Vendor: Cisco

Exam Code: 300-815

Exam Name: Implementing Cisco Advanced Call Control On-Premises (CLACC)

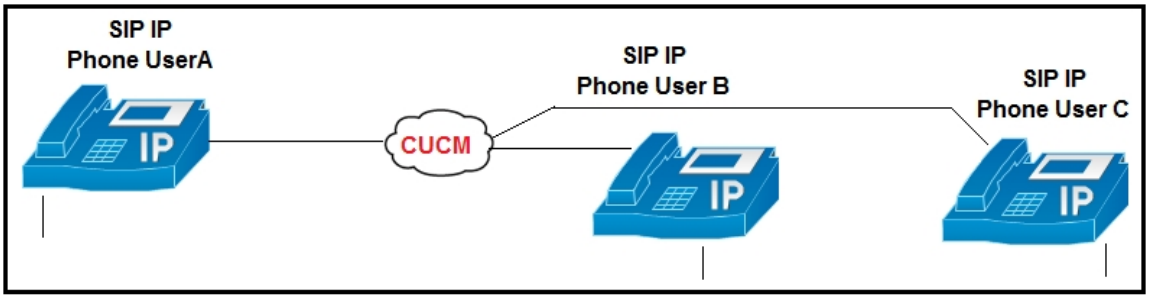
Version: DEMO

QUESTION 1
What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?

A. A partition can have multiple time schedules assigned. A time schedule contains one or more time periods.
B. A partition can have one time schedule assigned. A time schedule contains one or more time periods.
C. A partition can have multiple time schedules assigned. A time schedule contains only one time period.
D. A partition can have one time schedule assigned. A time schedule contains only one time period.

Answer: B
Explanation:
As mentioned earlier, partitions allow you to assign a time schedule to a partition. A time schedule contains a list of time periods.
The time schedule can include more than one time period. For example, you might create a time period for each holiday and then create a time schedule named Holidays that includes all the holiday time periods.

QUESTION 2
Refer to the exhibit. In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. Which two scenarios are correct? (Choose two.)



A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
B. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.
D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.
E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Answer: AD
Explanation:
User hold: With a user hold, a user presses the Hold button on a phone to explicitly place the caller on hold. If Phone A and Phone B are having a conversation, and the user of Phone B presses the Hold button, Phone B's user hold source is streamed to Phone A.

Network hold: A network hold occurs when a call is placed on hold as part of the processing of a supplementary service, such as park, transfer, or conference. If Phone B presses the Transfer button to transfer a call, Phone A still gets placed on hold but hears Phone B's network audio source while the rest of the transfer operations complete.

QUESTION 3
The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call.
You gather the H.225 and H.245 messages for one of the one-way audio calls.
Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).

A. H.245 Terminal Capability Set
B. H.245 Open Logical Channel
C. H.225 Connect
D. H.245 Open Logical Channel Ack

Answer: D
Explanation:
H245 Open Logical Channel contains only RTCP port, H245 Open Logical Channel ACk will have both RTP and RTP Port.
https://community.cisco.com/t5/collaboration-voice-and-video/h323-call-flow/ta-p/3160014

QUESTION 4
End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

A. Contact: header of the 200 OK response
B. Allow: header if the 200 OK response
C. o= line of SDP content
D. c= line of SDP content

Answer: D
Explanation:
c= attributes contained the connection media information. It contains the IP address that will be used for RTP.
https://tools.ietf.org/html/rfc4566#section-5.7

QUESTION 5
Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is one-way audio or no audio. You determine that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally.
What are two possible solutions? (Choose two.)

A. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 16384-32767
B. Ask the firewall administrator to change the ports to TCP.

C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.
D. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 20000-22000.
E. Go to System Parameters in Cisco Unified Communications Manager and change the range of media ports to 20000-22000.

Answer: CD
Explanation:
In this case, one solution is to change the ports on the SIP profile of the phones to match the ports on the firewall. The other solution is to ask the firewall admin to change his ports to match the ports in UCM.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/port/9_1_1/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91_chapter_01.html

QUESTION 6
Which description of RTP timestamps or sequence numbers is true?

A. The sequence number is used to detect losses.
B. Timestamps increase by the time "carrying" by a packet.
C. Sequence numbers increase by four for each RTP packet transmitted.
D. The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation).

