Chapter 3: Transport Layer

Our goals:

- understand principles behind transport layer services:
 - multiplexing/demultipl exing
 - o 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

<u>Chapter 3 outline</u>

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer

- 3.5 Connection-oriented transport: TCP
 - segment structure
 - reliable data transfer
 - flow control
 - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

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

Household analogy:

- 12 kids sending letters to 12 kids
- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill
- network-layer protocol
 - = postal service

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
 - congestion control
 - flow control
 - connection setup
- unreliable, unordered delivery: UDP
 - no-frills extension of "best-effort" IP
- services not available:
 - o delay guarantees
 - bandwidth guarantees



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Multiplexing/demultiplexing



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 mySocket1 = new
 DatagramSocket(12534);

DatagramSocket mySocket2 = new
DatagramSocket(12535);

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"

<u>Connection-oriented demux</u>

- TCP socket identified by 4-tuple:
 - source IP address
 - source port number
 - dest IP address
 - dest port number
- receiving 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

<u>Connection-oriented demux</u> (cont)



<u>Connection-oriented demux:</u> <u>Threaded Web Server</u>



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UDP: User Datagram Protocol [RFC 768]

- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
 - o 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

UDP: more





<u>Goal:</u> 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

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Internet Checksum Example

Note

 When adding numbers, a carryout from the most significant bit needs to be added to the result

Example: add two 16-bit integers



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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) init's sender, receiver state before data exchange

flow controlled:

 sender will not overwhelm receiver

TCP segment structure



Transport Layer 3-21

TCP seq. #'s and ACKs

<u>Seq. #'s:</u>

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

<u>ACKs:</u>

- seq # of next byte expected from other side
- o cumulative ACK
- Q: how receiver handles out-of-order segments
 - A: TCP spec doesn't say, - up to implementer

