

Cisco.300-815.v2022-11-14.q89

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NEW QUESTION: 1

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: 2

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

- A. in the telephony service configuration
- B. in the queuing configuration
- C. in Cisco Unified CM Enterprise Parameters
- D. in Cisco Unified CM Service Parameters

Answer: B (LEAVE A REPLY)

Reference:

<https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200453-Configure-CUCM-Native-Call-Queuing-Feat.html>

NEW QUESTION: 3

Refer to the exhibit.

Answer: D,E ([LEAVE A REPLY](#))

NEW QUESTION: 5

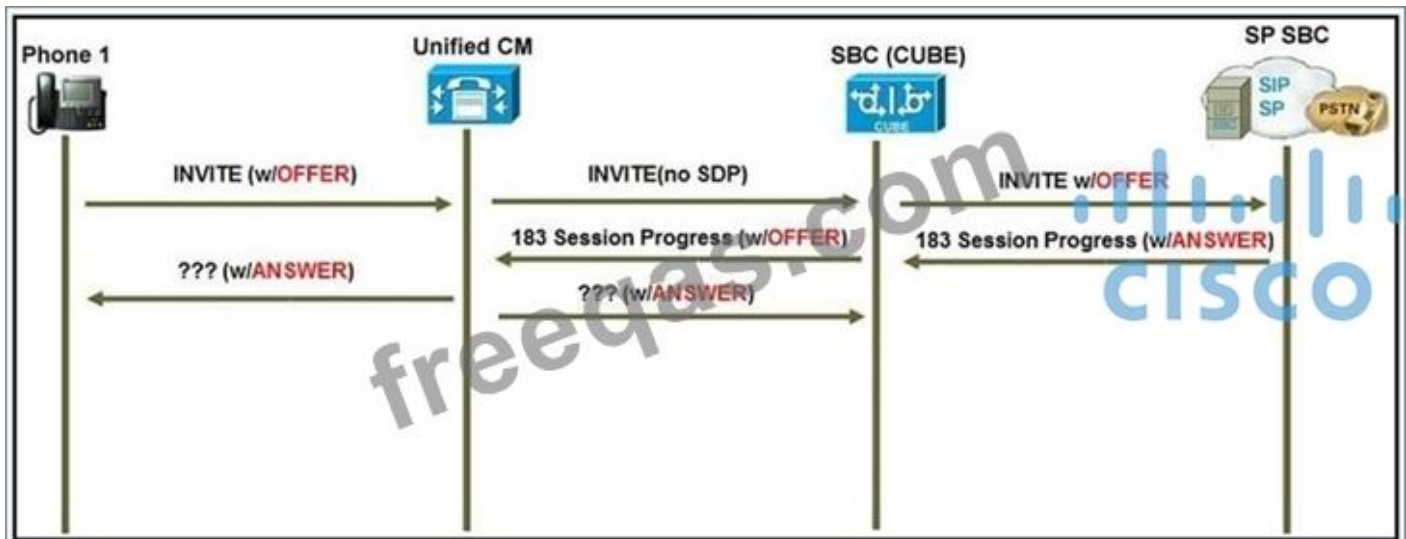
When a third-party SIP Phone System is dialed inbound across a Cisco Unified Border Element, DTMF is failing. The third-party vendor accepts only out-of-band DTMF. Which configuration should be added to the outgoing dial peer to resolve this issue?

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 6

Refer to the exhibit.



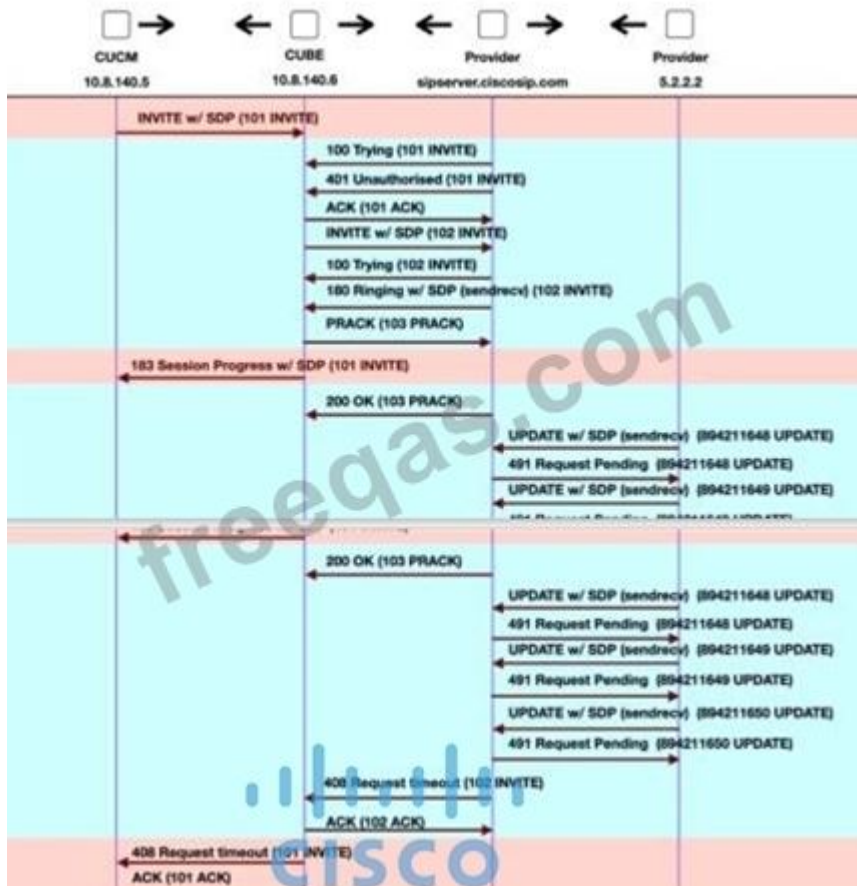
A user reports that when they call a specific phone number, no one answers the call, but when they call from a mobile phone, the call is answered. The engineer troubleshooting the issue is expecting the far-end gateway to cut through audio on the 183 Session Progress SIP message. Which SIP Profile configuration 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. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all 1xx Messages.
- C. Accept Audio Codec Preferences in Received Offer must be set to On.
- D. Early Offer for G Clear Calls must be enabled.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 7

Refer to the exhibit.



