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1. Where does an IP phone obtain the extension number and speed-dial settings from?
   A. the settings that are configured on the physical phone
   B. the registration file that the phone receives from the Cisco Unified Communications Manager
   C. the device and line configuration in Cisco Unified Communications Manager, during the registration process
   D. the default device profile that is configured in Cisco Unified Communications, Manager

   Answer: C

   Explanation:
   When we configure IP phone profile in CUCM that time we also configure extension number and speed dial as per requirement. When IP reachability gets establish between IP phone and CUCM then phone will download config file from CUCM during initial registration process.

   Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/3_1_2/ccmcfg/b06phone.html

2. Which web-based application that is accessed via the Cisco Unified Communications Manager Administration GUI generates reports for troubleshooting or inspecting cluster data?
   A. Cisco Unified Serviceability alarms
   B. Cisco Unified RTMT Trace and Log Central
   C. Cisco Unified RTMT Monitor
   D. Cisco Unified Reporting tool

   Answer: D

3. Which statement about device mobility is true?
   A. When local route groups are used, there is no need to configure device mobility groups or phone device CSSs as long as phone line CSSs are used.
   B. When local route groups are used, you must configure device mobility groups and phone device CSSs.
   C. When the device mobility group at the home device pool and roaming device pool are not the same, the Phone will keep the home region.
   D. When device mobility groups at the home device pool and roaming device pool are the same, the phone will keep the home MRGL setting.

   Answer: A

4. Refer to the SDI trace in the exhibit A PSTN call arrived at the MGCP gateway that is shown in the SDI trace.
   If the caller ID that is displayed on the IP phone is 087071 222 and the HQ_cing_pty_CSS contains the HQ_cing_pty_Pt partition, which exhibit shows the correct gateway digit manipulation?
   A. Exhibit A
   B. Exhibit B
   C. Exhibit C
   D. Exhibit D

   Answer: D

   Explanation:
   Actual incoming number is 14-087071 222 but next to this information in trace we can see two digits are
stripped which is international code hence D is valid answer.

5. When a database replication issue is suspected, which three tools can be used to check the database replication status? (Choose three.)
   A. Cisco Unified Communications Manager RTMT tool
   B. Cisco Unified Communications Manager Serviceability interface
   C. Cisco Unified Reporting
   D. Cisco Unified Communications Manager CLI interface
   E. Cisco IP Phone Device Stats from the Settings button
   F. Cisco Unified OS Administration interface
   Answer: A, C, D
   Explanation:
   Reference:

6. Which of these reasons can cause intrasite calls within a Cisco Unified Communications Manager cluster to fail?
   A. The route partition that is configured in the CCD requesting service is not listed in the calling phone CSS
   B. The trunk CSS does not include the partition for the called directory number.
   C. The MGCP gateway is not registered
   D. The calling phone does not have the correct CSS configured
   E. The calling phone does not have the correct partition configured.
   Answer: D
   Explanation:
   To make a successful call within CUCM cluster following condition should satisfy.
   Reference:

7. Refer to the exhibit.
   When a Cisco IP Communicator phone roams from San Jose (SJ) to RTP, the Cisco IP Communicator physical location and the device mobility group change from SJ to RTP. All route patterns are assigned a route list that points to the local route group. All device pools are configured to use the local route group. Which statement is true when the roaming phone places an AAR call?
   A. Since globalized call routing is not configured, then the SJ gateway will be used in this case
   B. The phone will use the AAR CSS that contains the SJ_PSTN partition. The call will egress at the SJ gateway
   C. The phone will use the AAR CSS that contains the RTP_PSTN partition. The call will egress at the SJ gateway
   D. The phone will use the AAR CSS that contains the SJ_PSTN partition. The call will egress at the RTP
E. The phone will use the AAR CSS that contains the RTP_PSTN partition. The call will egress at the RTP gateway.

Answer: D

Explanation:

Cisco Unified Communications Manager Version 7.0 introduced the Local Route Group feature. When using local route groups, gateway selection is totally independent of the matched route pattern and referenced route list and routegroup. The use of the Local Route Group feature makes no changes regarding roaming-sensitive settings. The application of these settings always makes sense when roaming between sites. The settings have no influence to the gateway selection and the dial rules that a user must follow. However, the dial plan-related part of Device Mobility changes substantially with the new dial plan concept. This concept allows a roaming user to follow the home dial rules for external calls but use the local gateway of the roaming site. In this case, when the device mobility group is not the same for San Jose and RTP, the Device Mobility related settings are not applied. The phone device keeps its San Jose-specific configuration. Despite the San Jose-specific configuration on the phone, the PSTN calls that originate from the roaming phone are routed via the local PSTN gateway (RTP GW) and are based on the route list and device pool local route group settings. The San Jose-specific dial plan is used. Also, AAR remains configured with the San Jose-specific configuration, but if the San Jose dial plan and San Jose AAR CSS permit and if the AAR group contains the prefix that can be applied in RTP, then AAR can work.

