Lecture 9: Transpor Layer Overview and UDP

COMP 332, Spring 2018

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Today

1. Announcements
   – homework 4 due Wed. at 11:59p

2. Transport layer
   – overview
   – multiplexing and demultiplexing
   – User Datagram Protocol (UDP)

3. Reliable data transport
   – principles
   – protocol v1.0

Transport Layer

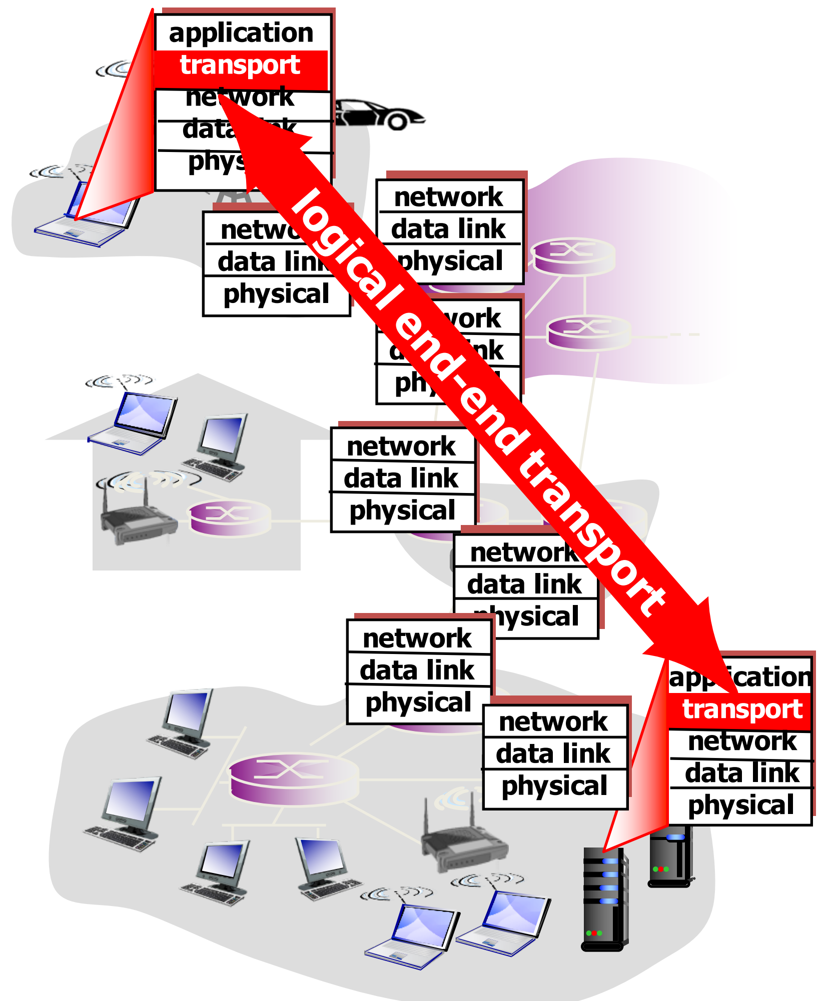
OVERVIEW

Why do we need a transport layer?

Application
Transport
Network
Link
Physical

- Logical communication between processes on end hosts
- Relies on, enhances, network layer services

- Logical communication between end hosts
- IP header does not contain port #s



What problems must transport layer address?

Why do we need a transport layer?

Transport layer services

Problem 1

- no port #s in IP header

How do packets get from host to process on host?

⇒ (De)Multiplexing

Problem 2

- IP is best effort
  - packets can be corrupted, dropped, duplicated, reordered, delayed

Pain for app developer to deal with

⇒Reliable data transfer

Problem 3

- IP gives no guidance about rate at which to send packets
  - sends whatever it receives immediately

Traffic can easily overwhelm network, host

⇒ Congestion, Flow control

Problem 4

- IP packets need to be reassembled into original message

Pain for app developer to deal with

⇒ Data stream

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UDP, TCP

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Transport layer protocols on Internet

TCP: reliable, in-order delivery

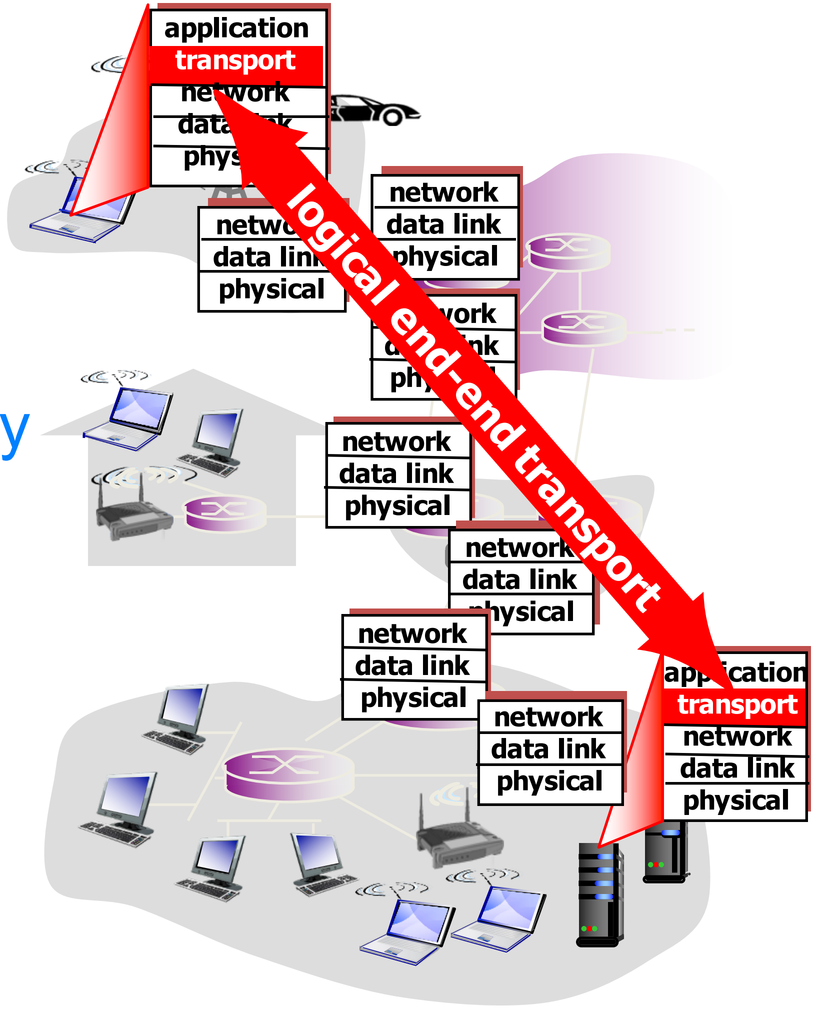
- connection-oriented
- congestion control
- flow control
- connection setup

