

300-815^{Q&As}

Implementing Cisco Advanced Call Control and Mobility Services
(CLACCM)

Pass Cisco 300-815 Exam with 100% Guarantee

Free Download Real Questions & Answers **PDF** and **VCE** file from:

<https://www.leads4pass.com/300-815.html>

100% Passing Guarantee
100% Money Back Assurance

Following Questions and Answers are all new published by Cisco
Official Exam Center

- ⚙️ **Instant Download** After Purchase
- ⚙️ **100% Money Back** Guarantee
- ⚙️ **365 Days** Free Update
- ⚙️ **800,000+** Satisfied Customers



QUESTION 1

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 2

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?

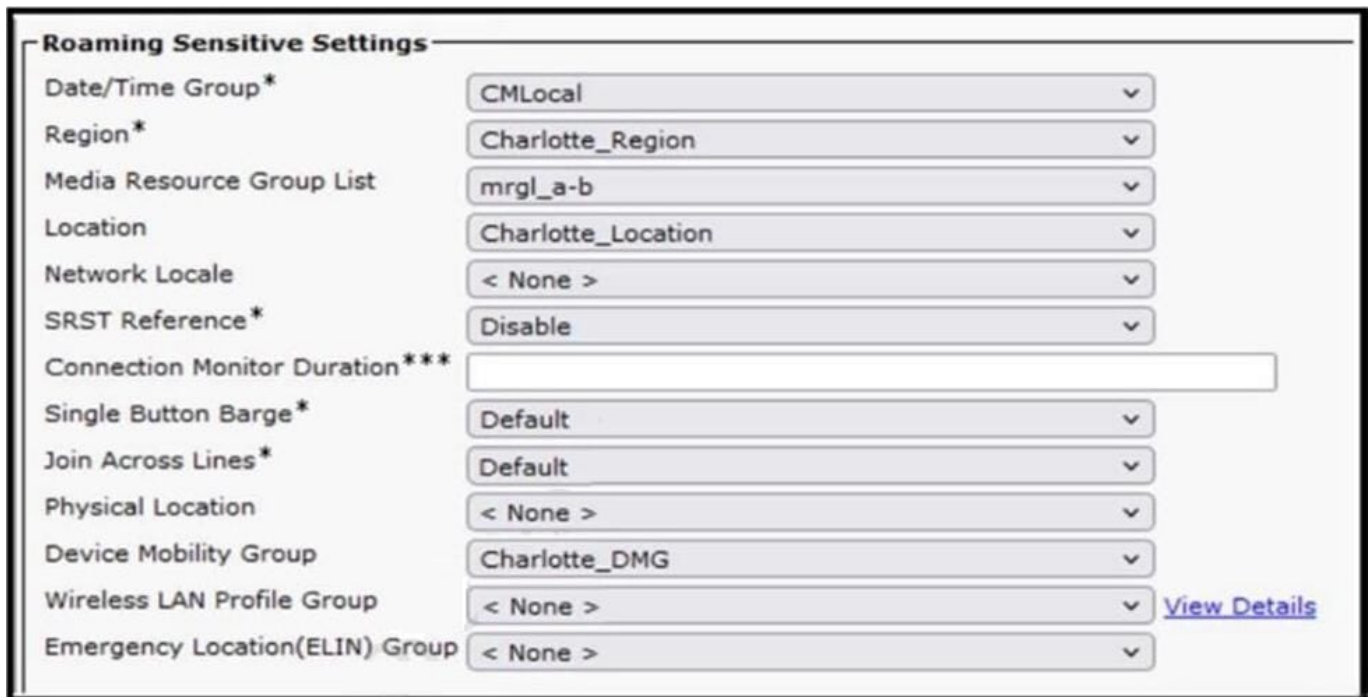
- A. voice service voip enable ip address trust list
- B. voice service voip ip address trusted list
- C. voice service voip ip address trusted authenticate
- D. voice service voip enable ip address trust authentication

Correct Answer: B

Reference:

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

QUESTION 3



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

- A. Join across lines is not supported with Device Mobility and must be disabled.
- B. The network locale must be changed from none to the Charlotte locale.
- C. The physical location must be updated from none to the appropriate location.
- D. The device must be added to the Device Mobility group.

Correct Answer: D

QUESTION 4

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

- A. Localize the calling and called numbers to E.164 format and globalize the called number in the gateway.
- B. Globalize the calling and called numbers to E.164 format and localize the called number in the gateway.
- C. Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
- D. Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.

Correct Answer: B

QUESTION 5

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

- A. Allow for VAD to be used for calls using the G7.29 codec, which reduces the usage of the WAN bandwidth and saves around 30% of bandwidth per call
- B. Configure the rtcp keepalive command in Cisco Unified Border Element to generate reports, which can be reviewed using the debug voip rtcp packet command.
- C. Encrypt the media to stop rogue devices from replying and putting that traffic back on the WAN, which avoids any extra bandwidth and ensures the quality of the calls.
- D. Gather statistics on a media connection and information such as packets sent, lost packets, jitter, feedback, and round-trip delay. This information can help isolate the type of audio quality issues and the direction of the affected traffic.
- E. Compress the headers of RTP traffic to lower the bandwidth consumption over the WAN, which allows more calls with less bandwidth consumed.

Correct Answer: BD

[300-815 PDF Dumps](#)

[300-815 VCE Dumps](#)

[300-815 Exam Questions](#)