Transport Layer – Part I

Computer Networks, Winter 2014/2015

Chapter 4: The Transport Layer

5: Application Layer

4: Transport Layer

3: Network Layer

2: Link Layer

1: Physical Layer

Chapter 4: The 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

Transport Layer

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
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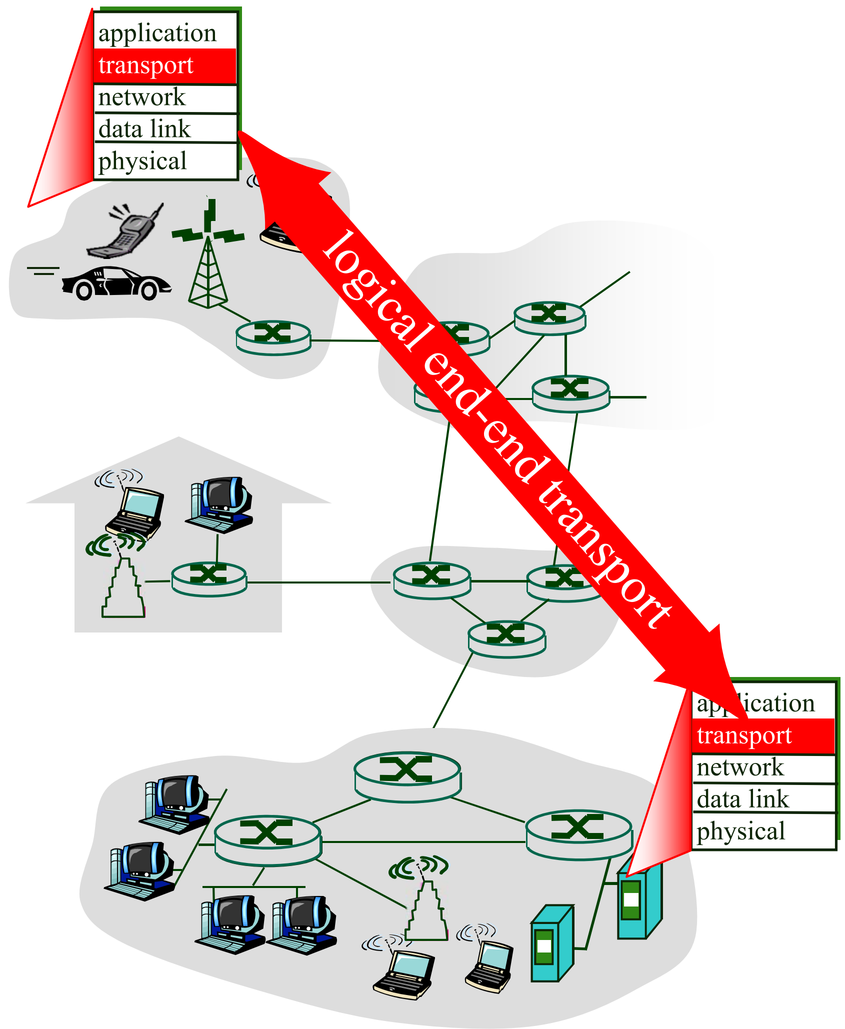
Transport services and protocols

- 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

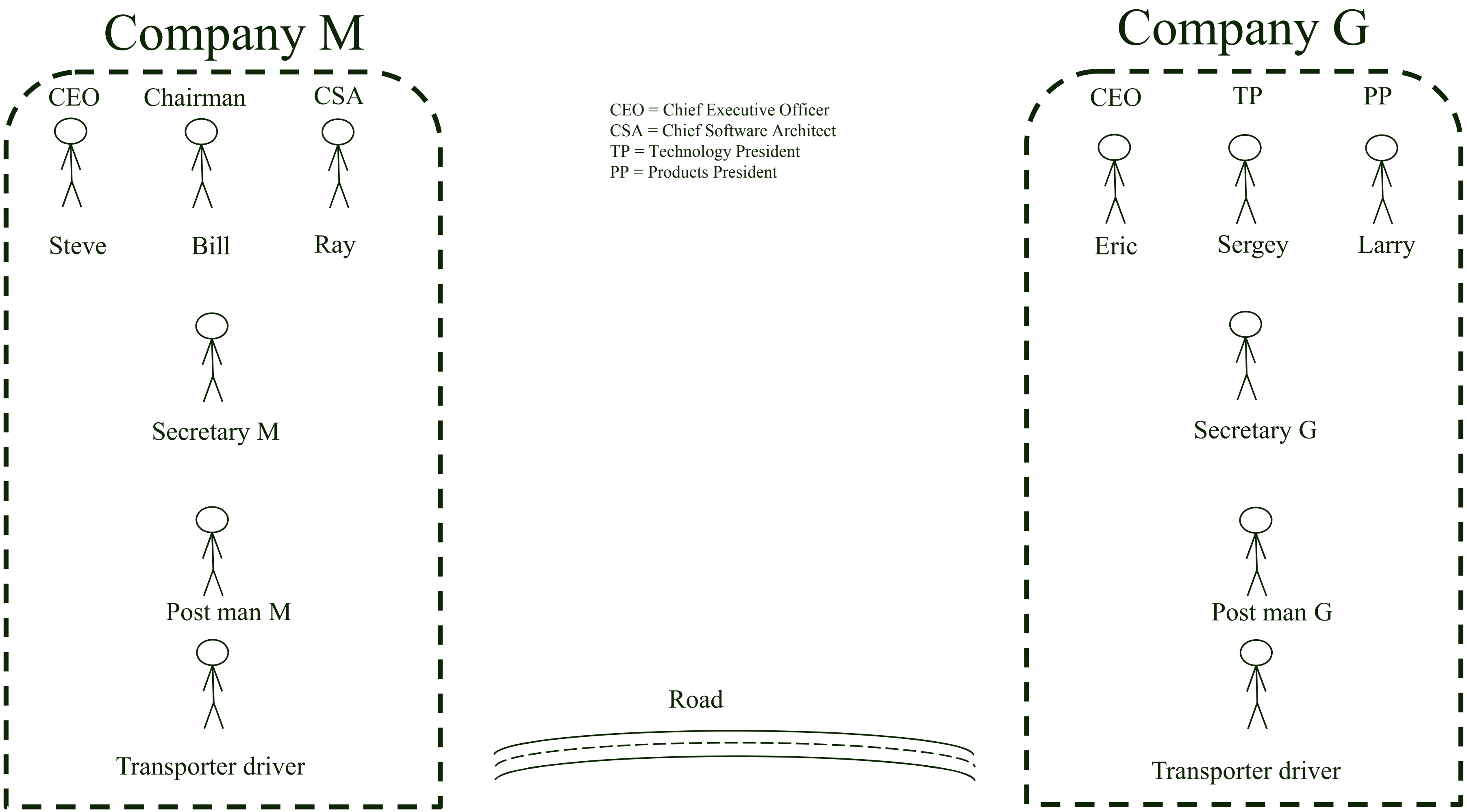


Transport vs. network layer

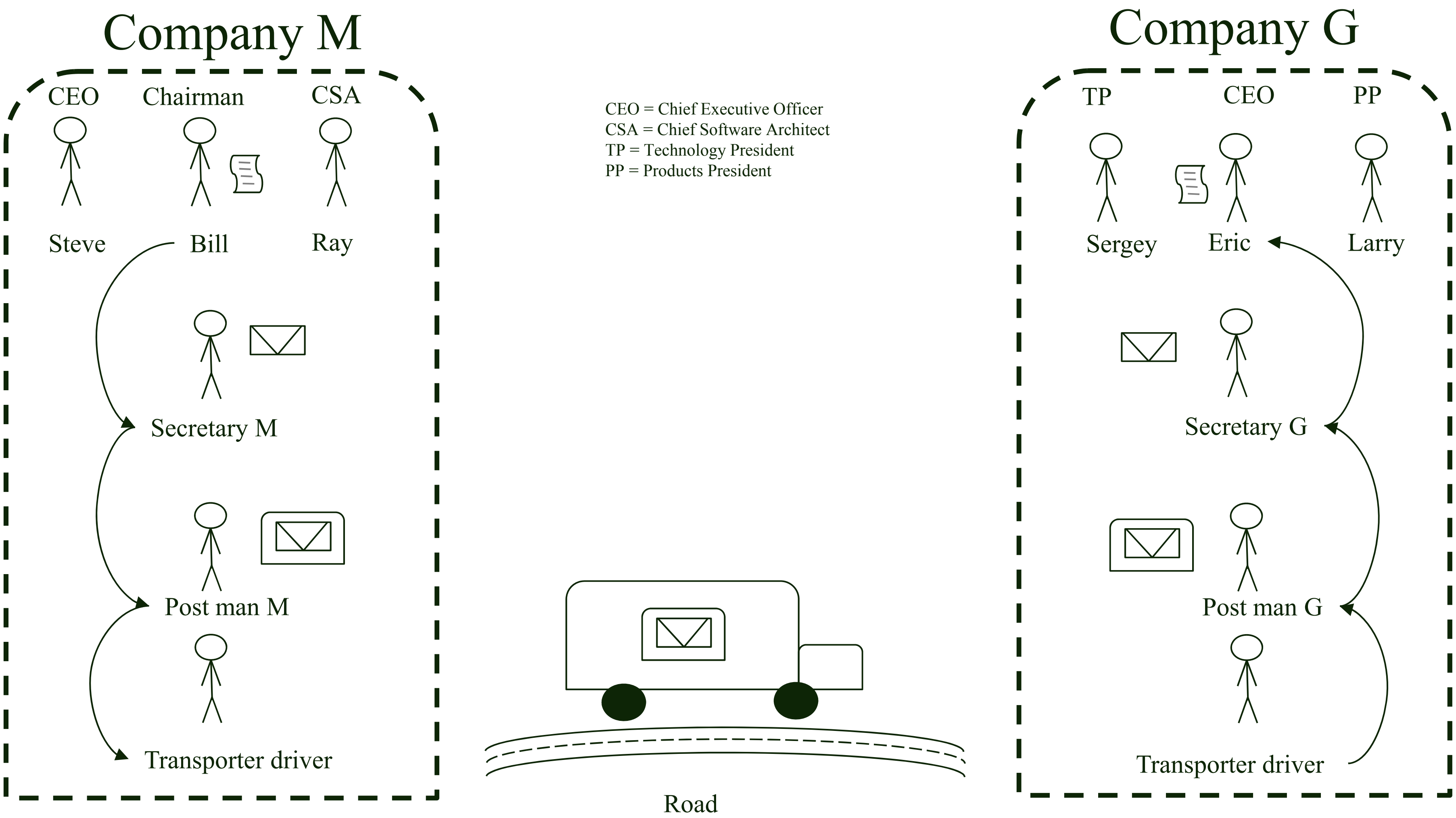
- *network layer:* logical communication between hosts

