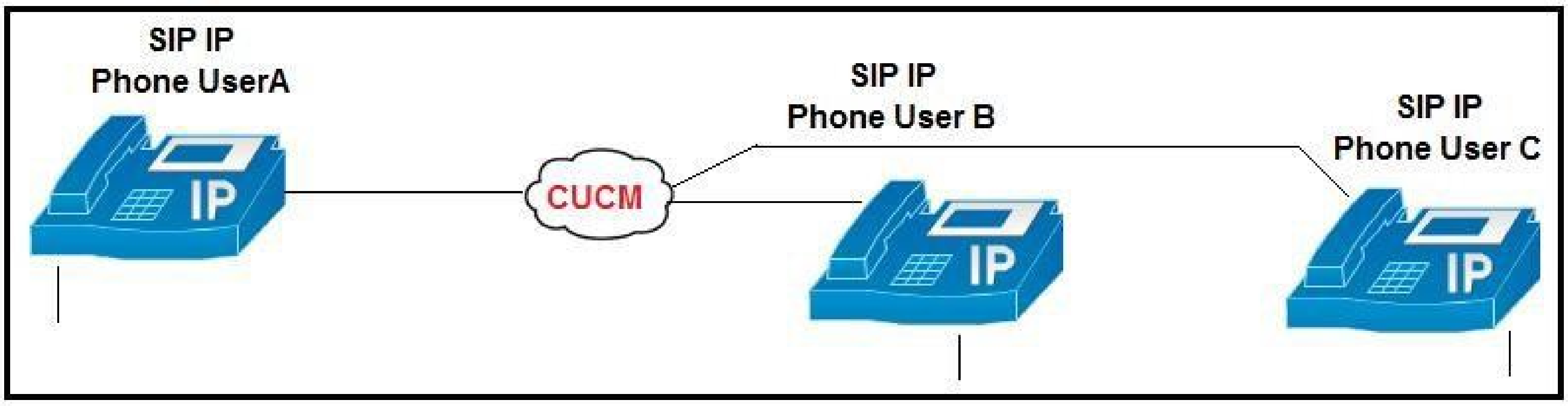
Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

NEW QUESTION 1



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. Which two scenarios are correct? (Choose two.)

A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
B. Phone_B sends a SIP-REFER message to the Cisco Unified CM 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.

Answer: AC

NEW QUESTION 2

```
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=ptime: 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
```

Refer to the exhibit. 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 Offe
C. DTMF is supported only in Early Offer.
D. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.
E. No DTMF is negotiated.

Answer: D

NEW QUESTION 3
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

Answer: AB

NEW QUESTION 4

What is first preference condition matched in a SIP-enabled incoming dial peer?

A. incoming uri
B. target carrier-id
C. answer-address
D. incoming called-number

Answer: A

NEW QUESTION 5
Which description of RTP timestamps or sequence numbers is true?

A. The sequence number is used to detect losses.
B. Timestamps increase by the time “carrying” by a packet.
C. Sequence numbers increase by four for each RTP packet transmitted.
D. The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation).

Answer: D

NEW QUESTION 6
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

Answer: AC

NEW QUESTION 7
You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?

A. configuration for a single SIP phone
B. configuration items common for all SIP phones
C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)
D. configuration for SIP registrar service

Answer: C

NEW QUESTION 8
For s SIP to SIP call flow, when does Cisco Unified Border Element require transcoding resources for DTMF?

A. interworking between an OOB method and RFC2833 for flow-around calls
B. interworking between h245-signal and rtp-nte
C. interworking between an OOB method and RFC2833 for flow-through calls
D. interworking between h245-alpha numeric and sip-kpml

Answer: A

NEW QUESTION 9
Where is the dtmf-relay command configured on Cisco Unified Border Element?

