

Name ID Section Seat No

Sirindhorn International Institute of Technology Thammasat University

Final Exam: Semester 2, 2012

Course Title: ITS413 Internet Technologies and Applications

Instructor: Steven Gordon

Date/Time: Thursday 28 February 2013; 13:30–16:30

Instructions:

- This examination paper has 16 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.
- Turn off all communication devices (mobile phone, etc.) and leave them at the front of the examination room.
- 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 examination paper is not allowed to be taken out of the examination room. A violation may result in score deduction.
- Reference material at the end of the exam may be used.

Question 1 [15 marks]

The following algorithms/events change the value of the Congestion Window ($cwnd$) in TCP congestion control:

- Additive Increase
- Slow Start
- Loss detected by Timeout
- Loss detected by Duplicated ACKs

(a) The following equations describe how the new Congestion Window is calculated. Match the correct algorithm/event above to the equation below by writing the name of the algorithm/event: [4 marks]

i. $cwnd_{new} = \text{minimum}(ssthresh, cwnd_{old})$ _____

ii. $cwnd_{new} = MSS$ _____

iii. $cwnd_{new} = cwnd_{old} + \frac{MSS * MSS}{cwnd_{old}}$ _____

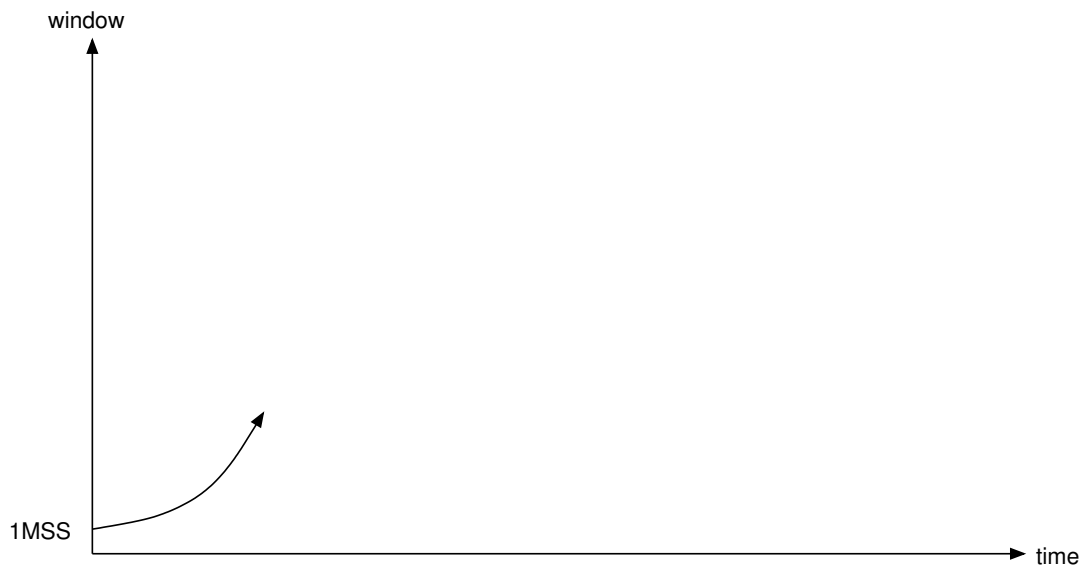
iv. $cwnd_{new} = cwnd_{old} + MSS$ _____

(b) In addition to the congestion window, TCP also has an advertised window. Explain the purpose of the two windows. [2 marks]

