

300-075

Number: 300-075
Passing Score: 900
Time Limit: 120 min
File Version: 1.0

Exam A**QUESTION 1**

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.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 2

Refer to the exhibit.

Which statement is true?

Exhibit:

```
*c xConfiguration Call Loop Detection Mode: "On"  
*c xConfiguration Call Routed Mode: "Always"  
*c xConfiguration Call Services CallsToUnknownIPAddresses: "Direct"  
*c xConfiguration Call Services Fallback Alias:  
*c xConfiguration CollaborationEdge Enabled: Off
```

- A. Endpoints can make calls to unknown IP addresses without the VCS querying any neighbors.
- B. If the VCS receives a call to an unknown IP address, it queries its neighbors for the remote address and if permitted, it routes the call through the neighbor.
- C. Endpoints that are registered directly to the VCS can call only an IP address of a system that is also registered directly to that VCS
- D. Dialing by IP address is not supported on VCS.

Correct Answer: A

Section: (none)

Explanation**Explanation/Reference:****QUESTION 3**

You want to avoid unnecessary interworking in Cisco TelePresence Video Communication Server, such as where a call between two H.323 endpoints is made over SIP, or vice versa. Which setting is recommended?

- A. H.323 - SIP interworking mode. Reject
- B. H.323 - SIP interworking mode. On
- C. H.323 - SIP interworking mode. Registered only
- D. H.323 - SIP interworking mode. Off
- E. H.323 - SIP interworking mode. Variable

Correct Answer: C

Section: (none)

Explanation**Explanation/Reference:****QUESTION 4**

Which three statements about configuring an encrypted trunk between Cisco TelePresence Video Communication Server and Cisco Unified Communications Manager are true? (Choose three.)

- A. The root CA of the VCS server certificate must be loaded in Cisco Unified Communications Manager.
- B. A SIP trunk security profile must be configured with Incoming Transport Type set to TCP+UDP.
- C. The Cisco Unified Communications Manager trunk configuration must have the destination port set to 5061.
- D. A SIP trunk security profile must be configured with Device Security Mode set to TLS.
- E. A SIP trunk security profile must be configured with the X.509 Subject Name from the VCS certificate.
- F. The Cisco Unified Communications Manager zone configured in VCS must have SIP authentication trust mode set to On.
- G. The Cisco Unified Communications Manager zone configured in VCS must have TLS verify mode set to Off.

Correct Answer: ACE

Section: (none)

Explanation**Explanation/Reference:**

QUESTION 5

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.

Correct Answer: BE

Section: (none)

Explanation

Explanation/Reference:

QUESTION 6

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. Create a neighbor zone in VCS with the Unified CM nodes listed as location peer addresses.
- B. Create a single traversal client zone in VCS with the Unified CM nodes listed as location peer addresses.
- C. Create one neighbor zone in VCS for each Unified CM node.
- D. Create a VCS DNS zone and configure one DNS SRV record per Unified CM node
- E. In VCS set Unified Communications mode to Mobile and remote access and configure each Unified CM node.

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 7

Which two options are configuration steps on Cisco Unified Communications Manager that are used when integrating with VCS Expressway servers?
(Choose two.)

- A. allowing numeric dialing from Cisco phones to Expressway
- B. configuring a device pool with video feature enabled
- C. allowing dialing to Expressway domain from Cisco phones
- D. creating an application user on Cisco Unified Communications Manager with assigned privileges
- E. adding the Expressway servers to the Application Servers list

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 8

Which two statements regarding IPv4 Static NAT address 209.165.200.230 has been configured on a VCS Expressway are true? (Choose two.)

- A. The Advanced Networking or Dual Network Interfaces option key has been installed.
- B. VCS rewrites the Layer 3 source address of outbound SIP and H.323 packets to 209.165.200.230
- C. VCS applies 209.165.200.230 to outbound SIP and H.323 payload messages.
- D. With static NAT enabled on the LAN2 interface, VCS applies 209.165.200.230 to outbound H.323 and SIP payload traffic exiting the LAN1 interface.

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 9

Which configuration does Cisco recommend for the peer address on the Expressway-C secure traversal zone when the Expressway-E has one NIC enabled?

- A. Expressway-E internal IP address
- B. Expressway-E external IP address
- C. Expressway-E internal FQDN
- D. Expressway-E external FQDN

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 10

If delegated credentials checking has been enabled and remote workers can register to the VCS Expressway, which statement is true?

- A. H.323 message credential checks are delegated.
- B. SIP registration proxy mode is set to On in the VCS Expressway.
- C. A secure neighbor zone has been configured between the VCS Expressway and the VCS Control.
- D. SIP registration proxy mode is set to Off in the VCS Expressway.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 11

Which two options should be used to create a secure traversal zone between the Expressway-C and Expressway-E? (Choose two.)

- A. Expressway-C and Expressway-E must trust each other's server certificate.
- B. One Cisco Unified Communications traversal zone for H.323 and SIP connections.
- C. A separate pair of traversal zones must be configured if an H.323 connection is required and Interworking is disabled.
- D. Enable username and password authentication verification on Expressway-E.
- E. Create a set of username and password on each of the Expressway-C and Expressway-E to authenticate the neighboring peer.

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 12

Which two statements regarding you configuring a traversal server and traversal client relationship are true? (Choose two.)

- A. VCS supports only the H.460.18/19 protocol for H.323 traversal calls.

- B. VCS supports either the Assent or the H.460.18/19 protocol for H.323 traversal calls.
- C. VCS supports either the Assent or the H.460.18/19 protocol for SIP traversal calls.
- D. If the Assent protocol is configured, a TCP/TLS connection is established from the traversal client to the traversal server for SIP signaling.
- E. A VCS Expressway located in the public network or DMZ acts as the firewall traversal client.

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 13

The administrator at Company X is getting user reports of inconsistent quality on video calls between endpoints registered to Cisco Unified Communications Manager. The administrator runs a wire trace while a video call is taking place and sees that the packets are not set to AF41 for desktop video as they should be.

Where should the administrator look next to confirm that the correct DSCP markings are being set?

- A. on the MGCP router at the edge of both networks
- B. the service parameters in the VCS Control
- C. the QoS service parameter in Cisco Unified Communications Manager
- D. on the actual Cisco phone itself because the DSCP setting is not part of its configuration file downloaded at registration
- E. The setting cannot be changed for video endpoints that are registered to Cisco Unified Communications Manager, but only when they are registered to the VCS Control.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 14

Which three commands are necessary to override the default CoS to DSCP mapping on interface Fastethernet0/1? (Choose three.)

- A. mls qos map cos-dscp 0 10 18 26 34 46 48 56
- B. mls qos map dscp-cos 8 10 to 2
- C. mls qos
- D. interface Fastethernet0/1 mls qos trust cos
- E. interface Fastethernet0/1 mls qos cos 1

F. interface Fastethernet0/2 mls qos cos 1

Correct Answer: ACD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 15

When video endpoints register with Cisco Unified Communications Manager, where are DSCP values configured?

- A. in Unified CM, under Enterprise Parameters Configuration
- B. in Unified CM, under Device > Device Settings > Device Defaults
- C. in Unified CM, under Service Parameters > Cisco CallManager Service > Cluster-wide Parameters
- D. DSCP parameters are always configured on each individual video endpoint.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 16

Which two options are valid service parameter settings that are used to set up proper video QoS behavior across the Cisco Unified Communications Manager infrastructure? (Choose two.)

- A. DSCP for Video Calls when RSVP Fails
- B. Default Intraregion Min Video Call Bit Rate (Includes Audio)
- C. Default Interregion Max Video Call Bit Rate (Includes Audio)
- D. DSCP for Video Signaling
- E. DSCP for Video Signaling when RSVP Fails

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 17

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.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 18

Which option is a benefit of implementing CFUR?

- A. CFUR is designed to initiate TEHO to reduce toll charges.
- B. CFUR can prevent phones from unregistering.
- C. CFUR can reroute calls placed to a temporarily unregistered destination phone.
- D. CFUR eliminates the need for COR on an ISR.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 19

Refer to the exhibit.

Which configuration change is needed to enable NANP international dialing during MGCP fallback?

Exhibit:

```
dial-peer voice 901 pots
destination-pattern 9011T
port 1/0:23
```

- A. Change the dial peer to dial-peer voice 901 voip.
- B. Change the dial peer to dial-peer voice 9011 pots.
- C. Add the command prefix 011 to the dial peer.
- D. Add the command prefix 9011 to the dial peer.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 20

Which command is needed to utilize local dial peers on an MGCP-controlled ISR during an SRST failover?

- A. ccm-manager fallback-mgcp
- B. telephony-service
- C. dialplan-pattern
- D. isdn overlap-receiving
- E. voice-translation-rule

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 21

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

Correct Answer: ABE

Section: (none)

Explanation

Explanation/Reference:

QUESTION 22

Which commands are needed to configure Cisco Unified Communications Manager Express in SRST mode?

- A. telephony-service and srst mode
- B. telephony-service and moh
- C. call-manager-fallback and srst mode
- D. call-manager-fallback and voice-translation

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 23

You need to verify if the Media Gateway Control Protocol gateway is enabled and active. Which command should you use for this purpose?

- A. show running-config
- B. show fallback-mgcp
- C. show gateway
- D. show ccm-manager fallback-mgcp
- E. how running-config gateway
- F. show fallback-mgcp ccm-manager

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 24

You want to perform Media Gateway Control Protocol gateway maintenance. For this purpose, you disable Media Control Gateway Protocol gateway using the no mgcp command. After you perform the maintenance, you want to enable the Media Control Gateway Protocol gateway. Which command should you use?

- A. enable mgcp
- B. mgcp
- C. mgcp enable
- D. mgcp yes
- E. activate mgcp
- F. mgcp active

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 25

Which command displays the detailed configuration of all the Cisco and dial peers of the Cisco Unified SRST router?

- A. show call-manager-fallback all
- B. show dial-peer voice summary
- C. show ephone summary
- D. show voice port summary

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 26

How many active gatekeepers can you define in a local zone?

- A. 1
- B. 2
- C. 5
- D. 10
- E. 15
- F. unlimited

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 27

Which statement about the function of the "+" symbol in the E.164 format is true?

- A. The "+" symbol represents the international country code.
- B. The "+" symbol represents the international call prefix.
- C. The "+" symbol matches the preceding element one or more times.
- D. The "+" symbol matches the preceding element zero or one time.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 28

What is an advantage of TEHO?

- A. TEHO implemented with ISRs eliminates PSTN toll charges.
- B. TEHO implemented with ISRs can reduce PSTN toll charges.
- C. TEHO implemented with AAR reduces toll charges.

D. TEHO implemented with CFUR reroutes calls.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 29

In a node-specific Service Advertisement Framework forwarder deployment model, what is the maximum number of Service Advertisement Framework forwarders that you can assign to a specific node?

- A. 1
- B. 2
- C. 3
- D. 4
- E. 5
- F. 6

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 30

Which statement is true regarding the configuration of SAF Forwarder?

- A. In a multisite dial plan, SAF Forwarders may exist in multiple autonomous systems.
- B. The client label that is configured in Cisco Unified Communications Manager must match the configuration on the SAF Forwarder router.
- C. There should not be multiple nodes of Cisco Unified Communications Manager clusters acting as SAF clients.
- D. The destination IP address must match the loopback address of the SAF router.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 31

Which three devices support the SAF Call Control Discovery protocol? (Choose three.)

- A. Cisco Unified Border Element
- B. Cisco Unity Connection
- C. Cisco IOS Gatekeeper
- D. Cisco Catalyst Switch
- E. Cisco IOS Gateway
- F. Cisco Unified Communications Manager

Correct Answer: AEF

Section: (none)

Explanation

Explanation/Reference:

QUESTION 32

Which component of Cisco Unified Communications Manager is responsible for sending keepalive messages to the Service Advertisement Framework forwarder?

- A. Call Control Discovery requesting service
- B. Hosted DNS service
- C. Service Advertisement Framework client control
- D. Cisco Unified Communications Manager database
- E. Service Advertisement Framework-enabled trunk
- F. gatekeeper

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 33

Which component is needed to set up SAF CCD?

- A. SAF-enabled H.323 intercluster (gatekeeper controlled) trunk
- B. SAF forwarders on Cisco routers
- C. Cisco Unified Communications cluster
- D. SAF-enabled H.225 trunk

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 34

Which statement about the host portion format in Cisco Unified Communications Manager URI dialing is false?

- A. The host portion cannot start or end with a hyphen.
- B. The host portion is not case sensitive.
- C. The host portion accepts characters a-z, A-Z, 0-9, hyphens, and periods.
- D. The host portion can have two periods in a row.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 35

Assume that local route groups are configured. When an IP phone moves from one device mobility group to another, which two configuration components are not changed? (Choose two.)

- A. IP subnet
- B. user settings
- C. SRST reference
- D. region
- E. phone button settings

Correct Answer: BE

Section: (none)

Explanation

Explanation/Reference:

QUESTION 36

Where do you specify the device mobility group and physical location after they have been configured?

- A. phones
- B. DMI
- C. device mobility CSS
- D. device pool
- E. MRGL
- F. locale

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 37

Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)

- A. The phone configuration is not modified.
- B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
- C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified Communications Manager device pool.
- D. If the DMGs are the same, the Device Mobility-related settings are also applied.

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 38

Which statement about setting up FindMe in Cisco TelePresence Video Communication Server is true?

- A. Users are allowed to delete or change the address of their principal devices.
- B. Endpoints should register with an alias that is the same as an existing FindMe ID.
- C. If VCS is using Cisco TMS provisioning, users manage their FindMe accounts via VCS.
- D. A VCS cluster name must be configured.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 39

Which feature allows you to specify which endpoints ring when someone calls a user on a specific destination ID?

- A. FindME
- B. Extension Mobility
- C. Speech Connect
- D. Single Number Reach

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 40

Which action configures phones in site A to use G.711 to site B, but uses G.729 to site C?

- A. Configure Cisco Unified Communications Manager regions.
- B. Configure Cisco Unified Communications Manager locations.
- C. Configure transcoder resources in Cisco Unified Communications Manager.
- D. Configure a gatekeeper.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 41

Which system configuration is used to set audio codecs?

- A. region
- B. location
- C. physical location
- D. licensing

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 42

Which task must you perform before deleting a transcoder?

- A. Delete the dependency records.
- B. Unassign it from a media resource group.
- C. Use the Reset option.
- D. Remove the device pool.
- E. Remove the subunit.
- F. Delete the common device configuration.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 43

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.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 44

Which system configuration is used to set a restriction on audio bandwidth?

- A. region
- B. location
- C. physical location
- D. licensing

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 45

The network administrator of Enterprise X receives reports that at peak hours, some calls between remote offices are not passing through. Investigation shows no connectivity problems. The network administrator wants to estimate the volume of calls being affected by this issue. Which two RTMT counters can give more information on this? (Choose two.)

- A. CallsRingNoAnswer
- B. OutOfResources
- C. LocationOutOfResources
- D. RequestsThrottled
- E. CallsAttempted

Correct Answer: BC

Section: (none)

Explanation

Explanation/Reference:**QUESTION 46**

The network administrator has been investigating bandwidth issues between the central office and remote sites where location-based CAC is implemented. What does the Cisco Unified Communications Manager "LocationOutOfResources" counter indicate?

