



**642-456**

**(Implementing Cisco Unified Communications Manager Part 2)**

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## Question: 1

**Which three statements are true in order for inbound PSTN calls to work in an H.323 gateway configured with Cisco Unified Communications Manager? (Choose three.)**

- A. The H.323 gateway should be registered with Cisco Unified Communications Manager.
- B. The command `h323-gateway voip tech-prefix` should be configured on the H.323 interface.
- C. The command `h323-gateway voip bind srcaddr` should be configured on the H.323 interface.
- D. A VoIP dial peer pointing to Cisco Unified Communications Manager should be configured.
- E. The command `h323-gateway voip id` should be configured under the H.323 interface.
- F. A pots dial peer with `direct-inward-dial` and `incoming-called number` should be configured.

Answer: C, D, F

## Question: 2

**Which two statements describe RSVP-enabled locations-based CAC? (Choose two.)**

- A. RSVP can be enabled selectively between pairs of locations.
- B. Using RSVP for CAC simply allows admitting or denying calls based on a logical configuration that is ignoring the physical topology.
- C. RSVP is topology aware, but only works with full mesh networks.
- D. An RSVP agent is a Media Termination Point that the call has to flow through.
- E. RSVP and RTP are used between the two endpoints.

Answer: A, D

## Question: 3

**An administrator was testing a new implementation of Cisco Unified Communications Manager Extension Mobility. When the administrator tried to log out of Cisco Unified Communications Manager Extension Mobility, the following message appeared on the user's IP phone:**

**"To set up speed dials and other services for your phone, please go to [https://ccm\\_ip@ccmuser/showHome.do](https://ccm_ip@ccmuser/showHome.do)"**

**What is the most likely cause for this message?**

- A. The administrator forgot to subscribe the IP phone to Cisco Unified Communications Manager Extension Mobility Service.
- B. The administrator did not associate the User Device Profile with Cisco Unified Communications Manager Extension Mobility Service.
- C. The Cisco Unified Communications Manager Extension Mobility checkbox was not selected for that particular phone.

D. The IP phone user needs to go into ccmuser and set up speed dials in order for Cisco Unified Communications Manager Extension Mobility to work.

Answer: B

Question: 4

**A branch site has a group of salespeople that take orders from customers. The site needs to be able to distribute calls evenly to the salespeople when connectivity to the main site is lost.**

**Which two configurations are correct? (Choose two.)**

- A. Configure hunt groups on the Cisco Unified CME, which is operating in SRST mode.
- B. All branch site IP phones will need to be preconfigured in the gateway.
- C. Only the salespeople's phones will need to be preconfigured in the gateway.
- D. The branch site will need a dedicated UCCX server to queue the customer calls.
- E. The gateway must be configured to use H.323 to communicate with the Cisco Unified Communications Manager cluster.
- F. A digital circuit will be required so that DNIS can be used to route the customer calls to the salespeople's queue.

Answer: A, C

Question: 5

**You have configured SRST at a remote site on an MGCP gateway. During testing, you find that IP phones are not registering with the SRST router when the IP WAN fails.**

**Which three potential problems need to be investigated? (Choose three.)**

- A. No dial peers have been added to the SRST router.
- B. No SRST reference address is included in the device pool.
- C. The proper service command has not been added to the SRST router.
- D. The max-ephones command is missing in the SRST router.
- E. The max-dn command is missing in the SRST router.
- F. The ccm-manager fallback-mgcp command is missing in the SRST router.

Answer: B, D, E

Question: 6

**Which feature addresses the issues with AAR for mobile users in Cisco Unified Communications Manager?**

- A. Extension Mobility
- B. Device Mobility
- C. Mobile Connect
- D. Mobile Voice for Access

Answer: B

Question: 7

Drop

- MGCP with PRI
- MGCP with CAS
- H.323 with PRI
- I.1323 with CAS

Calls can be preserved.

Calls will be dropped.

Answer:

- MGCP with PRI
- MGCP with CAS
- H.323 with PRI
- H.323 with CAS

Calls can be preserved.

Calls will be dropped.

Question: 8

**What does the addition of the Tcl paramspace command allow the application to do?**

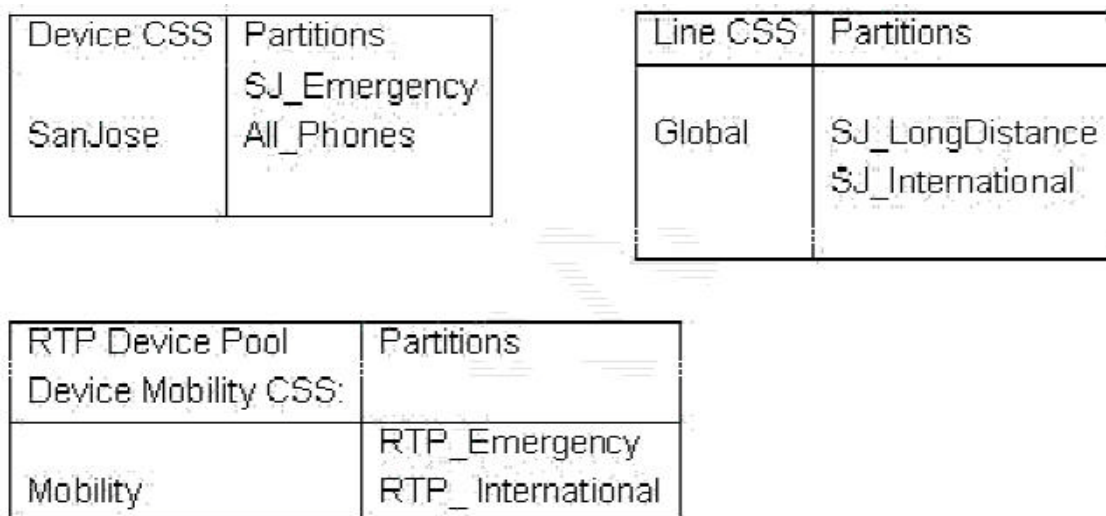
- A. Access the data on an external server
- B. Specify parameters for a single application
- C. Set aside memory for application variables

