300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

An administrator is trying to apply configuration changes on Cisco CME. When the users registered on Cisco CME to dial a local number to a PSTN call, the Cisco CME sends an incorrect number of digits. What translation rule fixes the issue and sends the correct number of digits?

A. voice translation-rule 1 rule 1 // // type any subscriber plan any isdn

B. voice translation-rule 1 rule 1 /^4...$/ /2404\0/ type any national plan any isdn

C. voice translation-rule 1 rule 1 /^4...$/ /9132404\0/ type any subscriber plan any isdn

D. voice translation-rule 1 rule 1 /^4...$/ /2404\0/ type any subscriber plan any isdn

Correct Answer: C

QUESTION 2

```
interface GigabitEthernet0/0/0
 description to CUCM
 ip address 10.10.150.1 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description to ITSP
 ip address 192.168.10.78 255.255.255.0
 negotiation auto
!
dial-peer voice 100 voip
 incoming called-number 8005532447
 session protocol sipv2
 codes g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 200 voip
 destination-pattern 8005532447
 session target ipv4:192.168.10.100
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 300 voip
 answer-address 8005532447
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
```

```
Received:
INVITE sip:8005532447010.10.150.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.150.11:5060;branch-z9hG4bK1046
From: <sip:1001010.10.150.11>;tag-23125042-8a7bedal-f
To: "CISCO SYSTEMS" <sip:8005532447010.10.150.1>;tag=
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02010.10.15
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK
CSeq: 103 INVITE
[..Omitted for brevity..]
Contact: <sip:1001010.10.150.11:5060>;
Content-Type: application/adp
Content-Length: 235

v=0
o=CiscoSystemsCCM-SIP 23125042 1 IN IP4 10.10.150.11
s=SIP Call
c=IN IP4 10.10.2.254
b=TIAS:64000
b=AS:64
t=0 0
m=audio 35023 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

  Calling Number=1001,(Calling Name-) (TON-Unknown, N
  Called Number-8005532447(TON-Unknown, NPI-Unknown),
  Calling Translated-FALSE, Subscriber Type Str-Unkno
  Incoming Dial-peer-100, Progress Indication-NULL(0)
```

Refer to the exhibit. An engineer is troubleshooting a call-establishment problem between Cisco Unified Border Element and Cisco UCM. Which command set corrects the issue?

A. SIP binding in SIP configuration mode: voice service voip sip bind control source-interface GigabitEthernet0/0/1 bind media source-interface GigabitEthernet0/0/1

B. SIP binding in dial-peer configuration mode: dial-peer voice 100 voip voice-class sip bind control source-interface

GigabitEthernet0/0/0 voice-class sip bind media source-interface GigabitEthernet0/0/0

C. SIP binding in dial-peer configuration mode: dial-peer voice 300 voip voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1

D. SIP binding in SIP configuration mode: voice service voip sip bind control source-interface GigabitEthernet0/0/0 bind media source-interface GigabitEthernet0/0/0

Correct Answer: B

QUESTION 3

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. Add the long-distance and international pattern\\'s partitions to the calling search space of the physical phone.

B. Remove the long-distance and international pattern\\'s partitions from the calling search space of the physical phone\\'s directory number.

C. Add the long-distance and international pattern\\'s partitions to the calling search space of the device phone.

D. Add the long-distance and international pattern\\'s partitions to the calling search space of the physical phone\\'s directory number.

