ECE257A HW4

Chi-Hsin Lo
UCSD ECE
A53311981

I. Problem 1 - True or false problems (If false, explain why).

(a) The shortest-path routing problem is an integer linear programming problem, so it is NP-hard.

(b) The wireless channel is reciprocal, so a routing protocol can use the forward direction packet delivery ratio equivalently as the backward direction packet delivery ratio.

(c) Unlike the ETX metric, the ETT metric takes into account the overhead of the ACK packet.

(d) Suppose Host A sends one segment with sequence number 38 and 4 bytes of data over a TCP connection to Host B. In this same segment the acknowledgment number must be 42.

(e) Consider congestion control in TCP. When the timer expires at the sender, the value of ssthresh is set to one half of its previous value.

(f) In general, a longer RTT causes lower average TCP throughput.

Answer :

(a) False. The shortest path Dijkstra algorithm using min heap has time complexity $\Theta((|V| + |E|)\log|V|)$ which is polynomial time.

(b) False. The link loss is asymmetric which means the forward direction packet delivery ratio is different from the backward direction packet delivery ratio.

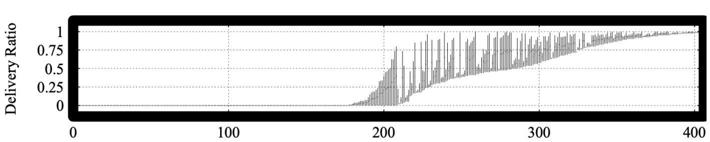

Fig. 1.

(c) False. The $ETX = \frac{1}{(df \times dr)}$ and $ETX(b) = \frac{1}{(df(b) \times dr)}$ and for packet of length S $ETT(b) = \frac{S}{b} \times ETX(b)$ so both ETX and ETT takes into account the overhead of the ACK packet, but ETX didn't take into account link with different bit rates.

(d) False. The acknowledgment number in the same segment is the next packet sequence number it expect from B. The acknowledgement number of this segment should be the sequence number of next byte Host A expect from Host B. 42 is the acknowledgment number B will send back to A with its next packet if it successfully receive the packet from A.

(e) False. When the timer expires at the sender (timeout), the value of ssthresh is set to half of cwnd (congestion window size) just before loss event.

(f) True. Estimated bandwidth $B = \frac{packet_size}{RTT \times \sqrt{\frac{2p}{3}}}$ where p is the packet loss rate.

II. Problem 2 - Mobile IP.

(a) DHCP can allocate IP address dynamically to users whenever users enter a new network (e.g., connect to the Ethernet switch using cables). Can DHCP itself solve the IP addressing problem for mobile wireless users? Why?

(b) Ideally, a mobile IP solution should be "transparent" to application layer. What does transparency mean? Why is it important? Between direct routing and indirect routing, which one satisfies "transparency" better?

(c) Compare the pros and cons of direct vs. indirect routing for mobile IP.

Answer :

(a) No. When DHCP server assigns a new IP address then user enter a new network, it doesn't let either the foreign agent or the home agent know, nor does it broadcast it, so a correspondent doesn't know where to send the packet, nor does the home agent. And the IP address is not for communication outside the local network.

(b) Transparency means the protocol is able to transmit the data over the network both the case the sender and receiver are in the same network or different network, and the sender doesn't need to know now where the receiver is and relevant network information. It is important because if when the receiver changes location and gets a new IP, the intermediate router needs to update its table and it's not scalable if there are millions of mobile. Plus, the constant changing of IP address will require TCP to reestablish connection, which will cause temporary disconnection which is unfriendly to user. Indirect routing satisfies transparency and direct routing does not. In indirect routing the packet is forwarded to the home agent with original address to send to the mobile through foreign agent, while in direct routing the sender gets the foreign address of mobile from the home agent.

(c) Reference : Lecture 13 slide 24 and 26

Direct vs. Indirect Routing		
Protocol	Pros	Cons
Direct Routing	Overcome triangle routing problem	Non-transparent, needs chaining and reconnection when mobile changes visited network
Indirect Routing	It's all transparent and ongoing connections can be maintained.	Triangle routing is inefficient when correspondent and mobile are in same network.

