Module 8
Transport Layer Protocols

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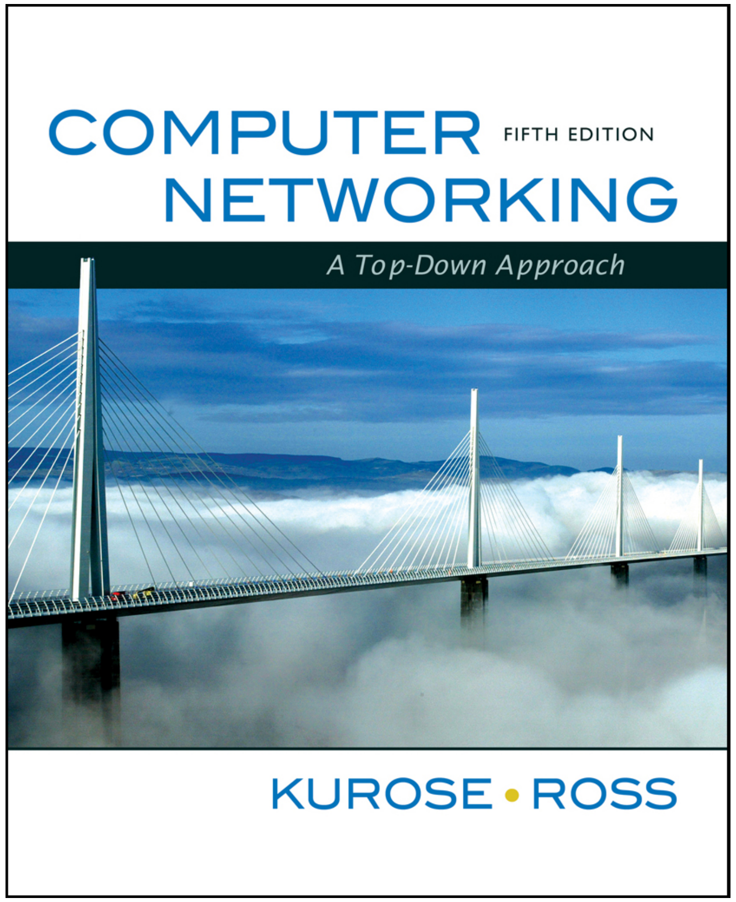

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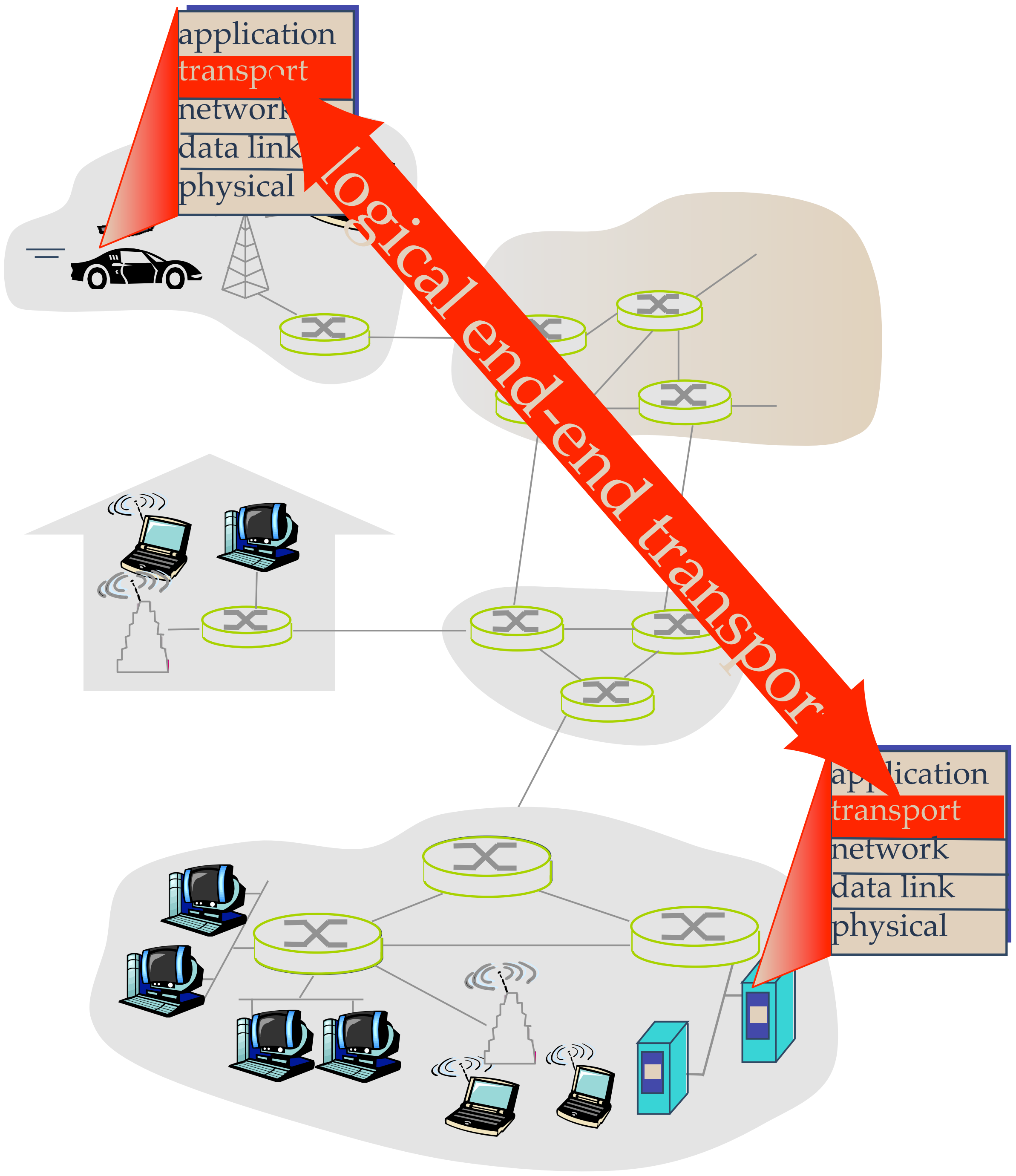
Thanks and enjoy! JFK/KWR

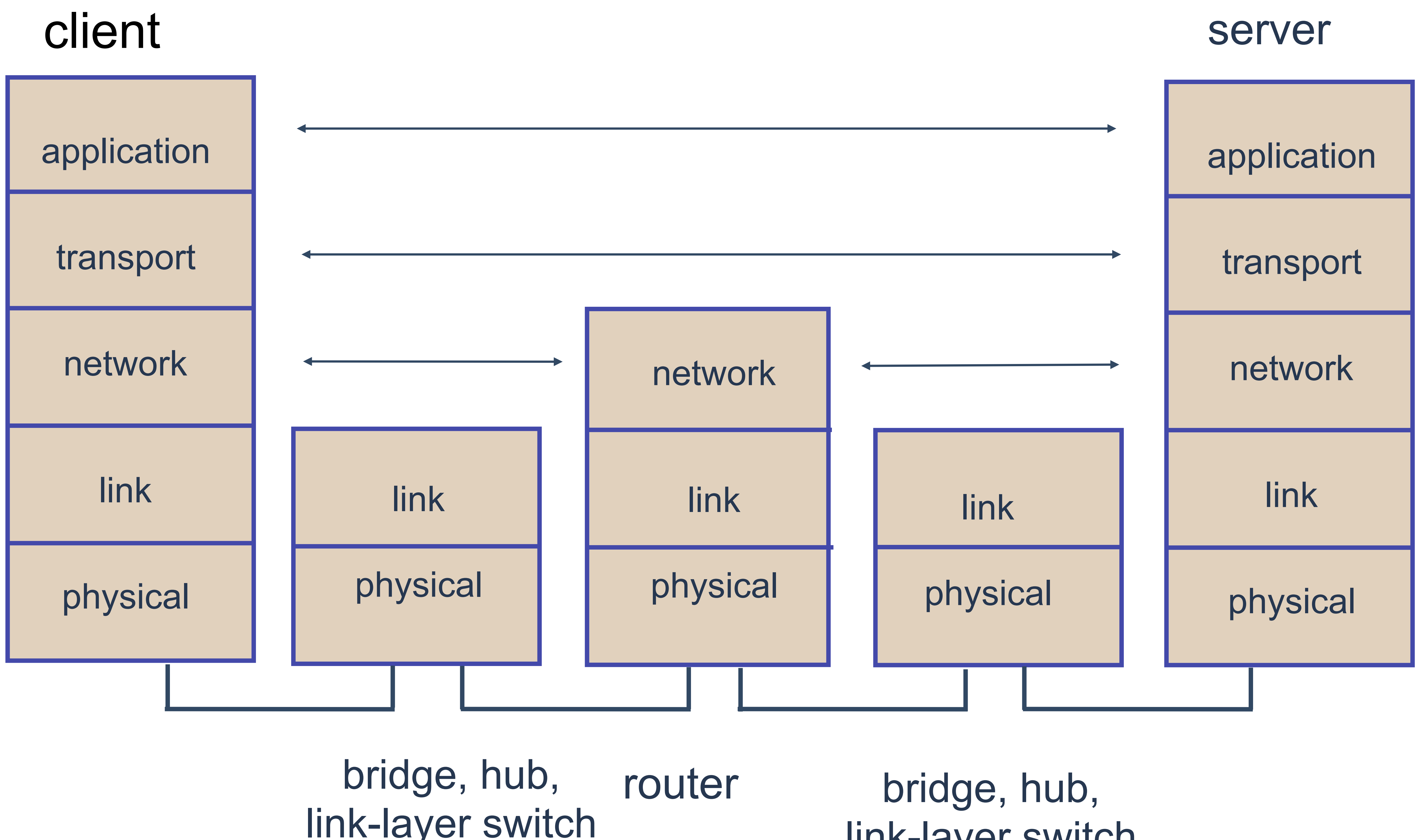
Computer Networking: A Top Down Approach
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

