300-815.VCEplus.premium.exam.60q

Number: 300-815
Passing Score: 800
Time Limit: 120 min
File Version: 1.0

Website: https://vceplus.com
VCE to PDF Converter: https://vceplus.com/vce-to-pdf/
Facebook: https://www.facebook.com/VCE.For.All.VN/
Twitter : https://twitter.com/VCE_Plus

300-815

Implementing Cisco Advanced Call Control and Mobility Services

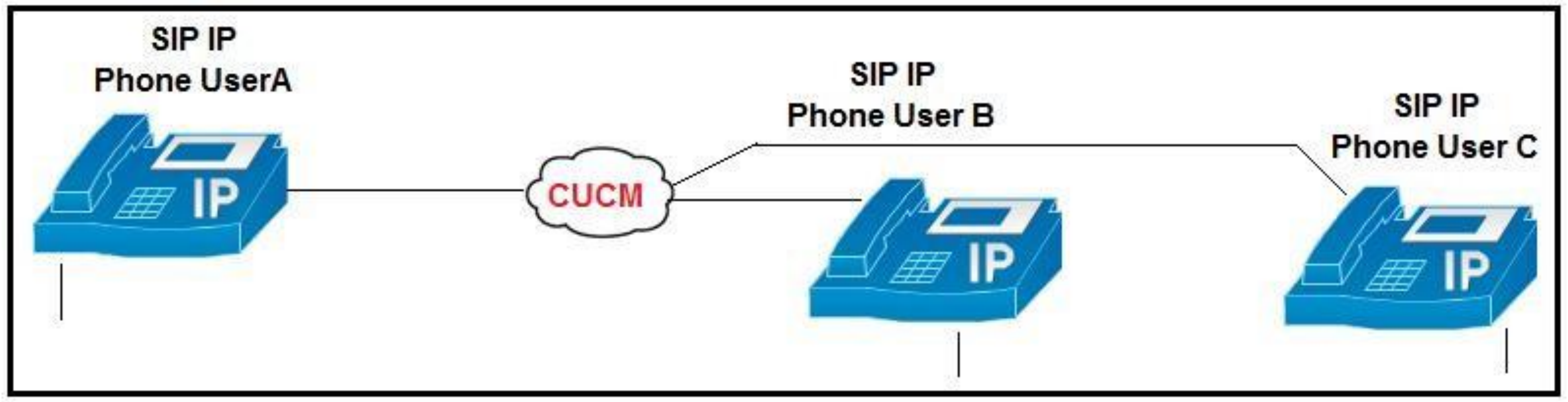
Version 1.0

Sections

1. Signaling and Media Protocols
2. CME/SRST Gateway Technologies
3. Cisco Unified Border Element
4. Call Control and Dial Planning
5. Cisco Unified CM Call Control Features
6. Mobility

Exam A

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.

Correct Answer: AC
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

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 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
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 3
The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call.

You gather the H.225 and H.245 messages for one of the one-way audio calls. Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).

A. H.245 Terminal Capability Set
B. H.245 Open Logical Channel
C. H.225 Connect

D. H.245 Open Logical Channel Ack

Correct Answer: B
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:
Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.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
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 5
When you troubleshoot H.323 call setup, which message informs you that the called party is being notified about the call?

A. ALERTING
B. PROCEEDING
C. CONNECT
D. RINGING

Correct Answer: C
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 6
End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

A. **Contact:** header of the 200 OK response
B. **Allow:** header if the 200 OK response
C. **o=** line of SDP content
D. **c=** line of SDP content

Correct Answer: C
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 7
Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?

A. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path.
B. Cisco Unified Communications Manager invoked media termination point resources.
C. The RTP traffic is arriving beyond the jitter buffer on the receiving end.
D. A firewall in the media path is blocking TCP ports 16384-32768.

Correct Answer: D
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 8 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
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 9 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

Correct Answer: A
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth-Explanation-of-Cisco-IOS-and-IO.html#anc8

QUESTION 10
Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is oneway audio or no audio. You determine that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally. What are two possible solutions? (Choose two.)

A. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 16384-32767
B. Ask the firewall administrator to change the ports to TCP.
C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.
D. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 20000-22000.
E. Go to System Parameters in Cisco Unified Communications Manager and change the range of media ports to 20000-22000.

Correct Answer: AC
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/port/9_1_1/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91_chapter_01.html

QUESTION 11
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
Section: Signaling and Media Protocols
Explanation

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

QUESTION 12 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).

