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Question 1

Question Type: MultipleChoice

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

Options:

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

Question 2

Question Type: MultipleChoice

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?

Options:

A- WAN connectivity

B- LAN connectivity

C- route pattern

D- route list and route group

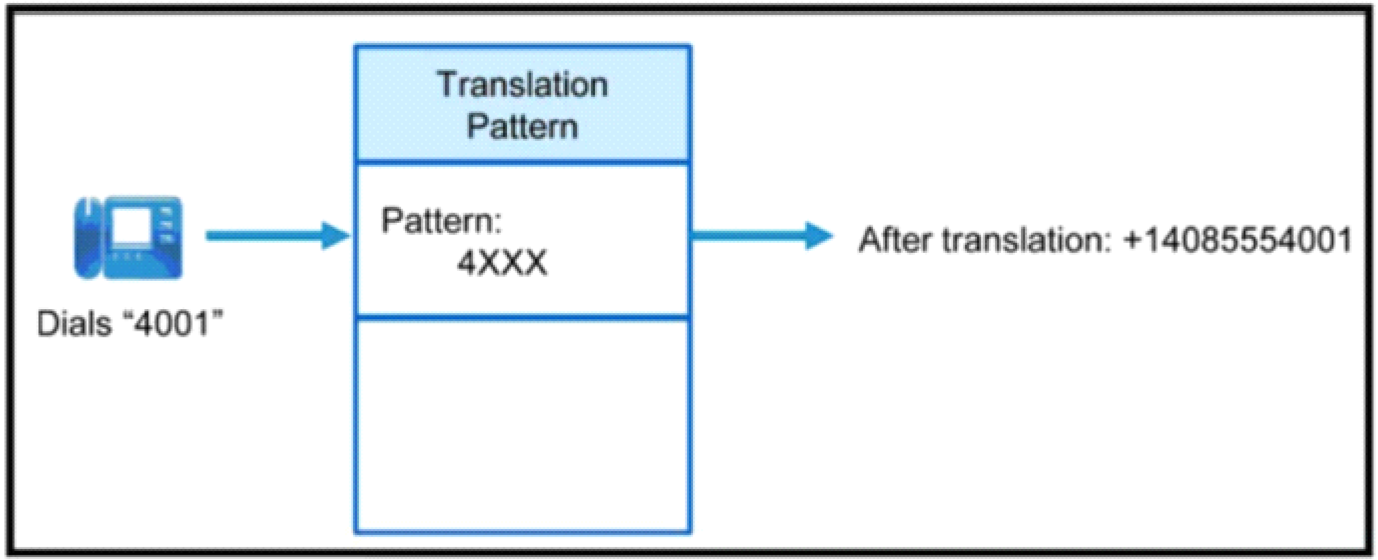
Answer:

A

Question 3

Question Type: **MultipleChoice**

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?

Options:

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

Question 4

Question Type: **MultipleChoice**

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.)

Options:

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

Question 5

Question Type: MultipleChoice

Refer to the exhibit.

```
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
```

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

Options:

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

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

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

D- rule 1/^9123548/^1/

Answer:

A

Question 6

Question Type: **MultipleChoice**

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
Via: SIP/2.0/UDP 10.10.150.11:5060;branch=z9hG4bK1046d36216b0de
From: <sip:1001@10.10.150.11>;tag=23125042-8a7beda1-fb5d-4d82-bdb6-4b07a7393aff-27428380
To: "CISCO SYSTEMS" <sip:8005532447@10.10.150.1>;tag=D9748182-FA5
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02@10.10.150.1
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:1001@10.10.150.11:5060>;
Content-Type: application/sdp
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,
```

An engineer is troubleshooting a call-establishment problem between Cisco Unified Border Element and Cisco UCM. Which command set corrects the issue?

Options:

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 source-interface 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

Question 7

Question Type: **MultipleChoice**

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?

Options:

A- Configure Single Number Reach.

B- Configure an access list.

C- Configure a mobility identity.

D- Configure Mobile Voice Access.

Answer:

B

Question 8

Question Type: MultipleChoice

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?

Options:

A- top down

B- circular

C- broadcast

D- longest idle time

Answer:

A