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Exam : 300-075

Title : Implementing Cisco IP Telephony & Video, Part 2 v1.0

Vendor : Cisco

Version : DEMO
NO.1 Which device is used to connect to the H.323 gatekeeper?
A. SIP trunk
B. H.323 gateway
C. H.323 trunk
D. MGCP gateway
Answer: C

NO.2 A new administrator at Company X has deployed a VCS Control on the LAN and VCS Expressway in the DMZ to facilitate VPN-less SIP calls with users outside of the network. However, the users report that calls via the VCS are erratic and not very consistent. What must the administrator configure on the firewall to stabilize this deployment?
A. The VCS Control should not be on the LAN, but it must be located in the DMZ with the Expressway.
B. The firewall at Company X requires a rule to allow all traffic from the DMZ to pass to the same network that the VCS Control is on.
C. A TMS server is needed to allow the firewall traversal to occur between the VCS Expressway and the VCS Control servers.
D. The firewall at Company X must have all SIP ALG functions disabled.
Answer: D

NO.3 Which action is performed by the Media Gateway Control Protocol gateway with SRST configured, when it loses connectivity with the primary and backup Cisco Unified Communications Manager servers?
A. The gateway falls back to the H.323 protocol for further call processing.
B. The gateway waits for the primary Cisco Unified Communications Manager server to come alive.
C. The MGCP calls are queued up until the Cisco Unified Communications Manager servers are online.
D. All MGCP call processing is interrupted until the Cisco Unified Communications Manager servers are online.
E. The gateway continues to make an attempt to connect to the backup Cisco Unified Communications Manager server.
F. The gateway continues with the MGCP call processing without any interruption.
Answer: A

NO.4 Refer to the exhibit. The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit. How many route lists and route groups should be configured for AAR at a minimum?
A. two route lists and two route groups for each site
B. None. The AAR CSS can point directly to the route pattern.
C. a single route list with a local route group for each site
D. a single route list and four route groups for each site

Answer: C

NO.5 Which code snippet is required for SAF to be initialized?
A. router eigrp
B. topology base
C. Service Family
D. External-Client

Answer: A

NO.6 What is the default value for the Drop Ad Hoc Conference service parameter?
A. When No On-Net Parties Remain in the Conference
B. When No Off-Net Parties Remain in the Conference
C. Drop Ad Hoc Conference When Creator Leaves
D. Never

Answer: D
NO.7 An engineer opens up a TAC case and wants to implement CCD/SAF service. Which menu navigation sequence must an engineer navigate to when configuring a hosted DN pattern?
A. Advanced Features > SAF > Hosted DN Pattern
B. Advanced Features > Call Control Discovery > Hosted DN Pattern
C. Call Routing > SAF > Hosted DN Pattern
D. Call Routing > Call Control Discovery > Hosted DN Pattern
Answer: D

NO.8 A CUCM cluster has been set up with one to one (1:1) call processing redundancy. What two occurrences will seen when the primary subscriber fails? (Choose two)
A. 125 SCCP phones per second will be able to register to the secondary subscriber.
B. 125 SIP phones per second will be able to register to the secondary subscriber
C. The secondary subscriber will start its CallManager service
D. The secondary subscriber will start its TFTP service
E. 40 SIP phones per second will be able to register to the secondary subscriber.
F. 40 SCCP phones per second will be able to register to the secondary subscriber.
Answer: A,E

NO.9 After forgetting to log out of his IP phone in the main office, an Extension Mobility user is unable to log in to a different IP phone at a remote office. Which option is a possible reason for the problem?
A. The user's Extension Mobility profile is misconfigured.
B. The user can log in to only one device at a time.
C. The device pool is misconfigured.
D. The phone at the remote location is a different model than the phone in the user's main office.
Answer: B

NO.10 An engineer must enable video desktop sharing between a Cisco Unified Communications Manager registered video endpoint and a Cisco VCS registered video endpoint. Which protocol must be enabled in SIP profile for VCS SIP trunk on Cisco Unified Communications Manager?
A. RDP
B. H.263
C. H.224
D. H.264
E. BFCP
Answer: E

NO.11 Refer to the exhibit. A user in RTP calls a phone in San Jose during congestion with Call Forward No Bandwidth (CFNB) configured to reach cell phone 4085550150. The user in RTP sees the message "Not Enough Bandwidth" on their phone and hears a fast busy tone. Which two conditions can correct this issue? (Choose two.)
A. The called phone (San Jose) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
B. The called phone (San Jose) needs to have AAR Group value of AAR under the AAR Settings.
C. The calling phone (RTP) needs to have AAR Group value of AAR under the AAR Settings.
D. The called phone (San Jose) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.
E. The calling phone (RTP) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
F. The calling phone (RTP) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

Answer: B, D

Explanation:
Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

NO.12 Refer to the exhibit. Which pattern will be advertised try the Cisco Unified Communications Manager?
A. 3XXX and the ToDID will be 0:+
B. 3XXX and the ToDID will be 0:.
C. 3XXX and the TnOID will be 0:44228822.
D. 3XXX and the ToDID will be 44228822.
E. 3XXX and the ToDID will be 0:+44228822.

