

## 300-815<sup>Q&As</sup>

Implementing Cisco Advanced Call Control and Mobility Services  
(CLACCM)

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## QUESTION 1

Refer to the exhibit.

### Intercluster Lookup Service Configuration

Role: Hub Cluster

Exchange Global Dial Plan Replication Data with Remote Clusters

Advertised Route String \* CCNP

Synchronize Clusters Every\* 10 (1-1440 minutes)

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#### ILS Authentication

Use TLS Certificates

Use Password

Password \* .....

Confirm Password \* .....

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### ILS Clusters and Global Dial Plan Imported Catalogs

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

ILS has been configured between two hubs this configuration. The hubs appear to register successfully, but ILS is not functioning as expected. Which configuration step is missing?

- A. Use TLS Certificates must be selected.
- B. The Cluster IDs have not been set to unique values.
- C. A password has never been set for ILS.
- D. Trust certificates for ILS have not been installed on the clusters.

Correct Answer: B

Reference: [https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/11\\_5\\_1\\_SU7/cucm\\_b\\_system-configuration-1151su7-1151su8/cucm\\_b\\_system-configuration-guide-1151su1\\_chapter\\_011001.pdf](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU7/cucm_b_system-configuration-1151su7-1151su8/cucm_b_system-configuration-guide-1151su1_chapter_011001.pdf)

## QUESTION 2

An administrator discovers that employees are making unauthorized long-distance and international calls from logged-off Extension Mobility phones when the authorized users are away from their desks. Which two configurations should the administrator configure in the Cisco UCM to avoid this issue? (Choose two.)

- A. Remove the long-distance and international pattern partitions from the calling search space of the physical phone.
- B. Add the long-distance and international pattern partitions to the calling search space of the physical phone's directory number.
- C. Remove the long-distance and international pattern partitions from the calling search space of the device profile.
- D. Add the long-distance and international pattern partitions to the calling search space of the physical phone.
- E. Add the long-distance and international pattern partitions to the calling search space of the device profile

Correct Answer: AE

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**QUESTION 3**

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

---

## QUESTION 4

A solution for a large company is being reconfigured to optimize for cost saving. The company has an extensive global QoS-enabled network with enough bandwidth to create a converged network. Local calls are relatively inexpensive in

countries the company have operations, but long distance and international calls are expensive.

Which type of configuration supported by Cisco UCM would help optimize cost control for this company?

- A. standard local route groups for mobile users
- B. Mobile and Remote Access
- C. high complex codec support like G.729 to minimize bandwidth usage
- D. tail end hop off

Correct Answer: D

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#### QUESTION 5

Refer to the exhibit.

```
voice class codec 100
  codec preference 1 g711alaw
  codec preference 2 g729r8
  codec preference 3 g729br8
  codec preference 4 g711ulaw
!
dial-peer voice 5002 voip
  session protocol sipv2
  session server-group 1
  incoming called-number 5...
  voice-class codec 100
  dtmf-relay rtp-nte
  no vad

m=audio 30104 RTP/AVP 0 9 124 116 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:124 iSAC/16000
a=rtpmap:116 iLBC/8000
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

The Cisco Unified Border Element receives an INVITE matching inbound dial peer 5002. The outbound dial peer supports only iLBC, and a Local Transcoding Interface is allocated. Based on the configuration and SDP from the

INVITE message, which codec is chosen by Cisco Unified Border Element for the inbound call leg?

- A. G.729r8
- B. G.711 A-law
- C. G.711 U-law
- D. G.729br8

Correct Answer: C

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**QUESTION 6**

An administrator is troubleshooting a one-way audio issue for a call that uses H.323 protocol in slow-start mode. The administrator requests that the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call is provided. The H.225 and H.245 messages for one of the one-way audio calls are gathered and the call flow has not invoked any media resources. Where is the RTP IP and port information for both sides found?

- A. H.245 Terminal Capability Set
- B. H.245 Open Logical Channel
- C. H.225 Connect
- D. H.245 Open Logical Channel Ack

Correct Answer: D

Reference: <http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html>

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**QUESTION 7**

Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?

- A. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path.
- B. Cisco UCM invoked media termination point resources.
- C. The RTP traffic is arriving beyond the jitter buffer on the receiving end.
- D. A firewall in the media path is blocking TCP ports 16384-32768.

Correct Answer: A

---

**QUESTION 8**

Refer to the exhibit.