UDP: unreliable, unordered delivery

- connectionless
- no-frills extension of best-effort IP

Q: What services are not available

- delay guarantees
- bandwidth guarantees



Transport Layer

MULTIPLEXING AND DEMULTIPLEXING

Transport layer

Provides

- logical communication between app processes running on different hosts

Transport protocols run in end systems

- send side
  - breaks app messages into segments (TCP) datagrams (UDP)
  - passes to network layer
- rcv side
  - reassembles segments or datagrams into messages
  - passes to app layer

Household analogy

12 kids in Alice's house sending letters to 12 kids in Bob's house

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

Multiplexing and demultiplexing

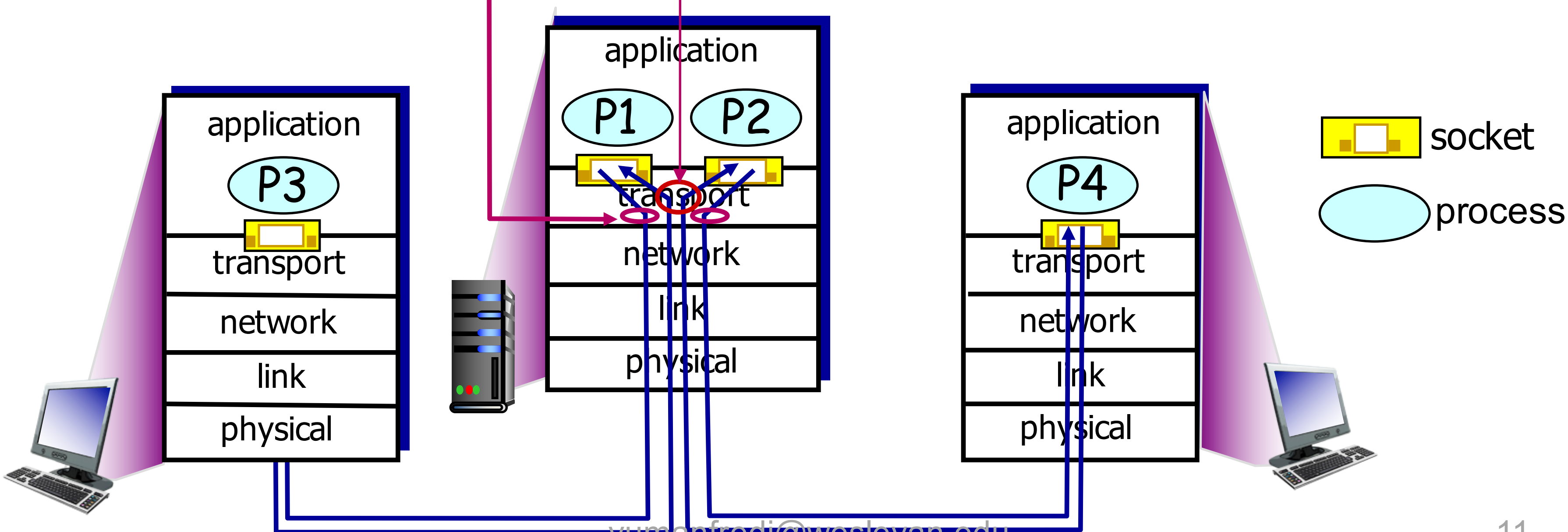
Determines which packets go to which app

Mux at sender

Handle data from multiple sockets, add transport header (later used for demultiplexing)

Demux at receiver

Use header info to deliver received segments to correct socket



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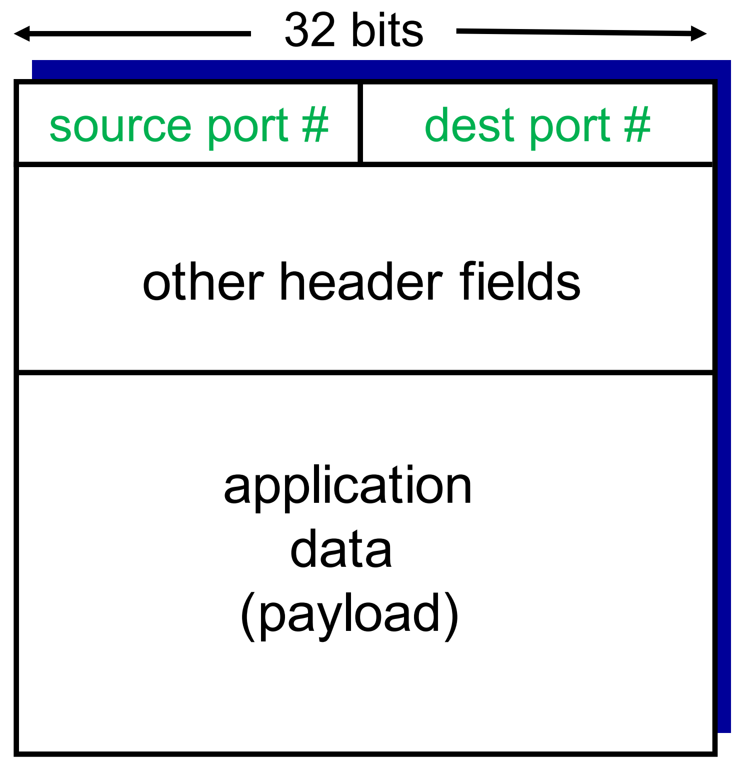
How demultiplexing works

Host receives IP packets

- packet header contains
  - source IP address
  - destination IP address
- packet payload is
  - one transport-layer segment or datagram
- transport-layer header contains
  - source port number
  - destination port number

