

CS3640

Transport Layer (1): Multiplexing and UDP

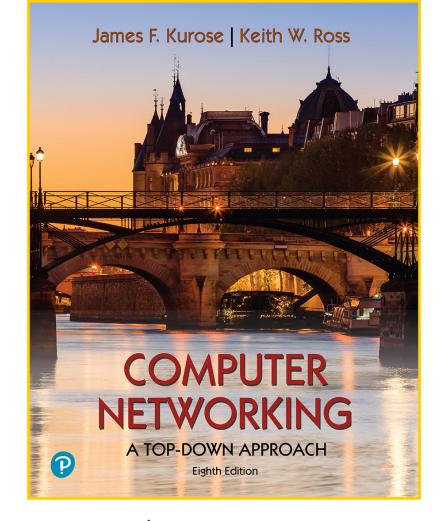
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Lecture goals

introduction to key transport layer services, and an overview of UDP

- Transport-layer services
- Multiplex and demultiplex
- UDP



Chapter 3.1 - 3.3

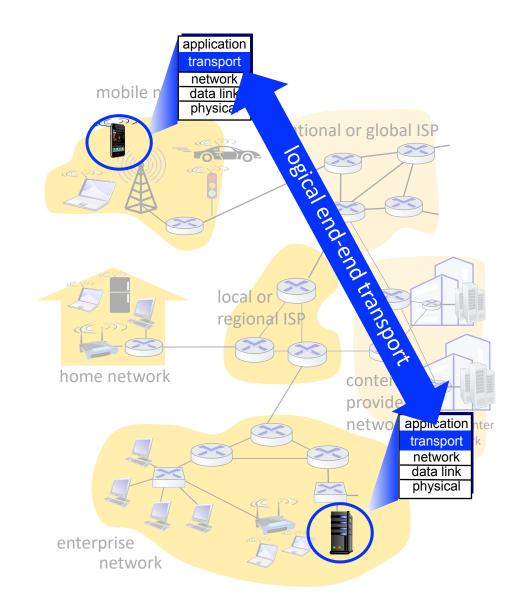


Transport-layer services

Key goal: to provide logical communication between application processes running on different hosts

Only end-systems need transport layer

- sender: breaks application messages into segments, and passes to the network layer
- receiver: reassembles received segments into messages, passes to the application layer



Transport vs. network layer services

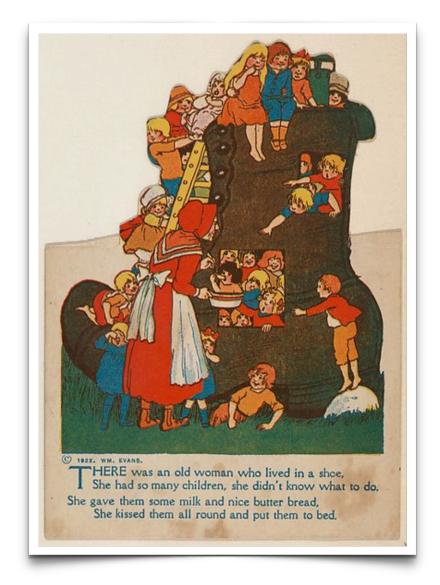
Network layer: logical communication b/w network hosts

Transport layer: logical communication b/w processes

a household analogy

Kids in Alice's house exchanging letters with kids in Bob's house

- houses == hosts
- kids == processes
- letters in envelopes == application messages
- postal service == network layer
- Alice and Bob == transport layer (mux-demux)



Transport-layer services of the Internet

	ТСР	UDP
multiplexing and demultiplexing	✓	✓
reliable data transfer	✓	
flow control	✓	
congestion control	✓	
timing and delay guarantees	0	
throughput guarantees	0	

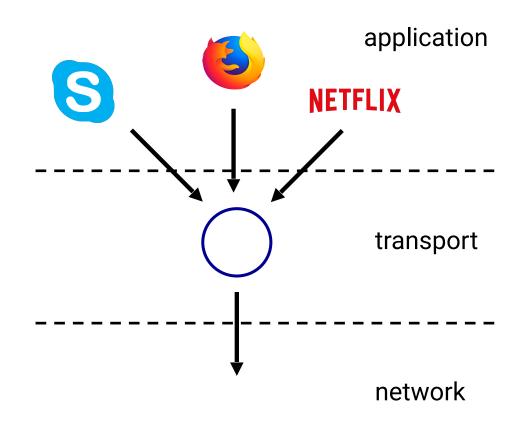
Q: why this particular choice of services by each protocol?

