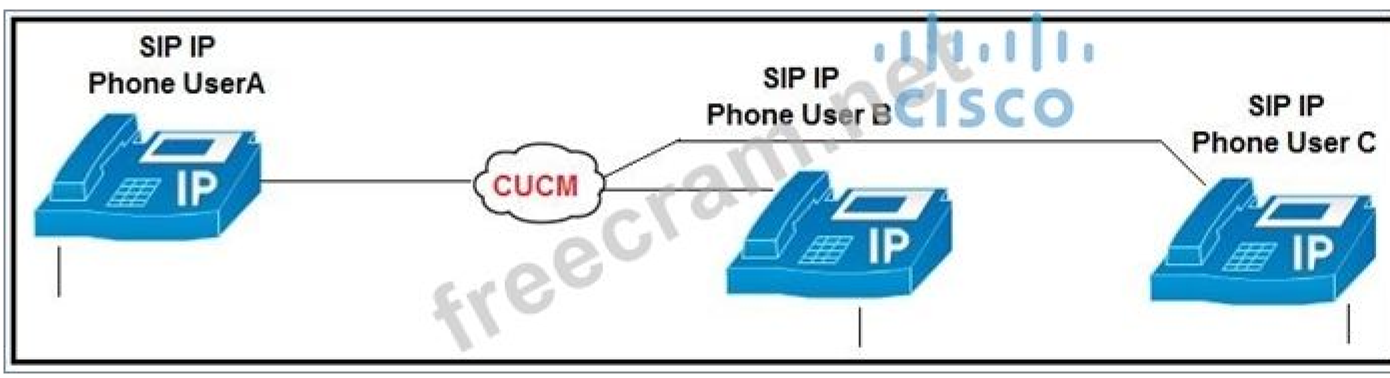
Cisco.300-815.v2023-06-28.q89

Exam Code:	300-815
Exam Name:	Implementing Cisco Advanced Call Control and Mobility Services
Certification Provider:	Cisco
Free Question Number:	89
Version:	v2023-06-28
# of views:	347
# of Questions views:	5210
https://www.freecram.net/torrent/Cisco.300-815.v2023-06-28.q89.html	

NEW QUESTION: 1

Refer to the exhibit.



In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C.

Which two scenarios are correct? (Choose two.)

A. 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.

B. 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.

C. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.

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. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.

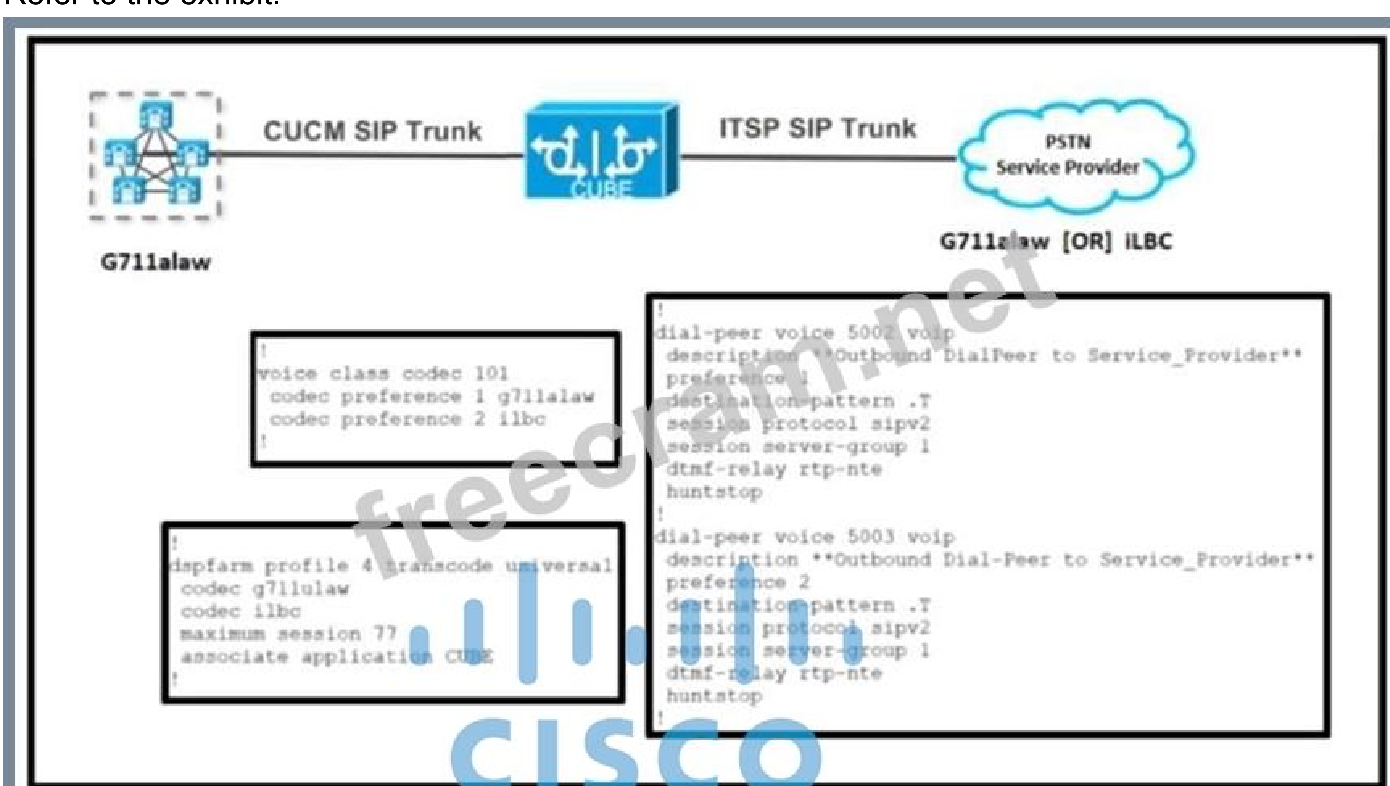
Answer: C,D (LEAVE A REPLY)

NEW QUESTION: 2

Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?
A. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page.
B. Enable the Detailed Call Analysis option under Enterprise Parameters for Unified CM.
C. Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis.
D. By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.
Answer: (SHOW ANSWER)

NEW QUESTION: 3
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. Cisco IP Manager Assistant
C. Call logs
D. CTI Manager traces
Answer: (SHOW ANSWER)

NEW QUESTION: 4
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 B
B. Option A
C. Option D
D. Option C
Answer: (SHOW ANSWER)

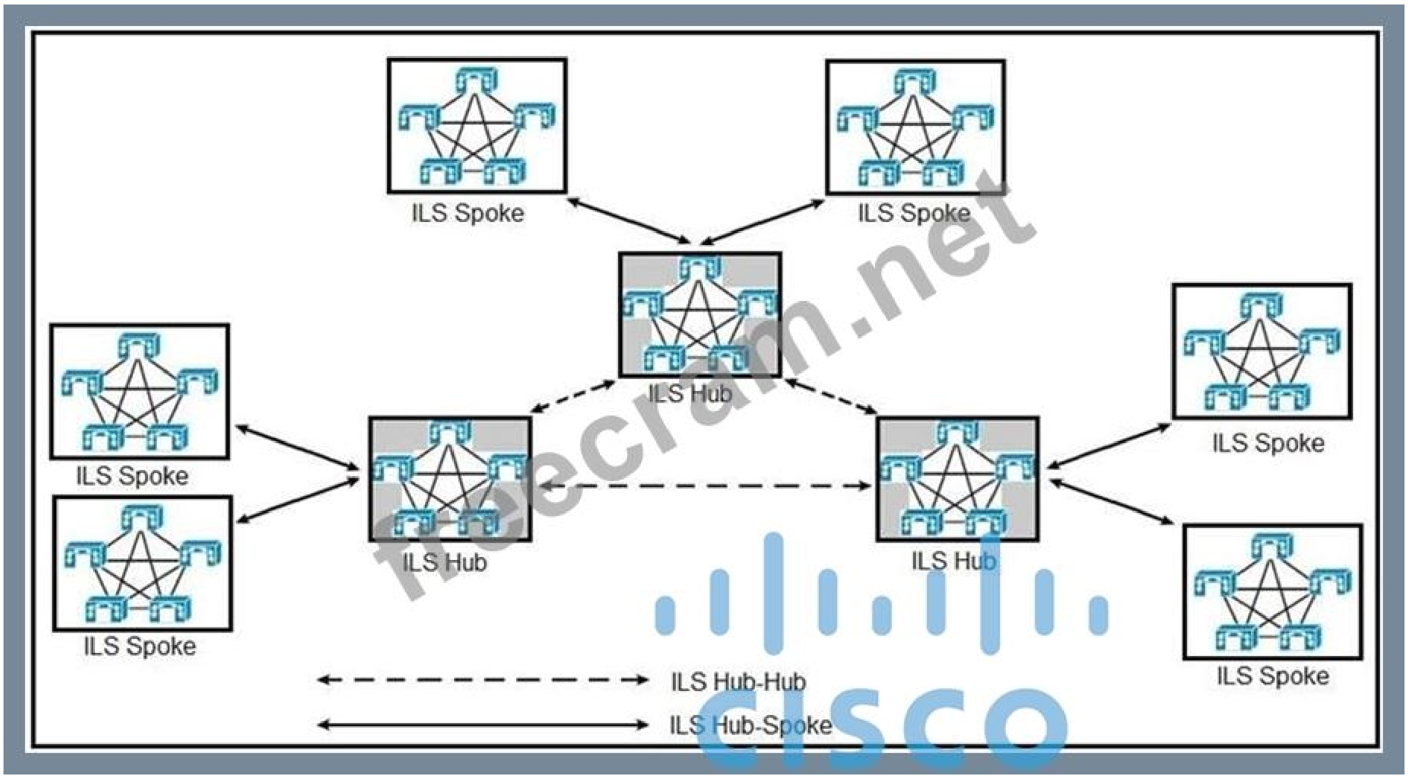
NEW QUESTION: 5
An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?
A. ILS Max Number of Learned Objects in Database
B. ILS Active Learned Object Upper Limit
C. Global Data Service Parameter Limit
D. Imported Dial Plan Replication Database Object Lower Limit
Answer: (SHOW ANSWER)
Section: Cisco Unified CM Call Control Features