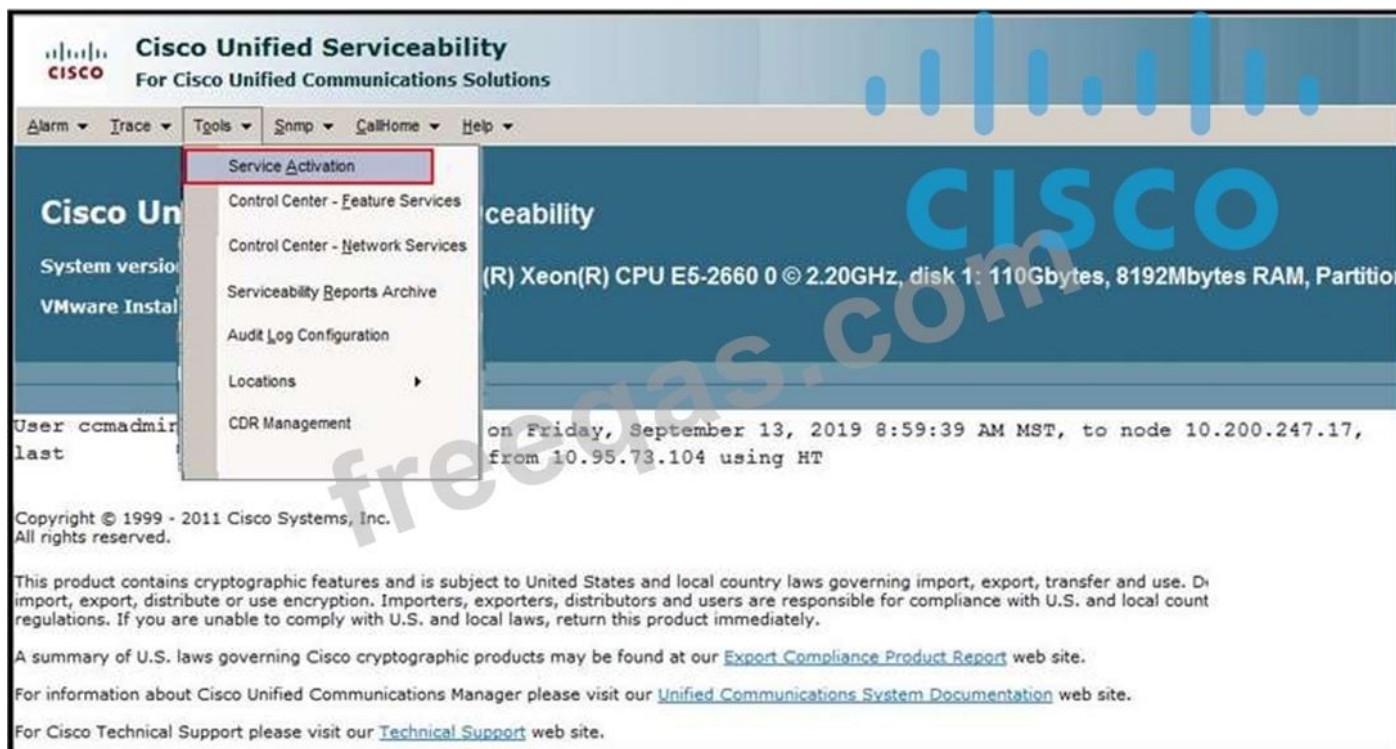
A Cisco Unified Border Element continues to send 180/183 with the required: 100rel header to Cisco UCM. and the call eventually disconnects How is the issue resolved?

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

Answer: B (LEAVE A REPLY)

NEW QUESTION: 8

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 CallManager service.
- C. Activate the Cisco Extended Functions service.
- D. Activate the Cisco Dialed Number Analyzer Server service.
- E. Activate the Cisco Dialed Number Analyzer service.

Answer: D,E (LEAVE A REPLY)

NEW QUESTION: 9

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?

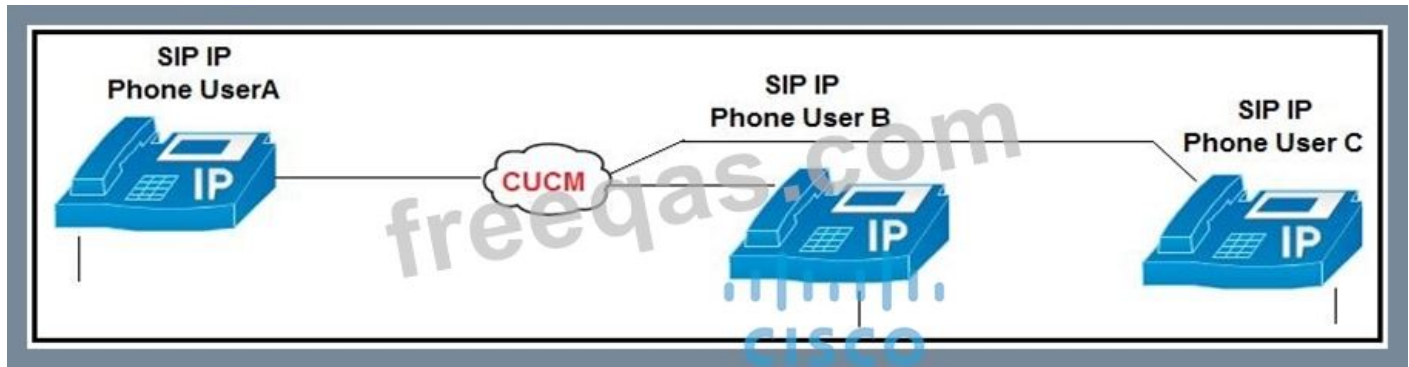
- A. Contact: header of the 200 OK response
- B. Allow: header of the 200 OK response
- C. o= line of SDP content
- D. c= line of SDP content

Answer: (SHOW ANSWER)

Section: Signaling and Media Protocols

NEW QUESTION: 10

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

Answer: C,E (LEAVE A REPLY)

NEW QUESTION: 11

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

Answer: C ([LEAVE A REPLY](#))

NEW QUESTION: 12

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.

Answer: A ([LEAVE A REPLY](#))

Reference:

<https://community.cisco.com/t5/collaboration-voice-and-video/taking-sip-call-trace-on-cisco-unified-cm-using-rtmt/ta-p/3161200>

NEW QUESTION: 13

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 14

When configuring hunt groups, where do you add the individual directory numbers that will be part of the group?

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

Answer: B ([LEAVE A REPLY](#))

Section: Cisco Unified CM Call Control Features

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_010101.html

NEW QUESTION: 15

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

Answer: (SHOW ANSWER)

NEW QUESTION: 16

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: A,C (LEAVE A REPLY)

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

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NEW QUESTION: 17

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: A,B (LEAVE A REPLY)

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: 18

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\([2-9]..[2-9].....\$\)/ \0/
- B. rule 1 /\([2-9]..[2-9].....\$\)/ \1/
- C. rule 1 /^+\([1].*\)/ /011\1/
- D. rule 1 /^+1\([2-9]..[2-9].....\$\)/ \1/

Answer: D (LEAVE A REPLY)

NEW QUESTION: 19

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

- A. The phone button template must have the Queue Status Softkey configured.
- B. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
- C. The maximum number of callers allowed in queue must be 10.
- D. A unicast music on hold audio source must be configured.
- E. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.

Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 20

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

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

Answer: A (LEAVE A REPLY)

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: 21

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. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".
- C. Change the destination-pattern on the outgoing dial peer to match "444333222".
- D. Set up translation-profile on the incoming dial peer to match incoming traffic.

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 22

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).

Answer: D ([LEAVE A REPLY](#))

Reference:

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

Answer: A,C ([LEAVE A REPLY](#))

NEW QUESTION: 24

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: B ([LEAVE A REPLY](#))

