300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

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.

Correct Answer: C

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configurationguide_chapter_010.html

QUESTION 2

Management wants to change the initial announcements for one of the existing call hunt groups. A new set of announcement audio file was provided. Which two configuration steps must the administrator take to accomplish this change? (Choose two.)

A. Identify the MOH audio source ID associated to one of the line group member\\'s “Network Hold MOH Audio Source”.

B. Identify the MOH audio source ID associated to “Network Hold Source and Announcements” under the Queuing section of the hunt pilot.

C. Identify the configured announcement names to change under the MOH audio source section, then upload the new files to the respective announcements under the Announcement section.

D. Identify the MOH audio source ID associated to one of the line group member\\'s “User Hold MOH Audio Source”.

E. Identify the configured announcement names to change under the Announcement section, and assign the uploaded files to the Queueing section of the hunt pilot.

Correct Answer: DE

QUESTION 3

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

Correct Answer: B

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200453-Configure-CUCM-Native-Call-Queuing-Featu.html

QUESTION 4

Which configuration must an administrator perform to display Translation Pattern operations in Cisco UCM SDL traces?

A. Enable the Detailed Call Analysis option under Enterprise Parameters for Cisco UCM.

B. Set up the Digit Analysis Complexity in Service Parameters for Cisco UCM to TranslationAndAlternatePatternAnalysis.

C. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco UCM Serviceability page.

D. By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.

Correct Answer: B

There is a bug CSCvb32314 in CUCM. Symptom:

Translation patterns are not present in cucm SDL traces even parameter "Digit Analysis Complexity" under Service parameters -> Call Manager -> Advanced is set to "TranslationAndAlernatePatternAnalysis". This is causing troubleshooting

of HCS looping dial plan issues (or complex dialplan issues) difficult as translation patterns applied before reaching endpoint are not visible.

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/hcs/12_5/HCS_Solution/Troubleshooting/chcs_b_troubleshooting-guide/chcs_m_troubleshooting-voice-application-components.html?dtid=osscdc000283