Correct Answer: D
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:
Reference: https://www.cs.columbia.edu/~hgs/rtp/faq.html

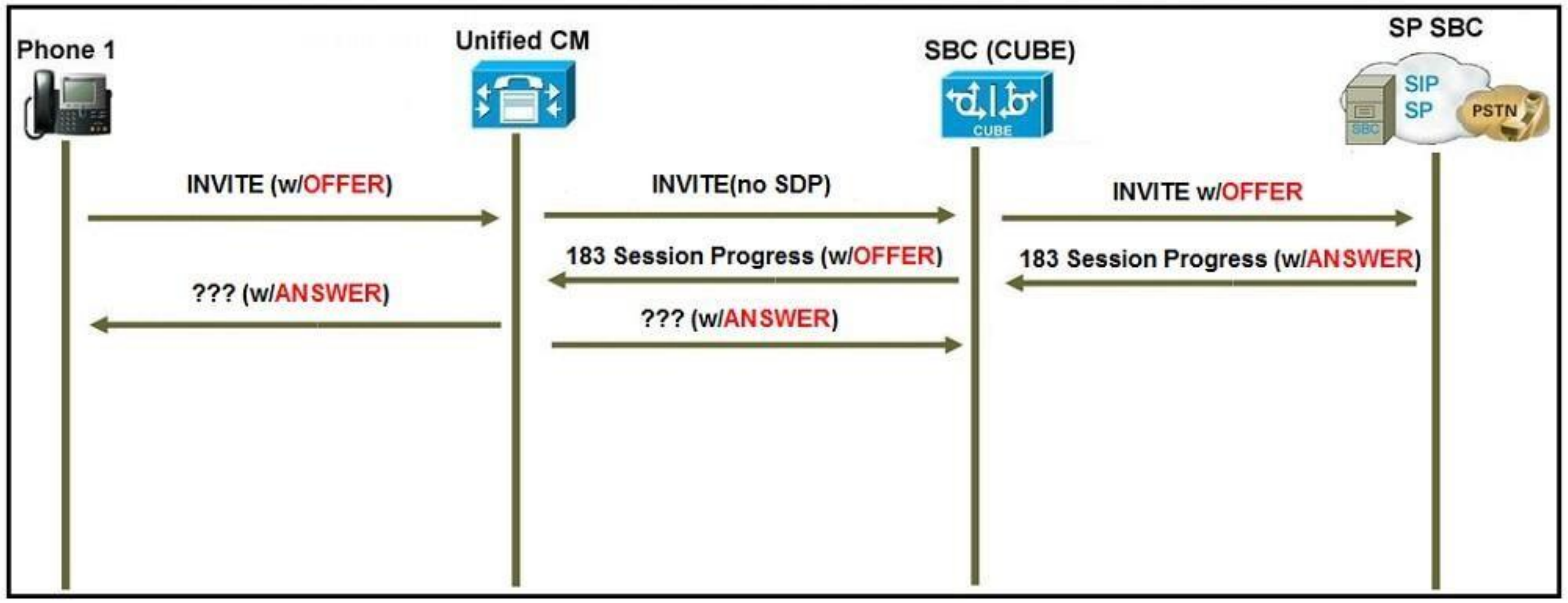
QUESTION 13 A support engineer is troubleshooting a voice network. When conducting a search for call setup details related to calling search space issues, which trace files should be investigated?

A. CallManager traces
B. CTI Manager traces
C. Cisco IP Manager Assistant
D. Call logs

Correct Answer: A
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 14



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
Section: Signaling and Media Protocols
Explanation

Explanation/Reference:

QUESTION 15
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
Section: CME/SRST Gateway Technologies
Explanation

Explanation/Reference:
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 16
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
Section: CME/SRST Gateway Technologies
Explanation

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

QUESTION 17
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

Correct Answer: C
Section: CME/SRST Gateway Technologies
Explanation

Explanation/Reference:
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_setting_up_using_sip.html

QUESTION 18
Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

A. **allow-connections sip to sip**
B. **voice service voip**
C. **voice register global**
D. **voice register dn**

Correct Answer: C
Section: CME/SRST Gateway Technologies
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/99946-cme-sip-guide.html

QUESTION 19 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

Correct Answer: A
Section: Cisco Unified Border Element Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35

QUESTION 20
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

Correct Answer: B
Section: Cisco Unified Border Element Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/dtmf-relay.html

