Internet Telephony (Midterm Exam)

April 30, 2003

1. Why is voice carried over UDP, not TCP? (5%) Describe the functions provided by RTP and RTCP. (10%)

2. Design the "Follow Me (隨身碼)" service in the H.323 system (Assume that the user is registered at different GKs). Please include
   (1) The message flow for Registration Procedure (10%)
   (2) The message flow for Call Delivery Procedure (10%)

3. SIP Extensions:
   (1) Describe the functions of SIP INFO method. (10%)
   (2) Why should the reliability of provisional responses be supported in SIP? (5%) How to achieve it? (5%)

4. You are asked to design a SIP-to-PSTN gateway based on MGCP. Draw the network architecture (10%) and describe the message flow for a SIP terminal calling a telephone on PSTN. (10%)

5. Describe the target of SIGTRAN in IETF. (5%) Also explain why SCTP is more suitable for SIGTRAN than TCP or UDP. (10%)

6. Call-setup time is one of the key measures for VoIP speech quality. Please describe two approaches in H.323 to reduce the call-setup time. (10%)