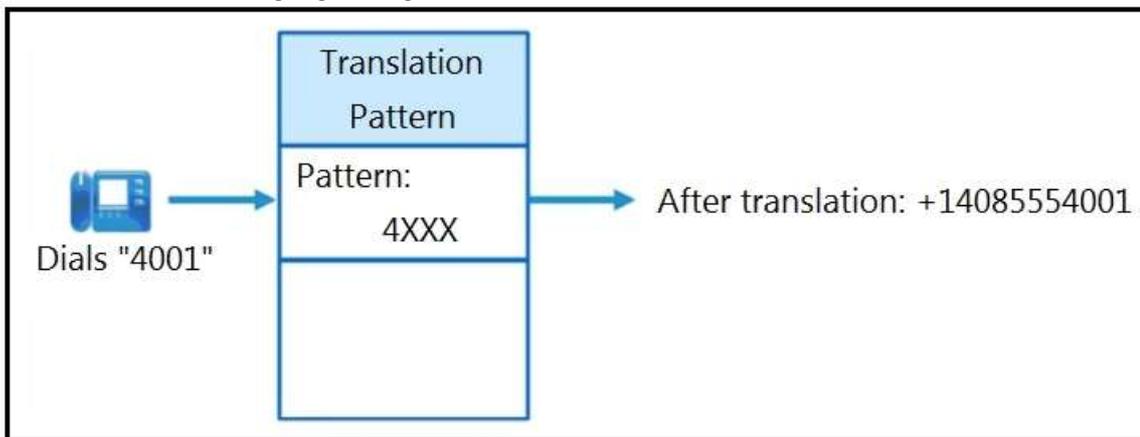


- **Vendor: Cisco**
- **Exam Code: 300-815**
- **Exam Name: Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)**
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**QUESTION 117**

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

**Answer: A**

**QUESTION 118**

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

**Answer: A**

**QUESTION 119**

An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM.

How should this be accomplished?

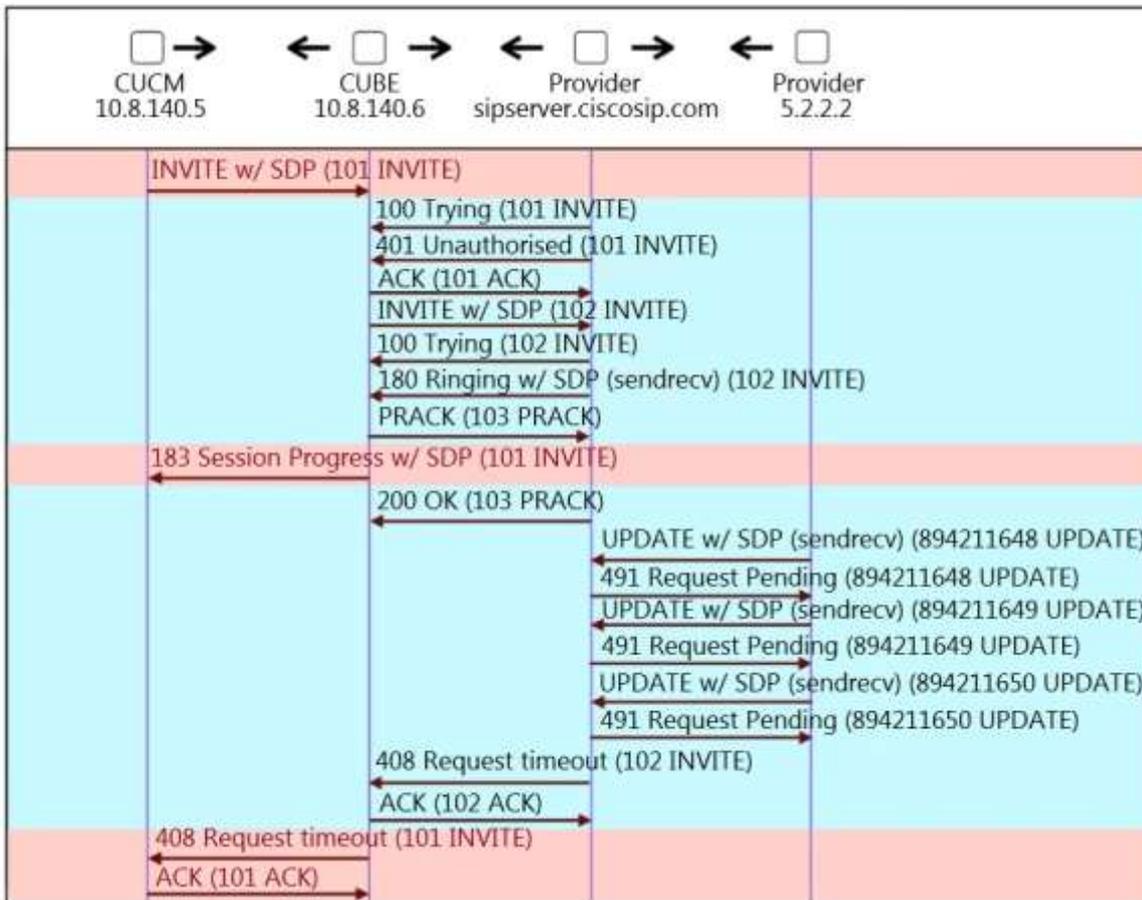
- A. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
- B. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.
- C. Localize the calling and called numbers to E.164 format and globalize 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 120**

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

**Answer: A**

**QUESTION 121**

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

**Answer: C**

**QUESTION 122**

Refer to the exhibit. Which INVITE is sent to 10.10.100.123 as a result of this log?

```
55697959.007 112:20:50.913 |AppInfo |RouteListCdr::createPartyTransformedCcSetupReqMsg
- before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222
crCdpnMask=33334444 crPrefixDigit= crDDI=2
55697959.008 112:20:50.913 |AppInfo |RouteListCdr::createPartyTransformedCcSetupReqMsg
- after DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 112:20:50.913 |AppInfo |RouteListCdr::transformed cdpn (without unconsumpt
digits) = 33334444, unconsumed digit=
```

- A. 

```
55698034.001 112:20:50.922 |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. 

```
55698034.001 112:20:50.922 |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. 

```
55698034.001 112:20:50.922 |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. A screenshot of a SIP INVITE message. The text is as follows:  
55698034.001 |12:20:50.922 |AppInfo |SIP TCP - wait\_SdISPISignal: 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>;rtag=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

**Answer: C**

#### QUESTION 123

An administrator has a requirement that the agents that receive calls from PSTN now must have a queue to hold the calls until they can be answered, but the customer cannot afford a contact center solution. The administrator discovered a feature called Native Call Queueing on Cisco UCM. Where must the administrator configure Native Call Queueing functionality?

- A. CTI route point
- B. hunt list
- C. hunt pilot
- D. hunt group

**Answer: C**

#### QUESTION 124

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

**Answer: D**

#### QUESTION 125

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

**Answer: B**

#### QUESTION 126

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  
ip address trusted list

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

**Answer:** A

**Explanation:**

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html)

**QUESTION 127**

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 128**

Drag and Drop Question

Drag and drop the steps from the left into the order to provision mobility users through LDAP on the right. Not all options are used.

Add and name a new template	step 1
Job Information > Run Immediately	step 2
Bulk Administration > Users > Update Users > Query	step 3
Configure the fields in the Feature Group Template Configuration window	step 4
User Management > User/Phone Add > Feature Group Template	
Apply the filter and select users to be assigned as mobility users	
Enable Mobility, Mobile Voice Access, Maximum Wait Time for Desk Pickup, and Remote Destination Limit.	

**Answer:**

Add and name a new template	User Management > User/Phone Add > Feature Group Template
Job Information > Run Immediately	Add and name a new template
Bulk Administration > Users > Update Users > Query	Enable Mobility, Mobile Voice Access, Maximum Wait Time for Desk Pickup, and Remote Destination Limit.
Configure the fields in the Feature Group Template Configuration window	Configure the fields in the Feature Group Template Configuration window
User Management > User/Phone Add > Feature Group Template	
Apply the filter and select users to be assigned as mobility users	
Enable Mobility, Mobile Voice Access, Maximum Wait Time for Desk Pickup, and Remote Destination Limit.	

**Explanation:**

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/11\\_5\\_1/cucm\\_b\\_feature-configuration-guide-cisco1151su8/cucm\\_b\\_feature-configuration-guide-cisco1151su8\\_chapter\\_01.html#task\\_77EEACB9BEBA958F465F4CE26BD76D36](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/cucm_b_feature-configuration-guide-cisco1151su8/cucm_b_feature-configuration-guide-cisco1151su8_chapter_01.html#task_77EEACB9BEBA958F465F4CE26BD76D36)

**QUESTION 129**

Drag and Drop Question

Drag and drop the commands from the bottom to the blanks in the code to implement a translation rule to allow only 11 digits to be received over a SIP trunk to a SIP provider. The Cisco UCM is currently sending calls to the Cisco Unified Border Element in E.164 format. Not all options are used.