Section: Cisco Unified Border Element

NEW QUESTION: 25

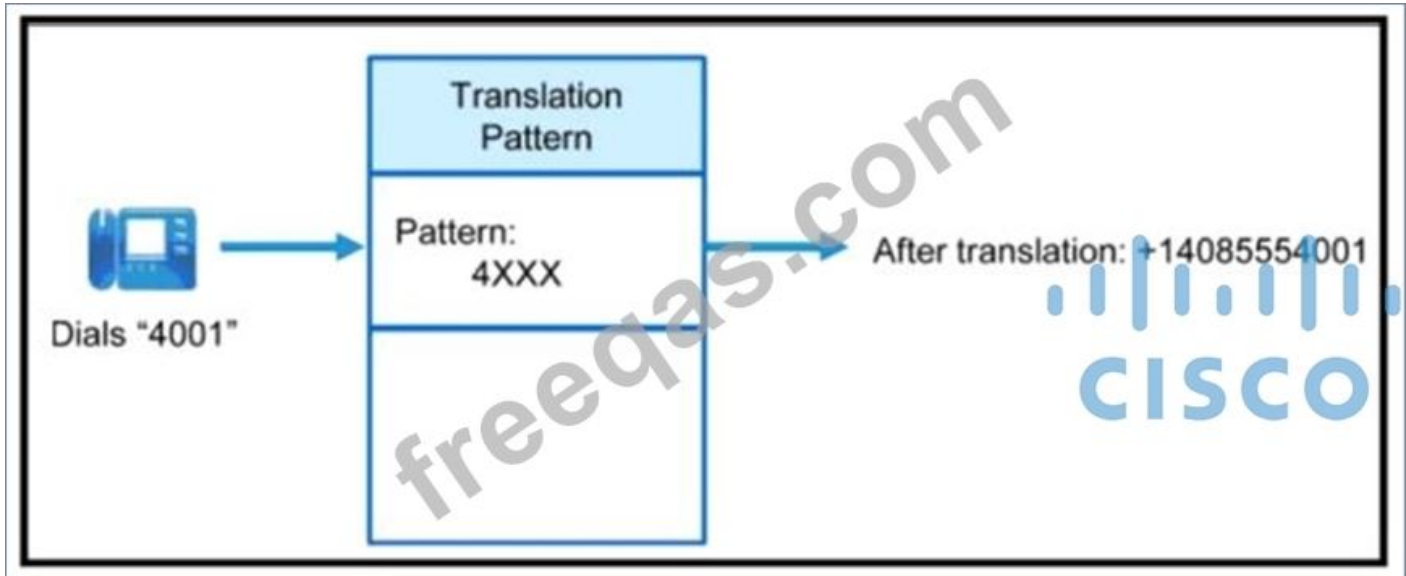
A customer routes PSTN calls to ITSP through a SIP trunk on Cisco UCM that forwards and receives calls to and from ITSP. ITSP is set to send an E.164 number when the customer's extension is four digits. Which action should be taken to route the incoming calls to four-digit extensions?

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 26

Refer to the exhibit.



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

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

Answer: C ([LEAVE A REPLY](#))

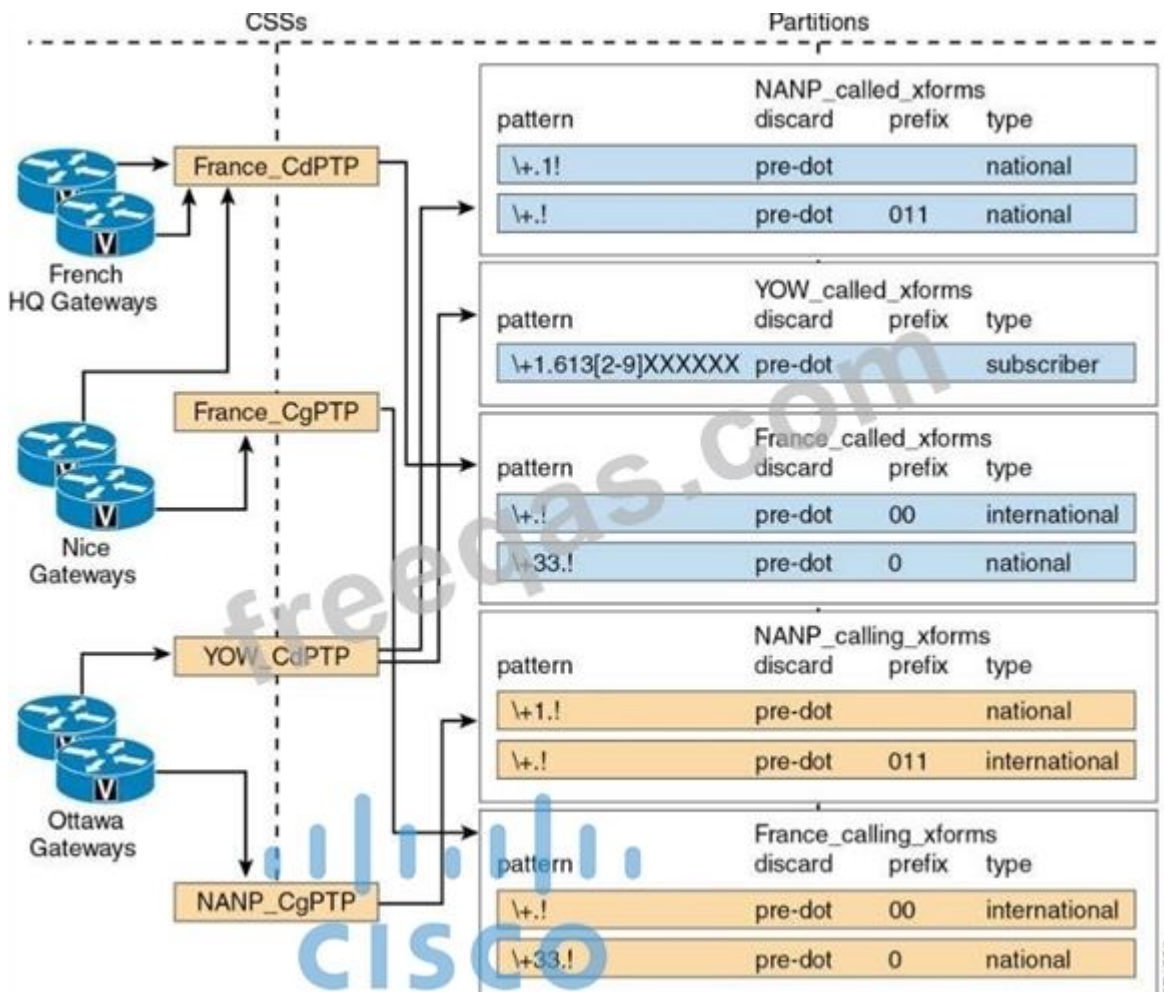
NEW QUESTION: 27

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.224 asn1
- C. debug H.323 asn 1
- D. debug H.225 media
- E. debug H.323 messages

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 28



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"

Answer: (SHOW ANSWER)

Section: Call Control and Dial Planning

Explanation/Reference:

NEW QUESTION: 29

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

- A. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service
- B. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility
- C. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service
- D. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 30

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

Refer to the exhibit. 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/

Answer: ([SHOW ANSWER](#))

Section: Cisco Unified Border Element

Explanation/Reference:

NEW QUESTION: 31

Which two types of distribution algorithm are within a line group? (Choose two.)

- A. random
- B. circular
- C. highest preference
- D. top down
- E. bottom up

Answer: B,D ([LEAVE A REPLY](#))

Section: Call Control and Dial Planning

Explanation/Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcfg/CUCM_BK_CDF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100011.html

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NEW QUESTION: 32

Refer to the exhibit.