III. Problem 3 - Routing models.

Consider the network topology shown below. The label "(x, y)" on each link means cost=x and capacity=y.

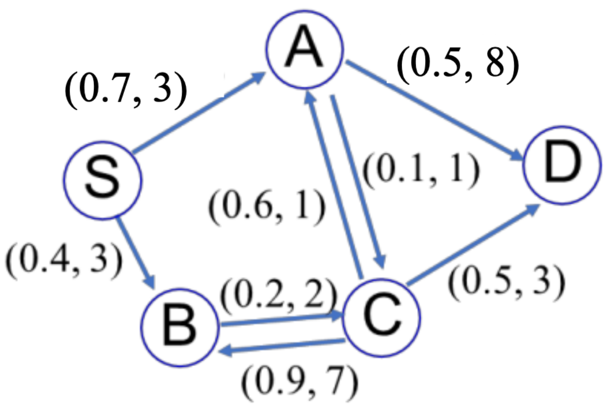

Fig. 2.

(a) Suppose the source node S wants to compute the shortest path from itself to all other nodes in the network. The shortest path is defined by hop count alone, without considering the cost and capacity of each link. Formulate the problem as a linear optimization problem.

(b) Now take into account the cost of each link, and reformulate the optimization problem.

(c) Now there are two sets of source-destination pairs, $S \to D$ and $B \to D$. All the links have cost 0, but limited capacity as indicated in the topology. Suppose a network operator wants to maximize the total throughput of these two source-destination pairs. Formulate the problem as an optimization problem.

Answer :

(a) Minimize $c^T x$, subject to $Ax = b, x_{ij} \in 0, 1$.

$$x = [x_{SA}, x_{SB}, x_{BC}, x_{CB}, x_{AC}, x_{CA}, x_{AD}, x_{CD}]^T$$

$$c = [1, 1, 1, 1, 1, 1, 1, 1]^T$$

$$b = [1, -1, -1, -1, -1]^T$$

$$A = \begin{bmatrix} 1 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ -1 & 0 & 0 & 0 & 1 & -1 & 1 & 0 \\ 0 & -1 & 1 & -1 & 0 & 0 & 0 & 0 \\ 0 & 0 & -1 & 1 & -1 & 1 & 0 & 1 \\ 0 & 0 & 0 & 0 & 0 & 0 & -1 & -1 \end{bmatrix}$$

(b) Minimize $c^T x$, subject to $Ax = b, x_{ij} \in 0, 1$.

$$x = [x_{SA}, x_{SB}, x_{BC}, x_{CB}, x_{AC}, x_{CA}, x_{AD}, x_{CD}]^T$$

$$c = [0.7, 0.4, 0.2, 0.9, 0.1, 0.6, 0.5, 0.5]^T$$

$$b = [1, -1, -1, -1, -1]^T$$

$$A = \begin{bmatrix} 1 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ -1 & 0 & 0 & 0 & 1 & -1 & 1 & 0 \\ 0 & -1 & 1 & -1 & 0 & 0 & 0 & 0 \\ 0 & 0 & -1 & 1 & -1 & 1 & 0 & 1 \\ 0 & 0 & 0 & 0 & 0 & 0 & -1 & -1 \end{bmatrix}$$

(c) Maximize $f_1 + f_2$, subject to $Ax = b, x \leq u$.

$$x = [x_{SA}, x_{SB}, x_{BC}, x_{CB}, x_{AC}, x_{CA}, x_{AD}, x_{CD}]^T$$

$$b = [f_1, 0, f_2, 0, -f_1 - f_2]^T$$

$$A = \begin{bmatrix} 1 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ -1 & 0 & 0 & 0 & 1 & -1 & 1 & 0 \\ 0 & -1 & 1 & -1 & 0 & 0 & 0 & 0 \\ 0 & 0 & -1 & 1 & -1 & 1 & 0 & 1 \\ 0 & 0 & 0 & 0 & 0 & 0 & -1 & -1 \end{bmatrix}$$

$$u = [3, 3, 2, 7, 1, 1, 8, 3]^T$$

IV. Problem 4 - Wireless routing.

(a) What are the main challenges for multi-hop routing in wireless networks, in comparison with multi-hop wireline networks?

