

300-815^{Q&As}

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

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 2

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 3

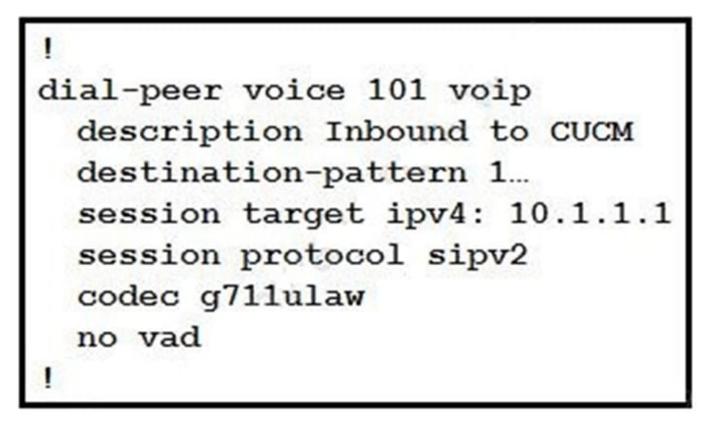
An administrator is configuring a SIP trunk to an ITSP. The SIP connection will traverse from a Cisco UCM to the ISTP through a Cisco Unified Border Element. The ITSP has indicated that they require an in-band method for DTMF. Which command on the outbound dial-peer to the ITSP will meet this requirement?

- A. router (config-dial-peer) dtmf-relay sip-notify
- B. router (config-dial-peer) dtmf-relay sip-kpml
- C. router (config-dial-peer) dtmf-relay h245-alphanumeric
- D. router (config-dial-peer) dtmf-relay rtp-nte



Correct Answer: D

QUESTION 4



Refer to the exhibit. When setting up a new connection to Cisco UCM, the engineer must use out-of-band DTMF. Which configuration meets this requirement?

- A. dtmf-relay h245-alphanumeric
- B. dtmf-relay sip-kpml
- C. dtmf-relay cisco-rtp
- D. dtmf-relay cisco-rtp
- Correct Answer: B

QUESTION 5

Refer to the exhibit.



```
voice translation-profile incoming
    translate called 999
1
voice translation-rule 999
    rule 1/\ (^[1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\) $ / / \2333\1/
1
dial-peer voice 999 voip
    translation-profile outgoing incoming
    session protocol sipv2
    incoming called-number
    dtmf-relay rtp-nte
    codec transparent
    destination dpg 888
    no vad
1
voice class dpg 888
    dial-peer 888
1
dial-peer voice 888 voip
    destination-pattern 888
    session protocol sipv2
    session target ipv4:192.168.0.1
    codec transparent
    dtmf-relay rtp-nte
    no vad
```

Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco UCM is expecting the called number to be delivered as "444333222". The administrator already verified that the IP address of the Cisco UCM is set up correctly, and there are no dial peers configured other than those shown in the exhibit. Which action should the administrator take to fix the issue?

A. Change the destination-pattern on the outgoing dial peer to match "444333222".

B. Set up translation-profile on the incoming dial peer to match incoming traffic.

- C. Create specific matching for "222333444" on the incoming dial peer.
- D. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".

Correct Answer: B

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