Free Questions for 300-815 by certsinside

Shared by Trujillo on 29-01-2024

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Question 1

Question Type: MultipleChoice

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?

Options:

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

Question 2

Question Type: **MultipleChoice**

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
```

Options:

A- Option A

B- Option B

C- Option C

D- Option D

E- Option E

Answer:

C, D

Question 3

Question Type: **MultipleChoice**

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?

Options:

A- G.711 A-law

B- G.711 U-law

C- G.729r8

D- G.729br8

Answer:

C

Question 4

Question Type: **MultipleChoice**

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?

Options:

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

Question 5

Question Type: MultipleChoice

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?

Options:

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

Question 6

Question Type: **MultipleChoice**

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?

Options:

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

Question 7

Question Type: MultipleChoice

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?

Options:

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.

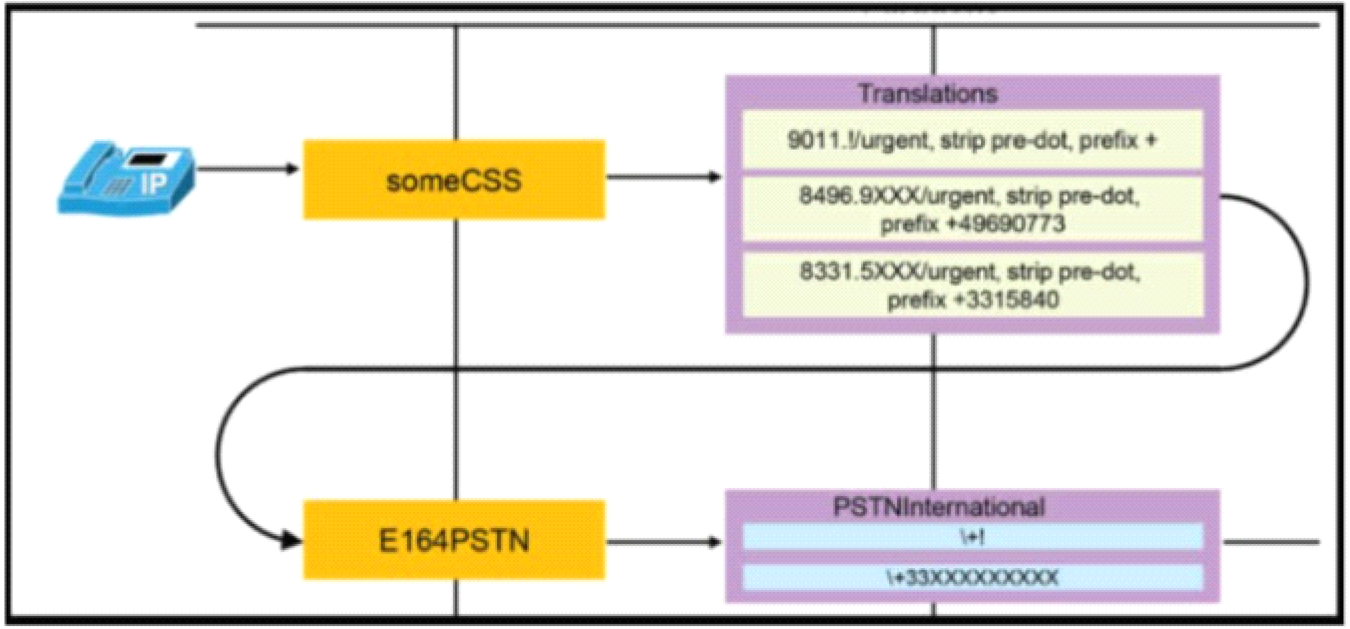
Answer:

C

Question 8

Question Type: MultipleChoice