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

Transport Protocol: Analogy



Transport Protocol: Analogy

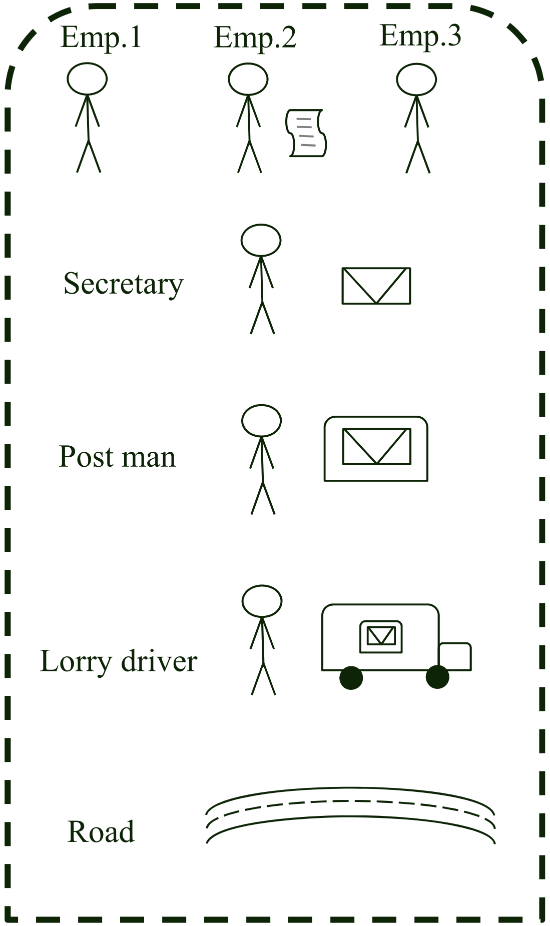


Transport Protocol: Analogy

- *Postal service (Network Layer):* logical communication between company buildings.
- *Secretary service (Transport Layer):* logical communication between employees of G und M.
  - relies on & enhances, postal services

Transport Protocol: Analogy

Company

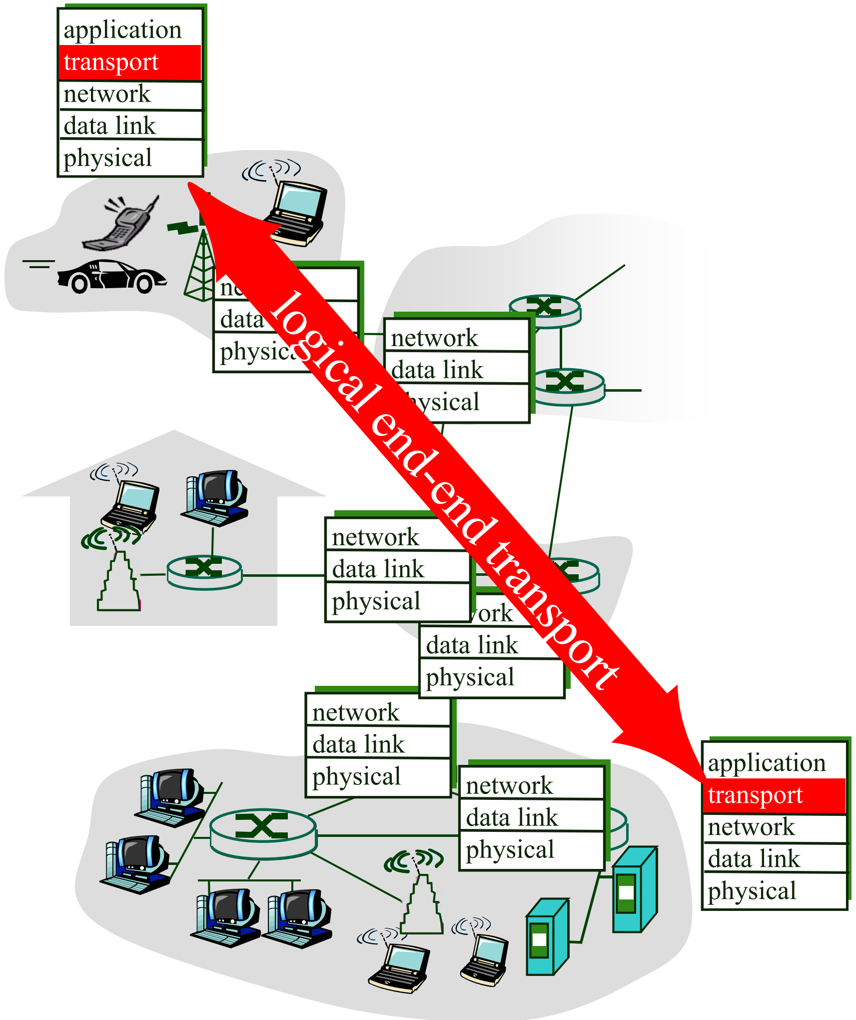


Host

- 5: App. Layer (processes)
- 4: Transport Layer (protocols)
- 3: Network Layer (protocol)
- 2: Link Layer (protocols)
- 1: Physical Layer (medium)

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



Excursus: Sockets

Socket API

- introduced in BSD4.1 UNIX, 1981
- explicitly created, used, released by apps
- client/server paradigm
- two types of transport service via socket API:
  - unreliable datagram
  - reliable, byte stream-oriented

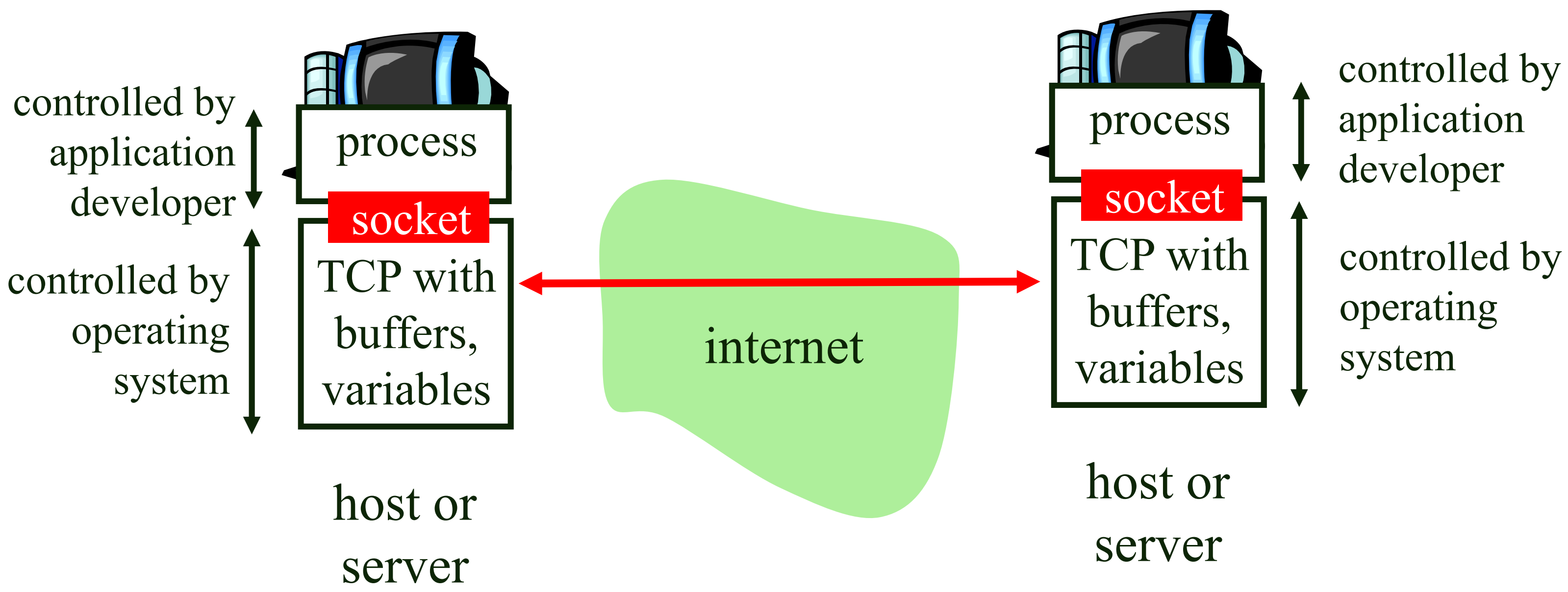
socket

a *host-local*, *application-created*, *OS-controlled* interface (a "door") into which application process can both send and receive messages to/from another application process

Excursus: Socket programming *with TCP*

Socket: a door between application process and end-end-transport protocol (UDP or TCP)

TCP service: reliable transfer of **bytes** from one process to another



Excursus: Socket programming *with TCP*

Client must contact server

- server process must first be running
- server must have created socket (door) that welcomes client's contact

Client contacts server by:

- creating client-local TCP socket
- specifying IP address, port number of server process
- When client creates socket: client TCP establishes connection to server TCP

- When contacted by client, server TCP creates new socket for server process to communicate with client
  - allows server to talk with multiple clients
  - source port numbers used to **distinguish** clients

application viewpoint

TCP provides reliable, in-order transfer of bytes ("pipe") between client and server

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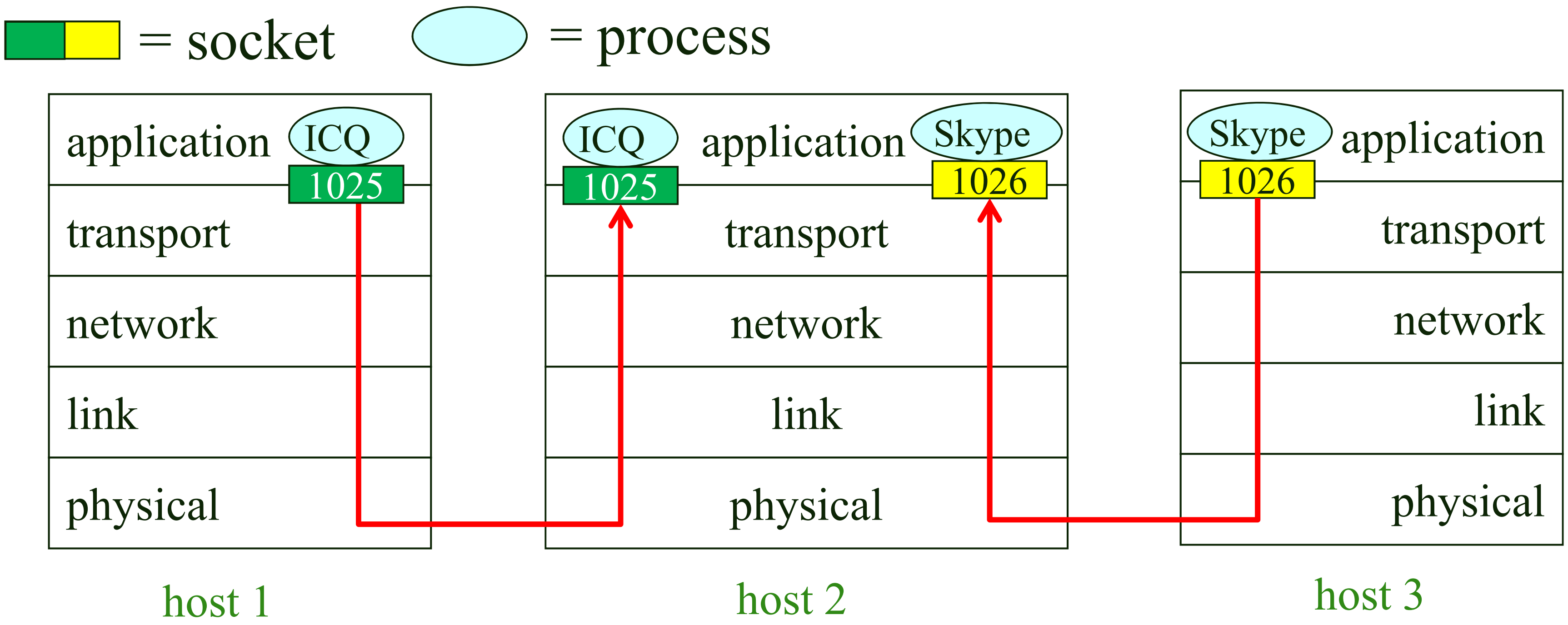
Multiplexing/demultiplexing

Multiplexing at send host:

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

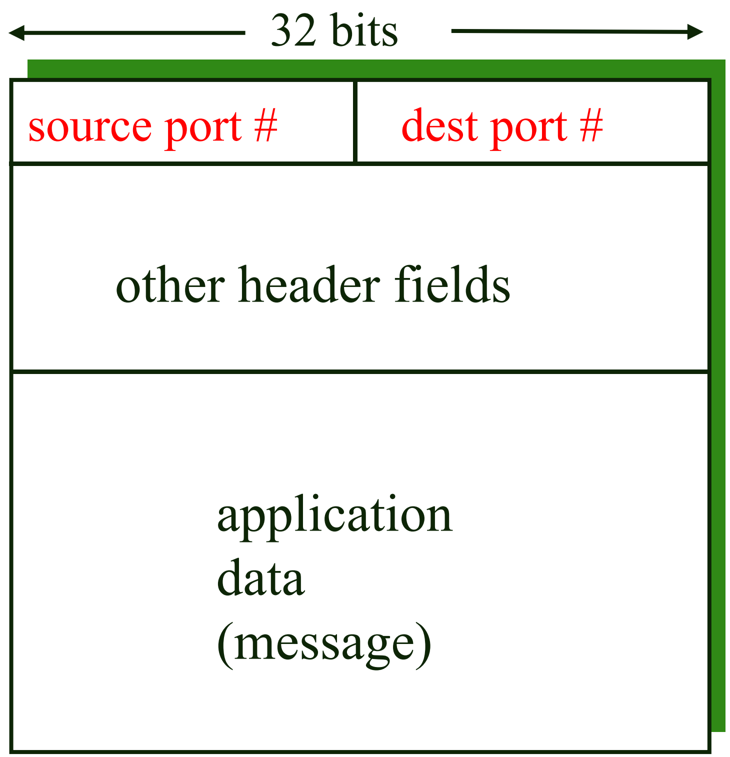
Demultiplexing at rcv host:

delivering received segments to correct socket



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 clientSocket =
   new DatagramSocket();

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

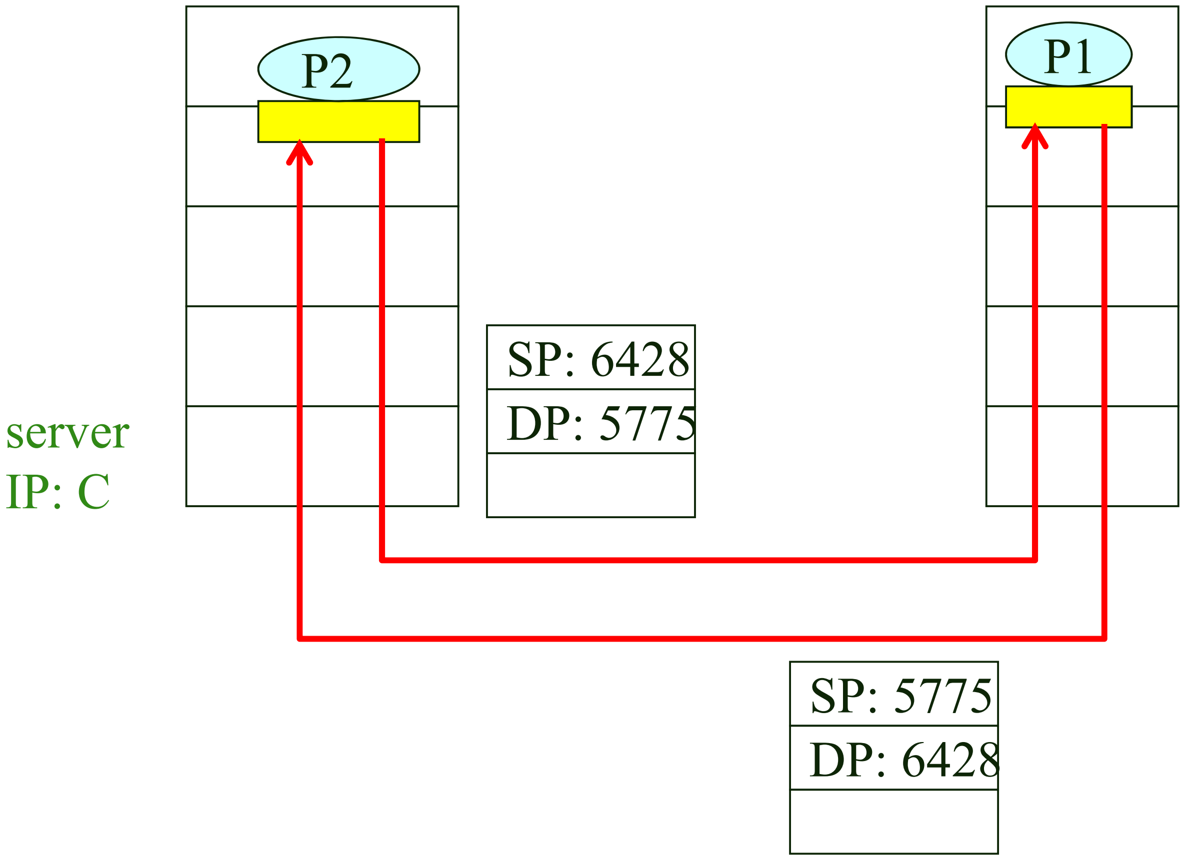
- UDP socket identified by two-tuple:

(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);
```



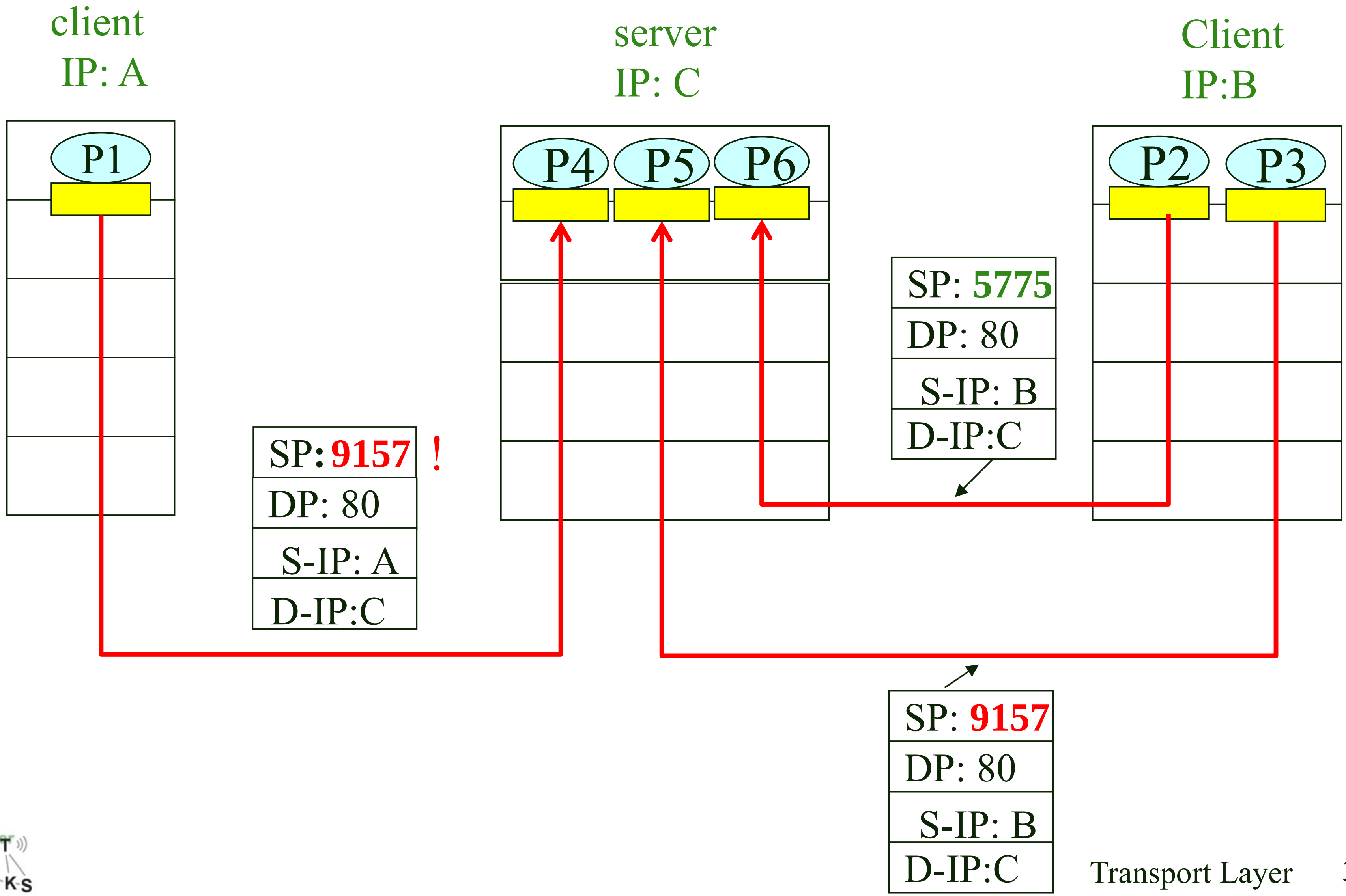
SP provides "return address"

Connection-oriented demux

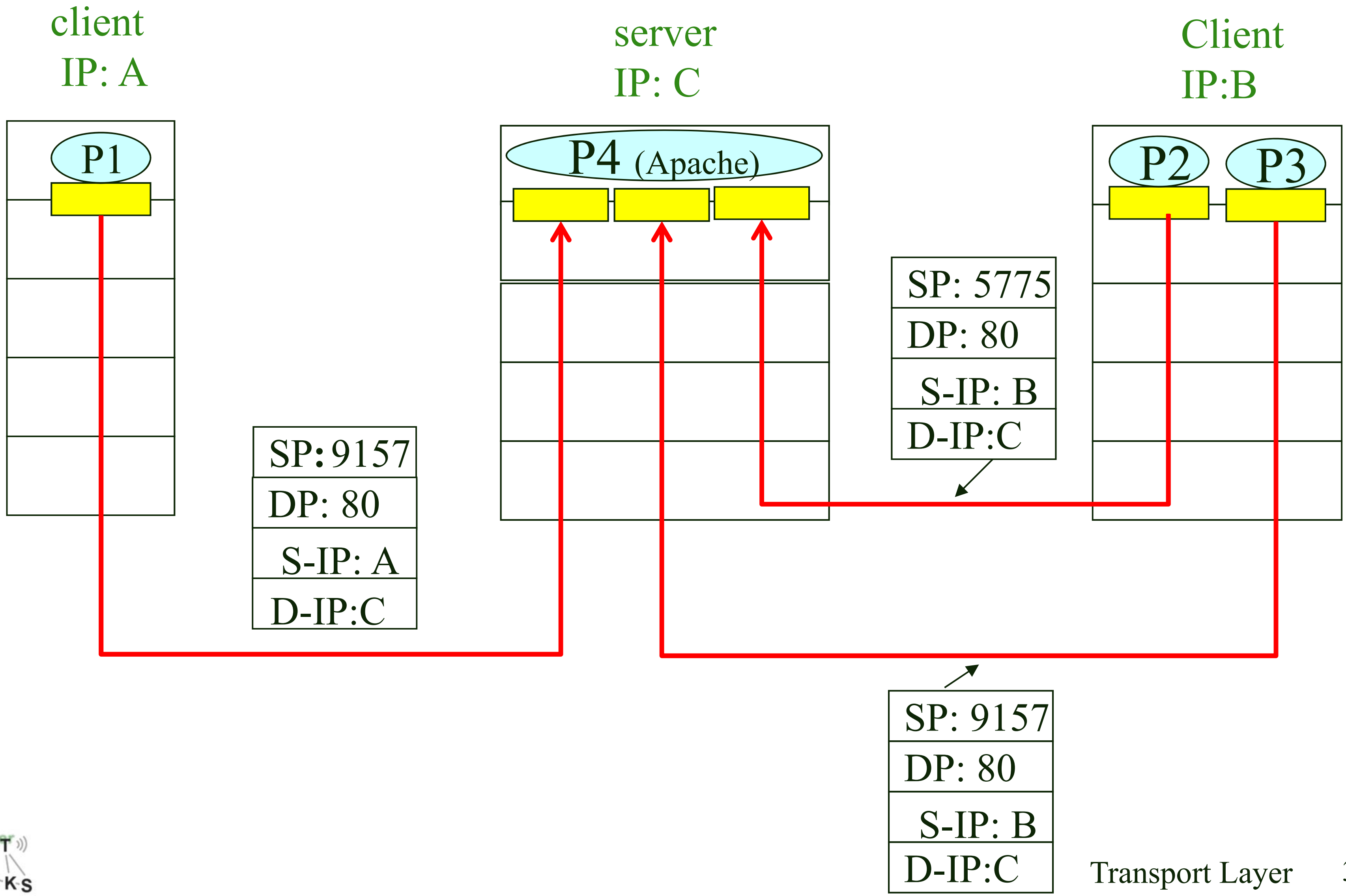
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host uses all four values to direct segment to appropriate socket

- 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

Connection-oriented demux (cont)



Connection-oriented demux (cont)



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The Problem with TCP

- TCP offers a **reliable** and **easy** to use transport protocol to programmers.
  - Congestion control
  - Retransmissions etc.
- However congestion control imposes transmission-rate **constraints**.
- If a traffic jam is detected on a path, sender **decreases** sending rate “dramatically”.
- **Problem**: One cannot “switch” off functions of TCP ex. Congestion control.

UDP: User Datagram Protocol [RFC 768]

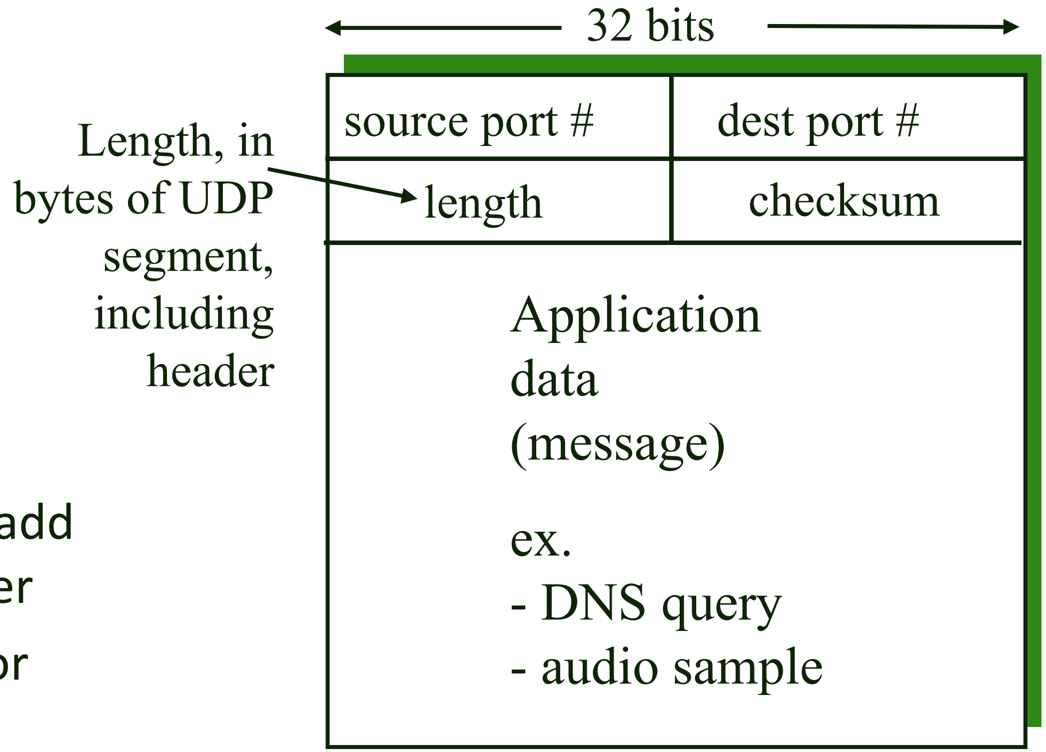
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled **independently** of others

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state (buffers & parameters) at sender, receiver
- small segment header (8 bytes v.s. 20 bytes)
- no congestion control & retransmission: UDP can blast away as fast as desired (e.g. used by VOIP)

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!
  - ex. ACK/NAK, retransmissions (non-trivial).



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 ….

UPD checksum example

- Lets take the word "hi" (8bit ASCII)
- Convert it to binary
  - h = 01101000
  - i = 01101001
- Add both words