```

voice translation-rule 1000
[ ]
!
voice translation-profile STRIP-PLUS
translate [ ] [ ]
    
```

rule 1 /+//	calling
100	rule 1 /\+//
called	1000

**Answer:**

```
voice translation-rule 1000
rule 1 /\+/ //
!
voice translation-profile STRIP-PLUS
translate called 1000
```

```
rule 1 /+/ //
calling
100
```

**QUESTION 130**

Refer to the exhibit. A call made through the Cisco Unified Border Element to pilot 2000 is failing. What is causing the call to fail?

```
!
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
```

TAG	TYPE	AD	MIN	OPER	PREFIX	DEST-PATTERN	PRE	PASS	SESS-SER-GRP\	OUT	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. 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.

**Answer: B**

**QUESTION 131**

A company was looking at the IT charges and saw many long-distance and international calls primarily to sites in North America and around the world. The administrator wants to optimize the PSTN expense. Which dial plan configuration reduces PSTN connectivity charges by using the IP network to bring the egress point to the PSTN as close as possible to the called number?

- A. translation patterns
- B. tail end hop off

- C. client matter codes
- D. dial rules

**Answer: B**

**QUESTION 132**

Refer to the exhibit. All calls from site A to site B are failing, and the issue has been identified as a media negotiation problem. Which configuration change resolves this issue?

```
46282041.005 |09:18:16.331 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=3,
PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) kbps=8, capACount=1, capBCount=7

46282041.006 |09:18:16.331 |AppInfo |DET-MediaManager-(1698821)::checkAudioPassThru,
param(bPostMTPAllocation=0,chkTrp=1), capCount(1,7), mtpPT=1, aPT=2

46282041.007 |09:18:16.331 |AppInfo |DET-MediaManager-(1698821)::preCheckCapabilities,
region1=RTP_Reg, region2=SJ_Reg, Pty1 capCount=1 (Cap,ptime)= (4,20), Pty2 capCount=7 (Cap,ptime)=
(4,20) (2,20) (6,20) (11,20) (12,20) (15,20) (16,20)

46282041.008 |09:18:16.331 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=0,
PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=0, B=0) kbps=8, capACount=1, capBCount=7

46282041.009 |09:18:16.331 |AppInfo |RegionsServer: applyCodecFilterIfNeeded - no codecs remained
after filtering so restored original 0 caps
```

- A. Increase the bandwidth allowance between the RTP\_Reg and SJ\_Reg regions to 64 kbps.
- B. Enable Early Offer on the SIP trunk.
- C. Create a new audio codec preference list with G.711 U-law 64k as the highest priority and apply it to RTP\_Reg and SJ\_Reg.
- D. Disable G.722 on all devices at both sites.

**Answer: C**

**QUESTION 133**

Which set of commands binds SIP media and signaling to interface GigabitEthernet0/0 when dial peer 1 is chosen for call routing?

- A. dial-peer voice 1 voip  
voice-class source interface GigabitEthernet0/0
- B. voice service voip  
bind sip source-interface GigabitEthernet0/0
- C. voice service voip  
sip  
bind all source-interface GigabitEthernet0/0
- D. dial-peer voice 1 voip  
voice-class bind control source-interface GigabitEthernet0/0  
voice-class sip bind media source-interface GigabitEthernet0/0

**Answer: C**

**QUESTION 134**

An administrator must control the number of calls to a remote specific site to reduce bandwidth constraints. The users on that remote site report bad quality of the calls passing through that WAN link. Which action must the administrator take in Cisco UCM to resolve the issue?

- A. Use RSPV.
- B. Use Location Bandwidth Manager.
- C. Use Expressway deployment.

D. Use Call Allow Controller.

**Answer: B**

**QUESTION 135**

Refer to the exhibit. When setting up a new connection to Cisco UCM, the engineer must use out-of-band DTMF. Which configuration meets this requirement?