(b) Consider the following network topology. Link label (x, y) indicates loss rate=x and link bit-rate $= y$ Mbps. Suppose links $A \leftrightarrow B$ and $D \leftrightarrow E$ can transmit concurrently, and any other pairs of links will interfere with each other if triggered simultaneously. Suppose A is the source node and E is the destination. What is the ETX metric for each link? What is the ETT metric? (You can ignore the ACK transmission time when calculating the ETT, as mentioned in lecture). Suppose each data packet is 1Kb. What is the optimal expected time to deliver 10 Mb of data? Is it equal to the sum of expected time on each link? Explain how you derived the answers in detail.

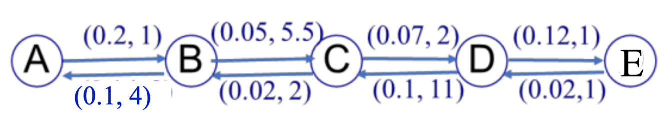

Fig. 3.

Answer:

(a) The main challenges are :

1. Highly dynamic network topology due to mobility
2. Highly dynamic link quality due to channel condition changes
3. The network cannot afford too frequent routing message exchange
4. Hop count no longer works, as attenuation due to distance affects more on packet delivery rate.

Reference : Lecture 14 slide 25

(b)

$$ETX = \frac{1}{(df \times dr)}$$

$$ETT(b) = \frac{S}{b} \times \frac{1}{(df(b) \times dr)}$$

For link $A \leftrightarrow B$

$$ETX = \frac{1}{(0.8 \times 0.9)} = 1.389$$

$$ETT(b) = \frac{S}{b} \times \frac{1}{(df(b) \times dr)} = \frac{S}{1} \times \frac{1}{(0.8 \times 0.9)} = 1.389S$$

For link $B \leftrightarrow C$

$$ETX = \frac{1}{(0.95 \times 0.98)} = 1.074$$

$$ETT(b) = \frac{S}{b} \times \frac{1}{(df(b) \times dr)} = \frac{S}{5.5} \times \frac{1}{(0.95 \times 0.98)} = 0.195S$$

For link $C \leftrightarrow D$

$$ETX = \frac{1}{(0.93 \times 0.9)} = 1.195$$

$$ETT(b) = \frac{S}{b} \times \frac{1}{(df(b) \times dr)} = \frac{S}{2} \times \frac{1}{(0.93 \times 0.9)} = 0.5975$$

For link $D \leftrightarrow E$

$$ETX = \frac{1}{(0.88 \times 0.98)} = 1.160$$

$$ETT(b) = \frac{S}{b} \times \frac{1}{(df(b) \times dr)} = \frac{S}{1} \times \frac{1}{(0.88 \times 0.98)} = 1.16S$$

For $S = 1Kb$, the optimal time for transmitting one packet is $(max(1.389, 1.16) + 0.195 + 0.5975)0.001 = 0.0021815$ seconds. The time optimal expected time of transmitting $10Mb$ data is 21.815 seconds. The sum of expected time on each link is $(1.389, 1.16 + 0.195 + 0.5975)0.001 * 10000 = 33.415$ seconds. They are not equal since $A \leftrightarrow B$ and $D \leftrightarrow E$ can transmit concurrently.

V. Problem 5 - Wireless TCP.

(a) TCP doesn't work well when an end-to-end path involves a wireless link. Why?

(b) What are the pros and cons of Split TCP and Snoop TCP?

(c) How does an ELN access point (or basestation) determine whether an uplink packet loss is due to congestion or poor link condition? How does it notify the TCP sender about a link loss?

Answer:

(a) TCP doesn't work well because concludes any packet loss to congestion and reduces cwnd accordingly. But in fact, packet loss can be due to other reasons like poor channel quality or mobility (When a mobile moves from one access point to another, packets sent with previous access point cannot arrive at the right place). In those cases, reducing cwnd may be too conservative and reacting the wrong way and cause unnecessary decrease to throughput.