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



Format of TCP/UDP segment/datagram

Connectionless demultiplexing (UDP)

Recall

- created socket has random host-local port # allocated:

```
sock1 = socket(AF_INET,SOCK_DGRAM)
port# allocated:9157
```

- when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

When host receives UDP datagram

- checks destination port # in UDP header on datagram
- directs UDP datagram to socket with that port #

IP packets with same destination IP and port #, but different source IP addresses and/or source port numbers: will still be directed to same socket at destination

Connectionless demultiplexing (UDP)

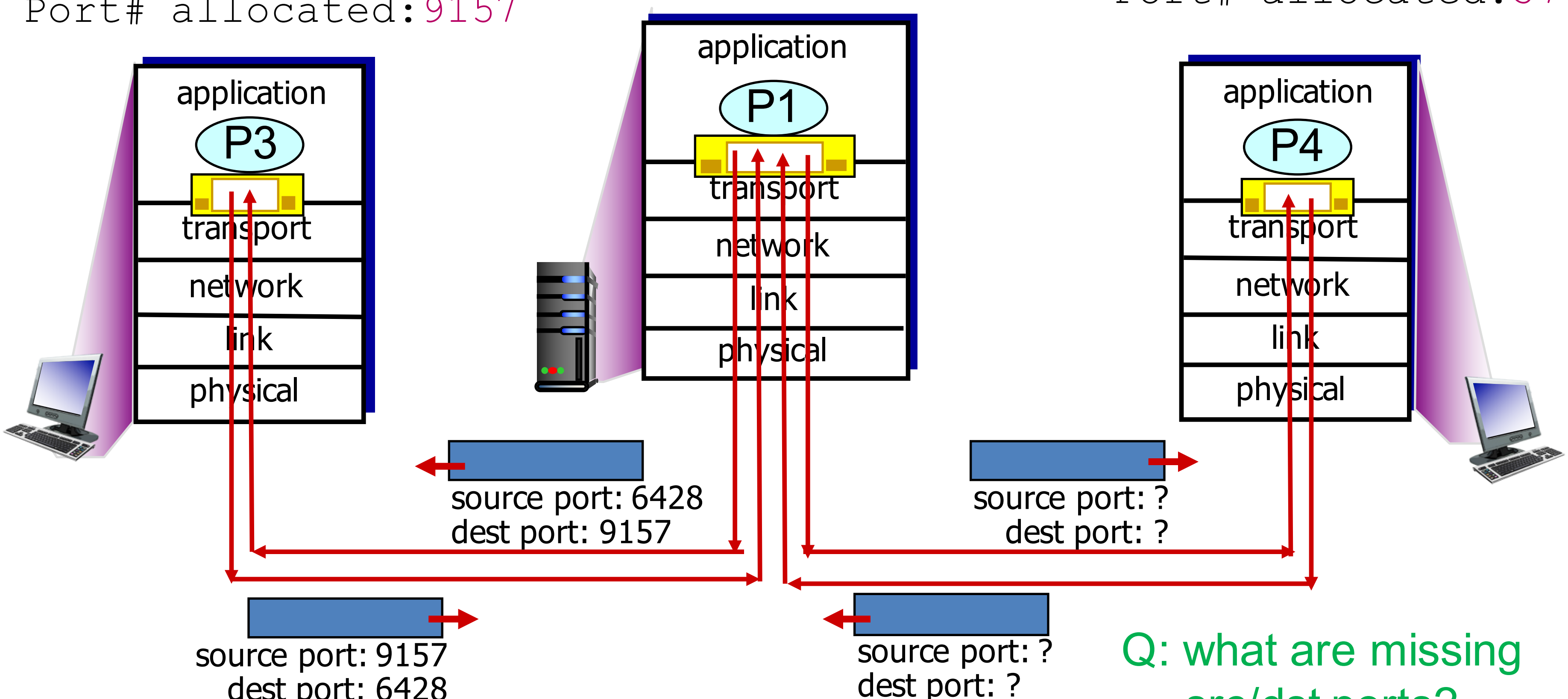
```
sock2 =
socket(AF_INET,
SOCK_DGRAM)
```

Port# allocated: 9157

```
server_sock =
socket(AF_INET,
SOCK_DGRAM)
server_sock.bind((
localhost,6428))
```

```
sock1 =
socket(AF_INET,
SOCK_DGRAM)
```

Port# allocated: 5775



Q: what are missing src/dst ports?

Connection-oriented demultiplexing (TCP)

TCP socket identified by 4-tuple

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

Demux

- receiver uses all four values to direct segment to appropriate socket

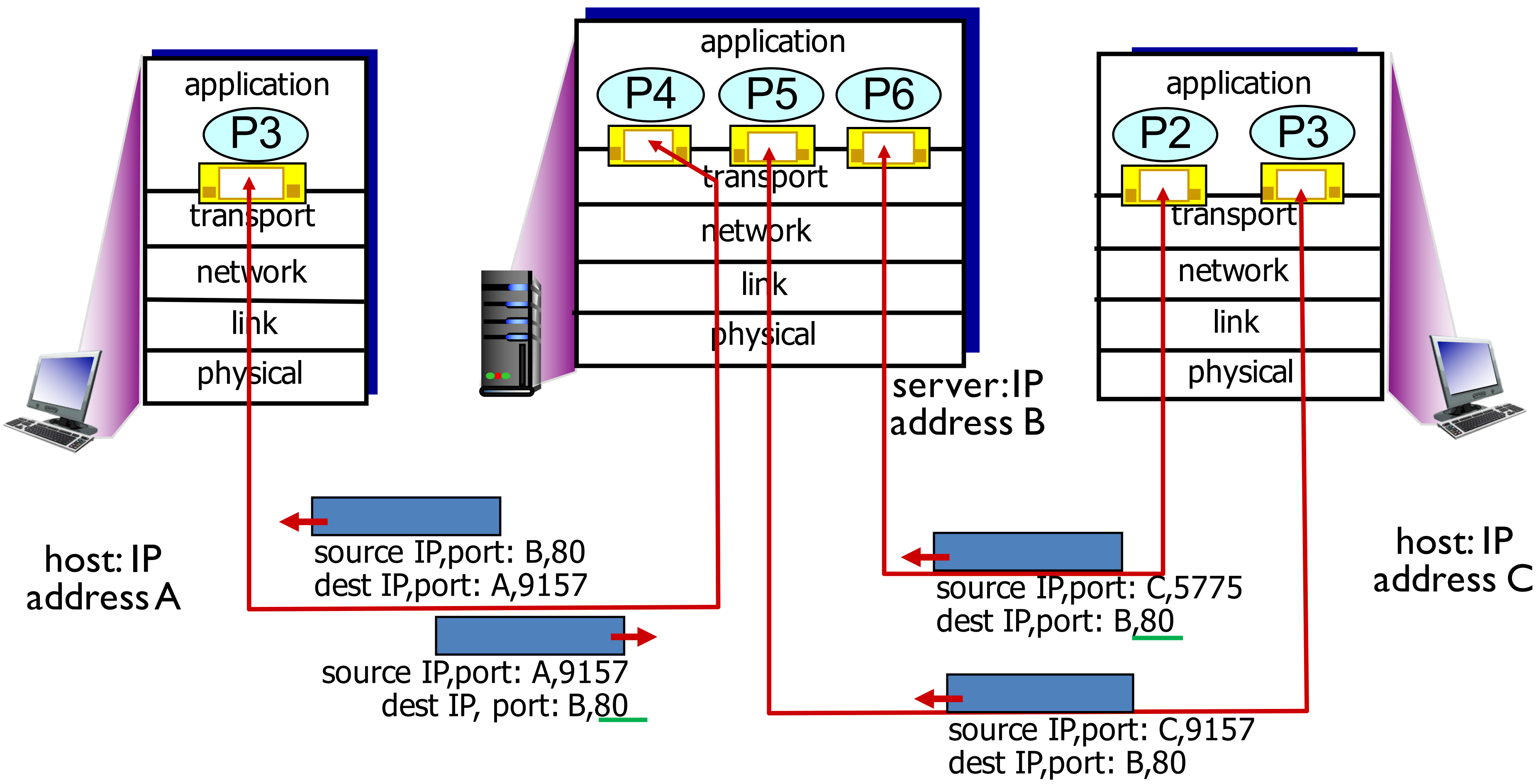
Server host

- may support many simultaneous TCP sockets
- each socket identified by its own 4-tuple

Web servers

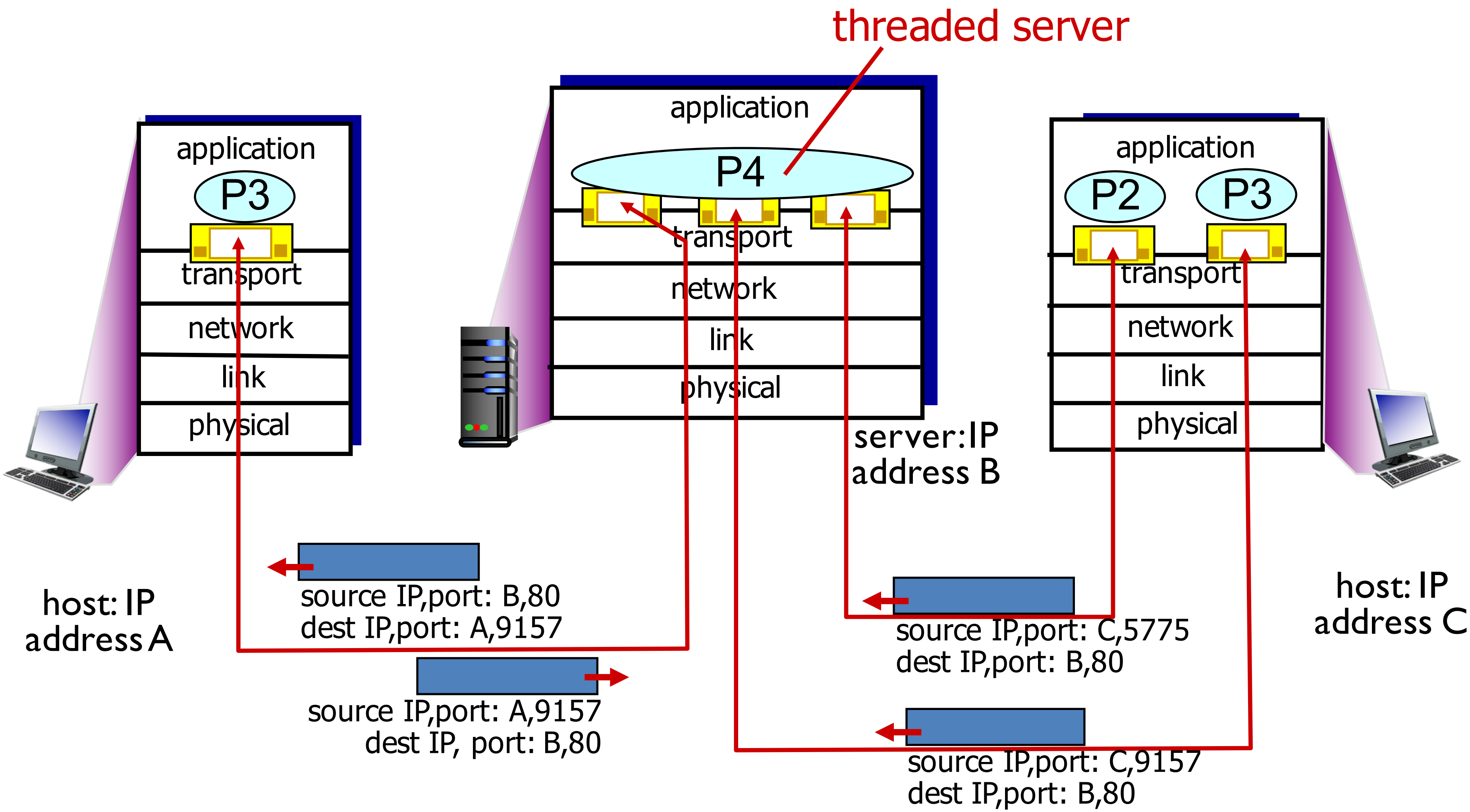
- have different sockets for each connecting client
- non-persistent HTTP will have different socket for each request

