University of Glasgow

Friday, 3 May 2013
9.30 am - 11.00 am
(1 hour 30 minutes)

DEGREES OF MSci, MEng, BEng, BSc, MA and MA (Social Sciences)

COMPUTING SCIENCE 3T: NETWORKED SYSTEMS 3

Answer all 3 questions

This examination paper is worth a total of 60 marks.

You must not leave the examination room within the first half hour or the last fifteen minutes of the examination.

INSTRUCTIONS TO INVIGILATORS: Please collect all exam question papers and return to the School together with the exam answer scripts

1. **(a)** In abstract terms, a network operates by sending some form of time-varying signal over a communications channel, to convey information from sender to receiver(s). We can visualise the information content of that signal by using Fourier analysis to derive a frequency domain representation of the signal. With the aid of a diagram, sketch how the frequency domain view of a complex, information rich, signal might differ from the frequency domain view of a simple signal carrying relatively little information. [4]

   (b) Nyquist showed that the maximum data rate, in bits per second, that can be supported by a perfect digital communications channel, is $R_{\max} = 2H \log_2 V$. Explain what is meant by a "perfect" channel, and outline what the terms H and V stand for in this expression. Discuss what is meant by the *baud rate* of the channel, and how it relates to $R_{\max}$. [6]

   (c) A real-world digital communications channel may be subject to various impairments that can cause bit errors. It is possible to detect and correct single bit errors in a word of data by using a Hamming code to generate check bits that can be sent along with the data as a form of forward error correction. For a signal comprising n data bits and k check bits, so that a total of $n+k$ bits are transmitted, describe what bits will be check bits, and what bits will be data, and explain how the check bits are calculated. Show how the 7-bit ($n = 7$) binary data item 1001000 is turned into an 11-bit ($n+k = 11$) binary value when protected by a Hamming code, highlighting what bits in the resulting value are check bits. [10]

2. **(a)** The two most widely used transport protocols in the Internet are TCP and UDP. Describe the service models provided by these two protocols, being sure to highlight their main differences. [8]

(b) You are implementing an interactive video conferencing application, to run over an IP-based network. You can implement this application using either TCP or UDP at the transport layer. Which of these two transport layer protocols would you choose? Discuss the advantages and disadvantages of the two approaches, and outline any cases where the other transport layer protocol would be more appropriate. [12]

3. **(a)** The Internet Protocol provides a connectionless, best effort, packet delivery service at the network layer of the protocol stack. Explain what is meant by *best effort* in this context, and discuss how this aspect of the delivery service impacts the transport layer. [4]

(b) One of the key roles of the network layer is *inter-networking* between different link layer technologies, to allow local area networks to be combined to form a seamless wide area network. Each link layer technology may have a different maximum transmission unit (MTU), and the network layer is responsible for handling this mismatch. Describe how IPv4 and IPv6 handle inter-networking between links with different MTUs. Discuss why this behaviour was changed when IPv6 was designed. [10]

(c) Network Address Translator (NAT) devices are widely used in the Internet. Describe the purpose of a NAT, and give four reasons why NAT devices are used in the Internet. [6]

END OF QUESTION PAPER