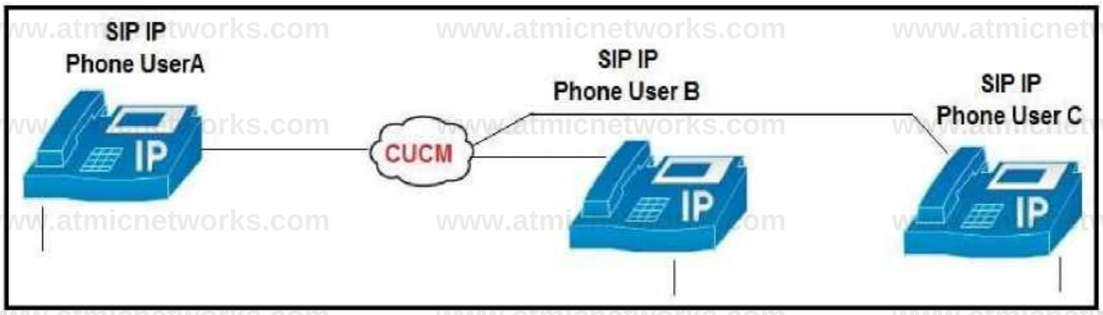
"Please note that these files may not be up to date. However, the questions will help you understand the exam format and typical question patterns."

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. 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: AD

Explanation:

Question: 2

Refer to the exhibit.

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

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

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

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

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

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

D. No DTMF is negotiated.

Answer: D

Explanation:

Question: 3

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: B

Explanation:

Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

Question: 4

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

A. DTMF

B. BFCP

C. VIDEO

D. FAX

E. AUDIO

Answer: AB

Explanation:

Question: 5

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

A. ALERTING

B. PROCEEDING

C. CONNECT

D. RINGING

Answer: A

Explanation:

Question: 6

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: D

Explanation:

Question: 7

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: C

Explanation:

Question: 8

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

A. debug H.323 messages

B. debug H.225 asn1

C. debug H.246 asn 1

D. debug H.225 media

E. debug H.323 asn 1

Answer: B

Explanation:

Question: 9

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

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

Question: 10

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: AC

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

Question: 11

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

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

Question: 12

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

Explanation:

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

Question: 13

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

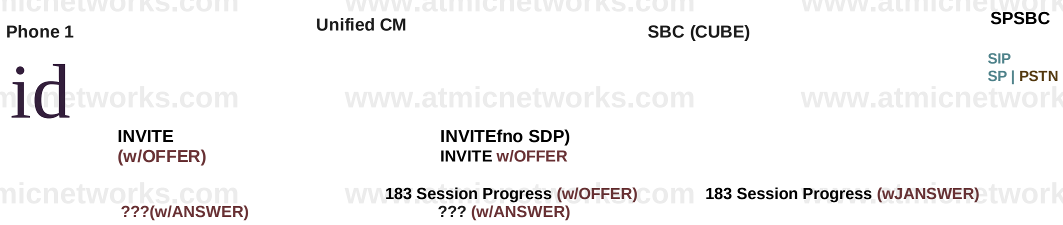
D. Call logs

Answer: A

Explanation:

Question: 14

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: C

Explanation:

Question: 15

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: AC

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

Question: 16

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: B

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

Question: 17

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

A. configuration for a single SIP phone

B. configuration items common for all SIP phones

C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)

D. configuration for SIP registrar service

Answer: A

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_setting_up_using_sip.html

Question: 18

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: B

Explanation:

Reference: https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified- communications-manager-express/99946-cme-sip-guide.html

Question: 19

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: A

Explanation:

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border- element/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35

Question: 20

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

A. in the voice-class VoIP configuration

B. in the VoIP dial peer

C. in global SIP configuration

D. in the VoIP or POTS dial peers

Answer: B

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cubebook/dtmf- relay.html

Question: 21

Refer to the exhibit.

```
voice translation-profile incoming translate called 999
1

voice translation-rule 999
                                                              A) s /     / \2333\1/

     rule 1/\ ("[1-2] [1-2] [l-2]\ } 333\ ([4-5] [4-5]
I
dial-peer voice 999 voip
     translation-profile outgoing incoming
     session protocol sipv2
     incoming called-number
     dtmf-relay rtp-nte codec transparent destination
     dpg 888 no vad

voice class dpg 888
     dial-peer 888

dial-peer voice 888 voip
     destination-pattern 888
     session protocol sipv2
     session target ipv4:192.168.0.1 codec transparent
     dtmf-relay rtp-nte no vad
```

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

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

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

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

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

Answer: B

Explanation:

Question: 22

Refer to the exhibit.

voice translation-rule 84
rule 1 r\ ([2-9]..[2-9].$\)/A2/

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/

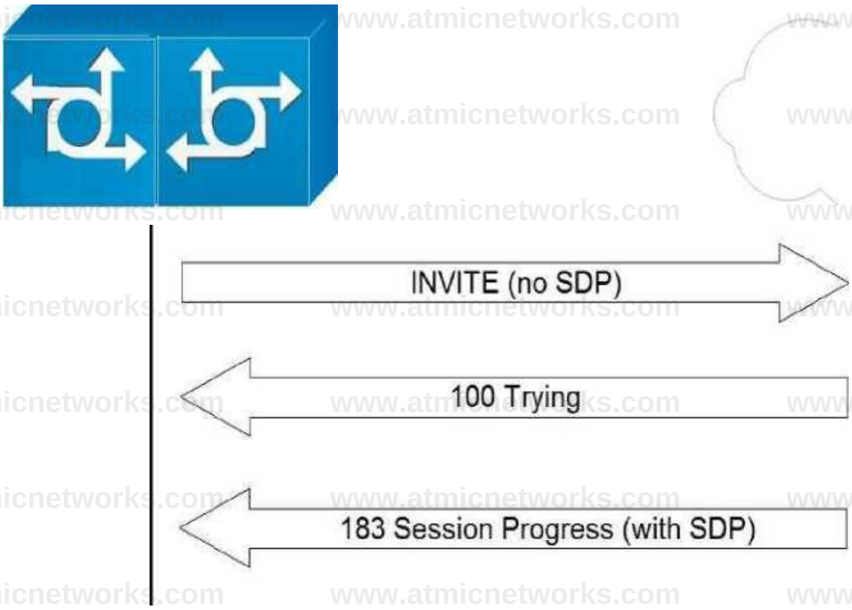
D. rule 1 /^\+1\([2-9]..[2-9]……$\)/ /\0/

Answer: B

Explanation:

Question: 23

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: AB

Explanation:

Question: 24

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

Explanation:

Reference: https://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

Question: 25

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: A

Explanation:

Question: 26

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

**

Answer: BE

Explanation:

Question: 27

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: AB

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

Question: 28

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: A

Explanation:

Question: 29

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: BC

Explanation:

Question: 30

Refer to the exhibit.

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

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

A. answer-address 555 ……..

B. codec g729

C. session-protocol sipv2

D. incoming called number 555

Answer: C

Explanation:

Question: 31

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: D

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_call_handling.html

Question: 32

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: CD

Explanation:

Question: 33

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

A. MTP

B. CCSIP

C. RTMT

D. OS Administration Page

Answer: C

Explanation:

Question: 34

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: D

Explanation:

Question: 35

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: D

Explanation:

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

Question: 36

Refer to the exhibit.

Route Patterns *(1–5 of 5)*

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

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

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

A. Adjust the service parameter T302 timet to the desired value.

B. Adjust the service parameter T204 timer to the desired value.

C. Check the Urgent Priority check box under 9.911 pattern.

D. Point the emergency pattern directly to the PSTN gateway.

Answer: C

Explanation:

Question: 37

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: CE

Explanation:

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

Question: 38

Refer to the exhibit.

Cisco Unified Serviceability
For Cisco Unified Communications Solutions

Alarm Trace ▼ | Tools ▼ j Snmp ▼ CallHome ▼ Help

Service Activation
Audit Log Configuration

Locations

Us Cisco U : Control Center - Feature Services CDR Management rceability
last Control Center - Network Services

System versio (R) Xeon(R) CPU E5-2660 0©2.20GHz, disk 1: HOGbytes, 8192Mbytes RAM, Partitior

VMware Instal Serviceability Reports Archive

on Friday, September 13, 2019 8:59:39 AM MST, to node 10.200.247.17, from 10.95.73.104 using HT

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Di import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local count regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.

For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.

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: DE

Explanation:

Question: 39

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

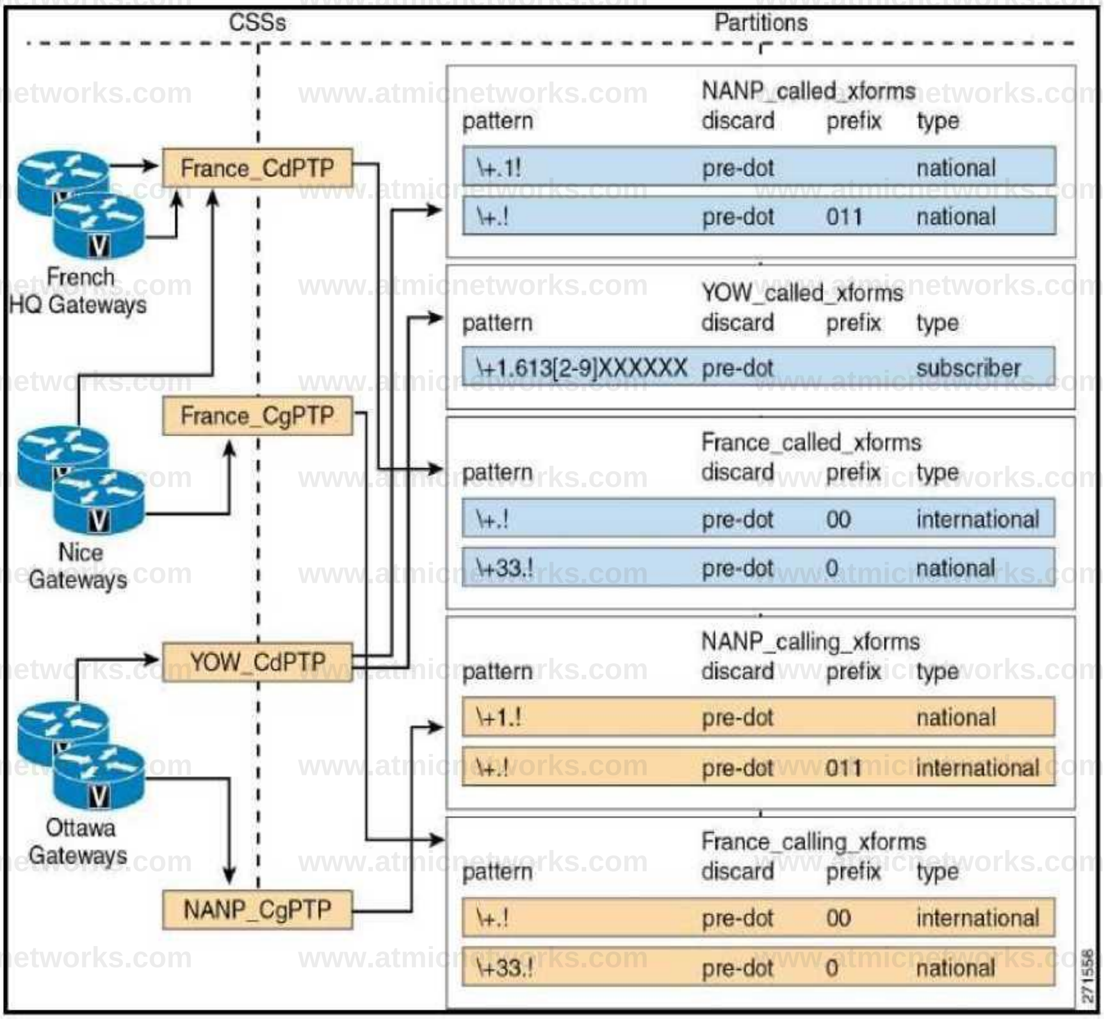
D. SDL trace

Answer: A

Explanation:

Question: 40

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: D

Explanation:

Question: 41

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

Explanation:

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

Question: 42

How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?

A. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.

B. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.

C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.

D. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.

Answer: D

Explanation:

Question: 43

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: BD

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

Question: 44

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: D

Explanation:

Question: 45

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

Explanation:

Reference: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/4_1_3/ccmsys/a07cpick.html#wp1022865

Question: 46

When locations-based Call Admission Control denies the call, which two masks can AAR apply when routing the call through the PSTN? (Choose two.)

A. AAR destination mask

B. called party transform mask

C. external phone number mask

D. +E.164 alternate number mask

E. enterrise alternate number mask

Answer: AC

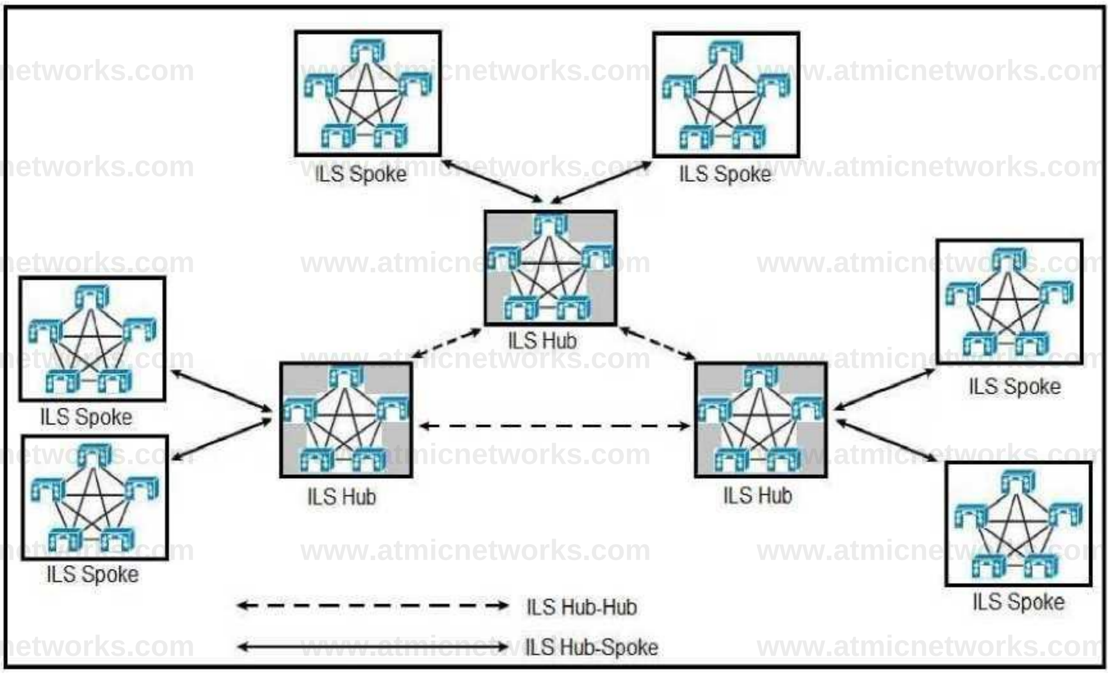
Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.ht ml

Question: 47

Refer to the exhibit.



How many maximum hops can an ILS update traverse?

A. 3

B. 6

C. 9

D. 12

Answer: A

Explanation:

Question: 48

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: A

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

Question: 49

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

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

Question: 50

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: B

Explanation:

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

Question: 51

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: CD

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

Question: 52

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: AC

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_01001101.html#CUCM_RF_C960BC9A_00

Question: 53

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 multiple time schedules assigned. A time schedule contains one or more time periods.

B. A partition can have one time schedule assigned. A time schedule contains one or more time periods.

C. A partition can have multiple time schedules assigned. A time schedule contains only one time period.

D. A partition can have one time schedule assigned. A time schedule contains only one time period.

Answer: A

Explanation:

Question: 54

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

Explanation:

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

Question: 55

A user reports that when they attempt to log out from the Cisco Extension Mobility service by pressing the Services button, they cannot log out. What is the most likely cause of this issue?

A. The Cisco Extension Mobility service has not been configured on the phone.

B. There might be a significant delay between the button being pressed and the Cisco Extension Mobility service recognizing it. It would be best to check network latency.

C. The user device profile has not been assigned to the user.

D. The user device profile is not subscribed to the Cisco Extension Mobility service.

Answer: D

Explanation:

Question: 56

What is a component of Cisco Unified Mobility?

A. Unified IVR

B. Mobile Connect

C. Smart Client Support

D. Single Number Connect

Answer: B

Explanation:

Question: 57

When the services key is pressed Cisco Extension Mobility does not show up. What is the cause of the issue?

A. The URL configured for Cisco Extension Mobility is not correct.

B. Cisco Extension Mobility Service is not running.

C. The phone is not subscribed to Cisco Extension Mobility Service.

D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

Answer: C

Explanation:

Question: 58

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

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

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

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

D. Assign the extension as a mobile extension.

Answer: B

Explanation:

Question: 59

What are the elements for Device Mobility configuration?

A. physical location, device pool, and Device Mobility group

B. device pool, Device Mobility group, and region

C. physical location. Device Mobility group, and region

D. device pool, Device Mobility group, and Cisco IP phone

Answer: A

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

Question: 60

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: C

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm- feature- configuration-guide_chapter_011101.html#CUCM_TK_A337E035_00

Question: 61

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 resources priority mode passthrough

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

C. voice-class sip early-offer forced.

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

Answer: D

Explanation:

Question: 62

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which two debugs must the administrator turn on? (Choose two).

A. debug H.245 asn1

B. debug H.323 message

C. debug H.225 asn1

D. debug H.225 media

E. debug H.323 asn1

Answer: AC

Explanation:

Question: 63

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 midcall-signaling block

D. voice-class sip conn-reuse

Answer: A

Explanation:

Question: 64

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor Incoming command configured? (Choose two.)

A. "ephone-dn' configuration mode

B. "voice register global" configuration mode

C. "telephony-service" configuration mode

D. "voice register pool" configuration mode

E. "dial-peer cor custom" configuration mode

Answer: AD

Explanation:

Question: 65

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. SIP route pattern

C. intercluster trunk

D. calling search space and partition

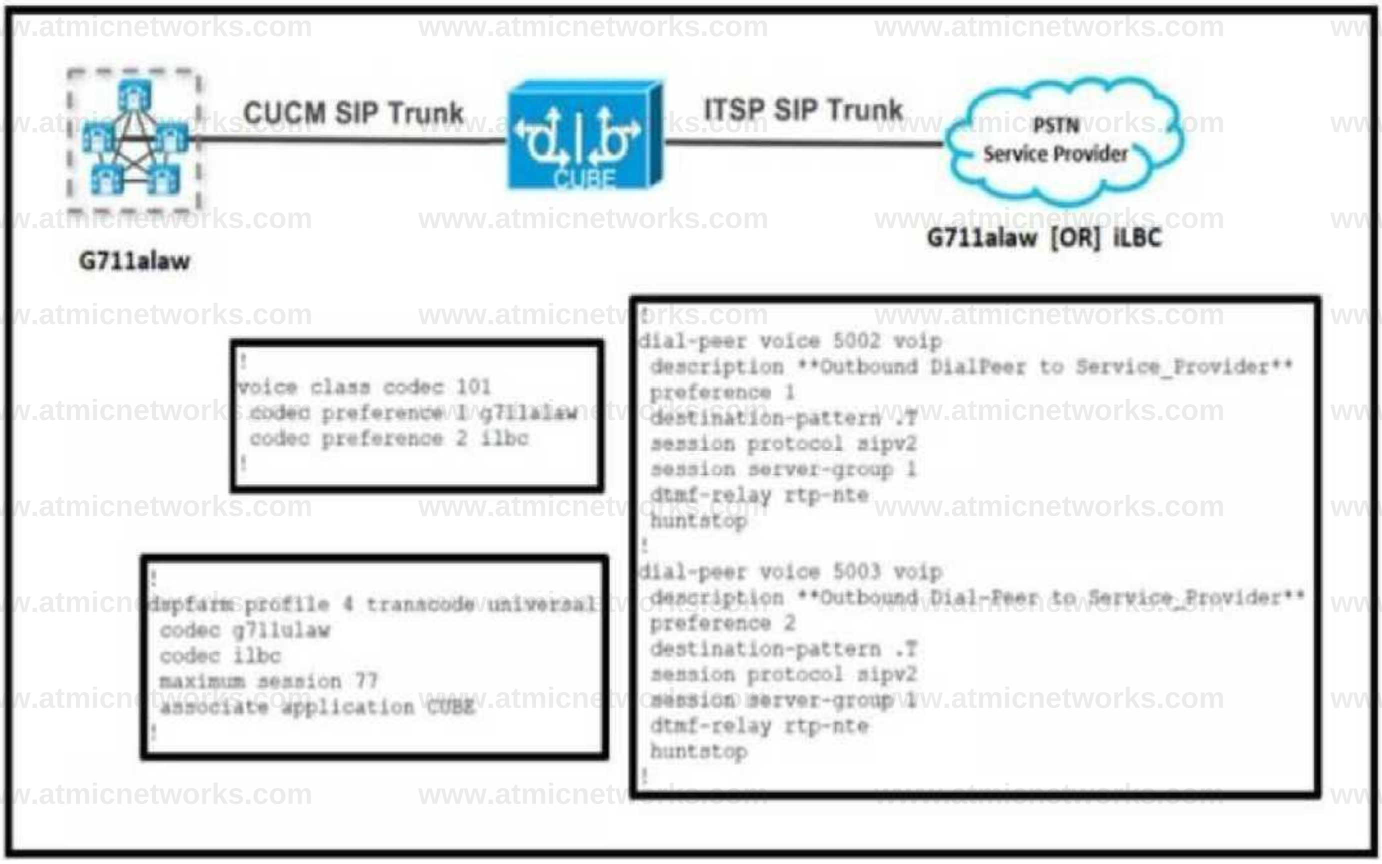
E. SIP trunk

Answer: BE

Explanation:

Question: 66

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 volp codec g711alaw ilbc f
dial-peer voice 5003 volp
codec g711alawllbc

dial-peer voice 5002 volp voice-class codec 101 offer-all
I
dial-peer voice 5003 volp voice-class codec 101 offer-all

dial-peer voice 5002 volp codec g711alaw
1
dial-peer voice 5003 volp codec llbc

dial-peer voice 5002 volp voice-class codec 101 t
dial-peer voice 5003 voip
voice-class codec 101

A. Option A

B. Option B

C. Option C

D. Option D

Answer: D

Explanation:

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

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

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

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

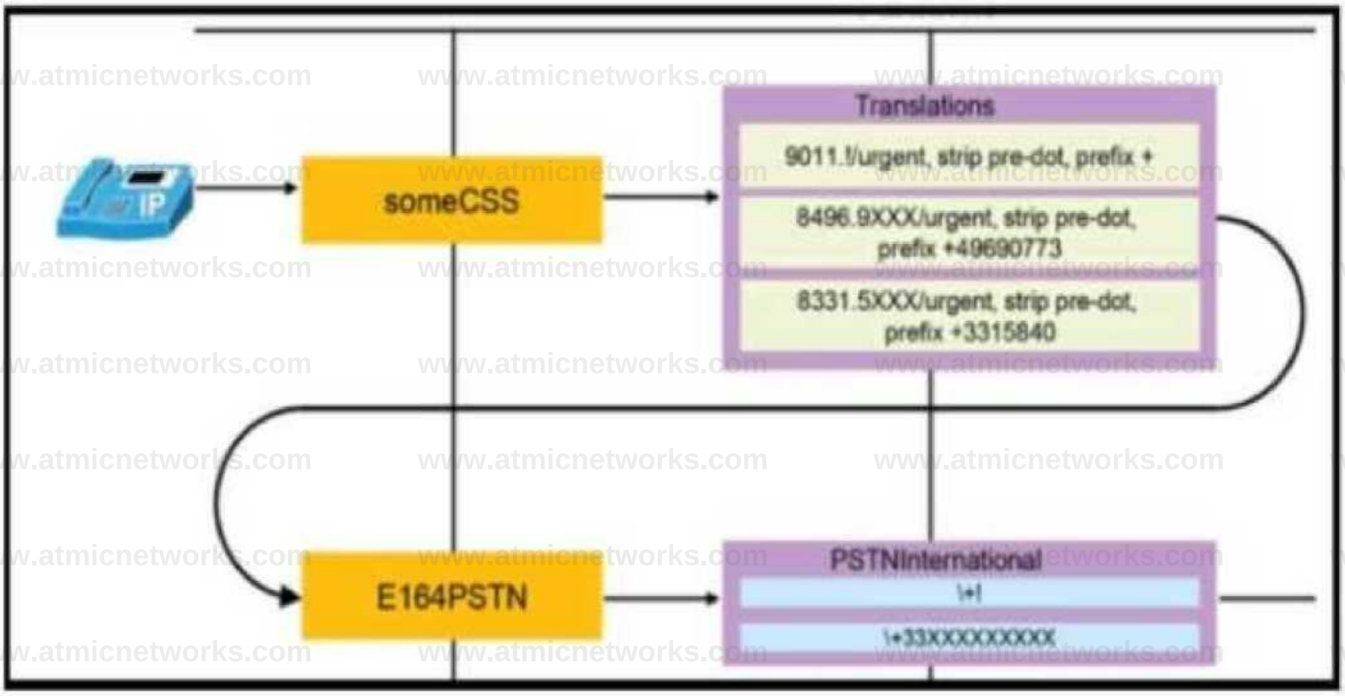
E. Add the long-distance & international pattern's partitions to the calling search space of the device profile

Answer: AE

Explanation:

Question: 68

Refer to the exhibit.



