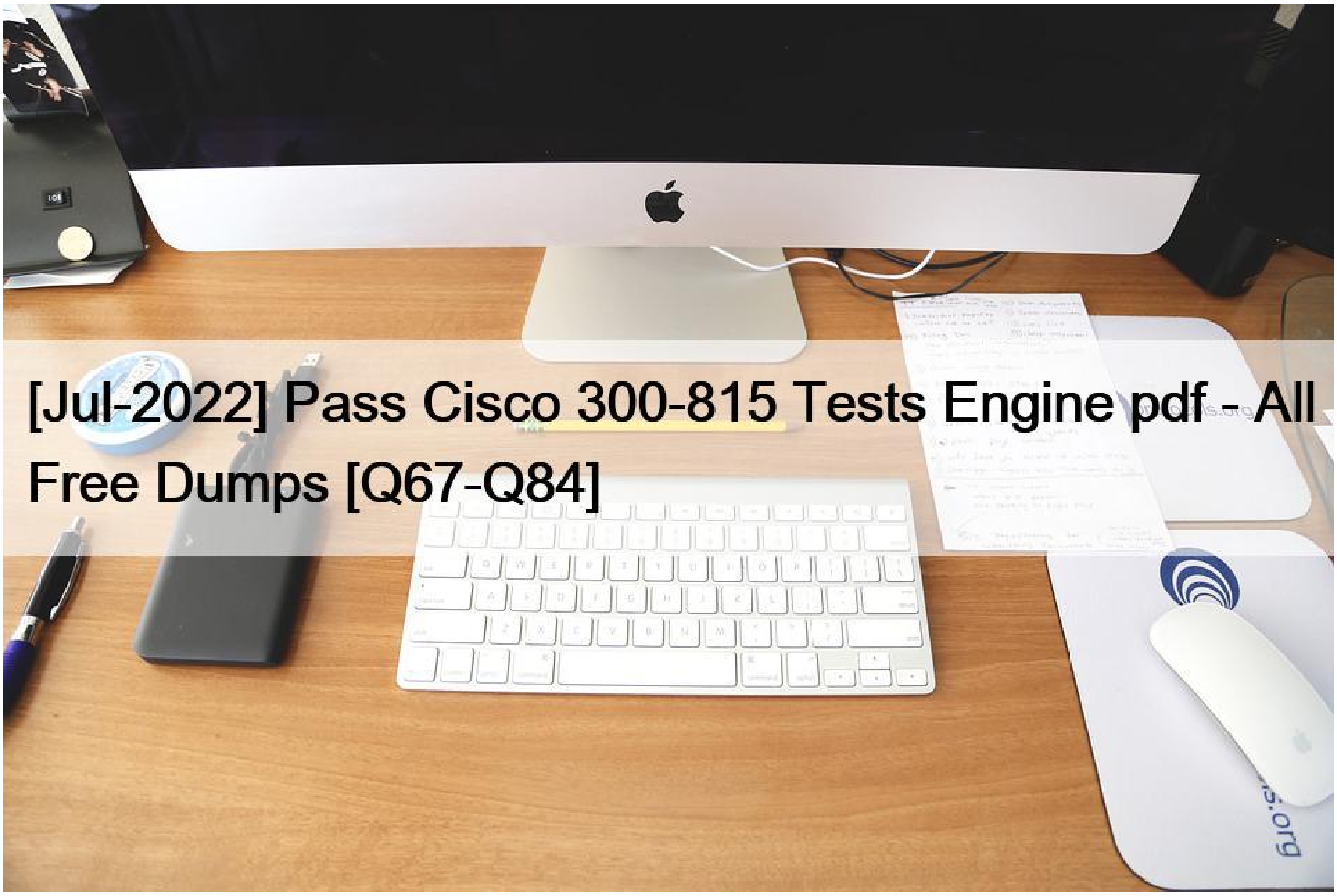
[Jul-2022 Pass Cisco 300-815 Tests Engine pdf - All Free Dumps [Q67-Q84



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Implementing Cisco Advanced Call Control and Mobility Services Practice Tests 2022 | Pass 300-815 with confidence!