Reference : Lecture 16 slide 9

(b) Reference : Lecture 16 slide 16, 17, 33, 34

Split TCP vs. Snoop TCP		
Protocol	Pros	Cons
Split TCP	1. No changes needed in wired network or content servers 2. Transmission errors do not propagate into the fixed network 3. Chance of using custom transport protocol for the hop between AP and mobile	1. Violation of end-to-end design principle 2. Large buffer space may be needed at AP to maintain TCP connection state 3. State must be forwarded to new AP on handoff (Causes extra lantency)
Snoop TCP	1. Downlink works without modification to mobile or server, follows end-to-end principle. 2. Crash does not affect correctness, only performance. 3. State migration is not required after AP handoff	1. Mobile host still needs to be modified at MAC and transport layers for NACK 2. Slight violation of the end-to-end principle

(c) ELN cannot distinguish whether the packet loss is due to congestion or poor link quality but determine them all as the later case. But is makes sense since the poor link quality case is more common and congestion is mostly handled by the MAC layer which cannot handle packet loss due to channel quality. AP keeps track of the gaps in the TCP packet sequence it received from the mobile sender. When it sees a duplicate ACK, it compares the ACK sequence number with its recorded gaps, and if they match, AP will set the ELN bit in the duplicate ACK and forward it to the sender.

Reference : Lecture 16 slide 37, 38

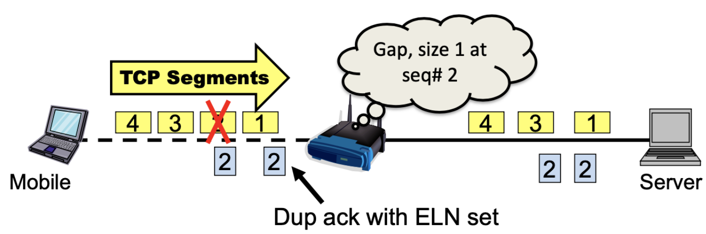

Fig. 4.

VI. Problem 6 - Mobile applications.

(a) HTTP underutilizes the network capacity when running over a network path that contains a cellular link. What's the reason behind?

(b) Video applications encounter more performance issues when running over wireless networks. What's the reason behind? What are the possible solutions? Why would they work?

Answer:

(a) TCP performs best when there is a steady stream of packet to drive the cwnd to converge to the true network capacity. When downloading the webpage, there are not enough data to make the cwnd converge, so it will underutilize the network capacity. Cellular link are wireless networks which have long RTT, which will cause TCP takes longer to converge to the true network capacity. Moreover, link loss over wireless cellular network makes the convergence even slower. Mobility of the devices also make the network capacity to change quickly, with long RTT it makes it even harder for TCP to follow the true network capacity.

Reference : Lecture 17 slide 8

(b) When it comes to video application, there are some additional challenges. Once the client starts playing the video, the playback must match original timing. But network delays are variable, so we will need client-side buffer to take care of this. Another challenge is client interactivity, the client may pause, fast-forward, rewind and jump through the video. The video packets may also be lost and retransmitted.

The first solution is **smart buffering**. The receiver may use a large buffer to smooth out the network capacity variation and play it out only after receiving a sufficient number of video frames, while this method is unsuitable for interactive videos which may need immediate playback.

The second solution is **active bandwidth probing**. Video server periodically sends some dummy data to estimate the end-to-end network capacity and then chooses the bit-rate that fits the network capacity. This requires modification to the video streaming protocol itself and costs extra network resource to probe frequently to keep track of the true network capacity.

The third solution is **physical layer informed mobile video streaming**. Since wireless link is often the bottleneck along the end-to-end path, it suffices to estimate the wireless link capacity based on PHY layer statistics such as signal strength, time/ frequency resource utilization. This requires the wireless link to provide PHY layer statistics which is already available in many WiFi and cellualar devices.

Reference : Lecture 17 slide 13, 16, 21, 22, 23