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?

Options:

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

Question 9

Question Type: MultipleChoice

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.)

Options:

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

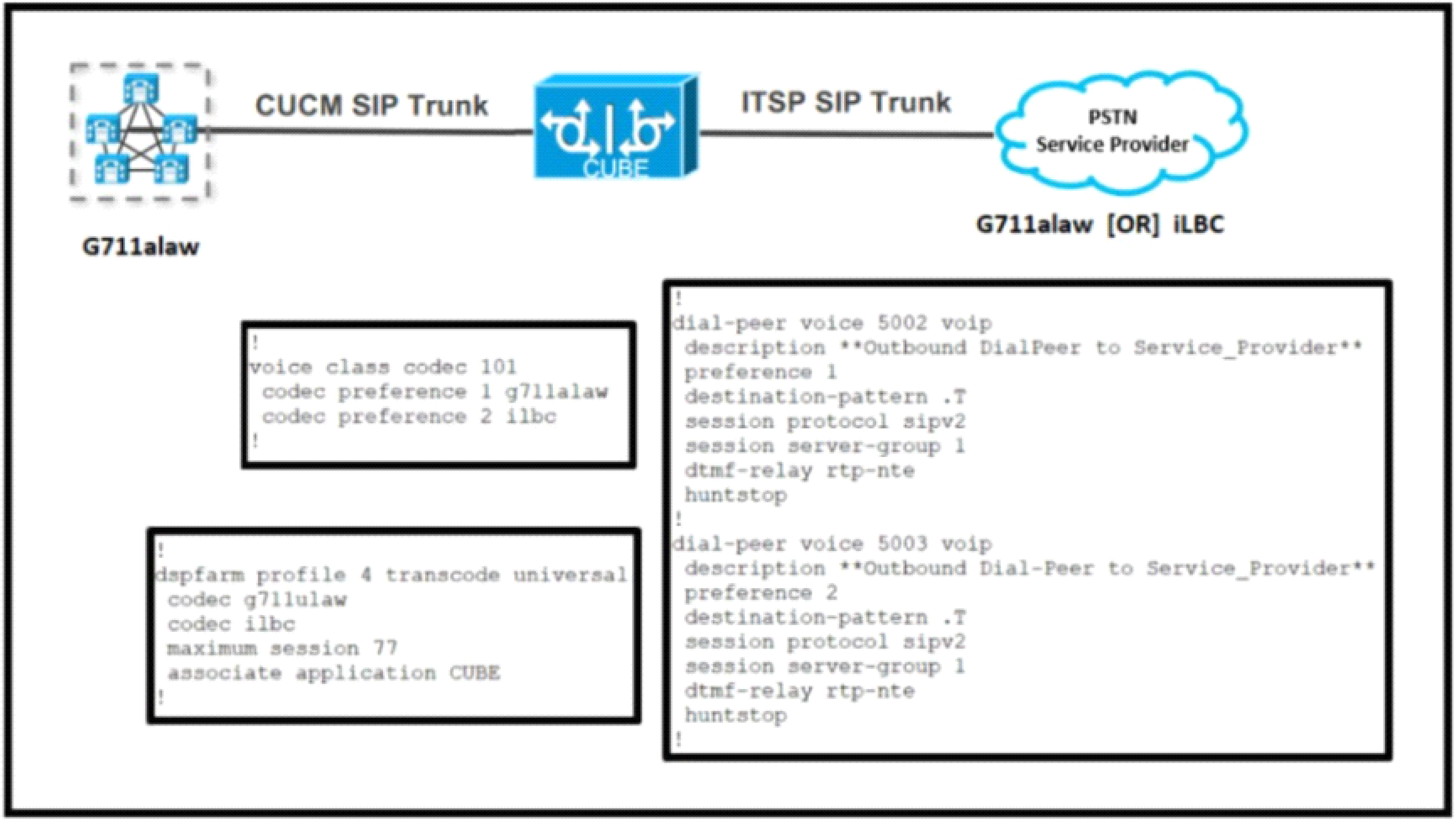
Answer:

A, E

Question 10

Question Type: **MultipleChoice**

Refer to the exhibit.



Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only G.711 A-law. The service provider reports that some destinations support only G.711 A-law while others support only iLBC. The service provider also allows only 20 active calls at a time Which configuration allows successful media negotiation for all calls using outbound dial peers 5002 and 5003?

```
dial-peer voice 5002 voip
codec g711alaw ilbc
!
dial-peer voice 5003 voip
codec g711alaw ilbc
```

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

```
dial-peer voice 5002 voip
codec g711alaw
!
dial-peer voice 5003 voip
codec ilbc
```

```
dial-peer voice 5002 voip
voice-class codec 101
!
dial-peer voice 5003 voip
voice-class codec 101
```

Options:

A- Option A

B- Option B

C- Option C

D- Option D

Answer:

D

Question 11

Question Type: **MultipleChoice**

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.)

Options:

A- directory URI partition

B- SIP route pattern

C- intercluster trunk

D- calling search space and partition

E- SIP trunk

Answer:

B, E