300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

Pass Cisco 300-815 Exam with 100% Guarantee

QUESTION 1

The Cisco UCM Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

A. translation patterns

B. device pools

C. CTI ports

D. CTI route points

E. IP phones

Correct Answer: CE

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/11_5_1/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115_chapter_01.html#CUCM_TP_A5DA99E0_00

QUESTION 2

An administrator is troubleshooting a one-way audio issue for a call that uses H.323 protocol in slow-start mode. The administrator requests that the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call is provided. The H.225 and H.245 messages for one of the one-way audio calls are gathered and the call flow has not invoked any media resources. Where is the RTP IP and port information for both sides found?

A. H.245 Terminal Capability Set

B. H.245 Open Logical Channel

C. H.225 Connect

D. H.245 Open Logical Channel Ack

Correct Answer: D

Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

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QUESTION 3

<sip: 1155@10.2.2.13>;privacy=off;reason=unconditional;counter=1;screen=no

and from the Cisco CUBE the logs show :

<sip:1155@10.3.3.25>;privacy=off;reason=unconditional;counter=1;screen=no

Refer to the exhibit. Users report that outgoing calls do not work on the new SIP trunk for outgoing calls. The solution consists of a Cisco UCM Cluster linked to a Cisco Unified Border Element where the SIP trunk is terminated. The provider required 10 digits. The logs show a line going toward the Cisco Unified Border Element. Which code snippet must be added to the configuration to meet the requirement?

A. request Invite sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the SIP translation profile configuration

B. request Invite sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the voice translation profile configuration

C. sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the voice translation profile configuration

D. request Invite sip-header Diversion modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the SIP profile configuration

Correct Answer: B

QUESTION 4

An administrator must control the number of calls to a remote specific site to reduce bandwidth constraints. The users on that remote site report bad quality of the calls passing through that WAN link. Which action must the administrator take in Cisco UCM to resolve the issue?

A. Use RSPV.

B. Use Location Bandwidth Manager.

C. Use Expressway deployment.

D. Use Call Allow Controller.

Correct Answer: B

QUESTION 5

A new deployment is using MVA for a specific user on the sales team, but the user is having issues when dialing DTMF. Which DTMF method must be configured in resolve the issue?

A. in-band

B. out-of-band

C. gateway

D. channel

Correct Answer: B