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Question 1

Question Type: MultipleChoice

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?

Options:

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 2

Question Type: MultipleChoice

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=47B5F70-43B
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
```

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?

Options:

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 3

Question Type: **MultipleChoice**

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?

Options:

A- Configure device pools.

B- Configure service parameters

C- Configure enterprise softkeys.

D- Configure class of control.

Answer:

B

Question 4

Question Type: MultipleChoice

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?

Options:

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 5

Question Type: MultipleChoice

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?

Options:

A- gateway

B- out-of-band

C- channel

D- in-band

Answer:

B

Question 6

Question Type: **MultipleChoice**

Refer to the exhibits.

Region Configuration **Related Links:** Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name* Region A

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Region B	Use System Default (Factory Default low loss)	8 kbps (G.729)	None	None
Region C	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
Region D	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Media Resource Group Information

Name* MRG2

Description Media Resource Group 2

Devices for this Group

Available Media Resources**

MOH_5
MTP_3
MTP_4
MTP_5
XCODER

Selected Media Resources*

ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP_2 (MTP)

```
#CallManager SDL Logs
|AppInfo  |DET-MediaManager-(22401)::preCheckCapabilities, caps mismatch! Xcoder Reqd. kbps(8),
    filtered A[capCount=0 (Cap,ptime)=], B[capCount=4 (Cap,ptime)= (11,220) (12,220) (15,220) (9,270)]
    allowMTP=0 numXcoderRequired=1 xcodingSide=1
|SdlSig   |MrmAllocateMtpResourceErr               |waitResourcesAllocated
    |MediaManager(6,100,144,22401)    |MediaResourceManager(6,100,142,1)
```

Regions have been configured for all major branches based on the available circuit bandwidth. Some calls from Region A endpoints to Region B endpoints are failing to connect. How is this issue resolved?

Options:

A- Update the calling search space for affected endpoints to none.

B- Add a media resource to transcode between available capabilities.

C- Update all regions to 8 kbps maximum audio bitrate.

