

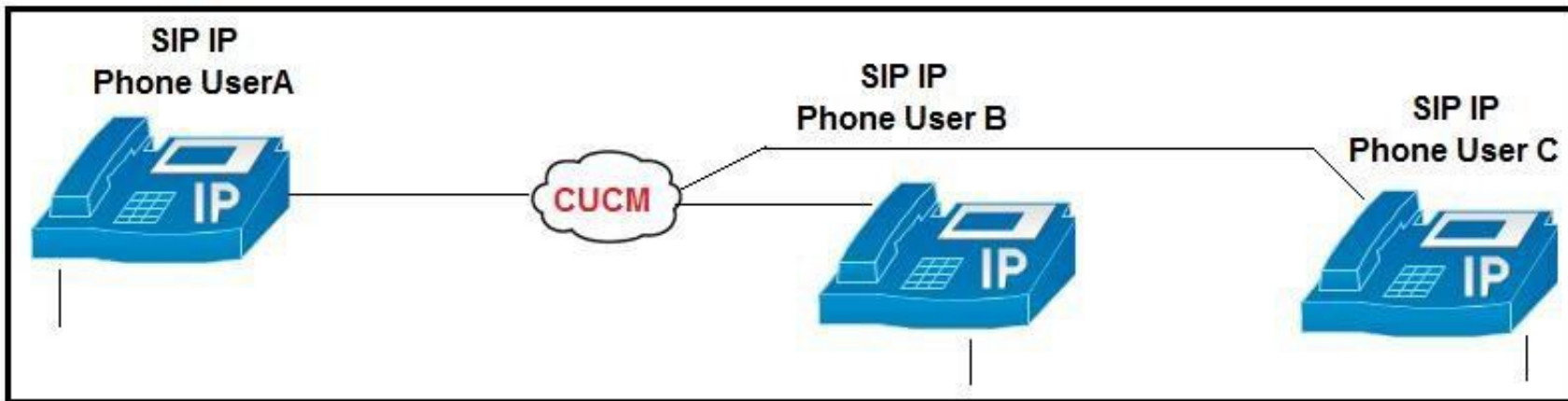
# Cisco

## Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)



#### NEW QUESTION 1



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.

**Answer:** AC

#### NEW QUESTION 2

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

**Answer:** AB

#### NEW QUESTION 3

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.

**Answer:** B

#### NEW QUESTION 4

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

**Answer:** A

#### NEW QUESTION 5

```
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”.

**Answer: B**

#### NEW QUESTION 6

```
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/

**Answer: B**

#### NEW QUESTION 7

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

**Answer: B**

#### NEW QUESTION 8

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. \*.\*

**Answer: BE**

#### NEW QUESTION 9

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.

**Answer:** AB

#### NEW QUESTION 10

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. address 555 .....
- B. codec g729
- C. session-protocol sipv2
- D. incoming called number 555.....

**Answer:** D

#### NEW QUESTION 10

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 voipRouter(conf-voi-serv)#allow-connections h323 to h323
- B. Router(config)#dial-peer voice 2 voipRouter(config-dial-peer)#no vad
- C. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections voice-mail mod
- D. Router(config)# voice service voipRouter(conf-voi-serv)#no supplementary-service sip moved-temporarily

**Answer:** A

#### NEW QUESTION 13

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.

**Answer:** A

#### NEW QUESTION 17



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.

**Answer:** DE

**NEW QUESTION 19**

In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

- A. Dialed Number Analyzer
- B. Real-Time Monitoring Tool
- C. SDI trace
- D. SDL trace

**Answer:** A

**NEW QUESTION 20**

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.

**Answer:** B

**NEW QUESTION 21**

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. enterprise alternate number mask

**Answer:** AC

**NEW QUESTION 23**

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

**Answer:** B

**NEW QUESTION 25**

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

**Answer:** B

**NEW QUESTION 29**

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

**Answer:** B

**NEW QUESTION 32**

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.
- C. It would be best to check network latency.
- D. The user device profile has not been assigned to the user.
- E. The user device profile is not subscribed to the Cisco Extension Mobility service.

**Answer:** D

**NEW QUESTION 36**

What is a component of Cisco Unified Mobility?

- A. Unified IVR
- B. Mobile Connect
- C. Smart Client Support
- D. Single Number Connect

**Answer:** B

**NEW QUESTION 41**

When the services key is pressed Cisco Extension Mobility does not show up. What is the cause of the issue?

- A. The URL configured for Cisco Extension Mobility is not correct.
- B. Cisco Extension Mobility Service is not running.
- C. The phone is not subscribed to Cisco Extension Mobility Service.
- D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

**Answer:** C

**NEW QUESTION 43**

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 locatio
- D. Device Mobility group, and region
- E. device pool, Device Mobility group, and Cisco IP phone

**Answer:** A

**NEW QUESTION 45**

.....

## Thank You for Trying Our Product

### We offer two products:

1st - We have Practice Tests Software with Actual Exam Questions

2nd - Questions and Answers in PDF Format

### 300-815 Practice Exam Features:

- \* 300-815 Questions and Answers Updated Frequently
- \* 300-815 Practice Questions Verified by Expert Senior Certified Staff
- \* 300-815 Most Realistic Questions that Guarantee you a Pass on Your First Try
- \* 300-815 Practice Test Questions in Multiple Choice Formats and Updates for 1 Year

**100% Actual & Verified — Instant Download, Please Click**  
**[Order The 300-815 Practice Test Here](#)**