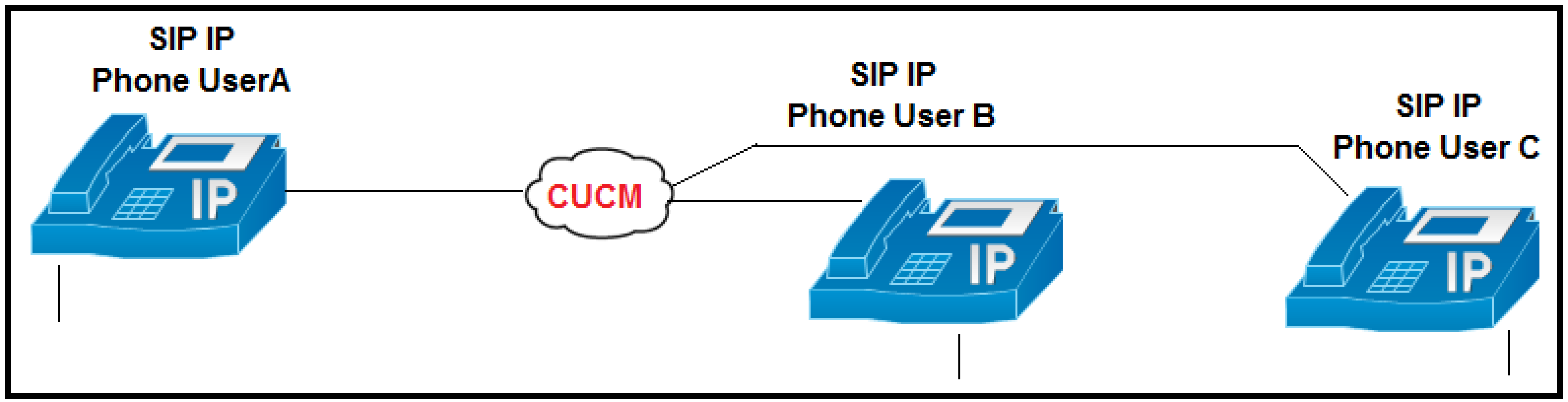
Actual exam question from Cisco's 300-815

Question #: 1

Topic #: 1

[All 300-815 Questions]



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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 2

Topic #: 1

[All 300-815 Questions]

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

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

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

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

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

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

D. No DTMF is negotiated.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 3
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 4

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 5

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 6
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 7

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 8

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 9
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 10
Topic #: 1
[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 11

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 12

Topic #: 1

[All 300-815 Questions]

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).

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 13
Topic #: 1
[All 300-815 Questions]

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

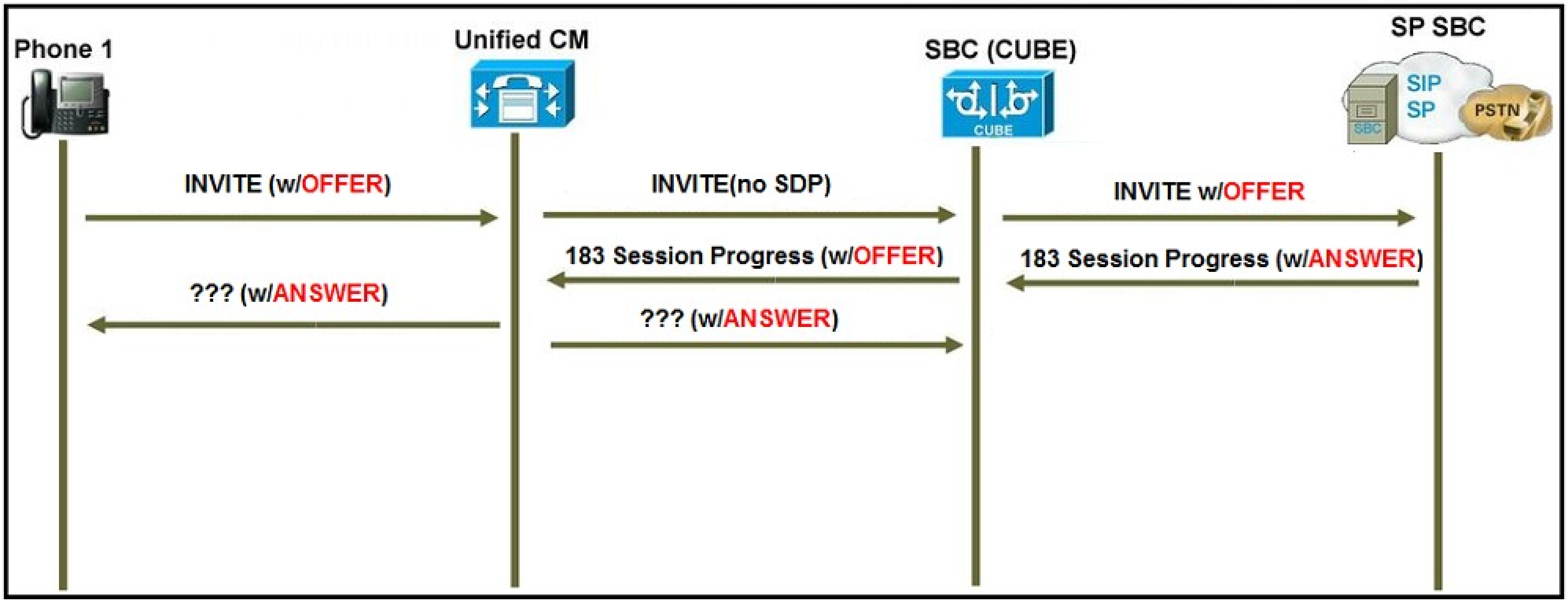
B. CTI Manager traces

C. Cisco IP Manager Assistant

D. Call logs

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 14
Topic #: 1
[All 300-815 Questions]



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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 15

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 16

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 17
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 18
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 19

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 20

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 21

Topic #: 1

[All 300-815 Questions]

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

Refer to the exhibit. 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".

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 22

Topic #: 1

[All 300-815 Questions]