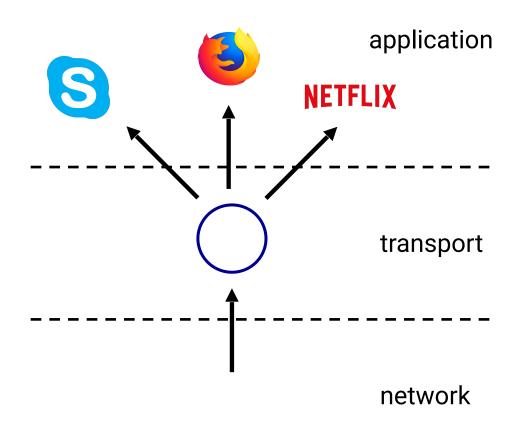
Q: are these even possible on the Internet?

Multiplexing / Demultiplexing

multiplexing

demultiplexing



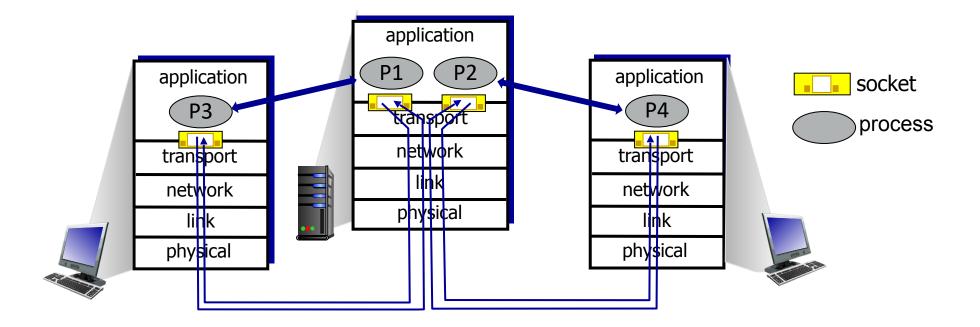


multiplexing at sender

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



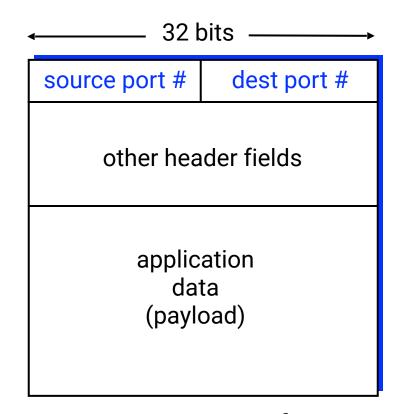
How demultiplexing works

1. host receives IP datagrams

- each datagram has source IP address, destination IP address, and carries one transport-layer segment
- the transport segment contains a source port number and destination port number

2. host directs the TCP/UDP segment to the appropriate socket

 based on the combination of IP addresses and port numbers



TCP/UDP segment format

Connectionless demultiplexing

1. when creating UDP socket, the application must specify the local port number

DatagramSocket mySocket = new DatagramSocket(12534);