E. Remove long-distance and international pattern\\'s partitions from the calling search space of the device phone.

Correct Answer: BE

QUESTION 4

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

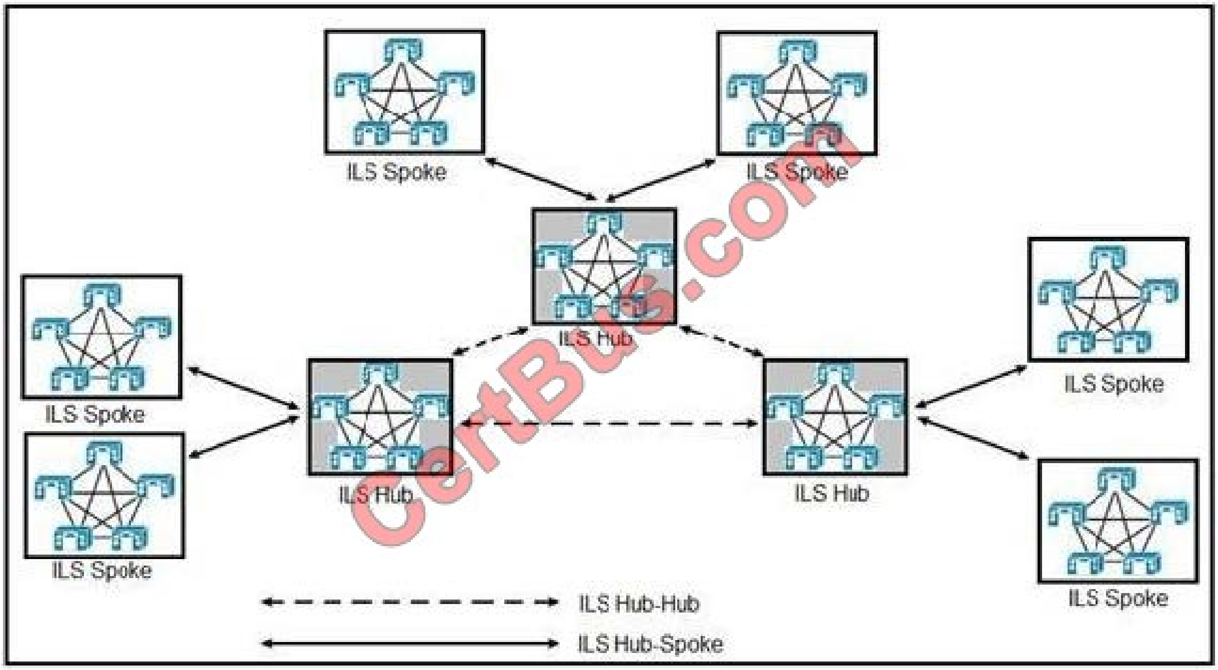
B. dial-peer voice 1 voip incoming called-number

C. dial-peer voice 1 voip incoming uri via

D. dial-peer voice 1 voip incoming uri request

Correct Answer: C

QUESTION 5



Refer to the exhibit. How many maximum hops does an ILS update traverse?

A. 3

B. 6

C. 9

D. 12

Correct Answer: A

QUESTION 6

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. H.323 messages

B. H.225 asn1

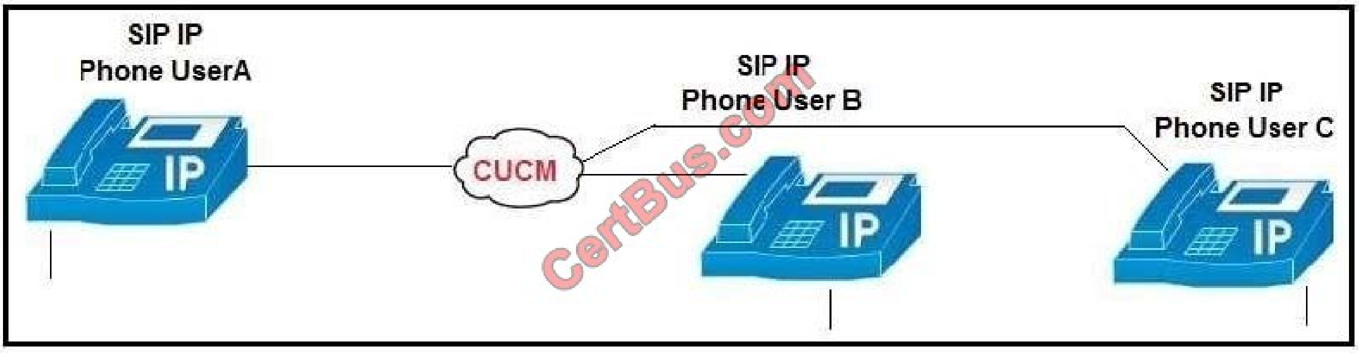
C. H.245 asn1

D. H.225 media

E. H.323 asn1

Correct Answer: BC

QUESTION 7



Refer to the exhibit. In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. What are two results from this action? (Choose two.)

