

## [October-2021 Real Exam Questions-Braindump2go 300-815 Dumps PDF 300-815 129Q Download[Q105-Q119]

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**QUESTION 105**The SIP session refresh timer allows the RTP session to stay active during an active call. The Cisco UCM sends either SIP-INVITE or SIP-UPDATE messages in a regular interval of time throughout the active duration of the call. During a troubleshooting session, the engineer finds that the Cisco UCM is sending SIP-UPDATE as the SIP session refresher, and the engineer would like to use SIP-INVITE as the session refresher. What configuration should be made in the Cisco UCM to achieve this?  
A. Enable SIP ReMXX Options on the SIP profile.  
B. Enable Send send-receive SDP in mid-call INVITE on the SIP profile.  
C. Change Session Refresh Method on the SIP profile to INVITE.  
D. Increase Retry INVITE to 20 seconds on the SIP profile.  
**Answer: C**  
**QUESTION 106**Refer to the exhibit. ILS has been configured between two hubs using this configuration. The hubs appear to register successfully, but ILS is not functioning as expected. Which configuration step is missing?

Cluster ID/Name	Last Contact Time	Role	Advertised Route String	USN Data Synchronization Status
StandAloneCluster	2/17/21 10:31 AM	Hub	CCIE	Not Applicable
StandAloneCluster		Hub (Local Cluster)	CCNP	Disabled

A. A password has never been set for ILS.  
B. Use TLS Certificates must be selected.  
C. Trust certificates for ILS have not been installed on the clusters.  
D. The Cluster IDs have not been set to unique values  
**Answer: D**  
**QUESTION 107**A new deployment is using MVA for a specific user on the sales team, but the user is having issues when dialing DTMF. Which DTMF method must be configured in resolve the issue?  
A. gateway  
B. out-of-band  
C. channel  
D. in-band  
**Answer: B**  
**QUESTION 108**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 IP routing is correct between the gateway and the IP phone.  
B. Verify that the route pattern is not blocking calls to the destination number.  
C. Verify that the dial peer of the gateway has the correct destination pattern configured.  
D. Verify that the route pattern has the correct calling-party transformation mask  
**Answer: C**  
**QUESTION 109**An engineer is configuring Cisco UCM to forward parked calls back to the user who parked the call if it is not retrieved after a specified time interval. Which action must be taken to accomplish this task?  
A. Configure device pools.  
B. Configure service parameters.  
C. Configure enterprise softkeys.  
D. Configure class of control.  
**Answer: B**  
**QUESTION 110**Refer to the exhibit. 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?

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.1.60.105:5060;branch=9hg4bk721ed5d4
From: "1001" <sip:1001@10.88.247.229>;tag=6cfa89726ac700b569ec133a-7e6cd8aa
To: <sip:2005@10.88.247.229>;tag=47B5F70-43B
Date: Fri, 19 Apr 2019 12:13:40 GMT
Call-ID: 6cfa8972-6ac7002b-5af19a5c-0de23108@10.1.60.105
CSeq: 101 INVITE
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
```

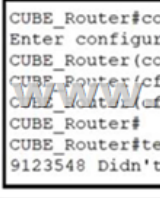
A. voice service voip sip send 180 sdp.  
B. voice service voip sip rehxx require 100rel.  
C. sip-ua disable-early-media 180D.  
D. voice service voip sip no reMxx  
**Answer: B**  
**QUESTION 111**Users are reporting that several inter-site calls are failing, and the message "not enough bandwidth" 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 Call Queuing so that the user waits until there is bandwidth availableC. Configure AAR to reroute calls that are denied by Call Admission Control through the PSTN.D. Reroute all calls through the PSTN and avoid using WAN.Answer:

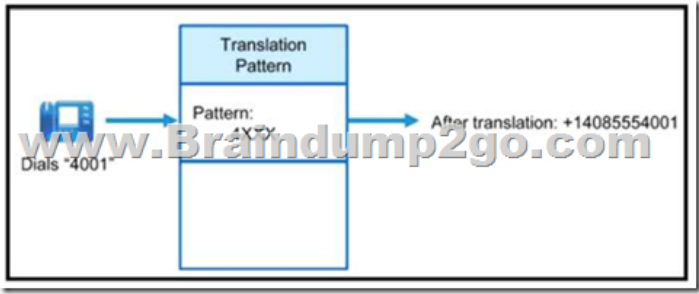
QUESTION 112An engineer must configure a Cisco UCM hunt list so that calls to users in a line group are routed to the first idle user and then the next.Which distribution algorithm must be configured to accomplish this task?A. top downB. circularC. broadcastD. longest idle timeAnswer: AQUESTION 113An administrator configured Cisco Unified Mobility to block access to remote destinations for certain caller IDs. A user reports that a blocked caller was able to reach a remote destination. Which action resolves the issue?A. Configure Single Number Reach.B. Configure an access list.C. Configure a mobility identity.D. Configure Mobile Voice Access.Answer: BQUESTION 114Refer to the exhibit. An engineer is troubleshooting a call-establishment problem between Cisco Unified Border Element and Cisco UCM. Which command set corrects the issue?



A. SIP binding in SIP configuration mode:voice service voip sipbind control source-interface GigabitEthernet0/0/0 bind media source-interface GigabitEthernet0/0/0B. SIP binding In SIP configuration mode:voice service volpsipbind control source-Interface GlgablEthernet0/0/1 bind media source-Interface GlgablEthernet0/0/1C. SIP binding In dial-peer configuration mode:dial-peer voice 300 voipvoice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source- interface GigabitEthernet0/0/1D. SIP binding in dial-peer configuration mode:dial-peer voice 100 volpvoice-class sip bind control source-interface GigabitEthernet0/0/0 voice-class sip bind media source-interface GigabitEthernet0/0/0Answer: DQUESTION 115 Refer to the exhibit. Which change to the translation rule is needed to strip only the leading 9 from the digit string 9123548?



A. rule 1 /9(\*/A)1/B. rule 1 /.\*(3548S)/1/C. rule 1 /9(d\*/1/D. rule 1/9123548/1/Answer: AQUESTION 116A 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 route group for each site and one global route list for PSTN calls that point to the local route group.B. Create a route group which has all the gateways and associate it to the device pool of every site.C. Create one global route list for PSTN calls that points to one global PSTN route group.D. Create a hunt group and assign it to each side route patternE. Assign one route group as a local route group in the device pool of the corresponding site.Answer: AEQUESTION 117Refer to the exhibit. A company needs to ensure that all calls are normalized to E164 format. Which configuration will ensure that the resulting digit string 14085554001 is created and will be routed to the E.164 routing schema?



A. Called Party Transformation Mask of + 14085554XXXXB. Called Party Transformation Mask of 1408555[35]XXXXC. Calling Party Transformation Mask of +1408555XXXXD. Calling Party Transformation Mask of +14085554XXXAnswer:

AQUESTION 118An engineer set up and successfully tested a TEHO solution on the Cisco UCM. PSTN calls are routed correctly using the IP WAN as close to the final PSTN destination as possible. However, suddenly, calls start using the backup local gateway instead. What is causing the issue?  
A. WAN connectivity  
B. LAN connectivity  
C. route pattern  
D. route list and route group

Answer: AQUESTION 119An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM. How should this be accomplished?  
A. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.  
B. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.  
C. Localize the calling and called numbers to E. 164 format and globalize the called number in the gateway.  
D. Globalize the calling and called numbers to E. 164 format and localize the called number in the gateway.

Answer: DResources From: 1. 2021 Latest Braindump2go 300-815 Exam Dumps (PDF & VCE) Free Share:

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