Number: 300-815
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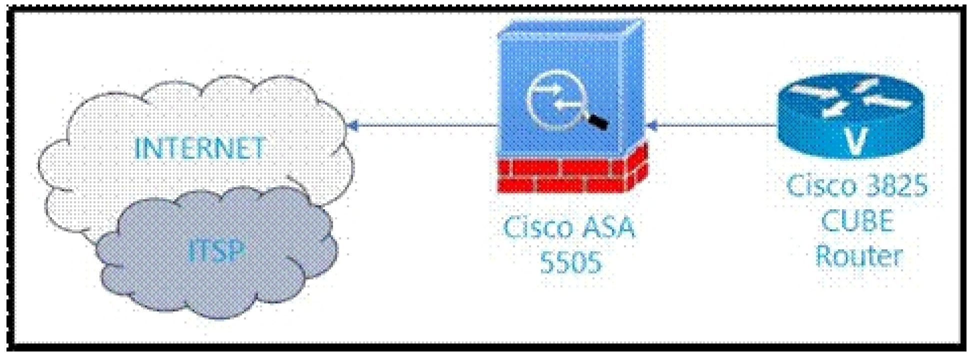
Exam Code: 300-815
Exam Name: Implementing Cisco Advanced Call Control and Mobility Services (CLASSM)

Exam A

QUESTION 1

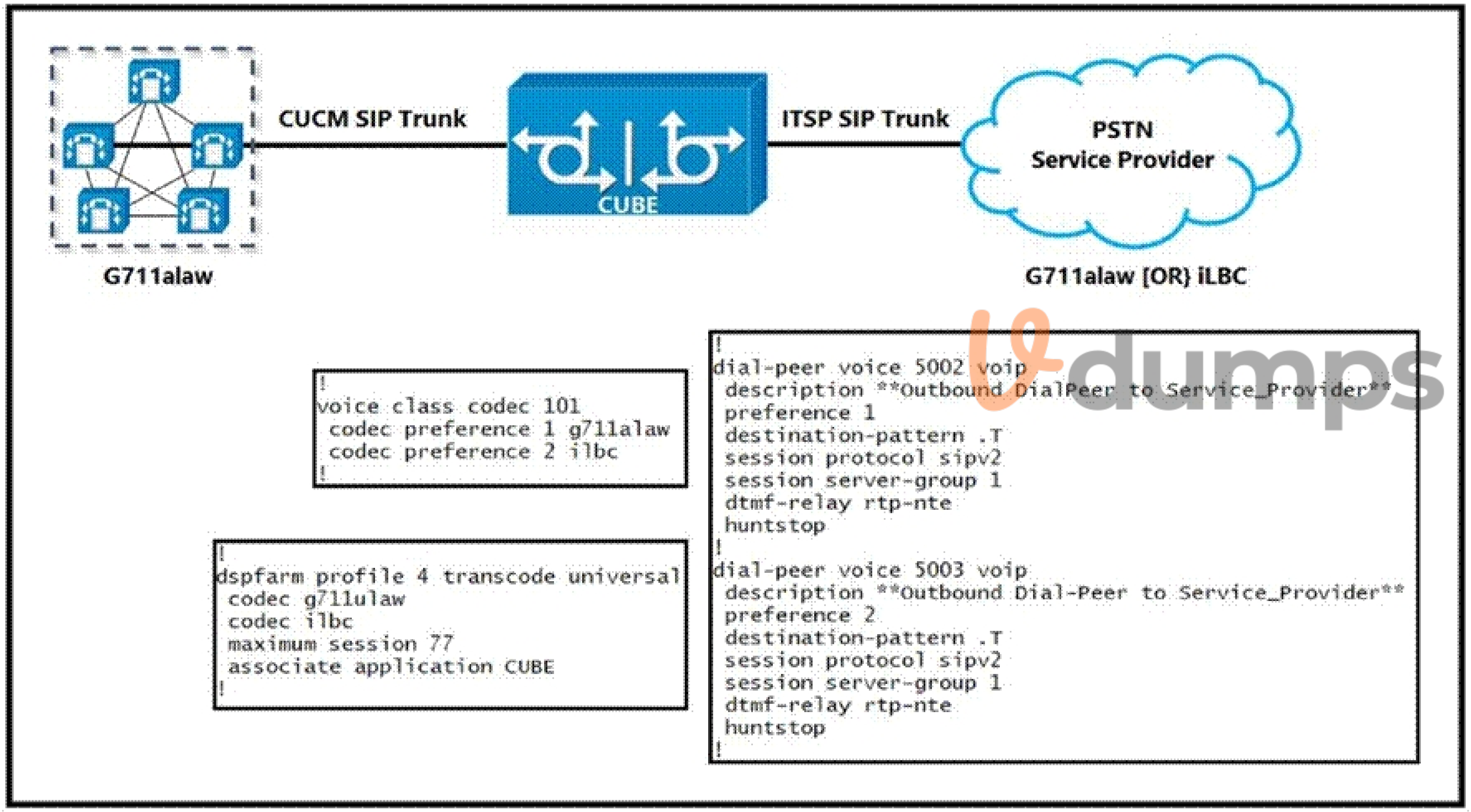


Refer to the exhibit. An administrator is troubleshooting a problem in which some outbound calls from an internal network to the Internet telephony service provider are not getting connected, but some others connect successfully. The firewall team found that some call attempts on port 5060 came from an unrecognized IP that has not been defined in the firewall rule. What should the administrator configure in the Cisco Unified Border Element to fix this issue?

A. use of port 5061 for SIP secure
B. access list allowing the firewall IP
C. bind signaling and media to the loopback interface
D. ip prefix-list to filter the unwanted IP address

Correct Answer: C
Section:

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
Section:

QUESTION 3

```
!
dial-peer voice 100 voip
 description Outbound to CUCM
 translation-profile outgoing CUCM
 session protocol sipv2
 session target ipv4:192.168.100.200
 voice-class sip transport switch udp tcp
 voice-class sip conn-reuse
 voice-class sip rel1xx disable
 voice-class sip session refresh
 voice-class sip midcall-signaling block
 voice-class sip early-media update block
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

Refer to the exhibit. An engineer is troubleshooting an issue where inbound calls to Cisco UCM with early media fail to establish. While investigating the issue, the engineer finds that Cisco UCM is set to require a PRACK, but the Cisco Unified Border Element is not sending it. Which command is causing this issue?

A. voice-class midcall-signaling block
B. voice-class sip rel1xx disable
C. voice-class sip early-media update block
D. voice-class sip conn-reuse

Correct Answer: B
Section:

QUESTION 4

```
voice class codec 100
 codec preference 1 g711alaw
 codec preference 2 g729r8
 codec preference 3 g729br8
 codec preference 4 g711ulaw
!
dial-peer voice 5002 voip
 session protocol sipv2
 session server-group 1
 incoming called-number 5...
 voice-class codec 100
 dtmf-relay rtp-nte
 no vad

m=audio 30104 RTP/AVP 0 9 124 116 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:124 iSAC/16000
a=rtpmap:116 iLBC/8000
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Refer to the exhibit. The Cisco Unified Border Element receives an INVITE matching inbound dial peer 5002. The outbound dial peer supports only iLBC, and a Local Transcoding Interface is allocated. Based on the configuration and SDP from the INVITE message, which codec is chosen by Cisco Unified Border

Element for the inbound call leg?

A. G.729r8
B. G.711 A-law
C. G.711 U-law
D. G.729br8

Correct Answer: C
Section:

QUESTION 5
Due to a shortage of physical interfaces on a device, the administrator requires that a loopback for RTP is used. Which command is required when using a loopback interface for RTP?

A. voice-class sip early-offer forced
B. voice-class sip bind control source-interface Loopback0
C. voice-class sip bind media source-interface Loopback0
D. voice-class sip resource priority mode passthrough

Correct Answer: C
Section:

QUESTION 6
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
Section:

QUESTION 7
When a third-party SIP Phone System is dialed inbound across a Cisco Unified Border Element, DTMF is failing. The third-party vendor accepts only out-of-band DTMF. Which configuration should be added to the outgoing dial peer to resolve this issue?

A. dtmf-relay rtp-nte
B. dtmf-relay cisco-rtp
C. dtmf-relay h245-signal
D. dtmf-relay sip-kpml

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

QUESTION 8
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: B, E
Section:

QUESTION 9
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: A, B
Section:
Explanation:
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 10
A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). Which route pattern should 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:

QUESTION 11
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:
Explanation:
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 12
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 "Network Hold MOH Source & Announcements" under Hunt Pilot's Queuing section.
B. Upload the audio file in "TFTP File Management" via OS Administration GUI.
C. Assign the uploaded audio file to the hunting Line Group member's "User Hold MOH Audio Source".
D. Assign the uploaded audio file to the hunting Line Group member's "Network Hold MOH Audio Source".
E. Upload the audio file in "MOH Audio File Management" via CM Administration GUI.

Correct Answer: A, E
Section:

QUESTION 13
```
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
Section:

QUESTION 14
An engineer has two Cisco UCM clusters and wants 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.

Correct Answer: D
Section:

QUESTION 15
A user requests a feature to send an active call to the mobile phone number on the physical phone. As an administrator, what should be configured in the Cisco UCM to accomplish this?

A. A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey "Join" to the physical phone's softkey template.
B. A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey "Mobility" to the physical phone's softkey template.
C. A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey "Join" to the physical phone's softkey template.
D. A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey "Mobility" to the physical phone's softkey template.

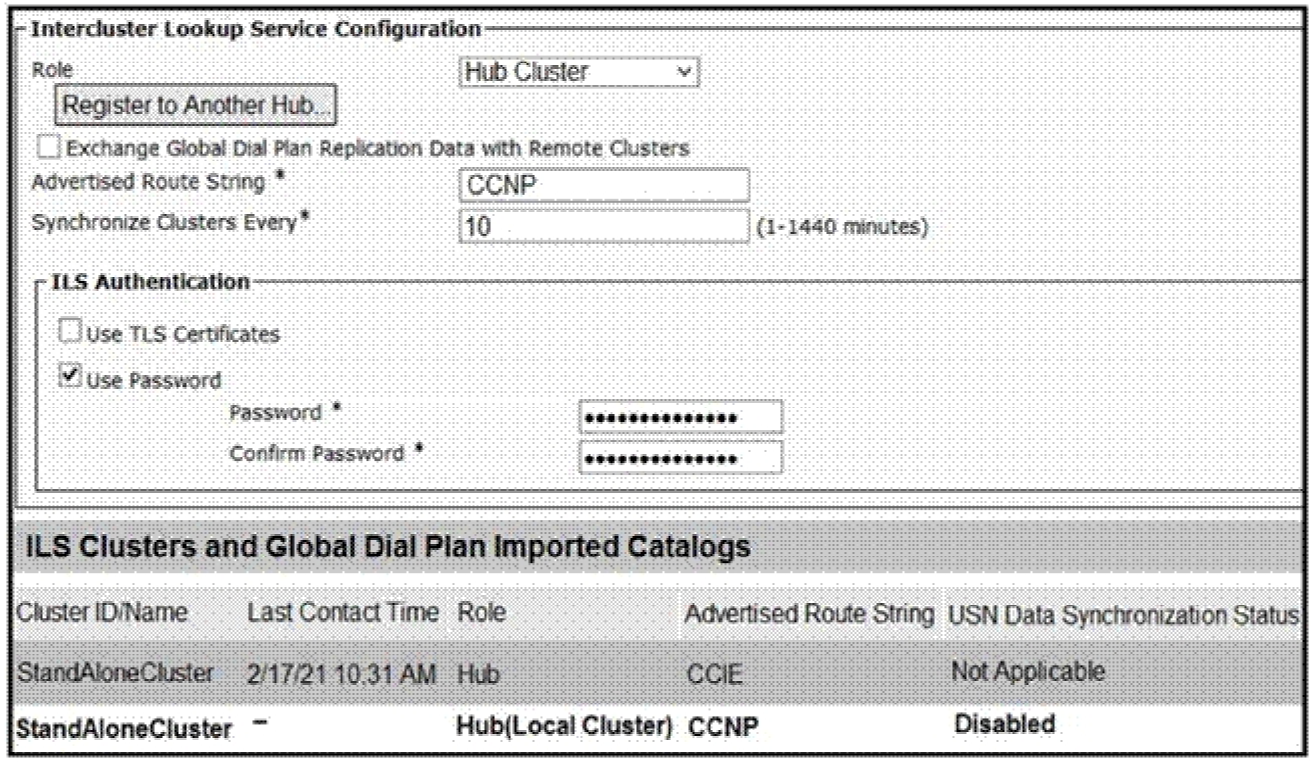
Correct Answer: D
Section:

QUESTION 16
ABC company has decided to implement hunt groups to help distribute calls between members. In order to implement this, the administrator must configure hunt list, hunt groups, and line groups on Cisco UCM. Which distribution algorithms should the administrator implement?

A. Top Down, Round Robin, Broadcast
B. Top Down, Circular, Broadcast
C. Top Down, Round Robin, Distribute
D. Sequential, Circular, Broadcast

Correct Answer: B
Section:
Explanation:
Reference: https://flylib.com/books/en/2.110.1/call_distribution_components.html

QUESTION 17



Refer to the exhibit. ILS has been configured between two hubs this configuration. The hubs appear to register successfully, but ILS is not functioning as expected. Which configuration step is missing?

A. Use TLS Certificates must be selected.
B. The Cluster IDs have not been set to unique values.
C. A password has never been set for ILS.
D. Trust certificates for ILS have not been installed on the clusters.

Correct Answer: B
Section:
Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1_SU7/cucm_b_system-configuration-1151su7-1151su8/cucm_b_system- configuration-guide-1151su1_chapter_011001.pdf

QUESTION 18
Cisco UCM has 100,000 entries in the database learned through the ILS Service. Parameter ILS Max Number of Learned Objects in Database value is set to 100,000. What will happen to learned data when the service parameter value is reduced to 50,000?

A. Cisco UCM does not write additional ILS learned objects to the database and will delete the last 50,000 entries learned to keep it to the service parameter value.
B. Cisco UCM does not write additional ILS learned objects to the database and keeps the existing database entries.
C. Cisco UCM will overwrite an entry for newly learned data and keep the parameter value at 100,000.
D. Cisco UCM does not write additional ILS learned objects to the database and will delete the first 50,000 entries learned to keep it to the service parameter value.

Correct Answer: B
Section:

QUESTION 19
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 cause of this issue?

A. The Cisco Extension Mobility service has not been configured on the phone.
B. Network latency should be checked since there might be a significant delay between the button being pressed and it being recognized by the Cisco Extension Mobility service.
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:

QUESTION 20
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:

QUESTION 21

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
Section:

QUESTION 22
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:

QUESTION 23
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:
Explanation:
Reference: https://www.ciscopress.com/articles/article.asp?p=1249228&seqNum=4

QUESTION 24
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:
Explanation:
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

QUESTION 25
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 & international pattern's partitions to the calling search space of the physical phone.
B. Remove the long-distance & international pattern's partitions to the calling search space of the physical phone's directory number.
C. Add the long-distance & international pattern's partitions to the calling search space of the device phone.
D. Add the long-distance & international pattern's partitions to the calling search space of the physical phone's directory number.
E. Remove long-distance & international pattern's partitions from the calling search space of the device phone.

Correct Answer: B, E
Section:

QUESTION 26
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
Section:

QUESTION 27

An engineer is troubleshooting Cisco Device Mobility and find that the phone has roamed to a building that is assigned to a different device pool but has not changed its device pool accordingly. What action resolves the issue?

A. Set correct Location under Current Device Mobility Settings.
B. Enable SRST under Current Device Mobility Settings.
C. Set Device CSS under Current Device Mobility Settings.
D. Set the correct subnet under Device Mobility info.

Correct Answer: D
Section:

QUESTION 28
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 & international pattern's partitions to the calling search space of the physical phone.
B. Remove the long-distance & international pattern's partitions to the calling search space of the physical phone.
C. Add the long-distance & international pattern's partitions to 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's directory number.
E. Remove long-distance & international pattern's partitions from the calling search space of the device profile.

Correct Answer: B, E
Section:

QUESTION 29
A user's phone is already configured for Single Number Reach, and the user wants a feature to move an active call from a mobile phone to a desk phone and vice-versa. As an administrator, which additional configuration should be made to fulfill the user's request?

A. Use Dialed Number Analyzer to determine if the user extension can dial the mobile phone.
B. Add the mobility key to the softkey template that the desk phone is using.
C. Check to make sure that the Resume softkey option appears on the desk phone.
D. Confirm that the desk phone is subscribed to Cisco Extension Mobility.

Correct Answer: C
Section:
Explanation:
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 30
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: B, E
Section:
Explanation:
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 31
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:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" ;tag=32412716~41f7
To:
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" ;tag=32412716~41f7
To:
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

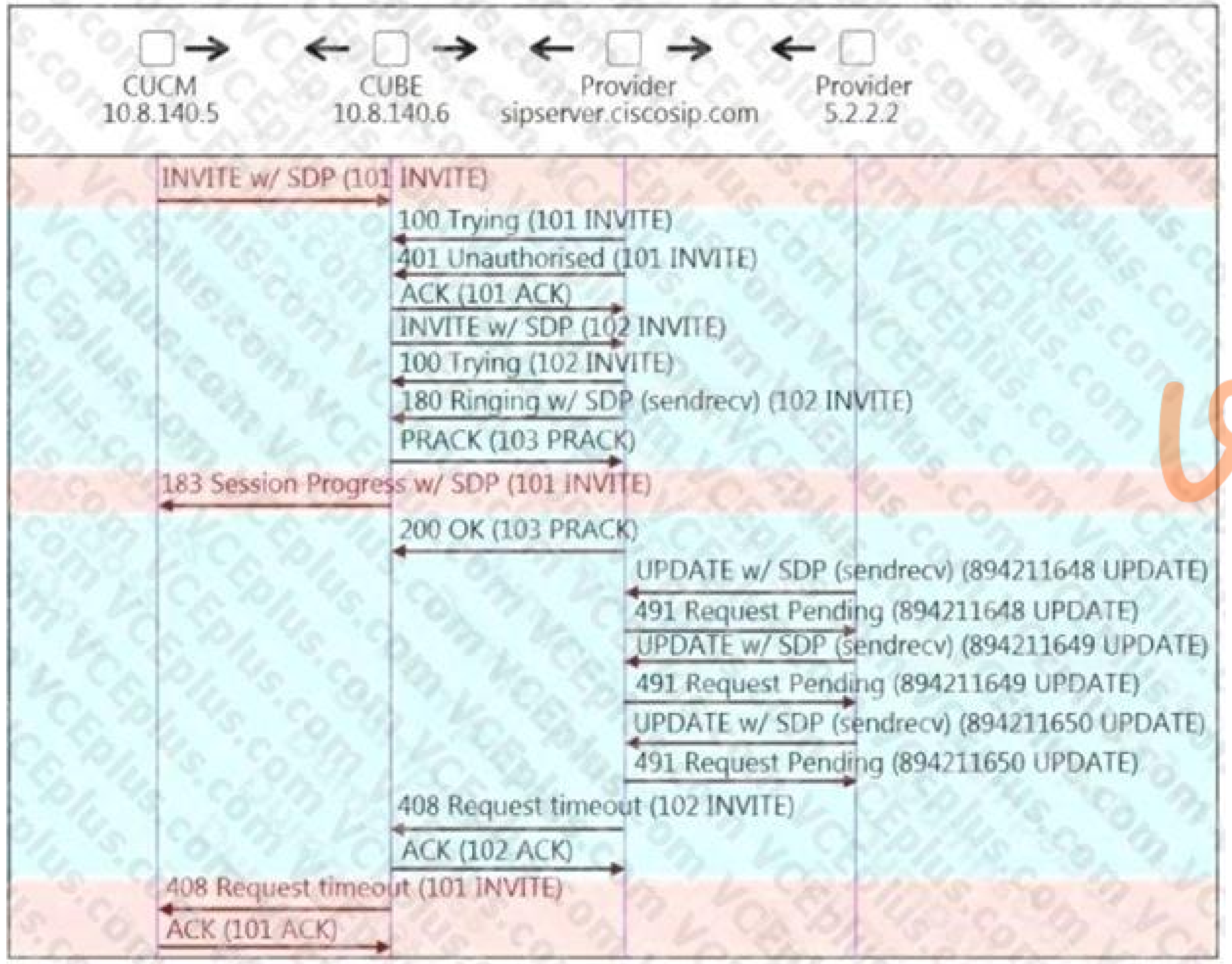
C. 55698034.001 |12:20:50.922 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" ;tag=32412716~41f7
To:
Date: 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: "11112222" ;tag=32412716~41f7
To:
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

Correct Answer: C
Section:

QUESTION 32
Refer to the exhibit.



A Cisco Unified Border Element continues to send 180/183 with the required: 100rel header to Cisco UCM, and the call eventually disconnects. How is the issue resolved?

A. Disable "SIP Rel1XX Options" and "Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.
B. Enable "SIP Rel1XX Options" and "Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.
C. Disable "Send send-receive SDP in mid-call INVITE" on the SIP Profile Configuration Page in Cisco UCM.
D. Enable "Early Offer support for voice and video calls" on the SIP Profile Configuration Page in Cisco UCM.

Correct Answer: D
Section:

QUESTION 33

Refer to the exhibit.

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.1.60.105:5060;branch=z9hG4bK721ed5d4
From: "1001" <sip:1001@10.88.247.229>;tag=6cfa89726ac700b569ec133a-7e6cd8aa
To: <sip:2005@10.88.247.229>;tag=47B5F70-438
Date: Fri, 19 Apr 2019 12:13:40 GMT
Call-ID: 6cfa8972-6ac7002b-5af19a5c-0de23108@10.1.60.105
CSeq: 101 INVITE
Require: 100rel
RSeq: 3344
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Remote-Party-ID: <sip:2005@10.88.247.229>;party=called;screen=yes;privacy=off
Contact: <sip:2005@10.88.247.229:5060>
Server: Cisco-SIPGateway/IOS-16.6.2
Content-Length: 0
```

An engineer is troubleshooting an issue with the caller not hearing a PSTN announcement before the SIP call has completed setup. How must the engineer resolve this issue using the reliable provisional response of the SIP?

A. voice service voip sip
   no rel1xx
B. sip-ua disable-early-media 180
C. voice service voip sip
   rel1xx require 100rel
D. voice service voip sip
   send 180 sdp

Correct Answer: C
Section:

QUESTION 34
A customer has multisite deployments with a globalized dial plan. The customer wants to route PSTN calls via the gateway assigned to each site. Which two actions will fulfill the requirement? (Choose two.)

A. Create one global route list for PSTN calls that points to one global PSTN route group.
B. Create a route group which has all the gateways and associate it to the device pool of every site.
C. Assign one route group as a local route group in the device pool of the corresponding site.
D. Create one route group for each site and one global route list for PSTN calls that point to the local route group.
E. Create a hunt group and assign it to each side route pattern.

Correct Answer: A, C
Section:
Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/dialplan.html

QUESTION 35
An engineer is troubleshooting local ringback on a Cisco SIP gateway. The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified in the SDP. Which configuration change must be made on the gateway to resolve the issue?

A. Router(config-sip-ua)# no disable-early-media 180
B. Router(conf-voi-serv)# no disable-early-media 180
C. Router(conf-voi-serv)# disable-early-media 180

D. Router(config-sip-ua)# disable-early-media 180

Correct Answer: D
Section:
Explanation:
Reference:
https://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_d2_ps5207_TSD_Products_Command_Refere nce_Chapter.html#wp1452642

QUESTION 36
An engineer must configure a Cisco UCM hunt list so that calls to users in a line group are routed to the first idle user and then the next. Which distribution algorithm must be configured to accomplish this task?

A. broadcast
B. top down
C. longest idle time
D. circular

Correct Answer: C
Section:
Explanation:
Reference: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_0_1/ccmcfg/b03lngrp.html

QUESTION 37
An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x and must enforce security and toll fraud prevention for the SIP line side on Cisco Unified CME. Which configuration must be used to start this process?

A. voice service voip enable
ip address trust list
B. voice service voip
ip address trusted list
C. voice service voip
ip address trusted authenticate
D. voice service voip enable
ip address trust authentication

Correct Answer: B
Section:
Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html

QUESTION 38
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 made through the Cisco Unified Border Element to pilot 2000 is failing. What is causing the call to fail?

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

Correct Answer: D
Section:

QUESTION 39
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
Section:

QUESTION 40
A new deployment is using MVA for a specific user on the sales team, but the user is having issues when dialing DTMF. Which DTMF method must be configured in resolve the issue?

A. in-band
B. out-of-band
C. gateway
D. channel

Correct Answer: B
Section:

QUESTION 41
An administrator is working on an issue between the customer's Cisco Unified Border Element and the service provider. The provider only wants to see mid-call signaling from the Cisco Unified Border Element for fax calls. Which command must be configured on Cisco Unified Border Element?

A. midcall-signaling passthru
B. no update-callerid
C. midcall-signaling passthru media-change
D. midcall-signaling preserve-codec

Correct Answer: C
Section:

QUESTION 42
A single site reports that when they dial select numbers, the call connects, but they do not get audio. The administrator finds that the calls are not routing out of the normal gateway but out of another site's gateway due to a TEHO configuration. What is the next step to diagnose and solve the issue?

A. Verify that the route pattern has the correct calling-party transformation mask.
B. Verify that IP routing is correct between the gateway and the IP phone.
C. Verify that the dial peer of the gateway has the correct destination pattern configured.
D. Verify that the route pattern is not blocking calls to the destination number.

Correct Answer: C
Section:

QUESTION 43
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\(\d*\)/ /\l/
B. rule 1 /^9\(.*\)/ /\l/
C. rule 1 /.*\(3548$\)/ /\l/
D. rule 1 /^9123548/ /\1/

Correct Answer: B
Section:

QUESTION 44
An engineer set up and successfully tested a TEHO solution on the Cisco UCM. PSTN calls are routed correctly using the IP WAN as close to the final PSTN destination as possible. However, suddenly, calls start using the backup local gateway instead. What is causing the issue?

A. route pattern
B. LAN connectivity
C. WAN connectivity
D. route list and route group

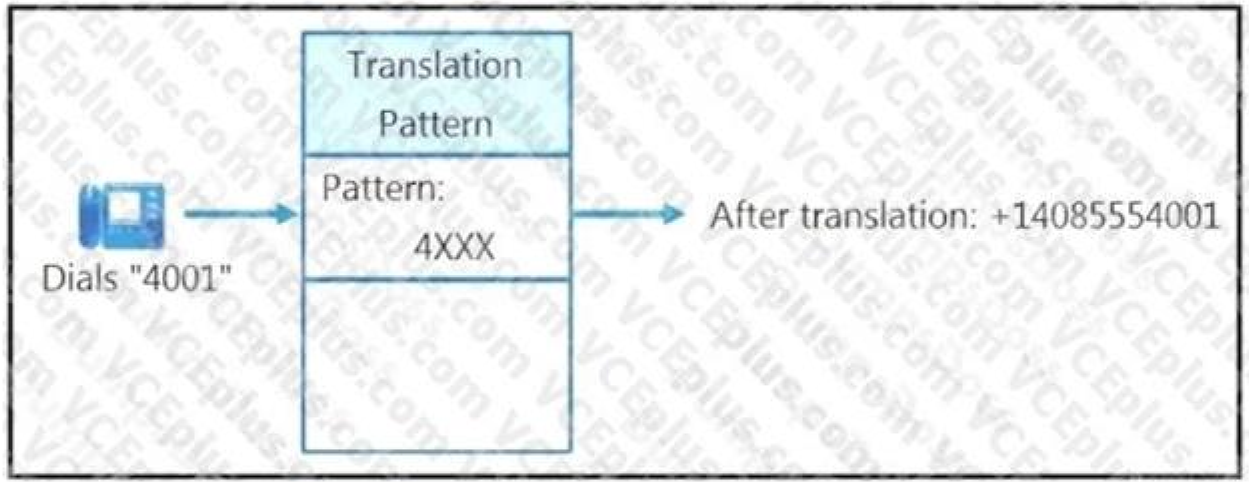
Correct Answer: C
Section:

QUESTION 45
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
Section:

QUESTION 46
Refer to the exhibit.



A company needs to ensure that all calls are normalized to + E164 format. Which configuration will ensure that the resulting digit string +14085554001 is created and will be routed to the E.164 routing schema?

A. Calling Party Transformation Mask of +14085554XXX
B. Calling Party Transformation Mask of +1408555XXXX
C. Called Party Transformation Mask of +1408555[35]XXX
D. Called Party Transformation Mask of +14085554XXX

Correct Answer: D
Section:

QUESTION 47
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 E.164 format and globalize the called number in the gateway.
B. Globalize the calling and called numbers to E.164 format and localize the called number in the gateway.
C. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
D. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.

Correct Answer: B
Section:

QUESTION 48
An engineer is configuring Cisco UCM to forward parked calls back to the user who parked the call if it is not retrieved after a specified time interval. Which action must be taken to accomplish this task?

A. Configure enterprise softkeys.
B. Configure device pools.
C. Configure class of control.
D. Configure service parameters.

Correct Answer: A
Section:

QUESTION 49

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 Enterprise Feature Access number for hunt group logout and set up an access number.
C. Set the service parameter Party Entrance Tone to "True."
D. Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.

Correct Answer: D
Section:

QUESTION 50
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
Section:
Explanation:
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucmfeature-configuration-guide_1052/
CUCM_BK_C3A84B33_00_cucmfeature-configuration-guide_chapter_010.html