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

Exam A

QUESTION 1
Where is the dtmf-relay command configured on Cisco Unified Border Element?

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

Correct Answer: B
Section:

QUESTION 2
Refer to the exhibit.

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

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

A. Change the destination-pattern on the outgoing dial peer to match "444333222".

B. Set up translation-profile on the incoming dial peer to match incoming traffic.

C. Create specific matching for ''222333444'' on the incoming dial peer.

D. Fix the voice translation-rule to match specifically number ''222333444'' and change it to ''444333222''.

Correct Answer: B
Section:

QUESTION 3

Refer to the exhibit.

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

Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border Element router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied, which is shown in the exhibit. Which rule modified DNIS in the format that the provider is expecting?

A. rule 1 /^/+\([^1].*\)/ /011\1/

B. rule 1/^\+1\([2-9]..[2-9]......$\)/ /\1/

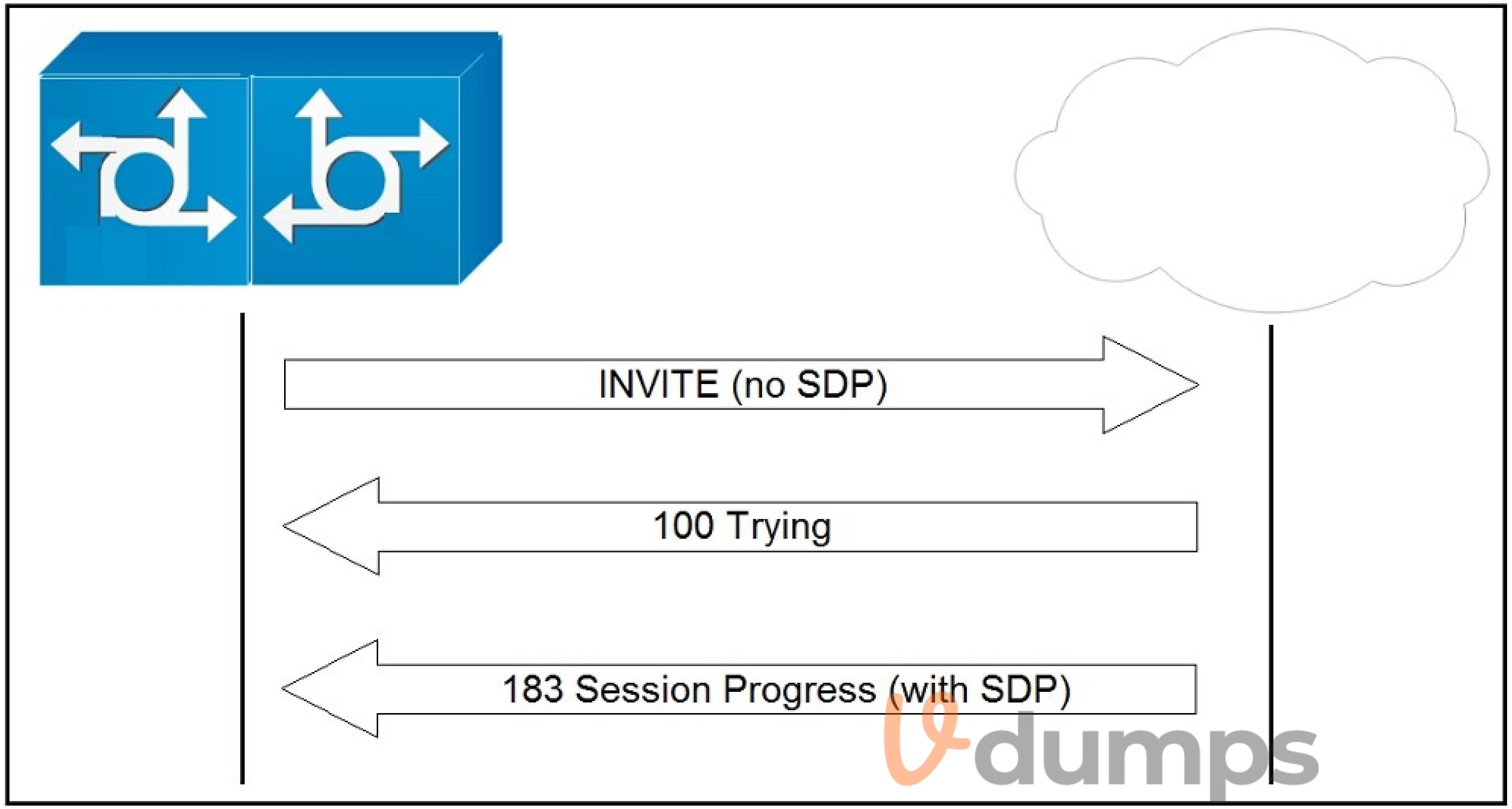
C. rule 1 /^\([2-9]..[2-9]......$\)/ /\1/

D. rule 1 /^\+1\([2-9]..[2-9]......$\)/ /\0/

Correct Answer: B
Section:

QUESTION 4

Refer to the exhibit.



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

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

Correct Answer: A, B
Section:

QUESTION 5
Which IOS command creates a SIP-enabled dial peer?

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

Correct Answer: B

Section:

QUESTION 6
A user in location X dials an extension at location Y. The call travels through a QoS-enabled WAN network, but the user experiences choppy or clipped audio. What is the cause of this issue?

A. missing Call Admission Control
B. codec mismatch
C. ptime mismatch
D. phone class of service issue

Correct Answer: A
Section:

QUESTION 7
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 8
Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.
B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.
E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Correct Answer: A, B
Section:

