300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

Some users report having issues dialing some external numbers when traveling to other locations within the company. The company has five locations in five cities in one country and has an egress gateway in each location for TEHO. The configuration has no specific entry stating that the roaming users are using the local gateway, but calls are going out. How is a verification of the call routing in such a specific configuration performed to further identify the problem?

A. device mobility

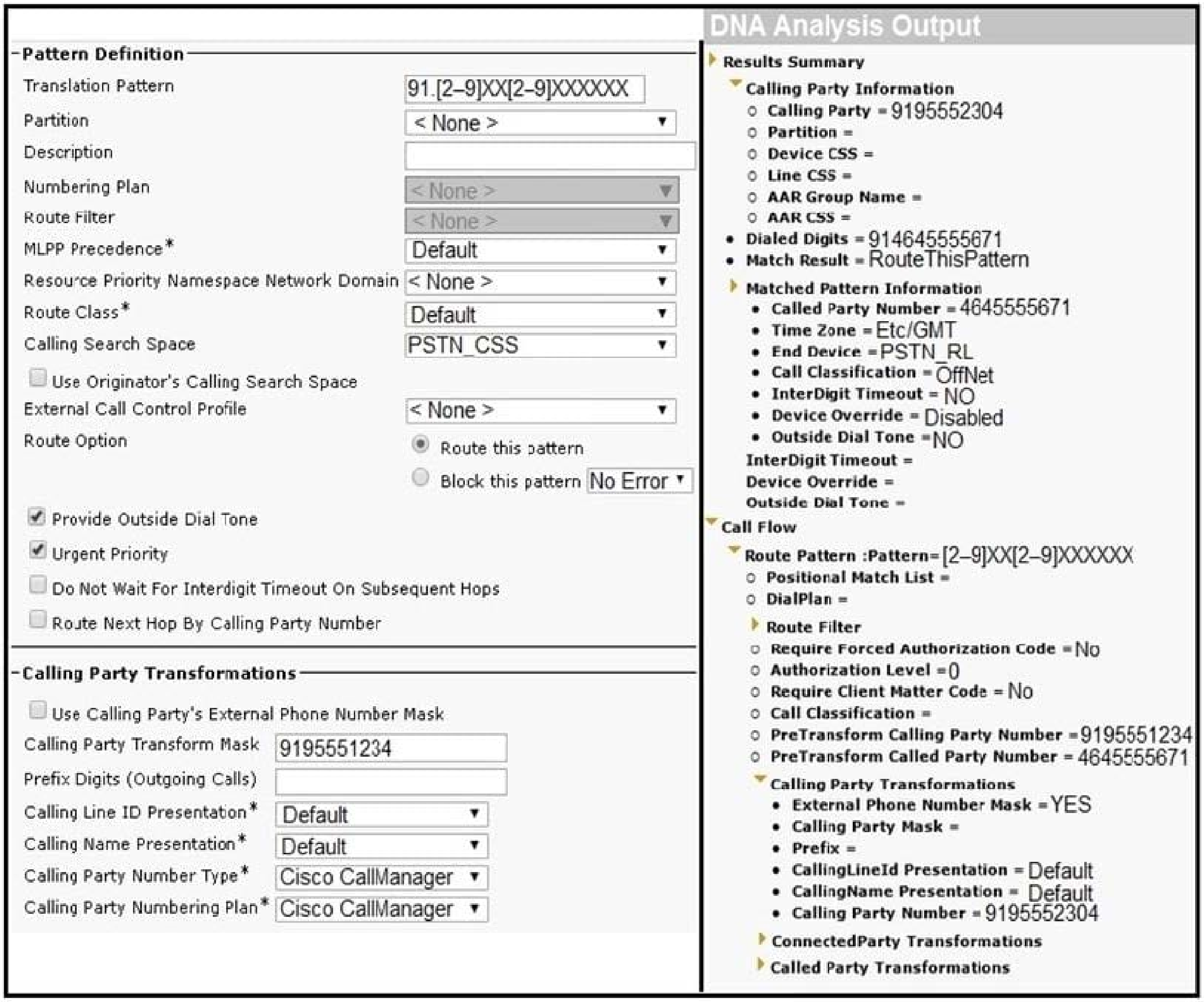
B. standard local route group

C. local route groups

D. TEHO

Correct Answer: A

QUESTION 2



Refer to the exhibit. For long-distance calls, users must prefix their dialed number with "91". The translation pattern was created to strip the 91 as the PSTN expects a 10-digit number. The PSTN also requires the calling number to be set to 9195551234. However, the service provider has said calls with a different calling number are being received. How is this issue resolved?