D- Increase the number of available media termination points.

Answer:

B

Question 7

Question Type: **MultipleChoice**

A user requests a feature to send an active call to the mobile phone number on the physical phone. As an administrator, what should be configured in the Cisco UCM to accomplish this?

Options:

A- A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey 'Mobility' to the physical phone's softkey template.

B- A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey 'Mobility' to the physical phone's softkey template.

C- A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey 'Join' to the physical phone's softkey template

D- A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey 'Join' to the physical phone's softkey template.

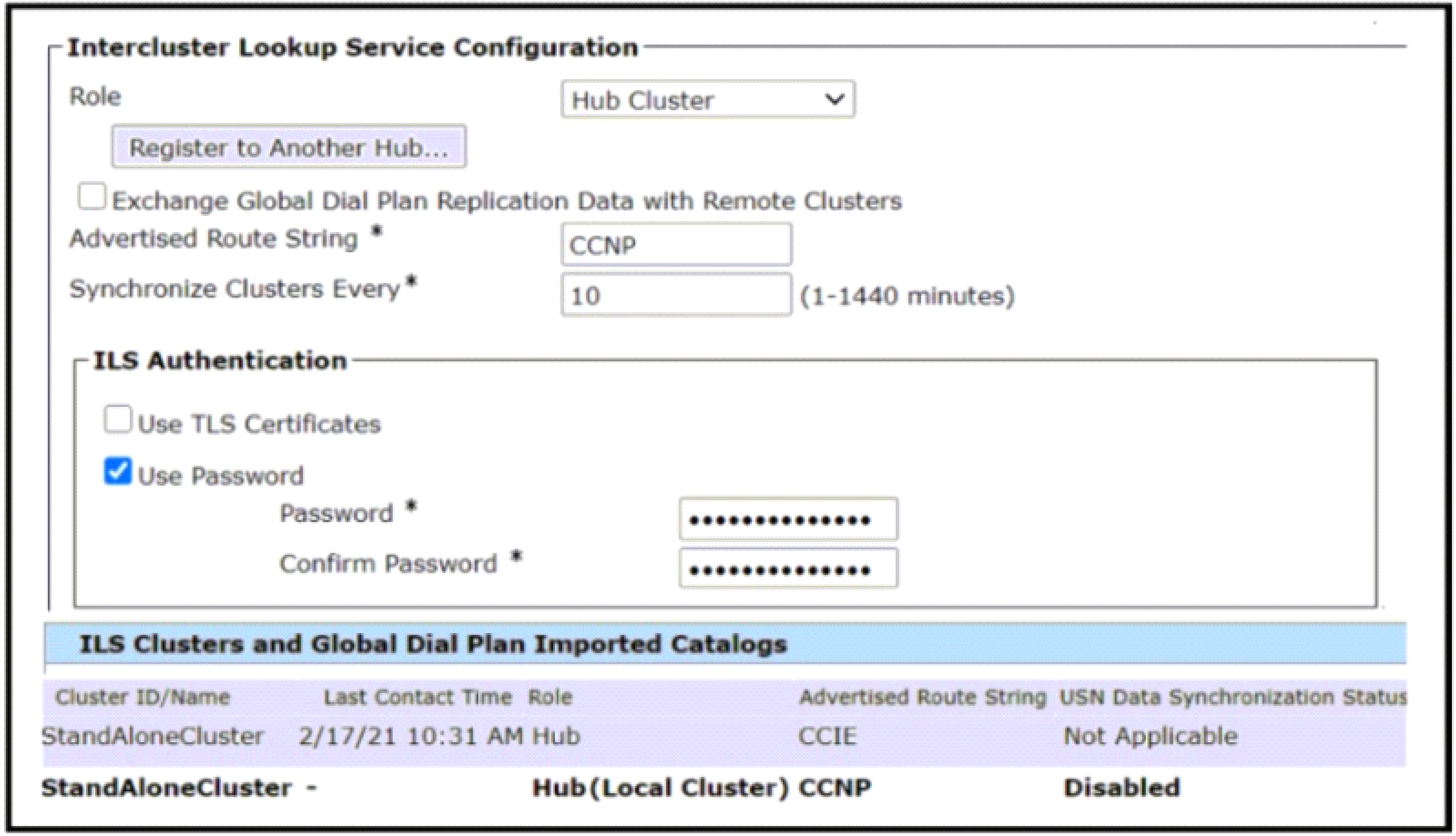
Answer:

B

Question 8

Question Type: **MultipleChoice**

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?

Options:

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 9

Question Type: MultipleChoice

Cisco UCM has 100,000 entries in the database learned through the ILS Service. Parameter ILS Max Number of Learned Objects in Database value is set to 100,000. What will happen to learned data when the service parameter value is reduced to 50,000?

Options:

A- Cisco UCM does not write additional ILS learned objects to the database and will delete the first 50,000 entries learned to keep it to the service parameter value.

B- Cisco UCM does not write additional ILS learned objects to the database and will delete the last 50,000 entries learned to keep it to the service parameter value.

C- Cisco UCM does not write additional ILS learned objects to the database and keeps the existing database entries.

D- Cisco UCM will overwrite an entry for newly learned data and keep the parameter value at 100,000.

Answer:

C

Question 10

Question Type: **MultipleChoice**

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-vers

a. As an administrator, which additional configuration should be made to fulfill the user's request?

Options:

A- Confirm that the desk phone is subscribed to Cisco Extension Mobility.

B- Check to make sure that the Resume softkey option appears on the desk phone.

C- Use Dialed Number Analyzer to determine if the user extension can dial the mobile phone.

D- Add the mobility key to the softkey template that the desk phone is using.

Answer:

D

Question 11

Question Type: MultipleChoice

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?

Options:

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