Module 2
Transport Layer Protocols
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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 vs. network layer

- **network layer**: enables logical communication between hosts
  - your laptop and CBC’s computer
- **transport layer**: enables logical communication between processes
  - relies on, enhances, network layer services
  - your laptop’s Web browser and CBC’s Web server
  - your laptop’s video player and CBC’s video server
  - relies on network-layer services
  - enhances network-layer services and provides different types of services to applications
    - e.g., reliability, confidentiality of messages
Transport Layer is End-to-End

client

<table>
<thead>
<tr>
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<th>link</th>
<th>physical</th>
</tr>
</thead>
</table>

server

<table>
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</table>

- bridge, hub, link-layer switch
- router
- bridge, hub, link-layer switch
## Internet Protocols

<table>
<thead>
<tr>
<th>Application</th>
<th>FTP</th>
<th>Telnet</th>
<th>NFS</th>
<th>SMTP</th>
<th>HTTP ...</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport</td>
<td>TCP</td>
<td>UDP</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Network</td>
<td>IP</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Data Link</td>
<td>X.25</td>
<td>Ethernet</td>
<td>Packet Radio</td>
<td>ATM</td>
<td>FDDI</td>
</tr>
<tr>
<td>Physical</td>
<td></td>
<td></td>
<td></td>
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<td></td>
</tr>
</tbody>
</table>
# Internet Apps: Their Protocols & Transport Protocols

<table>
<thead>
<tr>
<th>Applications</th>
<th>Data Loss</th>
<th>Throughput</th>
<th>Time Sensitive</th>
<th>Application Layer Protocol</th>
<th>Transport Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Email</td>
<td>No loss</td>
<td>Elastic</td>
<td>No</td>
<td>smtp</td>
<td>TCP</td>
</tr>
<tr>
<td>remote terminal access</td>
<td>No loss</td>
<td>Elastic</td>
<td>Yes</td>
<td>telnet</td>
<td>TCP</td>
</tr>
<tr>
<td>Web</td>
<td>No loss</td>
<td>Elastic</td>
<td>No</td>
<td>http</td>
<td>TCP</td>
</tr>
<tr>
<td>File transfer</td>
<td>No loss</td>
<td>Elastic</td>
<td>No</td>
<td>ftp</td>
<td>TCP</td>
</tr>
<tr>
<td>streaming multimedia</td>
<td>Loss tolerant</td>
<td>audio: 5kbps-1Mbps video: 10kbps-5Mbps</td>
<td>Yes, 100’s msec</td>
<td>Proprietary</td>
<td>TCP or UDP</td>
</tr>
<tr>
<td>Remote file server</td>
<td>No loss</td>
<td>Elastic</td>
<td>No</td>
<td>NFS</td>
<td>TCP or UDP (typically UDP)</td>
</tr>
<tr>
<td>Internet telephony</td>
<td>Loss tolerant</td>
<td>Elastic</td>
<td>Yes, few secs</td>
<td>SIP, RIP, Proprietary</td>
<td>TCP or UDP (typically UDP)</td>
</tr>
</tbody>
</table>
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

© Addison-Wesley Publishers 2000
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
Example: Java client (UDP)

Client process

Input: receives packet

Output: sends packet

Client UDP socket
Example: UDP Java client

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

class UDPClient {
    public static void main(String args[]) throws Exception {
        BufferedReader inFromUser =
            new BufferedReader(new InputStreamReader(System.in));
        DatagramSocket clientSocket = new DatagramSocket();
        InetAddress IPAddress = InetAddress.getByName("hostname");
        byte[] sendData = new byte[1024];
        byte[] receiveData = new byte[1024];

        String sentence = inFromUser.readLine();
        sendData = sentence.getBytes();
```
Example: UDP Java client (cont’d)

DatagramPacket sendPacket =
    new DatagramPacket(sendData, sendData.length, IPAddress, 9876);
clientSocket.send(sendPacket);

DatagramPacket receivePacket =
    new DatagramPacket(receiveData, receiveData.length);
clientSocket.receive(receivePacket);

String modifiedSentence =
    new String(receivePacket.getData());
System.out.println("FROM SERVER:" + modifiedSentence);
clientSocket.close();
import java.io.*;
import java.net.*;

class UDPServer {
  public static void main(String args[]) throws Exception {
    DatagramSocket serverSocket = new DatagramSocket(9876);

    byte[] receiveData = new byte[1024];
    byte[] sendData = new byte[1024];

    while (true) {
      DatagramPacket receivePacket = new DatagramPacket(receiveData, receiveData.length);
      serverSocket.receive(receivePacket);
    }
  }
}
Example: UDP Java server (cont’d)

String sentence = new String(receivePacket.getData());

InetAddress IPAddress = receivePacket.getAddress();

int port = receivePacket.getPort();

String capitalizedSentence = sentence.toUpperCase();

sendData = capitalizedSentence.getBytes();

DatagramPacket sendPacket = new DatagramPacket(sendData, sendData.length, IPAddress, port);

serverSocket.send(sendPacket);

end of while loop, loop back and wait for another datagram
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:
- Source port #
- Destination port #
- Length: in bytes of UDP segment, including header
- Checksum

Application data (message)
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

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<tbody>
<tr>
<td>P3</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

host 1

<table>
<thead>
<tr>
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<th>link</th>
<th>physical</th>
</tr>
</thead>
<tbody>
<tr>
<td>P1</td>
<td></td>
<td></td>
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</tr>
</tbody>
</table>

host 2

<table>
<thead>
<tr>
<th>application</th>
<th>transport</th>
<th>network</th>
<th>link</th>
<th>physical</th>
</tr>
</thead>
<tbody>
<tr>
<td>P2</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

host 3

= process
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:
- source port #
- dest port #
- other header fields
- application data (message)
Connectionless demultiplexing

• *recall*: create sockets with host-local port numbers:

