CS 455: INTRODUCTION TO DISTRIBUTED SYSTEMS
[NETWORKING]

The Receiver’s Buffer
Small it may be
But throttle the mightiest sender
It can

Not just the how much
But also the when
Or if at all

Shrideep Pallickara
Computer Science
Colorado State University

Topics covered in today’s lecture

- IP routing
  - IPv6
- UDP
- TCP
IPv6 versus IPv4: Key Differences

- Source and destination addresses are **128-bits** (16 bytes) in IPv6
- IPv6 treats Options as **extension headers**
- To simplify processing of packets in routers, IPv6 **did away with fragmentation**
  - Responsibility for packet fragmentation is at the end points
  - IPv6 hosts must perform: (1) path MTU discovery, (2) perform end-to-end fragmentation, OR (3) send packets no larger than the default MTU=1280
- As of 2014, IPv4 still carried >99% of worldwide Internet traffic
  - 2018: 24 countries (IPv6 > 15%) and 49 countries (IPv6>5%)
IPv6 Packet Header

<table>
<thead>
<tr>
<th>0</th>
<th>4</th>
<th>8</th>
<th>12</th>
<th>16</th>
<th>19</th>
<th>31</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version</td>
<td>Traffic Class</td>
<td>Flow Label</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Payload Length</td>
<td>Next Header</td>
<td>Hop Limit</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SourceAddr [16 bytes]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DestinationAddr [16 bytes]</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

IPv6 Packet Header is fixed at 40 bytes ... So there is no Header Length

IPv6 Packet Header: Some more details [1/2]

- **Version**: 4 bits [0110]
- **Traffic Class**: 6+2 bits
  - Differentiated Services for QoS
  - Anything that ends in 2 “1” bits is intended for experimental or local use
- **Flow Label** (20 bits)
  - If it is non-zero: Serves as a hint to routers and switches with multiple outbound paths that these packets should stay on the same path, so that they will not be reordered
- **Payload length** (16 bits): Size of payload including extension headers
IPv6 Packet Header: Some more details [2/2]

- Next Header (8 bits)
  - Specifies the type of the next header
- Hop Limit (8 bits)
  - Replaces the time-to-live field of IPv4
- Destination and Source Addresses (128-bits or 16 bytes each)
- Note: The IPv6 packet header has no checksum
  - Transport or application layer protocols are assumed to provide sufficient error detection

Structure of the IPv6 Packet

IPv6 Packet

IPv6 Header | Extension Headers | Upper Layer Protocol Data Unit (PDU)

Payload

PDU typically contains an upper layer protocol header and its payload. For e.g.: a TCP segment, UDP Datagram, or an ICMPv6 message
Extension Header

- If the Next Header field is non-zero
  - It defines an extension header
- Current extension header types
  - Information for routers, route definition, fragment handling, authentication, encryption, etc.
- Each extension header has a specific size and defined format
- If an extension header is present?
  - Follows the basic header and precedes the payload AND
  - Includes a Next Header

IPv6 Extension Headers: The chain of pointers using the Next Header field

- Each extension header must fall on a 64-bit (8-byte) boundary. Use Padding to get there if less than that.
UDP
SIMPLE DEMULTIPLEXER

User Datagram Protocol

- **Simplest** possible transport protocol
  - Extends host-to-host into process-to-process communications

- No additional functionality to best-effort service provided by underlying network

- Adds **demultiplexing**
  - Allows applications on a host to **share** the service
UDP identification of processes

- Processes *indirectly* identify each other
  - Abstract locator called *port*

- Source sends a message to a port
  - Destination receives messages from a port

- Process is identified by a *port on a particular host*

Format of a UDP header

<table>
<thead>
<tr>
<th></th>
<th>0</th>
<th>16</th>
<th>31</th>
</tr>
</thead>
<tbody>
<tr>
<td>SrcPort</td>
<td>DstPort</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Length</td>
<td>Checksum</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Data</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
A port is just an abstraction

- Typically implemented as a **message queue**
- When message arrives?
  - Protocol appends message to end of the queue

- **UDP**
  - If the queue is full, message is discarded
  - No flow-control mechanism

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UDP message queue: The port abstraction

[UDP diagram]

- Packets arrive
- Packets demultiplexed

Some work that UDP does do besides demultiplexing:

- Checksumming

- UDP header
- Message body

- **Psuedoheader**: From the IP header
  - Protocol number
  - Source IP address
  - Destination IP address

- UDP length
  - Used twice


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**Reliable Byte Stream**

**TCP**
Components of Reliable delivery

- **Acknowledgements**
  - Confirm receipt of data

- **Timeouts**
  - *Retransmit* if ACK not received within a specified time

- Use of ACKs and timeouts to implement reliable delivery
  - Sometime called ARQ (automatic repeat request)

Simplest ARQ is the stop-and-wait algorithm

- After transmitting one frame
  - Sender *waits* for ACK before transmitting the next frame

- If the ACK does not arrive after a period of time
  - Sender *retransmits* the original frame
Stop-and-wait

Stop-and-wait
Sliding window: Try to fill the network pipe

- Delay x Bandwidth product is 8 KB
- Data frames = 1KB
- Sender could transmit 9th frame
  - When ACK for the 1st frame arrives

Timeline for the sliding window
Transmission Control Protocol (TCP) [1/2]

- **Reliable, in-order** delivery of byte streams

- **Full duplex** protocol
  - Each connection supports a pair of byte streams
    - Flowing in different directions

- Includes **flow control** mechanism
  - Allows receiver to limit the data sender
    - Control how much data can be transmitted at a time
Transmission Control Protocol (TCP)

- Includes **multiplexing** mechanism
  - Multiple applications on a given host

- Implements a **congestion-control** mechanism
  1. *Throttle* how fast TCP sends data
  2. Keep sender from overloading the network

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Flow control and congestion control

- **Flow control** is an end-to-end issue
  - Don’t overrun capacity of receiver

- **Congestion control** is about how hosts & networks interact
  - Don’t cause switches and links to be overloaded
TCP: Setup and Teardown

- Two sides of the connection agree to exchange data
  - Establish shared state
  - 3 packets exchanged (SYN, SYN-ACK, ACK)

- Connection teardown
  - Let each host know it is OK to free the shared state
  - 4 packets exchanged (FIN, ACK, FIN, ACK)

TCP Segments & how they come about

- TCP
  - Accepts data from a data stream
  - Breaks it up into chunks
  - Adds a TCP header ... creating a TCP segment

- Segment is then encapsulated in an IP datagram

- TCP packet is a term that you will often hear
  - Segment is more precise, packets are generally datagrams, frames are at the link layer
How TCP manages a byte stream

Maximum Segment Size (MSS)

- To avoid fragmentation in the IP layer, a host must specify the MSS as equal to the largest IP datagram that the host can handle minus (the IP and TCP header sizes).

- The minimum requirements (in bytes) at the hosts are as follows:
  - IPv4: 576 – 20 – 20 = 536
  - IPv6: 1280 – 40 – 20 = 1220

- Each direction of the data flow can use a different MSS.
**TCP Header Format**

<table>
<thead>
<tr>
<th>0</th>
<th>4</th>
<th>10</th>
<th>16</th>
<th>31</th>
</tr>
</thead>
<tbody>
<tr>
<td>SrcPort</td>
<td>DestPort</td>
<td>Sliding Window Protocol</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SequenceNum</td>
<td>Acknowledgement</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>HdrLen</td>
<td>Flags</td>
<td>AdvertisedWindow</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Checksum</td>
<td>UrgPtr</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Options (variable)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Data</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

SourceAddr and DestinationAddr from IP

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**Relationship between SequenceNum, Acknowledgement and AdvertisedWindow**

Each byte of data has a sequence number.

SequenceNum contains sequence number for first byte of data in segment.
TCP Sliding Window

- Guarantees **reliable** delivery of data
- Data is delivered in **order**
- Enforces **flow control** between the sender and receiver

Sender has a **limit** on unacknowledged data
  - Limited to no more than **AdvertisedWindow** bytes of unacknowledged data

Receiver **selects** **AdvertisedWindow**
  - Based on memory set aside for connection's buffer space
TCP Send Buffer

LastByteAcked ≤ LastByteSent
LastByteSent ≤ LastByteWritten

Example:
LastByteWritten = 3000
LastByteSent = 2800
LastByteAcked = 2400
How many unacknowledged bytes?
(3000 - 2400) = 600

TCP Receive Buffer

LastByteRead < NextByteExpected
NextByteExpected ≤ LastByteRecvd + 1
Flow Control: Buffers are of finite size

MaxSendBuffer and MaxRcvBuffer

- Receiver **throttles** sender
  - Advertises a window
  - No bigger than what it can buffer

\[
\text{MaxSendBuffer} \leq \text{MaxRcvBuffer}
\]

LastByteRcvd – LastByteRead ≤ MaxRcvBuffer

AdvertisedWindow =

\[
\text{MaxRcvBuffer} - (\text{NextByteExpected} - 1) - \text{LastByteRead}
\]

The advertised window may potentially shrink

- If the process is reading data as fast as it arrives?
  - The advertised window *stays open*
    - i.e. AdvertisedWindow = MaxRcvBuffer

- If the receiving process falls behind?
  - Advertised window becomes *smaller* with every segment that arrives
  - Until it becomes 0
Flow Control: Buffers are of finite size
MaxSendBuffer and MaxRcvBuffer

- On the sender size, TCP adheres to the advertised window from the receiver

\[ \text{LastByteSent} - \text{LastByteAcked} \leq \text{AdvertisedWindow} \]

\[ \text{EffectiveWindow} = \] \[ \text{AdvertisedWindow} - (\text{LastByteSent} - \text{LastByteAcked}) \]

**EffectiveWindow should be > 0 before source can send more data**

Reliability is achieved by the sender detecting lost data and retransmitting it

- TCP uses two primary techniques to identify loss
  - Retransmission timeout (RTO)
  - Duplicate cumulative acknowledgements (DupAcks)
    - If the sender receives three duplicate acknowledgements, it retransmits the last unacknowledged packet
Selective Acknowledgements (SACK)

- Using SACK a receiver informs the sender of non-contiguous blocks of data that have been received and queued successfully.
- So the sender need retransmit only the segments that have actually been lost.

Issues with TCP
Protecting against wraparound: 32-bit sequence space

- TCP assumes each segment has a max lifetime
  - Maximum segment lifetime (MSL)
  - Currently this is 120 seconds

- Sequence number used on a connection might wrap-around
  - Within the MSL

<table>
<thead>
<tr>
<th>Bandwidth</th>
<th>Time until wraparound</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1 (1.5 Mbps)</td>
<td>6.4 hours</td>
</tr>
<tr>
<td>Ethernet (10 Mbps)</td>
<td>57 minutes</td>
</tr>
<tr>
<td>T3 (45 Mbps)</td>
<td>13 minutes</td>
</tr>
<tr>
<td>FDDI (100 mbps)</td>
<td>6 minutes</td>
</tr>
<tr>
<td>STS-3 (1.55 Mbps)</td>
<td>4 minutes</td>
</tr>
<tr>
<td>STS-12 (622 Mbps)</td>
<td>55 seconds</td>
</tr>
<tr>
<td>STS-24 (1.2 Gbps)</td>
<td>28 seconds</td>
</tr>
</tbody>
</table>

**STS**: Synchronous Transport Signal

**FDDI**: Fiber Distributed Data Interface
Keeping the pipe full

- AdvertisedWindow field (16-bits) must be big enough
  - To allow sender to keep the pipe full
  - 16 bit allows us a max window size of 64 KB ($2^{16}$)

- If receiver has unlimited buffer space?
  - AdvertisedWindow dictated by Delay x Bandwidth product

Required Window Size for 100 ms delay

<table>
<thead>
<tr>
<th>Bandwidth</th>
<th>Delay x Bandwidth Product</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1 (1.5 Mbps)</td>
<td>18 KB</td>
</tr>
<tr>
<td>Ethernet (10 Mbps)</td>
<td>122 KB</td>
</tr>
<tr>
<td>T3 (45 Mbps)</td>
<td>549 KB</td>
</tr>
<tr>
<td>FDDI (100 mbps)</td>
<td>1.2 MB</td>
</tr>
<tr>
<td>STS-3 (155 Mbps)</td>
<td>1.8 MB</td>
</tr>
<tr>
<td>STS-12 (622 Mbps)</td>
<td>7.4 MB</td>
</tr>
<tr>
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<td>14.8 MB</td>
</tr>
</tbody>
</table>

STS: Synchronous Transport Signal
FDDI: Fiber Distributed Data Interface
TCP extensions: Use 32-bit timestamp to extend sequence number space

- **Distinguish** between different incarnations of the same sequence number
- Timestamp not treated as part of sequence number
  - For ordering etc.
  - Just protects against wraparound

TCP Extension: Allow TCP to advertise larger window

- Fill larger \text{DELAY} \times \text{BANDWIDTH} pipes
- Include option defining \textit{scaling} factor
- Option allows TCP endpoints to agree that \texttt{AdvertisedWindow} counts \texttt{larger chunks}
A caveat regarding Options

- You cannot solve all problems with Options

- TCP Header has room for only 44 bytes of options
  - HdrLen is 4 bits long, so header length cannot exceed 16 x 32-bit = 64 bytes
  - Adding a TCP option that extends the space available for options?

The contents of this slide-set are based on the following references

- Understanding the IPv6 Header: https://www.microsoftpressstore.com/articles/article.aspx?p=2225063&seqNum=4