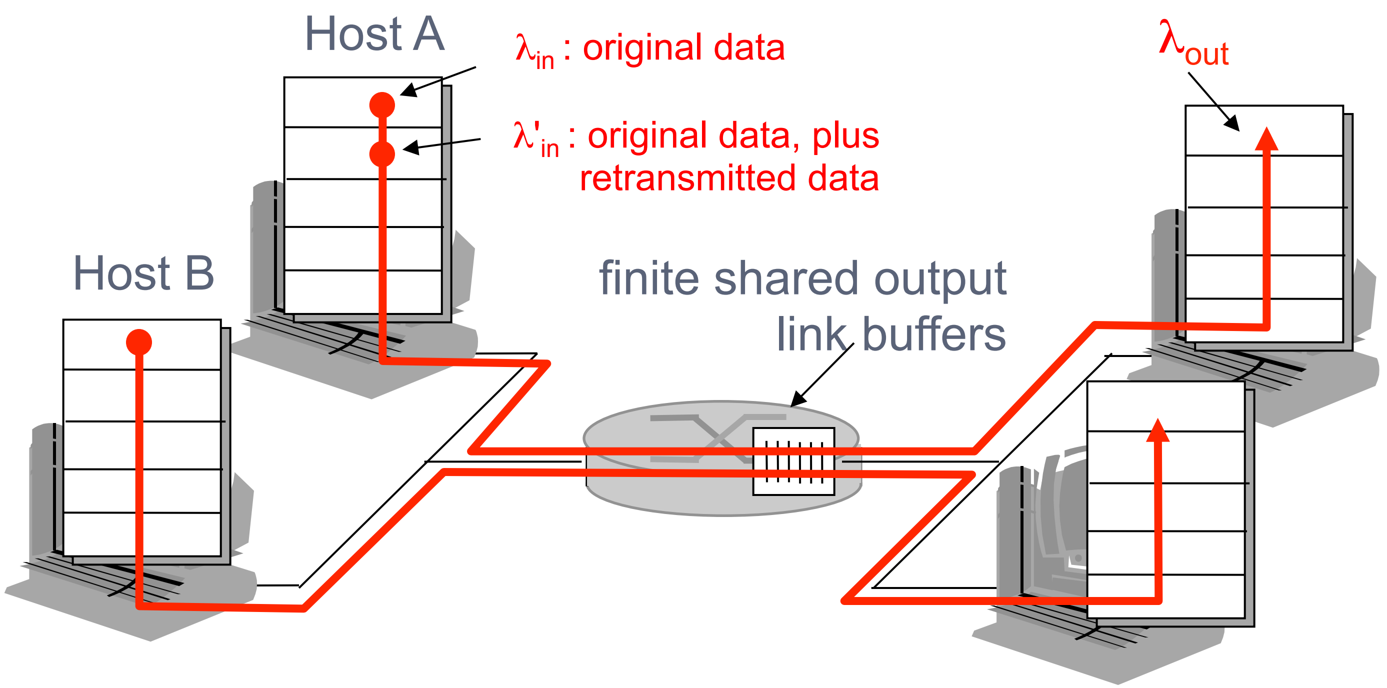




- **Large delays** when congested
- Maximum achievable throughput

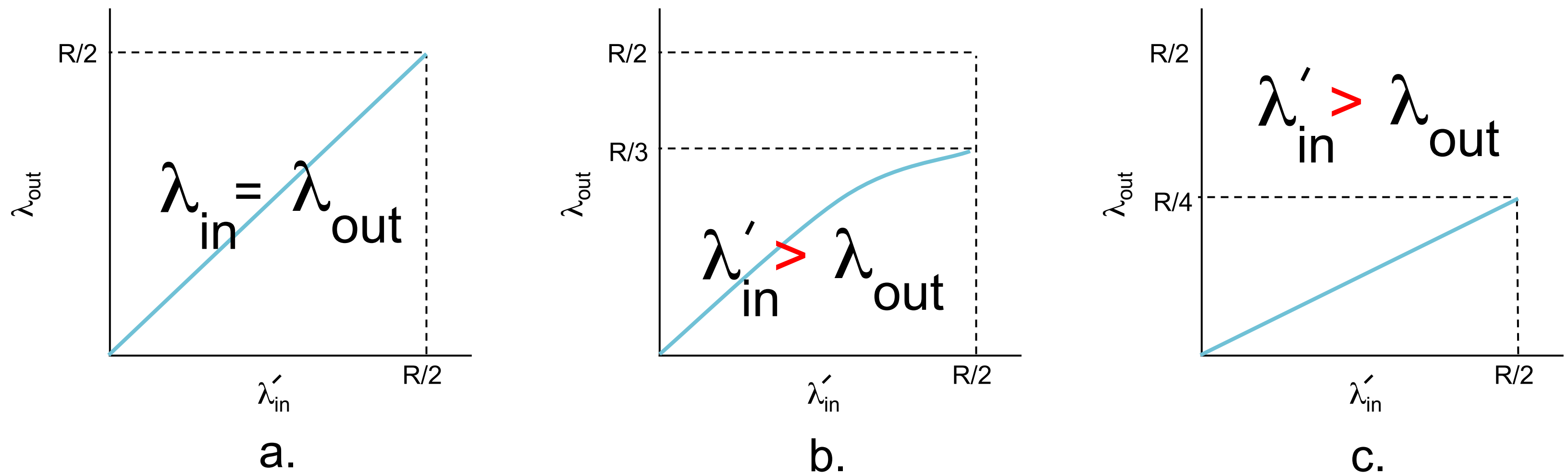
Causes/costs of Congestion: Scenario 2

- One router, **finite** buffers
- Sender retransmission of lost packet



Causes/costs of Congestion: Scenario 2

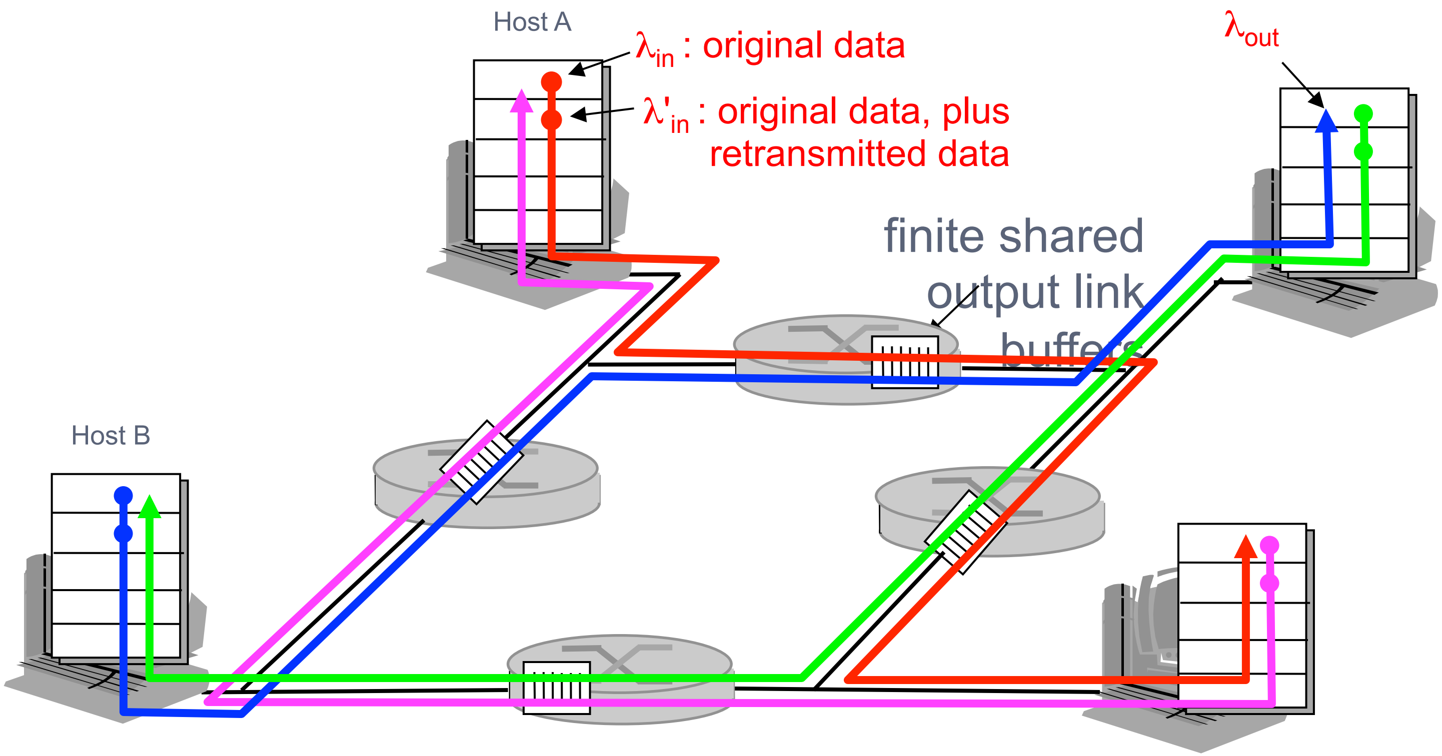
- Case a: Sender only transmits when it knows buffer space is available in router (unrealistic)
- Case b: Sender retransmits only when packet is known to be lost
  - New cost of congestion: More sender work (retrans) for given "goodput"
- Case c: Assume sender also retransmits when a packet is delayed (not lost), i.e. a premature timeout (bigger λ_{in}')
  - New cost of congestion: router output link carries multiple copies of packet



