

CSCE 463/612

Networks and Distributed Processing

Spring 2018

Transport Layer

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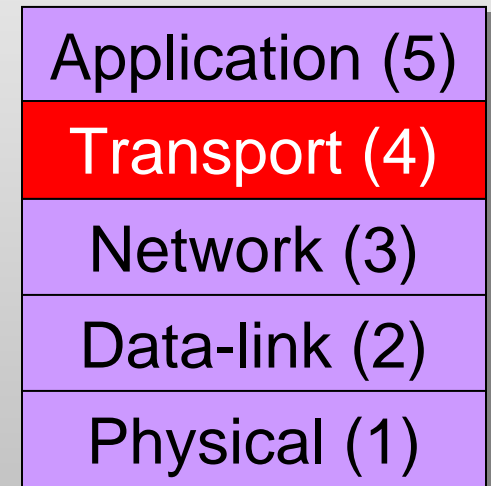
February 22, 2018

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Chapter 3: Transport Layer

Our goals:

- Understand principles behind transport layer services:
 - Multiplexing/demultiplexing
 - Reliable data transfer
 - Flow control
 - Congestion control
- Learn about transport layer protocols in the Internet:
 - UDP: connectionless transport
 - TCP: connection-oriented transport



Chapter 3: Roadmap

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

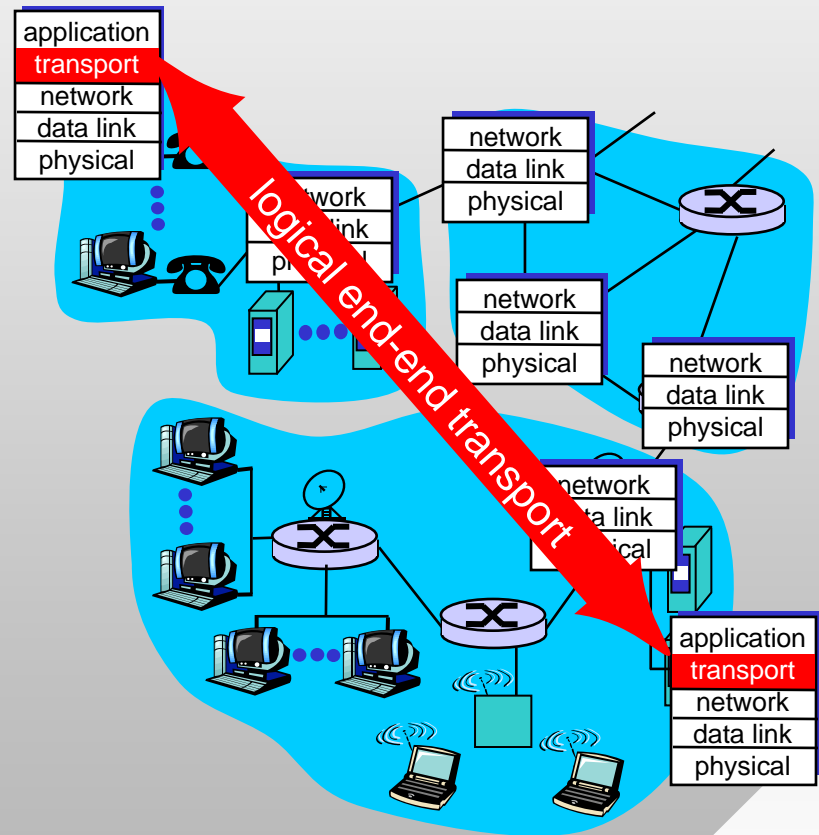
- Segment structure
- Reliable data transfer
- Flow control
- Connection management