```
!  
dial-peer voice 101 voip  
  description Inbound to CUCM  
  destination-pattern 1...  
  session target ipv4: 10.1.1.1  
  session protocol sipv2  
  codec g711ulaw  
  no vad  
!
```

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

**Answer: B**

**QUESTION 136**

An organization has decided to implement hunt groups to help with the distribution of calls between different members. The administrator must configure hunt groups on Cisco UCM. In the configuration, at which step does an administrator have the option to configure a distribution algorithm (top down, circular, longest idle time, broadcast)?

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

**Answer: B**

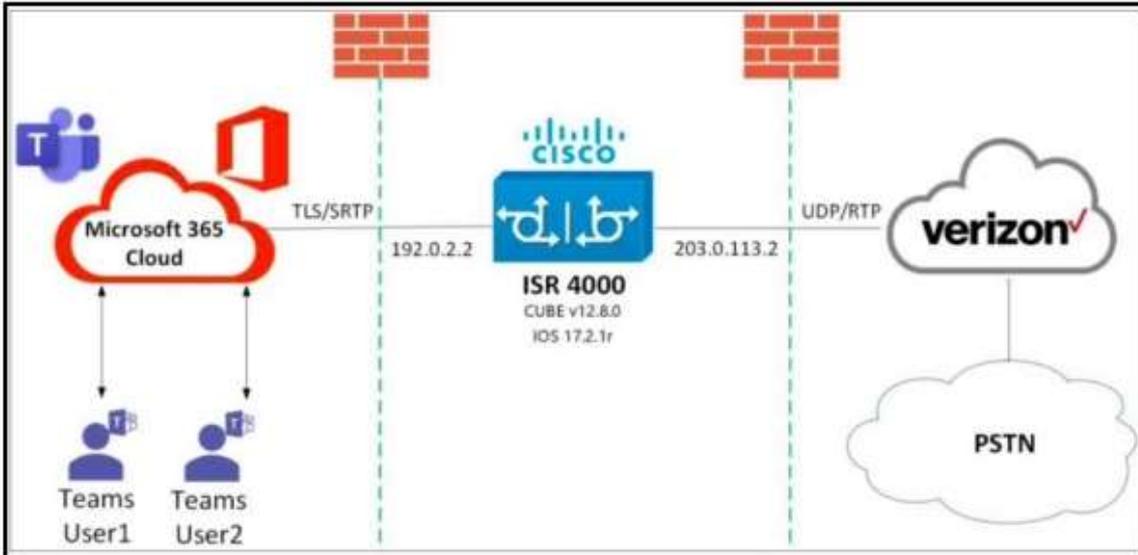
**Explanation:**



[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](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 137**

Refer to the exhibit. A company is using Microsoft Teams with Cisco Unified Border Element integration, but the administrator sees a one-way audio issue with Microsoft Teams. The administrator must modify the SIP profile to send the proper information on the SDP for IP address for media to match the internal and external interface. Which set of commands resolves the issue?

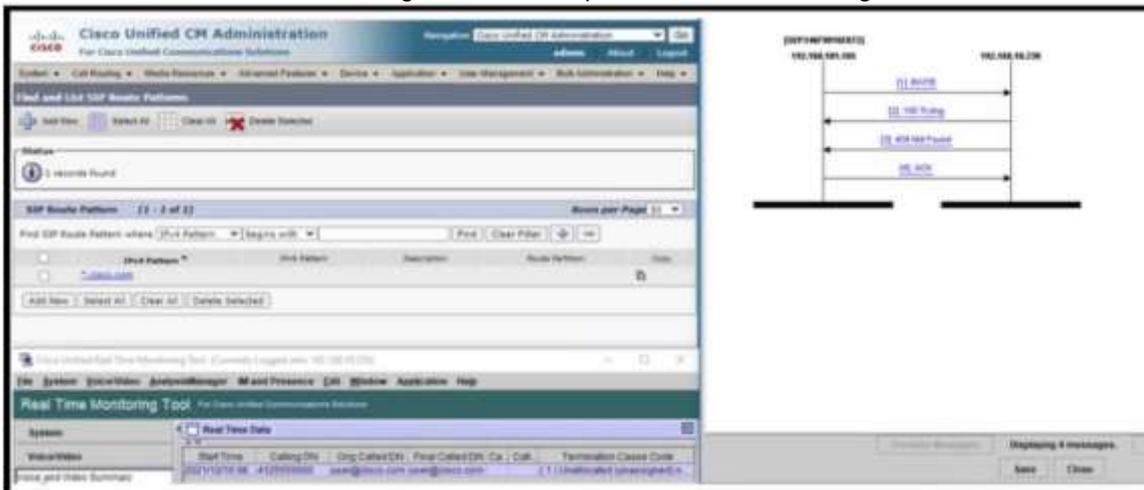


- A. voice class sip-profiles 1  
request INVITE sdp-header Connection-Info modify "2\76\1" "2.78.1"
- B. voice class sip-profiles 1  
request INVITE sdp-header Session-Owner modify "27\0\0" "27.3.0"
- C. voice class sip-profiles 1  
request INVITE sdp-header Connection add "2\76\1" "2.78.1"
- D. voice class sip-profiles 1  
request INVITE sdp-header Audio-Attribute modify "2\76\1" "2.71.1"

**Answer: A**

**QUESTION 138**

Refer to the exhibit. A collaboration engineer is troubleshooting a Cisco UCM issue where users report that calls placed to the domain cisco.com are failing and calls to sip.cisco.com are working. Which action resolves the issue?



The exhibit consists of two parts. On the left is a screenshot of the Cisco Unified CM Administration interface. The 'SIP Route Patterns' section is visible, showing a table with columns for 'SIP Pattern', 'SIP Pattern', 'Description', and 'Route Pattern'. There is a search bar and a 'Find' button. On the right is a SIP call flow diagram showing a sequence of messages between two endpoints. The messages are: INVITE, 200 OK, SIP INFO, and BYE.

- A. Change the "Route Partition" setting on the \*.cisco.com route pattern.
- B. Add a SIP route pattern for \*.\*.
- C. Select "Block Pattern" on the \*.cisco.com route pattern.
- D. Delete the \*.cisco.com route pattern.

**Answer: B**

**QUESTION 139**

Refer to the exhibit. This message is sent to the device being placed on hold for the Music On Hold audio setup. The held party reports receiving dead air rather than music when the call was put on hold. The software Music On Hold server on Cisco UCM is used in this scenario. Assume that the audio leg between the Music On Hold server and the held device uses G.711, and the relevant region relationship is configured for 64 kbps. What is the cause of the issue?

```
02904115.001 |09:08:07.093 |AppInfo |SIPtcp - wait_SdISPISignal: Outgoing SIP TCP message to 10.1.1.102 on
port 50244 index 22157
[166156,NET]
ACK sip:1315932e-ff29-4203-23e3-ef940216638e@10.1.1.102:50244;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.1.5:5060;branch=z9hG4bKb4fa1fa89d7e
From: <sip:1001@10.1.1.5>;tag=93016~bb788fb3-a1ef-4d03-a96d-651038e22050-28377951
To: <sip:1000@cucm1251.cisco.lab>;tag=7001b5dab46425b45ec6648a-25dc4f91
Date: Wed, 28 Jul 2021 13:08:07 GMT
Call-ID: cee27300-1ed10da7-b403-3251300e@10.1.1.5
User-Agent: Cisco-CUCM12.5
Max-Forwards: 70
CSeq: 103 ACK
Allow-Events: presence
Session-ID: 36ed016300105000a0002834a2824611;remote=49f3b76a00105000a0007001b5dab464
Content-Type: application/sdp
Content-Length: 412