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

NEW QUESTION: 6
Which configuration element of a hunt group allows for changing Calling Party Transformations settings?
A. line group
B. hunt pilot
C. route group
D. hunt list
Answer: (SHOW ANSWER)
Reference:
https://community.cisco.com/t5/ip-telephony-and-phones/call-alerting-on-hunt-group-as-shared-line/td-p/2658015

NEW QUESTION: 7
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. FAX
B. VIDEO
C. BFCP
D. AUDIO
E. DTMF
Answer: (SHOW ANSWER)

NEW QUESTION: 8
Refer to the exhibit.



How many maximum hops can an ILS updarte traverse?

A. 12

B. 3

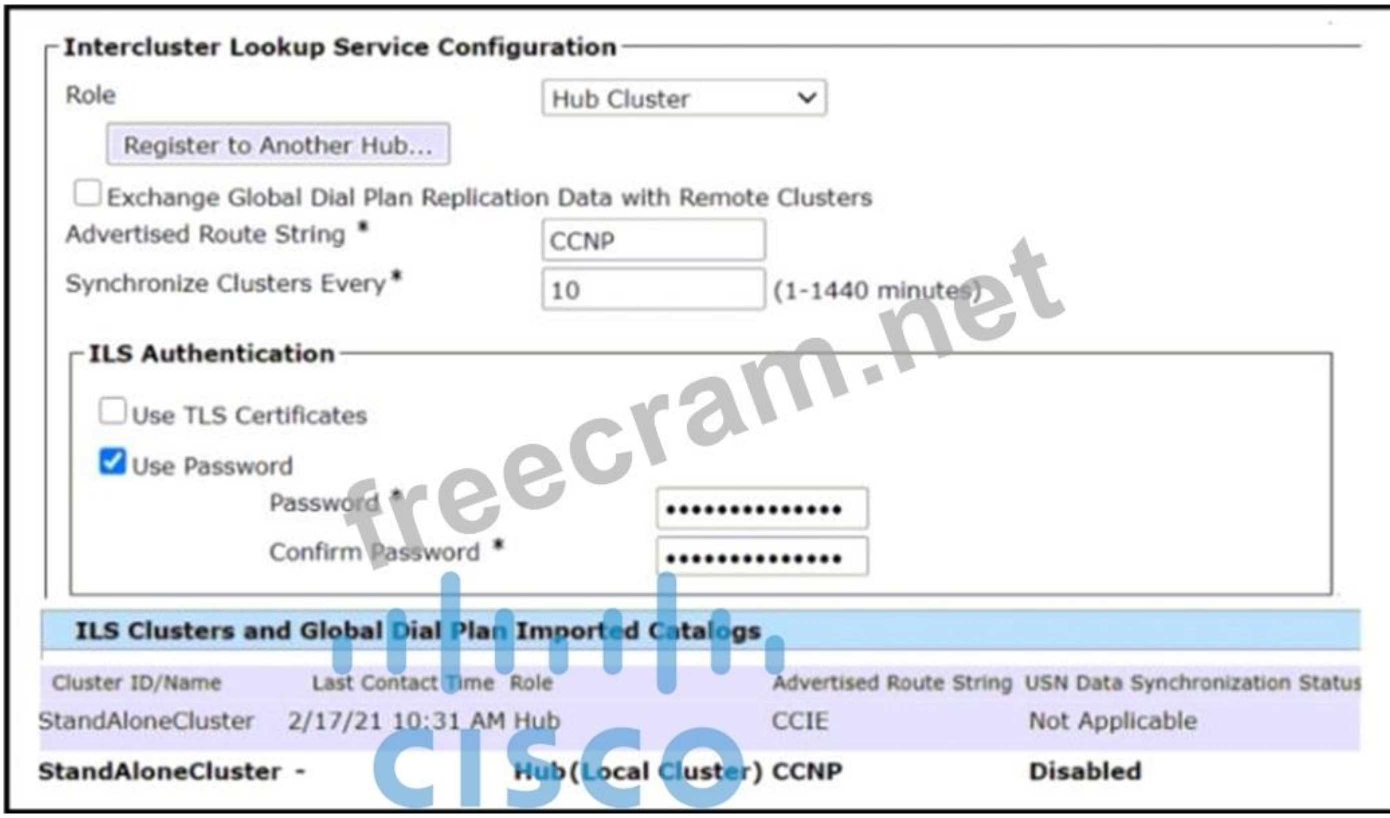
C. 6

D. 9

Answer: (SHOW ANSWER)

NEW QUESTION: 9

Refer to the exhibit.



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

A. Trust certificates for ILS have not been installed on the clusters

B. A password has never been set for ILS.

C. The Cluster IDs have not been set to unique values

D. Use TLS Certificates must be selected.

Answer: (SHOW ANSWER)

NEW QUESTION: 10

Refer to the exhibit.