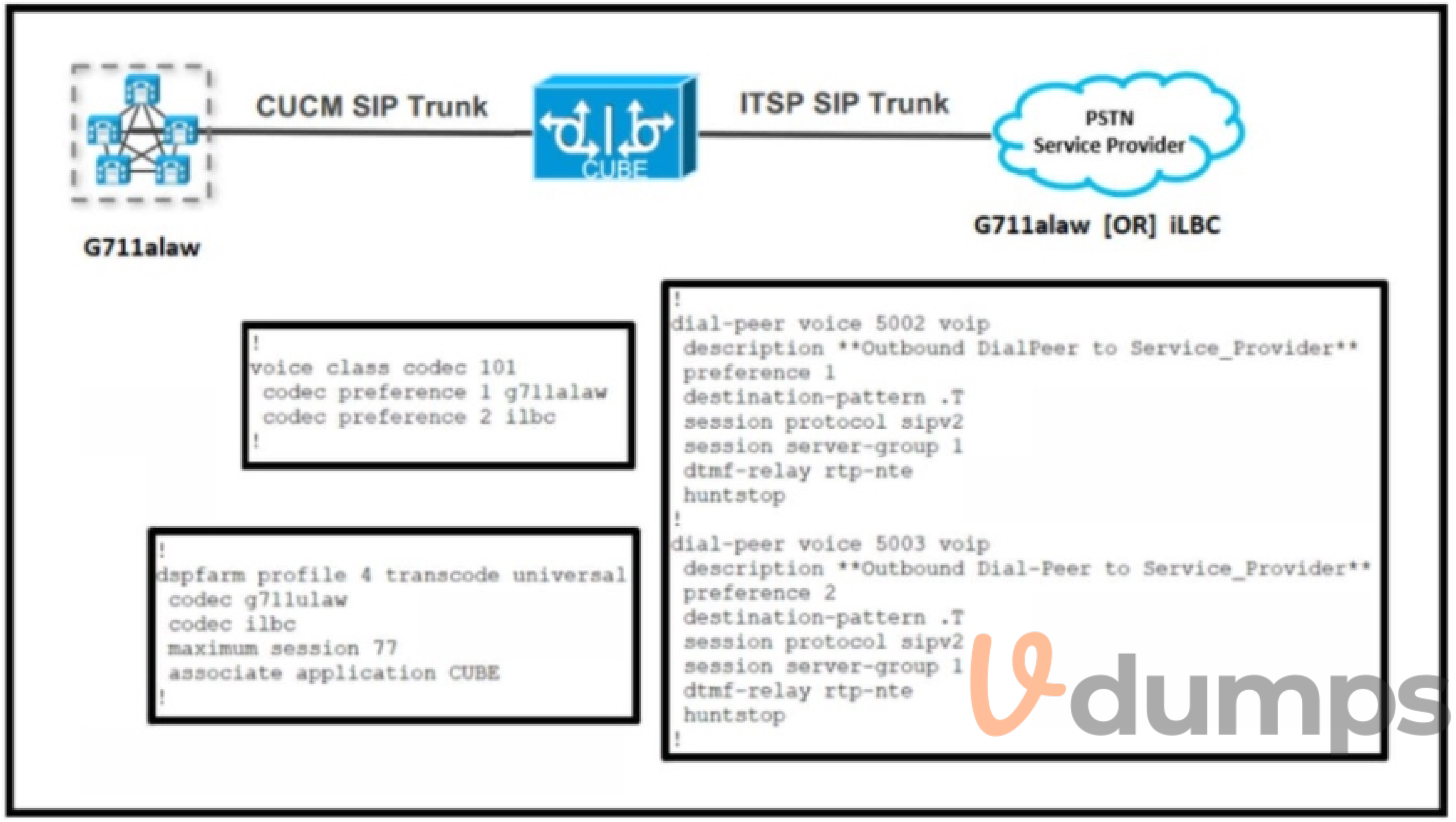
QUESTION 9
An engineer is troubleshooting local ringback on a Cisco SIP gateway The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified m the SDP Which configuration change must be made on the gateway to resolve the issue?

A. Router(conf-voi-serv)# dlisable-early-media 180
B. Router(conftg-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

Correct Answer: B

Section:

QUESTION 10
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?

- dial-peer voice 5002 voip
  codec g711alaw ilbc
  !
  dial-peer voice 5003 voip
  codec g711alaw ilbc
- dial-peer voice 5002 voip
  voice-class codec 101 offer-all
  !
  dial-peer voice 5003 voip
  voice-class codec 101 offer-all
- dial-peer voice 5002 voip
  codec g711alaw
  !
  dial-peer voice 5003 voip
  codec ilbc
- dial-peer voice 5002 voip
  voice-class codec 101
  !
  dial-peer voice 5003 voip
  voice-class codec 101

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

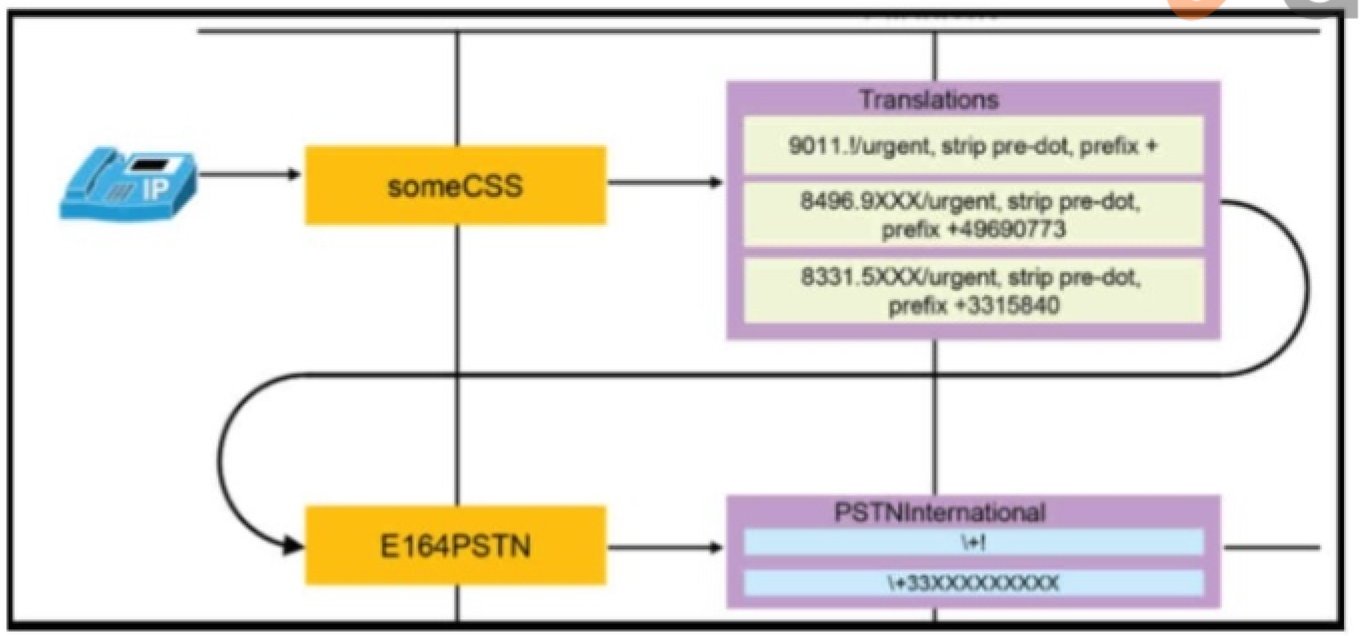
Correct Answer: D
Section:

QUESTION 11
An administrator discovers that employees are making unauthorized long-distance and international calls from logged-off Extension Mobility phones when the authorized users are away from their desks Which two configurations should the administrator configure in the Cisco UCM to avoid this issue? (Choose two.)