A. Change the partition of the translation pattern from none to pstn_pt.

B. Disable Use Calling Party\\'s External Phone Number Mask on the route pattern.

C. Enable Force Authorization Code on the route pattern.

D. Enable Use Calling Party\\'s External Phone Number Mask on the translation pattern.

Correct Answer: B

QUESTION 3

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. broadcast

B. top down

C. longest idle time

D. circular

Correct Answer: C

Reference: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcfg/b03lngrp.html

QUESTION 4

Which two extended capabilities must be configured on dial peers for fast start-to-early media scenarios (H.323 to SIP interworking)? (Choose two.)

A. DTMF

B. BFCP

C. VIDEO

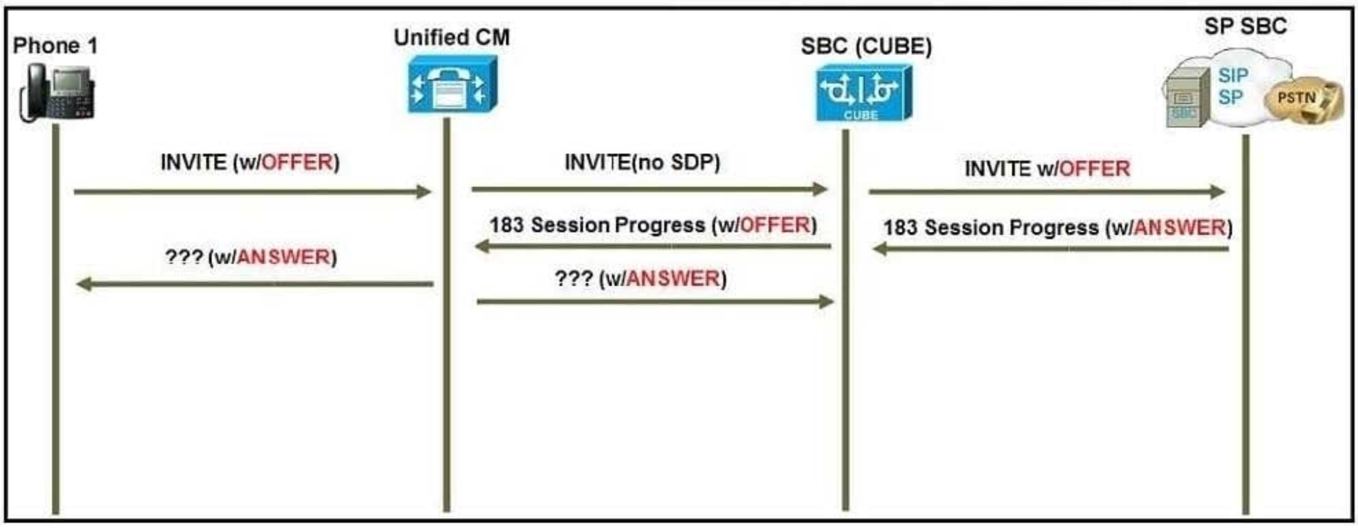
D. FAX

E. AUDIO

Correct Answer: AB

QUESTION 5

Refer to the exhibit.



A user reports that when they call a specific phone number, no one answers the call, but when they call from a mobile phone, the call is answered. The engineer troubleshooting the issue is expecting the far-end gateway to cut through audio on the 183 Session Progress SIP message. Which SIP Profile configuration element is necessary for the Cisco Unified Communications Manager to send acknowledgement of provisional responses?

A. Allow Passthrough of Configured Line Device Caller Information must be enabled.

B. Accept Audio Codec Preferences in Received Offer must be set to On.

C. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all 1xx Messages.

D. Early Offer for G Clear Calls must be enabled.

Correct Answer: C

QUESTION 6

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor incoming command configured? (Choose two.)