QUESTION 21

```
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”.

Correct Answer: B
Section: Cisco Unified Border Element
Explanation
Explanation/Reference:

QUESTION 22

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

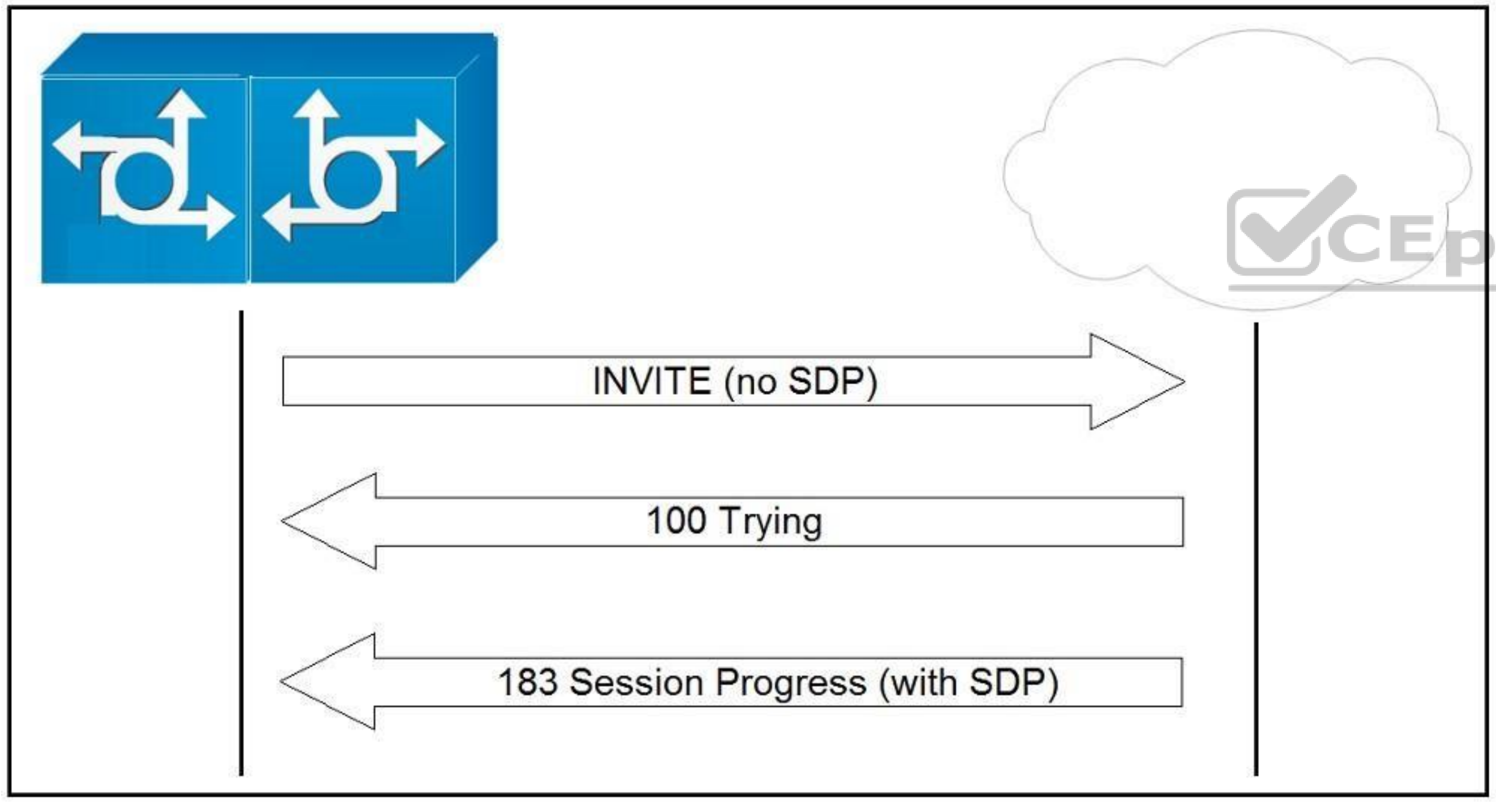
Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border 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 is shown in the exhibit. Which rule modified 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
Section: Cisco Unified Border Element Explanation

Explanation/Reference:

QUESTION 23



Refer to the exhibit. An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border Element. The customer’s carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)

A. Enanle PRACK.
B. Enable Early Offer on the Cisco Unified Border Element.
C. Enable the **supplementary-service media-renegotiate** command.
D. Enable Media Flow Around
E. Enable Mid-Call Signaling Consumption.

Correct Answer: AB

Section: Cisco Unified Border Element Explanation

Explanation/Reference:

QUESTION 24 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**

Correct Answer: B
Section: Cisco Unified Border Element Explanation

Explanation/Reference:
Reference: https://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

QUESTION 25
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: B
Section: Cisco Unified Border Element Explanation

Explanation/Reference:

QUESTION 26
An engineer must route all SIP calls in the form of <user>@example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

A. example.com@gateway.corporate.local
B. *@example.com
C. gateway.corporate.local
D. example.com
E. *.*

Correct Answer: BE
Section: Call Control and Dial Planning Explanation

Explanation/Reference:

QUESTION 27
Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)
A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.
B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.
E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Correct Answer: AB
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
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 28 A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?

A. 813510[012345]
B. 813510[12345]
C. 813510[^0-5]
D. 81XXXXX

Correct Answer: A
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 29 Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)

A. can be associated under the route group
B. can be associated only under the route list
C. chooses the route group that is configured under the device pool of the calling-party device
D. chooses the route group that is configured under the device pool of the called-party deviceE. can be assigned directly to the route pattern

Correct Answer: BD
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 30