v=0
o=CiscoSystemsCCM-SIP 93016 3 IN IP4 10.1.1.5
s=SIP Call
c=IN IP4 10.1.1.5
t=0 0
m=audio 4000 RTP/AVP 0
a=X-cisco-media:umoh
b=TIAS:64000
aptime:20
a=rtptime:0 PCMU/8000
a=inactive
```

- A. The bandwidth configured for this region relationship is too low and must be increased to 96 kbps or higher.
- B. The device that is placed on hold does not support G.711, and a transcoder could not be allocated for the call.
- C. Cisco UCM is sending a=inactive to the held device.
- D. The Music On Hold server does not support G.711 and a transcoder could not be allocated for the call.

**Answer: C**

#### **QUESTION 140**

Refer to the exhibit. An administrator configured Device Mobility but is receiving reports that local calls are failing when a user takes their device from the RTP location to the Charlotte location. The administrator confirmed that the correct subnet is configured under the Device Mobility info page. In addition, the roaming device pool has the correct Device Mobility calling search space selected. Which configuration change resolves the issue?

Roaming Sensitive Settings	
Date/Time Group*	CMLocal
Region*	Charlotte_Region
Media Resource Group List	mrgl_a-b
Location	Charlotte_Location
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	<input type="text"/>
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	Charlotte_DMG
Wireless LAN Profile Group	< None >
Emergency Location(ELIN) Group	< None >

[View Details](#)

- A. Join across lines is not supported with Device Mobility and must be disabled.
- B. The network locale must be changed from none to the Charlotte locale.
- C. The physical location must be updated from none to the appropriate location.
- D. The device must be added to the Device Mobility group.

**Answer: C**

**Explanation:**

Under Roaming Sensitive Settings, assign the parameters that you set up in the previous device mobility tasks:

>> Physical Location - From the drop-down list, select the physical location that you set up for this device pool. Device mobility uses this location to assign a device pool for a roaming device.

>> Device Mobility Group - From the drop-down list, select the device mobility group that you set up for this device pool.

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/12\\_5\\_1SU1/cucm\\_b\\_feature-configuration-guide-for-cisco1251SU1/cucm\\_b\\_feature-configuration-guide-for-cisco1251SU2\\_chapter\\_011.html#CUCM\\_TK\\_CC7C2AFB\\_00](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_5_1SU1/cucm_b_feature-configuration-guide-for-cisco1251SU1/cucm_b_feature-configuration-guide-for-cisco1251SU2_chapter_011.html#CUCM_TK_CC7C2AFB_00)

**QUESTION 141**

An administrator manages an environment with mobile users who commonly move between locations and take IP phones to remote sites. When mobile users are at one of the sites, users report issues with call quality, such as the audio cutting in and out. In addition, the affected area has a low bandwidth connection compared to other sites, so the administrator has put these devices in a region with an inter-region restriction of 8 kbps per call. Which feature must the administrator configure to mitigate the issue?

- A. Location Bandwidth Manager
- B. Extension Mobility
- C. Device Mobility
- D. Extension Mobility Cross Cluster

**Answer: A**

**QUESTION 142**

An administrator is configuring a new deployment using Cisco Unified CME. The SCCP phones register without any issues, but SIP phones are not registering. Assume that all other configuration is valid. Which code allows SIP phones to register to Cisco UCME?

- A. voice service voip  
allow-connections sip to h323
- B. voice service voip

- sip  
bind media source-interface Vlan100
- C. voice service voip  
sip  
bind control source-interface Vlan100
- D. voice service voip  
sip  
registrar server expires max 600 min 60

**Answer:** D

**QUESTION 143**

An administrator is configuring Cisco UCM and the system to send \*.webex.com traffic to a Cisco UCM Session Management Edition cluster. The administrator wants to limit which endpoints can reach \*.webex.com. Which two items must the administrator configure for the SIP route pattern? (Choose two.)

- A. calling party transformation
- B. partition of the SIP route pattern
- C. connected party transformation
- D. called party URI transformation
- E. destination SIP trunk of the SIP route pattern

**Answer:** BE

**Explanation:**

[https://www.cisco.com/c/en/us/td/docs/telepresence/tcs/7\\_1/admin/administration/tcs\\_7\\_0/cucm.html](https://www.cisco.com/c/en/us/td/docs/telepresence/tcs/7_1/admin/administration/tcs_7_0/cucm.html)

**QUESTION 144**

An administrator deployed a third-party H.323 gateway in a voice environment, but users report call failures when using features like call hold or call transfer. What are two reasons that these features fail? (Choose two.)

- A. The CSS of the transfer initiating line does not contain the partition of the supplementary feature extension (DirectTransfer or MoH Number).
- B. The MTP that is configured for use within the H.323 gateway configuration is configured as a trusted source, but the third-party gateway does not trust the signing root CA certificate of the MTP certificate.
- C. The MTP does not support the negotiated codec, and media renegotiating during the call is not supported.
- D. The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP.
- E. The third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.

**Answer:** CD

**QUESTION 145**

Refer to the exhibit. A collaboration engineer is troubleshooting an issue where a user of a Cisco UCM IP phone reports failed calls when trying to dial out to the PSTN. Which action resolves the issue?



- Assign a calling search space to the line or the device that has access to the route pattern.
- Deselect "Block this pattern" on the "Route Option" setting of the route pattern.
- Select the "Urgent Priority" setting on the route pattern.
- Instruct the user to not dial a "1" before their local area code.

**Answer: A**

**QUESTION 146**

Customers that call into a company's IVR report that when they try to select an option, none of the prompts work. The administrator determines that the calls are coming in across an H.323 gateway. While analyzing the dial peer that points toward Cisco UCM, the administrator notices that no DTMF method is configured. Which command resolves this issue?

- dial-peer voice 2 voip  
dtmf-relay sip-kpml
- dial-peer voice 2 voip  
dtmf-relay h245-alphanumeric
- dial-peer voice 2 voip  
dtmf-relay sip-notify
- dial-peer voice 2 pots  
dtmf-relay h245-alphanumeric

**Answer: B**

**QUESTION 147**

An engineer is configuring a Cisco Collaboration system for SIP endpoints and must enable Survivable Remote Site Telephony for these endpoints. Which code completes this configuration on the SRST gateway?

- call-manager-fallback  
max-conferences 8 gain -6  
ip source-address 10.10.10.100 port 2000  
max-ephones 100  
max-dn 200
- telephony-service  
max-conferences 8 gain -6  
ip source-address 10.10.10.100 port 2000  
max-ephones 100  
max-dn 200
- voice service voip  
default mode secure  
address hiding  
allow-connections sip to sip

sip registrar

- D. voice register global  
default mode  
no allow-hash-in-dn  
max-dn 100  
max-pool 200

**Answer:** A

**QUESTION 148**

Management wants to change the initial announcements for one of the existing call hunt groups. A new set of announcement audio file was provided. Which two configuration steps must the administrator take to accomplish this change? (Choose two.)

- A. Identify the MOH audio source ID associated to one of the line group member's "Network Hold MOH Audio Source".
- B. Identify the MOH audio source ID associated to "Network Hold Source & Announcements" under the Queuing section of the hunt pilot.
- C. Identify the configured announcement names to change under the MOH audio source section, then upload the new files to the respective announcements under the Announcement section.
- D. Identify the MOH audio source ID associated to one of the line group member's "User Hold MOH Audio Source".
- E. Identify the configured announcement names to change under the Announcement section, and assign the uploaded files to the Queueing section of the hunt pilot.

**Answer:** BC

**Explanation:**

<https://community.cisco.com/t5/unified-communications-infrastructure/cucm-11-0-hunt-pilot-queue-custom-initial-announcement-not/td-p/4008782>

**QUESTION 149**

An engineer is troubleshooting an intersite call between two endpoints where the call fails and the message "Not Enough Bandwidth" is displayed. G.729 codec is in use on both sites. First, calls are being properly routed, and the issue happens after the third call is established and the bandwidth utilization between the two sites is under 50%. Which configuration in Cisco UCM must be adjusted to resolve the issue?

- A. transcoder
- B. location
- C. route pattern
- D. translation pattern

**Answer:** B

**QUESTION 150**

Refer to the exhibit. An engineer is trying to set up a new deployment using the SIP provider with TCP for signaling. After troubleshooting, the customer notices that the matching incoming dial peer and outgoing dial peer did not generate the INVITE to the SIP provider. Why is the call failing?