- A. This counter represents the total number of times that a call on a particular Cisco Unified Communications Manager through the location failed due to lack of bandwidth.
- B. This counter represents the total number of times that a call through locations failed due to the lack of bandwidth.
- C. This counter represents the total number of failed video-stream requests (most likely due to lack of bandwidth) in the location where the person who initiated the video conference resides.
- D. This counter represents the total number of times since the last restart of the Cisco IP Voice Streaming Application that a Cisco Unified Communications Manager connection was lost.

Answer: B

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:**QUESTION 47****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 is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

A new DX650 IP phone with MAC address D0C7.8914.132D, IP address is 172.18.32.119 has been added to the Cisco Unified Communications Manager, but is not registering properly. What is causing this failure?

57-SX20 System information (exhibit):

SX20 System information:

System Information

General

Product: Cisco TelePresence SX20
Last boot: Last Wednesday at 21:43
Serial number: ABCD12345678
Software version: TC7.3.0
Installed options: PremiumResolution
System name: MySystem
IPv4: 192.168.1.128
IPv6: 2001:DB8:1001:2002:3003:4004:5005:F00F
MAC address: 01:23:45:67:89:AB
Temperature: 58.5°C / 137.3°F

H323

Status: Registered
Gatekeeper: 192.168.1.1
Number: 123456
ID: firstname.lastname@company.com

SIP Proxy 1

Status: Registered
Proxy: 192.168.1.2
URI: firstname.lastname@company.com

57-DX650 Configuration (exhibit):

DX650 Configuration:

"Pass Any Exam. Any Time." - www.actualtests.com

Cisco 300-075 Exam

DX650 Configuration

Modify Button Items

1	Line [1] - 3304 in Devices
2	Line [2] - Add a new DN
3	Redial
4	sx20-3@psl226.local
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
9	Add a new SD
10	Add a new SD
11	Add a new SD
12	Add a new SD
13	Add a new SD
14	Add a new SD
15	Add a new SD
----- Unassigned Associated Items -----	
16	Add a new SD

Product Type: Cisco DX650
Device Protocol: SIP

Real-time Device Status

Registration: Unregistered
IPv4 Address: 172.18.32.119
Active Load ID: slpdx650.10-2-3-26
Inactive Load ID: slpdx650.10-1-1-78
Download Status: None

Device Information

☒ Device is Active
☒ Device is trusted

MAC Address* D0C78914131D

Description DX650 Pod 3

Device Pool* Default

Common Device Configuration < None >

Phone Button Template* Cisco DX650 SIP

Softkey Template < None >

Common Phone Profile* Standard Common Phone Profile

Calling Search Space All-Devices

AAR Calling Search Space < None >

57 MRGL e DP (exhibit):

MRGL:

MRGL

Status
1 records found

Media Resource Group List (1 - 1 of 1)

Find Media Resource Group List where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^
<input type="checkbox"/>	MRGL-All

Add New Select All Clear All Delete Selected

DP:

DP

Status
3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Default Global CN Group	Region	Default	Default
<input type="checkbox"/>	Default	Default	Default	Default	Default
<input type="checkbox"/>	Default	Default	Default	Default	Default
<input type="checkbox"/>	GSM	Default	GSM	Default	Default

Add New Select All Clear All Delete Selected

Location e AARG e CSS (exhibit):

Locations

Locations (1 - 3 of 3)

Find Locations where Location begins with

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

AARG:

AARG

Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name begins with

no active query. Please enter your search criteria using the options above.

CSS:

CSS

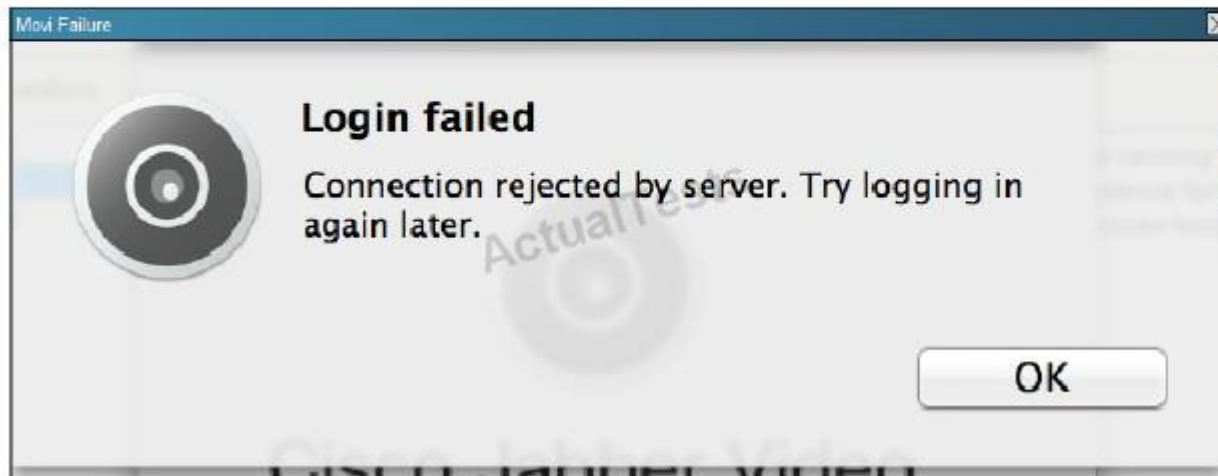
Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

<input type="checkbox"/>		CSS Name ^
<input type="checkbox"/>	A11-Devices	
<input type="checkbox"/>	All-Devices	

57-Movi Failure (exhibit):

Movi Failure:



57-Movi Settings (exhibit):

Mac Settings

Jabber Video

Sign-in Settings

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com

OK Cancel

- A. Device Pool cannot be default.
- B. The DX650 is the incorrect calling search space.
- C. The DX650 Phones does not support SIP.
- D. The location Hub_None has not been activated.
- E. The DX650's MAC address is incorrect in the Cisco UCM.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 48

cenario:

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 is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

What two issues could be causing the Cisco Jabber Video for TelePresence failure shown in the exhibit? (Choose two)

58-SX20 System information (exhibit):

SX20 System information:

System Information

General

Product: Cisco TelePresence SX20
Last boot: Last Wednesday at 21:43
Serial number: ABCD12345678
Software version: TC7.3.0
Installed options: PremiumResolution
System name: MySystem
IPv4: 192.168.1.128
IPv6: 2001:DB8:1001:2002:3003:4004:5005:F00F
MAC address: 01:23:45:67:89:AB
Temperature: 58.5°C / 137.3°F

H323

Status: Registered
Gatekeeper: 192.168.1.1
Number: 123456
ID: firstname.lastname@company.com

SIP Proxy 1

Status: Registered
Proxy: 192.168.1.2
URI: firstname.lastname@company.com

58-DX650 Configuration (exhibit):

Modify Button Items

1

Line [1] - 3304 in Devices

2

Line [2] - Add a new DN

3

Redial

4

☎

sx20-3@osl226.local

5

☎

Add a new SD

6

☎

Add a new SD

7

☎

Add a new SD

8

☎

Add a new SD

9

☎

Add a new SD

10

☎

Add a new SD

11

☎

Add a new SD

12

☎

Add a new SD

13

☎

Add a new SD

14

☎

Add a new SD

15

☎

Add a new SD

----- Unassigned Associated Items -----

16

☎

Add a new SD

Product Type: Cisco DX650

Device Protocol: SIP

Real-time Device Status

Registration: Unregistered

IPv4 Address: 172.18.32.119

Active Load ID: sipdx650.10-2-3-26

Inactive Load ID: sipdx650.10-1-1-78

Download Status: None

Device Information

☒ Device is Active
 ☒ Device is trusted

MAC Address * D0C78914131D

Description DX650 Pod 3

Device Pool * Default

Common Device Configuration < None >

Phone Button Template * Cisco DX650 SIP

Softkey Template < None >

Common Phone Profile * Standard Common Phone Profile

Calling Search Space All-Devices

AAR Calling Search Space < None >

58-MRGL e DP (exhibit):

www.vceplus.com - Download A+ VCE (latest) free Open VCE Exams - VCE to PDF Converter - VCE Exam Simulator - VCE Online - IT Certifications

MRGL:

Status
1 records found

Media Resource Group List (1 - 1 of 1)

Find Media Resource Group List where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^
<input type="checkbox"/>	MRGL-All

Add New Select All Clear All Delete Selected

DP:

Status
3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Device Group	Region	Cost/Ts
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	GSM	Default	GSM	CMLocal

Add New Select All Clear All Delete Selected

58-Location e AARG e CSS (exhibit):

Locations (1 - 3 of 3)

Find Locations where Location begins with

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

AARG:

AARG

Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name begins with

No active query. Please enter your search criteria using the options above.

CSS:

CSS

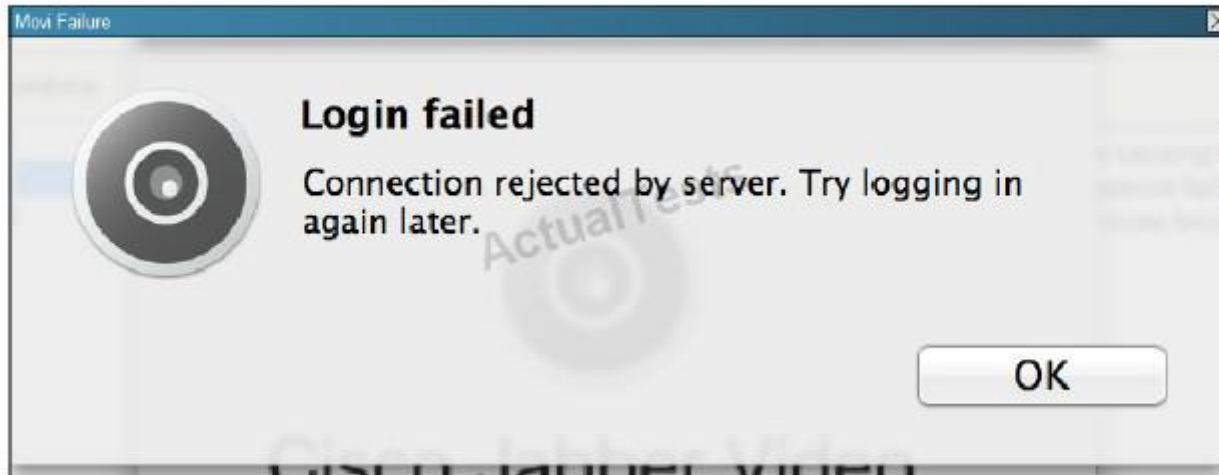
Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

<input type="checkbox"/>		CSS Name ^
<input type="checkbox"/>	All-Devices	
<input type="checkbox"/>	All-Devices	

58-Movi Failure (exhibit):

Movi Failure:



58-Movi Settings (exhibit):

Mac Settings

Jabber Video

Sign-in Settings

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com

OK Cancel

- A. Incorrect username and password.
- B. Wrong SIP domain configured.
- C. User is not associated with the device.
- D. IP or DNS name resolution issue.
- E. CSF Device is not registered.
- F. IP Phone DN not associated with the user.

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 49

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 is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

Which device configuration option will allow an administrator to assign a device to specific rights for making calls to specific DNS?

59-SX20 System information (exhibit):

SX20 System information:

System Information	
General	H323
Product:	Cisco TelePresence SX20
Last boot:	Last Wednesday at 21:43
Serial number:	ABCD12345678
Software version:	TC7.3.0
Installed options:	PremiumResolution
System name:	MySystem
IPvt:	192.168.1.120
IPv6:	2001:008:1001:2002:3003:4004:5005:F00F
MAC address:	01:23:45:67:89:AB
Temperature:	58.5°C / 137.3°F
	SIP Proxy 1
	Status: Registered
	Proxy: 192.168.1.2
	URI: firstname.lastname@company.com

59-DX650 Configuration (exhibit):

Cisco 300-075 Exam

Cisco DX650 Configuration	
<div>Modify Button Items</div>	
1	Line [1] - 3304 in Devices
2	Line [2] - Add a new DN
3	Redial
4	sd20-3@psl226.local
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
9	Add a new SD
10	Add a new SD
11	Add a new SD
12	Add a new SD
13	Add a new SD
14	Add a new SD
15	Add a new SD
----- Unassigned Associated Items -----	
16	Add a new SD

Product Type: Cisco DX650 Device Protocol: SIP	
Real-time Device Status Registration: Unregistered IPv4 Address: 172.18.32.119 Active Load ID: sipdx650.10-2-3-26 Inactive Load ID: sipdx650.10-1-1-78 Download Status: None	
Device Information <input checked="" type="checkbox"/> Device is Active <input checked="" type="checkbox"/> Device is trusted MAC Address * D0C78914131D Description DX650 Pod 3 Device Pool * Default Common Device Configuration < None > Phone Button Template * Cisco DX650 SIP Softkey Template < None > Common Phone Profile * Standard Common Phone Profile Calling Search Space All-Devices AAR Calling Search Space < None >	

59-MRGL e DP (exhibit):

MRGL:

MRGL

Status
1 records found

Media Resource Group List (1 - 1 of 1)

Find Media Resource Group List where Name begins with Find Clear Filter

	Name ^
<input type="checkbox"/>	MRGL-All

Add New Select All Clear All Delete Selected

DP:

DP

Status
3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name begins with Find Clear Filter

	Name ^	Date/Time	CM Group	Region	CM Local
<input type="checkbox"/>	Default		Default	Default	CM Local
<input type="checkbox"/>	Default		Default	Default	CM Local
<input type="checkbox"/>	GSM		Default	CM	CM Local

Add New Select All Clear All Delete Selected

59- Location e AARG e CSS (exhibit):

Locations (1 - 3 of 3)

Find Locations where Location begins with

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

AARG:

Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name begins with

No active query. Please enter your search criteria using the options above.

CSS:

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

<input type="checkbox"/>	CSS Name
<input type="checkbox"/>	All-Devices
<input type="checkbox"/>	All-Devices

59-Movi Failure (exhibit):

Movi Failure:



59-Movi Setings (exhibit):

Mac Settings

Jabber Video

Sign-in Settings

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com

OK Cancel

- A. Media Resource Group List
- B. Device Pool
- C. Location
- D. AAR Group
- E. Calling Search Space

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

QUESTION 50

Scenario:

There are two call control systems in this item.

The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone.

The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

Both of the Cisco TelePresence Video for Windows clients are able to log into the server but can't make any calls. After reviewing the exhibits, which of the following reasons could be causing this failure?

