Name .... ID .... Section ...... Seat No ......

Sirindhorn International Institute of Technology
Thammasat University

Final Exam: Semester 2, 2010

Course Title: ITS413 Internet Technologies and Applications

Instructor: Steven Gordon

Date/Time: Wednesday 9 March 2011; 13:30–16:30

Instructions:

- This examination paper has 19 pages (including this page).
- Conditions of Examination: Closed book; No dictionary; Non-programmable calculator is allowed
- Students are not allowed to be out of the exam room during examination. Going to the restroom may result in score deduction.
- Students are not allowed to have communication devices (e.g. mobile phone) in their possession.
- Write your name, student ID, section, and seat number clearly on the front page of the exam, and on any separate sheets (if they exist).
- The space on the back of each page can be used if necessary.
- Reference material at the end of the exam may be used.

Question 1 [34 marks]

Consider an IPTV network to be built across Bangkok by the company called *False Corporation*. The network will deliver the "triple-play service" to subscribers: video, voice and Internet across the single network. The video will include standard definition digital TV and video-on-demand, while voice will be using the G.726 codec. You are a consultant to *False*, advising them on the network design.

(a) Explain the difference between ADSL2+, FTTH and FTTN, as options for the service provider access network. State the transmission media they use and the advantages/disadvantages of each technology in your explanation. [3 marks]

(b) *False* will need to sell equipment for the home network. What network technology do you advise to be used inside the home network and why (give two reasons)? [3 marks]

(c) Draw a diagram illustrating the network topology of the entire network, showing at least: core network, service provider access network, home network, PSTN, Internet, TV network. Also identify the video headend, and two subscribers on separate access networks. [4 marks]

You estimate that each standard definition video session requires 5Mb/s of core network bandwidth. The core network capacity that can be allocated to video is planned to be 10Gb/s. (In the following three parts, if you do not have enough information to determine an exact number, then explain what will impact on the number).

(d) What is the maximum number of users and/or channels of standard definition TV that can be supported by the core network if *multiple unicast* is used as the delivery mechanism? Explain your answer, stating any assumptions. [2 marks]

(e) What is the maximum number of users and/or channels of standard definition TV that can be supported by the core network if *multicast* is used as the delivery mechanism? Explain your answer, stating any assumptions. [2 marks]

(f) What is the maximum number of users and/or channels of standard definition video-on-demand that can be supported by the core network? Explain your answer, stating any assumptions. [2 marks]

(g) If the core network supported multicast, then explain what IGMP is used for. [1 mark]

Now consider the voice traffic to be sent across the network. The G.726 codec produces a sample size of 20 Bytes and has a sample interval of 5ms. It generates a RTP packet every 20ms.

(h) What protocol should be used for signalling in the IPTV network? [1 mark]

(i) If 1Gb/s of the core network capacity is allocated to voice calls, what is the maximum number of voice calls supported? (You should assume the data link layer and physical layer contribute 5 Bytes of header in total to each IP datagram) [5 marks]

(j) An alternative to the G.726 codec is the G.711 codec which has 80 Byte samples, 10ms sample intervals and generates a RTP packet every 20ms. What is an advantage of using G.711 instead of G.726 in the network? [2 marks]

You need to recommend QoS mechanisms for the network. Two options available are service differentiation or soft QoS using DiffServ, and hard QoS using IntServ.

(h) Explain the difference between soft QoS and hard QoS. [2 marks]

(i) Explain how hard QoS is provided in IntServ. [2 marks]

Consider soft QoS being used to provide priority to video and voice packets from subscribers that have paid for a premium service. Inside the core network, routers check the DiffServ field (also called Type of Service (ToS) or DiffServ Code Point (DSCP)) in the IP header to determine the priority to give packets. There are two values of the DiffServ field available:

000000 Best effort traffic

101110 Voice and video of premium users

For packets sent by subscribers' computers, the DiffServ field is initially ignored. Instead a router on the edge of the core network classifies packets and marks the DiffServ field.