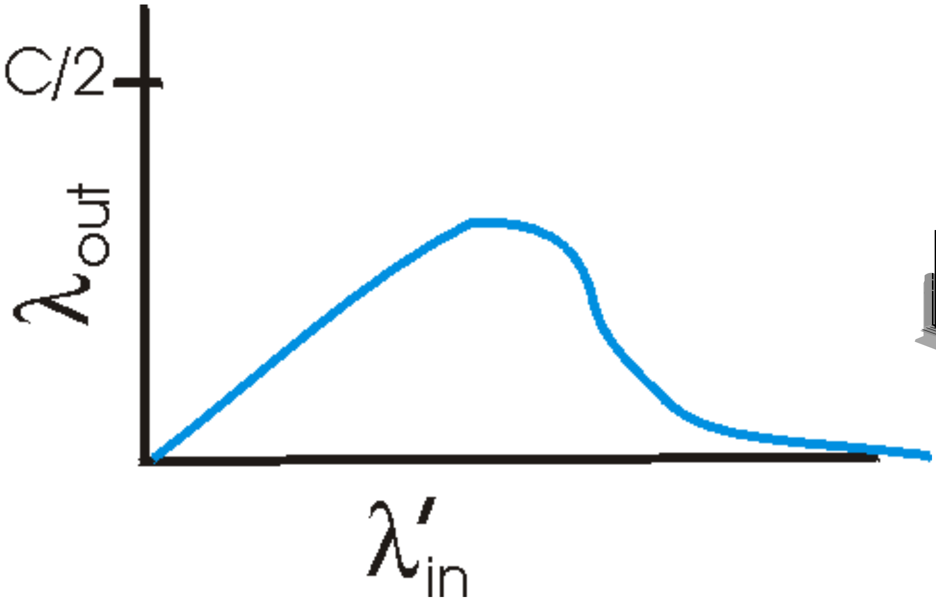
Causes/costs of Congestion: Scenario 3

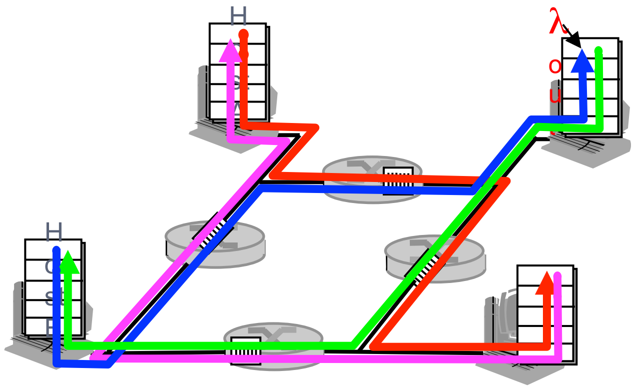
- Four senders
- Multihop paths
- Timeout/retransmit

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



Causes/costs of Congestion: Scenario 3





- A new cost of congestion
  - When packet dropped, any upstream transmission capacity used for that packet was wasted!