```

2899152: Mar 31 21:23:16.777: TC87F7F9668A9E8 setting property TCP_ALWAYS_PUSH (15) 7F7F94803528
2899153: Mar 31 21:23:16.777: tcp_uniqueport: using ephemeral max 65535
2899154: Mar 31 21:23:16.777: TCP: Random local port generated 30202, network 1
2899155: Mar 31 21:23:16.777: TC87F7F9668A9E8 bound to 10.10.10.40.30202
2899156: Mar 31 21:23:16.777: Reserved port 30202 in Transport Port Agent for TCP IP type 1
2899157: Mar 31 21:23:16.777: TCP: sending SYN, seq 974312871, ack 0
2899158: Mar 31 21:23:16.777: TCP0: Connection to 10.10.10.14:5060, advertising MSS 536
2899159: Mar 31 21:23:16.777: TCP0: state was CLOSED -> SYNSENT [30202 -> 10.10.10.14(5060)]
2899160: Mar 31 21:23:18.777: TCP0: RETRANS timeout timer expired
2899161: Mar 31 21:23:18.777: 10.10.10.40:30202 <-> 10.10.10.14:5060 congestion window changes
2899162: Mar 31 21:23:18.777: cwnd from 536 to 536, ssthresh from 65535 to 1072
2899163: Mar 31 21:23:18.777: TCP0: timeout #1 - timeout is 4000 ms, seq 974312871
--More--      2899164: Mar 31 21:23:18.777: TCP: (30202) -> 10.10.10.14(5060)
2899165: Mar 31 21:23:19.503: TCP866: state was ESTAB -> FINWAIT1 [22 -> 10.16.8.226(52145)]
2899166: Mar 31 21:23:19.503: TCP866: sending FIN
2899167: Mar 31 21:23:19.536: TCP866: state was FINWAIT1 -> FINWAIT2 [22 -> 10.16.8.226(52145)]
2899168: Mar 31 21:23:19.536: TCP866: FIN processed
2899169: Mar 31 21:23:19.536: TCP866: state was FINWAIT2 -> TIMEWAIT [22 -> 10.16.8.226(52145)]
2899170: Mar 31 21:23:21.777: Released port 30202 in Transport Port Agent for TCP IP type 1 delay 240000
2899171: Mar 31 21:23:21.777: TCP0: state was SYNSENT -> CLOSED [30202 -> 10.10.10.14(5060)]
2899172: Mar 31 21:23:21.777: TCB 0x7F7F9668A9E8 destroyed
  
```

- A. The ITSP is sending FIN.
- B. The ITSP is not sending the ACK.
- C. The ITSP is sending CLOSE state.
- D. The ITSP is not answering the SYN.

**Answer: D**

**QUESTION 151**

An administrator is configuring a SIP trunk to an ITSP. The SIP connection will traverse from a Cisco UCM to the ISTP through a Cisco Unified Border Element. The ITSP has indicated that they require an in-band method for DTMF. Which command on the outbound dial-peer to the ITSP will meet this requirement?

- A. router (config-dial-peer) dtmf-relay sip-notify
- B. router (config-dial-peer) dtmf-relay sip-kpml
- C. router (config-dial-peer) dtmf-relay h245-alphanumeric
- D. router (config-dial-peer) dtmf-relay rtp-nte

**Answer: D**

**QUESTION 152**

A new solution is configured to support internal, local, and international calling. Calling [+44 1111 1111] from one of the registered internal phones does not work. Local and internal calls seem to work without any problems. The configuration has patterns configured to match the failing dialed number [+44 ? ?]. The other configured patterns show [2] for internal numbers and [55] for local numbers. International numbers use E.164 as recommended. What is missing to make this solution work?

- A. 001 or 00 must be used instead of the + sign on Cisco UCM
- B. =+ cannot be used in a route pattern, only in a SIP pattern
- C. \ in front of the +
- D. / in front of the +

**Answer: C**

**QUESTION 153**

A customer must advertise some numbers through GDPR on Call Manager. The Cisco UCM already is advertising some URIs to other clusters with GDPR, but the customer wants to ensure that specific numbers are advertised. Where under the Cisco UCM Admin page > Call Routing > Global Dial Plan Replication must that set of numbers be configured?

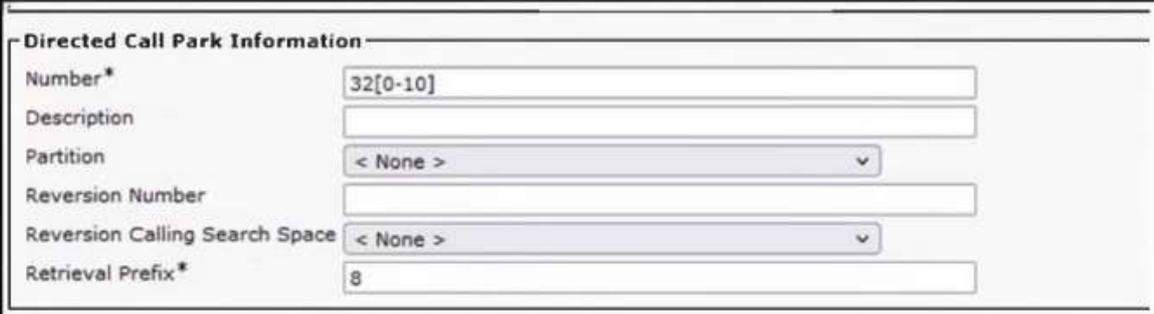
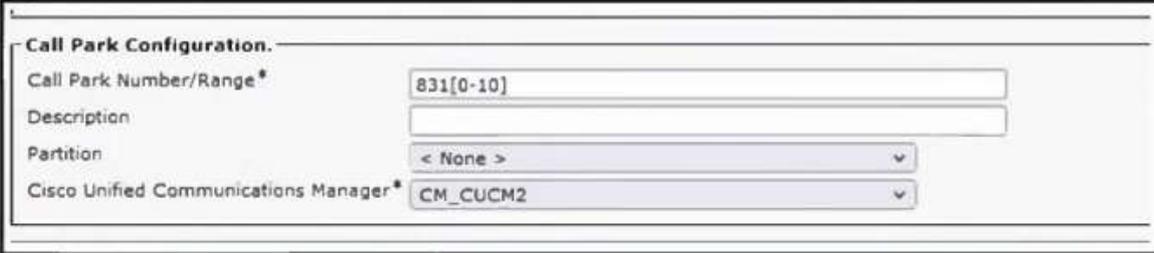
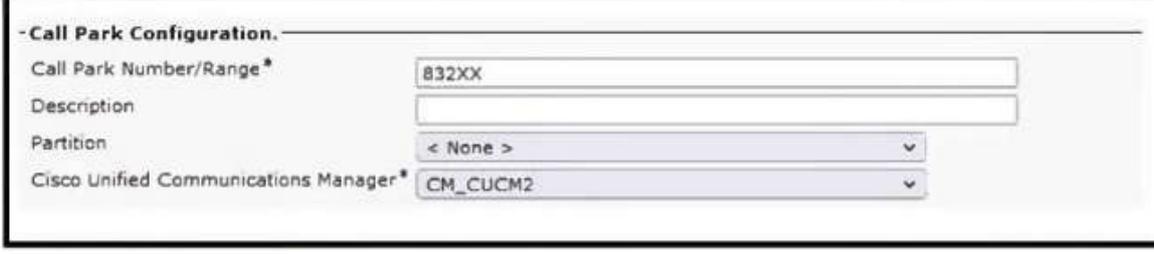
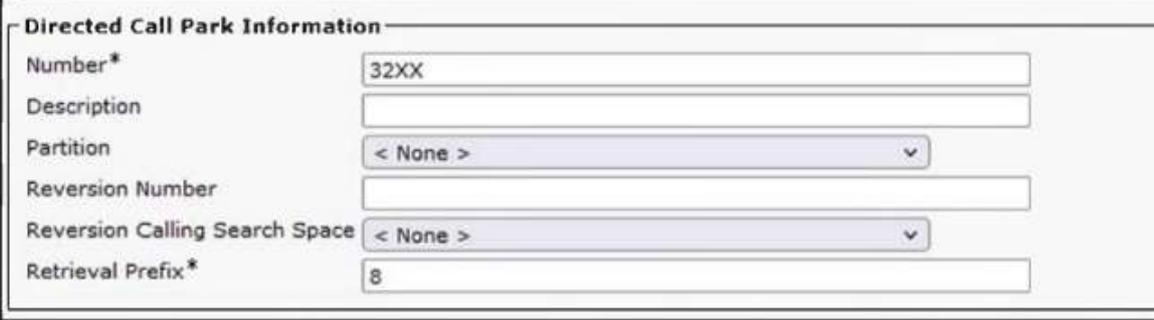
- A. Route Numbers
- B. Advertised Patterns

- C. Learned Numbers
- D. Route Patterns

**Answer: B**

**QUESTION 154**

A new Cisco UCM deployment is configured, and now the customer requires a set of 10 DNSs to park the calls starting with 8. The users must use the softkey to park the call. Which configuration must be performed to achieve this requirement?

- A. 
- B. 
- C. 
- D. 

**Answer: B**

**QUESTION 155**

An administrator troubleshoots call failure in a new deployment and finds that the SIP INVITE messages sent to the service provider contain a diversion header with the user's 4-cigit directory number. These 4-digit directory numbers range from 1000 to 9899. The service provider is rejecting the calls because it requires that the diversion header contain 10 digits. Which command on the Cisco Unified Border Element resolves this issue for all users?

- A. voice class sip-profiles 105

- request INVITE sip-header Diversion modify "andlt; sip:1(...)@"andlt; sip:263411\1@"
- B. voice class sip-profiles 105  
request INVITE sip-header Diversion modify "andlt; sip:(...)@"andlt; sip:263411\1@"
- C. voice class sip-profiles 105  
request INVITE sip-header Diversion add "andlt; sip:(...)@"andlt; sip:263411\1@"
- D. voice class sip-profiles 105  
request INVITE sip-header Diversion modify "andlt; sip:1(...)@"andlt; sip:263411\2@"

**Answer: B**

**QUESTION 156**

Some users report having issues dialing some external numbers when traveling to other locations within the company. The company has five locations in five cities in one country and has an egress gateway in each location for TEHO. The configuration has no specific entry stating that the roaming users are using the local gateway, but calls are going out. How is a verification of the call routing in such a specific configuration performed to further identify the problem?

- A. device mobility
- B. standard local route group
- C. local route groups
- D. TEHO

**Answer: A**

**QUESTION 157**

Refer to the exhibit. An administrator configured a new SIP trunk between Cisco UCM and Cisco Unified Border Element. Calls that use this new trunk are failing, and debugs from the Cisco Unified Border Element do not show any signaling received from Cisco UCM. The administrator verified that OPTIONS is enabled on the SIP profile configured on the trunk on Cisco UCM. The route pattern matched for this call is correctly configured to send calls to the Cisco Unified Border Element using this new trunk. What is the cause of this failure?