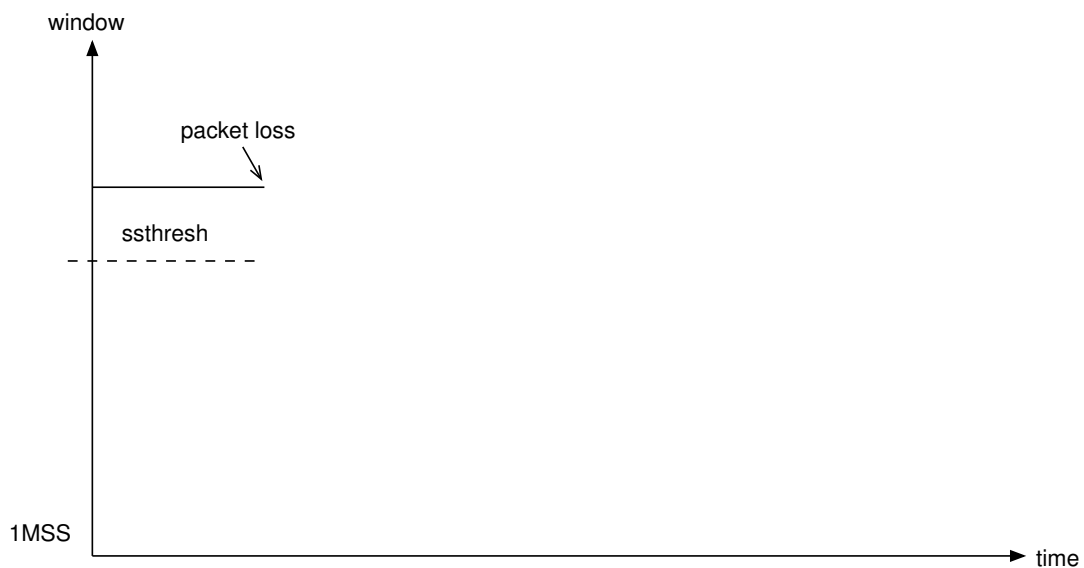
(c) How does the TCP source calculate the window from the congestion and advertised windows? [1 mark]

The following cases require you to draw a plot of the value of TCP source window over time. The plot does not have to be exact or to scale, but should clearly show the different phases (additive increase, slow start, responses to loss, etc.) and how they differ. That is, the shape of the plot should be clear. You should also show the slow start threshold on your plot. As a guide, the initial values of the window are given. You should assume the time on the horizontal axis is a long period, meaning the slow start threshold should be reached within that time.

- (d) Buffer space at receiver is infinite. No packet loss. [3 marks]



- (e) Buffer space at receiver is finite and greater than slow start threshold. Loss due to timeout occurs. Note that the diagram below shows the current value of the window when the loss occurs as a solid line. [3 marks]



- (f) Explain why TCP source responds differently when there is a loss due to a timeout compared to when there is a loss due to a 3rd duplicate ACK. [2 marks]

Question 2 [8 marks]

Consider the network from Client to Server, via three Routers, as shown below. Link data rates and one-way delays are the same in each direction. Ignore other delays.

C --- link1 --- R1 --- link2 --- R2 --- link3 --- R3 --- link4 --- S

- Link1: 15Mb/s capacity; 2ms one way delay
 - Link2: 20Mb/s capacity; 5ms one way delay
 - Link3: 10Mb/s capacity; 8ms one way delay
 - Link4: 15Mb/s capacity; 5ms one way delay
- (a) What is the value of the RTT? [1 mark]
- (b) If UDP is used to send data from client to server, what is the maximum throughput that can be achieved? (ignoring header overheads) [1 mark]
- (c) What is the Bandwidth Delay Product of the path? [2 marks]
- (d) How big should the server buffer be to achieve maximum TCP throughput? [2 marks]
- (e) What is the maximum TCP throughput if the server buffer is 40,000 Bytes? [2 marks]

Question 3 [11 marks]

Consider a company using VoIP for all internal phone calls. All computers are connected via switched 100Mb/s Fast Ethernet. The VoIP application uses the G.711 codec. The G.711 voice codec takes 100 samples of voice per second, with each sample 80 Bytes. A VoIP application using G.711 and RTP sends 2 voice samples per RTP packet (that is, 160 Byte payload).