- D. Share parameters between different call applications
- E. Consolidate all application parameters on a single call application list

Answer: D

Question: 9

Refer to the exhibit. When a Cisco IP Communicator Phone roams from SJ to RTP, the physical location for the Cisco IP Communicator Phone changes and the Device Mobility Group changes from SJ to RTP. All route patterns are assigned to partitions and configured to utilize the local gateway. After roaming to RTP, if the user dials 911, which statement about the call routing is true?

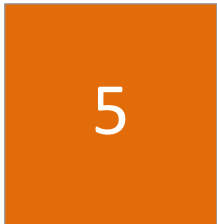


- A. The call will use the RTP gateway.
- B. The call will use the SJ gateway because the device's origin is SJ.
- C. Emergency calls will go through the RTP gateway as first priority. The SJ gateway will be used as backup.
- D. Emergency calls will always use the local gateway regardless of Device Mobility configurations.
- E. The call will use the SJ gateway because we keep the Device/Line CSS after roaming.

Answer: E

Question: 10

You have deployed a centralized call processing solution with a multicast MOH server at your central site on a different VLAN from your Cisco Unified Communications Manager servers and IP phones. When central site users place calls on hold, dead air or silence is heard.



**Which two actions will resolve this issue? (Choose two.)**

- A. Enable multicast routing on all the routers at the central site.
- B. Increase the TTL in the configuration of the MOH server to 2 so that packets can cross the VLAN boundary.
- C. Decrease the TTL configuration in the Cisco Unified Communications Manager server to 0 so that the multicast packets only go to the VLAN that contains the Cisco Unified Communications Manager server and IP phones.
- D. Enable multicast routing on only those router interfaces that connect the voice and MOH VLANs.
- E. Keep the TTL at 1 for the MOH server and increase the TTL for IP multicast routing to 2 on router interfaces.
- F. Configure ip-sparse mode on the router interfaces and increase the TTL on the routers to 2.

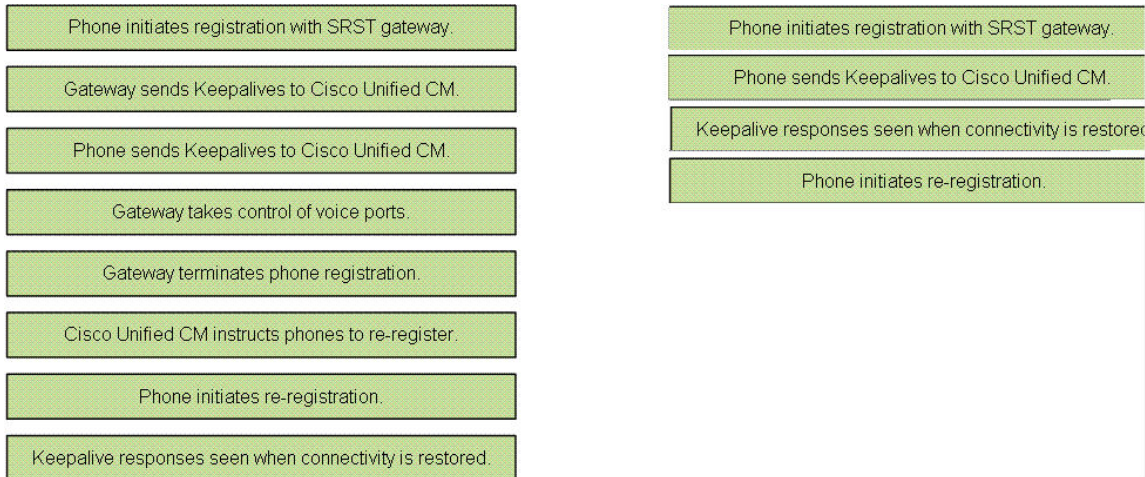
Answer: A, B

**Question: 11**

- Phone initiates registration with SRST gateway.
- Gateway sends Keepalives to Cisco Unified CM.
- Phone sends Keepalives to Cisco Unified CM.
- Gateway takes control of voice ports.
- Gateway terminates phone registration.
- Cisco Unified CM instructs phones to re-register.
- Phone initiates re-registration.
- Keepalive responses seen when connectivity is restored.

Four empty yellow rectangular boxes for selecting answers.

Answer:



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