Practice Exam Questions

300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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Exam 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

Version: 9.0

[Total Questions: 119]

Question No : 1

An engineer has two cisco UCM Clusters and wants to integrate them using ILS with TLS certificates. Cluster A (pub and 1 subscriber) will be the hub, and Cluster B (pub and 1 subscriber) will be the spoke. Both Clusters have self-signed certificates. The engineer has exchanged Publisher A and subscriber B Tomcat certificates, but the connection fails. What is the cause of the failure?

A. The password is incorrect.
B. Cluster IDs are not unique.
C. The tomcat certificate from Cluster B must be the publisher.
D. The engineer needs to exchange the CallManager certificate.

Answer: C

Question No : 2

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
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 No : 3

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor Incoming command configured? (Choose two.)

A. "ephone-dn' configuration mode
B. "voice register global" configuration mode
C. "telephony-service" configuration mode
D. "voice register pool" configuration mode

E. "dial-peer cor custom" configuration mode

Answer: A,D

Question No : 4

Refer to the exhibit.

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

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. call-blast
B. parallel
C. sequential
D. peer

Answer: D

Question No : 5

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

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- configuration-guide_chapter_011101.html#CUCM_TK_A337E035_00

Question No : 6

A company has users that are logged in to hunt groups. However, there is a requirement for hunt group configurations to provide an option to turn on audible ringtones when calls to a line group arrive at a phone that is logged out and on a break. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call does not ring at the phone of that line group member because of the logged-out status. Which action meets this requirement?

A. Configure the HLog softkey on the phone so that while a user is logged off, it plays an audible tone when a call is missed.
B. Set the service parameter Party Entrance Tone to True."
C. Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.
D. Set the service parameter Enterprise Feature Access number for hunt group logout and set up an access number

Answer: C

Question No : 7

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

Answer: C,E
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 No : 8

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

Answer: D

Question No : 9

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

Question No : 10

Refer to the exhibit.

```
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
 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 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:8005532447@10.10.150.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.150.11:5060;branch=z9hG4bK1046d36216b0de
From: <sip:1001@10.10.150.11>;tag=23125042-8a7beda1-fb5d-4d82-bdb6-4b07a7393aff-27428380
To: "CISCO SYSTEMS" <sip:8005532447@10.10.150.1>;tag=D974B182-FA5
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02@10.10.150.1
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 INVITE
[..Omitted for brevity..]
Contact: <sip:1001@10.10.150.11:5060>;
Content-Type: application/sdp
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, NPI=Unknown, Screening=User, Passed,
  Called Number=8005532447(TON=Unknown, NPI=Unknown),
  Calling Translated=FALSE, Subscriber Type Str=Unknown, FinalDestinationFlag=FALSE,
  Incoming Dial-peer=100, Progress Indication=NULL(0), Calling IE Present=TRUE,
```

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 GigabitEthernetO/0/0 bind media source-interface GigabitEthernetO/0/0
B. SIP binding In SIP configuration mode:
voice service volp
sip
bind control source-Interface GlgabltEthernetO/0/1 bind media source-Interface GlgabltEthernetO/0/1
C. SIP binding In dial-peer configuration mode:
dial-peer voice 300 voip
voice-class sip bind control source-interface GigabitEthernetO/0/1 voice-class sip bind media source-interface GigabitEthernetO/0/1
D. SIP binding in dial-peer configuration mode:
dial-peer voice 100 volp
voice-class sip bind control source-interface GigabitEthernetO/0/0
voice-class sip bind media source-interface GigabitEthernetO/0/0