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

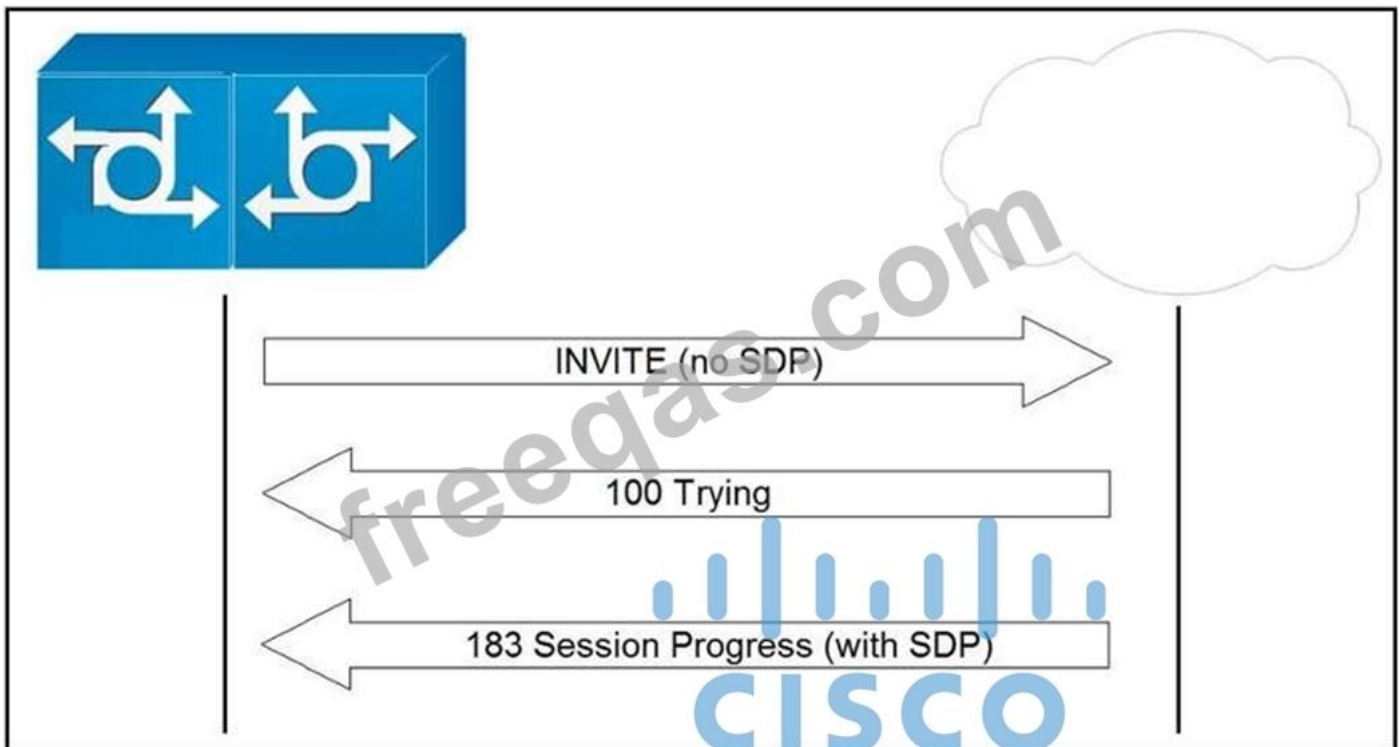
An engineer is troubleshooting an issue where inbound calls to Cisco UCM with early media fail to establish. While investigating the issue, the engineer finds that Cisco UCM is set to require a PRACK. but the Cisco Unified Border Element is not sending it. Which command is causing this issue?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 33

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 34

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

Answer: A ([LEAVE A REPLY](#))

Section: Call Control and Dial Planning

NEW QUESTION: 35

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

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

Answer: B (LEAVE A REPLY)

NEW QUESTION: 36

Refer to the exhibit.

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

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

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

Answer: D (LEAVE A REPLY)

NEW QUESTION: 37

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: 38

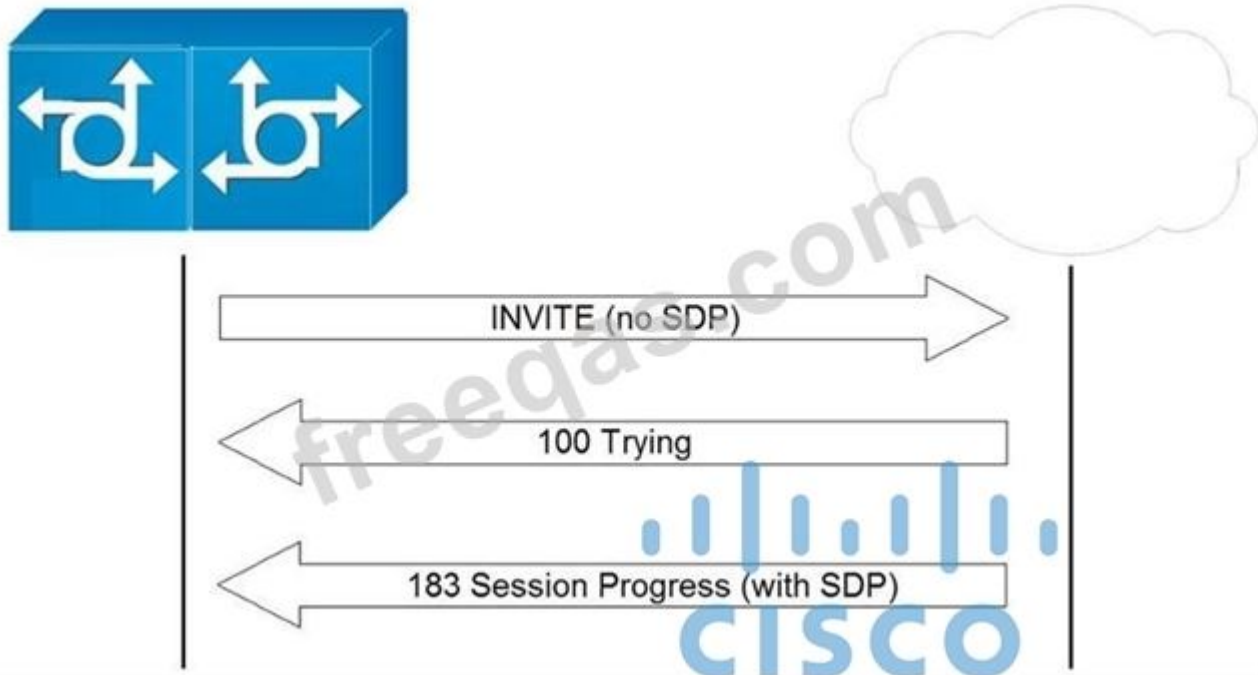
Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

- A. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.
- B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
- C. 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.
- D. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
- E. 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.

Answer: B,C (LEAVE A REPLY)

NEW QUESTION: 39

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

Answer: C,E (LEAVE A REPLY)

NEW QUESTION: 40

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!
- C. 12[2-5]
- D. 12X (urgent priority set)

Answer: D (LEAVE A REPLY)

NEW QUESTION: 41

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

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

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 42

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 43

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.246 asn 1
- C. debug H.225 media
- D. debug H.323 asn 1
- E. debug H.225 asn1

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 44

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]

Answer: ([SHOW ANSWER](#))

Section: Call Control and Dial Planning

NEW QUESTION: 45

Which IOS command creates a SIP-enabled dial peer?

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 46

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

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

Answer: (SHOW ANSWER)

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/dna/11_5_1/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115/CUCM_BK_CBA47A6E_00_cucm-dna-guide-115_chapter_01.html#CUCM_TP_A5DA99E0_00