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

A. Allow Device Override on the route pattern

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

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

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

Answer: C

Explanation:

Question: 69

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

A. Set correct Location under Current Device Mobility Settings

B. Enable SRST under Current Device Mobility Settings

C. Set the correct subnet under Device Mobility Info.

D. Set Device CSS under Current Device Mobility Settings.

Answer: C

Explanation:

Question: 70

A customer is using a SIP trunk to route calls to ITSP to decrease the possibility of downtime, the customer invested in a failover device How does the customer ensure reachability to ITSP, so that if one device on ITSP fails, the calls will be routed to another device?

A. Enable transmit security status on the SIP security profile

B. Enable ANAT on the SIP profile.

C. Monitor the link using network management toots, and if it fails, manually change the routing to another working device.

D. Enable SIP Option Ping on the SIP profile.

Answer: D

Explanation:

Question: 71

Calls to a particular extension are not routing to voicemail. The user reaches the voicemail system from the handset by pressing the Messages button Which configuration parameter causes this problem?

A. The voicemail pilot number for call forwarding is missing from the ephone-dn

B. The voicemail pilot number is missing from the call handling on Cisco Unity Express

C. The voicemail pilot number is missing from the telephony service configuration on Cisco UCME

D. The voicemail pilot number for call forwarding is missing from the ephone

Answer: A

Explanation:

Question: 72

Refer to the exhibit.

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.168.100.100:5060
From: <sip:+123456789@192.168.100.100>;
To: <sip:987654321@192.168.100.200>
Date: Fri, 28 Jun 2019 08:30:32 GMT
Call-ID: fce8c980-d151d028-19cf3-325900a@192.168.100.100
CSeq: 101 INVITE
Require: 100rel
RSeq: 101
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Contact: <sip:987654321@192.168.100.200:5060>
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 247

v=0
o=CiscoSystemsSIP-GW-UserAgent 4780 5245 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 16384 RTP/AVP 8 101
c=IN IP4 192.168.100.200
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

While troubleshooting call failures on the Cisco Unified Border Element, an administrator notices that messages are being sent to the service provider, but there is no response The administrator later learns that this SIP provider does not support PRACK. Which header should be removed from the SIP message to resolve this issue?

A. Content-Disposition: session:handling=required

B. Require 100rel

C. Contact <sip:9876S4321@l92.168.100.200:5060>

D. Content-Type: application/sdp

Answer: B

Explanation:

Question: 73

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

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 A-law

B. G.711 U-law

C. G.729r8

D. G.729br8

Answer: C

Explanation:

Question: 74

An IP Telephony administrator is deploying IP phones The administrator has an existing Cisco UCME router with several SCCP & SIP phones registered. The administrator receives a request for a new SIP phone with MAC address 1111 2222.3333 and directory number 2050 to be added in the Cisco UCME. Which two configurations should be added in CME to support this request? (Choose two)

voice register pool 1 id mac 1111.2222.3333 type 8941 number 2 dn 1

ephone-dn 2 number 2050

voice register pool 1 id mac 1111.2222.3333 type 8941 number 1 dn 2

voice register dn 2 number 2050

ephone 1 mac-address 1111.2222.3333 type 8941
button 1:2

A. Option A

B. Option B

C. Option C

D. Option D

E. Option E

Answer: CD

Explanation:

Question: 75

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 via

B. dial-peer voice 1 voip

incoming url request

C. dial-peer voice 1 voip

incoming called-number

D. dial-peer voice 1 voip

incoming url to

Answer: A

Explanation:

Question: 76

An engineer must configure call queuing under a Hunt Pilot. After the engineer receives the audio file that will be played to callers during queuing, which two steps should be taken to complete the configuration? (Choose two.)

A. Assign the uploaded audio file to the hunting Line Group member's "User Hold MOH Audio Source

B. Assign the uploaded audio file to the hunting Line Group member's "Network Hold MOH Audio Source".

C. Upload the audio file in "TFTP File Management" via OS Administration GUI

D. Assign the uploaded audio file to "Network Hold MOH Source & Announcements" under Hunt Pilot's Queuing section.

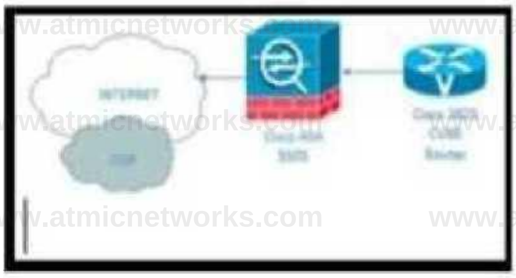
E. Upload the audio file in "MOH Audio File Management" via CM Administration GUI

Answer: CD

Explanation:

Question: 77

Refer to the exhibit.



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

A. use of port 5061 for SIP secure

B. access list allowing the firewall IP

C. ip prefix-list to filter the unwanted IP address

D. bind signaling and media to the loopback interface

Answer: D

Explanation:

Question: 78

Refer to the exhibit.

```
voice hunt-group 1 I
  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. call-blast

B. parallel

C. sequential

D. peer

Answer: D

Explanation:

Question: 79

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.224 asn1

C. debug H.246 asn 1

D. debug H.225 media

E. debug H.323 asn 1

Answer: B

Explanation:

Question: 80

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. The Client Matter Code feature provides the option to configure Authorization Level susch as in the Forced Authorization Code.

D. Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.

Answer: B, C

Explanation:

Question: 81

An administrator sees the voice register pool 1 command in a Cisco UCME configuration. Which configuration is occurring in this section?

A. configuration for a pool of SIP phones (similar to device pool on Cisco UCM)

B. configuration for a single SIP phone

C. configuration for SIP registrar service

D. configuration items common for all SIP phones

Answer: B

Explanation:

Question: 82

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 password is incorrect.

B. Cluster IDs are not unique.

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

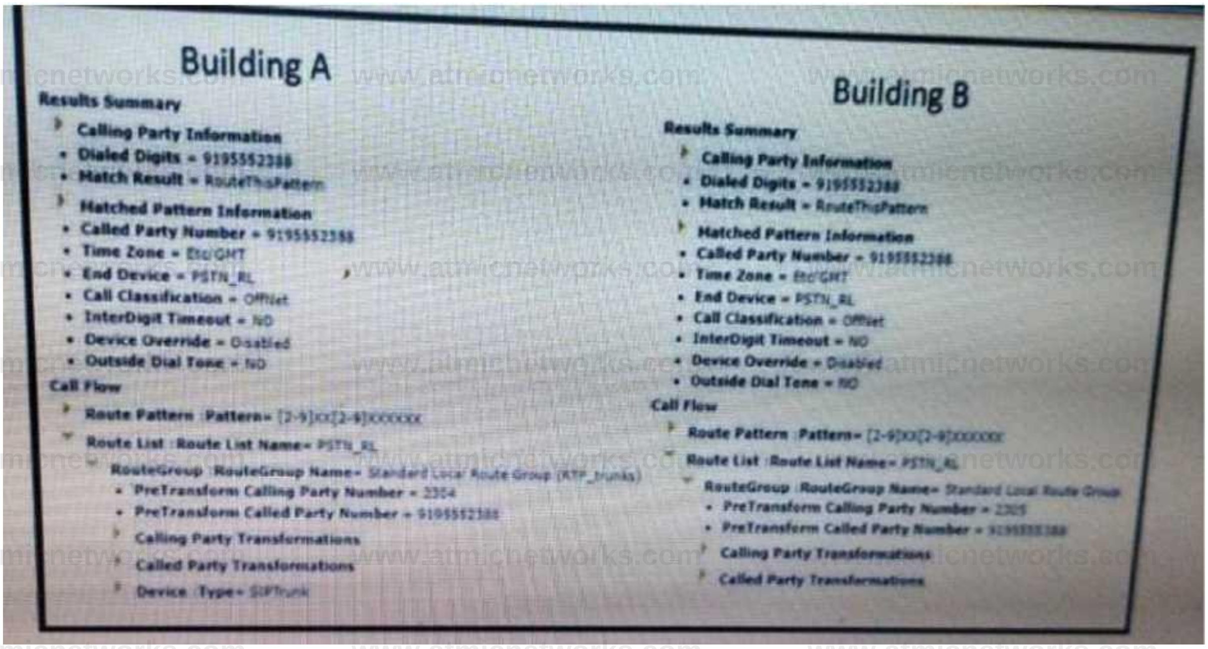
D. The engineer needs to exchange the CallManager certificate.

Answer: C

Explanation:

Question: 83

Refer to the exhibit.



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 is own device pool. The DNA tool is used to test the configuration. How is this issue resolved?

A. Change the partition of the route pattern

B. Add a sip trunk inside route group Standard Local Route Group.

C. Modify the route pattern to add a prefix of 91

D. Add a local route group on the device pool configuration.

Answer: D

Explanation:

Question: 84

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

A. Top Down, Round Robin, Broadcast

B. Top Down, Circular, Broadcast

C. Top Down, Round Robin, Distribute

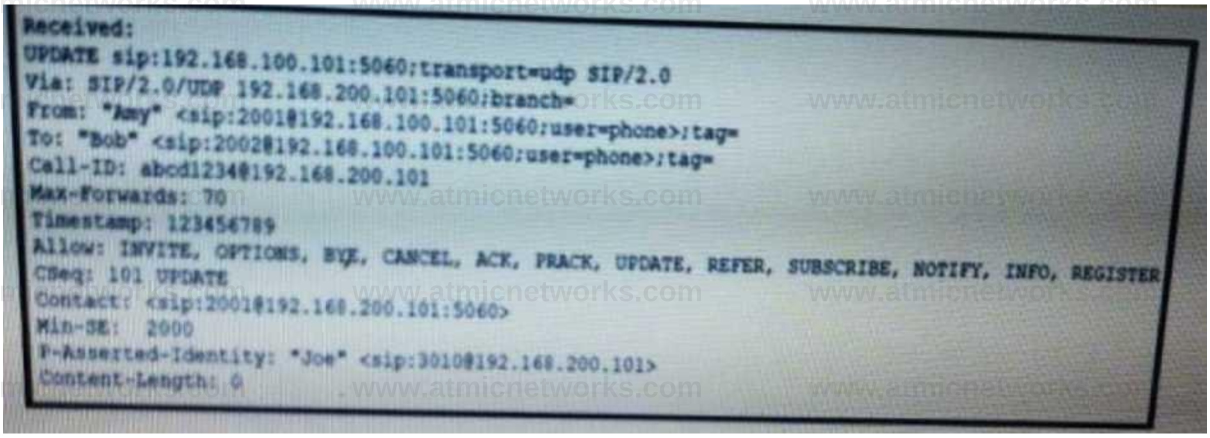
D. Sequantial, Circular, Broadcast

Answer: B

Explanation:

Question: 85

Refer to the exhibit.



```
Received:
UPDATE sip:192.168.100.101:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.200.101:5060;branch=
From: "Amy" <sip:2001@192.168.100.101:5060;user=phone>;tag=
To: "Bob" <sip:2002@192.168.100.101:5060;user=phone>;tag=
Call-ID: abcd1234@192.168.200.101
Max-Forwards: 70
Timestamp: 123456789
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 UPDATE
Contact: <sip:2001@192.168.200.101:5060>
Min-SE: 2000
P-Asserted-Identity: "Joe" <sip:3010@192.168.200.101>
Content-Length: 0
```

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. no update-callerId

C. no contact-passig

D. rel1xx require “100rel”

Answer: D

Explanation:

Question: 86

Signal number reach call phone that not answered are leaving voicemails on the cell phone rather the corporate mailbox. Which two options will resolve this issue? (Choose two.)

A. Check the Enable Extend and Connect checkbox

B. Check the Enable Unified Mobility features checkbox

C. Decrease the T302 timer

D. Decrease the T301 timer Decrease the Answer Too Late timer

Answer: B, D

Explanation:

Question: 87

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 rule to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

B. Set the Significant Digits to 4 on the SIP trunk.

C. 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.

D. Set the Significant Digits to 8 on the SIP trunk.

Answer: B

Explanation:

Question: 88

Refer to the exhibit.

```
SXFSea<lar/eMXO*O/ecOXO*0/vait_SirTuMr i TiMib|ir*d type*4IF_TXMES_MJUT_CCWKCT nlitKMO retxiee^ luck/TraMttrt^^awMIH/viiTraMHnhixXmtrMlXHi Fortum Xittnal Mop type*;
Staet/Treaapert/MO/eisTraaepexfaa^^aMCoaaeatreai hitiM TO Maa aX#M Cat aaar*10.10.1.11, part*1040, oeaaio^JO
Kack/TrMapart/taO/aipDtlataCoaaXBataBMi Delete# aeaaMM^aaOeoO* Meai4»10, eoarvio.10.1.11* part«lMO, tiaaapart^TO
•tack/lnfo/0*0/ece&p aroeee# aipapi peeae event: ccsip apt pet ■•• -IT* retaxaeOt 3 ill* MtTWCOl MM), tar avert 44 (S1PSSX tv iMTISim MM)
•taea/trrar/saOaoeeeeMO/eipTreaaportPeetSoMfoiXurei feet tap •*** tail are nep vita tconaixi raaaaaea
Maca/tata/aaO/cca&f^precea^aipMi.WM.maC! eceip_api_pe<_»M_tm retinal! 3 (StPjmKOKjtM), far evert it iSIPSPi rv ttac FMLUMjGQ) PteeVTafe/OeOaeeacOHO/reetp epi tracer# event: See# trrer far evert (CaeeaeMPO)
•tack/Crrer/Oad/ect^Kla^taM^aaf.faAlatti Seat trrer ta 10.10.1.11 <1940 far traarpart TO
Maet/Xafe/OaOaeeteOMO/oeeip^Mt^M^eaeee^fer^api^etti Ceteaoriaea eauaeiM« eat apery 1114
Xadt/Xafa/CaOaMOcOMO/eipTfXXaatxatadiacocaact) lunate call tiacaaeect (M) far autpotna call
3Xtlaa4Xer/aeBXa"3J4O*/tMX0"O/aeai*_api_aaXX_4iMeafteatMi ee#*>aes4iMwaaaM illif aa^earp^a&M.eauMiOOl)
```

Jiriaa41et/e«XevJ245»/ecalP*9/fkMOevxoeriD: iMtae to Sira Oy cefeli #sla
•caefe/ttatM/OBOweeaetMO/ekpirXOaaoe*****1 -aeetcieiO t State cMaape frea iSTATI ISU, BQUTaTt^OOHE) to i STAT1DISC0MKCTXB3, S'JSITITt KMi

An administrator has configured a SIP trunk between two Cisco UCM clusters. For calls that should use the trunk, the calls fail with a fast busy. The administrator checks the Cisco CallManager SDL traces and found that the cluster to which the calling device is registered never sends an INVITE to the destination cluster. The administrator also verifies that all nodes from both clusters are powered on, and the CallManager service is running. How is this issue resolved?

A. The administrator needs to disable OPTIONS pings on the SIP trunks for both clusters.

B. The administrator must allow connectivity so that TCP connections do not fail between the nodes.

C. The administrator needs to enable OPTIONS pings on the SIP trunks for both clusters.

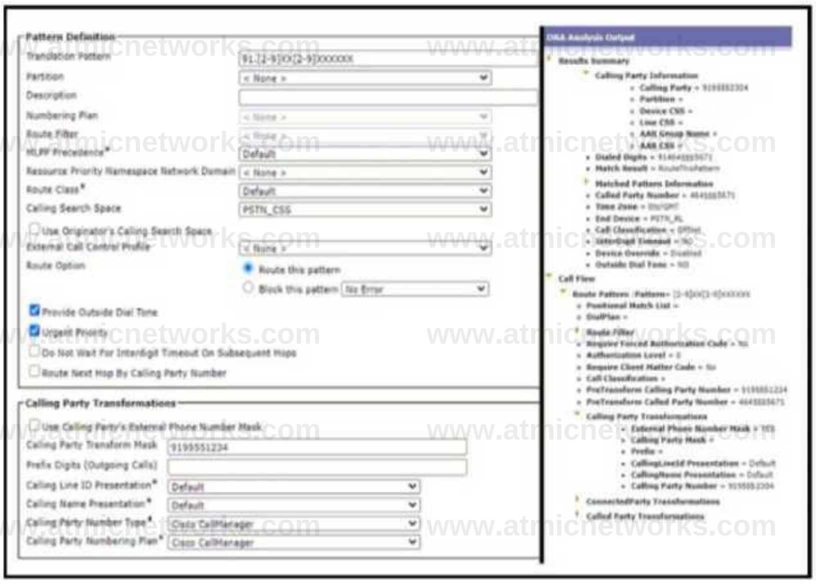
D. The administrator must associate the route pattern with a calling search space the device can dial.

Answer: B

Explanation:

Question: 89

Refer to the exhibit.



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. Change the partition of the translation pattern from none to pstn_pt.

B. Enable Force Authorization Code on the route pattern.

C. Disable Use Calling Party's External Phone Number Mask on the route pattern.

D. Enable Use Calling Party's External Phone Number Mask on the translation pattern.

Answer: C

Explanation:

Question: 90

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 h245-signal
B. dtmf-relay rtp-nte

C. dtmf-relay cisco-rtp

D. dtmf-relay sip-kpml

Answer: D

Explanation:

Question: 91

The SIP session refresh timer allows the RTP session to stay active during an active call. The Cisco UCM sends either SIP-INVITE or SIP-UPDATE messages in a regular interval of time throughout the active duration of the call. During a troubleshooting session, the engineer finds that the Cisco UCM is sending SIP-UPDATE as the SIP session refresher, and the engineer would like to use SIP-INVITE as the session refresher. What configuration should be made in the Cisco UCM to achieve this?

A. Enable SIP ReMXX Options on the SIP profile.

B. Enable Send send-receive SDP in mid-call INVITE on the SIP profile.

C. Change Session Refresh Method on the SIP profile to INVITE.

D. Increase Retry INVITE to 20 seconds on the SIP profile.

Answer: C

Explanation:

Question: 92

A user's phone is already configured for Single Number Reach, and the user wants a feature to move an active call from a mobile phone to a desk phone and vice-vers

a. As an administrator, which additional configuration should be made to fulfill the user's request?

A. Confirm that the desk phone is subscribed to Cisco Extension Mobility.

B. Check to make sure that the Resume softkey option appears on the desk phone.

C. Use Dialed Number Analyzer to determine if the user extension can dial the mobile phone.

D. Add the mobility key to the softkey template that the desk phone is using.

Answer: D

Explanation:

Question: 93

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

A. Cisco UCM does not write additional ILS learned objects to the database and will delete the first 50,000 entries learned to keep it to the service parameter value.

B. Cisco UCM does not write additional ILS learned objects to the database and will delete the last 50,000 entries learned to keep it to the service parameter value.

C. Cisco UCM does not write additional ILS learned objects to the database and keeps the existing database entries.

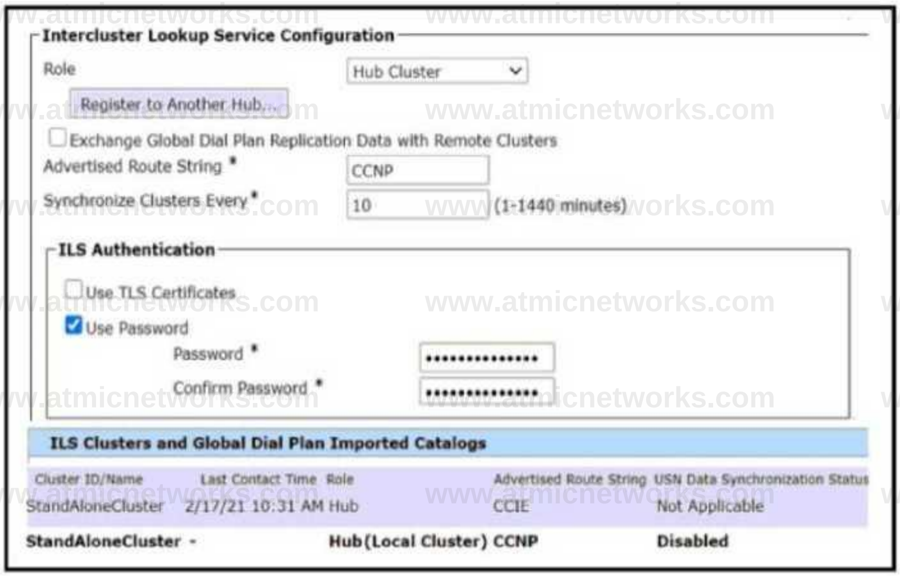
D. Cisco UCM will overwrite an entry for newly learned data and keep the parameter value at 100,000.

Answer: C

Explanation:

Question: 94

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. A password has never been set for ILS.

B. Use TLS Certificates must be selected.

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

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

Answer: D

Explanation:

Question: 95

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

A. A Remote Destination Profile having the same extension as the Mobile phone's number adds the physical phone's Directory Number to the RDP as a Remote Destination. Add the softkey "Mobility" to the physical phone's softkey template.

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

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

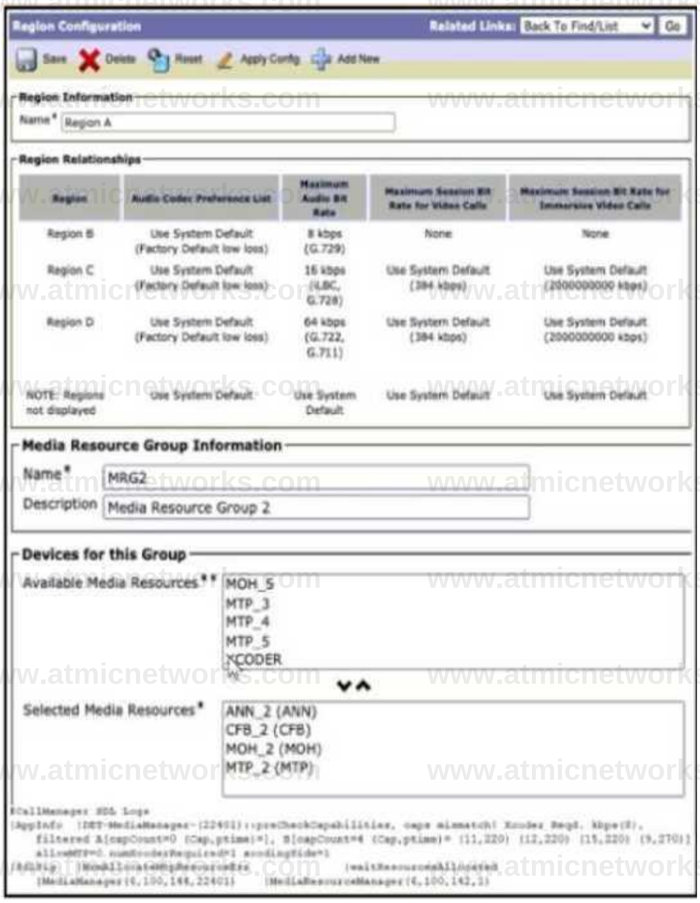
D. A Remote Destination Profile having the same extension as the physical phone's Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey "Join" to the physical phone's softkey template.

Answer: B

Explanation:

Question: 96

Refer to the exhibits.



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: B

Explanation:

Question: 97

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

A. gateway

B. out-of-band

C. channel

D. in-band

Answer: B

Explanation:

Question: 98

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 IP routing is correct between the gateway and the IP phone.

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

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: C

Explanation:

Question: 99

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

A. Configure device pools.

B. Configure service parameters

C. Configure enterprise softkeys.

D. Configure class of control.

Answer: B

Explanation:

Question: 100

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. sip-ua

disable-early-media 180

D. voice service voip sip no reMxx

Answer: B

Explanation:

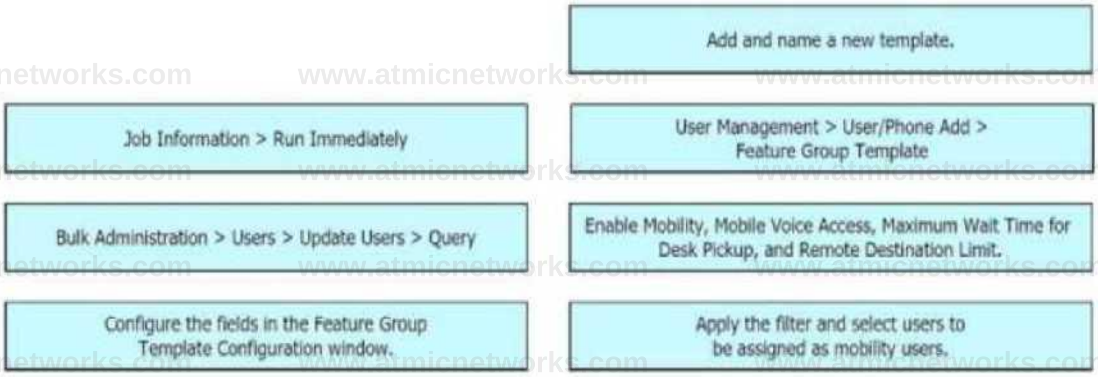
Question: 101

DRAG DROP

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.

Answer:

Explanation:



Question: 102

Users are reporting that several inter-site calls are failing, and the message "not enough bandwidth" is showing on the display. Voice traffic between locations goes through corporate WAN. and Call Admission Control is enabled to limit the number of calls between sites. How is the issue solved without increasing bandwidth utilization on the WAN links?

A. Disable Call Admission Control and let the calls use the amount of bandwidth they require.

B. Configure Call Queuing so that the user waits until there is bandwidth available

C. Configure AAR to reroute calls that are denied by Call Admission Control through the PSTN.

D. Reroute all calls through the PSTN and avoid using WAN.

Answer: C

Explanation:

Question: 103

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. top down

B. circular

C. broadcast

D. longest idle time

Answer: D

Explanation:

Question: 104

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 Single Number Reach.

B. Configure an access list.

C. Configure a mobility identity.

D. Configure Mobile Voice Access.

Answer: B

Explanation:

Question: 105

Refer to the exhibit.