Transport services & 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 Layer is End-to-End



Internet Protocols

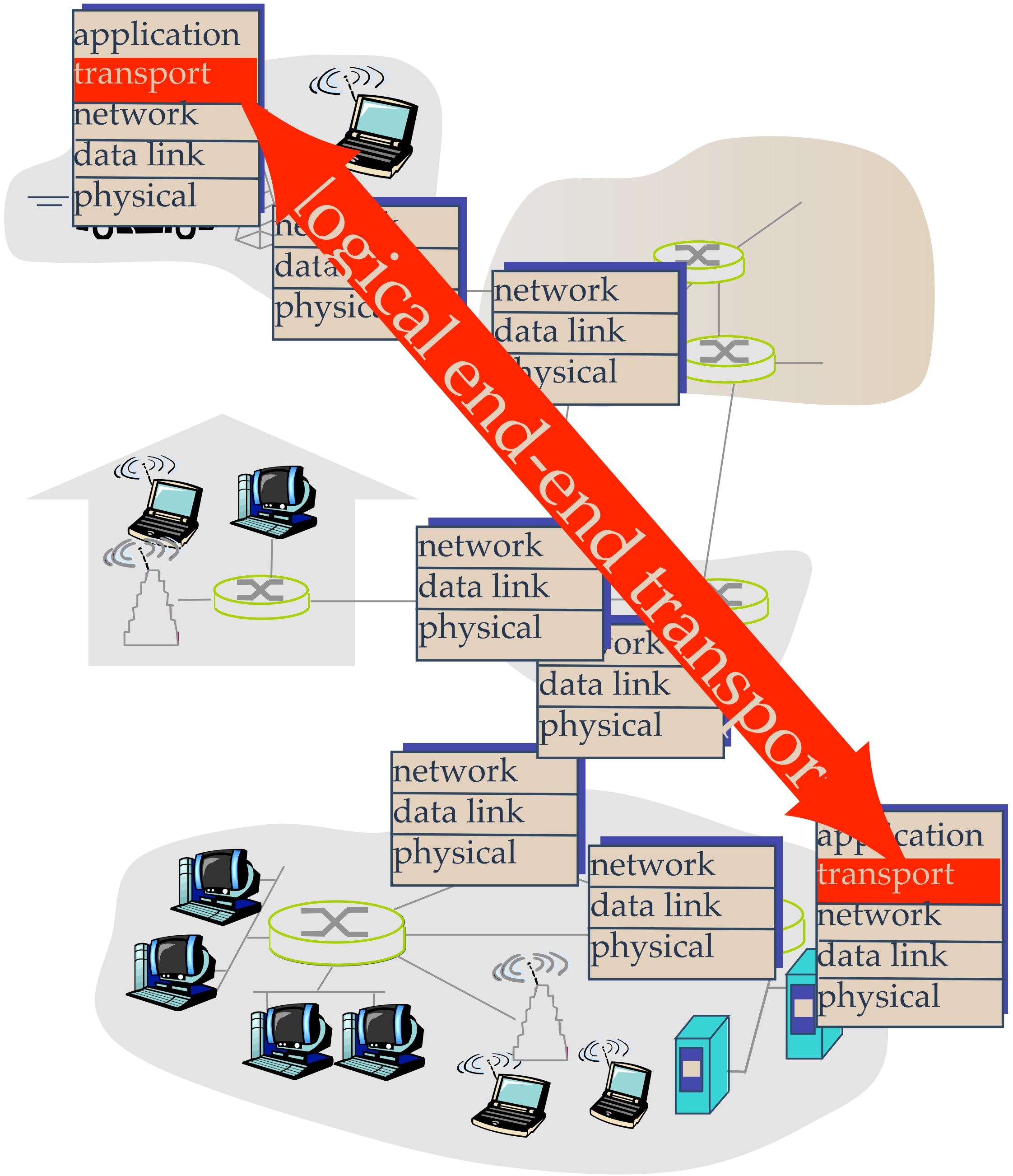
Application	FTP Telnet NFS SMTP HTTP …					
Transport	TCP			UDP		
Network	IP					
Data Link Physical	X.25	Ethernet	Packet Radio	ATM	FDDI	…

Internet Apps: Their Protocols & Transport Protocols

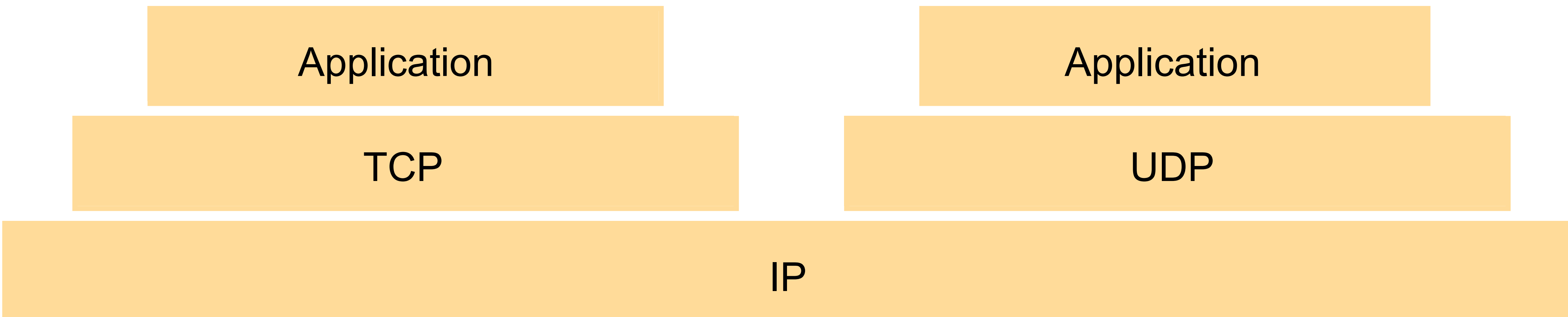
Applications	Data Loss	Throughput	Time Sensitive	Application Layer Protocol	Transport Protocol
Email	No loss	Elastic	No	smtp	TCP
remote terminal access	No loss	Elastic	Yes	telnet	TCP
Web	No loss	Elastic	No	http	TCP
File transfer	No loss	Elastic	No	ftp	TCP
streaming multimedia	Loss tolerant	audio: 5kbps-1Mbps video: 10kbps-5Mbps	Yes, 100's msec	Proprietary	TCP or UDP
Remote file server	No loss	Elastic	No	NFS	TCP or UDP (typically UDP)
Internet telephony	Loss tolerant	Depends on encoding, 32kbps typically	Yes, few secs	SIP, RIP, Prorietary	TCP or UDP (typically UDP)

Internet transport-layer protocols

- UDP: unreliable, unordered delivery
  - Process-to-process data delivery
    - Multiplexing/demultiplexing
  - End-to-end error checking
- TCP: reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- services not available:
  - delay guarantees
  - bandwidth guarantees