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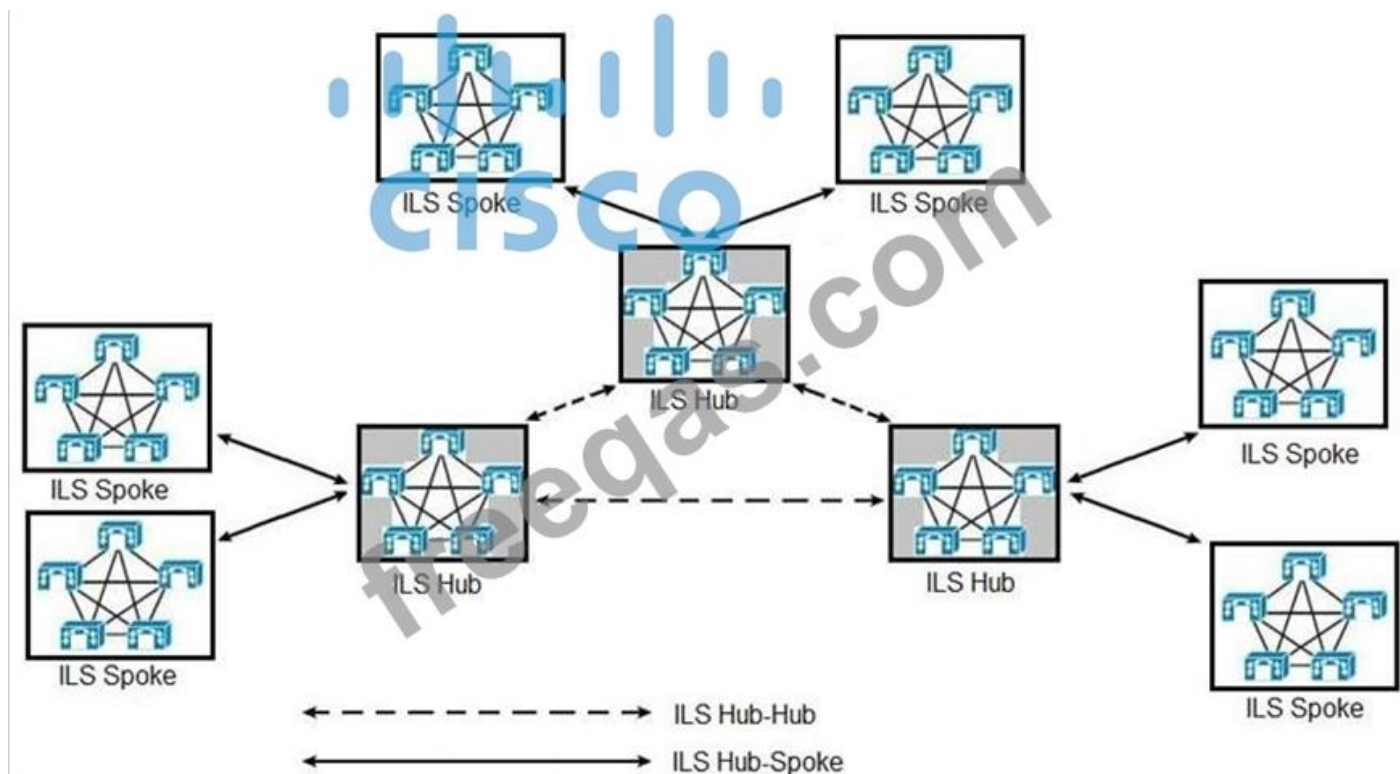
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NEW QUESTION: 47

Refer to the exhibit.



How many maximum hops can an ILS update traverse?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 48

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

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

Answer: D ([LEAVE A REPLY](#))

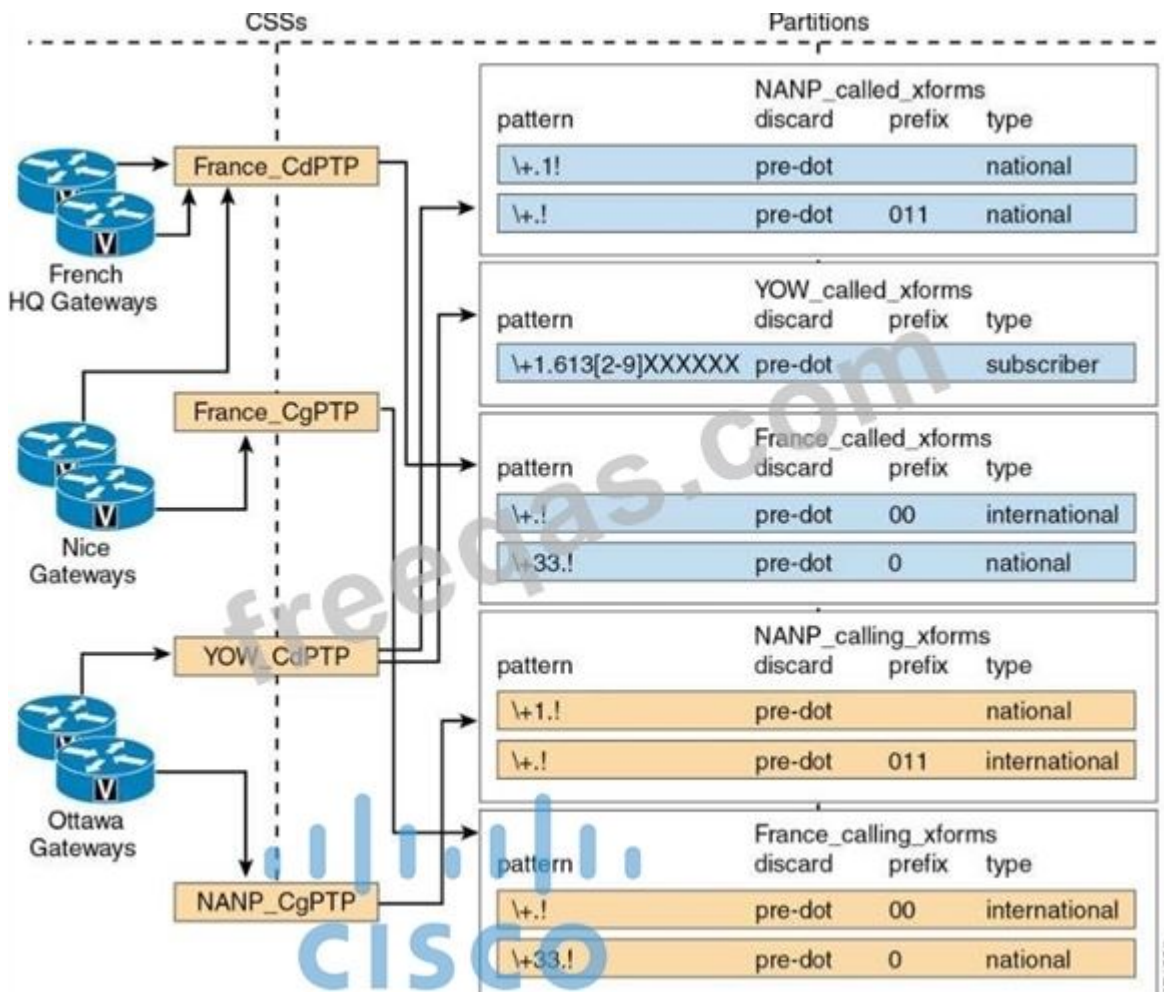
NEW QUESTION: 49

Refer to the exhibit. A standard local route group is configured for long-distance calls. Calls from building A succeed, but calls from building B fail. On the system. Each building has its own device pool. The DNA tool is used to test the configuration. How is this issue resolved?

- A. Modify the route pattern to add a prefix of 91
- B. Add a sip trunk inside route group Standard Local Route Group.
- C. Change the partition of the route pattern
- D. Add a local route group on the device pool configuration.

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 50



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"

Answer: (SHOW ANSWER)

Section: Call Control and Dial Planning

NEW QUESTION: 51

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. Set up translation-profile on the incoming dial peer to match incoming traffic.
- B. Create specific matching for "222333444" on the incoming dial peer.
- 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: A ([LEAVE A REPLY](#))

NEW QUESTION: 52



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.