```
!
dial-peer voice 1 voip
 description to ITSP
 destination-pattern 555…..
 session target ipv4:209.110.110.1
 incoming called-number .
 codec g711ulaw
!
!
```

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. answer-address 555 ……..

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

Correct Answer: D
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 31 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: A
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
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 32 An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

A. Incoming Port in SIP Information section of the SIP Trunk configuration.
B. Incoming Port in Security Information of the SIP Profile configuration.
C. Destination Port in SIP Information section of the SIP Trunk configuration
D. Incoming Port in SIP Trunk Security Profile configuration
E. Destination Port in SIP Trunk Security Profile configuration

Correct Answer: CD
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 33 In Cisco Unified Communications Manager, which tool do you use to check SIP traces?

A. MTP
B. CCSIP
C. RTMT
D. OS Administration Page

Correct Answer: C
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 34 If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern “123”?

A. 12!
B. 12X (urgent priority set)
C. 1XX (urgent Priority Set)
D. 12[2-5]

Correct Answer: B
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

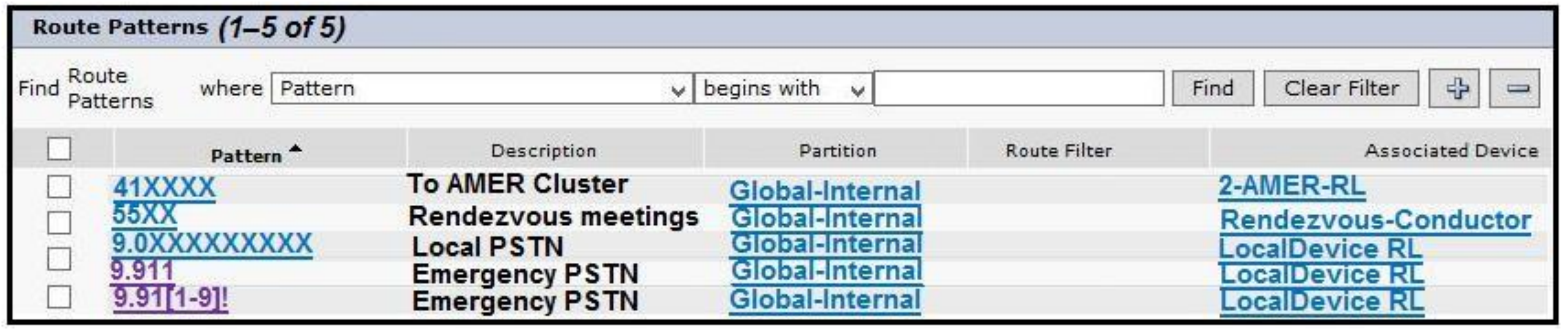
QUESTION 35 Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

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.

Correct Answer: A
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
Reference: https://community.cisco.com/t5/collaboration-voice-and-video/taking-sip-call-trace-on-cisco-unified-cm-using-rtmt/ta-p/3161200

QUESTION 36


Route Patterns (1–5 of 5)

Find Route Patterns where Pattern begins with Find Clear Filter