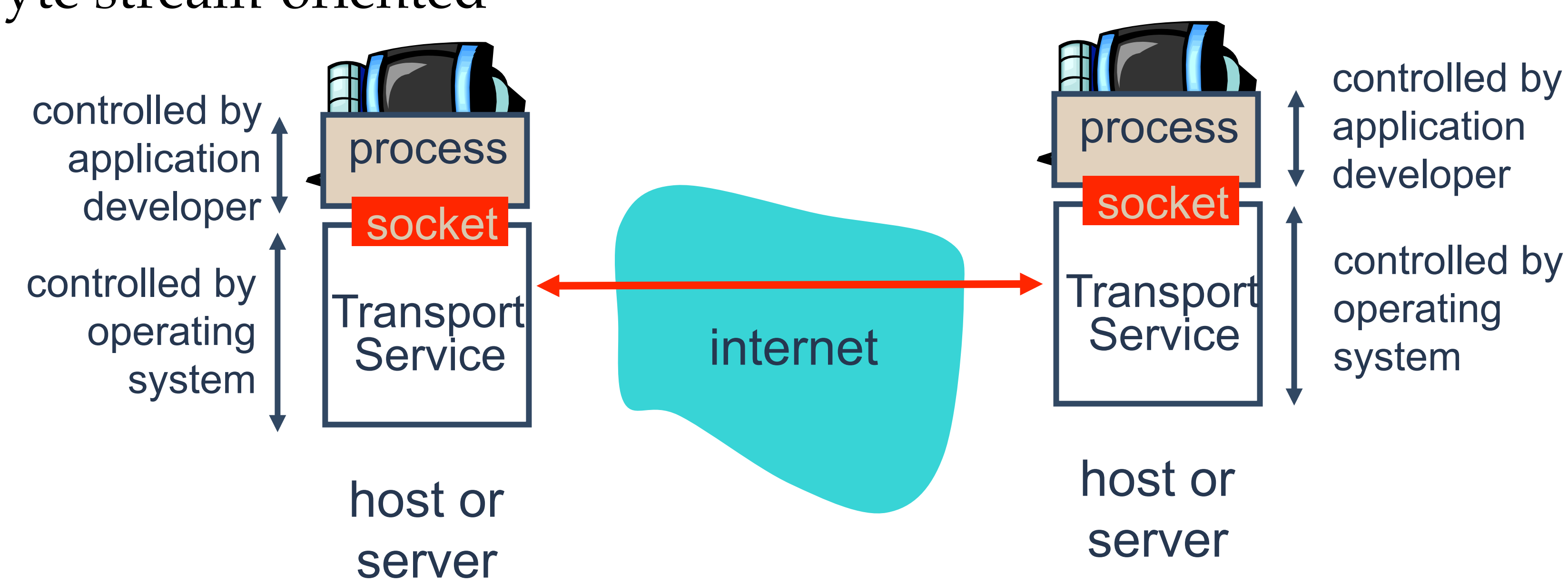


The programmer's conceptual view of a TCP/IP Internet



How Apps Access Transport Services

- Through sockets
- 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
- 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



UDP: User Datagram Protocol

- “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

Client/server socket interaction: UDP

Server (running on **hostid**)

create socket,
port= x.
serverSocket =
DatagramSocket()

read datagram from
serverSocket

write reply to
serverSocket
specifying
client address,
port number

Client

create socket,
clientSocket =
DatagramSocket()

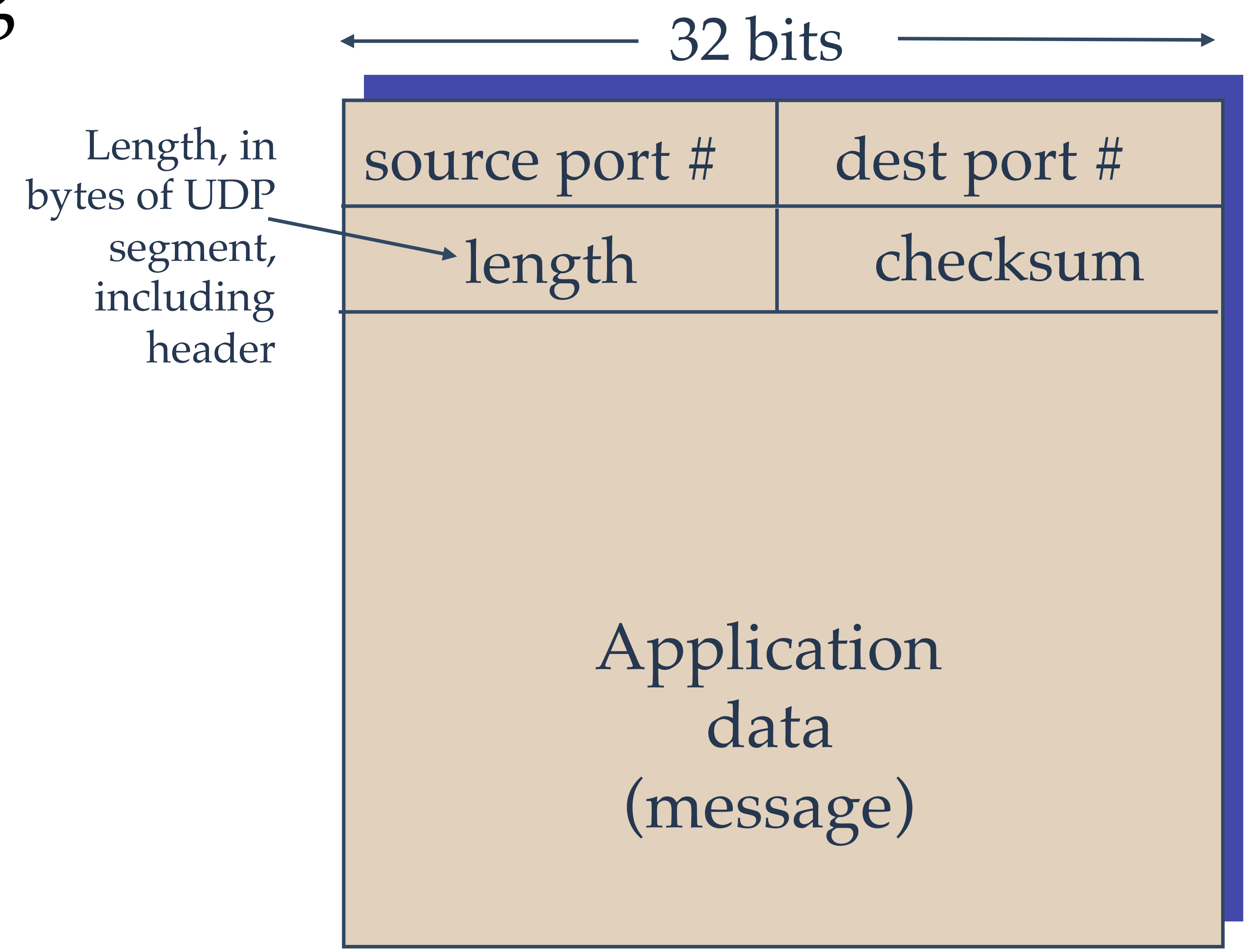
Create datagram with server IP and
port=x; send datagram via
clientSocket

read datagram from
clientSocket

close
clientSocket

UDP Use and Format

- 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:

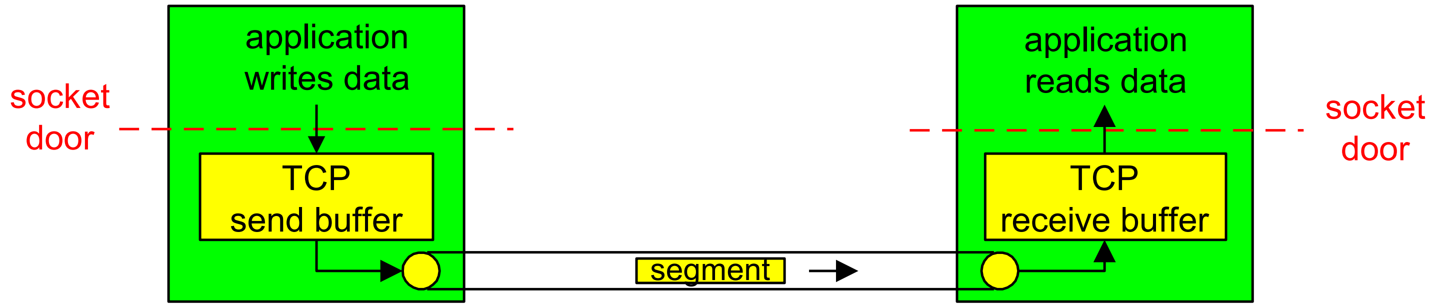
- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - ➡ NO - error detected
  - ➡ YES - no error detected.

