University College London

Department of Computer Science

M.Sc. in Data Communication Networks and Distributed Systems, 2000 D59 resit and

VIVE MSc Multimedia Systems

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EXAMINATION QUESTIONS

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QUESTION 1

a) Explain, with the help of diagrams, the basic components of a lossy compression scheme for motion video data in the H.261 video-telephony standard.

[12 marks]

b) Outline the trade-offs between quality, bandwidth, delay and loss which are behind the choice of inter-frame and intra-frame compression for typical schemes in use today (e.g. H.261, or MPEG II).

[13 marks]

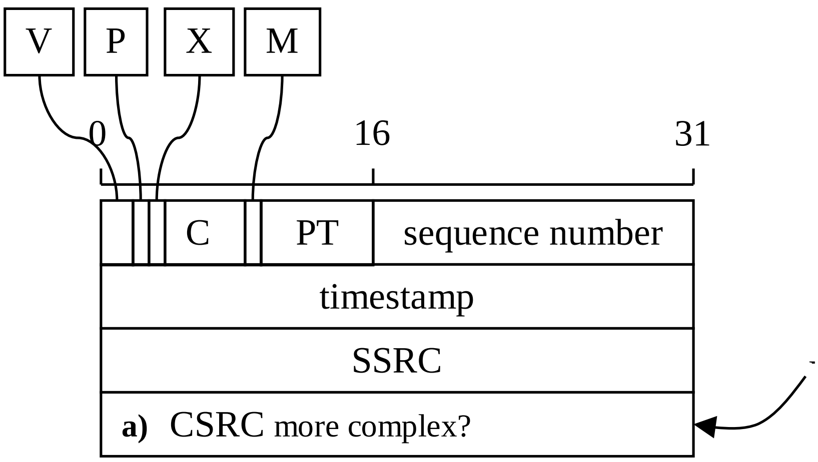
QUESTION 2

a) The World Wide Web is often enhanced by the use of proxy-cacheing systems. Outline the operation of these components.

[6 marks]

b) What are the characteristics of streamed media that make the use of caches potentially In the Internet, RTP (illustrated below) is used to carry audio, video and other time-structured continuous media over the UDP/IP service. Explain the use of the Synchronisation Source and timestamp and other relevant fields in reconstructing the best playout time at a receiver for single and multiple streams.

[10 marks]



QUESTION 3

You are a climber/reporter in the latest Everest expedition. A national TV station is sponsoring your place in the expedition, and in return you must report live every day using a multimedia satellite phone from camps along the route. As the team climbs, you are also contracted to record video sequences of the views, (such as the Khumbu ice fall), which can be uploaded later. Before the expedition, you need to specify just the remote systems the TV station needs to purchase, as you have your own special light-weight IP-based equipment to use on the mountain.

a) Draw a high-level host application software block diagram of the multimedia satellite phone (draw the diagram just for the audio) and the remote video file receiver.

[10 marks]

b) Calculate reasonably accurately the approximate bandwidth required from the satellite connection for the satellite phone (audio and video). State any assumptions you make.

[15 marks]

Assume:

Satellite phone audio: ADPCM codec (sampling frequency = 8kHz), which produces 4 bits per codeword. Assume that 160 bytes of codewords are sent in their own RTP/UDP/IP packet.

Satellite phone video: H.261 frame intra-coded video codec (with no inter-frame coding), which operates at 2 frames per second, and codes Y (352 x 288 pixels),U (176 x 144 pixels) and V (176 x 144 pixels) frames. Each pixel is coded using 8 bits per pixel. H.261 coding gives 20:1 compression. Assume that a maximum of 1024 bytes can be sent in each RTP/UDP/IP packet.

RTP/UDP/IP header: 40 bytes

QUESTION 4

a) Explain the steps in the development of the **Reverse Path Multicast (RPM)** mechanism starting from simple flooding. Your answer should include a clear explanation of each stage of refinement in the process from the simple flooding, and what the effects of the refinement are for the operation and behaviour multicast delivery. Also include a brief analysis of options for how the multicast delivery tree can be grown.

[12 marks]

b) Explain how the MBONE provides IP multicast capability across the Internet. Your answer should include a description of how MBONE connectivity is configured, any problems that may arise from such a configuration and how the range (scope) of multicast transmission is controlled.

[13 marks]

QUESTION 5

a) What are the advantages of a datagram network protocol (such as the Internet Protocol) compared to a purely circuit-switched network for general data applications?

Explain how some of these advantageous characteristics that you list for general data applications pose problems when trying to offer QoS (quality of service) capability to applications over datagram networks.

[6 marks]

b) Describe how the use of RSVP with the INTSERV (Integrated Services) model tries to solve these problems for IP-based networks. (Your answer should include a description of the purpose and the operation of RSVP.)

[9 marks]

c) List the drawbacks of the RSVP/INTSERV approach as described in your answer to part b) above. Describe briefly the principals and general features of an alternative QoS mechanism, DIFFSERV (Differentiated Services), which may help to overcome some of these drawbacks. (Your answer should consider the impact on the network and the applications when comparing the DIFFSERV approach to that of the RSVP/INTSERV approach.)

[10 marks]

QUESTION 6

a) Explain why TCP is unsuitable for real-time communication, making reference to the functions and service that TCP provides, as well as the mechanisms it uses during operation. Illustrate you answer by using a real-time voice flow as an example. Your answer should include a description of the relevant mechanisms that are used in the operation of TCP.

[15 marks]

b) Discuss why it might be useful for real-time applications to develop a flow-rate adaptation mechanism which is "TCP-like". Explain what it means to have "TCP-like" behaviour in this context and highlight the difficulties that could arise in using "TCP-like" behaviour for real-time flows.

[10 marks]

[10 marks]

END OF PAPER