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.1.60.105:5060;branch=z9hG4bK721ed5d4
From: "1001" <sip:1001@10.88.247.229>;tag=6cfa89726ac700b569ec133a-7e6cd8aa
To: <sip:2005@10.88.247.229>;tag=47B5F70-43B
Date: Fri, 19 Apr 2019 12:13:40 GMT
Call-ID: 6cfa8972-6ac7002b-5af19a5c-0de23108@10.1.60.105
CSeq: 101 INVITE
Require: 100rel
RSeq: 3344
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Remote-Party-ID: <sip:2005@10.88.247.229>;party=called;screen=yes;privacy=off
Contact: <sip:2005@10.88.247.229:5060>
Server: Cisco-SIPGateway/IOS-16.6.2
Content-Length: 0
```

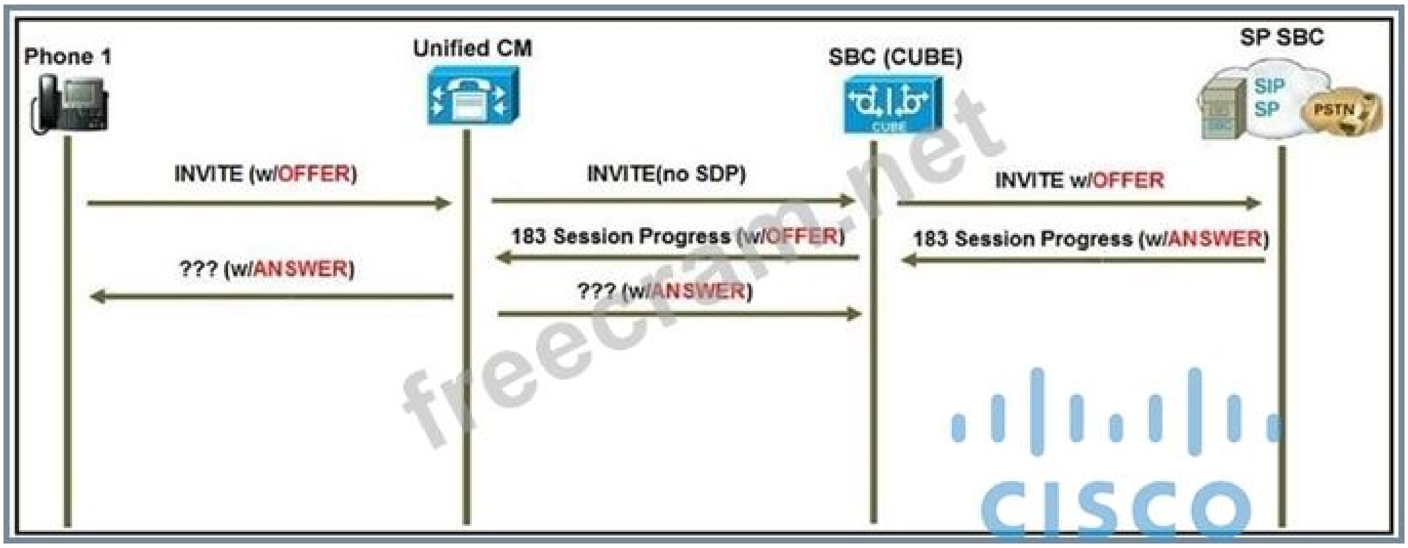
An engineer is troubleshooting an issue with the caller not hearing a PSTN announcement before the SIP call has completed setup. How must the engineer resolve this issue using the reliable provisional response of the SIP?

A. voice service voip sip send 180 sdp
B. voice service voip sip rehxx require 100rel
C. voice service voip sip no reMxx
D. sip-ua
disable-early-media 180
Answer: (SHOW ANSWER)

NEW QUESTION: 11
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.
Answer: (SHOW ANSWER)
Section: CME/SRST Gateway Technologies
Explanation/Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

NEW QUESTION: 12
Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?
A. The RTP traffic is arriving beyond the jitter buffer on the receiving end.
B. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path.
C. A firewall in the media path is blocking TCP ports 16384-32768.
D. Cisco Unified Communications Manager invoked media termination point resources.
Answer: (SHOW ANSWER)

NEW QUESTION: 13
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. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all
1xx Messages.

B. Accept Audio Codec Preferences in Received Offer must be set to On.

C. Early Offer for G Clear Calls must be enabled.

D. Allow Passthrough of Configured Line Device Caller Information must be enabled.

Answer: (SHOW ANSWER)

NEW QUESTION: 14

Refer to the exhibit.

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

m=audio 30404 RTP/AVP 0 9 124 116 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:124 iSAC/16000
a=rtpmap:116 iLBC/8000
a=maxptime:20
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 15

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

Answer: (SHOW ANSWER)

Section: Call Control and Dial Planning

NEW QUESTION: 16

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

A. hunt list

B. line group

C. route group

D. hunt pilot

Answer: (SHOW ANSWER)

NEW QUESTION: 17

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. Create specific matching for "222333444" on the incoming dial peer.

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 18

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

A. R2(config-ephone-dn)#park reservation-group 60

B. R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002

C. R2(config-ephone-dn)#park reservation-group 1

D. R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Answer: (SHOW ANSWER)

Section: Cisco Unified CM Call Control Features

NEW QUESTION: 19

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

A. line group

B. hunt pilot

C. route group

D. hunt list

Answer: (SHOW ANSWER)

Section: Cisco Unified CM Call Control Features

Explanation/Reference:

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

NEW QUESTION: 20

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.

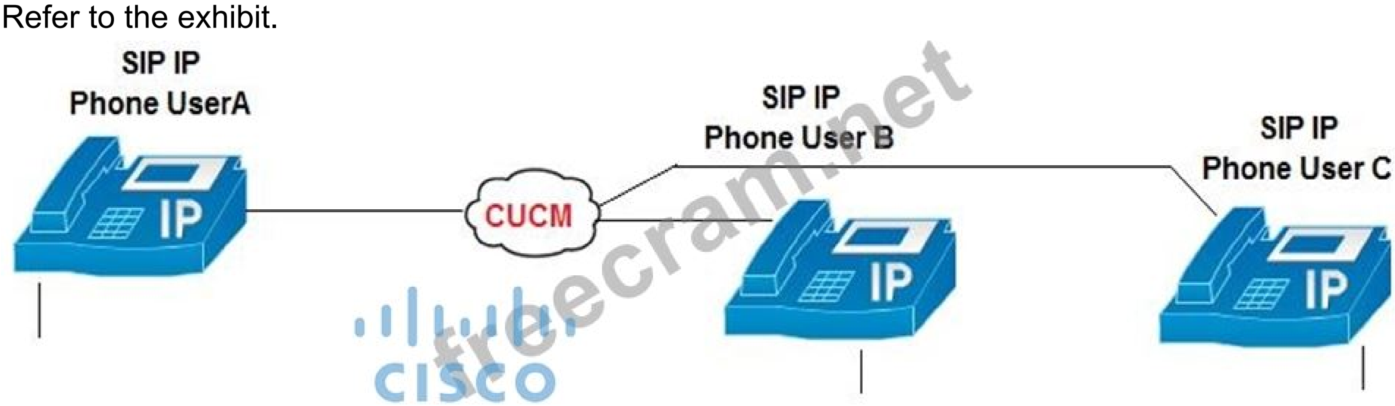
E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

NEW QUESTION: 21

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. 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.

B. 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.

C. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
D. 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.
E. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
Answer: (SHOW ANSWER)

NEW QUESTION: 22

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

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

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

A. The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.

B. There is SIP Delayed Offer. DTMF is supported only in Early Offer.

C. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.

D. No DTMF is negotiated.

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

NEW QUESTION: 23

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[^0-5]

C. 813510[12345]

D. 81XXXXX

Answer: (SHOW ANSWER)

NEW QUESTION: 24

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

A. 12X (urgent priority set)

B. 12!

C. 1XX (urgent Priority Set)

D. 12[2-5]

Answer: (SHOW ANSWER)

NEW QUESTION: 25

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. Activate the Cisco CallManager service.

B. Activate the Cisco Dialed Number Analyzer service.

C. Activate the Cisco Extended Functions service.

D. Activate the Cisco Dialed Number Analyzer Server service.

E. Restart the subscriber

Answer: (SHOW ANSWER)

NEW QUESTION: 26

Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?

A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service

B. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility

C. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service

D. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility

Answer: (SHOW ANSWER)

NEW QUESTION: 27