Answer: A,B (LEAVE A REPLY)

Section: Cisco Unified Border Element

NEW QUESTION: 53

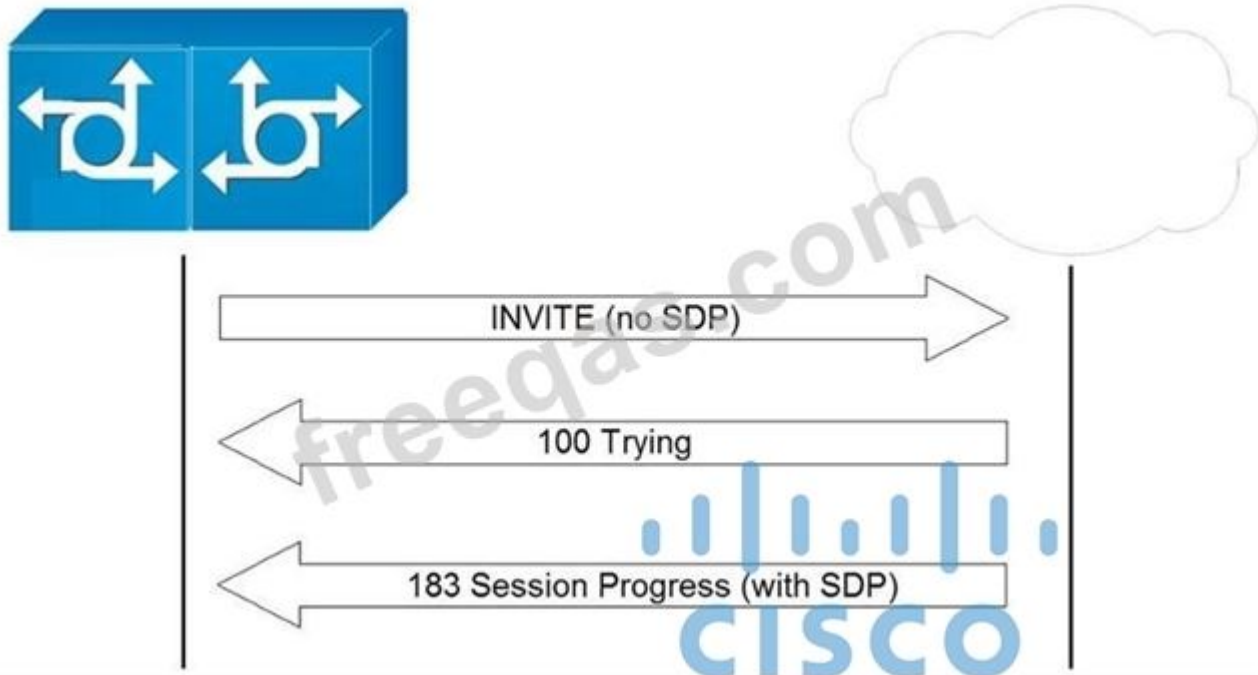
A user in location X dials an extension at location Y. The call travels through a QoS-enabled WAN network, but the user experiences choppy or clipped audio. What is the cause of this issue?

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

Answer: C (LEAVE A REPLY)

NEW QUESTION: 54

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

Answer: C,D (LEAVE A REPLY)

NEW QUESTION: 55

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

```

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

```

rule 1 /+/ //

calling

100

1000

called

rule 1 /\+/ //

Answer:

```

voice translation-rule 1000
  rule 1 /+/ //
!
voice translation-profile STRIP-8BUS
  translate called 1000

```

```

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

```

NEW QUESTION: 56

Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?

- A. System > Location Info
- B. Call Routing > Route/Hunt > Local Route Group Names
- C. System > Device Pool
- D. Call Routing > Emergency Location > Emergency Location (ELIN) Groups

Answer: B ([LEAVE A REPLY](#))

Reference:

<https://www.uccollabing.com/configuring-standard-local-route-group-cucm/>

NEW QUESTION: 57

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

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 58

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 dpq 888
  no vad
!
voice class dpq 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. Set up translation-profile on the incoming dial peer to match incoming traffic.
- B. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".
- C. Change the destination-pattern on the outgoing dial peer to match "444333222".
- D. Create specific matching for "222333444" on the incoming dial peer.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 59

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

(Choose two.)

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

Answer: C,D (LEAVE A REPLY)

NEW QUESTION: 60

Refer to the exhibit.



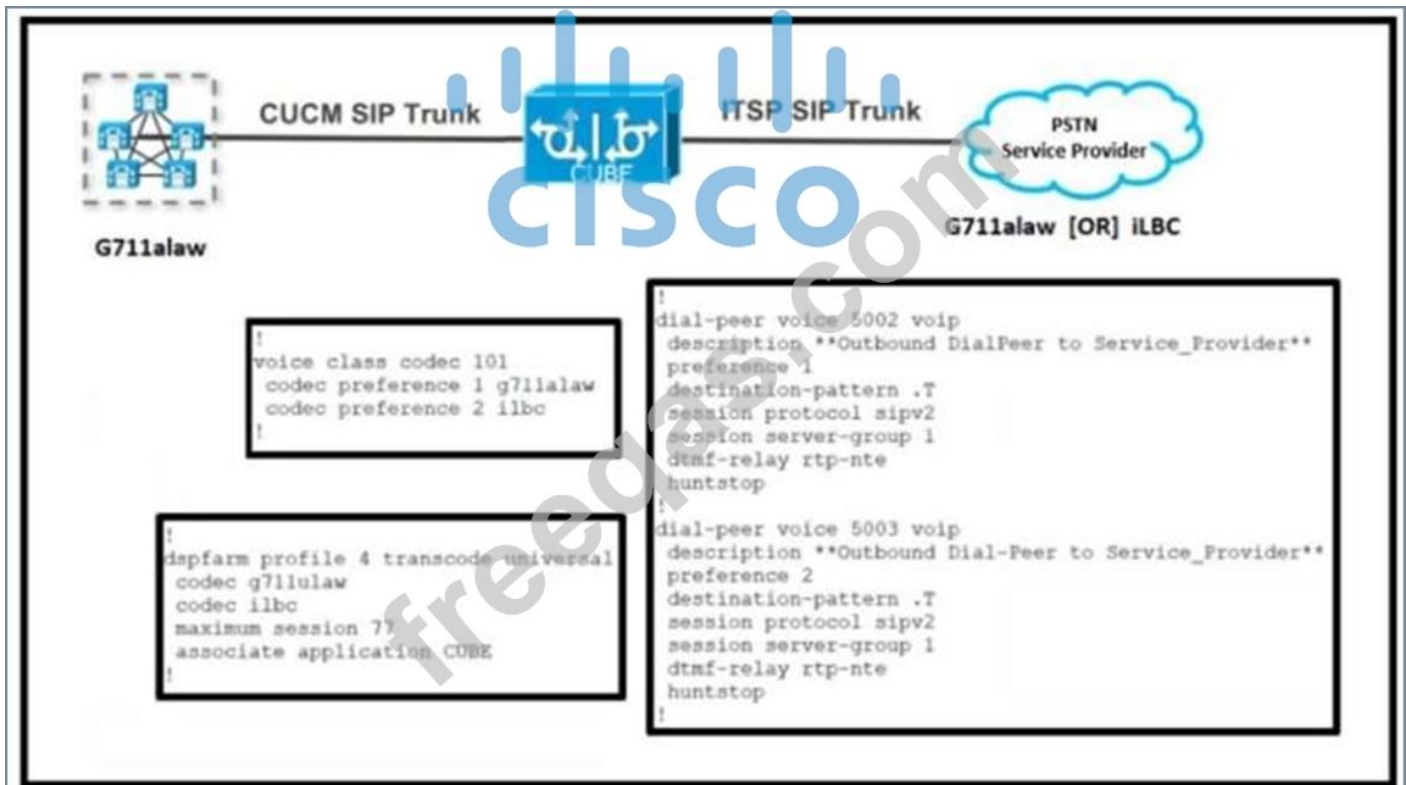
An administrator is troubleshooting a problem in which some outbound calls from an internal network to the Internet telephony service provider are not getting connected, but some others connect successfully. The firewall team found that some call attempts on port 5060 came from an unrecognized IP that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

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

Answer: A (LEAVE A REPLY)

NEW QUESTION: 61

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

Answer: D ([LEAVE A REPLY](#))

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NEW QUESTION: 62

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

- A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
- B. A unicast music on hold audio source must be configured.
- C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
- D. The maximum number of callers allowed in queue must be 10.
- E. The phone button template must have the Queue Status Softkey configured.

Answer: A,C (LEAVE A REPLY)

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_01001101.html#CUCM_RF_C960BC9A_00

NEW QUESTION: 63

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

Answer: (SHOW ANSWER)

NEW QUESTION: 64

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

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

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 65

Refer to the exhibits.

Region Configuration Related Links: [Back To Find/List](#)

Save

Region Information

Name*

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*

Description

Devices for this Group

Available Media Resources*

Selected Media Resources*

CallManager SDL Log

```
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 numXcodesRequired=1 soundingside=1]
[SDL]sig: [MemAllocateMtpResourceErr [waitResourcesAllocated
[MediaManager (6,100,144,22401) [MediaResourceManager (6,100,142,1)
```

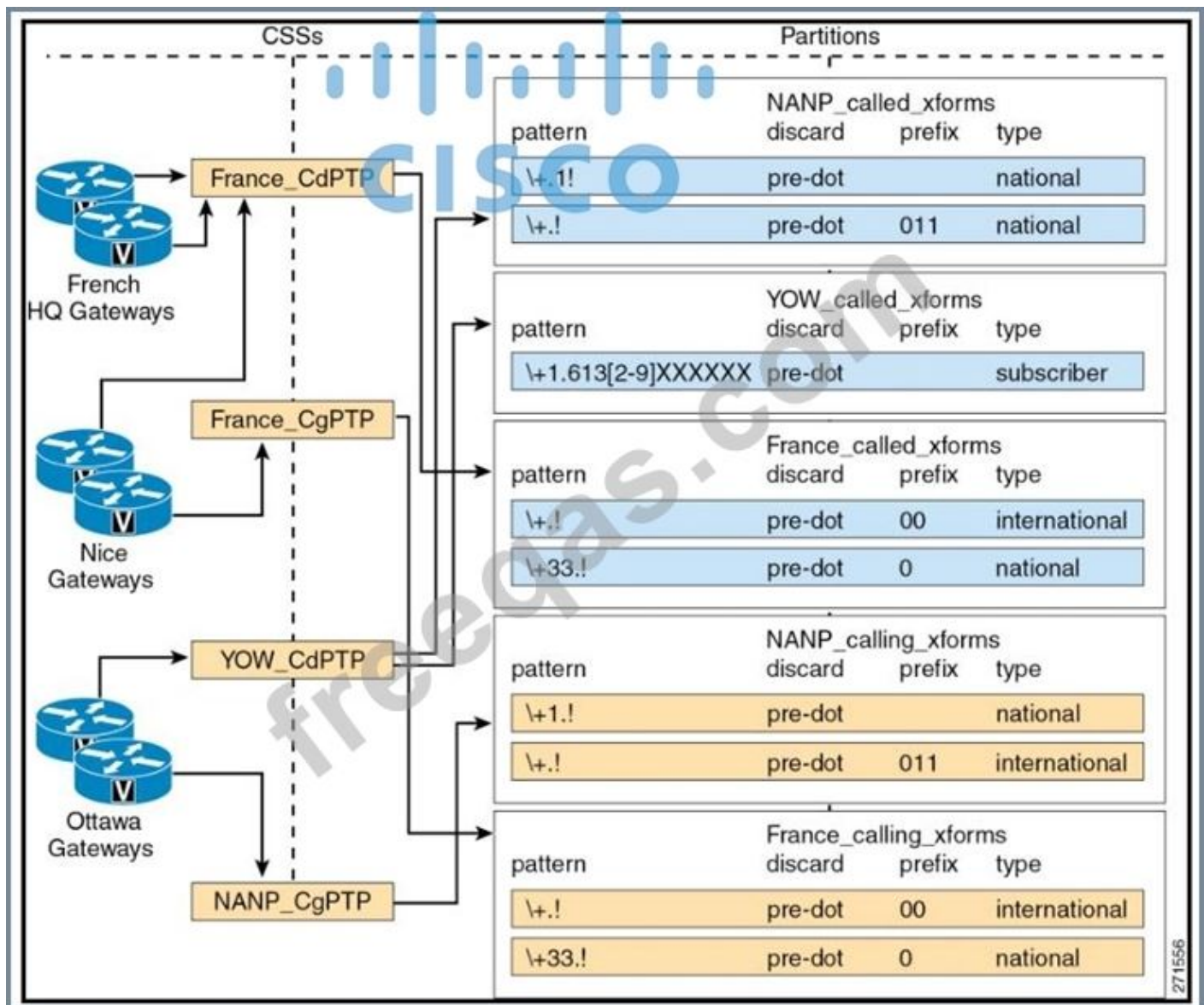
Regions have been configured for all major branches based on the available circuit bandwidth. Some calls from Region A endpoints to Region B endpoints are failing to connect. How is this issue resolved?

- A. Update the calling search space for affected endpoints to none.
- B. Add a media resource to transcode between available capabilities.
- C. Update all regions to 8 kbps maximum audio bitrate.
- D. Increase the number of available media termination points.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 66

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

Answer: A (LEAVE A REPLY)

NEW QUESTION: 67

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

- A. passwords
- B. FQDN of the servers defined in DNS
- C. TokenID

- D. TLS certificates
- E. username and secret key

Answer: A,D (LEAVE A REPLY)

NEW QUESTION: 68

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

Answer: C (LEAVE A REPLY)

Section: Signaling and Media Protocols

Explanation/Reference: <https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.html>

NEW QUESTION: 69

An administrator is implementing a new dial-plan on Cisco Unified Border Element. The administrator must ensure that incoming dial-peers are matched based on the IP address from where the incoming request originates. Which dial-peer configuration should be applied to accomplish this requirement?

- A. dial-peer voice 1 voip
incoming url request
- B. dial-peer voice 1 voip
incoming called-number
- C. dial-peer voice 1 voip
incoming url to
- D. dial-peer voice 1 voip
incoming url via

Answer: D (LEAVE A REPLY)

NEW QUESTION: 70

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

Answer: B,D (LEAVE A REPLY)

Section: Call Control and Dial Planning

NEW QUESTION: 71

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: A,C (LEAVE A REPLY)

Section: CME/SRST Gateway Technologies

Explanation/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: 72

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

Answer: A (LEAVE A REPLY)

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_call_handling.html

NEW QUESTION: 73

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. *.*

Answer: ([SHOW ANSWER](#))

Section: Call Control and Dial Planning

NEW QUESTION: 74

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

A. can be associated under the route group

B. chooses the route group that is configured under the device pool of the calling-party device

C. chooses the route group that is configured under the device pool of the called-party device

D. can be associated only under the route list

E. can be assigned directly to the route pattern

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 75

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).

Answer: D ([LEAVE A REPLY](#))

Section: Signaling and Media Protocols

Explanation/Reference: <https://www.cs.columbia.edu/~hgs/rtp/faq.html>

NEW QUESTION: 76

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

A. incoming uri

B. target carrier-id

C. answer-address

D. incoming called-number

Answer: A ([LEAVE A REPLY](#))

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>

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NEW QUESTION: 77

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/cmecmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

NEW QUESTION: 78

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. c= line of SDP content
- D. o= line of SDP content

Answer: D (LEAVE A REPLY)

NEW QUESTION: 79

Which two types of distribution algorithm are within a line group? (Choose two.)

- A. random
- B. circular
- C. highest preference
- D. top down
- E. bottom up

Answer: (SHOW ANSWER)

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcf/CUCM_BK_CDF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100011.html

NEW QUESTION: 80

Refer to the exhibit.

