S-38.3152 Networked Multimedia Protocols and Services

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Please write readably and answer in English.
There are three classes of questions: (a) expecting (relatively) short answers, (b) expecting more elaborate answers, and (c) a small design task. The questions are marked accordingly.

Questions:

1. [6p, a] Describe three aspects that contribute to the "mouth-to-ear" delay when sending voice over IP.

2. [6p, a] The RTP header contains a time stamp and a sequence number. Why both? Describe their functions and explain (e.g., by example) why one of them would be insufficient.

3. [6p, a] What is the purpose of an RTP packetization format for a codec? When would such a format use a payload header in addition to the RTP header?

4. [6p, b] a) What are the three sections comprising an SDP message (in conjunction with SAP)?
   b) Which of these does the SDP offer/answer model primarily use and how? Why are the others less relevant?

5. [6p, a] Contrast pro-active and re-active media-on-demand schemes. What are their respective advantaged and disadvantages?

6. [6p, b] a) Explain forking in SIP.
   b) Describe two problems that may arise from forking?

7. [6p, a] How does P2PSIP conceptually differ from original SIP? Which functions are the overlay nodes supposed to provide?

8. [6p, b] a) SIP supports personal presence information. Which (three) methods are used and what are their semantics? Sketch one simple example involving all three.
   b) How does a user learn that another user is interested in her presence state?

9. [6p, a] a) What is a self-signed certificate?
   b) Where is this useful? Why?

10. [6p, b] Explain the concept of a GRUU. Sketch/outline an example how a GRUU is obtained and used.

11. [12p,c] When calling the call center of a hotline you sometimes notice the announcement that your call may be monitored. Assume customer C calling in, a hotline assistant A taking the call, and a supervisor S who may want to monitor the call. Assume further that the RTP stream flows end-to-end between A and C without a sniffing device in the path. How can you enable S to initiate "entering" the call: both for actively supporting the assistant (so that S can also speak) and for passively monitoring the call (so that the assistant does not notice the presence). How does S learn about the call? Which components do you use? Which SIP methods and extensions do you use (but do not create *new* extensions)? Sketch a call flow to learn about and listen to (or enter) a call. Don't forget the media stream.
    (Note: there are many possible solutions.)