QUESTION 67

Refer to the exhibit.

```
SIP/2.0 200 OK
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=-IN IP4 10.10.10.11
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp: 110 0-16
a=ptime: 20

ACK sip:+123456789@10.10.20.20:5060 SIP/2.0
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=ptime:20
```

Users report that when they dial to Cisco Unity Connection from an external network, they cannot enter any digits. Assuming only in-band DTMF is supported, what is a reason for this malfunction?

* The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.
* There is SIP Delayed Offer. DTMF is supported only in Early Offer.
* The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.
* No DTMF is negotiated.

QUESTION 68

A user requests a feature to send an active call to the mobile phone number on the physical phone. As an administrator, what should be configured in the Cisco UCM to accomplish this?

* A Remote Destination Profile having the same extension as the Mobile phone’s number adds the physical phone’s Directory Number to the RDP as a Remote Destination. Add the softkey “Mobility” to the physical phone’s softkey template.
* A Remote Destination Profile having the same extension as the physical phone’s Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey “Mobility” to the physical phone’s softkey template.
* A Remote Destination Profile having the same extension as the Mobile phone’s number adds the physical phone’s Directory Number to the RDP as a Remote Destination. Add the softkey “Join” to the physical phone’s softkey template
* A Remote Destination Profile having the same extension as the physical phone’s Directory Number, add the mobile phone number to the RDP as a Remote Destination. Add the softkey “Join” to the physical phone’s softkey template.

QUESTION 69

An administrator is asked to configure egress call routing by applying globalization and localization on Cisco UCM. How should this be accomplished?
* Localize the calling and called numbers to PSTN format and globalize the calling and called numbers in the gateway.
* Globalize the calling and called numbers to PSTN format and localize the calling number in the gateway.
* Localize the calling and called numbers to E. 164 format and globalize the called number in the gateway.
* Globalize the calling and called numbers to E. 164 format and localize the called number in the gateway.

QUESTION 70

Signal number reach call phone that not answered are leaving voicemails on the cell phone rather the corporate mailbox. Which two options will resolve this issue? (Choose two.)
* Check the Enable Extend and Connect checkbox
* Check the Enable Unified Mobility features checkbox
* Decrease the T302 timer
* Decrease the T301 timer Decrease the Answer Too Late timer

QUESTION 71

Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)
* TokenID
* username and secret key
* TLS certificates
* passwords
* FQDN of the servers defined in DNS

QUESTION 72

The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call.

You gather the H.225 and H.245 messages for one of the one-way audio calls. Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).
* H.245 Terminal Capability Set
* H.245 Open Logical Channel
* H.225 Connect
* H.245 Open Logical Channel Ack

Section: Signaling and Media Protocols

Explanation/Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

QUESTION 73

A support engineer is troubleshooting a voice network. When conducting a search for call setup details related to calling search space issues, which trace files should be investigated?
* CallManager traces
* CTI Manager traces
* Cisco IP Manager Assistant
* Call logs

Section: Signaling and Media Protocols

QUESTION 74

You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?
* configuration for a single SIP phone
* configuration items common for all SIP phones
* configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)
* configuration for SIP registrar service
Reference:

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 75

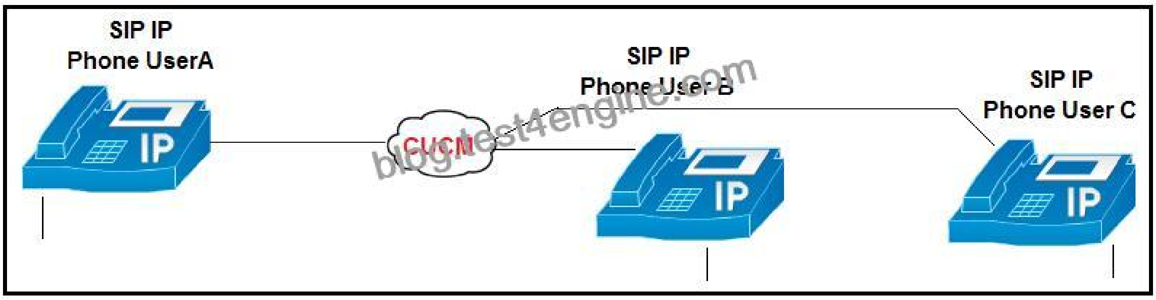
CollabCorp is a global company with two clusters, emea.collab corp and apac.collab.corp. URI dialing is implemented and working in each cluster. The company configured routing between clusters to make inter-cluster calls via URI. but this is not working as expected. Which two configuration elements should be checked to resolve this issue? (Choose two.)
* directory URI partition
* SIP route pattern
* intercluster trunk
* calling search space and partition
* SIP trunk

QUESTION 76

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?
* The Cisco Extension Mobility service has not been configured on the phone.
* There might be a significant delay between the button being pressed and the Cisco Extension Mobility service recognizing it. It would be best to check network latency.
* The user device profile has not been assigned to the user.
* The user device profile is not subscribed to the Cisco Extension Mobility service.

QUESTION 77

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.)
* Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
* Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
* 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.
* 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.
* 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.

QUESTION 78

An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?
* ILS Max Number of Learned Objects in Database
* ILS Active Learned Object Upper Limit
* Global Data Service Parameter Limit
* Imported Dial Plan Replication Database Object Lower Limit

QUESTION 79

Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)
* TokenID
* username and secret key
* TLS certificates
* passwords
* FQDN of the servers defined in DNS

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/11_5_1_SU1/cucm_b_system-configuration-guide-1151su1/cucm_b_system-configuration-guide-1151su1_chapter_011001.pdf

QUESTION 80

Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?
* Other Group Pickup
* BLF Call Pickup
* Group Call Pickup
* Directed Call Pickup

QUESTION 81

An engineer set up and successfully tested a TEHO solution on the Cisco UCM. PSTN calls are routed correctly using the IP WAN as close to the final PSTN destination as possible. However, suddenly, calls start using the backup local gateway instead. What is causing the issue?

* WAN connectivity
* LAN connectivity
* route pattern
* route list and route group

QUESTION 82

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?
* debug H.323 messages
* debug H.225 asn1
* debug H.246 asn 1
* debug H.225 media
* debug H.323 asn 1
Section: Signaling and Media Protocols

QUESTION 83

A single site reports that when they dial select numbers, the call connects, but they do not get audio. The administrator finds that the calls are not routing out of the normal gateway but out of another site’s gateway due to a TEHO configuration. What is the next step to diagnose and solve the issue?
* Verify that IP routing is correct between the gateway and the IP phone.
* Verify that the route pattern is not blocking calls to the destination number.
* Verify that the dial peer of the gateway has the correct destination pattern configured.
* Verify that the route pattern has the correct calling-party transformation mask

QUESTION 84

An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60 seconds?
* R2(config-ephone-dn)#park reservation-group 60
* R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002
* R2(config-ephone-dn)#park reservation-group 1
* R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002
Section: Cisco Unified CM Call Control Features

What Is 300-815 Exam and Who Is Its Targeted Audience?

This 300-815 Implementing Cisco Advanced Call Control as well as Mobility Services exam checks the applicant's understanding of various advanced level call control as well as mobility features. Particularly, such an exam is intended for those with a firm grasp of various signaling and media protocols, gateway technologies, Cisco Unified Border element, CM Call Control, and other related concepts. As for its details, the exam needs to be finished within 90 minutes. Also, the applicant requires an account on Pearson

VUE to schedule the exam. Thus, the candidate should select Proctored Exams and enter 300-815 as the test number on the Pearson VUE website. As you keep in mind, securing a passing score in the Cisco 300-815 test will push the entrant a step closer to earning the CCNP Collaboration endorsement.

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