QUESTION 5

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
Content-Length: 0
```

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 6

A customer routes PSTN calls to ITSP through a SIP trunk on Cisco UCM that forwards and receives calls to and from ITSP. ITSP is set to send an E.164 number when the customer\\'s extension is four digits. Which action should be taken to route the incoming calls to four-digit extensions?

A. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

B. Set the Significant Digits to 8 on the SIP trunk.

C. Set the Significant Digits to 4 on the SIP trunk.

D. Configure a voice translation rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

Correct Answer: C

QUESTION 7

When configuring hunt groups, where does the administrator add the individual directory numbers that should be part of the group?

A. route group

B. line group

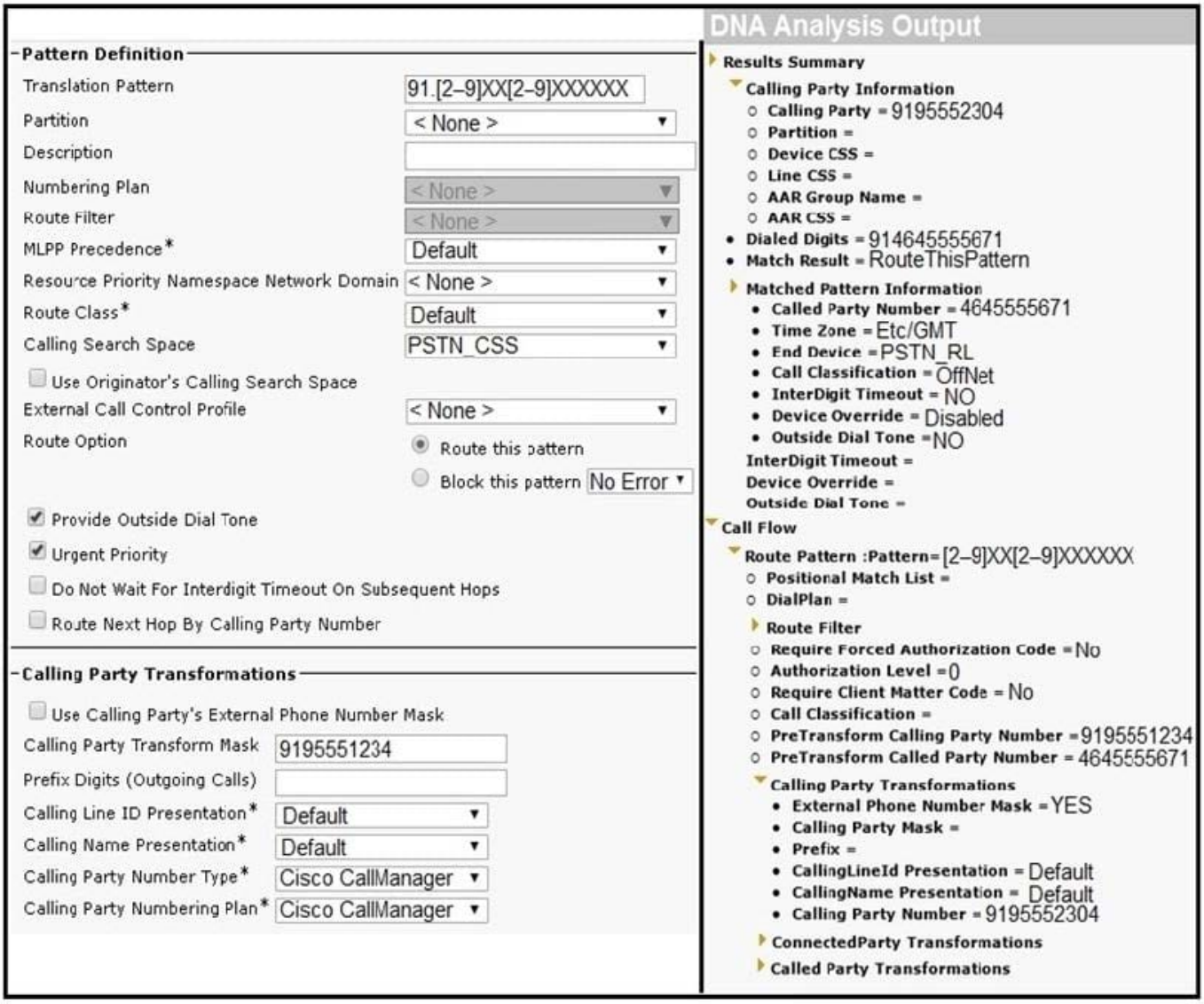
C. hunt list

D. hunt pilot

Correct Answer: B

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_010101.html

QUESTION 8



Refer to the exhibit. For long-distance calls, users must prefix their dialed number with "91". The translation pattern was created to strip the 91 as the PSTN expects a 10-digit number. The PSTN also requires the calling number to be set to 9195551234. However, the service provider has said calls with a different calling number are being received. How is this issue resolved?

A. Change the partition of the translation pattern from none to pstn_pt.