8. Refer to the exhibits.

Low latency queuing has been implemented on the HO and BR1 routers to allow five G.729 calls. Callers are still experiencing poor audio, in particular choppy and delayed audio during traffic congestion. This problem occurs even with just one active call.

Which two actions will solve the issue?

A. Change the codec type to G.711.

B. Configure RSVP call admission control.

C. Configure Link Fragmentation and Interleave on the WAN links.

D. Configure RTP header compression on the WAN links.

E. Increase the priority queue bandwidth to 80 Kb/s.

F. Configure location settings in Cisco Unified Communications Manager to 120 Kb/s.

Answer: C, D

Explanation:

Below link is very good to understand this concept.


9. Refer to the exhibit.

When calling 911, which gateway/route list is defined in the route pattern in Cisco Unified Communications Manager and used to route matched digits to the PSTN?

A. SEP002290BA361B

B. standardLocalRG
C. RouteListCdrc
D. LRG_RL
E. nodeld = 1
F. BRANCH
Answer: D
Explanation:
logs clearly showing route list name.

10. Which Cisco Unified Communications Manager troubleshooting tool can be used to look at detailed specific events, such as dial plan digit analysis, as they are happening?
A. traceroutes
B. RTMT real-time trace
C. Cisco Unified Communications Manager alerts
D. Cisco Unified Dialed Number Analyzer
E. RTMT performance log viewer
F. syslog output
Answer: B

11. Refer to the exhibits.
MOH has been configured to run from flash at the BR1 site. The HQ phones and MOH server are placed in the Default region through the Default device pool. The BR1 phones are placed in the BR1 region through the BR1 device pool. The region configuration between Default and BR1 only permits G.729 codec.
When an IP phone user at the HQ site places a BR1 caller on hold, the BR1 caller hears tone on hold. Which of the following can cause this issue?
A. Multicast routing is not enabled on the BR1 router.
B. The command ip pim separate-dense-mode is missing from interface VLAN 120 at the SRST router in BR1.
C. The MOH server is unable to stream MOH using G.711 codec because of the regions configuration.
D. The command route 10.1.120.1 must be added to the multicast moh 239.1.1.1 port 16384 command at the SRST router in BR1.
E. The Max Hops is too small in the MOH configuration
Answer: B
Explanation:
The router runs IP Multicast routing and IP PIM sparse-dense mode on any physical interface that must participate in multicast (PIM is in either sparse or dense mode, but the interface can be configured to forward sparse mode, dense mode, or both).

12. An IP phone that is connected through a Cisco Catalyst 3750 Series Switch is failing to register with the subscriber as a backup server. When the user presses the settings button on the phone, only the
Cisco Unified Communications Manager publisher shows as registered.

What is the most likely cause for this issue?

A. The phone does not have the correct Cisco Unified Communications Manager group in the device configuration page.
B. The Cisco Unified Communications Manager group that is applied through the device pool is misconfigured.
C. The ip-helper address command for the subscriber is not configured on the switch port.
D. The subscriber does not have the correct device pool configured.
E. The enterprise phone configuration does not have the call control redundancy enabled.

Answer: B

Explanation:
Yes if The Cisco Unified Communications Manager group that is applied through the device pool is misconfigured then IP phone doesn’t recognize the subscriber IP address.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_0_1/ccmcfg/b02devpl.html

13. Which step in the problem-solving model is important to accurately interview end users to get all the pertinent details of the problem?

A. Implement Action Plan
B. Define the Problem
C. Consider the Possibilities
D. Create Action Plan
E. Gather Facts
F. Observe Results
G. Restart Problem-Solving Process
H. Problem Resolved

Answer: E

Explanation:
http://www.cisco.com/en/US/docs/internetworking/troubleshooting/guide/tr901.html Step 2 Gather the facts that you need to help isolate possible causes. Ask questions of affected users, network administrators, managers, and other key people. Collect information from sources such as network management systems, protocol analyzer traces, output from router diagnostic commands, or software release notes.

14. Refer to the exhibit.
The exhibit shows the output of debug isdn q931. An inbound PSTN call was received by a SIP gateway that is reachable via a SIP trunk that is configured in Cisco Unified Communications Manager. The call failed to ring extension 3001. If the phone at extension 3001 is registered and reachable through the gateway inbound CSS, which three actions can resolve this issue? (Choose three.)