A. in the voice-class VoIP configuration
B. in the VoIP dial peer
C. in global SIP configuration
D. in the VoIP or POTS dial peers

Answer: B

NEW QUESTION 10

```
voice translation-profile incoming
     translate called 999
!
voice translation-rule 999
     rule 1/\ (^[1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\) $ /  / \2333\1/
!
dial-peer voice 999 voip
     translation-profile outgoing incoming
     session protocol sipv2
     incoming called-number
     dtmf-relay rtp-nte
     codec transparent
     destination dpg 888
     no vad
!
voice class dpg 888
     dial-peer 888
!
dial-peer voice 888 voip
     destination-pattern 888
     session protocol sipv2
     session target ipv4:192.168.0.1
     codec transparent
     dtmf-relay rtp-nte
     no vad
```

Refer to the exhibit. Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number “222333444” and Cisco Unified Communications Manager is expecting the called number to be delivered as “444333222”. The administrator already verified that the IP address of the Cisco Unified CM is set up correctly and there are no dial peers configured other than those shown in the exhibit. Which action must the administrator take to fix the issue?

A. Change the destination-pattern on the outgoing dial peer to match “444333222”.
B. Set up translation-profile on the incoming dial peer to match incoming traffic.
C. Create specific matching for “222333444” on the incoming dial peer.
D. Fix the voice translation-rule to match specifically number “222333444” and change it to “444333222”.

Answer: B

NEW QUESTION 10
Which IOS command creates a SIP- enabled dial peer?

A. voice dial-peer 20 sip
B. dial-peer voice 20 voip
C. dial-peer voice 20 pots
D. dial peer voice 20 sip

Answer: B

NEW QUESTION 15
Refer to the exhibit. An engineer configures Cisco Unified Border Element to connect the enterprise VoIP network with a SIP telephony provider. Calls are not working in either direction. What must be configured in the dial peer 1 to fix the issue?

A. address 555 ……..
B. codec g729
C. session-protocol sipv2
D. incoming called number 555…….

Answer: D

NEW QUESTION 20
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 voipRouter(conf-voi-serv)#allow-connections h323 to h323
B. Router(config)#dial-peer voice 2 voipRouter(config-dial-peer)#no vad
C. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections voice-mail mod
D. Router(config)# voice service voipRouter(conf-voi-serv)#no supplementary-service sip moved-temporarily