- (a) At what rate does the G.711 voice codec generate voice data? [2 marks]
- (b) How many packets per second does the VoIP application send? [2 marks]
- (c) Draw a protocol stack for a computer using the VoIP application. Consider all the protocols used by the VoIP application. [2 marks]

- (d) What is the bandwidth (or throughput) consumed on the company LAN to support each two-way voice call? [3 marks]
- (e) The G.729 codec has a sample size of 10 Bytes (instead of 80 Bytes in G.711—sampling rate is the same as G.711). Explain an advantage and disadvantage of using G.729 in the company network (compared to using G.711). [2 marks]

Question 4 [8 marks]

Explain the following delivery mechanisms and give an example application (or application protocol) that is commonly used by the mechanism.

(a) Unicast [2 marks]

(b) Broadcast [2 marks]

(c) Multicast [2 marks]

(d) Anycast [2 marks]

Question 5 [11 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. [2 marks]

- (e) Explain a disadvantage that FastTrack has compared to Gnutella. [2 marks]

Question 6 [9 marks]

Consider an IPTV network operated by a single network operator.

- (a) 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]

(b) Explain the difference between ADSL2+, FTTH and FTTN, as options for the service provider access network in an IPTV network. State the transmission media they use and an advantage the technology has (compared to the other two). [3 marks]

i. ADSL2+

ii. FTTH

iii. FTTN

(c) Explain why and how normal TV and video-on-demand are treated differently when delivered across a service providers core network. [2 marks]

(e) Where can a peer find the IP addresses of other peers in a swarm to obtain torrent *T*? [1 mark]

(f) Assume a peer may maintain 30 TCP connections to remote peers at a time. Some of the remote peers may be choked while the others are unchoked. Explain what this means. [2 marks]

(g) Which of the following P2P systems is closest to BitTorrent: Napster, Gnutella, FastTrack? Explain your answer. [2 marks]

Question 8 [8 marks]

- (a) Name two TCP congestion control algorithms available for the Linux kernel? [2 marks]

Explain what the following command(s) do when run in Linux. Be specific with your answer, mentioning the relevant values. Assume the computer(s) which the commands are run on are setup correctly. [1.5 marks each]

(b) `iperf -c 1.2.3.4 -u -b 10M -t 50`

(c) `iperf -s -w 20000`

(d) `sudo tc qdisc add dev eth0 root netem delay 100ms 5ms`

(e) `sudo iptables -A INPUT -m statistic --mode random --probability 0.03 -j DROP`