```
interface GigabitEthernet0/0/0
 description to CUCM
 ip address 10.10.150.1 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description to ITSP
 ip address 192.168.10.78 255.255.255.0
 negotiation auto
!
dial-peer voice 100 voip
 incoming called-number 8005532447
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 200 voip
 destination-pattern 8005532447
 session target ipv4:192.168.10.100
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
!
dial-peer voice 300 voip
 answer-address 8005532447
 session protocol sipv2
 codec g711ulaw
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
```

```
Received:
INVITE sip:8005532447@10.10.150.1:5060 SIP/2.0
To: "CISCO SYSTEMS" <sip:8005532447@10.10.150.1>
Date: Tue, 30 Mar 2021 22:14:00 GMT
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
[..Omitted for Brevity..]
Contact: <sip:1001@10.10.150.11:5060>;
Content-Type: application/sdp
Content-Length: 235

s=SIP Call
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

An engineer is troubleshooting a call-establishment problem between Cisco Unified Border Element and Cisco UCM.

Which command set corrects the issue?

A. SIP binding in SIP configuration mode:

voice service voip sip

bind control source-interface GigabitEthernetO/0/0 bind media source-interface

GigabitEthernetO/0/0

B. SIP binding In SIP configuration mode:

voice service volp

sip

bind control source-Interface GlgabltEthernetO/0/1 bind media source-Interface

GlgabltEthernetO/0/1

C. SIP binding In dial-peer configuration mode:

dial-peer voice 300 voip

voice-class sip bind control source-interface GigabitEthernetO/0/1 voice-class sip bind media sourceinterface

GigabitEthernetO/0/1

D. SIP binding in dial-peer configuration mode:

dial-peer voice 100 volp

voice-class sip bind control source-interface GigabitEthernetO/0/0 voice-class sip bind media source-interface GigabitEthernetO/0/0

Answer: D

Explanation:

Question: 106

Refer to the exhibit.

```
CUBERoutertconf t
Enter configuration commands, one per line. End with CNTL/Z
CUBE_Router(config)Ivoice translation-rule 999
CUBE_Router(cfg-translation-rule)irule 1 /*9(.*)/ //
CUBERouter(cfg-translation-rule)lend
CUBE_Routeri
CUBE_Routeritest voice translation-rule 999 9123548 9123548 Didn't match with any of rules
```

Which change to the translation rule is needed to strip only the leading 9 from the digit string 9123548?

A. rule 1 /^9\(.*\)/A1/

B. rulel /.*\(3548S\)/^1/

C. rulel /^9\(\d*\)/^1/

D. rule 1/^9123548/^1/

Answer: A

Explanation:

Question: 107

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 route group for each site and one global route list for PSTN calls that point to the local route group.

B. Create a route group which has all the gateways and associate it to the device pool of every site.

C. Create one global route list for PSTN calls that points to one global PSTN route group.

D. Create a hunt group and assign it to each side route pattern

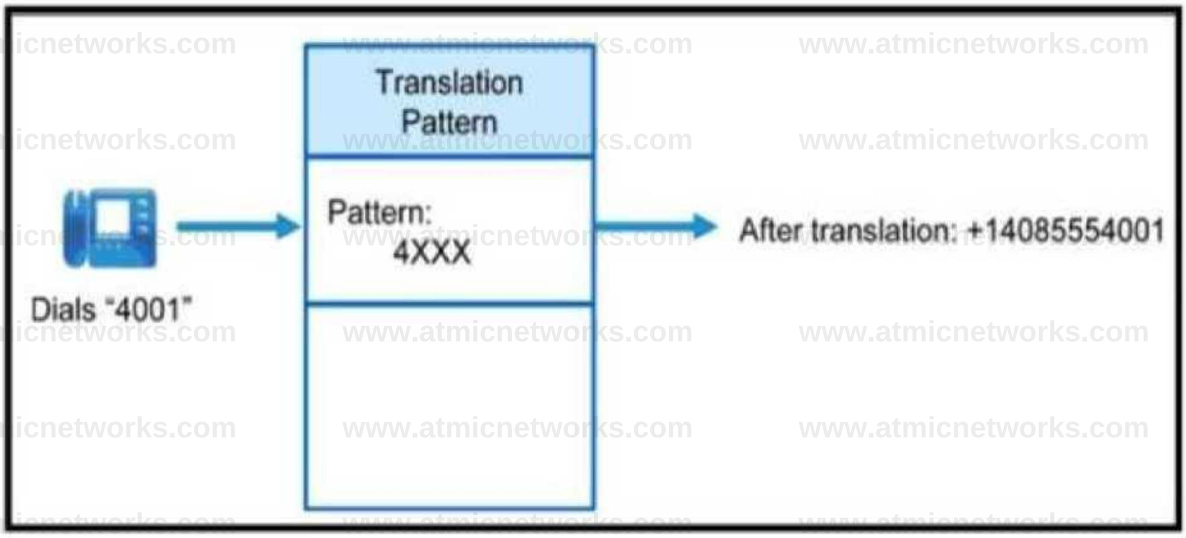
E. Assign one route group as a local route group in the device pool of the corresponding site.

Answer: A, E

Explanation:

Question: 108

Refer to the exhibit.



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

A. Called Party Transformation Mask of + 14085554XXX

B. Called Party Transformation Mask of ♦1408555[35)XXX

C. Calling Party Transformation Mask of +1408555XXXX

D. Calling Party Transformation Mask of +14085554XXX

Answer: A

Explanation:

Question: 109

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

A. WAN connectivity

B. LAN connectivity

C. route pattern

D. route list and route group

Answer: A

Explanation:

Question: 110

An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM. How should this be accomplished?

A. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.

B. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.

C. Localize the calling and called numbers to E. 164 format and globalize the called number in the gateway.

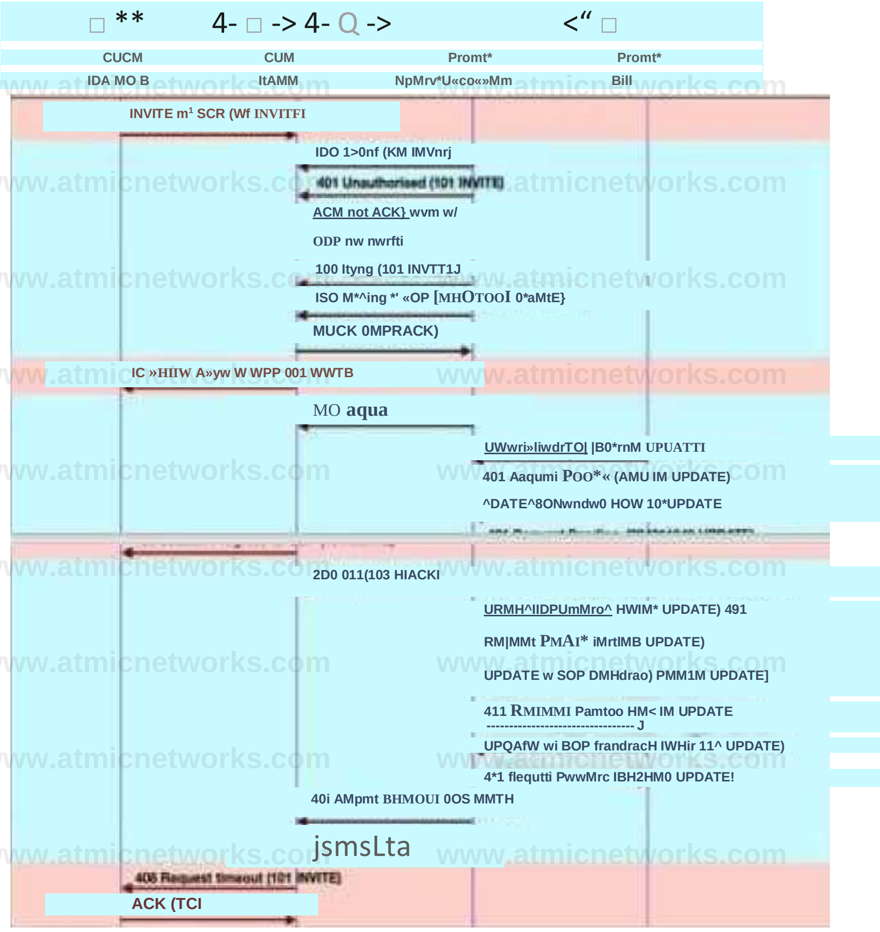
D. Globalize the calling and called numbers to E. 164 format and localize the called number in the gateway.

Answer: D

Explanation:

Question: 111

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 m Cisco UCM.

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

Answer: B

Explanation:

Question: 112

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 /^4...$/2404\0/ type any national plan any Isdn

B. voice translation-rule 1 rule 1 // // type any subscriber plan any isdn

C. voice translation-rule 1 rule 1 /^4...S/ /9132404 0/ type any subscriber plan any Isdn

D. voice translation-rule 1

rule 1 /^4...V /2404\0/ type any subscriber plan any isdn

Answer: D

Explanation:

Question: 113

Refer to the exhibit.

```
55697959.007 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222
crCdpnMask=33334444 crPrefixDigit= crDDI=2
55697959.008 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- after  DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 |12:20:50.913 |AppInfo  |RouteListCdrc::transformed cdpn (without unconsumpt
digits) = 33334444, unconsumed digit=
```

Which INVITE is sent to 10.10.100.123 as a result of this log?

A)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716-41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

B)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716-41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

C)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP
0.100.123 on port
195992 364,NET|
INVITE sip:33334444910.10.100.123:5060 SIP/2.0
Via: S1P/2.0/TCP 10.122.200.50:5060;branch-z9hG4bK268d6 Froa: -1000“ oip:1000910.122.200.50>;tag-32412716-4117 To: <sip:33334444910.10.100.123> Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100t2-265e57-67071d0a910.122.200.50
Supported: timer,resource—prior i t y,replaces
Min-SE; 1800
User-Agent: Cisco-CUCM12.0
```

D)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716-41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

A. Option A

B. Option B

C. Option C

D. Option D

Answer: C

Explanation:

Question: 114

Refer to the exhibit.

dial-peer voice 10 voip description inbound session protocol sipv2 incoming called-nuaber 2000 dust-relay rtp-nte no ved

dial-peer voice 20 voip description Outbound destination-pattern 2. session protocol sipv2 session target ipv<:192.168.100.101 voice-class sip options-keepalive dust-relay rtp-nte t

CUBEtshov dial-peer voice sussary dial-peer hunt 0
AD PRE PASS SESS-SER-GRP\ OUT
TAG TYPE MIN OPEN PREFIX DEST-PATTERN RR THRU SESS-TARGET STAT FORT KEEPALIVE VRF 10 voip up up 0 syst NA
20 voip up up 2. 0 syst ipvl:192.168.100.101 busyout NA

A call mode through the Cisco Unified Border Element to pilot 2000 is foiling. What is causing the call to foil?

A. No codecs are configured on the dial peers

B. The Cisco Unified Border Element is not receiving a response to its OPTION keepahves.

C. The destination pattern is incorrect for the dialed number.

D. VAD was not disabled on the outgoing dial poor.

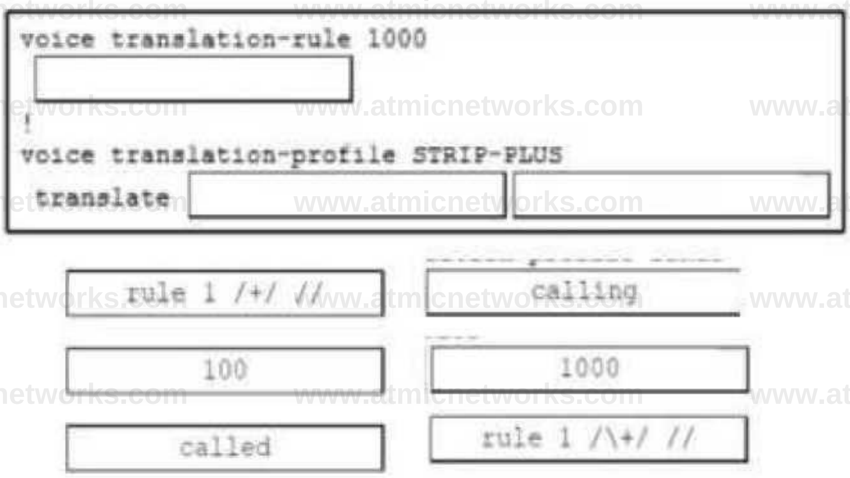
Answer: C

Explanation:

Question: 115

DRAG DROP

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:

Explanation:

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

Question: 116

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

A. midcall-signallng passthru

B. midcall-signaling preserve-codec

C. no update-callerid

D. midcall-signaling passthru media-change

Answer: D

Explanation:

Question: 117

An engineer is troubleshooting local ringback on a Cisco SIP gateway The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with

SDP to play ringback from the IP address specified m the SDP Which configuration change must be made on the gateway to resolve the issue?

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

B. Router(conftg-sip-ua)# disable-early-media 180

C. Router(con(-voi-serv)# no disable-early-media 180

D. Router(config-sip-ua)# no disable-early-media 180

Answer: B

Explanation:

Question: 118

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

A. voice service volp

Ip address trusted list

B. voice service volp

enablo ip address trust authentication

C. voice service volp

enable Ip address trust list

D. voice service volp

ip address trusted authenticate

Answer: D

Explanation:

Question: 119

A company has users that are logged in to hunt groups. However, there is a requirement for hunt group configurations to provide an option to turn on audible ringtones when calls to a line group arrive at a phone that is logged out and on a break. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call does not ring at the phone of that line group member because of the logged-out status. Which action meets this requirement?

A. Configure the HLog softkey on the phone so that while a user is logged off, it plays an audible tone when a call is missed.

B. Set the service parameter Party Entrance Tone to True."

C. Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.

D. Set the service parameter Enterprise Feature Access number for hunt group logout and set up an access number

Answer: C

Explanation:

Question: 120

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

A. Subscribe IP phones to the Extension Mobility Phone Service.

B. Consolidate Cisco UCM certificates by using Bulk Certificate Management.

C. Subscribe directory numbers to the Extension Mobility Phone Service.

D. Subscribe the device profile to the Extension Mobility Phone Service.

E. Define the default Device Template on IP phones when users log out.

Answer: D, E

Explanation:

Question: 121

An administrator troubleshoots call failure in a new deployment and finds that the SIP INVITE messages sent to the service provider contain a diversion header with the user's 4-digit directory number. These 4-digit directory numbers range from 1000 to 9999. The service provider is rejecting the calls because it requires that the diversion header contain 10 digits. Which command on the Cisco Unified Border Element resolves this issue for all users?

A)

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

B)

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

C)