Answer: D

Question No : 11

Refer to the exhibit.

```
SIPHandler/ccbId=0/scbId=0/wait_SIPTimer: TimerExpired type=SIP_TIMER_WAIT_CONNECT value=5000 retries=0
Stack/Transport/0x0xee9c8980/sipTransportPostInternalMsg: Posting Internal Msg type=1
Stack/Transport/0x0/sipTransportPostCloseConnection: Posting TCP conn close for addr=10.10.5.11, port=5060, connid=20
Stack/Transport/0x0/sipDeleteConnInstance: Deleted conn=0xe7ac06c0, connid=20, addr=10.10.5.11, port=5060, transport=TCP
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned: 2 (SIP_NETWORK_MSG), for event 64 (SIPSPI_EV_INTERNAL_MSG)
Stack/Error/0x0xee9c8980/sipTransportPostSendFailure: Posting send failure msg with tcb:(nil) reason=4
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned: 2 (SIP_NETWORK_MSG), for event 55 (SIPSPI_EV_SEND_FAILURE_MSG)
Stack/Info/0x0xee9c8980/ccsip_spi_process_event: Send Error for event(0xee9cb6b0)
Stack/Error/0x0/act_idle_send_msg_failure: Send Error to 10.10.5.11:5060 for transport TCP
Stack/Info/0x0xee9c8980/ccsip_set_cc_cause_for_spi_err: Categorized cause:38, category:186
Stack/Info/0x0xee9c8980/sipSPIInitiateDisconnect: Initiate call disconnect(38) for outgoing call
SIPHandler/ccbId=22609/scbId=0/ccsip_api_call_disconnected: ccb->cc_disc_cause (38): ccb->sip_disc_cause(503)
SIPHandler/ccbId=22609/scbId=0/findDevicePID: Routed to SIPD by ccbId/scbId
Stack/States/0x0xee9c8980/sipSPIChangeState: 0xee9c8980 : State change from (STATE_IDLE, SUBSTATE_NONE)  to (STATE_DISCONNECTING, SUBSTATE_NONE)
```

An administrator has configured a SIP trunk between two Cisco UCM clusters. For calls that should use the trunk, the calls fail with a fast busy. The administrator checks the Cisco CallManager SDL traces and found that the cluster to which the calling device is registered never sends an INVITE to the destination cluster. The administrator also verifies that all nodes from both clusters are powered on, and the CallManager service is running. How is this issue resolved?

A. The administrator needs to disable OPTIONS pings on the SIP trunks for both clusters.
B. The administrator must allow connectivity so that TCP connections do not fail between the nodes.
C. The administrator needs to enable OPTIONS pings on the SIP trunks for both clusters.
D. The administrator must associate the route pattern with a calling search space the device can dial.

Answer: B

Question No : 12

Refer to the exhibit.

```
!
dial-peer voice 10 voip
 description Inbound
 session protocol sipv2
 incoming called-number 2000
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 20 voip
 description Outbound
 destination-pattern 2.
 session protocol sipv2
 session target ipv4:192.168.100.101
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0
             AD                                  PRE PASS SESS-SER-GRP\  OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN    FER THRU SESS-TARGET    STAT PORT    KEEPALIVE    VRF
10     voip  up   up                              0  syst                                          NA
20     voip  up   up             2.               0  syst ipv4:192.168.100.101            busyout      NA
```

A call mode through the Cisco Unified Border Element to pilot 2000 is foiling. What is causing the call to foil?

A. No codecs are configured on the dial peers
B. The Cisco Unified Border Element is not receiving a response to its OPTION keepahves.
C. The destination pattern is incorrect for the dialed number.
D. VAD was not disabled on the outgoing dial poor.

Answer: C

Question No : 13

An engineer must configure call queuing under a Hunt Pilot. After the engineer receives the audio file that will be played to callers during queuing, which two steps should be taken to complete the configuration? (Choose two.)

A. Assign the uploaded audio file to the hunting Line Group member's "User Hold MOH Audio Source
B. Assign the uploaded audio file to the hunting Line Group member's "Network Hold MOH Audio Source".
C. Upload the audio file in "TFTP File Management" via OS Administration GUI
D. Assign the uploaded audio file to "Network Hold MOH Source & Announcements" under Hunt Pilot's Queuing section.
E. Upload the audio file in "MOH Audio File Management" via CM Administration GUI