Pattern	Description	Partition	Route Filter	Associated Device
41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL
55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor
9.0XXXXXXXXX	Local PSTN	Global-Internal		LocalDevice RL
9.911	Emergency PSTN	Global-Internal		LocalDevice RL
9.91[1-9]!	Emergency PSTN	Global-Internal		LocalDevice RL

Refer to the exhibit. Users report that when they dial the emergency number 9911 from any internal phone, it takes a long time to connect with the emergency operator. Which action resolves this issue?

A. Adjust the service parameter T302 timet to the desired value.
B. Adjust the service parameter T204 timer to the desired value.
C. Check the Urgent Priority check box under 9.911 pattern.
D. Point the emergency pattern directly to the PSTN gateway.

Correct Answer: C
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

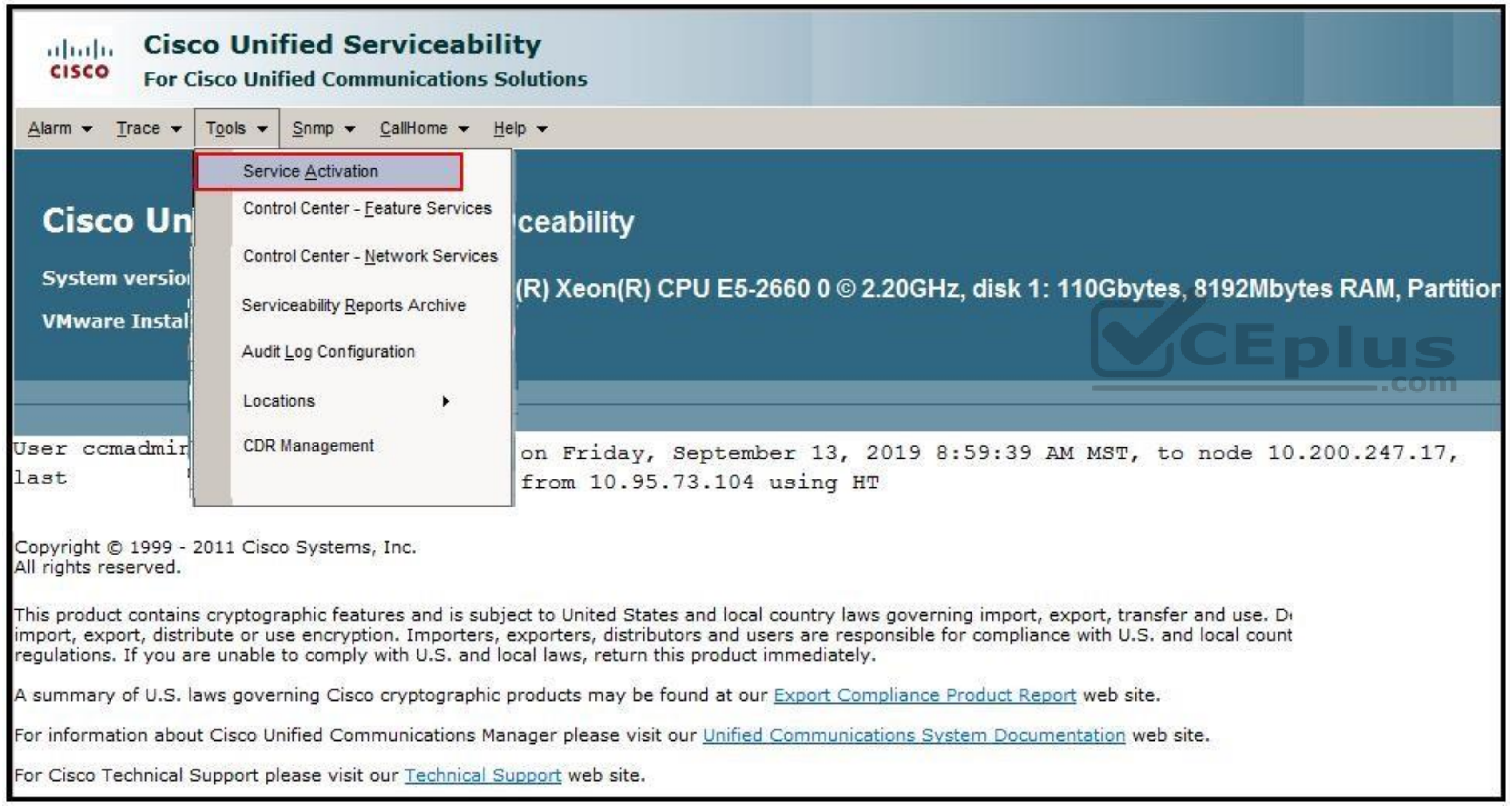
QUESTION 37
The Cisco Unified Communications Manager Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

A. translation patterns
B. device pools
C. CTI ports
D. CTI route points
E. IP phones

Correct Answer: CE
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/11_5_1/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115_chapter_01.html#CUCM_TP_A5DA99E0_00

QUESTION 38



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.

Correct Answer: DE

Section: Call Control and Dial Planning
Explanation

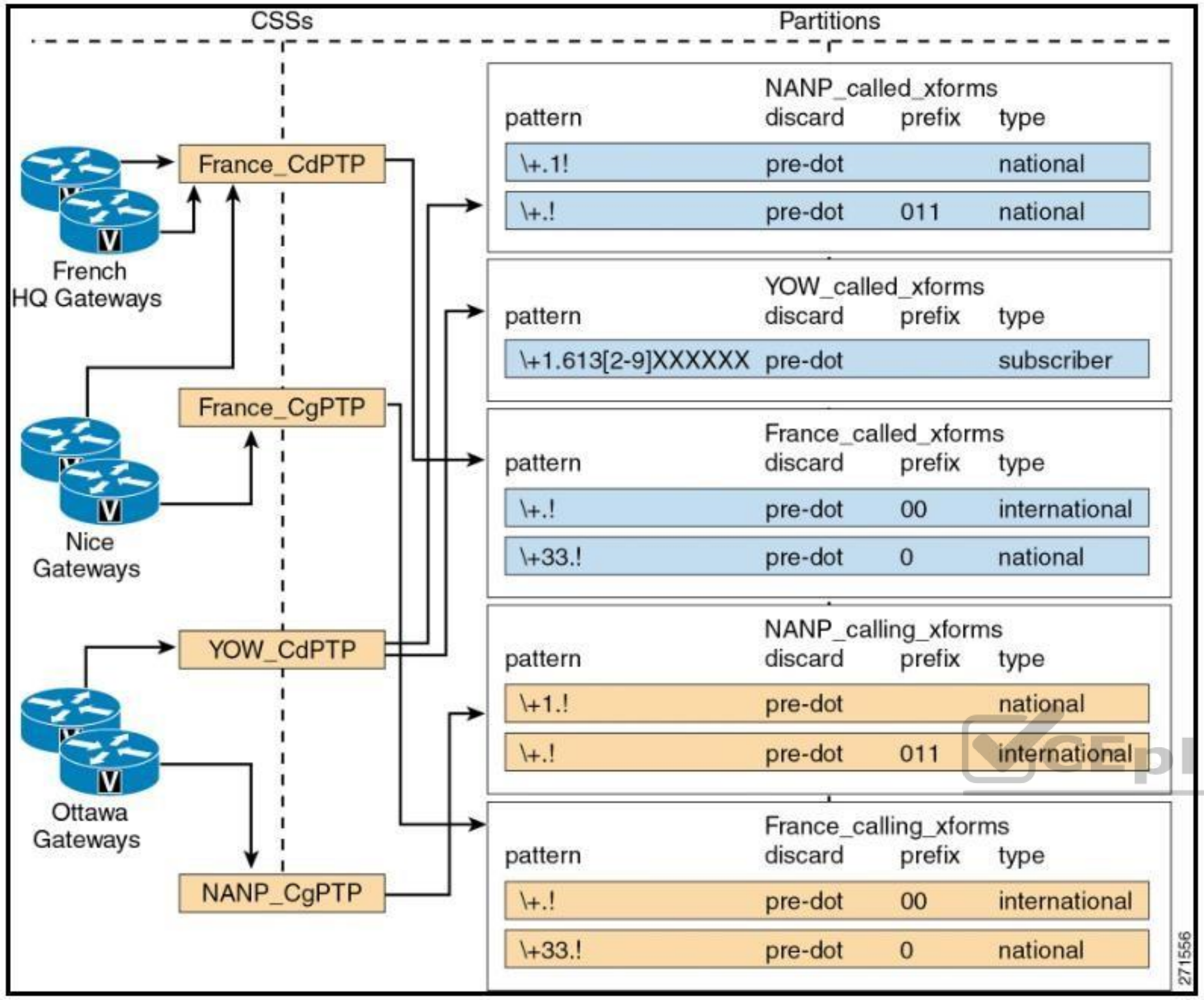
Explanation/Reference:

QUESTION 39 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

Correct Answer: A
Section: Call Control and Dial Planning
Explanation
Explanation/Reference:

QUESTION 40



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"

Correct Answer: D
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
QUESTION 41
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

Correct Answer: B
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