```
voice class sip-profiles 105
request INVITE sip-header Diversion modify “<sip:1(...)@" “<sip:2634H\2@”
```

D)

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

A. Option A

B. Option B

C. Option C

D. Option D

Answer: D

Explanation:

Question: 122

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

A. Configure mode srst under telephony-service configuration mode.

B. Configure mode esrst under telephony-service configuration mode.

C. Configure mode esrst under voice register global configuration mode.

D. Configure mode esrst under voice service voip configuration mode.

Answer: C

Explanation:

Question: 123

Refer to the exhibit.

Cisco Unified Communications Manager Dialed Number Analyzer Results

Expand All | Collapse All

Results Summary

Calling Party Information

- **Dialed Digits** - 730
- **Match Result** = BlockThisPattem
- **Route Block Cause** ■ Unallocated Number
- **Called Party Number** •

Matched Pattern Information

- **Pattern Type** =
- **Time Zone** • Etc/GMT
- **Outside Dial Tone** = NO

Call Flow

Translation Pattern Pattern = 7XX o **Partition** = Abbreviated o **Positional Match List** -
o **Calling Party Number** = 45731 o **Pre transform Calling Party Number** = 45731 o **PreTransform Called Party Number** - 730

Calling Party Transformations ' Connected Party Transformations '
Called Party Transformations • **Called Party Mask** = • **Discard Digits Instruction** ■ None • **Prefix** * 45

• **Called Number** = 45730

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

A. The abbreviated partition is missing from the calling search space.

B. The translation pattern is blocking the call.

C. The translation pattern Is misconfigured.

D. Directory number 45730 is unknown to Cisco UCM.

Answer: A

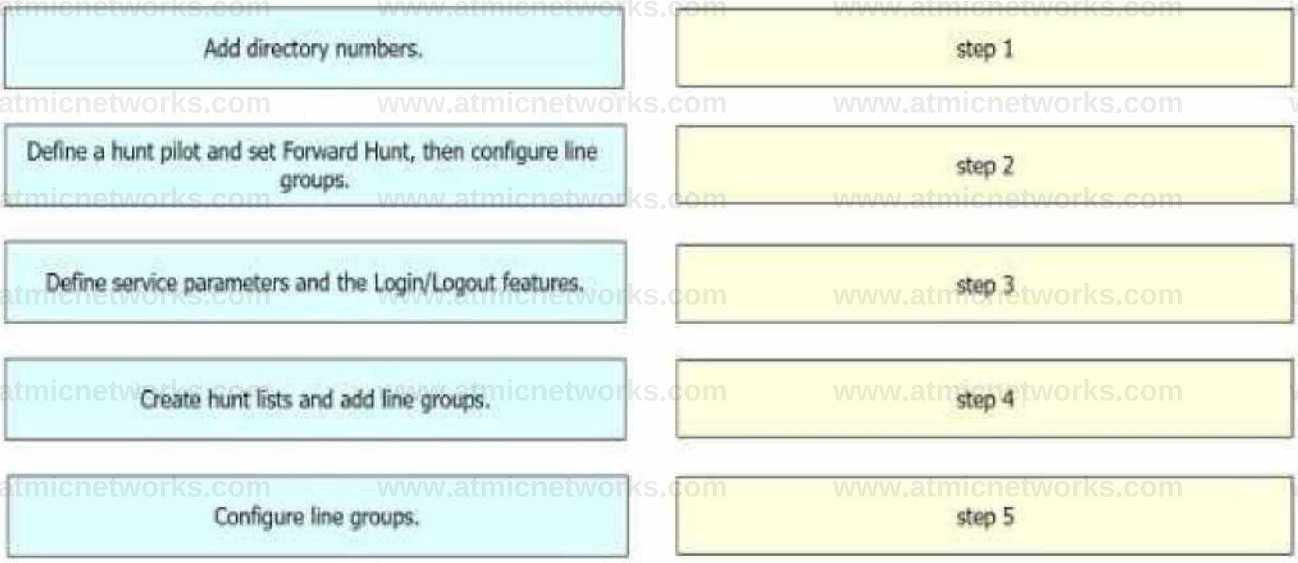
Explanation:

Question: 124

DRAG DROP

A customer wants to configure their team of 10 agents to answer incoming calls to a common central number. The phone for each team member must ring in sequence and ring four times before moving to the next member's phone.

Drag and drop the steps from the left into the order on the right to make this happen.



Answer:

Explanation:

Add directory numbers.

Configure line groups.

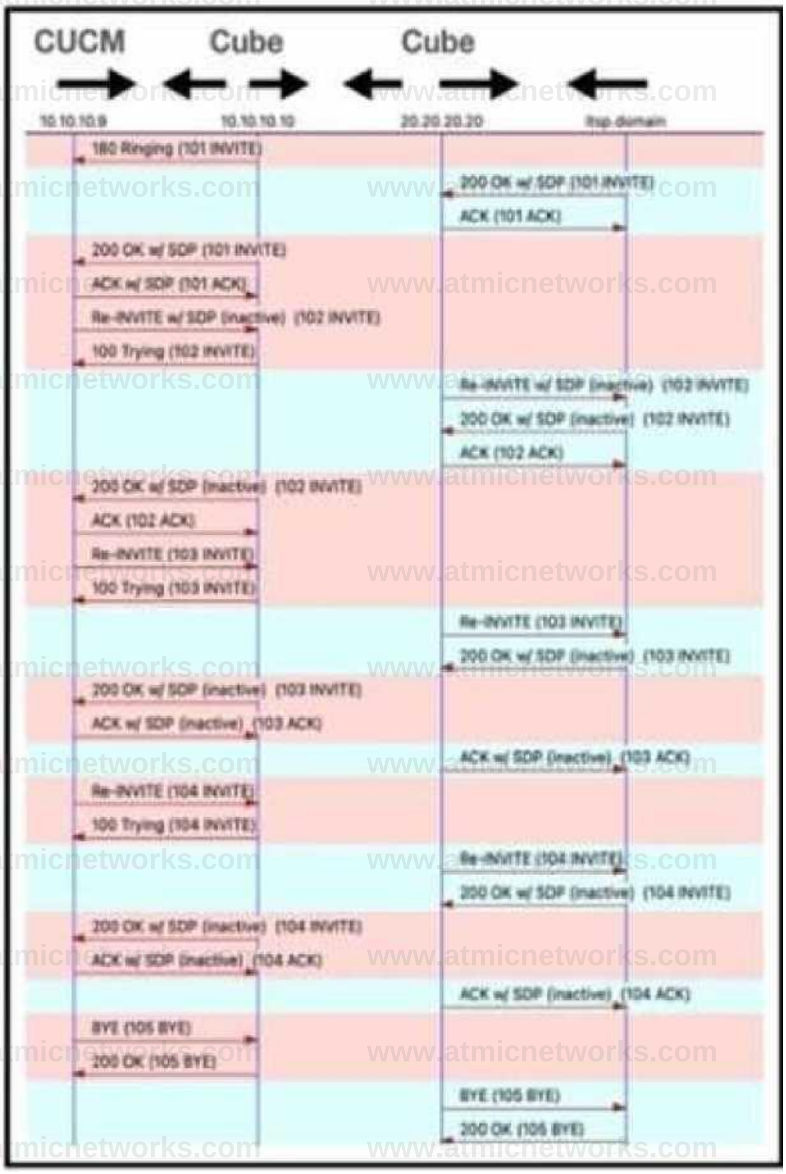
Create hunt lists and add line groups.

Define a hunt pilot and set Forward Hunt, then configure line groups.

Define service parameters and the Login/Logout features.

Question: 125

Refer to the exhibit.



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

A. Assign an MTP to the SIP trunk.

B. Restart the SIP trunk on Cisco UCM.

C. Upgrade the Cisco Unified Border Element to 15.2.

D. Upgrade the Cisco Unified Border Element device.

Answer: C

Explanation:

Question: 126

Refer to the exhibit.

Route Patterns *(1 - 2 of 2)*

Find Route Patterns where Pattern | begins with | \

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

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

A. Change the Cisco UCM routing logic to the longest-match algorithm.

B. Select Allow Overlap Sending for the \+! route pattern.

C. Mark the \+ [2-9JXX[2-91XXXXXX route pattern with urgent priority.

D. Set the interdigit timeout service parameter value to 0.

Answer: C

Explanation:

Question: 127

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

A. Set the "Retain Media on Disconnect with PI for Active Call" service parameter to False.

B. Set the "Retain Media on Disconnect with PI for Active Cal" service parameter to True

C. Configure a handoff number.

D. Add Other Dual-Mode Device.

Answer: A

Explanation:

Question: 128

Refer to the exhibit.

```
SIP/2.0183 Session Progress
Via: SIP/2.0/TCP 10.5.5.5:5060;branch=z6a3clK21d5aca213alb
From: <sip:1000@10.5.5.5>;tag=1103Weddb418-c7b4 dce4 Ob8b 9Olbaldb345c-13319871
To: <sip:2000@10.10.10.10>;tag=4C2F0432-8CIB
Date: Fri, 09 Apr 202101:41:33 GMT
Call-ID: c7011a00-d51ff310-31978-b42ab014@10.5.5.5
CSeq: 101 INVITE
RSeq:1541
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Contact: <sip:2000@ 10.10.10.10:5060;transport=tcp>
Supported: sdpanat
Supported: X-cisco-srtp-fallback
Server: Cisco-SIPGateway/IOS-15.7.3.M8
Content-Length: 330

v=0
o=CiscoSystemsSIP-GW-UserAgent 9841 5325 IN IP4 10.10.10.10 s=SIP Call
c=IN IP4 10.10.10.10
t=00
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:1010-15
a=rtpmap:19 CN/8000
```

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

for generating ringback in these problematic scenarios. Which action resolves the issue?

A. Enable early media on the appropriate SIP trunk on Cisco UCM.

B. Edit the configuration for SIP ReMXX Options in the appropriate SIP profile on Cisco UCM to Send PRACK if 1xx Contains SDP.

C. On Cisco Unified Border Element: voice service voip sip

Re1xx require 100rel

D. On Cisco Unified Border Element: voice service voip sip

Re1xx disable 100rel

Answer: B

Explanation:

Question: 129

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

A. Add a new standard local route group. Select a route group as the new standard local route group in Device Pools. Create a route pattern of 911 that points to a route list that contains the new standard local route group.

B. Create a new local route group name. In Device Pools, select a route group for that new local route group. Create a route pattern of 911 that points to a route list that contains the new local route group.

C. Set a new local route group name. Select a route group as the new local route group in Device Pools. Create a route pattern of 911 that points to the new local route group.

D. Add a new standard local route group. Select a route group as the new standard local route group in Device Pools. Create a route pattern of 911 that points to the new standard local route group.

Answer: A

Explanation:

Question: 130

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

A. It requires SIP trunk and CTI with recording application.

B. Cisco UCM integrates with recording destination via SIP trunk.

C. It uses BIB of IP phone to fork audio to the recording destination.

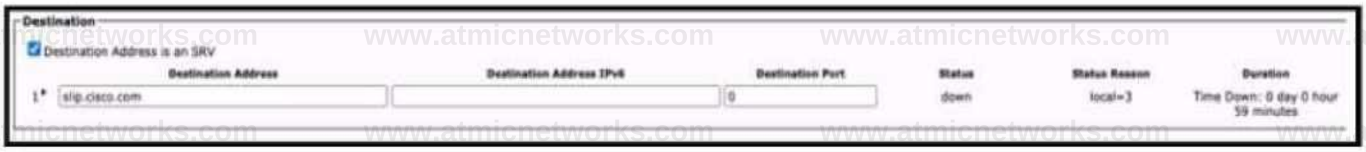
D. It initiates when the application (recorder) dictates that it must be initiated.

Answer: B

Explanation:

Question: 131

Refer to the exhibit.



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

A. Ensure that Cisco UCM can resolve the destination address via DNS.

B. Start the Cisco Call Manager service at the destination.

C. Ensure that the SIP Trunk Security Profile is configured to use UDP for transport.

D. Verify that the correct port numbers are used for the SIP trunk.

Answer: A

Explanation:

Question: 132

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

A. Configure two CSSs. then add a translation pattern with the secondary CSS to send calls with 9713 to the secondary gateway.

B. Configure two partitions and two CSSs, then add a translation pattern with the secondary CSS to send all calls with 9713 to the secondary gateway.

C. Create a Standard Route Group to dynamically route calls with prefix 9713 to the secondary gateway.

D. Configure a Route Group and a Route List to send calls with prefix 9713 to the secondary gateway.

Answer: A

Explanation:

Question: 133

Exhibit.

46282041.005 109:18:16.331 |Applnfo |DET-RegionsServer::matchCapabilities~ savedOption=3, PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=0, 8=0) kbps=8, capACount=l, capBCount=7

46282041.006 109:18:16.331 |Applnfo |DET-MediaManager-(1698821)::checkAudioPassThru, param(bPostMTPAIIocation=0,chkTrp=l), capCount(l,7), mtpPT=l, aPT=2

46282041.007 109:18:16.331 |Applnfo |DET-MediaManager-(1698821)::preCheckCapabilities, regionl=RTP_Reg, region2=SJ_Reg, Ptyl capCount=l (Cap,ptime)= (4,20), Pty2 capCount=7 (Cap,ptime)= (4,20) (2,20) (6,20) (11,20) (12,20) (15,20) (16,20)

46282041.008 109:18:16.331 |Applnfo |DET-RegionsServer::matchCapabilities--savedOption=0, PREF_NONE, regionA=(null) regionB=(null) latentCaps(A=O, 8=0) kbps=8, capACount=l, capBCount=7

46282041.009 109:18:16.331 | Applnfo | RegionsServer: applyCodecFilterlfNee^ed - no codecs remained after filtering so restored original 0 caps

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

A. Disable G.722 on all devices at both sites.

B. Enable Early Offer on the SIP trunk.

C. Increase the bandwidth allowance between the RTP_Reg and SJ_Reg regions to 64 kbps.

D. Create a new audio codec preference list with G.711 U-law 64k as the highest priority and apply it to RTP_Reg and SJ_Reg.

Answer: C

Explanation:

Question: 134

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

A. Change the "Maximum Wait Time for Mobility Pickup" setting from 4 to 6.

B. Change the "Maximum Wait Time for Desk Pickup" setting from 4 to 6.

C. Change the "Maximum Wait Time for Mobility Pickup" setting from 4000 to 6000.

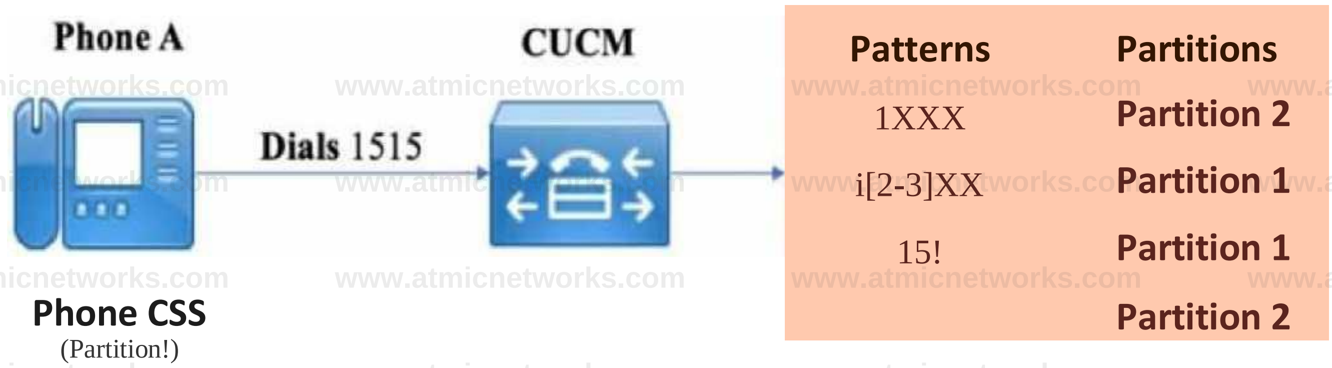
D. Change the "Maximum Wait Time for Desk Pickup" setting from 4000 to 6000.

Answer: D

Explanation:

Question: 135

Refer to the exhibit.



The Cisco UCM is configured with four route patterns. When phone A dials 1515, which pattern does Cisco UCM choose as the correct match?

A. 1XXX

B. 15

C. !

D. 1[2-3]XX

Answer: B

Explanation:

Question: 136

An engineer needs to configure a translation pattern in Cisco UCM to match dialed numbers that start with 9011 so that the calls can be routed with a single route pattern of VH If the engineer uses translation pattern 9011.! which two called party transformation settings must be configured in Cisco UCM? (Choose two.)

A. set Prefix Digits (Outgoing Calls) to +

B. set Called Party Transform Mask to +!

C. set Discard Digits to Predot

D. set Called Party Numbering Plan to International

E. set Called Party Number Type to International

Answer: B, C

Explanation:

Question: 137

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

A)

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

B)

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

C)

voice service voip
default mode secure
address hiding
allow-connections sip to sip sip registrar

D)

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

A. Option A

B. Option B

C. Option C

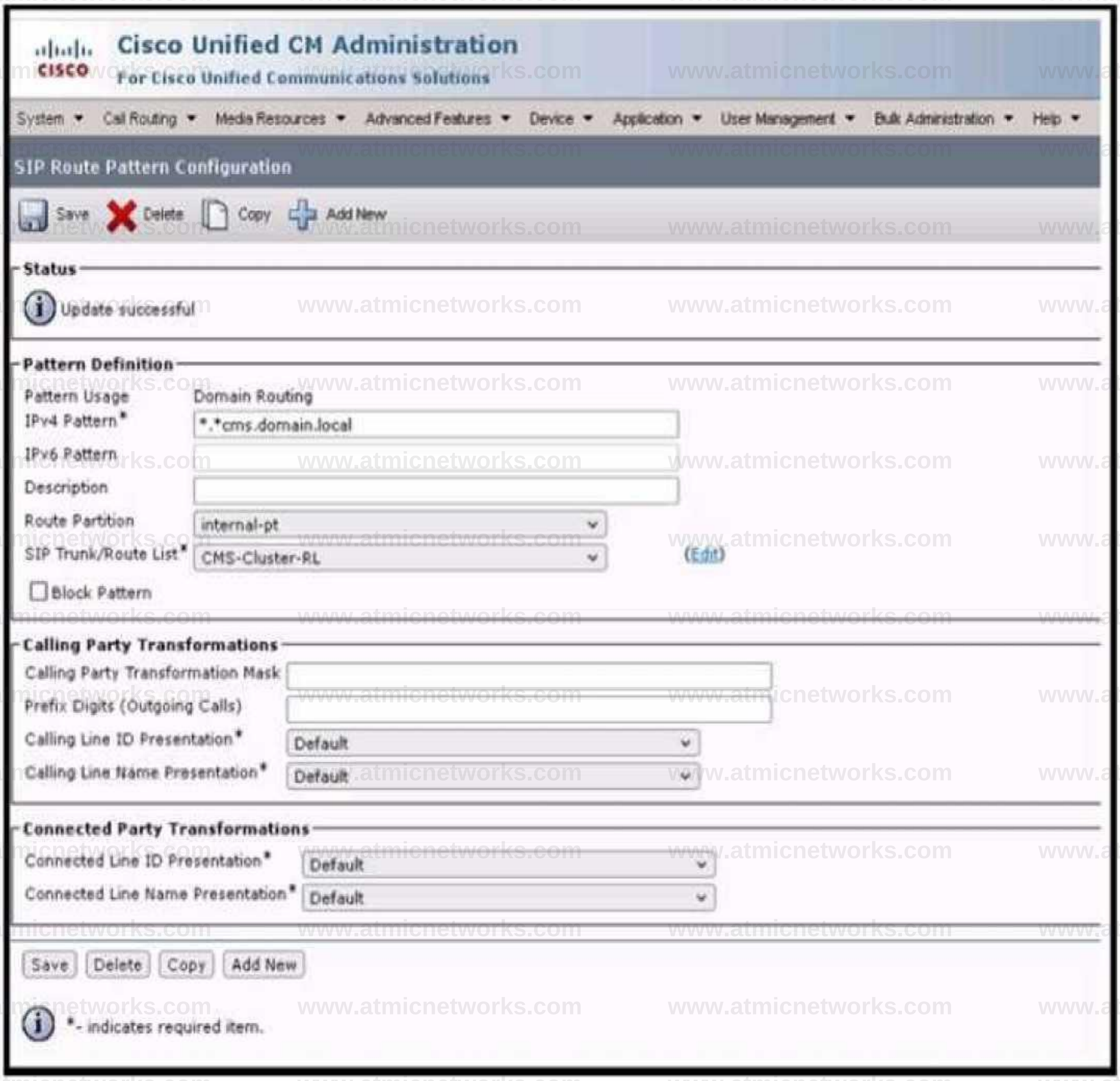
D. Option D

Answer: B

Explanation:

Question: 138

Refer to the exhibit.



The customer is troubleshooting an issue where users cannot dial the CMS SIP Route Pattern. The CMS URI they are attempting to dial is 7772002@cms.domain.local. Which IPv4 pattern must the customer enter to resolve the issue?

A. '@cms.domain.local

B. cms.domain.local

C. ".cms.domain.local

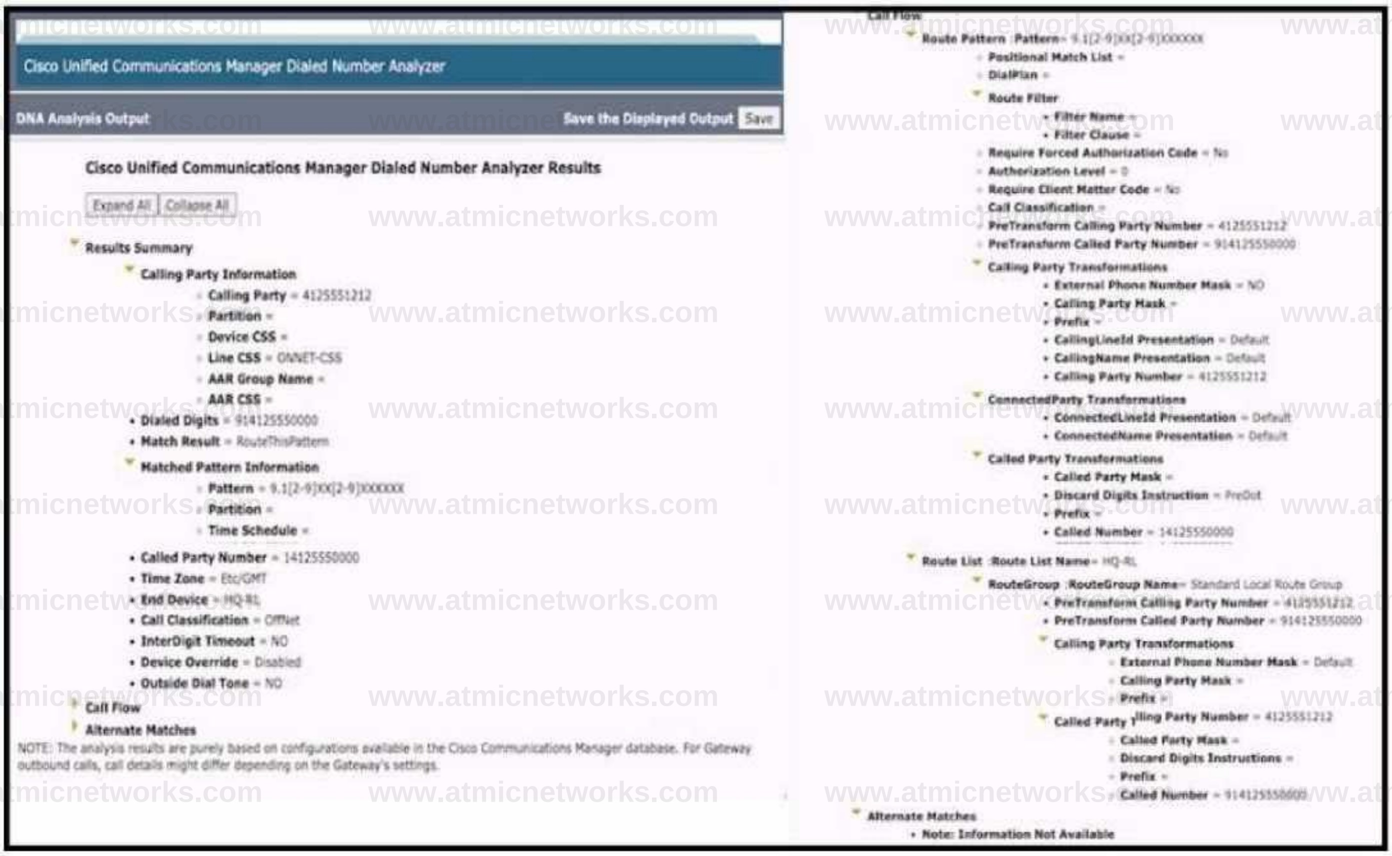
D. 7772002@cms.domain.local

Answer: B

Explanation:

Question: 139

Refer to the exhibit.



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

A. Deselect "Block this pattern" on the route pattern.

B. Change the calling search space of The user's line or device.

C. Change the "Call Classification" to "OnNet" on the route pattern.

D. Ensure that the user's phone is assigned to a device pool with the correct local route group settings.

Answer: D

Explanation:

Question: 140

A company wants to use a Cisco phone system without the need for a local generic server. It is maximum 30 phones using a vendor-independent, industry-standard call protocol. What is the first command that must be entered to add this type of phone when registering a Cisco phone to this device?

A. ephone 3

B. voice register pool 3

C. voice register sip ephone 3

D. voice register ephone 3

Answer: B

Explanation:

Question: 141

An administrator is configuring Cisco UCM and wants all inbound calls to the company's published DID number to ring on the receptionist's internal directory number. Which configuration step meets this requirement?

A. Create a translation pattern for the main number and map it to the receptionist's number.

B. Create a route pattern for the main number and map it to the receptionist's number.

C. Create a call park for the main number and map it to the receptionist's number.

D. Create a transformation pattern for the main number and map it to the receptionist's number.

Answer: A

Explanation:

Question: 142

A customer reports audio quality issues between video endpoints in the HQ location in California and one of the branches in Texas. Which two actions must RTCP take to troubleshoot this issue? (Choose two.)

A. Compress the headers of RTP traffic to lower the bandwidth consumption over the WAN. which allows more calls with less bandwidth consumed.

B. Configure the rtcp keepalive command in Cisco Unified Border Element to generate reports, which can be reviewed using the debug voip rtcp packet command.

C. Allow for VAD to be used for calls using the G7.29 codec, which reduces the usage of the WAN bandwidth and saves around 30% of bandwidth per call.

D. Encrypt the media to stop rogue devices from replying and putting that traffic back on the WAN, which avoids any extra bandwidth and ensures the quality of the calls.

E. Gather statistics on a media connection and information such as packets sent, lost packets, jitter, feedback, and round-trip delay. This information can help isolate the type of audio quality issues and the direction of the affected traffic.

Answer: A, E

Explanation:

Question: 143

A collaboration engineer is configuring call pickup for a Cisco UCM user that currently has a working BLF speed dial button for the teammate's extension. The user wants to see calls ringing on the teammate's extension and to answer those calls by pressing the BLF speed dial button. Which configuration step meets this requirement?

A. Set the call pickup group to <none> for both extensions.

B. Select the call pickup checkbox on the BLF speed dial settings.

C. Assign both extensions to the same partition.

D. Assign the extensions to different call pickup groups.

Answer: D

Explanation:

Question: 144

A customer reports that outbound calls from Cisco UCM to Cisco Unified Border Element are not getting early medi

a. During troubleshooting, the customer finds that Cisco UCM is sending a delayed offer SIP INVITE. Cisco Unified Border Element responds with a 183 Session Progress with SDP message. However, the customer finds that Cisco UCM does not respond to the 183 messages. Which action must the customer perform to resolve the problem?

A. On Cisco Unified Border Element, configure midcall-signaling passthru.

B. On Cisco UCM, change the "Early Offer support for voice and video calls" field on the SIP profile that is applied to the SIP trunk to Disabled.

C. On Cisco Unified Border Element, configure early-offer forced.

D. On Cisco UCM. change the "SIP ReHXX Options" field on the SIP profile that is applied to the SIP trunk to Send PRACK if 1xx Contains SDP.

Answer: C

Explanation:

Question: 145

A Collaboration engineer is implementing call processing capabilities on the Cisco UCM and needs to configure Call Queuing to place callers in a queue until hunt members are available to answer them. Which action must be completed before this feature can be configured?

A. Configure Hunt Pilot Queuing.

B. Service the Cisco IP Voice Media Streaming (IPVMS) application.

C. Configure Music On Hold.

D. Configure announcements.

Answer: A

Explanation:

Question: 146

What are two functions of the metadata in SIP recording? (Choose two.)

A. includes the details of the application media stream

B. contains the cryptographic context of the SRTP passthrough calls

C. identifies the session and media association time

D. carries the Agent Statistics Report

E. carries the communication session data (audio and video calls) that describes the call to the recording server

Answer: B, D

Explanation:

Question: 147

An engineer is configuring a Cisco UCM solution. The requirements state that some users in one location will receive calls from a number during work hours 09AM to 5PM, and another group will get the calls from the same number outside this defined timeslot. Users will also change the outgoing number when reaching out to customers based on the same time-of-day routing rules. Which feature is needed to allow for this type of configuration?

A. partitions

B. route groups

C. route lists

D. CSS

Answer: A

Explanation:

Question: 148

A company has users that frequently visit locations outside the office. These users need to be able to make calls and keep the company integrity by presenting a company caller ID. The users are equipped with company-paid mobile phones. The company uses Cisco UCM with Cisco Unified Border Element and Expressway-C and Expressway-E for external communications. Which technology must be configured to meet this requirement?

A. Cisco Unified SNR

B. Cisco Unified Mobility

C. Cisco FindMe

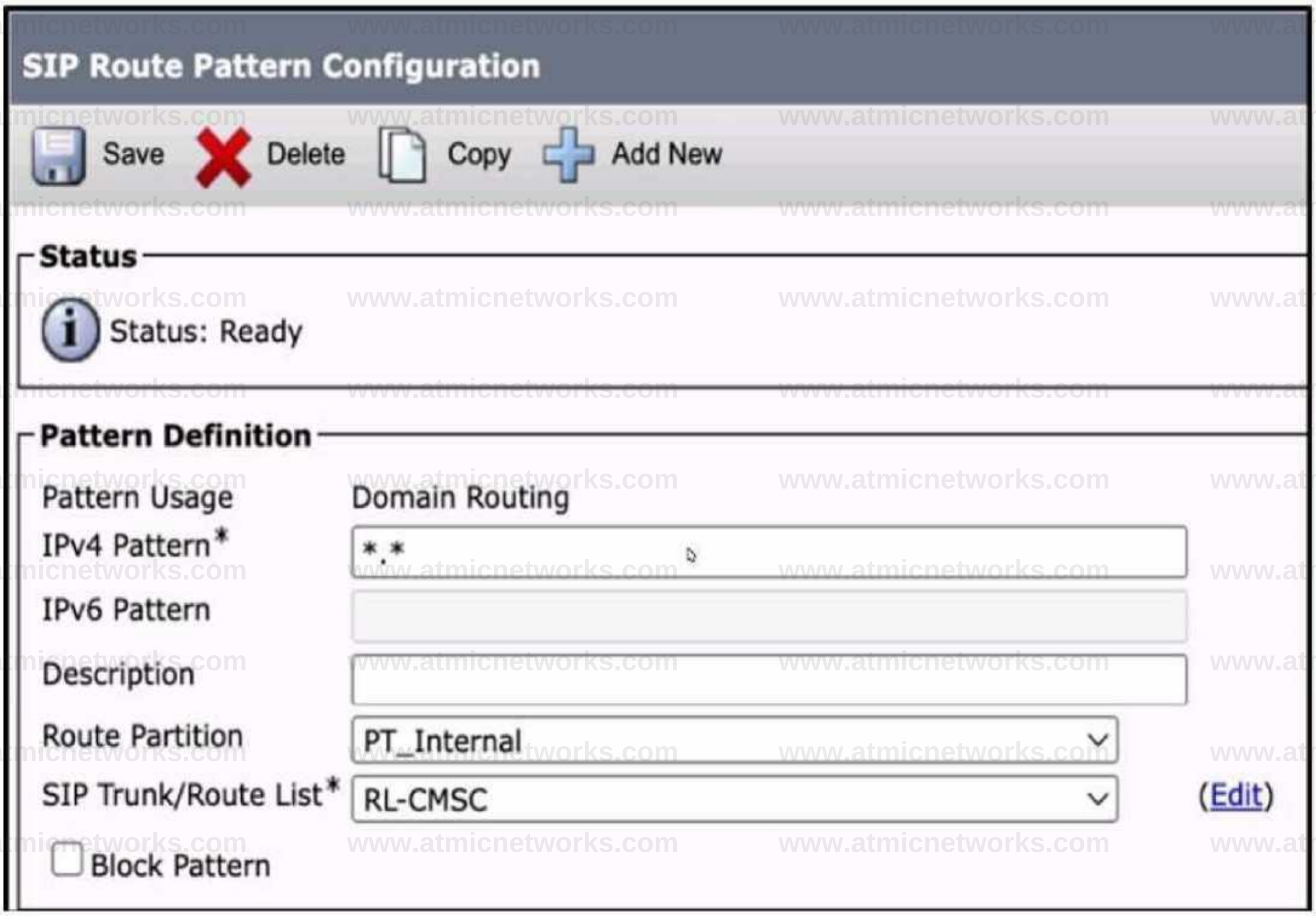
D. Cisco Roaming Mobility

Answer: B

Explanation:

Question: 149

Refer to the exhibit.



A collaboration engineer investigates an issue where the user calls that match the local Jabber domain are routed back to Cisco UCM. Which action resolves this problem?

A. Check the check box to block the pattern for the local domain.

B. Create another route SIP pattern for the local domain with the block pattern selected.

C. Create a translation pattern to block calls to the local domain.

D. Create a transformation pattern to block calls to the local domain.

Answer: B

Explanation:

Question: 150

A newly deployed site in Pennsylvania uses SIP-to-SIP legs on a Cisco Unified Border Element. The administrator wants to match the incoming dial peer based on the calling number. Which configuration is needed in the dial-peer?

A. answer-address

B. incoming called-number

C. destination calling

D. destination-pattern

Answer: B

Explanation:

Question: 151

An engineer configures several calling party transformation patterns in Cisco UCM so that calling numbers are shown in the +E.164 format. The route patterns match seven-digit internal numbers and prefix the numbers with 1408. The SIP trunk matches 1408XXXXXXX and prefixes the number with +. When users make outbound calls, the calling number still shows as seven digits. What is the cause of this problem?

A. The calling transformation on the SIP trunk is applied first, and then the calling transformation on the route patterns is applied.

B. Calling transformations cannot be configured on both route patterns and a SIP trunk.

C. The calling transformation on the route patterns is ignored because there is already a calling transformation on the SIP trunk.

D. The engineer must also apply a calling transformation at the route group details level.

Answer: D

Explanation:

Question: 152

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

A. Enter "!#" in the route pattern section tied to a route group and list.

B. Enter "*" in the SIP route pattern.

C. Enter "!#" in the SIP route pattern.

D. Enter ".*" in the route pattern section tied to a route group and list.

Answer: D

Explanation:

Question: 153

Refer to the exhibit.

<sip:1155@10.2.2.13>;privacy=off;reason=unconditional;counter=1;screen=no

and from the Cisco CUBE the logs show :

<sip:1155@10.3.3.25>;privacy=off;reason=unconditional;counter=1;screen=no

Refer to the exhibit. Users report that outgoing calls do not work on the new SIP trunk for outgoing calls. The solution consists of a Cisco UCM Cluster linked to a Cisco Unified Border Element where the SIP trunk is terminated. The provider required 10 digits. The logs show a line going toward the

Cisco Unified Border Element. Which code snippet must be added to the configuration to meet the requirement?

A. request Invite sip-header modify "<sip:1(...)@" "<sip:9135551\1@" under the SIP translation profile configuration

B. sip-header modify "<sip:1(...)@" "<sip:9135551\1@" under the voice translation profile configuration

C. request Invite sip-header Diversion modify "<sip:1(...)@" "<sip:9135551\1@" under the SIP profile configuration

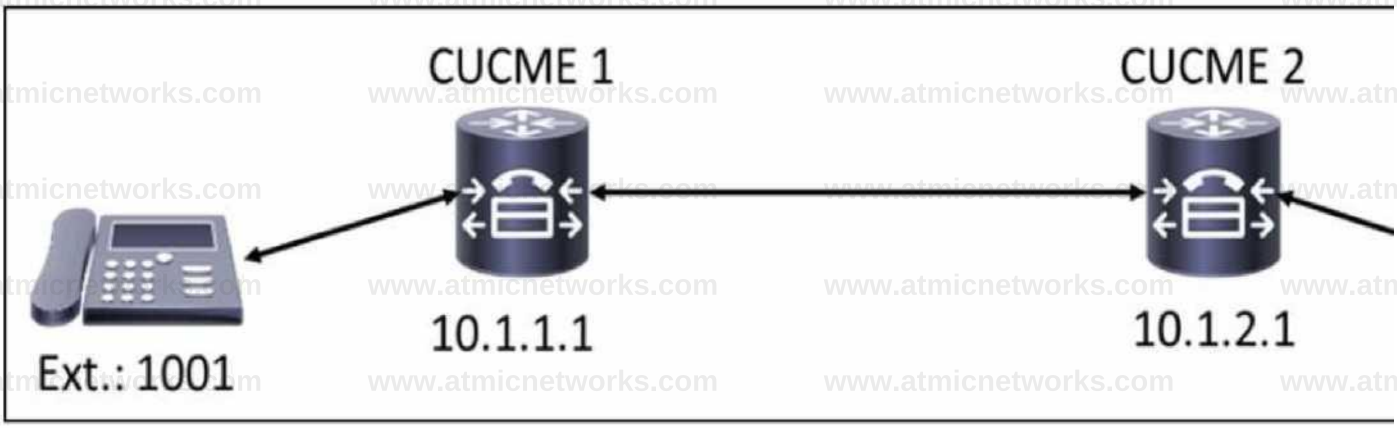
D. request Invite sip-header modify "<sip:1(...)@" "<sip:9135551\1@" under the voice translation profile configuration

Answer: C

Explanation:

Question: 154

Refer to the exhibit.



Refer to the exhibit. A company deploys an additional instance of Cisco UCME named CUCME 2 in a branch office. Phones in the main office are register of Unified CME named CUCME 1. Phones in the branch office are registered with their extension to CUCME 2. The engineer needs to configure CUCME the branch office using an 8-kbps codec. Which configuration should be applied to CUCME 1?

A)