A. "voice register pool " configuration mode

B. "ephone-dn " configuration mode

C. "dial-peer cor custom " configuration mode

D. "voice register global " configuration mode

E. "telephony-service " configuration mode

Correct Answer: AB

QUESTION 7

Which section under the Real-Time Monitoring Tool allows for reviewing the call flow and signaling for a SIP call in real time?

A. Analysis Manager > Inventory > Trace File Repositories

B. System > Tools > Trace and Log Central

C. Voice/Video > Session Trace Log View > Real Time Data

D. Voice/Video > Session Trace Log View > Open From Local Disk

Correct Answer: C

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-managercallmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.html

QUESTION 8

In Cisco UCM, which tool is used to check SIP traces?

A. MTP

B. CCSIP

C. RTMT

D. OS Administration Page

Correct Answer: C

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 two debugs should the Administrator turn on? (Choose two.)

A. debug H.245 asn1

B. debug H.323 message

C. debug H.225 asn1

D. debug H.225 media

E. debug H.323 asn1

Correct Answer: AC

QUESTION 10

An administrator is implementing a new dial-plan on Cisco Unified Border Element. The administrator must ensure that incoming dial-peers are matched based on the IP address from where the incoming request originates. Which dial-peer configuration should be applied to accomplish this requirement?

A. dial-peer voice 1 voip incoming uri to

B. dial-peer voice 1 voip incoming called-number

C. dial-peer voice 1 voip incoming uri via

D. dial-peer voice 1 voip incoming uri request

Correct Answer: C

QUESTION 11

A company is using Cisco Jabber on-premises to make B2B calls on video. The calls are using Cisco Expressway-C and Expressway-E and have been configured in Cisco UCM to be able to call any URI on the internet. The Jabber client also

has voice enabled and must be able to call local, regional, and international numbers.

Where must Cisco UCM be configured to meet this requirement for URI dialing?

A. Enter “!#” in the SIP route pattern.

B. Enter “.*” in the route pattern section tied to a route group and list.

C. Enter “*” in the SIP route pattern.

D. Enter “!#” in the route pattern section tied to a route group and list.

Correct Answer: B

QUESTION 12

An engineer is configuring a Cisco Collaboration system for SIP endpoints and must enable Survivable Remote Site Telephony for these endpoints. Which code completes this configuration on the SRST gateway?

A. call-manager-fallback max-conferences 8 gain –6 ip source-address 10.10.10.100 port 2000 max-ephones 100 max-dn 200

B. telephony-service max-conferences 8 gain –6 ip source-address 10.10.10.100 port 2000 max-ephones 100 max-dn 200

C. voice service voip default mode secure address hiding allow-connections sip to sip sip registrar

D. voice register global default mode no allow-hash-in-dn max-dn 100 max-pool 200

Correct Answer: A

QUESTION 13

A user reports that when they attempt to log out from the Cisco Extension Mobility service by pressing the services button, they cannot log out. What is cause of this issue?

A. The Cisco Extension Mobility service has not been configured on the phone.

B. Network latency should be checked since there might be a significant delay between the button being pressed and it being recognized by the Cisco Extension Mobility service.

C. The user device profile has not been assigned to the user.

D. The user device profile is not subscribed to the Cisco Extension Mobility service.

Correct Answer: D

QUESTION 14

Refer to the exhibit.

55697959.007 |12:20:50.913 |AppInfo |RouteListCdrc::createPartyTransformedCcSetupReqMsg - before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222 crCdpnMask=33334444 crPrefixDigit=

crDDI=2

55697959.008 |12:20:50.913 |AppInfo |RouteListCdrc::createPartyTransformedCcSetupReqMsg - after DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222 55697959.009 |12:20:50.913 |AppInfo |

RouteListCdrc::transformed cdpn (without unconsumpt digits) = 33334444, unconsumed digit=

Which INVITE is sent to 10.10.100.123 as a result of this log?

A. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:33334444@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "1000" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM12.0

B. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:33334444@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "11112222" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM12.0

C. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:11112222@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "1000" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM12.0

D. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41 [95992364,NET] INVITE sip:11112222@10.10.100.123:5060 SIP/2.0 Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae From: "11112222" ;tag=32412716~41f7 To: Date: Thu, 01 Apr 2021 17:20:50 GMT Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50 Supported: timer,resource-priority,replaces Min-SE: 1800 User-Agent: Cisco-CUCM12.0

Correct Answer: C

QUESTION 15

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