Answer: C,D

Question No : 14

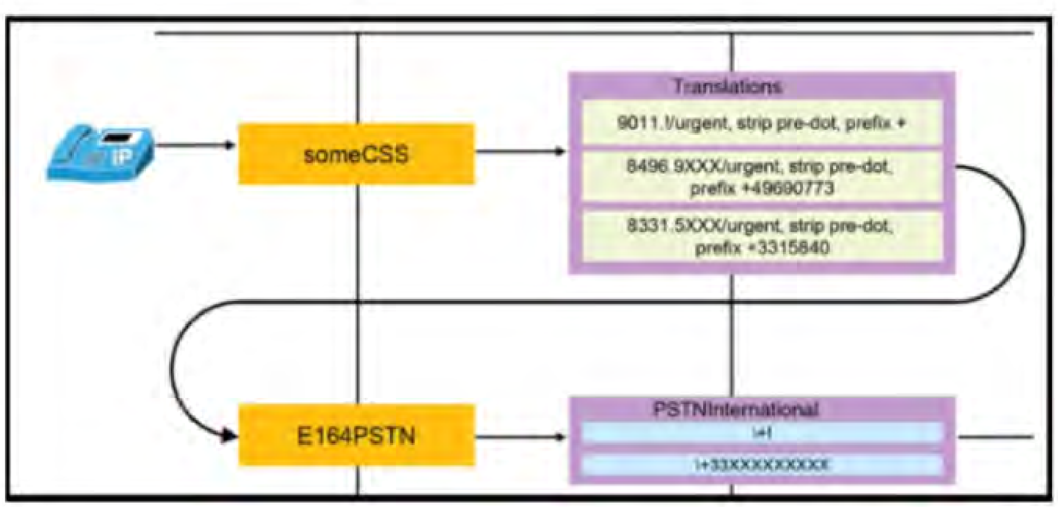
CollabCorp is a global company with two clusters, emea.collab corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI. but this is not working as expected. Which two configuration elements should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

A. directory URI partition
B. SIP route pattern
C. intercluster trunk
D. calling search space and partition
E. SIP trunk

Answer: B,E

Question No : 15

Refer to the exhibit.



A user dials 84969010 and observes that the call is not routed immediately. The administrator notices that after matching the fixed-length translation pattern, the call hits the \+! pattern and waits for interdigit timeout What should be configured to ensure that the call routes out immediately?

A. Allow Device Override on the route pattern
B. Route Next Hop By Calling Party Number on the translation pattern
C. Do Not Wait For Interdigit Timeout On Subsequent Hops on the translation pattern
D. Do Not Wait For Interdigit Timeout On Subsequent Hops on the route pattern

Answer: C

Question No : 16

Refer to the exhibit.

```
55697959.007 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222
crCdpnMask=33334444 crPrefixDigit= crDDI=2
55697959.008 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- after  DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 |12:20:50.913 |AppInfo  |RouteListCdrc::transformed cdpn (without unconsumpt
digits) = 33334444, unconsumed digit=
```

Which INVITE is sent to 10.10.100.123 as a result of this log?

A)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716-41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

B)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716-41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

C)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716-41f7
To: <sip:33334444@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

D)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

A. Option A
B. Option B
C. Option C
D. Option D

Answer: C

Question No : 17

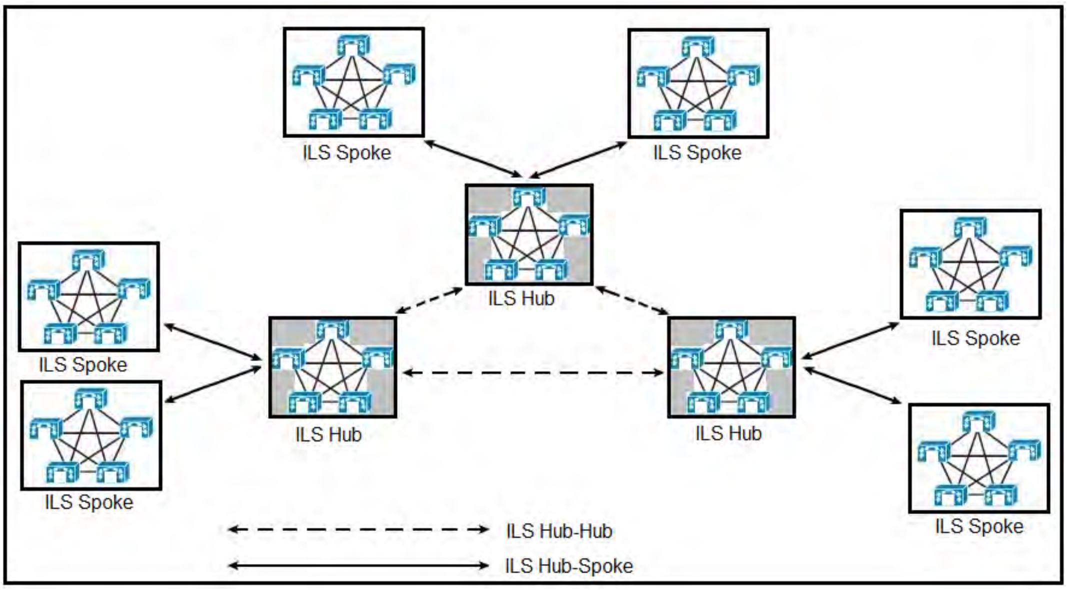
When the services key is pressed Cisco Extension Mobility does not show up. What is the cause of the issue?