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

Correct Answer: A, E
Section:

QUESTION 12
Refer to the exhibit.



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

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

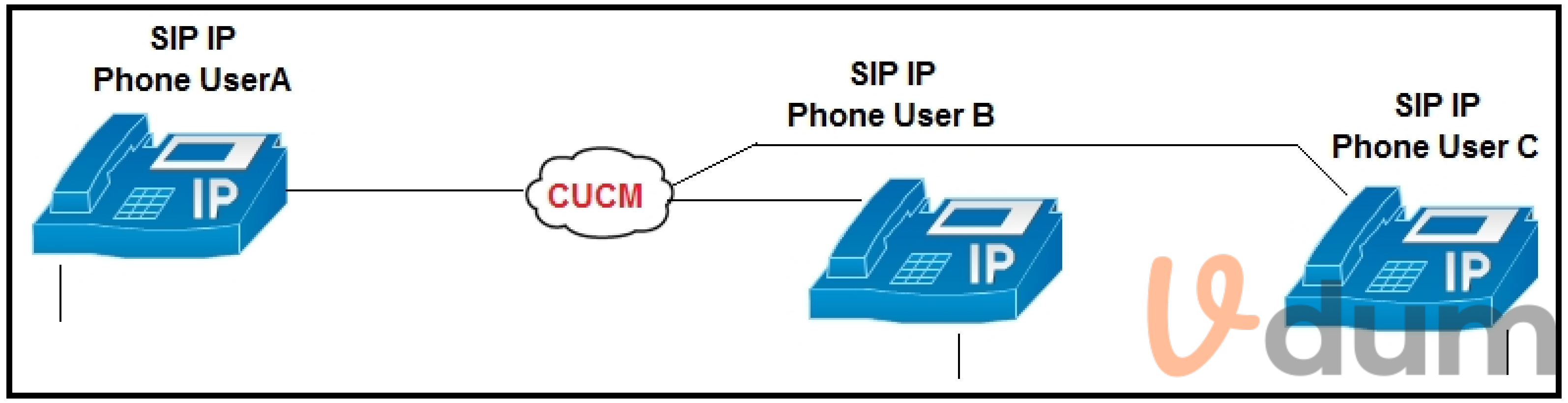
Correct Answer: C
Section:

QUESTION 13
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 the correct subnet under Device Mobility Info.
D. Set Device CSS under Current Device Mobility Settings.

Correct Answer: C
Section:

QUESTION 14
Refer to the exhibit.



In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. Which two scenarios are correct? (Choose two.)

A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
B. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.
D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.
E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Correct Answer: A, D
Section:

QUESTION 15
Refer to the exhibit.

```
SIP/2.0 200 OK
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=-IN IP4 10.10.10.11
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp: 110 0-16
a=ptime: 20

ACK sip:+123456789@10.10.20.20:5060 SIP/2.0
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=ptime:20
```

Users report that when they dial to Cisco Unity Connection from an external network, they cannot enter any digits. Assuming only in-band DTMF is supported, what is a reason for this malfunction?

A. The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.
B. There is SIP Delayed Offer. DTMF is supported only in Early Offer.
C. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.
D. No DTMF is negotiated.

Correct Answer: D
Section:

QUESTION 16
The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real- Time Transport Protocol traffic that had the one-way audio call.
You gather the H.225 and H.245 messages for one of the one-way audio calls. Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).

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

Correct Answer: B
Section:

QUESTION 17
Which two extended capabilities must be configured on dial peers for fast start-to-early media scenarios
(H.323 to SIP interworking)? (Choose two.)

A. DTMF
B. BFCP
C. VIDEO
D. FAX
E. AUDIO

Correct Answer: A, B
Section:

QUESTION 18
When you troubleshoot H.323 call setup, which message informs you that the called party is being notified about the call?

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

Correct Answer: A
Section:

QUESTION 19
End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

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

Correct Answer: D
Section:

QUESTION 20
Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?

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

Correct Answer: C
Section:

QUESTION 21
An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?

A. debug H.323 messages
B. debug H.225 asn1
C. debug H.246 asn 1
D. debug H.225 media
E. debug H.323 asn 1

Correct Answer: B
Section:

QUESTION 22
What is first preference condition matched in a SIP-enabled incoming dial peer?

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

Correct Answer: A
Section:

QUESTION 23
Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is one-way audio or no audio. You determine that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally. What are two possible solutions? (Choose two.)

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

Correct Answer: A, C
Section:

QUESTION 24
Which section under the Real-Time Monitoring Tool allows for reviewing the call flow and signaling for a SIP call in real time?

A. Analysis Manager > Inventory > Trace File Repositories
B. System > Tools > Trace and Log Central
C. Voice/Video > Session Trace Log View > Real Time Data
D. Voice/Video > Session Trace Log View > Open From Local Disk

Correct Answer: C
Section:

QUESTION 25
Which description of RTP timestamps or sequence numbers is true?