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

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

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

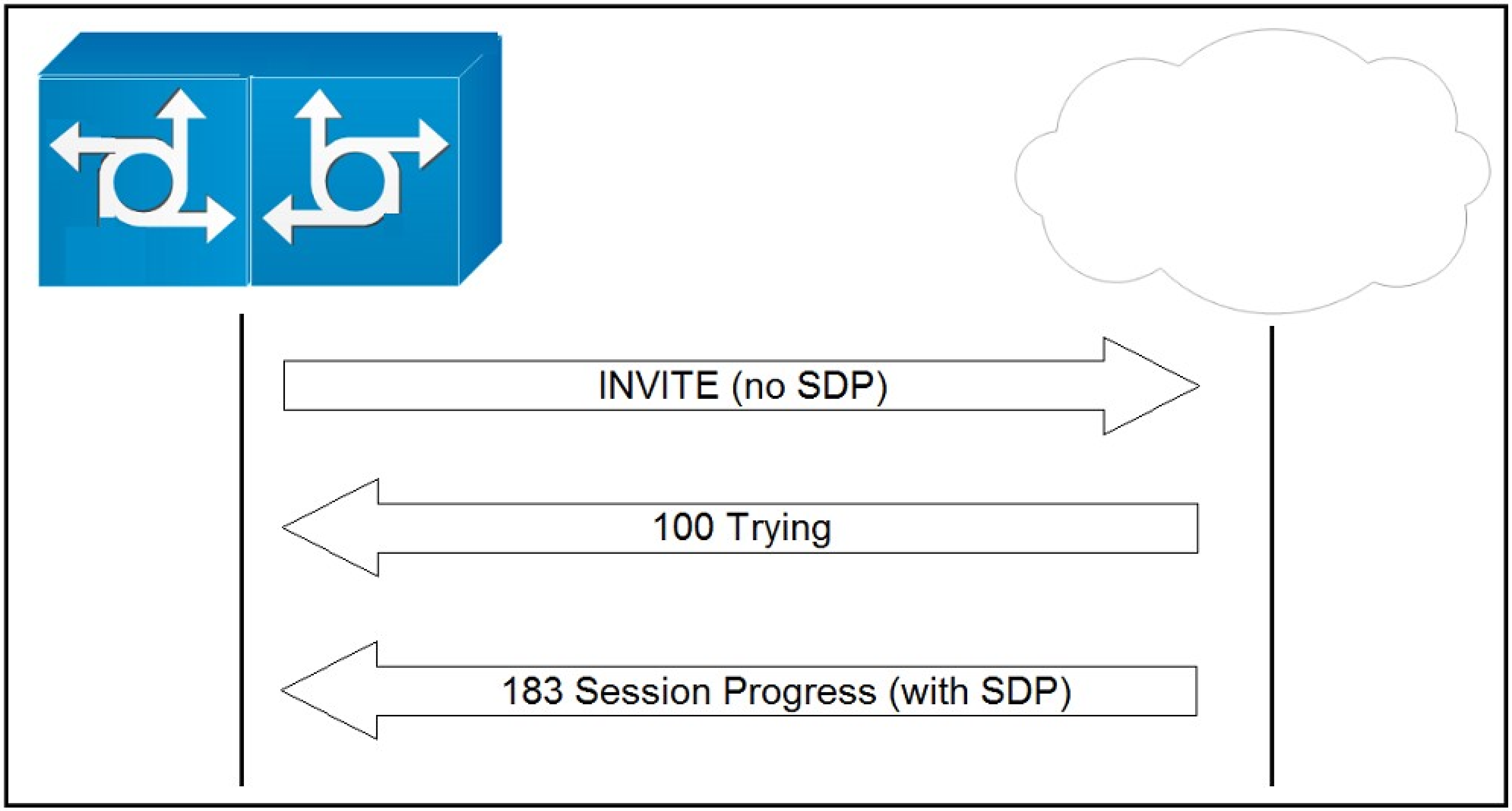
B. rule 1/^\+1\([2-9]..[2-9]"¦"¦$\)/ /\1/

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 23
Topic #: 1
[All 300-815 Questions]



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

A. Enanle PRACK.

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

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

D. Enable Media Flow Around

E. Enable Mid-Call Signaling Consumption.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 24

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 25

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 26

Topic #: 1

[All 300-815 Questions]

An engineer must route all SIP calls in the form of <user>@example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

