300-815^Q&As^

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

Which action is correct with respect to toll fraud prevention configuration in the Cisco Unified Communications Manager Express?

A. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.

B. Configure IP Address Trusted Authentication for Incoming VoIP Calls.

C. Configure the command no ip address trusted authenticate under "voice service voip".

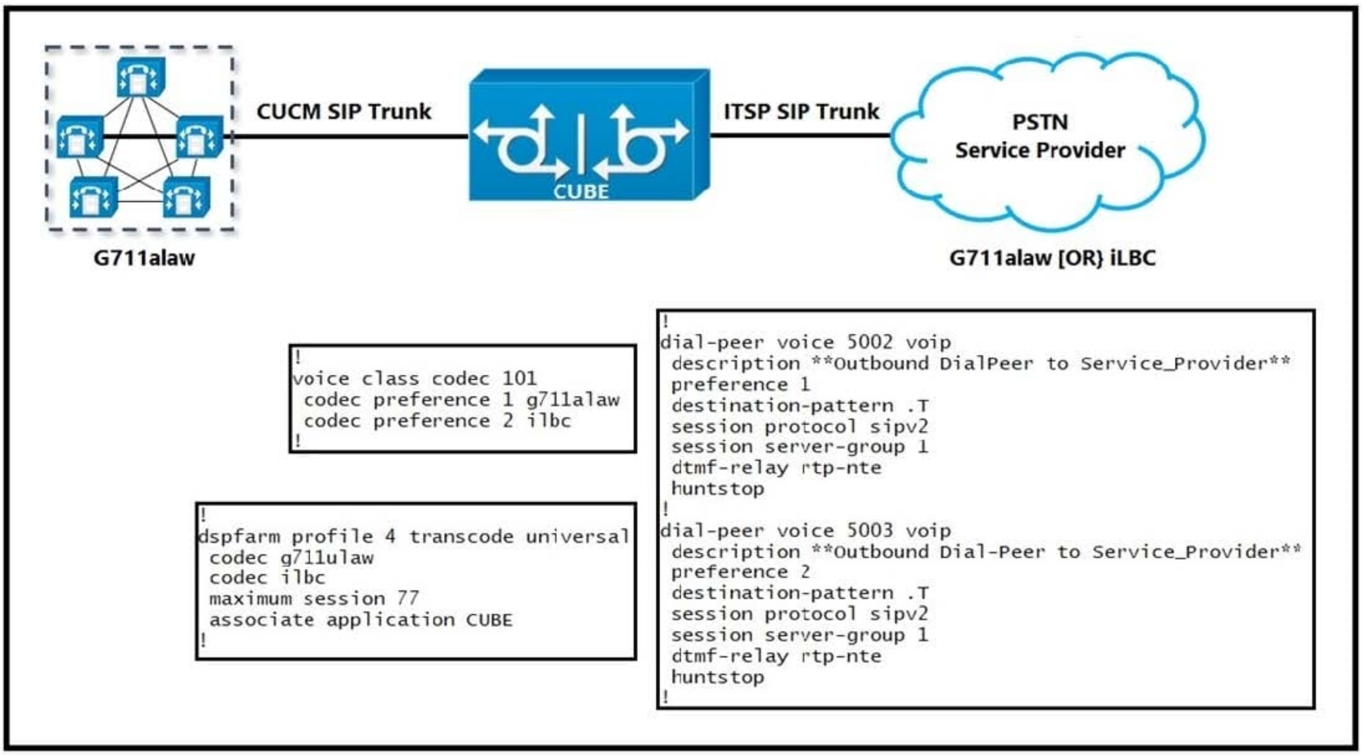
D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Correct Answer: B

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

QUESTION 2

Refer to the exhibit.



Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time. Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?

A. dial-peer voice 5002 voip codec g711alaw ilbc ! dial-peer voice 5003 voip codec g711alaw ilbc

B. dial-peer voice 5002 voip voice-class codec 101 offer-all ! dial-peer voice 5003 voip voice-class codec 101 offer-all

C. dial-peer voice 5002 voip voice-class codec 101 ! dial-peer voice 5003 voip voice-class codec 101

D. dial-peer voice 5002 voip codec g711alaw ! dial-peer voice 5003 voip codec ilbc

Correct Answer: B

QUESTION 3

A user in location X dials an extension at location Y. The call travels through a QoS-enabled WAN network, but the user experiences choppy or clipped audio. What is the cause of this issue?

