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Lab Exercise 6: Throughput, IP Fragmentation and Routing

Exercise 1: Setting up NS2 simulation for measuring TCP throughput

Question 1: Why the throughput achieved by flow tcp2 is higher than tcp1 between time span 6 sec to 8 sec?

Tcp1 is from n0 to n5 and tcp2 is from n3 to n5. Also, the 3rd and 4th flow compete more with tcp1 rather than tcp2. As a result, tcp2 has less RTT than tcp1. In other words, tcp2 should have higher share of the bandwidth and therefore has higher throughput recorded at n5.

Question 2: Why the throughput for flow tcp1 is fluctuating between time span 0.5 sec to 2 sec?

Because the flow tcp1 is in the slow start phase between time span 0.5 sec to 2 sec.

Question 3: Why is the maximum throughput achieved by any one flow capped at around 1.5Mbps?

This is because of the slow start phase and the congestion control. The main reason is that there are not any time span that one flow can use all the bandwidth except the flow tcp1. At the beginning, tcp1 is the only flow active but it is still in the slow start phase, and after 2 sec, it has to compete with other flows. In my opinion, there is not enough time for tcp1 to discover the maximum bandwidth, which is 2.5Mbps. Therefore the maximum throughput achieved by any one flow capped at around 1.5Mbps.

Exercise 2: Understanding IP Fragmentation

Question 1: Which data size has caused fragmentation and why? Which host/router has fragmented the original datagram? How many fragments have been created when data size is specified as 2000?

The 2000 data packet caused fragmentation. The data size caused fragmentation is 1500 bytes. Because in the 17th message, there is a [2 IPv4 Fragments (2008 bytes): #16(1480), #17(528)] segment, from those information, we can know that the first segment is 1480 bytes. Therefore 1480-8 = 1472, 1472 + IP header + ICMP header = 1500.

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▾ [2 IPv4 Fragments (2008 bytes): #16(1480), #17(528)]
      [Frame: 16, payload: 0-1479 (1480 bytes)]
      [Frame: 17, payload: 1480-2007 (528 bytes)]
      [Fragment count: 2]
      [Reassembled IPv4 length: 2008]
      [Reassembled IPv4 data: 080008f5d90500005b51dd800009a51108090a0b0c0d0e0f...]
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The host/router has fragmented the original datagram is e0:ac:cb:64:20:54
Two fragments have been created by router when data size is 2000.

Question 2: Did the reply from the destination 8.8.8.8. for 3500-byte data size also get fragmented? Why and why not?

Yes the reply from the destination 8.8.8.8 also get fragmented. Because it exceeds maximum transmission unit.

Question 3: Give the ID, length, flag and offset values for all the fragments of the first packet sent by 192.168.1.103 with data size of 3500 bytes?

1. ID: 0x7a7b length:1514 bytes flag: 0x01 offset:0
2. ID: 0x7a7b length:1514 bytes flag: 0x01 offset:1480
3. ID: 0x7a7b length:582 bytes flag: 0x00 offset:2960

Question 4: Has fragmentation of fragments occurred when data of size 3500 bytes has been used? Why and why not?

Yes. Because 3500 bytes is larger than the maximum transmission unit, so the data will be fragment.

Question 5: What will happen if for our example one fragment of the original datagram from 192.168.1.103 is lost?

According to the Wireshark, the packets are send by UDP. Therefore nothing will happen because UDP will do nothing about lost packets. If it is send by TCP, then I think the lost packets will be resend.

Exercise 3: Understanding the Impact of Network Dynamics on Routing

Question 1: Which nodes communicate with which other nodes? Which route do the packets follow? Does it change over time?

Node 0 communicate with node1
Node 1 communicate with node0, node2, node4
Node 2 communicate with node1, node 3
Node 3 communicate with node2, node 5
Node 4 communicate with node1, node 5
Node 5 communicate with node3, node 4
One route is from node 0 to node 1 to node 4 and to node 5. Another route is from node 2 to node 3 to node 5.
Yes, the route changed over time. The first route start at 0.5 sec, then at 0.6 sec, the second route begin to transmit from node 2, the route is the shown above. The first one stopped at 1.5 sec and the second one stopped at 1.6 sec.

Question 2: What happens at time 1.0 and at time 1.2? Does the route between the communicating nodes change as a result of that?

At time 1.0, there is something wrong between node 1 and node 4, the link is interrupted and the packets stopped at node 1._At time 1.2, the link between node 1 and node 4 recovered. In my opinion, the route does not change as a result of that, although the link is interrupted, the packet still try to go through the same route.

Question 3: Did you observe any additional traffic as compared to Step 3 above? How does the network react to the changes that take place at time 1.0 and time 1.2 now?

Yes, compared to step 3 above, while the link between node 1 and node 4 interrupted, the packets go through another route, which is from node 0 to node 1to node 2 to node 3 and then to node 5. After the link recovered, the packets go through the original route.
Basically, now the node 1 transmit packets to node 2 at tome 1.0 when the link between node 1 and node 4 is interrupted, and stopped to transmit to node 2 at time 1.2, because the link recovered.

Question 4: How does this change affect the routing? Explain why.

When a packet go from node 0 to node 1, node1 now choose to transmit packets to node 2 instead of node 4. In my opinion, this is because the cost between ode 1 and node 4 has been set to 3. Therefore, the node 1 choose lowest cost link to complete transmission.

Question 5: Describe what happens and deduce the effect of the line you just uncommented.

The line I just uncommented set more cost of the links and allow the nodes to set multiple paths if there is a low cost path. Therefore, after the node 2 starts to transmit packets, there are 2 possible route, one is from node 2 to node 3 and then to node 5, another route is from node 2 to node 1 to node 3 and then to node5. Since these routes have same cost, it have multiple path.