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**Vendor:**Cisco

**Exam Code:**300-815

**Exam Name:**Implementing Cisco Advanced Call  
Control and Mobility Services (CLACCM)

**Version:**Demo

### QUESTION 1

Cisco Extension Mobility does not show up when the services button is pressed. What is the cause of the issue?

- A. The URL configured for Cisco Extension Mobility is not correct.
- B. Cisco Extension Mobility Service is not running.
- C. The phone is not subscribed to Cisco Extension Mobility Service.
- D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

Correct Answer: C

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### QUESTION 2

The administrator sees the voice register pool 1 command in your Cisco UCME 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 UCM)
- D. configuration for SIP registrar service

Correct Answer: A

According to the the CME command reference, the answer should be 'A':

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#wp23397292](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_v1ht.html#wp23397292)  
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### QUESTION 3

Refer to the exhibit.

Building A	Building B
<b>Results Summary</b> <ul style="list-style-type: none"> <li>▶ Calling Party Information <ul style="list-style-type: none"> <li>● Dialed Digits = 9195552388</li> <li>● Match Result = RouteThisPattern</li> </ul> </li> <li>▶ Matched Pattern Information <ul style="list-style-type: none"> <li>● Called Party Number = 9195552388</li> <li>● Time Zone = Etc/GMT</li> <li>● End Device = PSTN_RL</li> <li>● Call Classification = OffNet</li> <li>● InterDigit Timeout = NO</li> <li>● Device Override = Disabled</li> <li>● Outside Dial Tone = NO</li> </ul> </li> </ul> <b>Call Flow</b> <ul style="list-style-type: none"> <li>▶ Route Pattern: Pattern = [2-9]XX[2-9]XXXXXX</li> <li>▼ Route List: Route List Name = PSTN_RL <ul style="list-style-type: none"> <li>▶ RouteGroup:RouteGroupName = Standard Local Route Group (RTP_trunks) <ul style="list-style-type: none"> <li>● PreTransform Calling Party Number = 2304</li> <li>● PreTransform Called Party Number = 9195552388</li> </ul> </li> <li>▶ Calling Party Transformations</li> <li>▶ Called Party Transformations</li> <li>▶ Device :Type = SIPTrunk</li> </ul> </li> </ul>	<b>Results Summary</b> <ul style="list-style-type: none"> <li>▶ Calling Party Information <ul style="list-style-type: none"> <li>● Dialed Digits = 9195552388</li> <li>● Match Result = RouteThisPattern</li> </ul> </li> <li>▶ Matched Pattern Information <ul style="list-style-type: none"> <li>● Called Party Number = 9195552388</li> <li>● Time Zone = Etc/GMT</li> <li>● End Device = PSTN_RL</li> <li>● Call Classification = OffNet</li> <li>● InterDigit Timeout = NO</li> <li>● Device Override = Disabled</li> <li>● Outside Dial Tone = NO</li> </ul> </li> </ul> <b>Call Flow</b> <ul style="list-style-type: none"> <li>▶ Route Pattern: Pattern = [2-9]XX[2-9]XXXXXX</li> <li>▼ Route List: Route List Name = PSTN_RL <ul style="list-style-type: none"> <li>▶ RouteGroup:RouteGroupName = Standard Local Route Group <ul style="list-style-type: none"> <li>● PreTransform Calling Party Number = 2305</li> <li>● PreTransform Called Party Number = 919555388</li> </ul> </li> <li>▶ Calling Party Transformations</li> <li>▶ Called Party Transformations</li> </ul> </li> </ul>

A standard local route group is configured for long-distance calls. Calls from building A succeed, but calls from building B fail. On the system, each building has its own device pool. The DNA tool is used to test the configuration. How is this issue resolved?

- A. Change the partition of the route pattern.
- B. Add a sip trunk inside route group Standard Local Route Group.
- C. Modify the route pattern to add a prefix of 91.
- D. Add a local route group on the device pool configuration.

Correct Answer: B

#### QUESTION 4

A single site reports that when they dial select numbers, the call connects, but they do not get audio. The administrator finds that the calls are not routing out of the normal gateway but out of another site's gateway due to a TEHO configuration. What is the next step to diagnose and solve the issue?

- A. Verify that the route pattern has the correct calling-party transformation mask.
- B. Verify that IP routing is correct between the gateway and the IP phone.
- C. Verify that the dial peer of the gateway has the correct destination pattern configured.
- D. Verify that the route pattern is not blocking calls to the destination number.