Connection-oriented demultiplexing (TCP)



3 segments, all destined to IP address: B,
dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demultiplexing (TCP)



Transport Layer

USER DATAGRAM PROTOCOL

UDP: User Datagram Protocol [RFC 768]

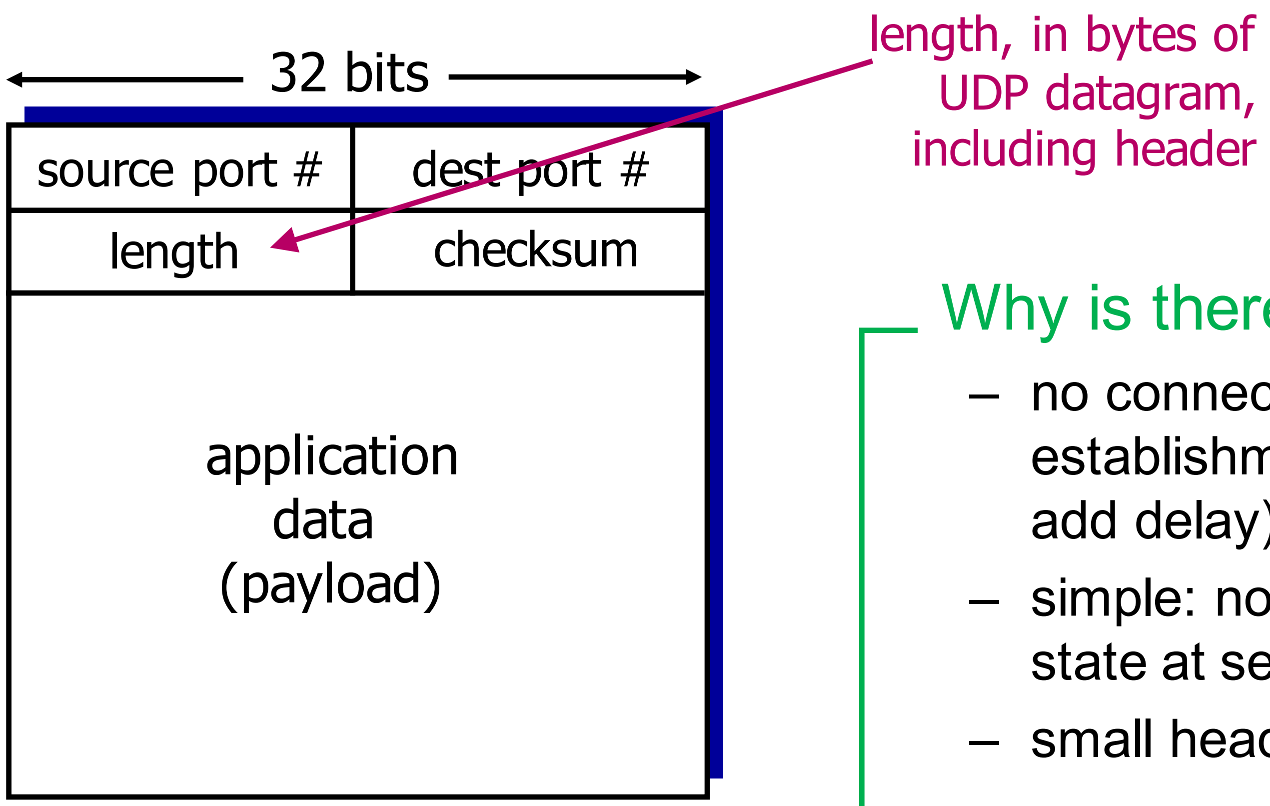
No frills Internet transport protocol

- best effort service, UDP segments may be
  - lost
  - delivered out-of-order to app
- reliable transfer over UDP
  - add reliability at application layer
  - application-specific error recovery!
- UDP uses
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS, SNMP

Connectionless

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

UDP datagram header



UDP datagram format

Why is there a UDP?

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

UDP error detection vs. recovery

Errors

- not just introduced during transmission over links
- can be introduced in memory, at router, at lower layer

UDP does not provide error recovery

- may drop datagram
- may pass datagram data to app with warning

UDP does provide error detection

- it's useful to know something damaged even if don't fix
- Q: How?
  - Checksum

UDP checksum

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

Sender

1. Views datagram contents, including header fields and user data, as sequence of 16-bit integers
   - skip checksum field
2. Computes checksum
   - adds 16-bit integers together using 1s complement arithmetic and then takes 1s complement of result
3. Puts checksum value in UDP checksum field

Receiver

1. Computes its own checksum over datagram including checksum in UDP header
2. Result should equal all 0s if no errors
   - NO: error detected
   - YES: no error detected
   - Q: can there still be errors?

Internet checksum example

Example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Summing these should give 0

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

Q: Why 1s complement? Why check for 0s?

- for efficiency: computed very fast in hardware
- independent of machine endianness