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

A. directory URI partition
B. intercluster trunk
C. calling search space and partition
D. SIP trunk
E. SIP route pattern
Answer: (SHOW ANSWER)

NEW QUESTION: 28
A customer has multisite deployments with a globalized dial plan. The customer wants to route PSTN calls via the gateway assigned to each site. Which two actions will fulfill the requirement? (Choose two.)
A. Create one global route list for PSTN calls that points to one global PSTN route group.
B. Create one route group for each site and one global route list for PSTN calls that point to the local route group.
C. Assign one route group as a local route group in the device pool of the corresponding site.
D. Create a route group which has all the gateways and associate it to the device pool of every site.
E. Create a hunt group and assign it to each side route pattern
Answer: (SHOW ANSWER)

NEW QUESTION: 29
Due to a shortage of physical interfaces on a device the administrator requires that a loopback for RTP is used.
Which command is required when using a loopback interface for RTP?
A. voice-class sip bind media source-interface Loopback0
B. voice-class sip early-offer forced.
C. voice-class sip resources priority mode passthrough
D. voice-class sip bind control source-interface Loopback0
Answer: (SHOW ANSWER)

NEW QUESTION: 30
Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?

A. Other Group Pickup
B. BLF Call Pickup
C. Group Call Pickup
D. Directed Call Pickup
Answer: A (LEAVE A REPLY)
Reference:
https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmsys/a07cpick.html#wp1022865

NEW QUESTION: 31

A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?

A. Assign the extension as a mobile extension.

B. Subscribe the phone to the Cisco Extension Mobility service.

C. Create the default device profiles for all phone models that are used.

D. Create the end user and associate it to the device profile.

Answer: (SHOW ANSWER)

NEW QUESTION: 32

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. session-protocol sipv2

B. codec g729

C. answer-address 555 ....

D. incoming called number 555.......

Answer: (SHOW ANSWER)

NEW QUESTION: 33
Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?
A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service
B. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service
C. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility
D. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility
Answer: (SHOW ANSWER)
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature- configuration-guide_chapter_011101.html#CUCM_TK_A337E035_00

NEW QUESTION: 34
Refer to the exhibit.


DN 1003 was the last to ring during the most recent call. Which hunting method ensures that DN 1005 is presented with the next call when the hunt pilot is dialed?
A. peer
B. sequential
C. parallel
D. call-blast
Answer: (SHOW ANSWER)

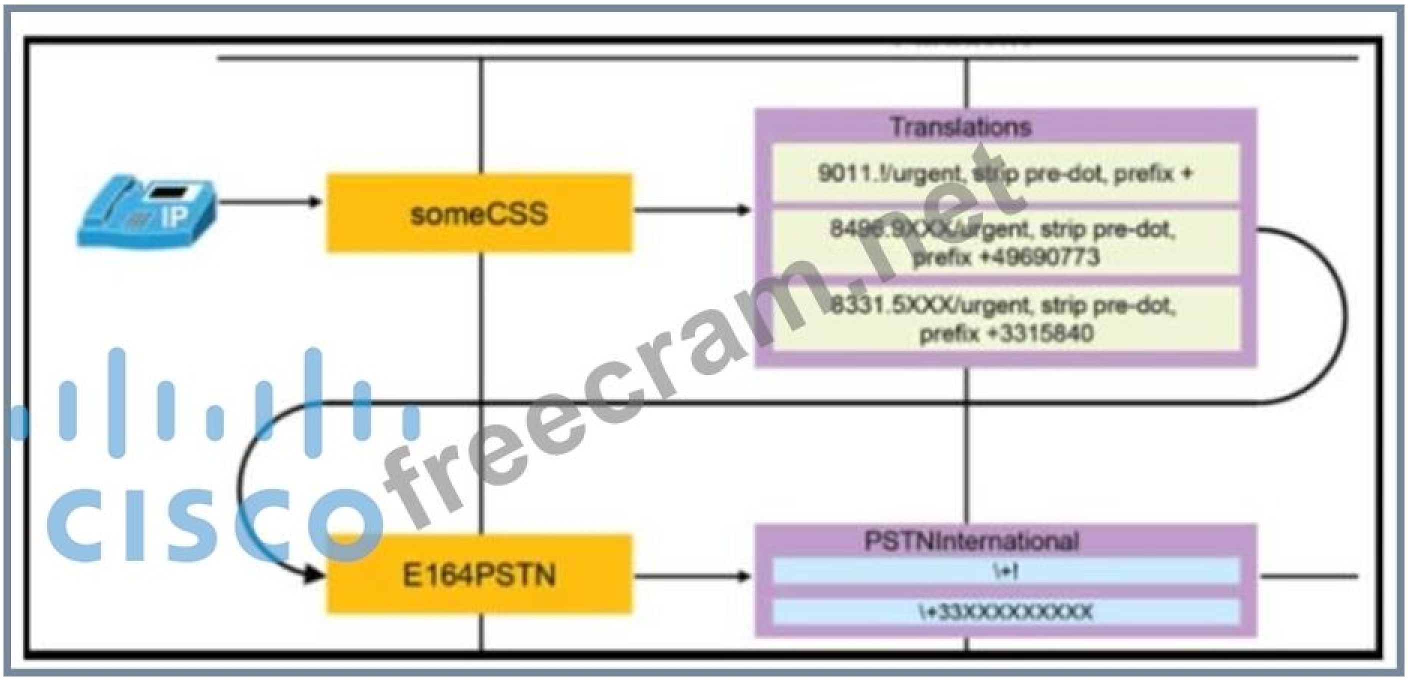
NEW QUESTION: 35
An administrator configured Cisco Unified Mobility to block access to remote destinations for certain caller IDs. A user reports that a blocked caller was able to reach a remote destination. Which action resolves the issue?

A. Configure a mobility identity.
B. Configure Mobile Voice Access.
C. Configure Single Number Reach.
D. Configure an access list.
Answer: (SHOW ANSWER)

NEW QUESTION: 36
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. c= line of SDP content
B. o= line of SDP content
C. Contact: header of the 200 OK response
D. Allow: header if the 200 OK response
Answer: (SHOW ANSWER)

NEW QUESTION: 37
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. codec mismatch
B. missing Call Admission Control
C. ptime mismatch
D. phone class of service issue
Answer: (SHOW ANSWER)

NEW QUESTION: 38
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. Do Not Wait For Interdigit Timeout On Subsequent Hops on the translation pattern

C. Do Not Wait For Interdigit Timeout On Subsequent Hops on the route pattern

D. Route Next Hop By Calling Party Number on the translation pattern

Answer: (SHOW ANSWER)

NEW QUESTION: 39

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

A. Verify that the route pattern is not blocking calls to the destination number.

B. Verify that IP routing is correct between the gateway and the IP phone.

C. Verify that the dial peer of the gateway has the correct destination pattern configured.

D. Verify that the route pattern has the correct calling-party transformation mask

Answer: (SHOW ANSWER)

NEW QUESTION: 40

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

A. Voice/Video > Session Trace Log View > Open From Local Disk

B. Voice/Video > Session Trace Log View > Real Time Data

C. Analysis Manager > Inventory > Trace File Repositories

D. System > Tools > Trace and Log Central
Answer: (SHOW ANSWER)

NEW QUESTION: 41
Which IOS command creates a SIP-enabled dial peer?
A. voice dial-peer 20 sip
B. dial-peer voice 20 voip
C. dial-peer voice 20 pots
D. dial peer voice 20 sip
Answer: B (LEAVE A REPLY)
Section: Cisco Unified Border Element
Explanation/Reference: https://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

NEW QUESTION: 42
An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?
A. Global Data Service Parameter Limit
B. Imported Dial Plan Replication Database Object Lower Limit
C. ILS Max Number of Learned Objects in Database