60-DP e Location (exhibit):

DP:

Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name begins with Find Clear Filter

<input type="checkbox"/>	Name	Device Pool Name	Device Pool ID	Device Pool Type	Device Pool Description	Device Pool Status
<input type="checkbox"/>	Default	Default	Default	Default	Default	Default
<input type="checkbox"/>	Default	Default	Default	Default	Default	Default
<input type="checkbox"/>	GSM	Default	Default	GSM	Default	Default

Add New Select All Clear All Delete Selected

Locations:

Locations (1 - 3 of 3)

Find Locations where Location begins with Find Clear Filter

<input type="checkbox"/>	Location
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

Add New Select All Clear All Delete Selected

60-CSS (exhibit):

Cisco 300-075 Exam

CSS

Calling Search Space (1 - 2 of 2)

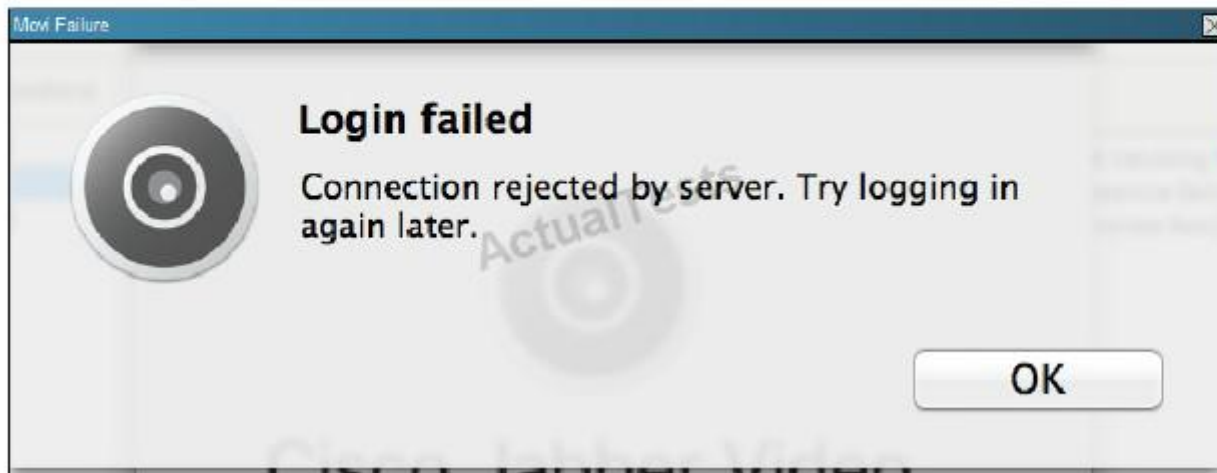
Find Calling Search Space where CSS Name begins with Find Clear Filter

<input type="checkbox"/>	CSS Name
<input type="checkbox"/>	All-Devices
<input type="checkbox"/>	All-Devices

Add New Select All Clear All Delete Selected

60-Movie Failure (exhibit):

Movie Failure:



60-Movie Settings (exhibit):

Move Settings

Jabber Video

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

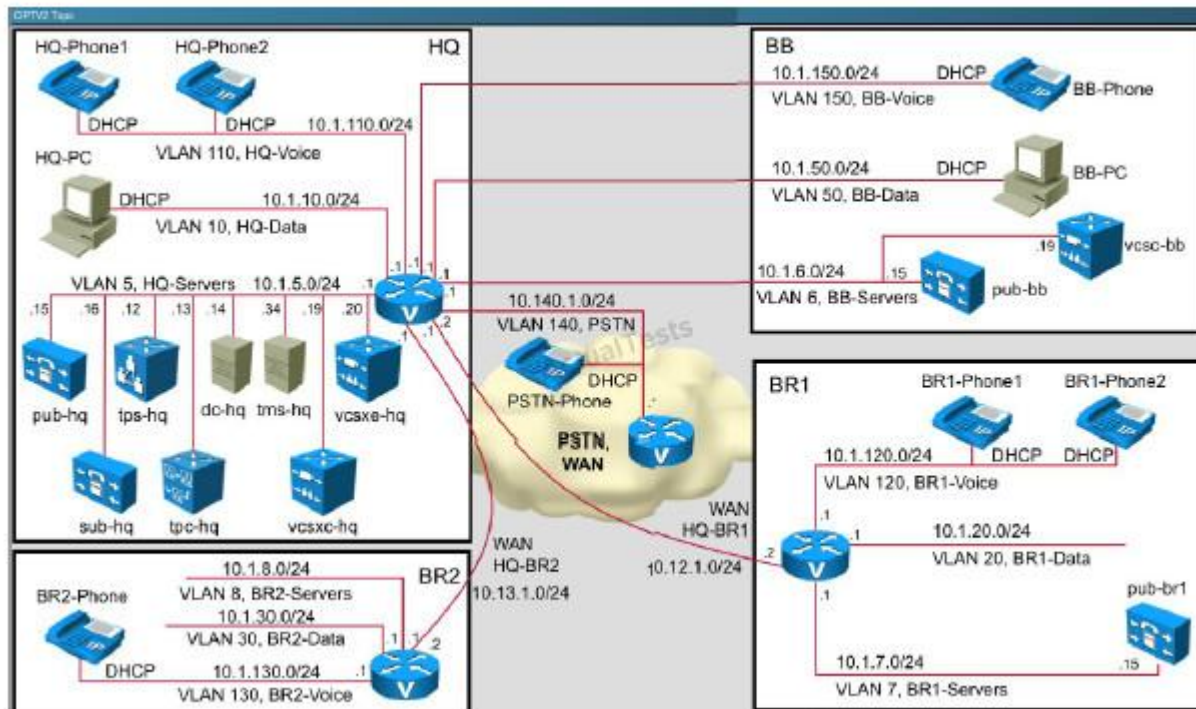
Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com


OK Cancel

60-CIPTV2 Topology (exhibit):



Subzone (exhibit):

Subzone:

- 
- **Name:** HQ
 - **Authentication policy:** Treat as authenticated
 - **Total bandwidth available** - Bandwidth restriction: Limited
 - **Total bandwidth available** - Total bandwidth limit (kbps): 512
 - **Calls into or out of this subzone** - Bandwidth restriction: Limited
 - **Calls into or out of this subzone** - Total bandwidth limit (kbps): 0
 - **Calls entirely within this subzone** Bandwidth restriction: Limited
 - **Calls entirely within this subzone** Total bandwidth limit (kbps): 200

60-Links (exhibit):

Links		You are here: Configuration > Bandwidth > Links					
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
<input type="checkbox"/> DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> VCS_HQ - toHQ	HQ	to HQ	to HQ pipe		1	128 kbps	View/Edit

60-Pipe (exhibit):

Name: to HQ pipe

Total Bandwidth available – Bandwidth restriction: Limited

Total Bandwidth available – Total bandwidth limit (kbps): 256

Calls through this pipe – Bandwidth restriction: Limited

Calls through this pipe – Per call bandwidth limit (kbps): 128

- A. Wrong username and/or password.
- B. Wrong SIP domain name.
- C. The TMSPE is not working.
- D. The bandwidth is incorrectly configured.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 51

Scenario:

There are two call control systems in this item.

The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone.

The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows

What two issues could be causing the Cisco Jabber Video for TelePresence failure shown in the exhibit? (Choose two)

61-DP (exhibit):

DP:

	Name ^	Cisco Unified CM Group	Region	Data/TI
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	GSM	Default	GSM	CMLocal

Buttons: Add New, Select All, Clear All, Delete Selected

61-Location (exhibit):

Locations:

Locations (1 - 3 of 3)

Find Locations where Location begins with Find Clear Filter + -

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

Add New Select All Clear All Delete Selected

61-CSS (exhibit):

CSS:

CSS

Calling Search Space (1 - 2 of 2)

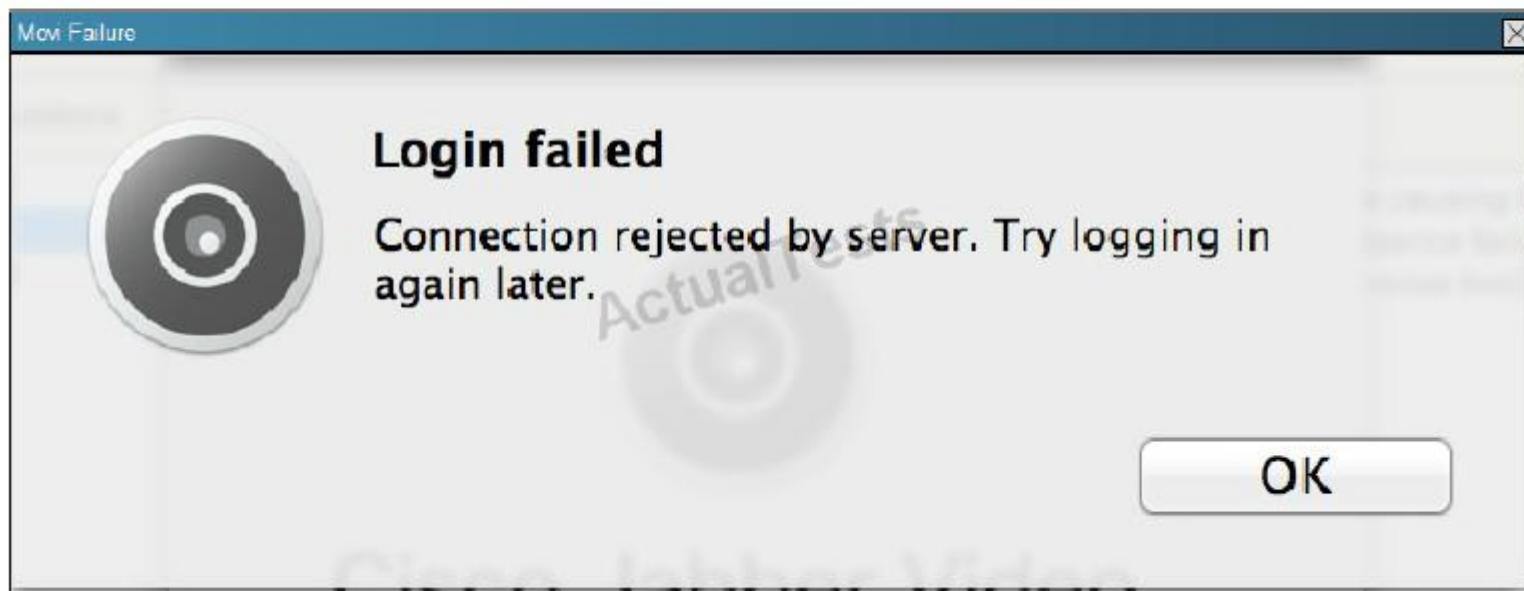
Find Calling Search Space where CSS Name begins with Find Clear Filter + -

	CSS Name ^
<input type="checkbox"/>	All-Devices
<input type="checkbox"/>	All-Devices

Add New Select All Clear All Delete Selected

Movie Failure (exhibit):

Movie Failure:



61-Movi Settings (exhibit):

Move Settings

Jabber Video

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

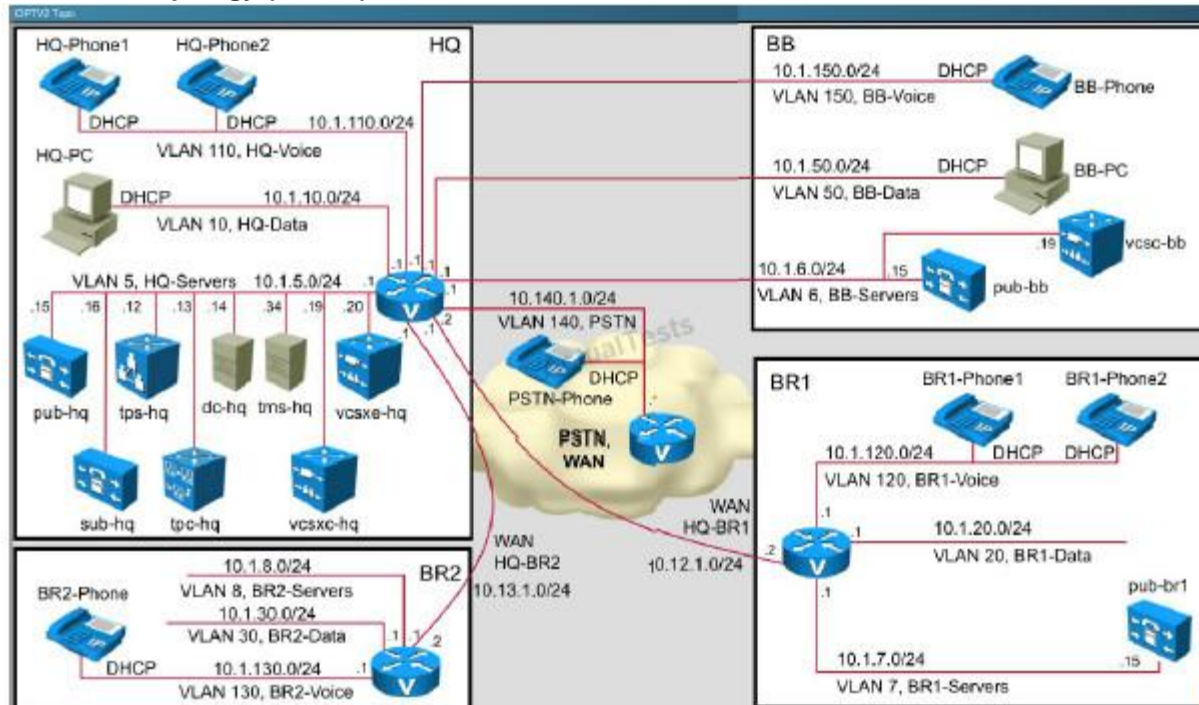
Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com

OK Cancel

61-CIPTV2 Topology (exhibit):



61-Subzone (exhibit):



- **Name:** HQ
- **Authentication policy:** Treat as authenticated
- **Total bandwidth available** - Bandwidth restriction: Limited
- **Total bandwidth available** - Total bandwidth limit (kbps): 512
- **Calls into or out of this subzone** - Bandwidth restriction: Limited
 - **Calls into or out of this subzone** - Total bandwidth limit (kbps): 0
 - **Calls entirely within this subzone** Bandwidth restriction: Limited
 - **Calls entirely within this subzone** Total bandwidth limit (kbps): 200

61-Links (exhibit):

Links

You are here: [Configuration](#) > [Bandwidth](#) > [Links](#)

Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
<input type="checkbox"/> DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> VCS: HQ - toHQ	HQ	to HQ	to HQ pipe		1	128 kbps	View/Edit

61-Pipe (exhibit):

Name: to HQ pipe

Total Bandwidth available – Bandwidth restriction: Limited

Total Bandwidth available – Total bandwidth limit (kbps): 256

Calls through this pipe – Bandwidth restriction: Limited

Calls through this pipe – Per call bandwidth limit (kbps): 128

- A. Incorrect username and password.
- B. Wrong SIP domain configured.
- C. User is not associated with the device.
- D. IP or DNS name resolution issue.
- E. CSF Device is not registered.
- F. IP Phone DN not associated with the user.

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 52

Scenario:

There are two call control systems in this item.

The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone.

The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

A third collaboration call fails between the backbone site and the HQ site. After reviewing the exhibits, which of the following reasons could be causing this failure?

