



Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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Found in 1998

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NEW QUESTION 1

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: B

NEW QUESTION 2

Which two extended capabilities must be configured on dial peers for fast start-to-early media scenarios (H.323 to SIP interworking)? (Choose two.)

- A. DTMF
- B. BFCP
- C. VIDEO
- D. FAX
- E. AUDIO

Answer: AB

NEW QUESTION 3

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 oneway 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: AC

NEW QUESTION 4

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

NEW QUESTION 5

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

NEW QUESTION 6

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

NEW QUESTION 7

Where is the dtmf-relay command configured on Cisco Unified Border Element?

- A. in the voice-class VoIP configuration
- B. in the VoIP dial peer
- C. in global SIP configuration
- D. in the VoIP or POTS dial peers

Answer: B

NEW QUESTION 8

Refer to the exhibit. Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222". The administrator already verified that the IP address of the Cisco Unified CM is set up correctly and there are no dial peers configured other than those shown in the exhibit. Which action must the administrator take to fix the issue?

- A. Change the destination-pattern on the outgoing dial peer to match "444333222".
- B. Set up translation-profile on the incoming dial peer to match incoming traffic.
- C. Create specific matching for "222333444" on the incoming dial peer.
- D. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".

Answer: B

NEW QUESTION 9

An engineer must route all SIP calls in the form of <user>@example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

- A. example.com@gateway.corporate.local
- B. *@example.com
- C. gateway.corporate.local
- D. example.com
- E. *.*

Answer: BE

NEW QUESTION 10

Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

- A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.
- B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
- C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
- D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.
- E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Answer: AB

NEW QUESTION 10

After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

- A. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections h323 to h323
- B. Router(config)#dial-peer voice 2 voipRouter(config-dial-peer)#no vad
- C. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections voice-mail mod
- D. Router(config)# voice service voipRouter(conf-voi-serv)#no supplementary-service sip moved-temporarily

Answer: A

NEW QUESTION 14

An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and outbound 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

NEW QUESTION 19

Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does

not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)

- A. Restart the subscriber
- B. Activate the Cisco Extended Functions service.
- C. Activate the Cisco CallManager service.
- D. Activate the Cisco Dialed Number Analyzer service.
- E. Activate the Cisco Dialed Number Analyzer Server service.

Answer: DE

NEW QUESTION 21

In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

- A. Dialed Number Analyzer
- B. Real-Time Monitoring Tool
- C. SDI trace
- D. SDL trace

Answer: A

NEW QUESTION 22

Refer to the exhibit. Within the North American Numbering Plan, gateways located in Ottawa, Canada and marked as “YOW” are assigned to the Calling Party Transformation CSS NANP_CgPTP, which contains partition NANP_calling_xforms. What is the calling-party number and the numbering type if the calling user +1613-555-1234 dials the number?

- A. calling number 613-555-1234 and numbering type “subscriber”
- B. calling number 011-1-613-555-1234 and numbering type “subscriber”
- C. calling number 011613-555-1234 and numbering type “international”
- D. calling number 613-555-1234 and numbering type “national”

Answer: D

NEW QUESTION 27

Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?

- A. System > Location Info
- B. Call Routing > Route/Hunt > Local Route Group Names
- C. System > Device Pool
- D. Call Routing > Emergency Location > Emergency Location (ELIN) Groups

Answer: B

NEW QUESTION 32

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: B

NEW QUESTION 37

Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?

- A. Other Group Pickup
- B. BLF Call Pickup
- C. Group Call Pickup
- D. Directed Call Pickup

Answer: A

NEW QUESTION 40

An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?

- A. ILS Max Number of Learned Objects in Database
- B. ILS Active Learned Object Upper Limit
- C. Global Data Service Parameter Limit
- D. Imported Dial Plan Replication Database Object Lower Limit

Answer: A

NEW QUESTION 45

When configuring hunt groups, where do you add the individual directory numbers that will be part of the group?

- A. route group
- B. line group
- C. hunt list
- D. hunt pilot

Answer: B

NEW QUESTION 46

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

- A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
- B. A unicast music on hold audio source must be configured.
- C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
- D. The maximum number of callers allowed in queue must be 10.
- E. The phone button template must have the Queue Status Softkey configured.

Answer: AC

NEW QUESTION 47

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

- A. in the telephony service configuration
- B. in the queuing configuration
- C. in Cisco Unified CM Enterprise Parameters
- D. in Cisco Unified CM Service Parameters

Answer: B

NEW QUESTION 48

Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?

- A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service
- B. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service
- C. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility
- D. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility

Answer: C

NEW QUESTION 52

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