A. example.com@gateway.corporate.local

B. *@example.com

C. gateway.corporate.local

D. example.com

E. *.*

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 27

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 28

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 29

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 30

Topic #: 1

[All 300-815 Questions]

```
!

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

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

A. answer-address 555 "¦"¦..

B. codec g729

C. session-protocol sipv2

D. incoming called number 555"¦"¦.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 31
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 32
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 33

Topic #: 1

[All 300-815 Questions]

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

A. MTP

B. CCSIP

C. RTMT

D. OS Administration Page

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 34
Topic #: 1
[All 300-815 Questions]

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]

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 35

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 36

Topic #: 1

[All 300-815 Questions]

Route Patterns ***(1–5 of 5)***

Find Route Patterns where Pattern begins with Find Clear Filter

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

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

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 37

Topic #: 1

[All 300-815 Questions]

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

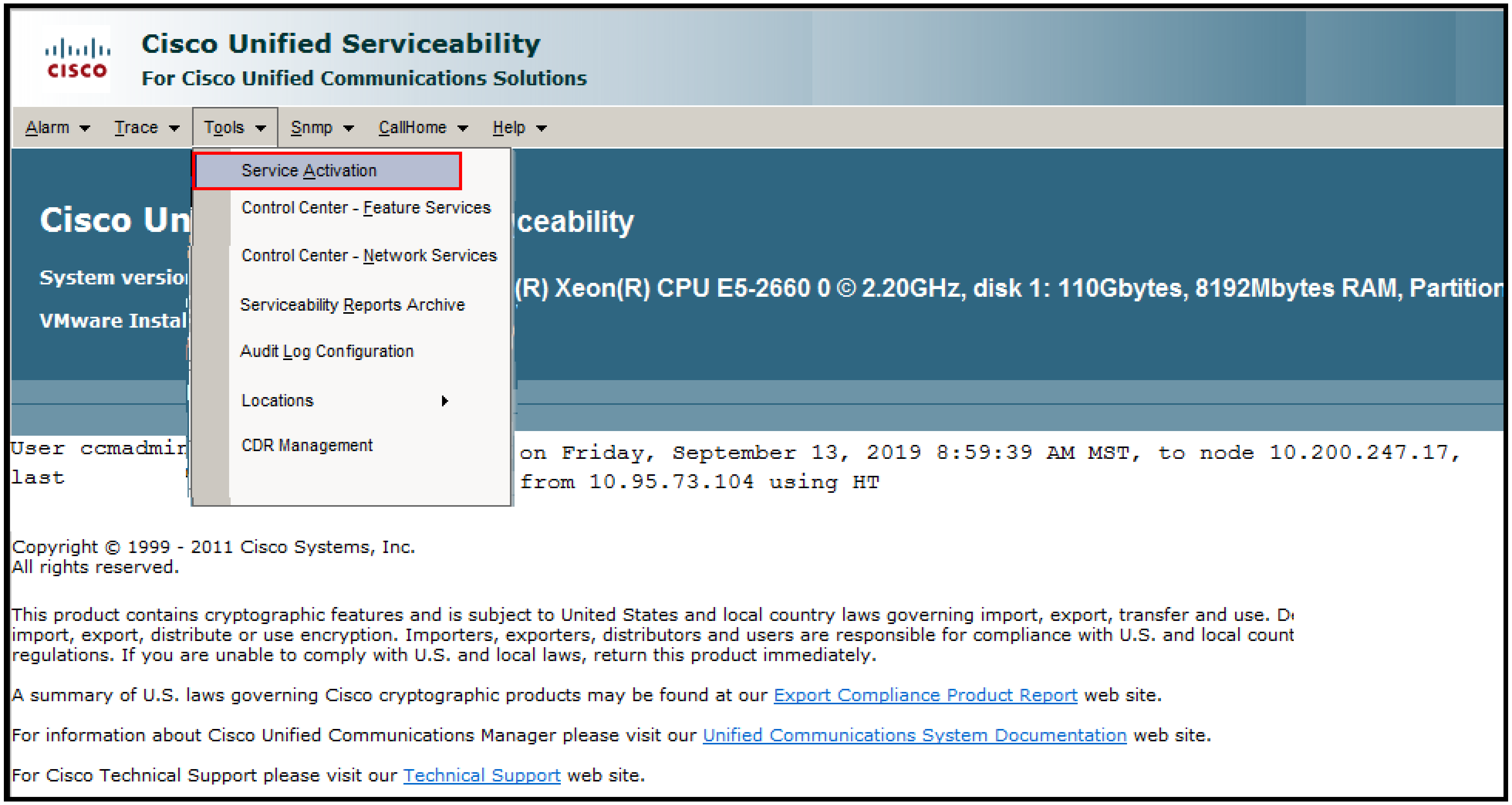
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 38

Topic #: 1

[All 300-815 Questions]



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

A. Restart the subscriber

B. Activate the Cisco Extended Functions service.

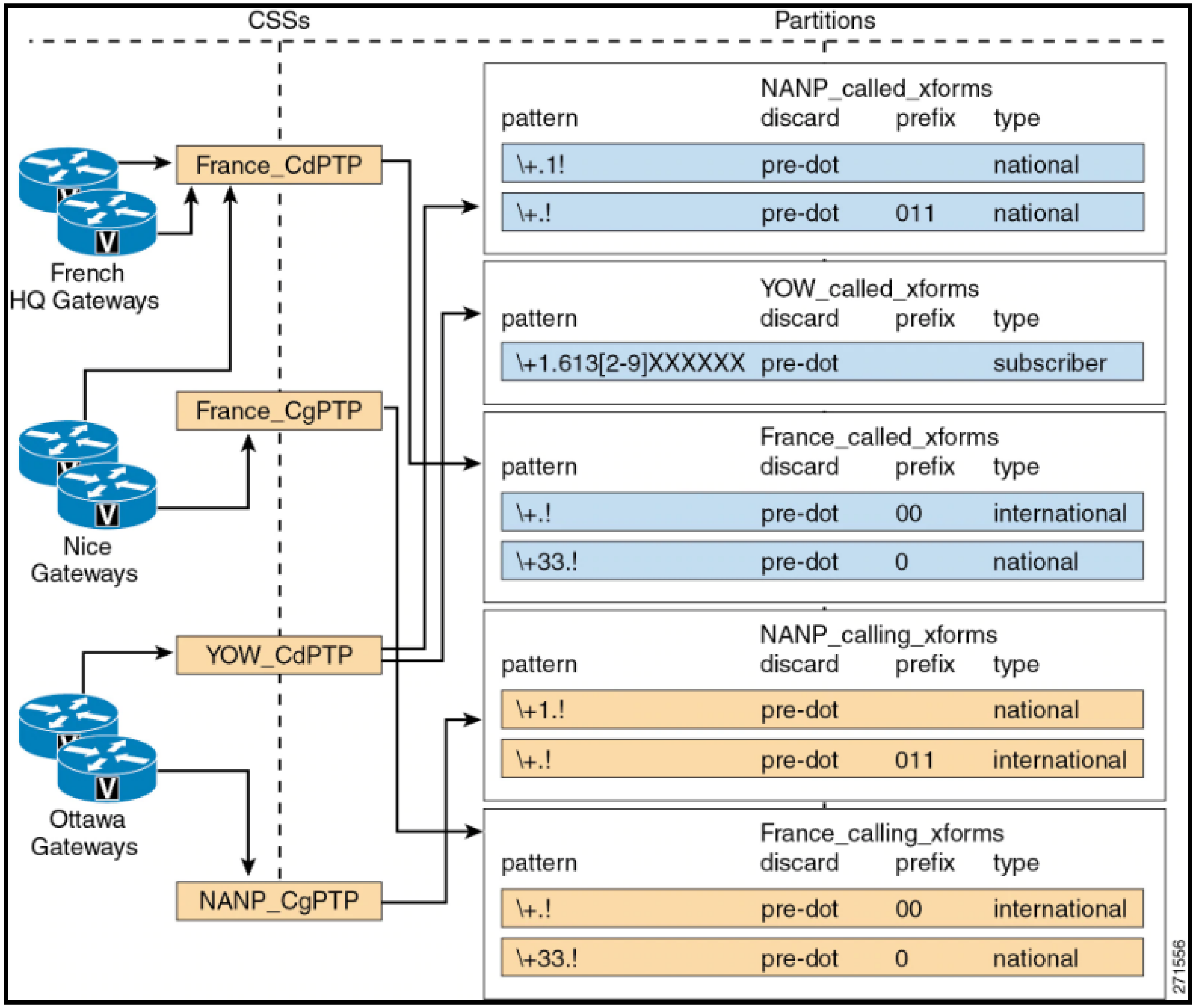
C. Activate the Cisco CallManager service.

D. Activate the Cisco Dialed Number Analyzer service.

E. Activate the Cisco Dialed Number Analyzer Server service.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 40
Topic #: 1
[All 300-815 Questions]



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"

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 41
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 42

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 43

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 44
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 45
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 46

Topic #: 1

[All 300-815 Questions]

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

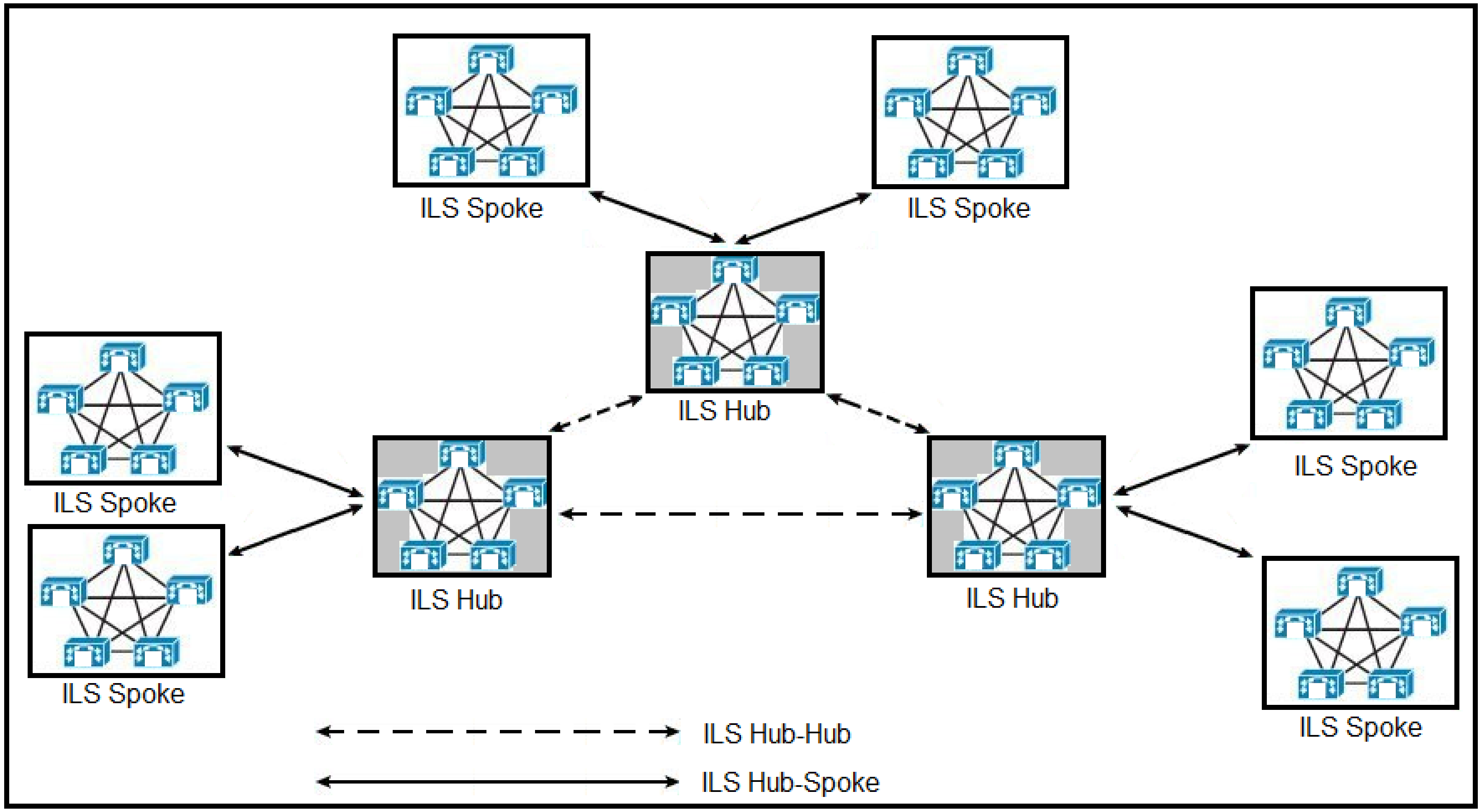
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 47

Topic #: 1

[All 300-815 Questions]



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

A. 3

B. 6

C. 9

D. 12

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 48
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 49

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 50

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 51

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 52

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 53

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 54
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 55

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 56

Topic #: 1

[All 300-815 Questions]

What is a component of Cisco Unified Mobility?

A. Unified IVR

B. Mobile Connect

C. Smart Client Support

D. Single Number Connect

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 58

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 59
Topic #: 1
[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 60

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 61
Topic #: 1
[All 300-815 Questions]