```

!
dial-peer voice 10 voip
description Inbound
session protocol sipv2
incoming called-number 2000
dtmf-relay rtp-nte
no vad
!
dial-peer voice 20 voip
description Outbound
destination-pattern 2.
session protocol sipv2
session target ipv4:192.168.100.101
voice-class sip options-keepalive
dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0

```

TAG	TYPE	MIN	OPER	PREFIX	DEST-PATTERN	PRE FER	PASS THRU	SESS-SER-GRP\ SESS-TARGET	OUT STAT	PORT	KEEPALIVE	VRF
10	voip	up	up			0	syst					NA
20	voip	up	up		2.	0	syst	ipv4:192.168.100.101			busyout	NA

A call made through the Cisco Unified Border Element to pilot 2000 is failing. What is causing the call to fail?

- A. The Cisco Unified Border Element is not receiving a response to its OPTION keepalives.
- B. The destination pattern is incorrect for the dialed number.
- C. VAD was not disabled on the outgoing dial peer.
- D. No codecs are configured on the dial peers.

Correct Answer: D

## QUESTION 9

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?

- A. debug H.323 messages
- B. debug H.225 asn1
- C. debug H.246 asn 1
- D. debug H.225 media
- E. debug H.323 asn 1

Correct Answer: B

**QUESTION 10**

Destination					
Destination Address in an SRV					
Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
* sip.cisco.com		0	down	local=3	Time Down: 0 day 0 hour 59 minutes

Refer to the exhibit. A collaboration engineer is troubleshooting an issue where external callers cannot leave voicemail messages. Also, internal users report hearing the reorder tone (fast busy) when they attempt to retrieve voicemail messages from their Cisco IP phones. Which action resolves the issue?

- A. Verify that the correct port numbers are used for the SIP trunk.
- B. Ensure that the SIP Trunk Security Profile is configured to use UDP for transport.
- C. Start the Cisco Call Manager service at the destination.
- D. Ensure that Cisco UCM can resolve the destination address via DNS.

Correct Answer: D

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**QUESTION 11**

What are two configuration features of the Standard Local Route Group deployment? (Choose two.)

- A. is associated under the route group
- B. is associated only under the route list
- C. chooses the route group that is configured under the device pool of the calling-party device
- D. chooses the route group that is configured under the device pool of the called-party device
- E. is assigned directly to the route pattern

Correct Answer: BC

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**QUESTION 12**

Refer to the exhibit.



Building A	Building B
<b>Results Summary</b> <ul style="list-style-type: none"> <li>▶ Calling Party Information           <ul style="list-style-type: none"> <li>● Dialed Digits = 9195552388</li> <li>● Match Result = RouteThisPattern</li> </ul> </li> <li>▶ Matched Pattern Information           <ul style="list-style-type: none"> <li>● Called Party Number = 9195552388</li> <li>● Time Zone = Etc/GMT</li> <li>● End Device = PSTN_RL</li> <li>● Call Classification = OffNet</li> <li>● InterDigit Timeout = NO</li> <li>● Device Override = Disabled</li> <li>● Outside Dial Tone = NO</li> </ul> </li> </ul> <b>Call Flow</b> <ul style="list-style-type: none"> <li>▶ Route Pattern: Pattern = [2-9]XX[2-9]XXXXXX</li> <li>▼ Route List: Route List Name = PSTN_RL           <ul style="list-style-type: none"> <li>▶ RouteGroup:RouteGroupName = Standard Local Route Group (RTP_trunks)               <ul style="list-style-type: none"> <li>● PreTransform Calling Party Number = 2304</li> <li>● PreTransform Called Party Number = 9195552388</li> </ul> </li> <li>▶ Calling Party Transformations</li> <li>▶ Called Party Transformations</li> <li>▶ Device :Type = SIPTrunk</li> </ul> </li> </ul>	<b>Results Summary</b> <ul style="list-style-type: none"> <li>▶ Calling Party Information           <ul style="list-style-type: none"> <li>● Dialed Digits = 9195552388</li> <li>● Match Result = RouteThisPattern</li> </ul> </li> <li>▶ Matched Pattern Information           <ul style="list-style-type: none"> <li>● Called Party Number = 9195552388</li> <li>● Time Zone = Etc/GMT</li> <li>● End Device = PSTN_RL</li> <li>● Call Classification = OffNet</li> <li>● InterDigit Timeout = NO</li> <li>● Device Override = Disabled</li> <li>● Outside Dial Tone = NO</li> </ul> </li> </ul> <b>Call Flow</b> <ul style="list-style-type: none"> <li>▶ Route Pattern: Pattern = [2-9]XX[2-9]XXXXXX</li> <li>▼ Route List: Route List Name = PSTN_RL           <ul style="list-style-type: none"> <li>▶ RouteGroup:RouteGroupName = Standard Local Route Group               <ul style="list-style-type: none"> <li>● PreTransform Calling Party Number = 2305</li> <li>● PreTransform Called Party Number = 919555388</li> </ul> </li> <li>▶ Calling Party Transformations</li> <li>▶ Called Party Transformations</li> </ul> </li> </ul>

A standard local route group is configured for long-distance calls. Calls from building A succeed, but calls from building B fail. On the system, each building has its own device pool. The DNA tool is used to test the configuration. How is this issue resolved?

- A. Change the partition of the route pattern.
- B. Add a sip trunk inside route group Standard Local Route Group.
- C. Modify the route pattern to add a prefix of 91.
- D. Add a local route group on the device pool configuration.

Correct Answer: B

### QUESTION 13

An administrator is configuring Cisco UCM and the system to send \*.webex.com traffic to a Cisco UCM Session Management Edition cluster. The administrator wants to limit which endpoints can reach \*.webex.com. Which two items must the administrator configure for the SIP route pattern? (Choose two.)

- A. calling party transformation
- B. partition of the SIP route pattern
- C. connected party transformation
- D. called party URI transformation
- E. destination SIP trunk of the SIP route pattern

Correct Answer: BE

## QUESTION 14

