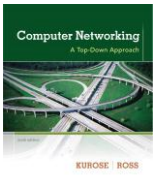
Chapter 3
Transport Layer



Computer Networking: A Top Down Approach
6th edition
Jim Kurose, Keith Ross
Addison-Wesley
March 2012

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Thanks and enjoy! JFK/KWR

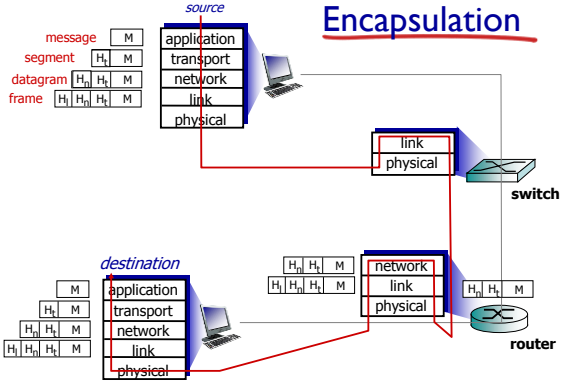
The course notes are adapted for Bucknell's CSCI 363
Xiannong Meng
Spring 2016

Chapter 3: Transport Layer

our goals:
- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

REVIEW LAYERED ARCHITECTURE

Encapsulation



Wireshark

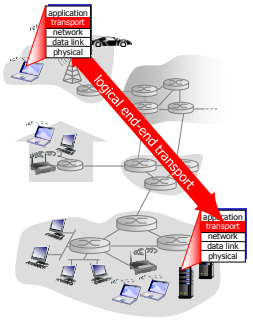
- *Wireshark* is a piece of software that can capture and record network traffic "on the wire."
- It also allows user to examine the traffic in a nice GUI.
- We'll use *Wireshark* to capture network traffic, and write our own analyzer.
- Our analyzer will peer into the layers and examine in detail the information in each captured frame.
- Demonstration of *Wireshark*.

Chapter 3 outline

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
  - recv 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

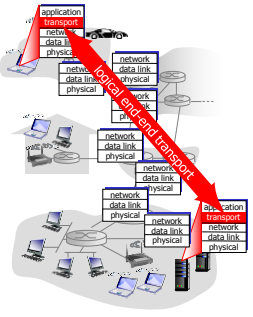
household analogy:

12 kids in Ann's house sending letters to 12 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann' multiplexing and Bill' demultiplexing to in-house siblings
- 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:
  - delay guarantees
  - bandwidth guarantees



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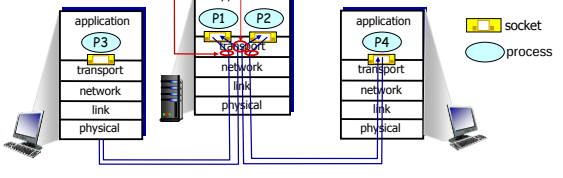
Multiplexing/demultiplexing

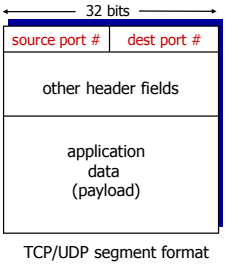
multiplexing at sender: handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver: use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one 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

recall: created socket has host-local port #:

```
udp_sock = socket(AF_INET,
  SOCK_DGRAM)
udp_sock.sendto(msg,(host,port))
```

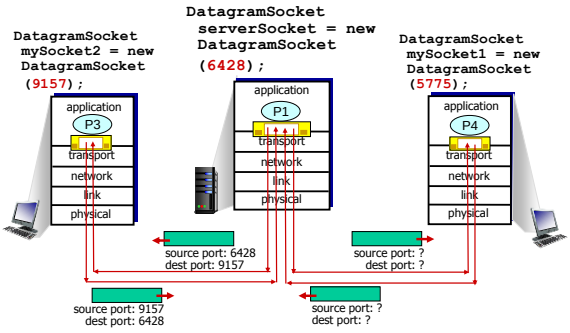
recall: when creating datagram to send into UDP socket, must specify

- destination IP address
- destination port #

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

IP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

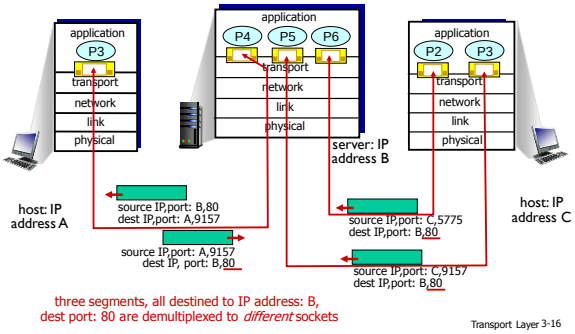
Connectionless demux: example



Connection-oriented demux

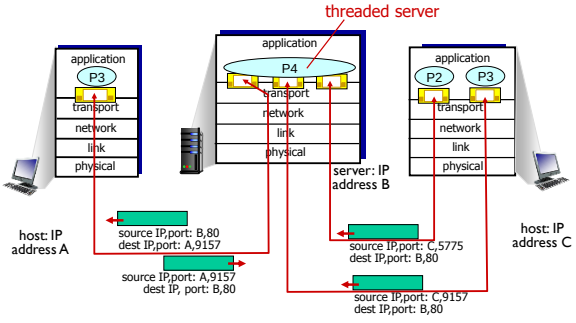
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



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

Connection-oriented demux: example



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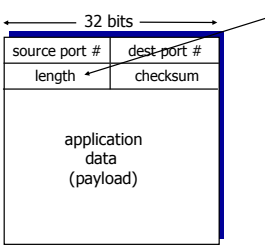
UDP: User Datagram Protocol [RFC 768]

- "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

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

UDP: segment header



UDP segment 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 Header File for C/C++

```
struct udphdr
{
  u_int16_t source;  /* src port number */
  u_int16_t dest;    /* dest port number */
  u_int16_t len;     /* total length in bytes */
  u_int16_t check;   /* check sum */
};
```

In the file : /usr/include/netinet/udp.h

UDP checksum

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

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one'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 ….

Internet checksum: example

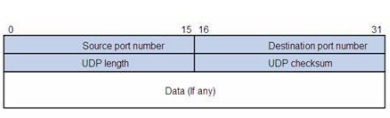
example: add two 16-bit integers

```
                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
```

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

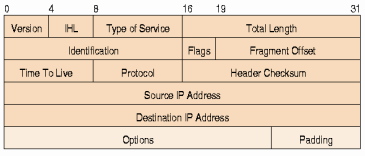
UDP packet format

- UDP header:
  - http://en.wikipedia.org/wiki/User_Datagram_Protocol
  - http://www.ietf.org/rfc/rfc768.txt



IP packet format

- IP header:
  - http://en.wikipedia.org/wiki/IPv4
  - http://www.ietf.org/rfc/rfc791.txt



UDP Checksum Computation

- According to RFC 768, http://www.faqs.org/rfcs/rfc768.html
- UDP checksum is computed as follows
  - Checksum is the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.