```
  01101000  (h)
+ 01101001 (i)
  ------------
  11010001 (h+i)
```

- UDP checksum works with 16 Bit words, but we use 8 Bits for simplicity

- The 1s complement is obtained by inverting ones to zeros and vice versa.
- 11010001 -> 00101110 (checksum)

UPD checksum example

- Check (unaltered bits):

  01101000 (h)

  \+ 01101001 (i)

  11010001 (h+i)

  \+ 00101110 (checksum)

  11111111 (OK)

- Check (altered bits):

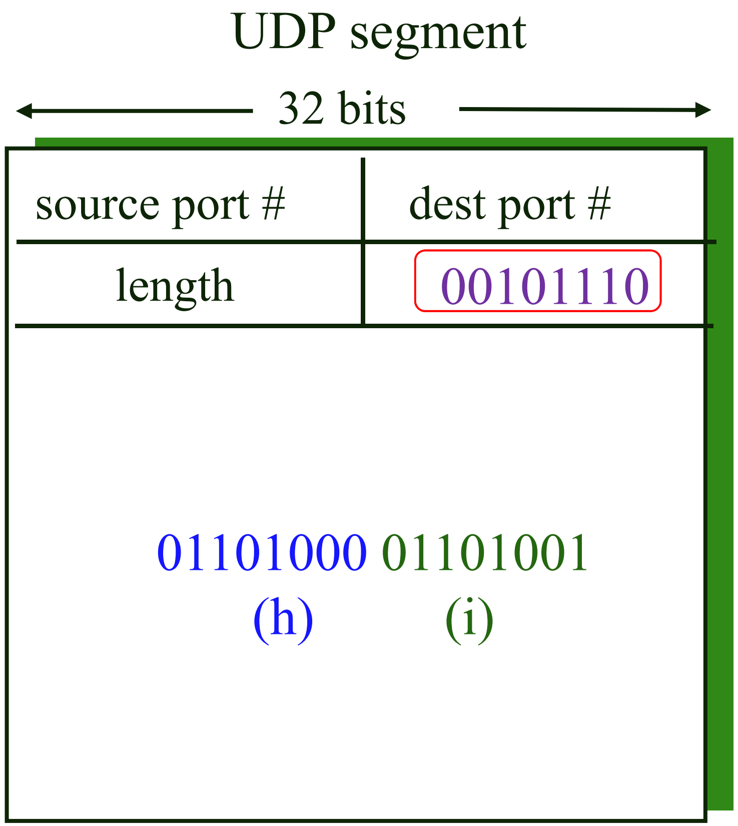
  01101000 (h)

  \+ 01101011 (i)

  11010011 (h+i)

  \+ 00101110 (checksum)

  100000001 (NOK!)



UDP checksum

- Why error detection in the first place?
- Link Layer provides CRC! (Ethernet)
- No guarantee for:
  - link-to-link reliability (e.g. non ethernet)
  - memory error detection on routers
- IP is designed to run on any layer 2 protocol (ethernet, PPP, 802.11, 802.16).
- End-to-end error detection is safety measure
- UPD does not recover from errors (discard/warning)

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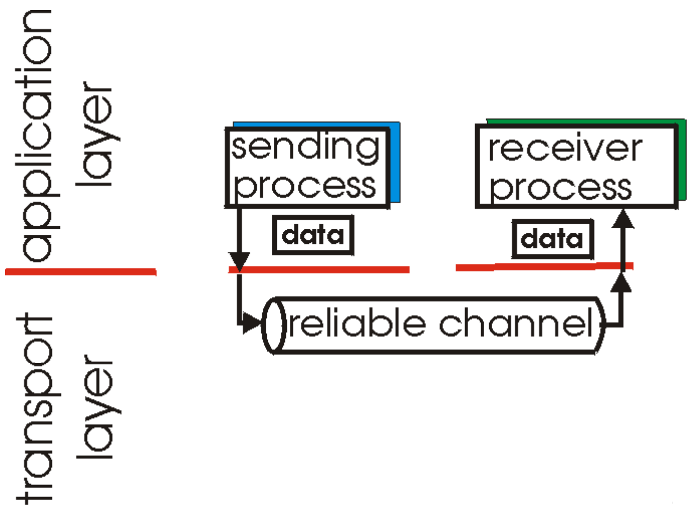
Principles of Reliable data transfer

- 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

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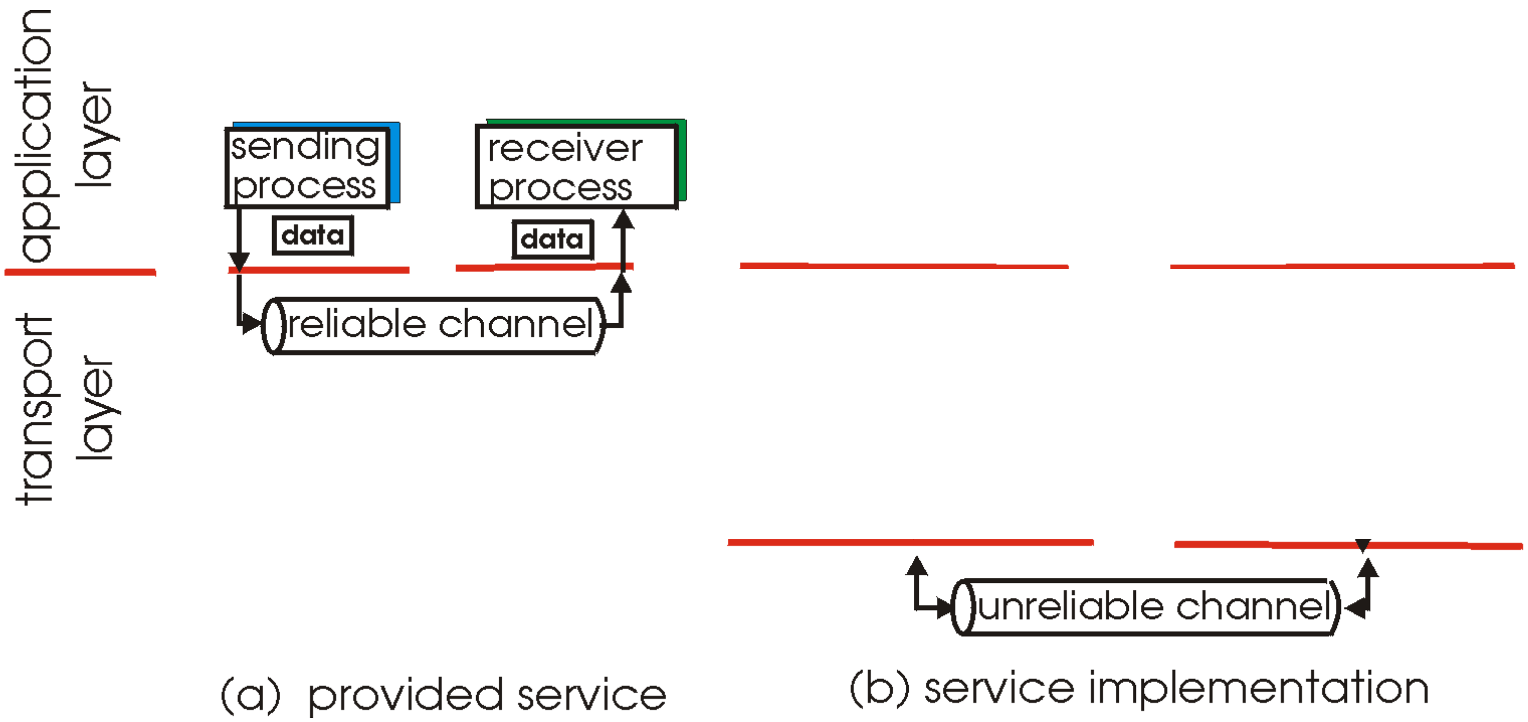
- top-10 list of important networking topics!



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Principles of Reliable data transfer