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

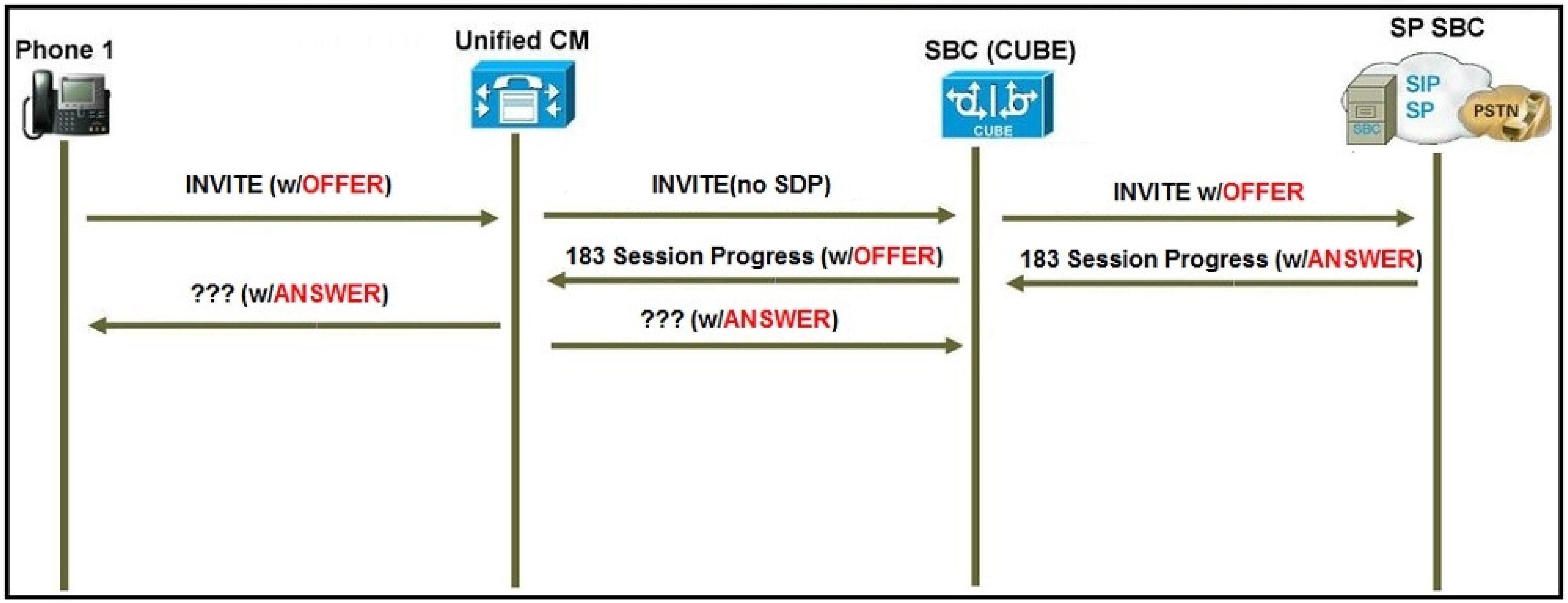
Correct Answer: D
Section:

QUESTION 26
A support engineer is troubleshooting a voice network. When conducting a search for call setup details related to calling search space issues, which trace files should be investigated?

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

Correct Answer: A
Section:

QUESTION 27
Refer to the exhibit.



A user reports that when they call a specific phone number, no one answers the call, but when they call from a mobile phone, the call is answered. The engineer troubleshooting the issue is expecting the far-end gateway to cut through audio on the 183 Session Progress SIP message. Which SIP Profile configuration element is necessary for the Cisco Unified Communications Manager to send acknowledgement of provisional responses?

A. Allow Passthrough of Configured Line Device Caller Information must be enabled.
B. Accept Audio Codec Preferences in Received Offer must be set to On.
C. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all 1xx Messages.
D. Early Offer for G Clear Calls must be enabled.

Correct Answer: C
Section:

QUESTION 28
A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

A. three-way conference
B. secure SIP lines
C. T.38 fax relay
D. transcoding
E. SIP trunk

Correct Answer: A, C
Section:

QUESTION 29
Which action is correct with respect to toll fraud prevention configuration in the Cisco Unified Communications Manager Express?

A. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.
B. Configure IP Address Trusted Authentication for Incoming VoIP Calls.
C. Configure the command no ip address trusted authenticate under ''voice service voip''.
D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Correct Answer: B
Section:

QUESTION 30
You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?

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

Correct Answer: A
Section:

QUESTION 31
Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

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

Correct Answer: B
Section:

QUESTION 32
For s SIP to SIP call flow, when does Cisco Unified Border Element require transcoding resources for DTMF?

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

Correct Answer: A
Section:

QUESTION 33
A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?

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

Correct Answer: A
Section:

QUESTION 34
Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)

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

Correct Answer: B, C
Section:

QUESTION 35
Refer to the exhibit.

```
!

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

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

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

Correct Answer: C
Section:

QUESTION 36
After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

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

Correct Answer: D
Section:

QUESTION 37
An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port
5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

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

Correct Answer: C, D
Section:

QUESTION 38

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

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

Correct Answer: C
Section:

QUESTION 39
If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern ''123''?

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

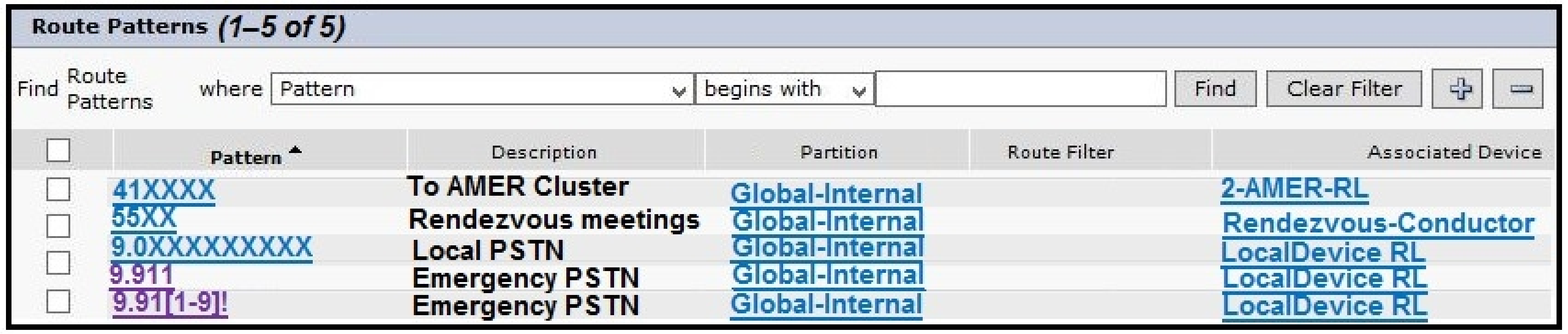
Correct Answer: D
Section:

QUESTION 40
Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

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

Correct Answer: D
Section:

QUESTION 41
Refer to the exhibit.


Route Patterns (1–5 of 5)

Find Route Patterns where Pattern begins with Find Clear Filter

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

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

A. Adjust the service parameter T302 timet to the desired value.
B. Adjust the service parameter T204 timer to the desired value.

C. Check the Urgent Priority check box under 9.911 pattern.
D. Point the emergency pattern directly to the PSTN gateway.

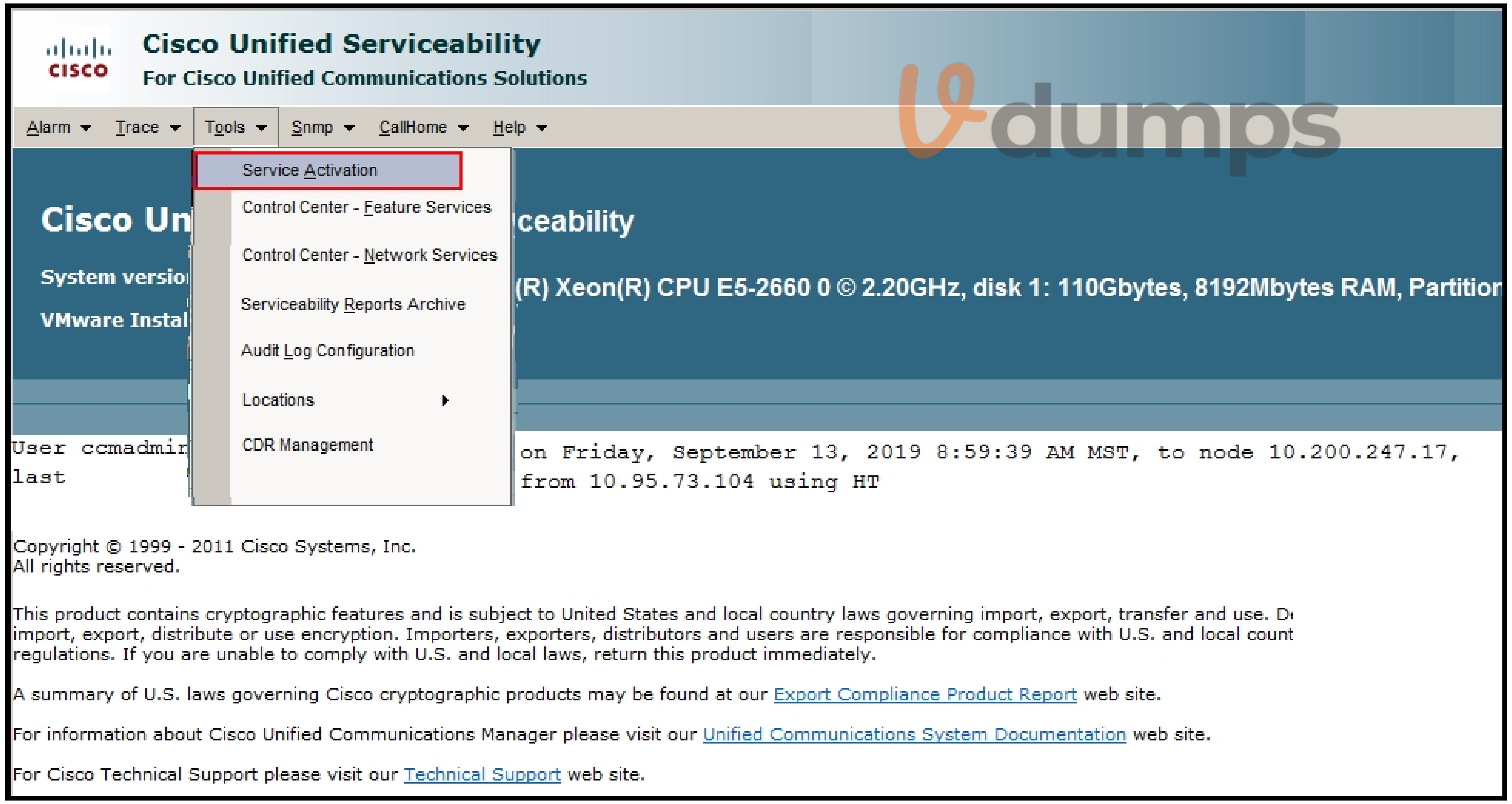
Correct Answer: C
Section:

QUESTION 42
The Cisco Unified Communications Manager Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

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

Correct Answer: C, E
Section:

QUESTION 43
Refer to the exhibit.



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

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

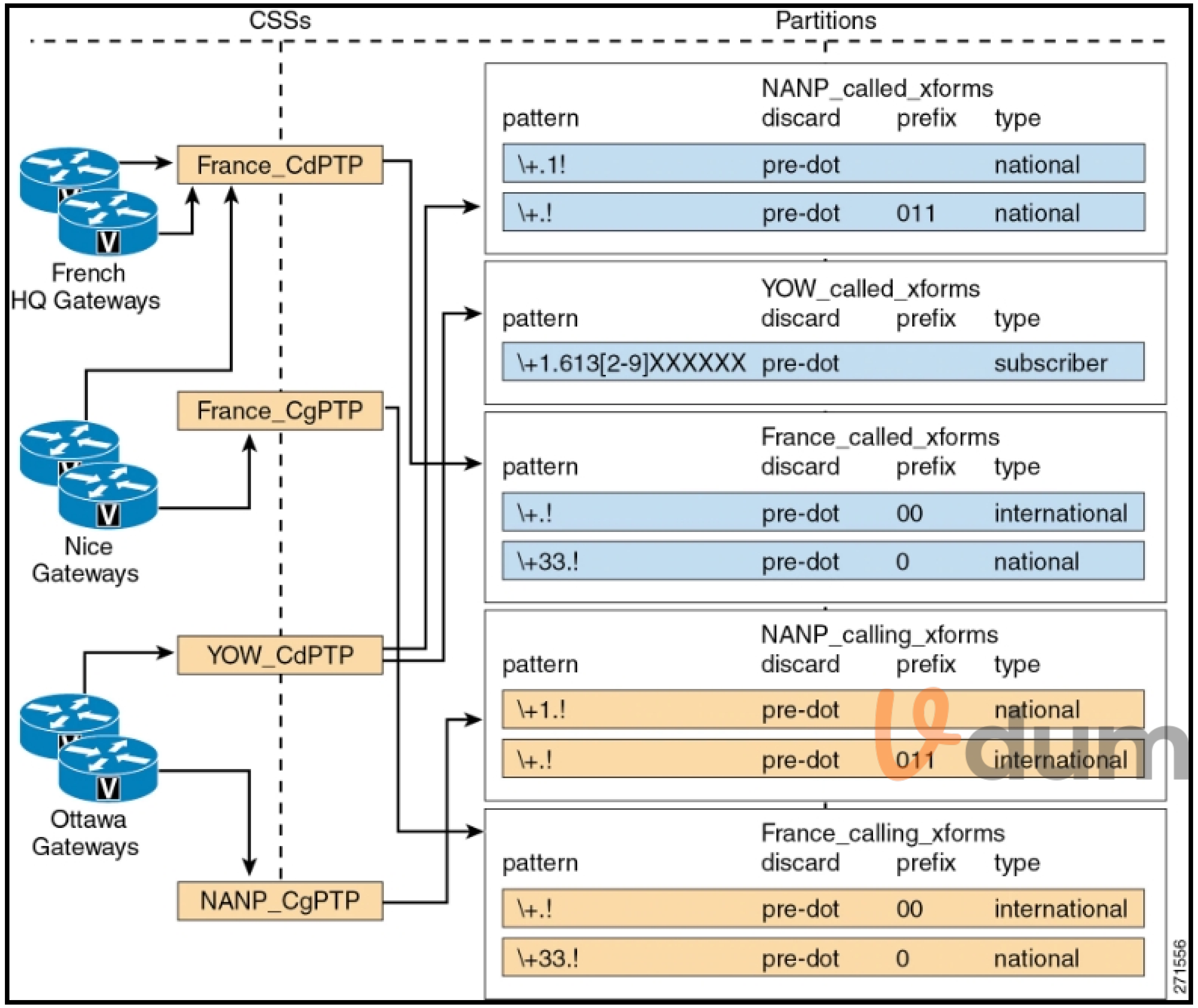
Correct Answer: D, E
Section:

QUESTION 44
In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

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

Correct Answer: A
Section:

QUESTION 45
Refer to the exhibit.



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

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

Correct Answer: D
Section:

QUESTION 46
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 volp Ip address trusted list
B. voice service volp enablo ip address trust authentication
C. voice service volp enable Ip address trust list
D. voice service volp ip address trusted authenticate

Correct Answer: D
Section:

QUESTION 47
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

Correct Answer: C
Section:

QUESTION 48
An engineer needs to deploy Cisco Extension Mobility. The engineer already activated the required services, configured the Extension Mobility Phone Service, and created an Extension Mobility device profile that is associated with end users. Which two additional steps are required to complete the configuration? (Choose two.)

A. Subscribe IP phones to the Extension Mobility Phone Service.
B. Consolidate Cisco UCM certificates by using Bulk Certificate Management.
C. Subscribe directory numbers to the Extension Mobility Phone Service.
D. Subscribe the device profile to the Extension Mobility Phone Service.
E. Define the default Device Template on IP phones when users log out.

Correct Answer: D, E
Section:

QUESTION 49
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-digit directory number. These 4-digit directory numbers range from 1000 to 9999. 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 add "<sip:(.....)@"
    "<sip:263411\1@"
```