62-DP (exhibit):

Status
3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where: Device Pool Name begins with

<input type="checkbox"/>	Name ▲	Cisco Unified CM Group	Region	Date/Time
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	GSM	Default	GSM	CMLocal

62-Location e CSS (exhibit):

Locations:

Locations (1 - 3 of 3)

Find Locations where Location begins with Find Clear Filter + -

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

Add New Select All Clear All Delete Selected

CSS:

CSS

Calling Search Space (1 - 2 of 2)

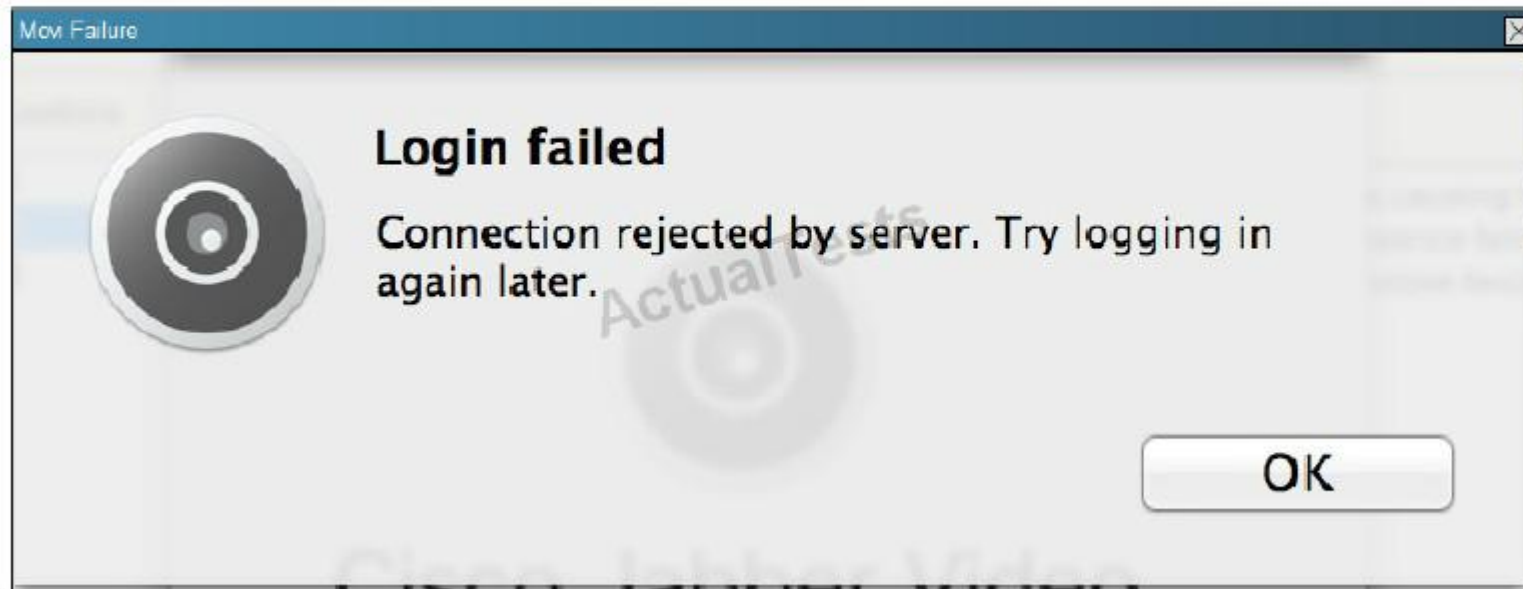
Find Calling Search Space where CSS Name begins with Find Clear Filter + -

	CSS Name ^
<input type="checkbox"/>	
<input type="checkbox"/>	A11-Devices
<input type="checkbox"/>	All-Devices

Add New Select All Clear All Delete Selected

62-Movie Failure (exhibit):

Movie Failure:



62-Movie Settings (exhibit):

Movie Settings:

Movie Settings

Jabber Video

Sign-in Settings

☐ Start Jabber Video when I log on to my computer

☐ Sign in automatically

Servers

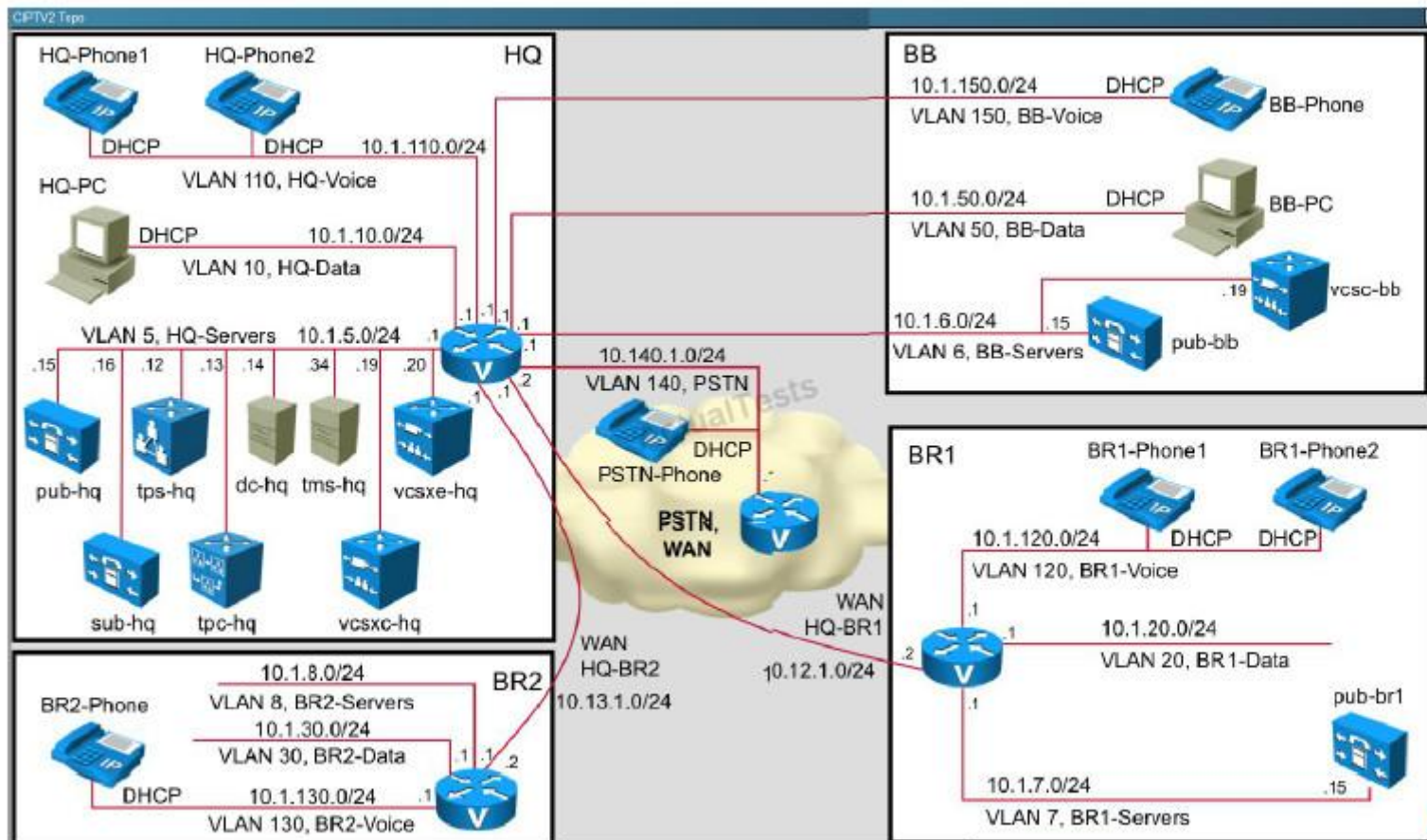
Internal Server
vcs.osl226.local

External Server
vcs.osl226.local

SIP Domain
osl226.com

OK Cancel

62-CIPTV Topology (exhibit):



62-Subzone (exhibit):

Subzone:



- **Name:** HQ
- **Authentication policy:** Treat as authenticated
- **Total bandwidth available - Bandwidth restriction:** Limited
- **Total bandwidth available - Total bandwidth limit (kbps):** 512
- **Calls into or out of this subzone - Bandwidth restriction:** Limited
 - **Calls into or out of this subzone - Total bandwidth limit (kbps):** 0
 - **Calls entirely within this subzone - Bandwidth restriction:** Limited
 - **Calls entirely within this subzone - Total bandwidth limit (kbps):** 200

62-Links (exhibit):

Links:

Links		You are here: Configuration > Bandwidth > Links					
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
<input type="checkbox"/> DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
<input type="checkbox"/> TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
<input type="checkbox"/> YCS HQ - toHQ	HQ	to HQ	to HQ pipe		1	128 kbps	View/Edit

62-Pipe (exhibit):

Name: to HQ pipe

Total Bandwidth available – Bandwidth restriction: Limited

Total Bandwidth available – Total bandwidth limit (kbps): 256

Calls through this pipe – Bandwidth restriction: Limited

Calls through this pipe – Per call bandwidth limit (kbps): 128

- A. Not enough bandwidth has been allocated.
- B. Device Pool.
- C. Location.
- D. The pipe is not functioning.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 53

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 control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

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.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 54

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 control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

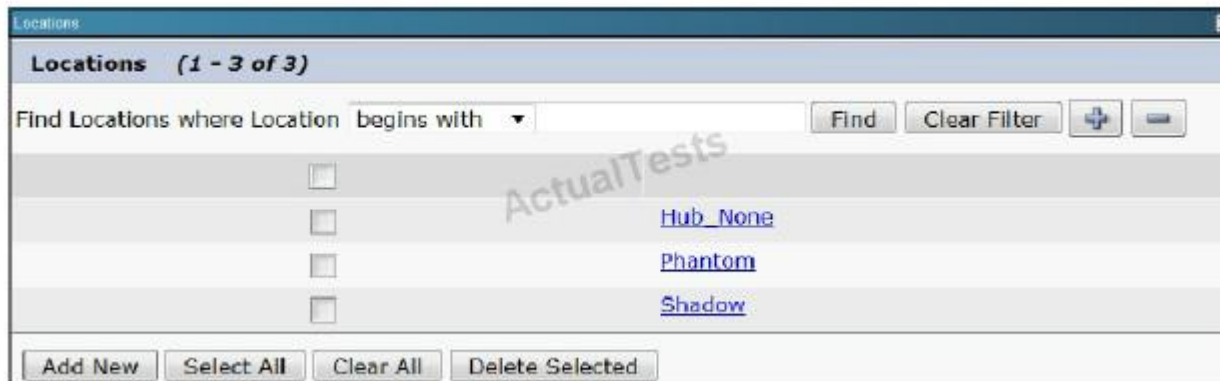
After configuring the CFUR for the directory number that is applied to BR2 phone (+442288224001), the calls fail from the PSTN. Which two of the following configurations if applied to the router, would remedy this situation? (Choose two.)

64-DP (exhibit):

DP:

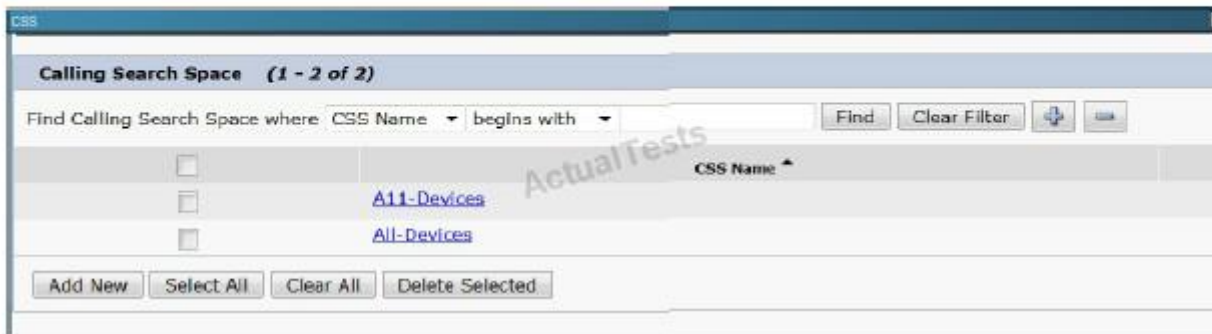
Name	Region	Date/Time
Default	Default	CMLocal
Default	Default	CMLocal
GSM	Default	CMLocal

64-Location (exhibit):

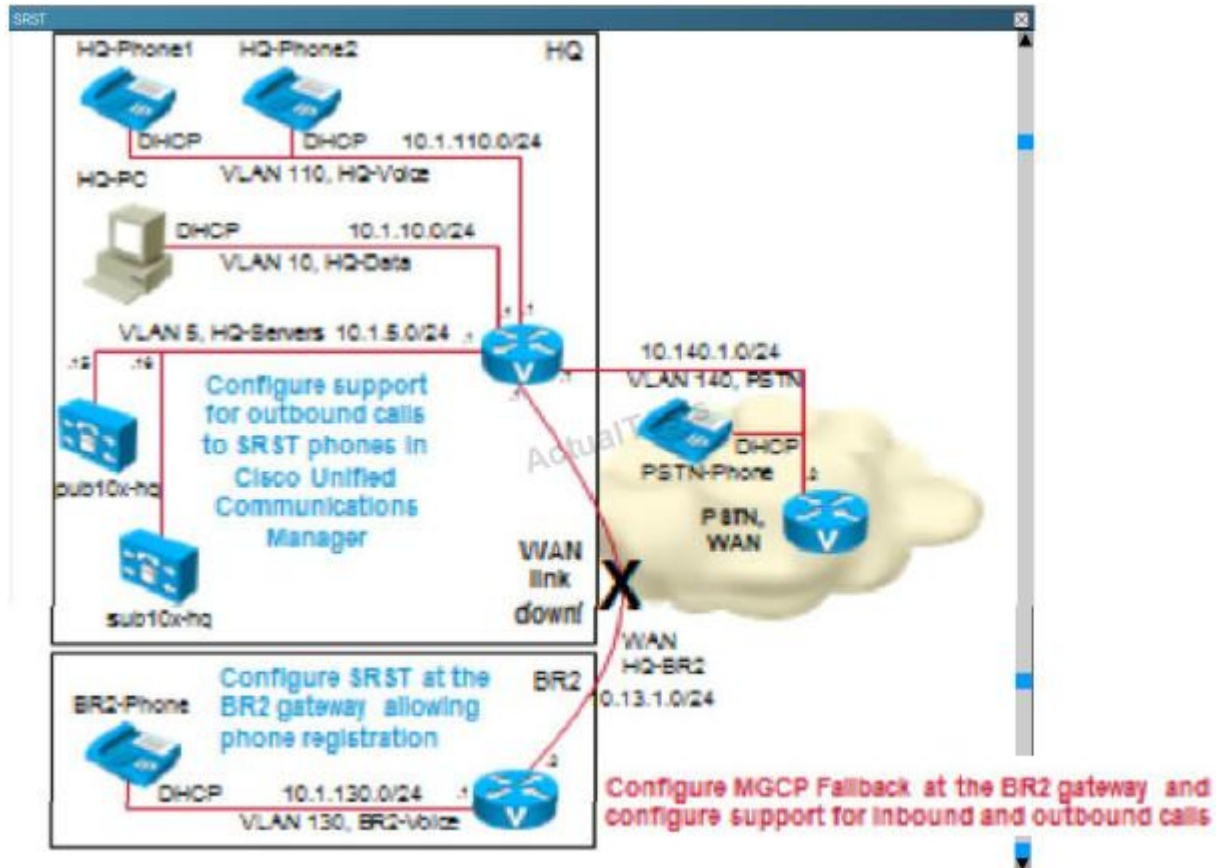


64-CSS (exhibit):

CSS:

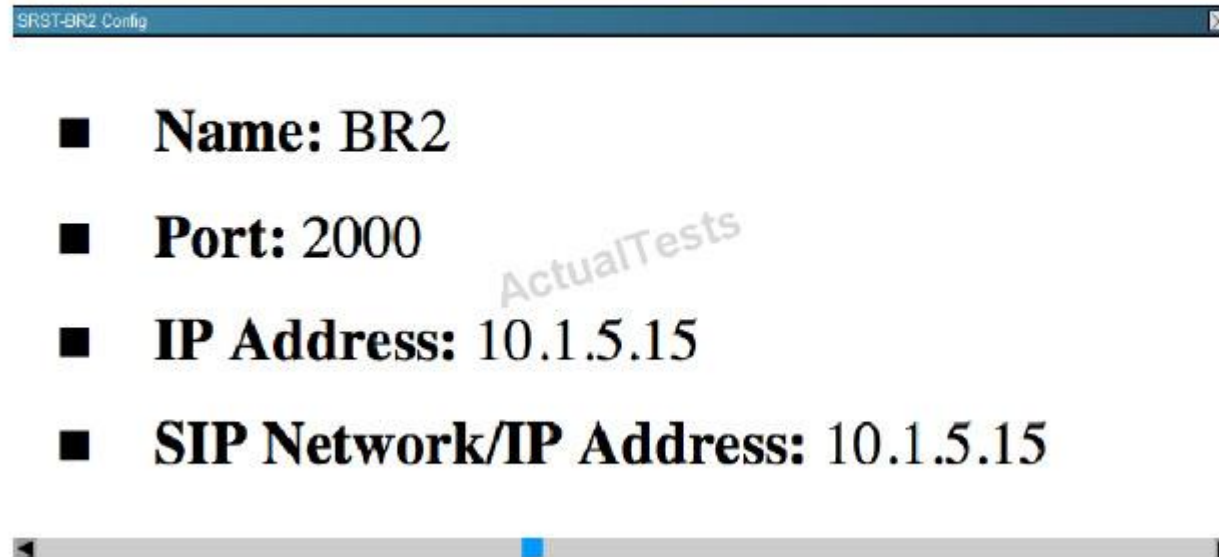


64-SRST (exhibit):



64-SRST-BR2-Config (exhibit):

SRST-BR2-Config:



64-BR2 Config (exhibit):

```
902 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 255.255.255
call-manager-fallback
  ip source-address 10.1.130.1
  max-dn 1 dual-line
  max-ephones 1
```

64-SRSTPSTNCall (exhibit):

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

- A. dial-peer voice 1 pots
 incoming called-number 228822...
 direct-inward-dial
 port 0/0/0:15
- B. dial-peer voice 1 pots
 incoming called-number 228822...
 direct-inward-dial
 port 0/0/0:13
- C. voice translation-rule 1
 rule 1/228821....S//+44&/
 exit
 !
 voice translation-profile pstn-in
 translate called 1
 !
 voice-port 0/0/0:15
 translation-profile incoming pstn-in
- D. voice translation-rule 1
 rule 1/228822....S//+44&/
 exit
 !
 voice translation-profile pstn-in
 translate called 1
 !
 voice-port 0/0/0:15
 translation-profile incoming pstn-in
- E. The router does not need to be configured.

