300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

<sip: 1155@10.2.2.13>;privacy=off;reason=unconditional;counter=1;screen=no

and from the Cisco CUBE the logs show :

<sip:1155@10.3.3.25>;privacy=off;reason=unconditional;counter=1;screen=no

Refer to the exhibit. Users report that outgoing calls do not work on the new SIP trunk for outgoing calls. The solution consists of a Cisco UCM Cluster linked to a Cisco Unified Border Element where the SIP trunk is terminated. The provider required 10 digits. The logs show a line going toward the Cisco Unified Border Element. Which code snippet must be added to the configuration to meet the requirement?

A. request Invite sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the SIP translation profile configuration

B. request Invite sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the voice translation profile configuration

C. sip-header modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the voice translation profile configuration

D. request Invite sip-header Diversion modify “andamp;lt;sip:1(…)@” “andamp;lt;sip:9135551\1@” under the SIP profile configuration

Correct Answer: B

QUESTION 2

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

QUESTION 3

```
02904115.001 |09:08:07.093 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.1.102 on
port 50244 index 22157
[166156,NET]
ACK sip:1315932e-ff29-4203-23e3-ef940216638e@10.1.1.102:50244;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.1.5:5060;branch=z9hG4bKb4fa1fa89d7e
From: <sip:1001@10.1.1.5>;tag=93016~bb788fb3-a1ef-4d03-a96d-651038e22050-28377951
To: <sip:1000@cucm1251.cisco.lab>;tag=7001b5dab46425b45ec6648a-25dc4f91
Date: Wed, 28 Jul 2021 13:08:07 GMT
Call-ID: cee27300-1ed10da7-b403-3251300e@10.1.1.5
User-Agent: Cisco-CUCM12.5
Max-Forwards: 70
CSeq: 103 ACK
Allow-Events: presence
Session-ID: 36ed016300105000a0002834a2824611;remote=49f3b76a00105000a0007001b5dab464
Content-Type: application/sdp
Content-Length: 412

v=0
o=CiscoSystemsCCM-SIP 93016 3 IN IP4 10.1.1.5
s=SIP Call
c=IN IP4 10.1.1.5
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
b=TIAS:64000
a=ptime:20
a=rtpmap:0 PCMU/8000
a=inactive
```

Refer to the exhibit. This message is sent to the device being placed on hold for the Music On Hold audio setup. The held party reports receiving dead air rather than music when the call was put on hold. The software Music On Hold server on Cisco UCM is used in this scenario. Assume that the audio leg between the Music On Hold server and the held device uses G.711, and the relevant region relationship is configured for 64 kbps. What is the cause of the issue?

A. The bandwidth configured for this region relationship is too low and must be increased to 96 kbps or higher.

B. The device that is placed on hold does not support G.711, and a transcoder could not be allocated for the call.

C. Cisco UCM is sending a=inactive to the held device.

D. The Music On Hold server does not support G.711 and a transcoder could not be allocated for the call.

Correct Answer: C

QUESTION 4

After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

A. Router(config)# voice service voip Router(conf-voi-serv)#allow-connections h323 to h323

B. Router(config)#dial-peer voice 2 voip Router(config-dial-peer)#no vad

C. Router(config)# voice service voip Router(conf-voi-serv)#allow-connections voice-mail mod

D. Router(config)# voice service voip Router(conf-voi-serv)#no supplementary-service sip moved-temporarily

Correct Answer: D

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_call_handling.html

QUESTION 5

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