TCP: Transport Control Protocol

- 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) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

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

TCP Client/server socket interaction

Server (running on **hostid**)

Client

create socket,
port=**x**, for
incoming request:
welcomeSocket =
ServerSocket()

TCP
connection setup

wait for incoming
connection request
connectionSocket =
welcomeSocket.accept()

create socket,
connect to **hostid**, port=**x**
clientSocket =
Socket()

send request using
clientSocket

read request from
connectionSocket

write reply to
connectionSocket

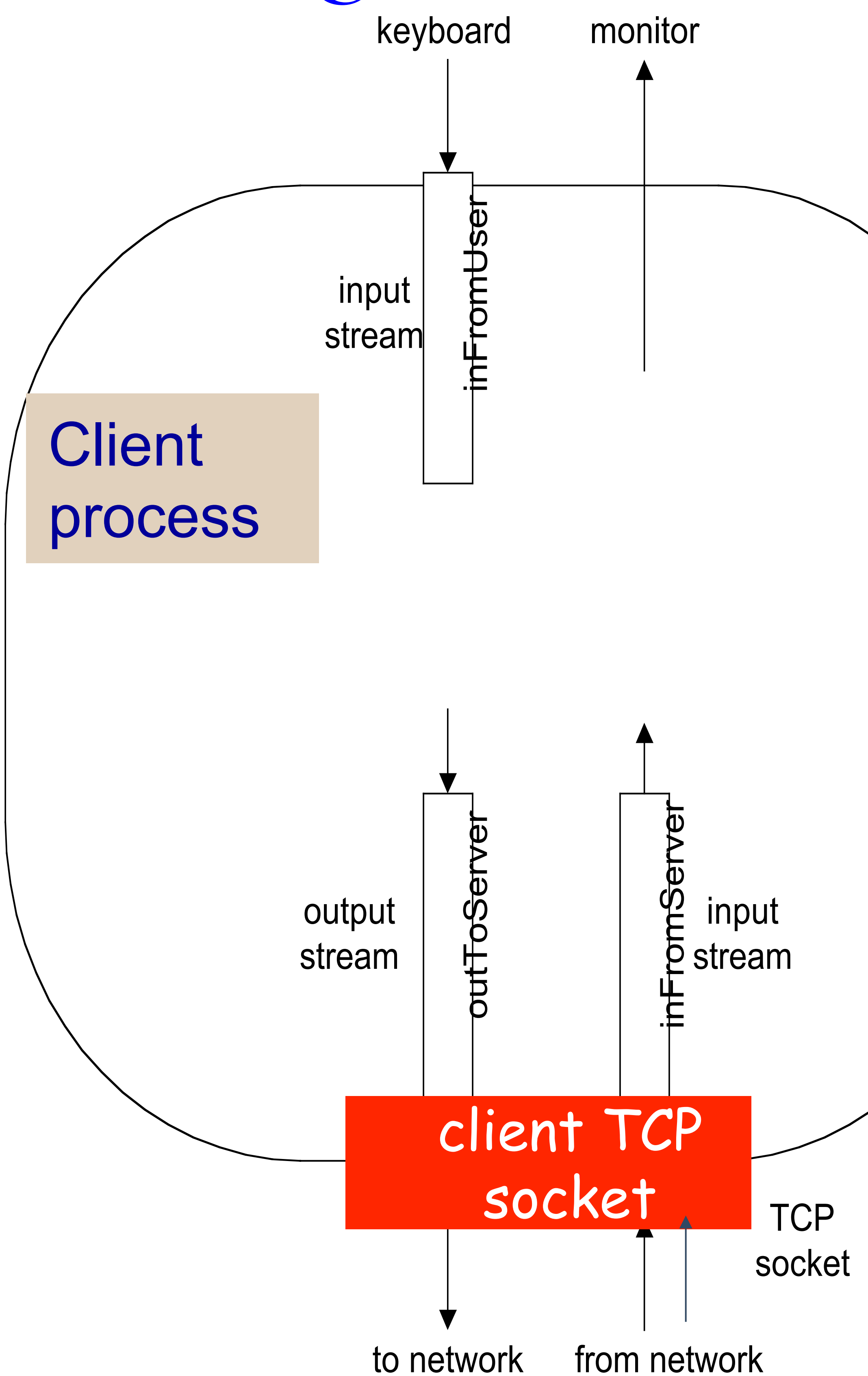
read reply from
clientSocket

close
connectionSocket

close
clientSocket

Stream Jargon

- stream is a sequence of characters that flow into or out of a process.
- input stream is attached to some input source for the process, e.g., keyboard or socket.
- output stream is attached to an output source, e.g., monitor or socket.



Socket programming with TCP

Example client-server app:

1) client reads line from standard input (**inFromUser** stream) , sends to server via socket (**outToServer** stream)
2) server reads line from socket
3) server converts line to uppercase, sends back to client
4) client reads, prints modified line from socket (**inFromServer** stream)

Example: TCP Java client

```
import java.io.*;
import java.net.*;
class TCPClient {

    public static void main(String argv[]) throws Exception
    {
        String sentence;
        String modifiedSentence;

        BufferedReader inFromUser =
          new BufferedReader(new InputStreamReader(System.in));

        Socket clientSocket = new Socket("hostname", 6789);

        DataOutputStream outToServer =
          new DataOutputStream(clientSocket.getOutputStream());
```

This package defines Socket() and ServerSocket() classes

server name, e.g., www.umass.edu

server port #

create input stream

create clientSocket object of type Socket, connect to server

create output stream attached to socket

Example: TCP Java client (cont'd)

create input stream attached to socket

send line to server

read line from server

close socket
(clean up behind yourself!)

```
        BufferedReader inFromServer =
          new BufferedReader(new
          InputStreamReader(clientSocket.getInputStream()));

         sentence = inFromUser.readLine();

         outToServer.writeBytes(sentence + '\n');

         modifiedSentence = inFromServer.readLine();

         System.out.println("FROM SERVER: " + modifiedSentence);

         clientSocket.close();

       }
   }
```

Example: TCP Java server

create
welcoming socket
at port 6789

wait, on welcoming
socket accept() method
for client contact create,
new socket on return

create input
stream, attached
to socket

```
import java.io.*;
import java.net.*;

class TCPServer {

  public static void main(String argv[]) throws Exception
   {
    String clientSentence;
    String capitalizedSentence;

    ServerSocket welcomeSocket = new ServerSocket(6789);

    while(true) {

       Socket connectionSocket = welcomeSocket.accept();

      BufferedReader inFromClient =
        new BufferedReader(new
        InputStreamReader(connectionSocket.getInputStream()));
```

Example: TCP Java server (cont'd)

create output stream, attached to socket

```
DataOutputStream  outToClient =
  new DataOutputStream(connectionSocket.getOutputStream());
```

read in line from socket

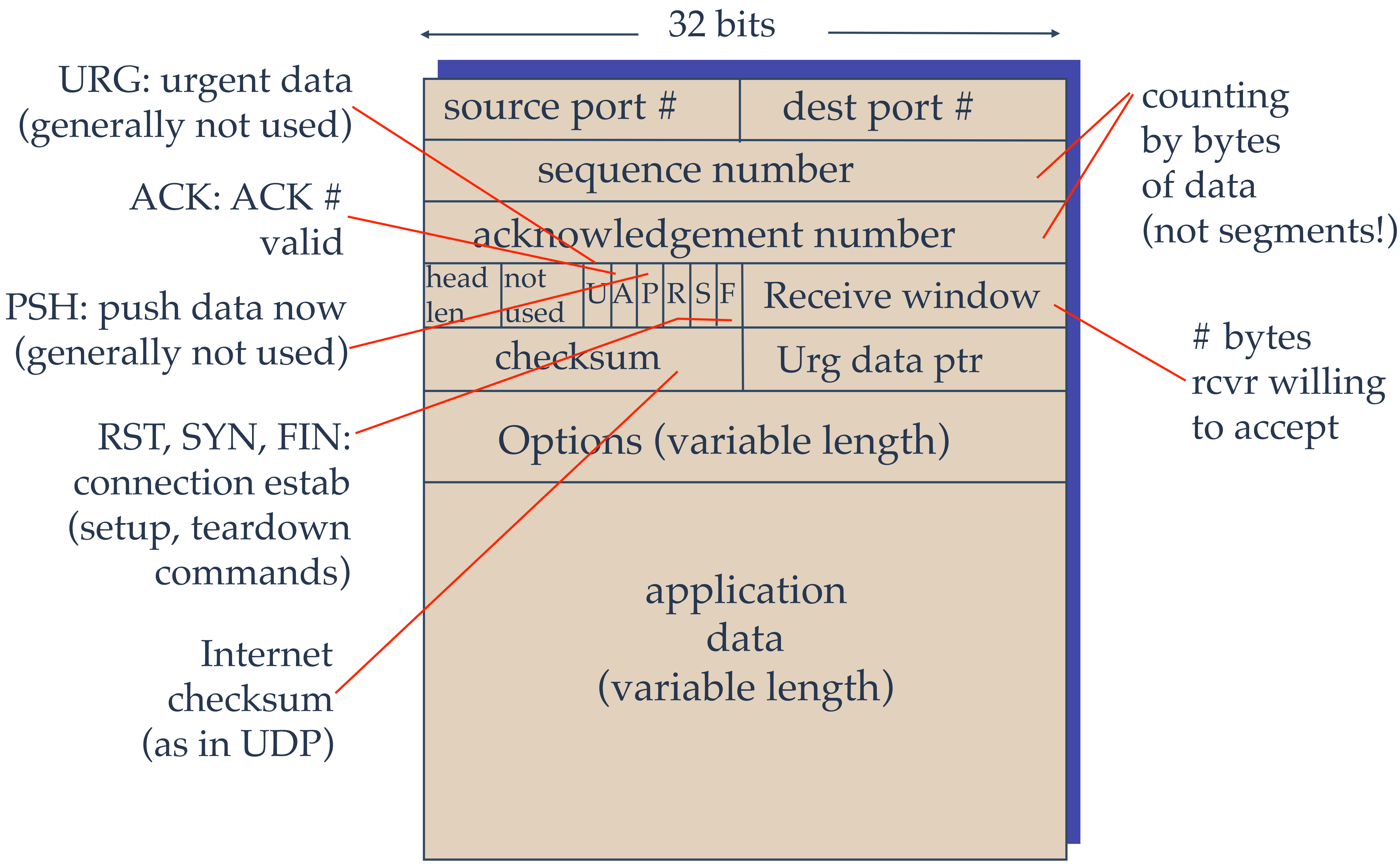
```
clientSentence = inFromClient.readLine();

capitalizedSentence = clientSentence.toUpperCase() + '\n';
```

write out line to socket

```
outToClient.writeBytes(capitalizedSentence);
      }
    }
}
```

