CSCE 463/612
Networks and Distributed Processing
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Transport Layer
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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

| application | P3 | | | | P1 | | | | | | P2 | | | | | | P4 | | | | | | application |
| transport | | | | | | transport | | | | | | transport |
| network | | | | | | network | | | | | | network |
| link | | | | | | link | | | | | | link |
| physical | | | | | | physical | | | | | | physical |

del1  del2  del3

Multiplexing at host 1:
Delivering data to correct application

Demultiplexing at host 2:
Gathering data from multiple sockets

Multiplexing at host 3:
Sending data to correct application
**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)
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
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

Port 80

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

32 bits

<table>
<thead>
<tr>
<th>source port #</th>
<th>dest port #</th>
</tr>
</thead>
<tbody>
<tr>
<td>length</td>
<td>checksum</td>
</tr>
</tbody>
</table>

Application data (message)

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
  --------------------------------
  wraparound:  1 1 0 1 1 1 0 1 1 1 0 1 1 0 1 1
  sum:         1 0 1 1 1 0 1 1 1 0 1 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) → (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