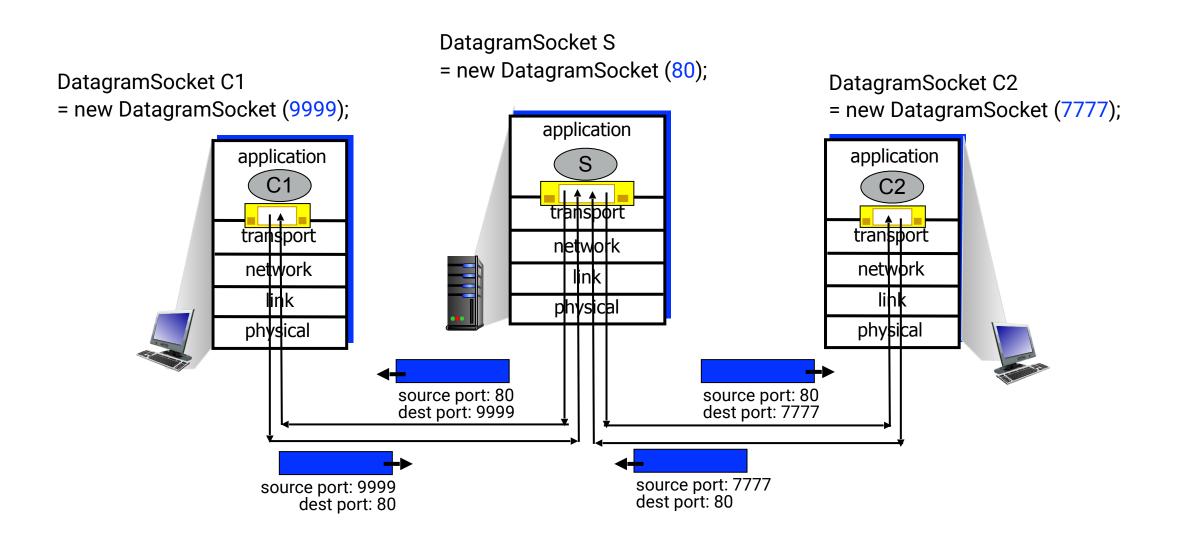
2. when sending a segment into UDP socket, the host must specify

destination IP address, destination port number

3. when a host receives a UDP segment, it will

- check destination port number, and direct it to the socket bound to that port
- demux is not dependent on either source IP address or source port number

Connectionless demultiplexing



Connection-oriented demultiplexing

1. connection-oriented sockets (TCP) are identified by the 4-tuple

- src IP address, src port number, dest IP address, dest port number
- this is done via connect() and accept() socket calls

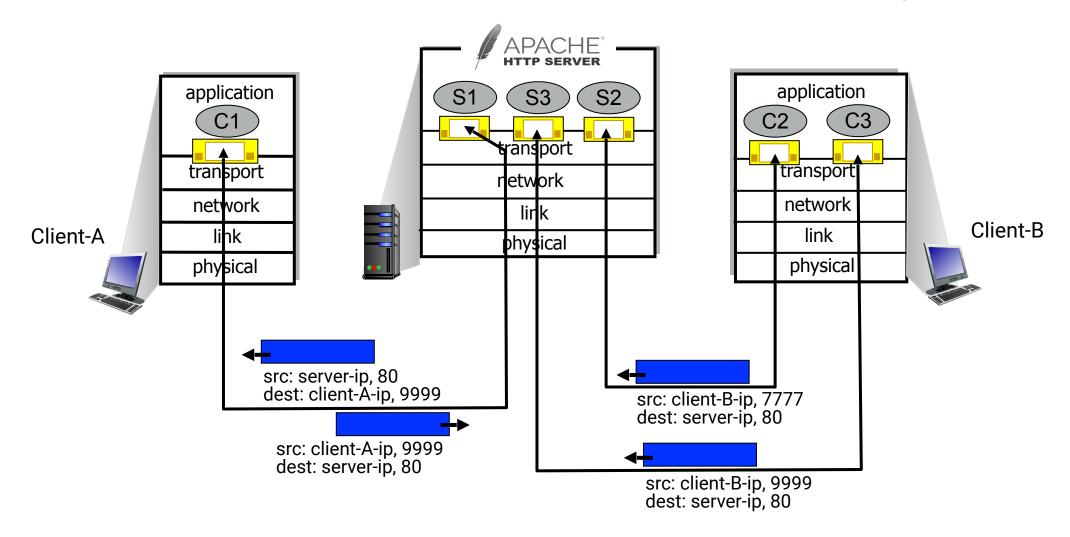
2. when host receives a TCP segment, it will

use all four values to direct the segment to the appropriate socket

3. server may support multiple simultaneous TCP connections, where

- each socket is uniquely identified by its own four tuple
- each socket is associated with a different client application