```
2899152: Mar 31 21:23:16.777: TCB7F7F9668A9E8 setting property TCP_ALWAYS PUSH (15) 7F7F94803528
2899153: Mar 31 21:23:16.777: tcp_uniqueport: using ephemeral max 65535
2899154: Mar 31 21:23:16.777: TCP: Random local port generated 30202, network 1
2899155: Mar 31 21:23:16.777: TCB7F7F9668A9E8 bound to 10.10.10.40.30202
2899156: Mar 31 21:23:16.777: Reserved port 30202 in Transport Port Agent for TCP IP type 1
2899157: Mar 31 21:23:16.777: TCP: sending SYN, seq 974312871, ack 0
2899158: Mar 31 21:23:16.777: TCP0: Connection to 10.10.10.14:5060, advertising MSS 536
2899159: Mar 31 21:23:16.777: TCP0: state was CLOSED -> SYNSENT [30202 -> 10.10.10.14(5060)]
2899160: Mar 31 21:23:18.777: TCP0: RETRANS timeout timer expired
2899161: Mar 31 21:23:18.777: 10.10.10.40:30202 <--> 10.10.10.14:5060 congestion window changes
2899162: Mar 31 21:23:18.777: cwnd from 536 to 536, ssthresh from 65535 to 1072
2899163: Mar 31 21:23:18.777: TCP0: timeout #1 - timeout is 4000 ms, seq 974312871
--More-- 2899164: Mar 31 21:23:18.777: TCP: (30202) -> 10.10.10.14(5060)
2899165: Mar 31 21:23:19.503: TCP866: state was ESTAB -> FINWAIT1 [22 -> 10.16.8.226(52145)]
2899166: Mar 31 21:23:19.503: TCP866: sending FIN
2899167: Mar 31 21:23:19.536: TCP866: state was FINWAIT1 -> FINWAIT2 [22 -> 10.16.8.226(52145)]
2899168: Mar 31 21:23:19.536: TCP866: FIN processed
2899169: Mar 31 21:23:19.536: TCP866: state was FINWAIT2 -> TIMEWAIT [22 -> 10.16.8.226(52145)]
2899170: Mar 31 21:23:21.777: Released port 30202 in Transport Port Agent for TCP IP type 1 delay 240000
2899171: Mar 31 21:23:21.777: TCP0: state was SYNSENT -> CLOSED [30202 -> 10.10.10.14(5060)]
2899172: Mar 31 21:23:21.777: TCB 0x7F7F9668A9E8 destroyed
```

Refer to the exhibit. An engineer is trying to set up a new deployment using the SIP provider with TCP for signaling. After troubleshooting, the customer notices that the matching incoming dial peer and outgoing dial peer did not generate the INVITE to the SIP provider. Why is the call failing?

- A. The ITSP is sending FIN.
- B. The ITSP is not sending the ACK.
- C. The ITSP is sending CLOSE state.
- D. The ITSP is not answering the SYN.

Correct Answer: D

## QUESTION 15

```
46282041.005 |09:18:16.331 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=3,
PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) kbps=8, capACount=1, capBCount=7

46282041.006 |09:18:16.331 |AppInfo |DET-MediaManager-(1698821)::checkAudioPassThru,
param(bPostMTPAllocation=0,chkTrp=1), capCount(1,7), mtpPT=1, aPT=2

46282041.007 |09:18:16.331 |AppInfo |DET-MediaManager-(1698821)::preCheckCapabilities,
region1=RTP_Reg, region2=SJ_Reg, Pty1 capCount=1 (Cap,ptime)=(4,20), Pty2 capCount=7 (Cap,ptime)=
(4,20) (2,20) (6,20) (11,20) (12,20) (15,20) (16,20)

46282041.008 |09:18:16.331 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=0,
PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) kbps=8, capACount=1, capBCount=7

46282041.009 |09:18:16.331 |AppInfo |RegionsServer: applyCodecFilterIfNeeded - no codecs remained
after filtering so restored original 0 caps
```

Refer to the exhibit. All calls from site A to site B are failing, and the issue has been identified as a media negotiation problem. Which configuration change resolves this issue?

- A. Increase the bandwidth allowance between the RTP\_Reg and SJ\_Reg regions to 64 kbps.
- B. Enable Early Offer on the SIP trunk.
- C. Create a new audio codec preference list with G.711 U-law 64k as the highest priority and apply it to RTP\_Reg and SJ\_Reg.
- D. Disable G.722 on all devices at both sites.

Correct Answer: C

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