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QUESTION 41

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 an SIP trunk to the ISR.
B. Configure Cisco Unified Communications Manager AAR.
C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
D. Configure Cisco Unified Communications Manager locations.

Answer: B

QUESTION 42

Which solution is needed to enable presence and extension mobility to branch office phones during a WAN failure?

A. SRST with MGCP fallback
B. SRST without MGCP fallback
C. Cisco Unified Communications Manager Express in SRST mode
D. SRST with VoIP dial peers to Cisco Unified Communications Manager Express

Answer: C

QUESTION 43

Which option configures the secondary dial tone option for SRST mode to let the users hear the dial tone for PSTN calls?

A. voice service voip

secondary dialtone 0

B. call-manager-fallback

secondary dialtone 0

C. dial-peer voice 1 pots

secondary dialtone 0

D. ccm-manager secondary dialtone 0

Answer: B

QUESTION 44

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 continues to make an attempt to connect to the backup Cisco Unified Communications Manager server.
B. The gateway falls back to the H.323 protocol for further call processing.
C. The gateway continues with the MGCP call processing without any interruption.
D. The gateway waits for the primary Cisco Unified Communications Manager server to come alive.
E. All MGCP call processing is interrupted until the Cisco Unified Communications Manager servers are online.
F. The MGCP calls are queued up until the Cisco Unified Communications Manager servers are online.

Answer: B

QUESTION 45

Which three commands are mandatory to implement SRST for five Cisco IP Phones? (Choose three.)

A. call-manager-fallback

B.   max-ephones

C.   keepalive

D.   limit-dn

E.   ip source-address

Answer: ABE

QUESTION 46

Which parameter should be set to prevent H.323 endpoints from registering to Cisco TelePresence Video Communication Server automatically?

A.   On the VCS, navigate to Configuration, Protocols, H.323, and set Auto Discover to off.

B.   On the VCS, navigate to Configuration, Protocols, H.323, and set Auto Registration to off.

C.   On the VCS, navigate to Configuration, Registration, Allow List, and set Auto Registration to off.

D.   On the VCS, navigate to Configuration, Registration, Configuration, and set Auto Registration to off.

Answer: A

QUESTION 47

Which two statements about configuring mobile and remote access on Cisco TelePresence Video Communication Server Expressway are true? (Choose two.)

A.   The traversal server zone on Expressway-C must have a TLS verify subject name configured.

B.   The traversal client zone and the traversal server zone Media encryption mode must be set to Force encrypted.

C.   The traversal client zone and the traversal server zone Media encryption mode must be set to Auto.

D.   The traversal client zone on Expressway-C Media encryption mode must be set to Auto.

E.   The traversal client zone and the traversal server zone must be set to SIP TLS with TLS verify mode set to On.

Answer: BE

QUESTION 48

Scenario:

There are two call control systems in this item. The Cisco UCM is controlling the DX650. the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS controll the the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

DP:

Locations:

CSS:

SRST:

SRST-BR2-Config:

- ■ **Name:** BR2
- ■ **Port:** 2000
- ■ **IP Address:** 10.1.5.15
- ■ **SIP Network/IP Address:** 10.1.5.15

BR2 Config:

```
voice service voip
 sip
  bind control source-interface
GigabitEthernet0/0/0.130
  bind media source-interface
GigabitEthernet0/0/0.130
registrar server
 !
voice register global
 max-dn 1
 max-pool 1
 !
voice register pool  1
  id network 10.1.130.0 mask 255.255.25
call-manager-fallback
  ip source-address 10.1.130.1
  max-dn 1 dual-line
  max-ephones 1
```

SESTPSTNCall:

At the HQ cluster, the CFUR for the directory number that is applied to
BR2 phone (+442288224001) has been configured:

- Forward Unregistered Internal Destination: **+442288224001**
- Forward Unregistered Internal Calling Search Space: **System_css**
- Forward Unregistered External Destination: **+442288224001**
- Forward Unregistered External Calling Search Space: **System_css**

Which device configuration option will allow an administrator to control bandwidth between calls placed between branches?

A.   Media Resource Group List
B.   Device Pool
C.   Location
D.   AAR Group
E.   Regions

Answer: C

QUESTION 49

A voicemail product that supports only the G.711 codec is installed in headquarters. Which action allows branch Cisco IP phones to function with voicemail while using only the G.729 codec over the WAN link to headquarters?

A.   Configure Cisco Unified Communications Manager regions.
B.   Configure transcoding within Cisco Unified Communications Manager.
C.   Configure transcoding resources in Cisco IOS and assign to the MRGL of Cisco IP phones.
D.   Configure transcoder resources in the branch Cisco IP phones.

Answer: C

QUESTION 50

Scenario:

There are two call control systems in this item. The Cisco UCM is controlling the DX650. the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS controll the the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

DP:

Locations:

CSS:

SRST:

SRST-BR2-Config:

■ **Name:** BR2

■ **Port:** 2000

■ **IP Address:** 10.1.5.15

■ **SIP Network/IP Address:** 10.1.5.15

BR2 Config:

```
BR2 Config                                                    ✕

 voice service voip
   sip
    bind control source-interface
 GigabitEthernet0/0/0.130
    bind media source-interface
 GigabitEthernet0/0/0.130
 registrar server
 !
 voice register global
  max-dn 1
  max-pool 1
 !
 voice register pool  1
   id network 10.1.130.0 mask 255.255.25
 call-manager-fallback
   ip source-address 10.1.130.1
   max-dn 1 dual-line
   max-ephones 1
```

SESTPSTNCall:

```
SRSTPSTNCall                                                   ✕

 At the HQ cluster, the CFUR for the directory number that is applied to
 BR2 phone (+442288224001) has been configured:

 ■  Forward Unregistered Internal Destination: +442288224001

 ■  Forward Unregistered Internal Calling Search Space: System_css

 ■  Forward Unregistered External Destination: +442288224001

 ■  Forward Unregistered External Calling Search Space: System_css
```

After adding SRST functionality the SRST does not work. After reviewing the exhibits, which of the following reasons could be causing this failure?

A.   Device Pool cannot be default.

B.   The Cisco UCM is pointing to the wrong IPv4 address of the BR router.

C.   The router does not support SRST.

D.   The SRST enabled router is not configured correctly.

Answer: A

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