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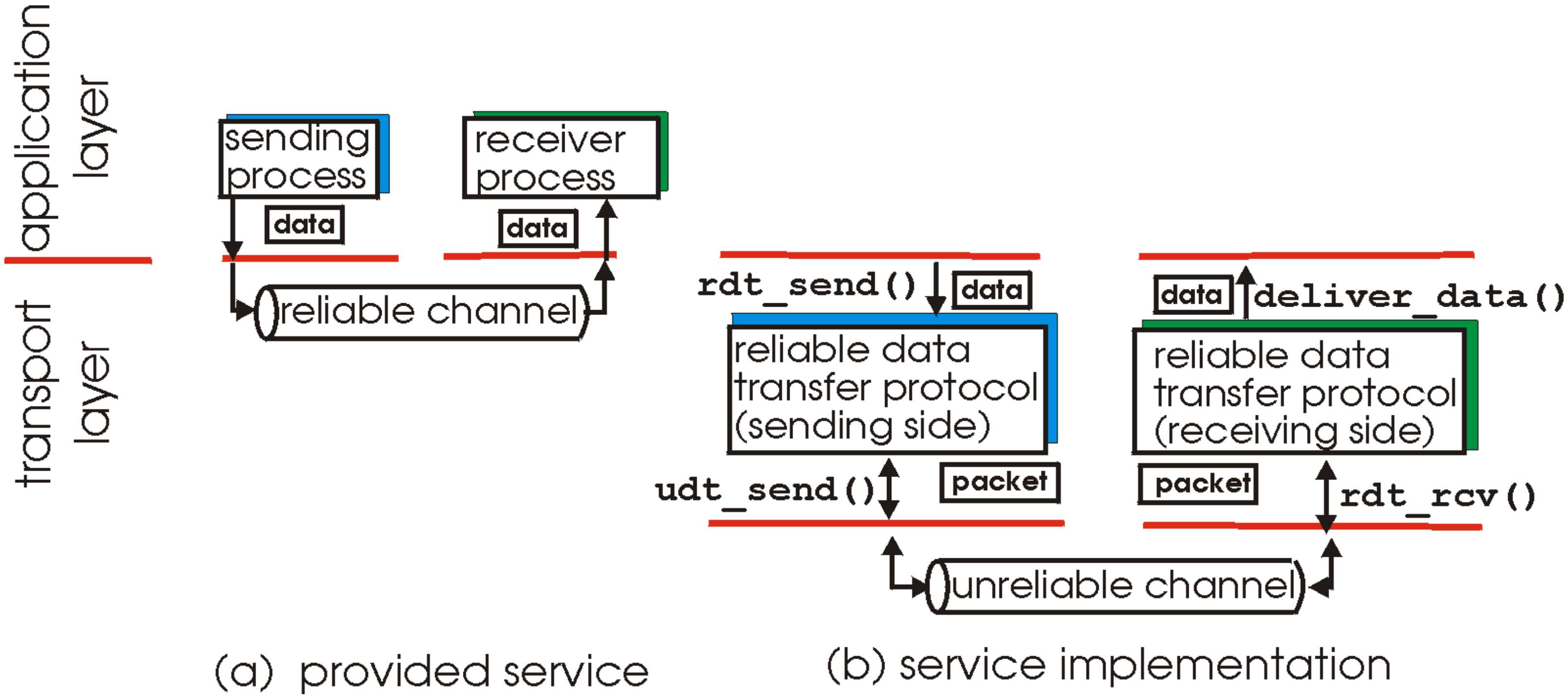
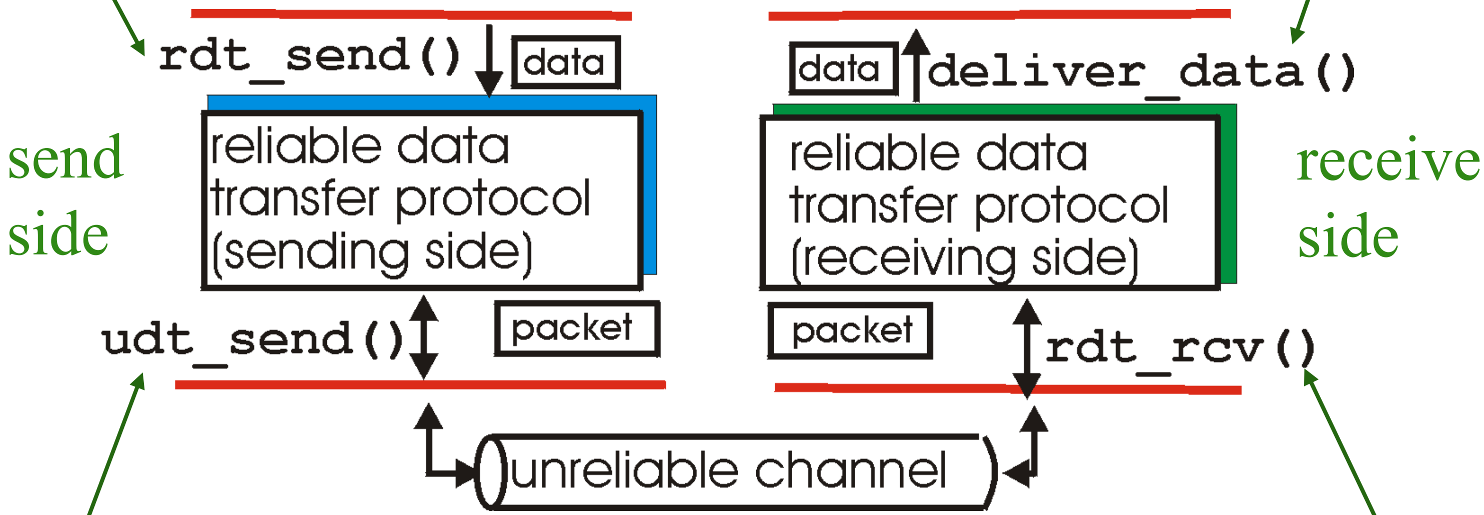


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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 upper



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

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

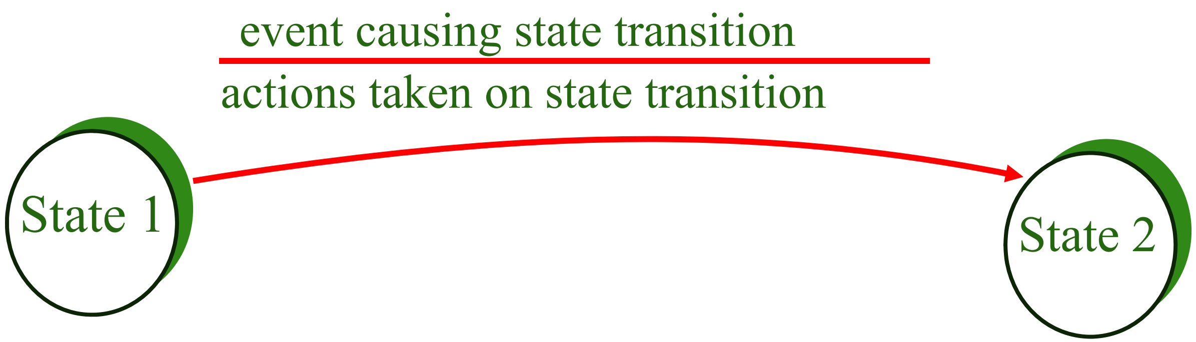
Reliable data transfer: getting started

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
- Use generic term "packet" rather than "segment"

Finite State Machine

- FSM is a model of behavior composed of a finite number of
  - states
  - transitions between states on events
  - actions taken upon events
- Necessary to define the behavior of our protocol, prior to implementation



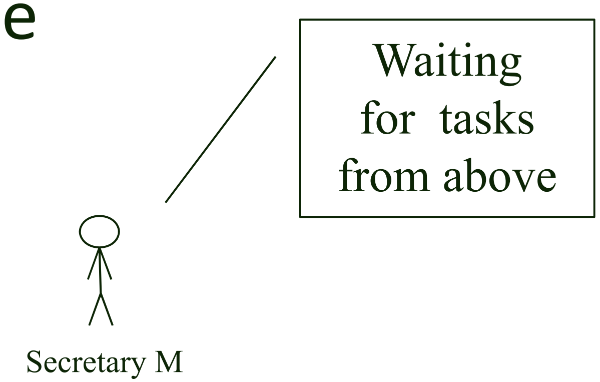
Rdt1.0: reliable transfer over a reliable channel

- **Assumption:** 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

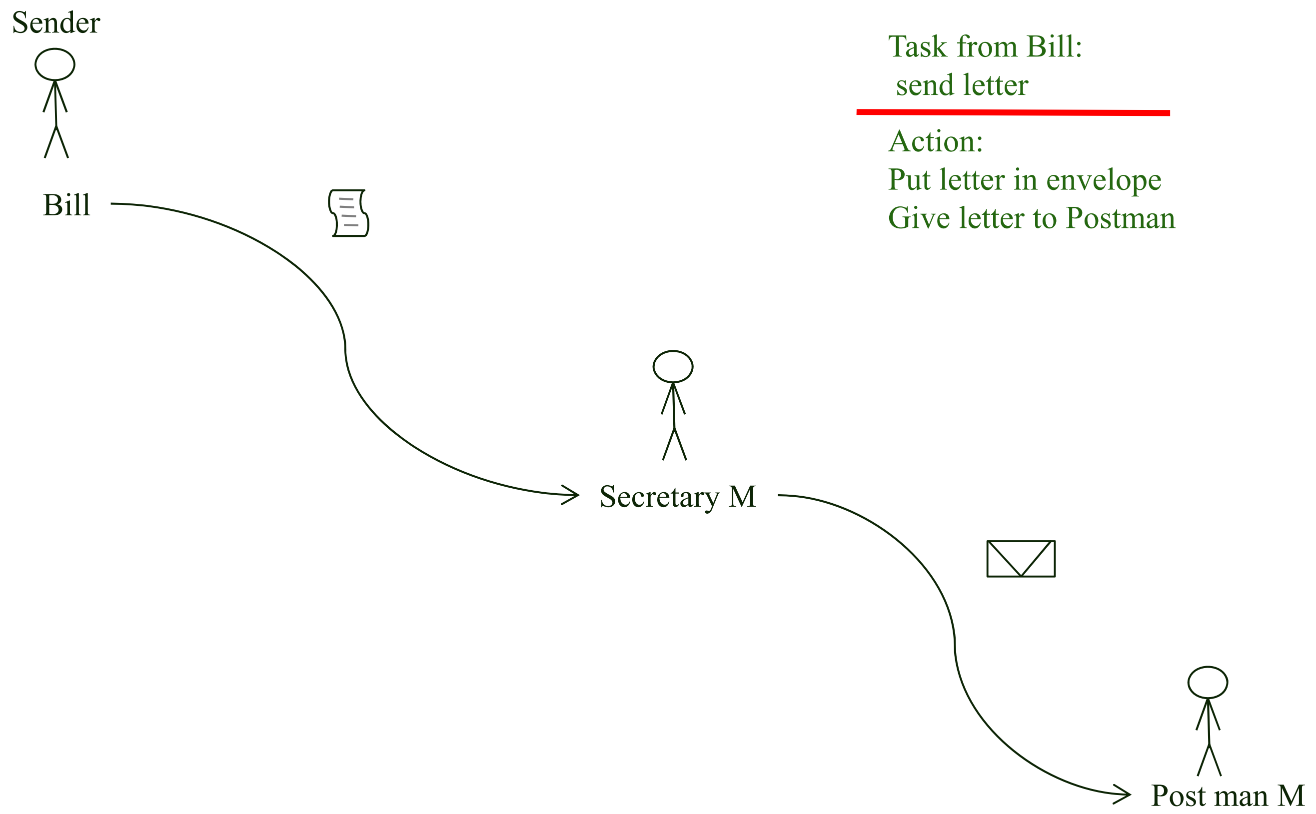
- We will first look at an analogy with the secretary then the state machines.

Rdt1.0: reliable transfer over a reliable channel (Analogy)

- The secretary from our previous example has one state
- He waits for tasks from his boss
- Task is sending letters



Rdt1.0: reliable transfer over a reliable channel (Analogy)



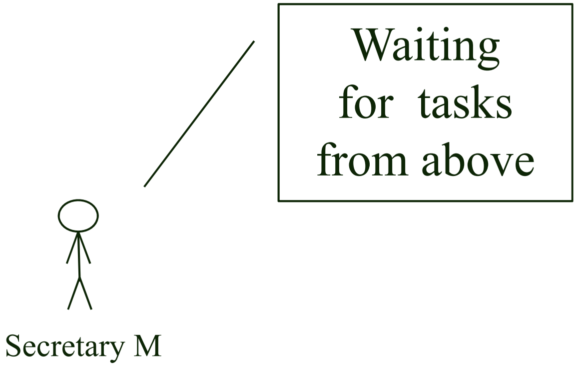
Task from Bill:
send letter

Action:
Put letter in envelope