D. ILS Active Learned Object Upper Limit
Answer: (SHOW ANSWER)

NEW QUESTION: 43
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
Answer: (SHOW ANSWER)
Reference:
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35

NEW QUESTION: 44
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
Answer: (SHOW ANSWER)
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_sip_isr4000.html

NEW QUESTION: 45
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.246 asn 1
B. debug H.225 media
C. debug H.323 asn 1
D. debug H.224 asn1
E. debug H.323 messages
Answer: (SHOW ANSWER)

NEW QUESTION: 46
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
Answer: (SHOW ANSWER)
Section: Signaling and Media Protocols
Explanation/Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

NEW QUESTION: 47
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
Answer: (SHOW ANSWER)
Section: Cisco Unified Border Element

NEW QUESTION: 48
How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?
A. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.
B. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.
C. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.
D. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.
Answer: (SHOW ANSWER)

NEW QUESTION: 49
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
Answer: (SHOW ANSWER)
Reference:
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/99946-cme-sip-guide.html

NEW QUESTION: 50

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

A. broadcast

B. circular

C. longest idle time

D. top down

Answer: (SHOW ANSWER)

NEW QUESTION: 51

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

A. The tomcat certificate from Cluster B must be the publisher.

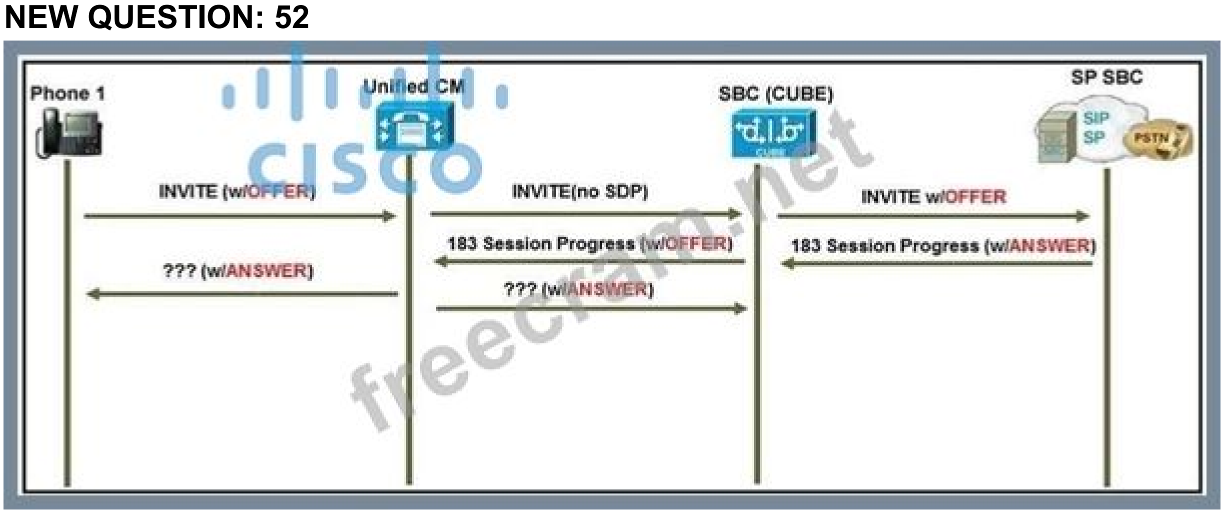
B. The password is incorrect.

C. Cluster IDs are not unique.

D. The engineer needs to exchange the CallManager certificate.

Answer: (SHOW ANSWER)

NEW QUESTION: 52



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?

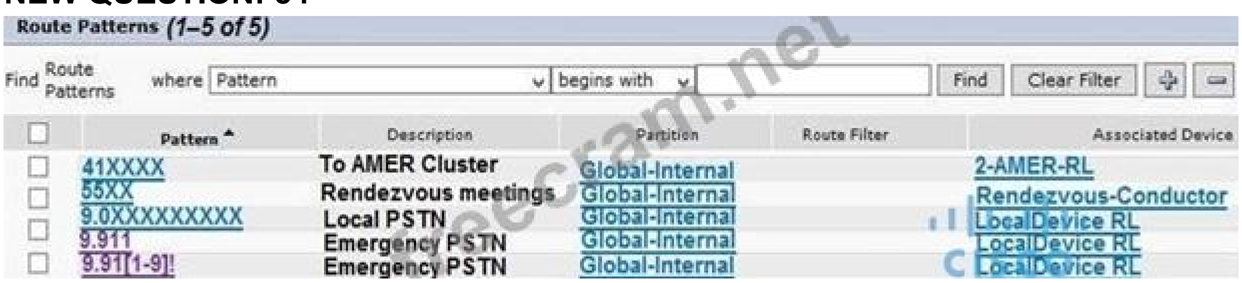
A. Allow Passthrough of Configured Line Device Caller Information must be enabled.

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.
Answer: (SHOW ANSWER)
Section: Signaling and Media Protocols

NEW QUESTION: 53
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. transcoding
C. SIP trunk
D. secure SIP lines
E. T.38 fax relay
Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 54

Route Patterns (1–5 of 5)

Find Route Patterns where Pattern begins with [Find] [Clear Filter]

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

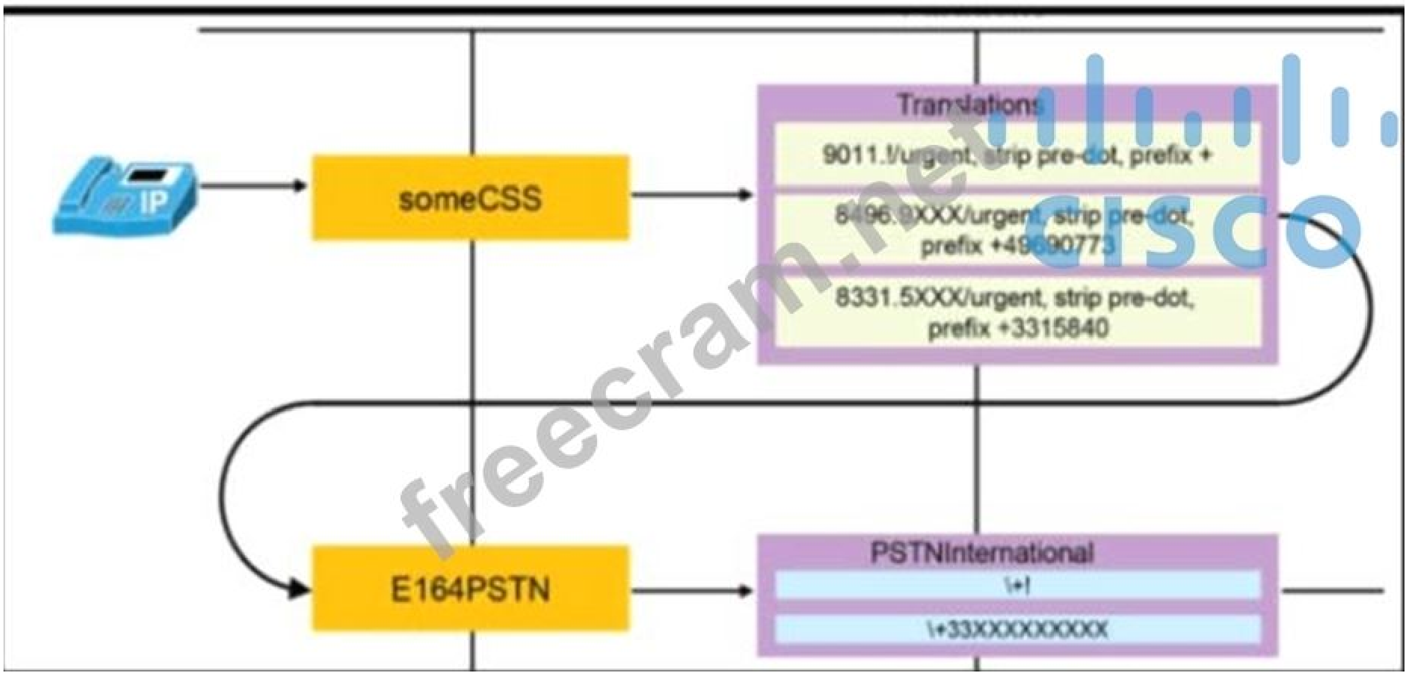
Refer to the exhibit. Users report that when they dial the emergency number 9911 from any internal phone, it takes a long time to connect with the emergency operator. Which action resolves this issue?
A. Adjust the service parameter T302 timet to the desired value.
B. Adjust the service parameter T204 timer to the desired value.
C. Check the Urgent Priority check box under 9.911 pattern.
D. Point the emergency pattern directly to the PSTN gateway.
Answer: (SHOW ANSWER)
Section: Call Control and Dial Planning

NEW QUESTION: 55
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
Answer: (SHOW ANSWER)

delay compensation).

NEW QUESTION: 56

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. Do Not Wait For Interdigit Timeout On Subsequent Hops on the translation pattern

B. Do Not Wait For Interdigit Timeout On Subsequent Hops on the route pattern

C. Allow Device Override on the route pattern

D. Route Next Hop By Calling Party Number on the translation pattern

Answer: (SHOW ANSWER)

NEW QUESTION: 57

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

A. incoming uri

B. target carrier-id

C. answer-address

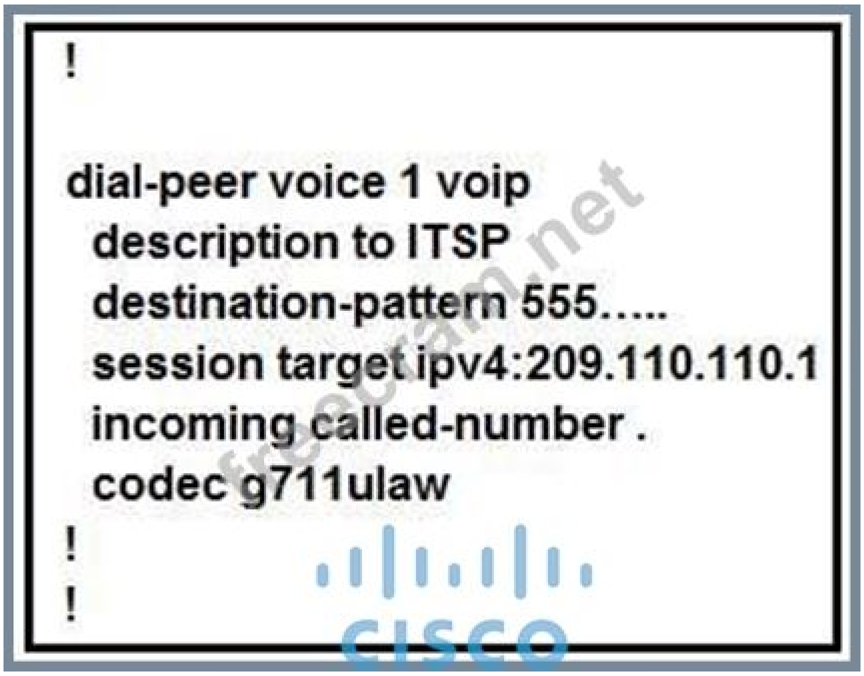
D. incoming called-number

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

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

NEW QUESTION: 58



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

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

A. answer-address 555 ....
B. codec g729
C. session-protocol sipv2
D. incoming called number 555.......
Answer: (SHOW ANSWER)
Section: Call Control and Dial Planning

NEW QUESTION: 59
What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?
A. A partition can have one time schedule assigned. A time schedule contains one or more time periods.
B. A partition can have one time schedule assigned. A time schedule contains only one time period.
C. A partition can have multiple time schedules assigned. A time schedule contains one or more time periods.
D. A partition can have multiple time schedules assigned. A time schedule contains only one time period.
Answer: (SHOW ANSWER)

NEW QUESTION: 60
Refer to the exhibits.

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name* Region A

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Region B	Use System Default (Factory Default low loss)	8 kbps (G.729)	None	None
Region C	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
Region D	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Media Resource Group Information

Name* MRG2

Description Media Resource Group 2

Devices for this Group

Available Media Resources**

MOH_5
MTP_3
MTP_4
MTP_5
XCODER

Selected Media Resources*

ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP_2 (MTP)

```
#CallManager SDL Logs
|AppInfo  |DET-MediaManager-(22401)::preCheckCapabilities, caps mismatch! Xcoder Reqd. kbps(8),
    filtered A[capCount=0 (Cap,ptime)=], B[capCount=4 (Cap,ptime)= (11,220) (12,220) (15,220) (9,270)]
    allowMTP=0 numXcoderRequired=1 xcodingSide=1
|SdlSig   |MxmAllocateMtpResourceErr              |waitResourcesAllocated
    |MediaManager(6,100,144,22401)     |MediaResourceManager(6,100,142,1)
```

Regions have been configured for all major branches based on the available circuit bandwidth. Some calls from Region A endpoints to Region B endpoints are failing to connect. How is this issue resolved?

A. Add a media resource to transcode between available capabilities.

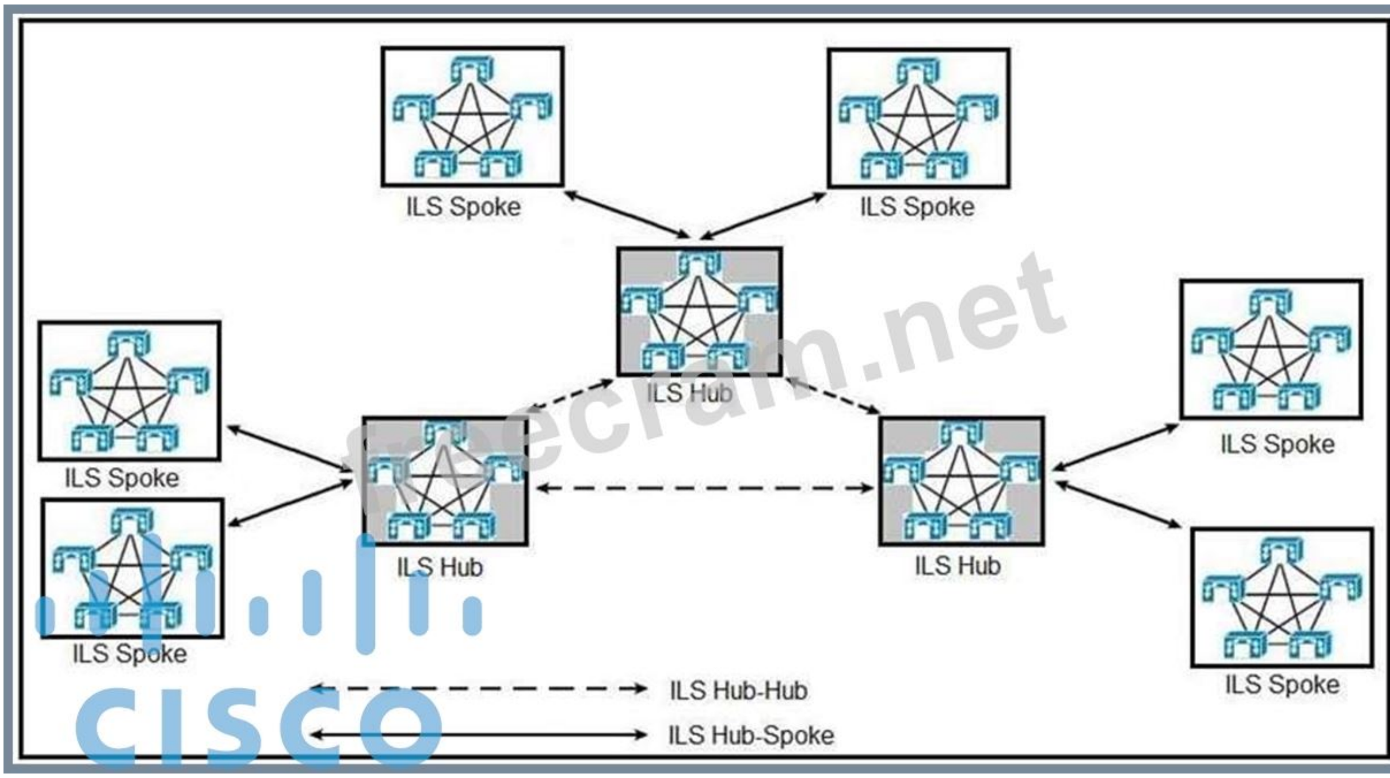
B. Increase the number of available media termination points.

C. Update the calling search space for affected endpoints to none.

D. Update all regions to 8 kbps maximum audio bitrate.

Answer: ([SHOW ANSWER]())

NEW QUESTION: 61



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

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

Answer: ([SHOW ANSWER]())
Section: Cisco Unified CM Call Control Features

NEW QUESTION: 62

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.
Answer: (SHOW ANSWER)
Section: Signaling and Media Protocols

NEW QUESTION: 63

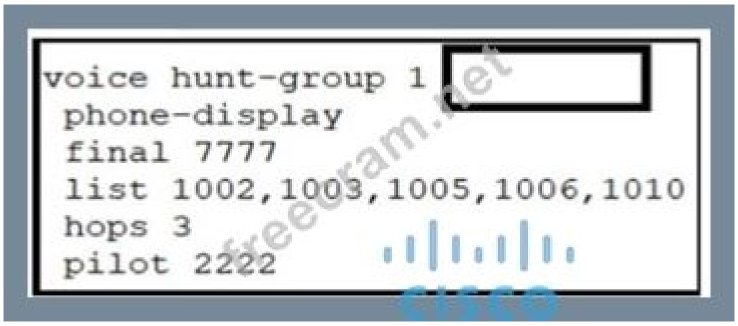
Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu.
Which two steps must be performed to resolve this issue? (Choose two.)
A. Restart the subscriber
B. Activate the Cisco Extended Functions service.
C. Activate the Cisco CallManager service.
D. Activate the Cisco Dialed Number Analyzer service.
E. Activate the Cisco Dialed Number Analyzer Server service.
Answer: D,E (LEAVE A REPLY)
Section: Call Control and Dial Planning

NEW QUESTION: 64
Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is one-way audio or no audio. You determine that

there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally.
What are two possible solutions? (Choose two.)
A. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 16384-32767
B. Ask the firewall administrator to change the ports to TCP.
C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.
D. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 20000-22000.
E. Go to System Parameters in Cisco Unified Communications Manager and change the range of media ports to 20000-22000.
Answer: (SHOW ANSWER)
Section: Signaling and Media Protocols
Explanation/Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/port/9_1_1/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-
91_chapter_01.html

NEW QUESTION: 65
Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)
A. TokenID
B. username and secret key
C. TLS certificates
D. passwords
E. FQDN of the servers defined in DNS
Answer: (SHOW ANSWER)
Section: Cisco Unified CM Call Control Features
Explanation/Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-
1151su1_chapter_011001.pdf

NEW QUESTION: 66
Refer to the exhibit.



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

DN 1003 was the last to ring during the most recent call. Which hunting method ensures that DN 1005 is presented with the next call when the hunt pilot is dialed?

A. parallel

B. call-blast

C. peer

D. sequential

Answer: (SHOW ANSWER)

NEW QUESTION: 67

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

A. Add the long-distance & international pattern's partitions to the calling search space of the device profile

B. Remove the long-distance & international pattern's partitions from the calling search space of the device profile.