Connection-oriented demultiplexing



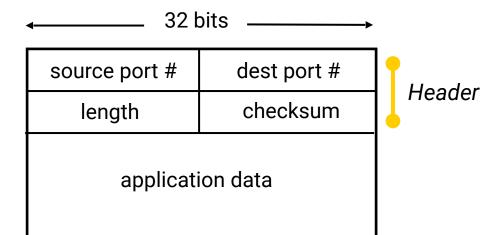
Three segments, all destined to <server-IP, 80> are demultiplexed to different sockets at the server



User Datagram Protocol

- a bare-bones datagram transport over IP
- defined in an RFC that is ~500 words
- mux-demux is the only service offered

UDP segment format



User Datagram Protocol

Introduction

This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

This protocol provides a procedure for application programs to send messages to other programs with a minimum of protocol mechanism. The protocol is transaction oriented, and delivery and duplicate protection are not guaranteed. Applications requiring ordered reliable delivery of streams of data should use the Transmission Control Protocol (TCP) [2].

Why UDP even exists?

connection-less	reduces latency and RTT delays	
stateless client and server	simpler implementation	
no reliability or in-order delivery	less overhead (due to small header size)	
no flow control	can send data as fast as desired	
no congestion control	can function in the face of congestion	

some applications do not need these services (e.g., video streaming needs a simple low latency transport) the associated **overhead** (e.g., SNMP needs to operate even during network congestions)

some applications can **efficiently implement** missing services by
themselves (e.g., HTTP/3 shifted to UDP
after absorbing many of the TCP features)

UDP checksum

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

sender

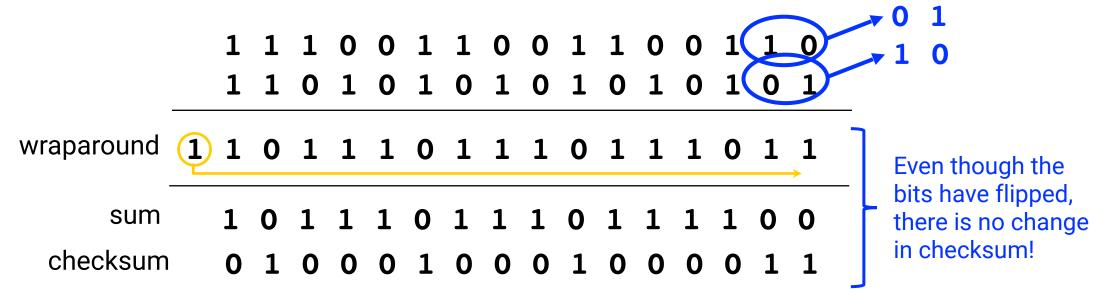
- treat contents of UDP segment as a sequence of 16-bit words
- add all the words (wrapping around any overflows) to produce a 16-bit result
- compute the checksum: by taking one's complement of the result
- put this value into UDP checksum field

receiver

- add all the 16-bit words including the checksum
- if no errors are introduced, then the sum will come out to be 1111 1111 1111 1111
- any zero in the sum indicates bit error(s)

UDP checksum shortcomings

example: add two 16-bit integers

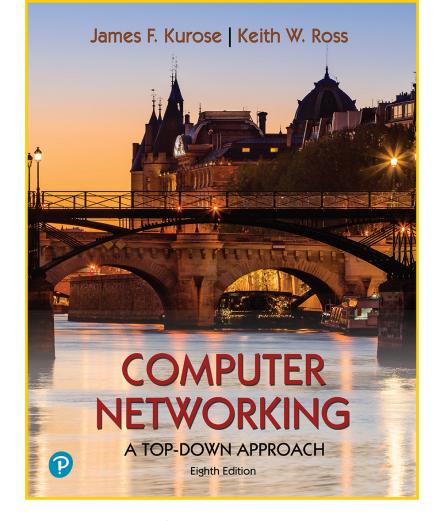


- ▶ This is a weak protection i.e., it can only detect single-bit errors
- Even when detected, UDP has no ability to correct the error
- ▶ UDP checksum is optional for IPv4 (if unused, set all the checksum bits to zero)

Next lecture

how to achieve reliable data transport over an unreliable network such as the Internet?

- Challenges and techniques
- Go-Back-N
- Selective Repeat



Chapter 3.4



Spot Quiz (ICON)