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)

Answer: C

Question No : 18

Refer to the exhibit.



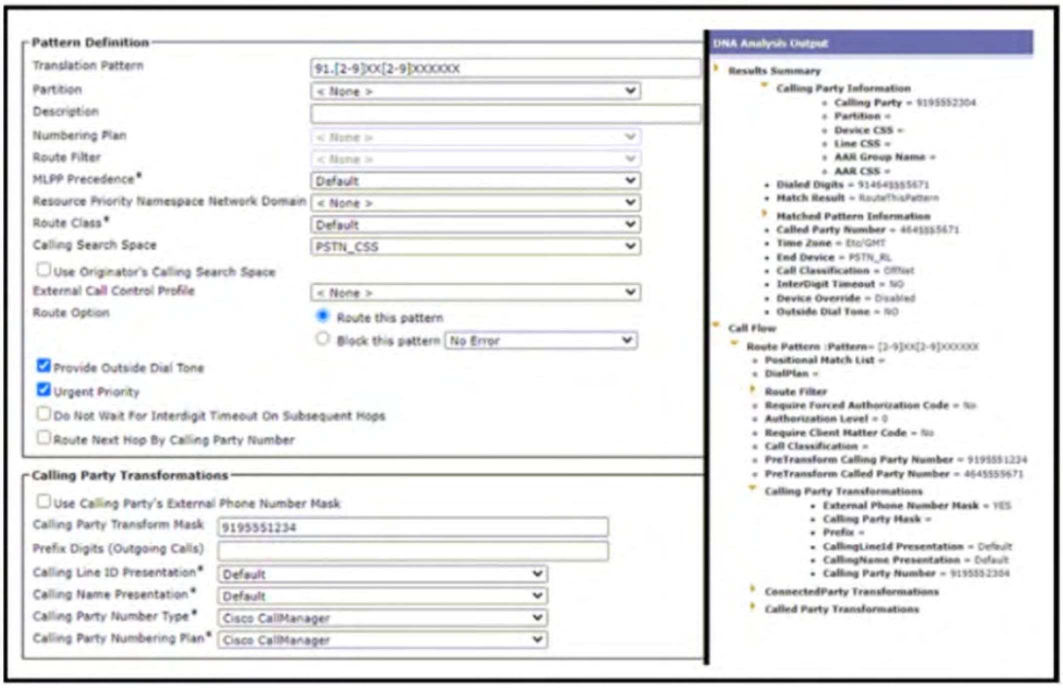
How many maximum hops can an ILS update traverse?

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

Answer: A

Question No : 19

Refer to the exhibit.



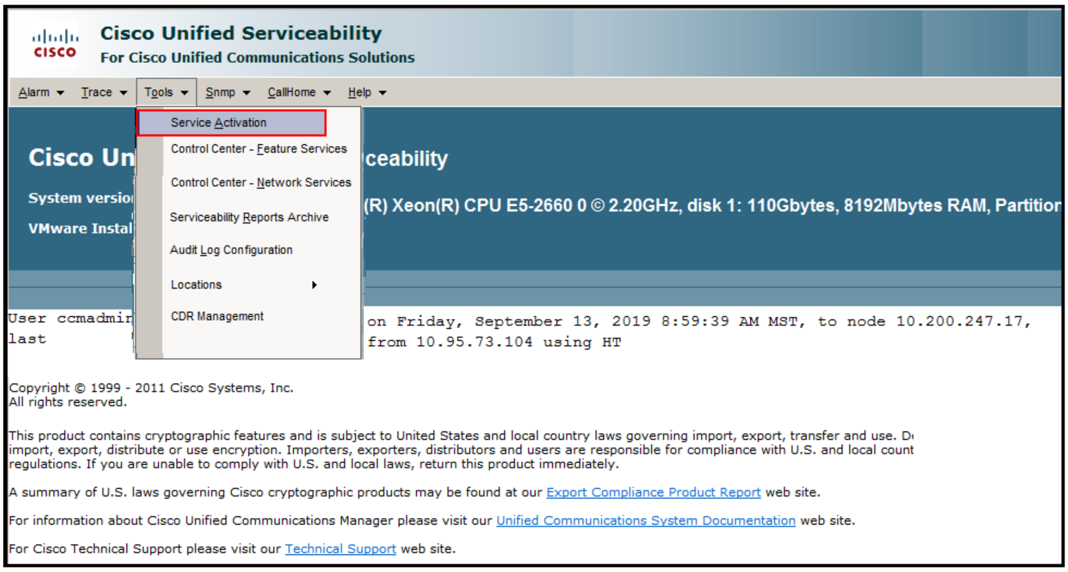
For long-distance calls, users must prefix their dialed number with “91.” The translation pattern was created to strip the 91 as the PSTN expects a 10- digit number. The PSTN also requires the calling number to be set to 9195551234. However, the service provider has said calls with a different calling number are being received. How is this issue resolved?

A. Change the partition of the translation pattern from none to pstn_pt.
B. Enable Force Authorization Code on the route pattern.
C. Disable Use Calling Party's External Phone Number Mask on the route pattern.
D. Enable Use Calling Party's External Phone Number Mask on the translation pattern.

Answer: C

Question No : 20

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: D,E

Question No : 21

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.

Answer: A

Question No : 22

Refer to the exhibit.

```
CUBE_Router#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
CUBE_Router(config)#voice translation-rule 999
CUBE_Router(cfg-translation-rule)#rule 1 /^9(.*)/ //
CUBE_Router(cfg-translation-rule)#end
CUBE_Router#
CUBE_Router#test voice translation-rule 999 9123548
9123548 Didn't match with any of rules
```

Which change to the translation rule is needed to strip only the leading 9 from the digit string 9123548?

A. rule 1 /^9\(.*\)/A1/
B. rulel /.*\(3548S\)/^1/
C. rulel /^9\(\d*\)/^1/
D. rule 1/^9123548/^1/

Answer: A

Question No : 23

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. Remove the long-distance & international pattern's partitions from the calling search space of the physical phone.
B. Add the long-distance & international pattern's partitions to the calling search space of the physical phone's directory number.
C. Remove the long-distance & international pattern's partitions from the calling search space of the device profile.

D. Add the long-distance & international pattern's partitions to the calling search space of the physical phone.
E. Add the long-distance & international pattern's partitions to the calling search space of the device profile

Answer: A,E

Question No : 24

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 /^4...$/2404\0/ type any national plan any Isdn
B. voice translation-rule 1 rule 1 // // type any subscriber plan any isdn
C. voice translation-rule 1 rule 1 /^4...S/ /9132404 0/ type any subscriber plan any Isdn
D. voice translation-rule 1
rule 1 /^4...V /2404\0/ type any subscriber plan any isdn

Answer: D

Question No : 25

A customer routes PSTN calls to ITSP through a SIP trunk on Cisco UCM that forwards and receives calls to and from ITSP. ITSP is set to send an E.164 number when the customer's extension is four digits. Which action should be taken to route the incoming calls to four-digit extensions?

A. Configure a voice translation rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.
B. Set the Significant Digits to 4 on the SIP trunk.
C. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.
D. Set the Significant Digits to 8 on the SIP trunk.
E. Configure a voice translation rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.
F. Set the Significant Digits to 4 on the SIP trunk.
G. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

H. Set the Significant Digits to 8 on the SIP trunk.

Answer: B

Question No : 26

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 one-way 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.

Answer: A,C
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 No : 27

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

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

Question No : 28

What are two configureation features of the Client matter code setting in the route pattern configuration? (Choose two.)

A. The client Matter Code feature supports overlap sending since the Cisco UCM can determine when to prompt the user for the code.
B. Selecting the Allow Overlap Sending setting disables the Require Client Matter Code setting.
C. The Client Matter Code feature provides the option to configure Authorization Level susch as in the Forced Authorization Code.
D. Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.

Answer: B,C

Question No : 29

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: A
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 No : 30

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.

Answer: D

Question No : 31

An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM. How should this be accomplished?

A. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
B. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.
C. Localize the calling and called numbers to E. 164 format and globalize the called number in the gateway.
D. Globalize the calling and called numbers to E. 164 format and localize the called number in the gateway.

Answer: D

Question No : 32