3.6 Principles of congestion control

3.7 TCP congestion control

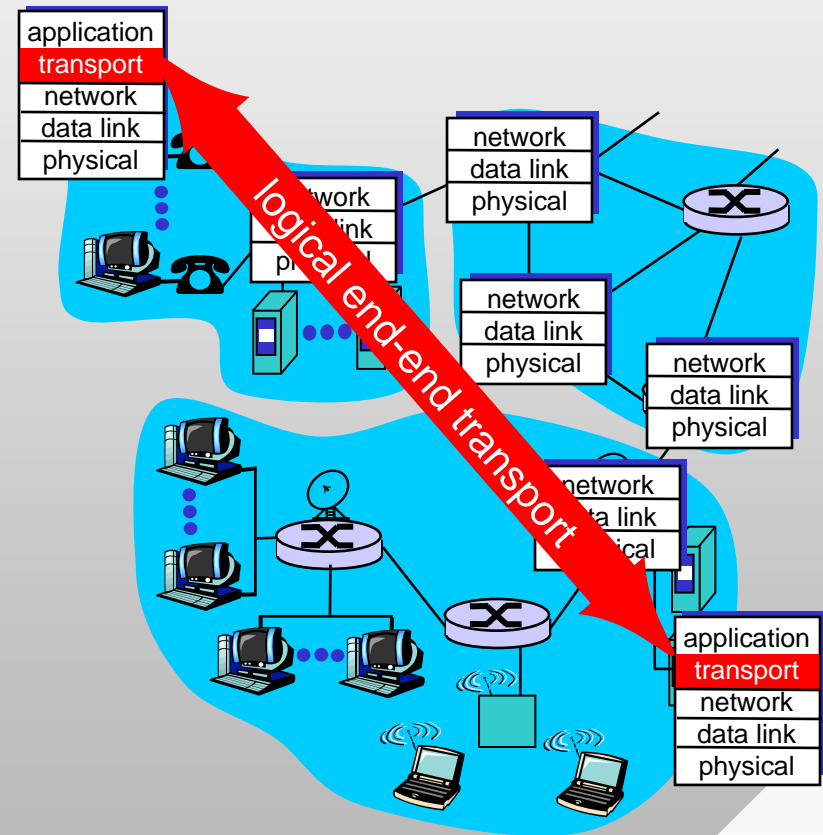
Transport Services and Protocols

- *Transport layer*: logical communication between **processes** on different hosts
 - Relies on and enhances network-layer services
- *Network layer*: logical communication between hosts



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 or loss guarantees
 - Bandwidth guarantees



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Multiplexing/Demultiplexing

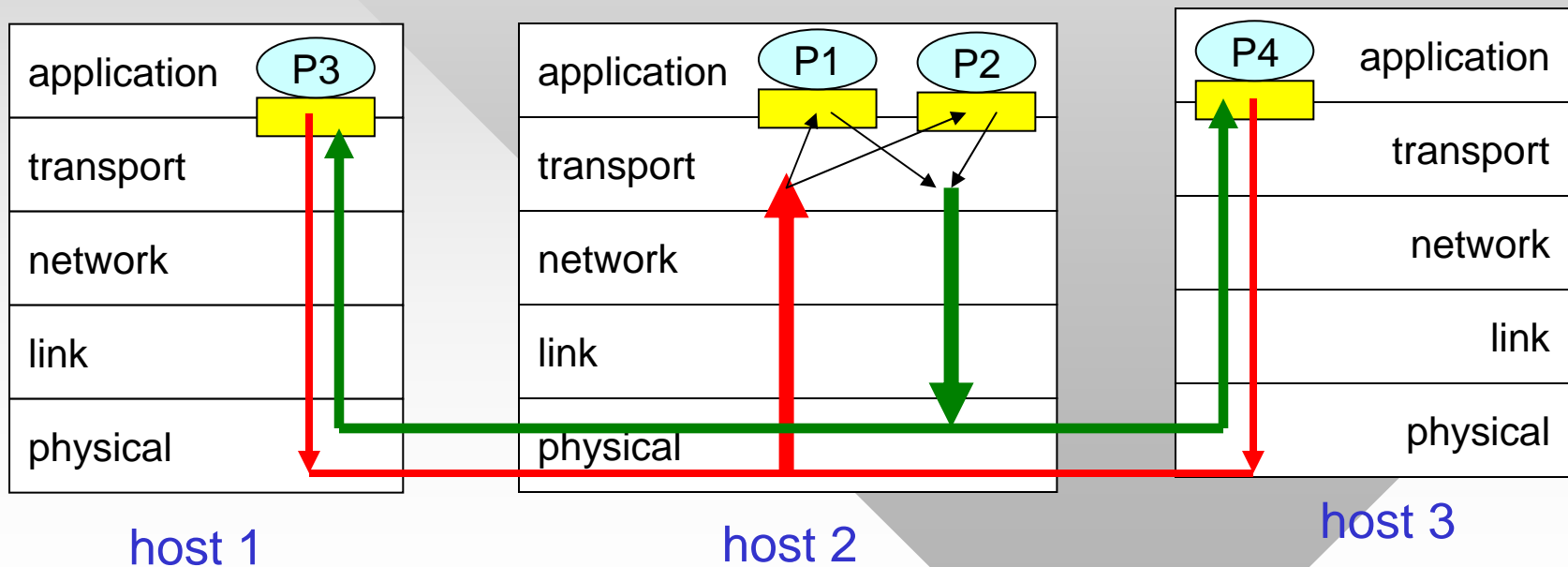
Demultiplexing at receiver host:

Delivering received segments to correct socket

Multiplexing at sender host:

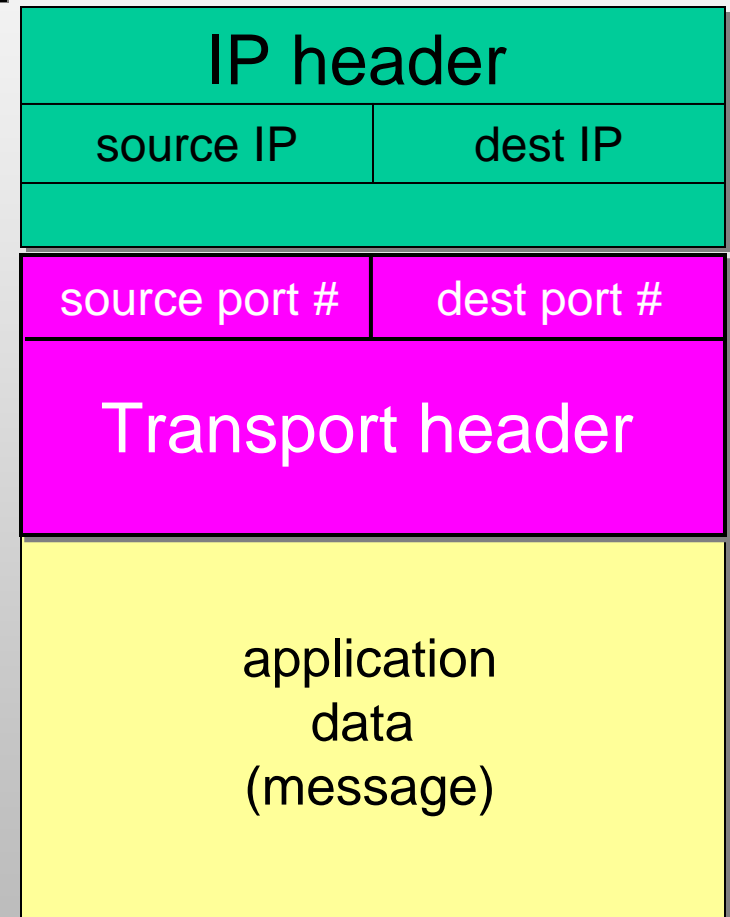
Gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

■ = socket ○ = process



How Demultiplexing Works

- **Host receives IP datagrams**
 - Each datagram has **source IP** address and **destination IP** address
- Each datagram carries one transport-layer header
 - Each transport header starts with source and destination port numbers
- Kernel uses port numbers to direct packets to appropriate socket or reject the message
 - Each port # is a 16-bit unsigned integer (1-65535)



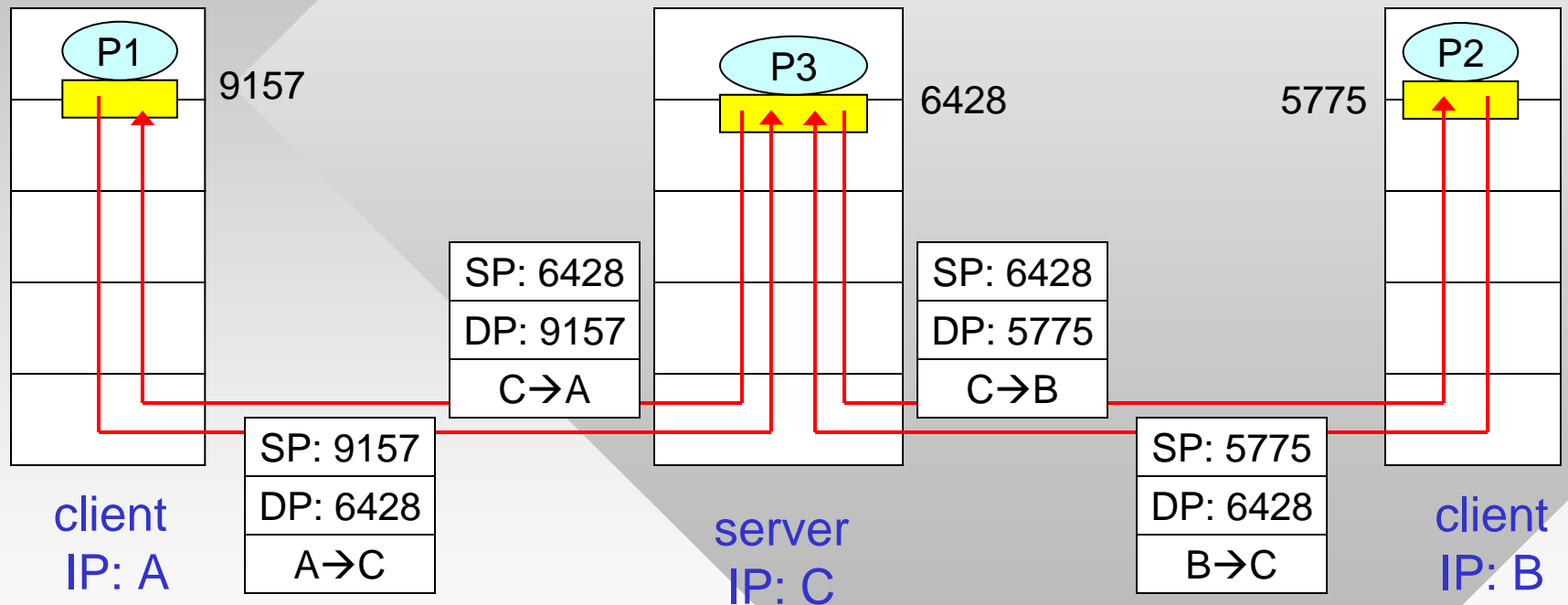
TCP/UDP segment format

Connectionless Demultiplexing

- Create a SOCK_DGRAM socket
- Bind the socket
 - Server: specify a well-known port (e.g., 53 for DNS)
 - Client: bind to port 0 (OS assigns next available #)
- Use sendto(), recvfrom()
- Target UDP socket is identified by a 2-tuple:
(dest IP address, dest port number)
- When host receives UDP segment:
 - OS checks destination port/IP in segment
 - Directs segment to the socket with a matching combination if socket is open; rejects otherwise
- IP datagrams with different source IP addresses and/or source port numbers may be directed to the same socket!

Connectionless Demultiplexing (Cont)

SP = source port, DP = destination port



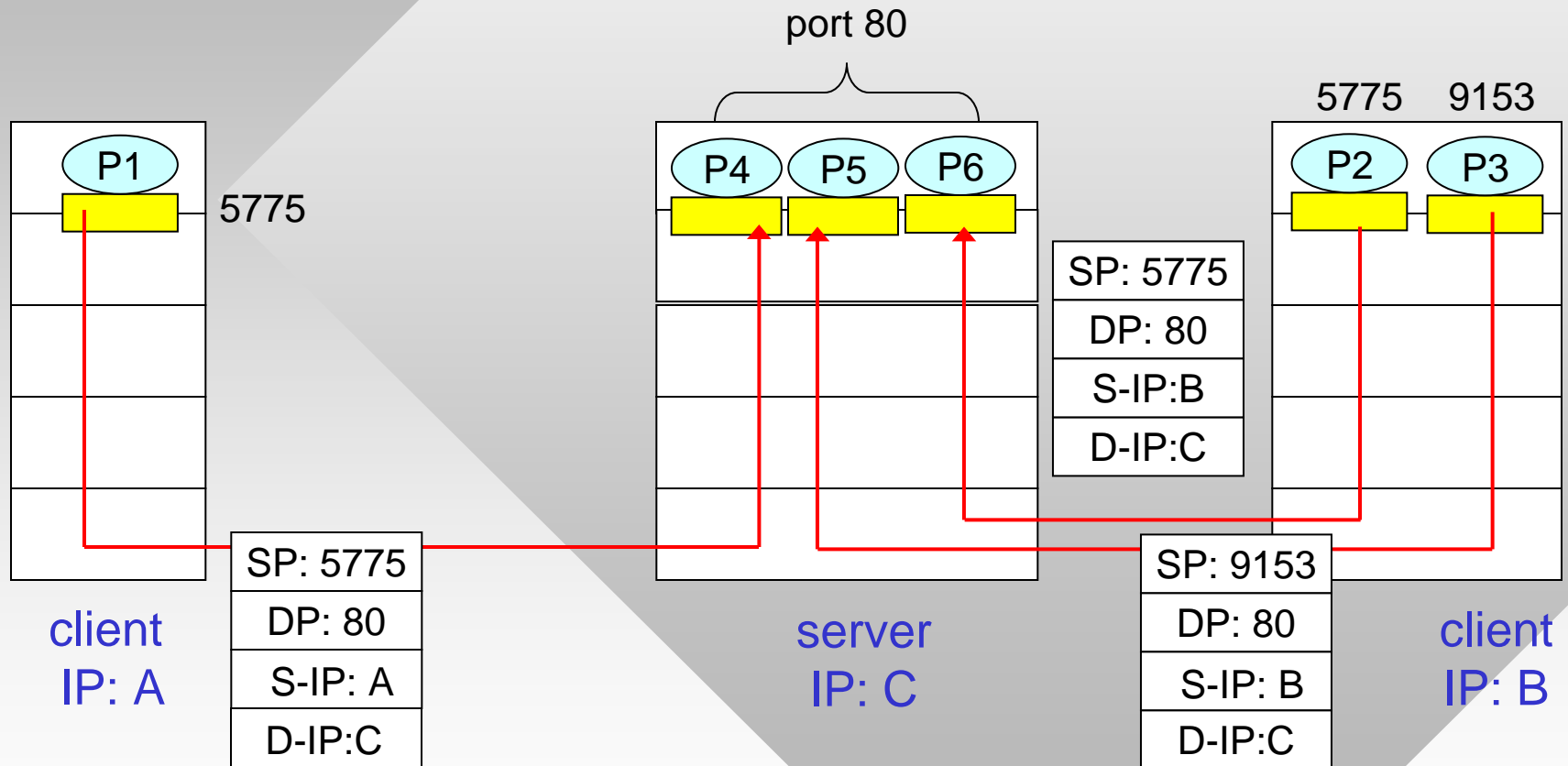
SP provides "return address"

Connection-Oriented Demultiplexing

- TCP socket identified by a 4-tuple:
 - Source IP address
 - Source port number
 - Destination IP address
 - Destination port number
- Receiver host uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets **on same port**:
 - Each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
 - All are on port 80
 - Non-persistent HTTP may have different socket for each request

Connection-Oriented Demultiplexing (Cont)