Looking at UDP in Wireshark

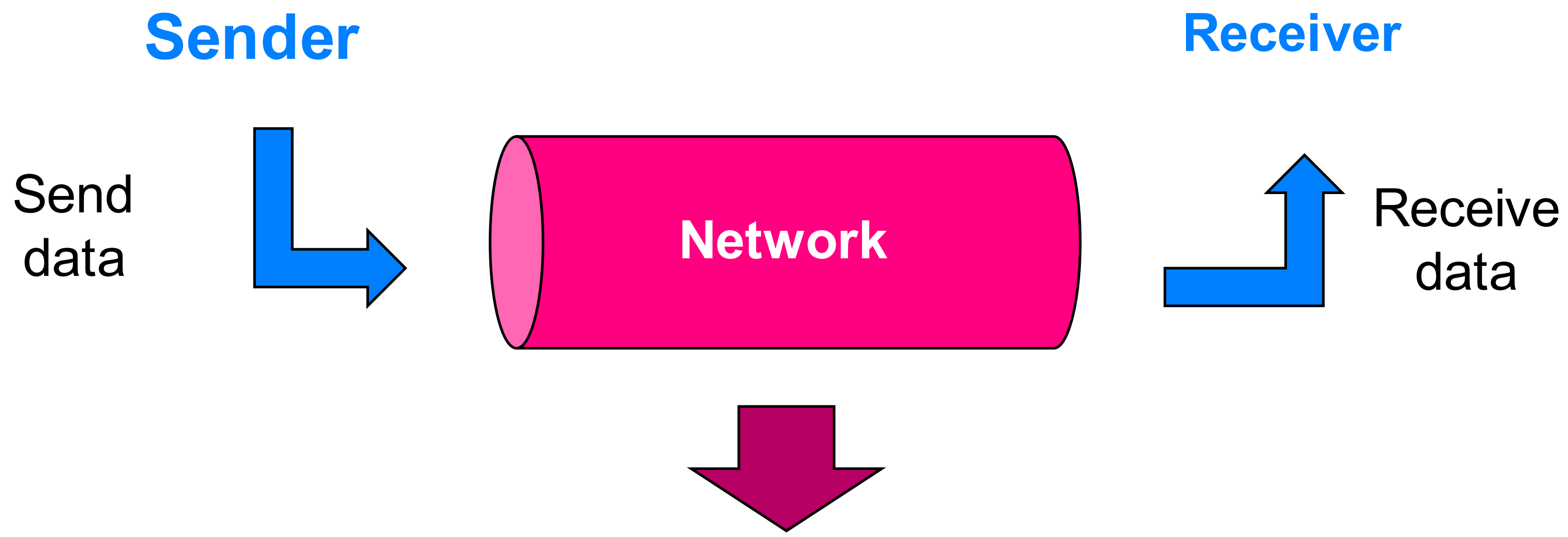
```
▶ Frame 237: 143 bytes on wire (1144 bits), 143 bytes captured (1144 bits) on ir
▶ Ethernet II, Src: JuniperN_1e:18:01 (3c:8a:b0:1e:18:01), Dst: 78:4f:43:73:43:2
▶ Internet Protocol Version 4, Src: intdns.wesleyan.edu (129.133.52.12), Dst: vm
▼ User Datagram Protocol, Src Port: 53 (53), Dst Port: 57332 (57332)
    Source Port: 53
    Destination Port: 57332
    Length: 109
  ▼ Checksum: 0x0f73 [validation disabled]
      [Good Checksum: False]
      [Bad Checksum: False]
    [Stream index: 1]
▶ Domain Name System (response)

0000  78 4f 43 73 43 26 3c 8a  b0 1e 18 01 08 00 45 00   xOCsC&<. ......E.
0010  00 81 87 f4 00 00 3e 11  01 b3 81 85 34 0c 81 85   ......>. ....4...
0020  bb ae 00 35 df f4 00 6d  0f 73 e6 72 81 80 00 01   ...5...m .s.r....
0030  00 01 00 00 00 00 03 32  32 37 03 31 39 30 02 33   .......2 27.190.3
0040  33 02 31 33 07 69 6e 2d  61 64 64 72 04 61 72 70   3.13.in- addr.arp
0050  61 00 00 0c 00 01 c0 0c  00 0c 00 01 00 01 51 8d   a....... ......Q.
0060  00 2d 14 73 65 72 76 65  72 2d 31 33 2d 33 33 2d   .-.serve r-13-33-
0070  31 39 30 2d 32 32 37 05  62 6f 73 35 30 01 72 0a   190-227. bos50.r.
0080  63 6c 6f 75 64 66 72 6f  6e 74 03 6e 65 74 00      cloudfro nt.net.
```

Reliable Data Transport

PRINCIPLES

Why can't we do the following?



Because Internet is unreliable channel

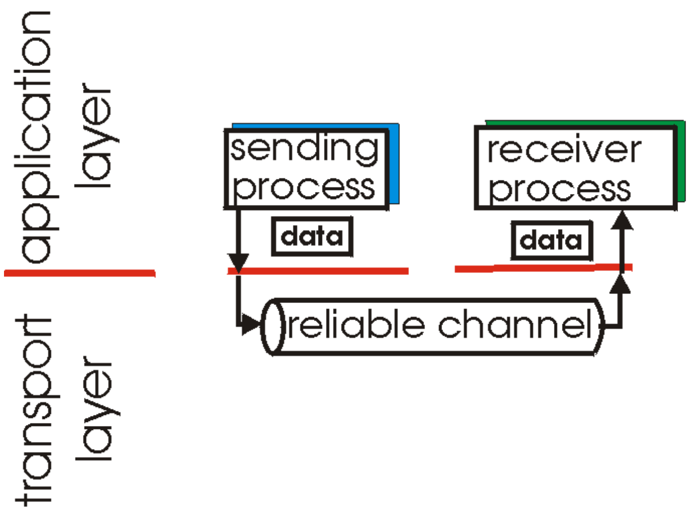
Packets can be corrupted, duplicated, reordered, delayed, lost

Q: What can we do?

Principles of reliable data transfer

Important in application, 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)

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Principles of reliable data transfer