```
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
```

Refer to the exhibit. While troubleshooting call failures on the Cisco Unified Border Element, an administrator notices that messages are being sent to the service provide, 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. Require: 100rel

B. Content-Type: application/sdp

C. Contact:

D. Content-Disposition: session;handling=required

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 62
Topic #: 1
[All 300-815 Questions]

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

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

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

D. Enable SIP Rel1XX Options on the SIP profile.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 63

Topic #: 1

[All 300-815 Questions]

```
Received
UPDATE sip:192.168.100.101:5060;transport=udp SIP/20.0
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 are 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-passing

D. rel1xx require "100rel"

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 64

Topic #: 1

[All 300-815 Questions]

```
SIPHandler/ccbId=0/scbId=0/wait_SIPTimer: TimerExpired
type=SIP_TIMER_WAIT_CONNECT value=5000 retries=0
Stack/Transport/0x0xee9c8980/sipTransportPostInternalMsg: Posting Internal Msg
type=1
Stack/Transport/0x0/sipTransportPostCloseConnection: Posting TCP conn close for
addr=10.10.5.11, port=5060, connid=20
Stack/Transport/0x0/sipDeleteConnInstance: Deleted conn=0xe7ac06c0, connid=20,
addr=10.10.5.11, port=5060, transport=ICP
Stack/Info/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type returned:
2 (SIP_NETWORK_MSG), for event 64 (SIPSPI_EV_INTERNAL_MSG)
Stack/Error/0x0xee9c8950/sipTransportPostSendFailure: Posting send failure msg
with tcb: (nll) reason=4
Stack/Infor/0x0/ccsip_process_sipspi_queue_event: ccsip_spi_get_msg_type
returned: 2 (SIP_NETWORK_MSG), for event 55 (SIPSPI_EV_SEND_FAILURE_MSG)
Stack/Info/0x0xee9c8980/ccsip_spi_process_event: Send Error for event
(0xee9cb8b0)
Stack/Error/0x0/act_idle_send_msg_failure: Send Error to 10.10.5.11:5060 for
transport TCP
Stack/Info/0x0xee9c8980/ccsip_set_oo_cause_for_spi_err: Categorized cause: 38,
category:186
Stack/Info/0x0xee9c8980/sipSPIInitiateDisconnect: Initiate call disconnect (38)
for outgoing call
SIPHandler/ccbId=22609/scbId=0/ccsip_api_call_disconnected: ccb->cc_disc_cause
(38): ccb->sip_disc_cause (503)
SIPHandler/ccbId=22609/scbId=0/findDevicePID: Routed to SIPD by ccbId/scbId
Stack/States/0x0xee9c8980/sipSPIChangeState: 0xee9c8980 : State change from
(STATE_IDLE, SUBSTATE_NONE) to (STATE_DISONNECTING, SUBSTATE_NONE)
```

Refer to the exhibit. 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 must associate the route pattern with a calling search space the device can dial.

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 66
Topic #: 1
[All 300-815 Questions]

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=

Refer to the exhibit. 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:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" ;tag=32412716~41f7
To:
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" ;tag=32412716~41f7
To:
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 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" ;tag=32412716~41f7
To:
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces

Min-SE: 1800 -

User-Agent: Cisco-CUCM12.0 -

D. 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" ;tag=32412716~41f7
To:
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

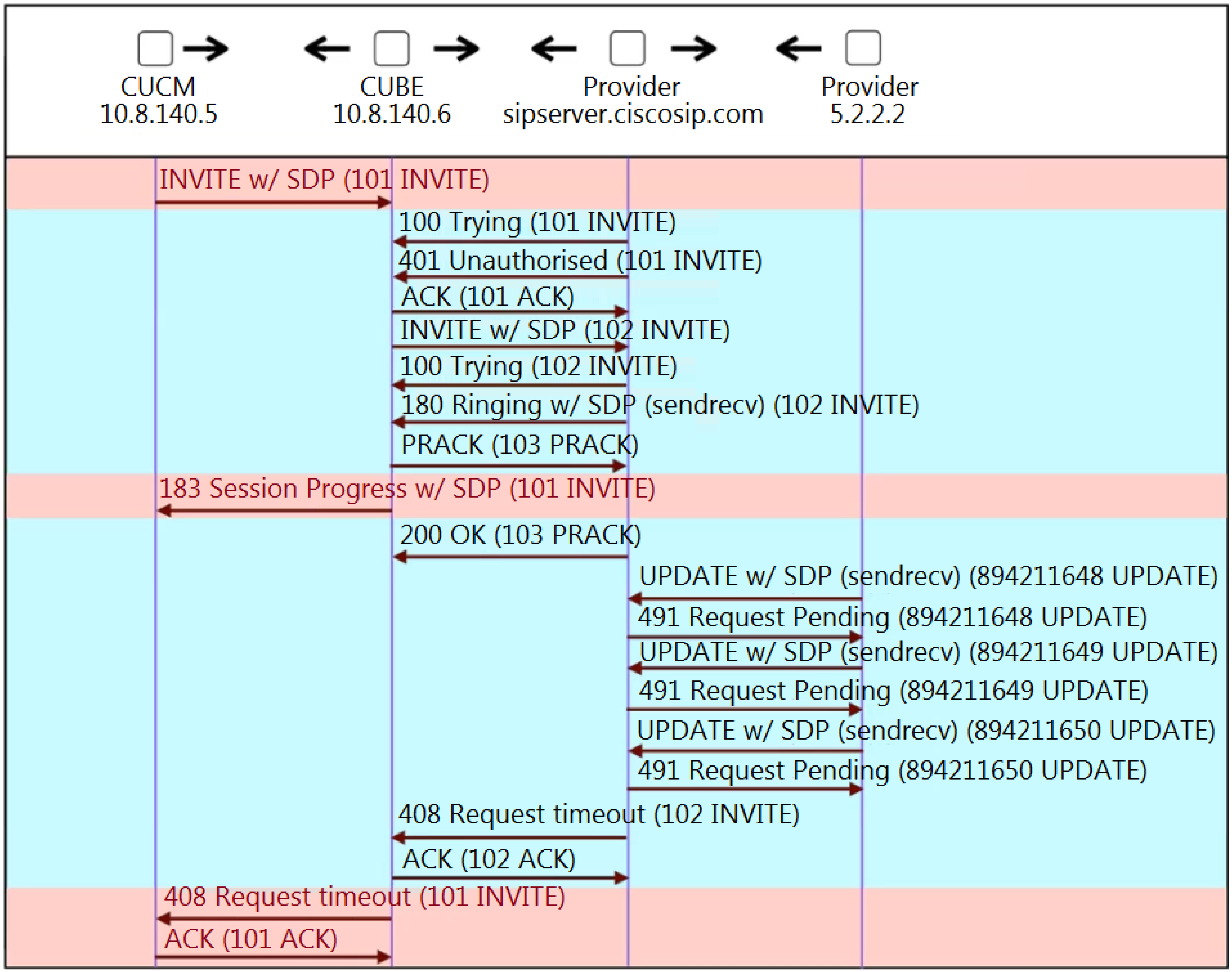
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 67

Topic #: 1

[All 300-815 Questions]



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. Disable "SIP Rel1XX Options" and "Early Offer Support" on the SIP Profile Configuration Page in Cisco UCM.

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 68
Topic #: 1
[All 300-815 Questions]

```
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=47B5F70-438
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
```

Refer to the exhibit. 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
no rel1xx

B. sip-ua
disable-early-media 180

C. voice service voip
sip
rel1xx require 100rel

D. voice service voip
sip
send 180 sdp

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 69
Topic #: 1
[All 300-815 Questions]

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.)