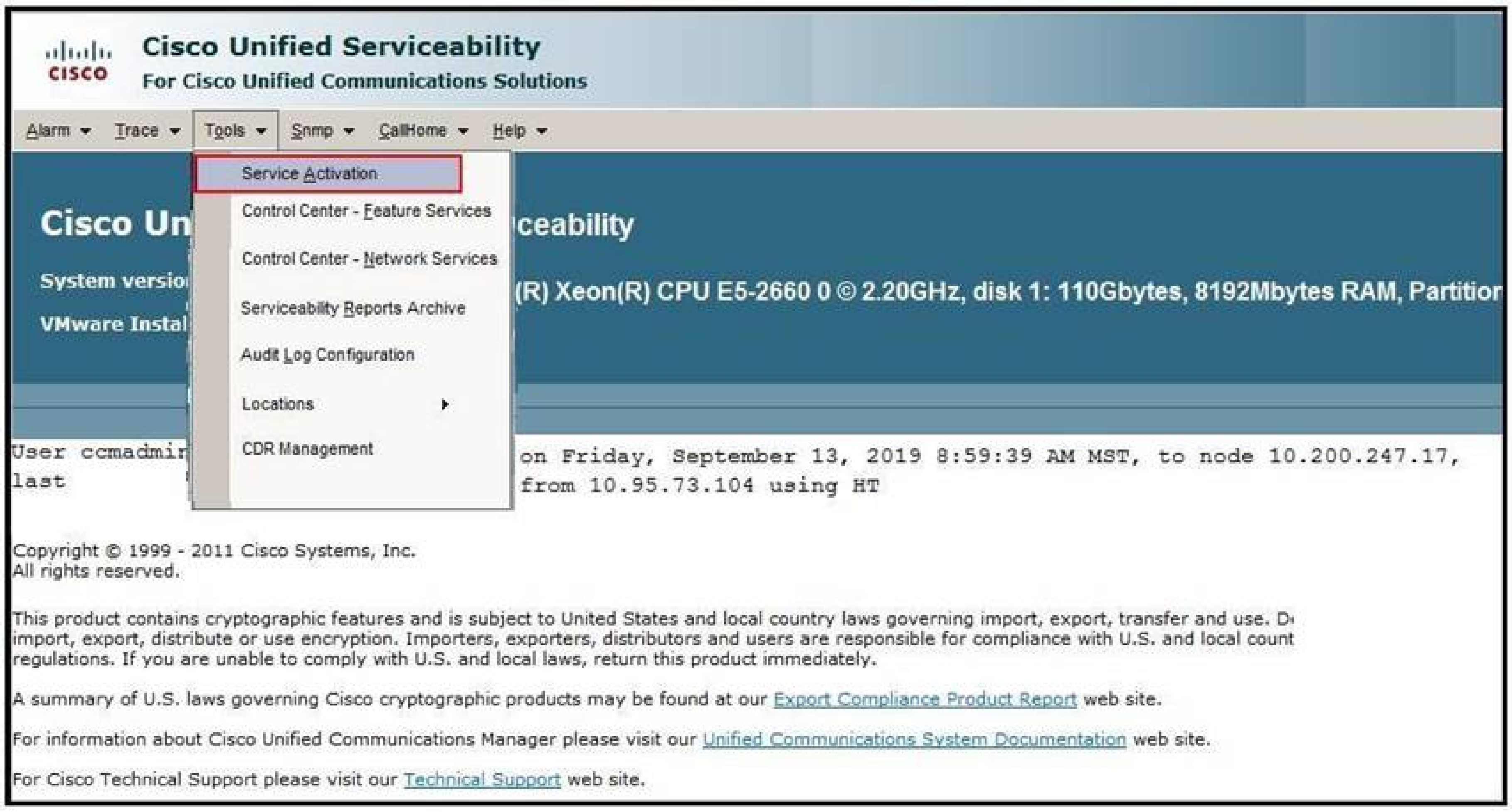
Answer: A

NEW QUESTION 25
Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

A. Enable the Detailed Call Analysis option under Enterprise Parameters for Unified CM.
B. Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis.
C. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page.
D. By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.

Answer: A

NEW QUESTION 29



Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)

A. Restart the subscriber
B. Activate the Cisco Extended Functions service.
C. Activate the Cisco CallManager service.
D. Activate the Cisco Dialed Number Analyzer service.
E. Activate the Cisco Dialed Number Analyzer Server service.