Reference: https://www.uccollabing.com/configuring-standard-local-route-group-cucm/

QUESTION 42 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: B
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:

QUESTION 43 Which two types of distribution algorithm are within a line group? (Choose two.)

A. random
B. circular
C. highest preference
D. top down
E. bottom up

Correct Answer: BD
Section: Call Control and Dial Planning
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcfg/CUCM_BK_CDF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100011.html

QUESTION 44
An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60 seconds?

A. R2(config-ephone-dn)#**park reservation-group 60**
B. R2(config-ephone-dn)#**park-slot timeout 60 limit 2 recall alternate 3002**
C. R2(config-ephone-dn)#**park reservation-group 1**
D. R2(config-ephone-dn)#**park-slot timeout 30 limit 2 recall alternate 3002**

Correct Answer: B
Section: Cisco Unified CM Call Control Features
Explanation
Explanation/Reference:

QUESTION 45 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

Correct Answer: A
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmsys/a07cpick.html#wp1022865

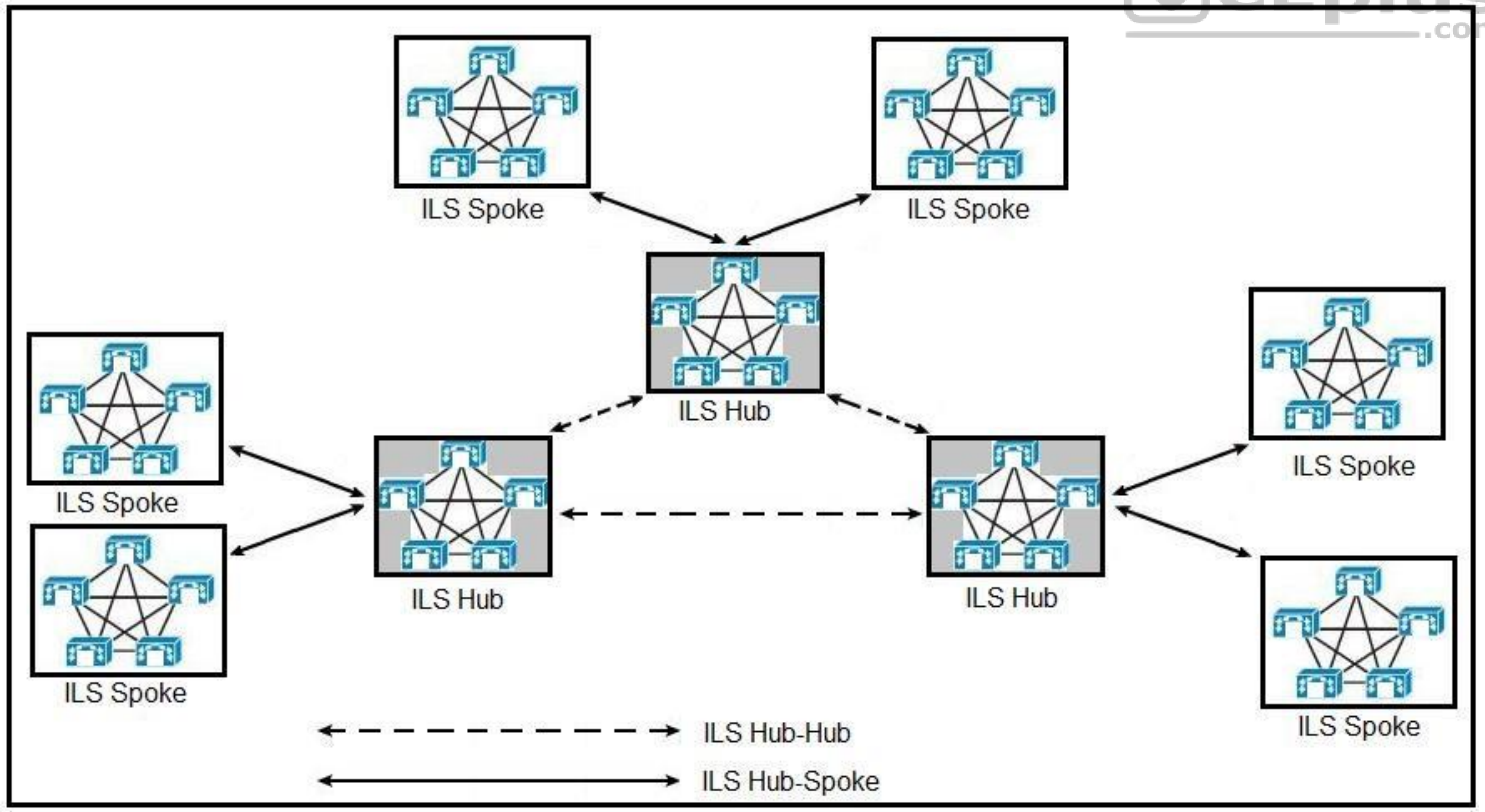
QUESTION 46 When locations-based Call Admission Control denies the call, which two masks can AAR apply when routing the call through the PSTN? (Choose two.)

A. AAR destination mask
B. called party transform mask
C. external phone number mask
D. +E.164 alternate number mask
E. enterrise alternate number mask

Correct Answer: AC
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

QUESTION 47



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

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

Correct Answer: A
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:

QUESTION 48
An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?

A. ILS Max Number of Learned Objects in Database
B. ILS Active Learned Object Upper Limit
C. Global Data Service Parameter Limit
