300-815 Q&As

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

Pass Cisco 300-815 Exam with 100% Guarantee

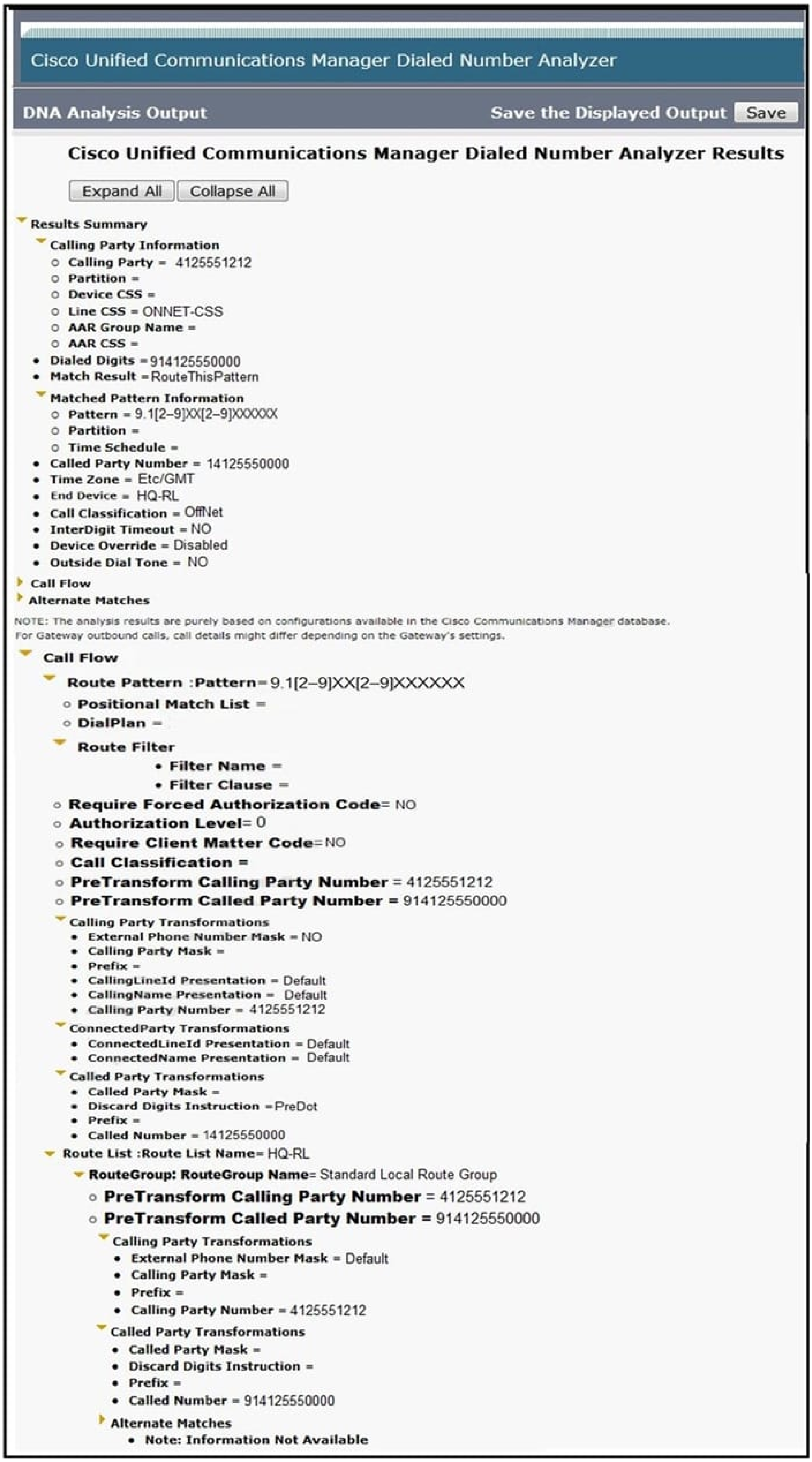
Free Download Real Questions & Answers **PDF** and **VCE** file from:

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

QUESTION 1


Cisco Unified Communications Manager Dialed Number Analyzer
DNA Analysis Output
Save the Displayed Output Save
Cisco Unified Communications Manager Dialed Number Analyzer Results
Expand All Collapse All
Results Summary
Calling Party Information
Calling Party = 4125551212
Partition =
Device CSS =
Line CSS = ONNET-CSS
AAR Group Name =
AAR CSS =
Dialed Digits = 914125550000
Match Result = RouteThisPattern
Matched Pattern Information
Pattern = 9.1[2–9]XX[2–9]XXXXXX
Partition =
Time Schedule =
Called Party Number = 14125550000
Time Zone = Etc/GMT
End Device = HQ-RL
Call Classification = OffNet
InterDigit Timeout = NO
Device Override = Disabled
Outside Dial Tone = NO
Call Flow
Alternate Matches
NOTE: The analysis results are purely based on configurations available in the Cisco Communications Manager database.
For Gateway outbound calls, call details might differ depending on the Gateway's settings.
Call Flow
Route Pattern :Pattern= 9.1[2–9]XX[2–9]XXXXXX
Positional Match List =
DialPlan =
Route Filter
Filter Name =
Filter Clause =
Require Forced Authorization Code= NO
Authorization Level= 0
Require Client Matter Code=NO
Call Classification =
PreTransform Calling Party Number = 4125551212
PreTransform Called Party Number = 914125550000
Calling Party Transformations
External Phone Number Mask = NO
Calling Party Mask =
Prefix =
CallingLineId Presentation = Default
CallingName Presentation = Default
Calling Party Number = 4125551212
ConnectedParty Transformations
ConnectedLineId Presentation = Default
ConnectedName Presentation = Default
Called Party Transformations
Called Party Mask =
Discard Digits Instruction = PreDot
Prefix =
Called Number = 14125550000
Route List :Route List Name= HQ-RL
RouteGroup: RouteGroup Name= Standard Local Route Group
PreTransform Calling Party Number = 4125551212
PreTransform Called Party Number = 914125550000
Calling Party Transformations
External Phone Number Mask = Default
Calling Party Mask =
Prefix =
Calling Party Number = 4125551212
Called Party Transformations
Called Party Mask =
Discard Digits Instruction =
Prefix =
Called Number = 914125550000
Alternate Matches
Note: Information Not Available

Refer to the exhibit. A collaboration engineer is troubleshooting an issue where the PSTN calls of a Cisco UCM IP phone user are not reaching the PSTN gateway. Which action resolves the issue?

A. Change the calling search space of the user\\'s line or device.

B. Change the “Call Classification” to “OnNet” on the route pattern.

C. Ensure that the user\\'s phone is assigned to a device pool with the correct local route settings.

D. Deselect “Block this pattern” on the route pattern.

Correct Answer: D

QUESTION 2

An administrator is troubleshooting a one-way audio issue for a call that uses H.323 protocol in slow-start mode. The administrator requests that the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call is provided. The H.225 and H.245 messages for one of the one-way audio calls are gathered and the call flow has not invoked any media resources. Where is the RTP IP and port information for both sides found?

A. H.245 Terminal Capability Set

B. H.245 Open Logical Channel

C. H.225 Connect

D. H.245 Open Logical Channel Ack

Correct Answer: D

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

---

QUESTION 3

An administrator is configuring a SIP trunk to an ITSP. The SIP connection will traverse from a Cisco UCM to the ISTP through a Cisco Unified Border Element. The ITSP has indicated that they require an in-band method for DTMF. Which command on the outbound dial-peer to the ITSP will meet this requirement?

A. router (config-dial-peer) dtmf-relay sip-notify

B. router (config-dial-peer) dtmf-relay sip-kpml

C. router (config-dial-peer) dtmf-relay h245-alphanumeric

D. router (config-dial-peer) dtmf-relay rtp-nte

Correct Answer: D

QUESTION 4

When an administrator troubleshoots H.323 call setup, which message gives an alert that the called party is being notified about the call?

A. ALERTING

B. PROCEEDING

C. CONNECT

D. RINGING

Correct Answer: A

QUESTION 5

An engineer is configuring a Cisco UCM solution. The requirements state that some users in one location will receive calls from a number during work hours 09AM to 5PM, and another group will get the calls from the same number outside this defined timeslot. Users will also change the outgoing number when reaching out to customers based on the same time-of-day routing rules. Which feature is needed to allow for this type of configuration?

A. partitions

B. CSS

C. route groups

D. route lists

Correct Answer: B