```
dial-peer voice 1 pots destination-pattern 2...$
 session target ipv4:10.1.2.1 session protocol sipv2
```

B)

```
dial-peer voice 1 voip
 destination-pattern 2...$
 session target ipv4:10.1.2.1
 codec g711ulaw
```

C)

```
dial-peer voice 1 voip
 destination-pattern 2...$
 session target ipv4:10.1.1.1
 session protocol sipv2
```

D)

```
dial-peer voice 1 voip destination-pattern 2... $ session target
  ipv4:10.1.2.1 session protocol sipv2
```

A. Option A

B. Option B

C. Option C

D. Option D

Answer: D

Explanation:

Question: 155

What does VoIP trace in a Cisco Unified Border Element do?

A. It stores allcall trace data in system memory for failed calls.

B. It stores allcan trace data in system memory (or successful and failed calls

C. It stores allcall trace data in system log files for successful and failedcalls.

D. It stores allcan trace data in system file logs for failed calls.

Answer: B

Explanation:

Question: 156

DRAG DROP

An organization configures a SIP trunk in Cisco UCM to connect to another system. These requirements must be met:

1. Use a specific IP address for SIP signaling.

2. Encrypt the signaling traffic.

3. Restrict which devices can use the SIP trunk. 4 Simplify SIP signaling.

Drag and drop the Cisco UCM configuration steps from the left onto the order on the right to achieve these goals.

enable TLS	Step 1
set the SIP trunk destination	Step 2
Assign a device pool	Step 3
configure SIP Early O'er	Step 4

Answer:

Explanation:

set the SIP trunk destination

assign a device pool

enable TLS

configure SIP Early Offer

Question: 157

Users in a small location report that audio calls work Tine, but only one video call works before the next one gets a busy signal. Three video conference rooms were recently installed by a third party. The audio and video devices are Cisco endpoints. The site uses video and audio calls, and a network investigation does not show saturation on the link The site is configured with QoS and CoS. All devices are registered on the headquarters Cisco UCM duster. What is the cause of this problem?

A. The subzone setting for video is too low.

B. The zone settings for video are too low.

C. The location setting for video is too low.

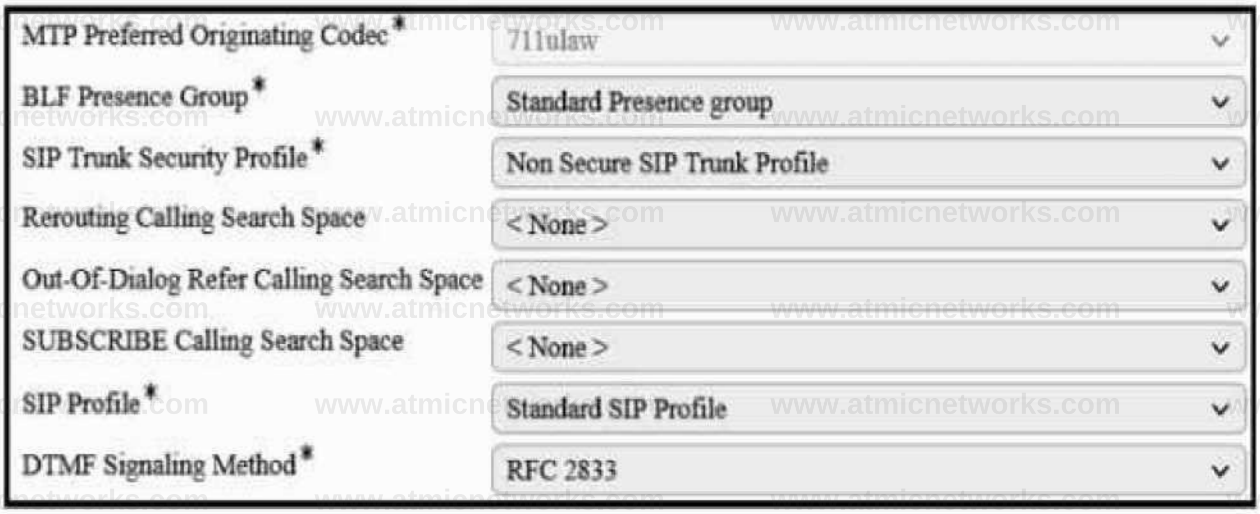
D. The pipe or link configuration for video is set to only one device.

Answer: D

Explanation:

Question: 158

Refer to the exhibit.



An administrator just Implemented SIP trunking on their Cisco UCM and reports that calls using the SIP trunk are using Media Termination Point resources unnecessarily. Which action resolves the issue?

A. Disable SIP Red XX Options.

B. Change to a range that does not result in MTP.

C. Change OTMF Signaling Method to "No Preference".

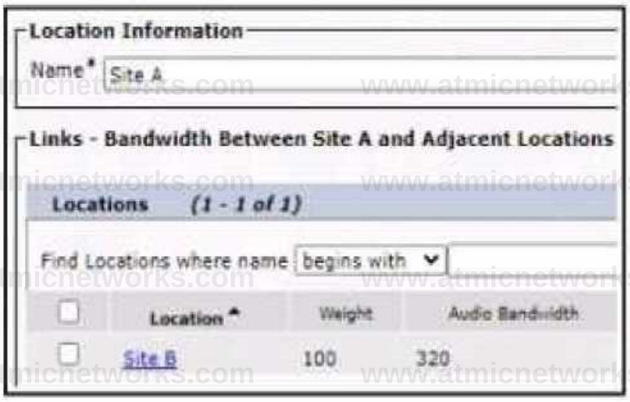
D. Change DTMF Signaling Method to "RFC 4733".

Answer: C

Explanation:

Question: 159

Refer to the exhibit.



Refer to the exhibit. An engineer deploys CAC to Cisco UCM. UptofiveG.711 calls must be supported on the WAN link between Site A and Site B. Users report that only four concurrent calls are possible between Site A and Site B. Why isn't the filth concurrent call successful?

A. The G.729 audio codec is negotiated between Site A and B instead of the G.711 codec.

B. The QoS configuration on the WAN link causes the fifth call to be dropped.

C. Cisco UCM is using alternative links because of the Weight value.

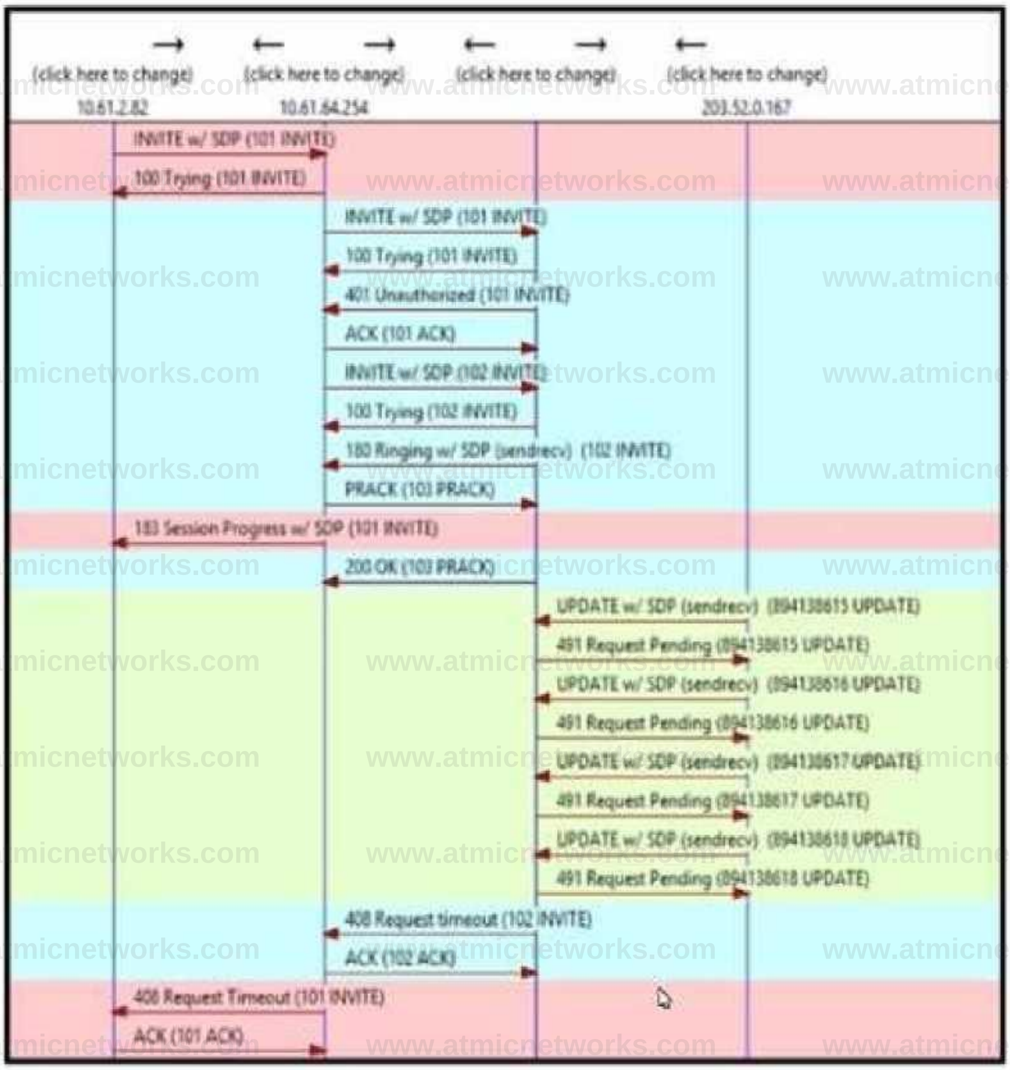
D. The Audio Bandwidth value is undersized and must be set to 400 kbps.

Answer: A

Explanation:

Question: 160

Refer to the exhibit.



Refer to the exhibit. Calls from users to the PSTN in an organization get disconnected with a 408 Request Timeout when the called party is unavailable to pick up the call. Which solution must be used to resolve this challenge?

A. Configure midcall-signaling preserve-codec.

B. Choose "Send PRACK" if 1xx contains SDP in the SIP profile.

C. Configure midcall-signaling passthru media-change

D. Choose "Send PRACK for all 1xx Messages" in the SIP profile.

Answer: C

Explanation:

Question: 161

Alter a new SIP trunk is configured from a third-party SIP trunk provider, the call does not seem to work. A troubles hooting tool and a tiace log from the Cisco Unified Border Element shows that there is communication with the provider and that it is sending out SDP information in the 2XX response. Why is the call not working?

A. The Cisco Unified Border Element does not have any PDVM resources to terminate the SIP trunk assigned.

B. The SIP provider expects a delayed offer.

C. The SIP provider expects an early offer.

D. The Cisco Unified Border Element must be licensed for third-party SIP trunks.

Answer: B

Explanation:

Question: 162

An SRST site must use pickup functionality on the Chicago remote site to allow the users to take incoming calls. The pickup command is configured under the call-manager-fallback section on the SRST router What are two results of that configuration? (Choose two.)

A. The PickUp soft key is on all SRST phones.

B. The GPickUp soft key is on all SRST phones.

C. Calls that come into one directory number can be picked up from another directory number.

D. Calls that come into a Cisco UCM registered phone can be picked up by a SRST phone.

E. Calls that ring an unassigned directory number are forwarded to the auto-attendant.

Answer: A, C

Explanation:

Question: 163

An administrator wants to use the Call Queuemg feature on Cisco UCM to allow new customers to wait in the queue while other agents are not available to take their call. Which configuration step enables this feature?

A. Call Routing > Route/Hunt > Call Park

B. Call Routing > Route/Hunt > Hunt Pilot

C. Call Routing > Route/Hunt > Line Group

D. Call Routing > Route/Hunt > Hunt List

Answer: B

Explanation:

Question: 164

Refer to the exhibit.