Give letter to Postman

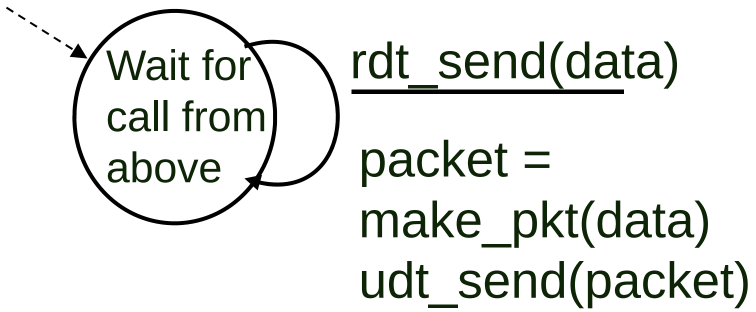
event → transition
transition → action

Rdt1.0: reliable transfer over a reliable channel (Analogy)

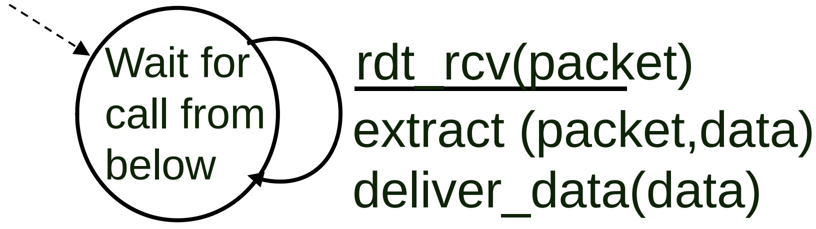
- The secretary goes back to his state, waiting for more tasks.



Rdt1.0: reliable transfer over a reliable channel



sender



receiver

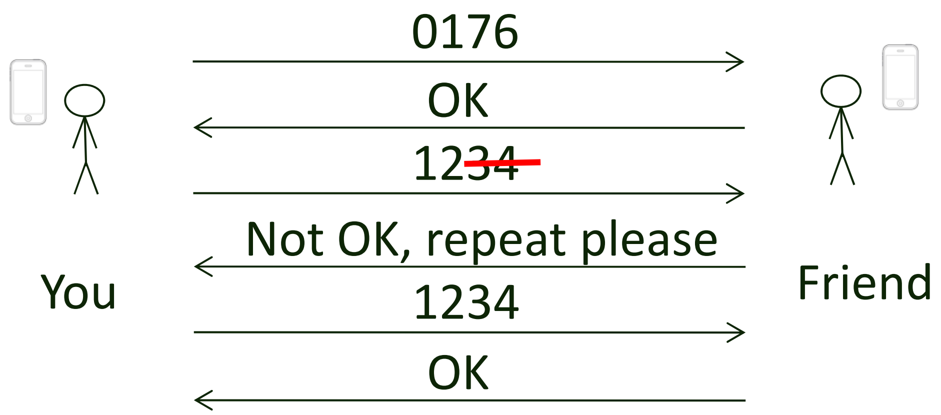
event → transition
transition → action

Λ = no event/action

Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors   00101110
- ***the* question**: how to recover from errors?

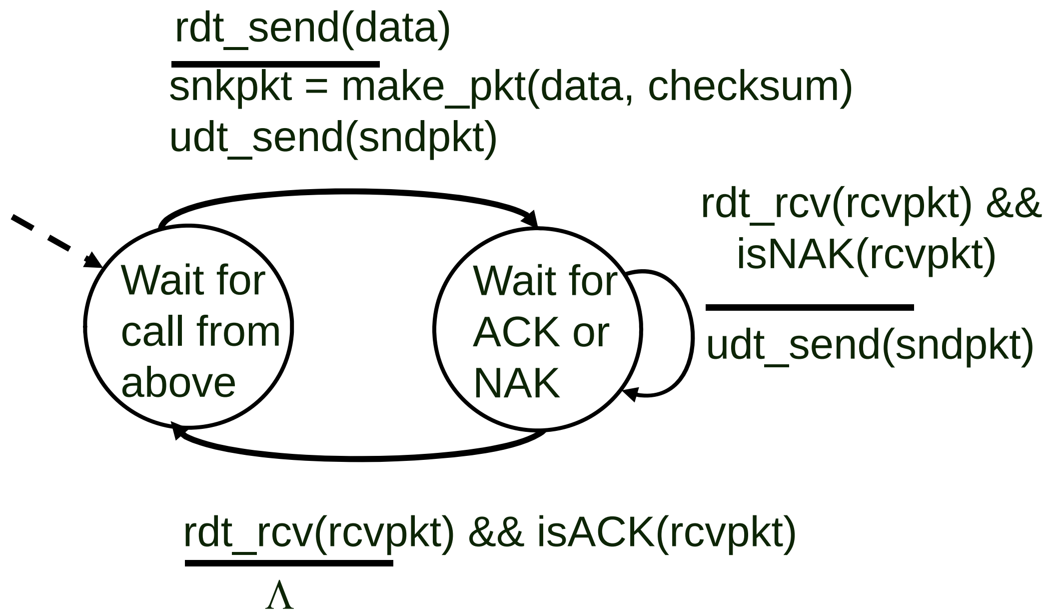
- **Analogy**:
  - Imagine you dictate phone number over cell phone to friend.
  - Bad reception may scramble your voice.



Rdt2.0: channel with bit errors

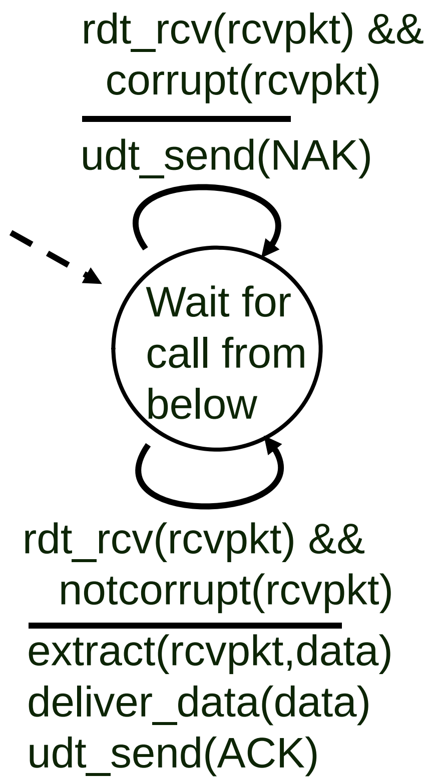
- *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
- *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender
- **A**utomatic **R**epeat re**Q**uest type of protocol (**ARQ**)

rdt2.0: FSM specification



sender

receiver



event → transition

transition → action

Λ = no event/action

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:

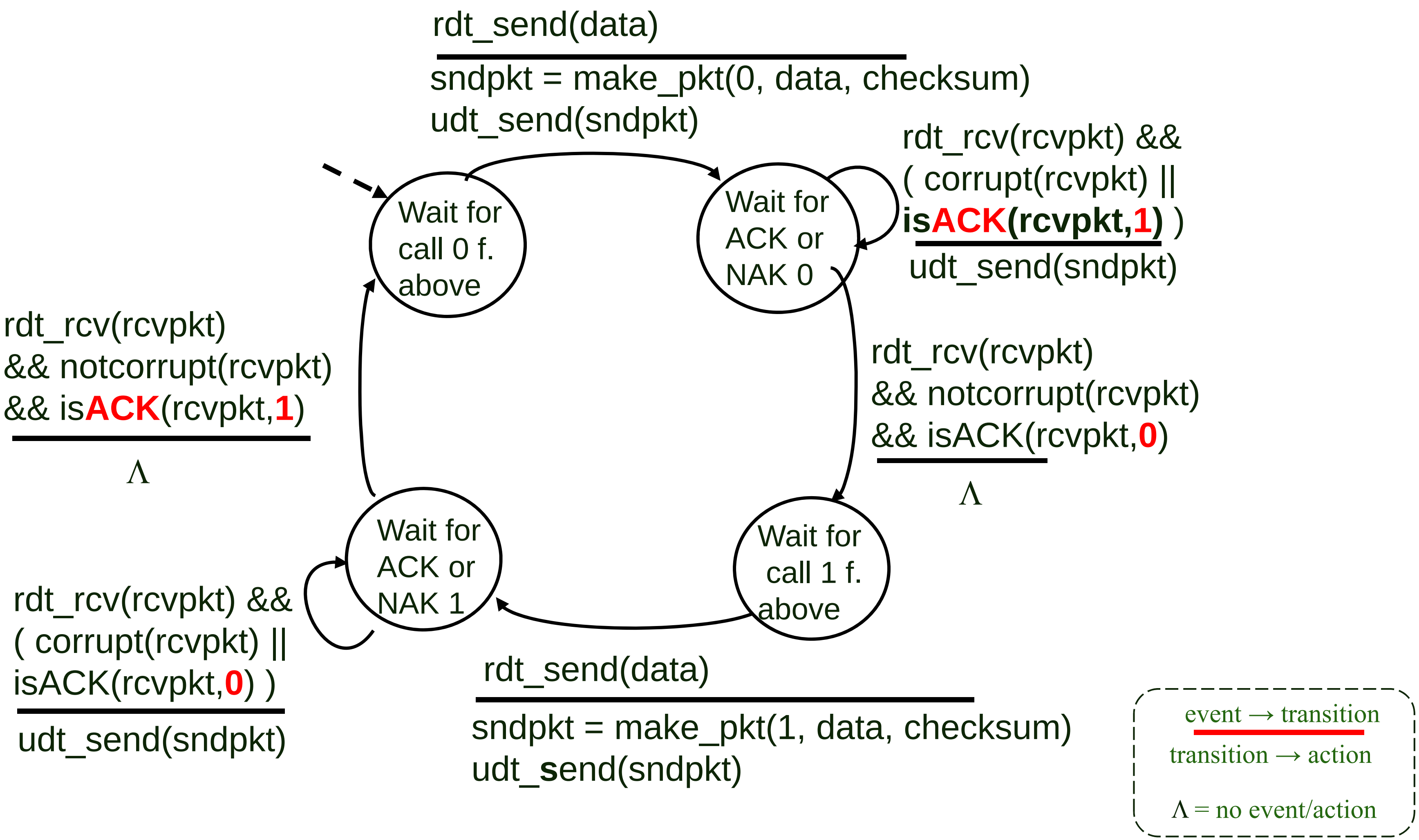
- sender retransmits current pkt if ACK/NAK garbled
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

Using only ACK + Sequence:

- We can **discard** NAK packets, by using only ACK + Seq.#