D. Imported Dial Plan Replication Database Object Lower Limit

Correct Answer: A
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU1/systemConfig/cucm_b_system-configuration-guide-1251su1/cucm_b_system-configuration-guide-1251su1_restructured_chapter_0100011.html#CUCM_TK_I7C708C2_00

QUESTION 49
When configuring hunt groups, where do you add the individual directory numbers that will be part of the group?

A. route group
B. line group
C. hunt list
D. hunt pilot

Correct Answer: B
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_010101.html

QUESTION 50
Which configuration element of a hunt group allows for changing Calling Party Transformations settings?

A. line group
B. hunt pilot
C. route group
D. hunt list

Correct Answer: B
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://community.cisco.com/t5/ip-telephony-and-phones/call-alerting-on-hunt-group-as-shared-line/td-p/2658015

QUESTION 51
Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

A. TokenID
B. username and secret key
C. TLS certificates
D. passwords
E. FQDN of the servers defined in DNS

Correct Answer: CD
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_011001.pdf

QUESTION 52
Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
B. A unicast music on hold audio source must be configured.
C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
D. The maximum number of callers allowed in queue must be 10.
E. The phone button template must have the Queue Status Softkey configured.

Correct Answer: AC
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide1201_chapter_01001101.html#CUCM_RF_C960BC9A_00

QUESTION 53
What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?

A. A partition can have multiple time schedules assigned. A time schedule contains one or more time periods.
B. A partition can have one time schedule assigned. A time schedule contains one or more time periods.