B)

```
voice class sip-profiles 105
    request INVITE sip-header Diversion modify "<sip:1(...)@"
    "<sip:263411\1@"
```

```
voice class sip-profiles 105
    request INVITE sip-header Diversion modify "<sip:1(...)@"
    "<sip:263411\2@"

voice class sip-profiles 105
    request INVITE sip-header Diversion modify "<sip:(....)@"
    "<sip:263411\1@"
```

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

Correct Answer: D
Section:

QUESTION 50
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 srst under telephony-service configuration mode.
B. Configure mode esrst under telephony-service configuration mode.
C. Configure mode esrst under voice register global configuration mode.
D. Configure mode esrst under voice service voip configuration mode.

Correct Answer: C
Section:

QUESTION 51
Refer to the exhibit.

A user cannot call number 730. which Is an abbreviated number for 45730. The engineer recently configured a translation pattern In Cisco UCM. The newly configured translation pattern of 7XX prefixes the digits 45. The engineer uses DNA to troubleshoot the issue. What is the reason for this problem?

A. The abbreviated partition is missing from the calling search space.
B. The translation pattern is blocking the call.
C. The translation pattern Is misconfigured.
D. Directory number 45730 is unknown to Cisco UCM.

Correct Answer: A
Section:

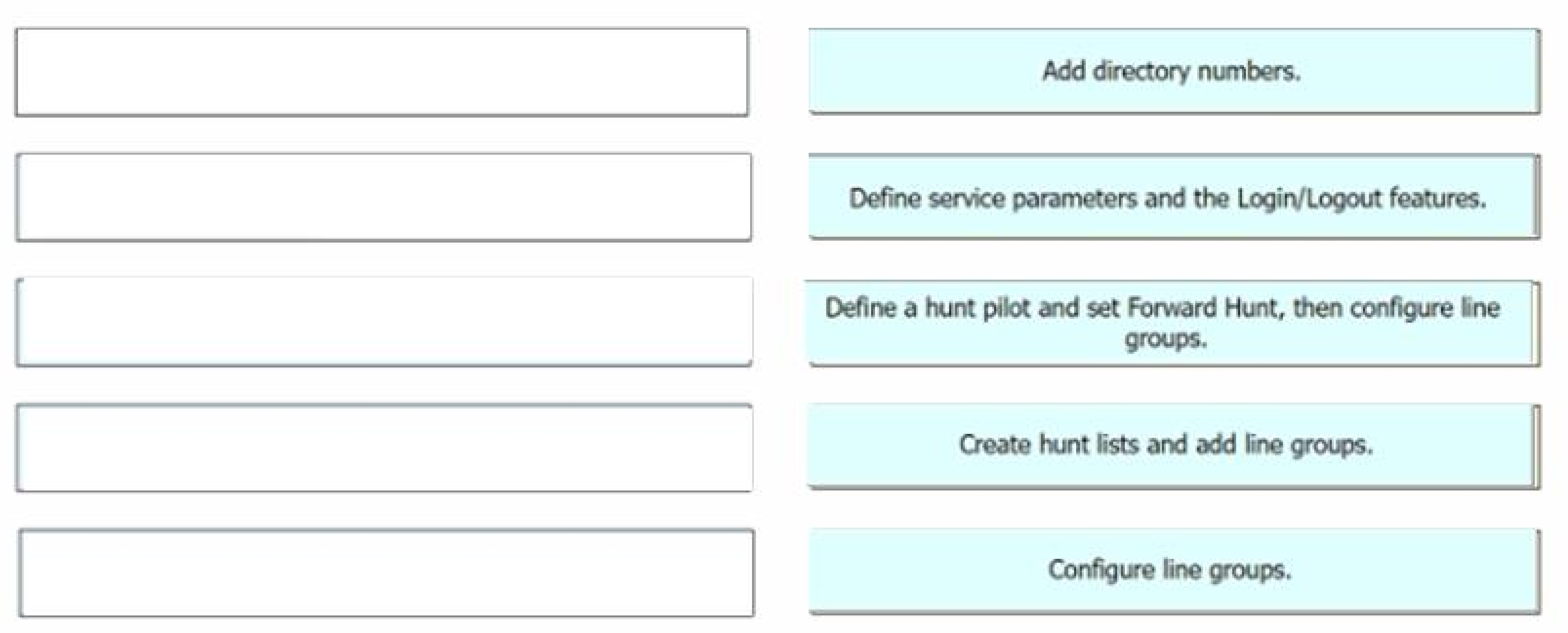
QUESTION 52
DRAG DROP
A customer wants to configure their team of 10 agents to answer incoming calls to a common central number. The phone for each team member must ring in sequence and ring four times before moving to the next member's phone. Drag and drop the steps from the left into the order on the right to make this happen.