(j) Explain how this edge router can classify packets, giving two examples that illustrate classifying based on type of user (premium or normal) and type of application (voice/video or other). [3 marks]

(k) Explain how other routers in the core network can provide priority to packets marked with DiffServ field '101110'. [2 marks]

Question 2 [9 marks]

Consider the times at which six packets were transmitted by a source and received by the destination (all times are relative to an initial clock value and measured in milliseconds (ms); the clocks at source and destination are synchronised):

Packet 1 Transmit time: 20; Receive time: 42

Packet 2 Transmit time: 40; Receive time: 62

Packet 3 Transmit time: 60; Receive time: 80

Packet 4 Transmit time: 80; Receive time: 103

Packet 5 Transmit time: 100; Receive time: 119

Packet 6 Transmit time: 120; Receive time: 140

(a) What is the average delay experienced in the network? [1 mark]

(b) What is the jitter experienced in the network? [2 marks]

(c) Playback buffers are often used to compensate for jitter. Explain how a playback buffer can be used in this case, and how it reduces the effect of jitter. [2 marks]

(d) Using a playback buffer, what is the preferred playback time of each of the 6 packets? [2 marks]

(e) One disadvantage of playback buffers is the additional complexity/memory needed. What is another disadvantage of using a playback buffer? Use the example six packets to explain. [2 marks]

Question 3 [14 marks]

(a) Explain the role of an indexer in a Bittorrent network. [1 mark]

(b) Explain the role of a tracker in a Bittorrent network. [1 mark]

(c) What application protocol does a Bittorrent client use to communicate with a tracker? [1 mark]

(d) Once a peer obtains a .torrent file, can Bittorrent be considered as a fully distributed (or de–centralised) system? Explain your answer. [2 marks]

Assume a peer, N_1, has joined a swarm and established two connections to peers N_2 and N_3. Peer N_1 wants to download a torrent with 100 pieces: $P_1, P_2, P_3, \ldots, P_{100}$. Each piece has 10 blocks (e.g. piece P_1 has blocks $B_{1,1}, B_{1,2}, \ldots, B_{1,10}$; piece P_2 has blocks $B_{2,1}, B_{2,2}, \ldots, B_{2,10}$). Peers N_2 and N_3 already have the following pieces:

N_2: $P_5, P_{10}, P_{11}, P_{13}, P_{20}, P_{23}, P_{30}, P_{39}, P_{64}$

N_3: All pieces *except* pieces P_5, P_{20}, P_{30}

In the Peer Exchange Protocol, after an initial *Handshake*, each peer exchanges a *Bitfield* message which indicates the pieces they have available. Then a peer may send a *Request* message to request a specific block, and receive a *Piece* message containing a specific block.

(e) Assuming only peers N_1, N_2 and N_3 are in the swarm, what is the availability of the torrent? Explain you answer. [2 marks]

(f) If there were another two seed peers in the swarm, N_4 and N_5, what would the availability be? [1 mark]

(g) In the *Bitfield* message sent from N_2 to N_1, what values will be included? [1.5 marks]

(h) Assume N_1 uses the "rarest-piece first" algorithm to select the ordering of pieces to download. If connected to N_2 and N_3 which pieces will N_1 NOT download first? [1.5 marks]

Each peer maintains four variables for each other peer it is connected to: *am_choking*, *am_interested*, *peer_choking*, *peer_interested*. Consider the values that peer N_1 maintains for the other two peers:

N_2: *am_choking=False*, *am_interested=True*, *peer_choking=False*, *peer_interested=True*

N_3: *am_choking=False*, *am_interested=True*, *peer_choking=True*, *peer_interested=False*

(i) Will N_1 send a *Request* message to N_2? Explain your answer. [1 mark]

(j) Will N_3 send a *Request* message to N_1? Explain your answer. [1 mark]

(k) Can N_1 download pieces from N_3? Explain your answer. [1 mark]

Question 4 [22 marks]

(a) Explain the difference between flow control and congestion control. [2 marks]

(b) Using any of the following variables, write an equation that gives the approximate sending rate, S, of a TCP source (when not packets are lost): round trip time, RTT; congestion window, *cwnd*; advertised window, *awnd*; slow start threshold, *ssthresh*; maximum segment size, *MSS*. [2 marks]

(c) If a link between two computers has a capacity of 10Mb/s and RTT of 5ms, what is the value its bandwidth-delay product? [2 marks]

(d) Ignoring congestion control and packet headers, if the receiver computer in part (c) had a maximum buffer size of 5KB, what do you think the approximate throughput will be for the TCP connection? [2 marks]

(e) If the maximum buffer size in part (d) was doubled to 10KB, then explain how it will impact on the TCP throughput. [2 marks]

(f) What method does a TCP source use to reduce its sending rate for congestion control? [1 mark]