The image shows two screenshots from a Cisco CallManager configuration interface. The left screenshot is the 'Pattern Definition' configuration page for a translation pattern. The 'Translation Pattern' is set to '91.[2-9]XX[2-9]XXXXXX'. The 'Partition' is set to '< None >'. The 'Route Class' is set to 'Default'. The 'Calling Search Space' is set to 'PSTN_CSS'. The 'Route Option' is set to 'Route this pattern'. The 'Calling Party Transformations' section shows 'Use Calling Party's External Phone Number Mask' is disabled, and the 'Calling Party Transform Mask' is set to '9195551234'. The right screenshot is the 'DNA Analysis Output' for the translation pattern. It shows the 'Results Summary' with 'Calling Party Information' including 'Calling Party = 9195552304' and 'Match Result = RouteThisPattern'. The 'Call Flow' section shows 'Route Pattern :Pattern= [2-9]XX[2-9]XXXXXX' and 'PreTransform Calling Party Number = 9195551234'.

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

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

Answer: B (LEAVE A REPLY)

NEW QUESTION: 81

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. AUDIO
- C. FAX
- D. BFCP
- E. VIDEO

Answer: A,D (LEAVE A REPLY)

NEW QUESTION: 82

```
!
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: D (LEAVE A REPLY)

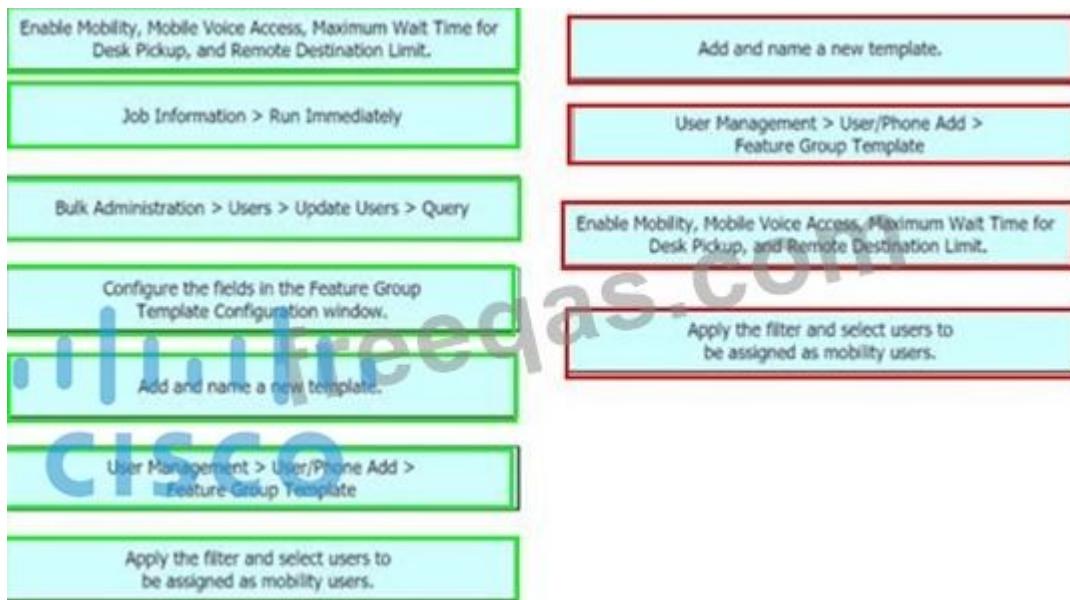
Section: Call Control and Dial Planning

NEW QUESTION: 83

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

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

Answer:



NEW QUESTION: 84

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

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

Answer: D (LEAVE A REPLY)

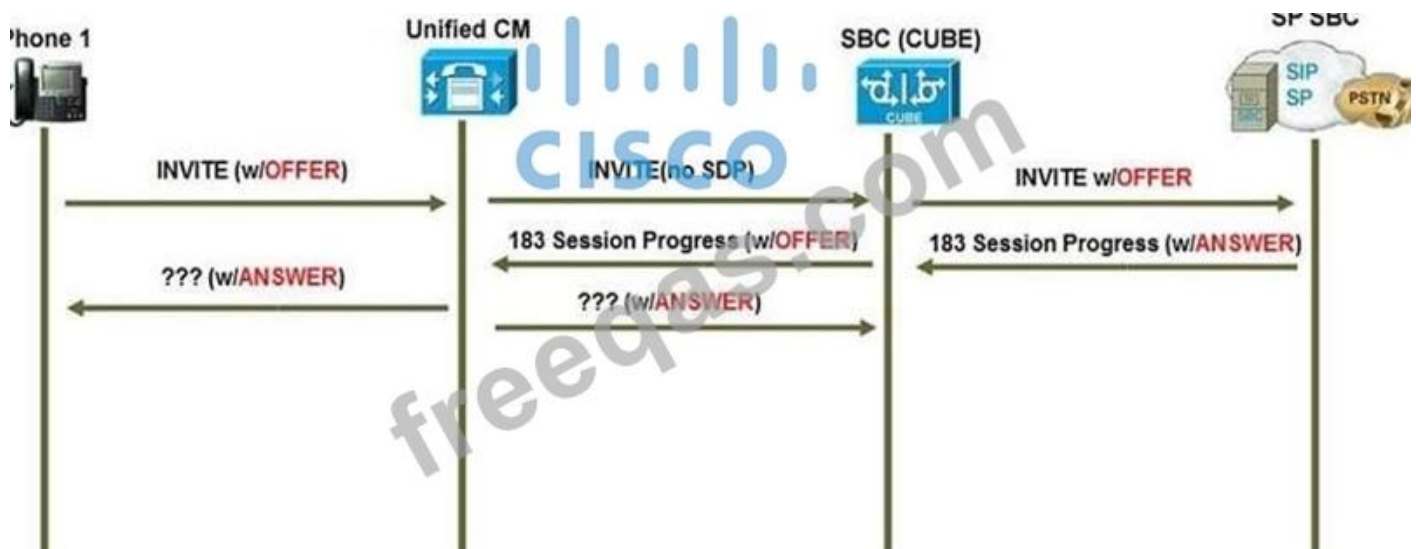
NEW QUESTION: 85

An administrator is trying to apply configuration changes on Cisco CME. When the users registered on Cisco CME to dial a local number to a PSTN call, the Cisco CME sends an incorrect number of digits. What translation rule fixes the issue and sends the correct number of digits?

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

Answer: (SHOW ANSWER)

NEW QUESTION: 86



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

Explanation

NEW QUESTION: 87

Refer to the exhibit. An engineer is troubleshooting an issue where inbound Calls are failing after they transferred. The provider reports that update is not supported, and this is causing the calls to fail. Which command should resolve this issue?

- A. no midcall-signaling passthru
- B. rel1xx require "100rel"
- C. no contact-passig
- D. no update-callerId

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 88

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 Device CSS under Current Device Mobility Settings.

- B. Enable SRST under Current Device Mobility Settings
- C. Set correct Location under Current Device Mobility Settings
- D. Set the correct subnet under Device Mobility Info.

Answer: D ([LEAVE A REPLY](#))

NEW QUESTION: 89

| Pattern | Description | Partition | Route Filter | Associated Device |
|---------------|---------------------|-----------------|--------------|----------------------|
| 41XXXX | To AMER Cluster | Global-Internal | | 2-AMER-RL |
| 55XX | Rendezvous meetings | Global-Internal | | Rendezvous-Conductor |
| 9.0XXXXXXXXXX | 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 timer 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: C ([LEAVE A REPLY](#))

Section: Call Control and Dial Planning

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