C. A partition can have multiple time schedules assigned. A time schedule contains only one time period.
D. A partition can have one time schedule assigned. A time schedule contains only one time period.

Correct Answer: A
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:

QUESTION 54
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
Section: Cisco Unified CM Call Control Features
Explanation

Explanation/Reference:
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 55
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 the most likely cause of this issue?

A. The Cisco Extension Mobility service has not been configured on the phone.
B. There might be a significant delay between the button being pressed and the Cisco Extension Mobility service recognizing it. It would be best to check network latency.
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
Section: Mobility
Explanation

Explanation/Reference:

QUESTION 56 What is a component of Cisco Unified Mobility?

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

Correct Answer: B
Section: Mobility
Explanation

Explanation/Reference:

QUESTION 57 When the services key is pressed Cisco Extension Mobility does not show up. 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
Section: Mobility
Explanation

Explanation/Reference:

QUESTION 58
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.

Correct Answer: B
Section: Mobility
Explanation

Explanation/Reference:

QUESTION 59 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 location. Device Mobility group, and region
D. device pool, Device Mobility group, and Cisco IP phone

Correct Answer: A
Section: Mobility
Explanation

Explanation/Reference:
Reference: https://www.ciscopress.com/articles/article.asp?p=1249228&seqNum=4

QUESTION 60
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

Correct Answer: C
Section: Mobility
Explanation

Explanation/Reference:
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_011101.html#CUCM_TK_A337E035_00