(g) What event(s) does a TCP source assume indicates decreased congestion? [1 mark]

(h) Explain the difference between how a TCP source increases its sending rate in the additive increase phase compared to the slow start phase. Also indicate when does the TCP source change from one of these phases to the other. [3 marks]

(i) In TCP, packet losses are detected by the source when either a timeout occurs or 3 duplicate ACKs are received. Explain how the TCP congestion control algorithm responds to each of these packet loss events. (You don't need to give exact algorithms, but should indicate how and why the congestion control algorithm responds differently to the two different events). [3 marks]

(j) Explain what is meant if TCP is described as "fair". [2 marks]

(k) Computers A, B and C share a bottleneck link with capacity 8Mb/s to the Internet. On computer A is a Bittorrent application that is downloading from 6 peers in a swarm. On computer B is a web browsing application that is downloading a file from a web server. Computer C also has a web browsing application downloading from a different web server. What is the approximate download rate achieved by the applications on each computer? (that is, give the download rate for application on A, application on B and application on C) [2 marks]

Question 5 [10 marks]

Two challenges of using P2P systems for sharing resources are: searching and data transfer. This question is only about searching.

(a) In Napster-like P2P systems an index is stored on a central server. What is the index (that is, what important information does it contain)? [2 marks]

(b) Where is the index information stored in a FastTrack P2P system? [2 marks]

(c) Explain how searching works in a FastTrack P2P system. As a guide, you should clearly explain the steps for all typical cases of where the queried resource may be located (e.g. the conditions when a response to a query is returned). You should state which nodes send the queries, and to what destinations and using what method. [3 marks]

(d) Explain an advantage that FastTrack has compared to Gnutella. [1.5 marks]

(e) Explain a disadvantage that FastTrack has compared to Gnutella. [1.5 marks]

Question 6 [11 marks]

You are designing a P2P system for file sharing within your company using a Chord Distributed Hash Table. The maximum number of users in the company is 1,000, while it is expected that there will be no more than 1,000,000 different resources (files) to be shared.

(a) You use a hash function that produces a k-bit hash value. What should the minimum value of k be in your network, and why? [2 marks]

Despite the minimum value given in part (a), the software you are using requires k to be 32. Assume this value for the following parts. Also assume of the 1,000 users, there are 5 users with the IDs: 205, 443, 444, 500 and 680. There are no other users with IDs between 205 and 680 (that is, the other 995 users have IDs outside this range).

(b) What is the maximum number of neighbours a peer may have in its routing table? [1 mark]

(c) Which keys may peer 500 index? [1 mark]

(d) Which, if any, of the other four peers (443, 444, 500 680) does peer 205 have a route to? Explain your answer. [3 marks]

(e) If a new peer with ID 300 joins the system, explain what needs to happen. [2 marks]

(f) Chord uses a specific algorithm for determining which neighbour to maintain routes to. An alternative would be for each peer to maintain routes to the next closest neighbour. Explain both an advantage and disadvantage of the Chord approach, compared to this alternative. [2 marks]