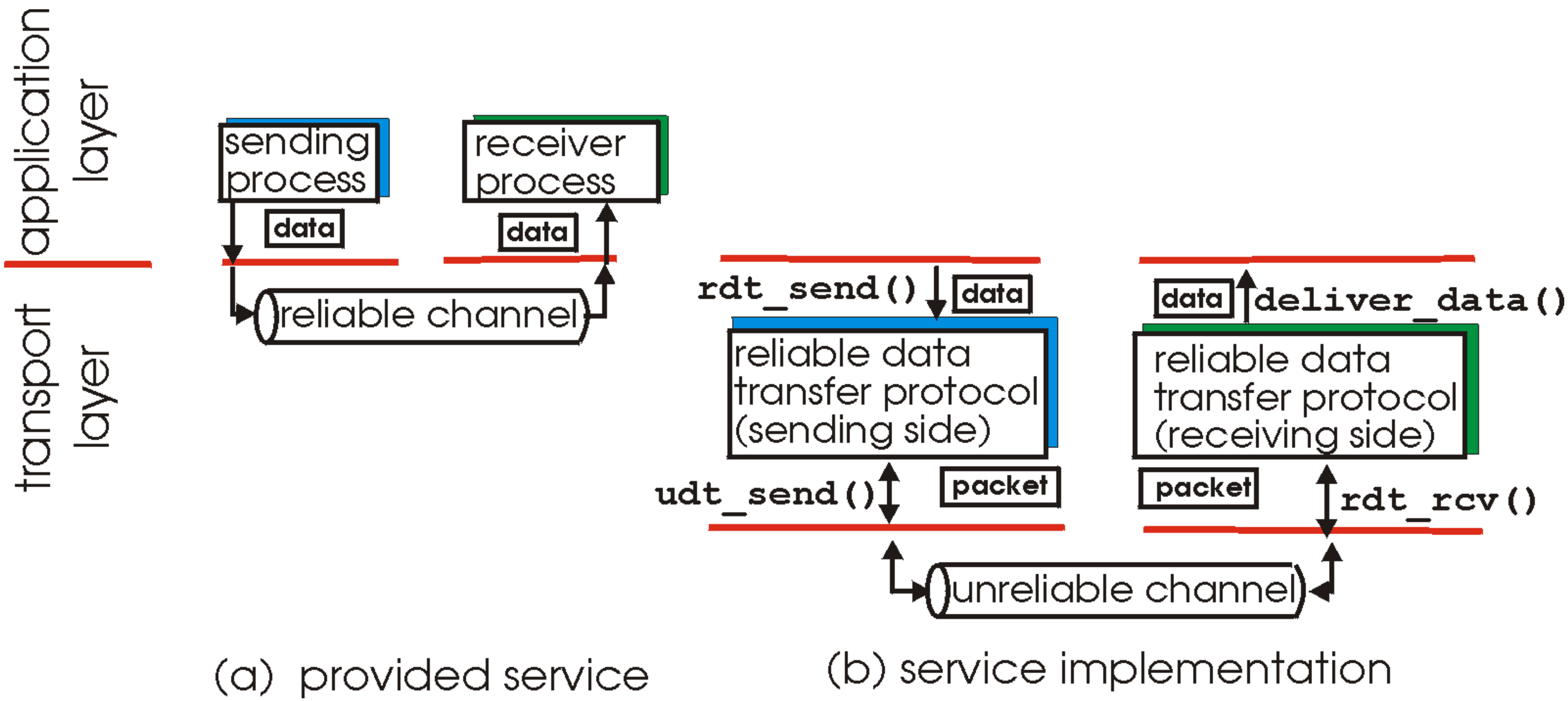
end of while loop, loop back and wait for another client connection

TCP segment structure



Principles of Reliable Data Transfer

- important in app., transport, link layers

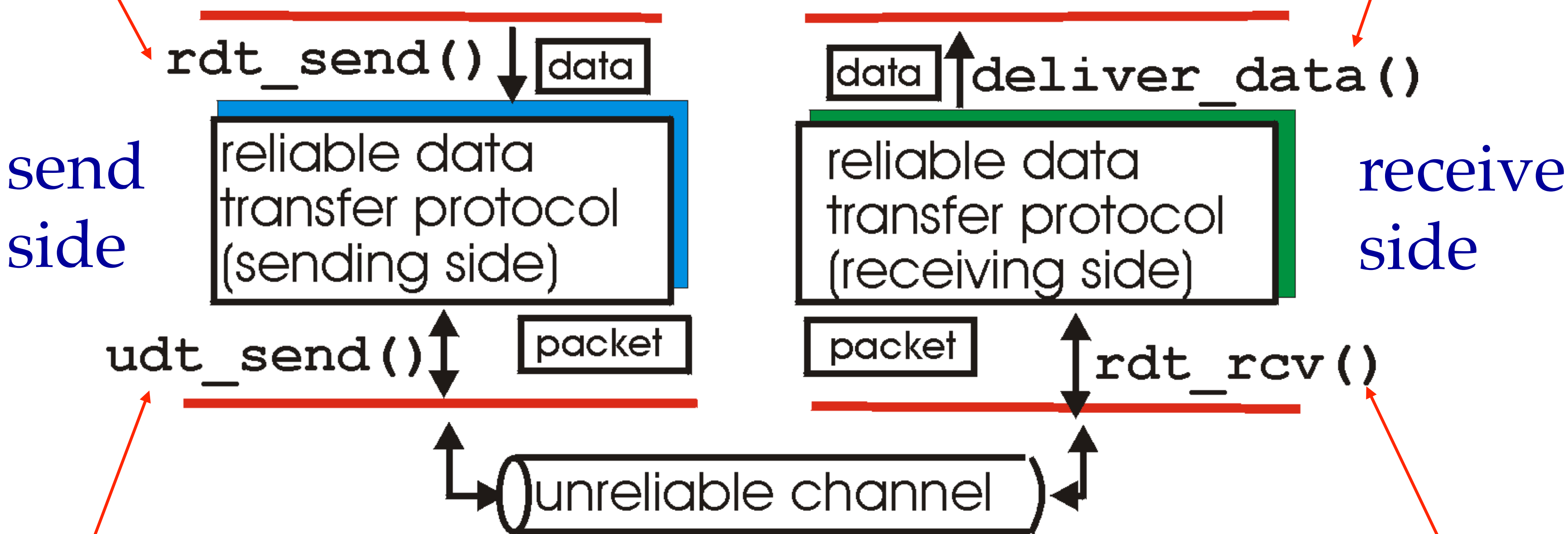


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol
- Note: slides use the term "packet" but at transport layer, these are segments

Principles of Relialable Data Transfer

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

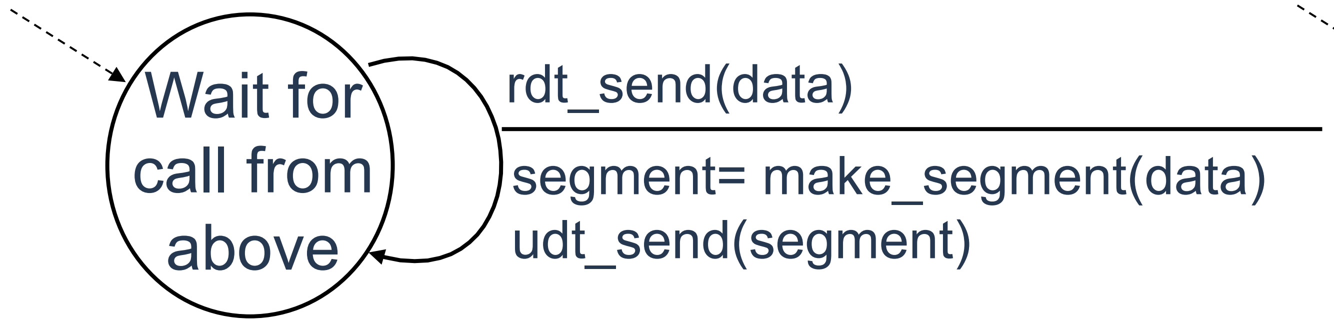


udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

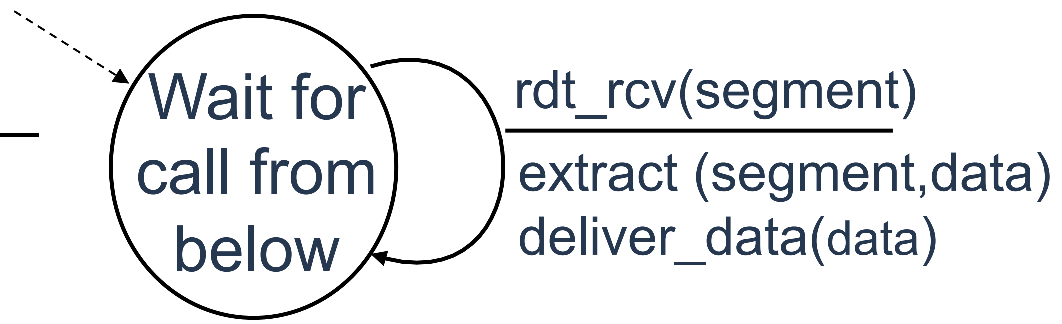
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable 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



sender



receiver

What can go wrong (1)?