```

03833226.000 |11:10:55.116 |SdISig |DaRes |info_da
|Cdcc(1,100,39,25) |Da(1,100,47,1) |[1,100,251,7289.65^14.48.81.100^* |[R:N-
H:0,N:0,L:0,V:0,Z:0,D:0] CI=28377985 Dial ForegoPotentialMatches OffNet
PretransformCgpn=tn=Onpi=Oti=1nd=+19195741001pi=Osi1
Cgpn=tn=Onpi=Oti=1nd=+19195741001pi=Osi1 DialingPartition=717ae636-e10b-2d56-6420-
8b1099f0fdb9 DialingPattern=91.[2-9]XXXXXX DialingRPRE={91}([2-9][0-9][0-9][0-9][0-9][0-9])
PID=(1,181,19) PretransformCgpn=tn=Onpi=Oti=1nd=915745000pi=Osi0 PretransformTags=ACCESS-
CODE:SUBSCRIBER PretransformPos=91:5745000 CollectedDigits=tn=Onpi=Oti=1nd=5745000pi=Osi0
Tags=SUBSCRIBER Pos=5745000 CalledPartyDevicePoolPkid= eccTransactionId=0
isCdpnNonPreemptable=F isCgpnNonPreemptable=F patternUsage =5 requestID =0
03833226.001 |11:10:55.116 |AppInfo |processCCMFeatureData: operationIdd=0
03833226.002 |11:10:55.116 |AppInfo |findUnfiredInterceptOnPattern numOfPatterns = 0
---
03833229.001 |11:10:55.116 |AppInfo |LineCdpc(46): -dispatchToAllDevices-, sigName=CcProceedReq,
device=SEP2834A2824611
---
03833232.001 |11:10:55.116 |AppInfo |LBMIF: CI: 28377985 ASSOC 28377986
03833232.002 |11:10:55.116 |AppInfo |LBMIF: CI: 28377986 ASSOC 28377985
---
03833235.001 |11:10:55.119 |AppInfo |SIPStationCdpc: CcProceedReq -
unicodeConnectedUnicodeDisplayName="" asciiConnectedDisplayName=""
03833235.002 |11:10:55.119 |AppInfo |SIPStationCdpc: CcProceedReq - unicodeCallingPartyName=""
asciiCallingPartyName="" callingParty="+19195741001" unicodeCalledPartyName=""
asciiCalledPartyName="" calledParty="5745000"
---
03833240.000 |11:10:55.119 |SdISig |CcRejInd |wait |Cc(1,100,38,1)
|SIPD(1,100,181,19) |[1,100,251,7289.65^14.48.81.100^* |[R:N-H:0,N:1,L:0,V:0,Z:0,D:0]
CI=28377986 Cl.branch=0 c.l=1 c.cid=8 c.cs=0 c.lc=0 c.r=0 CV=1459617833 rejectType=1 dtmDisconn=F
FDataType=0 opId=0 ssType=0 SsKey=0 invokeId=0 resultExp=F bpda=F transparentData=null
CanSupportSIPAndN=false TransId=0 AllowBitMask=0x0 UserAgentOrServer= OrigDDName=locale: 1
Name: UnicodeName: pi: 0 mCallerId= mCallerName= TrustRaid=T misMultiForkingEnabled=F
mediaCause=0 CAL={v=#####, m=#####, tDev=F, res=F, devType=0} CAL={v=#####6, m=#####6, tDev=F,
res=F, devType=0}

```

- A. The calling search space on the calling line/device does not contain the partition of the route pattern
- B. The called party number is not complete because it contains only 7 digits
- C. A connectivity issue between Cisco UCM and Cisco Unified Border Element prevents Cisco UCM from receiving a response to the OPTIONS it is sending to Cisco Unified Border Element
- D. There are multiple matches for the dialed number, and Cisco UCM is trying to send the call to SEP2834A2824611 rather than Cisco Unified Border Element

**Answer: C**

#### QUESTION 158

Refer to the exhibit. An administrator troubleshoots an issue where each user's 4-digit directory number is presented as the calling number when dialing out to the PSTN. This 4-digit calling number prevents the external user from calling the internal user back. Which two actions must the administrator perform to resolve the issue? (Choose two.)

```

Results Summary
  Calling Party Information
  • Dialed Digits = 915745000
  • Match Result = RouteThisPattern
  Matched Pattern Information
  • Called Party Number = 5745000
  • Time Zone = Etc/DMT
  • End Device = CSR1000v
  • Call Classification = OffNet
  • InterDigit Timeout = NO
  • Device Override = Disabled
  • Outside Dial Tone = NO

Call Flow
  Route Pattern (Pattern= 91.[2-9]XXXXXX)
    Positional Match List =
    DialPlan =
    Route Filter
    • Require Forced Authorization Code = No
    • Authorization Level = 0
    • Require Client Matter Code = No
    • Call Classification =
    • PreTransform Calling Party Number = 1001
    • PreTransform Called Party Number = 915745000
    Calling Party Transformations
      • External Phone Number Mask = NO
      • Calling Party Mask =
      • Prefix =
      • CallingLineId Presentation = Default
      • CallingName Presentation = Default
      • Calling Party Number = 1001
    ConnectedParty Transformations
    Called Party Transformations
  Device (Type= SIPTrunk)
    • End Device Name = CSR1000v
    • PortNumber =
    • Device Status = UnKnown
    • AAR Group Name =
    • AAR Calling Search Space =
    • AAR Prefix Digits =
    • Call Classification = Use System Default
    • Calling Party Selection = Originator
    • CallingLineId Presentation = Default
    • CallerID DN =

Alternate Matches
  
```

- Update the SIP profile on the SIP trunk by checking "enable external presentation name and number"
- Change the calling line ID presentation on the SIP trunk to allow.
- Create a calling party transformation pattern that matches the desired directory number and add a mask to modify the calling number
- Create a called party transformation pattern that matches the desired directory number and add a mask to modify the calling number
- Add a calling party transformation CSS to the SIP trunk.

Answer: CE

QUESTION 159

Refer to the exhibit. An administrator is trying to test outbound calls toward the ITSP but cannot complete the call and receives a SIP error. ITSP is consulted, and the issue is that the ANI that is being sent is not the DID provided 8005532447. Which configuration change sends the correct ANI on the INVITE sent to ITSP to fix the error?