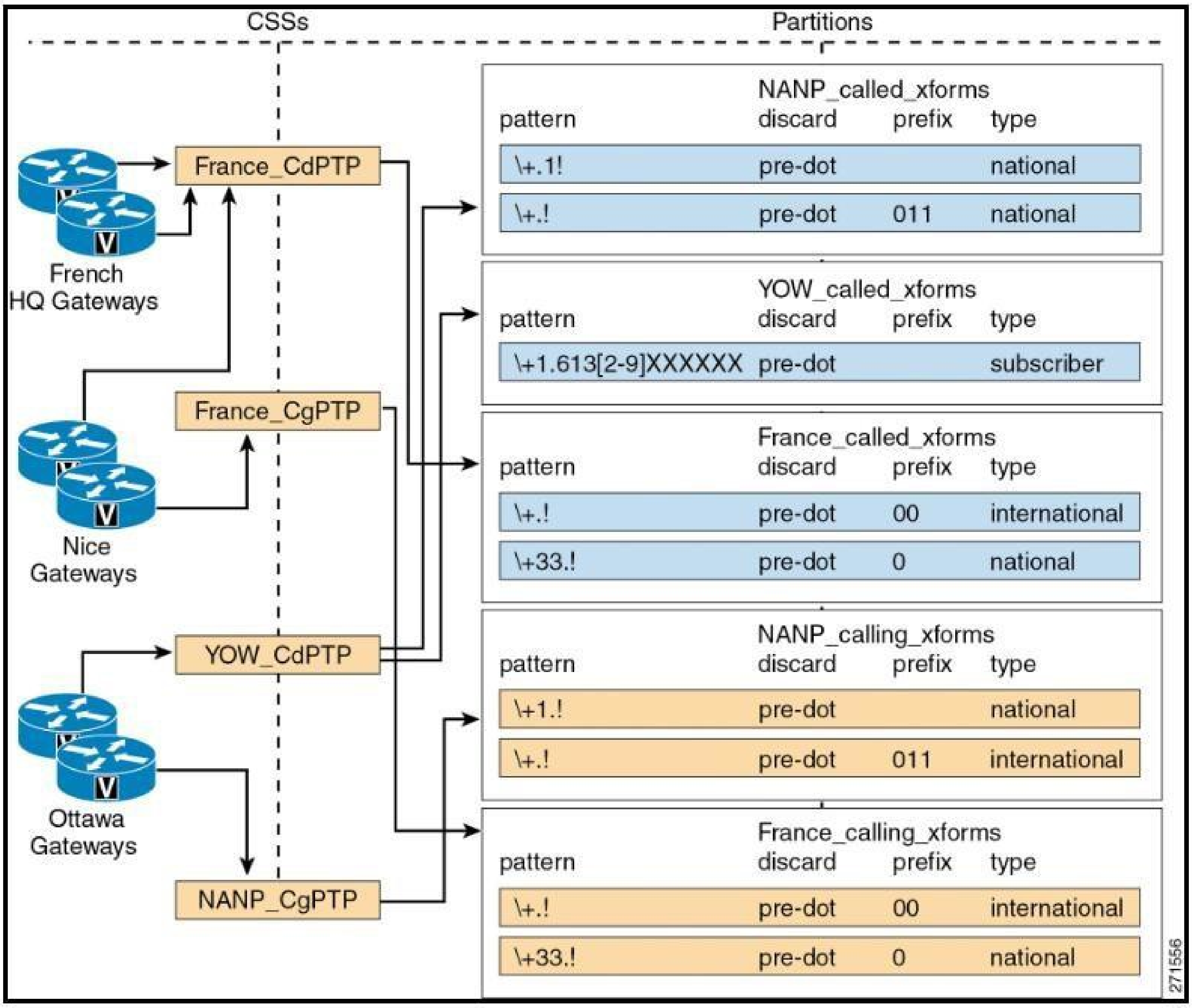
Answer: DE

NEW QUESTION 32
In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

A. Dialed Number Analyzer
B. Real-Time Monitoring Tool
C. SDI trace
D. SDL trace

Answer: A

NEW QUESTION 34



Refer to the exhibit. Within the North American Numbering Plan, gateways located in Ottawa, Canada and marked as "YOW" are assigned to the Calling Party Transformation CSS NANP_CgPTP, which contains partition NANP_calling_xforms. What is the calling-party number and the numbering type if the calling user +1613-555-1234 dials the number?

A. calling number 613-555-1234 and numbering type "subscriber"
B. calling number 011-1-613-555-1234 and numbering type "subscriber"
C. calling number 011613-555-1234 and numbering type "international"
D. calling number 613-555-1234 and numbering type "national"

Answer: D

NEW QUESTION 38
Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?

A. System > Location Info
B. Call Routing > Route/Hunt > Local Route Group Names
C. System > Device Pool
D. Call Routing > Emergency Location > Emergency Location (ELIN) Groups

