CCNA Voice Practice Tests – Describe VoIP Components and Technologies (1-5)

Topic 3 - Describe VoIP Components and Technologies Question 1 A SIP Trunk is a logical connection between an IP PBX and a Service Provider's application servers that allows voice over IP traffic to be exchanged between the two. When a call is placed from an internal phone to an external number, the PBX sends the necessary information to the SIP Trunk provider who establishes the call to the dialed number and acts as an intermediary for the call. All signaling and voice traffic between the PBX and the provider is exchanged using SIP and RTP protocol packets over the IP network. Which two statements about SIP trunk are true? A.A SIP trunk configuration is mandatory for a UC500 device. B.A SIP trunk is needed for internet access C.A SIP trunk is needed only for voice if you are planning on using VoIP through a service provider. D.A SIP trunk is not supported in a keyswitch configuration. Answer: C D Explanation: This question tests the SIP Trunk. A: SIP Trunk is one of the communication modes supported by UC500 device and it is not a mandatory configuration. B: SIP Trunk is used for the establishment of the end to end speech communication signaling, which does not apply to the Internet access. C: SIP Trunk is used for the establishment of the end to end speech communication signaling and it needs service provider to provide the function of SIP Proxy. D?SIP Trunk only support PBX system mode Question 2 You are CCNA VOICE associate in Lead2pass.com. One user from your company wants to use a signaling protocol on the voice gateways that require registration with the Cisco Unified Communications Manager. Which protocol should you suggest to him? A.SIP B.Frame relay C.SRTP D.MGCP Answer: D Explanation: This question tests the application of the communication signaling. There are two types of signaling commonly used to establish a connection: the traditional signaling and IP signaling. The traditional signaling is used for the communication between Gateway and CO and communication between CO and CO. IP signaling is used for the communication between CUCM and its components and communication between all IP voice devices. IP voice signaling can be divided into Client-Server mode and end-to-end mode. In Client-Server mode, all the actions registered in the call agent by UC components are decided by the call agent; In end-to-end mode, UC components only need to maintain communication with the call agent and it has the ability to initiate negotiations and call routing. This question requests voice gateway to register in the call agent (CUCM), so IP signaling of a Client-Server mode should be used. A: SIP: IP signaling of end-to-end mode. B: Frame relay is an access technology of WAN network, not the signaling of voice communication. C: SRTP intended to provide encryption, message authentication and integrity, and replay protection to the RTP data in both unicast and multicast applications, which is not a voice communication signaling. D: MGCP is the IP signaling of Client-Server mode, meeting the requirement of the question. D is correct. The Secure Real-time Transport Protocol (or SRTP) defines a profile of RTP (Real-time Transport Protocol), Question 3 What protocol needs to be enabled on an ATA if a fax machine is connected to the ATA? A.MGCP B.SCCP C.H323 D.SIP Answer: C Explanation: Under the network mode in which fax machine is connected to the ATA analog gateway and communicates with the external network, the fax-relay mechanism must be enabled in order to ensure that analog fax signals can transmit correctly after passing through IP networks. There are two kinds of fax-relay mechanism used by Cisco: one is fax-relay and the other is fax passthrough.ATA gateway only supports the relay mode of fax passthrough and it is necessary to enable H323 protocol. Question 4 You are CCNA VOICE associate in Lead2pass.com. You execute the show ephone command in the Cisco Unified Communications manager as below. What information will you get from the output?

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UCME#show ephone

ephone-1 Mac:0030.94C2.8A44 TCP socket:[2] activeLine:0 REGISTERED in SCCP ver 1
1 and Server in ver 8
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8
IP:10.3.130.10 50374 Telecaster 7960  keepalive 4 max_line 6
button 1: dn 1  number 5001 CH1   IDLE         CH2   IDLE         mwi
button 2: dn 3  number 5010 CH1   IDLE         CH2   IDLE


ephone-2 Mac:0003.E3C4.463C TCP socket:[-1] activeLine:0 DECEASED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:7
IP:10.3.130.12 49939 Telecaster 7960  keepalive 5162 max_line 6
button 1: dn 2  number 5002 CH1   DOWN         CH2   DOWN
button 2: dn 3  number 5010 CH1   IDLE         CH2   IDLE         shared
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A.There is only one IP phone. B.There are two registered IP phones. Shared number 5010 on line 2. Message waiting on shared line. C.There are two IP phones. Phone 2 is unregistered. Shared number 5010 on line 2. Message waiting on shared line D.There are two IP phones. Phone 2 is unregistered. Shared number 5010 on line 2. Message waiting on line 1 of phone 1. Answer: D Explanation: This question examines how to uses the show command to view the registration conditions of IP Phone in CCME. We can learn from the show result that Phone1 has registered to CCME via SCCP protocol. The number of line1 is 5001, which is used for mwi (message waiting indicator of CUE), and the number of line2 is 5010. Therefore, the shared number 5010 is on line 2 and the message waiting is on line1 of Phone1. As Phone2 is in a deceased state, Phone 2 failed to be registered.

Choose D. Question 5 What protocol needs to be enabled on an ATA if an analog telephone is connected to the ATA? A.MGCP B.SCCP   C.H323 D.SIP Answer: B Explanation: An analog telephony adapter (ATA) is a device used to connect one or more standard analog telephones to a digital and/or non-standard telephone system such as a Voice over IP based network. The ATA communicates with the server using a protocol such as H.323, SIP, MGCP, SCCP or IAX, and encodes and decodes the voice signal using a voice codec such as G.711, G.729, GSM, iLBC or others. However, if an analog telephone is connected to the ATA, Skinny Call Control Protocol (SCCP) must be enabled on an ATA.