



Vendor: Cisco

Exam Code: 642-457

Exam Name: Implementing Cisco Unified
Communications Manager, Part 2 v8.0 (CIPT2 v8.0)

Version: DEMO

QUESTION 1

When configuring Digit Manipulation, where exactly should you apply Digit Stripping?

- A. globally to all calls
- B. voice port
- C. Plain Old Telephone Systems (POTS) dial peer
- D. Voice Over Internet Protocol (VOIP) dial peer
- E. trunk group
- F. Non-Facility Associated Signaling (NFAS) interface

Answer: C

QUESTION 2

In a Centralized Call processing architecture, you have deployed Extension Mobility (EM) feature. After the deployment of EM, when one of the end-users tries to login to the IP phone, the Error 25 is displayed on the screen.

What three things should you do to resolve this issue? (Choose three.)

- A. upgrade the firmware of the IP Phone to the latest version
- B. activate EM feature service under Cisco Unified Serviceability
- C. associate EM Device profile with the end-user
- D. subscribe the MAC address of the IP Phone to EM Service
- E. update EM Phone Service URL to point to the publisher
- F. subscribe device profile to EM phone service in case the enterprise subscription of EM Service is disabled

Answer: BCD

QUESTION 3

Which command should you use on the gatekeeper to specify the address of a Cisco Unified Communications Manager IP address?

- A. gw-type-ip
- B. gw-type-prefix-ip
- C. gw-ip-cm
- D. gw-type-prefix
- E. gw-ip-type
- F. gw-prefix-ip

Answer: D

QUESTION 4

Which of the following are two functions that ensure that the telephony capabilities stay operational in the remote location Cisco Unified SRST router? (Choose two)

- A. Automatically detecting a failure in the network.
- B. Initiating a process to provide call-processing backup redundancy.
- C. Notifying the administrator of an issue for manual intervention.
- D. Proactively repairing issues in the voice network.

Answer: AB

QUESTION 5

Which three of the following are steps in configuring MGCP Fallback and Cisco Unified SRST?
(Choose three)

- A. Define the SRST reference for phones in the Device Pool configuration
- B. Enable and configure the MGCP fallback and Cisco Unified SRST features on the IOS gateways.
- C. Implement a simplified SRST dial plan on the remote-site-gateways to ensure connectivity for remote-site phones in SRST mode.
- D. Enable SIP trunking between both remote and hub sites to provide mesh coverage.
- E. Define the SRST reference in the configuration on the IP Phones.
- F. Enable and configure the MGCP fallback on the IOS gateway but not Cisco Unified SRST since it is enabled automatically.

Answer: ABC

QUESTION 6

What is the purpose of a SAF Client?

- A. To decode address information and route calls to and from the end points
- B. To pass IP information from the CUCM to the endpoint
- C. To learn about and advertise or subscribe information about SAF network services
- D. To reside in the Cisco IOS software, and to communicate with the SAF forwarder

Answer: C

QUESTION 7

How can the location setting be modified to resolve poor call quality?

- A. No adjustment to location setting is needed
- B. Mark the bandwidth between the locations as unlimited
- C. Decrease the audio bandwidth setting
- D. Remove the audio bandwidth setting

Answer: C

QUESTION 8

Cisco Unified border element is configured to support RSVP-based CAC. When is the RSVP path and reservation message sent and received?

- A. Immediately after the call setup message is received and the reservation message is received after H.245 capabilities negotiation is completed.
- B. The path and reservation messages are sent and received after the H.245 capabilities negotiation is completed.
- C. The path and reservation messages are sent and received immediately after the call setup message is received.
- D. The path is setup once the global command call rsvp-sync is configured.

Answer: C

QUESTION 9

What user profile is used to define the settings for a user on login?

- A. Device Profile
- B. Group Profile
- C. Pool Profile
- D. Specific Profile

Answer: A

QUESTION 10

Which minimum configuration is needed for the SAF Internal Client to register with this SAF Forwarder?

- A.

```
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopback1 transport tcp port 5060 i
```
- B.

```
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopback1 transport tcp port 5060 !
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx
!
profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2
i
```
- C.

```
router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
```

```
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopback1 transport tcp port 5060 !
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx
!
profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2
!
channel 1 vrouter SAF asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1
!
```

- D. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
!
channel 1 vrouter SAF asystem 1
- E. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base exit-sf-topology
exit-service-family i

Answer: A

QUESTION 11

Refer to the exhibit. Which pattern is advertised by the Cisco Unified Communications Manager?

Hosted DN Group	
— Hosted DN Group Info	
Name *	<input type="text" value="BR1_TEHO"/>
Description	<input type="text"/>
PSTN Failover Strip Digits	<input type="text" value="0"/>
PSTN Failover Prepend Digits	<input type="text"/>
<input type="checkbox"/> Use HostedDN as PSTN Failover	

Hosted DN Pattern	
Hosted DN Pattern Info	
Hosted Pattern*	<input type="text" value="+4420!"/>
Description	<input type="text"/>
Hosted DN Group*	<input type="text" value="BR1_TEHO"/> ▼
PSTN Failover Strip Digits	<input type="text" value="0"/>
PSTN Failover Prepend Digits	<input type="text"/>
<input checked="" type="checkbox"/> Use HostedDN as PSTN Failover	

- A. +4420!
- B. +4420! And the ToDID will be 0:
- C. +4420! And the ToDID will be +4420!
- D. +4420! And the ToDID will be 0:+4420!

Answer: D

QUESTION 12

Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?

- A. Use the command bandwidth 10.
- B. Use the command bandwidth 1280.
- C. Use the command bandwidth 160.
- D. Use the command bandwidth session 10.

Answer: B

QUESTION 13

Which action configures AAR to route the calls that have been rejected by the gatekeeper CAC through the PSTN?

- A. Configure Cisco IP Phones for AAR.
- B. Configure AAR to work with SRST.
- C. Configure AAR to work with CTI route points.
- D. This configuration is not possible using AAR.

Answer: D

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