GZ05 Coursework, March 2010.

1. The paper "A Survey of Packet-Loss Recovery Techniques for Streaming Audio" by Perkins, Hodson and Hardman (available from http://nrg.cs.ucl.ac.uk/mjh/AudioRecovery.pdf) gives a fairly complete discussion of how audio streams can be protected against packet loss.

   Some of the techniques may be applied to interactive video, some need modification to apply for video, and some don't apply at all. Choose two techniques that apply, two that don't apply, and one that can be modified, and discuss why you believe the choices apply or don't apply, or what modifications are needed.

2. You have just been employed as the chief software engineer for a company that aims to produce very high-quality Internet telephony software that uses an MP3 codec over a TFRC-based congestion-controlled RTP session. Draw a block diagram of the entire (one-way) audio pipeline from audio capture at the sender, through network transmission over the RTP session, to audio playout at the receiver. Indicate the feedback loops present in the pipeline, and what they control. Indicate also where the principle sources of delay arise, giving an explanation of why the delay occurs. In case it helps you, an MP3 audio frame compresses 1152 audio samples.

Your answer should not exceed 2000 words (approximately 4 pages of text) plus diagrams.

If it helps you to quote material from online (or other sources) this is OK, so long as you make it clear which parts of your answer are quotes, and so long as you cite all your sources. Remember that a fair amount of information available online is incorrect or confused, so use your judgment and justify your answers.