Reference Material

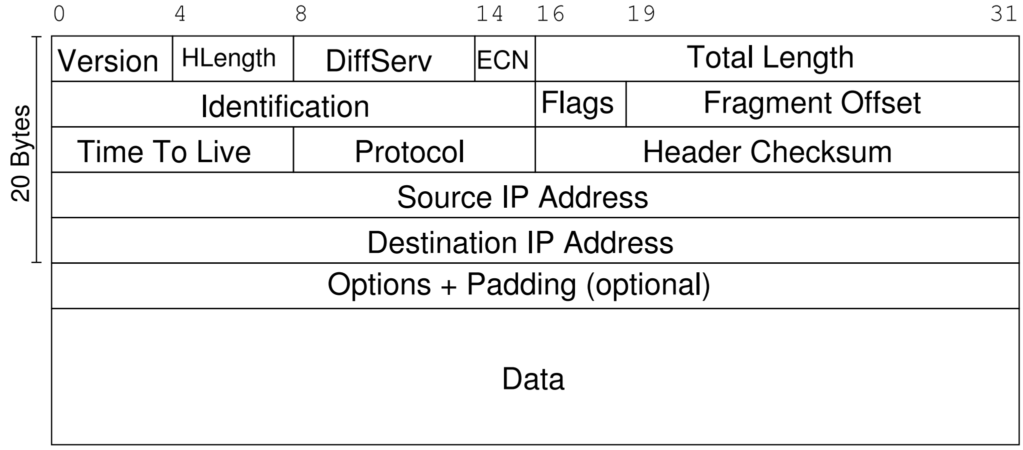

Figure 1: IP Datagram Format. Flags: Reserved, Don't Fragment, More Fragments

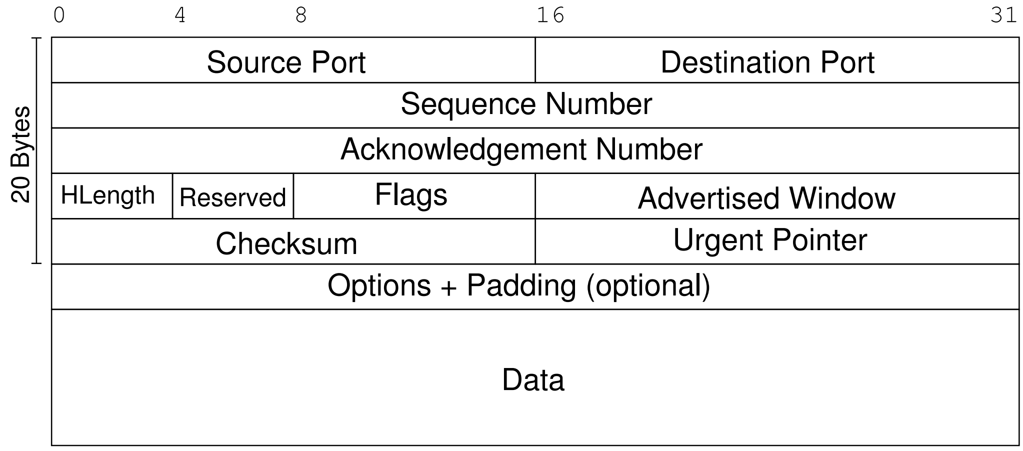

Figure 2: TCP Segment Format. Flags: CWR, ECE, URG, ACK, PSH, RST, SYN, FIN

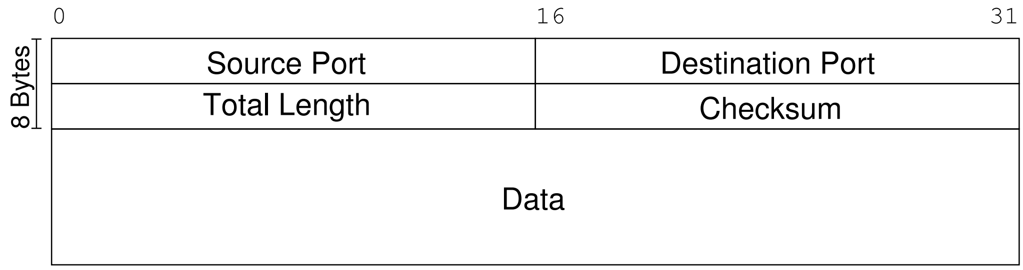

Figure 3: UDP Datagram Format

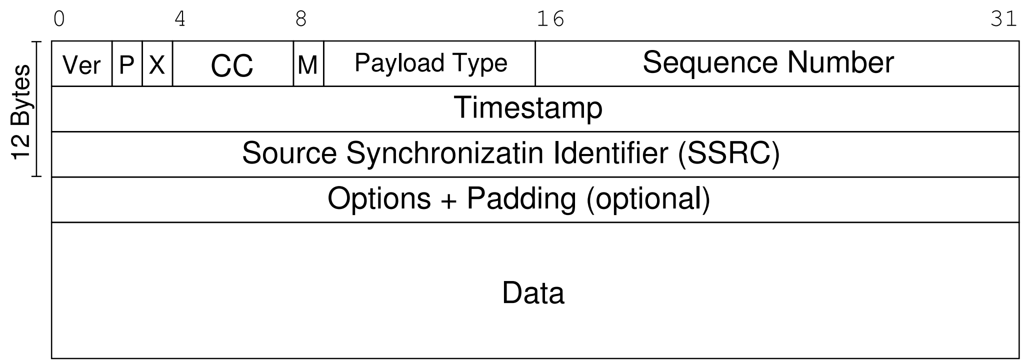

Figure 4: RTP Packet Format. P: Padding; X: Extension; CC: CSRC count; M: Marker