B. Disable Use Calling Party\\'s External Phone Number Mask on the route pattern.

C. Enable Force Authorization Code on the route pattern.

D. Enable Use Calling Party\\'s External Phone Number Mask on the translation pattern.

Correct Answer: B

QUESTION 9

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

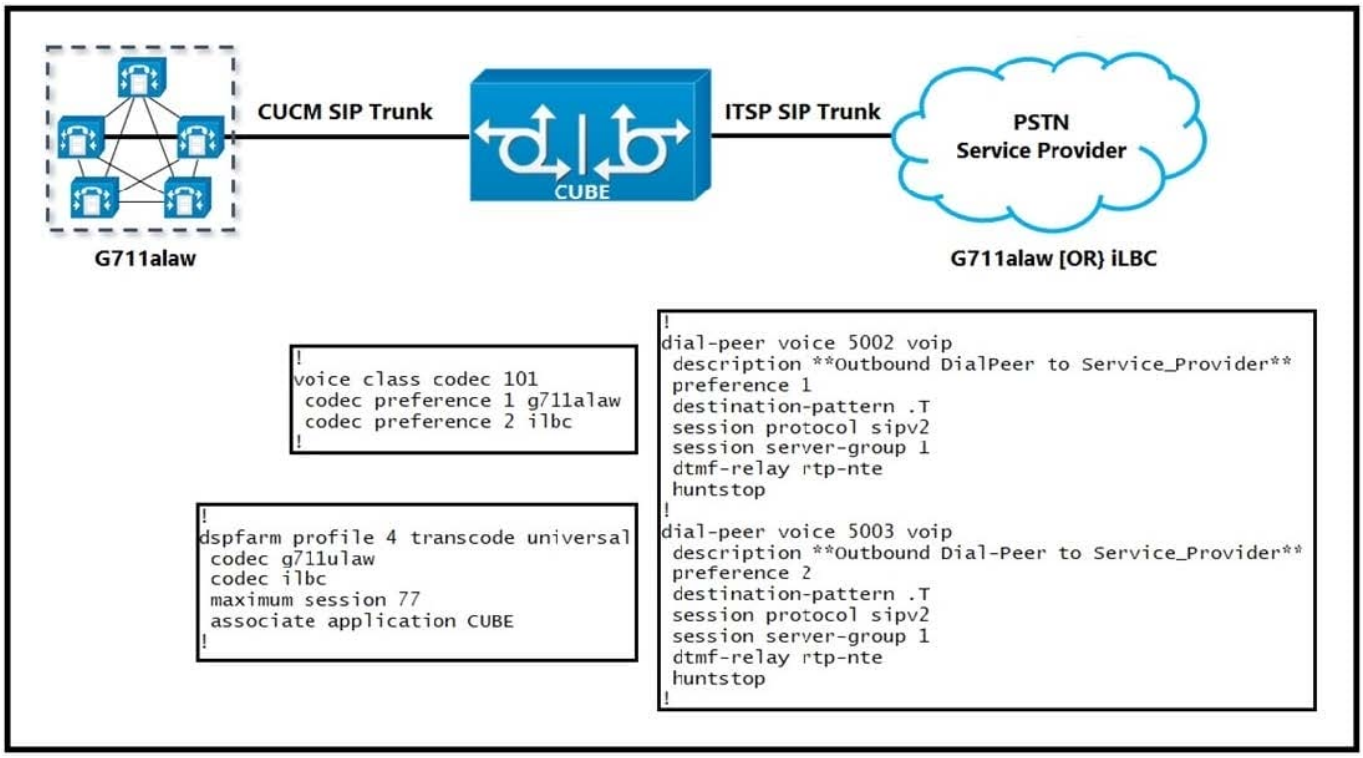
According to the the CME command reference, the answer should be \\'A\\':

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#wp2339729225

QUESTION 10

Refer to the exhibit.



Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time. Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?

A. dial-peer voice 5002 voip codec g711alaw ilbc ! dial-peer voice 5003 voip codec g711alaw ilbc

B. dial-peer voice 5002 voip voice-class codec 101 offer-all ! dial-peer voice 5003 voip voice-class codec 101 offer-all

C. dial-peer voice 5002 voip voice-class codec 101 ! dial-peer voice 5003 voip voice-class codec 101

D. dial-peer voice 5002 voip codec g711alaw ! dial-peer voice 5003 voip codec ilbc

Correct Answer: B

QUESTION 11

For a 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 H.245-signal and rtp-nte

C. interworking between an OOB method and RFC2833 for flow-through calls

D. interworking between H.245-alpha numeric and sip-kpml

Correct Answer: A

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35

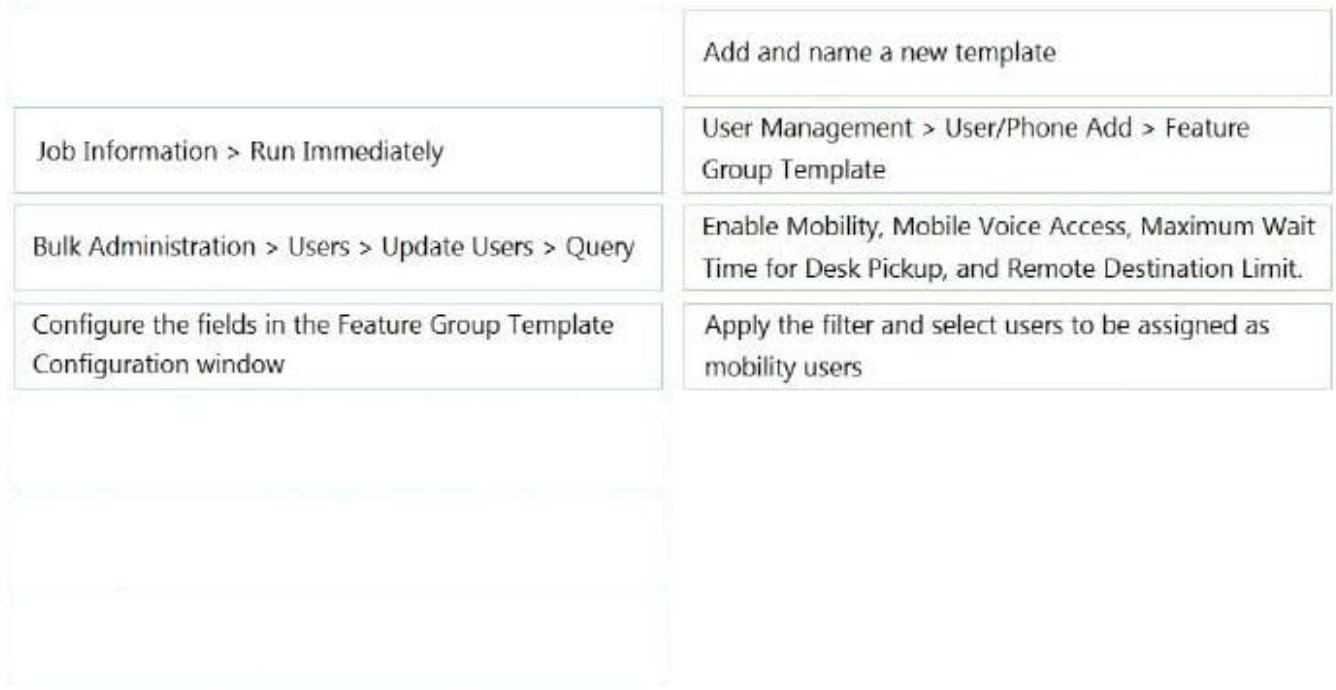
QUESTION 12

DRAG DROP

Drag and drop the steps from the left into the order to provision mobility users through LDAP on the right. Not all options are used.

Select and Place:

Correct Answer:



Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1/featureConfig/cucm_b_feature-configuration-guide-1251/cucm_b_feature-configuration-guide1251_chapter_010.html#task_77EEACB9BEBA958F465F4CE26BD76D36

QUESTION 13

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

Correct Answer: AB

QUESTION 14

A new solution is configured to support internal, local, and international calling. Calling [+44 1111 1111] from one of the registered internal phones does not work. Local and internal calls seem to work without any problems. The configuration has patterns configured to match the failing dialed number [+44 …. ….]. The other configured patterns show [2…] for internal numbers and [555 …] for local numbers. International numbers use E.164 as recommended. What is missing to make this solution work?

A. 001 or 00 must be used instead of the + sign on Cisco UCM

B. =+ cannot be used in a route pattern, only in a SIP pattern

C. \ in front of the +

D. / in front of the +

Correct Answer: C

QUESTION 15

CollabCorp is a global company with two clusters, emea.collab.corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI, but this is not working as expected. Which two configuration elements should be checked to resolve this issue? (Choose two.)

A. intercluster trunk

B. directory URI partition

C. SIP route pattern

D. calling search space and partition

E. SIP trunk

Correct Answer: CE

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