Answer: D
Explanation:
The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation). The sequence number is mainly used to detect losses. Sequence numbers increase by one for each RTP packet transmitted, timestamps increase by the time "covered" by a packet. For video formats where a video frame is split across several RTP packets, several packets may have the same timestamp. In some cases such as carrying DTMF (touch tone) data (RFC 2833), RTP timestamps may not be monotonic.

QUESTION 7
A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

A. three-way conference
B. secure SIP lines
C. T.38 fax relay
D. transcoding
E. SIP trunk

Answer: AC
Explanation:
The following supplementary services are supported as part of IPv6 in Unified SRST IP Phones:
Hold/Resume
Call Forward
Call Transfer
Three-way Conference (with BIB conferencing only)

Line to T1/E1 Trunk and Trunk to Line with Supplementary Service Features
Fax to and from PSTN (IPv4 ATA to ISDN T1/E1) for both T.38 Fax Relay and Fax Passthrough

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_sip_isr4000.html

QUESTION 8
Which action is correct with respect to toll fraud prevention configuration in the Cisco Unified Communications Manager Express?

A. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.
B. Configure IP Address Trusted Authentication for Incoming VoIP Calls.
C. Configure the command no ip address trusted authenticate under "voice service voip".
D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Answer: B
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_E435095B0E2845C78D60332753288337
* Direct-inward-dial isdn is not supported for incoming ISDN overlap dialing call.

QUESTION 9
You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?

A. configuration for a single SIP phone
B. configuration items common for all SIP phones
C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)
D. configuration for SIP registrar service

Answer: A
Explanation:
To enter voice register pool configuration mode and create a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST, use the voice register pool command in global configuration mode.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_v1ht.html#wp2339729225

QUESTION 10
Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

A. allow-connections sip to sip
B. voice service voip
C. voice register global
D. voice register dn

Answer: B
Explanation:

You have to enable the registrar service before you can provide the address in voice register global

voice service voip
allow-connections sip to sip

!--- Enable SIP to SIP calls.

sip
registrar server expires max 1200 min 300

!--- Enable Cisco IOS SIP registrar.

QUESTION 11
For s SIP to SIP call flow, when does Cisco Unified Border Element require transcoding resources for DTMF?

A. interworking between an OOB method and RFC2833 for flow-around calls
B. interworking between h245-signal and rtp-nte
C. interworking between an OOB method and RFC2833 for flow-through calls
D. interworking between h245-alpha numeric and sip-kpml

Answer: A
Explanation:
CUBE requires transcoding resources registered locally in these scenarios
...
Interworking between an OOB method and RFC2833 for flow-around calls

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html

QUESTION 12
An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

A. Incoming Port in SIP Information section of the SIP Trunk configuration.
B. Incoming Port in Security Information of the SIP Profile configuration.
C. Destination Port in SIP Information section of the SIP Trunk configuration
D. Incoming Port in SIP Trunk Security Profile configuration
E. Destination Port in SIP Trunk Security Profile configuration

Answer: CD
Explanation:
Incoming Port of a SIP Trunk Security Profile is Mandatory.
Destination Port is configured to the right of the Destination Address in the SIP Information portion of the SIP Trunk Configuration.

QUESTION 13
An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60

seconds?

A. R2(config-ephone-dn)#park reservation-group 60
B. R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002
C. R2(config-ephone-dn)#park reservation-group 1
D. R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Answer: D
Explanation:
The timeout and limit keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (park-slot timeout 10 limit 5) will park calls for approximately 50 seconds.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmepark.html#concept_6DE6A02D258F4BA48D2F94A1053563AF

QUESTION 14
A user's phone is already configured for Single Number Reach, and the user wants a feature to move an active call from a mobile phone to a desk phone and vice-versa. As an administrator, which additional configuration should be made to fulfill the user's request?

A. Use Dialed Number Analyzer to determine if the user extension can dial the mobile phone.
B. Add the mobility key to the softkey template that the desk phone is using.
C. Check to make sure that the Resume softkey option appears on the desk phone.
D. Confirm that the desk phone is subscribed to Cisco Extension Mobility.

Answer: B
Explanation:
Step 11 Configure a Mobility softkey for the phone user that uses Cisco Unified Mobility.
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0110111.html

QUESTION 15
Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

A. TokenID
B. username and secret key
C. TLS certificates
D. passwords
E. FQDN of the servers defined in DNS

Answer: CD
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_011001.pdf

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