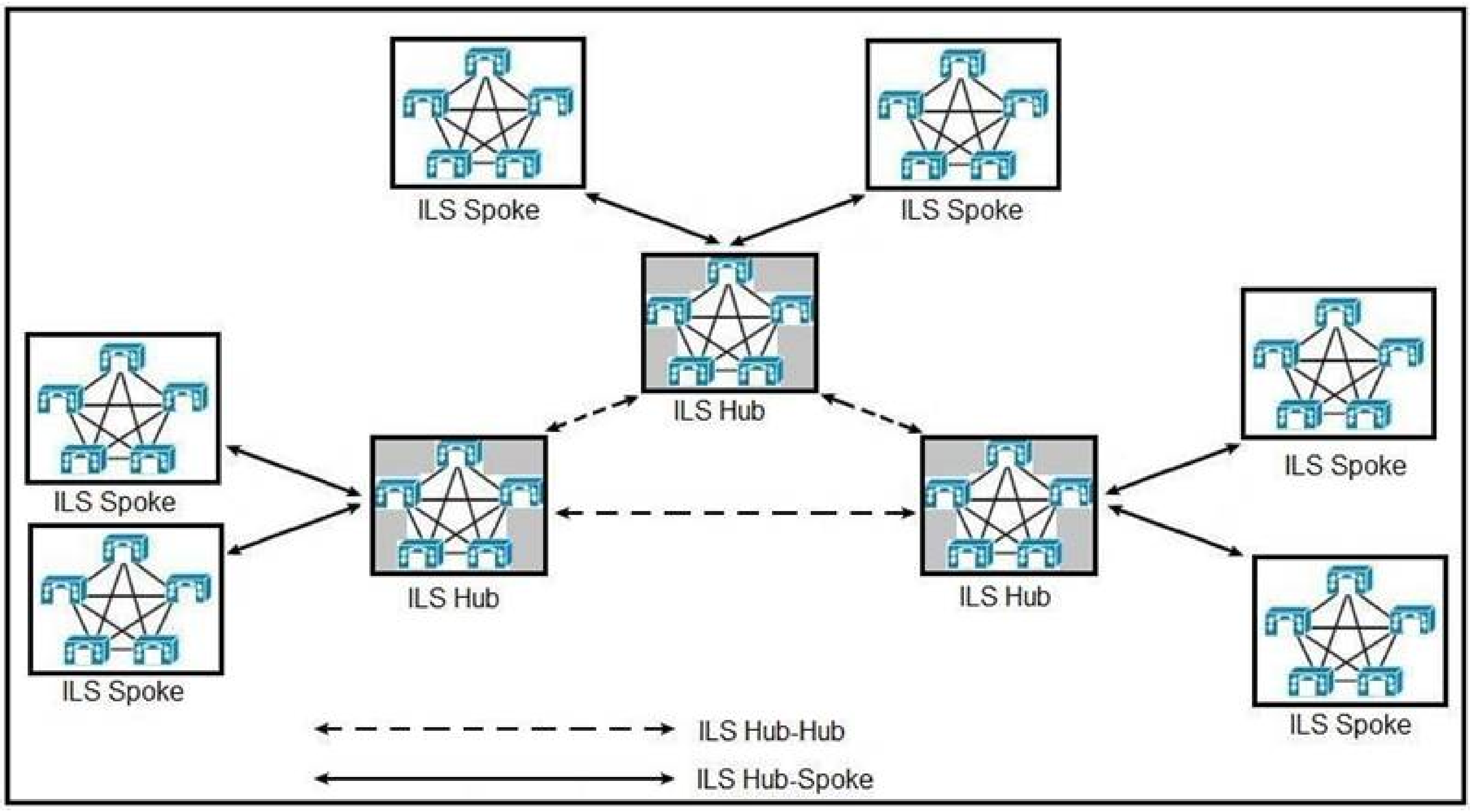
Answer: B

NEW QUESTION 43
Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?

A. Other Group Pickup
B. BLF Call Pickup
C. Group Call Pickup
D. Directed Call Pickup

Answer: A

NEW QUESTION 48



Refer to the exhibit. How many maximum hops can an ILS updarte traverse?

A. 3
B. 6
C. 9
D. 12

Answer: A

NEW QUESTION 49
What is a component of Cisco Unified Mobility?

A. Unified IVR
B. Mobile Connect
C. Smart Client Support
D. Single Number Connect

Answer: B

NEW QUESTION 50
A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?

A. Create the default device profiles for all phone models that are used.
B. Subscribe the phone to the Cisco Extension Mobility service.
C. Create the end user and associate it to the device profile.
D. Assign the extension as a mobile extension.

Answer: B

NEW QUESTION 51
What are the elements for Device Mobility configuration?

A. physical location, device pool, and Device Mobility group
B. device pool, Device Mobility group, and region
C. physical locatio
D. Device Mobility group, and region
E. device pool, Device Mobility group, and Cisco IP phone

Answer: A

NEW QUESTION 56
Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?

A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service
B. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service
C. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility
D. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility

Answer: C

NEW QUESTION 58
......

Thank You for Trying Our Product

We offer two products:

1st - We have Practice Tests Software with Actual Exam Questions

2nd - Questons and Answers in PDF Format

300-815 Practice Exam Features:

* 300-815 Questions and Answers Updated Frequently

* 300-815 Practice Questions Verified by Expert Senior Certified Staff

* 300-815 Most Realistic Questions that Guarantee you a Pass on Your FirstTry

* 300-815 Practice Test Questions in Multiple Choice Formats and Updatesfor 1 Year

100% Actual & Verified — Instant Download, Please Click
Order The 300-815 Practice Test Here