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 55

Scenario:

There are two call control systems in this item.

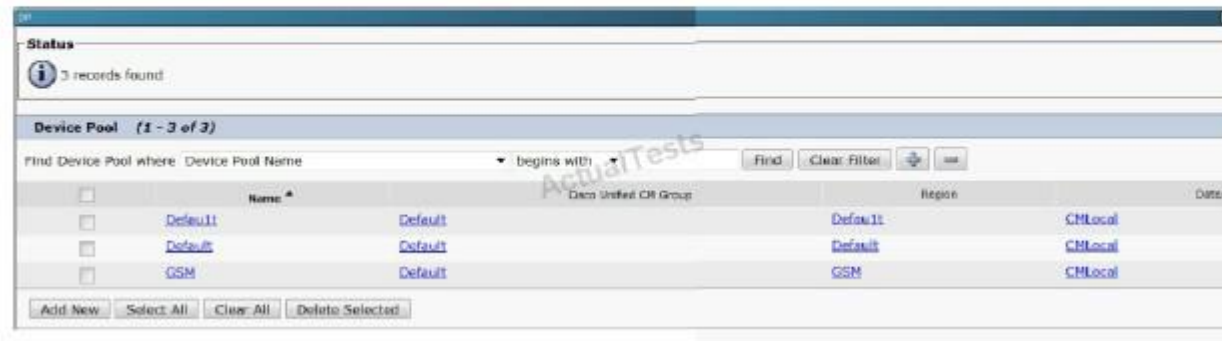
The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone.

The Cisco VCS and TMS control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows

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

65-DP (exhibit):

DP:



Name	Region	CM Local
Default	Default	CMLocal
Default	Default	CMLocal
GSM	GSM	CMLocal

65-Locations (exhibit):

Locations:

Locations (1 - 3 of 3)

Find Locations where Location begins with

<input type="checkbox"/>	Location
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

65-CSS (exhibit):

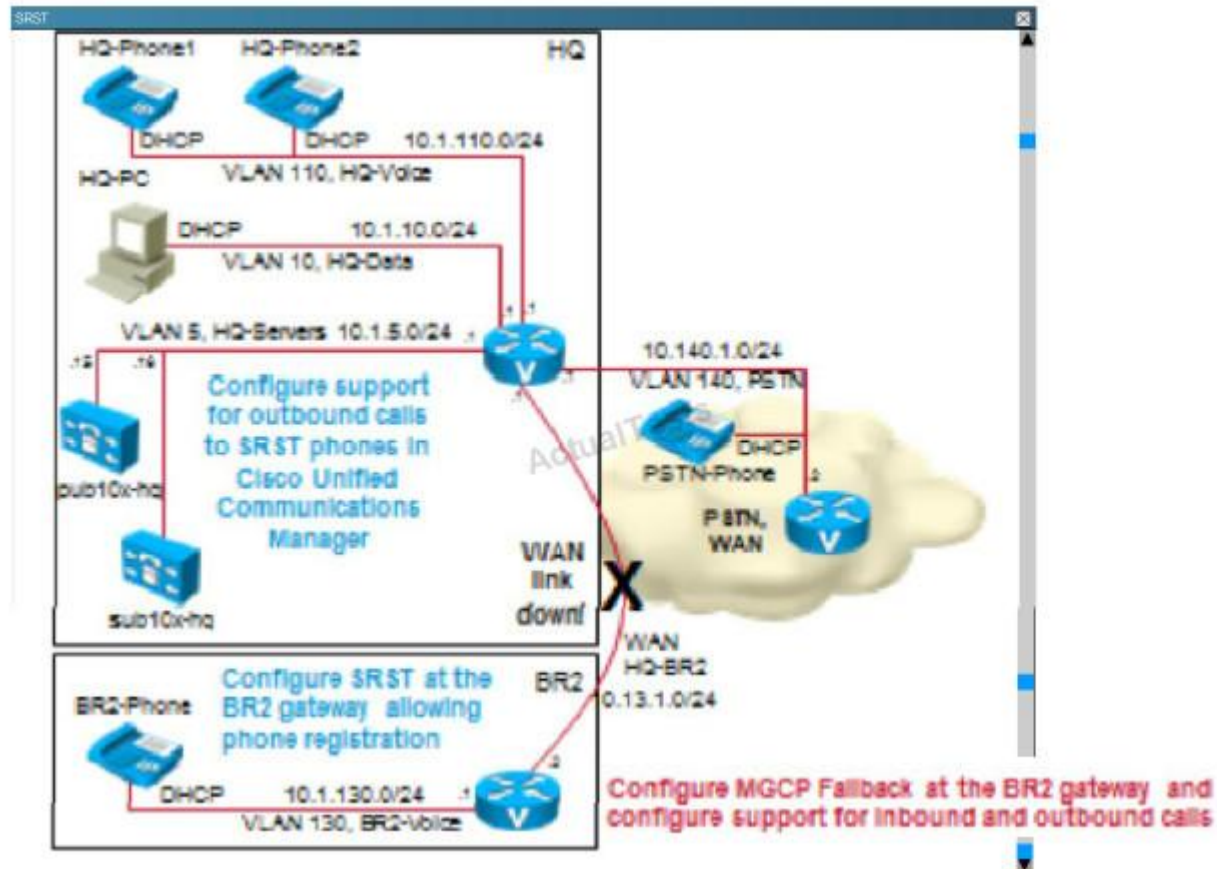
Cisco 300-075 Exam

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

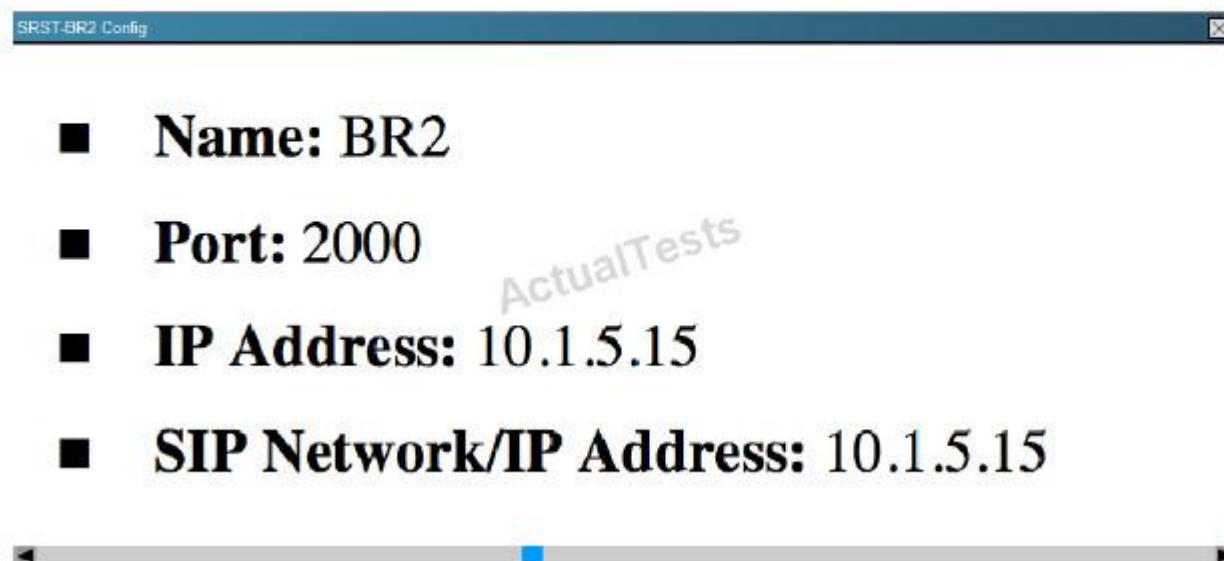
<input type="checkbox"/>	CSS Name
<input type="checkbox"/>	All-Devices
<input type="checkbox"/>	All-Devices

65-SRST (exhibit):



SRST-BR2-Config (exhibit):

SRST-BR2-Config:



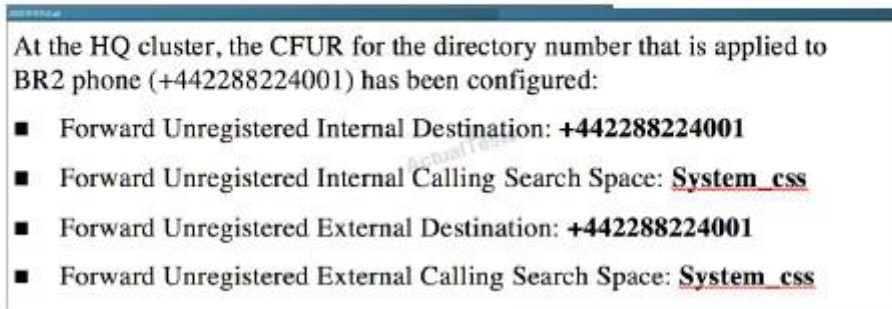
65-BR2 Config (exhibit):

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 255.255.255
call-manager-fallback
  ip source-address 10.1.130.1
  max-dn 1 dual-line
  max-ephones 1
```

65-SRSTPSTNCall (exhibit):

SRSTPSTNCall:



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

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 56

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 7965 and 9971 Video IP Phones.

The Cisco VCS and TMS control the Cisco TelePresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

Which three configuration tasks need to be completed on the router to support the registration of Cisco Jabber clients? (Choose three.)

66-DNS Server (exhibit):

DNS Server:

```
ip dns server
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.15
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.16
ip host pub10x-hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.collab10x.cisco.com 10.1.5.16
ip host pub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.hq.collab10x.cisco.com 10.1.5.16
ip host hq.cisco.com 10.1.5.1
```

66-DP (exhibit):

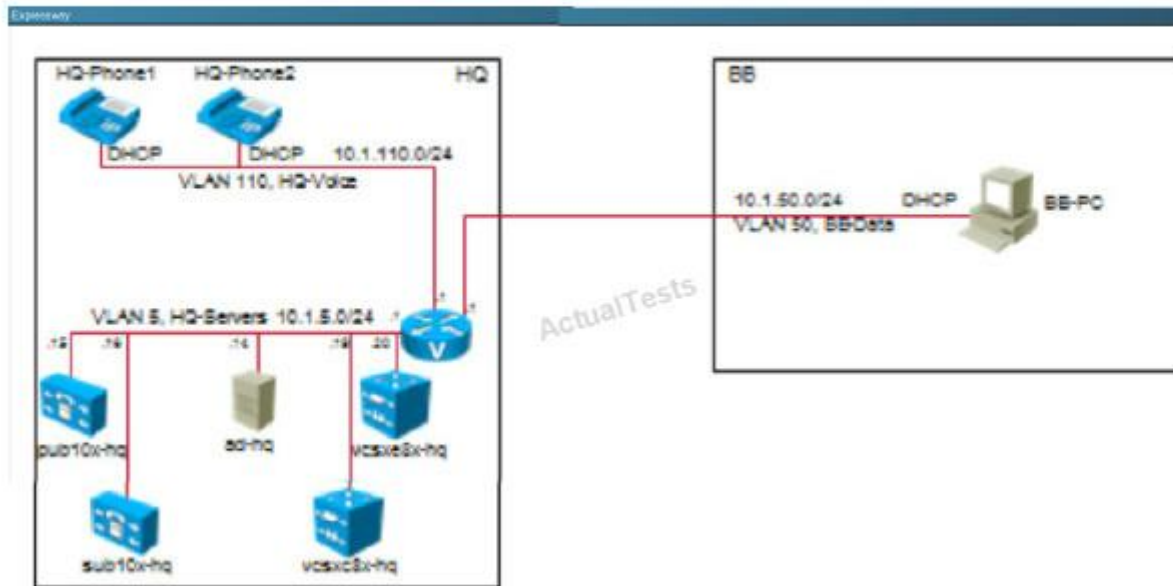
Device Pool:

Device Pool				
Status <i>3 records found</i>				
Device Pool (1 - 3 of 3)				
Find Device Pool where: Device Pool Name: <input type="text"/> begins with <input type="text"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="Add"/> <input type="button" value="Remove"/>				
<input type="checkbox"/>	Name	Device Pool Group	Region	Device Pool
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	GSM	Default	GSM	CMLocal
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/>				

66-Expressaway (exhibit):

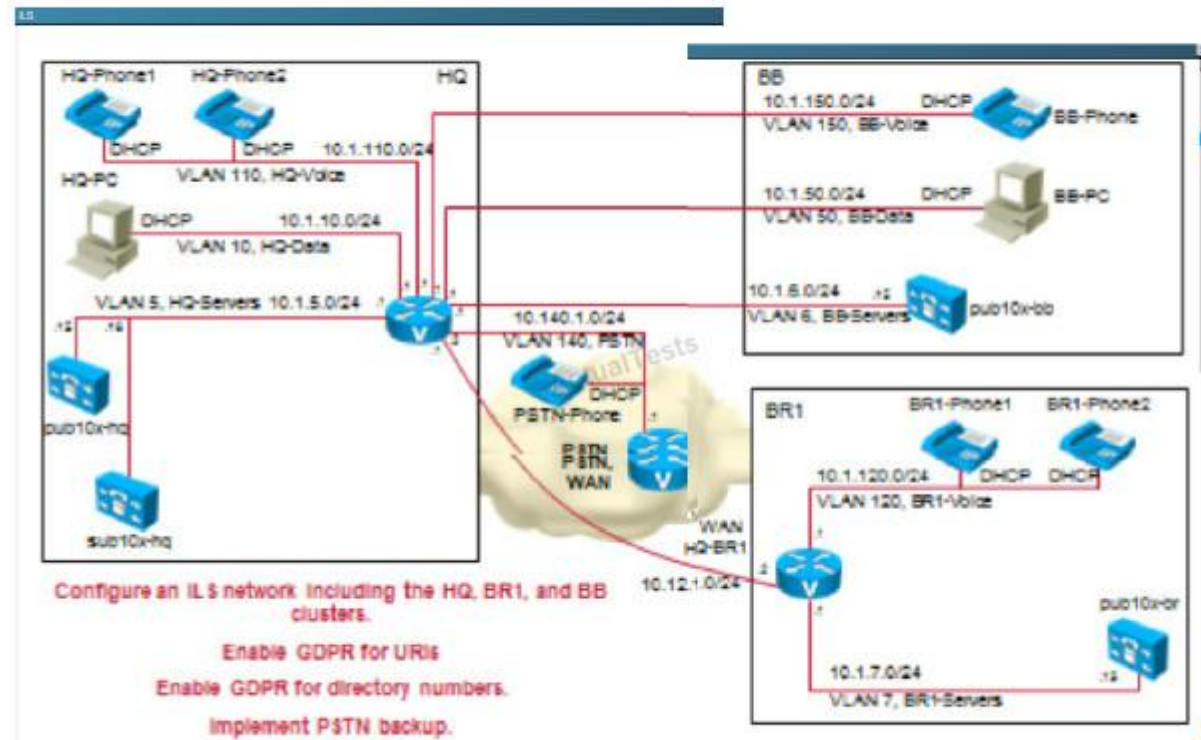
Cisco 300-075 Exam

Expressway:



66-ILS (exhibit):

ILS:



66-Location (exhibit):

Locations:

Locations (1 - 3 of 3)	
<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

66-MRA (exhibit):

Cisco 300-075 Exam

MRA:

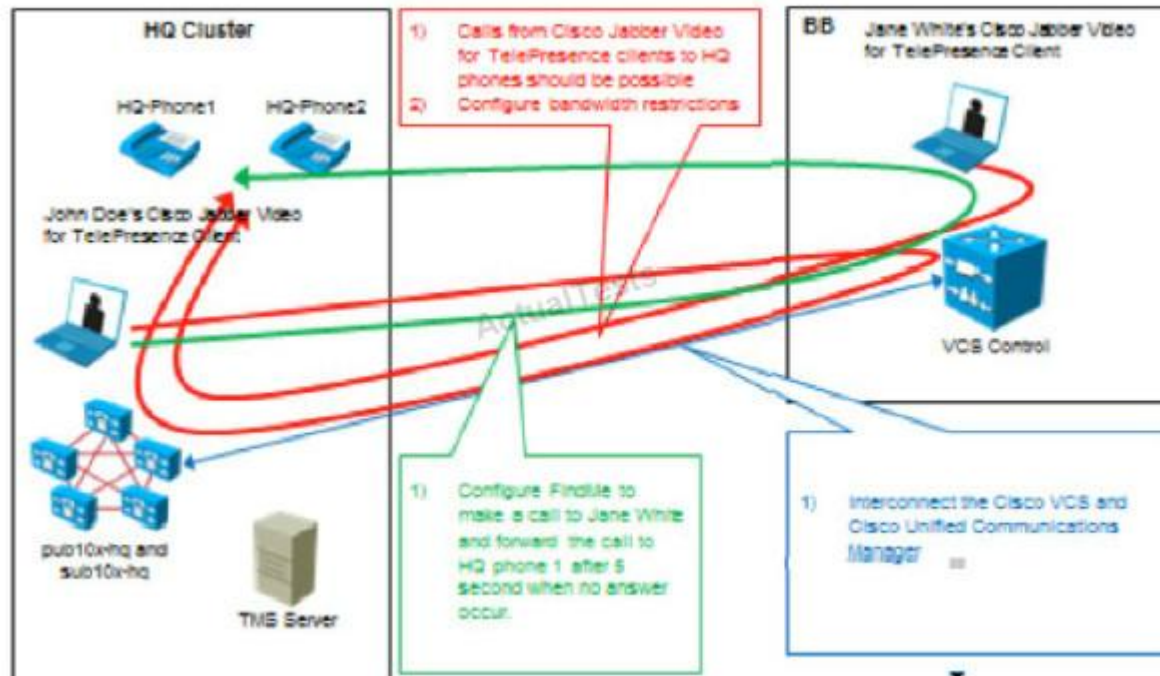
```
ip host _collab-edge.tls.hq.cisco.com SRV 1 1 8443 vcsxe8x-  
hq.hq.collab10x.cisco.com  
ip host vcsxe8x-hq.hq.collab10x.cisco.com 10.1.5.20
```

66-Speedi Dial (exhibit):

Speed Dial:

Phone	Speed Dial Button	Speed Dial Destination
HQ phone 1	1	hq2@cisco.lab
HQ phone 1	2	br1@cisco.lab
HQ phone 1	3	bb@cisco.lab
HQ phone 2	1	hq1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	3	bb@cisco.lab
BR1 phone 1	1	hq1@cisco.lab
BR1 phone 1	2	hq2@cisco.lab
BR1 phone 1	3	bb@cisco.lab
BB phone	1	hq1@cisco.lab
BB phone	2	hq2@cisco.lab

66-Sip Trunk (exhibit):



- A. The DNS server has the wrong IP address.
- B. The internal DNS Service (SRV) records need to be updated on the DNS Server.
- C. Flush the DNS Cache on the client.
- D. The DNS AOR records are wrong.
- E. Add the appropriate DNS SRV for the Internet entries on the DNS Server.

Correct Answer: BCE

Section: (none)

Explanation

Explanation/Reference:

QUESTION 57

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 7965 and 9971 Video IP Phones.

The Cisco VCS and TMS control the Cisco TelePresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

The intercluster URI call routing no longer allows calls between sites. What is the reason why this would happen?

67-DNS Server (exhibit):

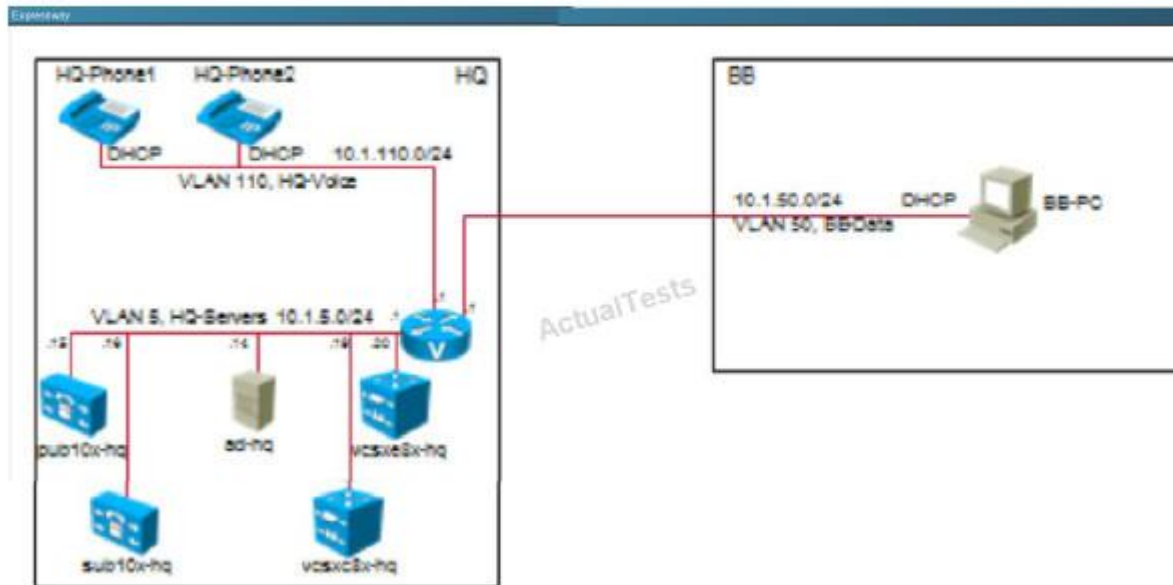
```
ip dns server
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.15
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.16
ip host pub10x-hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.collab10x.cisco.com 10.1.5.16
ip host pub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.hq.collab10x.cisco.com 10.1.5.16
ip host hq.cisco.com 10.1.5.1
```

67-Device Pool (exhibit):

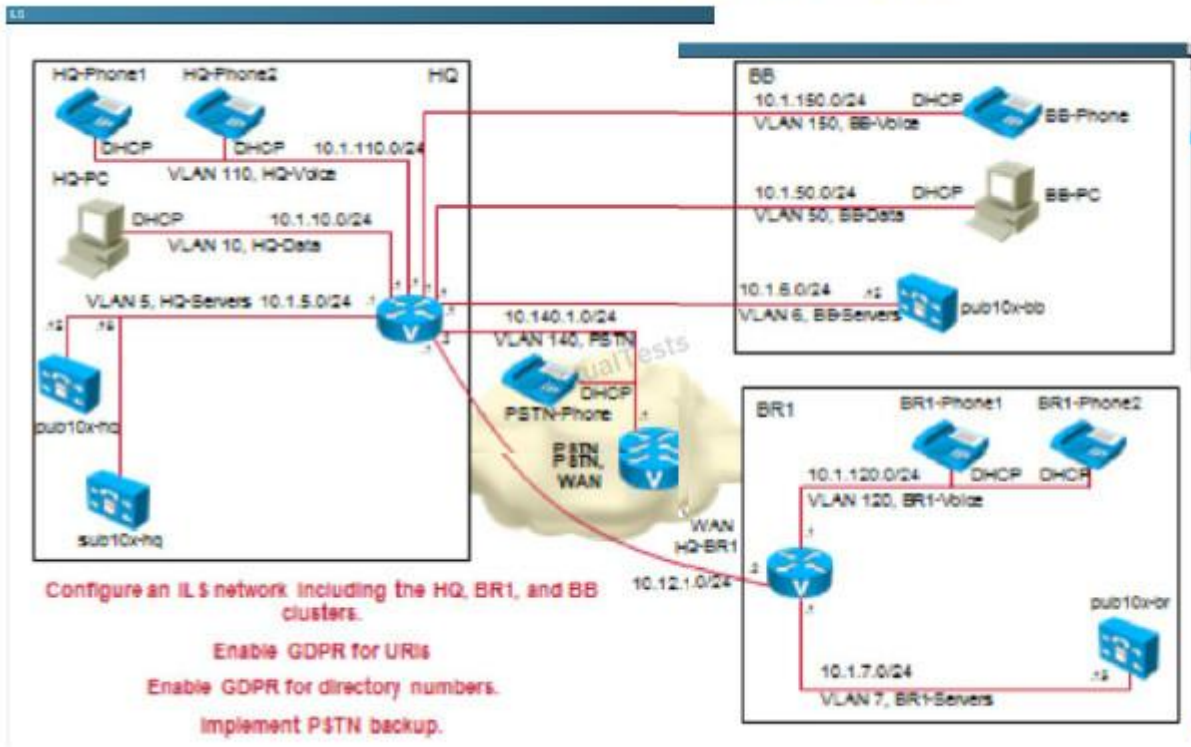
Device Pool					
Status					
3 records found					
Device Pool (1 - 3 of 3)					
Find Device Pool where: Device Pool Name begins with Find Clear Filter					
<input type="checkbox"/>	Name *	Device Pool Group	Region	CM Local	Date/Time
<input type="checkbox"/>	Default	Default	Default	CM Local	
<input type="checkbox"/>	Default	Default	Default	CM Local	
<input type="checkbox"/>	GSM	Default	GSM	CM Local	
Add New Select All Clear All Delete Selected					

67-Expressway (exhibit):

Expressway:

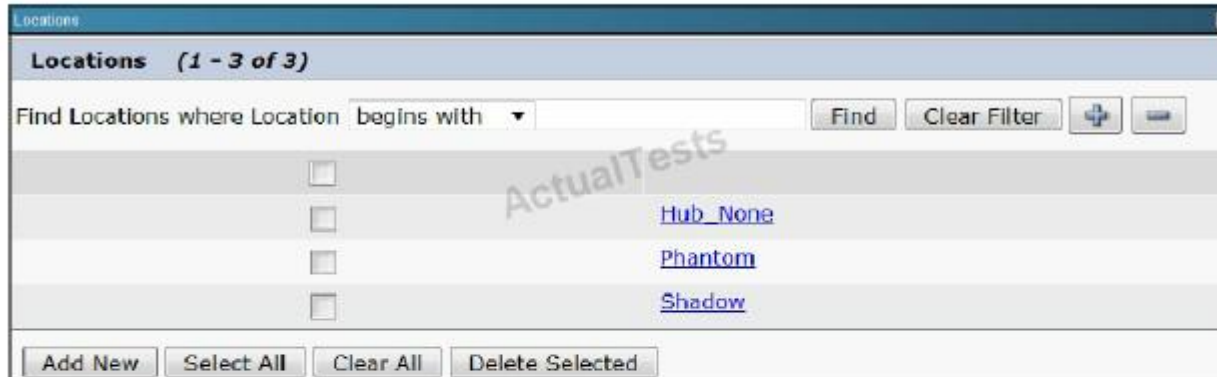


67-ILS (exhibit):



67-Location e MRA (exhibit):

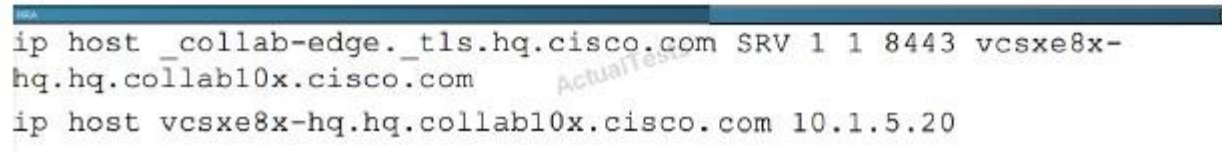
Locations:



Locations (1 - 3 of 3)	
<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

Buttons: Add New, Select All, Clear All, Delete Selected

MRA:



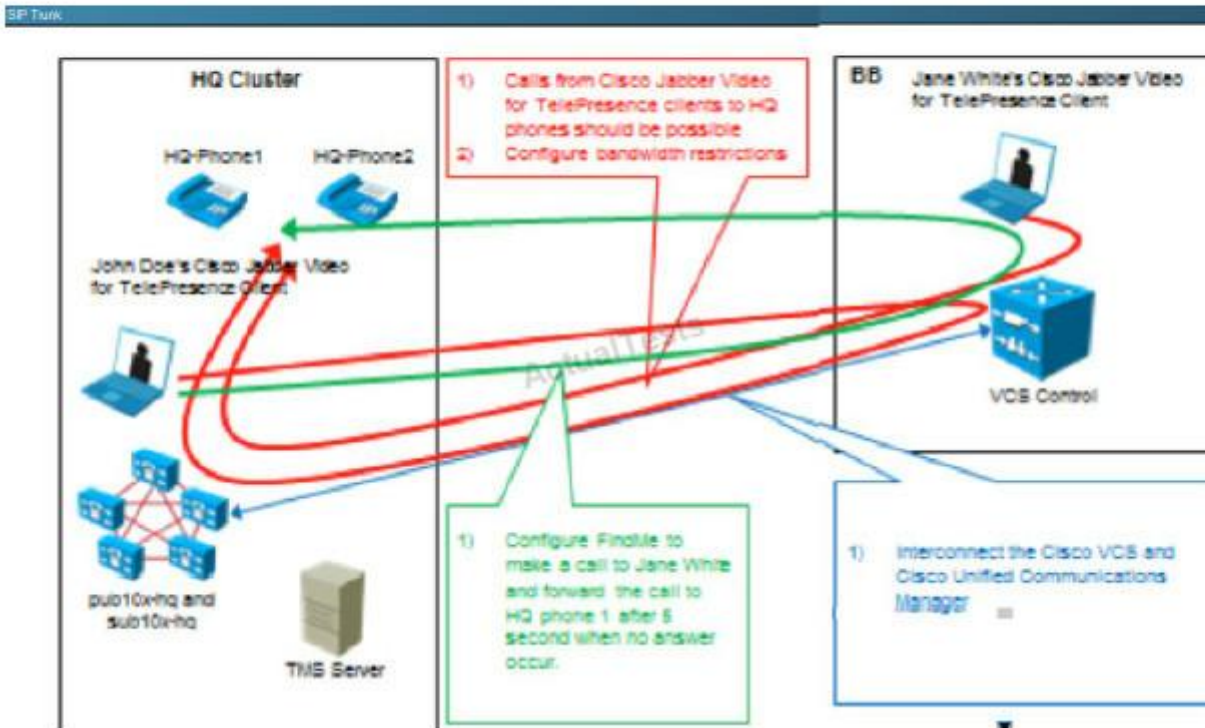
```
ip host _collab-edge.tls.hq.cisco.com SRV 1 1 8443 vcsxe8x-  
hq.hq.collab10x.cisco.com  
ip host vcsxe8x-hq.hq.collab10x.cisco.com 10.1.5.20
```

67-Sppedy dial (exhibit):

Phone	Speed Dial Button	Speed Dial Destination
HQ phone 1	1	hq2@cisco.lab
HQ phone 1	2	br1@cisco.lab
HQ phone 1	3	bb@cisco.lab
HQ phone 2	1	hq1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	3	bb@cisco.lab
BR1 phone 1	1	hq1@cisco.lab
BR1 phone 1	2	hq2@cisco.lab
BR1 phone 1	3	bb@cisco.lab
BB phone	1	hq1@cisco.lab
BB phone	2	hq2@cisco.lab

67-sip trunk (exhibit):

SIP Trunk:



- A. Wrong SIP domain configured.
- B. User is not associated with the device.
- C. IP or DNS name resolution issue.
- D. No SIP route patterns for cisco.lab exist.

Correct Answer: CD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 58

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 7965 and 9971 Video IP Phones.

The Cisco VCS and TMS control the Cisco TelePresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

What two tasks must be completed in order to support calls between the VCS controlled endpoints and the Cisco Unified CM endpoints? (Choose two.)

68-DNS (exhibit):

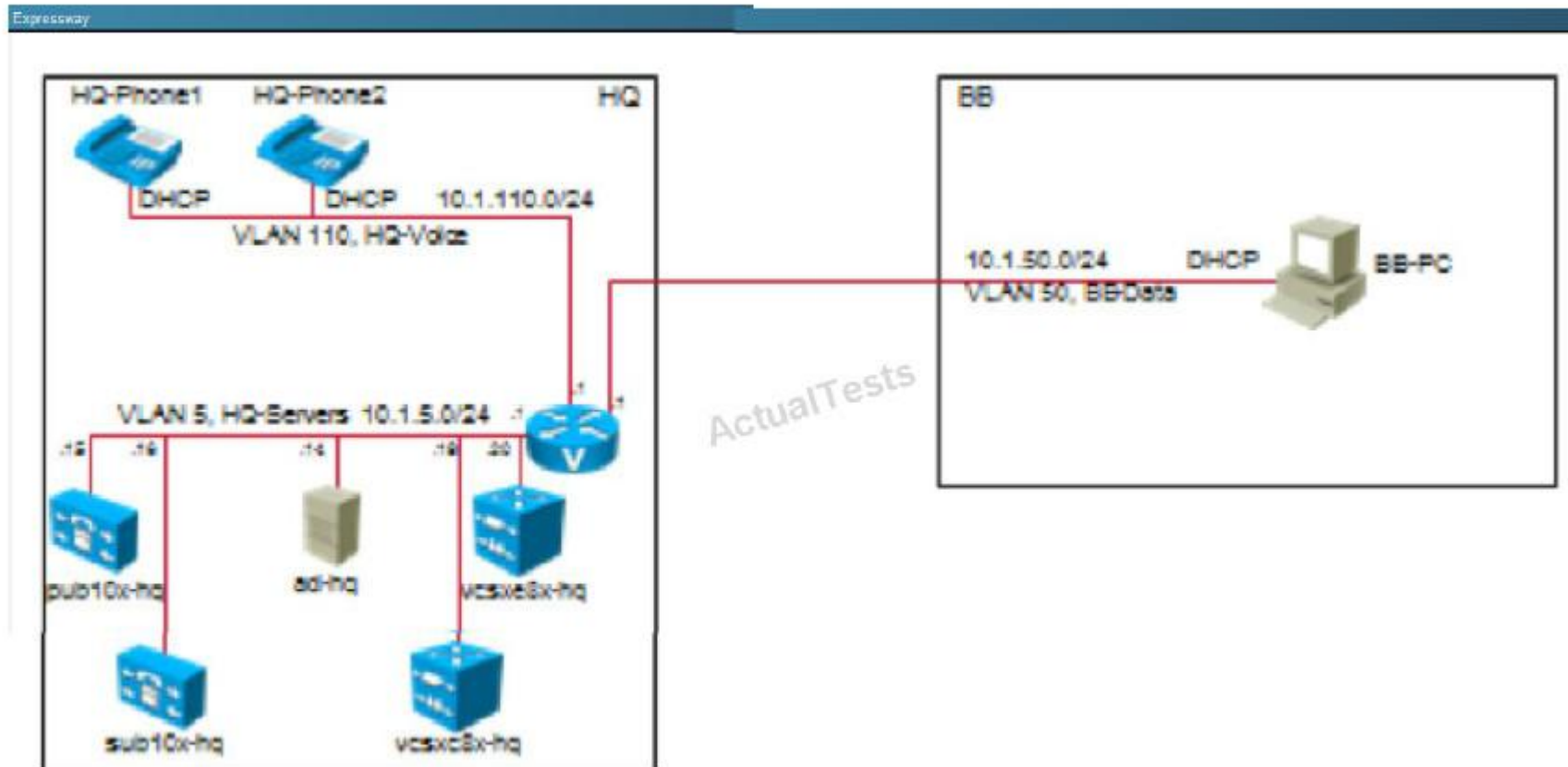
DNS Server:

```
DNS Servers
ip dns server
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.15
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.16
ip host pub10x-hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.collab10x.cisco.com 10.1.5.16
ip host pub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.hq.collab10x.cisco.com 10.1.5.16
ip host hq.cisco.com 10.1.5.1
```

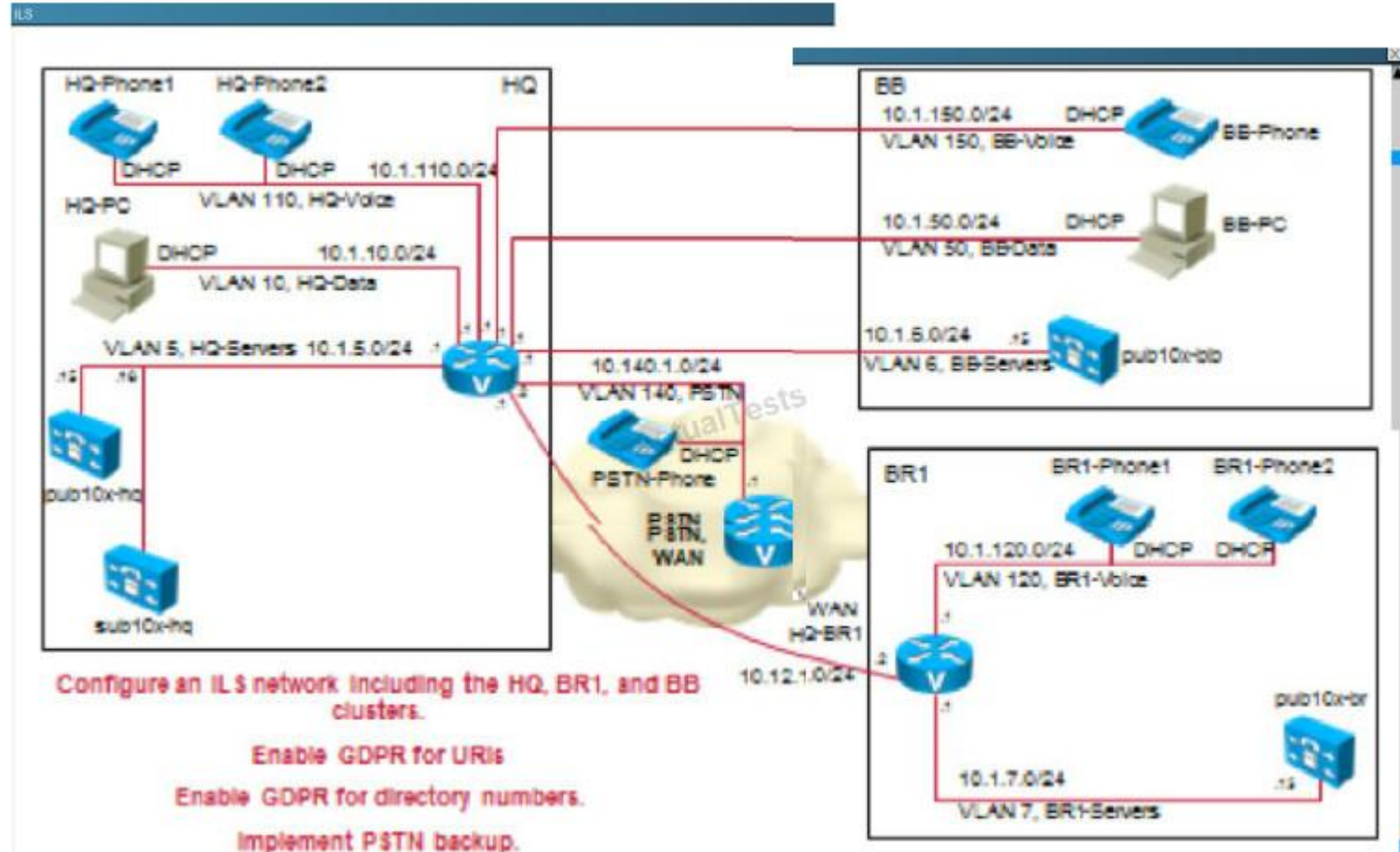
68-DP (exhibit):

Device Pool				
Status				
3 records found				
Device Pool (1 - 3 of 3)				
Find Device Pool where Device Pool Name <input type="text"/> begins with <input type="text"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="-"/>				
<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region	Date/Time
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	Default	Default	Default	CMLocal
<input type="checkbox"/>	GSM	Default	GSM	CMLocal
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/>				

68-Expressway (exhibit):



68-ILS (exhibit):



68-Locations (exhibit):

Locations

Locations (1 - 3 of 3)

Find Locations where Location begins with

<input type="checkbox"/>	
<input type="checkbox"/>	Hub_None
<input type="checkbox"/>	Phantom
<input type="checkbox"/>	Shadow

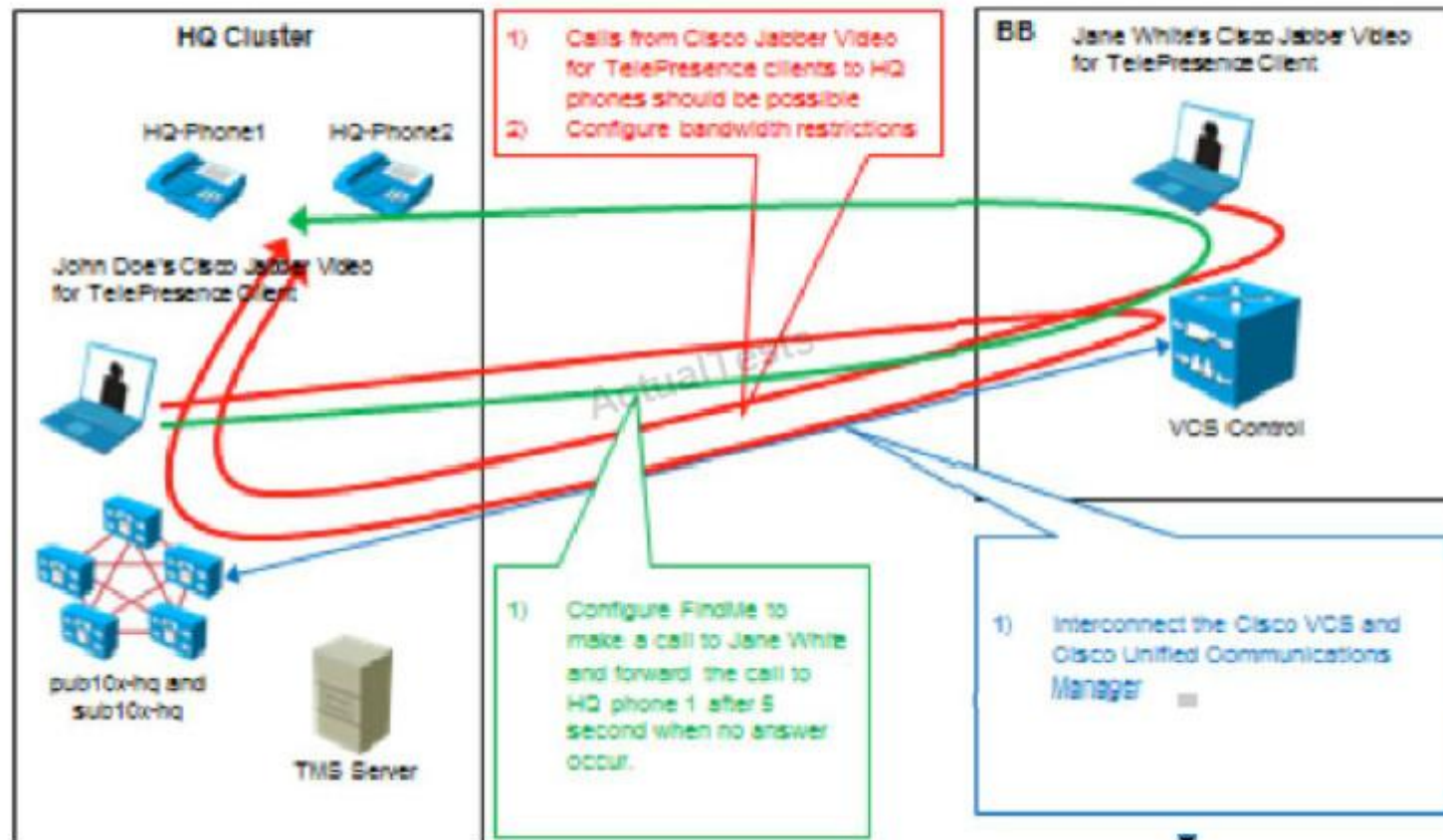
68-MRA (exhibit):