A. Phone_A sends a SIP-REFER message to the Cisco UCM with Phone_C information in the Refer-To section.

B. Phone_B sends a SIP-REFER message to the Cisco UCM with Phone_C information in the Refer-To section.

C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH, and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.

D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold, and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.

E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH, and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Correct Answer: AC

QUESTION 8

```
voice hunt-group 1
 phone-display
 final 7777
 list 1002,1003,1005,1006,1010
 hops 3
 pilot 2222
```

Refer to the exhibit. DN 1003 was the last to ring during the most recent call. Which hunting method ensures that DN 1005 is presented with the next call when the hunt pilot is dialed?

A. sequential

B. call-blast

C. peer

D. parallel

Correct Answer: A

QUESTION 9

Single Number Reach calls to a cell phone that not answered are leaving voicemails on the cell phone rather than the corporate mailbox. Which two options will resolve this issue? (Choose two.)

A. Check the Enable Extend and Connect checkbox.

B. Check the Enable Unified Mobility features checkbox.

C. Decrease the T302 timer.

D. Decrease the T301 timer.

E. Decrease the Answer Too Late timer.

Correct Answer: BE

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200447-Single-Number-Reach-Feature-for-Cisco-Un.pdf

QUESTION 10

Cisco Extension Mobility does not show up when the services button is pressed. What is the cause of the issue?

A. The URL configured for Cisco Extension Mobility is not correct.

B. Cisco Extension Mobility Service is not running.

C. The phone is not subscribed to Cisco Extension Mobility Service.

D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

Correct Answer: C

QUESTION 11

What are two configuration features of the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending since the Cisco UCM cannot determine when to prompt the user for the code.

B. Selecting the Allow Overlap Sending setting disables the Require Client Matter Code setting.

C. Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.

D. The Client Matter Code feature supports overlap sending since the Cisco UCM can determine when to prompt the user for the code.

E. The Client Matter Code feature provides the option to configure Authorization Level such as in the Forced Authorization Code.

Correct Answer: AB

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide100_chapter_010000.pdf

QUESTION 12

A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

A. three-way conference

B. secure SIP lines

C. T.38 fax relay

D. transcoding

E. SIP trunk

Correct Answer: AC

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_sip_isr4000.html

QUESTION 13

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 14

A customer is using a SIP trunk to route calls to ITSP. To decrease the possibility of downtime, the customer invested in a failover device. How does the customer ensure reachability to ITSP, so that if one device on ITSP fails, the calls will be routed to another device?

A. Enable SIP Option Ping on the SIP profile.

B. Monitor the link using network management tools, and if it fails, manually change the routing to another working device.

C. Enable ANAT on the SIP profile.

D. Enable transmit security status on the SIP security profile.

Correct Answer: A

QUESTION 15

```
voice translation-rule 84
        rule 1 /^\([2-9]..[2-9]......$\)/ /\2/
```

Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco UCM are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco UCM sends the calls to the Cisco Unified Border Element router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied. Which rule modifies DNIS in the format that the provider is expecting?

A. rule 1 /^\+\([^1].*\)/ /011\1/

B. rule 1/^\+1\([2-9]..[2-9]......$\)/ /\1/

C. rule 1 /^\([2-9]..[2-9]......$\)/ /\1/

D. rule 1 /^\+1\([2-9]..[2-9]......$\)/ /\0/

Correct Answer: B

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