Braindump2go First Try, First Pass!

Braindump2go Guarantee All Exams 100% Pass One Time!

- **Vendor:** Cisco
- **Exam Code:** 300-815
- **Exam Name:** Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)
- **New Updated Questions from Braindump2go (Updated in Sep/2020)**

Visit Braindump2go and Download Full Version 300-815 Exam Dumps

QUESTION 34
Which two features are part of Cisco Unified Mobility? (Choose Two)

A. Mobile Voice Access
B. Device Mobility
C. Enterprise Feature Access
D. Shared line
E. Extension Mobility Cross Cluster

Answer: AC

QUESTION 35
In RTMT, how many concurrent trace collections can you schedule to download the trace files to an SFTP or FTP server on your network?

A. three
B. four
C. five
D. six

Answer: D

QUESTION 36
Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

A. allow-connections sip to sip
B. voice service voip
C. voice register global
D. voice register dn

Answer: C
Explanation:
https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-express/99946-cme-sip-guide.html

QUESTION 37
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
Explanation:
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/200412-DTMF-Relay-and-Interworking-on- CUBE.html#anc35

QUESTION 38
Where is the dtmf-relay command configured on Cisco Unified Border Element?

A. in the voice-class VoIP configuration
B. in the VoIP dial peer
C. in global SIP configuration
D. in the VoIP or POTS dial peers

Answer: B
Explanation:
https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/dtmf-relay.html

QUESTION 39
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?

```
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
```

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

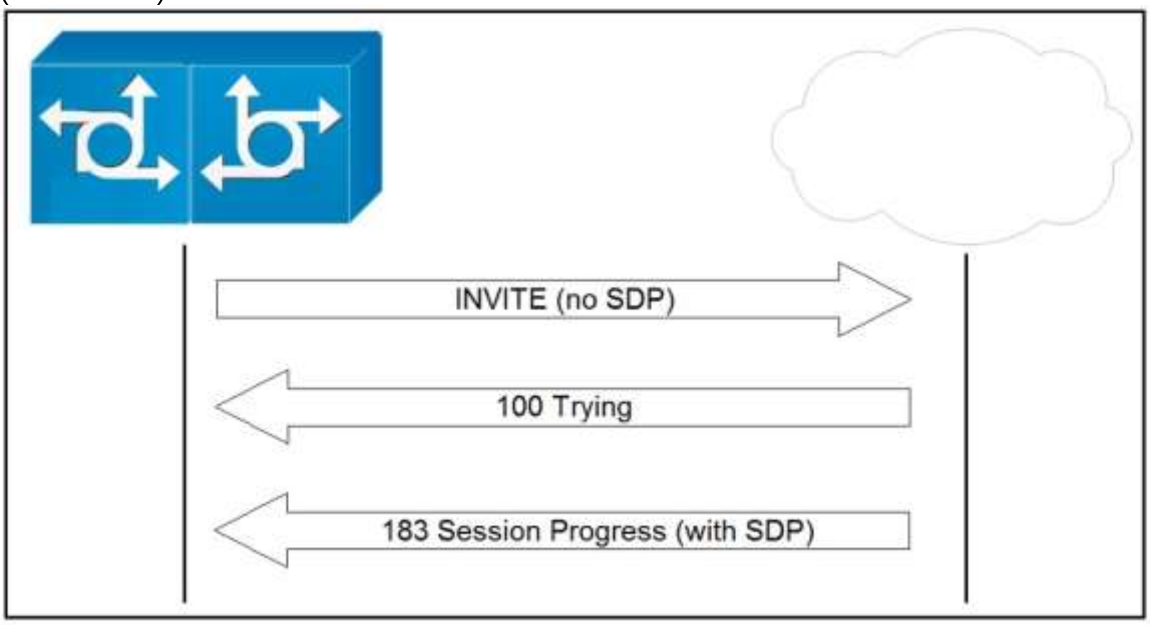
QUESTION 40
Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco 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?

```
voice translation-rule 84
      rule 1 /^\ ([2-9]..[2-9]......$\)/ /\2/
```

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

QUESTION 41
Refer to the exhibit. An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border Element. The customer's carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)



A. Enanle PRACK.
B. Enable Early Offer on the Cisco Unified Border Element.
C. Enable the supplementary-service media-renegotiate command.

D. Enable Media Flow Around
E. Enable Mid-Call Signaling Consumption.

Answer: AB

QUESTION 42
Which IOS command creates a SIP-enabled dial peer?

A. voice dial-peer 20 sip
B. dial-peer voice 20 voip
C. dial-peer voice 20 pots
D. dial peer voice 20 sip

Answer: B
Explanation:
https://www.ciscopress.com/articles/article.asp?p=664148&seqNum=6

QUESTION 43
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

QUESTION 44
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: DE