Add directory numbers.	step 1
Define a hunt pilot and set Forward Hunt, then configure line groups.	step 2
Define service parameters and the Login/Logout features.	step 3
Create hunt lists and add line groups.	step 4
Configure line groups.	step 5

Select and Place:

Add directory numbers.	step 1
Define a hunt pilot and set Forward Hunt, then configure line groups.	step 2
Define service parameters and the Login/Logout features.	step 3
Create hunt lists and add line groups.	step 4
Configure line groups.	step 5

Correct Answer:



Section:
Explanation:

QUESTION 53
Refer to the exhibit.

A no-way audio situation was experienced after the calling party on an outbound call placed the call on hold and then resumed from hold. Which action will resolve the issue?

A. Assign an MTP to the SIP trunk.
B. Restart the SIP trunk on Cisco UCM.
C. Upgrade the Cisco Unified Border Element to 15.2.
D. Upgrade the Cisco Unified Border Element device.

Correct Answer: C
Section:

QUESTION 54
Refer to the exhibit.

Route Patterns *(1 - 2 of 2)*

Find Route Patterns where Pattern begins with \

	Pattern	Description	Partition
☐	\+!	Route pattern for international dialing	International
☐	\+1[2-9]XX[2-9]XXXXXX	Route pattern for US national dialing	National

Users report high delays when calling national numbers in the United States. An engineer analyzes the route patterns in Cisco UCM. What must be done to resolve this issue?

A. Change the Cisco UCM routing logic to the longest-match algorithm.
B. Select Allow Overlap Sending for the \+! route pattern.
C. Mark the \+ [2-9JXX[2-91XXXXXX route pattern with urgent priority.
D. Set the interdigit timeout service parameter value to 0.

Correct Answer: C
Section:

QUESTION 55
An organization needs to ensure that the Cisco UCM can provide a resume softkey to users on their desk phones when they disconnect an anchored call on their mobile phone. Which solution must be used to accomplish these goals?

A. Set the 'Retain Media on Disconnect with PI for Active Call' service parameter to False.
B. Set the 'Retain Media on Disconnect with PI for Active Cal' service parameter to True
C. Configure a handoff number.
D. Add Other Dual-Mode Device.

Correct Answer: A
Section:

QUESTION 56
Refer to the exhibit.

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP 10.5.5.5:5060;branch=z6a3c1K21d5aca213a1b
From: <sip:1000@10.5.5.5>;tag=11031~feddb418-c7b4-dce4-0b8b-901ba1db345c-13319871
To: <sip:2000@10.10.10.10>;tag=4C2F0432-8C1B
Date: Fri, 09 Apr 2021 01:41:33 GMT
Call-ID: c7011a00-d51ff310-31978-b42ab014@10.5.5.5
CSeq: 101 INVITE
RSeq: 1541
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Contact: <sip:2000@10.10.10.10:5060;transport=tcp>
Supported: sdp-anat
Supported: X-cisco-srtp-fallback
Server: Cisco-SIPGateway/IOS-15.7.3.M8
Content-Length: 330

v=0
o=CiscoSystemsSIP-GW-UserAgent 9841 5325 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
```

An administrator received reports that users do not hear a ringback when they dial specific external numbers. For example, the Cisco Unified Border Element receives a delayed offer invite from Cisco UCM and responds with the 183 session progress in the exhibit. Assume that the PSTN is responsible for generating ringback in these problematic scenarios. Which action resolves the issue?

A. Enable early media on the appropriate SIP trunk on Cisco UCM.
B. Edit the configuration for SIP ReMXX Options in the appropriate SIP profile on Cisco UCM to Send PRACK if 1xx Contains SDP.
C. On Cisco Unified Border Element: voice service voip sip Re1xx require 100rel
D. On Cisco Unified Border Element: voice service voip sip Re1xx disable 100rel

Correct Answer: B
Section:

QUESTION 57
An engineer needs to deploy the local route group feature to a Cisco UCM server by using the standard local route group. For each device pool, the engineer needs to specify a different PSTN access for emergency calls. What must the engineer do?

A. Add a new standard local route group. Select a route group as the new standard local route group in Device Pools. Create a route pattern of 911 that points to a route list that contains the new standard local route group.
B. Create a new local route group name. In Device Pools, select a route group for that new local route group. Create a route pattern of 911 that points to a route list that contains the new local route group.
C. Set a new local route group name. Select a route group as the new local route group in Device Pools. Create a route pattern of 911 that points to the new local route group.
D. Add a new standard local route group. Select a route group as the new standard local route group in Device Pools. Create a route pattern of 911 that points to the new standard local route group.

Correct Answer: A

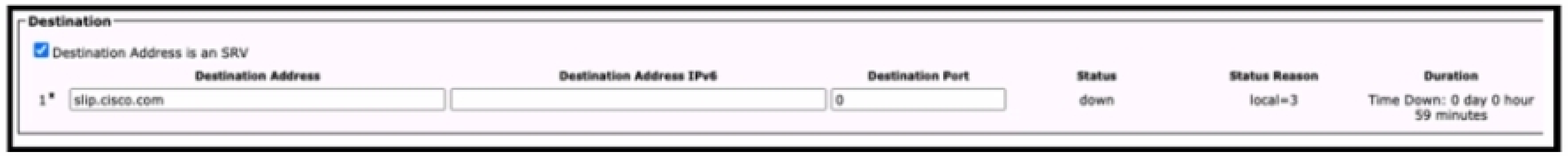
Section:

QUESTION 58
What is a capability of the gateway-preferred, network-based recording option in Cisco UCM?

A. It requires SIP trunk and CTI with recording application.
B. Cisco UCM integrates with recording destination via SIP trunk.
C. It uses BIB of IP phone to fork audio to the recording destination.
D. It initiates when the application (recorder) dictates that it must be initiated.

Correct Answer: B
Section:

QUESTION 59
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?

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

Correct Answer: A
Section:

QUESTION 60
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?
A)

```
telephony-service
max-conferences 8 gain -6
ip source-address 10.10.10.100 port 2000
max-ephones 100
max-dn 200
```

B)

```
call-manager-fallback
max-conferences 8 gain -6
ip source-address  10.10.10.100 port 2000
max-ephones 100
max-dn 200
```

C)

```
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
```

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

Correct Answer: B
Section:

QUESTION 61
The company Cisco UCM cluster has two different gateways for off-net calls. The current configuration uses 9 as a prefix to get to the main gateway. The secondary gateway is for any calls that start with 9713, but this is not yet configured. The admin does not want to add more route patterns other than the current 9 prefix for the gateway to the Cisco UCM. How must the Cisco UCM be configured to meet the requirements?

A. Configure two CSSs. then add a translation pattern with the secondary CSS to send calls with 9713 to the secondary gateway.
B. Configure two partitions and two CSSs, then add a translation pattern with the secondary CSS to send all calls with 9713 to the secondary gateway.
C. Create a Standard Route Group to dynamically route calls with prefix 9713 to the secondary gateway.
D. Configure a Route Group and a Route List to send calls with prefix 9713 to the secondary gateway.

Correct Answer: A
Section:

QUESTION 62
Exhibit.