Reference Material

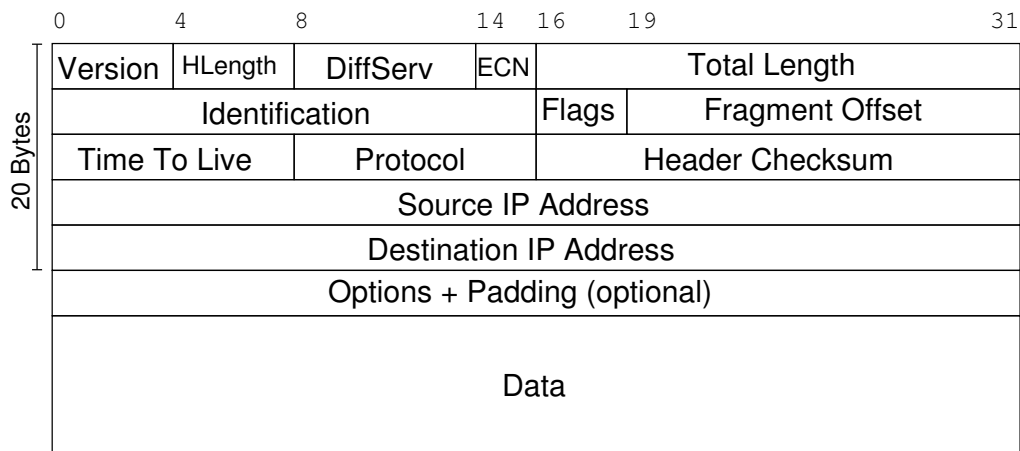


Figure 1: IP Datagram Format. Flags: Reserved, Don't Fragment, More Fragments

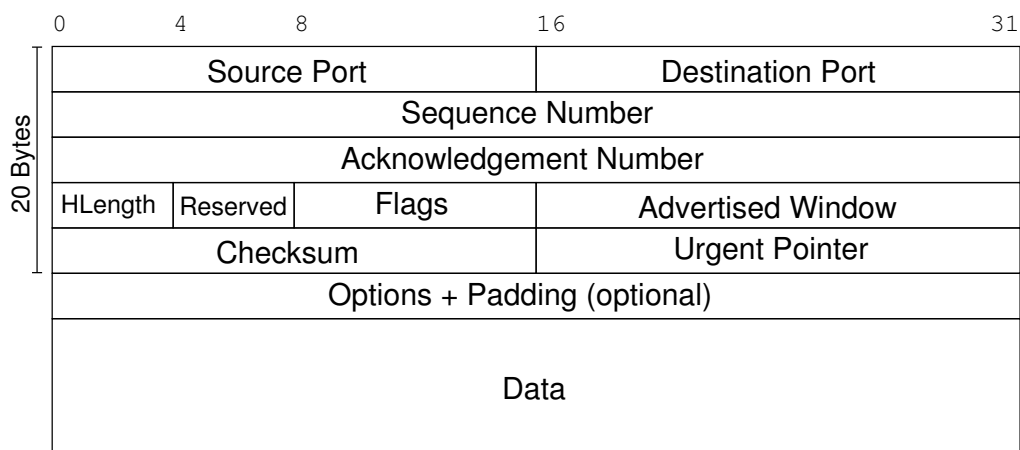


Figure 2: TCP Segment Format. Flags: CWR, ECE, URG, ACK, PSH, RST, SYN, FIN

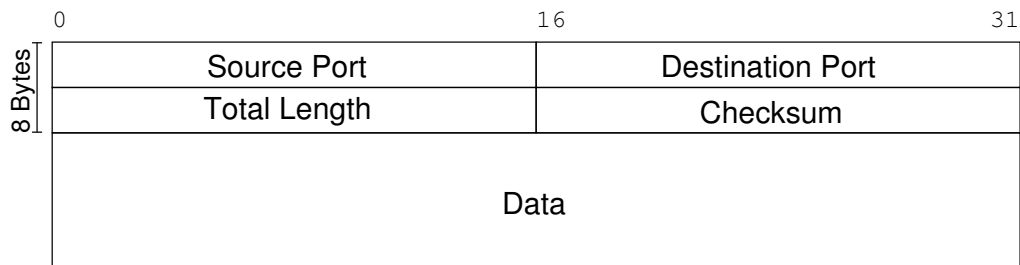


Figure 3: UDP Datagram Format

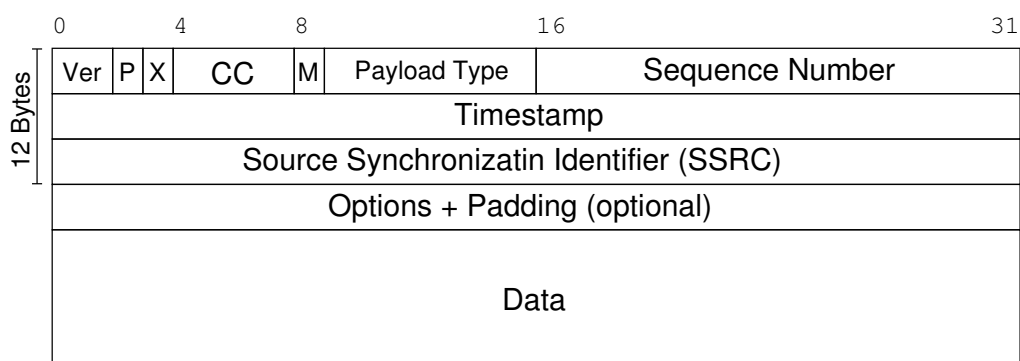


Figure 4: RTP Packet Format. P: Padding; X: Extension; CC: CSRC count; M: Marker