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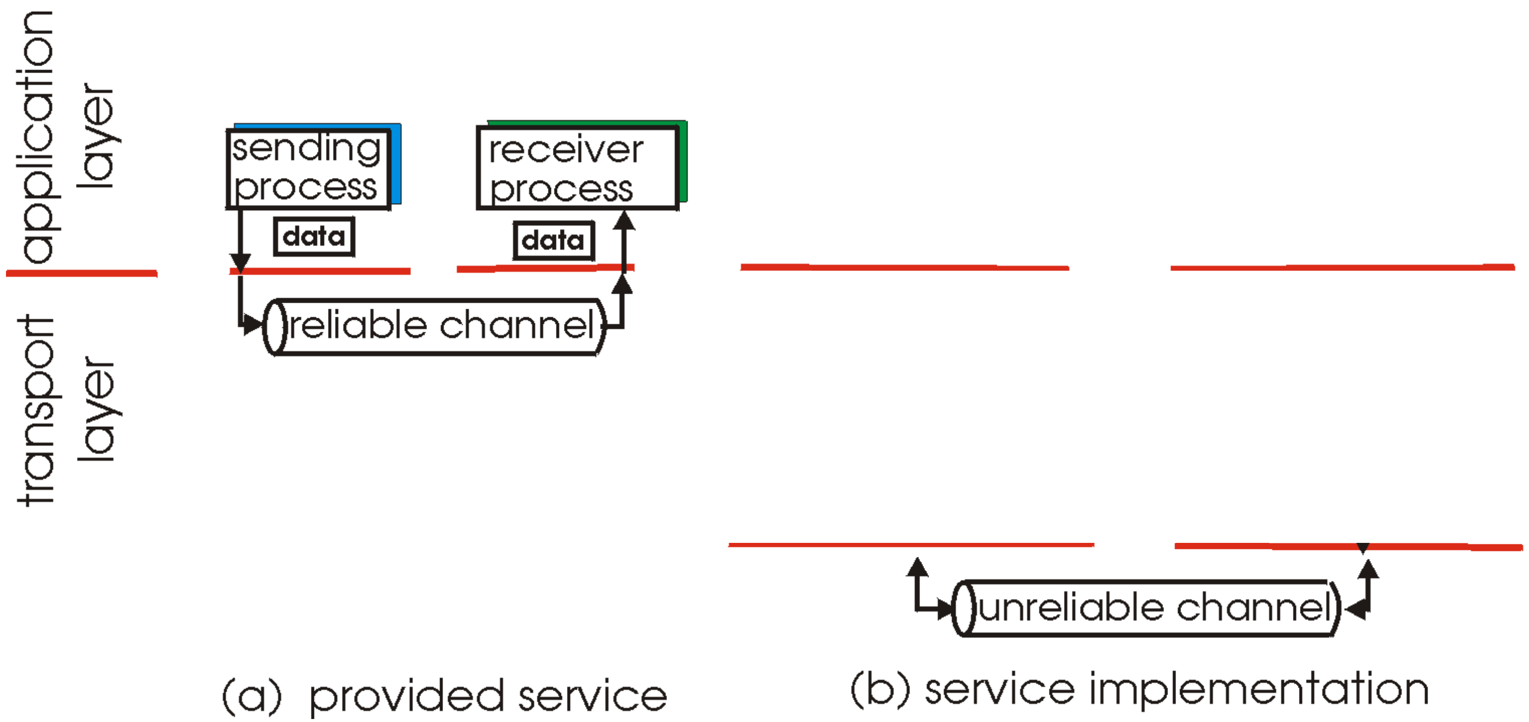

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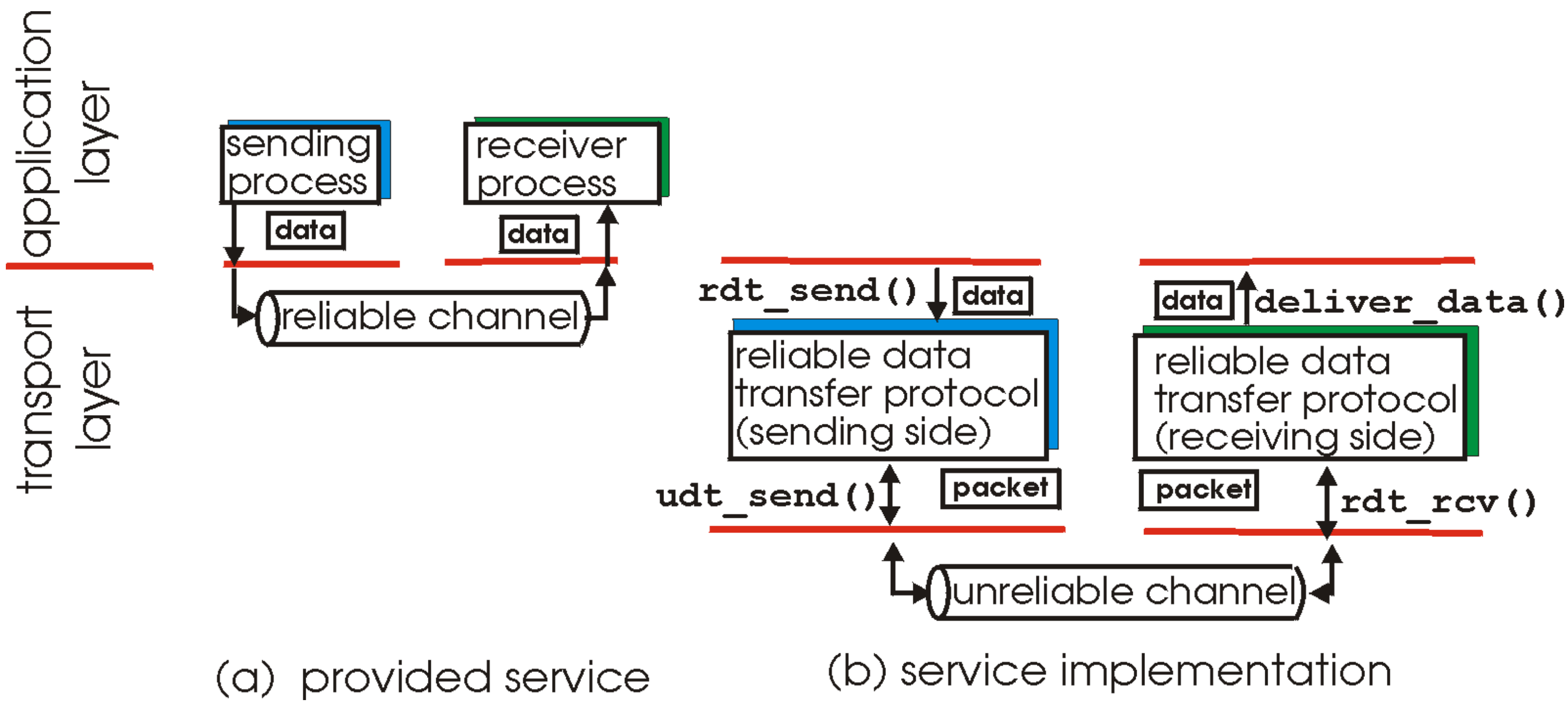
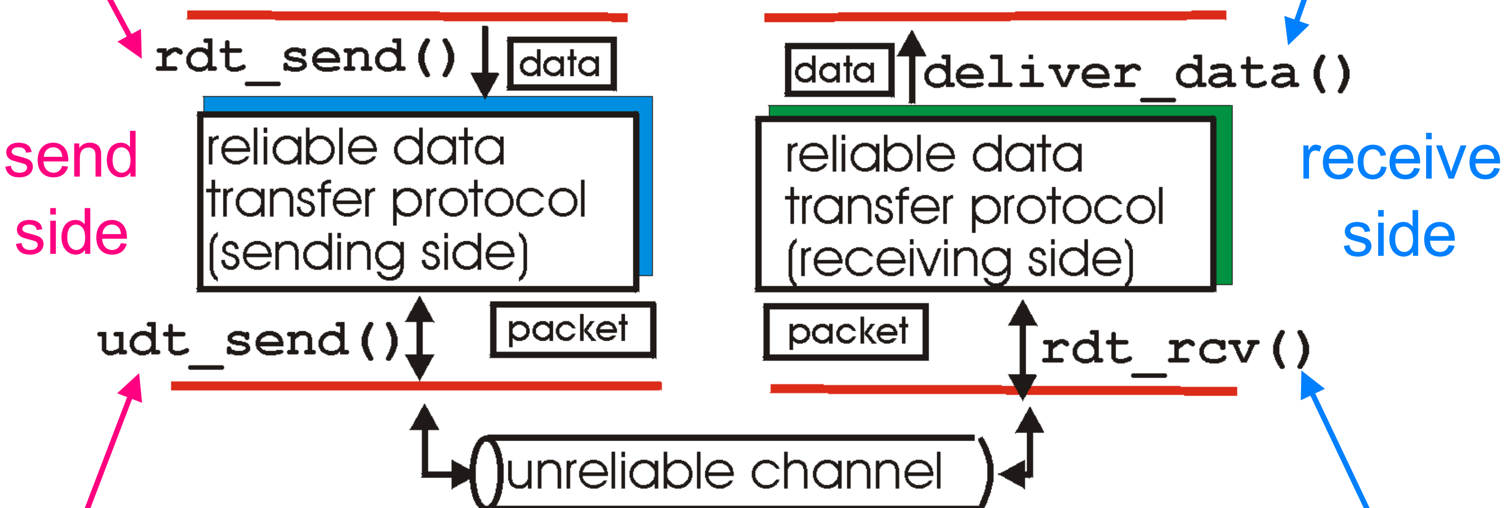


Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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

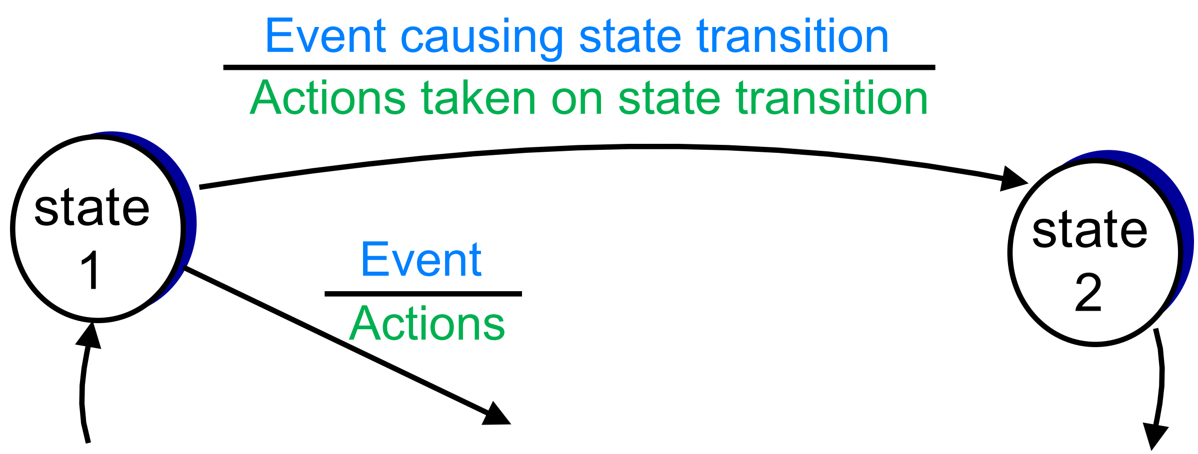
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

Our plan

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

State: when in this state, next state is uniquely determined by next event



Reliable Data Transport

PROTOCOL V1.0

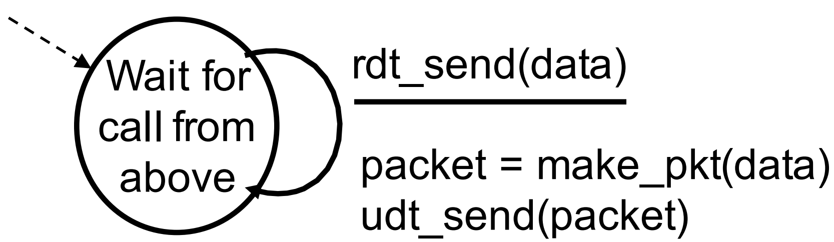
rdt1.0: reliable transfer over a reliable channel

Underlying channel perfectly reliable

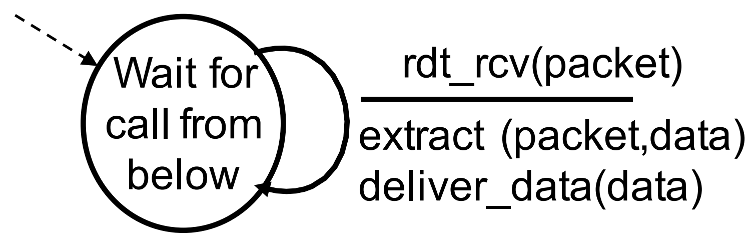
- no bit errors
- no loss of packets

Separate FSMs for sender, receiver:

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



sender



receiver

Reliable Data Transport

PROTOCOL V2.0

rdt2.0: channel with bit errors

Underlying channel may flip bits in packet

- checksum to detect bit errors
- Q: how to recover from errors?

How do humans recover from “errors” during conversation?

rdt2.0: channel with bit errors

Underlying channel may flip bits in packet

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- Q: 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
- feedback
  - control msgs (ACK,NAK) from receiver to sender

Continue rdt2.0 next lecture