  ```java
  DatagramSocket mySocket1 = new DatagramSocket(12534);
  DatagramSocket mySocket2 = new DatagramSocket(12535);
  ```

• *recall*: when creating datagram to send into UDP socket, must specify

  \[(\text{dest IP address, dest port number})\]

• 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
DatagramSocket serverSocket = new DatagramSocket(6428);

SP provides “return address”
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 (more in Chap 3)

application viewpoint

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

Server (running on hostid)

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

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

- read request from connectionSocket
- write reply to connectionSocket
- close connectionSocket

Client

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

- send request using clientSocket
- read reply from clientSocket
- 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

```java
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());
        BufferedReader inFromUser = new BufferedReader(new InputStreamReader(System.in));
        Socket clientSocket = new Socket("hostname", 6789);
        DataOutputStream outToServer = new DataOutputStream(clientSocket.getOutputStream());
```
Example: TCP Java client (cont’d)

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

sentence = inFromUser.readLine();
outToServer.writeBytes(sentence + '
');
modifiedSentence = inFromServer.readLine();
System.out.println("FROM SERVER: " + modifiedSentence);
clientSocket.close();
```
Example: TCP Java server

```java
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:
  ```java
dataOutputStream = new DataOutputStream(connectionSocket.getOutputStream());
```

- Read in line from socket:
  ```java
  clientSentence = inFromClient.readLine();
  ```

- Capitalize the sentence:
  ```java
capitalizedSentence = clientSentence.toUpperCase() + '\n';
  ```

- Write out line to socket:
  ```java
  outToClient.writeBytes(capitalizedSentence);
  ```

End of while loop, loop back and wait for another client connection.
TCP segment structure

- **URG**: urgent data (generally not used)
- **ACK**: ACK # valid
- **PSH**: push data now (generally not used)
- **RST, SYN, FIN**: connection estab (setup, teardown commands)
- **Internet checksum**: (as in UDP)

---

<table>
<thead>
<tr>
<th>Source Port #</th>
<th>Dest Port #</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Sequence Number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Acknowledgement Number</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Receive Window</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Checksum</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Urg Data Ptr</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Options (variable length)</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Application Data</strong> (variable length)</td>
<td></td>
</tr>
</tbody>
</table>
Multiplexing/demultiplexing

Demultiplexing at rcv host:
delivering received segments
to correct socket

= socket
= process

Multiplexing at send host:
gathering data from multiple
sockets, enveloping data with
header (later used for
demultiplexing)

Application
transport
network
link
physical

host 1

Application
transport
network
link
physical

host 2

Application
transport
network
link
physical

host 3
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
Connection-oriented demux (cont)
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 Reliable data transfer

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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
Reliable data transfer: getting started

**send side**

- **rdt_send()**: called from above, (e.g., by app.). Passed data to deliver to receiver upper layer
- **udt_send()**: called by rdt, to transfer packet over unreliable channel to receiver

**receive side**

- **deliver_data()**: called by rdt to deliver data to upper
- **rdt_rcv()**: called when packet arrives on rcv-side of channel

**Diagram Description**

- **rdt_send()**
  - Called from above (e.g., by app.)
  - Passed data to deliver to receiver upper layer

- **udt_send()**
  - Called by rdt to transfer packet over unreliable channel to receiver

- **deliver_data()**
  - Called by rdt to deliver data to upper

- **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

```
Wait for call from above
rdt_send(data)
    packet = make_pkt(data)
    udt_send(packet)