C. Add the long-distance & international pattern's partitions to the calling search space of the physical phone.

D. Remove the long-distance & international pattern's partitions from the calling search space of the physical phone.

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

Answer: (SHOW ANSWER)

NEW QUESTION: 68

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

A. Configure IP Address Trusted Authentication for Incoming VoIP Calls.

B. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.

C. Configure the command no ip address trusted authenticate under "voice service voip".

D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Answer: (SHOW ANSWER)

NEW QUESTION: 69

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

A. interworking between h245-alpha numeric and sip-kpml

B. interworking between an OOB method and RFC2833 for flow-around calls

C. interworking between h245-signal and rtp-nte

D. interworking between an OOB method and RFC2833 for flow-through calls

Answer: (SHOW ANSWER)

NEW QUESTION: 70

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 Open Logical Channel Ack

B. H.245 Terminal Capability Set

C. H.225 Connect

D. H.245 Open Logical Channel

Answer: D (LEAVE A REPLY)

NEW QUESTION: 71

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

A. Contact: header of the 200 OK response

B. Allow: header if the 200 OK response

C. o= line of SDP content

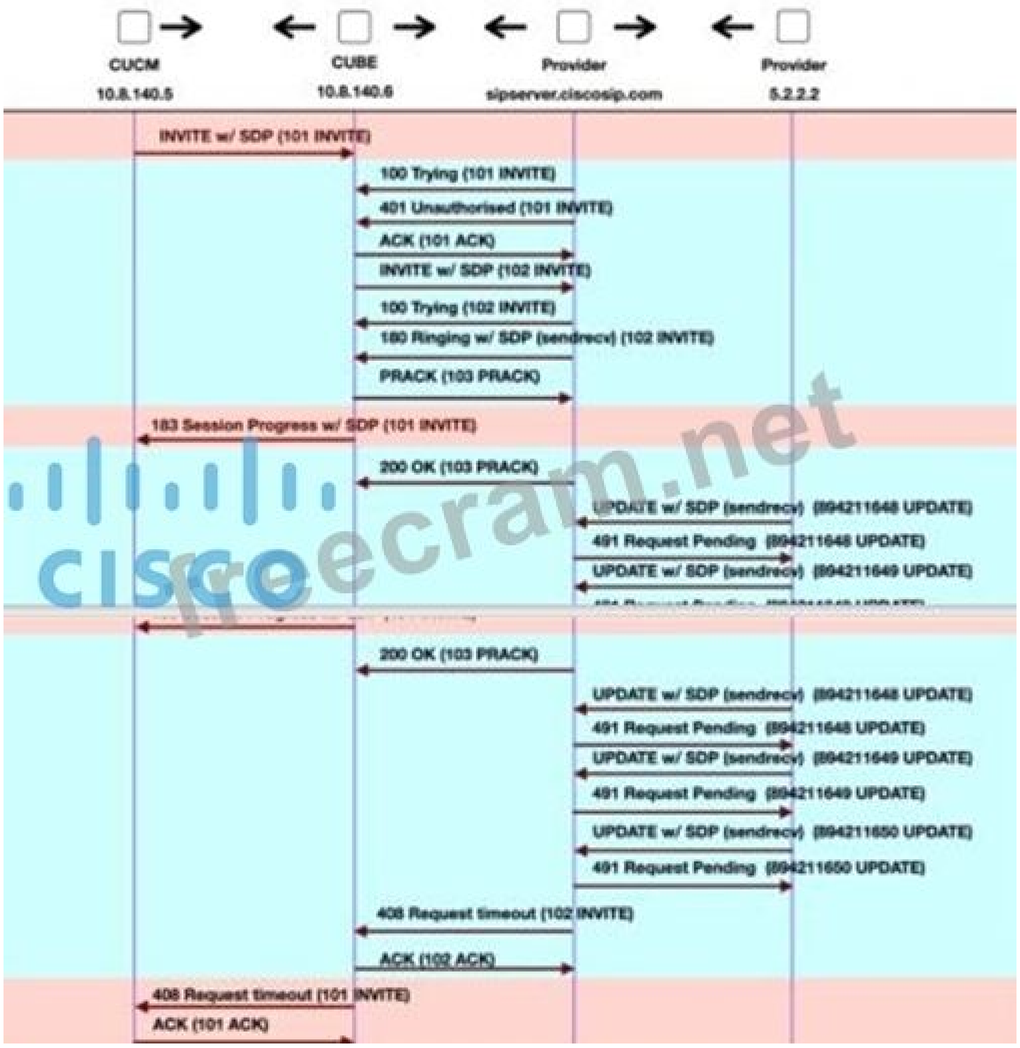
D. c= line of SDP content

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

NEW QUESTION: 72

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 *Early Offer support for voice and video calls" on the SIP Profile Configuration Page in Cisco UCM.

B. Enable 'SIP Rel1XX Options* and -Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.

C. Disable "Send send-receive SDP in mid-call INVITE* on the SIP Profile Configuration Page in Cisco UCM.

D. Disable "SIP Rel1XX Options* and 'Early Offer Support* on the SIP Profile Configuration Page m Cisco UCM.

Answer: (SHOW ANSWER)

NEW QUESTION: 73

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

A. 1XX (urgent Priority Set)

B. 12[2-5]

C. 12X (urgent priority set)

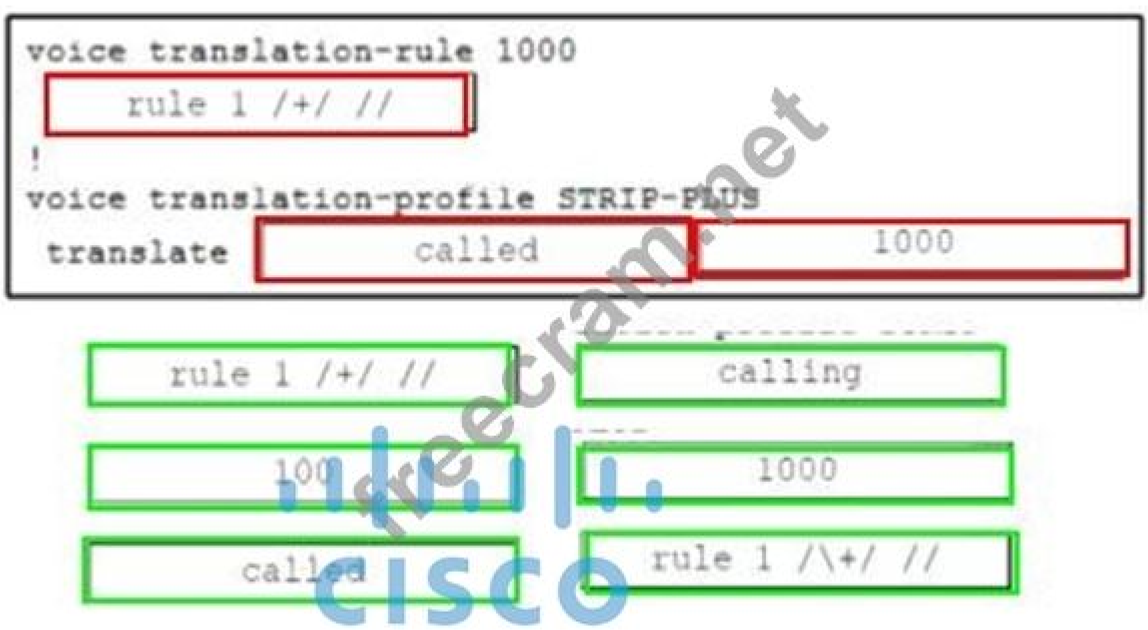
D. 12!

Answer: (SHOW ANSWER)

NEW QUESTION: 74

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.

Answer:



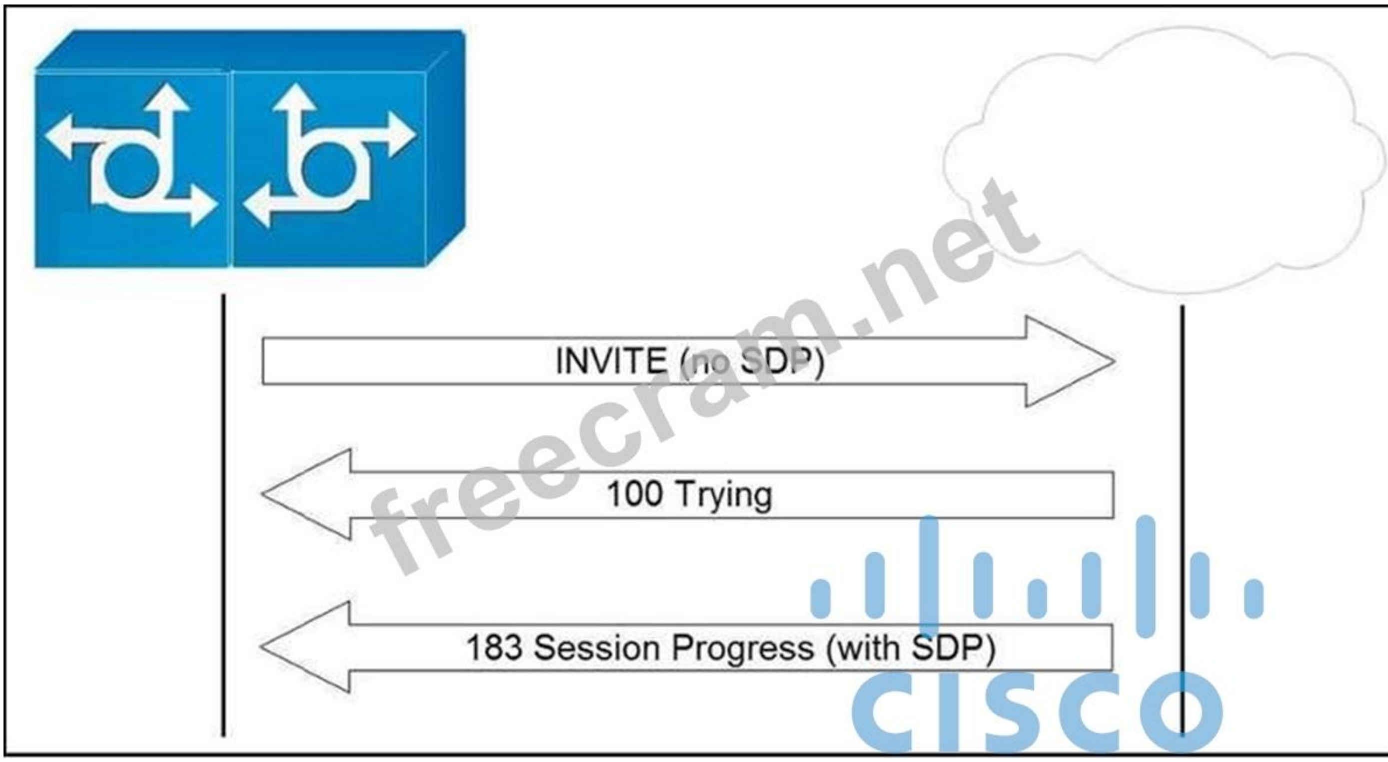
NEW QUESTION: 75

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 76



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

A. Enanle PRACK.

B. Enable Early Offer on the Cisco Unified Border Element.

C. Enable the supplementary-service media-renegotiate command.

D. Enable Media Flow Around

E. Enable Mid-Call Signaling Consumption.

Answer: (SHOW ANSWER)

Section: Cisco Unified Border Element

NEW QUESTION: 77

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 SIP Trunk Security Profile configuration

C. Incoming Port in Security Information of the SIP Profile configuration.

D. Destination Port in SIP Trunk Security Profile configuration

E. Destination Port in SIP Information section of the SIP Trunk configuration

Answer: (SHOW ANSWER)

NEW QUESTION: 78

The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323protocol (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.225 Connect

B. H.245 Open Logical Channel

C. H.245 Terminal Capability Set

D. H.245 Open Logical Channel Ack

Answer: (SHOW ANSWER)

NEW QUESTION: 79

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

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

NEW QUESTION: 80

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

