FINAL EXTRAORDINARY EXAM

NAME:

ACTIVITY 1 (4 PTS)

Please, consider the following communications scenario:

- There are 3 User Agents (A, B and C), an Application Server (AS) that acts as a B2BUA and a Media Server (MS)
- A, B y C represent human users
- AS is not an RTP entity (it does not handle media)
- UA's only support voice with PCMA codec
- Nodes IP addresses are (in order to be referenced in the SDP content) :
    i. AS: <IP-AS>
    ii. A: <IP-A>
    iii. B: <IP-B>
    iv. C: <IP-C>
    v. MS: <IP-MS>

AS implements a sequential search Service:

- When a call from A is received, it uses a MS to play an announcement in connected mode and route the call towards B
- B is alerted and it generates early media. It does not answer and after 10 seconds, the AS initiates a call to C
- C is alerted and it generates early media. Finally, C answers the call, and this is established with A

Please bear in mind that:

- Both B and C generate early media
- AS must avoid that the early media from B and C flows towards A so as not to interfere with the announcement being played by MS
- A, B, C and AS support both 100rel and UPDATE and they always use the 100rel extension to send any provisional response
- MS does not support 100rel nor UPDATE

Please represent the signaling flow indicating the name of SIP messages and the corresponding SDP content (c-line and m-line)

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ACTIVITY 2 (3 PTS)

In an Enterprise private voice network, there is an Asterisk server and two SIP devices with the corresponding extension numbers 100 and 101. The SIP devices are registered into the Asterisk server.

It is requested to build a simple voice menu application according to the following requirements:

a. The application will play a welcome announcement (lab_welcome.wav) and offer the caller to select amongst three options.
b. When the caller selects the first or second option, a specific announcement is played (option1.wav or option2.wav) and then the welcome announcement is played again.
c. Customer input must be allowed whilst the welcome announcement is being played and for 4 seconds after the announcement finishes
d. The third option informs the user that she can dial an extension number via DTMF to be connected with the corresponding extension. The dial plan needs to allow connecting to extensions 100 and 101. If extensions 100 or 101 do not take the call, then it is required to playback a new announcement: not-available.wav and hang up the call.

Please provide an example of the extensions.conf file used to create the voice menu.

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ACTIVITY 3 (3 PTS)

Please select the correct responses:

1. In the RTP header, the CSRC header:
    a. Conveys the value of the inter-arrival jitter
    b. Must always be present
    c. It is only present if mixers are used
    d. Its value is expressed in seconds
2. In a RTP audio transmission, if the media clock runs at the sampling frequency , the sampling rate is 8KHz and the frame size is 20 ms, which is the difference in the value of two consecutive timestamps
    a. None, they are the same
    b. 160
    c. 320
    d. 90KHz
3. If an SDP offer that contains a single m-line for audio also includes a value attribute equal to 'sendrecv': which are the possible values of the direction attribute in the SDP answer:
    a. sendrecv or inactive
    b. sendrecv, sendonly, recvonly or inactive
    c. sendonly
    d. recvonly or inactive
4. The Next Generation Network (NGN) as defined by ITU:
    a. It does not rely on IP packet-based transfer
    b. It does not provide broadband capabilities with e2e QoS
    c. It provides separation of control functions among bearer capabilities, call/session and application/service
5. In the ITU-T Release 2 NGN architecture
    a. The Transport stratum does not contain any control functions
    b. User Profiles only exist in the Service Stratum
    c. Content Delivery Functions are included in the Service Stratum
6. The Evolved Packet Core (EPC)
    a. It defines a Radio Access network for 4G
    b. It represents a converged core network for both 3GPP and non-3GPP accesses
    c. It is not based on a flat IP architecture
7. A Media Server (MS)
    a. It is an element that is used to interconnect several SIP networks
    b. It is an element that provides media services
    c. It is a SIP element that stores the AoR-CA relationship

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8. Triple-play
    a. It is a technology for online 3D multimedia games
    b. It is a MKT term that refers to the provision of mobile service over 2G, 3G and 4G
    c. It is a MKT term that refers to the combination of broadband Internet access, television and fixed telephony
9. SIP dialog Id
    a. It is an identification of the payload type contained in RTP packets
    b. It is identified by the o-line in the SDP content of SIP messages
    c. It is identified by the combination of CallID, From tag and To tag
10. In an SDP offer-answer exchange
    a. The number of media lines in the answer much be equal or greater than the number of media lines in the offer
    b. The number of media lines in the answer much be equal than the number of media lines in the offer
    c. The number of media lines in the answer much be lower than the number of media lines in the offer
11. The SIP configuration for an Asterisk server is defined in the file:
    a. asterisk.conf
    b. extensions.conf
    c. sip.conf
12. RTP and RTCP
    a. RTCP is a protocol used to carry the media samples over an IP network
    b. An RTP mixer does not change the SSRC of the RTP packets
    c. An RTP transport translator does not modify the RTP stream itself since it is only concerned with transport parameters