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

B.
```
ephone 1
 mac-address 1111.2222.3333
 type 8941
 button 1:2
```

C.
```
ephone-dn 2
 number 2050
```

D.
```
voice register dn 2
 number 2050
```

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 70
Topic #: 1
[All 300-815 Questions]

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. “voice register pool ” configuration mode

B. “ephone-dn ” configuration mode

C. “dial-peer cor custom ” configuration mode

D. “voice register global ” configuration mode

E. “telephony-service ” configuration mode

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 71
Topic #: 1
[All 300-815 Questions]

A customer has multisite deployments with a globalized dial plan. The customer wants to route PSTN calls via the gateway assigned to each site. Which two actions will fulfill the requirement? (Choose two.)

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

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 72

Topic #: 1

[All 300-815 Questions]

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

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

B. Router(conf-voi-serv)# no disable-early-media 180

C. Router(conf-voi-serv)# disable-early-media 180

D. Router(config-sip-ua)# disable-early-media 180

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 73
Topic #: 1
[All 300-815 Questions]

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

A. broadcast

B. top down

C. longest idle time

D. circular

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 74
Topic #: 1
[All 300-815 Questions]

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

A. voice service voip
enable ip address trust list

B. voice service voip
ip address trusted list

C. voice service voip
ip address trusted authenticate

D. voice service voip
enable ip address trust authentication

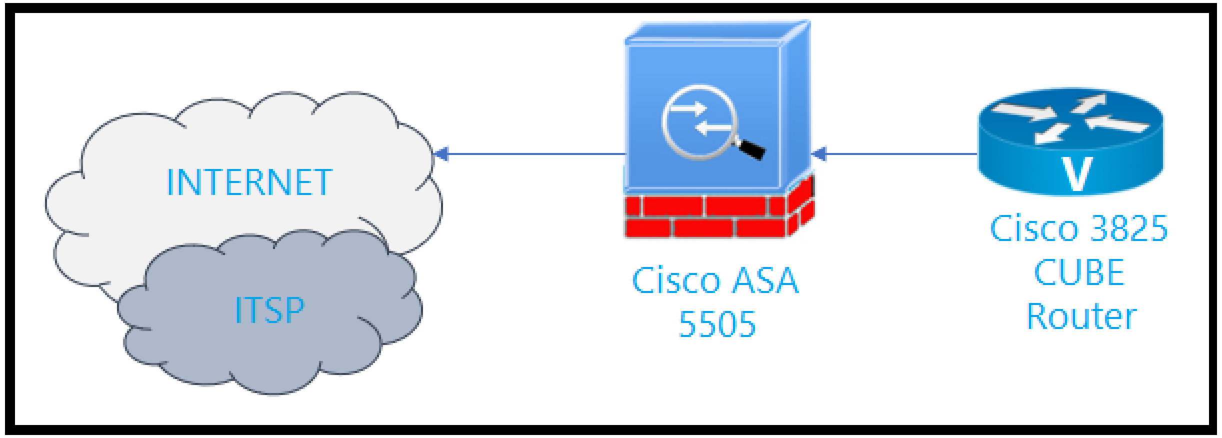
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 75

Topic #: 1

[All 300-815 Questions]



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. bind signaling and media to the loopback interface

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

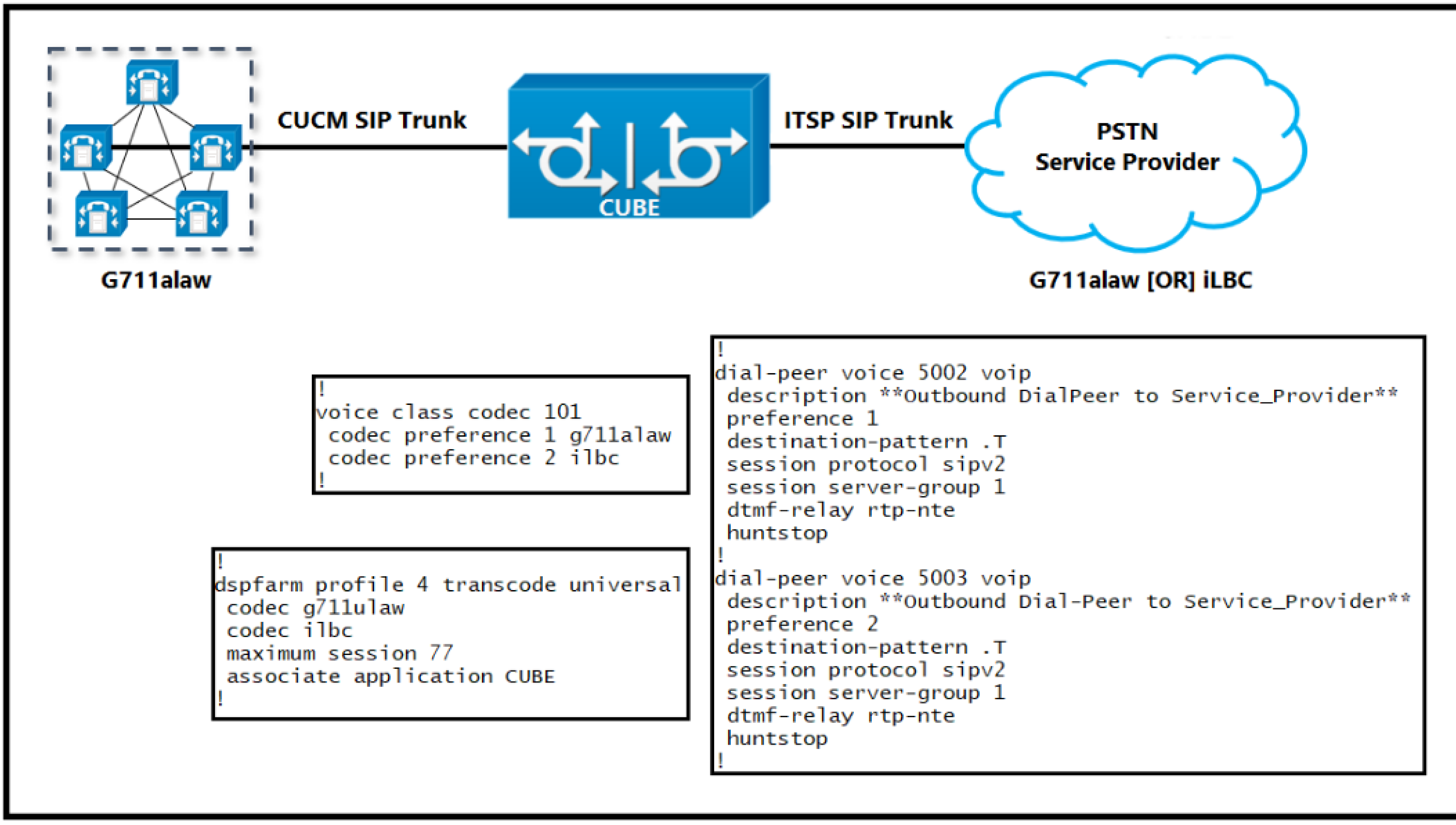
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 76

Topic #: 1

[All 300-815 Questions]



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?