- Underlying channel may flip bits in segment
- Error detection:
  - ➡ Checksum to detect bit errors
- Recovering from errors:
  - ➡ *acknowledgements (ACKs):* receiver explicitly tells sender that segment received OK
  - ➡ *negative acknowledgements (NAKs):* receiver explicitly tells sender that segment had errors
  - ➡ sender retransmits segment on receipt of NAK
- Stop-and-wait
  - ➡ Sender sends one segment, then waits for the receiver to respond
  - ➡ We will come back to this later

Handling duplicates

- What happens if ACK/NAK corrupted?
  - ➡ sender doesn't know what happened at receiver!
  - ➡ can't just retransmit: possible duplicate
- Sender retransmits current segment if ACK/NAK garbled
- Sender adds sequence number to each segment
  - ➡ For stop-and-go protocol a 1-bit sequence number with modulo-2 arithmetic is sufficient
- Receiver discards (does not deliver up to the application) duplicate segments

What can go wrong (2)?

- Segments (data or ACK) may be lost
- Sender waits a “reasonable” amount of time for ACK
  - Retransmits if no ACK received in this time
  - If segment (data or ACK) simply delayed (not lost)
    - Retransmission will be duplicate, but use of sequence numbers already handles this
    - Receiver must specify the sequence number of segment being ACKed
  - Requires countdown timer
- Sequence numbers
  - For data: byte stream “number” of first byte in segment’s data
  - For ACKs: if pipelined, segment number of next byte expected from other side
    - Cumulative ACK

What can go wrong (3)?

- Data segments may come out of order
- TCP specification does not say what to do
- Two alternatives
  - ➡ Receiver discards out of order segments
    - ✦ Simplifies receiver design
    - ✦ Wastes network bandwidth
  - ➡ Receiver keeps out of order segments and fills in the missing ones when they arrive
    - ✦ Usually this is what is implemented

TCP Sender Events

data rcvd from app:

- Create segment with sequence number
- start timer if not already running (think of timer as for oldest unacked segment)
- expiration interval: `TimeOutInterval`

timeout:

- retransmit segment that caused timeout
- restart timer

Ack rcvd:

- If acknowledges previously unacked segments
  - ➡ update what is known to be acked
  - ➡ start timer if there are outstanding segments

TCP sender
(simplified)

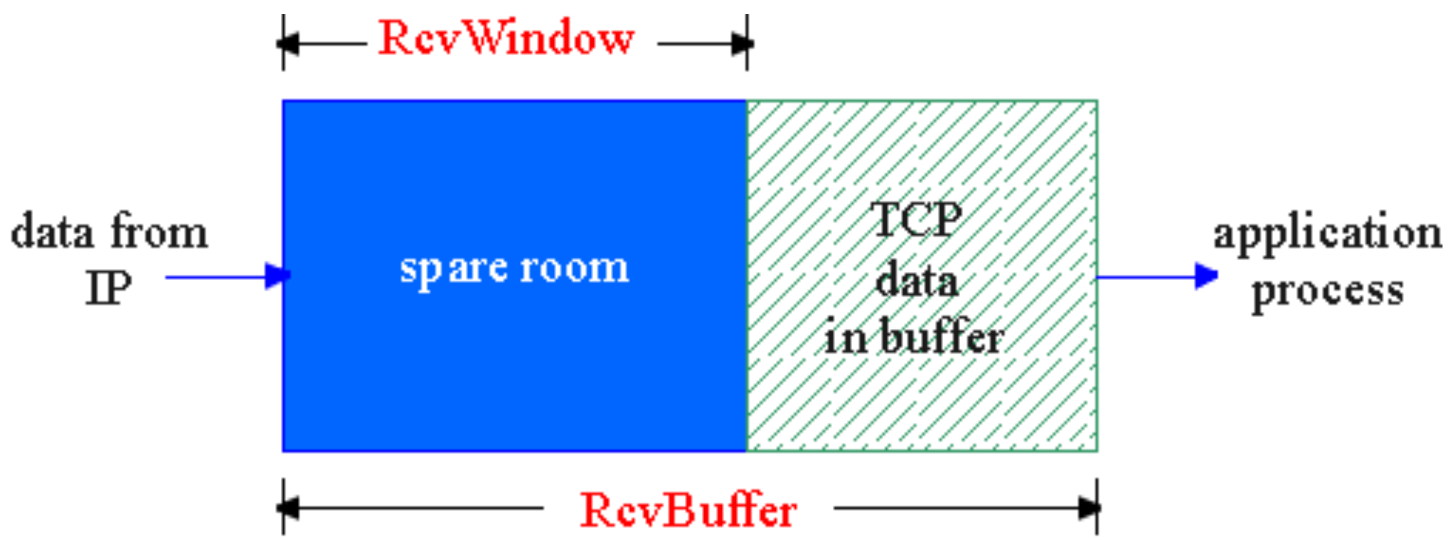
```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
  switch(event)

  event: data received from application above
      create TCP segment with sequence number NextSeqNum
      if (timer currently not running)
           start timer
      pass segment to IP
      NextSeqNum = NextSeqNum + length(data)

  event: timer timeout
      retransmit not-yet-acknowledged segment with
             smallest sequence number
      start timer

  event: ACK received, with ACK field value of y
      if (y > SendBase) {
           SendBase = y
          if (there are currently not-yet-acknowledged segments)
                  start timer
          }

 }  /* end of loop forever */
```

TCP Round Trip Time and Timeout

- How to set TCP timeout value?
  - longer than RTT
    - but RTT varies
  - too short: premature timeout
    - unnecessary retransmissions
  - too long: slow reaction to segment loss
- How to estimate RTT?
  - **`SampleRTT`**: measured time from segment transmission until ACK receipt
    - ignore retransmissions
  - **`SampleRTT`** will vary, want estimated RTT “smoother”
    - average several recent measurements, not just current **`SampleRTT`**

TCP Flow Control

- receive side of TCP connection has a receive buffer:



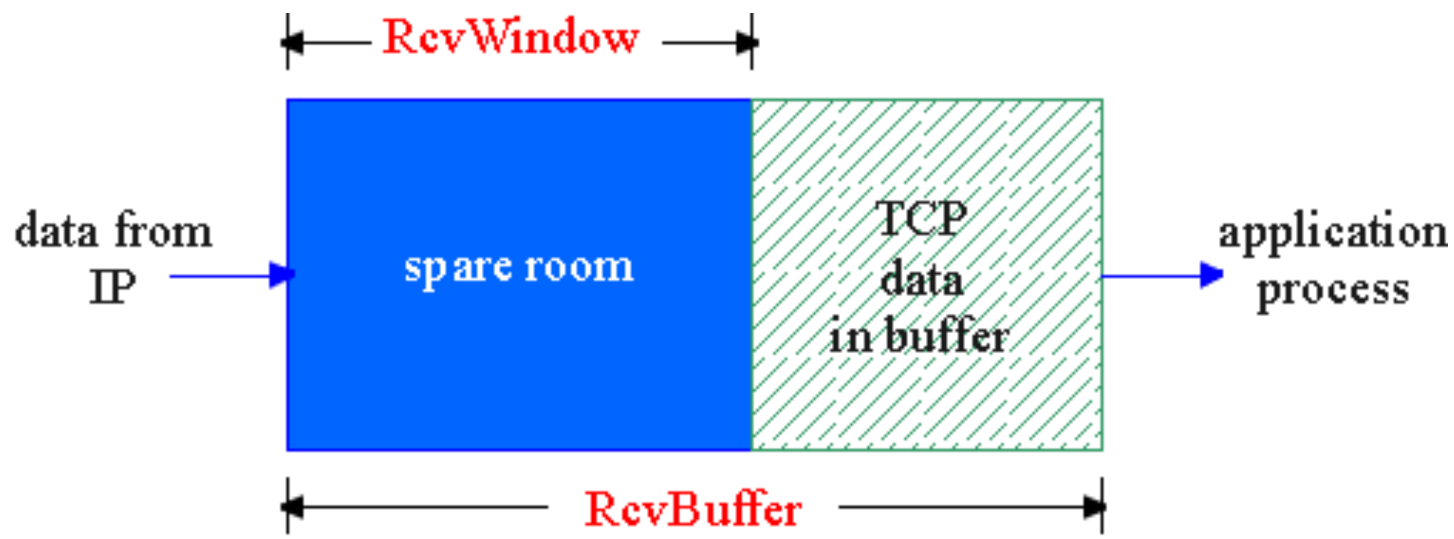
- app process may be slow at reading from buffer

flow control

sender won't overflow receiver's buffer by transmitting too much, too fast

- speed-matching service: matching the send rate to the receiving app's drain rate