Answer: B

NO.13 Which default DSCP marking value do telepresence endpoints use for video?
A. CS4
B. AF41
C. CS2
D. EF

Answer: A

NO.14 Which two bandwidth management parameters are available during the configuration of Cisco Unified Communications Manager regions? (Choose two.)
A. Max Audio Bit Rate
B. Max Video Call Bit Rate (Includes Audio)
C. Default Video Call Rate
D. Max Number of Video Sessions
E. Default Audio Call Rate

Answer: A,B

NO.15 Which command can be used to manually send the MGCP gateway to register with the secondary Cisco Unified Communications Manager server?
A. ccm-manager register backup
B. not supported
C. mgcp use backup
D. ccm-manager switchover-to-backup

Answer: D

NO.16 When considering Extension Mobility, what happens if a user logs into a phone for which the user does not have a user device profile?
A. If a default device profile for this phone has been configured, it is loaded.
B. Another user device profile is loaded.
C. The phone reboots with an error.
D. The user cannot log in.
Answer: A

NO.17 A voice engineer is enabling video capabilities between H.323 and SIP endpoints. Which component allows for standardized caller addresses between the endpoints?
A. policy service
B. search rules
C. transform
D. SIP route pattern
Answer: C

NO.18 Cisco Unified Communications Manager is configured with CAC for a maximum of 10 voice calls. Which action routes the 11th call through the PSTN?
A. Configure Cisco Unified Communications Manager AAR.
B. Configure Cisco Unified Communications Manager RSVP-enabled locations.
C. Configure Cisco Unified Communications Manager locations.
D. Configure an SIP trunk to the ISR.
Answer: A

NO.19 Which DSCP service parameter can be configured for Cisco TelePresence endpoints and Cisco Communication Manager Endpoints?
A. DSCP for audio calls
B. DSCP for audio portion of Cisco TelePresence calls
C. DSCP for audio portion of video calls
D. DSCP for video calls
Answer: C

NO.20 Which two features require or may require configuring a SIP trunk? (Choose two.)
A. Cisco Unified Mobility
B. SIP gateway
C. Cisco Device Mobility
D. registering a SIP phone
E. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
Answer: B,E
Explanation:
All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the

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ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones.

**NO.21** Refer to the exhibit. When a user presses a speed dial to +442079460255 when the SAF network is down, which event should occur?

<table>
<thead>
<tr>
<th>Pattern</th>
<th>TimeStamp</th>
<th>Status</th>
<th>Protocol</th>
<th>AgentID</th>
<th>IP Address</th>
<th>ToDID</th>
</tr>
</thead>
<tbody>
<tr>
<td>+442079460255</td>
<td>2010/05/05 09:48:03</td>
<td>Reachable</td>
<td>SIP</td>
<td>CID10 1.5.11</td>
<td>10.1.5.11 (5DE60)</td>
<td>0</td>
</tr>
<tr>
<td>+442079460255</td>
<td>2010/05/05 09:48:03</td>
<td>Reachable</td>
<td>H323</td>
<td>CID10 1.5.11</td>
<td>10.1.5.11 (476067)</td>
<td>0</td>
</tr>
</tbody>
</table>

A. The call will reroute via the PSTN with the constructed PSTN number as 442079460255.
B. The call will reroute via the PSTN with the constructed PSTN number as 00442079460255.
C. The call will fail because the ToDID is 0:
D. The call will fail because the called number will be 2079460255.
E. The call will reroute via the PSTN with the constructed PSTN number as +442079460255.

**Answer:** E

**NO.22** Which two results occur in SRST mode if the pickup commands is configured under the call manager fallback section of a router?

A. The PickUp soft key is on all SRST phones.
B. Calls coming into a Cisco Unified Communications Manager registered phone can be picked up by a SRST phone.
C. The GPickUp soft key in on all SRST phones.
D. Calls ringing an unassigned directory number are forwarded to the auto attendant.
E. Calls coming into one directory number can be picked up from another directory number.

**Answer:** A,E

**Explanation:**
The pickup command was introduced to enable the PickUp soft key on all Cisco Unified IP Phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST. Configuring the pickup command enables the PickUp soft key on all SRST phones. You can then press the PickUp key and answer any currently ringing IP phone that has a DID called number that matches the configured telephone-number. This command does not enable the Group PickUp (GPickUp) soft key.


**NO.23** The administrator at Company X is trying to set up Extension Mobility and has done these steps:
- Set up end users accounts for the users who need to roam
- Set up a device profile for the type of phones users will be allowed to log in

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Users have reported to the administrator that they are unable to log in to the phones designated for Extension Mobility. Which two options are the two reasons for this issue? (Choose two.)
A. The user device profile is not associated to the correct end user.
B. The username must be numeric only and must match the DN.
C. The user must ensure that their main endpoint is online and registered, otherwise they cannot log in elsewhere.
D. Extension Mobility has not been enabled under Enterprise Parameters.
E. The Extension Mobility service has not been enabled under the Cisco Unified Serviceability Page.
Answer: A,E

NO. 24 Which two features are part of Cisco Unified Mobility? (Choose Two)
A. Device Mobility
B. Mobile Voice Access
C. Extension Mobility Cross Cluster
D. Enterprise Feature Access
E. Shared line
Answer: B,D

NO. 25 Which two actions ensure that the call load from Cisco TelePresence Video Communication Server to a Cisco Unified Communications Manager cluster is shared across Unified CM nodes? (Choose two.)
A. In VCS set Unified Communications mode to Mobile and remote access and configure each Unified CM node.
B. Create a single traversal client zone in VCS with the Unified CM nodes listed as location peer addresses.
C. Create a neighbor zone in VCS with the Unified CM nodes listed as location peer addresses.
D. Create one neighbor zone in VCS for each Unified CM node.
E. Create a VCS DNS zone and configure one DNS SRV record per Unified CM node.
Answer: C,E

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