Correct Answer: C

QUESTION 5

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Route Pattern. The top navigation bar includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. Below the navigation bar are icons for Save, Delete, Copy, and Add New. The main configuration area is divided into sections: Status, Pattern Definition, Calling Party Transformations, and Connected Party Transformations. In the Pattern Definition section, the following fields are visible: Pattern Usage\* (Domain Routing), IPv4 Pattern\* (\*.cms.domain.local), IPv6 Pattern (empty), Description (empty), Route Partition (internal-pt), SIP Trunk/Route List\* (CMS-Cluster-RL), and a Block Pattern checkbox. The Calling Party Transformations section includes Use Calling Party's External Phone Mask (checkbox), Calling Party Transformation Mask (empty), Prefix Digits (Outgoing Calls) (empty), Calling Line ID Presentation\* (Default), and Calling Line Name Presentation\* (Default). The Connected Party Transformations section includes Connected Line ID Presentation\* (Default) and Connected Line Name Presentation\* (Default). At the bottom, there are buttons for Save, Delete, Copy, and Add New, and a legend indicating that an asterisk (\*) denotes a required item.

Refer to the exhibit. The customer is troubleshooting an issue where users cannot dial the CMS SIP Route Pattern. The CMS URI they are attempting to dial is 7772002@cms.domain.local. Which IPv4 pattern must the customer enter to resolve the issue?

- A. 7772002@cms.domain.local
- B. \*@cms.domain.local
- C. \*.cms.domain.local
- D. cms.domain.local

Correct Answer: B

### QUESTION 6

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.

Correct Answer: AB

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### QUESTION 7

A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). Which route pattern should be configured to block only the numbers in this range?

- A. 813510[012345]
- B. 813510[12345]
- C. 813510[^0-5] D. 81XXXXX

Correct Answer: A

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### QUESTION 8

An administrator is configuring a new deployment using Cisco Unified CME. The SCCP phones register without any issues, but SIP phones are not registering. Assume that all other configuration is valid. Which code allows SIP phones to register to Cisco UCME?

- A. voice service voip allow-connections sip to h323
- B. voice service voip sip bind media source-interface Vlan100
- C. voice service voip sip bind control source-interface Vlan100
- D. voice service voip sip registrar server expires max 600 min 60

Correct Answer: D

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### QUESTION 9

An engineer is troubleshooting local ringback on a Cisco SIP gateway. The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified in the SDP. Which configuration change must be made on the gateway to resolve the issue?

- A. Router(config-sip-ua)# no disable-early-media 180
- B. Router(conf-voi-serv)# no disable-early-media 180
- C. Router(conf-voi-serv)# disable-early-media 180
- D. Router(config-sip-ua)# disable-early-media 180

Correct Answer: D

Reference: [https://www.cisco.com/en/US/docs/ios/12\\_3t/voice/command/reference/vrht\\_d2\\_ps5207\\_TSD\\_Products\\_Command\\_Reference\\_Chapter.html#wp1452642](https://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_d2_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1452642)

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### QUESTION 10

A company has users that are logged in to hunt groups. However, there is a requirement for hunt group configurations to provide an option to turn on audible ringtones when calls to a line group arrive at a phone that is logged out and on a break. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call does not ring at the phone of that line group member because of the logged-out status. Which action meets this requirement?

- A. Configure the HLog softkey on the phone so that while a user is logged off, it plays an audible tone when a call is missed.
- B. Set the service parameter Enterprise Feature Access number for hunt group logout and set up an access number.
- C. Set the service parameter Party Entrance Tone to "True."
- D. Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.

Correct Answer: D

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### QUESTION 11

Which section under the Real-Time Monitoring Tool allows for reviewing the call flow and signaling for a SIP call in real time?

- A. Analysis Manager > Inventory > Trace File Repositories
- B. System > Tools > Trace and Log Central
- C. Voice/Video > Session Trace Log View > Real Time Data
- D. Voice/Video > Session Trace Log View > Open From Local Disk

Correct Answer: C

Reference: <https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.html>

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#### **QUESTION 12**

Which IOS command creates a SIP-enabled dial peer?

- A. voice dial-peer 20 sip
- B. dial-peer voice 20 voip
- C. dial-peer voice 20 pots
- D. dial peer voice 20 sip

Correct Answer: B

Reference: <https://www.ciscopress.com/articles/article.asp?p=664148andseqNum=6>