A. dial-peer voice 5002 voip
codec g711alaw ilbc
!
dial-peer voice 5003 voip
codec g711alaw ilbc

B. dial-peer voice 5002 voip
voice-class codec 101 offer-all
!
dial-peer voice 5003 voip
voice-class codec 101 offer-all

C. dial-peer voice 5002 voip
voice-class codec 101
!
dial-peer voice 5003 voip
voice-class codec 101

D. dial-peer voice 5002 voip
codec g711alaw
!
dial-peer voice 5003 voip
codec ilbc

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 78

Topic #: 1

[All 300-815 Questions]

```
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
```

Refer to the exhibit. 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.729r8

B. G.711 A-law

C. G.711 U-law

D. G.729br8

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 80
Topic #: 1
[All 300-815 Questions]

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 uri to

B. dial-peer voice 1 voip
incoming called-number

C. dial-peer voice 1 voip
incoming uri via

D. dial-peer voice 1 voip
incoming uri request

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 81
Topic #: 1
[All 300-815 Questions]

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 rtp-nte

B. dtmf-relay cisco-rtp

C. dtmf-relay h245-signal

D. dtmf-relay sip-kpml

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 82

Topic #: 1

[All 300-815 Questions]

```
!
dial-peer voice 10 voip
 description Inbound
 session protocol sipv2
 incoming called-number 2000
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 20 voip
 description Outbound
 destination-pattern 2.
 session protocol sipv2
 session target ipv4:192.168.100.101
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0
            AD                              PRE  PASS  SESS-SER-GRP\    OUT
TAG   TYPE  MIN   OPER PREFIX   DEST-PATTERN   FER  THRU  SESS-TARGET      STAT PORT   KEEPALIVE  VRF
10    voip  up    up                         0    syst                                         NA
20    voip  up    up            2.           0    syst  ipv4:192.168.100.101          busyout  NA
```

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

A. The Cisco Unified Border Element is not receiving a response to its OPTION keepalives.

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

C. VAD was not disabled on the outgoing dial peer.

D. No codecs are configured on the dial peers.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 83

Topic #: 1

[All 300-815 Questions]

```
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:8005532447010.10.150.1:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.150.11:5060;branch-z9hG4bK1046d36216b0de
From: <sip:1001010.10.150.11>;tag-23125042-8a7bedal-fb5d-4d82-bdb6-4b07a7393aff-27428388
To: "CISCO SYSTEMS" <sip:8005532447010.10.150.1>;tag=D974B182=FAS
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02010.10.150.1
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 103 INVITE
[..Omitted for brevity..]
Contact: <sip:1001010.10.150.11:5060>;
Content-Type: application/adp
Content-Length: 235

v-0
o-CiscoSystemsCCM SIP 23125042 1 IN IP4 10.10.150.11
s-SIP Call
c-IN IP4 10.10.2.254
b-TIAS:64000
b-AS:64
t-0 0
m-audio 35023 RTP/AVP 0 101
a-ptime:20
a-rtpmap:0 PCMU/8000
a-rtpmap:101 telephone-event/8000
a-fmtp:101 0-15

  Calling Number-1001,(Calling Name-) (TON-Unknown, NPI-Unknown, Screening-User, Passed,
  Called Number-8005532447(TON-Unknown, NPI-Unknown),
  Calling Translated-FALSE, Subscriber Type Str-Unknown, FinalDestinationFlag-FALSE,
  Incoming Dial-peer-100, Progress Indication-NULL(0), Calling IE Present-TRUE,
```

Refer to the exhibit. 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 GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1

B. SIP binding in dial-peer configuration mode:
dial-peer voice 100 voip
voice-class sip bind control source-interface GigabitEthernet0/0/0 voice-class sip bind media source-interface GigabitEthernet0/0/0

C. SIP binding in dial-peer configuration mode:
dial-peer voice 300 voip
voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1

D. SIP binding in SIP configuration mode:
voice service voip
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 85
Topic #: 1
[All 300-815 Questions]

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

A. midcall-signaling passthru

B. no update-callerid

C. midcall-signaling passthru media-change

D. midcall-signaling preserve-codec

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 86

Topic #: 1

[All 300-815 Questions]

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. intercluster trunk

B. directory URI partition

C. SIP route pattern

D. calling search space and partition

E. SIP trunk

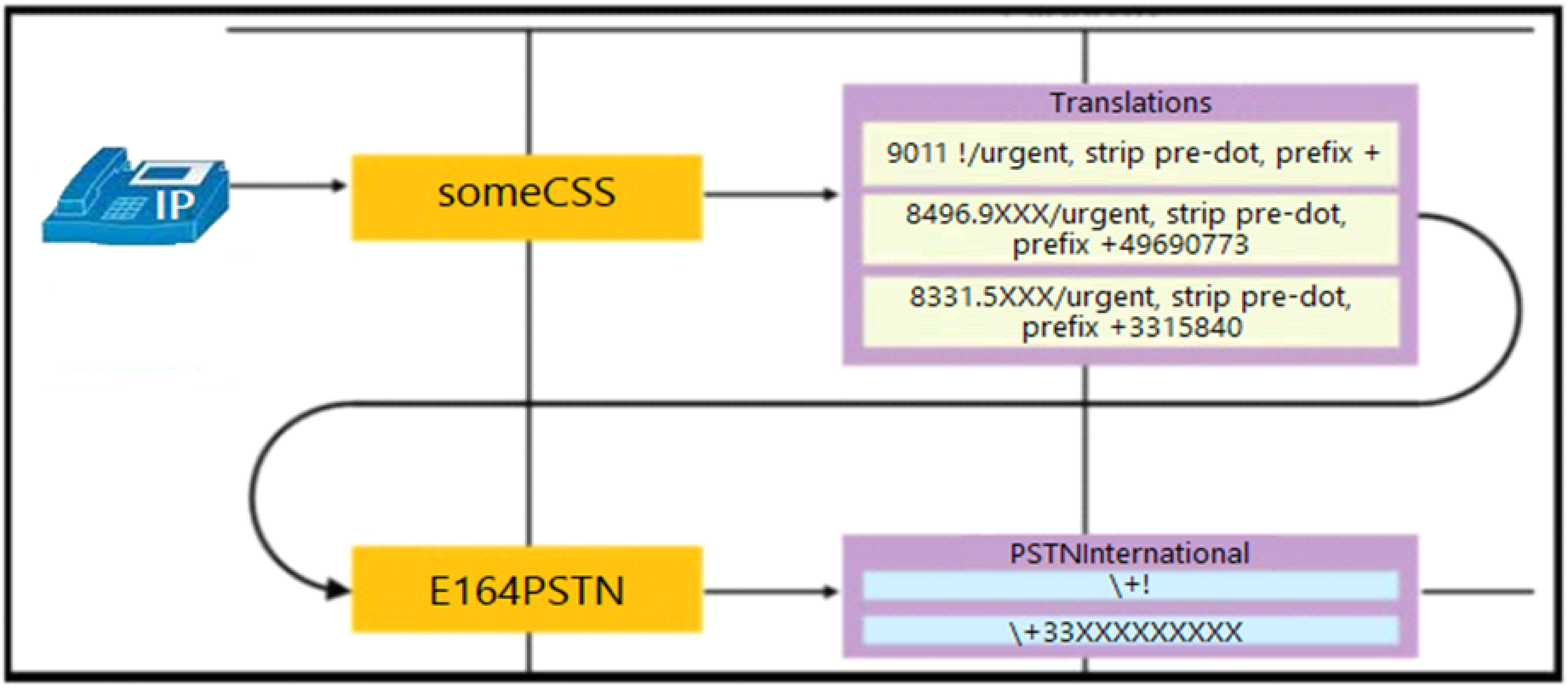
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 87

Topic #: 1

[All 300-815 Questions]



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

A. Allow Device Override on the route pattern

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 88

Topic #: 1

[All 300-815 Questions]

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 SIP Option Ping on the SIP profile.

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

C. Enable ANAT on the SIP profile.

D. Enable transmit security status on the SIP security profile.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 89

Topic #: 1

[All 300-815 Questions]

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

A. Configure a voice translation profile to map the E.164 number to four digits and assign it to the incoming dial-peer on Cisco Unified Border Element.