TCP Flow control: how it works



- spare room in buffer (ignoring out-of-order segments)

  `RcvWindow = RcvBuffer-[LastByteRcvd - LastByteRead]`

- Receiver advertises spare room by including value of **`RcvWindow`** in segments
- Sender limits unACKed data to **`RcvWindow`**
  - ➡ guarantees receive buffer doesn't overflow

TCP Connection Management

Recall: TCP is a connection-oriented protocol

- sender, receiver establish "connection" before exchanging data segments
- initialize TCP variables:
  - Sequence numbers
  - 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

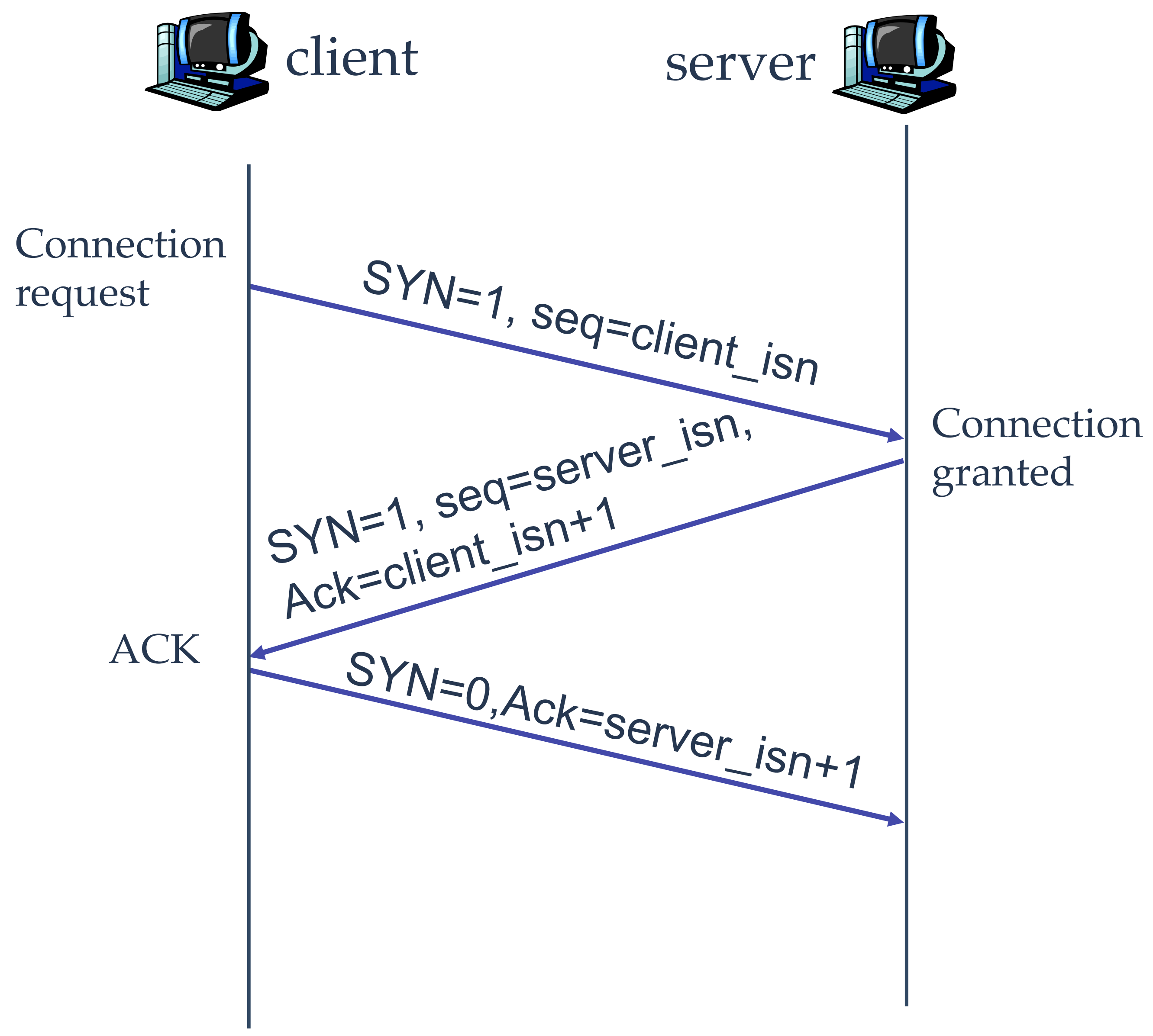
Step 1: client host sends TCP SYN segment to server

- specifies initial sequence number
- no data

Step 2: server host receives SYN, replies with SYNACK segment

- server allocates buffers
- specifies server initial sequence number

Step 3: client receives SYNACK, replies with ACK segment, which may contain data



Closing a TCP Connection

client closes socket:

```
clientSocket.close();
```

Step 1: client end system sends TCP FIN control segment to server

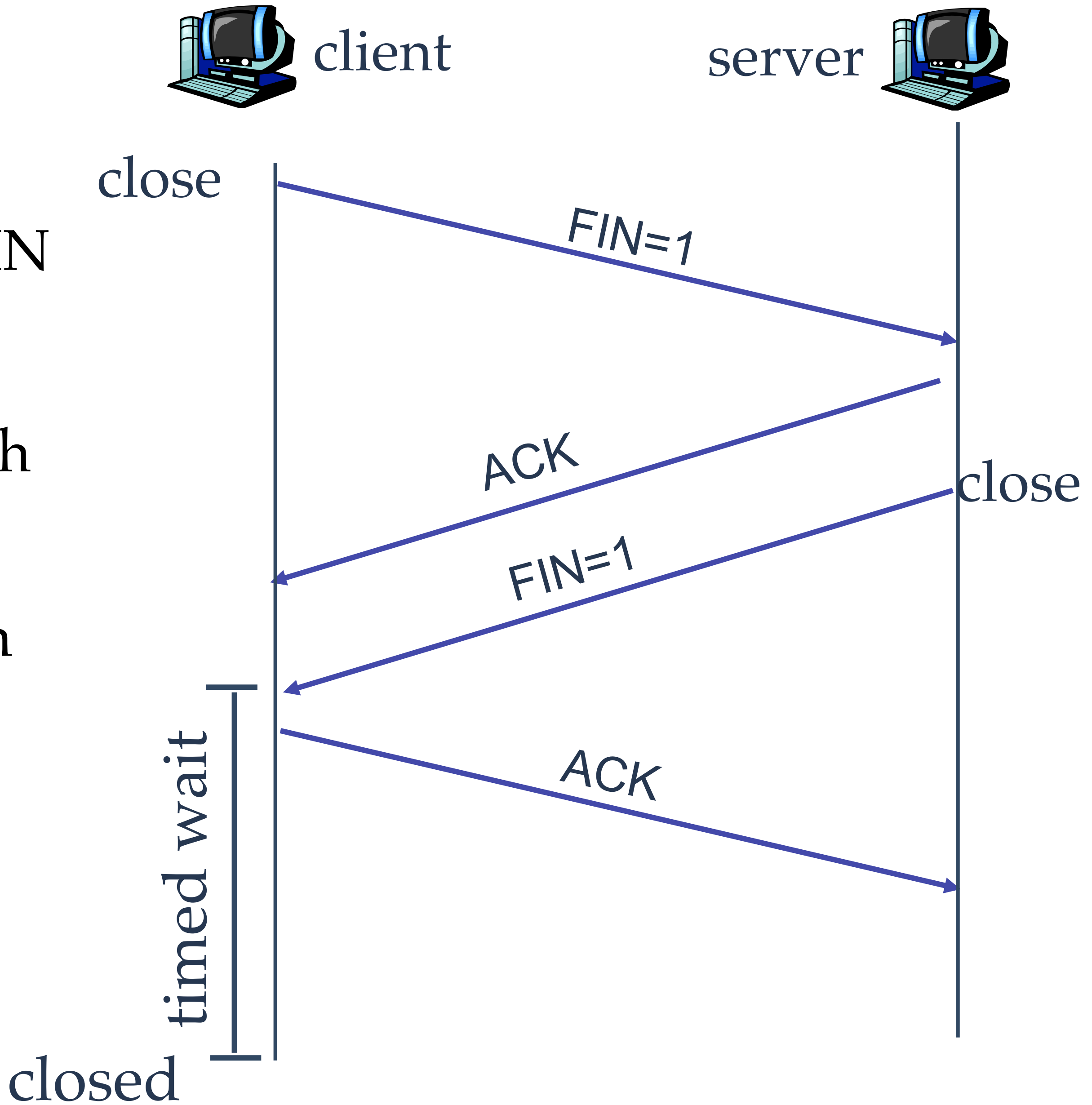
Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.

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.

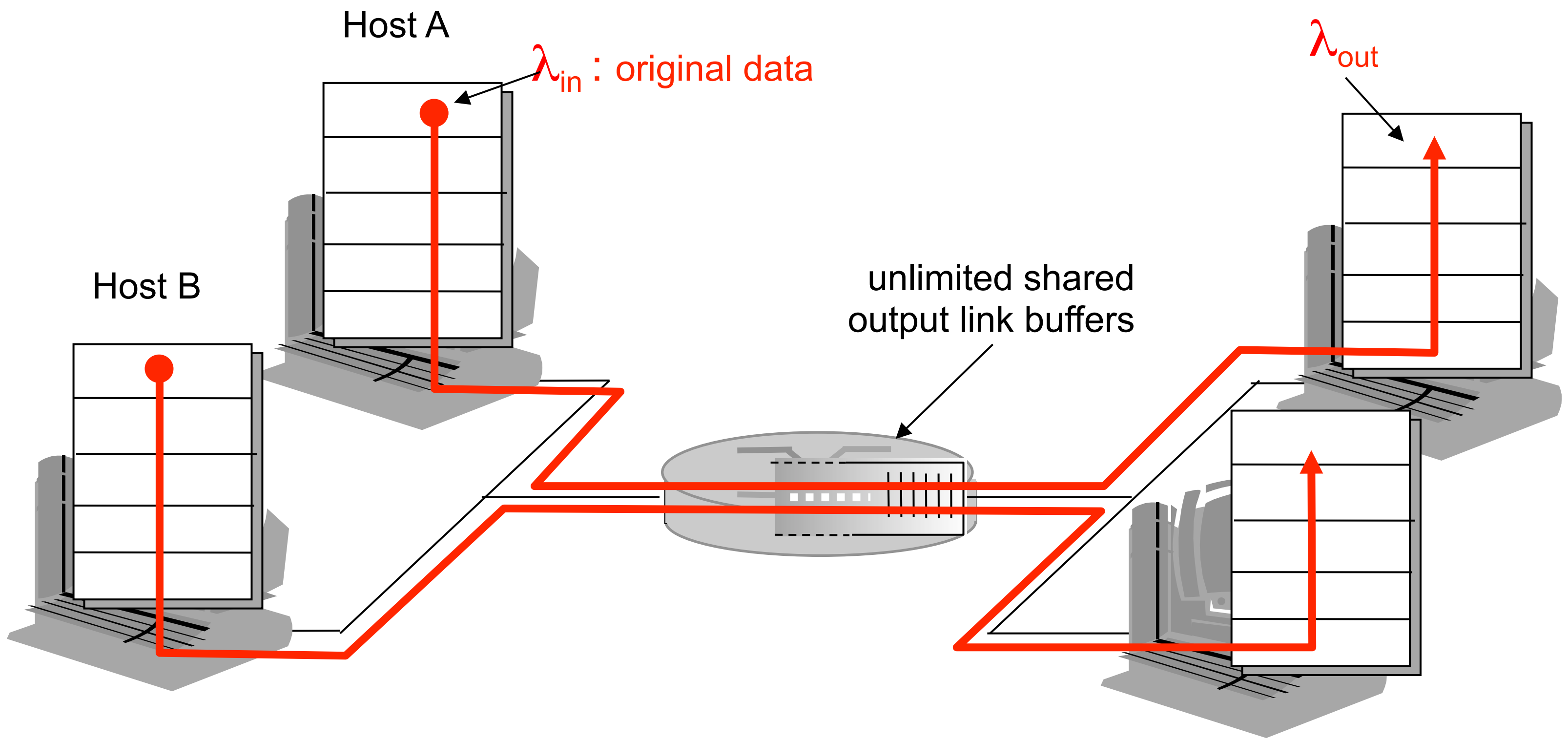


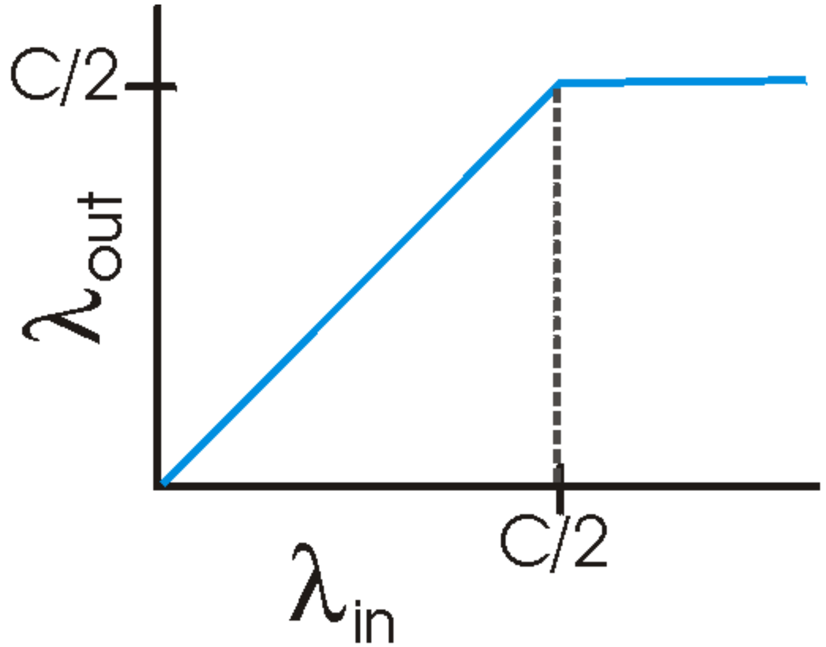
Principles of Congestion Control

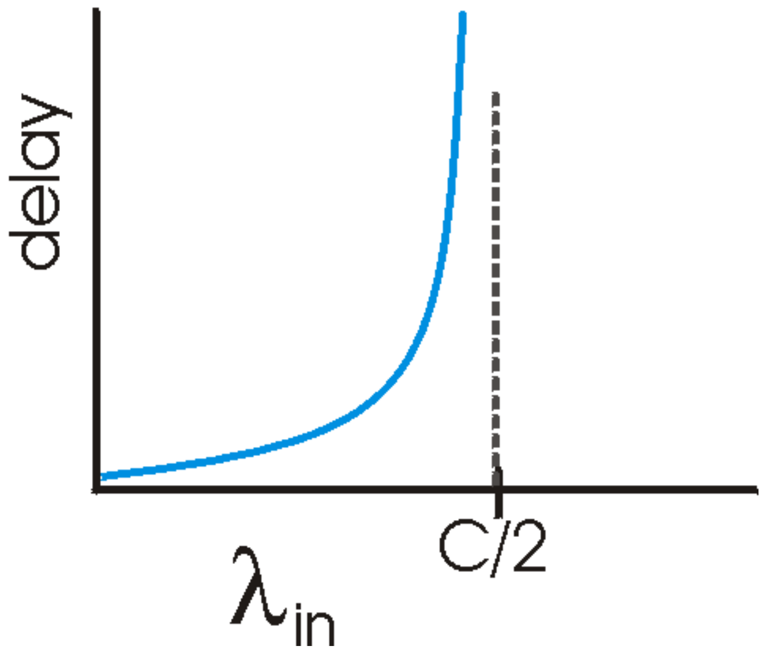
- Congestion: informally, "too many sources sending too much data too fast for the *network* to handle"
- different from flow control!
- manifestations:
  - ➡ lost packets (buffer overflow at routers)
  - ➡ long delays (queueing in router buffers)

Causes/costs of congestion: Simple Scenario

- two senders, two receivers
- one router, infinite buffers
- no retransmission







- large delays when congested
- maximum achievable throughput

Causes and Effects of Congestion

- With finite buffers at routers, packets may be dropped as λ_{in} increases
  - ➡ Retransmission needed
  - ➡ Offered load $\lambda'_{in} > \lambda_{in}$
  - – More work (retransmissions) for given λ_{out}
  - – Unneeded retransmissions: link carries multiple copies of segment
- With multi-hop connections, upstream routers receive two types of traffic:
  - ➡ Forwarded traffic from downstream routers
  - ➡ Traffic they may receive directly from hosts
  - ➡ As λ'_{in} increases, more dropped packages and more transmissions
    - ✦ λ_{out} will approach 0
  - – When packet dropped, any upstream transmission capacity used for that packet was wasted!

Approaches to Congestion Control

Two broad approaches towards congestion control:

1. end-end congestion control:
   - no explicit feedback from network
   - congestion inferred from end-system observed loss, delay
   - approach taken by TCP
2. 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