300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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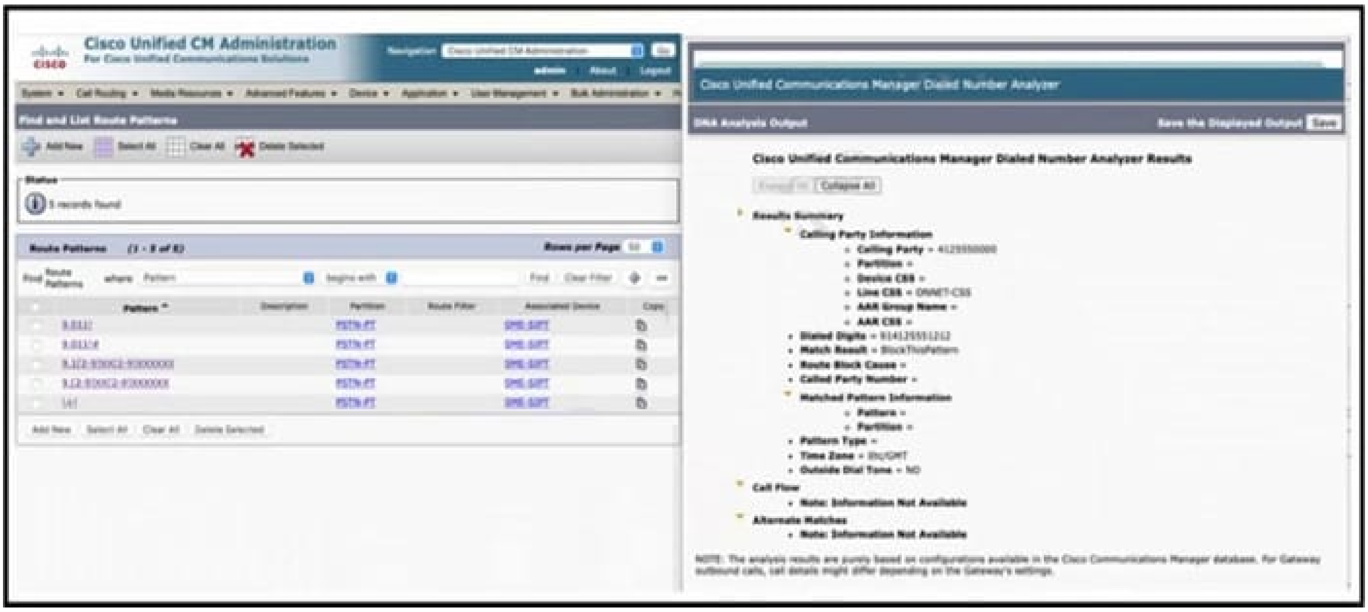
QUESTION 1



Refer to the exhibit. A collaboration engineer is troubleshooting an issue where a user of a Cisco UCM IP phone reports failed calls when trying to dial out to the PSTN. Which action resolves the issue?

A. Assign a calling search space to the line or the device that has access to the route pattern.

B. Deselect “Block this pattern” on the “Route Option” setting of the route pattern.

C. Select the “Urgent Priority” setting on the route pattern.

D. Instruct the user to not dial a “1” before their local area code.

Correct Answer: A

QUESTION 2

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:

Add and name a new template	step 1
Job Information > Run Immediately	step 2
Bulk Administration > Users > Update Users > Query	step 3
Configure the fields in the Feature Group Template Configuration window	step 4
User Management > User/Phone Add > Feature Group Template	
Apply the filter and select users to be assigned as mobility users	
Enable Mobility, Mobile Voice Access, Maximum Wait Time for Desk Pickup, and Remote Destination Limit.	

Correct Answer:

	Add and name a new template
Job Information > Run Immediately	User Management > User/Phone Add > Feature Group Template
Bulk Administration > Users > Update Users > Query	Enable Mobility, Mobile Voice Access, Maximum Wait Time for Desk Pickup, and Remote Destination Limit.
Configure the fields in the Feature Group Template Configuration window	Apply the filter and select users to be assigned as mobility users

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 3

Due to a shortage of physical interfaces on a device, the administrator requires that a loopback for RTP is used. Which command is required when using a loopback interface for RTP?

A. voice-class sip early-offer forced

B. voice-class sip bind control source-interface Loopback0

C. voice-class sip bind media source-interface Loopback0

D. voice-class sip resource priority mode passthrough

Correct Answer: C

QUESTION 4

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 5

A customer has multisite deployments with a globalized dial plan. The customer wants to route PSTN calls via the gateway assigned to each site.

Which two actions will fulfill the requirement? (Choose two.)

A. Create one global route list for PSTN calls that points to one global PSTN route group.

B. Create a route group which has all the gateways and associate it to the device pool of every site.

C. Assign one route group as a local route group in the device pool of the corresponding site.

D. Create one route group for each site and one global route list for PSTN calls that point to the local route group.

E. Create a hunt group and assign it to each side route pattern.

Correct Answer: AC

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/dialplan.html

QUESTION 6

Which built-in bridge configuration option can be set on the individual IP phones to ensure that the cluster-wide service parameter is used?

A. automatic

B. default

C. on

D. off

Correct Answer: B

QUESTION 7

Some users report having issues dialing some external numbers when traveling to other locations within the company. The company has five locations in five cities in one country and has an egress gateway in each location for TEHO. The configuration has no specific entry stating that the roaming users are using the local gateway, but calls are going out. How is a verification of the call routing in such a specific configuration performed to further identify the problem?

A. device mobility

B. standard local route group

C. local route groups

D. TEHO

Correct Answer: A

QUESTION 8

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 9

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 10

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

Correct Answer: CD

QUESTION 11

End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

A. Contact: header of the 200 OK response

B. Allow: header if the 200 OK response

C. o= line of SDP content

D. c= line of SDP content

Correct Answer: D

QUESTION 12

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

QUESTION 13

An administrator deployed a third-party H.323 gateway in a voice environment, but users report call failures when using features like call hold or call transfer. What are two reasons that these features fail? (Choose two.)

A. The CSS of the transfer initiating line does not contain the partition of the supplementary feature extension (DirectTransfer or MoH Number).

B. The MTP that is configured for use within the H.323 gateway configuration is configured as a trusted source, but the third-party gateway does not trust the signing root CA certificate of the MTP certificate.

C. The MTP does not support the negotiated codec, and media renegotiating during the call is not supported.

D. The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP.

E. The third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.

Correct Answer: CD

QUESTION 14

What is first preference condition matched in a SIP-enabled incoming dial peer?

A. incoming uri

B. target carrier-id

C. answer-address

D. incoming called-number

Correct Answer: A

Reference: https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc8

QUESTION 15

An IP Telephony administrator is deploying IP phones. The administrator has an existing Cisco UCME router with several SCCP and SIP phones registered. The administrator receives a request for a new SIP phone with MAC address 1111.2222.3333 and directory number 2050 to be added in the Cisco UCME. Which two configurations should be added in CME to support this request? (Choose two.)

A.

```
voice register pool 1
 id mac 1111.2222.3333
 type 8941
 number 2 dn 1
```

B.

```
ephone 1
 mac-address 1111.2222.3333
 type 8941
 button 1:2
```

C.

```
ephone-dn 2
 number 2050
```

D.

```
voice register dn 2
 number 2050
```

E.

```
voice register pool 1
 id mac 1111.2222.3333
 type 8941
 number 1 dn 2
```

A. Option A

B. Option B

C. Option C

D. Option D

E. Option E

Correct Answer: DE