300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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QUESTION 1

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

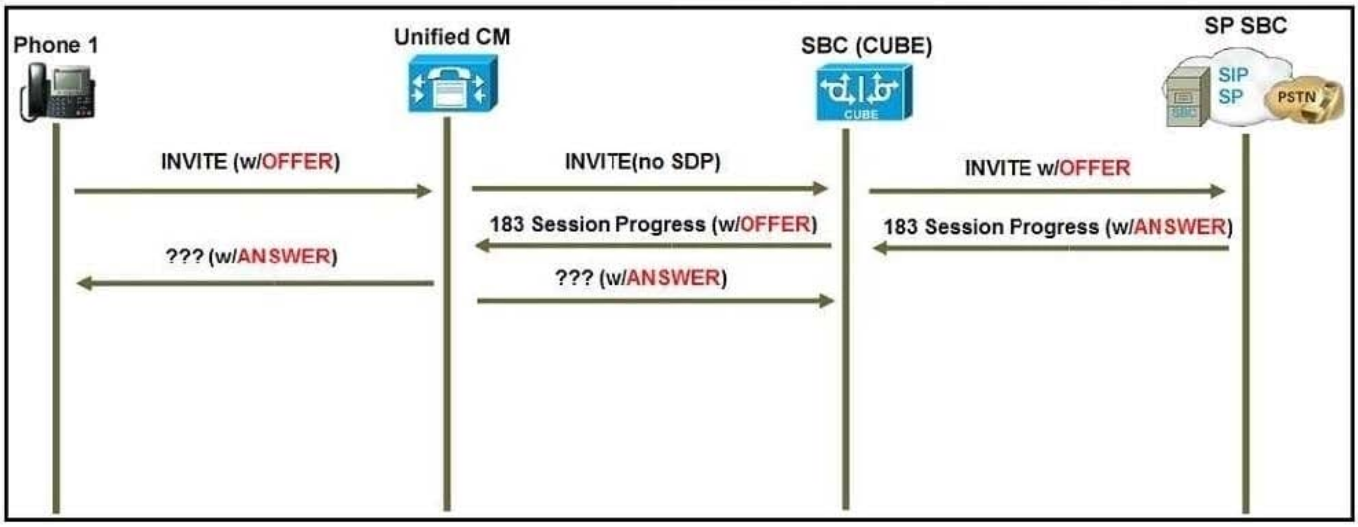
D. calling search space and partition

E. SIP trunk

Correct Answer: CE

QUESTION 2

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.

Correct Answer: C

QUESTION 3

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
 codes 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-z9hG4bK1046
From: <sip:1001010.10.150.11>;tag-23125042-8a7bedal-f
To: "CISCO SYSTEMS" <sip:8005532447010.10.150.1>;tag=
Date: Tue, 30 Mar 2021 22:14:00 GMT
Call-ID: C57C1746-90D511EB-826BBE69-C6943E02010.10.15
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK
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, N
  Called Number-8005532447(TON-Unknown, NPI-Unknown),
  Calling Translated-FALSE, Subscriber Type Str-Unkno
  Incoming Dial-peer-100, Progress Indication-NULL(0)
```

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

Correct Answer: B

QUESTION 4

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

A. voice class request sip-header diversion

B. sip-header contact modify

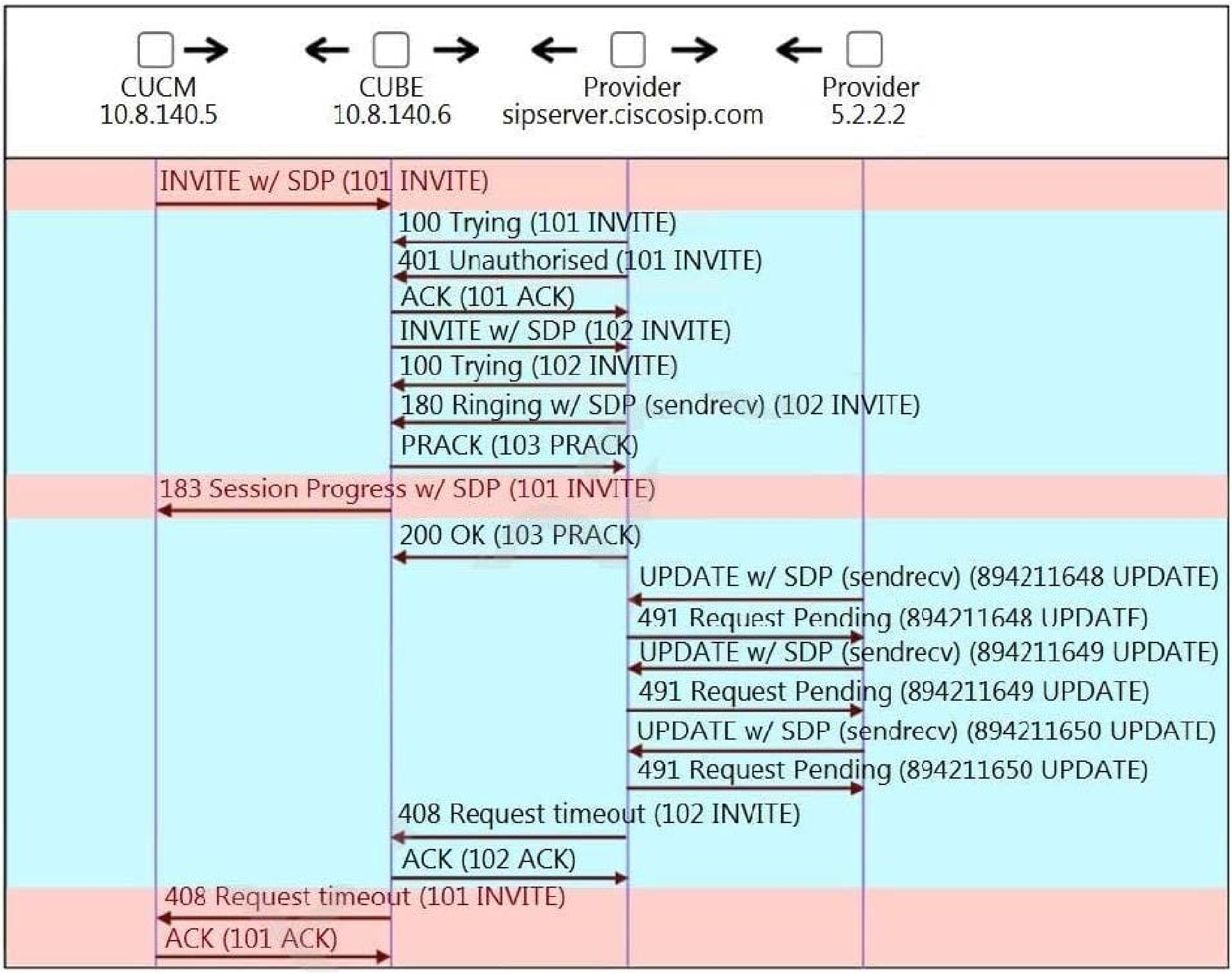
C. voice class request sip-header modify

D. request invite sip-header diversion modify

Correct Answer: D

QUESTION 5

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.

Correct Answer: D