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

stop and wait
Sender sends one packet, then waits for receiver response

rdt2.2: sender, handles garbled ACKs



rdt2.2: receiver, handles garbled ACKs

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)

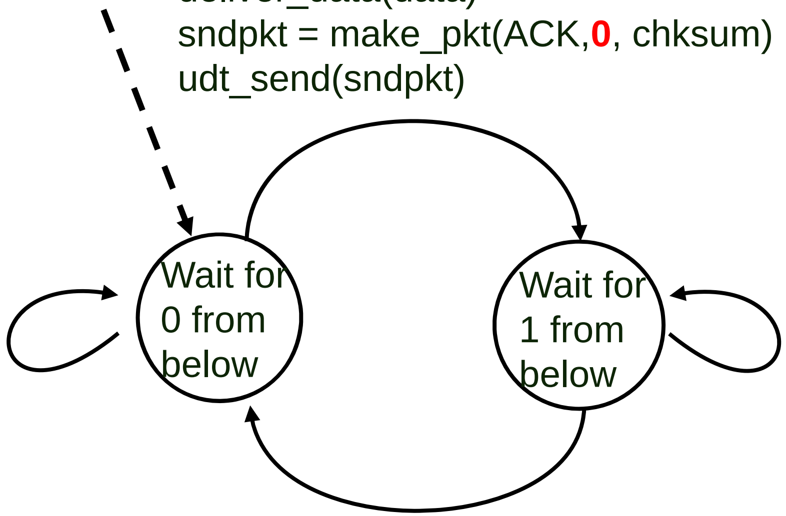
---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK,**0**, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
has_seq1(rcvpkt)

---

udt_send(sndpkt)



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---

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&& has_seq1(rcvpkt)

---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, **1**,chksum)
udt_send(sndpkt)

event → transition

---

transition → action

Λ = no event/action

rdt2.2: discussion

Sender:

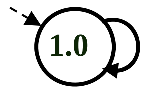
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

Receiver:

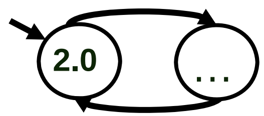
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK received OK at sender

rdt: What do we have so far?

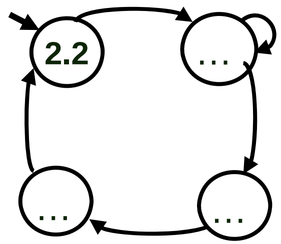
- rdt 1.0
  - simple transfer over reliable channel (unrealistic)



- rdt 2.0
  - bit error prone channel (more realistic)
  - checksum (data), ACK/NAK, retransmit
  - **but what if ACK corrupt**?



- rdt 2.2
  - checksum (data & **ACK**)
  - retransmit if ACK corrupt
  - **but what if data OK, but ACK corrupt?** -> **duplicate**
