

# 300-815<sup>Q&As</sup>

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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#### **QUESTION 1**

Refer to the exhibit.

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.1.60.105:5060; branch=z9hG4bK721ed5d4

From: "1001" <sip:1001@10.88.247.229>;tag=6cfa89726ac700b569ec133a-7e6cd8aa

To: <sip:2005@10.88.247.229>;tag=47B5F70-438

Date: Fri, 19 Apr 2019 12:13:40 GMT

Call-ID: 6cfa8972-6ac7002b-5af19a5c-0de23108@10.1.60.105

CSeq: 101 INVITE Require: 100rel

RSeq: 3344

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,

INFO, REGISTER

Remote-Party-ID: <sip:2005@10.88.247.229>;party=called;screen=yes;privacy=off

Contact: <sip:2005@10.88.247.229:5060> Server: Cisco-SIPGateway/IOS-16.6.2

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An engineer is troubleshooting an issue with the caller not hearing a PSTN announcement before the SIP call has completed setup. How must the engineer resolve this issue using the reliable provisional response of the SIP?

A. voice service voip sip no rel1xx

B. sip-ua disable-early-media 180

C. voice service voip sip rel1xx require 100rel

D. voice service voip sip send 180 sdp

Correct Answer: C

#### **QUESTION 2**

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.

Correct Answer: B

Reference: https://www.cisco.com/c/en/us/td/docs/voice\_ip\_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.



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### **QUESTION 3**

Which set of commands binds SIP media and signaling to interface GigabitEthernet0/0 when dial peer 1 is chosen for call routing?

- A. dial-peer voice 1 voip voice-class source interface GigabitEthernet0/0
- B. voice service voip bind sip source-interface GigabitEthernet0/0
- C. voice service voip sip bind all source-interface GigabitEthernet0/0
- D. dial-peer voice 1 voip voice-class bind control source-interface GigabitEthernet0/0 voice-class sip bind media source-interface GigabitEthernet0/0

Correct Answer: C

#### **QUESTION 4**

Users are reporting that several inter-site calls are failing, and the message "not enough bandwidth" is showing on the display. Voice traffic between locations goes through corporate WAN, and Call Admission Control is enabled to limit the number of calls between sites. How is the issue solved without increasing bandwidth utilization on the WAN links?

- A. Disable Call Admission Control and let the calls use the amount of bandwidth they require.
- B. Configure AAR to reroute calls that are denied by Call Admission Control through the PSTN.
- C. Reroute all calls through the PSTN and avoid using WAN.
- D. Configure Call Queuing so that the user waits until there is bandwidth available.

Correct Answer: B

#### **QUESTION 5**

An administrator is configuring Meet-me conferencing in a Cisco UCM deployment and has created the Meet-me number and ensured that it is in a partition accessible by all devices. Which two additional steps must the administrator perform? (Choose two.)

- A. Ensure that conferencing-initiating devices are using a media resource group list that contains at least one Cisco UCM conference bridge.
- B. Disable Early Media on the SIP profile of all devices that will use Meet-me conferencing.
- C. Enable Meet-me conferencing in enterprise parameters.
- D. Ensure that all devices have G.729 enabled.



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E. Update the softkey template on all phones to ensure that they contain the Meet-me softkey.

Correct Answer: AE

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