```
CUBE! shew sic-ua status
SIF User Agent Status
sir User Agent for UDE : ENABLED
Sir User Agent for TCP : ENABLED
Sir User Agent for TLS over TCP : ENABLED
SI? User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): ENABLED 192.1€:.IC.1
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIE DNS SAV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event^ in NOTIFYs: 2000 ms
SIP support for ISDN SUSFEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
protocol mode is ipv4
```

Refer to the exhibit An engineer deploys a Cisco Unified Border Element for SIP trunk interconnection to an ITSP The engineer performs test calls but there is no audio when the called party picks up the phone. What must the engineer do to resolve the issue?

A. Disable SIP User Agent bind status (media).

B. Decrease the SIP max-forwards value to 16.

C. Disable SIP early-media for 180 responses with SDP

D. Enable Check media source packets.

Answer: B

Explanation:

Question: 165

Refer to the exhibit.

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

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

A. Change to a valid range.
B. Change DTMF Signaling Method to "RFC 4833*.

C. Make sure the destination number is configured correctly and the device is registered.

D. Disable SIP Rel1XX Options.

Answer: C

Explanation:

Question: 166

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

A. Set all dusters to standalone on the ILS configuration page.

B. Install multiserver CA-signed Tomcat certificates on all cluster publishers.

C. Enable password-based authentication in conjunction with certificate-based authentication.

D. Ensure that the cluster ID for all clusters is the same.

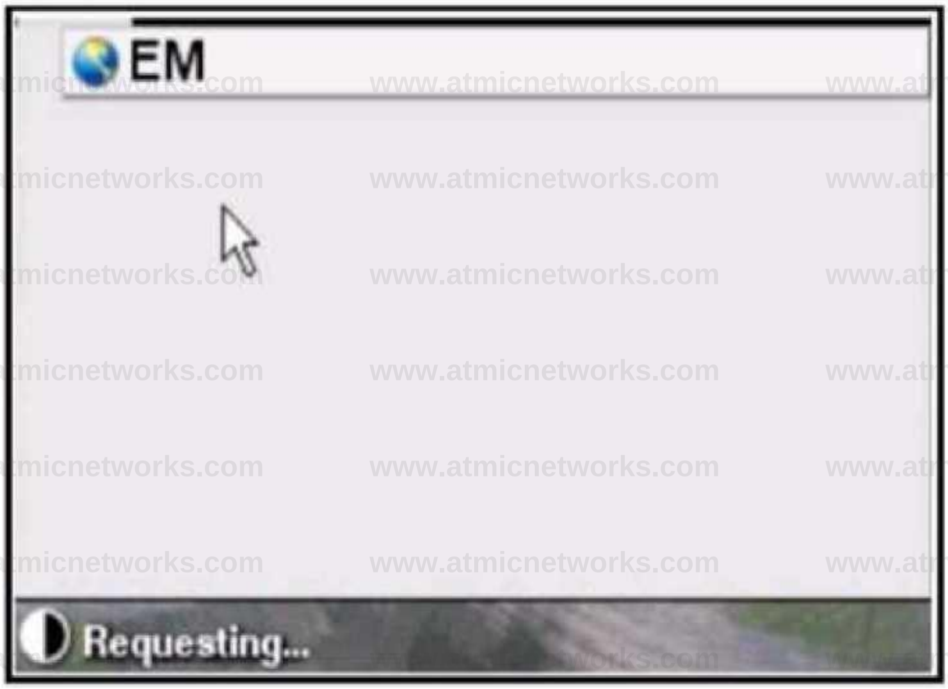
E. Install multiserver CA-signed CallManager certificates on all cluster publishers.

Answer: B, C

Explanation:

Question: 167

Refer to me exhibit.



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

A. IP phone services Extension Mobility does not have the correct service URL configured.

B. IP phone services Extension Mobility has XML Service Category configured.

C. The Cisco IP phone has not checked the Extension Mobility checkbox.

D. The Cisco IP phone is not subscribed to an Extension Mobility service.

Answer: A

Explanation:

Question: 168

An engineer Is configuring a Cisco Unified Border Element. The SIP ITSP requires a DNIS of 10-digits and DTMF tones sent in the RTP stream on the outbound leg of a call. Which code completes this configuration?

A)

dial-peer voice 10 volp destination-pattern [2-9]..p-9]....4 session protocol sipv2 voice-class codec 1
sip-inlo

B)

dial-peer voice 10 voip destination-pattern [2-9]..[2-9],.....$ session protocol sipv2
voice-class codec 1
sip-kpml

C)

dial-peer voice 10 voip destination-pattern [2-9].[2-9],.....$ session protocol slpv2
voice-class codec 1
sip-notify

D)

dial-peer voice 10 voip destination-pattern [2-9]..[2-9]......$ session protocol sipv2 voice-class codec 1
ftp-nte

A. Option A

B. Option B

C. Option C

D. Option D

Answer: D

Explanation:

Question: 169

A company with only 15 users needs to have a phone system without deploying dedicated servers for this purpose. Using the local Cisco router with Cisco UCME capabilities, which configuration is needed to support SIP phones?

A. voice-register ephone

B. ephone

C. voice-register pool

D. voice-register SIP

Answer: B

Explanation:

Question: 170

Users report slrange behaviors when Ihey call external numbers from the company phone system. The company is using Cisco 8865 phones with Cisco UCM connected to a gateway with two T1 circuits. There is no indication of the possible problem, but the connection seems to be established Which command meets the requirement to capture all the data needed to debug the problem?

A. debug mgcp media tracelevel critical all

B. debug mgcp media tracelevel critical verbose

C. debug mgcp media tracelevel critical

D. debug mgcp media tracelevel moderate

Answer: D

Explanation:

Question: 171

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

Update the softkey template on all phones to ensure that they contain the Meet-me softkey.

A. Ensure that conferencing-initiating devices are using a media resource group list that contains at least one Cisco UCM conference bridge.

B. Enable Meet-me conferencing In enterprise parameters.

C. Ensure that all devices have G.729 enabled.

D. Disable Early Media on the SIP profile of all devices that will use Meet-me conferencing.

Answer: A, B

Explanation:

Question: 172

A Cisco UCM-based solution uses URI-based calling for Cisco Jabber clients. The solution has several clusters in different locations. The solution being configured must have a secure method o* exchanging addresses between the clusters. Which action ensures that a configuration complies with the requirements?

A. Assign a self-signed certificate to the ILS service trunk.

B. Apply a secure security profile to the trunk between the clusters.

C. Select the TLS option on the SIP trunk between the clusters.

D. Exchange Tomcat certificates

Answer: C

Explanation:

Question: 173

The same digits sent in OOB, as well as in-band on the Cisco Unified Border Element, are being interpreted as duplicate digits by the receiving end. Which configuration must an engineer implement to resolve this issue?

A. DTMF Relay SIP-KPML

B. DTMF Relay SIP-Notify

C. DTMF Relay RTP-nte

D. DTMF Relay Digit-Drop

Answer: D

Explanation:

Question: 174

An engineer configured extension mobility to allow users to roam between phones while taking their numbers and specific device settings with them. Users are now getting "Error:- XML Error [4| Parse Error" on their phones. Which action will resolve this issue?

A. Set service provisioning lo internal.

B. Lower the number of concurrent requests.

C. Verify that the URL Is correctly configured.

D. Set service provisioning to default.

Answer: C

Question: 175

When an administrator troubleshoots H.323 call setup, which message gives an alert that the called party is being notified about the call?

A. ALERTING

B. PROCEEDING

C. CONNECT

D. RINGING

Answer: C

Explanation:

Question: 176

In Cisco UCM, which tool is used to check SIP traces?

A. MTP

B. CCSIP

C. RTMT

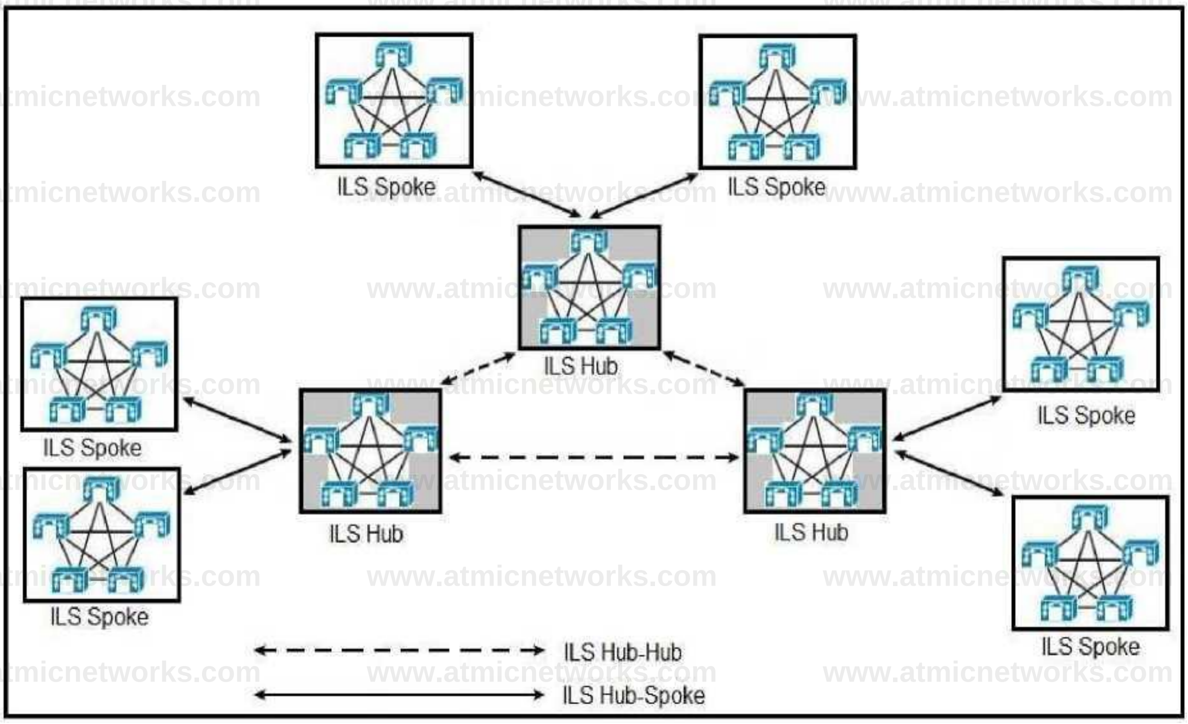
D. OS Administration Page

Answer: C

Explanation:

Question: 177

Refer to the exhibit.



Refer to the exhibit. How many maximum hops does an ILS update traverse?

A. 3

B. 6

C. 9

D. 12

Answer: A

Explanation:

Question: 178

When configuring hunt groups, where does the administrator add the individual directory numbers that should be part of the group?

A. route group

B. line group

C. hunt list

D. hunt pilot

Answer: B

Explanation:

Question: 179

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

A. translation patterns

B. tail end hop off

C. client matter codes

D. dial rules

Answer: B

Explanation:

Question: 180

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

A. dial-peer voice 1 voip

voice-class source interface GigabitEthernet0/0

B. voice service voip

bind sip source-interface GigabitEthernet0/0

C. voice service voip

sip

bind all source-interface GigabitEthernet0/0

D. dial-peer voice 1 voip

voice-class bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

Answer: C

Explanation:

Question: 181

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

A. Use RSPV.

B. Use Location Bandwidth Manager.

C. Use Expressway deployment.

D. Use Call Allow Controller.

Answer: B

Explanation:

Question: 182

Refer to the exhibit.

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

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

A. dtmf-relay h245-alphanumeric

B. dtmf-relay sip-kpml

C. dtmf-relay cisco-rtp

D. dtmf-relay rtp-nte

Answer: B

Explanation:

Question: 183

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

A. hunt group

B. line group

C. hunt list

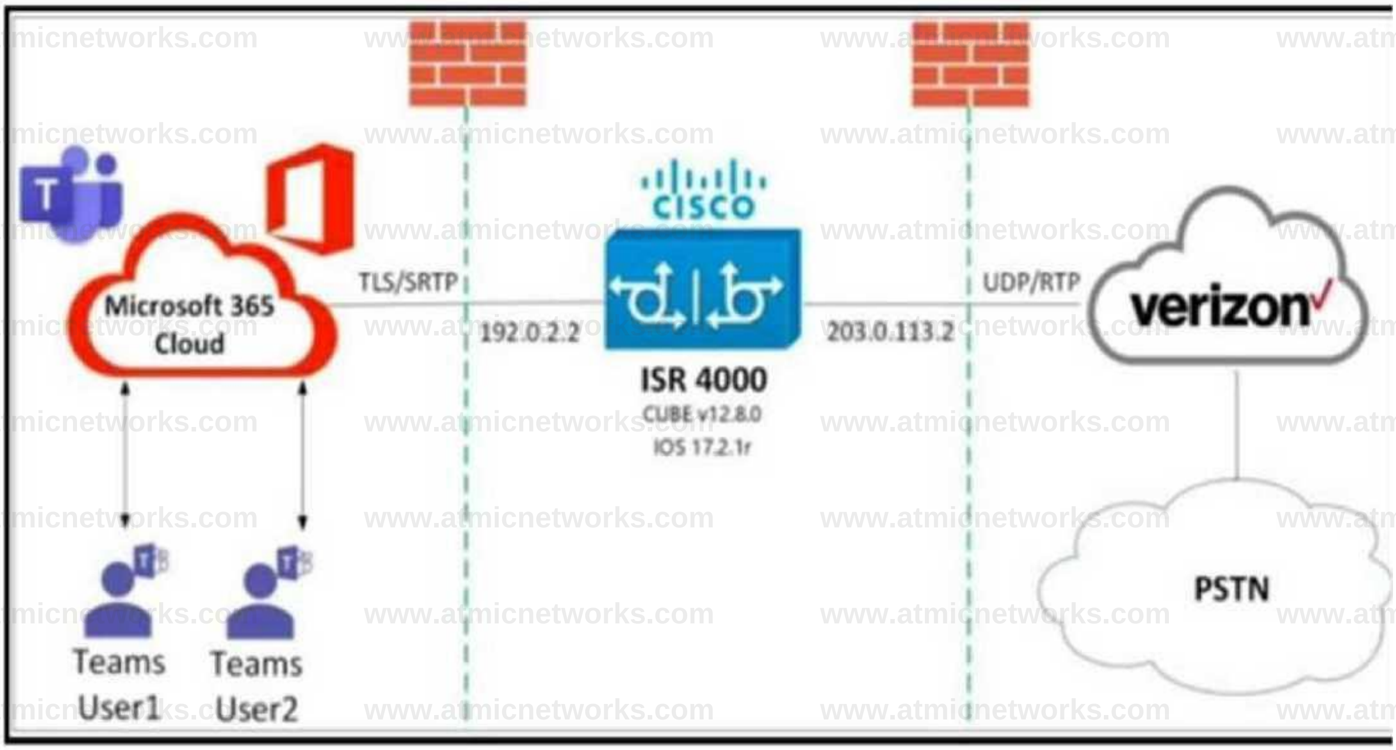
D. route list

Answer: C

Explanation:

Question: 184

Refer to the exhibit.



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

A.

voice class sip-profiles 1

request INVITE sdp-header Connection-Info modify “2\.76\.1” “2.78.1”

B.

voice class sip-profiles 1

request INVITE sdp-header Session-Owner modify “27\.0\.0” “27.3.0”

C.

voice class sip-profiles 1

request INVITE sdp-header Connection add “2\.76\.1” “2.78.1”

D.

voice class sip-profiles 1

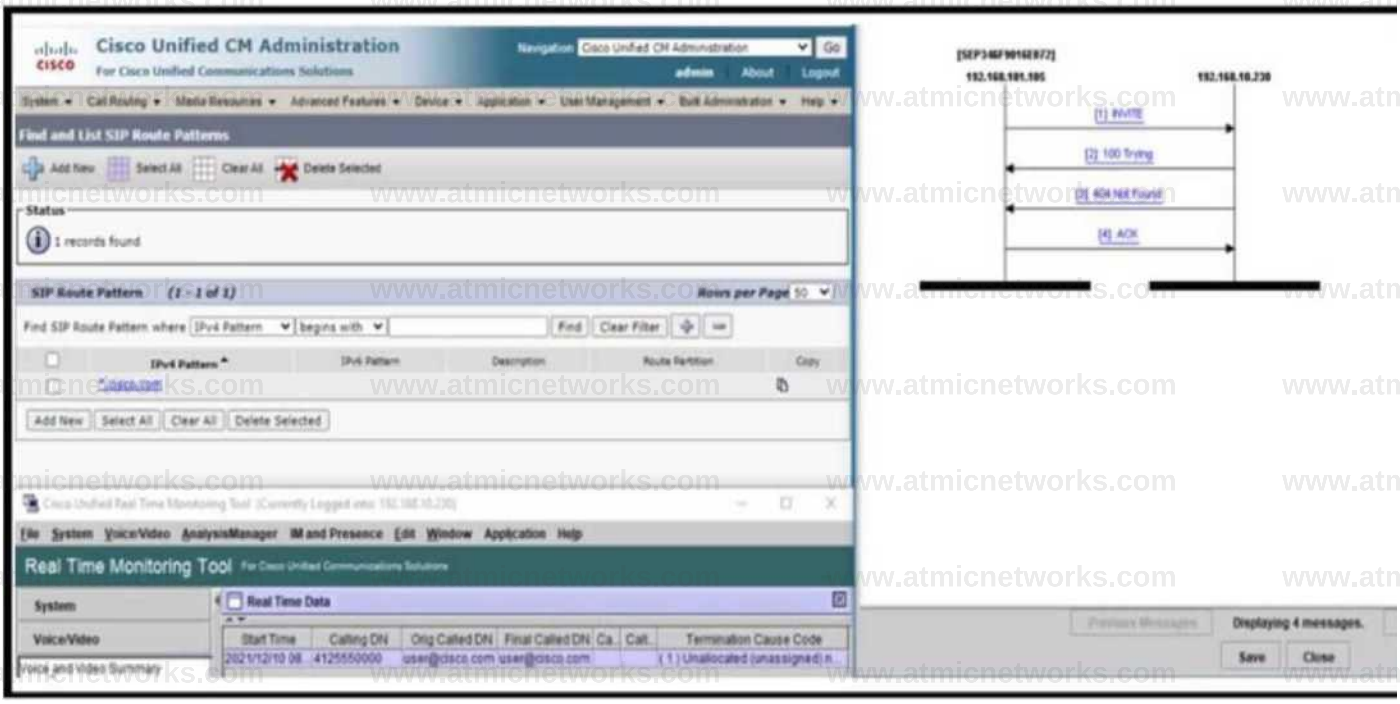
request INVITE sdp-header Audio-Attribute modify “2\.76\.1” “2.71.1”

Answer: C

Explanation:

Question: 185

Refer to the exhibit.



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

A. Change the “Route Partition” setting on the *.cisco.com route pattern.

B. Add a SIP route pattern for *.*.

C. Select “Block Pattern” on the *.cisco.com route pattern.

D. Delete the *.cisco.com route pattern.

Answer: B

Explanation:

Question: 186

Refer to the exhibit.