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

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

D. 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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 90

Topic #: 1

[All 300-815 Questions]

Building A

Results Summary
- Calling Party Information
- Dialed Digits = 9195552388
- Match Result = RouteThisPattern
- Matched Pattern Information
- Called Party Number = 9195552388
- Time Zone = Etc/GMT
- End Device = PSTN_RL
- Call Classification = OffNet
- InterDigit Timeout = NO
- Device Override = Disabled
- Outside Dial Tone = NO

Call Flow
- Route Pattern: Pattern = [2–9]XX[2–9]XXXXXX
- Route List: Route List Name = PSTN_RL
- RouteGroup:RouteGroupName = Standard Local Route Group (RTP_trunks)
- PreTransform Calling Party Number = 2304
- PreTransform Called Party Number = 9195552388
- Calling Party Transformations
- Called Party Transformations
- Device :Type = SIPTrunk

Building B

Results Summary
- Calling Party Information
- Dialed Digits = 9195552388
- Match Result = RouteThisPattern
- Matched Pattern Information
- Called Party Number = 9195552388
- Time Zone = Etc/GMT
- End Device = PSTN_RL
- Call Classification = OffNet
- InterDigit Timeout = NO
- Device Override = Disabled
- Outside Dial Tone = NO

Call Flow
- Route Pattern: Pattern = [2–9]XX[2–9]XXXXXX
- Route List: Route List Name = PSTN_RL
- RouteGroup:RouteGroupName = Standard Local Route Group
- PreTransform Calling Party Number = 2305
- PreTransform Called Party Number = 9195555388
- Calling Party Transformations
- Called Party Transformations

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

A. 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.

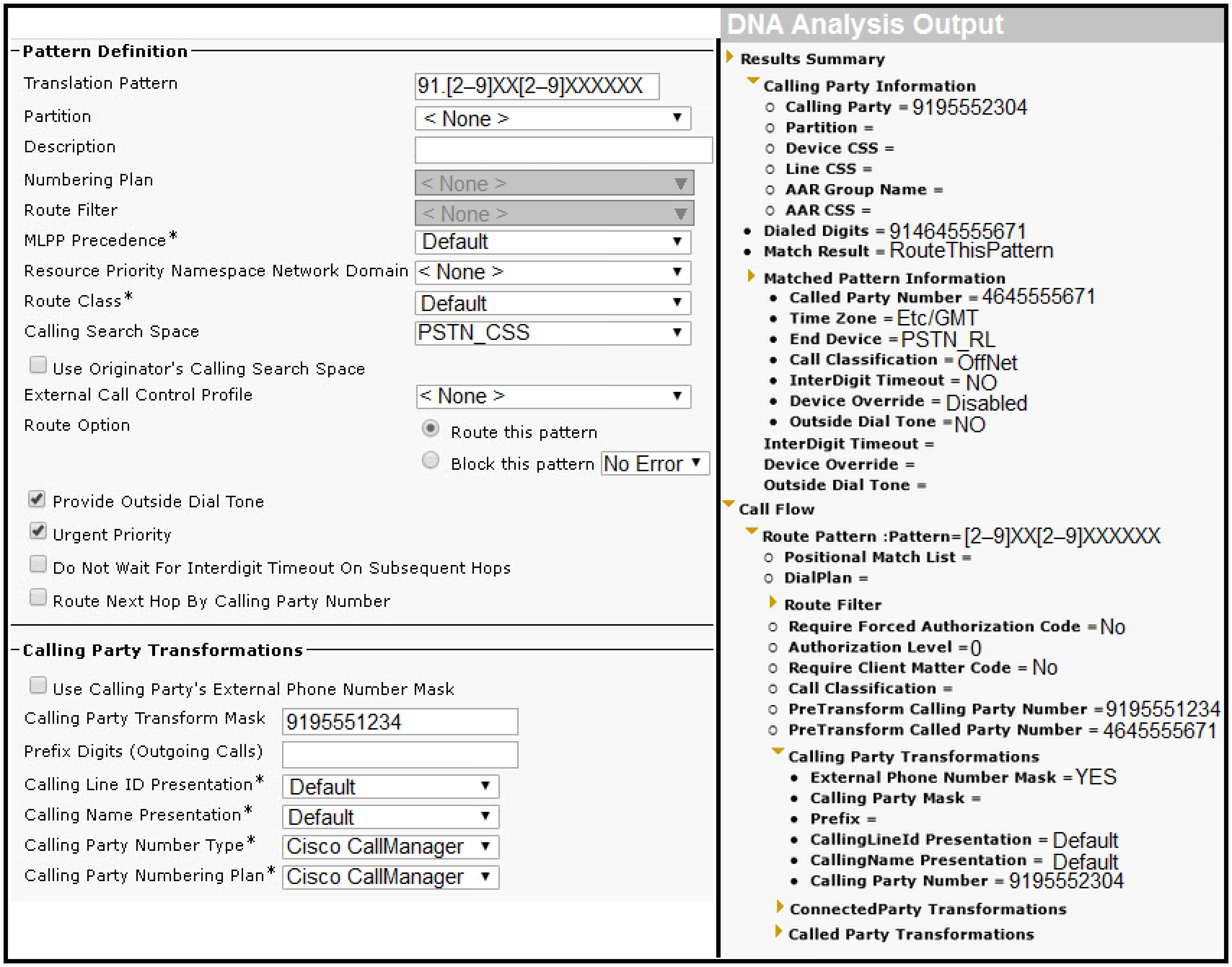
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 91

Topic #: 1

[All 300-815 Questions]



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

C. Enable Force Authorization Code on the route pattern.

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 92

Topic #: 1

[All 300-815 Questions]

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

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 93

Topic #: 1

[All 300-815 Questions]

```
CUBE_Router#conf t
Enter configuration commands, one per line. End with CNTL/Z.
CUBE_Router(config)#voice translation-rule 999
CUBE_Router(cfg-translation-rule)#rule 1 /^9(.*)/ //
CUBE_Router(cfg-translation-rule)#end
CUBE_Router#
CUBE_Router#test voice translation-rule 999 9123548
9123548 Didn't match with any of rules
```

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

A. rule 1 /^9\(\d*\)/ /\l/

B. rule 1 /^9\(.*\)/ /\l/

C. rule 1 /.*\(3548$\)/ /\l/

D. rule 1 /^9123548/ /\1/

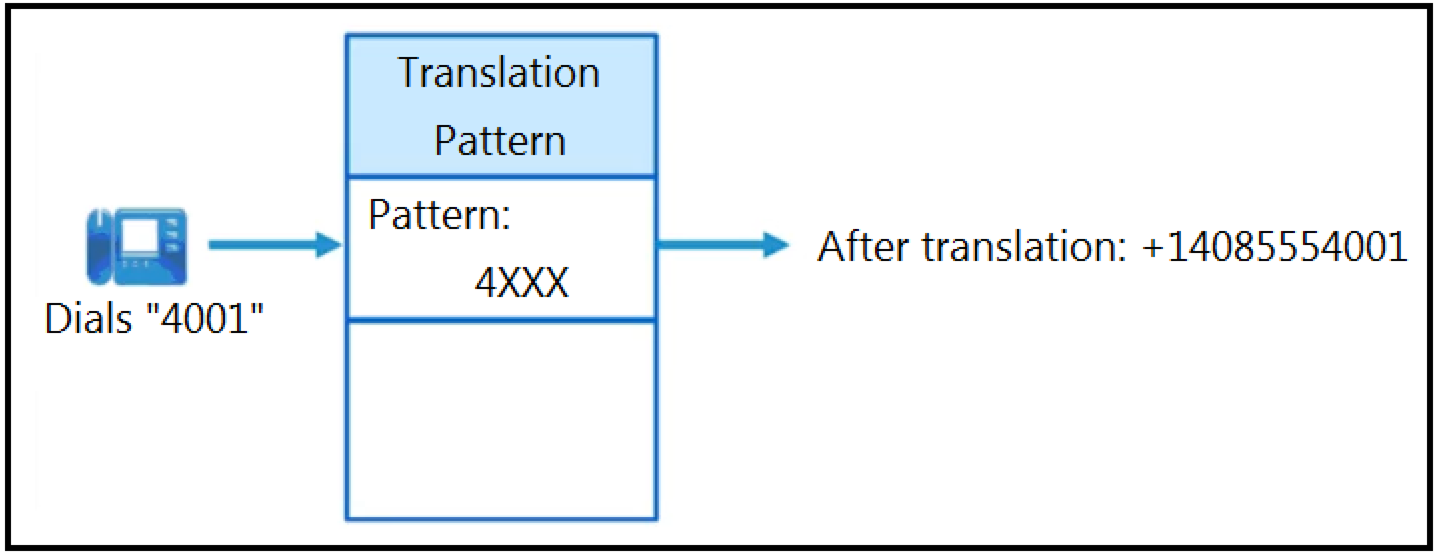
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 96