```
ip host _collab-edge._tls.hq.cisco.com SRV 1 1 8443 vcsxe8x-
hq.hq.collab10x.cisco.com
ip host vcsxe8x-hq.hq.collab10x.cisco.com 10.1.5.20
```

68-Speed Dial (exhibit):

Phone	Speed Dial Button	Speed Dial Destination
HQ phone 1	1	hq2@cisco.lab
HQ phone 1	2	br1@cisco.lab
HQ phone 1	3	bb@cisco.lab
HQ phone 2	1	hq1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	2	br1@cisco.lab
HQ phone 2	3	bb@cisco.lab
BR1 phone 1	1	hq1@cisco.lab
BR1 phone 1	2	hq2@cisco.lab
BR1 phone 1	3	bb@cisco.lab
BB phone	1	hq1@cisco.lab
BB phone	2	hq2@cisco.lab

68-Sip Trunk (exhibit):



- A. Media Resource Group List.
- B. Configure a SIP trunk on the Cisco Unified CM to point to the Cisco VCS.
- C. Configure a neighbor zone on the Cisco Unified CM to point to the Cisco VCS.
- D. Configure a SIP trunk on the Cisco VCS to point to the Cisco Unified CM.
- E. Configure a neighbor zone on the Cisco VCS to point to the Cisco Unified CM.

Correct Answer: CD

Section: (none)

Explanation**Explanation/Reference:****QUESTION 59**

A video endpoint is configured with SIP only. What does a video endpoint use to register with the VCS Control?

- A. IP address
- B. SIP URI
- C. MAC address
- D. system name

Correct Answer: B

Section: (none)

Explanation**Explanation/Reference:****QUESTION 60**

Which zone will the VCS Control use to route calls to the VCS Expressway?

- A. neighbor zone
- B. DNS zone
- C. traversal client zone
- D. ENUM zone

Correct Answer: C

Section: (none)

Explanation**Explanation/Reference:****QUESTION 61**

You want to configure Cisco VCS SIP endpoints and H.323 endpoints so that they communicate with one another. To do this, which format must you use in the Search Rule?

- A. name@domain

- B. IP Address (192.168.100.0)
- C. name@hostname
- D. name@IP Address (192.168.100.0)

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 62

When you connect a Cisco VCS Control to Cisco Unified Communications Manager by using a SIP trunk, which mechanism do you use to verify that the trunk has an active connection?

- A. OPTIONS ping
- B. DNS tracing
- C. Continuous ping
- D. Dynamic DNS

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 63

Which DNS SRV Records must be configured on the external DNS server in a mobile remote access scenario with Cisco Expressway?

- A. _collab-edge._tls.example.com
- B. _collab-edge._udp.example.com
- C. _cisco-uds._tcp.example.com
- D. _cuplogin._tcp.example.com

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 64

Company X currently uses a Cisco Unified Communications Manager, which has been configured for IP desk phones and Jabber soft phones. Users report however that whenever they are out of the office, a VPN must be set up before their Jabber client can be used. The administrator for Company X has deployed a Collaboration Expressway server at the edge of the network in an attempt to remove the need for VPN when doing voice. However, devices outside cannot register. Which two additional steps are needed to complete this deployment? (Choose two.)

- A. A SIP trunk has to be set up between the Expressway-C and Cisco UCM.
- B. An additional interface must be enabled on the Cisco UCM and placed in the same subnet at the Expressway.
- C. The customer firewall must be configured with any rule for the IP address of the external Jabber client.
- D. The Expressway server needs a neighbor zone created that points to Cisco UCM.
- E. Jabber cannot connect to Cisco UCM unless it is on the same network or a VPN is set up from outside.

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 65

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 must have all SIP ALG functions disabled.
- C. 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.
- D. A TMS server is needed to allow the firewall traversal to occur between the VCS Expressway and the VCS Control servers.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 66

The VCS Expressway can be configured with security controls to safeguard external calls and endpoints. One such option is the control of trusted endpoints via a whitelist.

Where is this option enabled?

- A. on the voice-enabled firewall at the edge of the network
- B. on the VCS under Configuration > registration > configuration
- C. on the TMS server under Registrations > whitelist
- D. on the VCS under System > configuration > Registrations

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 67

Which statement about configuring the Cisco VCS Control and Cisco VCS Expressway is true?

- A. You do not need to configure search rules for traversal calls.
- B. You need to configure the firewall to allow communication from the Cisco VCS Expressway to the Cisco VCS Control.
- C. The username on the Cisco VCS Control and Cisco VCS Expressway are local and do not need to match.
- D. The Cisco VCS Expressway is the Traversal Server.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 68

Company X wants to implement RSVP-based Call Admission Control and move away from the current location-based configuration. Where does the administrator go to create a default profile?

- A. System > Call Manager > Clusterwide > Service Parameters > RSVP
- B. System > Service Parameters > RSVP
- C. System > Service Parameters > Call Manager > Clusterwide parameters > RSVP

D. on each MGCP gateway at all remote locations

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 69

Where can you change the clusterwide DSCP setting for Cisco Unified Communications Manager?

- A. enterprise parameters
- B. service parameters
- C. enterprise phone configuration
- D. Ethernet configuration

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 70

Refer to the exhibit.

What is the correct value to use for the "DSCP for TelePresence Calls" Cisco CallManager service parameter?

Exhibit:

Cisco 300-075 Exam

```
Router#sh policy-map interface serial0/3/0
Serial0/3/0

Service-policy output: VOICE-VIDEO

  queue stats for all priority classes:

    queue limit 64 packets
    (queue depth/total drops/no-buffer drops) 0/0/0
    (pkts output/bytes output) 0/0

  Class-map: VOICE (match-all)
    0 packets, 0 bytes
    5 minute offered rate 0 bps, drop rate 0 bps
    Match:  dscp ef (46)
    Priority: 10% (153 kbps), burst bytes 3800, b/w exceed drops: 0

  Class-map: VIDEO (match-all)
    0 packets, 0 bytes
    5 minute offered rate 0 bps, drop rate 0 bps
    Match:  dscp af41 (34)
    Queueing
    queue limit 64 packets
    (queue depth/total drops/no-buffer drops) 0/0/0
    (pkts output/bytes output) 0/0
    bandwidth 25% (384 kbps)

  Class-map: TELEPRESENCE (match-all)
    0 packets, 0 bytes
    5 minute offered rate 0 bps, drop rate 0 bps
    Match:  dscp af32 (28)
    Queueing
    queue limit 64 packets
    (queue depth/total drops/no-buffer drops) 0/0/0
    (pkts output/bytes output) 0/0
    bandwidth 25% (384 kbps)
```

- A. 28 (011100)
- B. 34 (100010)
- C. 41 (101001)
- D. 46 (101110)

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 71

Which two statements about remote survivability are true? (Choose two.)

- A. SRST supports more Cisco IP Phones than Cisco Unified Communications Manager Express in SRST mode.
- B. Cisco Unified Communications Manager Express in SRST mode supports more Cisco IP Phones than SRST.
- C. MGCP fallback is required for ISDN call preservation.
- D. MGCP fallback functions with SRST.

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 72

Which two options enable routers to provide basic call handling support for Cisco Unified IP Phones if they lose connection to all Cisco Unified Communications Manager systems? (Choose two.)

- A. SCCP fallback
- B. Cisco Unified Survivable Remote Site Telephony
- C. Cisco Unified Communications Manager Express
- D. MGCP fallback
- E. Cisco Unified Communications Manager Express in SRST mode

Correct Answer: BE

Section: (none)

Explanation**Explanation/Reference:****QUESTION 73**

When considering Cisco Unified Communications Manager failover, how many backup servers can be configured in a Cisco Unified Communications Manager Group?

- A. 1
- B. 5
- C. 2
- D. 4
- E. 3
- F. 6

Correct Answer: C

Section: (none)

Explanation**Explanation/Reference:****QUESTION 74**

Which three CLI commands are used when configuring H.323 call survivability for all calls? (Choose three.)

- A. voice service voip
- B. telephony-service
- C. h323
- D. call preserve
- E. call-router h323-annexg
- F. transfer-system

Correct Answer: ACD

Section: (none)

Explanation**Explanation/Reference:**

QUESTION 75

When configuring Cisco Unified Survivable Remote Site Telephony, which CLI command enables this feature on the router?

- A. call-manager-fallback
- B. ccm-manager redundant-host
- C. ccm-manager sccp local
- D. ccm-manager switchback

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 76

How long is the default keepalive period for SRST in Cisco IOS?

- A. 45 sec
- B. 30 sec
- C. 60 sec
- D. 120 sec

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 77

Which option is a valid test scenario to verify that Cisco Unified Communications Manager failover works and endpoints register with the backup call manager?

- A. During a predetermined maintenance window, stop the Cisco IP Phone Services service on the primary Unified CM. Devices should reregister with the backup Unified CM in the Cisco CallManager Group.
- B. During a predetermined maintenance window, stop the Unified CM service on the Publisher. Devices should reregister with the backup Publisher in the Cisco CallManager Group.
- C. During a predetermined maintenance window, stop the TFTP service on the primary call manager. Devices should reregister with the backup Unified CM in the Cisco CallManager Group.

- D. During a predetermined maintenance window, stop the Unified CM service on the primary call manager. Devices should re-register with the backup Unified CM in the CallManager Group.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 78

Which three commands can be used to verify SRST fallback mode? (Choose three.)

- A. show telephony-service all
- B. show telephony-service ephone-dn
- C. show telephony-service ephone
- D. show telephony-service voice-port
- E. show telephony-service tftp-bindings

Correct Answer: ABC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 79

Company X has a Cisco Unified Communications Manager cluster and a Cisco Unity Connection cluster at its head office and implemented SRST for its branch offices. One Monday at 2:00 pm, the WAN connection to a branch office failed and stayed down for 45 minutes. That day the help desk received several calls from the branch saying their voicemail was not working but they were able to make and receive calls.

Why did the users not realize the WAN was down and prevented access to their voicemail?

- A. All the phones should have started ringing the instant the WAN connection failed to signal the start of SRST mode.
- B. All calls should have dropped when the WAN failed so users would be instantly aware.
- C. Although the phones were still working, the users should have noticed that the phone displays said "SRST Fallback Active".
- D. The voice administrators at the head office did not call the users to notify them.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 80

What are two important considerations when implementing TEHO to reduce long-distance cost? (Choose two.)

- A. on-net calling patterns
- B. E911 calling
- C. number of route patterns
- D. caller ID

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 81

Which two statements about the use of the Intercluster Lookup Service in a multicluster environment are true? (Choose two.)

- A. Cisco Unified Communications Manager uses the ILS to support intercluster URI dialing.
- B. ILS contains an optional directory URI replication feature that allows the clusters in an ILS network to replicate their directory URIs to the other clusters in the ILS network.
- C. Directory URI replication does not need to be enabled individually for each cluster.
- D. To enable URI replication in a cluster, check the Exchange Directory URIs with Remote Clusters check box that appears in the SIP trunk configuration menu.
- E. If the ILS and directory URI replication feature is disabled on a cluster, this cluster still accepts ILS advertisements and directory URIs from other neighbor clusters; it just does not advertise its local directory URIs.

Correct Answer: AB

Section: (none)

Explanation

Explanation/Reference:

QUESTION 82

In Cisco Unified Communications Manager, where do you configure the +E.164 number that is advertised globally via ILS?

- A. ILS configuration under Advanced Features
- B. +E.164 alternate number under Directory Number Settings
- C. Device Information under Phone Configuration
- D. Route Pattern under Route/Hunt

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 83

When implementing a dial plan for multisite deployments, what must be present for SRST to work successfully?

- A. dial peers that address all sites in the multisite cluster
- B. translation patterns that apply to the local PSTN for each gateway
- C. incoming and outgoing COR lists
- D. configuration of the gateway as an MGCP gateway

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 84

Which code snippet is required for SAF to be initialized?

- A. Service Family
- B. External-Client
- C. router eigrp
- D. topology base

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:**QUESTION 85**

When using SAF, how do you prevent multiple nodes in a cluster from showing up in the Show Advance section of the SAF Forwarder configuration?

- A. Configure the publisher node only in the SAF Forwarder configuration page.
- B. Append an @ symbol at the end of the client label value in the SAF Forwarder configuration page.
- C. Configure the correct node in the EIGRP configuration of the gateway router that is associated with the Cisco Unified Communications Manager node.
- D. Configure the SAF Security Profile Configuration to support only a single node.

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:**QUESTION 86**

Which statement about the SAF Client Control is correct?

- A. The SAF Client Control is a configurable inherent component of Cisco Unified Communications Manager.
- B. The SAF Client Control is a non-configurable inherent component of Cisco Unified Communications Manager.
- C. The SAF Client Control is a non-configurable inherent component of the Cisco IOS Routers.
- D. The SAF Client Control is a configurable inherent component of the Cisco IOS Routers

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:**QUESTION 87**

If you want to delete a SAF-enabled trunk from Cisco Unified Communications Manager Administration, what must you do first?

- A. Disassociate the trunk from the CCD advertising service or CCD requesting service.
- B. Delete the trunk from the CCD requesting service node.
- C. Place the Cisco Unified Communications Manager node in standby mode.

D. Redirect CCD advertising and requesting services to another Cisco Unified Communications Manager.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 88

Which functionality does ILS use to link all hub clusters in an ILS network?

- A. Fullmesh
- B. Automesh
- C. ILS updates
- D. multicast

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 89

Which option is known as the location attribute that the global dialplan replication uses to advertise its dial plan information?

- A. location controller
- B. route pattern
- C. route string
- D. URI

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 90

Which three steps are required when configuring extension mobility in Cisco Unified Communications Manager? (Choose three.)

- A. Create the extension mobility IP Phone Service.
- B. Check the Home Cluster checkbox on the End User Configuration page.
- C. Check the Enable Extension Mobility checkbox on the Directory Number Configuration page.
- D. Unsubscribe all other services from the Cisco IP Phone.
- E. Create a user Device Profile.
- F. Subscribe the extension mobility IP Phone Service to the user Device Profile.

Correct Answer: AEF

Section: (none)

Explanation

Explanation/Reference:

QUESTION 91

How many Cisco Unified Mobility destinations can be configured per user?

- A. 1
- B. 10
- C. 4
- D. 6

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 92

When configuring Cisco Unified Mobility, which parameter defines the access control for a call that reaches out to a remote destination?

- A. Calling Party Transformation Calling Search Space under Remote Destination Profile Information
- B. User Local under Remote Destination Profile Information
- C. Rerouting Calling Search Space under Remote Destination Profile Information
- D. Rerouting Calling Search Space under Remote Destination information
- E. Calling Search Space under Phone Configuration

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 93

Which two bandwidth management parameters are available during the configuration of Cisco Unified Communications Manager regions? (Choose two.)

- A. Default Audio Call Rate
- B. Max Audio Bit Rate
- C. Default Video Call Rate
- D. Max Video Call Bit Rate (Includes Audio)
- E. Max Number of Video Sessions

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 94

In Cisco Unified Communications Manager, where do you configure the default bit rate for audio and video devices?

- A. Enterprise Parameters
- B. Region under Region Information
- C. Cisco CallManager service under Service Parameter Configuration
- D. Enterprise Phone Configuration

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 95

Which module is the minimum PVDM3 module needed to support video transcoding?

- A. PVDM3-32
- B. PVDM3-64
- C. PVDM3-128
- D. PVDM3-192

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 96

When implementing a Media Termination Point, what determines the number of sessions that is supported on each DSP?

- A. the codecs that are used in universal transcoding mode
- B. the size of the cluster that is being designed
- C. the number of full-duplex media streams
- D. the Cisco Unified Communications Manager node setting

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference: