Transport Layer, Part 2
Transmission Control Protocol

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Introduction

- TCP provides
  - reliable, stream delivery service
  - full-duplex (bidirectional) communication
  - flow/error control
  - congestion control
  - connection establishment and destruction
- TCP does not include application program interfaces (API).
  - API is provided by operating systems.
Stream Delivery Service

- Data from an application are treated as a stream of bytes
- The stream is divided into segments for delivery.
  - this division is up to TCP; application data boundaries are ignored
- Conceptually, every byte has a sequence #.
- Only the sequence # of the first byte of the segment is sent.
- Window sizes are also in bytes (as opposed to the # of frames).

Forced Delivery

- Imagine that you telnet to site.gmu.edu and issue an ls command.
- While you wait for responses, your TCP module considers three bytes (‘l’, ‘s’, and return) too small a segment and decides to wait for additional bytes before sending a segment.
- Applications sometimes need a way to force TCP to transmit data immediately.
- This is achieved by setting a PSH bit to 1 in TCP header
Urgent Data

- TCP also provides a way to urgently send a part of the data stream to the remote process, regardless of the position of that part of data in the stream.
- This is achieved by encapsulating the urgent data in a segment whose URG bit is set to 1.
- Also, a Urgent Pointer in TCP header allows you to specify where the urgent data ends.
- Applications?

TCP Segment Format

<table>
<thead>
<tr>
<th>0</th>
<th>15</th>
<th>16</th>
<th>31</th>
</tr>
</thead>
<tbody>
<tr>
<td>Source Port</td>
<td>Destination Port</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sequence Number</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Acknowledgment Number</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4-bit HLEN</td>
<td>unused</td>
<td>U</td>
<td>A</td>
</tr>
<tr>
<td>G</td>
<td>K</td>
<td>H</td>
<td>T</td>
</tr>
<tr>
<td>Window Size</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>TCP Checksum</td>
<td>Urgent Pointer</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Data</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
The Concept of Connections

- Unlike UDP, TCP uses connections, not ports, to identify message queues
  - A connection is identified by a pair of endpoints.
  - An endpoint is an (IP address, port) pair.
- Examples: three connections involving site.gmu.edu (129.174.40.83):
  - (128.9.4.33, 2000) ↔ (129.174.40.83, 1069)
  - (196.83.4.22, 64) ↔ (129.174.40.83, 25)

Connection Establishment

- Naïve solution: one party sends a connection request message, and the other either accepts or rejects the request.
- However, consider the “storage” property of the Internet.
  - each router uses buffers to store packets waiting for transmission
  - during congestion, a packet could be buffered in the network for a prolonged period of time
  - it looks as if the packet is temporarily stored in the network
A Horrible Scenario

- Image you establish a connection to a bank server to transfer 1 million dollars to a friend.
- The connection messages are “stored” in the Internet.
- Your machine times out and retransmits.
- The new set of messages arrive at the bank server before the original, probably because they follow different paths; 1 million transferred.
- Later, the original request arrives at the server, a second million transferred.

Solution

- Choose the initial sequence numbers (ISN) of a connection based on current time.
- The ISN is included in the connection request message.
- If the request message has to be retransmitted, a new ISN shall be chosen.
- This enables us to recognize obsolete request messages.
Three-Way Hand Shaking

- Machine A initiates a connection to machine B.
- A sends a segment with ISN $x$, chosen according to A’s clock, and with SYN bit set to 1, this is the connection request message.
  - the ACK flag is set to 0
  - this segment is still a regular TCP segment; it can carry data too
  - however, if it does contain data, machine B cannot deliver the data to application until the connection is successfully established

- B returns a segment whose Ack=$x+1$, Seq=$y$, the ISN of B chosen according to B’s clock, and SYN=1.
  - this is the connection acceptance message.
  - if the first segment from A contains $b$ bytes of data, then Ack=$x+b+1$.
  - this segment itself can contain data too.
- After receiving the acceptance, A considers the connection established.

- A then sends B a segment with Ack=y+1 and SYN=0.
  - this is the connection confirmation message
  - all segments from this point on will have SYN=0

- After receiving the confirmation, B considers the connection established.

**Visualize It**
Example

- A to B: SYN=1, ACK=0, seq=100
- B to A: SYN=1, ACK=1, seq=523, ack=101
  - when A receives this segment, it considers the connection successful
- A to B: SYN=0, ACK=1, seq=101, ack=524, data
  - after B receives this segment, it considers the connection successful
  - connection is now full-duplex

A Second Example

- A to B: SYN=1, ACK=0, seq=100
  - *segment lost due to transmission errors*
- A to B: SYN=1, ACK=0, seq=336
- B to A: SYN=1, ACK=1, seq=523, ack=337
  - *segment lost due to transmission errors*
- B to A: SYN=1, ACK=1, seq=880, ack=337
- A to B: SYN=0, ACK=1, seq=337, ack=881, data
Connection Termination

- Consider a connection between $A$ and $B$.
- $A$ terminates the connection by setting $\text{FIN}=1$ in its last outgoing segment.
  - $\text{SYN}=0$, $\text{FIN}=1$, $\text{seq}=x$, $b$ bytes of data
- $A$ considers the connection terminated after receiving the Ack of its last segment.
  - $\text{SYN}=0$, $\text{FIN}=0$, $\text{ack}=x+b+1$
- Both $\text{SYN}$ and $\text{FIN}$ counted as one byte in the sequence space.

Half-Closed Connections

- After $A$ terminates the connection, the connection is still considered open by $B$ and is said to be **half-close**.
  - when a connection is half-close, data can flow in only one direction ($B$ to $A$)
  - $A$ must continue receiving data from $B$ and returning Ack's.
- **B** is responsible for terminating the **B**-to-**A** direction by setting **FIN**=1 in **B**’s last outgoing segment:
  - **SYN**=0, **FIN**=1, seq=x, b bytes of data
- **B** considers the connection closed after receiving the Ack of its last segment:
  - **SYN**=0, **FIN**=0, **ack**=x+b+1
  - the connection is then completely terminated

- It is up to **B** to decide whether it would terminate the connection upon receiving the **FIN** from **A**.
- Either endpoint can close the connection first, regardless who initiates the connection.
Example

- Consider a telnet session where the user logs in to server B via his/her PC A.
- The user closes the session by typing “exit\n”.
- A to B: FIN=1, seq=1060, ack=7777, data="exit\n"
- B to A: FIN=1, seq=7777, ack=1066
- A to B: FIN=0, seq=1066, ack=7778

Half-Close Example

- The last segment from client A to database server B is a request message, 10-byte long.
- A to B: FIN=1, seq=2405, ack=8002, 10-byte data
- B to A: FIN=0, seq=8002, ack=2405+10+1=2416, no data
The server process reads the request and sends a long reply, corresponding to multiple segments.
- the connection is half close when the reply is in transit

Consider the last reply segment, assumed 100 bytes.

- B to A: FIN=1, seq=9125, 100-byte data
- A to B: FIN=0, ack=9125+100+1=9226

TCP Acknowledgments

- TCP acknowledges the next byte expected (not the last one received).
- Ack=x indicates all bytes up to and including x-1 has been successfully received.
- SYN=1 occupies a “byte” in the sequence number space; so does FIN=1.
- A pure Ack is simply a segment with no data
- A receiver could postpone acks up to 200ms for chances of piggybacking.
Exercise

☐ Assume that all bytes up to 999 have been received and acked. Give the acks in response to segments
  – Seg(1000, 200)
  – Seg(1300, 100)
  – Seg(1400, 300)
  – Seg(1700, 100)
  – Seg(1200, 100)

Exercise

☐ Give the acks in response to segments
  – Seg(1000, 200, SYN=1)
  – Seg(1300, 100)
  – Seg(1400, 300)
  – Seg(1700, 100, FIN=1)
  – Seg(1201, 99)
**TCP Sliding Window Protocol**

- The sliding window protocol of TCP must work with dynamic conditions.
  - Unlike the DLL, round-trip times between the two endpoints are unknown a priori.
  - Moreover, round-trip times not fixed, due to fluctuations in background traffic.
- Time-out intervals must constantly adjusted according to present RTT.

**RTT and Timeout Interval**

- If the timeout interval is too short, unnecessary retransmissions occur.
- If the interval is too long, performance suffers when packets are actually lost (waiting for too long to retransmit).
- Recall that round-trip times are not known a priori and could change (perhaps widely) during the lifespan of the connection.
- We need to measure RTT in order to derive timeout intervals.
An Early Solution

- When an outgoing packet is stored in the window, the current time \( T_1 \) is recorded.
- When its ack arrives at time \( T_2 \) we obtain an RTT sample: \( \text{New}_\text{Sample}=T_2-T_1 \).
- A variable RTT is maintained for each conn.
- When an ack arrives
  \[
  \text{RTT} = \alpha \times \text{RTT} + (1 - \alpha) \times (\text{New}_\text{Sample})
  \]
  \[
  \text{Timeout} = \beta \times \text{RTT}
  \]
where \( \alpha = 0.9 \) and \( \beta = 2 \).

Problem?

- What is your choice of timeout interval if the past three RTT samples are
  - 10, 9.1, and 10.1 milliseconds
  - 1, 20, and 9 milliseconds
RTT Variances

- Associated with each connection is a second variable, DEV, to estimate the mean deviation of round trip times.

\[
\text{DIFF} = \text{SAMPLE} - \text{RTT}
\]

\[
\text{RTT} = \text{RTT} + \delta \ast \text{DIFF}
\]

\[
\text{DEV} = \text{DEV} + \rho \ast (|\text{DIFF}| - \text{DEV})
\]

\[
\text{Timeout} = \text{RTT} + \tau \ast \text{DEV}
\]

where \( \delta \) is recommended to be 1/8, \( \rho \) 1/4, and \( \tau \) 4.

Retransmission Ambiguity Problem

- Consider a packet with seq=x, first transmitted at time \( T_1 \) and then retransmitted at time \( T_2 \).
- An Ack(x) arrives at time \( T_3 \).
- What is the value of the new RTT sample?
  \( T_3 - T_2 \) or \( T_3 - T_1 \)?

Naïve Solution: discard all samples pertaining to packets that have been retransmitted.

Problem: assuming real RTT=30 ms, and current timeout=20 ms
Kern's Algorithm

- When computing the RTT variable, ignore all round-trip time samples that correspond to retransmitted segments.
- Further, when an ACK that corresponds to a retransmitted packet arrives, double the timeout interval.
- Derive the timeout interval from variable RTT when a valid RTT sample arrives.

<table>
<thead>
<tr>
<th></th>
<th>0</th>
<th>10</th>
<th>20</th>
<th>30</th>
<th>40</th>
<th>50</th>
<th>60</th>
<th>70</th>
<th>80</th>
<th>90</th>
<th>100</th>
<th>110</th>
</tr>
</thead>
</table>

Put Them Together

When an Ack arrives:

if (it is ambiguous)

Timeout = 2*Timesout

else

DIFF = SAMPLE – RTT
RTT = RTT + δ*DIFF
DEV = DEV + ρ*(|DIFF| - DEV)
Timeout = RTT + τ*DEV

endif