```
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
```

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 the issue?

A. Disable G.722 on all devices at both sites.
B. Enable Early Offer on the SIP trunk.
C. Increase the bandwidth allowance between the RTP_Reg and SJ_Reg regions to 64 kbps.
D. 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.

Correct Answer: C
Section:

QUESTION 63
The company implemented Cisco Unified Mobility on the Cisco UCM. The users are satisfied and can transfer voice calls between devices. Mobile users want to extend the timer that controls the time given to pick up a voice call on an IP phone from 4 to 6 seconds. Which configuration change satisfies this requirement?

A. Change the 'Maximum Wait Time for Mobility Pickup' setting from 4 to 6.
B. Change the 'Maximum Wait Time for Desk Pickup' setting from 4 to 6.
C. Change the 'Maximum Wait Time for Mobility Pickup' setting from 4000 to 6000.
D. Change the 'Maximum Wait Time for Desk Pickup' setting from 4000 to 6000.

Correct Answer: D
Section:

QUESTION 64
Refer to the exhibit.

```
14280314.001 |17:18:26.418 |AppInfo  |SIPTcp - wait_SdlSPISignal:
Outgoing SIP TCP message to 10.122.44.183 on port 49458 index 217
[445204,NET]
SIP/2.0 404 Not Found
Via: SIP/2.0/TCP 10.122.44.183:49458;branch=z9hG4bK00001847
From: "4049" <sip:4049@10.122.44.125>;tag=005056b0186f00e100000603-00006aac
To: <sip:4009@10.122.44.125>;tag=148579~8b971ab6-3fcf-4fb3-b506-529257fa1198-40264034
Date: Wed, 08 Feb 2023 23:18:26 GMT
Call-ID: 005056b0-186f000b-00000400-00006c9a@10.122.44.183
CSeq: 101 INVITE
Allow-Events: presence
Reason: Q.850;cause=1
Server: Cisco-CUCM12.5
Session-ID: 00000000000000000000000000000000;remote=00005dec00105000a000005056b0186f
Content-Length: 0
```

An administrator just upgraded to Cisco UCM to version 14 and started SIP implementation with some new SIP trunks. During the testing, an error was reported when making a call. Which action resolves the issue?

A. Change to a valid range.
B. Change DTMF Signaling Method to 'RFC 4833*.
C. Make sure the destination number is configured correctly and the device is registered.
D. Disable SIP Rel1XX Options.

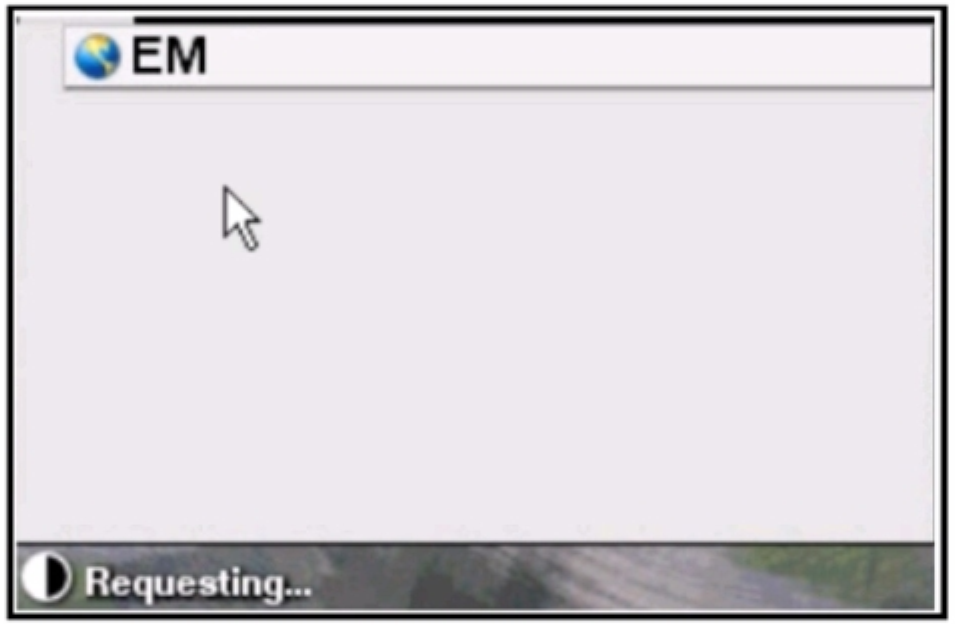
Correct Answer: C
Section:

QUESTION 65
An administrator is configuring an Intercluster Lookup Service between 10 Cisco UCM clusters. Due to security requirements, certificate-based authentication must be used. Due to the deployment size, the administrator wants to avoid manually exchanging certificates between the clusters. Which two steps must be followed to meet these requirements? (Choose two)

A. Set all dusters to standalone on the ILS configuration page.
B. Install multiserver CA-signed Tomcat certificates on all cluster publishers.
C. Enable password-based authentication in conjunction with certificate-based authentication.
D. Ensure that the cluster ID for all clusters is the same.
E. Install multiserver CA-signed CallManager certificates on all cluster publishers.

Correct Answer: B, C
Section:

QUESTION 66
Refer to me exhibit.



A user cannot access Extension Mobility even after clicking the services button and selecting EM service. There is nowhere to place the username and password to sign In for this service. What Is the cause of the issue?

A. IP phone services Extension Mobility does not have the correct service URL configured.
B. IP phone services Extension Mobility has XML Service Category configured.
C. The Cisco IP phone has not checked the Extension Mobility checkbox.
D. The Cisco IP phone is not subscribed to an Extension Mobility service.

Correct Answer: A
Section: