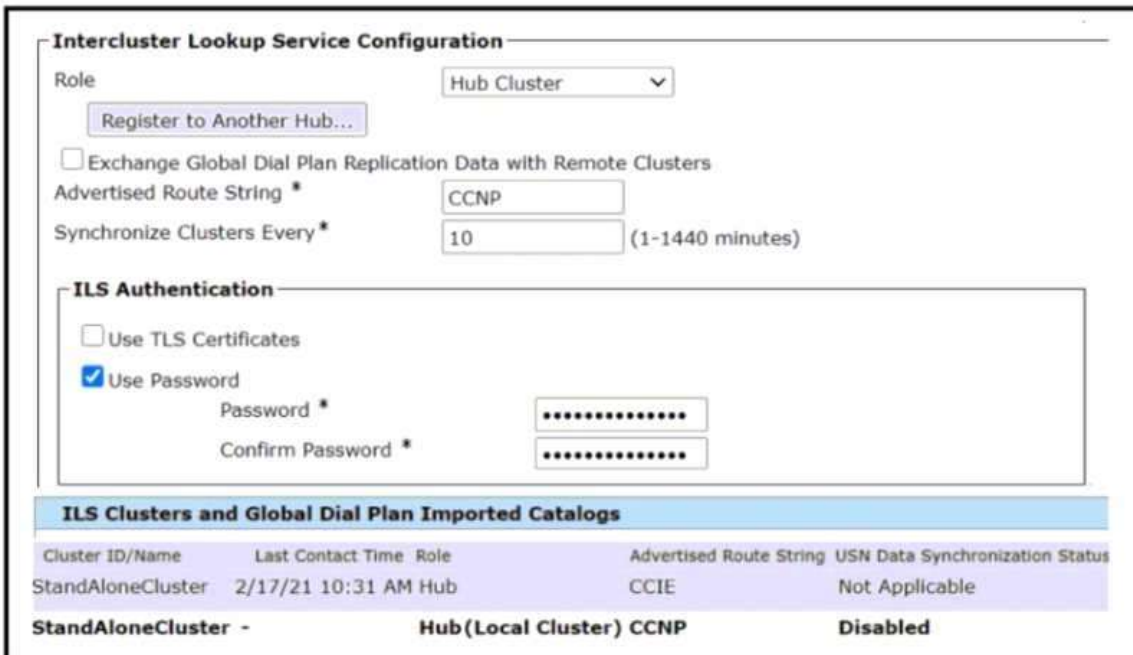➢ **Vendor: Cisco**

➢ **Exam Code: 300-815**

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

➢ **New Updated Questions from Braindump2go (Updated in October/2021)**

Visit Braindump2go and Download Full Version 300-815 Exam Dumps

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?

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 lo 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=z9hG4bK721ed5d4
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
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
```

A. voice service voip sip send 180 sdp
B. voice service voip sip rehxx require 100rel
C. sip-ua disable-early-media 180
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 available
C. 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: C

QUESTION 112
An 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 down
B. circular
C. broadcast
D. longest idle time

Answer: A

QUESTION 113
An 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: B

QUESTION 114
Refer 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 sip
    bind control source-interface GigabitEthernetO/0/0
    bind media source-interface GigabitEthernetO/0/0
B. SIP binding In SIP configuration mode:
    voice service volp
    sip
    bind control source-Interface GlgabltEthernetO/0/1
    bind media source-Interface GlgabltEthernetO/0/1
C. SIP binding In dial-peer configuration mode:
    dial-peer voice 300 voip
    voice-class sip bind control source-interface GigabitEthernetO/0/1
    voice-class sip bind media source- interface GigabitEthernetO/0/1
D. SIP binding in dial-peer configuration mode:
    dial-peer voice 100 volp
    voice-class sip bind control source-interface GigabitEthernetO/0/0
    voice-class sip bind media source-interface GigabitEthernetO/0/0

Answer: D

QUESTION 115
Refer to the exhibit. Which change to the translation rule is needed to strip only the leading 9 from the digit string 9123548?


```
CUBE_Router#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
CUBE_Router(config)#voice translation-rule 999
CUBE_Router(cfg-translation-rule)#rule 1 /^9(.*)/ //
CUBE_Router(cfg-translation-rule)#end
CUBE_Router#
CUBE_Router#test voice translation-rule 999 9123548
9123548 Didn't match with any of rules
```

A. rule 1 /^9\(.*\)/A1/
B. rulel /.*\(3548S\)/^1/
C. rulel /^9\(\d*\)/^1/
D. rule 1/^9123548/^1/

Answer: A

QUESTION 116
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 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 pattern
E. Assign one route group as a local route group in the device pool of the corresponding site.

Answer: AE

QUESTION 117
Refer 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?
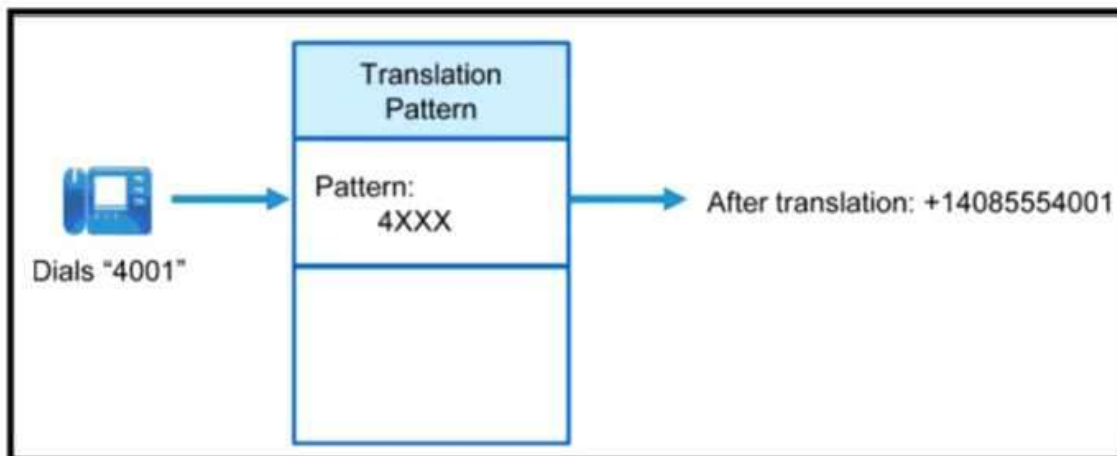
A. Called Party Transformation Mask of + 14085554XXX
B. Called Party Transformation Mask of 1408555[35)XXX
C. Calling Party Transformation Mask of +1408555XXXX
D. Calling Party Transformation Mask of +14085554XXX

Answer: A

QUESTION 118
An 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: A

QUESTION 119
An 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: D