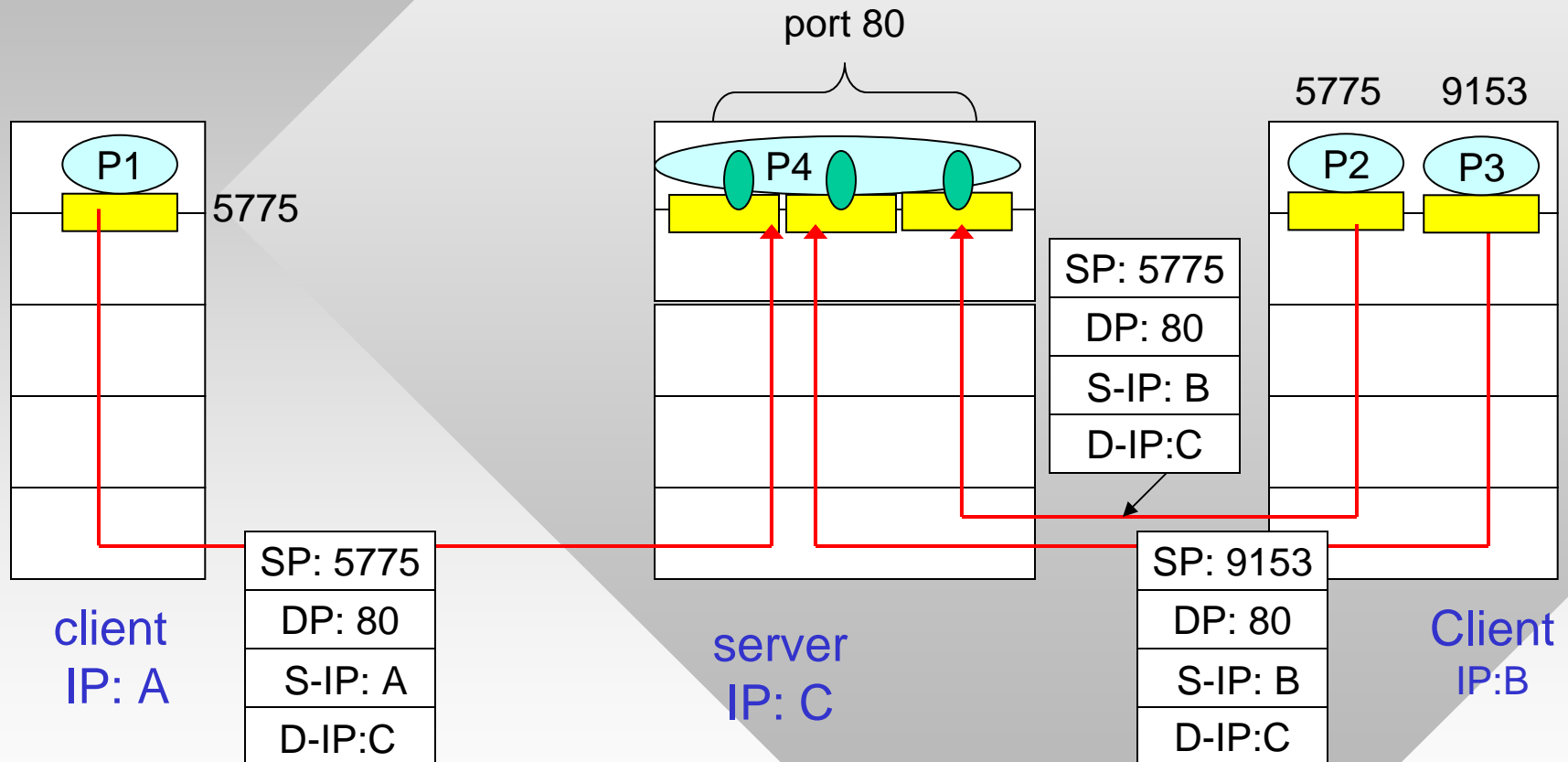
Web server spawns a new process per connection



SP = source port, DP = destination port;
S-IP = source IP, D-IP = destination IP

Connection-Oriented Demultiplexing (Cont)

Web server spawns a new thread per connection



SP = source port, DP = destination port;
S-IP = source IP, D-IP = destination IP

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UDP: User Datagram Protocol [RFC 768]

- Standardized in 1980
 - Hasn't changed much
- **Best-effort** service
- UDP segments may be:
 - Lost or corrupted
 - Delivered out of order to the application
- **Connectionless:**
 - No handshaking between UDP sender and receiver
 - Each UDP segment handled independently of others

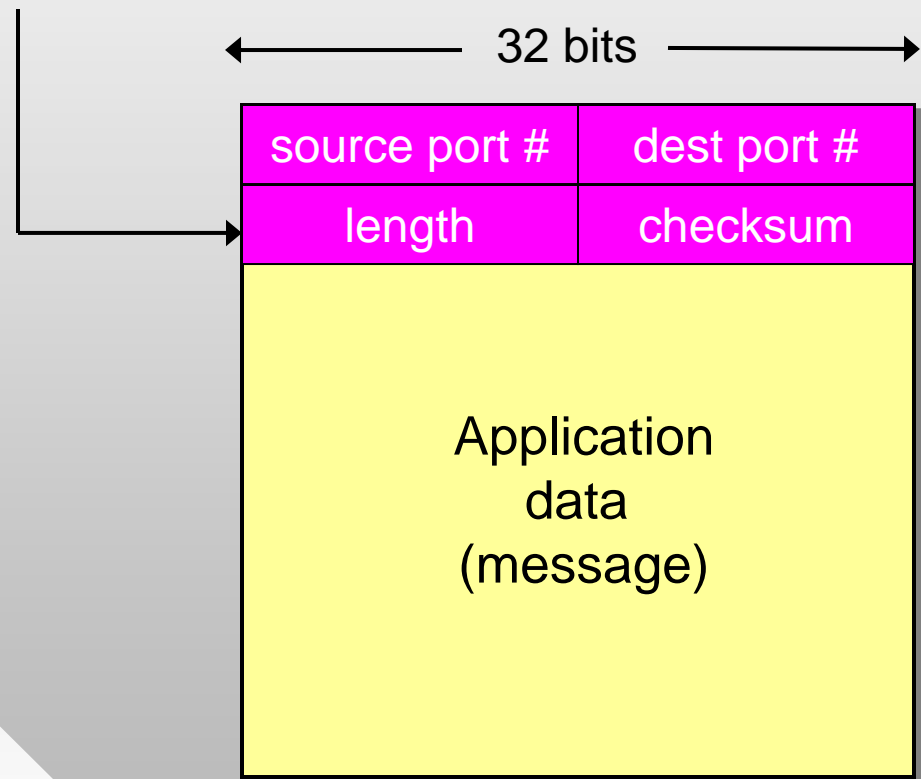
Why is there a UDP?

- Overhead: no connection establishment or retransmission
- Simplicity: no connection state at sender/receiver
- Small segment header
- No congestion control
 - For short transfers, this is completely unnecessary
 - In other cases, desirable to control rate directly from application

UDP: More

- Often used for streaming multimedia apps
 - Loss tolerant
 - Rate sensitive
- Other UDP uses
 - DNS
 - SNMP
 - NFSv2 (1989)
- Reliable transfer over UDP: add reliability at application layer
 - Application-specific error recovery

Length (in bytes) of
UDP segment,
including header



UDP segment format

UDP Checksum

Goal: detect “errors” (e.g., flipped bits) in transmitted segment (packet)

Sender (simplified):

- Set checksum = 0 in hdr
- Treat packet contents as a sequence of 16-bit integers (padded with 0s to 2-byte boundary)
- **Checksum:** add all integers, then XOR with 0xffff
- Sender puts checksum value into UDP checksum field

Receiver:

- Sum all 16-bit words in entire received segment with the checksum field in it
- Check if result = 0xffff
 - NO - error detected
 - YES - no error detected
- Idea: $(x \text{ XOR } 0xffff) + x = 0xffff$
- *Are undetected errors possible nonetheless?*

UDP Checksum Example

- Note on 1's complement addition:
 - When adding numbers, a carryout from the most significant bit needs to be added to the result
- 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	
<hr/>																	
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
<hr/>																	
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	

UDP Checksum (Cont)

- How many corrupted bits does UDP detect?
- Example of undetected single-bit corruption?
 - Not possible
- Example of undetected 2-bit corruption?
 - Two words (0, 5) result in checksum 5 XOR 0xffff
 - Suppose 0 is corrupted to become 1 and 5 is corrupted to become 4, then the checksum is the same
- Example of undetected 3-bit corruption?
 - Two words (1, 1) \rightarrow (0, 2)
- What if the transmitted words are 0 and 12?
 - Can two-bit corruption produce the same checksum?
 - If yes, how many ways can (0,12) be affected by 2-bit corruption so as to avoid detection?

Wrap-up

- Is there a pair of integers (x,y) that allow the UDP checksum to detect **any** 2-bit corruption?
- Data-link and physical layers are often assumed to have their own checksums and error correction
 - Why is transport-level checksum important then?
- Reasons:
 - 1) Lower layers do not always implement error checking
 - Even then, implementation bugs may affect the result
 - 2) Corruption may occur in router RAM or faulty hardware, outside the control of data-link protocols