```
02904115.001 |09:08:07.093 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.1.1.102 on
port 50244 index 22157
[166156,NET]
ACK sip:1315932e-ff29-4203-23e3-ef940216638e@10.1.1.102:50244;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.1.1.5:5060;branch=z9hG4bK4fa1fa89d7e
From: <sip:1001@10.1.1.5>;tag=93016~bb788fb3-a1ef-4d03-a96d-651038e22050-28377951
To: <sip:1000@cucm1251.cisco.lab>;tag=7001b5dab46425b45ec6648a-25dc4f91
Date: Wed, 28 Jul 2021 13:08:07 GMT
Call-ID: cee27300-1ed10da7-b403-3251300e@10.1.1.5
User-Agent: Cisco-CUCM12.5
Max-Forwards: 70
CSeq: 103 ACK
Allow-Events: presence
Session-ID: 36ed016300105000a0002834a2824611;remote=49f3b76a00105000a0007001b5dab464
Content-Type: application/sdp
Content-Length: 412

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

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

A. The bandwidth configured for this region relationship is too low and must be increased to 96 kbps or higher.

B. The device that is placed on hold does not support G.711, and a transcoder could not be allocated for the call.

C. Cisco UCM is sending a=inactive to the held device.

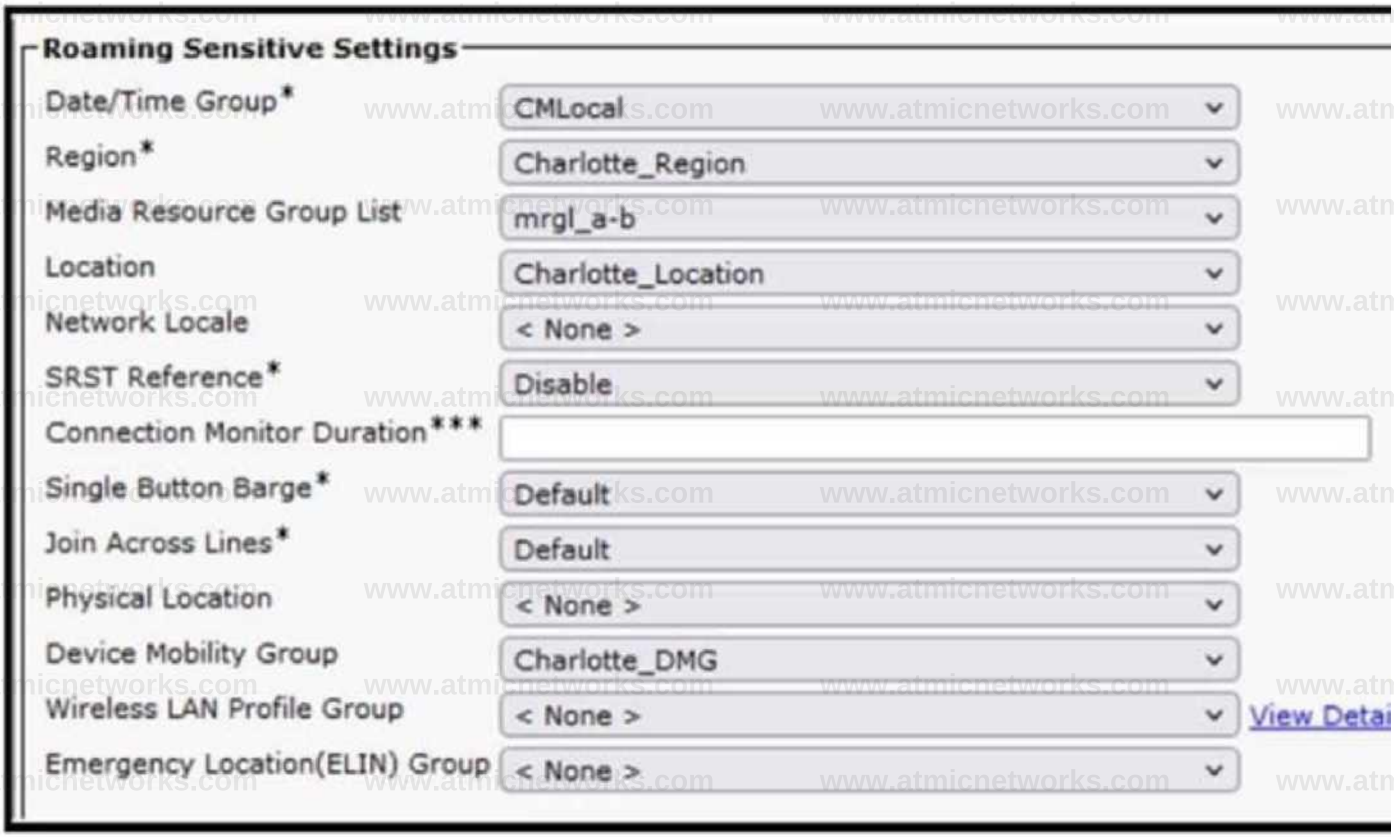
D. The Music On Hold server does not support G.711 and a transcoder could not be allocated for the call.

Answer: C

Explanation:

Question: 187

Refer to the exhibit.



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

A. Join across lines is not supported with Device Mobility and must be disabled.

B. The network locale must be changed from none to the Charlotte locale.

C. The physical location must be updated from none to the appropriate location.

D. The device must be added to the Device Mobility group.

Answer: D

Explanation:

Question: 188

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

A. Location Bandwidth Manager

B. Extension Mobility

C. Device Mobility

D. Extension Mobility Cross Cluster

Answer: A

Explanation:

Question: 189

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

A. voice service voip

allow-connections sip to h323

B. voice service voip

sip

bind media source-interface Vlan100

C. voice service voip

sip

bind control source-interface Vlan100

D. voice service voip

sip

registrar server expires max 600 min 60

Answer: C

Explanation:

Question: 190

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

A. calling party transformation

B. partition of the SIP route pattern

C. connected party transformation

D. called party URI transformation

E. destination SIP trunk of the SIP route pattern

Answer: B

Explanation:

Question: 191

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

A. The CSS of the transfer initiating line does not contain the partition of the supplementary feature extension (DirectTransfer or MoH Number).

B. The MTP that is configured for use within the H.323 gateway configuration is configured as a trusted source, but the third-party gateway does not trust the signing root CA certificate of the MTP certificate.

C. The MTP does not support the negotiated codec, and media renegotiating during the call is not supported.

D. The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP.

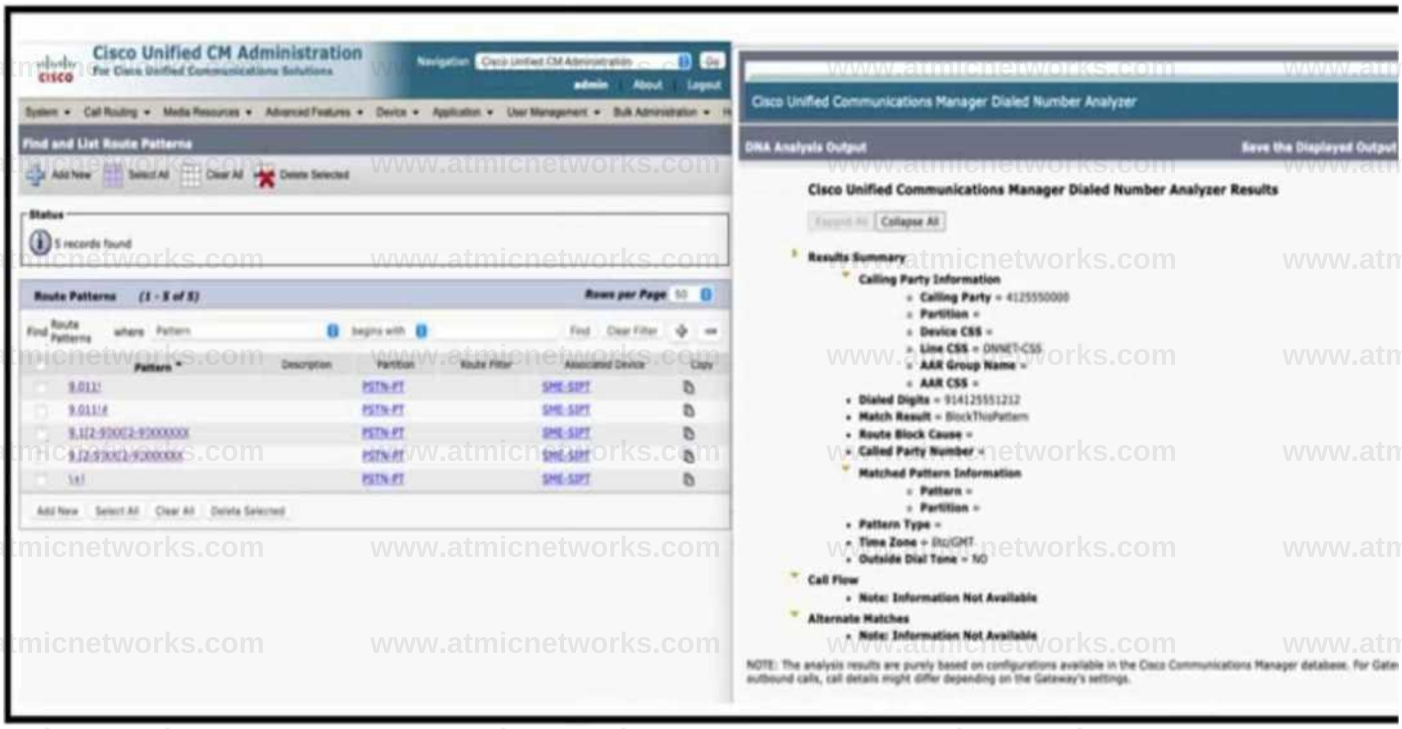
E. The third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.

Answer: C, D

Explanation:

Question: 192

Refer to the exhibit.



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

A. Assign a calling search space to the line or the device that has access to the route pattern.

B. Deselect “Block this pattern” on the “Route Option” setting of the route pattern.

C. Select the “Urgent Priority” setting on the route pattern.

D. Instruct the user to not dial a “1” before their local area code.

Answer: A

Explanation:

Question: 193

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

A. dial-peer voice 2 voip

dtmf-relay sip-kpml

B. dial-peer voice 2 voip

dtmf-relay h245-alphanumeric

C. dial-peer voice 2 voip

dtmf-relay sip-notify

D. dial-peer voice 2 pots

dtmf-relay h245-alphanumeric

Answer: B

Explanation:

Question: 194

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

A. Identify the MOH audio source ID associated to one of the line group member's "Network Hold MOH Audio Source".

B. Identify the MOH audio source ID associated to "Network Hold Source & Announcements" under the Queuing section of the hunt pilot.

C. Identify the configured announcement names to change under the MOH audio source section, then upload the new files to the respective announcements under the Announcement section.

D. Identify the MOH audio source ID associated to one of the line group member's "User Hold MOH Audio Source".

E. Identify the configured announcement names to change under the Announcement section, and assign the uploaded files to the Queueing section of the hunt pilot.

Answer: D, E

Explanation:

Question: 195

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

A. transcoder

B. location

C. route pattern

D. translation pattern

Answer: B

Explanation:

Question: 196

Refer to the exhibit.

```
2899152: Mar 31 21:23:16.777: TCB7F7F9668A9E8 setting property TCP_ALWAYSPUSH (15) 7F7F94B03528
2899153: Mar 31 21:23:16.777:  tcp_uniqueport: using ephemeral max 65535
2899154: Mar 31 21:23:16.777: TCP: Random local port generated 30202, network 1
2899155: Mar 31 21:23:16.777: TCB7F7F9668A9E8 bound to 10.10.10.40.30202
2899156: Mar 31 21:23:16.777: Reserved port 30202 in Transport Port Agent for TCP IP type 1
2899157: Mar 31 21:23:16.777: TCP: sending SYN, seq 974312871, ack 0
2899158: Mar 31 21:23:16.777: TCP0: Connection to 10.10.10.14:5060, advertising MSS 536
2899159: Mar 31 21:23:16.777: TCP0: state was CLOSED -> SYNSENT [30202 -> 10.10.10.14(5060)]
2899160: Mar 31 21:23:18.777: TCP0: RETRANS timeout timer expired
2899161: Mar 31 21:23:18.777: 10.10.10.40:30202 <---> 10.10.10.14:5060   congestion window changes
2899162: Mar 31 21:23:18.777: cwnd from 536 to 536, ssthresh from 65535 to 1072
2899163: Mar 31 21:23:18.777: TCP0: timeout #1 - timeout is 4000 ms, seq 974312871
 --More--         2899164: Mar 31 21:23:18.777: TCP: (30202) -> 10.10.10.14(5060)
2899165: Mar 31 21:23:19.503: TCP866: state was ESTAB -> FINWAIT1 [22 -> 10.16.8.226(52145)]
2899166: Mar 31 21:23:19.503: TCP866: sending FIN
2899167: Mar 31 21:23:19.536: TCP866: state was FINWAIT1 -> FINWAIT2 [22 -> 10.16.8.226(52145)]
2899168: Mar 31 21:23:19.536: TCP866: FIN processed
2899169: Mar 31 21:23:19.536: TCP866: state was FINWAIT2 -> TIMEWAIT [22 -> 10.16.8.226(52145)]
2899170: Mar 31 21:23:21.777: Released port 30202 in Transport Port Agent for TCP IP type 1 delay 2
2899171: Mar 31 21:23:21.777: TCP0: state was SYNSENT -> CLOSED [30202 -> 10.10.10.14(5060)]
2899172: Mar 31 21:23:21.777: TCB 0x7F7F9668A9E8 destroyed
```

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

A. The ITSP is sending FIN.

B. The ITSP is not sending the ACK.

C. The ITSP is sending CLOSE state.

D. The ITSP is not answering the SYN.

Answer: D

Explanation:

Question: 197

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

A. router (config-dial-peer) dtmf-relay sip-notify

B. router (config-dial-peer) dtmf-relay sip-kpml

C. router (config-dial-peer) dtmf-relay h245-alphanumeric

D. router (config-dial-peer) dtmf-relay rtp-nte

Answer: D

Explanation:

Question: 198

A new solution is configured to support internal, local, and international calling. Calling [+44 1111 1111] from one of the registered internal phones does not work. Local and internal calls seem to work without any problems. The configuration has patterns configured to match the failing dialed number [+44]. The other configured patterns show [2…] for internal numbers and [555 …] for

local numbers. International numbers use E.164 as recommended. What is missing to make this solution work?

A. 001 or 00 must be used instead of the + sign on Cisco UCM

B. =+ cannot be used in a route pattern, only in a SIP pattern

C. \ in front of the +

D. / in front of the +

Answer: C

Explanation:

Question: 199

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

A. Route Numbers

B. Advertised Patterns

C. Learned Numbers

D. Route Patterns

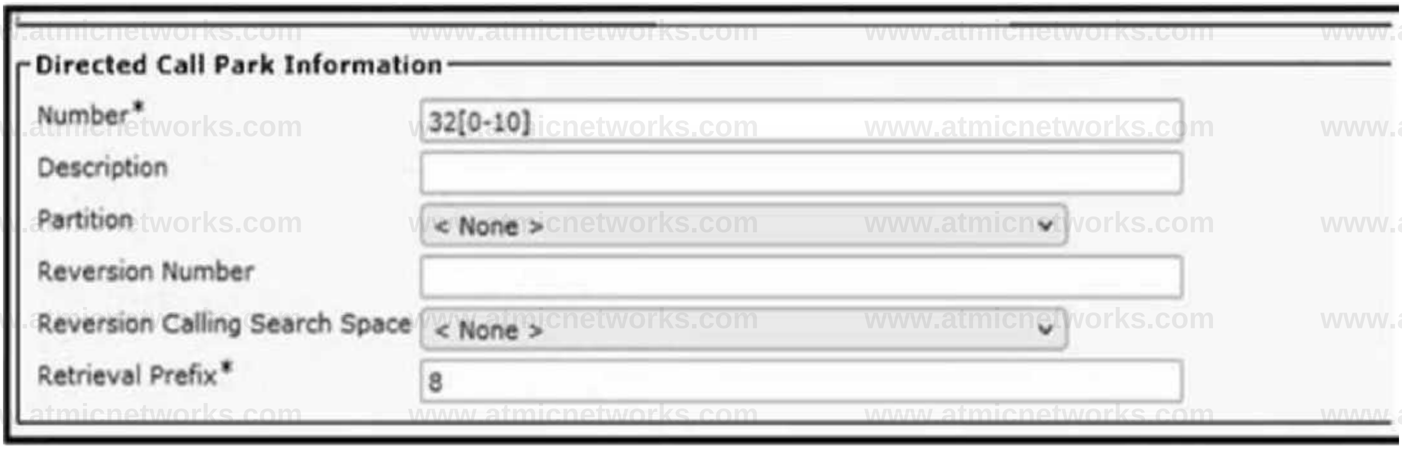
Answer: B

Explanation:

Question: 200

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

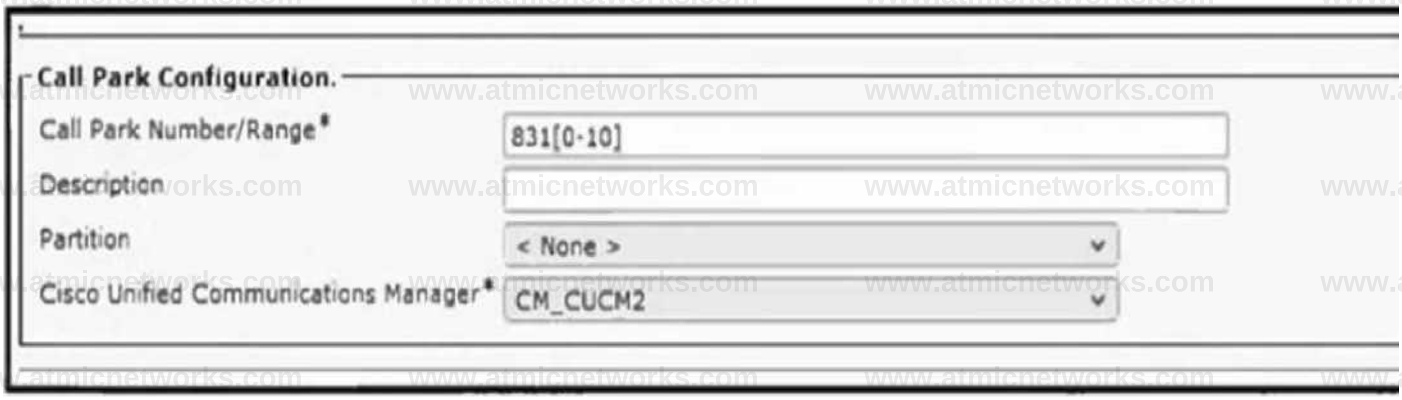
A)



Directed Call Park Information

Number*	32[0-10]
Description	
Partition	< None >
Reversion Number	
Reversion Calling Search Space	< None >
Retrieval Prefix*	8

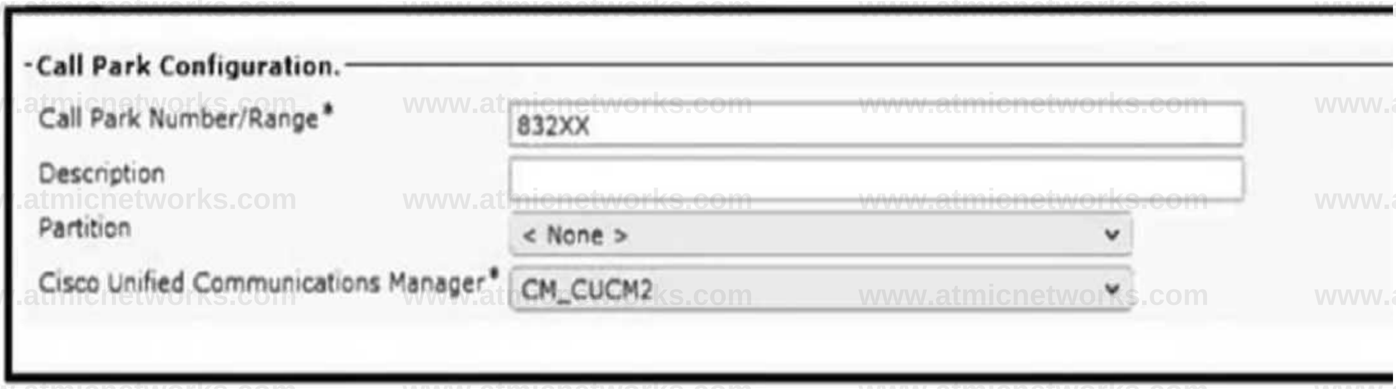
B)



Call Park Configuration.

Call Park Number/Range*	831[0-10]
Description	
Partition	< None >
Cisco Unified Communications Manager*	CM_CUCM2

C)



Call Park Configuration.

Call Park Number/Range*	832XX
Description	
Partition	< None >
Cisco Unified Communications Manager*	CM_CUCM2

D)

A. Option A

B. Option B

C. Option C

D. Option D

Answer: B

Explanation:

Question: 201

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

A. device mobility

B. standard local route group

C. local route groups

D. TEHO

Answer: A

Explanation:

Question: 202

Refer to the exhibit.

0 3 8 3 32 2 6 000 111 10 SS 116 ISdlStf IDihi I^foda
Cdccll,100.39 25) 1041.100,47,1) 11.100,251.7269 65*14 48 81100** [R N-
M O.N 0.10.v 0 7 0.0 0| 0*213 7 7985 0 4 FotetoPotert a Matchet Off Net
Prrt ranjformCpntn "On pfOt“Ind=* 19195741001 p'CvI
Cgpr=tn=Onp.=OV:lrd=«19195741OOlp =0^1 O4mfPartdiPAs717M6)M10b*2dS& 6420- 8blO99Wdb9
D^hr^AtlernHl |2-9|KXXXXX 0i4><'tmt>|91N|2*9||(M!(0-9n0*9]|fr9^^ 910*11.181.19)
Pretran»<ormCdon»tn*Orp«*0'i*Ind*91S745000pi'Ot© PretranpormTaft'ACCESS- CODE SUBSCRIBER
PretrantformPoi:91 5745000 ejected (>4.ts=tn=0nc.=0ti= |nd = S74 5000p.=<X 0
TaptSUMCRiBER Poi«S74SOOO Ca edP*ftvDew<ePcoPk.ds cceTnmpCtion d*0 <»CdpoNo^Preamptab'e*F
itCtpnNonPfeemp!abie'FpatterrU$a|e *S'tQueftiO <0 03833226 001 |11 10 55 116 AppInfo
IproceuCCMFeatureData operatIoHeidd*© 031332 26 002 111 10 55 116 |Appi*fo
MdUnfftdInttfCtptOnPptternnumOfPaffemi *0

03833229 001 111 10 55 116 IAppirfo |bneCdpc(46) -dspatchToA' Devcei-.sifftaineaCcProceedRec. devK e=Sf P28 MA2824611

03833232 001 111 10 55 116 IAppmfo IBM F CI 28377985 ASSOC 28377986
03833232 002 111 10 55 116 IAppinfo |1BM F Ct 28377986 ASSOC 28377985

03133235 001 111 10 55 119 IAppiMo |SiPStat»oKdfc CcProceedRep«
uricodeComecledUrxodeOttplayNamts* au (ConnertedC'tpiiyName:'
03833235 002 111 10 55 119 | Appmfo ISiPStatorCdk CcProceedReo * ur codeCa ^fPartyNames
tKMCaibn<PaftyName«" raihngParfy*'* 19193741001 un^odeCaitedPartyName* tKMCa iedPart^Namea c4kdParty> 5745000

03833240 000 111 10S5 119 |SdlSg ICcRejtfid |wa<! |Cdl,100,38,1)
|SIPO{1,100.181.191 |1.1OO,251.7289 66*14 48 81 100** ||R N-H O.N I.I AV 0.7 O.D 0]
0=28377986 CI branch =0 c 1=1 C C^=8 CO*0 C k=0 C r=O CV'1459617833 rejectType?! dtmO $conn-F
FOataTypeaO op*d*O »Type »0 St*ey»O »nvoktM>O r« iu t€xp*F bpdMFTraftapArcntDatainu
CanSwpport$iP7andNHal$« TrantidrO Al c*B^Mitir*OiO Ut<rA<entOrS«<ver! Or <ODNamt*loal«' 1 Name
Un.<odeName pi 0mCa!iertd= mCa trName: TnjjtR«ifl:T muMul! Forli>fl|Erab ed=F
rnedaC8uw>OCAl*{*HffMH. m«ffffffff tDe^F. r«*F. devType*©) CAi*(v*fffffff6. m»fffffw tDev*F. r«sF, devTypc^O]

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

A. The calling search space on the calling line/device does not contain the partition of the route pattern

B. The called party number is not complete because it contains only 7 digits

C. A connectivity issue between Cisco UCM and Cisco Unified Border Element prevents Cisco UCM from receiving a response to the OPTIONS it is sending to Cisco Unified Border Element

D. There are multiple matches for the dialed number, and Cisco UCM is trying to send the call to SEP2834A2824611 rather than Cisco Unified Border Element

Answer: C

Explanation:

Question: 203

Refer to the exhibit.