Wait for call from below
rdt_rcv(segment)
    extract (segment, data)
    deliver_data(data)
```

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

• 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 segment if ACK/NAK garbled
- Sender adds **sequence number** to each segment
  - For data: byte stream “number” of first byte in segment’s data
  - For ACKs: segment number of next byte expected from other side
    - Cumulative ACK
- Receiver discards (does not deliver up to the application) duplicate segments
- **Stop and wait**
  - Sender sends one segment, then waits for the receiver to respond
  - We will come back to this later
NAK-free Protocol

• It is possible to eliminate NAKs as negative acknowledgement
• instead of NAK, receiver sends ACK for last segment received OK
  ➡ receiver must *explicitly* include sequence number of segment being ACKed
• duplicate ACK at sender results in same action as NAK: *retransmit current packet*
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
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: retransmission scenarios

Host A
Seq=100, 20 bytes data

Host B
Seq=92, 8 bytes data

Host A
Seq=92, 8 bytes data

Host B
Seq=92, 8 bytes data

Host A
ACK=100

Host B
ACK=100

Host A
ACK=100

Host B
ACK=120

Host A
Seq=100, 20 bytes data

Host B
Seq=92, 8 bytes data

Host A
Seq=92 timeout

Host B
Seq=92 timeout

lost ACK scenario

premature timeout

X

loss
TCP retransmission scenarios (more)

Host A
Seq=92, 8 bytes data
ACK=100

Host B
Seq=100, 20 bytes data
ACK=100

X
loss

Cumulative ACK scenario
## TCP ACK Generation

<table>
<thead>
<tr>
<th>Event at Receiver</th>
<th>TCP Receiver action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed</td>
<td>Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK</td>
</tr>
<tr>
<td>Arrival of in-order segment with expected seq #. One other segment has ACK pending</td>
<td>Immediately send single cumulative ACK, ACKing both in-order segments</td>
</tr>
<tr>
<td>Arrival of out-of-order segment higher-than-expect seq. #. Gap detected</td>
<td>Immediately send <em>duplicate ACK</em>, indicating seq. # of next expected byte</td>
</tr>
<tr>
<td>Arrival of segment that partially or completely fills gap</td>
<td>Immediate send ACK, provided that segment starts at lower end of gap</td>
</tr>
</tbody>
</table>
Fast Retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.
- if sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires
Resending segment after triple duplicate ACK

Host A

timeout

Host B

resend 2\textsuperscript{nd} segment
Fast retransmit algorithm:

**event:** ACK received, with ACK field value of y

if (y > SendBase) {
    SendBase = y
    if (there are currently not-yet-acknowledged segments)
        start timer
} else {
    increment count of dup ACKs received for y
    if (count of dup ACKs received for y = 3) {
        resend segment with sequence number y
    }
}

a duplicate ACK for already ACKed segment  

fast retransmit
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 Round Trip Time and Timeout

\[
\text{EstimatedRTT} = (1 - \alpha) \times \text{EstimatedRTT} + \alpha \times \text{SampleRTT}
\]

- Exponential weighted moving average
- Influence of past sample decreases exponentially fast
- Typical value: \(\alpha = 0.125\)

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr
TCP Round Trip Time and Timeout

Setting the timeout

- EstimatedRTT plus “safety margin”
  - large variation in EstimatedRTT -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

\[
\text{DevRTT} = (1-\beta) \times \text{DevRTT} + \\
\beta \times |\text{SampleRTT} - \text{EstimatedRTT}|
\]

(typically, \(\beta = 0.25\))

Then set timeout interval:

\[
\text{TimeoutInterval} = \text{EstimatedRTT} + 4 \times \text{DevRTT}
\]
TCP Flow Control

• receive side of TCP connection has a receive 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

• app process may be slow at reading from buffer
**TCP Flow control: how it works**

- **Spare room in buffer (ignoring out-of-order segments)**
  \[ \text{RcvWindow} = \text{RcvBuffer} - [\text{LastByteRcvd} - \text{LastByteRead}] \]

- 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
  
  ```java
  Socket clientSocket = new Socket("hostname","port number");
  ```
- server: contacted by client
  
  ```java
  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

Client: SYN=1, seq=client_isn
Server: SYN=1, seq=server_isn, Ack=client_isn+1
ACK: SYN=0, Ack=server_isn+1
Connection granted
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.
TCP Connection States

TCP client lifecycle

TCP server lifecycle
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

\[ \lambda_{\text{out}} \] vs \[ \lambda_{\text{in}} \]

\[ \text{delay} \] vs \[ \lambda_{\text{in}} \]
Causes and Effects of Congestion

• With finite buffers at routers, packets may be dropped as $\lambda_{in}$ increases
  ➡ Retransmission needed
  ➡ Offered load $\lambda_{in}' > \lambda_{in}$
  ‒ More work (retransmissions) for given $\lambda_{out}$
  ‒ 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 $\lambda_{in}'$ increases, more dropped packages and more transmissions
    ✦ $\lambda_{out}$ 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
TCP congestion control

- **Approach:** increase transmission rate (window size), probing for usable bandwidth, until loss occurs
  - **Additive increase:** increase \( cwnd \) by 1 MSS (maximum segment size) every RTT until loss detected
  - **Multiplicative decrease:** cut \( cwnd \) in half after loss
- **Algorithm has three components:**
  - **Slow start**
    - Initially set \( cwnd = 1 \) MSS
    - double \( cwnd \) every RTT
    - done by incrementing \( cwnd \) for every ACK received
  - **Congestion avoidance:** After timeout or 2 duplicate ACKS
    - \( cwnd \) is cut in half
    - window then grows linearly
  - **Fast recovery:** After a timeout
    - \( cwnd \) set to 1 MSS;
    - window then grows exponentially
    - to a threshold, then grows linearly