```

Sent:
INVITE sip:9011919999321453@cisco.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.14:5060;branch=z9hG4bKFE45715AD
Remote-Party-ID: "User A" <sip:4444@cisco.com>;party=calling;screen=no;privacy=off
From: "User A" <sip:4444@cisco.com>;tag=E141986A-D65
To: <sip:9011919999321453@cisco.com>
Date: Thu, 31 Jan 2019 19:34:47 GMT
Call-ID: 1A0EC126-24C611E9-81D8D031-86D18CDB@cisco.com
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 1127492352-0000065536-0000014911-0203951114
User-Agent: Cisco-SIPGateway/IOS-15.4.3.S7
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1548963287
Contact: <sip:4444@192.168.1.14:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 668

v=0
o=CiscoSystemsCCH-SIP 113060473 1 IN IP4 192.168.1.16
s=SIP Call
c=IN IP4 192.168.1.14
b=TIAS:384000
b=AS:384
t=0 0
m=audio 24500 RTP/AVP 9 0 8 116 18 101
-Output omitted

007008: Jan 31 19:34:47.246: //1655739/43342800000/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 403 From: URI not recognized
Via: SIP/2.0/UDP 192.168.1.14:5060;branch=z9hG4bKFE45715AD
From: "User A" <sip:4444@192.168.2.1>;tag=E141986A-D65
To: <sip:9011919999321453@cisco.com>;