Congestion Control Approaches

Two broad approaches to congestion control:

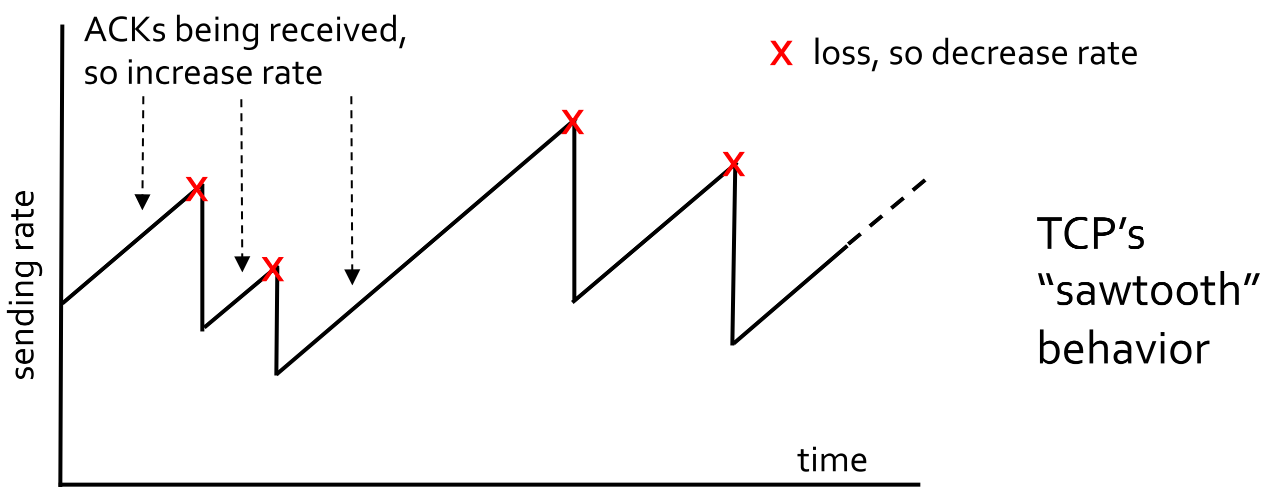
- **End-end congestion control:**
  - No explicit feedback from network
  - Congestion inferred from end-system observed packet loss and delay
  - **Approach taken by TCP**
- **Network-assisted congestion control:**
  - Routers provide feedback to end systems
  - Single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - Explicit rate sender should send at

TCP Congestion Control

- Goal: TCP sender should transmit **as fast as possible**, but without congesting network
- How do we find the rate just below congestion level?
  - Decentralized approach – each TCP sender sets its own rate, based on *implicit* feedback:
  - ACK indicates segment received (a good thing!)
    - Network not congested, so increase sending rate
  - Lost segment – <u>assume</u> loss is due to congested network, so decrease sending rate

TCP Congestion Control: Bandwidth Probing

- Probing for bandwidth
  - Increase transmission rate on receipt of ACK, until eventually loss occurs, then decrease transmission rate



- How fast to increase or decrease?

Summary

User Datagram Protocol (UDP) Characteristics

- UDP is a connectionless datagram service.
  - There is no connection establishment: packets may show up at any time.
- UDP packets are self-contained.
- UDP is unreliable:
  - No acknowledgements to indicate delivery of data.
  - Checksums cover the header, and only optionally cover the data.
  - Contains no mechanism to detect missing or mis-sequenced packets.
  - No mechanism for automatic retransmission.
  - No mechanism for flow control or congestion control (sender can overrun receiver or network)

TCP Characteristics

- TCP is connection-oriented.
  - 3-way handshake used for connection setup
- TCP provides a stream-of-bytes service
- TCP is reliable:
  - Acknowledgements indicate delivery of data
  - Checksums are used to detect corrupted data
  - Sequence numbers detect missing, or mis-sequenced data
  - Corrupted data is retransmitted after a timeout
  - Mis-sequenced data is re-sequenced
  - (Window-based) Flow control prevents over-run of receiver
- TCP uses congestion control to share network capacity among users