A. Change the significant digits for inbound calls to 4 on the SIP trunk configuration in Cisco Unified Communications Manager.
B. Configure the digit strip 4 on the SIP trunk under Incoming Called Party Settings in Cisco Unified Communications Manager.
Communications Manager.

C. Configure a translation pattern in Cisco Unified Communications Manager that can be accessed by the trunk CSS to truncate the called number to four digits.

D. Configure a called-party transformation CSS on the gateway in Cisco Unified Communications Manager that includes a pattern that transforms the number from ten digits to four digits.

E. Configure a voice translation profile in the SIP Cisco IOS gateway with a voice translation rule that truncates the number from ten digits to four digits.

F. Configure the Cisco IOS command `num-exp 2288223001 3001` on the gateway ISDN interface.

Answer: A, C, E

15. Which of these is used by the Cisco IP phone to relay to the switch the information regarding how much power is needed?

A. the Cisco Discovery Protocol
B. IEEE 802.10 protocol
C. Cisco IP phones always use a fixed power consumption based on the resistor, which is specific to the model
D. The switch model determines how much power is consumed by the different phone models

Answer: A

Explanation:

If CDP is enabled on the switch, 15.4W is initially allocated, and then further refined when the CDP message is received from the PD

Link:

16. Refer to the exhibit.

Assume a centralized Cisco Unified Communications Manager topology with the headquarters at RTP and remote located at the U.K. All route patterns are assigned a route list that contains a route group pointing to the local gateway. RTP route patterns use the RTP gateway, and U.K. route patterns use the U.K. gateway.

When a U.K. user logs into an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000, which statement about the emergency call is true?

A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
C. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
D. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
E. The call will fail.

Answer: B

17. Which issue would cause an MGCP gateway to fail to register with Cisco Unified Communications Manager?

A. missing the configuration command `isdn bind-13 ccm-manager` under the ISDN interface
18. Refer to the exhibits.

The HG_MRG that is shown in the exhibit is assigned to an MRGL, which is configured at the HQ phones. A call exists between two HQ phones that use G.711 codec. When one of the HQ users attempts to conference a BR phone across the WAN, the conference fails. The SDI trace shows an error “No transcoder device configured.”

Which statement indicates the correct resolution or reason for the issue?

A. The BR phone does not have access to the HO_Conf bridge
B. The BR phone does not have access to the CFB_2 bridge
C. The BR phone does not have access to a transcoder
D. The CFB_2 bridge should be removed from the HQ_MRG and assigned to an MRG that is not assigned to an MRGL
E. The CFB_2 bridge should be listed last in the HO_MRG

Answer: E

Explanation:

In the group MRG_HQ are two conference system in the following sequence is entered:

1. Software = CFB_2
2. Hardware = HO_Conf

It is as always the first group CFB_2 used. But as they only support G711 calls the call will fail. Only the conference originator need access to the transcoder. See TVOICE V 2 6-71

19. Refer to the exhibits.

When a remote Cisco Unified Communications Manager learns the advertised patterns that are shown in the exhibit, which patterns would be shown in the Cisco Unified Communications Manager RTMT tool?

A. 2XXX and the ToDiD will be 0:+498950555
B. 2XXX and the ToDiD will be 0+498953121
C. +4989505552XXX and the ToDiD will be 0:
D. +498953121 2XXX and the ToDiD will be 0:
E. Both +4989505552XXX and +4989531 21 2XXX will be advertised with ToDiD of 0:

Answer: A

20. Refer to the exhibits.

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When the IP phone 2001 places a call to 9011 49403021 56001, the call is sent to the Cisco Unified Border Element as 40302156001 which is what the ITSP expects to receive. The ITSP support personnel claim that they never saw the call. Issuing the debug CCSIP message command on the Cisco Unified Border Element results in the message "SIP/2 0 404 Not Found".

Refer to the Cisco Unified Border Element configuration, debug voice dial and ccsp messages exhibits.

Which situation can cause this issued?

A. The Cisco Unified Border Element is configured as an MGCP gateway also so that the call is attempted via the PSTN
B. The command allow-connections sip to h323 is missing
C. SIP error 404 means that a codec mismatch occurred Cisco Unified Communications Manager is sending the call as an early offer with G.711 codec.
D. The Cisco Unified Communications Manager is misconfigured. The SIP invite should be sent to the ITSP at 10.1.2.1.2. The debug ccsp message shows the SIP invite being sent to 10.12.1.2.

Answer: B

Explanation:

As we can see in logs, the call is between two different signaling devices i.e. SIP and H.323 hence The command allow-connections sip to h323 is mandatory.

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