' Retells Summary

Calm* P*t> lafrwmattan . Dialed Difit* - $13745000 • Match Retell ■ Route? big Pattern

Hatched Pattern Informaboe

. Called Party Number * 57 43000

- Time Zoea - Etc'OMT
- Ind Device • CSRICOOv
- Cal daeeiflcaliM • Offset
- let er Digit Timeout • NO
- Device Overtide ■ Disabled
- Outage Dial Tone • NO

Cal Flow

Route Pattern Palter*- 9 1J2 9])OOOOOC PoHtwaal Hatch Lot - DralPtae -

▶ Route Filter

ftaf awe Forced Authoruabon Code ■ No Authoruatioe Level • 0 Require Clieut Halter Code ■ No Can OatnfkatkM ■ PieTianiform Calling Party Member -1001 PreTiaaiform Called Party Number - 913743000 Catting Party Traer to me t tarts

- External Phone Number Math ■ NO
- Callleg Party Hath - ■ Prefa ■
- * CaBinqUneld Prenerrtabon ■ Default
- Cal ling Name Presentation ■ Default
- Calling Party Number - 1001 Connected Party Transformations

Catted Party Transformations

Device Type- SIPTmuk

End Owe Name n CSR1 OOOv PortRumber • Devree Stain ■ UnKnown AAR Group Nam* • AAR Calling Search Space ■ AAR Prefa Digits ■ Catt C1»S#fk*Iron ■ Uie Syttem Default Catting Party Selection ■ Originator Calergliaeld Presentation e Default CallerID £>N -

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

A. Update the SIP profile on the SIP trunk by checking “enable external presentation name and number”

B. Change the calling line ID presentation on the SIP trunk to allow.

C. Create a calling party transformation patter that matches the desired directory number and add a mask to modify the calling number

D. Create a called party transformation pattern that matches the desired directory number and add a mask to modify the calling number

E. Add a calling party transformation CSS to the SIP trunk.

Answer: C, E

Explanation:

Question: 204

Refer to the exhibit.

Sent:
INVITE sip:9811919999321453^<lsco.coa:5e68 SIP/2.B
Via: S1P/2.I/U0P 192.168.1.14:5668;branch=i9hG4bKFE45715AD
Reaote-Party-ID: "User A" <sip:44448<isco.coa*;party«callmg;screen-no;privacy-of♦ Froa: 'User A" «cip:44440cisco.coa>:tag=E141986*-D6S
To: <s>p:981191999932)453«c>sco.coa>
Date: Thu, 31 Jan 20)9 19:34:47 WT
Call-ID: 1A6EC126-24(6) 1E9-8108D631-8601BCD88C isco. coa
Supported: IBBrel,tiaer,resource-priority,replaces,sdp-anat Min-SE: 1888
Cisco-Guid: 1127492352-9888865536-6666614911-6263951114
User-Agent: Cisco-SIPGateway/105-15.4.3.S7
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER CSeq: 181 INVITE
Tiaestaap: 1548963287
Contact: <Slp:4444el92.168.1.14:5166*
Expires: 188
Al low-Events: telephone-event
Max-Forwards: 69
Session-Expires: 1886
Content-Type: applicatlon/sdp
Content-01 spoilt ion: session;hand 11 ng-required Content-Length: 668

v-e o-CiscoSysteasCCM-SIP >13068473 1 IN IP4 192.168.1.16 s-SIP Call
C=1N IP4 192.168.1.14 b*TIAS:384888 b-AS:384 t-a a
■•audio 24588 RTP/AVP 9 8 8 1)6 18 161
-Output Malted

667888: Jan 31 19:34:47.246: //16SS739/43342B866ieO/SlP/Msg/ccsipOlsplayMsg: Received:
SIP/2.6 463 Free: URI not recognised
Via: SIP/2.0/U0P 192.168.1.14:Sa66;branch*B9hG4bKFE4571SA0 Free: "User A" clip: 44446)192.168.2.1*;tag*E14198«A-065
To: <sip:98119199993214S3«K isco.co«»;

Call-ID: lA0EC126-24C611E9-B108D831-86D18CDB9clSCO.eoa
CSeq: 161 INVITE
Yiaestaap: 1548963287
Server; Provider/2.6
Content-length: 6

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

A.

voice class sip-profiles 2

request INVITE sip-header To modify "sip:(.*)@" sip:8005532447@

B.

voice translation-rule 3

rule 1/.*/ /8005532447/

C.

voice class sip-profiles 1

request INVITE sip-header Diversion modify "sip:(.*)@" "sip:8005532447@"

D.

voice translation-rule 4

rule 1/^.*\(8005532447\)/ /\1/

Answer: B

Explanation:

Question: 205

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

A. decrease the T302 timer in Service Parameters from the default value

B. enable Urgent Priority on the 9.[2-9]XXXXXX pattern

C. enable Urgent Priority on the 9.911 pattern

D. enable Device Override on both route patterns

Answer: C

Explanation:

Question: 206

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

A. voice class request sip-header diversion

B. sip-header contact modify

C. voice class request sip-header modify

D. request invite sip-header diversion modify

Answer: D

Explanation:

Question: 207

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

A. route group defined in Standard Local Route Group section of Cisco UCM service parameters

B. route group from the phone B device pool

C. route group from the phone A device pool

D. standard local route group defined in the route group configuration

Answer: C

Explanation:

Question: 208

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

A. CTI route point

B. hunt list

C. hunt pilot

D. hunt group

Answer: C

Explanation:

Question: 209

A solution for a large company is being reconfigured to optimize for cost saving. The company has an extensive global QoS-enabled network with enough bandwidth to create a converged network. Local calls are relatively inexpensive in countries the company have operations, but long distance and international calls are expensive.

Which type of configuration supported by Cisco UCM would help optimize cost control for this company?

A. standard local route groups for mobile users

B. Mobile and Remote Access

C. high complex codec support like G.729 to minimize bandwidth usage

D. end hop off

Answer: D

Explanation:

Question: 210

Refer to the exhibit.

Cisco Unified Communications Manager Dialed Number Analyzer

DNA Analysis Output — **Save the Displayed Output** [Save]

Cisco Unified Communications Manager Dialed Number Analyzer Results

[Expand All] [Collapse All]

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

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

- **Call Flow**
  - **Route Pattern :Pattern**= 9.1[2-9]XX[2-9]XXXXXX
    - **Positional Match List** =
    - **DialPlan** =
    - **Route Filter**
      - **Filter Name** =
      - **Filter Clause** =
    - **Require Forced Authorization Code**= NO
    - **Authorization Level**= 0
    - **Require Client Matter Code**= NO
    - **Call Classification** =
    - **PreTransform Calling Party Number** = 4125551212
    - **PreTransform Called Party Number** = 914125550000
    - **Calling Party Transformations**
      - **External Phone Number Mask** = NO
      - **Calling Party Mask** =
      - **Prefix** =
      - **CallingLineId Presentation** = Default
      - **CallingName Presentation** = Default
      - **Calling Party Number** = 4125551212
    - **ConnectedParty Transformations**
      - **ConnectedLineId Presentation** = Default
      - **ConnectedName Presentation** = Default
    - **Called Party Transformations**

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

A. Change the calling search space of the user's line or device.

B. Change the "Call Classification" to "OnNet" on the route pattern.

C. Ensure that the user's phone is assigned to a device pool with the correct local route settings.

D. Deselect "Block this pattern" on the route pattern.

Answer: D

Explanation:

Question: 211

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

A. Run RTMP on the Cisco router.

B. Verify that the MGCP service is running under serviceability in Cisco UCM.

C. Run the debug mgcp media command on the Cisco router.

D. Run the debug H.323 gateway command on the Cisco router.

Answer: C

Explanation:

Question: 212

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

A. It forks RTP media to the recorder.

B. It provides advanced capabilities, such as speech analytics.

C. It sets up a new SIP session

D. It identifies the participant change due to transfers during the call.

Answer: D

Explanation:

Question: 213

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

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

A. CUBE is not configured forTEHO in the specific country

B. A local route group is not added to the route pattern.

C. The users are missing this specific gateway at the device pool level.

D. Regions in Cisco UCM are not configured correctly.

Answer: C

Explanation:

Question: 214

Which elements does Cisco cloud mobility for collaboration include?

A. Cisco Webex Collaboration Cloud Services

B. Cisco Collaboration Cloud

C. Cisco Collaboration Cloud and Cisco Webex Collaboration Cloud Services

D. Cisco Collaboration Cloud and Cisco Mobile and Remote Access Collaboration Cloud Services

Answer: D

Explanation:

Question: 215

The sales department must answer phones when other sales members are not at their desks. The administrator knows that configuring Call Pickup allows the sales users to answer all the calls in the

department by pressing only the softkey. Which call pickup configuration meets this requirement?

A. Standard Call Pickup

B. Group Call Pickup

C. Other Group Call Pickup

D. Directed Call Pickup

Answer: B

Explanation:

Question: 216

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

A. Subscribe all device profiles to the Extension Mobility phone service.

B. Delete the identity trust list file from the phone(s).

C. Change the Extension Mobility URL from HHTP to HTTPS.

D. Restart the Cisco Extension Mobility and Cisco Extension Mobility application services.

Answer: A

Explanation:

Question: 217

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

A. automatic

B. default

C. on

D. off

Answer: B

Explanation:

Question: 218

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

A. Forward Hunt No Answer and Forward Hunt Busy must be enabled while Call Queuing is enabled.

B. Alter Call Queuing is enabled. Forward Hunt No Answer and Forward Hunt Busy must be disabled manually.

C. Before Call Queuing is enabled. Forward Hunt No Answer and Forward Hunt Busy must be disabled manually.

D. When Call Queuing is enabled. Forward Hunt No Answer and Forward Hunt Busy are disabled automatically.

Answer: B

Explanation:

Question: 219

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

A. 913520[012345]

B. 913520X

C. 913520[1–5]

D. 913520!

Answer: A

Explanation:

Question: 220

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

this requirement?

A. Change the “Maximum Wait Time for Desk Pickup” setting from 4 to 6

B. Change the “Maximum Wait Time for Mobility Pickup” setting from 4 to 6.

C. Change the “Maximum Wait Time for Mobility Pickup” setting from 4000 to 6000.

D. Change the “Maximum Wait Time for Desk Pickup” setting from 4000 to 6000.

Answer: C

Explanation:

Question: 221

A company is experiencing an issue where calls from the Cisco UCM via Cisco Unified Border Element to the ITSP fail with a “500 server internal error” right after a 183 session in progress is sent back from the ITSP.

What must be enabled to resolve the issue?

A. delayed offer on Cisco UCM

B. early offer at the CUBE

C. delayed offer on CUBE

D. early offer at the ITSP

Answer: B

Question: 222

An organization needs a solution to meet these requirements for its Cisco UCM system:

Modify calling party numbers for outbound calls.

Add a prefix for internal calls.

Normalize incoming calls to a consistent format.

Apply different patterns for different device pools.

Which Cisco UCM feature must be used to accomplish these goals?

A. Route patterns

B. Call routing rules

C. Transformation patterns

D. Translation patterns

Answer: D

Explanation:

Question: 223

An engineer is configuring call queuing to place callers in a queue until hunt members are available to answer them. The engineer also must set the option to automatically log off an agent that does not answer a queuing-enabled hunt pilot call.

Which Cisco UCM menu is used to accomplish this goal?

A. Call Routing > Route/Hunt > Line Group

B. Call Routing > Route/Hunt > Hunt Group

C. Call Routing > Route/Hunt > Hunt Pilot

D. Call Routing > Route/Hunt > Call Queuing

Answer: C

Explanation:

Question: 224

An engineer is configuring the Extension Mobility feature on Cisco UCM and has created the phone service by setting the service name, URL, and service type.

What is the next step in the configuration?

A. Subscribe to Extension Mobility

B. Create an Extension Mobility device profile for users

C. Configure the change credential IP phone service

D. Associate a device profile to a user

Answer: B

Explanation:

Question: 225

A voice administrator enabled secure media between multiple Cisco UCM clusters that are connected via a SIP trunk.

Which configuration parameter in the SIP trunk must be enabled to allow encrypted media traffic to traverse the SIP trunk?

A. Uncheck the RTP Allowed check box in the SIP Trunk configuration

B. Check the RTP Allowed check box in the SIP Trunk configuration

C. Check the SRTP Allowed check box in the SIP Trunk configuration

D. Check the Allow iX Application Media check box in the SIP Profile

Answer: C

Explanation:

Question: 226

Refer to the exhibit.

```
voice class tenant I
credentials number 1111 username test password 7 071B245B5D1D realm ipvoice.jp
authentication username test password 7 06120A3258
registrar ipv4:1.1.1.1 expires 120
sip-server ipv4:l.l.l.l
outbound-proxy ipv4:10.64.86.35:9057
error-passthni
```

Refer to the exhibit. A Collaboration engineer is implementing call routing and must configure the Cisco Unified Border Element to register to multiple registrars using a SIP trunk listen port.

Which code snippet completes the configuration?

A. session transport tcp

B. session transport tls

C. dtmf-relay rtp-nte

D. bind control source-interface GigabitEthernet0/0

Answer: D

Explanation:

Question: 227

Refer to the exhibit.

```
dial-peer voice 120 voip
 description ******Dial Peer Outgoing to offshore****
translation-profile outgoing OUT
 max-bandwidth 240
 destination-pattern 32T
 progress_ind alert enable 8
 progressjnd progress enable 8
 progress_ind connect enable 8
 session protocol sipv2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0.200
 dtmf-relay rtp-nte no vad
```

Refer to the exhibit. An engineer configured Call Admission Control on Cisco Unified Border Element to reject SIP calls when the aggregate bandwidth that is required for the calls exceeds the threshold. However, users complain of dropped calls when they place calls on hold during peak periods.

Which solution resolves this challenge?

A. Bind media to the loopback interface

B. Bind media to interface GigabitEthernet0/0.200

C. Enable the midcall-exceed feature on the dial peer

D. Enable the error-code-override feature on the dial peer

Answer: C

Explanation:

Question: 228

Refer to the exhibit.

```
voice register dialplan 1
 type 7940-7960-others
pattern 1 2... timeout 10 user ip
pattern 2 1234 user ip button 4
pattern 3 65...
pattern 4 1...!
            !
voice register pool 1
id mac 0016.9DEF.1A70
type 7961GE
number 1 dn 1
number 2 dn 2
dtmf-relay rtp-nte
codec g711ulaw
```

Refer to the exhibit. An engineer is implementing the Cisco UCME and must set up a dial plan for the 7961 SIP phones to recognize digit strings dialed by users.

Which code snippet completes the configuration?

A. "button 1:1" under voice register pool

B. "number 1 dialplan 1" under the voice register pool

C. "number 1 dp 1" under the voice register pool

D. "dialplan 1" under the voice register pool

Answer: A

Explanation:

Question: 229

An engineer uses the standard local route group to configure call routing for the organization. The engineer configured an additional local route group with a unique name.

Which action must be taken to ensure that each device in the system is provisioned to know its local route group?

A. Configure route filters

B. Add the local route group to a route list

C. Associate the local route group with a device pool

D. Add the local route group to a translation pattern

Answer: C

Explanation:

Question: 230

An administrator is attempting to set up redundancy to allow users to make calls even if all Cisco UCM nodes go down or are unreachable by the IP phones.

Which feature must the administrator configure to accomplish this goal?

A. Survivable Remote Site Telephony

B. Cross-Origin Resource Sharing

C. Cisco UCM Redundancy Group

D. Hot Standby Router Protocol

Answer: A

Explanation:

Question: 231

A company is using a H.323 gateway for PSTN calls from Cisco phones connected to a Cisco UCM cluster. There are reports of problems getting audio on these incoming calls from the PSTN. The phone rings, but nothing is heard and the call disconnects.

Which command troubleshoots this issue?

A. debug voice h323 inout

B. debug h245 pstn

C. debug h245 asn1

D. debug voice sip inout

Answer: C

Explanation:

Question: 232

A company is using third-party phones on the internal network at a location behind a Cisco Unified Border Element. These phones seem to work with only two types of codecs, which are high- complexity codecs.

Which configuration ensures that calls are completed with the correct and working media each time?

A. Configure a codec preference list on the Cisco Unified Border Element with the voice class codec command, set the codec preference list codecs, and add it to the dial peer.

B. Configure a codec priority list on the Cisco Unified Border Element with the voice class codec command, set the codec priority list codecs, and add it to the dial-peer voice.

C. Configure a codec preference list on the Cisco Unified Border Element with the VoIP class codec command, set the codec preference list codecs, and add it to the dial peer.

D. Configure a codec preference list on the Cisco Unified Border Element with the voice class codec command, set the codec priority list codecs, and add it to the VoIP peer.

Answer: B

Explanation:

Question: 233

A Collaboration engineer wants to deploy Global Dial Plan Replication for advertising its global dial plan data, including the global dial plan data that was configured locally and any data that was learned from other clusters. The engineer is configuring the inter-cluster lookup service on the Cisco UCM and activated the ILS service and configured the cluster IDs.

What is the next step in the configuration?

A. Activate ILS on the spoke cluster.

B. Configure remote clusters.

C. Enable TLS authentication between clusters.

D. Activate ILS on the hub cluster.

Answer: B

Explanation:

Question: 234

An engineer is reviewing the dial peer configuration on the Cisco Unified Border Element after users reported that DTMF digits are not recognized when they make an outbound SIP call to a phone system that supports only out-of-band DTMF.

Which configuration under outbound dial peer resolves the issue?

A. dtmf-relay sip-notify

B. dtmf-relay h245-signal

C. dtmf-relay rtp-nte

D. dtmf-relay h245-alphanumeric

Answer: A

Explanation:

Question: 235

Refer to the exhibit.

Refer to the exhibit. An engineer troubleshoots why a new SIP trunk does not come up on Cisco UCM. Which action resolves this issue?

A. The device cube.abc.com must have the SIP stack turned up.

B. The destination port must be set to 5061.

C. The ACL that is configured on cube.abc.com must include the Cisco UCM IP address.

D. The DNS server that Cisco UCM is pointing to needs an A Record added for cube.abc.com.

Answer: D

Explanation:

Question: 236

Users report that calls to some locations do not work. The phone is calling, the recipient picks up, but it does not connect. These users are in the headquarters and registered to the company Cisco UCM with default values for calling (G.711). The problem locations are smaller remote locations with low bandwidth capacity, so the phones are configured with iLBC.

Which command should be used to troubleshoot the issue?

A. debug dsp all

B. show dsp active

C. show dspfarm dsp active

D. debug voice dspfarm

Answer: C

Explanation:

Question: 237

Users of a Cisco UCME report that calls lose audio after they are forwarded to some users. Calling point-to-point seems to function fine, but depending on the type of phone, some calls lose audio when forwarded. Looking at the trace header that shows:

Allow: INVITE,BYE,CANCEL,ACK,INFO,PRACK,COMET,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER

What is missing and is a reason that the SRTP stream stops working?

A. EARLY-OFFER

B. FORWARD

C. UPDATE

D. REDIRECT

Answer: C

Explanation:

Question: 238

An engineer is configuring Cisco UCME to support SIP endpoints. The engineer wants to limit the system to 20 SIP phones and 50 directory numbers. Which code completes this configuration?

A.

telephony-service

mode cme

max-ephone 20

max-dn 50

B.

voice register global

mode cme

max-ephone 20

max-dn 50

C.

telephony-service

mode cme

max-pool 20

max-dn 50

D.

voice register global

mode cme

max-pool 20

max-dn 50

Answer: B

Explanation:

Question: 239

Some Cisco mobility users cannot resume calls on the desk phone after they hang up on a mobile phone. The users have Cisco IP phones registered to the Cisco UCM. Which set of configuration steps resolve the issue?

A. Add voice call disc-pi-off on the voice gateway. On Call Manager, set the Retain Media on Disconnect with PI for Active Call service parameter to True.

B. Add voice call disc-pi-on on the voice gateway. On Call Manager, set the Retain Media on Disconnect with PI for Active Call service parameter to False.

C. Add voice call disc-pi-off on the voice gateway. On Call Manager, set the Retain Media on Disconnect with PI for Active Call service parameter to False.

D. Add voice call disc-pi-on on the voice gateway. On Call Manager, set the Retain Media on Disconnect with PI for Active Call service parameter to True.

Answer: A

Explanation:

Question: 240

A company wants to optimize the ability to answer the phone for customers that call different departments. Each of the three departments consists of five people. The company needs a simple round-robin style queue for each department to ensure that the phone rings to a new person after four rings each. What is the most efficient configuration to meet this requirement?

A. Configure three forward groups for each line in each department.

B. Configure three hunt pilots and set them up with round robin.

C. Configure three hunt groups and set them up with round robin.

D. Configure three Cisco Unified Contact Center Express agents and configure round robin.

Answer: C

Explanation:

Question: 241

A user has a problem calling a specific PSTN number. An engineer investigates the issue on a Cisco UCM server and a Cisco Unified Border Element router, and discovers that the ITSP rejects the outbound SIP INVITE with a 422 Session Timer Too Small SIP message. What must the engineer do to
resolve the issue?

A. Decrease the SIP Session Expires Timer service parameter in Cisco UCM.

B. Decrease the Min-SE SIP header timer on the appropriate Cisco Unified Border Element dial peer.

C. Increase the Min-SE SIP header timer on the appropriate Cisco Unified Border Element dial peer.

D. Increase the SIP Session Expires Timer service parameter in Cisco UCM.

Answer: C

Explanation:

Question: 242

Refer to the exhibit.

Refer to the exhibit. During a troubleshooting session, an engineer was presented with this call flow output diagram after a user reported issues with a call via the SIP ITSP to a counterpart. What is occurring with this call flow?

A. The user is experiencing no-way audio after resuming the call from hold.

B. The user is experiencing one-way audio after resuming the call from hold.

C. The user call was terminated due to network issues.

D. The user call proceeded normally after resuming from hold.

Answer: A