A. voice-class sip bind media source-interface Loopback0

B. voice-class sip bind control source-interface Loopback0

C. voice-class sip early-offer forced.

D. voice-class sip resources priority mode passthrough
Answer: (SHOW ANSWER)

NEW QUESTION: 81
What is first preference condition matched in a SIP-enabled incoming dial peer?
A. incoming uri
B. target carrier-id
C. answer-address
D. incoming called-number
Answer: (SHOW ANSWER)
Section: Signaling and Media Protocols
Explanation
Explanation/Reference: https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In- Depth-Explanation-of-Cisco-IOS-and-IO.html#anc8

NEW QUESTION: 82
What are two configureation features of the Client matter code setting in the route pattern configuration? (Choose two.)
A. The client Matter Code feature supports overlap sending since the Cisco UCM can determine when to prompt the user for the code.
B. Selecting the Allow Overlap Sending setting disables the Require Client Matter Code setting.
C. Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.
D. The Client Matter Code feature provides the option to configure Authorization Level susch as in the Forced Authorization Code.
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 83
Calls to a particular extension are not routing to voicemail. The user reaches the voicemail system from the handset by pressing the Messages button Which configuration parameter causes this problem?
A. The voicemail pilot number for call forwarding is missing from the ephone-dn
B. The voicemail pilot number for call forwarding is missing from the ephone
C. The voicemail pilot number is missing from the call handling on Cisco Unity Express
D. The voicemail pilot number is missing from the telephony service configuration on Cisco UCME
Answer: A (LEAVE A REPLY)

NEW QUESTION: 84

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

A. R2(config-ephone-dn)#park reservation-group 1

B. R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002

C. R2(config-ephone-dn)#park reservation-group 60

D. R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Answer: (SHOW ANSWER)

NEW QUESTION: 85

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.

Answer: (SHOW ANSWER)

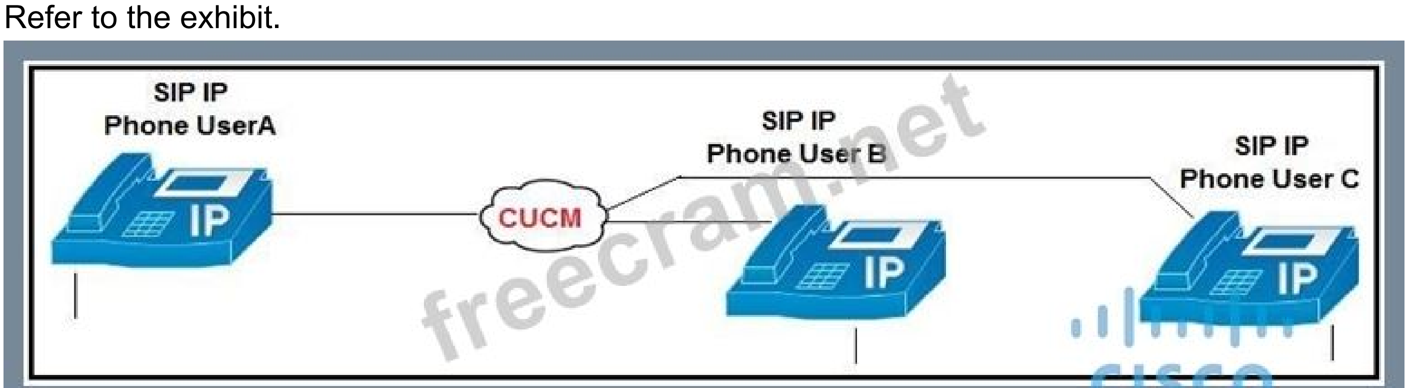
Section: Call Control and Dial Planning

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features- services-guide-100_chapter_010000.pdf

NEW QUESTION: 86

Refer to the exhibit.



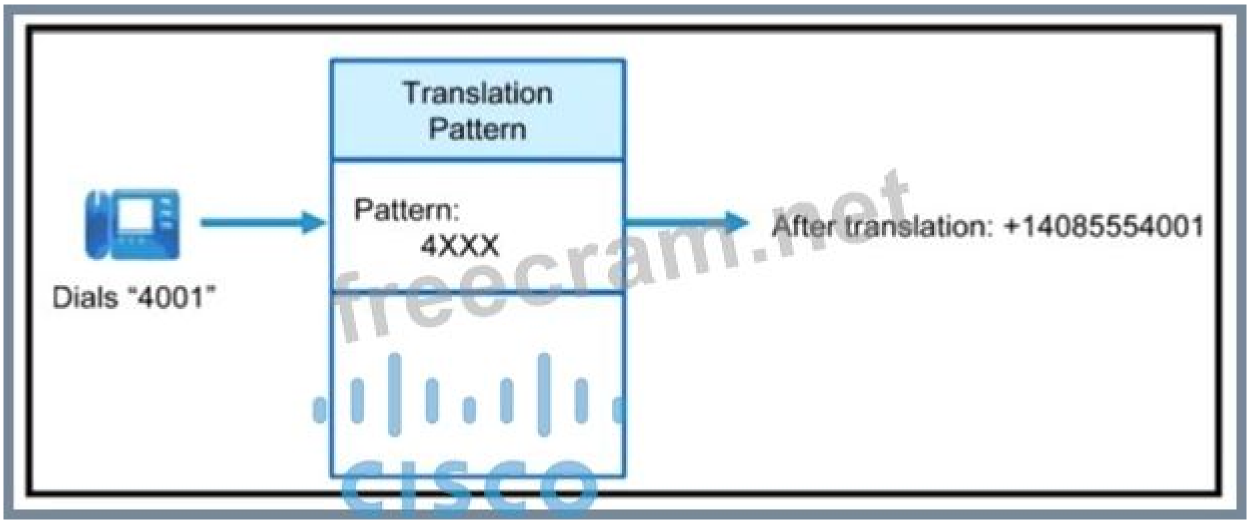
In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C.

Which two scenarios are correct? (Choose two.)
A. 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.
B. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
C. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
D. 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.
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_B User Hold MOH Audio Source settings.
Answer: (SHOW ANSWER)

NEW QUESTION: 87
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
Answer: (SHOW ANSWER)
Section: Call Control and Dial Planning

NEW QUESTION: 88
Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?
A. Call Routing > Route/Hunt > Local Route Group Names
B. Call Routing > Emergency Location > Emergency Location (ELIN) Groups
C. System > Device Pool
D. System > Location Info
Answer: (SHOW ANSWER)

NEW QUESTION: 89
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. Calling Party Transformation Mask of +1408555XXXX
C. Calling Party Transformation Mask of +14085554XXX
D. Called Party Transformation Mask of + 1408555[35)XXX

Answer: (SHOW ANSWER)