Call-ID: 1A0EC126-24C611E9-81D8D031-86D18CDB@cisco.com
CSeq: 101 INVITE
Timestamp: 1548963287
Server: Provider/2.0
Content-Length: 0

```

- A. voice class sip-profiles 2  
request INVITE sip-header To modify "sip:(.\*)@" sip:8005532447@
- B. voice translation-rule 3  
rule 1/.\*/ /8005532447/
- C. voice class sip-profiles 1  
request INVITE sip-header Diversion modify "sip:(.\*)@" "sip:8005532447@"
- D. voice translation-rule 4  
rule 1/^\(8005532447\) / \1/

**Answer: B**

**QUESTION 160**

An administrator has configured two route patterns, 9.911 and 9.[2-9]XXXXXX. When a user dials 9911. Cisco UCM waits for the T302 timer before routing the call. How will the administrator force interdigit timeout and route the call as soon as the user has finished dialing 9911, without waiting for the T302 timer to expire?

- A. decrease the T302 timer in Service Parameters from the default value
- B. enable Urgent Priority on the 9.[2-9]XXXXXX pattern
- C. enable Urgent Priority on the 9.911 pattern
- D. enable Device Override on both route patterns

**Answer: C**

**QUESTION 161**

Calls are not working when sent from a Cisco Unified Border Element to a service provider. After investigating the logs, the engineer notes that the Cisco Unified Border Element is sending the extension only. How is the issue addressed in the configuration?

- A. voice class request sip-header diversion
- B. sip-header contact modify
- C. voice class request sip-header modify
- D. request invite sip-header diversion modify

**Answer: D**

**QUESTION 162**

Phone A calls to phone B, but phone B has Call Forward All set to a PSTN number. The route list responsible for the phone B call to the PSTN has a standard local route group configured. Which route group must be used to send the call to the PSTN?

- A. route group defined in Standard Local Route Group section of Cisco UCM service parameters
- B. route group from the phone B device pool
- C. route group from the phone A device pool
- D. standard local route group defined in the route group configuration

**Answer: C**

**QUESTION 163**

Refer to the exhibit. A collaboration engineer is troubleshooting an issue where external callers cannot leave voicemail messages. Also, internal users report hearing the reorder tone (fast busy) when they attempt to retrieve voicemail messages from their Cisco IP phones. Which action resolves the issue?

Destination						
Destination Address in an SRV						
Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration	
* sip.cisco.com		0	down	local=3	Time Down: 0 day 0 hour 59 minutes	

- A. Verify that the correct port numbers are used for the SIP trunk.
- B. Ensure that the SIP Trunk Security Profile is configured to use UDP for transport.
- C. Start the Cisco Call Manager service at the destination.
- D. Ensure that Cisco UCM can resolve the destination address via DNS.

**Answer: D**

**QUESTION 164**

A solution for a large company is being reconfigured to optimize for cost saving. The company has an extensive global QoS-enabled network with enough bandwidth to create a converged network. Local calls are relatively inexpensive in countries the company have operations, but long distance and international calls are expensive. Which type of configuration supported by Cisco UCM would help optimize cost control for this company?

- A. standard local route groups for mobile users
- B. Mobile and Remote Access
- C. high complex codec support like G.729 to minimize bandwidth usage
- D. tail end hop off

**Answer: D**

**QUESTION 165**

Refer to the exhibit. A collaboration engineer is troubleshooting an issue where the PSTN calls of a Cisco UCM IP phone user are not reaching the PSTN gateway. Which action resolves the issue?

Cisco Unified Communications Manager Dialed Number Analyzer

DNA Analysis Output
Save the Displayed Output

**Cisco Unified Communications Manager Dialed Number Analyzer Results**

▼ **Results Summary**

- ▼ **Calling Party Information**
  - Calling Party = 4125551212
  - Partition =
  - Device CSS =
  - Line CSS = ONNET-CSS
  - AAR Group Name =
  - AAR CSS =
- Dialed Digits = 914125550000
- Match Result = RouteThisPattern
- ▼ **Matched Pattern Information**
  - Pattern = 9.1[2-9]XX[2-9]XXXXXX
  - Partition =
  - Time Schedule =
- Called Party Number = 14125550000
- Time Zone = Etc/GMT
- End Device = HQ-RL
- Call Classification = DRNet
- InterDigit Timeout = NO
- Device Override = Disabled
- Outside Dial Tone = NO

▼ **Call Flow**

▼ **Alternate Matches**

NOTE: The analysis results are purely based on configurations available in the Cisco Communications Manager database. For Gateway outbound calls, call details might differ depending on the gateway's settings.

▼ **Call Flow**

- ▼ **Route Pattern : Pattern=9.1[2-9]XX[2-9]XXXXXX**
  - Positional Match List =
  - DialPlan =
  - ▼ **Route Filter**
    - Filter Name =
    - Filter Clause =
  - **Require Forced Authorization Code= NO**
  - **Authorization Level= 0**
  - **Require Client Matter Code=NO**
  - **Call Classification =**
  - **PreTransform Calling Party Number = 4125551212**
  - **PreTransform Called Party Number = 914125550000**
  - ▼ **Calling Party Transformations**
    - External Phone Number Mask = NO
    - Calling Party Mask =
    - Prefix =
    - CallingLineId Presentation = Default
    - CallingName Presentation = Default
    - Calling Party Number = 4125551212
  - ▼ **ConnectedParty Transformations**
    - ConnectedLineId Presentation = Default
    - ConnectedName Presentation = Default
  - ▼ **Called Party Transformations**
    - Called Party Mask =
    - Discard Digits Instruction = PreDot
    - Prefix =
    - Called Number = 14125550000
  - ▼ **Route List :Route List Name= HQ-RL**
    - ▼ **RouteGroup: RouteGroup Name= Standard Local Route Group**
      - **PreTransform Calling Party Number = 4125551212**
      - **PreTransform Called Party Number = 914125550000**
      - ▼ **Calling Party Transformations**
        - External Phone Number Mask = Default
        - Calling Party Mask =
        - Prefix =
        - Calling Party Number = 4125551212
      - ▼ **Called Party Transformations**
        - Called Party Mask =
        - Discard Digits Instruction =
        - Prefix =
        - Called Number = 914125550000
    - ▼ **Alternate Matches**
      - Note: Information Not Available

- A. Change the calling search space of the user's line or device.
- B. Change the "Call Classification" to "OnNet" on the route pattern.

- C. Ensure that the user's phone is assigned to a device pool with the correct local route settings.
- D. Deselect "Block this pattern" on the route pattern.

**Answer: C**

**Explanation:**

The DNA output clearly says "RouteThisPattern" in the match result, which means that "Block This Pattern" is not checked. The 9 is being stripped by the Discard Digit in the Called Party Transformation section being set to PREDOT. Then it is added again in the Called Party Transformations assigned to the RL's Standard Local Route Group. The user dialed the 9, so CSS has already matched an appropriate Pattern, what needs to be done is either the PREDOT on the Route Pattern needs to be removed or the Called Party Transformation needs to be removed.

#### **QUESTION 166**

An engineer is implementing survivability for a collaboration environment. The environment is utilizing a centralized Cisco UCM at the headquarters and Cisco IOS-XE gateways in the remote branches. Which action must the engineer take to enable both shared lines and B-ACD for SIP branch phones during a WAN outage?

- A. Configure mode esrst under voice register global configuration mode.
- B. Configure mode srst under telephony-service configuration mode.
- C. Configure mode esrst under voice service voip configuration mode.
- D. Configure mode esrst under telephony-service configuration mode.

**Answer: A**

#### **QUESTION 167**

Users report silence on the line when they try to connect to external voice numbers. The company is using Cisco 8865 phones with Cisco UCM connected to a gateway with two T1 circuits. The MGCP server seems to be responding. Which action must the engineer take to begin to investigate the media for this type of issue?

- A. Run RTMP on the Cisco router.
- B. Verify that the MGCP service is running under serviceability in Cisco UCM.
- C. Run the debug mgcp media command on the Cisco router.
- D. Run the debug H.323 gateway command on the Cisco router.

**Answer: C**

#### **QUESTION 168**

What is a function of the metadata carried in SIP sessions between the recording client and the recording server?

- A. It forks RTP media to the recorder.
- B. It provides advanced capabilities, such as speech analytics.
- C. It sets up a new SIP session
- D. It identifies the participant change due to transfers during the call.

**Answer: D**

#### **QUESTION 169**

Some users report poor quality when they travel to another continent and make calls. This issue applies only to one continent and not to others, where typically the dialing is fast and quality is clear. Users experience the same result at home when they call the same numbers in that specific continent. It seems like some users do not exist in the correct PSTN gateway when making calls to a specific country. The company is using TEHO to save on the cost of international or long-distance calling.

They are also using a globalized dial plan. What is the cause of the issue?

- A. CUBE is not configured for TEHO in the specific country.
- B. A local route group is not added to the route pattern.
- C. The users are missing this specific gateway at the device pool level.
- D. Regions in Cisco UCM are not configured correctly.

**Answer:** D

**QUESTION 170**

An administrator is configuring Meet-me conferencing in a Cisco UCM deployment and has created the Meet-me number and ensured that it is in a partition accessible by all devices. Which two additional steps must the administrator perform? (Choose two.)

- A. Ensure that conferencing-initiating devices are using a media resource group list that contains at least one Cisco UCM conference bridge.
- B. Disable Early Media on the SIP profile of all devices that will use Meet-me conferencing.
- C. Enable Meet-me conferencing in enterprise parameters.
- D. Ensure that all devices have G.729 enabled.
- E. Update the softkey template on all phones to ensure that they contain the Meet-me softkey.

**Answer:** AE

**QUESTION 171**

Which elements does Cisco cloud mobility for collaboration include?

- A. Cisco Webex Collaboration Cloud Services
- B. Cisco Collaboration Cloud
- C. Cisco Collaboration Cloud and Cisco Webex Collaboration Cloud Services
- D. Cisco Collaboration Cloud and Cisco Mobile and Remote Access Collaboration Cloud Services

**Answer:** C

**Explanation:**

Cisco Cloud Mobility for Collaboration includes both the Cisco Collaboration Cloud (which provides core collaboration services such as voice, video, and presence) and Cisco Webex Collaboration Cloud Services (which support messaging, meetings, and team collaboration). Together, these components enable seamless communication across devices and locations.

**QUESTION 172**

A company is using Cisco Jabber on-premises to make B2B calls on video. The calls are using Cisco Expressway-C and Expressway-E and have been configured in Cisco UCM to be able to call any URI on the internet. The Jabber client also has voice enabled and must be able to call local, regional, and international numbers. Where must Cisco UCM be configured to meet this requirement for URI dialing?

- A. Enter "!#" in the SIP route pattern.
- B. Enter ".\*" in the route pattern section tied to a route group and list.
- C. Enter "\*" in the SIP route pattern.
- D. Enter "!#" in the route pattern section tied to a route group and list.

**Answer:** A

**Explanation:**

To allow URI dialing to any domain over SIP (for B2B video or voice calls via Expressway), Cisco UCM must have a SIP route pattern that matches any URI format. The pattern !# is used in SIP route patterns to match any SIP URI, enabling routing of calls like user@example.com to the appropriate SIP trunk (e.g., to Expressway-C).

**QUESTION 173**

The sales department must answer phones when other sales members are not at their desks. The administrator knows that configuring Call Pickup allows the sales users to answer all the calls in the department by pressing only the softkey. Which call pickup configuration meets this requirement?

- A. Standard Call Pickup
- B. Group Call Pickup
- C. Other Group Call Pickup

D. Directed Call Pickup

**Answer: B**

**QUESTION 174**

An administrator configured Extension Mobility on Cisco UCM. Users report that logins are successful, but the phones do not have an Extension Mobility option after logging in. The administrator verified that Extension Mobility is enabled on the devices and that the log-out profile is valid. Which action must the administrator take to resolve the issue?

- A. Subscribe all device profiles to the Extension Mobility phone service.
- B. Delete the identity trust list file from the phone(s).
- C. Change the Extension Mobility URL from HHTP to HTTPS.
- D. Restart the Cisco Extension Mobility and Cisco Extension Mobility application services.

**Answer: A**

**QUESTION 175**

Which built-in bridge configuration option can be set on the individual IP phones to ensure that the cluster-wide service parameter is used?

- A. automatic
- B. default
- C. on
- D. off

**Answer: B**

**QUESTION 176**

An engineer has a hunt group with some overflow and many calls are going to voicemail. To reduce the number of calls forwarded to voicemail, the engineer must create Call Queuing to make callers wait before going to voicemail. How will Call Queuing affect Forward Hunt No Answer and Forward Hunt Busy?

- A. Forward Hunt No Answer and Forward Hunt Busy must be enabled while Call Queuing is enabled.
- B. After Call Queuing is enabled, Forward Hunt No Answer and Forward Hunt Busy must be disabled manually.
- C. Before Call Queuing is enabled, Forward Hunt No Answer and Forward Hunt Busy must be disabled manually.
- D. When Call Queuing is enabled, Forward Hunt No Answer and Forward Hunt Busy are disabled automatically.

**Answer: D**

**QUESTION 177**

When route patterns are defined as precisely as possible on a Cisco UCM, the control and reliability of the calling plan is increased. A route pattern must be added to support calls to this number range: 9135200 to 9135205. The calls are on-net and no translation patterns are configured. Which configuration meets these requirements?

- A. 913520[012345]
- B. 913520X
- C. 913520[1-5]
- D. 913520!

**Answer: A**