

## [September-2020Braindump2go 300-815 PDF Dumps 300-815 70Q Free Offer[Q31-Q49

Sep/2020 Latest Braindump2go 300-815 Exam Dumps with PDF and VCE Free Updated Today! Following are some new 300-815 Real Exam Questions!  
QUESTION 31A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)  
A. three-way conference  
B. secure SIP lines  
C. T.38 fax relay  
D. transcoding  
E. SIP trunk  
Answer: ACE  
Explanation:

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cusrst/admin/sccp\\_sip\\_srst/configuration/guide/SCCP\\_and\\_SIP\\_SRST\\_Admin\\_Guide/srst\\_sip\\_isr4000.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_sip_isr4000.html)  
QUESTION 32Which 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  
Explanation:

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept\\_ECC4F4E7ED0F45C594B703EEF34762F2](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2)  
QUESTION 33You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?  
A. configuration for a single SIP phone  
B. configuration items common for all SIP phones  
C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)  
D. configuration for SIP registrar service  
Answer: C  
Explanation:

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cusrst/admin/sccp\\_sip\\_srst/configuration/guide/SCCP\\_and\\_SIP\\_SRST\\_Admin\\_Guide/srst\\_setting\\_up\\_using\\_sip.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_setting_up_using_sip.html)  
QUESTION 34Which 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 35In 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 36Which 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 37For 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 38Where 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>

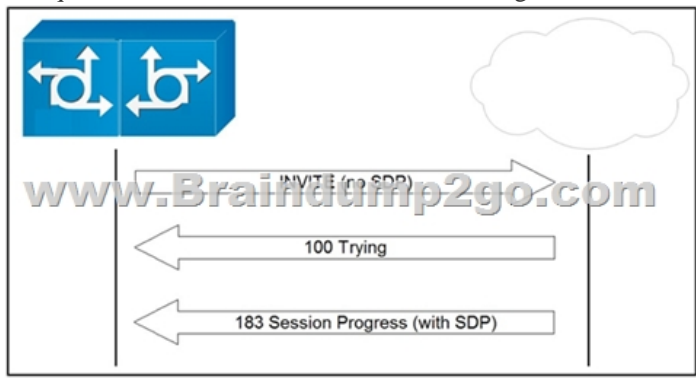
QUESTION 39Refer to the exhibit. Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco Unified Communications Manager 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) )
!
dial-peer voice 999 voip
  translation-profile outgoing incoming
  session protocol sipv2
  incoming called-number
  dtmf-relay rtp-nte
  codec transparent
  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?



A. rule 1 //+([1].\*)/ /0111/ 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].....\$)/ /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. Enable 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: A, B  
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: D, E

QUESTION 45 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: A, B  
Explanation:

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/10\\_0\\_1/ccmfeat/CUCM\\_BK\\_F3AC1C0F\\_00\\_cucm-features-services-guide-100/CUCM\\_BK\\_F3AC1C0F\\_00\\_cucm-features-services-guide-100\\_chapter\\_010000.pdf](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_010000.pdf)  
QUESTION 46 A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?  
A. 813510[012345] B. 813510[12345] C.

813510[0-5]D. 81XXXXXAnswer: AQUESTION 47Which three CLI commands are used when configuring H.323 call survivability for all calls? (Choose three.)A. voice service voipB. telephony-serviceC. h323D. call preserveE. call-router h323-annexG. transfer-systemAnswer: ACDQUESTION 48Which command displays the detailed configuration of all the Cisco Unified IP phones, voice ports, and dial peers of the Cisco Unified SRST router?A. show call-manager-fallback allB. show dial-peer voice summaryC. show ephone summaryD. show voice port summaryAnswer: AQUESTION 49Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)A. can be associated under the route groupB. can be associated only under the route listC. chooses the route group that is configured under the device pool of the calling-party deviceD. chooses the route group that is configured under the device pool of the called-party deviceE. can be assigned directly to the route patternAnswer: BDResources From:1.2020 Latest Braindump2go 300-815 Exam Dumps (PDF & VCE) Free Share: <https://www.braindump2go.com/300-815.html>2.2020 Latest Braindump2go 300-815 PDF and 300-815 VCE Dumps Free Share: <https://drive.google.com/drive/folders/1IHjHEsMRfmKZVssEobUIr0a8XtPy0qWv?usp=sharing>3.2020 Free Braindump2go 300-815 PDF Download:[https://www.braindump2go.com/free-online-pdf/300-815-Dumps\(34-44\).pdf](https://www.braindump2go.com/free-online-pdf/300-815-Dumps(34-44).pdf)  
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