Topic #: 1

[All 300-815 Questions]



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

A. Calling Party Transformation Mask of +14085554XXX

B. Calling Party Transformation Mask of +1408555XXXX

C. Called Party Transformation Mask of +1408555[35]XXX

D. Called Party Transformation Mask of +14085554XXX

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 97

Topic #: 1

[All 300-815 Questions]

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.

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 100

Topic #: 1

[All 300-815 Questions]

An engineer is configuring Cisco UCM to 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 enterprise softkeys.

B. Configure device pools.

C. Configure class of control.

D. Configure service parameters.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 101
Topic #: 1
[All 300-815 Questions]

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 "Network Hold MOH Source & Announcements" under Hunt Pilot's Queuing section.

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 102

Topic #: 1

[All 300-815 Questions]

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

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

A. sequential

B. call-blast

C. peer

D. parallel

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 103
Topic #: 1
[All 300-815 Questions]

An engineer has two Cisco UCM clusters and wants 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.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 104

Topic #: 1

[All 300-815 Questions]

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 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.

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 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.

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 105
Topic #: 1
[All 300-815 Questions]

ABC company has decided to implement hunt groups to help distribute calls between members. In order to implement this, the administrator must configure hunt list, hunt groups, 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. Sequential, Circular, Broadcast

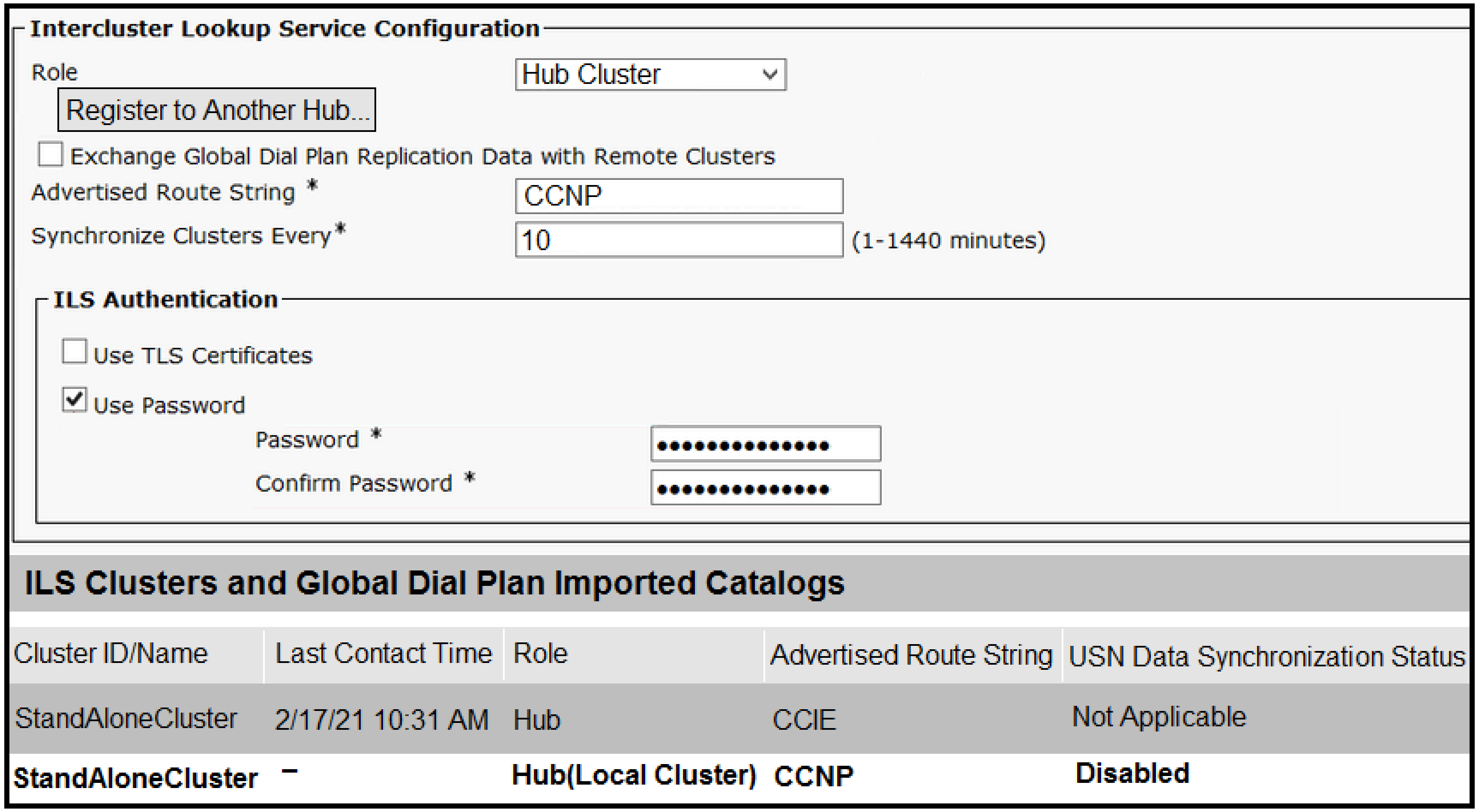
Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 106

Topic #: 1

[All 300-815 Questions]



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

A. Use TLS Certificates must be selected.

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

C. A password has never been set for ILS.

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 107

Topic #: 1

[All 300-815 Questions]

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 last 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 keeps the existing database entries.

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 108
Topic #: 1
[All 300-815 Questions]

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 Enterprise Feature Access number for hunt group logout and set up an access number.

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 109
Topic #: 1

[All 300-815 Questions]

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

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

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 110

Topic #: 1

[All 300-815 Questions]

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 telephony service configuration on Cisco UCME.

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 111

Topic #: 1

[All 300-815 Questions]

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 Device CSS under Current Device Mobility Settings.

D. Set the correct subnet under Device Mobility info.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 112

Topic #: 1

[All 300-815 Questions]

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

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

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

C. Add the long-distance & international pattern's partitions to 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's directory number.

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 113

Topic #: 1

[All 300-815 Questions]

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

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

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

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 114

Topic #: 1

[All 300-815 Questions]

Single Number Reach calls to a cell phone that not answered are leaving voicemails on the cell phone rather than 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.

E. Decrease the Answer Too Late timer.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 115

Topic #: 1

[All 300-815 Questions]

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.

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 116

Topic #: 1

[All 300-815 Questions]

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 an access list.

B. Configure Single Number Reach.

C. Configure Mobile Voice Access.

D. Configure a mobility identity.

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 117

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815
Question #: 118
Topic #: 1
[All 300-815 Questions]

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

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

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

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

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

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

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

B. Enable Early Offer on the SIP trunk.

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

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 119

Topic #: 1

[All 300-815 Questions]

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

Show Suggested Answer

Actual exam question from Cisco's 300-815

Question #: 120

Topic #: 1

[All 300-815 Questions]

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.

Show Suggested Answer