  - introduce sequence numbers (more states)
  - slimed down: discard NAK by introducing seq. in ACK
  - **but what if channel looses packets?**



rdt3.0: channels with errors *and* loss

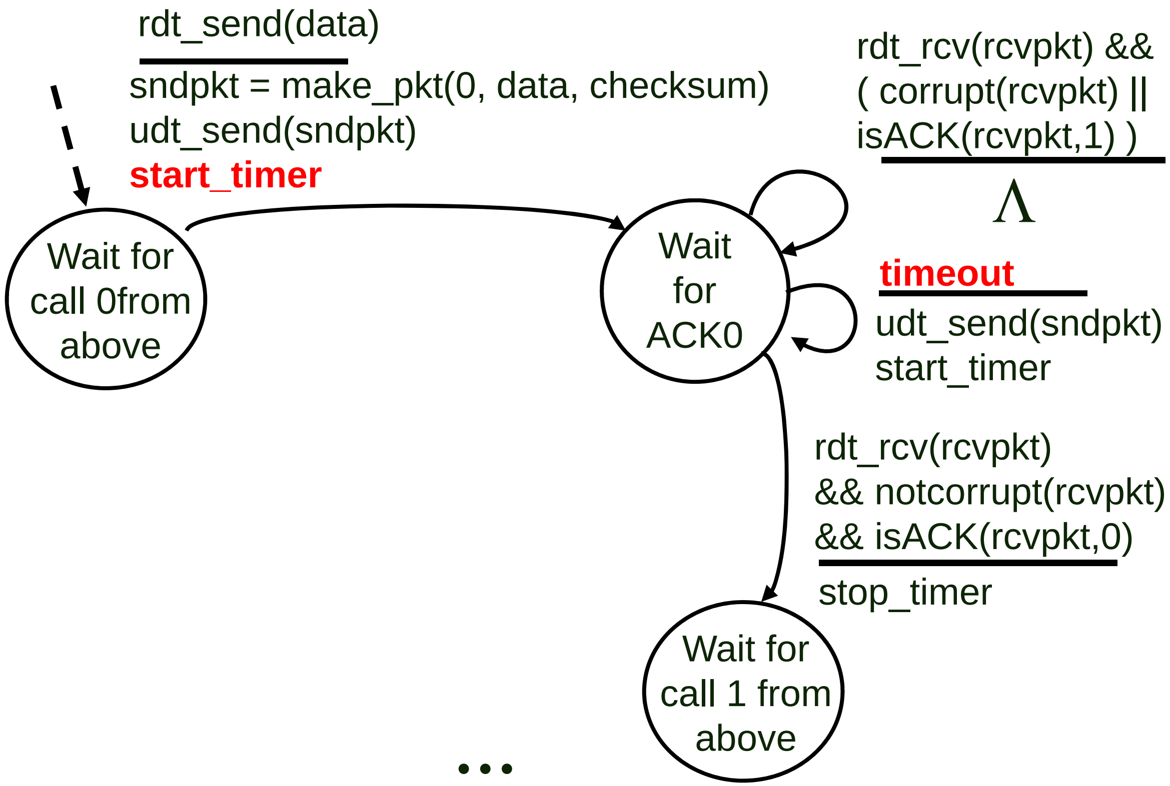
New assumption: underlying channel can also lose packets (data or ACKs)

- 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

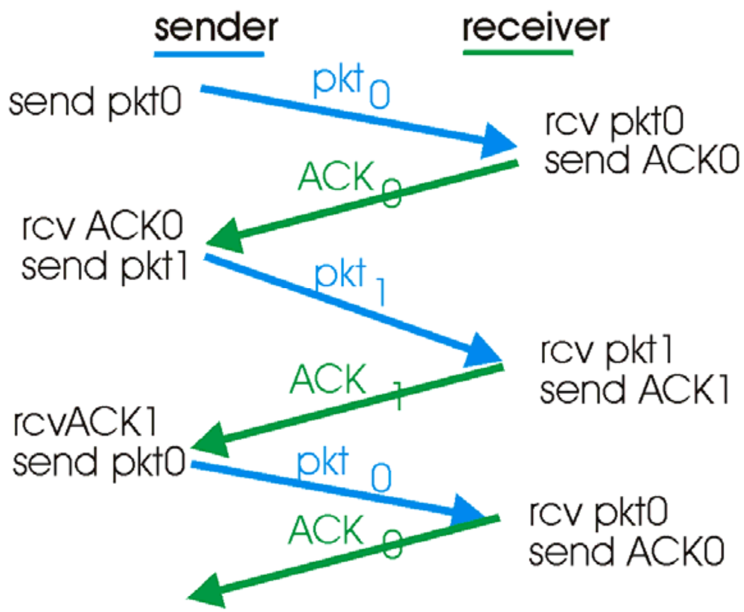


event → transition

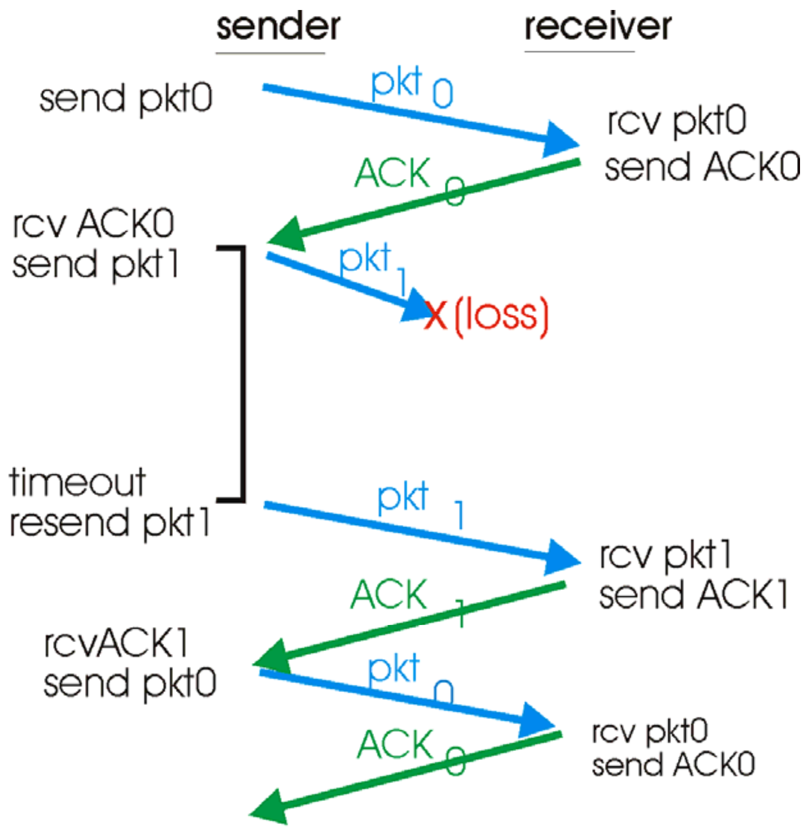
transition → action

Λ = no event/action

rdt3.0 in action

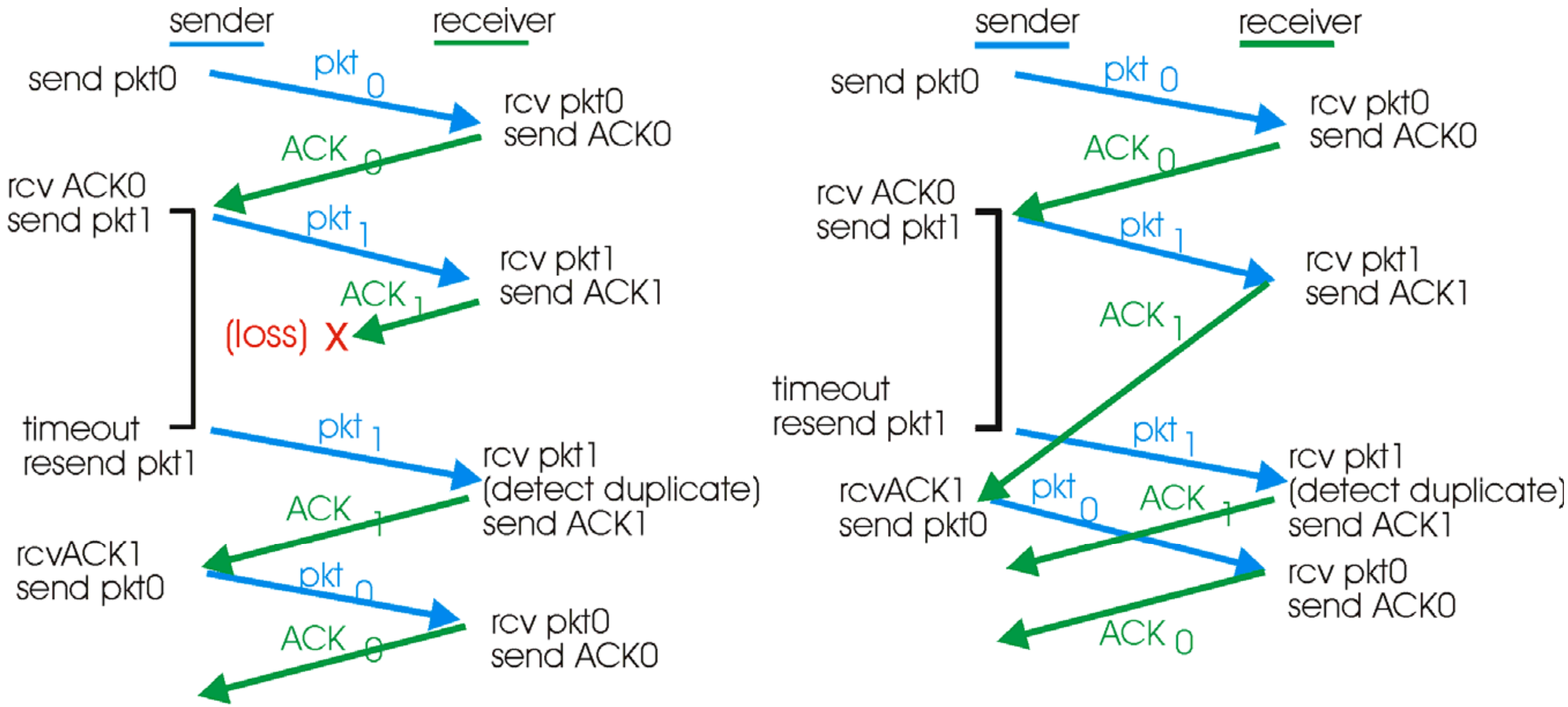


(a) operation with no loss



(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:

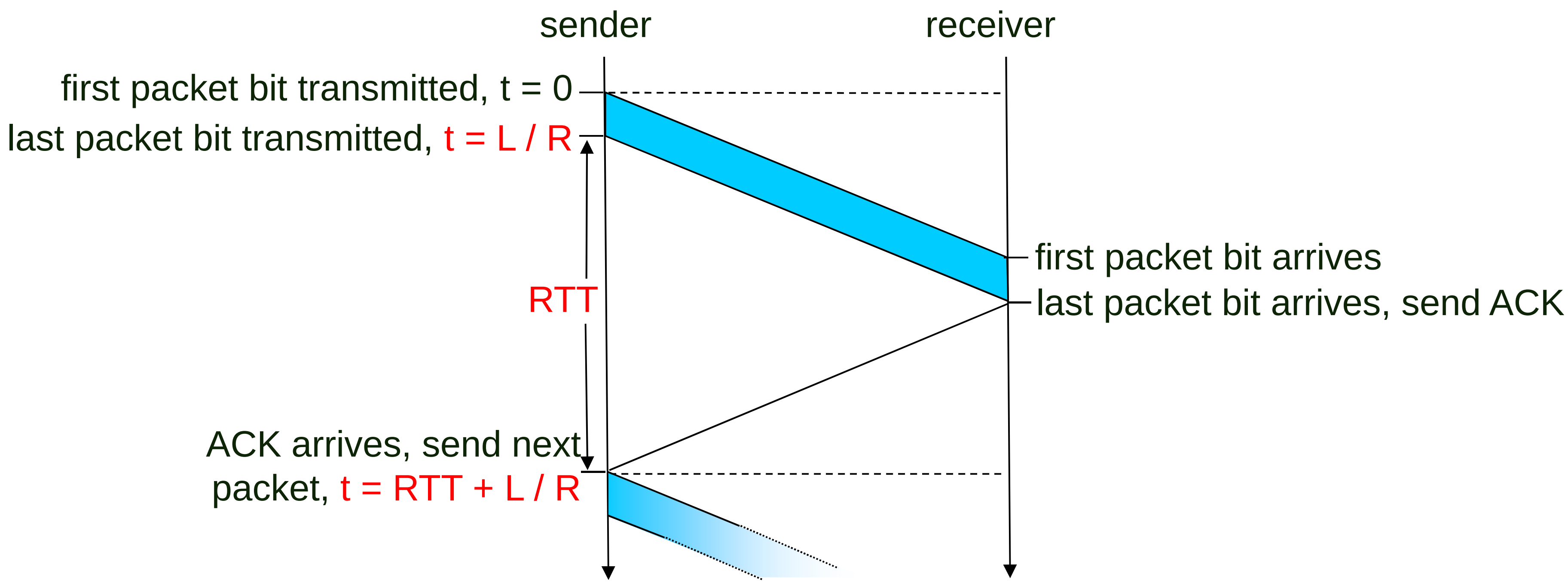
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\,\text{microseconds}$$

- 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

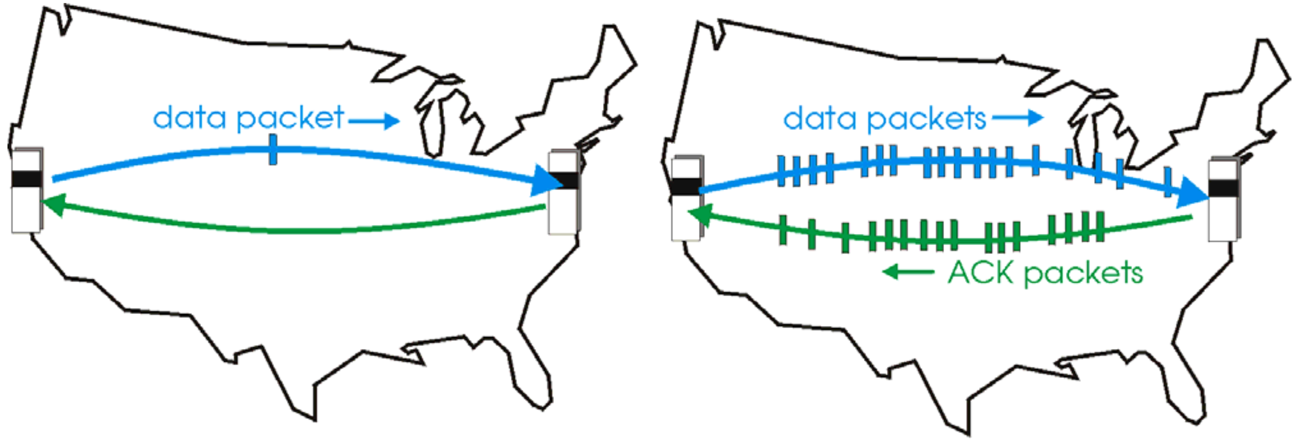


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

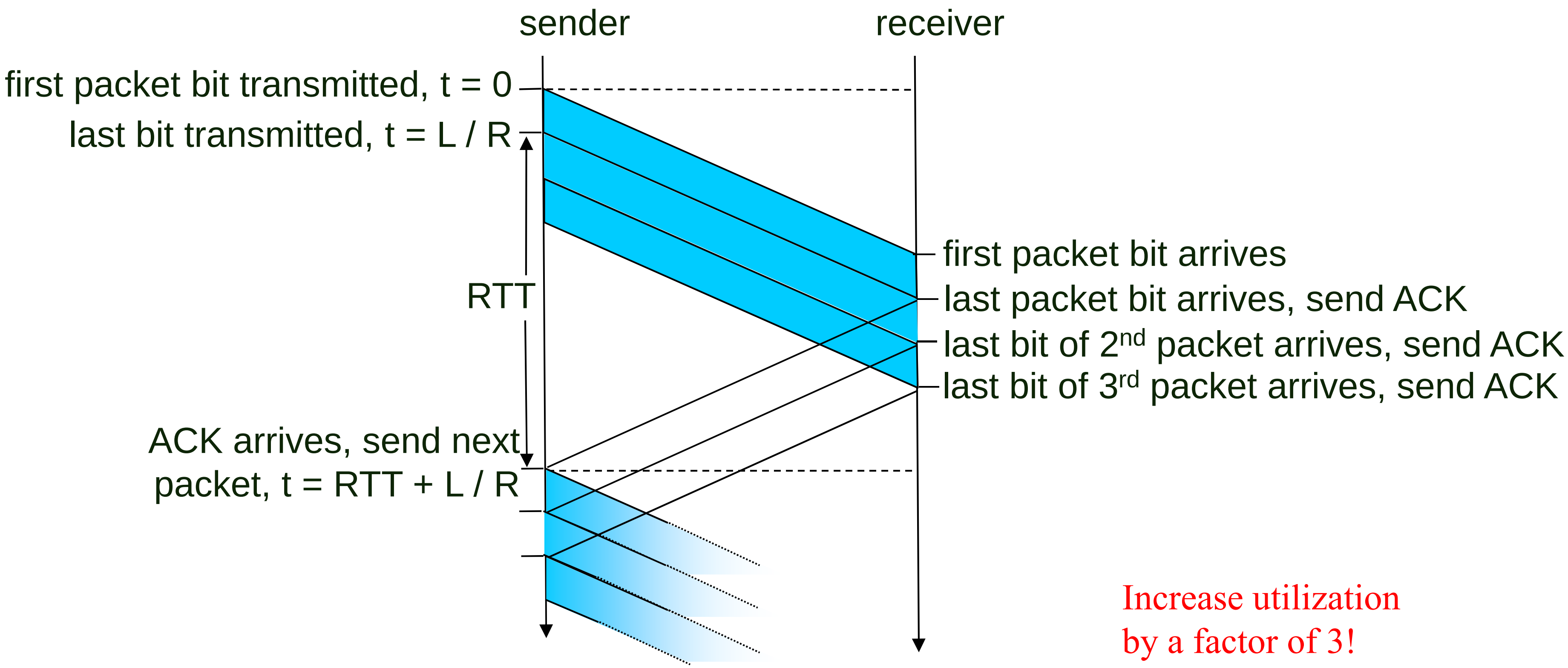
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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

Go-Back-N (GBN) Demonstration

- Protocol Demo (Link)
- http://media.pearsoncmg.com/aw/aw_kurose_network_2/applets/go-back-n/go-back-n.html

Chapter 4: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer

<u>Next:</u>

  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

Thank you

Any questions?