A. missing Call Admission Control

B. codec mismatch

C. ptime mismatch

D. phone class of service issue

Correct Answer: A

QUESTION 4

An administrator has configured two route patterns, 9.911 and 9.[2-9]XXXXXX. When a user dials 9911. Cisco UCM waits for the T302 timer before routing the call. How will the administrator force interdigit timeout and route the call as soon as the user has finished dialing 9911, without waiting for the T302 timer to expire?

A. decrease the T302 timer in Service Parameters from the default value

B. enable Urgent Priority on the 9.[2-9]XXXXXX pattern

C. enable Urgent Priority on the 9.911 pattern

D. enable Device Override on both route patterns

Correct Answer: C

QUESTION 5

An SRST site must use pickup functionality on the Chicago remote site to allow the users to take incoming calls. The pickup command is configured under the call manager-fallback section on the SRST router. What are two results of that configuration? (Choose two.)

A. Calls that ring an unassigned directory number are forwarded to the auto-attendant.

B. The PickUp soft key is on all SRST phones.

C. The GPickUp soft key is on all SRST phones.

D. Calls that come into one directory number can be picked up from another directory number.

E. Calls that come into a Cisco UCM registered phone can be picked up by a SRST phone.

Correct Answer: BD

QUESTION 6

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 7

How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?

A. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.

B. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.

C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.

D. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.

Correct Answer: D

QUESTION 8

A new solution is configured to support internal, local, and international calling. Calling [+44 1111 1111] from one of the registered internal phones does not work. Local and internal calls seem to work without any problems. The configuration has patterns configured to match the failing dialed number [+44 …. ….]. The other configured patterns show [2…] for internal numbers and [555 …] for local numbers. International numbers use E.164 as recommended. What is missing to make this solution work?

A. 001 or 00 must be used instead of the + sign on Cisco UCM

B. =+ cannot be used in a route pattern, only in a SIP pattern

C. \ in front of the +

D. / in front of the +

Correct Answer: C

QUESTION 9

Calls to a particular extension are not routing to voicemail. The user reaches the voicemail system from the handset by pressing the Messages button. Which configuration parameter causes this problem?

A. The voicemail pilot number for call forwarding is missing from the ephone-dn.

B. The voicemail pilot number is missing from the telephony service configuration on Cisco UCME.

C. The voicemail pilot number is missing from the call handling on Cisco Unity Express.

D. The voicemail pilot number for call forwarding is missing from the ephone.

Correct Answer: A

QUESTION 10

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

A. in the telephony service configuration

B. in the queuing configuration

C. in Cisco Unified CM Enterprise Parameters

D. in Cisco Unified CM Service Parameters

Correct Answer: B

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200453-Configure-CUCM-Native-Call-Queuing-Featu.html

QUESTION 11

```
!
dial-peer voice 101 voip
 description Inbound to CUCM
 destination-pattern 1...
 session target ipv4: 10.1.1.1
 session protocol sipv2
 codec g711ulaw
 no vad
!
```

Refer to the exhibit. When setting up a new connection to Cisco UCM, the engineer must use out-of-band DTMF. Which configuration meets this requirement?

A. dtmf-relay h245-alphanumeric

B. dtmf-relay sip-kpml

C. dtmf-relay cisco-rtp

D. dtmf-relay cisco-rtp

Correct Answer: B

QUESTION 12

Refer to the exhibit.

```
SIP/2.0 200 OK
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=-IN IP4 10.10.10.11
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp: 110 0-16
a-ptimc: 20

ACK sip:+123456789@10.10.20.20:5060 SIP/2.0
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=ptime:20
```

Users report that when they dial to Cisco Unity Connection from an external network, they cannot enter any digits. Assuming only in-band DTMF is supported, what is a reason for this malfunction?

A. The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.

B. There is SIP Delayed Offer. DTMF is supported only in Early Offer.

C. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.

D. No DTMF is negotiated.

Correct Answer: D

QUESTION 13

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 an access list.

B. Configure Single Number Reach.

C. Configure Mobile Voice Access.

D. Configure a mobility identity.

Correct Answer: C

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configurationguide_chapter_010.html

QUESTION 14

```
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

QUESTION 15

CollabCorp is a global company with two clusters, emea.collab.corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI, but this is not working as expected. Which two configuration elements should be checked to resolve this issue? (Choose two.)

A. intercluster trunk

B. directory URI partition

C. SIP route pattern

D. calling search space and partition

E. SIP trunk

Correct Answer: CE